Listen

LISTEN, INC. 1995-2020

25

Years of Innovation

SoundCheck®
User Manual

VERSION 18

18

Loudspeakers & Microspeakers

Smart Devices

Headphones & Headsets

Smartphones & Tablets

Automotive

Microphones

Wireless Devices

Telephones

Hearing Aids

Audio Electronics

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PN: 8010 REV082820 SOUNDCHECK® VERSION 18 USER MANUAL

REV 082820

Contents

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Installing SoundCheck®

System Requirements

SoundCheck 18.1 is a 64-bit application and only 64-bit operating systems are supported.

SoundCheck uses your computer's CPU to perform all calculations and signal processing. Because of this, the speed of your computer directly affects the overall performance of the SoundCheck system. In addition, all stimulus waveforms and measured waveforms are played and/or stored in memory for optimum performance. This means that long test signals and measurements require more memory and longer processing time than short test signals and measurements. Therefore, it is recommended that 8 GB of RAM be used for measurements of 4 to 10 seconds and 16 GB of RAM (or more) for measurements longer than 10 seconds. More system memory is also required for the following: Low frequency measurements, using more than 2 channels per measurement and high bandwidth measurements (more than 48 kHz sample rate).

Minimum Computer Requirements

Before buying a series of new computers for use with SoundCheck, we recommend that you test one with all the related hardware, including the audio interface. Test the audio interface with the SoundCheck Self Test sequence to insure that it is compatible with the computer. We recommend that you purchase a high quality computer according to the guidelines below.

- Supported operating systems:
  - Windows® 10 - 64-bit. SoundCheck may work in Windows® 7 - 64-bit but it is no longer validated. No other versions of Windows are supported.
  - macOS® Catalina 10.15. Versions of macOS prior to 10.13.6 are not supported.
- Both Intel and AMD processor motherboards require changes to BIOS for proper operation with Windows.

  See ***Hardware Compatibility List on page 557***.
- 8 GB of RAM minimum (16 GB or more recommended for large WAV files or high resolution measurements below 50 Hz).
- 2 GB of free hard-disk space required for complete software installation
- Do not connect audio interfaces through USB hubs. Connect directly to the USB port on the computer
- Listen only supports Thunderbolt audio interfaces on Windows® 10. Windows® 10 has superior support for Thunderbolt devices, hot-plugging is possible and no 3rd party software is required.
- Dante support is only available for Windows® 10.
- Refer to ***Hardware Compatibility List on page 557***, for details regarding audio interfaces and operating systems. The audio interfaces listed in the Hardware Compatibility List have been tested with SoundCheck on computer systems supplied by Listen. Note that some computers may not be compatible with all audio interfaces.

SoundCheck Installation on macOS

- The Mac version requires **optional module 2100**
- SoundCheck 18 validated with macOS Catalina, 10.15. Versions prior to 10.13.6 are not supported
- SoundCheck 18 default Hardware Editor settings determined with macOS Catalina 10.15

  Using a different macOS may require different Hardware Editor Latency Values than those included in the default hardware files (HAR). Follow the instructions in the audio interface setup documentation included with the approved drivers on our website to determine the proper latency value for the Hardware Editor.

  See **https://support.listeninc.com/hc/en-us/sections/200370694-Drivers**.

Computer Setup

Important! SoundCheck requires that you have Administrative Rights Enabled in Windows, for any account that it is installed on.

Important! ***Refer to*** **Hardware Compatibility List on page 557** ***for information on approved audio interfaces.***

After installing SoundCheck please follow the instructions in ***Windows Setup Recommendations on page 12***.

Backup

It is highly recommended that you make a backup of your SoundCheck critical folders on a regular basis. A backup should always be made prior to installing a new version or update to SoundCheck. We recommend that the following folders be included in any backup:

- Sequence files (.SQC)
- Data (If the default Data folder in SoundCheck is the location for your data files)
- Results (If the default Results folder in SoundCheck is the location for your result files)
- WAV files
- **SoundCheck 18.ini** (The preferences that were last used in SoundCheck.) See ***SoundCheck 18.ini File on page 53***.

Upgrading From an Earlier Version

If you are upgrading from an earlier version of SoundCheck 18 (or Beta version) you should copy the old installation folder and name it "SoundCheck 18 OLD" before installing the new version. This is a precautionary measure to keep you from overwriting sequence steps that you have customized.

Installing SoundCheck 18 overwrites the contents of the SoundCheck 18 folder.

If upgrading from SoundCheck 17.x (or earlier), the folder name does not need to be changed. This way you can run both versions of SoundCheck without disturbing any tests you have already created. If you want to run your SoundCheck 17.x Sequences in the new version, you can use the Setup Wizard to convert those sequences for SoundCheck 18. See ***Convert Sequences From Previous Version on page 36***.

Rules - Installing SoundCheck

- Do not copy Steps and Sequences folders from previous version and paste them into SoundCheck 18. See ***Convert Sequences From Previous Version on page 36***.
- SoundCheck 18 sequences are not backward compatible. They will not work in previous versions of SoundCheck. Sequences from SoundCheck 4.x, 5.x and 6.x will run in SoundCheck 18 but will require updating to conform to the new Multichannel parameters.
- DAT files created with SoundCheck 18 are not viewable in versions of SoundCheck prior to and including SoundCheck 6.0x. The updated DAT file format in SoundCheck 18 is not compatible with versions of SoundCheck prior to and including SoundCheck 6.0x. The DAT file format was updated in SoundCheck 6.1.
- Status.dat files for SoundCheck 8.x and later will not work with previous versions of SoundCheck

Software Requirement

LabVIEW 2018 (LVRT) is required as part of the installation process.

NI Visa 19.5 is required for Listen Hardware such as: AmpConnect ISC, AudioConnect and SoundConnect 2. It is automatically installed with the SoundCheck installer for Windows and must be manually installed for macOS.

NI Visa is also required if you intend to use a GPIB controller, an external footswitch or external buzzer with SoundCheck (optional equipment).

DAQmx 19.6 is an optional component that must be downloaded and installed separately. It is required in order to use NI DAQmx hardware.

SoundCheck Software Installation

1. Close all running applications before installation.

   We recommend that you temporarily disable antivirus software during the SoundCheck installation process. Once complete, re-enable your antivirus software.

   Make sure your IT department has allowed permission for the installation of SoundCheck.

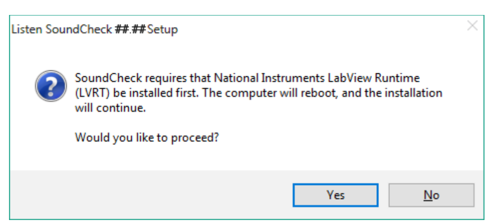
2. **SoundCheck 18 is only available as a 64-bit application**. It requires a 64-bit version of Windows. It cannot be installed on a 32 bit version of Windows.

3. The SoundCheck installer is typically supplied as a downloadable file. You can also install from disc by placing a SoundCheck installation disc in the DVD-ROM drive of your computer.

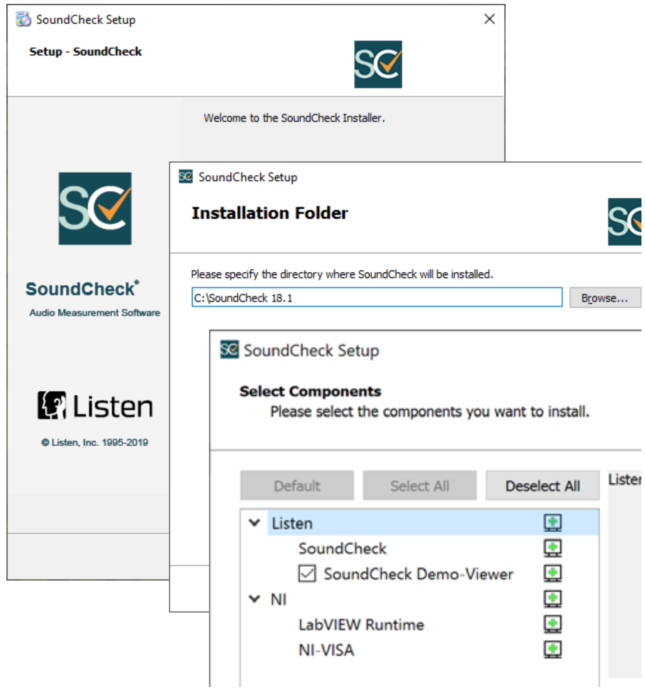
   - If your computer is set to “Auto Run”, you will be prompted to install the software
   - You may also be prompted to run “**Start.exe**”
   - Click “**Install SoundCheck**”

4. For first time installations, LabVIEW Run Time (*LabVIEW 2018*) is required as part of the installation process. **This will require a computer reboot before SoundCheck is installed.**

   The SoundCheck installation will resume after reboot.



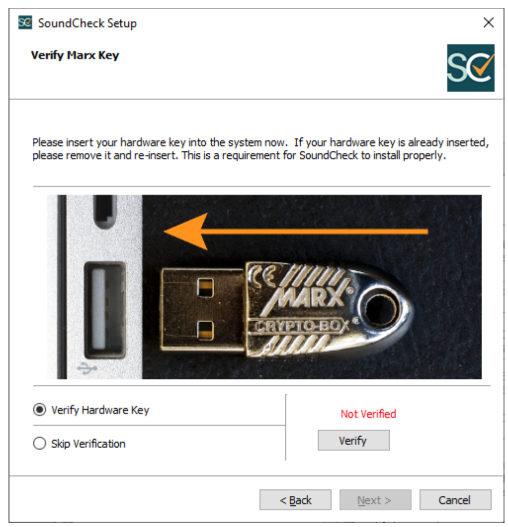
5. Follow the prompts for:



- **Welcome Screen**
- **Installation Location Prompt -**

  Windows - by default installs to: C:\SoundCheck 18

  macOS - by default installs to: /Applications/SoundCheck 18
- **Select Components** - allow you to select if the Demo-Viewer is installed
- **License Agreement** (See ***Listen Software License Agreement on page i***)
- **Verify Hardware Key** (Does not appear if the Hardware Key is already inserted)



Confirm that the hardware key has been inserted, then click **Install** to continue the installation.
or
Select "**Skip Hardware Key...**" if you are installing the Demo Viewer without a Hardware Key.
See ***SoundCheck Demo Viewer - Windows only on page 10***.

- **Installation Complete**

SoundCheck 18 Setup Wizard

6. Open SoundCheck. The **Setup Wizard** runs automatically at startup. The wizard can also be run by clicking **File** on the SoundCheck Main Screen and selecting **Setup Wizard** (Ctrl+Z).

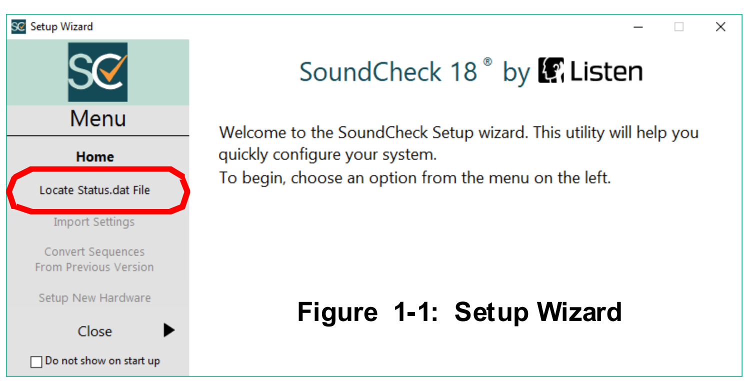


Figure 1-1: Setup Wizard

7. Proceed to

   Setup Wizard on page 35.

macOS - User-Approved Kernel Extension Loading

Introduction

Starting in macOS 10.13 'High Sierra', Apple introduced a system that will automatically prevent users from installing software that wasn't downloaded from the App Store unless the user manually allows this from System Preferences. Once the user has allowed the installation manually, all other software by that developer will be allowed to pass automatically without having to repeat the steps.

Approval is automatically granted to third-party KEXTs that were already present when upgrading to Mac OS High Sierra.

In-Depth Explanation

This feature enforces that only kernel extensions approved by the user will be loaded on a system. When a request is made to load a KEXT that the user has not yet approved, the load request is denied and Mac OS presents the alert shown in ***Figure 1-2***.

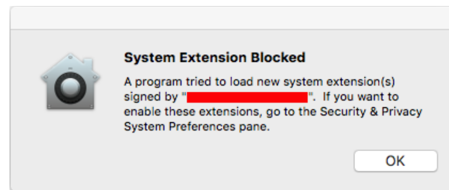


Figure 1-2: Blocked Kernel Extension

This prompts the user to approve the KEXT in **System Preferences** > **Security & Privacy** as shown in ***Figure 1-3***.

This approval UI is only present in the Security & Privacy preferences pane for 30 minutes after the alert. Until the user approves the KEXT, future load attempts will cause the approval UI to reappear but will not trigger another user alert.

The alert shows the name of the developer who signed the KEXT so the user has some information to decide whether to approve the KEXT. This name comes from the Subject Common Name field of the Developer ID Application certificate used to sign the KEXT. Because of this, developers are encouraged to provide an appropriate company name when requesting KEXT signing identities.

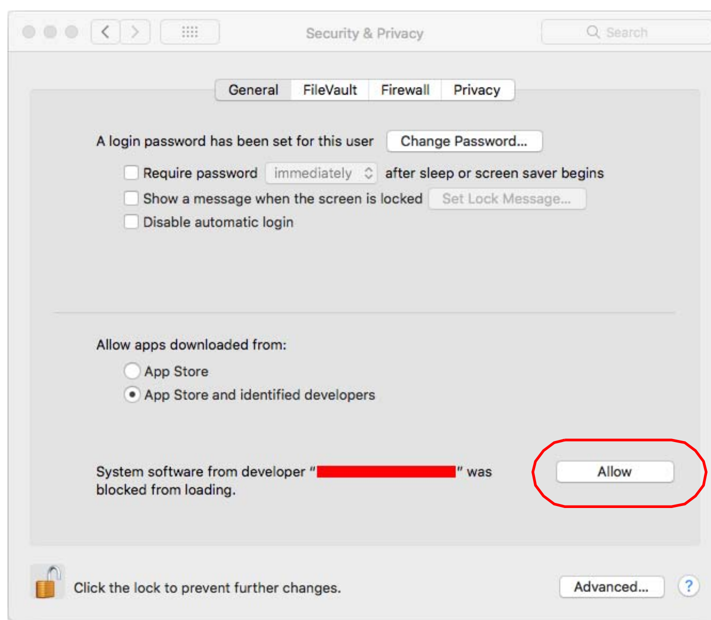


Figure 1-3: User Approval To Load A KEXT

When the user approves a KEXT, they are at the same time approving these other KEXTs signed by the same Team ID:

If the approved KEXT is located in an application's bundle, all other KEXTs signed by the same Team ID in the same application's bundle are also approved.

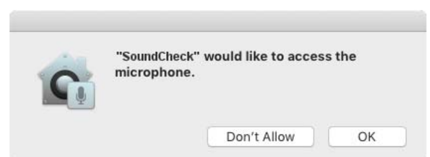
If the approved KEXT is located in the app's sub-directory inside /Library/Application Support, all other KEXTs signed by the same Team ID found in that same sub-directory are also approved.

All KEXTs in /Library/Extensions signed by the same Team ID are also approved.

Once approved, the KEXT will immediately be loaded or added to the prelinked kernel cache, depending on what action was blocked. Subsequent requests to load the KEXT will proceed silently as on previous Mac OS versions.

Approved KEXTs are tracked in a system-wide policy database through the team identifier in the KEXT's code signature and the bundle identifier from the KEXT's Info.plist, so updating a KEXT that has already been approved will not trigger a new approval request.

macOS Mojave Security Features



There are new security features in macOS Mojave that require you to allow access to the "microphone" the first time SoundCheck (or any audio and recording software) is launched.

The prompt is confusing as the prompt, "microphone", refers to any input device, including audio interfaces. In order for SoundCheck to be able to record audio, you must enable access to the "microphone". If you do not see the prompt and are unable to record audio in SoundCheck, you will need to enable recording by opening the "Security & Privacy" panel in "System Preferences".

Silent Installation

Silent or unattended installation is available as of SoundCheck 16. You can call the installer from an **Elevated Command Prompt**. This is useful when installing SoundCheck on several computers at once.

Example:

1. Copy the SoundCheck executable to a local drive or note the path to the file if it is on a server.
2. Click the **Windows icon** and type CMD in the search field.
3. Right-click **cmd.exe** and select **Run as administrator** as shown in ***Figure 1-4***. This is an Elevated Command Prompt.
4. In the cmd.exe window enter the full path & file name of the installer plus **/S** and click **Enter**:

   C:\sc18.10.###_setup.exe /S (/S is case sensitive)
5. The LabVIEW Runtime Engine window will open and close if this is the first time the current version of SoundCheck has been installed.
6. The Hardware Driver will open and close followed by a SoundCheck splash screen.
7. The SoundCheck shortcut is added to the desktop when the install is finished.
8. Open SoundCheck and the Setup Wizard opens so you can select the status.dat location and hardware setup. See ***Setup Wizard on page 35***.

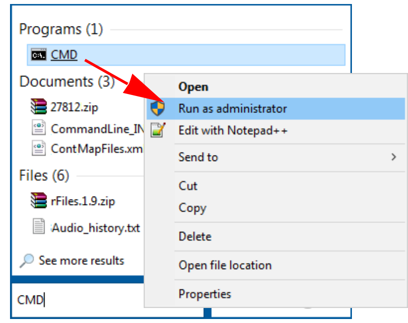


Figure 1-4: Elevated CMD Prompt

Listen Hardware Drivers for Windows

Device Driver Installation Wizard

The Driver Installation runs automatically when the SoundCheck installation runs. Drivers are included for Listen Hardware such as: AmpConnect 621, AudioConnect, SoundConnect 2, AmpConnect ISC and DC Connect. All Portland Tool and Die device drivers such as the BTC-4148/4149 are included as well.

AmpConnect ISC and DC Connect

As of SoundCheck 13, new drivers for AmpConnect ISC & DC Connect have been included in the SoundCheck installation process. The new driver will not work in versions prior to SoundCheck 13. To use AmpConnect or DC Connect with SoundCheck 12 (and previously supported versions), you will need to manually Rollback the device driver in Windows Device Manager. Please refer to the latest AmpConnect ISC or DC Connect manual for driver rollback procedures.

AmpConnect ISC in SoundCheck 13

The AmpConnect driver used in SoundCheck 18 creates the following limitations for installations of SoundCheck 13 on the same system:

- In SC13 you cannot control the AmpConnect headphone output
- In SC13 the serial number is removed from AmpConnect audio device name, requiring a relink of audio channels in the hardware editor

If you need to use AmpConnect in SoundCheck 13:

- Go to Add/remove programs in Windows
- Select “AmpConnect USB SC driver” and click **Uninstall**

AudioConnect and SoundConnect 2

These drivers are included in the SoundCheck 18 software installation.

AudioConnect 4x4

Important:* *The Windows drivers are not included in the SoundCheck 18 software installation. The AudioConnect 4x4 interface must be connected to the computer when the drivers are installed. Drivers are provided with the device and are available on the Listen website; https://support.listeninc.com/hc/en-us/sections/200370694-Drivers.

Drivers for macOS

- Only hardware with Core Audio drivers or Mac specific drivers can be used
- Listen hardware such as AmpConnect ISC, AudioConnect and AudioConnect 4x4 use Core Audio drivers
- DC Connect is not available on macOS

Please refer to ***Hardware Compatibility List on page 557*** for information on Audio Interfaces approved for use with SoundCheck Mac.

Hardware Key

- **SoundCheck 18 requires a hardware key as shown in *Figure 1-5*. This is 64-bit OS compatible.**
- **The SoundCheck hardware key must be connected to the computer in order to open SoundCheck**
- **The SoundCheck Demo Viewer is installed along with the full version. You can preview the software and view data without the hardware key.**

Figure 1-5: Hardware Key

- **Versions of SoundCheck prior to SoundCheck 13 will not work with this key**

Warning! Do not lose the hardware key!

- Do not lose the hardware key for the SoundCheck system
- It unlocks the functionality of your SoundCheck software
- For insurance purposes, this key represents the full value of your system and should be noted in your company's list of assets
- We recommend that it be securely attached to the computer to avoid loss or theft

Hardware Key Installation

- Insert the included hardware key into the USB port of the computer when prompted during the SoundCheck installation process. The computer will recognize the new hardware key as the driver is included in the software installation.

Important! ***Do not to remove the hardware key while SoundCheck is running. The hardware key can be damaged. If damaged, it will need to be returned to Listen, Inc. for replacement.***

Note: If the USB key is moved from one USB port to another, the driver will automatically re-install for the new USB port.

Electrostatic Discharge (ESD) Precautions

Use the following precautions to help neutralize the difference in electrical charge between your hardware key and the computer, before contact is made. This should help to protect your hardware key from ESD or Static Shock.

- Use a rubber mat that has been specifically designed as an electrical insulator. Do not use a mat designed to decrease electrostatic discharge as protection from electrical shock.
- Use a grounded wrist strap designed to prevent static discharge
- Use antistatic or electrostatic discharge (ESD) preventive clothing or gloves
- Avoid touching the USB contact pins
- For grounding purposes, verify that your computer provides excellent conductivity between the power supply, the case, the mounting fasteners, and the mainboard.

Normal Mode with Hardware Key

With a valid Hardware Key and no Acquisition Channels enabled (0 Channels):

- You can not open, edit, apply or insert an Acquisition Step into a sequence.
- Acquisition Steps in the Sequence are skipped.

With a valid Hardware Key and Acquisition Channels enabled (2, 4, 8, 16, or 32):

- When inserting or editing an Acquisition Step you are limited to the number of hardware channels enabled on the Hardware Key. A warning is issued if you try to exceed this number.
- If you open a sequence that uses a number of hardware channels that is greater than the number of channels enabled on the Hardware Key, a detailed warning is issued when the sequence is pre-loaded, indicating that the sequence was created using more channels than is available. Acquisition steps such as this will be skipped when the sequence is run.

Hardware Key Laser ID

The Laser ID of the Hardware Key currently connected to the system is shown on the SoundCheck Main Screen.

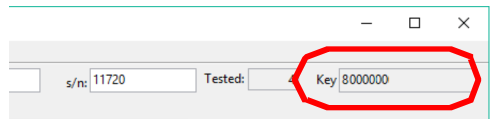


Figure 1-6: Hardware Key Laser ID

SoundCheck Demo Viewer - Windows only

The Demo Viewer is installed along with the full version of SoundCheck. Shortcuts are created on the desktop as shown in ***Figure 1-7***.

- The Demo Viewer is not available for macOS

Figure 1-7: Shortcuts

All data generated in the demo viewer is randomly adjusted in level, Randomized, and is therefore not valid.

The SoundCheck wall paper will change to indicate this as shown in *Figure 1-8*.

- Recall, save and print data is randomized
- Create and modify test sequence but Save sequence is disabled
- 2 channels of input/output hardware channels
- A valid Hardware Key is required to register ActiveX components during SoundCheck installation. The Hardware Key is not required to use ActiveX in Demo Mode.
- All other functionality of SoundCheck is available

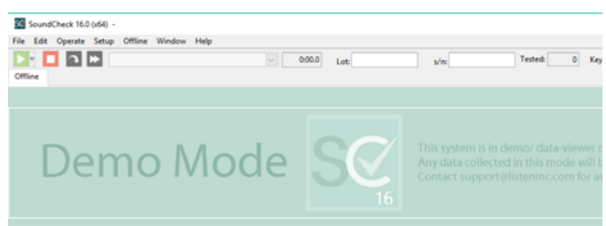


Figure 1-8: Demo Wallpaper

SoundCheck executes the test sequence and adds random values to the data displayed or saved. All data generated in the demo viewer is randomly adjusted in level and is therefore not valid.

Optional Modules and Protected Sequences

Once SoundCheck is installed, with the proper status.dat file and hardware key, a list of Optional Modules and Protected Sequences is available. Click on the **Help** menu of the SoundCheck Main Screen and then select Optional Modules. The current list of modules and sequences available is found in ***Figure 1-9: Optional Modules List***. This indicates which modules and sequences are currently enabled on the system as well as items that can be added.

Note: Acquisition is now an option that allows only a specific number of hardware channels.

Optional Modules

Installed

2000 Limits Editor
2001 Harmonic Distortion
2002 Sequence Editor
2003 Spect Analyzer (Scope FFT)
2004 Post-Processing
2005 RTA Spect Analyzer
2006 Time Selective Response
2007 Loudness Rating
2008 Attack and Release
2009 Statistics
2010 Save to Database
2011 Polar Plot
2012 Equation Editor
2013 EQ a Wav File
2014 Signal Generator
2015 Multimeter
2016 Loose Particle
2017 Stimulus
2018 Stepped Sine
2019 IM Distortion
2020 Multitone
2021 Transfer Functions

Not Installed

2022 2 Channel Acquisition
2023 4 Channel Acquisition
2024 8 Channel Acquisition
2025 16 Channel Acquisition
2026 32 Channel Acquisition
2027 64 Channel Acquisition
2029 SC ONE - Acoustics
2030 Perceptual Rub & Buzz
2031 Zwicker Loudness Rating
2032 Waveform Filter
2033 Active Speech Level
2034 Calibration Editor
2035 Reserved
2036 Distortion Analyzer
2037 Frequency Counter
2038 Reserved
2040 Strip Chart Recorder
2099 SC Win
2100 SC Mac
1301 SoundMap CSD
1300 SoundMap Full

3000 Hearing Aid Application for ANSI 3.22-1996
3102 EN50332-1 Max SPL for Head-phones
3103 IEEE 1329, Clause 10 Seq (JB03xx)
3104 TIA-470 Seq (JB04xx)
3105 IEEE 269, Clause 9 Seq (JB05xx)
3106 Hearing Aid Magnetic Comp Seq (JB06xx)
3107 TIA USB Headsets Seq (JB07xx)
3108 TIA-810 Legacy Narrowband Seq (JB08xx)
3109 TIA-920 Legacy Wideband Seq (JB09xx)
3110 ETSI TBR 38 Seq (JB10xx)
3111 TIA-920+ Narrow/Wideband Seq (JB0905)
3112 (JB12xx) Reserved
3113 (JB13xx) Reserved
3114 (JB14xx) Reserved
3121 Background Noise Sequences ETSI ES202396-1

Figure 1-9: Optional Modules List

Windows Setup Recommendations

High Performance Computer Settings

The SoundCheck computer is a core part of a test and measurement system. In general, test instruments do not automatically shut down to save power. Computer operating systems and the computer BIOS have power save settings that should be changed.

We recommend that Windows be set to the High Performance plan and USB ports are set to “Never Sleep”. Depending on the computer mother board, changes to automatic power saving may need to be made in the system BIOS as well.

Sleep Settings

Windows 10:

- Click **Start** > Settings > System > Power & Sleep
- Related Settings: Select **Additional power settings**
- Select **High Performance**
- Select **Change Plan Settings**
- Put the computer to sleep: Select **Never**

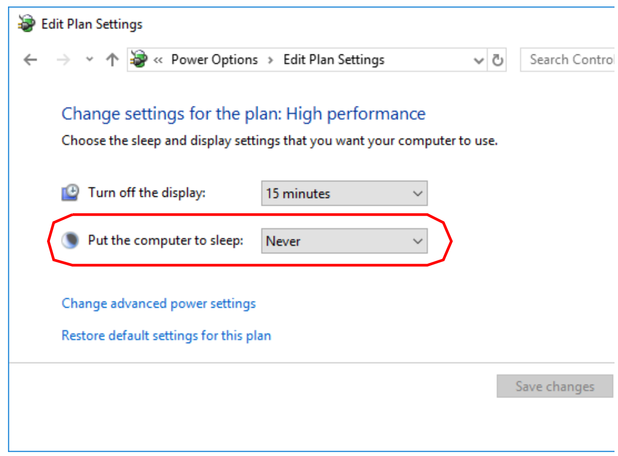


Figure 1-1: Select High Performance

USB Selective Suspend

- Next click on **Change advanced power settings**
- Scroll down to **USB settings**
- USB selective suspend setting: Select **Disabled**
- Click **OK** to save settings
- Close all remaining options windows

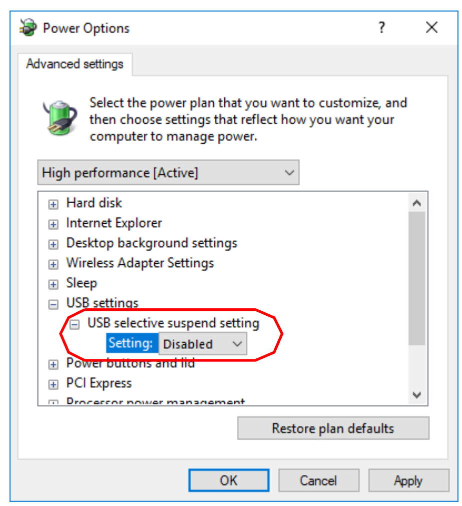


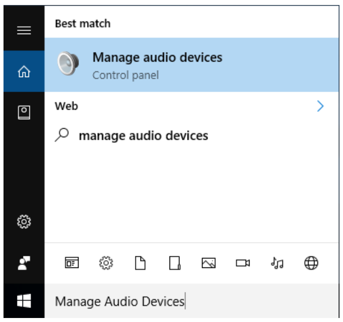
Figure 1-2: USB Selective Suspend Disabled

Audio Device System Settings

The following settings are recommended to prevent Windows systems sounds from inadvertently playing through a device under test or artificial mouth. Some system sounds are capable of damaging some transducers.

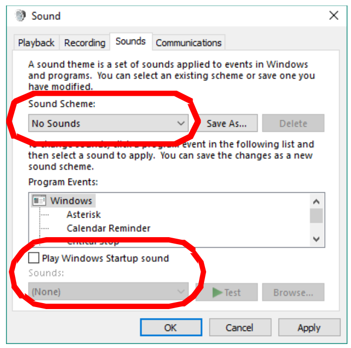
<table>
<tr>
<td>

1. Click **Windows Start**, Type **"Manage Audio Devices"** and hit Enter. This opens the Sound Control Panel.



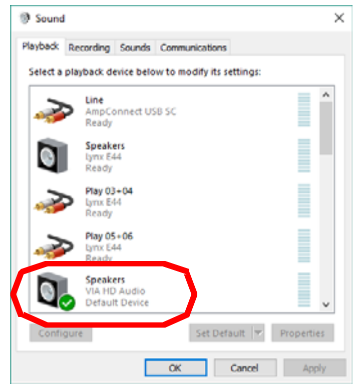
</td>
<td>

2. Click the **Sounds tab** and set the default scheme to No Sounds
3. Uncheck "Play Windows Startup sound"



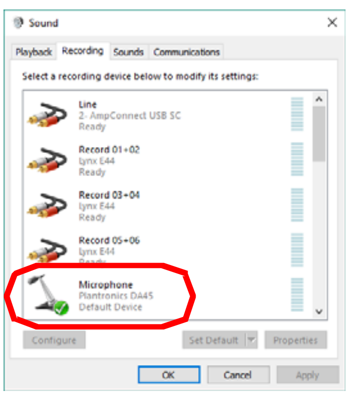
</td>
<td>

4. Select the **Playback tab** and set the default device to the motherboard audio interface (no SoundCheck devices)



</td>
</tr>
<tr>
<td>

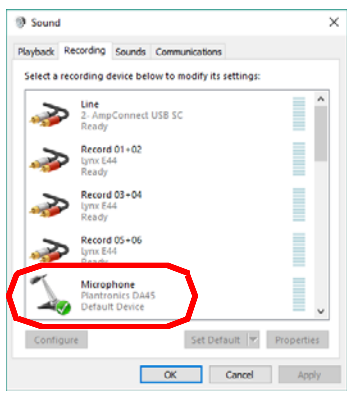
5. Select the **Recording tab** and set the default device to the motherboard audio interface (or any other device that is not used for test and measurement)



6. Click **OK** and close the remaining Control Panel windows

</td>
<td>

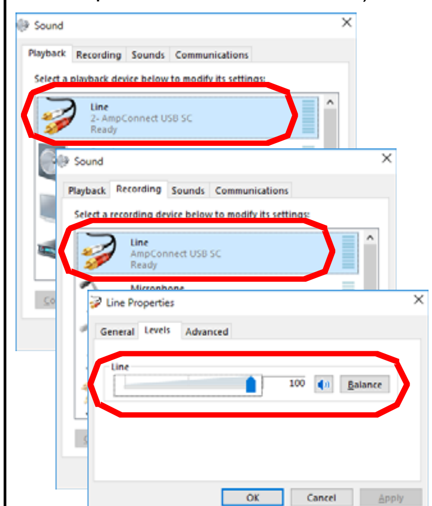
7. Select the **Recording tab** and set the default device to the motherboard audio interface (or any other device that is not used for test and measurement)



8. Click **OK** and close the remaining Control Panel windows

</td>
<td>

9. On the **Recording** and **Playback** tabs set the levels for all WDM / WASAPI devices used with SoundCheck to 100%. (e.g.: AmpConnect, AudioConnect)

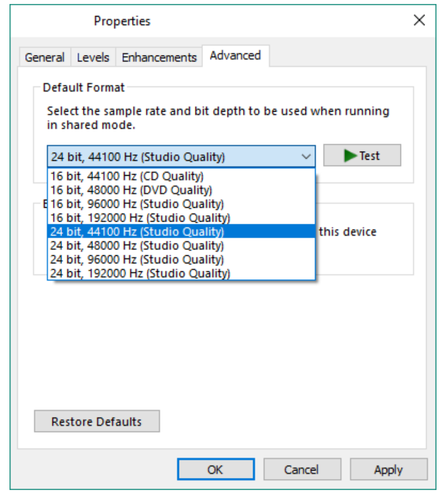


</td>
</tr>
</table>

WDM / WASAPI Device Sample Rate

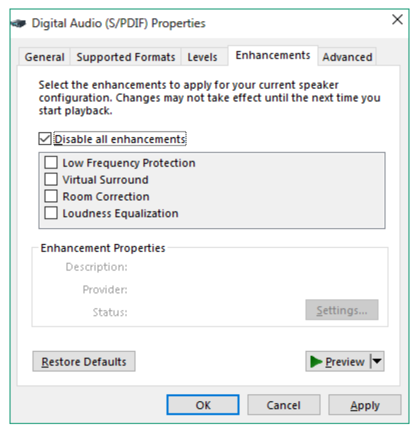
Note: ***When using WDM / WASAPI audio interfaces with SoundCheck, you will need to set the sample rate in the SoundCheck Hardware Editor and in the Windows Play and Record panels for the audio interface.***

1. Double click on a device in Playback or Recording
2. Select the **Advanced Tab**
3. Set the Sample Rate and Bit Depth to match the SoundCheck Hardware Editor



Audio Device Enhancements

Some audio devices feature Enhancement settings. When testing such a device, these enhancements should be shut off.



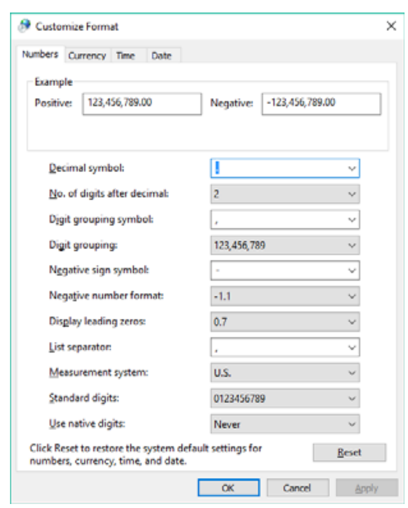
European Decimal Notation

The “Decimal Display Format” can be changed to European Style in Windows. (comma in place of period)

SoundCheck must be closed before changing the decimal format. Changes will be seen the next time it is started.

Windows Instructions

1. Click **Start**
2. Select **Control Panel**
3. Select **Clocks, Language and Region Options**
4. Click **Change the date, time or number format**
5. Click **Additional Settings**
6. Change the Decimal Symbol to "," (comma) and Digit Grouping Symbol to "." (period)
7. Click **Apply** and click **OK** to close editing windows



Change SoundCheck Default

If you want to overrule your regional settings, use the following method.

You can change the default behavior by changing the SoundCheck INI file found in the root of your SoundCheck folder. This changes the **decimal mark** based on the language of the OS. This forces SoundCheck to always use a dot for decimal notation.

1. Exit SoundCheck
2. Add the following line to **SoundCheck 18.ini** file.

   useLocaleDecimalPt=False
3. Save the changes to the .INI file and start SoundCheck

Windows Display - Text Size

The settings of the Windows display resolution effects the display windows and information tabs in SoundCheck. The following settings should always be used for a SoundCheck system.

1. SoundCheck must be closed
2. In Windows 10, Right-click the desktop and select **Display Settings**
3. Open the Display menu
4. Under **Scale and layout** select **100%** (Recommended)

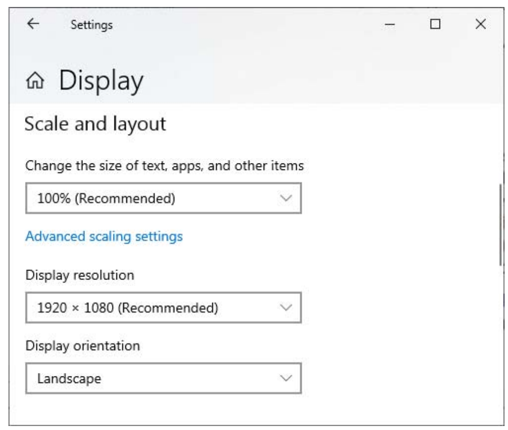


Figure 1-3: Use 100%

This is the typical default setting for Windows but it is sometimes changed when individual users are trying to make icons larger on the Windows desktop or making program menu fields larger. Setting this higher than 100% causes fields in SoundCheck to overlap and in some cases become "not visible".

Medium (125%) and Large (150%) should not be used. Menus will not be readable.

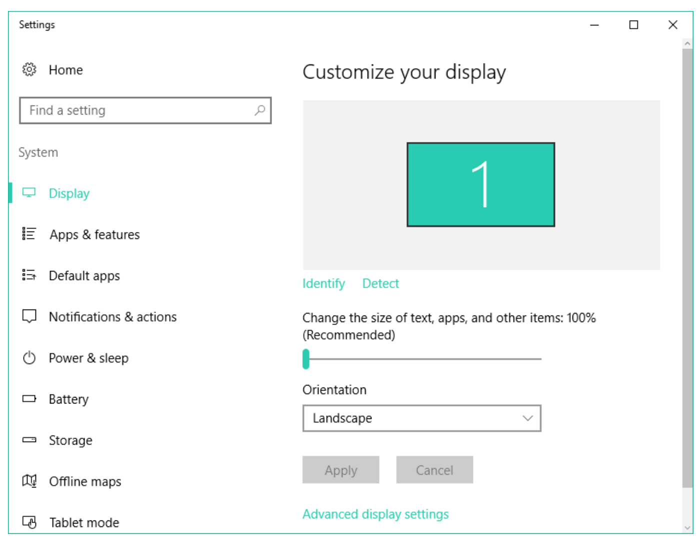


Figure 1-4: Windows 10 Use 100%

macOS Settings

In order to prevent errors when recalling and saving data during sequence runs, the **"Put hard disks to sleep when possible"** option must be unchecked in **Preferences > Energy Saver**.

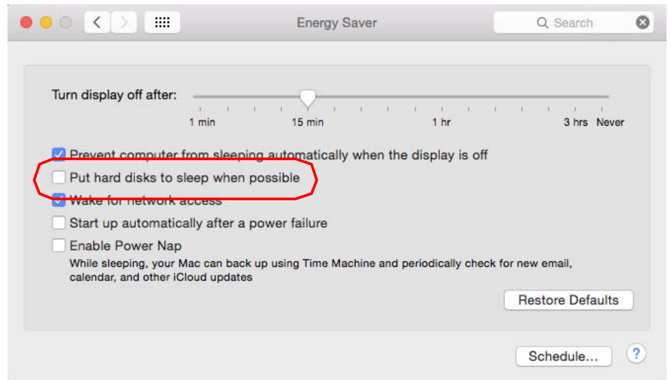


Figure 1-10: Energy Saving Settings

macOS Functions

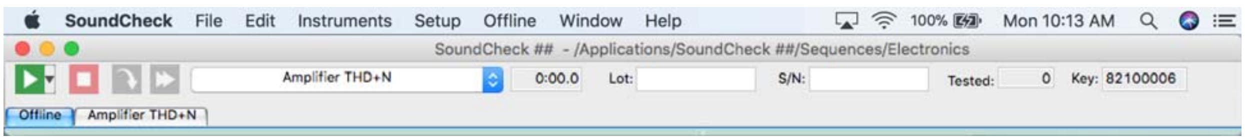


Figure 1-11: Main Screen Tool Bar

Due to differences in the macOS menus, you will notice that SoundCheck menus behave differently.

- The Main Screen is simply the SoundCheck Tool Bar aligned at the top of the Mac screen as shown in ***Figure 1-11***. The SoundCheck background is hidden intentionally.
- The top menu changes depending on which SoundCheck window is open. For an example, refer to ***Memory List on page 18***.

macOS Differences

- The Mac version requires **optional module 2100**
- Demo Viewer is not available
- External software control using TCP/IP is available. ActiveX is not available.
- Microsoft Office is not supported (currently not supported by LabVIEW)

  This affects:
  - No Autosave to Excel
  - No Autosave to database
  - No Report functions from Memory List: Word, Excel or HTML
  - No printing to Word or Excel
- National Instruments DAQmx hardware is not compatible with Mac OS. This includes NI 4461 and cDAQ devices.
- Some Keyboard Shortcuts are not available. Refer to ***Mac Keyboard Shortcuts on page 593***.
- Serial Port **Footswitch control** and **Serial Port External Buzzer** are not supported

Operation

Sequence files are cross platform compatible.

This includes:

- Calibration - .CAL
- Data - .DAT
- Sequence - .SQC
- All step templates

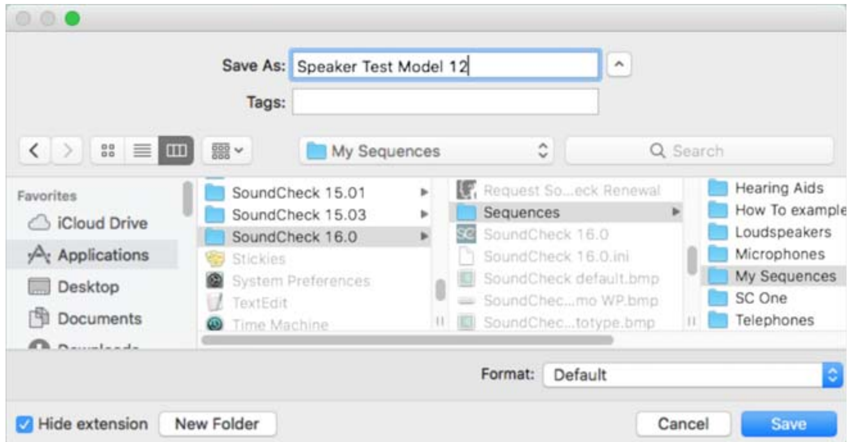


Figure 1-12: Sequence Save

Acquisition Step

When using Audio Interfaces with unstable Latency, Acquisition steps should use a minimum Record Padding setting of 200 mSec as shown in ***Figure 1-13***.

Auto Delay **MUST** be used in Analysis Steps.

Please refer to ***Hardware Compatibility List on page 557*** for information on Audio Interfaces approved for use with SoundCheck Mac.

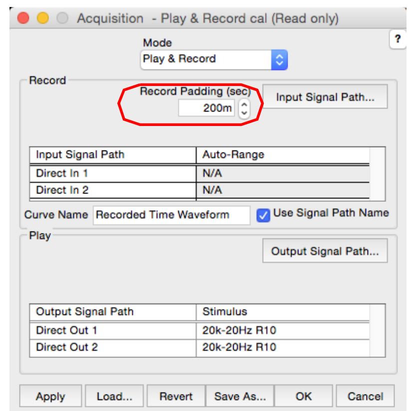


Figure 1-13: Acquisition Record Padding

Select Current Folder

When selecting a folder location in Autosave, Recall and other operations, click **Open**. This is the same function as clicking "**Current Folder**" in Windows.

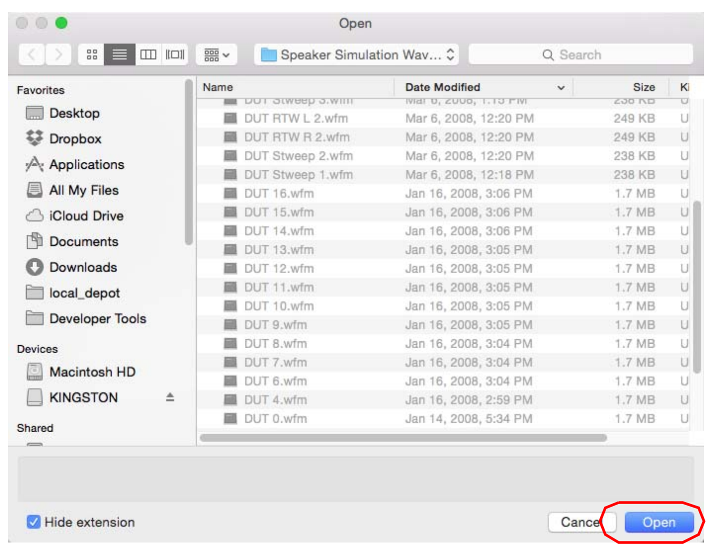


Figure 1-14: Click Open to Select Folder

Memory List

The functions for the Memory List are at the top of the Main Screen as shown in ***Figure 1-15***.

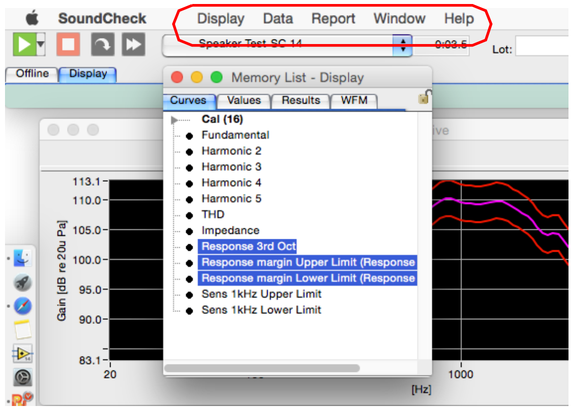


Figure 1-15: Memory List Functions

New Features SoundCheck ® 18.1

AmpConnect 621 Control

SoundCheck 18.1 introduces control for the new AmpConnect 621 interface. This all-in-one multichannel interface (6-in, 2-out) has no front-panel controls and is exclusively controlled via the software. This minimizes potential operator error and ensures seamless integration with test sequences and virtual instruments.

See ***AmpConnect 621 Message Configuration on page 294***.

Auto Delay+ Algorithm

The new Auto Delay+ algorithm offers increased accuracy for open loop measurements, especially in cases where the delay is longer than 1 second. It is particularly useful where testing involves sending test signals via the cloud, such as smart speakers, automotive audio, hearables, etc.

See ***Auto Delay+ on page 169***.

Power Sum of Waveform

This has been updated so that it is normalized by the sampling rate.

See ***Power Sum of Waveform on page 244***.

New Impedance Measurement Method Selections

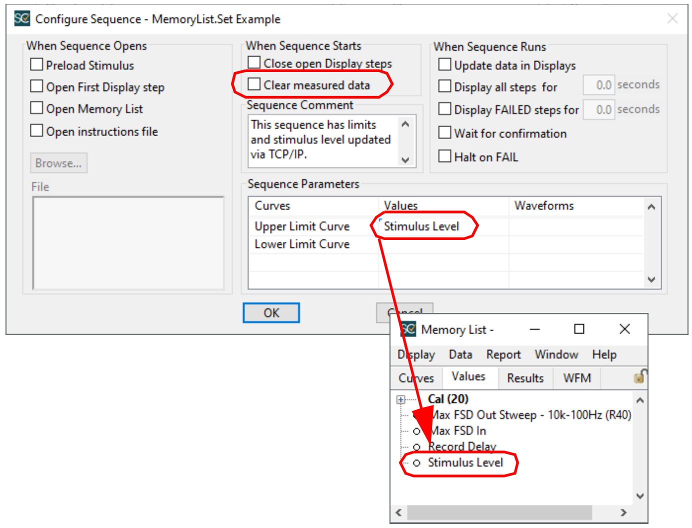
Selections have been expanded to included AmpConnect 621 along with others.

See ***Electrical Tab - Impedance on page 204***.

New Features from SoundCheck 18.0

SoundCheck18's new features focus on automation and simplicity. Automation features include the ability to pass test configuration data to and from external programs, and control of MEMS interfaces via a sequence. Setup and test development is simplified with 'plug and play' functionality for Listen hardware, simplified gain control, and improved sampling rate management.

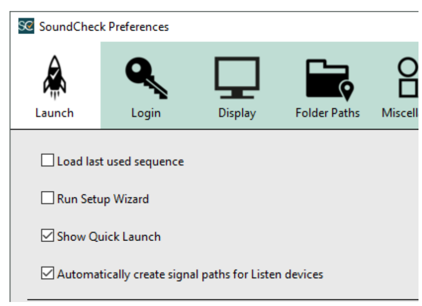
Improved Automation Simplifies Test Sequences



Soundcheck's new **Sequence Parameters** feature allows the user to pass test configuration data into the memory list from external programs using TCP/IP commands. By externally storing parameters such as limits, test levels, and test signals, a single sequence can be used for multiple products, simplifying the number of sequences that a large organization needs to maintain, and reducing test configuration time. It is also useful for applications where a sequence needs to be run many times with different parameters, for example, testing voice recognition with a range of voices or test levels. This functionality is particularly useful where SoundCheck tests are run as part of a larger automated test framework controlled using Python, C#, Visual Basic, LabVIEW, etc.

See ***Sequence Parameters on page 445*** and ***MemoryList.Set TCP/IP Command on page 493***.

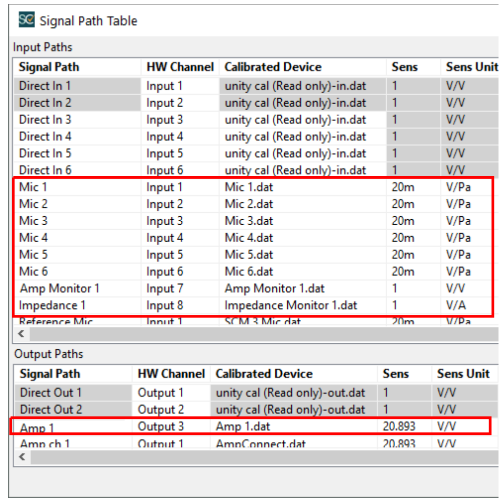
Reduced Set Up Time with Plug and Play Listen Hardware



Plug and Play, aka Automatically Create Signal Paths for Listen Devices

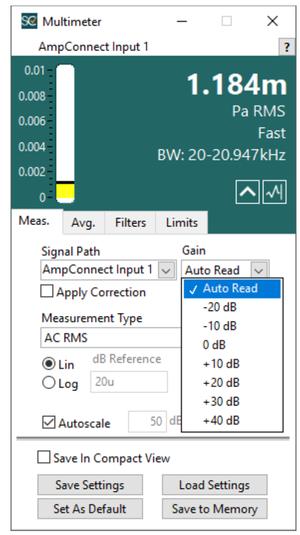
Improved hardware setup enables you to plug in and make measurements faster. Listen and Portland Tool & Die Hardware are now true 'plug-and-play' which minimizes configuration time, particularly with multichannel interfaces. Simply connect your AmpConnect 621, AmpConnect ISC, BTC-4149 or other Listen interface via the USB cable and the software auto-populates parameters such as sampling rate and calibration values, and creates input and output signal paths on the next start of SoundCheck. Multiple hardware items are stored within the software, so that they are immediately recognized when connected, and the software will automatically default to the correct device.

When substituting devices of the same type, SoundCheck will re-use the configuration previously set up, either manually or during a sequence, so that signal paths and sequence configurations do not need to be updated. This makes deploying a sequence over multiple stations simpler and faster and will offer significant time-savings for large-scale operations who need to configure multiple stations or move hardware around between stations.



See ***Automatically Create Signal Paths for Listen Devices on page 46***.

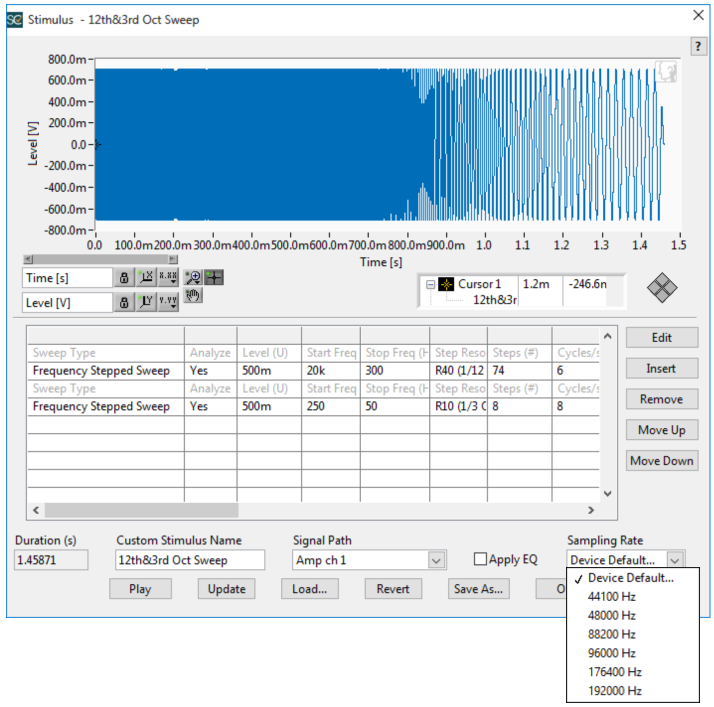
Simplified Gain Control Selection



Auto Read and Auto Range gain settings are now easily accessible as drop-down options in the gain menu in the Acquisition Editor, and Auto Read is available in Virtual Instruments. Setting the gain to Auto Read, means that SoundCheck will automatically adjust the gain based on manual input from the user, making it much faster to make quick gain adjustments while developing sequences. When the gain is set to Auto Range, SoundCheck will automatically identify the correct level of gain to optimize the measurement signal to noise ratio. This makes it fast and simple to optimize tests in the R&D lab for the correct gain before transferring them to production.

See ***Gain Field on page 142*** and ***Gain - Auto Read on page 457***.

Improved sampling rate management



Many audio interfaces have a latency that is dependent on the sampling rate, and SoundCheck 18 brings improved communication with these devices.

Firstly, for any hardware device, the user can specify the latencies for various sampling rates, either by entering them manually or importing from a file. SoundCheck will then automatically use the correct latency for any sampling rate specified in the stimulus editor.

See ***Sampling Rate/Latency Table on page 68***.

Secondly, the sampling rate can now be set in the Stimulus Editor. This is useful for users who need to do iterative tests involving multiple sampling rates and/or set the sampling rate via a sequence.

See ***Sampling Rate on page 114***.

Finally, when using WAV files, the sampling rate no longer needs to be manually set to match the sampling rate of the WAV file. This makes it simpler to switch between WAV files, and is useful for anyone making measurements with WAV file signals such as speech or music.

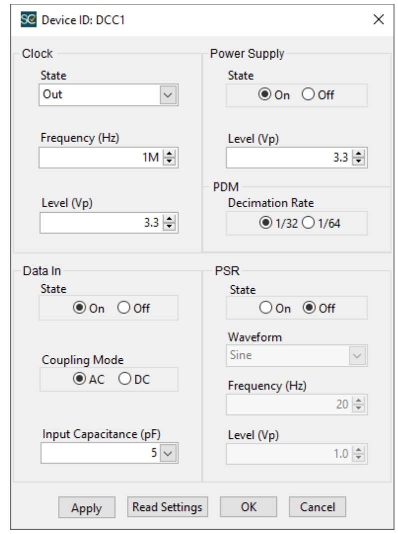
See ***WAV Sample Rate on page 114*** and ***WAV File playback on page 462.***

New Digital I/O Control

Digital I/O control is now a separate message step. This means that all controls for I/O settings are completely separate from other AmpConnect settings, minimizing the possibility of other parameters such as gain or routing being accidentally modified. The new settings offer a uniform appearance and switching scheme across all devices with digital I/O control, so that sequences written, for example, with NI hardware, are easily converted to work with AmpConnect.

See ***Digital I/O Message on page 290***.

MEMS Interface control



The Portland Tool & Die MEMS microphone interfaces can now be fully controlled from within SoundCheck. This offers faster setup of the measurement interfaces with SoundCheck as well as greater measurement reliability as the device settings can be built into the test sequence. Sequences using the R&D grade DCC-1448 interface will also work seamlessly with the production-grade PQC-1448, making it much simpler to transition sequences from the R&D lab to the production line.

See ***Portland Tool & Die DCC-1448/9 and PQC Control on page 285***.

WASAPI Driver support

SoundCheck 18 now includes WASAPI driver support in Windows 10. This driver offers full multichannel support, allowing each individual channel to display separately in the hardware editor, rather than as channel pairs.

See ***The essential hardware settings for all channels of a multichannel Hardware Configuration can be viewed and edited in the table as shown in Figure: 8-6. on page 63***.

TEDS support

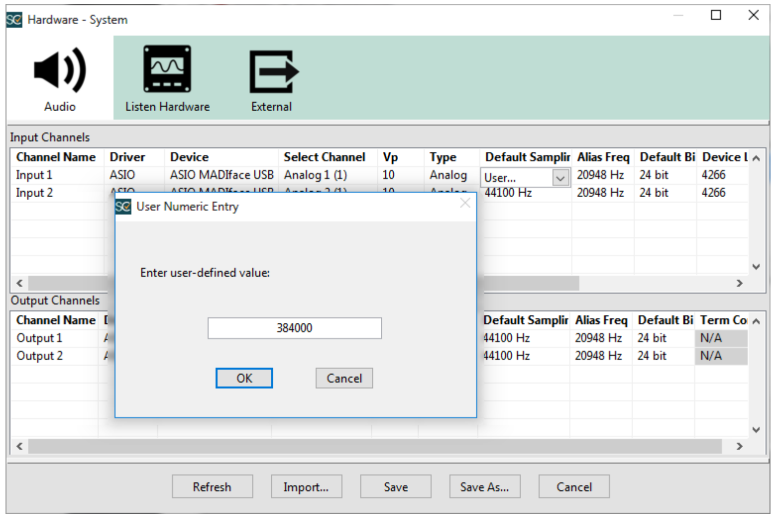
TEDS Support (with compatible Listen hardware) enables automatic identification, configuration and calibration of TEDS microphones and accelerometers, saving time on initial hardware setup and whenever hardware is changed.

dB ASL

Memory List value units can be set to dB ASL and will be properly scaled when used in a Stimulus Step when using the **MemoryList.Set** command. See ***MemoryList.Set Command on page 493***.

User Defined Sample Rate

384/352 kHz Sample Rate Interfaces



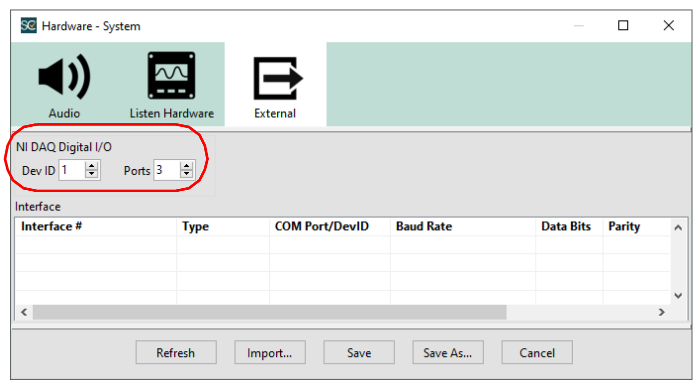
For audio interfaces that support sampling rates above 192 kHz you will need to select **User...** in the **Default Sampling Rate** drop-down list and then enter the rate supported by the interface. Once the rate is set for the input channels, the output channels will automatically update to the new sample rate.

Sample rates up to 384 kHz are supported.

NI 4461 Interface

The NI 4461 operates at a sample rate of 51.2 kHz. Select **User...** from the Sampling Rate drop-down list. Enter a sample rate of **51200** and click **OK** to save the new sampling rate.

NI DAQ Digital I/O



Since only one NI DAQ Digital I/O device can be selected at a time the selection options have been simplified.

Select the Device ID

The **Dev ID** must correspond to the Device ID shown in the NI MAX application. NI MAX is available in the DAQmx installer.

Set the number of Ports

This must be the maximum number of ports available on the NI DAQ device.

The remaining settings are made in Message Steps in a SoundCheck sequence.

See ***NI DAQ Digital I/O on page 76*** for more information on Digital I/O Hardware Editor settings.

See ***Digital I/O Message on page 290*** for more information on Digital I/O Message Steps.

Introduction

Congratulations on your purchase of SoundCheck®, created by Listen, Inc. SoundCheck is the first dedicated electroacoustic test and measurement system for production line quality control testing as well as research and development applications. SoundCheck was developed by some of the most knowledgeable and skilled engineers and software programmers in the world. Our goal is to provide fast and accurate testing with an intuitive user interface. Listen will constantly be looking for new ways to improve SoundCheck. Customer feedback helps us develop better sound measurement solutions and is greatly appreciated. Please call or email us at **support@listeninc.com**.

Listen Hardware Support

As of SoundCheck 14, support for the control of AudioConnect, SoundConnect 2, as well as the Portland Tool & Die BTC-4148/ 4149 Bluetooth interface are available.

SoundCheck 18 supports AmpConnect ISC™, Listen's integrated hardware interface which replaces a power amplifier, microphone power supply, impedance box and digital I/O card in your testing setup. AmpConnect ISC can be fully controlled either via the sequence editor, or directly via a control panel that replicates the appearance of the front of the hardware.

See ***AmpConnect ISC Control In SoundCheck on page 296.***

Please refer to the AmpConnect ISC manual for more information.

Note: SoundCheck 10.11 and above is required to use AmpConnect ISC™. For driver change note See ***AudioConnect and SoundConnect 2 on page 5***.

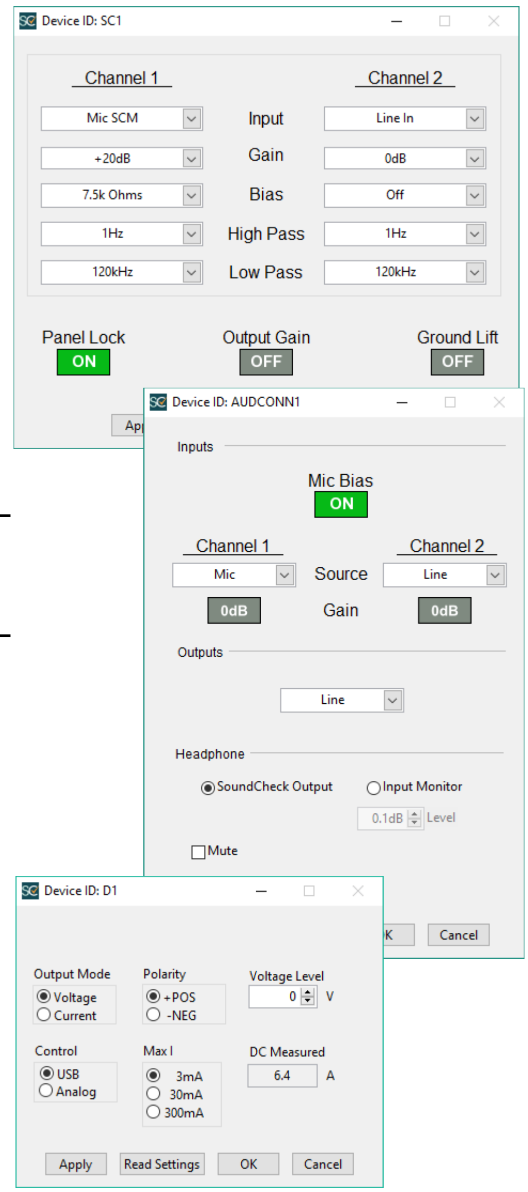


Figure 3-1: Listen Hardware Control

Full Multichannel Acquisition

An unlimited number of hardware channels enables you to use as many channels simultaneously as your hardware and computer memory can support.

This offers advantages for many types of testing including:

- Production testing of surround sound electronics
  - all channels can be measured simultaneously, enabling an entire surround sound test to be carried out much faster.
- Stereo headset testing
  - Two channels of acoustic signal and two channels of impedance can be measured simultaneously, increasing the speed of test.
- Multichannel devices
  - Pro audio mixers and other multichannel devices can be tested faster using SoundCheck.
- Batch mode testing
  - Multiple channels can be utilized for batch-mode testing of multiple devices at once, such as microphones.

The multichannel capability of SoundCheck also means that you can play and record simultaneously on different devices (audio interfaces, or data acquisition devices). This allows an NI data acquisition card (such as the PXI/PCI 4461 and 4462) to be used in conjunction with an audio interface, combining the flexibility of a Windows Multimedia environment with the high accuracy of the NI hardware.

SoundCheck communicates with Windows Multimedia devices in real time by sending WAV files. It is the only audio test system that offers complete control of Windows multimedia devices, making it the ideal solution for testing audio electronics and multimedia devices such as IP phones, MP3 players and Bluetooth headphones.

SoundCheck ONE™

SoundCheck ONE is an entry-level SoundCheck system which is essentially a lower cost, simplified, version of SoundCheck coupled with the AmpConnect ISC or AudioConnect hardware. SoundCheck ONE offers the capability to test loudspeakers, microphones and headphones within predetermined sequence templates.

Although the user interface is the same as in the full version of SoundCheck, rather than using the Sequence Editor, SoundCheck ONE users are supplied with sequence templates. These templates serve as the starting point for all SoundCheck ONE tests and can be used to generate as many product specific sequences as desired by selecting parameters such as the stimulus signal, characteristics to be measured, frequency range, level and limits.

SoundCheck 18 hardware keys can be programmed to also work in SoundCheck ONE mode. This enables you to easily switch between SoundCheck 18 and SoundCheck ONE.

Refer to ***SoundCheck ONE™ on page 485*** for more information.

Global System Hardware and Calibration

- **System Hardware** - One Hardware Configuration to define and configure data acquisition equipment for all sequences
- **System Calibration** - One System Configuration to define the sensitivity of the input or output transducers along with any needed EQ and Correction curves

Test Sequence

SoundCheck allows you to develop tests or modify existing tests from our extensive library.

Each test, referred to as a ***"Sequence"*** is essentially a script. A Sequence is a series of ***"Steps"***, with each step performing a specific task. An extremely simple Sequence might have the following structure:

- **Stimulus Step** - Define and generate the signal to be sent to the DUT
- **Acquisition Step** - To play the Stimulus and record the DUTs response
- **Analysis Step** - For example, to calculate frequency response of the DUT
- **Limits Step** - To apply Pass/Fail criteria on acquired data
- **Display Step** - To Display data and results

Many other Step types are available, including Post-Processing, Printing, Statistics, etc. Each step type is clearly defined and explained later in the manual.

See ***Sequence Editor on page 435***.

Virtual Instruments

In addition to running pre-defined Sequences, you can also generate stimuli and analyze data using stand-alone ***"Virtual Instruments"*** (optional). These can be launched from the "Instruments" menu, and replicate familiar laboratory equipment. These include the following: Signal Generator, Multimeter, Oscilloscope, FFT-Spectrum, Real-Time Analyzer, Distortion Analyzer, Frequency Counter and Strip Chart.

See ***Virtual Instruments on page 455***.

Operating Principles

SoundCheck operates on the same principles as a traditional, stand-alone measurement system consisting of a *Signal Generator*, *RMS Multimeter*, tracking filter, and level recorder. With SoundCheck, all of these functions are implemented in software as VI's, or *Virtual Instruments*.

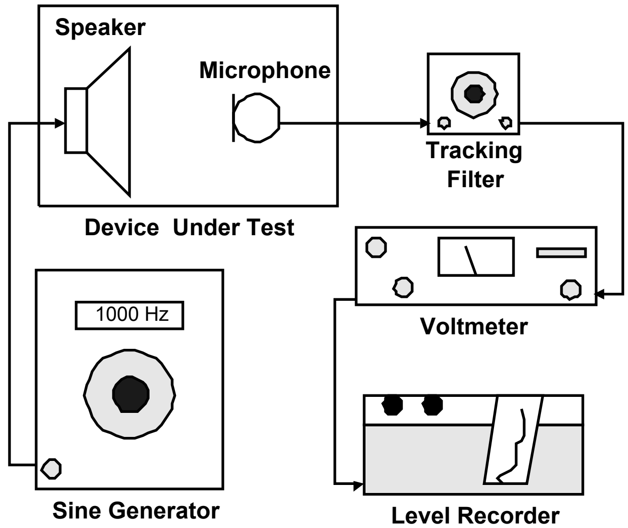


Figure 3-2: Traditional Hardware-based Test System

The advantages of software-based instruments are numerous. SoundCheck takes advantage of today's high-speed personal computers, professional audio interfaces, data acquisition cards and Windows software platforms. This saves thousands of dollars in hardware cost compared with traditional audio test and measurement systems. The system is modular, which means you can easily upgrade as your needs change.

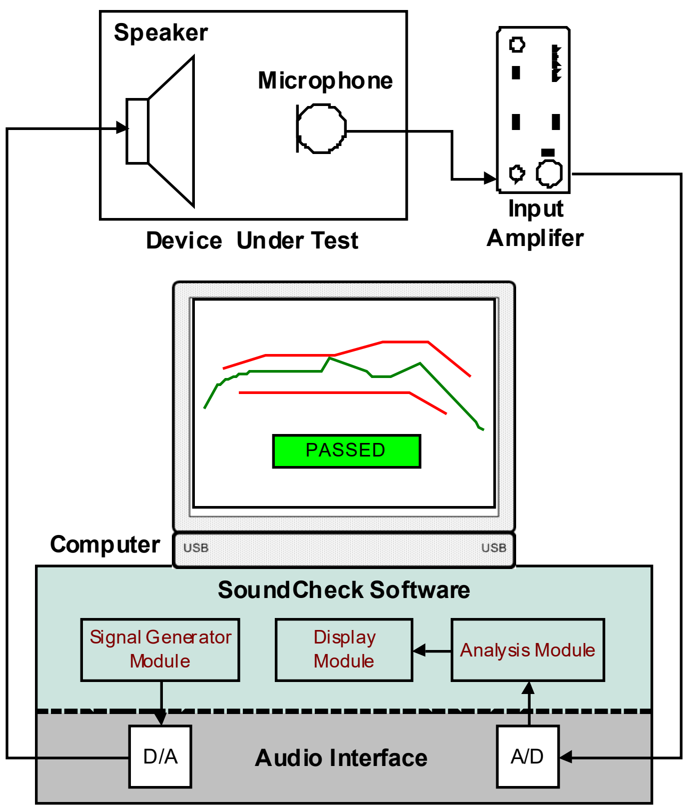
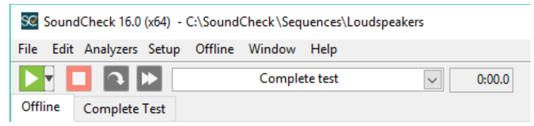


Figure 3-3: SoundCheck Software-based Test System

Sequences



Running a Sequence performs most measurements in SoundCheck. A Sequence is made up of individual steps, or operations that are strung together to create an overall test. Custom test procedures can be written or modified using the *Sequence Editor*. Typically, sequence names are product model numbers or device names. SoundCheck includes example sequences to aid in developing new sequences.

See ***Sequence Editor on page 435***.

Steps

A step is a unique operation that is executed in the order it occurs in a sequence. To edit a step in the active sequence:

- Choose **Setup** from the main screen drop-down list
- Select the category for the Step in the sequence to be edited (e.g., Stimulus, Acquisition, etc)
- You can also open the Sequence Editor and select a step from the right side of the editor.
- A step can also be saved as a template in the library on the left side of the Sequence Editor. The step templates are then available for use in other sequences. Every Step has a *Step Category* and a *Step Editor*.

As of SoundCheck 12, all attributes and fields of a step in the active sequence are linked to that sequence. Changes to the steps in the active sequence appear only in that sequence.

Sequence Run Status

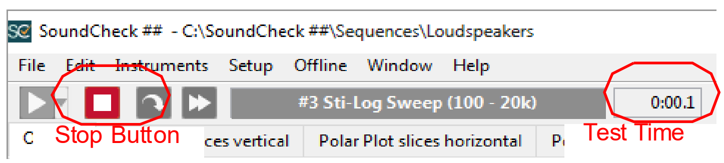


Figure 3-4: Stop Button

- After pushing **Start**, the **Stop** button turns red to indicate that the sequence is running
- The Test Time field shows the elapsed time of the sequence run
- Click the **Stop** button at any time during the sequence run to halt operation
- You can also hit the **Escape** key on the keyboard to **Stop**

See ***SoundCheck Main Screen on page 41***.

Test Equipment Setup for Typical Applications

Note: Chose the proper input and output Hardware Channels that correspond to the Signal Paths used in the selected test sequence.

Loudspeaker Setup

1. Connect an output of the audio interface to the input of the power amplifier.
2. Connect the output of the power amplifier to the loudspeaker under test.
3. Connect the microphone preamp cable to the microphone input on the microphone power supply (e.g., SoundConnect).
4. Connect the output of the microphone power supply to an input on the audio interface.
5. Select the appropriate sequence in SoundCheck and click **Start**.

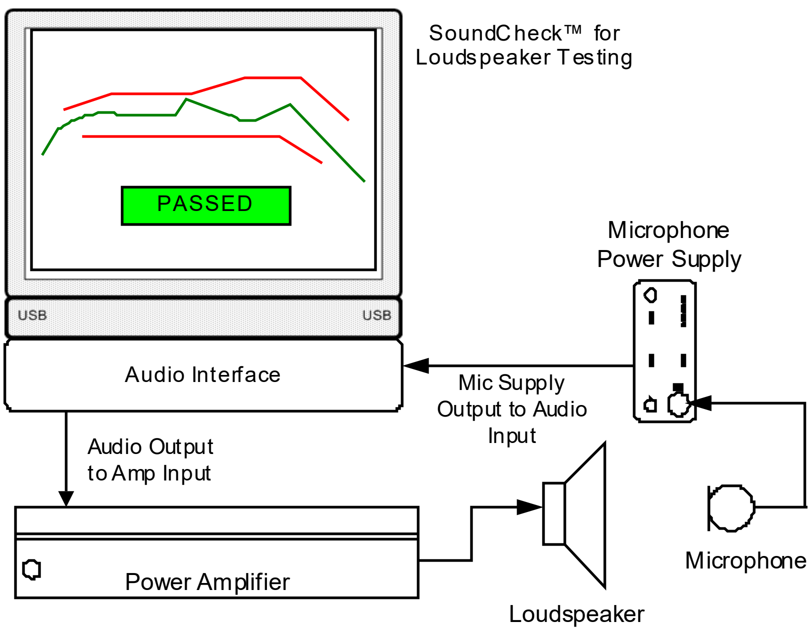


Figure 3-5: Loudspeaker Equipment Setup

Earphone/Headphone Setup

Note: Chose the proper input and output Hardware Channels that correspond to the Signal Paths used in the selected test sequence.

---

1. Connect an output of the audio interface to the input of the power amplifier.
2. Connect the output of the power amplifier to the earphone/headphone under test.
3. Connect the ear simulator preamp cable to the microphone input on the microphone power supply (e.g., SoundConnect).
4. Connect the output of the microphone power supply to an Input on the audio interface.
5. Select the appropriate sequence in SoundCheck and click **Start**.

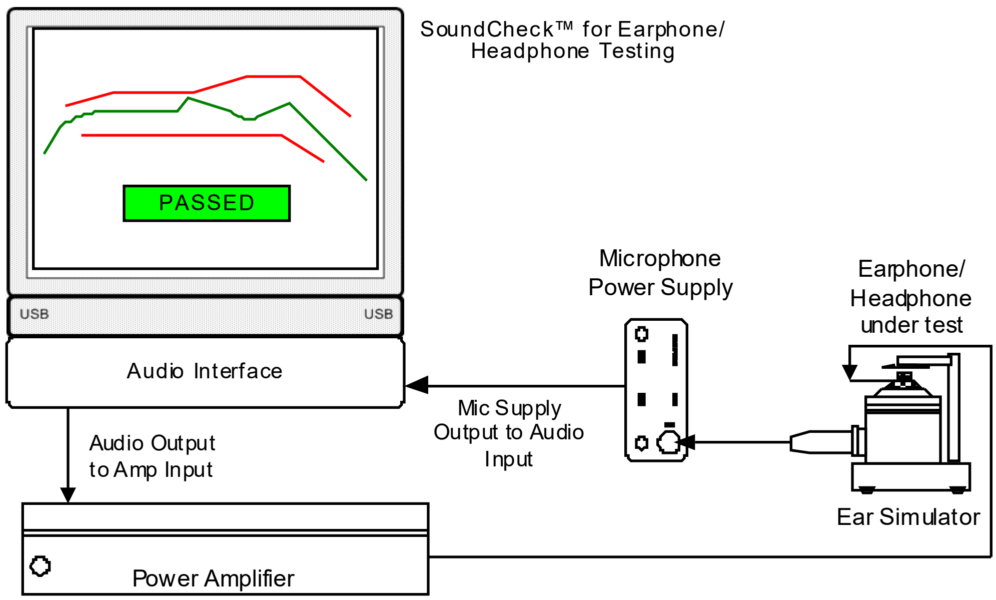


Figure 3-6: Earphone Equipment Setup

Microphone Setup

> ***Note:*** Chose the proper input and output Hardware Channels that correspond to the Signal Paths used in the selected test sequence.

1. Connect an output of the audio interface to the Input of the Power Amplifier.
2. Connect the output of the power amplifier to the mouth simulator.
3. Connect the microphone under test either to the direct input of the microphone power supply (e.g., SoundConnect BNC Input), or if no additional gain is needed, connect directly to the input of the audio interface.
4. If using a microphone power supply, connect the output to an Input on the audio interface.
5. Select the appropriate sequence in SoundCheck and click **Start**.

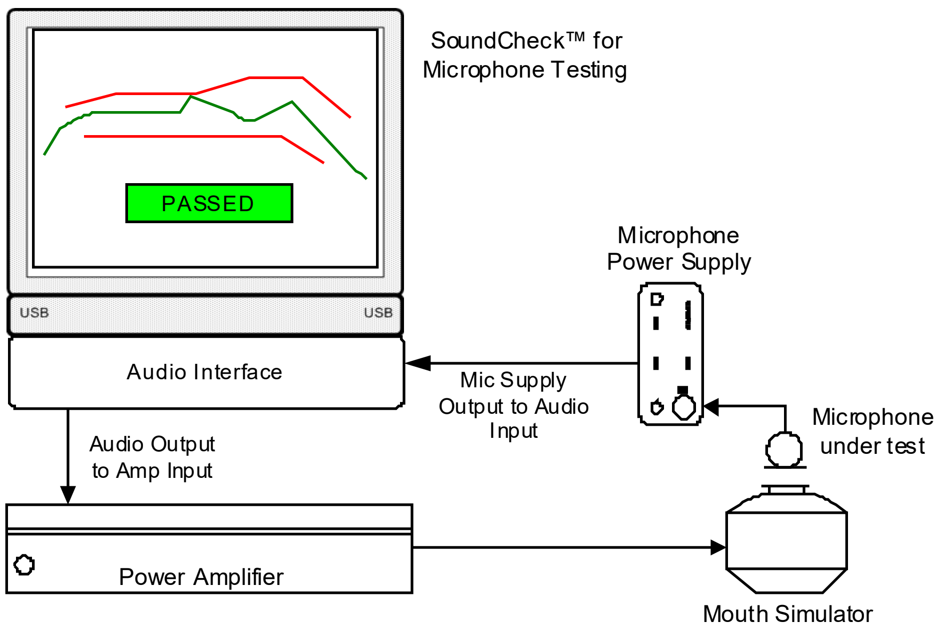


Figure 3-7: Microphone Equipment Setup

Hearing Aid Setup

Note: Chose the proper input and output Hardware Channels that correspond to the Signal Paths used in the selected test sequence.

1. Connect an output of the audio interface to the Input of the power amplifier.
2. Connect the output of the power amplifier to the anechoic chamber.
3. Connect the hearing aid under test to a coupler, sealing the transmitter of the hearing aid towards a calibrated microphone.
4. Connect this mic to the microphone input on the microphone power supply (e.g., SoundConnect).
5. Connect the output of the power supply to an input on the audio interface.
6. Select the appropriate sequence in SoundCheck and click **Start**.

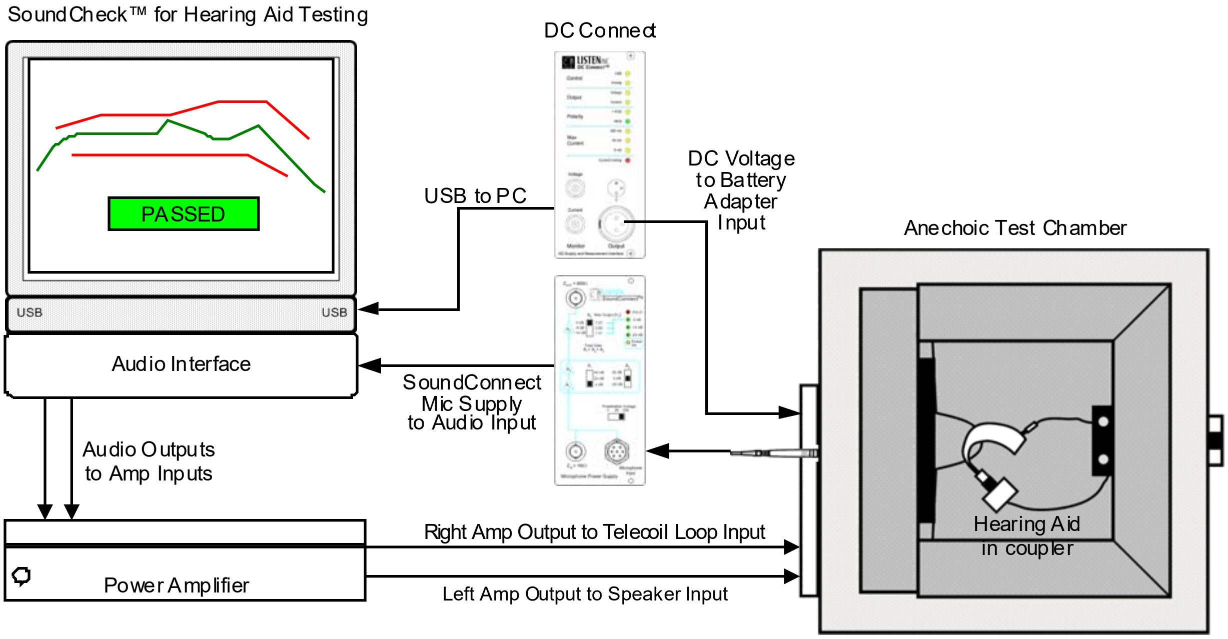


Figure 3-8: Hearing Aid Equipment Setup

Note: DC Connect is an optional hardware item available from Listen, Inc.

Telephone/Cell Phone Setup

Note: Chose the proper input and output Hardware Channels that correspond to the Signal Paths used in the selected test sequence.

1. Connect an output of the audio interface to the input of the power amplifier.
2. Connect the output of the power amplifier to the mouth simulator cable for the head and torso simulator.
3. Connect the microphone (Ear Simulator) to the microphone input of the microphone power supply (e.g., SoundConnect).
4. Connect the output of the microphone power supply to an input on the audio interface.
5. Connect an output of the audio interface to the input of the Telephone interface to send signal to the device in the positioner.
6. Select the appropriate sequence in SoundCheck and click **Start**.

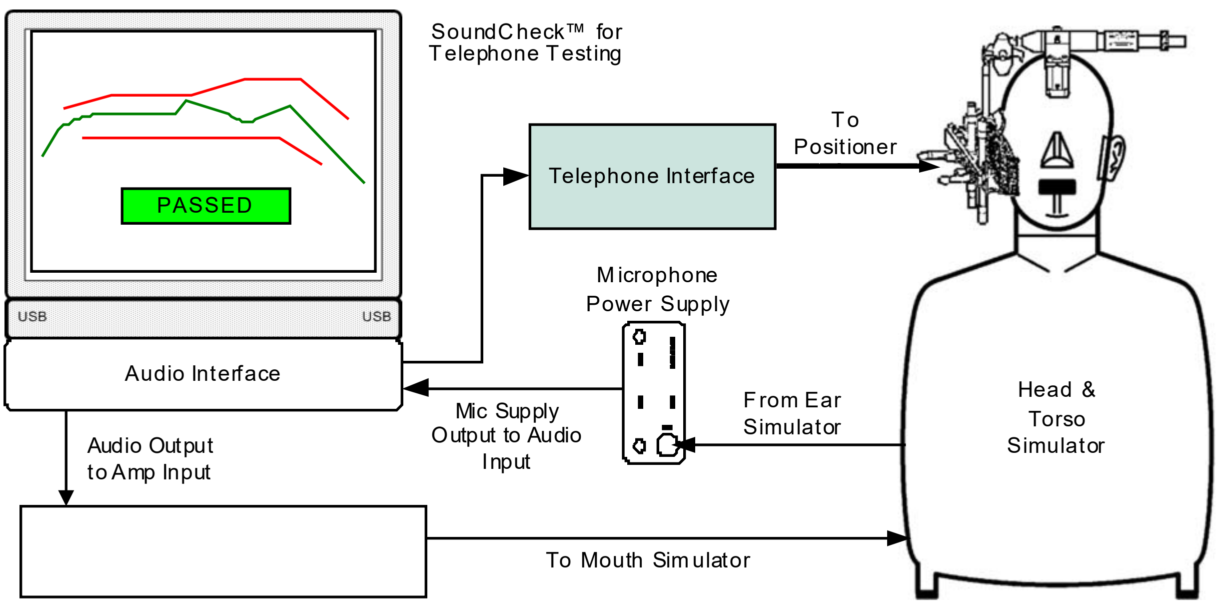


Figure 3-9: Telephone Equipment Setup

Setup Wizard

SoundCheck®, when installed at Listen onto a purchased computer, is configured with the appropriate audio interface driver. If a different audio interface is installed, you will have to configure it manually.

Note: Please refer to ***Minimum Computer Requirements on page 1***.

If only the software is purchased, you should use this section to setup your audio interface for use with SoundCheck.

Note: For complete computer setup recommendations, please refer to ***Windows Setup Recommendations on page 509***.

First Run

The Setup Wizard runs the first time SoundCheck runs. The wizard can also be run by clicking **File** on the SoundCheck Main Screen and selecting **Setup Wizard** (Ctrl+Z).

- Click on the items in the Menu to select a function
- Check box for "Do not show on start up"
- Assists in setting up new hardware, including auto-detection of audio interface
- Allows you to transfer over sequences as well as hardware and calibration settings from previous versions of SoundCheck

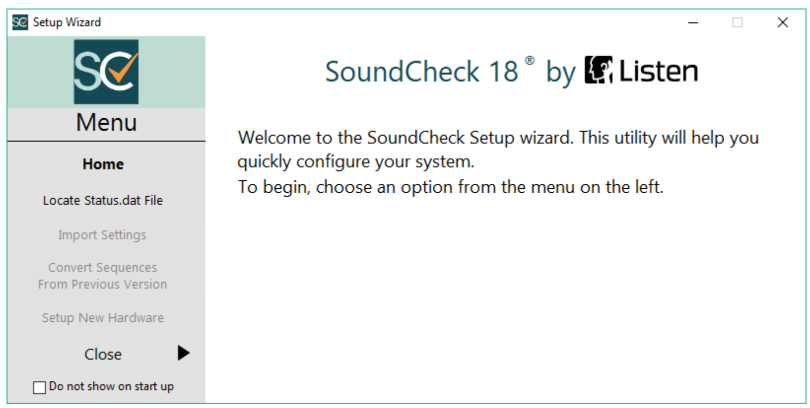


Figure 4-1: Setup Wizard Greeting

Locate Status.dat File

This allows you to navigate to the appropriate status.dat file for your hardware key.

- The *status.dat* file is normally sent by email from Listen, Inc.

Automatically Select Status.dat

If the status.dat path is pointed to a folder containing multiple status.dat files, the software will automatically load the file that corresponds to the hardware key that is currently plugged in.

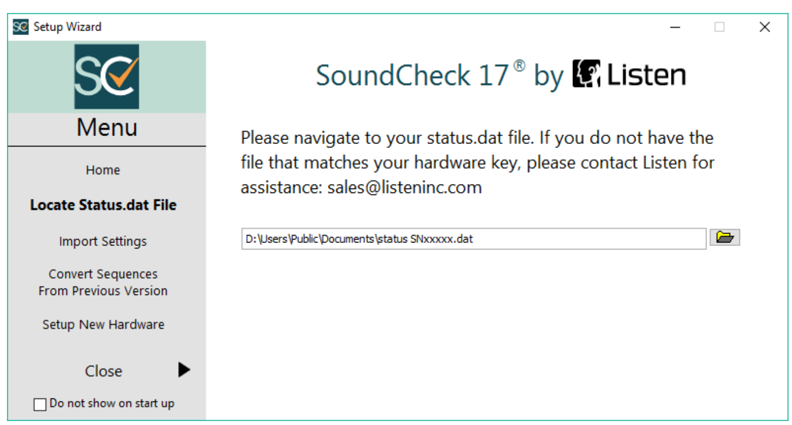


Figure 4-2: Locate Status.dat File

Import Settings

You can import Preferences as well as Hardware and Calibration setups from previous installations of SoundCheck.

- You will be prompted to choose if Signal Paths should be overwritten and then if Calibrated Device Files should be imported
- Select or ignore import of Preferences or Hardware/ Calibration setups

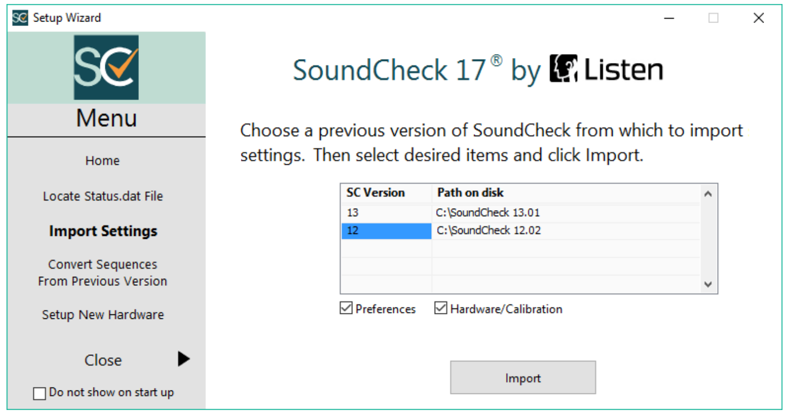


Figure 4-3: Import Settings

Note: ***Calibration .DAT files from devices not in use in System Calibration will not be copied into the new version of SoundCheck. These files will need to be manually imported later.***

Convert Sequences From Previous Version

Right-click to:

- Add a sequence or a directory of sequences to the list
- Remove a sequence from the list
- Clear the list
- Select the Destination Folder for the converted sequences
- Click **Convert**

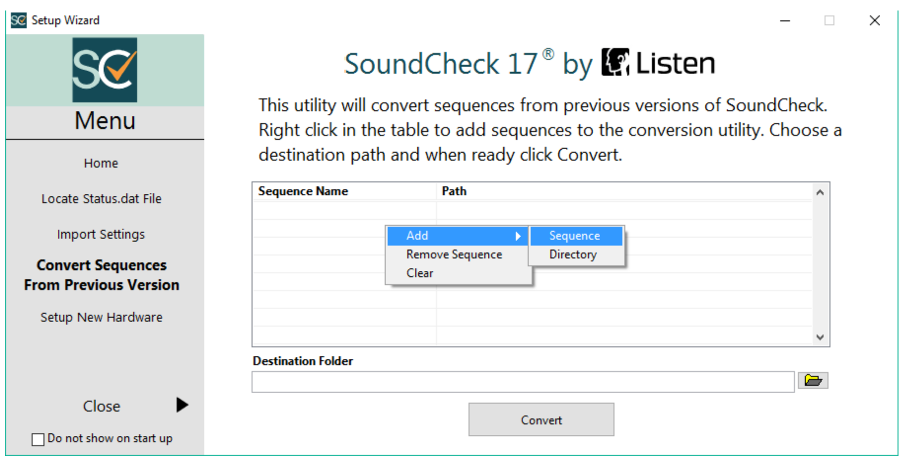


Figure 4-4: Convert Sequences

Setup New Hardware

You are prompted when new Listen hardware is detected.

- "**Automatically Create Signal Paths for Listen Devices**" is enabled by default in Preferences > Launch. The Vp value fields in the Hardware Editor will automatically update for supported Listen Hardware.
- ***See Automatically Create Signal Paths for Listen Devices on page 46***
- Hardware from other manufacturer's is not detected automatically
- HAR files are available for Listen approved hardware and can be imported in the Hardware Editor after the device drivers have been installed. See ***Audio Interface on page 38***.
- Click on the **Arrow button** to open the Hardware Editor or Self Test sequence

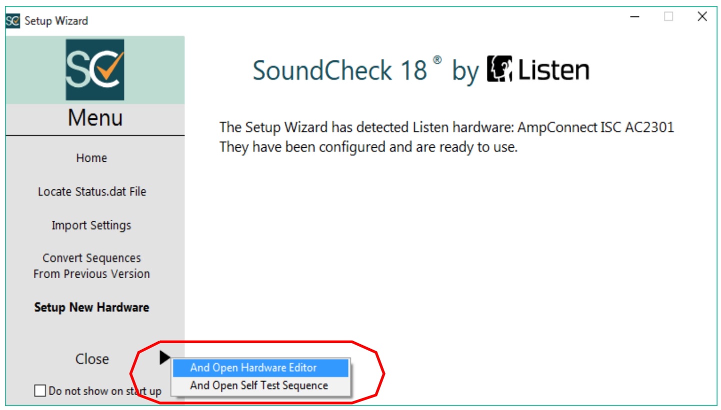


Figure 4-5: Setup New Hardware

Creating Sequences

A sequence tutorial is available in the main SoundCheck manual in the Sequence Editor Chapter.

See ***Creating a New Sequence on page 449.***

Audio Interface

Important! ***Before setting up an audio interface for use with SoundCheck, please refer to*** ***Hardware Compatibility List on page 557. This contains important information regarding approved audio interfaces.***

For manual setup of the audio interface in SoundCheck:

1. Open SoundCheck and select **Hardware** from the **Setup** menu.
2. Click **Import** and browse the list of Hardware Steps in the **C:*SoundCheck 18\Steps\Hardware*** folder
   - Steps are configured for individual audio interfaces and are named by brand and/or model of that audio interface
   - If the audio interface is not listed, the setup for the audio interface will have to be created manually

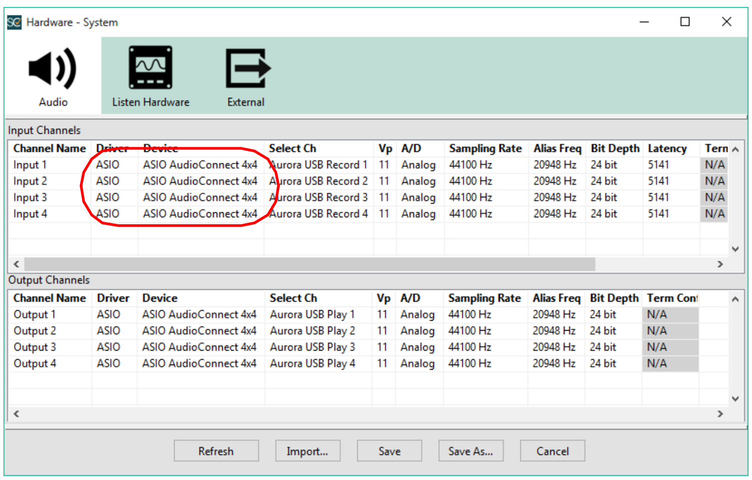


Figure 4-6: Hardware Editor

Note: Please contact Listen for the most recent list of Hardware Configurations, or for help creating a new step.

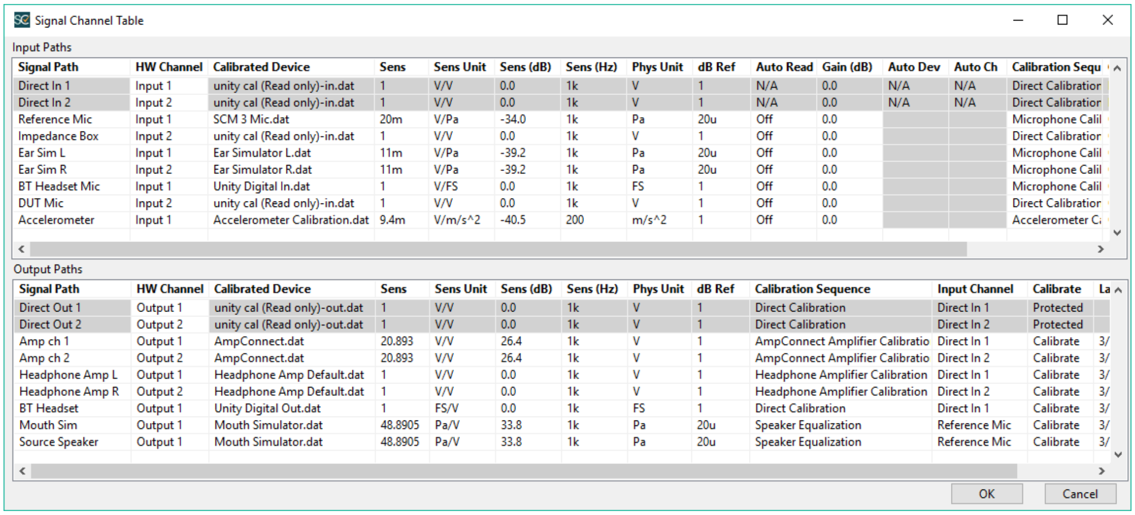
When using a audio interface with Balanced Inputs and Outputs please follow the calibration and wiring guidelines found in ***Balanced Audio Interface Calibration Connections on page 578***.

Note: When using the Digital Audio Labs “CardDeluxe“ in conjunction with Listen’s “SoundConnect“ microphone power supply, the Max In value of the Input Hardware Channel must be multiplied by **1.125**. This is to account for the impedance difference between the SoundConnect and the CardDeluxe.

A Note About Calibration

The Calibration Configuration includes all devices in the signal path including the Audio Interface, Amplifier, and/or Microphone.

The Calibration Editor allows you to create a Signal Path for each device that might be used on the system. The Table View of the editor acts as a database of all calibrated devices.


Signal Channel Table

Input Paths

Signal Path	HW Channel	Calibrated Device	Sens	Sens Unit	Sens (dB)	Sens (Hz)	Phys Unit	dB Ref	Auto Read	Gain (dB)	Auto Dev	Auto Ch	Calibration Sequ
Direct In 1	Input 1	unity cal (Read only)-in.dat	1	V/V	0.0	1k	V	1	N/A	0.0	N/A	N/A	Direct Calibratior
Direct In 2	Input 2	unity cal (Read only)-in.dat	1	V/V	0.0	1k	V	1	N/A	0.0	N/A	N/A	Direct Calibratior
Reference Mic	Input 1	SCM 3 Mic.dat	20m	V/Pa	-34.0	1k	Pa	20u	Off	0.0			Microphone Calil
Impedance Box	Input 2	unity cal (Read only)-in.dat	1	V/V	0.0	1k	V	1	Off	0.0			Direct Calibratior
Ear Sim L	Input 1	Ear Simulator L.dat	11m	V/Pa	-39.2	1k	Pa	20u	Off	0.0			Microphone Calil
Ear Sim R	Input 2	Ear Simulator R.dat	11m	V/Pa	-39.2	1k	Pa	20u	Off	0.0			Microphone Calil
BT Headset Mic	Input 1	Unity Digital In.dat	1	V/FS	0.0	1k	FS	1	Off	0.0			Microphone Calil
DUT Mic	Input 2	unity cal (Read only)-in.dat	1	V/V	0.0	1k	V	1	Off	0.0			Direct Calibratior
Accelerometer	Input 1	Accelerometer Calibration.dat	9.4m	V/m/s^2	-40.5	200	m/s^2	1	Off	0.0			Accelerometer Ca

Output Paths

Signal Path	HW Channel	Calibrated Device	Sens	Sens Unit	Sens (dB)	Sens (Hz)	Phys Unit	dB Ref	Calibration Sequence	Input Channel	Calibrate	La
Direct Out 1	Output 1	unity cal (Read only)-out.dat	1	V/V	0.0	1k	V	1	Direct Calibration	Direct In 1	Protected	
Direct Out 2	Output 2	unity cal (Read only)-out.dat	1	V/V	0.0	1k	V	1	Direct Calibration	Direct In 2	Protected	
Amp ch 1	Output 1	AmpConnect.dat	20.893	V/V	26.4	1k	V	1	AmpConnect Amplifier Calibratio	Direct In 1	Calibrate	3/
Amp ch 2	Output 2	AmpConnect.dat	20.893	V/V	26.4	1k	V	1	AmpConnect Amplifier Calibratio	Direct In 2	Calibrate	3/
Headphone Amp L	Output 1	Headphone Amp Default.dat	1	V/V	0.0	1k	V	1	Headphone Amplifier Calibration	Direct In 1	Calibrate	3/
Headphone Amp R	Output 2	Headphone Amp Default.dat	1	V/V	0.0	1k	V	1	Headphone Amplifier Calibration	Direct In 2	Calibrate	3/
BT Headset	Output 1	Unity Digital Out.dat	1	FS/V	0.0	1k	FS	1	Direct Calibration	Direct In 1	Calibrate	3/
Mouth Sim	Output 1	Mouth Simulator.dat	48.8905	Pa/V	33.8	1k	Pa	20u	Speaker Equalization	Reference Mic	Calibrate	3/
Source Speaker	Output 1	Mouth Simulator.dat	48.8905	Pa/V	33.8	1k	Pa	20u	Speaker Equalization	Reference Mic	Calibrate	3/

OK Cancel

Figure 4-7: Calibration Table

The accuracy of your SoundCheck system depends upon accurate calibration of your input and output devices. Nominal calibration values for many devices typically used are included with SoundCheck. For more accurate measurements, calibration of individual signal paths should be performed for each device. Frequency of calibration of these devices depends upon the stability of the device.

For more information refer to ***Calibration Configuration on page 79***.

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SoundCheck Main Screen

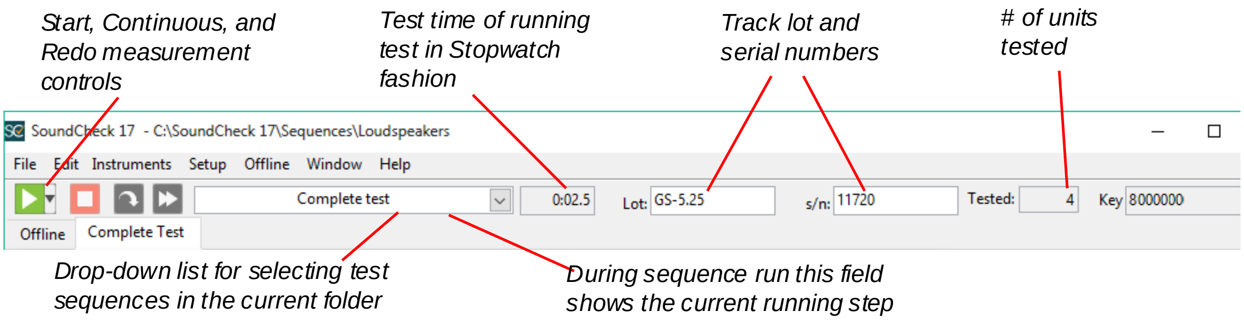


Figure 5-1: SoundCheck Main Screen

Quick Launch Menu

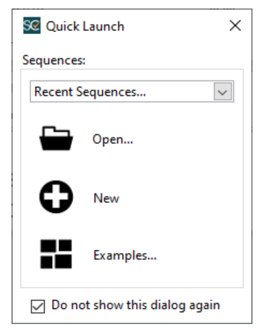


Figure 5-2: Quick Launch Menu

The optional *Quick Launch Menu* provides a simple interface for choosing a sequence to open or starting a new sequence. It also allows easy access to recently used sequences as well as examples.

- **Recent Sequences**: Select a recently used sequence to open
- **Open**: Opens a Windows browser so you select a sequence to open (Same as the File > Open function)
- **New**: This opens the Sequence Editor with a blank sequence.

  See ***Creating a New Sequence on page 449*** for more information.
- **Examples**:

  Opens the **Example Sequence Browser** as shown in ***Figure 5-3***. You can select from any of the default sequences included with SoundCheck. The Sequence Information field provides an explanation of the selected sequence. You can also select View Sequence Document to open the PDF file included with that sequence.

  See ***Default Sequence List on page 615***.

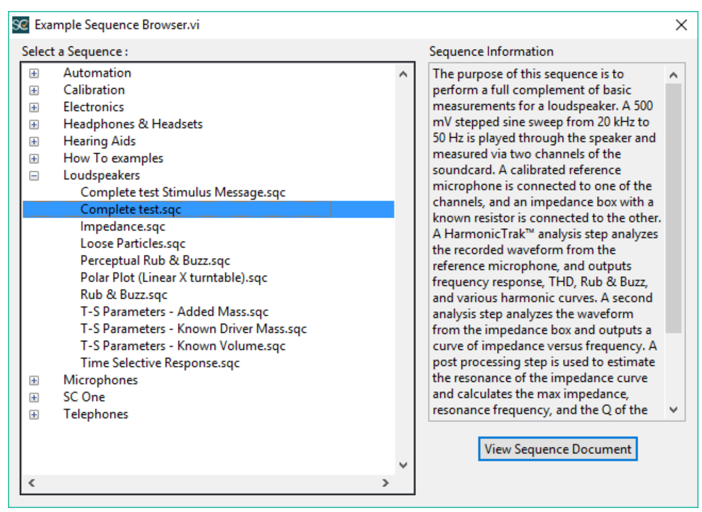
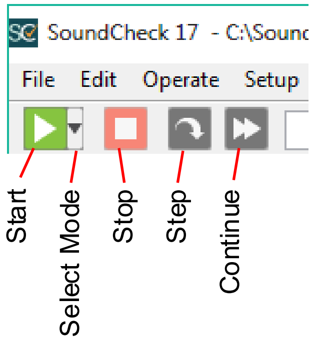


Figure 5-3: Example Sequence Browser

Control Buttons



Start - To start a measurement, Left-click the Green **Start** button. You can also click the **F2 key** on your keyboard, or use an optional foot switch or bar code reader.

Select Mode (down arrow) - allows you to select the following options:

●**Start** - Runs the sequence one time

●**Cont.** (Continuous) - Repeats the sequence until **Stop** is selected

●**Redo** - Runs the same measurement again and overwrites the results of the previous measurement

Stop - Ends the run of the sequence at the current running step

- Click the **Stop** button at any time during the sequence run to halt operation
- You can also hit the **Escape** key on the keyboard to **Stop**

Step- Runs one step of the sequence at a time. Available when Breakpoints are set in a sequence.

See ***Debugging Tools on page 444***.

Continue - Runs the remaining steps in a sequence when:

A previous step is set to **Wait for Confirmation**. Click the **Enter key** to continue.

See ***Configure Step on page 446***.

Breakpoints are set in a sequence. Click the **Enter key** to continue. The sequence will continue to the next Breakpoint or to the end of the sequence. You can also click the **Step** button to run one step at a time.

See ***Debugging Tools on page 444***.

The keyboard shortcuts will also change "**Select Mode**" and the Start Icon will change accordingly:

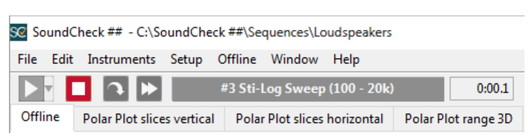
Start = F2, **Continuous =** F3, **Redo** = F4, **Continue** = Enter and **Stop =** Esc

See ***Keyboard Shortcuts on page 591***.

Note: Clicking **Stop** ends the sequence run at the current running step. The sequence does not run to the end.

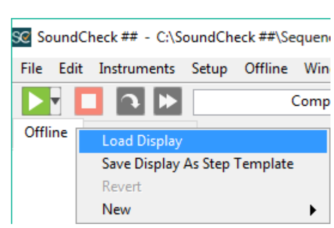
Running Step Display

When a sequence is running, the **Current Running Step** is displayed in the sequence name field.



Offline Tab Display

The Offline Tab is available with or without a sequence loaded. This allows you to open, process, and view data without loading a test sequence. In the Offline Tab, data can be examined or analyzed without affecting the layout of the display steps of the sequence. It minimizes the risk of accidentally editing sequences. This is also useful to customers who view data on a regular basis, but may not be opening a sequence.



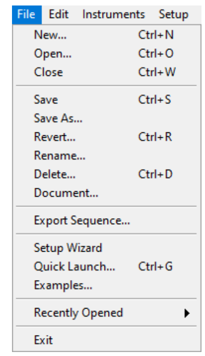
See ***Offline Tab on page 338***.

Drop-down Lists

All SoundCheck functions are divided into individual modules, accessible from drop-down lists.

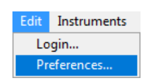
Left-click on a menu heading and then click on the desired selection.

File



- Create a New sequence or Open an existing sequence
- Save changes to a sequence or Save As to save with a new name
- Revert - Discard recent changes and restore the sequence to the last saved version
- Rename - Change the current name of the sequence
- Delete - Erase the current sequence from hard disk
- Document... - Allows you to export a list of the steps of the active sequence along with information regarding the configuration of the steps.
- Export Sequence - See ***Exporting Sequences on page 451***.
  - Mass Export - Multiple sequences can be exported in a single operation. See ***Mass Export on page 52***.
- Setup Wizard - Opens Setup Wizard.
- Quick Launch - Opens Quick Launch.
- Examples - Opens Example Sequence Browser. See ***Example Sequence Browser on page 41***.
- Recently Opened - Shows a list of the last 10 sequence opened

Edit



- Open the **Login** window to change the Access Level or User Name
- **Preferences** allows you to customize and maintain various SoundCheck operations and functions including status.dat file location.

  See ***Preferences on page 45***.

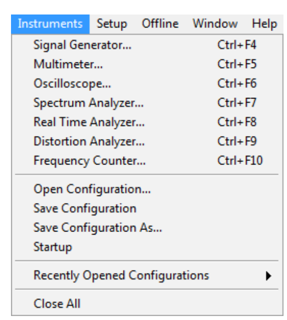
  See ***Folder Paths on page 49***.

Instruments

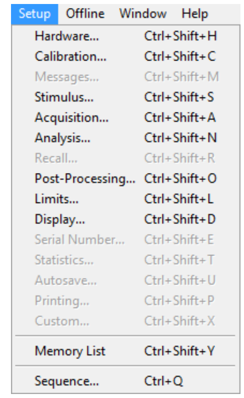
- Select **Instruments** (formerly Operate) from the main screen
- Select single or multiple Virtual Instruments

  See ***Virtual Instruments on page 455***

- Open or Save Virtual Instrument Configuration files (.VIC) Setup
- Startup - Select a Startup Configuration to launch when SoundCheck opens



Setup



- Open System **Hardware** and System **Calibration**

  See ***Page 59*** for Hardware and ***Page 79*** for Calibration
- Open steps that are used in the Active Sequence

  (Grayed out steps are not in use)
- Open the Memory List or Sequence Editor

Offline

The Offline Menu on the SoundCheck Main Screen features steps that can be used to process data without affecting the active sequence.

- The processed data can be saved
- Changes to steps can only be saved by selecting Save As, giving the step a new name and saving it to the appropriate SoundCheck step folder.
- None of the changes to a step will be saved with the active sequence when it is closed and saved

This allows you to try different functions and ideas, modify the display to show data in a different ways and then save the results.

1. In ***Figure 5-4***, **Post-Processing** has been selected from the **Offline** menu.
2. The **Curve Division** step is selected.
3. Curves for Operand A and B are selected from the existing Memory List of the Active Sequence.
4. After pressing the **Apply** button, the result shows up in the **Memory List** as **Protected Data**.

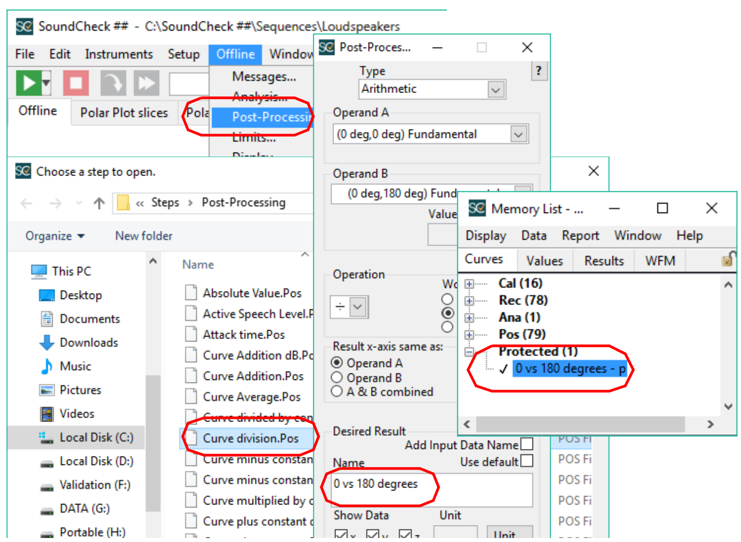
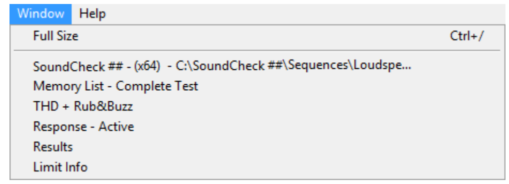


Figure 5-4: Offline Menu Example

Rules

Steps called from the Offline menu must be opened from their default location: **C:\SoundCheck 18\Steps folders**. You should not use steps from a different SoundCheck folder or from a stand-alone folder on your local drive or network. Any steps that you wish to use that are not part of the default SoundCheck installation should be copied to the appropriate folders in the SoundCheck 18 directory you are operating from.

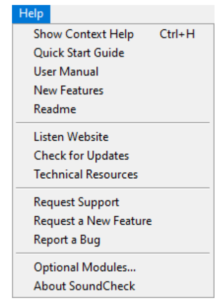
Window



- Promote SoundCheck Main Screen to fill the desktop

  (the Main Screen can be resized manually)
- Shows the Open windows for the current sequence. You can select specific windows that may have become hidden.

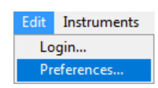
Help



- Turn on Pop Up Context Sensitive Help - Ctrl+H
- Open PDF files for: Quick Start Guide, SoundCheck Instruction Manual, New Features and Readme.txt file
- Open Listen Website
- Check for Updates
- Technical Resources - Opens the Support website with links to: Sequences & Resources, User Documentation, Drivers, Knowledgebase and Register / Update Product
- Request support, request new features or report a bug
- Show the Optional Modules installed on the system
- About SoundCheck - Shows the full version number of SoundCheck which should be included when submitting a support request

To display context sensitive help, choose **Help** from the drop-down list or click on the question mark (**?**) in the upper right corner of an editor. Move your cursor over the control or field of interest and the help text window will show information for that control.

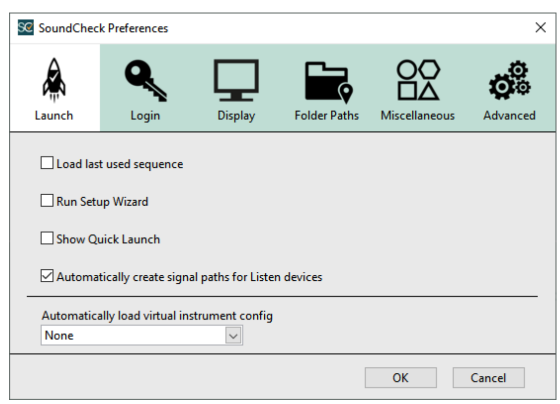
Preferences



The Preferences Menu consolidates program-wide preferences such as folder paths and background wallpaper into a single menu. This menu also includes new options such as toggling sequence documentation and whether or not to automatically load a sequence on startup.

Click **Edit** on the Main Screen and select **Preferences**.

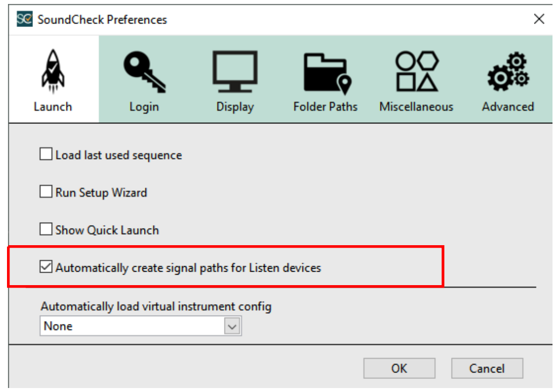
Launch



Select:

- Load last used sequence
- Run Setup Wizard when SoundCheck Starts
- Show Quick Launch Menu when SoundCheck Starts
- Automatically Create Signal Paths for Listen Devices
- Automatically load virtual instrument config

Automatically Create Signal Paths for Listen Devices

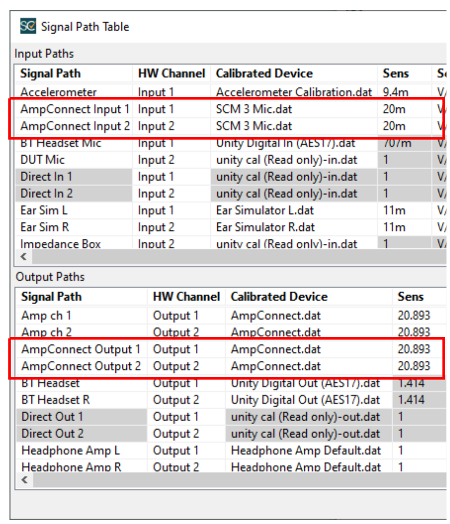


As of SoundCheck 18, this replaces **Automatic Startup Configuration**. This allows SoundCheck to detect and maintain Listen hardware as well as Portland Tool & Die Hardware so that most users will never have to open the Hardware Editor or modify hardware settings.

SoundCheck will scan for available devices on startup and will only automatically configure Listen Hardware devices if **"Automatically Create Signal Paths for Listen Devices"** is selected.

When selected, **"Automatically Create Signal Paths for Listen Devices"** will affect the following:

- It is selected by default
- It affects the **Audio** tab of the Hardware Editor
- It can be turned off in order to prevent automatic changes
- **Calibration Editor Signal Paths** are automatically created for new Listen Hardware devices as shown in ***Figure 5-5***
- After automatic configuration you can further edit the hardware configuration. For example, this allows you to use automatic configuration to detect and configure the AudioConnect 4x4, but then easily change its configuration to 192 kHz sample rate.
- Input and Output Vp values are stored in the firmware of Listen Hardware such as AmpConnect ISC and AudioConnect. These values are automatically loaded in the Hardware Editor when SoundCheck discovers new hardware on startup or when you click **Refresh**. Theses values are fixed and should not be edited.
- It will not overwrite the settings of an existing audio channel where those settings are allowed values. Only Vp and Sampling Rate are checked for validity. The same is true for **Refresh** in the Hardware Editor.
  - If you are using audio interfaces other than those made by Listen, **"Automatically Create Signal Paths for Listen Devices"** will not affect your “already configured channels” as long as the device is connected.
- For Listen Hardware, new channels will be added in the Hardware Editor “**After**” the channel numbers of existing hardware



Signal Path	HW Channel	Calibrated Device	Sens
Accelerometer	Input 1	Accelerometer Calibration.dat	9.4m
AmpConnect Input 1	Input 1	SCM 3 Mic.dat	20m
AmpConnect Input 2	Input 2	SCM 3 Mic.dat	20m
BT Headset Mic	Input 1	Unity Digital In (AES17).dat	707m
DUT Mic	Input 2	unity cal (Read only)-in.dat	1
Direct In 1	Input 1	unity cal (Read only)-in.dat	1
Direct In 2	Input 2	unity cal (Read only)-in.dat	1
Ear Sim L	Input 1	Ear Simulator L.dat	11m
Ear Sim R	Input 2	Ear Simulator R.dat	11m
Impedance Box	Input 2	unity cal (Read only)-in.dat	1

Signal Path	HW Channel	Calibrated Device	Sens
Amp ch 1	Output 1	AmpConnect.dat	20.893
Amp ch 2	Output 2	AmpConnect.dat	20.893
AmpConnect Output 1	Output 1	AmpConnect.dat	20.893
AmpConnect Output 2	Output 2	AmpConnect.dat	20.893
BT Headset	Output 1	Unity Digital Out (AES17).dat	1.414
BT Headset R	Output 2	Unity Digital Out (AES17).dat	1.414
Direct Out 1	Output 1	unity cal (Read only)-out.dat	1
Direct Out 2	Output 2	unity cal (Read only)-out.dat	1
Headphone Amp L	Output 1	Headphone Amp Default.dat	1
Headphone Amp R	Output 2	Headphone Amp Default.dat	1

Figure 5-5: Auto Signal Paths

Note: When **Refresh** is clicked in the Hardware Editor, SoundCheck scans for hardware updating the Audio and Listen Hardware Tabs but will not create signal paths.

Automatically load virtual instrument config

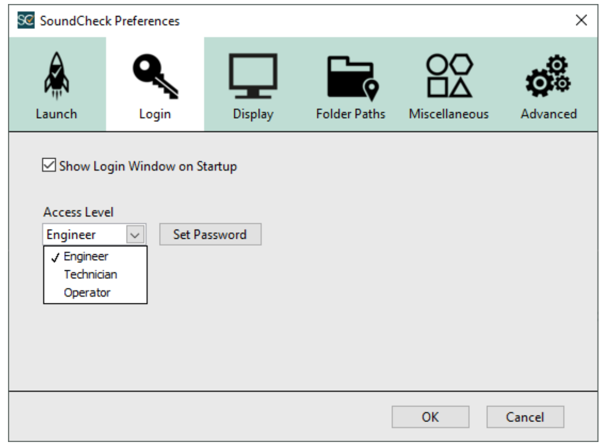
Select a saved virtual instrument configuration to load at startup.

See ***Virtual Instrument Configuration on page 458***.

Login

- Show or Hide Login Window on Startup
- Select Access Level: Engineer, Technician or Operator
- Set the Password for each Access Level

See ***Login on page 57***.



Display

- **Display Image** - Select to use an image file as a Main Screen background
- **Background Color** - Sets the color of the Main Screen background
- Use the Folder Icon to browse for a Wallpaper Image file (.BMP, .JPG or .JPEG)

Use the drop-down list to position the image.

- **Center** - The graphic is not resized and is placed in the center of the Main Screen background
- **Tile** - The graphic is not resized and is duplicated, filling the Main Screen background
- **Stretch** - The graphic is stretched to fill the Main Screen background

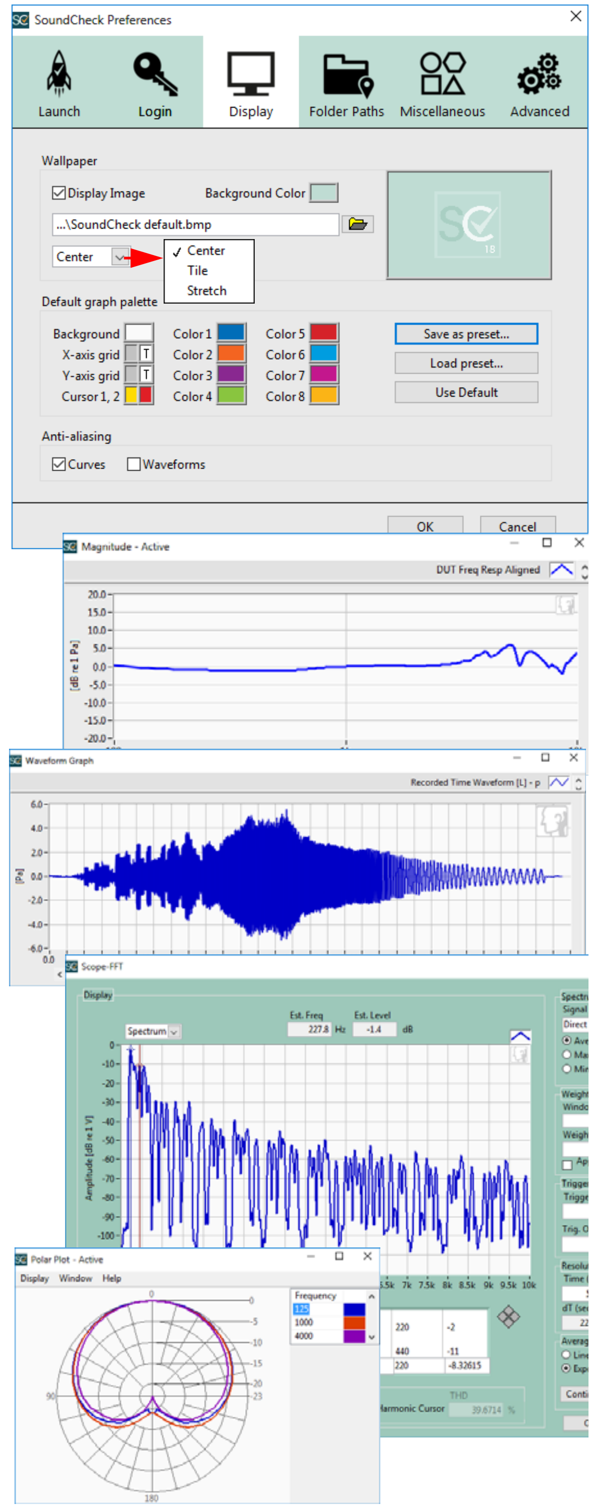


Figure 5-6: Default Graph Palette

Default graph palette

- Settings apply to any new display created
- White background is the default for new displays as of SoundCheck 17

Background - Sets the background color for all new instances of the following:

- Displays
- Virtual Instruments
- SoundMap (except default Intensity Plot)
- Settings do not change pre-existing displays

X-axis grid/Y-axis grid - The left box sets the major grid line color and the right box sets the minor grid line color. By default the minor grid lines are set to transparent (T).

Colors 1 - 8

Color 1 - Used for first curve in Display Graph windows as well as Limits Editor windows. After the first color is used, colors are sequentially selected from the list as new curves are added to a display graph.

Save as Preset - Allows you to save the current settings in a preset file (*.palette). This allows you to have different color sets for different applications.

Load Preset - Open a previously created preset file

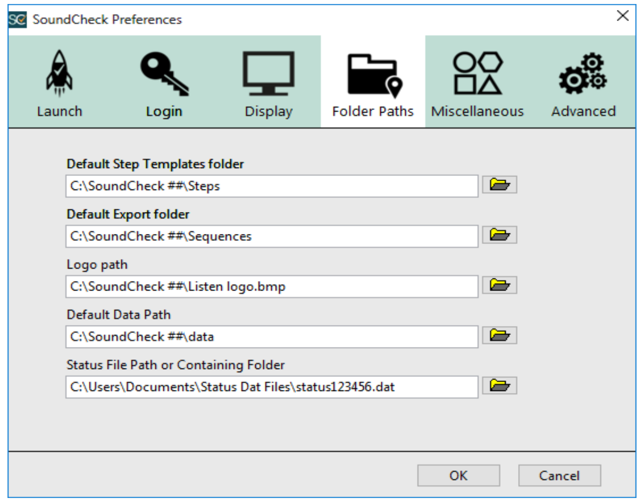
Use Default - Select to revert to the SoundCheck 18 default color set

Anti-Aliasing

This is a visual smoothing option for graphed data. Older or lower performance computers may experience a slow down when updating displays with a large amount of data. Anti-Aliasing can be shut off to speed up display rendering. Anti-Aliasing Waveforms is off by default.

Folder Paths

The **Folder Paths** dialog window allows you to customize where files are located as well as file selection.



- **Default Step Templates Folder** - Location of Step Templates
- **Import Export folder** - Default directory for exporting sequences
- **Logo path** - Logo file used for printing
- **Default Data Path** - Default path used in Autosave and Recall Steps
- **Status File Path or Containing Folder** - Allows you to select a specific status.dat file or set the location for several status.dat files.

  If the status.dat path is directed to a folder containing multiple status.dat files, with the Key ID in the file name, the software will automatically load the file that corresponds to the hardware key that is currently plugged in.

  This makes it easier when setting up multiple SoundCheck systems which have a common path for status.dat files or using status.dat files with different functionality for a single hardware key.

1. Click **Browse** and select a status.dat file

   or click on **Current Folder** to allow SoundCheck to automatically select.
2. Click **OK** to continue.
3. **OK** to close the Folders path dialog

- SoundCheck will automatically switch to using the new status.dat file
- SoundCheck does not have to restart once the new Status file has been selected

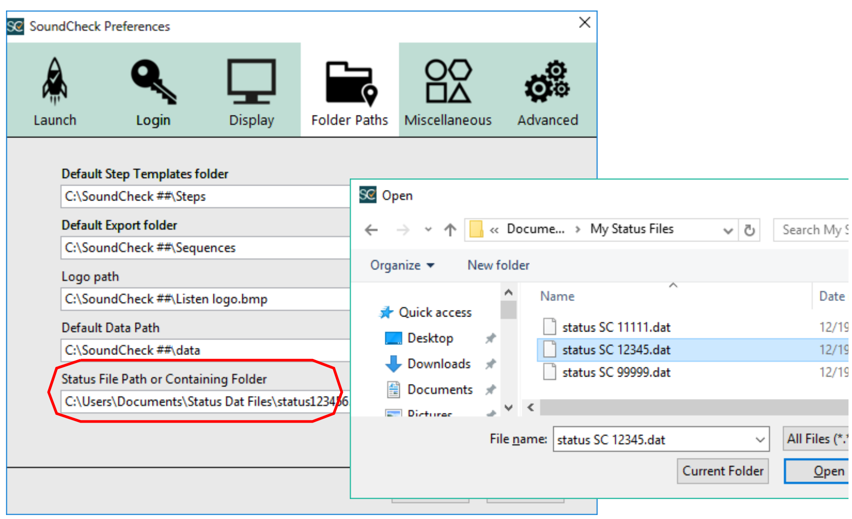


Figure 5-8: Status File Path

The status.dat file must have the Key ID in the file name; e.g., "status SC 1111.dat". This allows for use of multiple hardware keys on one system.

You can also switch between SoundCheck full version and SoundCheck ONE, if you have a SoundCheck ONE status.dat file. In this case you will need to select the specific status.dat file instead of selecting the folder.

Wrong Status.dat Warning

If an invalid status.dat file is selected, the SoundCheck wall paper will change to indicate that “Data is Randomized”. There is also a message in the upper right corner of the SoundCheck Main Screen indicating the condition of the status.dat file. See *Figure 5-9*.

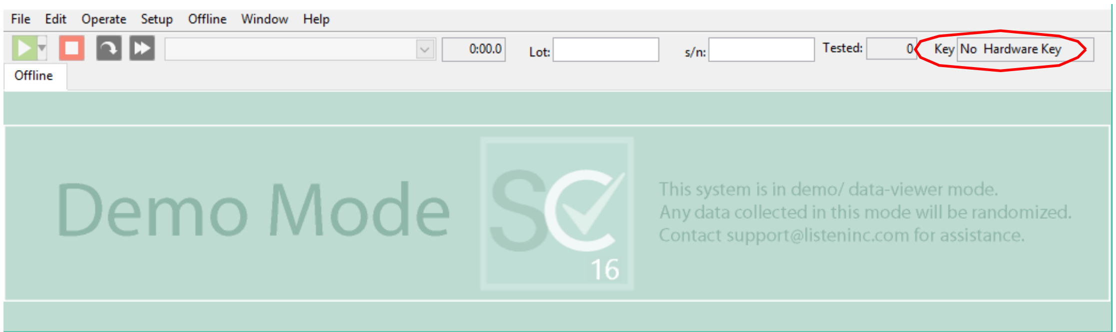


Figure 5-9: Data Randomized Warning

Miscellaneous

- Show Sequence Documentation

  This allows you to globally turn off Show Sequence Documentation in all sequences.

- Show ‘Protect Data’ option when switching sequences

  Turns off the warning: “This action will remove any unused pre-run curves and unprotected measured data from the Memory List.”

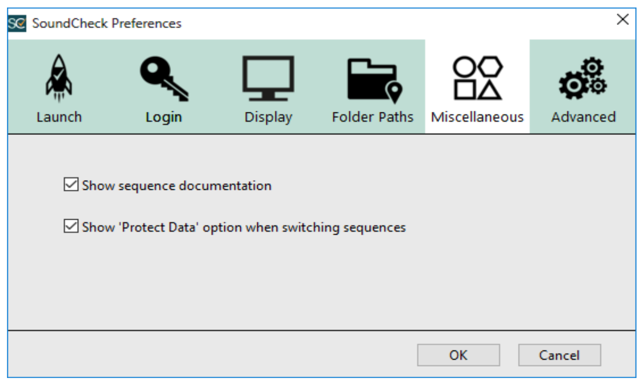


Figure 5-10: Miscellaneous Tab

Advanced

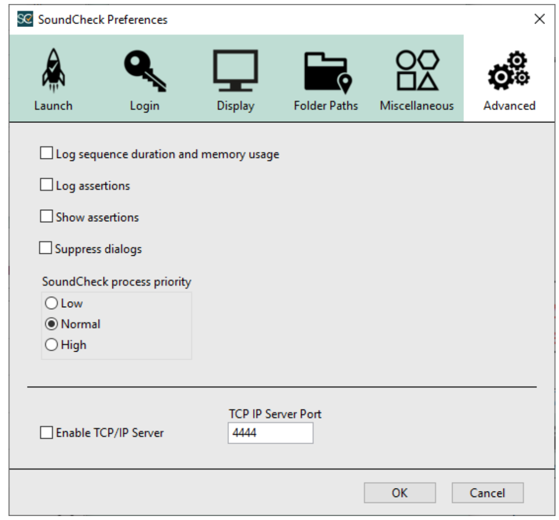


Figure 5-11: Advanced Tab

- **Log sequence duration and memory usage**:

  Produces a log file in the SoundCheck folder containing the length of time the sequence ran and the memory consumption of each sequence run

  C:\SoundCheck 18.1\Time Mem Log.txt

- **Log assertions**:

  Creates log of failed programmer sanity checks (programmer errors)

- **Show assertions**:

  Show failed programmer sanity checks (programmer errors) on screen

- **Suppress dialogs**:

  When controlling SoundCheck from another application, select **Suppress dialogs** to allow fully unattended operation. This will prevent SoundCheck generated pop-up dialogs from halting or pausing sequence operation. This should only be used when controlling SoundCheck from another source since it will hide critical notices and error messages normally used by SoundCheck to inform the user that some action is required.

Process Priority

Raising the Process Priority level helps to minimize fluctuation in sequence execution time. The default value is set to **Normal**. When the priority is set to "**Realtime**", SoundCheck takes priority over all other applications. We do not recommend using "**Realtime**" as it can dramatically change the performance of the system. The appropriate values are listed below.

SCProcessPriority=

- Low
- BelowNormal
- Normal
- AboveNormal
- High
- Realtime

Important! ***The options above are for troubleshooting and should only be used if directed to by Listen Support.***

Enable TCP/IP Server

Check to allow TCP/IP Server (Automatically updates SoundCheck **18**.ini file with True/False status and port number)

TCP IP Server Port: The Port number of this SoundCheck system

See ***Controlling SoundCheck with TCP/IP on page 491*** for more information.

Mass Export

Multiple sequences can be exported in a single operation. This saves time when upgrading from one version of SoundCheck to another and when transferring your sequences from one system to another.

See ***Exporting Sequences on page 451***.

- Click **File** on the SoundCheck Main Screen
- Open the appropriate sequence folder
- Select the sequences you want to export as shown in ***Figure 5-12***

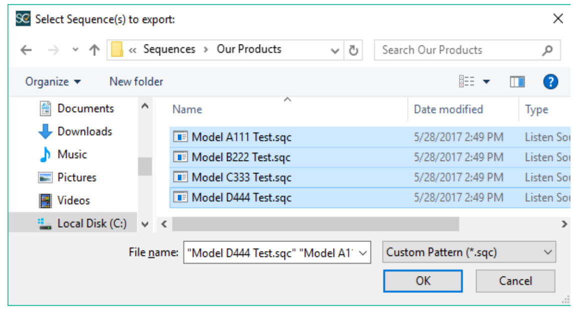
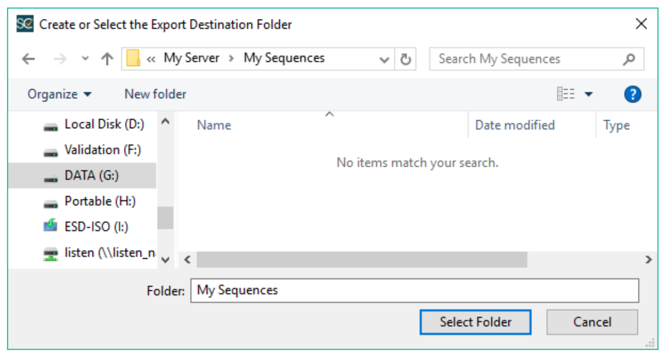
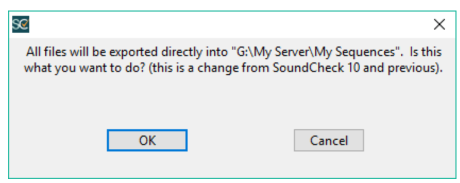


Figure 5-12: Select Sequences

- Navigate to and open the export destination folder (or create a new one)
- Click **Current Folder**. We do not recommend exporting directly to your desktop, but a folder on your desktop is OK.



- You will be prompted to confirm the export destination
- Click **OK** to export



SoundCheck 18.ini File

The **SoundCheck 18.ini** file is used to store various settings for SoundCheck including the settings made in the preferences menu.

For example, when changing sequences, you are prompted to preserve unprotected data in the *Memory List* as shown in ***Figure 5-13***. When "**Don't ask me again**" is selected, SoundCheck will use the selection, Discard or Protect, as the default action each time the sequence is changed. The "**Keep Unprotected Data**" prompt will no longer appear. You can review this setting in the ***SoundCheck*** **18*.ini*** file, found in the root of the SoundCheck folder.

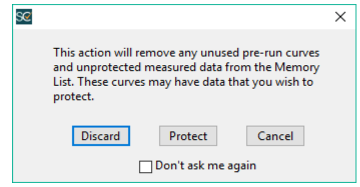
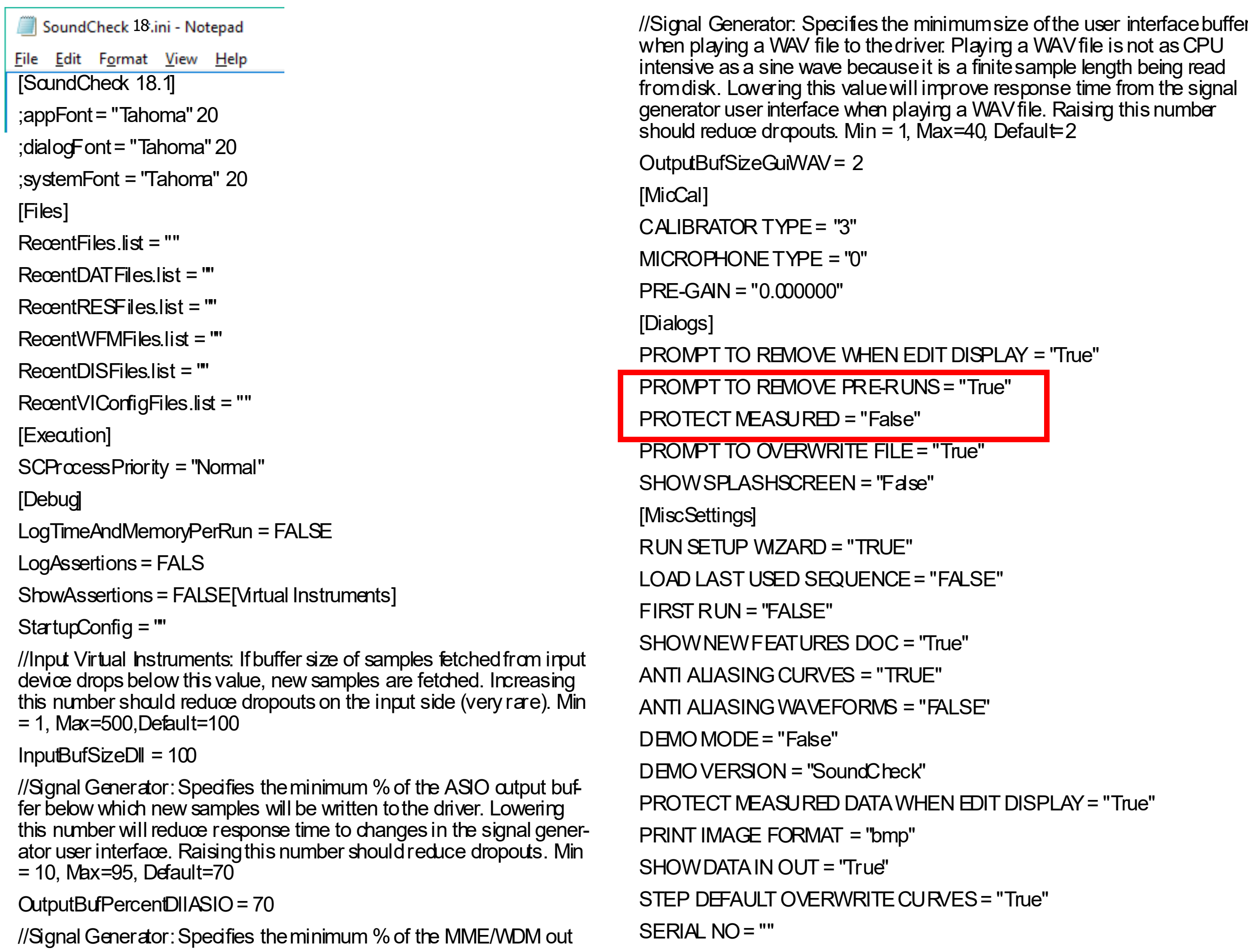


Figure 5-13: Prompt to Keep Unprotected Data



SoundCheck 18.ini - Notepad
File Edit Format View Help

```
[SoundCheck 18.1]
;appFont = "Tahoma" 20
;dialogFont = "Tahoma" 20
;systemFont = "Tahoma" 20
[Files]
RecentFiles.list = ""
RecentDATFiles.list = ""
RecentRESFiles.list = ""
RecentWFMFiles.list = ""
RecentDISFiles.list = ""
RecentVIConfigFiles.list = ""
[Execution]
SCProcessPriority = "Normal"
[Debug]
LogTimeAndMemoryPerRun = FALSE
LogAssertions = FALS
ShowAssertions = FALSE[Virtual Instruments]
StartupConfig = ""
//Input Virtual Instruments: If buffer size of samples fetched from input device drops below this value, new samples are fetched. Increasing this number should reduce dropouts on the input side (very rare). Min = 1, Max=500,Default=100
InputBufSizeDll = 100
//Signal Generator: Specifies the minimum % of the ASIO output buffer below which new samples will be written to the driver. Lowering this number will reduce response time to changes in the signal generator user interface. Raising this number should reduce dropouts. Min = 10, Max=95, Default=70
OutputBufPercentDllASIO = 70
//Signal Generator: Specifies the minimum % of the MME/WDM out
//Signal Generator: Specifies the minimum size of the user interface buffer when playing a WAV file to the driver. Playing a WAV file is not as CPU intensive as a sine wave because it is a finite sample length being read from disk. Lowering this value will improve response time from the signal generator user interface when playing a WAV file. Raising this number should reduce dropouts. Min = 1, Max=40, Default=2
OutputBufSizeGuiWAV = 2
[MicCal]
CALIBRATOR TYPE = "3"
MICROPHONE TYPE = "0"
PRE-GAIN = "0.000000"
[Dialogs]
PROMPT TO REMOVE WHEN EDIT DISPLAY = "True"
PROMPT TO REMOVE PRE-RUNS = "True"
PROTECT MEASURED = "False"
PROMPT TO OVERWRITE FILE = "True"
SHOW SPLASHSCREEN = "False"
[MiscSettings]
RUN SETUP WIZARD = "TRUE"
LOAD LAST USED SEQUENCE = "FALSE"
FIRST RUN = "FALSE"
SHOW NEW FEATURES DOC = "True"
ANTI ALIASING CURVES = "TRUE"
ANTI ALIASING WAVEFORMS = "FALSE"
DEMO MODE = "False"
DEMO VERSION = "SoundCheck"
PROTECT MEASURED DATA WHEN EDIT DISPLAY = "True"
PRINT IMAGE FORMAT = "bmp"
SHOW DATA IN OUT = "True"
STEP DEFAULT OVERWRITE CURVES = "True"
SERIAL NO = ""
```

Figure 5-14: Example of *SoundCheck* 18*.ini*

In ***Figure 5-14***, you can see two entries in the ***SoundCheck*** **18*.ini*** file labeled ***Prompt to remove pre-runs*** and ***Protect Measured***. When ***Prompt to remove pre-runs*** is set to **False**, the dialog is disabled. SoundCheck is then using the ***Protect Measured*** field to determine whether measured curves are protected when sequences are changed. When ***Protect Measured*** is set to **True**, the *Memory List* will preserve all measured curves from one sequence to the next. When it is set to **False**, all unprotected data is discarded when the sequence is changed. See ***Sequence Editor on page 435*** for more information on changing sequences.

Page intentionally left blank

Controls and Details

SI Units

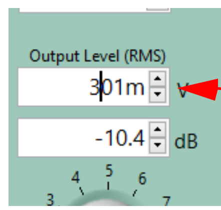
SoundCheck uses *SI Units* throughout the system. It is important to note that values entered, such as 0.1 Volts, will change to 100 m when *SI Units* are selected in the *Preferences* for a Virtual Instrument. Values can also be entered directly with *SI Units* by typing 150 m. It is then important to note the following table of abbreviations for *SI Units* that should be used in SoundCheck.

Symbol	Name	Factor	Symbol	Name	Factor
m	milli	10^{-3}	k	kilo	10^{3}
u	micro	10^{-6}	M	mega	10^{6}
n	nano	10^{-9}	G	giga	10^{9}
p	pico	10^{-12}	T	tera	10^{12}
f	femto	10^{-15}	P	peta	10^{15}
a	atto	10^{-18}	E	exa	10^{18}
z	zepto	10^{-21}	Z	zeta	10^{21}
y	yocto	10^{-24}	Y	yotta	10^{24}

Figure 6-1: SI Units

Numeric Fields

Highlight the value in a numeric field by dragging the mouse cursor over the number while holding down the left-button, or by repeatedly pressing the **Tab** button on the keyboard until you find the correct entry field. You can enter the correct value by highlighting the value in the numeric field and entering the correct number using the keyboard, or by using the left-button of your mouse to click on the **up/down arrow keys** next to the numeric field to increase/decrease the present value.



With the cursor placed after the 3, click the Up Arrow key to raise the value to 401 mV

Another method is to place the blinking cursor to the right of the value you want to increase or decrease. In this example, the cursor is placed in the Voltage field. Using the up/down arrows on the keyboard allows you to change the value in 1, 0.1 or 0.01 increments, depending on which digit the cursor is placed next to. By using the Page Up/Down keys you can change the output level in 1 dB increments. In this example, the up arrow will increment the level by 1.00 V. Pressing the up arrow once increases the level to 1.01 V.

By putting the cursor in the **Frequency** field you can use the **Page Up/Down** keys to change the frequency in R80 or 24th octave steps.

Right-click Functions

Many of SoundCheck's settings can be found under **Right-click** functions. Refer to the following list for some of the main Right-click functions.

Convert Sequences From Previous Version on page 36

Right-click Functions on page 65

Right-click Function on page 89

Right-click Functions on page 115

Virtual Instruments on page 149

Right-click - Memory List on page 323

Right-click Graph on page 340

Resolution

In the Signal Generator click **Preferences** and then **Resolution** to change the settings as shown in ***Figure 6-2***.

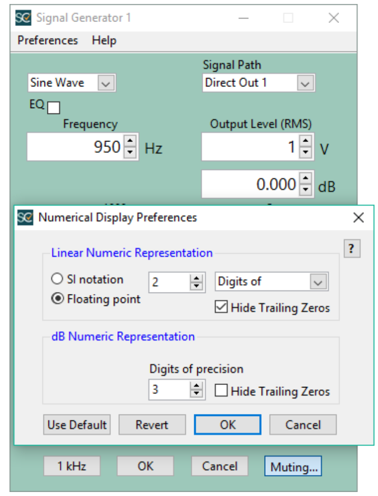
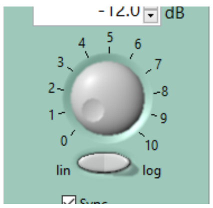


Figure 6-2: Preferences - Signal Generator

Knobs



Instead of entering the desired value with the keyboard, you can also dial in the value with the virtual knob located below the numeric field. Simply place the cursor over the knob, hold down the left mouse button and rotate the knob to change the level.

Graphs and Cursors

The cursor can be moved by Left-click-Hold on the **cursor marker (+)** and dragging it to the desired point on the measurement curve. The cursor will snap to the closest curve you drag to.

The XY coordinates of a cursor are displayed next to the cursor on the display. The XY coordinate box can be moved so that it does not cover the graph line. See ***Cursors on page 340***.

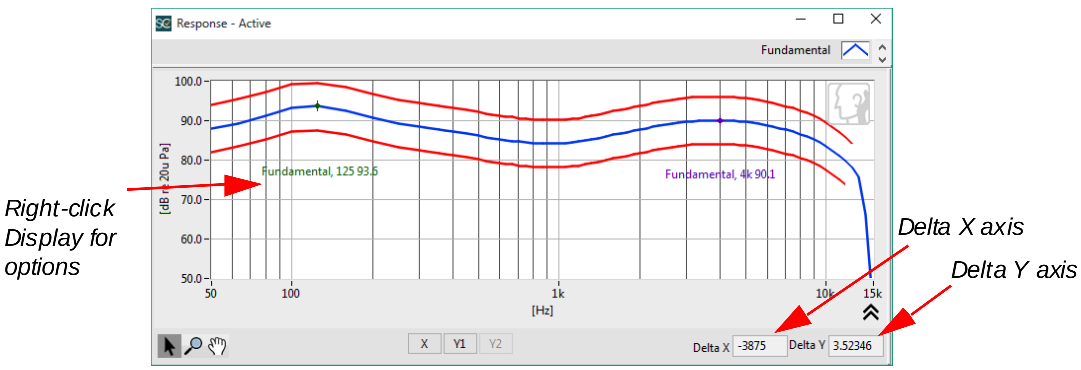


Figure 6-3: Frequency Response Graph

Modifying Graph Display

Right-click a Display and select **Preferences** to show the Graph Preferences editor.

As of SoundCheck 17, new display window backgrounds are white. Displays in existing sequences are not affected.

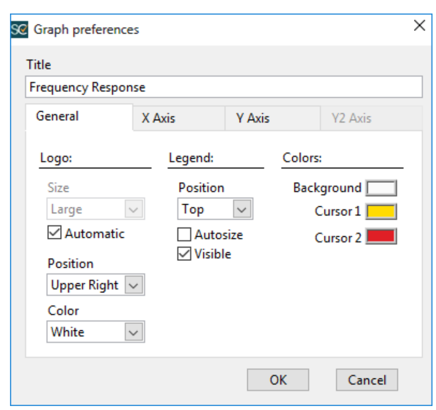


Figure 6-4: Graph Preferences

Cursor Color and Grid Color

Cursor color is controlled by through the Graph Preferences window.

See ***Display Editing on page 339***.

Login

To view and change the system's login settings, select **Login** from the **Edit** drop-down list on the SoundCheck® Main Screen. The *Login* screen also appears when first starting the SoundCheck program. *Login* allows the SoundCheck administrator to prevent use of certain test system functions by other users. For instance, unauthorized users can be blocked from editing Calibration or Sequences with password protection.

Access Level

There are three access levels: Engineer, Technician, and Operator.

- **Engineer** - The Engineer level is the highest access level, and therefore all functionality is accessible
- **Technician** - The Technician level allows access to Calibration procedures, and may measure, print and save data
- **Operator** - The Operator level can only measure and print

Figure 7-1: Login

User Name

The User Name entered in **Login** can be stored with data, results, and included with printouts. It is useful for keeping track of who performed the measurement. The user name is typed in at the login prompt or can be scanned in with a barcode reader. Once the user name has been entered SoundCheck will remember the name and use it for all future sessions. The name is remembered when SoundCheck is closed and re-opened. If a new user name needs to be entered, simply open the *Login* screen from the **Edit** menu in SoundCheck and enter a new operator name, or enter a new name in the *Login* screen upon opening SoundCheck.

Password

The default passwords are not specified (blank) when SoundCheck is shipped. You must click **Setup** to create passwords. See ***Login Setup on page 58.***

Passwords are case sensitive, so be careful with capital letters. Click **Setup** to open the *Login Setup* dialog in order to create new passwords. (Always keep a written copy of your passwords and keep them in a secure place.)

Login Setup

The *Login Setup* dialog allows you to change the login password or bypass the user login.

This can be changed at any time by selecting **Edit** on the SoundCheck Main Screen and then selecting **Login.**

Bypasses Login Screen

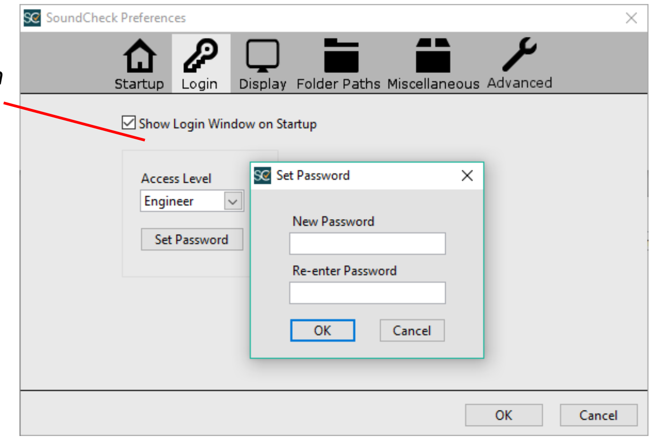


Figure 7-2: Login Setup

- Select **Edit** from the SoundCheck Main Screen and then select **Login** to access the Login Screen

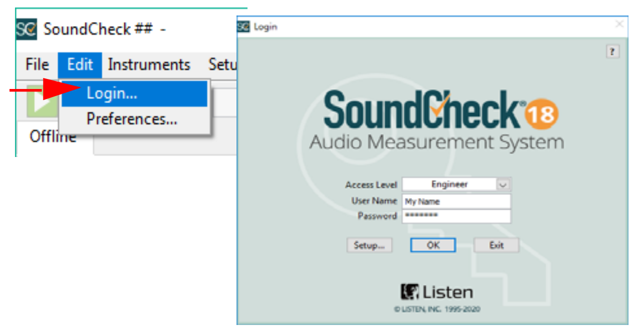


Figure 7-3: Login Screen

Change a Password

- Select **Edit** from the Main Screen
- Select **Login**
- Select Access Level “Engineer” and enter the password
- **Setup**
- In the Login **Edit** window select the desired Access Level and enter the new password
- Click **OK** to close the windows

Hardware Configuration

Hardware - System.Har

This serves as a database for all of the hardware that the specific SoundCheck system uses.

To edit the Hardware Configuration, click **Setup** on the SoundCheck Main Screen and then select **Hardware** as shown in ***Figure 8-1*** (shortcut **Ctrl+Shift+H).**

> ***Note:*** As of SoundCheck 11, Hardware is a “System Level” configuration. It is unique to a specific SoundCheck system and is used by all sequences. Settings from earlier step versions can be imported in the Hardware Configuration Editor.

> ***Note:*** Hardware (.HAR) Steps created with SoundCheck 16 and later are not backward compatible with previous versions of SoundCheck.

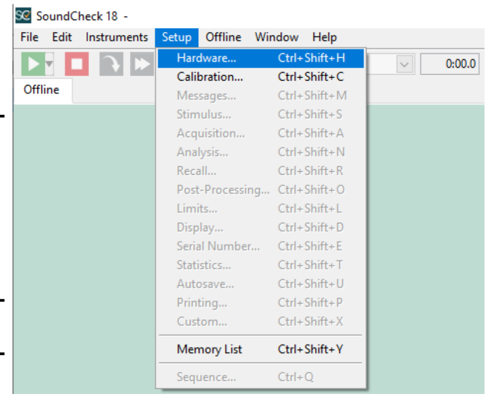


Figure 8-1: Setup - Hardware

Features

Listen hardware is configured automatically but can be adjusted, making setup faster and more flexible. This allows you to more easily add 3rd party hardware to use along with Listen Hardware.

- Easier integration of Listen and 3rd party hardware
- 3rd party hardware settings are preserved (not overwritten) when adding Listen Hardware
- Easier integration of Listen and 3rd party hardware

  See ***Listen Hardware Page on page 70***.

Hardware Compatibility

SoundCheck will work with a variety of audio interfaces, including other multimedia sound devices such as the Bluetooth headsets and USB microphones shown under Windows Sound and Audio Devices Properties. There are a wide variety of audio interfaces available with varying degrees of performance but we recommend that you use one of the audio interfaces certified by Listen.

See ***Appendix 1: Hardware Compatibility List on page 557***.

SoundCheck has been validated to work with the National Instruments 4461 data acquisition cards using the DAQmx driver. Other National Instruments IEEE/GPIB DAQmx devices may be compatible. Please contact **support@listeninc.com** for a current list of DAQmx compatible hardware.

See ***Appendix 2: PXI/PCI 4461 Installation on page 567*** for information on setting up the NI 4461.

Other manufacturers' cards can be used, but require some knowledge of LabVIEW programming to create a Custom VI Step inside SoundCheck.

See ***Creating a Custom VI and Custom Step on page 431***.

- I/O devices that are not manufactured by NI must conform to DAQmx and must be recognized by NI MAX, in order to be used with SoundCheck
- As of SC 9.1, Digital I/O functions are compatible with DAQmx devices
- DAQmx devices cannot be used simultaneously with ASIO audio interfaces

Hardware Editor Rules For Production Lines

When setting up multiple SoundCheck systems that will use the same sequence, it is important to follow some basic rules:

- The minimum number of Audio Interface channels should be the same on all systems
- Calibration Editor channels must use the same naming convention across all systems.

  See ***Naming - Best Practices on page 88***.
- When controlling Listen Hardware with Message Steps, e.g.: AmpConnect ISC, AudioConnect and SoundConnect 2, the same hardware must be used across all systems so that the Message Steps perform the specified commands. See ***Listen Hardware Control Message on page 273***.
- AudioConnect 4x4 does not appear in the Listen Hardware Tab since there are no functions to control
- The Listen Hardware **Startup Default** settings should be the same across all systems.

  See ***Listen Hardware Page on page 70***.
- External Interfaces must use the same Interface Numbering scheme across all SoundCheck systems so that Message Steps communicate correctly. See ***External Interface on page 292***.
- NI Daq device ports must have the same minimum number of ports. Each port number must be configured with the same Input or Output status across all SoundCheck systems.

  See ***NI DAQ Digital I/O on page 76***.
- Select **Save As** in the Hardware Editor to save the System.HAR file so it can be imported to other SoundCheck Systems. We recommend that you save it with a name identifying the specific hardware in use. Note that the Hardware Interface Vp and Latency values for a specific interface should be entered on each SoundCheck system after importing the settings for Listen Hardware and External Interfaces.
- Hardware (.HAR) Steps created with SoundCheck 16 and later are not backward compatible with previous versions of SoundCheck

Audio Page

The following settings will affect how WAV files are created and played by SoundCheck, such as in the Stimulus and Acquisition Editors.

Figure 8-2 shows the default System Hardware Configuration for AudioConnect. This shows the hardware channels available when using this audio interface and default Vp values. Channels and other values are Grayed out until the actual device is selected in the **Device** column.

Hardware settings for other audio interfaces or from previous sequences can be imported into the Hardware Editor as well.

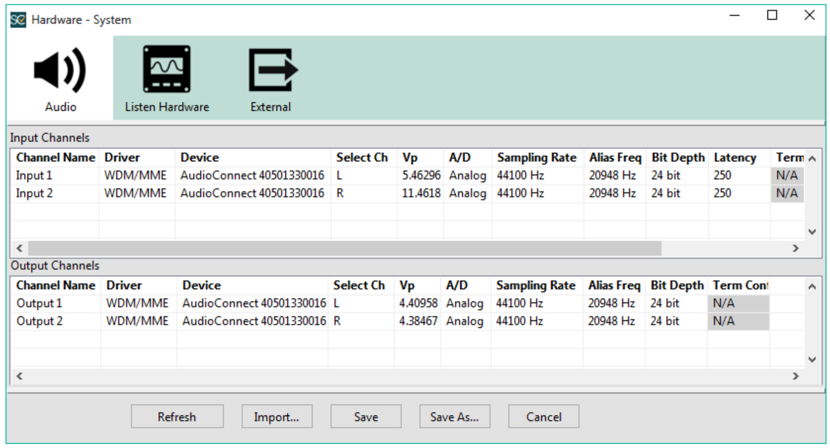


Figure 8-2: System Hardware Table

If device channels in the list become obsolete they can be deleted. ***This will of course affect signal paths in the Calibration Editor that link to that hardware device and any sequence that might use one of the deleted signal paths.*** See ***Warning Deleting Hardware Channel on page 63***.

Each Channel Name should be unique to avoid confusion when editing a sequence or adding hardware at a later date.

Automatic Startup Configuration

Note: As of SoundCheck 18, "**Automatic Startup Configuration**" has been replaced with "**Automatically Create Signal Paths for Listen Devices**" found in **Preferences > Launch**. See ***Automatically Create Signal Paths for Listen Devices on page 46***.

Selection Buttons

Five buttons at the bottom of the editor allow you to:

- **Refresh** - Click to scan for Listen hardware and automatically configure discovered hardware channels. Input and Output Vp values are stored in the firmware of Listen Hardware such as AmpConnect ISC and AudioConnect. These values are automatically loaded in the Hardware Editor when SoundCheck discovers new hardware on startup or when you click **Refresh**. Theses values are fixed and should not be edited.
- **Import** - Allows you to import hardware settings from the Hardware Configuration examples provided with SoundCheck or from other SoundCheck sequences. This will overwrite the settings of current channels in the table if they have the same channel name.
- **Save** - Save changes to hard disk and closes the Hardware Editor
- **Save As** - Allows you to save the configuration for a specific hardware setup. You can create different hardware configurations so you can easily switch between them using the Import function.
- **Cancel** - Discard any changes made while the editor window was open
- **X Close** - Clicking on the Windows **X Close button** discards changes and closes the editor

Importing Hardware Settings

This can be useful when sequences are created on a different SoundCheck system, with different hardware.

When Import is selected, the Hardware Configuration Editor will check for Input and Output channel name duplication. You are prompted to select Yes or No to overwrite existing channels.

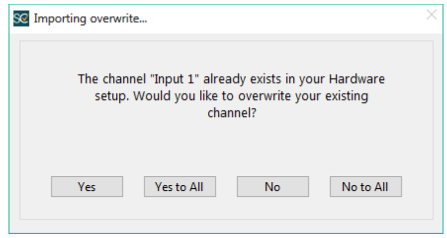


Figure 8-3: Import Overwrite Message

- **Yes/Yes to All**: The current channel settings will be replaced with the settings from the imported channel(s)
- **No/No to All**: You can choose to not update individual channels or all channels
- **New Channels**: If the imported Hardware Configuration has channels with different names from the current configuration, the channels will be added to the Hardware Configuration.

Note: When channels are added to the Hardware Configuration, each channel must have a unique name. The Hardware Configuration cannot be saved if there are duplicate channel names.

The example in ***Figure 8-4*** shows new Input and Output channel names after importing channels from a hardware file for another device.

- The new channels can be renamed. In this case they might be named Input 3, Input 4, Output 3 and Output 4. This flexibility allows the Hardware Configuration to be used as a database for any device that you might have available, even if it is not always connected to the system.
- Right-click a channel and select **Rename**
- The settings for the Lynx audio interface are grayed out and cannot be edited since the device is not present in the system

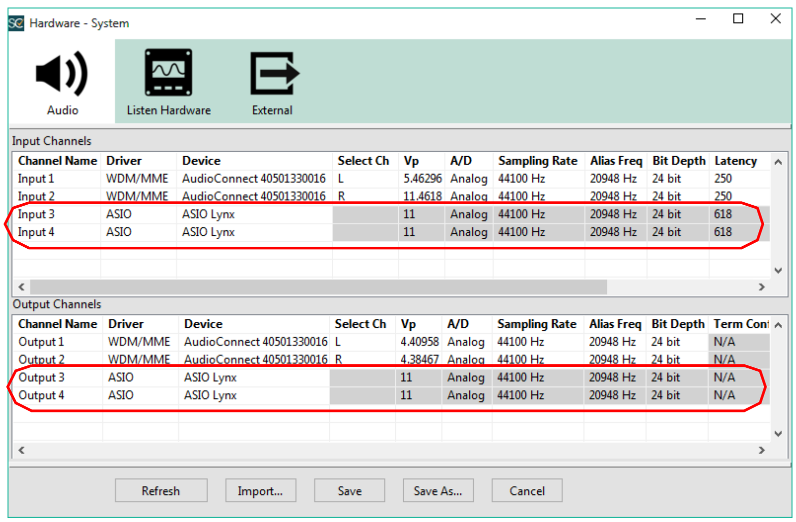


Figure 8-4: Import Channels

Warning Deleting Hardware Channel

Hardware paths connected to Signal Paths cannot be deleted. If you try to delete a connected channel you will see a warning screen showing the associated Signal Path.

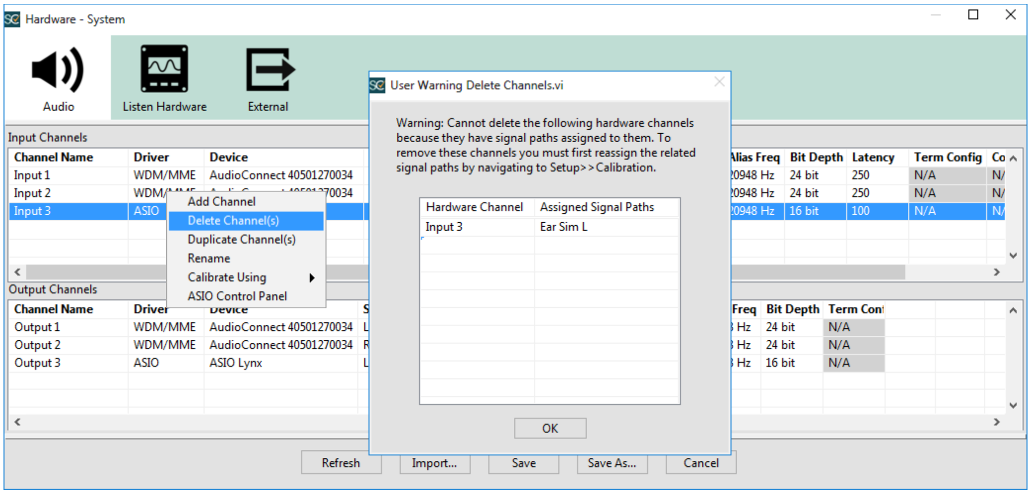


Figure 8-5: Warning Deleting Hardware Channel

The essential hardware settings for all channels of a multichannel Hardware Configuration can be viewed and edited in the table as shown in ***Figure: 8-6***.

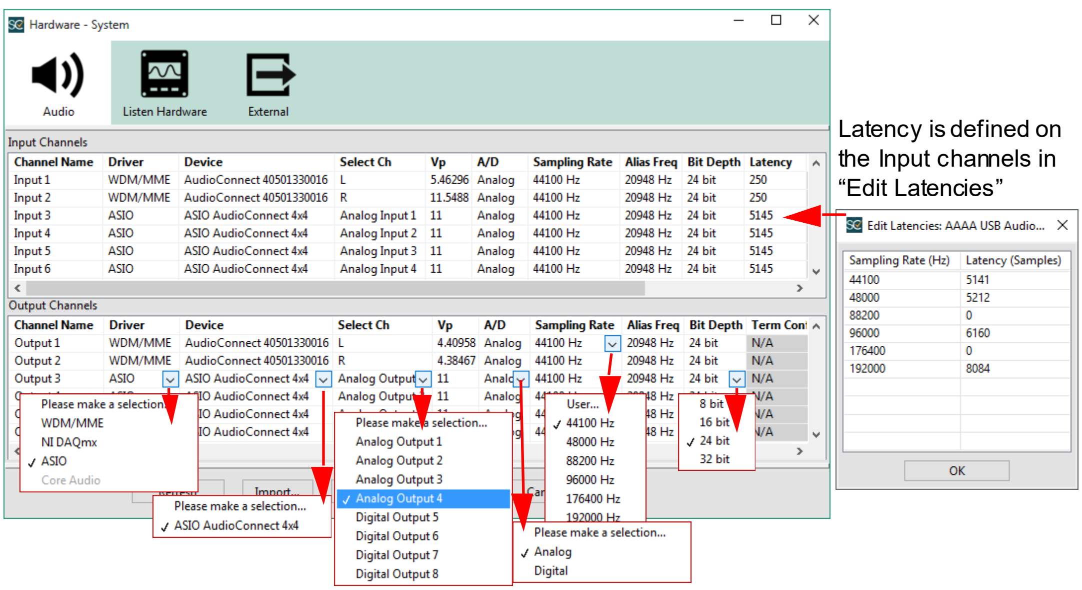


Figure: 8-6 Hardware Editor

Waveforms

The Hardware & Calibration info of a channel are attached to any waveform that was played or recorded through that channel. This allows it to be analyzed on another computer. Please refer to ***Calibration Configuration on page 79*** for more information.

Sort

In the table view of the Hardware Editor you can sort the table by Left-clicking on a column header. This simplifies viewing and editing when you have a lot of channels. For example, sort by Channel Name or sort by Input Channel.

Hardware Table Fields

The Hardware Configuration Table view allows you to add, delete, duplicate and edit hardware settings for all devices available on the SoundCheck system. Each device is defined in a row with column headings of:

- **Channel Name** - Defined by user. Default = Input 1..n, Output 1..n
- **Driver** - Audio Interface or NI DAQmx
    - **WDM/MME** - Select when using an audio interface that only has WDM or MME drivers
    - **WASAPI** - Very useful for multichannel audio interfaces that do not allow for multichannel use with WDM drivers. Allows you to use devices that support WASAPI simultaneously with a separate ASIO device.
    - **DAQmx** - Select to setup an NI DAQmx compatible Digital I/O device. See ***NI DAQmx on page 77*** for more information.
    - **ASIO** - If the installed audio interface features ASIO drivers, this can be selected to take advantage of the benefits of ASIO. Only one ASIO device can be defined in the Hardware Editor at a time.
    - **Core Audio** - Available only on macOS
- **Device Name** - The name of the audio interface as it appears in the Windows Multimedia Stack
- **Select Ch** (Channel 1..n or L/R of Audio Interface Channel Pair) Click on the drop-down list to select a channel.
- **Vp (max voltage)** - Determined in Hardware Channel Calibration process or entered by user

  (See ***Input / Output Vp Values on page 67*** and ***Audio Interface Calibration on page 69***)
- **Analog/Digital** - Set by user
- **Sampling Rate** - Click on the Sample Rate field and select from the available rates for the device

  (For details see ***Sampling Rate on page 66***)
    - For devices using WDM or WASAPI drivers, some listed sampling rates may not be compatible with SoundCheck
    - For Devices that support 384 kHz sample rate see ***User Defined Sample Rate on page 66***
- **Alias Freq** - Anti-aliasing filter frequency, automatically determined but can be edited by user

  (See ***Alias Freq (Alias free freq limit (Hz)) on page 66***)
- **Bit Depth** - Select from list
- **Latency** - This value is tied to the selected Sample Rate. Latencies for Listen Hardware devices are entered in the default HAR files provided with SoundCheck. Third party interface HAR files have the latency for 44.1 kHz entered in the table. Other latencies must be determined in the hardware channel calibration process according to the buffer settings used for your device, on your system.

  For details see ***Latency on page 67*** and ***Latency Adjustment on page 68***.

  For more information on Hardware Channel Calibration see ***Audio Interface Calibration on page 69***.

Right-click Functions

Right-click a line of the table to Add, Delete, Duplicate or Rename hardware channels.

Multiple channels can be selected. Click and hold on the **shift key** to select a range of channels or the control key to select specific channels.

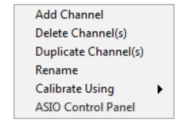


Figure: 8-7 Right-click

- **Add Channel** - Create a new channel in the drop-down list
- **Delete Channel(s)** - Remove the selected channel(s) from the list. This will affect any Acquisition Step in the sequence that uses this channel. See *Warning Deleting Hardware Channel on page 63*.
- **Duplicate Channel(s)** - Creates a duplicate of the selected channel(s) and appends "- Copy" to the name
- **Rename** - Change the name of the selected channel
- **Calibrate Using** - Select the proper input or output channel to calibrate with. This automatically starts the Audio Interface Calibration sequence.
- **ASIO Control Panel** - Right-click the **Channel Name Field** and select **ASIO Control Panel.** (grayed out when no ASIO device is present)

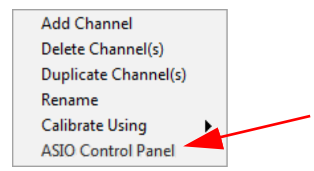


Figure 8-8: Hardware Using ASIO

ASIO

- Audio interfaces with ASIO drivers are supported as of SoundCheck 10.

  (Previous versions of SoundCheck only support WDM / MME drivers.)
- ASIO is a driver standard geared towards Pro Audio equipment, which allows for more stable latencies
- ASIO drivers offer excellent audio interface control and allow for use of a wide range of professional audio multichannel audio interfaces

When using ASIO devices, the Hardware Editor in SoundCheck looks similar to WDM / WASAPI audio interfaces. The channel number is selected after selecting the device name. Right-click a **Channel Name** and select **ASIO Control Panel** to adjust ASIO settings for that device.

ASIO Control Panel

ASIO devices have their own control panel and the placement of controls will vary. The buffer size of the device, and in some cases USB Streaming Mode, is set here. This is directly related to the **Latency Value** in the Hardware Editor in SoundCheck. The example shown is the control panel for an AudioConnect 4x4 audio interface.

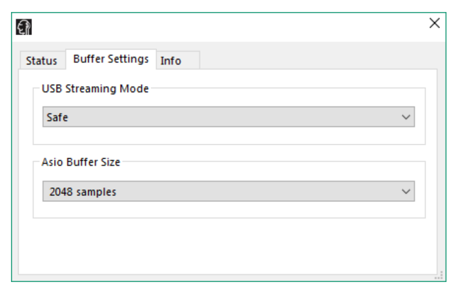
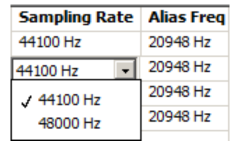


Figure 8-9: ASIO Control Panel

- USB Streaming Mode is set to Safe
- The Buffer Size is set to 2048 Samples. Other interfaces may require larger or smaller buffer sizes. The buffer size is also dependent on the Sample Rate and the number of channels used simultaneously. Please refer to the instructions for your audio interface.

Some ASIO control panels also allow you to set the gain settings for the device. These values must be set before calibrating the ASIO audio interface in SoundCheck.

Sampling Rate



Sampling Rate refers to the audio interface's sampling rate in samples per second (e.g., 8000, 11025, 22050, 32000, 44100, 48000, 96000, 192000, etc.). Refer to the documentation provided with the audio interface for appropriate sampling rates. SoundCheck will check the available sample rates and bit depths for the device and only show those that work.

See ***Figure: 8-6 Hardware Editor on page 63.***

Note: Audio Interfaces with WDM driver (Windows Driver Model) support all sample rates through "**sample rate conversion**". For SoundCheck we recommend that you only use the sample rates supported by the audio interface. If the sample rate between the audio interface and device under test are different, use **Frequency Shift** and **Resample** Post Processing steps as indicated in ***Resampling on page 263***. WASAPI does not support "**sample rate conversion**".

Note: Using a higher sample rate proportionally increases the amount of memory required.

User Defined Sample Rate

For audio interfaces that support sampling rates above 192 kHz you will need to select **User...** in the **Default Sampling Rate** drop-down list and then enter the rate supported by the interface. Once the rate is set for the input channels, the output channels will automatically update to the new sample rate.

Sample rates up to 384 kHz are supported.

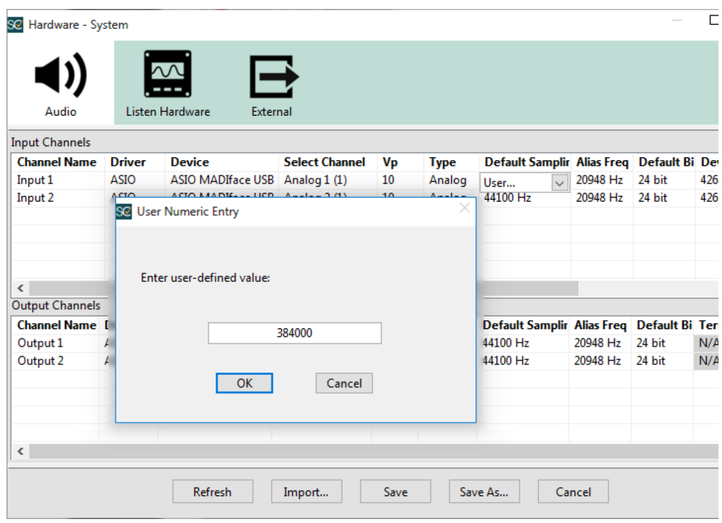
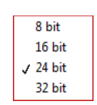


Figure 8-10: User Defined Sample Rate

Alias Freq (Alias free freq limit (Hz))

This shows the limit of the upper frequency range to which the audio interface can measure. It is dependent on the Sample Rate setting and the filter applied by the selected audio interface. This allows you to account for the differences between anti-alias filters in different devices. The default is typically 47.5% of sampling rate. This should prevent anti-alias components of most audio interfaces from creeping into measurements.

Bit Depth



Select the bit depth used by the A/D and D/A converters of the audio interface. Typical values are 16 and 24 bits (sometimes 32 bit words are used to convey 24 bits of information as in the Lynx Studio and RME audio interfaces).

24-bit audio will provide the greatest dynamic range (approximately 120 dB). However, when using 24-bit audio, more computer memory is required. If there is insufficient RAM, SoundCheck may slow down significantly, because the computer is using the hard drive as virtual memory. If this occurs, you may need to upgrade the computer's RAM.

Device Selection - Input or Output

Select the device to be used for Input or Output from the drop-down list. These are stored by device name with the Hardware Configuration.

If the device is not available (Hardware removed from PC or sequence imported on another PC without the proper hardware), a warning message will appear. Open the editor to update the Hardware Table or select Ignore.

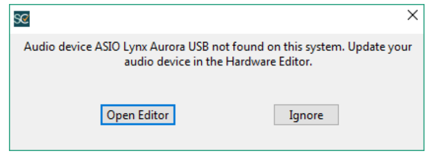


Figure 8-11: Hardware Not Found

Input / Output Vp Values

This defines the maximum peak voltage that corresponds to 0 dBFS (Full Scale Deflection). This value is different for all interfaces. The Input Vp can also vary depending on the input type, e.g.: Line In vs Mic In.

All audio interfaces provided by Listen Inc. have predefined Hardware Configurations in SoundCheck.

When using Listen Hardware, such as AudioConnect or AmpConnect, the Vp values for the input and output channels are stored on the device and automatically updated in the Hardware Editor when "**Automatically Create Signal Paths for Listen Devices**" is selected. These Vp values are fixed and should not be edited. See ***Automatically Create Signal Paths for Listen Devices on page 46***.

If you have a different audio interface, the Vp values for each "Output / Input channel set" can be measured and entered into the Hardware Table. See ***Audio Interface Calibration on page 69*** for instructions. You will also need to update the Sample Rate/Latency Table as shown in ***on page 69***.

Latency

This is the time delay or latency between input and output hardware channels when operating in full-duplex mode (record and play simultaneously). This value is expressed in seconds and in samples. The latency expressed in seconds is calculated from the integer number of samples. SoundCheck uses samples in measurements, since there are no rounding errors. (For audio interfaces that Listen provides, this value will be known.)

Most audio interfaces cannot record and play simultaneously. There is almost always a delay between the two and the delay should not vary from measurement to measurement. The audio interfaces that Listen provides are certified to have high performance in making audio-related measurements. If you are using an audio interface that Listen, Inc has not certified, the measurement performance of SoundCheck may be severely compromised!

The Latency is determined in Hardware Channel Calibration process and entered in the **Edit Latencies** window. Latencies for Listen Hardware devices are included in the default HAR files provided with SoundCheck. Third party interface HAR files have the latency for 44.1 kHz entered in the table. Other latencies must be determined in the hardware channel calibration process according to the buffer settings used for your device, on your system. See ***on page 69*** for instructions.

Important! If the latency is not consistent, (as with WDM / WASAPI audio interface drivers), **Auto Delay** must be enabled in the **Analysis Steps** of all sequences or measurements will NOT be reliable. See ***Transition Discard Time - Time Tab on page 172***.

Sampling Rate/Latency Table

The table allows you to set the Latency for all of the available sample rates for the audio interface in use. For Listen Hardware these values are already filled in for the typically used sample rates. Values in the table can be manually updated for 3rd party hardware. Click on the drop down arrow next to the value in the **Latency** field of the Hardware Editor. Select **Edit** and the Latency Table will open. Click in a Latency field to enter the latency value for that sample rate.

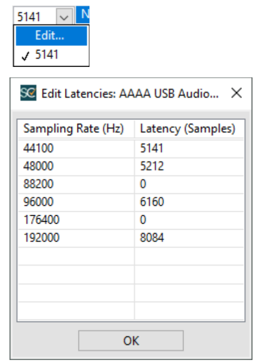


Figure 8-12: Edit Latencies Table

Latency Adjustment

When measuring **Absolute Phase** we recommend that you run the "**Self Test**" sequence from the Calibration sequence folder with the **Sample Rate** set to the desired value in the **Hardware Editor**.

Note: This only applies when using an audio interface with ASIO driver. WDM/WASAPI drivers do not allow for consistent Latency. These devices require that you use Auto Delay or Auto Delay+ in any Analysis Step.

To determine the Latency value for a Sample Rate you must also consider the **ASIO Buffer Size** (and **USB Streaming Mode** if applicable). These values vary from one model of audio interface to another. These values should be set in the **ASIO Control Panel** for the audio interface. Instructions for Listen approved audio interfaces are included with the approved drivers which are available on the Listen Support website: ***https://support.listeninc.com/hc/en-us/sections/200370694-Drivers***

Instructions

1. Open the Hardware Editor. Change the **Sample Rate** to the value you need to measure Latency for. Click on the drop down arrow next to the value in the **Latency** field. Select **Edit** and the **Edit Latencies** table will open.
2. Set the **Latency** for the desired sample rate to 0 (zero), click **OK** and then click **Save**.
3. Make sure the sample rate of the audio interface has updated. Change the **ASIO Buffer/USB Streaming mode** for the audio interface in the **ASIO Control Panel** (if applicable).
4. Run the **Self Test** sequence from the Calibration folder in SoundCheck. The **Results** display shows the **Audio Interface Latency** for the new Sample Rate/ Buffer Size. (Some audio interfaces may require "fine tuning" of the Latency value by following Step 5 and then repeating Step 4.)

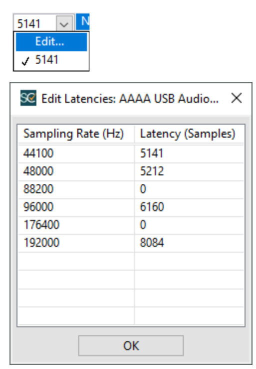


Figure 8-13: Edit Latencies Table

5. Enter this value in the Latency field of the **Hardware Editor - Sample Rate - Edit Latencies table** by repeating steps 1 and 2. The same process is used for other required Sample Rates.
6. You can run the Self Test sequence again to verify the new Latency value.
7. All channels, analog or digital, must have the same **Latency** value across all channels of the audio interface. This insures the system will work correctly if they are used simultaneously in a sequence.

Hardware Configurations

You can click **Save As** in the Hardware Editor to save different hardware configurations so that they can easily be recalled.

A/D - Analog/Digital Selection

When set to **Analog**, the Max Input and Output Vp values of the audio interface can be set. (Values determined by the Audio Interface Calibration Sequence.)

When set to **Digital** the Input/Output values change to 100% FSD (Full Scale Deflection) of the audio device selected. The Vp value should be set to 1.

Audio Interface Calibration

Right-click a channel and select **Calibrate Using** then select the proper input or output channel. The Audio Interface Calibration Sequence will run automatically on the selected Input/Output channel signal chain.

See ***Figure: 8-14***.

The Audio Interface Calibration process will instruct you on the required connections and procedure.

See ***Balanced Audio Interface Calibration Connections on page 578*** when using a balanced audio interface.

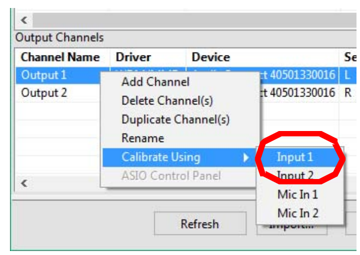


Figure: 8-14 Calibrate Using...

When finished, the specific maximum Vp values for the Input and Output channel set tested are entered into the appropriate fields of the Hardware Table.

Rules

- Close your sequence before calibrating audio interface channels in the Hardware Editor.
- Only one channel set, Output to Input, is calibrated at a time. A four channel audio interface will require four different calibration tests, one for each channel.
- The calibration sequence should not be modified!

Listen Hardware Page

Listen Hardware is automatically added to this page when it is detected by SoundCheck. Once added, items will remain on the page until manually removed. Disconnected items are noted as “Disconnected” in the **Status** column.

The fields below apply to all Listen Hardware devices.

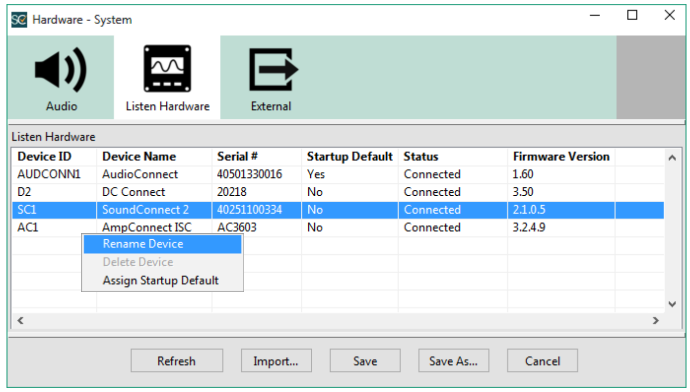


Figure 8-15: Listen Hardware

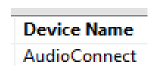
Device ID

Shows the device identification for all devices that have been connected to the SoundCheck system. Device IDs can be changed to make it easier to keep track of multiple devices of the same Device Name.

- **Rename** - The device name default can be changed by Right-clicking on the device line and selecting **Rename Device**
- **Delete Device** - Right-click the device line to remove a device from the table

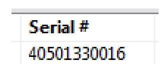
Device Name

Shows the product name of the device



Serial Number

Shows the Serial number of the connected device



Startup Default

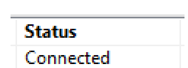
Right-click the device line to set the initial settings that the Listen hardware device will adopt when SoundCheck opens

Setting the Startup Default gain to the maximum available for the selected Listen Hardware device is not recommended if you plan to override the gain setting in an Acquisition Step noting the following:

Important! ***Switching Listen Hardware from Maximum Gain to Minimum Gain in the Acquisition Step is not recommended. This does not allow the input gain circuit sufficient time to stabilize. If you need to switch from Max Gain to 0 or Minimum Gain we recommend that you use a Listen Hardware Message step with a 500 mSec wait time to allow for settling.***

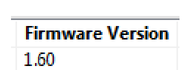
Status

Indicates if the device is connected and recognized by SoundCheck or disconnected



Firmware Version

Shows the firmware of the connected device



AmpConnect ISC™

The AmpConnect ISC **Startup Default** controls in ***Figure 8-16*** are the same as the AmpConnect ISC **Message Step**.

Refer to the ***AmpConnect ISC Manual*** for more detailed information.

Important! ***After installing SC 14, prior versions of SoundCheck will not have control over the device. Additionally, the serial number of the AmpConnect ISC audio interface will not be read properly which changes the name of the device in the Hardware Editor.***

For details on step settings see ***AmpConnect ISC Message on page 277***.

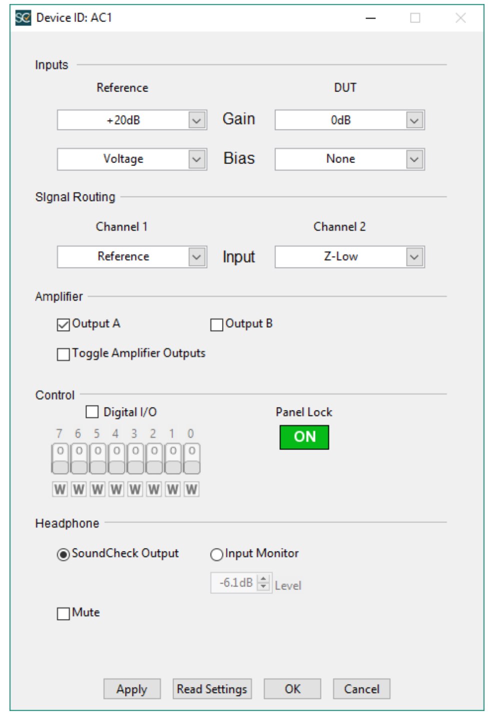


Figure 8-16: AmpConnect ISC Startup Default

AmpConnect 621™

The AmpConnect 621 **Startup Default** controls in ***Figure 8-17*** are the same as the AmpConnect 621 **Message Step**.

Refer to the ***AmpConnect 621 Manual*** for more detailed information.

For details on step settings see ***AmpConnect 621 Message on page 275***.

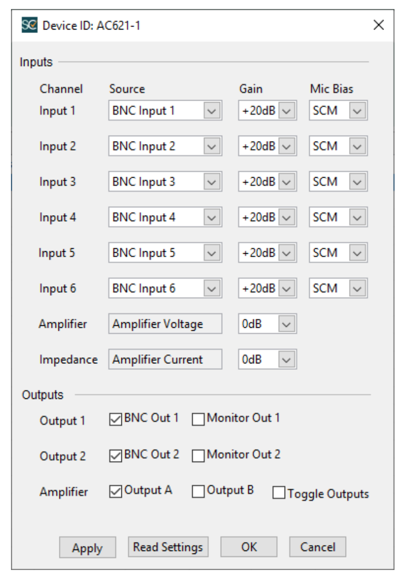


Figure 8-17: AmpConnect 621 Startup Default

AudioConnect™

The **Startup Default** controls in ***Figure 8-18*** are the same as the AudioConnect **Message Step**.

The AudioConnect audio interface can be controlled through a SoundCheck Message Step. This allows you to change the input and output configuration of the device during the operation of a SoundCheck sequence.

For details on step settings see ***AudioConnect Message on page 274***.

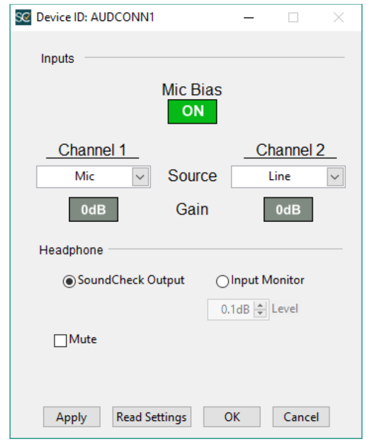


Figure 8-18: AudioConnect Startup Default

Listen Hardware Digital I/O Message

Digital I/O for Listen hardware devices such as AmpConnect 621™ and AmpConnect ISC™ is now a separate **Message Step**.

This allows Digital I/O changes to be separate from audio channel settings. This improves use of Digital I/O steps in a sequence as these steps no longer interfere with electro-acoustic settings.

The new settings in the Digital I/O Message Step allow Listen hardware to have the same switching scheme and appearance as other Digital I/O devices. This makes translation of steps from NI Digital I/O device to Listen hardware easier.

See ***Digital I/O Message Step on page 295***.

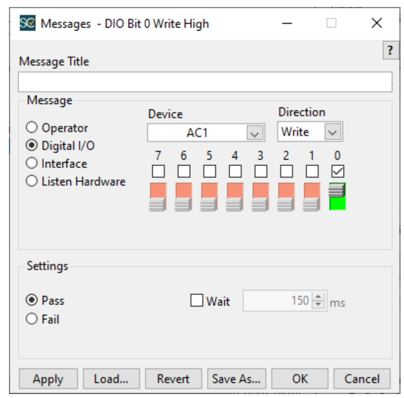


Figure 8-19: Digital I/O Message

SoundConnect 2™

SoundConnect 2 is a compact and rugged USB controlled microphone power supply and conditioning amplifier.

It can be controlled through a SoundCheck Message Step. This allows you to change the input signal routing, Gain and High/Low Pass filters during the operation of a SoundCheck sequence.

Right-click the SoundConnect 2 line and select **Assign Startup Default**. The controls in ***Figure 8-20*** are the same as the SoundConnect 2 Message Step.

For details on step settings see ***SoundConnect 2 Message on page 274***.

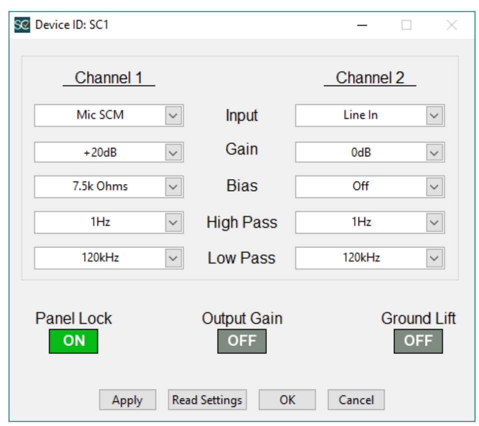


Figure 8-20: SoundConnect 2 Startup Default

DC Connect[TM]

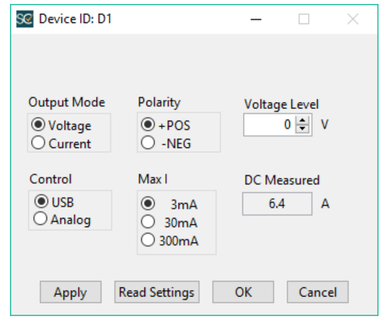


Figure 8-21: DC Connect Startup Default

The Listen *DC Connect* is a USB-controlled DC power supply and measuring amplifier used for measuring the DC voltage and current consumption on DC-powered audio devices. The Listen Hardware table shows any DC Connect device that has been setup in the Hardware Editor. SoundCheck supports the use of only one DC Connect at a time.

DC Connect can be controlled in SoundCheck by using Message Steps as well as the *Stimulus Editor* and *Acquisition Editor*.

Important! As of SoundCheck 13, after installing SoundCheck, you cannot use DC Connect with versions prior to SoundCheck 13, unless you manually switch the drivers in Windows Device Manger. Download the latest DC Connect manual from the Listen website for a step by step procedure.

The DC Connect startup default editor is shown in ***Figure 8-21***. This allows you to set DC Connect to 0 VDC output when SoundCheck is launched, along with other device settings.

For details on step settings see ***DC Connect Message on page 280***.

Multiple DC Connects

As of SoundCheck 16.2, you can control and measure with more than one DC Connect. DC Connects are automatically given Device IDs in the Hardware Editor as they are added to the system. Once the relationship between the Serial number is established the DC Connects can be connected in any order and the Hardware Editor retains the relationship between Device ID and Serial Number. The example in ***Figure 8-22*** shows Serial # 20237 is Device ID “**D1**”.

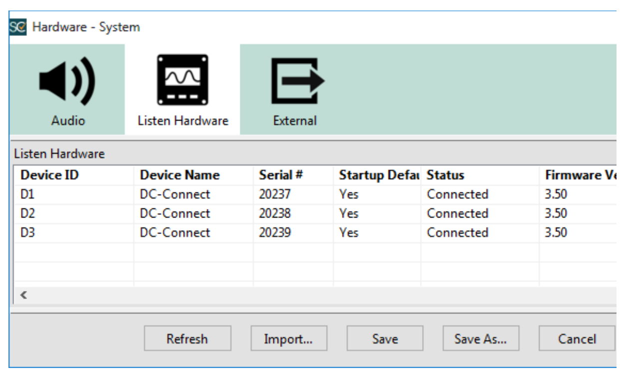


Figure 8-22: Hardware Editor

Adding a new DC Connect to the example in ***Figure 8-22*** would create a new entry with a Device ID of “**D4**”.

If the new device is intended to replace a Device ID in the list, you must remove the old Device and then manually change the new device by Right-clicking on the Device ID and editing it.

Proper Device ID maintenance is critical to retain the proper relationship between each device and how it is used in Message and Acquisition Steps. Changing Device IDs in Listen Hardware will break device assignments and data names in pre-existing sequences. When changing DC Connects on a system it is best to keep the Device IDs consistent so they match Device IDs in sequences.

Portland Tool & Die BTC-4148/4149 & PQC-3048 Message Step Controls

The BTC-4148/4149 is a complete interface for measuring and characterizing Bluetooth audio devices including handsets, headsets, speakers, car kits and other devices with Bluetooth audio input or output. The PQC-3048 is a similar interface designed for production line use.

As of SoundCheck 17, the new APTX HD codec for high resolution Bluetooth testing is supported and requires the latest version of the BTC-4149 Bluetooth Interface, which is fully integrated with, and controlled by, SoundCheck.

The BTC interface can be controlled through a SoundCheck Message Step. This allows you to change the pairing of devices and the profiles used when testing, during the operation of a SoundCheck sequence.

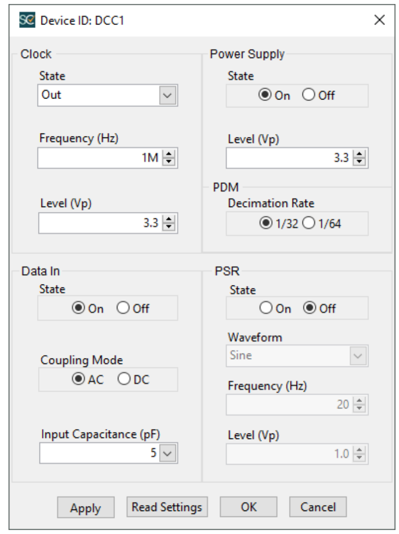


Figure 8-23: PT&D Message Step

Startup Defaults

When SoundCheck starts, Startup Default settings are written to the DCC/PQC device. This ensures that the device is ready to take measurements from a specified setting.

Message Step

DCC/PQC Message Steps can be used to change the setup of the test interface during the run of a sequence or as a standalone off line step.

Quick Setup

The DCC/PQC control allows for faster setup of the measurement interfaces with SoundCheck as well as greater consistency in measurements as the device settings can be built into the test sequence. This allows for a seamless transition between R&D and production testing. Sequences from product development using the DCC can easily be used for the production line with the PQC.

In the Listen Hardware Table, Right-click the BTC-4148/4149 line and select **Assign Startup Default**.

The controls in ***Figure 8-24*** are the same as the BTC-4148/4149 Message Step.

- **A2DP Profile** - Select SBC, aptX or

  aptX-HD (only available when using the BTC-4149)
- **HFP Profile** - Select CVSD or mSBC
- **Audio Source** - Only available in Startup Default
  - Select USB or SPDIF
- **Role** - Only available in Startup Default
  - Source - Use when connecting to transducer
  - Sink - Use when connecting to a phone or laptop

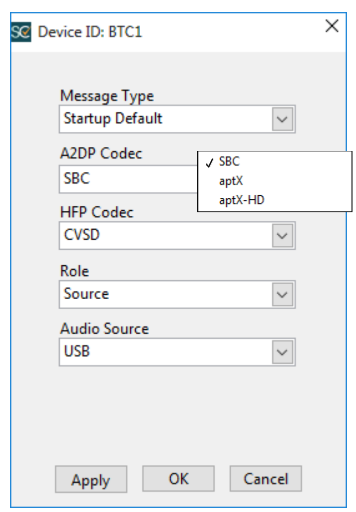


Figure 8-24: BTC-4148 Startup Default

For details on step settings see ***Portland Tool & Die BTC/BQC-4148/4149 Message on page 281***.

External Hardware Page

Interface Table

Select which Computer Interface type is being used to control external devices, such as a multiplexer or turntable. To send or receive IEEE-488 (GPIB) and RS232 commands, use the *Message Editor*.

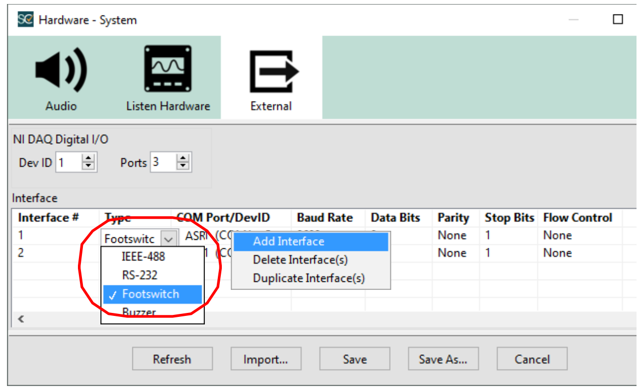


Figure 8-25: External Interface Setup

Interface

Each external interface is assigned an Interface Number which is used to identify the device in Messages Steps used in SoundCheck sequences.

Right-click the Interface Number line and select **Add Interface**.

This can be a mixture of IEEE-488, RS-232, Serial Footswitches or Serial Buzzers.

See ***Serial Port Control on page 581*** for footswitch and buzzer wiring.

Note: The IEEE interface card must have a LabVIEW driver.

Important! The Interface Number for external devices must be the same across multiple SoundCheck systems that share a common Message Step in a sequence. Message Steps use these device numbers to identify the device used in the step. See External Interface on page 292***. Any change in the order of devices in the Hardware Editor will cause communication errors with the external devices. See*** Hardware Editor Rules For Production Lines on page 60***.***

Type

Type of communication interface, such as IEEE-488, RS232, Footswitch, or Buzzer. If the Footswitch is installed and configured in the **Hardware Configuration**, it can control the **Start** button of SoundCheck and the **Continue** button on the *Main Screen*. A second Footswitch can control the **Redo** and **Stop** button.

COM Port

Communication port number as defined by the *Windows System Properties Device Manager*.

Baud rate

Transfer rate speed of the communication port (in bits per second).

Data bits

Specify the data bits for this port. Consult your hardware manufacturer for more information.

Parity

Specify the parity for this port if your serial device requires this setting.

Stop bits

Specify the stop bits for this port if your serial device requires this setting.

Flow control

Specify the flow control for this port if your serial device requires this setting.

- None - Default setting
- Xon/Xoff - This allows the RS232 device to tell the computer port when it is ready to send or receive
- Hardware - Typically not used. It may require pull up resistors on specific control lines. Please consult the documentation for your RS232 device if this is required.

Important! SoundCheck cannot access a COM Port that is also be being referenced by Windows or another app. Make sure that no other software applications are using the COM Port that is used in the Hardware Configuration.

NI DAQ Digital I/O

SoundCheck can interact with PLC controls on a production line by using Digital I/O Message steps in a sequence. These steps control a Digital I/O device such as the NI-65xx DAQmx series from National Instruments.

The NI DAQ Digital I/O fields on the External tab allow you to select the Device ID and number of Ports for the device in use. These settings must be made before setting up a Message Step to control the Digital I/O device.

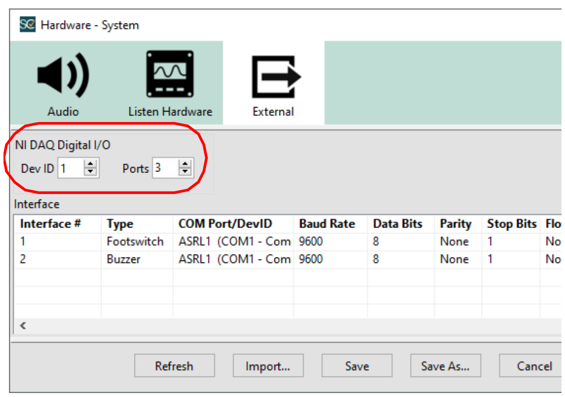


Figure 8-26: External

Dev ID

Device identification number for the digital I/O board. To determine the ID number in Windows, look at the *System Device Manager* to locate your board.

For DAQmx devices, look in the NI MAX software application to determine the device ID.

No. of Ports

Set the number of Input/Output Ports available on the digital I/O board.

Remember that the port count starts at 0, e.g.: A three port device has ports 0, 1, and 2. In this case you would select 3 in the Ports field.

Note: DAQmx devices require that all ports are setup and defined in each Message Step that controls the device. Even though you may only be using two ports of a four port device, you must define all ports.

Port Configuration

As of SoundCheck 18, port configuration is controlled in the Messages Steps that control the Digital I/O device. See ***Digital I/O Message on page 290***.

NI DAQmx

This sets the properties of the NI 4461 Analog Data Acquisition Card. (optional hardware for testing electronics devices such as audio interfaces, amplifiers, preamps, etc.)

Notes:

- The NI 4461 card requires the installation of NI-DAQmx. Use the latest approved DAQmx version from the SoundCheck installation DVD. ***See "Hardware Compatibility List" on page 557.***
- DAQmx devices cannot be used simultaneously with ASIO audio interfaces.
- As of SoundCheck 11.0, the RTA virtual instrument is compatible with NI DAQmx devices. Also note that the Multimeter and Scope/FFT cannot be used simultaneously with NI DAQmx devices.

