

SP395A

SoundPro
Audio Integrator
Operation Manual
(Firmware version 1.67)



SENCORE
3200 Sencore Drive, Sioux Falls, SD 57107

WARNING

PLEASE OBSERVE THESE SAFETY PRECAUTIONS

There is always a danger present when using electronic test equipment.

Unexpected voltages can be present at unusual locations in defective equipment and distribution systems. Become familiar with the equipment with which you are working, and observe the following safety precautions.

Every precaution has been taken in the design of the SoundPro SP395A Audio Integrator to insure that it is as safe as possible. However, safe operation depends on you, the operator.

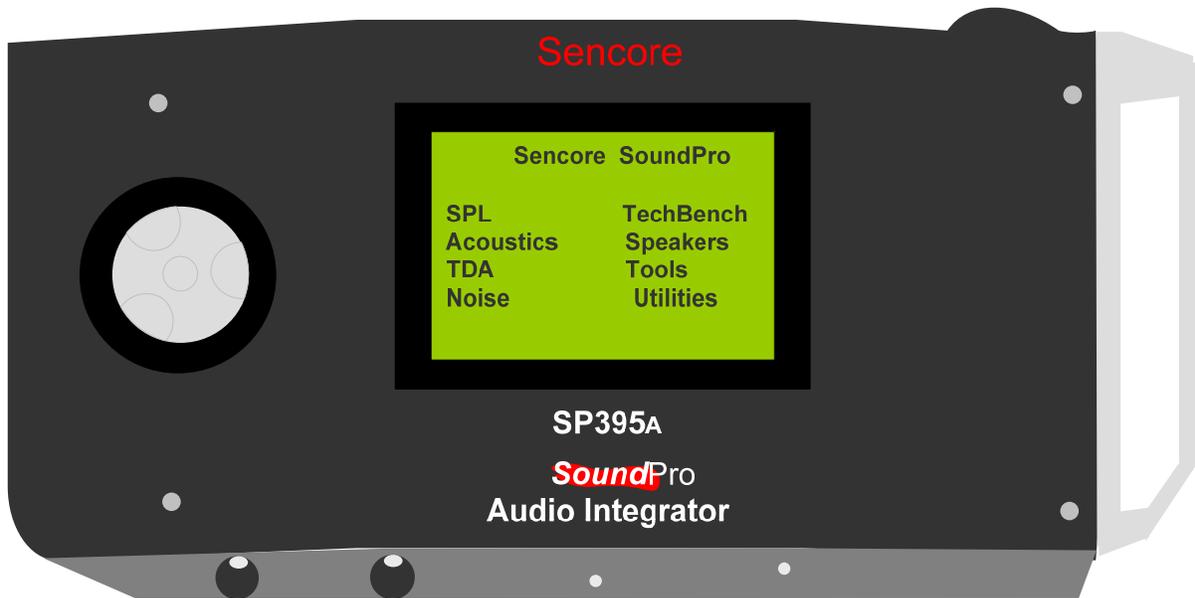
1. **Never exceed the limits of the SoundPro SP395A** as given in the specifications section or other special warnings provided in this manual.
2. **Always be sure that your equipment is in good working order.** Broken or frayed test leads or cables can be extremely dangerous and may expose you to high voltages.
3. **Remove test leads** immediately following measurements to reduce the possibility of shock.
4. **Do not work alone when working under hazardous conditions.** Always have another person available in case of an accident.
5. **Never assume that a cable shield is at earth ground potential.** Both static and electrical voltages can be present on a cable's sheath.
6. **Always follow standard safety procedures.**

When in doubt, be careful.

Sencore reserves the right to make changes in specifications and other information contained in this publication without prior notice, and the reader should in all cases consult Sencore to determine whether any such changes have been made. This manual may not be reproduced, and is intended for the exclusive use of Sencore users.

SP395A

SoundPro
Audio Integrator
Operation Manual
(Firmware version 1.67)



SENCORE

3200 Sencore Drive, Sioux Falls, SD 57107

TABLE OF CONTENTS

SAFETY PRECAUTIONS	inside front cover
DESCRIPTION	1
Introduction	1
SP395A Audio Integrator Features	1
Specifications	4
Front Panel Control/Descriptions	9
Bottom Panel – Input Description	9
Top Panel – Output Description	10
SP395A Audio Integrator Accessories	11
Firmware Add-on Test Functions	13
OPERATION	14
Power ON/OFF Switch	14
Power Input Jack	14
Battery	14
Installing/Removing Battery	15
Battery Charging & Battery Life	15
User Interface – Control Knob	16
Menu Navigation	16
Menu Overview	17
Inputs & Outputs	18
Audio Inputs	18
Audio Outputs	19
Audio Monitoring Output	19
Input Gain Control - Preamplifiers	20
Menu Toolbars	21
Top Tool Bar	21
Bottom Tool Bar	22
Headphone/Speaker Monitor Control	22
Memory Control	23
Main Output & Generator Control	23
SPL	24
Sound Level Meter	24
LEQ	26
Dosimeter	27
Sound Study Graph	29
Acoustics	31
Real Time Analyzer	31
FFT Analyzer	35
Energy Time Graph	38
Reverb Decay Time	43
Multi-Band Decay	47
TDA - Time Delay Analysis	50

Noise Tools	57
ALCONS	57
RaSTI	60
STI-PA.....	63
Noise Curves.....	65
Transmission Loss	67
Audio Stethoscope.....	69
Tech Bench	70
Level/Freq. Meter	71
Signal Generator	72
Amplitude Sweep	74
Distortion Meter	76
Phase Meter	80
Crosstalk Meter	81
Speakers	82
Speaker Polarity.....	82
Speaker Distortion.....	84
Impedance Meter	86
Impedance Sweep	87
Cable Tester.....	89
Tools	91
USB Preamp Display & Controls.....	91
USB Preamp Operation.....	92
USB Output Operation.....	93
Auto HT Tests.....	94
Auto Time Delay Test	102
Utility Function	104
Audio Scope	104
Clear Memories	106
Save Settings	106
Setup & Calibration	107
Setup – Unlock Firmware	107
Setup – General	108
SPL Calibration.....	109
PC Interface/About.....	110
Computer Interface	111
Uploading Memories to the Computer.....	111
Upgrading Firmware in SP395A.....	112
TerraLink Software: Real-Time Computer Interface.....	113
APPENDIX A - dB SPL Scale	114
APPENDIX B ANSI Weighting Curves	115
APPENDIX C ANSI Balanced Noise Criterion Curves	116
APPENDIX D Serial Protocol for Accessing SP395A Memories	117
WARRANTY AND SERVICE INFORMATION	inside back cover

DESCRIPTION

Introduction

The Sencore SP395A SoundPro Audio Integrator is a quality, high-end audio and acoustic analyzer designed for audio professionals, commercial audio installers, and audio system engineers. Its audio analyzing functions make it attractive for virtually anyone involved with audio acoustics or audio systems testing.

The Audio Integrator employs a powerful DSP engine. All functions, filters, and processing are implemented in DSP firmware algorithms. As each function is selected, loaded into internal memory and executed, the analyzer takes on a completely new personality. Over 25 different test functions become possible in one instrument.

The SP395A is especially good at multi-tasking. It employs independent input, DSP processing, output, monitor, and USB circuit

sections. For example, you can run the Sound Study Graph to record SPL over time, while monitoring the input in headphones, and simultaneously feeding the input signals to a PC through the USB audio interface.

The Integrator uses precision 24-bit A/D and D/A converters, and internal 64-bit processing. The input and output amplifier stages are fully balanced, low noise, high-resolution gain stages. The result is precision measurements for accurate audio/acoustic analysis.

SP395A Audio Integrator Features

The SP395A provides multiple audio analyzing features including:

- **Sound Level Meter or (SPL) Sound Pressure Level:** Measures loudness of ambient sound according to ANSI Type 1 A and C weighting networks and ANSI Class 1 octave and 1/3 octave band filters. Selectable slow, fast, impulse, or peak sound level measurement averaging modes.
- **Leq:** Measures linear averaged Sound Pressure Level over a specified period of time to determine if noise standards are being met.
- **Dosimeter:** Measures noise dosage level according to ANSI S1.25-1991 and ASA 98-1991 specifications for noise studies, machinery noise evaluations, and occupational noise analysis.
- **Sound Study Graph:** Analyzes and records SPL noise levels over a specified time period from 1 minute to 24 hours, and graphs the results. Automatically runs multiple sound studies storing up to 40 sound study tests.
- **Real Time Analyzer (RTA):** An audio spectrum analyzer which graphs the dB level of each audio octave or 1/3 octave band. Fast DSP audio analyzing can be used for live music or sound analysis. An ANSI X curve weighting network for motion picture theater setup is included. RTA graphs can be stored to memory or compared with the “difference” mode for before and after analysis.
- **FFT Analyzer:** Digital signal processing with three FFTs that graph the dB sound level with 1/6 or 1/12 octave band resolution to 20 kHz. RTA graphs can be stored to memory or compared with the “difference” mode for before and after analysis.
- **Energy Time Graph:** Graphs the arrival of sound energy to the microphone input and/or shows the decay of sound energy in dB over time. Starts with sound initiated from the

SP395A generator or an external sound pickup such as a handclap. Valuable in setting time delays from speaker to listening positions and analyzing sound decay for reflections or decay time.

- **Reverb Decay Time (RT60):** Measures and computes reverb decay time or the time it takes the sound in a room to decrease 60 dB. The measurement computes wideband RT60 (20Hz-20 kHz) or the RT60 of any octave band.
- **Multi-Band Decay:** Measures, computes and displays the reverb decay time RT60 for seven audio octave bands simultaneously to identify differences in room sound absorption characteristics in each band.
- **Time Delay Analysis:** Times a swept audio generator (20-20,000Hz) and tracking audio 1/12 octave band level meter, with a variable filter Q of 6 to 150, graphing the detected level. Measurement eliminates room reverberations or echoes for accurate frequency response testing of speakers or microphones.
- **Speech Intelligibility:** Measures speech intelligibility using the most popular industry standards including ALCONS, RaSTI and STI-PA. A CD with the proper test signal for these tests is provided.
- **Noise Curve:** Measures noise in a room and characterizes it to a numeric value taking into account how the human ear hears. Performs standard NC or NCB (balanced noise curves), RC (Room Criteria), and PNC (preferred noise curve) ANSI tests.
- **Transmission Loss:** Measures sound transmission loss through a partition. Computes STC, RW and OITC standard transmission loss measurements by comparing 1/3 octave RTAs from the source and received sound.
- **Audio Stethoscope:** Combines the microphone input and sensitive audio preamplifier with the audio monitoring headphones to form an audio stethoscope. Weighted filters and octave or 1/3 octave band filters may be selected.
- **Level/Freq. Meter:** Measures two channel left and right audio input levels from -120 dBu to +40 dBu and sine wave frequency from 10 Hz to 32 kHz. Measures dBu, dBV, Vave, Vrms, Vp-p and dBr.
- **Signal Generator:** Generates audio test signals including sine wave to 22 kHz, square wave, pink noise and white noise with variable output level. Also outputs octave or 1/3 octave band limited pink noise.
- **Amplitude Sweep:** Plots the frequency response of audio equipment by sweeping the generator through its frequency range and graphing the amplitude input level of each 1/12 octave.
- **Distortion Meter THD:** Measures THD+N (Total Harmonic Distortion plus Noise) of an incoming signal by comparing the level of the incoming signal with the level of harmonics and noise.
- **Distortion Meter IMD:** Measures IMD (Inter-modulation Distortion) of an incoming signal by outputting a multi-frequency output test tone and measuring the distortion components. Performs SMPTE IMD and DIN IMD tests.
- **Phase Meter:** Measures the phase difference between left and right input sine waves of the same frequency.
- **Crosstalk Meter:** Measures the amount of dB isolation between the left and right input audio sources or the signal bleeding from one channel into the other.
- **Speaker Polarity:** Determines the polarity of the speakers to determine proper wiring.
- **Speaker Distortion:** Tests speaker distortion, using the THD+N test function, to identify speaker defects or performance problems.
- **Impedance Meter:** Measures the impedance of an audio device or component such as a speaker. The frequency of the test is variable. Also indicates watts or amplification required at a particular voltage.

- **Impedance Sweep:** Plots the impedance of a load by sweeping the generator through its frequency range and graphing the impedance at each 1/12 octave step.
- **Cable Tester:** Provides a cable attenuation test at 1 kHz and 20 kHz, and a polarity test to identify defective or improperly wired cables.
- **USB Preamp Control:** Serves as a calibrated USB microphone preamp. Use as a sound card for a PC based analysis program or to route audio from a PC to the Audio Integrator main outputs.
- **Auto HT Tests:** Provides an automatic sequence of acoustic tests of a home theater by remotely controlling the Sencore DAG5161 generator and capturing measurements to memory.
- **Auto Time Delay Test:** Provides a time delay test from all speakers in a home theater system when using a special audio test selection on the Sencore DAG5161 generator.
- **Audio Scope:** Provides a voltage vs. time graph of the audio waveform to look for distortion or clipping.

Specifications

SoundCore™ Module

Sound Level Meter (SLM): Measures ambient sound energy levels at microphone input
Displays true RMS SPL

Averaging: Meets ANSI S1.4
Slow
Fast
Impulse
Peak
Avg – Equal time-averaged SPL

Sound Study Graph (SSG): 1 minute to 24 hour continuous recorded SPL measurement
“Strip Chart Recorder”
Peak or Avg mode
Auto-Saves to internal memory

SPL/SSG Measurements

Weighting networks: ANSI Type 1 A, C, Flat, octave, or 1/3-octave band filtered
Level Range: 25-140 dBA with MIC-TP2 mic
Level Accuracy: ± 0.1 dB at 1 kHz with MIC-TP2 mic
Level Resolution: 0.1 dB
Frequency Range: 20 Hz to 22 kHz

Real-Time Analyzer (RTA): Plots sound energy levels vs. octave band filtered spectrum
ANSI-compliant 1/3 octave digital filter-based
FIR Filters: Full octave, 1/3 octave frequency resolution.
Weighting networks: Flat, A, C, X-curve, difference mode.

FFT Analyzer (FFT): Plots sound energy levels vs. octave band filtered spectrum
ANSI-compliant 1/12 octave resolution
Three 1024-point cascaded FFTs
Weighting networks: Flat, A, C, difference mode.

RTA, FFT Analyzer

Display range: 30 dB or 15 dB window within 0 dB to 165 dB range, adjustable in 5 dB steps
Level range: 25-140 dBA with MIC-TP2 mic
Level accuracy: ± 0.1 dB at 1 kHz with MIC-TP2 mic
Level resolution: ± 0.1 dB
Frequency range: 20 Hz – 22 kHz
Frequency accuracy: ± 0.5 dB 10 Hz to 22 kHz, ± 0.2 dB 100 Hz to 16 kHz
Averaging: 1 sec, 3, sec, 6 sec, 10 sec, 30 sec, 60 sec, equal time-weighted, peak hold

Energy-Time Graph (ETG):	Graphs impulse sound decay energy vs. time or distance Identifies arrival time of direct sound and audio reflections Displayed in mS (milliseconds), ft (feet), m (meters)
Decay time ranges:	From 0-15 mS to 0-7680 mS
Accuracy:	Calculates RT10-RT60 from cursor points on decay curve Better than $\pm 1\%$ of full scale
Reverb Decay Time:	Calculates time for residual pink noise test signal to decay 60 dB in room Also RT10, RT20
Range:	0-15 sec
Accuracy:	$\pm 2\%$ of full signal
Resolution:	1 mSec
Weighting networks:	A, C, Flat, octave, or 1/3-octave band filtered.
Ambient Noise:	Ambient noise level must be at least 26 dB lower than test signal
RT60 Time:	Extrapolated from Decay Time measurement.
Impedance Meter:	
Accuracy:	1 ohm to 50 ohms, $\pm 2\%$, 50 Hz – 12.5 kHz 1 ohm to 8 kohms, $\pm 10\%$, 20 Hz – 20 kHz
Speaker Polarity:	Tests absolute speaker phase (polarity) by analyzing a proprietary waveform Shows polarity of mid-frequency wavefront
Range:	0-100 ft, speaker to SoundPro
Cable Tester:	Tests XLR, 1/4", and RCA audio cables, balanced and unbalanced Performs analog transmission test to test transmission loss at 1 kHz and 20 kHz
Resolution:	0.1 dB Performs polarity test
Signal Generator:	
Signals:	Sine, square, gaussian white noise, pseudo-random pink noise, band-filtered pink noise
Frequency Range:	
Sine:	1 Hz to 22 kHz; fine, 1/3 octave, or one octave steps
Square:	1 Hz to 6 kHz; fine, 1/3 octave, or one octave steps
Noise bandwidth:	Full (20 Hz to 20 kHz), octave, or 1/3-octave band filtered
Frequency Accuracy:	50 ppm; 0.005%
Level range:	-35 dBu to 17 dBu
Maximum Sine Output Level:	Balanced: +16 dBu Unbalanced: +10 dBu
Level Accuracy:	± 0.1 dBu at 1 kHz, ± 0.2 dBu, 16 Hz to 20 kHz
Typical:	± 0.05 dBu at 1 kHz, ± 0.1 dBu, 16 Hz to 20 kHz
Sine distortion:	< 0.01% THD at full-scale output
Audio Monitor:	Source independent channel for monitoring Software volume control Headphone or built-in internal speaker output

TechBench™ Module

Level Meter:	Stereo level meter reads in dBu, dBv, Vp-p, Vrms, or dB (reference)
Frequency Range:	10 Hz to 22 kHz
Auto-level ranging:	
Level Range:	-120 dBu to +40 dBu
Level Accuracy (-85 dBu to +40 dBu):	±0.2 dB, 10 Hz to 22 kHz
Typical:	±0.01 dB, 100 Hz to 16 kHz; ±0.05 dB, 10 Hz to 22 kHz
Level Resolution:	0.01 dBu
Frequency Counter:	
Frequency Range:	16 Hz to 40 kHz
Accuracy:	±5%
Resolution:	1 Hz
Amplitude Sweep:	Graphical display of 1/3 and 1/12 frequency response User adjustable frequency range
Impedance Sweep:	Graphical display of 1/3 and 1/12 octave impedance values User adjustable frequency range
Distortion Meter:	THD+N – Computes Total Harmonic Distortion plus Noise Compares level of output harmonics and noise to single test frequency Test Frequencies: 63 Hz, 125 Hz, 250 Hz, 500 Hz, 1 kHz, 2 kHz, 4 kHz IMD – Computes Inter-modulation Distortion Compares levels of output sum and difference frequencies to dual test frequencies Test Types: SMPTE 60/7kHz, DIN 250/8kHz. Inputs: Speaker, microphone, line level. Range: 0.02% to 50% THD+N Accuracy: ±5% of reading
Phase Meter:	Shows the phase difference in degrees between 2 sine waves
Crosstalk Meter:	Displays the amount of isolation that exists between two audio sources Displayed in dB
Audio Scope:	Shows audio signal on an oscilloscope-like screen Dual-trace, auto-triggering AC-only input coupling (down to approx. 3 Hz) X-Y mode for phase display Sample rate: Hz Accuracy: ±1 display pixel

Speech Intelligibility Module: (RaSTI and STI-PA test waveforms included on CD)

ALCONS:	Computes Percent Apparent Loss of Consonants
RaSTI:	Calculates Rapid Speech Transmission Index
STI-PA:	Computes Speech Transmission Index using STI-PA method (STI for Public Address)

Time-Delay Analysis Module: (TerraLink software included)

Time-Delay Analysis:	Measures frequency response using log-swept sine wave.
-----------------------------	--

Noise Curves Module

NC Curves:	Displays Noise Criteria data, based on ANSI standard S12.2-1995 (ASA 115-1995) Displays limiting band, SPL by octave band
RC Curves:	Displays Room Criteria rating
PNC Curves:	Displays Preferred Noise Criteria rating
STC/NIC Measurement:	1/24th octave Sound Transmission Class/Noise Isolation Class measurement

Multi-Band Decay Module

Multi-Band Decay:	Displays seven octave-band filtered RT60 measurements simultaneously
--------------------------	---

Audio Stethoscope Module

Audio Stethoscope:	Filters incoming left input audio through to the headphone or speaker
Filters:	Flat (no filter), A or C ANSI Weighted, octave, 1/3 octave filters

Audio Hardware

Sample Rate:	48 kHz
Input Gain Ranges:	-50 dB to +50 dB
Maximum Rated Input:	+40 dBu

Preamplifier

Distortion:	-90 dB THD+N
Phantom Power output:	48 VDC \pm 1 VDC

ADC/DAC

Cirrus Logic:	CS4224 48 kHz 24-bit 105 dB Audio CODEC
----------------------	---

USB Preamplifier

Burr-Brown Texas Instruments® USB Audio CODEC
USB 1.1 full-speed
USB isochronous data format

Standards

Sound Level Meter: ANSI S1.4-1983 (ASA 47-1983)

Leq: ANSI S1.43-1997

Dosimeter: ANSI S1.25-99 (ASA 98-1991) Class 2

Digital Filters: ANSI S1.11-1986 1/1, 1/3 octave digital filters

All filters, weighting networks, response rates, and algorithms meet or exceed ANSI Type 1 specifications

General/Environmental

Input Impedances

XLR and balanced inputs: 40 kohms nominal

RCA unbalanced inputs: 20 kohms nominal

Output Impedances

XLR and balanced outputs: 300 ohms nominal

RCA unbalanced outputs: 150 ohms nominal

Non-Volatile Memory: 40 graph data storage locations

Display: 64x128 pixel blue LCD with backlight

Size: 9 1/2" x 5 1/2" x 2 1/4" (HWD)

Weight: 2 lbs. 5 oz.

Temperature Range: 5°C to 45°C, non-condensing

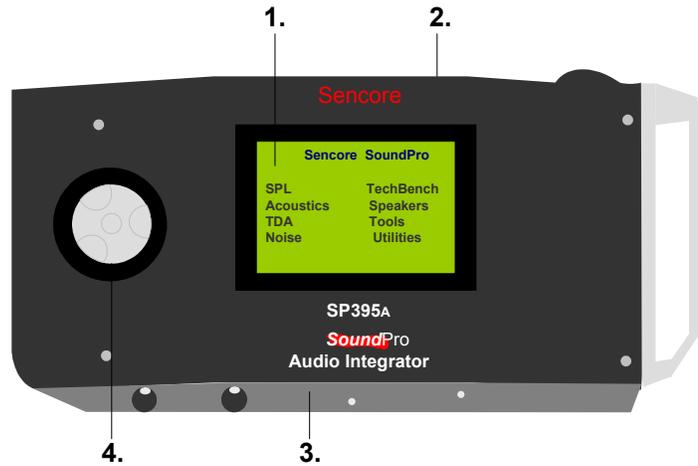
Power

Battery: Li-Ion pack provides over 5 hours of service, depending on usage

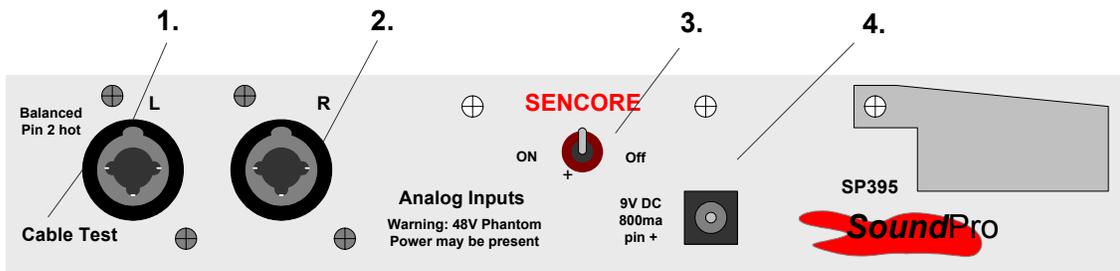
AC adapter/charger: 120/130/230/240 VAC, 9-12VDC, 800 mA (Nominal 9V)

Front Panel Control/Descriptions

- Display Screen:** Provides a visual interface to the user of all instrument functions and test results.
- Bottom Panel (when viewing display) - Input Panel:** Contains power input jack, audio input jacks, and on/off switch. See Front Side Panel Description section for details.
- Top Panel (when viewing display) - Output Panel:** Contains Computer, USB and analog audio jacks and speaker. See Left Side Panel Description section for details.
- Control Knob:** Provides navigation through instrument menus. All test operations and user selections are performed using this control. This control is both rotated and pressed. Rotate the control to move the screen cursor or increment through menu selections. Momentarily pressing inward (clicking the control knob) makes a selection or change.

