Listen

Transforming Sound into Knowledge

Sequence Note

Open Loop Sequence

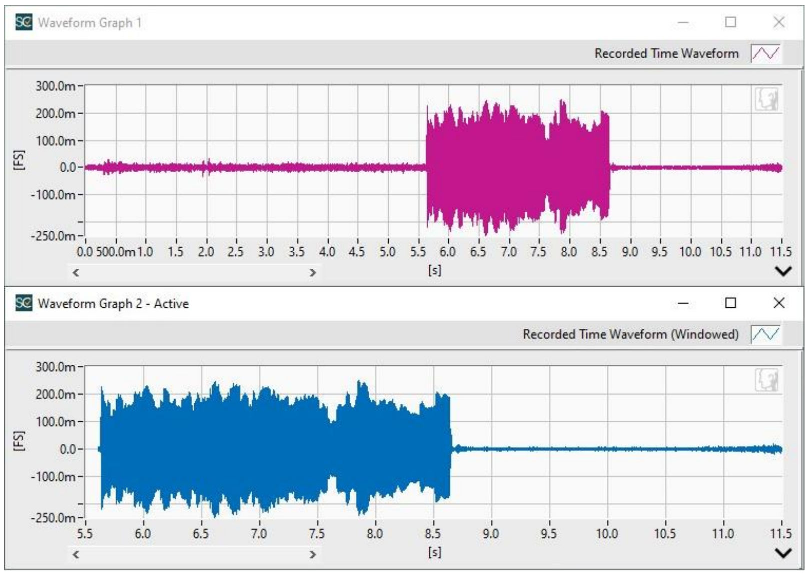
Introduction

This sequence demonstrates the two most common microphone measurements, frequency response and sensitivity, of a microphone embedded in a recording device. Typically, when measuring a microphone, the response of the device can be captured simultaneously with the stimulus. However, with devices such as voice recorders and wireless telephone forming a closed loop can be cumbersome or impossible. This sequence demonstrates how to measure such a device by recording the signal on the device under test, transferring that recording to the computer running SoundCheck and then using a Recall step to import the recorded waveform and analyze it.

This specific sequence, v3, is an improvement on the prior versions. The v1 release required that the audio file containing the recorded response waveform be manually windowed outside of SoundCheck before being analyzed. The v2 release utilized a new feature in SoundCheck 14, using values from the memory list to semi-automatically trim the waveform before analysis. This v3 release completely automates waveform editing.

Final Display for Open Loop Microphone Sequence

The sequence begins by prompting the operator to enter an intersection value. This value is used to determine the start time coordinate of the recorded waveform from the DUT. By default, it is set at 100ms FS and depending on the ambient noise in your test environment, you may have to edit this value to capture the true start time of the waveform. Next, the operator will be prompted to begin recording on the device under test and then a stepped sine sweep is generated from 10 kHz to 100 Hz through a source that has previously been calibrated to produce 1 Pascal across the sweep range. The operator is then prompted to transfer the recording to the computer and then to load the file into SoundCheck via a Recall sequence step. At this point, a series of post processing steps automatically window the waveform to an acceptable range for use in the remaining steps of the sequence.



DUT waveform before Windowing (Top) and after Windowing (Bottom)

The recorded signal is then sample rate converted and frequency shifted to match the stimulus and then analyzed with a HarmonicTrak analysis step, which then calculates the response curve. A post processing step is then used to extract the level at 1 kHz, the sensitivity value. Limits are set around both the frequency response and the sensitivity. The default limits values should be adapted to your particular device.

The final display shows two graphs. The top X-Y graph displays the data at its absolute level in dBFS/Pa (since the imported recording is digital, the results will be in FS or dBFS rather than Volts or dBV). The lower graph shows the windowed recorded signal analyzed by the software. In addition, the calculated sensitivity at 1kHz and the frequency response margin is also displayed.

Software Requirements

- SoundCheck 18 Basic

Hardware Requirements

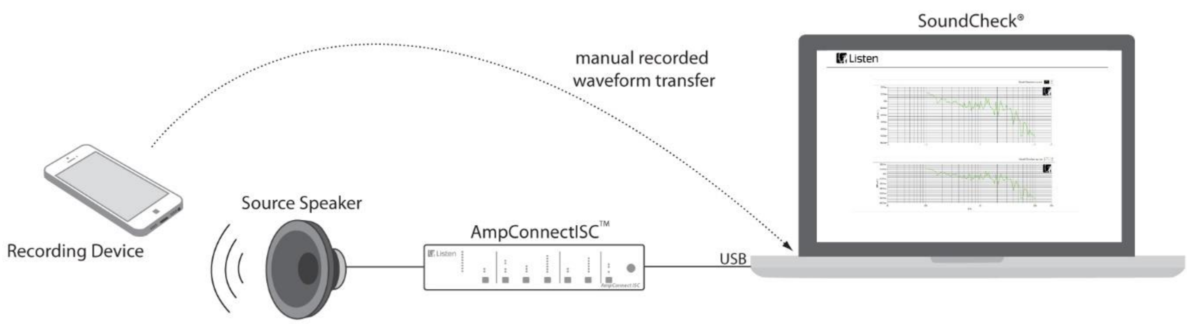
- Audio Interface - Listen AudioConnect, AmpConnect(integrated power amp) or similar
- Power Amplifier - Listen SCAmp, AmpConnect or similar

Hardware Setup & Calibration

1. Calibrate the source speaker as per the instructions in the SoundCheck manual.

You are ready to start the sequence

System diagram



Sequence Logic

Type	Step Name	#	Out	In	
Sti	10k-100Hz (R40)	1	Source Speaker		
Mes	Intersection Level	2			// Sets default value for Intersection Step 7
Mes	Operator Message - Begin Recording	3			// Prompt operator to begin recording.
Acq	Play Only	4	Source Speaker	Direct In 1	// Generate the stimulus
Mes	Operator Message - Transfer Recording to Computer	5			// Prompt operator to stop recording and copy the wav file.
Rec	Recall waveforms	6			// Import the wav file
Pos	Intersection	7			// Determines the waveform signal start time
Pos	Curve minus constant	8			
Pos	Calculate Waveform Stop Time	9			// Add the stimulus length in time to the start time to arrive at the stop time.
Pos	Windowing	10			// Window the response from the recalled .wav file.
Pos	Re-sample	11			// Convert recording to 44.1 kHz
Pos	Frequency Shift	12			// Align the stimulus and response waveforms
Ana	Fundamental	13			// Derive the frequency response
Pos	Curve Average	14			// Finds the sensitivity value at 1kHz
Lim	Microphone Sensitivity	15			// Arbitrary limits for the sensitivity value
Lim	Microphone Response	16			// Arbitrary limits for response (floating 20 dB window)
Dis	Microphone Response	17			

Notes:

1. Step 10, Post Processing-Re-sample, assumes that the stimulus was generated at 44.1 kHz sample rate. If your hardware is configured for a different sample rate you must edit this step to match the sample rate of your confirmation.
2. Step 2 – Intersection Level – This value (FS) is used to detect the start of the stimulus signal in the DUT waveform. Its default value is 100 mFS which should be fine for most quiet test environments. In less quiet environments, ambient noise in the recording may result in a false intersection values which can produce bad results. In such cases, the default value should be adapted to your actual test environment.