Important: ***Output and input sample rates must match in the hardware editor. The NI 4461 clock defaults to the output sample rate.***

When the NI-Daq Acquisition device is selected the *Hardware Editor* has the following constraints:

- Digital In and Out are not available. Only Analog is available.
- Input and Output Vp values are in Volts and relate to the Sensitivity of the channel. ***See "Input and Output Vp" on page 78.***
- Bit Depth is fixed at 24 Bit

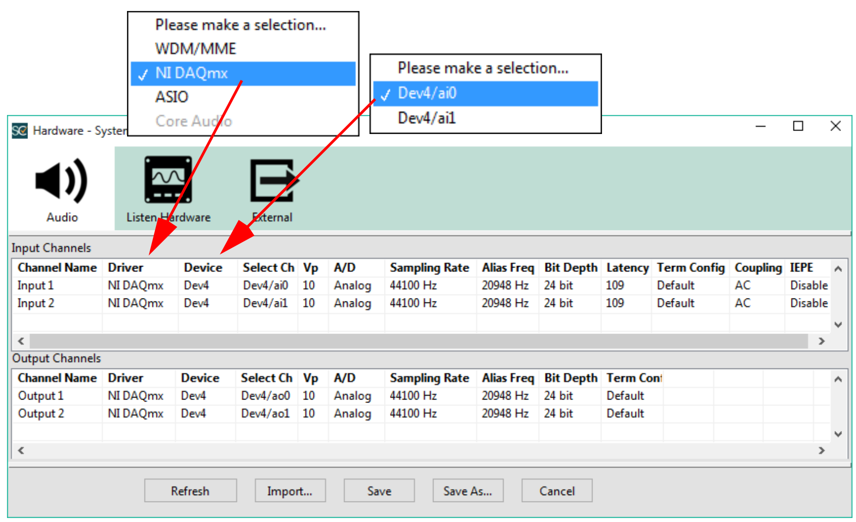


Figure 8-27: NI 4461

Driver: Select NI DAQmx

Device: Select proper device ID

Select Ch: Select Channels according to Device ID, Select ai0 (Input 1) or ai1 (Input 2)

Vp: Set the Vp value for desired sensitivity (***See "Input and Output Vp" on page 78.***)

Sample Rate: The NI 4461 operates at a sample rate of 51.2 kHz. Select **User...** from the Sampling Rate drop-down list. Enter a sample rate of **51200** and click **OK** to save the new sampling rate.

Latency: Set according to chart (***See "Sample Rate / Latency" on page 78.***)

Term Config: Select Default, RSE, NRSE, Differential or Pseudo-Differential

Coupling: Select AC, DC or GND

IEPE: Enable to turn on ICP power for the NI 4461 Inputs. *See* ***Figure 8-27***.

The IEPE check box on the Audio Input tab is used to turn on ICP power for the inputs of the NI 4461. This will power the SCM microphone as well as any ICP powered transducer (4 mA current standard).

Input and Output Vp settings

Input Range

Gain (dB)	Vp (Hardware Config Setting)[1]
30	0.316
20	1.00
10	3.16
0	10.0
-10	31.6
-20	42.4

[1] Each input channel gain is independently set in the Hardware Config.

Output Range

Attenuation (dB)	Vp (Hardware Config Setting)[2]
0	10.00
-20	1.0
-40	0.1

[2] Each output channel attenuation is independently set in the Hardware Config.

Figure 8-28: Input and Output Vp

Latency and Sample Rate

The NI 4461 operates at a default sample rate of 51.2 kHz. Loading the default **HAR** file for the NI 4461 will enter the **Sample Rate** and **Latency** for you.

You can also manually edit the settings in the Hardware Editor. The Latency of the 4461 will change as the Sample Rate of the Hardware Configuration changes. The following chart shows the sample rates supported in SoundCheck and the Latency values.

See ***Latency Adjustment on page 68*** for instructions on determining the best Latency Value for your system and changes to the Edit Latencies table.

Sample Rate (Hz)	Latency (Samples)
200k	100
192,000	100
176,400	100
96,000	114
88,200	114
51,200	108
48,000	109
44,100	109
32,000	109
16,000	90
8,000	80

Figure 8-29: Sample Rate / Latency

Calibration Configuration

The accuracy of your SoundCheck system depends upon accurate calibration of your input and output devices. Nominal calibration values for many devices typically used are included with SoundCheck. For more accurate measurements, calibration of individual signal paths should be performed for each device. Frequency of calibration of these devices depends upon the stability of the device.

Important! As of SoundCheck 11, Calibration is a "System Level" configuration. It is unique to a specific SoundCheck system and is used by all sequences. Individual Calibration Steps are no longer used. Settings from these steps can be imported in the Calibration Configuration Editor. See ***Input Tab on page 83.***

The *Calibration Editor* (**Ctrl+Shift+C**) is used to calibrate the complete SoundCheck® test setup including signal conditioning (e.g., amps and preamps) and transducers (e.g., microphones and sound sources). This allows absolute measurements of acoustic, electroacoustic, electrical, and electronic devices. This step calibrates the entire system. See ***Figure 9-15*** for a system diagram.

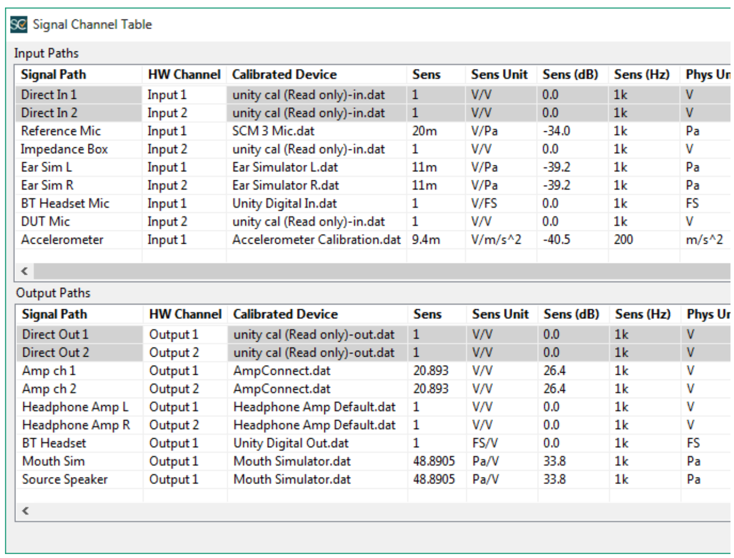
The System Calibration Configuration input/output sensitivities and units are also used for calibrating the virtual instruments accessed in SoundCheck's **Instruments** drop-down list.

System.Cal

This is the database for all signal path setups and calibration data for the system.

Figure 9-1 shows the table view of the Calibration Configuration supplied with SoundCheck.

- Channel setups can be imported from previous SoundCheck sequences
- Obsolete channels can be deleted. (Note that this can effect the curves used by this and other sequences.)
- Best Naming practices should be followed when creating new calibration items. See ***Naming - Best Practices on page 88.***
- Each line defines the following for use in a sequence or virtual instrument:
  - Input or Output path
  - Calibration data file
  - Physical connection to hardware
- Tab View shows information for selected Input/Output channels.
- Table View shows all input and output channels.


Signal Channel Table

Input Paths

Signal Path	HW Channel	Calibrated Device	Sens	Sens Unit	Sens (dB)	Sens (Hz)	Phys Ur
Direct In 1	Input 1	unity cal (Read only)-in.dat	1	V/V	0.0	1k	V
Direct In 2	Input 2	unity cal (Read only)-in.dat	1	V/V	0.0	1k	V
Reference Mic	Input 1	SCM 3 Mic.dat	20m	V/Pa	-34.0	1k	Pa
Impedance Box	Input 2	unity cal (Read only)-in.dat	1	V/V	0.0	1k	V
Ear Sim L	Input 1	Ear Simulator L.dat	11m	V/Pa	-39.2	1k	Pa
Ear Sim R	Input 2	Ear Simulator R.dat	11m	V/Pa	-39.2	1k	Pa
BT Headset Mic	Input 1	Unity Digital In.dat	1	V/FS	0.0	1k	FS
DUT Mic	Input 2	unity cal (Read only)-in.dat	1	V/V	0.0	1k	V
Accelerometer	Input 1	Accelerometer Calibration.dat	9.4m	V/m/s^2	-40.5	200	m/s^2

Output Paths

Signal Path	HW Channel	Calibrated Device	Sens	Sens Unit	Sens (dB)	Sens (Hz)	Phys Ur
Direct Out 1	Output 1	unity cal (Read only)-out.dat	1	V/V	0.0	1k	V
Direct Out 2	Output 2	unity cal (Read only)-out.dat	1	V/V	0.0	1k	V
Amp ch 1	Output 1	AmpConnect.dat	20.893	V/V	26.4	1k	V
Amp ch 2	Output 2	AmpConnect.dat	20.893	V/V	26.4	1k	V
Headphone Amp L	Output 1	Headphone Amp Default.dat	1	V/V	0.0	1k	V
Headphone Amp R	Output 2	Headphone Amp Default.dat	1	V/V	0.0	1k	V
BT Headset	Output 1	Unity Digital Out.dat	1	FS/V	0.0	1k	FS
Mouth Sim	Output 1	Mouth Simulator.dat	48.8905	Pa/V	33.8	1k	Pa
Source Speaker	Output 1	Mouth Simulator.dat	48.8905	Pa/V	33.8	1k	Pa

Figure 9-1: System Calibration Table

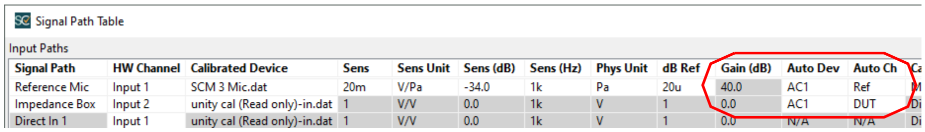
For more information on calibration procedures and specifics, please refer to ***Calibrating SoundCheck on page 92***.

Features

Calibration Editor Signal Paths are automatically created for new Listen Hardware devices by default. See ***Automatically Create Signal Paths for Listen Devices on page 46***.

Auto Dev and Auto Ch Fields

As of SoundCheck 18, fields have been added to support Listen Hardware devices such as AmpConnect and SoundConnect 2.



Signal Path	HW Channel	Calibrated Device	Sens	Sens Unit	Sens (dB)	Sens (Hz)	Phys Unit	dB Ref	Gain (dB)	Auto Dev	Auto Ch
Reference Mic	Input 1	SCM 3 Mic.dat	20m	V/Pa	-34.0	1k	Pa	20u	40.0	AC1	Ref
Impedance Box	Input 2	unity cal (Read only)-in.dat	1	V/V	0.0	1k	V	1	0.0	AC1	DUT
Direct In 1	Input 1	unity cal (Read only)-in.dat	1	V/V	0.0	1k	V	1	0.0	N/A	N/A

- **Auto Dev** - Select the Listen hardware device from the drop down list to associate with the Signal Path
- **Auto Ch** - Select the device channel from the drop down list to associate with the Signal Path
- See ***Listen Hardware - Auto Device / Auto Channel on page 84***
- As of SoundCheck 18, Auto Read has been removed from the Calibration Editor Input Tab

- Accounts for all of the equipment in the measured chain:
  - Amplifiers
  - Measurement Microphones
  - Sound Sources
  - Preamplifiers
- Calibration information for each channel includes:
  - Magnitude and phase response
  - Sensitivity
  - dB reference units
- The Calibration Table serves as a database of stored devices
- Each Signal Path assigns a set of calibration data to a physical Hardware Input/Output channel
- Calibration information can be shown in Table View making it easier to review multiple devices
- Custom curves are easily imported
- A single calibrated device can be associated with any number of hardware channels
- Calibration History allows you to view past calibration data for a device

Sort

In the table view of the Calibration Editor you can sort the table by Left-clicking on a column header. This simplifies viewing and editing when you have a lot of channels. For example, sort by Signal Path or sort by HW Channel.

TEDS support

TEDS Support (with compatible Listen hardware) enables automatic identification, configuration and calibration of TEDS microphones and accelerometers, saving time on initial hardware setup and whenever hardware is changed.

Calibration Basics

System Calibration Structure

Input and Output Signal Paths are defined in the Calibration Editor.

A Signal Path links a Calibrated Device (mic, amp, etc.) to a physical Hardware Channel (audio interface in/out).

The following diagram shows the basic structure of a channel in System Calibration.

System Calibration Layout

Input Channels

1. Input Signal Path	2. Calibrated Device	3. Hardware Channel	4. Listen Hardware Only
Name of Channel	Unique DAT file with correction data for a specific input device	Input channel of audio interface (Named in Hardware Step)	Auto Dev and Auto Ch selections allows Gain to automatically update

Output Channels

1. Output Signal Path	2. Calibrated Device	3. Hardware Channel	4. Input Signal Path
Name of Channel	Unique DAT file with correction data for a specific output device	Output channel of audio interface (Named in Hardware Step)	Input Signal Path used for calibration (Named in Calibration Step)

The next diagram shows how this structure would be used for specific devices.

System Calibration Example

Input Channels

Signal Path	Calibrated Device	Hardware Channel	Listen Hardware Auto Dev	Auto Ch
Reference Mic	SCM 2 Mic.dat	Input 1	AudConn1	Channel 1
Direct In 2	Unity Cal (Read Only).dat	Input 2	AudConn1	Channel 2

Output Channels

Signal Path	Calibrated Device	Hardware Channel	Input Signal Path
Amp Ch 1	SC Amp- L.dat	Output 1	Direct In 1
Amp Ch 2	SC Amp- R.dat	Output 2	Direct In 2

This structure allows you to have one calibrated device associated with many hardware channels, or have many calibrated devices associated with one hardware channel. Of course, only one signal path can be used with a hardware channel when sending/receiving signal.

The Input and Output Tabs divide the calibration settings for the Signal Paths into two groups. Both groups can have multiple Input or Output Signal Paths.

- Direct In 1 and 2, and Direct Out 1 and 2 are **Protected Paths** that cannot be removed or edited

Each Signal Path is linked to a set of calibration data (calibrated device) as well as a physical hardware channel. This information is also available in the Table View.

SoundCheck Signal Flow

The diagram in *Figure 9-2* shows the successive transfer functions that occur in the signal chain, from the output of SoundCheck back to the input.

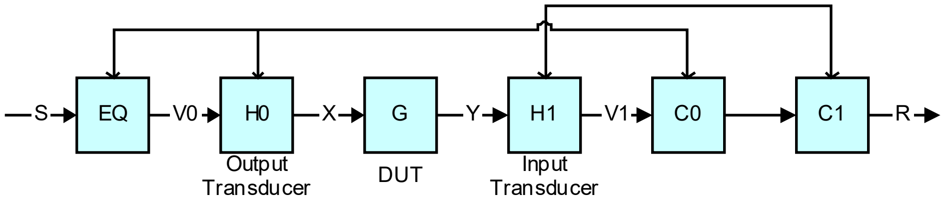


Figure 9-2: Signal Flow

1. **S**: Stimulus created in Stimulus Step
2. **EQ**: Equalization is applied to the stimulus to correct for the response of the output transducer (e.g., Mouth) [NAME eq-out.dat, NAME eq-out.dat]
3. $\mathbf{V_0}$: Electrical stimulus played out the audio interface
4. **X**: Physical input to the DUT (G)
5. **Y**: Physical output from the DUT (G)
6. $\mathbf{V_1}$: Electrical response acquired by the audio interface
7. $\mathbf{C_0}$: Correction out for the output transducer [NAME corr-out.dat, NAME corr-out.dat] (applied after analysis)
8. $\mathbf{C_1}$: Correction in for the input transducer (e.g., Microphone) [NAME corr-in.dat, NAME corr-in.dat] (applied after analysis)
9. **R**: Response calculated with output and input transducers compensated

- The **EQ** correction curve is used to compensate for the response of the output transducer. This is applied to the stimulus before it is played out of the audio interface.
- $\mathbf{C_0}$ is used to fine-tune the compensation for the output transducer, after the measurement has been made. The net result of this two part compensation is: $\mathbf{EQ.H_0.C_0} = 1$
- **C1** is used to compensate for the input transducer: $\mathbf{H_1.C_1} = 1$
- This way, **R** is the response of the DUT to the stimulus **S**
- $\mathbf{R = S.\ EQ.\ H_0.G.H_1.C_0.C_1 = G.\ S}$

All of the Correction Curves can be viewed in the Memory List by selecting an XY Display from the **Display** drop-down list. The following curves will be present for both Input and Output Signal Paths:

Inputs

- corr-in.dat
- sens-in.dat
- gain-in.dat

Outputs

- corr-out.dat
- eq-out.dat
- sens-out.dat

Input Tab

Input Signal Paths

Shown in *Figure 9-3*. These are made up of the following:

- The Calibrated Input Device
- The sensitivity of that specific device and its units
- The Gain of the specified Listen Hardware device

  See *Input Hardware Channel on Page 85* for more information.
- The type of calibration that is used for the device
- The hardware channel that it is connected to

The Signal Path names are used in any editor where "Use Signal Path Name" can be selected, e.g., in the Acquisition step, when "Use Signal Path Name" is selected, the Input name will be: "**Recorded Time Waveform [Input Signal Path Name]**".

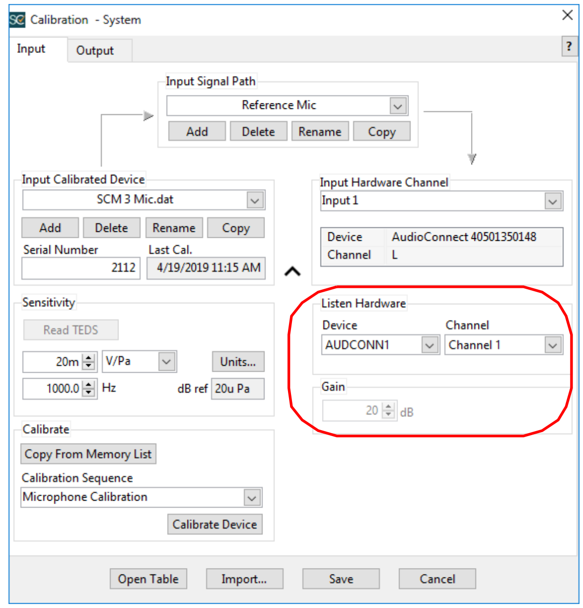
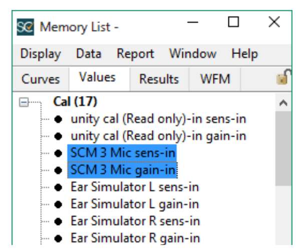


Figure 9-3: Input Channel Definition

Input Calibrated Device

The transducer or input device used for acquiring signal. This may be a Microphone, Direct audio interface input, USB or Bluetooth device, etc. The DAT file associated with this device is where the correction curves for that device are stored. The DAT file is created when a new Calibrated Device is added to the list.

Sensitivity and Gain



Calibration input sensitivity is divided into two parts:

- Sensitivity of the transducer
- Preamplifier gain

These values show up in the Memory List following the naming of the Calibrated Input Device, e.g.:

- SCM 3 Mic sens-in (Sensitivity)
- SCM 3 Mic gain-in (Preamp Gain)

Layout

- **Serial Number**: Allows you to enter the Serial Number of the specific devices connected
- **Sensitivity**: Shows the measured sensitivity that is acquired through the **Calibrate Device** process. This value can also be manually entered.
- **TEDS** - TEDS (with compatible Listen hardware) allows automatic update of calibration data for TEDS microphones and accelerometers
- **Units** - Input physical units and the dB reference are entered by clicking on **Units**. Refer to *Common Units for Inputs and Outputs on Page 90*.
- **Input Hardware Channel**: Only the connected device name and channel are shown
- **Listen Hardware**: When creating a new signal path, the channel selected in the **Input Hardware Channel drop down** automatically updates the device selected in Listen Hardware.
- **Gain**: The total gain in dB of the preamplifier the transducer is connected to. This value can be entered manually or automatically read from Listen hardware devices by selecting **Auto Dev and Auto Ch**.

Listen Hardware - Auto Device / Auto Channel

As of SoundCheck 18, fields have been added to support Listen Hardware devices such as AmpConnect, Audio Connect and SoundConnect 2.

This option requires that Listen Hardware is connected via USB and is selected in the **Device** drop-down list. SoundCheck automatically reads the gain from a **single channel** of USB-connected Listen Hardware and updates the Calibration Editor. This allows the preamp gain to be changed manually or through **Listen Hardware Message Steps** in a sequence. This can be used to optimize dynamic range without needing to re-calibrate the input path.

- **Auto Dev** - Select the Listen hardware device from the drop down list to associate with the Signal Path
- **Auto Ch** - Select the device channel from the drop down list to associate with the Signal Path
- As of SoundCheck 18, Auto Read has been removed from the Calibration Editor Input Tab

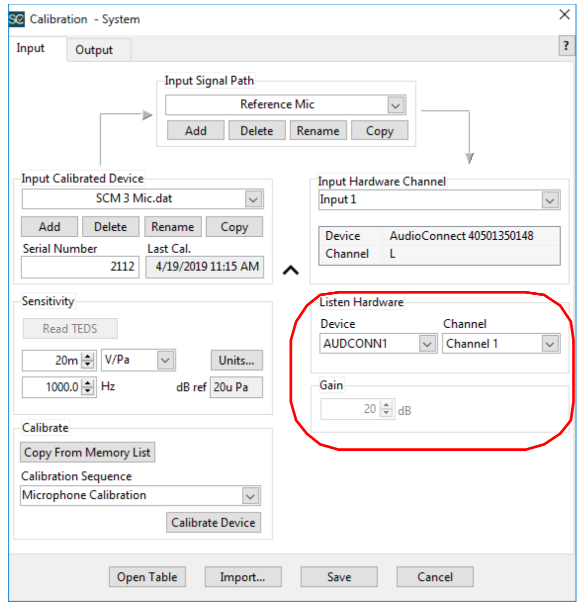
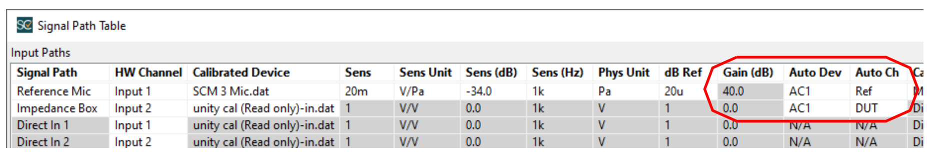


Figure 9-4: Listen Hardware - Auto Dev and Auto Ch


Signal Path Table

Input Paths

Signal Path	HW Channel	Calibrated Device	Sens	Sens Unit	Sens (dB)	Sens (Hz)	Phys Unit	dB Ref	Gain (dB)	Auto Dev	Auto Ch	Ca
Reference Mic	Input 1	SCM 3 Mic.dat	20m	V/Pa	-34.0	1k	Pa	20u	40.0	AC1	Ref	M
Impedance Box	Input 2	unity cal (Read only)-in.dat	1	V/V	0.0	1k	V	1	0.0	AC1	DUT	Di
Direct In 1	Input 1	unity cal (Read only)-in.dat	1	V/V	0.0	1k	V	1	0.0	N/A	N/A	Di
Direct In 2	Input 2	unity cal (Read only)-in.dat	1	V/V	0.0	1k	V	1	0.0	N/A	N/A	Di

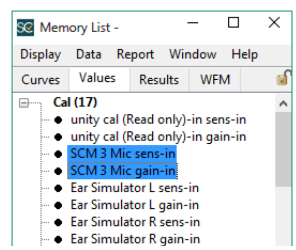
Sensitivity and Gain

Calibration input sensitivity is divided into two parts:

- Sensitivity of the transducer
- Preamplifier gain

These values show up in the Memory List following the naming of the Calibrated Input Device, e.g.:

- SCM 3 Mic sens-in (Sensitivity)
- SCM 3 Mic gain-in (Preamp Gain)



Signal Path Name Change

Changing an **Input Signal Path name** in the **Calibration Editor** affects all sequences that use that Signal Path Name.

This can occur when sending sequences to a SoundCheck system that uses different Signal Path Names or when modifying the Signal Paths on your SoundCheck system.

When opening a sequence that uses a changed signal path name, SoundCheck will prompt you as shown in ***Figure 9-5***. This allows you to add the missing signal paths or replace them with ones already present on the system.

Remember: If any Analysis Step is set to "Use Signal Path Name", subsequent steps in the sequence will need to be

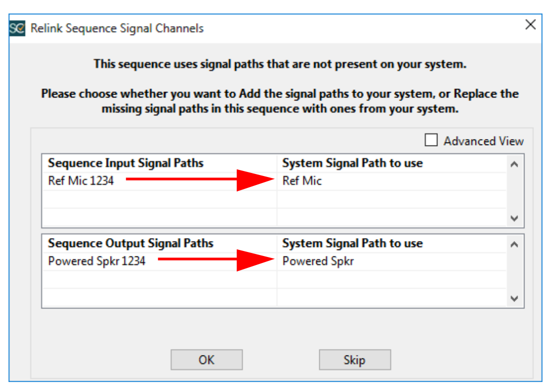


Figure 9-5: Relink Prompt

redirected to the new data names in the Memory List. This may also require that you rebuild Groups of data in the Memory List.

For recommendations on Signal Path naming see ***Naming - Best Practices on page 88.***

Input Hardware Channel

Allows you to select which channel of the audio interface the Calibrated Device is connected to. In this case the SCM 3 Mic is connected to Input 1 of the AudioConnect audio interface.

Copy from Memory List

Allows you to select a curve from the Memory List to be used as a Correction or Equalization curve

- The default Reference Frequency is 1 kHz as shown in ***Figure 9-6***.

Importing Correction Curves

If an imported curve does not have a 0dB value at 1 kHz you must change the calibration Reference Frequency in the editor to a point on the correction curve that is at 0dB.

This may also have to be done when importing Diffuse Field or other such correction curves without data at 1 kHz.

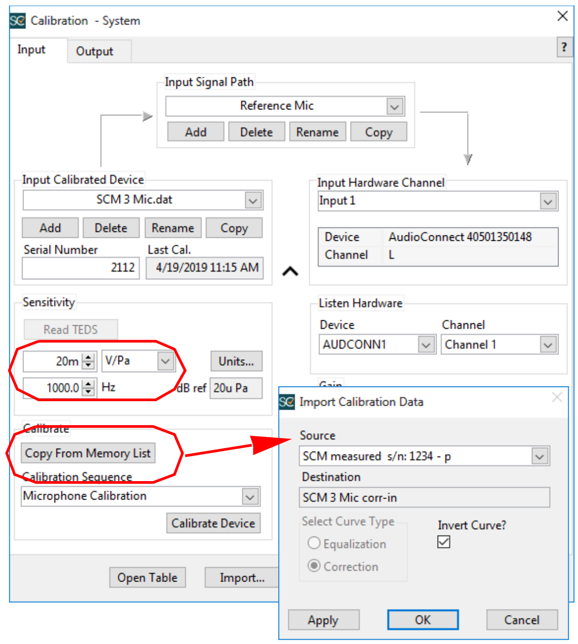


Figure 9-6: Copy From Memory List

Calibration Sequence

Select the sequence that runs when the Calibrate button is selected

Calibrate Device

Run the selected calibration sequence

Note: Calibration Sequences are never run on their own. They are always run from the Calibration Configuration Editor.

Open Table

Opens the Calibration Editor Table View. See ***Table View on Page 87*** for more information

Import

Allows you to import settings from SoundCheck sequences from previous versions or other systems. This imports settings for both Input and Output tabs.

Save

Stores calibration changes to hard disk

Cancel

Discard all changes made to the Calibration Configuration

Copy From Memory List - Input

This allows you to overwrite the **Calibrated Input Device** correction file with a new curve from the Memory List. Select **Invert Curve** if it is the fundamental response curve as opposed to a Reciprocal Curve.

See ***Figure 9-8: SCM 3 Correction Curve***.

Note: The EQ & correction curves will be normalized to 0dB at the frequency specified in the Sensitivity field [Sens(Hz)]. Units are automatically set to + dB re 1 so that pure gain is applied.

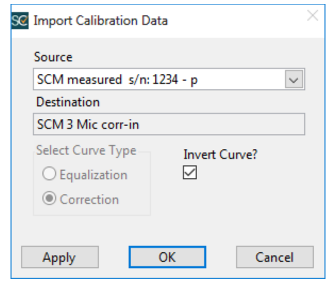


Figure 9-7: Copy From Memory List - Input

The curve in the example is from the DAT file that comes with an SCM 3 microphone. This is a response curve of the SCM 3 and is normalized to 0dB at 1 kHz. In this case, to create a reciprocal curve, **Invert Curve** must be checked.

Click **Apply** to overwrite the Destination Curve with the Source Curve selected. Click **OK** to leave the selection window and click **OK** to close the editor.

Subsequent measurements will apply this new correction curve to the input response, provided that “**Apply Correction In**“ has been checked in the Analysis Step. See ***Apply Correction on Page 162*** for more information.

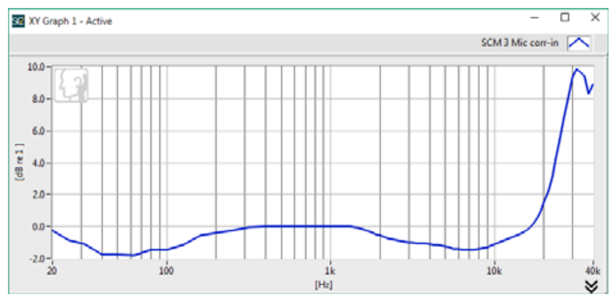


Figure 9-8: SCM 3 Correction Curve

Copy From Memory List - Output

Similarly, curves can be copied in the Output Tab as well. These can be used as correction curves or as equalization curves. For more information see ***Equalization and Correction Curve on Page 93***.

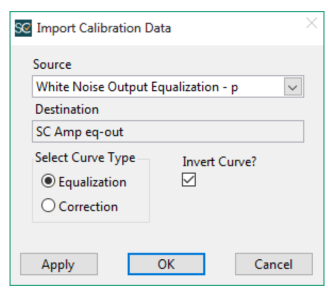


Figure 9-9: Copy From Memory List - Output

Output Tab

Output Signal Paths are made up of the following:

- The Calibrated Output Device
- The sensitivity of that specific device and its units
- The hardware channel that it is connected to
- The type of calibration that is used for the device

The **Calibration Sequence** selected is the sequence that runs when the Calibrate button is selected.

The functions in the Output Tab are identical to those in the Input Tab with one addition:

- **Input Signal Path** - Must be selected so that the system knows where the Output Device is connected to
- The functions of Open Table, Import, Save and Cancel are the same as the Input Tab.

  See ***Input Tab on Page 83*** for more information.

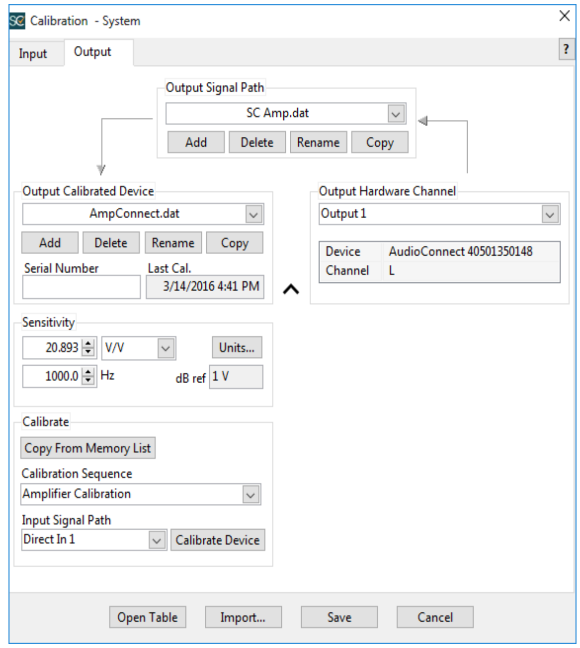
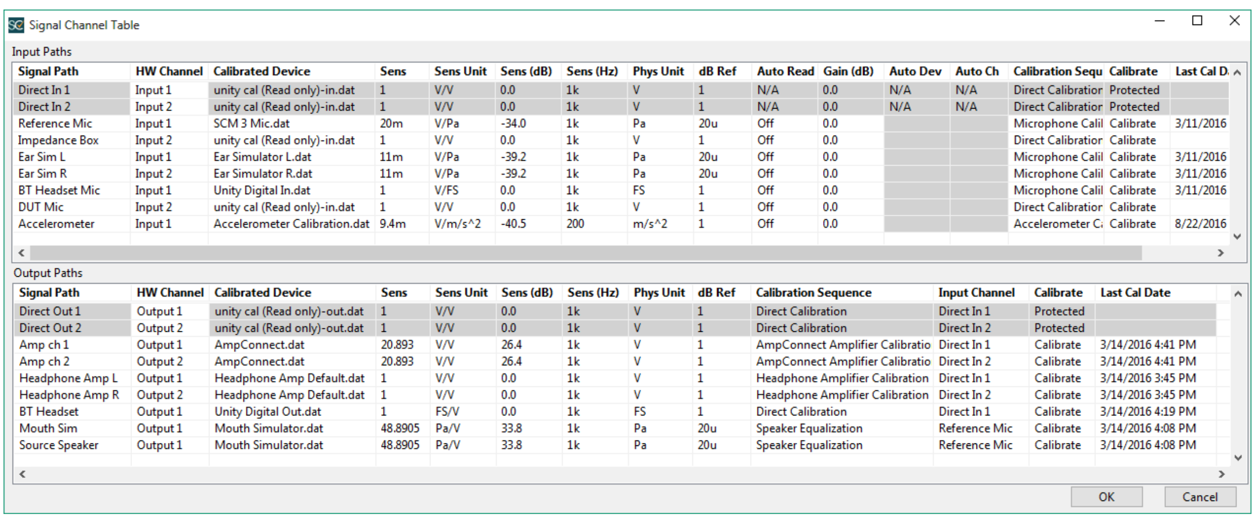


Figure 9-10: Output Channel Definition

Table View



Signal Channel Table

Input Paths

Signal Path	HW Channel	Calibrated Device	Sens	Sens Unit	Sens (dB)	Sens (Hz)	Phys Unit	dB Ref	Auto Read	Gain (dB)	Auto Dev	Auto Ch	Calibration Sequ	Calibrate	Last Cal D.
Direct In 1	Input 1	unity cal (Read only)-in.dat	1	V/V	0.0	1k	V	1	N/A	0.0	N/A	N/A	Direct Calibratior	Protected	
Direct In 2	Input 2	unity cal (Read only)-in.dat	1	V/V	0.0	1k	V	1	N/A	0.0	N/A	N/A	Direct Calibratior	Protected	
Reference Mic	Input 1	SCM 3 Mic.dat	20m	V/Pa	-34.0	1k	Pa	20u	Off	0.0			Microphone Calil	Calibrate	3/11/2016
Impedance Box	Input 2	unity cal (Read only)-in.dat	1	V/V	0.0	1k	V	1	Off	0.0			Direct Calibratior	Calibrate	
Ear Sim L	Input 1	Ear Simulator L.dat	11m	V/Pa	-39.2	1k	Pa	20u	Off	0.0			Microphone Calil	Calibrate	3/11/2016
Ear Sim R	Input 2	Ear Simulator R.dat	11m	V/Pa	-39.2	1k	Pa	20u	Off	0.0			Microphone Calil	Calibrate	3/11/2016
BT Headset Mic	Input 1	Unity Digital In.dat	1	V/FS	0.0	1k	FS	1	Off	0.0			Microphone Calil	Calibrate	3/11/2016
DUT Mic	Input 2	unity cal (Read only)-in.dat	1	V/V	0.0	1k	V	1	Off	0.0			Direct Calibratior	Calibrate	
Accelerometer	Input 1	Accelerometer Calibration.dat	9.4m	V/m/s^2	-40.5	200	m/s^2	1	Off	0.0			Accelerometer C:	Calibrate	8/22/2016

Output Paths

Signal Path	HW Channel	Calibrated Device	Sens	Sens Unit	Sens (dB)	Sens (Hz)	Phys Unit	dB Ref	Calibration Sequence	Input Channel	Calibrate	Last Cal Date
Direct Out 1	Output 1	unity cal (Read only)-out.dat	1	V/V	0.0	1k	V	1	Direct Calibration	Direct In 1	Protected	
Direct Out 2	Output 2	unity cal (Read only)-out.dat	1	V/V	0.0	1k	V	1	Direct Calibration	Direct In 2	Protected	
Amp ch 1	Output 1	AmpConnect.dat	20.893	V/V	26.4	1k	V	1	AmpConnect Amplifier Calibratio	Direct In 1	Calibrate	3/14/2016 4:41 PM
Amp ch 2	Output 2	AmpConnect.dat	20.893	V/V	26.4	1k	V	1	AmpConnect Amplifier Calibratio	Direct In 2	Calibrate	3/14/2016 4:41 PM
Headphone Amp L	Output 1	Headphone Amp Default.dat	1	V/V	0.0	1k	V	1	Headphone Amplifier Calibration	Direct In 1	Calibrate	3/14/2016 3:45 PM
Headphone Amp R	Output 2	Headphone Amp Default.dat	1	V/V	0.0	1k	V	1	Headphone Amplifier Calibration	Direct In 2	Calibrate	3/14/2016 3:45 PM
BT Headset	Output 1	Unity Digital Out.dat	1	FS/V	0.0	1k	FS	1	Direct Calibration	Direct In 1	Calibrate	3/14/2016 4:19 PM
Mouth Sim	Output 1	Mouth Simulator.dat	48.8905	Pa/V	33.8	1k	Pa	20u	Speaker Equalization	Reference Mic	Calibrate	3/14/2016 4:08 PM
Source Speaker	Output 1	Mouth Simulator.dat	48.8905	Pa/V	33.8	1k	Pa	20u	Speaker Equalization	Reference Mic	Calibrate	3/14/2016 4:08 PM

OK Cancel

Figure 9-11: Table View

Table View at the bottom of the editor to open the complete list of Signal Paths. The "Last Cal Date" field cannot be edited. This updates when a new calibration is run on a Signal Path. Similarly the "Sens Unit" field can only be changed by selecting a new setting under "Physical Unit".

The following fields can be edited directly:

- Signal Path
- Sens
- Sens (dB)
- Sens (Hz)
- dB Ref

The following fields have drop down selection menus:

- Calibrated Device
- Phys Unit - Button opens Units Setup menu
- HW Channel
- Calibration Sequence
- Input Channel (Output Channels Only)
- Calibrate (Button)

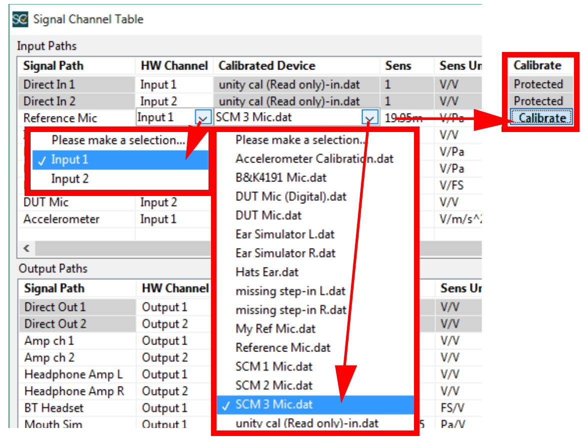


Figure 9-12: Table Drop Down Example

Click **Calibrate** to run the selected calibration sequence. See ***Figure 9-12***. Remember that the Input Signal Path must be selected prior to running the output calibration.

Note: All calibration procedures are specialized sequences located in the folder *C:\SoundCheck 18.1\Sequences\Calibration*. You can create your own calibration sequences as well. If you save the new calibration sequence in the calibration input or output sub-folders, it will appear in the respective drop-down list in the Calibration Editor. See ***Reference Codec & dBm0 on Page 106*** for more information.

Naming - Best Practices

Signal Path - This name is used in the sequence (i.e.; Analysis Step - The Channel name can be appended to the output curve name). A generic name should be used that indicates the type of device being used. Since the name is used in the sequence, it is hard to change once the sequence is created. Changing this name could effect how subsequent steps in the sequence operate.

Calibrated Device - Specific name of the device being used (i.e.; type of microphone). It is also the name of the .DAT file for the specific calibration data. This can be changed easily since the name is not used in sequence steps. This allows you to have several different possible devices to use under the generic Signal Path name.

Hardware Channel - This is the name of the physical channel of the audio interface. In this example, the Ref Mic is an SCM 3 which is connected to Input 1 of the audio interface. (This is named in the System Hardware Configuration, e.g., Input 1).

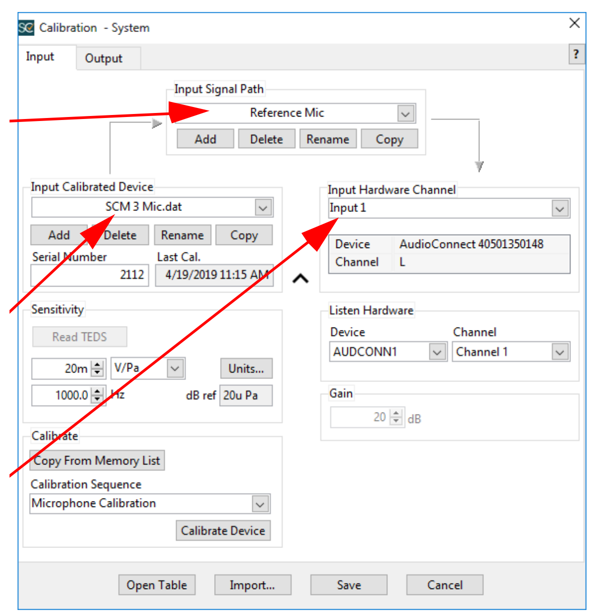


Figure 9-13: Naming Convention

Right-click Function

Right-click the table to open the Channel Modification window. With this, you can Add, Delete Signal Paths to the table as well as Copy a channel to a new row of the table.

Rules

- Calibrated Device .DAT files marked as “Read Only” are displayed grayed out and cannot be modified. *Figure 9-14* shows items with “Gray” fields indicating the properties of the .DAT files are set to “Read Only”. The files shown are set to “Read Only” by default.
- DAT files for devices that are calibrated should never be set to “Read Only”, e.g.: Reference Mic, Mouth Sim, Amp Ch 1, etc.

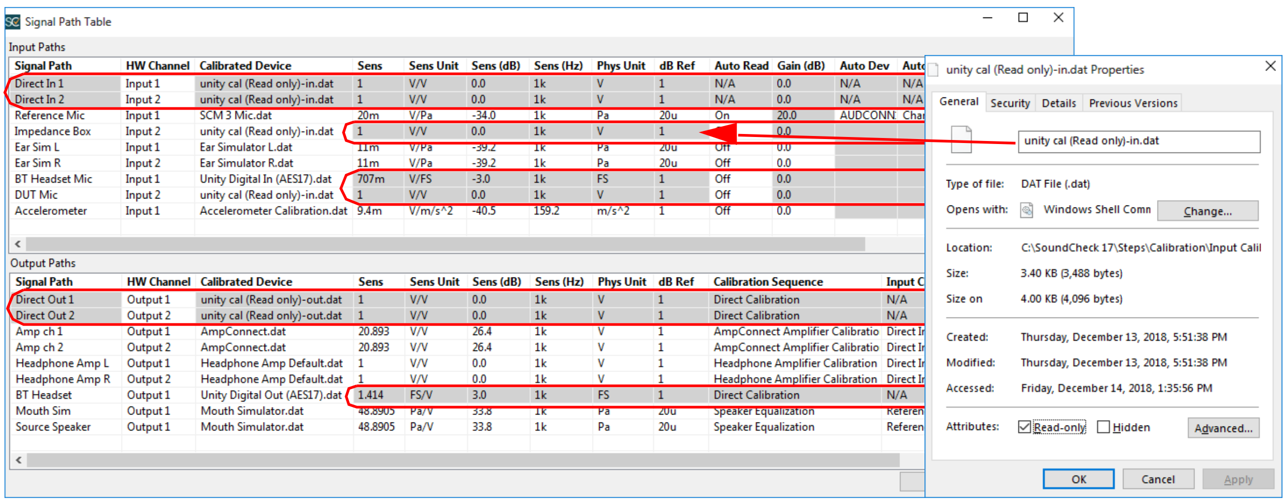


Figure 9-14: Read Only Files

Defining the Units

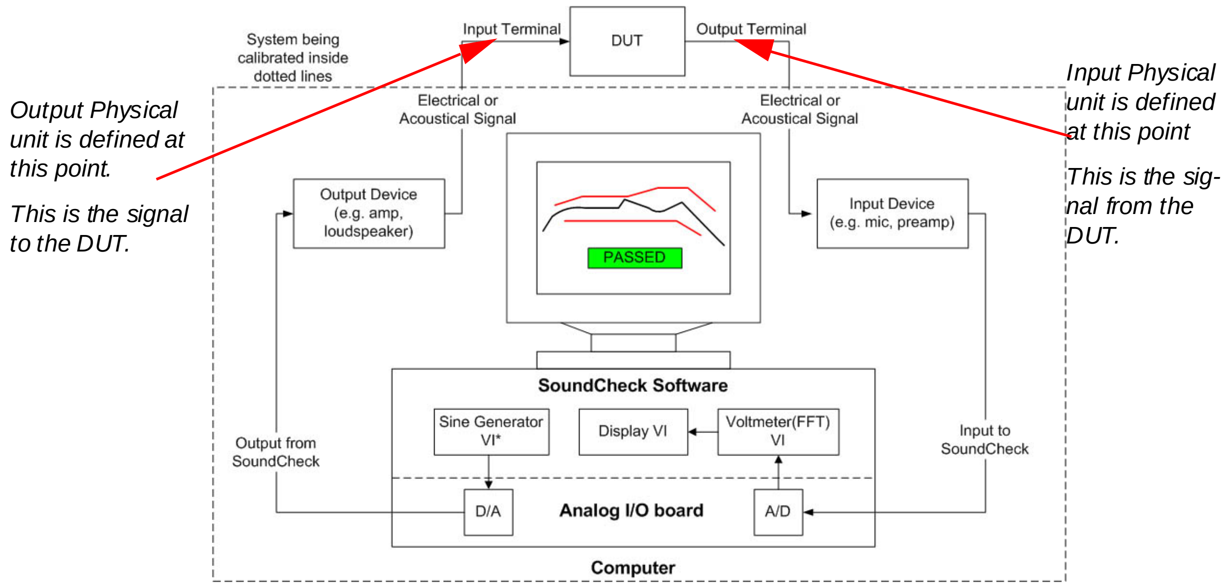


Figure 9-15: Input and Output Calibration of a Measurement System

The ***Figure 9-15*** shows the Input and Output Calibration of a Measurement System and Definition of the Input and the Output Terminals of the DUT.

- The Physical Units are specifically related to each Calibrated Device in the Calibration Editor.
- They are dependent on the type of device that is used in the input or output of the calibration.

DUT Type	**Input: Signal from DUT**		**Output: Signal to DUT**		**Response Unit Examples**
	Unit	**Input Signal Conditioning**	**Unit**	**Output Signal Conditioning**	
Loudspeaker	Pa	Microphone	V	Amplifier	Pa/V
Microphone	V	Microphone (Itself)	Pa	Mouth or Anechoic Chamber	V/Pa
Amplifier, analog electronics	V	Direct	V	Amplifier (no longer Direct In)	V/V
Motor, Fan, Bearings, etc.	G	Accelerometer	V	DUT Itself or Shaker	G/V
Haptic Sensor	N	Force Transducer	V	DUT Itself	V/N
Hearing Aid	Pa	Microphone (in coupler)	Pa	Anechoic Chamber	Pa/Pa
Telephone	Pa	Microphone (in coupler)	Pa	Mouth	Pa/Pa

Figure 9-16: Common Units for Inputs and Outputs

The Calibration Sequence selected determines what type of calibration sequence is performed after selecting **Calibrate**.

The ratio of the audio interface signal to the Device Under Test Unit is automatically determined by SoundCheck. You are required to determine the units of the DUT in the Calibration. SoundCheck uses the physical units when displaying measurement data.

Define Linear and Logarithmic Units

Linear units are absolute (e.g., Voltage, Watt, Pascal, G). Logarithmic units are relative to a reference level (e.g., dB re 20µPa). For example:

$$dBSPL = 20\log\frac{?\,Pa}{20\,\mu Pa}$$

$$dBV = 20\log\frac{?V}{1V}$$

Convert Linear to Logarithmic Units

To convert linear units to logarithmic units, the following examples are useful references:

$$0\,dBV = 20\log\frac{1V}{1V}$$

$$94\,dBSPL = 20\log\frac{1\,Pa}{20\,\mu Pa}$$

Calibrating SoundCheck

Rules

- Do not open the Calibration Sequences found in the *Calibration* folder. These sequences are accessed from the *Calibration Editor* when performing a calibration.
- Close your sequence before running a calibration process, including calibrating audio interface channels in the Hardware Editor.

The calibration of SoundCheck enables response measurements to be performed directly in terms of the input and output terminals of the Device Under Test (DUT). The **Calibrate Device** function in the **System Calibration Configuration** measures the sensitivity or gain of any external device such as a microphone or amplifier in the measurement chain. Once this is done, the values and units in the **Calibration Setup** menu correspond to the signal level at the DUT and not at the connectors of the audio interface in the computer. Default calibration sequences are included with SoundCheck. You can also create custom calibration sequences. Please refer to ***Reference Codec & dBm0 on Page 106*** for more information.

- Sensitivity is a relative measurement of output to input. By definition, an electroacoustic transducer converts either voltage into acoustical output (e.g., loudspeaker); acoustical input into an electrical voltage (e.g., microphone); or both, an acoustical input into electrical voltage and back again to an acoustical output (e.g., hearing aid). As another example, analog electronics, which usually amplify, attenuate, or shape the electrical signal (e.g., Preamplifiers, Amplifiers or Signal Processors).
- Calibrating the Input establishes the correlation between the input voltage to the audio interface and the measured units. Input sensitivity is represented as Volts per measured unit (e.g., V/Pa).
- Calibrating the Output establishes the correlation between the output voltage of the audio interface and the measured units. Output sensitivity is represented as measured units per Volt (e.g., Pa/V).

Note: The units selected for calibration can be redefined when displaying data and results. Select units for displaying data in the *Analysis Editor* (See *Units on Page 164*). In addition, the Device (e.g., microphone or amp) Sensitivity is measured when **Calibrate Device** is clicked.

Note: We recommend that you Not use periods or commas in step names. This is known to cause a problem with **System Calibration Configuration** not saving the calibration information when the configuration is saved.

Note: Upon completing a calibration sequence, the newly acquired date is stored to the calibrated device file. Any Signal Paths that are set to this calibrated device will use the updated data.

Calibration History

This feature allows you to view past calibration data for a device. This data can be used to identify trends such as “changing sensitivity”. The history is stored in the .DAT file for the calibrated device. Each time you recalibrate and save the **System Calibration Configuration**, a new entry is created in the calibration history. Theses entries are tracked by calibration date (e.g., SCM 2 Mic corr-in 9/17/2008 10:16 AM). The history can be viewed by opening the **Memory List**, selecting **File** and then **Open Data**. Browse to the calibrated device **.DAT** file (e.g., SCM 2 Mic.dat), select and click **OK**. The calibration history is then loaded into the Memory List.

Digital Signals

When an audio interface channel is set to digital in the *Hardware Editor*, the Units for that channel in the *Calibration Editor* will change such that V (Volts) is replaced by FS (Full Scale). This is because the input/output values of the audio interface are expressed relative to 100% Full Scale Deflection instead of Volts. The choice of physical unit remains the same. Sensitivities are expressed in Unit/FS for output or FS/Unit for inputs.

The input and output of the audio interface are normalized to 100% FS. Any digital signal cannot exceed +1/-1 FS, where 1 represents the maximum number of bits selected in the System Hardware Configuration.

Please refer to ***Reference Codec & dBm0 on Page 106*** for more information.

Equalization and Correction Curve

Correction vs Equalization can be a little tricky to understand at first. Let's start with the input side.

The input correction will typically be a correction curve for a reference microphone. This would be utilized when performing loudspeaker measurements or when that reference mic is being used to calibrate a speaker for microphone measurements.

All output calibrated devices have two curves, the correction and the EQ. The EQ is only used in cases where you want to change the stimulus before it gets played, e.g.: equalizing a loudspeaker. EQ (if selected in stimulus) will modify the voltage that gets sent out of the audio interface on a per frequency basis in accordance with the curve. If calibrated correctly, this results in a flat acoustical output from the speaker.

The output correction curve is optionally applied in the analysis step as a sort of post processing operation. This is a mathematical correction that is performed after the signal has already been played and acquired. If you have equalized a speaker you will find that the correction is very small. It is just the residual part of the speaker response that the calibration sequence could not perfectly flatten. To view this look in your memory list and graph the eq-out and corr-out curves for a given calibrated device.

The time when the output correction is more important is in loudspeaker testing. In this case we are calibrating an amplifier. Because the amplifier has a flat magnitude response there is no need to use equalization. However, we do want to account for the amp's phase response, which we do by generating the correction.

Input Signal Path

The **System Calibration Configuration** loads a complete set of EQ and correction curves into memory from the calibration folders. When a Calibration Sequence is run, from the **System Calibration Configuration**, these curves are updated. These curves are used to correct for the response of devices in the input or output signal chain. The correction curves can be displayed since they are selectable items in the *Memory List*.

An example of this is to correct the measured signal for the measurement microphone’s frequency and phase response (typically on its calibration chart) or for telephone measurements, where the DRP to ERP correction curve is needed to compensate for the microphone’s position in the artificial ear. The correction file is named according to the Calibrated Device:

<Calibrated Device Name> corr-in.dat for Input Signal Path correction

See ***Copy From Memory List - Input on Page 86*** for more information.

Output Signal Path

An equalization curve will equalize the stimulus when the EQ check box is selected in the *Stimulus Editor* or *Signal Generator*.

See ***Stimulus Editor on Page 113***.

The output correction curve is applied when it is selected in the *Analysis Editor*.

The Output Signal Path correction and equalization files are named according to the Calibrated Device:

<Calibrated Device Name> corr-out.dat for Output Signal Path correction

<Calibrated Device Name> eq-out.dat for Output Signal Path equalization

See ***Copy From Memory List - Output on Page 86*** for more information.

The files are updated when an Output calibration is run and the step is saved.

If you Import a frequency response curve to use for Correction or Equalization you have to "Invert" it by selecting **Invert Curve** as shown in ***Figure 9-17***. If the curve is the result of a process that creates a reciprocal of the response, it will not need to be inverted.

Note: EQ out correction curves are populated with data when the Speaker Equalization or Simulated Free Field calibration sequences are selected in the output calibration process.

Important! Correction Curve Units should be set to + dB re 1 so that pure gain is applied.

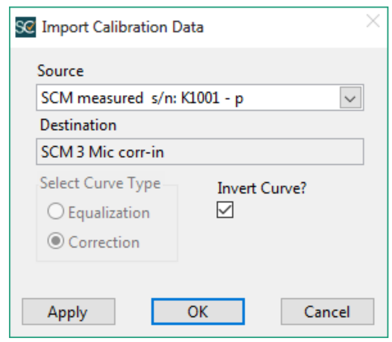


Figure 9-17: Copy From Memory List - Correction Curve

SI units are used throughout *SoundCheck* 18.1. For example, to display decibel values referenced to 20 microPascals, the dB ref value would be twenty (20) with a "**u**" added at the end and the unit would be Pa resulting in 20 µPa. See ***SI Units on Page 55*** for more information.

For output calibration, the measured response will automatically be stored and used to correct any future measurements. (e.g., if the amplifier's magnitude and phase responses are not perfectly flat, the system will correct the measured response as if it were perfectly flat.)

Input Calibration

Direct Calibration Sequence

Direct refers to the input of the audio interface. Direct Calibration should be used when there is no signal conditioning between the device under test (DUT) and the audio interface. For example, when measuring electronics such as amplifiers, the direct input sensitivity should be set to 1 V/V or 0 dBV.

Note: Direct Calibration cannot be done on the default input channels Direct In 1 and 2, which are protected. Create a new Direct In channel that uses a new Unity Gain **Calibrated Input Device** as shown in ***Figure 9-18***.

Clicking **Calibrate Device** will bring up a *Multimeter* virtual instrument. Check the input calibration of the test system by applying a known signal source, e.g., 1 VRMS at 1 kHz from an external signal generator, into the audio interface. You should read the same level, e.g., 1 volt, on the SoundCheck *Multimeter* virtual instrument.

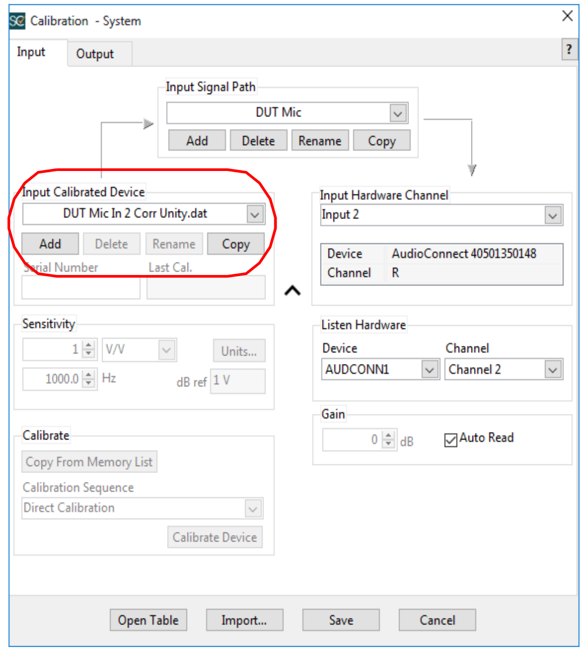


Figure 9-18: Direct In Calibration

Note: Please be aware that the input impedance of the audio interface and the output impedance of the signal generator can affect the reading.

Attenuator Calibration Sequence

The Attenuator setting should be used when there is an attenuator or pre-amplifier between the DUT and the input of an audio interface. For example, this would be used to optimize the signal-to-noise ratio of a measurement when using a measuring amplifier with gain and/or attenuation, such as the Listen SoundConnect™ microphone power supply. This allows for a wider dynamic range of test levels and a better match to the input range of the audio interface.

Clicking **Calibrate** will open the manually-controlled *Signal Generator* and *Multimeter* instruments. Use these virtual instruments to measure the input gain or attenuation of your measuring amplifier.

Microphone Calibration Sequence

Use the Microphone setting when measuring the sound pressure level from the DUT (e.g., earphone or mouth simulator) with a measurement microphone, e.g., SCM 3 microphone.

Enter the correct sensitivity from the calibration chart or click **Calibrate** to measure its sensitivity with an acoustic calibrator. This is the preferred method since it takes into account the entire Input Signal Path including signal conditioning from the microphone power supply.

Accelerometer Calibration Sequence

Used when making measurements with an accelerometer or other vibration transducer, e.g.: B&K 4533-B, where force units are used. dB re 1 m/s2, dB re 1 N and dB re 1 g are currently supported.

As with Microphone Calibration above, the sensitivity can be entered manually or click **Calibrate** to measure its sensitivity with an accelerometer calibrator such as the B&K 4294.

Microphone Calibration Procedure

This procedure will allow you to check your measured microphone’s sensitivity against the microphone manufacturer’s specifications.

1. Enter the gain or attenuation in dB that corresponds to the settings on your microphone power supply or measuring amplifier in the **Gain** field. This gain will be used in the Gain field of the Transducer Calibration window.

   If you are using Listen Hardware you can select **Auto Dev** and **Auto Ch** to automatically get the gain for the selected channel from the device. Using **Auto Dev** and **Auto Ch** in Calibration allows you to change the gain of the input without having to re-calibrate the mic. Current supported hardware includes AudioConnect and SoundConnect 2.

   Figure 9-19 shows AudioConnect is set to 20 dB of gain on Channel 1. This gain is automatically updated in the Transducer Calibration window.

   If you are using a Brüel & Kjær Nexus, see ***Calibrating using a Brüel & Kjær Nexus on page 97***.

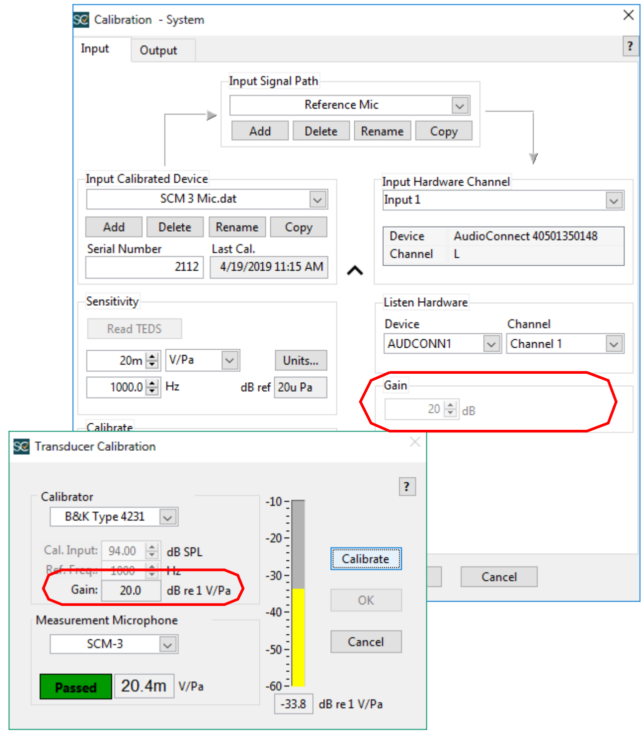


Figure 9-19: Microphone Calibration

2. Open the *Calibration Editor* and select the Input tab. The **Calibration Sequence** should be set to **Microphone**.

3. Click **Calibrate**

4. Select your calibrator model # from the drop-down list or select **Other Calibrator** and enter the acoustic calibrator’s reference level and frequency.

- The microphone calibrator's reference level should be indicated in its specifications as a given dB SPL value (relative to 20µPa) at a reference frequency.
- e.g., for the Brüel & Kjær Type 4231 Acoustic Calibrator:
  - Sound Pressure Level: 94.00 dB ± 0.20 dB
  - Frequency: 1000 Hz ± 0.1%
  - (When calibrating an artificial ear mic, you may need to select “B&K 4231 and UA1546”.)

5. Select your measurement microphone model number from the drop-down list. If your microphone is not listed in the drop-down list, choose **Add New Mic**.

6. Place your acoustic calibrator on your reference microphone and click **Calibrate** to measure its sensitivity.

   - The measured sensitivity of your reference microphone is displayed under **Measured Sensitivity** in mV/Pa. If the measured sensitivity is outside the manufacturer’s specifications, a flashing FAILED message will appear. Check first to see if your connections are correct or if the calibrator is turned on before assuming something is wrong with the microphone.

- The FAILED message can also appear if the Calibrator's frequency is not correct. If a Reference Frequency of 1000 Hz is entered, but the Calibrator's actual frequency is 1008 Hz, the Calibration may Fail. To verify the Calibrator's frequency, use the *Spectrum Analyzer* under the **Instruments** menu on the SoundCheck Main Screen.
- The meter on the right side indicates the corresponding dB level relative to 1 Volt per Pascal. If it varies by a few tenths of a dB from your last calibration measurement, do not be alarmed, this is normal. If it varies by more than 1 dB or failed the sensitivity test, you may want to have your microphone checked by a qualified calibration lab.

Add New Mic

This allows you to enter a different microphone into the list. The sensitivity limits can be entered for each new mic so that calibrations are made according to the microphone's specifications.

- The check box for **1/2" Free Field Microphone** should be checked only when using this type of microphone

  The calibration level is set to 93.85 dB SPL @ 1 kHz when measured in a pressure field, (such as a B&K 4231 Acoustical Calibrator).

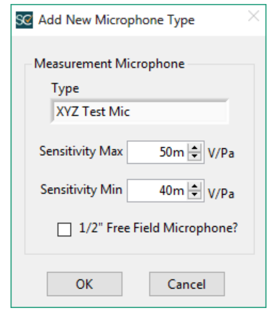


Figure 9-20: Add New Mic

Calibrating using a Brüel & Kjær Nexus

Method 1: Nexus Unity Gain

- Set the Microphone Input Sensitivity of the Nexus to 100 mV/Pa (or other appropriate input value)
- Set the Nexus Output to the same level: 100 mV/Pa
- As long as the input and output levels of Nexus are the same, the **Gain** value for the SoundCheck Microphone Calibration Editor is 0 dB
- Changes in the Nexus output setting can be easily converted to Gain in dB. A change of a factor 10 in the Nexus output is equivalent to an increase of 20 dB, e.g.: 100 mV/Pa to 1 V/Pa = 20 dB of gain.

Method 2

The Brüel & Kjær Nexus Type 2690 is designed to provide an output voltage regulated in 10 dB steps (e.g., 100 mV/unit, 316 mV/unit, etc.). To ensure proper calibration using SoundCheck, you must do the following to enter the proper **Gain** field value:

1. Enter the transducer sensitivity in Nexus per the Brüel & Kjær instructions.
2. Choose the Nexus output level you want (e.g., 1.00 volt per Pascal).
3. Enter the Gain value in SoundCheck using the following equation:

$$Gain = 20 Log\left(\frac{NexusOutputVoltage}{TransducerSensitivity}\right)$$

Example:

A B&K microphone is used with a sensitivity of 50 mV/Pa and the Nexus is set to an output of 1.00 V/Pa.

$$Gain = 20 Log\left(\frac{1.00V}{0.050V}\right)$$

Accelerometer Calibration Procedure

This procedure will allow you to check your measured accelerometer's sensitivity against the accelerometer manufacturer's specifications.

1. Enter the gain or attenuation in dB that corresponds to the settings on your microphone power supply or measuring amplifier in the **Gain** field. This gain will be used in the Gain field of the Transducer Calibration window.

   If you are using Listen Hardware you can select **Auto Dev** and **Auto Ch** to automatically get the gain for the selected channel from the device. Using **Auto Dev** and **Auto Ch** in Calibration allows you to change the gain of the input without having to re-calibrate the mic. Current supported hardware includes AudioConnect and SoundConnect 2.

2. Open the *Calibration Editor* and select the Input tab. The **Calibration Sequence** should be set to **Accelerometer**.

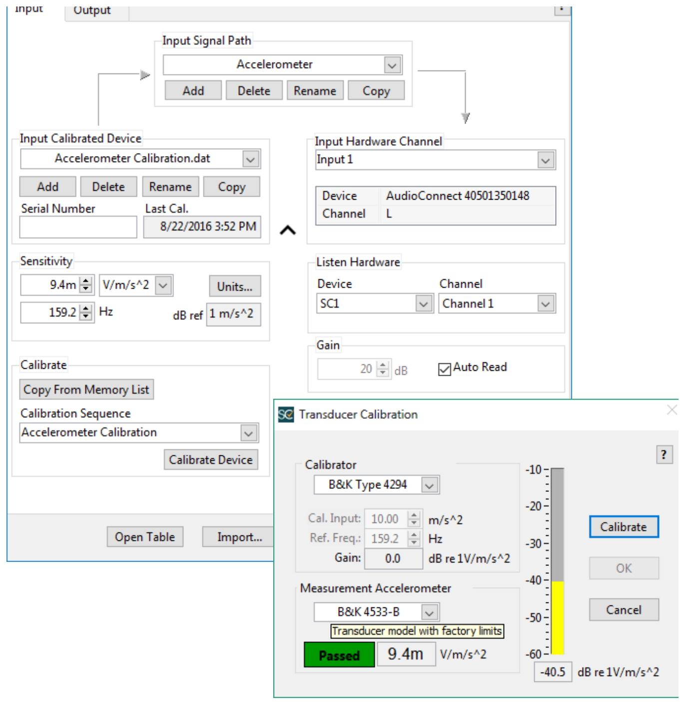


Figure 9-21: Accelerometer Calibration

3. Select your calibrator model # from the drop-down list or select **Other Calibrator** and enter the calibrator's reference level and frequency.

- The accelerometer calibrator's reference level should be indicated in its specifications as a given V/m/s^2, at a reference frequency, e.g.: for the Brüel & Kjær Type 4294 Calibrator:
  - Cal Input: 10 m/s^2
  - Frequency: 159.2 Hz

4. Select your measurement accelerometer model number from the drop-down list. If your accelerometer is not listed in the drop-down list, choose **Add New Accelerometer**. The add process is the same as shown in ***Add New Mic on Page 97***.

5. Mount the accelerometer on the accelerometer calibrator according to the manufacturer's instructions. Click **Calibrate** to measure the sensitivity.

   - The measured sensitivity of your reference accelerometer is displayed under **Measured Sensitivity** in V/m/s^2. If the measured sensitivity is outside the manufacturer's specifications, a flashing FAILED message will appear. Check first to see if your connections are correct or if the calibrator is turned on before assuming something is wrong with the accelerometer.
   - The FAILED message can also appear if the Calibrator's frequency is not correct. If a Reference Frequency of 159.2 Hz is entered, but the Calibrator's actual frequency is 162 Hz, the calibration may Fail. To verify the Calibrator's frequency, use the *Spectrum Analyzer* under the **Instruments** menu on the SoundCheck Main Screen.
   - The meter on the right side indicates the corresponding dB level relative to 1 V/m/s^2. If it varies by a few tenths of a dB from your last calibration measurement, do not be alarmed, this is normal. If it varies by more than 1 dB or failed the sensitivity test, you may want to have your accelerometer checked by a qualified calibration lab.

Output Calibration

Amplifier

The Amplifier setting should be used when there is an amplifier between the output of the audio interface and the device under test. This might be required for loudspeaker measurements, to drive difficult loads (e.g., low impedance devices) or to test at levels above 2 VRMS.

Enter the gain of your amplifier or run the Calibration test to measure it. If you are going to measure it, make sure your input is calibrated first, and then follow the Amplifier Calibration Procedure (below).

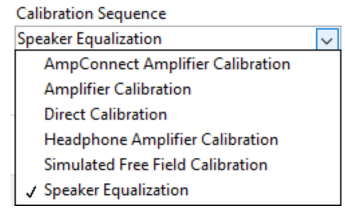


Figure 9-22: Output Calibration Sequence List

Important! For Amplifier Calibration, select an Input Signal Path in the *Calibration Editor* that is set to 1 V/V, to get an accurate calibration.

Important! Note that the AmpConnect Amplifier Calibration sequence must be used to correctly calibrate AmpConnect ISC.

Headphone Amplifier Calibration Procedure

The procedure is the same as noted below, except that the sequence used is called **Headphone Amplifier Calibration** and the wiring requirements are different. Please refer to the AudioConnect and AmpConnect ISC manuals for wiring examples.

Amplifier Calibration Procedure

A general wiring diagram and outline of the steps for amp calibration can be found in ***Connection Procedures on page 571***.

1. In SoundCheck, open the System Calibration Editor from the **Setup** drop-down list on the SoundCheck Main Screen.

Important! Do not open the Amplifier Calibration sequence. Run the calibration from the System Calibration Editor.

---

2. Select the Output Tab and select the Output Signal Path to calibrate. In the example in ***Figure 9-23*** "**Amp ch 1**" is selected.

3. Select the proper Calibrated Device: "**Crown Amp-L.dat**"

   If a new device is used with the system, click **Add** and enter a name. Calibration values will be stored for this new device when the calibration process is complete.

4. Select the Output Hardware Channel that provides signal to the Amplifier Input: "Output 1"

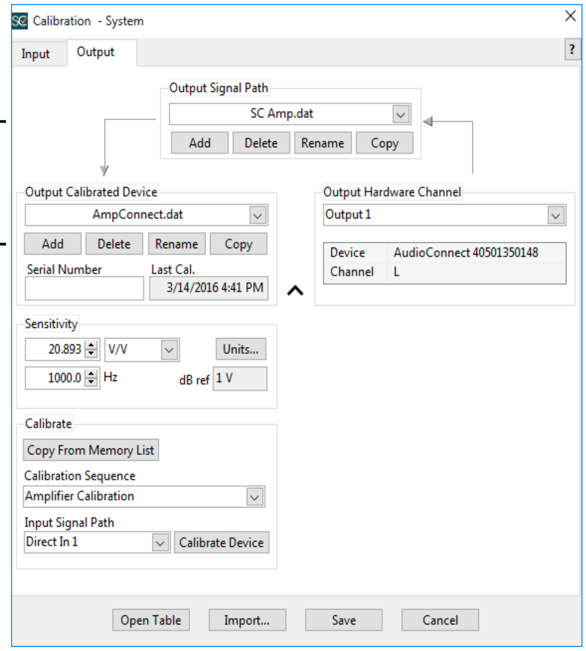


Figure 9-23: Amp Calibration

5. Calibration Sequence should be set to "**Amplifier Calibration**"

6. The Input Signal Path should be set to a Direct Input that is set for Unity Gain. The sensitivity of this channel must be unity gain in order to get an accurate calibration.

7. **Turn the power amplifier off.**

8. Connect SoundCheck **Output 1** to the input of the Amplifier.

9. Connect the corresponding output of the amplifier to SoundCheck **Direct In 1**. The amplifier output should not have an added Load.

10. Turn the power amplifier on. If it has a gain control, set it to maximum. This is the most stable position for a gain control.

Note: Some applications may require a lower gain amp. In that case, setting the volume control to a lower level is acceptable, but less stable. If anyone bumps the control, the calibration will be off.

---

11. Click **Calibrate** to measure the amplifier's sensitivity (gain) and frequency response.

    The measured sensitivity of your amplifier is automatically entered in the output sensitivity field. If the measured sensitivity fails, check your wiring and connections and try calibrating again.

    If the measured response margin fails, check to see that the amplifier is not connected to anything other than the audio interface and that it is properly grounded. If there is a bump around 120 Hz (or 100 Hz if line frequency is 50 Hz), you might be picking up hum due to poor grounding or bad cabling.

    For more troubleshooting information, go to www.listeninc.com and click **Support**.

Direct

This is for calibrating low gain electronics with a sensitivity range between 700 mV/V and 1.4 V/V.

This creates a Correction Out curve for the device which can be used in Analysis Steps to correct for the output response of the device, e.g.: correcting for the frequency response of an audio interface or transducer preamp.

Headphone Amp

The **Headphone Amp** calibration process is the same as **Amplifier** calibration. The Headphone Amp sequence is setup to account for the lower sensitivity typical of headphone amplifiers. These amps may have no gain or negative gain.

Simulated Free Field Calibration

This is required when using Frequency Log Sweep Stimulus and Time Selective Response Analysis.

Speaker Equalization

This calibration sequence prompts you to input a Stimulus Level, in dB, for the calibration signal. Usually this is the level that will be used in the actual test sequence.

The sequence will Loop to "Fine Tune" the equalization curve and output correction.

Use when calibrating one of the following devices:

- Mouth Simulator
- Anechoic Chamber/speaker
- Anechoic Test Box

Note: ***Speaker Equalization must be done at the same resolution or higher than the stimulus resolution of test sequence. This can be done by making a copy of the Speaker EQ sequence and changing the resolution of the 2 stimulus steps. If using a Compound Resolution Stimulus Step in the test sequence, the resolution in the Speaker Equalization sequence should be no less than the highest resolution of the Compound Stimulus.***

Mouth Simulator Calibration and Correction Example

When calibrating a Mouth Simulator the recommended calibration sequence is **Speaker Equalization**. This is normally for use with an acoustic mouth simulator or sound source for testing microphones close to the source (e.g., vocal mics) with a swept sine at a constant sound pressure level.

- Output Signal Path set to **Source Speaker**
- **Mouth Simulator.dat** selected under Calibrated Device
- Output Hardware Channel set to **Output 1**
- **Speaker Equalization** selected under Calibration Sequence
- Input Signal Path set to **Reference Mic**

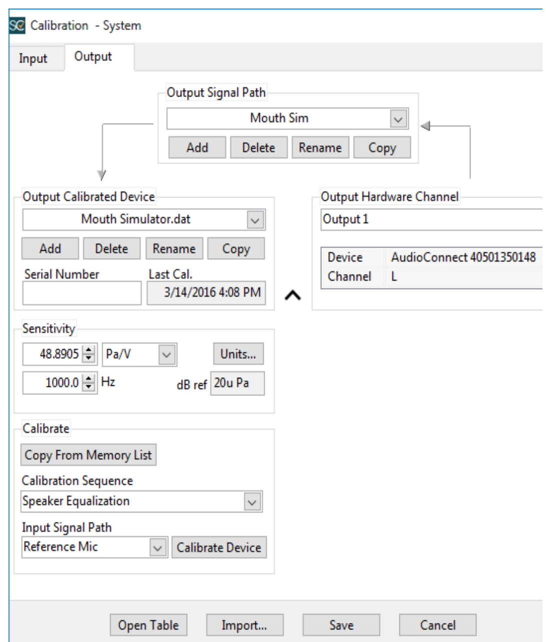


Figure 9-24: Mouth Simulator Calibration

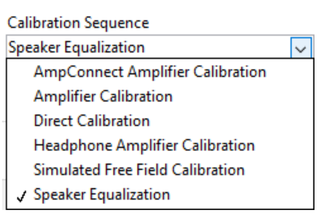


Figure 9-25: Calibration Sequence Menu

1. Using a calibrated reference microphone (e.g.; SCM 3), place the microphone in the same position that you intend to measure the Device Under Test.
2. Click the **Calibrate Device** button at the bottom of the **Output Tab** to begin the calibration procedure.

3. The first step in the calibration procedure is a message stating that the reference microphone needs to be calibrated first. Click **Enter**.

4. Enter the level in dB SPL for the calibration signal. This should be the level that will be used in the test sequence. This example uses a level of 80 dB SPL. Click **OK** or **Enter** to continue.

   The system will play back a short sine tone to get the sensitivity of the speaker. This allows the sequence to automatically adjust the stimulus level to 80 dB SPL.

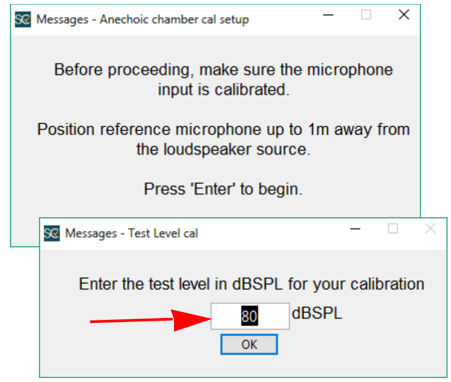


Figure 9-26: Set Signal Level

5. Set the lowest frequency to equalize down to. In this example, 100 Hz. Click **OK** or **Enter** to continue.

6. Set the highest frequency to equalize up to. In this example, 10 kHz. Click **OK** or **Enter** to continue.

Important! Do not try to EQ beyond the designed range of the loudspeaker or mouth simulator. Damage can occur due to excessive low or high frequency output, as the correction process attempts to create a Flat Output Response from the speaker.

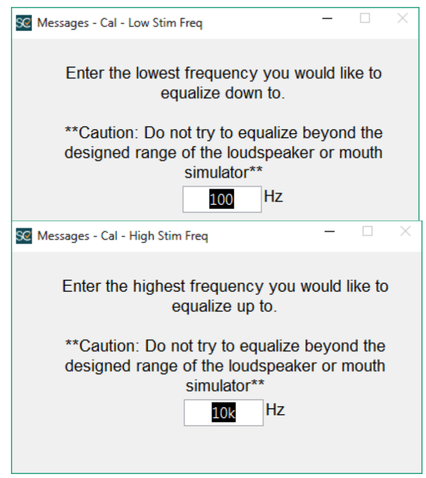


Figure 9-27: Set EQ Stop Points

7. SoundCheck will measure the speaker's response and sensitivity. These are compared to preset upper and lower limits. ***Figure 9-28*** shows the response before it is corrected.

8. If the Mouth frequency response is within acceptable limits, the sequence will continue. If the response is not within limits you will be prompted to stop the equalization process and change the equalization range.

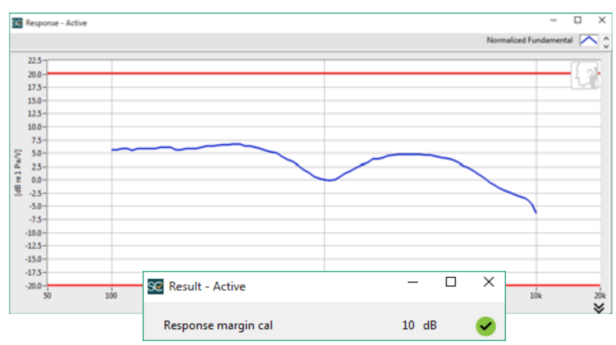


Figure 9-28: Response Before Correction

9. SoundCheck will generate an equalized stimulus and play it through the speaker.

   The typical equalized response for a Mouth Simulator is ±0.5 dB from 100 Hz to 10 kHz.

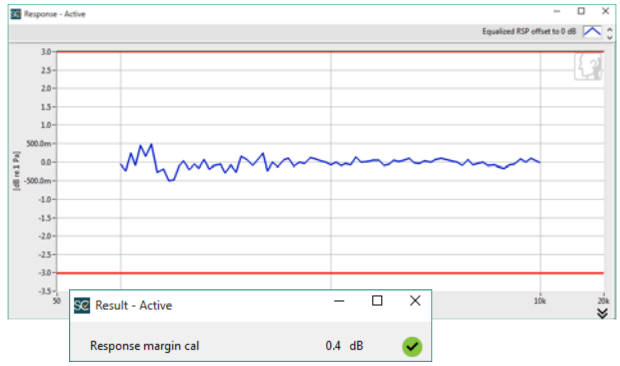


Figure 9-29: Equalized Response

10. If the equalized response or sensitivity are outside of the limits set in the calibration sequence, the sequence will prompt you to stop or attempt equalization anyway.

    If both pass, a sequence prompt asks if the EQ's Response is flat enough.

    Selecting "**No**" will run another pass of equalization to "fine tune" the correction curve. This can be run as many times as desired, but at some point the correction will show no further improvement.

    Select "**Yes**" to complete the calibration and correction process.

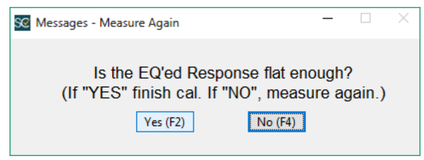


Figure 9-30: Prompt To Measure Again

11. Once the calibration procedure is complete, SoundCheck will update the Mouth sensitivity. You can then choose **Save** to overwrite the original **Mouth Simulator.dat** calibration. Select **Rename** under Calibrated Device to give the calibration a different name.

12. Select **Save** to store the changes to the System Calibration Configuration.

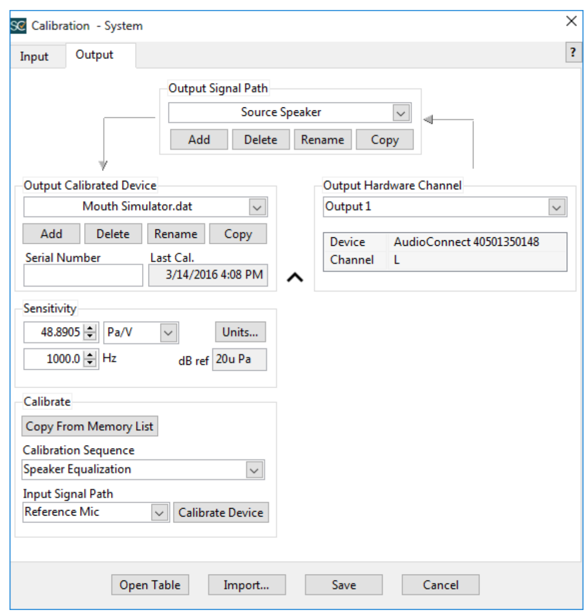


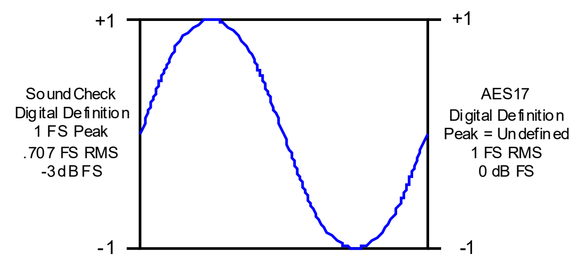
Figure 9-31: Updated Mouth Sensitivity

Digital Channel Calibration

FS and dB re 1 FS Definition

Digital Full Scale (**FS**) and dB relative to 1 FS (**dBFS**) though widely used are not normatively defined. In SoundCheck, **FS** as a unit is used to express peak and **RMS** amplitude. When used to define peak amplitude the maximum values are +1 and -1 FS which correspond to the maximum positive and negative digital codes in a digital audio sample stream. 1 FS RMS or 0 dB re 1 FS correspond to the RMS amplitude of a square wave that reaches the maximum positive and negative digital codes. The maximum amplitude of a sine wave in this scheme is therefore **.707 FS** or **-3.01 dB re 1 FS**. This represents the maximum amplitude of a sine wave where the peak positive and negative deflections reach the maximum positive and negative digital code.

The other commonly used definition of FS comes from the Audio Engineering Society standard 17 (AES17) which defines FS as the RMS amplitude of a sine whose peak positive and negative values reach the maximum digital code of the system.



In AES17, the maximum amplitude of a sine wave is 1 FS or 0 dBFS and the maximum amplitude of a square wave is 1.414 FS or +3 dBFS.

The AES17 definition of FS cannot be used to express peak signal amplitudes. For a signal of given amplitude the AES17 and SoundCheck definition of FS differ by 3.01 dB.

This difference in scaling between SoundCheck and AES17 can be compensated for by using FS (AES17) when importing a .WAV file or Unity Digital (AES) calibration with digital inputs and outputs.

See ***WAV File General Rules on page 336***.

As of SoundCheck 14, **Unity Digital In (AES17)**.dat and **Unity Digital Out (AES17)**.dat are included as calibrated devices. When these devices are used in the Calibration Editor, they adjust the sensitivity to match the AES17 definition and include the proper units. See ***Figure 9-32***.

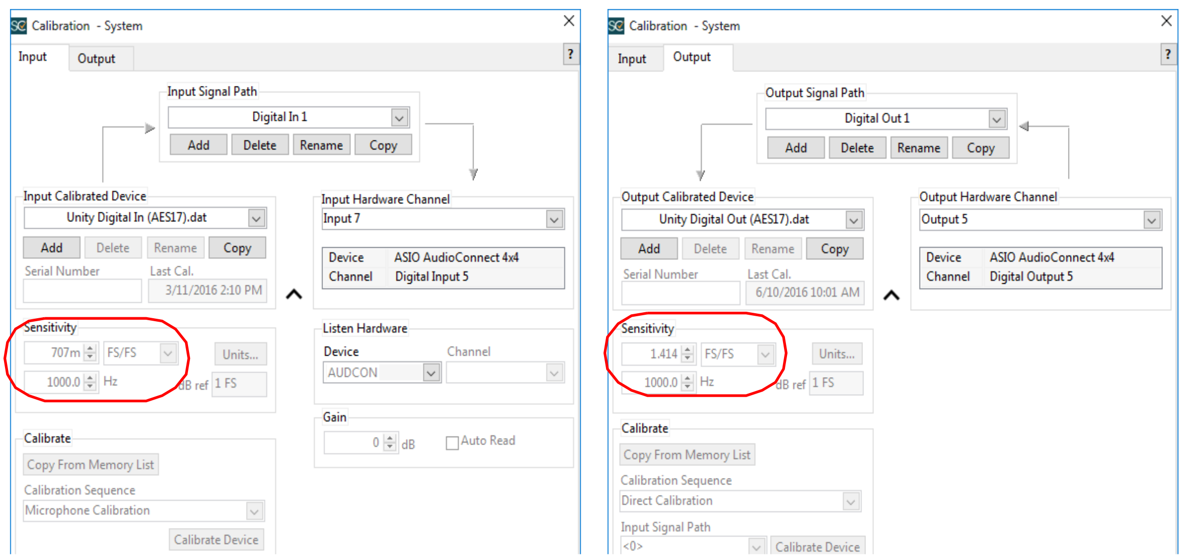


Figure 9-32: Calibration Editor Digital Channels

WAV Analysis

When analyzing a WAV file no signal path is used and results will be scaled according to the SoundCheck definition of FS. In order to scale the results according to the AES17 definition of FS you must add 3 dB to your results using a Post Processing step.

Reference Codec & dBm0

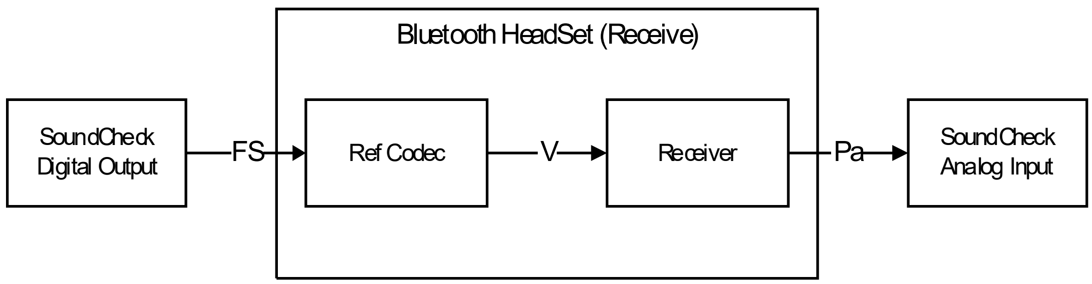


Figure 9-33: Bluetooth Test - Signal Chain

When performing a Receive test on a Bluetooth Headset, SoundCheck sends a digital audio signal to the DUT. SoundCheck then records and analyzes the acoustic output of the DUT.

Therefore, assuming that the DUT operates in a linear manner, a digital sine wave in yields an acoustic sine wave out. Moreover, you will be able to measure the overall Receive Gain of the DUT in Pa/FS.

Definition: 1 Full Scale = maximum absolute value of the digital wave. e.g., 32767 ≡ 1 FS for a 16 bit encoded wave. (FSD: Full Scale Deflection)

Normally, in telephone testing, the output units are expressed in Pa/Volt. Since the input signal is digital audio we need to translate the FSD units to Volts using a virtual reference CODEC.

The ref CODEC of the blue tooth device handles the conversion of the digital signal into volts (Virtual Volts).

The ref CODEC states that a digital sine at zero dBm0 applied at the input yields 0.775 Vrms at the output of the CODEC.

The ref level of zero dBm0 is defined differently, depending on which coding law is used: A-law or µ -law (mu-law).

A-law

0 dBm0 is the level of a sine signal which is 3.14 dB below saturation. That means that the absolute peak amplitude of the sine is 10^(-3.14/20) = 0.6966 FS and the rms value is 0.4927 FS, equivalent to –3.14 -3.01 = -6.15 dB FS.[1]

$$0\text{dBm0} \equiv -6.15\text{dBFS} = 0.4927\text{FS}$$

μ-law

0 dBm0 the level of the sine signal is 3.17 dB below saturation. That yields: 0.6942 FS pk ≡ 0.4910 FS rms, equivalent to –6.18 dB FS.

$$0\text{dBm0} \equiv -6.18\text{dBFS} = 0.4910\text{FS}$$

Note: dBm0: A reference voltage of 775 mV applied to a load of 600 Ohm yields 1 mW.

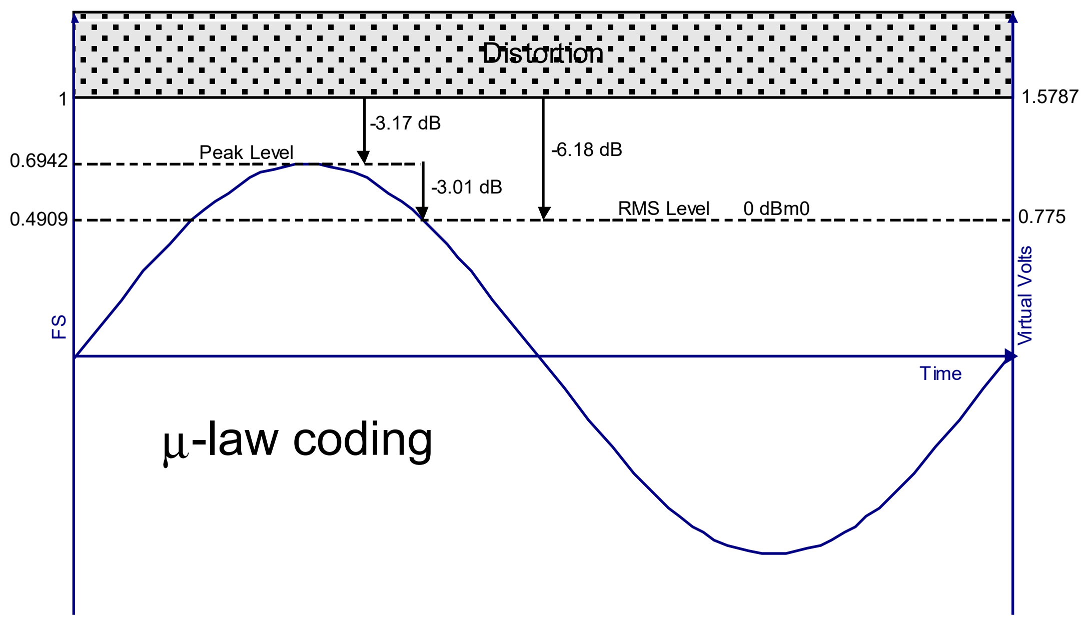


Figure 9-34: μ-law Coding - Sine @ 0 dBm0

1. 3.01 dB ≡ √2, crest-factor of sine.

Bluetooth Sequence Setup Example

The following example is from the Bluetooth Receive sequence which is part of the default sequences included with SoundCheck. It is located in the Headphones & Headsets folder.

Hardware Settings

- In System Hardware create an input channel and an output channel for the Bluetooth interface
- Select Digital for both
- Set the Sampling Rate and Bit Depth appropriately for the device under test (typically 8 kHz and 16 bit)
- The Max Out value is automatically set to100% since Digital Output has been selected

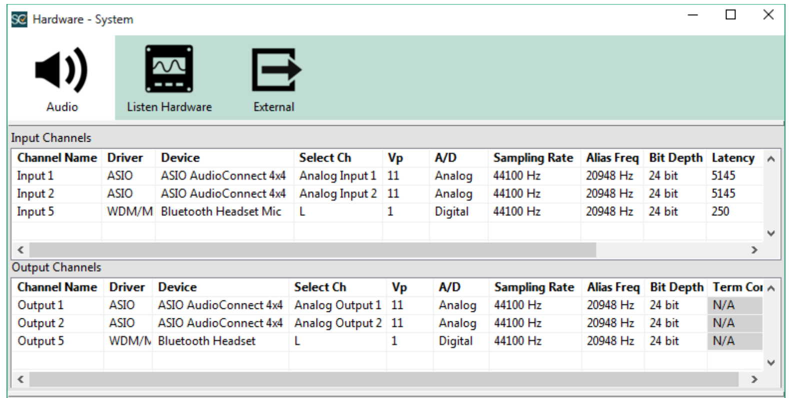


Figure 9-35: Bluetooth Headset

System Calibration Configuration

- In the **Output Signal Path** of the Calibration Editor, select the 'BT Headset' Signal Path from the list
- Set the **Output Hardware Channel** used for this Bluetooth interface
- In **Output Calibrated Device**, select Unity Digital Out (AES17).dat if testing purely digital data
- For µ -law: Select "**Virtual Code-Out (u Law).dat**" which has a sensitivity of 1.578 V/FS (0.775/0.491 = 1.578)
- For A-law: Set the Output Calibration Sensitivity to 1.573 V/FS (0.775/0.4927 = 1.573)
- If using either codec, set the Output Calibration Units dB ref to: 0.775 V (0 dBm0)
- In the end, 0 dB = 0.775 "Volts". The saturation point for µ-law is +3.17 dB above this level. For A-law it is +3.14 dB.
- Anything above saturation (1 FS) is distortion
- You can also use vV (virtual Volt) as an output calibration unit

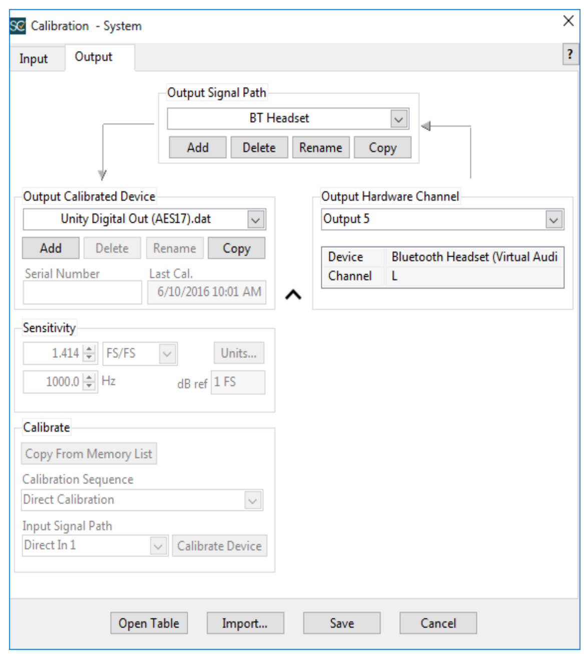


Figure 9-36: Calibration Bluetooth Headset

Units

Units Overview

SoundCheck® 18.1 is set up to provide you with a great deal of flexibility regarding measurement units. Units can be defined in five different places. The *Calibration Editor*, *Analysis Editor*, *Post-processing Editor*, *Display Editor*, and *Message Editor* allow you to make changes to the units of the curves, single values and results generated by the sequence.

SI Units are used throughout SoundCheck 18.1. For more explanation on SI Units please refer to ***SI Units on page 55***.

Calibration Editor

The **Units** button in the *Calibration Editor* will set the units for only the *Signal Generator*, *Multimeter*, *FFT*, and *Real Time Analyzer (RTA)* virtual instruments. In the example below, the microphone that is connected to the Left Signal Path has a sensitivity of 14 mV (0.014 Volts) per Pascal. By clicking **Units**, you can choose the units for the decibel reference. In ***Figure 10-1*** the reference has been chosen to be 20 µPa. This enables the *Multimeter* to display the measurement in both dB re 20 µPa and absolute units (Pascals).

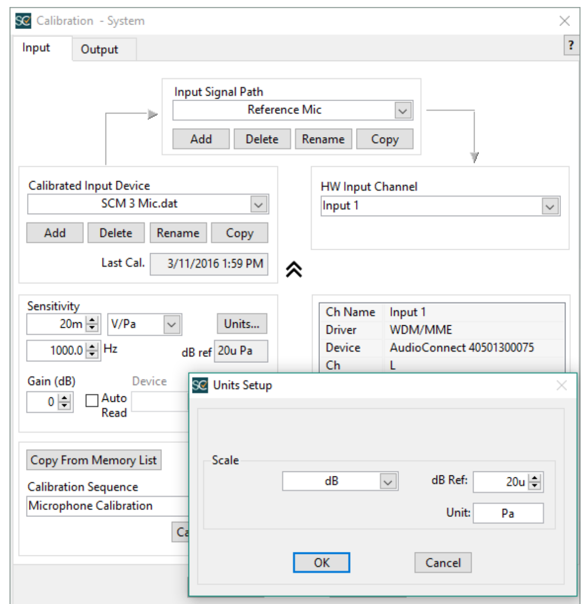


Figure 10-1: Input Signal Path Units Affect Units of Virtual Instruments

In ***Figure 10-2***, we switch the Signal Path to Accelerometer which has a decibel reference of dB re 1 m/s^2.

The *Multimeter* will now read out in dB re 1 m/s^2.

The virtual instruments adopt the unit of the selected channel.

Any changes made to the units in the *Calibration Editor* will change the units that are displayed in the virtual instruments.

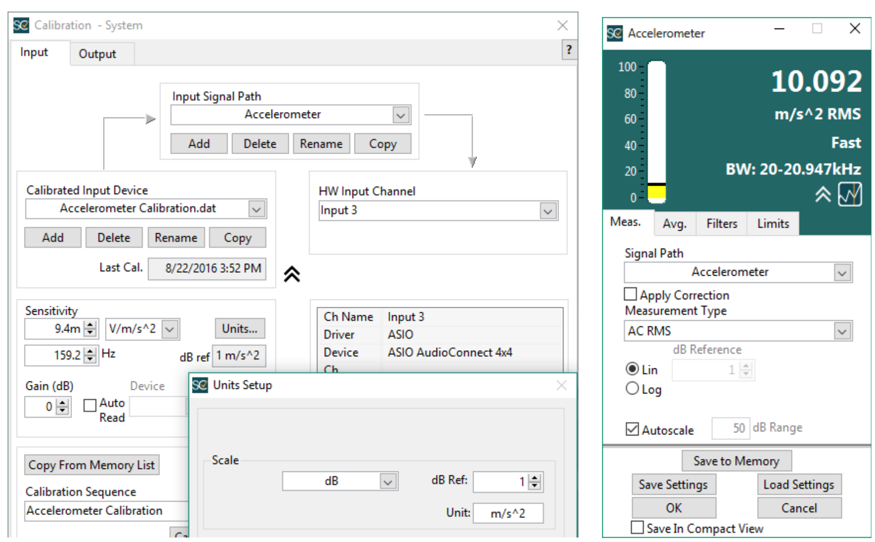


Figure 10-2: Virtual Instrument Units

Several common dB references are already pre-programmed in SoundCheck.

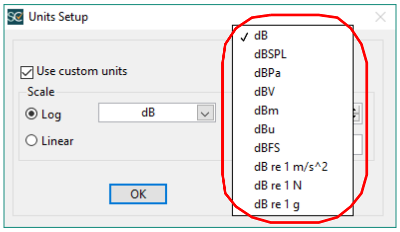


Figure 10-3: Common Decibel References

dB SPL	Re 20 µPa
dB Pa	Re 1 Pa
dB V	Re 1 Volt
dB m	Re 1 mWatt
dB u	Re 0.775 V (600 Ohm load)
dB FS	Re 1 FS
dB m/s^2	Re 1 m/s^2
dB N	Re 1 N
dB g	Re 1 g

Note: The **Use Custom Units** check box is available in the following editors: *Analysis*, *Post-Processing*, *User Equation and Memory List*. If the box is left unchecked, the units from the *Input* and *Output Calibration* editors will be used. If the **Analysis-Response Measurement** is set to **Absolute,** the units will be that of the Input in the Calibration Configuration. If it is set to **Relative** then the units will be the Input Units over the Output Units. Please refer to ***Defining the Units on page 90*** for more information on Physical Units.

Note: When **Use Custom Units** is checked, the units can be modified to be any type desired.

Analysis Editor

The *Analysis Editor* enables you to process measured time signals using a variety of analysis algorithms. The curves and single values created by this step will use the units indicated in the **Units** tab. The units selected here apply to the **DUT response** and typically agree with the units selected for your input and output devices in the System Calibration Configuration. For more details on how to use this editor, please refer to *Analysis Editor on page 157*.

The **Units** button in the *Analysis Editor* will set the units for the data generated by that particular Analysis Step (e.g., frequency response).

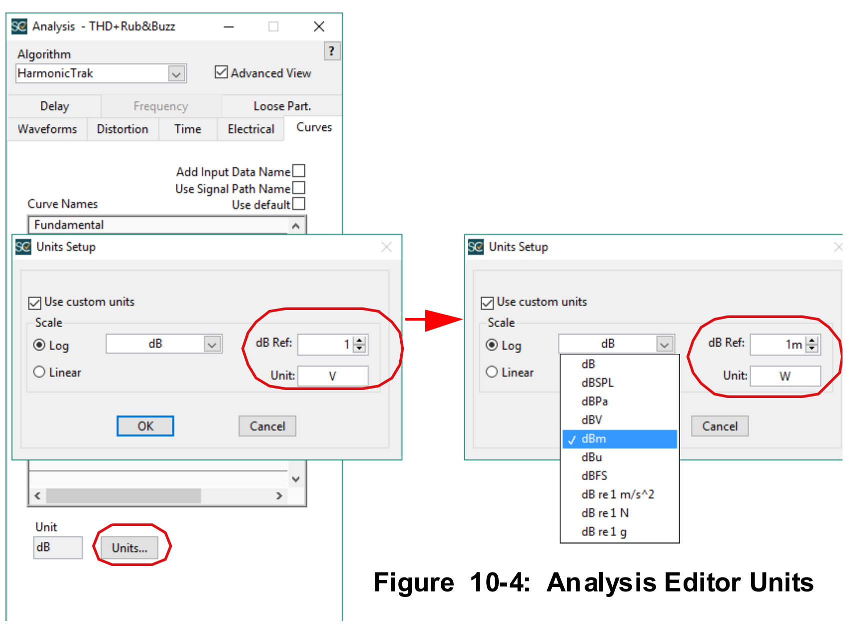


Figure 10-4: Analysis Editor Units

In the example in ***Figure 10-4*** the units were changed from electrical voltage decibels (dB V) to electrical power decibels (dBm) by choosing “dBm” from the drop-down list.

The corresponding data display will show the measured curves in dBm (dB re 1 mW).

Display Editor – Memory List

Use the *Display Editor* to format the presentation of data on the screen using six types of display windows. Refer to***Display Editing on page 339***.

Typically the units for the curves and single values are set up in the *Analysis Editor*. However, the units can be temporarily modified if a curve needs to be rescaled, for example by changing the decibel reference. These changes will disappear when the sequence is run again, unless the curve or value with the new units is protected in the *Memory List*. Refer to ***Display Editing on page 339*** for more information on Protected Items.

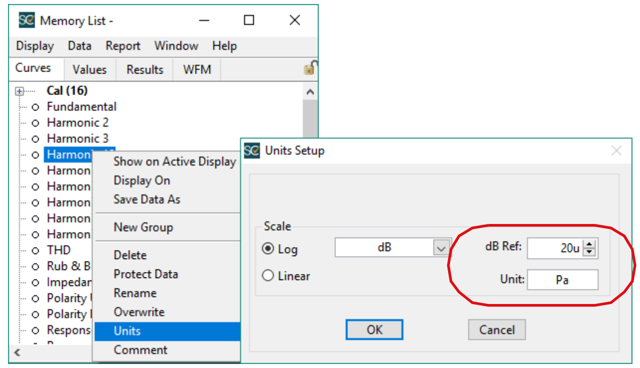


Figure 10-5: Memory List Units

In ***Figure 10-5***, the units of the y-axis for the curve *Harmonic 10* are modified in the *Display Editor*. To do this, open the *Memory List* and choose a curve or single value with values in the y-axis. Note that only one curve can be selected in the current tab of the *Memory List*. The *Units Setup* window will then appear and user-defined units can be entered once the **Use Custom Units** box has been checked. These new units are only valid while the display is open. Once the test sequence is run again, the units for the measured data will revert back to the units defined in the *Analysis Editor*.

Post-Processing Editor

By clicking **Units** in the *Post-Processing Editor*, the *Units Setup* screen appears. The default units in this editor are that of Operand A, as modified by the post-processing operation. In this example, the units of Operand A are dBm re 1.00 mW. Since the operation being performed is Reciprocal Value, the new units will be dB re 1.00 x 103 Watts since this is the reciprocal of 1 mW. This unit can be changed by clicking **Use custom units** in the *Units Setup* window. The post-processed result will then be displayed in these user-defined units.

Each mode of the *Post-Processing Editor* has a **Units** button located as in ***Figure 10-6***, except the User Equation. In the User Equation mode, you must designate the units of the equation results on the right of the editor, and set the units for any **User Defined Constants** created on the left side of the Editor. Units cannot be changed on curves or single values created by other steps in the sequence.

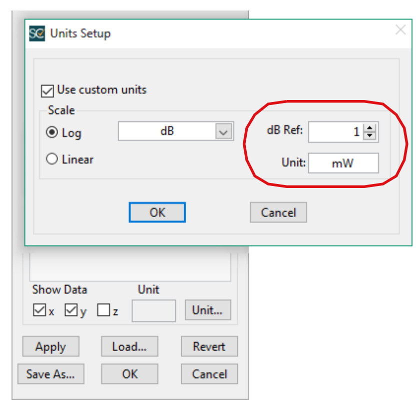


Figure 10-6: Post-Processing Units

The units can also be changed in the User Equation Post Processing step. After selecting an Input Curve or Value, and then filling in the other associated fields, click on the “**Select Unit**” field to open the Units Setup window.

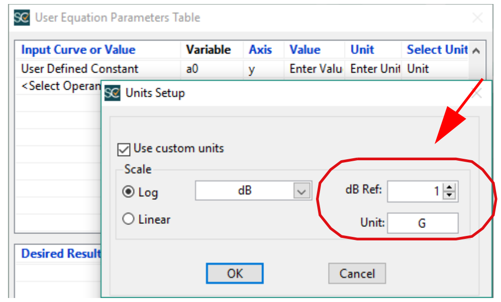


Figure 10-7: Equation Editor Units

Message Step Editor

The **Units** button becomes available when the *Message Step* is set to ask the Operator for a numerical value during the sequence. The *Units Setup* screen appears when this button is clicked. There are no units for this value by default.

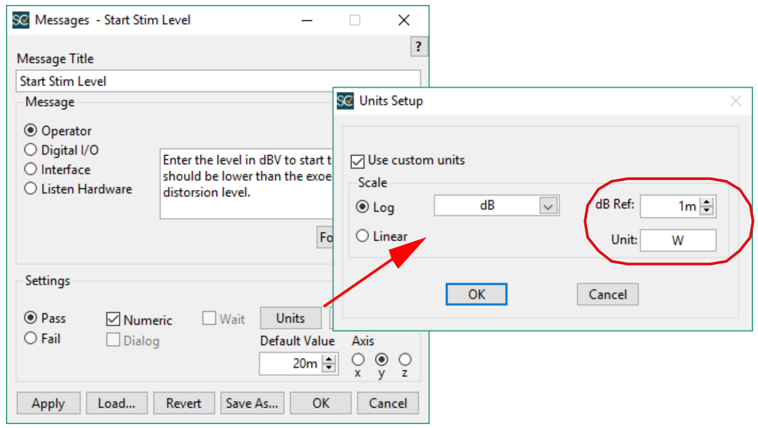


Figure 10-8: Message Editor Units

Stimulus Editor

Features

- As of SoundCheck 18, the Stimulus Editor has only one view mode. The Advanced View selection has been removed.
- **Output Path** is now a field in the editor. The **Output Path** button has been removed.
- The Sampling Rate of the Stimulus Step is set in a field of the editor
- As of SC 8.1, when "**Memory List Selection**" is selected in the Stimulus Editor, a message will pop up as a reminder to "shut off" Preload Stimulus in the Sequence Configuration. See ***Configure Sequence on page 445***.
- The *Stimulus Editor* (Ctrl+Shift+S) creates or loads a WAV file that the Acquisition Step can play.

See ***Stimulus Type on page 116***.

Frequency Stepped Sweep (Stweep™) Excitation Signal Parameters

The Stweep stimulus offers faster and more accurate measurements.

Typically a digitally generated stepped-sine excitation will contain discontinuities because the frequencies do not change at a zero phase and amplitude crossing. By using an integer number of cycles at each frequency step, the STWEEP ensures that transition from frequency to frequency is always smooth. This ensures significantly less transducer settling time and results in faster and more accurate measurements.

Sweep Equalization for Minimized Transients

In stepped sine, amplitude and frequency sweeps, selecting equalization now also enables a smooth transition between steps. These smooth transitions minimize the transient response in the device under test. This results in shorter test times, and is particularly useful for microphone testing where a source speaker needs to be equalized. Select "**Apply EQ**" in a Stimulus Step.

See ***Sweep Equalization for Minimized Transients on page 121***.

Sweeping Hi Frequency to Low Frequency

The default sweep direction of Stimulus Steps using Frequency Stepped Sweep is from Hi to Low.

A device's fundamental frequency response should not change with frequency sweep direction or test signal since it is a linear approximation. Since loudspeakers are inherently non-linear, extra care should be taken in getting a reliable and repeatable linear response, e.g.: the fundamental.

The non-linear response, e.g.: distortion, is a different matter and is even more affected by how the test signal is applied. The goal is to get the loudspeaker in a stable state before measuring it in order to get repeatable results.

When sweeping from low to high frequencies, the loudspeaker sees a burst of energy at the first low frequency and can continue ringing throughout the measurement before reaching a stable state. When sweeping from high to low frequencies, the first high frequency has very little energy compared to a low frequency. It reaches stability faster and consequently there is less chance for the loudspeaker to ring throughout the measurement.

The other advantage to sweeping from high to low frequencies has to do with system delays but that should not matter when testing a simple loudspeaker driver without any electronics.

Stimulus Settings

The following uses a stepped sine sweep (Stweep™) as an example. Some Stimulus Types will require different settings.

To view and change the Stimulus step settings, select **Stimulus** from the **Setup** drop-down list on the SoundCheck Main Screen or double click the **Stimulus Step** in the Sequence Editor.

The *Stimulus Editor* will appear indicating the current test signal settings for the selected step.

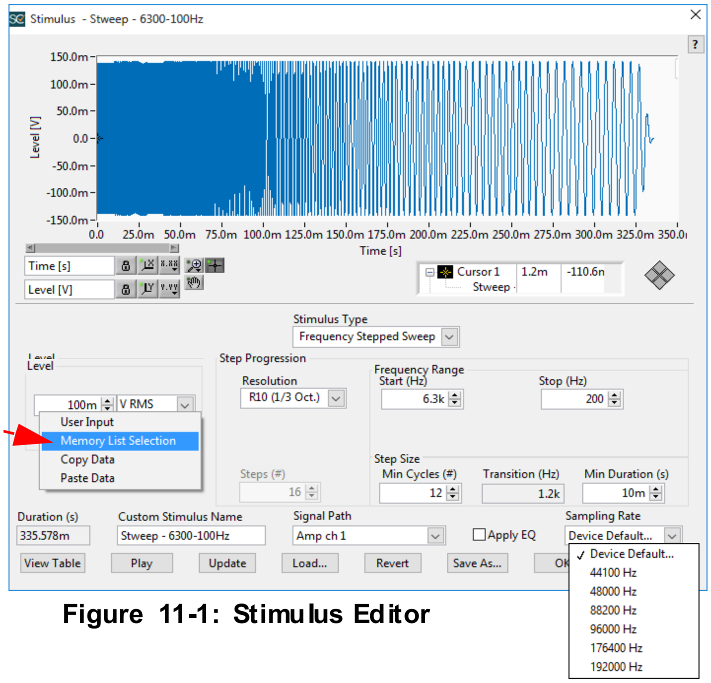


Figure 11-1: Stimulus Editor

Sampling Rate

Sampling rates are no longer limited to using the sample rate specified in the Hardware Editor. Select a supported sampling rate in the **Sampling Rate** field.

This is especially helpful when playing WAV files as the Stimulus Editor no longer requires that you specify the sample rate.

Available sample rates for the selected Signal Path are listed in the Sampling Rate drop-down list as shown in ***Figure 11-1***.

WAV Sample Rate

When a WAV file is selected, the sample rate of the WAV file automatically sets the sample rate of the Hardware Channel associated with the selected Signal Path. As long as the audio interface supports the sample rate of the selected WAV file, the WAV file will play without having to open the Hardware Editor.

The Sampling Rate field is grayed out when WAV is selected as the Stimulus Type.

All available sample rates for the audio interface, up to 192 kHz, are displayed in the drop-down list.

The Hardware Editor channels for the selected Signal Path will change to the selected sample rate automatically.

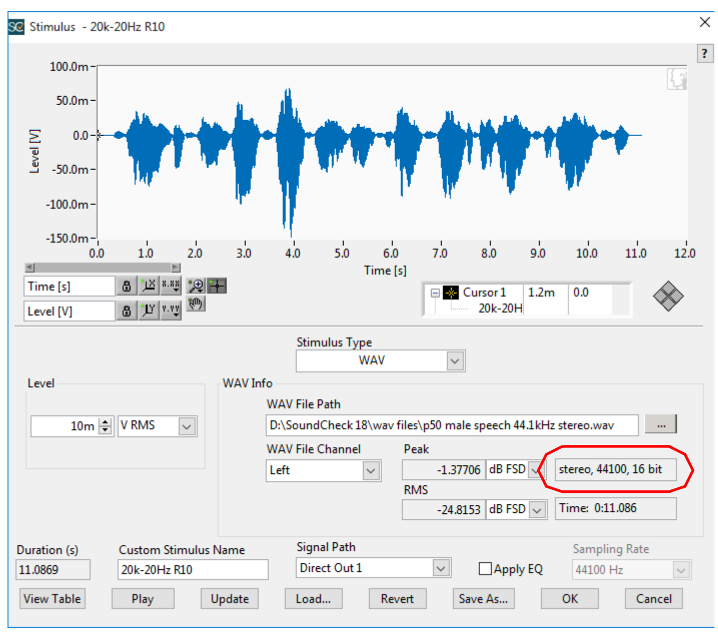


Figure 11-2: WAV File Stimulus

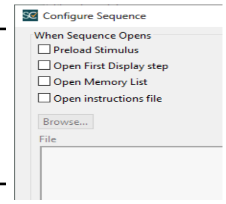
Example: In ***Figure 11-2***, a WAV file with a sample rate of 44.1 kHz is selected in the Stimulus Editor but the associated hardware channel has a sample rate of 96 kHz. The sample rate of the hardware channel will automatically change to allow playback of the WAV file.

Right-click Functions

Right-click the fields for Level, Start Frequency and Stop Frequency to select **Memory List Selection** or **User Input**.

The Memory List item must be created or added to the Memory List before the Stimulus Step occurs in the order of steps in the sequence.

When setting the units for the stimulus, remember to set the units in the Message Step that is used to generate the Memory List item. *See "Numeric Message" on page 289.*



Important! When selecting "**Memory List Selection**" in the Stimulus Editor, the sequence must be configured to NOT Preload the Stimulus. In the Sequence Editor select Sequence from the drop-down list and select **Configure Sequence**. Uncheck "**Preload Stimulus**".

Level

This allows you to set the stimulus level dynamically during the sequence execution. Examples: The default sequence, "Headphones", shows an example of how the Level is entered through a Message Step at the start of the sequence. The Speaker Equalization sequence also shows how the level is determined through Post Processing.

Start and Stop Frequencies from Memory List Values

Right-click the Start or Stop Frequency fields and select **Memory List Selection** as shown in *Figure 11-3*.

Stimulus frequencies can be entered from a prompt in the sequence, automatically generated, or even automatically incremented. This is particularly useful when you need to use a different test frequency range for each test run. You can also automatically increment the stimulus frequency and perform sequential measurements at different frequencies, e.g.: testing Max SPL.

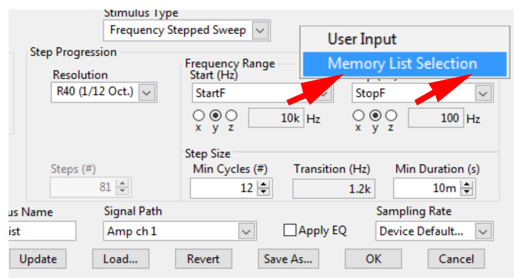
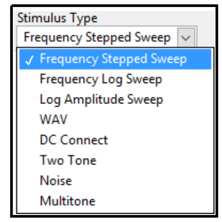


Figure 11-3: Memory List Selection

Important! ***Select the Y axis in the Stimulus Editor for Start/Stop Frequency values. The Index generated by Step Configuration is always a Y axis value. See Index (Loop Index) on page 447 for more on setting this in Step Configuration.***

Stimulus Step Controls

Stimulus Type



Select the type of stimulus the step will use from the drop-down list. Note that DC Connect stimulus can only be created using Listen's DC Connect™ interface.

- **Stepped-sine Frequency sweep (Stweep™)** – A stepped-sine sweep using an integer number of cycles at each frequency step, ensuring a smooth transition from frequency to frequency. This ensures significantly less transducer settling time and results in faster and more accurate measurements.

- **Frequency Log Sweep** - An optional continuous log sine sweep used for simulated free-field measurements

- **Log Amplitude Sweep** - At a single frequency. As of SoundCheck 8, the Log Sweep can use a faster sweep rate, e.g., 100 ms/decade, which dramatically decreases the measurement time.

- **WAV file** - Allows playback of a Windows audio file, user selectable

> ***Note:*** The WAV file must have the same sample rate as the System Hardware configuration. See ***Figure 11-4***. See ***WAV File Types on page 336*** for supported WAV file types.

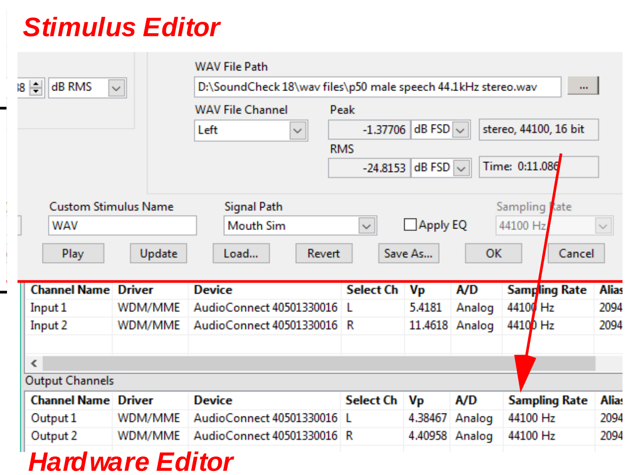


Figure 11-4: WAV file sample rate

- When opening a multichannel WAV file, each WAV file channel is numbered and selected from the **WAV Channel** drop-down list

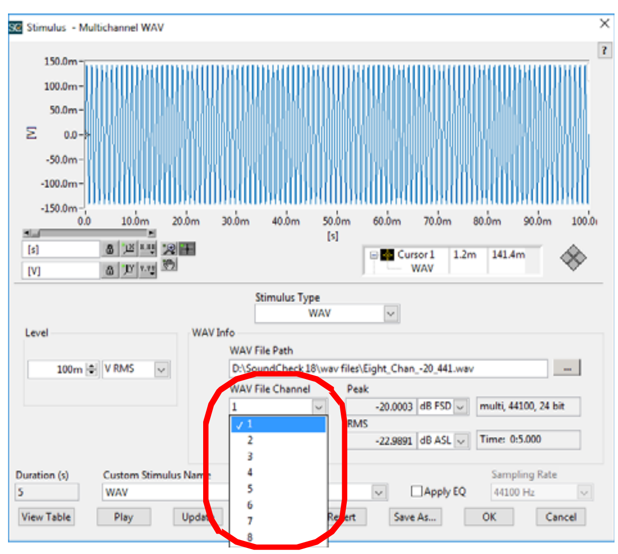


Figure 11-5: Multichannel WAV Files

- **DC Connect** – Allows SoundCheck® to control DC Connect™ for making DC voltage/current measurements (requires DC Connect™ optional hardware). Multiple DC Connects are not supported in the Stimulus Editor. See **Using Multiple DC Connects** in the latest DC Connect manual.
- **Two Tone** – Two simultaneous stimuli to use for IM and Difference distortion measurements
- **Noise** – Featuring Pink and White Noise, with scalable Duration (s) and Band limits: Fmin and Fmax. MLS (Maximum Length Sequence) noise - equalized and band limited in the same way as white noise. SoundCheck includes MLS stimulus to enable direct comparison with other measurement systems.
- **Multitone** – An ensemble of tones regularly spaced in frequency. The amplitudes are equal. The phases follow a deterministic mathematical law and are optimized to lower the crest-factor. An example of the step setup is shown in ***Figure 11-35: Multitone Stimulus - Frequency Rounding***.
- **Memory List Selection** - (Right-click Function) Allows the Level and Start/Stop Frequencies to be read from Memory List values. See ***Right-click Functions on page 115***.

Step Progression

Resolution

Determines the number of measurable frequencies, including the start and stop frequency.

- Sweep Low to Hi, Hi to Low or Single Frequency
- You can choose standard ISO frequency steps such as R10, which corresponds to preferred 1/3 octave center frequency steps, or choose User Defined linear or log step sizes.
    - Steps (#) - Shows the number of steps according to the selected resolution. If **User Defined** is selected, the number of steps is manually set in this field. The number of steps will determine how many unique frequencies are generated.

      See **Step Size** below.

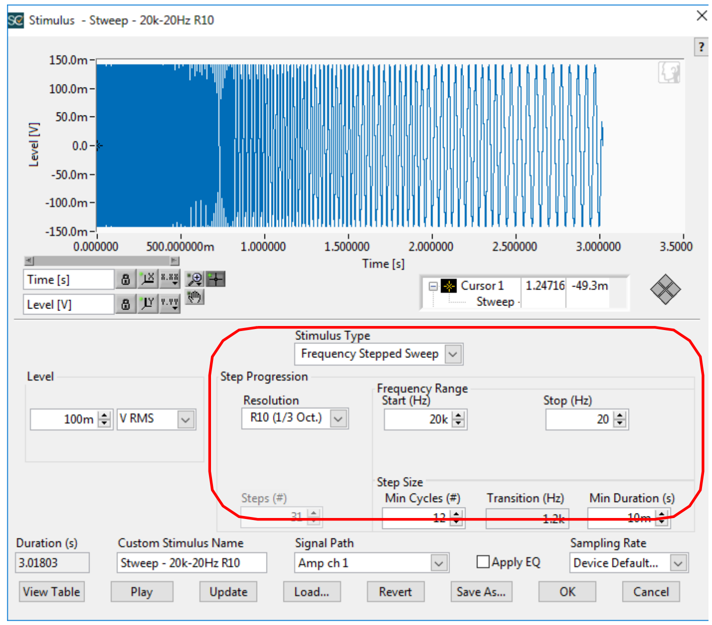


Figure 11-6: Step Progression

Start Frequency

First excitation frequency in the Stweep. Generally it is best to set this to the highest frequency you want to measure, so that the measurement sweeps from high to low frequency in order to minimize transducer settling time (low frequencies have more energy than high frequencies).

Stop Frequency

Last excitation frequency in the Stweep. Generally it is best to set this to the lowest frequency you want to measure, so that the measurement sweeps from high to low frequency in order to minimize transducer settling time.

Step Size

Min Cycles (#) & Min Duration (s)

To ensure proper measurement accuracy, each sine step must have a minimum number of cycles AND a minimum duration. To properly measure the level of a sine, you need at least three (3) cycles. In the presence of background noise, it is likely more cycles will be required. Because of noise, settling time and input-output delay, you also need a minimum duration to achieve a precise measurement.

In regard to noise only, the S/N ratio measurement increases by three (3) dB each time the duration is doubled. The Stweep algorithm ensures that each step has a duration that is greater than the Min Duration AND contains an integer number of sine cycles that is at least equal to Min Cycles.

As the Stweep covers the specified range of frequencies, each step has a number of cycles equal to Min Cycles in frequencies below the transition frequency, and a duration equal to Min Duration in frequencies above the transition frequency. By adjusting these two parameters, it is possible to optimize the total duration of your Stweep as well as the measurement accuracy.

- **Min Cycles** – Minimum number of cycles of sine at each step (See ***Figure 11-6***)
- **Min Duration** – Minimum Dwell time or the minimum time in seconds at each step

---

Note: Stimulus steps using the Stweep excitation from SC4.x and SC5.x will have a Min Duration value of zero (0) s.

- **Transition** - The frequency at which the Minimum Duration exactly matches the Minimum Cycles as selected in the editor. Below that frequency the Min Cycles condition is applied. Above that frequency the Min Duration condition is applied.
- **Example:** If a Stweep of 20k to 20 Hz is set to have eight (8) Cycles minimum per step and 10 mSec minimum duration, then the transition frequency will be 800 Hz. Below 800 Hz, all steps will be eight (8) cycles long and above 800 Hz all steps will be 10 ms long. If you do the math, at 20 Hz, eight cycles will require 400 ms and at 20 kHz, 10 ms contains 200 cycles.

User Defined Stimulus Frequency Points

The following formula are used to determine the number of frequency points in a User Defined Stimulus.

For example, a step size of one (1) allows only one frequency (the stop frequency) to be entered. In ***Figure 11-7***, a 100-cycle, 1 kHz tone has been generated.

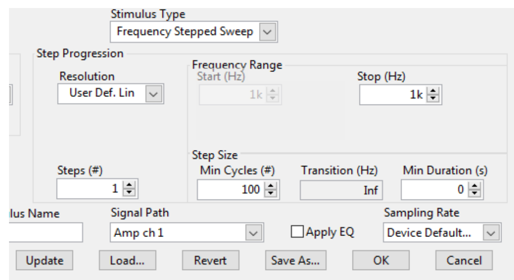


Figure 11-7: *User Defined* Step size

Log Formula	Linear Formula
$f_n = f_{start}\left(\frac{f_{stop}}{f_{start}}\right)^{\frac{n}{N-1}}$	$f_n = f_{start} + (f_{stop} - f_{start})\frac{n}{N-1}$

Where:

- N is the number of steps
- n is the frequency index from 0 to N-1

Level

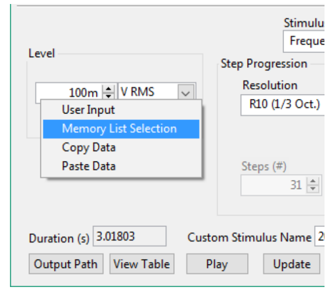


Figure 11-8: Level

Enter the correct test level in either dB or linear units. The output level and units are influenced by the output calibration sensitivity and units. The level set in the *Stimulus Editor* is the RMS (root mean square) value at the terminals of the device under test (e.g., if an amplifier is powering a loudspeaker and its gain has been entered in the output calibration sensitivity field, the output level in the *Stimulus Editor* will be the output level coming out of the amplifier).

- Define the output level in linear or dB units based on the selected Output Signal Path
- Select User Input, Memory List Selection or Copy/Paste Data
- Copy/Paste Data is used to copy the value for use in another step

Duration (s)

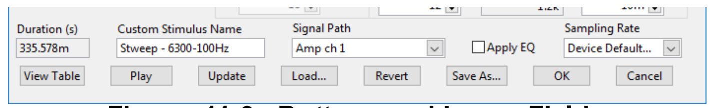


Figure 11-9: Buttons and Lower Fields

Shows the total duration in seconds required to complete the playback of the stimulus. This field is updated after clicking on the Update or Play buttons.

This calculated time is dependent on the test parameters selected in the *Stimulus Editor* (for example, step size and cycles (#); this does not include system overhead time for data and display processing). See ***Figure 11-9***.

Custom Stimulus Name

Names the stimulus waveform created in the Memory List. This item is selected in the Acquisition and Analysis Steps.

Apply EQ

As of SoundCheck 16, Apply EQ also features “Sweep Equalization for Minimized Transients”.

See ***Sweep Equalization for Minimized Transients on page 121***.

When checked, this applies the EQ curve to the Stimulus Signal for the Output Signal Path that the Stimulus is played out of. See ***Figure 11-9***.

- If you click **Play**, it will use the EQ curve for the Output Signal Path selected in the Output Path section.
- When the sequence is run, it will use the EQ curve for the Output Signal Path selected in the Acquisition Step.

The EQ curve is a correction for the response of the Output Signal Path device selected in the System Calibration Configuration. This curve is created in the Calibration process for each Output Signal Path. The curve is always present, even though it may be a Flat Curve.

(The curve can also be imported. See ***Copy From Memory List - Output on page 86***.)

Important! A complete calibration of the Output Signal Path must be run in order to store an EQ curve.

Important! Equalization curves are applied, no matter what is selected in the Output Calibration Sequence field of the System Calibration Configuration, e.g., Amplifier Calibration, Speaker Equalization, etc. The “Apply EQ” selection must be checked in the Stimulus Step, Signal Generator VI or “Acquisition Step using Generator” in order to utilize this function. **See “Calibration Configuration” on page 79.**

Sweep Equalization for Minimized Transients

In stepped sine, amplitude and frequency sweeps, selecting equalization will also enable a smooth transition between steps. These “smooth transitions” minimize the transient response in the device under test. This results in shorter test times, and is particularly useful for microphone testing where a source speaker needs to be equalized.

Select “**Apply EQ**” in a Stimulus Step.

The example SoundMap 3D plots show the difference between a sine sweep without Sweep Equalization compared to the same stimulus with “**Apply EQ**” on.

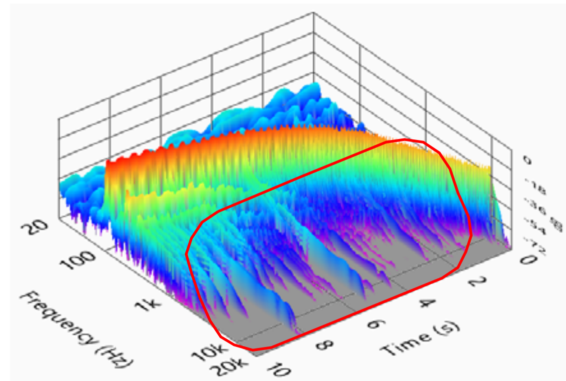


Figure 11-10: Without Sweep Equalization

The ringing circled in **Figure 11-10** is not present in **Figure 11-11**.

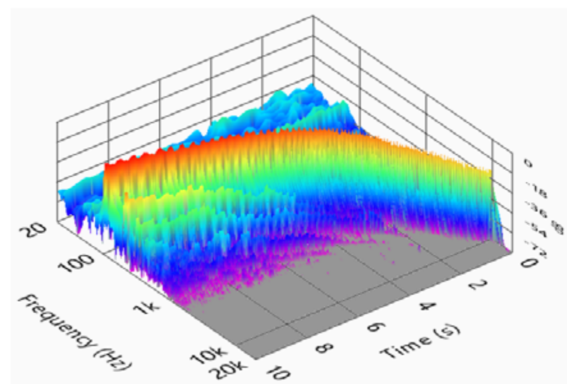


Figure 11-11: With Sweep Equalization

Advanced View

Used to show or hide advanced settings

Output Path

This allows you to select the Output Signal Path for this Stimulus Step.

This determines the Units of the Level field, the max/min frequencies for Frequency Range and the amplitude range of the stimulus.

Any Output Signal Path defined in the System Calibration Configuration can be selected as the Output Signal Path for the stimulus.

Note: As of SoundCheck 8, a separate Stimulus Step is required for each Output Signal Path “**with a unique Calibrated Device**” used in the sequence. Using this method, each Stimulus Step will conform to the settings of that Channel in the System Calibration Configuration. To create signals on multiple channels, multiple Stimulus Steps are used, each one referencing a different Output Signal Path. A single Acquisition Step is used to start the multiple stimulus signals.

Important! When multiple output devices use Unity Gain, a single Stimulus Step can be used.

The Acquisition Step will determine which channel the Stimulus is played out of when the sequence runs.

Multiple channels can be selected and the same stimulus can be used for those channels.

*The **Play** button in the Stimulus Editor uses the Output Path specified in the Stimulus Editor.*

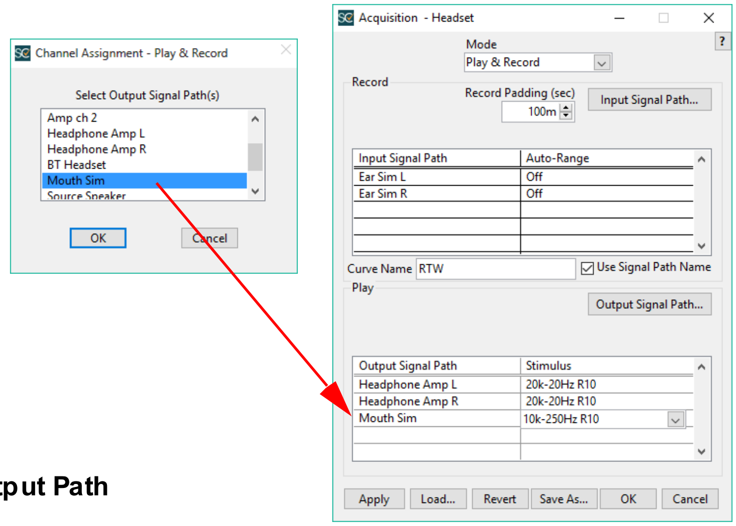


Figure 11-12: Output Path

View Table

Used to create and modify Compound Stimulus, e.g.; 1/12th oct High with 1/3rd oct Low in a single sweep

See ***Signal Parameters for Amplitude Sweep Excitation on page 128*** and ***Compound Stimulus - WAV File With Trigger Tone on page 126***.

Play

Plays the stimulus that is present in the **Stimulus Editor** to the Output Signal Path selected in the Output Path section. This allows you to hear the signal from the editor, without having to run the entire sequence.

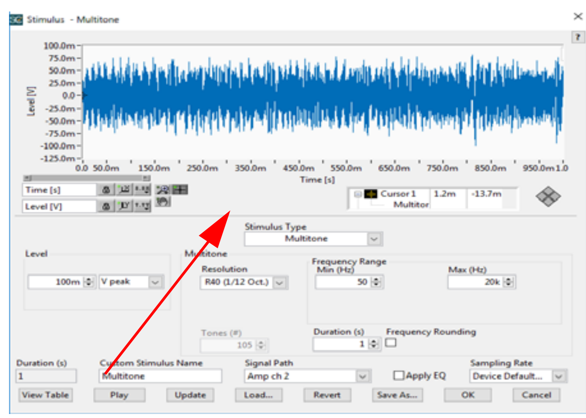


Figure 11-13: Channel Selection

Update

Click to update the stimulus display after making changes

Load

Loads the settings from a previously saved Stimulus Step

Revert

Discards any changes made to the Stimulus Editor since the last time it was saved

Save As

Allows you to save the Stimulus Step to the Steps library

OK

Accept changes and closes the Stimulus Editor. Save the sequence to save the changes to disk.

Cancel

Closes the editor and discards all changes made since the last save

Analyze Yes/No Option

The '**Analyze**' option in the Stimulus Editor allows you to choose whether or not segments of a compound stimulus will be analyzed or ignored by an Analysis Step. This feature is particularly useful for testing that requires a Conditioning Tone, Voice Activation or Trigger Tone.

Compound Stimulus Rules

When creating a compound stimulus it is important to note the following rules.

1. The Stimulus Type used must be suitable for the type of Analysis applied. Please see ***Algorithm Application Overview on page 159*** for information on what type of stimulus to use with each analysis type.
2. Like stimulus types can be chained together and these segments set to "**Analyze - Yes**". In this case, each segment will be analyzed.
3. Any number of unlike segments set to "**Analyze - No**" that occur before the first "**Analyze - Yes**" segment is allowed.
4. When using different stimulus types, only the first stimulus segment set to "**Analyze - Yes**" will be analyzed. If the second stimulus segment is of a different type than the first segment, it will not be analyzed. The "**Analyze - Yes**" setting is ignored.

   Example: A sine sweep segment combined with a pink noise segment cannot be analyzed using HarmonicTrak or Heterodyne algorithms.
5. Gaps of silence set to "**Analyze - No**" that occur after the first segment set to "**Analyze - Yes**" will end the analysis chain. No segments after this will be analyzed.

Chirp Trigger Example

The Triggered Record – Chirp Trigger function in SoundCheck allows you to test the output of devices without analog inputs such as smart speakers, wearables, smart home devices, tablets and cellphones.

The Compound Stimulus Step shown in ***Figure 11-14*** is used in the example sequence.

1. The first segment of the stimulus is the Trigger Tone or "Chirp" set to "**Analyze - No**"
2. The second segment is the test signal set to "**Analyze - Yes**"

The Stimulus Step **Analyze Yes/No** setting will tell the subsequent Analysis Step to process only the test signal and ignore the trigger tone.

A stimulus WAV file is created in SoundCheck and transferred to the device under test, where it is played back and the response recorded in SoundCheck as if the stimulus were played directly from SoundCheck. The Acquisition step is triggered by the chirp in the stimulus file.

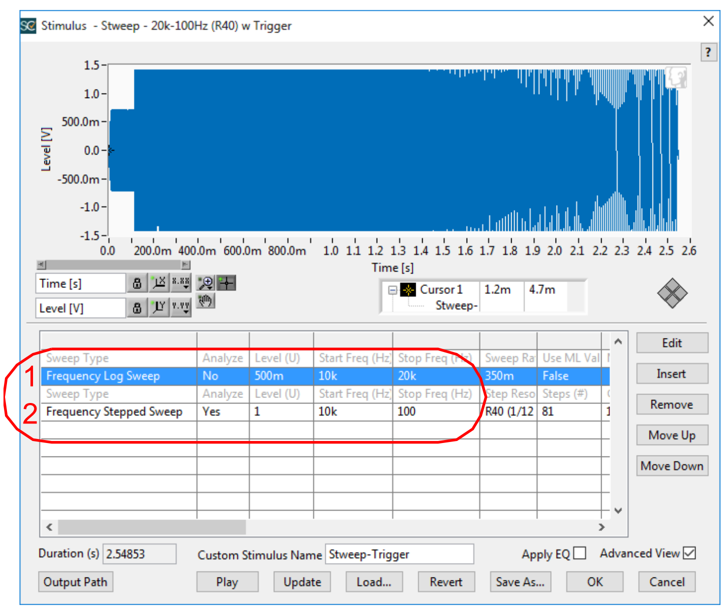


Figure 11-14: Chirp Trigger Compound Stimulus

For more information, download the sequence from our website:

http://content.listeninc.com/triggered_record_chirp

Compound Stimulus - Stweep Optimized

Combinations of different Stweep resolutions can be created using the *View Table* function of the Stimulus Editor. Each line in the stimulus can have its own Cycles (#), Level, Start/Stop Frequency, etc. This allows you to create a variety of stimuli that can accentuate the DUT's linear and non-linear characteristics.

In this case, we will use the default Stimulus Step "**Stweep - 12th&3rd Oct**" as shown in ***Figure 11-15***.

If the stimulus was in a single resolution of R40 the duration would be 2.58 seconds. Instead, this optimized stimulus has a duration of approx. 1.46 seconds.

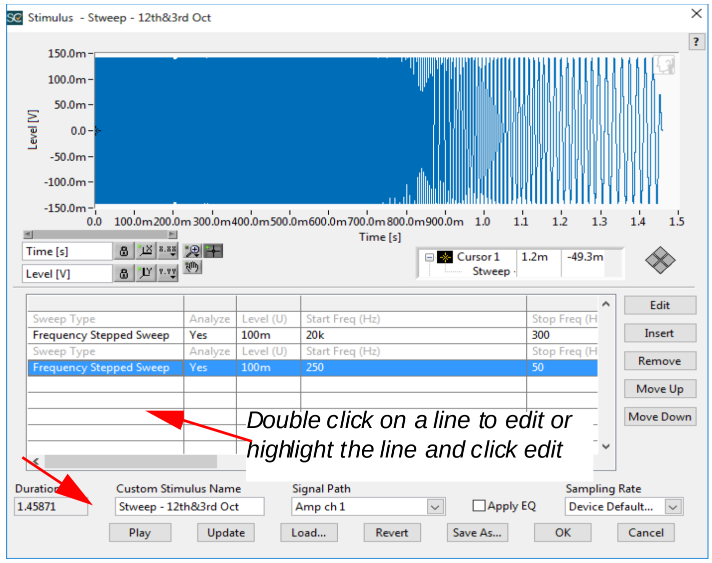


Figure 11-15: Stweep - 12th&3rd Oct

Table Buttons

- **Edit:** Each line can be edited separately
- **Insert:** Add new lines to the table
- **Move Up/Move Down:** Any line can be moved independently
- Highlight a line and click ***Edit*** or double click on a line to open the *Line Editor* window. ***Figure 11-16*** shows how the Line Editor in Table Mode allows editing of each line.
- Click **Update** to show the changes in the graphic display and to see the new duration.

Stimulus Line 1:

- Resolution: R40
- Start: 20k Hz
- Stop: 300 Hz
- Min Cycles: 6
- Min Duration: 10m

Stimulus Line 2:

- Resolution: R10
- Start: 250 Hz
- Stop: 50 Hz
- Min Cycles: 8
- Min Duration: 10m

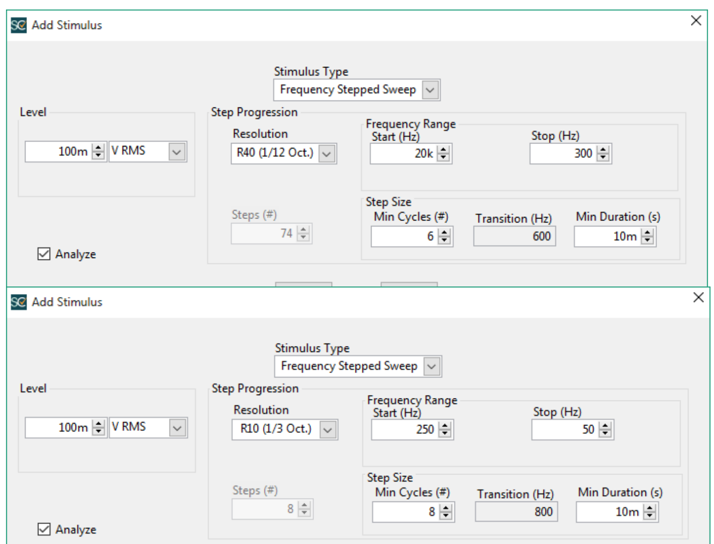


Figure 11-16: Table Editing

Both lines are set to **Analyze** and the **Level** is 100mV.

Note: Use "**Memory List Selection**" for the **Level** values in a Compound Stimulus so that editing the level is easier. You only need to set the level once, at the start of the sequence. A Message Step creates the value "**Level**" in the Memory List, which is then used in the Stimulus Step. This Message Step must occur before any Stimulus Step that uses the Memory List value it creates. The Message Step does not need to be set to "**Display Step when run**" in Step Configuration.

ISO Frequency Points

Each Start Frequency must be at the next ISO frequency point, for that resolution, following the stop frequency of the previous step.

> Example:
>
> The Line 1 Stweep ends at 300 Hz. Line 2, with a resolution of R10, must start at 250 Hz.

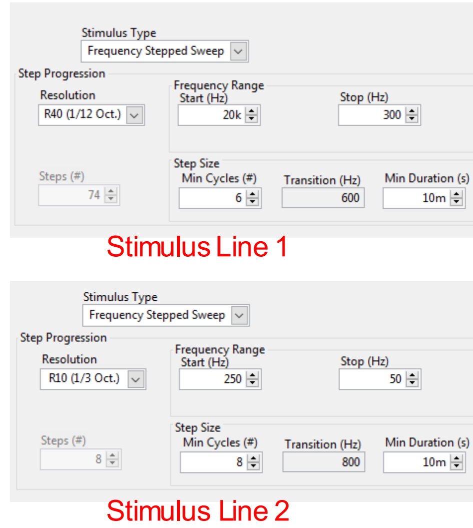
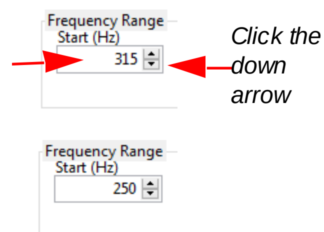


Figure 11-17: ISO Frequency Points

You can easily find the appropriate start frequency:



- Enter the stop frequency from the previous step, 300 Hz, and click **Enter**. The frequency automatically jumps up to the nearest frequency point, 315 Hz.
- Click on the **down arrow** next to the field and value will switch to the next available ISO point that is below 300 Hz (250 Hz).
- This prevents an overlap of frequencies between the two lines of stimulus.

A complete ISO frequency chart for all available resolutions can be found in ***Windows Keyboard Shortcuts on page 591***.

Compound Stimulus - WAV File With Trigger Tone

In this example a 1 kHz Trigger Tone is added to a WAV file.

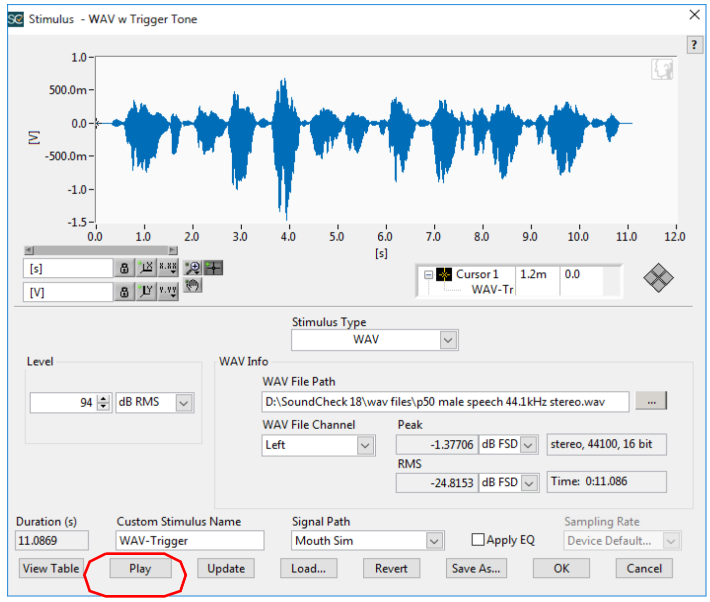


Figure 11-18: WAV Stimulus

Step 1

Start with a Stimulus Step set to play a WAV file. *Figure 11-18* shows the initial step. The table is not visible since there is only one stimulus.

- Click **View Table**

- Click **Insert** and the **Add Stimulus** window will open
- Enter the stimulus parameters for the new line as shown in *Figure 11-19*

  Level: 84 dB

  Analyze: Unchecked

  Resolution: R10

  Start Frequency: 1 kHz (Automatically "grayed out" when Stop Frequency is the same)

  Stop Frequency: 1 kHz

  Min Cycles: 1

  Min Duration: 500 mSec

  Analyze: Unchecked

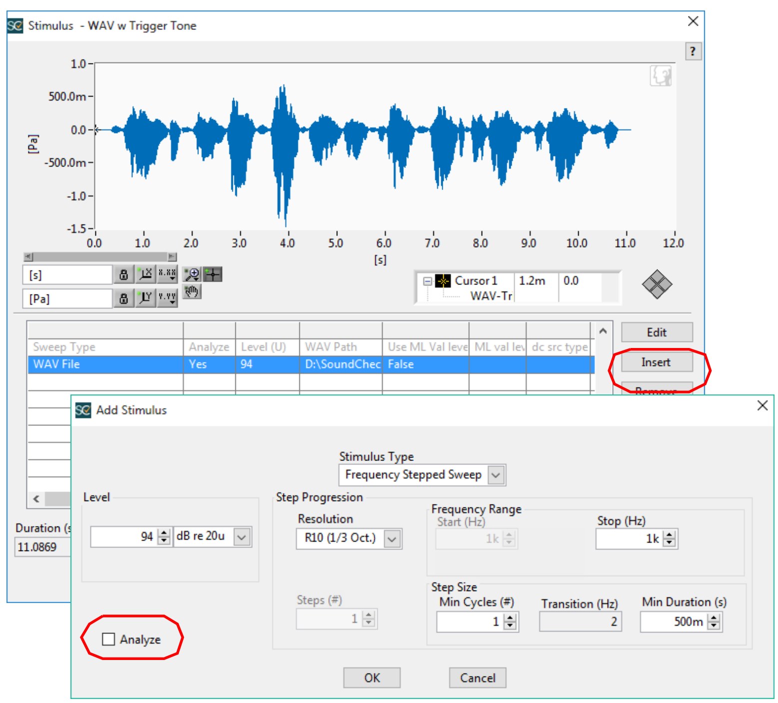


Figure 11-19: Insert Frequency Sweep

Analyze must not be selected so that only the recorded response of the WAV file is analyzed.

See *Analyze Yes/No Option on page 123*.

- Click **OK** to close the line editor

Step 3

- Click **Update** to show the actual **Duration** of the stimulus and to update the waveform view

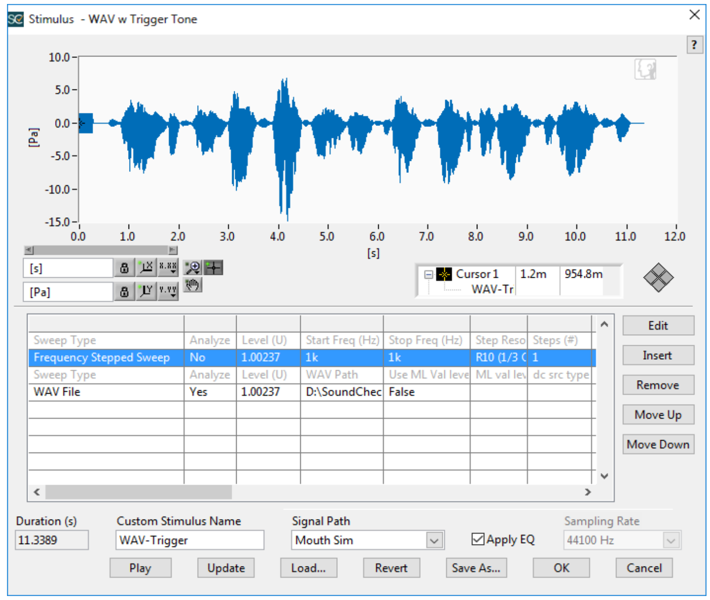


Figure 11-20: Compound Stimulus

Note: Use "**Memory List Selection**" for the **Level** values in a Compound Stimulus so that editing the level is easier. You only need to set the level once, at the start of the sequence. A Message Step creates the value "**Level**" in the Memory List, which is then used in the Stimulus Step. This Message Step must occur before any Stimulus Step that uses the Memory List value it creates. The Message Step does not need to be set to "**Display Step when run**" in Step Configuration.

Signal Parameters for Amplitude Sweep Excitation

Frequency

Enter the frequency you want to sweep in amplitude. You can only choose a single frequency here. You can also select a value from the Memory List, as long as it contains an X value.

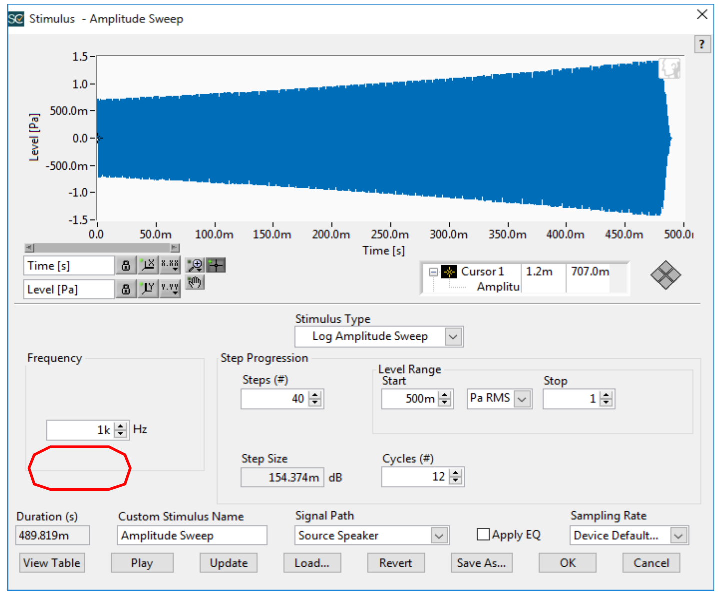


Figure 11-21: Amplitude Sweep

Start Level & Stop Level

Choose the amplitude level, either in linear or dB units. (Memory List selection is not available.)

Cycles (#)

Choose the number of cycles of the selected frequency for each step.

Steps

The number of steps is the total number of equal level increments needed to go from Start Level to Stop Level. The progression is done in dB.

The Start Level counts for one step. Then, the step size (in dB) is given by:

(Stop Level dB – Start Level dB)/(# of Steps -1).

TIP: To increase the level in integer increments, (e.g., 1 dB steps), enter the Start and Stop levels and select the proper unit (dB). Increment the value in the **Steps** field until the value in the **Step Size** field is 1. By putting the cursor to the right of the **Steps Value**, you can use the **Up/Down** arrow keys of the keyboard to quickly scroll to the desired value.

WAV File Excitation

SoundCheck can load any stereo or mono WAV file to be used as an excitation signal. See ***System Requirements on page 1*** for information regarding large WAV files and computer memory. See ***WAV File Types on page 336*** for more information on supported WAV file types.

Important! The sampling rate of the WAV file must match the sampling rate of the System Hardware configuration.

The Stimulus Step creates a signal for only one channel at a time. To output a stereo WAV file with a different signal on each channel, you need to use two Stimulus Steps with the proper channel assignment; one Stimulus Step plays the left channel of the WAV file, the other plays the right. You also need to assign them to two appropriate channels in the *Acquisition Editor*.

You need to play a WAV using the ***Stimulus Step*** if you want to capture the Time Record of the WAV as a test signal in the *Acquisition Editor* later in the sequence. In ***Figure 11-22***, WAV File is selected as the Stimulus Type. The Duration refers to the actual length of the WAV file. In this case, an 11 second sample of artificial speech is displayed.

Note: Using the WAV File option in the *Stimulus Editor* limits you to the Broadband or Spectrum Algorithms in the *Analysis Editor*.

Level

The Output Level field allows you to set the playback level of the WAV file. The level is set in physical units. The output units will vary depending on the output units of the System Calibration Configuration. For example, if using an artificial mouth or anechoic test box the output level will be Pa rms. For an amplifier or direct output the level will be V rms.

The output level will be the actual level out of the calibrated output transducer or device. This requires an accurate calibration of the output signal chain. (See ***Calibration Configuration on page 79*** for instructions on output calibration.)

The drop-down list next to the Level field has the following selections:

- RMS level (Pa rms, V rms)
- dB level
- Peak level

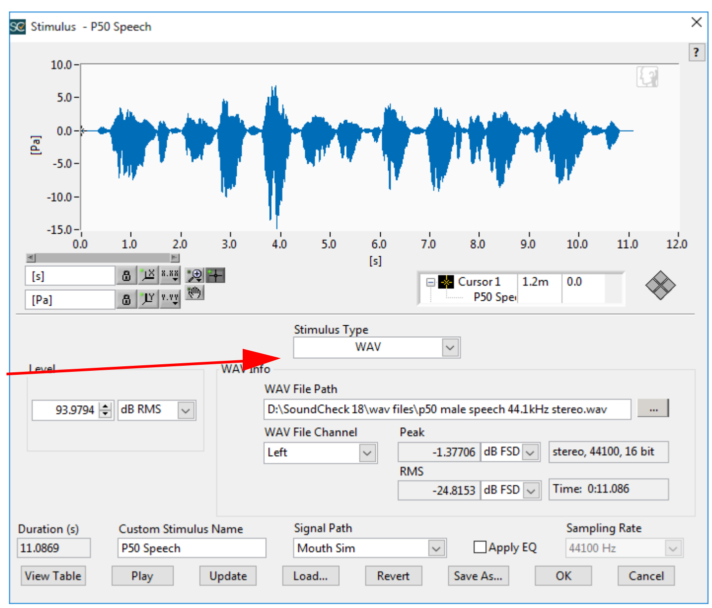


Figure 11-22: P50 Artificial Speech

Memory List Selection

This option gets the WAV playback level from a Memory List value.

WAV Info

This section shows the properties of the selected WAV file. These values are for reference only, they cannot be changed.

- **Peak**: The maximum absolute value of the file (in dB FS, or %FS).
- **RMS**: The RMS value of the entire wave file (in dB FS, or %FS).
- **WAV format**: Stereo/mono, sampling rate, bit depth.
- **Time**: Total duration of the wave file in mm:ss.ms.
- **WAV file Channel**: Allows you to choose the left or right side of a Stereo WAV file.

Apply EQ

This allows you to create an Equalized version of the WAV file. Refer to ***Equalize a WAV file (Requires optional module 2013 - EQ a Wav File) on page 463*** for more information.

When checked, this applies the EQ curve to the WAV file for the Output Signal Path that the Stimulus is played out of.

- If you click **Play**, it will use the EQ curve for the Output Signal Path selected in the Output Path section.
- When the sequence is run, it will use the EQ curve for the Output Signal Path selected in the Acquisition Step.

The EQ curve is a correction for the response of the Output Signal Path device selected in the System Calibration Configuration. This curve is created in the Calibration process for each Output Signal Path. The curve is always present, even though it may be a Flat Curve. (The curve can also be imported. See ***Copy From Memory List - Output on page 86***.)

Important! A complete calibration of the Output Signal Path must be run in order to store an EQ curve.

Important! Equalization curves are applied, no matter what is selected in the Output Calibration Sequence field of the System Calibration Configuration, e.g., Amplifier Calibration, Speaker Equalization, etc. The “Apply EQ” selection must be checked in the Stimulus Step, Signal Generator VI or “Acquisition Step using Generator” in order to utilize this function. **See “Calibration Configuration” on page 79.**

More information on the use of WAV files in SoundCheck can be found in: ***WAV File playback on page 462*** and ***WAV File Types on page 336***.

DC Connect

Multiple DC Connects are not supported in the Stimulus Editor. See **Using Multiple DC Connects** in the latest DC Connect manual.

Control Method: USB

This method is used to produce a steady voltage or current output from DC Connect.

The Stimulus Step in ***Figure 11-23*** will send the following settings to the DC Connect™ device assigned to this channel:

- Control Method = USB
- Source Type = Voltage
- Polarity = Positive
- Max Current = 30mA range
- Level = 100mV

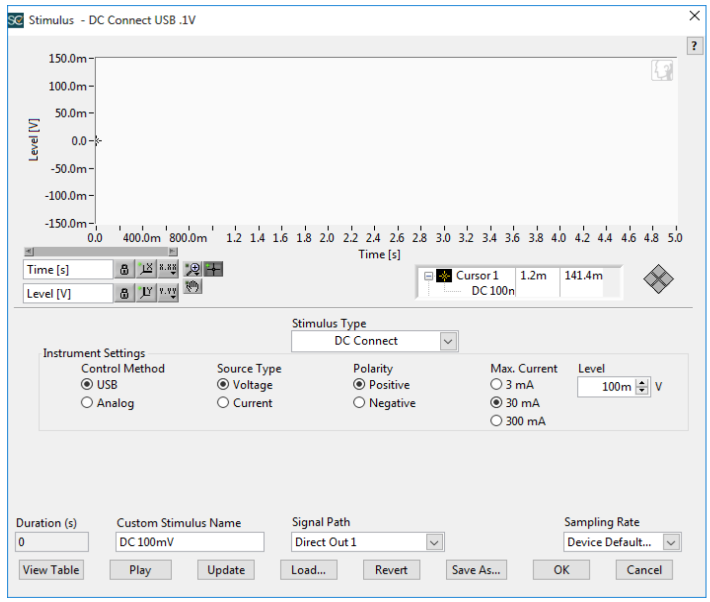


Figure 11-23: DC Connect USB Control

Preload Stimulus

- If your Sequence Configuration is set to **Preload Stimulus** (which is a default setting), the first Stimulus Step of this type will run when the sequence is loaded, and the DC Connect LEDs will show all the settings designated in the fields noted above before the sequence runs.
- If **Preload Stimulus** is not selected, you will see the LEDs on the front panel of the DC Connect light up to match the fields noted above when the Stimulus Step runs.

Important! *Preload Stimulus in Sequence Configuration will make DC Connect operational even if the sequence Start button has not been clicked. For example, if DC Connect is set to provide 9 VDC to power the DUT, the 9 V will be applied to the DUT once the sequence is loaded into memory. This will take place BEFORE you click* ***Start****.*

Use the *Level* field to set the output voltage source or current source level. That level will be output when the Stimulus Step runs, which again, may be when the sequence is loaded. If Source Type is set to voltage, the Level unit is V (volts). If the Source Type is set to current, the Level unit is mA (milliamps).

Important! DC Connect levels and settings cannot be changed with subsequent Stimulus Steps when Preload Stimulus is selected. DC Connect will use the settings of the first Stimulus Step until a new sequence is loaded. Subsequent Stimulus Steps will be ignored.

DC Output - rate of change

When controlling DC Connect via USB with SoundCheck, the maximum source level stepping rate is about 5 steps per second. When controlling the source level with an audio interface (Analog Control), the maximum stepping rate is 500 steps per second. This means that the Time/Step, as set in the Stimulus Editor, can be as small as 2 mSec.

Dynamic Control In Sequence

If "Preload Stimulus" is not selected in the Configure Sequence menu, DC Connect will switch to the settings of each DC Connect Stimulus Step as they occur in the sequence. This can be useful if the output of DC Connect needs to change during the operation of the sequence.

The example sequence in ***Figure 11-24***, shows a DC Connect stimulus set to output 1.5 VDC when the sequence is started. After the Broadband RMS Analysis is complete a second DC Connect Stimulus switches to 0 VDC, shutting the device under test off, until another test is made. The Configure Sequence window for the overall sequence shows that Preload Stimulus is not checked.

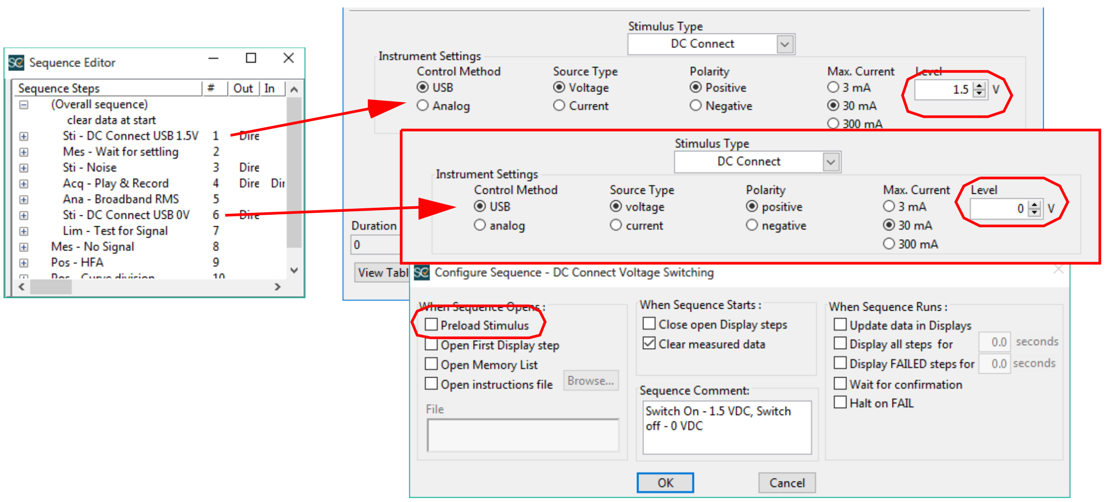


Figure 11-24: DC Connect Voltage Switching in Sequence

Control Method: Analog

Multiple DC Connects are not supported when using DC Connect in Analog Mode. See **Using Multiple DC Connects** in the latest DC Connect manual.

This method allows you to have Dynamic control over the output level of DC Connect. Analog Control provides a much faster rate of change than what is available with USB control. See ***DC Output - rate of change on page 132.***

When the Control Method is set to analog, the **Level** field is replaced with the **Settings/Sweep** selector. The remaining settings allow for selection of Source Type, Output Polarity and Maximum Current. Confirmation of these settings is indicated by LEDs on the DC Connect front panel.

When **Analog Sweep** is selected, the controls allow you to construct a voltage source (or current source) **stepped** waveform. ***Do not use a compound stimulus as it is not compatible.***

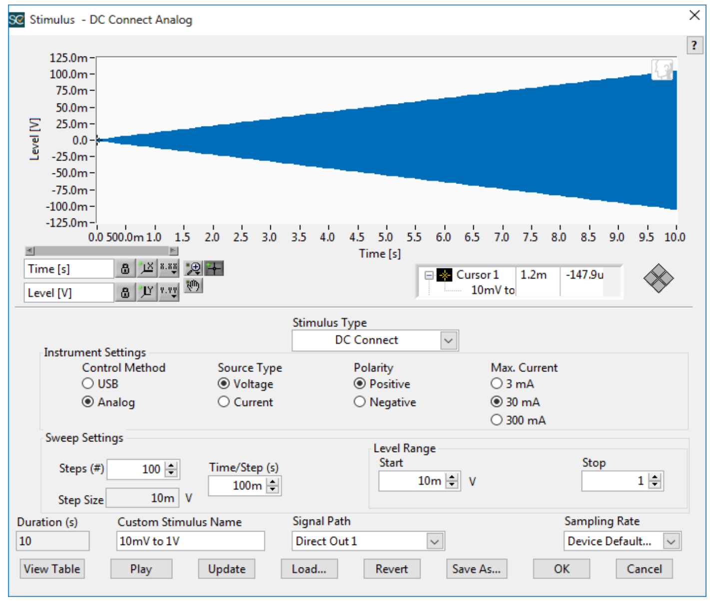


Figure 11-25: Analog Sweep Settings

These are the same controls used in the audio log amplitude sweep stimulus type, and they operate in the same way regarding number of steps, Start and Stop levels, and the View Table function. Here the units are V or mA. See **Signal Parameters for Amplitude Sweep Excitation on page 128**.

Note: Please refer to the DC Connect Instruction Manual for more information on this product. This can be found on the Listen website; **https://support.listeninc.com/hc/en-us/sections/200836954-Hardware-User-Manuals**.

Two Tone Stimulus

When you play two tones in a non-linear system, they interact in such a way that frequencies are present at the output that were not part of the stimulus. These by-product frequencies are different linear combinations of the two original frequencies and are called Intermodulation Products. Intermodulation Products are highly undesirable since they have no harmonic relationship to the original signal.

Types of two-tone stimulus available:

- **Intermodulation stimulus**: This superimposes a sweeping frequency tone against a fixed frequency tone. The fixed slave tone is usually a low frequency tone.

  For IM distortion, the Stweep Master and fixed Slave frequencies should be set according to the equation:

  $$f_{min} \geq f_{slave} + 14/T$$

  where T= the minimum step duration and $\mathbf{f_{min}}$ is the minimum frequency of the Master Tone.

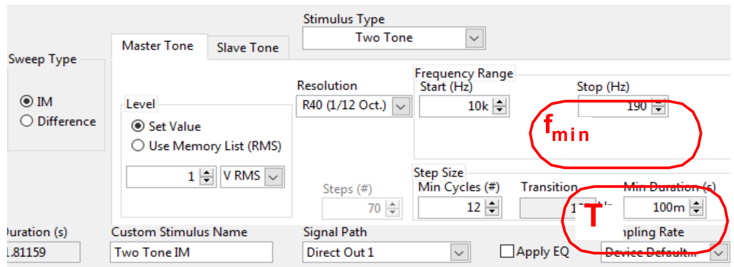


Figure 11-26: Two Tone IM Stimulus

  This method insures that there are at least 14 beat cycles in each step and the closest IM products can be resolved with good accuracy.

  For example: If $\mathbf{f_{slave}}$ is set to 43.1 Hz and T= 100 ms, then $\mathbf{f_{min}}$ should be set to at least 183 Hz in order to yield good results. In the Stimulus Editor, pick the closest value available depending on the resolution of the stimulus. For the example in ***Figure 11-26***, the minimum frequency for a resolution of R40 is 190 Hz. (Refer to ***Stweep Table - ISO Stepped-sine Frequencies on page 592***.)

  As a sanity check, in the Analysis Step, select the stimulus waveform as the “Response Waveform In”. This allows you to compare it to itself. Apply the analysis and there should be no IM distortion in the result.

- **Difference Frequency stimulus**: This consists of two sweeping tones which are separated by a specified frequency interval. This frequency interval can be a fixed difference or a fixed ratio. Similar to Intermodulation stimulus, the difference between the two frequencies should be greater than or equal to 14/T.

Application to Loudspeaker measurements

Intermodulation distortion is useful to detect amplitude and Doppler modulations on high frequencies when a low frequency signal produces a large excursion of the diaphragm.

Difference Frequency distortion is useful to detect distortion at high frequencies, where single tone harmonic distortion would fall far out of the frequency range of the loudspeaker or the ear.

For more details on these techniques see: Steve Temme, “Audio Distortion Measurements”, Bruel & Kjaer, Application Note BO 0385-11.

Note: An example sequence is included in the Default Sequences: C:\SoundCheck 18.1\Sequences\How To examples\IM Distortion.sqc and Diff Distortion.sqc.

IM Distortion Master and Slave Tone Settings

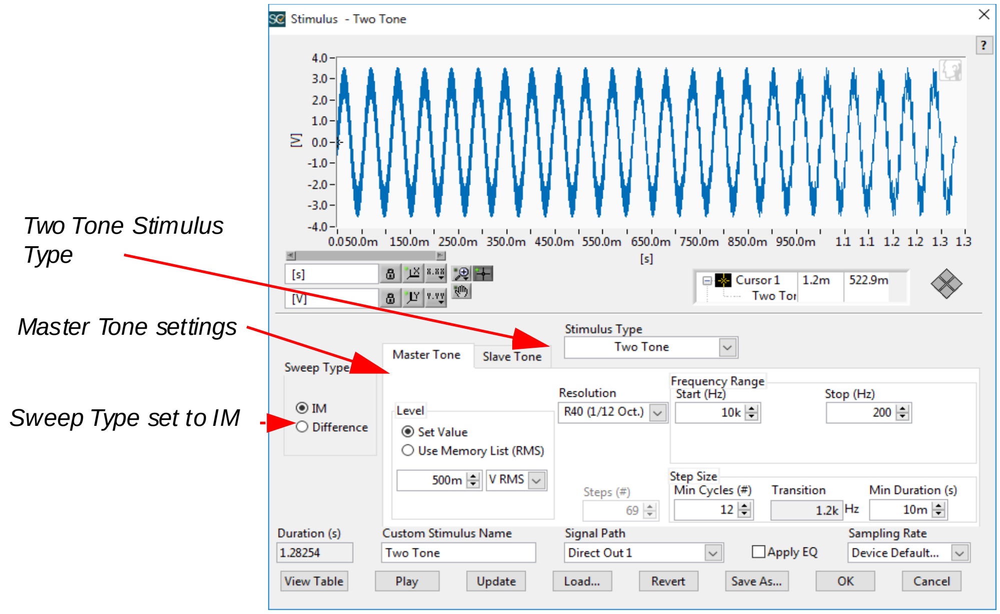


Figure 11-27: Stimulus for IM Distortion – Master Tone Settings

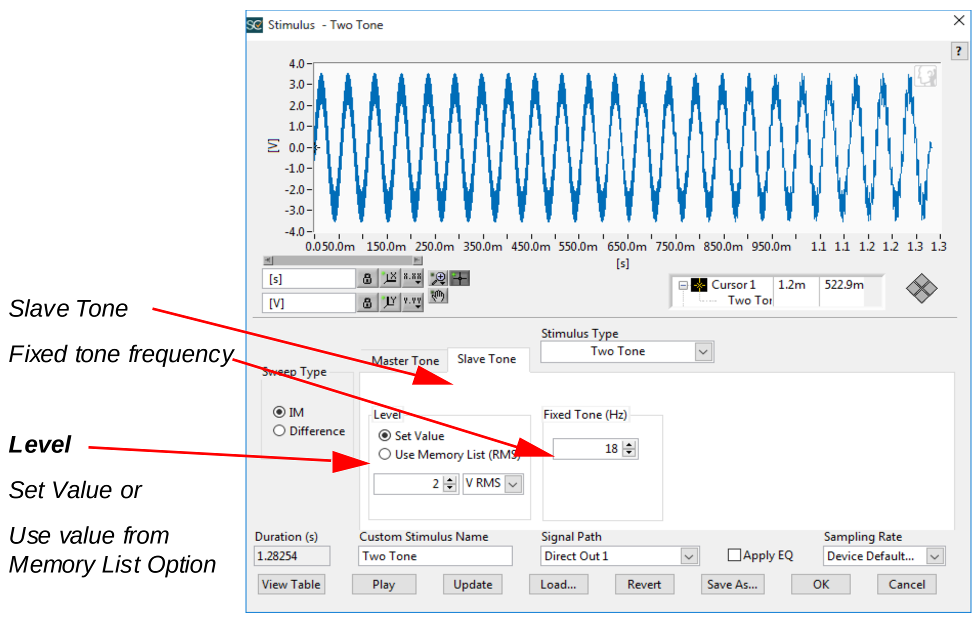


Figure 11-28: Stimulus for IM Distortion – Slave Tone Settings

Sweep Type - Difference

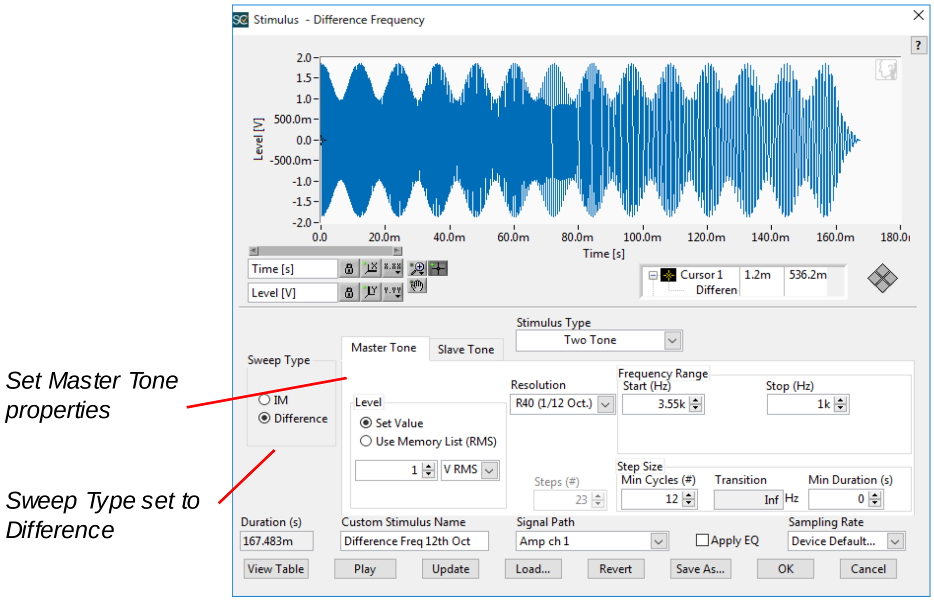


Figure 11-29: Stimulus for Difference Distortion – Master Tone Settings

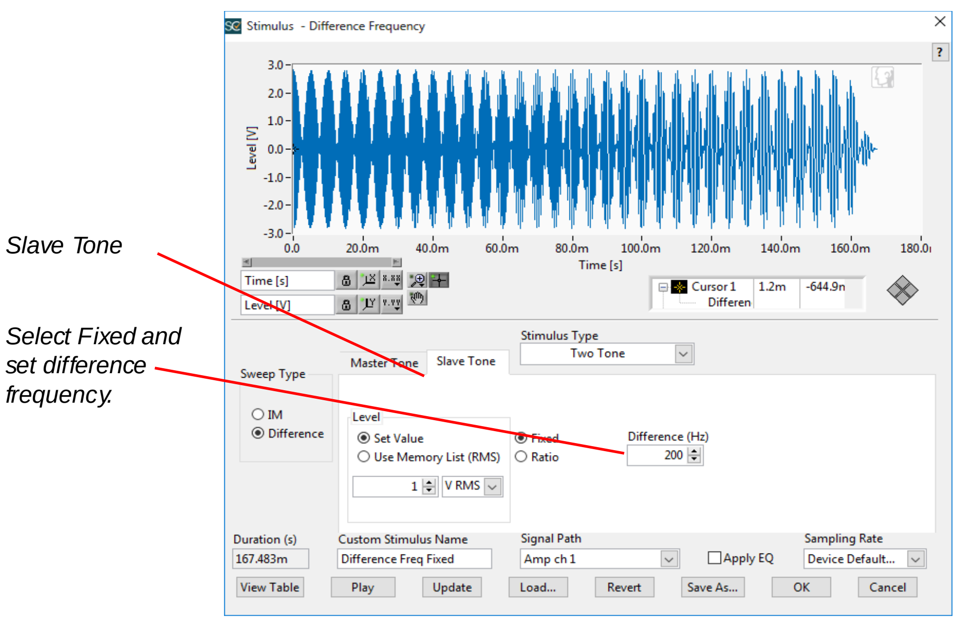
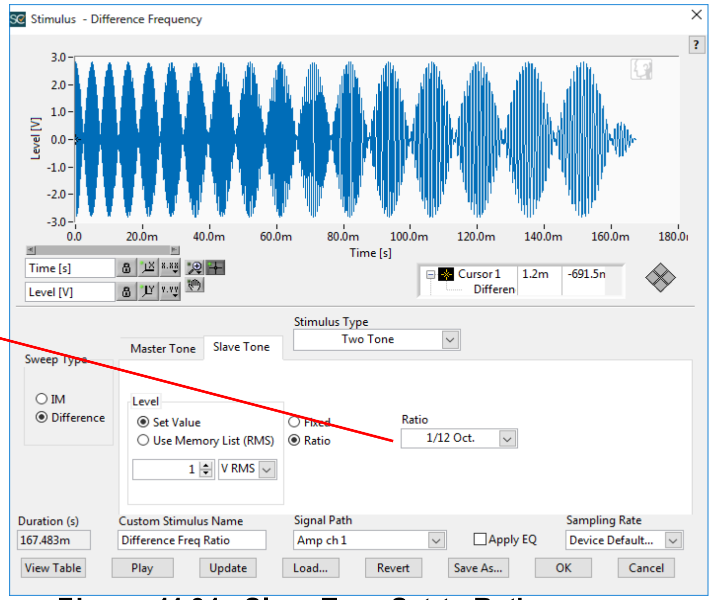


Figure 11-30: Stimulus for Difference Distortion – Slave Tone Settings



As an alternative the Slave Tone can be set to Ratio.

Figure 11-31: Slave Tone Set to Ratios

Active Speech Level Stimulus Control

(requires active speech module p/n 2033)

In telephony standards (IEEE, TIA), speech stimuli levels are often specified in terms of Active Speech Level (ASL). In ASL, the pauses are ignored and the level is set only for the active portion.

- Select WAV under Stimulus Type
- Select the WAV file to use

  The RMS level of the WAV file itself can be shown in **dB ASL**

  Remember that **ASL** is a weighting and not a unit
- From the **Level** drop-down list select dB ASL, as shown in ***Figure 11-32***, and set the level of the stimulus
- The Active Speech Module calculates the ASL of the WAV file rather than its average level

  ASL is calculated according to P56 Method B

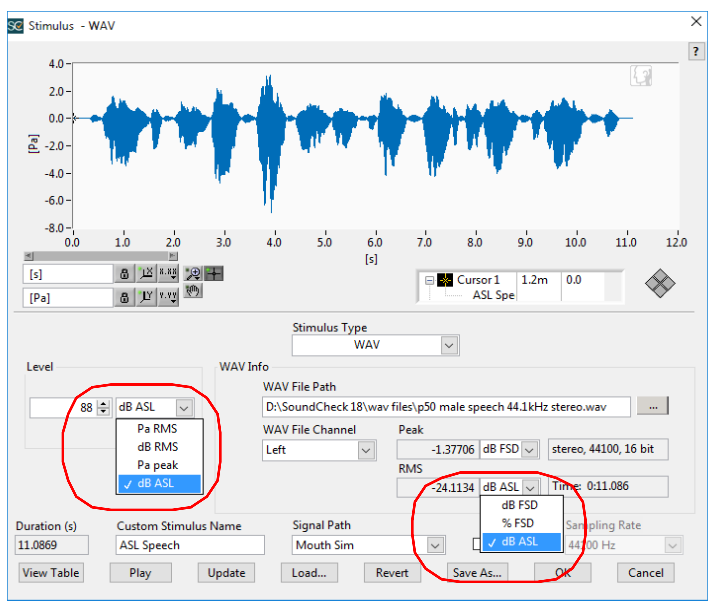


Figure 11-32: Active Speech Level

Noise

Pink and White noise stimulus is available with variable Duration, Band limits (Freq-min and Freq-max) and control of RMS Level in physical unit (linear or dB).

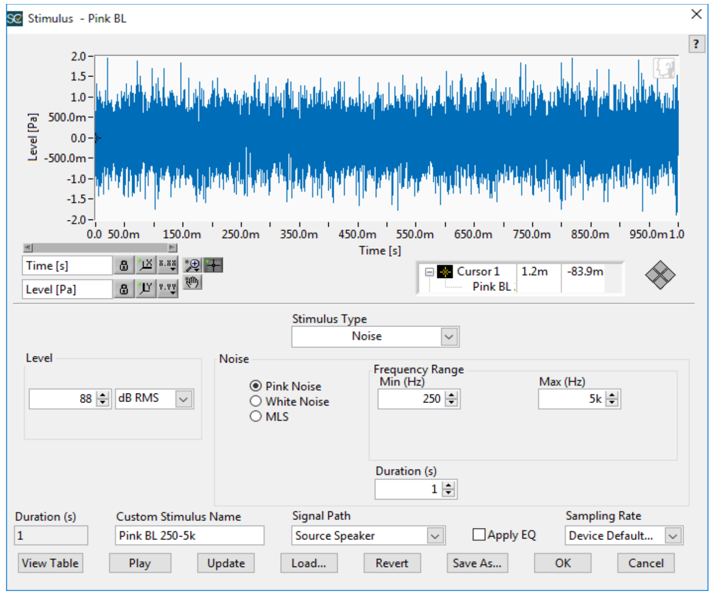


Figure 11-33: Band Limited Pink Noise

- Pink Noise: Noise with a continuous frequency spectrum and with equal power per constant percentage bandwidth. For example, equal power is any one-third octave band.
- White Noise: Noise with a continuous frequency spectrum and with equal power per unit bandwidth. For example, equal power in any band of 100 Hz width.
- MLS (Maximum Length Sequence): A special kind of white noise with low crest factor. Technically it is a square wave with randomly varied duty cycle.
  Therefore, the crest-factor is 1, which is only true for full-band MLS (DC to Nyquist). If the signal is filtered or band-limited, the crest factor may rise up to 4, as with standard white noise. In SoundCheck the MLS signal is intended to be used as a stimulus for transfer function measurements (Analysis algorithm = transfer functions). It will yield about the same results as the standard (Gaussian) White Noise also available in Stimulus. Notably, it is a legacy stimulus made available in SoundCheck for comparison purposes with other acoustic measurement systems.

The white noise and pink noise used in SoundCheck have a Gaussian amplitude distribution. Theoretically, the crest-factor is infinite. Of course that doesn't happen in a WAV file. Practically, the expected crest-factor can be anywhere between 3 and 5. The longer the wave file, the greater the chance of being close to 5.

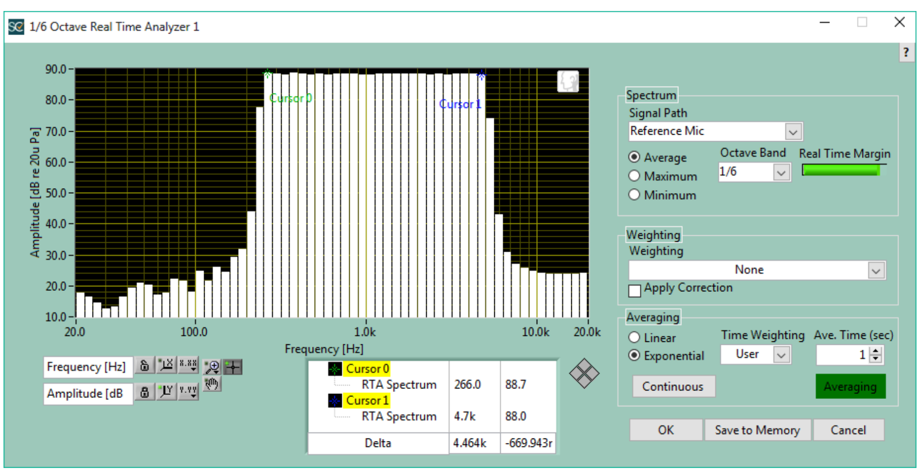


Figure 11-34: RTA display of Band Limited Pink Noise

Multitone

A Multitone is an ensemble of tones that are regularly spaced in frequency. The amplitudes are equal and the phases follow a deterministic mathematical law so that they are optimized to lower the crest-factor.

- Choice of Fmin, Fmax
- Choice of resolution (frequency spacing): R10 to R80, User Defined Log and User Defined Linear
- Control of Duration (s)
- Control of global RMS or Peak level

Frequency Rounding

An option has been created to align the frequencies of tones in order to round integer values (e.g.multiple of 5). The **Rounding Value** chosen in the Stimulus Step must be the same as the **Curve Resolution** specified in the Analysis Step **Frequency tab**. This helps to avoid leakage and makes the multitone analysis more accurate. The example in ***Figure 11-35*** shows the Stimulus Rounding Value set to 5 Hz with the corresponding Analysis Step set to 5 Hz Curve Resolution. By doing this, the spacing of the stimulus tones match the FFT spacing in Analysis.

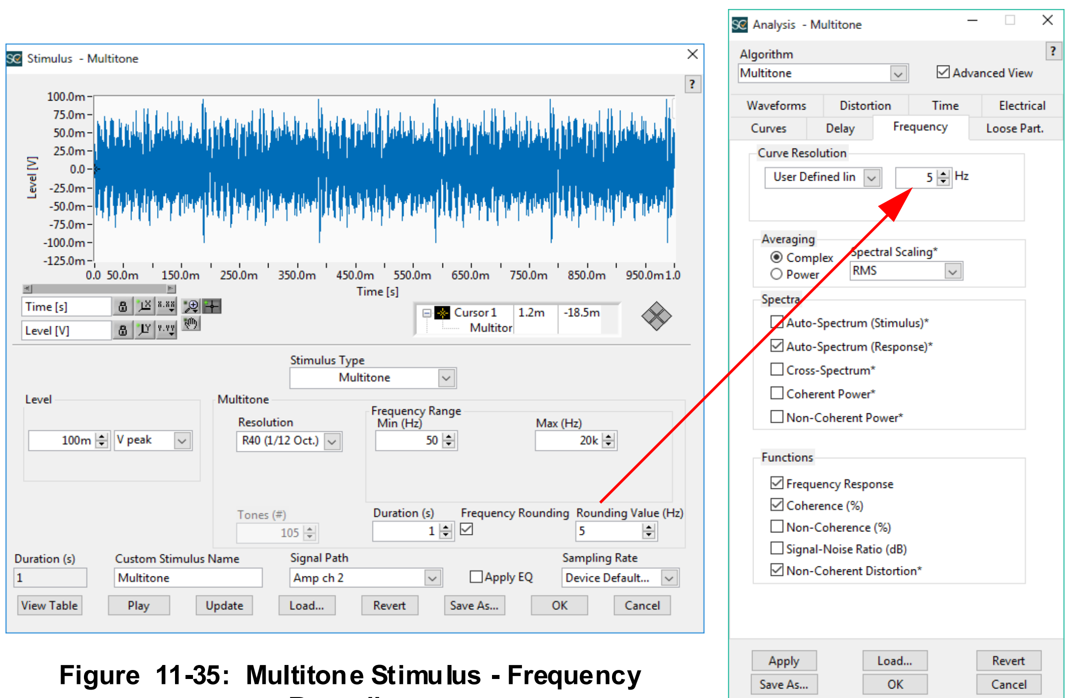


Figure 11-35: Multitone Stimulus - Frequency Rounding

The Stimulus Editor has two unique fields for Multitone Stimulus:

- Check-box for Frequency Rounding
- Rounding value control in Hz. Min rounding value= 0.1 Hz. Default= 5 Hz. All frequencies are rounded to the nearest Rounding Value. Frequencies that are duplicated, as a result of the rounding process, are deleted.

Multitone is repeatable (Not Random) which is useful for MP3 testing. You can now recreate in memory, a perfect copy of the multitone stimulus that has been transferred to an MP3 player. By playing the stimulus on the MP3 player and comparing it to the stimulus in memory, the response of the MP3 player can be analyzed.

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Acquisition Editor

Acquisition determines how test signals (Stimuli) are played and how measured signals are recorded or analyzed in a sequence. The flexibility of the Acquisition Step allows measurements to be made in a variety of ways.

The Acquisition Step establishes:

- The relationship between the Input/Output Signal Paths defined in the System Calibration Configuration
- The Stimulus that is to be applied on each Output Signal Path
- The resulting Waveform at each Input Signal Path

A simple table interface is used to show the Signal Path names and the Input/Output waveforms.

Select Acquisition (Ctrl + Shift + A) from the **Setup** drop-down list on the main SoundCheck® menu bar to view and change the Acquisition Steps in a sequence.

Features

- Level & Cross-Correlation Trigger - Offers improved performance over “Level & Frequency Trigger”. Requires optional module 2006 - Time Selective Response.

  See ***Level & Cross-Correlation Trigger (optional module required) on page 147***.
- Frequency Trigger - Allows frequency-based triggering of acquisition from an external source using a trigger tone at the beginning of a test sweep. This allows for improved accuracy over previous level-based triggering when capturing responses from a device where you don’t have direct access to the microphone or speaker. This is otherwise known as Open Loop testing for Smart Speakers and other voice-controlled devices. See ***Level & Frequency on page 146***.
- Listen Hardware Gain - When using Listen hardware, you can automatically adjust the microphone preamplifier gain in order to optimize the dynamic range of the test
- Table interface for selecting Play and Record Signal Paths
- Multiple Signal Paths are selected by holding down the **Control** key while clicking on Signal Paths in the table
- Capability to mix different sampling rates within a measurement, using different output and input interfaces for each of the different sample rates
- Max FSD - Record level monitoring allows you to optimize the dynamic range of the measurement, resulting in a better SNR ratio and therefore more accurate measurements
- Multiple Virtual Instruments can be opened simultaneously in the acquisition step for automation.

  See ***Virtual Instruments on page 149.***

Record Level Monitoring - Max FSD

The Input and Output Max FSD values are added to the Memory List for recorded waveforms. These values can be used to optimize the signal to noise and to show that the signal is in a “comfortable” operating range; i.e., not clipping.

Gain Field

This feature allows those using Listen hardware to automatically adjust the microphone preamplifier gain in order to optimize the dynamic range of the test.

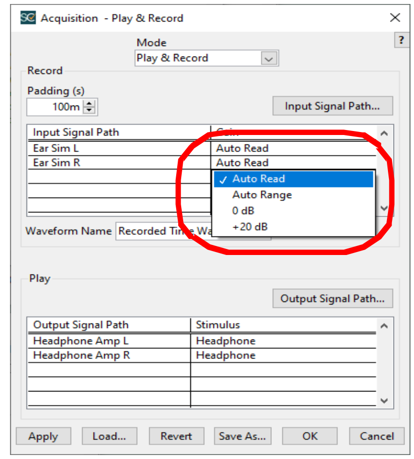


Figure 12-1: Listen Hardware Gain

- Available for input signal paths only
- Overrides the Startup Default Gain setting
- This can only be selected if **Auto Dev / Auto Ch** is set for the selected input signal path in the Calibration Editor

  See ***Listen Hardware - Auto Device / Auto Channel on page 84***.
- Click on the **Gain** field drop down to select available options
- **Auto Read**

  Reads the gain value from the selected Listen Hardware device channel
- **Auto Range**

  When selected, SoundCheck will monitor the digital headroom of the audio interface (Max FSD) and if necessary increase or decrease the preamp gain, selecting the optimal setting for maximum signal to noise ratio. If a change to the gain is made, the acquisition step runs again with the new settings.
- **Select Value**

  You can also select a gain from values available for that Listen Hardware device

---

Important! ***Switching Listen Hardware from Maximum Gain to Minimum Gain in the Acquisition Step is not recommended. This does not allow the input gain circuit sufficient time to stabilize. If you need to switch from Max Gain to 0 or Minimum Gain we recommend that you use a Listen Hardware Message step with a 500 mSec wait time to allow for settling.***

Play & Record

This used to simultaneously play a stimulus and record the time response. This mode is typically used when measuring the frequency response of a device such as a speaker or microphone. The Recorded Time Waveform (RTW) is then used in an Analysis Step, along with the Stimulus Waveform, to derive the response of the device under test, e.g., magnitude, phase, distortion, etc.

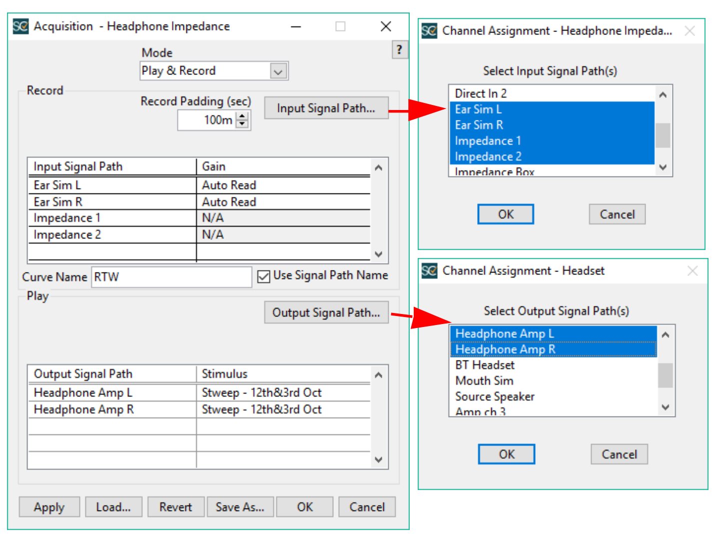


Figure 12-2: Play and Record Acquisition Step

Curve Name - Waveform Name

This field allows you to use a custom name for the acquired recorded time waveform. (Not available in all modes.)

When acquiring a time signal using Record, Play & Record or Oscilloscope virtual instrument mode, the recorded time waveform will appear in the **WFM** tab of the *Memory List*.

Stimulus waveforms and Recorded Time Waveforms can be save to disk in Autosave Steps or manually from the Memory List WFM tab. Saving as a Waveform allows you to keep the physical units associated with the file.

Signal Path

The Signal Path Name, as seen in the input or output section of the Acquisition Step, is assigned in the System Calibration Configuration.

The default name, "Recorded Time Waveform", is used unless a custom name is entered.

When an Play and Record Acquisition Step is added to a sequence, you are prompted to select the Input Signal Path(s) and Output Signal Path(s). This can also be edited by clicking on the Input/Output Signal Path buttons. (To select multiple Signal Paths, hold down the **Control** key while clicking on Signal Paths in the drop-down list.)

Tip: When appending *Long Channels* names to *Waveform Names* it may be a good idea to shorten the name. RTW is used in place of Recorded Time Waveform in ***Figure 12-2:***.

Use Signal Path Name

This allows you to change the name of the recorded time waveform by appending the **Signal Path Name** to the **Waveform Name**. The example in ***Figure 12-3:*** shows the original name "Recorded Time Waveform".

By selecting "Use Signal Path Name", [Reference Mic] is appended to the waveform name.

The resulting waveform appears in the *Memory List WFM Tab*.

For more information on the Memory List refer to ***Display Editor and Memory List on page 321***.

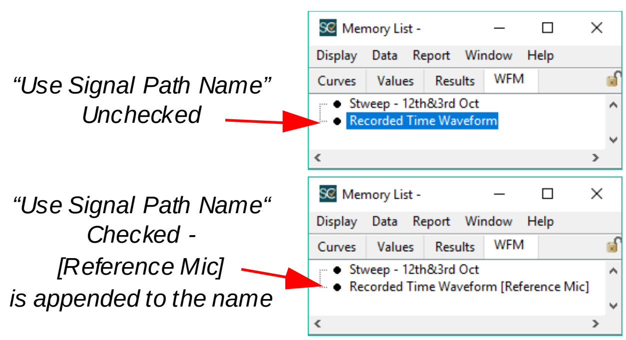


Figure 12-3: Waveform Name in Memory List

Stimulus Selection

You can choose the specific stimulus to play for each Output Signal Path. Any stimulus in memory can be selected. (Only available in Play & Record mode.)

The stimulus is selected from the drop-down list as shown in ***Figure 12-4:***.

> ***Important:*** ***The sampling rate and the bit depth must be the same for all inputs and outputs of a specific audio interface.***

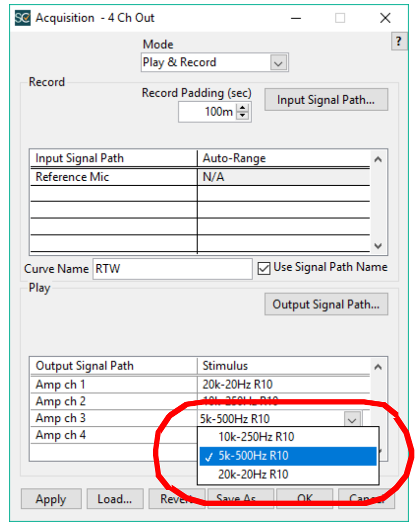


Figure 12-4: Select Stimulus

Time (sec) - Record Mode

Time (sec) specifies the total record time. This is typically used for Open Loop measurements or when measuring background noise or noise from the DUT. The minimum Record time is 50 mSeconds.

See ***Record Only on page 145***.

Record Padding (sec) - Play and Record Mode

This allows you to add extra time to the Record process to capture information beyond the length of the stimulus. This can be used to compensate for "Time of Flight" when there is a great distance between the mic and the source. (Especially useful for Telephony measurements where there can be a long delay due to the signal chain.)

Record Delay (sec) - Virtual Instrument Mode only

The Record Delay (sec) is the time in seconds before the acquisition of signal starts.

> ***Important:*** ***SI units are used throughout SoundCheck. A time of 0.5 seconds will be represented as 500 mSec.***

Input Signal Path/Output Signal Path

Input and Output Signal Paths are selected here. Any signal path that has been set up in the System Calibration Configuration can be selected. Refer to ***Calibration Configuration on page 79*** for more information.

Configure Record / Configure Generator

Configure Record / Configure Generator is used to set the properties of the Virtual Instruments that are selected in the **Mode** field. (Not available in all modes.) Refer to ***Virtual Instruments on page 455*** for more information.

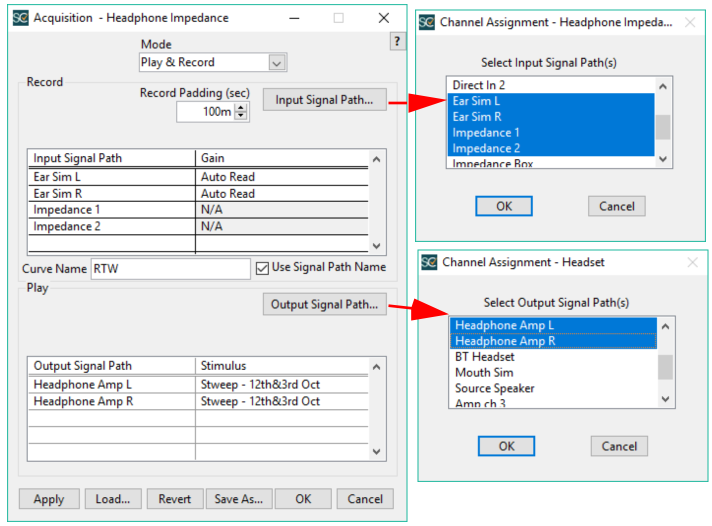


Figure 12-5: Signal Paths

Mode

Measurements can be made in the following ways:

- **Record** a time signal from a device under test (DUT).
- **Play** a stimulus into a DUT **and Record** a DUT's time response.
- Use **Virtual Instruments** such as the Multimeter, Oscilloscope, Spectrum Analyzer, or Real Time Analyzer (RTA) in a sequence. (Refer to ***Virtual Instruments on page 455*** for more details.)
- Use the Signal Generator in combination with the Multimeter, Oscilloscope, Spectrum Analyzer, or RTA.
- **Read from DC Connect** (Optional hardware required)

The Acquisition Step allows you to combine SoundCheck's Virtual Instruments to play test signals and acquire signals for analysis. The following are the different Acquisition modes available:

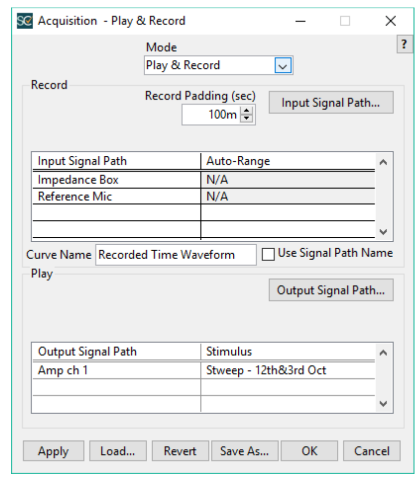


Figure 12-6: Play and Record Acquisition Step

Play & Record

Simultaneously play a stimulus and record the time response. This common mode is used when measuring a device's response to a test signal and then analyzing its response (e.g., magnitude, phase, distortion) in an Analysis Step. See ***Figure 12-6***.

Record Only

When the Acquisition Mode is set to Record, Triggering can be used to automatically capture a spectrum when the signal level crosses the threshold set by the **Trigger Level** field.

See ***Triggered Record Parameters on page 146***.

Triggered Record Parameters

The trigger options are similar to what is available in the Scope FFT Virtual Instrument along with the addition of selecting **Trigger Types**.

Mode

Must be set to **Record** and **Triggered** must be selected

Time (sec)

This is the total time acquired by the acquisition step AFTER a successful trigger. This time must include the "**Total Length of the stimulus**" along with some overhead to ensure that the whole signal is captured.

Input Signal Path

Any active signal path from the Calibration Table can be used

Curve Name

The name for the recorded time waveform that appears in the Memory List waveform tab. Check "Use Signal Path" to append the path name to the Curve Name, e.g.: RTW [Reference Mic].

Trigger Type

<u>*Level*</u>

- Start acquisition when the signal level at the input goes above the value set in **Trigger Level**

<u>*Level & Frequency*</u>

Level & Frequency is very useful for "Open Loop" testing where the device under test does not have a direct connection to the SoundCheck audio interface, e.g. testing Smart Speakers and other voice-controlled devices.

- **Frequency Trigger** allows frequency-based triggering of acquisition from an external source using a trigger tone at the beginning of a test sweep. This allows for improved accuracy over previous level-based triggering when capturing responses from a device where you don't have direct access to the microphone or speaker.
- First, the step waits for the signal to exceed the **Trigger Level** value. The step then looks at 100 mSec of the input signal to see if a pure sine tone, greater than 200 Hz, is present. Once these two parameters are satisfied, acquisition begins.
    - *The input signal must contain a single sine tone, above 200 Hz, that is at least 6 dB above the noise floor of the spectrum*

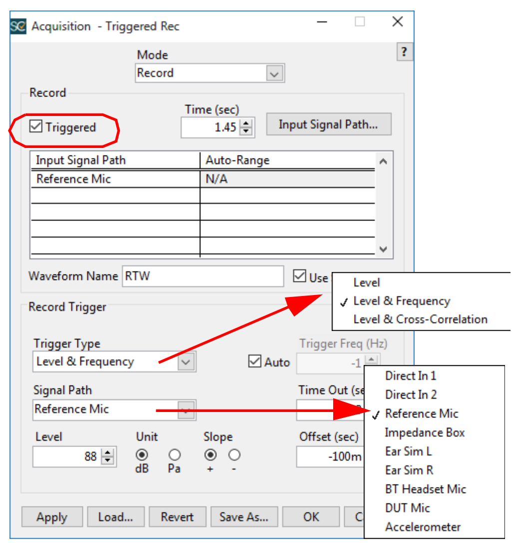


Figure 12-7: Triggered Record

- When **Auto** is selected the step triggers on the first pure tone detected and reports the frequency in the field, ***Trigger Freq (Hz)***. *See* ***Figure 12-7***. This is also added to the Memory List Values tab and called "**Trig Freq**". This allows you to display the detected frequency. This is only available in Auto mode.
    - If no sine tone is detected the Trigger Freq field will show "-1". The Trig Freq value in the Memory List will be unpopulated (empty circle).
- Uncheck **Auto** to manually enter the frequency of the expected trigger tone in the field, ***Trigger Freq (Hz)***. This should be +/- 10 Hz of the expected trigger frequency.

Level & Cross-Correlation Trigger *(optional module required)*

The Cross-Correlation Trigger requires a Log Frequency Sweep: optional module **2006 - Time Selective Response**.

- AKA "Chirp Trigger"
- Offers improved performance over Level & Frequency Trigger
  - More robust
  - Less susceptible to false triggers since it is looking for the exact 'chirp' in the input signal
- Requires Log Frequency Sweep set as **"Analyze - No"** in the compound stimulus. This must be elected in the **Stimuli** field.

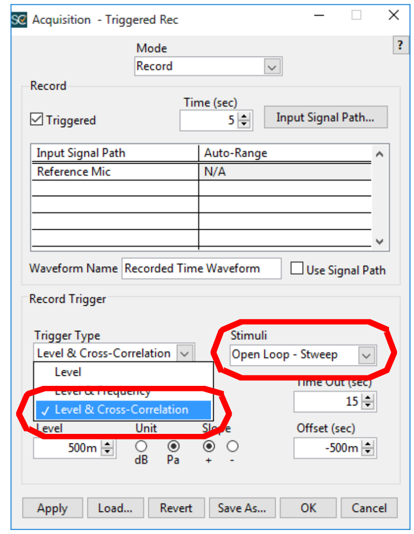


Figure 12-8: Level & Cross-Correlation Trigger

- A log frequency sweep from 1 kHz to 5 kHz with a duration of 50 mSec to 200 mSec is recommended.
- The frequency range of the log frequency sweep must be in the pass band of the device under test:

  e.g.: Testing a subwoofer would require a lower start and stop frequencies. A longer Chirp may be required to have enough cycles for accurate triggering.