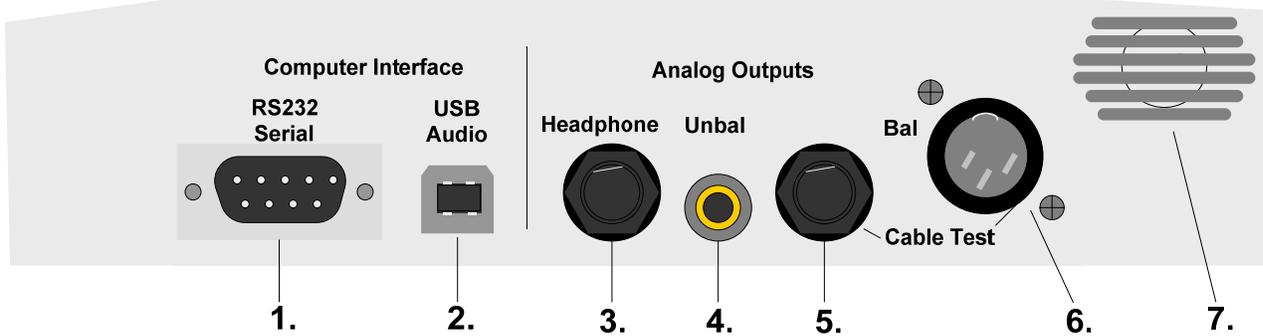


Bottom Panel – Input Description



- Left Analog Input:** Combination XLR and ¼ inch Balanced Microphone /Line input. Used as main input for most tests. May have 48 Volt Phantom Power switched to jack by instrument. Used as Cable Test Input.
- Right Analog Input:** Combination XLR and ¼ inch Balanced Microphone /Line input. May have 48 Volt Phantom Power switched to jack by instrument.
- Power Switch:** Turns the instrument power on or off. Down is Off.
- Power Adapter Input:** Powers the instrument and charges the internal rechargeable battery. Use a Sencore approved adapter nominal 9V @ min 800 mA. (center pin positive)

Top Panel – Output Description



1. **RS232 Computer Interface Connector:** Serial DE-9 connector. Interfaces instrument to computer for use with automation software and for the purpose of updating unit firmware to factory changes.
2. **USB Audio Connector:** Provides USB connection for digital audio signals.
3. **Stereo Headphone (Hdpn) Output Jack:** ¼ inch headphone jack. Provides stereo audio monitoring with headphone.
4. **RCA Phono Audio Output Jack:** Provides unbalanced audio output through female RCA phono jack.
5. **Balanced mono Audio Output Jack:** Provides balanced audio output through a ¼ inch headphone jack (TRS).
6. **Balanced XLR Audio output:** XLR jack male, Provides a balanced mono audio output. Jack also used for Cable Test output.
7. **Internal Speaker:** Provides audio output to monitor input audio to instrument or audio generated and output by instrument.

SP395A Audio Integrator Accessories		
Sencore Part #s	Description	Photo
133G527	<p>TerraLink Viewer Software / Speech Intelligibility Audio CD provides TerraLink – viewer only capability – along with the audio CD files needed for the STIPA and RASTI speech intelligibility tests.</p> <p>supplied</p>	
13G122	<p>XLR Audio Cable – a microphone cable for interfacing balanced audio inputs and outputs to the SP395A.</p> <p>supplied</p>	
13G123	<p>USB Cable – a 6' connection cable (type A to type B), connects the unit's USB connector to a USB port on a computer PC USB port.</p> <p>supplied</p>	
13G124	<p>Serial Interface Adapter Cable – a 6' computer serial interface adapter cable (straight through), connects between the unit's RS232 jack and a computer serial port.</p> <p>supplied</p>	
17G65	<p>Lithium Ion Battery – install battery to power the instrument. The battery is removable, supplies up to 5 hours of continuous power per charge, depending on use.</p> <p>supplied</p>	
39G1156	<p>Standard Measurement Microphone meets or exceeds ANSI Type 2 specifications. Microphone Holder Clip mounts on a standard microphone stand and holds the test microphone.</p> <p>Both are supplied.</p>	

<p>39G1171</p>	<p>Impedance Test Cable 1/4 inch mono plug to clip connections – provides a test lead connection when using the Impedance Meter and Impedance Sweep test functions. supplied</p>				
<p>60G80</p>	<p>AC Power Adapter/ Charger plugs into the SP395A to charge the battery or to provide AC operation. supplied</p>				
<p>CC309</p>	<p>Soft Carry Case padded carrying case to transport unit; includes extra storage capacity for supplied accessories. supplied</p>				
<p>SP-PMIC1 optional</p>	<p>Precision Low Noise Microphone – meets or exceeds ANSI Type 1 specifications optional</p>				
<p>SPSW-TL optional</p>	<p>Software – provides a real-time PC connection/interface with the instrument. Plots memories and displays FFT, RTA, SSG and SLM Meter results in real-time. (Requires purchase and authorization of an UNLOCK code from Sencore) <i>Included as part of the Time Delay Analysis (TDA) Firmware Add-on Module.</i> optional</p>	<p>Terra Link Software Unlock Code</p>			
<table border="1" style="width:100%; border-collapse: collapse;"> <tr> <td style="width:33%; padding: 2px;">http://www.sencore.com</td> <td style="width:33%; padding: 2px;">mailto:sales@sencore.com</td> <td style="width:33%; padding: 2px;">1.800.736.2673</td> </tr> </table>			http://www.sencore.com	mailto:sales@sencore.com	1.800.736.2673
http://www.sencore.com	mailto:sales@sencore.com	1.800.736.2673			

Firmware Add-on Test Functions

The Audio Integrator comes standard with a solid core of audio analyzing tests. These tests are fundamental to most audio and acoustic analysis work. The core tests are referred to as the SP395A's "SoundCore" measurements.

The Audio Integrator contains, in its internal operating software or firmware, instructions for performing many more audio and acoustic tests than provided by the standard "SoundCore" measurements. These additional tests may be purchased based upon individual user and company requirements. The additional available tests are divided into packages or "Firmware Modules" for easy implementation or purchase.

Firmware Module	Function	Menu
TechBench	Amplitude Sweep	TechBench
TechBench	Distortion Meter	TechBench
TechBench	Phase Meter	TechBench
TechBench	Crosstalk	TechBench
TechBench	Audio Scope	Utilities
TechBench	Impedance Sweep	Speaker
Noise Curves	NCB, RC, PNC	Noise Tools
Noise Curves	Transmission Loss (RW, STC, OITC)	Noise Tools
Speech Intelligibility	ALCONS, RASTI, STI-PA	Noise Tools
Time Delay Analysis (includes TerraLink)	TDA	TDA
Audio Stethoscope	Audio Stethoscope	Noise Tools
Multi-Band Decay RT60	Multi-Band Decay	Acoustics

The optional functions are available for use upon purchasing the firmware module and entering the supplied unlock code. The functions contained within a single firmware module may appear on more than one menu. The table lists the firmware modules and the functions they contain, along with the Audio Integrator menu that the function appears in.

To purchase an unlock code for a firmware module, contact Sencore. For instructions on how to unlock a firmware module, see the Setup & Calibration section of this manual.

Specifications, accessories and software add-on test functions are subject to change without notice.

OPERATION

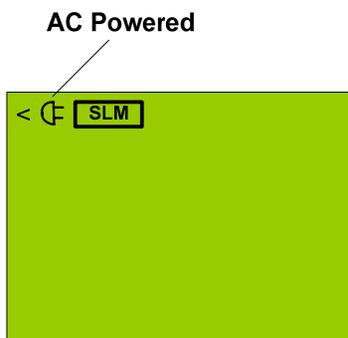
Power On/Off Switch

The power switch is mounted at the center of the bottom panel. The power switch turns the Audio Integrator on or off, whether the unit is powered by the internal battery or the AC adapter. When you turn on the power switch, the unit begins initialization. During this time software is loaded from memory locations within the Audio Integrator including updates. The unit is ready to use as soon as the main menu screen appears, no warm-up time is needed. The Audio Integrator can be powered off at any time – there is no required shutdown procedure.

Power Input Jack

The Power Input Jack is located close to the power switch on the bottom panel. The DC Input Jack requires the input of a DC voltage to power the instrument and charge the internal battery. The SP395A requires a 9-12v DC power adapter, regulated or unregulated, rated at a minimum of 800ma. The input connector is a 2.1mm coaxial power connector with the positive polarity voltage on the center pin of the connector.

An AC power adapter/charger is supplied to power the Audio Integrator and charge the internal battery. Use only the Sencore-supplied power adapter, if possible. Using an improper AC adapter can damage the unit and/or improperly charge the battery. Using an improper AC power adapter/charger voids the unit warranty.

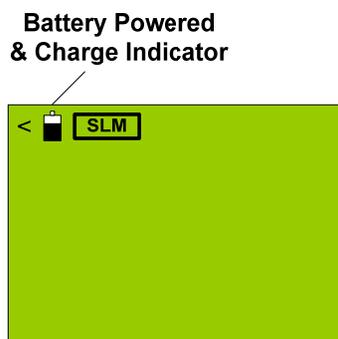


The AC plug symbol in the display screen beside the < (back) field indicates the unit is powered from the AC power adapter/charger.

When the AC power adapter is plugged in, the battery level indicator shows an AC power plug symbol. When the AC power adapter is removed, the indicator shows a battery symbol with a relative battery charge level.

Battery

A lithium-ion rechargeable battery is supplied with the Audio Integrator. The battery is charged whenever the AC adapter is plugged in, or can be charged using an external charger.



Battery operation is indicated with a battery symbol near the < (back) field. The symbol indicates the approximate charge level of the battery.

Battery operation is indicated in the Audio Integrator screens with a battery symbol located in the upper left of the screen adjacent to the < (back field). The symbol also indicates the relative charge of the battery. If the battery symbol is filled to the top, the battery is at or near full charge. A battery symbol that shows little or no fill indicates that the battery is nearing a discharge level in which it must be recharged.

Installing / Removing Battery

The battery can be removed for replacement. The battery is installed in a small compartment in the bottom of the Audio Integrator. To remove the battery you must open the battery door or cover plate. Unscrew the thumb screw holding the plate to the instrument case. The door contains a cutout so if you push the door away from the screw the door can be hinged open. Use the nylon strap that surrounds the battery to lift it up and out of the compartment. The battery terminals plug into the small circuit board lining the top side of the battery. Gently pull the circuit board away from the battery to disconnect the battery from the circuit board.

To install a battery, reverse the removal process. The battery is polarized and needs to be installed correctly. Line up the flat side of the battery with the flat side of the circuit board. Push the board male connector pins gently and straight into the battery terminals. Seat the battery into the instrument, close the battery door and secure it by tightening the retainer screw.



The battery may be removed from the rear battery compartment.

Warning: Observe proper polarity. Do not plug the battery into the connector board the wrong way, or the battery may be damaged.

Battery Charging & Battery Life

The battery is charged anytime that the AC adapter is plugged into the Integrator. It charges at the same rate, whether the unit is turned on or not. You can also charge the battery in an external lithium-ion battery charger. You cannot overcharge the battery, as long as you charge it in the instrument or use an approved external charger. The battery requires about 6 hours to fully charge.

Charger	Charge Time
Charge in Integrator	6.0 hours

The Audio Integrator uses different amounts of battery power, depending on what functions are being used. The phantom power uses a significant amount of power. Also, the headphone monitor amplifier consumes power when it is enabled. Expect 4-5 hours of run time on a full charge.

	Battery Life
Maximum run time	5.5 hours
Phantom power on	4.5 hours

Warning: Storing the unit with the battery in a state of discharge for long periods of time can shorten the life of the battery, or possibly destroy it. We recommend that the battery be kept charged at all times. The battery cannot be overcharged, so it is fine to leave the instrument plugged in, or to leave the battery plugged into an external charger when not in use.

User Interface – Control Knob

All the Audio Integrator operations and user selections are performed using the front panel control knob. This control is either rotated or momentarily pressed (clicking) the control knob. Turning the control knob moves a display highlight (white box with dark text) between menu options and fields on the screen. Clicking a highlighted choice selects it. To exit from a function and return to the previous menu or screen, highlight and click on the “<” symbol located in the upper left corner of the display.

To change the value of a selected field, click the control knob. The highlighted field will either toggle to a new value (fields with only a few possible values) or will change to an underlined “control lock” (data with many possible values).

To change the value of a “control-lock” field, click the control knob to “lock” (underline) the field and then turn the knob to change the value. When the desired value is shown, click the control knob again to unlock the control highlight. Turning the knob then moves you to the next field.



All user selections are made using the control knob either by rotating it to select a menu field or by momentarily pressing or “clicking” the knob to increment values or lock to a field.

Menu Navigation

The main menu with eight main function categories appears shortly after the Audio Integrator is turned on. By default the first item or category is highlighted. Turn the control knob to highlight the different menu items and click the control knob to select the item or category. A sub-menu appears with fields that list test functions that can be selected for that category. To select a function in the sub-menu, turn the control knob to highlight the desired field or function and click the control knob (press down momentarily). To exit from a sub-menu back to the main menu move the highlight to the upper left “<” field and click the control knob.



Click the < field located in the upper left corner of every display to return to the previous menu.

Menu Overview

All user selections to configure the Audio Integrator are performed with indications in the display screen. The Audio Integrator includes a main menu with eight categories of tests in which to choose from. Choosing a category in the main screen results in a submenu. The submenu lists all the tests available in that category.

Note: Some of the tests shown in the submenu screens may not be included in your Audio Integrator. These test features may be purchased as Software Additions to your instrument.



- 1. Main Menu:** Opening menu screen listing eight categories of Audio Integrator tests.
- 2. SPL Submenu:** Appears when the SPL selection is made in the main menu. This submenu lists the sound level measurements available for selection using the Audio Integrator.
- 3. Acoustics Submenu:** Appears when the Acoustics selection is made in the main menu. This submenu lists the acoustic analyzing tests available for selection using the Audio Integrator. *Note: The Multi-Band Decay tests is available as a firmware option. It appears in submenu but cannot be implemented without purchasing an unlock code.*

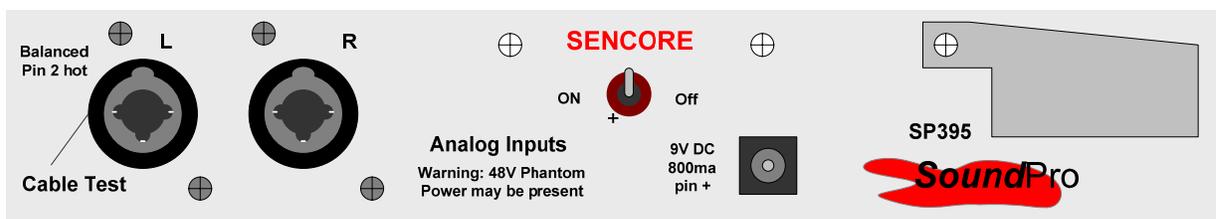
4. **TDA Submenu:** Appears when the TDA selection is made in the main menu. This submenu shows the TDA test functions available on the Audio Integrator.
Note: The TDA tests are optional firmware add-on test functions.
5. **Noise Tools Submenu:** Appears when the Noise selection is made in the main menu. Provides noise related measurements using the Audio Integrator. *Note: The ALCONS, RaSTI, and STI-PA are speak intelligibility tests available as part of the Speech Intelligibility firmware option. The Audio Stethoscope is available as a Stethoscope firmware option.*
6. **TechBench Submenu:** Appears when the TechBench selection is made in the main menu. Provides selection of audio bench testing functions. *Note: The Amplitude Sweep, Impedance Sweep, Distortion Meter, Phase Meter, and Crosstalk Meter are part of the TechBench firmware option.*
7. **Speakers Submenu:** Appears when the Speaker selection is made in the main menu. The submenu provides selection of speaker related tests.
8. **Tools Submenu:** Appears when the Tools selection is made in the main menu. This submenu provides a selection for USB Preamp Control. It further provides automated home theater (HT) testing and automatic time delay measurements when used with the Sencore DAG5161 generator.
9. **Utilities Submenu:** Appears when the Utilities selection is made in the main menu. The Utilities Functions submenu provides user options to save settings, calibrate and interface the Audio Integrator. *Note: The Audio Scope function is part of the TechBench firmware add-on option.*

All test functions available for the Audio Integrator are listed in the submenus. This includes all functions which are part of the standard or “Sound core” tests and test functions which can be added with firmware add-on purchases. If you select a test in the submenu that has not been added with a firmware purchase, the instruments indicates an “unlock required” message.

Inputs & Outputs

Audio Inputs

The Audio Integrator receives audio signal inputs from a microphone or line input. The input signal jacks are located on the bottom of the instrument. There are two input jacks which are labeled left (L) and right (R). The input jacks are a combination connector which accepts a balanced XLR connector male input and a ¼ inch stereo mono headphone male connector. This provides compatibility with most audio microphone cables and line input cables.



Left side input signal connectors are a combination XLR and ¼ inch mono headphone jack

The left (L) input should be considered as the main channel input when using the Audio Integrator. Some test functions, such as the cable test and audio stethoscope, only use one input. In these cases you always use the left (L) input connector when performing the test. Be aware that the different test functions of the Audio Integrator are setup for a mic. level input and others for a line level input. In each case, the input signal is taken from the left and right input jacks. For the acoustics test functions, a microphone is normally connected, and for the TechBench tests, a line level (or lower) electrical audio signal is anticipated by the SP395A.

The acoustics functions, which expect input from a microphone, use the microphone SPL calibration that is stored in the Utilities Calibration section. Electrical functions, which expect their input from a line level source, use the dBu calibration. Keep this in mind to prevent potential confusion over measurement results. For example, if a 0 dBu electrical signal is plugged into the Sound Level Meter, you may see results like +150 dB SPL, depending on the sensitivity calibration that is stored. Conversely, if you are using a microphone with the Amplitude Sweep function, you will see very low results, perhaps in the -70 to -60 dBu range.

Audio Outputs

The Audio Integrator provides 3 output jacks for connecting analog audio output to sound systems. You may use either the XLR, 1/4" TRS, or RCA phono jack. These jacks are located on the top panel of the instrument. All carry the same signal. The output range of the Audio Integrator is approximately -72 dBu to +17 dBu.



The left side panel contains three analog audio outputs including an XLR, 1/4 TRS headphone mono, and unbalanced RCA phono jack.

Use the XLR male output jack when a balanced XLR audio output is needed. The 1/4 inch TRS headphone jack output is available when this connection is desired. This connector is wired in parallel with the XLR output jack. It provides a TRS (tip, ring, sleeve) balanced audio output. One side of the balanced output is routed to the RCA phono to form an unbalanced output. Use this jack when you desire connection to an RCA connection cable.

Audio Monitoring Output

The Audio Integrator provides a headphone output jack for monitoring output audio. It is a stereo output headphone jack with the purpose of simultaneously monitoring both a right and left audio channel. The stereo headphone jack powers 32 ohm or greater headphones. The headphone jack can also serve as a line level output if needed.

A speaker is built into the Audio Integrator for the purpose of monitoring the output audio. The speaker is located on the left side panel of the instrument. The internal speaker outputs the mixed left and right outputs taken from the headphone output.

The audio output from the headphone jack and speaker is routed through separate output monitoring circuits within the Integrator. Audio to the headphone or speaker is independently switched on or off in the monitor part of the menu screen. The audio level output to the headphone and speaker is controlled in the monitor part of the menu screen. Internal generated audio or audio to the input jacks of the Audio Integrator may be routed to the monitor circuits. Right, left, stereo, mixed (mono) audio may be selected. See the Bottom Toolbar section of this manual.

Input Gain Control - Preamplifiers

The left and right audio inputs of the Audio Integrator feed into separate audio preamplifiers and gain control circuits. These circuits provide audio measurement over a wide range of audio levels and provide great sensitivity. There are thirteen gain ranges that are available to set the level of both the left and right input audio signal. The gain ranges of the left and right inputs are controlled independently in the top toolbar section of the test menu. See the Top Tool Bar Section of this manual.

The selected input level or gain control setting establishes an overall amplification of the input signal and best signal-to-noise ratio. The gain setting must be properly chosen for proper testing results. You should select the range which provides the most amplification without overdriving the sensitive input pre-amplifiers. The table below defines the clip points of each gain setting, both in dBu and related for available standard microphones.

Clip Points Table

Gain Setting	Line Input (dBu)	MIC-TP2 (dB SPL)	MIC-TP1 (dB SPL)
58	-45		
52	-39		
46	-34		
40	-29	110	80
34	-23		
28	-16		
20	-9	130	100
10	1	140	110
-10	15	160*	130
-20	25	170*	140
-30	35	180*	150*
-40	45	190*	160*

* Beyond microphone maximum level

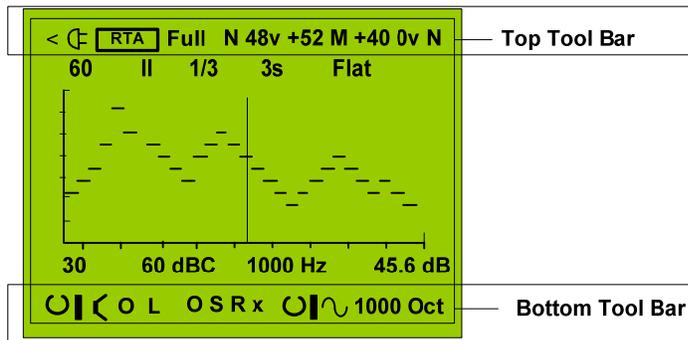
Note: To get the lowest SPL or noise levels you should use the maximum input gain setting possible without clipping. In many cases this will be 42-58. At this setting you will have the most sensitivity, but the input may clip at normal room SPL levels, depending on the sensitivity of the microphone.

The Audio Integrator provides both manual and automatic range selection for the left and right audio channels. The auto-gain function selects an input range for best measurement. If the input level gets within 2dB of clipping, the gain is set to the next least sensitive gain (if possible). If the input level is below that level at which the most accurate measurements can be made, the gain is increased in steps until the maximum input gain is reached.

The Audio Integrator provides an indication when the input level is approaching or overdriving the input preamplifier. The range number in the top tool bar of the menu blinks. When you see this select a higher gain setting in the top toolbar for that channel.

Menu Toolbars

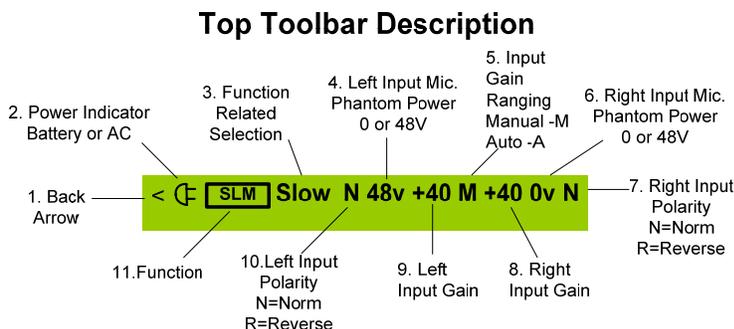
The Audio Integrator uses “toolbars” to provide access to often-used features and controls. Toolbars along the top edge and bottom edge of the screen provide user selections which are common to many of the test functions. For example, you can adjust input gain, turn phantom power on and off, and control the generator output and the headphone monitor from any screen using the top SP395A. It is important that you understand the use of the toolbars to properly operate the SP395A. Individual items which can be selected and changed are referred to as fields in this manual.



Toolbars along the top edge and bottom edge of the screen provide user selections which are common to many of the test functions.

Top Tool Bar

The top toolbar provides a function identification field and is used to control the SP395A inputs. You can manually select input gain, turn on phantom power, or invert the phase of either channel independently. You can also enable auto-gain scaling for both inputs.



The Top Tool Bar provides function identification and selections for the L & R inputs.

Top Toolbar Icons

- Back Arrow.** Clicking on this icon exits the currently running function and returns to the function menu.
- Battery Indicator.** When operated on the internal battery, this field shows the relative battery level, or remaining charge, in the battery. When operated in AC mode, this field shows the AC power symbol.
- Screen Title.** This is the abbreviated name of the function that is currently running.
- Function related selection – Not on all menus
- Left Input Polarity** The N/R icon changes the left input from normal (N) polarity to reverse (R) polarity.
- Left Input Phantom Power** Click to select the icon, then turn clockwise to enable phantom power (48v), or counter-clockwise to disable phantom power (0v).