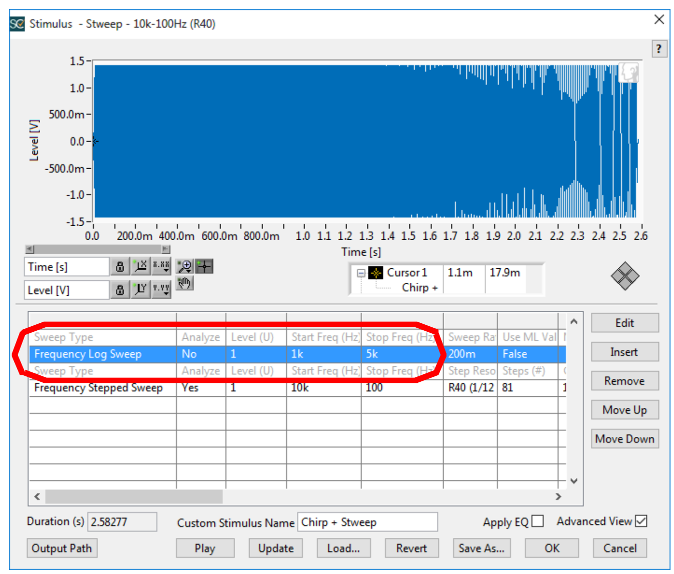


Figure 12-9: Compound Stimulus with 'Chirp'

Signal Path

This is the signal path used for triggering. The Recorded Time Waveform from this path will not be output to the Memory List unless it is also added to the Input Signal Path table.

- The example in ***Figure 12-10*** shows Reference Mic selected as the **Input Signal Path**
- The Reference Mic path is used for the measurement. In this case, the result waveform "**RTW [Reference Mic]**" is used in subsequent analysis steps.
- Direct In 1 is used for the **Record Trigger** > **Signal Path**

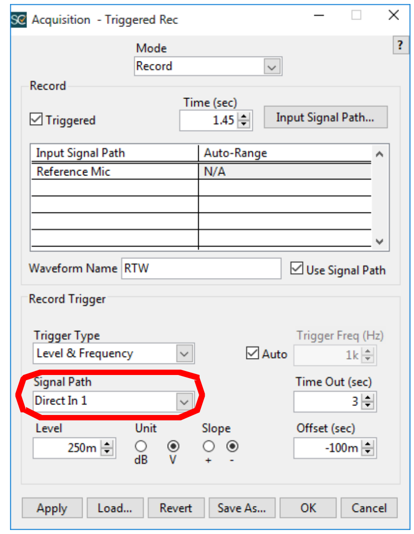


Figure 12-10: Trigger Signal Path

Level

This is the threshold level required for a successful trigger. The step triggers at the instant the signal crosses the threshold.

Unit

- This can be set to either Physical Units or dB
- The Physical Units will update according to the selected **Trigger Signal Path,** e.g.: Pa for Microphone or V for Direct In
- Linear values can be positive or negative
- If dB is selected, the trigger threshold is always a positive linear value
- The trigger value has the same dB reference as the **Trigger Signal Path**

Slope

Selects whether the acquisition is triggered on a positive going signal or a negative going signal

Offset

Sets the amount of time that the Acquired Signal is shifted, relative to the point at which it was triggered.

A negative offset indicates that the signal will be shifted to the right of "Zero Time Start" by the time that is in the field, e.g., -100 mSec

Time Out

This is the amount of time the Acquisition Step waits for a successful trigger to occur before continuing to the next sequence step.

If the "Time Out" is exceeded, the state of the step will be "Fail". This can then be used in step configuration to **Halt on Fail** or **Jump to** a different step in the sequence.

The example in ***Figure 12-10*** shows the Time Out set to 3 Seconds. If the trigger does not occur within 3 seconds, the step state will "Fail" and the configuration of the step shown in ***Figure 12-11*** will stop the sequence operation (Halt on Fail).

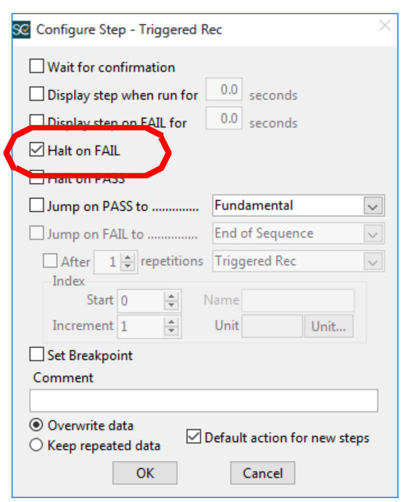


Figure 12-11: Halt on Fail

Virtual Instruments

Multiple Virtual Instruments can be opened simultaneously in the Acquisition Step for automation in a sequence.

- Select **Virtual Instruments** from the **Mode** drop-down list
- This displays a table showing all the virtual instruments in that step
- **Right-click** a row to add or configure an instrument or select Configure All
- Virtual instrument configuration files can be loaded into the acquisition step for quick setup in any sequence
- Under **Panel**, select **Show** to have the Virtual Instrument visible during sequence run or **Hide** to have it run in the background

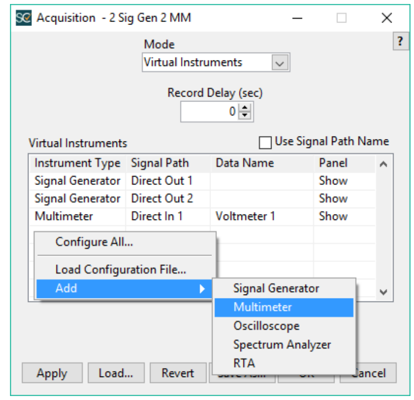


Figure 12-12: Multiple Virtual Instruments

Show/Hide

- You can't **Hide** a Virtual Instrument that is configured to run Continuously or Exponential. The editor must be set to Duration or Linear as shown in ***Figure 12-13*** in order to see the Show/Hide option.

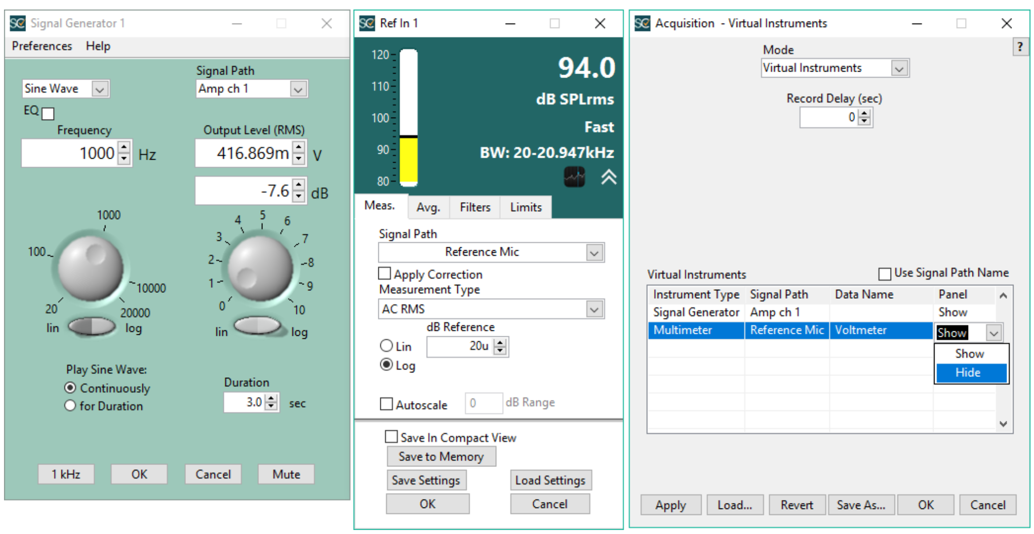


Figure 12-13: Show/Hide Panel

Rules - Virtual Instrument Acquisition Step Configuration

- When the virtual instruments in a step are set to Run Continuously, they are expecting you to hit the Enter key in order to stop the operation and close the step.
- If the virtual instruments in a step are set to Run **for Duration**, **Linear**, etc, the step will automatically close when the virtual instruments finish.
- In either case, the step should not be configured to **Wait for Confirmation**. Right-click the step in the Sequence Editor and Select **Configure Step** as shown in ***Figure 12-14*** to change the setting.

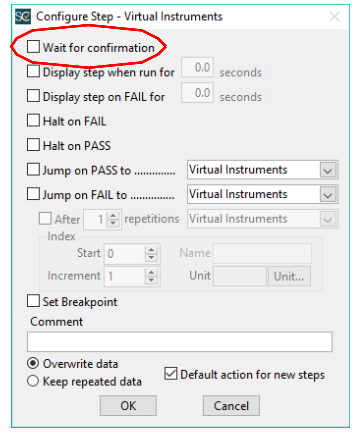


Figure 12-14: Step Configuration

Signal Generator

The *Signal Generator* can be used for manual control of a test signal during a sequence, (e.g., audible tuning of a transducer), or playing a WAV file (Refer to the ***Virtual Instruments on page 455***). Specify a Frequency and Output Level for a Sine Wave, or the file path and percentage of the originally recorded level of a WAV file.

- Select **Stimulus type** from the drop-down list
- Select **Signal Path**
- Set **Stimulus Frequency**, and Level in physical units
- Play a sine wave or a WAV file continuously or for a specified duration of time
- WAV files can also be configured to play a user defined (N) number of times
- The WAV file can also be equalized to a target spectrum stored in the System Calibration Configuration (Refer to ***Signal Generator on page 460*** for more information).

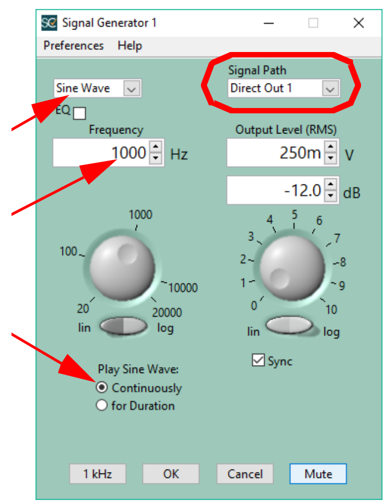


Figure 12-15: Signal Generator Options

- Sync allows you to Start, Stop and Mute multiple signal generators by clicking on only 1 button. It also synchronizes the phase of sine signals and the start of WAV files.

The EQ output function can be applied to all types of output signals available in the Signal Generator (Sine Wave as well as WAV and noise). When the EQ box is checked the EQ Out Correction curve that is created in the output calibration process is applied to the output signal. This allows you to equalize the response of an artificial mouth or anechoic chamber. EQ out correction curves are populated with data when the Speaker Equalization or Simulated Free Field calibration sequences are selected in the output calibration process.

See ***Equalization and Correction Curve on page 93*** in the Calibration chapter.

Multimeter

The *Multimeter* can be used for measuring overall signal level. Choose between linear and exponential averaging and the number of averages you wish to make.

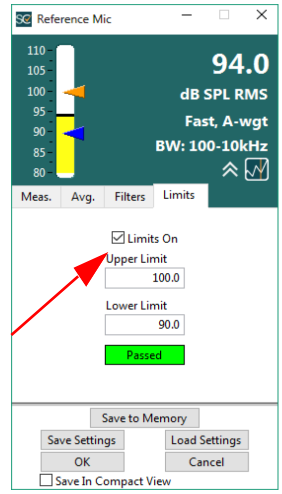


Figure 12-16: Multimeter Options

- **Linear -** The *Multimeter* will run for the duration set in the Time field
  - **Gain** - See ***Gain - Auto Read on page 457***
- **Exponential -** The *Multimeter* will run continuously until you click **OK** or **Cancel** in the *Multimeter* window
  - Averaging Time: Fast (250 mSec), Slow (2 Sec) or User Defined
- **Limits On -** The Multimeter Limits function can be used to set a visual Pass/Fail indicator on the multimeter. A Limit Step should be added after the Multimeter Acquisition Step to read the value and put the Result in the Memory List.
- **Hide Panel (Acquisition Step window) -** The *Multimeter* is not shown during the sequence run. If Exponential Averaging is selected, the **Hide Panel** option is not available since the Virtual Instrument requires user intervention to be stopped.
- **Data Name (Acquisition Step window) -** Enter a custom name for the Multimeter Value in the Memory List

For more information on the individual controls of the *Multimeter* refer to ***Multimeter on page 466***.

- The Multimeter step can generate is the Linear or Log values.
- These values are dependent on the calibrated values of the signal path selected in the Multimeter.
- The Pass/Fail state of the Multimeter Acquisition can be used for conditional branching.
- In some cases you may need to put the Meter in a "loop" while the DUT is being adjusted to produce a passing level. Have the second Limit Step "Jump on fail" back to the first Limit Step. When the level passes, the sequence will continue past the loop.

Important:* *The* Multimeter *frequency range is the full range of the audio interface based on its sampling rate. The broadband dynamic range of most audio interfaces is limited by their DC offset. Their AC dynamic range is typically much greater and sometimes it makes more sense to use the* Spectrum Analyzer *and power sum the "frequency range of interest" in a Post-Processing Step.

Oscilloscope (Scope-FFT/Time)

The *Oscilloscope* can be used to make sure the recorded waveform does not look clipped or distorted. Note that delta µs cannot be changed, since this is based on the sampling rate of the audio interface.

If you want the *Oscilloscope* to run for a preset time, choose **Lin** and the number of averages multiplied by the record time will determine the total measurement time.

If you choose **Exp**, the *Oscilloscope* will run continuously until you click **OK** or **Cancel** in the *Oscilloscope* window. If running as a step in a sequence you can click **Enter** or **Stop**.

If you do not want the *Oscilloscope* to appear during the sequence, check **Hide Panel**. As with the Multimeter, if exponential averaging is selected, the **Hide Panel** option is not available since the Virtual Instrument requires user intervention to be stopped. Refer to ***Virtual Instruments on page 455*** regarding functions available for the *Oscilloscope* via the Acquisition Step. The minimum Time value is 50 mSec.

Calculate Spectrum

- Off - FFTs are not being done in the background which makes it faster
- On - Collect data while the Scope is running and then switch to FFT scope to view or save the spectrum
- All spectrum are summable. As of SoundCheck 18, you can also calculate the Power Sum of a Waveform.

Gain

The Gain control for the Input Signal Path and Trigger Signal Path allows you to automatically read the gain from Listen hardware as well as Portland Tool & Die hardware. You can also select a gain value available for that device. It will automatically switch to the new gain when the step is run.

- See *Gain - Auto Read on page 457*

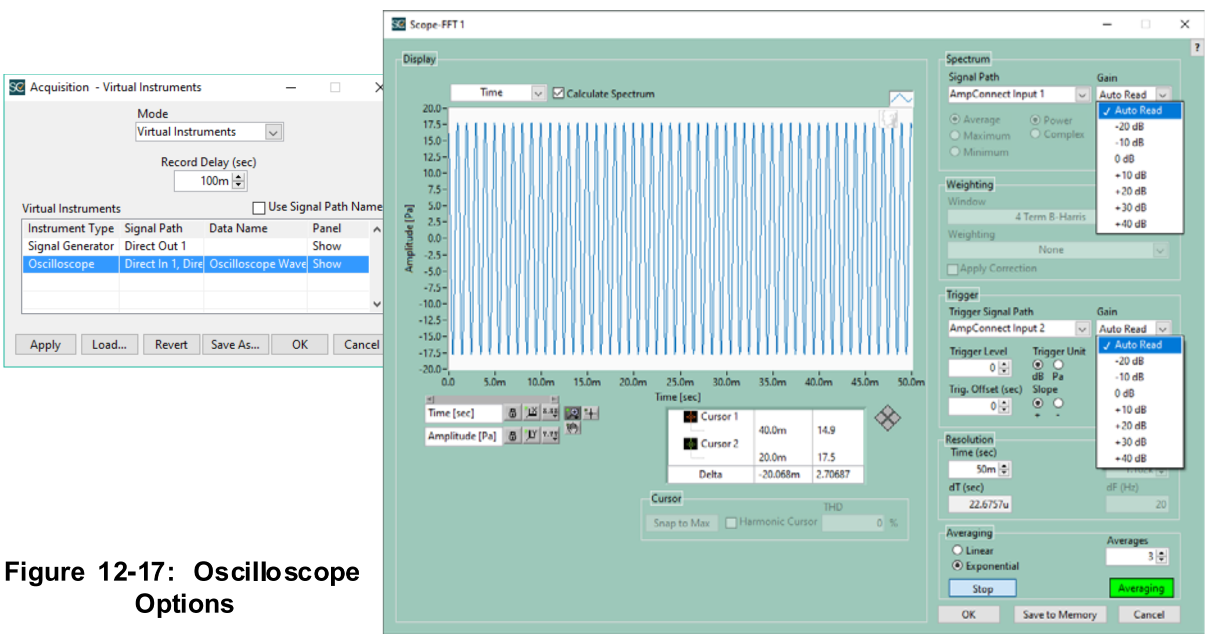


Figure 12-17: Oscilloscope Options

Important! ***Switching Listen Hardware from Maximum Gain to Minimum Gain in the Acquisition Step is not recommended. This does not allow the input gain circuit sufficient time to stabilize. If you need to switch from Max Gain to 0 or Minimum Gain we recommend that you use a Listen Hardware Message step with a 500 mSec wait time to allow for settling. See*** ***Listen Hardware Control Message on page 273******.***

Spectrum Analyzer (Scope-FFT/Spectrum)

Important: ***All spectrum are summable. As of SoundCheck 18, you can also calculate the Power Sum of a Waveform.***

The *Spectrum Analyzer* can be used for analyzing pure tones or noise coming from the DUT, since these signals are typically flat when plotted on a linear frequency scale. Lin will average for only the number of averages entered (e.g., 3). Exp applies an exponential time weighting where *Time Sec* is the exponential time constant tau (τ). Power averages the RMS amplitude of each FFT bin and excludes phase information. Complex averages the RMS amplitude of each FFT bin but includes phase information. Note that the delta µs is based on the audio interface's sampling frequency. Because of this, it can only be changed via the audio interface driver and/or switching jumper cables on the audio interface itself.

If you do not want the Spectrum Analyzer to appear during the sequence, check **Hide Panel**. As with the *Multimeter*, if exponential averaging is selected, the **Hide Panel** option is not available since the Virtual Instrument requires user intervention to be stopped. Refer to ***Virtual Instruments on page 455*** regarding functions available for the *Spectrum Analyzer* via the Acquisition Step.

The "**Snap to Max**" button on the Scope-FFT control panel moves Cursor 1 to the peak of the acquired spectrum. The Estimated Frequency and Estimated Level are shown in the fields at the top of the Spectrum Display. This function is available when the mode is set to Time or Spectrum but the cursor location is only shown when the mode is set to Spectrum as shown in ***Figure 12-18:***.

The FFT record length is set in seconds and/or number of Spectral lines. (Minimum value of 50 mSec.)

The Estimated Frequency and Level are shown in either mode. Clicking on the Harmonic Cursor will then plot and show the Harmonics on the FFT display as well as calculate the THD.

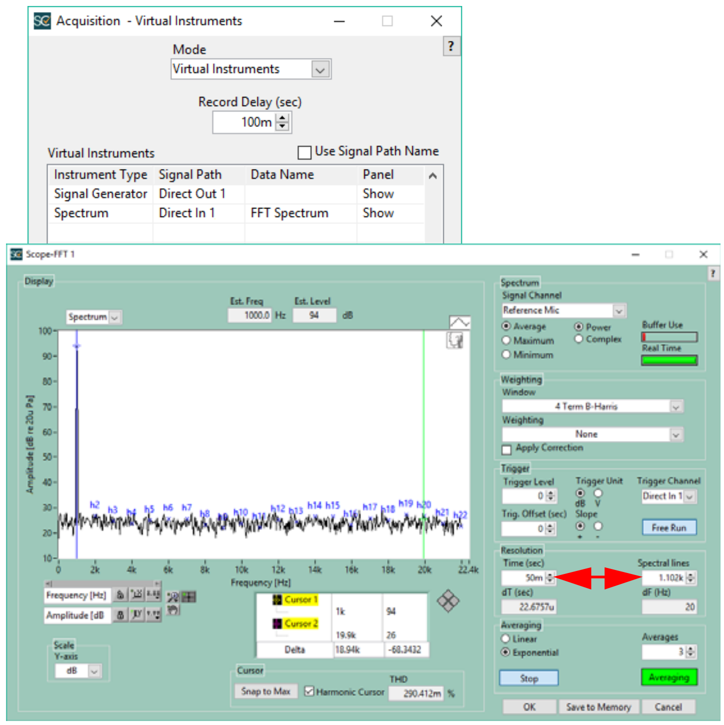


Figure 12-18: Spectrum Analyzer Options

Save to Memory

When "**Save to Memory**" is selected, the FFT Spectrum will be added to the Curves tab in the Memory List. If Snap to Max is selected before hand, the FFT Cursor values will be added to the Memory List: Est. Freq, Est. Level and THD. This value can then be shown in a Display Table as shown in ***Figure 12-19:***.

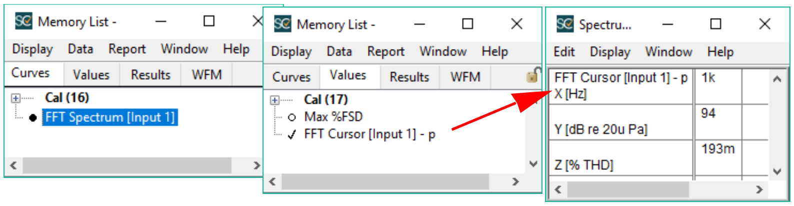


Figure 12-19: Memory List - Snap to Max Values

Real Time Analyzer

The *Real Time Analyzer* (RTA) analyzes a signal, using constant-percentage bandwidth (1-Nth octave) filters. This can be used to analyze background noise, since it is usually flat when plotted on a log frequency scale. The RTA has 1/1, 1/3, 1/6, 1/12, and 1/24 octave digital recursive filters. The upper frequency range is based on the audio interface's sampling frequency. The highest frequency that can be measured will be no more than one-half the audio interface sampling rate (Fsample). For example, if you are using an audio interface with a 44.1 kHz maximum sampling rate, the maximum measurement you can acquire will be at approximately 22 kHz. The Detector Time determines the time duration of the window to be sampled. Fast sampling averages every 0.25 Sec of data, and Slow averages every 2 Sec of data. You can input your own averaging time by using Other and entering a value in the field to the right.

As with the above Virtual Instruments, you can choose Linear or Exponential averaging. Additionally, you can choose A, B, or C Weighting to display, but not save, the overall weighted level of your measurement (this setting will not affect the display or saving of the acquired data).

If you do not want the RTA to appear during the sequence, check **Hide Panel**. As with the *Multimeter*, if exponential averaging is selected, the **Hide Panel** option is not available since the Virtual Instrument requires user intervention to be stopped. Refer to ***Virtual Instruments on page 455*** regarding functions available for the *Real Time Analyzer* via the Acquisition Step.

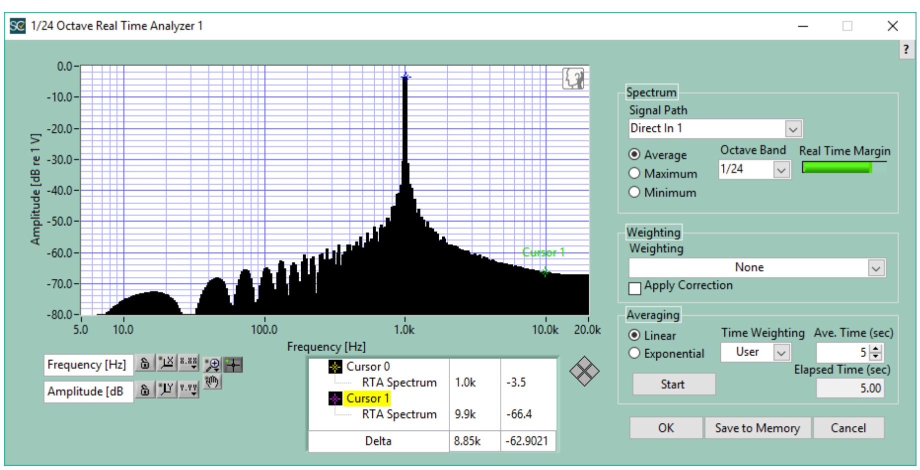


Figure 12-20: RTA Options

Signal Generator & Multimeter

The *Signal Generator* and *Multimeter* are typically used for trimming output or input levels on electronic devices, e.g., active loudspeakers or crossovers. Refer to ***Virtual Instruments on page 455*** regarding functions available for the *Signal Generator* and *Multimeter* via the Acquisition Step.

The Record Delay (sec) allows for a wait time before the Multimeter is started.

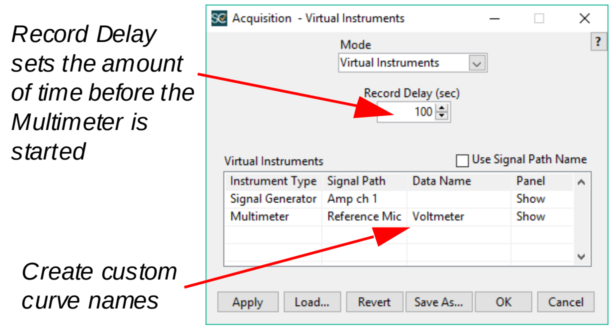


Figure 12-21: Generator and Multimeter

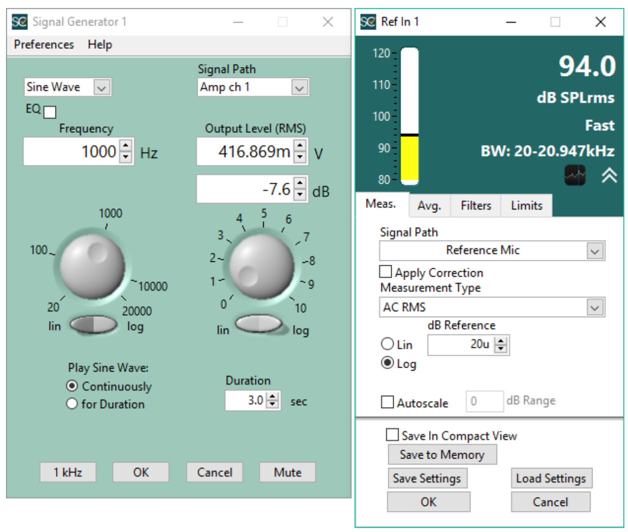


Figure 12-22: Signal Generator & Multimeter

Generator & Oscilloscope or Generator & Spectrum Analyzer

The *Signal Generator* and *Oscilloscope* or *Signal Generator and Spectrum Analyzer* are used to play and analyze a test signal in a sequence. The *Signal Generator* produces a constant frequency sine wave, or plays a WAV file. The *Oscilloscope* displays the time signal on the screen. The *Oscilloscope* will add the Oscilloscope time record to the WFM tab of the *Memory List*. The *Spectrum Analyzer* will add the FFT Spectrum to the **Curves** tab of the *Memory List*. Typically, white noise is used as a test signal when analyzing a DUT with a spectrum analyzer, since it is flat when analyzed with constant bandwidth (FFT) filters. Refer to ***Virtual Instruments on page 455*** regarding functions available for the *Oscilloscope* via the Acquisition Step.

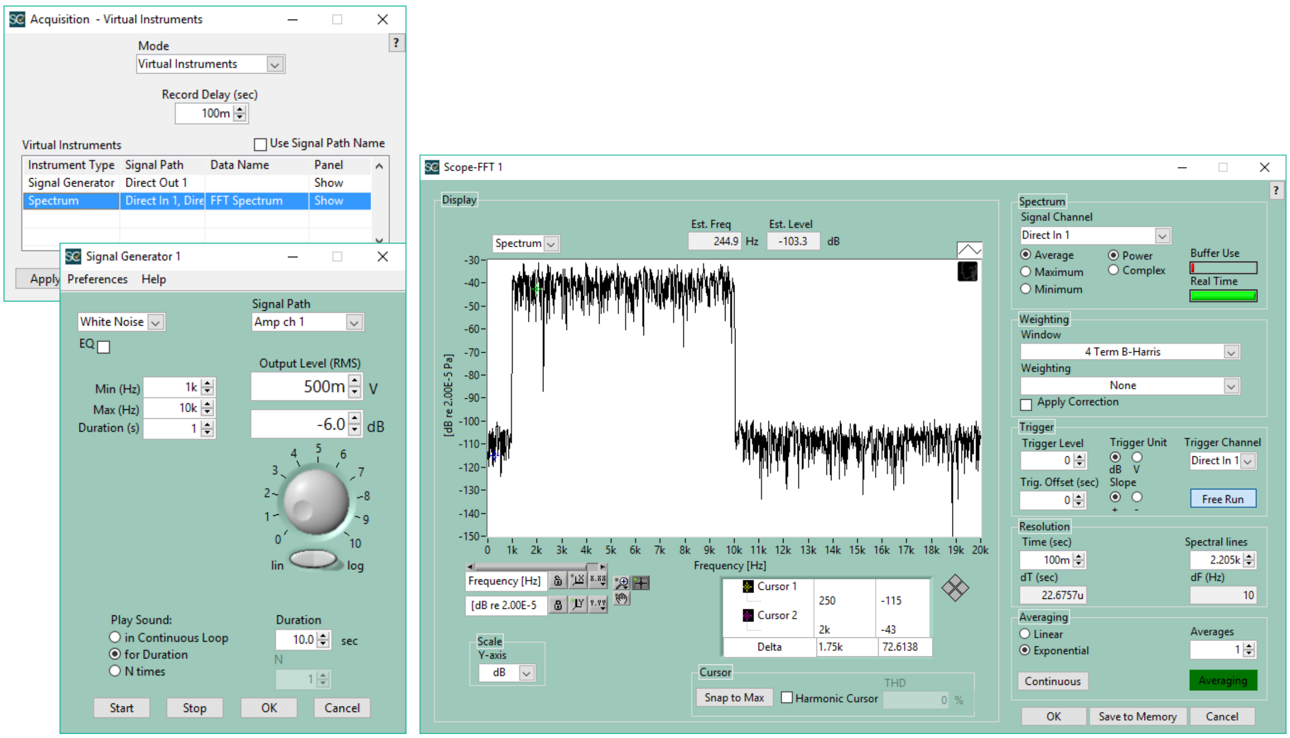


Figure 12-23: Combine Signal Generator and Oscilloscope or Spectrum Analyzer

Generator & Real Time Analyzer

The *Signal Generator and Real Time Analyzer* will play and analyze the test signal. Typically, pink noise is used as a test signal when analyzing a DUT with an RTA, since it is flat when analyzed with constant percentage bandwidth (Nth octave) filters. Check **Hide Panel** if you do not want the *Signal Generator* or *RTA* to open when the sequence is run. As with the *Multimeter*, if exponential averaging is selected, the **Hide Panel** option is not available, since the Virtual Instrument requires user intervention to be stopped. The *RTA* will add the **1/Nth Octave RTA** to the **Curves** tab of the *Memory List*. Refer to ***Virtual Instruments on page 455*** regarding functions available for the *Signal Generator* or *RTA* via the Acquisition Step.

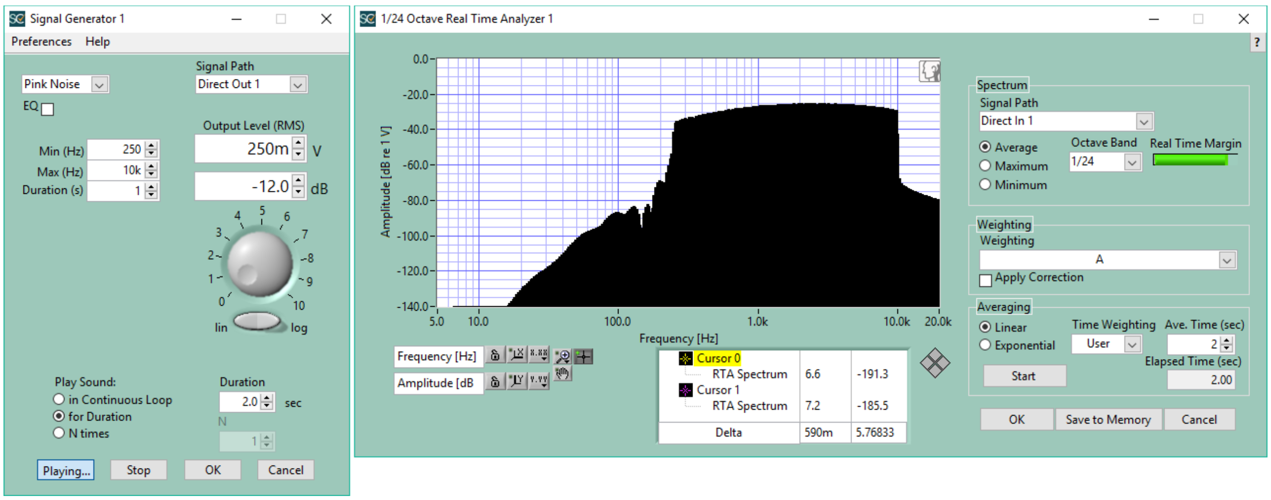


Figure 12-24: Generator & RTA

Read from DC Connect™

When you select **Read from DC Connect** from the **Mode** drop-down list, the Acquisition Step will read a single measured value from the selected DC Connect device over the USB interface.

As of SoundCheck 16.2, the Read from DC Connect Acquisition Step allows you to select which DC Connect to read from. Multiple DC Connects can be selected.

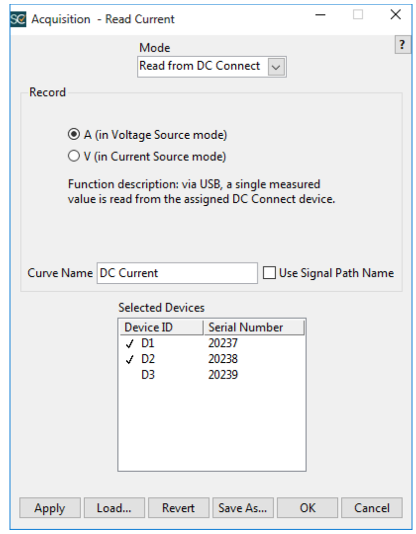


Figure 12-25: DC Connect

Important: ***If a DC Connect is disconnected from the system, the Acquisition Editor field for that device will be grayed out.***

This type of Acquisition Step would likely follow a Stimulus Step or Message Step that sets the DC Connect output voltage or current source level.

Record

If multiple DC Connects are selected, all DC Connect Message Steps with settings for the measurement must occur before the Acquisition Step so each unit is set to the proper Output Mode, Control Mode, etc.

Select which mode DC Connect should operate in:

- **A** (in Voltage Source mode) to measure current
- **V** (in Current Source mode) to measure voltage

If DC Connect is set up as a voltage source, choose units of **Amps DC**, since the current draw of the load is being measured.

If DC Connect is set up as a current source, choose units of **Volts**, since the voltage across the load that results from the applied source current is being measured.

Curve Name

Enter the name for the data that will appear in the Memory List.

Do not check **Use Signal Path Name**. If multiple DC Connects are selected, the Device ID will be appended to the Curve Name.

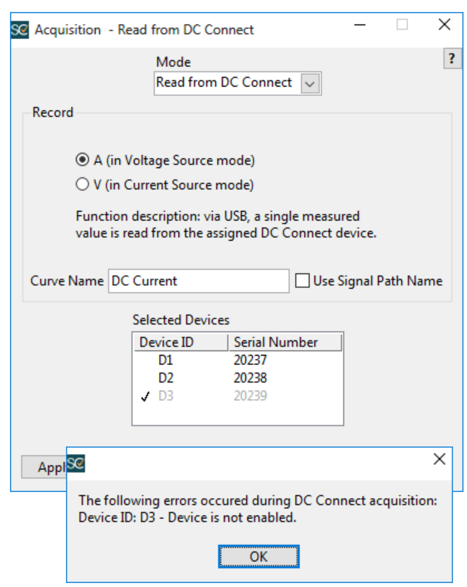


Figure 12-26: DC Connect Disconnected

Selected Devices

Select the appropriate devices in the list. The example in ***Figure 12-25*** shows devices D1 and D2 selected. The resulting Memory List values will be named:

- DC Current (D1)
- DC Current (D2)

At least one device must be selected in the list. If an expected device is not connected, it will be grayed out in the Selected Devices field. If you try to run the sequence with a disconnected device, a warning prompt will show the disconnected device. See ***Figure 12-26***.

Analysis Editor

The *Analysis Editor* (**Ctrl+Shift+N**) enables you to process measured time signals using a variety of analysis algorithms. Choose an algorithm based on the type of measurement needed.

- See ***Algorithm Application Overview on page 161*** for more information on what measurements are possible with each algorithm
- Please note that the HarmonicTrak and Time Selective Response algorithms are optional and only available in advanced versions of SoundCheck®
  - HarmonicTrak - Requires optional module 2001
  - Time Selective Response - Requires optional module 2006

The Analysis setup allows for a variety of measurement types which generate many different types of curves that can be viewed from the *Memory List* and further processed in SoundCheck.

Note: Clicking **Apply** in the Analysis Editor allows you to change the settings in the editor and apply a new analysis without making a new measurement. The result of the new calculations will appear immediately in the ***Memory List***.

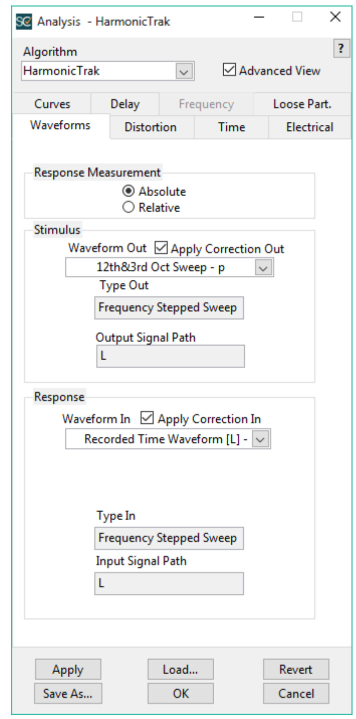


Figure 13-1: Analysis Editor

Latest Features

- Auto Delay+ is a new delay algorithm that is able to detect delays of -0.5 Seconds to any positive delay - ***Auto Delay+ on page 169***.
- New Impedance controls including AmpConnect 621 Impedance - ***Electrical Tab - Impedance on page 204***
- Impedance Box vs. AmpConnect 621, ISC / SC Amp measurement method is selectable - ***Impedance Measurement Method on page 204***
- Optimized THD+N algorithm with Traditional and Synthetic Notch Filters - ***THD + Noise on page 192***
- Simplified Polarity measurement - ***Simplified Polarity Test on page 170***
- Curve Resolution is selectable in any FFT based algorithm - ***Frequency Tab - Curve Resolution on page 188***
- Maximum Valid Harmonic indication - ***Max Valid Harmonic on page 179***

Navigating the Analysis Editor

View Modes

The Analysis editor has two modes to view step information: Simple and Advanced.

Figure 13-2 shows the Distortion Tab low level information that is provided in Simple View compared to the information shown in Advanced.

Simple view is the default step setting.

All tabs will show additional settings that can be accessed by switching to Advanced View as shown in *Figure 13-3*. All field settings on these tabs are active whether the view is Advanced or Simple.

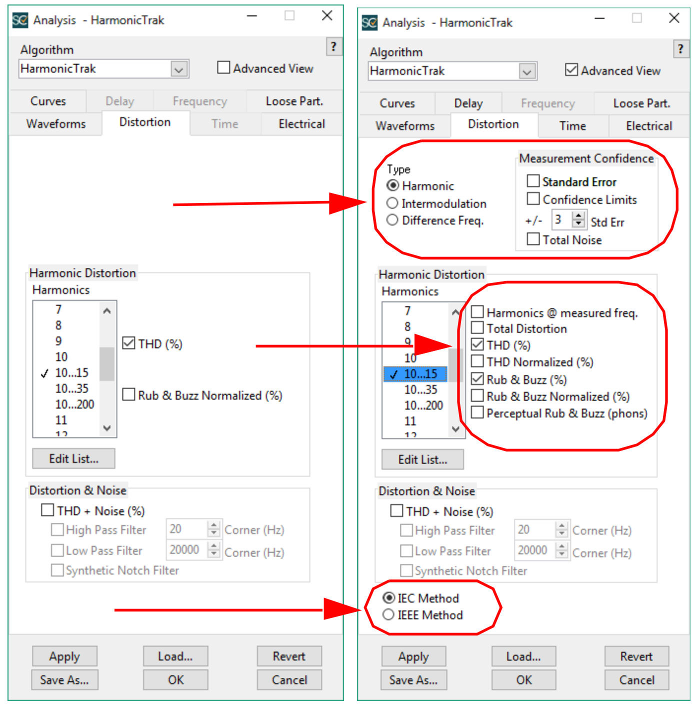


Figure 13-2: Simple vs Advanced View

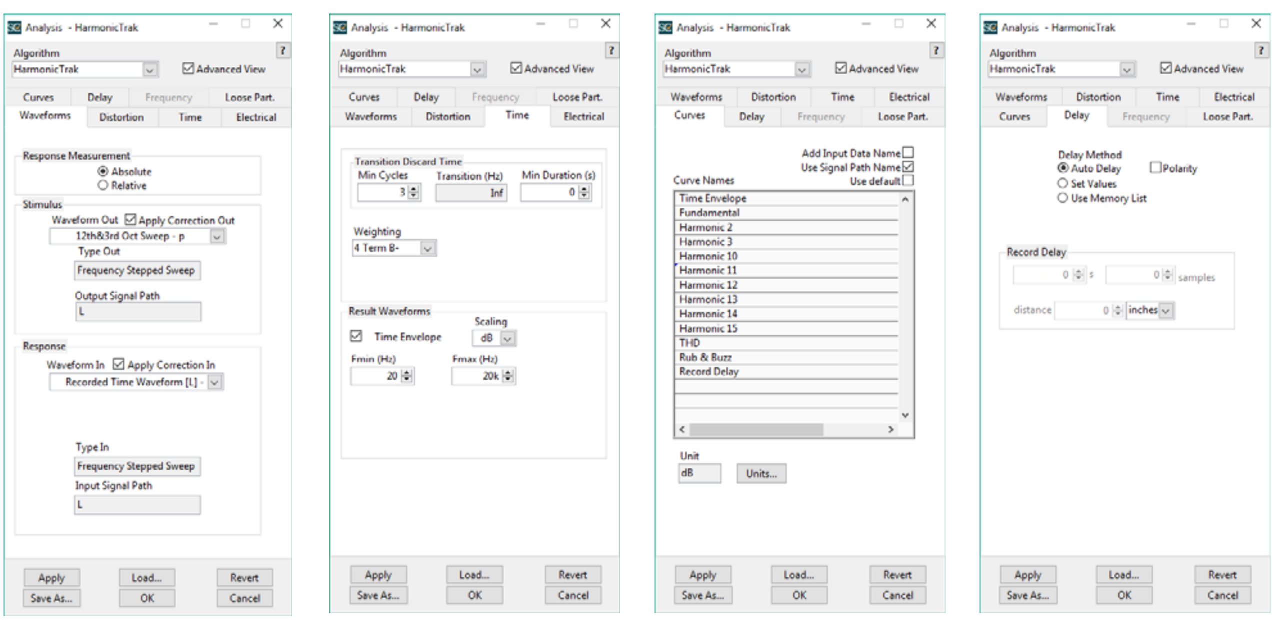
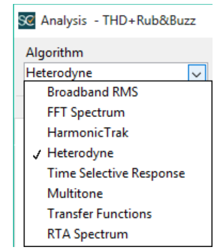


Figure: 13-3 Analysis Editor - Advanced View

Algorithm Application Overview

You can select an algorithm from the drop-down list at the top of the Analysis Editor.



The following table shows the algorithms available in SoundCheck along with suggestions for use of each.

Algorithm	Feature	Typical Stimulus	Desired Measurement	Application
Broadband RMS page 171	No filtering other than bandwidth of sound card	Stweep or None	Unfiltered Frequency Response, DC values	Some Hearing Aid standards, DC Connect measurements
FFT Spectrum page 171	Single Channel	Noise or None	Response spectrum	Most often used for noise measurements - background noise of microphones or electronics
HarmonicTrak™ page 174	Tracks level and phase of any user-selected harmonics	Stweep	Frequency Response, Phase, Harmonic Distortion, Impedance	Near field acoustic or electronics measurements
Heterodyne page 176	Measure frequency and phase response with optimal accuracy	Stweep	Frequency Response, Phase, Impedance	High precision frequency response measurements
Time Selective Response (TSR) page 177	Simulated Free Field - Suppress the effects of reflections in an ordinary room, Fast	Log Sweep	Frequency Response, Phase, Harmonic Distortion, Impedance, Impulse Response	Measurement of acoustic devices in a real room
Multitone page 184	Plays a group of tones simultaneously	Multitone	Frequency Response, Phase, Non-Coherent Distortion, Impedance	Fast frequency response measurements
Transfer Functions - Dual Channel Analysis page 182	Can be used with program material	Noise, speech or music	Frequency Response, Phase, Non-Coherent Distortion, Impedance, Impulse Response	General purpose: near field acoustic or electronics measurements
RTA Spectrum page 186	Nth octave spectrum	Noise, speech or music	Input, Output Nth Octave Spectrum and Nth Octave Frequency Response	Very useful for telephony and devices with DSP, Some Hearing Aid standards

Figure 13-4: Algorithms and Suggested Use

Note: Time Envelope and Loose Particles are available in all algorithms. See ***Appropriate Algorithm vs. Desired Measurement on page 162***.

Appropriate Algorithm vs. Desired Measurement

You can select options from the *Analysis Editor* to add curves and single values in the sequence.

Desired Measurement	Broadband RMS	FFT Spectrum	Harmonic Trak	Heterodyne	Time Selective Response	Transfer Functions	Multitone	RTA Spectrum
Frequency Response			X	X	X	X	X	X
Noise	X	X						X
Harmonic Distortion			X		X			
Non-Coherent Distortion						X	X	
Impedance			X	X	X	X	X	
Loose Particles	X	X	X	X	X	X	X	X
Impulse Response					X	X		
Time Envelope	X	X	X	X	X	X	X	X
DC	X							

Figure 13-5: Appropriate Algorithm vs. Desired Measurement

For example, checking the **Loose Particles** box will add a **Loose Particle Count** single value to the *Memory List*. In the *Analysis Editor*, many of the options become available only when an appropriate algorithm is selected. ***Figure 13-6*** shows a chart displaying available measurements when each algorithm is active.

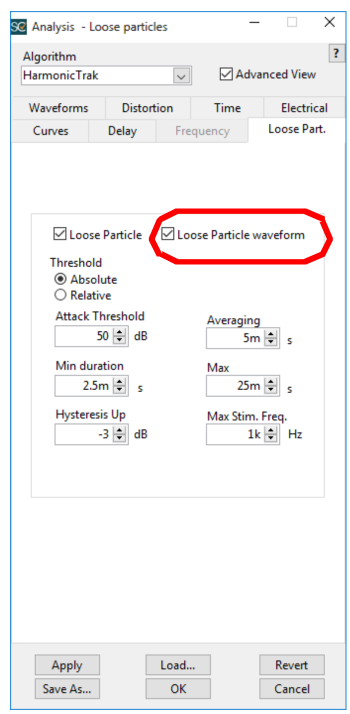


Figure 13-6: Loose Particle Waveform

Analysis Editor Tabs

This section covers four of the Analysis Tabs that are setup in the same way for all algorithms: Waveforms, Curves, Delay and Time. Distortion and Frequency are covered under each algorithm. Electrical and Loose Particles are covered at the end of the chapter.

Time Tab

All algorithms allow for a Result Curve of **Time Envelope**.

Enter the Fmin and Fmax frequencies in Hz.

Details of the Time Tab for each algorithm are covered in the description of each algorithm.

Waveforms Tab

- Eight Algorithms for analyzing time domain signals
- Response Measurement
  - Relative response is the response level divided by the stimulus level
  - Absolute Response is the response level only
- Stimulus and Response waveform selection
  - All waveforms in the Memory List appear in the Waveform drop-down lists
  - Select the desired Stimulus from the **Waveform Out** drop-down list
  - Select the Response waveform that is related to the selected Stimulus from the **Waveform In** menu
  - Analysis can be applied to any waveform in the Memory List by selecting it in the **Response - Waveform In** section
  - Select **Apply Correction In or Out** as required for the type of measurement. See ***Apply Correction on page 164***.

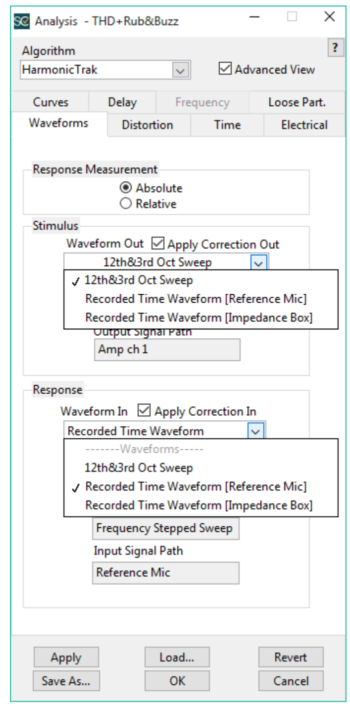


Figure 13-7: Analysis Setup

Clicking ***Apply*** allows you to change the settings and apply a new analysis without making a new measurement. The result of the calculation is immediately updated in the **Memory List.**

Relative or Absolute

Relative Response is the response level divided by the stimulus level. The response is stimulus level independent or "normalized" to the input level. When increasing the input level, the relative response amplitude level will not change if the system is linear because the output level will increase by the same amount as the stimulus level.

- Sensitivity measurements use Relative.
- This is also useful when looking at compression.

Absolute Response is the response level only and is stimulus level dependent. i.e.: If the stimulus increases from 1 Volts to 2 Volts, there will be a 6 dB output level increase.

- Max SPL measurements use Absolute.

Apply Correction

Checking the **Apply Correction** boxes will apply the input or output correction curves to the input or output signal, as part of the analysis process. This is particularly useful for removing the response curve of an amplifier from a loudspeaker test as well as removing the response of the measurement microphone. This is not the same thing as an Equalization Curve. (See ***Equalization and Correction Curve on page 93***.) It occurs in Analysis, after the signal has been acquired.

Apply Correction Out

Generally, this is always checked unless you have a very specific exception. Here are the two most common scenarios:

Microphone Test - When testing microphones you have likely equalized a loudspeaker. In this case the output correction is the remaining few tenths of a dB that the equalization was unable to flatten. In this case the EQ will be accounting for 99% of the loudspeaker response, and the correction out accounts for that last 1%.

Loudspeaker Test - When testing a loudspeaker using an external amplifier, the amplifier needs to be calibrated. During Amplifier Calibration SoundCheck creates an output correction curve that compensates for the magnitude and phase response of the amplifier. The magnitude is typically very flat, except at the extremes. The phase curve is not flat. Selecting **Apply Correction Out** compensates for the phase response as well as the frequency response.

This correction is applied in Analysis after the acquisition has occurred.

Please refer to ***SoundCheck Signal Flow on page 82*** which covers the order of operations in the SoundCheck signal chain.

Apply Correction In

This is usually checked if you are using a reference microphone in your test and you have imported the correction curve for that specific microphone (usually supplied by the manufacturer).

Another less common usage is to import a correction curve for an ear simulator, like a free-field or diffuse-field correction curve for a head and torso manikin.

Waveform Batch Processing

Waveform batch processing is a powerful analysis tool that enables a group of waveforms to be analyzed with just one step in a sequence, rather than having to program multiple analysis steps. This significantly simplifies sequences with multichannel acquisition. The feature is also available in offline mode where multiple waveforms can be grouped together in a custom group and the batch processing operand applied to all simultaneously. Such offline analysis may be useful for detailed analysis of production line data. To help distinguish the output curves and values, the name of the response waveform can be appended to the resulting data by selecting "**Add Input Data Name**" on the Curves Tab.

The following procedure should be used for Waveform Batch Processing:
(See ***Figure 13-8***)

1. In the Memory List of the sequence create a **Custom Group** and add the Recorded Time Waveforms

   This example shows a group named "Multichannel". The number of waveforms in the group is indicated by the value in parenthesis (4), which is added automatically.

2. In the **Analysis Step** > Waveform Tab > under Waveform In > select - **custom group: Multichannel (4)**, from the drop-down list

3. Switch to the Curves Tab, select "**Add Input Data Name**" to append this to the **Output Curve Name**

   This creates unique names for each of the output curves. As an alternative, "**Use Signal Path Name**" can be used for curve naming.

4. After the Analysis step runs the output curves are populated in the Memory List

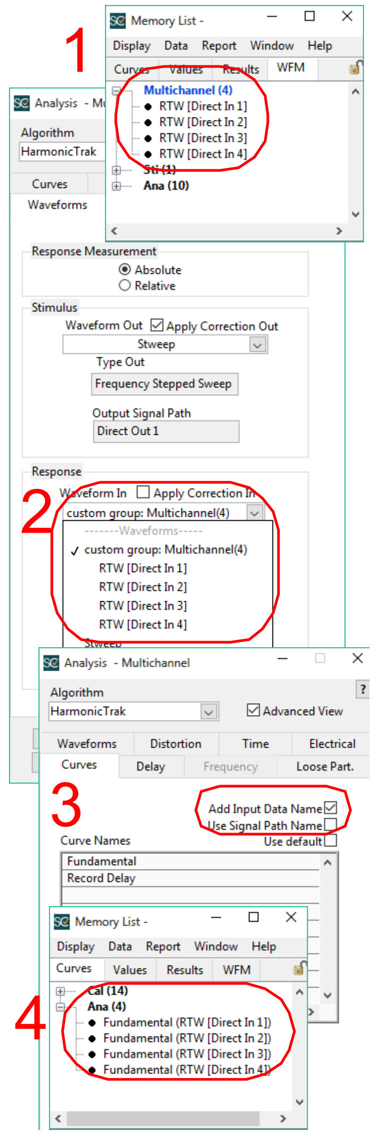


Figure 13-8: Batch Process Procedure

Curves Tab

Curve Names

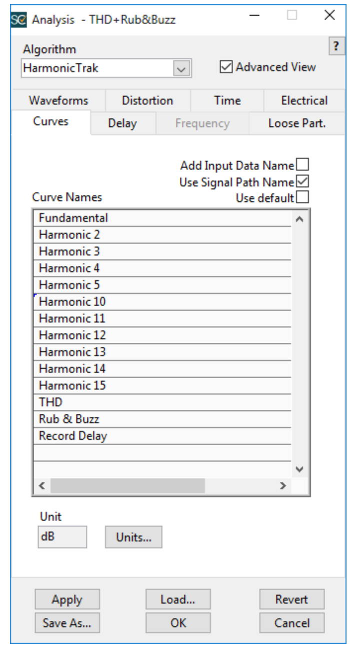


Figure 13-9: Use Default Curve Names

Memory List curve names created in the Analysis Step can be modified in the **Curve Names** tab. This may be especially helpful when using multiple Analysis Steps to measure multiple Stimuli, in a single sequence or in subsequences. If two Analysis Steps in the same sequence have the same curve name, the 2nd Analysis Step will append a 2 in front of the curve name, e.g.; 2-Fundamental, 3-Fundamental, and so on.

Highlight a default name and enter a custom name for curves and single values.

- **Add Input Data Name -** Select to append the Input Waveform name to the curve name, e.g.; Fundamental + (Recorded Time Waveform)
- **Use Signal Path Name -** Select to append the Signal Path name to the curve name, e.g.; Fundamental + (Dut Mic)
- **Use Default -** The **Custom Curve Names** field will be grayed out and all changes to curve names will revert back to their original state. Curve names cannot be edited while this is selected.

Note: Changing curve names in an existing sequence may affect your displays.

For example:

- **Fundamental (DUT Mic)** was originally selected to display on the XY Graph
- The curve name is then changed to **Model ABC Frequency Response (DUT Mic)** and the display will cease to display **Fundamental (DUT Mic)**
- You then need to open your Display Step and Memory list. Then drag **Model ABC Frequency Response (DUT Mic)** to the appropriate XY Graph.

Units

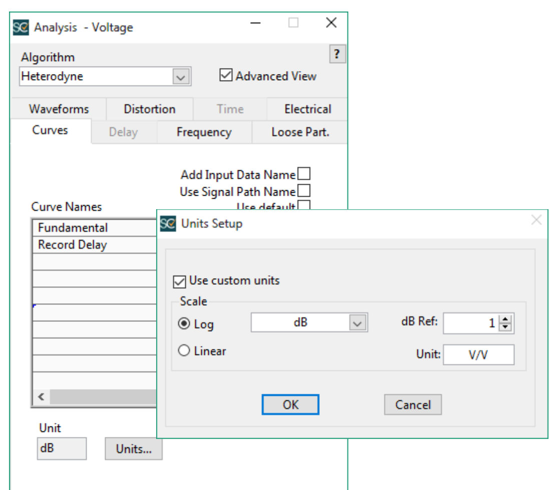


Figure 13-10: Set Units

Abs/relative - the measurement can be relative or absolute (Waveform Tab). ***See "Relative or Absolute" on page 163.***

Default Unit

- Absolute: The frequency curves are in the physical unit of the response waveform (e.g Pa). The dB ref comes from the input signal path used in Acquisition.
- Relative: The frequency curves are in relative unit (e.g., Pa/V). The unit is the ratio of the response unit over stimulus unit, with a dB ref of 1. All algorithms are subjected to Rel/Abs except Spectrum, Time Envelope & Loose Particles, which are always in abs unit.

Custom Unit

- Select "**Use custom units**" to enter the Scale details:
  Log or Linear, dB drop down, dB Ref and Unit. Be aware, the math applied for absolute/relative stays the same. See Units chapter: ***Analysis Editor on page 110*** for more details.

Curve Naming Best Practices

Selecting **Add Input Data Name** and **Use Signal Path Name** can create long file names that are difficult to read in the Memory List. We recommend that shorter names should be used in the Calibration Editor and the Acquisition Steps if you plan to append the Input Data Name or Signal Path Name.

For example:

Calibration Signal Path = **Reference Mic SCM 3 SN 1234**.

Selecting **Use Signal Path Name** as in ***Figure 13-9*** results in a Memory List name of:

Fundamental [Reference Mic SCM 3 SN 1234].

An alternative would be to name the Signal Path

"***RefMic-SN1234*** " as shown in ***Figure 13-11***, which yields a Memory List name of:

Fundamental [RefMic-SN1234]

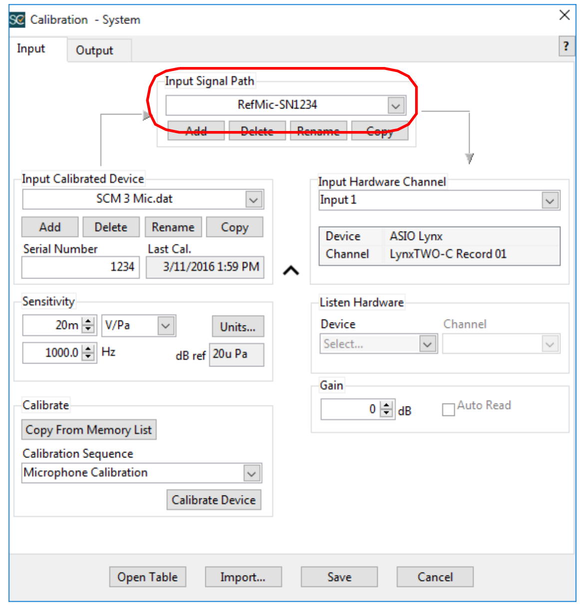


Figure 13-11: Input Signal Path Name

Long names in Acquisition Steps can cause similar problems when **Use Signal Path Name** is selected in the step. This can result in a Memory List data name of:

Fundamental (Recorded Time Wave Form [Reference Mic SCM 3 SN 1234])

1. Further post processing of the data can result in even longer names, so shortening names in Analysis can be beneficial.
2. It can be helpful to abbreviate the Waveform Name in Acquisition to **RTW**, especially when **Use Signal Path Name** is selected in Acquisition.
3. Additionally, shortening the Curve Names in the **Analysis Curves Tab** can help make data names shorter and easier to manage.

Waveform Batch Processing

You must use **Add Input Data Name** or **Use Signal Path Name** when using **Waveform Batch Processing** in order to get a unique name for each file analyzed. *See **Waveform Batch Processing on page 165**.*

Custom Curve Name

This is helpful when using multiple Analysis Steps in a single sequence or in subsequences.

If two Analysis Steps have the same curve names, the result curves from the 2nd Analysis Step will have "2-" prepended to the curve name, e.g.; 2-Fundamental, 2-THD, etc. Giving them different names in the two Analysis Steps avoids this problem.

Delay Tab

To perform Analysis algorithms accurately, it is important that Stimulus waveforms and Response waveforms are in time alignment with each other. Soundcheck can automatically calculate the delay using either Auto Delay or Auto Delay+ modes. The user can also specify the delay by selecting **Set Values** or **Use Memory List** from the **Delay Method** drop down menu.

For signals paths with inconsistent delays, such as Bluetooth interfaces, Auto Delay or Auto Delay+ must be used.

Auto Delay can be used to measure time of flight or distance between the microphone and the device under test, however this requires that signal path latencies are stable and are accounted for in the Hardware Table.

Auto Delay

In Auto Delay mode, SoundCheck automatically calculates and compensates for the delay between the stimulus and response waveform. This delay value is output to the memory list as '**Record Delay**'.

In this method the peak of an averaged impulse response, calculated using the entire stimulus and response waveforms, is used to determine the delay amount.

For some extreme scenarios, Auto Delay may give inaccurate results, these scenarios include:

- **High background noise**: noisy production environment
- **Strong reflection**: side wall reflections are stronger than the direct sound wave
- **Sampling rate errors**: sample rate variation in Bluetooth devices

This method can only compensate for delays between -1s and +1s. For larger delays, we recommend using Auto Delay+ or windowing the response waveform.

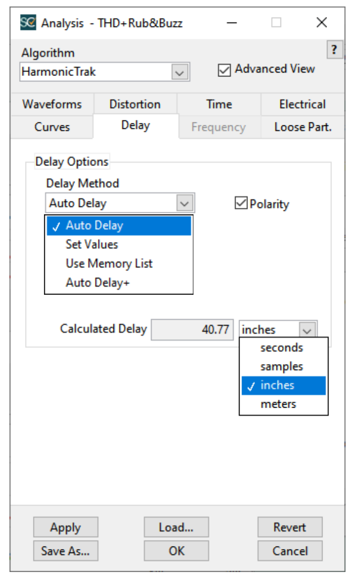


Figure 13-12: Auto Delay

Polarity

This will calculate the simple Polarity value and add it to the Memory List. See ***Simplified Polarity Test on page 170***.

- Positive polarity is indicated by 1 and Negative (inverted) polarity is indicated by -1

Set Values

This allows you to specify a fixed delay offset for the response waveforms. The delay offset can be specified either as time (in seconds), samples, or distance (in feet or meters).

Note: Calculating 'Polarity' is not available in this mode.

Memory List Selection

This allows you to choose a Memory List Value to use as the source for your delay offset. In addition to the Memory List Value, you will need to specify the axis to use, and the units applied to the value.

Note: Calculating 'Polarity' is not available in this mode.

Auto Delay+

Auto Delay+ is a new delay algorithm that is able to detect delays of -0.5 seconds to any positive delay. Listen recommends using Auto Delay+ for Open Loop testing where delays can be greater than 1 second.

In this algorithm, a reference frame is selected from the Analysis section of the Stimulus waveform and is compared against the Response waveform using a sliding window. For stimuli with non-stationary tones (i.e., STweep, Two Tone, Noise, and WAV-file), a Generalized Cross-Correlation with Phase Transform (GCCPHAT) is used for comparison. For stimuli with stationary tones (i.e., Logarithmic Amplitude Sweep and Multitone), an impulse response is used.

In '**Advanced View**' mode, you can view/set some of the underlying parameters used by the algorithm. By default, Parameter Selection option is set to Automatic; here SoundCheck attempts to identify the best 1s **Stimulus Waveform Reference Window**, which is chosen from sections of the Stimulus set to Analyze (see ***Analyze Yes/No Option on page 123***). SoundCheck also attempts to select the best **Response Waveform Search Range** to perform the algorithm on.

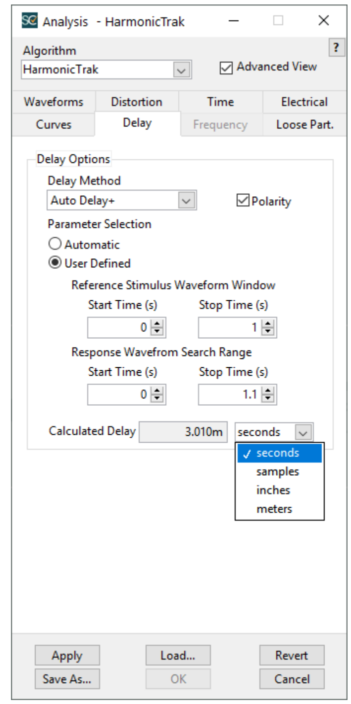


Figure 13-13: Auto Delay+

Note: The algorithm assumes that the response waveform contains the full length of the stimulus waveform within it.

Setting Parameter Selection to **User Defined** allows you to change the algorithm parameters, this can be helpful in scenarios where Auto Delay+ may have failed to give accurate results. Some recommendations for settings are:

- Select a **Stimulus Waveform Reference Window** that has large frequency variations over time and is in the passband of the device under test.
- For single tones, multitone, or log sweep it is recommended to include either the starting or ending edge of the stimulus.
- Longer durations may improve the accuracy of the algorithm, however, this comes at the cost of increased processing time.
- For the **Response Waveform Search Range**, select a reasonable section of the response waveform where you would expect to find the Reference Stimulus Window selected.
- Low frequency Log Stweeps, such as those used for testing Subwoofers may not give accurate results with Auto Delay+ in Automatic mode. In this case, select **User Defined** mode and set the Stimulus Reference Window to the entire length of the Stimulus.

The calculated delay value is output to the Memory List as 'Record Delay'.

Calculating 'Polarity' is available in this mode. See ***Polarity on page 168*** for more information.

Simplified Polarity Test

A polarity test is often used to verify that a device is wired correctly. The quick polarity test is performed in an analysis step, and uses the impulse response from the **Auto Delay** function *(See* ***Figure 13-14****)*. It analyzes the peak of this impulse response and measures if it is negative or positive to determine overall polarity. It is a simple and easy alternative to phase domain testing for simple devices or single drivers where the phase does not change more than 180 degrees. Polarity measurement using phase response is still available as an alternative method for more complex devices.

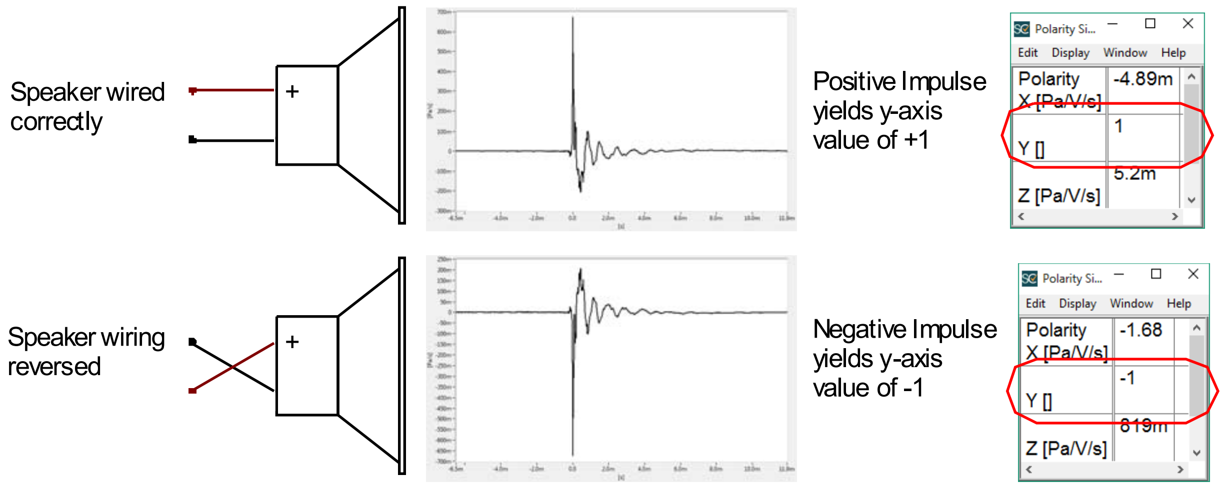


Figure 13-14: Speaker Polarity Impulse

- Auto Delay must be selected on the Delay tab in order to use this feature
- The polarity function is available in any of the analysis algorithms
- Polarity will appear as a value in the Memory List
- A value of +1 shows that the device under test has a positive polarity
- A value of -1 shows the device under test has a negative polarity
- This value can then be used in a limit step with a lower limit set to zero to show when a device has been wired correctly or has correct polarity

Note: The limit step allows you to determine if a value of +1 is a pass or fail, depending on the overall phase of the measurement chain. (e.g., the microphone output is inverted)

Note: Polarity measurement using phase response is still available as an alternative method for more complex devices.

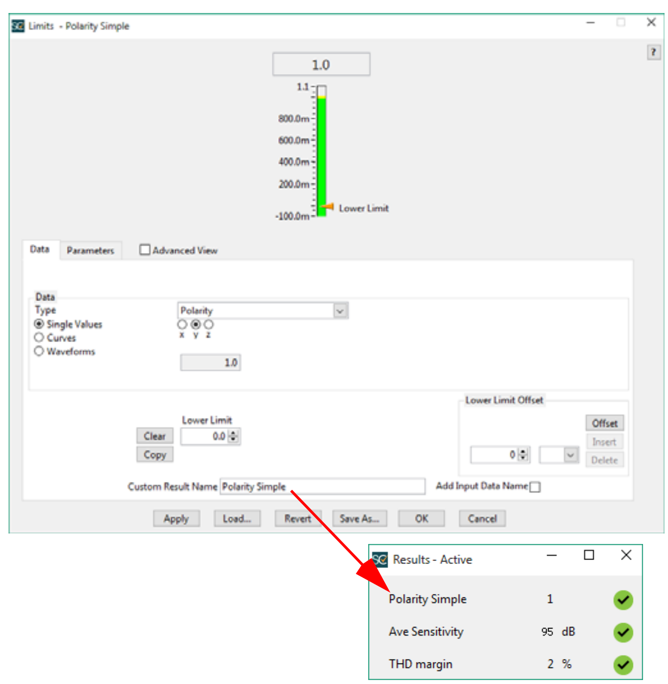


Figure 13-15: Limit Step and Results

Broadband RMS

This algorithm does not utilize any filtering. This measures the total RMS energy at each excitation frequency. The upper frequency limit used to determine the total RMS energy is based on the audio interface's sampling rate.

For example, at a sampling rate of 44.1 kHz, the upper frequency limit is approximately 22 kHz. Please note that this value is also influenced by the audio interface's anti-aliasing filter. Typically the highest usable frequency is 45% of the maximum sampling rate.

Uses include measuring buzzers and other harmonically-rich devices. You must use this algorithm to measure OSPL on hearing aids according to **ANSI S3.22** and other similar standards.

Note: When Broadband RMS is selected, **Apply Correction In** (on the Waveform tab) are disabled. No correction is applied if either is selected.

FFT Spectrum

The spectrum algorithm calculates an averaged FFT of the response waveform. This is performed according to the Frequency Resolution Weighting and Overlap settings in the editor. This algorithm can be used to measure the background noise prior to applying a stimulus to the Device Under Test (DUT).

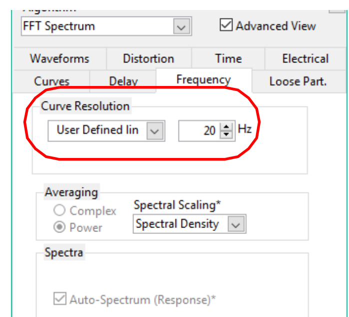


Figure 13-16: Default Resolution

Resolution

The default resolution of 20 Hz requires a measurement length of at least 0.1 Second. A resolution of 1 Hz requires a measurement length of at least 1 Second. The resolution is inversely proportional to the time required.

- When acquiring the signal with a Play/Record Acquisition step the following should be selected in the Waveform Tab of the Analysis Step:
  - Waveform In = Recorded Time Waveform of the Acquisition
  - Waveform Out = Stimulus Waveform of the Acquisition
- The resolution is determined by the length of the stimulus waveform (See ***Figure 13-17***)

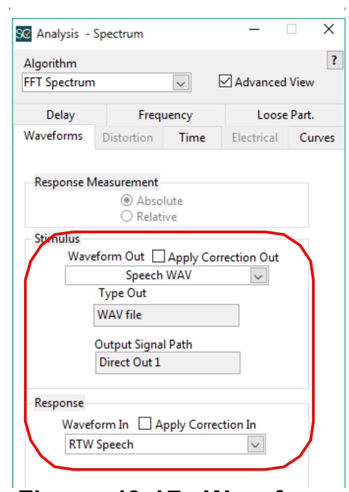


Figure 13-17: Waveform Selection Play/Record

FFT Spectrum - Signal Acquisition

- When acquiring the signal with a Record Only Acquisition step (See *Figure 13-18*)
- The Waveform In and Out should be the Recorded Time Waveform of the Acquisition

---

Note: Warning! A resolution smaller than 1 Hz can require a great amount of system memory. Out of memory errors may result.

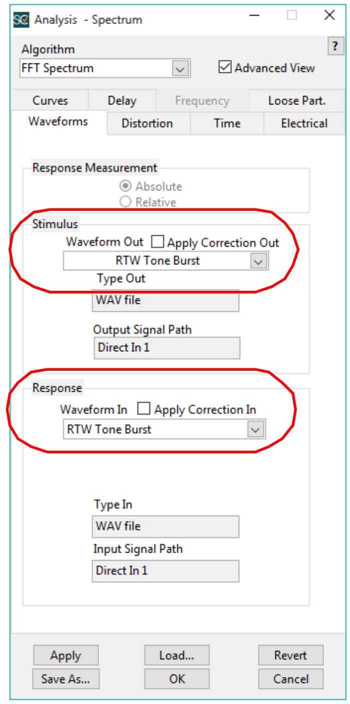


Figure 13-18: Waveform Selection Record Only

FFT Spectrum Settings

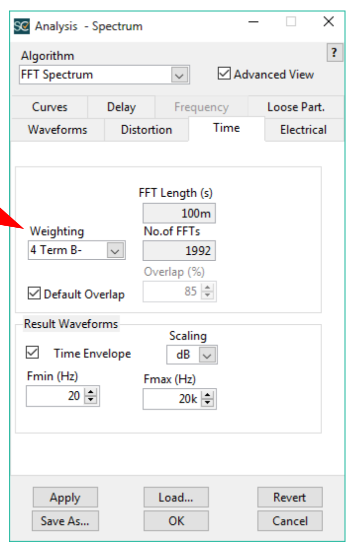
Weighting - Time Tab

The Weighting function selected will also set default values for the Overlap Percentage.

Figure 13-19 shows the selections for Spectrum Analysis. It also includes a table of Weighting and the Default Overlap Percentage.

Select proper weighting window type

Refer to ***Table of Weighting Window Applications on page 174*** *for more information*



Weighting Type	Default Overlap %
None	0
Hanning	75
4 Term Blackman-Harris	85
7 Term Blackman-Harris	90
Flat Top	90

Figure: 13-19 Weighting and Default Overlap %

Spectral Scaling - Frequency Tab

- RMS - The result is the RMS level according to the FFT resolution. This is best used for Pure Tone/Sine measurements.
- Spectral Density - The result is independent of the frequency resolution of the FFT. This is used for noise based measurements, e.g., Self Noise, noise stimulus, speech or music.

  The unit of the result will be:

$$\frac{\text{InputUnits}}{\sqrt{\text{Hz}}}$$

Note: Custom Units can be used to simplify the Spectral Density unit, e.g., “dB V“ instead of “dB V/sqrt(Hz)”.

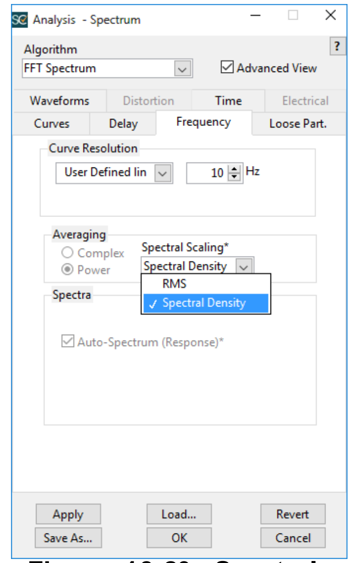


Figure 13-20: Spectral Scaling

HarmonicTrak™

This module allows multiple harmonics to be measured in one stepped-sine sweep (Stweep™). You would also choose this algorithm in order to measure the DUT's THD and/or Rub & Buzz distortion characteristics. To choose a specific range of Harmonics, click **Edit List** in the Editor.

See ***THD w H2, H3 on page 188***.

This measures the levels of any harmonic or sub-harmonic when using sinusoidal excitation. This algorithm also includes Two Tone distortion and normalized distortion.

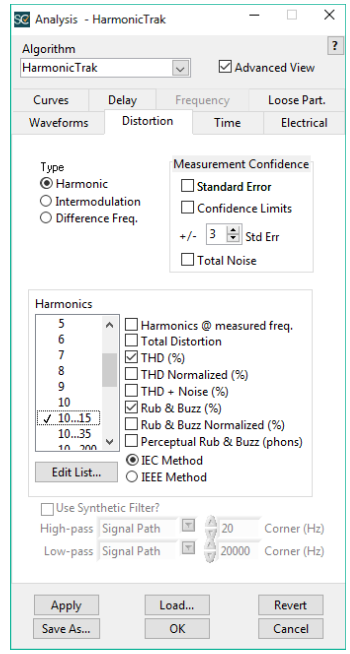


Figure 13-21: HarmonicTrak

Weighting - Time Tab

HarmonicTrak uses a step FFT analysis: at each sine step of the response, a weighting window is applied and an FFT is performed.

The available Weighting Window types are: Hanning, 4 Term Blackman-Harris and 7 Term Blackman-Harris.

The performances of the windows are shown in ***Figure 13-22: Table of Weighting Window Applications***.

Weighting Window Type	Min Cycles per Step	Min H2	Lowest Measurable Distortion	Comment
Hanning	5	-40 dB	1.0%	General purpose, high-speed but sidelobe attenuation is low compared to other windows
Blackman-Harris 4 Term	10	-90 dB	0.003%	More precise but require more time. Good for most electroacoustic measurements.
Blackman-Harris 7 Term	15	-120 dB	0.0001%	Most precise due to greatest attenuation of side lobes. Best window to use when measuring very low distortion devices, such as electronic circuits and products.

Figure 13-22: Table of Weighting Window Applications

Transition Discard Time - Time Tab

This applies to Broadband RMS, HarmonicTrak and Heterodyne algorithms.

Most devices exhibit some transient phenomena between each step of a stepped sweep. Transition Discard Time allows you to exclude these transients for the measurement and return the true steady state response of the device under test. The analysis algorithm will ignore the beginning of the step defined as a number of cycles of the fundamental stimulus or a fixed amount of time, which ever is longer in time.

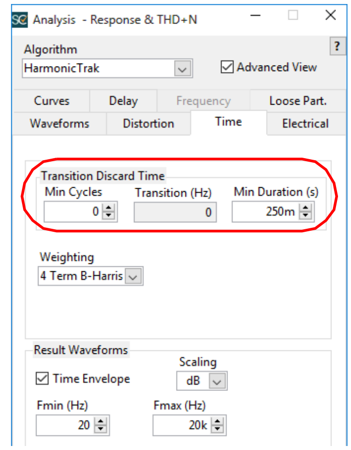


Figure 13-23: Transition Discard Time

Minimum Cycles

This is the number of cycles that are discarded at the beginning of each frequency step, up to the Transition Frequency.

Transition (Hz)

Below this frequency the analysis algorithm will discard the Minimum Cycles. Above this frequency the Minimum Duration will be discarded.

Minimum Duration

This is the minimum amount of time that will be discarded from the beginning of each step.

Note: The Stimulus Step settings for "**Min Cycles**" and "**Min Duration**" (per step) must be greater than the Transition Discard Time settings in Analysis. If either of the discard settings are equal to or greater than the corresponding values in the Stimulus Step, there will be nothing left over for SoundCheck to analyze. No data will be passed to the Memory List.

THD+N Minimum Duration Settings

For accurate THD+N you need to have sufficient dwell time in both Stimulus and Analysis Steps. It is recommended to use the following as a minimum:

- THD+N Analysis Step > Time Tab > **Min Duration** of at least 250 mSec
- Stimulus Step > Step Size > **Min Duration** of at least 1 Sec
- See the **Amplifier THD+N** example sequence in SoundCheck

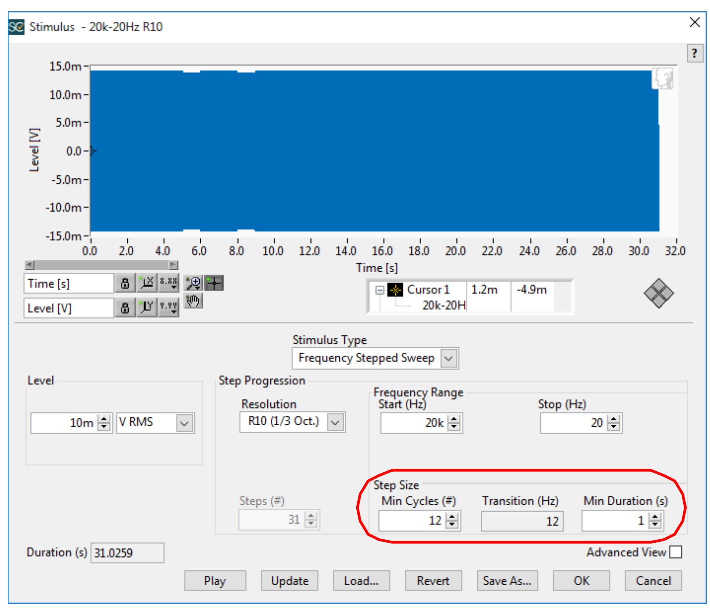


Figure 13-24: Min Duration

Warning Message

If the settings noted above are out of balance the following warning will appear. Adjust one or both of the Min Duration fields noted above.

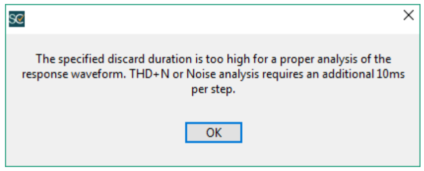


Figure 13-25: Min Duration Warning

Heterodyne

This is the same algorithm used in the Brüel & Kjær Type 2010 and 2012 analyzers. This algorithm is very accurate but, unlike HarmonicTrak, it cannot measure the harmonics and fundamental simultaneously. It serves as an alternative to HarmonicTrak, when distortion measurements are not required or when the distortion module is not available.

Heterodyne measures just the fundamental response using sinusoidal excitation with excellent background noise rejection and fast calculation speed.

- Detection in synchronous
- A Quadrature Detector is employed, i.e., mixing or multiplication is by both sine and cosine, in order to obtain the phase for the complete complex steady state response.

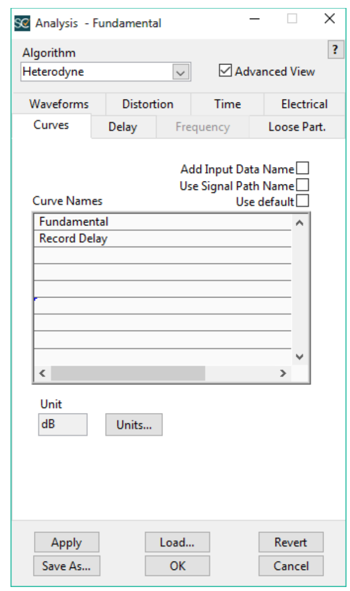


Figure 13-26: Heterodyne

Weighting - Time Tab

The available Weighting Window types are: Hanning, 4 Term Blackman-Harris and 7 Term Blackman-Harris.

The performances of the windows are shown in ***Table of Weighting Window Applications on page 174***.

Time Selective Response (TSR)

Requires optional module **2006 - Time Selective Response**

This module enables time selective measurements with a logarithmically swept sine wave, which can be used for simulated free-field measurements in non-anechoic environments. By removing the stimulus from the response waveform (deconvolution operation) the global impulse response is calculated directly, and from that, the frequency response is determined. The Impulse response of the fundamental can be displayed using linear or log amplitude.

In just one sweep, the free-field response of the fundamental and harmonics can be measured and analyzed.

See ***Time Selective Measurements With Log Sweep on page 599*** for technical details.

The result of the deconvolution process can be added to the Memory List by selecting "Deconvolved Response" in the Time Tab.

See ***Deconvolved Response on page 180*** for more information.

Note: The time window used by the Time Selective Response algorithm has a 10% taper at each end. The Fundamental Impulse Response must be inside these tapers. See ***Figure 13-31: TSR Window - Cosine Taper on Pg 180*** for more information.

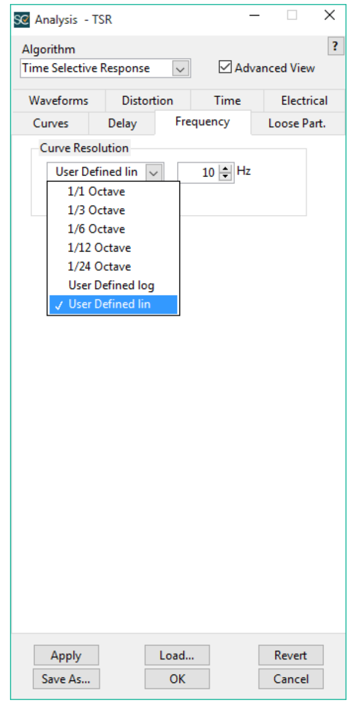


Figure 13-27: Curve Resolution

TSR requires a LogTSR Stimulus

This algorithm is only available when using a **Frequency Log Sweep** in the Stimulus Step of the sequence.

- Please note that the Frequency Log Sweep must sweep from low to high frequency.

Curve Resolution

This is the Base Resolution for the analysis. e.g.: 10 Hz

The Resolution can be set to a number of predetermined ranges or User Defined Log or User Defined Lin. This allows you to manually set the resolution of the result curve as shown in ***Figure 13-27***.

Time Selective Response - Time Tab

Start and Stop Time Selection

You can enter values in the Start and Stop Time fields from the Time Tab. This determines the Windowed Resolution.

- See ***Example:*** below
- See ***TSR Window on page 180*** for instructions

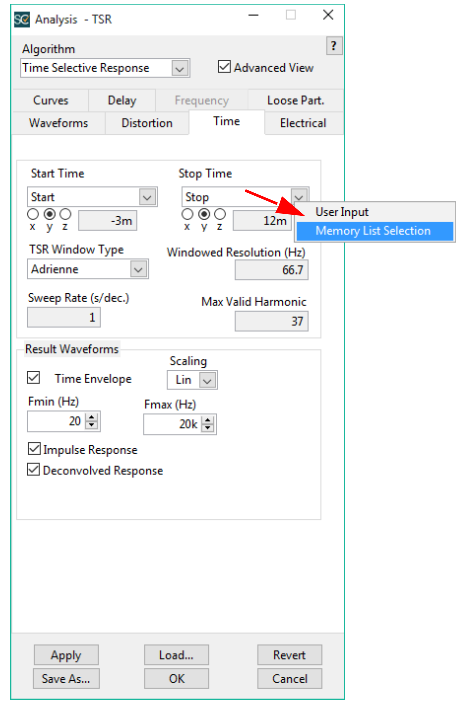


Figure 13-28: Memory List Start/Stop Time

Memory List Selection

This allows the window to be set dynamically during the sequence using recalled or calculated values rather than fixed.

You can use two message steps prior to the TSR Analysis Step to set the desired Start and Stop times. The operator can be prompted to enter the Start and Stop times during the run of the sequence.

Right-click on the Start and Stop Time fields and click on **Memory List Selection** as shown in ***Figure 13-28***.

TSR Window Type

Adrienne is the default. Other available types are: None, Cosine Tapered, Half-Cosine Tapered, Exponential, Half-Hanning and Half-BH4.

Windowed Resolution (Hz)

This can be considered "Smoothing" rather than interpolation. e.g.: 66.7 Hz

Example:

Figure 13-27 and ***Figure 13-28*** show the Curve Resolution set to 10 Hz and the Windowed Resolution at 66.7 Hz. The Windowed Resolution always takes priority over Curve Resolution.

BT=1 (Bandwidth x Time equals unity)

This means that the “truly realizable frequency resolution” is determined by the reciprocal of the measured signal duration (F=1/T). So even if you measure for many seconds, what matters is the time from the beginning of your impulse response to the end of the time window. Think of the actual frequency response Curve Resolution as oversampling and the time Windowed Resolution as smoothing.

Good old fashion chart recorders had something similar relating to the pen and paper speed. If the paper speed was too fast, the effect was like curve smoothing. If the pen moved too fast, the effect was like oversampling.

Max Valid Harmonic

The maximum valid harmonic is the maximum harmonic order that can be time separated from its immediate harmonic neighbors. This is a function of the Start Time, Stop Time and Sweep Rate.

(This is the Maximum Harmonic that can be selected while still having a valid result.)

While Log TSR sweeps are a popular test method, users unfamiliar with this method may inadvertently miss important measurement details by sweeping too fast.

For example, a given combination of window size and speed may be adequate for measuring Rub & Buzz, but insufficient to enable analysis of individual harmonics. The new indicator shows the maximum harmonic that can be selected independently of its neighbors. The indicator, while it does not place any restrictions on your ability to define the speed and window size, will offer an advisory when the settings are such that individual harmonics will not be accurately calculated.

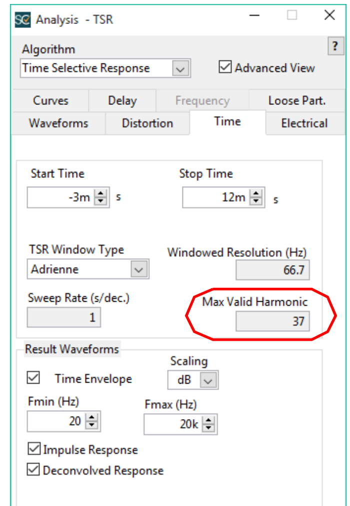


Figure 13-29: Max Valid Harmonic

Time Envelope

For Time Selective Response the **Time Envelope** is used to view the magnitude of the response time signal. This is useful in analyzing the effects of compression in an electrical circuit and/or an electroacoustic transducer. The Scaling can be set to dB or Linear.

The magnitude is calculated only between Fmin and Fmax. This helps to create a cleaner, smoother or "less noisy" envelope.

Note: To avoid ripple effects in the time domain, Fmin and Fmax must be outside the stimulus bandwidth.

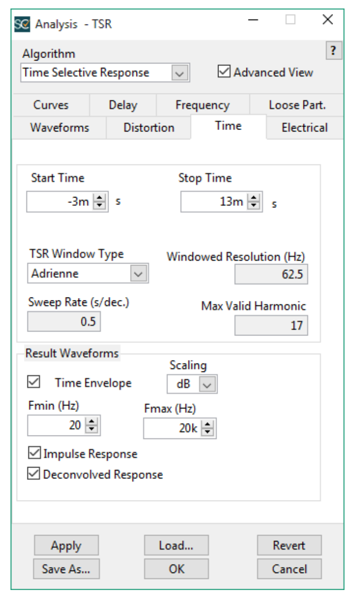


Figure 13-30: Define Envelope & Impulse Response Units

Impulse Response

The Impulse Response is the time domain response of a system to an idealized infinitely short impulse. An impulse response is the time domain equivalent of a frequency response function, and can be computed using the Inverse Fourier Transform on a frequency response function. This can only be accessed when Time Selective Response or Dual Channel is the chosen algorithm.

The units for the impulse response can either be dB, where the resulting display is an Energy Time Curve, or linear values. With linear units, the impulse response will look like a *ring-down curve* that you would see on an oscilloscope. To change the impulse response units, select the **Units** tab.

Deconvolved Response

The Deconvolved Response is the response waveform divided by the stimulus waveform, in the spectral domain. It shows the Linear or Fundamental Impulse Response of the DUT, the Harmonic Impulse Response that occurs before it and the reflections (if any) that occur after it. The deconvolved Response is a result of the Time Selective Response algorithm. This helps you to properly position the time window so that the Fundamental Impulse Response is between the start and stop time, while leaving the Harmonic Impulse Response and Reflections outside the window. This can be tested by clicking the **Apply** button before using the algorithm in a sequence.

TSR Window

The Cosine Taper window used by the Time Selective Response algorithm, has a 10% taper at each end. The Fundamental Impulse Response must be inside these tapers. The example in ***Figure 13-31: TSR Window - Cosine Taper*** shows a 100 mSec window set on an impulse response. The taper of the TSR window disregards the first and last 10 mSec of the impulse.

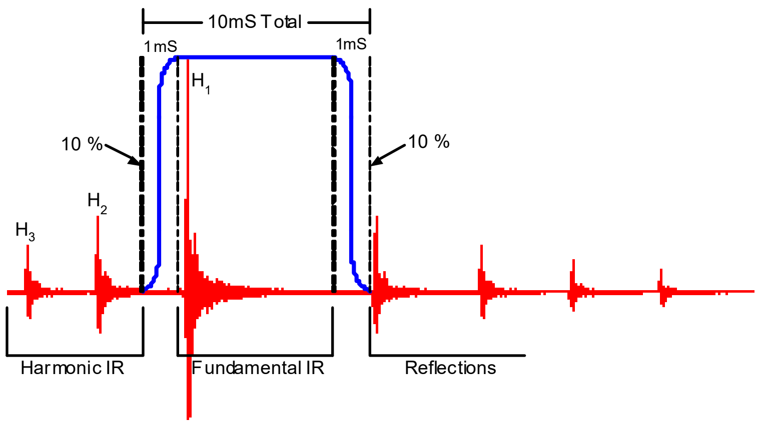


Figure 13-31: TSR Window - Cosine Taper

Definitions of each of the window types can be found in ***Weighting and Window Types on page 597***.

Note: As of SoundCheck 8, you can choose other window types, as shown in ***Figure 13-32***.

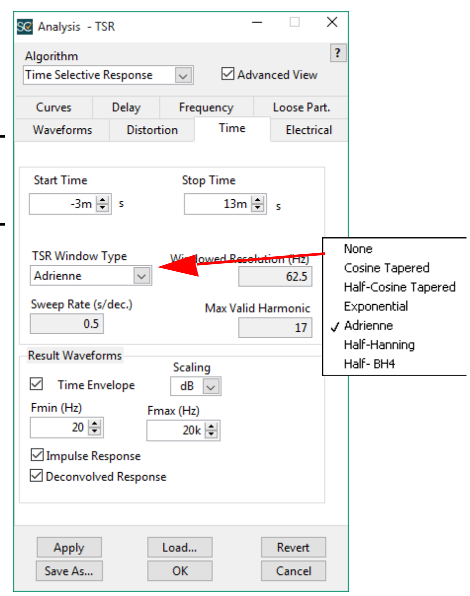


Figure 13-32: TSR Window Type

The **TSR Window** is output in the Memory List as a waveform. This can be displayed on top of the **Impulse Response** or the **Deconvolved Response** waveforms to check the time alignment as shown in ***Figure 13-33***.

The fundamental impulse response must fall inside the window. If it does not, simply edit the Start and Stop times in the Analysis Editor as shown in ***Figure 13-32***. The start time should be slightly before the start of the impulse (0.20 mSec for Adrienne) and the stop time should be before the first reflection.

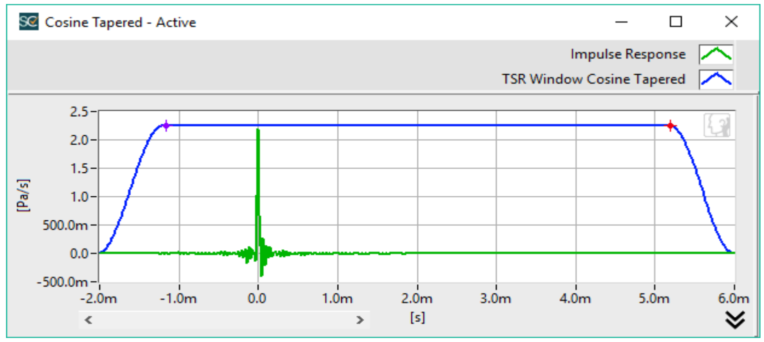
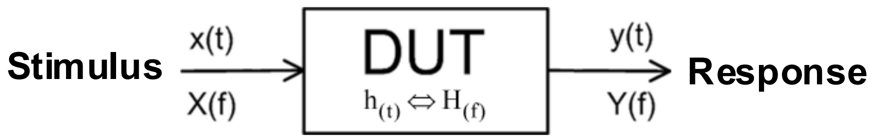


Figure 13-33: Impulse Response and TSR Window

Transfer Functions - Dual Channel Analysis

Dual Channel spectrum analysis yields the true transfer function between the input and output.

This technique should be used mainly with broadband stimulus such as noise. However, it could also be used with arbitrary waveforms such as voice or music.



Spectral Scaling

- Spectral Density should be used and is necessary for determining NonCoherent Distortion. Refer to ***Spectral Scaling - Frequency Tab on page 173*** for more information.

A scan FFT is performed in parallel on x(t) and y(t), which yields to two series of spectrum: $\{X_0, X_1, \dots, X_{N-1}\}$ and $\{Y_0, Y_1, \dots, Y_{N-1}\}$.

By averaging these spectrum together we get:

Auto-spectrum:

$$G_{XX} = E\lfloor \bar{X}X \rfloor$$

Stimulus Average Power Spectrum

E[.] is the mean and $\bar{X}$ denotes the complex conjugate of X.

Auto-spectrum:

$$G_{YY} = E\lfloor \bar{Y}Y \rfloor$$

Response Average Power Spectrum

Cross-spectrum:

$$G_{XY} = E\lfloor \bar{X}Y \rfloor$$

Stimulus/Response Average Cross Spectrum

The average cross-spectrum between stimulus & response.

Frequency Response:

$$H_1 = \frac{G_{XY}}{G_{XX}}$$

An unbiased estimator when noise is present at the output.

Impulse Response:

$$h_{1(t)} = \Im^{-1}[H_{1(f)}]$$

Response of the DUT to an ideally short impulse. $\Im^{-1}$ is the inverse Fourier Transform.

Coherence & Non-coherence:

$$\gamma^2 = \frac{|G_{XY}|^2}{G_{XX}G_{YY}}$$

$1-\gamma^2$, these functions give you the degree of linear relationship between the stimulus and response for each frequency.

Coherent Power (CP)

$CP = \gamma^2 \cdot G_{YY}$ CP is the part of G_{YY} which is linearly related to the stimulus.

Non-Coherent Power (NCP)

$NCP = (1 - \gamma^2) \cdot G_{YY}$ NCP is the part of G_{YY} which is not linearly related to the stimulus, such as noise and distortion.

Signal to Noise Ratio:

$SNR = \frac{\gamma^2}{1 - \gamma^2} = \frac{CP}{NCP}$ The Signal to Noise Ratio is the ratio at each frequency, between the power linearly related to the stimulus and the part non-linearly related, such as noise and distortion.

Non-Coherent Distortion in %

$NCD_{(f)} = 100 \sqrt{\frac{NCP}{\sum_f G_{YY}}}$ The NCD gives the proportion of noise and distortion which is present in the Response Spectrum G_{YY}. It is a function of frequency. Making a power sum of it with Post-processing will give you a single Noise & Distortion number. The NCD function used along with multitone or noise stimulus will give you a global assessment of the non-linearities of your DUT.

Auto Correlation of Stimulus:

$C_{xx(t)} = \Im^{-1}[G_{XX(f)}]$ The peak value is equal to the total power of the stimulus.

Stimulus Auto Correlation

Auto Correlation of Response:

$C_{yy(t)} = \Im^{-1}[G_{YY(f)}]$ The peak value is equal to the total power of the response.

Response Auto Correlation

Cross-Correlation of Stimulus and Response:

$C_{xy(t)} = \Im^{-1}[G_{XY(f)}]$ The position of the peak yields the delay of the input to the output.

Stimulus & Response Cross-Correlation

Multitone

The results are similar to Dual Channel Analysis but the Frequency Response is the amplitude and phase of only the tones that are present.

Because all the tones are analyzed in parallel, Multitone is the fastest way to get the Frequency Response.

Regarding distortion; because a Multitone stimulus has a rich frequency content (like "real-life" signals, e.g., music), it produces more realistic distortion components.

Using Multitone, the Non-Coherent Distortion curve yields a quick and global distortion assessment of the DUT.

In addition, one can make the Power Sum (using Post-Processing) of the Non-Coherent Distortion curve to get a single percentage. That number quantifies the global Distortion & Noise present at the DUT and can be used as a quality figure.

Spectral Scaling

- In most cases, RMS should be used since **Multitone** is a collection of fixed sine tones. Spectral Density should be used and is necessary for determining Non-Coherent Distortion. Refer to ***Spectral Scaling - Frequency Tab on page 173*** for more information.

Applying EQ

In Dual Channel and Multitone Analysis, the frequency response of the DUT is normally obtained by dividing the Response Spectrum (Y) by the Spectrum at the input of the DUT (X).

$$\text{Frequency Response} = \frac{Y}{X}$$

When EQ is applied in the Stimulus Step, the Spectrum at the input of DUT (X) is assumed to be the Stimulus Spectrum (S) divided by the EQ Curve (EQ).

$$\text{Frequency Response} = \frac{Y}{S/EQ}$$

Complex vs. Power Averaging - Frequency Tab

For Dual Channel and Multitone Analysis, the cross-spectrum yields the average phase relationship between input and output. The cross-spectrum is calculated when the Complex Averaging option is selected from the Frequency tab of the Analysis Editor. See ***Figure 13-34***.

- Complex is selected by default

Cross-spectrum cannot be used when making measurements on devices with non-stable phase due to frequency shift or jitter (e.g., Bluetooth headsets or MP3 players).

As an alternative, select Power, which gives an estimate of the frequency response based on the Auto-Spectra only. In this case, the cross-spectrum is not calculated so phase information is no longer available and the list of analysis functions is reduced.

When Power is selected, the following functions are available in the Analysis Editor:

- Auto Spectrum - Stimulus and Response
- Frequency Response (Magnitude Only)
- Time Envelope (Time Tab)
- Auto Correlation - Stimulus and Response (Time Tab - Dual Channel only)

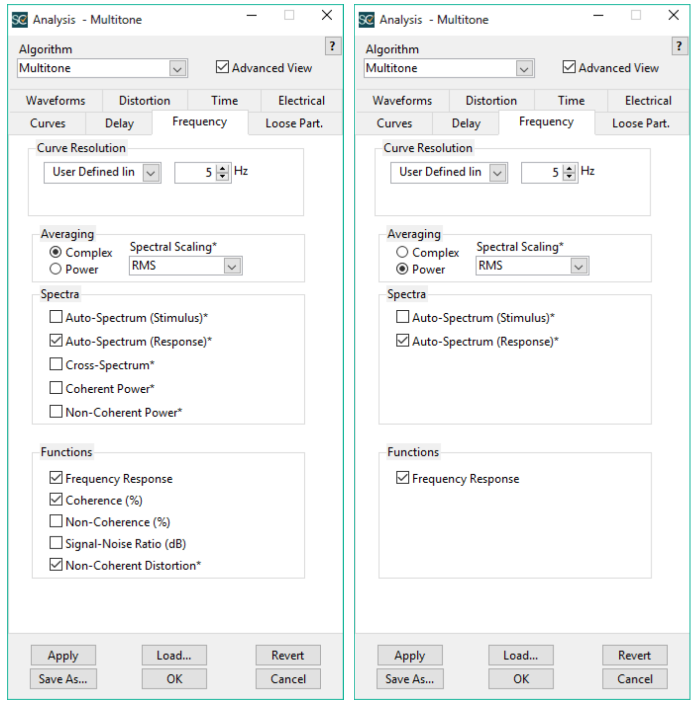


Figure 13-34: Complex vs. Power Averaging

The Complex and Power options apply only to Dual Channel and Multitone Analysis.

Note: A faster frequency shift algorithm with output of the jitter curve in the Memory List offers increased testing speeds for digital devices which have their own digital clock.

RTA Spectrum

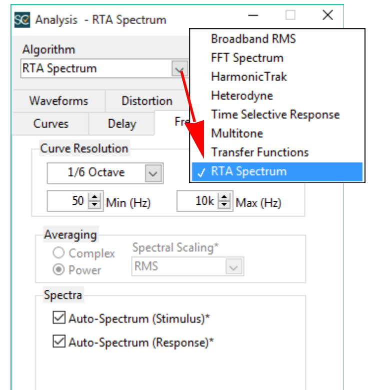


Figure 13-35: RTA Spectrum

In addition to calculating the RTA spectrum of the response waveform, the RTA analysis algorithm now allows the option to calculate the spectrum of the stimulus as well as the overall frequency response (comparing the response to the stimulus). This is useful when analyzing non-stationary signals, for example speech signals in telephony where compensation needs to be made for a non-flat stimulus.

The RTA Algorithm applies the RTA filter bank on the selected response waveform and yields the average 1/n octave spectrum in the Memory List. The advantage of this method over the RTA virtual instrument is that the analysis is perfectly synchronized with the stimulus, and the averaging time fits exactly to the length of the waveform. This eliminates the need for several stimulus repetitions and reduces the total test time.

This algorithm yields the exact same response spectrum measured with the RTA virtual instrument, and it conforms to the **ANSI S1.11 - 2004 class 0** standard.

This can also be used to perform a multichannel acquisition and analysis of the stimulus, which yields a synchronized RTA spectrum for all channels, with only one run of the stimulus. It is sometimes difficult to start the RTA virtual instrument at exactly the right time to capture the desired spectrum. This process eliminates such a synchronization problem. For example:

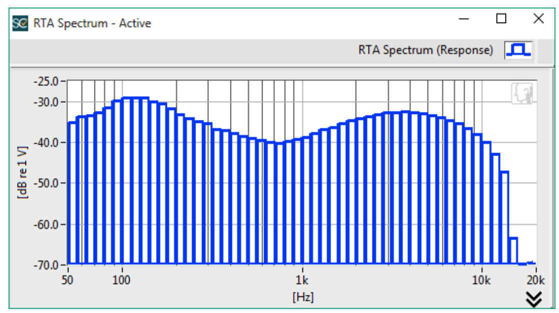


Figure 13-36: RTA Spectrum Graph

- Play P50 speech only one time
- Record two channels (send and receive or left and right)
- Both spectrums are synchronous since they were recorded at the same time
- The RTA spectrum for both channels requires only two analysis steps
- The algorithm processing time has been optimized to reduce sequence runtime

In addition to calculating the RTA spectrum of the response waveform, the RTA analysis algorithm also allows the option to calculate the spectrum of the stimulus as well as the overall frequency response (comparing the response to the stimulus).

- Spectra - Select to output either of the RTA curves to the Memory List
- Functions - Select Frequency Response to output the curve to the Memory List

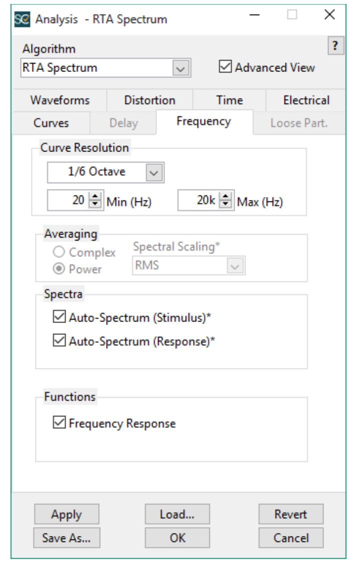


Figure 13-37: RTA Spectrum Settings

Applying EQ

In RTA Spectrum Analysis, the frequency response of the DUT is normally obtained by dividing the Response Spectrum (Y) by the Spectrum at the input of the DUT (X).

- When EQ is applied in the Stimulus Step, the Spectrum at the input of DUT (X) is assumed to be the Stimulus Spectrum (S) divided by the EQ Curve (EQ).

$$\text{Frequency Response} = \frac{Y}{X} = \frac{Y}{S/EQ}$$

Analysis Settings

Frequency Tab - Curve Resolution

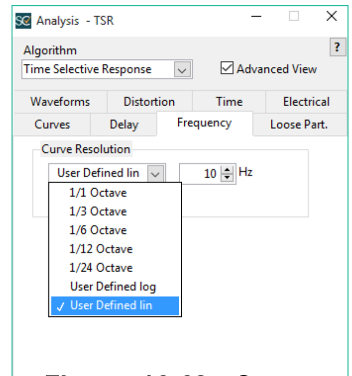


Figure 13-38: Curve Resolution

Curve Resolution is available for all FFT based frequency curves: LogTSR, Spectrum, Transfer Functions (Dual Channel) and Multitone.

Selecting **1/Nth Octave** resolution automatically sets the resolution of the curves that are output by the sequence, and performs smoothing on those curves. This resolution change reduces the number of points in the resulting curve, making the sequence operate faster.

- Preset selections:

1/1, 1/3, 1/6, 1/12, 1/24 octave (ISO frequencies)

- User Defined lin (Hz)

For backward compatibility, Analysis Steps from previous versions of SoundCheck will copy the frequency resolution value from the old version into the **User defined lin** resolution field.

- User Defined log (1/n oct)

Allows you to define the 1/Nth octave resolution.

Distortion Tab

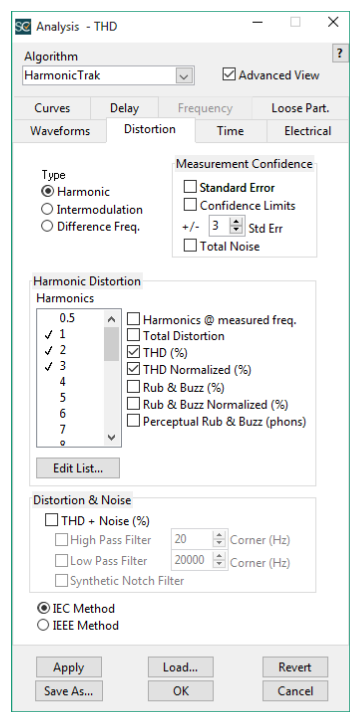


Figure 13-39: THD w H2, H3

Distortion is only available when using the HarmonicTrak and Time Selective Response algorithms.

<u>Harmonic Distortion</u>

Harmonics

The Harmonics field value indicates the harmonic number; 1 is the fundamental, 2 is the 2nd harmonic, etc. Select which harmonics to measure, either individually or as a group. You can also measure sub-harmonics, e.g., 0.5 harmonic. Use the **Edit List** button to modify list preferences.

Note: Checking 1 in the Harmonics list of the Analysis Editor does not change the analysis process. The Fundamental is always available, even if 1 is not checked.

Data added to Memory List is shown in the Curves Tab.

Default names are:

Fundamental, Harmonic 2, Harmonic 3, etc.

These can be shortened if you are checking “Add Input Data Name” or “Use Signal Path Name”.

Edit List

This button enables modification of the list of harmonics. To add a single harmonic, enter the single harmonic value (e.g., 5) and click **Add**. To enter a range (e.g., 10 through 15) enter the starting harmonic in the box to the left of the **To** button. Then click **To** and enter the ending harmonic. Clicking **Add** will add the harmonic family to the list.

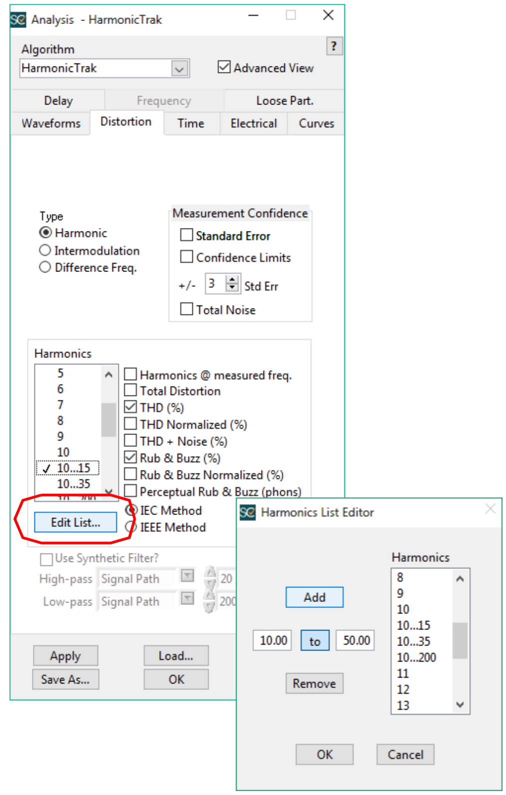


Figure 13-40: Edit List

Harmonics Plotted at Actual Measured Frequency

Traditionally, the harmonics of the signal are displayed at the excitation frequency. When **Harmonics at measured frequency** is selected, harmonics will be displayed on the XY Graph at the actual measured frequency. By displaying the shifted harmonics and the Fundamental, one can determine whether harmonic amplitude levels increase or decrease due to amplitude changes in the frequency response or are actual increases or decreases in distortion energy. THD and Rub & Buzz distortion calculated using the amplitude normalized distortion method removes the effects of DUT's frequency response modifying distortion levels.

For more information, see: *How to Graph Distortion Measurements* by Steve F. Temme, found at the Listen website, https://www.listeninc.com/how-to-graph-distortion-measurements/

In ***Harmonics at Actual Measured Frequency on page 189*** you can see the harmonics plotted at actual measured frequencies.

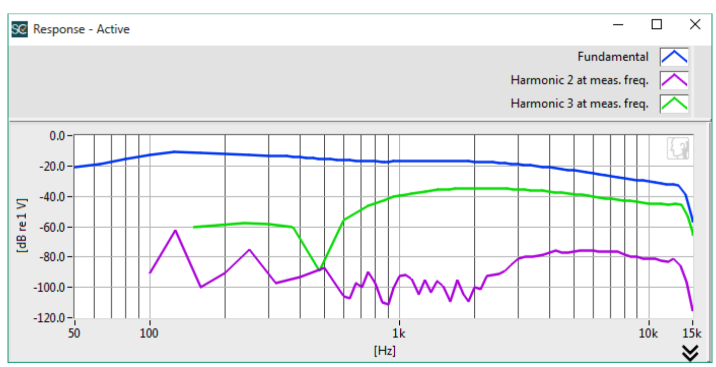


Figure 13-41: Harmonics at Actual Measured Frequency

Total Distortion

Total Distortion (TD) is the power sum of all the harmonics selected in the Analysis Step. It is presented in the units of the Harmonics themselves. It represents the level of distortion at each excitation frequency. The calculation is based on the Harmonics selected in the *Analysis Editor*.

$$TD = \sqrt{(H_2^2 + H_3^2 + \ldots + H_n^2)}$$

Total Distortion

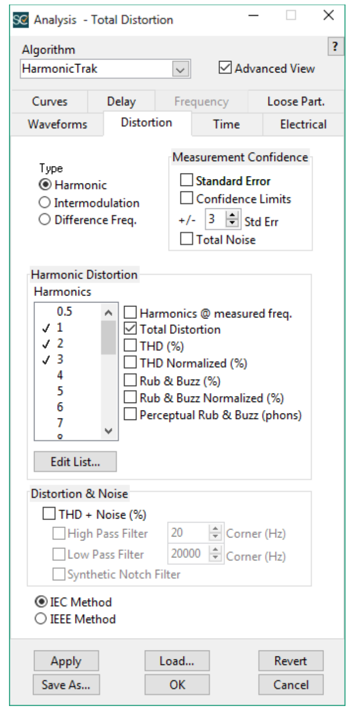


Figure 13-42: Total Distortion

References:

- *Values in the equations are RMS engineering units, unless labeled in %. However, the final distortion curves are frequently shown in dB (SoundCheck default).*
- *All above equations are a function of frequency (f)*
- *n indicates the distortion order*
- *F is the Fundamental (aka H_1)*
- *H_n is the harmonic of n^{th} order*

Total Harmonic Distortion

Total Harmonic Distortion (THD) is the percentage of the total signal that is affected by distortion due to the harmonics. Total Distortion (TD) is referenced in the equations. See ***Total Distortion on page 190***.

Selecting IEC method means that the distortion ratio is the power sum of the distortion components divided by the total input power (fundamental + distortion components). The square root of this ratio is presented in percent. The distortion is always <100%.

$$\%\,\mathrm{THD} = 100 \times \frac{\mathrm{TD}}{\sqrt{\mathrm{F}^2 + \mathrm{TD}^2}}$$

Total Harmonic Distortion in % (IEC)

Selecting IEEE method means that the distortion ratio is the power sum of the distortion components divided by the power of the fundamental. The square root of this ratio is presented in percent. The distortion may be >100%.

(The IEEE standard allows for this method and what is described as the IEC method mentioned above.)

$$\%\,\mathrm{THD} = 100 \times \frac{\mathrm{TD}}{\mathrm{F}}$$

Total Harmonic Distortion in % (IEEE)

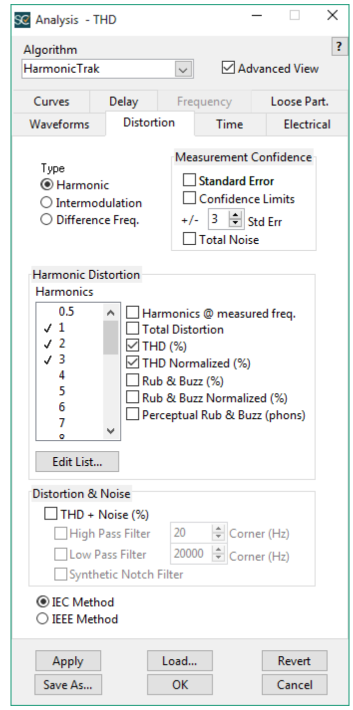


Figure 13-43: THD

(See ***Intermodulation and Difference Distortion on page 199*** for more options).

The THD measurement only takes into account the harmonics selected in the **Analysis Step Distortion** tab.

Note: Checking 1 in the Harmonics list of the Analysis Editor does not change the analysis process. The Fundamental is always available, even if 1 is not checked.

References:

- *Values in the equations are RMS engineering units, unless labeled in %. However, the final distortion curves are frequently shown in dB (SoundCheck default).*
- *All above equations are a function of frequency (f)*
- *n indicates the distortion order*
- *F is the Fundamental (aka H_1)*
- *H_n is the harmonic of n^{th} order*
- *N is Noise*

THD + Noise

Total Harmonic Distortion plus Noise (THD+N) is a measurement that expresses the ratio of all the harmonic and noise energy to the total signal. DC is excluded.

As of SoundCheck 16, the THD+N algorithm has been improved as follows:

- Accurate THD+N measurements can now be made with far shorter stimulus times
- The notch filter used complies with AES17 and will produce identical results to alternative measurement systems
- Optionally, a Synthetic Notch Filter will produce more accurate and much faster results
- User defined High Pass and Low Pass filters are available to define the bandwidth of the measurement

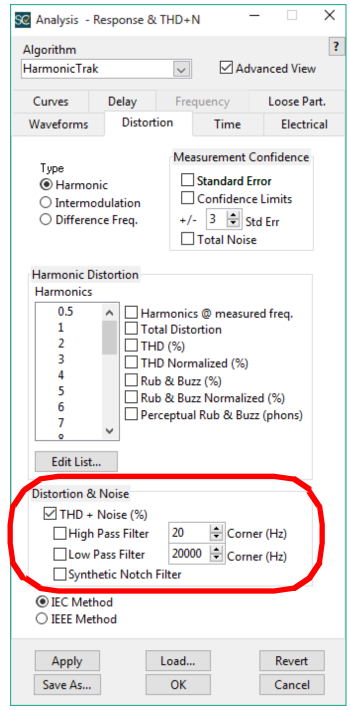


Figure 13-44: THD+N

Controls:

Checking THD+Noise (%) produces a curve in the Memory List.

- If High Pass Filter is unchecked, no High Pass filter is applied
- If the Low Pass Filter is not checked a default low pass filter defined by the Anti-alias Frequency set in Hardware Editor is used
- If High Pass Filter is checked, you are able to set the low frequency limit of the measurement
- High-pass filter is Elliptic, Brick Wall, 5th order. The Corner Frequency is set in the Corner (Hz) field. Default value is 10 Hz.
- Low-pass filter is Elliptic, Brick Wall, 8th order. The Corner Frequency is set in the Corner (Hz) field. Default value is 20 kHz.

Note: The THD+N algorithm requires a Stimulus Step Minimum Duration of 20 mSec. Any time discarded in Analysis requires a corresponding increase in Stimulus Step Duration.

Measurement Accuracy and Repeatability

The key to measuring THD+N accurately is to understand that the Noise component is a broadband measurement. The Step Size defined for the stimulus must be sufficiently long in time to accurately measure a signal at the High Pass Frequency. In addition, if the device is noise dominated, to produce repeatable measurements will require a long Step Size in order to average the random noise.

Example:

- The high pass frequency is set to 20 Hz.
- The Step Size cannot be less than 1 cycle or 50 mSec.
- The practical minimum Step Size would be 3 cycles or 150 mSec.
- To account for typical transients between sweep steps, would normally require 5 cycles or 250 mSec.
- For a typical noise dominated electronic device, dwell times of 1 second or more may be required for repeatable measurements.
- These time constants will scale with the High Pass frequency.

See ***Transition Discard Time - Time Tab on page 175***.

THD+N Equations

THD+N IEC

$$\% \text{ THD+N} = 100 \times \frac{\sqrt{H_2^2 + H_3^2 + \ldots + H_n^2 + N^2}}{\sqrt{F^2 + H_2^2 + H_3^2 + \ldots + H_n^2 + N^2}}$$

Total Harmonic Distortion in % (IEC)

THD+N IEEE

$$\% \text{ THD+N} = 100 \times \frac{\sqrt{H_2^2 + H_3^2 + \ldots + H_n^2 + N^2}}{F}$$

Total Harmonic Distortion in % (IEEE)

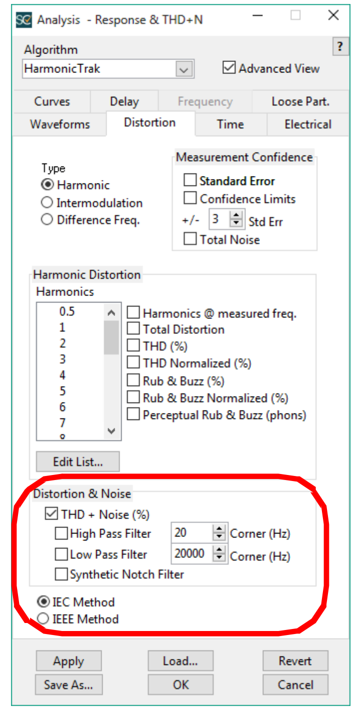


Figure 13-45: THD+N

Virtual Instrument THD+N Options

THD+N Residual - The level of all the noise and distortion products in the measurement bandwidth

SINAD - Is the reciprocal of THD+N, if and only if THD+N is calculated without High and Low Pass filters in the Analysis Editor

$$\text{SINAD} = \frac{P_{signal} + P_{noise} + P_{distortion}}{P_{noise} + P_{distortion}}$$

SINAD (Signal to noise and distortion ratio)

References:

- *Values in the equations are RMS engineering units, unless labeled in %. However, the final distortion curves are frequently shown in dB (SoundCheck default).*
- *All above equations are a function of frequency (f)*
- *n indicates the distortion order*
- *F is the Fundamental (aka H_1)*
- *H_n is the harmonic of n^{th} order*
- *N is Noise*
- *P_n is the Average Power of the n component*

Rub & Buzz

Rub & Buzz is the power sum of all harmonics selected above the 9th harmonic divided by the fundamental. Harmonics 10 and higher are the main contributors to the audible rub and buzz even though lower order harmonics may be higher in level. For more information refer to: ***Are you Shipping Defective Speakers to your Customers*** **by Steve F. Temme**, found on the Listen website.

$$Rub\ \&\ Buzz\ (f)\ =\ 100\sqrt{\frac{\sum\limits_{n \geq 10} |H_n(f)|^2}{|H_1(f)|^2}}$$

Rub and Buzz in %

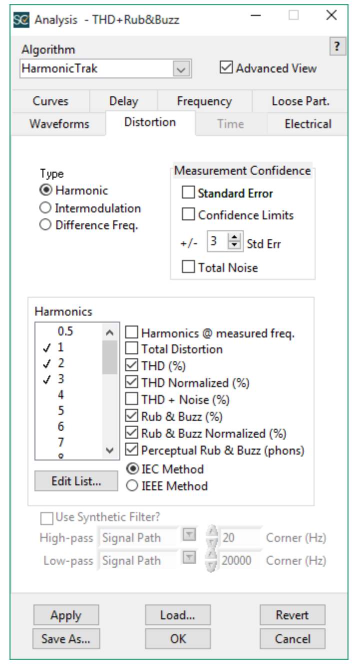


Figure 13-46: Rub and Buzz

Normalized THD or Normalized Rub and Buzz

An alternate algorithm for the THD and Rub & Buzz measurements is calculated using the harmonics after re-plotting them at the actual measured frequency of their signals.

See ***Harmonics Plotted at Actual Measured Frequency on page 189***.

The % THD and % Rub & Buzz is then calculated using the following methods:

THD Normalized Equation

$$Normalized\ THD\ (f)\ =\ 100\sqrt{\sum_{n \neq 1}\left|\frac{H_n(f)}{H_1(nf)}\right|^2}$$

Normalized THD in %

Rub & Buzz Normalized Equation

$$Normalized\ Rub\ \&\ Buzz\ (f)\ =\ 100\sqrt{\sum_{n \geq 10}\left|\frac{H_n(f)}{H_1(nf)}\right|^2}$$

Normalized Rub and Buzz in %

The values of THD Normalized and Rub & Buzz Normalized can be compared to the Harmonic “**n**” shifted curves (where “**n**” is from 2 to your highest harmonic requested).

The formulas above use the IEEE method. The IEC method can also be used. Please Refer to ***Intermodulation and Difference Distortion on page 199*** for information on IEC vs. IEEE.

References:

- *Values in the equations are RMS engineering units, unless labeled in %. However, the final distortion curves are frequently shown in dB (SoundCheck default).*
- *All above equations are a function of frequency (f)*
- *n indicates the distortion order*
- *H_1 (or F) is the Fundamental*
- *H_n is the harmonic of n^{th} order*

Rules - Normalized THD/Normalized Rub and Buzz

- If the stimulus waveform is from an Amplitude Sweep stimulus step, the following Harmonics options are grayed out and not available:
  - Harmonics @ measured freq.
  - THD Normalized (%)
  - Rub & Buzz Normalized (%)
- Even though the response measurement may go to 20 kHz, Normalized THD measurements stop at 10 kHz. For a normalized distortion measurement, the maximum measured frequency is the stimulus frequency divided by the highest order harmonic being measured.

  For example: If you are sweeping up to 20 kHz and measuring the 2nd through 5th harmonic (as is common for THD measurements):
  - The **2nd** harmonic distortion product will be measured up to 10 kHz, the **3rd** up to 6.67 kHz, the **4th** up to 5 kHz and the **5th** up to 4 kHz.
  - The measurement will stop at 10 kHz as there are *no normalized harmonic distortion components calculated above this frequency*. It should also be noted that above 4 kHz, the harmonics are not included.
- In other words, ***it is impossible to normalize (ratio) the "harmonics at their measured frequencies" to the fundamental, at stimulus frequencies not present in the measurement.***

  i.e., If the stimulus range does not include the frequency range of high order harmonics.
  In regular Rub and Buzz, the ratio of the harmonics to the fundamental are compared at the stimulus frequency but still have to be within the passband (Alias free Freq) of the sampling rate.

  Please refer to the following papers on the Listen website: ***Harmonic Distortion Measurement: The effects of sampling rate and stimulus frequency on the measured harmonic frequency (including THD and Rub & Buzz)*** ***by Steve F. Temme***.

  How to Graph Distortion Measurements **by Steve F. Temme**.

Measure Relative to Fundamental Only

THD and Rub & Buzz measurements can be calculated using one of two methods. Choose IEC to include the fundamental and all the harmonics (Total Distortion) in the denominator (typically used in Europe), or IEEE to use the fundamental (first harmonic, typically used in the USA). These are selected in the bottom section of the *Analysis Editor - HarmonicTrak Algorithm - Distortion Tab* as shown in ***Figure 13-47***.

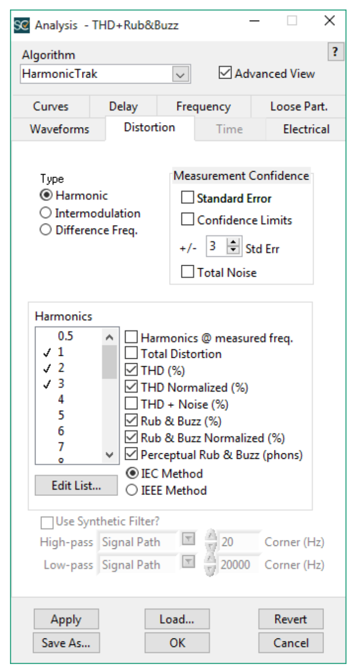


Figure 13-47: THD and Rub & Buzz Selections

Perceptual Rub & Buzz - CLEAR™ Distortion Measurement

Requires optional module **2030 - Perceptual Rub & Buzz**.

The new CLEAR (Cepstral Loudness Enhanced Algorithm for Rub & Buzz) algorithm from Listen offers true Perceptual Rub & Buzz analysis for production line applications.

This is selected in the bottom section of the *Analysis Editor - Distortion Tab* as shown in ***Figure 13-48***.

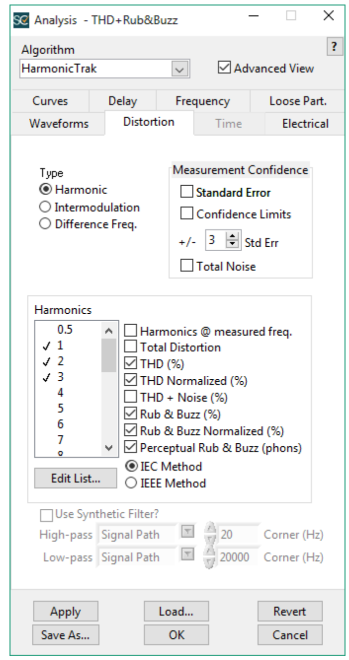


Figure 13-48: Perceptual Rub & Buzz

> ***Note:*** The Distortion Tab is only available for the *HarmonicTrak* and *Time Selective Response* algorithms.

It uses a simplified auditory perceptual model to measure the loudness of Rub & Buzz distortion in phons rather than the more traditional dB SPL and % distortion units. These better identify whether distortion due to manufacturing defects can be heard by the listener than conventional measurements. In addition to a result which corresponds more accurately to the human ear, this new test method also offers two significant advantages for use on the production line:

- It is less sensitive to transient background noises than traditional methods, therefore is reliable in noisy environments
- It is much simpler to set limits than when using conventional distortion measurements

> ***Important! As of SoundCheck 11.0, a threshold was added to the Perceptual Rub & Buzz algorithm that will output zero if that threshold isn't met. In this case it is normal to see a flat line at 0 Phons. See Comparison on page 197.***

Perceptual vs. Conventional Rub & Buzz

Conventional Rub & Buzz detection has been widely used on the production line since Listen introduced it in SoundCheck Version 1 back in 1996. It offers excellent identification of Rub and Buzz defects caused by manufacturing problems, and will continue to do so. In recent years, some manufacturers have moved to a defect detection model where they prefer only speakers with audible faults to fail QC checks. This is because yields are higher when only speakers with audible faults are rejected rather than any faults at all. Perceptual Rub & Buzz offers a means of identifying and precisely quantifying this with all the benefits and reliability of an automated test system.

Perceptual Rub & Buzz using the CLEAR™ algorithm shows audible distortion more clearly. Traditional Rub & Buzz measurements do not take into account the insensitivity of the human ear to low and high frequencies, therefore it is more difficult to identify problem areas and set limits on a production line.

Comparison

Figure 13-49: Example A - Good Speaker:

- Rub and Buzz is a low percentage - Red line
- Perceptual Rub and Buzz is below the threshold of perception across the range of the measurement, so the resulting curve is a flat line at 0 Phons - Black dotted line

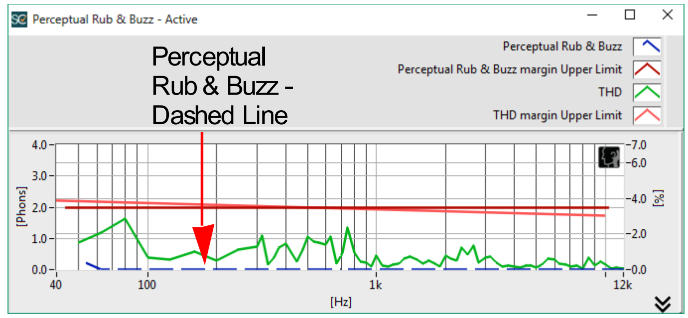


Figure 13-49: Example A - Good Speaker

Figure 13-50: Example B - Bad Speaker:

- Rub and Buzz is very high at low frequencies with an 8% peak at 160 Hz - Red line
- The Perceptual Rub and Buzz curve shows the more audible cone breakup of 8.9 Phons at 530 Hz, along with the noticeable low frequency distortion at 125 Hz - Black dotted line

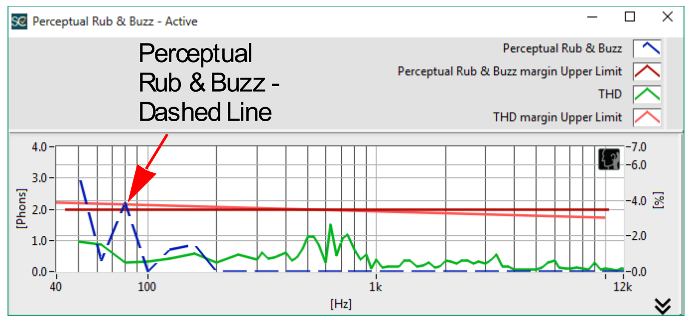


Figure 13-50: Example B - Bad Speaker

The CLEAR Rub & Buzz Detection Algorithm

Listen's CLEAR™ Rub & Buzz detection algorithm uses true perceptual analysis to 'hear' any faults in the speaker. It offers many advantages over other 'perceptual' Rub & Buzz analysis systems:

- True Perceptual Rub & Buzz: The CLEAR Algorithm is a true perceptual Rub & Buzz algorithm. Based on well-proven psychoacoustic principles, it accurately replicates the human ear using mathematical models found in MP3 encoders that mimic the way that both the ear and the brain interpret sound. This results in close to 100% correlation to the human ear.
- Less sensitive to transient background noise: A significant advantage of our Perceptual Rub & Buzz algorithm is that it is very insensitive to transient background noise – tests show that it offers far more consistent results with high background noise levels than other Rub & Buzz measurement methods. This makes it ideal for noisy factory environments.
- Flexible: The CLEAR Rub & Buzz detection system is extremely flexible. While it can of course be configured for a simple pass/fail result, it can also offer detailed results including defect analysis and offers a calibrated loudness value rather than simply a comparison to a reference.
- Better Correlation to Human Ear: Testing carried out by an independent laboratory shows excellent correlation to the human ear.

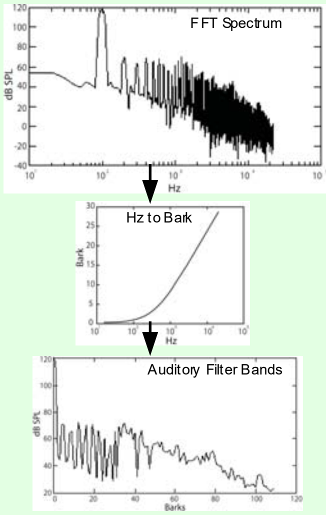
More details of the research leading to the development of this algorithm are presented in the paper: ***'Practical Measurement of Loudspeaker Distortion Using a Simplified Auditory Perceptual Model'***, found on the Listen website.

CLEAR™ Algorithm For Perceptual Rub & Buzz Analysis

Stimulus applied to transducer

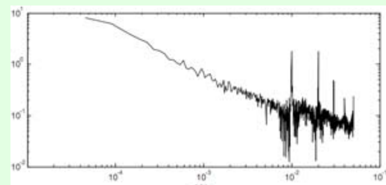
The algorithm uses a sine wave stimulus because this is widely accepted as the standard test signal for production line testing in the loudspeaker industry.

Response Spectrum



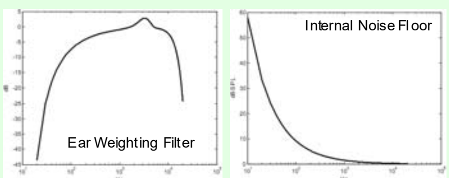
Auditory filter bands are applied to the response signal to convert the FFT spectrum (constant bandwidth) to a Bark scale (auditory filter bands). This replicates the way the ear filters sound.

Auditory Filter Bands



Harmonic Structure

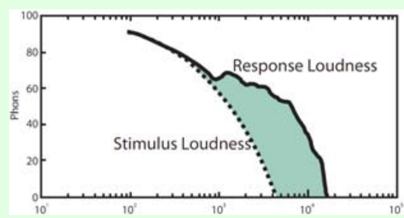
The harmonic structure of the response is quantified using the power cepstrum (a cepstrum is a spectrum of a log spectrum). A strong and extended harmonic structure is a signature of Rub & Buzz.

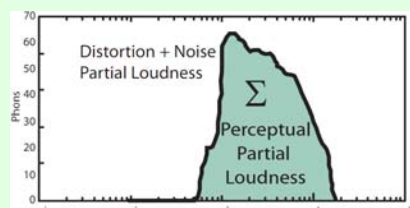


An ear weighting filter compensates for the transfer function of the outer to inner ear, and the internal noise of the ear (noise floor due to blood flow) is added. Together these model the frequency response of the ear.

Ear Weighting

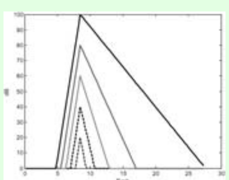
Internal Noise





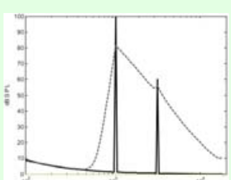
The fundamental and its masking effects are subtracted out from the result for the response signal to give the distortion of the speaker plus noise. This is summed over the frequency range to give the perceptual partial loudness (in phons) for a single tone of the input signal.

Frequency Spreading



A frequency spreading function is applied. This is a simple mathematical representation of auditory masking curves.

Partial Loudness Calculation



This is how the algorithm mimics the psychoacoustic filters of the ear in hearing Rub & Buzz defects. These complex curves change with frequency and level.

PARTIAL LOUDNESS combined with HARMONIC STRUCTURE

The result of the harmonic analysis (a percentage measurement) is combined with the perceptual distortion for each frequency. This accentuates the rub & buzz, making it easier to identify and set limits.

PERCEPTUAL RUB & BUZZ

Intermodulation and Difference Distortion

When you play two tones in a non-linear system, they interact in such a way that you get new frequencies at the output. These frequencies are different linear combinations of the two original frequencies and are called orders. This is the case when music is played through a loudspeaker. These orders are particularly annoying because they have no harmonic relationship with the original frequencies.

Two types of two-tone distortion are commonly used.

- Intermodulation distortion (IM): this distortion occurs when a high frequency tone is superimposed on a high-level, low frequency tone. The high frequency signal is modulated by the low frequency.
- Difference frequency distortion (DF): this distortion arises when 2 tones are separated by a small frequency difference. This distortion is similar to harmonic distortion but is especially noticeable when the 2 tones are at high frequencies.

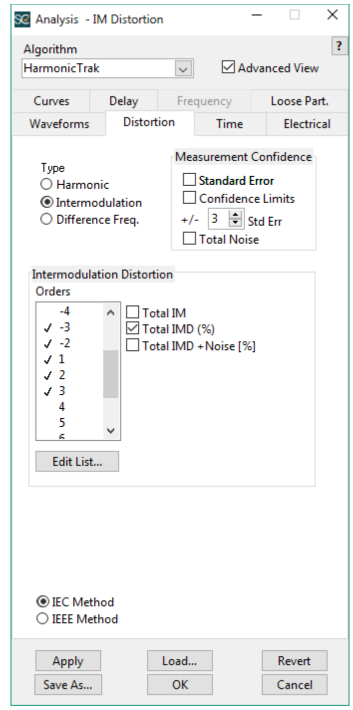


Figure 13-51: Total IMD

Application to Loudspeakers Measurements

Intermodulation distortion is used to detect amplitude and Doppler modulations that occur when low frequency signals produce large excursions of the speaker voice coil.

Difference frequency distortion is used to detect distortion at high frequencies, where single tone harmonic distortion would fall far out the frequency range of the loudspeaker or of the ear.

For more details on these techniques see: Steve Temme, ***Audio Distortion Measurements***, Bruel & Kjaer, Application Note BO 0385-11, found on the Listen website.

IM (or difference frequencies) are measured using step FFT analysis in a method similar to HarmonicTrak. For each step frequency, processing is applied on the entire spectrum of the signal.

IM Distortion Formulas

Total IM

$$\text{Total IM} = \sqrt{\sum_{n>1}(H_n + H_{-n})^2}$$

Total IM Distortion dB (IEC)

$$\text{Total IM} = \sqrt{\sum_{n>1}(H_n^2 + H_{-n}^2)}$$

Total IM Distortion dB (IEEE)

Total IMD %

$$\%\,\text{IMD} = 100 \times \frac{\text{Total IM}}{(F_1 + F_2)}$$

Total IM Distortion % (IEC)

$$\%\,\text{IMD} = 100 \times \frac{\text{Total IM}}{\sqrt{F_1^2 + F_2^2}}$$

Total IM Distortion % (IEEE)

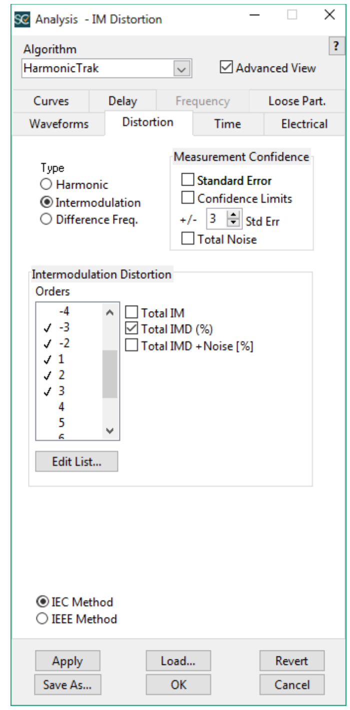


Figure 13-52: Total IMD

References:
- *All above equations are a function of frequency (f)*
- *Values in the equations are RMS engineering units, unless labeled in %. However, the final distortion curves are frequently shown in dB (SoundCheck default).*
- *n indicates the distortion order*
- *F1 is the IM Fundamental*
- *F2 is the IM Fixed Tone*
- *H_n are the n^{th} distortion orders*
- *N is Noise*

Difference Distortion Formulas

$$DFD = \sqrt{\sum_{n<0,\ \text{even}} H_n^{\,2} \; + \sum_{n>1,\ \text{odd}} (H_n + H_{-n})^2}$$

Total Diff Distortion in dB (IEC)

$$DFD = \sqrt{\sum H_n^{\,2}}$$

Total Diff Distortion in dB (IEEE)

$$TotalDFD = 100 \times \frac{DFD}{(F_1 + F_2)}$$

Total DF Distortion in % (IEC)

$$TotalDFD = 100 \times \frac{DFD}{\sqrt{F_1^{\,2} + F_2^{\,2}}}$$

Total DF Distortion in % (IEEE)

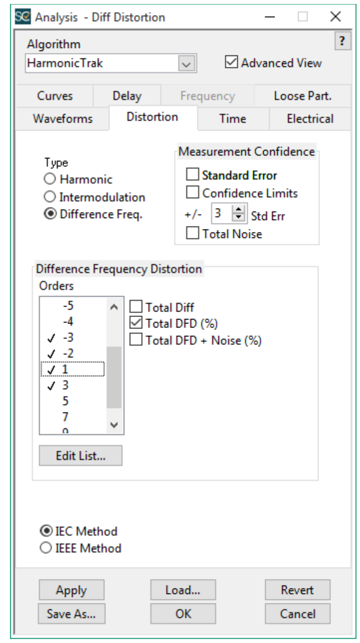


Figure 13-53: Total DFD

References:

- *All above equations are a function of frequency (f)*
- *Values in the equations are RMS engineering units, unless labeled in %. However, the final distortion curves are frequently shown in dB (SoundCheck default).*
- *n indicates the distortion order*
- *F1 is the Diff Upper Fundamental*
- *F2 is the Diff Lower Fundamental*
- *H_n are the n^{th} distortion orders*
- *N is Noise*

Total Distortion + Noise

The power sum of TD and Noise (IM or Diff)

$$\text{TotalD} + \text{N} = 100 \times \frac{\sqrt{\text{TD}^2 + \text{N}^2}}{\text{F}_1 + \text{F}_2}$$

Total Distortion + Noise in % (IEC)

$$\text{TotalD} + \text{N} = 100 \times \sqrt{\frac{\text{TD}^2 + \text{N}^2}{\text{F}_1^2 + \text{F}_2^2}}$$

Total Distortion + Noise in % (IEEE)

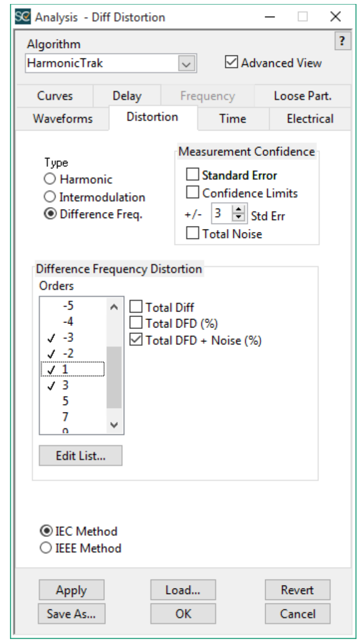


Figure 13-54: Total DFD + Noise

References:

- *All above equations are a function of frequency (f)*
- *Values in the equations are RMS engineering units, unless labeled in %. However, the final distortion curves are frequently shown in dB (SoundCheck default).*
- *n indicates the distortion order*
- *H_n are the n^{th} distortion orders*
- *N is Noise*

For IM Distortion:

- *F1 is the IM Fundamental*
- *F2 is the IM Fixed Tone*

For Diff Distortion:

- *F1 is the Diff Upper Fundamental*

Confidence and Noise

When making any measurement there is always a level of measurement uncertainty, partly due to noise. That noise adds randomness to the level of the measurement. Therefore a measurement is only an estimate of the true value. Since we know the level of noise, it is possible to calculate the Standard Error (σ) of the estimated level.

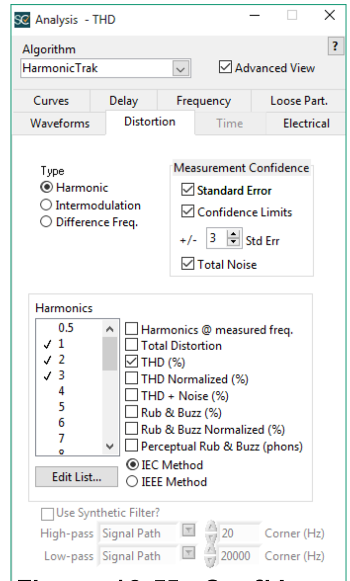


Figure 13-55: Confidence

Confidence Limits

If we consider one measurement with a value of x and a standard error of σ, then the confidence that the true value will be within the limits $\pm n\sigma$ is as follows:

- $[x-\sigma , x+\sigma]$ with 68% confidence
- $[x-2\sigma , x+2\sigma]$ with 97% confidence
- $[x-3\sigma , x+3\sigma]$ with 99.7% confidence.

These are the Confidence Limits, as selected in ***Figure 13-55***.

Standard Error

The Standard Error for the Fundamental as well as every selected harmonic or order can be displayed. The Standard Error is calculated in the same units as the Fundamental and the resulting curve is based on one standard deviation (σ).

There is a choice of having the Standard Error as a single curve (σ) or the Confidence Limits as 2 curves (measured curve $\pm a\ \sigma$). The factor a, is chosen by the user.

Total Noise

A Noise curve can also be displayed. This is the RMS level of Total Noise for each stimulus frequency.

Note: Please refer to the How To Example sequence: Confidence and Noise

This can also be used for frequency pairs when measuring Intermodulation and Difference Frequency distortion.

For more information on the Stimulus required for these measurements please Refer to ***Two Tone Stimulus on page 134***.

Note: Please refer to the How To Example sequences: IM Distortion.sqc and Diff Distortion.sqc.

Measurement Confidence Rules:

Here are some general recommendations to help to increase the measurement confidence and improve the repeatability of THD and THD+N results.

- Increase the input gain if Max FSD is below -30 dB
- Change the sweep direction to “High to Low”
- Increase the bit depth to 24 Bit
- Increase the min cycles/min duration in the Stimulus Step

Electrical Tab - Impedance

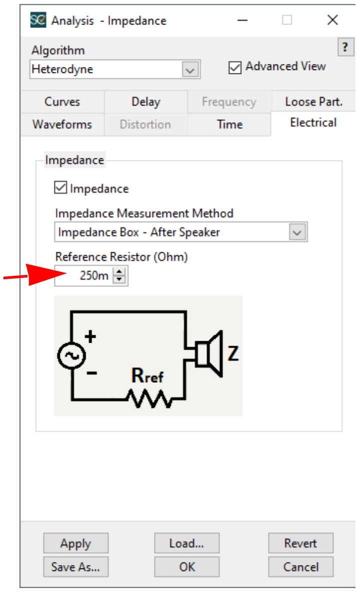


Figure 13-56: Reference Resistor Value

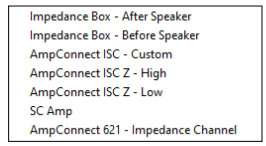
Impedance measures the voltage level across a known reference resistor and calculates the impedance.

SI units are used throughout SoundCheck 18.1.

0.25 Ohms is represented as 250 milliohms as shown in ***Figure 13-56***.

Impedance Measurement Method

Select **Impedance Box** if using the Impedance Measurement Interface box from Listen, Inc. as shown in ***Figure 13-56***. Note that you can select if the box is placed before or after the speaker in the circuit.



This method allows you to enter the value of the reference resistor used in the Impedance Box. Other methods have fixed values depending on the Listen hardware used.

Ref Resistor

Value of the reference resistor, in Ohms, placed in series with the transducer under test. See ***Impedance Setup on page 205*** for more details.

AmpConnect 621™, AmpConnect ISC™ and SC Amp™ Impedance Measurement Methods

Each of the Listen hardware devices have their own measurement methods. AmpConnect ISC is used in ***Figure 13-57***.

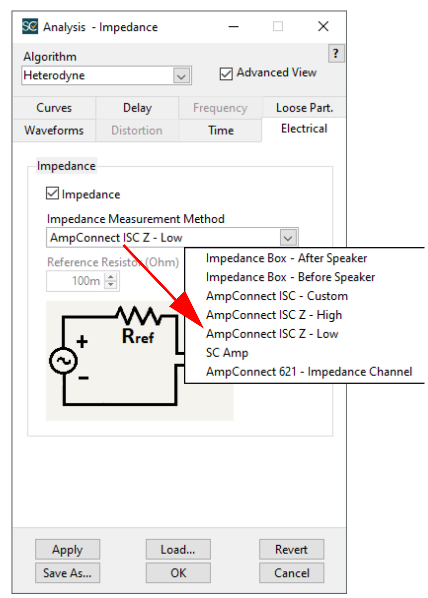


Figure 13-57: Impedance - AmpConnect ISC

Note: ***The AmpConnect front panel or AmpConnect Message Step/Startup Default must be set to the same Reference Resistor value as set in the Analysis Step of the sequence.***

The default value used in the default sequence **Complete test using AmpConnect** is Z-Low. For AmpConnect ISC you can also select Z-High to use a 1 Ohm reference resistor.

SC Amp™

As of SoundCheck 18.1, you can select **SC Amp** from the Impedance Measurement Method drop down. This uses a 0.1 Ohm reference resistor as in previous versions. Prior sequences that use the **AmpConnect ISC/SC Amp - Z Low** method will still work correctly as the resistor value and position in the circuit are the same.

Impedance Setup

Listen offers an optional **Impedance Measurement Interface Box** for connecting the power amplifier, transducer, and SoundCheck system. This features dual banana leads, alligator clips, a removable cable for the impedance measurement channel input and a removable cover for easy access when changing the reference resistor. For more information, please contact Listen, Inc.

(Refer to ***Loudspeaker Test Connections with Impedance Box on page 576*** for a detailed drawing of the impedance box)

To measure the transducer's impedance, a small resistor is connected in series with the transducer between its negative terminal and ground. A general rule of thumb is that the DUT Impedance should be somewhere between 20 to 40 times greater than the Reference Resistor. Refer to ***Impedance Measurement Details on page 206*** for more information.

- For example, use a 0.25 Ohm resistor for an 8 Ohm loudspeaker. This way both the acoustic response (e.g., left channel) and impedance response (e.g., right channel) of the transducer can be measured at the same time.

To measure the current flow through the resistor, connect Input 2 of the audio interface across the Reference Resistor as shown in ***Figure 13-58***.

Important! **WARNING!** Make sure to connect the ground of the output from the amplifier to the ground on Input 2 of the audio interface. The positive lead of Input 2 should be connected to the negative terminal of the transducer, the same as the resistor.

Enter the value of current sensing resistor in series with the transducer in the Ref. Resistor field of the Analysis Editor as shown in ***Figure 13-56***. The formulas below apply only to the Impedance Box Method.

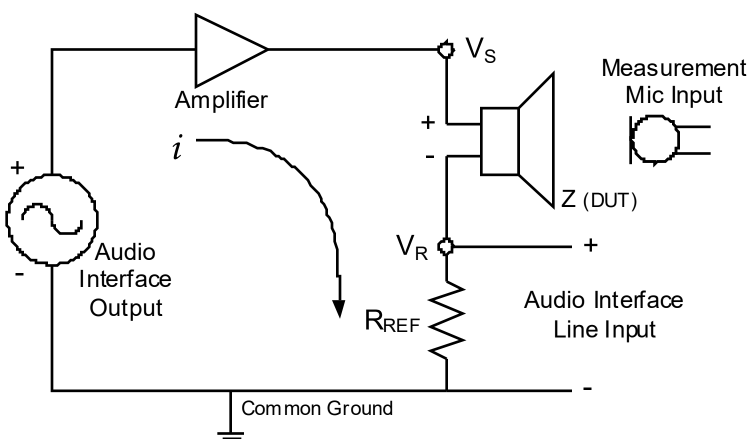
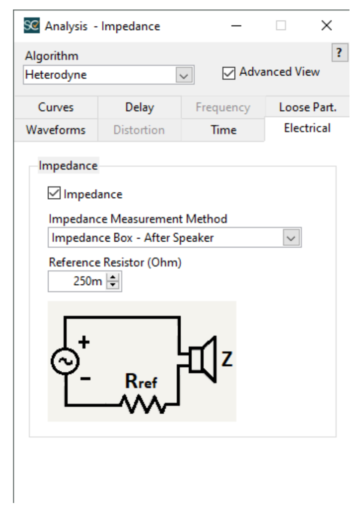


Figure 13-58: Impedance Box Circuit



- Z = impedance of device under test (e.g., loudspeaker)
- VS = voltage out of amplifier (measured during calibration)
- VR = voltage across resistor (e.g., 0.1 W reference)
- i = current running through DUT and reference resistor

$$Z = \frac{(V_S - V_R)}{i} \qquad i = \frac{V_R}{R_{REF}} \Rightarrow \quad Z = \frac{(V_S \times R_{REF})}{V_R} - R_{REF}$$

Impedance Measurement Details

When measuring impedance with SoundCheck, it helps to know the expected impedance of the Device Under Test.

This allows you to select a **Reference Resistor** value that provides a signal level to return to SoundCheck that is significantly above the noise floor of the system, but not so high that it overloads the inputs. If the Impedance curve as measured in SoundCheck appears to have jagged edges, this would indicate that the signal across the Reference Resistor is too low.

You also have to take into consideration the signal drop at the DUT due to the added Reference Resistor. This signal drop is not included in the Amplifier Calibration process, so it needs to be as small as possible.

The idea is to minimize the Signal Drop at the DUT while maintaining a sufficient signal level across the Reference Resistor.

Rules - Impedance Measurement

- A general rule of thumb is that the DUT Impedance should be somewhere between 20 to 40 times greater than the Reference Resistor.

To calculate the Signal Drop at the DUT:

$$\text{DUT Signal Drop (dB)} = 20 \times \log_{10}\left(\frac{\text{DUT Imp}}{\text{DUT Imp} + \text{Ref Res}}\right)$$

Signal Drop at DUT in dB

To calculate the Signal Level at the Reference Resistor:

$$\text{Ref Res Level (dBV)} = 20 \times \log_{10}\left(\left(\frac{\text{Ref Res}}{\text{DUT Imp} + \text{Ref Res}}\right) \times \text{Stimulus Level}\right)$$

Level Across Reference Resistor in dBV

The charts in ***Figure 13-59*** and ***Figure 13-60*** show how the **Drop at the DUT** and **Level across Ref Res** change, depending on the equation variables.

For example:

If you are testing an 8 Ω loudspeaker with a Stimulus of 3 Volt and a Ref Res of 0.25 Ohms:

- Signal Drop at DUT (due to added resistance) = -0.267 dB
- Level across Ref Res = -20.83 dBV or 91 mVolts

Speaker Imp (Ohms)	Ref Res (Ohms)	Stimulus (V)	Drop at DUT (dB)	Ref Res Level (dBV)	Ref Res Level (V)
2	0.25	3	-1.023	-9.54	0.333
4	0.25	3	-0.527	-15.07	0.176
8	0.25	3	-0.267	-20.83	0.091
16	0.25	3	-0.135	-26.72	0.046
32	0.25	3	-0.068	-32.67	0.023
64	0.25	3	-0.034	-38.66	0.012
150	0.25	3	-0.014	-46.04	0.005
250	0.25	3	-0.009	-50.47	0.003

Figure 13-59: Reference Resistor 0.25 Ohm

By changing the Ref Res to 1 Ohm the values are:

- Signal Drop at DUT = -1.023 dB
- Level across Ref Res = -9.54 dBV or 333 mVolts

Speaker Imp (Ohms)	Ref Res (Ohms)	Stimulus (V)	Drop at DUT (dB)	Ref Res Level (dBV)	Ref Res Level (V)
2	1	3	-3.522	0.00	1.000
4	1	3	-1.938	-4.44	0.600
8	1	3	-1.023	-9.54	0.333
16	1	3	-0.527	-15.07	0.176
32	1	3	-0.267	-20.83	0.091
64	1	3	-0.135	-26.72	0.046
150	1	3	-0.058	-34.04	0.020
250	1	3	-0.035	-38.45	0.012

Figure 13-60: Reference Resistor 1 Ohm

The 1 Ohm Ref Res provides a sufficient level to SoundCheck, but there is a large signal drop of -1.023 dB at the DUT.

The 0.25 Ohm Ref Res only presents a drop of -0.267 dB, but still has a signal level well above the noise floor of the audio interface: 91 mVolts. **The 0.25 Ohm Ref Res would be a better choice in this case.**

Note: As the DUT impedance goes up or down significantly, you will want to scale the Ref Res value accordingly.

Reference Information

For more information on Impedance please refer to:

Practical Impedance measurements using SoundCheck found on the Listen website.

Headphone Impedance Testing

The larger impedance of headphones will of course require the use of a larger reference resistor.

Important! The amplifier used to drive the headphones should have an output impedance of near zero Ohms. Headphone amplifiers, which tend to have higher output impedances (5 to 30 Ohms), should not be used.

If the DUT impedance is 150 Ohms with a stimulus of 0.5 V, the calculator chart shows that a Ref Res of 15 yields sufficient signal while presenting a reasonable drop at the DUT.

Speaker Imp (Ohms)	Ref Res (Ohms)	Stimulus (V)	Drop at DUT (dB)	Ref Res Level (dBV)	Ref Res Level (V)
2	15	0.5	-18.588	-7.11	0.441
4	15	0.5	-13.534	-8.07	0.395
8	15	0.5	-9.173	-9.73	0.326
16	15	0.5	-5.745	-12.33	0.242
32	15	0.5	-3.339	-15.94	0.160
64	15	0.5	-1.829	-20.45	0.095
150	15	0.5	-0.828	-26.85	0.045
250	15	0.5	-0.506	-30.96	0.028

Figure 13-61: Headphone Impedance vs Reference Resistor

Measuring Left and Right Headphone Impedance Simultaneously

Due to the common ground between left and right headphones, feedback loops can occur when measuring headphone impedance. In this case, you can use two Impedance Boxes connected to a stereo amplifier. Both Impedance Boxes must use the same resistor value. The boxes should be wired as follows:

Simply **reverse** the Red and Black connectors for both impedance boxes at both ends:

- Red Left and Red Right to **Amplifier Common**
- Black L and R to **Amplifier L and R**

  then

- Red Left and Red Right to **Headphone Common**
- Black L and R to **Headphone L and R**

This puts the load resistor in the positive side of the signal path.

The Impedance Analysis Step in the SoundCheck sequence must also be set as shown in *Figure 13-62*.

- Click on the Analysis Step **Electrical** tab and **Impedance Box - Before Speaker**. This method expects the load resistor to be in the **positive side of the signal path.**
- Under **Reference Resistor**, enter the value of the resistor in the impedance boxes.

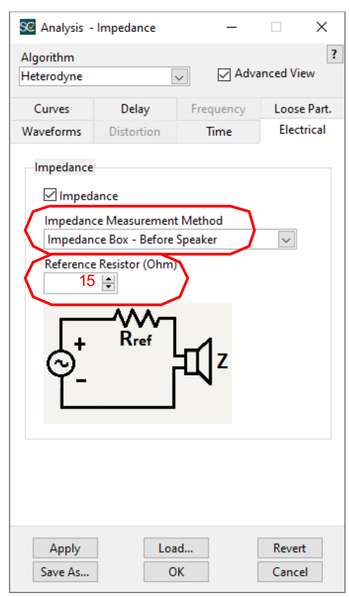


Figure 13-62: Impedance Box Before Speaker

Electrical Tab - DC

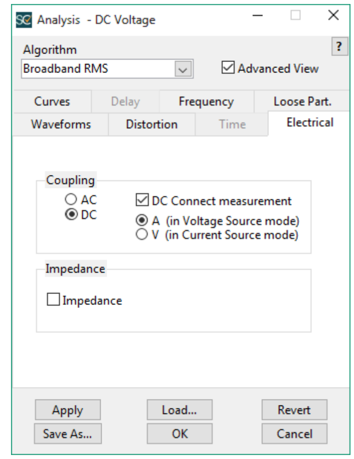


Figure 13-63: DC Coupling

If your audio interface or data acquisition card is DC coupled, select **DC** coupling to measure the DC voltage.

If the audio interface is not DC coupled, the DC current or voltage waveform can be read from a DC Connect™† instrument using your AC coupled, audio interface. An audio interface input is connected to the DC Connect Analog Monitor back-panel output. The Analysis Step is preceded by an Acquisition Step set to Record or Play & Record. Be sure to assign the Analysis and the Acquisition Step's Input Signal Path to the audio interface input channel that you've connected to the DC Connect™.

† DC Connect, made by Listen, Inc., is a USB-controlled DC voltage / current source and measuring amplifier. See the Listen website for more details. ***https://www.listeninc.com/products/***

When you select the **DC Connect measurement** check box, you can select:

- **A** (in Voltage Source mode) to measure current
- **V** (in Current Source mode) to measure voltage

The Analysis Step will create a DC Current Waveform or DC Voltage Waveform.

For example, when this type of Analysis Step follows a Play & Record Acquisition Step, the X-axis of the DC Current Waveform matches the X-axis of the stimulus signal. The stimulus will likely be an audio amplitude/ frequency sweep, or a sweep of the DC voltage.

If you also check the **Time Envelope** check box, you create a Time Envelope curve whose units are also in **A** or **V**. The x-axis of this curve is time.

For more information on this use of DC Connect, please refer to the DC Connect manual, Analog Control Example chapter.

Loose Particle Tab

During the manufacturing process of a loudspeaker, some loose particles of foreign material may stay trapped in the gap behind the diaphragm or dust cap. During operation at low frequencies, these particles randomly hit the diaphragm making a click or pop noise. This algorithm detects loose particles as impulses in the sound emitted by a loudspeaker during a measurement. Loose particle defects are easier to catch at low frequencies (typically at or below resonance) where maximum driver displacement occurs.

A Sine Sweep stimulus should be used for this type of measurement. For greater accuracy, a Stepped Sine Sweep (Stweep) should be used. For this reason, it performs best when used with HarmonicTrak, Heterodyne and Time Selective Response algorithms.

The Loose Particle algorithm in SoundCheck offers better noise immunity in production and other noisy environments as limits float with the normalized background noise rather than being set absolutely.

In addition to simplifying limit setting, false rejections due to sudden increases in background noise are less likely. There is also a setting for choosing a maximum stimulus frequency, above which the loose particle envelope is not calculated. As loose particles tend to present themselves during the low frequency portion of a stimulus sweep, this feature further prevents false rejects.

Loose Particle features:

- The loose particles algorithm is optimized for speed and offers a cleaner envelope, which makes it easier to set limits
- In Basic View the only available parameter to edit is Attack Threshold
- The Loose Particle tab is always available in the Analysis Editor, for all of the Analysis Algorithms

---

Note: While the Loose Particles function is available for all Analysis Algorithms, it is important to use a stimulus with a repetitive, steady state tone in order to avoid false particle detection.

The Loose Particles example sequence included with SoundCheck (Loudspeakers folder) shows the basic settings for use in a loudspeaker test. ***Figure 13-64*** shows the difference between the Loose Particle Envelopes of a good speaker and a bad one.

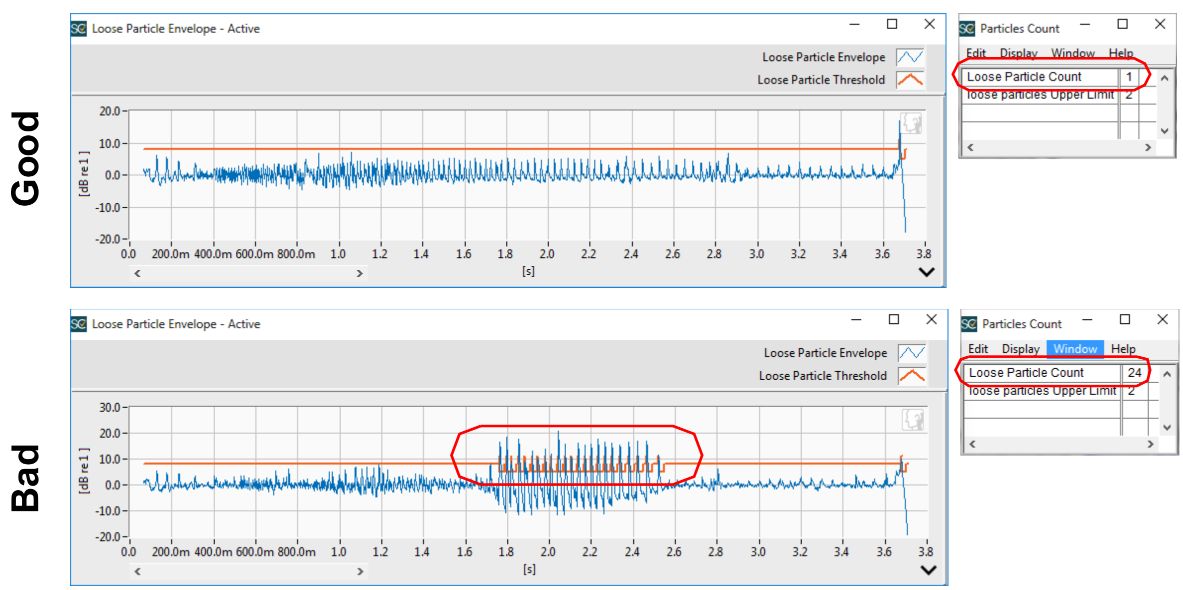


Figure 13-64: Good vs Bad Loose Particle Envelopes

In Advanced View the following parameters are available with the noted default values in ***Figure 13-65***:

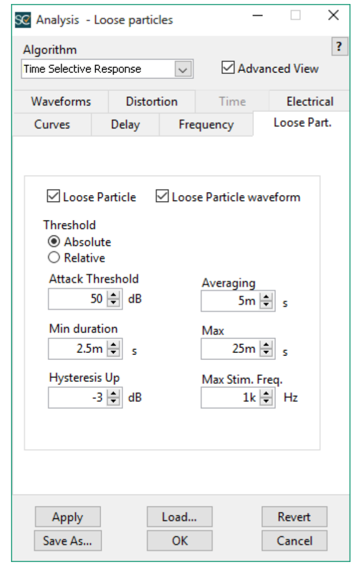


Figure 13-65: Loose Particle Detection Settings

- Loose Particle Waveform: Select to output a separate waveform to the Memory List that is the Recorded Time Waveform minus the Stimulus Waveform. This allow you to hear the sound of just the Loose Particles and see when they occur.
- Threshold: Select Absolute or Relative envelope

  Relative envelope has a steady state at 0. This makes the Threshold become relative to the steady state level as well. This allows you to utilize a standard detection threshold independent of the test level.

  Absolute envelope will reflect the Response level as before.
- Attack Threshold: 50 dB (default) See ***Attack Threshold & Hysteresis on page 211***
- Averaging Time: 5 ms (default)

  This is the width of the running rms averaging applied on the time signal to generate the time envelope used to detect loose particles.
- Minimum Duration: 2.5 ms (default) See ***Min/Max Duration on page 212***
- Max Duration: 25 ms (default)
- Hysteresis: -3 dB (default) See ***Attack Threshold & Hysteresis on page 211***
- Max Stimulus Frequency

  Essentially this is a Low Pass Filter. It allows you to choose a maximum stimulus frequency, above which the loose particle envelope is not calculated. As loose particles tend to present themselves during the low frequency portion of a stimulus sweep, this feature further prevents false rejects.

  A typical default value for max Stimulus frequency would be 1 kHz.

Attack Threshold & Hysteresis

All peaks that exceed the threshold are counted as loose particles. You must define this threshold and set it high enough to exclude random background noise. Establishing the threshold level will require some trial and error. Use a known good speaker (or Golden Speaker) to choose a level above the background noise. Then, measure a loudspeaker with known loose particle defects to verify that the threshold is exceeded.

To avoid false peak detection, the hysteresis level should be set to a value greater than the background noise.

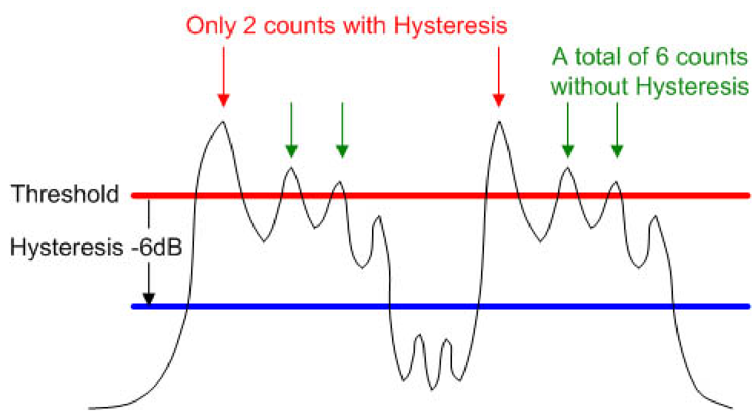


Figure 13-66: Threshold and Hysteresis

The example in ***Figure 13-66*** shows 6 points that cross the threshold. Hysteresis is set to -6 dB so that it ignores the peak over threshold until the signal is 6 dB below threshold. In this example, only 2 loose particles are counted.

Note: The Loose Particle Detection Algorithm gives you the total number of individual peaks that exceed the threshold (Particle Count). Sometimes you will get false peaks from transient background noise events such as a box dropping or an air gun going off. This can be limited by setting the Min/Max Duration levels. The hysteresis level can be used to ignore steady state background noise.

Min/Max Duration

Allows you to ignore transients which do not fit within the Max and Min limits.

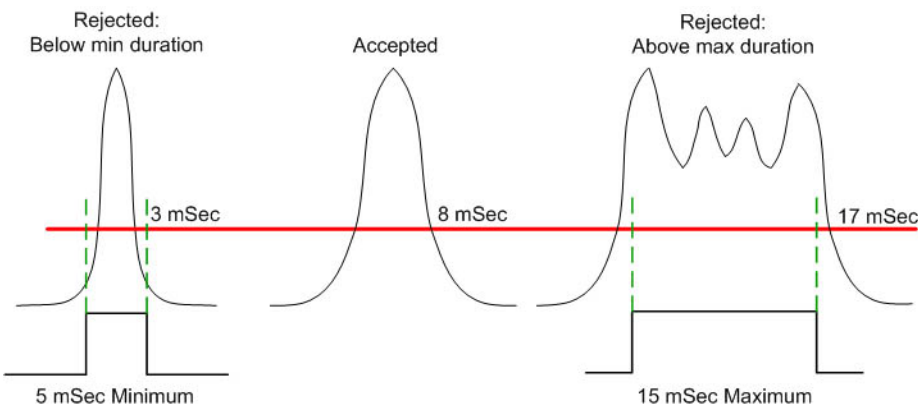
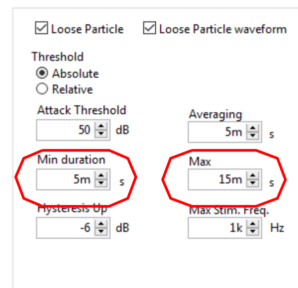


Figure 13-67: Min and Max Duration



The example in ***Figure 13-67*** shows three transients of different durations.

- Transient 1 is rejected since it falls below the 5 mSec minimum
- Transient 2 is accepted since it falls between the limits, 5 to 15 mSec
- Transient 3 is rejected since it is above the 15 mSec maximum

For more information regarding Loose Particle Analysis please refer to two AES papers presented by Listen, Inc.:

Enhancements for Loose Particle Detection in Loudspeakers **by Pascal Brunet & Steve Temme; Listen, Inc.**

Loose Particle Detection in Loudspeakers, found on the Listen website.

A Single Value Limit Step is used to obtain a Pass/Fail verdict for the DUT.

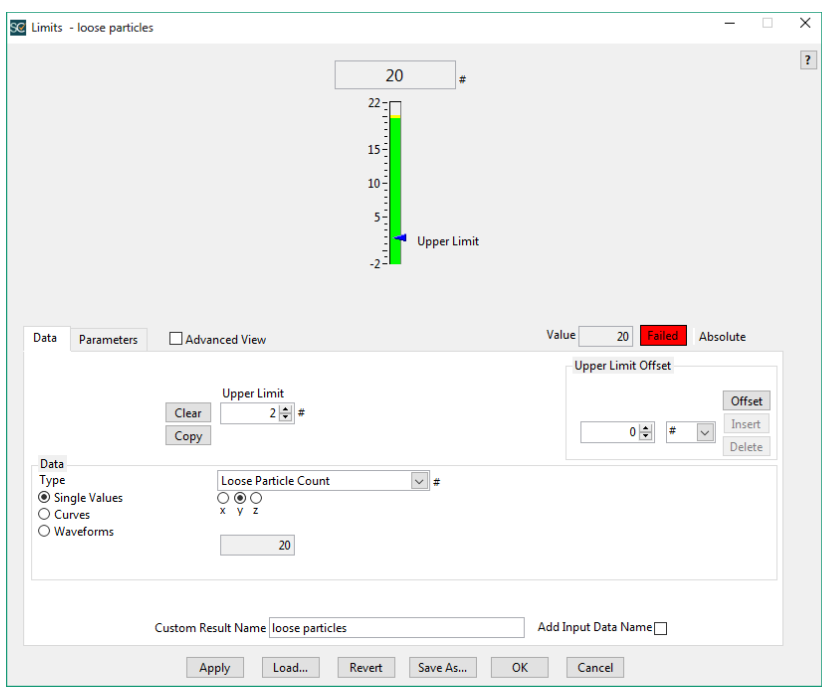


Figure 13-69: Loose Particle Count Limits

Autosave Editor

Information in the *Memory List* can be automatically saved to disk using an Autosave Step (**Ctrl+Shift+U**). To view and change the Autosave settings, select **Autosave** from the **Setup** drop-down list on the SoundCheck Main Screen. You can also create a new Autosave Step, or insert an Autosave Step into an existing sequence using the *Sequence Editor*.

The *Autosave Editor* is divided into four major sections – Save, Format, Test Information, and Filename.

Save

You can choose to save **only Data** (curves & values), **only Results**, or **only Waveforms**. Separate Autosave Steps must be used for each Memory List category. The files can be saved to the same folder.

> **Data** - (Curves and Values) Cannot be saved to WAV
>
> **Results** - Cannot be saved to WAV
>
> **Waveforms** - Cannot be saved to Database or Excel

Figure 14-1 shows three Autosave steps saving files to a folder. In this case, the files have different extensions (.DAT, .RES and .WFM), so the file names can be the same without the possibility of files being overwritten.

When saving to Excel, Data and Results can be appended to a single Excel file. Each data item will appear in its own worksheet.

See ***Excel Mode on page 364*** for another method of saving to Excel.

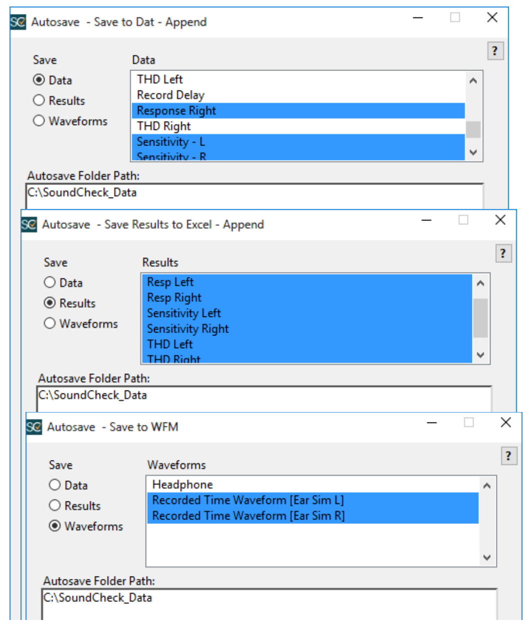


Figure 14-1: Saving Data, Results and Waveforms

Data and Results must be in separate TXT files

If you intend to import SoundCheck data into other applications that use .TXT files, Data and Results must be in separate files. You can do this by specifying different file names in the File Name field of each Autosave Step. This has very little effect on the speed of a measurement.

Autosave Folder Path

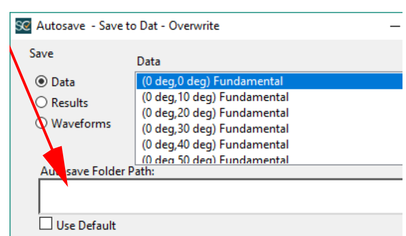
- Select **Use Default** to use the "Default Data Path" that is selected in the **Edit > Preferences > Folder Paths** options on the SoundCheck Main Screen. See ***Folder Paths on page 49***.
- You can specify the file path by unchecking **Use Default** and Browsing to file location
- When converting a sequence to SoundCheck 18.1, Autosave and Recall Steps will need to be updated if the file location on the new computer is not identical to the original computer.

Rules

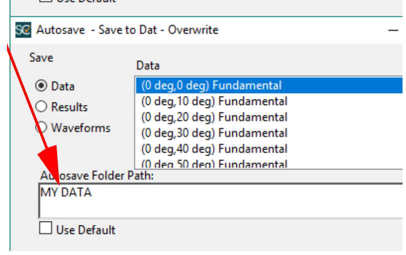
- The Autosave Step(s) must follow the sequence steps that create information to be saved to disk when the sequence runs
- When saving waveforms for reprocessing in SoundCheck they should be saved as WFM files, not WAV. Details of the stimulus (resolution, level, bandwidth, etc.) are included in the metadata of the WFM file.
- Separate Autosave Steps must be used for each Memory List category: Data (curves & values), Results and Waveforms
- **Excel** must be installed on the computer when saving to Excel. Cloud based installations are not recognized by SoundCheck.
- As of SoundCheck 14.01, Excel Macro-enabled files with the XLSM file extension are allowed. The XLSM file extension is used in the generated file.
- Separate Autosave Steps must be used to save Data and Results. When converting sequences from versions prior to SoundCheck 8, Autosave Steps will need to be updated if a single step is used for both Data and Results.
- When using Autosave and Recall steps with the same data set, the formatting of the steps must match. As of SoundCheck 8, the functionality of the Autosave and Recall Steps has been matched. This assures that data saved with an Autosave Step can easily be accessed by a Recall Step. Refer to ***Recall Editor on page 227*** for more information.
- It is not possible to save both Data and Results to the same text file
- When converting a sequence to SoundCheck 18.1, Autosave and Recall Steps will need to be updated if the file location on the new computer is not identical to the original computer
- Waveform and WAV files cannot be saved to Excel

Rules - Relative File Path Rules in Autosave

- This indicates that the location of the saved file is relative to the folder path of the sequence, e.g., an exported sequence folder. This is useful when sharing sequences with other SoundCheck users as it keeps the data in the same location as the sequence. The relative path can even include sub-folders.
- Delete any text in the File Path field and leave it blank. This indicates that the file will be saved in the same folder as the sequence file location.
- This also applies to the Template File location



- Sub-folders are indicated by just the name of the folder (no back slash): **My DATA**.
- If the sub-folder does not exist, SoundCheck will automatically create it.
- A 2nd level of sub-folder does require a back slash: **My DATA\Product 1**



Format

Select the type of information you want to save from the drop-down list. The file type selected defines the options that are available in the other sections of the editor. All options are linked to the format selected; many fields will toggle between active and inactive as the file type is changed.

Select from the drop-down list:

- **Database** - save to an SQL database
- **Excel** - Each data item is saved or appended to a separate worksheet of the Excel file, .XLS or .XLSX
- **MATLAB** - Data is saved as a standard MAT-file
- **SoundCheck** - Save as .DAT (curves & values), .RES (results) or .WFM (waveforms)
(Separate Autosave Steps are required for each)

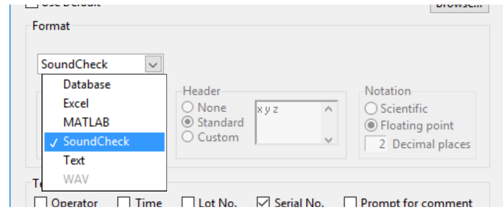


Figure 14-2: Select Format

- **Text** - .TXT delimited text file
- **WAV** - Any waveform in the Memory List can be saved as a WAV file. The WAV file will be saved with the sample rate and bit depth that are set in the System Hardware configuration.
See ***WAV File Types on page 336*** for more information on supported WAV file types.
- You must use a separate Autosave Step in the sequence for each **Format** required

Database

Save directly to database. You are required to enter a Universal Data Link (UDL) or Data Source Name (DSN) to gain access to a database either on the local machine or on a connected server.

As of SoundCheck 17, save to database has a redesigned schema featuring more efficient use of data types and the use of BLOBs (Binary Large OBjects).

- Fewer tables
- 3x More space efficient
- 4x Faster

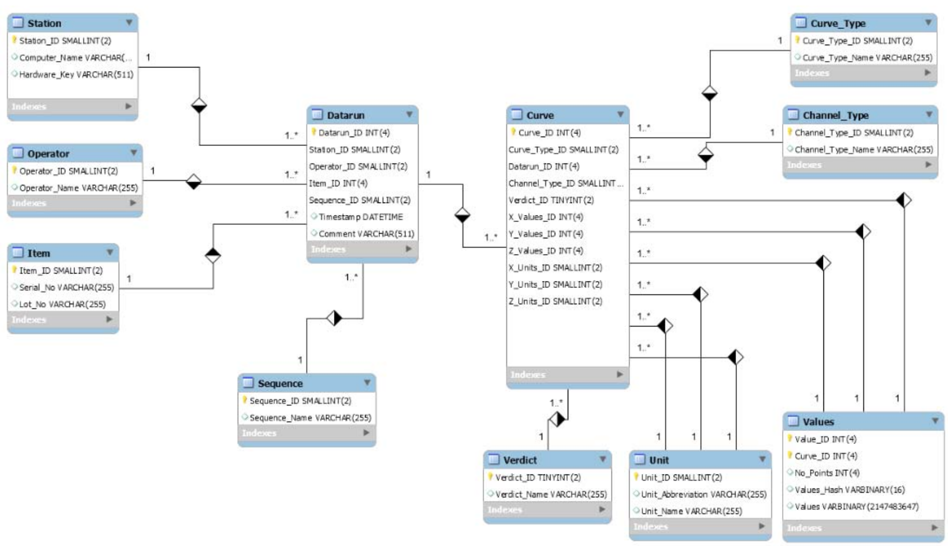


Figure 14-3: Database Schema

The example in ***Figure 14-4*** shows a table created by SoundCheck. You can find a more extensive description of the relational table structure in ***Relationship of Access Tables for SoundCheck on page 507***.

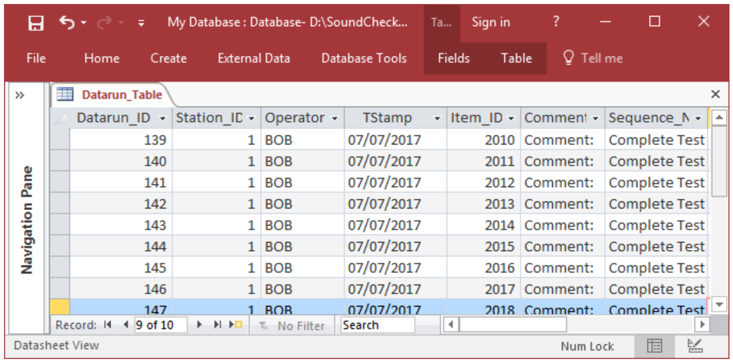


Figure 14-4: Database Table

- UDL or DSN - You must browse for a Universal Data Link (UDL) or Data Source Name (DSN) to access the target database. See ***Database Setup on page 533*** for more information on using DSNs and UDLs.

  You must select a pre-existing UDL or DSN file. Create a UDL or DSN with the help of your company database administrator or consult your database application manual to create one, using the correct provider for your database software. When you browse for this file, SoundCheck will verify the connection to your database and create the set of tables for your SoundCheck data.

Note: The fastest way to save directly to a database is to save single values or results, such as the Pass/Fail verdict of a Limit Step. The curves can be saved to a *.DAT file for subsequent analysis. By adding Serial Number to the curve in the DAT file, it will be easy to correlate the curve to the items saved in the database.

Excel

Saves selected curves, values and/or results to an Excel file. You can save to a new Excel file or append to a previously created file. Each selected data item is saved on a separate worksheet in an Excel workbook. Each worksheet will be named according to the *Memory List name*. The example in ***Figure 14-5*** will have worksheets named:

Response Right

THD Right

Sensitivity - L

Sensitivity - R

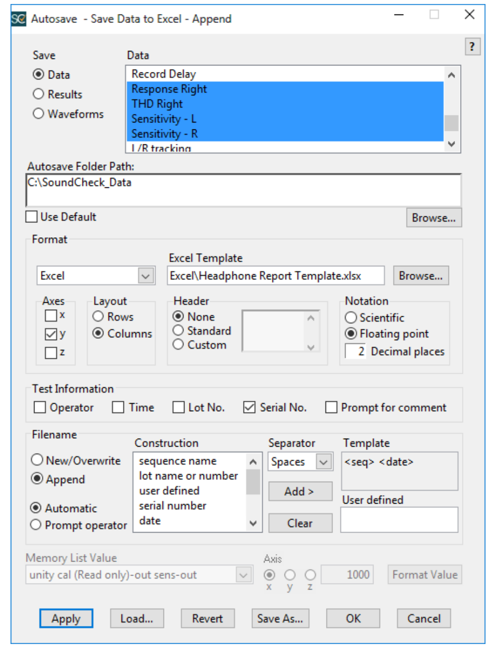


Figure 14-5: Autosave - to Excel

See ***Excel Template on page 218*** for a note on Excel templates.

See ***Excel Template Tutorial on page 601***.

Note: SoundCheck saves the x-axis values at least once when saving to Excel. If you want to append y and or z values to an Excel workbook, DO NOT check the **x-axis** box. The x-axis will be written in the first row or column, depending on the layout. Subsequent curves will then only contain y and/or z data, since the x values would be redundant.

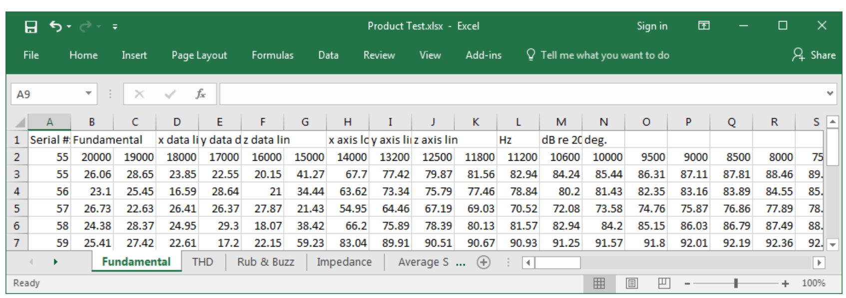


Figure 14-6: Save to Excel in Rows

Important! Excel files that are Targets of an Autosave Step should not be open when the sequence is run. Open files may prevent data from being saved. Do not attempt to close Excel manually when it is opened by an Autosave Step. Let SoundCheck close Excel in the run of the sequence. Office 97 and 2000 are no longer compatible with SoundCheck.

Note: Waveform and WAV files cannot be saved to Excel.

Excel Template

If Excel is selected from the File Type choices, you have the option of browsing to a pre-defined Excel template to arrange or analyze data. This template can utilize a master worksheet to collect data from other worksheets in the Excel workbook.

The master worksheet can then be used for presentation and graphing of the data in any format that can be utilized in Excel. Refer to ***Excel Template Tutorial on page 601*** for step by step instructions.

Note: If an error exists in a cell of an Excel template, SoundCheck cannot create a new Excel file for saving data. SoundCheck will open and close Excel but is not able to report an error. If this occurs, check the Excel template, repair the broken cells, save the template and run the SoundCheck test again.

Important! Excel files that are Targets of an Autosave Step should not be open when the sequence is run. Open files may prevent data from being saved. Do not attempt to close Excel manually when it is opened by an Autosave Step. Let SoundCheck close Excel in the run of the sequence. Office 97 and 2000 are no longer compatible with SoundCheck.

Note: As of SoundCheck 14.01, Excel Macro-enabled files with the XLSM file extension are allowed. The XLSM file extension is used in the generated file.

MATLAB

Curves and Values Data (*.DAT), Results (*.RES) and Waveforms (*.WFM) are saved to standard .MAT files.

Note that no other options can be selected under Format or Test Information. This information is not compatible with .MAT files.

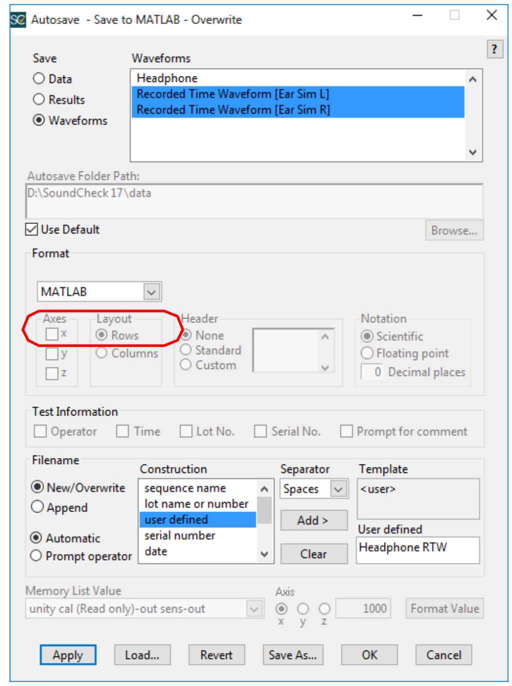


Figure 14-7: MATLAB

SoundCheck

Curves and Values Data (*. DAT), Results (*.RES) and Waveforms (*.WFM) are saved to binary files. The x, y, and z data are stored automatically in the DAT format. DAT files can be recalled for post processing and WFM files can be re-analyzed in SoundCheck.

Text Files

Save output to a text file. Output can be imported into other programs such as a filter design program. When **Text** is selected, you can then choose to store either frequency or time headers (x), amplitude (y), and/or phase data (z). A test saving five 3-D curves (x, y, and z axes) of 100 points each takes under 200 mSec, and creates a file size of 15KB. Data and Results must be saved using separate Autosave Steps.

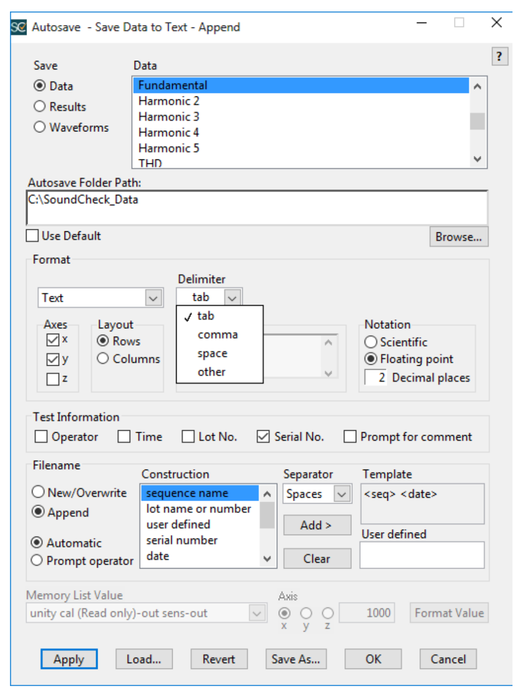


Figure 14-8: Text File

Rules for Text Files

- SoundCheck saves the x-axis values at least once when saving to a text file
- If you want to append y and or z values to a text file, **DO NOT check the x-axis box**. The x-axis will be written in the first row or column, depending on the layout. Subsequent curves will then only contain y and/or z data, since the x values would be redundant.
- If the X axis values change, the x axis will be saved, even if the x axis box is unchecked. If multiple curves are saved to one text file, the x axis will always appear, even if x axis is unchecked.
- See ***Delimiter on page 221***
- Multiple WFMs or WAVs should not be saved to one TXT file. Use separate TXT files for each.

WAV

SoundCheck will create a *.WAV file of the selected Waveform. When saving a waveform to WAV file, three **Scaling Options** are available:

- **Normalize to peak**: Saves the waveform so that the peak value of the WAV file is 100% Full Scale (FS), regardless of the level of the waveform
- **Sound Card Values**: The raw digital audio sample values recorded by the audio interface are saved. Absolute calibration in physical units, e.g. V or Pa, are lost.

  The WAV file is scaled according to the full scale deflection of the audio interface digital level.
- **User defined**: Scales the waveform relative to a user defined Maximum Level in physical units

  WAV File = Input Waveform (in physical units) / Maximum Level

  The resulting WAV cannot be scaled so that its peak value is above 100% FS.

  This makes it simple to return the measurement back to the physical unit. This can be used to export data for customized mathematical analysis using other tools such as MATLAB™.

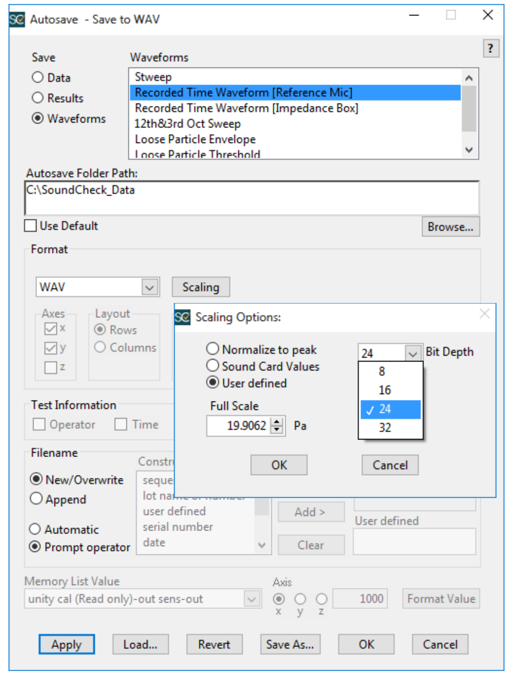


Figure 14-9: WAV File Scaling

Axes

The Axes options are not available for some formats and are grayed out.

Choose one or more axes to be saved. To store only the magnitude data (e.g., decibel values), check just the Y-axis box. Selecting axes only applies to Text and Excel file types. For Dat, Res, and Database; x, y, and z-axes are automatically stored.

Header

- **None** – For a *.TXT file, the first row or column will contain the frequencies. The Data is in subsequent rows or columns based on the Display being used.
- **Standard** – Header information related to the data and/or results will be the first row or column. (Curve name, axis units, freq. points, etc.)

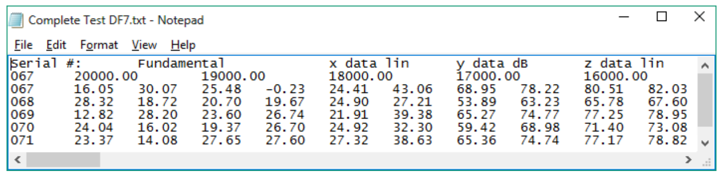


Figure 14-10: Text File With Standard Header

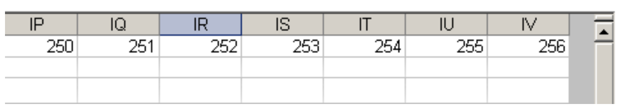
- **Custom** – Allows user to define header for compatibility with other programs and personal preferences. When using this option, you can choose among tab, comma, space, or other as the delimiter.

  For tab, use `\t` between the custom header fields (e.g., `Header1\tHeader2\tHeader3`).

Layout

- **Rows** – Aligns data in rows with headers above each column.
- **Columns** – Aligns data in columns with headers along each row.

Important! Excel .XLS files are limited to 256 Columns. Rows are unlimited. In Excel 2007 and later the .XLSX file maximum worksheet size is 1048576 rows by 16384 columns.



Delimiter

- When saving a *.TXT file, you can choose to separate data values using commas, tabs, spaces, or a user defined character
- Space Delimited can only be used with files that have NO HEADER INFO. If there are spaces in the header info, the Autosave Step will reject the file.

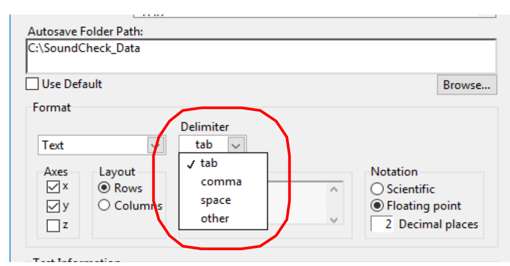


Figure 14-11: TXT Delimiter

Notation

- **Scientific** – Scientific notation is used. (1.03E+2)
- **Floating Point** – Floating Point notation is used. (102.86)
- **Decimal places** – Enter the desired precision for your data.

Test Information

These settings will need to be configured after you insert an Autosave Step into a new sequence. When Database is selected as your file type, Operator, Time Stamp, Lot No, and Serial No will automatically be saved to the database. See ***Database Setup on page 533*** for more information on what is stored in your database. All other file types allow you to save only the selected test information.

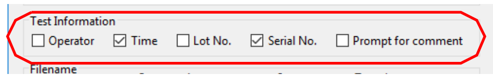


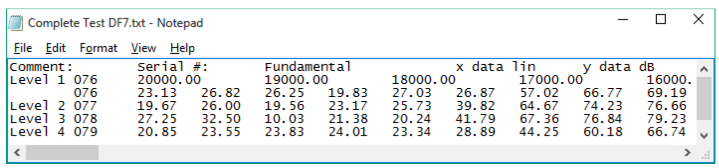
Figure 14-12: Test Information

- **Operator** – Keeps the Operator name (login name) with the curves or results being saved.
- **Time** – Attaches a time and date stamp to the information being saved (up to one second resolution).
- **Lot No.** – The lot number entered on the SoundCheck Main Screen is recorded with the data.
- **Serial No.** – The serial number entered on the SoundCheck Main Screen is recorded with the data. To have a unique serial number assigned to each row or column of data, choose the **SN auto increment** step in the **Serial No** sequence step category.
- **Prompt for comment** – After the test has run, the operator can enter a text note. This appears as a separate field on the same line as the test data.

Note: As of SoundCheck 15, the clipboard is cleared after each Autosave Step. You will not be able to **Copy** the text from one Comment field and **Paste** it into another Autosave prompt.

Figure 14-13 shows an example of a Text file with the lot number and serial number added. Other Test information would be added to the left of the serial number, and would also follow the row or column format selected by the user.

If a *.DAT, *.RES or *.WFM file is chosen, all Test Information will be appended to the curve or result name.


Complete Test DF7.txt - Notepad

File Edit Format View Help

```
Comment:        Serial #:       Fundamental             x data lin      y data dB
Level 1 076     20000.00        19000.00        18000.00        17000.00        16000.
        076     23.13   26.82   26.25   19.83   27.03   26.87   57.02   66.77   69.19
Level 2 077     19.67   26.00   19.56   23.17   25.73   39.82   64.67   74.23   76.66
Level 3 078     27.25   32.50   10.03   21.38   20.24   41.79   67.36   76.84   79.23
Level 4 079     20.85   23.55   23.83   24.01   23.34   28.89   44.25   60.18   66.74
```

Figure 14-13: Text File With Comment

Filename

The Autosave Step saves the selected curves, values or results to a file whose type is pre-selected. The name of the file being saved can also be determined in the *Autosave Editor*.

- **New** – Every time the Autosave Step runs, it will create a new file and overwrite an existing file of the same name without prompting you. The first time SoundCheck is asked to copy over an existing file, it will ask if the file should be replaced. Select **Always Replace without Prompting** to disable this message in the future.
    - You can change this setting back to enable prompting by exiting SoundCheck and opening ***SoundCheck 18.ini*** from the SoundCheck directory. Find the entry **PROMPT TO OVERWRITE FILE=False** and set it to **PROMPT TO OVERWRITE FILE=True**. Save the ***SoundCheck 18.ini*** file and open SoundCheck.)
- **Append** – If a file of the same name exists in the same folder, the Autosave Step will append the new data to it. If a file of the same name does not exist in the folder, Autosave will create a new file. It the name template includes the date, the file will only be appended if the current date is the same as the date in the file name. If not, a new file will be created.
- **Automatic** – SoundCheck automatically stores the file to the specified location using the constructed filename template. This option can also be used to append multiple tests to the same file (e.g., as a table).
    - The example in ***Figure 14-14***, **Option** is set to **Automatic**
    - From the **Construction** drop-down list select "Sequence Name" and click **Add**. Select "Lot Number" and click **Add**.
    - To erase the Template name, click **Clear**
    - To make a User Defined name (e.g., Prototype or Pilot Run), click **User Defined** from the **Construction** drop-down list. Click **Add**. Any text in the **User Defined** field will be added to the filename.
- **Prompt Operator** – SoundCheck will prompt the Operator to enter a filename (not including the filename extension)

Construction & Template

Choose the item(s) to add to the filename template. In the following example:

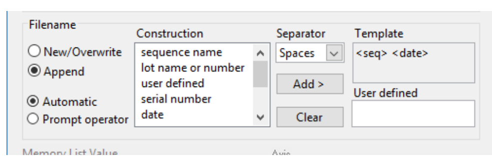


Figure 14-14: Filename Construction

- The sequence name is "***Autosave***"
- The lot number entered on the SoundCheck Main Screen is "***Demo 99"***
- The template adds these two together to form the filename: "***Autosave Demo 99.txt***"

Choose from the options listed in the **Construction** list box to build a filename for your data.

- **Sequence name** – uses current sequence name for filename. The sequence name is typically the model number of the product being tested.
- **Lot name or number** – uses current lot name for filename.

- **User defined** – text entered in the User Defined field is added to the filename, e.g., DUT model name or model number. Entries in this box will only be applied to the filename when **<user>** is added to the **Template** field. Invalid characters are shown in ***Figure 14-15***.
- **Serial number** – uses current serial number for filename.

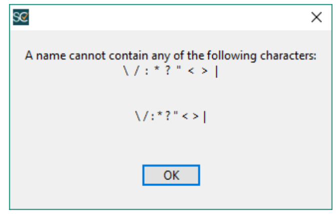


Figure 14-15: Character Type Error

> ***Note:*** If the test sequence automatically increments the serial number (SN auto increment step), a separate file will be created for each measurement. To store all the measurements in one file, do not use this text string (<sn>) as part of the file name.

- **Date** – uses current date for filename.
- **Time** – uses current time for filename. This has one-second resolution and will generate a unique filename each time the sequence is run. For this reason it is not appropriate to be used with the "Append" option.

> ***Note:*** If you want to append a file and include date/time stamp information, check the **Time** check box in the **Test Information** section of the *Autosave Editor*.

- **Data** or **Results** – append "*data*" or "*results*" to the filename to distinguish data files from results files.
- **Memory List Selection** – a value from the Memory List can be used as part of the filename. See ***Memory List Value Example on page 226*** for an example. This could be the Loop Index of a Step Configuration, e.g., the Degrees that a turntable turns for each increment of a polar plot measurement. The Loop Index field is always a Y axis value. See ***Step Configuration on page 297*** and ***Index (Loop Index) on page 447*** for more information.
- **User Name** – the User Name from the SoundCheck Login can be added to the file name.

Separator

Both the Autosave and Recall Steps feature a **Separator** to add to the file name template. This insures that the filename saved in the Autosave Step can be accessed in a Recall Step. The selected separator will be used between every item added to the template.

The following options are available:

- Spaces - a single space will be added between each item in the template.
- Underscores - a single underscore will be added between each item in the template.
- None - no space or underscore is added between items in the template.

Apply Button

The Apply Button allows you to test the action of the Autosave Step without having to run the sequence.

In a Sequence

Using the Autosave Step with the *Sequence Editor*, the Autosave Step can be inserted into any existing test sequence. It should be inserted after any analysis or post processing. In this example, it has been placed immediately before the Display Step.

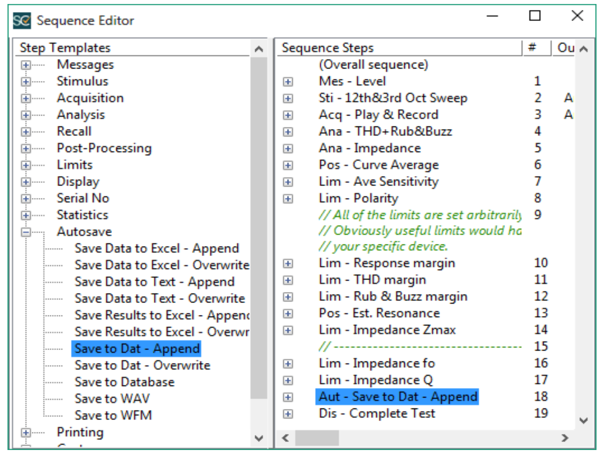


Figure 14-16: Autosave Step in Sequence

With the variety of file types available to save user data, keep in mind that one Autosave Step is needed for each type of file you wish to create or append. If you wish to save curves to a *.TXT file and results to a *.RES file, two steps must be inserted in the sequence to accomplish this. Similarly, you may wish to use three Autosave Steps to save the Time Response in a *.WAV file, the curves to an *.XLS file and then record all the results information to database.