7. **Left Input Gain** There are 12 selectable input gain settings. Click and turn to select the gain for this channel. A blinking value or digits indicate clipping and the need to reduce the gain selection number.
8. **Auto-Gain.** Select manual (M) or auto (A). In auto-gain, the gain setting for the channel is automatically adjusted to prevent clipping and to have enough gain for accurate measurements. Note that in some functions, such as Leq, you may want to hold the gain at a fixed level. In this case, choose manual (M).
9. **Right Input Gain** There are 12 selectable input gain settings. Click and turn to select the gain for this channel. A blinking value or digits indicate clipping and the need to reduce the gain selection number.
10. **Right Phantom power** Click to select the icon, then turn clockwise to enable phantom power (48v), or counter-clockwise to disable phantom power (0v).
11. **Right channel polarity reverse** The N/R icon changes the left input from normal (N) polarity to reverse (R) polarity.

Bottom Tool Bar

The bottom toolbar contains three functional control sections. The sections include:

1. Headphone/Speaker Monitor
2. Memory
3. Output Generator

All or part of the bottom tool bar sections or selections within each section may be present in the various test functions of the Audio Integrator.

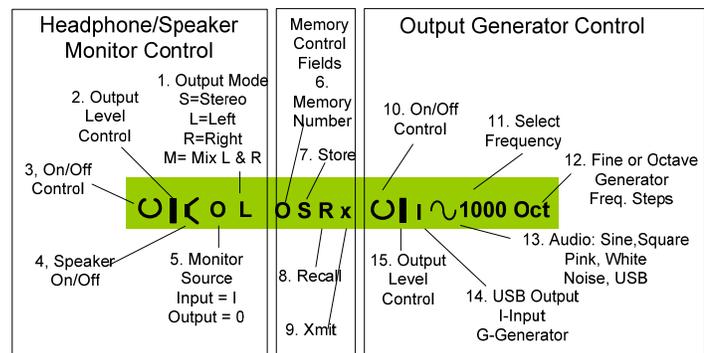
The Bottom Tool Bar provides headphone/speaker, memory, and internal generator control.

Headphone/Speaker Monitor Control

The five icons at the left end of the bottom toolbar control the monitor output.

1. **Monitor Output On/Off** The 1/0 icon turns the analog monitor output on and off. This controls both the headphone jack and the built-in speaker.
2. **Level** The bar level gauge icon controls the analog output level. Every turning step of the control knob changes the output level by approximately 1 dB.
3. **Speaker On/Off.** This icon turns the speaker on and off. The speaker can be turned off, even when the headphone output is enabled. To hear the speaker output, both the Monitor On/Off and the Speaker On/Off must be turned on.
4. **Source** The I/O source icon determines the source for the monitor output. The I selects the Input for monitoring, the O selects the Output signal (usually the internal digital generator) for monitoring. Therefore, in Input mode you will hear exactly what is being received, as selected by the input select field on the top toolbar, and in Output mode, an exact copy of the digital output signal will be routed to the headphone jack.
5. **Mode.** The monitor mode field changes between stereo (S), left mono (L), right mono (R), and the mixed left and right channels (M).

Bottom Toolbar Description



Headphone Warning: Be careful when wearing headphones and changing input connections, input gain, and phantom power. Although the analog output is usually muted in situations that may cause high output, unexpected loud volume may occasionally occur at the headphone output. We recommend turning off the output before changing functions, and in other situations that may cause loud volume.

Memory Control

Some test functions support saving measurement data to internal, non-volatile memory. The SP395A provides memory storage of RTA, FFT, ETG, Dosimeter, Sound Study Graphs, Amplitude Sweep and Impedance Sweep functions. These functions use the four memory control icons, in the center of the bottom toolbar. Note that a memory must be empty (cleared) before measurement data can be stored into it. The Clear (C) icon can be used to erase the contents of a memory.

6. **Memory Number** The memory number icon selects a memory location for storing, recalling, or clearing. There are memory numbers 0 through 39 or 40 locations. When the memory number field is selected, the type of data stored in that location (or “---” if empty) appears on the bottom of the graph.
7. **Store** Click the S (Store) icon to store the results of the current function. The memory location must be available (empty) before storing results. If the function does not support memories, the icon does nothing.
8. **Recall** Click the R (Recall) icon to retrieve the results stored in the current memory location and to display them on the screen. If the current function does not support memories, the icon does nothing.
9. **Xmit** Click the x (transmit) icon to jump to the transmit screen for download of memories to a PC.

Main Output and Generator Control

The five icons that control the main output and signal generator are located on right end of the bottom toolbar. Some fields control only the generator, while others control features of the main output section.

10. **On/Off** The 1/0 icon turns the generator on and off. This field also controls functions that use special output signals, such as the transparency test.
11. **Select Frequency** This field changes the frequency of the generator output, as determined by the Step Size field.
12. **Step Size** This field selects how the frequency field changes the generator frequency. Choices are octave steps (Oct), one-third octave steps (1/3), or fine mode (Full). In full mode the generator frequency is selectable in 1 Hz steps to 10 kHz and 2 Hz steps above 10 kHz.
13. **Waveform** This icon selects the generator waveform. It can also select the USB input as a signal source. The available waveforms are sine, square, white noise, and pink noise. The USB selection sets up the USB input as a source for the main output.
14. **USB Output** sets the USB digital to output signal from the mic input (I) or internal generator (G).
15. **Level** The bar level gauge controls the digital generator output level. Every turning step of the control knob changes the output level by exactly 1 dB.

In some functions the output generator is required to be in set conditions to perform the test function. Examples include the Impedance Meter, the Cable Tester, and the Sweep tests. In these functions, some icons on the bottom toolbar will not appear. However, even if generator settings, levels, and other items are changed, the settings that were in effect most recently will re-appear when a function is selected that uses the normal toolbars.

SPL

The SPL main menu selection provides several functions that are used in measuring Sound Pressure Level (SPL). SPL typically uses a microphone as an input. The standard input for the SPL tests in the left (L) input.

The SPL tests that are available on the SP395A include:

- Sound Level Meter:** Measures sound pressure level in dB.
- Leq:** Measures linear averaged sound pressure level over time.
- Dosimeter:** Measures sound dosage according to ANSI or ASA specifications.
- Sound Study Graph:** Graphs sound level changes over time.



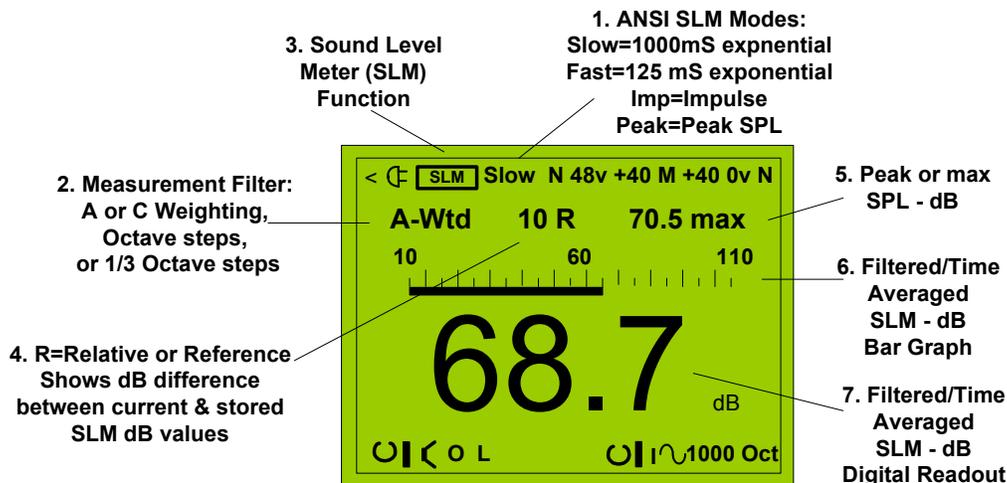
Sound Level Meter

A sound pressure level meter measures the changes in air pressure created by sound waves and displays this pressure as a dB level, relative to the threshold of hearing sound pressure. Weighting curves are added to SPL measurements to make SPL readings correspond to perceived loudness. The SP395A's Sound Pressure Level function measures the loudness of the ambient sound level in standard dB SPL measurement units, auto ranged from 30 to 130 dB (A-weighted). Use the SLM function when you need to know objective sound volume, such as when checking for sound level uniformity, balancing

speaker output levels, or adjusting room sound levels.

The SPL level is a true-RMS measurement, using ANSI Type 1 standard display time averages, ANSI Type 1 standard A and C weighting networks, ANSI Class 1 octave and 1/3 octave band filters, and ANSI Slow, Fast, and Impulse averaging. SPL measurements are made using the supplied microphone connected to the Microphone Input. The large, digital display shows the ambient sound level in standard units of dB SPL and the level is also shown on an analog bar graph meter (one pixel equals one dB).

SLM Measurement Screen Description



The SLM measurement screen.

The following fields and parameters are displayed in the SLM function:

1. **Averaging** - Three ANSI-standard averaging modes are available. These modes use exponential decay averaging, in which the more recent sounds have more bearing in the average.
 - Slow** – 1000 mS exponential decay, time-averaged RMS SPL.
 - Fast** – 125mS exponential decay, time-averaged SPL.
 - Impulse** – 35 mS exponential decay, time-averaged SPL
 - Peak** – Peak, rather than RMS SPL.
 Highlight the Averaging field and press the control knob to toggle between these four modes.
2. **Weighting** - Standard unweighted (flat), A-weighted, and C-weighted filters are available. These weighting filters make SPL readings correspond more closely to what our ears hear. Ten ANSI Class 1 octave-band filters – 31, 63, 125, 250, 500, 1k, 2k, 4k, 8k, 16 kHz; and 30 ANSI Class 1 1/3 octave-band filters are also available. Highlight the field and press the control knob. Rotate the knob to the desired filter and click to select it.
3. **Test Function:** Identifies the current test selected.
4. **R - Relative (Seat to Seat SPL)** Clicking on the “R” (relative) field sets the currently displayed SPL reading as the relative SPL reference. Then, the numeric dB field will show the difference between the current reading and the stored reference. This is useful to show relative seat-to-seat SPL readings.
5. **Max – Max Hold Clear** The maximum SPL reading is constantly updated and displayed. To reset the displayed value, highlight and click on the "Max" field.
6. **Bar Graph SPL:** Indicates SPL measurement in bar graph presentation.
7. **Digital Readout:** Indicates SPL measurement with a digital numeric value.

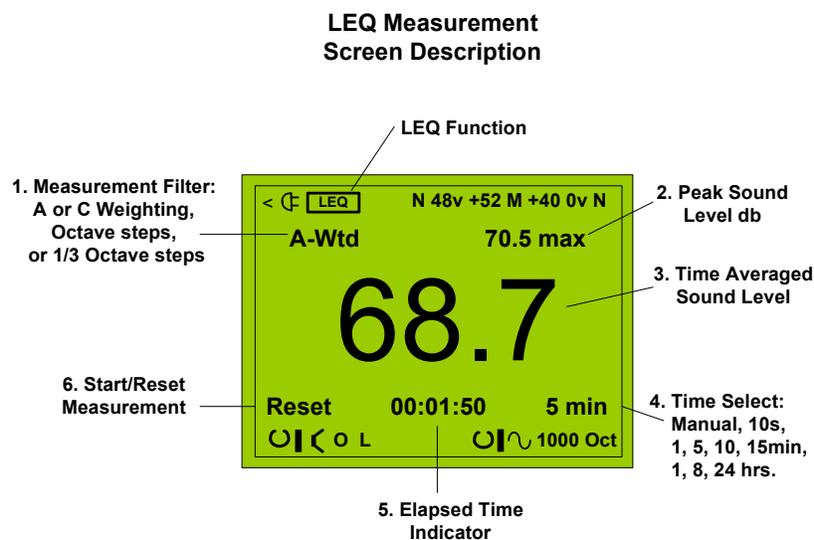
SLM Operation

1. **Connect a test signal to the audio system input.** SPL tests require a constant-level signal, such as a single-frequency test tone or wide-band pink noise. To use the SP395A as the signal source, connect a cable from an output connector to the desired audio system input. **Caution: Preset the amplifier gain to minimum to prevent speaker damage when the SPL test is turned on.** There will be no output until the signal generator is turned on in the bottom Toolbar.
2. **Position the microphone.** Position the microphone for SPL testing. For example, when setting speaker level balance, position the microphone in the center of the listening position.
3. **Select one of the four available ANSI-standard averaging modes:**
 - Use Slow (1000 mS exponential decay time, time-averaged RMS) for most SPL measurements. This averages transients and provides the best indication of the sound level that our ears hear.
 - Use Fast (125 mS exponential decay time, time-averaged RMS) to follow fast audio changes.
 - Use Impulse mode, (35 mS exponential decay time, time-averaged RMS) to measure noise spikes.
 - Use Peak mode to show peak SPL, rather than RMS.
4. **Select the weighting filter.** Select one of 38 filtering modes:
 - C-weighting for louder SPL levels, including most system measurements.
 - A-weighting for low SPL levels, or for measurements that need to correlate to noise-induced hearing damage.
 - Flat (un-weighted).

- Any of nine ANSI Class 1 octave-band filters – 31, 63, 125, 250, 500, 1000, 2000, 4000, and 8000Hz.
 - Any of 25 ANSI Class 1 1/3 octave-band filters.
5. **If you are using the Audio Integrator as the signal source for the audio system:**
- Turn the generator on by clicking the "on/off" field.
 - Adjust the generator output level.
6. **Read the SPL level.**

Leq

Leq is defined as the linear average of the sound pressure level over a period of time. You can use this to determine if noise standards are being met. The LEQ function computes and displays the time-averaged sound level. When an A-weighting filter is used to measure the time-averaged effects of noise, this measurement is often called LAeq.



Leq measurement screen.

The following fields and parameters are displayed in the Leq Screen:

1. **Weighting** - Standard unweighted (flat), A-weighted, and C-weighted filters are available. These weighting networks make SPL readings correspond more closely to what our ears hear. Ten ANSI Class 1 octave-band filters – 31, 63, 125, 250, 500, 1k, 2k, 4k, 8k, 16 kHz; and 30 ANSI Class 1 1/3 octave-band filters are also available. Highlight the field and press the control knob. Rotate the knob to the desired filter and click to select it.
2. **Max SPL** - The SPL reading in the upper right is a peak-hold SPL reading. Highlight the "Max" field and click the control knob to reset this reading.
3. **Linear Averaged Sound Level dB** – Leq measurement value.
4. **Time Duration of Test** - Select time interval of test including 10 sec, 1, 5, 10, or 15 min. or 1, 8, or 24 hrs. Can be set to Manual for manual starting and stopping of the test interval.
5. **Elapsed Time** - This field shows the time that the input sound level has been averaged, in hours, minutes, and seconds since the start of the test.
6. **Start/Reset** - This field starts the time or interval for the test measurement. This field also resets the time interval to start a new test during or after a previous Leq test interval.

Leq Operation:

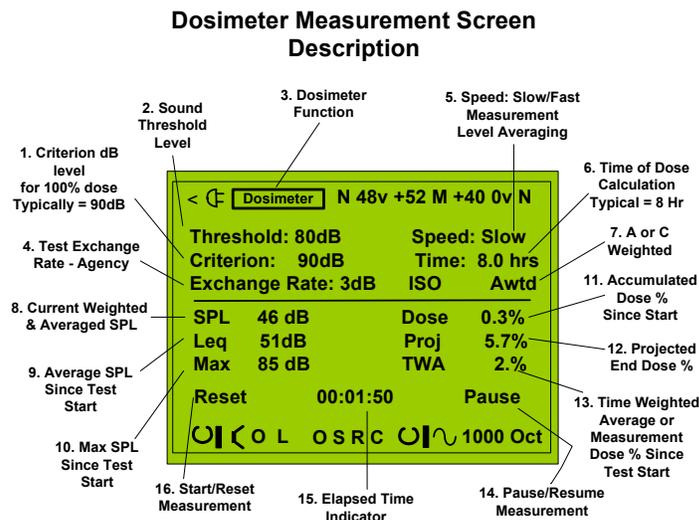
1. **Connect a microphone** to the L input. Apply phantom power and set the gain range.

2. **Select the desired filter:** Choose from A or C -weighting, an octave or one-third octave band, or flat (unweighted). A-weighting is often desired when noise is being measured.
3. **Select the time.** Choose from 10seconds, 1, 5, 10, or 15minutes, or 1, 8, or 24 hours. Or, select Manual and start and stop the LEQ test yourself.
4. **Click Start.** After a 2 second delay, the SP395A starts averaging the incoming sound. It stops automatically at the end of the selected time, or, if you are in Manual mode, continues to run until you click Reset.

Note: To avoid the possibility of incorrect averages, do not use Auto-gain in the Leq function. Pick a range that will handle the highest expected SPL without clipping through the length of the LEQ test.

Dosimeter

The SP395A can be used as a Noise Dosimeter for noise surveys, hearing conservation programs, community noise studies, machinery noise evaluations, and occupational noise analysis. The Noise Dosimeter meets ANSI S1.25-1991 and ASA 98-1991 specifications.



The Dosimeter test screen.

The following fields and parameters are displayed in the Dosimeter test screen.

Control Fields

1. **Criterion - Criterion Level** The Criterion Level is the 8 hour dB exposure level that is considered a 100% Dose. For most regulating agencies, a 100% dose occurs with an average sound level of 90dB over an 8-hour period.
2. **Threshold Level** –
3. **Test Function:** Identifies current test function as Dosimeter
4. **Exchange Rate** - Three Exchange Rates are available: 3 dB ISO, 4 dB DOD, and 5 dB OSHA. The Exchange Rate affects how sound energy is averaged over time. The ISO 3 dB standard, used by most users, causes the reported Dose to double for every 3 dB increase in the time-weighted average sound energy. The DOD 4 dB standard, used by the U.S. military, causes the reported Dose to double for every 4 dB increase in the time-weighted average sound energy. The OSHA (and MSHA) 5 dB standard causes the reported Dose to double for every 5 dB increase in the time-weighted average sound energy.

5. **Speed – Averaging Speed.** Slow (1000 mS time constant) and Fast (125 mS time constant) averaging speeds are available. Highlight the field and press the control knob to select the desired averaging speed.
6. **Time - Criterion Time.** Sets the Criterion Time from 0.1 hour to 24.0 hours, in 0.1 hour steps. Highlight the field and press the control knob. Rotate the knob to select the desired time.
7. **Weighting**– Standard A-weighted, and C-weighted filters are available. Highlight the field and press the control knob to select the desired filter.

Display Fields -

As the test is running, the elapsed time is shown on the screen. The fields displayed are:

8. **SPL** – Current Sound Pressure Level, using selected weighting and averaging speed.
9. **Leq** – Equal-weighted average SPL from start of test. Also referred to as the time-averaged sound pressure level.
10. **Max** – Maximum SPL since beginning of test, using selected weighting and averaging speed.
11. **Dose** – Noise Dose accumulated since start of test. For example, running at the Criterion Level for 1/2 of the Test Length would equal a 50% dose.
12. **Proj** – Projected Dose. This is defined as the dose that would be reported at the end of the Criterion Time Length, if no more sound above the Threshold was measured.
13. **TWA – Time-Weighted-Average.** The average sound level for the measured time period, spread out over the selected workday period (Time), which is normally set to 8 hours.
14. **Pause/Resume** – Permits test to be paused and then to be resumed.
15. **Elapsed Time** – Time that has lapsed since the start of the test period.
16. **Start/Reset** – Starts or resets the test.

Dosimeter Operation

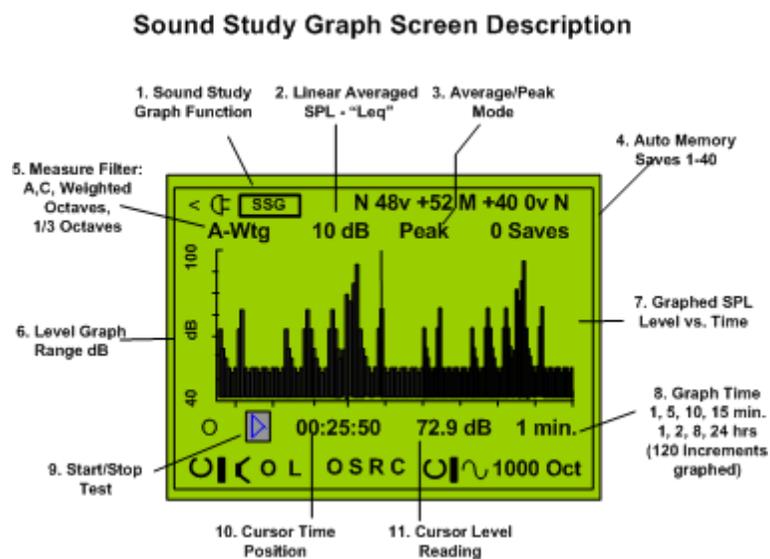
1. **Set the Criterion Level.** This is the level that will result in a 100% dose reading if maintained for the entire test. The criterion level is normally 90dB, but may be overridden if desired.
2. **Set the Threshold Level.** Any sound below this level does not contribute to the measured dose. Normally, this is set to 80dB. You can override this setting if desired. It may not be set higher than the Criterion Level.
3. **Set the Averaging Speed.** Normally, Slow (1 second time constant) is used, but may be changed to Fast if desired.
4. **Set the Criterion Time.** This time is normally set to 8 hours to match the typical work day. Dosimeter standards are all written around 8 hour work days. All Dose calculations are referenced to this time.
5. **Set the Exchange Rate.** This is defined as the number of dB equaling a doubling (or halving) of the Dose. For example, if a 90 dB dose for 8 hours equals 100%, with a 5dB exchange rate, an 85 dB dose for 8 hours would equal a 50% dose. 95dB for 8 hours would equal a 200% dose. This field also displays OSHA, DOD (US Department of Defense), and ISO, when the Exchange Rate specified by those agencies is selected.
6. **Start the test.** Click on “Start” to begin the test.
7. **Pause / Resume.** You can use this field to pause the test whenever desired, for example to not include an event or time period in the Dosimeter test period.

The Dosimeter results may be stored in a memory, or recalled to examine a previous test.

Sound Study Graph

The Sound Study Graph function graphically displays the SPL level over a period of time. Use this function to analyze noise levels over time, or to record concert or PA sound levels. It can run in peak or average mode, and any of the A, C, octave-band or 1/3 octave-band filters may be selected. The time can range from 1 minute to 24 hours. When the function runs, it divides the time period into 120 equal divisions, sampling the sound level at each period. The SP395A computes either the average or peak SPL for that period, and draws a bar on the graph. This continues until the total time for the test has expired.

As the function is running, the LEQ is computed and displayed. After the graph is complete, the full LEQ is displayed, and stored with the graph. The graph can then be stored in one of the memories, if desired.



The Sound Study Graph function graphically displays the SPL level over a period of time

The following fields and parameters are displayed in the Sound Study Graph test screen.

- Test Function** – Shows the selected test function.
- LEQ** - The accumulated LEQ (time-average SPL) is displayed in this field. When a test is complete, or when a memory is recalled, this field shows the overall LEQ for the entire graph.
- RMS/Peak Mode** - Switches between running the test in RMS average mode or in absolute peak mode.
- Saves - Automatic Memory Save.** If desired, set the number of automatic memory saves. This is the number to the left of the word "Save." When multiple saves are specified, the sound study graph automatically performs multiple sound studies or the specified period and saves each to memory.
- Weighting** - Any of 43 weighting filters are available, including unweighted (Flat); A-weighted; C-weighted; ten ANSI Class 1 octave band filters– 31, 63, 125, 250, 500, 1k, 2k, 4k, 8k, 16 kHz; or 30 ANSI Class 1 1/3 octave band filters.
- Graph Range** - The vertical scale is calibrated in dB, and covers a 60 dB range. Each division equals 10dB. The bottom range limit can be changed from 10 dB to 80 dB SPL.
- Graphed Data vs. Time** – Sound Level data graphed over the specified time period.

8. **Graph Length Time** - This is the field in the lower-right corner. You can select 1, 5, 10, or 15 minutes, or 1, 2, 8, or 24 hours.
9. **Start/Stop** – Starts the sound study and begins graphing sound level samples. Stops the test.
10. **Cursor Time Position** – Indicates the time represented by the position of the cursor on the graph.

Sound Study Graph Operation

To perform a Sound Study:

1. **Connect microphone** to L input. Apply phantom power and set the gain range.
2. **Select the desired filter:** Choose from A or C -weighting, an octave or one-third octave band, or flat (unweighted). A-weighting is often desired when noise is being measured.
3. **Select Avg (Average) or Peak measurement.** (Use Avg. for most applications)
4. **Select the time.** Choose from 10seconds, 1, 5, 10, or 15minutes, or 1, 8, or 24 hours. Or, select Manual and start and stop the LEQ test yourself.

Note: If you want to do multiple Sound Study Graphs to extend the study time or store results for analysis set the Memory field to the desired number of saves. Be sure to select a starting memory number in the bottom toolbar.

5. **Start/Stop** - Click on the start icon. The function will now run. While running, either the peak or average SPL is shown on the second line of the screen. As the bars are drawn, you can use the cursor to read the exact level and time of each bar.

Saves - Automatic Memory Save. You can use the automatic memory save feature to store the results of a test to memory at the end of the test. To store one test, set the “Save” field to “1.” To store a test and automatically run more tests, set the Save field to any number between 2 and 40. Then, as each test completes, its results will be stored in the next available unused memory, and a new test will begin. Test results will be stored only in unused memory locations. You can use the Memory Clear function to erase memories to make room for up to 40 test results.

Overloads - If any overloads occur while running, a small dot will appear above the dB bar. These overload values are stored in memory, along with the graph value.

ACOUSTICS

This main menu selection includes the SP395A's Acoustical Analysis functions. These functions are typically used to analyze sound or acoustic parameters in a room or space. These functions typically use a microphone for input. The standard input source is the left input unless otherwise noted. Also, phantom power is normally used to power the microphone.

The SP395A Acoustic Tests Include:

Real Time Analyzer (RTA)
FFT Analyzer
Energy Time Graph
Reverb Decay Time
Multi-Band Decay



The Acoustics Tests menu.

Real-Time Analyzer

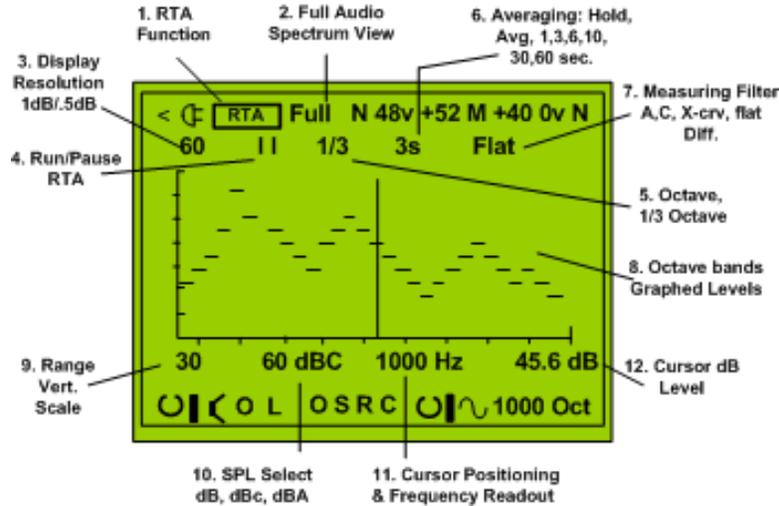
The Real Time Analyzer or RTA is essentially an audio spectrum analyzer which graphs the dB level of each audio octave or 1/3 octave band. Each band is graphed as a vertical bar, with the height representing the SPL level of the individual octave or sub-octave bands within the audio spectrum. The graph displays either a 15 dB or a 30 dB range. The display is updated approximately twice per second, with selectable signal averaging times from 1 to 60 seconds.

The filter-based RTA responds much more quickly to changing sound levels than does the FFT Analyzer function (below). The RTA function has frequency resolution to 1/3 octave. You can use the fast-responding RTA function to analyze the frequency response and levels of live voice and music. The 1/3 octave resolution is particularly useful, as this closely duplicates the resolution of the human auditory system.

The RTA function includes an ANSI X curve weighting network for motion picture theater setup is included. The SP395A RTA function uses ANSI Class 1 filters to separate the spectrum of the audio signal at the microphone or left line input into octave band or 1/3 octave band segments. RTA graphs can be stored to memory or compared with the "difference" mode for before and after analysis.

The internal signal generator can be used to provide a pink noise test signal while using the RTA function. RTA graphs can be stored and recalled for viewing and analyzing at a later time, and the data can also be downloaded to a PC, or can be viewed in real time with the TerraLink software.

RTA Measurement Screen Description



The RTA measurement test screen.

RTA Display and Controls

1. **Test Function** - Identifies the test function (RTA)
2. **Full Bandwidth** – Indicates all octave bands are graphed in the RTA. This field is not selectable.
3. **Graph Resolution** - Click the dB number at the top of the left graph axis to change the vertical graph resolution between 1 dB/pixel (30 dB range) to 0.5 dB/pixel (15 dB range). The resolution number changes to be either 30 or 15 dB higher than the range number at the bottom of the axis.
4. **Run/Pause** – Start the RTA or pause it by clicking the Run/Pause field. Pause the test to "freeze" the displayed RTA graph so that you can easier make cursor measurements or store the graph to memory.
5. **Octave Mode** – Select octave or 1/3 octave band filters, as indicated by Oct or 1/3.
6. **Averaging** – Select an exponential decay averaging mode with a time constant of 1, 3, 6, 10, 30 or 60 seconds; a mode that averages equally over time (cumulative); or peak hold mode. A numeric value indicates an exponential decay mode, with a time-constant equal to the value. Exponential decay averaging means that more recent sounds have more bearing in the displayed average. A longer time constant produces a more stable display, but a slower response to changing sounds. Cumulative average mode is indicated by AVG. Peak hold mode is indicated by HLD.
7. **Overlay Test Filter** – Select Flat response (normal), the A or C weighting curve, or the motion picture ANSI X-curve (PH22.202M-1984), used for setting up motion picture theaters. Note that these overlay filters actually modify the data on the screen, unless Flat is selected. This means that a flat input signal will show up like the A or C curve, with the lows and highs attenuated accordingly. This can be useful in community noise studies, to determine what frequency bands may be exceeding a standard referenced to A or C weighting. You may also select a "Diff" (difference) mode to show the change in RTA which occurred from a previous RTA captured into memory.
8. **Graphed RTA test data** – Bar graphed data.

9. **Graph Range** – dB range of graphed levels. Adjust the dB range as needed so that the display fits on the vertical axis, by selecting and changing the dB value at the bottom of the vertical axis.
10. **Average SPL** – The digital SPL measurement that is displayed along the bottom of the graph is calculated from an average of the RTA bands and is therefore slightly less accurate than the standard SPL function. It is provided here for convenience while using the RTA display. You can select **dB SPL** – the unweighted full-band SPL level; **dB A** - the A-weighted SPL; or **dB C** - the C-weighted SPL. Turn the control knob until the desired weighting value is displayed, then click to select the new value.
11. **Measurement Cursor** – Click the control knob on the cursor control field, then turn the control knob to move the cursor to any measurement bar to read out the exact frequency and dB level. Cursor measurements can be made in the Run or Pause mode, or on recalled RTA graphs.
12. **Cursor Level** – Indicates the level SPL of the cursor position (octave or 1/3 octave band) on the graph. The measurement Cursor allows you to analyze the RTA graph display in detail. It is an inverse-highlighted vertical bar that you can position horizontally along the graph. The center frequency of the selected band and the band's dB SPL level are displayed in the lower right portion of the graph. Cursor measurements can be made in the Run or Hold mode, or on recalled RTA graphs. To position the Measurement cursor along the graph, highlight the frequency "Measurement Cursor" field and click on it (underlined). Rotate the control knob to move the cursor.

RTA Operation

1. **Position the microphone.** A measurement microphone is normally placed at the desired measurement location and pointed at the ceiling to be omni directional to all system loudspeakers. Apply phantom power to the microphone and select the proper gain range.
2. **(Optional) Connect a constant-level pink noise signal to the audio system.** To use the SP395A as a pink noise signal source, connect a cable from an SP395A output connector to the desired input on the audio system. There will not be an output from the system until the RTA and Signal generator are turned on from the RTA display. The output level of the RTA pink noise is adjustable in the Signal Generator bottom toolbar menu section.
Caution: Preset the amplifier gain to minimum to prevent speaker damage when the RTA test is turned on.
3. **Select the octave band filter resolution.** Use 1/3 octave for most RTA analysis, as this resolution corresponds most closely to the resolution of the human auditory system.
4. **Select the averaging time constant.** Longer times provide a more stable display, while slower times respond more quickly to changing sounds and noise. Use 6 second (or longer) averaging when performing low frequency equalization.
5. **Select the desired weighting.** Use Flat (no weighting) for most applications.
6. **Turn on the pink noise.** If you are using the SP395A as the signal source, turn on both the RTA function and the generator. ("Gen on" field).
7. **Adjust the Graph Range and Graph Resolution.** Adjust the range and resolution so that the entire graph display fits on the vertical axis. Depending on the SPL difference between the highest and lowest levels, you may need to use 35 dB resolution to see the entire graph.
8. **Use the measurement cursor to read the exact frequency and dB level of any frequency band on the graph.** Pause the RTA, if desired, to freeze the display.

Storing an RTA graph to memory:

Real-Time Analyzer graphs can be stored in any of the 40 non-volatile memories, using the memory control icons in the center of the bottom toolbar. Use the memory number field to select a memory number. See the Bottom Toolbar section of this manual for details on the memory toolbar section. When the memory number field is selected, the type of memory stored (or “---” if empty) appears on the bottom of the graph. The memory type for the Real Time Analyzer is labeled “RTA”.

To store an RTA to memory:

1. **Select a memory** number in the memory number field. Note: The display indicates if a previous test screen is stored or “- -” if empty.
2. **Store to memory** - Click on the S (Store) icon to store the function data to the current memory number.
3. **Recall from memory** - Click on the R (Recall) icon to recall function data from a selected memory number. If the data type does not match the current function, a “Wrong type” error message will appear momentarily. A recalled graph can be analyzed exactly the same as a live graph - by changing the octave band filter size, graph amplitude range, and using the cursor to read individual band frequencies and dB levels. Memory data can also be transferred to a PC using the Transmit Data function.
4. **To Clear memory**. Click on the C (Clear) icon to clear function data from the current memory number.

RTA - Difference Mode

You can operate the RTA in a difference (Diff) mode. This allows you to see how current conditions differ from a stored memory RTA. It is especially useful to compare seating position RTAs or to judge how changes to the room affected the acoustic performance.

To use the RTA difference mode, you must first either recall or save a memory. Then, change the overlay field to ‘Diff’ and the display will switch to difference mode. The center of the screen will be set to 0 dB, with a ± 15 or ± 7.5 dB scale, depending on the graph resolution selected (top left corner of RTA graph). To exit difference mode, change the ‘Diff’ in the overlay field to “Flat.”

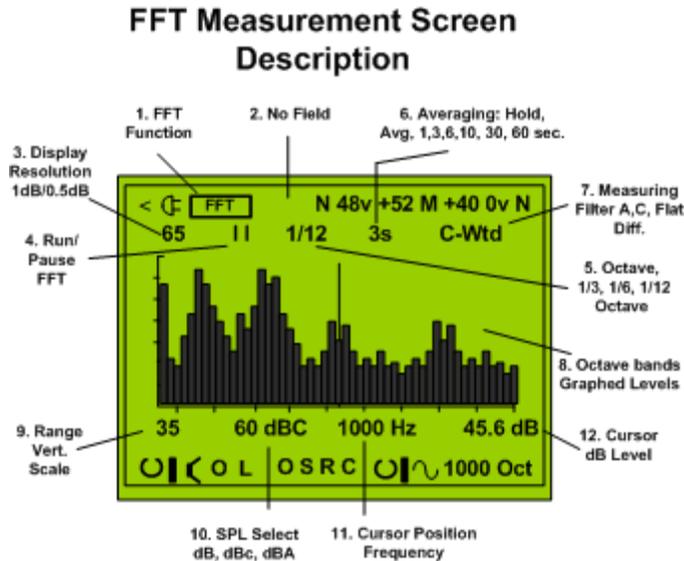
Real-time Interface

The RTA function can be used with the TerraLink computer software program to provide real time measurements on a PC screen. See the TerraLink Control – Real Time Computer Interface section of this manual for more information.

FFT Analyzer

The SP395A FFT Analyzer function processes the audio received at the microphone or left line input through three cascaded FFT stages that resolve the audio spectrum into octave, 1/3, 1/6, or 1/12 octave bands. Compared to the filter-based RTA (above), this function is slower to update, but has much finer frequency resolution.

Each band is graphed as a vertical bar, with the height representing the SPL level of the individual octave or sub-octave bands within the audio spectrum. The graph displays either a 30 dB or a 15 dB range. The display is updated approximately twice per second, with selectable signal averaging times from 1 to 60 seconds.



The internal signal generator can be used to provide pink noise while using the FFT Analyzer function. FFT Analyzer graphs can be stored and recalled for viewing and analyzing at a later time, and the data can also be sent to a computer.

Use the SP395A FFT Analyzer function to analyze the frequency response of audio systems and listening rooms, and to monitor the results as you re-position speakers, adjust room treatments, and change equalizer settings for the most acceptable response.

FFT Analyzing Test Screen

FFT Display and Control Fields:

- Test Function** - Identifies the test function (FFT)
- This field is empty on the FFT screen. No selection
- Graph Resolution** - Click the dB number at the top of the left graph axis to change the vertical graph resolution between 1 dB/pixel (30 dB range) to 0.5 dB/pixel (15 dB range). The resolution number changes to be either 30 or 15 dB higher than the range number at the bottom of the axis.
- Run/Pause** – Start the FFT or pause it by clicking the Run/Pause field. Pause the test to "freeze" the displayed FFT graph so that you can easier make cursor measurements or store the graph to memory.
- Octave Mode** – Select octave or sub-octave band. Select octave, 1/3 octave, 1/6 octave, or 1/12 octave bands, as indicated by Oct, 1/3, 1/6, or 1/12.
- Averaging** – Select an exponential decay averaging mode with a time constant of 1, 3, 6, 10, 30 or 60 seconds; a mode that averages equally over time (cumulative); or peak hold mode. A numeric value indicates an exponential decay mode, with a time-constant equal to the value. Exponential decay averaging means that more recent sounds have more bearing in the displayed average. A longer time constant produces a more stable display, but a slower response to changing sounds. Cumulative average mode is indicated by AVG. Peak hold mode is indicated by HLD.
- Overlay Test Filter** – Select Flat response (normal), the A or C weighting curve, or the difference (Diff). Note that these overlay filters actually modify the data on the screen,

unless Flat is selected. This means that a flat input signal will show up like the A or C curve, with the lows and highs attenuated accordingly. This can be useful in community noise studies, to determine what frequency bands may be exceeding a standard referenced to A or C weighting. You may also select a “Diff” (difference) mode to show the change in RTA which occurred from a previous RTA captured into memory.

8. **Graphed FFT test data** – Test data by band showing bar graph levels.
9. **Graph Range** – dB range of graphed levels. Adjust the dB range as needed so that the display fits on the vertical axis, by selecting and changing the dB value at the bottom of the vertical axis.
10. **Average SPL** – The digital SPL measurement that is displayed along the bottom of the graph is calculated from an average of the RTA bands and is therefore slightly less accurate than the standard SPL function. It is provided here for convenience while using the RTA display. You can select **dB SPL** – the unweighted full-band SPL level; **dB A** - the A-weighted SPL; **dB C** - the C-weighted SPL. Turn the control knob until the desired weighting value is displayed, then click to select the new value.
11. **Measurement Cursor** – Click the control knob on the cursor control field, then turn the control knob to move the cursor to any measurement bar to read out the exact frequency and dB level. Cursor measurements can be made in the Run or Pause mode, or on recalled FFT graphs.
12. **Cursor Level** – Indicates the level SPL of the cursor position (octave or 1/3 octave band) on the graph. The measurement Cursor allows you to analyze the FFT graph display in detail. It is an inverse-highlighted vertical bar that you can position horizontally along the graph. The center frequency of the selected band and the band's dB SPL level are displayed in the lower right portion of the graph. Cursor measurements can be made in the Run or Hold mode, or on recalled FFT graphs. To position the Measurement cursor along the graph, highlight the frequency "Measurement Cursor" field and click on it (underlined). Rotate the control knob to move the cursor.

Overlay – Select Flat response (normal), or the A or C-weighting curve. Note that the A or C overlay filters actually modify the data on the screen. This means that a flat input signal will show up like the A or C curve, with the lows and highs attenuated accordingly. This can be useful in community noise studies, to determine what frequency bands may be exceeding a standard referenced to A or C weighting.

To use the FFT function:

1. **Position the microphone.** A measurement microphone is normally placed at the desired measurement location and pointed at the ceiling to be omni directional to all system loudspeakers. Apply phantom power to the microphone and select the best gain range in the top toolbar section.
2. **(Optional) Connect a constant-level pink noise signal to the audio system.** To use the SP395A as a pink noise signal source, connect a cable from an SP395A output connector to the desired input on the audio system. There will not be an output from the system until the FFT and signal generator are turned on from the FFT display. The output level of the pink noise is adjustable in the Signal Generator bottom toolbar section. **Caution:** *Preset the amplifier gain to minimum to prevent speaker damage when the FFT test is turned on.*
3. **Select the octave band filter resolution.** Use 1/6 or 1/12 octave for most FFT analysis, as this resolution provides the best resolution to discern differences in the audio sub-octave bands.

4. **Select the averaging time constant.** Longer times provide a more stable display, while slower times respond more quickly to changing sounds and noise. Use 6 second (or longer) averaging when performing low frequency equalization.
5. **Select the desired weighting.** Use Flat (no weighting) for most applications.
6. **Turn on the pink noise.** If you are using the SP395A as the signal source, turn on both the FFT function and the generator. ("Gen on" field).
7. **Adjust the Graph Range and Graph Resolution.** Adjust the range and resolution so that the entire graph display fits on the vertical axis. Depending on the SPL difference between the highest and lowest levels, you may need to use 35 dB resolution to see the entire graph.
8. **Use the measurement cursor to read the exact frequency and dB level of any frequency band on the graph.** Pause the FFT if desired, to freeze the display.

Storing an FFT graph to memory:

FFT graphs can be stored in any of the 40 non-volatile memories, using the memory control icons in the center of the bottom toolbar. Use the memory number field to select a memory number. See the Bottom Toolbar section of this manual for details on the memory toolbar section. When the memory number field is selected, the type of memory stored (or "---" if empty) appears on the bottom of the graph. The memory type for the FFT is labeled "FFT".

To store an FFT measurement screen to memory:

1. **Select a memory number** in the memory number field. Note: The display indicates if a previous test screen is stored or "- -" if empty.
2. **Store to memory** - Click on the S (Store) icon to store the function data to the current memory number.
3. **Recall from memory** - Click on the R (Recall) icon to recall function data from a selected memory number. If the data type does not match the current function, a "Wrong type" error message will appear momentarily. A recalled graph can be analyzed exactly the same as a live graph - by changing the octave band filter size, graph amplitude range, and using the cursor to read individual band frequencies and dB levels. Memory data can also be transferred to a PC using the Transmit Data function.
4. **To Clear memory.** Click on the C (Clear) icon to clear function data from the current memory number.

FFT - Difference Mode

You can operate the FFT Analyzer in a "Diff" or Difference mode. This allows you to see how current conditions differ from a stored memory. This allows you to see how current conditions differ from a stored memory FFT. It is especially useful to compare seating position FFTs or to judge how changes to the room affected the acoustic performance

To use the Difference mode, first either recall or save a memory. Then, change the overlay field to 'Diff' and the display will switch to Difference mode. The center of the screen will be set to 0 dB, with a ± 15 or ± 7.5 dB scale, depending on the graph resolution selected (top left corner of FFT Analyzer graph). To exit Difference mode, change the 'Diff' in the overlay field to "Flat."

Real-time Interface

The FFT function can be used with the TerraLink computer software program to provide real time measurements on a PC screen. See the TerraLink Control – Real Time Computer Interface section of this manual for more information.

Energy Time Graph

Sound travels only about 1130 feet/second. Time delays caused by even relatively short distances can cause audible effects. An energy-time graph shows time relations between sounds.

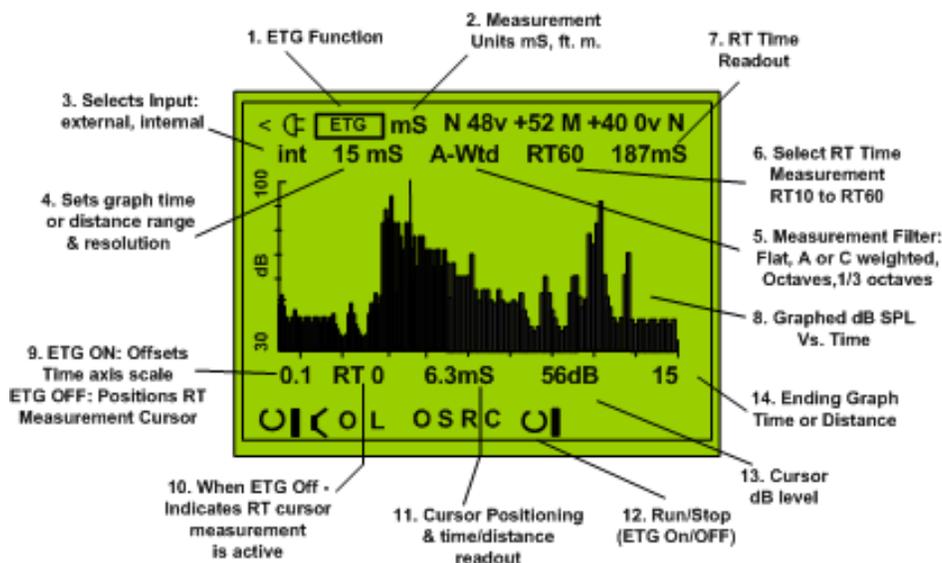
The SP395A Energy-Time Graph (ETG) function applies an exciting pulse through an external amplifier and speaker, and displays direct sound from the speaker, plus the energy decay of the sound reaching the microphone. The graph represents the instantaneous SPL levels at specific times after the exciting pulse occurs, and is calibrated vertically in dB (SPL) and horizontally in time or distance.

Use the ETG function to measure direct sound arrival times, identify room surface reflections, measure Initial Time Delay Gap (ITDG) to characterize room acoustics, accurately set A/V processor time delay and to calibrate speaker time delays in sound reinforcement systems. The energy-time graph is typically viewed at a short time/distance to check direct sound arrival time from a speaker, identify early speaker reflections, check PA speaker phasing, or determine ITDG; and at a long time/distance to view the room energy decay rate.

The Energy-Time Graph shows the decay pattern for sound reaching the microphone or line input over time after a pulse has been generated and played through a speaker, or after a pulse has been detected externally. For an internally-generated pulse, initial delay time is also shown. The graph is calibrated in units of dB SPL vertically, and units of time horizontally. Optionally, the graph can show units of distance on the horizontal axis.

The Energy-Time Graph can also be used to measure the reverberation decay RT60 time for a room. RT60 is defined as the time it takes for a loud sound in a room to decay by 60dB.

ETG Measurement Screen Description



Energy Time Graph (ETG) test screen.

ETG measurement screen and control fields:

1. **ETG Function** – Indicates ETG test function has been selected.
2. **Sound travel time or distance unit of measurement** – measurement units include milliseconds (mS), feet (ft), or meters (m). This field controls what measurement units are displayed along the horizontal (X) axis of the display

3. **Trigger Mode** - The ETG graph shows the time relationships of sounds, referenced to a sound pulse trigger. This trigger pulse is graphed at time/distance 0, at the far left side of the graph. Two trigger modes are provided:
 - **Int - Internal trigger:** Use this mode when the SP395A's generator output is connected to an amplified speaker. The graph updates repeatedly with each new output pulse. This is the most common way to use the ETG function.
 - **Ext - External trigger:** Use this mode to trigger the ETG using an external sound impulse, such as a handclap, starter pistol, or balloon pop. In this mode, the graph updates only when it detects a trigger pulse of sufficient SPL level (about 75 dB).
4. **Horizontal Measurement Range** – This field controls the full-scale resolution of the measurement that is displayed by the horizontal (X) axis. The selected full scale range is also displayed in the **Horizontal Full Scale** field. The following full scale range selections are available:
 - Millisecond ranges - 15, 30, 60, 120, 240, 480, 960
 - Feet ranges - 16, 33, 67, 135, 271, 542, 1084
 - Meter ranges - 5, 10, 20, 41, 82, 165, 330
5. **Weighting** – Any of 36 weighting filters are available, including unweighted (Flat); A-weighted; C-weighted; ten ANSI Class 1 octave band filters – 31, 63, 125, 250, 500, 1k, 2k, 4k, 8k, 16 kHz; or 23 ANSI Class 1 1/3 octave band filters.
6. **RT Select** – RT60 is defined as the time it takes a sound to decay by 60dB. It is often used to characterize rooms for reverberation decay time. The ETG function allows you to calculate reverberation times from a sound energy decay slope, using the RT Cursor and the Measurement Cursor. Select from the following: RT10, RT20, RT30, RT40, RT50, and RT60.
7. **RT Time** – This field shows the desired RT10-60 time that is extrapolated from the points marked on the decay slope by the RT Cursor and the Measurement Cursor. Measured decay times that are less than the desired decay range (i.e.; 60 dB for RT60) are mathematically projected, at the measured decay rate, to the desired RT measurement. The RT time is shown in milliseconds.
8. **Measurement data** - Graphed dB SPL vs. time with triggered starting point at left of graph.
9. **RT Cursor/Offset** – This is a dual-purpose field.
 - When the ETG is running, the field just below the left graph axis is an offset value. You can use this to offset the starting time of the display to a time greater than 0 mS, to see the details of a reflection, for example, that are beyond the normal window time width.
 - When the ETG is paused, changing this field moves the RT Cursor to set an RT calculation starting point on the ETG curve. The RT Cursor is the small marker that looks like a short "T," located just above the graph. You can move the cursor to any point to the left of the measurement cursor, to position it at the start of the decay slope.
10. **RT Cursor Measurement Enabled** – Indicates the RT calculation (based upon the position of the cursor) is active or available. This occurs when the ETG function is stopped.
11. **Measurement Cursor** – The Measurement Cursor allows you to analyze the ETG graph in detail. It is an inverse-highlighted vertical bar that you can move horizontally along the graph. As you move the cursor, the time (or sound travel distance) from the source pulse, and the dB level relative to the maximum value received, are displayed in the Measurement Cursor data field. The Measurement Cursor is also used to set the end time for the RT calculation.