The default Autosave sequence that is delivered with SoundCheck uses a Serial Number Step to automatically increment the serial number before proceeding to the Autosave Step. (The sequence can be found in the “How To examples“ folder.) The functions of the *Sequence Editor* (such as Step Configuration of the Limit Steps) could be adjusted to jump over the Serial Number and Autosave Steps if the device under test failed the Limit Step conditions. In a sequence without jumps, data for all items tested would be saved, regardless whether the DUT passes or fails.

Note: Autosave Steps from SoundCheck 4.13 and earlier may need to be revised. Previous versions allowed you to choose two file types in a single Autosave Step. If you copy a 4.13 or earlier sequence into the new folders, the data file type (*.DAT) will be used in the sequence and the results file type (.RES) will be ignored.

Memory List Value Example

The following example uses the “**Polar Plot (Turntable)**” example sequence that is included with SoundCheck.

Figure 14-17 shows the configuration of the first Rotate Speaker Message Step.

- It is set to create a value in the Memory List named “Angle“.
- The starting value of “Angle” is 0 degrees.
- Each time the step runs the value “Angle” is incremented 10 degrees.
- After 18 repetitions, (180 degrees) the step instructs the sequence to jump to the second Rotate Speaker Message.

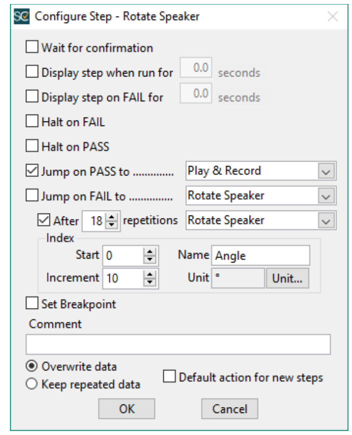


Figure 14-17: Loop Index

If you add an Autosave Step to the sequence that saves the fundamental curve at each angle, the Filename of the step can be set as shown in ***Figure 14-18*** through ***Figure 14-20***.

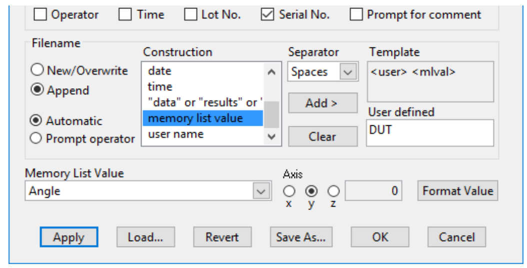


Figure 14-18: Memory List Value

- The **User Defined** name entered is “DUT“.
- The **Separator** is set to Spaces.
- **Memory list value** is added to the Template. This opens the Memory List Value Field.
- **Angle** is chosen in the Memory List Value field.
- The Axis of the value is set to “Y”.
- The **Format Value** window is shown in ***Figure 14-18***. It is important to note that **Hide Trailing Zeros** is checked.
- The .DAT files are saved starting at “DUT 0.dat“ as shown in ***Figure 14-20***.

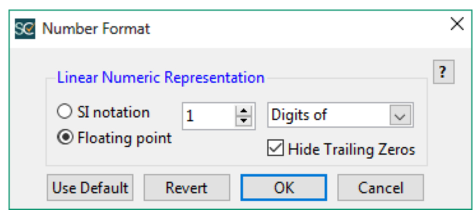


Figure 14-19: Number Format

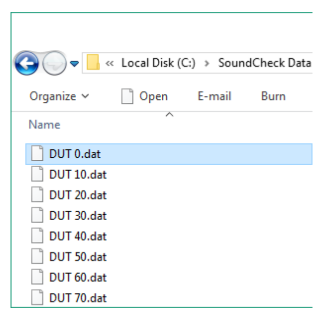


Figure 14-20: Output Example

Recall Editor

The *Recall Editor* (**Ctrl+Shift+R**) allows you to open previously saved data into your current sequence for post-processing or display. Any SoundCheck® data or results file (marked by *.DAT and *.RES file extensions) can be accessed and entered into the current *Memory List*. Waveform (WFM) files saved with SoundCheck can be recalled as well.

The Recall Step uses the same controls as the Autosave Step:

- Specify File Path - A specific file can be picked from any available folder location.
- Prompt Operator - You can choose the file when prompted during the sequence run.
- Automatic - Allows the sequence to recall saved data with the same rules or criteria used in the Autosave Step which saved the data. This can be useful when running Statistics, e.g., Recall all curves with the model name; “xyz“.
- File selection can be limited to “Last Only” or any specific curve from the list.
- Automatic File Addressing using Index values from the Memory List. The Index number generated by the Configuration of a Step can be used in the File Name Template for Dynamic file naming.
- The Apply Button is used to test the validity of a file path/address, e.g., Test recalling files from a network folder.

Note: DAT files created with SoundCheck 18.1 are not viewable in versions of SoundCheck prior to and including SoundCheck 6.0x. The DAT file format was updated in SoundCheck 6.1.

Note: Recalled data can be added to a Custom Group. See ***Sorting and Grouping on page 331***.

File Path

- When a sequence is exported, dependent files are exported to the selected folder along with the .SQC file, i.e.; DAT, RES, WFM, CSV and TXT files that are the object of the Recall Step.
- When converting a sequence to SoundCheck 18.1, Autosave and Recall Steps will need to be updated if the file location on the new computer is not identical to the original computer.

Specify File Path

The default behavior for the Recall Step is Specify File Path. The step will operate in the same manner as in previous versions of SoundCheck. (***Figure 15-1:*** shows that individual curves can be selected in the list.)

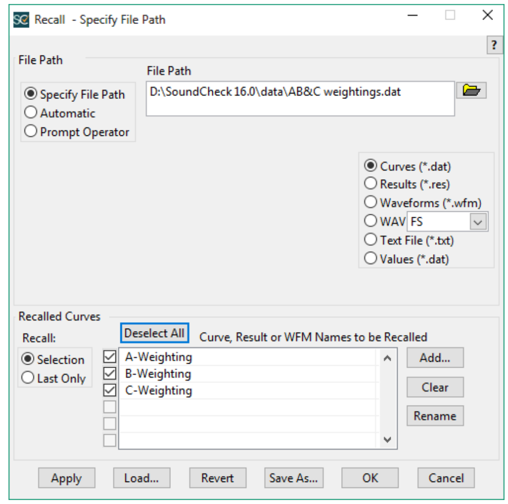


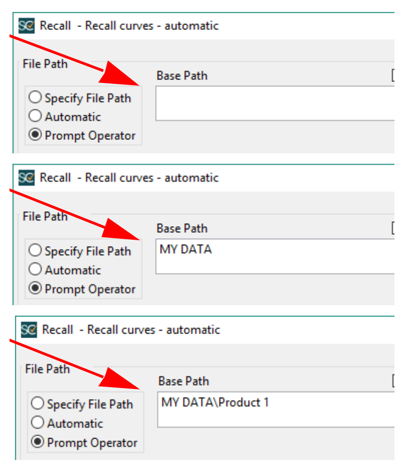
Figure 15-1: Specify File Path

Exact File Path

By indicating the exact path in the File Path field, the file will be recalled from that location, even if the sequence is saved to a new location. Select the **Folder Browse** button to the right of the file path window to select the proper folder.

Rules - Relative File Path Rules in Recall Editor

- This indicates that the location of the file selected is relative to the folder path of the sequence, e.g., from an exported sequence folder. This is useful when sharing sequences with other SoundCheck users as it keeps the data in the same location as the sequence. The relative path can even include sub-folders.
- This is for **Automatic** and **Prompt Operator** selections only.
- Delete any text in the File Path field and leave it blank. This indicates that the file to be recalled is in the same folder as the sequence file location.
- Sub-folders are indicated by just the name of the folder (no back slash): **My DATA**.
- If the sub-folder does not exist, SoundCheck will automatically create it.
- A 2nd level of sub-folder does require a back slash: **My DATA\Product 1**



Automatic

With Automatic mode, only the Base Path for the file is specified. The **Base Path** is the location the step will open files from. The file name is created using the controls for the Template field.

In the **Construction List**, the option(s) for the file name are selected and added to the **Template**. This forms the full filename, which is to be recalled by the step.

Figure 15-2: shows a **User Defined** name in the Template. The Serial Number entered is “1”. When the step runs, it will look for any file, in the Base Path, with the name “DUT 1.dat“.

The standard Autosave options for Filename Construction apply to the Recall Step. Please refer to ***Filename on page 223*** for a description of each of the Filename Construction options.

If using a Memory List Value to recall an integer as part of the User Defined name, you must click on **Format Value** and set the following: **Floating Point**, 0 (zero) and **Digits of Precision**.

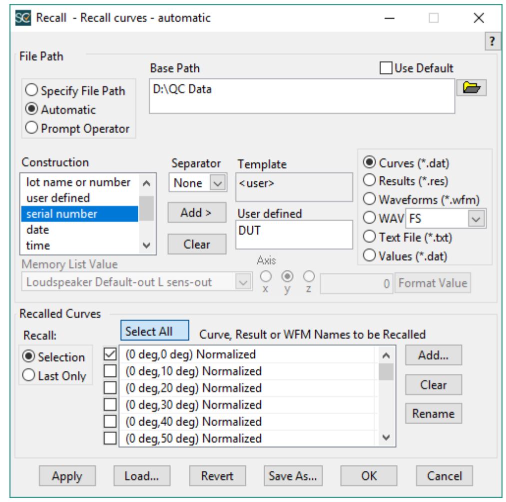


Figure 15-2: Automatic

Separator

The separator used in the Recall Step must match what is used in the Autosave Step.

The **Memory List Value** option has the same function as in Autosave, so that users can select a Memory List value (such as loop index) to be added to the Recall file name. See ***Memory List Value Example on page 226*** for an example of this type of construction.

Prompt Operator

With a Prompt Operator step, a “Select File” window will open when the step runs in the sequence. The operator can then select any file available for recall. **Base Path** allows for a default directory to be specified. This directs the operator to a specific folder for file selection.

Under **Recalled Curves**, a custom name has been entered. When “Pro Mic 2.dat” is opened, the curve will be added to the Memory List as “Fundamental“.

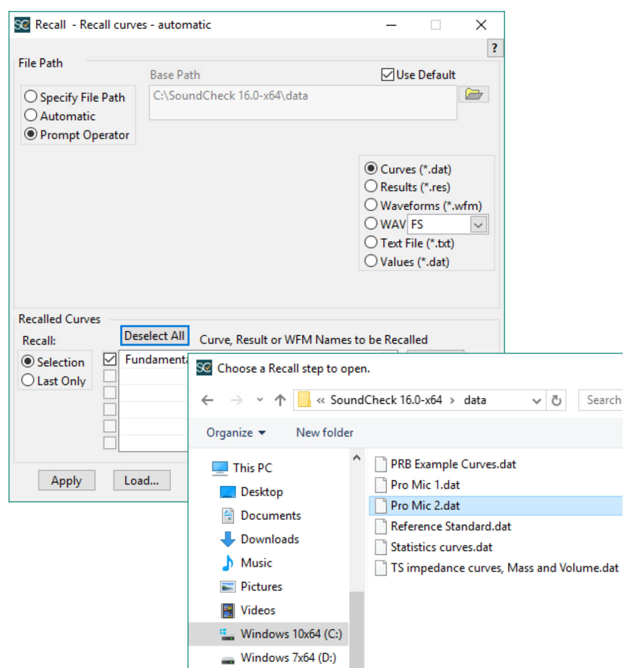


Figure 15-3: Prompt Operator

File Types

The following files types can be recalled:

Curves (.DAT), Results (.RES), Waveforms (.WFM and .WAV), Text (.TXT and .CSV) and Values (.DAT).

As of SoundCheck 17, CSV files can be recalled in addition to TXT files as shown in ***Figure 15-5:***. A separate IMP file for the CSV format can be created allowing for automatic recall of CSV files.

WAV files must match the sample rate of the System Hardware configuration in order to be used.

See ***WAV File Types on page 336*** for more information on supported WAV types.

WAV Recall

Recalled WAV files can now be scaled so that subsequent analysis will provide results in FS and dBFS as defined by AES17.

- When recalling a WAV file, select units of FS or FS(AES17) from the drop-down list as shown in ***Figure 15-4:***
- When opening a stereo WAV file, SoundCheck will automatically split the file into two waveforms, adding [L] or [R] to the file names

WAV File Scaling Options:

- 'FS' – SoundCheck default value, the max amplitude of a digital sine wave is -3 dBFS
- 'FS (AES17)' – Value corresponding to AES17 standard definition, the max amplitude of a digital sine wave is 0 dBFS

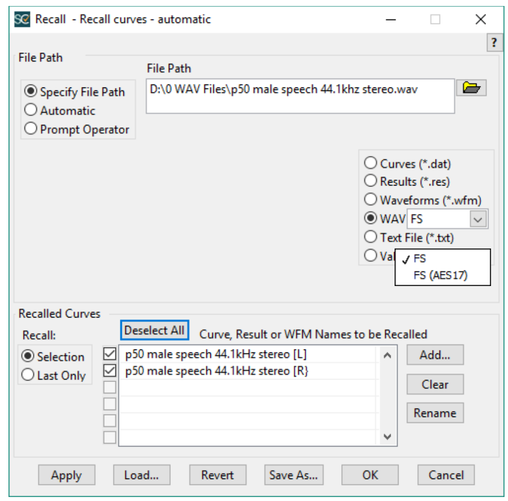


Figure 15-4: WAV Recall

Data Import Wizard

As of SoundCheck 14, the first time you recall text in the step, the Data Import Wizard runs. Once the text import settings are correct, you can create an .IMP file which saves the settings used in the Data Import Wizard. When the Recall Step runs in a sequence the .IMP file allows the step to run with no operator action required.

The Data Import Wizard also runs when importing text files in the Memory List. For more information refer to ***Data Import Wizard Tutorial on page 611***.

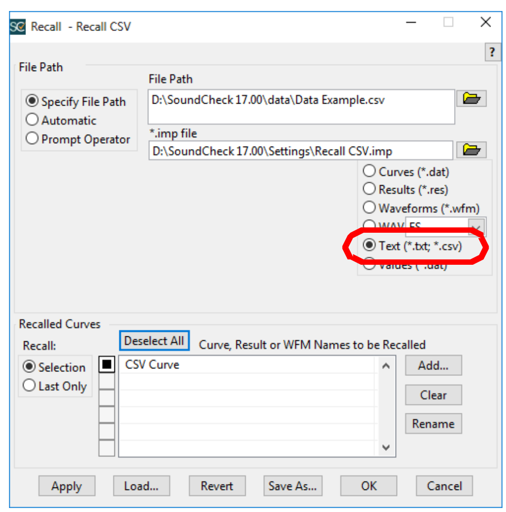


Figure 15-5: Import Wizard

File Path Info

Browse to the .DAT or .RES file you wish to reference. The file path and file name will appear in the **File Path** field.

Curve Names to be Recalled

After selecting the file, the *Recall Editor* will display the names of the curves, results or waveforms saved to that file in the **Curve or Result Names to be Recalled** text box. These names will be inserted into the *Memory list* as placeholders for data that will be created during the sequence run. These names will include any Test Information that was saved with the file. See ***Autosave Editoron page 213*** for more information.

The **Add…** and **Clear** buttons on the right hand side of the editor allow you to edit your selected list of data. The **Add…** button will allow you to add a name (an empty curve, single value or result) to the **Curve Names to be Recalled** text box. You may wish to do this when you know the data will be created later in the sequence. **Clear** will empty the text box of all information, removing placeholders from the *Memory List* until the step is executed in the sequence.

Order of Data

The recall data list is only generated the first time you point to the file. If the order of the data changes after that it will be populated into the wrong names in the memory list.

This can occur when the order of Autosave Steps changes, putting the data into the .DAT file in the wrong order. It can also happen when you manually save the data from the Memory List.

Pointing directly to the file again refreshes the list to fix the issue.

Recall in a Sequence

The **Recall** selection is the only information specific to the step.

You can choose to enter **All Curves** into the current sequence, or use the **Last Curve Only** choice to recall only the most recent curve or result saved to the file.

Note: When a *SoundCheck 18.1* sequence that contains a Recall Step(s) is exported, a copy of the recalled file(s) is also exported to the *Exported Sequence Folder*. The file path to this data will change once the sequence is exported (to point to the file when it is in the exported folder that the "*.SQC" file is in).

An example of the use of the *Recall Editor* can be found in the ***Limits in Reference to Standard*** sequence located in the ***How To examples*** folder.

In the example sequence, the curve of a reference standard “golden” loudspeaker is stored and then recalled by the sequence in order to compare it to the response of the speaker under test.

- The Recall step must be placed before the Post Processing step that will be using the recalled data.
- An example of the Post Processing step can be found in **Figure 15-8: Curve used in Post Processing**.

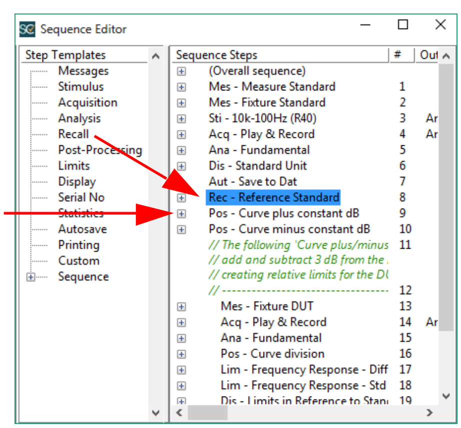


Figure 15-6: Recall Step in Sequence

The *Reference Standard.dat* file is located by browsing to the *Data* folder. The proper curve is then selected. Note that in this example **Last Curve Only** has been selected to use the latest Reference Standard calibration. This limits the list to only one item.

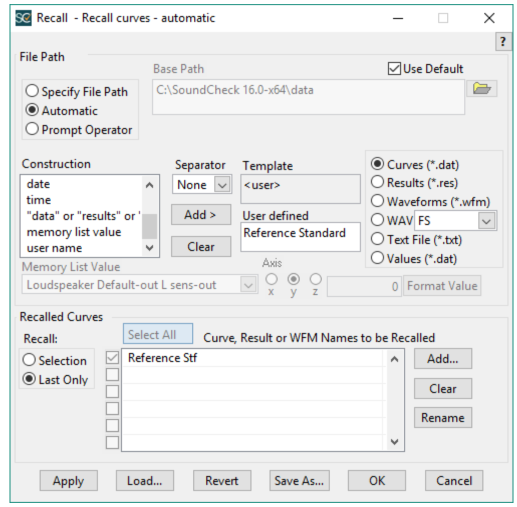


Figure 15-7: Last Only Selected

This curve can now be used in the Post Processing Step to perform a curve division calculation. ***Figure 15-8:*** shows the recalled curve selected as Operand B in the Post Processing step.

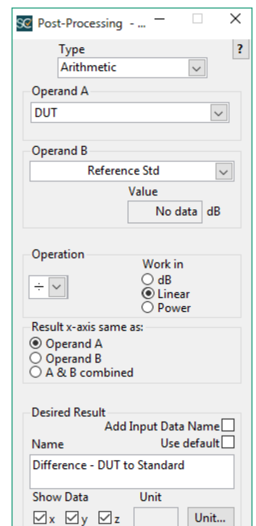


Figure 15-8: Curve used in Post Processing

Rename

By clicking on Rename, the name of any curve or result can be given a custom name. The example in ***Figure 15-9*** shows that the name has been changed from “Reference Std” to “Golden Unit“.

By selecting New Curve, the original item will remain in the list and a copy of the item will be added, using the new name.

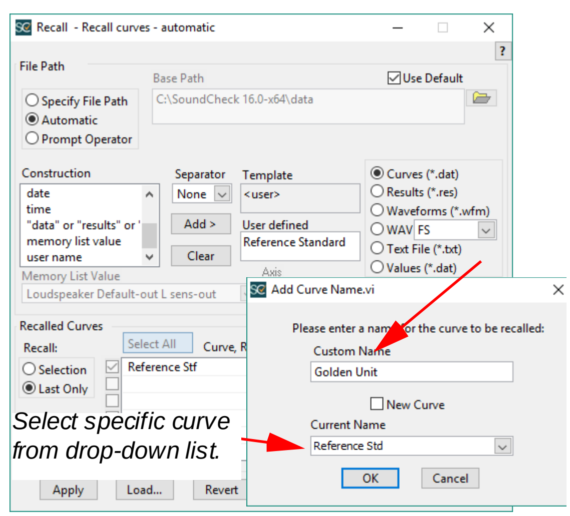


Figure 15-9: Custom Curve Name

Clear All

By clicking on Clear All, all the curves in the list are unchecked. This can be used to de-select a large number of curves and then select the desired curve(s).

Add

This allows you to add curve names to the list, as replacement names for existing items to be recalled. This is useful for changing a long list of curve names; e.g., Polar Plot Curves.

Clear

This will clear the contents of the Names to be Recalled. You can then add custom names using the Add button. It is important to know the number of curves in the list and the order of the curves, so that a corresponding number of new names are added.

Renaming Example

The example in ***Figure 15-10:*** shows 20 curves are present in the Polar.Dat file. After clicking the **Clear** button, 20 new names can be added to the list. These new names will replace the old curve names in the Memory List.

1. The Polar.dat file contains named from "Fundamental" to "19-Fundamental"
2. Click Clear to empty the list
3. Click the Add button to enter new curve names
4. Each curve must be entered one at a time

Note that Polar Plot curves must be entered in the Memory List in the proper order: 0 Degrees to 180 Degrees or 0 Degrees to 360 Degrees.

This puts "**place markers**" in the Memory List for each of the curves, without actually having to recall the Polar.dat file. When the sequence runs, the new step names will be used.

This can be used to open polar plot.DAT files and rename the curves with more descriptive names.

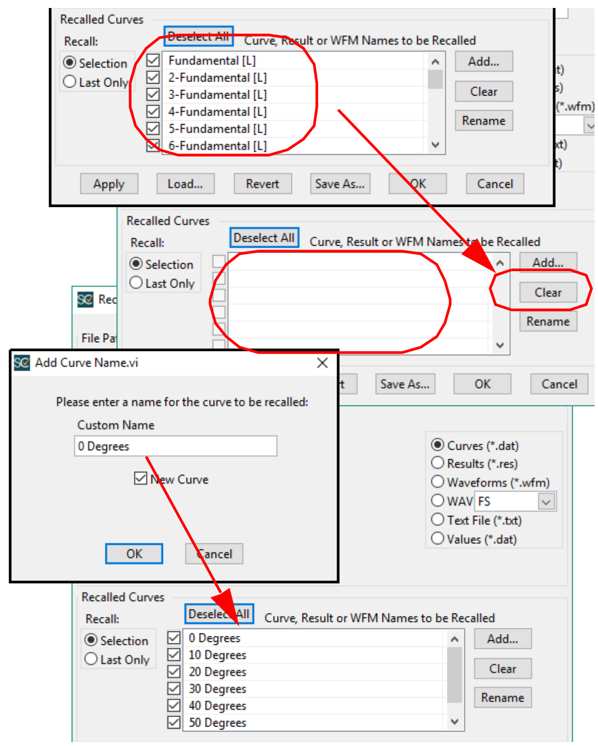


Figure 15-10: Renaming Example

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Post-Processing Editor

Note: All spectrum are summable. As of SoundCheck 18, you can also calculate the Power Sum of a Waveform.

To view and change the system's post-processing settings; select **Post-Processing** from the **Setup** drop-down list on the SoundCheck Main Screen, or use the shortcut **Ctrl+Shift+O**. Post-Processing allows a variety of operations to be applied to measurement data, including additional calculations, smoothing, and statistics.

The ***Post Processing Use Chart on page 269*** provides an overview of the Post Processing functions available in SoundCheck along with examples of use. Post Processing and the Equation Editor are optional modules and may not be available depending on the modules enabled on your Hardware Key. They can always be purchased from Listen and added to a Hardware Key license if needed.

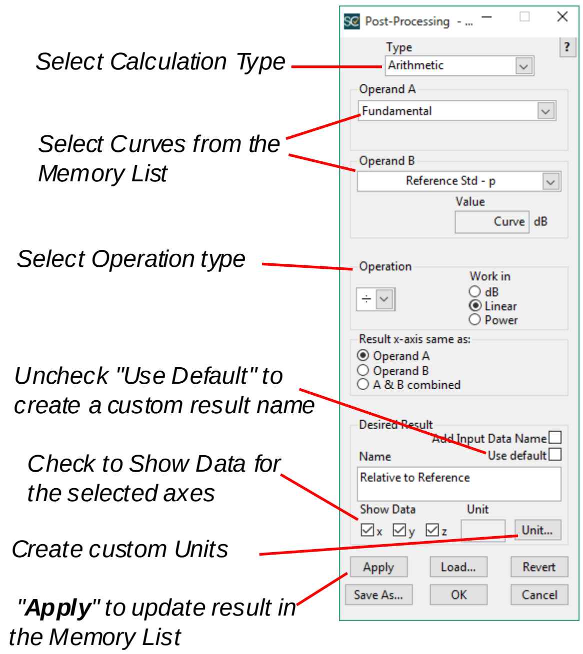

Figure 16-1: Post-processing Editor

Search Range

Many post processing operations in SoundCheck have a search range, which allows the user to select discrete points or ranges along the x-axis over which to perform the desired calculation. Examples include scalar functions such as average, max and min, as well as windowing intersection, and more.

As of SoundCheck 14, the search range function uses a very simple table control and allows memory list values to be selected. This means that search range parameters can be variables that are dynamically calculated by the sequence.

If Search Range is unchecked, this indicates it is set to "All" points in the selected data.

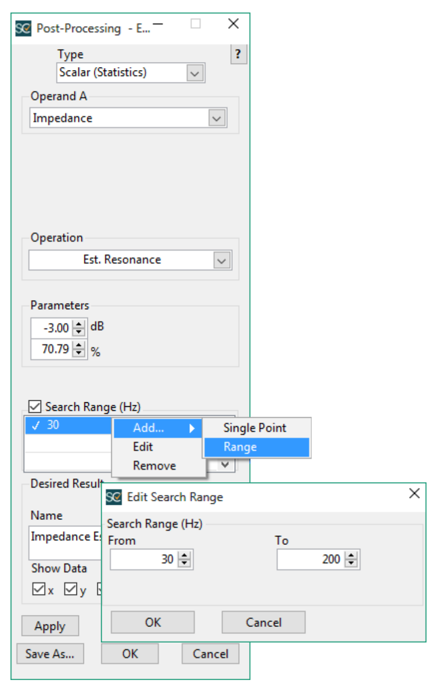

Figure 16-2: Search Range

Batch Processing

This allows you to select a group of items from the Memory List to Post Process.

Any Custom Group created in the Memory List can be used for the appropriate Post-processing type. Please refer to ***Sorting and Grouping on page 331*** for instructions on creating a Custom Group.

Figure 16-1 shows a group of 4 curves selected in Operand A.

- **Add Input Data Name** is checked. The curves are labeled, Mic 1 through Mic 4.
- The result of this step will be 4 values added to the Memory List. The values are labeled: "Sens @ 1 kHz (Mic 1)" to "Sens @ 1 kHz (Mic 4)"
- When using Batch Processing it is recommended that you use the **Add Input Data Name** option so that the results are clearly named.

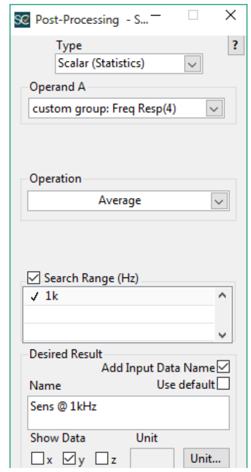

Figure 16-3: Batch Processing

Desired Result

This is the name of the newly created curve. If **Use default** is checked, the Result name is automatically generated. A new name will be created using the original curve name concatenated with the mathematical operator, e.g., <Operand A><mathematical operator><Operand B>. This applies to all types of Post Processing steps except the User Equation. See ***User Equation (optional module required) on page 256*** for more details.

When this box is unchecked, a custom name can be entered manually. In ***Figure 16-1*** the result has been named *Relative to Reference*.

Average Curve/Waveform

The Average Curve/WFM Post Processing function allows you to get the average curve (or waveform) of a selected group of data in the Memory List.

The example in ***Figure 16-4*** shows the output of the Average Curve/WFM function when using the **Statistics curves.dat** example data included with SoundCheck.

The blue curve shows the Power Average of the selected curves. The curve name shows the result of selecting "Use default".

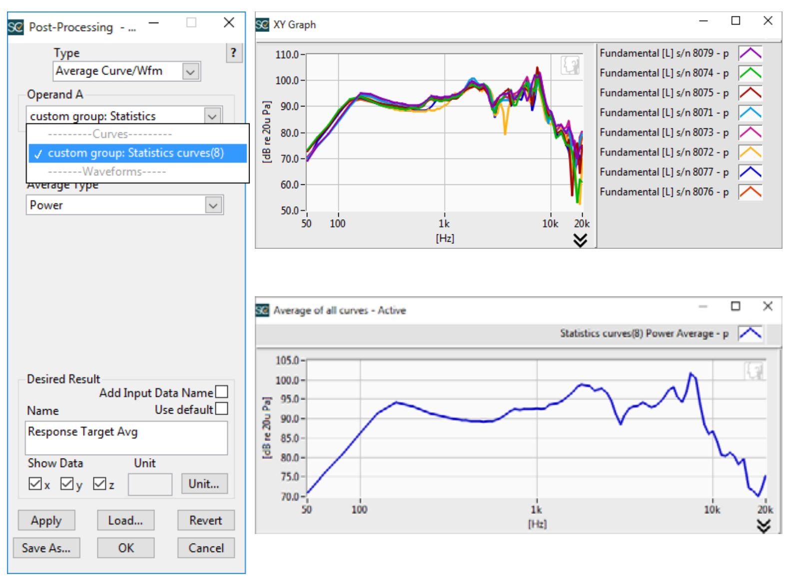

Figure 16-4: Average Curve/Waveform

Average Type

Power - excludes the phase - Typically used for curves that are not from the same spatial position, e.g.: Mic Array.

Complex - includes phase - Recommended when all data is acquired from the same spatial position.

Arithmetic

Allows block arithmetic operations (e.g., addition, subtraction, multiplication, and division) to be performed between two complex data sets (magnitude and phase vs. frequency) or waveforms. Operand A and B can be any curve, waveform or value chosen from the *Memory List*. Mathematical operations are performed in parallel on Operand A and B: Point by point operation. Interpolation is performed when the frequency axis of the two operands do not match.

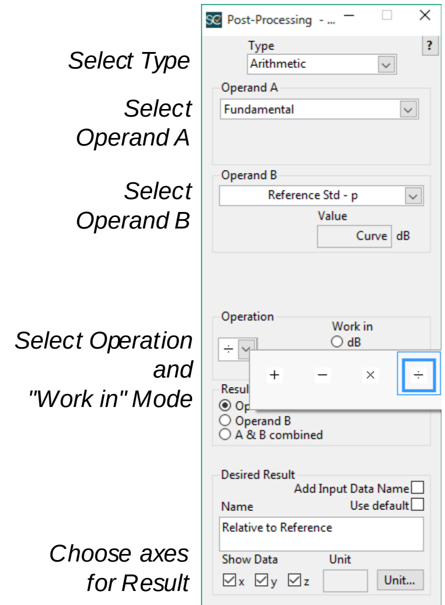

Figure 16-5: Arithmetic Post-processing

- **Available Operations**: Addition, Subtraction, Multiplication and Division
- **Work in mode**: Mathematical operations can be performed on the dB or power values instead of Linear values (for Y axis)

  Example of Work in dB: 90 dB + 90 dB = 180 dB (The math is applied on the dB values.)

  Example of Work in Linear: 90 dB + 90 dB = 96 dB (The math is applied on the linear values.)

  Example of Work in Power: 90 dB + 90 dB = 93 dB (The math is applied on the power values.)

Note: Multiplication or Division of dB to dB values is NOT allowed.

- Units are combined according to the operation chosen. For more information on the combination of units, refer to ***Equation Editor Functions on page 595***. (The units of the final result can be modified by clicking *Units* in the editor.)
- **Result x-axis same as**: Allows you to set the x-axis scale from the selected operand or combination of operands.

X-axis combination:

Operand type		X-axis Combin				
A	B	A	B	A&B	Output type	Rule applied
Curve	Point	x			Curve	Y_B, Z_B is applied as a complex constant on the entire curve A regardless of X_B.
Curve	Point		x		Point	Curve value is applied on point value @ X_B using interpolation.
Curve	Point			x	Curve	Result curve is the same as A except at $X_{B,}$ where points A and B are combined.
Curve	Curve	x			Curve	Points with X_A only.
Curve	Curve		x		Curve	Points with X_B only.
Curve	Curve			x	Curve	Points with X_A & X_B combined.
Point	Point	x			Point	One point with $X_{A.}$.
Point	Point		x		Point	One point with X_B.
Point	Point			x	Curve	Two points of same values @ X_A & X_B.

You can choose **A & B combined** as your x-axis, instead of choosing to combine magnitude values regardless of frequency. This allows you to splice curves with different ranges together. This works best when the range of one curve ends when the range of the next curve begins.

See ***Windowing on page 258***.

Figure 16-6 shows a Low Frequency and a High Frequency curve (in the XY Graph of the *Display Editor*).

Figure 16-7 shows the resulting curve when the two curves are added and the x-axis is the same as A & B combined.

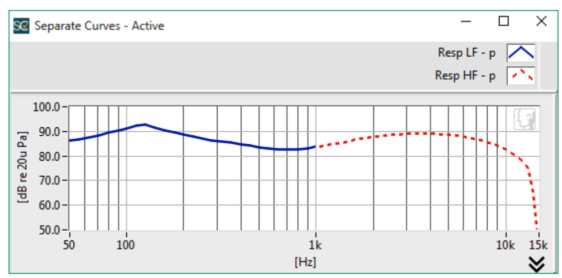

Figure 16-6: Separate Curves

Microphone or loudspeaker measurements that require two different measurements to acquire the proper response data can be combined into one curve using this technique.

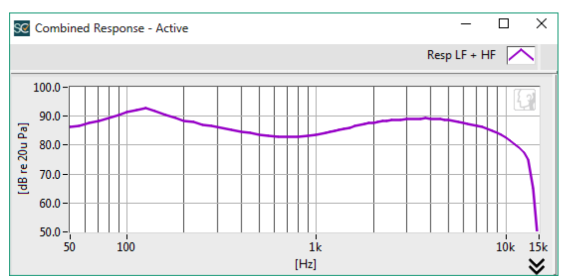

Figure 16-7: Combined Curves

Constant

Allows any curve, value or waveform in the *Memory List* (Operand A) to be modified by block arithmetic operations (e.g., addition, subtraction, multiplication, and division). Operand B can either be a single value or a user defined constant. The operation is made on real data. The constant is applied on only one axis: each value of a single axis of Operand A is combined with the constant, Operand B.

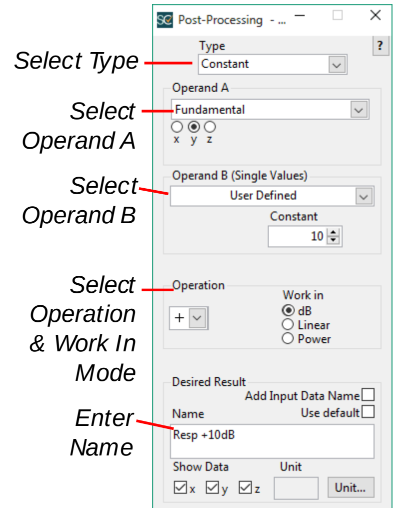

Figure 16-8: Curve Shifting Using a Constant

Rules - Axis choices for Operand A in Post Processing

- **X** – Modifies (+, -, x, /) Operand A by the factor entered in the Constant value box, ONLY ON THE X Axis. e.g., Frequency
- **Y** – Modifies (+, -, x, /) Operand A by the amount entered in the Constant value box, ONLY ON THE Y Axis. e.g., Magnitude
- **Z** – Modifies (+, -, x, /) Operand A by the amount entered in the Constant value box, ONLY ON THE Z Axis. e.g., Phase curve

  Operand B can be a Single Value item from the *Memory List*. In this case you can chose the x, y or z value in the Operand B selection field. This value can then be applied to Operand A as per the rules stated above.

  Work in mode is the same as in Arithmetic except: Multiplication or Division of dB to dB values is allowed. Units combination is the same as in Arithmetic. Refer to ***Arithmetic on page 237*** for more information.

Unary

Allows unary operations to be performed on a complex data curve as well as waveforms.

Single values, curves and waveforms are available as Operand A.

However, it should be noted that not all operations are valid with all operand types. Please refer to ***Figure 16-10***.

All operations allow you to select the axis and Default or Custom Name for the Desired Result.

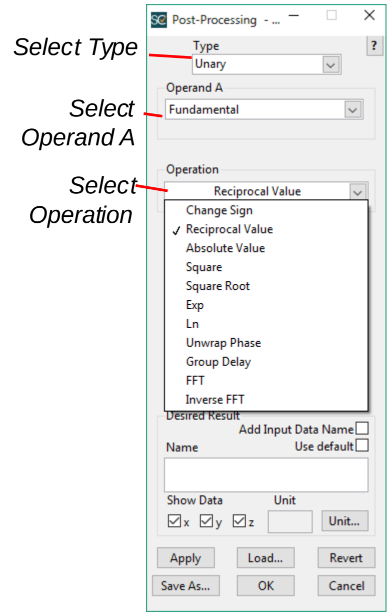

Figure 16-9: Unary Options

Name	Operation	Operand A Curves	 Values	 Waveforms	Result Type	Typically Used For
Change Sign	Polarity inverter for phase	Y	Y	Y	Same as Operand A	To correct for a microphone with inverted polarity
Reciprocal Value	Calculates the Inverse (1/X) of Operand A	Y	Y	N	Same as Operand A	Inverting curves used as correction curves
Absolute Value	Returns the positive valued magnitude of Operand A	Y	Y	N	Same as Operand A	Math operation
Square	Returns the Square of Operand A (Operand A x Operand A)	Y	Y	Y	Same as Operand A	Math operation
Square Root	Returns the Square Root of Operand A	Y	Y	Y	Same as Operand A	Math operation
Exp	Calculates the exponential of Operand A ($exp^{Operand\ A}$)	Y	Y	N	Same as Operand A	Math operation
Ln	Returns the natural Logarithm of Operand A	Y	Y	N	Same as Operand A	Math operation
Unwrap Phase	Returns the Unwrapped phase of Operand A	Y	N	N	Curve	Allows Unwrapped Phase to be exported as a curve
Group Delay	Calculates the Group Delay (negative derivative of Phase) of Operand A	Y	N	N	Curve	Loudspeaker or other frequency response function analysis
FFT	Calculates the FFT of a Waveform	N	N	Y	Curve	Spectrum analysis
Inv FFT	Returns a real-valued time signal (waveform) from a complex (Mag & Phase) spectrum or response	Y	N	N	Waveform	Time Domain Analysis

Figure 16-10: Unary Operations Chart

Unary Operations

- **Change Sign**: Polarity inverter for phase
  - Curves: Only affects the phase (Z axis). Magnitude remains unchanged.
  - Values: Axis is selectable. Typically used to invert polarity on the Z axis
  - Waveforms: Inverts polarity of the waveform on the Y axis

- **Reciprocal Value**: Calculates the Reciprocal (1/X) or Inverse of Operand A
  - Curves: Creates inverse of curve, swapping positive and negative values of the y and z axis
  - Values: Allowed to select one axis to process; x, y or z. For log data in dB, calculates the inverted transfer function (Input/Output). Useful for setting target equalization curves.
  - Waveforms: Not allowed

- **Absolute Value**: Returns the non-negative value of Operand A
  - Curves: Returns the magnitude (all values positive) of Operand A
    - Works on complex frequency domain data and sets all phase values to zero
  - Values: Allowed to select one axis to process; x, y or z. Returns the magnitude of the item selected.
  - Waveforms: Not allowed

- **Square**: Returns the Square of Operand A: Operand A x Operand A
  - Select axis for result
  - Curves:
  - Values: Allowed to select one axis to process; x, y or z
  - Waveforms:

- **Square Root**: Returns the Square Root of Operand A
  - Select axis for result
  - Curves:
  - Values: Allowed to select one axis to process; x, y or z
  - Waveforms:

- **Exp**: Calculates the exponential of Operand A ($\exp^{\text{Operand A}}$)
  - Curves: Convert dB back to real values, calculate the exponential and convert the result back to dB
  - Values: Allows you to select one axis to process; x, y or z
  - Waveforms: Not allowed

- **Ln**: Returns the Natural Logarithm of Operand A
  - Curves: To extract the all-pass portion (pure delay) from the linear phase of a response. Many steps are required, but this is a key element.
  - Values: Allowed to select one axis to process; x, y or z
  - Waveforms: Not allowed

- **Unwrap Phase**: Allows Unwrapped Phase to be exported as a curve. It also allows Unwrapped Phase to be shown in a Display without having to select Unwrapped Phase from the Right-click Options of the display window. See ***Unwrap Phase on page 341*** for details and an example.
  - Options: Under "**Show Data**", X, Y and Z must be selected
  - Returns Unwrapped Phase as a curve. This operation can only be performed on curves.

- **Group Delay**: Group Delay is the time it takes a signal to pass through a device with respect to frequency. It is commonly used to align multi-way speaker systems, e.g.: a woofer and a tweeter that have different acoustic centers.
  - The derivative of the 'unwrapped phase' as delay (not absolute time) vs. frequency
  - This is relative to the reference frequency which is the first frequency measured
  - Options: Under "Show Data", X, Y and Z must be selected to show the desired axis
  - Smoothing width in Hz: Increasing width can obscure details
  - This operation can only be performed on curves

$$\tau = \frac{1}{2\pi}\frac{d\varphi}{df}$$

- **FFT**: Calculates the FFT of a waveform
  - Select the Impulse Response Waveform from the Memory List. The Waveform must be in **linear Y units only** (not dB)
  - Select the Weighting window: None (Rectangular), Cosine Tapered, Exponential, etc
  - Check Search Range and Right-click in the field
    - Select Add, Edit or Remove
    - Select Single Point or Range
    - In the Select Search Range section, edit the time range on which to apply the FFT and weighting
    - ***Data outside of the time range is ignored***
  - The result will be a frequency curve with a linear X axis, default Y unit in dB and Phase in degrees
  - If **Search Range** is unchecked, this indicates it is set to "All" points in the selected data
  - This operation can only be performed on waveforms

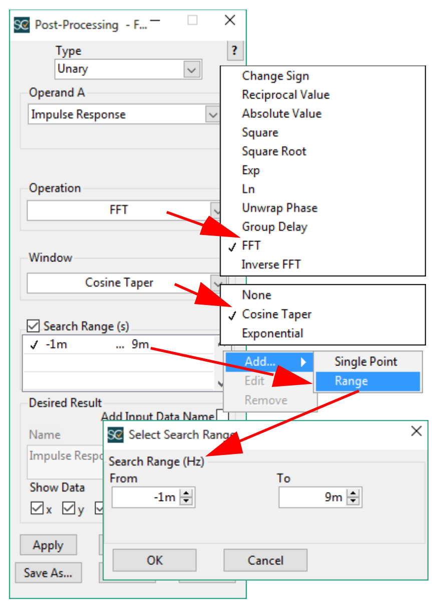

Figure 16-11: FFT and Inverse FFT

- **Inv FFT**: Returns a real-valued time signal (waveform) from a complex (Mag & Phase) spectrum or response.
  - Select the response curve from the Memory List
  - The output is a Linear or dB (envelope) Waveform
  - This operation can only be performed on curves

Scalar (Statistics)

This allows calculations of a single value from a specified curve or waveform. Again, not all operations are valid for all operand types.

- None of these functions can be applied on single values.
- IEEE and ITU loudness operations can only be performed on frequency curves.

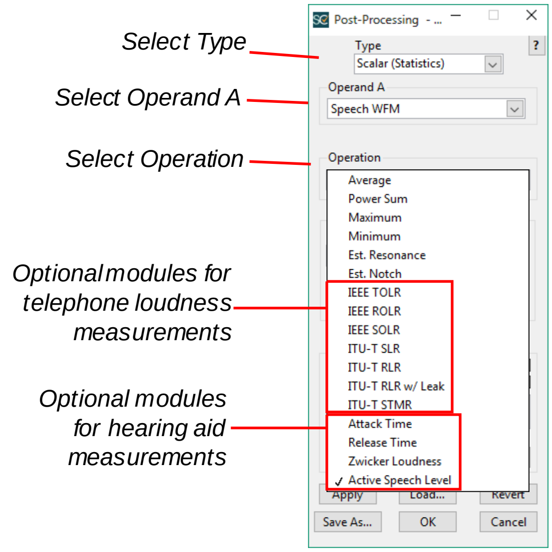

Figure 16-12: Scalar Operations

Average

This returns the average of a curve as a **single value**.

Calculates the mean of the curve Y values $\bar{y} = \sum_{i=1}^{N} \left(\frac{y_i}{N}\right)$ over the search range, regardless of the units (e.g., averaging a curve with 3 points, 10, 12 and 14 dB, will yield a value of 12 dB.

Power Sum of Curve

Calculates the square root of the sum of the squares of each Y value in a spectrum.

$$\text{Curve PwrSum} = \sqrt{\frac{1}{B}\left(\sum_i Y_i^{\,2}\right)}$$

This is dependent on the Bandwidth Value, “B”. The Bandwidth Value varies according to the weighting applied in the Analysis Step and the analysis method selected.

e.g., The square root of the sum of the power of each RTA bin, or the sum of the power of a specific frequency band of an FFT spectrum.

Note: All spectrum are summable.

Power Sum of Waveform

You can also calculate the Power Sum of a Waveform.

- The Z axis of the result shows the power sum
- The Y axis of the result shows the power sum in dB
- The X axis is excluded from the result
- The formula below shows that the result is normalized by the sampling rate

$$\text{Waveform PwrSum} = \frac{\sum_i (x_i)^2}{\text{sampling rate}}$$

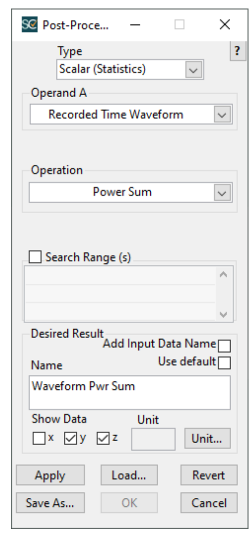

Figure 16-13: Waveform Power Sum

Maximum

Finds the maximum curve Y value in the specified search range and returns X, Y and Z values at that point.

Minimum

Finds the minimum curve Y value in the specified search range and returns X, Y and Z values at that point.

Est. Resonance

Finds the resonance frequency, amplitude, and quality factor (Q) of a peak in a curve. The calculation is based on an algorithm that fits a quadratic polynomial to sequential groups of data points. The number of data points used in the fit is specified by the width control in terms of either how many dB down from the peak or % of the peak. For most woofers, -3 dB will suffice, but for low Q drivers (such as tweeters) -1 dB may be required in order to resolve the resonance frequency from the fitted curve.

Est. Notch

Finds the antiresonance frequency, amplitude, and quality factor of a dip or notch in a curve.

IEEE-661 OLR

This calculates the average of the Frequency Response from Start Freq to Stop Frequency The average is taken of the amplitude raised to the power of (1/Exponent). The Frequency Response must be given in dB. The average is calculated using the trapezoid rule. Values for Frequency Response at the exact Start and Stop Frequencies are found by interpolation.

To calculate OLR according to **ANSI/IEEE 661**, the Start Freq must be no higher than 300 Hz and the Stop Freq must be no lower than 3300 Hz. The Frequency Response shall (according to the standard) cover this frequency range and include at least 12 measurement points within the frequency range.

$$\mathrm{LoudnessRating} = -20\log\left[\frac{\displaystyle\sum_{i=2}^{N}\left(\log\frac{f_i}{f_{i-1}}\right)\cdot\frac{\left(10^{\frac{S_i}{20\alpha}}+10^{\frac{S_{i-1}}{20\alpha}}\right)}{2}}{\log\left(\frac{f_N}{f_1}\right)}\right]^{\alpha}$$

- α = Loudness compression exponent. The default value, based on the standard, is 2.2.
- N = number of measured frequencies
- f_i = frequency at index i. The frequency range is usually 300 - 3300 Hz
- S_i = electroacoustic sensitivity of the path at frequency fi. This sensitivity is usually expressed in units of dB mV/Pa or dB Pa/mV

  A correction may be applied to the sensitivities in these formula to account for leakage of (legacy) Type 1 ear simulators, or for impedance, depending on the application and/or applicable performance standards.

ITU-T SLR & RLR (optional module required)

Calculates the send and receive loudness ratings according to the **ITU-T Rec. P79 Loudness Rating** from the values in the Frequency Response. Requires optional module **2007 - Loudness Rating**.

Formula: $$\text{LoudnessRating} = -\frac{10}{m}\log\left(\sum_{i=N_1}^{N_2} 10^{\frac{m(S_i - W_i)}{10}}\right)$$

- m = Loudness growth exponent
- i = frequency band index (ISO R10 1/3-octave bands)
- Wi = frequency weighting in dB
- Si = electroacoustic sensitivity of the path at frequency fi
- N = frequency band number (typically 4-17 where band no. 1 = 100 Hz)

  A correction may be applied to the sensitivities in these formula to account for leakage of (legacy) Type 1 ear simulators, or for impedance, depending on the application and/or applicable performance standards.

ITU-T STMR (optional module required)

Calculates the sidetone loudness ratings according to the **ITU-T Rec. P79 Loudness Rating** from the values in Frequency Response. Requires optional module **2007 - Loudness Rating**.

For IEEE standards, the exponent can be user defined. For ITU standards, both the exponent and the weighting curve can be user defined. This is for future use and modifications of standards. The step is set to **Use Default Values** by default.

- Select **Type**: Scalar
- Select **Operand A**: Fundamental
- **Operation**: ITU-T options; SLR, RLR, RLR with Leak and STMR
- **Weighting Function**: Select "Use Default Values" or select an item from the Memory List
- **Search Range**

  If Search Range is unchecked, this indicates it is set to "All" points in the selected data.

  Check Search Range and Right-click in the field

  - ♦ Select Add, Edit or Remove
  - ♦ Select Single Point or Range
  - ♦ In the Select Search Range section, edit the frequency range **From** and **To** values
- Select Name option: Use Default or Custom Name for output

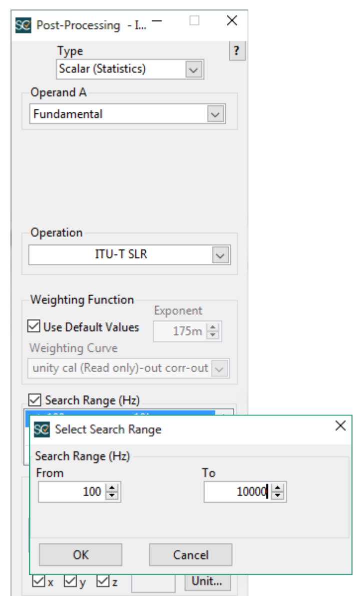

Figure 16-14: ITU-T SLR Settings

ITU Wideband Weighting Curves (optional module required)

The default ITU weighting curves used for loudness rating are narrowband (200-4 kHz). They are from **ITU Recommendation P.79, Table 1**.

Requires optional module **2007 - Loudness Rating**.

Wideband weighting curves (100-10 kHz) can be used instead. They are from **ITU Recommendation P.79, Table A.2**. They are intended for use on devices that can operate both in narrowband and wideband modes.

Wideband weighting curves are available in the SoundCheck Data folder: C:\SoundCheck 18.1\data\.

Figure 16-15 shows an example of how the weighting curves are used.

1. Open the RLR or SLR DAT file in the Memory List
2. The weighting curve can then be selected in the Post Processing step
3. Exponent is same as default
4. If Search Range is unchecked, this indicates it is set to “All” points in the selected data. RLR is calculated on each point in the weighting curve, according to the standard.

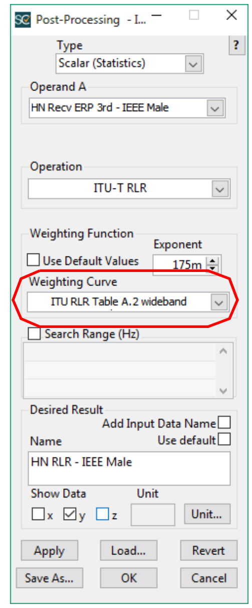

Figure 16-15: Wideband Weighting Curve

5. For better results, Operand A should be the received frequency response in 1/3rd octave resolution. This can be the result of a 1/3rd octave RTA measurement, or conversion from another format using “band-averaging” as described in **IEEE 269-2012**
6. For use in a sequence, use a Recall Step to load the RLR or SLR DAT file into the Memory List

Attack and Release (optional module required)

This function can be used to test the time it takes for the signal from a DUT to stabilize after a sudden change in signal level. Requires optional module **2008 - Attack and Release**.

- In this example, Stimulus is set to 2 kHz, with 2 different stimulus levels (***Figure 16-16***)
- Time Envelope must be turned on in the Analysis Step of the sequence (***Figure 16-16***)
- See the default sequence “**Release Time**“ in the Hearing Aid folder for an example

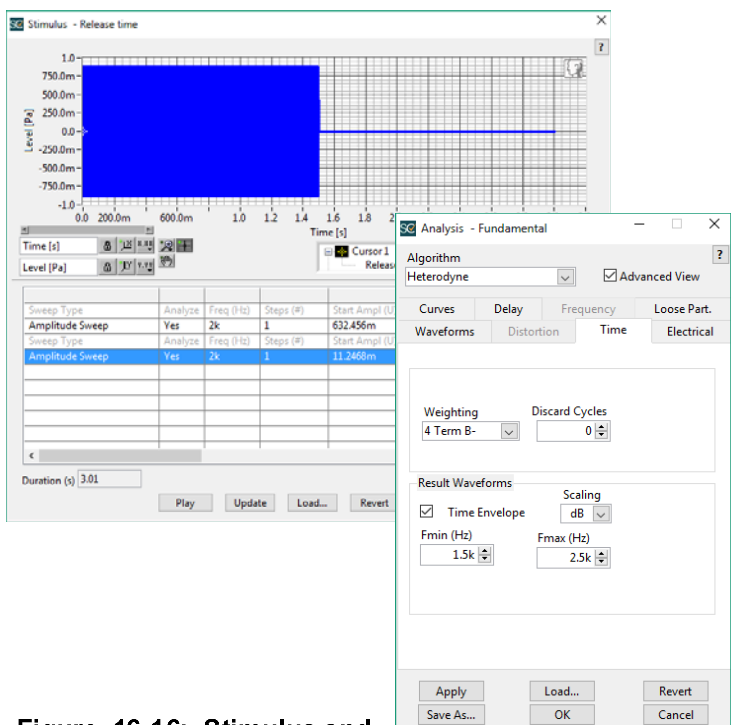

Figure 16-16: Stimulus and Analysis Step Settings

Attack Time

Figure 16-17 shows the settings for the Attack Time function in Post Processing. By making an initial measurement of the device under test, you can look at the **Attack Time Envelope** to determine if the Search Range of the example sequence is correct for that particular device.

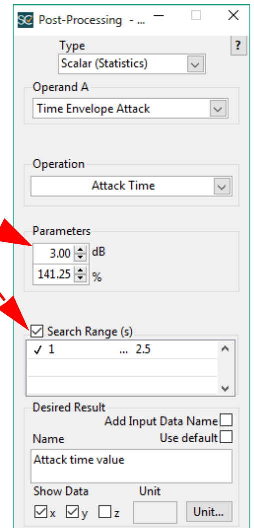

Figure 16-17: Attack Time

In this case, the Attack level is set to 3 dB and the search range is set at 1 to 2.5 seconds.

> ***Note:*** The Search Range of the Post Processing step must be set to start before the onset of the transient and end well after the point were the signal has become stable. An initial measurement of the device will provide and Attack Envelope so that you can determine what the actual range should be. See *Figure 16-18*. Right-click the **Search Range** field to edit.

> ***Important!*** In order to avoid "turn on transients" the Search Range should never start at 0 mSec.

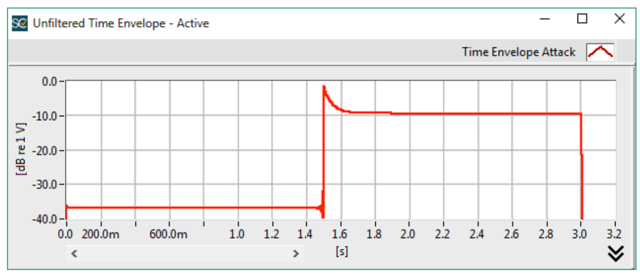

Figure 16-18: Attack Envelope

The process starts by searching the Attack Time Envelope in reverse, from the Search Range End Point as specified in the Post Processing step. The green cursor in *Figure 16-19* indicates this point. From here, the Post Processing step searches back to the point where the signal is 3 dB greater than the stable level. The red cursor indicates this point. The difference in level between the two points is shown in the Delta Y field (Δ **y**); 3 dB as specified in the Parameters section of the Post Processing step.

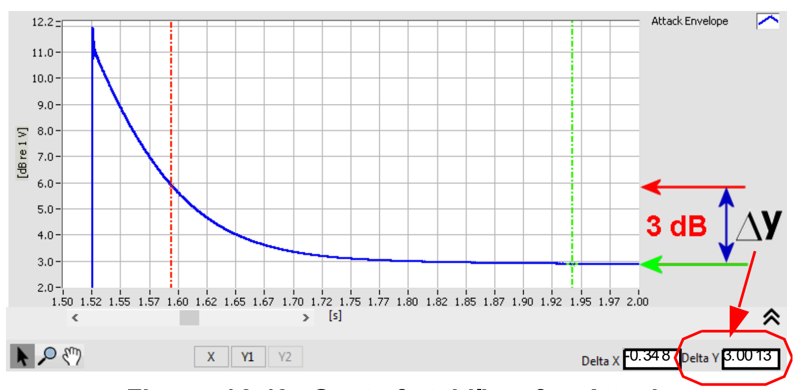

Figure 16-19: Start of stability after Attack

The next step is to determine the actual attack time. Cursor 1 and Cursor 2 are positioned

This is the time from the onset of the transient to the "3 dB up" point. ***Figure 16-20*** shows the attack time in the Delta X – Time box (Δ **t**).

(Delta X and Delta Y are visible by clicking the **Double Down Arrow** at the right side of the display window.)

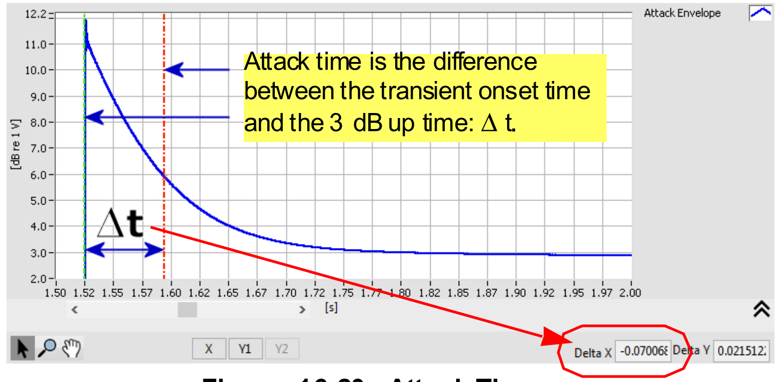

Figure 16-20: Attack Time

Release Time

In the example sequence, the Release level Parameter is set to –4 dB with a Search Range of 1 to 2.5 seconds. The Search Range may need to be increased if testing a device with a long release time.

The Release Time function in the Post Processing step works in a similar fashion to the Attack Time.

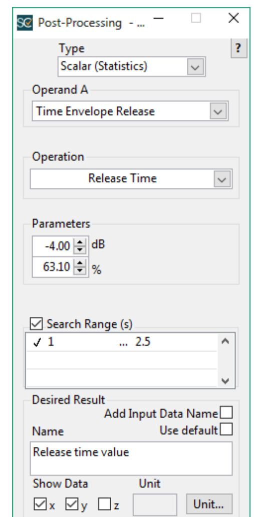

Figure 16-21: Release Time

It searches in reverse for the point where the signal becomes stable after release. From this point it then searches for the point where the envelope level drops below the stable level, by the value specified in the Parameters section of the Post Processing Step.

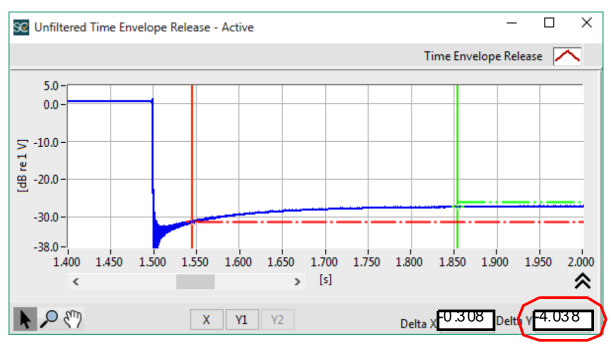

Figure 16-22: Start of stability after Release

The Release time is then determined to be the time from where the level of the envelope drops below the threshold level to the point where the envelope level is 4 dB down from the stable level.

This method of determining the attack and release time, as well as the values noted in the setup of the sequence, is in accordance with **ANSI standard S3.22 – 1996, section 6.15.2**. The actual Attack and Release Limits will vary according to the specifications of the device under test.

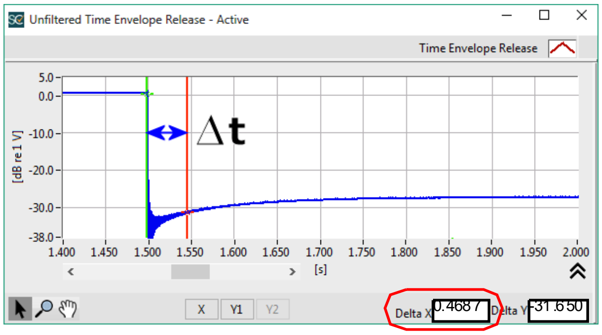

Figure 16-23: Release Time

Zwicker Loudness (optional module required)

Zwicker Loudness calculates the overall perceived loudness of a sound. This post-processing operation uses a psycho-acoustic model which takes into account the nonlinearity of the human ear to sound at different frequencies and levels. It provides the capability to measure the perceived loudness of complex sounds, e.g., telephone ring tones. Requires optional module **2031 - Zwicker Loudness Rating**.

This is the actual loudness of the sound recorded, with a value in PHONS and SONES.

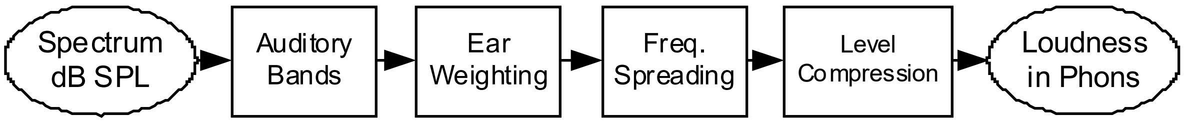

Figure 16-24: Zwicker Loudness Process

The Zwicker Loudness process from ***Figure 16-24*** is detailed below.

1. The spectrum in dB SPL is recorded. It must be a "**Calibrated Acoustic Pressure**" in Pascals or dB SPL.
2. It is regrouped into auditory bands, according to Bark Scale
3. The spectrum is remapped
4. A frequency weighting is then applied to take into account the response of the ear
5. A frequency masking effect is applied
6. A mathematical compression is applied to achieve the final loudness number
7. This yields the actual loudness in PHONS, by power summing the compressed level

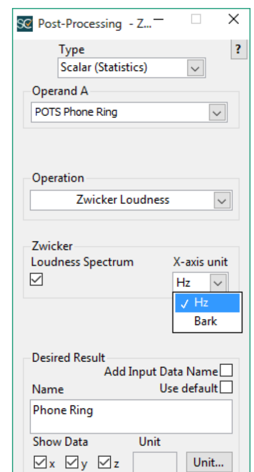

Figure 16-25: Zwicker Settings

- Zwicker Loudness uses an up-to-date algorithm that conforms to the **ITU-R Rec. BS.1387 PEAQ** standard. It is similar to **ISO532**, which is simpler and only processes 3rd Octave spectrums.
- Zwicker Loudness is intended for use with broadband signals such as speech. It is not for use with single tones.
- Zwicker can be applied to any spectrum, as long as it is a calibrated acoustic spectrum in Pascals or dB SPL.
- The curve must be consistent in resolution, e.g.: 5 Hz, 3rd Octave, etc. The frequency resolution of the spectrum is not necessarily important. The algorithm converts the spectrum to power spectral density before calculating the loudness.
- Zwicker Loudness also features an option to output the Loudness Spectrum to the Memory List. The units for the X-axis are selectable between Hz and Bark scale (See ***Bark Scale on page 253***). The loudness spectrum allows you to determine which frequencies are responsible for the loudness, and is useful for analysis of signals such as: telephone ring tones, speech and music.

Zwicker Example

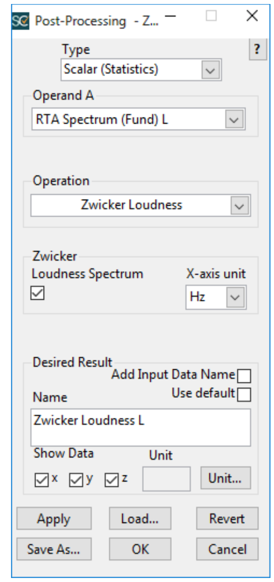

Figure 16-26: Zwicker Example

This example shows the Zwicker Loudness results for a set of headphones.

- The Post-Processing step in ***Figure 16-26*** shows "RTA Spectrum (Fund) L" selected under **Operand A.**
- The response curves in this example were acquired using Pink Noise stimulus and RTA Spectrum Analysis
- Two Post Processing steps are required to create separate results with unique names for left and right headphones
- **Loudness Spectrum** is checked and the X-axis unit is set to Hz
- The **Show Data** check boxes allow you to select which axis' to show up in the Loudness Value

Output

The Zwicker Loudness Post Processing output is shown in ***Figure 16-27***.

- Headphones were measured with Pink Noise stimulus and RTA Spectrum analysis
- The 12th Oct RTA Response is used as Operand A in the Zwicker Post Processing step
- Zwicker Loudness Spectrum - This shows which frequencies are responsible for the loudness
- Separate Loudness values are shown for Left and Right Headphones
- X = Loudness in Phons
- Y = Power sum of original spectrum
- Z = Loudness in Sones

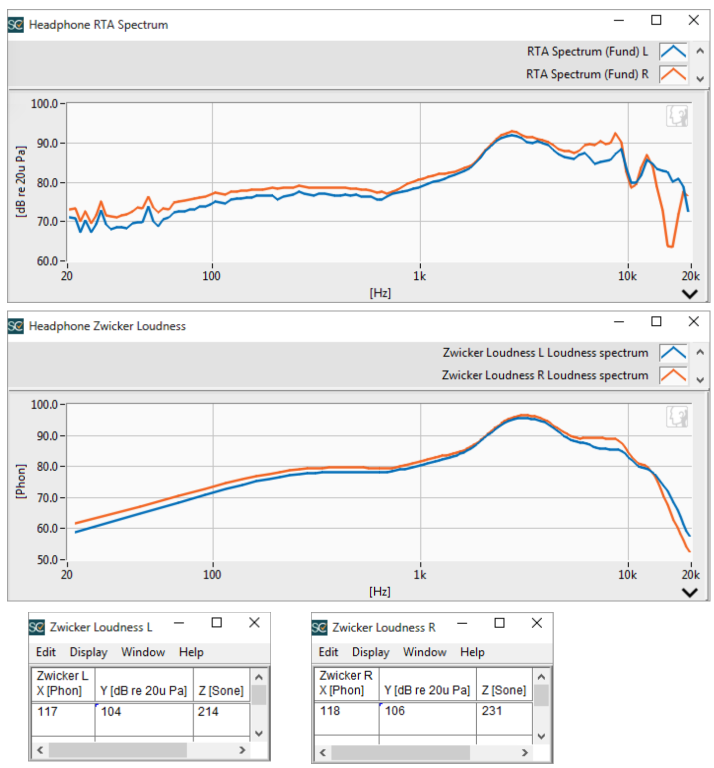

Figure 16-27: Zwicker Loudness Output

Bark Scale

The auditory pitch scale is expressed in Barks. The relationship between Barks and Frequency (Hz) is shown in *Figure 16-28*.

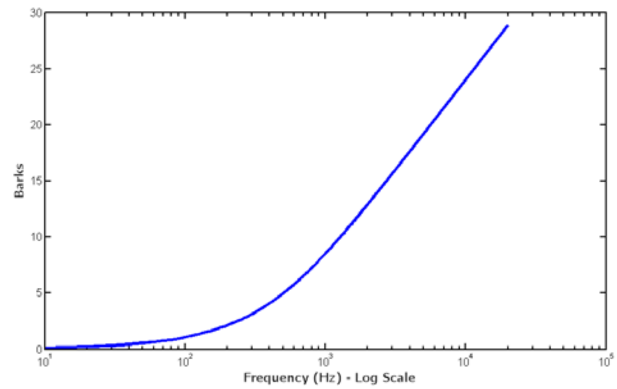

Figure 16-28: Bark Scale

Active Speech Level - P56 (optional module required)

Used with a speech signal, this operation evaluates levels for only the parts of the waveform where speech is actually present. Silent gaps are excluded but short interruptions that are part of continuous speech are included.

Requires optional module **2033 - Active Speech Level**.

It is widely used in telephony applications, i.e.: testing to **ITU-T P.56 05/93 Method B**.

The Active Speech Level of the WAV file can be set in the Stimulus Editor.

See ***Stimulus Settings on page 114***.

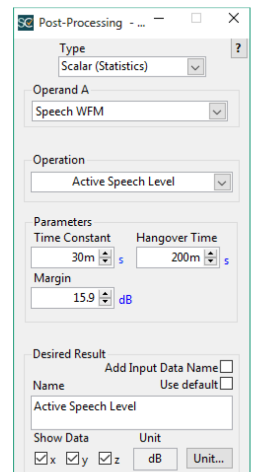

Figure 16-29: Active Speech Level

Parameters:

- **Operand A -** This should be a speech waveform
- **Time Constant (Sec)** - Time constant of exponential averaging used to smooth the envelope of the speech signal (30 ms default)
- **Hangover Time (Sec)** - Allowable time for silence during active speech. Longer silent gaps between active speech sections are ignored and left out of the calculation. (200 ms default)
- **Margin (dB)** - Difference, in dB, between threshold of activity and active speech level. When the level of the background noise is high, the margin can be reduced in order to exclude the noise. (15.9 dB default)

Output:

The Active Speech Level value in the Memory List shows the following:

- **X - Activity Factor (%):** Percentage of time during waveform where speech is active
- **Y - Active Speech Level (dB):** Mean power of speech measured over the aggregate time of activity
- **Z - Long-term Level (dB):** Mean power of the waveform measured over its entire duration

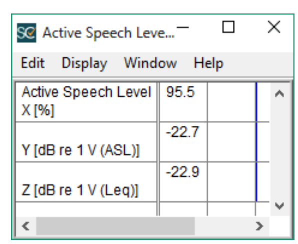

Figure 16-30: Value in Memory List

Smoothing

Selects the degree of smoothing for the displayed curve. Smoothing corresponds to a running average on a frequency axis with a window of the given width (1/n octave or number of Hertz). The running average can be weighted using a Hanning window in order to get a smoother curve result.

- Smoothing applies a running weighted average on the curve with a defined width. The width can be expressed in 1/n octave or linear terms.
- Two different weightings can be applied: None (Rectangular) or Hanning. The weighting is symmetrical around the middle point and doesn't include zero end points.
- The smoothed curve has the same number of points as the original
- The curve can have an uneven spacing in frequency domain, because interpolation is used
- Smoothing is applied on power values (y^2) and unwrapped phases
- The curve end points are unchanged. To do this, a symmetrical reduction of smoothing width at the edges is used.

Note: To keep both the beginning and end points of the smoothed curve untouched, the smoothing width is gradually reduced down to zero when it reaches the extremities of the curve. In ***Figure 16-32***, the two curves merge at 20 kHz.

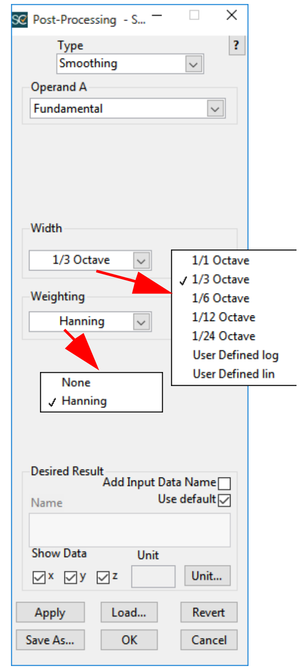

Figure 16-31: Smoothing

The display in ***Figure 16-32*** compares an 8196-line Fundamental curve with the same curve smoothed to 1/3 octave.

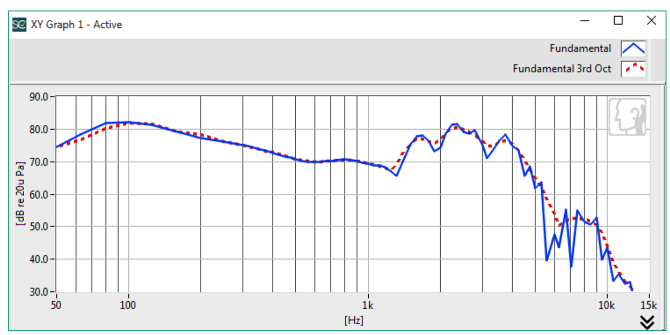

Figure 16-32: One-third Octave Curve Smoothing

Intersection (search)

This is used to find the intersection points between two curves or waveforms, or between a curve/waveform and a single point.

Search Up - Determines the first intersection point

Search Down - Determines the last intersection point

Return all intersection points - Yields a curve of all of the intersection points. (This can also be displayed as a table of values in a *Display Step*.)

Search Range allows you to narrow the range of the search or to exclude regions that a search should not occur within.

If Search Range is unchecked, this indicates it is set to “All” points in the selected data.

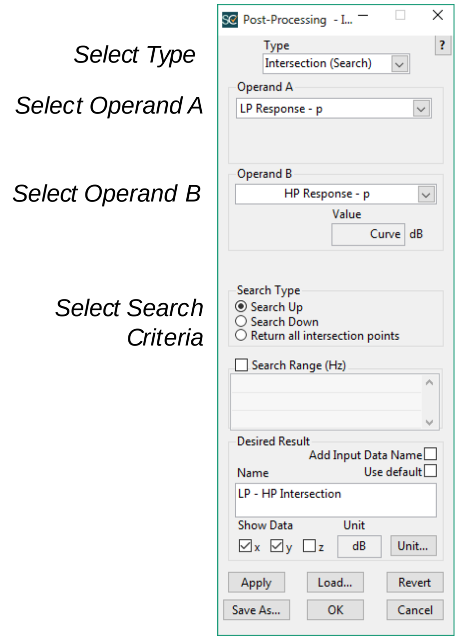

Figure 16-33: Intersection of Two Curves

Result

The Intersection Value is available in the Memory List.

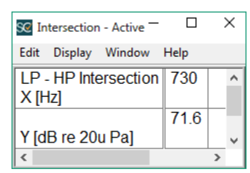

Figure 16-34: Intersection Value

User Equation (optional module required)

Requires optional module **2012 - Equation Editor**.

This section allows you to build an arithmetic equation using constants and the values generated by the sequence. You can build an equation (similar syntax to writing an equation in Excel) and define the variables and results at the top of the editor. Build this equation by selecting a curve, single value name, or defining a constant from the pull down lists that appear in the top table of the *User Equation Parameters* window.

Assign variable names for these data in the **Variables** column, using numbers and letters, but no spaces. Choose which axis of the curve or value you wish to use. The **Values** and **Units** columns are only accessible if you are in a user defined constant row.

Curves and Values from the *Memory List* have sequence-determined values and units. Use these variables in the equation box in the *Post Processing Editor*. ALL CALCULATIONS ARE PERFORMED IN LINEAR UNITS IN THE REAL DOMAIN. Units drawn from the *Memory List* are analyzed, and if a log scale is in use, it will be first converted to linear before being used in calculation. The Editor will then convert back to the log scale to output results, if desired. Define the units of the results by clicking **Units**.

Complex Math operations are not supported. You will need to break the operation down to its arithmetic equivalent.

Operations on the Z axis (phase) are not supported in the Equation Editor.

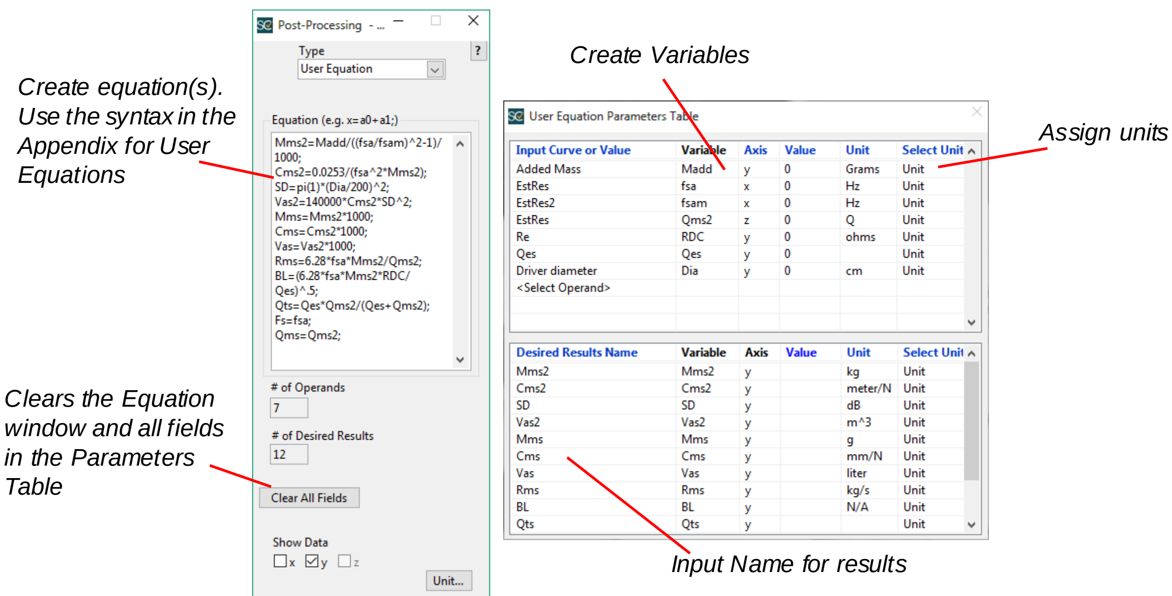

Figure 16-35: Assign Variable Names to Curves and Values From the Memory List

You can find a list of valid function syntax for this equation box in ***Equation Editor Functions on page 595***. Each equation must produce only one new variable as the result. In the equation box, the result must be on the left, an equal sign immediately to the right of the result, and then the remainder of the equation. A semicolon (;) should end every individual equation.

Each result will be listed on the right hand of the editor, and will appear in the *Memory List* as you have named it. You can use numbers and letters, but no spaces. You must assign unit information for this new value, created in your equation.

User Equation Syntax

Each equation must start with the result variable, followed by the equal sign, then the operators, variables and constants that will produce your result. Equations shall be separated by a semi colon.

Help menus are available for the ***User Equation (optional module required) on page 256*** section of SoundCheck. Press **Ctrl+H** on your keyboard or Click the "**? mark**" on the editor to open the *Context Help* window.

See ***Figure 16-36: Equation Context Help***.

It will give information on the last item your mouse has scrolled over. Press **Ctrl+H** again to make the *Context Help* window disappear, or click the close box button in the upper right hand corner.

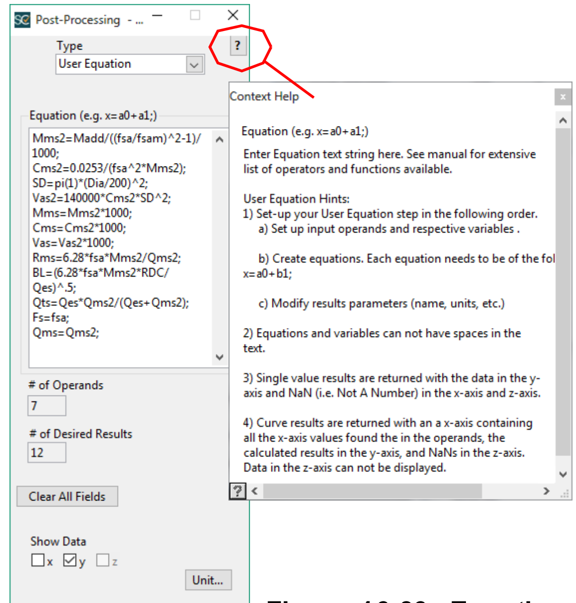

Figure 16-36: Equation Context Help

In general, use the following procedures:

1. Define the Input Operands

- Define the variables of input operands. Default is "a0, a1 …aX".
- Define the units of user defined variables.

2. Create Equation

- The desired results list will be generated by the equation.
- Default curve names are the same as the result variables but can be changed manually by the user.

3. Modify results parameters

- Change the name of the desired results. (This name will appear in the Memory List.)
- Define the units of the desired results. In the column labeled Select Units and then click Units… to change the units for that line.
- See ***Figure 16-35: Assign Variable Names to Curves and Values From the Memory List***

Note: When creating or editing very large equations, it may be helpful to use a text editor such as Wordpad since it has more space for typing. The equation can then be copied and pasted into the **Equation** field in SoundCheck. A useful tool in Wordpad is Find and Replace.

Windowing

This type allows you to trim the curve selected in Operand A with respect to the X axis.

1. Select a range of X values to include in the new curve from the **Search Range** list box, or define a new range by clicking **Edit Range**.

   (See *Search Range*)

2. Name the new windowed curve before leaving the step.

3. By default, the new curve will be named as seen in ***Figure 16-37: Frequency Window Options***, with the range appended to the original name.

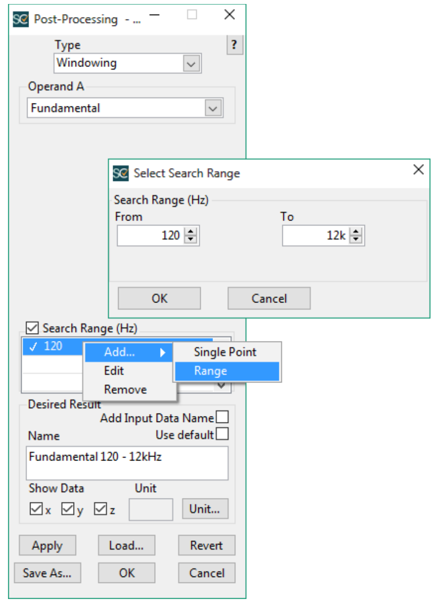

Figure 16-37: Frequency Window Options

You can select **All Points** to include the entire acquired range of the test signal. This feature allows you to splice curves together cleanly. You can trim curves first by setting the Frequency Window and then splice them in an additional Post Processing step.

- If Search Range is unchecked, this indicates it is set to “All” points in the selected data.
- Check Search Range and Right-click in the field
    - ♦ Select Add
    - ♦ Select Range
    - ♦ In the Select Search Range section, edit the frequency range **From** and **To** values

Resolution

A curve can be resampled according to a new resolution. e.g., a Linear Spectrum can be resampled to 1/3 octave.

The Resolution Post-Processing operation should be used on signals acquired with HarmonicTrak, Heterodyne and Time Selective Response Analysis modes.

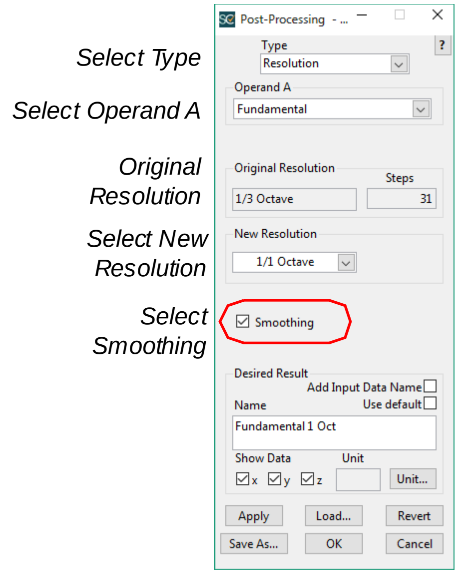

Figure 16-38: Resolution

Smoothing

There is no need to use a separate Smoothing Step before a Resolution Step as this operation is available with the step.

- The smoothing function is applied before resampling
- The smoothing width is the same as the selected resolution

Operation

Smoothing Off: The Resolution step creates the output curve by linearly interpolating the magnitude and phase data (y and z axes) of Operand A at frequencies (x-axis values) defined by the selected resolution.

Smoothing On: Resolution will first linearly interpolate Operand A data to the highest possible resolution. (This is the equivalent of the original curve being "User defined lin" with a value of 1 Hz).

Next, the Resolution Step resamples the interpolated data at the selected resolution to create the result curve in the Memory List.

Smoothing is only likely useful if the final curve resolution is higher than or "not a mathematical factor" of the original resolution. See ***Example 2*** and ***Example 3***.

Example 1

Operand A is smoothed at 1/24th (R80) octave resolution and the desired resolution is 1/3rd octave (R10). Smoothing is optional.

The values corresponding to the 1/3rd octave frequencies will be picked out of Operand A to create the result curve in the Memory List.

Example 2

Operand A is smoothed at 1/24th (R80) octave resolution and the selected resolution is User Defined, 1/10th octave. Smoothing is optional.

Without smoothing: New 1/10th octave frequencies will be calculated and the original values in Operand A will be used to linearly interpolate new values for the output curve.

With smoothing: Operand A will be used to interpolate values for every frequency between the original 1/24th octave frequencies. Then the calculated values that correspond to the 1/10th octave frequencies will be used to create the output curve.

Example 3

Operand A is non-standard 1/10th octave resolution and the selected result resolution is standard R40 1/12th octave. In this case, smoothing should applied.

Resolution will interpolate values for basically every frequency between the original 1/10th octave frequencies. Then, just the values corresponding to 1/12th octave frequencies will be picked out to create the result curve in the Memory List.

The example in *Figure 16-39* shows a comparison of the original spectrum to two different results: **1/3 octave with no smoothing and 1/3 octave with smoothing**.

Smoothing yields a curve that better tracks the mid-line of the original spectrum.

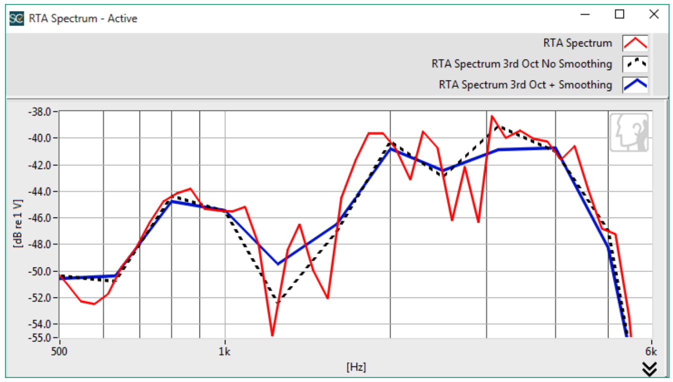

Figure 16-39: Resolution Comparison

ISO or RTA Frequencies

In measurement resolutions above 1/3rd octave, ISO and RTA frequency points no longer coincide.

When the “New Resolution” field is set to 1/6TH octave or above, you can select either ISO or RTA frequencies. This is useful for comparing measurements made with HarmonicTrak to measurements made with the RTA virtual instrument.

- When set to ISO, the result can be compared to a curve made with the HarmonicTrak algorithm
- When set to RTA, the result can be compared to measurements made with the RTA virtual instrument

When using a Stweep with a resolution of R40 (1/12th Octave), the ISO frequency points of the measured response curve can be converted to RTA frequency points. The result of the example in *Figure 16-40* can be compared to a curve made with the RTA virtual instrument.

Conversely, RTA curves can be converted to ISO curves.

Please refer to *Appendix 7: Windows Keyboard Shortcuts on page 591*, for a chart of ISO frequencies according to resolution.

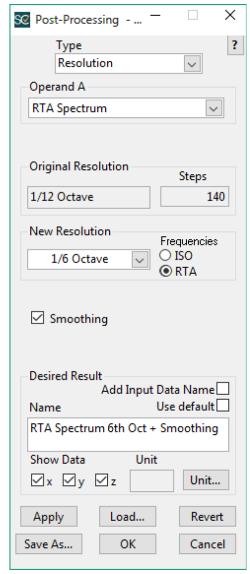

Figure 16-40: Resolution: ISO and RTA frequency

Directivity Index

This function is used to calculate the Directivity Index from a set of off axis response curves. These curves must be measured on an arc, around the DUT (based on Polar Response). This process yields a single curve that indicates the directivity at specific frequencies.

- **Reference Curve**: This is the On Axis measurement (0 degrees).
- **Weighting Function**: This is a means of applying a weighting curve that will emphasize/de-emphasize measurement values according to the angle of the measurement. This is applied to the group of selected curves. The standard is: w(q)=|Sine q|.
- **Formula Applied**: For a set of off-axis frequency responses L(f, q) and a defined weighting function w(q), the directivity index function is defined by:

$$\mathrm{Index}(f) = 10\log\left[\frac{|L(f,0)|^2 \cdot \int_0^{\max\theta} w(\theta)\,d\theta}{\int_0^{\max\theta} w(\theta)\cdot|L(f,\theta)|^2\,d\theta}\right]$$

Directivity Index

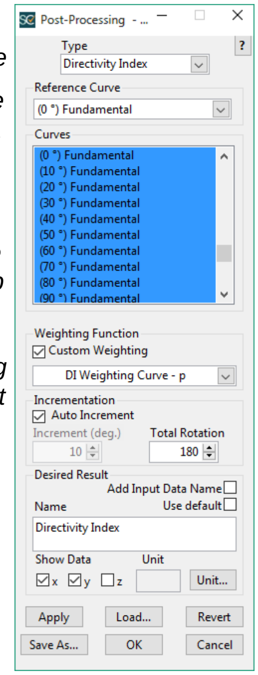

Figure 16-41: Directivity Index

where Maxθ is the total range of angle.

The formula used for Directivity Index is the one described in [1], with the assumption that the DUT is symmetrical around its nominal axis.

[1] Beranek LL. Acoustics. New York: McGraw-Hill Electrical and Electronic Engineering Series, McGraw Hill; 1954

Incrementation:

- **Auto Increment**: equally divides the selected curves across the Total Rotation in degrees,

  e.g.; If Total Rotation is set to 180 degrees and there is one Reference curve and nineteen Off Axis curves, Auto Increment will calculate an angle of 10 degrees between each measurement.
- **Increment (deg.)**: Sets the Angle of increment in degrees for each measured curve
- **Desired Result**: This can use the default name or a user selected name

Nth Octave Synthesis

This operator is meant to transform an FFT Spectrum into an RTA Spectrum. The transform is done by making a power sum of all adjacent FFT lines that are encompassed in the target Nth-octave bands. The final RTA Spectrum has the same frequencies as the RTA analyzer.

- This can be used after an analysis step using the Spectrum or Dual Channel algorithms. e.g., A response auto-spectrum with 1 Hz resolution can then be transformed into a 3rd octave spectrum, from 20 Hz to 20 kHz.
- It can be used to reduce the resolution of an RTA spectrum, e.g., going from a 1/24th octave to a 1/3rd octave spectrum.
- The algorithm can be used to convert the Summable Spectrum saved from a “pre-SoundCheck 7 FFT Spectrum Analyzer”.

Any FFT Spectrum acquired with SoundCheck can be processed with Nth Octave Synthesis.

The example in ***Figure 16-42*** shows the Spectrum (acquired with the Spectrum Analysis module) being synthesized to an RTA Spectrum with a resolution of 3rd Octave. The result is compared to the Summable Spectrum in the XY Graph. The FFT Spectrum is in Blue (lower) and the RTA Spectrum is in Red (upper).

Note: This operation should not be used on a frequency response result (Fundamental from HarmonicTrak), because one cannot make a power sum of the ratios of output over input. In the case of frequency response, use the Resolution and Smoothing operations in Post Processing.

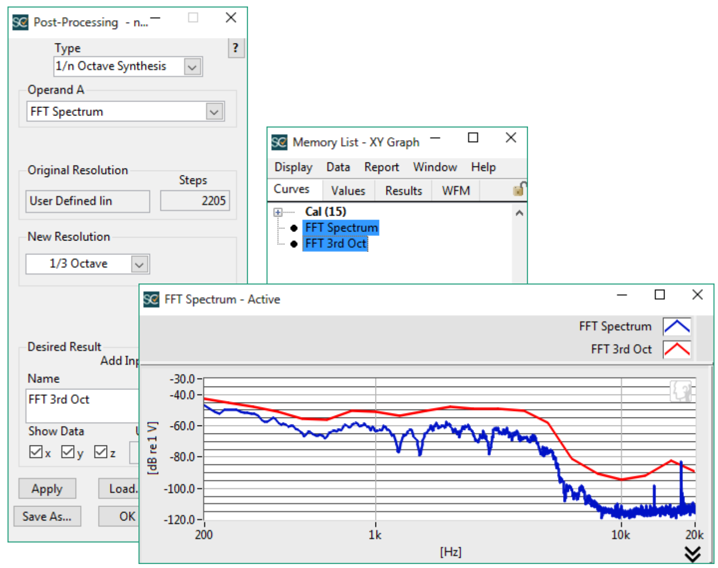

Figure 16-42: 1/Nth Octave Synthesis

FFT Spectrum			nth Octave RTA Low Frequency Cut-off			
Resolution	**Minimum Time Required**	apply nth octave synthesis	**R10 Cut-off (1/3rd oct)**	**R20 Cut-off (1/6th oct)**	**R40 Cut-off (1/12th oct)**	**R80 Cut-off (1/24th oct)**
100 Hz	10 mS	>	2000 Hz	4000 Hz	8000 Hz	16000 Hz
10 Hz	100 mS	>	200 Hz	400 Hz	800 Hz	1600 Hz
1 Hz	1 S	>	20 Hz	40 Hz	80 Hz	160 Hz

Figure 16-43: FFT to RTA Nth Octave Synthesis

A minimum of 5 lines is enforced to ensure ANSI Nth octave filter compliance. That constraint gives a lower frequency cut-off for the operation. The chart in ***Figure 16-43*** shows the relationship between the resolution of the FFT Spectrum and the result of applying Nth Octave Synthesis to get the RTA Spectrum. The chart shows the resulting low frequency cut-off point.

Resampling

This function changes the sampling rate of a waveform up or down. The original sampling rate is displayed and the new sampling rate is entered by the user.

For example, when measuring Bluetooth devices that only support a sample rate of 8000 Hz, the measured waveform can be resampled to 44.1 kHz as shown in ***Figure 16-44***.

Since the original sample rate of the DUT may not be exactly 8 kHz, the result waveform should have **Frequency Shift** applied as well. (See below.)

This will correct for any sample rate error.

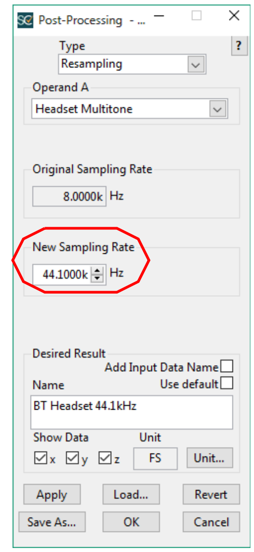

Figure 16-44: Resampling

Frequency Shift

This allows you to adjust the sample rate of a recorded time waveform to a specific sample rate.

This can be used to match the sample rate of the device under test to the sample rate of the stimulus used in SoundCheck which is useful for testing “Open Loop” devices.

When a stimulus is played back as a WAV file on a digital device, such as a Smart Speaker or MP3 player, the device may not play the file at its original sample rate. The frequencies present in the response are artificially offset, which corrupts the analysis. In order to perform the analysis, the response waveform must be adjusted so its frequency points match the stimulus waveform.

The example sequence, **C:\SoundCheck 17\Sequences\Microphones\Open Loop Microphone.sqc**, shows an example of how Frequency Shift is used.

In ***Figure 16-45***, Frequency Shift is used to change the measured Response to match the original stimulus.

After the Response Waveform is shifted, the Analysis Step will yield an accurate frequency response.

- Dual channel or Multitone analysis will then have the correct coherence function
- Frequency Shift can also be used with a Stweep stimulus and HarmonicTrak Analysis
- See ***Frequency Shift Rules on page 264***.

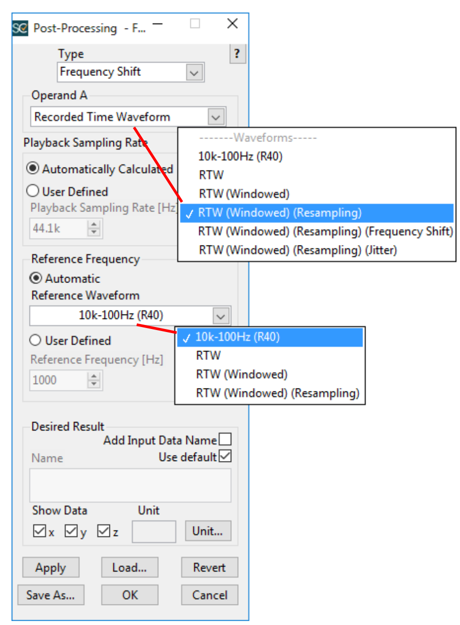

Figure 16-45: Frequency Shift

Waveform Selection

Operand A: Select the Input waveform that should be corrected, e.g.; The response waveform - RTW.

Playback Sampling Rate: This is the method used to estimate the playback sampling rate of Operand A

- **Automatically Calculated**: The playback sampling rate is calculated by comparing the frequency content of Operand A to the selected **Reference Frequency**
- **User Defined**: Use if the sampling rate of Waveform A is already known

Reference Frequency

- Reference Waveform: Select the Stimulus Waveform to be used. This should be the stimulus used in the Acquisition Step that generates the waveform in Operand A.
- Reference Frequency Selection
  - Automatic: It automatically chooses a frequency from the Reference Waveform for synchronization. This frequency is chosen from the first Stweep or Multitone in the Reference Waveform.
  - User Defined mode: Enter the Reference Frequency to use for synchronization. This frequency must be present in the Reference Waveform.

    The Reference Frequency entered is automatically coerced to the closest frequency found in the Reference Waveform.
  - A stationary tone with more cycles will yield a more accurate playback sampling rate. Longer tone duration will yield more cycles. Higher frequency tones will also yield more cycles without increasing the duration.

    e.g.: A 10 mSec, 1 kHz tone has 10 cycles, but a 10 mSec, 5 kHz tone has 50 cycles. The 5 kHz, 10 mSec tone would provide a more reliable frequency shift reference.

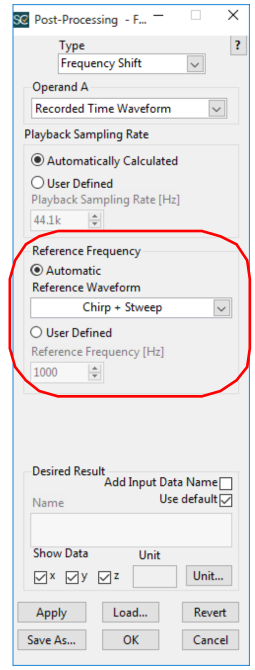

Figure 16-46: Reference Frequency

Desired Result

Enter a custom name for the new "Shifted Waveform" or select **Use Default**.

<u>Memory List - WFM Result</u>

The new waveform is the result of Operand A being shifted to match the sampling rate of the Reference Waveform. This new waveform should be used in subsequent Analysis steps.

Other Example Sequences

C:\SoundCheck 17\Sequences\Headphones & Headsets\Bluetooth Headset - Send.sqc

C:\SoundCheck 17\Sequences\Headphones & Headsets\Bluetooth Headset - Recieve.sqc

Frequency Shift Rules

- Frequency Shift only works with steady state tones; the stimulus needs to have a Stweep or Multitone segment
- Regardless of the **"Analyze - No"** selection in the Stimulus Editor, the reference frequency can only be from the first stweep or multitone in a compound stimulus
- Using WAV files for non-sine base signals requires that a steady state tone be prepended to the WAV stimulus in the Stimulus Editor, e.g.: A 150 mSec, 1 kHz tone, set to "**Analyze - No"**
- See the following example sequence included with SoundCheck:

  C:\SoundCheck 17\Sequences\Microphones\Open Loop Microphone.sqc

Resampling and Frequency Shift Uses

SoundCheck cannot analyze response signals that do not match the sample rate of the corresponding stimulus. Resampling and Frequency Shift post-processing steps allow you to synchronize waveforms that do not have the same sample rate or where generated and recorded by systems with different sample clock sources.

Resampling and Frequency Shift may be required when:

- Different audio interfaces operating at different sample rates are used to generate the stimulus and record the response waveforms: e.g. AmpConnect ISC (44.1 kHz) generates the stimulus and DCC-1448 (48 kHz) records it.
- The sample rate of the device under test does not match the SoundCheck sample rate, e.g.: Bluetooth Headsets that have a fixed sample rate of 8 kHz
- The audio interface(s) and DUT do not have a common clock source, which means that the resulting waveforms may not match in sample rate.
- In general, first apply a Resampling Step, then apply a Frequency Shift Step.

See ***Resampling on page 263*** and ***Frequency Shift on page 263***.

Time Domain Waveform Filter (optional module required)

Arbitrary Waveform Filter

The Waveform Filter post-processing operation allows you to choose a curve from the Memory List, use it as the frequency response and apply it to a waveform in the Memory List. The result is a new waveform that has its spectral content shaped by the selected curve. There is also an option for minimum phase and inverting the curve. This may be used, for example, for measuring the A-weighted peak acoustic pressure of a waveform. The A-weighting curve is first applied to the acoustic waveform via the waveform filtering post processing step, and then the peak value of the resulting waveform is measured. This method is used in the **IEEE 269** and **TIA 920** telephony standards.

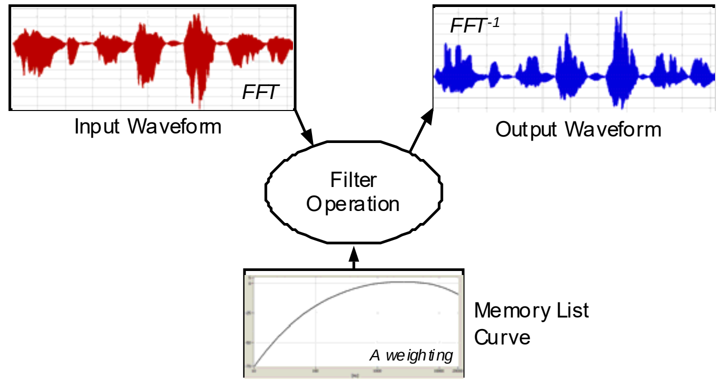

Figure 16-47: Waveform Filter Process

Requires optional module **2032 - Waveform Filter**.

The example in ***Figure 16-47*** shows the process of using an arbitrary waveform filter:

- A target waveform is selected from the Memory List: P50 Speech
- A curve from the Memory List is then applied: A Weighting curve
- The result is a new waveform that is shaped by the A weighting curve: P50 (Filtered-min phase)

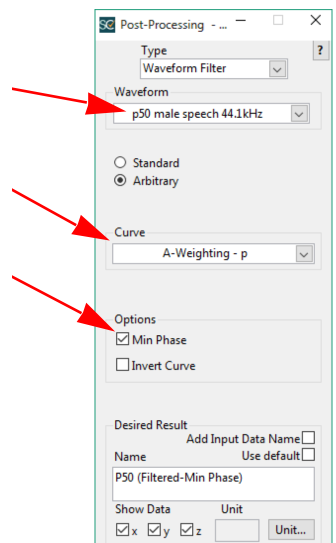

Figure 16-48: Waveform Filter

Filter Options:

- **Min Phase** - use when selected curve for filter has no phase information
- **Invert Curve** - useful for creating the inverse of the selected curve when using it as a correction curve
- The filtering process can be applied on the incoming waveform and can be used in a sequence in real-time
- This is the same algorithm used in the **Stimulus Step** for creating Equalization Curves
- Waveform filtering can also be useful for applying ERP to DRP correction

Standard Waveform Filter

In addition to the Arbitrary waveform filters (introduced in SoundCheck 11.0), a selection of Standard Waveform filters are also available, including Butterworth high-pass, low-pass, band-pass and band-stop filters. These standard filters are useful for conditioning stimulus and response waveforms when you are making time domain measurements, or for band pass analysis in the time domain (e.g., speech intelligibility or attack and release testing). The arbitrary filters are often used for telephony and hearing aid applications and can also be used when you need to listen to the processed time signal for subjective evaluation.

Both the Arbitrary and Standard waveform filters require Optional Module 2032: Waveform Filter.

When the post processing step is applied, the filtered waveform is output to the Memory List.

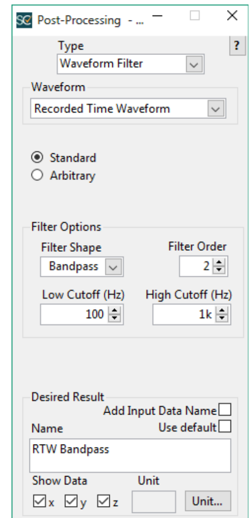

Figure 16-49: Standard Waveform Filter

Filter Options:

- Filter Shape - Select Lowpass, Highpass, Bandpass and Bandstop

  Lowpass and Highpass filters only show one **Cutoff** field.

  Bandpass and Bandstop show Low Cutoff and High Cutoff fields.

- Cutoff (Hz) – the corner frequency for lowpass and highpass filters (nominal 3 dB – down point)
- Low Cutoff (Hz) – the lower corner frequency for bandpass and bandstop filters (nominal 3 dB – down point)
- High Cutoff (Hz) - the upper corner frequency for bandpass and bandstop filters (nominal 3 dB – down point)
- Filter Order - Enter 1 to n as a value for the filter order

  The slope of the attenuation is proportional to the filter order: An order “n” will result in an attenuation rate of 20*n dB/ decade = 6*n dB/ octave. e.g.; A 5th order filter will have an attenuation slope of 100 dB/decade. In other words, the higher the order of the filter, the steeper the attenuation.

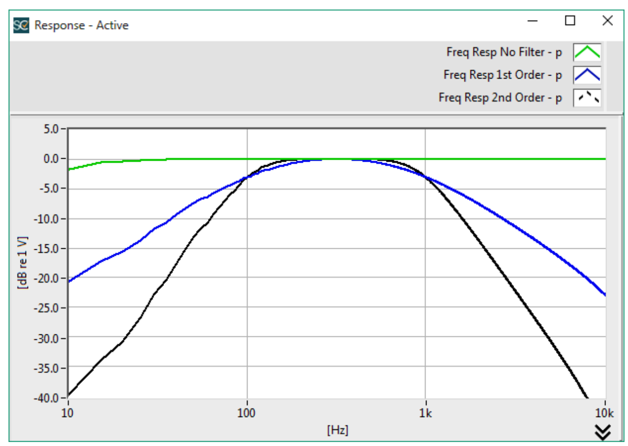

Figure 16-50: Comparison of Filters

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Post Processing Use Chart

Post P Type	Operation	Description	Example use	Operand A Axis X	Y	Z	Result Type	Post P Step Template
Unary	Change Sign	Polarity inverter for phase	Change polarity e.g. polarized mics which invert time signal	Y	Y	Y	Same as Operand A	none
Unary	Reciprocal Value	Calculates the Inverse (1/X) of	Inverting curves used for equalization and correction curves	Y	Y	N	Same as Operand A	Reciprocal
Unary	Absolute Value	Returns the positive valued magnitude of Operand A	Removes phase e.g. when adding harmonics to calculate total distortion	Y	Y	N	Same as Operand A	Absolute Value
Unary	Square	Returns the Square of Operand A (Operand A x Operand A)	e.g. first step in calculating RMS and removes phase	Y	Y	Y	Same as Operand A	none
Unary	Square Root	Returns the Square Root of Operand A	e.g. last step in RMS calculation	Y	Y	Y	Same as Operand A	none
Unary	Exp	Raises the Operand to the user input power (exponent)	Math operation	Y	Y	N	Same as Operand A	none
Unary	Ln	Returns the natural Logarithm of Operand A	Math operation	Y	Y	N	Same as Operand A	none
Unary	Unwrap Phase	Returns the Unwrapped phase of Operand A	Allows Unwrapped Phase to be treated as a separate curve from magnitude	Y	N	N	Curve	Unwrap Phase
Unary	Group Delay	Calculates the Group Delay (negative derivative of Phase) of Operand A	converts phase response to delay in seconds vs. frequency	Y	N	N	Curve	Group Delay
Unary	FFT	Calculates the FFT of a Waveform	Spectrum analysis	N	N	Y	Curve	FFT
Unary	Inv FFT	Returns a real-valued time signal (waveform) from a complex (Mag & Phase) spectrum or response	Time Domain Analysis	Y	N	N	Waveform	See FFT
Arithmetic	"+, -, x, /"	Allows you to perform basic math operations on data. Work In allows math to be done on the y axis dB or Power values as well as Linear values.	Block math operations are performed between two complex data sets (magnitude and phase vs. frequency) or waveforms. Multiplication or Division of dB to dB values is NOT allowed	Y	Y	Y	Same as Operand A	Curve Addition dB, Curve Addition, Curve division, Curve Subtraction dB, Curve Subtraction
Constant	"+, -, x, /"	Allows you to apply a single value constant from the Memory List or from the User Defined field to the selected axis of the data.	Multiplication or Division of dB to dB values is allowed	Y	Y	Y	Same as Operand A	Curve divided by constant, Curve minus constant dB, Curve multiplied by constant, Curve plus constant dB, Curve plus constant
Scalar (Statistics)	Average	Calculates the mean of the curve Y values over the search range, regardless of the units.	Can be used to determine the Sensitivity from a Fundamental curve at a single frequency or series of frequencies	Y	N	Y	Value	Sensitivity & HFA (OSPL 90 sequence)

Post P Type	Operation	Description	Example use	Operand A Axis			Result Type	Post P Step Template
				X	Y	Z		
Scalar (Statistics)	Power Sum of Curve	Calculates the square root of the sum of the squares of each Y value in a spectrum		Y	N	Y	Value	Power sum
Scalar (Statistics)	Power Sum of Waveform	Result Z axis shows power sum, Result Y axis shows power sum in dB		N	Y	Y	Value	Power sum
Scalar (Statistics)	Maximum / Minimum	Finds the max or min Y value of a curve in the specified search range and returns X, Y and Z values at that point.		Y	N	Y	Value	Maximum, Minimum
Scalar (Statistics)	Est. Resonance	Finds the resonance frequency, amplitude, and quality factor (Q) of a peak in a curve.		Y	N	N	Value	Est. Resonance
Scalar (Statistics)	Est. Notch	Finds the antiresonance frequency, amplitude, and quality factor of a dip or notch in a curve.		Y	N	N	Value	none
Scalar (Statistics)	IEEE TOLR	This calculates the average of the Frequency Response from Start Freq to Stop Freq according to IEEE Standard.	Transmitting Objective Loudness Rating	Y	N	N	Value	none
Scalar (Statistics)	IEEE ROLR	This calculates the average of the Frequency Response from Start Freq to Stop Freq according to IEEE Standard.	Receiving Objective Loudness Rating	Y	N	N	Value	none
Scalar (Statistics)	IEEE SOLR	This calculates the average of the Frequency Response from Start Freq to Stop Freq according to IEEE Standard.	Sidetone Objective Loudness Rating	Y	N	N	Value	none
Scalar (Statistics)	ITU-T SLR	Calculates the send loudness rating according to the ITU-T Rec. P79 from the Frequency Response.	For IEEE standards, the exponent can be user defined. For ITU standards, both the exponent and the weighting curve can be user defined.	Y	N	N	Value	none
Scalar (Statistics)	ITU-T RLR	Calculates the receive loudness rating according to the ITU-T Rec. P79 from the Frequency Response.	For ITU standards	Y	N	N	Value	none
Scalar (Statistics)	ITU-T RLR w/ Leak	Calculates the receive loudness rating according to the ITU-T Rec. P79 from the Frequency Response.	For ITU standards	Y	N	N	Value	none

Post P Type	Operation	Description	Example use	Operand A Axis X	Y	Z	Result Type	Post P Step Template
Scalar (Statistics)	ITU-T STMR	Calculates the sidetone loudness rating according to the ITU-T Rec. P79 from the Frequency Response.	For ITU standards	Y	N	N	Value	none
Scalar (Statistics)	Attack Time / Release Time	Used to test the time it takes for the signal from a DUT to stabilize after a sudden change in signal level.	Requires a time envelope waveform in dB.	N	N	Y	Value	Attack Time & Release time
Scalar (Statistics)	Zwicker Loudness	Calculates the overall perceived loudness of a sound.	Used to measure the perceived loudness of complex sounds, e.g., telephone ring tones.	Y	N	N	Curve & Value	Zwicker Loudness
Scalar (Statistics)	Active Speech Level	Used with a speech signal, this operation evaluates levels for only the parts of the waveform where speech is actually present.	Widely used in telephony applications, i.e.: testing to ITU-T P.56 05/93 Method B.	N	N	Y	Value	Active Speech Level
Intersection	same	Used to find the intersection points between two curves or waveforms, or between a curve/ waveform and a single point.		Y	Y	N	Value	Intersection
Smoothing	same	corresponds to a running average on a frequency axis with a window of the given width: 1/n octave.		Y	Y	N	Curve or	Smoothing
Directivity Index	same	Used to calculate the Directivity Index from a set of off axis response curves (Polar Response).		Y	N	N	Curve	Directivity Index
1/n Octave Synthesis	same	Used to transform an FFT Spectrum into an RTA Spectrum.		Y	N	N	Curve	none
Resolution	same	Curves acquired with HarmonicTrak, Heterodyne and Time Selective Response Analysis can be resampled according to a new resolution: 1/3rd Octave.		Y	N	N	Curve	Resolution
Windowing	same	Used to trim the selected curve according to the selected X axis range.		Y	Y	N	Curve or	Windowing
Frequency Shift	same	Allows you to match the sample rate of the device under test to the sample rate of the stimulus used in SoundCheck.	Commonly used to account for sample rate inaccuracies of portable audio devices. Apply after using a Resampling step.	N	N	Y	Waveform	Frequency Shift

Post P Type	Operation	Description	Example use	Operand A Axis			Result Type	Post P Step Template
				X	Y	Z		
Resampling	same	Changes the sampling rate of a waveform up or down.	Example: measuring Bluetooth devices with a sample rate of 8000 Hz, the waveform can be resampled to 44.1 kHz.	N	N	Y	Waveform	none
Waveform Filter	same	Allows you to choose a Memory List Curve, apply it to a Memory List Waveform and create a new waveform shaped by the spectral content of the selected curve.	Example: measuring the A-weighted peak acoustic pressure of a waveform. Used in the IEEE 269 and TIA 920 telephony standards	N	N	Y	Waveform	Waveform Filter
User Equation Module	same	Allows you to build an arithmetic equation using constants and the values generated by the sequence.	Complex Math operations are not supported. You will need to break the operation down to its arithmetic equivalent.	Y	Y	Y		none
Average Curve/WFM	same	The average curve (or wave-form) of a selected group of data in the Memory List	Make an average response curve from a group of response curves without using a Statistics step	Y	Y	Y		none

Message Step

The *Message Step* (**Ctrl+Shift+M**) is used in a test sequence to:

- Provide messages to the operator
- Enable the operator to input information needed to complete the test (e.g., a reference resistance value when measuring loudspeaker impedance)
- Communicate to devices through the PC's RS232 or IEEE-488 (GPIB) interface
- Messages Steps can be run from the Offline menu as well as from a sequence

To view and change the system's message settings, select **Messages** from the **Setup** drop-down list on the main SoundCheck® menu bar. The *Message Setup* dialog provides user prompts, control of external equipment and calls to other programs.

The four types of message allow you to interact with the Operator, a Digital I/O card, an external Interface connected to the system or Listen Hardware.

Listen Hardware Control Message

Message steps can be used to control the functions of the following Listen hardware products:

- AmpConnect 621, AmpConnect ISC, AudioConnect, SoundConnect 2 and DC Connect
- AudioConnect 4x4 is not selectable since there are no functions to control
- Also included is software control for the Portland Tool & Die BTC/BQC-4148/4149 Bluetooth interface
- Select **Listen Hardware** in the Message Editor and select from the **Device ID** drop-down list shown in ***Figure 18-2***

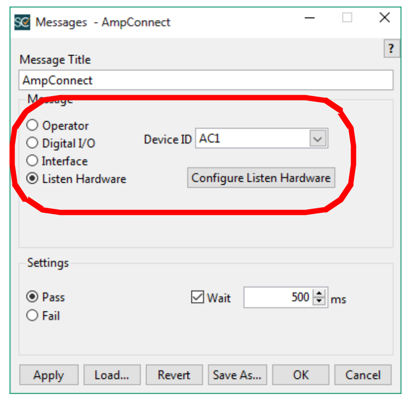


Figure: 18-1 Listen Hardware

Specific details on how a device is controlled through Message Steps is covered in the manuals for each device. The settings of the Listen Hardware Message Step are the same as the Startup Default found in the **Hardware Editor > Listen Hardware** page.

We recommend that you set the **Startup Default** for each piece of Listen Hardware. See ***Listen Hardware Page on page 70***. The alternative is to always use Listen Hardware Message Steps at the beginning of each sequence to configure the devices for use in the sequence. (AudioConnect requires either a Startup Default or a Message Step in order to be used properly in SoundCheck.)

Important! ***Turning Mic Bias on at the beginning of a sequence may require the use of "Wait Time" in the first panel of the AmpConnect Message Step. As an alternative, we recommend that Bias should be turned on in the Startup Default settings for Listen Hardware. See Listen Hardware Page on page 70.***

Common Controls

- **Apply** - This is used to send the setting to the device without having to run the sequence
- **Read Settings** - Click to load the current settings from the device
- **OK** - Exit the step editor. Save the sequence in order to save the step changes.
- **Cancel** - Close the step editor and discard changes

AudioConnect™ Message

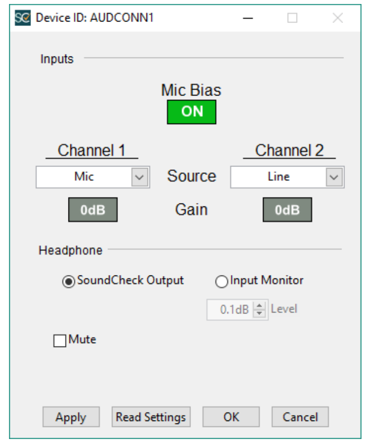


Figure: 18-2 AudioConnect Control

The following controls are available in an AudioConnect Message Step (details are in the AudioConnect manual):

- **Mic Bias -** Turns mic bias voltage on for both Mic Input channels
- **Source** - Set to Line or Mic independently for each channel
- **Gain** - Set to 0 dB or +20 dB independently for each channel

  This gain value is used in Calibration Configuration > **Auto Dev/ Auto Ch**. See ***Listen Hardware - Auto Device / Auto Channel on Page 84***.

Headphone

- **SoundCheck Output** - Sends the SoundCheck Stimulus to the headphone output. Gain is grayed out when SoundCheck Output is selected and fixed at 0.1 dB.
- **Input Monitor** - Allows you to monitor the Left Input of the internal audio interface

  Gain - Allows you to adjust the level of the Headphone Out. The default setting is 0 dB. (Available only when Input Monitor is selected.)
- **Mute** - Mute signal to headphone output

SoundConnect 2™ Message

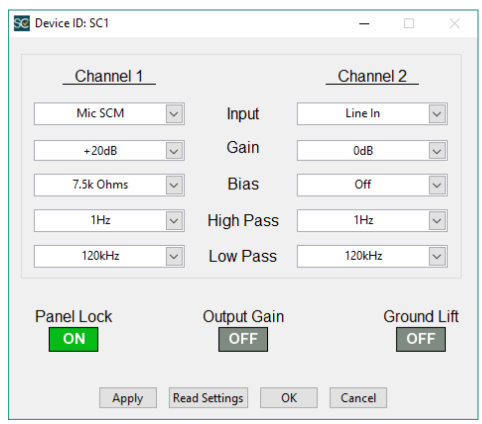


Figure: 18-3 SoundConnect 2 Control

The following controls are available in an SoundConnect Message Step (details are in the SoundConnect 2 manual):

- **Input** - Muted, Line In, Mic In, Mic SCM, Mic IEPE and Lemo
- **Gain** - Select: -20 dB, -10 dB, 0 dB, +10 dB, +20 dB, +30 dB, +40 dB

  This gain value is used in Calibration Configuration > **Auto Dev/Auto Ch**. See ***Listen Hardware - Auto Device / Auto Channel on Page 84***.
- **Bias** - No Bias, 7.5 kOhm (for SCM mics), 2.2 kOhm (typically used for testing electret capsules)
- **High Pass** - 1 Hz, 10 Hz, 20 Hz, 100 Hz
- **Low Pass** - 22.4 kHz, 120 kHz

Overall settings for unit:

- **Panel Lock** - When On the front panel buttons are disabled
- **Output Gain** - When on the Output Gain of channels 1 and 2 are increased by 6 dB. (Refer to the SoundConnect 2 manual for an explanation of this feature.)
- **Ground Lift** - Off = Chassis Ground (default). Allows you to interrupt the connection between the Line In/Out Grounds and the Chassis Ground. Only select **Lift** when you are trying to resolve noise issues due to a Ground Loop.

AmpConnect 621™ Message

AmpConnect 621 is a USB multichannel test interface for use with SoundCheck 18.1 and later.

- 6 BNC Inputs - Unbalanced Microphone/Line Inputs with selectable Voltage and IEPE bias
- 2 BNC Outputs - Unbalanced Line outputs
- 1 Mono power amplifier with switchable A/B binding post outputs
- Dedicated Impedance measurement interface
- 8 channel Digital I/O (See ***Digital I/O Message on page 290***)
- Supports sample rates from 44.1 kHz to 192 kHz
- ASIO drivers (also compatible with WASAPI) and compatible with Core Audio drivers in macOS

AmpConnect 621 Message Configuration

The settings in **Configure Startup Default** and the **AmpConnect 621 Message Step** are identical.

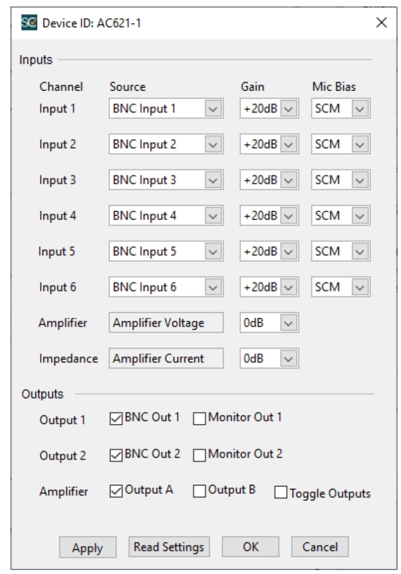


Figure 18-4: Startup Default

Signal Routing - Inputs

- This allows you to route the following to the audio interface inputs 1 through 6:
  - BNC Input 1 through 6
  - Monitor Out 1 or 2 (Loop back for self test) [Monitor Out 1 or 2 must be enabled in the Output section]
  - Amplifier (internal loop back for amp calibration)
  - Impedance (internal current sense circuit)
- **Gain**: -20 dB, 0 dB, +10 dB, +20 dB, +30 dB
- **Mic Bias**: Off, SCM and IEPE

SoundCheck Acquisition Step Gain Field

Important! ***Switching Listen Hardware from Maximum Gain to Minimum Gain in the Acquisition Step is not recommended. This does not allow the input gain circuit sufficient time to stabilize. If you need to switch from Max Gain to 0 or Minimum Gain we recommend that you use a Listen Hardware Message step with a 500 mSec wait time to allow for settling.***

Signal Routing - Outputs

- **BNC Out 1 and 2** - Routes the Output of the internal audio interface to the BNC outputs.
- **Monitor Out 1 and 2 -** Audio interface outputs are internally available so that Monitor In can be selected in the Input Source section to make an internal loop back.
  - **Monitor Out 1** can be routed to the odd number Input channels: 1, 3 and 5
  - **Monitor Out 2** can be routed to the even number Input channels: 2, 4 and 6
  - **Gain** and **Mic Bias** are automatically off (grayed out) when **Monitor Out** is selected
- **Amplifier -** Select Output A, Output B, A+B or Off. You can also Toggle Amplifier Outputs.

Toggle Amplifier Outputs

The Toggle control can be used to switch the Amplifier Output as a test sequence runs in **Continuous** or Loop operation. This can be used to test a loudspeaker on one test fixture while setting up a different speaker on another fixture. (The fixtures and test microphones should be identical.)

Toggle Outputs: Switches so that the first pass of the sequence uses Amplifier Output A and the second pass uses Amplifier Output B.

Microphone Bias "On Time"

If you turn the microphone bias voltage on during the run of a sequence, you may need to set a "Wait" period before proceeding to an Acquisition Step. The microphone connected to the BNC input may require 2 seconds (approx) to fully turn on after voltage is applied. You must allow for the microphone turn on time before an Acquisition Step using that mic is run in the sequence. See ***Sequence Editor > Step Configuration*** in the SoundCheck manual for instructions on setting "Wait time" in a Message Step.

Digital I/O Control

All Digital I/O functions for AmpConnect 621 are available in Digital I/O Messages Steps.

See ***Digital I/O Message on page 290***.

AmpConnect ISC™ Message

AmpConnect ISC is fully controllable via SoundCheck which means that adjustments of parameters such as gain can be included in test sequences. All Front Panel settings can be changed via USB during the run of a test sequence. An eight-bit digital I/O port provides digital control and/or status monitoring of external devices for operator feedback, test fixture control, etc. Digital I/O is only accessed by USB through SoundCheck. The following example shows how to use a Message Step in SoundCheck to control AmpConnect ISC.

Important! As of SoundCheck 13, after installing SoundCheck you cannot use AmpConnect ISC with versions prior to SoundCheck 13, unless you manually switch the drivers in Windows Device Manger. See the AmpConnect ISC manual for a step by step driver rollback procedure.

Note: In SoundCheck ONE, you do not have access to creating a new Message Step. Only the settings in the existing Message Step can be changed.

Configure AmpConnect

Configure Listen Hardware to set the front panel settings of AmpConnect. The standard front panel controls are explained in the ***Front Panel Functions*** chapter of the AmpConnect ISC Manual.

Inputs

- **Gain - Select:** -20 dB, -10 dB, 0 dB, +10 dB, +20 dB, +30 dB, +40 dB
- **Bias - Select:** None, Voltage (for SCM Mic) and IEPE

Signal Routing - Input

- **Select:** Quiet, Reference, DUT, Impedance (Z-High/Low), Amp and 1 V Sine

Signal Routing - Amplifier

- **Select:** Output A or Output B
- Toggle Amplifier Output - See ***Toggle Outputs on page 278***.

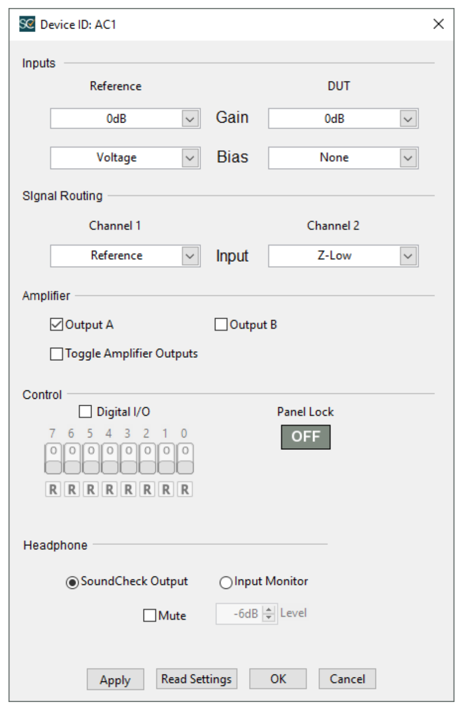


Figure 18-5: AmpConnect ISC Panel

Control

Digital I/O

> ***Note:*** ***Enable Check Box:*** *As of SoundCheck 18, AmpConnect ISC Digital I/O can be controlled by a separate Digital I/O Message Step in a sequence. See* ***Digital I/O Message on page 290******.***

- Digital I/O control is available in this step for backward compatibility. You must select the Digtial I/O check box to enable it in the step.
- Bit 0 through 7: Shows the state that the Bit expects to see when set to Read. If the actual state of the input agrees with the setting, the step result will be “Pass“. If it does not agree, the result will be “Fail“.
- When set to write, this is the state that the Bit will change to when the Message Step runs in the sequence
- **Important!**: Unused Bits must be set to **Write** with the switch **Off** or low
- AmpConnect ISC must be connected to a SoundCheck system via USB to use Digital I/O

Panel Lock

- When **On (Green)** the front panel buttons are disabled

Toggle Outputs

The Toggle control can be used to switch the Amplifier Output as a test sequence runs in **Continuous** or **Loop** operation. This can be used to test a loudspeaker on one test fixture while setting up a different speaker on another fixture. (The fixtures and test microphones should be identical.)

- Toggle Outputs: Switches so that the first pass of the sequence uses Amplifier Output A and the second pass uses Amplifier Output B.

Headphone Monitoring

This allows you to monitor the input and output signal on headphones, during the run of a test.

- Input Monitor - Listen to signal at input of SoundCheck audio interface
- Input Monitor Gain - Headphone output level can be adjusted: -60 dB to +4 dB
- SoundCheck Output - Listen to output of SoundCheck audio interface
- SoundCheck Output Gain - Fixed at -6.1 dB. This accounts for added gain in the headphone amp and adjusts the actual headphone output to 0 dB (Unity).
- Mute - Silence Headphone output
- A **Message Step** must be added to the beginning of a sequence to set the AmpConnect ISC Headphone output to it’s maximum level. Inserting this step turns off the **27 dB output pad** on the Headphone Output (AmpConnect ISC default).

Dynamic headphones typically need very little voltage, e.g., 100 mV to produce high sound pressure levels. A dedicated headphone amplifier, like the AmpConnect ISC headphone amplifier, is recommended to limit sound pressure levels. A normal power amplifier can output a voltage level that will damage most headphones if used at full gain. The maximum output voltage of the Headphone Output is limited to 1.5 Vp (1.06 V rms)

WARNING! When monitoring signals with headphones: **REMOVE HEADPHONES** before making any changes to AmpConnect or SoundCheck settings to prevent hearing damage due to high sound pressure levels.

Sequence Example for AmpConnect ISC

The Loudspeaker Impedance Test Sequence will require the settings as shown in ***Figure 18-6***. (Refer to the ***Single Loudspeaker Test*** section of the SoundCheck ONE chapter for wiring suggestions.)

- **Panel Lock** is OFF. This enables the front panel controls.

  Input Section

- **Reference In** Bias is set to Voltage
- **DUT In** Bias is set to None
- **Gain** set to 0 dB for both

  Signal Routing

- **Reference Mic** is routed to audio interface Channel 1
- **Amplifier** output is set to Output A
- **Z-Low** is routed to audio interface Channel 2. This is used for the Impedance measurement input in the SoundCheck sequence.

  Control

- **Digital I/O** is disabled

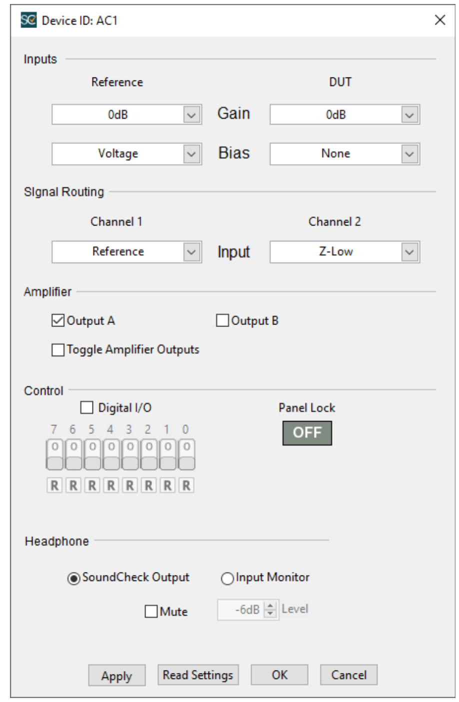


Figure 18-6: AmpConnect ISC Settings

Click **Apply** to send these settings to AmpConnect. This allows you to check the settings without having to run the sequence. Verify the settings on the front panel of the AmpConnect.

When the sequence runs, AmpConnect ISC will automatically change to these settings.

Note: Overload of the Front Panel **Input Level** or **Output Level** indicators currently requires a hard reset of AmpConnect ISC. Turn the AmpConnect ISC power switch off and then back on to reset.

Custom Step Conversion

SoundCheck 9 users controlling AmpConnect with the Custom Step will need to use the following procedure to re-create the step in SoundCheck 18.1 .

- Open the AmpConnect Custom Step(s) in SoundCheck 9
- Write down the settings for the step. (The AmpConnect ISC manual has a blank front panel drawing which makes notation of settings easier.)
- In SoundCheck 18.1, edit the Hardware Configuration for AmpConnect. See ***AmpConnect ISC on page 71***
- Make a new Message Step in SoundCheck 18.1 and enter the settings from the front panel notation
- SoundCheck 18.1 will not open AmpConnect Custom Steps

DC Connect™ Message

The following controls are available in a DC Connect Message Step (details are in the DC Connect manual):

- **Output Mode** - Voltage or Current
- **Control** - USB or Analog
- **Polarity** - Pos or Neg
- **Max I** - Select maximum current range for measurement
- **Voltage Level** - Set output voltage level
- **DC Measured** - Shows the DC value measured by DC Connect

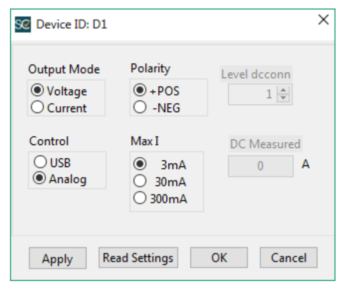


Figure: 18-7 DC Connect Control

Portland Tool & Die BTC/BQC-4148/4149 Message

Initial communication with the Portland Tool & Die BTC/BQC instruments is setup in Hardware Editor > Listen Hardware Tab > **Startup Default** settings as shown in ***Figure 18-8***.

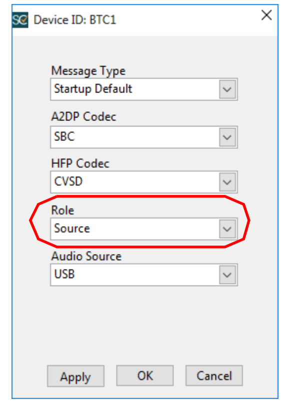


Figure: 18-8 BTC Startup Default

This allows you to send settings to the BTC/BQC when SoundCheck opens. The **Role** of the BTC-414x can only be set in the **Startup Default**. The BQC interfaces only operate in **Source Role.**

In this case, the BTC-414x is set to **Source** as shown in ***Figure 18-8***.

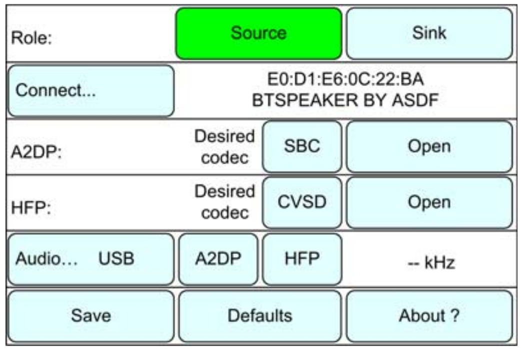


Figure: 18-9 BTC-414x Front Panel

Portland Tool & Die BTC/BQC instruments can also be controlled by selecting **Listen Hardware** in a Message Step. Click the **Device ID** drop-down list to select the interface you want to use.

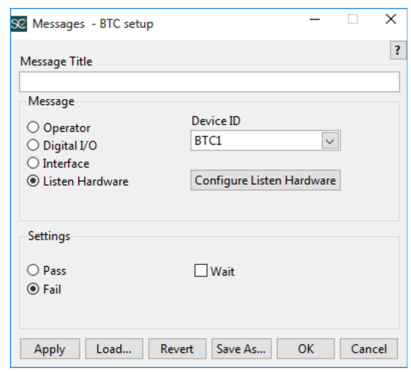


Figure: 18-10 Select Device ID

As of SoundCheck 17, the new **aptX HD codec** for high resolution Bluetooth testing is supported. This requires the BTC-4149 Bluetooth Interface, which is fully integrated with, and controlled by, SoundCheck.

BTC - BQC Functional Differences

Some functions of the BTC interface are not available on the BQC interface as noted below.

	BTC-4149	**BQC-4149**
Test Bluetooth Source Devices (phones and dongles)	Yes	No
User Interface	Control via SoundCheck USB and front panel touchscreen	Control via SoundCheck Message Steps and PC control app
Digital Audio Interface	SPDIF / USB	USB
Bluetooth Specification	Bluetooth 3.0 compliant	
Bluetooth Modes	Source, Sink	Source only
Codecs:		
A2DP	SBC, aptX, aptX-HD	SBC only

Note: BTC and BQC devices have different Device IDs. This means that Message Steps setup for BTC must be manually reassigned to control a BQC.

The following Message Type selections for the BTC/BQC-414x are presented in the order they typically appear in a sequence. The settings of each step are applied to the BTC/BQC interface when the step is run in a sequence or when the Apply button is pressed. (Details of use are in the BTC-4148/4149 manual.)

Codec Selection Message Step

When the message is set to Codec Selection, the field selections change to those shown in *Figure 18-11*.

Select an item from both profile drop downs.

- **A2DP Profile** - Select SBC, aptX or aptX-HD
  (aptX HD is only available when using the BTC-4149)
- **HFP Profile** - Select CVSD or mSBC

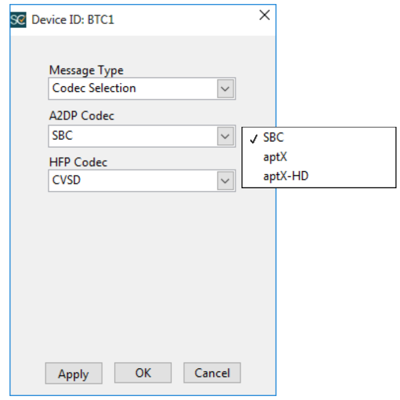


Figure: 18-11 Codec Selection

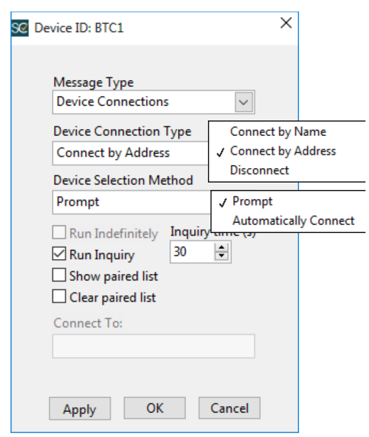
Device Connections Message Step

Set **Message Type** to **Device Connections**. This allows you to choose which device to connect to.

<u>*Device Connection Type*</u>

- Connect by Name - Uses the “Friendly Name” of the device
- Connect by Address - Uses the Bluetooth address of the device
- Disconnect - This disables the audio and closes both profiles. You can also select the *“Clear the paired list”* check box at the same time.

  (These options are typically used at the end of the sequence to make SoundCheck ready to test a different device.)

<table>
<tr><td colspan="2" rowspan="2"></td><td colspan="2">Device Connection Type</td><td rowspan="4">
Figure: 18-12 Device Connections</td></tr>
<tr><td>Connect by Name</td><td>Connect by Address</td></tr>
<tr><td rowspan="2">Device Connection Method</td><td>Automatic</td><td>BTC/BQC will scan for available devices until finding a match to the friendly name given in the Connect To: field.

If Run Indefinitely is checked, this operation will never timeout, otherwise the operation will time-out after the time in seconds specified by Inquiry Time.</td><td>No scanning will occur and the address specified in the Connect To: field will be used to open audio channels.</td></tr>
<tr><td>Prompt</td><td>Not Supported

Connect To: field is grayed out</td><td>A dialog prompt will be shown allowing an operator to select a specific device.

If Run Inquiry is selected, BTC/BQC will scan for available devices.

If Show Paired List is selected, the history of devices previously paired will be shown.</td></tr>
</table>

Device Selection Method

Prompt:

- Run Indefinitely - BTC-414x will scan for BT devices with no time out (not recommended)
- Run Inquiry - BTC-414x will scan for any available Bluetooth devices
- Inquiry time - Set time limit to search for devices
- Show paired list - Shows history of devices previously paired with BTC-4148/4149
- Clear paired list - Removes devices from history list. Usually done at the end of sequence.

Automatically Connect:

- Fill in the field - **Connect To:** (This field is grayed out when **Prompt** is selected.)

See ***Example of Prompt on page 283***.

Note: You can also enter the MAC address for a device by using a TCP/IP command. See ***on page 509***.

Example of Prompt

With the Message Step in ***Figure 18-12*** set to **Prompt**, when the step runs a window opens to allow you to select devices that have been found. See ***Figure 18-13***.

Name or Address Field

- This is populated by selecting an item from the **Available Devices** field
- You can also type the address in manually or scan it in using a bar code reader
- The Refresh Icon allows you to run the inquiry again. **Inquiry time** is set in the Message Step as shown in ***Figure 18-12***.

Available Devices

- Shows all devices found during the scan

Paired Devices field

- This will be filled in once the pairing process has been completed in the next step

  See ***Audio Connections Message Step*** below.

Click **OK** to close the window and continue with the sequence.

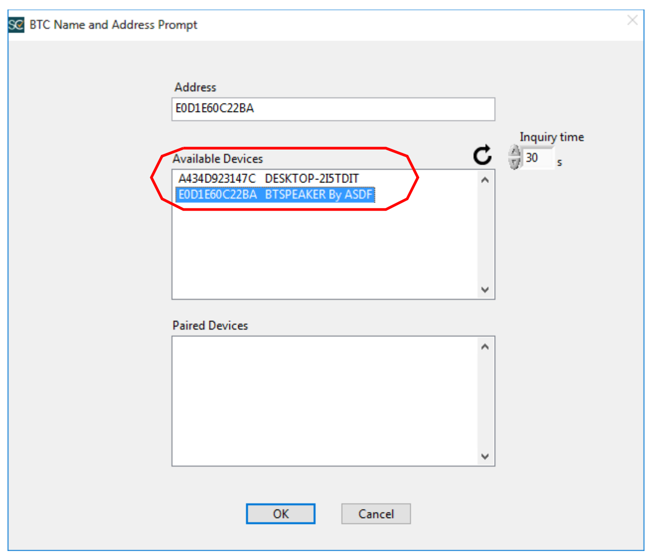


Figure: 18-13 Prompt Example

BTC-4148/4149 Front Panel Update
Note that the BTC front panel will change to show the result of the Device Connection step. See ***Figure 18-14***.

- Connect button - changes to **Close All**
- Connected device address - E0:D1:E6:OC:22:BA
- Connected device name - BTSPEAKER BY ASDF
- Selected codec - button turns green and shows codec, **Close** button becomes active
- A2DP status - button turns green
- Current Sample Rate - 48 kHz

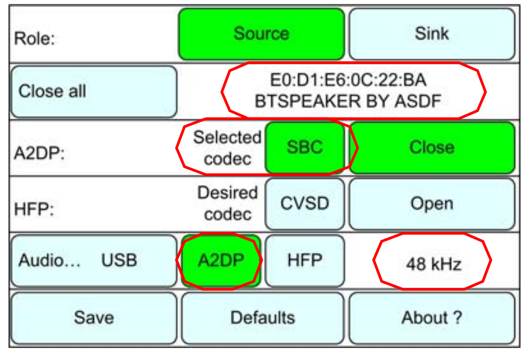


Figure: 18-14 BTC Front Panel Update

Audio Connections Message Step

A2DP and HFP Connections
From the two connection type drop-down lists select the profiles you would like to use.

- Connected - Connect using the selected profile
- No change - Allows you to leave BTC/BQC-414x in the state set in a previous step

Audio Channel
This is where the profile is enabled.

- Select A2DP, HFP or Close audio channel

The pairing process starts when this step is Applied or run in the sequence.

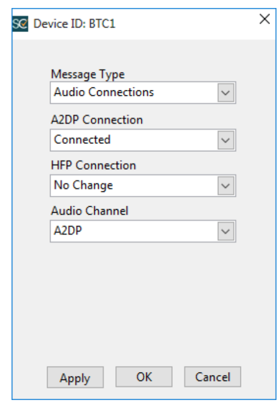


Figure: 18-15 Audio Connections

Portland Tool & Die DCC-1448/9 and PQC Control

DCC-1448A or PQC-3048 settings can be controlled in a Message Step.

Startup Defaults

When SoundCheck starts, Startup Default settings are written to the DCC/PQC device. This ensures that the device is ready to take measurements from a specified setting.

Message Step

DCC/PQC Message Steps can be used to change the setup of the test interface during the run of a sequence or as a standalone off line step.

Quick Setup

The DCC/PQC control allows for faster setup of the measurement interfaces with SoundCheck as well as greater consistency in measurements as the device settings can be built into the test sequence. This allows for a seamless transition between R&D and production testing. Sequences from product development using the DCC can easily be used for the production line with the PQC.

DCC-1448/9 Settings

<u>*Clock*</u>

- State - In, Out, Hi and Low
- Frequency (Hz) - User defined
- Level (Vp) - User defined, 1.4 to 5.5 Vp

<u>*Data In*</u>

- State - On/Off
- Coupling Mode - AC or DC
- Input Capacitance (pF) - 5, 50, 100, 200, 400 pF

<u>*Power Supply*</u>

- State - On/Off
- Level (Vp) - 0.1 to 5.5 Vp

<u>*PDM*</u>

- Decimation Rate - 1/32 or 1/64

<u>*PSR*</u>

- State - On/Off
- Waveform - Sine or Square
- Frequency (Hz) - User defined, 20 to 20 kHz
- Level (Vp) - User defined, 0 to 1 Vp

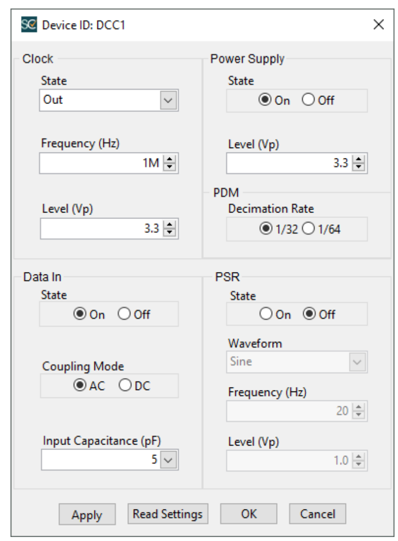


Figure: 18-16 DCC-1448/9 Control

PQC-3048 Settings

The available settings for the PQC-3048 are the same as the DCC-1448/9 interface.

- Clock - Frequency only
- Power Supply - Level only
- PDM - Decimation Rate - 1/32 or 1/64

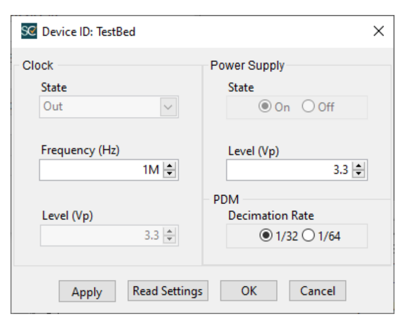


Figure: 18-17 PQC-3048 Control

Operator Message - Dialog

Display messages or instructions on the computer screen for the operator to read (e.g., *test signal is not present* or *the DUT needs to be placed in a test fixture*). You can send a text message to the operator during the sequence, or poll the user for information during the sequence (such as temperature or humidity conditions).

Pass/Fail

Sets the Pass or Fail verdict the message is associated with, e.g., No Signal Detected – Fail.

Wait

Control length of time (in milliseconds) the Message Step waits before finishing. This can be used to ensure commands or devices settle before continuing to measure.

Format

Text Formatting is available Operator Message Steps. This allows you to change the settings for the text that will be displayed so the message is easier to read.

Default Value

By setting the Default Value to "Yes (F2)" in the Message Step, the Key Focus is to the Yes button by default. This means that the Yes button will be "Highlighted" each time the step runs in the sequence. The operator can then click the Enter key to answer Yes. (As well as the F2 key or clicking on Yes in the message.)

Note: The Message Step must be configured to "Display step when run" in order for the Dialog to display in the sequence.

Note: Operator messages can be displayed in local languages based on the Windows operating system in use on the machine running SoundCheck. See ***Display Local Language Characters on page 288***.

The Dialog Message allows you to select one of two conditions: yes or no. This can be used to determine functionality of the test system, (*Was the test signal audible?*) or to determine how the sequence will proceed (*Do you want to save this data?*).

Figure: 18-18 Operator Message - Dialog, shows an example of a Message that can be used to prompt the operator to enter a Yes/No answer regarding Visual Inspection.

Format Text
This allows you to set the font, font size, font color and attributes.

The Dialog Window that the operator sees will resize according to the font size chosen for the text. Only one font format can be set in each message step. In other words, you cannot mix text colors or font sizes.

Memory List Item
This result will show up on the Memory List.

The name of the result is the **Name of the Message Step**.

The Apply Button in the message editor allows you to test the message to see if it is displayed as required. This also allows you to test the communication of the message step with external devices that are connected via Serial - RS232 or GPIB.

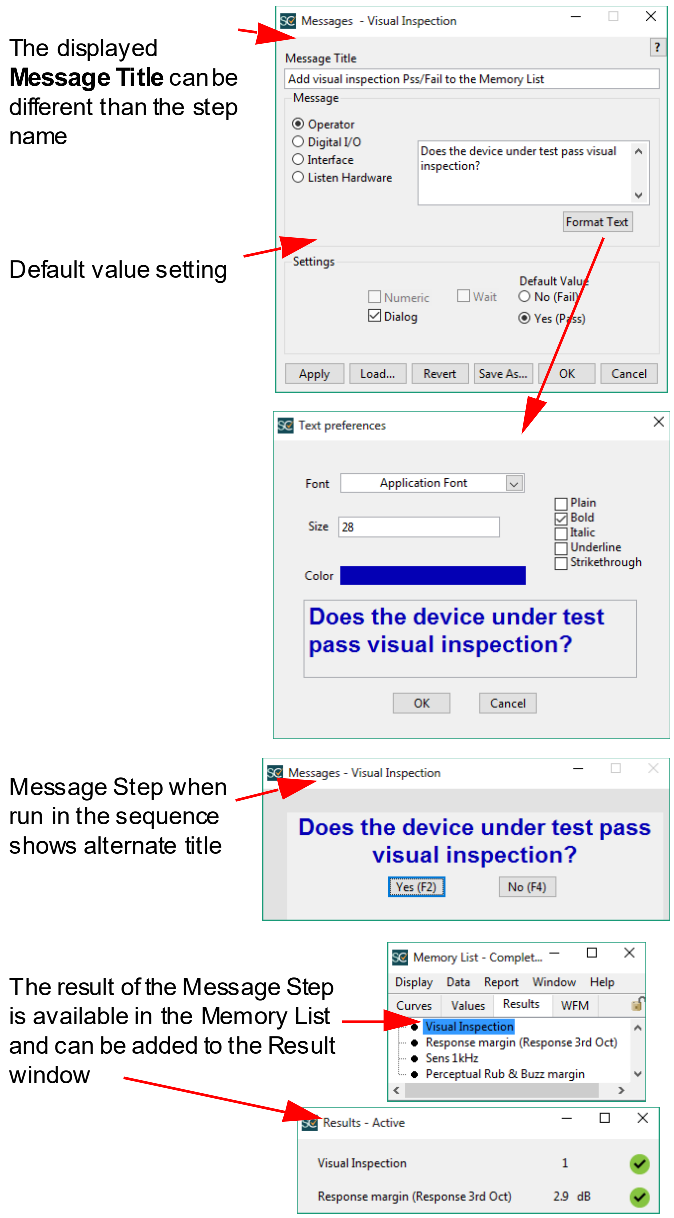


Figure: 18-18 Operator Message - Dialog

You can also Right-click a Message step in the Sequence Editor to test the action of the step without having to run the sequence.

The Default Value setting is available when the Operator Message Step is set to Dialog or Numeric.

Note: If the step is configured to “not display when run”, the default value for either Dialog (T/F) or Numeric (#), is sent to the Memory list for that step's result/value.

Display Local Language Characters

In order to display Local Language Characters in a SoundCheck Message Step or Text Display, the following changes must be made to your Windows operating system. In this example Simplified Chinese is selected.

Note: These instructions apply to Windows 7 operating systems. The instructions may vary in other versions of Windows.

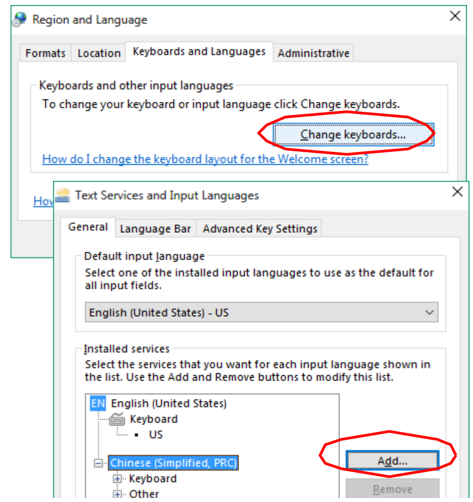
---

1. Install the Chinese (Simplified) Language Pack via Windows Updates. For detailed instructions see:

   https://support.microsoft.com/en-us/help/14236/language-packs

2. Go to Control Panel > Clock, Language and Region. Select Keyboards and other input methods

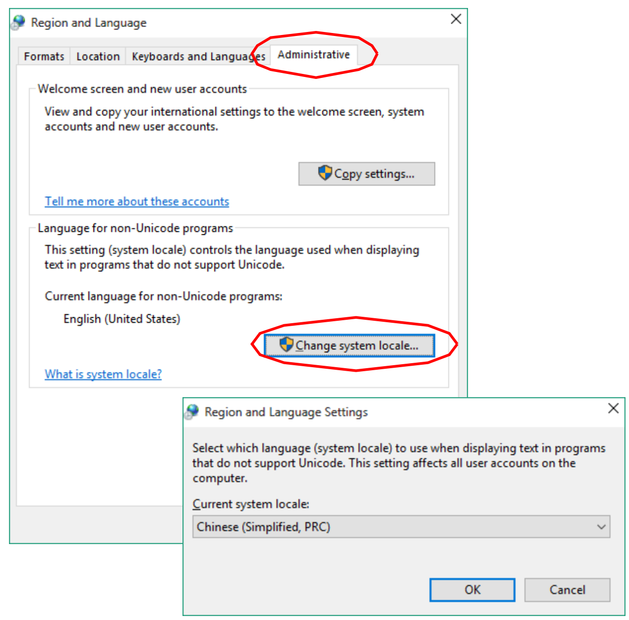
3. Click the Change Keyboards button and add Chinese (Simplified, PRC)



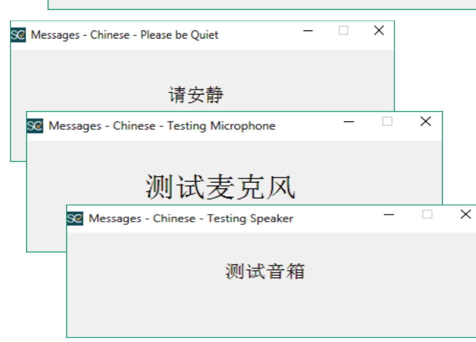
4. Select the Administrative Tab, click the Change system locale button and select **Chinese (Simplified, PRC)**

5. Click **OK** to exit all editors

6. You must restart your computer for these changes to take effect



7. You can now enter and display Simplified Chinese characters in SoundCheck Message Steps and Text Displays.



Numeric Message

When the Operator Message is set to Numeric, a number can be entered by the operator for use as a value in the sequence.

This prompts the operator when a numeric value needs to be entered in a sequence. An example of a Message Step setup to request a numeric during the sequence is shown in ***Figure 18-19***.

The example shown in ***Figure 18-19***, shows how the Numeric Message prompts the operator for a Stimulus Level. Since the name of the Message Step is “**Stimulus Level**“, an item is added to the Memory List called **Stimulus Level**. This item can then be used in Stimulus Step. This will allow the operator to change the stimulus level on each run of the sequence.

When Apply is selected in the Message Editor, the value is sent to the Memory List.

- The specific axis of a numeric value to be stored can be selected, e.g., Y axis only. Other axes will have NAN as a value and will be shown this way when selected in a Display Table.
- Units are shown next to numeric inputs when specified.
- Preload Stimulus must be Off in the Sequence Configuration. See ***Preload Stimulus vs Memory List Selection on page 445***.

See ***Units on page 109*** for more information on using the *Units* dialog box.

The Memory List item is named the same as the Message Step name

The message as it is displayed during the run of the sequence

In the Stimulus Editor, Right-click the level field and select **Memory List Selection**.

Select “**Stimulus Level**” from the Memory List.

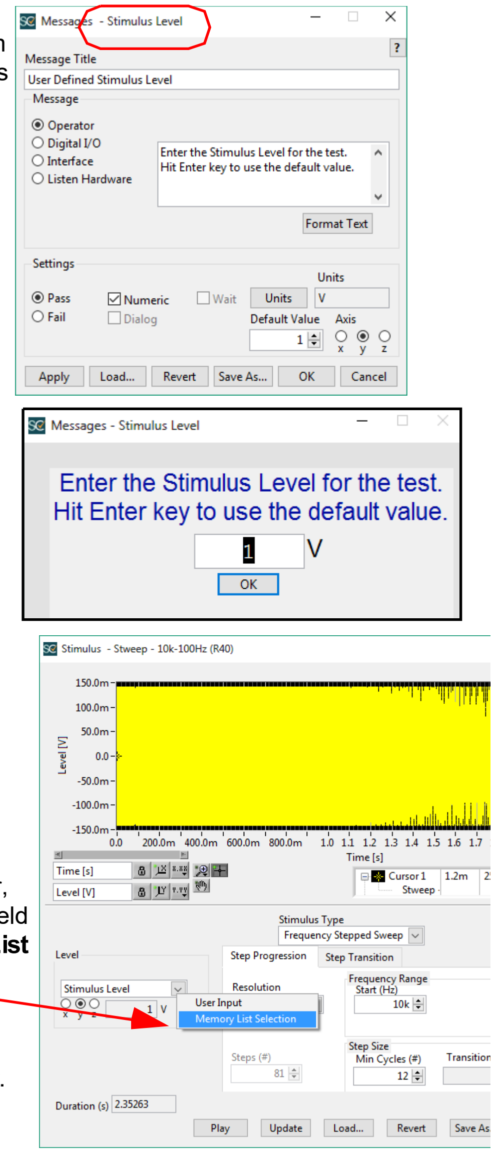


Figure: 18-19 Operator Message - Numeric

Digital I/O Message

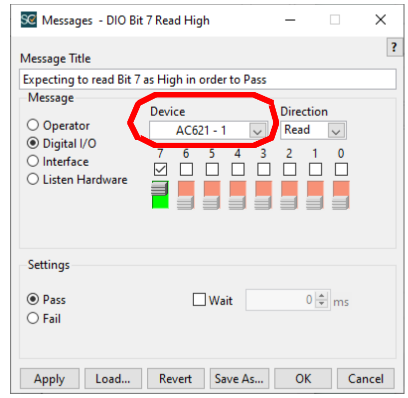


Figure 18-20: Digital I/O AC621

SoundCheck can interact with PLC controls on a production line by using Digital I/O Message steps in a sequence. These steps control a Digital I/O device such as the NI-65xx DAQmx series from National Instruments. The step allows for 8 Bits of digital I/O that can be written and read directly in SoundCheck. This allows SoundCheck to accomplish such tasks as:

- Automatically start a test on a device once it has been loaded into a test fixture
- Used to control process indicator lights, relay boards and detect the state of switches, e.g.: Confirm that a test chamber is closed by verifying the state of a door switch
- Communicate with PLCs in automated production lines
- Receive TTL signals from other test equipment

When using a National Instruments Digital I/O device such as the NI-6503, NI-6528, NI USB-6501, etc, **NI DAQ Digital I/O** must be enabled in the SoundCheck Hardware Editor External Tab.

See ***NI DAQ Digital I/O on page 76***.

Controls are the same for Listen hardware with Digital I/O (AmpConnect 621 / ISC) and NI DAQ.

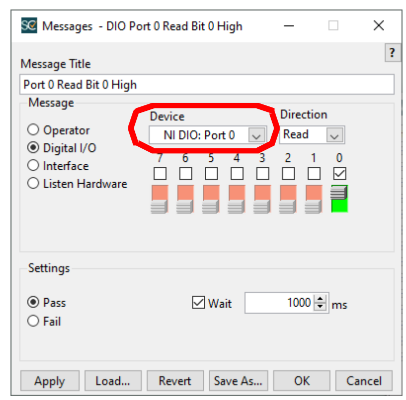


Figure 18-21: Digital I/O NI DAQ

- Select the **Digital I/O** radio button
- **Device** - Select the device ID from the drop-down menu
- **Direction** - Set the communication direction for the step: Read or Write

---

Note: On some devices the function of ports may be fixed: e.g., Ports 0, 1 are Read and Ports 2, 3 are Write. The Message Step settings much match the fixed state of the device ports.

- **Enable Check Boxes** - Select which Bits are to be used in the step

Digital I/O States

- Each Bit in a Message Step can be enabled or disabled using the **Check Box** above the Bit Switch. This allows you to ignore unnecessary bit settings when determining the Pass/Fail state of the Digital I/O Message. Unused Bits are grayed out.
- The expected state of the Bit is indicated by the switches:
  - In the Digital I/O Message Step, you click on the **Switches** to set the state. See ***Digital I/O Rules on page 291***.
  - High - On - Green
  - Low - Off - Red

Note: Individual Digital I/O Message Steps are required in a SoundCheck sequence to perform various Digital I/O operations. Each Digital I/O Message Step can control individual Bits as well as controlling all Bits.

Read vs Write

- Read: A High bit means that the step expects the selected bit(s) to see a "High State" at the Digital I/O port(s) to produce a PASS condition when the step is run
- Write: A High bit means that the step will change the state of the selected bit to High

The bit settings are applied when the Message Step runs in the sequence. Corresponding Message Steps with opposite Bit States may be required, e.g.: switch **Bit 0 High** at the beginning of the sequence and the switch **Bit 0 Low** at the end. Step Configuration is used to setup conditional branching allowing the Message Steps to control the flow of the sequence. See ***Configure Step on page 446***.

Pass/Fail

Set the Pass/Fail condition the message should show when it runs in the sequence. This can be used in Conditional Branching. See ***Configure Step on page 446*** and ***Conditional Branching Rules - Sequence Editor on page 448***.

Wait

You can set an amount of time for the step to pause before proceeding to the next step in the sequence. This may be needed to allow for settling time or operation time of external devices.

Digital I/O Rules

- Digital I/O Message Steps can be set to Read (Input) or Write (Output)
- Best practice: Use a Digital I/O Message Step at the top of a sequence to "initialize" the Bits to a desired starting state
- Bits are at ground when set Low, Message Step switch Off
- Bits are at +5 V when set High, Message Step switch On
- When set to Read (Input), enabled Bits set to High (On) are pulled high through an internal 100K resistor
  - The setting of the Bit indicates the state that you expect to read. That state is compared to the actual state of the physical Bit line to produce the PASS/FAIL result for the step.
  - The PASS/FAIL state of a Digital I/O Message Step set to **Read** is determined at the time the step is executed in the sequence.
  - If the state of any enabled Bit disagrees with the actual state of the selected Bit line, the step will issue a "FAIL" verdict when the step occurs during sequence run. This verdict is then used by the **Step Configuration** to perform Conditional Branching or sequence logic. See ***Step Configuration*** in the main SoundCheck Manual for more information.
  - If the state of all enabled Bits agree with the actual state of the Bit lines, the step will issue a "PASS" verdict.
- Physical lines for Bits 0 thru 7 have an internal 470 Ω resistor in series with each
- Digital I/O control is not available in SoundCheck ONE
- The Pass/Fail radio buttons are not used

External Interface

Communicate with other devices via a computer interface such as RS232 or IEEE-488 (GPIB). Select the Interface Number and the Interface Type field will update according to the matching device number in the Hardware Editor - External Tab. For more information see *Interface Table on page 75*. When setting up multiple devices in the Hardware Editor - External Tab, the Interface Number order must be the same.

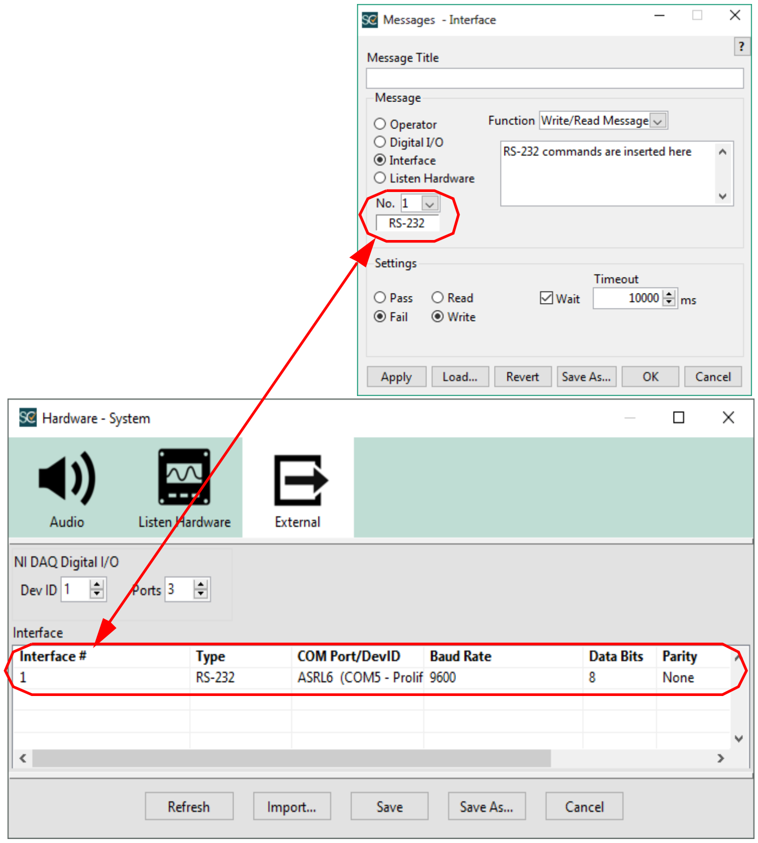


Figure 18-22: Hardware Editor - External

RS232

Setting the Interface Type to RS232 enables SoundCheck to send control messages to an external device that accepts RS232 commands. Add a Message Step to the sequence to send a command to the selected device.

Choose the interface type (RS232 or IEEE-488) by selecting the device number

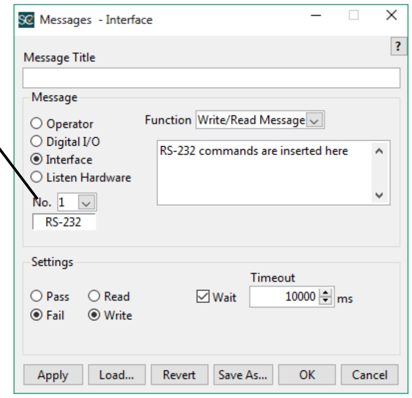


Figure 18-23: RS232 - *Write/Read Message*

RS232 Interface Actions

1. Invoke RS232 command based on the PASS/FAIL status of the previous step in the sequence.
2. Read message from another instrument or write (send) message to another instrument.
3. Wait for n number of milliseconds before executing the RS232 step configured in the *Message Step Editor*.

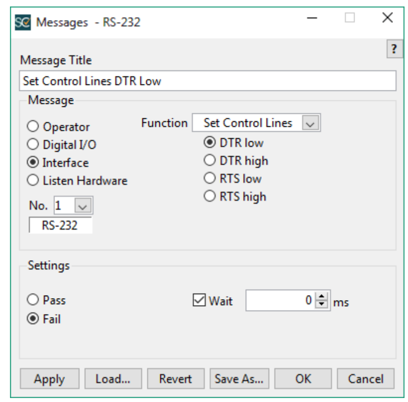


Figure 18-24: RS232 - *Set Control Lines*

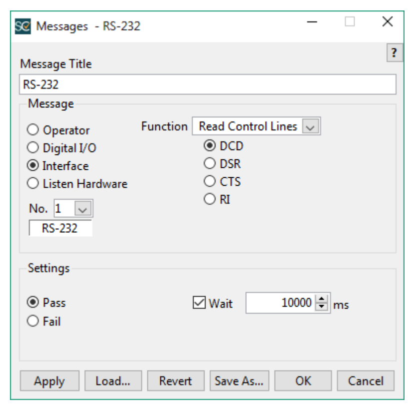


Figure 18-25: RS232 - *Read Control Lines*

IEEE-488 (GPIB)

Setting the Interface Type to IEEE-488 enables SoundCheck to send control messages to an external device that accepts IEEE-488 commands.

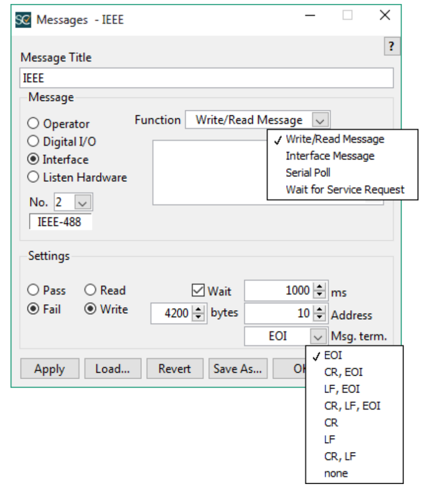


Figure 18-26: IEEE Message Setup

Output Message
IEEE commands that SoundCheck uses to control an addressed device. Messages must be entered in full (e.g., including Header, Header Separator, Data and Data Separator where appropriate). Only ASCII characters are allowed.

Interface Message

The IEEE commands listed below. These commands are used to setup and control the interface itself, rather than a particular device.

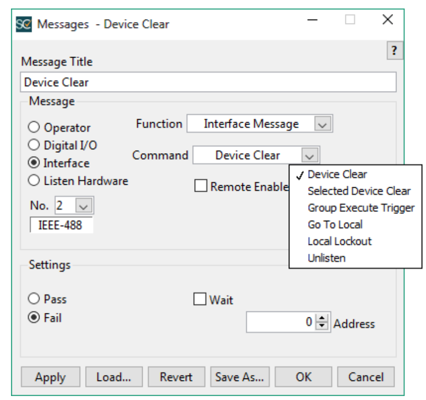


Figure 18-27: Interface Message Choices

- **Device Clear (DCL)** – causes all connected devices that implement the command to return to a predefined device-dependent state.
- **Selected Device Clear (SDC)** – sets all devices currently addressed as listeners to a predefined device-dependent state. Otherwise identical to Device Clear.
- **Group Execute Trigger (GET)** – provides a means of triggering devices simultaneously. GET causes all capable devices, which are currently addressed as listeners, to initiate a preprogrammed action (e.g., trigger, start a sweep etc.)
- **Go To Local (GTL)** – returns all devices currently addressed as listeners to local control. A device will return to remote when it is again addressed as a listener with REN true.
- **Unlisten (UNL)** – unaddresses all current listeners connected to the bus. UNL is used to guarantee that only the desired listeners are addressed.

Remote Enable
Enables SoundCheck to be controlled.

Serial Poll
SoundCheck interrogates an addressed device to ascertain the state of each bit in its status byte. A serial poll is typically used to synchronize SoundCheck with an external device, such as a turntable, to make sure it is in the desired position before making a measurement.

Syntax for Sending RS232 (serial) or IEEE-488 (GPIB) Commands in SoundCheck

Visible Characters:

In SoundCheck, you need to use a specific syntax in the *Message Step Editor* to send generic ASCII commands to your device. Any visible characters in the command field of the *Message Step Editor* are sent through without being altered. For example, the command SYSTEM would be sent to the device as the ASCII byte stream for the characters shown in the example on the right.

S
Y
S
T
E
M

which would be:

83
89
83
84
69
77

Non-visible (Termination) Characters:

Some devices require a termination character at the end of each sent command, which most often is not a visible character. To send any non-visible characters using the SoundCheck *Message Step*, you must enclose the decimal ASCII code for that character in brackets like this: **<10>**

Note: Some external devices state that termination characters are entered as: (CR) or [CR]. Always replace parenthesis or braces with bracket arrows: <CR>. Termination characters must use this format in SoundCheck message steps.

Here are a few characters that are not visible but are often used in serial communication, followed by their decimal ASCII codes:

- Line Feed (or \n): 10
- Carriage Return (or \r): 13
- Tab: 9

To include any of these characters in the command string, you must surround the decimal ASCII code for that character in brackets (e.g., line feed is <10> and carriage return is <13>). Parenthesis () and Bar Braces [] are not allowed. To produce an end-of-line character (\r\n), you need to combine both line feed and carriage return characters:

Line Feed + Carriage Return: <10><13>

Example:

When sending commands to a device like the Agilent 34401a, you must terminate each command with a line feed character. To initiate RS232 control over this device, you must send a SYSTEM:REMOTE command before sending any other commas.

In the SoundCheck *Message Step*, this command would look like:

SYSTEM:REMOTE<10>

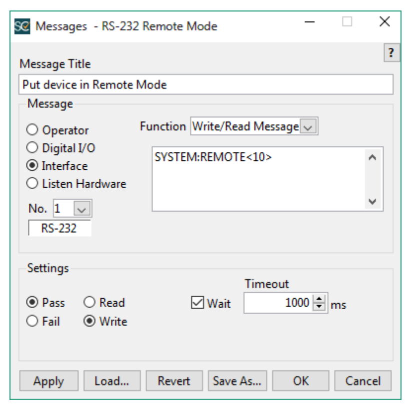


Figure 18-28: Message Step RS232 Example

Reading RS232 Messages

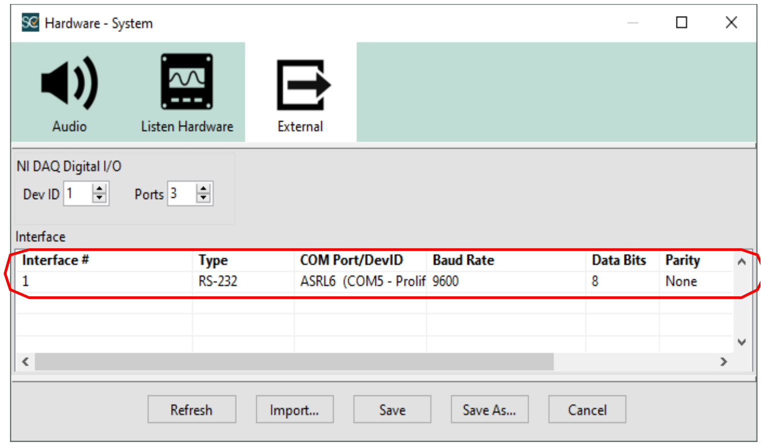


Figure 18-29: Hardware Config - RS232 Settings

SoundCheck can "Poll" the RS232 input for a "start" command. The trigger can be any alpha/numeric character or a simple string. (Single characters are preferred since there is less chance that there will be a communication error during the transmission of one character.) The Message Step can be set to automatically "Loop" back to itself, until the RS232 string has been received.

Once a sequence has been started, a Message Step can be set to look for input on the Serial Port specified in the Message Editor. This device must be Enabled in the System Hardware configuration before it can be selected in the Message Step. See ***Figure 18-29***.

The Hardware Configuration must match the RS232 settings of the device that is sending RS232 data. This is extremely important since incorrect settings can cause the SoundCheck computer to shut down when a message is sent.

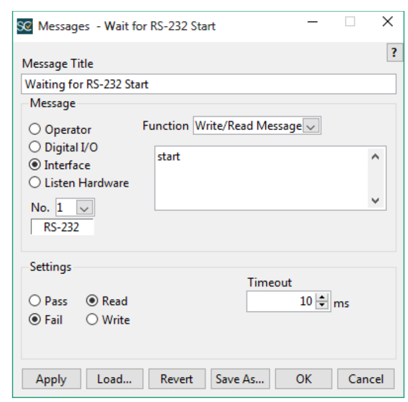


Figure 18-30: RS232 Settings

The Message Step is then set according to the example in ***Figure 18-30***.

- Select "Interface"
- Select number of RS232 line that is set in the System Hardware Configuration
- Select Function: Write/Read Message
- Enter the character string that will be sent from the external device. In this example: "start" (case sensitive)
- Settings: Select "Pass" and "Read" with a Timeout that is slightly greater than the amount of time needed for the external device to send its string of characters. (Using single characters for a trigger is preferred.) Do not set the Timeout to 0 mSec. This can cause a communication error.

Click **OK** to exit or Save As to give the step a new name.

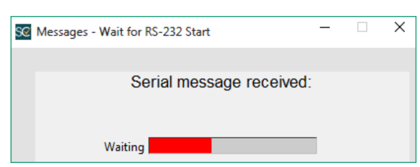


Figure 18-31: Displayed Message Step

When the sequence is run, the Message Step will appear as shown in. When the serial message has been received, the text will be displayed and the step will close so the that the sequence can continue. The Configuration of the Message Step must be set in order for it to Loop and Display.

Step Configuration

The Step Configuration of the Message Step should be set as shown in ***Figure 18-32***. (Step Configuration is found in the **Sequence** drop-down list of the Sequence Editor window.)

- "Display step when run" should be checked if you want to see the "Waiting for message prompt" during the run of the sequence. Normally this can be set to 0 seconds.
- Check "Jump on FAIL" and select the same message step. This creates the loop that recycles until the RS232 text has been received.
- Select "Overwrite Data" so that only one instance of the Index value (Loop Index) is created in the Memory List.

  See ***Index (Loop Index) on page 447***.

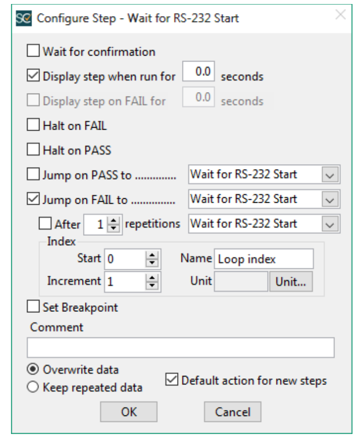


Figure 18-32: Message Step Configuration

The Message Step must be positioned in the sequence so that all desired measurement operations occur after it.

Figure 18-33 shows the Message Step in the sequence along with its configuration settings. Once the RS232 string has been received, (and the message status changes to "Passed") the sequence will proceed to the next step (Sti - 20k-20 Hz R10) and the operation of the sequence will finish.

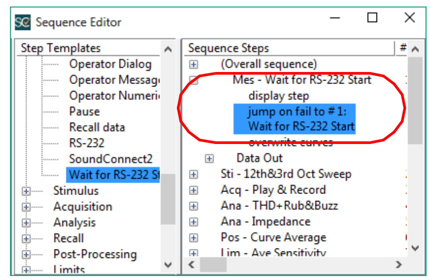


Figure 18-33: RS232 Step In Sequence

Testing with HyperTerminal

A second computer can use Windows HyperTerminal to send RS232 Test Commands. The Message Step in SoundCheck, that is reading RS232, must be configured to "Display step when run", long enough for all of the characters sent to be present in the RS232 buffer of the SoundCheck computer. It is important to note that characters are sent from HyperTerminal as soon as they are typed.

A message of "start" may take a person 2 or 3 seconds to type into HyperTerminal. If the Message Step is not open for this period of time, all of the characters may not make it into the buffer before the step recycles. Since the complete string was not read, the step status will indicate "Failed" and the loop will continue. This is one of the reasons that single characters are recommended for use as triggers.

Connection Type

The connection between two computers should be a Serial Port Null Modem cable. Other devices with RS232 outputs may use a standard serial port cable. Please refer to the documentation for that device for the correct cable type.

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Limits Editor

To view and change the system's test limits, select **Limits** from the **Setup** drop-down list on the main SoundCheck® menu bar, or use the shortcut **Ctrl+Shift+L**. You can use this step to set Pass/Fail bounds for your measured curves, waveforms and single values. A curve or value that crosses the upper or lower limit set by the step will give a Fail result, while a curve or value that lies between or equal to the set boundaries will Pass. There are many different ways of applying limits to measurement data.

Features include:

- Floating limit curves to fixed data
- Floating data to fixed limit curves
- Setting Absolute limits
- Allowing data to float on the x axis
- Data from the Memory List can be used to create limit curves which can be offset by a predetermined amount in the Limits Editor
- Data from the Memory List can be used as a limit curve, e.g., +/- 3 sigma of running statistics.

The Limits Editor functions are divided between two tabs; Data and Parameters. ***Figure 19-1*** shows the settings available under the two tabs.

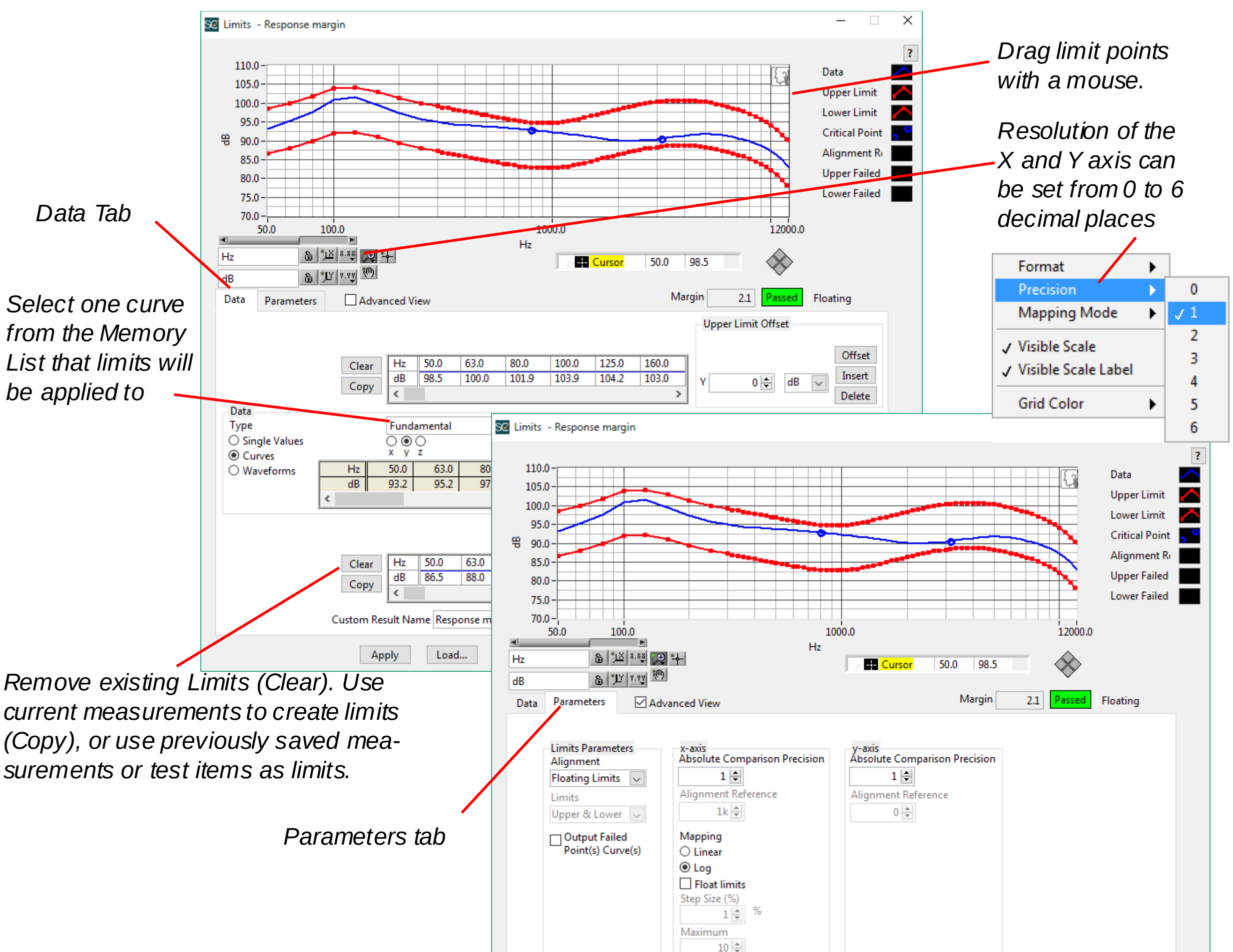


Figure 19-1: Limits Editor Tabs: Data and Parameters

Features

The Limits Editor offers two views. Basic view shows only the commonly adjusted settings, and by clicking on the Advanced tab, many more are revealed. This keeps the software simple for novice and production line users while retaining the flexibility required in R&D applications.

- Batch Processing allows you to select a Custom Group of multiple curves to apply limits on
- **Custom Result Name** allows you to change the default name for the limit curves in the Memory List without having to rename the step
- **Add Input Data Name** appends the name of the selected data to the Limit name in the Memory List
- Click on **Apply** and the limit curves and results are updated in the Memory List. Settings can be changed and tested without having to run a new measurement. Note the additional condition in ***Save or OK Warning after Apply on page 312***.
- Dynamic Selection of limit curves in Memory List, so that limits can come from another measurement or Recalled DAT file
- Show measurement curve on table (disabled for waveform)
- SI Units are implemented
- The editor incorporates the look and behavior of the Display Step.
- Easy "dragging" of limit points
- Graph or Plot items such as the Mapping Mode or Precision of the X and Y axis are stored when the step is saved
- Failed Points can be stored as curves in the Memory List

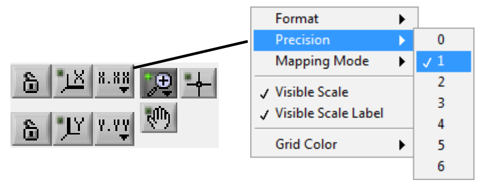
Note: Limits can be applied to single data items or a Custom Group created in the Memory List. Sequences from SoundCheck 6.01 (and previous versions) that applied limits to multiple curves in one step (e.g.; Self Test) will need to be updated to work in SoundCheck 18.1.

Precision of Limits Display

The precision of the Limits Editor display is set through the graph controls located below the graph. Click on **x.xx** or **y.yy**, click on **Precision** and select the number of decimal places to display. This changes only the precision of the graph. It does not change the precision of how the limits are applied to the data.

See ***Absolute Comparison Precision on page 315*** for an explanation of changing the precision of the applied limits.

Note: Setting the Absolute Comparison Precision overrides the Display Precision set by the user.



Format and Mapping Mode selections are also available.

Format - Select Decimal, Scientific, Engineering, etc.

Mapping Mode - Select Linear or Logarithmic

Critical Points

One critical point is generated for the upper limit and a second is generated for the lower limit. These points show up as yellow circles on the Limits Editor Display but are not passed on to the Memory List. This is to be used as a visual marker to aid in building and editing limit steps.

In the case where the data is within the bounds of the limit curve, this point shows where the data is closest to the limit curve.

In the case where the data lies outside the bounds of the limit curve, this point shows the greatest deviation of the data relative to that limit curve.

Limits Editor Summary Table

Limits Type	Data	Tolerance(s)
Individual Points	Stays fixed at measured level.	Same as Absolute but with no interpolation between data and limits resolution.
Absolute	Stays fixed at measured level.	Stays fixed at values entered in Limits Editor.
Floating Limits	Stays fixed at measured level.	Moves up or down such that maximum number of points in data curve can fit between tolerances.
Floating Data	Moves up or down such that maximum number of points in data curve can fit between tolerances.	Stays fixed at values entered in *Limits Editor*.
Aligned Limits	Stays fixed at measured level. Used in some telephone (e.g., **TIA 470B**) and military standards. (not used in new **TIA 470C** standard)	Moves up or down to align the Reference x, y value to the measurement curve.
Aligned Data	Moves up or down to the anchor point (X and Y values entered in **Alignment Reference** section of *Limits Editor*). An example is anchoring the 1 kHz measured curve value at 0 dB.	Stays fixed at values entered in *Limits Editor*.

Using the Limits Editor

You only need to specify the minimum number of X-Y data pairs (or knee points) that define the shape of the tolerance curve. All other points in between the specified knee points of the limit curve will be interpolated and compared to see if any point on the measurement curve intersects. In the example below, a frequency response curve for a telephone headset is shown in green. The upper and lower limits (red) were entered in the table manually. After the points are entered you can click on any of the limit curve points on the graph to move them. Holding down the control key restricts the movement of the graph point so it can only move along the y axis.

Frequencies and dB levels can be entered/deleted manually in table

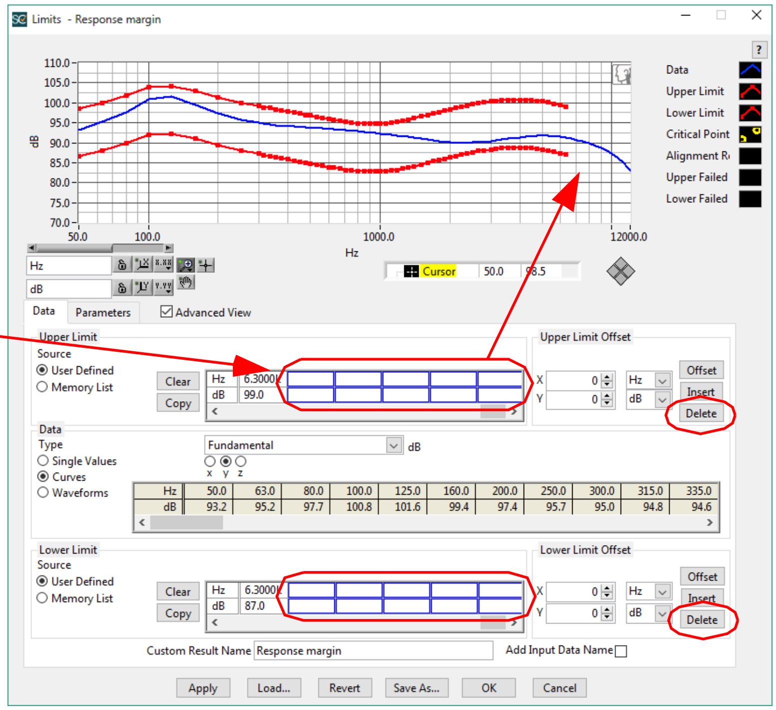


Figure 19-2: Entering Limits in Table

Upper and Lower Limits

Note: Floating Limits or Floating Data will require upper and lower limits.

Select whether to apply an upper, lower limit or both. Distortion, for example, should only require an upper limit, but response limits will require both an upper and lower limits if Floating Limits or Floating Data is selected.

See ***Alignment on page 310*** for more information.

Clear

Deletes all of the limit data in the cells. This allows you to enter new limits data manually or copy Data from a Memory List curve into the cells (single value or curve).

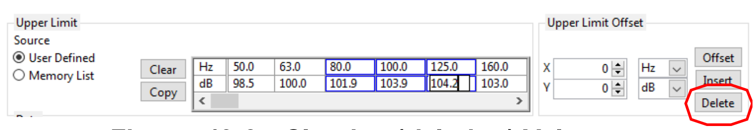


Figure 19-3: Clearing (deleting) Values

To select a range of cells to delete, left mouse click and hold on the first cell and drag the mouse to highlight the desired cells. The selected cells will be highlighted in blue. Click the **Delete** button at the right of the editor.

Copy

Copies the selected data curve from the *Memory List* and inserts the curve values into the **Upper or Lower Limits** table. In the example in ***Figure 19-4***:

1. Select Advanced View
2. In the Data section, select a data curve from the Memory List.
3. Click **Copy** on both Upper and Lower Limits table buttons. Data points are copied into these fields.
4. Enter the **Offset** for the Upper and Lower tables. +6 dB for the Upper and -6 dB for the Lower.
5. Click the **Offset** button for both tables. The upper limit is offset by +6 dB and the lower limit is offset by -6 dB.

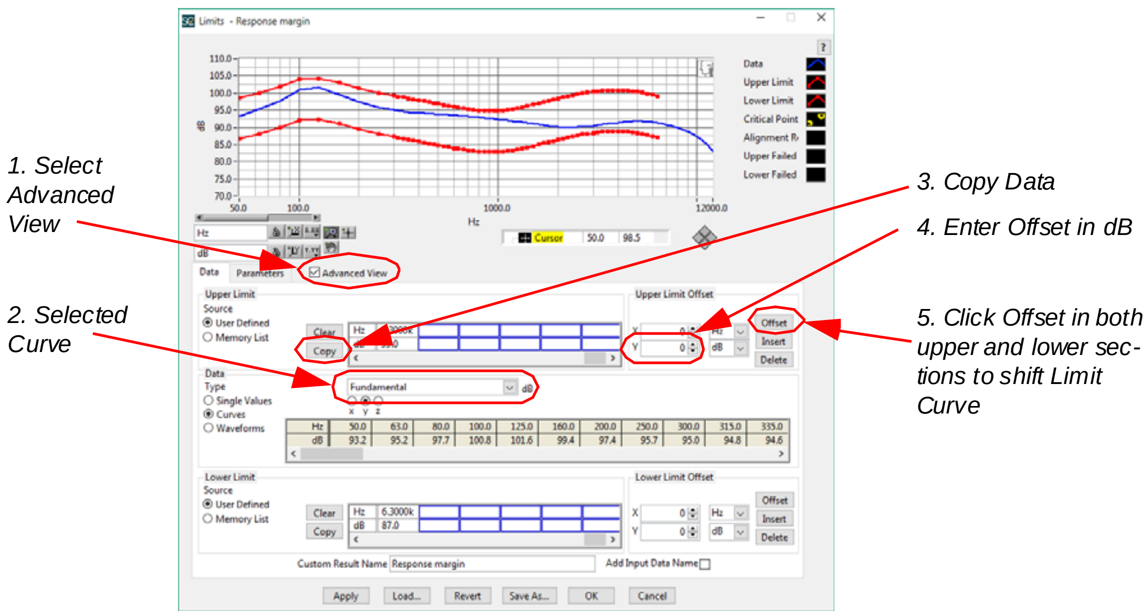


Figure 19-4: Creating Limits by Using Data From Memory List

Note: SoundCheck may have difficulty copying very large arrays (>2000 points). For example, copying an FFT Spectrum into the Limits Table may cause you difficulty. Computer hardware will dictate full capability.

Delete

Removes those points that have been selected in the **Upper Limit** and/or **Lower Limit** curves. To select one or more points, Left-click on the first UPPER cell in the range and then click on the last Lower cell in the range. In the example in ***Figure 19-5: Deleting Values in Limits Table***, points above 6300 Hz will be deleted. You can also Left-click the mouse and drag it to the right to highlight both frequency and amplitude values. **A blue border surrounds the cells in the table that will be deleted.**

Clicking the **Delete** button will remove these cells from the table. By deleting the same cells in **Lower Limit**, the resulting Pass/Fail limits are shown in ***Figure 19-5: Deleting Values in Limits Table***.

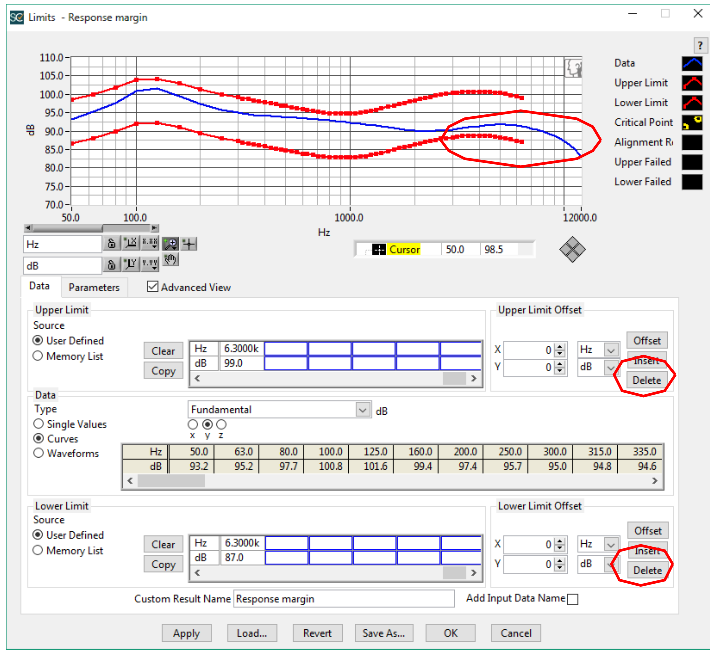


Figure 19-5: Deleting Values in Limits Table

Offset

Enables you to move or offset the entire or selected section of a limit curve(s) by a specified amount. The values you want to offset must be highlighted in blue. Enter the number you want to offset by (usually you will offset in the Y-axis, leave the X-axis field at 0.0 to make no changes there) and then click **Offset**. You can select all the values in the limit table by clicking the **horizontal line** between the two unit specifications.

To select the entire table, hold down the control key and click on any cell in the table.

Offset by Percent

Offset by percent allows you to set the percent difference from a mean value that is acceptable for upper and/or lower limits. In ***Figure: 19-6 Offset by Percent*** the mean value is 1 and the limits should be + or - 10%.

1. Enter the mean value (1) in the Upper and Lower Limit value boxes.
2. Enter the Percent of offset in the Upper and Lower Limit Offset boxes and select **%**.
3. Click Offset for the Upper and Lower Limits.
4. The Upper and Lower Limit values are updated in the value boxes and on the meter or graphs.

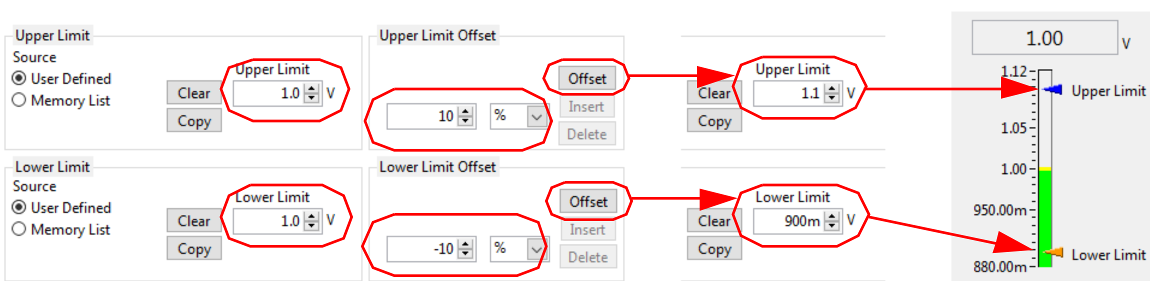


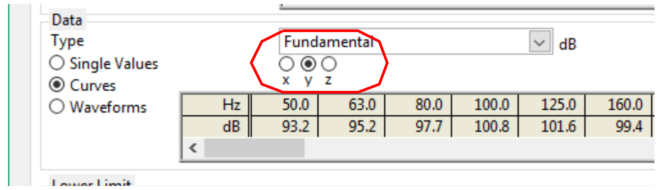
Figure: 19-6 Offset by Percent

Insert

X-Y cells can be added to the limit table by clicking on a cell and then clicking on the Insert button.

Pass/Fail Tolerance Axis

You can choose which axis will be compared to a Pass/Fail tolerance. Typically it is the y (magnitude) axis, but tolerances can be generated for the other axes. The units for the tolerance curves or values are automatically assigned based on the x, y, or z-axis units.



Dynamic Limits - Recall from Memory List

Limits can be called from an item in the Memory List by clicking on the drop-down list. These limits could be recalled from a DAT file or from a previous measurement in the sequence. The example in ***Figure: 19-7 Limits from Memory List*** shows the upper and lower limits being called from 2 files; +3 Sigma and -3 Sigma.

Only applicable data types from the Memory List are available to be used as dynamic limits for any given data type. i.e; Only single values and curves can be applied as limits on curves. If a curve is selected as data, a waveform cannot be selected from the Memory List.

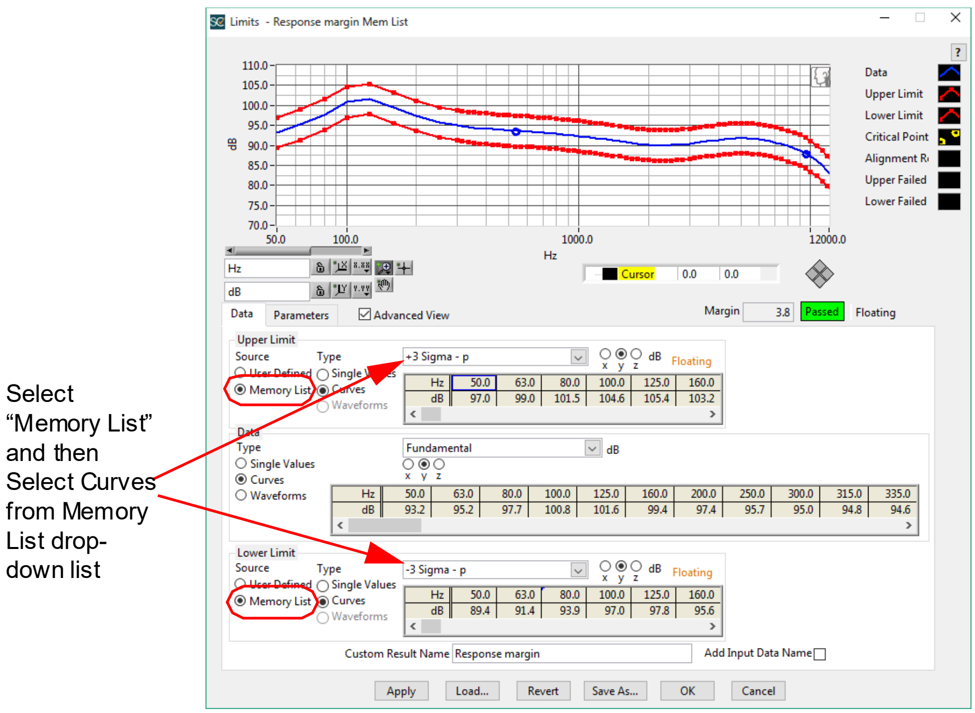


Figure: 19-7 Limits from Memory List

Note: No offset is available when using dynamic limits.

Data Tab Settings

The following choices are available in the Data selection field:

- Single Values - Any single number value or group of values from the Memory List
- Curves - Any curve or group of curves in the Memory List
- Waveforms - WFM and WAV files and groups
- **Batch Processing** allows you to select a group of curves as shown in ***Figure 19-5***. Any Custom Group created in the Memory List can be used for the appropriate limit type. Please refer to ***Sorting and Grouping on page 331*** for instructions on creating a Custom Group.

Figure 19-8: Batch Processing in Limits Editor

Single Values

Single Values is used when applying limits to items from the Memory List that are single number values. This can be used for comparing the measured level at a specified frequency to a lower and upper threshold or a single value created from a Post-processing Step, such **Curve Average** or **Loose Particle Count**.

The use of Single Value is not limited to values acquired by a measurement. ***Figure 19-9*** shows limits applied to the value of "Diaphragm Diameter". This number was entered by a technician during the execution of the sequence. The number was passed from a Message step to the Memory List.

User defined limit values can be entered manually in the numeric input field or by clicking and dragging the Upper and/or Lower Limit arrows. If either the upper or lower limit is defined from the Memory List, the ability to click and drag the arrows is disabled.

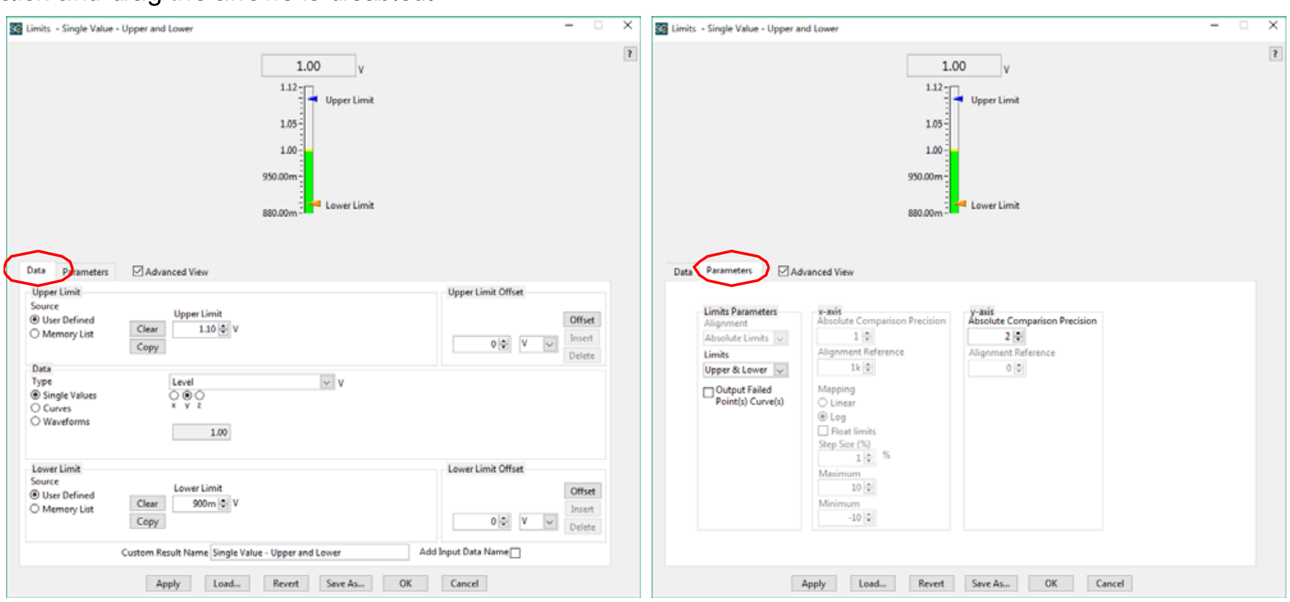


Figure: 19-9 Single Values

Offset for single value (linear or %)

- Linear allows you to select specific upper and/or lower values. Offset by percent allows you to select the percent difference from a target mean value that is acceptable for upper and/or lower limits.

  See ***Figure: 19-6 Offset by Percent*** for an example.

Curves

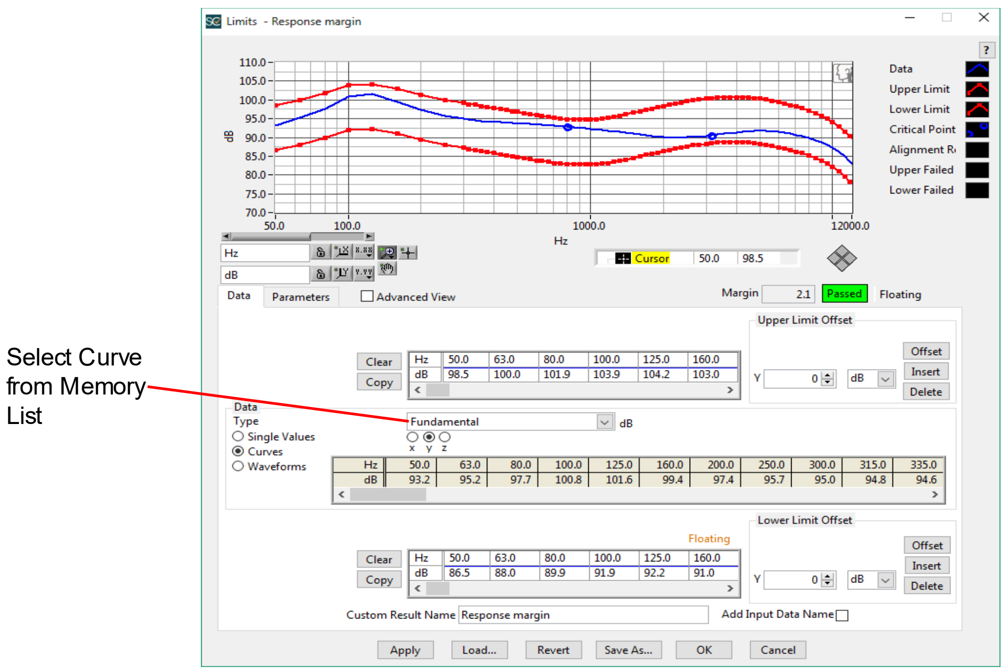


Figure: 19-10 Curve selected, User Defined Limits

Waveforms

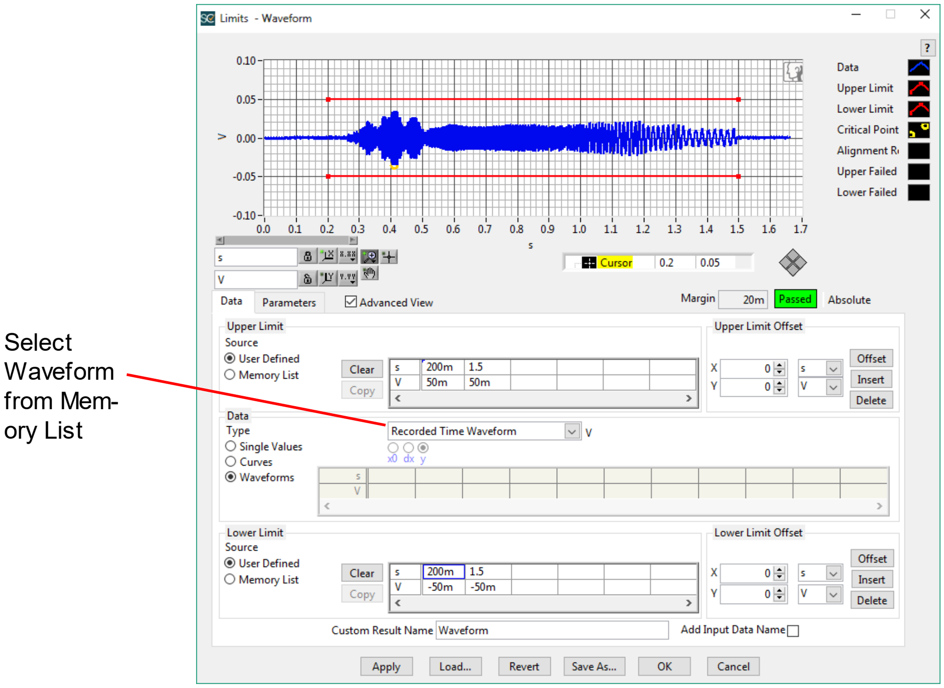


Figure: 19-11 Waveform

Rules - Waveform in Limit Steps

Because waveforms have so many points, the following rules apply:

- User defined limits may be created as X-Y (Time-Amplitude) "knee points" by the user.
- User defined limits will be converted to waveforms and output to the Memory List so you can plot data and limits together.
- Dynamic limits may be applied to waveforms. However, You can not copy a waveform in the Limits Editor to create "static Limits" (i.e., limits saved with the step) because of the large waveform size.
- A Single Value may be applied as a limit on a waveform.
- User will not be able to view the waveform data in the table to which the limits are to be applied in the Limits Editor. This will only be visible in the graph.

Multimeter Limit Steps

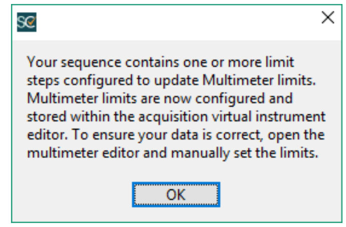


Figure: 19-12 Multimeter Update

Important! **Sequences prior to SoundCheck 15:** Multimeter Limit steps that were used to set the Limit Range of the Multimeter Virtual Instrument Acquisition Step require changes. The Multimeter Acquisition Step must be modified to set the Limit Range. The Limit Step used for setting the limit range can be deleted from the sequence. When opening sequences from prior versions you will encounter the message as shown in ***Figure: 19-12 Multimeter Update***.

A separate Limit Step is then used after the Multimeter Acquisition Step to add the Results to the Memory List. See ***Limits Tab on page 468*** for more information.

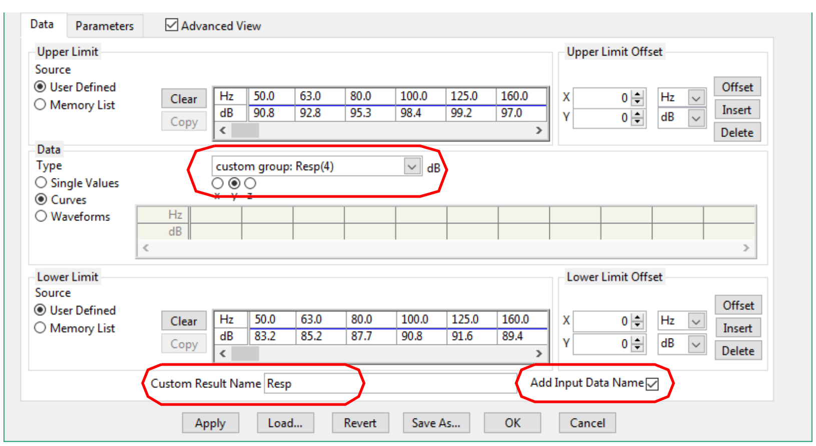
Batch Processing

This allows you to select a group of curves from the Memory List to apply limits on. ***Figure 19-13*** shows the custom group **Array** at the top of the Memory List. This custom group is then selected in the Limit Step.

Any Custom Group created in the Memory List can be used for the appropriate limit type. Please refer to ***Sorting and Grouping on page 331*** for instructions on creating a Custom Group.

Custom Result Name

The default name for a Limit Step is the name of the step itself. Custom Result Name allows you to change that name. This name will only appear in the sequence it is used in. The example in ***Figure 19-13*** shows a Limit Step with the name "**Resp**". The Memory List shows the new limit name, "**Resp lower limit**...". You can have a unique name for the limits in the Memory List, for every instance of the step.



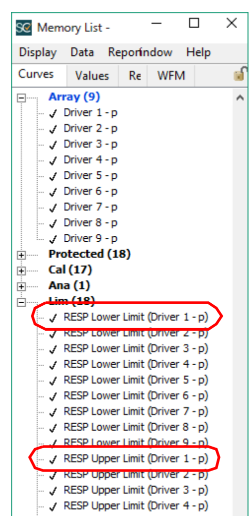


Figure: 19-13 Naming Features

Add Input Data Name

When using the Batch Limits feature to apply limits to a Custom Group of curves, it is recommend that you use Add Input Data Name to make it easier to keep track of the resulting limit curves in the Memory List.

Figure 19-13 shows the nine data curves in the Custom Group: Array. Each curve name, in parenthesis, is appended to the limit curve names, e.g.; "**RESP lower limit (Driver 1)**" through "**RESP lower limit (Driver 9)**".

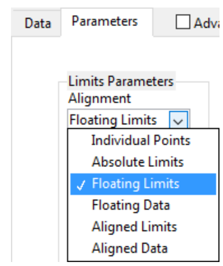
Parameters Tab - Settings

Note: Advanced View is enabled in examples to show all features.

Note: The Resolution for X & Y can be set independently.

Alignment

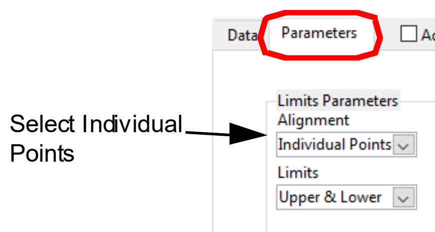
Five methods of Aligning Limits to the selected data can be found on the Parameters Tab.



- Individual Points
- Absolute Limits
- Floating Limits
- Floating Data
- Aligned Limits
- Aligned Data

Individual Points

The measurement is compared to the absolute values of the individual tolerance points. Only when the measurement exceeds the tolerance at specific limit points will a failure be indicated. In ***Figure 19-14: Individual Limit Points*** the level at 1700 Hz is above the level of the limits at 1000 and 2000 Hz, but the limit status indicates Passed. There is no interpolation between limit points.



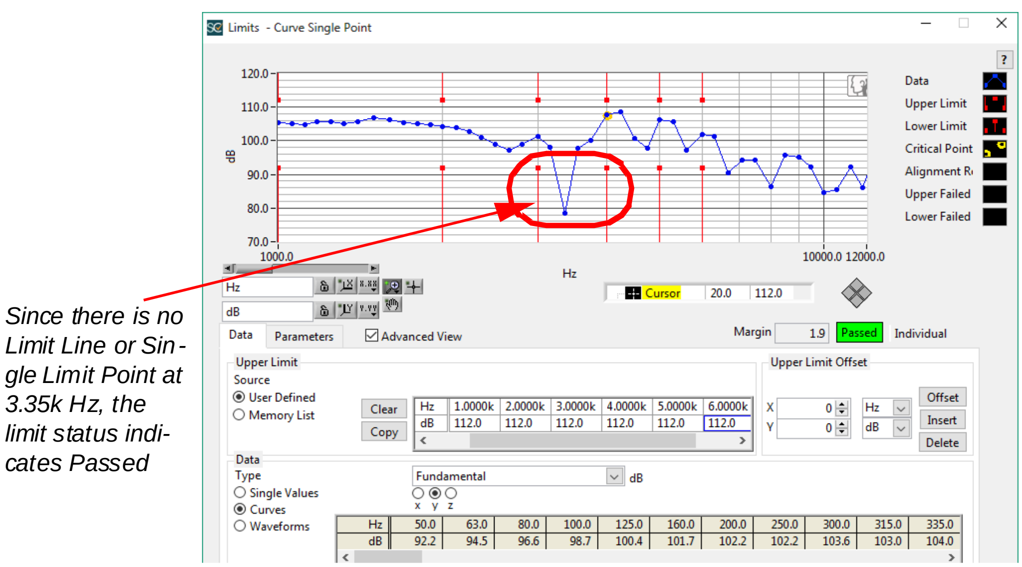


Figure 19-14: Individual Limit Points

Absolute Limits

The measurement is compared to the absolute values of the tolerance limit(s). When **Apply** is clicked, SoundCheck will highlight those points where the data exceeds the upper and lower limits by the greatest amount. In ***Figure 19-15: Curve Compared to Absolute Limits***, the measured curve (Test Monitor) failed by -1.0 dB at 4250 Hz (note the small yellow circle at 4250 Hz).

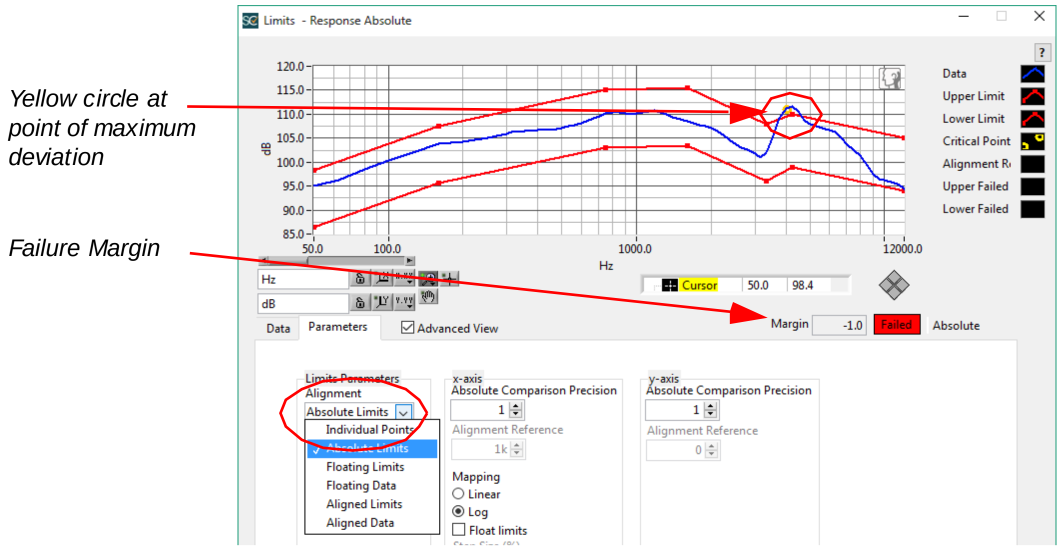


Figure 19-15: Curve Compared to Absolute Limits

Floating Limits

The tolerance limits will float (shift up and down) in reference to the measurement curve such that the maximum number of data points will fit between the tolerances. By clicking **Apply**, the curve that previously failed now passes by 0.8 dB, because the tolerances have shifted so the measured data fits between the limits. When **Floating Limits** is chosen, SoundCheck will execute a "Best Fit" curve fitting routine between the upper and lower limits. Yellow circles will highlight those points closest to the limits (in this case 4250 Hz).

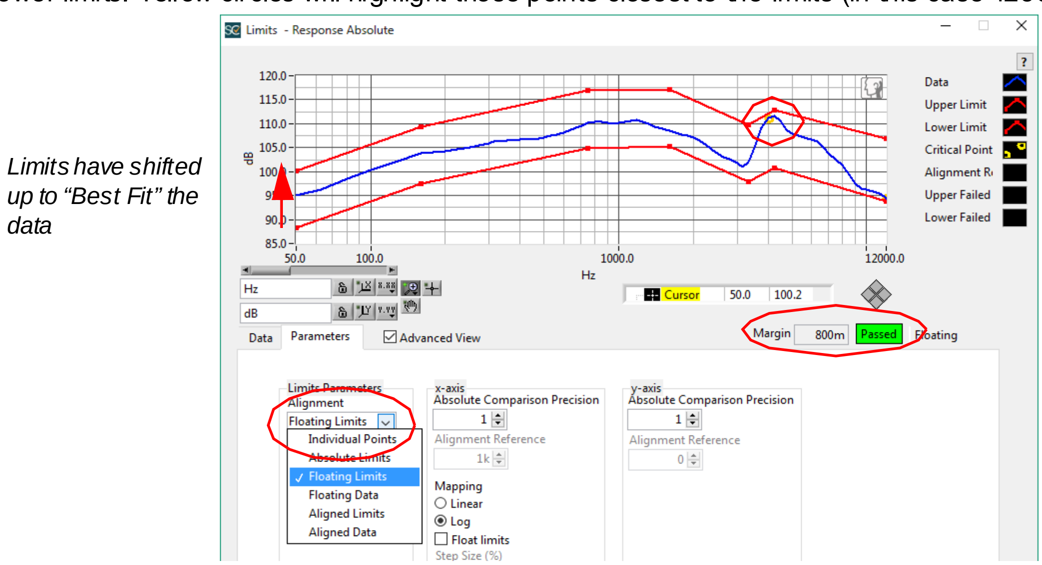


Figure 19-16: Curve Compared to Floating Limits

Save or OK Warning after Apply

When using Floating Limits or Floating Data a warning message will appear when you click on “**Ok**” or “**Save as”**, after clicking Apply. This warning appears because “Apply” may have moved the limits. You are then prompted to:

- Save Current - Save using the new position of the limits.
- Cancel & Revert - Undo the modified limit step settings and return to the editor.
- Save Previous - Save the limits as they were before the last time the Apply button was clicked.

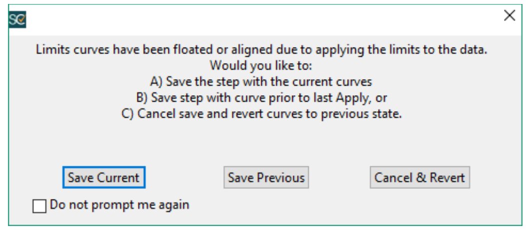


Figure: 19-17 Warning - Floated/ Aligned

Output to Memory List

- Upper and Lower Limits are added to the Memory List for the data item selected
- Limits are named according to the name entered in the **Custom Result Name field**
- **Add Input Data Name** will append the data name to the **Custom Result Name**, e.g., “Response margin Upper Limit (Fundamental)”
- If a Custom Group is selected for Batch Limits, limits are created for each item in the group. Selecting “Add Input Data Name“ is recommended when selecting a Custom Group. See ***Sorting and Grouping on page 331***.

Note: Floating Limits or Floating Data always create upper and lower limits.

Floating Limits on y axis

Limit curves will adjust to “Best Fit“ the selected data. This adjustment is made each time the data curve changes for each sequence run.

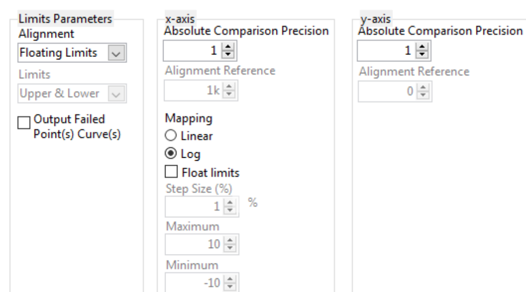


Figure: 19-18 Y axis Floating Limits

Floating Limits on x axis

In many instances, electroacoustic transducers will exhibit sharp resonance or anti-resonance in their frequency response curves.

For example, a small shift or reflection due to a slight change in microphone position can cause false rejects. SoundCheck can take into account these slight changes in peaks and dips in the frequency response curve by floating an X-axis tolerance.

When the x-axis **Float Limits** box is checked, SoundCheck will shift the frequency response curve by the amount specified in the **Step Size** box. See ***Figure: 19-19 X axis Floating Limits***.

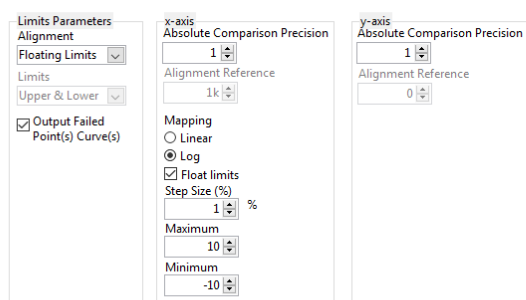


Figure: 19-19 X axis Floating Limits

The X-axis tolerance can be either logarithmic or linear. If it is set to **Log**, the step size (amount that SoundCheck will shift the curve in the X direction) is in percent (%). If you want to shift the curve by 1/24th of an octave, you would set this value to 3 percent. **Maximum** and **Minimum** refer to the greatest amount, in percent, that the curve will be shifted. If **Lin** is chosen, the step size is in Hz as well as the **Maximum** and **Minimum** frequency limits (e.g., ± 10 Hz).

Floating Data

The amplitude of the Limit Curves stays fixed as it does when set to **Absolute**. The data points will float (shift up and down) in reference to the upper and lower tolerances, such that the maximum number of data points will fit between these tolerances. The example in ***Figure 19-20: Data Floats and Tolerance Curves Stay Fixed*** shows a reference point of 1 kHz at 82.3 dB. In the first frame, the Data curve is obviously outside the range of the limits even though the shape of the response is correct. By selecting Float Data from the Parameters Tab, the Upper and Lower Limit Curves will remain at their current levels. After clicking **Apply** the Data curve has shifted to best fit the Limit Curves. Note that the reference point of 1 kHz is now at 77.3 dB.

Note: A copy of the Floating Data is passed to the Memory List with “Floated“ appended to the end of the name.

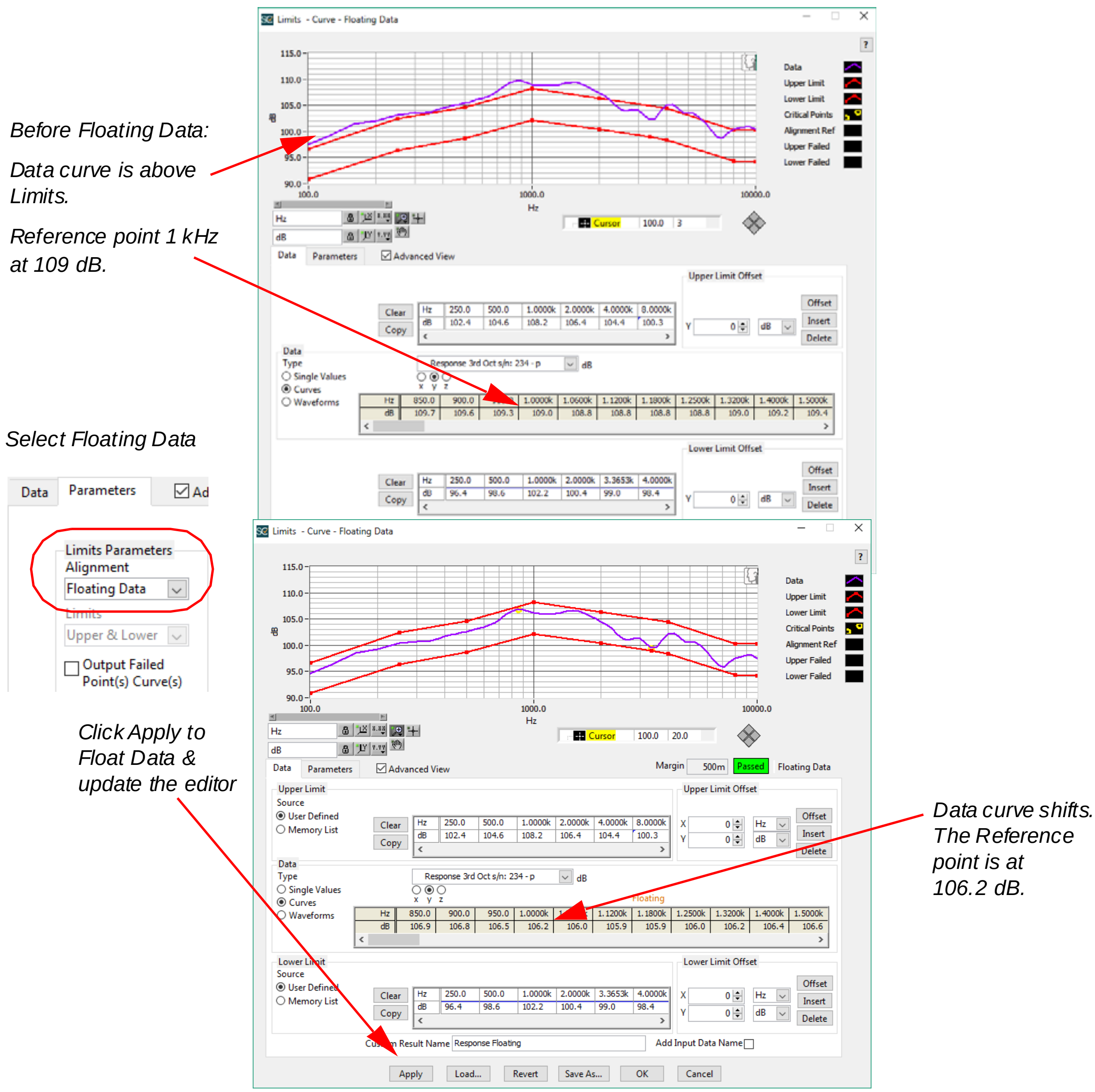


Figure 19-20: Data Floats and Tolerance Curves Stay Fixed

Aligned Limits

Positions the tolerance limits relative to the curve by a specified offset at a user-defined reference point in the **Align Reference** fields. The **Alignment Reference** numeric fields are enabled when this tolerance type is selected. See ***Figure: 19-21 Aligned Limits***.

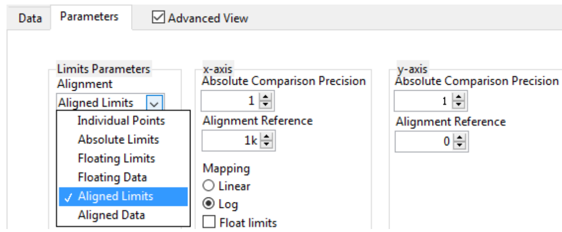


Figure: 19-21 Aligned Limits

- Select a frequency point on the Data Curve and enter it in the <u>x axis Alignment Reference field</u>
- Select the midpoint of the Upper and Lower Limits curves, at the frequency point entered in the x axis field. Enter this in the <u>y axis Alignment Reference field</u>
- On each sequence run, the limits will adjust so that selected center point of the limits always tracks the specified frequency point on the data curve
- Used in older telecom and military standards

Output to Memory List

- Upper and Lower Limits are added to the Memory List for the data item selected
- Limits are named according to the name entered in the **Custom Result Name field**
- **Add Input Data Name** will append the data name to the **Custom Result Name**, e.g., “Response margin Upper Limit (Fundamental)”
- If a Custom Group is selected for Batch Limits, limits are created for each item in the group. Selecting “Add Input Data Name“ is recommended when selecting a Custom Group. See ***Sorting and Grouping on page 331***.

Aligned Data

Moves the selected data curve so that the specified x axis point of the curve is at the specified y axis point on the graph. Limit curves are not adjusted.

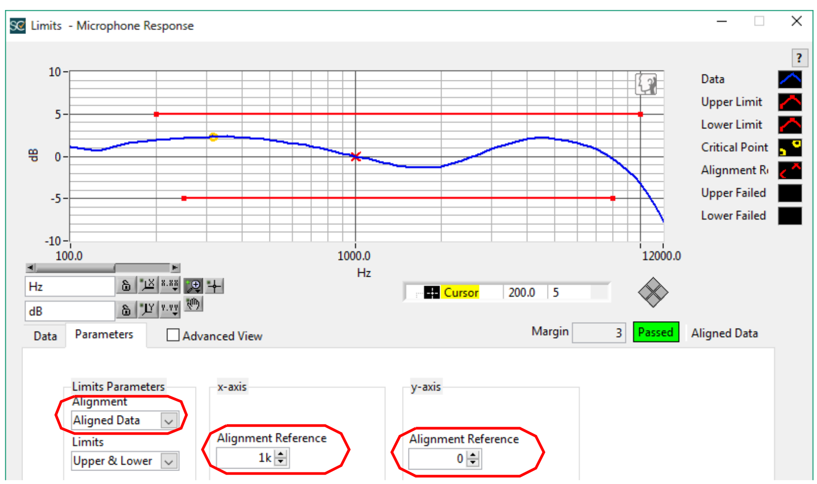


Figure 19-22: Aligned Data Selection

- On each sequence run, the data curve will adjust so that selected center point of the limits always tracks the specified frequency point on the data curve
- Commonly used for microphone frequency response, as shown in the “Microphone” example sequence. ***Figure 19-22*** shows the response, Normalized to 0 dB at 1 kHz.

Output to Memory List

- Upper Limits, Lower Limits or both are added to the Memory List for the data item selected
- Limits are named according to the name entered in the **Custom Result Name field**

- A new version of the data curve is created with “Aligned” append to the data curve name, e.g., “Fundamental Aligned“. This appears in the Limits Group when Autogroup is turned on.
- **Add Input Data Name** will append the data name to the **Custom Result Name**, e.g., “Response margin Upper Limit (Fundamental)”
- If a Custom Group is selected for Batch Limits, selected limits are created for each item in the group. Selecting “Add Input Data Name“ is recommended when selecting a Custom Group.

Absolute Comparison Precision

To pass or fail the device under test to within 1 dB, choose **Precision = 0**; within 0.1 dB choose **Precision = 1**; within 0.01 dB choose **Precision = 2**, etc. The default precision is one (1) decimal place. In this case, a data point of 0.05 is rounded up to 0.1 and then the limit, which has also be rounded to the selected precision, is applied. When the data equals the limit, a Pass verdict is returned. Only when the data is outside the bounds of the limits is a Fail verdict returned.

Note: Regardless of the resolution set in the Limits step, the default resolution for the Results Display window is two (2) digits of precision.

Note: Setting the Absolute Comparison Precision overrides the Display Precision set by the user. See ***Precision of Limits Display on page 301***.

The example in ***Figure: 19-23 Absolute Comparison Precision 2 decimal places*** show the Absolute Comparison Precision set to 2 decimal places on the y axis. The value of 77.79 dB at 6.3 kHz is outside the limit tolerance by 0.01 B (-10m dB). In this case the Limit step returns a Failed verdict.

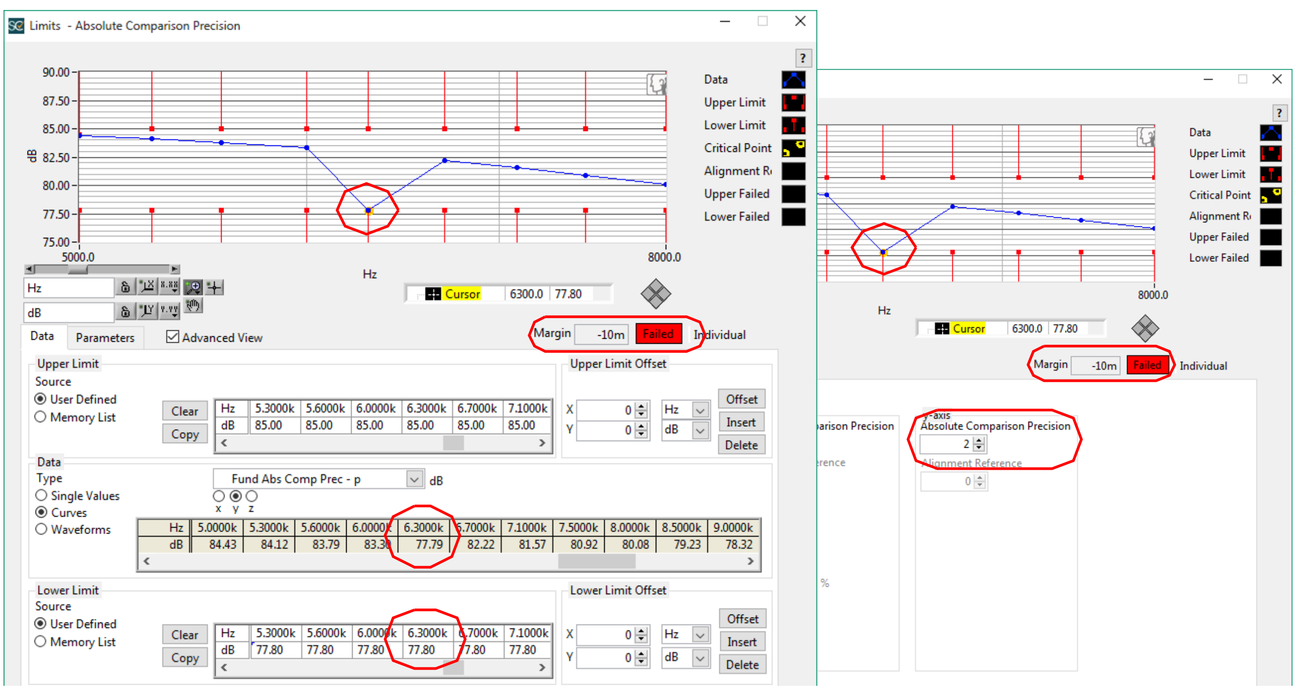


Figure: 19-23 Absolute Comparison Precision 2 decimal places

By changing the Absolute Comparison Precision to 1 decimal place the y axis value is rounded to 77.8 dB. The data values have not changed. Only the precision of the application of the data has changed. Since the Limit value and the Data value are the same, a Pass verdict is returned with a margin of 0 dB. This is shown in ***Figure: 19-24 Absolute Comparison Precision 1 decimal place***.

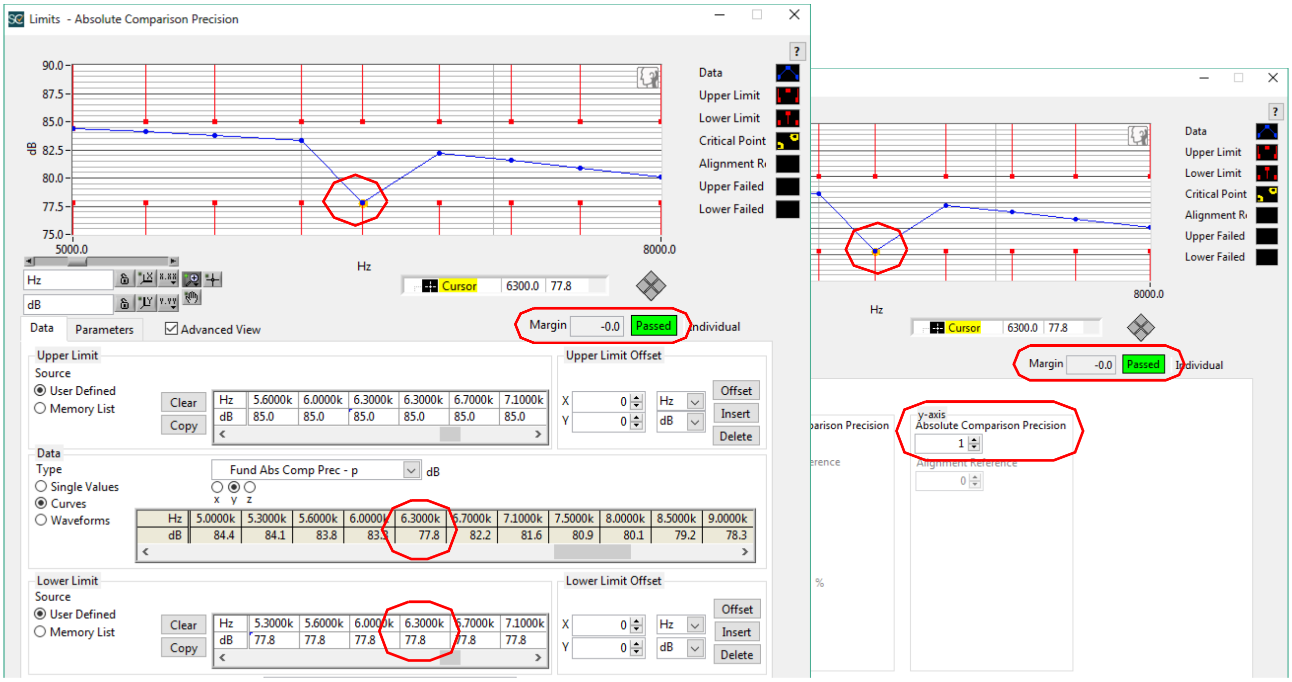


Figure: 19-24 Absolute Comparison Precision 1 decimal place

X Axis - Log vs. Linear Interpolation

If a data point lies between two points along a limit curve, SoundCheck must interpolate along the Limit Curve in order to determine how the data point is compared to the limit curve. When Log is selected, the data value is compared to the limit curve using a Log, X axis scale. See *Figure 19-25: Log Interpolation*. The data point is 4.4 dB at 1000 Hz. The Lower Limit curve starts at 5 dB at 100 Hz and ends at 3 dB at 10 kHz. In this case, log interpolation compares the data to a value of 4 dB at 1 kHz. This is the default setting for the Limit Step.

The result is PASS since the Data Point is ABOVE the ***Interpolated Point*** along the limit curve.

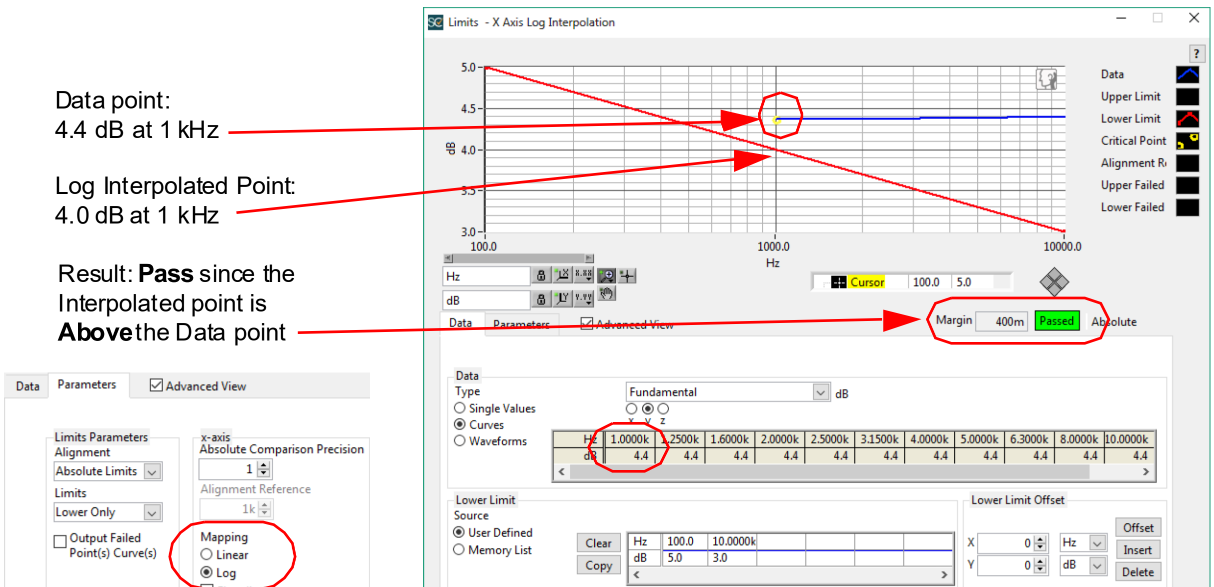


Figure 19-25: Log Interpolation

Using the same limit curve and data point, *Figure 19-26: Linear Interpolation* shows the Linear Interpolation result. The Limit Curve is interpolated to be **4.82 dB at 1 kHz**. The X axis of the graph has been switched to Linear to match the Linear Mapping selected in Parameters. The result is **FAIL** since the data point is BELOW the ***Interpolated Point*** along the limit curve.

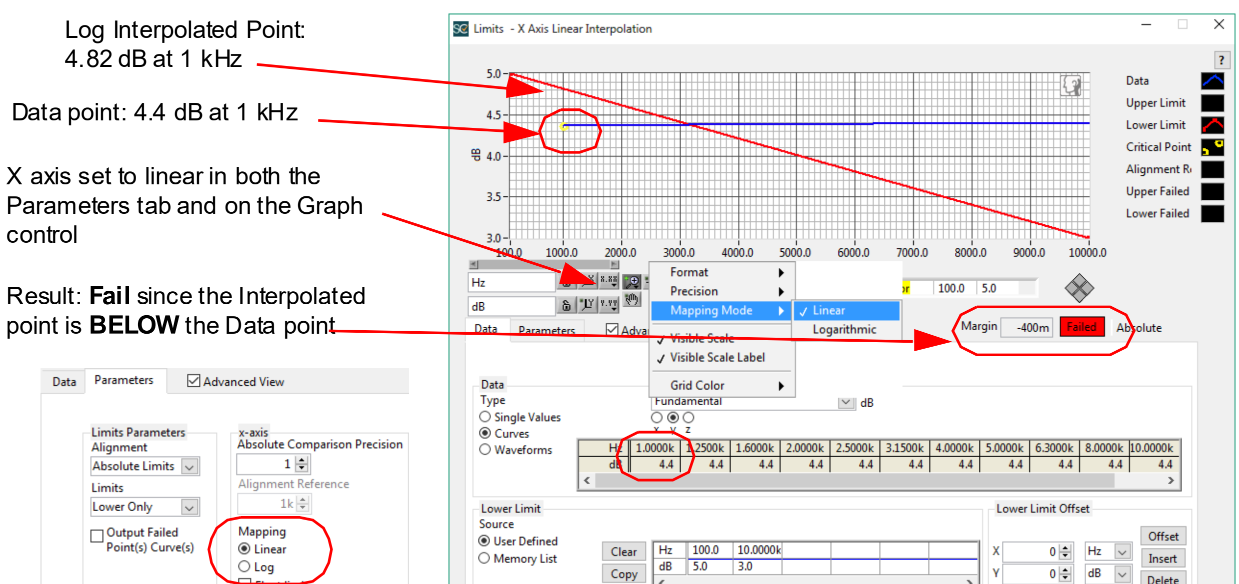


Figure 19-26: Linear Interpolation

This shows why the display X axis mapping and the interpolation control should be set to the same scale. This way, the interpolation matches the visual representation of the curves. This should prevent false visual failures.

- Interpolation along the Y axis is always Linear regardless of the settings of the X Y Graph Display
- For no interpolation between limit points, select Individual Points under Limits Parameters - Alignment. *See* ***Individual Points on page 310***.
- For Displays with Linearity Limits, where both axis' are dB, select Linear Mapping for the X axis on both the Parameters Tab and on the Limit Graph Selection. See ***Precision of Limits Display on page 301***.

Failed Points

When Failed Points is selected on the Parameters tab the points of failure are stored in the Memory List as a curve.

- One curve for Upper Limit and another for Lower Limit
- If a Custom Group is selected, Upper and Lower Failed Points are created for each curve in the group
- Curve of failed points contains all (and only) the points that exceed the limit curve
  - If only one failed point is produced, the curve will not show up in a Display Step unless the curve is set to Solid Points (See ***Figure 19-26: Linear Interpolation***)
  - If the limit step is executed and there are no failed points (PASS situation), the failed points curve is populated with a value of NaN (Not a Number).
  - If the limit step is *not* executed, the failed points curve will remain empty

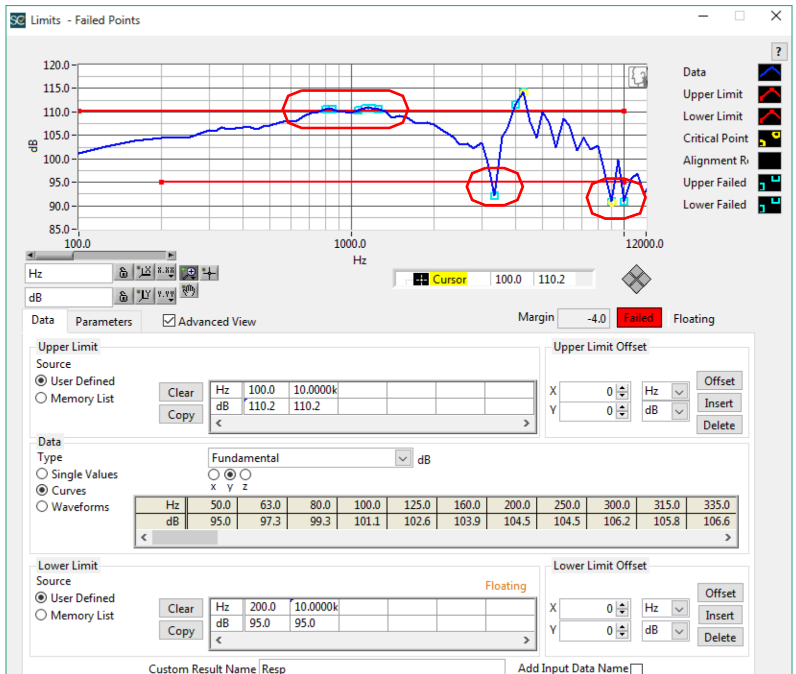


Figure: 19-27 Failed Points

Display Step example of Failed Points

Green and Purple Failed Points curves are set to have “Square Point Style”.

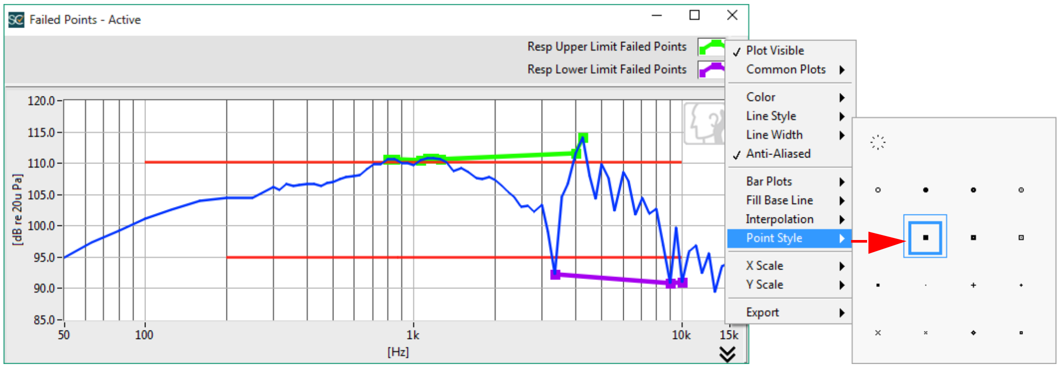


Figure: 19-28 Failed Points

Comparison of Absolute Limits, Floating Limits and Floating Data

The following three figures show how the same response curve can have limits applied in three different ways.

Absolute Limits

The measured curve is outside of absolute value of the limits, so the condition is **FAIL**. The level of the response is 108.9 dB at 1 kHz.

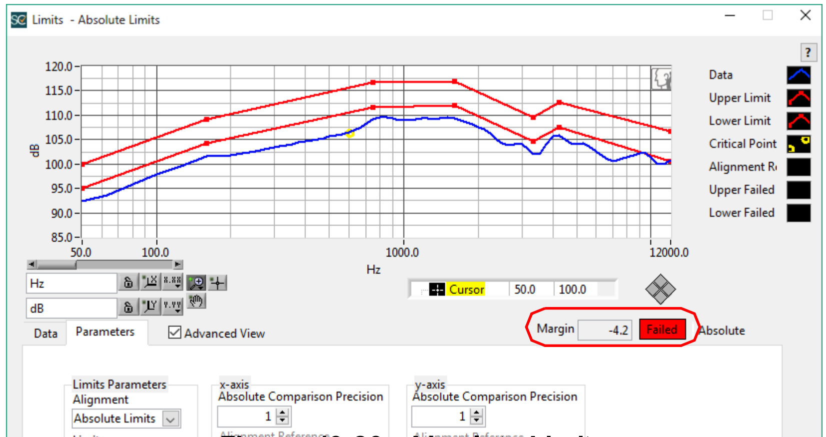


Figure 19-29: Absolute Limits

Floating Limits

The measured curve stays at its measured level, 108.9 dB at 1 kHz, but the limits shift downward to match this level (Best Fit to Average).

No values exceed the limits, so the result is a **PASS** verdict.

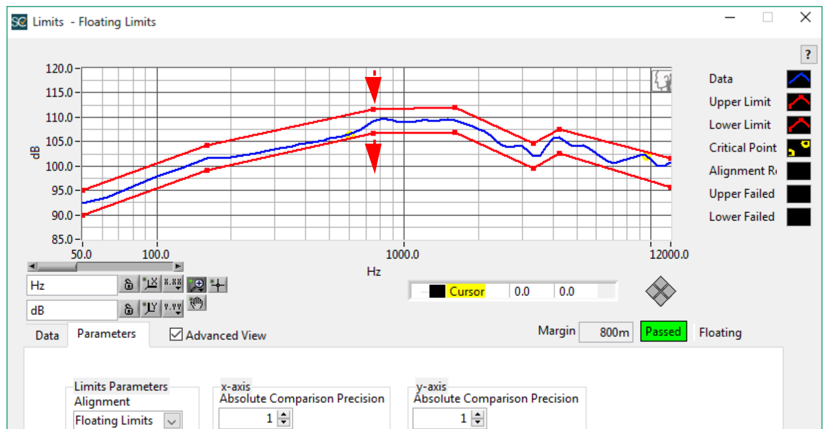


Figure 19-30: Floating Limits

Floating Data

The measured curve shifts upward to fit in between the absolute values of the limit curves. The 108.9 dB level at 1 kHz shifts upward to 113.9 dB. No values exceed the limits so the result is a **PASS** verdict.

A copy of the Floated Data is added to the Memory List with “Floated” appended to the data name.

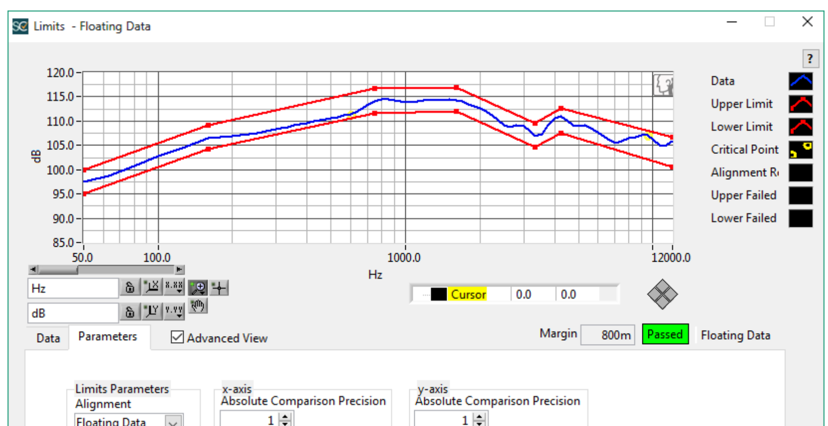


Figure 19-31: Floating Data

Display Editor and Memory List

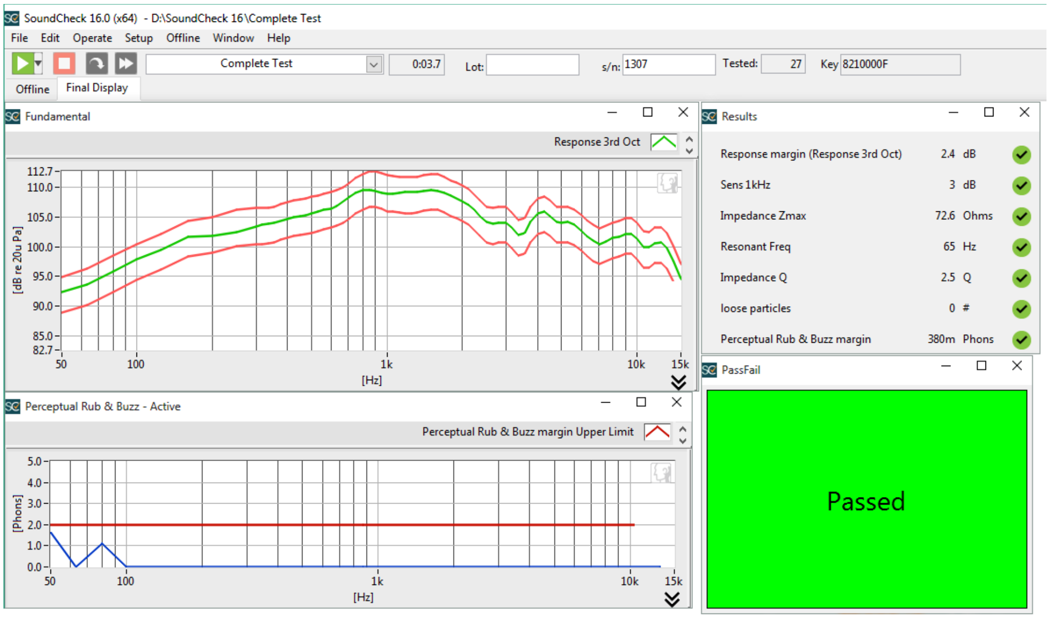


Figure 20-1: Display Step Example

The Display Editor (**Ctrl+Shift+D**) allows you to control the “on screen” display of:

- XY Graphs for data
- Waveform graphs
- Polar plots (optional)
- Result windows
- Table of values
- Text boxes
- Pictures

To edit an existing Display Step in a sequence, choose **Display** from the **Setup** menu.

See ***Display Editing on page 339*** for more information on displays and their properties.

Memory List

The control center of the *Display Editor* is the **Memory List**. Select **Memory List** from the **Setup** menu in the main SoundCheck® window (Ctrl+Shift+Y).

- To edit a Display, click on the desired Display Tab on the Main Screen.
- When a sequence contains multiple Display Steps you can select **Display** from the **Setup** menu and select a Display Step from the list (Ctrl+Shift+D). See ***Display Editing on page 339***.
- Open the **Memory List** to show the data available for displays.

The **Memory List** can produce seven types of *Display* windows as shown in ***Figure 20-2***. See ***Display Editing on page 339***.

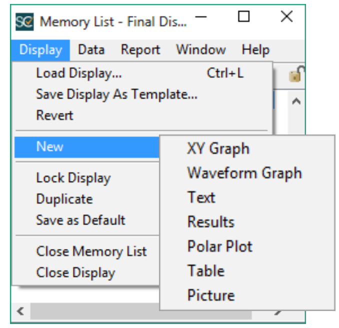


Figure 20-2: Display Drop down

Tabs (Curves, Values, Results and WFM)

Data in the *Memory List* is divided into four tabs: **Curves**, **Values**, **Results** and **WFM (Waveforms)**. The data content of these items is generated by the sequence can be accessed by clicking on any of these tabs. Names with an empty circle contain no data and act as placeholders when creating a sequence. After the sequence is run, the circle will be filled indicating that data is in memory and can be displayed. If one or more steps are added to the sequence after it is run, the associated names will be preceded by an open circle until the sequence is run. Data items can also be filled by clicking on the Apply button in some editors and by recalling data from disk.

Only certain types of data are appropriate for each display.

- **Curves** - Can be added to an XY Graph, Table or Polar Plot
- **Values** - Can be added to a Table
- **Results** - Can be added to a Results display or a Table
- **WFM** - Can be added to a Waveform Graph or a Table

For example, if the display consists of only XY Graphs, and a result item is selected from the **Results** tab of the *Memory List*, you will not be able to "Left-click and Drag" the item to a display.

Memory List Data Items

Prior to running a sequence, the only curves in the Memory List that contain data, are the calibration curves (filled circles at top of list as in ***Figure 20-3***).

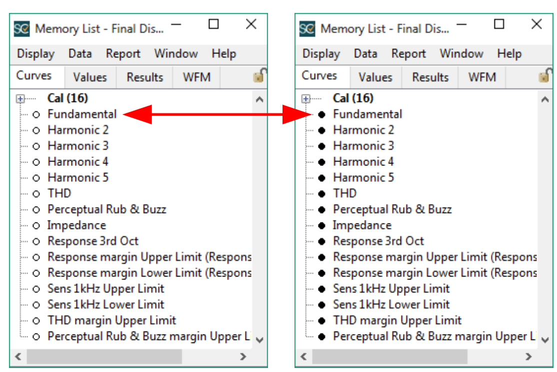


Figure 20-3: Empty vs Filled Circle

- **Empty Circle** - Item contains no data
- **Filled Circle** - Item contains data
- **Blue Group Heading** - Data added to Custom Group. See ***Sorting and Grouping on page 331***.
- **Check Mark** - Protected data
- **Empty Diamond** - Autoprotected & contains no data
- **Filled Diamond** - Autoprotected & contains data
- **Grey Group Heading** - Protected data. See ***Auto Grouping General Rules - Memory List on page 333***.

After the sequence has been run, the Memory List indicates that the curves contain data (filled circles).

Memory List Features

- Clicking on items in the Memory List does not break the relationship of those items to a display
- Simply double clicking on a memory list item will open it in its default display window
- You can use Drag and Drop to add other items to an Active Display
- Autogroup items in the Memory List and create Custom Groups (See ***Sorting and Grouping on page 331***)
- **Right-click** items to set properties (See ***Right-click - Memory List on page 323***)
- **Right-click** an item to display it on the *Active Display*, on a different display or create a new display
- **Alt+click** an item to add or remove it from an *Active Display*
- The Protect and Autoprotect functions append the serial number to the Protected Data name. This offers a quick and easy way to annotate data by typing a comment into the **Serial Number** field on the SoundCheck Main Screen.
- Open Multiple DAT Files in the Memory List
- Lock Display to prevent unwanted changes to displays - See ***Lock Display Off on page 324***
- Report Generator - Replaces printing to Word, Excel, HTML or image file from the Memory List Report Menu. Templates are now used to add flexibility and repeatability when printing data. See ***Report Generator on page 361*** for more information.

Note: **Word and Excel** must be installed on the computer. Cloud based installations are not recognized by SoundCheck.

- Set Text Size and Text Color of a Value Table. See ***Table - Preferences on page 358***.

Right-click - Memory List

Right-click a single item or several items in any of the Memory List Tabs and select:

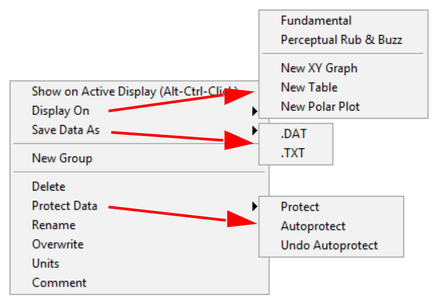


Figure 20-4: Right-click Menu

- **Show on Active Display** - For data that is not linked to the active display, select an item, Right-click it and select **Show**.
  - Select a range of items by selecting the top item, hold the Shift Key and select the bottom item. Right-click to Show or select another function.
  - An alternative is to hold the **Alt Key** and Left-click on an item to add it to the selected display window.
  - Hold **Alt** and **Ctrl** then Left-click on items to add them
- **Remove from Active Display** - Use the above methods to break the connection of the items to the active display
- **Display On** - Add the selected item(s) to an existing display
- **New Group** - Creates a new group in the Memory List

  See ***Sorting and Grouping on page 331***.

The following features are available in the Right-click menu as well as in the **Data** drop-down list: **Delete, Protect, Autoprotect, Undo Autoprotect, Rename, Overwrite, Units** and **Comment**. Explanations can be found starting under ***Data Menu on page 326***.

Memory List Drop-down Lists

Display Menu

Open display layouts, data, results and Waveforms previously saved to disk. **Lock Display** prevents display overwrite.

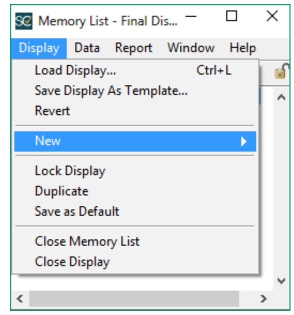


Figure 20-5: Display Menu

- **Load Display** - Opens a new display in place of the currently selected Display Tab.
  - This display is added to the active sequence and will overwrite the existing step.
  - The sequence editor is updated the next time the sequence is run or when it is saved.
- **Save Display As Template** - Renames the step in the sequence and saves the current display as a template. You are prompted for a file name and the location to save the file. The default location is the Display folder in SoundCheck. The sequence must be saved in order to remember the name change in the active sequence.
- **Revert** - Discard changes, returning display to last saved version
- **New** - Open a blank display for: XY Graph, Waveform Graph, Text, Results, Polar Plot, Table and Picture
- **Lock Display** - This control prevents you from closing the displays out of habit, as one would in other windows applications. Locked displays can be edited, but the changes are lost when the sequence is changed or the application is closed.

  When the lock is off, displays can be edited. For Technician and Operator login level, the lock is always on. The Lock feature can only be unlocked by the Engineer user level.

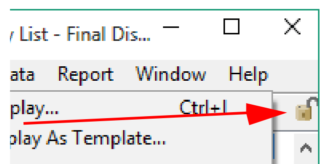


Figure 20-6: Lock Display Off

Display Lock Rules

When Display Lock is on:

- **Locking the display affects all displays in all sequences (until it is explicitly unlocked)**
- The display windows cannot be closed, resized or "minimized". The Minimize/Maximize buttons are removed from the display headers.
- New displays can be opened, but will not be saved in memory. The display will revert back to the “Last Saved State“ when the sequence is run. The ONLY exception to this rule is when the display is left open, if you have added or moved some display windows, they will stay in their new positions until they are closed (either by File->Close in the Memory List, by the sequence, or by another display opening).
- None of the temporary edits (like positioning) will be saved in memory. You will not be prompted to save display-related changes upon sequence exit
- Changes to the displays are not saved when the sequence is saved
- You can select Save Display as Template from the Memory List - Display menu
- A new display cannot be loaded into Active Display Tabs (you can load a display in the Offline Tab)
- The lock state is remembered in the ***SoundCheck 18.ini*** file
- Duplicate - Opens a copy of a display window, retaining the Properties of the display but not the link to Memory List items.

  This allows you to “Clone“ the Active Display type. This way, all displays within a sequence can have the same preferences. The Line Style settings; width, color, etc, are not duplicated.

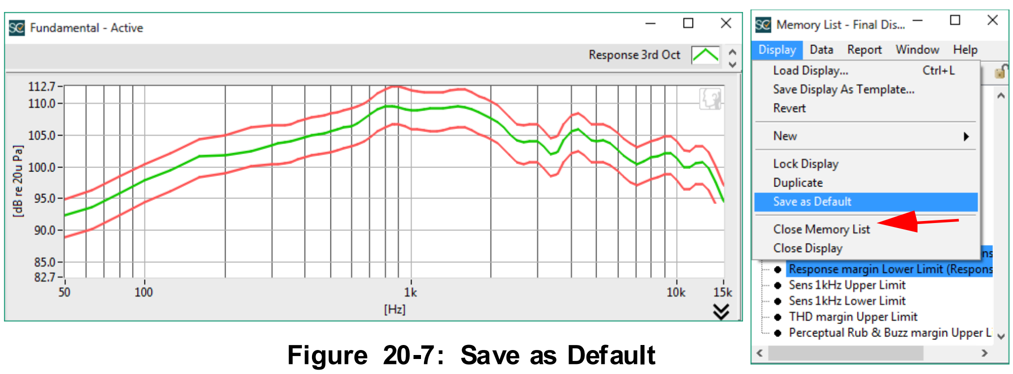


Figure 20-7: Save as Default

- **Save as Default** - This allows you to save the Display Preferences of the Active Display, as the default settings for that type of display. There is a default template for each type of display; XY Graph, Waveform, etc.
- **Close Memory List** - Closes only the Memory List, leaves display windows open
- **Close Display** - Closes all display windows and the Memory List. Allows you to clear the display windows from the desktop without deleting the display layout.

Data Menu

The **Memory List** can be used to recall saved *.DAT and *.RES files and display saved curves, values and results in the current *Memory List*. See ***Recall Editor on page 227***.

You can also open text (*.TXT) files. This will automatically open the **Data Import Wizard** *(See* ***Importing text from a saved file on page 611****)*.

- **Open Data** - Opens selected .DAT file

  Multiple DAT files can be opened at once from the Memory List. This is convenient when running statistics on a batch of curves and working with large numbers of files. If a DAT file contains multiple curves, a new *Group* is created in the Memory List, which contains the curves for that DAT file. When opening data, hold down the **Control Key** to select individual files. Click on the first item in the list and hold down the **Shift Key** while clicking on the last item to select that range of items.

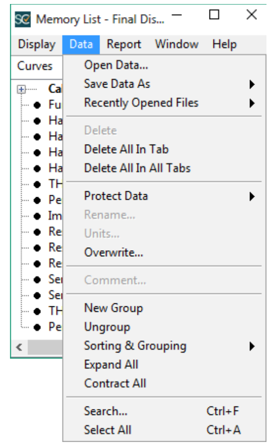


Figure 20-8: Data Drop down

  - All items opened from disk are Protected Data, denoted with a check mark.
  - Protected Data can only be added to Protected Groups. See ***Auto Grouping General Rules - Memory List on page 333***.
  - When opening a DAT file in the Memory List, a Protected Group is added to the Memory List that contains all the curves from the DAT file. The curves are not automatically added to the Active Display. The curves can be individually selected (or selected as a group) so that only the required curves show up in the display. This makes loading a large number of curves into memory much faster. Grouping makes it easier to manage a large number of curves.

Note: DAT files created with SoundCheck 18.1 are not viewable in versions of SoundCheck prior to and including SoundCheck 6.0x. The DAT file format was updated in SoundCheck 6.1.

  - When opening a multichannel WAV file in the Memory list, each WAV file channel is named using the file name and channel number. All files are automatically grouped as shown in ***Figure 20-9***.

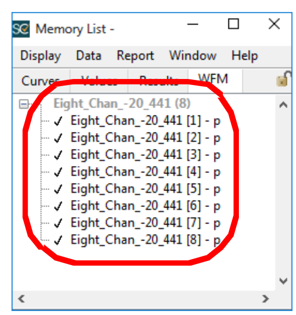
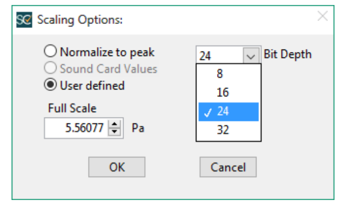


Figure 20-9: Multichannel WAV Files

- **Save Data as** - Saves the data selected in the Memory List to a .DAT and allows you to specify the name for the file
    - If the hardware is set to 24 bit, the waveform saved will be 24 bit
    - **Lock Display** prevents display overwrite
    - The item name is used as the default file name when saving an item to disk
    - Waveforms can be saved to a WAV file:
        - You are prompted to select the Bit Depth to save as and the scaling
        - Select two Waveforms (WFM), select "Save As” then select .WAV". This creates a single stereo WAV file. The first waveform selected will be the left and the second will be the right.
        - If three or more WFM files are selected, an error message will indicate that this is not allowed.



    - Save multiple Waveforms to a single *.WFM file.
    - You can save curves, values or results to a *.TXT file

If save to a *.TXT file is selected, you will be asked to make formatting choices, as shown in *Figure 20-10*.

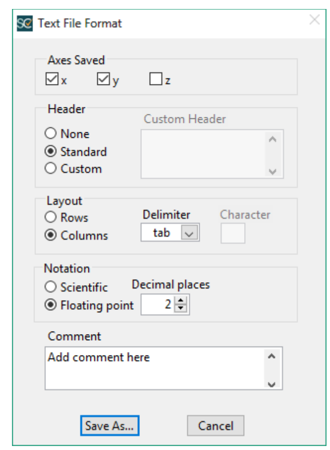


Figure 20-10: Formatting Choices

The options available in this window are identical to those of the **Autosave Editor** when saving to a Text file. You can choose which axes to save, whether to include a standard or custom header, to orient the data into rows or columns, and the notation and precision of the numerical values.

See *Autosave Editor on page 213*.

- **Recently Opened Files** - Shows the most recent Displays, DAT, RES and WFM files (useful when opening the same files on a regular basis)
- **Delete** - will remove only the selected item(s) from the active Memory List Tab (Available in both the **Right-click** and **Data Menu**)
- **Delete All In Tab** - removes all of the items from the active Memory List Tab (**Data Menu** Only)
- **Delete All In All Tabs** - removes all of the items from all tabs: Curves, Values, Results and WFM (**Data Menu** Only)

- **Protect Data** - Items in the Memory List can be protected so that subsequent runs of the sequence do not overwrite the data

  Any unprotected Curves, Values, Results and WFMs generated by a sequence are overwritten in memory each time the sequence is run. To keep the current data and results in memory as more tests are run, highlight the item(s), **Right-click** or click on the **Data Menu** and select **Protect**. Protecting items will keep them in the *Memory List* until they are deleted or the application is restarted.

  Protected Data will be shown in the *Memory List* with the same name as the original curve, with a suffix of "-p" appended to the data name. Protected items are also identified with a Check Mark to the left of the name, which only appears in the Memory List.

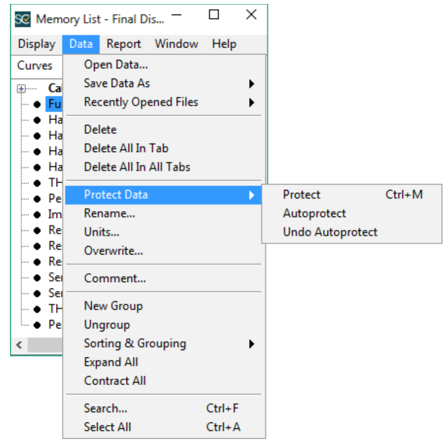


Figure 20-11: Protect Data

  You can also choose to Protect any information after the sequence has been run. Protected data is not discarded when changing sequences.

- The Protect and Autoprotect functions append the serial number to the Protected Data name. This offers a quick and easy way to annotate data by typing a comment into the **Serial Number** field.
- The protected data should be removed by using the **Delete** option in the **Data Menu** or the **Right-click Menu**.
- Protected Data can only be added to Protected Groups. See ***Auto Grouping General Rules - Memory List on page 333***.
- **Autoprotect data** - Used to keep data contents of items in memory every time a sequence is run

  When an item is highlighted in the *Memory List* and **Autoprotect** is selected, the icon to the left of the item(s) changes to a diamond shape (♦).

  Autoprotecting allows you to designate items that should be protected before the sequence is run. A copy of the item is generated each time the sequence is run and will be protected in the *Memory List* (until **Undo Autoprotect** is selected).

  Each time the sequence runs, the selected item will be protected and numbered in ascending order.

  If a serial number is entered in the Serial Number field, either manually or with a Serial Number step, that number will be appended to the Autoprotected items.

  Choosing Autoprotect for the Fundamental curve will create a diamond ♦ next to that curve, and all information will be protected.

  ♦Fundamental [L] is a place holder for information, indicating that the item is to be Protected. Each time the sequence is run the data for Fundamental [L] is appended to the end of the Memory List and protected.

  Protected Data will have checkmarks on the left, and items that are protected multiple times, with the same name, will be given a prefix number.

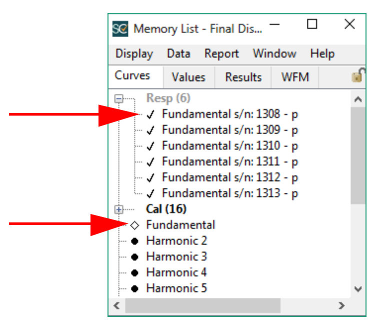


Figure 20-12: Fundamental Autoprotected

Note: If the Serial Number Step is used at the beginning of the sequence, the Autoprotected data has the serial number automatically appended to the data name, for all autoprotected items in the Memory List.

Protected Data - Protected Group Rules

When opening a DAT file (containing one or multiple curves) from the Data Menu in the Memory List, it is important to remember that some Offline, Limit and Post Processing functions may not be available.

- Limit Steps that manipulate data such as Aligned Data cannot be used on Protected Data
- Post processing of Protected Groups is not allowed but post processing of individual data items is allowed
- Using a Recall Step in a sequence will allow you to open data and perform post processing and limit functions with no restrictions

Autoprotect/Undo Autoprotect Rules

- The Autoprotect and Undo Autoprotect functions require that a Display Step is in the sequence
- The Display MUST be open to access the Autoprotect and Undo functions in the Memory List
- **Autoprotect with multiple Displays: The Autoprotect state of a Memory List item is <u>unique for each Display Step</u> in a sequence.** You should only autoprotect data on one display, unless you have a step in between the displays which regenerates the data such that there is new data to autoprotect on the second display.

Remember: Autoprotect renames the data, e.g.: 2 - Fundamental - p, 3 - Fundamental - p, etc.

- The display does not need to be configured to "**Display Step When Run**". See ***Configure Step on page 446***.
- Once the item is selected for Autoprotect, the Diamond marker is only visible while the display is open. Items are still Autoprotected, but not marked with a Diamond.
- Autoprotected data will remain in the Memory List after changing sequences. The data is deleted when SoundCheck is closed or the data is selected and deleted manually.

When **Autogrouping** is enabled, a "Protected Group" (Grey Text) is automatically created. See ***Figure 20-12***. See ***Auto Grouping General Rules - Memory List on page 333***.

- **Undo Autoprotect** - Selected Items are no longer Autoprotected in subsequent measurements

To stop Autoprotecting an item, highlight the original name (with the diamond to the left [♦]) and select **Undo Autoprotect**. The Protected Data will remain in the *Memory List*, but all measurements following this action will only remain in the list until the next time the sequence is run. Future measurements will no longer protect the item automatically.

The data and results are in Memory and will only be saved to disk by using commands in the **Data Menu** or using one or more Autosave Steps.

- **Rename** - Creates a Protected copy of the item and prompts you for a new name

  The new item appears in the Memory List with a Check Mark and a "-p" suffix.

  A Memory List item can be renamed at any time. Renaming an item will automatically protect it, since a curve of that name may not exist in the sequence. Select the item(s) from the current tab to rename. **Right-click** an item and select **Rename**. (Or select **Rename** from the **Data Menu** of the *Memory List*.)

- **Units** - To increase flexibility, the units of Memory List Items can be temporarily changed

  These changes disappear each time the sequence is executed. To retain the Unit changes the Item must be Protected. Otherwise, the unit information will revert to the values created by the sequence, on the next sequence run. Unit changes are not available on Autoprotected Items. The sequence Units will be used in the Autoprotected item.

TIP: To temporarily change the Units of a curve in a display (or table), Right-click the curve in the Memory List and select **Units**. The new Units will appear in the display. This is a one time change. The next time the sequence runs, the units will revert back to the original units.

- **Overwrite** - Allows you to select a Subject item from the Memory List and then choose another item as the Target to be overwritten

Note: In general, overwriting a correction in or out curve with a Reference Calibration File is handled in the Calibration Editor by using the “Copy from Memory List” button.

In special cases, you might want to overwrite a curve in the Memory List manually.

The data of a Reference.DAT file (magnitude and phase of a curve) can be used to replace the contents of a Correction In or Out.DAT in the Memory List.

This can be used in special cases for importing custom correction curves and/or EQ curves.

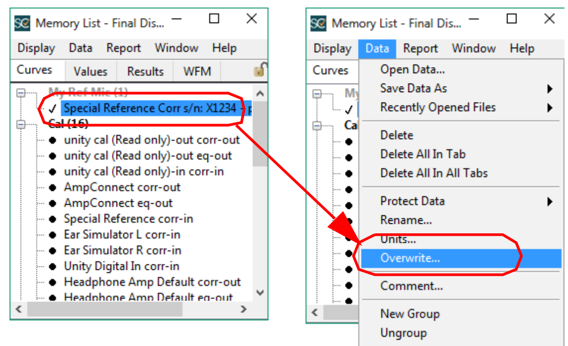


Figure 20-13: Overwrite a curve - step 1

1. In ***Figure 20-13***, we have loaded a Microphone Correction Curve into the Memory List (See ***Data Menu on page 326***).
2. **Right-click** the curve and click **Overwrite** or select **Overwrite** from the **Data** drop-down list.
3. In ***Figure 20-14*** the Calibration Curve for the Input is selected:

   Special Reference corr-in
4. OK to replace the data of this curve with the data from “**Special Reference Corr s/n: X1234**“.
   - The name of the target curve does not change, only the data changes
   - The new curve can be used in the Calibration Editor to correct for the response of the microphone in future measurements

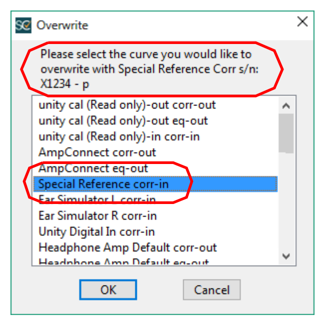


Figure 20-14: Overwrite - step 2

- **Comment** - Comments can be attached to Curves in the Memory List. They appear in green text, to the right of the item, in the Memory List.
  - Comments are normally added to curves or items that have **already been protected** or recalled from disk
  - Comments on an Autoprotected source curve (♦) only appear on the next curve that is protected. Subsequent runs will have no comment. This will also leave a copy of the comment on associated Limits. (Remove these by entering a blank comment on the source curve)
  - The comment is saved with the data when saved as a .DAT file
  - When importing the .DAT file, the comment will still be attached to the data

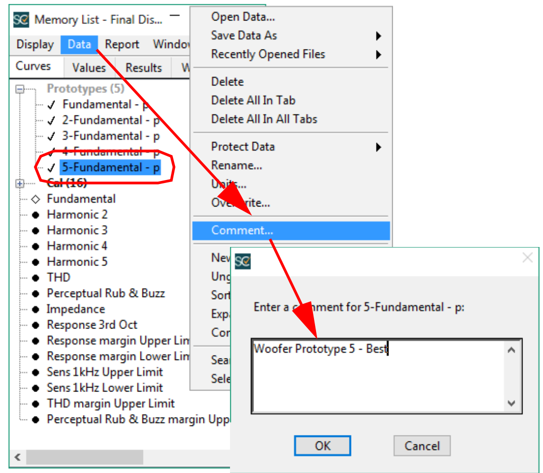


Figure 20-15: Comment Protected Memory List Item

Right-click an item and select **Comment** or select an item and then click on **Comment** from the **Memory** drop-down list. See ***Figure 20-15***. The comment can then be entered in the editing window.

Long comments will not "text wrap" in the Memory List, so the entire comment may not be easily visible. Use the **Comment** function to view the text in the editor window.

Important! Comments should only be added to **Protected** or **Recalled items** in the Memory List. Adding comments to Unprotected items is not recommended. Comments added to empty items, before sequence run, will be erased. Comments added to Unprotected items after sequence run will duplicate the comments to related Limits and other related items.

Sorting and Grouping

In order to improve organization and ease of use, items in the Memory List can be grouped and sorted.

- Data can be sorted alphabetically by **Name** or by the **Order** in which generated by sequence
- Data cannot be manually sorted by moving an item in the list
- Data can be set to Auto Group by Category or Step
- Sort and Auto Group are continuously applied to all data in the Memory List so data is automatically sorted into the correct place
- **Custom Groups** can be created allowing you to put selected data into a custom named group
  - Items can be added to a Custom Group through the menu option or by clicking on an item and dragging it to a group
  - Items can also be removed from a Custom Group by clicking and dragging them to the root of the Memory List Tab
  - Custom Groups can also have Subgroups
  - Custom Groups always appear at the top of the list
  - Group names should not contain Parenthesis, ( ), or other types of brackets

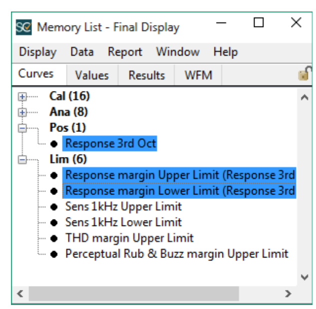


Figure 20-16: Group Curves

- Protected Data can only be added to Protected Groups. See ***Auto Grouping General Rules - Memory List on page 333***.
- Grouping is constantly updated as steps are added to a sequence
- Sorting and Grouping functions are independent for the four items tabs: Data, Values, Results and WFM
- **New Group** - Selected items will be added to a **Custom Group** and you are prompted to enter a name for the group
  - Uncheck "Group Selected Items" to create a Group without adding items to it

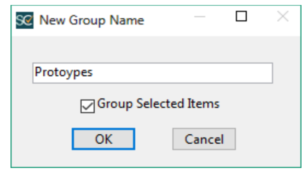


Figure 20-17: Group

- **Ungroup** - Removes the Group heading and returns the Items to the root of the Memory List
- **Expand All** - Opens all Groups in the Memory List (show all Items)
- **Contract All** - Closes all Groups in the Memory List
- **Sorting and Grouping**

  Clicking on any of the following functions will Undo all Autogroup in the current Tab. Sort functions will return the grouped items to the root of the Memory List. This does not effect **Custom Groups** (User Groups).

  - **Sort By Order** - Arranges all items in the current Tab in the order created by the sequence
  - **Sort By Name** - Arranges all items in the current Tab in alphabetical order
  - **Autogrouping Off -** Autogroups are dissolved, returning Autogrouped items in the current Tab to the root of the Memory List and sorted according to the Sort function (Custom Groups are not effected)
  - **Autogroup By Category** - Automatically creates groups for items in the current Tab based on the data type of the Items
  - **Autogroup By Step** - Automatically creates groups for items in the current Tab based on the Sequence Step that created the items

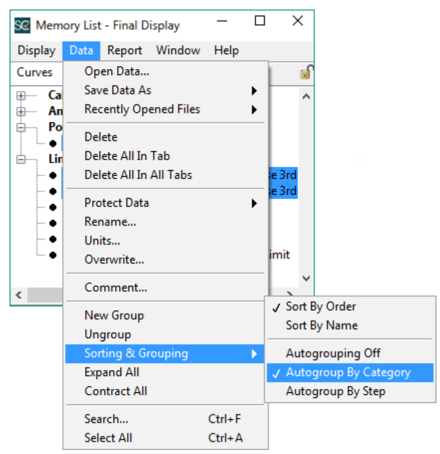


Figure 20-18: Sorting

Auto Grouping General Rules - Memory List

- Memory List items can be moved to Custom Groups. The group title is in **Blue Text**.
- Protected Data (noted with a check mark before the curve name) can only be added to Protected Groups indicated by **Grey Text**, not Custom Groups (Blue Text)
- Turning off auto grouping does not remove Custom Groups
- Memory List items cannot be moved between Auto Groups
- Switching between sort by category or sort by step does not break sorting of Custom Group items
- Memory List items can be moved back to auto groups if the Custom Group remains (is not deleted)
- When all items in a Protected group are deleted, the group is not visible in the Memory List until new items fill the group
- When saving an Autogroup of items, the default file name will be the first item in the group

- **Search** - The **Data** drop-down list has a search function feature to find and highlight all items which match a search string. Advanced searching through the use of regular expressions is also possible as described in the context sensitive help window of the search string (See ***Figure 20-19***). This is particularly useful when the Memory List contains a large number of curves such as in production line applications.

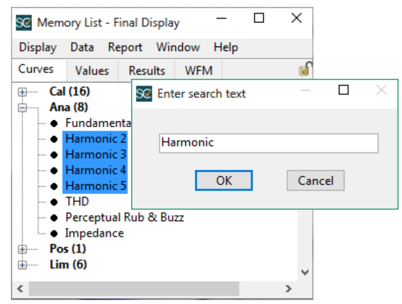


Figure 20-19: Search Function

Example: A user wants to display a group of curves from a large list, all on the same graph, even though each curve has an appended number, e.g., 2-Fundamental, 3-Fundamental, etc.

Use the Memory List **Search** function to find all the curves with the characters, "Fun". Then **Right-click** on any of the highlighted curves and select **Display On - Graph or New Group**.

- **Select All** - Highlights all of the items in the current *Memory List* tab

  From the **Data Menu** click on **Select All** (not available in Right-click Menu). This is useful for adding all Results to a Results display for an Overall Pass/Fail verdict, or to add a long series of curves to a XY Graph or Polar Plot.

Report Menu

- **Load Logo** - select .BMP or .JPG file for the heading of a report
- **Report Setup**- Opens the Report Generator

  Create and print company reports in Word, Excel, HTML and image files. The Report Menu is available in the Memory List - Report Menu, and in each of the Display Windows. See ***Report Generator on page 361*** for more information.

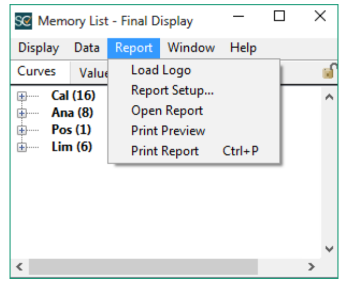


Figure 20-20: Report Generator

Note: **Word and Excel** must be installed on the computer. Cloud based installations are not recognized by SoundCheck.

- **Open Report** - Opens a report based on the current display and the settings in Report Setup
- **Print Preview** - Opens the report in Print Preview
- **Print Report**- Sends the report to the default system printer

Right-click

Right-click a single item or several items in any of the Memory List Tabs and select:

- **Add/Remove from Active Display** - (Press Alt+Left-click on an item to break the connection of all items to the active display)
- **Display On** - Add the selected item(s) to an existing or new display window

  The options vary depending on which Memory List Tab is selected. The example in ***Figure 20-21*** shows that a curve can be added to:

  - One of the current display windows in the "Display On" list
  - **New XY Graph**- Creates a graph for the selected items
  - **New Table**- Creates a table for the selected items
  - **New Polar Plot** - Creates a polar plot for the selected items
- **New Group** - Creates a new group in the Memory List

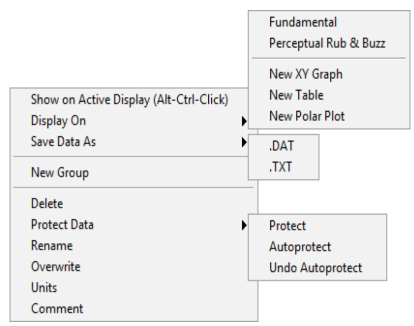


Figure 20-21: Right-click Menu

The following features are available in the **Right-click** menu as well as in the **Data Menu**. They follow the same explanations and rules. For more information see ***Data Menu on page 326***.

- **Rename** - Creates a Protected copy of the item and prompts you for a new name
- **Overwrite** - Allows you to select a Subject item from the Memory List and then choose another item as the Target to be overwritten
- **Delete** - will remove only the selected item(s) from the active Memory List Tab (Available in both the Right-click and Data Menus)
- **Units** - To increase flexibility, the units of Memory List Items can be temporarily changed
- **Comment** - Comments can be attached to Curves in the Memory List
- **Protect** - Items in the Memory List can be protected so that subsequent runs of the sequence do not overwrite the data
- **Autoprotect** - Used to keep data contents of Items in memory every time a sequence is run
- **Undo Autoprotect** - Selected Items are no longer Autoprotected in subsequent measurements

Right-click a Group

- **Show on Active Display** - Add all items in group to display
- **Display On** - Select display to add group items to
- **Save Data As** - Save all items in group as selectable data type
- **Ungroup** - Removes the selected Custom Group heading and returns the Items to the root of the Memory List (Will not remove Auto Groups)
- **Delete Group and Data** - Deletes the selected Custom Group heading and all of the Items under that heading (Cannot be Undone)

  Will not delete Autogroups but will clear data in items
- **Rename** - Change the name of the selected Group
- **New Group** - Creates a new Group and adds the selected Group to it

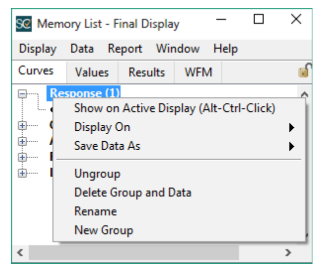


Figure 20-22: Right-click on Group

Window Menu

- Full Size - Set the SoundCheck Main Screen to fill the computer desktop (Does not allow application window to be resized)
- Clicking on an Open Window Title brings that display window to the front - useful when smaller windows are inadvertently hidden behind larger display windows

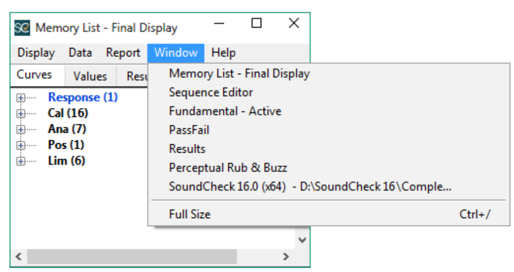


Figure 20-23: Window Menu

Help Menu

- Context Help menus are available for many items in SoundCheck. Press **Ctrl+H** to open the *Context Help* window or select it from the **Help** drop-down list. This will give information on the last item the mouse has scrolled over. Press **Ctrl+H** again to make the *Context Help* window disappear, or click the close box button in the upper right hand corner.

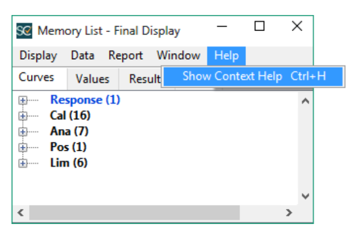
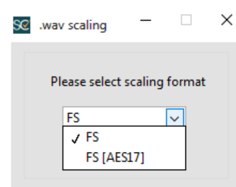


Figure 20-24: Help Menu

WAV File Types

- SoundCheck will read and write 8, 16, 24 and 32 bit WAV files.
- 8 bit WAV files in PCM format are stored as unsigned integers, each sample ranging from 0-255. However, there are two other compressed formats, A-law and mu-law (commonly used in telephony). These files are stored as 8 bit files, but decompress to 16 bit Integer when read.
- SoundCheck will read 8 bit files in all three formats, PCM, A-law and mu-law, but writes only in PCM format. See table below.
- As of SoundCheck 16.01, WAV files that do not conform to the standard WAV file header can be read
- As of SoundCheck 17, **importing** multichannel WAV files is supported in the Memory List, Stimulus Editor and Signal Generator

WAV File General Rules



- When opening a WAV file, SoundCheck will prompt you to select units of FS or FS(AES17)
  - 'FS' – SoundCheck default value, the max amplitude of a digital sine wave is -3 dBFS
  - 'FS (AES17)' – Value corresponding to AES17 standard definition, the max amplitude of a digital sine wave is 0 dBFS
- When opening a stereo WAV file, SoundCheck will automatically split the file into two waveforms in the memory list, adding [L] or [R] to the file names
- 16 bit files are stored as 16 bit Integer PCM format WAV. SoundCheck reads and writes 16 bit WAV files in PCM format.
- 24 bit files are stored as 32 bit Integer PCM format WAV. SoundCheck reads and writes 24 bit WAV files in PCM format.
- 32 bit files are stored as 32 bit Integer PCM format WAV ranging from -2147483648 to +2147483648. SoundCheck reads PCM and IEEE float but writes only PCM format.
- The sample rate of the WAV file must match the sample rate of the System Hardware Configuration

WAV File Format Table

SoundCheck reads and writes WAV files according to the following table.

Bit Depth	Read	Write
8	8 bit PCM uncompressed unsigned 8 bit A-law compressed unsigned 8 bit mu-law compressed unsigned	8 bit PCM uncompressed unsigned
16	16 bit Integer PCM	16 bit Integer PCM
24	32 bit Integer PCM	32 bit Integer PCM
32	32 bit Integer PCM 32 bit IEEE Floating Point	32 bit Integer PCM

A-law and mu-law WAV files are mostly used in Telephony ***(https://en.wikipedia.org/wiki/G.711)***.

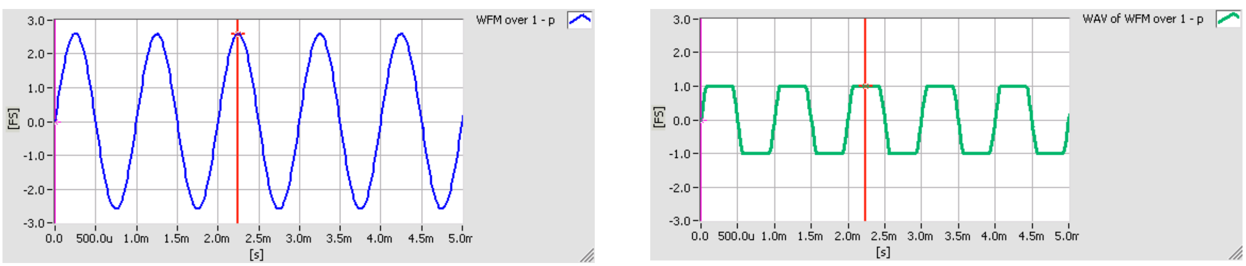
Most popular audio editing programs can write WAV files in these formats.

WAV File Scaling

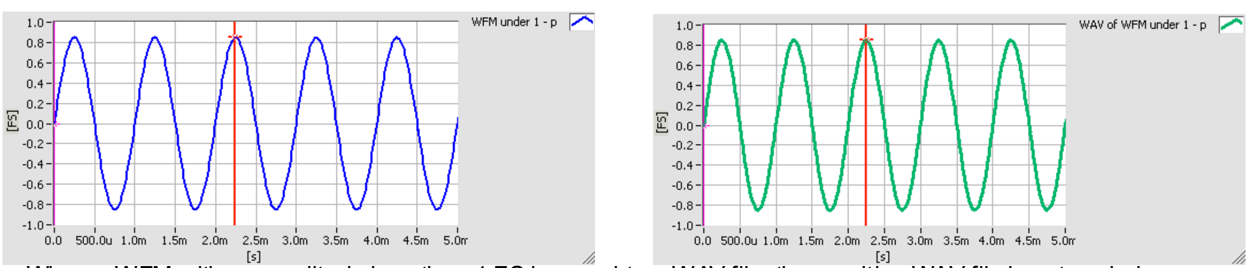
In SoundCheck versions 6.11 and higher, WAV files are scaled according the following rules. (In versions 6.00, 6.01, and 6.10 WAV files were normalized to +/- full scale deflection when saved).

Rules for scaling Waveforms when they are saved as WAV files:

- A Waveform is not scaled or normalized if the amplitude unit is FS, FSD, dB re:FS, or dB re:FSD.
- Waveforms with dB re:[any unit] get converted to their linear version upon being saved as a WAV.
- If the waveform has units based on Full Scale Deflection (dB FS or dB FSD), but the Absolute Linear Peak Amplitude is greater than 1.0, the data of the resultant WAV file will be clipped.
- For any waveform with units that are not based on FSD (e.g., Pa, dB SPL, V/V, etc.), data will be normalized to +/- 1.0 FS (or 0 dB FSD), which is the maximum allowed amplitude of a WAV file. The scaling either increases or decreases the amplitude values of the data so that the Peak value of the data in the WAV file is +/- 1.0 FS.
- It is possible for a SoundCheck user to convert the units of a waveform from anything to FS (either directly from the Memory List or via Post-Processing), so its peak amplitude could be greater than +/- 1.0. In this case, no scaling occurs. Saving such a waveform as a WAV file means that there are points lying outside the allowable range. Upon saving this WAV file, these points will be coerced to either +1.0 or -1.0 FS (i.e., clipped). See ***Figure 20-25***.



When a WFM with an amplitude greater than 1 FS is saved to a WAV file, the resulting WAV file is clipped.



When a WFM with an amplitude less than 1 FS is saved to a WAV file, the resulting WAV file is not scaled.

Figure 20-25: WFM saved as WAV examples

- When acquiring data on a digital channel or creating a stimulus for a digital channel, the waveform will, by definition, have an absolute peak amplitude of less than or equal to 1.0 FS. Saving these WFMs as WAV files will not result in clipping.
- When saving the waveform as a WFM, the data is never scaled, normalized, or clipped.
- In SoundCheck, when a WAV is opened from disk, it has units of FS. (Peak allowable range of +/- 1.0)

More information on the use of WAV files in SoundCheck can be found in: ***WAV File Excitation on page 129*** and ***WAV File playback on page 462***.

Displays

The SoundCheck Main Screen tool bar allows you to quickly switch between display steps when viewing sequence results.

Offline Tab

The **Offline Tab** is available with or without a sequence loaded. This allows you to open, process, and view data without loading a test sequence. In the Offline Tab, data can be examined or analyzed without affecting the layout of the display steps of the sequence. It minimizes the risk of accidentally editing sequences. In this mode, virtual instruments such as the signal generator, multimeter etc. can also be used, without a sequence being open.

This is especially useful for those who view data on a regular basis:

- Development engineers who run a series of experiments and then post-process the data
- Production engineers who need to view large sets of data from the factory

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Note: The Lock Display function does not apply to the Offline Tab.

Display Tabs

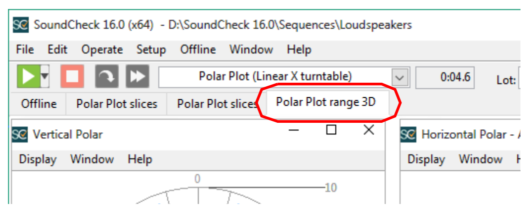
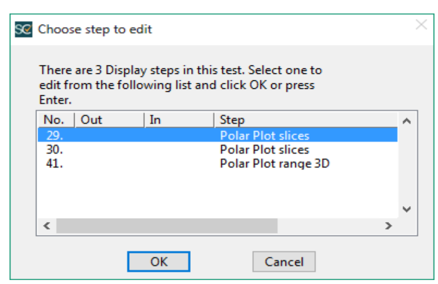


Figure 20-26: Display Tabs

The Display Steps in a sequence are always available via the Display Tabs on the SoundCheck Main Screen as shown in ***Figure 20-26***. This allows you to easily manage multiple displays in a sequence. When a sequence is loaded, new tabs are added for each display step in the sequence.

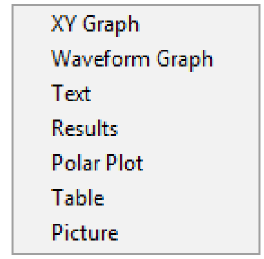
- When the Memory List is Unlocked, any changes to the display windows, in any of the tabs, are saved when the sequence is saved. See ***Lock Display Off on page 324***.
- If you remove the display windows and save the sequence, those windows are deleted from the display tab
- In a sequence with multiple displays, clicking **Setup** and then **Display** opens a drop-down list of available Display Steps to choose from. Selecting a step opens that Display Tab.
- In a sequence with only one display, simply click on that Display Tab to edit
- You can Revert the display to its most recently saved form
- Opening a Display Step DOES NOT open the Memory List
- Opening the Memory List DOES NOT open a Display Step
- Closing the Display Step will close the display windows for that tab as well as the Memory List. This is not saved with the sequence.



For more information see ***Memory List Display Menu on page 324***.

Display Editing

You can easily display info from the Memory List in a sequence by adding a Display Step. Under the **Display Menu - New**, you can choose from six types of displays:



- XY Graph
- Waveform Graph
- Table
- Results
- Polar Plot (Requires optional module 2011)
- Text Box
- Picture

A Display Step can have many display windows, in any combination. The only limitation is in organizing the displays so that they can be seen on the computer screen.

- To view the data associated with the items in the *Memory List*, select one or more items under the **Curves**, **Values**, **Results** or **WFM** tabs.
- Select a single item by Left-clicking on it once
- To select more than one item, hold down the Control key while making multiple selections
- To de-select an item, click on it a second time while holding down the Control key
- To select a range of curves highlight the first curve in the range, hold down the Shift key and then select the last curve in the range

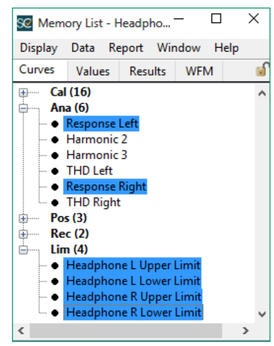
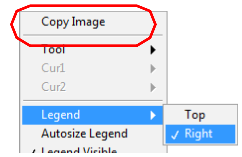


Figure 20-27: Multiple Items Selected

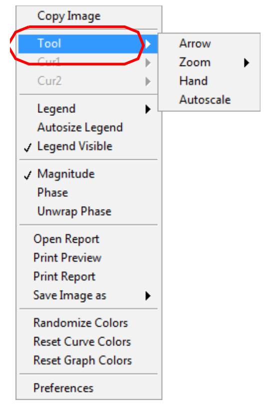
Right-click Graph

Copy Image

This takes a screen shot of only the selected graph window and puts it in the Windows Clipboard. This graphic can be cut and pasted into a document or image editor to save it.



Tool

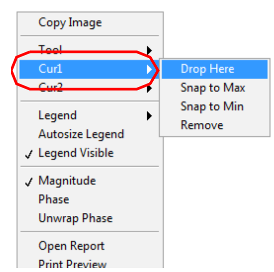


- Arrow - When active you can click on the XY axis to change the values. You can also click on the cursor to drag it to a location.
- Zoom - Click to zoom into a specific part of the display
  - XY allows you to drag a box to select zoom area
  - X allows you to drag between two points on the X Axis
  - Y allows you to drag between two points on the Y Axis
- Hand - Click on the graph and move it within the graph window
- Autoscale - Select to set both X and Y axis to the maximum extent of the graph window

Cursors

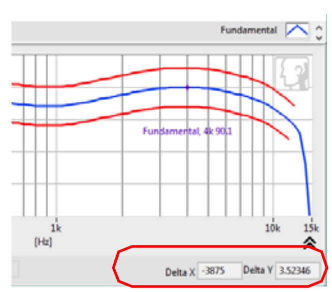
Cur1 and Cur2

Right-click a graph, Select Cur1 or Cur2, then select:



- Drop Here - Places the cursor on the closest curve
  - Drag the cursor to any point on the curve
  - Drag the cursor to a different curve
- Snap to Max - The cursor jumps to the maximum value of the curve
- Snap to Min - The cursor jumps to the minimum value of the curve
- Remove the cursor from the graph window

The cursor can be moved by Left-clicking and dragging the cursor marker (+) to the desired point on the curve. The cursor will snap to the closest curve you drag to.



- The XY coordinates of a cursor are displayed next to the cursor on the display. The XY coordinate box can be moved so that it does not cover the graph line.
- Cursors can be placed on the graph to note specific coordinates or find the X/Y difference between 2 cursors.
- Delta X and Delta Y are displayed in the bottom right corner of the graph window.

Legend

Legend - The curve legend can be placed on the Top of the graph or on the Right as shown in ***Figure 20-28***.

Autosize Legend - Insures that the full title of curves or waveforms in the legend is visible. When Autosize is disabled, placing the mouse pointer over a curve or waveform name will display the entire name.

Legend Visible - Allows you to hide the legend

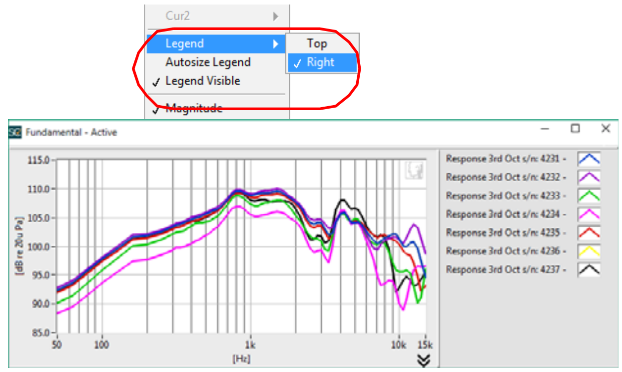
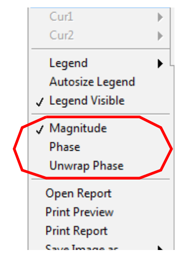


Figure 20-28: Legend On Right

Magnitude

- Shows the XY relationship of the selected curve



Phase

- Shows the XZ relationship of the selected curve

Unwrap Phase

- Allows you to show the phase as a continuous plot, even if the phase exceeds $\pm$ 180 degrees.

Note: Magnitude, Phase and Unwrap Phase are not available on Waveform Graphs.

In a wrapped phase graph, the phase offset of the device under test is only expressed between 180 and -180 degrees.

If the device has a fixed delay that exceeds the wavelength of the highest frequency of interest, then the results in the phase graph will "wrap around" as shown in ***Figure 20-29***.

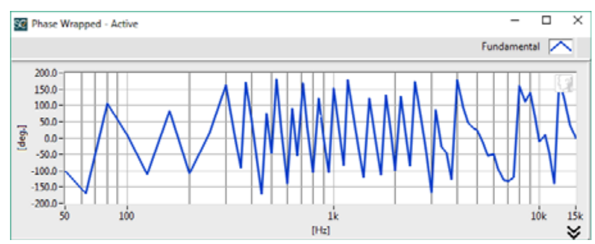


Figure 20-29: Wrapped Phase

When you un-wrap the phase you take this into account and plot phase offset values greater than 180 degrees.

This allows phase to be shown as a continuous plot.

Figure 20-30 shows the same Fundamental phase curved with Unwrap Phase selected.

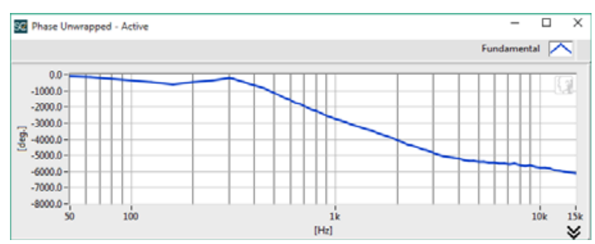


Figure 20-30: Unwrap Phase

Note: Unwrap Phase can also be accomplished in a Post Processing step if you want to save the Unwrap Phase data or apply Limits to Unwrapped phase. See ***Unary on page 240***.

Open Report

- Follows same rules from Memory List Report Generator. See ***Report Generator on page 361***.

Print Preview

- Opens report in print preview

Print Report

- Sends report to selected default printer. No print preview or print setup.

Save Image as

- You can save the current view of the selected display window as a JPG or PNG file. The graph header and footer can be expanded so that they are included in the image.

Randomize Colors

This function is especially useful in displays with a large number of curves.

Right-click the graph area and select **Randomize Colors**.

It creates a random list of colors for the curves displayed in the active graph. These new colors are linked to the curves in the Memory List. If one of the curves is added to a different display, it will use the same color assigned by “Randomize Color”.

For information on graph colors see ***Default graph palette on page 48***.

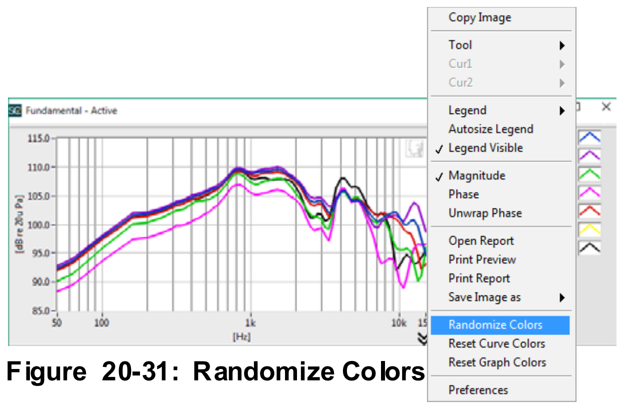


Figure 20-31: Randomize Colors

Reset Curve Colors

Right-click the graph area and select **Reset Curve Colors**.

Sequences or data from prior versions of SoundCheck will have their colors already assigned. When these sequences are loaded, they will not use colors from the custom color palette. You can reassign these curve or waveform colors by selecting the "Reset Curve Colors". When you reset colors, the curves and waveforms will use the colors defined by the **Default graph palette** in SoundCheck Main Screen Preferences > Display tab.

See ***Display on page 48***.

Reset Graph Colors

Right-click the graph area and select **Reset Graph Colors** to set the background and grid line colors to those selected in SoundCheck Main Screen Preferences > Display tab.

This allows you to reset the background, gridline, and cursor colors for a display so that they match the color palette defined in the SoundCheck display preferences. This may also be useful when loading displays from pre-existing sequences where the display theme does not match new sequences.

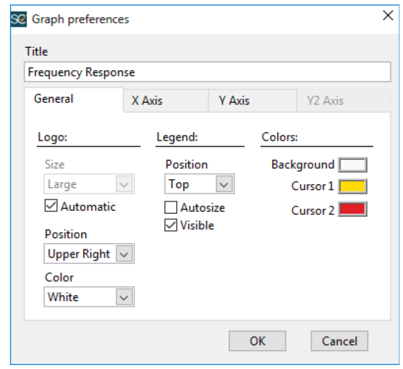
See ***Display on page 48***.

Preferences

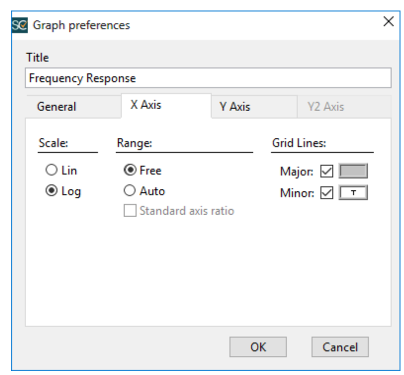
Right-click the graph window and select **Preferences**. Click on a tab to change graph settings.

As of SoundCheck 17, new display window backgrounds are white. Displays in existing sequences are not affected.

General

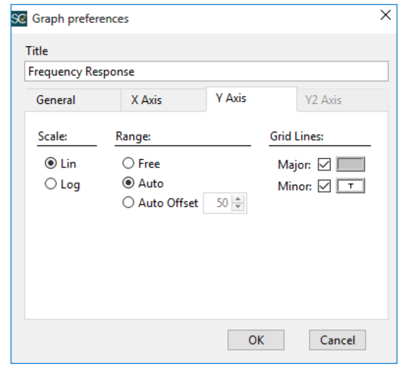


- **Title** - Enter a name for the graph window. This name appears on exported graphics for reports.
- **Logo** - The Listen Logo is displayed in all graph windows
  - Autosize - The logo will change size as the graph window is changed. When Autosize is not selected you can select fixed sizes of small, medium and large.
  - Position - Select: Upper Right or Left, Lower Right or Left
  - Color - Select black logo background or white logo background. The default is white which is better for printing.
- **Legend**
  - The Legend can be Autosized so that it alway shows all of the curves selected for a graph. This disables the re-sizing bar on the graph.
  - Visible allows chose whether or not to show the legend
  - Legend Position allows you to place the legend on top or on the right side of the graph
- **Colors** - Select colors for: Background, Cursor 1, Cursor 2, X and Y Major and Minor divisions. See ***Default graph palette on page 48***.



- **X Axis**
  - Scale
    - Select Linear or Log Axis
  - Range
    - Select Free or Autoscale
    - Autoscale will automatically scale the graph to fit the full X Axis extents of the graph information
    - Free allows you to set fixed Max and Min values in the graph window
    - Standard axis ratio (requires that Auto Offset is on in Y Axis Range control). See ***Standard Axis Ratio on page 344*** for instructions.
  - Grid Lines
    - Show Major and/or Minor Grid line check boxes
    - Major and Minor Grid Color selection boxes. By default the minor grid lines are set to transparent (T).
    - When the Major or Minor divisions are not selected in the X Axis/Y Axis Tabs, the color is set to transparent.

- **Y Axis**
  - Scale
    - Select Linear or Log Axis
  - Range
    - Select Free or Autoscale
    - Autoscale will automatically scale the graph to fit the full Y Axis extents of the graph information
    - Free allows you to set fixed Max and Min values
    - Auto Offset - Used to set the Y Axis scale to a fixed range. The graph will always perform a "Best Fit" on full extents of the graph information. In this example the Y Axis range is 30 dB.



Note: Y Axis 2 is available if two graph items have different Unit Sets. Only two unit sets are supported on a graph window. If an third type is added you will see the following message: "Cannot plot these data together! You must select curves with a maximum of two different unit sets, including phase."

  - Grid Lines
    - Show Major and/or Minor Grid line check boxes
    - Major and Minor Grid Color selection boxes. By default the minor grid lines are set to transparent (T).
    - When the Major or Minor divisions are not selected in the X Axis/Y Axis Tabs, the color is set to transparent.

Standard Axis Ratio

This allows the XY Graph to display a user-defined dB range per decade of frequencies. The example below insures that the XY Graph will always be 50 dB per decade, regardless of actual size of the screen.

Note: IEC 263 specifies 10, 25, and 50 dB/decade. B&K chart paper conforms to this standard.

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1. **Right-click** the display and select **Preferences.**
2. Click on the **Y Axis** tab, click the **Auto Offset** radio button and enter the proper decibel range as in ***Figure 20-32***. The typical ranges are 25 or 50 however any value can be used.

   In this example the Y Axis will always have a scale of 50 dB.

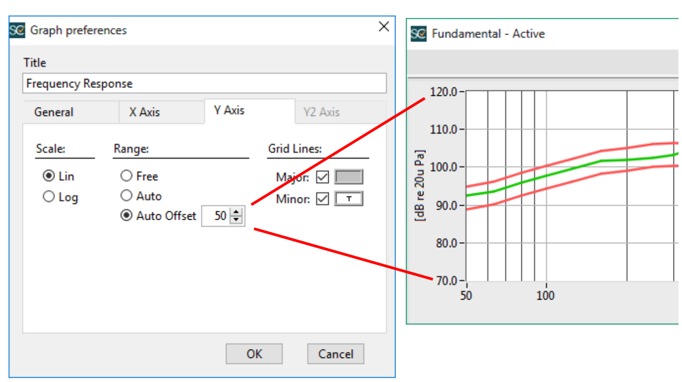


Figure 20-32: Y Axis Auto Offset

3. Select the **X Axis** tab, click **Standard axis ratio** as in ***Figure 20-33***. Make sure that the X Axis is set to Free and NOT Autoscale.

4. Click **OK** to close

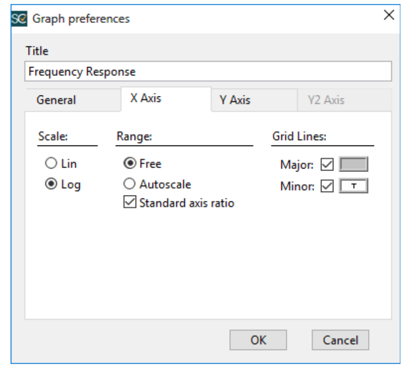
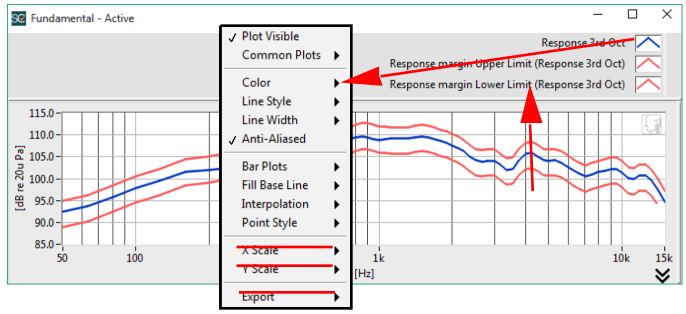


Figure 20-33: X Axis Standard Axis Ratio

Graph Header

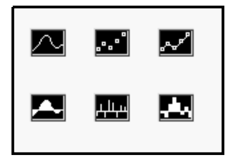


Click the up/down arrows to the right of the Plot Legend to select other curves.

You can also Left-click and pull down the divider bar at the top of the graph to expand the header to show all the curves in the Plot Legend.

Right-click a graph line in the Plot Legend to adjust its properties.

Plot Visible - The selected plot can be hidden.



Common Plots - Select the line style for the selected item: Continuous Line, Dots, Connected Dots, Fill Baseline, Vertical Lines and Bar Plot.

Color - You have several options using the color spectrum bars: Greyscale, Soft Colors and Bright Colors. You can also click on the **artist palette** in the lower right and create a custom color. The selected color is shown in the bottom left hand box.

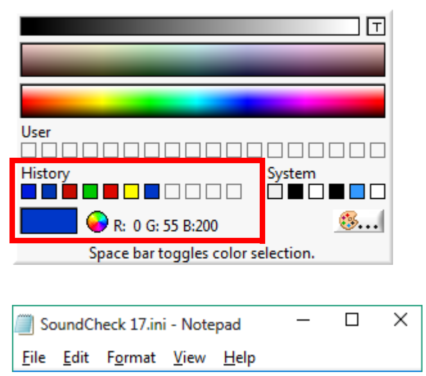
Color History

You can view and edit the Color History of the Display Editor in the **SoundCheck 18.ini** file.

This shows a list of up to 11 colors used in the most recent run sequence.

Colors in the list follow the standard HEX color code with “00” prepended to the value.

The color history and INI file text are shown in ***Figure 20-34***.



[SoundCheck 18.1]
colorHistoryItemA=000411D4
colorHistoryItemB=00CA0D00
colorHistoryItemC=00E84D00
colorHistoryItemD=00FFD900
colorHistoryItemE=0012FF00

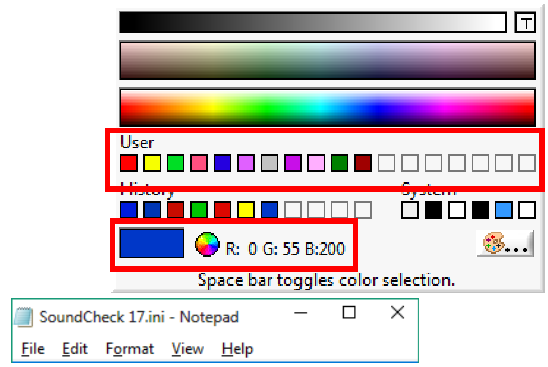
Figure 20-34: Color History

User Colors

18 Colors can be defined in the **SoundCheck 18.ini** file.

Colors in the list follow the standard HEX color code and can be named as shown in ***Figure 20-35***. This also shows the text of the INI file used in the example.

This is a continuous string with no line breaks. All color items must be within quotes and semi-colons must be used as delimiters.



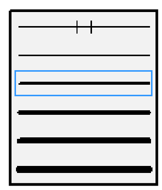
[SoundCheck 18.1]

colorUserItem="Limits_Red=FF0000;Yellow=F7FF00;L_Green=00DE25;Red=FF5080;Blue=2500DF;Violet=E060FF;Lt._Gray=C3C3C3;Dark_Violet=C80FE8;Light Violet=FFAFFF;Dark Green=008000;Dark Red=A00000"

Figure 20-35: User Colors

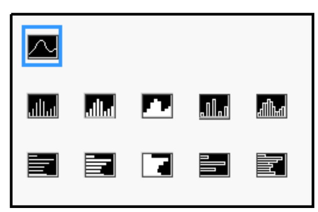
Line Style - Set the selected line to solid or dashed

Line Width - Set the width of the selected item

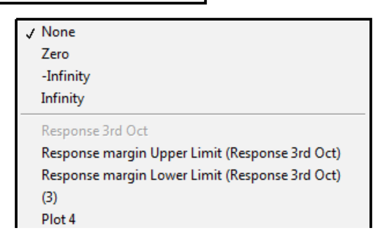


Anti-Aliased - Select this item to make plot lines appear smoother. Note that anti-aliased lines can make sequences run slower if a large number of lines are drawn.

Bar Plots - Select between Line Style and a variety of Bar Plots. Useful for display of RTA Spectrums.



Fill Base Line - This is used to fill the area above or below the selected item in the Plot Legend.



This is useful for creating masks as in ***Figure 20-36***, making it easier to see the response curve inside the limits mask.

- Infinity - Fills the area above the selected item
- -Infinity - Fills the area below the selected item
- Zero - Fills the area from the selected down to zero
- Memory List Items - You can fill the area from the item to another curve

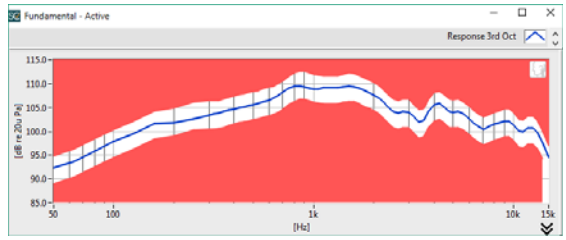
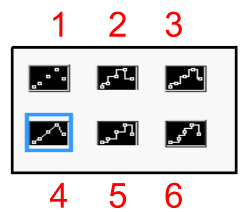


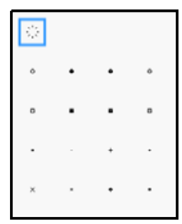
Figure 20-36: Fill Base Line Mask

Interpolation - Determines how the line between known points will be drawn. The most commonly used method in SoundCheck sequences is number 4: Point to Point line.



1. Points only - No interpolation between data points
2. Right edge of step is aligned to data point
3. Center of step is aligned to data point
4. Point to point - (Linear interpolation) is the straight line between each data point
5. Left edge of step is aligned to data point
6. Vertically centered to data point

Point Style - Select None or a variety of point styles.



X Scale - This function is not supported and should not be used.

Y Scale - This function is not supported and should not be used.

Export - This function is not supported and should not be used.

Important: ***As of SoundCheck 8, Display Windows do not allow the user to export data to Excel. This has been replaced with the Report Function. See Report Generator on page 361 for more information.***

Graph Footer

To view the tools, click on the **Expand Footer** icon (down arrow) in the bottom right corner of the graph.

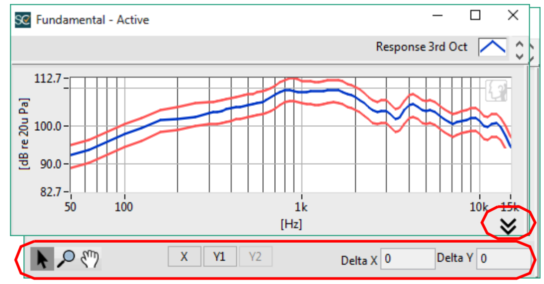


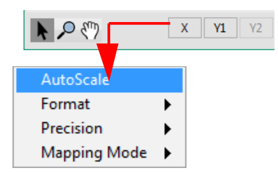
Figure 20-37: Expand Footer

Tools

Select the tool icons in the Graph Footer:

- Arrow - When active you can click on the **XY axis** to change the values. You can also click on the **cursor** to drag it to a location.
- Magnifying Glass - Click to zoom into a specific part of the display.
  - XY allows you to drag a box to select zoom area
  - X allows you to drag between two points on the X Axis
  - Y allows you to drag between two points on the Y Axis
- Hand - Click on the graph and move it within the graph window
- Autoscale - Select to set X or Y axis to the maximum extent of the graph window

Axis Controls



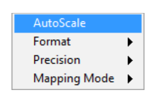
Left-clicking on the X, Y1 or Y2 buttons acts as a “One-time” AutoScale.

You can also set the X and Y axis to automatically AutoScale with the Right-click functions noted below.

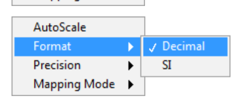
Y2 is only enabled when there is a graph item requiring a Y Axis that is different from Y1, e.g.; Y1 = Frequency Response and Y2 = Impedance

Right-click the X, Y1 or Y2 buttons to open the properties for that axis.

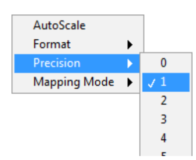
- AutoScale - Set the preference to automatically scale the graph so the selected axis always fits the screen.



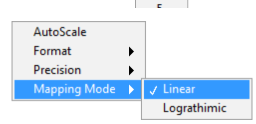
- Format - The appearance of the numbers on the axis can be changed. This is set to Decimal by default.
  - Decimal - Standard decimal point number
  - SI - International system of units



- Precision - Set the number of decimal points that should be represented on the axis.

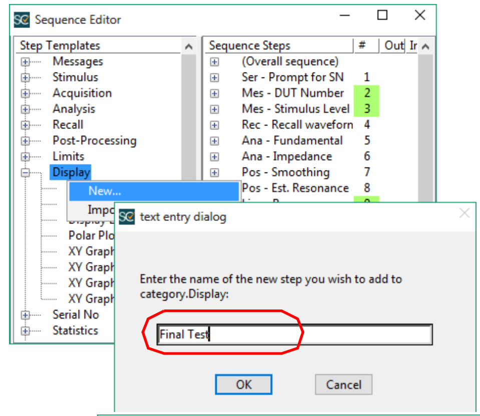


- Mapping Mode - Sets the scale of the axis to linear or logarithmic

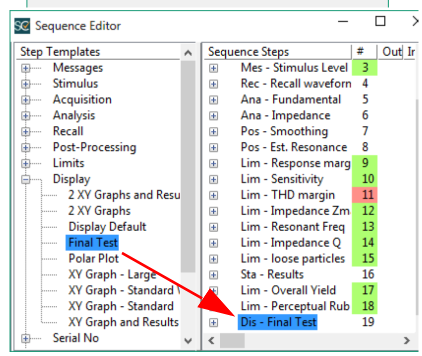


Creating a Display Step

1. Open the sequence editor
2. Open the Display Category folder in the library
3. Right-click the area under Display and select **New**
4. Type in a name for the Display Step. This example is named “Final Test”.
5. Click **OK**.



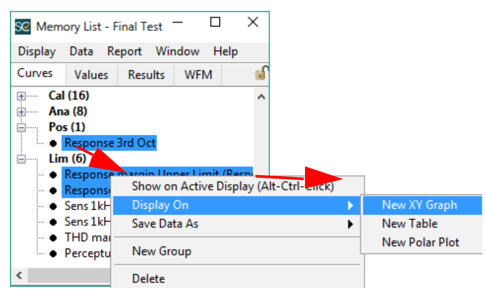
6. Left-click on the new display template and drag it into the Active Sequence. Make sure it is at the end of the sequence. You can Left-click on the Display step to drag it to a new location in the sequence.



Display Step Rules

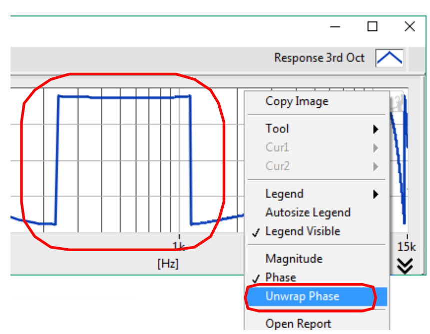
When using multiple Display Steps, each Display Step in the sequence must have a unique name or data will not be displayed correctly. This includes sub-sequences.

7. Open the **Memory List**. In this example we have already run the sequence so that we have data ready to view in display windows.
8. Open the Curves tab. Simply double clicking on a memory list item will open it in its default display window. You can use Drag and Drop to add other items to that window.
9. As an alternative, select multiple data items for the XY Graph. Right-click the selected items. Select **Display On** and select **New XY Graph**.
10. A new graph display opens on the SoundCheck Main Screen in the **Final Test Tab**.



11. You can modify the settings of the graph following the instructions seen in ***Display Editing on page 339***.
12. Right-click the graph of the new display. Select **Preferences** and name the window. In this case it is named **Response**.
13. Next click on just the **Response 3rd Oct** curve.

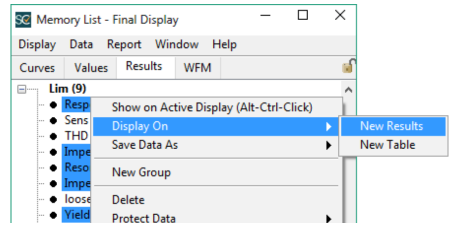
    Right-click and select **Display On** and select **New XY Graph**. This creates another XY Graph in the **Final Test Tab**.
14. Right-click the graph of the new display. Select **Preferences** and name the window. In this case it is named **Phase**.
15. The example phase curve is not displayed as a continuous curve, shown in the example to the right.



16. Right-click the graph again and select **Unwrap Phase**. This makes the phase a continuous curve as shown in ***Figure 20-38***. See ***Unwrap Phase on page 341*** for more information.

17. In the Memory List, click on the **Results** tab.

18. Select all of the result items. Right-click the group. Select **Display On** and select **New Results**.



19. The final display has three windows for Response, Phase and Results.

20. Other windows can be added in the same manner.

21. See ***Display Examples on page 352*** for more information on available displays.

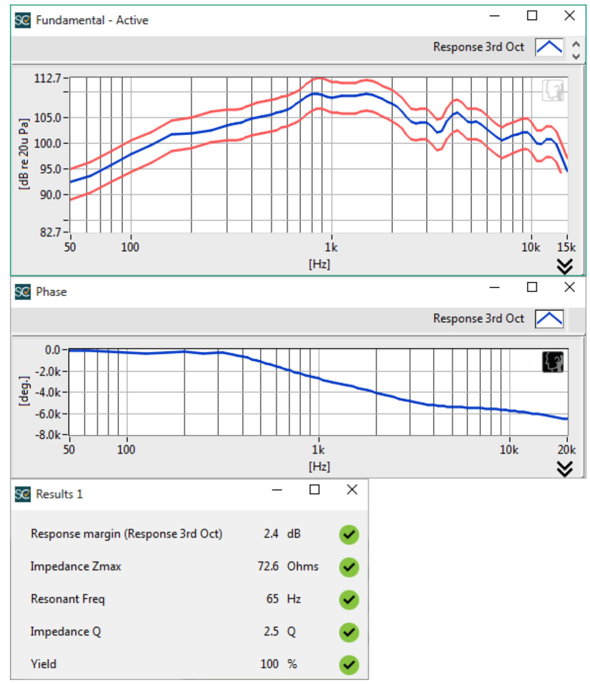


Figure 20-38: New Display

Display Examples

XY Graph

The XY Graph can only be used to display curves.

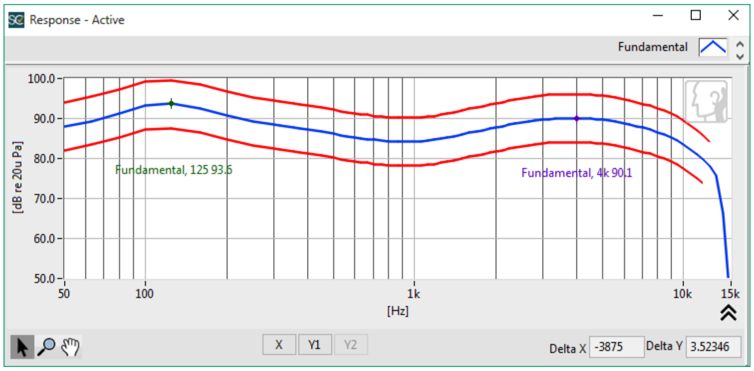


Figure 20-39: XY Graph

- Multiple curves may be shown on one graph, with one or two Y-axes in use.
- You can change the colors of a line on the graph by Right-clicking on the name of the target curve and choosing **Color**.
- Other XY Graph properties such as Line Width, Line Style, and Point Style may be configured through the *Plot Legend* as well. See ***Graph Header on page 346***.

See ***Display Editing on page 339***.

Results

The *Results* window displays the test results, margins, and actual yield percentage if one or more Statistics Steps are in the sequence.

The Display Step must occur after the Statistics Steps.

Only items in the **Results** tab of the *Memory List* can be displayed in this window.

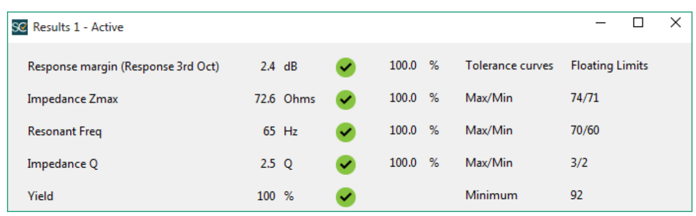


Figure 20-40: Results Window

Export to Excel

Note: To export the information to Excel you must export the entire display from the Report function in the Memory List. See ***Report Menu on page 334***.

Note: **Excel** must be installed on the computer. Cloud based installations are not recognized by SoundCheck.

Results - Display Menu

Set the options for the Results window by Right-clicking on the window and then selecting **Preferences**.

The name of the *Results* display window can be changed in the **Preferences** dialog box. Use **Show Limits** and **Show Yield** to display more information about the limits that produce the pass/fail result. Change the notation and precision (decimal places) to further customize the numerical values.

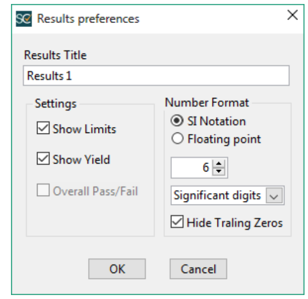


Figure 20-41: Results - Preferences

The **Overall Pass/Fail** selection performs a “Boolean AND” function on the selected results and shows the final Pass or Fail verdict.

When **Overall Pass/Fail** is selected, the verdict is shown as a Green “Passed” or Red “Failed” window as shown in ***Figure 20-42***.

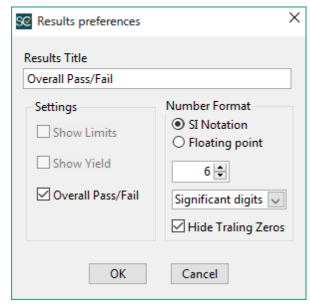


Figure 20-42: Overall Pass/Fail

Polar Plot (optional module required)

The Polar Plot allows you to visualize directionality characteristics of the device under test. The list of Responses in the *Memory List* should be in order (from lowest angle to highest or highest to lowest). SoundCheck will assume that all measurements have been taken consecutively. When all the curves are selected (there must be more than one to show data on the Polar Plot), there will be one color for each.

Requires optional module **2011 - Polar Plot**.

By default:

- The Polar Plot is Autoscaled
- It displays the magnitude of the response at a frequency of 1000 Hz
- The curves will be set at an interval of 10 degrees
- These default settings may be altered by selecting **Preferences** from the **Display Menu** of the *Polar Plot* window

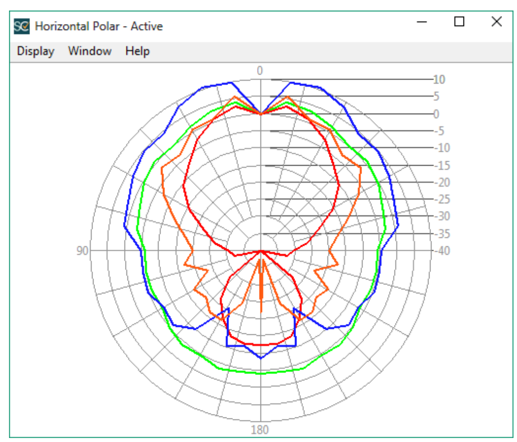
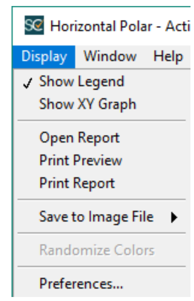


Figure 20-43: Polar Plot Display

Note: The “**Polar Plot (Linear X turntable)**” default sequence has several display examples.

Adding Polar Curves to Display Rules

- Click on the Polar Display to make it the "Active Display"
- If the number of curves for the polar plot changes you must remove the curves from the Polar Display, Re-select the curves in the Memory List and add them back to the Polar Display. This keeps the curves in the proper order so they are displayed correctly.
- Select the group of curves for the Polar Plot in the Memory List. ***They must be selected in order, from 0 degrees to the last degree measured and added as a group.***
- The curves should be from 0 to 180 degrees or 0 to 360 degrees
- Right-click on the curves and select "**Show On Active Display**"



Polar - Display Menu

The **Display Menu** in the *Polar Plot* display window allows you to:

- **Show Legend** - of Frequencies next to the Polar Plot as in ***Figure 20-43***
- **Show XY Graph** - in the *Polar Plot* window as shown in ***Figure 20-50***

  A check mark next to these items means they are visible.
- **Open Report** - Opens a report based on the current display and the settings in Report Setup. See ***Report Generator on page 361***.
- **Print Preview** - Opens the report in Print Preview
- **Print Report** - Sends the report to the default system printer
- **Save to Image File** - Allows you to save the display window as a JPG or BMP file

Preferences

Display Title

Allows you to enter a unique name for the polar window.

Tabs

There are two tabs in the *Polar Plot Preferences* window, Polar Plot and XY Graph. Click on the tab to access the options for the desired graph.

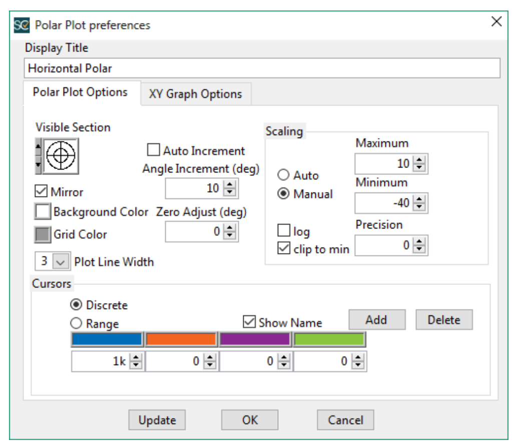


Figure 20-44: Polar Plot Settings

Polar Plot Options

Visible Section

This is used to change the viewable portion of the 360 degrees available. Use the arrow buttons to scroll through the options, or Left-click the current selection to view the possible displays.

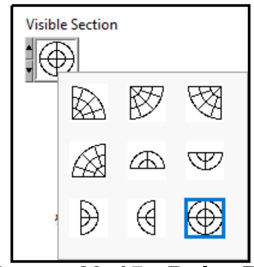


Figure 20-45: Polar Plot Quadrants

Mirror

If the curves represent less than 360 degrees of data, you can choose to Mirror the information around the vertical axis of the polar plot. The data will start at 0° at the North position of the circular grid, and proceed counter-clockwise. By clicking the **Mirror** check box, the right side of the polar display will be a Mirror Image of the left side of the display.

Background and Grid Color

The color of the plot background or plot grid lines can be selected by Left-clicking on the color control icon, and then selecting a new color from the color palette.

See ***Figure 20-46***.

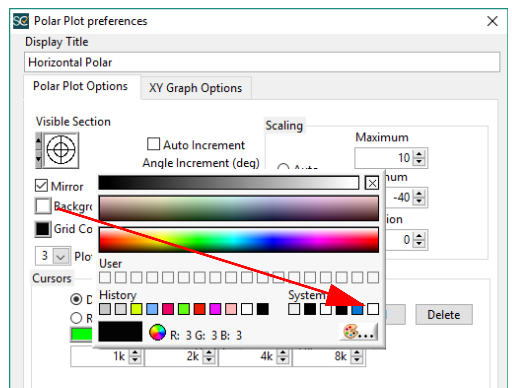


Figure 20-46: Background and Grid Color

Angle of Increment

The *Polar Plot* display assumes that the first curve in the series of curves is at the 0° mark of the rotation.

- **Auto Increment** requires that you enter the Total Rotation, in degrees, of the polar measurements. The number of points is then automatically divided and displayed in even increments. The And Increment field changes so you can enter the **Total Rotation**. Auto Increment is off in the default sequence example.
- **Angle Increment** is used to set the degrees of rotation used when rotating the turntable. This is to Increment the curves so they are equally spaced from zero degrees.

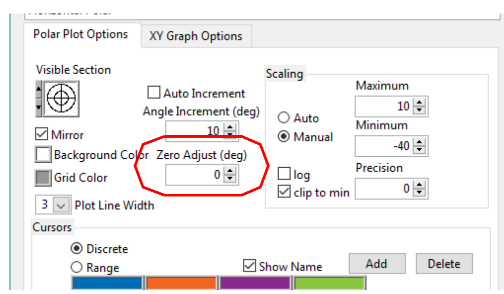


Figure 20-47: Zero Adjust

- **Zero Adjust** is used to reflect the true angle of the first measurement. Every other curve is assumed to be taken in increments as set in the Angel Increment field above.
- **Angle Increment** should match the degrees of rotation used in rotating the turntable. The default sequence step configuration is set to 10 degrees.

Scaling

The Polar Plot, like the XY Graph, is set to Autoscale the curves by default. You can set this to **Manual** and set a minimum and maximum range. As curves are added or removed to the display using the *Memory List*, this value will not change.

The **Precision** field sets the number of decimal places displayed on the Polar Plot.

You can show magnitude on a log scale by checking the **Log** box, or make the minimum value of the Polar Plot the minimum magnitude in the selected range of curves by using the **clip to min** check box.

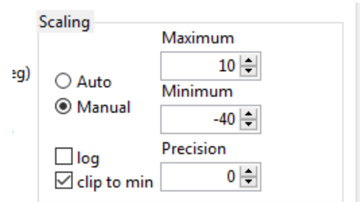


Figure 20-48: Autoscale - Manual

Cursors

Once all the curves to be displayed are selected, one line will be visible. This line represents the response level for each curve at a frequency of 1000 Hz. If the XY Plot is displayed, the selected curves will be visible with one cursor labeled 1000. This is a discrete cursor. Other discrete cursors can be added at other frequencies. An additional line will be added to the Polar Plot with each discrete cursor. The default frequency increments are one-octave widths (e.g., 1000, 2000, 4000, 8000, etc.)

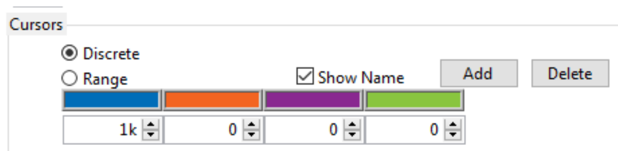


Figure 20-49: Cursors

To see the response over many frequencies, select the **Range** option. You can set the start and end values of the range, and one line will be added to the Polar Plot for each measured frequency. These frequencies are shown in the legend to the right of the Polar Plot.

Polar Plot Example

- A color is assigned to each Cursor Point within the designated range
- Each measured point within the range will create a vertical line in the polar plot

You can add an XY graph to the *Polar Plot* display window as noted in ***Polar - Display Menu on page 354***.

This graph (See ***Figure 20-50***) will show all the curves that have been selected from the Memory List as well as the **Cursor Lines** selected in the Polar Plot Options tab. See ***Polar Plot Options on page 355***.

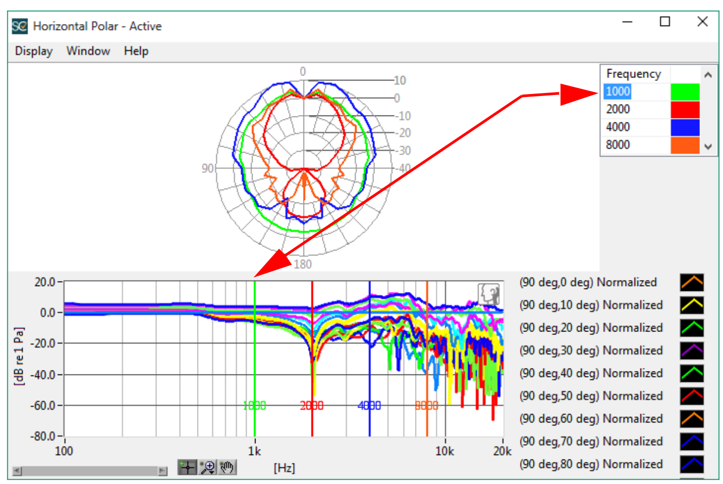


Figure 20-50: Polar Plot with XY Graph

XY Graph Options

This is a subset of the functions available in the *XY Graph Preferences* as described in ***Display Editing on page 339***.

There are no settings for:

- auto-offset
- standard axis ratio

(See ***Randomize Colors on page 342*** for information on how to easily change the plot colors.)

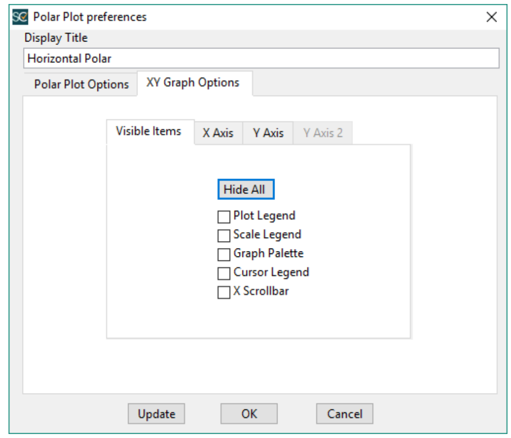
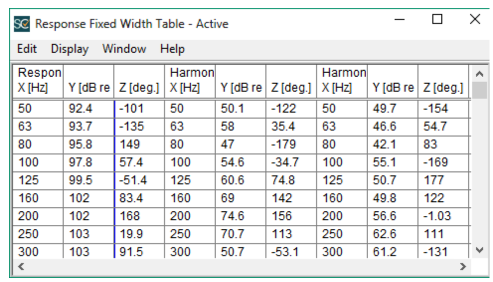


Figure 20-51: XY Graph Options

Table

A Table displays the numerical values, the name, and units of the Curves, Values, Results or WFM selected from the *Memory List*. Any combination of these three types of information can be displayed in the same table.



Respon X [Hz]	Y [dB re	Z [deg.]	Harmon X [Hz]	Y [dB re	Z [deg.]	Harmon X [Hz]	Y [dB re	Z [deg.]
50	92.4	-101	50	50.1	-122	50	49.7	-154
63	93.7	-135	63	58	35.4	63	46.6	54.7
80	95.8	149	80	47	-179	80	42.1	83
100	97.8	57.4	100	54.6	-34.7	100	55.1	-169
125	99.5	-51.4	125	60.6	74.8	125	50.7	177
160	102	83.4	160	69	142	160	49.8	122
200	102	168	200	74.6	156	200	56.6	-1.03
250	103	19.9	250	70.7	113	250	62.6	111
300	103	91.5	300	50.7	-53.1	300	61.2	-131

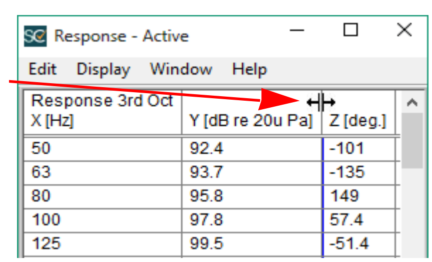
Figure 20-52: Table of Curves, in Column Format

Table - Display Menu

Table properties can be set by choosing **Preferences** from the **Display Menu**.

You can change the title of the *Table* display window, and set the notation and precision for the numerical values in the table. By default, the names selected from the *Memory List* for display on the Table are displayed in columns. This can be changed to Rows by selecting **Transpose** in the **Table Format** section.

When the Memory List Items are displayed in column format, the width of all the columns can be manually adjusted. Put the mouse cursor over the grid lines in the table until it becomes a **double-sided arrow**, then drag it left or right.



Response 3rd Oct X [Hz]	Y [dB re 20u Pa]	Z [deg.]
50	92.4	-101
63	93.7	-135
80	95.8	149
100	97.8	57.4
125	99.5	-51.4

Figure 20-53: Manual Resize

Table - Preferences

This allows for a great deal of flexibility in how table data is presented. From the Active Table window select **Display** and then **Preferences**.

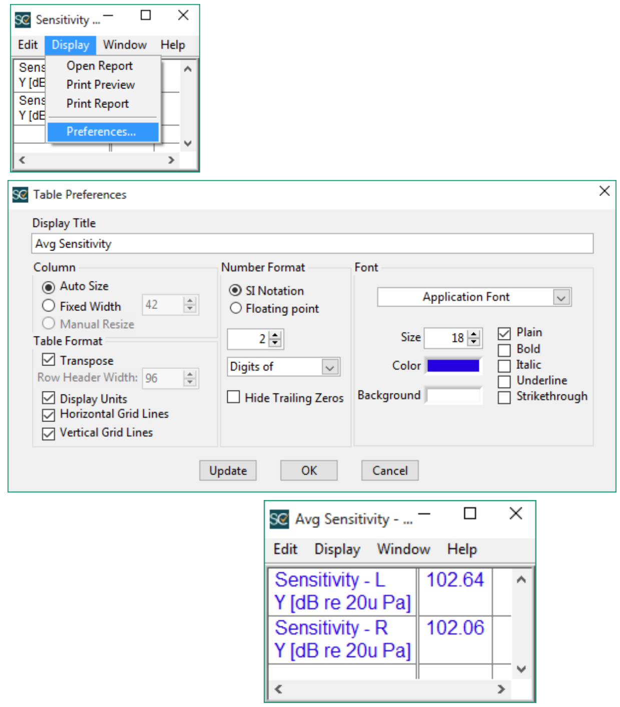


Figure 20-54: Table Preferences

Column

- Auto Size sets the column width to widest table cell (default setting)
- Fixed Width allows you to enter a value for the cell width
- Manual Resize allows you to click on a column boundary and move it to the desired width

Table Format

- Transpose changes the x and y orientation of the data in the table

  When **Transpose** is selected, you can set the width of the row header (which then contains the name and unit information of the curves, values and results). *The rest of the table is no longer manually adjustable.*
- Row Header Width can be set to a specific value
- Display Units show the units along with the column header (default setting)

Number Format

- Choose between SI Notation or Floating Point
- Enter value for Significant Digits
- Hide Trailing Zeros

Font

- Choose from a drop-down list of fonts
- Set Size, Color and Attributes of selected font (See ***Figure 20-54***)
- Set color of the Background of the table

Text

The **Text** box can be used to annotate tests. The text box allows you to enter free-form notes into the display layout, although it is not a full-fledged text editor.

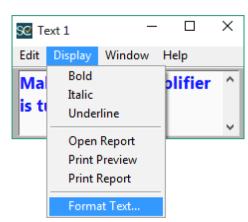


Figure 20-55: Display Menu Options

Local Language Characters

Text can also be entered in Local Language Characters. See ***Display Local Language Characters on page 288***.

Text - Display Menu

The **Display Menu** in the *Text* display window allows you to change the formatting of the words and numbers of the display. The size, font, style, and color of the displayed text can be changed as well as the title of the *Text* display window.

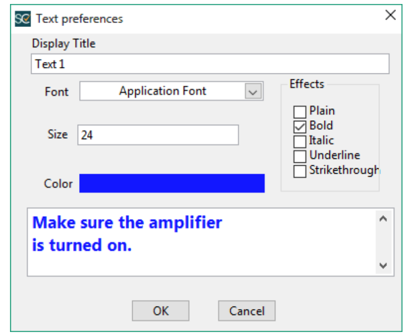


Figure 20-56: Text Preferences

Picture

The *Picture* display window allows jpeg (*.JPG) or bitmap (*.BMP) images to be opened in the display.

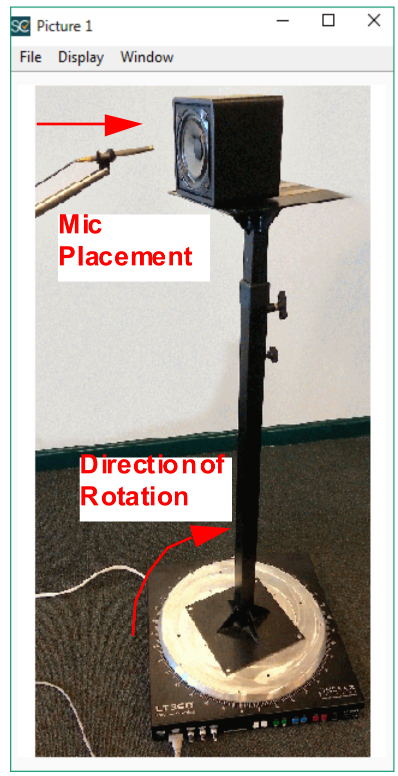


Figure 20-57: Picture Display

Picture - File Menu

The **File** menu of the Picture Display window allows you to select an image to display.

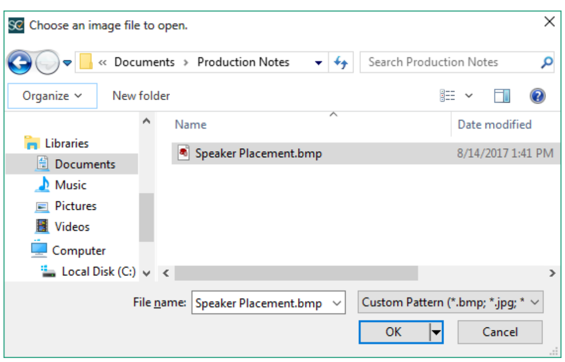


Figure 20-58: Select an Image

Picture - Display Menu - Preferences

- **Picture Title** - Set the name of the window
- **Image Path** - Select the image file to be displayed
- **Resize Window to Picture** - The display window automatically adjusts according to the size of the image file
- **Resize Picture to Window** - The image file automatically adjusts to the size of the display window. Left-click on the border of the window and drag it to adjust the window size.

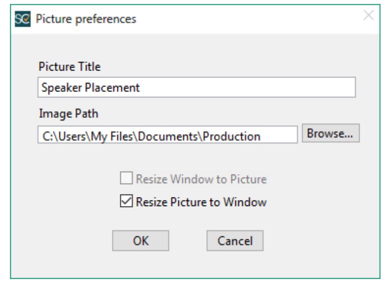


Figure 20-59: Preferences

Waveform

Waveforms can be displayed on a graph. They cannot be displayed on the same graph as a curve.

The Preferences for Waveform graphs are the same as XY Graphs. See ***XY Graph on page 352***.

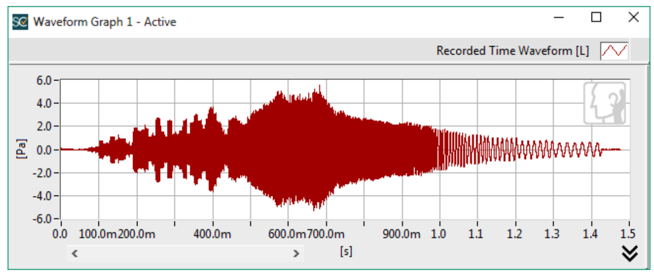


Figure 20-60: Waveform Graph

Report Generator

Create and print company reports in Word, Excel, HTML and image files. The **Report Menu** is found in the Memory List and **Open Report** is in each of the Display Windows.

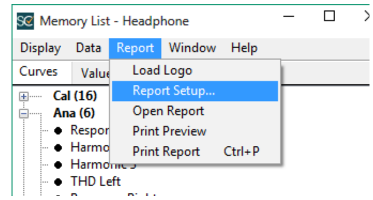


Figure 20-61: Report Setup

- **Load Logo** - Company logos can be added to the heading of a report. Click here to select .BMP or .JPG file.
- **Report Setup** - Select the Export Settings for the report:
  - Report Selection - Word, Excel or HTML: Export settings change depending on the type of report selected
  - Report title - None, Standard or Custom
  - Template - Must follow rules as noted in ***Report Template Rules on page 361***
  - Test Information - Operator, Time, Lot Number and Serial Number
  - Option to Create New File or Append to Existing File

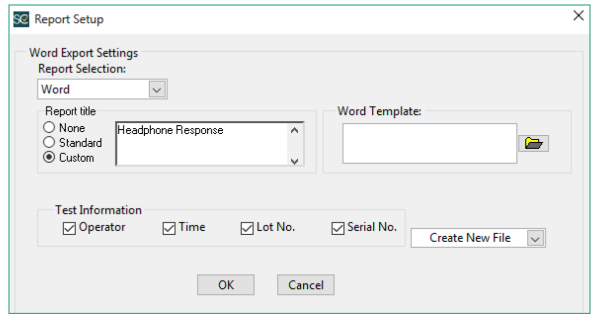


Figure 20-62: Report Setup

- **Open Report** - Opens a report based on the current display and the settings in Report Setup
- **Print Preview** - Opens the report in Print Preview
- **Print Report** - Sends the report to the default system printer

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Note: When printing to PDF Writer or PDF Distiller, the file is not automatically named.

Note: **Excel** must be installed on the computer. Cloud based installations are not recognized by SoundCheck.

Report Template Rules

When creating a template, the following fields must be present or an error message will display indicating that the template is corrupt or invalid. These fields are case sensitive and must be included in every report even if they are not used.

- 'seq_name' - Sequence Name
- 'operator' - Operator's Name
- 'time' - Time the sequence was run
- 'lot' - Lot number
- 'serial' - Serial number
- Graph name (case-sensitive, no spaces)

Regarding graph names: Bookmarks in Word and Excel use the name of the Display Window: Response, Distortion, Harmonics, Results, etc. The name in the title bar of the display will be used as a bookmark. Excel and Word do not allow spaces in bookmark names, e.g., "Waveform Graph 1" should be "WaveformGraph1" (case-sensitive).

Creating Templates

Word and Excel Bookmark instructions can be found at: **http://office.microsoft.com/** by searching for Work Bookmark or Excel Bookmark.

REPORTS USING EXCEL

An Excel worksheet will appear on the desktop with the data filled in. This file is named and saved to retain the data. The other option, **Excel format…** is used to set up the way the data is exported to Excel. These preferences are saved when the Display is saved. If the preferences for exporting to Excel are not set, default settings will be used.

You can choose which axes to include. This may become useful, for example, if all the data has the same X-axis. In that case, it is recommended to avoid taking up disk space by omitting the X-axis from all Exports after the first one. Data exported to Excel will be displayed in columns by default, but export to rows can also be used. If the data requires a Header, the Standard Header can be used, which includes unit information. A custom header can also be created.

You can choose to export the data in scientific or floating point notation. Test Information such as Operator name, Time of test, Lot number, and Serial number can be exported with the numerical data. All data will be exported to a new Excel spreadsheet, each time, by default. Data can also be appended to an existing Excel file that was created by SoundCheck. Each different name of a curve, value, or result will be saved to an individual worksheet in the Excel workbook. The worksheet will have the same name as the curve, value, or result saved there.

Note: If selected, time information is exported with one second resolution, but the default Time/Date format in Excel is one minute resolution. To display the complete time information, format the cell, row, or column in Excel that contains the data. For example to show hour, minute, and second information format the row or column to HH:MM:SS.

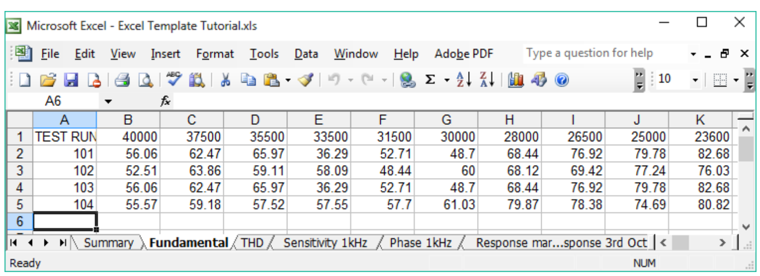


Figure 20-63: XY Data Curves In Excel

Print Step

A Print Step in a sequence allows you to produce a printed report each time the sequence runs. Printing can be done in the background with no operator action required. A standard SoundCheck® report is shown in ***Figure 21-1: Printing Editor Standard Report***. This can be sent to your printer, or reports can be output to Word, Excel, HTML or Images.

A print step must be inserted after a Display Step in the sequence and the Display Step must be configured to "Display when run".

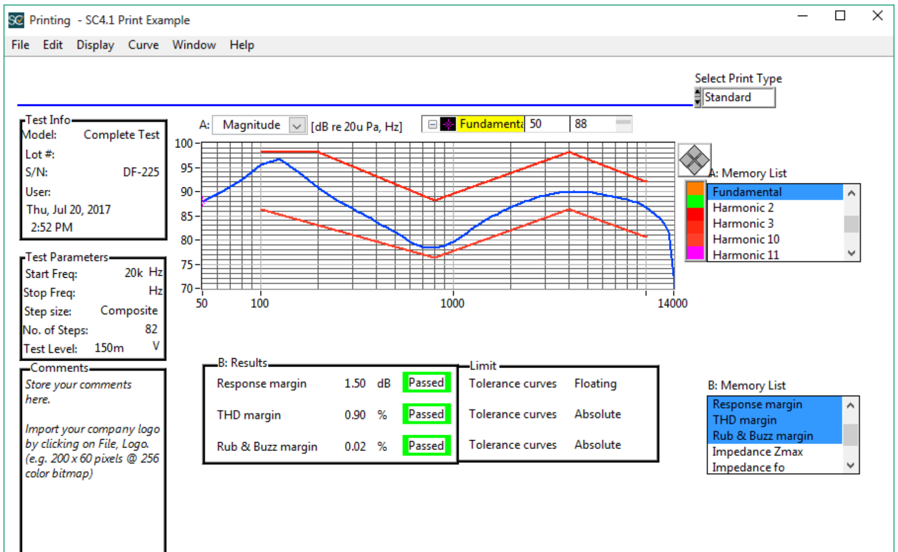
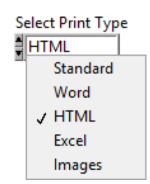


Figure 21-1: Printing Editor Standard Report

- The Test Info and Test Parameters fields on the left side of the printout are linked to the test information of the sequence. All of the fields can be modified except for the date, which is fixed.

Printing is also available in the Offline menu on the SoundCheck Main Screen. Reports can be created in the Memory List. See ***Report Generator on page 361.***

Print Type Modes



There are five modes to choose from in the **Select Print Type** field; Standard, Word, HTML, Excel and Images.

Important! A Display Step MUST come before the Printing Step in the sequence and be open when the Printing Step runs.

Standard Mode

This mode allows you to setup a report in the Print Editor. You can choose any combination of two graphs, tables, and/or results displays for your printout.

sends a report to your printer formatted as *Printing Editor*.

This mode is compatible with the *Printing Editor* available in previous versions of SoundCheck.

Report Generator Modes

The Report Modes for Word, Excel, HTML and Images each create a report based on the layout of the currently open Display Step that occurs prior to the Printing Step in the sequence. The Filename options are the same as those used for Autosave and Recall.

Rules for the generation of reports are the same as those for the Display Step. See ***Report Generator on page 361*** for more information.

Note: **Word and Excel** must be installed on the computer. Cloud based installations are not recognized by SoundCheck.

Excel Mode

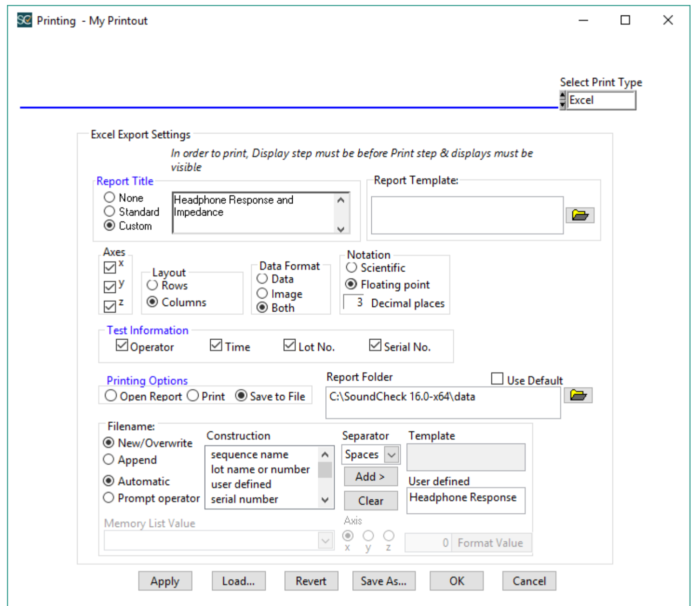


Figure 21-2: Printing Editor - Excel Mode

Some options are common to all print modes. The rules for setting the common options are the same as those for Word. See ***Word Mode on page 365*** and ***Report Template Rules on page 371*** for more information.

The options for setting Axes, Layout, Data Format and Notation are the same as Autosave and Recall Steps.

Axes - Choose axes to include in export. The X axis is always included on the first save to Excel.

Layout - Set in Columns or Rows.

Data Format - Save Data, Images or Both.

Notation - Scientific or Floating Point.

Note: As of SoundCheck 14.01, Excel Macro-enabled files with the XLSM file extension are allowed. The XLSM file extension is used in the generated file.

Note: Excel .XLS files are limited to 256 Columns. Rows are unlimited. In Excel 2007 and later the .XLSX file maximum worksheet size is 1048576 rows by 16384 columns.

IP	IQ	IR	IS	IT	IU	IV
250	251	252	253	254	255	256

Important! When using an Excel file (XLS) as a template in the Printing Step Editor, the SoundCheck sequence name must not be longer than 31 characters. This is because Excel has a limitation of 31 characters maximum for worksheets. The sequence name is used as the name for the first worksheet in the Excel workbook created by the Printing Step. In the end, short sequence names are recommended.

Word Mode

Print to Word options:

- Report Title - None, Standard or Custom
- Report Template and Path (See ***Report Template Rules on page 371***)
- Test Information to include:
  - Operator
  - Time
  - Lot Number
  - Serial Number
- Printing Options
  - Open Report - Opens the report in Word (Word must be installed)
  - Print - Prints the file to the default printer (opens Word in the background)
  - Save to File - Saves Word file to the location specified in “Report Folder“. Check “Use Default“ to use the “Default Data Path“. See ***Folder Paths on page 49***.

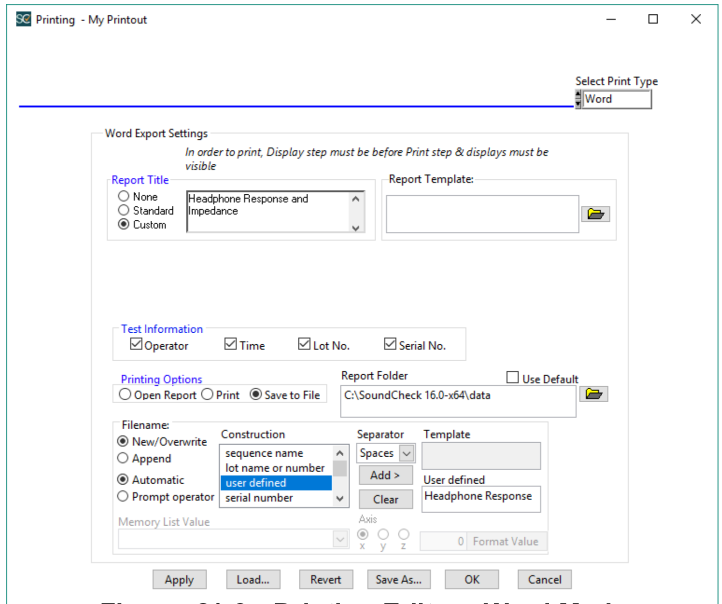


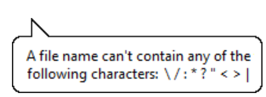
Figure 21-3: Printing Editor - Word Mode

Save to File Options

- Filename for the report
  - New/Overwrite - Creates a report using the Template name
  - Append - Add to an existing report
  - Automatic - Creates the file when the step is run
  - Prompt Operator - Requires operator to enter name for report

Automatic Filename uses the same options as found in the Autosave and Recall Steps:

- Construction - Select an item and click Add> to put the item in the Template
- Separator - Specify what is to be added between items: Spaces, Underscores or None
- When Memory List is added to the Template, the Memory List Value field is enabled. This allows you to select a value from the Memory List to add to the name template; e.g., Index number of Loop. The Axis and number Format of the value can be set as well.
- User Defined - User entered text
- Clear button empties the Template field
- The file names must not contain the following characters: **\ / : * ? < > |** or the files will not be saved



Images Mode

Some options are common to all print modes. The rules for setting the common options are the same as those for Word. See ***Word Mode on page 365*** for more information.

Image Format - Select JPG or BMP.

BMP files have a higher resolution but a larger file size.

When printing JPG or BMP files of display windows, the window names must not contain the following characters: **\ / : * ? < > |** or the files will not be saved.

A file name can't contain any of the following characters: \ / : * ? " < > |

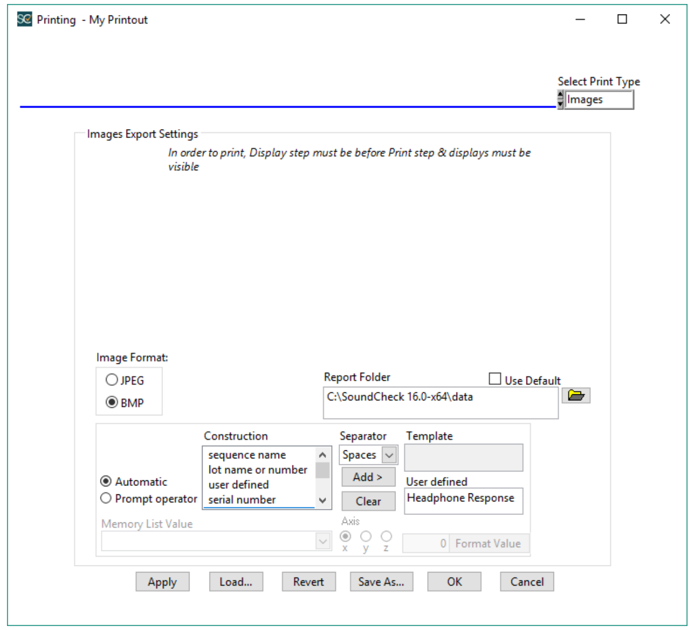


Figure 21-4: Printing Editor - Image Mode

HTML Mode

HTML mode requires that you designate a path and filename for your HTML file, and decide whether you would like to simply create the HTML file and save it to disk, or send it directly to a printer.

HTML files can be edited with any type of HTML editor.

The individual graphic files associated with the HTML file, are uniquely named but not according to the output file name.

Some options are common to all print modes. The rules for setting the common options are the same as those for Word. See ***Word Mode on page 365*** for more information.

The file names must contain the following characters: **\ / : * ? < > |** or the files will not be saved.

A file name can't contain any of the following characters: \ / : * ? " < > |

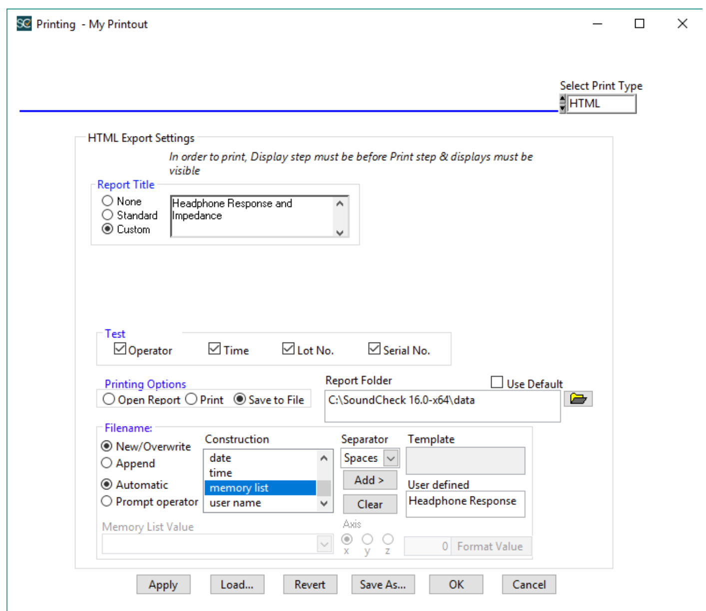


Figure 21-5: Printing Editor - HTML Mode

Note: As of SoundCheck 8, it is no longer necessary to set the **Page Setup** options in Internet Explorer.

The default graphic format for HTML mode is BMP. The format can be set to JPG or BMP by editing the "***SoundCheck 18.ini***" file. This file is located in the root of the SoundCheck folder.

Standard Mode

Many functions of the *Printing Editor* are contained in the menu bar. The following sections describe what options are available.

File

In the **File** menu, you can open saved data or results files to be displayed for printing. Additionally, Data or Results in the current sequence can be saved to a *.DAT, .RES, *.TXT, or *.WAV file.

Click **Logo** to load your company logo to customize your reports.

Print Preview shows a preview of your report only if you are in Standard mode.

Print sends the report in the Print Editor to the selected printer without having to run the sequence. This allows you to make adjustments to the Print Step and then test the results.

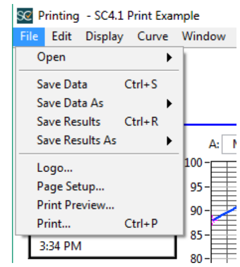


Figure 21-6: File Menu

Page Setup allows you to choose page formatting and printer options.

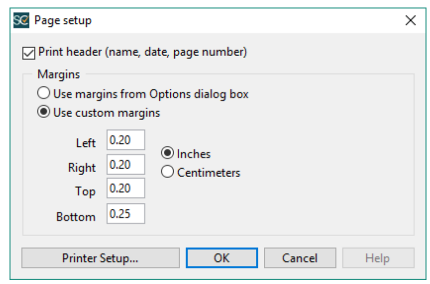


Figure 21-7: File - Page Setup

Display

The Display menu enables you to configure your printout while in Standard mode. (The **Display** drop-down list is disabled for all other modes.)

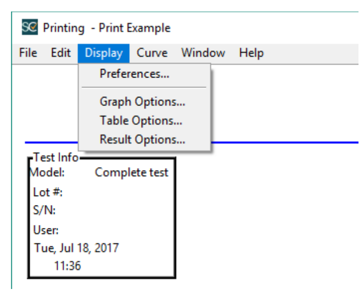


Figure 21-8: Display Menu Options

The **Preferences** command opens the dialog box shown in ***Figure 21-9***. Here, you have the option of enabling one or two displays (marked Display A and Display B). If you choose to show one display, options for Display B will become disabled, and Display A will be the only display. For each display you can choose to show a Graph, a Table, or Results.

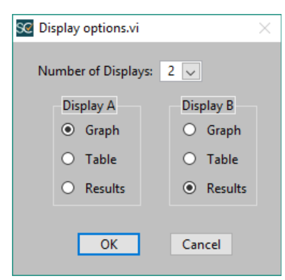


Figure 21-9: Preferences

Figure 21-10 shows a graph and a table on the *Standard Print* report. Notice that unlike the Display Step, the Printing Step has two *Memory Lists*, one for each type of display. XY Graphs and Tables can display curves and single values, and a Results display can display the result margins and verdicts.

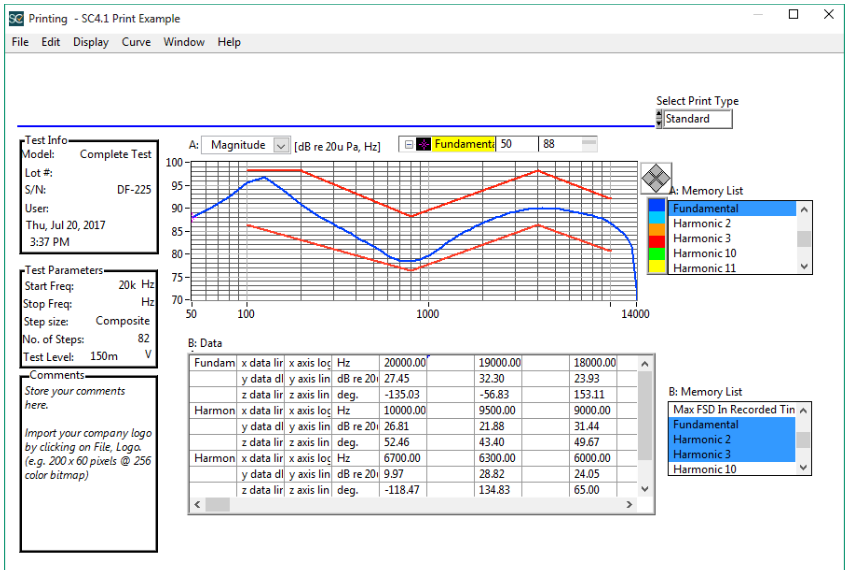


Figure 21-10: Show Graph and Table

Use **Graph Options**, *Figure 21-11*, to adjust scale options on the x or y-axis. You can remove grid lines or change from a linear to a logarithmic scale. You can also cancel the default Autoscaling of the graphs, allowing you to choose your own scale.

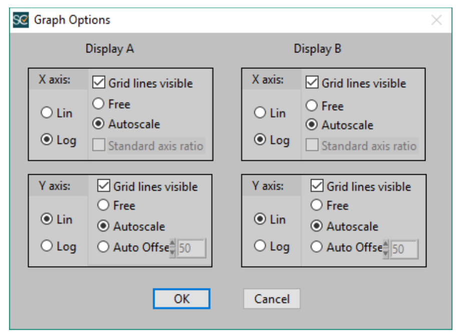


Figure 21-11: Configure Graphs

In the Y Axis options, **Auto Offset** is available. Here you can set the visible range on the Y-axis. This enables the XY Graph to display a user-defined dB range per decade of frequencies. When this is selected on the Y-Axis box, the **Standard axis ratio** becomes available on the X-Axis box. Selecting this option will ensure that the aspect ratio set by the **Auto Offset** will remain constant. Since you cannot alter the graph size in the *Printing Editor*, the Standard axis ratio is set by default.

The table data can be transposed from rows to columns, and the width of the columns can be controlled for each of the two possible tables.

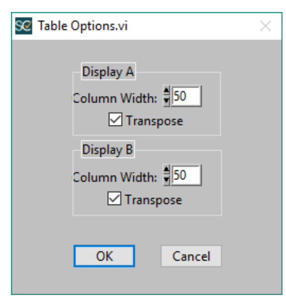


Figure 21-12: Table Options

Show Limits displays the upper and/or lower bound of your Limits Step in the results display. The results can be displayed individually, or by choosing **Overall Pass/Fail**, you can perform a Boolean “AND” on the results selected for display, and receive one Pass or Fail notice. To choose **Overall Pass/Fail** in this example, you would first need to uncheck the **Show Limits** box, and then the **Overall Pass/Fail** box will become available. The choices in ***Figure 21-13: Results Options*** are mutually exclusive.

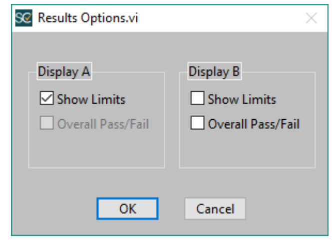


Figure 21-13: Results Options

Curve

In the **Curve** menu, you can alter the properties of the curves and single values in the *Memory List* box of the *Printing Editor*. **Delete** enables you to delete curves, single values or results selected from the *Memory List* box on either Display A or Display B (called “Graph A” and “Graph B” in ***Figure 21-14: Curve menu options***. You can rename one selected curve, single value or result at a time in either Display A or Display B. Similarly, you can change the units of the Y axis of one curve or single value (not valid for results) at a time from either of the two displays.

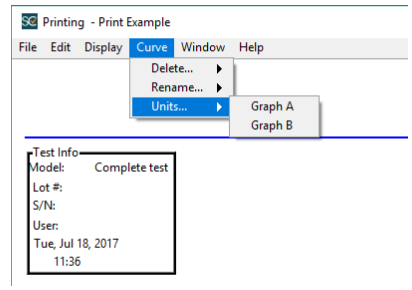


Figure 21-14: Curve menu options

Page intentionally left blank

Report Templates

Report Template Rules

- Office 2003, 2007, 2010 and 2016 are supported for generating reports. Office 97, 2000, XP and 2002 are not compatible.
- Word and Excel must be installed on the computer. Cloud based installations are not recognized by SoundCheck.

When creating a template, Bookmarks must be added to the template so that these items are automatically placed in the correct location in the report. These fields are **case sensitive**.

- **logo** - Uses the file specified under **Edit > Preferences > Folder Paths > Logo Path**. Maximum size of 260 x 60 pixels
- **seq_name** - Sequence Name, Custom Name or None, depending on what is selected in the Report Title field
- **operator** - The User Name used in the SoundCheck Login window
- **time** - Date and Time the sequence was run
- **lot** - Lot number from the SoundCheck Main Screen
- **serial** - Serial number entered on the SoundCheck Main Screen or from a Serial Number Step that occurs prior to the Print Step/Report generation
- **Display Window Name** - Each Display window must have an individual bookmark or it will be ignored when the report is generated. (case-sensitive, no spaces)
- Bookmarks in Word and Excel use the Graph Title set in Graph Preferences of the Display Window: Response, Distortion, Harmonics, Results, etc. HOWEVER - Spaces are not allowed in Excel or Word bookmarks.
  - Example: SoundCheck graph title "**Waveform Graph 1**" should be bookmarked with the spaces removed; "**WaveformGraph1**" (case-sensitive)
- Only letters and numbers are allowed. No other characters, punctuation or hyphens can be used.

In addition, the following rules apply:

- Reports require that a Display Step, along with all its windows, be open in SoundCheck prior to generating the report. The properties of each of the windows must be set in advance since the report will use a Bitmap of each display window.
- The graphic of a Display Window cannot be automatically scaled in Word or Excel. The graphic size is set in the SoundCheck display. The Excel template can have its page scaling set to reduce the size of report graphics.
- A Report Template is matched to a sequence since the Bookmark names must match the titles of the Display windows
- Changing the name of a Display in SoundCheck will break the link to the Bookmark in the template
- The Report Setup in a display window or the Memory List cannot be saved with the sequence. The Report Title is not remembered after you switch sequences or close SoundCheck. The information will need to be re-selected the next time SoundCheck is run.
- Reports can be in Landscape or Portrait orientation
- Bookmarks can be added to the Header or Footer of the template
- Tables can be used to group graphics with text

For more examples of report templates, refer to the Report Templates folder in SoundCheck.
The “Self Test“ sequence has template examples in Landscape and Portrait orientation.

Manually Creating Reports

Reports can be created from the Memory List for the whole display, or from individual display windows.

1. In the Memory List click on **File > Report Setup**.
2. Under **Report Selection** select a report type from the drop-down list. Selections are: Word, HTML and Excel. The Export Settings will change depending on the Report Type selected.
3. For this example, Word is selected.
4. Under **Report Title** you can choose the title name that will be used on the report. This is the value that is used for the template field - "seq_name". Selections are:

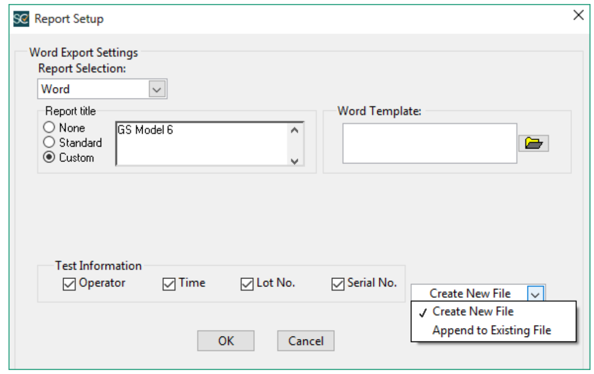


Figure 22-1: Report Setup

- None - report title will be blank.
- Standard - report title will be the name of current sequence.
- Custom - text entered into the field on the right will be used for the report title.

  For this example, **Custom** is selected and the name entered is "GS Model 6".

5. Under Test Information, select which of the four items are to be included on the report. These correspond to the required fields as noted in ***Report Template Rules on page 371***.
6. Click **OK** to close the Report Setup menu.
7. Click **Open Report** and a generic report opens in Word as shown in ***Figure 22-2:***.

The logo is included at the top along with a table that contains the Report Title and the four items selected under **Test Information**. Bitmaps of all of the display windows follow.

> ***Note:*** The manual report setup cannot be saved with the sequence. Changes to the table or graphic size and position are not remembered. The Report Title is not remembered after you switch sequences or close SoundCheck. The information will need to be re-selected the next time SoundCheck is run.

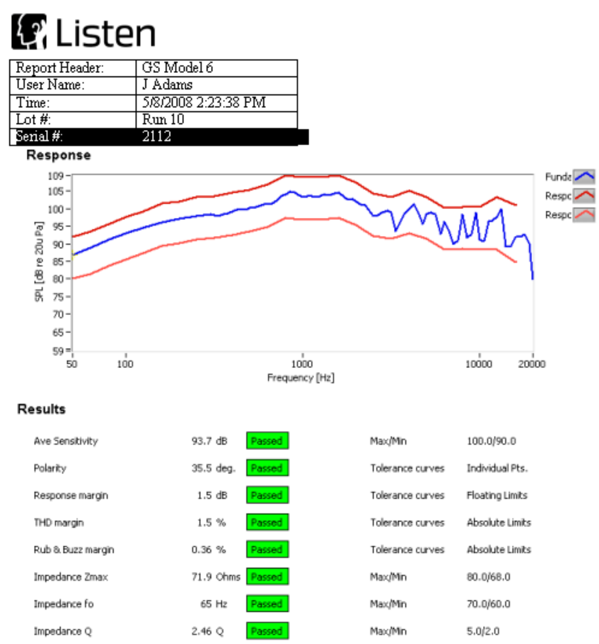


Figure 22-2: Generic Report, Page 1

Creating a Word Template

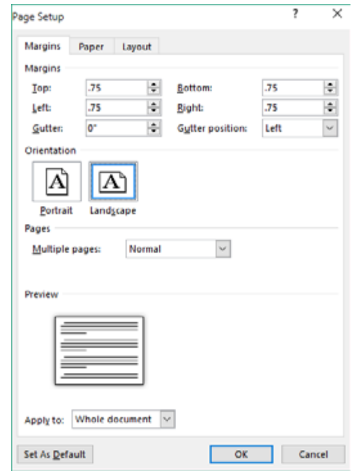


Figure 22-3: Page Setup

For the next example, a Word Template is created so that the layout of the report can be customized.

1. Open Word and create a new document named "Speaker Test Template.doc".
2. In Word, select **Page Setup**.
3. Set the Orientation of the page to "Landscape".
4. Set the Margins as desired.
5. Add a page number field to the Footer of the document. Click on (**View - Header and Footer**)

Create Bookmarks

Bookmarks are used in Word to set the location of text and graphic fields that SoundCheck will use as targets when generating the report.

The window names from the generic report in the previous example are used to create a report template for this sequence.

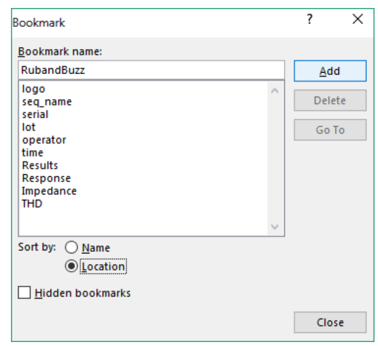


Figure 22-4: Set Bookmarks

1. **Insert** and then **Bookmark**.
2. Under **Bookmark name** type in "logo" and then click **Add**. Since this is the first Bookmark entered, it will appear in the upper left corner of the document. The Bookmark appears as a simple "I" bracket as long as **Show Bookmarks** is checked under: **Tools - Options - View**. (The size of the logo must be set in advance.)
3. Set the location of the next Bookmark in the document. In this case, a Tab is entered so that the next field is moved away from the logo.
4. **Insert** and then **Bookmark**. Enter the Report Title field name, **seq_name** and then click **Add**.
5. Set the location of the next Bookmark in the document. Hit **Enter** to move to a new paragraph.
6. Type "**Serial Number:** ". Click on **Insert** and then **Bookmark**. Enter the field name, **serial** and then click **Add**.
7. Repeat the procedure to enter the other required field names as noted in ***Report Template Rules on page 371***. Enter the bookmarks for the display windows as well. The bookmarks must be in place for the display window graphics to be added to the document.

8. The Word example in ***Figure 22-5:*** shows the final template for this sequence. The document is simply a collection of bookmarks as listed in ***Figure 22-4:***. Text can be added before or after the bookmark so that a text marker for the graphic is included.

9. Save and close the word template, "Speaker Test Template.doc" to the Report Template folder in SoundCheck: **C:\SoundCheck 18.1\Report Templates\Word**.

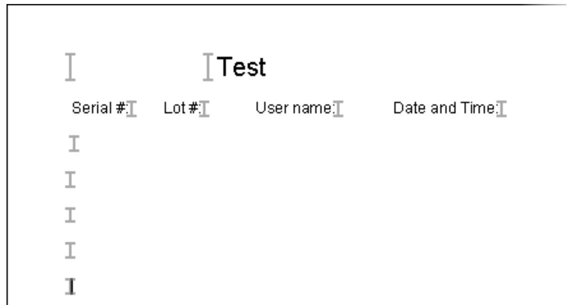


Figure 22-5: Template Bookmark Layout

Manual Report With Template

10. In SoundCheck, the Report Setup will be modified as in ***Figure 22-6:***. Select the template from the previous step. The example title has been changed to "GS 6.5 Woofer".

11. Click **OK** to close the setup window.

12. Click **Open Report** to generate a new report using the template.

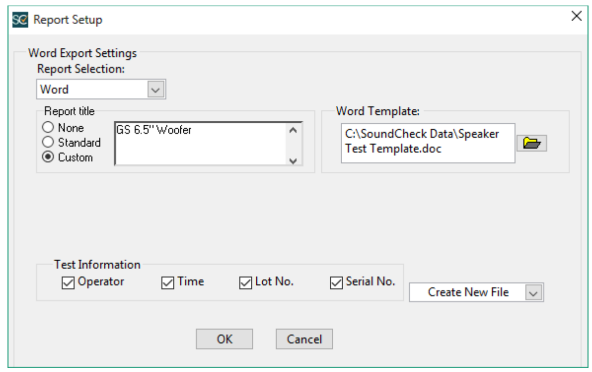


Figure 22-6: Report Setup 2

13. The report example in ***Figure 22-7:*** shows the two pages of the report. The Bookmarks are visible on the screen but are not printed. The size of each display graphic is set in the SoundCheck Display. If the graphics in the report are too large, change the size in SoundCheck and re-open the report.

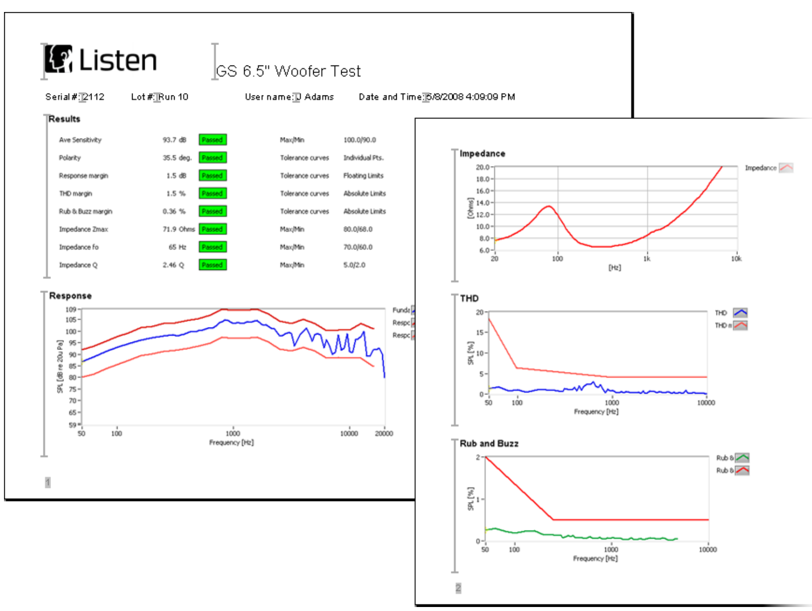


Figure 22-7: Report Example in Word

Print Step Reports

By adding a Print Step to a sequence, generating reports can be an automatic process. Reports can be saved to Word without being visible to the user.

In the following process, the same sequence from previous examples has been used. The same template can then be used in the Print Step.

1. Insert a Print Step to the sequence. It must occur after a Display Step that is configured to “Display Step when run“. If not, an error will occur when the sequence is run. See ***Figure 22-8:***. (See ***Adding Multiple Steps on page 440*** for more information.)

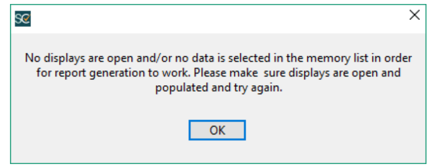


Figure 22-8: Error

The settings in the upper section are the same as the **Report Setup** in the previous example: ***Manual Report With Template on page 374***.

2. Under **Select Print Type**, choose **Word** from the drop-down list. See ***Figure 22-9:***.

3. Under **Report Title** select **Custom** and enter ”GS 65 Woofer”.

4. Under **Report Template** select “Speaker Test Template.doc“.

5. Under **Test Information** select all four items.

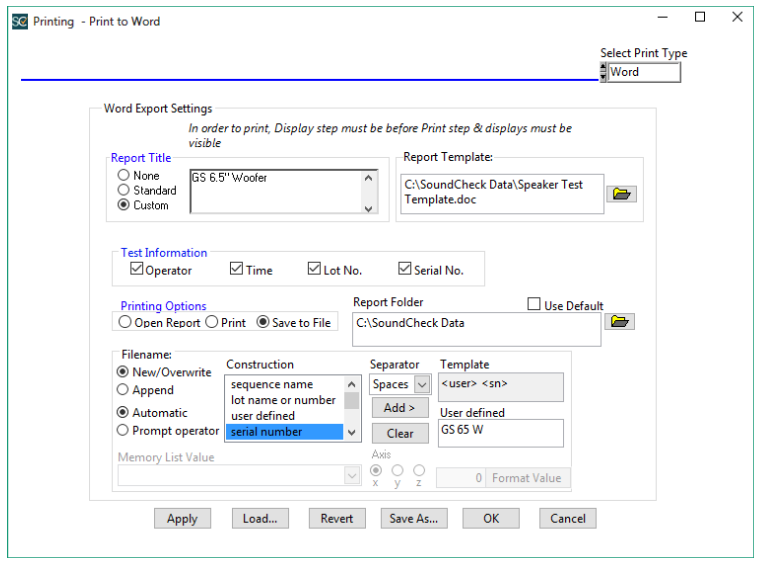


Figure 22-9: Print Step Setup

The settings in the lower section follow the same rules as found in ***Word Mode on page 365***.

6. Under **Printing Options** select **Open Report** then click **Apply** to test the report. A new report will open in word so that you can verify the above settings. The report will be the same as the previous example: ***Manual Report With Template on page 374***.

7. Under **Printing Options** change the selection to **Save File**. The filename options are now enabled.

8. Under **Filename** select **New/Overwrite** to create a new report for each measurement.

9. Select **Automatic** to enable the Construction fields. Select **User Defined** and click **Add**. Select **Serial Number** and click **Add**. The **Separator** should be set to “Spaces”. In the **User Defined** field enter “GS 65 W“. For every sequence run, with a new serial number, a new report will be generated.

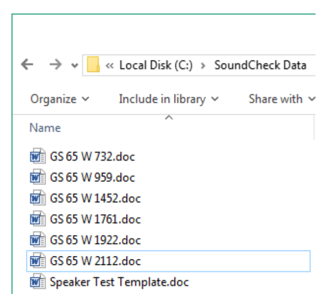


Figure 22-10: Report Folder

10. Click **Apply** and a new report is generated in the selected Report Folder. Remember: The Display Step of the sequence must be open in order to generate a report. As seen in ***Figure 22-10:***, ten reports were generated after testing ten speakers. Each file has the serial number appended to the name.

File Path

The Report Template and Report Folder fields do not follow Relative File Path rules. The path must be re-entered if the sequence is run from a different computer.

Creating an Excel Template

Note: When printing to an Excel template, it is important to keep the File Name short since there is a maximum File Name Length when saving to some versions of Excel.

Defined Names are used in cells as field markers for information and graphics when SoundCheck saves to Excel using a template. In this example, we have created a new Excel file named "Speaker Test Template.xlsx".

1. Right-click a cell where you want to locate a field marker. (A1)
2. Select **Define Name** and enter the name in the editor. See ***Figure 22-11:***.
3. In the Define Name window enter "logo" and click **OK**.
4. The Defined Name can now be seen in the Name Box above column A.
5. Select cells for the other required field markers and markers for the graphics of the report.

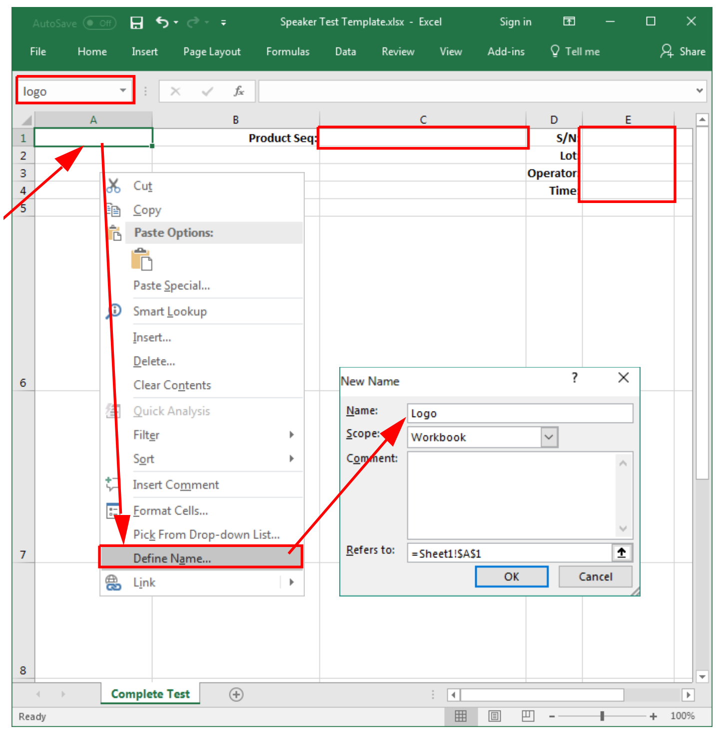


Figure 22-11: Excel Template

6. In this example, the markers are in the following fields:

(Cells A6 through A10 must be sized to allow enough room for the graphics.)

7. Save and close the template.

A1 - logo	C1 - seq_name
A6 - Results	E1 - serial
A7 - Fundamental	E2 - lot
A8 - Impedance	E3 - Operator
A9 - THD	E4 - time
A10- RubandBuzz	

In our example sequence we have changed the Print Step to **Excel** as the **Print Type**. See ***Figure 22-12:***.

The settings for an Excel Print Step are covered in ***Excel Mode on page 364***, ***Word Mode on page 365*** and ***Autosave Editor on page 213***.

8. Under **Report Template** select “Speaker Test Template.xls”.
9. Under **Printing Options** select **Open Report** and click **Apply**.
10. Note that the **Data Format** is set to **Both**. This will save both the graphics and the raw data to the report. The data for each of display windows will appear in a separate worksheet in the Excel file.
11. The Excel Report is opened and the size of the cells can be Fined Tuned to fit the display graphics. See ***Figure 22-13:***.

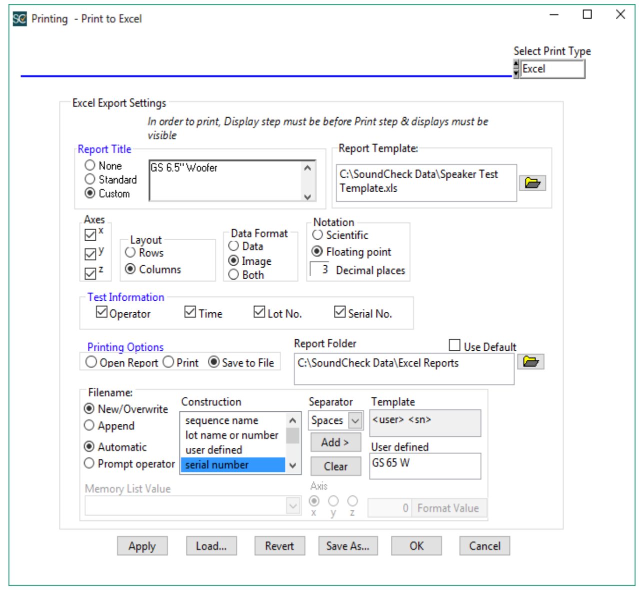


Figure 22-12: Print Step - Excel

12. Once the cells in the Excel report have been adjusted, delete the graphics and text from the variable fields. Then delete the worksheets for the display data. This leaves the Defined Names and Cell Titles for the template.
13. Save and close the Excel template.
14. In the Print Step in SoundCheck, set **Printing Options** to **Save to File**. When the sequence is run, a new Excel spreadsheet is created for each new device tested: “GS 65 W 2001.xls“.

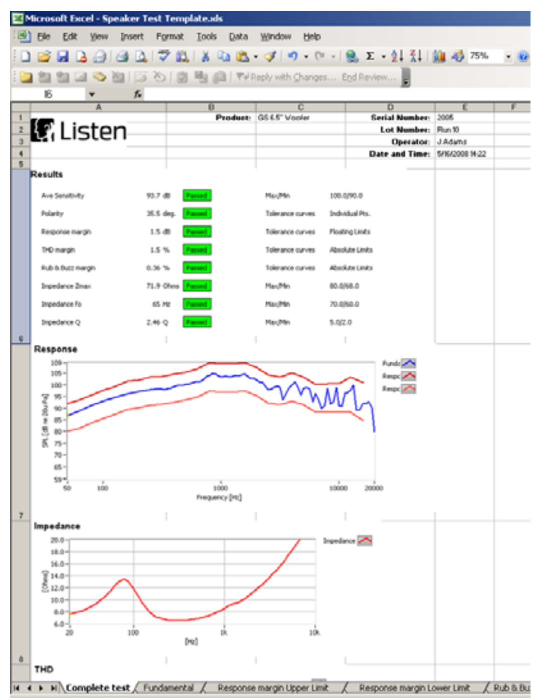


Figure 22-13: Excel Template With Data

Important! The font size of the cells in the Excel template cannot be adjusted. The setting is over written by SoundCheck. The font size is fixed at 10 point.

Important! The size of graphics is not remembered when they are adjusted in the Excel template. Display windows must be scaled in the SoundCheck display.

Note: As of SoundCheck 14.01, Excel Macro-enabled files with the XLSM file extension are allowed. The XLSM file extension is used in the generated file.

Serial Number Editor

You can choose to automatically increment the serial number each time the sequence is run, or to prompt the operator to manually enter the serial number of the DUT. When used in a sequence along with the Autosave Step (See ***Autosave Editor on page 213***) the updated serial number can be saved with measurement data. The serial number also appears at the top of the main SoundCheck® window and in reports generated by the Printing Step. To view and change the system's serial number settings, select **Serial Number** from the **Setup** drop-down list on the SoundCheck Main Screen, or use the shortcut **Ctrl+Shift+E**.

When exiting SoundCheck, the last recorded serial number is saved in the **SoundCheck *18*.ini** file. The next time SoundCheck is opened, the last serial number is automatically recalled to the S/N field of the SoundCheck Main Screen. This enables the operator to continue measuring a lot that was not finished before their shift was over.

Auto Increment

With **Auto Increment** selected, the S/N prefix appears along with the incremented number. Using a step configured as in ***Figure 23-1: Serial Number Setup***, the serial number for the first three items tested would be; ABC1, ABC2 and ABC3.

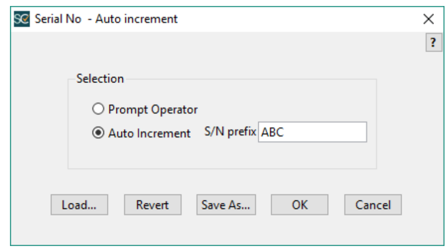


Figure 23-1: Serial Number Setup

Prompt Operator

When **Prompt Operator** is selected, any alpha-numeric combination can be entered by the user. This number is then used to identify the data. Once the number is entered, you can click **OK** or **Enter** to continue running the sequence.

Note: The Serial Number Step must precede the Autosave Step in a sequence for the serial number to be correctly recorded with the measurement data.

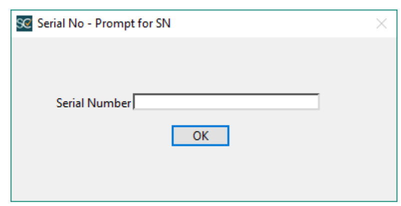


Figure 23-2: Serial Number Prompt

Note: As of SoundCheck 7, the Serial Number category in the Step Library has been changed from ***"Serial #"*** to ***"Serial No"***. To use steps from previous versions of SoundCheck, copy the steps from the *Serial #* folder of SoundCheck x.x to the *Serial No.* folder in the SoundCheck 18.1 directory. These can then be used as step templates.

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Statistics Editor

The Statistics Editor allows you to perform a variety of statistical measurements on the data that is produced. These measurements include: minimum, maximum, mean, standard deviation, Cp, Cpk, and Best/Worst fit to Average for curves.

The editor can operate in either of two modes depending on the application. The Online mode allows for the use of the statistics step in a sequence. You can use this mode to determine the standard deviation after each run and determine whether the spread of the results is outside of acceptable limits. Statistical calculations are made upon the consecutive runs of a sequence when using Online mode.

Online Mode

The values calculated by the Statistics Step are created from curves, values, and results generated during the run of the current sequence.

Important! If the Hardware, Calibration, Acquisition, Analysis or Statistics Steps are changed, all unsaved statistical data is lost.

It is important to understand the circumstances that will enable you to keep building upon the current statistics values. The first time the sequence is run, the statistical values begin to fill. This first run produces curves and values that are based on only one set of data. In other words, the curve Minimum, Maximum, Mean, and Standard Deviation are all identical curves the first time the sequence is run.

The second time the sequence runs, all the statistics calculations are performed using the current and previous data together. The third time, the algorithm integrates the current run values with the running statistics calculated from the last two runs, and so on.

Redo

When Redo is selected from the SoundCheck® Main screen, the last measurement gets overwritten and the statistics recalculated to include the new measurement. This is an unlimited Redo, so it can be selected as many times as is necessary.

A change in the lot number, changing the sequence or changing certain steps of the current sequence (See note above) will reset and empty all your Statistics curves. Any statistics values from this point on will not take old curves/values into account.

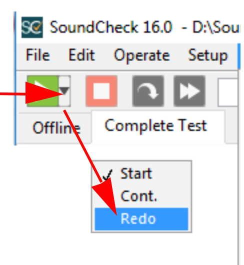


Figure: 24-1 Unlimited Redo

Offline Mode

You can use the Offline Display Tab to calculate statistics on previously measured data. The Offline mode is used outside of a sequence and can be used on protected data or other previously collected data. Offline mode allows you to compare each of the selected curves to each other, rather than to previous runs. The results are output to the Memory List in the form of protected data.

If you want to calculate the average of a group of impedance curves, you can use the Offline mode to compare these curves.

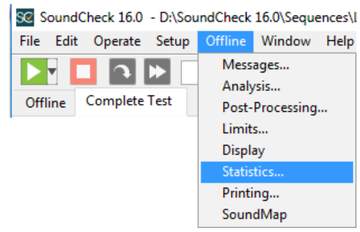


Figure: 24-2 Offline Mode

Note: Histogram, Best and Worst Fit to Average options are only available in Offline mode, and can be used to determine "Golden Units" and "Outliers".

When using stored *.DAT files to create statistics, verify that the *.DAT file contains multiple tests of the same measurement. Please note that any statistical analysis becomes more valid with a large number of samples. Depending on the variability of your measurements, you may need 60 to 100 tests. You can adjust your Sigma value based on the number of samples available.

Note: Mean, Maximum, Minimum, and Standard Deviation calculations are done on the numerical values as they are stored in the file. No conversion is made to use a linear scale.

Note: Protected Data can only be added to Protected Groups. See ***Auto Grouping General Rules - Memory List on page 333***.

Important! Any display windows added to the Offline Tab are temporary. They are not stored with a sequence.

Statistical Process Control

Statistical Process Control measurements are also available in SoundCheck. These options allow users to track the quality of the production by comparing deviations to user-defined limits. For more information See ***SPC - Statistical Process Control on page 387***.

- Select Statistics from Offline Menu
- Select Data, Results or both
- Work In mode allows for calculation in Linear, dB or Power units
- Select items from Memory List
- Select statistical processes to run for Data and Results
- Click Apply and the statistics results are updated in the Memory List

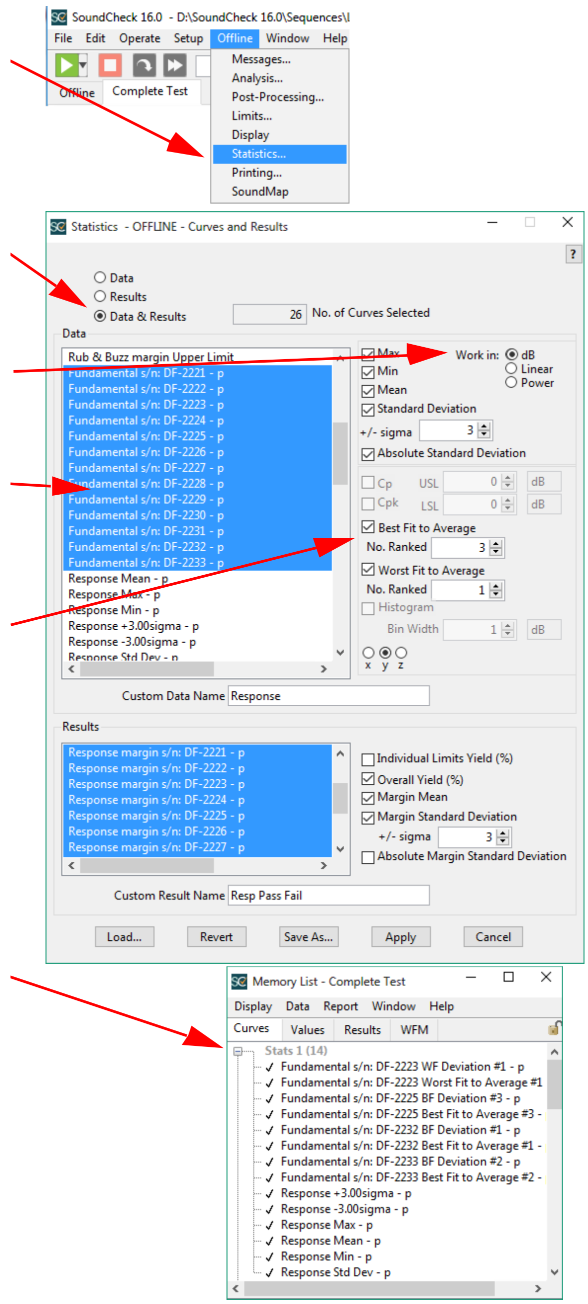


Figure: 24-1 Statistics Offline

Choosing What Statistics to Create

You can decide what curves, values and/or results to analyze statistically. In ***Figure 24-3: Choosing Data and/or Results*** the top area of the editor indicates the number of times this particular step has run in this sequence. All statistical values calculated are based on the "**No. of Curves Selected**".