12. **Run/Stop** – Starts and stops the ETG function. – Click on this field to turn On the ETG function to run the test and Off to Pause the current display. When Run is selected, the test pulses the room with sound energy and collects the resulting energy decay data. Select Pause to stop the test and "freeze" the currently displayed graph. This allows you to make cursor measurements and RT calculations, and to save the graph to memory. The display graphs the decay of the test pulse, as measured at the microphone. If the graph does not show a decay, the test pulse may be too quiet compared to the noise in the room, or the graph width may be too short.
13. **Cursor level dB** – indicates level dB of sound indicated at position of cursor.
14. **Ending Graph Time or Distance** – Stop the test when you have a good graph, and use the cursor control field to move the Measurement Cursor. The time and level of the cursor position are displayed below the graph. Measure any peak directly, to compute room reflections or to phase speaker arrays. To move the Measurement Cursor, highlight the "Measurement Cursor" field and click on it (underline). Rotate the control knob to move the Measurement Cursor and read the corresponding time/distance and dB level.

Memory

Energy-Time graphs can be stored in any of the 40 non-volatile SP395A memories, using the memory control icons in the center of the bottom toolbar. To access the memories, follow these steps:

- **Select a memory number.** Use the memory number field to select a memory number. When the memory number field is selected, the type of memory stored (or “---” if empty) appears on the bottom of the graph. The memory type for the FFT Analyzer is labeled “FFT.”
- **Store to memory.** Click on the S (Store) icon to store the function data to the current memory number.
- **Recall from memory.** Click on the R (Recall) icon to recall function data from the current memory number. If the data type does not match the current function, a “Wrong type” error message will appear momentarily. A recalled graph can be analyzed exactly the same as a live graph - by changing the octave band filter size, graph amplitude range, and using the cursor to read individual band frequencies and dB levels. Memory data can also be transferred to a PC using the Transmit Data function.
- **Clear memory.** Click on the C (Clear) icon to clear data from the current memory number.

ETG Operation

Producing an Energy-Time Graph (follow these setup steps for all ETG applications.)

1. **Trigger Mode.** Select “Int” (Internal) to set the SP395A to generate its own excitation pulse. Plug any of the SP395A audio output into an amplifier and speaker and turn the level down at the amplifier!
2. **Select “Ext”.** (External) if you wish to trigger the ETG using a external signal, such as a handclap or starter pistol.
3. **Connect the SP395A Output to the amplifier input (for internal trigger).** Connect a cable from an SP395A output connector to the input of the audio amplifier. Apply the signal to each speaker individually. **Caution: Preset the amplifier gain to minimum to prevent speaker damage when the ETG test is turned on.** You will not hear an output from the speakers until the test is turned on.

4. **Units/Time.** Select the desired units (time, in milliseconds, or equivalent distance, in feet or meters) and duration of the graph window.
5. **Select the Horizontal Axis Measurement.** Use a distance display (ft, m) for locating reflections and the mS time display for ITDG and direct sound arrival time measurements.
6. **Select the Horizontal Axis Range.** To locate reflections, select a distance that is just longer than the total distance from the speaker, to the rear wall, to the microphone location. Use 60 mS for measuring ITDG in a typical size home theater room, and use longer times for large rooms.
7. **Weighting.** Select one of the 38 available filters, including A or C weighting, ANSI octave band filters, or ANSI 1/3 octave band filters. In most cases, A-weighting, which rolls off most of the low frequencies, will give you more reliable results.
8. **Select the desired weighting.** A-weighting often provides sharper timing peaks because it rolls off most of the low frequency background noise and reflections.
9. **Run/Pause.** Turn the function “on” by turning on the generator and carefully turn up the power amplifier (if you are using Internal trigger mode) to obtain the highest possible signal reading without overloading the SP395A or creating excessive distortion in the speaker.
10. **Turn on the test. *Be sure the amplifier volume is set to minimum before turning on the ETG test.***
11. **Adjust the amplifier volume for a good display.** Increase the amplifier volume to obtain a high signal reading on the ETG graph without signal clipping or speaker distortion. If the signal level picked up by the microphone is too loud, the display will indicate "Overload".
12. **RT Cursor/Offset.** While the ETG is running, if you wish to offset the starting time of the display to a time greater than 0 mS, change the Offset field, just below the left graph axis. This would allow you to see the details of a reflection that is beyond the window time width, for example, without increasing the graph time and decreasing the graph resolution.

Identifying Loudspeaker Reflections:

1. Follow these steps to obtain an Energy-Time graph. Enable only one loudspeaker at a time to display early reflections from that loudspeaker to the listening position.
2. Set the Horizontal measurement to either feet or distance.
3. When you have a representative graph, turn off the generator to freeze the graph.
4. **Analyze the data from the graph:**
 - **Direct sound** -The first peak is the direct sound and is the distance (or time) that the sound traveled in a straight line from the speaker to the microphone.
 - **Reflections** -The next peak(s) are reflections. Each peak is the distance that a reflection traveled from the speaker to the microphone. The peaks that occur within 10 milliseconds after the direct sound (a reflection path 10 feet longer than the direct sound path) are called early reflections. In a home theater, the early reflections coming from the side walls should be at least 10 dB lower than the direct sound level to prevent stereo imaging degradation.

To measure Initial Time Delay Gap (ITDG)

ITDG is the time between the direct sound and the first lateral sound reflection that is within 10 dB of the direct sound. This time clues the brain as to the room size - longer ITDGs mentally equate to larger spaces, while too short ITDGs cause early reflection problems. For good home theater acoustics, the ITDG for all speakers should be at least 10 mSec. ITDG time can be lengthened by adding sound absorbing material to the speaker mirror points to reduce early reflections.

1. Follow the steps outlined above for "Identify Loudspeaker Reflections".
2. Set the Horizontal measurement to time (mS).
3. After you obtain a good display, turn off the generator to freeze the graph.
4. Set the cursor to the direct sound (first peak). Note the dB level and the time.
5. Move the cursor towards the right until you encounter a next peak that has an SPL level that is within 10 dB of the direct sound level. Measure this time.
6. Subtract the time of the reflection that is within 10 dB of the direct sound level (time in step #5) from the direct sound arrival time (time in step #4).

Measuring RT Times

Reverberation time is how long it takes for the sound pressure level to decay by a specified dB amount after the sound source is removed. The ETG function allows you to determine RT times from RT10 to RT60.

1. Follow steps 1-6 above for obtaining an Energy-Time graph. Use A-weighting.
2. Select the desired RT time (RT10, RT20, RT30, RT40, RT50, or RT60).
3. Position the RT cursor to the start of the decay slope (to the right of the initial sharp decay).
4. Position the measurement cursor to the end of the decay slope.
5. Read the equivalent RT time for the decay slope.

RT60 Time ETG Measurement

Once you have a good decay graph, turn off the generator to freeze the graph. Select the RT Cursor by clicking on the field below the vertical dB axis, and move the RT Cursor until it is above the start of the linear decay slope, just after the initial sharp decay. This cursor will only move up to the main Measurement Cursor, so you may need to move the main cursor first. Then, select the main cursor and move it to the end time of the decay slope. The RT60 time for the decay slope you have selected will be computed and shown in the RT Time field, at the upper right.

Real-time Interface

The ETG function can be used with the TerraLink computer software program to provide real time measurements on a PC screen. See the TerraLink Control – Real Time Computer Interface section of this manual for more information.

Reverb Decay Time

The Reverb Decay Time function computes the reverb decay time for a room, referenced to the standard RT60 time. RT60 is defined as the time that it takes sound to decay by 60dB. The RT60 time may be computed full spectrum (20-20kHz), with no filtering (Flat), with A or C weighting filters, with any of ten standard octave band filters, or with any of 30 standard 1/3 octave band filters. An "octave band" is defined as one octave of sound, centered around one of the ISO standard octave frequency values, such as 1000Hz. Ear protection is recommended when using this function.

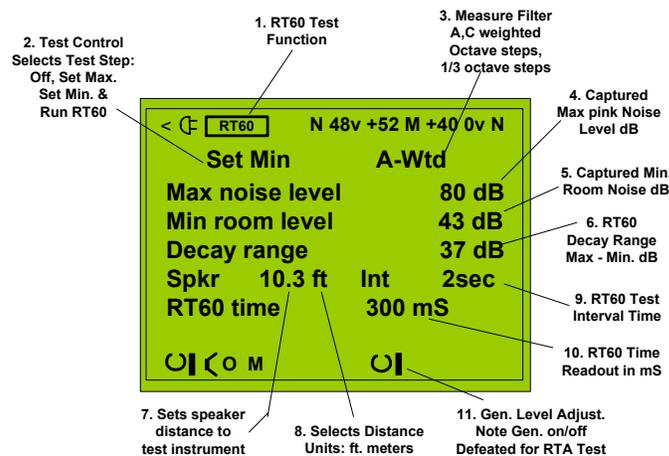
When a steady sound is applied to a room, the sound energy builds until it reaches a constant level. When the sound source is removed, the sound energy doesn't immediately cease, but decays over a short period of time. The length of time the sound energy reflects off multiple surfaces inside a room before it is absorbed by the room's contents is called decay time. The time needed for the sound level to decay by 60 dB after the sound source is removed is called reverberation time, or RT60.

Optimum decay time depends largely on the room's size, whether you are listening to speech or music, and listener preferences. As heard by a listener, decay time, or reverberation, creates room ambiance - the sense that the performance is taking place in either a large or small space. A decay time that is too long causes echoes, while a decay time that is too short makes the room sound "dead" and unnatural. Music requires longer RT60 times for best sound, while speech requires shorter RT60 time to be intelligible. An average-sized home theater should typically have an RT60 time between 0.3 and 0.6 seconds.

A room's volume (physical dimensions) and the sound absorption qualities of objects inside the room (i.e.: carpet, drapery, wall treatment, people etc.) determine its decay time. Make RT60 decay time measurements with all of the furnishings in place and, if possible, with someone seated in each listening position. If the final acoustical panels aren't available, place temporary panels at the speaker mirror points to eliminate early reflections.

The SP395A Reverb Decay Time function works by first measuring the room's ambient noise level, then the sound level with the test pink noise is applied. The difference in levels is the decay range (at least 26 dB is required for the test to continue). After the decay range dB is determined, the function outputs an interval of pink noise, followed by silence. As soon as the pink noise is stopped, the test measures the time needed for the room's sound level to drop by the decay range dB amount. This time is displayed as the Decay time reading. The RT60 time is determined by extrapolating the decay time to 60 dB.

Reverb Decay Time (RT60) Measurement Screen Description



The Reverb Decay Time (RT₆₀) Measurement Screen.

Reverb Decay Time Test Screen and Control Fields:

1. **RT60 Test** – indicates the selected test function or screen.
2. **Test Control** – The Reverb Decay function requires setting the ambient noise and the maximum sound (pink noise on) levels to determine the Decay range dB. The Test Control field allows the operator to set these levels and to control the test. Highlight and click this field until it displays "**Set Min**". Then click the control knob to store the ambient noise level. The field then changes to "**Set Max**" and the pink noise test signal turns on. Adjust the generator output, or the amplifier volume, to adjust the "Max noise level" to be at least 26 dB higher than the "Min noise level." Click to store the maximum noise level. The field now changes to "**Run RT60**" and the test begins. Click again to stop the test. When the field displays "**Off**", clicking again repeats the setup sequence. Turning the test "Off" holds the currently displayed readings.
3. **Weighting** – Any of 36 weighting filters are available, including unweighted (Flat); A-weighted; C-weighted; ten ANSI Class 1 octave band filters– 31, 63, 125, 250, 500, 1k, 2k, 4k, 8k, 16 kHz; or 30 ANSI Class 1 1/3 octave band filters. A-weighting will roll off most of the low frequencies. If in doubt about which filter to use, we recommend using A-weighting.
4. **Captured Max Noise Level** – This is the maximum sound energy in the room with the pink noise test signal applied. Adjust the amplifier's volume and/or SP395A generator output to increase the sound level to approximately 75-85 dB. At this level the Max Noise reading should be at least 26 dB higher than the Min Room level reading (number in the Decay Range field). To store the Max Noise level, set the Test Control field to "**Set Max**" and click the Test Control field.
5. **Min Room Level** – This is the level of the ambient room noise. The minimum room noise level must be at least 26 dB lower (indicated by a number display in the Decay Range field) than the Max Noise Level reading. If the Min Room level is greater than 50 dB, you need to lower the ambient noise for best test results. To store the Min Room level, set the Test Control field to "**Set Min**," then click the Test Control field.
6. **Decay Range** – This field shows the dB difference between the Max Noise and Min Room levels, minus 6 dB to allow for 6 dB test headroom. The Reverb Decay test measures the time taken for the sound energy to decrease by this amount. The display automatically updates when the Min Room level has been set, and displays "--" if the

difference between the max and min levels is less than 20 dB. Adjust the sound system's amplifier volume and/or the SP395A test generator output, or reduce the ambient background noise to obtain a decay range greater than 20 dB. Larger dB decay ranges produce better test results. If possible, try to attain a decay range of 35-45 dB.

7. **Speaker Distance (Value)** – Use this field to enter the approximate distance that the SP395A microphone is from the nearest speaker. The reverb decay test uses this information to ignore any possible early sounds that are not part of the normal test, allowing the test to provide more accurate measurements with transient noise spikes. Turn the control knob to highlight the Speaker Distance Value field. Click on the field and turn the control knob to set the distance in 0.1 increments from 4 ft. to 999.9 ft. or the equivalent range in meters. Click to select the desired value.
8. **Speaker Distance (Units)** – selects the unit of measurement used to set speaker distance either feet (ft) or meters (m). To set the speaker distance units, turn the control knob to highlight the Speaker Distance Units field, and click to select **m** (meters) or **ft** (feet).
9. **Test Interval** – This field sets the noise on-time duration (the off and on time are equal) from 2 to 15 seconds in 1-second steps. This determines the longest RT60 time that can be measured. Larger rooms require longer test intervals to allow the sound to build to maximum and the sound to decay.
10. **RT60 Time** – This field shows the calculated RT60 reverberation time. The number is extrapolated from the measured decay time by linearly extending the decay time and dB decay range to a 60 dB drop in SPL.
11. **Generator Level** – This field allows you to make coarse adjustments of the pink noise level. You may need to adjust the generator level to set Max Noise level for an acceptable decay range. Highlight the field and click to select it (underlined). Turn the control knob to the desired level then click to select the level setting.

Reverb Decay Time Operation

1. **Position the SP395A microphone in the center of the listening area.**
2. **Connect any SP395A line output** into an amplifier and speaker. Turn the level down at the amplifier!
3. **Select the Reverb Decay Time function and select A-weighting.** A-weighting rolls off most of the effects of low frequency room modes and provides a better indication of delay time.
4. **Speaker Distance.** Set the “Speaker” field to the approximate distance from the speaker to the SP395A’s microphone.
5. **Set the Test Interval.** Use short test times for small rooms, and longer test times for larger rooms and rooms with minimal sound absorption. Choose a test interval long enough to include the expected room reverberation time.
6. **Set Minimum Room Noise - Highlight the Test Control field and select "Set Min."** Allow the SP395A to read the average background noise level in the room. Click to capture the minimum room level (ambient noise). If this level is greater than 50 dB, you should attempt to decrease the ambient noise for best test results and better room acoustics.
7. **Set Maximum Room Level.** When you capture the minimum room noise (Set Min), the display changes to "Set Max" and pink noise is applied to the amplifier input. Adjust the amplifier's volume and/or the SP395A output level for a Max noise level of approximately 75-85 dB. This should produce a Decay Range reading of at least 26 dB (30dB or greater is recommended for a more accurate reading). You want the sound as loud as possible, without damaging your system or overloading the SP395A

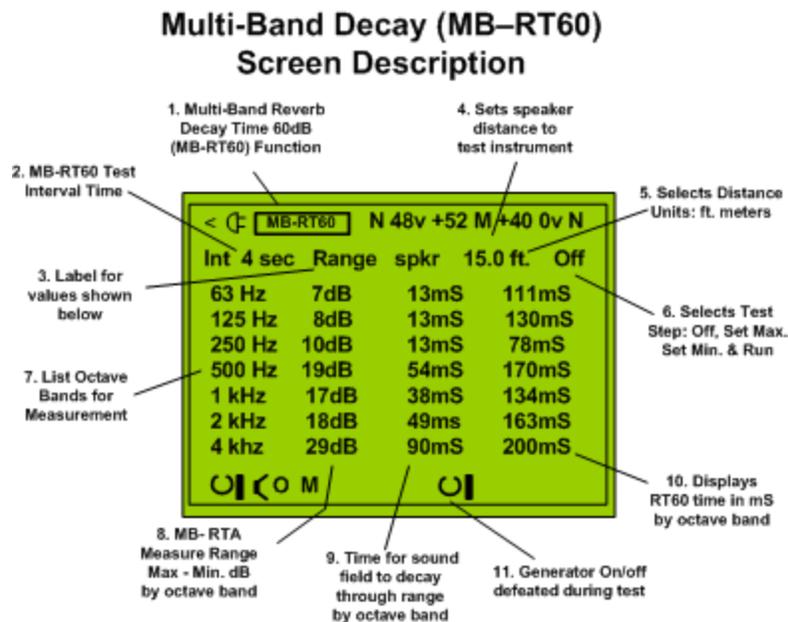
8. **Run RT60 - Click the Test Control field to display "Run RT60."** The pink noise generator will cycle on and off as readings are taken. The "Decay Time" field displays the actual time in milliseconds for the sound field to decay by the Decay range amount. The RT60 time field shows the extrapolated time for the sound to decay in the room, if a full 60dB decay range were available.
9. **Stop.** To stop the analysis and freeze the current test reading, change the mode from "Run RT60" to "Stop".

A full room analysis would involve running this test for all of the octave bands and recording the results.

Note: When looking at reverb times, keep in mind that low frequency energy can vary quite a bit and still be inaudible to the human ear. This can cause the results to vary in the lower octaves. Sources of low-frequency sound include heating and air-conditioning systems, traffic, and even people walking in other parts of the building.

Multi-Band Decay

The Multi-Band Decay function computes the RT60 reverberation time for 7 octave bands simultaneously. RT60 is defined as the time that it takes sound to decay by 60dB. Ear protection is recommended when using this function.



The Multi-Band RT60 (MB-RT60) test measurement screen.

MB-RT60 Test Measurement Screen and Control fields:

1. **Test Function** – Identifies the current test function MB-RT60 (Multi-band RT60).
2. **Test Interval** – Selects the time duration of the test as it alternates between pink noise on and off to perform the MB-RT60 measurement.
3. **Column Description** – Labels the columns of values below this field. Not selectable.
4. **Speaker Distance (Value)** – Use this field to enter the approximate distance that the SP395A microphone is from the nearest speaker. The reverb decay test uses this information to ignore any possible early sounds that are not part of the normal test, allowing the test to provide more accurate measurements with transient noise spikes. Turn the control knob to highlight the Speaker Distance Value field. Click on the field and turn the control knob to set the distance in 0.1 increments from 4 ft. to 999.9 ft. or the equivalent range in meters. Click to select the desired value.
5. **Speaker Distance (Units)** – selects the unit of measurement used to set speaker distance either feet (ft) or meters (m). To set the speaker distance units, turn the control knob to highlight the Speaker Distance Units field, and click to select **m** (meters) or **ft** (feet).
6. **Test Control/Mode** – increments the MB-RT60 test function through its testing steps. Steps include: Off, Min, Max, and Run steps.
 - **Off: Stops the MB-RT60 test.**
 - **Min: Sets Minimum Room Noise – Highlight the Test Control field and select "Min."** The SP395A reads the average background noise level in the room during this step. If this level is greater than 50 dB, you should attempt to decrease the ambient noise for best test results and better room acoustics.

- **Max:** Sets maximum room sound level using pink noise - This is the maximum sound energy in the room with the pink noise test signal applied. Adjust the amplifier's volume and/or SP395A generator output to increase the sound level to 80 – 90 dB. It is desirable to get at least 26 dB higher than the Min room level for best test results. This can be challenging as low frequency room noise tends to limit the decay range in the lower bands. To store the Max Noise level, set the Test Control field to "Max" and click the Test Control field to "Run"
 - **Run: Click the Test Control field to display "Run".** The pink noise generator will cycle on and off as readings are taken. The "Decay Time" field displays the actual time in milliseconds for the sound field to decay by the Decay range amount. The RT60 time field shows the extrapolated time for the sound to decay in the room, if a full 60dB decay range were available.
7. **Octave Bands** – lists the octave bands that are being measured by the SP395A as part of the multi-band RT60 analysis.
 8. **Decay Range** – This column of values shows the dB difference between the Max Noise and Min Room levels for each of the octave bands listed minus 6 dB to allow for 6 dB test headroom. The multi-band reverb decay test measures the time taken for the sound energy to decrease by this amount in each band.
 9. **Decay Time** – This column of values shows the time it takes for the sound to decay through the decay range for each of the octave bands listed.
 10. **RT60 Measurement per. Octave** – This column of values shows the extrapolated time for the sound to decay 60 dB (RT60) for each of the octave bands.
 11. **Generator On/Off** – This indicates whether the Generator is on or off. The Generator is defeated during the test.

Multi-Band Decay RT60 Operation:

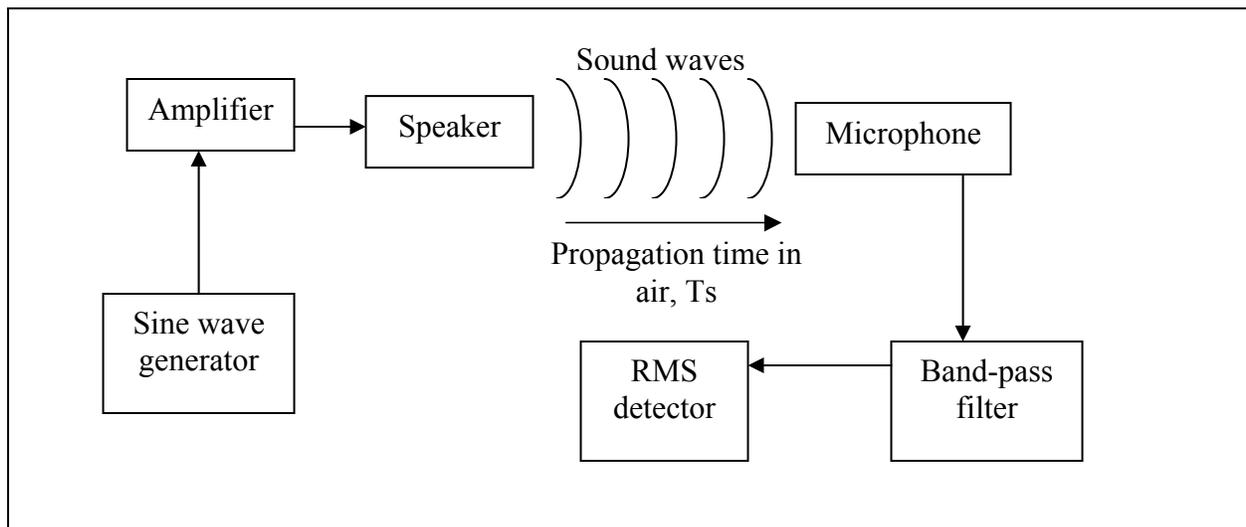
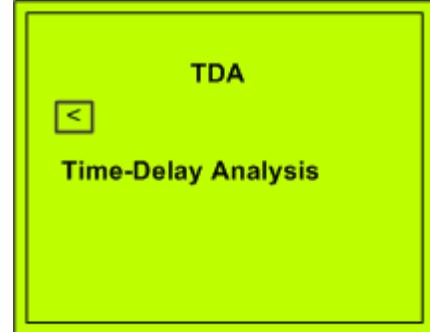
1. **Select the SP395A Multi-Band Decay function and position the microphone in the center of the listening area.**
2. **Connect any SP395A line output** into an amplifier and speaker. Turn the level down at the amplifier! Apply phantom power to the microphone and select the proper gain range on the SP395A.
3. **Set the Test Interval.** Use short test times for small rooms, and longer test times for larger rooms and rooms with minimal sound absorption. Choose a test interval long enough to include the expected room reverberation time.
4. **Speaker Distance.** Set the "Speaker" field to the approximate distance from the speaker to the microphone.
5. **Set Minimum Room Noise. Highlight the Test Control field and select "Min."** Allow the SP395A to read the background noise level in the room for each of the octave bands. If this level is greater than 50 dB in any band, you should attempt to decrease the ambient noise for best test results and better room acoustics. Click on "Min" to capture these minimum room levels.
6. **Set Maximum Test Level.** Adjust the amplifier's volume and/or the SP395A output level for a Max noise level of approximately 85 dB or higher. You want the sound as loud as possible, without damaging your system or overloading the SP395A. With the Test Control field highlighted showing "MAX", click the field to capture the maximum pink noise levels. The field increments to "RUN" and begins to cycle through the test.
Note: You may want to use earplugs so you can increase the volume to higher levels enabling a greater decay range.