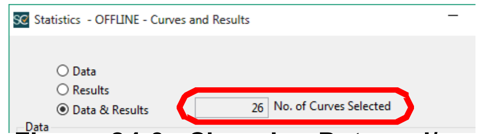


Figure 24-3: Choosing Data and/or Results

You can choose to calculate statistics on Data (curves and values), Results or both. The Data or Results items are selected form the Data and Results lists, respectively. Data items will include the curve and/or value names displayed in the *Memory List*. Each selected name in the list box will have between one and four calculations performed, which depends on how many statistics boxes are checked.

Statistics for Curves and Values

The **Data** section of the *Statistics Editor* (See ***Figure: 24-1 Statistics Offline***) contains all the options for statistical measurements that this step may perform.

- **Work in mode**: Statistics can be calculated using Linear (RMS), dB or Power Units for the Y axis.

  Linear Example: 90 dB + 90 dB = 96 dB (The math is applied on the linear values.)

  dB Example: 90 dB + 90 dB = 180 dB (The math is applied on the dB values.)

  Power Example: 90 dB + 90 dB = 93 dB (The math is applied on the power values.)

- **Max** – Compares the current Y (magnitude) and Z (phase) value at point X (frequency or time) with the existing maximum for point X. If the new value is higher, the maximum value is rewritten. The *Memory List* is updated with a new curve or value whose name ends in Max (e.g., *Fundamental [L] Max*). This enables you to track the upper extreme of the range of the DUT.

  Note: To view the Z values in the Display Step, select **Phase** from the **Display** menu in the *XY Graph*, or use a table to display the numerical values of the curve or single value.

- **Min** – Compares the current Y and Z value at point X with the existing minimum for point X. If the new value is lower, the minimum value is rewritten. The *Memory List* is updated with a new curve or value whose name ends in Min (e.g., *Fundamental [L] Min*). This enables you to track the lower extreme of the range of the DUT.

- **Mean** - Calculates the mean Y and Z value at every X point along the curve. If a single value, this command calculates the average single value. The *Memory List* is updated with a new curve or value whose name ends in Mean (e.g., *Fundamental [L] Mean*). Variation in the mean after many tests have been run indicates a new factor has been introduced that may be causing problems.

  Note: This is not to be confused with *Statistics: Average in the Post-Processing Editor*, a single value which is the average of all the Y values in one curve.

- **Standard Deviation** - The standard deviation is a measure of the dispersion of the Y and Z dimension of the selected curve or value. SoundCheck uses the equation, where x_i is the current Y or Z value at point X, M is the mean of all the past values at point X, and n is the number of values in the set.

  $$\sigma = \sqrt{\frac{\sum(\chi_i - M)^2}{n-1}}$$

  Choosing this option creates two new curves. These may be viewed as bounds around the curve being measured. For example, if **Fundamental [L]** is chosen from the **Data** list box, and the **Standard**

Deviation box is checked, you will see two new curves in the *Memory List*: *Fundamental [L] +1.00sigma* and *Fundamental [L] -1.00sigma* (this is assuming 1 sigma was indicated, as in ***Figure: 24-1 Statistics Offline***). When these two curves are displayed on the XY Graph with the *Fundamental [L] Mean* curve, the Mean curve should fall directly between the two Standard Deviation curves. Future Fundamental curves have a 68% chance of falling within these bounds. If 2 sigma is chosen, you know that there is a 95% chance new curves will fall between these bounds. At 3 sigma, there is a 99% chance that new curves will fall within the bounds. Responses that fall outside the Standard Deviation of the Mean may indicate erratic problems with the DUT or the test environment.

Please note that Standard Deviation values become more accurate when a large number of samples are taken.

- **Absolute Standard Deviation** - The absolute standard deviation is the pure σ value. The curve that is the result of this calculation is available in the Memory List.
- **Histogram**
  - Available only in Off Line Statistics, on single values (cannot be used on curves)
  - Select which axis of value to apply statistics on: X, Y or Z
  - Select Bin Width - Sets the width of the bar plots of the Histogram Curve and sets the resolution of the counting process. Enter any value greater than 0.

  The example in ***Figure 24-5*** shows a Bin Width of 1. The bar at 90-91 Pa shows the number of samples that fall in that bin.

  - Enter Custom Name for the resulting curves

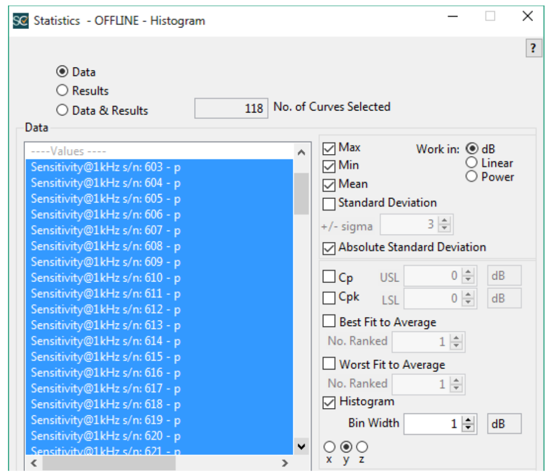


Figure 24-4: Offline Histogram Settings

  - Users will be able to select a series of single values and output a histogram curve ***Figure 24-5: Histogram Display***
  - Histogram and Distribution (Gaussian bell) curves will appear as new Curves in the memory list
  - These curves can be added to a display

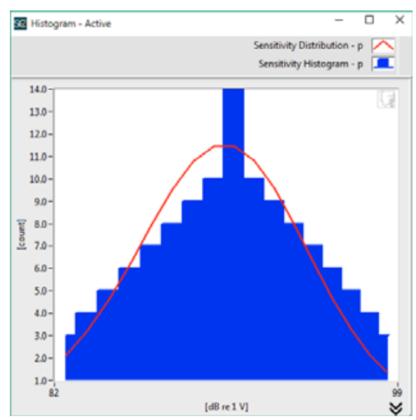


Figure 24-5: Histogram Display

Histogram Display Setup

The setup of a Histogram Display window is different than other display types. Follow the example in ***Figure 24-6*** when creating a Histogram Display.

1. Add Histogram curves from the Memory List to a new X Y Graph.
2. Right-click the Histogram Curve in the Legend.

- Select **Common Plots** and click Bar Plots
- Select Fill Base Line and click "-Infinity"
- Set the curve colors as well

3. Right-click an X axis value to open the X Axis Tools.

- Set **Format** to Decimal
- Set **Precision** as needed
- Set **Mapping Mode** to Linear
- Do the same for the Y axis

4. Right-click the graph of the display and select **Preferences**.

- Enter the **Graph Title**
- Set the desired Plot Area color on the Colors tab
- Set Legend properties as desired
- Turn off Major and Minor Grid Lines for both the X and Y axis
- Set both X and Y axis to **Free.**
- Click **OK** to exit
- You will need to double click on the X and Y axis values on the graph to change the graph window scaling

5. From the Memory List click **Display** and select **Save Display as Template** to save the layout for future use.

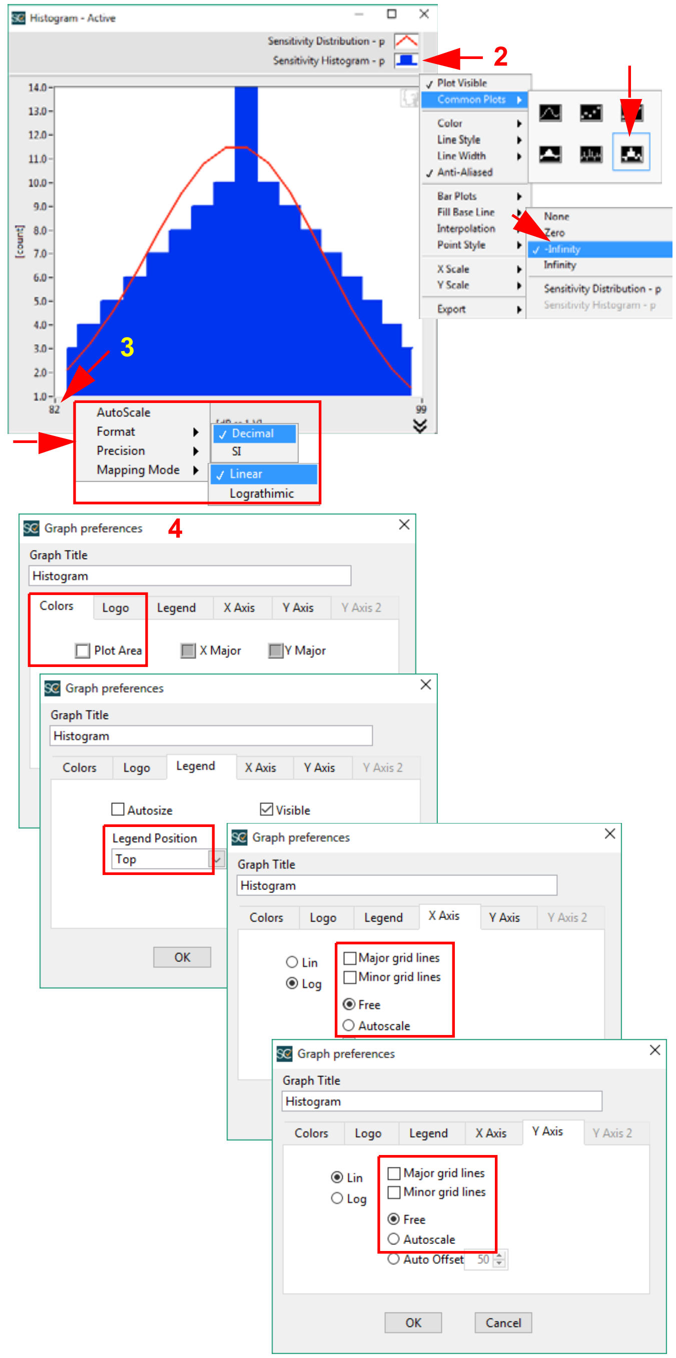


Figure 24-6: Histogram Display Setup

SPC - Statistical Process Control

Statistical Process Control can be used to monitor the quality of the production run.

Statistics can be calculated on a specific scalar parameter (e.g., THD or sensitivity @ 1 kHz) for a group of loudspeakers (e.g., the production of the day). The Statistics module calculates the capability indexes and the results are made available in the Memory List. These can then be saved using an Autosave Step. The history of these indexes can be used to check the trend of production using some external software (e.g., EXCEL).

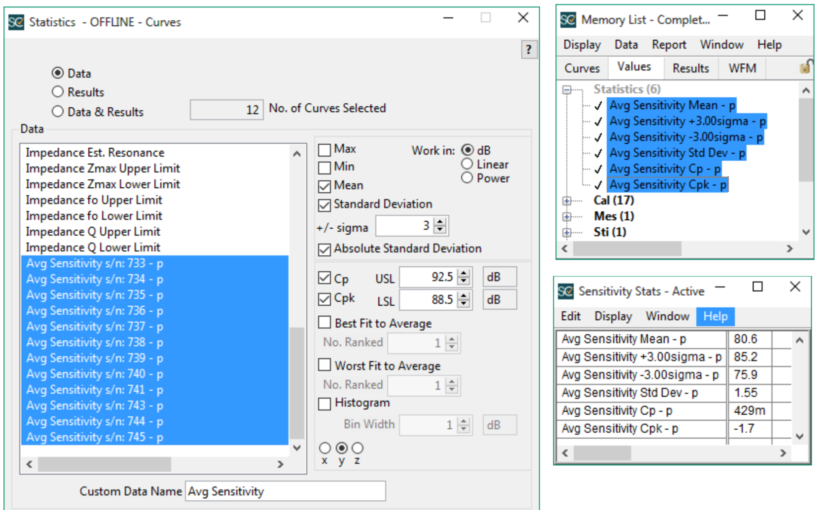


Figure: 24-7 USL/LSL Settings

Menu choices:

- USL: Upper Specification Limit. The maximum tolerance of the production limits. Constants set by the user. e.g., 92.5 dB SPL. (See **Figure: 24-7 USL/LSL Settings**)
- LSL: Lower Specification Limit. The minimum tolerance of the production limits. Constants set by the user. e.g., 88.5 dB SPL.
- Process Capability indexes Cp and Cpk. These indexes are added to the existing list: min, max, mean, sigma. These indexes are calculated in real time and appear in the Memory List.

Cp - A measure of process performance

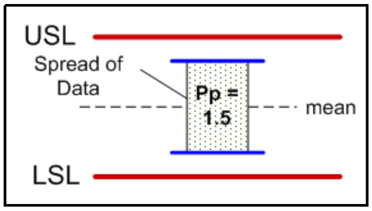
Essentially it is a measure of variance/spread of one's data with respect to specification limits. Values greater than 1 indicate that the 6 sigma range is within the limits. If Cp is equal to 1, the range of 6 sigma exactly equals the range of the limits. If it is less than one, the 6 sigma range exceeds the limits. Ideally or to be safe, the process would yield a result of 1.33 (8 sigma) or higher. This means that your measured parameter will not exceed the limits more than 0.0063% of the time. To the right is a table of Cp values and the related Percentage of Failure. The formula for Cp is:

$$C_p = (USL - LSL)/(6\sigma)$$

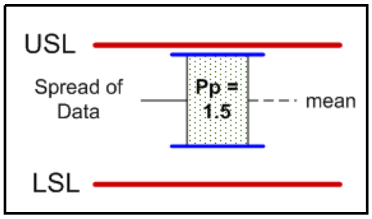
Cpk - Process performance index

This is similar to Cp, except that it takes into consideration any off-centered alignment of the data. The 6 sigma range may be much smaller than the limits range, but the mean could still be close to one of the limits. This would result in a high Cp but a low Cpk. The formula for Cpk is:

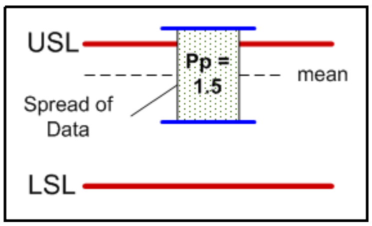
$$C_{pk} = \mathrm{Min}[(USL - \mathrm{mean}), (\mathrm{mean} - LSL)]/(3\sigma)$$



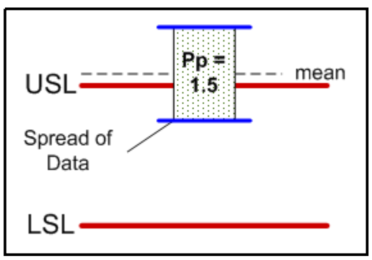
If the spread of data is Well Centered, then Cpk equals Cp. This would be a good situation. In this example, Cp = 1.5 and Cpk =1.5 on day 1.



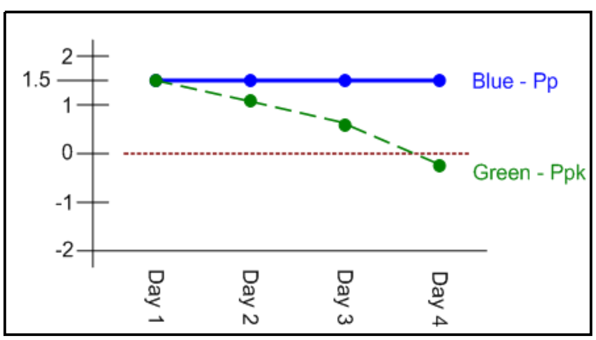
If the spread of data is Off Centered, Cpk is less than Cp. This is a warning sign of a bad trend. In this example, Cp = 1.5 and Cpk =1.05 on day 2.



If the spread of data is outside the USL/LSL limits, Cpk will be below 1. This of course, indicates production problems. In this example, Cp = 1.5 and Cpk = 0.66 on day 3.



If the mean is outside the USL/LSL limits, Cpk will be less than zero. This indicates a serious problem. In this example, Cp = 1.5 and Cpk = -0.28 on day 4.



The graph shows the Cp and Cpk values plotted over the course of the days indicated in the above examples. An ideal situation would be to have Cp and Cpk equal on all days. In this case, Cp (solid blue line) is constant indicating that the spread of the data is consistent. Cpk (dashed green line) is unfortunately not consistent. Tracking the two values in such a way gives you a quick check on your production performance for a particular line.

Note: Careful selection of USL and LSL numbers is essential to have meaningful Cp and Cpk numbers. Calibration of the SoundCheck system is also critical.

Note: Cpk - Process capability index (and Cp): SoundCheck provides Cpk (and Cp) instead of Ppk (and Pp), but they are mathematically identical. Once a process is put into a state of statistical control, process capability is described using process capability indices, which uses the same formula as Cpk (and Cp). The indices are named differently to call attention to whether the process under study is believed to be in control or not. For more information, please visit: **www.isixsigma.com**.

Best Fit to Average

You can select the number of Best or Worst Fit curves that you want to produce. For example, if you have five hundred curves representing the impedances of five hundred speakers, you might want to determine which ten of those were the closest to the average (to find "Golden" reference unit). To do this, you should enter "10" in the "No. Ranked" field in the editor. This will determine which ten curves are the closest to the average, and rank them according to how close they are.

In addition to determining which curves are the best fit, the option also produces a deviation curve for each of the "best fit" curves. These curves are the arithmetic difference between the mean curve and the "best fit" curves.

The best fit is the curve, which minimizes the quadratic distance to the average curve:

$$\varepsilon = \sum_i |X_i - Y_i|^2$$

, where X is the average curve and Y is one of the result curves.

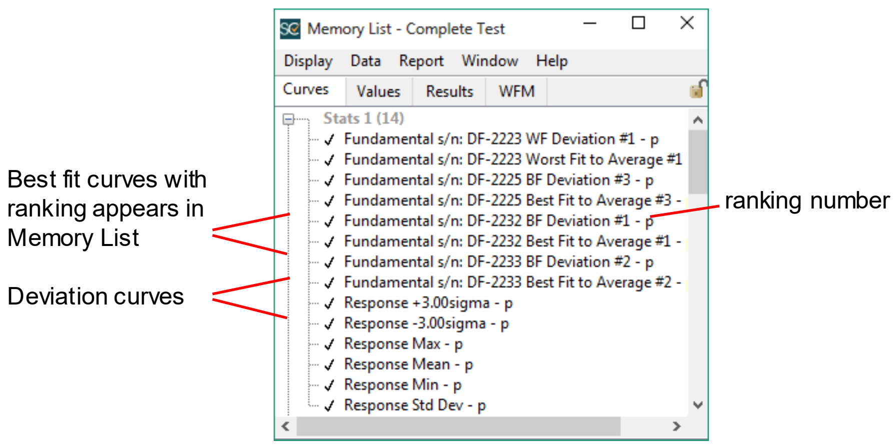


Figure: 24-8 Memory List - Best Fit to Average

The curves selected in **Statistics Offline on page 383**, that meet the Best Fit criteria, appear in the Memory List. The numbers of the curves are indicated along with their ranking. See **Figure 24-8** for an example of the items created in the Memory List.

In addition to the Best Fit to Average function, a Worst Fit to Average function is also included. This function maximizes the quadratic distance to the average curve using the same formula as shown above. With this option, you can isolate a specified number of outliers. As with the Best Fit function, this option will generate a group of curves whose quantity is specified in the editor along with a group of corresponding deviation curves. This function can be useful in determining which of the tests represent useless data. You can remove these curves from their data group and re-perform the statistical measurements in order to get results that are more meaningful.

Statistics for Results

The **Results** section of the *Statistics Editor* (See ***Statistics Offline on page 383***) holds the options for Statistical calculations you can perform on a Result, the Margin or the Verdict.

- **Individual Limits Yield [%]** - Each Result chosen from the list box has its own Verdict (Pass/Fail). Selecting **Individual Limits Yield** will create a running tally of the success rate of each result, as the sequence runs multiple times. This value will reflect the percent of times the Individual Limit has passed and can be viewed next to the Limit Result in the *Results Display* (See ***Results on page 352***), or as a Value in a Table of the Display Step. You could use this value to set a Limits Step, and alert the operator when a Result is failing over 75% of the time.

- **Overall Yield [%]** -A Boolean AND function is performed on the Verdicts of the result names selected in the Statistics Step. A PASS or FAIL Verdict is recorded for that test run. The next time the sequence runs, the new value (after the Boolean AND) will be compared with the previous. For example, after two runs, if one run passed and one failed, the Overall Yield will be 50%.
- **Margin Mean** - Each time the sequence is run, the Margin mean value is recalculated. There will be a unique Margin Mean for each Result selected, named e.g., *Response Margin [L] Margin Mean*. Tracking the Margin Mean can give advance notice that a particular process is drifting towards an out-of-specification condition (e.g., if the Margin Mean is getting smaller over time).
- **Margin Standard Deviation** - The standard deviation is a measure of the dispersion of the result margins. The equation used is the same as the one used for calculating the standard deviation for a curve or value. When choosing this option, two new values are created, named *Response Margin +1.00 sigma* and *Response Margin -1.00 sigma* (assuming 1 sigma was specified). These may be viewed as bounds about the margin being estimated. When these two values are displayed on the Table with the Response Margin Mean value, the Margin Mean value should fall between the two Standard Deviation values. If you choose one 1 sigma (one Standard Deviation) then the bounds created will indicate that 68% of future runs will fall with in these bounds. If 2 sigma is chosen, you know that there is a 95% chance new Margin values will fall between these bounds. At 3 sigma, there is a 99% chance that new Margin values will fall within the bounds. Response signals that fall outside the Standard Deviation curves may indicate a new issue in the DUT or the test environment.

Note: Please note that Standard Deviation values become more accurate when a large number of samples are taken.

- Absolute Margin Standard Deviation - Applies only to the Margin of the Results. It is calculated in the same method as Absolute Standard Deviation.

Adding Statistics Steps to the Sequence

In the case that more than one Statistics Step exists within a sequence, the Statistical values remain completely exclusive to the step. For example, you can insert a jump condition into your sequence, and run a Statistics Step called *Passing Stats* if a Limits Step passes and *Failing Stats* if the Limits Step fails. In this way, you can keep your averages, maxima and minima comprised of only Passing, or of only Failing data.

Verdict of the Step

In the sequence, the Verdict of the Statistics Step is dependent on whether you have selected an Overall Yield calculation in that step. If the **Overall Yield** box has been checked, the step's Verdict will be Pass if all the Results selected are Pass. If any of the Result verdicts are Fail, the Overall Yield will fail, causing the step verdict to be Fail. If **Overall Yield** has not been selected in the step, the step will pass by default.

Note: All Mean and Standard Deviation calculations are done on the numerical values of the data or result. No conversion is made to linear units.

Rules - Statistics

- The Statistics Step must occur before the Display Step that shows the Yield Results tied to those statistics.

Statistics Example Sequence

The Statistics sequence located in the “How to Example” sequence folder can be used as a template when creating a new sequence. These steps can be added to an already existing sequence as well.

Figure 24-9 and ***Figure 24-10*** show the settings of the Statistics steps in the sequence.

The first step is used to calculate statistics on the Fundamental curve and the Response Limits results.

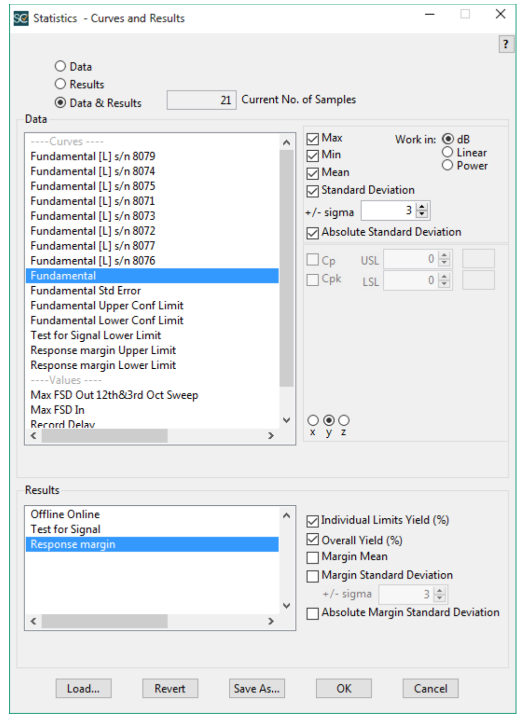


Figure: 24-9 Statistics Step 1

The second Statistics Step is used to calculate Process Performance on the Max SPL value.

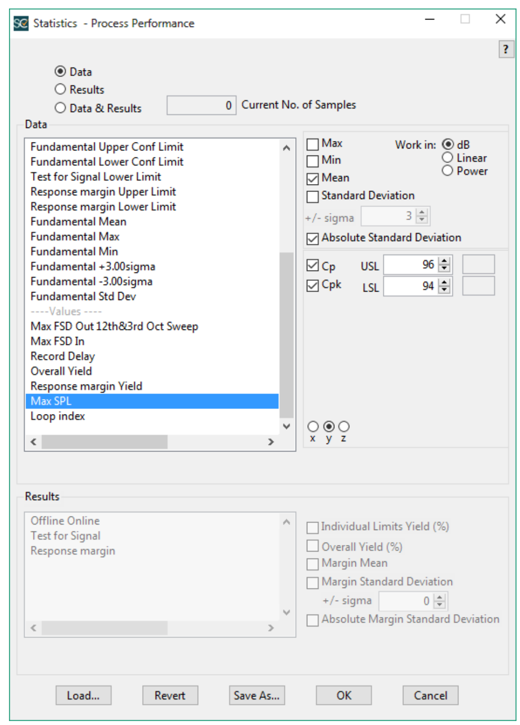


Figure: 24-10 Statistic Step 2

Figure 24-11 shows the results screen of a sample run of speakers.

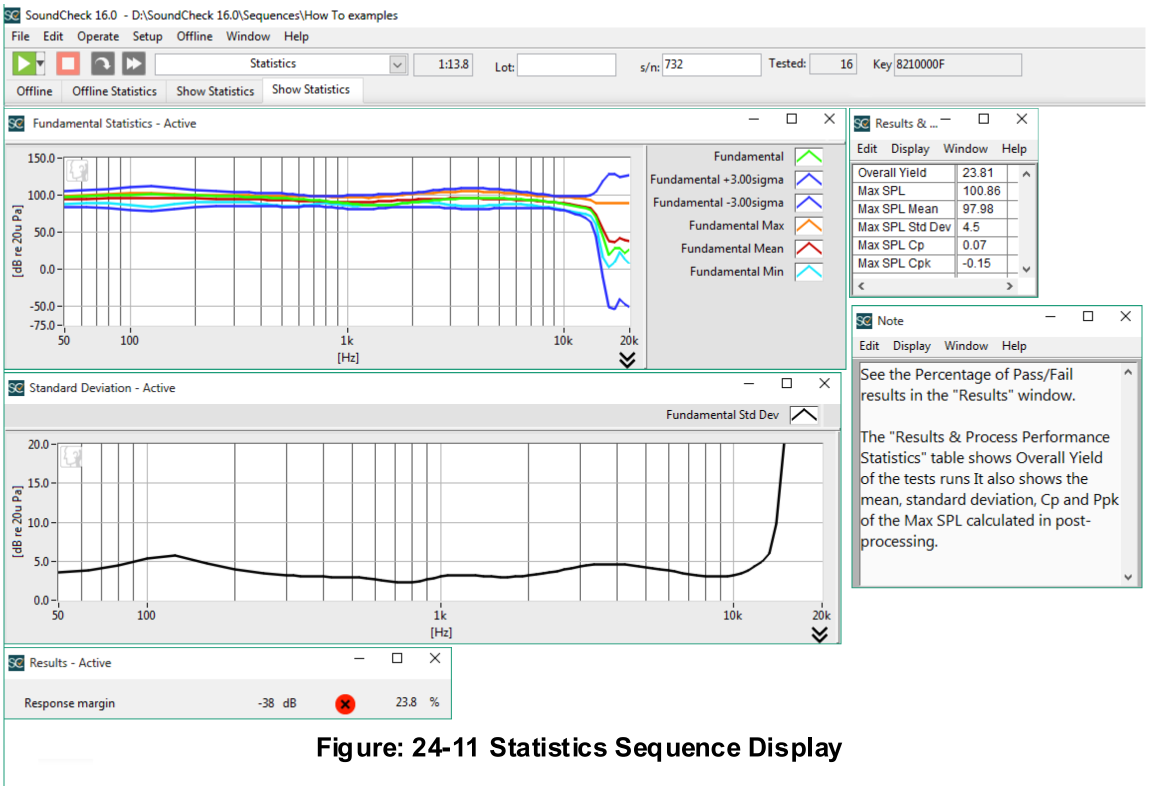


Figure: 24-11 Statistics Sequence Display

SoundMap™

Time Frequency Analysis Introduction

SoundMap™ Time Frequency Analysis is a module which enables detailed analysis of signals simultaneously in both the time and frequency domain. This off-line analysis module can read measurement data from any WAV file or any waveform file created with SoundCheck [.WAV, .TIM (MLSSA), .WFM, .TXT and .MAP]. SoundMap offers the following transforms:

- Short Time Fourier Transform (STFT)
- Cumulative Spectral Decay (CSD)
- Wigner-Ville
- Wavelet

These transforms are ideal for loose particle detection, Rub & Buzz detection and impulse response analysis of loudspeakers. They are also used for identification of transient effects such as drop out in digital devices including VoIP, Bluetooth headsets or transient distortion in MP3 players.

Example data is included in SoundCheck: C:\SoundCheck 18.1\data\SoundMap\Demo Data

When opening SoundMap™, the initial display is a Time-Frequency Analysis window which displays the time signal to be analyzed. From this, you can select which of the four algorithms to use, and define the analysis parameters. The time signal to be analyzed is shown in ***Figure: 25-1***.

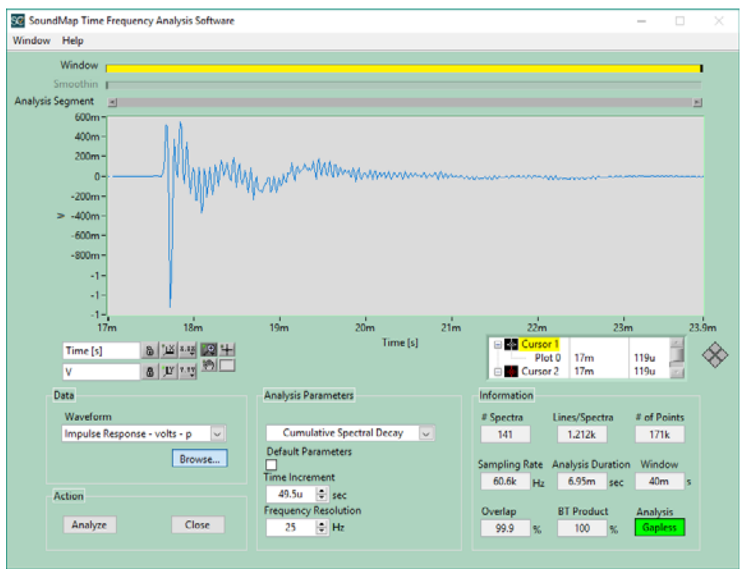


Figure: 25-1 Analysis Window

Displays

SoundMap™ offers a variety of display options including:

- 3D waterfall plot
- Intensity map with time and frequency slices
- Global Energy Spectrum
- Instantaneous spectrum
- Partial Average Spectrum
- Group delay
- Time envelope
- Partial time envelope
- Frequency time curve
- Instantaneous frequency

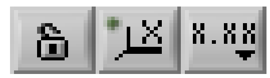
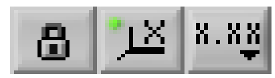
Controls

Axis Controls

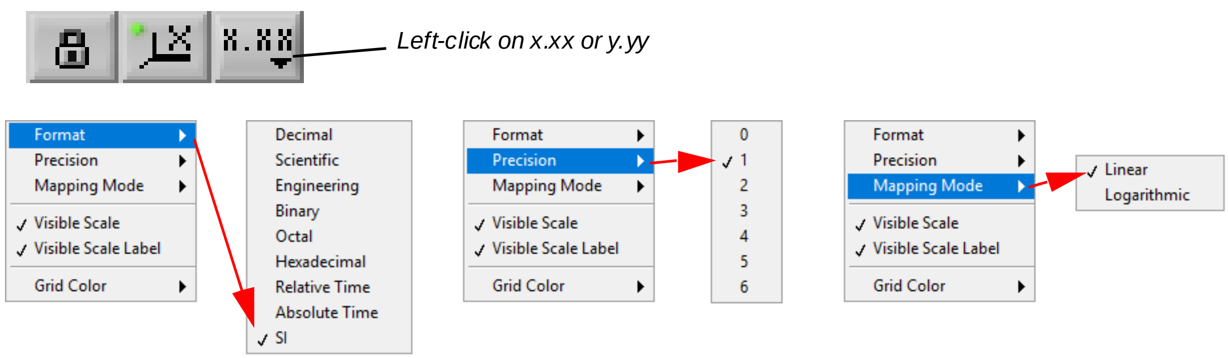
These controls allow you to move the graph, zoom in or out to select the area to be analyzed, and move the cursor by dragging the mouse target along the curve.

Advanced Graph Controls

Autoscale: Left-click on the Lock symbol to turn Autoscale on and off. The Green light indicates that Autoscale is on (as well as the Lock/Unlock symbol). Clicking on the X or Y axis symbol, autoscales the axis without turning autoscale on. This is a "one shot" autoscale.

Autoscale OFF for X axis (Unlocked)	
Autoscale ON for X axis (Locked)	

X and Y-axis formatting: Choose value type, number of decimal places and **Log** or **Linear** scale by Left-clicking on the "x.xx" or "y.yy" buttons.



Grid Color: Changes the gridline color for X or Y axis.

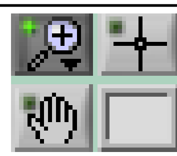
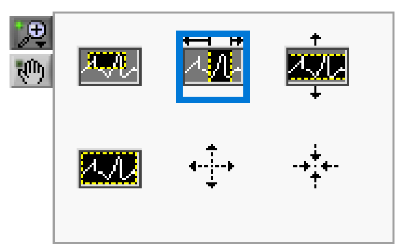
Zoom: Choose to zoom in to different areas of the graph		**Cursor**: Click to activate cursor control tool
Hand: Choose to move graph with hand control		**Graph Background**: Click to change color of background

Figure 25-2: Axis Scaling, Zoom and Style Controls

<u>Zoom</u>

To zoom different sections of the graph, place the mouse pointer on top of the magnifying glass and Left-click. This will open the zoom window, allowing you six (6) different choices.



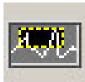
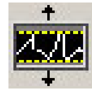
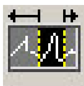
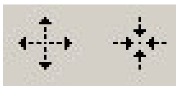
 Zoom in any area of graph	 Zoom vertically along the Y axis
 Zoom horizontally along the X axis	 Undo zoom
	Left- the zoom-in or zoom-out buttons. Move mouse pointer inside graph and single click. Graph will zoom in or out automatically.

Figure: 25-3 Zoom Modes

Cursor Controls

By clicking on a cursor line on the graph, or on the cursor icon in the control box, you can make that cursor active. Once active, the cursor is highlighted and its attributes can be modified.

(**Snap to** is an unused function.)

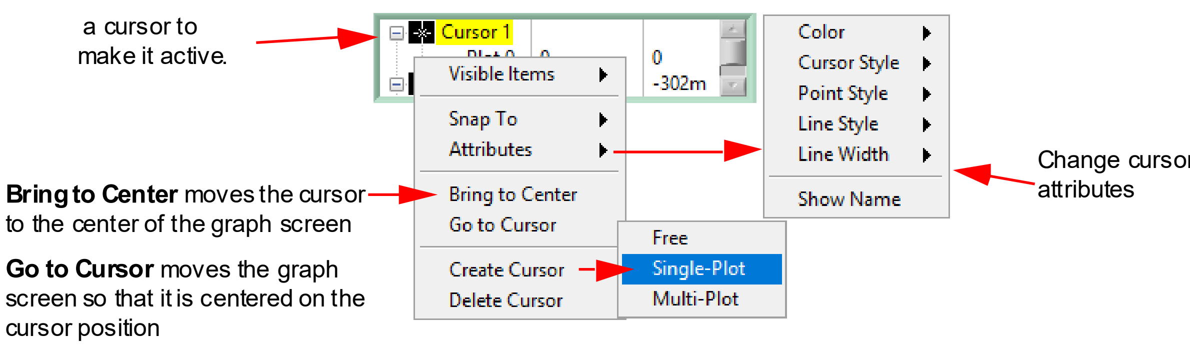
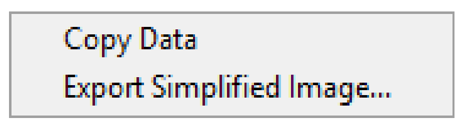


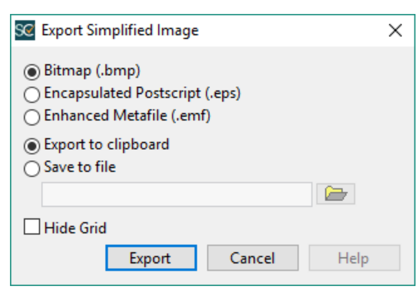
Figure: 25-4 Cursor Controls

Right-click Graph

Right-click any graph in the Analysis or Map windows:



- **Copy Data** - The selected graph is copied to the clipboard so it can be pasted into a report
- **Export Simplified Image** - You can choose to save as .BMP, .EPS or .EMF. Choose to Export to clipboard or Save to file. Hide Grid simply removes the grid lines of the graph.



Short Time Fourier Transform (STFT)

The Short Time Fourier Transform is a general purpose algorithm which enables observation of the spectral changes of a signal over time. This method is ideal for the detection of manufacturing defects such as:

- Loose particles and Rub & Buzz in loudspeakers
- Measurement of settling time and ringing in devices including loudspeakers and telephones
- Analysis of dropouts, discontinuities and instabilities in digital devices

Figure: 25-5 shows the STFT Analysis of a loudspeaker with loose particles.

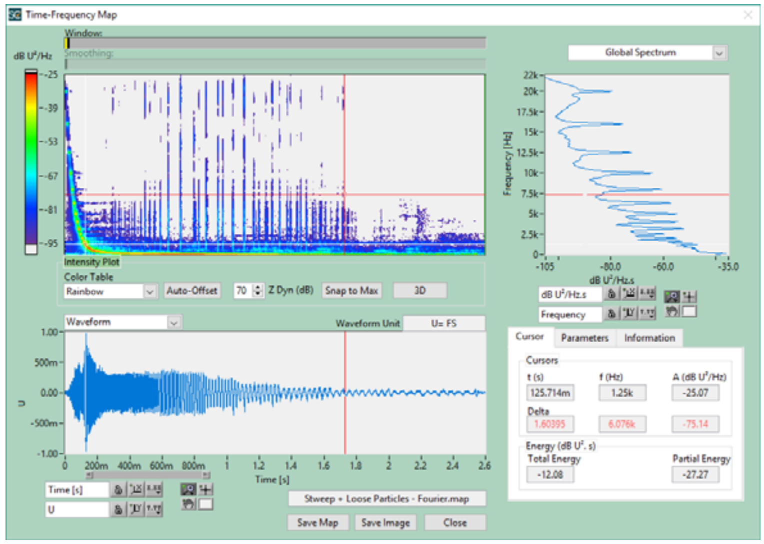


Figure: 25-5 STFT Analysis

Cumulative Spectral Decay (CSD)

Cumulative Spectral Decay is the traditional tool for impulse response analysis of loudspeakers. It calculates the “ringing” of the loudspeaker for each frequency using the impulse response.

Data can be output in a variety of formats including the widely-used three-dimensional ‘waterfall plots’.

An example of a 3D waterfall plot is shown in ***Figure: 25-6***. This is from the analysis of an impulse response of a loudspeaker.

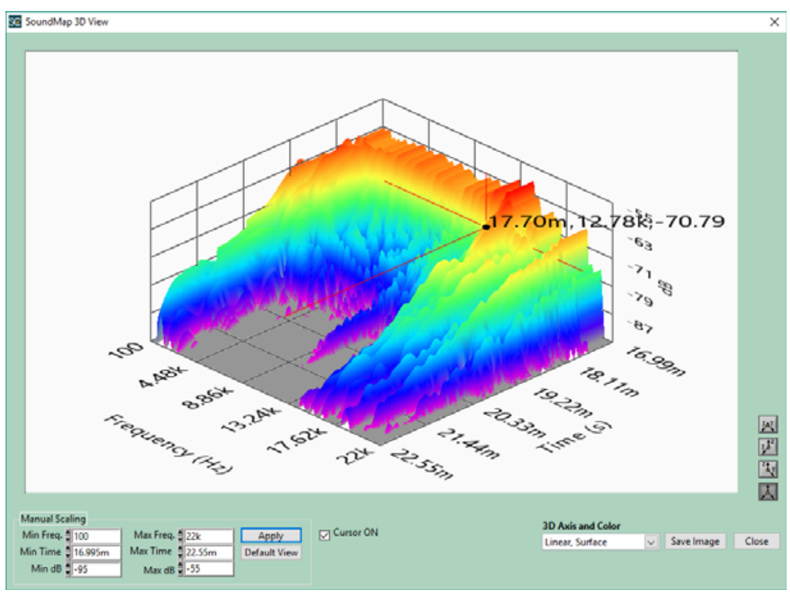


Figure: 25-6 3D Waterfall Plot

Wigner-Ville

Wigner-Ville is the ultimate algorithm for detailed analysis of very short events.

- Fine analysis of transients or in-depth observation of rapidly evolving signals are two examples.
- This algorithm offers an output resolution down to one spectrum per sample
- Wigner-Ville provides the best resolution of all the algorithms
- It complements the more commonly used analysis methods discussed above
- ***Figure: 25-7*** show the analysis of the impulse response of a loudspeaker with a time-slice at 3.69 kHz and the group-delay curve

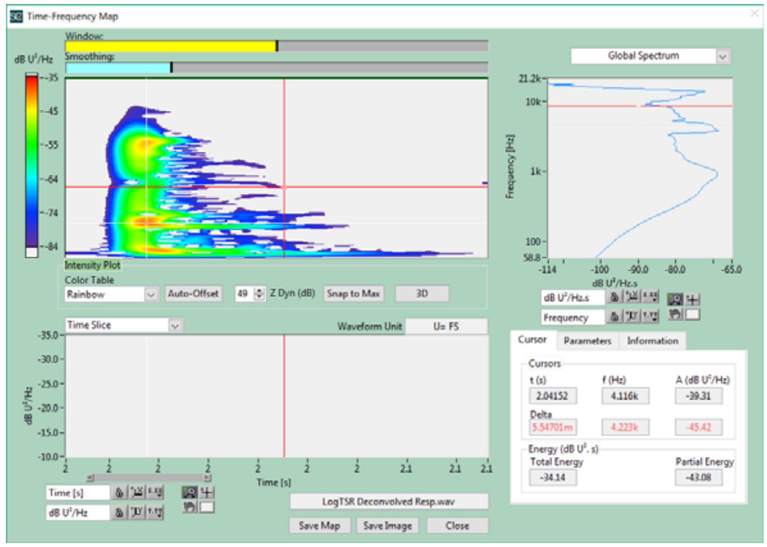


Figure: 25-7 Wigner-Ville Analysis

Wavelet

Wavelet analysis differs from CSD and STFT analysis in that it uses constant percentage bandwidth rather than constant frequency bandwidth. This offers better time resolution at high frequencies and better frequency resolution at the lower end of the spectrum. This is advantageous as it is more psycho-acoustically significant and it is easy to see the entire 20 Hz – 20 kHz spectrum in one picture.

- Applications for wavelet analysis are generally the same as for STFT analysis described above
- The algorithm selected depends on whether constant frequency or constant percentage bandwidth is preferred
- Wavelet analysis presented as a time-frequency map to show Bluetooth dropout is shown in ***Figure: 25-8***

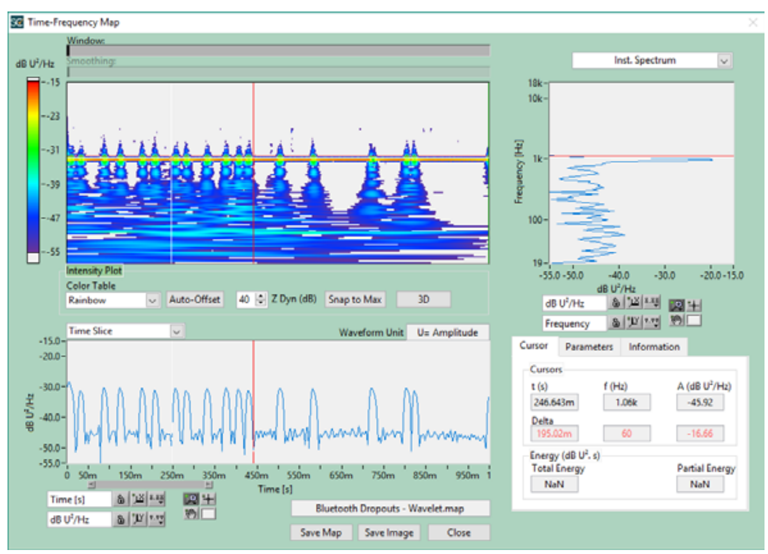
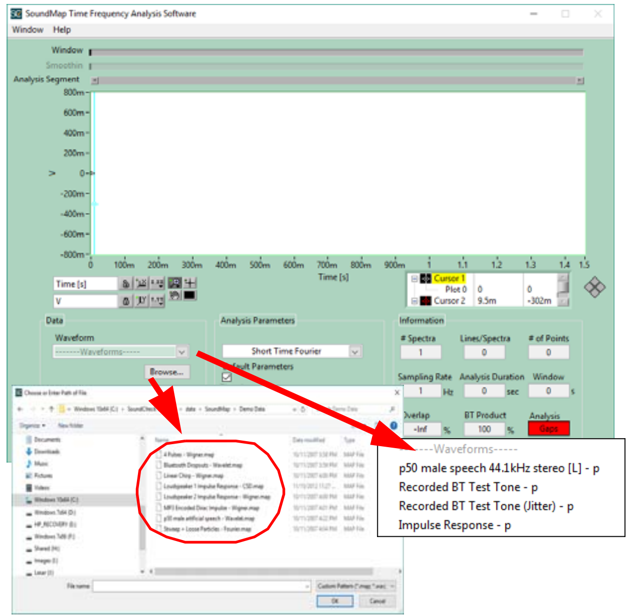


Figure: 25-8 Wavelet Analysis

Time-Frequency Analysis Window

The Analysis Window is displayed when SoundMap is started. This window allows you to select the file to be analyzed, the region of the file to be analyzed and the type of transform to be used.

Data Selection



Select a Waveform from the Memory List or click on the **Browse** button and navigate to the file.

Allowable files types are:

- .MAP - SoundMap data file. Opening a .MAP file automatically opens the Time-Frequency Map window using waveform information stored in the MAP file.
- .WAV - Standard Windows PCM file. See ***WAV File Types on page 336*** for more information on supported WAV file types.
- .TIM - MLSSA time file.
- .WFM - SoundCheck waveform (This must be a linear format waveform - y axis not in dB.)
- .TXT - a text file with specific format (See ***Text File Format on page 421***).

A Waveform file may contain more than one waveform. You will be prompted to select which file should be opened for analysis.

Important! The waveform must be in linear units such as V or Pa, not dB V or dB Pa.

Bargraph Indicators

The bargraph above the Intensity Display shows information on the following:

- **Window** - A graphic display of the window size that is applied to the waveform being analyzed.
- **Smoothing** - The amount of time smoothing applied when using the Wigner-Ville algorithm.
- **Analysis Segment** - Shows the location and block size of the waveform portion being analyzed in relation to the full waveform.

Axis Controls

These controls allow you to move the graph, zoom in or out to select the area to be analyzed, and move the cursor by dragging the mouse target along the curve. The operation of these controls is outlined in ***Controls on page 394***.

Analysis Process

Figure: 25-9 shows the breakdown of how the analysis window moves along a waveform in Time Resolution steps. For each step, a Spectra is created. The resulting Multispectrum is used in the Time-Frequency Map and 3D Waterfall Window.

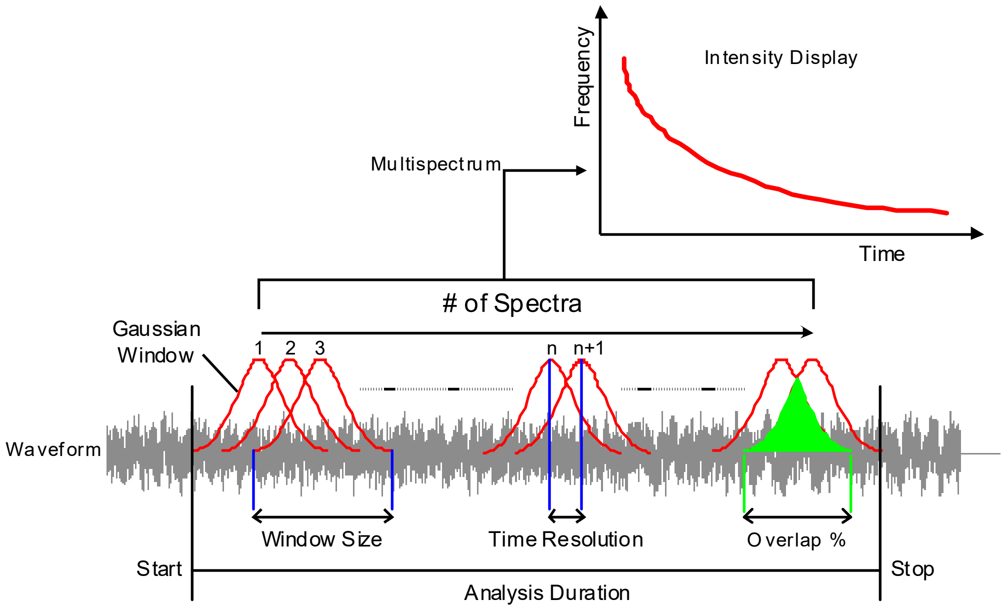
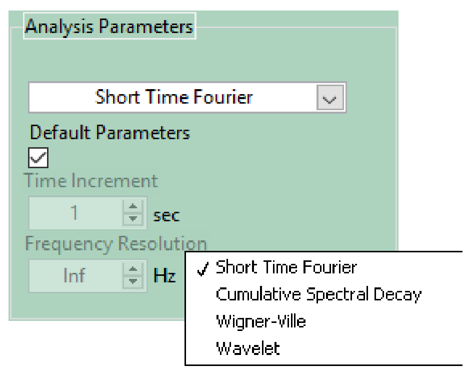


Figure: 25-9 Analysis Process

Analysis Parameters



Algorithm

You can select one of four available algorithms:

- Short Time Fourier Transform - STFT: General purpose algorithm. Not recommended for low frequency analysis.
- Cumulative Spectral Decay - CSD: Specifically for impulse response analysis.
- Wigner-Ville - For tightly focused analysis of waveform details. Sharper resolution than STFT (Super-STFT).
- Wavelet - Much better for general acoustic measurements (logarithmic frequency spacing).

Default Parameters

Using the Default Parameters allows SoundMap to calculate a sensible number of spectra and size of spectra for the block of time data displayed. The default values depend on the algorithm chosen and the current block duration selected in the graph window.

Time Resolution

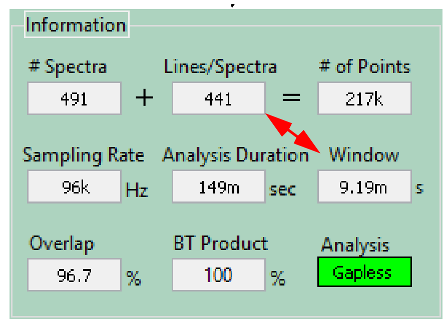
Decreasing the Time Resolution increases the number of spectra in the analysis.

Frequency Resolution

For the first three algorithms this is set in Hz. For Wavelet this is set in octaves, selected from a drop-down list; 1/3, 1/6, 1/12, 1/24, and user defined.

Setting a lower Frequency Resolution increases the analysis window size. The Frequency Resolution is inversely proportional to the window size, i.e.: changing the Frequency Resolution from 100 Hz to 10 Hz will make the window 10 times larger.

Information



- **# of Spectra** - The number of spectra calculated in the analysis segment. This is directly proportional to the Time Resolution.
- **Lines/Spectra** - The number of frequency lines calculated according to the Frequency Resolution specified.
- **# of Points** - The **# of Spectra** multiplied by the **Lines/ Spectra**. This is the number of points in the Time Frequency Map (similar to the number of pixels in a digital photo).
- **Sampling Rate** - The sampling rate of the selected file.
- **Analysis Duration**- Time length of the analyzed portion.This is equal to the **# of Spectra** multiplied by the **Time Resolution** (1000 spectra x 1 mSec Time Resolution = 1 Second Analysis Duration).
- **Window** - Shows the size of the analysis window as determined by the Frequency Resolution setting. For Wavelet, this will be the minimum window size.
- **Overlap** - The amount one analysis window overlaps the previous.
- **BT Product** - Time Resolution multiplied by the Frequency Resolution. Normally this is 100% but for Wigner-Ville this value varies according to the degree of smoothing applied. See ***Algorithm Definitions on page 418***. (BT=1 or Bandwidth x Time equals unity)
- **Analysis** - This shows the Analysis Completeness. **"Gapless"** means that the Overlap of the Analysis Window is sufficient to ensure that no information is lost between windows. This helps to prevent missing short term transient details. If **"Gaps"** are indicated, some data is lost and the curves for Global Spectrum, Partial Spectrum and Group Delay, as well as the values for Total Energy and Partial Energy cannot be calculated. These curves and values will not be available in the Time-Frequency Map. See ***Frequency Display on page 402*** and ***Cursor, Parameters and Information on page 403***.

Important! Using a high number of Points (great number of spectra and/or a very high accuracy in frequency resolution) can use all available computer memory and lead to program instability.

Analyze

Click **Analyze** to run the selected algorithm on the portion of the file selected in the graph display.

Click **Exit** to close the SoundMap program.

Time-Frequency Map

The Time-Frequency Map is available after clicking “Analyze” or after opening a .MAP file.

The Map is an arrangement of three displays and a set of Tabs. The Tabs contain information about the Map and how it was analyzed.

The top left window is the Time-Frequency Intensity Plot.

The top right window is the Frequency Display which shows the different frequency functions.

The bottom left window is the Time Display. Different time functions can be displayed as well.

The information panel at the bottom right side of the screen shows current cursor positions and values as well as information on how the analysis was processed.

Intensity Map Display

This provides a 3 dimensional map of the multispectrum created in the analysis process. The three axis of the map are:

- Vertical Scale - Frequency (axis on right). Linear Scale except for Wavelet, which is Logarithmic.
- Horizontal Scale - Time in seconds (axis on the bottom).
- Color Scale - Indicates the level of power at a specific time and a specific frequency (scale on the left).

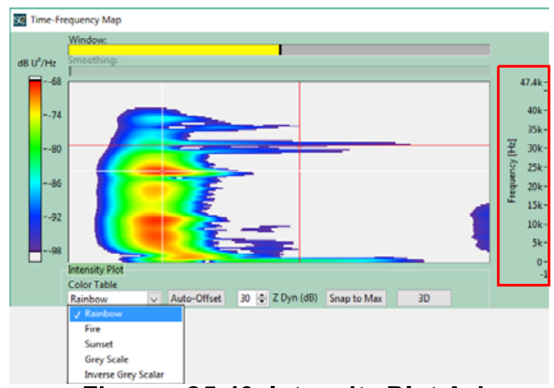


Figure: 25-10 Intensity Plot Axis

The three display windows share the X and Y axis. As the cursor is moved in the Intensity display, the horizontal cursor line is linked to the Frequency Display. Changes in the vertical cursor line are also linked in the Time Display.

The graduated color scale shows level in dB U. In the example, the color scale transitions from bright red at a high level (-63 dB U), to dark blue at a low level (-93 dB U).

Note: U is the unit of the waveform used in analysis. All dB are at a reference of 1.

Color Table

There are five different color settings for the plot and color scale thermometer.

- Rainbow
- Fire
- Sunset
- Grey Scale
- Inverse Grey Scale

Auto-Offset - Applies the dynamic range to the Map and side curves, below the maximum value. You can modify the max and min scale numbers by double clicking on a number and editing it. Level values that are below the minimum of the color scale will be displayed with the background color (not visible).

Dynamic range: The dynamic range of the display is set by typing a value into the **Z Dyn (dB)** field or by clicking on the up/down arrows.

Snap to Max sets the horizontal and vertical lines of Cursor 1 to the absolute peak energy point of the Intensity Plot.

The **3D** button opens the 3D Waterfall display. See ***3D View on page 404***.

Frequency Display

The Frequency Display axis are transposed compared to the conventional way of looking at level vs frequency. This is done so that the spectrum can be viewed in direct relation to the Intensity Plot. Cursors 1 and 2 of the Frequency Display are linked to the cursors in the Intensity Plot.

The following display modes are available:

- Instantaneous Spectrum (U^2/Hz) - Shows the power density spectrum slice at the time location of cursor 1.
- Global Spectrum (U^2.s/Hz) - The sum of all spectra of the multispectrum in the Intensity Plot.
- Partial Spectrum (U^2.s/Hz) - The sum of all spectra of the multispectrum that are between the two vertical cursors.
- Group Delay (s) - The time of arrival for the energy of each frequency. This shows the *mean time of arrival* of each horizontal time slice over frequency. See ***Multispectrum Exploitation: on page 421*** for formula.

The X and Y axis controls function the same as those in the Analysis Window. See ***Controls on page 394*** for more information.

Time Display

The following display modes are available:

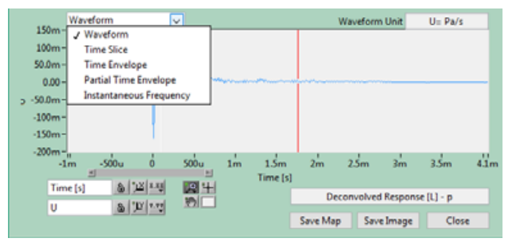


Figure: 25-11 Time Display

- Waveform - Shows the waveform of the analyzed segment.
- Time Slice (U^2/Hz) - The horizontal time slice is taken at the frequency location of cursor 1.
- Time Envelope (U^2) - The sum of all the horizontal time slices in the Intensity Map Display.
- Partial Time Envelope (U^2) - The sum of all the horizontal time slices between the two horizontal cursors in the Intensity Map Display.
- Instantaneous Frequency - This is the plot of frequency vs time which shows the frequency location of the energy at each time point (shows the *mean time of arrival* for each spectra vs time).

The X and Y axis controls function the same as those in the Analysis Window. See ***Controls on page 394*** for more information.

The name of the analyzed file is shown below the Time Display.

- **Save** - Click “Save Map” to store the current analysis as a .MAP file. The analyzed portion of the waveform is stored as part of the Map file. This allows you to open the Map file and change the analysis and display parameters.
- **Save Image** - This saves the entire Time-Frequency Map screen as a .JPG or .BMP file
- **Close -** Closes the Time-Frequency Map and returns you to the Analysis Window

Cursor, Parameters and Information

The Cursor tab shows the values for the current locations of the cursors.

- **t (s)** - The time at cursor 1.
- **f (Hz)** - The frequency at cursor 1.
- **A (dB U^2/Hz)** - The power level at cursor 1.

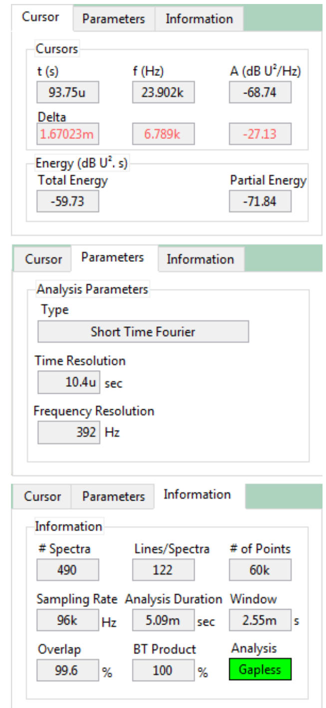


Figure: 25-12 Info Tabs

Delta

- **t (s)** - The time difference between cursor 1 and cursor 2.
- **f (Hz)** - The frequency difference between cursor 1 and cursor 2.
- **A (dB U^2/Hz)** - The power level difference between cursor 1 and cursor 2.

Energy (dB U^2. s)

- **Total Energy** - This is the sum of the entire multispectrum analyzed, both in time and frequency. This is equal to the energy of the waveform analyzed.
- **Partial Energy** - This is the sum of the multispectrum that occupies the area between both the horizontal and vertical cursor lines.

The Parameters tab shows the Analysis Parameters that were used to create the current Map. These are reference values and cannot be edited.

The Information tab shows the information from the original Analysis window that was used to create the current Map. These are reference values and cannot be edited.

3D View

The 3D View or Waterfall Plot allows you to display the analyzed segment in a three-dimensional window showing Level vs Frequency vs Time.

Additional controls on the right side of the window allow you to show 2D displays showing aspects of the current analysis.

Note: The units of the display are in dB FS.

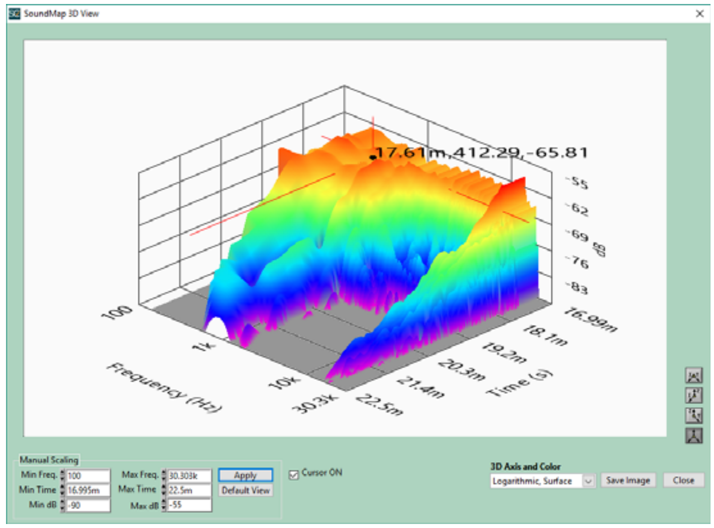
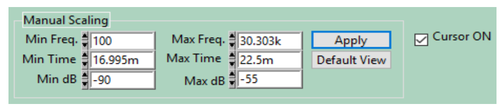


Figure: 25-13 3D View Waterfall

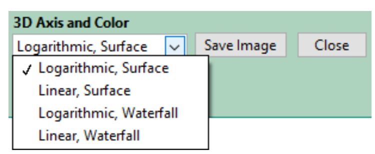
Manual Scaling



- **Min Freq, Max Freq** - Adjust the Frequency Axis range
- **Min Time, Max Time** - Adjust the Time Axis range
- **Min dB, Max dB** - Adjust the vertical scale
- **Apply** - Click **Apply** to use the new axis settings.
- **Default View** - Click to return to the original display angle. This does not change the time or frequency axis ranges.
- **Cursor On** - Activates Cursor

3D Axis and Color

Select from four options for 3D display:



- Logarithmic, Surface - Color Intensity with log frequency scale
- Linear, Surface - Color Intensity with linear frequency scale
- Logarithmic, Waterfall - Monochrome with log frequency scale
- Linear, Waterfall - Monochrome with linear frequency scale

Save Image - Save 3D view to .JPG or .BMP

Close 3D View window

2D View Controls

The 2D control buttons to the right side of the window allow you to select 2d Intensity Plots from the current 3D display. This allows you to put the cursor on a specific point of interest and then switch back to the 3D view.

- Time vs Frequency (X,Y)
- dB vs Time (Z,X)
- dB vs Frequency (Z,Y)
- 3D View (X,Y,Z) - Returns to the 3D view

Rotate

Change the viewing angle of the display.

- Put the cursor on the plot.
- Click and hold on the left mouse button.
- Rotate the plot by moving the mouse.
- Release the mouse button and the plot will be redrawn using the new viewing angle.
- Default View: Click to return to the original display angle. This does not change the time or frequency axis ranges.

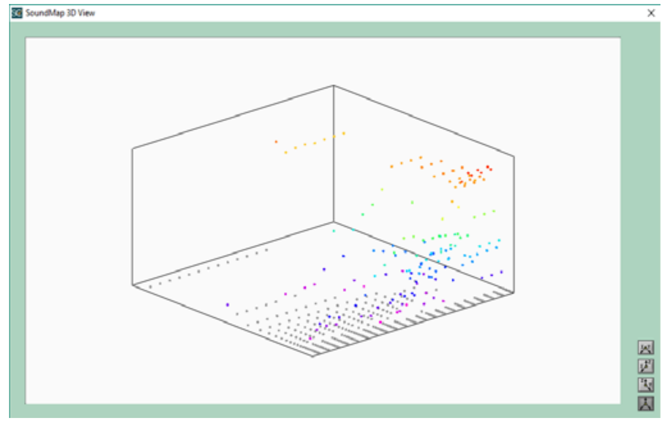


Figure: 25-14 Rotate Viewing Angle

Zoom

You can zoom in and out of the plot view.

- Hold down the “Shift key”. The cursor changes to a magnifying glass.
- Left-click and hold on the mouse button.
- Move the mouse up and down while Left-clicking on the mouse to zoom in and out.

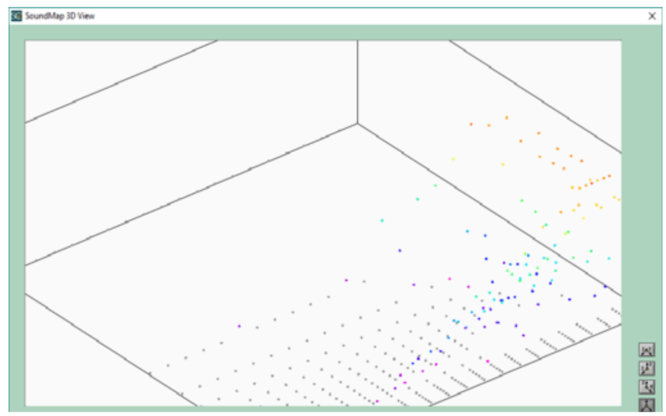


Figure: 25-15 Zoom In/Out

Move

The plot can be moved on the screen.

- Hold down the “Control key”. The cursor changes to cross hairs.
- Left-click and hold on the mouse button.
- Move the mouse to drag the plot to a new location.

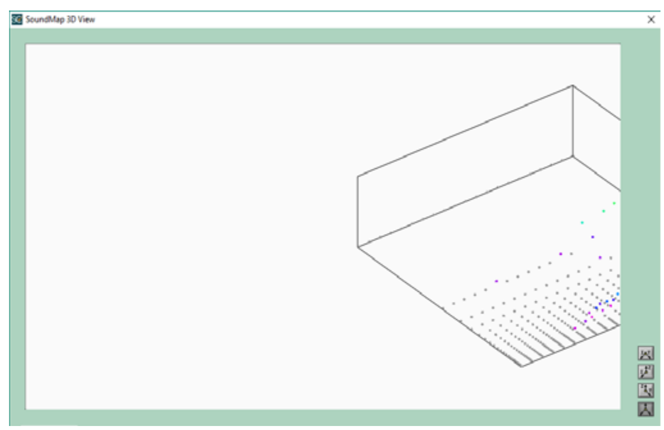


Figure: 25-16 Move Plot

3D Axis and Color

You can choose from 4 different types of plot.

- Logarithmic, Surface - Logarithmic frequency axis 3 dimensional surface map. In addition, the level is color coded using a rainbow scale.
- Linear, Surface - Linear frequency axis 3 dimensional surface map.
- Logarithmic, Waterfall - Logarithmic frequency axis, traditional black and white waterfall plot.
- Linear, Waterfall - Linear frequency axis, traditional black and white waterfall plot.

The spectrum shown are a subset of the complete multispectrum. This is done for display clarity.

The examples in ***Figure: 25-17*** show the two waterfall versions of the same analysis shown in ***Figure: 25-13***.

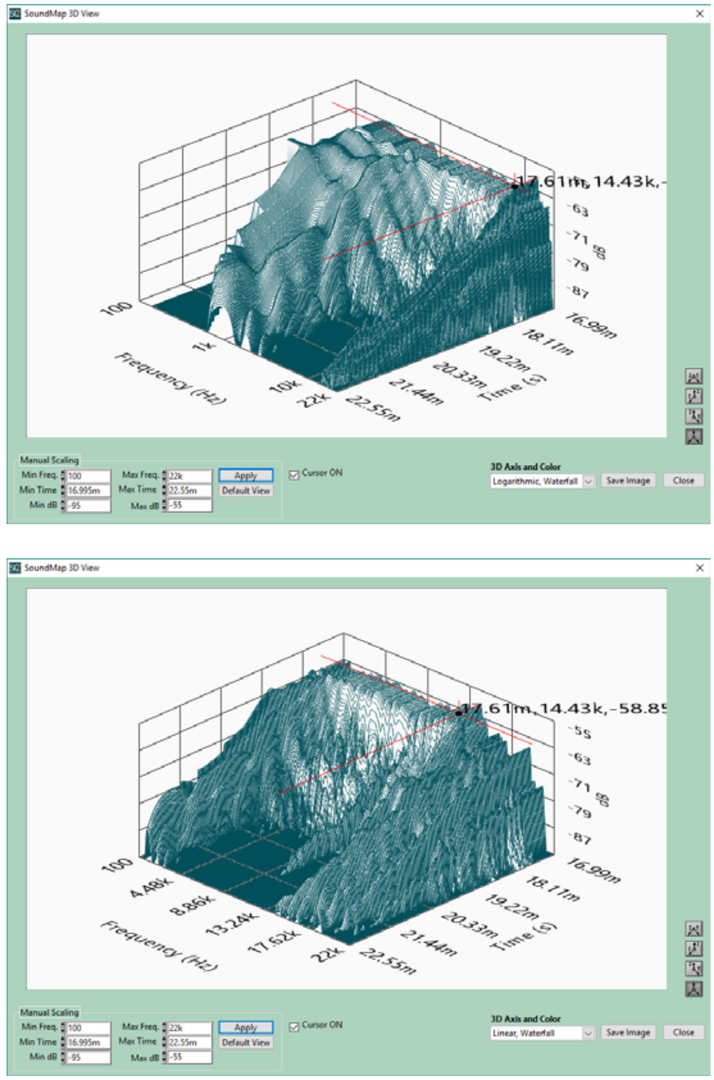


Figure: 25-17 3D View Waterfall Versions

Analysis Examples

The following section shows the four analysis types and examples of how they can be used. The example .MAP files are installed with SoundMap and can be found in the Demo Data folder. Please use these .MAP files while reading this section of the manual.

The “Linear Chirp” and “4 Pulses” files are provided for educational purposes. They do not represent typical signals to be analyzed.

Linear Chirp

Open the “**Linear Chirp - Wigner.map**” file. This file was created using a linear sine sweep from near 0 Hz to approximately 22 kHz. The diagonal response line in the Intensity Display shows how the frequency content changes over time. By moving Cursor 1, you can track the frequency response in the **Intensity Display** and see the **time of arrival** of each frequency in the **Time Display**.

(Time Display set to “Time Slice“.)

The Frequency Display shows the Instantaneous Power Spectrum at that point in time. See ***Figure: 25-18***.

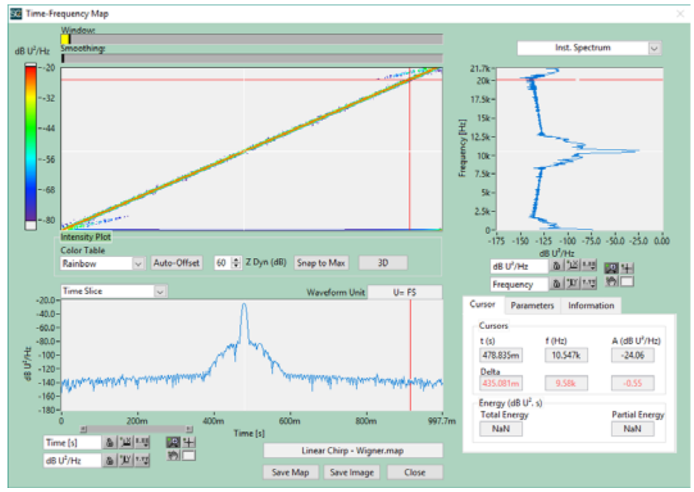


Figure: 25-18 Linear Chirp - Wigner-Ville Analysis

Click **Close** and change the Analysis type to Short Term Fourier (STFT).

Click **Analyze.**

With this analysis method there is less resolution in the **Intensity Display** but the **Time and Frequency Displays** are clearer. See ***Figure: 25-19***.

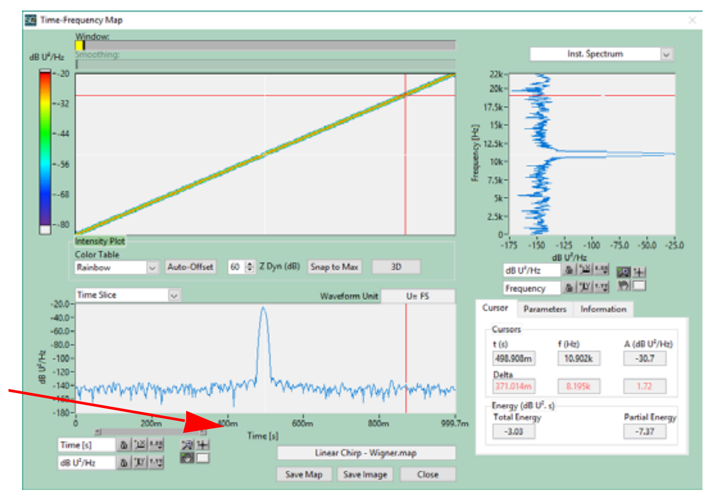


Figure: 25-19 Linear Chirp - STFT Analysis

If you increase the Frequency Resolution in the **Analysis Window**, you can see that the width of the **Intensity Display** gets smaller. Experimentation with Frequency Resolution values will help you optimize the resolution of the analysis of future waveforms.

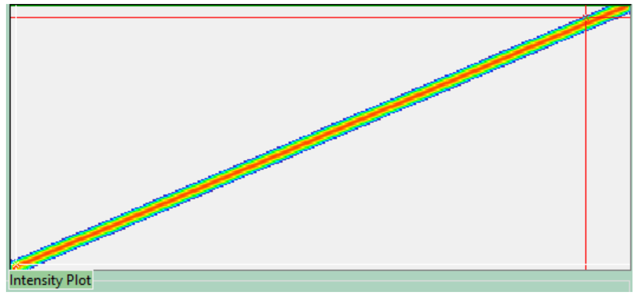


Figure: 25-20 STFT

Change the Analysis type back to Wigner-Ville. Use the default Frequency Resolution but decrease the **Smoothing** to 1 mSec.

Click Analyze and you will see the **Intensity Display** is more narrow, compared to the width in ***Figure: 25-18***.

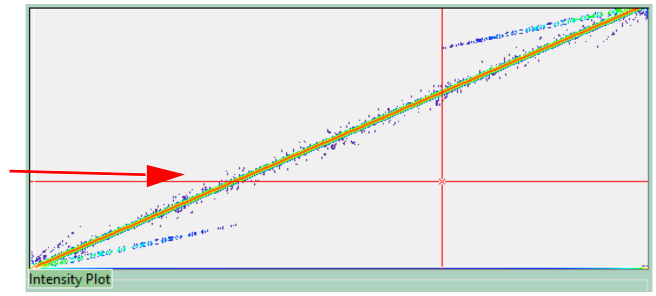


Figure: 25-21 Wigner-Ville

4 Pulses

Open the file, "**4 Pulses - Wigner.map**". This is from a waveform of two overlapping tone bursts followed by a copy of the same pair of tone bursts. The Analysis Type is Wigner-Ville.

In the Time Display you can't determine the difference between the two tones. In the Frequency Display you can see two tones but you don't see their occurrence in time. Only in the Intensity Display do you see four distinct pulses.

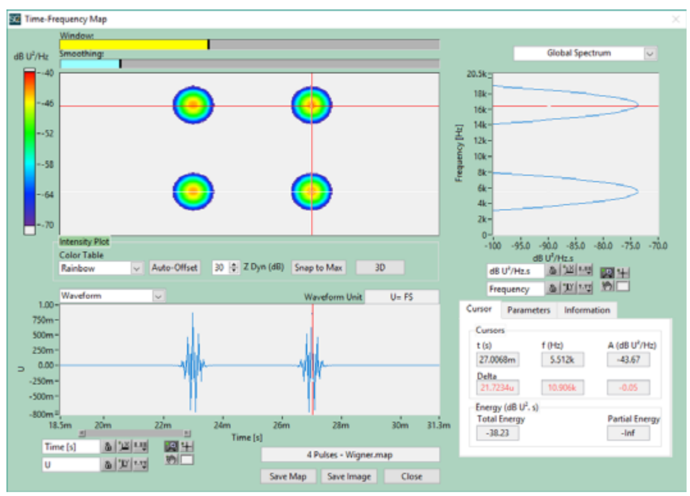
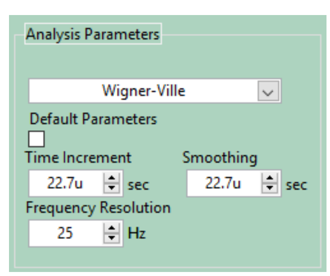


Figure: 25-22 4 Pulses - Wigner-Ville Analysis

1. Click **Close** and uncheck **Default Parameters** in the Analysis Window.
2. Change the **Smoothing** and **Time Resolution** values to 0. Note that the field automatically changes to 22.7 µSec. This is the minimum value for these fields. 22.7 µSec is the sampling interval.
3. Increase the **Frequency Resolution** from 100 Hz to 25 Hz.



4. Click **Analyze** to see the changes in the Intensity Display. The resulting Analysis Window will encompass both sets of pulses. This introduces interferences in the Intensity Display. Note the “Ghosting“ of pulses in the example.

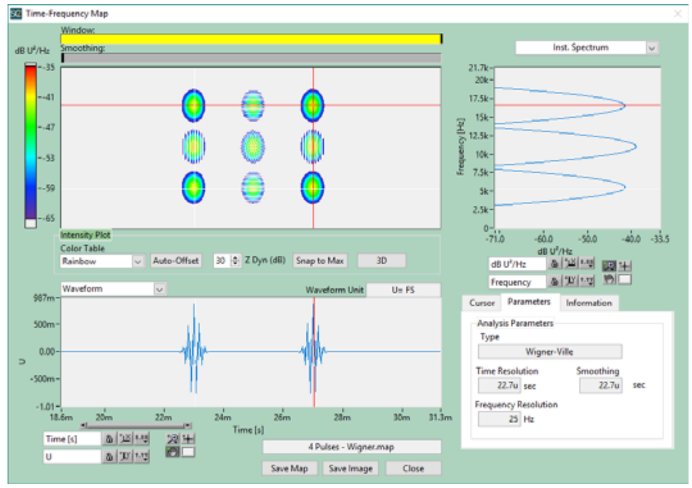


Figure: 25-23 Ghost Interference

General Rules For Wigner-Ville:

- The interferences or “Ghost“ will always occur at the midpoint between two components of the signal.
- The ripple pattern or “Beating“ is perpendicular to the axis drawn between the two components. In the example, you can see the ghosts on the horizontal, vertical and diagonal axes. The center ghost pattern is a product of both diagonal axes.
- The Global Spectrum shows that the ghost interference adds no residual energy to the Total Spectrum.
- The frequency of the ripple pattern in inversely proportional to the distance between the two components. As the distance grows, the beating of the ripple becomes more rapid.
- In the Time Display you can see that the ghost interference adds no residual energy in the time domain.
- To eliminate the ghost that appears between two successive components, the Analysis Window size should be less than the distance between the two components.
- To eliminate the ghost that appears between two simultaneous components, the Smoothing should be long enough to encompass 10 periods of the ghost ripple pattern.

Although the interferences are the representation of real phenomena, such as: time beating between two simultaneous frequencies or harmonic patterns due to signal periodicity, they may obscure the distribution of a complex signal.

In the end, these ghost interferences are caused by having excess resolution and/or insufficient smoothing in the analysis process. Decreasing the **Frequency Resolution** from 25 Hz to 100 Hz removes the interference along the frequency axis. Increasing the **Smoothing** from 22.7 µSec to 1 ms removes the interference along the time axis.

Short Term Fourier Transform (STFT)

Open the file, “**Stweep + Loose Particles - Fourier.map**”. This map was made from a Stweep on a loudspeaker with loose particles under the dust cap.

The log curve of the Stweep is visible as the red curve in the Intensity Display.

Along the time axis, the loose particle impacts are very obvious as shown by the randomly spaced, vertical, blue lines.

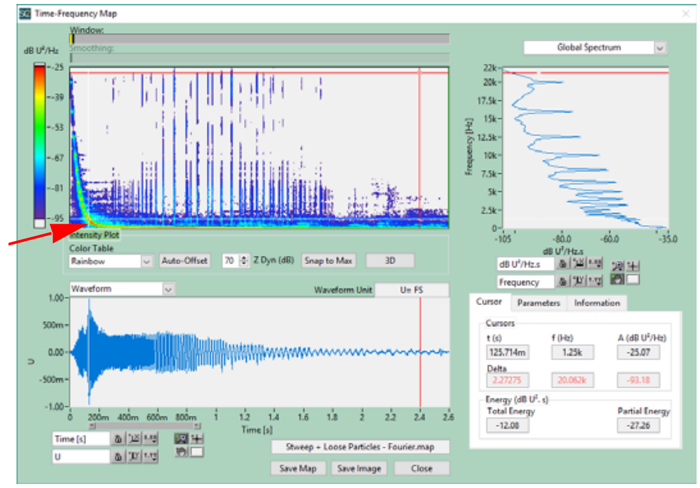


Figure: 25-24 STFT

Note that you cannot see the loose particles in the Global Spectrum. Only by zooming into the Time Display can you see the transients added to the waveform.

Switching the analysis type to Wigner-Ville will increase the resolution so the transients are more defined.

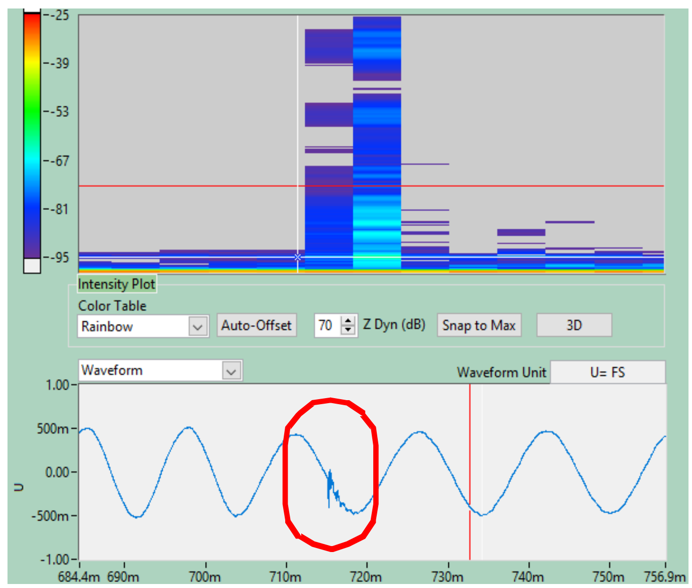
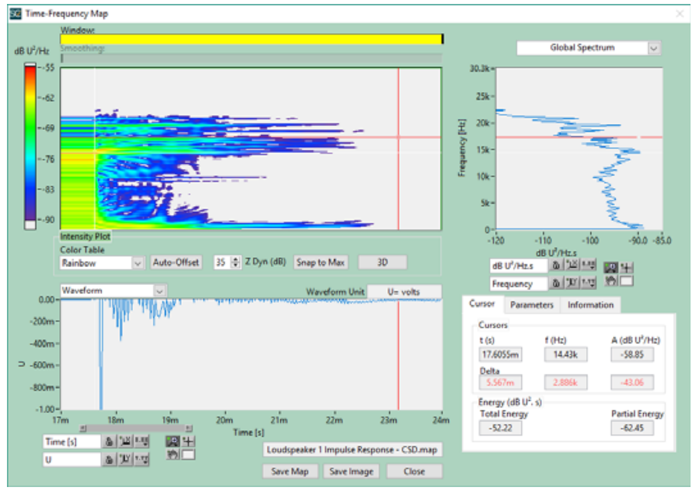


Figure: 25-25 Time Display Zoom

Cumulative Spectral Decay (CSD)

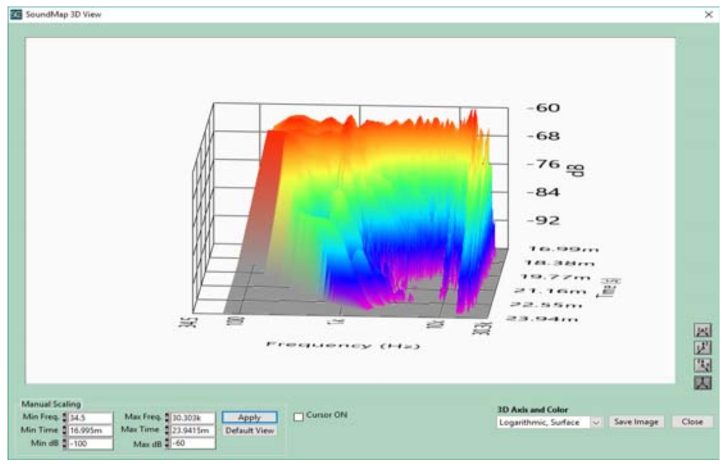
Open “**Loudspeaker 1 Impulse Response - CSD.map**“. This is the impulse response of a Loudspeaker analyzed by Cumulative Spectral Decay. This was made from a MLSSA time file (.TIM).



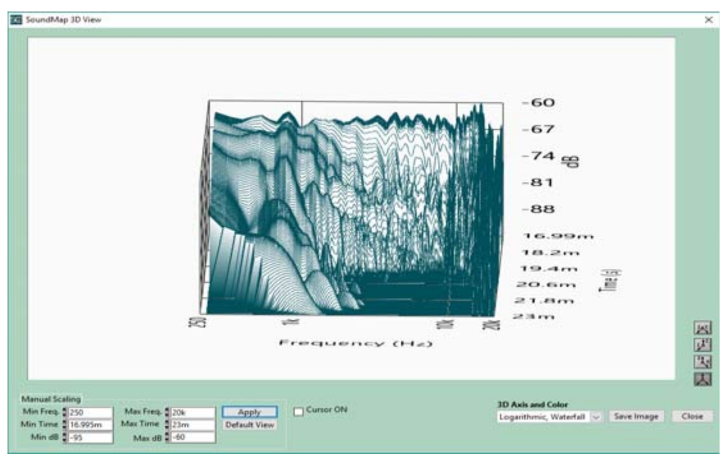
Notice the high frequency ringing between 15 kHz and 20 kHz. There is also ringing below 1000 Hz.

Move Cursor 1 before the start of the impulse response (before 17.5 ms). The **Instantaneous Spectrum Display** shows the actual frequency response of the device under test. This is one of the special features of using CSD.

Next, click the **3D** button to show the Waterfall Display. The default view shows a log waterfall plot with the intensity indicated by color.



Clicking on **Logarithmic, Waterfall** under ***3D Axis and Color*** shows a black and white waterfall display that is equivalent to a MLSSA display.



Under **Manual Scaling**, enter min/max frequencies of 250 and 20 kHz. Enter min/max times of 17 and 23 mSec. Then click **Apply**. This allows the waterfall plot to fill the grid so that the screen is optimized.

Any of the display modes can be rotated on any of the three dimensional axes. The cursor can be switched on and moved to highlight any point in three dimensions to indicate its coordinates.

Click “**Save Image**“ to save a screenshot of the display.

Clicking **Default View** allows you to return to the default three dimensional orientation. This does not change the Min/Max Frequency or Min/Max Time settings.

CSD Analysis Window

With CSD analysis the end point of the analysis window is fixed at the endpoint of the analysis segment. The start point of the window is what changes, as shown in ***Figure 25-26***. As the window moves from one analysis point to another, it gets smaller. As this happens frequencies that have less than one cycle within the remaining analysis window are removed because they cannot be reliably measured.

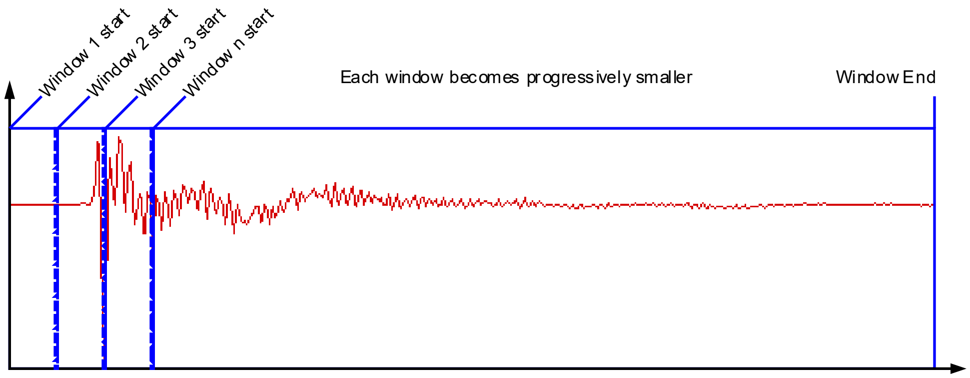


Figure: 25-26 CSD Analysis Window

Wigner-Ville

Loudspeaker 2 Impulse Response

Open "**Loudspeaker 2 Impulse Response - Wigner.map**". This example was made from the impulse response of a two way, near-field monitor. The Global Spectrum shows the overall frequency response. There is a large resonance at 23 kHz. The large spread of energy at this frequency indicates the loudspeaker is ringing at this frequency. There is also a long ring at 31 kHz.

Below 3 kHz the signal not only rings but lags behind the high frequency, as evident from the **Group Delay Display**. This indicates that the woofer and tweeter are not time aligned.

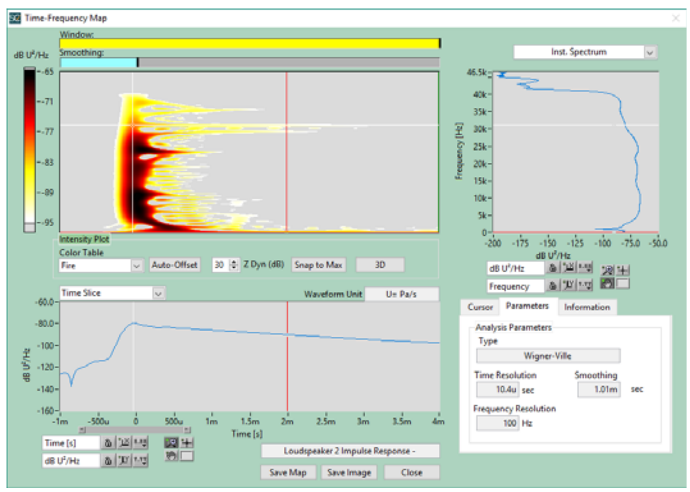
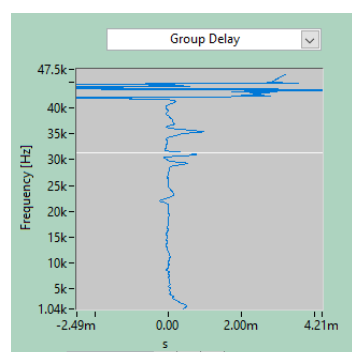


Figure: 25-27 Wigner-Ville

Set the **Frequency Display** to **Group Delay**. The delay peaks are seen here. Ideally, this would be a straight, vertical line.



The ring time or decay time for specific frequencies is also viewed in the **Time Slice Display**. By comparing the position of Cursor 1 to Cursor 2, you can find the decay rate at specific frequencies.

This can be seen in ***Figure: 25-28***.

- Set Cursor 1 at the peak of the signal at 31.1 kHz.
- Move Cursor 2 to a Delta of approximately 2 ms and then adjust it to the point where the Frequency Delta is 0 Hz.
- The Amplitude Delta in this example is 10.38 dB. This indicates that the time constant of the resonance at 31.1 kHz is about 0.83 ms.

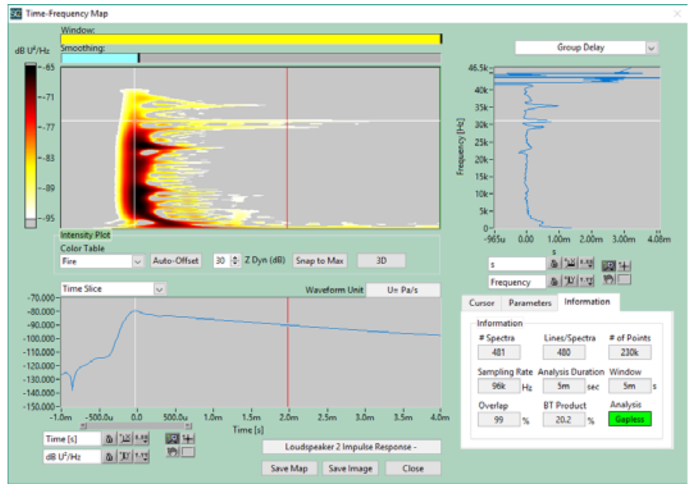


Figure: 25-28 Decay Rate

$$\frac{2\,\mathrm{ms}}{10.38\,\mathrm{dB}} \times 4.34\,\mathrm{dB} = 0.83\,\mathrm{ms}$$

Note: 4.34 dB is exp(1) expressed in dB.

In the **Time Display**, **Time Slice** shows the ringing that occurs between two frequencies. The beating pattern can easily be seen by setting Cursor 1 at approximately 30.1 kHz, between the two frequency peaks as shown in ***Figure: 25-29***.

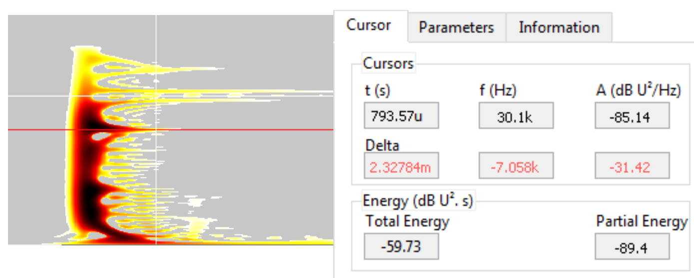


Figure: 25-29 Set Cursor 1 between HF peaks

The Time Slice in ***Figure: 25-30*** shows the beating pattern.

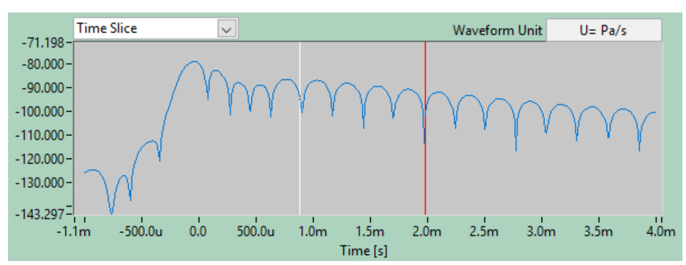


Figure: 25-30 Beating Pattern

To reduce the amount of information in the Intensity Display, you can decrease the **Frequency Resolution** in the **Analysis Window**. This will tend to smooth the distribution of energy along the vertical frequency axis of the Intensity Display.

The example in ***Figure: 25-31*** has been reanalyzed with a Frequency Resolution of 250 Hz.

It is now easier to see the main features of the distribution. Notice that the duration of the ringing frequencies is shorter. This shows that too much smoothing can throw away useful information. It is recommended to always start with a high definition and then smooth gradually. This way you can determine when you have gone to far and return to a more appropriate resolution.

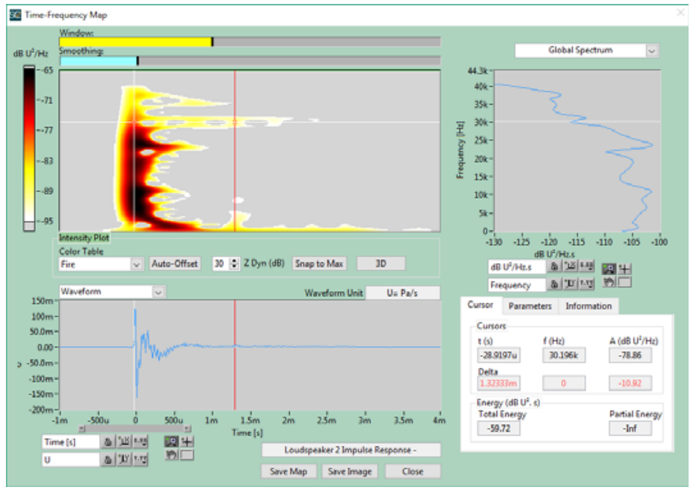
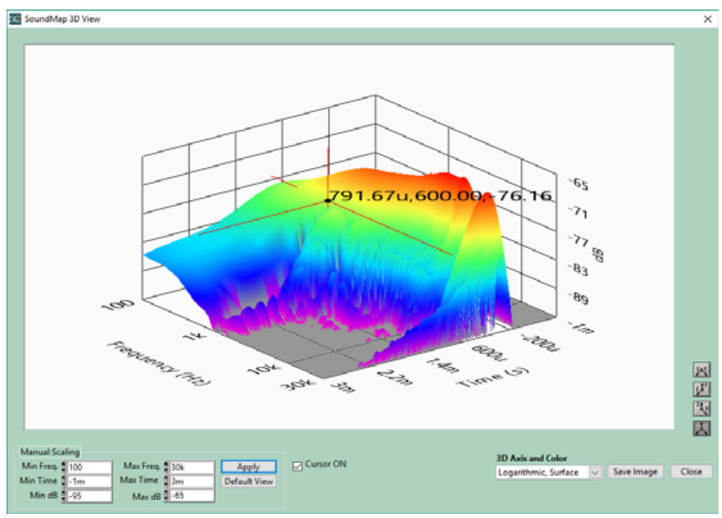


Figure: 25-31 Lower Frequency Resolution Analysis

Select **3D** to show the same information as the **Intensity Map**, with a more intuitive method of viewing.

Note that the Min/Max Frequency and Min/Max Time values have been adjusted to so that the 3D plot is filled with information.



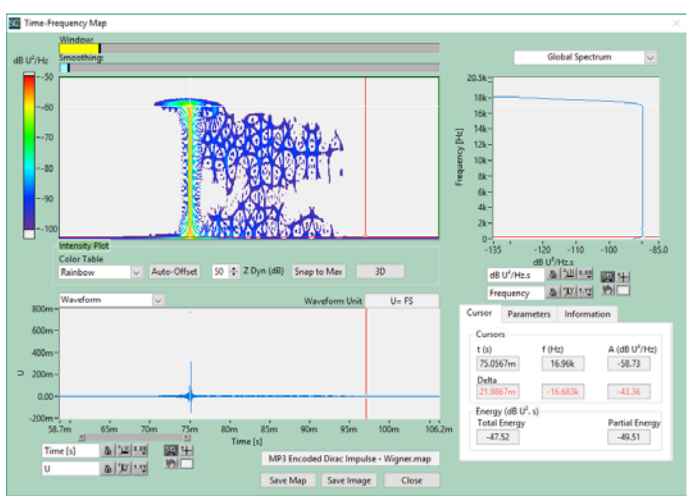
MP3 Encoded Dirac Impulse

Open “**MP3 Encoded Dirac Impulse - Wigner.map**”.

This was made by encoding a Dirac impulse WAV file to MP3 and then decoding it back to WAV. This is a single impulse at 44100 Hz that is 1 sample long with a level of 0 dB Full Scale.

Instead of seeing a single vertical bar in the **Intensity Display**, there is a lot of added noise after the signal and even some added noise before the signal.

The pre-ringing in the high and low frequencies is due to the low-pass filtering that occurs in the MP3 encoding process.



The **Global Spectrum Display** shows the high frequency cut off above 17 kHz.

The signal after the impulse shows the noise created by the encoding process. This is most likely quantization noise due to perceptual encoding.

You can get the spectrum of the noise part by selection it with the two vertical cursors and using the **Partial Spectrum Display** as shown in ***Figure: 25-32.***

Notice that the noise is not flat. There is less energy at approximately 5.53 kHz.

Using Cursors 1 and 2 you can see the duration of the noise in the **Waveform Display**. This is approximately 20 ms.

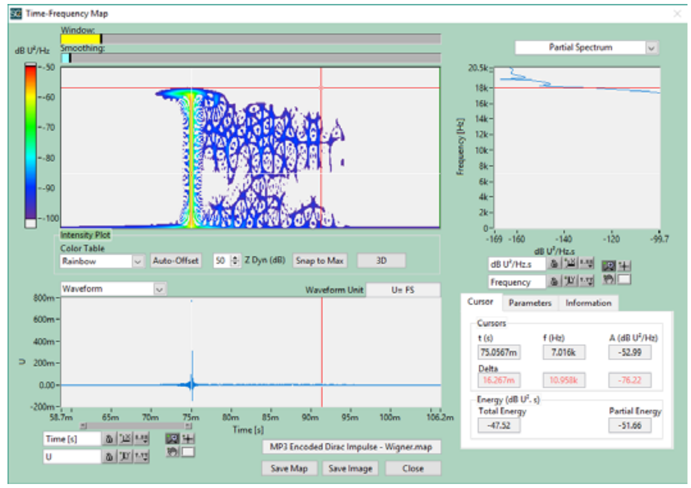


Figure: 25-32 MP3 Noise

Wavelet

p50 Male Artificial Speech

Open "**p50 male artificial speech - Wavelet.map**". This file is the Wavelet Analysis of the p50 speech file that is included with SoundCheck.

The frequency axis of Wavelet Analysis is logarithmic. Logarithmic is the more typically used type of axis for electroacoustic signals.

You can see fundamental of each syllable of the speech waveform as well as each harmonic.

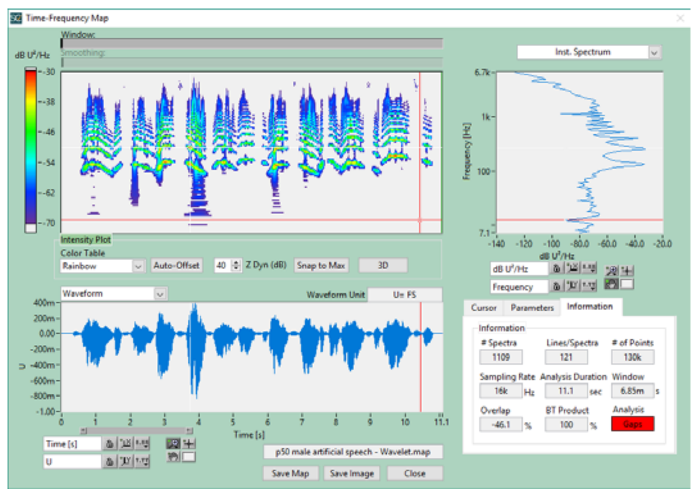


Figure: 25-33 p50 Speech

Bluetooth Dropouts

Open “**Bluetooth Dropouts - Wavelet.map**“. This is a Bluetooth headset measurement using a steady sine wave at 1 kHz.

The constant signal at 1 kHz can be seen in the Instantaneous **Spectrum Display**.

Due to intermittent loss of signal, you can see the effect on the constant tone in the **Intensity Display**.

You can easily see the dropouts by looking at the Time Slice Display. Each dropout appears as a vertical spike over time. Move the Cursor 1 horizontal line to the top of the dropout peaks as shown in ***Figure: 25-34***. The cursor value shows that the peaks of the drop out extend above 1.15 kHz.

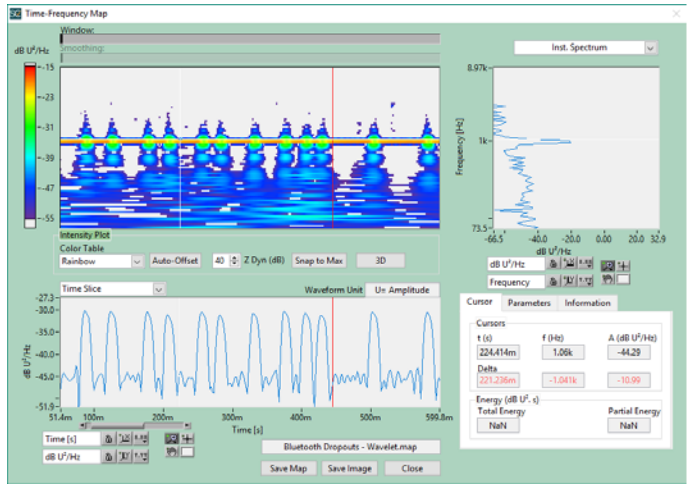
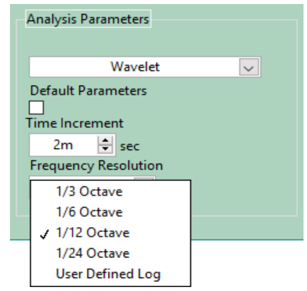


Figure: 25-34 Bluetooth Dropouts

With this information you can now make a 12th octave filter, centered at approximately 1.2 kHz, to act as a dropout detector when testing this device in SoundCheck.

Note: In SoundCheck this is done by limiting the Analysis Step - Time Envelope values to around 1.2 kHz.

The resolution of the analysis can be changed to 1/6th octave by closing the Intensity Display, unchecking **Default Parameters** and then selecting **1/6 octave** from the **Frequency Resolution** drop-down list.



Click **Analyze**.

Wavelet Analysis has low time resolution/ high frequency resolution in the low frequencies and high time resolution/low frequency resolution in the high frequencies. This time resolution is logarithmic, as in an RTA.

As you increase frequency resolution, the time resolution decreases. The two are inversely related.

You can see this by moving Cursor 1 from high frequencies to low frequencies while looking at the **Time Slice Display**.

High frequencies are very defined as shown in the **Time Slice Display** of ***Figure: 25-35***.

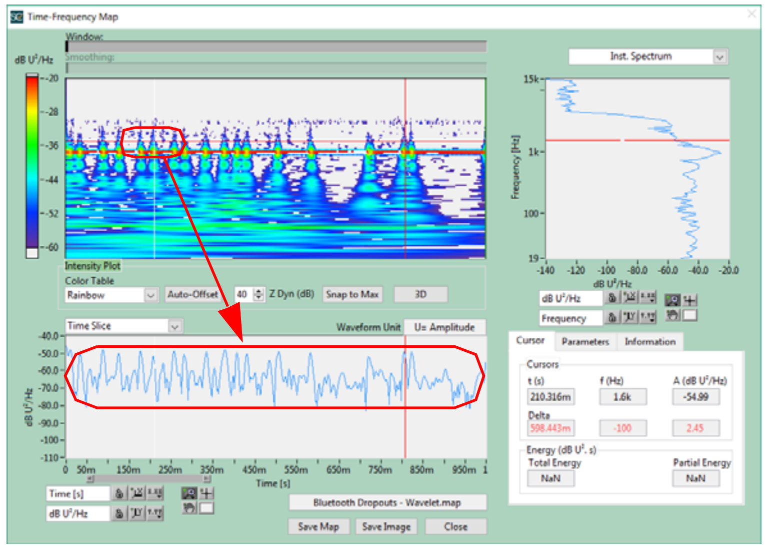


Figure: 25-35 High Frequencies

As you move lower in frequency, the details diminish which can be seen in the **Time Slice Display** of ***Figure: 25-36***

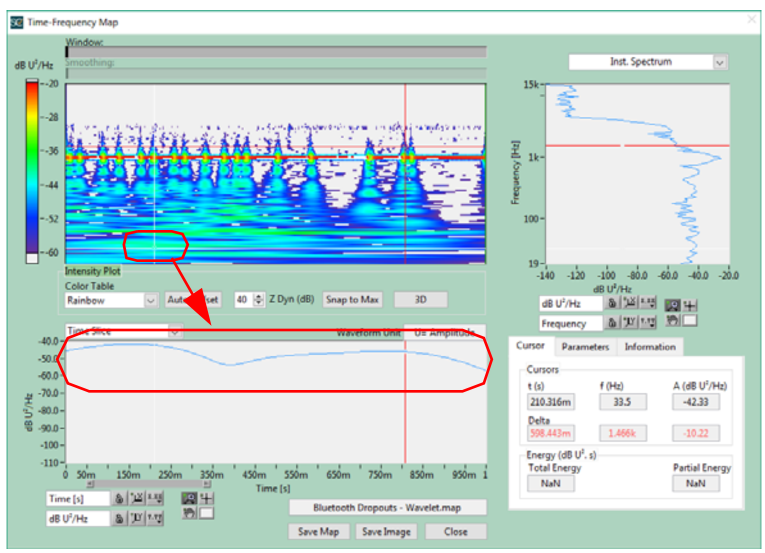


Figure: 25-36 Low Frequencies

Algorithm Definitions

Short Time Fourier Transform (STFT) Definition

SoundMap uses the following formula: $S_x(t, f) = \left| \int h(u) \times (t + u) e^{-i2\pi uf} du \right|^2$

With:

- x(t) = time function to analyze
- h(u)= weighting function for time window

Window

The window used is a truncated Gaussian function. Gaussian functions provide the best time-frequency precision.

Analysis Completeness

If the time increment is below a certain threshold, there is no loss of data and the energy of the signal is conserved. The analysis is then complete. The threshold of completeness is the windows rms duration.

BT Product

It is the normalized product of the window duration by the window bandwidth. It is always 100% for the STFT. It is called the Heisenberg-Gabor limit in the signal processing world. (BT=1 or Bandwidth x Time equals unity)

Cumulative Spectral Decay Transform Definition

At each frequency, the CSD time-frequency distribution yields the temporal decay of a “stopping tone burst” of that frequency. This is applied on the analyzed impulse response. This allows you to see which frequencies are “ringing“ in the Device Under Test, including the room.

CSD is meaningful only when applied on an impulse response. It should not be used on other types of waveforms.

CSD Transform

With x(t) being the signal to analyze, the CSD is defined in SoundMap for a frequency, f, as the squared magnitude of the convolution of x by the stopping tone burst $1_{[-T,\, 0]}(t)\mathrm{Exp}(j2\pi ft)$.

$$C(t, f) = \left| \int_t^{t+T} x(\tau) e^{-j2\pi f\tau}\, d\tau \right|^2$$

The CSD can be expressed as a STFT with a rectangular, right-sided window.

Low frequency limit

Because the analyzed data gets shorter as the integration window slides, the frequency limit for each spectrum gets lower. Under this limit the values are meaningless and set to zero. The limit is equal to the inverse of the effective duration of the integration.

Analysis Completeness

The completeness is determined in the same way as for the STFT.

BT Product

The BT Product is always 100%. (BT=1 or Bandwidth x Time equals unity)

Wigner-Ville Definition

SoundMap uses the following formula, which is a Smoothed Pseudo Wigner-Ville Transform (SPWT):

$$W_x(t, f) = \iint h\left(\frac{\tau}{2}\right) g(u)\ \tilde{x}\left(t + u + \frac{\tau}{2}\right)\ \tilde{x}^*\left(t + u - \frac{\tau}{2}\right) e^{-i2\pi\tau f} du\ d\tau$$

With:

- $\tilde{x}(t)$ = analytical signal from time signal x(*t*)
- h = window function
- g = smoothing function

The window function h is used to limit the time integration range for practical implementation. Each Wigner-Ville spectrum is localized around its time location.

The smoothing function, g, is used to smooth the values along the time axis.

With a smoothing value equal to the sampling interval, the result is an effective time resolution down to the sample. In this case, there may be a lot of interference on a complex signal. See ***BT Product on page 419***.

Analytical Filtering

The analytical filtering removes the negative frequencies of the signal. It is used in the Wigner-Ville analysis to avoid frequency aliasing.

Window

SoundMap uses two separate Gaussian windows for windowing and smoothing. When the window and smoothing functions have the same duration, the SPWT is equivalent to an STFT.

Analysis Completeness

With the SPWV, the Analysis Completeness depends only on the smoothing. The threshold of completeness is the rms duration of the smoothing.

BT Product

With the SPWT, it is the normalized product of the smoothing duration multiplied by the window bandwidth. The two can be adjusted independently.

When BT is smaller than 100%, the time-frequency resolution is better than with the other transforms, but interference appears between components of the signal. This is the price paid when breaking the Heisenberg-Gabor limit. (BT=1 or Bandwidth x Time equals unity)

The smaller the BT, the better the time-frequency resolution, but interference is stronger. Increasing the smoothing allows you to smooth out the time aligned interferences. Increasing the frequency resolution value smooths out the frequency aligned interferences. When the BT Product is 100%, the result is equivalent to an STFT.

Wavelet Transform Definition

SoundMap uses the spectral expression of the continuous Wavelet Transform:

$$\mathrm{WLT}_x(t, f) = \left| \frac{1}{\sqrt{f}} \cdot \int X(v) \cdot \Psi^*\left(\frac{v}{f}\right) \cdot e^{j2\pi vt} dv \right|^2$$

With:

- X(í) = spectrum of the signal to analyze
- Ø(í) = wavelet spectrum

For the wavelet, SoundMap uses a Gaussian analytical wavelet (modified Morlet wavelet) defined in the frequency domain as:

$$\Psi\left(\frac{v}{f}\right) = e^{-\frac{\pi}{2}\left(\frac{v - f}{\gamma f}\right)^2}$$

Frequency Scale

The center frequencies of the analyzing wavelets will follow the standardized, 1/3, 1/6, 1/12, 1/24 octave (RTA) frequencies. The center frequency of each band will use the RTA scale so the analysis can be compared to an acquisition from the SoundCheck RTA analyzer.

Analysis Completeness

As for STFT, the threshold of completeness is the rms duration of the shortest wavelet (maximum analysis frequency).

BT Product

The BT Product is 100%. (BT=1 or Bandwidth x Time equals unity)

Multispectrum Exploitation:

Global Energy Spectrum †*:* sum of all spectrum of the multispectrum. It is the energy spectrum of the analyzed signal.

Sub-Total Spectrum †*:* sum of all spectrum of the multispectrum, between the two vertical cursors. It is the energy spectrum of the portion of signal that lies between the two vertical cursors.

Energy Calculus †*:* the total energy of the analyzed signal is obtained by summing of all time-frequency values of the multispectrum. The partial energy is obtained by summing the multispectrum values in the time-frequency region delimited by the two cross-cursors.

Energy Time Curve: sum of all time slices of the multispectrum. This yields the time envelope of the signal.

Partial Energy Time Curve: sum of all time slices of the multispectrum, between the two horizontal cursors. This yields the time envelope of the filtered signal, limited to the frequency band delimited by the two horizontal cursors.

Mean Group Delay †*:* center of gravity of each time slice over frequency. It is the time of arrival of the energy for each frequency.

$$T_x(v) = \frac{\int t\,\rho_x(t, v)\,dt}{\int \rho_x(t, v)\,dt}$$

Mean Instantaneous Frequency: center of gravity of each frequency slice over time. It is the frequency location of the energy at each time.

$$F_x(t) = \frac{\int v\,\rho_x(t, v)\,dv}{\int \rho_x(t, v)\,dv}$$

Note: †: *These functions are only available when Analysis Completeness is attained.*

Text File Format

- Physical unit string <CR-LF> (Must be linear units such as V or Pa, not dB V or dB Pa.)
- Sampling rate in Hz <CR-LF> (Such as 44.1 kHz sampling rate)
- Real Values (floating point) <CR-LF> .. <CR-LF> (one value per line. For a sampling rate of 44.1 kHz there will be 44100 value lines.)

page intentionally left blank

Custom Steps Included With SoundCheck

As of SoundCheck 16, the Exit Status of custom steps is now indicated. Like other SoundCheck steps, the Sequence Editor now highlights in red or green whether a custom step completed successfully or not.

Outline Ethernet

As of SoundCheck 16, ethernet control for the Outline ET250-3D turntable is included in a custom step.

> ***Note:*** In order to configure the Outline Ethernet custom step, it must first be inserted into a sequence from the Template Library.

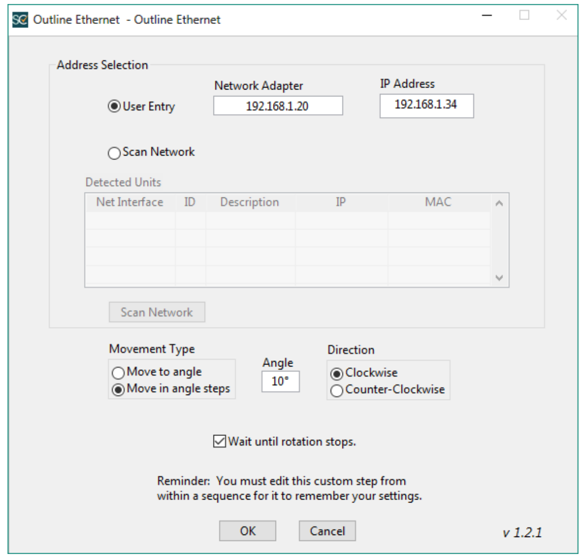


Figure: 26-1 Outline Ethernet Control

Address Selection

Scan Network

Under **Address Selection**, select Scan Network and then click the **Scan Network** button. SoundCheck will search for the turntable and automatically populate the address information into the Detected Units table. If the host computer is on both a LAN and wireless network, multiple instances of the turntable may be visible in the Detected Units table. This is the recommended method.

User Entry

If you are writing a sequence which will be used at another location, you may want to manually enter the known Network Adapter address.

Enter the host computer's Ethernet IP address in the Network Adapter field and the turntable's address in the IP address field. Note that the computer Ethernet IP address must be set in Windows prior to using the custom step in SoundCheck. If the computer is connected to a network using an ethernet connection, a separate ethernet interface will be required for the connection to the turntable.

Movement Type

Move to Angle

Input a specific target angle into the Angle entry box, e.g.: Enter 0° to return the turntable to its Home position.

Move in Angled Steps

Input the desired angular increment of rotation into the Angle entry box. This would be used to move the turntable in 10° increments for measurement of polar data.

Direction

Specify clockwise or counter-clockwise rotation of the turntable

Wait Until Rotation Stops

This will pause the operation of the SoundCheck sequence until the turntable stops moving before continuing on to the next step.

For complete setup instructions, please refer to the example sequence and sequence note included with SoundCheck.

Instrument Open Close

This demonstrates opening and closing Virtual Instruments from a Custom Step. It serves as a template so you can create a custom step that combines your LabVIEW code with the operation of virtual instruments.

1. Open the necessary Virtual Instruments in SoundCheck. Set them as required for use with your code. See ***Virtual Instruments on page 455***.
2. Save them as a .VIC file. See ***Virtual Instrument Configuration on page 458***.
3. Open your LabVIEW development system.
4. Open "**Instrument Open Close.vi**" to use as a template.
5. Before doing anything else, select "**Save As**" from the file menu and give the template a new name, e.g.: "My Code with SigGen.vi".
6. It must be saved in your current SoundCheck installation, e.g.: C:\SoundCheck 18.1\System\Custom VIs\.
7. The example shows where you should enter your code.
8. Specify the location of the .VIC file.

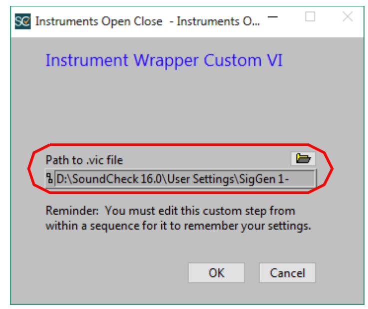


Figure: 26-2 Instrument Open Close

Note: The file path to the .VIC file is absolute.The VIC file will need to be copied to new versions of SoundCheck and the file path used in your Custom Step will need to be updated when you update to a new version of SoundCheck.

9. Next, open the LabVIEW VI template "SoundCheck\System\Custom VIs**Instrument Open Close Editor.vi**".
10. Select "**Save As**" and use the same name from the previous step. Make sure it contains ' Editor.vi' after the custom name. For example: and "**My Code with SigGen Editor.vi**"
11. Edit the new custom step in **LabVIEW 2018**. A place marker for your code has been left in the template as shown in ***Figure 26-3***.
12. For more information refer to the instructions in ***Creating a Custom VI and Custom Step on page 431***.

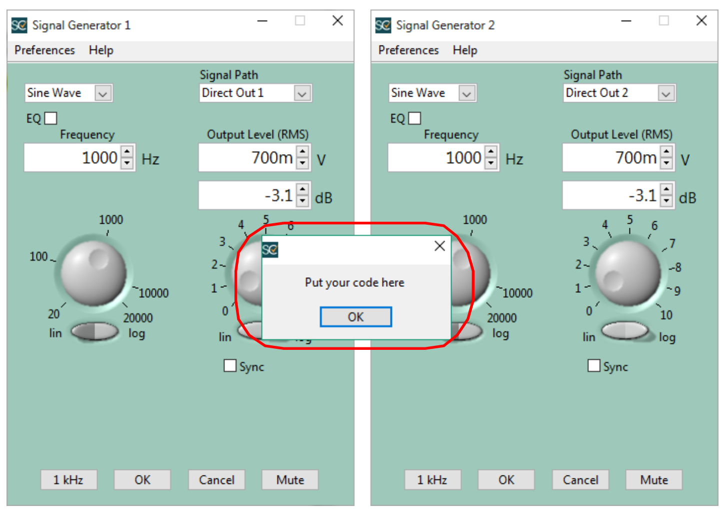


Figure: 26-3 Put Your Code Here

See ***Instrument Open Close Custom Step on page 492*** for instructions for use in TCP IP.

System Custom Step - Windows

Included in SoundCheck Steps library is the Custom Step, System.cus. This allows you to run Command Line operations as part of a sequence. This includes:

- executable files (.EXE)
- batch files (.BAT)

Command Line Field

If the executable is not in a directory listed in the Windows PATH environment variable, the command line must contain the full path to the executable.

As of SoundCheck 17, when executing Batch Files or Executable Files that require a Command Line Interface, **"cmd/c"** no longer needs to be added before the file name in the **Command Line Field** as shown in ***Figure 26-4***.

- Batch files may require that the Working Directory is specified
- When the **System** Step is executed in the sequence, it will perform the functions in the batch file

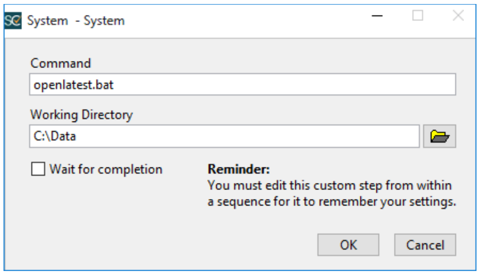


Figure: 26-4 System Step - Command Line

The following example batch file is useful for opening the latest Excel file, according to the "Date Modified", from the specified folder. Batch file contents:

```
@echo off
for /f "eol=: delims=" %%F in ('dir /b /od *.xls') do @set "newest=%%F"
"%newest%"
```

In order to open the latest Word document you would change *.xls to *.doc.

Windows Executable Files

When running an executable file that runs from the standard desktop, only the name of the executable needs to be in the **Command Line Field** as shown in ***Figure 26-5***. The Working Directory is not required for this type of executable file.

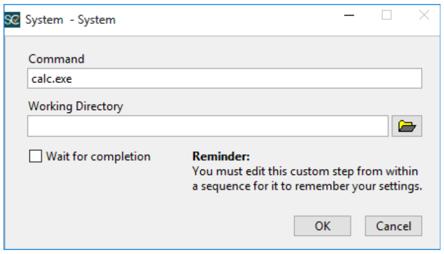


Figure: 26-5 Open Calc.exe

Working Directory

Working Directory is the file system directory from which you want to execute the command.

Note: Do not use working directory to locate the executable you want to run. Working Directory applies to the executable only after it launches.

Wait for completion
When checked, SoundCheck will wait for the operation called in the Command Line field to complete or be closed, depending on type of operation called. When this is left unchecked, SoundCheck will execute the Command Line and continue to the next step in the sequence.

Run as administrator

The next step is optional depending on where the batch file is located and what the batch file tries to access. When you launch SoundCheck, it launches under the current user account. By default, this account only has read/write access to its own user folder. So you can run SoundCheck as the administrator by Right-clicking on the shortcut and clicking “Run as administrator”. This will open up the privileges. Another way to work around that is to move the batch file and restrict it’s “movement” to the account user’s folder.

System Custom Step - macOS

Included in SoundCheck Steps library is the Custom Step, System.cus. This allows you to run Command Line operations as part of a sequence. This includes:

- application files (.APP)
- script files (.SH)

Command Line Field

When calling a script in the Command Line you must enter “**bash**” before the script name. The example in ***Figure 26-6*** calls “**script.sh**” from the Working Directory:

“**/Users/support/Documents**”

Working Directory
Working Directory is the file system location of the application or script that is being called.

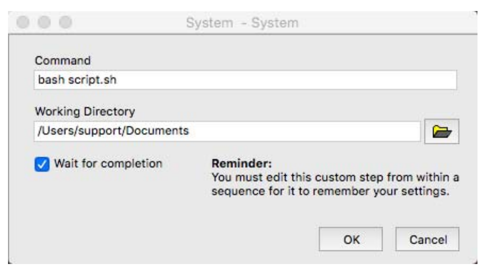


Figure: 26-6 Run Script

Wait for completion
This must remain checked in macOS.

The example in ***Figure 26-7*** shows how to open an application file.

In the Command Line enter:

“**open -a calculator**” (**-a** specifies that the target of the open command is an application)

In the Working Directory enter the location of the app:

“**/Applications**”

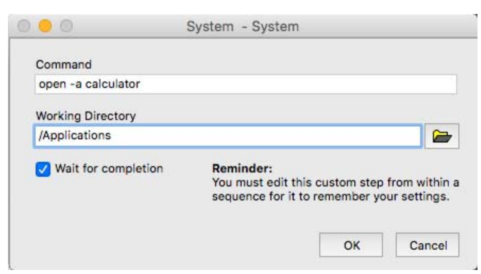


Figure: 26-7 Open Calculator

Mixer Volume

Each Mixer Volume Step in a sequence, allows you to control the input and output levels of a 2 channel WDM / WASAPI or Core Audio device.

For example, the volume of a headset may be controlled while testing it, or the levels of a device can be fixed at unity gain for consistent calibration.

Figure 26-8 shows the Input and Output of a Bluetooth Headset set to 100%. This will require two Mixer Volume steps in the sequence.

- Compatible with Windows and macOS
- Compatible with WDM / WASAPI or Core Audio devices
- Separate steps are required for Playback and Record
- Separate steps are required for each device being controlled
- The steps cannot be renamed in a sequence. The name must remain “Mixer Volume” for each instance. Use Step Comments to name the function of each step.

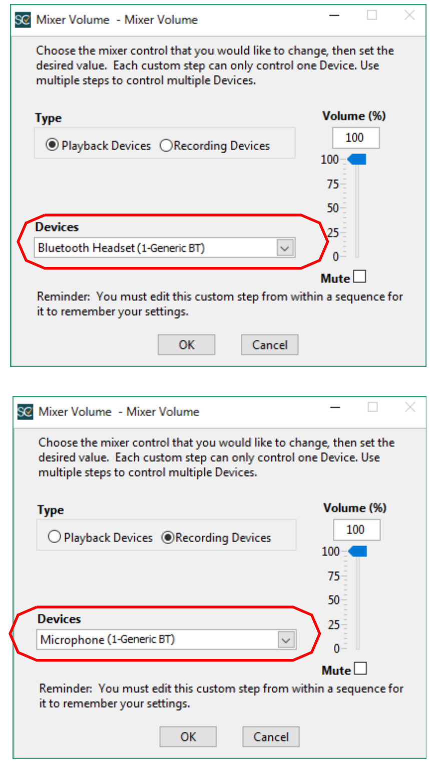
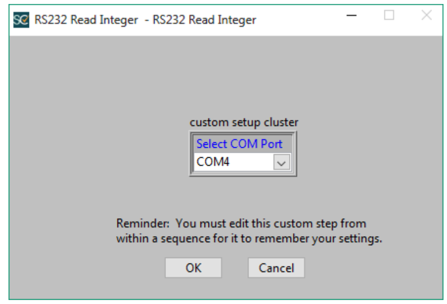
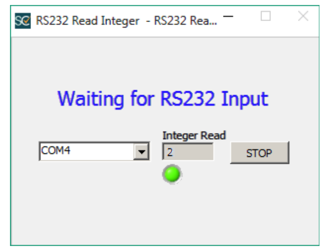


Figure: 26-8 Mixer Volume

RS232 Read Integer



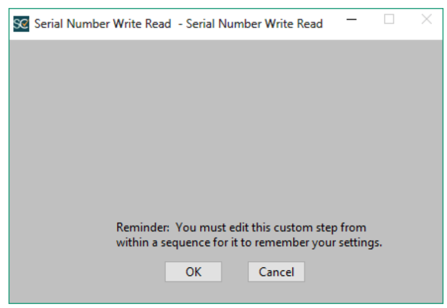
- Intended for use as a programming example only. Not for use “as is” in a sequence.
- Reads the integer value from an external device connected via RS232
- Generates a value in the memory list: Read Integer



Serial Number Write Read

- Intended for use as a programming example only. Not for use “as is” in a sequence.
- This step writes the value "1234" to the serial number field
- Next, it reads the serial number and writes it to the memory list in a value named "Serial Num"

It is useful as a template for creating your own custom vi's which read/write the serial number field of SoundCheck.

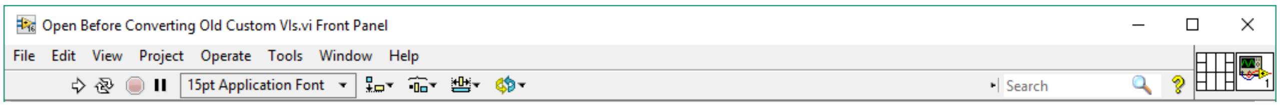


Open Before Converting Old Custom VIs

This serves as a tool to be used when updating your Custom VIs to the latest version of SoundCheck and LabVIEW.

- Easier to use Custom VI templates
- Updating Custom vis from previous SoundCheck versions is easier

Important! This must not be used in a sequence!



To convert Custom VI's from a previous version of SoundCheck:

1. Copy the Custom VI's into the Custom VI folder of this SoundCheck version.

   Example: C:\SoundCheck 18.1\System\Custom VIs\

2. Open the version of LabVIEW which is appropriate for the SoundCheck version and bitness you are upgrading to. In LabVIEW open the vi, "**Open Before Converting Old Custom VIs.vi**". (SoundCheck 18.1 requires LabVIEW 2018)

3. From the File menu of "Open Before Converting Old Custom VIs.vi", open the custom vi(s) you are converting.

4. They should automatically relink to the appropriate vi and ctl files. These are located in the file "**custom and SC Run Seq Sub VIs(xXX).llb**" found in the root of the SoundCheck folder.

5. Save the vi's, Start SoundCheck and verify the functions of your vi.

Issues you may encounter:

Some user Custom VI's may refer to "Global Data Stack Database Dynamic.vi". You should relink to "Global Data Stack Database.vi"

Some user Custom VI's may read the "Abort" global from previous versions of SoundCheck. Instead, you must use "Query Sequence Abort Flag.vi"

Figure: 26-9 Open Before Converting Old Custom VIs

When opening your vi you should see a solid arrow as in ***Figure 26-9***. This indicates that the vi opened correctly.

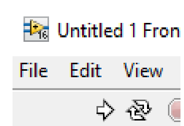


Figure: 26-9 Solid Arrow

If you see a broken arrow (***Figure 26-10***), debugging is required.

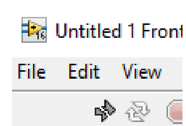


Figure: 26-10 Broke n Arrow

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Creating a Custom VI and Custom Step

Important! The contents of this chapter requires the use of LabVIEW. It is suggested that users have an advanced level of experience with the LabVIEW development environment in order to use these examples.

Creating a Custom VI for SoundCheck®

SoundCheck provides a way for you to integrate your own LabVIEW code into your SoundCheck sequence as a step. Included with SoundCheck are templates to create your own Custom Steps which can be run in a SoundCheck sequence. Once the files are created according the instructions, you can use those steps in the SoundCheck sequence editor.

Note: LabVIEW 2018 English Language version is required to create custom steps and step editors for SoundCheck 18.1. The "**Save for Previous Version**" option in LabVIEW is not recommended.

---

your step name here.vi

1. Make sure the SoundCheck application is closed.
2. Open your LabVIEW development system.
3. Open the LabVIEW VI template: C:\SoundCheck 18.1\System\Custom VIs**your step name here.vi**.

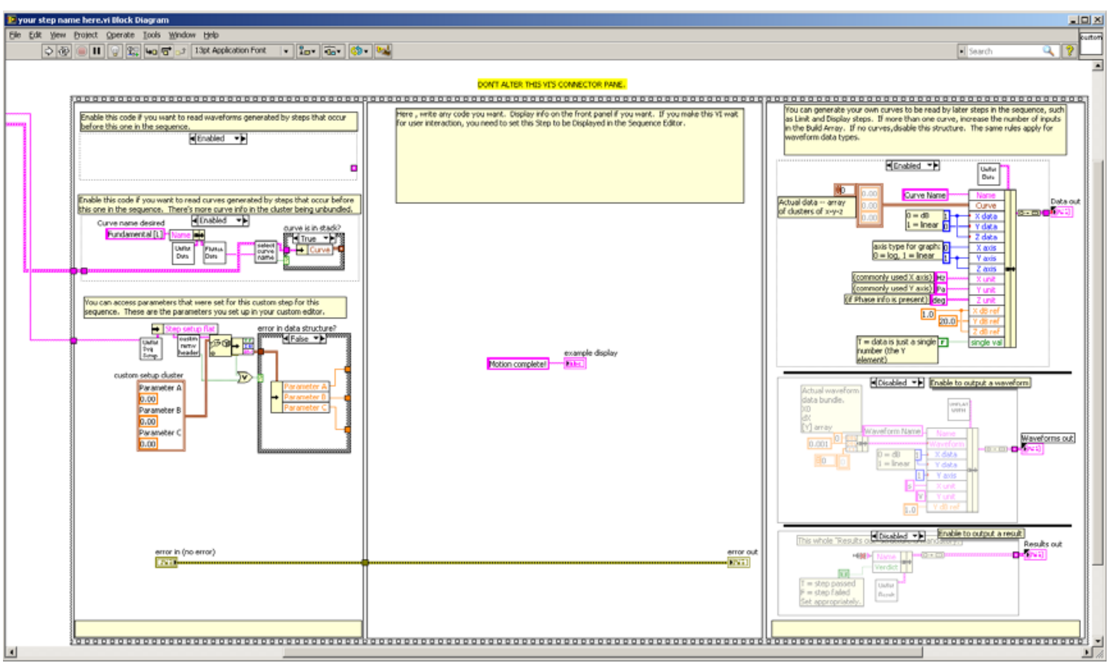


Figure: 27-1 Open VI Template

4. Before doing anything else, select "**Save As**" from the file menu and give the template a new name. For example: 'RS232 Read Integer.vi' (This example is installed with SoundCheck by default.) The new name cannot be the same as a pre-existing custom vi.

5. This must be saved in the "**Custom VIs**" folder. Make sure it has the "**.vi**" extension (lower case).

6. Make note of the VI name. The step name used must be the same as the VI name (minus the ".vi"). It is also CASE SPECIFIC.

7. Combine your code with the existing code in the SoundCheck template.

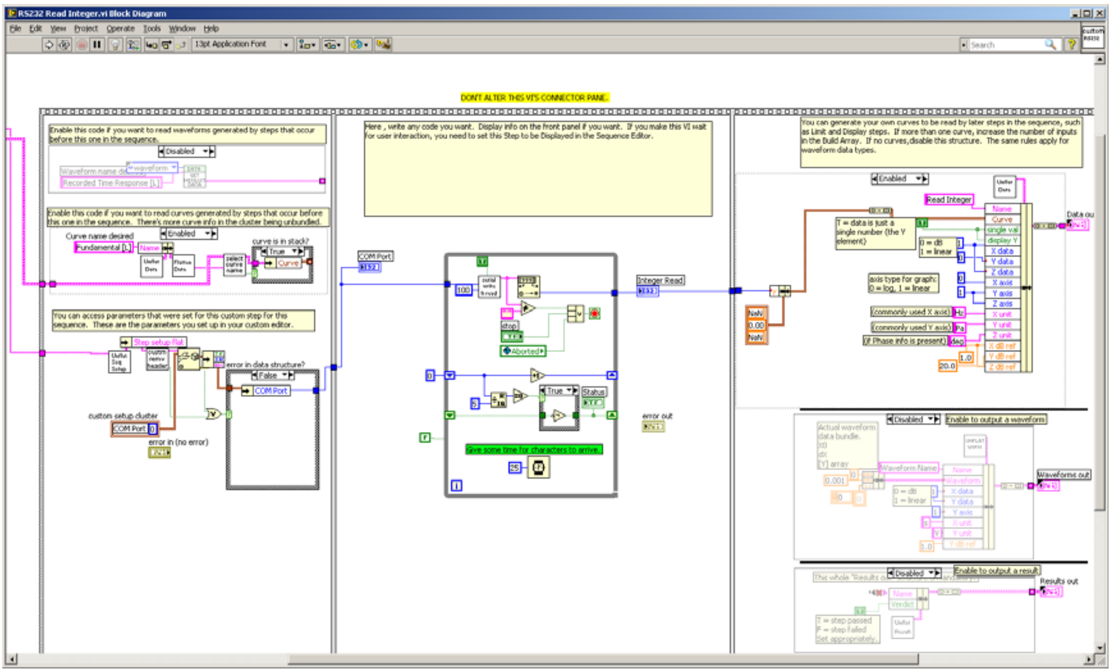


Figure: 27-2 Add LabVIEW Code To Template

8. Save the VI.

your step name here Editor.vi

9. Next, open the LabVIEW VI template "C:\SoundCheck 18.1\System\Custom VIs**your step name here Editor.vi**".

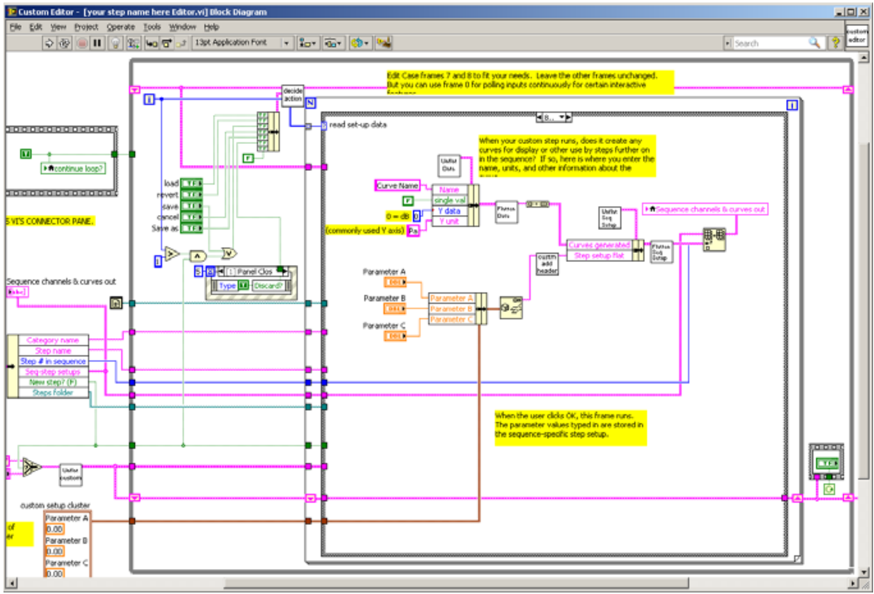


Figure: 27-3 Open VI Editor Template

10. Select "**Save As**" and use the same name from the previous step. Make sure it contains ' Editor.vi' after the custom name. For example: 'RS232 Read Integer Editor.vi' (This example is installed with SoundCheck by default.)

11. Follow the instructions highlighted in yellow on the VI diagram, and "wire in" all the custom code.

Rules

- The two VIs, **Step Name.vi** and **Step Name Editor.vi**, work together within SoundCheck and must be located in the **SoundCheck 18.1\System\Custom VIs folder**.

  In this example, 'RS232 Read Integer Editor.vi' is used to create the custom step, and 'RS232 Read Integer.vi' is called when the custom step is run in the sequence to execute the custom code.

- The curves listed in the 'Curves Generated' array in the custom editor must have exactly the same name as those created by the custom vi. This ensures that the placeholder for the curve in the Memory List is filled with the correct data when the sequence is run.

- If the VI has any subVIs, put them in the folder: "...\Custom VIs\subVIs\". If the folder does not exist, it must be created and must be located in the "SoundCheck 18.1\System\Custom Vis" folder.

- If you make your custom VI wait for user interaction, such as clicking a **Done** button, the Custom Step must be configured to "**Display step when run**" as shown in ***Figure 27-4***. Otherwise, SoundCheck may get stuck inside the custom VI, in an infinite loop. (See ***Sequence Editor on page 435***)

- If you call a sub vi in a custom vi, and that subvi has the VI Property “**Separate compiled code from source file**" turned on, SoundCheck will not be able to load the custom vi.

Creating a Custom Step

1. Start SoundCheck and open the *Sequence Editor*.
2. Select "**Custom**" from the left hand Step Category menu. By default, this list contains the example step 'RS232 Read Integer'.
3. Select "**New...**" from the **Step** menu and enter the name for the new step, using the same name as your step VI and step Editor VI.

Important! The step name should be exactly the custom VI name, minus the ".vi" (e.g., RS232 Read Integer).

In this example, the new step would be called *RS232 Read Integer* (do not include the .CUS extension in this dialog). This opens your *Custom Editor* of the same name (*RS232 Read Integer Editor.vi*).

4. A Custom Step is created and saved when you click **OK**. The new step can now be used in a sequence.

Note: The settings of the Custom Step must be made after inserting it into a sequence.

Using your Custom Step in a Sequence

1. Open SoundCheck and open the sequence editor.
2. Insert the step in a sequence and then open it from the right side of the editor ***Figure 27-4***.

   If you have used the **Curves Generated** option in your Editor, you will add names to the *Memory List* with your step. These curve, value, and result names will allow steps, such as Limits and Display Steps, that occur later in the sequence, to act on your custom curves. Open the Memory List to view the curves, values, and results generated by your sequence.
3. Edit and save the parameters for the step as you would any other step in SoundCheck, and save the sequence.

To edit the parameters of a Custom Step in a sequence, select "Custom" from the Setup menu, or use the shortcut **Ctrl+Shift+X**. If there is a Custom Step in the sequence, either method will open the editor for the Custom Step.

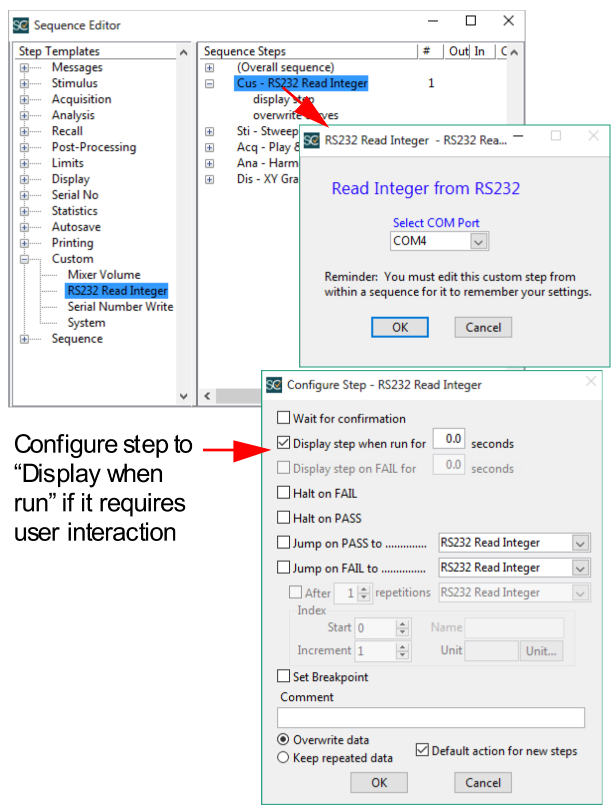


Figure: 27-4 Custom Step In Sequence

Sequence Editor

What is a sequence?

- A script to carry out a "sequence" of events
- A series of individual Steps
- The sequence (.SQC file) contains all of the steps and settings for those steps. It does not contain the settings for Hardware and Calibration. (See ***Hardware - System.Har on page 59 and System.Cal on page 79***)
- A sequence contains steps, test variables and Conditional Branching instructions. (See ***Configure Step on page 446***)

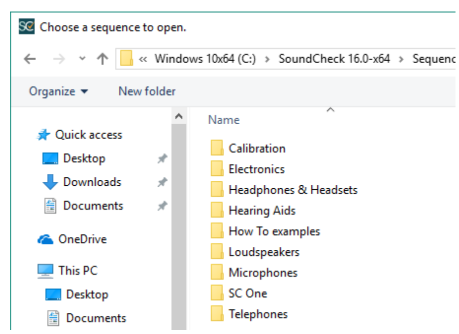
What is a step?

  - These are the building blocks of a sequence
  - Each step is from a specific category
  - Each step in the library of the sequence editor is a template for use in the active sequence on the right side of the editor
  - The file extension of each step file matches the step type, e.g., Stimulus Step = .STI, Acquisition Step = .ACQ
- After a step is inserted into the active sequence, it has no connection back to the step in the library.
- The step information for the active sequence is saved in the .SQC file (See ***Single-file Sequence Format on page 436***)

Default Sequences

SoundCheck comes with a library of sequences and step templates that can be used with minor or no modifications.

- These sequences serve as templates for making new sequences
- Step Templates are independent of sequences and are reusable



Please refer to ***Default Sequence List on page 615*** which includes descriptions of all the sequences included in the SoundCheck installation.

Note: Application specific sequences are also available on the Listen, Inc. website.

Features

- The Sequence Editor runs in parallel with the rest of SoundCheck (See ***Sequence Operation on page 438***)
  - Editor can remain open while running a sequence
  - Current running step is highlighted allowing you to see the progress of the sequence run
  - The SoundCheck Main Screen menu can be accessed
  - Editor can be moved to a second monitor
- The step templates pane now features an expandable tree structure for template categories and can be fully collapsed to maximize sequence space. See ***Step Template Library on page 441***.
- You can add single or multiple steps to a sequence simply by dragging and dropping. Steps can also be moved in a sequence with drag and drop. All other functions are accessed by Right-clicking on the step.
  - **Drag and drop** to insert and re-order single or multiple steps
  - Intuitive Right-click menu for all functions (See ***Right-click Functions on page 443***)
- **Sequence Debugging Tools** (See ***Debugging Tools on page 444***)
  - Steps with Pass/Fail (Conditional Branching) (Mes, Lim, Dis) are highlighted green or red in the editor after they are run
  - **Breakpoints** can be inserted into a sequence for debugging purposes. Control buttons on the SoundCheck Main Screen can then be used to advance the sequence one step at a time or run the remaining steps in the sequence. See ***Debugging Tools on page 444***.
    - Insert as many break points as desired
    - New buttons for run a single step or continue from breakpoint
    - Steps and sequence may be edited while at a breakpoint
  - '**Run from here**' option allows you to run only a portion of the sequence
- **Undo** functionality has also been added to the Sequence Editor both as a Right-click and the familiar Ctrl+Z shortcut. This allows you to quickly revert changes made when developing and editing sequences.

Single-file Sequence Format

As of SoundCheck 12:

- The sequence file (.SQC) contains all sequence parameters and steps. Individual Step files are no longer required.
- All attributes and fields of a step in the active sequence are linked to that sequence
  - Changes to the steps in the active sequence appear only in that sequence
  - Changes are not linked back to the step template in the Step Template Library of the Sequence Editor. See also ***Right-click Functions on page 443***.
- Selecting Save As when editing a step in the active sequence, saves the changes in the sequence and makes it a template in the library
- Sharing and updating sequences is greatly simplified. Only a single SQC file is required, rather than an entire folder full of step files. See ***Exporting Sequences on page 451***.

Converting Sequences

As of SoundCheck 12, sequences are no longer imported. The Setup Wizard allows you to convert sequences from a previous version to the Single File Sequence format. Master sequences and sub-sequences from versions prior to SoundCheck 12 should be handled as follows:

1. Export the sequence from SoundCheck 11 (or previous) into its own folder
2. In SoundCheck 18.1, open each sub-sequence (from the exported folder) and click save. This will convert the sub-sequences to the single file format.
3. Open the master sequence and click save so that it is in single file format
4. See ***Exporting Sequences on page 451***

Sequence Editor Interface

The *Sequence Editor* (optional module 2002) allows you to create and customize sequences to fit your specific testing needs. Sequences can involve a few steps for a straightforward measurement (e.g., loudspeaker frequency response), or include dozens of steps for elaborate tests. To view and change the current sequence, select **Sequence** from the **Setup** drop-down list on the main SoundCheck® menu bar, or use the keyboard shortcut **Ctrl+Q**.

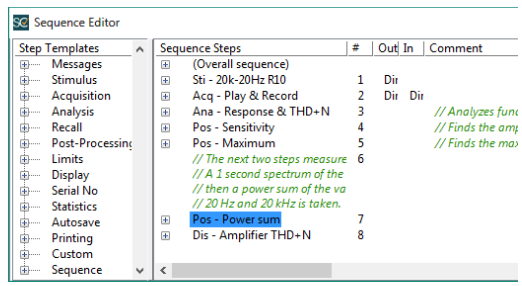


Figure 28-1: Sequence Editor

The right side of the Sequence Editor shows the Active Sequence being edited. Here you can expand or collapse step details.

The overall sequence configurations are listed first (Overall sequence) and take precedence over any conflicting step configurations.

Any individual step configurations are displayed immediately after the step. You can expand the configuration info by clicking the **+** button next to the step.

The left side shows the Step Template Library which gives you a variety of preset steps normally used in SoundCheck. You can also create your own step templates.

The editor window allows the right side to be expanded for easier viewing of long step names.

Note: Settings for the window and column size of the sequence editor window are stored in the **SoundCheck *18*.ini** file under [WinColBounds]. This will be recalled the next time SoundCheck is run.

Relative File Paths

File path controls in a sequence step can be set relative to the folder path of the sequence itself. This is useful when sharing sequences with other SoundCheck users and locations, as it makes it easy to move the sequence and all of its related files (recalled DAT files, WAV file stimuli, etc.) from one place to another. The relative path can even include sub-folders. See ***Rules - Relative File Path Rules in Recall Editor on page 228***.

File Menu

The **File** menu on the SoundCheck Main Screen has the following sequence related functions:

- **New** - Create new sequences
- **Save** current sequence being edited
- **Save As** - Save current sequence with a new name and optional change location folder
- **Revert** allows you to discard all changes made since the last time the sequence was saved to disk.
- **Delete** - Deletes the selected sequence from the sequence folder. This does not go to the Recycle Bin. Deleting is permanent.

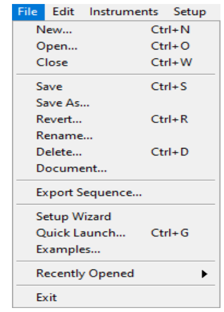


Figure 28-2: Sequence Menu

Document Sequence - Allows you to export a list of the steps of the active sequence along with information regarding the configuration of the steps and sequence. See ***Document Sequence on page 452***.

Sequence Operation

The Sequence Editor can remain open while a sequence is running. This allows you to see the progress of the sequence as the steps run.

- As a step is running it is highlighted in Yellow. See Display Step in ***Figure 28-3***.
- Steps configured with Pass/Fail Conditional Branching functions are highlighted in Green or Red after they have run. (.MES, .LIM and .DIS)
- Breakpoints can be set to pause the sequence run for debugging. Steps are marked with a red dot. See ***Debugging Tools on page 444***.

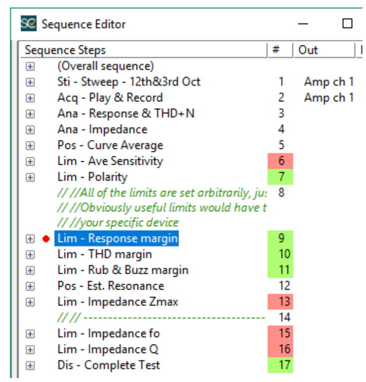


Figure 28-3: Step Highlighting

- The SoundCheck Main Screen menu can be accessed while the editor is open. The Start, Stop, Step and Continue buttons are located in the top left corner.

Note: **As of SoundCheck 13, the Status window has been removed. Use the Enter key or the Continue button as shown in *Figure 28-4*.**

See ***SoundCheck Main Screen on page 41*** for more information on Sequence Control Buttons. Also see ***Keyboard Shortcuts on page 591***.

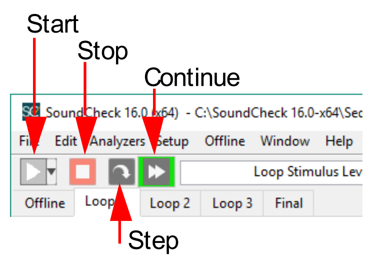


Figure 28-4: Sequence Control

- The editor can be moved to a second monitor to leave more room for display windows.

Editing Sequences

Every sequence is comprised of several steps. The *Sequence Editor* allows you to configure the sequence as a whole, as well as to configure individual steps.

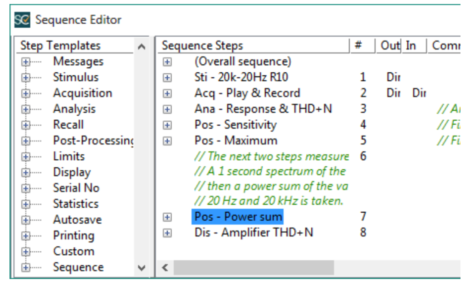


Figure 28-5: Sequence Editor

The active sequence name is shown on the SoundCheck Main Screen as shown in ***Figure 28-6***.

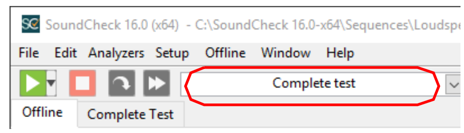


Figure 28-6: Active Sequence

All the steps are listed in the order that they are executed. Steps can also be configured with one or more Conditional Branches (or jumps) to change the order of execution. See ***Configure Step on page 446***.

- You can change the order of these steps by clicking on a step and dragging it to a new location in the sequence. SoundCheck will prompt you if you are moving steps that are part of a jump.
- Double click on a step to open its editor.
- As the sequence runs, each step is highlighted in Yellow as it is processing.
- Steps marked with a Red Dot are set as Breakpoints. See ***Debugging Tools on page 444***.

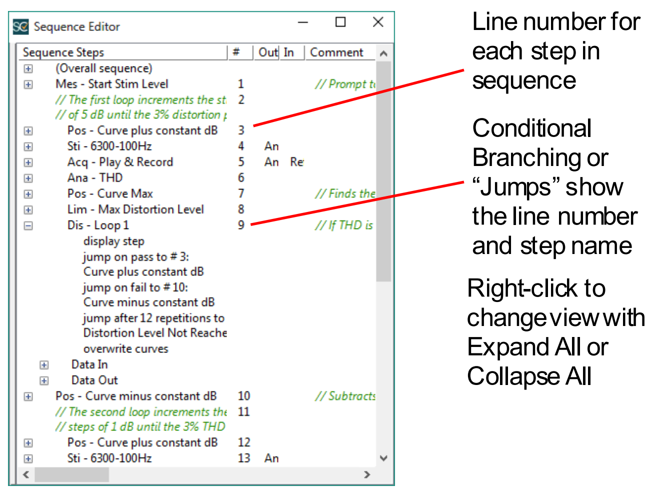


Figure 28-7: Sequence Editor

- You cannot edit sub-sequences from the master sequence. Instead, open the sub-sequence itself in the *Sequence Editor* to edit it.

Insert/Remove Steps

You can also remove or insert steps into the current sequence.

Right-click a step (or select several steps), Right-click them and select **Remove**

To insert a Step Template:

1. The default Step Template location is the Steps folder of your current SoundCheck installation.
2. Choose the desired Category from the drop-down list, Left-click on a step and drag it into the Active Sequence.
3. You can then “Drag and Drop” the new step in the correct position in the sequence.

Note: You can only browse to the *Steps* directory specified in Main Screen > Edit > Preferences > Folder Paths. See ***Preferences on page 45***.

After the step is added to the active sequence it is no longer connected to the Step Template Library. Changes to the step in the sequence appear only in that sequence.

Note: In the Step Template Library you can select **Sequence** to show the steps of a saved sequence. This is an easy short-cut for re-using steps from another sequence.

Adding Multiple Steps

It is possible to “Add a Block“ of steps to a sequence. This is very useful when adding steps from another saved sequence.

Figure 28-8 shows an example.

- In the Step Template Library select **Sequence** and pick a sequence from the list
- Select a group of steps from the sequence in the library. Left-click and drag them to the Active Sequence
- They will be added above the cursor line that appears when you “mouse over” the Active Sequence. In this example, the steps were inserted above the Acquisition Step.

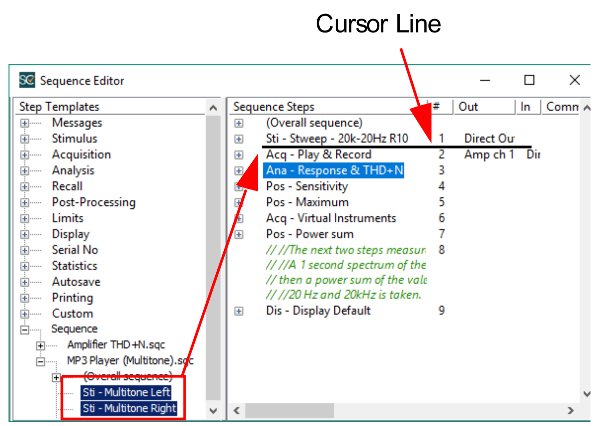


Figure 28-8: Add Multiple Steps

Editing Steps

You can edit a step in the sequence by simply double-clicking on the step in the **Active Sequence** list box.

You can also click **Setup** on the SoundCheck Main Screen, select a Category and select a step to edit.

Step Template Library

The Step Template Library contains preset steps for use in building sequences. The steps are arranged by Category as shown in ***Figure 28-9***.

- Double clicking a step name in the Step Template Library will open that step so you can review its properties or edit its default settings.
- To save changes click **Save As** in the editor.
- You can give the step a new name or leave the name the same and overwrite the existing step.

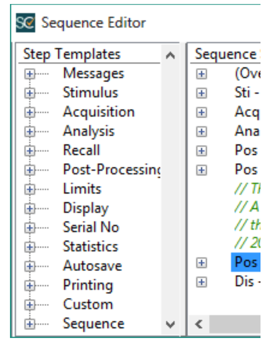


Figure 28-9: Step Template Library

Categories

- **Messages** - Specifies text based, digital I/O based messages, or RS232 and/or IEEE messages.
- **Stimulus** - Specifies the test signal (sine-based, WAV file)
- **Acquisition** - Selects method of data acquisition, e.g., Play and Record, Record only, Real Time Analyzer, etc.
- **Analysis** - Selects the analysis algorithm
- **Recall** - Recalls a saved data file (SoundCheck-specific *.DAT, *.RES, *.WFM and standard *.WAV files) into the *Memory List*
- **Post-Processing** - Allows for complex data operations
- **Limits** - Checks measurement data against preset tolerances
- **Display** - Selects how the data and results will be presented
- **Serial No** - Selects automatic incrementing of a serial number, or a prompt for the user to enter a Serial Number
- **Statistics** - Calculates running statistics of curves and results that are updated with each test run

Abbreviation	Associated Editor
Mes	Messages
Sti	Stimulus
Acq	Acquisition
Ana	Analysis
Rec	Recall
Pos	Post-Processing
Lim	Limits
Dis	Display
Ser	Serial Number
Sta	Statistics
Aut	Autosave
Pri	Printing
Cus	Custom
Seq	Sub-sequence

Figure 28-10: Category Abbreviations

- **Autosave** - Saves data, results and/or wave forms to disk in one of five formats
- **Printing** - Selects how the data and results will be printed
- **Custom** - Allows you to integrate your own LabVIEW code into the sequence as a step. See ***Creating a Custom VI and Custom Step on page 431***.
- **Sequence** - Allows you to copy a step from a pre-existing sequence into the selected sequence. See ***Adding Multiple Steps on page 440***.
  - ***This step will retain the settings it had in the saved sequence in the library.***
  - You can also insert a sequence file, making it a sub-sequence. You can only select sequences that are in the same folder as the current Active Sequence. See ***Insert Sub-sequence on page 443***.

Step Template Menu

Right-click a Step Template category and select:

- **New** - Create new steps in the selected category
- **Import Step Templates** allows you to bring steps in from previous versions of SoundCheck or other SoundCheck folders. The entire folder of steps must be selected. If you want specific steps it is recommended that you make a copy of the original folder and remove the unwanted files before importing the folder.

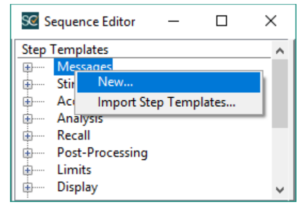


Figure 28-11: Step Template Menu

Rules - Sub-sequences

- You can insert an entire sequence into another sequence. This makes it a sub-sequence. The sub-sequence must be in the same folder location as the master sequence.
- Batch Processing and Memory List Grouping is not allowed in Sub-sequences. If you want to use a sequence which contains batch processing, as a sub-sequence, it will need to be re-configured to use individual steps that do not require Memory List Groups and Batch Processing.
- **Reminder:** Rules for Overwrite Data/Keep Repeated Data in Step Configuration applies to data in sub-sequences. Either use unique names for Data and Waveforms or, select “Add Input Data Name” or “Use Signal Path Name”. Refer to ***Overwrite vs Keep repeated data on page 447*** for more information.

  Example: When all Acquisition Steps use the name “Recorded Time Waveform” AND the sub-sequence acquisitions are configured to “Keep Repeated Data”, the resulting names are:

  Recorded Time Waveform, 2- Recorded Time Waveform, 3-Recorded Time Waveform, etc.

  The Analysis Steps then point at the first waveform generated, “Recorded Time Waveform”, instead of the waveform that analysis is paired with in the Sub-sequence.

  The best way to get around this is to use unique names for the acquisition waveforms in each sub-sequence.
- When using the Memory List Sorting and Grouping function, “Autogroup By Category”, in the Master Sequence, the Curves, Values, Results and Waveforms from the Sub-sequence are automatically grouped under the heading “**Seq**” for each Memory List Tab.
- The sub-sequence when inserted into the master sequence is given a step number in relation to the other steps of the master sequence
- Steps of sub-sequences are numbered separately, relative to the first step of the sub-sequence. Other steps of the Master Sequence resume their numbering after the sub-sequence.
- Steps of a Sub-sequence are indented in relation to steps in the Master Sequence.
- You cannot edit sub-sequences from the master sequence. Instead, open the sub-sequence itself in the *Sequence Editor* to edit the steps.

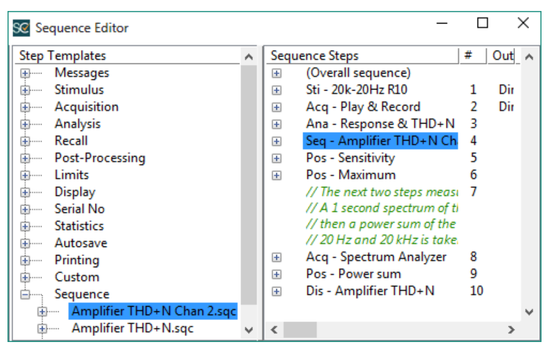


Figure 28-12: Sequence Step Numbering

- When exporting the Master Sequence, its sub-sequences are exported as well. This will also export the Associated Files of the Sub-sequences. See ***Exporting Master Sequence and its Sub-sequences on page 451***.
- See ***Converting Sequences on page 437*** for information on converting Master and Sub-sequences from previous versions of SoundCheck

Insert Sub-sequence

- In the active sequence, select the step that you want the Sub-sequence to occur before
- Right-click the step and select **Insert Step**
- Select **Category** and **Sequence**
- Choose a sequence
- The sub-sequence must be in the same folder location as the master sequence.
- Sub-sequences show up in the active sequence as a single step. The step category is identified as “Seq“.

In this example, the Display step was selected and the sub-sequence was inserted before it. See ***Figure 28-13***.

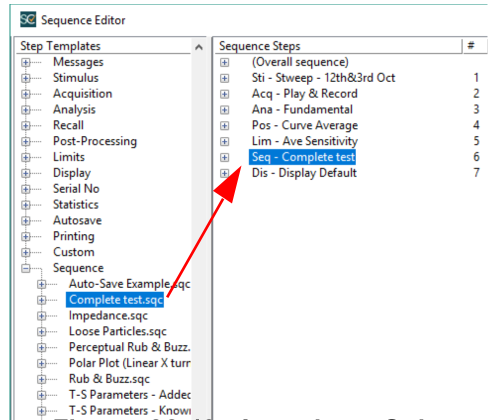


Figure 28-13: Inserting a Sub-sequence

Right-click Functions

Right-clicking on a step or overall sequence opens a selection menu. This makes it easier to change the configuration of any step or the overall sequence.

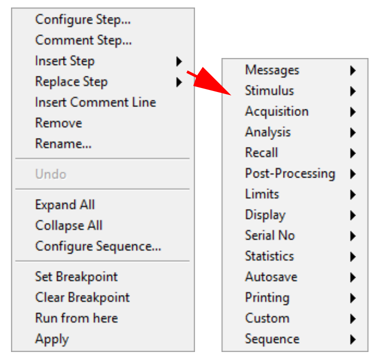


Figure 28-14: Right-click

Steps

- **Configure Step** - Open step configuration window to set options for how the step works in the sequence. See ***Configure Step on page 446***.
- **Comment Step** - Comments appear in the Comment column of the sequence editor as well as the sequence Documentation
- **Insert Step/Replace Step** - Allows you to select a step from the Step Library. See ***Figure 28-8***.
- **Remove** - Removes the selected step(s) from the sequence. Select multiple steps by holding down the CTRL key while selecting Steps.
- **Rename** - Change the name of the selected step
- **Undo [Name of Last Edit]** - Revert last change(s) made. This remembers the order that changes are made to the sequence and allows you to step back through multiple layers of Undo. You can also use Ctrl+Z as long as the Sequence Editor window is Active (blue title bar).

Sequence

- **Expand All / Collapse All** - Used to Show or Hide the configuration of all steps in the sequence
- **Configure Sequence** - Opens the sequence configuration window

Debugging Tools

These functions are typically used to test and fix sequence operation.

- **Set Breakpoint** - Select a step for the sequence to stop on. This step is denoted by a **Red Dot**. You can set as many breakpoints in a sequence as required.
- **Clear Breakpoint** - Removes the Breakpoint on the selected step(s).
- **Run from here** - Run the sequence from the selected point. This allows you to run to the end of the sequence or to the next Breakpoint. After the sequence stops at a Breakpoint, you can select "R**un from here**" again to continue sequence run.

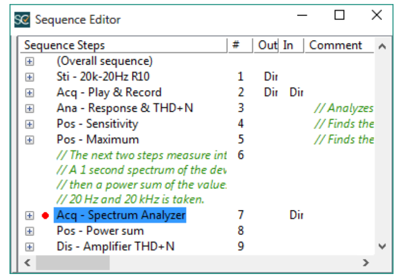
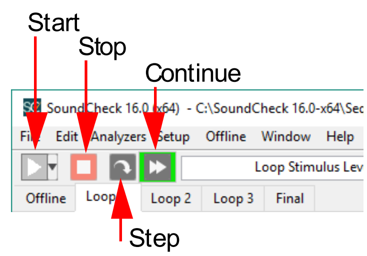


Figure 28-15: Set Breakpoint

- **Apply** - Allows you to Apply the action of selected step without having to open the step. For .MES, .LIM and .DIS steps this will show the Pass/Fail state of the step by highlighting the step number in Green or Red. (Clicking Apply in the step editor will not change the step highlighting.)

When a sequence run pauses at a breakpoint, the Step and Continue buttons become available to either advance one step at a time or run the rest of the sequence.



While the sequence is paused at a breakpoint, you can edit the steps in the sequence. This allows you to trouble shoot a sequence one section at a time, without having to run the entire sequence.

Breakpoints can also be set in Step Configuration. See ***Configure Step on page 446***.

Configure Sequence

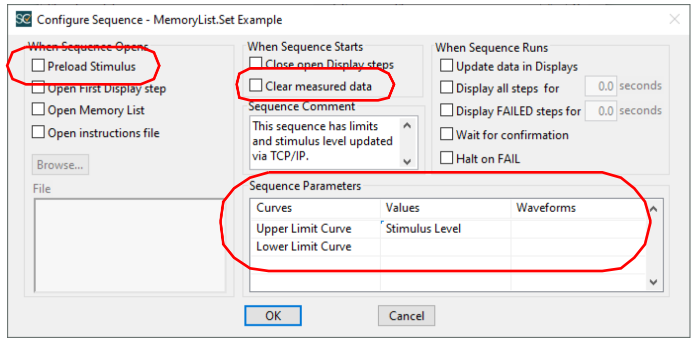


Figure 28-16: Configure Sequence

Double click on the first line of the sequence, **Overall sequence**, at the top of the active sequence or Right-click the Sequence Editor to open the **Configure Sequence** window.

Sequence Parameters

As of SoundCheck 18, you can create empty Memory List items that will be populated with data using the "**MemoryList.Set**" command. The example in ***Figure 28-16*** shows how Curves and Values are created in the **Sequence Parameters** section.

These empty items show up in the **Memory List** as shown in ***Figure 28-17*** and will be populated with data when the **MemoryList.Set** command is used. See ***Controlling SoundCheck with TCP/IP on page 491*** and ***MemoryList.Set TCP/IP Command on page 493*** for more information.

- **Preload Stimulus** and **Clear measured data** must be unchecked in order to use the **MemoryList.Set** command.

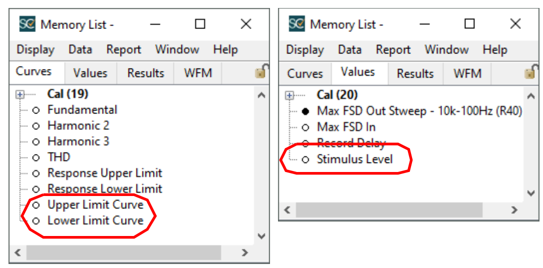
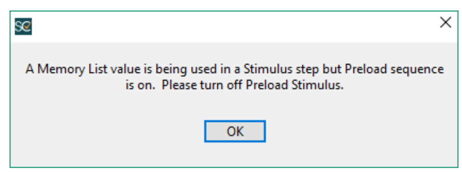


Figure 28-17: MemoryList.Set Items

When Sequence opens

- **Preload Stimulus** - Creates and loads the Stimulus Waveform into memory when the sequence opens.

  <u>**Preload Stimulus vs Memory List Selection**</u>
  When "**Memory List Selection**" is selected in the Stimulus Editor, a message will pop up as a reminder to "shut off" Preload Stimulus.



- **Open First Display Step** when sequence opens
- **Open Memory List** when sequence opens
- **Open Instruction File** - PDF or other document types can be tied to a sequence so that the file opens when the sequence opens. Example sequences included with SoundCheck use this function to open sequence notes and instructions.
  - **Browse to Location** - Select the file to link to sequence
  - **File** - Shows the linked File Name and path

When Sequence Starts

- **Close Open Display Step** - Only the SoundCheck Main Screen is visible after sequence start
- **Clear Measured Data** - Used to clear displays of curves and values. Helps prevent confusion during sequence run.
- **Sequence Comments** - This field can be used to include notes about the sequence; version, date created, author, etc.

When Sequence Runs

- **Update Data in Displays** - Data is shows in the display in the order it is acquired, analyzed or processed rather than waiting until the sequence run completes.
- **Display all steps for ___ Time** - Useful for debugging. Allows you to see the operation of each step for a preset time.
- **Display FAILED Steps for ___ Time** - Same as above except that it only applies to steps that issue a FAILED verdict.
- **Wait for confirmation** - Waits for you to click Continue (Enter) or Stop (Esc) on the SoundCheck Main Screen.
- **Halt on Fail** - Stops the sequence run when the verdict of any step is FAIL.

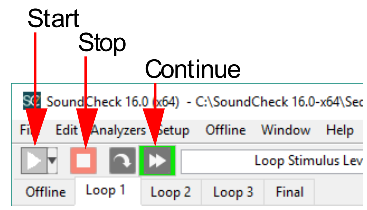


Figure 28-18: Stop or Continue Buttons

Configure Step

You can configure individual steps to fine tune their role in the sequence.

Configuring individual steps allows implementation of loops and **Conditional Branching** in the sequence.

Conditional Branching Example

Right-click a step and select **Configure Step**.

The example in ***Figure 28-19*** shows that if the result of the "Imp Test for Signal" step is **Pass**, it is configured to "Jump" to step 6, "Est. Resonance".

If the result is **Fail**, the sequence continues to the "No Signal" step and the sequence stops since this step is set to Halt on Fail.

Using these options, you can alter the operation and order of step execution in the sequence based on the outcome of a particular step. See ***Conditional Branching Rules - Sequence Editor on page 448*** for more information.

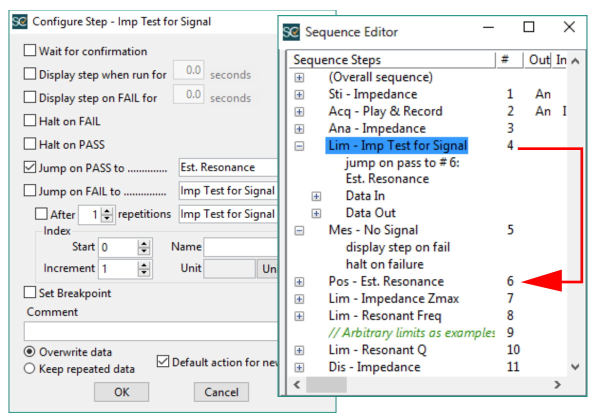


Figure 28-19: Conditional Branching

Options

- **Wait for Confirmation** - The sequence will pause after the selected step is completed and wait for user input before continuing. Click the **Continue** or **Stop** buttons on the SoundCheck Main Screen as shown in ***Figure 28-20***.

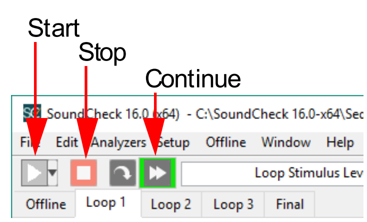


Figure 28-20: Stop or Continue Buttons

- **Display step when run for / Display Step on FAIL for** - Eliminates the need for user confirmation. Enter the amount of time you wish the sequence to pause while the step is displayed. The sequence will resume once the step times out.
- **Halt on FAIL / Halt on Pass -** Allows the sequence to be stopped based on the result of the step (PASS or FAIL).
- **Jump on PASS to / Jump on FAIL to** - Allows for **Conditional Branching**. Depending on the PASS/FAIL result of the step, the sequence will jump to a different point in the sequence. The drop-down lists contain the names of all the steps in the sequence; you can jump forward or backward in the sequence.

You can also elect to loop through part of the sequence for a given number of iterations and jump to a selected step. See ***Conditional Branching Rules - Sequence Editor on page 448*** for more information.

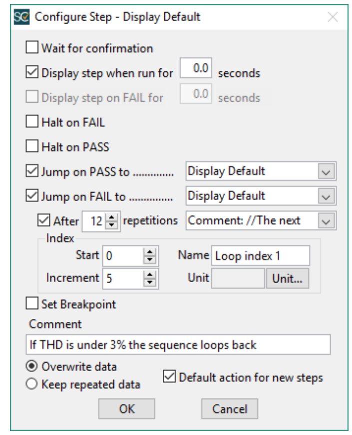


Figure 28-21: Configure Step - Loop Stimulus Level

- **After "n" repetitions** - Sequence operation jumps to the selected step for a set number of runs. This increments the Index value as noted below. The Index value appears in the Memory List and can be used to control stimulus level, set turntable angle for polar plots and other uses. (Requires that "Jump on PASS or FAIL" is selected.)

Index (Loop Index)

You can create a Memory List value that increments according to the following settings:

- Start - Starting value of Memory List item
- Name - Name of Index value created in Memory List
- Increment - Amount the item will increase or decrease after each repetition
- Unit - Log or Linear units can be used. Uses standard editor.
- See **Loop Stimulus Level** or **Polar Plot (Linear X turntable)** example sequences and sequence notes for more information.
- The Index is a "Y axis" value. When using the Index value to update Start and Stop Frequencies in the Stimulus Editor, select Y axis. See ***Start and Stop Frequencies from Memory List Values on page 115*** for info on how this is used in a Stimulus Step.

- **Set Breakpoint** - Allows you to set a Breakpoint to halt sequence operation. Useful for debugging. Looping sequences. The step will have a red dot next to the step name in the Active Sequence to indicate the Breakpoint. See ***Debugging Tools on page 444***.
- **Comment Step** - Comments appear in the Comment Column of the sequence editor as well as the sequence Documentation.

Overwrite vs Keep repeated data

- **Overwrite data** - This is selected so that the *Memory List* data generated by the step will be overwritten in the each time a sequence loop occurs. This helps reduce the number of curves that appear in the Memory List. This can be set on a per-step basis. In other words, you can overwrite data for some curves in a loop and keep all repeated data for other curves. **This is the default setting.**
- **Keep repeated data** - Allows you to keep the repeated measured curves in memory. They will be added to the Memory List with an iteration prefix, e.g., 2-Fundamental, 3-Fundamental, etc. Curves are only kept when the iteration logic produces a PASS.
- **Default Action for new steps** - Any **New Step** added to the sequence, after this point in time, will use the "Overwrite data/Keep repeated data" setting of this step. This applies to any new step added, independent of its location in the sequence.

Conditional Branching Rules - Sequence Editor

- Stimulus, Acquisition and Analysis steps should be configured to “Overwrite Data” when these steps are within a Loop

  The default state of steps when added to a sequence is “**Overwrite Data**”. When steps in a loop are configured to “Keep Repeated Data”, redundant curves can be generated in the Memory List, e.g., Fundamental [L], 2-Fundamental [L], 3-Fundamental [L], etc. In most cases, steps should be set to Overwrite Data so that only one instance of the curves is created. They will be updated each time the sequence runs.

- The final step of a sequence should not be used in a Loop

  Jump on Pass or Fail may create an endless loop

  In order to execute a jump at the end of a sequence, add a dummy display or message step at the end of the sequence, e.g., Message Step named “End of Sequence“. Then configure the previous step to jump to the desired location.

- Step Configuration in the Sequence Editor shows the Line number and Step name for the target of a “Jump“

  Refer to ***Figure 28-22*** for an example

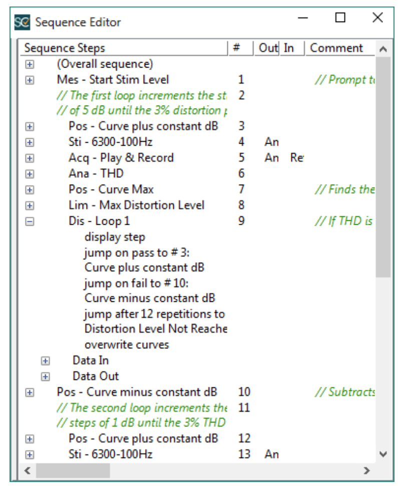


Figure 28-22: Step Configuration Info

For an example of how Conditional Branching and Loops work, open the **Loop Stimulus Level** sequence from the SoundCheck Sequences - How To Examples folder.

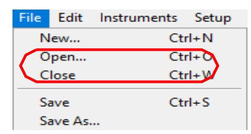
Creating a New Sequence

This is the basic procedure required to complete a sequence. You can alter it to fit your test needs.

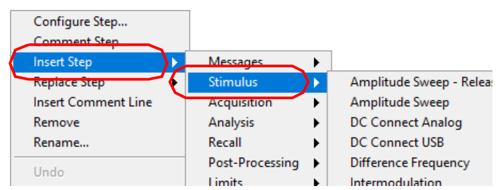
Note: The order of the Step Categories in the Template Library is also the order that the steps are normally used in a sequence: Stimulus, Acquisition, Analysis, etc.

Note: Steps can be added to the sequence from the Step Template Library on the left side of the editor. The step can be opened and edited from the Active Sequence on the right hand side of the editor. Changes to the step are stored in the sequence and are not saved back to the Step Template Library.

Note: After opening any of these sequence steps in the active sequence, they can be saved under a different name by clicking on **Save As** in the Step Editor. You can also Right-click a step and select **Rename**. This change is not saved in the library.



1. On the SoundCheck Main Screen select **File > New**. This opens the Sequence Editor with a blank Active Sequence. Click **File** again and select **Save** to name the sequence. You can also browse to save the sequence in a different folder.
2. The sequence will use the system **Hardware** and **Calibration** configurations.
3. Right-click in the blank Active Sequence and select **Insert Step**.
   - Select the Stimulus Category and select the Stimulus Step to be used for the test.
   - You are prompted to select the Output Signal Path.



Note: ***You can also drag and drop steps from the Step Template Library into the Active Sequence.***

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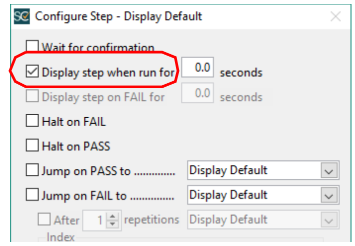
4. Right-click in the blank Active Sequence and select **Insert Step** again.
   - Select **Acquisition** and select the method for playing the stimulus signal and recording the DUT (device under test) response and drag it into the Active Sequence.
   - You are prompted to select the Input and Output Signal Path.
5. Repeat the process of Right-clicking on the Active Sequence to insert steps. You can always change the order of the steps by clicking on them, using the “Drag and Drop” method.
6. Once a step is inserted in the sequence you can Right-click it and select **Rename** to give it a unique name to identify its function in the sequence.
7. Insert an **Analysis** step.
   - Select the type of signal analysis that will be performed.

8. Insert a **Display** step.

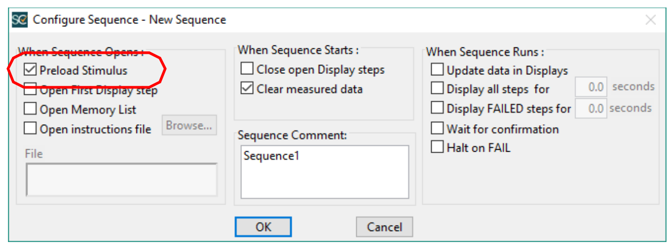


   - Select a step to display your data
   - Add display windows as described in ***Creating a Display Step on page 350*** and ***Display Examples on page 352***
   - Right-click the Display Step and select **Configure step**. See ***Configure Step on page 446*** for more information
   - Select **Display step when run**. This is required in order for the display to be shown when the sequence is run.

Note: ***When using multiple Display Steps, each Display Step in the sequence must have a unique name or data will not be displayed correctly.***

---

9. At the top of the Sequence Editor double click on “Overall sequence”. This opens the Configure Sequence window. You can also Right-click the sequence and select **Configure Sequence**. See ***Configure Sequence on page 445*** for more information.



   - Select **Preload Stimulus** to optimize test speed

10. Insert other types of steps as noted above.

Note: ***Remember that you can Drag and Drop steps to change the order that they occur in the sequence.***

---

11. Double click to open each step in the Active Sequence (right hand side of the editor) to fine tune the operation of each step.
12. Right-click a step to Configure the Step. See ***Configure Step on page 446*** for more information.
13. Now you can save and run your sequence.

Exporting Sequences

Sequences developed on one SoundCheck 18.1 installation can be used by other SoundCheck 18.1 (and later) systems using the **Export Seq** command from the **File** menu. This will copy the saved sequence to a selected folder.

Associated Files

In addition to the .SQC file, it also exports the following:

- Any *.DAT, *.RES or *.WFM files from Recall Steps set to ***"Specify File Path"***
- Picture files being used in the *Display Steps*
- WAV files used in the Acquisition or Stimulus Steps
- Calibration .DAT files associated with the signal paths used in the sequence
- Any instruction files from the Sequence Configuration

The sequence can be run from the exported folder. This folder can be on a network or on the local PC.

Exporting Master Sequence and its Sub-sequences

When exporting the Master Sequence, its sub-sequences are exported as well. This will also export the Associated Files of the Sub-sequences.

Example - Exporting Sequences

To export sequences, do the following:

1. Make sure the sequences you want to export have been saved.
2. Select **File→Export Seq…**.
3. Navigate to the sequence file(s) location, select the desired .SQC file(s) and click **OK**
4. Open the folder where the exported sequences will reside. This can be a folder on the local hard drive or anywhere on a network.
5. Once you have opened the destination folder, click on **Select Cur Dir**
6. SoundCheck will then export the sequence files as noted above

Document Sequence

From the SoundCheck Main Screen click **File > Document Sequence**.

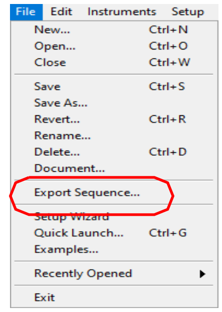


Figure 28-23: Document Sequence

Allows you to export a list of the steps of the active sequence along with information regarding the configuration of the steps and sequence.

- Export the list of steps of a sequence to a text file in 2 text formats or to an Excel file: (See ***Figure 28-24***)
    1. Delimited txt file; tab, comma, semicolon and other
    2. Space-aligned for fixed width text editors
- The Documentation setup window allows you to select:
    1. Summary: Shows the Simple step information first. For sequences with sub-sequences, only the master sequence setup is shown.
    2. Current View: same layout as on screen
    3. Expanded Details: Steps names, configuration and settings:

        a. Line number

        b. Output Signal Path / Input Signal Path

        c. Category

        d. Step title

        e. Step configuration
- The Comment window gives you a space to enter general notes about the sequence and it's use. This info shows up in the Quick-Start menu.

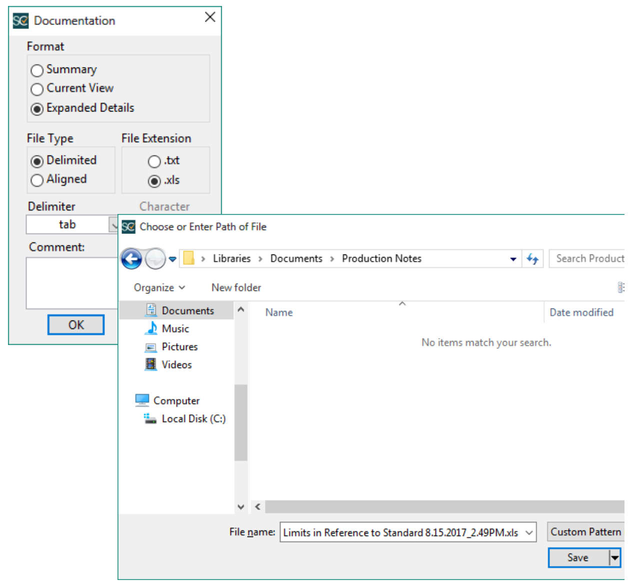


Figure 28-24: Documentation Editor

An example of the output from the Documentation Editor is shown in ***Figure 28-25***.

Documentation Spreadsheet

	SoundCheck	www.listeninc.com			
	Sequence:	Limits in Reference to Standard			
	Fri Mar 23 2018	11:48 AM			
Type	Step Name	#	Out	In	
	(Overall sequence)				
	preload Stimulus				
	display instructions				
	clear data at start				
	close display at start				
Mes	Measure Standard		1		// Measure standard or use stored data
	display step				
	jump on pass to # 2: Fixture Standard				
	jump on fail to # 8: Reference Standard				
	overwrite curves				
Mes	Fixture Standard		2		
	display step				
	wait for confirmation				
	overwrite curves				
Sti	10k-100 Hz (R40)		3Amp ch 1		
	Data Out				
		Max FSD Out 10k-100 Hz (R40)			
		10k-100 Hz (R40)			
Acq	Play & Record		4Amp ch 1	Reference Mic	
	Data In				
		10k-100 Hz (R40)			
	Data Out				
		Max FSD In			
		Recorded Time Waveform			

Sequence Title

Date & time of documentation

Comment Lines are denoted by a double slash (//) starting in the first column.

Figure 28-25: Documentation in Excel spreadsheet

The documentation example shows the result of selecting the “Expanded View” format. If “Summary“ were selected, the step configuration information below each step would be omitted.

Sub-sequence Notation

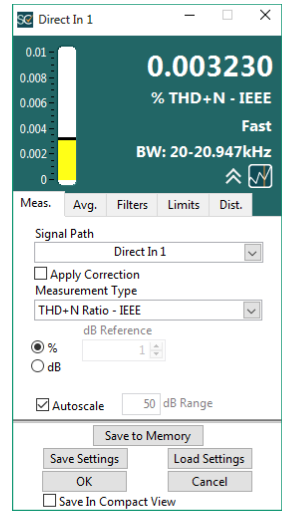
- The sub-sequence title is given a step number in relation to the other steps of the master sequence
- Steps of sub-sequences are numbered separately, relative to the first step of the sub-sequence. Other steps of the Master Sequence resume their number after the sub-sequence.

page intentionally left blank

Virtual Instruments

Features

Distortion Analyzer with THD and Optimized THD+N

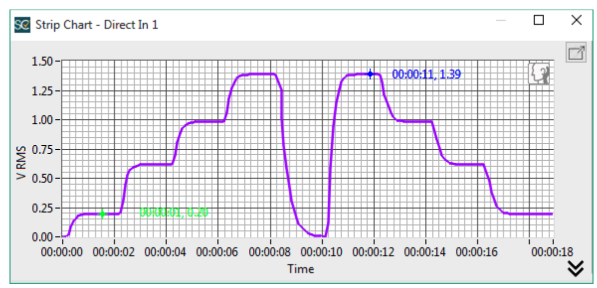


Real time distortion analyzer that allows you quickly get distortion values and plot them over time with the Strip Chart Recorder.

- THD Ratio - IEEE
- THD Ratio - IEC
- THD Residual
- THD+N Ratio - IEEE
- THD+N Ratio - IEC
- THD+N Residual
- SINAD
- Weighting Filter options:
  - Available selections: A, B, C, and "Memory List"
  - Any arbitrary curve from the Memory List can be used
- New Save to Memory List option

See ***Distortion Analyzer on page 483***.

Strip Chart Recorder (optional module)

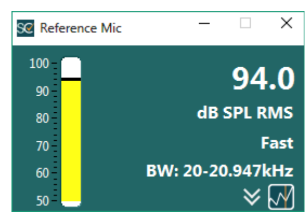


This is available in the Multimeter, Distortion Analyzer and Frequency Counter.

The Strip Chart plots the data from the instrument on a graph, where X axis is time and Y axis is the output of the meter.

See ***Strip Chart Recorder on page 469***.

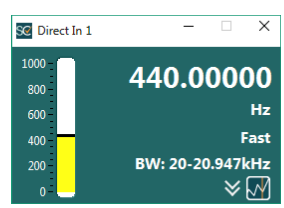
Multimeter



- New - Save to Memory List option. See ***Common Instrument Controls on page 457***.
- New - Linear-Repeating Averaging function
- New - Bandpass Filter option with Auto or Fixed frequency selections
  - Auto: Tracks the most prominent tone
  - Fixed: Allows you to specify the frequency
  - The Q is the same value, for either mode, regardless of frequency

For more information on the Multimeter See ***Multimeter on page 466***.

New Frequency Counter



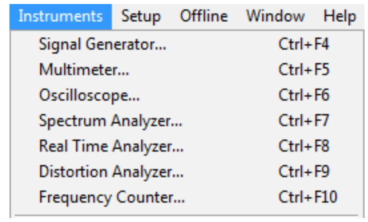
The new high resolution frequency counter offers an accurate and clear visual indication of frequency, determining the dominant signal in a selected signal path and returning a precise frequency measurement.

See ***Frequency Counter on page 484***.

Instrument List

Select **Instruments** (formerly Operate) from the SoundCheck main screen.

The Virtual Instruments offered by SoundCheck® allow you to manually operate the different components of the test system as stand-alone instruments:

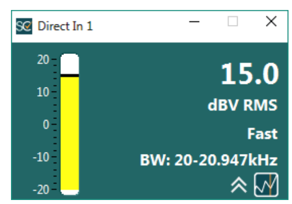


- **Signal Generator** (Ctrl+F4) - generate sine waves, noise and WAV files.
  - The EQ output function can be applied to all types of output signals available in the Signal Generator. (Requires optional module 2013 - EQ a Wav File)
- **Multimeter** (Ctrl+F5) - display the weighted or unweighted RMS level of an input signal
- **Oscilloscope** (Ctrl+F6) - view time waveforms
- **Spectrum Analyzer** (Ctrl+F7) - narrow-band frequency analysis
- **Real Time Analyzer** (Ctrl+F8) - Nth octave analysis
- **Distortion Analyzer** (Ctrl+F9)
- **Frequency Counter** (Ctrl+F10)

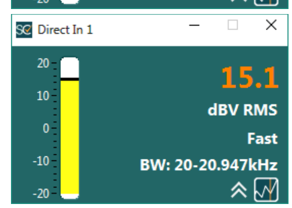
Overload Indicator

An overload indicator is included in all Instruments. This appears when the amplitude of the input signal exceeds the range of the hardware.

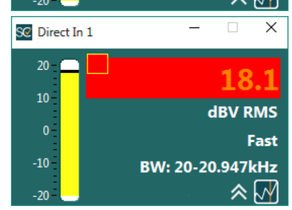
- If the signal is within the Max FSD tolerances the value characters of the meter are White



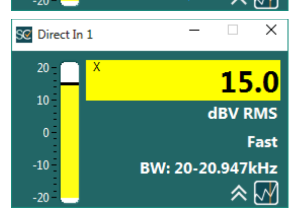
- When the input signal is within 3 dB of Max FSD, the meter reading will switch from White to Orange characters. See ***Record Level Monitoring - Max FSD on page 141***.



- When the input signal is actively overloading the hardware, the **Overload Indicator** becomes visible and flashes Red



- After the overload condition ceases, the indicator stops flashing, but it remains highlighted in yellow to let you know that an overload condition had been present in the current data acquisition session



The **Overload Indicator** is reset (closed) by clicking on the X (upper left corner), or by stopping and starting the instrument. For the Spectrum Analyzer, Oscilloscope and RTA, click on the Overload Indicator to close it.

Common Instrument Controls

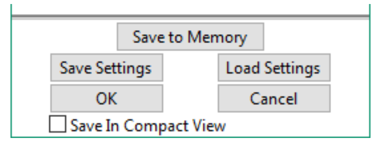


Figure 29-1: Common Controls

The following controls are used on all of the Instrument meters and analyzers:

- Save to Memory - Saves the acquired data to the Memory List
- Save Settings - Allows you save the current settings of the Instrument to a **.VIC** file
- Set as Default - Saves the current settings as the default settings for this type of Instrument
- Load Settings - Allows you load settings from a **.VIC** file
  - If the selected .VIC file does not contain settings for the current instrument, an error is returned
  - If the .VIC file contains multiple configurations for the current instrument, the first entry is used
- Close - Closes the Instrument
- Save In Compact View - Sets the Instrument to open in Compact View when you select **Save Settings** or **Save as Default**

Gain - Auto Read

All input Virtual Instruments have a drop down menu for setting Gain.

When using Listen Hardware or Portland Tool & Die Hardware you can select **Auto Read** in order to automatically update the Virtual Instrument with the gain setting for the selected channel.

It also allows you to override the gain setting by selecting any of the gain values available for the selected channel.

The gain of the hardware device changes when the Gain value is selected in the Virtual Instrument.

This also allows you to set the Gain value in a **Virtual Instrument Acquisition Step** without the need for a prior Message Step.

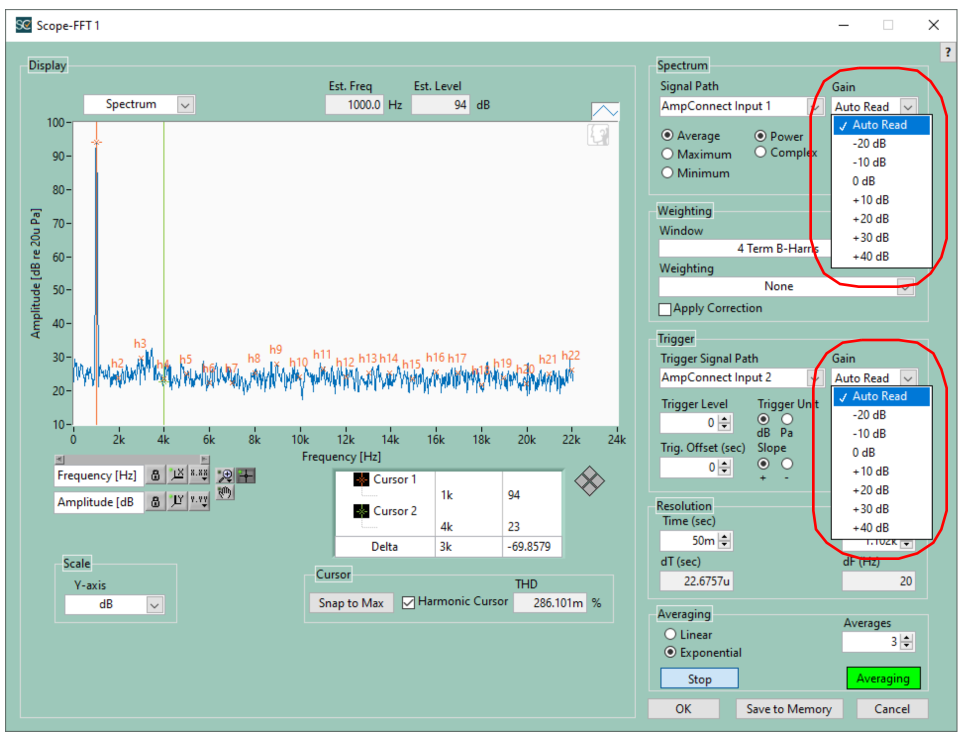


Figure 29-2: Gain Auto Read

Trigger

Instruments with a **Trigger** function have a Gain field as well.

Virtual Instrument Configuration

The set up of the multiple instruments can be saved in a Virtual Instrument Configuration file otherwise known as .VIC. This allows you to save or recall a set of instruments that might be used on a regular basis.

> ***Note:*** The Signal Generator output will be muted when a .VIC file is opened.

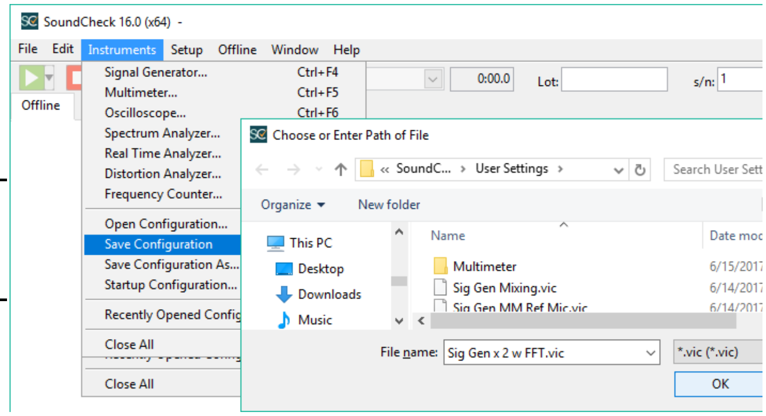


Figure 29-3: Save Config

Virtual Instrument Configurations are managed from the Instruments Menu on the SoundCheck Main Screen. See ***Figure 29-3***.

- Click **Instruments** Menu
- **Save Configuration** or **Save Configuration As** to save current Virtual Instrument setup
- **Open Configuration** to open a .VIC file
- **Start Up Configuration** - Opens the selected virtual instrument set up when SoundCheck is opened. See ***Figure 29-4***.
- **Recently Opened Configurations** - List of recently used configurations

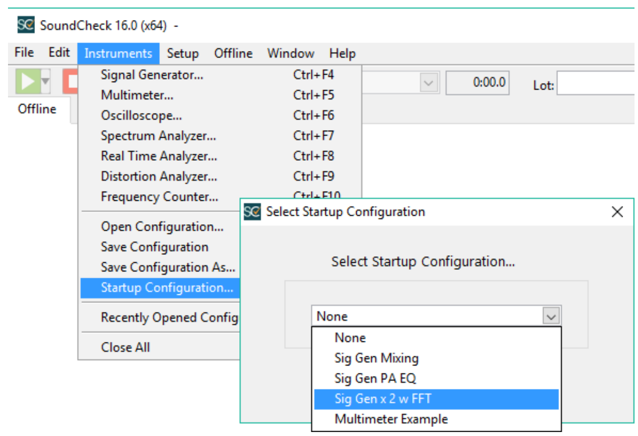


Figure 29-4: Startup Config

Opening Multiple Instances of Instruments

Several instances of the same virtual instrument or combinations of instruments can be opened at the same time.

- For multichannel applications, a new instance of the VI can be opened for each channel
- Both the Spectrum Analyzer and the RTA can run simultaneously while running a sequence so real time live data can be viewed
- Two Signal Generators can be used to mix signals (e.g., pink noise and a sine sweep) for signal conditioning
- Waveforms in the memory list may also now be played directly from the Signal Generator VI

Instrument Operation Time Rules

There is no deterministic link between the input and output of virtual instruments. They are completely independent. In fact, all Input Instruments are independent from each other. The only deterministic synchronization implemented between instruments in SoundCheck is “Sync” for Signal Generators (output side only).

The best way to guarantee that the Signal Generator is playing for the entire duration of the Multimeter record operation is by setting up your acquisition as follows:

- Include a record delay value that is long enough to allow the Signal Generator to open and start playing
- Configure the Multimeter to record for the duration of interest
- Make sure the Signal Generator is configured to play for long enough to cover:

  a. The Record Delay entered in the Acquisition Step field

  b. The record duration set in the Multimeter Time field

  c. The time it takes for the Multimeter to open and close

The example in ***Figure 29-5*** shows the Signal Generator set to run for 3.8 seconds.

This covers the times for:

Record Delay (1 Sec) + Multimeter Time (2 Sec) + Multimeter Open & Close (0.8 sec)

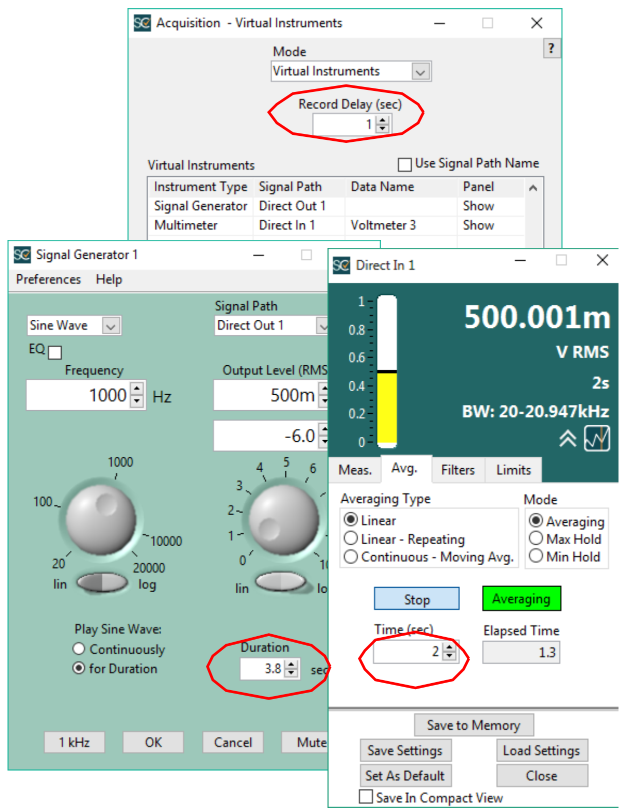


Figure 29-5: Total Acquisition Time

These methods should be used when using any of the virtual instruments as stand-alone instruments or when using them in an Acquisition Step.

The values used in this example are for demonstration purposes only. You should determine the optimal record delay, signal generator duration and multimeter duration times for your setup. They will vary based on OS, audio driver, and number of input and output channels acquiring during the step.

Signal Generator

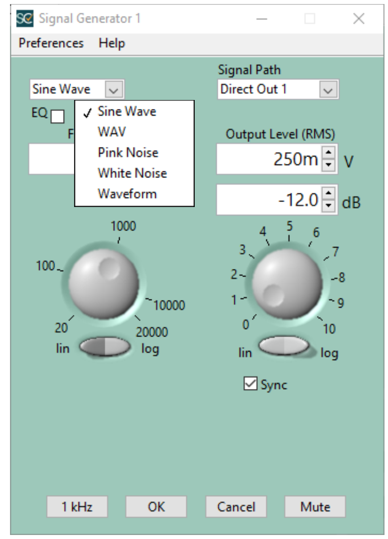


Figure 29-6: Signal Generator - Sine Wave

Choose the Output Signal Path, Frequency, and Output Level of the sine wave or WAV file to play. The *Signal Generator* always opens with **Mute** enabled to prevent playback of inadvertent, excessively high levels. When you are ready to play your signal, click the blinking **Muting** button to disable muting.

Features

- Frequency steps are in non-integer values. A decimal frequency can be entered as: 37.3 Hz.
- Bandwidth is limited only by the sampling rate of the audio interface. DC is not supported in the Signal Generator.
- Drop-down list - select stimulus types
  - Sine Wave
  - WAV (See ***WAV File playback on page 462***)
  - Pink Noise
  - Waveform (See ***Play Waveform Option on page 461***)
- EQ - Apply EQ uses the correction curve associated with the selected Signal Path (See Below)
  - The EQ output function can be applied to all types of output signals available in the Signal Generator. (Requires optional module 2013 - EQ a Wav File)
- Drop-down list - select Output Signal Path
- Max level and units defined in Calibration Editor, Output Signal Path
- Vary frequency using linear or logarithmic scale
- Vary level with linear or logarithmic scale
- Reference 1 kHz tone
- Sync
- Temporarily mute output

As of SoundCheck 15, you can unconditionally sync WAV file and Noise playback with other WAV file/Noise generators and Sine Wave generators. (Previous versions of SoundCheck allowed you to Sync multiple Sine Wave generators and multiple WAV file or Noise players, the latter only under specific conditions.)

- Sync only guarantees matching first sample output on the audio interface. Anything beyond that is up to the user. That is, they are not "synced" in terms of overall Signal Generator configuration:
  - Frequency and level are unique to and controlled individually on each synced generator.
- Changing the frequency on one synced signal generator will mute all synced generators. This is for phase matching reasons. Syncing signal generators guarantees the signals match at Phase 0/Time 0. Changing the frequency would change the phase relationship between two signals, so all generators need to be reset. Muting all synced generators effectively forces the sync operation to reset. Un-muting any synced generator is a signal for all in the group to start playing again in sync.
- Changing the level of one synced signal generator has no effect on any other synced signal generator.
- The same rules generally apply to synced WAV file/Noise Signal Generators. Clicking "Stop," modifying the WAV file path, or changing the Noise parameters will stop all synced generators. "Start" will start them all. Other than that, each one is configured individually.

EQ

The EQ output function can be applied to all types of output signals available in the Signal Generator (Sine Wave as well as WAV and noise). (**Requires optional module 2013 - EQ a Wav File**)

When the EQ box is checked the EQ Out Correction curve that is created in the output calibration process is applied to the output signal. This allows you to equalize the response of an artificial mouth or anechoic chamber. EQ out correction curves are populated with data when the Speaker Equalization or Simulated Free Field calibration sequences are selected in the output calibration process. See ***Equalization and Correction Curve on page 93***.