- This should produce a Decay Range reading of at least 26 dB. The greater the decay range the more accurate is the extrapolated RT60 measurement. You may find that low frequency room noise does not make it possible to achieve a 26 dB range in the lowest octave bands.
7. **Run.** Upon setting the maximum pink noise level the test begins to “RUN”. The pink noise generator cycles on and off as readings are taken. The “Range” column displays the decay range obtained for each octave band. The “Decay Time” column displays the actual time in milliseconds for the sound field to decay by the decay range amount. The RT60 time column shows the extrapolated time for the sound to decay in the room for each octave band, if a full 60dB decay range were available.
 8. **Stop.** To stop the analysis and freeze the current test readings, change the test control mode from “Run” to “Off”

Note: When looking at reverb times, keep in mind that low frequency energy can vary quite a bit and still be inaudible to the human ear. This can cause the results to vary in the lower octaves. Sources of low-frequency sound include heating and air-conditioning systems, traffic, and even people walking in other parts of the building.

TDA – Time Delay Analysis

This menu includes the Time Delay Analysis functions of the SP395A. Richard Heyser's ground-breaking work on acoustics in the late 1960s created new paradigms for analyzing audio. The Audio Integrator incorporates one version of this work. As implemented in the SP395A, TDA is measured directly, using a log swept sine wave output and a narrow band-pass filter on the input.



Visual representation of the TDA test setup.

The time for the sound to travel from the speaker to the microphone (T_s) is approximately 1 mS for every 1 foot of separation (1130 ft/S or 0.885 mS/ft). The sine wave generator is swept through the portion of the audio spectrum we wish to test. Since the sound arriving at the microphone from the speaker is delayed by T_s , the band-pass filter sweep also needs to be delayed by T_s , so that at any given instant, the filter is tuned to the test frequency that is arriving at the microphone.

This allows the detector to 'hear' only the direct sound arriving at the microphone. Without the band-pass filter, all sound reaching the microphone would be summed in the RMS detector, and ambient noise or late-arriving signals due to reverberation or echoes in the room would contribute to false readings. With the band-pass filter sweeping in step with the generator, only one frequency at a time (depending on the filter bandwidth) is measured by the RMS detector.

Essentially, since the reverberant field contains all of the previously generated frequencies, attenuated at the RT60 rate, the filter allows the system to attenuate those frequencies and send to the measurement detector only the direct-sound frequency currently arriving from the speaker.

In the SP395A, 1/12 octave data points are selected for the TDA graphs. As the generator output sine wave is being swept, the input band-pass filter is also being swept, delayed by T_s . As each 1/12 octave boundary is reached, the RMS energy accumulated since the last boundary is recorded; then a new data point sum is begun. This adds a small amount (1/12 octave) of smoothing to the data.

This TDA measurement method is ideal for measuring the frequency response of speakers. In essence, TDA looks at the speaker as if it were in an anechoic chamber (ideally speaking), and the effect of the room on the measurement is reduced. The amount of reduction depends on the steepness of the band-pass filter and the speed of the sweep. The filter in the SP395A has variable Q (steepness), and can be set from a Q of 6 to a Q of 150. More information about filter Q and setting up the TDA function will be detailed below.

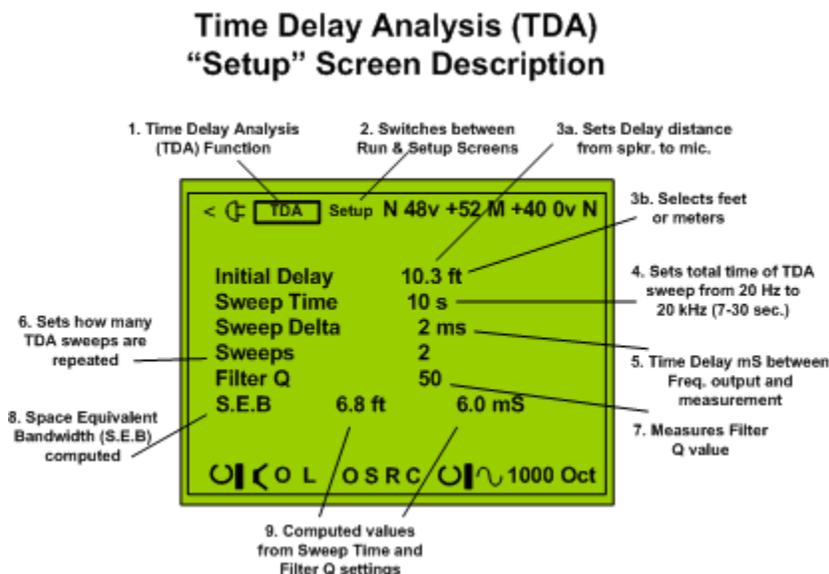
Room Response

An interesting thing happens if the filter sweep is delayed from the generator sweep by more than the T_s time. Say, for example, that the microphone was 8 feet from the speaker. With a speed of sound in air of 1130 ft/s, the time for sound to travel from the speaker to the microphone (T_s) is 0.007 seconds, or 7 mS. If we add 100 mS to the T_s time, for example, delaying the input filter sweep by 107 seconds from the generator output, the resulting graph shows the level of the swept sine wave 100 mS after it is first produced at the loudspeaker. We see the level of each frequency in the swept sine wave after 100 mS of decay in the room. To see the decay at any other time, just add that time to T_s and run the sweep.

Say now that we run 10 sweeps, each with a 50 mS greater delay. If the resulting 10 graphs are plotted in 3D, side by side, we get a good picture of the decay pattern of the room, by frequency, over 500 mS of time. Room modes appear as ridges that remain at a higher level after other frequencies have decayed. This multiple set of time-delayed graphs, showing us the room response over time, is sometimes called a 3D waterfall graph.

TDA Parameters

The SP395A's TDA function has great flexibility. You can set the distance from the microphone to the speaker (Initial Delay), the time for each sweep, the additional delay time between consecutive sweeps (Sweep Delta), the number of sweeps to run, and the Q of the filter. You can run the selected sweeps, page back and forth through the resulting graphs, use the measurement cursor to inspect data points on the graphs, and store the graphs to the internal memories at a specified starting memory number.



TDA Setup screen.

Time Delay Analyzer Setup Screen and Controls:

The TDA function includes a setup menu screen and a test menu screen. When you enter the TDA function, the Setup screen is displayed first. This screen has the fields that you use to setup

the TDA test. You can switch directly between the Setup Screen and the Run Screen within the Setup/Run field. All setup screen fields are described below.

TDA Setup Screen and Control Fields:

1. **Test Function** – identifies the current test function TDA (Time Delay Analysis).
2. **Setup/Run** – switches between the TDA Setup Screen and the TDA Run screen. The TDA Setup Screen provides selection for configuring the TDA test. The TDA Run screen provides the test results.
- 3a. **Initial Delay of Distance (Value)** – Enter the speaker to microphone distance so that the TDA function can compute T_s and offset the input filter correctly with respect to the sweep generator. Selects distance in 0.1 increments from 0 ft. to 999.9 ft.
- 3b. **Initial Delay or Distance (Units)** – selects the unit of measurement used to set speaker distance either feet (ft) or meters (m). Select between **m** (meters) or **ft** (feet).
4. **Sweep Time** – Sets the time for the generator to sweep the TDA through the range of 20 Hz to 20 kHz. Variable from 7-30 seconds.
5. **Sweep Delta** – This field sets the additional delay time between the generator sweep and the filter sweep for each consecutive sweep. Note that, if you are running a single sweep, this field has no effect.
6. **Number of Sweeps** – Specifies how many subsequent sweeps are performed ranging from 1 to 20.
7. **Filter Q** – The Filter Q field controls the steepness or sharpness of the band-pass filter skirts ranging from 2 – 50.
8. **Space Equivalent Bandwidth (S.E.B)** – This field is not selectable.
9. **Space Equivalent Values in ft. and mS** – Indicates S.E.B. values computed from the Sweep Time and Filter Q selections. Values describe a distance and/or time in which reflected sound is significantly reduced in level 3 dB.

Initial Delay – Enter the speaker to microphone distance so that the TDA function can compute T_s and offset the input filter correctly with respect to the sweep generator. However, this delay value is not as critical as it is with other time delay analysis methods, since we are sweeping an actual filter, rather than deriving the response through convolution of an impulse response, or with a dual FFT transfer function. These other methods require very accurate T_s measurements. With the SP395A TDA method, an incorrect T_s time simply results in a graph that is shifted slightly on the frequency axis.

You can simply measure the distance with a tape measure (in feet or meters), or you can use the Energy Time Graph to measure the initial delay time.

Sweep Time – Sets the time for a single sweep, from 7 seconds to 30 seconds. A shorter sweep time completes faster, but a longer sweep time gives more low frequency energy, and more accurate results. Sweep time interacts with filter Q to determine the Space Equivalent Bandwidth. See below.

Sweep Delta – This field sets the additional delay time between the generator sweep and the filter sweep for each consecutive sweep. Note that, if you are running a single sweep, this field has no effect.

Number of Sweeps – You can set the number of sweeps from 1 to 20. Normally, you run one sweep if you are testing a transducer, and you run multiple sweeps if you are testing room decay and wish to create a set of curves.

For example, if you want to create a waterfall graphs to show the decay pattern for a room, you may want 10 sweeps, spaced out in 50 mS steps, to examine the first 500 mS of sound decay in the room. In that case, you would select 50 in the Sweep Delta field and 10 in the Sweeps field. If, for example, your room has a long decay time, you may want to examine a longer decay time in the room. You could create 10 sweeps that cover 4 seconds by selecting a Sweep Delta of 250 mS. Each sweep would then have an additional 250 mS delay from the previous sweep.

Filter Q – The Filter Q field controls the steepness of the band-pass filter. A higher Q results in a filter with steeper slopes. In most cases, you will want to use a high Q value. A lower Q might be used to create a smoother curve, to see the big picture more easily. However, more information is shown with a higher Q filter.

Note also that the SP395A TDA function uses a constant-Q filter. Other TDA platforms work with the equivalent of a constant-frequency filter. Constant Q can result in fewer problems with comb-filtering.

Space Equivalent Bandwidth – It is useful to be able to control the distance beyond which TDA will reject input. In some systems this is called “time windowing,” referring to the time that it takes reflections to bounce off room surfaces. In TDA, we define a distance, called the “space equivalent bandwidth” (SEB). This distance is defined as the distance beyond which input is attenuated by at least 3 dB. This distance is a function of sweep speed and filter Q.

For example, with a Q of 36 and a sweep time of 8 seconds, the SEB is 7.3 feet. Thus, if your speaker is at least 7.3 feet from any other surface, the test results will effectively remove the effects of the room from the measurement. In fact, since the filter decays very rapidly, in many cases you will get good results at up to twice the SEB.

Also, the shorter the SEB distance, the more the rejection of spurious noise.

Space Equivalent Bandwidth is computed on the Setup screen and changes as you change the Sweep Time and Filter Q.

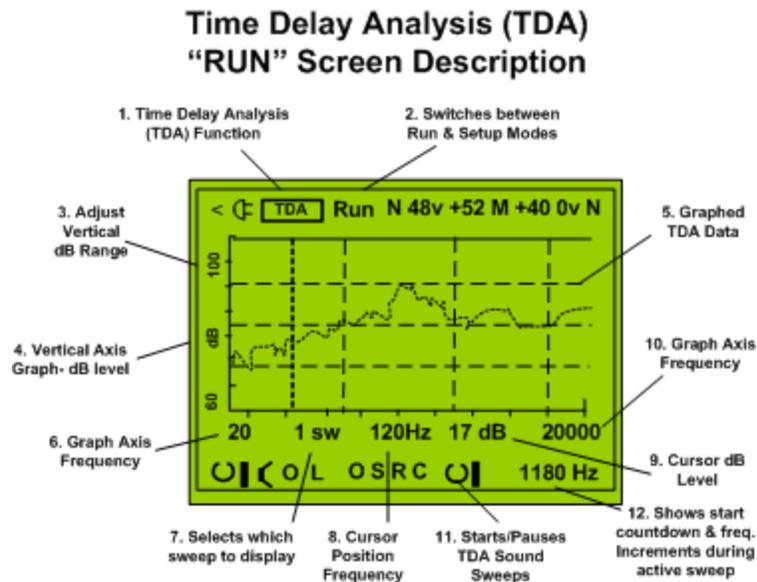
The formula for SEB is as follows:

$$\text{SEB} = \text{B} * [\text{c} / (\text{df}/\text{dt})]$$

Where B is the filter bandwidth in Hz, c is the speed of sound, and df/dt is the sweep rate of the filter in units of Hz/sec. The units of SEB are the same as the distance units used in the speed of sound.

Time Delay Analysis - TDA Run Screen

The TDA function includes a Setup menu screen and a Run menu screen. When you enter the TDA function, the Setup screen is displayed first automatically. After you have made the proper TDA setup, switch to the Run screen. You can switch directly between the Setup Screen and the Run Screen within the Setup/Run field. To Run the TDA test, click on the word "Setup." The Run screen displays the sweep graph and has the fields that you use to run the TDA test, make measurements, and store the graphs to memory. All Run screen fields and control fields are described below.



Time Delay Analysis (TDA) run or test screen.

TDA Run Screen and Control Fields:

1. **Test Function** – identifies the current test function TDA (Time Delay Analysis).
2. **Setup/Run** – switches between the TDA Setup Screen and the TDA Run screen. The TDA Setup Screen provides selection for configuring the TDA test. The TDA Run screen provides the test results.
3. **Graph Window** – This field selects the upper limit or maximum level of the graph establishing the vertical graph range.
4. **Vertical Scale dB** – Not selectable. Indicates 40 dB range of measurement graph.
5. **TDA Measurement Graph** – test data graphed 1/12 octave vs. dB.
6. **Graph x axis scale** – Indicates the lower limit or minimum frequency range of the graph. Not selectable or variable.
7. **Sweep Number** – This field selects which sweep to display from the current sweep set. The number of sweeps is determined on the Setup page.
8. **Cursor Position** – Selects the cursor for measurement and moves it to the desired frequency on the graph for level analysis.
9. **Cursor Level** – Indicates the dB level at the selected cursor position on the graph.
10. **Graph x axis scale** – Indicates the upper limit or maximum frequency range of the graph. Not selectable or variable.
11. **Run/Stop** – To start the TDA test, click the generator On/Off icon. To cancel a running test, click the icon again.
12. **Audio Sweep Countdown/Frequency Information Field** – This field shows the countdown before a sweep begins and the generator frequency as a sweep is running. After the sweeps are completed, it shows the millisecond offset of the currently displayed sweep.

How to Use TDA to Measure Speaker Response

1. **Connect the output.** Connect the output of the SP395A to an amplifier and to the speaker that you wish to test. It is advisable to have a gain control at or before the amplifier, since the only other way to change the speaker level during the TDA testing is to switch to the signal generator function and set the level from there.
2. **Set up the speaker.** Set the speaker at least 4 feet from any surface.
3. **Set up the microphone.** Mount the microphone on a mic stand and point it at the speaker. The microphone should be as far as possible from any solid surface, to minimize reflections. The TDA filters will help eliminate reflections, but they will work better given better input.
4. **Set the Initial Delay.** Set this field to the speaker to microphone distance. Determine the speaker to microphone distance either by measuring, or by running the Energy Time Graph.
5. **Set the Sweep Time.** Start with 8 seconds. This is a good trade-off between shorter SEB and more low frequency energy.
6. **Set the Sweep Delta.** This field has no effect since we are running only one sweep.
7. **Set the number of sweeps.** Set this field to 1.
8. **Set the Filter Q.** Set this field to 36.
9. **Run the test.** Click on the generator On/Off icon on the bottom toolbar to begin the test.
10. **Read the finished graph.** Check the graph for level. You can read the results directly on the screen using the cursor. If the warning “Overload” appears, reduce the speaker volume and run the test again. If you get “Low input” or the level is below 60 dB or so, increase the speaker volume and run the test again.

How to Use TDA to Measure Room Response

1. **Connect the output.** Connect the output of the SP395A to an amplifier and speaker. It is advisable to have a gain control at or before the amplifier, since the only other way to change the speaker level is to switch to the signal generator function and set the level from there.
2. **Set up the microphone.** Mount the microphone on a mic stand and point it at the speaker.
3. **Set the Initial Delay.** Set this field to the speaker to microphone distance. Determine the speaker to microphone distance either by measuring, or by running the Energy Time Graph.
4. **Set the Sweep Time.** Start with 10 seconds.
5. **Set the Sweep Delta.** Choose the Sweep Delta and the number of sweeps to cover the room time decay period that you wish to examine. For example, if the room decay time is just under one second, and you would like to see that decay time split into ten sweeps, choose a Sweep Delta of $1/10^{\text{th}}$ second (100 mS).
6. **Set the number of sweeps.** Set this field to the number of sweeps that you want to run, as determined in the step above.
7. **Set the Filter Q.** Set this field to 17. This will result in a smoother set of graphs.
8. **Run the test.** Click on the generator On/Off icon on the bottom toolbar to begin the test.
9. **Read the results.** Check the graph for level. You can read the results directly on the screen using the cursor. If the warning “Overload” appears, stop the test, reduce the speaker volume, and run the test again. If you get “Low input” or the level is below 60dB or so after one sweep, stop the test, increase the speaker volume, and run the test again. When the test is complete, you can use the sweep number field to page through the stored graphs, to observe the pattern of sound decay in the room.

How to Use TDA to Measure Microphone Response

1. **Connect the output.** Connect the output of the SP395A to an amplifier and to the speaker. It is advisable to have a gain control at or before the amplifier, since the only other way to control the speaker level is to switch to the signal generator function and set the level from there.
2. **Set up the microphone.** Mount the microphone on a mic stand and point it at the speaker. Determine the speaker to microphone distance, either by measuring, or by running the Energy Time Graph. Enter this number in the Initial Delay field in the TDA setup menu.
3. **Set the sweep time.** Start with 20 seconds. Use a longer time for the most accurate results, or a shorter time for a quick reading.
4. **Set the filter Q.** Set this field to 50.
5. **Set the number of sweeps.** Set this field to 1.
6. **Run the test.** Click on the generator On/Off icon on the bottom toolbar to begin the tests
7. **Read the results.** Check the graph for level. If you get the word “Overload”, reduce the speaker volume. If you get “Low input” or the level is below 60dB or so, raise the speaker volume. You can read the results directly on the screen using the cursor.

Storing TDA Sweeps into Memories

The TDA function can run up to 20 sweeps in a group and store them to the internal SP395A non-volatile memory locations. Stored sweeps can then be recalled and viewed.

After running one or more sweeps, you can store them to memory. To do this, select the starting memory location on the bottom toolbar, and click S (Store). Note that when the memory location is selected, the type of memory stored (or “---” if empty) appears on the bottom of the graph. As many sweeps as are part of this set are stored in consecutive memory locations. Note that if you leave the TDA function without storing the sweeps to memory, the measurement data is lost.

Recalling TDA Sweeps from Memories

To recall a set of TDA sweeps from memory, to view them on the SP395A, select the first memory location in the series and click the R (Recall) icon. If the memory number selected is the first (or only) memory in a TDA series, the series will be read into working memory and you can then click on the Sweep number field (Information Field) to scroll through the sweeps.

Note that when the memory number field is selected, the type of memory stored (or “---” if empty) will appear on the bottom of the graph. Only “TDA” type memories can be recalled to the TDA function. If you see “Tm,” that means that the selected memory is not the first sweep,

but one of the later sweeps, in a TDA set of sweeps.

TDA – TerraLink Software

The SP395A’s TDA function is designed to work hand-in-hand with TerraLink PC software. On the SP395A screen, you can see only one sweep graph at a time. To view a set of room response sweeps, you page back and forth through the sweeps and watch the decay pattern, to see how the decay varies by frequency as you move through time. Although this works, seeing multiple graphs in 3D gives a much more intuitive and quicker understanding.

To use TerraLink for TDA analysis, just perform the sweeps in the SP395A, store them to non-volatile memory, and transfer them to TerraLink. From TerraLink, you can bring up a set of TDA sweeps as a three-dimensional graph. The surface of the graph represents the decay curve for the room. You can rotate this graph and view it from the top, or look at in perspective. You can also switch between wire-frame view and solid view.

As with other TerraLink graphs, you can store TDA sweeps in project files on the computer hard drive, to keep an archive of TDA information, and to print or share with others. You can also define a reference curve in TerraLink, and compare a TDA curve against the reference. This is particularly useful for doing production line testing of transducers, such as speakers and microphones.

Noise Tools

The Noise Tools menu contains functions that measure speech intelligibility, room noise levels and transmission loss. The Noise Tools menu includes:



ALCONS
RaSTI
STI-PA
Noise Curves
Transmission Loss
Audio Stethoscope

The Noise Tools menu.

The ALCONS, RaSTI, AND STI-PA are speech intelligibility tests that gauge how discernable speech is at a particular position in a room in respect to the speaker. Noise Curves measure noise in a room characterizing it to a single numeric value taking into account how the human ear hears. Transmission Loss measures sound reduction (transmission loss) through a partition, wall, door etc. The Audio Stethoscope combines the microphone input and sensitive audio preamplifier with the audio monitoring headphones to form an audio stethoscope. Weighted filters and octave or 1/3 octave band filters may be selected.

Speech Intelligibility - ALCONS

ALCONS is a speech intelligibility test. Percent Articulation Loss of Consonants (%ALCONS) is a speech intelligibility measure that is intended to mimic the loss of consonants of speech, which has been shown to be related to loss of intelligibility of speech. It is based on reverberation time (RT60) and room sound level measurements.

Two different formulas are used, depending on whether the distance from speaker to listener is less than or greater than the critical distance.

Critical distance is defined as:

$$D_c = 0.2 * \text{SQRT}(V/T).$$

At less than the critical distance, this formula is used:

$$\text{ALCONS} = (200 * D^2 * T^2) / (V + a)$$

Beyond the critical distance, this formula is used:

$$\text{ALCONS} = (9 * T) + a$$

Where,

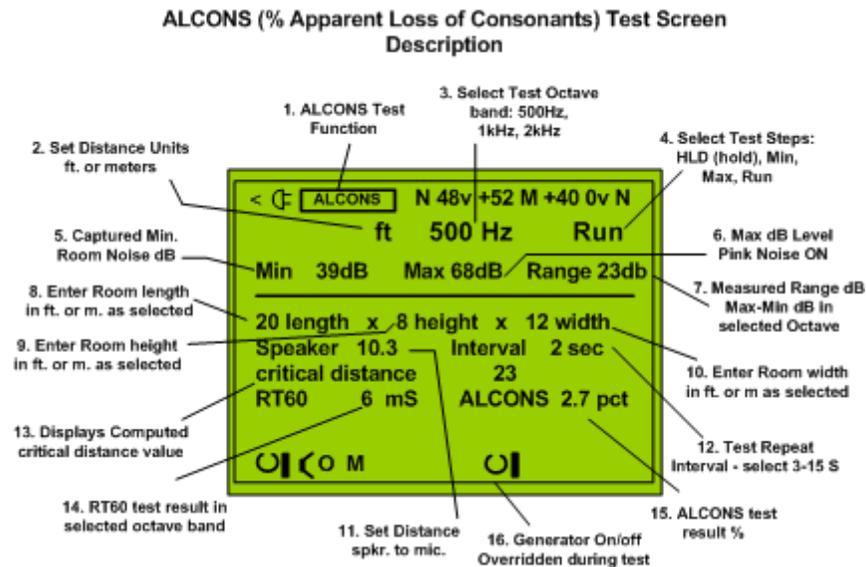
T = RT60 time, in seconds

V = Volume of the room, in cubic meters

D = Speaker to Listener distance

a = Correction constant, typically 1.5%.

In operation, an amplifier and loudspeaker are positioned at the “speaker” position, and the SP395A microphone is positioned at the “listener” position. The % ALCONS function measures RT60 in a speech octave band, and computes the % ALCONS.



The ALCONS Speech Intelligibility test display screen.

The ALCONS test screen & control fields:

- Test Function** – Identifies the current test function as ALCONS.
- Distance Units – Speaker Distance (Units)** – selects the unit of measurement used to set speaker distance from speaker to listener (microphone) in either feet (ft) or meters (m). To set the speaker distance units, turn the control knob to highlight the Distance Units field, and click to select **m** (meters) or **ft** (feet).
- Octave Band** – Selects 500 Hz, 1 kHz or 2 kHz octave band. Selects the audio octave band in which the ALCONS test is performed. The audio output from the generator matches the band selected.
- Testing Control/Steps** – Increments the test function through 4 positions or steps including Hold (HLD), Min, Max and Run to complete the ALCONS test.
 - HLD:** Stop the test & holds the test results for analysis.
 - Min:** Sets Minimum Room Noise – Highlight the Test Control field and select "Min." The SP395A reads the average background noise level in the room during this step. If this level is greater than 50 dB, you should attempt to decrease the ambient noise for best test results and better room acoustics.
 - Max:** Sets maximum room sound level using pink noise – This is the maximum sound energy in the room with the octave pink noise test signal applied. Adjust the amplifier's volume and/or SP395A generator output to increase the sound level to 80 – 90 dB. It is desirable to get the noise level considerably higher than the Min room level dB for improved decay range and test accuracy. This can be challenging as low frequency room noise tends to limit the decay range in the lower bands. You may want to wear ear plugs to enable higher noise settings. To store the Max Noise level, set the Test Control field to "Max" and click the Test Control field to "Run"
 - Run:** Click the Test Control field to display "Run". The pink noise generator will cycle on and off as readings are taken. The "Decay Time" field displays the actual time in milliseconds for the sound field to decay by the Decay range amount. The RT60 time field shows the extrapolated time for the sound to decay in the room, if a full 60dB decay range were available.
- Minimum Room Noise Level** – The ambient room noise level. The minimum room noise level must be at least 30 dB lower (indicated by a number display in the Decay Range

- field) than the Max Noise Level reading. If the Min Room level is greater than 50 dB, you need to lower the ambient noise for best test results. To store the Min Room level, set the Test Control field to "**Set Min**," then click the Test Control field.
6. **Maximum Room Noise Level** – This is the maximum sound energy in the room with the octave band pink noise test signal applied. Adjust the amplifier's volume and/or SP395A generator output to increase the sound level to 80-90 dB to achieve the highest possible decay range. *You may want to wear ear plugs.*
 7. **Decay Range** – The dB difference between the Max Noise and Min Room levels for the selected octave band. This is the range of the RT60 test. Try to achieve the highest range possible by reducing room noise or increasing the octave band pink noise. A range of 30dB or higher is required to perform the ALCOM calculation.
 8. **Room Dimension length** – Enter length of room in which the measurement is to be taken.
 9. **Room Dimension height** – Enter ceiling height of the room in ft. or meters.
 10. **Room Dimension width** – Enter width of the room in ft. or meters.
 11. **Sound Travel Distance Spkr to Mic** – Enter the distance from the speaker to the listener or microphone.
 12. **Test Interval** – Time interval of cycling octave band noise on and off and updating measurement results. The test continues to repeat in the "RUN" mode.
 13. **Critical Distance Calculation** – Determined by mathematical formula based upon room dimensions, speaker to listener distance, and RT60 results.
 14. **RT60 Measurement Readout** – The RT60 calculated measurement of the room. Indicates the extrapolated time for the sound to decay 60 dB (RT60) in the measured octave band.
 15. **ALCONS Measurement Readout** – ALCON test measurement readout in %.
 16. **Generator** – On/off function overridden during ALCONS test. Level Output is adjustable.

ALCONS Function Operation

1. **Connect output.** Connect an amplified speaker to any SP395A output. Put this at the "speaker" location.
2. **Position microphone.** Set up the SP395A microphone at the "listener" position. Use a stand, if possible.
3. **Units.** Select the unit of measurement used to specify the speaker to microphone (ft or m).
4. **Octave Band.** Select the octave band. 2 kHz is recommended, although 500Hz and 1 kHz are also available.
5. **Room Dimensions.** Enter the room dimensions (length x height x width) in the units previously selected.
6. **Speaker Distance.** Enter the distance between the speaker and the microphone. Needed for critical distance and ALCONS calculations.
7. **Time Interval.** Enter the RT60 pulse time interval. This is related to the RT60 time. Ensure that the interval time is greater than the expected or measured RT60 time for the room, to allow the pink noise pulse to decay as much as possible.
8. **Mode.** Now, move to the "Hld" field, and change this to "Min". Allow several seconds for the SP395A to measure the noise floor in the room in the octave band selected.
9. **Generator Level.** Click the "Min" field to "Max". The generator is now turned on. Increase the volume level in the room using the generator level icon or the amplifier volume control to get as much level as possible without overloading the amplified speaker. The available decay range is computed and displayed.

Note: You must have at least 30dB of range available to compute the RT60 and ALCON % for the room.

10. **Run test.** Change the “Max” field to “Run” to begin the test. The Critical Distance is computed, and the correct %ALCONS formula is selected based on whether the microphone is within the Critical Distance. The resulting loss of consonants is displayed as a percent.

Speech Intelligibility Results

Use this table as a guideline for evaluating the results of the %ALCONS and RaSTI tests.

%Alcons	RaSTI	Rating
57.70	0.20	Bad
44.00	0.25	Bad
33.56	0.30	Poor
25.59	0.35	Poor
19.52	0.40	Poor
14.89	0.45	Fair
11.35	0.50	Fair
6.60	0.60	Fair
5.04	0.65	Good
3.84	0.70	Good
2.93	0.75	Good
2.23	0.80	Excellent
1.70	0.85	Excellent
1.30	0.90	Excellent
0.99	0.95	Excellent
0.76	1.00	Excellent

Speech Intelligibility – RASTI

Rapid Speech Transmission Index (RaSTI) is one of several speech intelligibility testing methods. RaSTI calculates an index value (0.00 to 1.00, with 1.00 being perfect intelligibility) by evaluating the loss of modulation, known as the Modulation Transfer Function (MTF), of special waveforms.

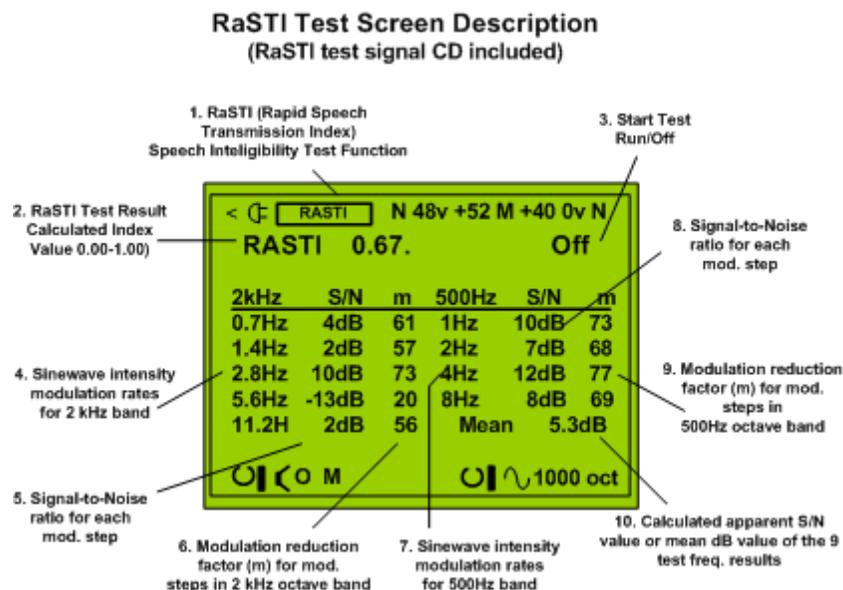
The RaSTI waveforms are a subset of the Speech Transmission Index (STI) waveforms, which are based on octave-band limited pink noise, covering 7 octave bands from 125Hz to 8 kHz, each modulated 100% in amplitude (loudness) by sine waves, in 14 different frequencies ranging from 0.63Hz to 12.5Hz, for a total of 98 waveforms.

STI is calculated by measuring the amount of modulation that is lost (MTF) in each of the 98 waveforms, and converting to an “apparent” signal to noise ratio in dB, limiting these to +/- 15dB, and averaging and normalizing these values.

RaSTI simplifies the above method by selecting only 9 waveforms total, using the 500Hz and 2kHz octave bands. RaSTI measurements use a transmitter to output a composite signal of all 9 modulation waveforms, and a separate receiver which analyzes the composite waveform. The SP395A analyzes the results of a standard RASTI test signal, which is supplied on a CD. By using a separate test signal on a CD, the test signal can be routed to a system that has a number of speakers such as an emergency evacuation system, and the SP395A can be carried around to measure each speaker individually.

In operation, an amplifier and speaker are set up at the “speaker” position, and the SP395A microphone is set up at the “listener” position. The RaSTI test waveform is played from a CD or other source, and the test is performed by the SP395A RASTI analyzing function.

After 10 seconds of analysis, the results are shown on the screen, and the RaSTI value is read directly. See the table, below the ALCONS function description, for a summary of the results of these tests.



RaSTI speech intelligibility test screen and control fields.

RaSTI test screen and control fields.

1. **Test Function** – Identifies RASTI as the selected function and measurement screen.
2. **RASTI Test Value** – This is the RASTI measurement or index value after a valid test is performed. The readout range is .00 to 1.
3. **Off/Run** – This field begins the test by selecting “RUN” and turns the test function off by selecting “OFF.”
4. **Column 2 kHz Intensity Modulation Rates** – Five Modulation signals of the 2 kHz test waveform used by RASTI to measure modulation reduction factor (m) and compute S/N values.
5. **S/N Ratio Measurements** – values measured by the RASTI test for each of the modulation signals.
6. **Modulation Reduction Factor (m)** – values measured by the RASTI test for each of the modulation signals.

7. **Column 500Hz Intensity Modulation Rates** – Four Modulation signals of the 500Hz test waveform used by RASTI to measure modulation reduction factor (m) and compute S/N values.
8. **S/N Ratio Measurements** – values measured/computed by the RASTI test for each of the modulation signals.
9. **Modulation Reduction Factor (m)** – values measured by the RASTI test for each of the modulation signals
10. **Mean Value** – Mean value of the apparent S/N ratio values measured/computed within the RASTI test.

RaSTI Function Operation

1. **Apply a test signal to the system.** Connect an amplified speaker to a CD player and insert the RASTI test signal CD. The location of the speaker is the “speaker” location.
2. **Microphone** - Set up the SP395A microphone at the “listener” position. Use a stand, if possible.
3. **Level** - Using the SPL meter function, set the level of the test signal as measured by the SP395A to the “nominal” listening level. In most cases this will be 75-85dB SPL.
4. **RASTI** - Select the RASTI function on the SP395A.
5. **Begin Test** Click on the ON/OFF field changing the “Off” to “Run”. This begins the test.
6. **Wait For test to complete** - Allow at least 12 seconds for the analysis to be completed.
7. **RASTI Result** - read the RASTI value on the bottom of the screen.

Speech Intelligibility Results

Use this table as a guideline for evaluating the results of the %ALCONS and RaSTI tests.

%Alcons	RaSTI	Rating
57.70	0.20	Bad
44.00	0.25	Bad
33.56	0.30	Poor
25.59	0.35	Poor
19.52	0.40	Poor
14.89	0.45	Fair
11.35	0.50	Fair
6.60	0.60	Fair
5.04	0.65	Good
3.84	0.70	Good
2.93	0.75	Good
2.23	0.80	Excellent
1.70	0.85	Excellent
1.30	0.90	Excellent
0.99	0.95	Excellent
0.76	1.00	Excellent

RASTI Caveats

RaSTI can be used to evaluate intelligibility, and is considered superior to some other methods, particularly ALCONS. However, it is possible to get false results. To get the best possible results, watch out for these problems:

Out of band noise. Since RaSTI looks only at the 500Hz and 2000Hz octave bands, it will not take into account noise outside of these bands. For example, imagine a very loud signal at 4 kHz. This could cause terrible intelligibility yet show a good RaSTI score.

Non-linear systems. Driving a system into distortion, or running the signal through compressors or other non-linear systems can cause false readings.

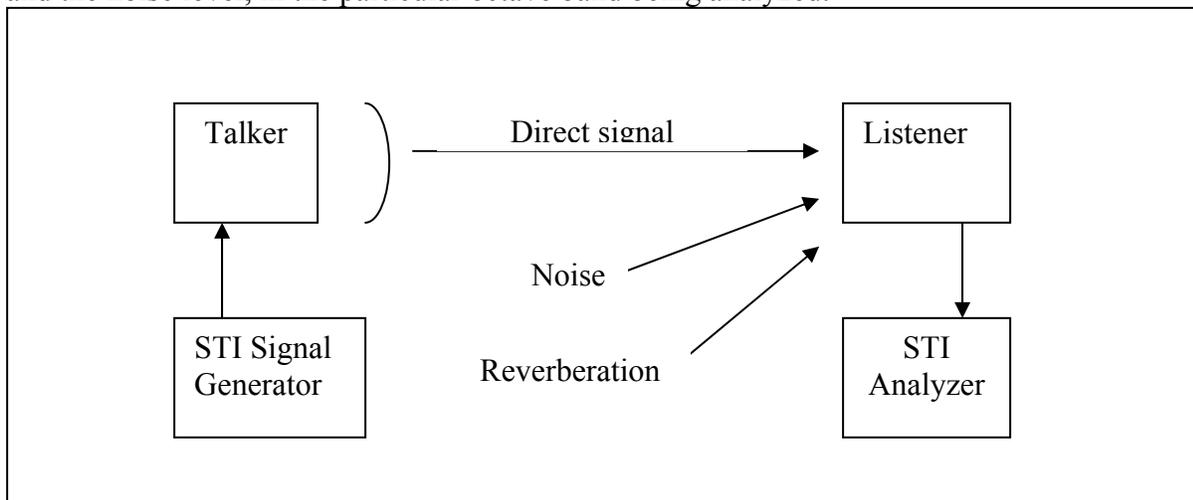
Bad “talker” level. If the talker level is not within normal limits, very low room noise can cause a bad score, or a very loud test signal can mask room noise. Set the talker (test signal) level at expected speech or system signal levels.

If these cases are avoided, the results will correlate well with STI.

Speech Intelligibility – STI-PA

STIPA is a speech intelligibility measurement derived from the more extensive STI standard. STI specifies 98 data points, each an octave band of pink noise modulated by a slow sine wave envelope. Octave bands from 125Hz to 8000Hz are specified, and for each octave band, up to 9 different sine wave modulations. For a single data point, imagine generating octave band pink noise and changing the volume from off to maximum to follow a slow sine wave. The combined STI signal for a set of points sounds like a steam locomotive.

For each data point, the amount of the modulation that is lost in an acoustic space is measured, either directly or indirectly, and from this “loss of modulation” the speech intelligibility is derived. Things that can affect loss of modulation are primarily reverberation time in the room and the noise level, in the particular octave band being analyzed.

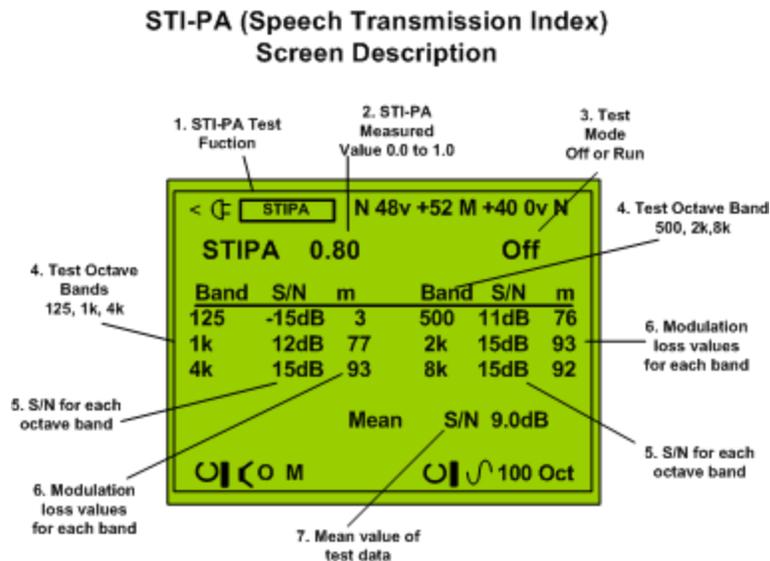


After the modulation loss numbers have been computed for each data point, they are combined mathematically to arrive at a single value. This value is referred to as “STI” and ranges from 0.0 (no speech intelligibility) to 1.0 (best possible speech intelligibility).

STI-PA Description

STI-PA is a subset of STI that uses 9 data points, spanning the 125 Hz to 8000Hz octave bands, but using only one modulation speed for each data point. STI-PA has been shown statistically to correlate well with STI, even though it has 1/10th the data points.

The SP395A STI-PA implementation is unique, using an STI-PA excitation signal and proprietary DSP algorithms for determining loss of modulation. The STI-PA test signal is played from the supplied CD, and the signal is sampled for 15 seconds. The STI-PA value is calculated and shown on the SP395A test screen.



STIPA Measurement screen and control fields descriptions.

STIPA Test Screen and Control Fields:

1. **Test Function** – Identifies STIPA as the selected function and measurement screen.
2. **STIPA Test Value** – This is the STIPA measurement or index value after a valid test is performed. The readout range is .00 to 1.
3. **Off/Run** – This field begins the test by selecting “RUN” and turns the test function off by selecting “OFF.”
4. **Octave Test Bands** – Column listings of the octave bands within the CD test signal used by the STIPA test to measure modulation reduction factor (m) and compute S/N values.
5. **S/N Ratio Measurements** – values measured by the STI PA test for each of the modulation signals.
6. **Modulation Reduction Factor (m)** – values measured by the STIPA test for each of the bands.
7. **Mean Value** – Mean value of the apparent S/N ratio values measured/computed within the STIPA test.

STI-PA Function Operation

To use STI-PA, you need to play the STI-PA test signal through a speaker, and then use the SP395A microphone to measure the STI-PA signal in the room. If you are testing a specific installed system, such as a fire evacuation sound system, send the signal through the system amplifier and speaker.

- **Set up the connections.** Connect the STI-PA test signal to the speaker system. Play the test signal CD and adjust for appropriate level. In some cases, a test level SPL will be specified. Otherwise, typically a nominal level such as 70-80 dB SPL is used.
- **Start the test.** Click on the Run field to begin the test. Wait for 15 seconds for the test to complete. While the test is running, the test microphone should remain in the same position, and the noise level in the room should remain stable.
- **Read the results.** After the test is complete, you will see a set of signal to noise (s/n) and m values. Also, the final STI value is shown. If a condition was detected that prevents getting an accurate value, you may see the message “low input” or “overload” on the screen, and the data values are not shown.

Noise Curves

All rooms have noise: sound that is present without any energy applied by the audio system. Noise affects the dynamic range of the reproduced audio. Noise levels that are too high interfere with our ability to hear sounds, especially softer sounds, and may require extra volume from the amplifier.

How our ears respond to noise depends on its frequency and loudness. Noise measurements follow ANSI standard S12.2-1995 Noise Criteria (NC) weighting curves (see Appendix C) that are weighted for the frequencies and sound pressure levels where our ears are most sensitive. NC curves specify equal-loudness levels of noise over the audio spectrum. They are numbered for easy reference, with lower numbers corresponding to less noise. The NC rating of a room is the number of the lowest curve that is not exceeded by any of the background noise.

There are actually several standards used to measure noise curves. The SP395A supports three of these standards. SP395A Noise Curve tests are based on standard NC or NCB (balanced noise curves), RC (Room Criteria), and PNC (preferred noise curve) ANSI tests.

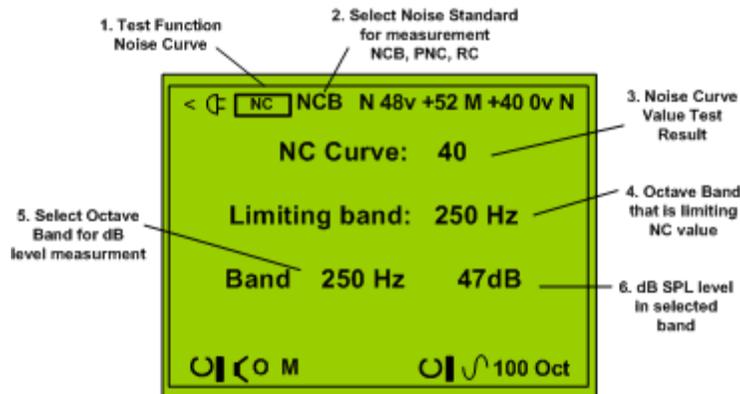
The NC (also known as NCB, or balanced noise curves) value is based on ANSI standard S12.2-1995 (ASA 115-1995) Criteria for Evaluating Room Noise, developed first by Beranek in 1957. It uses ANSI Class 1 octave-band filters to divide the sound spectrum into 9 octave bands, measuring the SPL level of each band. Then, the results are compared to a table of values defined in the specification. The table lists noise criteria numbers by row, with each row having values defined for each octave band. The NC band is defined as the lowest band number for which none of the octave-band SPL values for the row are exceeded.

A similar computation is RC (Room Criteria), based on the same ANSI standard. For this measurement, a different table of values is used, based on a study by Blazier in 1981, primarily to rate noise in offices. This system was not intended for use in very quiet spaces below RC 25. Typically, RC is used in evaluating HVAC systems, while NC is used for more general purposes, including motion picture theater spaces and offices.

An additional method is to compute PNC (preferred noise criteria). This computation was derived by Beranek, Blazier, and Figwer in 1971, but is not based on an ANSI standard, and has not been widely accepted.

The SP395A Noise Criteria Test automatically measures the SPL noise levels at the standard octave band frequencies and compares them to ANSI standard NC curves (Appendix C), RC, or PNC standards. The display shows the NC rating, the weighted noise energy level in the individual octave bands, and the band that is limiting the NC rating. Reducing the noise in this band may allow you to achieve a lower NC rating. You should perform NC measurements under typical and worst case conditions, including having the room's ventilation system running and outside noises such as street traffic corresponding to the typical listening times.

NC Curve Screen Description



Noise Curve test screen and field descriptions.

Noise Curves Test Screen and Control Fields:

1. **Test Function** – Identifies Noise Curves (NC) as the selected function and measurement screen.
2. **Test Standard** – Selects the standard used for determining the noise curve value. Select from Balanced Noise Curves (NCB), Room Criteria (RC) or Preferred Noise Curve (PNC).
3. **Noise Curve Result** – Measured Noise Curve value test result.
4. **Limiting Octave Band** – Indicates the octave band in which the level of noise is exceeding the standard noise curve. Decreasing the room noise in this band improves the NC value.
5. **Select Octave Band** – Selects one of the nine octave bands used in the NC Curve tests for measurement.
6. **Octave Band SPL Level** – Displays the SPL level of the selected octave band.

Noise Curves Operation

1. **Connect the microphone** to the SP395A and position it in the center of the listening position. The microphone is the only input that can be used for the noise criteria test.
2. **Select the Noise Criteria function.** Select the standard used for NC testing.
3. **Quiet the room and read the NC background noise rating.** The value in the NC Curve field is the room's background noise rating.
4. **Read the octave band that is limiting the NC rating.** The number in the Limiting Band field indicates the octave band that contains the noise level that is limiting the NC rating. If you can lower the noise level in this band, the NC rating will drop to a lower level.
5. **View the noise SPL level for any octave band.** The number in the Octave Band field indicates the noise level in that particular band. Use this function to view the noise level in the limiting band as you track down possible sources of that noise. Also use this function to plot the noise levels against the NC curve chart in Appendix C, to check whether there are equivalent levels of noise in multiple octave bands or primarily in one band.