The level and units are determined by the calibration setup. For example, if you have calibrated the output sensitivity to include the gain of an amplifier in the path of the output signal, the level indicated will be the level into the DUT. The output level is in physical units. Refer to ***Calibration Configuration on page 79*** for more information on Physical Units.

Pink or White Noise

When playing Pink or White Noise you can select the Duration of the noise and the number of times it can be looped.

- Select **in Continuous Loop** to playback until you hit **Stop**
- Select **Duration** and enter the playback time in the **Duration field**

  or

- Select **N Times** and enter the number of times for file playback in the **N field**

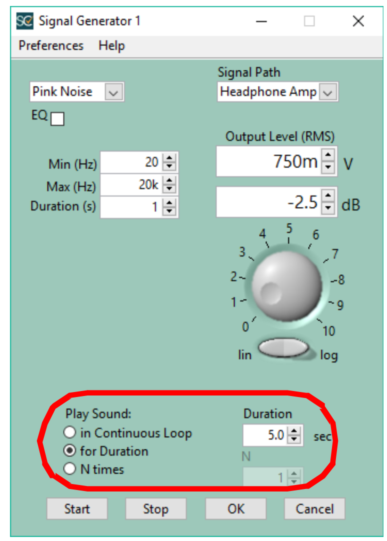


Figure 29-7: Play N Times

Play Waveform Option

Any waveform available in the memory list can be played from the Signal Generator. Select Waveform from the first drop-down list.

The Memory List Waveform is selected from the drop-down list under Waveform Name.

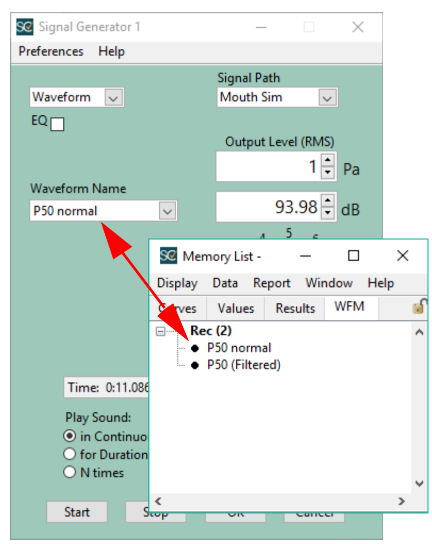


Figure 29-8: Play Waveform

WAV File playback

WAV file streaming in the signal generator with real time equalization removes memory limitations on the length of test signals. Longer test signals such as those required for analysis of speech and music can easily be accommodated. This applies to the standalone Signal Generator as well as the Signal Generator when used in an Acquisition Step. (WAV file streaming is not available in the Stimulus Step.)

As of SoundCheck 18, the sample rate of the WAV file automatically sets the sample rate of the Hardware Channel associated with the selected Signal Path.

Important! See ***Instrument Operation Time Rules on page 459*** regarding syncing and operation time with multiple instruments.

Level

The Output Level field allows you to set the playback level of the WAV file. The level is set in physical units. The output units will vary depending on the output units of the System Calibration Configuration. For example, if using an artificial mouth or anechoic test box the output level will be Pa rms. For an amplifier or direct output the level will be V rms. This requires an accurate calibration of the output signal chain. (See ***Calibration Configuration on page 79*** for instructions on output calibration.)

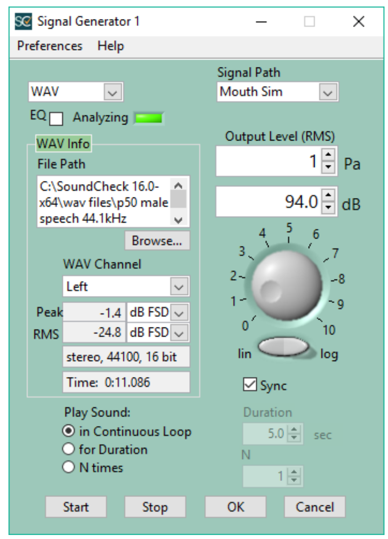


Figure 29-9: Signal Generator - WAV File

The drop-down list next to the Level field has the following selections:

- RMS level (Pa rms, V rms)
- dB level

WAV Info

- File Path - Browse and select a WAV file.
- WAV Channel - Select which channel of stereo WAV file to use for playback.

  The following section shows the properties of the selected WAV file. These values are for reference only, and cannot be changed.

  When opening a multichannel WAV file, each WAV file channel is numbered and selected from the **WAV Channel** drop-down list
- **Peak**: the maximum absolute value of the file (in dB FS, or %FS)
- **RMS**: the RMS value of the entire wave file (in dB FS, or %FS)
- **WAV format** stereo/mono, sampling rate, bit depth
- **Time**: total duration of the wave file in mm:ss.ms

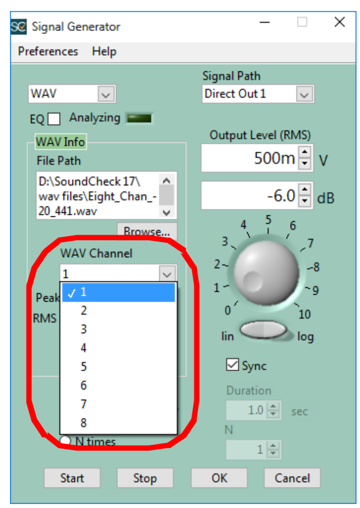


Figure 29-10: Multichannel WAV Files

Note: As of SoundCheck 10.0, WAV file playback “N times“ is available when using NI DAQmx hardware devices such as the PCI/PXI 4461.

More information on the use of WAV files in SoundCheck can be found in ***WAV File Excitation on page 129*** and ***WAV File Types on page 336***.

Why use an equalized WAV file?

Many modern electroacoustic products, in particular mobile phones, incorporate nonlinear digital signal processing for noise suppression and speech encoding. Usually these products must be tested using complex excitation signals such as real or simulated speech. Prior to their use, these special signals need to be equalized to compensate for the non-flat response of the mouth simulator or loudspeaker used as the output device.

Equalize a WAV file (Requires optional module 2013 - EQ a Wav File)

In order to obtain an accurate output level the WAV file should be equalized as shown in ***Figure 29-11***. The selected WAV file is equalized by applying the EQ curve defined in the System Calibration Configuration.

The System Calibration Configuration must have an equalization curve associated with the selected Output Signal Path. EQ Out Correction curves are populated with data when the Speaker Equalization or Simulated Free Field calibration sequences are selected in the output calibration process. See ***Equalization and Correction Curve on page 93***.

Make sure that the desired Output Signal Path has an appropriate EQ curve available in the Memory List.

The EQ curve is unique to the Output Signal Path. The EQ curve is available in the Memory List and can be manipulated via Post-processing or calibration.

(For more information on generating or manipulating EQ curves refer to ***Calibration Configuration on page 79***.)

How to example

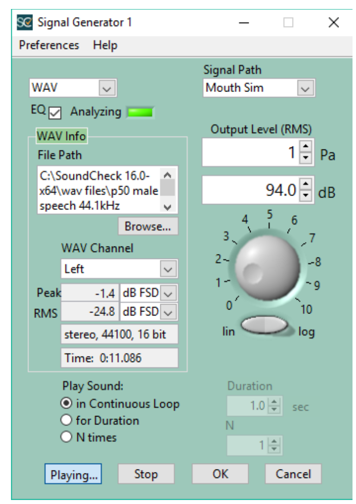


Figure 29-11: WAV File EQ

1. Open the *Signal Generator* by choosing it from SoundCheck's **Instruments** menu. (Open the *RTA* as well if you wish to see the measured signal change as the equalization is applied.)

2. Select **WAV file** from the radio button in the upper left hand corner of the *Signal Generator*. Browse to the WAV file you would like to equalize, and set your Output Level and number of times for the file to play. Press **Start** to play the file, and watch the unequalized response on the *RTA*. The WAV file will stop playing when the number of times to play has passed, or, if you have chosen **Continuous Loop**, click **Stop** when you are satisfied.

3. Select the **EQ** option at the top of the *Signal Generator*, make sure your Output Level is appropriate, and the WAV file is set to play a fixed number of times (for Duration will play the WAV file once).

4. The **Analyzing** Status light indicates that the EQ'd signal is being calculated. When the light goes out the process is complete.

5. Press the **Start** button and watch the equalized response of your WAV file on the *RTA*.

Important! As of SoundCheck 8, an equalized version of the WAV file is no longer created. The equalization is done in real time.

---

6. Return to editing your sequence. Add a Virtual Instrument Acquisition Step to play the WAV file (e.g., P50 speech.wav) through the *Signal Generator.* The EQ box must be checked in order for SoundCheck to apply the EQ.
7. Record the response through the *RTA*, as in ***Figure 29-12***.

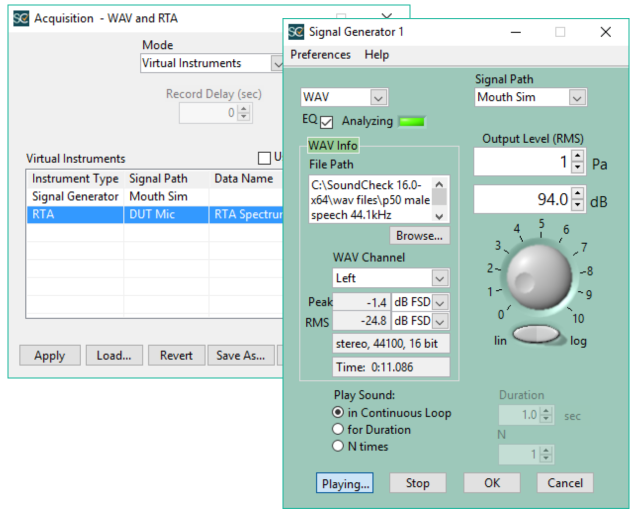


Figure 29-12: Equalize WAV in Sequence

Signal Generator Synchronization

Multiple signal generators can be synced in real time and in a sequence so they will start and stop simultaneously. This is important when checking the phasing of multiple channels such as loudspeaker and microphone arrays.

- ***Figure 29-13*** shows three Signal Generators, all set to Sync
- Output on multiple channels or on the same channel
- Clicking **Start** or **Unmute** on a signal generator in the group will start all signal generators in the group
- Clicking **Stop** or **Mute** operates all generators in the group
- The output level of signal generators can be changed independently while the group is running
- Changing the Signal Path or Frequency on one generator in the group will Stop/Mute the whole group
- Different signal types can be mixed as shown in ***Figure 29-14***
- Sync allows you to Start, Stop and Mute multiple signal generators by clicking on only 1 button
- It synchronizes the phase of sine signals and the start of WAV files

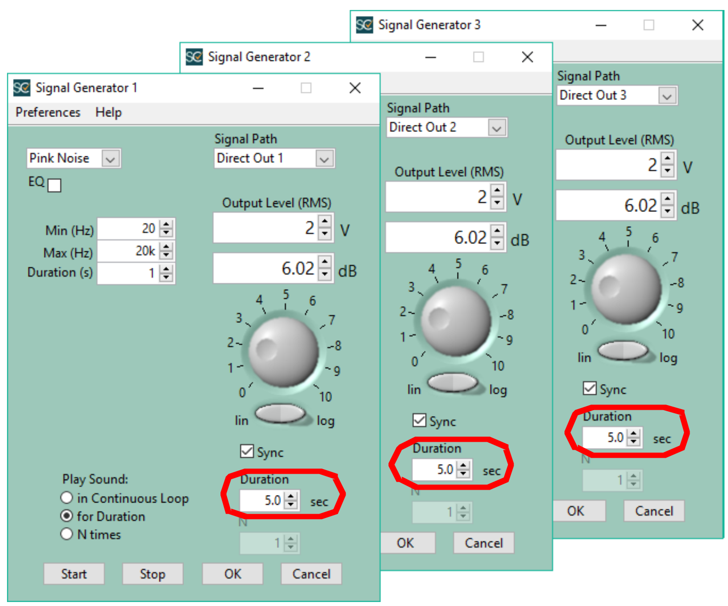


Figure 29-13: Signal Generator Synchronization

Important! See ***Instrument Operation Time Rules on page 459*** regarding syncing and operation time with multiple instruments.

Signal Generator Mixing

In *Figure 29-14*, two Signal Generators are open, both sending to Direct Out 1 but with different frequencies set in their respective control panels.

- Signal Generator 1 is set to 1000 Hz
- Signal Generator 2 is set to 2000 Hz
- The Scope FFT shows the mix of the two signal generators

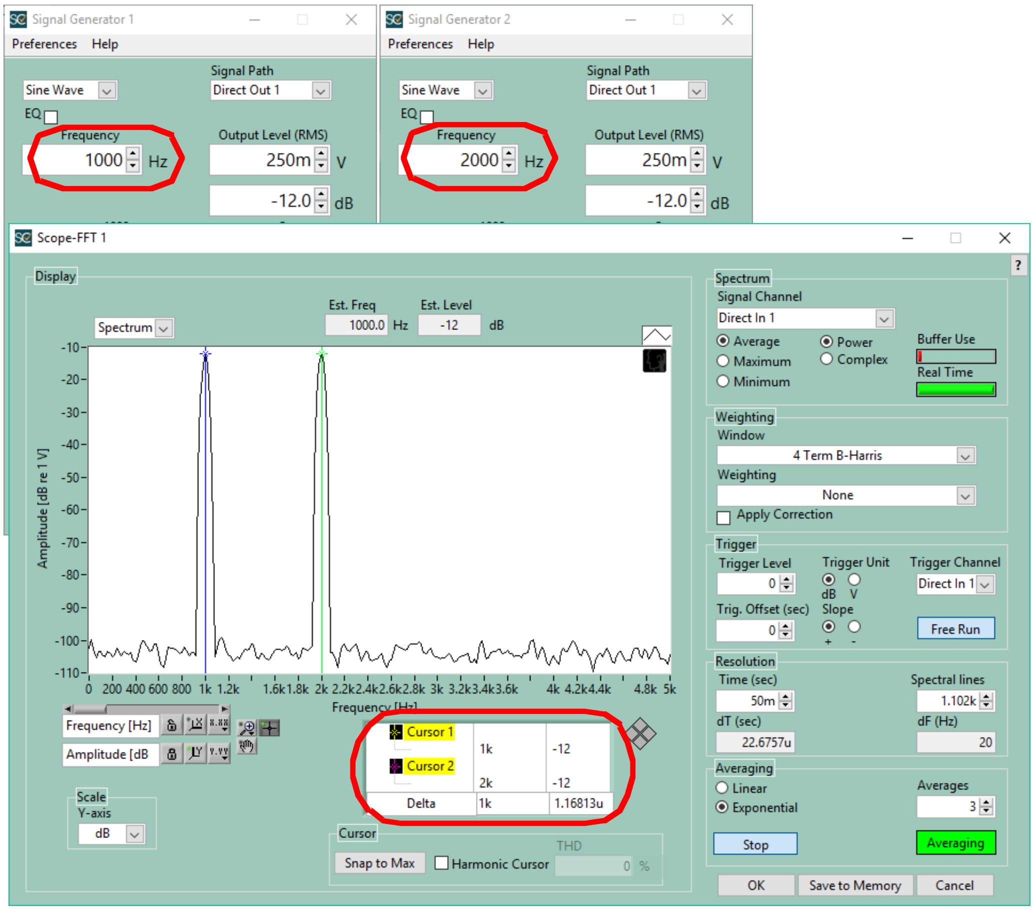


Figure 29-14: Multiple Signal Generator Mixing

Of course, other types of signals can be mixed. Signal Generator 1 can play a 1 kHz sine wave while a WAV file is played from Signal Generator 2.

Note: When playing WAV files, the sample rate of the WAV file automatically sets the sample rate of the Hardware Channel associated with the selected Signal Path.

Note: See ***WAV File Types on page 336*** for more information on supported WAV file types.

Important! See ***Instrument Operation Time Rules on page 459*** regarding syncing and operation time with multiple instruments.

Multimeter

The Multimeter allows for measurement of input signal in real time.

The level and units are determined by the Signal Path Calibration setup. For example, if you have calibrated the input sensitivity to include the sensitivity of a microphone used to measure the input signal, the level indicated will be the absolute SPL level at the microphone position.

Meter Display

- **Compact View Button** - (double Up or Down arrows) Minimizes the Multimeter to just the Thermometer and Numeric display. This helps to save space on a computer monitor.
- **Strip Chart Recorder Button** - Opens and Closes the Strip Chart Recorder. ***See "Strip Chart Recorder" on page 469.***
- **Thermometer** - When Limits are active, a Yellow bar indicates passing. A Red bar indicates the level is failing the set limits.
- **Overload** - The numeric display area will turn yellow when the signal into the audio interface exceeds the Max In Vp level

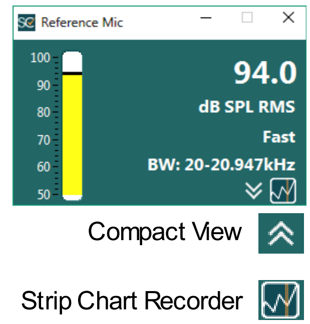


Figure 29-15: Buttons

Resolution

The Display Resolution can be set by Right-clicking on the numeric display of the meter and setting the values for Linear Numeric and dB Numeric as in ***Figure 29-18***.

<u>Linear Numeric Field</u>

- Choose between SI notation and Floating point
- Select Digits of precision or Significant digits
- Check to Hide Trailing Zeros

<u>dB Numeric Field</u>

- Set Digits of precision. Setting a greater precision may be required when using Multimeter Limits in a sequence as noted above.
- Check to Hide Trailing Zeros

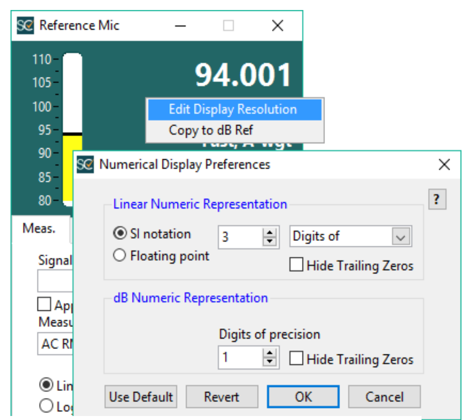


Figure 29-16: Numeric Display Preferences

Measurement Tab

- **Signal Path** - Select any signal path available in the Calibration Editor as the Input to the Multimeter
- **Apply Correction** - Allows you to apply input correction curves associated with the selected, calibrated signal path
- Measurement Type
  - Select AC RMS, AC Peak or DC RMS (DC requires a DC coupled audio interface)
  - Peak - Within 100 mSec time blocks
- **Strip Chart Button** - This opens the Strip Chart Recorder for the Multimeter. Circled in ***Figure 29-17***.
- **Autoscale** - Default is 100 dB scale. You can manually change the upper and lower scale values to “Zoom In’. This is not remembered when a configuration is saved.
- **Save in Compact View** - Changes the meter to Compact view when you click **Save Settings** or **Set as Default**.

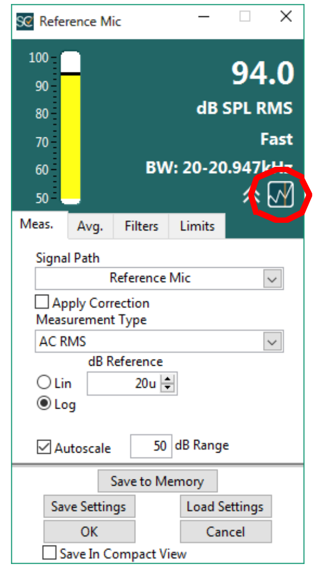


Figure 29-17: Multimeter

Averaging Tab (Avg.)

- **Averaging** - The meter shows the Average value, updated according to the Time or Time Weighting selected
- **Max Hold** and **Min Hold** - The meter collects data for the duration the meter is running and shows the Maximum or Minimum value
- Linear
  - Click “**Start**” to run the meter
  - The *Multimeter* will run for the duration set in the **Time** field
  - The “Averaging” indicator is green when the Multimeter is running
  - The Elapsed Time window shows the amount of acquisition time
- Linear - Repeating
  - Measures the input signal for the duration set in the **Time** field. Example: If Time is set to 1 second, the meter will measure for 1 second and then repeat. This is particularly useful when used with the Strip Chart Recorder to output a curve with measurements taken in blocks of time. The **Value** saved to the Memory List is the last value taken.
  - The Strip Chart Recorder Duration field will set the length of the acquired curve. See ***Strip Chart Recorder (optional module) on page 455***.
  - This will run in a loop until you click **Stop**. Only the data from the most recent acquisition is saved to the Memory List from a Virtual Instrument Acquisition Step or when you click “Save to Memory”.
- Continuous - Moving Avg.
  - Click “**Continuous**” to run the meter
  - The Multimeter will run continuously until you click **OK** or **Cancel** in the Multimeter window
  - Averaging Time: Fast (250 mSec), Slow (2 Sec) or User Defined
  - The Strip Chart Recorder Duration field will set the length of the acquired curve

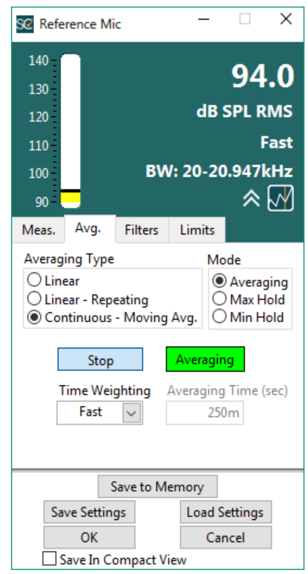


Figure 29-18: Averaging

Filters Tab

- **Weighting** - Select A, B, C or any curve available in the Memory List
- Check to enable the **High Pass Filter** and/or **Low Pass Filter** and enter values in Hz
- The selected frequency will be the **-0.1 dB point** of a dynamically calculated filter
- Bandpass Filter option with Auto or Fixed frequency selections
  - Auto: Tracks the most prominent tone
  - Fixed: Allows you to specify the frequency
  - The Q is the same value, for either mode, regardless of frequency

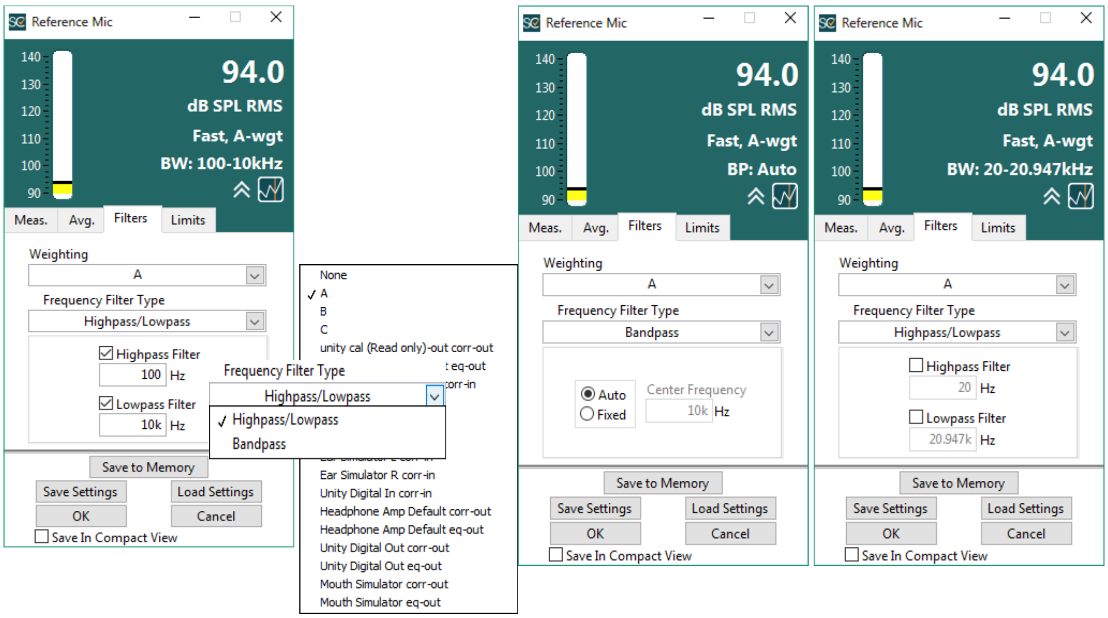


Figure 29-19: Filters

Limits Tab

Limits can be used when the Multimeter is opened from the Instruments menu. The limit settings are remembered when the Multimeter is saved in a Configuration. The Pass/Fail indicator is only for visual reference.

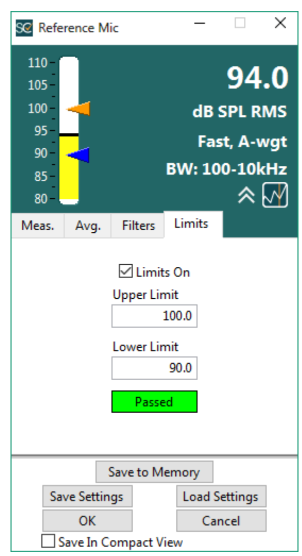


Figure 29-20: Limits

- Check **Limits On** to enable
- Upper and Lower Limit values are entered manually
- When the condition of the limits is “Passing”, the Multimeter Thermometer is yellow. When failing, the thermometer turns red.
- Pass/Fail conditions can be set in Step Configuration to use for Conditional Branching (Jump on Pass/Fail to ...). See ***Configure Step on page 446***.

Strip Chart Recorder

The Strip Chart Recorder is an optional module available for the following Instruments:

- Multimeter
- Distortion Analyzer
- Frequency Counter

Click the Strip Chart button to open the feature.

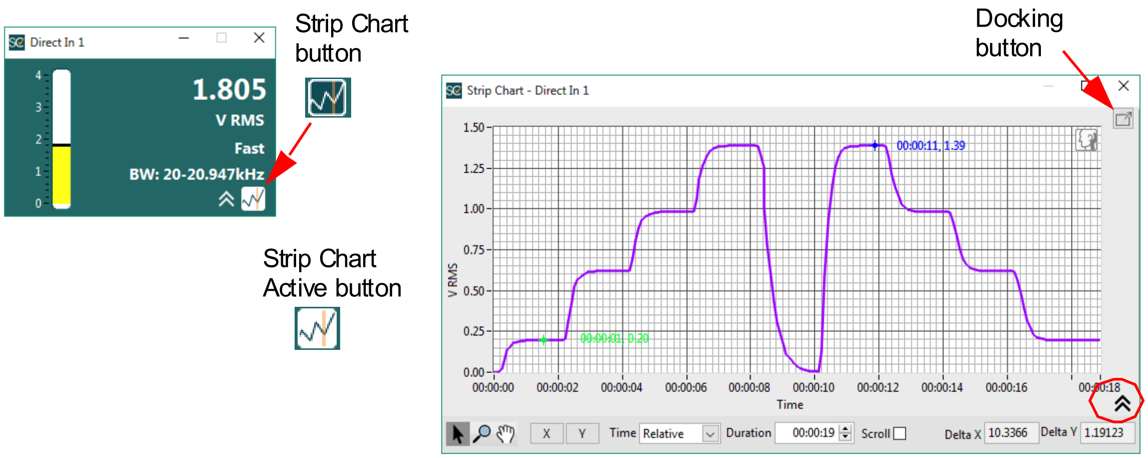
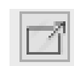


Figure 29-21: Strip Chart

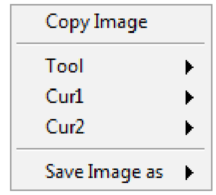
The chart window can be “Undocked” from the meter, resized and moved on the desktop as an independent window. Click the **Docking** button as shown in ***Figure 29-21***.

The Strip Chart can be “Docked” to it’s Instrument by clicking the **Docking** button. 

Click the Expand Menu button on Strip Chart for the following controls: 

- Cursor Arrow - Click on an X or Y axis endpoint value to change the extents of the graph
- Magnifying Glass - Zoom on X, Y or and X and Y axis
- Hand - Move the graph within the window. (Autoscale X and Y should be shut off when using this.)
- XY Autoscale buttons
- Time
  - Relative - Shows the time according to the start of signal acquisition
  - Actual - Shows the clock time of the SoundCheck Computer. **Note: Data saved to the Memory List uses Relative Time values.**
- Duration - Sets the maximum window size. The window scales to show as much data as possible until the Duration value is reached. The Strip Chart will then stop unless Scroll is selected.
- Scroll
  - Off - The Strip Chart recorder stops after the Duration value has been reached
  - On - The Strip Chart data will scroll to the left once the Duration value has been reached. The Strip Chart window maximum width is set by the Duration value. This is also the available data that is saved to the Memory List when you click **Save to Memory**.
- Delta X and Delta Y - Shows the X and Y axis difference between Cursor 1 and Cursor 2

- Right-click the Strip Chart window to place cursors, access the Zoom Tool and to export the window as a JPG or BMP file. The remaining controls are not available.
- Cur1/Cur2 - Two cursors are available in the Strip Chart window
- Tool - An alternate means of accessing the Cursor Arrow, Zoom, Hand and Autoscale controls



Note: Syncing of multiple Strip Chart Recorders is not supported

Averaging Tab

The Strip Chart Recorder follows the selected Averaging Mode. See ***Averaging Tab (Avg.) on page 467***.

Linear - Makes one measurement that is integrated over the Time set in Averaging. The Strip Chart shows all the measurements that comprise that measurement.

Linear Repeating - Makes one measurement every X seconds, depending on the Time set.

Continuous - Makes a continuous measurement but the Strip Chart Output Curve is set by the Duration Field.

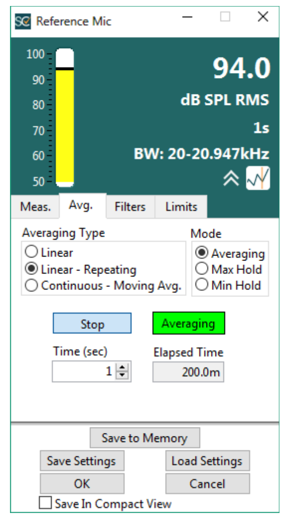


Figure 29-22: Linear Repeating

Save to Memory

In the meter that is linked to the Strip Chart, click on **Save to Memory** to send the data to the Memory List.

- Only the data in the window is saved
- If Scroll is selected, data that scrolls past the left edge of the window is not saved
- You can click **Save to Memory** more than once while scrolling to save “Snapshots” of data
- When used in a sequence, the data is saved to the Memory List when acquisition is complete

The Value and Waveform data will be named with the Meter Input name, e.g.: Multimeter - Direct In 1 and saved to the Memory List Values and WFM tabs. If multiple data Snapshots are taken while scrolling the data will be Autoprotected and the order number prepended to the data name.

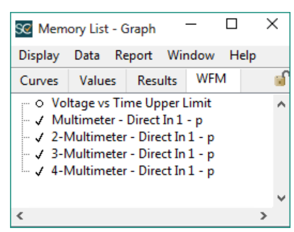


Figure 29-23: Memory List

When used in a sequence, Limits can be applied to the Strip Chart waveform (or value) as shown in ***Figure 29-24***.

This allows you to apply limits to measurements of Level vs Time, Distortion vs Time or Frequency vs Time.

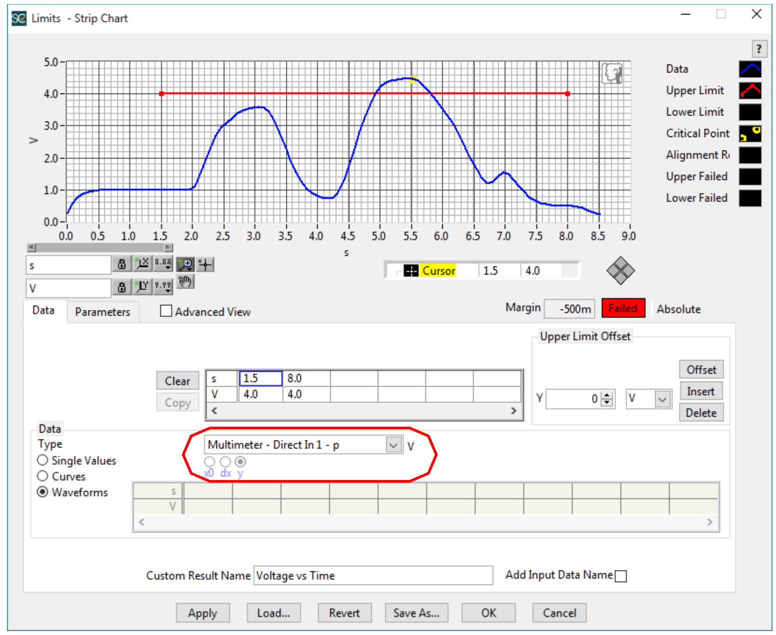


Figure 29-24: Limits On Strip Chart Waveform

Spectrum Analyzer

The spectrum analyzer allows detailed analysis of spectral components. The weighted spectrum is now available. The weighting functions include: none, A, B, C or any other curve in the SoundCheck *Memory List*.

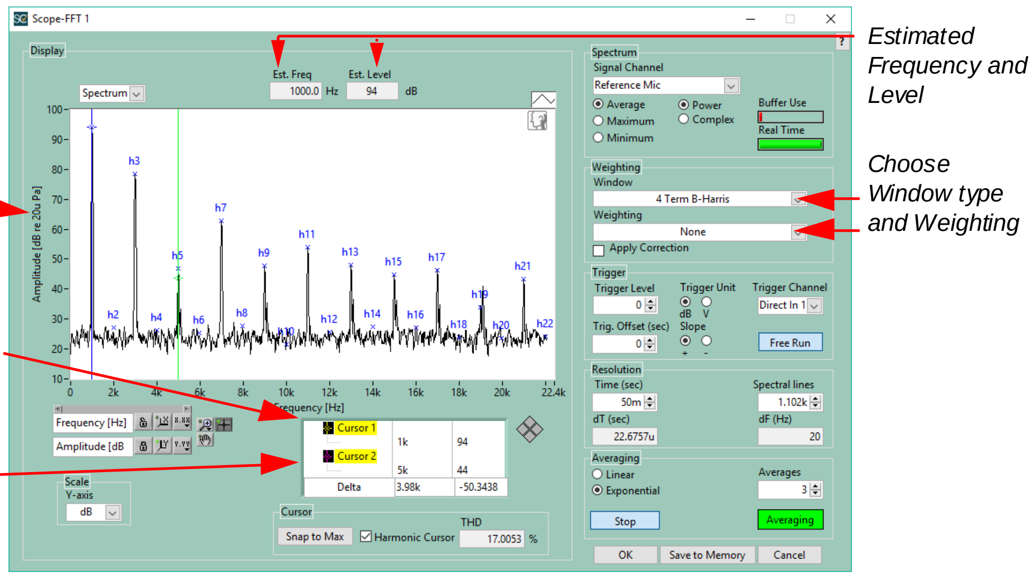
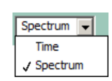


Figure 29-25: FFT Screen Overview

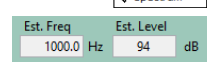
FFT Controls

Display (Mode Select and Cursor Readout Controls)

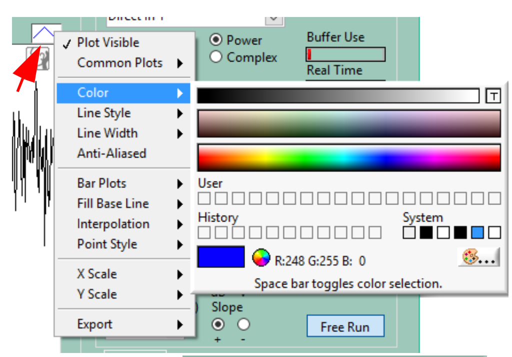
- **Spectrum/Time**: Selects mode of operation - frequency or time analysis. The Oscilloscope will run in the time domain, while the Spectrum Analyzer will run in the frequency domain.



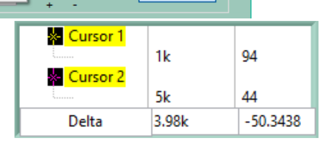
- **Est. Freq**: Displays interpolated frequency estimate, in Hz, at Cursor 1. The interpolated frequency value is significantly greater in resolution than the measurement bandwidth.



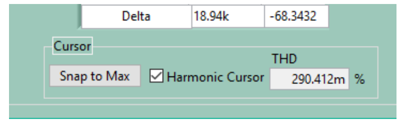
- **Est. Level**: Displays the calculated level estimate at Cursor 1, in units RMS. In order to get valid results for Est. Frequency and Est Level, Cursor 1 must be positioned before or during measurement. The interpolated level value is significantly great in resolution than the direct reading in the Cursor 1 legend.
- **Spectrum** (Display properties): Allows you to change the color and attributes of the Spectrum Line. For more information on these controls refer to ***Display Editor and Memory List on page 321***.



- **Cursor 1 & 2**: Shows the position on the X and Y-axis of the two cursors. Cursor position can be fine tuned by using the Diamond control to the right of the cursor panel.



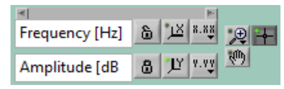
- The **Snap to Max** button on the Scope-FFT control panel moves Cursor 1 to the peak of the acquired spectrum. The Estimated Frequency and Estimated Level are shown in the fields at the top of the Spectrum Display. This function is available when the mode is set to Time or Spectrum but the cursor location is only shown when the mode is set to Spectrum. The Estimated Frequency and Level are shown in either mode. Clicking on the Harmonic Cursor will then plot and show the Harmonics on the FFT display as well as calculate the THD.



- **Delta**: Difference (in relevant units) between the position of Cursor 1 and Cursor 2.
-

Axis Scaling, Zoom and Style Controls

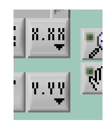
- **Curve Attributes**: Controls curve attributes. See ***Introduction on page 25*** for more information on these graph controls.



- **Autoscale**: Left-click on the Lock symbol to turn Autoscale on and off. The Green light indicates that Autoscale is on (as well as the Lock/Unlock symbol). Clicking on the X or Y axis symbol, autoscales the axis without turning autoscale on. This is a "one shot" autoscale

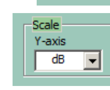
	
Autoscale OFF	Autoscale ON

- **Precision**: Allows you to set the level of precision for the display of values.



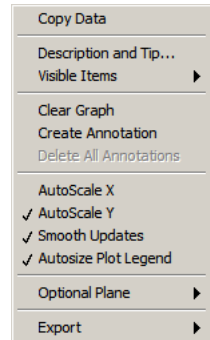
Scale

- Y axis: Provides choice of dB (relative) or linear (absolute) units.



Right-click Functions

Right-click the graph:



- Copy Data - Creates a screen shot of the graph side of the instrument window, including Cursor section and annotations
- Description and Tip - *Not used*
- Visible Items - Select/De-select items to show in the graph section of the instrument
- Clear Graph - *Not used*
- Create Annotation - Allows you to post markers on the graph that are included when exporting a screen shot of the graph
- Delete All Annotations - Removes all annotations from the graph
- Smooth Updates - Selected by default to minimize flicker as the display changes
- Autosize Plot Legend - Always on
- Optional Plane - *Not used*
- Export
  - Data to Clipboard - Copies the graph data to the clipboard so that it can be pasted into another application
  - Data to Excel - Opens Excel and copies the graph data to a new worksheet
  - Simplified Image - Sends a black and white capture of only the graph window to the clipboard. Annotations are included.

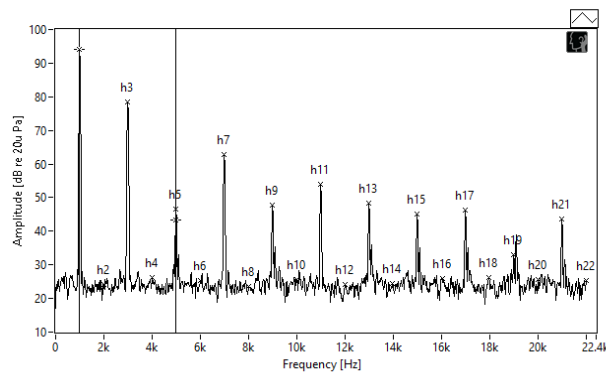
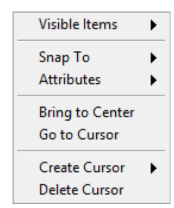


Figure 29-26: Export Simplified Image

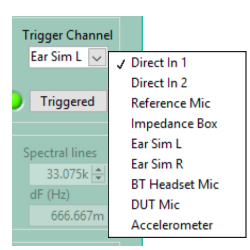
Right-click a Cursor name:



- Visible Items - Turn the Vertical Scroll Bar on and off (Other controls are not used)
- Snap To - *Not used*
- Attributes - The Standard Line Attribute settings are available. Refer to ***Cursor Color and Grid Color on page 56***.
- Bring to Center - Moves the selected cursor to the center of the spectrum display
- Go to Cursor - Centers the spectrum on the selected cursor
- Create Cursor - *Not used*
- Delete Cursor - *Not used*

Trigger Controls

Triggering can be used to automatically capture a spectrum when the signal level exceeds the value set in the “Trigger Level” field.



- **Trigger Channel**: Sets the Input Signal Path used for Trigger

  Trigger Level: Sets level of measurement trigger, in either Physical Units or dB. If dB is chosen, the trigger threshold is on the positive value of the waveform. The trigger level is a Peak value. The trigger value has the same dB reference as the trigger signal path.

- **Trig. Offset (sec)**: Sets the amount of time that the Acquired Signal is shifted, relative to the point at which it is triggered. A negative offset indicates that the signal will be shifted to the right by the time that is in the field e.g., -100 mSec.
- **Slope**: Selects whether Positive going signal triggers before negative going signal
- Indicator light next to **Triggered** button indicates when the signal is triggering measurement

Resolution

When the Display type is set to Time, only the Time [sec] field is available.

When the Display type is set to Spectrum both the Time and Spectral lines fields are available. See ***Figure 29-27***.

- Time [sec] indicates the total amount of time allowed before the next trigger will be allowed. Measurement record length in seconds (sec).

  As **Averaging Time** increases, the filter bandwidth gets more narrow. ***Figure 29-27*** shows the effect of increasing the Averaging time from 50 ms to 500 ms.

- dT [sec]: Sampling Interval of the time signal. This is the inverse of the sampling rate determined in the System Hardware configuration.
- Spectral lines: Number of FFT lines from 0 Hz to 0.5x (half) of the audio interface sampling frequency.
- dF [Hz]: Resolution/spacing of Spectral Lines.

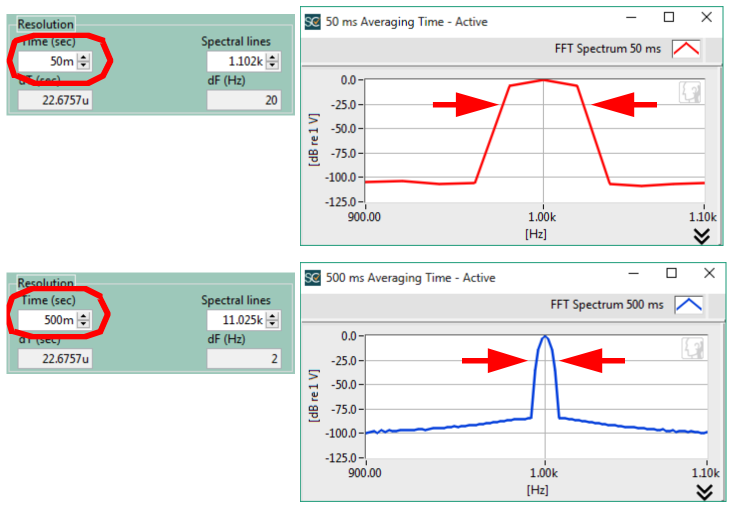
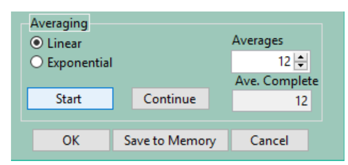


Figure 29-27: Averaging Time

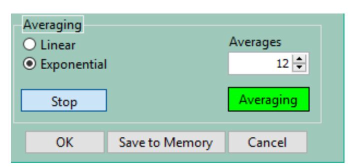
Averaging Controls

- **Linear/Exponential**: Different averaging modes

  Linear will average for only the number of averages entered (e.g., 12) and then stops measuring. When Linear averaging is selected, the Averages indicator will appear, specifying the number of averages currently completed.

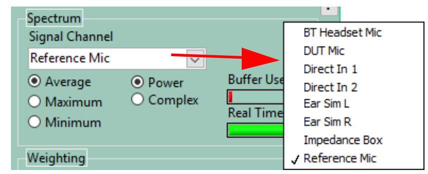


  Exponential allows for continuous measurement. Averages, in this case, sets the number of spectrum that are averaged together for each measurement display. If this number is set to 1, there will be no averaging. This will then be the raw spectrum.



- **Start Button** (Available in Linear mode): Starts the acquisition of data. Acquisition ends when the number of averages is complete.
- **Continuous** (Available in Exponential mode): Starts the continuous measurement of spectrum and ends when the Stop button is clicked. The Continue button allows you to resume a previously started measurement.

Spectrum



- **Input Channel**: Any predefined Input Signal Path will be available for input.
- **Average**: Displays the average spectrum, either Linear or Exponential, as determined in the Averaging Field.
- **Power**: The result is the average Power of each FFT bin, excluding phase information.
- **Complex**: The result is the average of the complex value of each FFT bin (amplitude and phase); this must be used in conjunction with the trigger controls. The signal must be very stable when using triggering, otherwise slight random variations in phase from one trigger cycle to the next will cause synchronous components to be underestimated in amplitude. A greater signal to noise ratio can be obtained by using this.
- **Maximum**: Displays the maximum of each spectral line.
- **Minimum**: Displays the minimum of each spectral line.

Weighting Controls

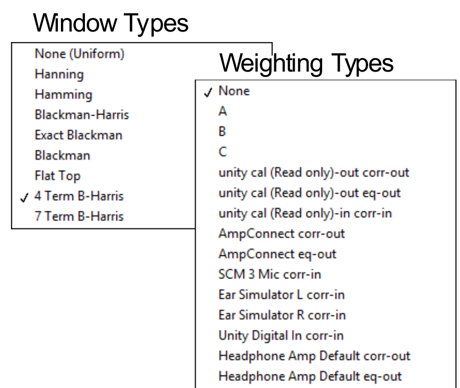
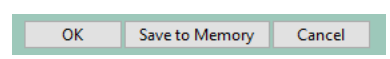


Figure 29-28: Window and Weighting Types

- **Window**: User selectable time window. See ***Figure 29-28***.
- **Weighting**: Allows you to select A, B or C weighting. Additionally, a curve from the *Memory List* can be used as a weighting function. This is applied to the measured spectrum, e.g., frequency domain.
- Apply **Correction**: Applies the correction curve associated with the Input Signal Path. Refer to ***Calibration Configuration on page 79*** for information on the creation of Calibration Curves.

Operation Control Buttons



- **Save to Memory**: Saves the current measurement data to the Memory List. The curve will appear in the Memory List as "**FFT Spectrum [L]**". Change the file name by Selecting "Rename" from the **Memory** drop-down list in the Memory List. The FFT Spectrum displays the amplitude corrected for the noise energy bandwidth of the time window used in the FFT calculation (e.g., Hanning, 4-Term Blackman-Harris, etc.). The FFT spectrum should be used when measuring pure tones and/or sinusoidal distortion components, such as harmonics. The FFT Spectrum is used when using the Power Sum post-processing function. Typical applications include determining the total power in a frequency band when using Pink or White noise or program material.

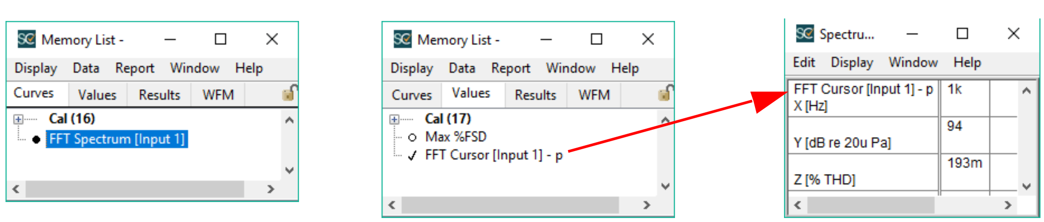
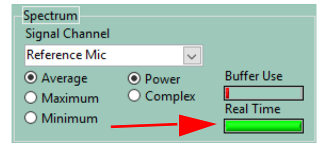


Figure 29-29: Memory List - Snap to Max Values

- If "Snap to Max" is selected before clicking "Save to Memory", the FFT Cursor values will also be added to the Memory List: Est. Freq, Est. Level and THD. This value can then be shown in a Display Table as shown in ***Figure 29-29***.
- **Cancel**: Closes the *Spectrum Analyzer*, any changes to user defined fields will not be stored

- **OK**: Closes the *Spectrum Analyzer*, storing all settings.

Measurement Status



- **Buffer Use**: This shows memory buffer use. If this indicator is not solid red, there is no data loss (all data processed). Real time analysis is performed.
- **Real Time**: This shows the update rate of the data display. A solid green bar indicates that the data is being displayed as quickly as it being acquired. There will be circumstances where the display processor cannot keep up with the actual data processing. Since the display may not be able to be updated after each completion of a new average, this field would be a partial green bar. The actual time data is still processed even though the display is not updated with each new average. After the number of required averages has been calculated, the final screen is the cumulative average of all averages.

Measurements

Listen's Scope/FFT instrument allows straightforward measurements of acoustic, vibration and electrical signals. The procedure to make a measurement is relatively simple:

1. Set up the Y-axis, X-axis, time window, and units in the **Scale** and **Weighting** sections.

   In the **Trigger** and **Averaging** sections:

2. Select the proper Input Signal Path.
3. Input the Trigger Level and the Slope.
4. Determine the Time record length. For lower frequencies and higher resolution, a longer record length is needed. The delta Hz will automatically be updated to reflect changes in record length and number of FFT Lines.
5. Enter the number of FFT Lines desired and select type of averaging (Linear or Exponential).
6. Select and type in the number of Averages needed.

Running a Measurement

After set up of the above sections, pushing **Start** in linear averaging mode will start the measurement and averaging and will stop the analyzer when finished (when the requested number of averages is reached). In Exponential averaging mode, this button becomes the **Continuous** button which, when pressed, starts exponential averaging. To stop exponential averaging, press the **Continuous** button again. Pressing **Start** in linear averaging mode while running will stop the measurement and averaging. Use the cursors and/or saving facilities to analyze and view data.

Note: The **Estimated Frequency** and **Level** fields only update during measurements. When the measurement is finished (as in the case of linear averaging), the fields will reflect the last cursor position before the completion of the last average.

Using the Graph Cursors

When the cross (+) cursor appears (selected using the Return to Cursor button), click on the graph cursor and drag it to the frequency line of interest to read the frequency and level data. Use the other cursor facilities to zoom, scale and change the graph attributes. Refer to ***Introduction on page 25*** for tips on using graph controls.

FFT Windowing Types

An FFT analysis must be made on a time record or measurement of finite length. The measurement is then limited to a specified window. Spectral leakage occurs when the acquired data does not exactly correspond to one of the spectrum frequency lines. This leakage leads to amplitude accuracy errors as well as obscuring adjacent frequency peaks. For these reasons it is important to apply a Windowing function to obtain more useful information from a measurement. Various windowing types affect the results of the measurement in different ways.

None (Uniform)

Also referred to as Rectangular. No windowing is applied to the measurement. This works well with transients that are shorter in length than the measurement time. Due to the flat characteristic in the time domain, all parts of the signal are equally weighted.

Hanning

This is a smooth window function, which is one period of a cosine^2 function and tapers to zero at the beginning and end of the measurement. Hanning is recommended for the analysis of noisy signals. Its main advantage is that it has excellent frequency selectivity.

Blackman-Harris

This window has a low ripple (<0.87 dB) in the pass-band and a low skirt (<-80 dB) in the stop-band. Blackman-Harris is recommended for harmonic and order analysis. Its main advantage is that it has an excellent dynamic range combined with good frequency selectivity.

Flat-top

This window has very little ripple (<0.01 dB) in the pass-band (in the frequency domain). The window's main use is for level measurement of sinusoid (calibration), due to its negligible amplitude errors. Its main advantage is that it has excellent amplitude accuracy.

Note: Audio Analyzer Type 2012, Brüel & Kjær Technical Documentation, BE 1074-12, 1994.

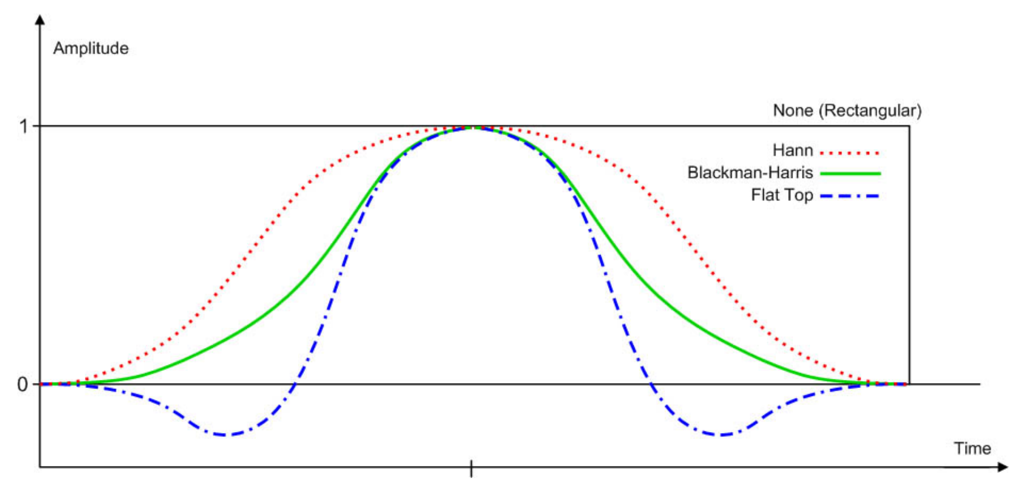


Figure 29-30: Windowing Types

Signal Content	Window Type
Sine wave or combination of sine waves	Hanning
Sine wave (amplitude accuracy is important)	Flat Top
Narrow-band random signal (vibration detail)	Hanning
Broad-band random (white noise)	None (Uniform)
Closely spaced sine waves	None (Uniform), Hamming
Signals with harmonics	Blackman-Harris
Unknown content	Blackman-Harris

Figure 29-31: Table of Applications vs. Window Types

Oscilloscope

The *Oscilloscope* allows you to display and analyze the waveform of the signal from your transducer. This instrument draws a graph of the instantaneous input signal as a function of time. You can configure the settings of the SoundCheck *Oscilloscope* to take a linear average of the input signals as well. The units of the Y axis of the *Oscilloscope* are determined by the System Calibration Configuration. Controls for the *Oscilloscope* are similar to those in the *Spectrum Analyzer* and are defined in more detail in the *Spectrum Analyzer* section.

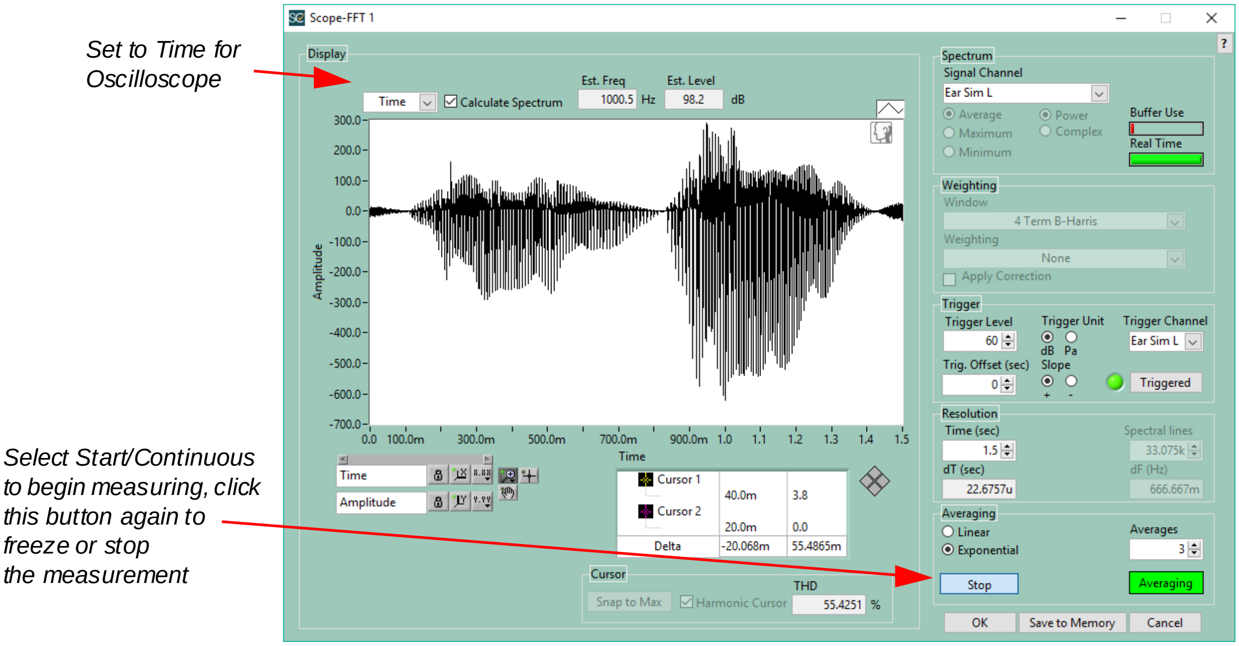


Figure 29-32: Oscilloscope

- **Triggering**: All of the controls available in Spectrum are available except for the following: Average – Maximum - Minimum, Power - Complex, Window, Weighting, Apply Correction, Spectral Lines, dF, THD and X & Y Axis scale. Triggering can be used to automatically capture a spectrum when the signal level exceeds the value set in the “Trigger Level” field. Refer to ***Spectrum Analyzer on page 472*** for more information.
- Calculate Spectrum
  - Off - FFTs are not being done in the background which makes it faster.
  - On - Collect data while the Scope is running and then switch to FFT scope to view or save the spectrum. The spectrum acquired by Calculate Spectrum will be available in the Memory List.

Controls

- **Save to Memory**: Saves the current measurement waveform to the Memory List. The data will appear in the Memory List as “**Oscilloscope Waveform [L]**”. Change the file name by selecting “Rename“ from the **Memory** drop-down list in the Memory List.

Right-click Functions

- Refer to ***Right-click Functions on page 474***

Real Time Analyzer

The *Real Time Analyzer* (*RTA*) allows you to analyze a signal using **Constant-Percentage Bandwidth** (1-Nth octave) filters. This method of frequency analysis is inherently different than using the *FFT*. The *FFT* approach operates on a whole block of time data; e.g., a time block of 100 ms in length. The recursive digital filtering utilized in the *RTA* is a continuous process. For every input (each time sample from the audio interface), an output data value is obtained. This way, the *RTA* functions like a bank of analog 1-Nth octave filters that are wired in parallel.

An *RTA* should be used when

- Analyzing non-linear audio devices (such as cell phones)
- Using complex signals (such as simulated or actual speech)
- Testing devices using pink noise or tone bursts

This algorithm conforms to the **ANSI S1.11 - 2004 class 0** standard.

Note: As of SoundCheck 10.0, the RTA instrument is compatible with NI DAQmx devices.

Note: When using a National Instruments DAQmx interface hardware you cannot open Multimeter and RTA instruments simultaneously.

To use the *RTA*, select **Real Time Analyzer** (**Ctrl+F8**) from the **Instruments** drop-down list.

The *RTA* will start as soon as it opens. The *RTA* wakes up in the same mode it was last used (e.g., Exp averaging with a Slow time constant using 1/3 octave filters).

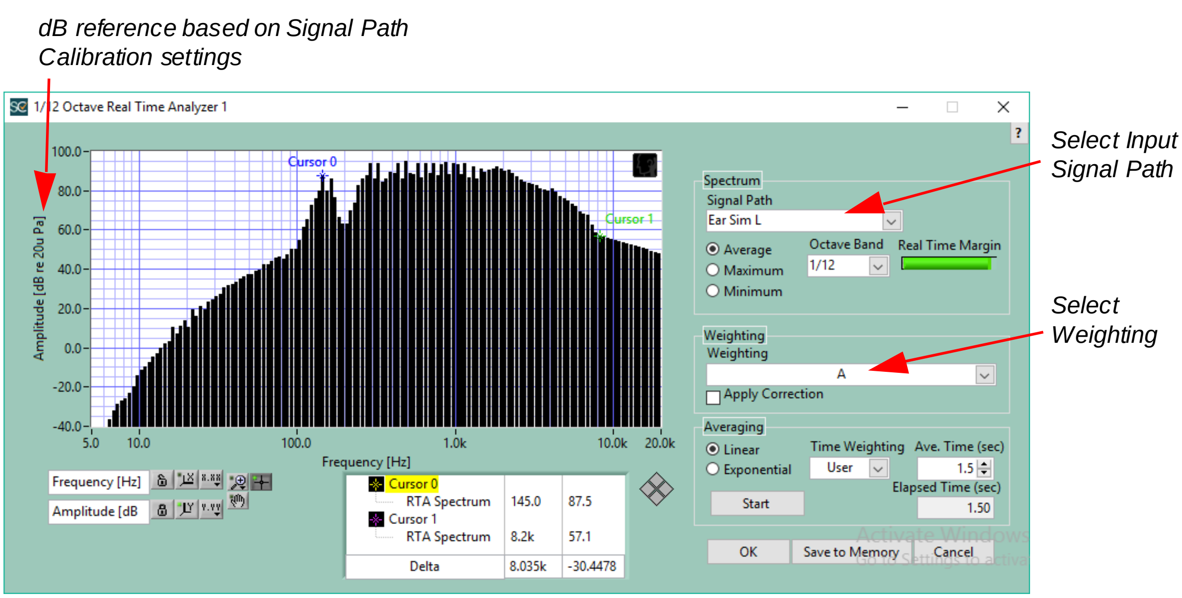


Figure 29-33: 1/Nth-Octave Real Time Analyzer

Spectrum

Octave Band (was Filter Width)

The *RTA* has 1/1, 1/3, 1/6, 1/12, and 1/24 octave digital recursive filters. The upper frequency range is based on the audio interface's sampling frequency. The highest frequency that can be measured will be no more than one-half the audio interface sampling rate (F_{sample}). To measure beyond 22 kHz, choose a sampling rate higher than 44.1 kHz.

Because of the different filter widths, the highest filter that is displayed will typically be lower than $F_{sample}/2$. For example, the highest 1/3 octave filter that can be used for a 44.1 kHz sampling rate is 16 kHz. To measure closer to the actual upper limit of 22 kHz, you must use filters that are narrower (e.g., 1/12 or 1/24 octave).

The following controls operate the same as in the *Spectrum Analyzer*. Refer to ***Spectrum Analyzer on page 472*** for more information.

- Input Signal Path – Any predefined channel will be available for input
- Average
- Maximum
- Minimum
- Buffer Use
- Real Time Indicator
- Weighting
- Apply Correction

Averaging

- **Linear**: Linear will average for only the number of averages entered (e.g., 13) and then stops measuring. When Linear averaging is selected, the **Averages** indicator will appear, specifying the number of averages currently completed.
- **Exponential**: Exponential averaging is a continuous process. It is equivalent to a running average. As the averaging time gets longer, the response of the filters slows down.
- **Choice of Averaging Time**: Fast (250 ms), Slow (1 s), User Defined (s).

Controls

- **Save to Memory**: Saves the current measurement data to the Memory List. The curve will appear in the Memory List as "**1/X Octave RTA [L]**". (1/X changes depending on the resolution of the measurement.) Change the file name by Selecting "Rename" from the **Memory** drop-down list in the Memory List.
- The standard Scale Legend, Graph Palette and Cursor Legend appear below the graph. See ***Axis Scaling, Zoom and Style Controls on page 473*** for more information.

Right-click Functions

- Refer to ***Right-click Functions on page 474***

Distortion Analyzer

The Distortion Analyzer measures the distortion or distortion and noise characteristics of the signal on the selected signal path.

Measurement tab controls (Meas.)

Signal Path control - Select any input signal path available in the Calibration Configuration

Apply Correction control - Select to apply input correction is applied

Measurement Type

Select the type of distortion to be measured and then select the appropriate harmonics from the Distortion (Dist.) Tab. See ***Figure 29-34***.

- THD Ratio IEEE - Same as in HarmonicTrak
- THD Ratio IEC - Same as in HarmonicTrak

  See ***Total Harmonic Distortion on page 188***

- THD Residual - The power sum of the harmonics selected in the Dist. Tab.
- THD+N Ratio IEEE - Same as in HarmonicTrak
- THD+N Ratio IEC - Same as in HarmonicTrak

  See ***THD + Noise on page 189***

- THD+N Residual - The level of all the noise and distortion products in the measurement bandwidth
- SINAD - Is the reciprocal of THD+N, if and only if THD+N is calculated without High and Low Pass filters in the Analysis Editor

  See ***Virtual Instrument THD+N Options on page 190***

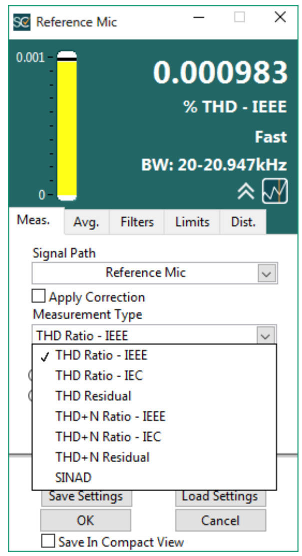


Figure 29-34: Meas. Tab

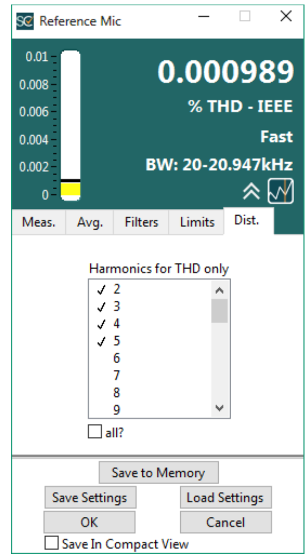


Figure 29-34: Dist. Tab

Averaging Tab (Avg.)

Averaging Type

When setting the averaging time, be aware that averaging times greater than 250 mSec my be required to produce repeatable THD+N measurements.

When selecting Continuous - Moving Avg, the default averaging time is Fast (250 mSec).

When selecting Linear or Linear - Repeating, the default averaging time is 1 second as shown in ***Figure 29-35***.

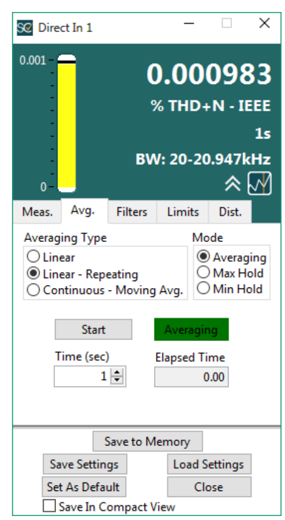


Figure 29-35: Avg. Tab

The remaining tab controls are the same as the Multimeter.

See ***Averaging Tab (Avg.) on page 467***, ***Filters Tab on page 468*** and ***Limits Tab on page 468***.

For more details on THD+N Optimized, refer to ***THD + Noise on page 189***.

Frequency Counter

The Frequency Counter returns a precision frequency measurement of the dominant signal in the selected signal path.

Combined with the Strip Chart recorder this can be used to determine if the device under test is playing back audio at a constant rate.

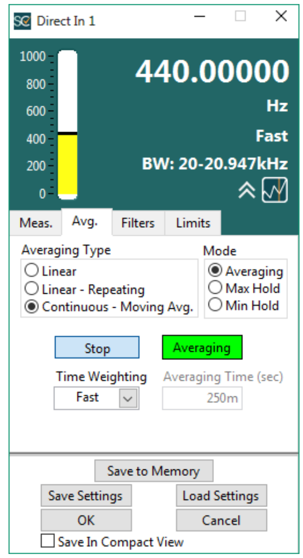


Figure 29-36: Frequency Counter

The remaining tab controls are the same as the Multimeter. See ***Averaging Tab (Avg.) on page 467***, ***Filters Tab on page 468*** and ***Limits Tab on page 468***.

SoundCheck ONE™

Introduction

SoundCheck ONE is an entry-level SoundCheck system which is essentially a lower cost, simplified, version of SoundCheck coupled with the AmpConnect ISC or AudioConnect. SoundCheck ONE offers the capability to test loudspeakers, microphones and headphones within predetermined sequence templates.

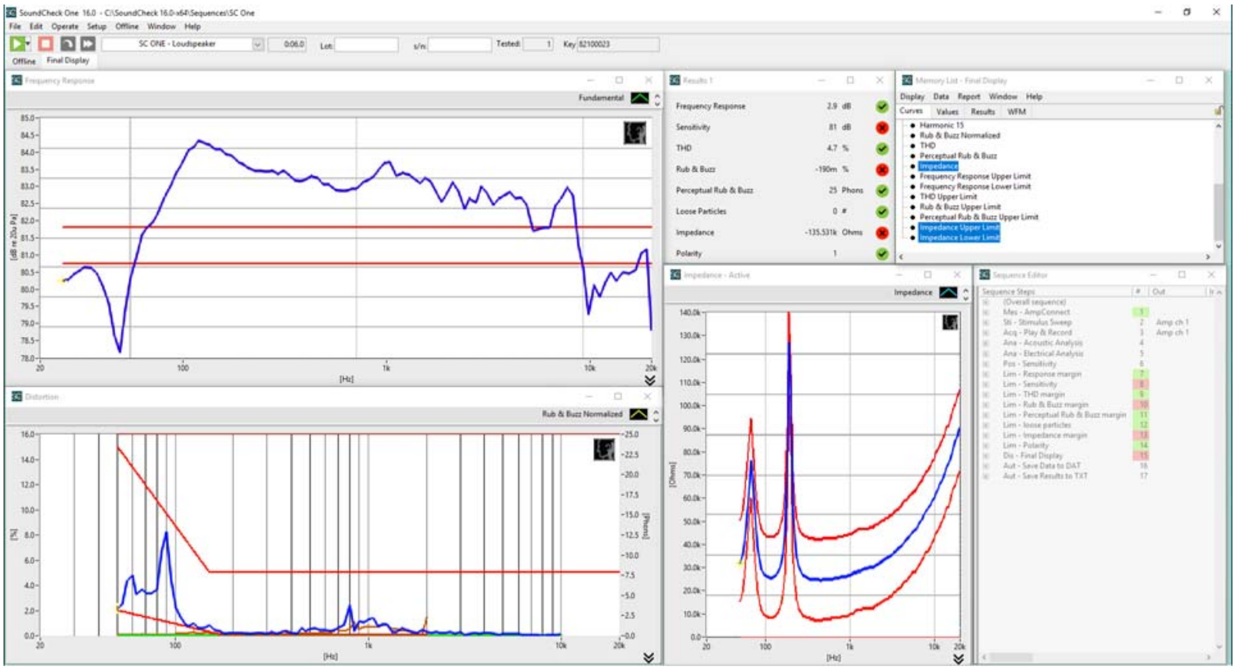


Figure 30-1: Final Display

Although the user interface is the same as in the full version of SoundCheck, rather than using the Sequence Editor, SoundCheck ONE users are supplied with sequence templates. These templates serve as the starting point for all SoundCheck ONE tests and can be used to generate as many product specific sequences as desired by selecting parameters such as the stimulus signal, characteristics to be measured, frequency range, level and limits.

SoundCheck ONE is aimed at customers who do not own a full version of SoundCheck and need a low cost and easy to set up system for basic production line tests of loudspeakers, microphones or headphones. While it offers the same accuracy, advanced algorithms and speed as the regular version of SoundCheck, its flexibility and test customization capabilities are restricted. It is a good entry point for a company testing their products for the first time or moving up from a more rudimentary test system. It can be upgraded to the full version of SoundCheck at any time for an additional fee.

Setup Wizard

The Setup Wizard runs when you start SoundCheck ONE. You can check “Do not show this dialog again” to stop the wizard from running at each startup. See ***Setup Wizard on page 35*** for details.

As of SoundCheck 18, hardware setup is simplified by "**Automatically Create Signal Paths for Listen Devices**" in **Preferences > Launch**. AmpConnect ISC or AudioConnect are detected when SoundCheck starts and Vp values are loaded automatically from the connected Listen Hardware device.

(SoundCheck ONE requires either AmpConnect ISC or AudioConnect.)

AudioConnect Hardware Setup

Before calibrating the reference mic and/or before the first run of the sequence, open the Hardware Editor and select the Listen Hardware Tab. (This requires that the user has signed in with Engineer or Technician Access Level.)

Right-click AudioConnect Device ID and select **Assign Startup Default**. Set the channels as instructed in the Sequence Template - Sequence Note you are using.

Typical settings for each template:

Loudspeaker

- Inputs: Channel 1 Mic In, Channel 2 Line In
- Mic Bias: On for SCM microphone

Microphone

- Inputs: Channel 1 Mic In, Channel 2 Line In
- Mic Bias: On for SCM microphone

Headphone

- Inputs: Channel 1 and 2 Line In
- Mic Bias: Off

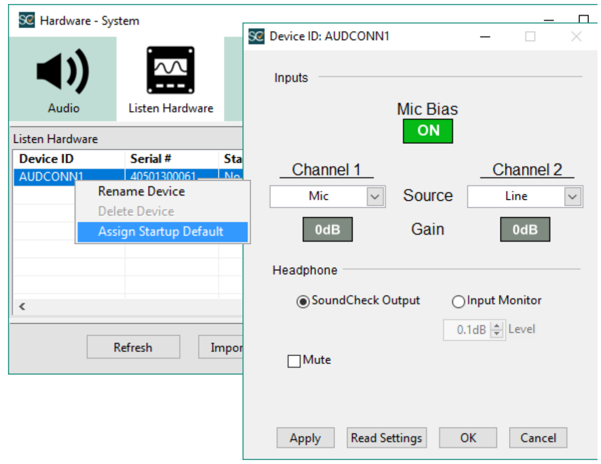


Figure 30-2: Assign Startup Default

Template Sequences

SoundCheck ONE template sequences are used to create customized sequences that are specific to a given product. The provided templates serve as a starting point, containing all the necessary steps to perform the essential measurements for their test application.

The basic process of using SoundCheck ONE:

- Select the template for your interface: AmpConnect ISC or AudioConnect
- Choose the appropriate template for the application
- Modify any necessary settings
- Save the template as a new sequence, e.g.: “Product Name rev ##.sqc”
- There is no limit as to how many custom sequences can be saved from these templates

The typical setup for a SoundCheck ONE system is to have a separate test sequence for each product model that will be tested. Each sequence can have its own unique settings such as: stimulus range/level, tolerance limits, graphical displays, and data saving. Unlike the full version of SoundCheck, sequences in SoundCheck ONE cannot have their steps and layout modified, however the settings within the steps can be changed.

AmpConnect/AudioConnect Self Test for SC ONE

Included in the SC ONE sequence folder are Self Test sequences for AmpConnect ISC and AudioConnect. These are used to verify the operation of the hardware. They should not to be used as template sequences.

Setup & Calibration

The required calibration will depend on which application is being tested. Choose from the three sub-headings below:

Loudspeaker:

AmpConnect: The calibration for the AmpConnect power amplifier is fixed, so no additional steps are required.

AudioConnect: Calibrate the power amplifier according to the instructions in ***Amplifier Calibration Procedure on page 100***.

The Reference Microphone should be calibrated in either case.

See ***Microphone Calibration Procedure on page 96***.

> ***Prior to calibrating the microphone, the gain of the AmpConnect/AudioConnect Reference Mic channel must match the gain used in the AmpConnect/AudioConnect Message Step of the test sequence. See Input Hardware Channel on page 85.***

Microphone:

The SC ONE Microphone sequence uses a substitution method to account for the response of the reference speaker.

1. Calibrate the Reference Microphone. See ***Microphone Calibration Procedure on page 96***.
2. Open the sequence, C:\SoundCheck 18.1\Sequences\SC ONE\SC ONE Microphone (Measure Reference).sqc
3. Position the reference microphone at the test point in front of the reference speaker
4. Run the sequence to measure and record the response of the speaker
5. Remove the reference microphone and replace it with the DUT
6. Open and run the SC ONE Microphone sequence

Note: *These steps should be run at a regular interval to ensure accurate calibration of the reference speaker as environmental conditions change.*

Headphones:

AmpConnect ISC: The headphone amplifier provides a unity gain output, so no calibration is required for the output. The ear simulators or couplers (signal paths **“Ear Sim L”** and **“Ear Sim R”**) should be calibrated according to the instructions in ***Microphone Calibration Procedure on page 96***.

AudioConnect: The headphone amp (or external power amp) should be calibrated to account for gain and frequency response. In this case, the two channels should be calibrated independently for more accurate results. Use the Headphone Amplifier Calibration sequence and follow the instructions for **Headphone Amplifier Calibration** in the AudioConnect Manual.

Generating SoundCheck ONE Sequences

1. Refer to the sequence note PDF files for the SoundCheck ONE sequences, included in the Sequences > SC ONE folder, for complete instructions on setup and use.

2. Open one of the three template sequences which will serve as the starting point (Loudspeaker, Microphone or Headphones). There are versions for AudioConnect and AmpConnect ISC, named accordingly. "**Microphone (Measure Reference)**" is used to store the Reference Microphone Fundamental and Sensitivity, before using the "**Microphone**" measurement sequence.

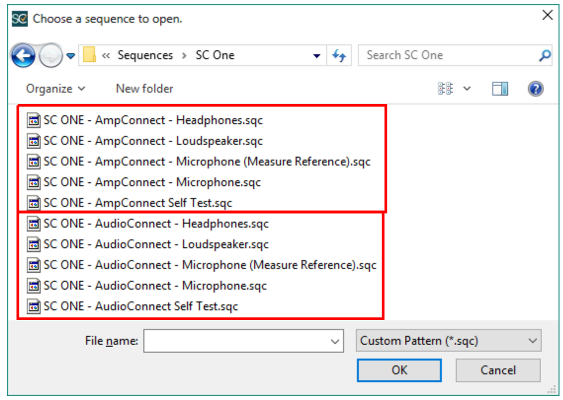


Figure 30-3: Template Sequences

Important!: All sequences must be generated from one of these templates. Sequences cannot be created from scratch.

---

3. Use **File > Save As** to save a copy of the sequence with a new name. The new sequence file can be stored anywhere on the system. It does not need to be stored in the same folder as the template.

Note: ***The templates are Read-only files and cannot be overwritten.***

---

4. To modify sequence parameters click **Setup** from the SoundCheck Main Screen then choose the step category that you would like to modify (See the examples below).

Note: *The Sequence Editor can be opened to watch the progress of a sequence, but steps can only be edited through the **Setup** drop-down list.*

---

5. Click the green start button on the toolbar to run the sequence and see the result.

6. When finished making changes save the sequence (**File > Save**) to store your changes to disk.

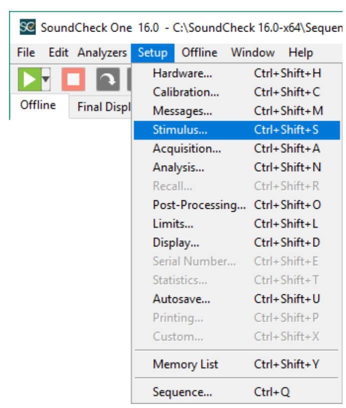


Figure 30-4: Setup Drop-down List

Note: *For more information on the Template Sequences, please refer to the sequence note: "SoundCheck ONE Templates" found in the SoundCheck ONE sequence folder: C:\SoundCheck 18.1\Sequences\SC One.*

Sequence Editing

- The Sequence Editor is not used. All steps are grayed out. You can use it to view the progress of a sequence.
- All steps are accessed by using the **Setup** drop-down list on the SoundCheck Main Screen as noted in Step 3 of ***Generating SoundCheck ONE Sequences on page 488***
- Steps cannot be added or removed from a sequence
- Step parameters can be modified
- Breakpoints cannot be added to a sequence

Stimulus

To edit the stimulus level and frequency range in a SoundCheck ONE sequence click **Setup > Stimulus**. This will open the stimulus editor. Make the desired changes then click 'OK'. The changes will be saved to disk when the sequence is saved.

For details on Stimulus Step editing see ***Stimulus Editor on page 113***.

> ***Note:*** **SoundCheck ONE uses only Frequency stepped-sine sweep (Stweep™). No other stimulus types are available. See** ***Frequency Stepped Sweep (Stweep™) Excitation Signal Parameters on page 113.***

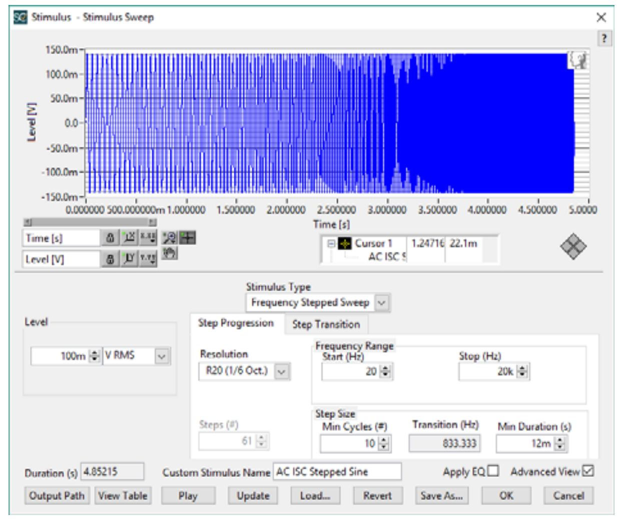


Figure 30-5: Stimulus Step

Limits

To edit the limits in a SoundCheck ONE sequence click **Setup > Limits**. You will see a list of all the different limit steps in the sequence. Choose the one you would like to edit, and click 'OK'. The editor will open where you can make any desired changes then click 'OK'. The changes will be saved to disk when the sequence is saved.

For details on Limit Step editing see ***Limits Editor on page 299***.

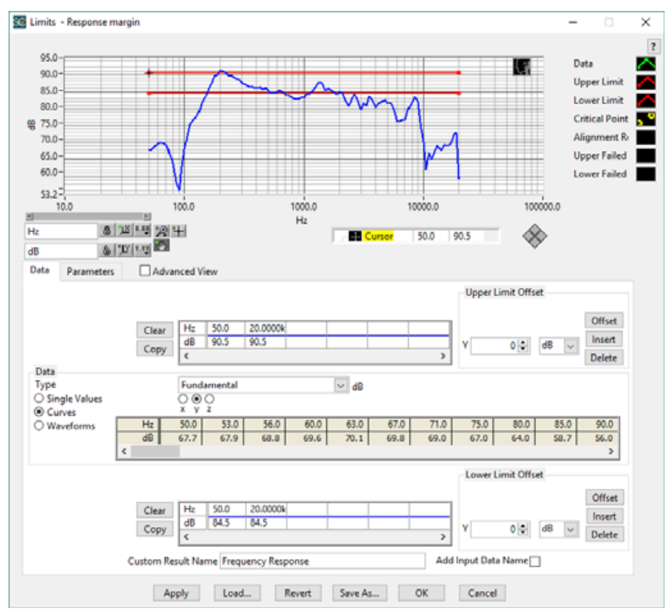


Figure 30-6: Response Limits Step

Display

To edit the display in a SoundCheck ONE sequence, first click the tab for 'Final Display'. The windows for the sequence display will then open.

For details on Display editing see ***Display Editor and Memory List on page 321***.

Add or remove windows, modify what data is shown via the Memory List, and change display preferences of the individual windows. The entire display is configurable.

Once all necessary changes have been made click **File > Save** to save the sequence.

Note: *If you use both SoundCheck ONE and SoundCheck full version, you can switch between the two versions by selecting the appropriate status.dat file under preferences. See* ***Folder Paths on page 49****. SoundCheck ONE operation mode is shown on the Main Screen title bar. The desktop wallpaper is the same as the full version of SoundCheck.*

Important!: If you save a sequence in SoundCheck Full Version, it cannot be opened in SoundCheck ONE.

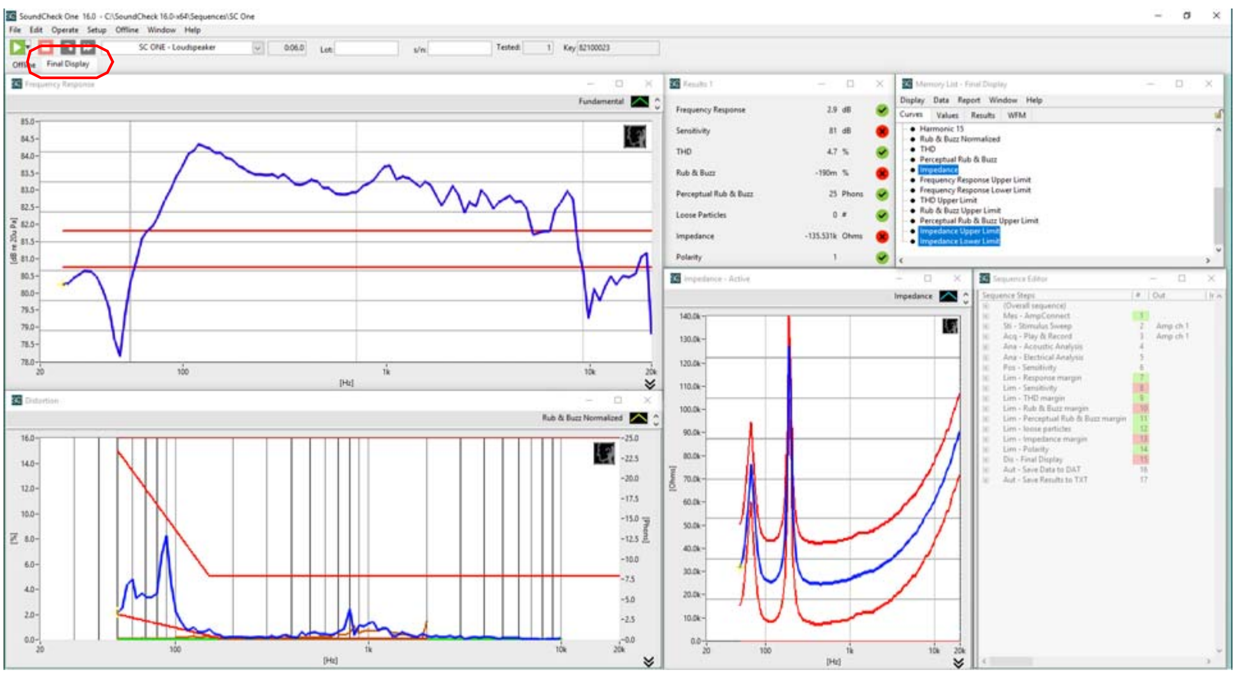


Figure 30-7: Final Display

Controlling SoundCheck with TCP/IP

Overview

As of SoundCheck 15, you can control SoundCheck through TCP/IP.

External control of SoundCheck is simpler, yet more powerful with the new TCP/IP control. This offers many advantages over the previous ActiveX controls (still available), such as the ability to connect to SoundCheck via any programming language, on any operating system, either locally or through a network. It also features a more powerful and expandable command format for interacting with SoundCheck. This is extremely valuable for anyone who needs to control SoundCheck from an external program, for example as part of an overall test plan or factory automation. For example, one computer can control multiple SoundCheck systems, simplifying production line measurements. This helps when integrating with LabVIEW Test Stand.

TCP/IP control uses a JSON data format so that commands return information in a format that is easily parsed by any programming language.

- You can receive test data and results
- SoundCheck sequences can be opened and run through this API, but not modified
- Only one connection to SoundCheck can be used at any time

Setup

- Click **Edit** on the Main Screen and select **Preferences**
- Select the Advanced tab
- Check Enable TCP/IP Server. This automatically updates the SoundCheck *18*.ini file with the Enable status and Port #.

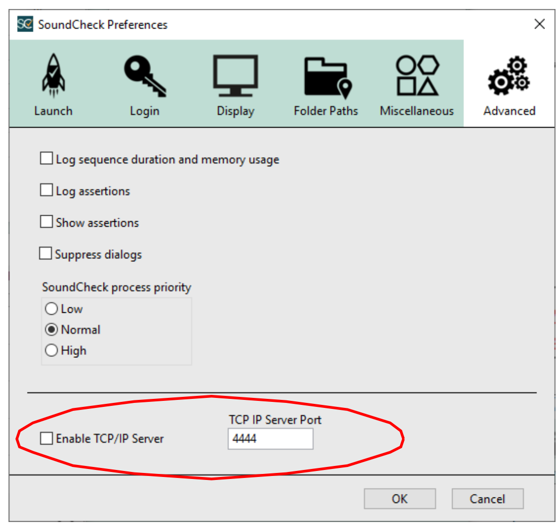


Figure 31-1: Advanced Tab

Windows Security

The first time you enable TCP/IP in SoundCheck Preferences, you may get a Windows Security Alert prompting you to allow SoundCheck to communicate on the selected network.

Click **Allow Access** and continue.

Manual Setup

The “Enable TCP/IP” and “Port #” settings are stored in the **SoundCheck *18*.ini** file found in the root of the SoundCheck folder.

[External Control]

- TCP IP SERVER ENABLED = TRUE or FALSE
- TCP IP SERVER PORT # = 4444

These settings can be modified manually if necessary.

Suppress Dialogs

For unattended operation of SoundCheck the **SUPPRESS DIALOGS** function can be enable in **Preferences** > **Advanced**. See ***Advanced on page 51*** info on Preferences.

For example:

When autosaving a file, SoundCheck may prompt, “Do you want to overwrite the existing file?”.

or when changing sequences, SoundCheck may prompt, “Do you want to Protect or Discard data?”

These Dialog Messages will interrupt TCP IP communication with SoundCheck.

Setting Suppress Dialogs to True allows you to disregard all dialog prompts from SoundCheck.

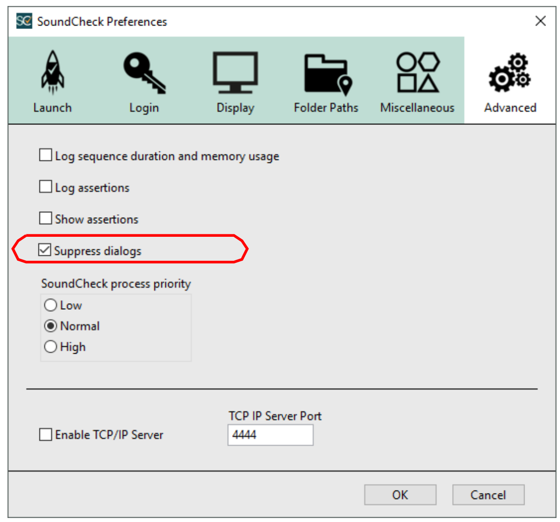


Figure 31-2: Suppress Dialogs

Server IP address

If running on the local computer this is “127.0.0.1” or “localhost”. If accessing a computer on the network, it is the IP address of the target PC, e.g.: 192.168.0.107. You will need to know the IP address of the computers running SoundCheck in order to control them over a network.

Instrument Open Close Custom Step

When calling the new custom step from TCP IP the command should be formatted as:

- Instruments.OpenVICFile(path, mute?)
- Instruments.CloseAll()

Note: When you log in to SoundCheck as “Operator” (***See “Login” on page 57.***), clicking on anything in SoundCheck that initiates a modal Popup Window will stop TCP IP communications. This includes clicking on Displays and Main Menu buttons.

MemoryList.Set Command

Sequence Parameters

As of SoundCheck 18, the **Sequence Parameters** section found in **Sequence Editor > Configure Sequence** will allow you to create Memory List items that will be populated with data using the “**MemoryList.Set**“ command.

The example to the right shows how Curves and Values created in the **Sequence Parameters** section show up in the Memory List as empty items.

Note: **Configure Sequence > Clear measured data** must be unchecked to avoid clearing the data from MemoryList.Set.

Available Sequence Parameters

- Curves
- Values
- Waveforms

Using this method allows you to create data placeholders that are available as soon as the sequence is opened.

This can be used to update stimulus levels or limits without having to open the sequence.

MemoryList.Set TCP/IP Command

This command allows you to easily write data from an external control program to the SoundCheck Memory List for use in sequences.

The **MemoryList.Set** TCP/IP command will fill existing Curves, Values, Results and Waveforms in the Memory List with data. The data is passed to SoundCheck via TCP/IP.

This command is the functional opposite of MemoryList.Get.

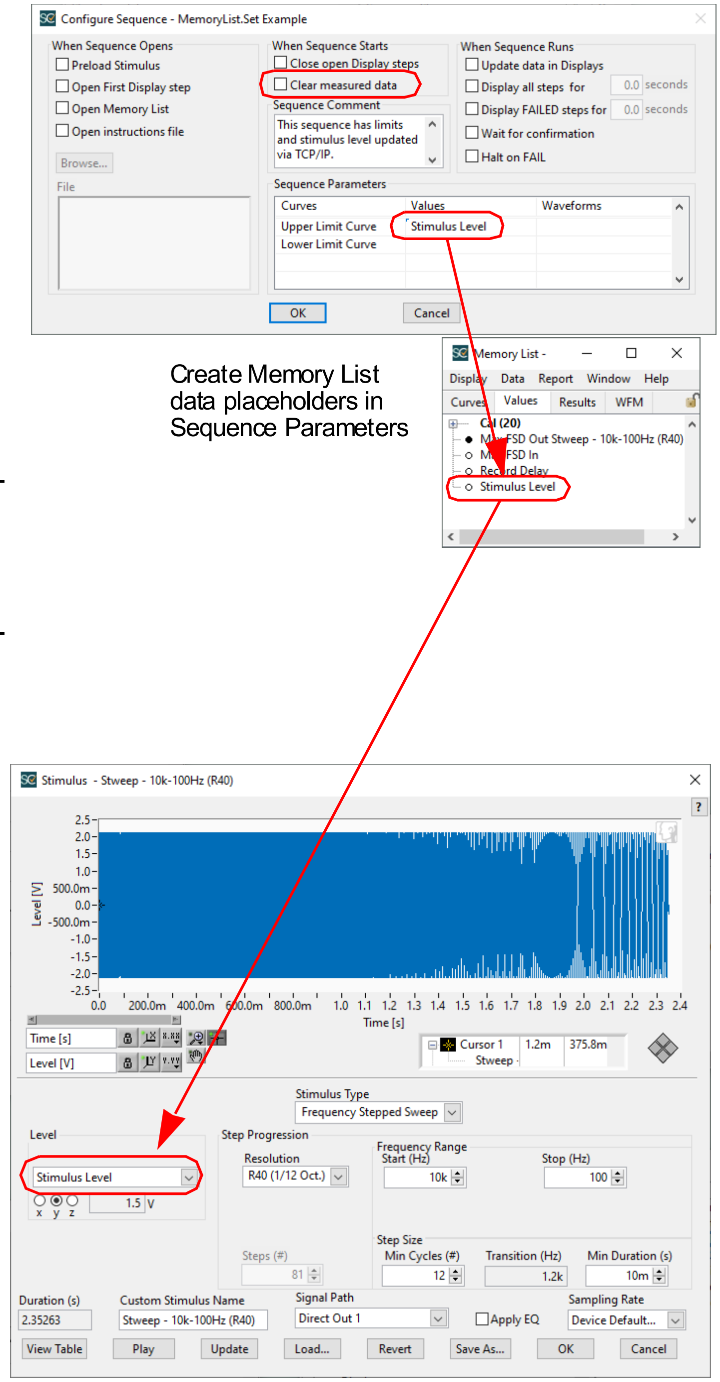


Figure 31-3: Sequence Parameters For MemoryList.Set

Controlling SoundCheck with TELNET

Windows Telnet Setup

Windows 10 no longer includes Telnet Server by default as Telnet is considered an insecure protocol. In order to use Telnet, Port 23 must be open on the firewall. In general, Telnet is not be required in order to use TCP IP. These instructions are included for legacy support purposes.

Windows users will need to enable Telnet. Telnet is enabled by default on Mac computers.

1. Click Start > Control Panel
2. Click Programs and Features
3. Click Turn Windows features on or off
4. In the Windows Features dialog box, check the **Telnet Client** check box
5. Click **OK**. The system installs the appropriate files. This will take a few seconds to a minute.

Using Telnet

1. Edit SoundCheck Preferences. See ***Setupon page 491***.
2. Open SoundCheck 18.1
3. **macOS**
   - Use the Terminal editor
   - enter: "**telnet localhost 4444**" and hit Return

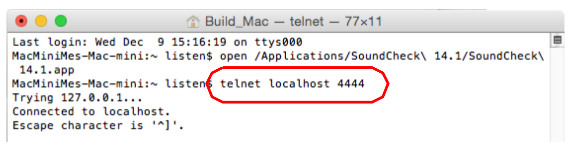


Figure 31-4: Mac Telnet Editor

   Windows OS
   - Open a Command Line window
   - enter: "**telnet localhost 4444**" and hit Return

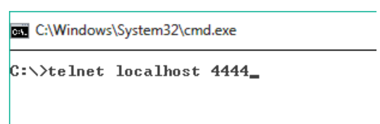


Figure 31-5: Windows Telnet

4. "**Connected to SoundCheck**" confirms that the connection to SoundCheck is complete.

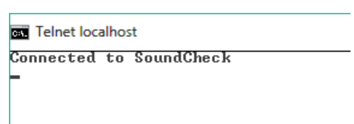


Figure 31-6: Connected

Then it is a matter of entering commands with the proper syntax and then executing them. See ***Command Set Definition on page 500*** & ***Command List and Return Format on page 501***.

Run a Sequence and Retrieve Data

Now that you have connected to SoundCheck through Telnet, you can start to execute commands.

The following example covers manually opening a sequence, running a sequence from Telnet and retrieving results.

1. From the SoundCheck File menu, open the **Calibration/Self Test.sqc** file.

   (Later you'll learn how to do this from Telnet as well.)

2. Open Telnet as shown in ***Using Telnet on page 494***.

   Telnet should show: **Connected to SoundCheck**

3. To Run the sequence, type the following then hit Enter/Return:

   Sequence.Run

   You will need to click on the SoundCheck window and answer any prompts from the sequence.

4. Telnet returns a JSON response:

   {"cmdCompleted":true,"returnData":...

   The response format is defined in ***Command List and Return Format on page 501***.

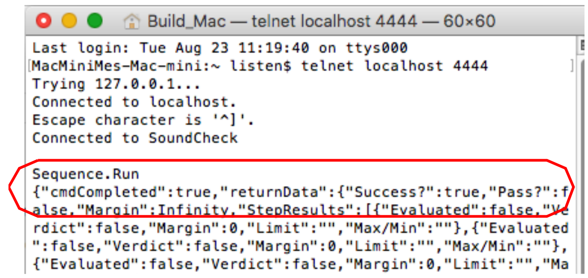


Figure 31-7: Sequence.Run Example

5. After the sequence has run you can retrieve a curve using: **Command:** MemoryList.Get('Curve', 'Parameter'). See ***on page 504***.

6. Type the following and hit Enter/Return:

   MemoryList.Get('Curve', 'Fundamental [Direct In 1]')

   The JSON response is:

   {"cmdCompleted":true,"returnType":"MemoryListCurve","returnData":{"Found":true,"Curve":{"Name":"Fundamental [Direct In1]","XData":[20000,16000,...,31.5,25,20],"Data":[0.3442260151137711,...,3.0393952906618722],"ZData":"","XUnit":"Hz","YUnit":"V/V","ZUnit":"deg.","XDataScale":"Lin","YDataScale":"dB","ZDataScale":"Lin","XdBRef":1,"YdBRef":1,"ZdBRef":1,"XAxisScale":"Log","YAxisScale":"Lin","ZAxisScale":"Lin","Protected":false}},"errorType":"","errorDescription":"","originalCommand":"MemoryList.Get('Curve', 'Fundamental [Direct In 1]')","resolvedCommand":"MemoryList- Get('Curve', 'Fundamental [Direct In 1]')"}

Every command sent via TCP/IP gets back a JSON formatted text string. The definitions for these are found in ***Command Returns on page 500***.

C# Example App

This shows what can be created using C# and how it can be used to control SoundCheck.

The example executable is in the main SoundCheck folder:

C:\SoundCheck 18\External Control Examples\C Sharp\Application\C Sharp Example.exe.

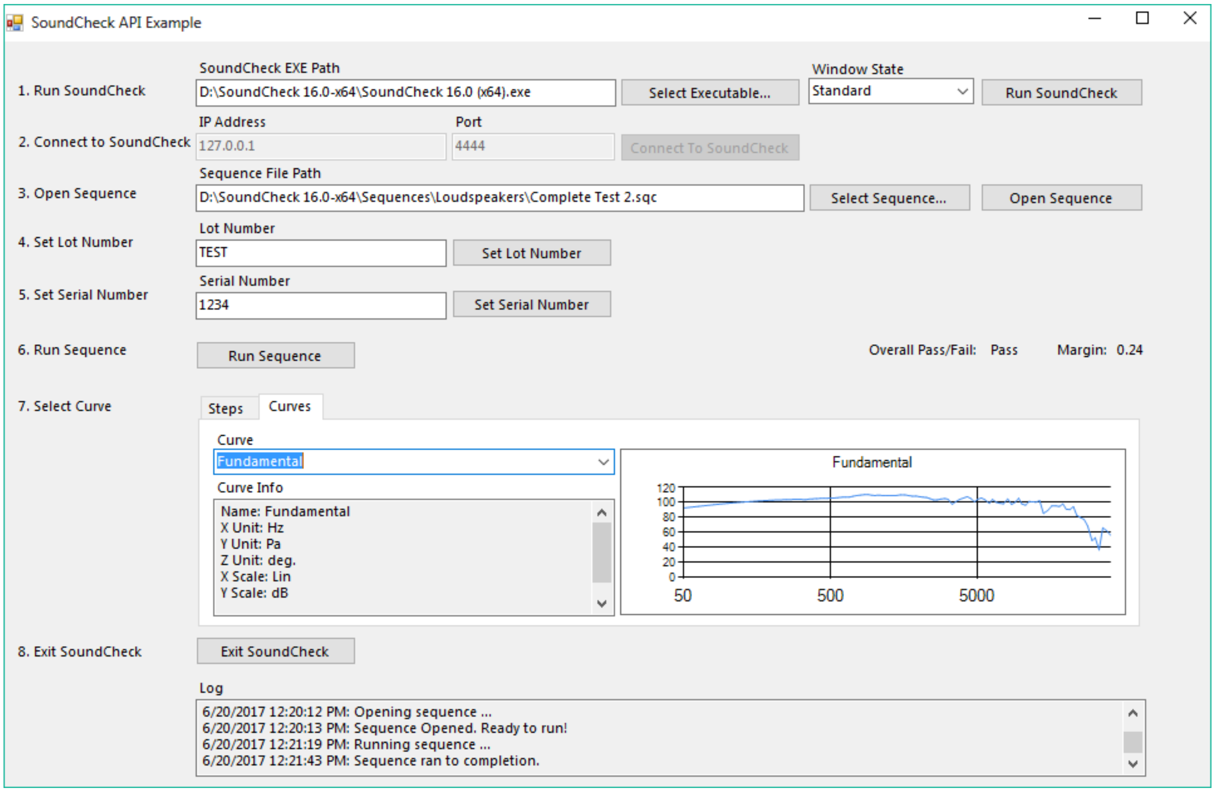


Figure 31-8: C# API Example

1. Run SoundCheck - Click “Select Executable” and navigate to: **C:\SoundCheck 18\SoundCheck 18.exe**. Click “Run SoundCheck”. You can also have SoundCheck running and then open the C Sharp Example.exe.

   Window State - Allows you to select how SoundCheck opens: Standard, Hidden or Minimized.

2. Connect to SoundCheck - Enter the IP Address and Port number of the PC to control, e.g.: **127.0.0.1 and 4444**. Click “**Connect to SoundCheck**”.

   These fields are disabled as soon as the app has connected to SoundCheck, and enabled again if it is determined that the connection has been lost.

3. Open Sequence - Click “**Select Sequence**” and navigate to your sequence folder and select a sequence, e.g.: **Complete test.sqc**

   Click “**Open Sequence**” to load it into SoundCheck.

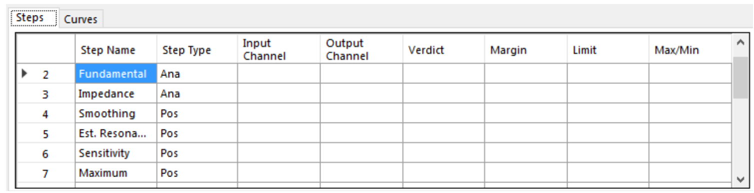
4. Set Lot Number - Click “**Set Lot Number**” to send data to the Lot Number field on the SoundCheck Main Toolbar.

5. Set Serial Number - Click “**Set Serial Number**” to send data to SoundCheck

6. Click “**Run Sequence**”

7. **Curves** - Allows you to query curves from the Memory List. This is only a preview window. It does not allow you to view multiple curves with limits.

Steps - Shows the order of steps in the sequence along with channel settings and Limits results.



8. Click "**Exit SoundCheck**" when finished

Log Window - This shows all of the activity occurring with the C Sharp App.

LabVIEW Example App

The example app shows what can be created in *LabVIEW 2018* to control SoundCheck.

It also works as a control panel to test the operation of the available commands.

The example executable is in the main SoundCheck folder:

C:\SoundCheck 18\External Control Examples\LabVIEW\Application\ **LabVIEW Example.exe**

The LabVIEW example uses the same Command Set Definition as the C# Sharp example. See ***Command Set Definition on page 500*** and ***Command List and Return Format on page 501***.

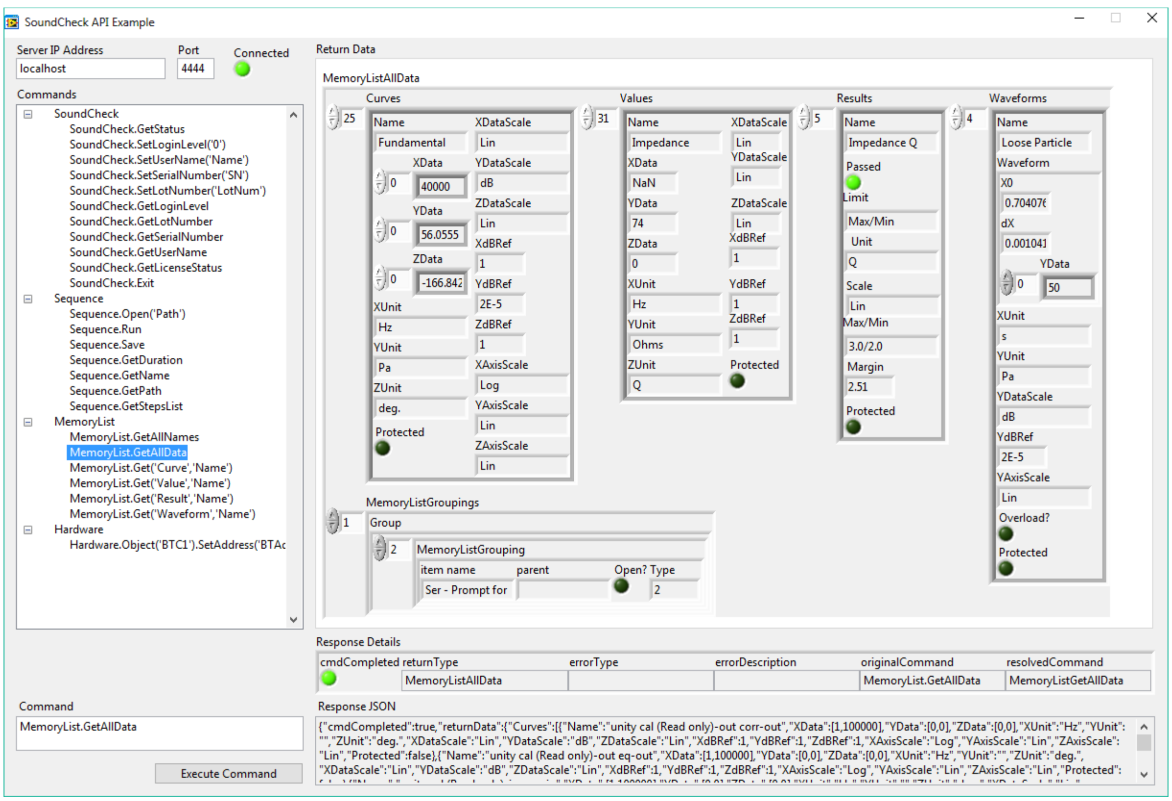


Figure 31-9: LabVIEW API Example

Server IP address: If running on the local computer this is "127.0.0.1" or "localhost". If accessing a computer on the network, it is the IP address of the target PC, e.g.: 192.168.0.123. You will need to know the IP address of the computers running SoundCheck in order to control them over a network.

Port: Port number of the selected SoundCheck system

Connected: Shows that the selected system is responding

Commands: List of available commands and the proper syntax to use when entering the command

Command: Enter a command from the list and click Execute Command

Return Data: Defines the format for each of the listed data types

It shows information based on the most recently run MemoryList Command, e.g.: "MemoryList.GetAllData" shows the data fields in ***Figure 31-9***. The Up/Down arrows to the left of each data type allow you to cycle through the data from the sequence run.

Response Details: Shows if a command completed and the type of data returned as well as showing details on errors.

Response JSON: Shows the data contents of the Response JSON. See ***Command Returns on page 500***.

Each data return type has a uniform standard which is defined in ***Command List and Return Format on page 501***.

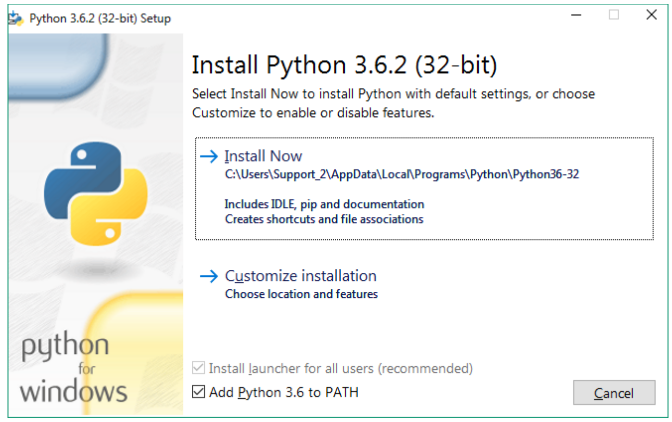
LabVIEW Return Library

Located in C:\SoundCheck 18\API\LabVIEW Return Library

This collection of VIs is included to help you get data out of the JSON Response.

Note: LabVIEW 2018 or later is required to use these VIs.

Python Example



Python is an object-oriented scripting language that offers some advantages over C++.

The example included with SoundCheck shows off the simplicity of SoundCheck control via Python.

- Compatible with Python 2 and 3.
- See ***SoundCheck TCP/IP Python Library (soundcheck_tcpip) on page 511***

***Important:* Before running any of the example scripts included with SoundCheck, please follow the Setup Notes in the Readme.txt file found in: <SoundCheck root>\External Control Examples\Python.**

The example script included with SoundCheck will:

- Open SoundCheck
- Automatically run the “Complete Test” sequence
- After the sequence is complete, results are passed back to the python script for further processing

Requirements

Before running the Python script you will need to:

- Uncheck “**Run Setup Wizard**” in Preferences > Startup
- Uncheck “**Show Quick Launch**” in Preferences > Startup
- Uncheck “**Show Login Window on Startup**” in Preferences > Login
- Check “**Enable TCP/IP Server**” in Preferences > Advanced
- See ***Preferences on page 45*** for more information

Example

Set the drive letter to the root where the python script is stored. In this case, **C:.**

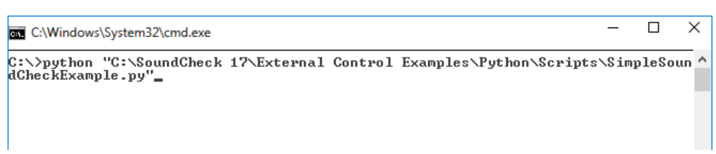
From a Command Line, run the Python script by calling:

python “<**SoundCheck root**>\External Control Examples\Python\Scripts\SimpleSoundCheckExample.py"

e.g.:

python “C:\SoundCheck 18\External Control Examples\Python\Scripts\SimpleSoundCheckExample.py”

The quotes must enclose the path to the python script.



Command Set Definition

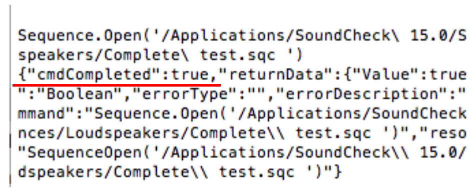
The commands sent to SoundCheck via TCP/IP must conform to these rules. This is used in C Sharp, Telnet or any other communication method, via TCP/IP.

- SoundCheck commands are built of period (.) separated command segments, where each segment may take parameters
- All parameters must be enclosed by single-quotation marks, a.k.a. apostrophe ('). This includes both string and numeric values.
- All commands must be completed with the \r\n (Carriage Return + Line Feed) sequence. You don't need to add these when using Telnet or any other terminal app that automatically adds Carriage Return & Line Feed.

  Please note that you will need to do this only when programming your own TCP interface for sending commands to SoundCheck.
- The parentheses after a command are only needed if the command needs parameters. If there are no parameters associated with a command or command segment, and parentheses are added anyway, SoundCheck will ignore them.
- Commands ARE case sensitive

Command Returns

Every command sent gets back a JSON formatted text string with the following fields:



- cmdCompleted: true or false

  True indicates that the command was recognized as a valid SoundCheck command and was executed to completion. If the command is not recognized, it times out, or there is an error while trying to execute the command, cmdCompleted will be false. It does not indicate what the result was, e.g.: **Sequence.Save** - cmdCompleted = true can be returned even if the sequence is marked "Read Only" and cannot be saved.
- returnData - This field contains the data that is returned by SoundCheck after executing a command. This may be the result of a query, e.g.: **Sequence.GetName**, or it may indicate if a requested operation successfully completed, e.g.: **SoundCheck.SetSerialNumber**. **returnData** will be different for each command.

  See ***Command List and Return Format on page 501***.
- returnType - This field indicates the data type of the returnData, e.g.: Boolean, String, StringArray, etc.
- errorType - If an error occurs, this field shows the error type, e.g.: Timeout, Unknown Command, etc.
- errorDescription - If an error occurs, this field shows the error description
- originalCommand - This is the original command that was sent to SoundCheck
- resolvedCommand - This is the actual resolved command that SoundCheck executed

  originalCommand and resolvedCommand fields are provided for troubleshooting.

Command List and Return Format

The following legend may be used to determine the data type of a field by examining the value in the example JSON data, returned by SoundCheck API.

For example, if you see “false” in the result, you can interpret that as a Boolean, which may be either false or true.

Data Type	Value
Boolean	false
String	"string"
String (with three possible values: "a", "b", or "c")	"a/b/c"
Number (Integer)	0
Number (Double)	0.1

The following list of commands are currently available for use with SoundCheck. Every command issued gets back a Return. Some Returns show as “Void” which simply means that no data was returned. An acknowledgment is still returned indicating that the command completed.

SoundCheck Commands	
Command/Parameter	Return Data
Command: SoundCheck.SetLoginLevel('Parameter') **Parameter:** Login level (0, 1, or 2) 0 = Engineer, 1 = Technician, 2 = Operator **Description:** Used to set the user login level	Void
Command: SoundCheck.SetUserName('Parameter') **Parameter:** User Name **Description:** Used to set the user name of the currently logged in user	Void
Command: SoundCheck.SetSerialNumber('Parameter') **Parameter:** Serial Number **Description:** Used to set the serial number of the device under test	Void
Command: SoundCheck.SetLotNumber('Parameter') **Parameter:** Lot Number **Description:** Used to set the lot number for a batch of devices to be tested	Void
Command: SoundCheck.GetLoginLevel **Parameter:** None **Description:** Used to get the login level of the currently logged in user. It returns an integer: 0 = Engineer, 1 = Technician, 2 = Operator.	{"Value":0}
Command: SoundCheck.GetLotNumber **Parameter:** None **Description:** Used to get the lot number of the batch of devices currently being tested	{"Value":"String"}
Command: SoundCheck.GetSerialNumber **Parameter:** None **Description:** Used to get the serial number for the device under test	{"Value":"String"}

SoundCheck Commands	
Command/Parameter	Return Data
Command: SoundCheck.GetUserName **Parameter:** None **Description:** Used to get the user name of the currently logged in user	{"Value":"String"}
Command: SoundCheck.GetLicenseStatus **Parameter:** None **Description:** Used to get the license status of SoundCheck. It indicates whether a valid hardware key was found, and if so, what the key ID is.	{"Valid":false,"KeyID":"String"}
Command: SoundCheck.GetStatus **Parameter:** None **Description:** Used to query SoundCheck for its status and determine if SoundCheck is busy, or available to execute a command such as, run a sequence.	{"Busy":false}
Command: SoundCheck.Exit **Parameter:** None **Description:** Used to request SoundCheck to exit.	Void
Command: SoundCheck.SetFloatStrings('Parameter1','Parameter2','Parameter3') **Parameter:** 1. String representing "Not a Number" 2. String representing "Positive Infinity" 3. String representing "Negative Infinity" **Description:** For floating point numbers, SoundCheck uses "NaN" for "Not a Number", "Infinity" for "Positive Infinity", and "-Infinity" for "Negative Infinity". If your programming environment use different strings to represent these values, send this command with all three parameters. For example, for clients written in Python and MATLAB will need to send SetFloatStrings('NaN', 'Inf', '-Inf'). JavaScript does not support any of these values, so the command should be SetFloatStrings('null','null','null'). A client written in any language that uses "NaN", "Infinity", and "-Infinity", which are the same strings that SoundCheck uses, does not need to send the SetFloatStrings command, if it is the only TCP client connecting to SoundCheck.	Void
Command: Instruments.OpenVICFile(path, mute?) **Parameter:** 1. 'x:\file path\VicFile.vic' 2. 'TRUE' or 'FALSE' **Command:** Instruments.CloseAll() **Parameter:** None **Description:** Allows you to open and close virtual instrument configuration files (.VIC)	Void

Sequence Commands	
Command/Parameter	Return Data
Command: Sequence.Open('Parameter') **Parameter:** Sequence Path **Description:** Used to request SoundCheck to open a sequence. If a sequence is already open, SoundCheck may display a dialog to save or discard the existing sequence. That dialog needs to be closed for this command to complete, otherwise it will time out.	{"Value":false}

Command: Sequence.Run or
Sequence.Run('Parameter')
Optional Parameter: Timeout in milliseconds
Description: Used to request SoundCheck to run the currently open sequence. SoundCheck has a default timeout of 5 minutes to run a sequence. If a longer sequence needs to be run, the optional timeout may be included. For example, for a timeout of 10 minutes, the command will be:
Sequence.Run('600000')
	{
	"Success?": false,
	"Pass?": false,
	"Margin": 0.1,
	"StepResults": [{
	"Evaluated": false,
	"Verdict": false,
	"Margin": 0.1,
	"Limit": "String",
	"Max/Min": "String"
	}]
	}

Command: Sequence.Save **Parameter:** None **Description:** Used to request SoundCheck to save the currently open sequence.	{"Value":false}
Command: Sequence.GetDuration **Parameter:** None **Description:** Used to get the duration of the last run sequence.	{"Value":0.1}
Command: Sequence.GetName **Parameter:** None **Description:** Used to get the name of the currently open sequence.	{"Value":"String"}
Command: Sequence.GetPath **Parameter:** None **Description:** Used to get the path of the currently open sequence.	{"Value":"String"}
Command: Sequence.GetStepsList **Parameter:** None **Description:** Used to get a list of all the steps in the currently open sequence.	[{ "Name": "String", "Type": "String", "InputChannelNames": ["String"], "OutputChannelNames": ["String"] }]

<table>
<tr><th colspan="2">Memory List Commands</th></tr>
<tr><td>Command/Parameter</td><td>Return Data</td></tr>
<tr><td>Command: MemoryList.GetAllNames

Parameter: None

Description: Used to get names of all curves, values, results, and waveforms in the Memory List</td><td>{
"Curves": ["String"],
"Values": ["String"],
"Results": ["String"],
"Waveforms": ["String"]
}</td></tr>
<tr><td>Command: MemoryList.Get('Curve', 'Parameter')

Parameter: Curve Name

Description: Used to get data for a specific curve from the Memory List.

The response indicates whether or not the curve and its data were found.</td><td>{
"Found": false,
"Curve": {
"Name": "String",
"XData": [0.1],
"YData": [0.1],
"ZData": [0.1],
"XUnit": "String",
"YUnit": "String",
"ZUnit": "String",
"XDataScale": "dB/Lin/Pwr",
"YDataScale": "dB/Lin/Pwr",
"ZDataScale": "dB/Lin/Pwr",
"XdBRef": 0.1,
"YdBRef": 0.1,
"ZdBRef": 0.1,
"XAxisScale": "Log/Lin",
"YAxisScale": "Log/Lin",
"ZAxisScale": "Log/Lin",
"Protected": false
}
}</td></tr>
</table>

Memory List Commands	
Command/Parameter	Return Data

Command: MemoryList.Get('Value', 'Parameter')
 	
Parameter: Value Name
 	
Description: Used to get data for a specific value from the Memory List.
 	
The response indicates whether or not the value and its data were found.
	{
	"Found": false,
	"Value": {
	"Name": "String",
	"XData": 0.1,
	"YData": 0.1,
	"ZData": 0.1,
	"XUnit": "String",
	"YUnit": "String",
	"ZUnit": "String",
	"XDataScale": "dB/Lin/Pwr",
	"YDataScale": "dB/Lin/Pwr",
	"ZDataScale": "dB/Lin/Pwr",
	"XdBRef": 0.1,
	"YdBRef": 0.1,
	"ZdBRef": 0.1,
	"Protected": false
	}
	}

Command: MemoryList.Get('Result', 'Parameter')
 	
Parameter: Result Name
 	
Description: Used to get data for a specific result from the Memory List.
 	
The response indicates whether or not the result and its data were found.
	{
	"Found": false,
	"Result": {
	"Name": "String",
	"Passed": false,
	"Limit": "String",
	"Unit": "String",
	"Scale": "String",
	"Max/Min": "String",
	"Margin": 0.1,
	"Protected": false
	}
	}

Memory List Commands	
Command/Parameter	Return Data

Command: MemoryList.Get('Waveform', 'Parameter')
Parameter: Waveform Name
Description: Used to get data for a specific waveform from the Memory List.
The response indicates whether or not the waveform and its data were found.
	{
	"Found": false,
	"Waveform": {
	"Name": "String",
	"Waveform": {
	"X0": 0.1,
	"dX": 0.1,
	"YData": [0.1]
	},
	"XUnit": "String",
	"YUnit": "String",
	"YDataScale": "dB/Lin/Pwr",
	"YdBRef": 0.1,
	"YAxisScale": "Log/Lin",
	"Overload?": false,
	"Protected": false
	}
	}

Command: MemoryList.GetAllData
Parameter: None
Description: Used to get data for all curves, values, results, and waveforms in the Memory List
	{
	"Curves":[<See data format for "Curve" in Return Data for Command: MemoryList.Get('Curve', 'Parameter')>],
	"Values":[<See data format for "Value" in Return Data for Command: MemoryList.Get('Value', 'Parameter')>],
	"Results":[<See data format for "Result" in Return Data for Command: MemoryList.Get('Result', 'Parameter')>],
	"Waveforms":[<See data format for "Waveform" in Return Data for Command: MemoryList.Get('Waveform', 'Parameter')>]
	}

Memory List Commands	

Command/Parameter
 	
Description: Used to fill pre-existing curves, values, results, and waveforms in the Memory List with data.
 	
This cannot create data in the Memory List.
 	
It can overwrite pre-existing data in the Memory List. See ***Sequence Parameters on page 445*** for information on creating Memory List data placeholders.
 	
Command: MemoryList.Set('Curve', 'Data')
 	
Description: Used to fill pre-existing **curves** in the Memory List with data.
	The entire format shown below must be used when creating data to send to SoundCheck.
	
	{
	"Curve": {
	"Name": "String",
	"XData": [0.1],
	"YData": [0.1],
	"ZData": [0.1],
	"XUnit": "String",
	"YUnit": "String",
	"ZUnit": "String",
	"XDataScale": "dB/Lin/Pwr",
	"YDataScale": "dB/Lin/Pwr",
	"ZDataScale": "dB/Lin/Pwr",
	"XdBRef": 0.1,
	"YdBRef": 0.1,
	"ZdBRef": 0.1,
	"XAxisScale": "Log/Lin",
	"YAxisScale": "Log/Lin",
	"ZAxisScale": "Log/Lin",
	"Protected": false
	}
	}

Command: MemoryList.Set('Value', 'Data')
 	
Description: Used to fill pre-existing **values** in the Memory List with data.
 	
This cannot create data in the Memory List.
 	
It can overwrite pre-existing data in the Memory List. See ***Sequence Parameters on page 445*** for information on creating Memory List data placeholders.
	{
	"Value": {
	"Name": "String",
	"XData": 0.1,
	"YData": 0.1,
	"ZData": 0.1,
	"XUnit": "String",
	"YUnit": "String",
	"ZUnit": "String",
	"XDataScale": "dB/Lin/Pwr",
	"YDataScale": "dB/Lin/Pwr",
	"ZDataScale": "dB/Lin/Pwr",
	"XdBRef": 0.1,
	"YdBRef": 0.1,
	"ZdBRef": 0.1,
	"Protected": false
	}
	}

Memory List Commands	
Command/Parameter	The entire format shown below must be used when creating data to send to SoundCheck.

Command: MemoryList.Set('Result', 'Data')
Description: Used to fill pre-existing **results** in the Memory List with data.
This cannot create data in the Memory List.
It can overwrite pre-existing data in the Memory List. See ***Sequence Parameters on page 445*** for information on creating Memory List data placeholders.
	{
	"Result": {
	"Name": "String",
	"Passed": false,
	"Limit": "String",
	"Unit": "String",
	"Scale": "String",
	"Max/Min": "String",
	"Margin": 0.1,
	"Protected": false
	}
	}

Command: MemoryList.Set('Waveform', 'Data')
Parameter: Waveform Name
Description: Used to fill pre-existing **waveforms** in the Memory List with data.
This cannot create data in the Memory List.
It can overwrite pre-existing data in the Memory List. See ***Sequence Parameters on page 445*** for information on creating Memory List data placeholders.
	{
	"Waveform": {
	"Name": "String",
	"Waveform": {
	"X0": 0.1,
	"dX": 0.1,
	"YData": [0.1]
	},
	"XUnit": "String",
	"YUnit": "String",
	"YDataScale": "dB/Lin/Pwr",
	"YdBRef": 0.1,
	"YAxisScale": "Log/Lin",
	"Overload?": false,
	"Protected": false
	}
	}

Hardware Commands	
Command/Parameter	Return Data
Command: Hardware.Object('Parameter1').SetAddress(' Parameter2') **Parameter:** 1. Bluetooth Device Name - BTC device ID (e.g. 'BTC1') 2. Bluetooth Address - (e.g. 'AA:BB:CC:DD:EE:FF') **Description:** Used to set the bluetooth address for a Portland Tool & Die BTC or BQC device	Void

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SoundCheck TCP/IP Python Library (soundcheck_tcpip)

The SoundCheck TCP/IP Python library is included for users who want to control SoundCheck with Python scripts. This library greatly simplifies the process of launching SoundCheck, running sequences, and interacting with the Memory List.

Getting Started

Before using soundcheck_tcpip you must install it using **pip**.

From a Terminal (macOS) or DOS shell (Windows), CD to ${SOUNDCHECK_ROOT}/External Control Examples/Python where ${SOUNDCHECK_ROOT} is the location of your SoundCheck installation (e.g., C:\SoundCheck 18).

Run:

```
pip install soundcheck_tcpip-18.0.0-py3-none-any.whl
```

Classes and Methods

SoundCheck Installation (SCInstallation)

The SCInstallation object manages an installed version of SoundCheck.

SCInstallation Methods	
Method	**Description / Example Code**

construct_installation(
`  version,`
`  path`
)
	Returns a `SCInstallation` object for the `version` of SoundCheck located at `path` where `version` is a list containing the major and minor version numbers (e.g., [18, 0])
	
	`from soundcheck_tcpip.soundcheck.installation`
	`    import construct_installation`
	
	`sc_inst = construct_installation([18,0], 'C:\SoundCheck 18')`

launcher_path()	**Returns the full path to the SoundCheck executable**
	
	`from soundcheck_tcpip.soundcheck.installation`
	`    import construct_installation`
	
	`sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')`
	`sc_path = sc_inst.launcher_path()`

Launch Methods	
Method	**Description / Example Code**

launch()	**Launches SoundCheck specified by a `SCInstallation` object**
	
	`from soundcheck_tcpip.soundcheck.installation`
	`    import construct_installation`
	
	`sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')`
	`sc_inst.launch()`

Launch Methods	
Method	**Description / Example Code**

terminate()	**Terminates the SoundCheck that was launched by a `SCInstallation` object**
	
	`from soundcheck_tcpip.soundcheck.installation`
	`    import construct_installation`
	
	`sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')`
	`sc_inst.launch()`
	`sc_inst.terminate()`

is_running()	**Returns True if SoundCheck specified by a `SCInstallation` object is running**
	
	`from soundcheck_tcpip.soundcheck.installation`
	`    import construct_installation`
	
	`sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')`
	`running = sc_inst.is_running()`

INI File Methods	
Method	**Description / Example Code**

ini_path()	**Returns the full path to a `SCInstallation` INI file**
	
	`from soundcheck_tcpip.soundcheck.installation`
	`    import construct_installation`
	
	`sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')`
	`sc_path = sc_inst.ini_path()`

get_ini_option(
`   section,`
`   option,`
`   default`
)
	Attempt to read an `option` **from a** `SCInstallation` **INI file where `section` is a string identifying where in the INI file where the `option` is located. If the `option` is not found in the ini file,** `get_ini_option` **returns the `default` value.**
	
	`from soundcheck_tcpip.soundcheck.installation`
	`    import construct_installation`
	
	`sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')`
	`option = sc_inst.get_ini_option('Dialogs', 'SHOW`
	`SPLASHSCREEN')`

set_ini_option(
`   section,`
`   option,`
`   value`
)
	Attempt to write an `option` **to a** `SCInstallation` **INI file where `section` is a string identifying where in the INI file where the** `option` **is located and** `value` **is written to the INI for option.**
	
	`from soundcheck_tcpip.soundcheck.installation`
	`    import construct_installation`
	
	`sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')`
	`sc_inst.set_ini_option(`
	`    'Dialogs', 'SHOW SPLASHSCREEN', 'FAlSE'`
	`)`

File Import Methods	
Method	**Description / Example Code**

import_ini(ini_file)	**Import an INI file into an installation where** `ini_file` **is a string containing the path to the INI file**
	
	`from soundcheck_tcpip.soundcheck.installation`
	`    import construct_installation`
	
	`sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')`
	`sc_inst.import_ini('C:\Users\Me\MySoundCheckINI.ini')`

import_har(har_file)	**Import a hardware file into an installation where** `har_file` **is a string containing the path to the hardware file**
	
	`from soundcheck_tcpip.soundcheck.installation`
	`    import construct_installation`
	
	`sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')`
	`sc_inst.import_har('C:\Users\Me\MySoundCheckHAR.har')`

import_cal(cal_file)	**Import the calibration file into an installation where** `cal_file` **is a string containing the path to the calibration file**
	
	`from soundcheck_tcpip.soundcheck.installation`
	`    import construct_installation`
	
	`sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')`
	`sc_inst.import_cal('C:\Users\Me\MySoundCheckCAL.cal')`

Miscellaneous Methods	
Method	**Description / Example Code**

server_details()	**Returns the IP address and port number for an** `SCInstallation` **TCP/IP server**
	
	`from soundcheck_tcpip.soundcheck.installation`
	`    import construct_installation`
	
	`sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')`
	`ip_address, port = sc_inst.server_details()`

SoundCheck TCP/IP Control (SCControlTCPIP)

The SCControlTCPIP object manages the TCP/IP connection to a SoundCheck installation.

<table>
<tr><th colspan="2">Construction</th></tr>
<tr><th>Method</th><th>Description / Example Code</th></tr>
<tr><td>SCControlTCPIP(
<code>installation</code>
)</td><td>Creates an SCControlTCPIP object from the <code>SCInstallation</code> object, <code>installation</code>, and enables its TCP/IP server where:
<pre>
from soundcheck_tcpip.soundcheck.installation
    import construct_installation
from soundcheck_tcpip.soundcheck.controller
    import SCControlTCPIP

sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')
sc_tcpip = SCControlTCPIP(sc_inst)
</pre></td></tr>
</table>

<table>
<tr><th colspan="2">Launch Methods</th></tr>
<tr><th>Method</th><th>Description / Example Code</th></tr>
<tr><td>launch(timeout=30)</td><td>Starts SoundCheck and waits for it to be ready to respond to external commands. If SoundCheck does not respond within <code>timeout</code> seconds, an exception is raised
<pre>
from soundcheck_tcpip.soundcheck.installation
    import construct_installation
from soundcheck_tcpip.soundcheck.controller
    import SCControlTCPIP

sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')
sc_tcpip = SCControlTCPIP(sc_inst)
sc_tcpip.launch()
</pre></td></tr>
<tr><td>close(timeout=5)</td><td>Sends SoundCheck a shutdown command and force exit if SoundCheck does not exit within <code>timeout</code> seconds
<pre>
from soundcheck_tcpip.soundcheck.installation
    import construct_installation
from soundcheck_tcpip.soundcheck.controller
    import SCControlTCPIP

sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')
sc_tcpip = SCControlTCPIP(sc_inst)
sc_tcpip.close()
</pre></td></tr>
</table>

Sequence Methods	
Method	**Description / Example Code**

open_sequence(
`path,`
`timeout=10`
)
	Sends a command for SoundCheck to open the sequence located at `path`. If the sequence does not open within `timeout` seconds, an exception is raised
	
	`from soundcheck_tcpip.soundcheck.installation`
	`    import construct_installation`
	`from soundcheck_tcpip.soundcheck.controller`
	`    import SCControlTCPIP`
	
	`sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')`
	`sc_tcpip = SCControlTCPIP(sc_inst)`
	`sc_tcpip.launch()`
	`sc_tcpip.open_sequence('C:\Users\Me\Sequences\My Seq.sqc')`

current_sequence()	**Returns the name of the currently loaded sequence**
	
	`from soundcheck_tcpip.soundcheck.installation`
	`    import construct_installation`
	`from soundcheck_tcpip.soundcheck.controller`
	`    import SCControlTCPIP`
	
	`sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')`
	`sc_tcpip = SCControlTCPIP(sc_inst)`
	`sc_tcpip.launch()`
	`sc_tcpip.open_sequence('C:\Users\Me\Sequences\My Seq.sqc')`
	`sqc_name = sc_tcpip.current_sequence()`

get_sequence_path()	**Returns the full path name of the currently loaded sequence**
	
	`from soundcheck_tcpip.soundcheck.installation`
	`    import construct_installation`
	`from soundcheck_tcpip.soundcheck.controller`
	`    import SCControlTCPIP`
	
	`sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')`
	`sc_tcpip = SCControlTCPIP(sc_inst)`
	`sc_tcpip.launch()`
	`sc_tcpip.open_sequence('C:\Users\Me\Sequences\My Seq.sqc')`
	`sqc_path = sc_tcpip.get_sequence_path()`

Sequence Methods	
Method	**Description / Example Code**

get_sequence_steps_list()	**Returns a list of dicts containing the name, type, input channels, and output channels of the currently loaded sequence. The format of the response is:**
	
	`[`
	`    {`
	`        'Name': step1_name,`
	`        'Type': step1_type,`
	`        'InputChannelNames': [list of input channel names],`
	`        'OutputChannelNames': [list of output channel names]`
	`    },`
	`    .`
	`    .`
	`    .`
	`]`
	
	`from soundcheck_tcpip.soundcheck.installation`
	`    import construct_installation`
	`from soundcheck_tcpip.soundcheck.controller`
	`    import SCControlTCPIP`
	
	`sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')`
	`sc_tcpip = SCControlTCPIP(sc_inst)`
	`sc_tcpip.launch()`
	`sc_tcpip.open_sequence('C:\Users\Me\Sequences\My Seq.sqc')`
	`sqc_steps = sc_tcpip.get_sequence_steps_list()`

run_sequence(
`path=None,`
`timeout=300,`
`iterations=1,`
`respond=True`
)
	Sends a command for SoundCheck to open and run the sequence located at `path` **for the specified number of** `iterations`**. If the sequence does not open and run within** `timeout` **seconds, an exception is raised. If** `respond` **is** `True`**, a list of responses from each sequence iteration is returned, otherwise** `None` **is returned.**
	
	`from soundcheck_tcpip.soundcheck.installation`
	`    import construct_installation`
	`from soundcheck_tcpip.soundcheck.controller`
	`    import SCControlTCPIP`
	
	`sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')`
	`sc_tcpip = SCControlTCPIP(sc_inst)`
	`sc_tcpip.launch()`
	`sc_tcpip.open_sequence('C:\Users\Me\Sequences\My Seq.sqc')`
	`sc_tcpip.run_sequence()`

save_sequence()	**Saves the currently loaded sequence to disk.**
	
	`from soundcheck_tcpip.soundcheck.installation`
	`    import construct_installation`
	`from soundcheck_tcpip.soundcheck.controller`
	`    import SCControlTCPIP`
	
	`sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')`
	`sc_tcpip = SCControlTCPIP(sc_inst)`
	`sc_tcpip.launch()`
	`sc_tcpip.open_sequence('C:\Users\Me\Sequences\My Seq.sqc')`
	`sc_tcpip.save_sequence()`

Sequence Methods	
Method	**Description / Example Code**

get_sequence_duration()	**Returns the time (in seconds) that it took for the last sequence run.**
	
	`from soundcheck_tcpip.soundcheck.installation`
	`    import construct_installation`
	`from soundcheck_tcpip.soundcheck.controller`
	`    import SCControlTCPIP`
	
	`sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')`
	`sc_tcpip = SCControlTCPIP(sc_inst)`
	`sc_tcpip.launch()`
	`sc_tcpip.open_sequence('C:\Users\Me\Sequences\My Seq.sqc')`
	`sc_tcpip.run_sequence()`
	`duration = sc_tcpip.get_sequence_duration()`

Memory List Methods	
Method	**Description / Example Code**

get_memlist_names(
`data_type=None`
)
	Returns a dict of name lists keyed by data_type where `data_type` **can be one of** `'Curves'`, `'Values'`, `'Results'`, **or** `'Waveforms'`. **If** `data_type` **is not specified,** `get_memlist_names` **returns the names of all items of all data types as a list of strings.**
	
	`from soundcheck_tcpip.soundcheck.installation`
	`    import construct_installation`
	`from soundcheck_tcpip.soundcheck.controller`
	`    import SCControlTCPIP`
	
	`sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')`
	`sc_tcpip = SCControlTCPIP(sc_inst)`
	`sc_tcpip.launch()`
	`sc_tcpip.run_sequence('C:\Users\Me\Sequences\My Seq.sqc')`
	`ml_names = sc_tcpip.get_memlist_names()`

get_result(name)	**Gets the result from the Memory List identified by** `name`
	
	`from soundcheck_tcpip.soundcheck.installation`
	`    import construct_installation`
	`from soundcheck_tcpip.soundcheck.controller`
	`    import SCControlTCPIP`
	
	`sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')`
	`sc_tcpip = SCControlTCPIP(sc_inst)`
	`sc_tcpip.launch()`
	`sc_tcpip.run_sequence('C:\Users\Me\Sequences\My Seq.sqc')`
	`result = sc_tcpip.get_result('My Result')`

Memory List Methods	
Method	**Description / Example Code**

get_value(name)	**Returns a dict representing the value in the Memory List identified by `name`. The format of the response is:**
	
	`{`
	`    'Name': name,`
	`    'XData': X data,`
	`    'YData': Y data,`
	`    'ZData': Z data,`
	`    'XUnit': X units,`
	`    'YUnit': Y units,`
	`    'ZUnit': Z units,`
	`    'XDataScale': X data scale,`
	`    'YDataScale': Y data scale,`
	`    'ZDataScale': Z data scale,`
	`    'XdBRef': X dB reference,`
	`    'YdBRef': Y dB reference,`
	`    'ZdBRef': Z dB reference,`
	`    'XAxisScale': X axis scale,`
	`    'YAxisScale': Y axis scale,`
	`    'ZAxisScale': Z axis scale,`
	`    'Protected': protected status`
	`}`
	
	`from soundcheck_tcpip.soundcheck.installation`
	`    import construct_installation`
	`from soundcheck_tcpip.soundcheck.controller`
	`    import SCControlTCPIP`
	
	`sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')`
	`sc_tcpip = SCControlTCPIP(sc_inst)`
	`sc_tcpip.launch()`
	`sc_tcpip.run_sequence('C:\Users\Me\Sequences\My Seq.sqc')`
	`value = sc_tcpip.get_value('My Value')`

<table>
<tr><th colspan="2">Memory List Methods</th></tr>
<tr><th>Method</th><th>Description / Example Code</th></tr>
<tr><td>get_curve(name)</td><td>Returns a dict representing the curve from the Memory List identified by <code>name</code>. The format of the response is:
<pre>
{
    'Name': name,
    'XData': X data,
    'YData': Y data,
    'ZData': Z data,
    'XUnit': X units,
    'YUnit': Y units,
    'ZUnit': Z units,
    'XDataScale': X data scale,
    'YDataScale': Y data scale,
    'ZDataScale': Z data scale,
    'XdBRef': X dB reference,
    'YdBRef': Y dB reference,
    'ZdBRef': Z dB reference,
    'XAxisScale': X axis scale,
    'YAxisScale': Y axis scale,
    'ZAxisScale': Z axis scale,
    'Protected': protected status
}

from soundcheck_tcpip.soundcheck.installation
    import construct_installation
from soundcheck_tcpip.soundcheck.controller
    import SCControlTCPIP

sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')
sc_tcpip = SCControlTCPIP(sc_inst)
sc_tcpip.launch()
sc_tcpip.run_sequence('C:\Users\Me\Sequences\My Seq.sqc')
curve = sc_tcpip.get_curve('My Curve')
</pre></td></tr>
</table>

<table>
<tr><th colspan="2">Memory List Methods</th></tr>
<tr><th>Method</th><th>Description / Example Code</th></tr>
<tr><td>get_waveform(name)</td><td>Returns a dict representing the waveform from the Memory List identified by <code>name</code>. The format of the response is:

<pre>
{
    'Name': name,
    'Waveform': {'X0': initial X, 'dX': delta X, 'YData': Y
data},
    'XUnit': X units,
    'YUnit': Y units,
    'YDataScale': Y data scale,
    'YdBRef': Y dB reference,
    'YAxisScale': Y axis scale,
    'Overload?': overload status
    'Protected': protected status
}

from soundcheck_tcpip.soundcheck.installation
    import construct_installation
from soundcheck_tcpip.soundcheck.controller
    import SCControlTCPIP

sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')
sc_tcpip = SCControlTCPIP(sc_inst)
sc_tcpip.launch()
sc_tcpip.run_sequence('C:\Users\Me\Sequences\My Seq.sqc')
waveform = sc_tcpip.get_waveform('My Waveform')
</pre></td></tr>
<tr><td>set_value(
<pre>
   name,
   x_data=0.0,
   y_data=0.0,
   z_data=0.0,
   x_unit='',
   y_unit='',
   z_unit='',
   x_data_scale='Lin',
   y_data_scale='Lin',
   z_data_scale='Lin',
   xdb_ref=1,
   ydb_ref=1,
   zdb_ref=1,
   protected=False
</pre>
)</td><td>Sets the value in the Memory List identified by <code>name</code>. Note that the value must exist in the Memory List for it to be set.

<pre>
from soundcheck_tcpip.soundcheck.installation
    import construct_installation
from soundcheck_tcpip.soundcheck.controller
    import SCControlTCPIP

sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')
sc_tcpip = SCControlTCPIP(sc_inst)
sc_tcpip.launch()
sc_tcpip.set_value('My Value', y_data=4.321, y_unit='Hz')
sc_tcpip.run_sequence('C:\Users\Me\Sequences\My Seq.sqc')
</pre></td></tr>
</table>

Memory List Methods	
Method	**Description / Example Code**

set_curve(
`name,`
`x_data=np.array([]),`
`y_data=np.array([]),`
`z_data=np.array([]),`
`x_unit='',`
`y_unit='',`
`z_unit='',`
`x_data_scale='Lin',`
`y_data_scale='Lin',`
`z_data_scale='Lin',`
`xdb_ref=1,`
`ydb_ref=1,`
`zdb_ref=1,`
`x_axis_scale='Lin',`
`y_axis_scale='Lin',`
`z_axis_scale='Lin',`
`protected=False`
)
	Sets the curve from the Memory List identified by `name`. Note that the curve must exist in the Memory List for it to be set.
	
	`from soundcheck_tcpip.soundcheck.installation`
	`    import construct_installation`
	`from soundcheck_tcpip.soundcheck.controller`
	`    import SCControlTCPIP`
	`import numpy as np`
	
	`sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')`
	`sc_tcpip = SCControlTCPIP(sc_inst)`
	`sc_tcpip.launch()`
	`sc_tcpip.set_curve('My Curve', y_data=np.random.rand(4096))`
	`sc_tcpip.run_sequence('C:\Users\Me\Sequences\My Seq.sqc')`

set_waveform(
`name,`
`x0=0.0,`
`dX=1.0,`
`y_data=np.array([]),`
`x_unit='',`
`y_unit='',`
`y_data_scale='Lin',`
`ydb_ref=1,`
`y_axis_scale='Lin',`
`overload=False,`
`protected=False`
)
	Sets the waveform from the Memory List identified by `name`. Note that the waveform must exist in the Memory List for it to be set.
	
	`from soundcheck_tcpip.soundcheck.installation`
	`    import construct_installation`
	`from soundcheck_tcpip.soundcheck.controller`
	`    import SCControlTCPIP`
	`import numpy as np`
	
	`sc_inst = construct_installation([18, 0], 'C:\SoundCheck 18')`
	`sc_tcpip = SCControlTCPIP(sc_inst)`
	`sc_tcpip.launch()`
	`sc_tcpip.set_waveform('My Curve', x0=0.0, dX=0.1,`
	`y_data=np.random.rand(4096))`
	`sc_tcpip.run_sequence('C:\Users\Me\Sequences\My Seq.sqc')`

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Controlling SoundCheck® From ActiveX - DEPRECATED

Important! ActiveX control is being replaced by TCP/IP control. SoundCheck 17 will be the last version in which ActiveX is supported. We recommend that you use the TCP/IP interface instead.

Please refer to ***Controlling SoundCheck with TCP/IP on page 491*** for more information. The following information is provided for legacy purposes and to assist with converting existing ActiveX controls to TCP/IP.

SoundCheck Run Seq.vi has been replaced with a version that uses TCP/IP:

C:\SoundCheck x.x\API\SoundCheck Run Seq TCP\SoundCheck Run Seq TCP.vi.

- SoundCheck must be running, with TCP/IP enabled, before running the vi
- You must enable TCP/IP at port 4444 in SoundCheck preferences before running the vi

The new vi outputs standardized data structures which are easier to work with than the old SoundCheck Run Seq vi.

For Windows users who are relying on the old SoundCheck Run Seq.vi, a new version has been included for compatibility purposes. This vi starts SoundCheck and converts the new data formats to match the old data format. This vi will not return waveforms from the Memory List. "SoundCheck Run Seq.vi" is found in the root of the SoundCheck folder.

"**SoundCheck Run Seq TCP.vi**" is run inside of "**SoundCheck Run Seq.vi**" after the code to open and close SoundCheck. The name of the top level vi has not been changed for legacy/upgrading purposes.

ActiveX Control - Legacy Examples (Windows Only)

- You can run SoundCheck test sequences from any programming language that supports ActiveX
- You can receive test data and results
- SoundCheck is opened in this process, but the SoundCheck window can be hidden
- SoundCheck sequences can be opened and run in this mode, but not modified
- A valid Hardware Key is required to register ActiveX components during SoundCheck installation. The Hardware Key is not required to use ActiveX in Demo Mode.

Important!* *Please make sure only one version of SoundCheck is installed on the computer before using ActiveX control. Otherwise, the wrong version might get called.

In terms of Microsoft's® Component Object Model (COM), SoundCheck is an ActiveX Server, while the software to control SoundCheck is an ActiveX Client. It is important that the developer be familiar with these programming concepts before attempting to use SoundCheck's API. Listen, Inc. does not provide a static library to link to.

Examples are in included in the SoundCheck installation folder: C:\SoundCheck 18.1\External Control Examples_Legacy ActiveX Examples

- **Visual Basic (2010):** See ***Visual Basic Example on page 525***
- **C#:** See ***C # Example on page 526***

LabVIEW's ActiveX Library

SoundCheck is written in LabVIEW. There are many ways to programmatically obtain SoundCheck's ActiveX object, but the most efficient way is to import its **Type Library** into your project. The SoundCheck program folder contains a file: "SoundCheck 18.1.tlb". This is the SoundCheck Type Library which contains a superset of **LabVIEW 2018**, plus SoundCheck's interfaces. You add the TLB as a reference into your project.

The main ActiveX interface used to control SoundCheck is the **Call Method** of the VI object. The **Call Method** defines inputs to a VI, runs the VI, and then receives the outputs from the VI. The **Call** is synchronous, so your program execution will stay on that line of code until the **Call** is completed.

Starting Up SoundCheck

SoundCheck can be started via a shell object as a command line invocation, or when the COM object is created. In our examples, ActiveX is not used to start SoundCheck. Instead, a Windows® **"command line"** command is used.

> **Example:** In Visual Basic the line of code is: **Shell "C:\SoundCheck 18.1\SoundCheck 18.1.exe"**

If your SoundCheck folder is not in the root directory, replace the beginning of the path, with the path to the SoundCheck folder on your system.

Command Line Options

SoundCheck interprets three command line options:

- **-m** – Minimize SoundCheck immediately after it starts up (SoundCheck main screen is accessible)
- **-h** – Hide the SoundCheck Main Screen so that it cannot be brought into view even by clicking the SoundCheck **Task Bar Button** (Prevents access to the SoundCheck main screen options)

Note: The only restriction on these options is that **-m** and **-h** cannot be used together.

Visual Basic Example

- The example executable file can be run to show its general use. The source code files are included in the example folder.
- C:\SoundCheck 18.1\External Control Examples_Legacy ActiveX Examples\VB2010 Example.exe

This line of code is used to start SoundCheck. The Main window is hidden.

```
Shell """C:\SoundCheck 18.1\SoundCheck 18.1.exe"" -h -s"
```

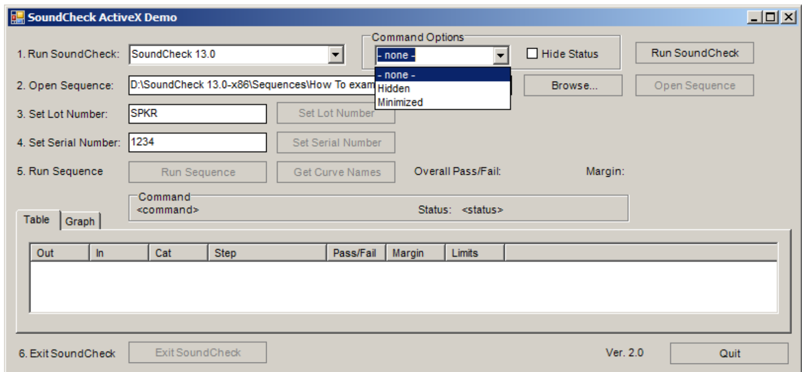


Figure 33-1: Visual Basic Panel

- Select Command Options:
  - None - SoundCheck opens in a normal window
  - Hidden - SoundCheck is hidden from view
  - Minimized - SoundCheck is opened but the window is minimized

1. Select SoundCheck version to run - SoundCheck 18.1, then click **Run SoundCheck**
2. Select sequence to open - ActiveX & Test Stand example, then click **Open Sequence**
3. Enter Serial Number and click **Set Serial Number**
4. Enter Lot Number and click **Set Lot Number**
5. Click **Run Sequence - Get Curve Names**
6. Click **Exit** SoundCheck to close

C # Example

- The example executable file can be run to show its general use. The source code files are included in the example folder.
- C:\SoundCheck 18.1\External Control Examples_Legacy ActiveX Examples\C Sharp\Application\C Sharp Example.exe

1. Launch SoundCheck. Check SoundCheck Main Screen and dismiss open dialog windows.
2. Press any key to load sequence. Check SoundCheck Main Screen and dismiss open dialog windows.
3. Run Sequence

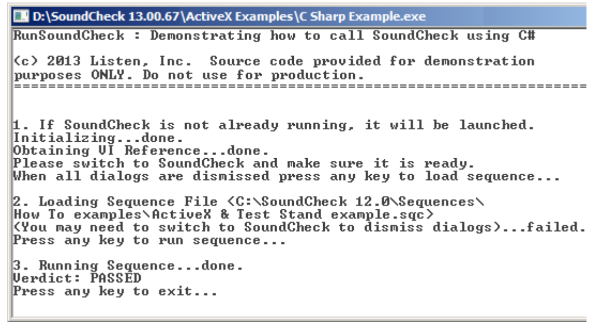


Figure 33-2: C # Example

Creating the VI ActiveX Object and Calling SoundCheck

Here are some Visual Basic lines of code that illustrate how to create the VI object you need and then how to use it to communicate with SoundCheck.

Only paramValues (0) is an input to ControlSC.vi. You must set the remaining seven paramValues to dummy values prior to making the Call. After the Call returns, paramValues (1) through (7) will contain values returned by the VI.

```
Set lvApp = CreateObject("SoundCheck111.Application")
lvPath = "ControlSC.vi"
Set lvVI = lvApp.GetVIReference(lvPath)
paramNames(0) = "Command"
paramNames(1) = "Success?"
paramNames(2) = "Pass?"
paramNames(3) = "Margin"
paramNames(4) = "Table"
paramNames(5) = "Xdatapoints"
paramNames(6) = "Ydatapoints"
paramNames(7) = "Zdatapoints"
paramValues(0) = "serial " & txtSerialNum
paramValues(1) = False
paramValues(2) = False
paramValues(3) = 0#
paramValues(4) = ""
paramValues(5) = ""
paramValues(6) = ""
paramValues(7) = ""
lvVI.Call paramNames, paramValues
```

API Specification (Windows)

SoundCheck's ControlSC.vi

The VI that you Call is ControlSC.vi, which is embedded in the application's executable file. ControlSC.vi has the following inputs and outputs (not all outputs are returned by every command):

Parameter / Function	Input or Output?	Type	Description
[0] - Command	Input	string	One of seven commands directing SoundCheck perform an action. The command word may be followed by parameters that are separated by spaces, all in the same string.
[1] - Success?	Output	boolean	Tells whether or not the command and action were successful.
[2] - Pass?	Output	boolean	Tells whether or not the test sequence passed or failed overall, typically as configured in the final Display Step.
[3] - Margin	Output	double float	The margin of **Pass** or **Fail** in the last Limits Step of a test sequence.
[4] - Table	Output	string	Multi-purpose table of information, in standard tab-delimited spread-sheet format, in which columns are separated by tabs and rows are separated by CR-LF.
[5] - Xdatapoints	Output	array of double floats	Array of X values of requested data curve. Example: Frequency in Hz.
[6] - Ydatapoints	Output	array of double floats	Array of Y values of requested data curve. Example: Magnitude in dB.
[7] - Zdatapoints	Output	array of double floats	Array of Z values of requested data curve. Example: Phase in deg.

Common Properties of the *Success?* Returned Parameter

For all seven commands, the "Success?" parameter returns a FALSE if SoundCheck was busy or if the command was not understood.

For example, SoundCheck is busy and cannot process any commands if it is not finished processing the last command that was sent or a test sequence is running.

If "Success?" returns a FALSE, SoundCheck is probably busy, and therefore it is suggested practice to reissue the command until Success? = True.

Note: Do not reissue the command more than about 10 times, which should take a total of less than one second.

The Seven Commands of the “Command” Input

1. **Open** - **Action**: Loads a desired test sequence into SoundCheck and prepares it for execution. Unloads the previously loaded sequence. If this command is not issued, SoundCheck will load its default sequence at start-up.

Parameter / Function	Input	Output	Remarks
[0] - Command	“open <sqc>”	string	<sqc> must contain full path of sequence file
[1] - Success?	FALSE	boolean	TRUE if the requested sequence was opened successfully. FALSE if: • Opening a sequence not currently permitted • Sequence file not found or path invalid • Sequence file corrupt • One or more steps in the sequence not found or is corrupt
[2] - Pass?	FALSE	boolean	Not used, but input should still be initialized to FALSE
[3] - Margin	0.0#		Not used, input value should be zero.
[4] - Table	NULL	string	Table – One row for each step in the sequence. Four columns, as follows: • Output channel • Input channel • Step category (3-letter abbreviation) • Step name
[5] - Xdatapoints	NULL		Not used
[6] - Ydatapoints	NULL		Not used
[7] - Zdatapoints	NULL		Not used

Example: *open C:\SoundCheck 18.1\Sequences\Loudspeakers\Fundamental.sqc*

2. **Lot** - **Action**: Sets the Lot Number in SoundCheck. This Lot Number will remain in force until it is changed.

Parameter / Function	Input	Output	Remarks
[0] - Command	“lot <number>”	string	<number> - any alphanumeric characters
[1] - Success?	FALSE	boolean	TRUE if the Lot Number was set successfully
[2] - Pass?	FALSE	boolean	Not used; but input should still be initialized to FALSE
[3] - Margin	0.0#	double float	Not used; input value should be zero
[4] - Table	NULL	string	Table – One row for each step in the sequence. Four columns, as follows: ● Output channel ● Input channel ● Step category (3-letter abbreviation) ● Step name
[5] - Xdatapoints	NULL		Not used
[6] - Ydatapoints	NULL		Not used
[7] - Zdatapoints	NULL		Not used

Example: *lot SC200108*

3. **Serial** - **Action**: Sets the Serial Number in SoundCheck. This Serial Number will remain in force until it is changed (the sequence may be configured to change it as well).

Parameter / Function	Input	Output	Remarks
[0] - Command	“serial <sn>”	string	<sn> - serial number (any alphanumeric characters)
[1] - Success?	FALSE	boolean	TRUE if the Serial Number was set successfully.
[2] - Pass?	FALSE	boolean	Not used; but input should still be initialized to FALSE
[3] - Margin	0.0#	double float	Not used; input value should be zero.
[4] - Table	NULL		Not used
[5] - Xdatapoints	NULL		Not used
[6] - Ydatapoints	NULL		Not used
[7] - Zdatapoints	NULL		Not used

Example: *serial LSX00844*

4. Run - **Action**: *Runs the test sequence currently loaded in SoundCheck.*

Parameter / Function	Input	Output	Remarks
[0] - Command	"run"	string	
[1] - Success?	FALSE	boolean	TRUE if the sequence ran to completion, regardless of whether or not it passed. FALSE if running a sequence not currently permitted, or Sequence aborted at some point.
[2] - Pass?	FALSE	boolean	TRUE if sequence passes
[3] - Margin	0.0#	double float	Overall margin
[4] - Table	NULL	string	One row for each step in the sequence. Three columns, as follows: - Pass or FAIL - Margin - Limit-Max/Min info
[5] - Xdatapoints	NULL		Not used
[6] - Ydatapoints	NULL		Not used
[7] - Zdatapoints	NULL		Not used

5. **Names - Action**: Returns a list of curve names generated by the last test sequence run. If no sequence was run or if the sequence did not generate any curves, the list is empty.

Parameter / Function	Input	Output	Remarks
[0] - Command	"names"	string	
[1] - Success?	FALSE	boolean	TRUE if a list of curve names was returned, even if it was empty.
[2] - Pass?	FALSE	boolean	TRUE if sequence passes
[3] - Margin	0.0#	double float	Not used
[4] - Table	NULL	string	One column: the curve names
[5] - Xdatapoints	NULL		Not used
[6] - Ydatapoints	NULL		Not used
[7] - Zdatapoints	NULL		Not used

Command string: *names*

6. **Curve - Action**: Returns a binary representation of the data from the requested curve, and other curve info.

Parameter / Function	Input	Output	Remarks
[0] - Command	"curve <cn>"	string	<cn> - curve name obtained from "names" command
[1] - Success?	FALSE	boolean	TRUE if the requested data was returned. FALSE if the requested curve was not found among the curves generated by the last sequence run.
[2] - Pass?	FALSE	boolean	Not used
[3] - Margin	0.0#	double float	Not used
[4] - Table	NULL	string	17 rows, 2 columns. A table of information about the data values including units and log scaling. The first column contains the item names, the second column the item values: ♦ N points - number of data points in the curve ♦ X data - "dB" or "lin" ♦ Y data - "dB" or "lin" ♦ Z data - "dB" or "lin" ♦ X axis - "log" or "lin" ♦ Y axis - "log" or "lin" ♦ Z axis - "log" or "lin" ♦ X unit - such as "Hz" ♦ Y unit - such as "Pa", "V" (floating point numbers are used so prefixes such as mV are not used) ♦ Z unit - such as "deg" ♦ X dB ref - dB reference, in floating point or scientific notation ♦ Y dB ref - dB reference, in floating point or scientific notation ♦ Z dB ref - dB reference, in floating point or scientific notation ♦ Single val? - "True" or "False", True meaning that only the Y value of only the first data point is of interest.
[5] - Xdatapoints	NULL	double float array	Xdatapoints
[6] - Ydatapoints	NULL	double float array	Ydatapoints
[7] - Zdatapoints	NULL	double float array	Zdatapoints

Example: "*curve Fundamental [L]*"

7. **Exit** - **Action**: *Exit SoundCheck. SoundCheck and LabVIEW Run-Time will quit and close.*

Parameter / Function	Input	Output	Remarks
[0] - Command	“exit”	string	
[1] - Success?	FALSE	boolean	TRUE if SoundCheck started exiting.
[2] - Pass?	FALSE	boolean	Not used
[3] - Margin	0.0#	double float	Not used
[4] - Table	NULL		Not used
[5] - Xdatapoints	NULL		Not used
[6] - Ydatapoints	NULL		Not used
[7] - Zdatapoints	NULL		Not used

Database Setup

Important! As of SoundCheck 17, the procedure for setting up a database has changed. The following instructions should be reviewed even if you already have a database working with SoundCheck.

Requirements

- Database V2 requires SoundCheck 17 or later
- Database V2 and Database Schema 1 both require optional module 2010 - Save to Database
- **SoundCheck 18 is only available as a 64-bit application**. It requires a 64-bit version of Windows. It cannot be installed on a 32-bit version of Windows. This also requires 64-bit database applications and drivers.

Database V2 Features

Using Autosave Steps with databases has been improved. As of SoundCheck 17, a new database schema is available with the following features:

- Fewer tables
- More efficient use of data types and the use of BLOBs (Binary Large OBjects)
- More space efficient
- Faster transfer of data

All Database Users

Regardless of the database schema used, all users should review:

- ***Supported Databases on page 534***
- ***Creating a UDL Definition for SQL Server on page 541***

New Database Users

- ***Database V2 Setup on page 537***
- ***SoundCheck DB v2 Schema on page 538***

Existing Access Database Users

While Database V2 offers significant improvements, Database Schema 1 is available for users with existing databases.

If you are using Microsoft Access please refer to the sections below.

- ***Database Schema 1 for Access on page 546***
- ***Creating an ODBC Connection for MS Access on page 547***

Supported Databases

As of SoundCheck 17, all database applications must be 64-bit. SoundCheck's Autosave feature currently supports the following databases:

- Microsoft SQL Server 64-bit - 2005, 2008, 2014 and 2016
- Microsoft Access 64-bit - 2003, 2007, 2010, 2013 (*.MDB files)

Database Setup

You need to perform the following steps in order to make a successful database connection.

1 Determine the Data Store

The database schema, storage method and storage location must be considered first.

Please review the information in the following sections before setting up your schema.

The Database Store on page 535

Creating the Database Connection on page 536

2 Ensure that the data store contains the correct schema

The schema for your database must be generated before you can connect and save to it.

- When using Database V2 follow the instructions in ***Database V2 Setup on page 537***
- When using MS Access follow the instructions in ***Database Schema 1 for Access on page 546***

3 Create the appropriate Connect Descriptor

Database clients use a connect descriptor to define the location of the database and the name of the database service. It's the key to connecting SoundCheck to the database.

Database V2

- ***Creating a UDL Definition for SQL Server on page 541***
- ***Creating an ODBC Connection for SQL Server on page 542***

Database Schema 1

- ***Creating a UDL Definition for MS Access on page 549***

4 Configure SoundCheck to use the Connect Descriptor

The SoundCheck Autosave step must be configured to use the appropriate Connect Descriptor for the database.

Database V2

- ***SoundCheck Autosave Database V2 on page 545***

Database Schema 1

- ***SoundCheck Autosave for Access DB on page 550***

The Database Store

The following data storage scenarios are supported:

- Local data storage
- Remote data storage

Local storage means that data is stored on a hard disk of the computer running SoundCheck. SoundCheck can write to a local database such as SQL Server Express or to an Access database file (mdb).

If you plan on integrating this data into your enterprise, then SQL Server would be more appropriate. This may require System Administrators' assistance in setting up the database.

If you want to set up a quick way of saving data for your own use and are familiar with MS Access, then use a .MDB file.

SQL Server for Local Storage

If you choose SQL Server Express, the database engine must be installed. SQL Server Express is freely distributed by Microsoft at no cost. Download SQL Server Express 64-bit.

SQL Server for Remote Storage

Remote storage means that the data will be stored outside the computer, which must be connected to a network. If this scenario matches your environment, you need to know the following information:

- The name of the database server
- The name of the database table
- The authentication parameters (account name and password)

No database software is required for remote storage, but drivers are required.

Since SQL Server is an enterprise class database, its installation should be performed by a qualified Database Administrator (DBA), and is beyond the scope of this document.

It is recommended that the SQL script, **C:\SoundCheck 18\Database\createSchema.sql**, be forwarded to the DBA for installation of Database V2.

MS Access for Local Storage

Installation of MS Access drivers are required for interfacing to .MDB files.

Previous databases using the ***Jet 4.0 Database Engine*** must upgrade to **Microsoft Access Database Engine** for 64-bit compatibility.

This can be downloaded separately:

https://www.microsoft.com/en-us/download/confirmation.aspx?id=54920

The MS Access application is not required to be installed for writing to a .MDB file, but you do need it to create a new .MDB file.

Rather than set up your own database, please use the one included with SoundCheck.

See ***Database Schema 1 for Access on page 546***.

Make sure that the file is on a local disk, not a shared network disk, as this will significantly affect performance.

Creating the Database Connection

Regardless of data store, SoundCheck relies upon Open Database Connectivity (ODBC) or Universal Data Links (UDL) to describe how database connections are to be established. The descriptor specifies where the storage is, and which driver SoundCheck will use to communicate with the database.

ODBC Connection Rules

SoundCheck 18 is only available as a 64-bit application. It requires a 64-bit version of Windows. It cannot be installed on a 32-bit version of Windows.

Windows includes **ODBC Data Source** apps for both 64 and 32-bit. SoundCheck 18 requires **ODBC Data Sources 64-bit.**

You must use a 64-bit version of MS Access along with the 64-bit ODBC tool.

- The 64-bit version of the Odbcad32.exe file is located in the %systemdrive%\Windows\System32 folder
- Both 32-bit and 64-bit files are named Odbcad32.exe but they are not the same
- The 64-bit file is required for SoundCheck 18
- **NOTE: The default installation of MS Office is 32-bit. You will need to install MS Office 64-bit instead.**

DSN vs UDL

SoundCheck can use either set of drivers to connect to a database. Which set you will use most likely comes from other software requirements, if any. For example, there may be a 3rd party application that requires one or the other. In either case, SoundCheck can use the same drivers to make setup easier.

Important: ***When using a database that requires SQL Server authentication with a Login ID and Password, you must use a "UDL" file. A DSN does not store the login ID and password.***

DSN

A Data Source Name (DSN) descriptor is required for database connections using ODBC drivers. A DSN can be of **System**, **User** or **File** types. Choose **System DSN** if you are not sure which one to use. A DSN does not store the login ID and password.

UDL

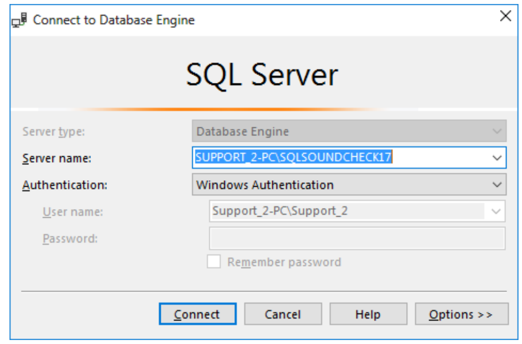
A UDL descriptor is required for database connections using OLE DB drivers, as well as ODBC drivers. Like a File DSN, UDL connection parameters are stored as a connection string in a text file. UDL files also allow you to store the Login ID and Password for the database.

The most straightforward method is to create a UDL file.

Database V2 Setup

Creating a Schema

1. An SQL Server must be installed before creating the schema.
2. Install **Microsoft SQL Server Management Studio**.
3. Open **C:\SoundCheck 18\Database** and double : **Create Schema.SQL.**
   SQL Server Management Studio will open.
4. You are prompted to select the SQL Server installed on the system.



5. In SQL Server Management Studio, click on the SQL code in the right hand side of the window. This makes the **Execute** button active.
6. Click **Execute** to run the SQL code.
7. This creates the **SoundCheck DB v2** schema, e.g.: files and tables, as shown in ***Figure 34-1***.

   (Note that the schema does not show up in Management Studio until it is refreshed.)

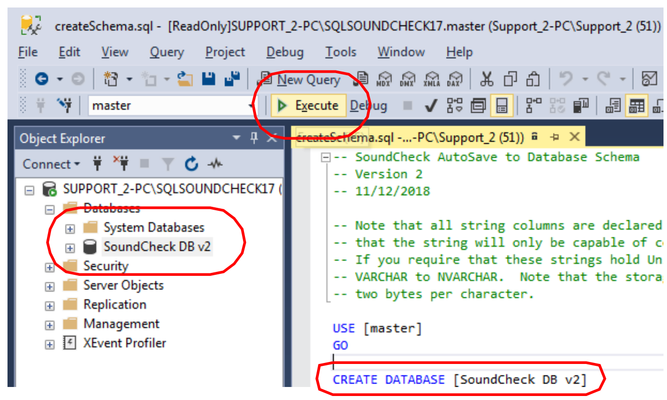


Figure 34-1: SQL Server Management Studio

Database Name

To change the name of the database edit the **createSchema.sql** file and click **Execute** to generate a new schema, e.g.:

Change: CREATE DATABASE [**SoundCheck DB v2**]

to: CREATE DATABASE [**Enter Custom Name DB**]

SoundCheck DB v2 Schema

The tables of the database are related as per the schema shown in *Figure 34-2*.

- Datarun table joins to Station table on the “Station_ID”
- Datarun table joins to Curve table on the “Curve_ID”

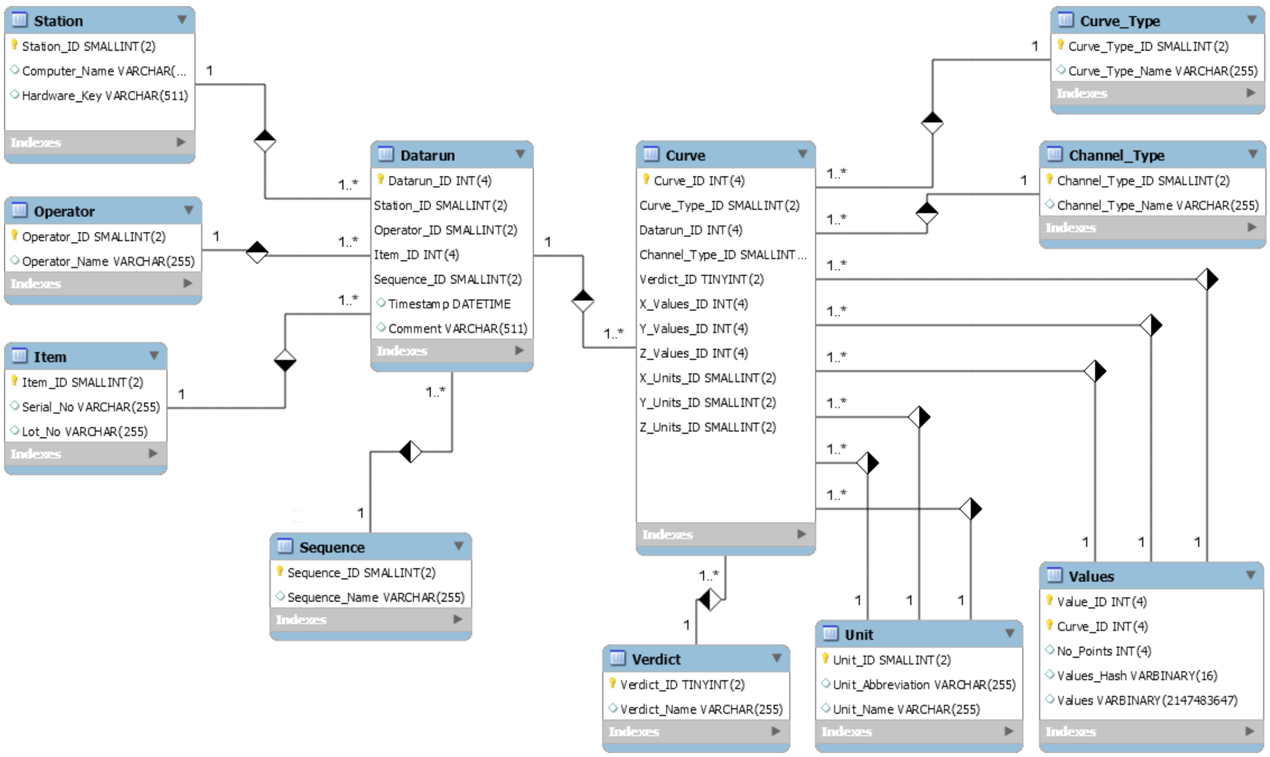


Figure 34-2: Database Schema

SQL Server Management Studio

This application shows the tables, their columns and the size of the tables, e.g.: [INT], [SMALLINT], [VARCHAR], etc. See ***Figure 34-3***.

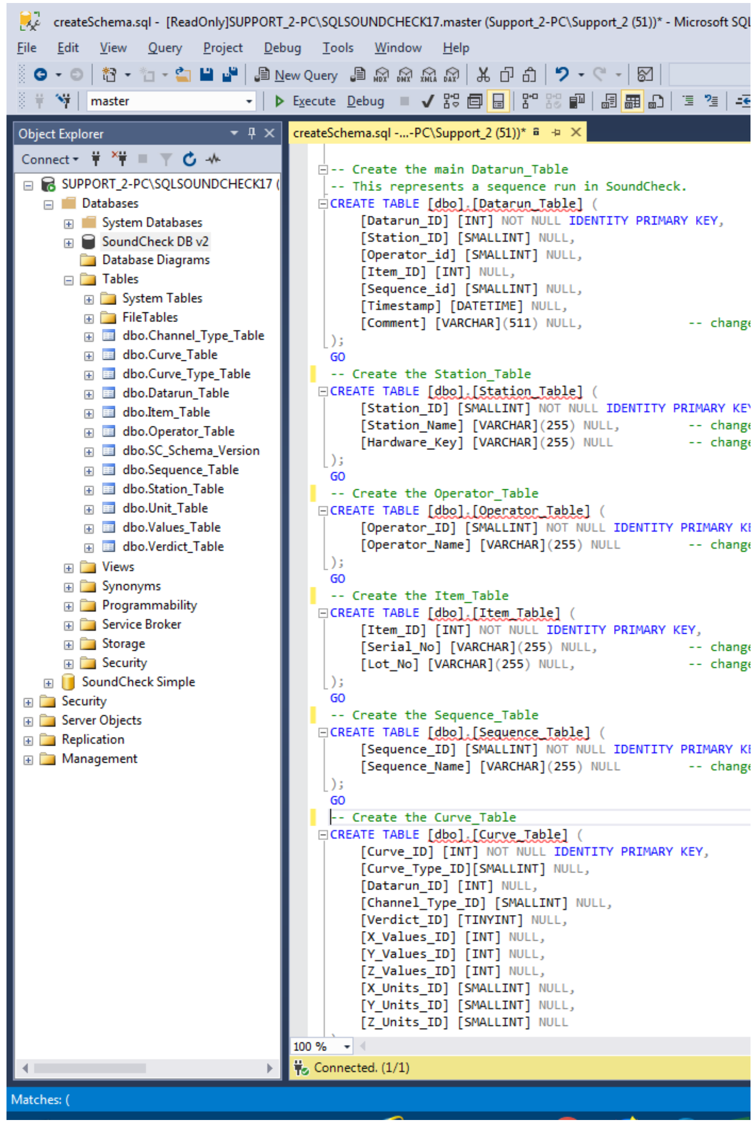


Figure 34-3: SQL Server Management Studio

Note: All string columns are declared as VARCHAR where each character takes up 1 byte. This implies that the string will only be capable of containing ASCII characters. If you require that these strings hold Unicode characters, which take up 2 bytes, change VARCHAR to NVARCHAR, IN ALL TABLES. This may be required for non-english language support.

Table Examples

The Curve_Table:

```
CREATE TABLE [dbo].[Curve_Table] (
        [Curve_ID] [INT] NOT NULL IDENTITY PRIMARY KEY,
        [Curve_Type_ID][SMALLINT] NULL,
        [Datarun_ID] [INT] NULL,
        [Channel_Type_ID] [SMALLINT] NULL,
        [Verdict_ID] [TINYINT] NULL,
        [X_Values_ID] [INT] NULL,
        [Y_Values_ID] [INT] NULL,
        [Z_Values_ID] [INT] NULL,
        [X_Units_ID] [SMALLINT] NULL,
        [Y_Units_ID] [SMALLINT] NULL,
        [Z_Units_ID] [SMALLINT] NULL
```

(XYZ)_Values_ID - The IDs for the tables that contain the Binary Large Objects (BLOBs)

Some tables are pre-populated, e.g.: the Curve_Type_Table is pre-poplulated with the “Fundamental” curve and the Channel_Type_Table is pre-populated with left and right channels, “L” and “R”.

The Values_Table:

```
CREATE TABLE [dbo].[Values_Table] (
        [Values_ID] [INT] NOT NULL
IDENTITY(0,1) PRIMARY KEY,
        [No_Points] [INT] NULL,
        [Values_Hash] [BINARY](16) NULL,

        [Values] [VARBINARY](MAX) NULL
```

- Identifies the row of data in the table

- Number of points in the x, y or z values for the curve

- Calculated identifier used internally to prevent duplicate data and speed up data identification

- Varbinary is the type of data used for storing BLOBs

Creating a UDL Definition for SQL Server

Follow these directions to create an UDL file that connects to SQL Server (any edition).

1. Make sure that a database that contains SoundCheck's schema has been created and available on the network.

   See ***Creating a Schema on page 537***.

   If you are using local storage, make sure that SQL Server Express is installed and the schema has been created.

2. Create a new Microsoft Data Link file from Windows Explorer. If this choice is not available, you can create an empty text file and rename it with a .udl extension, e.g.: ***SoundCheck_Data.udl***. The file will then be displayed as Microsoft Data Link document.

3. Double-click the ***SoundCheck_Data.udl*** file. The Data Link Properties dialog is displayed as in ***Figure 34-4***.

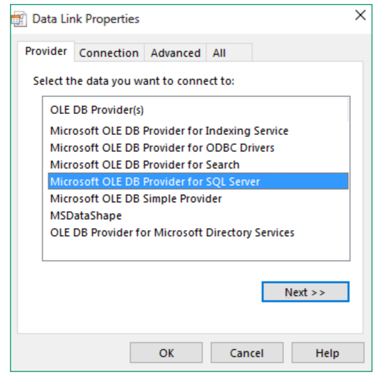


Figure 34-4: Provider Tab

4. Click on the **Provider tab**.

5. Select **Microsoft OLE DB Provider for SQL Server**.

6. Select **Next** > to go to the **Connection** tab.

   - Select a server from the drop-down list or enter a server name
   - Select the Use Windows NT Integrated security radio button, unless specified otherwise
   - Select the database that contains the SoundCheck schema, e.g.: **SoundCheck_Data**, from the drop-down list

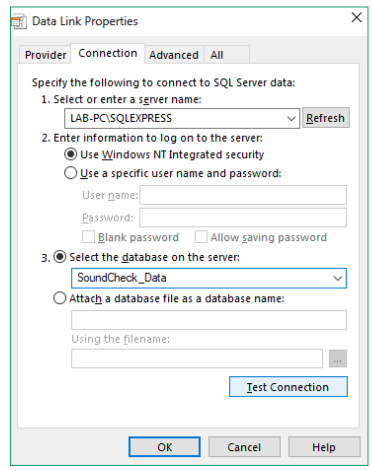
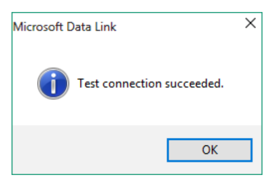


Figure 34-5: Connection Tab

7. Select the **Test Connection** button. A success dialog is displayed.



Creating an ODBC Connection for SQL Server

Follow these directions to create a **DSN** that connects to SQL Server (any edition).

Important: ***When using a database that requires SQL Server authentication with a Login ID and Password, you must use a "UDL" file. A DSN does not store the login ID and password.***

---

1. Make sure that a database that contains SoundCheck's schema has been created and is available on the network. If you are using local storage, make sure that SQL Server Express is installed and the schema is created. (See ***Creating a Schema on page 537***)

2. Open OBDC Data Source Administrator from the Control Panel. **Administrative Tools > Setup ODBC Data Sources (64-bit)** or search Program Files for "Data Sources (ODBC) 64 Bit"

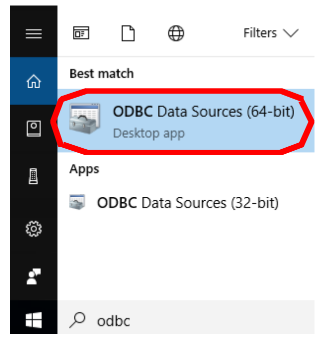


Figure 34-6: ODBC Data Sources

3. Select **System DSN** tab

   Select **Add**… button. The **Create New Data Source** dialog is displayed:

4. Select **SQL Server** from the list.

5. Select **Finish**. The Create a New Data Source to SQL Server dialog is displayed as in ***Figure 34-7***.

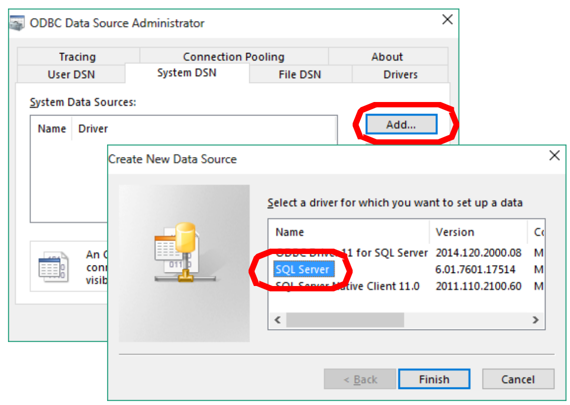


Figure 34-7: New Data Source

6. Enter a name and description for the selected data source as shown in ***Figure 34-8***.

7. In the **Server** drop-down list, select (local) for local storage, or select the database server on the network. This can be found through the drop-down list or by manually entering the location.

8. Do not click **Finish**. Select **Next** > to open Login verification.

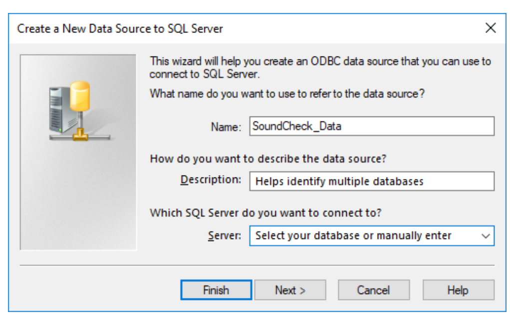


Figure 34-8: Name Data Source

9. The wizard prompts to determine how the SQL Server should verify the network login as shown in ***Figure 34-9***. Select **Next** > to accept the defaults, or consult your IT administrator.

Important: ***When using a database that requires SQL Server authentication with a Login ID and Password, you must use a "UDL" file. A DSN does not store the login ID and password.***

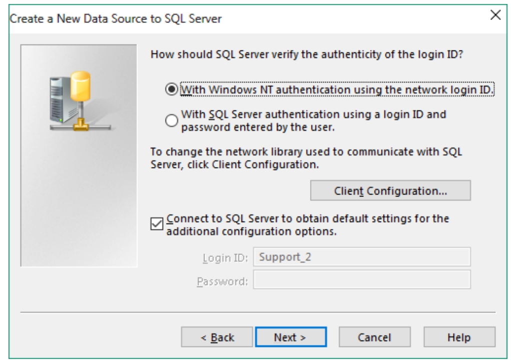


Figure 34-9: Verify Login

10. Check "**Change the default database to:**" and select the database that contains the SoundCheck schema, ***SoundCheck DB v2***, from the drop-down list as in ***Figure 34-10.***

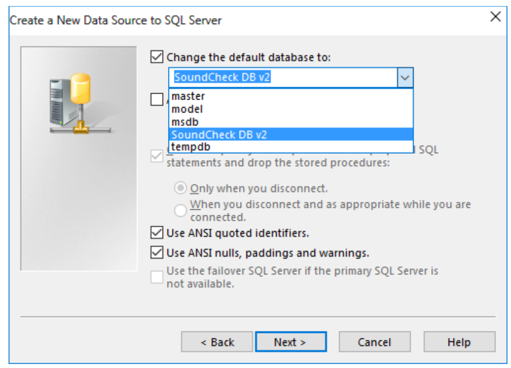


Figure 34-10: Select Database

11. Select **Next**. The next dialog in the wizard appears ***Figure 34-11.***

12. Unless specified by your IT administrator, use the defaults.

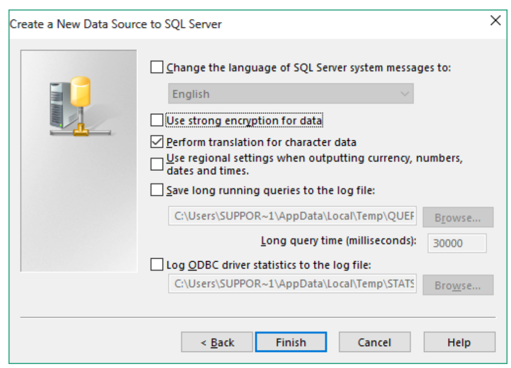


Figure 34-11: Use Defaults

13. Select **Finish**.

    A summary of the new ODBC data source is displayed as in ***Figure 34-12.***

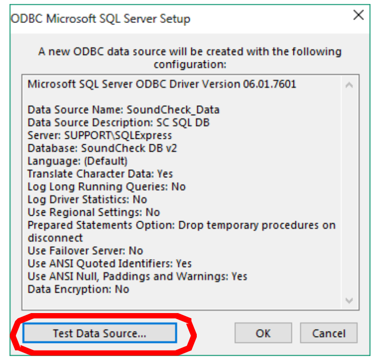


Figure 34-12: ODBC Data Source Summary

14. Select **Test Data Source** to make sure the configuration works.

15. A confirmation dialog is displayed as in ***Figure 34-13***.

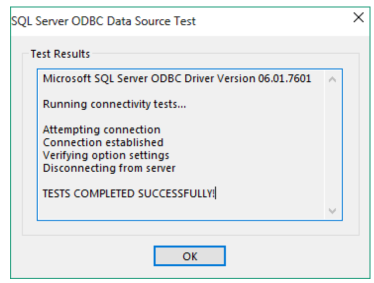


Figure 34-13: Test Confirmation

16. Select **OK** to close the windows. The new data source will be listed in the **System DSN** tab.

17. Select **OK** to close the OBDC Data Source Administrator.

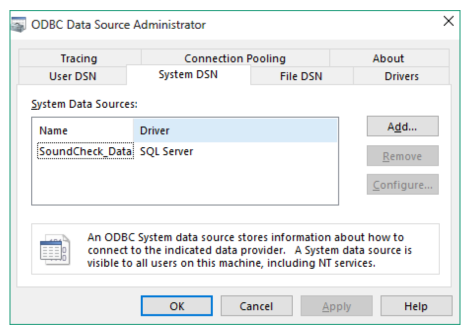
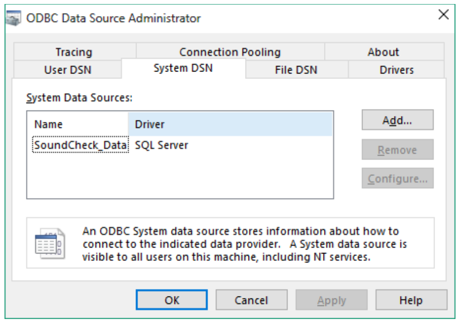


Figure 34-14: New System DSN

SoundCheck Autosave Database V2

Using DSN

- In the Autosave Step **UDL or DSN** field, **Type In** the System DSN name from the **ODBC Data Source Administrator > System DNS** tab as shown in ***Figure 34-15***.



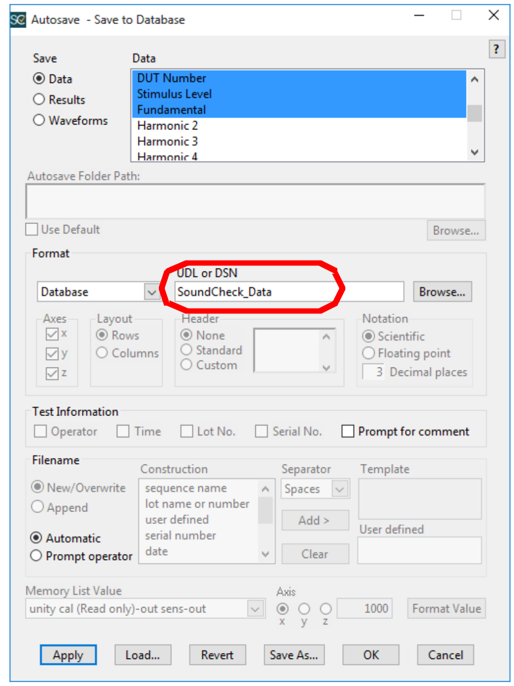


Figure 34-15: Type the DSN Name

Using UDL

- If using a password protected database, a UDL file must be used instead of a DSN. Click the **Browse** button to select the file, e.g.: SoundCheck_Data.udl as shown in ***Figure 34-16***.

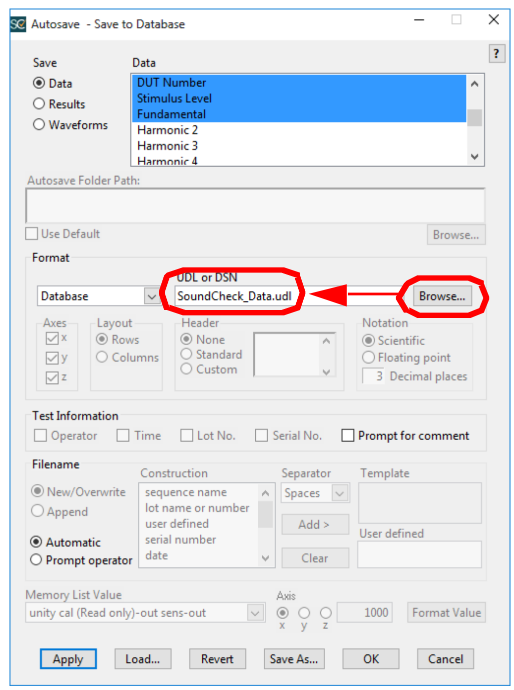


Figure 34-16: Browse to UDL File

Database Schema 1 for Access

Database V2 is not compatible with Microsoft Access.

An Access example database is included in **C:\SoundCheck 18\Database**. Note that data saved in Database Schema 1 is not compatible with Database V2.

If you are creating a new database, we recommend that you use Database V2 and follow the instructions in ***Database Setup on page 533***.

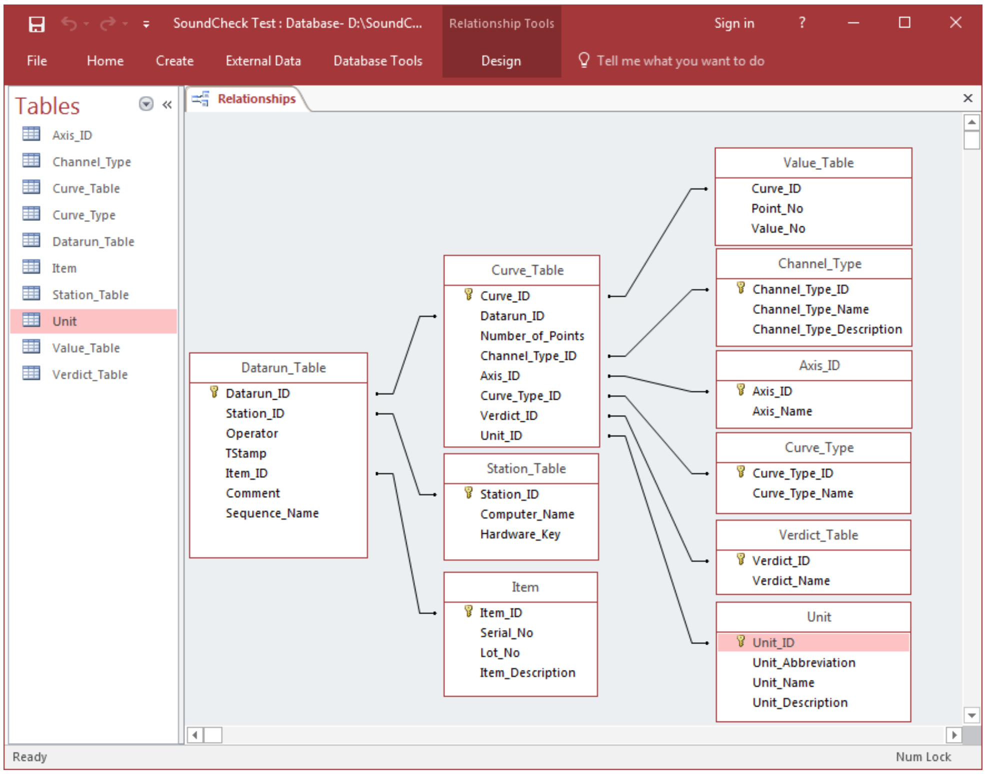


Figure 34-17: Database Schema 1 Access Table Relationships

Database Schema 1 Access table relationships are shown in ***Figure 34-15***. This provides for storage of information using MS Access but does not take advantage of optimizations offered in Database V2.

Creating an ODBC Connection for MS Access

Example Database

Rather than set up your own database, please use the one included with SoundCheck.

- Copy "C:\SoundCheck 18\Database**Blank SoundCheck Database.mdb**" and paste it in the desired location.
- Rename it so it is relevant to your application, e.g.: **SoundCheck_Data.mdb**
- Point your ODBC driver to it. The file must not be set to Read Only.

  If you start from our blank database, you will not need to open Access to start writing to the database.
- The database schema can be stored on the local hard drive or on a server.
- If you are using local storage, make sure SQL Server Express is installed.
- Use SoundCheck 18 with 64-bit MS Office 2010 (or later) installed
- Open OBDC Data Source Administrator from the Control Panel. **Administrative Tools > Setup ODBC Data Sources (64-bit)** or search Program Files for “Data Sources (ODBC) 64 Bit”.

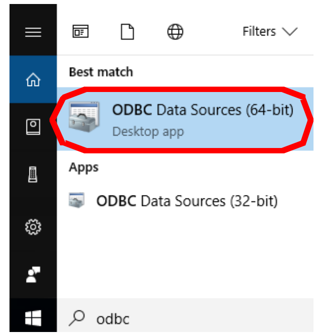


Figure 34-18: ODBC Data Sources

- Click the **System DSN** tab. See ***Figure 34-19***.
- Click **Add** to create a new DSN
- Select the **Microsoft Access Driver** from the list and click **Finish**

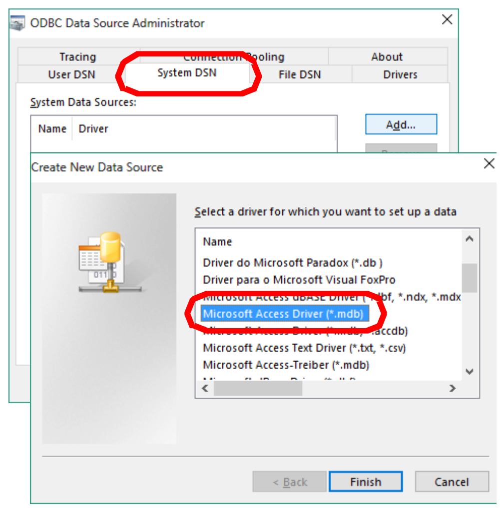


Figure 34-19: Create DSN

- In the **Data Source Name** field enter "SoundCheck DSN"
- In the Database section click **Select**
- Navigate to the location of your .MDB file: "C:\SoundCheck 18\Database\SoundCheck_Data.mdb"
- Click **OK** to exit out of the open windows

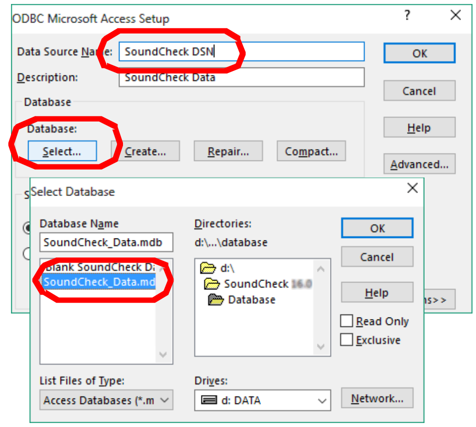


Figure 34-20: Select MDB

Creating a UDL Definition for MS Access

1. Create a new Microsoft Data Link file from Windows Explorer. If this choice is not available, you can create an empty text file and rename it with a .udl extension. The file will then be displayed as Microsoft Data Link document.

2. The Microsoft Jet 4.0 OLE Database Provider is not available in the 64-bit Windows operating system. See ***MS Access for Local Storage on page 535***.

3. Navigate to Start>All Programs>Accessories>Command Prompt

4. Type the following command:

   C:\Windows\syswow64\rundll32.exe "C:\Program Files\Common Files\System\Ole DB\oledb32.dll",OpenDSLFile C:\SoundCheck 18\Database\SoundCheck_Data.udl

- C:\SoundCheck 18\Database\SoundCheck_Data.udl is the file path to the UDL file you have created. This will open the UDL file.
- Click the **Provider** tab, the 64-bit database providers should now show up.

5. Double-click the file. The **Data Link Properties** dialog is displayed.

6. Click on the **Provider** tab as in ***Figure 34-21***.

7. Select a **Provider** that is compatible with Access.

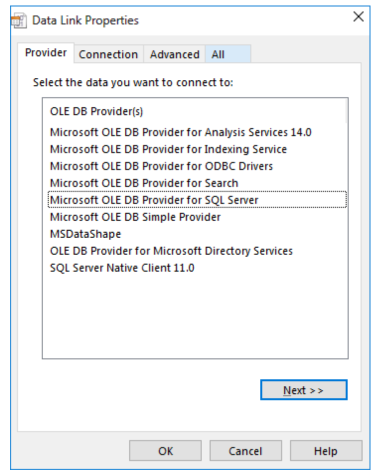


Figure 34-21: Provider Tab

8. Select **Next** > to go to the **Connection** tab.

   For Item 1, specify the Access .MDB file as in ***Figure 34-22***.

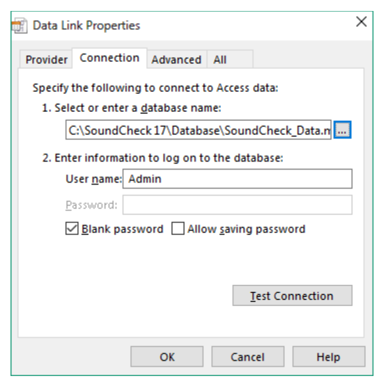


Figure 34-22: Connection Tab

9. Select **Test Connection** button. A success dialog is displayed:

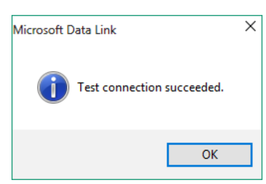


Figure 34-23: Success

SoundCheck Autosave for Access DB

Using DSN

- Open the default sequence, "**C:\SoundCheck 18\Sequences\How To examples\Autosave.sqc**"
- Edit the **Autosave to Database** step and manually enter "**SoundCheck DSN**" as the DSN name.

  Do not browse to the .MDB file.
- Run the sequence and store data to the Access database. Use Access to look at the **Datarun_Table** to confirm that data was written.

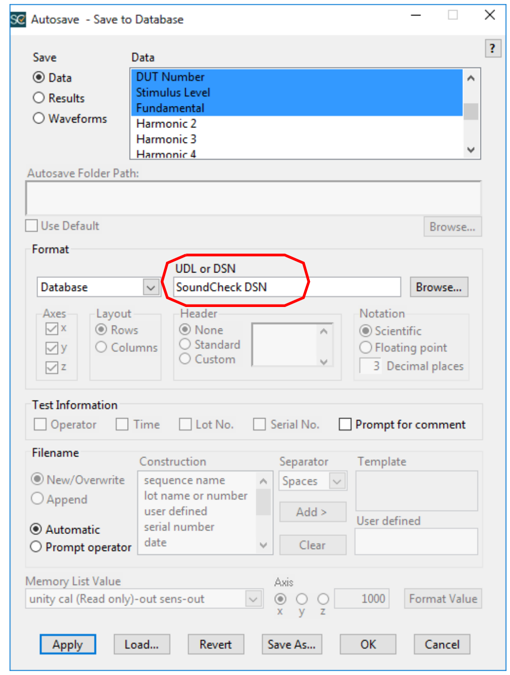


Figure 34-24: Autosave to Database

Using UDL

- If using a password protected database, a UDL file must be used instead of a DSN. Click the **Browse** button to select the file, e.g.: **SoundCheck_Data.udl** as shown in ***Figure 34-16***.

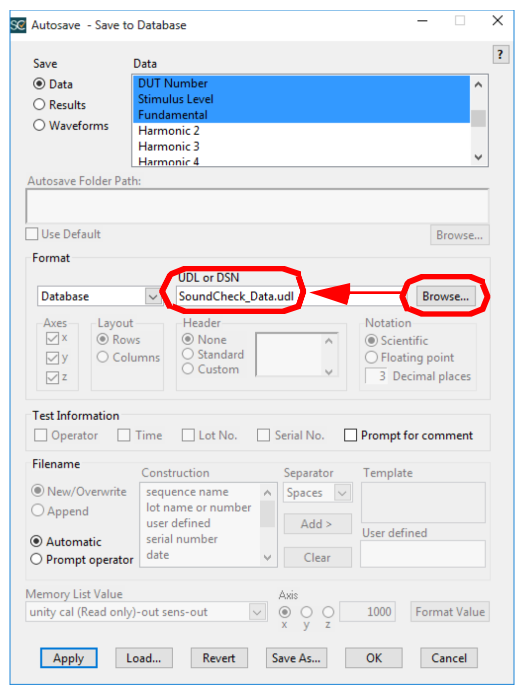


Figure 34-25: Browse to UDL File

Data File Format

SoundCheck® *.DAT and *.WFM file binary formats for most commonly used versions

1 DAT Binary Data File Format – SoundCheck 4.13 (DAT v2)

Key:

B=bytes, b=bits, uint=unsigned integer, float=floating point number in IEEE standard format

Note: Strings do not have a termination character

ONCE AT BEGINNING OF FILE

4B 32b uint number of curves (curve structures) in file

BEGINNING OF FIRST CURVE STRUCTURE

4B 32b uint number of bytes in curve structure: header, data, and descriptors

64-BYTE HEADER USED TO GET INFO Prior TO LabVIEW CLUSTER UNFLATTEN

16B string SoundCheck flattened cluster type, "Data", right padded with spaces

2B 16b uint SoundCheck version number for this cluster type

1B 8b uint number of dimensions in data array (obsolete in SoundCheck, now set to 0)

42B string curve name, right padded with spaces

3B reserved for SoundCheck, binary 0 for now

FLATTENED LabVIEW CLUSTER

4B 32b uint number of chars (N) in curve name

NB string curve name

4B 32b uint number of X-Y-Z data points (N) in curve

N*3*8B 64b float data points in X-Y-Z X-Y-Z X-Y-Z order

2B 16b uint Xdata (0=dB, 1=linear)

2B 16b uint Ydata (0=dB, 1=linear)

2B 16b uint Zdata (0=dB, 1=linear)

2B 16b uint Xaxis (0=log, 1=linear)

2B 16b uint Yaxis (0=log, 1=linear)

2B 16b uint Zaxis (0=log, 1=linear)

4B 32b uint number of chars (N) in Xprefix

NB string Xprefix SI prefix for unit

4B 32b uint number of chars (N) in Yprefix

NB string Yprefix SI prefix for unit

4B 32b uint number of chars (N) in Zprefix

NB string Zprefix SI prefix for unit

4B 32b uint number of chars (N) in Xunit

NB string Xunit

4B 32b uint number of chars (N) in Yunit

NB string Yunit

4B 32b uint number of chars (N) in Zunit

NB string Zunit

8B 64b float X dB ref

8B 64b float Y dB ref

8B 64b float Z dB ref

1B boolean single-value flag (0=normal curve, 1=single value)

BEGINNING OF SECOND CURVE STRUCTURE

4B 32b uint number of bytes in curve structure: header, data, and descriptors

2 DAT Binary Data File Format – SoundCheck 5.54 (DAT v3)

Key:

B=bytes, b=bits, uint=unsigned integer, float=floating point number in IEEE standard format

Note: Strings do not have a termination character

ONCE AT BEGINNING OF FILE

4B 32b uint number of curves (curve structures) in file

BEGINNING OF FIRST CURVE STRUCTURE

4B 32b uint number of bytes in curve structure: header, data, and descriptors

64-BYTE HEADER USED TO GET INFO Prior TO LabVIEW CLUSTER UNFLATTEN

16B string SoundCheck flattened cluster type, "Data", right padded with spaces

2B 16b uint SoundCheck version number for this cluster type

1B 8b uint number of dimensions in data array (obsolete in SoundCheck, now set to 0)

42B string curve name, right padded with spaces

3B reserved for SoundCheck, binary 0 for now

FLATTENED LabVIEW CLUSTER

4B 32b uint number of chars (N) in curve name

NB string curve name

4B 32b uint number of X-Y-Z data points (N) in curve

N*3*8B 64b float data points in X-Y-Z X-Y-Z X-Y-Z order

2B 16b uint Xdata (0=dB, 1=linear)

2B 16b uint Ydata (0=dB, 1=linear)

2B 16b uint Zdata (0=dB, 1=linear)

2B 16b uint Xaxis (0=log, 1=linear)

2B 16b uint Yaxis (0=log, 1=linear)

2B 16b uint Zaxis (0=log, 1=linear)

4B 32b uint number of chars (N) in Xprefix

NB string Xprefix SI prefix for unit

4B 32b uint number of chars (N) in Yprefix

NB string Yprefix SI prefix for unit

4B 32b uint number of chars (N) in Zprefix

NB string Zprefix SI prefix for unit

4B 32b uint number of chars (N) in Xunit

NB string Xunit

4B 32b uint number of chars (N) in Yunit

NB string Yunit

4B 32b uint number of chars (N) in Zunit

NB string Zunit

8B 64b float X dB ref

8B 64b float Y dB ref

8B 64b float Z dB ref

1B boolean single-value flag (0=normal curve, 1=single value)

1B boolean protected flag (0=unprotected, 1=protected)

1B boolean display X flag (0=do not display, 1=display)

1B boolean display Y flag (0=do not display, 1=display)

1B boolean display Z flag (0=do not display, 1=display)

4B 32b uint Plot Color (RGBa)

BEGINNING OF SECOND CURVE STRUCTURE

4B 32b uint number of bytes in curve structure: header, data, and descriptors

3 DAT Binary Data File Format SoundCheck 6.01-7.01 (DAT v6

Key:

B=bytes, b=bits, uint=unsigned integer, float=floating point number in IEEE standard format

---

Note: Strings do not have a termination character

ONCE AT BEGINNING OF FILE

4B 32b uint number of curves (curve structures) in file

BEGINNING OF FIRST CURVE STRUCTURE

4B 32b uint number of bytes in curve structure: header, data, and descriptors

64-BYTE HEADER USED TO GET INFO Prior TO LabVIEW CLUSTER UNFLATTEN

16B string SoundCheck flattened cluster type, "Data", right padded with spaces

2B 16b uint SoundCheck version number for this cluster type

1B 8b uint number of dimensions in data array (obsolete in SoundCheck, now set to 0)

42B string curve name, right padded with spaces

3B reserved for SoundCheck, binary 0 for now

FLATTENED LabVIEW CLUSTER

4B 32b uint number of chars (N) in curve name

NB string curve name

4B 32b uint number of X-Y-Z data points (N) in curve

N*3*8B 64b float data points in X-Y-Z X-Y-Z X-Y-Z order

2B 16b uint Xdata (0=dB, 1=linear)

2B 16b uint Ydata (0=dB, 1=linear)

2B 16b uint Zdata (0=dB, 1=linear)

2B 16b uint Xaxis (0=log, 1=linear)

2B 16b uint Yaxis (0=log, 1=linear)

2B 16b uint Zaxis (0=log, 1=linear)

4B 32b uint number of chars (N) in Xunit

NB string Xunit

4B 32b uint number of chars (N) in Yunit

NB string Yunit

4B 32b uint number of chars (N) in Zunit

NB string Zunit

8B 64b float X dB ref

8B 64b float Y dB ref

8B 64b float Z dB ref

1B boolean single-value flag (0=normal curve, 1=single value)

1B boolean protected flag (0=unprotected, 1=protected)

1B boolean display X flag (0=do not display, 1=display)

1B boolean display Y flag (0=do not display, 1=display)

1B boolean display Z flag (0=do not display, 1=display)

4B 32b uint Plot Color (RGBa)

1B 8b uint Plot Interpolation (0-5)

1B 8b uint Plot Point Style (0-16)

1B 8b uint Plot Line Style (0-4)

4B 32b uint Plot Point Color (RGBa)

1B 8b uint Plot Line Width (0-5)

1B 8b uint Plot Bar Plot Style (0-10)

2B 16b int Fill Baseline (-1 – 32767)

4B 32b uint number of chars (N) in Test Info

NB string Test Info

BEGINNING OF SECOND CURVE STRUCTURE

4B 32b uint number of bytes in curve structure: header, data, and descriptors

4 WFM Binary File Format – SoundCheck 6.01-7.01 (WFM v3)

Key:

B=bytes, b=bits, uint=unsigned integer, int = signed integer, float=floating point number in IEEE standard format

Note: Strings do not have a termination character

ONCE AT BEGINNING OF FILE

4B 32b uint number of waveforms (waveform structures) in file

BEGINNING OF FIRST WAVEFORM STRUCTURE

4B 32b uint number of bytes in waveform structure: header, data, and descriptors

64-BYTE HEADER USED TO GET INFO Prior TO LabVIEW CLUSTER UNFLATTEN

16B string SoundCheck flattened cluster type, "Waveform", right padded with spaces

2B 16b uint SoundCheck version number for this cluster type

1B 8b uint number of dimensions in waveform array (obsolete in SoundCheck, now set to 0)

42B string waveform name, right padded with spaces

3B reserved for SoundCheck, binary 0 for now

FLATTENED LabVIEW CLUSTER

4B 32b uint number of chars (N) in waveform name

NB string waveform name

8B 64b float X0 Waveform start value

8B 64b float dX Waveform increment

4B 32b uint number of points (N) in waveform

N*4B 32b float waveform points

2B 16b uint Xdata (0=dB, 1=linear)

2B 16b uint Ydata (0=dB, 1=linear)

2B 16b uint Yaxis (0=log, 1=linear)

4B 32b uint number of chars (N) in Xunit

NB string Xunit

4B 32b uint number of chars (N) in Yunit

NB string Yunit

8B 64b float Y dB ref

1B boolean display Y flag (0=do not display, 1=display)

1B boolean display X flag (0=do not display, 1=display)

1B boolean overload flag (0=no overload, 1=overload)

1B boolean protected flag (0=unprotected, 1=protected)

4B 32b uint number of flattened steps (N) in sequence history (for SoundCheck use)

N*(4B 32b uint number of chars (M) in flattened step string, MB string) (for SoundCheck use)

4B 32b int Waveform Channel Number (-1 – N Channels)

4B 32b uint Plot Color (RGBa)

1B 8b uint Plot Interpolation (0-5)

1B 8b uint Plot Point Style (0-16)

1B 8b uint Plot Line Style (0-4)

4B 32b uint Plot Point Color (RGBa)

1B 8b uint Plot Line Width (0-5)

1B 8b uint Plot Bar Plot Style (0-10)

2B 16b int Fill Baseline (-1 – 32767)

4B 32b uint number of chars (N) in Test Info

NB string Test Info

BEGINNING OF SECOND WAVEFORM STRUCTURE

4B 32b uint number of bytes in waveform structure: header, data, and descriptors

Appendix 1: Hardware Compatibility List

The devices in the following list are approved for use with SoundCheck. Other devices may be compatible but have not been verified for use with SoundCheck and are not supported.

SoundCheck is validated using Windows® 10 - 64-bit. No other versions of Windows are supported.

These audio interfaces allow for repeatable delay, therefore they can be used for measurement of absolute phase.

Audio Interface Current	Windows 10-64 Driver Type	Audio Interface Discontinued	Windows 10-64 Driver Type
AmpConnect 621	ASIO	**Fireface 800 (discontinued)**	ASIO
Lynx E44/E22	ASIO	**Multiface II (discontinued)**	ASIO
Lynx Aurora (n) 8/32 LT-USB & LT-TB	ASIO	**Lynx Aurora 8/16 LT-USB & LT-TB (discontinued)**	ASIO
Fireface UC	ASIO		
Fireface UCX USB & Firewire	ASIO		
Fireface 802 USB & Firewire	ASIO		
NI 4461 PCI & PXI	DAQmx		

The following audio interfaces require the use of *Auto Delay* in SoundCheck Analysis Steps to compensate for large and changing latencies. This is normal when using devices with WDM / WASAPI drivers. These devices should not be used when measuring absolute phase.

Audio Interface Current	Windows 10-64 Driver Type	Audio Interface Discontinued	Windows 10-64 Driver Type
AudioConnect 4x4	ASIO	**LynxTwo (discontinued)**	WDM [3,4]
AudioConnect & AmpConnect ISC	WDM [1,6]	**MultiFace II (discontinued)**	WDM [3]
DCC-1448	WDM [2]	**CardDeluxe (discontinued)**	ASIO Win 10 [3]
PQC-3048	WDM [2]		
PIO-9216	WDM [2]		
BTC-4148/4149 + BQC-4148/4149	WDM[5]		
Lynx E44/E22 & Aurora series	WDM [3,4]		
Fireface 802 USB	WDM		
NI cDAQ, NI 9260 & NI 9234 modules	DAQmx		

1. Can only be used at 44.1 kHz sampling rate for all operating systems.
2. Can only be used at 48 kHz sampling rate for all operating systems.
3. Sampling rate must be changed in the audio interface mixer/control panel program AND in the SoundCheck Hardware configuration.
4. MME drivers are not supported.
5. Can only be used at 44.1 kHz and 48 kHz sampling rates for all operating systems.
6. Does not support WASAPI driver.

For additional information on drivers see ***Approved Drivers - Windows on page 559***.

macOS®

These audio interfaces allow for repeatable delay which can be used for measurement of absolute phase.

Audio Interface	macOS - High Sierra 10.13.6 or later
AmpConnect 621	Core Audio
Lynx Aurora (n) 8-16 LT-USB	Core Audio
Lynx Aurora (n) 8-32 LT-TB	Lynx Core Audio Thunderbolt
Fireface UC	RME Intel Driver
Fireface UCX	RME Intel Driver
Fireface 802 USB	RME Intel Driver
Lynx Aurora 8-16 LT-USB (discontinued)	Core Audio
Lynx Aurora 8-16 LT-TB (discontinued)	Lynx Core Audio Thunderbolt
Fireface 800 (discontinued)	RME Intel Driver

The following audio interfaces require the use of Auto Delay in SoundCheck Analysis Steps to compensate for large and changing latencies. These devices should not be used when measuring absolute phase.

Audio Interface	macOS - High Sierra 10.13.6 or later
AudioConnect 4x4	Core Audio
AudioConnect	Core Audio [1]
AmpConnect ISC	Core Audio [1]
DCC-1448	Core Audio [2]
PQC-3048	Core Audio [2]
PIO-9216	Core Audio [2]
BTC-4148/4149 + BQC-4148/4149	Core Audio [3]

1. Can only be used at 44.1 kHz sampling rate for all operating systems.
2. Can only be used at 48 kHz sampling rate for all operating systems.
3. Can only be used at 44.1 kHz and 48 kHz sampling rates for all operating systems.

For additional information on drivers see ***Approved Drivers - Mac on page 560***.

Approved Drivers - Windows

MFG	Device	Connection	Driver Type	Driver Ver	Firmware	Default ASIO / USB Buffers	Default Latency		Default Chan Trim – In/Out	Sample Rate Auto Update
Listen	AmpConnect 621	USB	ASIO		0.68	Automatic per sample rate			NA	Yes
Listen	AmpConnect ISC (3)	USB	WDM	Native Win	3.3.0.1 (3) AUAB: 1.17	NA	250		NA	44.1 kHz only
Listen	AudioConnect	USB	WDM	Native Win	1.61	NA	250		NA	44.1 kHz only
Listen	AudioConnect 4x4	USB	ASIO	3.34	28, LTusb 10	2048 / Safe Set In ASIO (1)	5145		NA	Yes
Lynx Studio	Aurora (n) 8-32 LT-TB	Thunderbolt	ASIO	2.24B	1.7, 5.5	256	612		+4 dBu/+4 dBu	Yes
Lynx Studio	Aurora (n) 8-16 LT-USB	USB	ASIO	3.34	28, LTusb 10	1024 / Standard Set In ASIO (1)	2565		+4 dBu/+4 dBu	Yes
Lynx Studio	E44, E22	PCI	ASIO	2.24B	21	256	538		+4 dBu/+4 dBu	Yes
NI	NI 4461	PCI/PXI	DAQmx	DAQmx 19.6	NA	NA	108		NA	Yes
NI	cDAQ 9260 & 9234 modules	USB	DAQmx	DAQmx 19.6	NA	NA	71		NA	Yes
Portland Tool & Die (PTD)	DCC 1448	USB	WDM	Native Win	1.28	NA	250		NA	48 kHz only
PTD	PIO-9216	USB	WDM	Native Win	1.07	NA	NA		NA	48 kHz only
PTD	PQC-3048	USB	WDM	Native Win	1.27	NA	NA		NA	48 kHz only
PTD	BTC-4148/49, BQC-4148	USB	WDM	Native Win	1.32 1.8	NA	NA		NA	44.1 kHz & 48 kHz
RME Audio	Fireface 802	Firewire	ASIO	3.124	34/27/7	256	610		+4 dBu/+4 dBu	Yes
RME Audio	Fireface 802	USB	ASIO	1.168	13/9/7/9	1024	2116		+4 dBu/+4 dBu	Yes
RME Audio	Fireface UC	USB	ASIO	1.166	123/136	1024	2167	2114	-10/+4 (5)	Yes
RME Audio	Fireface UCX	Firewire	ASIO	3.125	46, 27	256	612		+4 dBu/+4 dBu (5)	Yes
RME Audio	Fireface UCX	USB	ASIO	1.168	46	1024	2116		+4 dBu/+4 dBu (5)	Yes

Note: The Fireface UC has two different pc board revisions. Units with serial number 23682169 and earlier will have a default latency of 2167. Units with serial number 23682170 and after will have a default latency of 2114. There is also a difference in Vp values. HAR files for both hardware versions are available in SoundCheck 16.01 and later. If you need to verify the Latency value, run the Self Test sequence as indicated in the Fireface UC setup instructions included with the driver from the Listen Website, ***https://support.listeninc.com/hc/en-us***.

See ***Footnotes on page 560***.

Approved Drivers - Mac

MFG	Device	Connection	Driver Type	Driver	Firmware	Default Latency	Default Chan Trim – In/Out	Sample Rate Auto Update
Listen	AmpConnect 621	USB	Core Audio	Native Mac	0.68	Automatic	NA	Yes
Listen	AudioConnect (4)	USB	Core Audio	Native Mac	1.61	0	NA	44.1 kHz only
Listen	AudioConnect 4x4 (4)	USB	Core Audio	Native Mac	28, LTusb 10	1208	+4 dBu/+4 dBu	No
Listen	AmpConnect ISC (3, 4)	USB	Core Audio	Native Mac	3.3.0.1 (3) AUAB: 1.17	0	NA	44.1 kHz only
Portland Tool & Die (PTD)	DCC 1448 (4)	USB	Core Audio	Native Mac	1.28	0	NA	48 kHz only
PTD	PIO-9216 (4)	USB	Core Audio	Native Mac	1.07	0	NA	48 kHz only
PTD	PQC-3048 (4)	USB	Core Audio	Native Mac	1.27	0	NA	48 kHz only
PTD	BTC-4148/49, BQC-4148 (4)	USB	Core Audio	Native Mac	1.32, 1.8	0	NA	44.1 kHz & 48 kHz
RME Audio	Fireface UC USB (5)	USB	RME USB	3.18	136/124	1265 1234	+4 dBu/+4 dBu	No
RME Audio	Fireface UCX USB	USB	RME USB	3.18	46	1269	+4 dBu/+4 dBu	No
RME Audio	Fireface 802 (5)	USB	RME USB	3.18	17/11/07	1273	+4 dBu/+4 dBu	No
Lynx Studio	Aurora (n) 8-32 LT-TB	Thunderbolt	Lynx TB	58i	1.7, 5.5	1218	+4 dBu/+4 dBu	No
Lynx Studio	Aurora (n) 8-16 LT-USB	USB	Core Audio	Native Mac	10	1207	+4 dBu/+4 dBu	No

Footnotes

See ***Hardware Configurations on page 563*** for individual audio interface settings.

(1) Requires change to audio interface defaults for proper operation. Follow setup guide provided with approved driver from www.listeninc.com.

(2) Windows 8.x is no longer supported

(3) For SoundCheck 10.11 and above, AmpConnect requires firmware version 3.2.4.6 or later. See ***AmpConnect ISC PN: 4042 on page 563***.

(4) Auto Delay **MUST** be used in SoundCheck Analysis Steps.

(5) Requires changes to default mixer settings.

Note: *The Fireface UC has two different pc board revisions. Units with serial number 23682169 and earlier will have a default latency of 1234. Units with serial number 23682170 and after will have a default latency of 1265. HAR files for both hardware versions are available in SoundCheck 16.01 and later. If you need to verify the Latency value, run the Self Test sequence as indicated in the Fireface UC setup instructions included with the driver from the Listen Website,* **https://support.listeninc.com/hc/en-us**.

Discontinued Hardware

The following hardware has been discontinued by the manufacturer.

Approved Drivers - Windows

MFG	Device	Connection	Driver Type	Driver Ver	Firmware	Default ASIO / USB Buffers	Default Latency	Default Chan Trim – In/Out	Sample Rate Auto Update
DAL	CardDeluxe Win 10 64Bit	PCI	ASIO	5.10.3523	NA	30mSec	2697	Set Jumpers: +4dBu In /-10dBV Out	No
DAL	CardDeluxe Win 7 64Bit	PCI	WDM	5.10.3523	NA	NA	250		No
Lynx Studio	Aurora 8-16 LT-USB	USB	ASIO	3.34	28, LTusb 10	1024 / Standard (1)	1232	+4 dBu/+4 dBu	Yes
Lynx Studio	Aurora 8-16 LT-TB	Thunderbolt	ASIO	2.24B	28, 6.2	256	1207	+4 dBu/+4 dBu	Yes
Lynx Studio	LynxTwo, A, C	PCI	ASIO	2.0.23J	Build 24	256 (1)	618	+4 dBu/+4 dBu	Yes
RME Audio	Fireface 800 (discontinued)	Firewire	ASIO	3.114	2.77	256	653	(5)	
RME Audio	Multiface	PCI, PCIe & PCMCIA	ASIO	4.06	55	256	623	Front panel switch	Yes

Approved Drivers - Mac

MFG	Device	Connection	Driver Type	Driver	Firmware	Default Latency	Default Chan Trim – In/Out	Sample Rate Auto Update
RME Audio	Fireface 800	Firewire	RME FW	3.27	33	1265		No
Lynx	Aurora 8-16 LT-TB	Thunderbolt	Lynx TB	Build 58i	2016.05.18	1207	+4 dBu/+4 dBu	No
Lynx	Aurora 8-16 LT-USB	USB	Core Audio	Native Mac	2015.07.23	1232	+4 dBu/+4 dBu	No

Footnotes

See ***Hardware Configurations on page 563*** for individual audio interface settings.

(1) Requires change to ASIO defaults for proper operation. Follow setup guide provided with approved driver from www.listeninc.com.

(2) Windows 8.x is no longer supported

(3) For SoundCheck 10.11 and above, AmpConnect requires firmware version 3.2.4.6 or later. See ***AmpConnect ISC PN: 4042 on page 563***.

(4) Auto Delay **MUST** be used in SoundCheck Analysis Steps.

(5) Requires changes to default mixer settings.

Minimum Computer Requirements - Windows

Before buying a series of new computers for use with SoundCheck, we recommend that you purchase a single computer so it can be tested with all the related hardware, including the audio interface. Test the audio interface using the SoundCheck Self Test sequence to insure that it is compatible with the computer. We recommend that you purchase a high quality computer according to the guidelines below.

Note that some computers may not be compatible with all audio interfaces.

- Supported operating systems:
  - Validated in Windows® 10 - 64-bit. No other versions of Windows are supported. SoundCheck may work in Windows® 7 - 64-bit but it is no longer validated.
  - Validated in macOS® Catalina 10.15. Versions of macOS prior to 10.13.6 are not supported.
- Celeron processors are not recommended
- To take advantage of using multiple virtual instruments, a multi-threaded processor is recommended, e.g.; Intel multicore processor / AMD multicore processor.
- 8 GB of RAM minimum (16 GB or more recommended for large WAV files or high resolution measurements below 50 Hz).
- 2 GB of free hard-disk space required for complete software installation
- Listen only supports Thunderbolt audio interfaces on Windows® 10. Windows® 10 has superior support for Thunderbolt devices, hot-plugging is possible and no 3rd party software is required.
- Dante support is only available for Windows® 10

Windows Computer Setup Recommendations

BIOS Settings: Hyper-threading, SpeedStep (Cool n Quiet) and C-States:

Problems may occur with audio and system performance on computers with Intel processors and motherboards using chipsets that employ Hyper-threading, SpeedStep (EIST) and/or C-State functions.

While these functions work to improve power management and energy saving, they can have detrimental effects on the performance of a computer used for SoundCheck.

If your system experiences performance issues, please follow the instructions below.

Speedstep – Allows the system to dynamically increase/decrease its clock speed between its minimum clock and its normal operating frequency, as well as voltage, in order to optimize for power consumption. While this helps save energy, it can unfortunately result in audio dropouts.

C-states – In order to save power, this reduces clock speed by adjusting the multiplier and to some extent, the processor voltage. With multicore processors this can result in a single core partially shutting down to multiple cores completely shutting down. This can cause large jumps in CPU usage as the processor adjusts to these changes while processing audio.

In BIOS, turn off SpeedStep and all C-States (C1E; C3; C6). This may require a BIOS update.

Dell computers do not always allow control of these functions in BIOS. Please contact Dell support for information on disabling Hyper-threading, SpeedStep and C-States.

Hyper-threading – This can cause problems with system performance when SoundCheck is running. It can interfere with real-time audio processes on some motherboards.

If the system is experiencing problems with performance we recommend that you shut off hyper-threading.

AMD Processors - Cool and Quiet is the equivalent to Speed Step and should be shut off along with C1E.

Windows Settings

Set Windows power management scheme to high-performance. When Windows tells the processor to go into low power mode, it can cause glitches in the audio stream. Please refer to: **Installing SoundCheck for Windows > Computer Setup**, in the SoundCheck Instruction Manual for more recommendations.

Hardware Configurations

Most audio interfaces cannot record and play simultaneously. There is almost always a delay between the two and the delay should not vary from measurement to measurement. The audio interfaces that Listen provides are certified to have high performance in making audio-related measurements. If you are using an audio interface that Listen, Inc has not certified, the measurement performance of SoundCheck may be severely compromised!

Important:* *Do not connect audio interfaces through USB hubs. Connect directly to computer USB port.

Listen Hardware

AmpConnect 621 PN: 4046

- Driver: Included in SoundCheck installer. Firmware: 0.68

AudioConnect PN: 4050

- Driver: Uses native USB audio driver in Windows. Large and changing latencies are to be expected. You must use Auto Delay in Analysis Steps. Firmware: 1.58 and later required.
- Prior to S/N 40501270108, the headphone output polarity is inverted for both channels. Polarity tests using the headphone out will need to be adjusted accordingly. Updating firmware to 1.61 corrects the polarity to match the audio outputs.

AmpConnect ISC PN: 4042

1. SoundCheck (or SoundCheck ONE) 10.11 as a minimum is required in order to control AmpConnect ISC™ via USB. AmpConnect ISC requires firmware version 3.2.4.6 or later to work with SoundCheck 10.11 and higher. AmpConnect units with S/N 367 and above have this firmware pre-installed. Units with S/N 366 and prior may require a firmware update. Please contact **support@listeninc.com** for instructions on determining the firmware version.

2. Units with S/N AC432 and after (AUAB audio board firmware: 1.14), operate at 44.1 kHz and 24 bit depth in Windows 7 and above. (Prior to that S/N, 44.1 and 48 kHz are supported) [Windows XP: AmpConnect can only be used at 44.1 kHz sampling rate AND 16-bit]

3. As of SoundCheck 13, a new driver for AmpConnect ISC has been included in the SoundCheck installation process. The new driver will not work in versions prior to SoundCheck 13. To use AmpConnect ISC with SoundCheck 12 (and previously supported versions), you will need to manually rollback the device driver in Windows Device Manager.

4. As of S/N 1536, the “3 dB Down Point” default jumper position is 2 Hz (Jumper removed).

5. As of serial number AC2402, the default jumper position sets the XLR inputs to Single Ended by putting a jumper across pins 2 & 3 of J20 and J28. This also makes the XLR input single ended since the jumper connects pin 3 of the XLR to pin 1 (common).

6. SoundCheck 14 requirements: Default Windows WDM audio driver, minimum of SoundCheck 14 control driver, minimum of firmware 3.2.4.6.

7. After installing SC 14, SC 13 users will not have control over the Headphone Amp. Other controls will work correctly. Additionally, the serial number of the AmpConnect audio interface will not be read properly, which changes the name of the device in the Hardware Editor.
8. Prior to S/N AC4277 (AUAB audio board firmware: 1.17), the headphone output polarity is inverted for both channels. Polarity tests using the headphone out will need to be adjusted accordingly.

AudioConnect 4 x 4 PN: 4051

- Driver: See ***Approved Drivers - Windows on page 559 and Approved Drivers - Mac on page 560***
- Firmware: Board 29, LT-USB 11
- Default ASIO buffer is 2048 & USB Streaming Mode is Safe (Hardware Step > ASIO control panel)
- Prior to SoundCheck 14, large and changing latencies are to be expected. You must use Auto Delay in Analysis Steps.

Portland Tool and Die

DCC-1448 PN: 5810 - MEMS Digital Microphone Measurement Interface Configuration

PQC-3048 PN: 5811 - Production Line MEMS Digital Microphone Measurement Interface

PIO-9216 PN: 5813 - Programmable Digital Serial Audio Data Interface

- Driver: Uses native USB audio driver in Windows and Core Audio in macOS
- Operates at 48 kHz and 24 bit depth (Select 16 bit depth if using Windows XP)

BTC-4148 PN: 5814 - Bluetooth Audio Measurement Interface

BTC-4149 PN: 5816 - Bluetooth Audio Measurement Interface

BQC-4148 PN: 5815 - Bluetooth Audio QC Interface

BQC-4149 PN: 5817 - Bluetooth Audio QC Interface

- Driver: Uses native USB audio driver in Windows and Core Audio in macOS
- Operates at 44.1 kHz & 48 kHz, and 24 bit depth (Select 16 bit depth if using Windows XP)

SoundCheck requires that input and output sample rates match.

You can either:

- Use 48 kHz for your output Hardware Configuration in SoundCheck

Or

- If the output configuration cannot be set to 48 kHz, for example because you are using an AmpConnect ISC, you can use **Re-sampling** and **Frequency Shift** Post-Processing steps in your sequence to align the stimulus and response waveforms.
- DCC-1448 can be used as a clock source by connecting its SPDIF Out to the SPDIF In of the SoundCheck audio interface. Then set the audio interface to synchronize its clock to the SPDIF Input. Doing so insures that input and output are synchronous and will insure that re-sample and frequency shift steps are not required.

Do not connect audio interfaces through USB hubs. Connect directly to computer USB port.

NI 4461 PCI and PXI

- Tested with DAQmx 19.6 which can be found on the SoundCheck DVD
- Install full version of DAQmx including Measurement and Automation Explorer

NI cDAQ, NI 9260 & NI 9234 modules

- Tested with DAQmx 19.6 which can be found on the SoundCheck DVD

- Install full version of DAQmx with Measurement and Automation Explorer
- Requires that SoundCheck sequences use Auto Delay on the Delay Tab of all Analysis Steps
- Triggered record acquisition is supported as of SoundCheck 16.1

LynxTwo/E44/E22

- Windows 10 – 64-bit
- Driver and Firmware refer to ***Approved Drivers - Windows on page 559***.
- **Important!** - Open the **ASIO Control Panel** from the SoundCheck **Hardware Configuration** Editor and then Turn Off "**Double Buffer Output**"
- If you see periodic drop outs in SoundCheck Acquisition, increase the buffer size to the next highest value. The latency value must be updated in the SoundCheck Hardware Editor.
- **Maximum Channels** - The default is "Unlimited". We recommend changing this to 4 or 8 channels in order to save system resources. This limits the virtual channels of the device and limits the number of channels that can be selected in the SoundCheck Hardware Editor. When using a 192 kHz sample rate in SoundCheck, this may be essential. Otherwise, severe dropouts may occur.
- **203 kHz Maximum Sample Rate** - The sample rate of 200 kHz is available in the Sample Rate field of the Hardware Table but is not valid for the Lynx TWO. Instead, use the Input and Output Tabs of the editor. There you can select "User" in the sample rate field of each channel and change the sample rate to 203 kHz.

Lynx Aurora (n) 8-16 LT-USB or 8-32 LT-TB Interface

- USB: Windows 10 – 64-bit. Thunderbolt: Windows 10 64-bit only
- Driver and Firmware refer to ***Approved Drivers - Windows on page 559***.
- LT-USB - Currently you must set the ASIO control panel to a buffer of 1024 and Safe in order to match the latency values provided in the default hardware (HAR) file. LT-TB does not use USB buffers.

Lynx Aurora 8-16 with LT-USB or LT-TB Interface - discontinued

- USB: Windows 10 – 64-bit. Thunderbolt: Windows 10 64-bit only
- Driver and Firmware refer to ***Approved Drivers - Windows on page 559***.
- LT-USB - Currently you must set the ASIO control panel to a buffer of 2048 and Safe in order to match the latency values provided in the default hardware (HAR) file. LT-TB does not use USB buffers.

Lynx issues with Intel motherboards using SpeedStep and C-States:

Problems can occur with PCI audio interfaces on Intel motherboards using chipsets that employ SpeedStep and C-State functions.

- In BIOS, turn off SpeedStep and all C-States (C1E; C3; C6) – may require BIOS update.
- Dell computers do not always allow control of these functions in BIOS. Please contact Dell support for information on disabling SpeedStep and C-States.

Set Windows power management scheme to high-performance. When Windows tells the processor to go into low power mode, it can cause glitches in the audio stream.

RME (Multiface II, Fireface UC/UCX, Fireface 800)

Driver versions tested (firmware update may be required).

- Driver and Firmware refer to ***Approved Drivers - Windows on page 559***.

Configuration Details

The Multiface II does not use Firewire. It requires an RME PCI, PCI Express, ExpressCard or CardBus interface card. **Do not connect it to a Firewire port!** It will damage the Multiface II box beyond repair.

- Fireface UCX and Fireface 800/802 are not compatible with all firewire chipsets: Texas Instruments and VIA chipsets are generally known to work.
- Fireface UC and UCX are both compatible with USB 3 (USB 2 transfer rate)
- Fireface 802 USB will produce small changes in latency. Auto Delay should be used in SoundCheck Analysis Steps.
- The Fireface UC has two different pc board revisions. Units with a serial number before 23682170 will have a default latency of 2167. Units with serial number 23682170 and after will have a default latency of 2114. The default HAR file provided with SoundCheck is for S/N 23682170 and later. See ***Approved Drivers - Windows on page 559***. If you need to verify the Latency value, run the Self Test sequence as indicated in the Fireface UC setup instructions included with the driver from the Listen Website, ***https://support.listeninc.com/hc/en-us***.

Do not connect audio interfaces through USB hubs. Connect directly to computer USB port.

- Windows 7 & 10: ASIO driver supported, WDM driver not recommended

PCI Express Interface Card for Laptop

- BIOS settings, "PCI Express Power Management" should be disabled

ASIO Configuration

- The sample rate of the RME interfaces automatically updates to the rate set in the SoundCheck Hardware Editor.
- In Windows 7, if using Fireface UCX or 800/802, you must Rollback the system's 1394 OHCI Compliant (firewire) device driver to the legacy version in Device Manager as shown below.

CardDeluxe - discontinued

Please install SoundCheck BEFORE installing the CardDeluxe drivers. The SoundCheck installation process sets the customized configuration of the CardDeluxe for use with SoundCheck.

If the CardDeluxe driver is already installed, you will need to configure the driver manually. Refer to the CardDeluxe instructions in the driver folder on the SoundCheck install DVD or on the Listen, Inc. website.

Windows 7-64 & 10-64 bit:

- Driver: Windows 7 Must use WDM Driver - Version 5.10.3523. Windows 10 Must use ASIO Driver - Version 5.10.3523.Note issues from chart on page 1 of this document
- Must also set the sample rate and the bit depth to 24-bit mode via the CardDeluxe control panel

AudioFire 12 - discontinued

- Driver: Version 5.8 (Firmware update may be required. Requires internet connection.)
- In Windows 7, you must Rollback the system's 1394 OHCI Compliant (firewire) device driver to the legacy version in Device Manager as shown under **IEEE 1394 Legacy Driver** (below).
- The sample rate must be set in the AudioFire mixer, settings tab. The sample rate selection in the SoundCheck Hardware Editor must agree with this setting.

Appendix 2: PXI/PCI 4461 Installation

Always use the latest NI DAQmx driver that has been approved for use with SoundCheck.

See "Hardware Compatibility List" on page 557.

- ***When SoundCheck is off or not running, the default state of the NI 4461 is "Open Outputs". This means that noise will be present on the outputs of the card.***

  As a safety precaution we recommend that you shut off any amplifier connected to the output of the NI 4461 before starting SoundCheck. The amplifier should be shut off before exiting SoundCheck as well.

- ***Please be advised that phase measurements are inconsistent from one system restart to the next.*** *This is a known issue to National Instruments.*
  - Output signal path calibration is recommended after every computer restart to account for phase changes in the NI 4461
  - **Apply Correction Out** must be selected in any Analysis Step that involves phase

PXI 1031 Chassis Identification

The following is for installation of the PXI 4461 card in a PXI 1031 Chassis.

Figure 2-1 shows the Chassis Identification process when using the PXI 1031 chassis and a PXI 4461 card.

- Click on the "Chassis Unidentified" device
- Select Identify As
- Select PXI-1031 from the list

This is not required when installing the PCI 4461 in a computer.

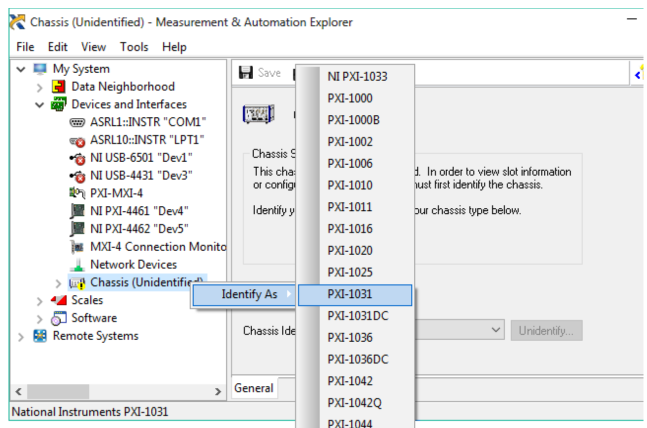


Figure 2-1: Chassis Identification for PXI 1031

Note: ***Some NI interface cards may require the PXI platform services driver. Please refer to the NI documentation.***

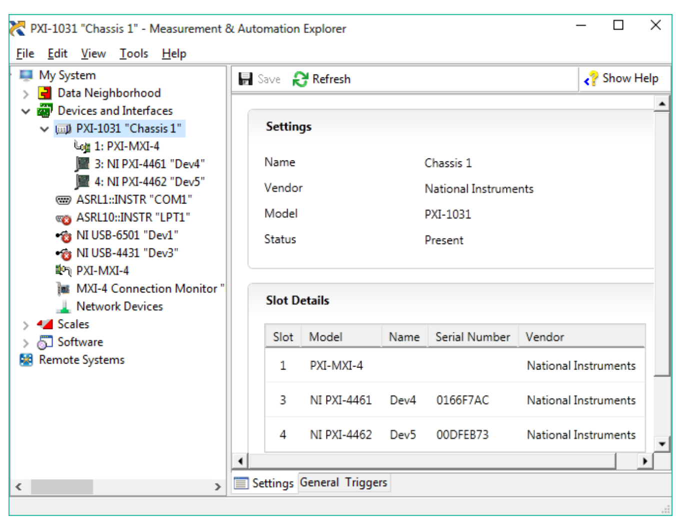


Figure 2-2: PXI 4461 recognized in NI Explorer

NI 4461 Install and Setup

Follow the instructions provided by NI for installing the NI 4461 card. The following screen shots are a step by walk-through of the NI 4461 install process. Instructions are included on each screen of the install process.

Open NI Max, Measurement and Automation Explorer to verify that the 4461 has been installed correctly.

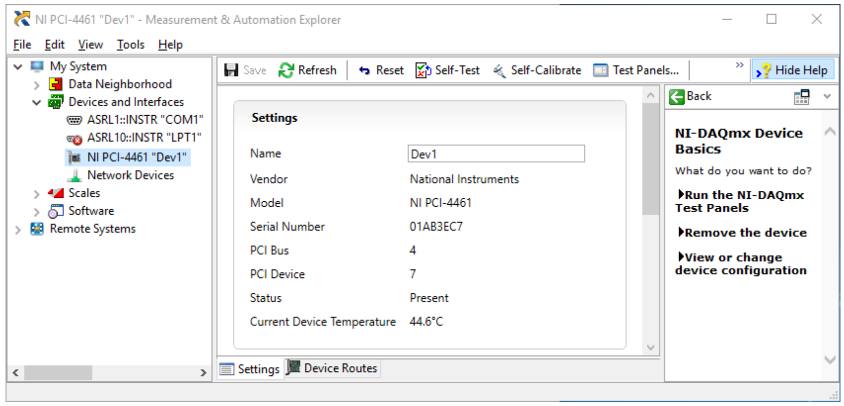


Figure 2-3: PCI 4461 DAQ-MX Properties

Self Calibration

Select the 4461 from the NI-DAQmx device list as shown in *Figure 2-4*. Click on the **Self-Calibrate** button at the right hand side of the window.

When the process is finished you should get a response indicating the device was calibrated successfully.

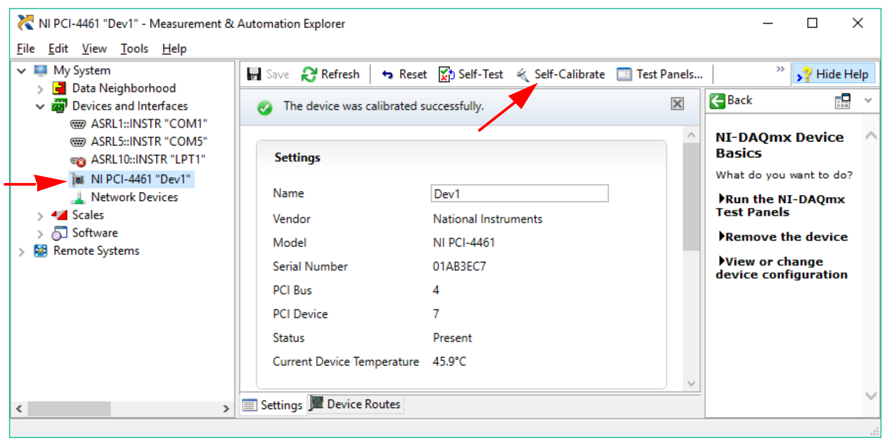


Figure 2-4: Self-Calibrate

Self Test

Self Test can be run by clicking on Self Test in the NI Max screen or by Right-clicking on the NI 4461 in the Device list and selecting **Self Test** from the drop-down list.

When the process is finished you should get a response indicating that Self Test was successful.

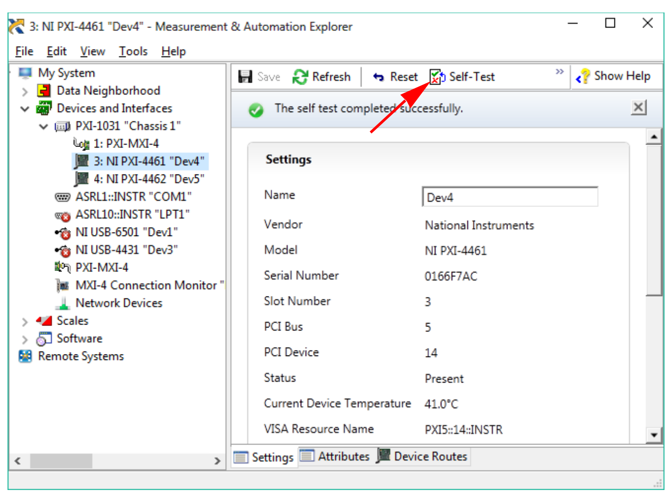


Figure 2-5: Self Test

SoundCheck® Hardware Configuration

The System Hardware configuration should be setup according to the following examples.

Setting the Vp values for the NI 4461 in the Hardware Editor actually sets the hardware input range and sends the Gain or Attenuation levels to the NI 4461 card. See ***Input and Output Vp settings on page 570*** **for more information.**

In the SoundCheck Hardware Editor, click the **Import Button** and select "**NI PCI-4461.Har**" from the appropriate operating system folder. Use the same file for the PXI-4461. This automatically sets up the Hardware Editor with the basic settings for the 4461 interface.

Depending on the number of DAQmx devices available, you may need to click on the **Device Fields** and select the appropriate Device Number from the list.

This should be the same Device ID as shown in the NI Explorer in ***Figure 2-3***.

Select the proper Input and Output channels from the **Select Ch Fields**.

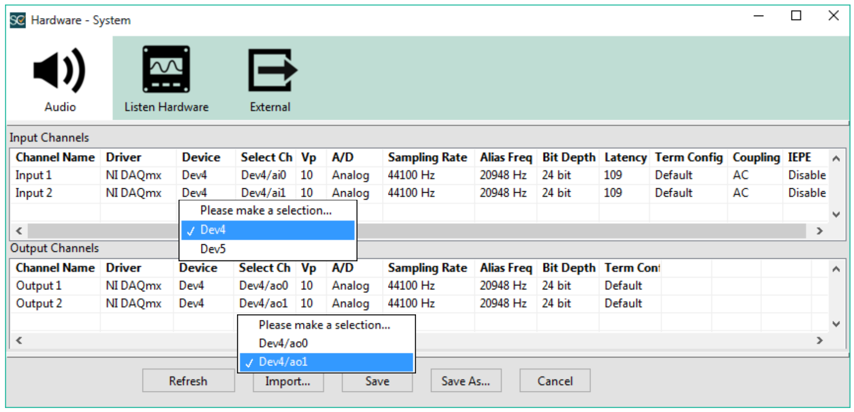


Figure 2-6: NI 4461 Hardware Editor

Vp Values

The In (Vp) and Out (Vp) values in the System Hardware configuration sets the resolution of the NI 4461.

If the signal level of the Device Under Test is low, select a low value from the list, e.g.: 1.

If the levels are high, select a larger value, such as 10. This is the default value in the **NI PCI-4461.Har** file.

Input Vp values can be different from Output Vp values. Refer to the table in ***Figure 2-7*** for a list of allowable Vp settings. Using values other than what are listed in the table will produce unreliable results.

Max FSD

Note: ***As you increase the input gain, the maximum input signal limit decreases proportionally. Check the Max FSD value in the Memory List to make sure the input of the NI 4461 is not clipping or under-loaded (signal too low which increases the noise floor). In general, if you are measuring distortion, Max FSD should be greater than -30 dB FSD. If Max FSD reaches 0 dB FSD the signal is being clipped.***

Hardware Defaults

- Sampling Rate: 44100 Hz (Only one rate can be selected for all Input and Output channels)
- Bit Depth: 24 Bit
- Latency: 109 (This will change if the Sampling Rate is changed. ***See "Sample Rate / Latency" on page 570.***)
- Term Config: Default
- Coupling: AC
- IEPE: Disable - Enable only if connecting an IEPE powered transducer to the input of the 4461.

Input and Output Vp settings

In the Hardware Configuration Editor you must enter the **Vp** value from the chart in ***Figure 2-7*** that corresponds to the Gain or Attenuation required on the NI 4461. These settings are sent to the NI 4461 after saving the Hardware Configuration.

Input Range

NI 4461 Gain (dB)	SoundCheck Hardware Setting Vp Value [1]
30	0.316
20	1.00
10	3.16
0	10.0
-10	31.6
-20	42.4

Output Range

NI 4461 Attenuation (dB)	SoundCheck Hardware Setting Vp Value [2]
0	10.00
-20	1.0
-40	0.1

Figure 2-7: Input and Output Vp

[1, 2] The input gain and output attenuation is set independently for each channel in System Hardware.

Latency and Sample Rate

The Latency of the 4461 will change as the sample rate of the System Hardware configuration changes. The following chart shows recommended Latency values for the sample rates supported in SoundCheck.

Sample Rate (Hz)	Latency (Samples)
200k	100
192,000	100
176,400	100
96,000	114
88,200	114
48,000	109
44,100	109
32,000	109
16,000	90
8,000	80

Figure 2-8: Sample Rate / Latency

Note: Output and Input sample rates must match in the Hardware Editor. The NI 4461 clock defaults to the output sample rate.

Appendix 3: Connection Procedures

Amp Calibration - Single Ended Connections

1. Make sure amplifier is OFF.

2. to 3. Connect Output 1 of the audio interface to the input of the amplifier. (amp channel 1)

4. to 5. Connect the output of the amplifier (amp channel 1) to Input 1 of the audio interface.

Important: ***The ground of the banana adapter must be connected to the amplifier minus (-) connection or a "loop" condition will occur that could result in damage to the amp and audio interface.***

6. Turn the amplifier **ON**.

Follow the procedure outlined in ***Amplifier Calibration Procedure on page 100***.

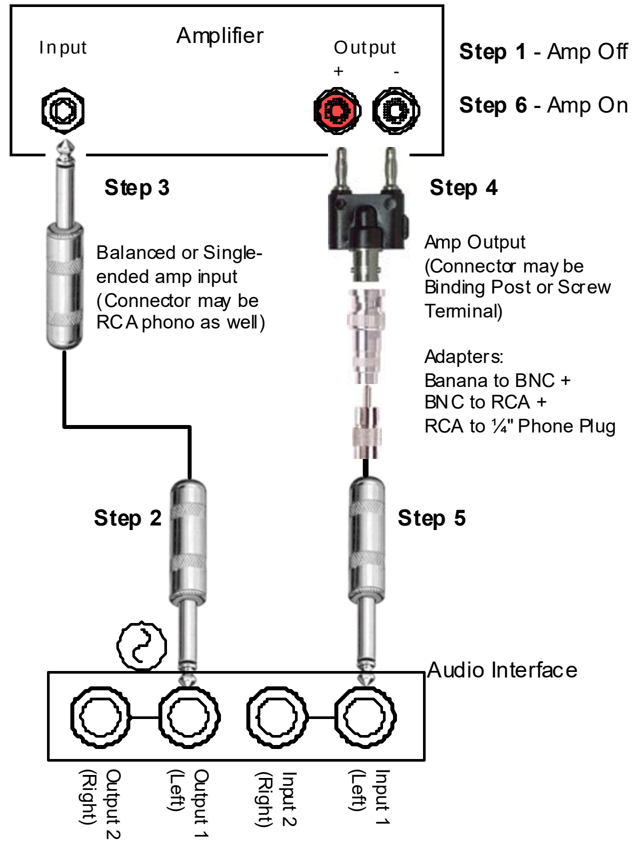


Figure 3-1: Amplifier Calibration Connections

Amp Calibration - XLR Balanced Connections

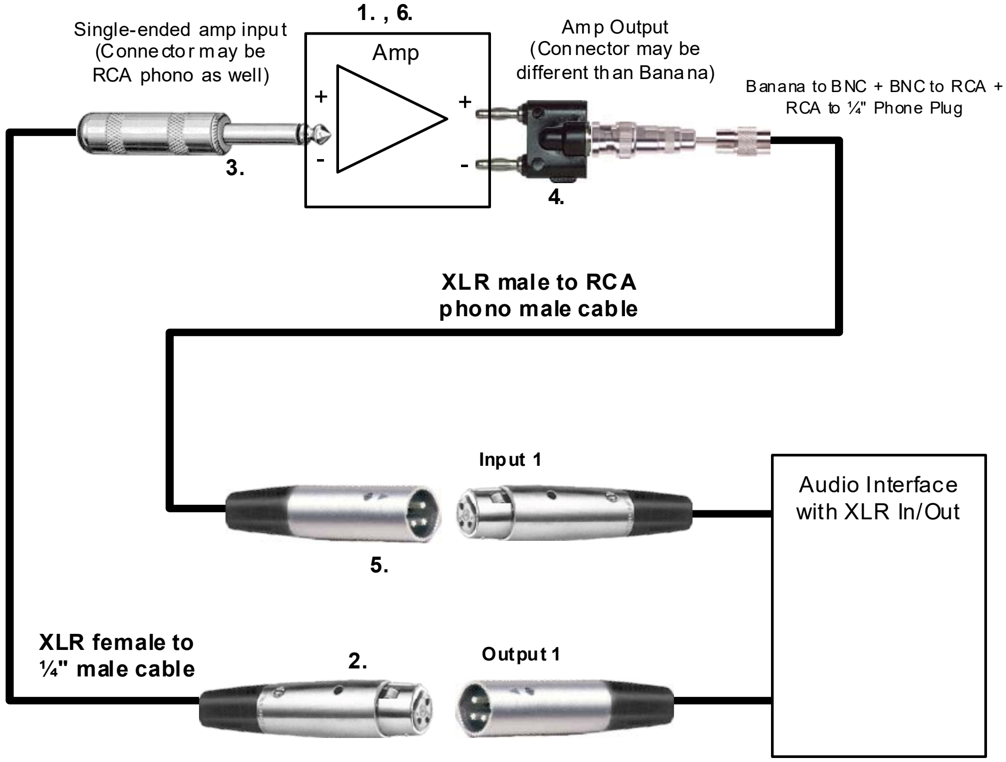


Figure 3-2: Amplifier Calibration Connections - Balanced

- **1.** Make sure amplifier is OFF.
- **2. to 3.** Connect Output 1 of the audio interface to the input of the amplifier. (amp channel 1) (Connection may be balanced or single ended.)
- **4. to 5.** Connect the output of the amplifier (amp channel 1) to the **I**nput 1 of the audio interface.

Note: The ground of the banana adapter (**4**) must be connected to the amplifier minus (-) connection or a "loop" condition will occur that could result in damage to the amp and audio interface.

- **6.** Turn the amplifier **ON**.
- Follow the amplifier calibration procedure outlined in the Calibration chapter: ***Amplifier Calibration Procedure on page 100***.
- In SoundCheck®, open the Calibration Configuration Editor. Do not open the Amplifier Calibration sequence. The Calibration Operation is run from the Calibration Editor.
- Select the Amplifier Output channel to calibrate. Make sure the Input Signal Path is set the proper Direct In as used in Step 4/5. It must be an Input Signal Path that is set for Unity Gain.
- Click the **Calibrate** button in the *Calibration Editor* to calibrate the amplifier.
- After receiving a **PASS** notification from the calibration sequence, click **Save As** to save the new amplifier gain settings to disk with a new name or close the *Calibration Editor* and click **File > Save** to save the entire sequence using the existing step name.

Amp Calibration - Bridged Connections

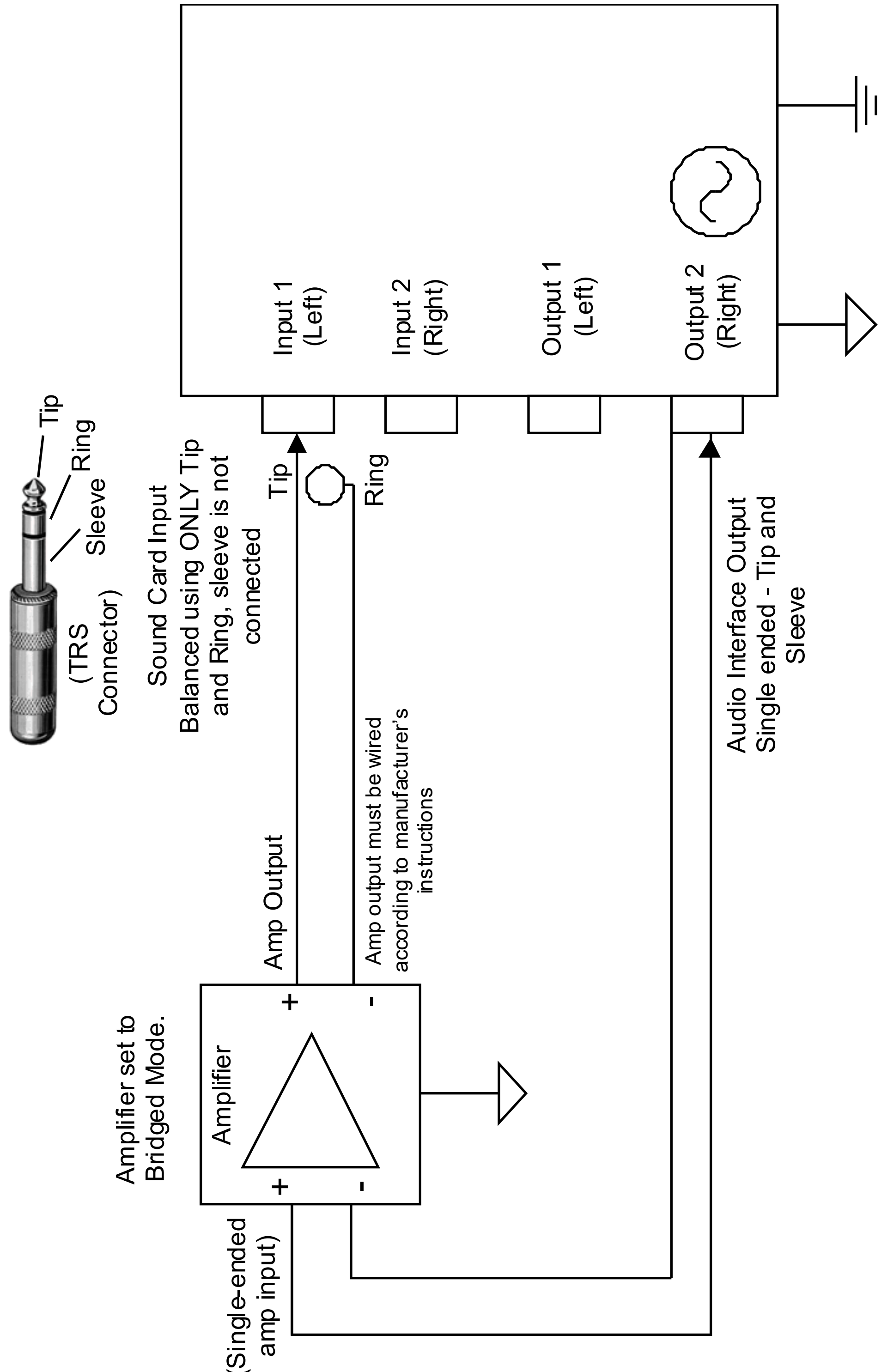


Figure 3-3: Bridged Amp Calibration Connections

Mic Calibration - SoundConnect™ Connections

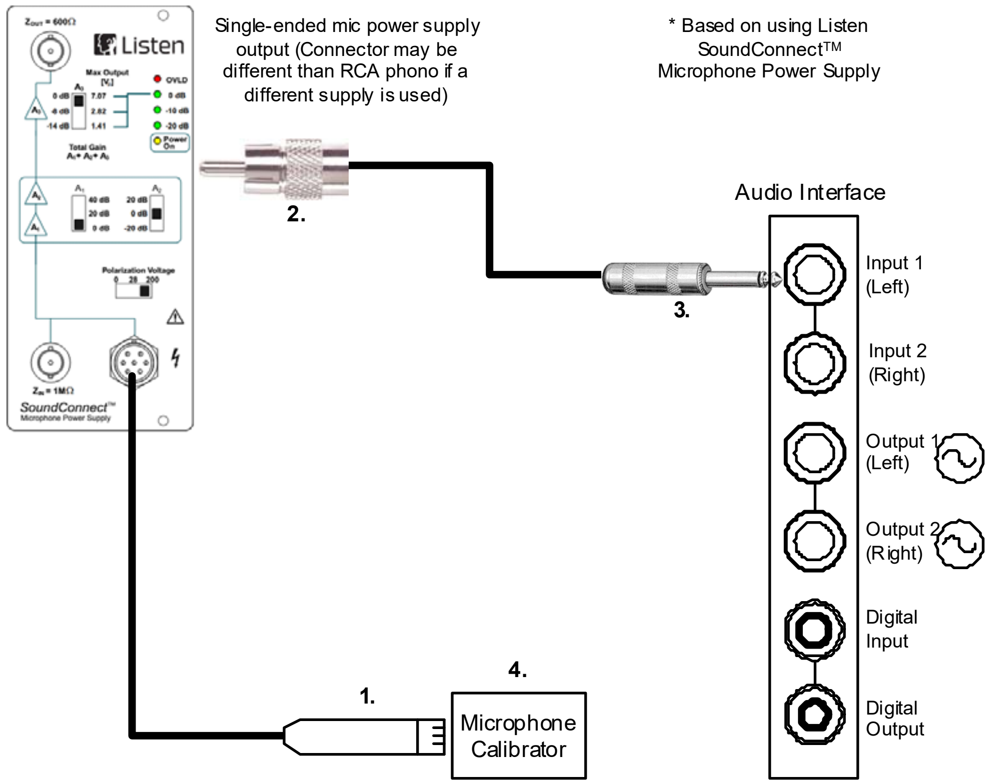


Figure 3-4: Microphone Calibration Connections

1. Plug microphone into front of SoundConnect. Make sure appropriate polarization voltage is selected.
2. Plug RCA phono (or other connector type appropriate for your power supply) to output connector on back of SoundConnect.
3. Plug 1/4-inch jack into Input 1 of audio interface. This should be an input with a Unity Gain Signal Path in the Calibration Configuration of SoundCheck.
4. Insert microphone into calibrator and turn the calibrator on.
5. Click **Calibrate** in the Input section of the SoundCheck® *Calibration Editor* to calibrate the microphone. Make sure units are V/Pa and dB re 20 µ.
6. After calibration is successful, click **OK** to close the Microphone Calibration window, then click Save to close the editor and save the new microphone gain settings to disk.

Loudspeaker Test Connections

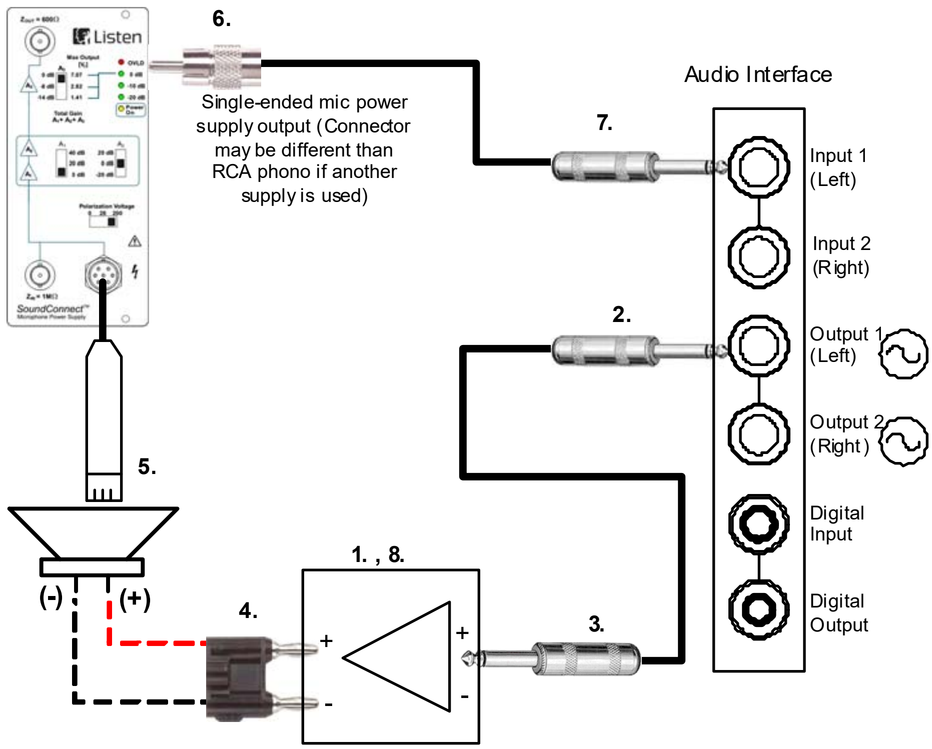


Figure 3-5: Loudspeaker Test Connections

- **1.** Make sure amplifier is OFF before connecting any cables.
- **2. to 3.** Connect Output 1 of the audio interface to the amp input.
- **4.** Connect the amp output to the DUT (loudspeaker). This example uses a banana connector.
- **5.** Plug the measurement microphone into the front of SoundConnect. Make sure the appropriate polarization voltage is selected and mic has been calibrated. (See ***Calibration Configuration on page 79.***)
- **6. to 7.** Connect the output of the mic power supply to Input 1 of the audio interface. Make sure the mic power supply is ON.
- **8.** Turn the amplifier ON. You are now ready to run your loudspeaker test sequence. Turn the amplifier OFF before shutting down the PC.

Note: If an external footswitch has been supplied, plug it into the computer's COM port (DB 9 connector on back of PC). Make sure the footswitch is enabled in the system Hardware Configuration. See ***Hardware Configuration on page 59***. Make sure that **Hardware Type > External Interface** is chosen and the selected interface is **Footswitch**. NI VISA is required and is installed during the SoundCheck installation. This is required for the footswitch to operate properly.

Loudspeaker Test Connections with Impedance Box

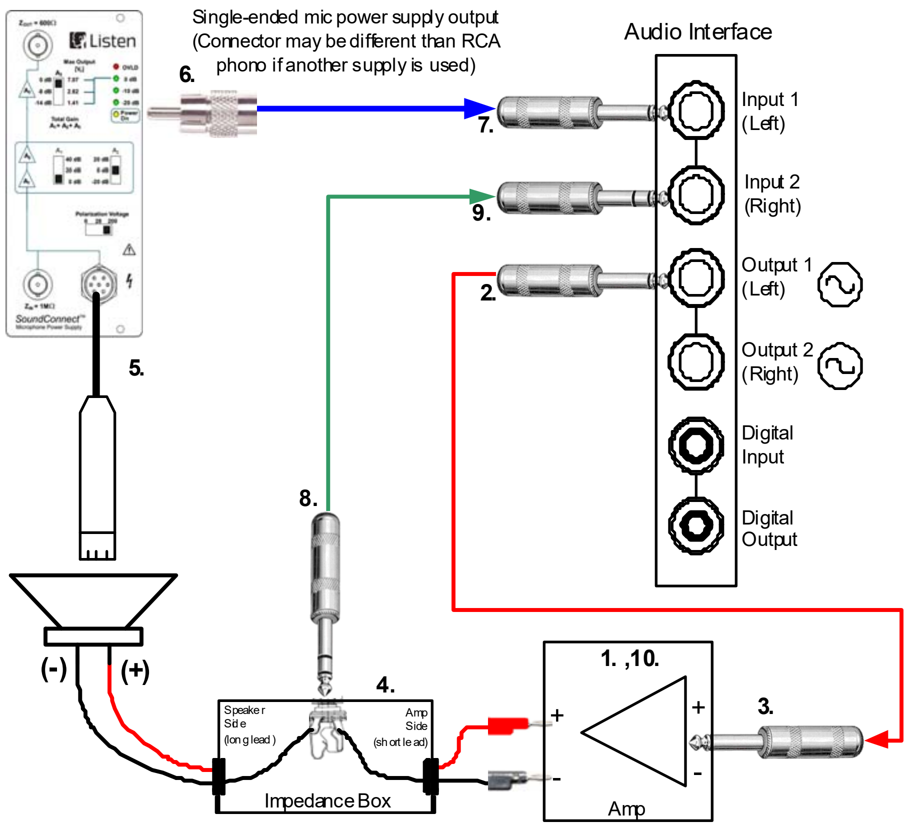


Figure 3-6: Loudspeaker Test Connections with Impedance Box

Important: ***The Impedance Measurement Interface shown in Figure 3-6: is available from Listen. Part number: 4009 with 1/4" connector cable and 4010 with XLR connector cable.***

- **1.** Make sure the amplifier is OFF before connecting any cables.

Note: The amplifier should be calibrated before making any measurements. Refer to the ***Calibration Configuration on page 79*** chapter for details.

- **2. to 3.** Connect Output 1 of the audio interface to the amplifier input.
- **4.** Connect the short Impedance Box leads to the amplifier output channel and then connect the long Impedance Box leads to the loudspeaker.
- **5.** Plug the microphone into the mic input of SoundConnect. Make sure the appropriate polarization voltage is selected and that the mic has been calibrated.
- **6. to 7.** Connect the RCA output on the back of SoundConnect to Input 1 of the audio interface. (The BNC output on the front of SoundConnect can be used as an alternative.) Make sure the mic power supply is ON.
- **8. to 9.** Use a TRS cable (Tip-Ring-Sleeve) to connect the Impedance Box output to Input 2 of the audio interface. (XLR audio interfaces will use a 1/4" TRS to XLR Male cable.)

- **10.** Turn the amplifier ON. You are now ready to run your loudspeaker test sequence. Turn the amplifier OFF before shutting down the PC to avoid unwanted transients from potentially damaging the loudspeaker.

Detailed Drawing of Impedance Box

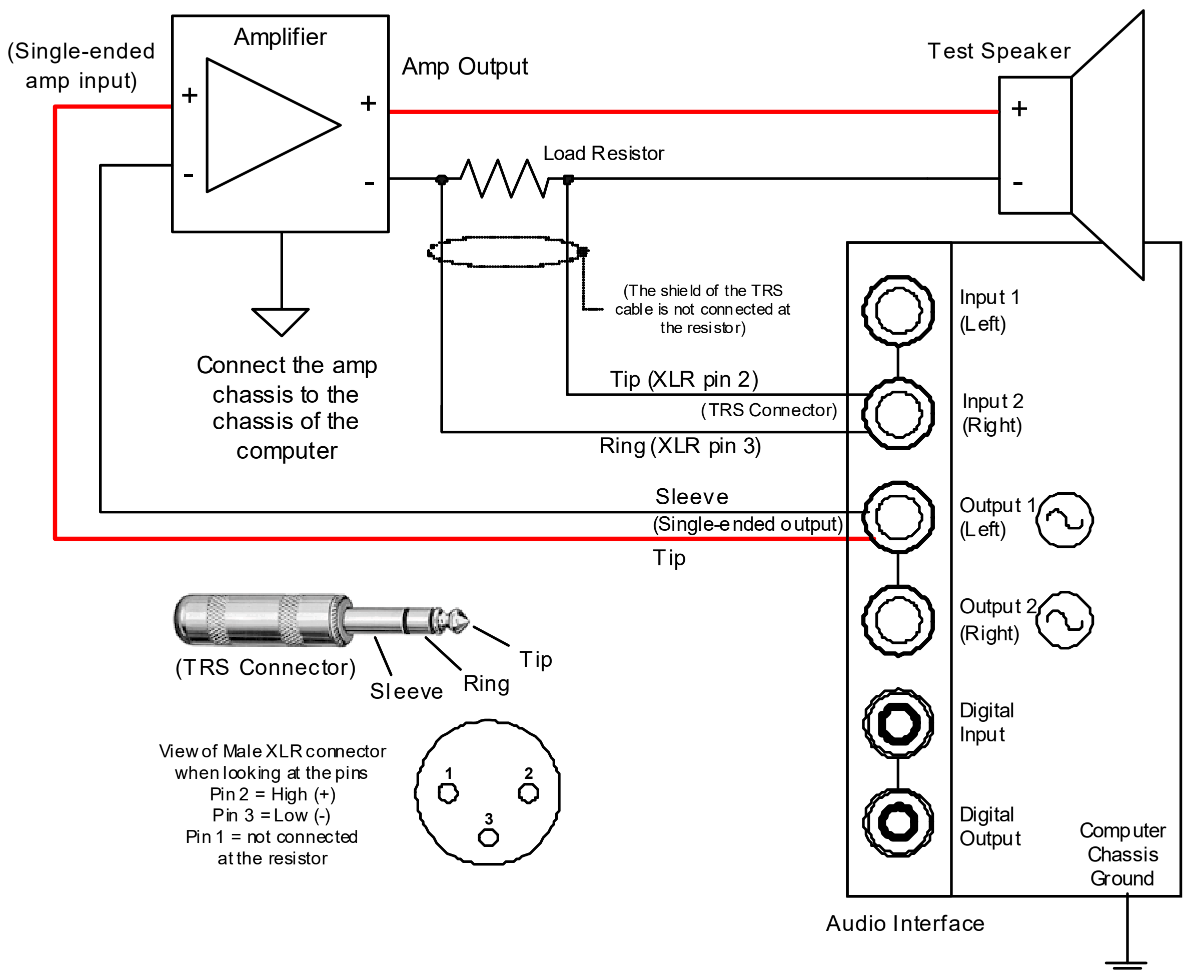


Figure 3-7: Impedance Box Layout

Important: ***In order to maintain a good signal to noise ratio, the Load Resistor value should not be more than 100 times different than the load being measured. Please refer to*** ***Impedance Setup on page 202*** ***for more information.***

Balanced Audio Interface Calibration Connections

When calibrating an audio interface with Balanced inputs and outputs it is important to follow the wiring procedure noted in ***Balanced Audio Interface Calibration Connection on page 578***.

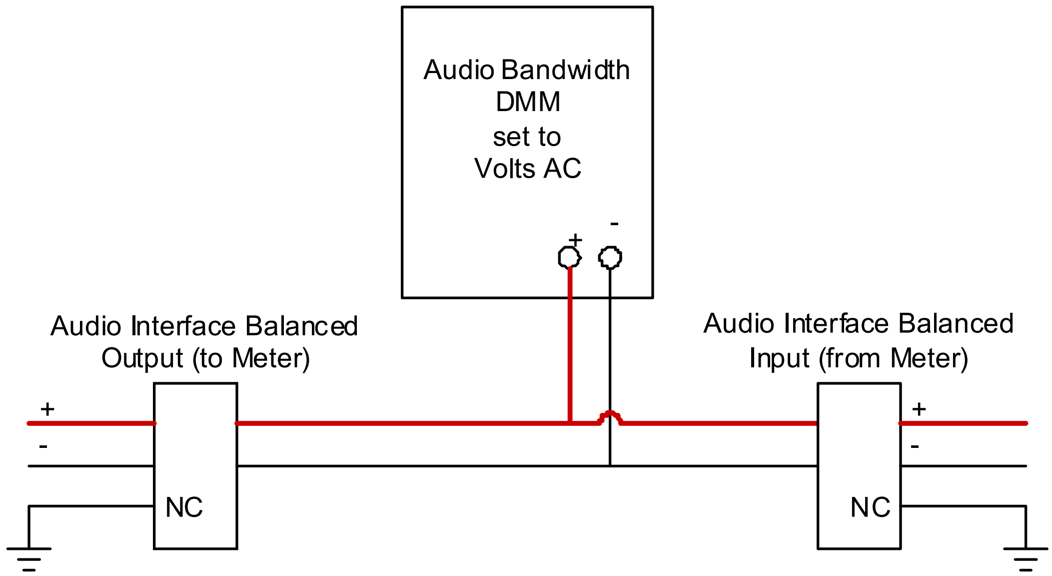


Figure 3-8: Balanced Audio Interface Calibration Connection

- For the purpose of calibration, do not connect the ground of the XLR connectors. Leave them "Floating". (NC = Not Connected)
- Do not short the Low (-) of a balanced output to ground. With some Active Output devices this will result in distortion on the High (+) signal.
- (The low (-) of the balanced input can be tied to ground. Shorting the low input insures that it is not a source of noise and usually causes no problem.)
- Note that many commonly available XLR to BNC adapters short pins 1 and 3 together internally. This can cause problems in the calibration process.

Important: ***When an audio interface is calibrated for Balanced mode but used in Single Ended mode, there will be a 6 dB drop in the output level. Only one line of the balanced output is being used. See*** ***Balanced Output to Single Ended Input on page 579.***

When calibrating and/or using a Balanced Audio Interface in Single Ended mode it is important to follow the guidelines outlined in the following section.

Balanced vs Single Ended Connections

Single Ended or Unbalanced outputs can typically be connected to Balanced inputs in either of the two methods shown in ***Figure 3-9*** and ***Figure 3-10***.

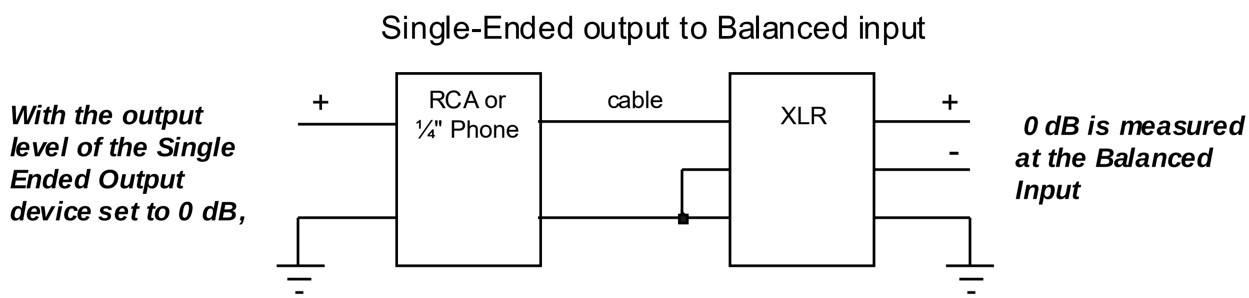


Figure 3-9: Single Ended Output to Balanced Input 1

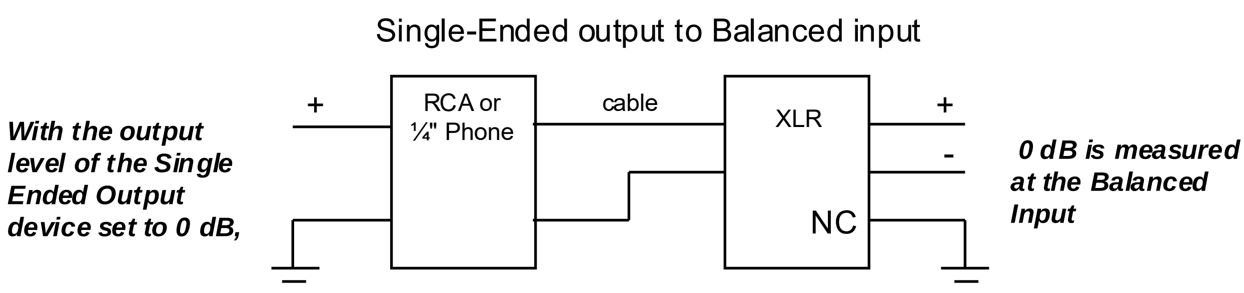


Figure 3-10: Single Ended Output to Balanced Input 2

Balanced output to Single Ended input wiring is a different matter. It is important to not short the Low (-) of a balanced output to ground. With some devices with Active Balanced Outputs this will result in distortion on the High (+) signal. (Transformer balanced outputs can have the low connected to ground.)

Figure 3-11 shows the suggested wiring for connecting a Balanced output to a Single Ended input. The measured level at the Single Ended input is down by 6 dB since only one line of the Balanced output it used.

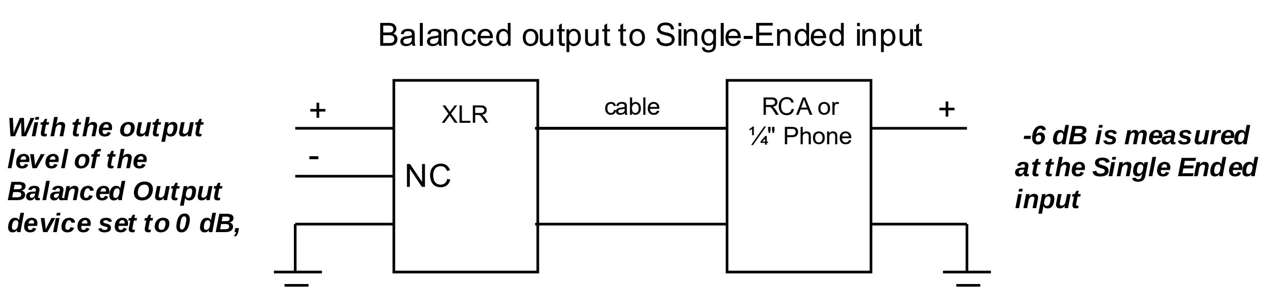


Figure 3-11: Balanced Output to Single Ended Input

Important: ***(NC = Not Connected)***

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Appendix 4: Serial Port Control

Note that many computers no longer have 9 Pin Serial Ports. You can also use a USB to Serial Adapter to take advantage of the functions in this chapter. At the time of this publication, the recommended adapters would be:

- Trendnet TU-S9 USB to Serial Adapter
- Dynex DX-UBDB9 USB to Serial Adapter
- USB to Serial Adapter using the PL2303 Prolific chip set and driver

Footswitch and Buzzer Control Via Serial Port

Important: ***Use of the footswitch and buzzer with SoundCheck® requires that NI Visa is installed on the system. This can be found on the SoundCheck installation CD under Additional Software. (Footswitch and Buzzer control cannot be used with Windows NT.)***

Serial Port Pin Out Definition

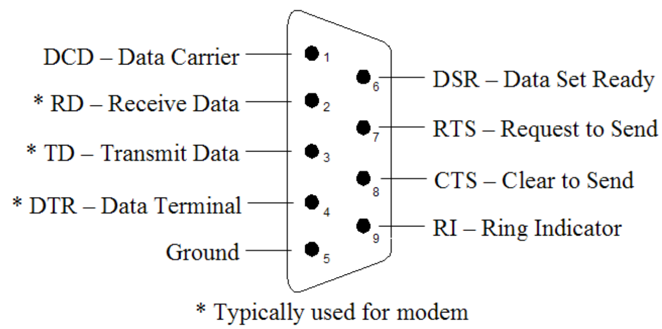

Figure 4-1: Serial Port Pin Out

Buzzer On/Off Message

The output of the serial ports of the computer can be used to control remote devices such as a Piezo Buzzer. The voltages are generally 11.2 VDC when the Line is high and -11.2 VDC when the line is low. The standard wiring for a Piezoelectric Buzzer is to connect the Positive lead to the DTR Line and the Negative lead to Ground. Similarly, a second device can be connected across the RTS Line and Ground to receive completely separate On/Off messages. Footswitches or other types of external devices cannot be used with a second buzzer on the same COM port, e.g., Buzzer A and Buzzer B can be on COM Port 1 and Footswitch 1 and Footswitch 2 can be on COM Port 2.

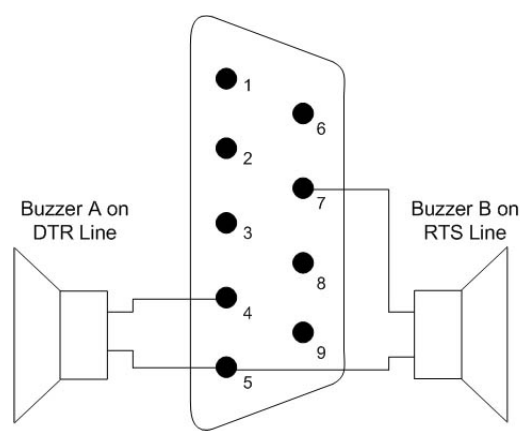

Figure 4-2: Buzzer A and B Wiring

Important: **USB serial port adapters can be used but not all adapters are compatible with NI Visa.**

Insert a Message Step into a sequence and rename it "**Buzzer ON**".

Right-click the step and select **Configure Step**. Check Display step when run and set the time to 0.0 seconds.

Select "Wait for confirmation" only if the buzzer is to remain on until the operator selects "Continue" on the Main Screen

Check-off "Display step when run" and set the time to 0.0 seconds

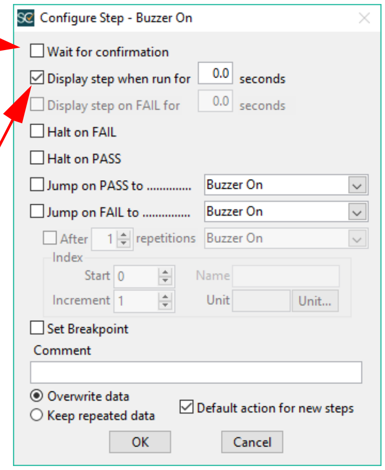

Figure 4-3: Configure Message Step

Open the SoundCheck Hardware Editor from the SoundCheck Main Screen.

- Select the **External** page
- Enter the **COM Port** number for the Serial Port that is used. This can be found in the Windows Device Manager.
- In this case the Serial port is 9
- Under Type select Buzzer
- Click **Save** to close the editor

The Interface settings are simply added to the System Hardware Configuration.

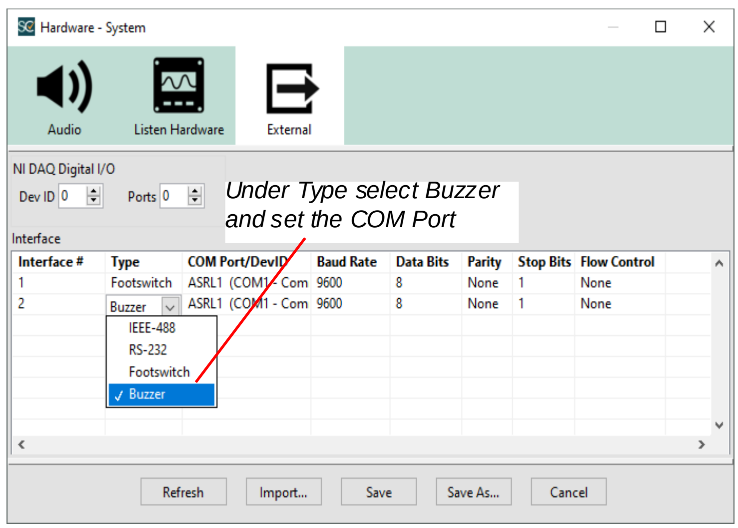

Figure 4-4: Hardware Config - Interfaces

Setting the Message steps in the Sequence

Double click on the **Buzzer On** step on the right of the *Sequence Editor*. Function should be set to **Set Control Lines**. **Interface**, Number **2** and **DTR high** are selected. Under **Setup** the following should be selected: **Pass/Fail** and **Wait**. The wait time can then be set for the length of time the buzzer should sound. This will also be the amount of time before the next step of the sequence is executed.

- Select Interface
- Select Interface number (must be setup in Hardware Editor - External page)
- Select Pass/Fail - this is the state that the message will report when the step runs
- Select Wait time - this is the amount of time in mSec that the step will wait before moving to the next step in the sequence. This is the length of time the buzzer will sound.
- Function - Select “Set Control Lines”
- Select DTR high

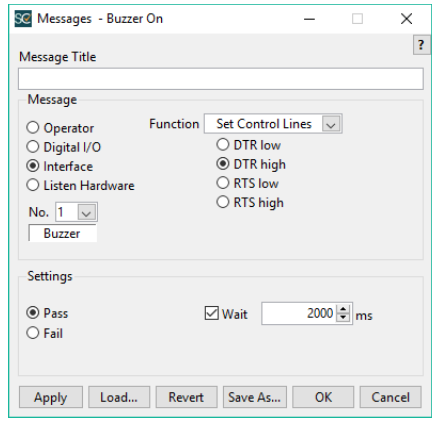

Figure 4-5: Message Step - Interface Settings

Insert the Buzzer Off message below (after) the Buzzer On Step in the **Sequences** section. Double click on it to edit the step. The message section needs to be set to match the settings in the Buzzer On step. **DTR low** should be selected to shut the buzzer off. Under **Setup**, **Pass** or **Fail** must be selected. Select **Wait** only if the *Buzzer Off* message is to be displayed for a selected amount of time.

- Select Interface
- Select Interface number (must be setup in Hardware Editor - External page)
- Select Pass/Fail - this is the state that the message will report when the step runs
- Select Wait time - Off messages do not necessarily need any Wait time
- Function - Select “Set Control Lines”
- Select DTR low

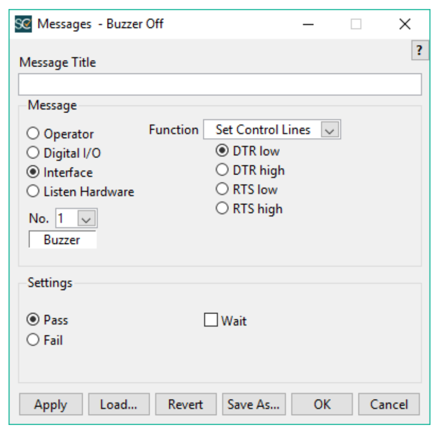

Figure 4-6: Message Step - Interface Settings

Important: ***Footswitches and a Buzzer can be connected to the same Serial Port connector but there is usually limited room for wiring in standard connector housings.***

Important: ***Two Buzzers can be connected to the same Serial Port at the same time with no footswitches. Only one Buzzer can be connected to a Serial Port when two Footswitches are in use on the same port.***

Remote Control Switch

The serial port connections can also be used as a Remote Control Switch. A foot switch, PLC, test box closure switch, or similar can be wired across pins 7 and 8 of the serial port to act as a Start and Continue switch. This replicates pressing the F2 key on the keyboard.

A second switch can be added across pins 6 and 7 for the Stop and REDO function (F4 key).

- These switch functions are pre-set in the SoundCheck system and cannot be changed
- These functions do not require the use of a Message Step in a sequence

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Important: ***When two switches are used on a single serial port, a buzzer cannot be used on the same serial port. Use a second serial port if needed.***

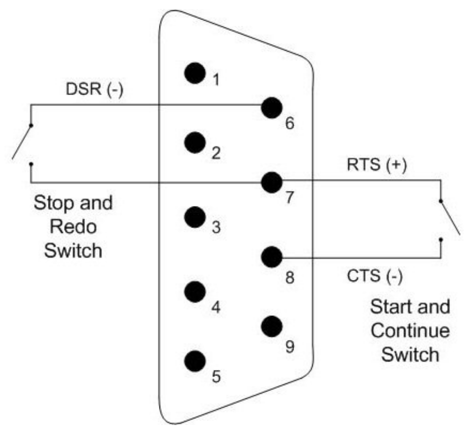

Figure 4-7: Serial Port Switch Wiring

1. To enable the Remote Control Switch function
   - Click **Setup** on the SoundCheck Main Screen
   - Open the Hardware Editor and select **External Hardware**

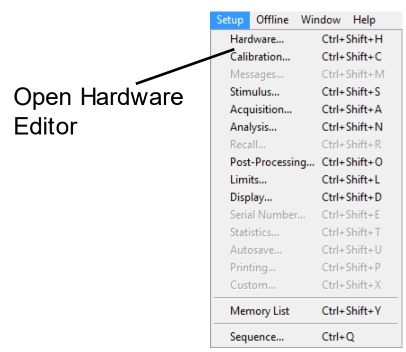

Figure 4-8: Open Hardware Configuration

2. Right-click an empty line in the Interface section and click **Add Interface**.
   - Under **Type** select **Footswitch**
   - Under **COM Port enter the COM Port number for the Serial Port** (Found in Windows Device Manager)
3. **Save** to close the *Hardware Configuration*

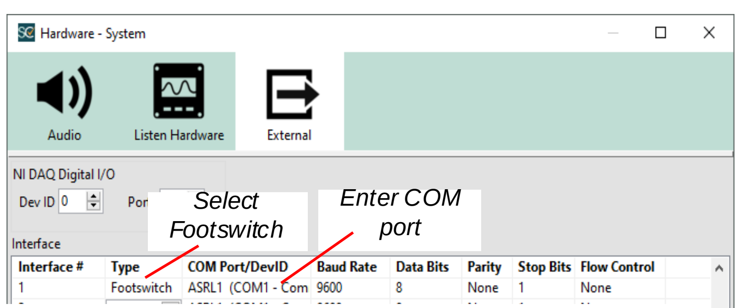

Figure 4-9: Hardware Setting For Footswitch

Appendix 5: System Verification Using SoundCheck

This section will enable users of SoundCheck to verify that their PC-based electroacoustic test system is working properly.

Open the Self Test sequence from the Calibration folder in SoundCheck. ***Figure 5-1***

This sequence requires that at least two channels of input and output are setup in your System Hardware configuration.

(See ***Hardware Configuration on page 586.***)

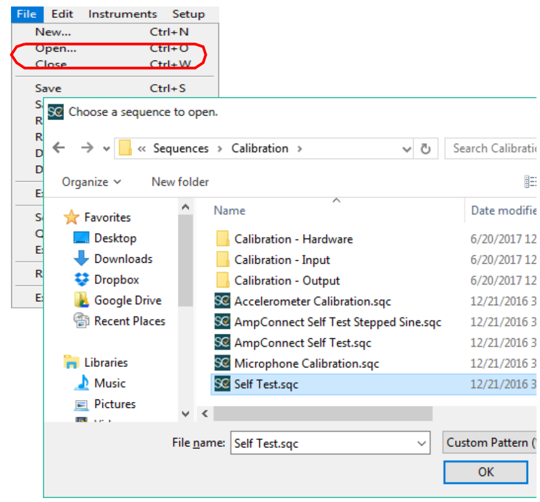

Figure: 5-1 Open Self Test

The Self Test sequence checks to see if your SoundCheck installation is working properly and verifies the performance of your audio interface. By looping back the audio interface output directly into the audio interface input, the audio interface's Frequency Response, Sensitivity, THD, Play/Record Delay, and Self Noise are measured.

The sequence is setup to test two channels of the selected audio interface as shown in ***Figure 5-2***.

Note that audio interfaces using WDM / WASAPI drivers will not have a consistent play/record delay, so the Delay results will show up as a failure. This is normal for these devices. Auto Delay must be used in Analysis Steps when using WDM / WASAPI.

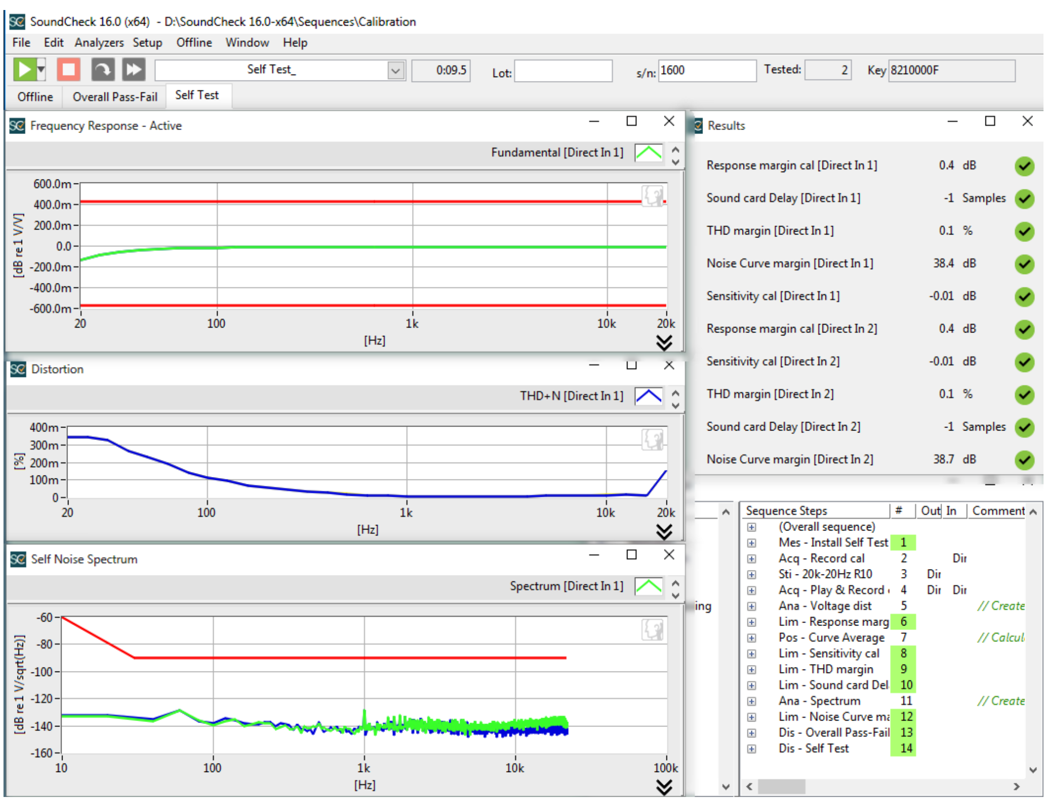

Figure: 5-2 Completed Sequence Showing Passing Results

Hardware Configuration

Make sure that the correct System Hardware settings are entered for your audio interface.

The default settings shown are for the AudioConnect.

To run Self Test on AudioConnect both input channels must be set to Line In by opening an AudioConnect Message Step or by changing the Input Selection in the Startup Configuration of AudioConnect in the Listen Hardware Tab.

See ***AudioConnect on page 72***.

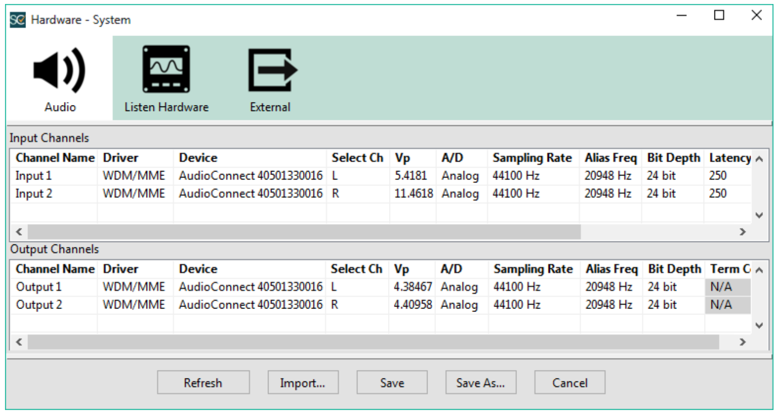

Figure: 5-3 Default Hardware Settings

Connect Output 1 to Input 1 and Output 2 to Input 2 of the audio interface. ***Figure 5-4***

WARNING! Do not connect the amplifier. The test level could damage the amplifier/loudspeaker if connected while running Self Test.

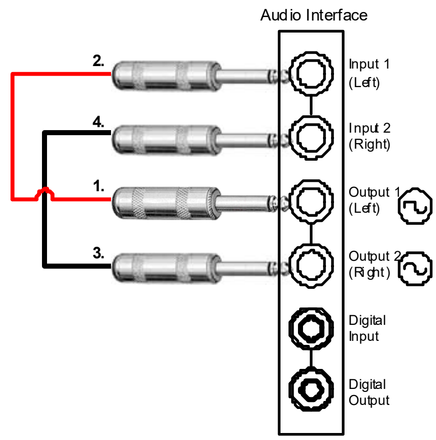

Figure: 5-4 Loop Back Wiring

Sequence Parameters

The sequence will test the audio interface using a Frequency Stepped Sweep with the following settings:

- Frequency Stepped Sweep
- Amplitude 1 V. (This level will provide a good signal-to-noise ratio for most audio interfaces. For the LynxTwo, try increasing the level to 5 V.)
- 20 kHz-20 Hz (To measure to higher frequencies, remember to increase the sampling rate for the audio interface in the Hardware Configuration)
- 1/3 Oct Steps - This gives you 31 frequency points
- 24 Cycles/Step (if you change the number of cycles, you should stay above an effective 15 cycles/step to insure proper THD+N measurement)

The results as shown in ***Figure 5-2*** cover the following parameters:

- Frequency Response: 20 to 20 kHz, +/- 0.5 dB
- Sensitivity @ 1 kHz: +/- 1 dB
- THD (%) & THD+N (%): 20 to 20 kHz, 0.03%
- Audio interface delay: +/- 0.05 ms
- Overall Noise Level Limits: [0 Hz, -70 dB], [30 Hz, -90 dB], [22 kHz, -90 dB]

Appendix 6: Verifying SoundConnect™ Performance

Checking the performance of the SoundConnect™ using the Amplifier THD+N sequence (Electronics folder). The *Generator and FFT* instruments will be used as well.

Connect the audio interface to SoundConnect™ as shown in ***Figure 6-1***.

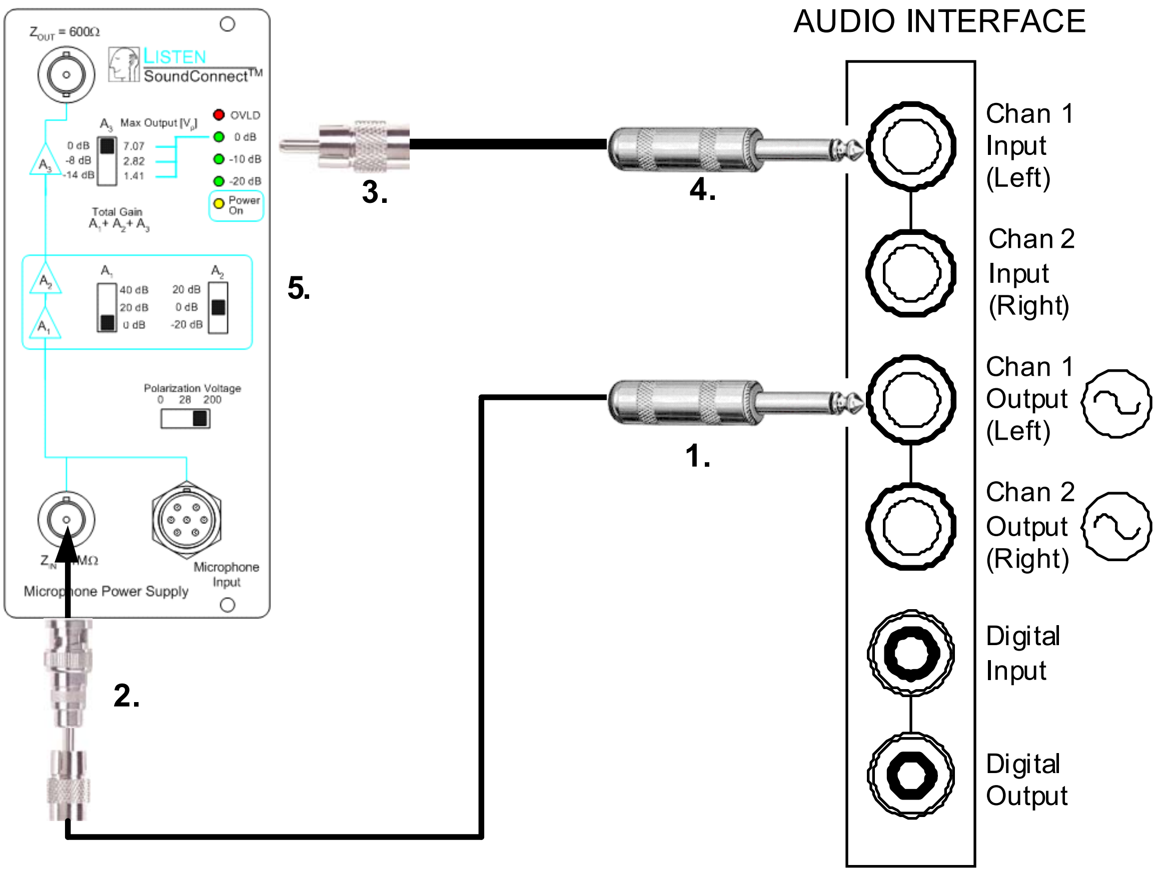

Figure 6-1: SoundConnect Performance Test Connections

- ***1 to 2*** Connect the Chan 1 Out of the audio interface to the BNC input on the SoundConnect front panel. In this example a 1/4 inch-to-RCA cable and a BNC male to RCA female adapter is used.
- ***3 to 4*** Connect the RCA output connector on the back of SoundConnect to the Chan 1 In of the audio interface. A 1/4 inch-to-RCA cable is used in the example. The BNC out on the front of SoundConnect can also be used. The two jacks are connected in parallel and should not be used simultaneously.
- ***5*** Set A1, A2, and A3 switches to 0 dB.

Run the Amplifier THD+N sequence to check the frequency response and sensitivity of the SoundConnect.

The SoundConnect™ frequency should be very flat (less than 1.0 dB variation).

Even though the audio interface frequency and phase response may be within spec, ground loops may occur. A more detailed analysis can be done using the built-in signal generator and FFT functions within SoundCheck®.

SoundConnect 2™ Connections

SoundConnect 2™ can be tested using the Amplifier THD+N sequence as noted on the previous page. The wiring requirements are different. Since the sequence is only setup to test a single channel, the test can be run on SoundConnect 2, channel 2 as well.

Connect the audio interface to SoundConnect 2™ as shown in ***Figure 6-2***.

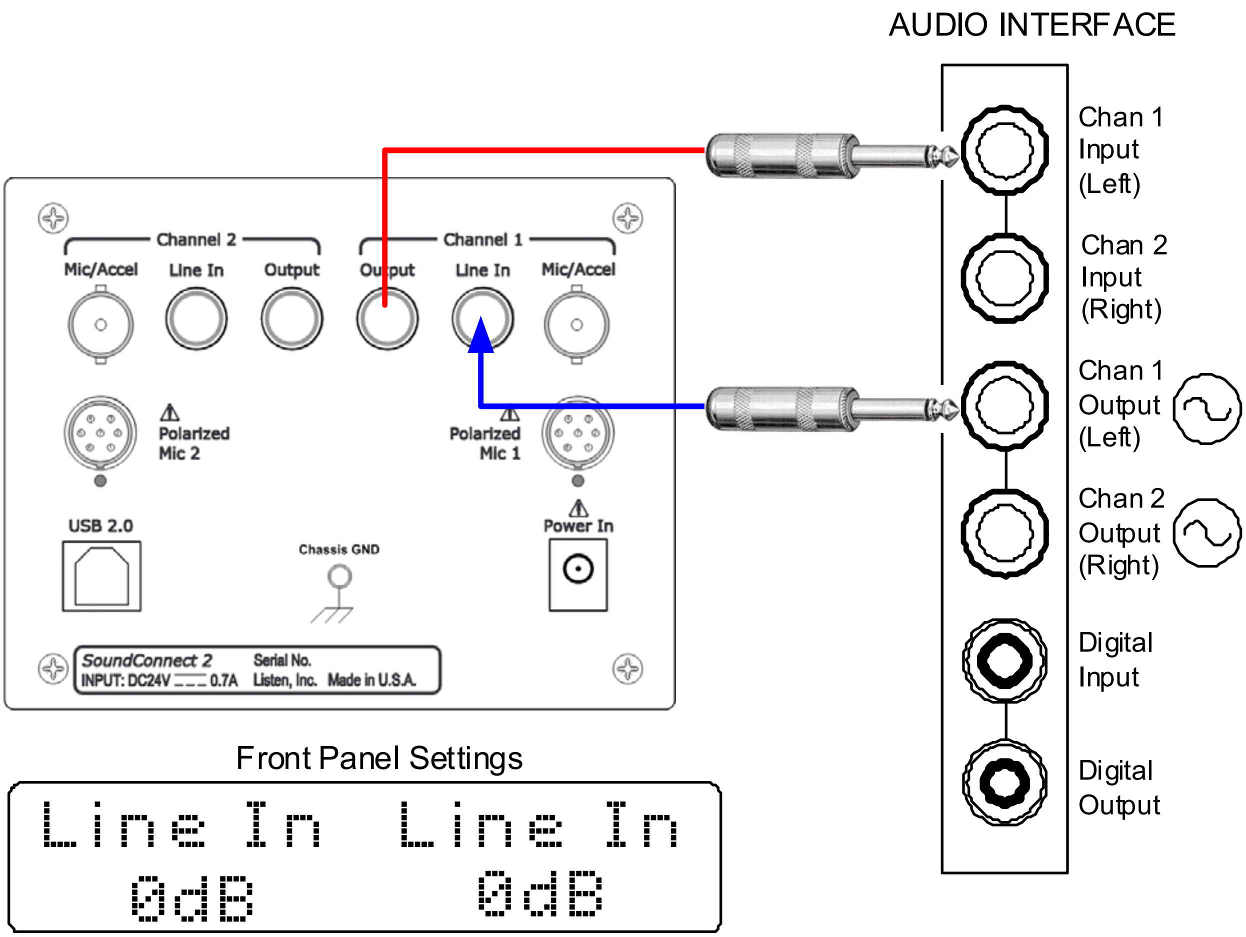

Figure 6-2: SoundConnect 2 Test Connections

- *Connect SoundConnect 2 Output 1 to Audio Interface Input 1*. We recommend using a balanced 1/4" TRS cable.
- Connect Output 1 of the audio interface to Input 1 of SoundConnect 2, again using a balance 1/4" TRS cable.
- The SoundConnect 2 Front Panel should be set to Line In and 0 dB of gain for both channels.

Run the Amplifier THD+N sequence to check the frequency response and sensitivity of the SoundConnect 2.

The SoundConnect 2™ frequency should be very flat (less than 1.0 dB variation).

Repeat the process to test Channel 2.

Even though the audio interface frequency and phase response may be within spec, ground loops may occur. A more detailed analysis can be done using the built-in signal generator and FFT functions within SoundCheck®.

Noise Floor and Ground Loop Detection

1. Open the *Signal Generator* and *FFT* instruments by clicking **Instruments** and click on **Signal Generator** and **Spectrum Analyzer**. You can also use the keyboard shortcuts (**Crtl** + **F4** and **Ctrl** + **F7**).

2. Set the Generator Output Level to 1.000 V, Frequency to 1000, and Channel to Direct Out 1.

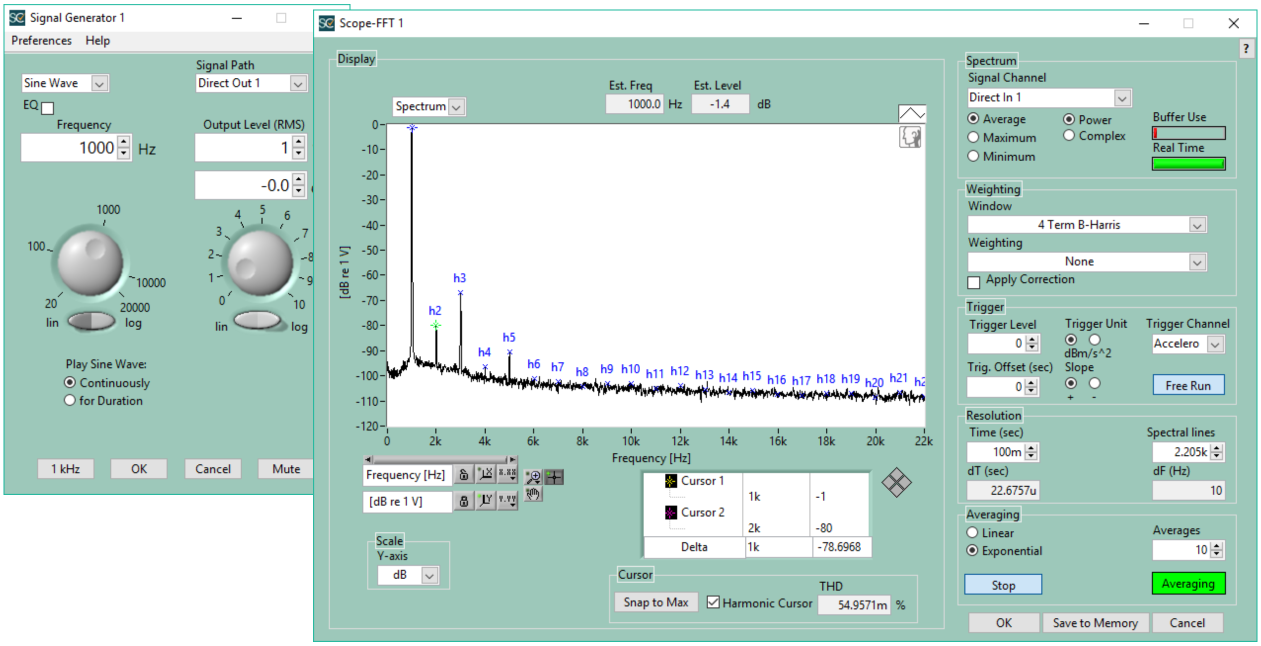

Figure 6-3: Noise Floor

3. Set FFT controls as shown in ***Figure 6-3***. The Input Signal Path should be set to Direct In 1.

4. **Mute** in **Signal Generator** so it no longer is blinking red. A 1.0 Volt, 1 kHz sine wave will be sent to the SoundConnect. Click **Start** on the *FFT* screen and the frequency spectrum will appear. If the SoundConnect and interconnecting cables are in good working condition, the FFT spectrum should look like the one above. The FFT spectrum should only have a 1 kHz signal. If any harmonics are present, they should be at least 60 dB below the 1 kHz level. Use the **Snap to Max** button to set Cursor 1 to the peak of the signal and then click on the **Harmonic Cursor** to show the THD.

To check for any ground loops or electrical line frequency interference (50 Hz or 60 Hz), reduce the frequency range of the FFT display from 20 kHz to 1 kHz. To do this, place the mouse cursor over the 20000 number and highlight it by double-clicking using the left mouse button. Enter 1k and the press then press the **Enter** key on the computer keyboard. The display will look like ***Figure 6-4***.

If there is any electrical interference, there will be significant spikes at 60 Hz (or 50 Hz) and associated harmon-ics.

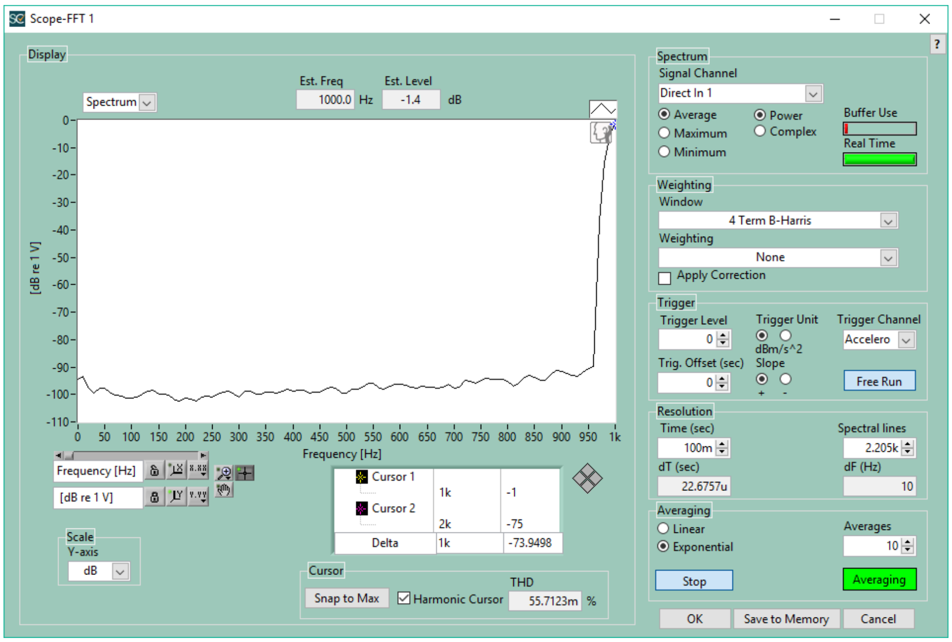

Figure 6-4: Check for Ground Loop

Windows Keyboard Shortcuts

Main Screen Buttons	Shortcut
Start Sequence	F2
Continuous	F3
Redo	F4
Cursor to Lot # Field	F7
Cursor to Serial # Field	F8
Continue	Enter
Stop	Esc

FILE (Alt+F)	Shortcut
New	Ctrl+N
Open	Ctrl+O
Close	Ctrl+W
Save	Ctrl+S
Save As	None
Revert	Ctrl+R
Rename	None
Delete	Ctrl+D
Export Seq	Ctrl+E
Setup Wizard	None
Exit	None

EDIT (Alt+E)	Shortcut
Login	None
Preferences	None

Instruments (Alt+A)	Shortcut
Signal Generator	Ctrl+F4
Multimeter	Ctrl+F5
Oscilloscope	Ctrl+F6
Spectrum Analyzer	Ctrl+F7
RTA	Ctrl+F8
Distortion Analyzer	Ctrl+F9
Frequency Counter	Ctrl+F10

Signal Generator (Ctrl+F4)

Mute	F5
Toggle between Frequency and Output Level	F8
Increase	Page Up
Decrease	Page Dn

SETUP (Alt+S)	Shortcut
Hardware	Ctrl+Shift+H
Calibration	Ctrl+Shift+C
Messages	Ctrl+Shift+M
Stimulus	Ctrl+Shift+S
Acquisition	Ctrl+Shift+A
Analysis	Ctrl+Shift+N
Recall	Ctrl+Shift+R
Post-Processing	Ctrl+Shift+O
Limits	Ctrl+Shift+L
Display	Ctrl+Shift+D
Serial Number	Ctrl+Shift+E
Statistics	Ctrl+Shift+T
Autosave	Ctrl+Shift+U
Printing	Ctrl+Shift+P
Custom	Ctrl+Shift+X
Memory List	Ctrl+Shift+Y
Sequence	Ctrl+Q

SEQUENCE	Shortcut
Sequence Editor	Ctrl+Q
New…	Ctrl+N
Save	Ctrl+S
Delete	Ctrl+D

OFFLINE (Alt+N)	Shortcut
None	None

Memory List (Ctrl+Shift+Y)

Memory List must be highlighted

FILE	Shortcut
Load Display	Ctrl+L
Search	Ctrl+F
Select All	Ctrl+A
Protect	Ctrl+M
Full Size	Ctrl+/
Print Current Display	Ctrl+P

HELP (Alt+H)	Shortcut
Show Context Help	Ctrl+H
User Manual	None
Additional Documentation	None
Quick Launch Menu…	Ctrl+G
Listen Inc Website	None
Request Support	None
Request a New Feature	None
Report a Bug	None
Optional Modules…	None
About SoundCheck…	None

Stweep™ Table - ISO Stepped-sine Frequencies

R10	R20	R40	R80	R10	R20	R40	R80	R10	R20	R40	R80	R10	R20	R40	R80
20	20	20	20	125	125	125	125	800	800	800	800	5000	5000	5000	5000
			20.6				128				825				5150
		21.2	21.2			132	132			850	850			5300	5300
			21.8				136				875				5450
	22.4	22.4	22.4		140	140	140		900	900	900		5600	5600	5600
			23				145				925				5800
		23.6	23.6			150	150			950	950			6000	6000
			24.3				155				975				6150
25	25	25	25	160	160	160	160	1000	1000	1000	1000	6300	6300	6300	6300
			25.8				165				1030				6500
		26.5	26.5			170	170			1060	1060			6700	6700
			27.2				175				1090				6900
	28	28	28		180	180	180		1120	1120	1120		7100	7100	7100
			29				185				1150				7300
		30	30			190	190			1180	1180			7500	7500
			30.7				195				1220				7750
31.5	31.5	31.5	31.5	200	200	200	200	1250	1250	1250	1250	8000	8000	8000	8000
			32.5				206				1280				8250
		33.5	33.5			212	212			1320	1320			8500	8500
			34.5				218				1360				8750
	35.5	35.5	35.5		224	224	224		1400	1400	1400		9000	9000	9000
			36.5				230				1450				9250
		37.5	37.5			236	236			1500	1500			9500	9500
			38.7				243				1550				9750
40	40	40	40	250	250	250	250	1600	1600	1600	1600	10000	10000	10000	10000
			41.2				258				1650				10300
		42.5	42.5			265	265			1700	1700			10600	10600
			43.7				272				1750				10900
	4.5	45	45		280	280	280		1800	1800	1800		11200	11200	11200
			46.2				290				1850				11500
		47.5	47.5			300	300			1900	1900			11800	11800
			48.7				307				1950				12200
50	50	50	50	315	315	315	315	2000	2000	2000	2000	12500	12500	12500	12500
			51.5				325				2060				12800
		53	53			335	335			2120	2120			13200	13200
			54.5				345				2180				13600
	56	56	56		355	355	355		2240	2240	2240		14000	14000	14000
			58				365				2300				14500
		60	60			375	375			2360	2360			15000	15000
			61.5				387				2430				15500
63	63	63	63	400	400	400	400	2500	2500	2500	2500	16000	16000	16000	16000
			65				412				2580				16500
		67	67			425	425			2650	2650			17000	17000
			69				437				2720				17500
	71	71	71		45	450	450		2800	2800	2800		18000	18000	18000
			73				462				2900				18500
		75	75			475	475			3000	3000			19000	19000
			77.5				487				3070				19500
80	80	80	80	500	500	500	500	3150	3150	3150	3150	20000	20000	20000	20000
			82.5				515				3250				
		85	85			530	530			3350	3350				
			87.5				545				3450				
	90	90	90		560	560	560		3550	3550	3550				
			92.5				580				3650				
		95	95			600	600			3750	3750				
			97.5				615				3870				
100	100	100	100	630	630	630	630	4000	4000	4000	4000				
			103				650				4120				
		106	106			670	670			4250	4250				
			109				690				4370				
	112	112	112		710	710	710		450	4500	4500				
			115				730				4620				
		118	118			750	750			4750	4750				
			122				775				4870				

Appendix 8: Keyboard Shortcuts Mac

Mac Keyboard Shortcuts

Main Screen Buttons	Shortcut
Start Sequence	F2
Continuous	F3
Redo	F4
Cursor to Lot # Field	F7
Cursor to Serial # Field	F8
Continue	Enter
Stop	Dbl Click Esc

FILE	Shortcut
New	CMD+N
Open	CMD+O
Close	CMD+W
Save	CMD+S
Save As	None
Revert	CMD+R
Rename	None
Delete	CMD+D
Export Seq	None
Setup Wizard	None
Exit	None

EDIT	Shortcut
Login	None
Preferences	None

OPERATE	Shortcut
Signal Generator	CMD+4
Multimeter	CMD+5
Oscilloscope	CMD+6
Spectrum Analyzer	CMD+7
RTA	CMD+8

Signal Generator

Mute	Dbl Click F5
Toggle between Frequency and Output Level	None
Increase	Page Up
Decrease	Page Dn

SETUP	Shortcut
Hardware	CMD+Shift+H
Calibration	CMD+Shift+C
Messages	CMD+Shift+M
Stimulus	CMD+Shift+S
Acquisition	CMD+Shift+A
Analysis	CMD+Shift+N
Recall	CMD+Shift+R
Post-Processing	CMD+Shift+O
Limits	CMD+Shift+L
Display	CMD+Shift+D
Serial Number	CMD+Shift+E
Statistics	CMD+Shift+T
Autosave	CMD+Shift+U
Printing	CMD+Shift+P
Custom	None
Memory List	CMD+Shift+Y
Sequence	None

SEQUENCE	Shortcut
Sequence Editor	CMD+E
New...	CMD+N
Save	CMD+S
Delete	CMD+D

OFFLINE	Shortcut
None	None

Memory List

Memory List must be highlighted

FILE	Shortcut
Load Display	CMD+L
Search	CMD+F
Select All	CMD+A
Protect	CMD+M
Full Size	CMD+/ (toggles)

HELP	Shortcut
Show Context Help	None
User Manual	None
Additional Documentation	None
Quick Launch Menu...	CMD+G
Listen Inc Website	None
Request Support	None
Request a New Feature	None
Report a Bug	None
Optional Modules...	None
About SoundCheck...	None

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Appendix 9: Equation Editor Functions

+ and -	addition and subtraction
* and /	multiplication and division
^	Exponentiation

User Equation Syntax

Function	Corresponding LabVIEW Function	Description
abs(x)	Absolute Value	Returns the absolute value of x.
acos(x)	Inverse Cosine	Computes the inverse cosine of x in radians.
acosh(x)	Inverse Hyperbolic Cosine	Computes the inverse hyperbolic cosine of x.
asin(x)	Inverse Sine	Computes the inverse sine of x in radians.
asinh(x)	Inverse Hyperbolic Sine	Computes the inverse hyperbolic sine of x.
atan(x)	Inverse Tangent	Computes the inverse tangent of x in radians.
atanh(x)	Inverse Hyperbolic Tangent	Computes the inverse hyperbolic tangent of x.
ci(x)	Cosine Integral	Computes the cosine integral of x where x is any real number.
cos(x)	Cosine	Computes the cosine of x, where x is in radians.
cosh(x)	Hyperbolic Cosine	Computes the hyperbolic cosine of x.
cot(x)	Cotangent	Computes the cotangent of x (1/tan(x)), where x is in radians.
csc(x)	Cosecant	Computes the cosecant of x (1/sin(x)), where x is in radians.
exp(x)	Exponential	Computes the value of e raised to the x power.
expm1(x)	Exponential (Arg) - 1	Computes one less than the value of e raised to the x power ((e^x) - 1).
floor(x)	Round To -Infinity	Truncates x to the next lower integer (largest integer £x).

gamma(x)	Gamma Function	G (n + 1) = n! for all natural numbers n.
int(x)	Round To Nearest	Rounds x to the nearest integer.
intrz(x)	-	Rounds x to the nearest integer between x and zero.
ln(x)	Natural Logarithm	Computes the natural logarithm of x (to the base of e).
lnp1(x)	Natural Logarithm (Arg +1)	Computes the natural logarithm of (x + 1).
log(x)	Logarithm Base 10	Computes the logarithm of x (to the base of 10).
log2(x)	Logarithm Base 2	Computes the logarithm of x (to the base of 2).
pi(x)	Represents the value = 3.14159...	pi(x) = x * ppi(1) = ppi(2.4) = 2.4 * p
rand( )	Random Number (0 - 1)	Produces a floating point number between 0 and 1 exclusively.
sec(x)	Secant	Computes the secant of x, where x is in radians (1/cos(x)).
si(x)	Sine Integral	Computes the sine integral of x where x is any real number.
sign(x)	Sign	Returns 1 if x is greater than 0, returns 0 if x is equal to 0, and returns -1 if x is less than 0.
sin(x)	Sine	Computes the sine of x, where x is in radians.
sinc(x)	Sinc	Computes the sine of x divided by x (sin(x)/x), where x is in radians.
sinh(x)	Hyperbolic Sine	Computes the hyperbolic sine of x.
spike(x)	Spike Function	spike(x) returns: 1 if 0£ x £10 for any other value of x.
sqrt(x)	Square Root	Computes the square root of x.
square(x)	Square Function	square (x) returns: 1 if 2n£ x£ (2n + 1)0 if 2n + 1£ x £(2n + 2)where x is any real number and n is any integer.
step(x)	Step Function	step(x) returns: 0 if x < 01 if any other condition obtains.
tan(x)	Tangent	Computes the tangent of x, where x is in radians.
tanh(x)	Hyperbolic Tangent	Computes the hyperbolic tangent of x.

Appendix 10: Weighting and Window Types

TSR Window Type

As of SoundCheck 8, the TSR window waveform is output in the Memory List and can be displayed on top of the impulse response or the deconvolved response to check the time alignment. The Start and Stop times of the window are set in the Analysis Step. See *__Time Selective Response (TSR) on page 174__* for more information. The window is used to select the portion of the Deconvolved Response that contains the fundamental impulse response (linear impulse response). This impulse response contains the frequency response of the device under test.

The following examples use the same impulse response as a means of comparison.

None

Also referred to as Rectangular. No windowing is applied to the measurement. This works well with transients that are shorter in length than the measurement time. Due to the flat characteristic in the time domain, all parts of the signal are equally weighted.

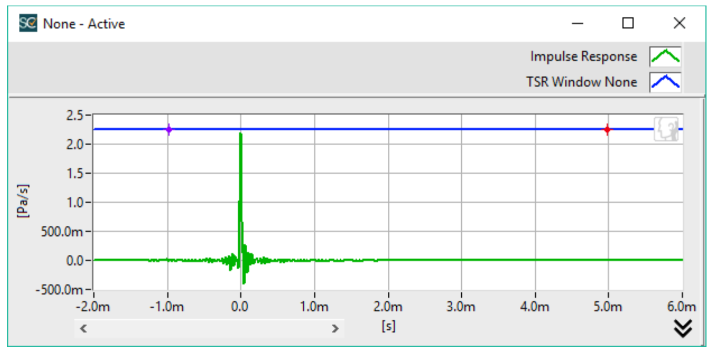


Figure 10-1: None

Cosine Tapered

The Cosine Taper window used by the Time Selective Response algorithm, has a 10% taper at each end. The Fundamental Impulse Response must be inside these tapers. In the example, the taper of the window disregards the first and last 1 mSec of the impulse. In versions of SoundCheck prior to version 8, this was the fixed window type for TSR.

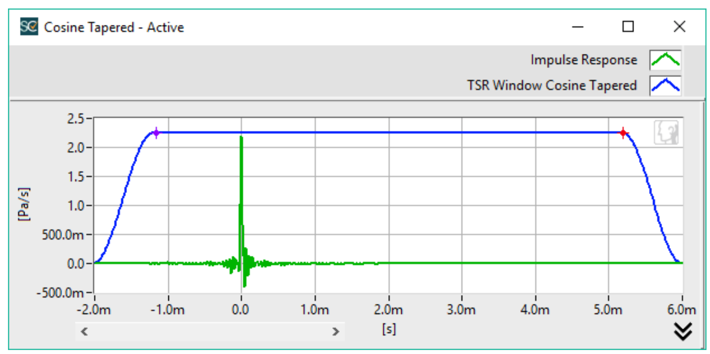


Figure 10-2: Cosine Tapered

Half-Cosine Tapered

This window has only a trailing Cosine taper. It is well suited for impulses which have no content before zero time. The taper removes any slight discontinuity at the very end.

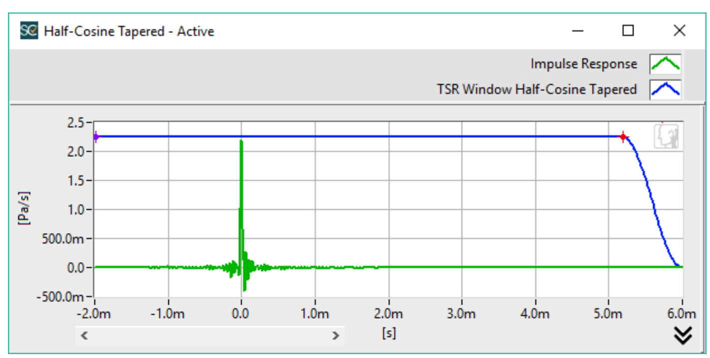


Figure 10-3: Half-Cosine Tapered

Exponential

This window is useful for impulses with a long decay rate which exceeds the duration of the time window. The exponential curve applies an additional damping that forces a smoothing of the impulse in the time window. The start value of the exponential weighting is 100% and the final value is 10%."

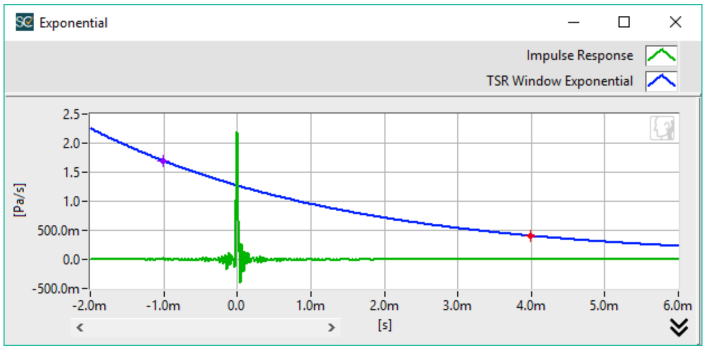


Figure 10-4: Exponential

Adrienne

This is an optimized window for time selective response measurements. The leading and trailing tapers are both half Blackman-Harris functions. For optimal results the beginning of the flat portion should be 0.2 mSec before the impulse.

This window is specified in the British Standard, "Road traffic noise reducing devices - test method for determining the acoustic performance - part 5: intrinsic characteristics - in situ values of sound reflection and airborne sound insulation" BS CEN/TS 1793-5:2003.

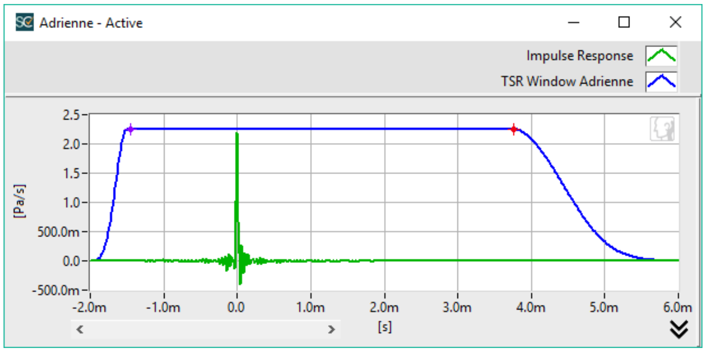


Figure 10-5: Adrienne

Half-Hanning

This applies a Half-cosine taper to the end of the window. It is well suited for long impulses that exceed the duration of the time window.

The damping applied is stronger than what is found in the Exponential window, yielding smoother results in the frequency domain.

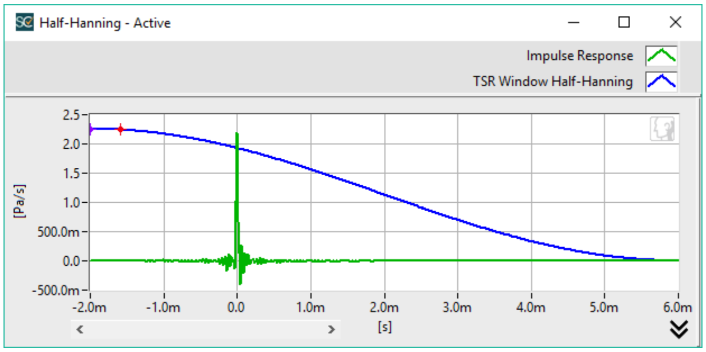


Figure 10-6: Half-Hanning

Half-BH4

This is a Half Blackman-Harris function (4 terms). The application is the same as for exponential and Half-Hanning. It has the strongest damping and therefore yields the greatest smoothing in the frequency domain.

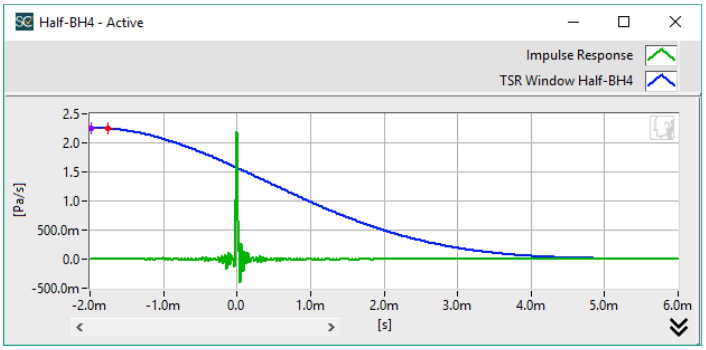


Figure 10-7: Half Blackman-Harris 4 Term

Appendix 11: Time Selective Measurements With Log Sweep

A time selective measurement of a frequency response is (directly or indirectly) based on a measurement of the impulse response, where a well-defined time window is applied to the impulse response. The frequency response is simply the Fourier transform of the impulse response. Time selective measurements are often used in electroacoustics to make simulated free-field measurements of transducers. This is to isolate the directly transmitted, free field sound from reflections due to the surroundings. By using a time window applied to the impulse response, it is possible to obtain results similar to those obtained in a non-echoic environment.

The accuracy of such measurements depends on several factors. Some basic properties apply generally to all implementations, while some others relate to the actual algorithm and specific implementation.

The most basic requirement for simulated free-field measurements is that the reflections must not arrive so early that they overlap the impulse response of the direct sound. If the reflections arrive so early that they overlap the direct sound, the time window will cut away some of the impulse response of the direct sound, and the measurement will loose accuracy. This applies to all time selective methods.

If the system is perfectly linear then the impulse response can be obtained by a direct measurement, simply by applying a short pulse and recording the resulting "impulse response". In practice, however, such measurements will usually suffer from either a poor S/N ratio due to the low energy in the short pulse or suffer from overloading if the pulse is increased to improve the S/N ratio. The crest factor of the excitation is simply too high for practical use.

Various types of noise signals have a much lower crest factor and combined with cross-spectrum or cross-correlation analysis, the response of the linear behavior can be measured. Most systems, however, are not perfectly linear. The non-linearity not only sets limits for which test signals can be used, but also introduces the need for characterizing this non-linearity, e.g., measuring the distortion.

The Time Delay Spectrometry (TDS) introduced by Heyser uses a linear swept sine for time-selective measurements. Like most other swept sine algorithms, it is based on the assumption that the response at a certain point in time represents the response to just one particular frequency. That is approximately correct if the sweep rate is low enough. A delay in the system under test, however, will result in a frequency shift (proportional to the delay) of the measured response. If the signal follows different paths with different delays (e.g., direct and reflected sound from a loudspeaker), the signal measured will contain slightly different frequencies. Tracking the response with a narrow bandpass filter therefore makes it possible to isolate one path (e.g., just the direct sound from a loudspeaker) from the others. It is proven that such a tracking bandpass filter is simply equivalent to applying a time window to the impulse response. In addition, a tracking bandpass filter can also be used to track the harmonics of the swept signal, thereby measure harmonic distortion as well (and still in a time selective way). However, the sweep rate must be limited in order to measure the fundamental correctly, measuring harmonics at low frequencies put further constraints on the maximum sweep rate.

The Time Selective Response (TSR) introduced by Brüel & Kjær also uses a linear swept sine, but removes the limitation on the sweep rate (by mathematical refinement of the algorithm) so the fundamental response is measured correctly even with very fast sweeps. For measuring distortion at low frequencies, the same constraints still applies to the maximum sweep rate for TSR, as for TDS.

Both TDS and TSR, with the tracking bandpass filter approach, are linked to using a linear sweep. The linear sweep, however, is not very ideal, if the measurements shall cover a broad frequency range:

- Often the S/N ratio at low frequencies is critical, but the linear sweep has relatively little energy at low frequencies: Half of the time (and thereby half of the energy) is used in the highest octave, only one fourth of the time (and of the energy) is used in the second highest octave, etc. In order to achieve a sufficient S/N ratio at low frequencies a very slow sweep has to be used, wasting time (and energy) at high frequencies.
- The linear sweep also becomes very slow, if the sweep rate has to be kept very low in order to measure distortion at low frequencies - eventually even slower than required for a sufficient S/N ratio.

The log TSR implemented in SoundCheck® uses a logarithmically swept sine for fast time selective measurements of both the fundamental response and of distortion. The logarithmically swept sine is much more suitable for electroacoustic measurements:

- The logarithmic sweep uses the same time (and energy) for every octave, which is much more suitable achieving a good S/N ration for all frequencies in typical electroacoustic measurements.
- The logarithmic sweep also provides a sweep rate, which is low at low frequencies but increases with the frequency. That makes it possible to measure distortion also at low frequencies without making the whole sweep very slow. As a logarithmically swept sine is used the "tracking bandpass filter" method is not applicable for the analysis. Instead, Cross-Correlation analysis is used. By doing the Cross-Correlation of the response signal with a special energy weighted version of the excitation signal, the impulse response is found directly, and from that the frequency response is easily calculated as well.

It is thereby possible to obtain a unique combination of features:

- Time selectivity.
- The very suitable energy distribution of the logarithmic sweep (increased energy at low frequencies).
- The capability of measuring distortion due to the sine based nature of the signal.
- An effective way measure distortion even at low frequencies due to the sweep rate of the logarithmic sweep being low only at low frequencies just where it is needed.

***Important:* *The time windows used by the Time Selective Response algorithm are defined in* *Weighting and Window Types on page 597*.**

Please refer to the following AES paper for more information on this measurement technique:

“***Simulated Free Field Measurements”***, found on the Listen website.

Appendix 12: Excel Template Tutorial

Excel templates are a great way to keep your data organized when you are running the same sequence repeatedly and want to save several pieces of data each time. They allow you to customize the layout of data and even create additional worksheets with graphs, statistics, or summaries.

Note: **Excel** must be installed on the computer. Cloud based installations are not recognized by SoundCheck.

Step 1 – Write Sequence

- Before you can begin developing a template your sequence should be able to collect all the data that should be saved to Excel
- SoundCheck relies on the data names that are visible in the memory list to create the worksheets in Excel
- When saving to an Excel file, a new worksheet is created for each data item
- The names for all Curves, Values and Results should be finalized before they are saved to Excel
- If the data names are changed after you've created the Excel template, the Excel template will have to be re-created

Step 2 – Create Autosave Step(s)

- Insert an **Autosave Step** in the sequence, and configure it to **Save Data** to Excel.

Important: ***If you are saving Data and Results, you will need at least 2 Autosave Steps set to Append.***

In this case, the steps are set to Append Data so that all the information is added to one Excel file.

- Choose settings for all of the parameters, but select 'None' for the Header
- If saving curves you may wish to only select the **'Y' axis**. This will result in the **X axis** being printed once at the beginning of the Excel file, and only the **Y axis** data will append after that.

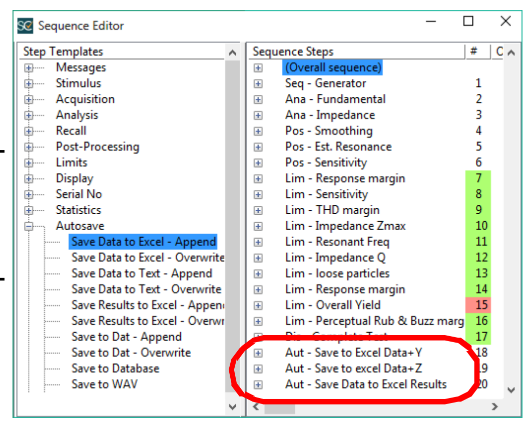


Figure 12-1: Autosave Steps

Depending on the sequence, there may be multiple **Autosave Steps** appending to the same file. The benefit of this is that you can save different data types with unique formats but keep the data confined to a single file. **Data** and **Results** are autosaved via different steps. ***Figure 12-1*** shows 3 separate Autosave Steps in the Sequence Editor.

The Autosave Step in ***Figure 12-2*** is saving the bulk of the data to Excel.

- The Y axis is always saved on the first save to Excel

In this example, only the Y axis is saved since the Z axis values are not needed.

Other settings used for all three Autosave Steps:

Autosave Folder Path: C:\SoundCheck\Data (or other folder on your local drive)

Format: Excel

Header: None

Layout: Rows

Notation: Floating point, 2 Decimal places

Test Information: Serial No.

Filename: Append and Automatic

Filename Template: <Seq> (Sequence Name)

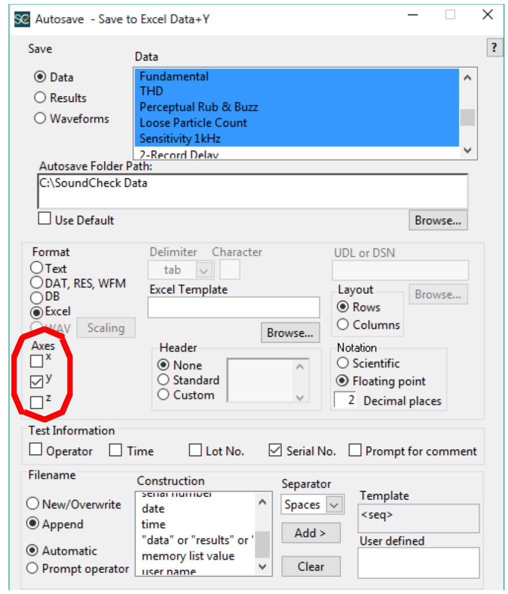


Figure 12-2: Save Y Axis

The step in ***Figure 12-3*** is nearly identical to the previous step except for the axis choice. Only the Z axis is saved since the data is phase information.

These two steps are set to append to the same file. This is done by selecting Excel as the data format, and choosing the same data folder and filename settings. In this particular example the first Autosave Step will open Excel, save the data, and then close Excel. The second step will repeat this procedure, appending to the file that was just created.

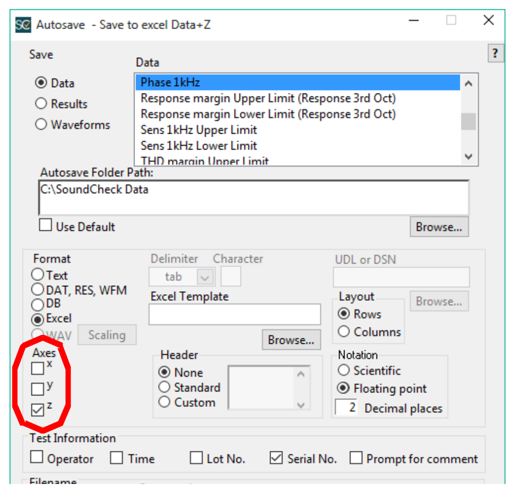


Figure 12-3: Save Z Axis

The third Autosave Step in *Figure 12-4* is set to save the Results of the selected limit steps. Otherwise the settings are identical to the first two steps, and the data will again be saved to the single Excel file.

Note: ***Saving Data and Results requires two separate Autosave steps.***

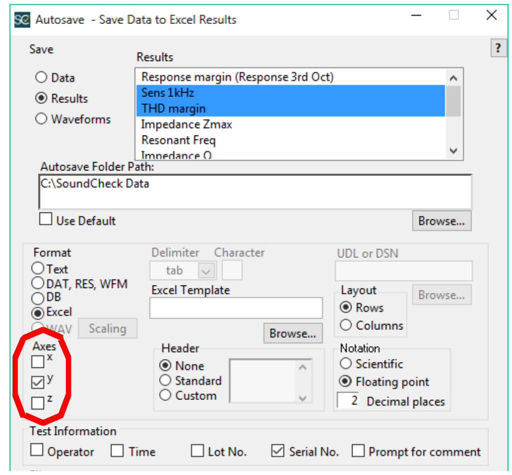


Figure 12-4: Save Results

Step 3 – Create Initial Excel File

Now that the autosave step(s) have been created, it is time to run the sequence so that an **Excel file** is created. This initial file is used to create the template.

Using an Excel file with real data saves a lot of time in Step 4 since this initial file has the correct structure for data including worksheet names and data headers.

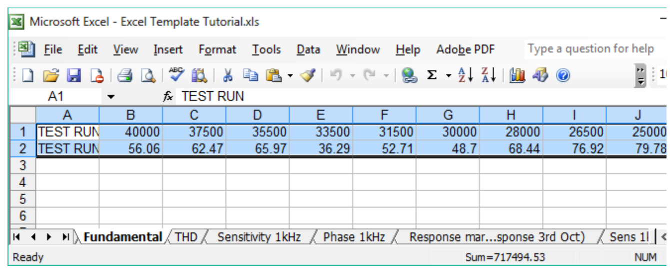


Figure 12-5: Initial Excel File Example

Figure 12-5 shows the “Fundamental Tab“ of the initial saved file from the example sequence.

Step 4 – Create Template

Open the **Initial Excel File** and save it as a new file, adding the word **“Template”** to the end of name. This identifies it for use in the next step and to prevent it from accidentally being overwritten. **For example: Sequence Name Template.xls**

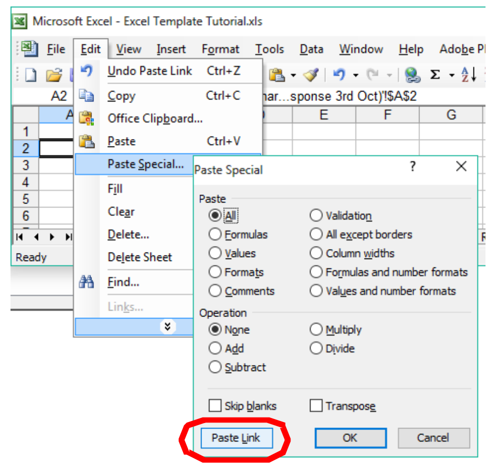
- This must be an .XLS or .XLSX file, not an .XLT Excel template. XLSM files are supported as of SC 14.01.

Create New Summary Worksheet

Excel allows you to link data from several worksheets to cells on a Master Page. This allows you to condense and format data to suit your needs.

In the example template file, a '**Summary**' worksheet has been added. It has been moved so it is the first worksheet in the Excel file. All the required data from each worksheet can be presented on a single page instead of having to scan through each of the tabs. This is especially useful for individual product reports or creating an overall pass/fail report for a production run.

Summary for Production Run

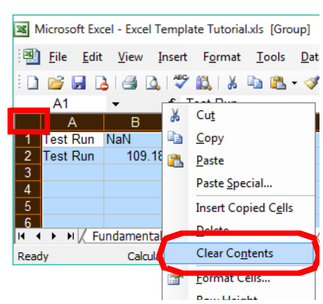


- Copy the entire Column of data from a worksheet for each item you want on the Summary Page. This ensures that the data from **all sequence runs** will be shown on the Summary Page.
- Go to the Summary Page and click on the Column Letter where you would like that data to appear
- Select Edit > Paste Special > Paste Link
- Do this for each Data Column that should appear on the Summary
- See example in ***Figure 12-8***

Summary for Single DUT

- Copy individual cells of data from a worksheet for each item needed and paste in the desired cell on the Summary Page using "Edit > Paste Special > Paste Link" as noted above
- You can also use the graph function in Excel, e.g.: Response or Distortion graphs

Prepare Template for New Data



- Go through each **Tab** or Worksheet of the Excel Template that generated by SoundCheck. Do not include the Summary Page. Perform the following operation
  - Select all the <u>Initial Data Cells</u> by clicking on box in the upper left corner of the first data worksheet
  - Select multiple worksheet by holding the **Shift Key** and clicking on all of the Worksheets along the bottom.

    DO NOT INCLUDE THE SUMMARY PAGE!
  - **Right-click** the selected region(s) and select **Clear Contents**. (Do not use the Delete key. It is not the same function.) You can also go to the Excel Edit menu, select **Clear** and then select **Contents**.
  - This Clears the data from the cells but retains the <u>formating and data pointers required by Excel</u>

- *Figure 12-6* shows the all the worksheets are present in the template but the data lines are empty. This must be done for every worksheet in the template.

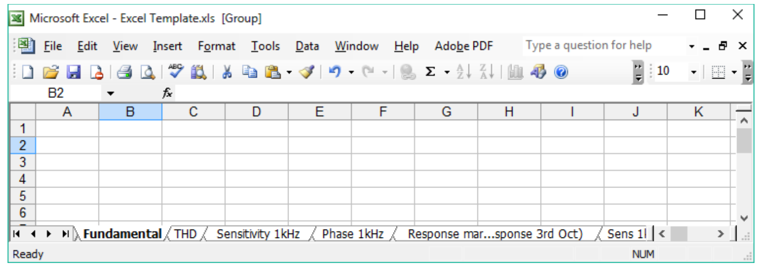


Figure 12-6: Clear Contents to Create Template

Step 5 – Use the Template in the Sequence

In SoundCheck open the Sequence Editor and set each of the Autosave Steps to use the new **Excel Template**. When the sequence is run, SoundCheck will enter the data into this template but, <u>save it as a new Excel file</u> according to your Filename settings. If the Autosave Step is set to Append, SoundCheck will continue appending to an existing file.

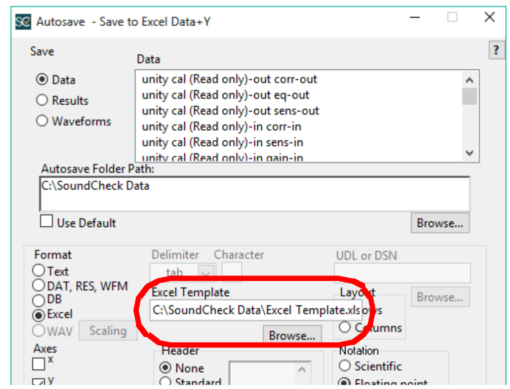


Figure 12-7 shows the new Excel file after saving date on four units in a production run.

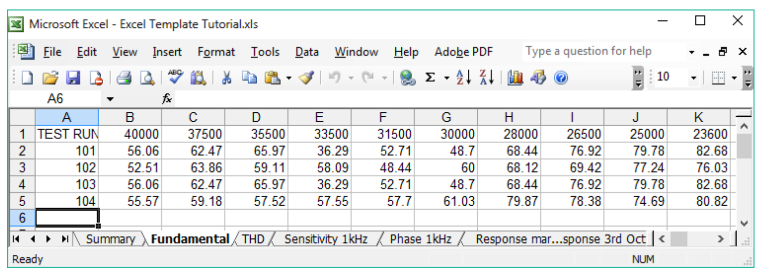


Figure 12-7: New Production Run Excel File

Figure 12-8 shows the Summary Tab which links to data from the other worksheets.

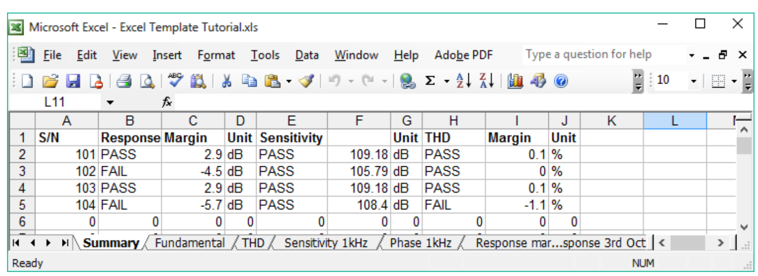


Figure 12-8: Summary Tab

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Appendix 13: Barcode Reader Integration

At its most basic functionality, a barcode reader can be used to input characters scanned from a barcode directly into any available field in SoundCheck. This is useful for quickly entering long and/or complex serial numbers or lot numbers in SoundCheck.

It is also possible to scan a barcode that performs an action in SoundCheck.

- Scan a barcode that starts a SoundCheck sequence
- Scan a barcode that opens the Signal Generator

This is possible because a barcode reader converts data read from the barcode into ASCII data that is sent to your PC. Essentially, it's emulating keyboard activity. This is great for production line systems where a keyboard is not available to the operator and you want to simplify and minimize operator contact with the PC.

We can take advantage of this by creating barcodes which represent the keyboard shortcuts available in SoundCheck. Referencing the first example, starting a sequence through the keyboard is done by pressing **F2** on the keyboard. Therefore, by generating a barcode which contains the same ASCII data as **F2**, we are able to mimic that function key being pressed by scanning that barcode. See ***Keyboard Shortcuts on page 591***.

Programming

Most barcode readers need to be programmed to be able to read a barcode and emulate a key stroke. This is done by scanning a series of barcodes in the bar code reader user manual. Sometimes an extended manual or programming guide for the reader needs to be downloaded from the manufacturer's website.

Important! The barcode reader must be able to "Emulate Keyboard Functions" so that SoundCheck receives keystrokes when a barcode is scanned.

Before you buy a barcode reader for SoundCheck control, contact the technical support department of the bar code reader manufacturer and ask if it supports:

- Keyboard Emulation, Keyboard Wedge or Does it act as an HID (Human Interface Device)
- Code 39 Full ASCII and Code 39 Extended
- Function Key Mapping (See below)
- Does the reader come pre-programmed to read Function Key barcodes from their programming guide (cut and paste F key barcodes from their PDF file)

Symbology and Function Key Mapping

The first thing that needs to be activated is the symbology called "**Code 39 Full ASCII**" and Code 39 Extended. Either of these may be required.

Bar Code Rules

Some bar code readers allow you to create special rules allowing you to emulate key strokes. The process usually involves scanning a list of bar codes in the bar code reader instruction manual. Instructions will vary depending on the make and model of bar code reader. This is only a general example.

Example

Scan the bar codes in the order listed.

1. Begin New Rule
2. Specific String at Start
3. Capital letter F code
4. Number 2 code
5. End of Message
6. Send F2 Key
7. Save Rule

With barcode creation software, make a barcode for “F2” (the letter F and the number 2 in one bar code), label it “Start Sequence” and print it out. Scanning this bar code will emulate hitting the F2 key on the keyboard.

Repeat the process for other function keys. See ***Keyboard Shortcuts on page 591***.

Check with the manufacturer regarding utilities that will assist with Function Key Mapping or programming the reader.

Advanced

Function Key Mapping creates a “preset” in the barcode reader for selected keys on the keyboard, such as the function keys. The mapping uses ASCII values that are rarely used. This is what allows a barcode reader to send key strokes to the computer, such as **F2**, which is used to start a sequence is SoundCheck.

Some barcode readers also have “Advanced Data Editing” functionality. This allows you to program the barcode reader to manipulate the data read by the barcode reader. You can program prefix and suffix commands, so that when a bar code is scanned, other commands can be “chained“ together to control SoundCheck. For example:

- The barcode reader is programmed with the prefix **F8** (which puts the cursor in the serial number field) and the suffix **F2** (which starts a SoundCheck sequence)
- Scanning a serial number barcode on your DUT automatically adds the number to the serial number field and then starts the currently open sequence

Barcode Software

You will also need a software application to create barcodes.

- The software must be able to create Code 39 barcodes

  The example in ***Figure 13-1*** shows a Code39 barcode for “capital F + 2”. The barcode reader is programmed to read this code and send the “F2 key” to the computer.

Code39

F2

Figure 13-1: Code39 of Capital F + 2

- You will need to create a separate barcode for each keyboard function required (F2, F3, F4, etc.) See ***Keyboard Shortcuts on page 591***.
- These barcodes and instructions for use can be used to make to a ”Barcode Command Sheet” that is printed out and posted with each SoundCheck system

There are a number of Barcode Software applications available that provide the basic functions required, such as Code 39. There are also free barcode applications available.

Appendix 14: Running Sequences from a Network Drive

As of SoundCheck 12, the sequence file (.SQC) contains all sequence parameters and steps. Individual Step files are no longer required. See ***Single-file Sequence Format on page 436***.

- Sequences can be shared with workstations on the network
- All systems must use the same version of SoundCheck as sequences are not backward compatible
- Each workstation will use its own System Hardware and System Calibration Configurations, which will have unique values
- Once edited, sequences can be marked as "**Read-Only**" to prevent unwanted changes on the workstations (Right-click the sequence or folder of sequences, select Properties and check Read-Only)

Important! The only downside to this practice is that the Master and Workstation PCs can only open sequences if the network is operational. The reliability of the network should be considered before implementing this system.

- Server - A sub-server used solely for SoundCheck (preferred) or the main network server
- Master PC - Used to create and edit SoundCheck sequences, storing them on the network
- Workstation PC - A SoundCheck system that opens sequences stored on the server

Master PC Configuration

The **Master PC** is the computer that sequences are developed and/or edited on. This requires the *Sequence Editor* (optional module 2002).

1. Any exported sequences from previous versions should be saved with SoundCheck 18.1 on the Master PC. Doing this saves all of the steps within the sequence file as specified in ***Single-file Sequence Format on page 436***.

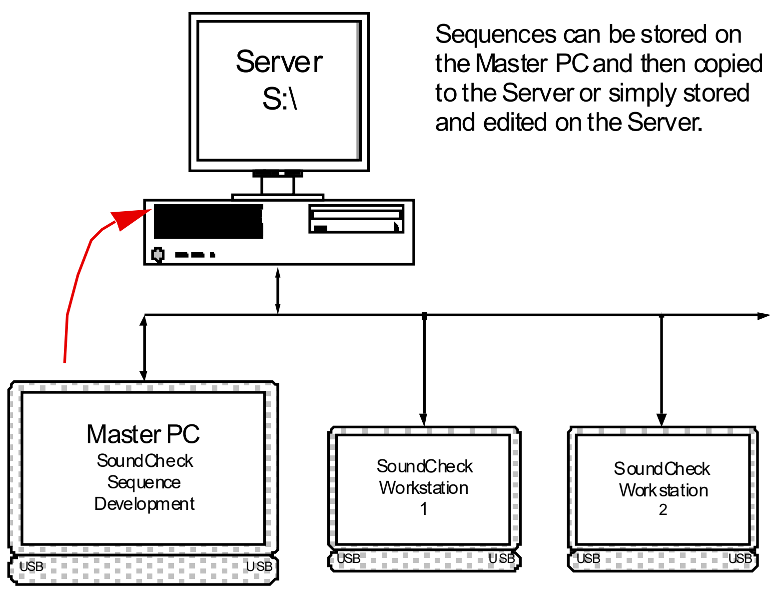
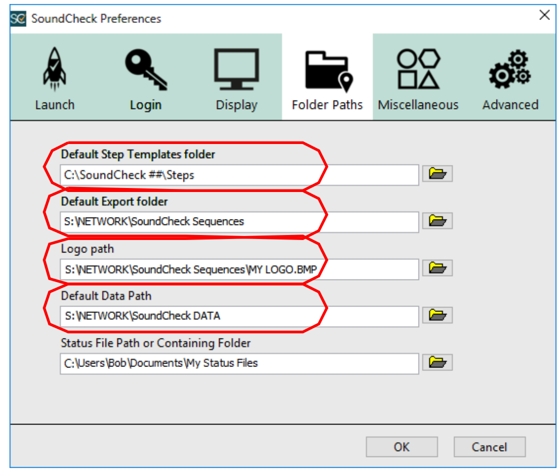


Figure 14-1: SoundCheck Network Scheme

2. On the SoundCheck Main Screen, click on **Edit** and then select **Folder Paths**.

3. Step Templates can remain on the local PC. They are not linked to sequences.
4. Browse to the location for the Import/Export folder. In this case it is on the network. Click on **Select Cur Dir**.
5. Browse to the location of the Logo for use in print-outs. Click on **Select Cur Dir**.
6. Status.dat files can also be located on the network.
7. The Master PC can now open sequences from the network. On the SoundCheck main window click on **File** and then **Open**. Browse to the network sequence location and open a sequence.



Note: Changes to the Master PC System Hardware and System Calibration Configurations will not be replicated on the Workstation PCs.

Workstation PC Configuration

8. For existing workstations: Make sure the sequences unique to that workstation have been exported and backed up. ***See "Exporting Sequences" on page 451.***
9. The **Hardware Configuration** of the Workstation PC should be set up for the specific audio interface used on each machine. Note that the physical hardware can be different on each machine. ***See "Hardware Configuration" on page 59.***

   Of course, multichannel sequences will require multichannel hardware.
10. Each workstation must have the same Signal Path naming convention as the Master PC. ***See "Naming - Best Practices" on page 88.***

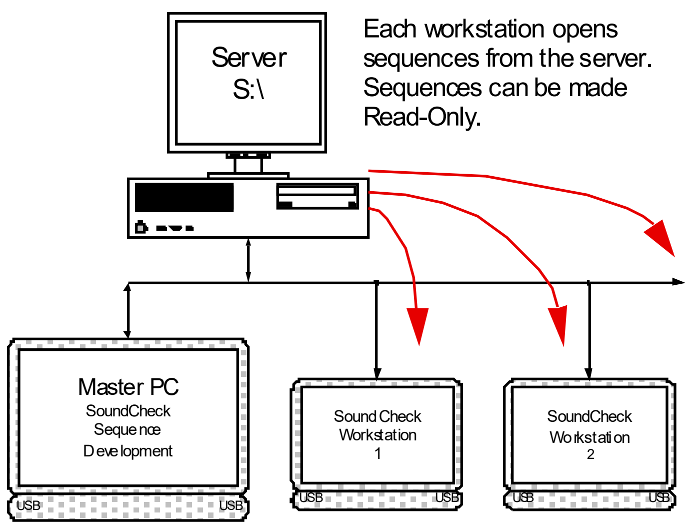


Figure 14-2: Workstation Configuration

As long as the "**Signal Path naming convention**" of the workstations matches the Master PC, the sequence will access the hardware and work as expected.

Each workstation will have the same channel names and channel structure, but will have unique calibration values for each input and Output Signal Path.

11. See ***Calibrating SoundCheck on page 92*** for instructions on input and output calibration.

Set the folder paths in the same way the Master PC is setup. Generally, sequences are not edited on the workstation computers. If this is the case, the Sequence Editor module is not needed for the workstations.

Appendix 15: Data Import Wizard Tutorial

Importing text from a saved file

1. From the Memory List click **Data**
2. Select **Open Data**
3. Select a **.TXT** file to start wizard

The first Import Wizard screen (***Figure 15-1***) will display the contents of the selected text file. The default settings are:

- Delimited
- Column delimiter - tab
- Data is in - columns
- Data offset: 0 columns and 0 rows

The text file shown in ***Figure 15-1*** is tab delimited in a row format and includes a header row of frequency values.

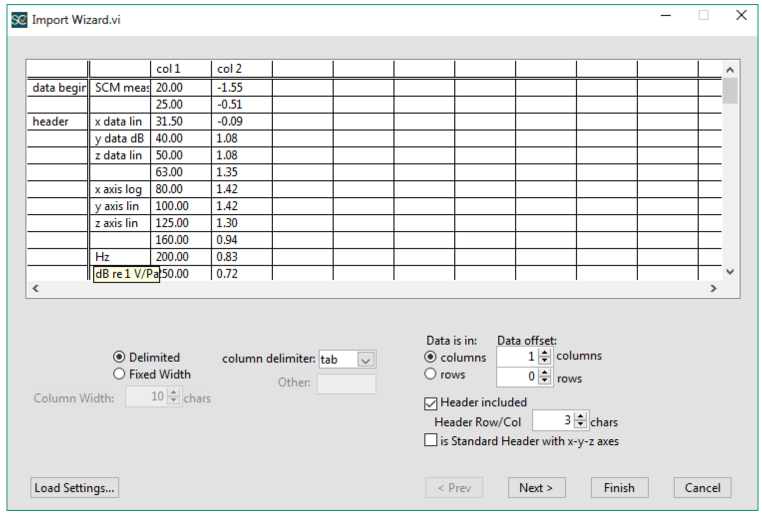


Figure 15-1: Data Import Wizard Screen 1

To configure the import wizard, do the following:

1. Choose whether data is delimited or fixed width. If the data is delimited, choose the type of delimiter (tab, comma, space, other).
2. Choose the data format (columns or rows).
3. Choose whether the data file includes a header. The **row/col:** box refers to how many rows (or columns) the header is offset from the first row (or column). In ***Figure 15-1***, the header information is in the first row, therefore the offset value is 0. Check the **Standard Header** box if the data file was stored using the SoundCheck Standard header format. (See ***Autosave Editor on page 213*** for more details regarding the standard header format.)
4. Increment the Data offset if the data begins at a row (or column) other than Row 1 (or Column 1). If the first row or column contains the data (as in ***Figure 15-1***), then the offset should be 0.
5. Click **Next** when finished.

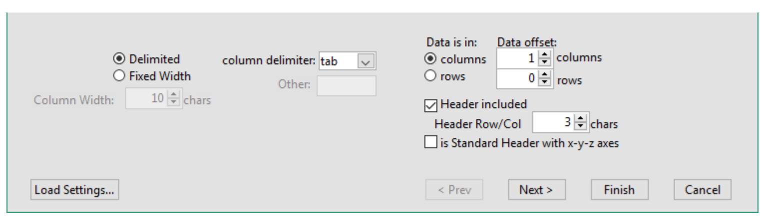


Figure 15-2: Configuring Import Wizard Screen 1

The Import Wizard table in ***Figure 15-3*** shows if the X-axis/Y-axis appears in Columns or Rows.

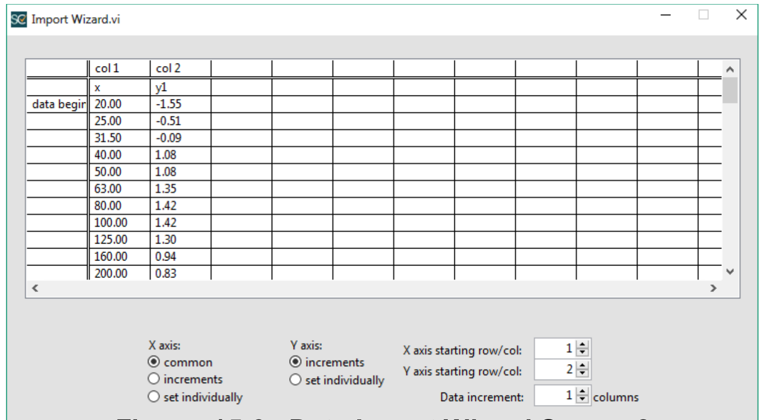


Figure 15-3: Data Import Wizard Screen 2

X-axis

This is typically set to **common**

Y-axis

Typically set to **increments**.

If the X-axis is set to **increments** , SoundCheck will interpret the first row (or column) to be the X-axis and the second row (or column) to be the corresponding data values.

For files containing multiple data sets:

The X and Y-axes are grouped in pairs (e.g., rows 1 and 2 are the X and Y data respectively for Data Set #1

Rows 3 & 4 are the X and Y data respectively for Data Set #2, etc.)

The **set individually** option allows you to choose which rows or columns are X-axis values.

Click **Next** once the appropriate settings have been made.

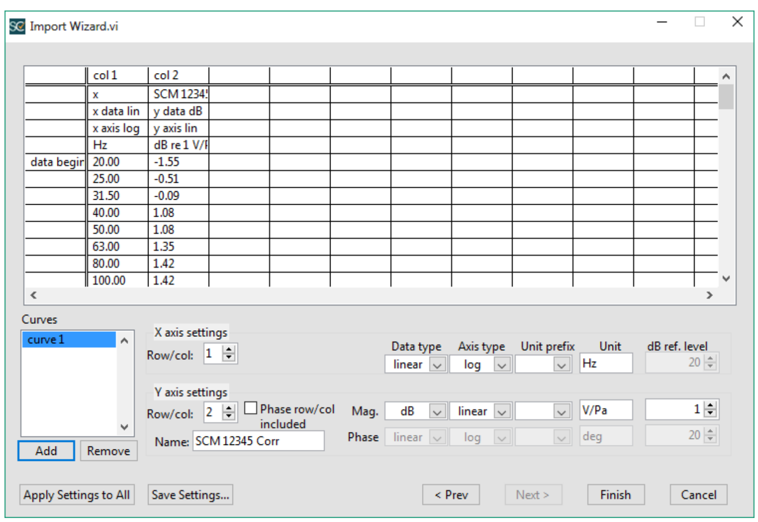


Figure 15-4: Data Import Wizard Screen 3

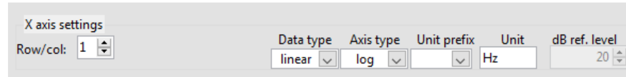
Note: The import wizard keeps imported data in memory. Depending on the computer hardware, you may not be able to import large data arrays such as an FFT spectrum or time waveform.

Note: The example in ***Figure 15-4*** shows only X and Y data. For files with both Magnitude and Phase, see ***Phase Data on page 613***.

The following is a breakdown of the settings in ***Figure 15-4***.

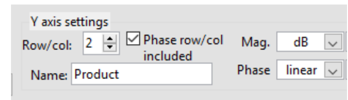
X axis settings

- Location of X-axis values
- X-axis data type and units. Typically the X-axis values are linear (Hz) and displayed on a logarithmic axis.



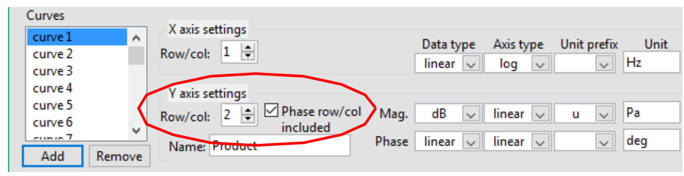
Y axis settings

Location of Y-axis values. The Row/col: value is based on which curve is highlighted in the Curves list. In this case, curve 1 (the highlighted curve) is located in Row 2.



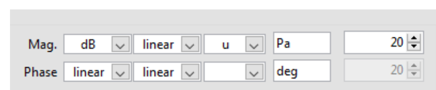
Phase Data

When importing a curve with Magnitude and Phase data, select **"Phase row/col included"**.



Note: When importing a reference curve with Magnitude and Phase you may need to remove extra Curves created by the wizard. Then select "Phase row/col included". This will enable the Phase fields for editing and setup the proper Z-axis column for phase data.

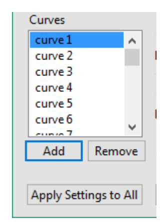
Y-axis data type and units. Typically, the Y-axis values are in dB and displayed on a linear axis. The decibel reference value can be entered.



Note: When importing correction or equalization curves, choose Units with no prefix or unit name and set the dB reference to 1.00.

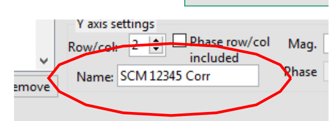
Curves

Data curves are available in the imported text file. Curves can be added or removed by clicking **Add** and **Remove**. Click **Apply Setting to All** to apply the settings for a highlighted curve; e.g., Units of dB re 20 µPa to all the curves in the list.



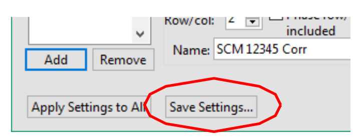
Name

Allows you to create a custom name for a curve. The highlighted curve(s) can be individually named by typing in the Name field.



Save Settings

The import configuration should be saved using **Save Settings…** This allows you to recall these settings in the future by using the **Load Settings...** button shown in ***Figure 15-1***. Once settings are loaded, click **Finish** and data is immediately imported.



Apply Settings to All

Apply data/axis type, magnitude, units to all curves for import

Add

Add new curve to import

Remove

Remove curve from import list

Importing Correction Curves From Other Manufacturers

If an imported curve does not have a 0 dB value at 1 kHz you must change the calibration Reference Frequency in the Calibration Editor to a point on the correction curve that is at 0 dB.

This may also have to be done when importing Diffuse Field or other such correction curves without data at 1 kHz.

See ***Importing Correction Curves on page 85***.

Appendix 16: Default Sequence List

The following list of sequences is included with SoundCheck.

Sequence	Description
Calibration	
Accelerometer Calibration	This is not to be used as a sequence. It is only a placeholder for Calibration Editor functions.
AC621 Get Latencies	Used to determineoptimal Hardware Editor Latency values for AmpConnect 621.
AmpConnect 621 Self Test	Used to test the functionality and connections of AmpConnect 621. Use a Stweep Stimulus.
AmpConnect ISC Self Test	Used to test the functionality and connections of AmpConnect. Use a Stweep Stimulus.
Microphone Calibration	This is not to be used as a sequence. It is only a placeholder for Calibration Editor functions.
Self Test	A diagnostic tool used to check the settings and performance of the audio interface (sound card)
Electronics	
Amplifier THD+N	Tests the gain response and THD+N for an amplifier or other electronics
MP3 Player (Multitone)	Uses a multitone stimulus to test various parameters of a portable music player
Headphones and Headsets	
Bluetooth Headset - Receive	Subsequence used in Bluetooth Headset sequence for testing the receive side
Bluetooth Headset - Send	Subsequence used in the Bluetooth Headset sequence for testing the send side
Bluetooth Headset	Uses a multitone stimulus to test a Bluetooth headset in both the send and receive directions
Headphones	An example headphone sequence that measures frequency response and distortion
Hearing Aids	
Frequency Response	Basic frequency response measurement (gain versus frequency) for a hearing aid
Input vs Output	Uses amplitude sweeps at several frequencies to generate the input-output curves for a hearing aid
OSPL 90	Performs the OSPL 90 test from the ANSI and IEC hearing aid standards
Release Time - 1996	Tests hearing aid release time according to ANSI S3-2003
How To Examples	
ActiveX & Test Stand example	Can be called by the ActiveX example code located at C:\SoundCheck x.x\External Control Examples_Legacy ActiveX Examples\VB2010\VB2010 Example.exe
Autosave	Demonstrates several methods for using the autosave step
Average Sensitivity	A very basic sequence that calculates the average sensitivity from a response curve
Complex Averaging in Loop	Calculates a complex average (magnitude and phase) over multiple measurements
Confidence & Noise	Demonstrates the confidence feature in the analysis editor by using 3 stimuli of varying lengths
Diff Distortion	An example for testing difference frequency distortion
Dual Channel Analysis	Measures frequency response with three different methods: stepped sine, log sweep + TSR, pink noise and transfer function
IM and Diff Distortion	An example for testing intermodulation distortion
Limits in Reference to Standard	Measures a reference standard and uses its response to automatically generate limits for subsequent measurements
Loop Stimulus Level	A complex example of sequence logic that searches for the stimulus level that will generate 3% THD in the DUT

Multitone Analysis	A basic example of the multitone stimulus and analysis
Power Averaging	Uses the Real Time Analyzer (RTA) to measure sound power
Statistics	Demonstrates the various uses of the statistics step
THD at Actual Measured Frequency	Plots the harmonics at their measured frequency and generates a normalized distortion curve
Virtual Instrument Acquisition	Walks through each of the virtual instruments and demonstrates their uses in a simple electrical loop back
Loudspeakers	
Complete Test Using AmpConnect 621	An example of a loudspeaker production test, with AmpConnect 621 controls, that measures frequency response, distortion, impedance, and sensitivity all in one sweep. Requires AmpConnect 621.
Complete Test Using AmpConnect ISC	An example of a loudspeaker production test, with AmpConnect ISC controls, that measures frequency response, distortion, impedance, and sensitivity all in one sweep. Requires AmpConnect ISC.
Complete test	An example of a loudspeaker production test that measures frequency response, distortion, impedance, and sensitivity all in one sweep
Impedance	Measures impedance of a loudspeaker
Loose Particles	Demonstrates the loose particle algorithm
Perceptual Rub & Buzz	Demonstrates the Perceptual Rub & Buzz function which measures distortion based on human hearing models and masking curves (Analysis Editor)
Polar Plot (Linear X turntable)	Measures the polar response and directivity of a loudspeaker in the horizontal and vertical axes by automatically controlling a Linear X turntable
Polar Plot (Outline ET250-3D)	Measures the polar response and directivity of a loudspeaker in the horizontal and vertical axes by automatically controlling an Outline ET250-3D turntable
Rub & Buzz	An example of measuring rub & buzz
Time Selective Response	An example of measuring frequency response using a log sweep and the TSR algorithm
T-S Parameters – Added Mass T-S Parameters – Known Mass T-S Parameters – Known Volume	Thiele-Small Parameters: Generates Thiele-Small parameters via the added mass or fixed volume methods
Triggered Record Using WAV File	An example for testing devices without an analog input such as tablets, cellphones and MP3 players
Microphones	
Open Loop Microphone	Demonstrates the two most common microphone measurements, frequency response and sensitivity, on a microphone embedded in a recording device
Mic substitution	A basic example of the two most common microphone measurements: frequency response and sensitivity
Microphone Self Noise	Measures the self noise of a microphone
Microphone	Measures frequency response and sensitivity of a microphone with an equalized stepped sine sweep
SC ONE	
SC ONE - AmpConnect - Headphones SC ONE - AmpConnect - Loudspeaker SC ONE - AmpConnect - Microphone (Measure Reference) SC ONE - AmpConnect - Microphone	SoundCheck ONE template sequences serve as a starting point for making new sequences. They contain all the necessary steps to perform the essential measurements for their test application. The sequences in this group are for use with AmpConnect ISC.
SC ONE - AmpConnect Self Test	Used to test the functionality and connections of AmpConnect ISC
SC ONE - AudioConnect - Headphones SC ONE - AudioConnect - Loudspeaker SC ONE - AudioConnect - Microphone (Measure Reference) SC ONE - AudioConnect - Microphone	SoundCheck ONE template sequences serve as a starting point for making new sequences. They contain all the necessary steps to perform the essential measurements for their test application. The sequences in this group are for use with AudioConnect.

SC ONE - AudioConnect Self Test	Used to test the functionality and connections of AudioConnect
Telephones	
Receive	Measures response, distortion, and loudness for the receive side of a telephone
Send	Measures response, distortion, and loudness for the send side of a telephone

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SoundCheck® 18 Available Functionality

Industry Standard Sequences

(Please contact Listen for a complete list)

• IEEE • ITU • ANSI • IEC • TIA •TBR • ISO • AES • ALMA

SoundCheck Virtual Audio Test Bench

- Run multiple instances of each virtual instrument simultaneously
- Save to Memory List available in all meters

Manual Signal Generator

- Sine
- Pink and White Noise with user-defined frequency range
- Streaming WAV file from disk with RMS, Peak level and active speech level calibration
- Equalization using calibration measurements or any user-defined curve

Multimeter

- AC RMS, AC Peak and DC levels (Average, Max Hold, Min Hold) with overload indication
- Linear and Continuous-Moving averaging time (fast, slow, and user-defined)
- Linear and Exponential averaging time (Fast, Slow, and user-defined)
- A, B, C and user-defined weightings
- Selectable Max/Min limits with Pass/Fail indication
- Save and Recall specific multimeter settings
- Fixed or auto-tracking bandpass filter option
- 'Linear Repeating' averaging mode

Oscilloscope

- Triggering
- Delta cursor
- Selectable graphical zoom
- View spectrum of current waveform

Spectrum Analyzer

- FFT with arbitrary number of spectral lines (only limited by computer speed and memory)
- Hanning, Hamming, Blackman-Harris, Exact Blackman, Blackman, Flat top, 4 Term Blackman-Harris, and 7 Term Blackman-Harris windows
- Triggering
- Complex or power averaging
- Average, Maximum, Minimum level with overload and real time indicators
- Selectable averaging time (Linear & Exponential)
- A, B, C and user-defined weightings
- Pure tone frequency and amplitude extraction with "snap to max"
- Delta and Harmonic cursor with THD readout
- Selectable graphical zoom
- View last waveform of current spectrum
- Save to memory of current spectrum

Real-Time Analyzer

- 1/1, 1/3, 1/6, 1/12, 1/24 octave filters with true digital recursive filters
- Average, Maximum, Minimum level with overload and real time indicators
- A, B, C and user defined weighting filters
- Complies with ANSI S1.11 and IEC 1260
- Linear and Exponential averaging time (Fast, Slow, and user-defined)

Real Time Distortion Analyzer

- continuous real time measurement of output distortion
- Select THD / THD+N, THD / THD+N Residual and SINAD
- A, B, and C weighting filters along with user-defined arbitrary weighting functions
- Distortion over time using optional strip chart recorder

Frequency Counter

- High resolution frequency measurement
- Frequency value can be saved to the Memory List for use in a sequence

Strip Chart Recorder

- Provides measurement over time capability for the multimeter, distortion analyzer and frequency counter

- Plot continuously or for a predefined amount of time and plot instantaneous results or repeating averages
- Curves and Values can be saved to the Memory List for use in a sequence

SoundCheck Step Editors

Hardware

- Windows Multimedia devices including sound cards with ASIO drivers (PCI, PCMCIA, USB, Fire wire), Bluetooth, and VoIP.
- Apple's Core Audio devices (SoundCheck Mac version)
- NI DAQmx data acquisition cards including NI 4461
- Calibration and settings including sampling rate, bit depth, analog or digital audio, maximum voltage, and alias free frequency limit.
- I/O cards for TTL relay control
- Computer interfaces control with RS-232, GPIB (IEEE-488), footswitch and buzzer.
- Configuration for AmpConnect™, AudioConnect™, AudioConnect 4×4™, SoundConnect 2™, DC Connect™, BTC - 4148/4149 and BQC-4148/4149
- Multichannel configuration with table view of channels

Calibration

- Measure Input and Output sensitivities for transducers, amplifiers, and signal conditioning devices using built-in routines and store calibration history
- Calibration with external, absolute source including acoustic, vibration, or voltage.
- User-defined physical units (e.g. Pa, V, G, etc.)
- Complex Equalization (amplitude and phase) of input and output devices (e.g. microphones and amplifiers). If the output device is an acoustic source, (e.g. loudspeaker, mouth simulator), SoundCheck can automatically equalize any test signal including arbitrary signals.
- Import of EQ/Correction curves for transducers

Messages

- Message steps initiated based on Pass/Fail conditions
- Display text message in local language, input numeric values, Yes/No dialog
- Digital I/O
- IEEE-488 and RS-232
- Controls for AmpConnect 621™, AmpConnect ISC™ AudioConnect™, AudioConnect 4×4™, SoundConnect 2™, DC Connect™, BTC-4148/4149 and BQC-4148/4149 setup

Stimulus

- Log sweep ("Farina" sweep)
- Sine (stepped – any linear or logarithmic resolution, and amplitude sweep)
- Two-tone (two sweeping tones for Difference Frequency Distortion or one fixed and one sweeping tone for IM)
- Multitone with linear or logarithmic spacing
- Noise (pink, white, MLS with user-defined bandpass range)
- Arbitrary (any WAV file)
- Equalization
- DC (requires Listen DC Connect or National Instruments hardware)
- Sweep Equalization for Minimized Transients - Selecting equalization enables a smooth transition between steps in stepped sine amplitude and frequency sweeps

Acquisition

- Play & Record, or any combination of Signal Generator, Multimeter, RTA, and FFT
- Capture response time waveform as WAV file
- Simultaneous acquisition using up to 64 channels
- Triggered record, Frequency Trigger, Level & Freq Trigger and Level & Cross-Correlation Trigger
- Record level monitoring

Analysis

Time

- Impulse Response
- Auto-Correlation
- Cross-Correlation

- Time Envelope
- Loose particle detection
- Automatic delay compensation

Frequency

- FFT & DFT (any size), and Nth octave resolution
- Hann, Blackman-Harris and Flat Top windows
- Auto-spectrum & Cross-spectrum
- Spectral Scaling: RMS or Power Density
- Frequency and phase response including harmonics
- Complex or power averaging
- Relative or absolute response
- Coherent Output Power
- Coherence & Non-Coherence
- Signal-to-Noise Ratio
- Measurement Confidence
- Impedance

Algorithms

- Broadband RMS to measure unfiltered level of an AC or DC signal
- Average FFT Spectrum
- Time Selective Response ('Farina" method) to measure free-field and impulse response of fundamental AND harmonics. This includes deconvolved time response and choice of time windows.
- Heterodyne to measure frequency and phase response with optimal accuracy
- HarmonicTrak™ Algorithm tracks level and phase of any user-selected harmonics including sub-harmonics or intermodulation products; no limit to number of harmonics
- Loose particle detection
- Multitone
- RTA Spectrum & frequency response
- Transfer function between any two channels

Distortion

- THD and Rub & Buzz
- Normalized THD and Rub & Buzz distortion (harmonics compared to amplitude of fundamental at measured frequency)
- Perceptual Rub & Buzz in phons
- THD + Noise
- Intermodulation or Difference Frequency
- Difference Frequency
- Non-Coherent Distortion

Autosave

- Save to SoundCheck binary formats: .dat, .res and .wfm
- WAV
- SQL & MS Access databases
- MATLAB
- MS Excel
- ASCII text

Recall

- Automatically recall data or results

Post-processing

- Complex math: Addition, subtraction, multiplication, division, offset by constant (X, Y, or Z dimensions), change sign, reciprocal, absolute value, square, square root, exponential, and logarithm
- Scalar (Ave, Power, Max, Min, Resonant Frequency, Q, Notch, Loudness)
- Windowing (time and frequency)
- User-defined Equations (e.g. Thiele-Small parameters)
- Change resolution to Nth octave or user defined linear or logarithmic
- Nth Octave synthesis
- Power Summation of any user-defined frequency range
- Group delay
- Unwrapped phase
- Search range to find intersection of two curves (e.g. -3dB points of crossover network)
- Curve smoothing with 1/3, 1/6, 1/12, and 1/24 octave or user-defined linear or log resolution
- Loudness rating according to IEEE and ITU-T; example sequences for TIA and other industry standards included
- Attack & Release time calculates the time for the response signal to rise or decay, respectively, by a user-defined amplitude in dB or linear units
- Directivity Index
- Resampling and sampling rate correction

- FFT / Inverse FFT
- Zwicker loudness (level and spectrum)
- Standard & arbitrary waveform filtering
- Active speech level (ITU-T P.56)
- Average curve

Statistics

- Average
- Max, Min, Mean
- Standard Deviation with user-defined sigma
- Process Control (Cp & Cpk)
- Best and Worst Fit to Average with ranking
- Yield
- Histogram and bell curve fit

Limits

- Pass/Fail
- Absolute
- Floating (x & y directions)
- Aligned to a pre-defined value (e.g. 0 dB at 1 kHz)
- Dynamic, using live measurements
- Waveforms, single values, and curves
- Control of significant digits
- Margin, Critical and Failed points

Display

- Multiple displays on separate tabs allow viewing of curves, single values, and test results with PASS/FAIL indicators
- Display windows can be saved to Word or as HTML document
- Curves with different units (e.g. dBSPL and Ohms) can be displayed in one graph with no limit to number of curves displayed in a single graph
- Displays include XY graph, table, results, text, polar plots and embedded images
- Default template and duplicate
- Save to image file
- Lock Memory List and displays to protect display layout
- Memory List data search function

Printing

- Report generation to Excel, Word, HTML or image files
- Use of templates (Excel, Word)
- Print preview
- Direct print to file or printer
- Automatic file naming

Custom Steps

Outline ET250-3D Turntable Control

- Ethernet control for the Outline ET250-3D turntable

Instrument Open Close

- Template for creating a custom step combining your LabVIEW code with SoundCheck virtual instruments

System Custom Step

- Run Command Line operations as part of a sequence

Mixer Volume

- Control input / output levels of a 2 channel WDM / WASAPI / Core Audio device

RS232 Read Integer

- Programming example that reads the integer value from an external RS232 device

Serial Number Write Read

- Programming example for creating a custom vi to read or write to the SoundCheck serial number field

Empty Memory List Item

- Creates place holder for data for use with LabVIEW

Open Before Converting Old Custom VIs

- Tool for updating your Custom VIs to latest version of SoundCheck / LabVIEW

Automation

- Pass test configuration to and from external programs
- TCP/IP commands
- Python command library

Glossary

Term	Definition
A2DP	A Bluetooth profile that allows for streaming stereo audio between, e.g., an MP3 player and headphones or speakers. (aka: Bluetooth Audio Streaming)
Absolute Standard Deviation	The standard deviation calculated at each point of the curve. See Statistics chapter for formula.
algorithm	A procedure for solving a mathematical problem in a finite number of steps that frequently involves repetition of an operation. Algorithm here refers specifically to a procedure that encodes audio information (so that it can be sent at high speed across data lines) and decodes the transmitted information into audio at the receiving end.
Amplitude	The instantaneous magnitude of an oscillating quantity such as sound pressure. The peak amplitude is the maximum value.
Anechoic	Without echo
Area of Audibility	The area within which a specific sound or sounds are audible.
aptX	Bluetooth codec offering better quality than SBC. Used in A2DP. ***See "A2DP" on page 623.***
aptX HD	Bluetooth codec supporting 24-bit/48 kHz audio with a bitrate of 576 kbps. aka: aptX Lossless. ***See "A2DP" on page 623.***
Best/Worst Fit to Average	Determines which of the selected curves is the closest/furthest from the average of the curves. See Statistics chapter for formula.
Bluetooth	Bluetooth® is a wireless standard for exchanging data between devices using UHF radio waves
Center Frequency	See IEEE 269 and/or B&K Frequency Analysis Text Book definition
Conditional Branching	Used in a sequence to Jump over steps in a sequence according to the Pass/Fail criteria of a step.
Cp (statistics)	A measure of process performance. The relationship of the +/- 6? value to the user specified limits. See Statistics chapter for formula.
Cpk (statistics)	The same as Cp except that it takes into consideration how centered the data is with respect to the limits. See Statistics chapter for formula.
CPB	Constant Percentage Bandwidth.
Crest Factor	crest factor = absolute peak value / rms value = max[|x|]/rms(x)
CVSD	Continuously variable slope delta modulation. Bluetooth codec. A compromise between simplicity, low bitrate, and quality. Used in HFP. ***See "HFP" on page 625.***
dB	See decibel
dB (A) or dBA	A sound-level meter reading with an A-weighting network simulating the human-ear response at a loudness level of 40 phons.
dB (B)	A sound-level meter reading with a B-weighting network simulating the human-ear response at a loudness level of 70 phons. C34
dBm0	dBm0 is a digital level. 0 dBu = 775 mV = 1 mW@600 Ω. A reference voltage of 775 mV yields 1 mW with a load of 600 Ohm.
dBSPL	A sound-level meter reading with no weighting network in the circuit, i.e., flat. The reference level is 20 µPa.
Decade	Ten times any quantity or frequency range. The range of the human ear is about 3 decades.

Decibel	A logarithmic form of any measured physical quantity and commonly used in the measurement of sound. The decibel provides the possibility of representing a large span of signal levels in a simple manner as opposed to using the basic unit Pascal. The difference between the sound pressure for silence versus a loud sound is a factor of 1,000,000:1 or more, and it is not practical to use these large numbers. Doubling of Sound Pressure = 6 dB. Doubling of Sound Power = 3 dB. Doubling of Perceived Sound Level = 10 dB (approximately).
Decibel	dB---the term used to identify ten times the common logarithm of the ratio of two like quantities proportional to power or energy. (See Level, sound transmission loss.) One decibel corresponds to a power ratio of 100.1.
Directivity index (DI)	The difference between sound pressure level in any given direction in the acoustic far field and the average sound pressure level in that field.
DUT	Device Under Test
Equal loudness contour	A contour representing a constant loudness for all audible frequencies. The contour having a sound pressure level of 40 dB at 1,000 Hz is arbitrarily defined as the 40-phon contour.
Equalization	The process of adjusting the frequency response of a device or system to achieve a flat or other desired response.
Far field	That part of the sound field in which sound pressure decreases inversely with distance from the source. This corresponds to a reduction of approximately 6 dB in level for each doubling distance.
Feedback, acoustic	Unwanted interaction between the output and input of an acoustical system, e.g., between the loudspeaker and the microphone of a system.
FFT	Efficient algorithm to calculate the Fourier Transform.
Filter, band pass	A filter that passes all frequencies between a low-frequency cutoff point or a high-frequency cutoff point.C55
Filter, high pass	A filter that passes all frequencies above a cutoff frequency.
Filter, low pass	A filter that passes all frequencies below a certain cutoff frequency.
Fletcher-Munson Curve	Our sensitivity to sound depends on its frequency and volume. Human ears are most sensitive to sounds in the midrange. At lower volume levels humans are less sensitive to sounds away from the midrange, bass and treble sounds "Seem" reduced in intensity at lower listening levels.
Fourier analysis	Application of the Fourier transform to a signal to determine its spectrum.
Free field	An environment in which a sound wave may propagate in all directions without obstructions or reflections. Anechoic rooms can produce such an environment under controlled conditions.
Frequency	The number of times per second that the sine wave of sound repeats itself. It can be expressed in cycles per second, or Hertz (Hz). Frequency equals Speed of Sound / Wavelength.
Frequency Masking	Principle where louder sounds render soft sounds inaudible in nearby frequency bands. This is the principle behind perceptual encoding.
Frequency response	The changes in the sensitivity of a circuit, device, or room with frequency.
FSD	Full Scale Deflection
full duplex	Telco communication that is bi-directional. ISDN is full duplex, so each end of the connection can simultaneously transmit to the other.
Fundamental	The lowest frequency of a note in a complex wave form or chord.
G.711	Refers to the transmission of audio via a POTS (Plain Old Telephone) circuit. Frequency response is limited to about 3.5 kHz.
Gain	To increase in level. The function of a volume control.
GUI	Graphical User Interface

Handshaking	Protocols usually implemented in hardware that let one data device tell another that conditions are right (or wrong) for communications. A simple example: a printer telling a computer that it is OK to print.
Harmonics	Also called overtones, these are vibrations at frequencies that are multiples of the fundamentals. Harmonics extend without limit beyond the audible range. They are characterized as even-order and odd-order harmonics. A second-order harmonic is two times the frequency of the fundamental; a third order is three times the fundamental; a fourth order is four times the fundamental; and so forth. Each even-order harmonic second, fourth, sixth, etc.- is one octave or multiples of one octave higher than the fundamental; these even-order overtones are therefore musically related to the fundamental. Odd-order harmonics, on the other hand third, fifth, seventh, and up-create a series of notes that are not related to any octave overtones and therefore may have an unpleasant sound. Audio systems that emphasize odd-order harmonics tend to have a harsh, hard quality.
Hearing Range (human)	A healthy young person generally can hear frequencies from approximately 20 Hz to 20000 Hz, and sound pressure levels from 0 dB to 130 dB or more (threshold of pain). The smallest perceptible change is 1 dB.
Hearing sensitivity	The human ear is less sensitive at low frequencies than in the midrange. Turn your volume knob down and notice how the bass seems to "disappear". To hear low bass requires an adequate SPL level. To hear 25 Hz requires a much higher SPL level than to hear 250 Hz.
Hertz	The unit of frequency, abbreviated Hz. The same as cycles per second.
HFP	Hands Free Profile. Bluetooth profile that allows for voice transmission between, e.g., a mobile phone and a wireless headset or a car kit.
High-pass filter	See Filter, high pass.
Impedance	The opposition to the flow of electric or acoustic energy measured in Ohms (Ω).
Impulse	A very short, transient, electric or acoustic signal.
Impulse response	Sound pressure versus time measurement showing how a device or room responds to an impulse.
In phase	Two periodic waves reaching peaks and going through zero at the same instant are said to be "in phase."
Infrasound	Frequencies below 20 Hz. Humans perceive frequencies below about 20 Hz as pressure rather than sound.
Inverse-square law	Under far field/free field conditions, sound intensity varies inversely with the square of the distance from the source. In pure spherical divergence of sound from a point source in free space, the sound pressure level decreases 6 dB for each doubling of the distance.
Loudness	The subjective judgment of intensity of a sound by humans. Loudness depends upon the sound pressure and frequency of the stimulus. Loudness was defined by Fletcher and Munson (1933) as a physiological description of the magnitude of an auditory sensation. The definition of loudness was later refined as a definition of the attribute of auditory sensation corresponding most closely to the physical measurement of sound intensity, but is not always accurate. Loudness is a subjective quantity and all measurement techniques are based on assumptions and interpretation.
Masking	The process by which the threshold of audibility for a sound is raised by the presence of another (masking) sound. A masking noise is one that is intense enough to render inaudible or unintelligible another sound that is also present.
Max (statistics)	The maximum value at each point of the curves being compared.
Mean (statistics)	The average value at each point of the curves being compared
Microphone	An acoustical-to-electrical transducer by which sound waves in air are converted to electrical signals.
Min (statistics)	The minimum value at each point of the curves being compared.

mSBC	A mono version of the SBC Bluetooth codec. aka: Wide Band Speech (WBS). Used in Bluetooth HFP. ***See "HFP" on page 625.***
NaN	Not a Number
Near field	Locations close to the sound source between the source and the far field. The near field is typically characterized by large sound pressure level variations with small changes in measurement position from the source. This is a physical region in space where the inverse square law does not apply.
Noise	Traditionally, noise has been defined as unwanted, undesired, or unpleasant sound. This makes noise a subjective term. Sounds that may be unwanted and undesired by some may be wanted and desirable by others. Noise is sound, as defined in this document: a pressure variation, etc. In order to keep terms used in soundscape management as non-subjective as possible, sounds should be classified as either appropriate or inappropriate, rather than as "noise." or "sound." The appropriateness of any sound in a given area of a park will depend on a variety of factors, including the management objectives of that area.
Noise Free Interval (natural sounds only)	The length of the continuous period of time during which only natural sounds are audible. Though little research has been conducted to relate how this measure correlates with visitor judgments or with common experiences in park settings, it should provide a reasonable measure of the existence and availability of periods with only natural sounds. It is also a metric that requires no acoustics knowledge to be meaningful. Over the coming years of soundscape data collection, the NPS will acquire such data and develop an understanding of how this metric can best be used to aid in assessing and managing park soundscapes.
Octave	An octave is a doubling or halving of frequency. 20 Hz-40 Hz is often considered the bottom octave. For each octave lower in frequency that a speaker tries to reproduce, the speaker needs to move four times as much air!
Octave Band	The segment of the frequency spectrum separated by an octave.
Octave bands	Frequency ranges in which the upper limit of each band is twice the lower limit. Octave bands are identified by their geometric mean frequency, or center frequency.
One-third octave bands	Frequency ranges where each octave is divided into one-third octaves with the upper frequency limit being 2* (1.26) times the lower frequency. Identified by the geometric mean frequency of each band.
Open Loop Test	Stimulating and capturing responses from a device where you don't have direct access to the microphone or speaker.
OSPL 90	Output Sound Pressure Level 90: The output (saturation) sound pressure level with a 90dB SPL input level, which is measured over a frequency range. See OSPL 90 example sequence.
Peak sound pressure level	LPK[nd] ---ten times the common logarithm of the square of the ratio of the largest absolute value of the instantaneous sound pressure in a stated frequency band during a specified time interval to the reference sound pressure of 20 micro pascals.
Percent Time Above Natural Ambient	The amount of time that sound levels from human-caused sound(s) are greater than sound levels of natural ambient sounds in a given area. This measure is not specific to the hearing ability of a given animal, but a measure of when and how long human-caused sound levels exceed natural ambient sound levels.
Percent Time Audible	The amount of time that various sound sources are audible to animals, including humans, with normal hearing (hearing ability varies among animals). A sound may be above natural ambient sound pressure levels, but still not audible to some animals. This information is essential for measuring and monitoring human-caused noise in national parks. These data can be collected by either a trained observer (attended logging) or by making high-quality digital recordings (for later playback). Percent Time Audible is useful because it is a measure that is understandable without any acoustics knowledge. It is a measure that can be specific to a given animal, and it is a metric that correlates well with park visitor judgments of annoyance and with visitor reports of interference from certain noise sources with natural quiet and the sounds of nature.
Phase	Phase is the measure of progression of a periodic wave. Phase identifies the position at any instant which a periodic wave occupies in its cycle. It can also be described as the time relationship between two signals.

Phase shift	The time or angular difference between two signals.
Phon	The loudness level in phons of any sound is defined as being numerically equal to the dBSPL of a 1000 Hz tone that is judged by the average observer to be equally loud.
Pink noise	Noise with a continuous frequency spectrum and with equal power per constant percentage bandwidth. For example, equal power is any one-third octave band.
Pitch	A subjective term for the perceived frequency of a tone.
Polarity	The positive or negative direction of an electrical, acoustical, or magnetic force. Two identical signals in opposite polarity are 180 degrees apart at all frequencies. Polarity is not frequency dependent.
POTS	Plain Old Telephone Service. Standard analog phone lines used for voice and computer modem operation.
Power Sum	Calculates the square root of the sum of the squares of each Y value in a spectrum. See ***Power Sum of Curve on page 244***.
Pressure Microphone	Used to measure the sound pressure on the microphone diaphragm. Typically used to measure the sound pressure in a coupler or measure the sound pressure on a wall.
Pressure zone	As sound waves strike a solid surface, the particle velocity is zero at the surface and the pressure is high, which creates a high-pressure layer near the surface.
Pure tone	A tone with no harmonics. All energy is concentrated at a single frequency.
Random noise	A noise signal, commonly used in measurements, which has constantly shifting amplitude, phase, and a uniform spectral distribution of energy.
Reflection	For large surfaces compared to the wavelength of impinging sound, sound is reflected much as light is reflected, with the angle of incidence equaling the angle of reflection.
Refraction	The bending of sound waves traveling through layered media with different sound velocities.
Resistance	The quality of electrical or acoustical circuits that results in dissipation of energy through heat.
Resonance	A natural periodicity, or the reinforcement associated with this periodicity.
Resonant frequency	Any system has a resonance at some particular frequency. At that frequency, even a slight amount of energy can cause the system to vibrate. A stretched piano string, when plucked, will vibrate for a while at a certain fundamental frequency. Plucked again, it will again vibrate at that same frequency. This is its natural or resonant frequency. While this is the basis of musical instruments, it is undesirable in music-reproducing instruments like audio equipment.
Response	See frequency response.
Reverberant sound field	The sound in an enclosed or partially enclosed space that has been reflected repeatedly or continuously from the boundaries.
Reverberation	The persistence of sound in an enclosed or partially enclosed space after the source of sound has stopped; by extension, in some contexts, the sound that so persists.
Reverberation room	A room so designed that the reverberant sound field closely approximates a diffuse sound field, both in the steady state when the sound source is on, and during the decay after the source of sound has stopped.C125
Reverberation time	The tailing off of a sound in an enclosure because of multiple reflections from the boundaries.
Root Mean Square (RMS)	Square root of the average of the squares of the signal. Measures the power of a signal.
RS232	electronic specification for serial data connections between digital terminal equipment (DTE) and data communications equipment. (DCE) Signal is unbalanced. May be either synchronous or asynchronous. Can reside on multiple connector sizes, but most commonly appear on DB9/ DB25 connectors. 50' transmission limit.
S/PDIF	Sony/Philips Digital InterFace. Standard 2-channel digital audio interface found on many consumer-oriented products.

sample rate	The rate at which an analog signal is sampled, or digitized. For instance, when digitizing audio for a CD, the audio is captured at a sample rate of 44.1 kHz, or 44,100 times per second, creating a very close, but not perfect, digital representation of the analog waveform.
SBC	Low-complexity **subband codec**. The only Bluetooth codec that must be supported for A2DP. *See "A2DP" on page 623.*
SC	SoundCheck®
Self-noise, n	Extraneous non-acoustical signals, generated or induced in a measurement system.
Signal-to-noise (SN) ratio	The range or distance between the noise floor (the noise level of the equipment itself) and the test signal or program material.
Sine wave	A periodic wave related to simple harmonic motion.
Sone	The unit of measurement for subjective loudness.
Sound	A wave motion in air, water, or other media. It is the rapid oscillatory compressional changes in a medium that propagate to distant points. It is characterized by changes in density, pressure, motion, and temperature as well as other physical properties. Not all rapid changes in the medium are sound (e.g., wind distortion on a microphone diaphragm).
Sound attenuation	The reduction of the intensity of sound as it travels from the source to a receiving location. Sound absorption is often involved as, for instance, in a lined duct. Spherical spreading and scattering are other attenuation mechanisms.
Sound energy, E	[ML2T-2]; J-energy added to an elastic medium by the presence of sound, consisting of potential energy in the form of deviations from static pressure and of kinetic energy in the form of particle velocity.
Sound insulation	The capacity of a structure to prevent sound from reaching a receiving location. Sound energy is not necessarily absorbed; impedance mismatch, or reflection back toward the source, is often the principal mechanism.
Sound intensity, I	[MT-3]; W/m2 the quotient obtained when the average rate of energy flow in a specified direction and sense is divided by the area, perpendicular to that direction, through or toward which it flows. The intensity at a point is the limit of that quotient as the area that includes the point approaches zero.
Sound isolation	The degree of acoustical separation between two locations, especially adjacent rooms.
Sound level	Of airborne sound, a sound pressure level obtained using a signal to which a standard frequency-weighting has been applied.
Sound Level	The *weighted* sound pressure level obtained by frequency weighting, generally A- or C-weighted. The weighting used must be clearly stated: For L Aeq, "A" denotes that A-weighting was used, and "eq" indicates that an equivalent level has been calculated. Hence, L Aeq is the A-weighted, energy-equivalent sound level.
Sound Level Floor	The lowest amplitude measurable by sound monitoring equipment. Most commercially available sound level meters and microphones can detect sound levels down to about 15 to 20 dBA; however, there are microphones capable of measuring sound levels below 0 dBA.
Sound power level, Lp	Of airborne sound, ten times the common logarithm of the ratio of the sound power under consideration of the standard reference power of 1 pW. The quantity so obtained is expressed in decibels.
Sound power, W	[ML2T-3]; W--in a specified frequency band, the rate at which acoustic energy is radiated from a source. In general, the rate of flow of sound energy, whether from a source, through an area, or into an absorber.
Sound Pressure	Fluctuations in air pressure caused by the presence of sound waves. Sound pressure is the instantaneous difference between the actual pressure produced by a sound wave and the average barometric pressure at a given point in space. Not all pressure fluctuations detected by a microphone are sound (e.g., wind over the microphone). Sound pressure is measured in Pascals (Pa), Newtons per square meter, which is the metric equivalent of pounds per square inch.

Sound Pressure Level (LP or SPL)	The logarithmic form of sound pressure. In air, 20 times the logarithm (to the base 10) of the ratio of the actual sound pressure to a reference sound pressure (which is 20 micropascals, and by convention has been selected to be equal to the assumed threshold of human hearing). It is also expressed by attachment of the word decibel to the number. A 10 dB increase in SPL represents a perceived doubling in loudness sensation and a 3 dB increase is typically a "just noticeable difference" to an average listener.
Sound Speed	The speed of sound in air is about 344 m/sec (1,130 ft/sec or 770 mph) at 70° F at sea level.
Sound waves	Sound waves can be thought of like the waves in water. Frequency determines the length of the waves; amplitude or volume determines the height of the waves. At 20 Hz, the wavelength is 56 feet long! These long waves give bass its penetrating ability, (why you can hear car boomers blocks away).
Spectrum	the distribution of the energy of a signal versus frequency.
Spectrum (Frequency Spectrum)	The amplitude of sound at various frequencies. It is given by a set of numbers that describe the amplitude at each frequency or band of frequencies.
Spectrum analyzer	An instrument for measuring, and usually recording, the spectrum of a signal.
Speech intelligibility	A measure of sound clarity that indicates the ease of understanding speech. It is a complex function of psycho acoustics, signal-to-noise ratio of the sound source, and direct-to-reverberant energy within the listening environment.
Standard Deviation (statistics)	The plus/minus sigma values evaluated on each point of the curves. See Statistics chapter for formula.
STFT	Short Term Fourier Transform
Stweep	Stepped Sine Sweep stimulus signal
Timbre	The quality of a sound that distinguishes it from other sounds of the same pitch and volume. The distinctive tone of an instrument or a singing voice.
Time Weighting	The response speed of the detector in a sound level meter. For Slow response, the response speed is 1 second. Slow time weighting is frequently used in environmental sound measurements. Fast response time is 1/8 second (0.125). This is less frequently used, but will detect changes in sound levels more rapidly. Fast and Slow time weightings were developed, in part, to slow needle movement (called a "decay" factor) in analog meters so investigators could read and record sound levels. New digital sound level meters, while changing numbers rapidly on the screen, store sound level data in memory for later analysis. This means the ability to read numbers on the screen is less important. Hence, the most accurate "weighting" is none. Generally, 1-second Leq data are appropriate; however, when measuring sudden onset sound events such as sonic booms, more frequent data (many readings per second) may be appropriate.
Tone burst	A short signal used in acoustical measurements to make possible differentiating desired signals from spurious reflections.
Total harmonic distortion (THD)	Refers to a device adding harmonics that were not in the original signal. For example: a device that is fed a 20 Hz sine wave that is also putting out 40 Hz, 80 Hz, etc. Not usually a factor in most modern electronics, but still a significant design problem in loudspeakers.
Transient response	The ability of a component to respond quickly and accurately to transients. Transient response affects reproduction of the attack and decay characteristics of a sound.
Transients	Instantaneous changes in dynamics, producing steep wave fronts.
Ultrasound	Sounds or a frequency higher than 20,000 Hz.
Watt	The unit of electrical or acoustical power. 1 watt = 1 joule per second
Wattage	Is the unit of power used to rate the output of audio amplifiers. For a wattage number to have meaning the distortion level and impedance must also be specified.
Wave	A particular type of disturbance that travels through a medium by virtue of the elastic properties of that medium.

Wavelength	Wavelength is the distance a wave travels in the time it takes to complete one cycle. A wavelength can be measured between successive peaks or between any two corresponding points on the cycle. Wavelength (ft) = Speed of Sound (ft) / Frequency (Hz). (speed of sound at sea level is 331.4 meters/second or 1087.42 feet/second).
Weighting	Adjustment of the unweighted frequency response to account for a given human psycho acoustic
White noise (ANS)	Noise with a continuous frequency spectrum and with equal power per unit bandwidth. For example, equal power in any band of 100 Hz width.

These definitions were derived from several sources, including:

Listen, Inc.

Acoustic Alliance. 2001. Glossary of Terms, Acoustic Alliance Products and Services Catalog. Provo, UT.

American National Standards Institute. 1976. Standard Acoustical Terminology, S1.1. American National Standards Institute, NY, NY.

Bruel & Kjaer. 2002. Environmental Noise. Bruel & Kjaer Sound and Vibration Measurement. Naerum, Denmark.

Everest, F. A. 2001. Master Handbook of Acoustics. McGraw-Hill, New York, NY.

Hirschorn, M. 2002. Noise Control Reference Handbook. Sound & Vibration, Bay Village, OH.

Kelso, D. and A. Perez. 1983. Noise Control Terms Made Somewhat Easier. Minnesota Pollution Control Agency, St. Paul, MN.

U. S. Environmental Protection Agency. 1976. About Sound. Environmental Protection Agency, Washington, D. C.

INDEX

A

B

C

D

E

F

P

Q

R

S

T

U

V

W

X

Y

Z

Page intentionally left blank

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