Transmission Loss

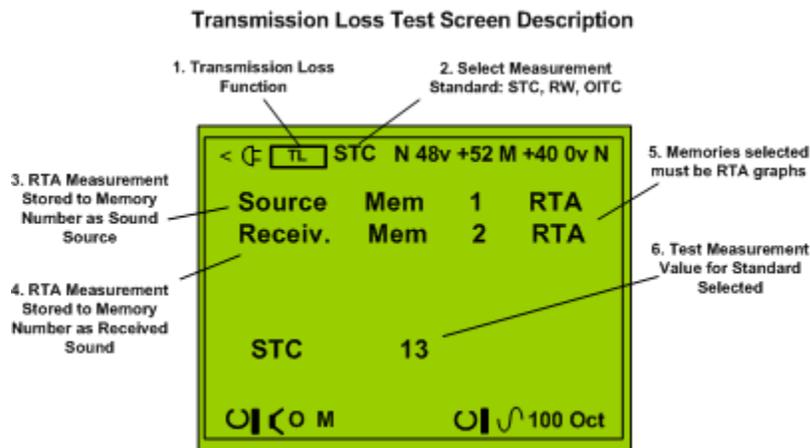
Transmission Loss is used to measure the amount of sound isolation that a partition provides. Typically, a noise source is placed on one side of the partition, and the room noise level is measured in 1/3 octave bands, sometimes at more than one point. Then, the measuring device is moved to the other side of the partition (presumably in another room) and the same measurement is taken. Then, the transmission loss (TL) at each 1/3 octave bands is computed, and a formula is applied to derive a numeric number representing the relative noise reduction of the partition. The idea is that this number can be used to effectively compute the amount of reduction in sound level that a partition (wall) will provide in actual use.

There are several ways of measuring transmission loss. One key issue is whether or not the measurement is being done under laboratory conditions, with carefully controlled room reverb time and control of sound leaking around the partition, or whether the measurements are being taken in the field under minimal control characteristics.

Several standards apply here, including ASTM 90, which talks about how to measure the transmission loss data, ASTM 413 (or ISO 717-1), to define a laboratory measurement, and ASTM 336, which takes the lab measurement and applies it to field work.

The SP395A calculates laboratory values including STC, RW, and OITC. STC (Sound Transmission Class) is the ASTM (US) measurement, RW is based on the European standard, and OITC (Outdoor-Indoor Transmission Class) is based on an A-weighting curve, and is considered by some to be more modern.

Please note that even if you are doing an "STC" measurement, technically the measurement is an NIC measurement since it is being done in the field.



The Transmission Loss test screen and control fields.

The Transmission Loss test screen and control fields:

1. **Test Function** – Identifies Transmission Loss (TL) as the selected function and measurement screen.
2. **Test Standard** – Selects a standard used for determining a Transmission Loss value. Select from Balanced Noise Curves (NCB), Room Criteria Outdoor/Indoor Transmission Class(OITC) or Sound Transmission Class (STC)

3. **Source Memory** – Selects the memory location of the RTA screen captured on the sound source side of the partition.
4. **Receive Memory** – Selects the SP395A memory location of the RTA test captured on the sound receive side of the partition.
5. **Memory Type** – Indicates if the selected memory originated from the SP395A RTA, FFT, Sweep or other functions. The Transmission Loss test requires RTA memories for proper analysis.
6. **Test Measurement Value** – Indicates the value of the test result for the standard measurement selected.

Transmission Loss test operation

The Transmission Loss test requires performing the RTA test on both the signal source side and receiving side of the partition. The RTA tests should be stored to memory locations using the bottom toolbar of the SP395A. Then switch to the Transmission Loss Test and enter the stored memories containing the RTA test results both on the signal source side of the partition and receive side of the partition.

To perform the Transmission Loss Test:

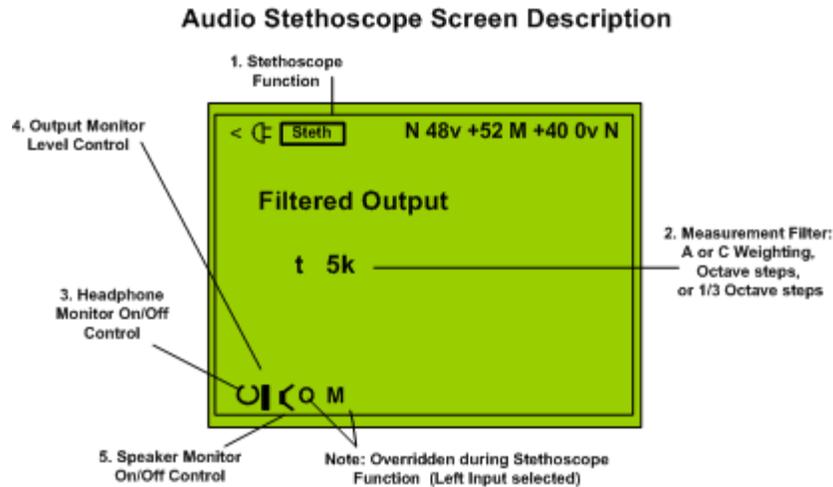
1. **Setup.** Set up the source room with an amplified speaker. Place the microphone approximately 1 meter from the partition.
2. **Apply Noise Source.** Generate a sufficient sound level (pink noise) in the room.
3. **Saving the source room RTA.** Select the RTA test function on the SP395A. Setup for a 1/3 octave RTA. Store the RTA test results to a memory location using the bottom center toolbar. Note: Make sure that the averaging time on the RTA is slow enough that the display is not visibly bouncing around. If you wish, use Avg (Average) mode to average the results in multiple locations before storing the curve. Note the memory number for later entry into the Transmission Loss test screen. (Example: Memory 1)
4. **Save the receiving room RTA.** With the sound generator still running at the same level, move the microphone to the other side of the partition. Like before, store this 1/3 octave RTA into a memory. Note this memory number for later entry into the Transmission Loss test screen.
5. **Perform Transmission Loss Analysis.** (This part of the analysis can be done off-line, as it uses the memories stored above). Go to the Transmission Loss function and select the desired measurement, STC, RW, or OITC. Now select the source room RTA memory as Memory 1, and the Receiving room memory as Memory 2.
6. **Read the test result.** The Transmission Loss test result is displayed in the bottom center just above the bottom toolbar.

Note: (Refer to ASTM 90 standard for full details on transmission loss testing.)

Audio Stethoscope

The Audio Stethoscope combines the microphone input and sensitive audio preamplifier with the audio monitoring headphones to form an audio stethoscope. Weighted filters and octave or 1/3 octave band filters may be selected and headphones used to listen to the filtered signal

The Audio Stethoscope is a tool that can be used to aid critical listening so that you can find problems with noise isolation, or to listen to some sounds while filtering out others. You can use the Audio Stethoscope to listen to the sound leaking around doors or other partitions at specific frequencies, to find areas that need better isolation. Also, you can listen to the noise in a space by frequencies, to listen for problems with HVAC systems or other noise sources.



Stethoscope test screen and description of control fields.

Stethoscope test screen and control fields:

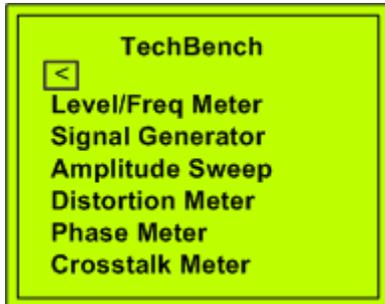
1. **Test Function** – Identifies Stethoscope (Steth) as the selected function and measurement screen.
2. **Measurement Filter** – Selects measurement filter applied to input from left channel. Includes selection of A weighted, C weighted, octave or sub-octave filters.
3. **Headphone On/Off** – Turns on or off audio from the left input to the headphone jack.
4. **Headphone/Speaker Monitor Level** – Increases or decreases volume out of the speaker or headphone.
5. **Speaker Monitor On/Off** – Turns on or off audio pickup from the left input to the internal speaker of the SP395A.
6. **Audio Monitor Source/Mode** – Overridden during Stethoscope function. Left input selected.

Stethoscope Operation

1. **Set up the microphone.** Connect the microphone to the left input. Turn on phantom power (if needed) and adjust the input gain as needed. Typically for full gain (+58) for most sensitivity.
2. **Plug in headphones.** Plug in any good-quality headphones into the 1/4" headphone jack. We recommend that you use closed-ear models to avoid feedback and to get the maximum isolation.
3. **Turn on the monitor output.** Using the bottom toolbar, turn on the monitor and adjust the volume as required.
4. **Select the filter.** Select the filter field to listen to the frequency ranges that you want to listen to. Note that whenever you change filters, the output is momentarily muted.

TechBench Menu

The TechBench menu contains test functions which are primarily oriented towards audio equipment testing and calibration. The TechBench menu includes the following tests:



Level/Freq. Meter
Signal Generator
Amplitude Sweep
Distortion Meter
Phase Meter
Crosstalk Meter

The TechBench menu showing the available test functions.

The SP395A assumes that tests of audio equipment involve standard line level audio inputs. Therefore, while the same left and right inputs are used on the right side panel of the instrument, line level audio is expected.

The SPL, Acoustics, TDA and Noise Tools functions, are designed for microphone inputs. These functions use the microphone SPL calibration stored within the SP395A. Electrical functions, such as the TechBench, Speaker, and Utilities functions expect line level source inputs. These functions use a dBu calibration within the SP395A. Keep this in mind to prevent potential confusion over measurement results. For example, if a 0 dBu electrical signal is plugged into the Sound Level Meter, you may see results like +150 dB SPL, depending on the sensitivity calibration that is stored. Conversely, if you are using a microphone with the Amplitude Sweep function, you will see very low results, perhaps in the -70 to -60 dBu range.

When performing TechBench measurements of audio signals you typically would not be using a microphone. Therefore, remove phantom power to the SP395A inputs by changing both the left and right phantom voltage from 48V to 0V as indicated in the top toolbar.

Use the auto-gain input feature of the SP395A as it provides the optimum range from most measurements. To select auto gain, change the M to an A in the top toolbar on the left and/or right channel.

All functions that display level data allow you to select the display units as dBu ($0 \text{ dBu} = 0.775 \text{ V rms}$, or the equivalent voltage level to dBm, referenced to a 600mw power level), dBV (where 0dBV is referenced to 0.0 Volts RMS), Volts average, Volts RMS or Volts Peak-to-Peak.

Level/Freq. Meter

The Level Meter measures and displays the Left and Right audio input levels simultaneously. You can use either XLR connectors or 1/4 inch connectors to the either or both of the audio inputs. The level meter measures input signals over a wide range from -120 dBu to +40 dBu. The resolution varies with the input range providing resolution to .001 uV.

Levels are metered using various display units including V rms, Vp-p, dBu, dBV or dBr.

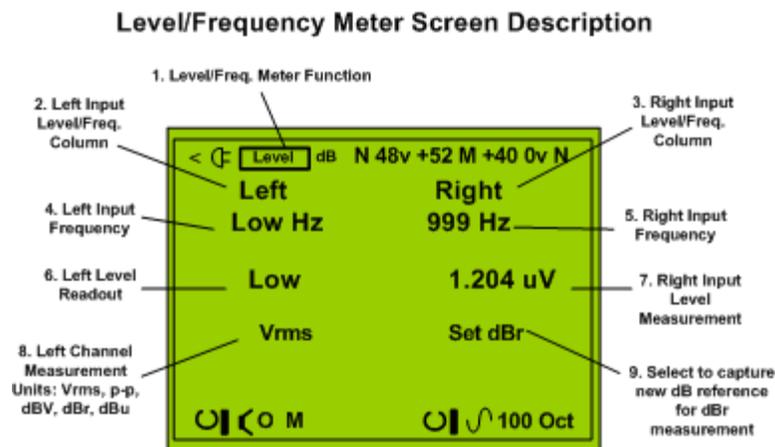
dBu is based upon 0 dBu being equivalent to .775 V rms.

dBm is based upon 0dBm being 600 mW of power.

0dBV is based upon 0 V as the reference level.

All measurements are performed in averaging mode, which is the same method used by the meters on most tape recorders, consoles, and outboard equipment, so the SoundPro readings conform to the readings of most recording studio signal level meters.

The Level/Freq. meter contains a frequency counter to measure the frequency of the incoming audio sine-wave signals. The frequency counter measures and displays the frequency of both the right and left audio channel input simultaneously. The frequency counter range is from 10 Hz to 32 kHz.



Level/Frequency Meter test screen and field descriptions.

Level/Freq. Meter Display Screen

1. **Test Function** – Identifies Level Frequency Meter (Level) as the selected function and measurement screen.
2. **Left Input** – The left column lists the level/frequency of the left audio input.
3. **Right Input** – The right column lists the level/frequency of the right audio input.
4. **Left Input Frequency** – Indicates the frequency of the left audio input sine-wave.
5. **Right Input Frequency** – Indicates the frequency of the right audio input sine-wave.
6. **Left Channel Input Level** – Indicates the left audio input level measurement value.
7. **Right Channel Input Level** – Indicates the right audio input level measurement value.
8. **Measurement Units** – Selects the unit of measurement for the level readout on both the left and right meter.
9. **dBr Reference Level Capture** – Select and click to capture a reference level for both the left and right channel for dBr measurements. Select dBr as the measurement unit to meter how the left and right audio input levels are changing compared to the captured reference level.

Level/Frequency Meter Operation:

1. **Apply left and/or right audio into the SP395A's inputs.**
2. **Select the Level Meter Test function.**
3. **Select the desired display units.** The same units are selected for both inputs.
4. **Select the measuring range.** It is advised to select auto-ranging or "A" in the top center toolbar so the SP395A can select the best range automatically. You may hear some artifacts of connecting a strong signal to an input of the meter when it has auto-ranged to maximum gain. Connecting a strong signal may cause the display to momentarily say "OVER" or "UNDER" and you may hear the signal bleed through to the internal speaker.

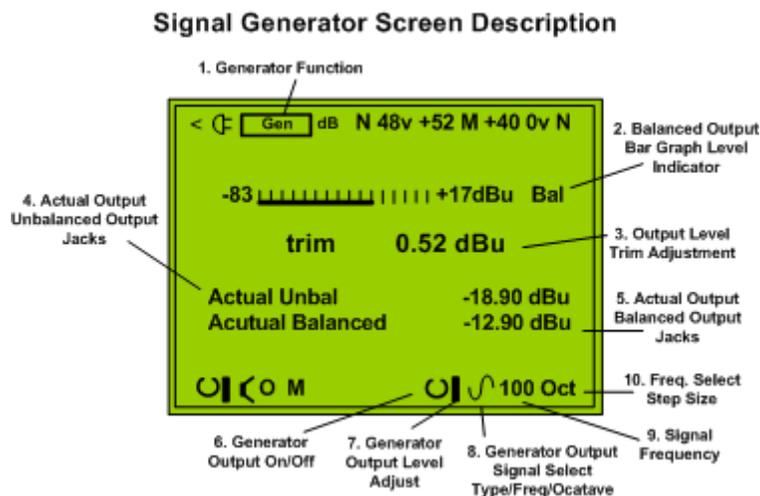
Note: When the signal is improperly ranged or weak, the display reads "LOW" or "LOW Hz."

Signal Generator

The SP395A contains a general-purpose, full-featured, crystal-controlled signal generator that creates test tones and standard noise waveforms. Most of the features of the signal generator are accessed from any function, using bottom toolbar. These include On/Off, level (in 0.5dB steps), waveform selection (including USB), and frequency.

The Signal Generator function selection within the TechBench menu provides a bar graph indication of output level. It further provides a measured level of the actual output to the balance and unbalanced outputs. It lets you trim the output level + or - 1 dBu.

Note: You can store default values for generator waveform type, frequency, and level, by using the Save Defaults function in the Setup & Calibration function.



Generator Screen and description of fields.

Generator screen and field descriptions:

1. **Test Function** – Identifies Generator as the menu selection. Function permits generator output level and output signal analysis and selection.
2. **Bar Graph Level Indicator** – Indicates generator output level in dBu.
3. **Output Level Trim Adjustment** – Offsets the output level by + or - 1 dBu in .01 dBu steps.

4. **Meter Audio Output Level Unbalanced** – Indicates the actual output on the unbalance output jack.
5. **Meter Audio Output Level Balanced** – Indicates the actual output on the balanced output jack.
6. **Generator Output On/Off** – Turns on or off the generator.
7. **Generator Output Level Adjust** – Adjusts the level output of the generator. Level metered by bar graph level indicator.
8. **Generator Output Signal Type** – Selects the generator output signal type. Sine, Square, Pink, White, USB.
9. **Generator Output Signal Frequency** – Adjusts the frequency of the sine or square wave.
10. **Step Size** - Selects how the generator increments frequency selections, octave, 1/3 octave, or full.

Signal Generator Operation:

1. **Connections.** Plug any SP295 balance or unbalanced audio output into the desired destination.
2. **Generator On / Off** Turn the generator “on”. Select the headphone symbol in the bottom toolbar and click the control knob until a line appears in the middle of the symbol.
3. **Output Level** Click on the bottom toolbar level indicator and rotate the knob to increase or decrease the generator output level.
4. **Signal Type:** Click on the signal type indicator in the bottom toolbar to select a sine wave, square wave, white noise, pink noise, or USB symbol. When USB is selected, the USB source (typically a computer) is routed directly to the main SP395A output. Level in this case is controlled from the computer, not the SP395A.
5. **Frequency** Select a desired sine-wave or square-wave frequency by changing the frequency field at the bottom right of the toolbar. The way that the frequency changes is controlled by the Step Size field.

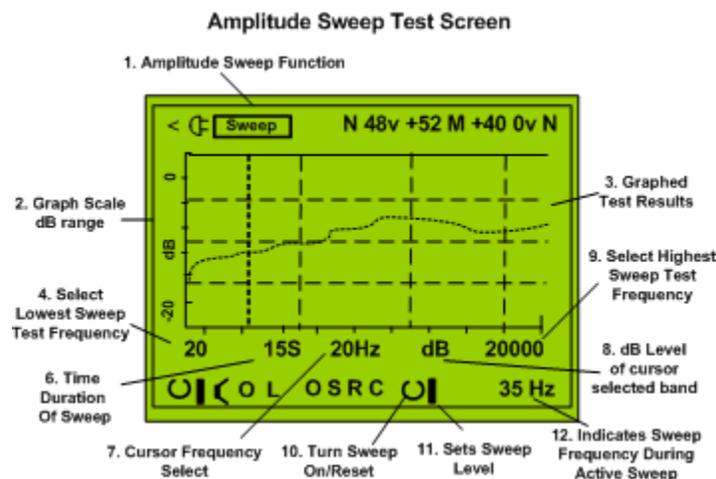
Note: For pink noise, the octave or 1/3 octave step size selection determines if the generator outputs octave or 1/3 octave band limited pink noise.

Amplitude Sweep

The Amplitude Sweep function is a non-acoustic, electrical test, used to measure the audio frequency response of a device under test. This function varies the SoundPro generator output frequency once per millisecond, in integer Hz values only, which results in a sonically smooth sweep. The generator output is fed to the input of the device under test and the output of the device is fed back to the SP395A's measurement input. As the generator is swept smoothly, the measuring circuit averages the results over each of the 1/12 octave bands within the desired sweep range.

This function measures the frequency response of a device by measuring the average level of each 1/12 octave band in the selected sweep range while the generator smoothly sweeps the sine wave. You can set the starting and ending frequencies to any ISO standard 1/3 octave frequency, and the sweep time from 10 to 99 seconds.

Amplitude Sweep Display and Controls



Amplitude Sweep test screen and description of control fields.

1. **Test Function** – Identifies Amplitude Sweep as the menu selection.
2. **Graph Scale Range dBu** – Indicates range of dB level scale in dBu. Selects the highest level of the dB range scale in 5 dB increments from -80 to +60 dBu.
3. **Test Results Graph** – Shows the test results of the amplitude dBu vs. frequency for the device being tested.
4. **Graph Scale Resolution dBu** – Selects the graph resolution. 20 dBu, 40 dBu, or 80 dBu.
5. **Minimum Sweep Frequency** – Indicates minimum frequency of graphed amplitude sweep test.. Adjustable field in octave steps to 10 kHz.
6. **Sweep Time Interval** – Time to sweep through selected frequency range. Selectable from 10 – 99 seconds.
7. **Cursor Frequency** – Indicates frequency at the position of the graph cursor. Select the field and vary the control knob to move and position the cursor to the desired frequency step.
8. **dB Level** – Indicates dB level of the cursor position on the graph or test results level.
9. **Maximum Sweep Frequency** – Indicates the maximum frequency graphed in the Amplitude Sweep test. Adjustable field in octave steps to 40 Hz.

10. **Generator On/Off** – Turns the generator on or off. Starts the frequency sweep to perform the Amplitude Sweep test.
11. **Generator Output Level** – Adjusts the output level of the generator used during the Amplitude Sweep test.
12. **Generator Sweep Frequency** – Indicates the generator frequency as it begins and sweeps through the selected range.

Amplitude Sweep Operation

1. **Set up Output** Connect any of the SP395A's outputs to the input of the device being tested.
2. **Set up Input** Select the appropriate input range that provides the highest range while avoiding clipping. Turn off phantom power if it is on. Connect the output of the device being tested to the LEFT input.
3. **Set up sweep endpoints.** Using the fields below the graph to each side of it, set the desired sweep start and stop frequencies. The lowest frequency is 20 Hz, the highest frequency is 20kHz.
4. **Start / Stop** Start the test by turning on the generator. Wait for the sweep to finish. You can stop a test by turning off the generator while the test is running.
5. **View results.** You can change the graph window by using the fields to the left of the graph window. The top field moves the entire graph up and down, in 5dB steps, and the bottom field changes the bottom dBu setting, in effect changing the vertical dB range of the graph. Together these fields give you the flexibility to set up the screen the way that you want it for a particular application.
6. **Cursor** Use the cursor to display the exact dB level of any point on the screen.

Memories

The test results of the Amplitude Sweeps can be stored in any of the 40 non-volatile memories, using the icons on the bottom toolbar. To access the memories, follow these steps:

1. **Select a memory number** Use the memory number field to select a memory number. When the memory number field is selected, the type of memory stored (or “---“ if empty) will appear on the bottom of the screen. The memory type for the Amplitude Sweeps is “Amp”.
2. **Store memory.** Click on the S (Store) icon to store the memory to the current memory number.
3. **Recall memory.** To recall a memory, click the R (Recall) icon. If the memory is the wrong type, an error message will appear momentarily.
4. After recalling a memory, you can use the graph control fields and the cursor to read out individual data values. These memories can be transferred to the PC/Mac using the Transmit Data function located in the Utilities menu, using either the ATB Interface or TerraLink program.
5. **Real-time Interface**
6. The Amplitude sweep can be used with TerraLink software. The software enables you to see the test results in real time on the screen of a PC or Mac computer. See the TerraLink section of this manual for more information.

Distortion Meter

The Distortion Meter analyzes audio equipment for distortion. There are two separate distortion measurements supplied, THD+N (total harmonic distortion plus noise) and IMD (inter-modulation distortion). The SP395A measures THD+N of an incoming signal by comparing the level of the incoming signal sine-wave with the level of harmonics and noise. It measures IMD by comparing the level of an incoming multi-frequency test tone to the level of its inter-modulation distortion components.

Distortion Meter Display Screen and Controls

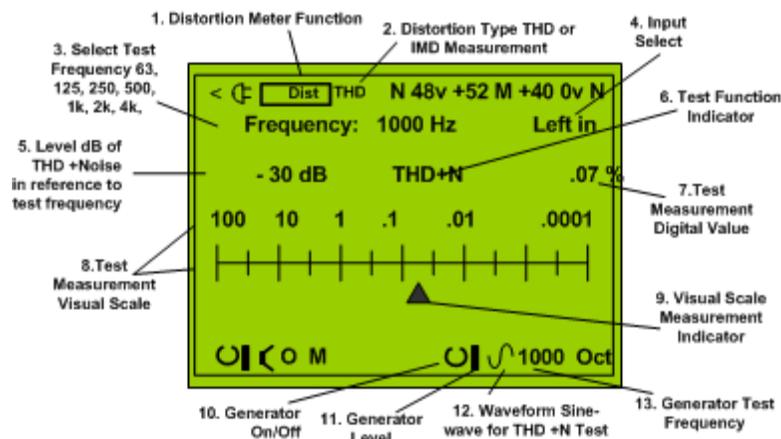
The Distortion Meter tests have slightly different control test screens. The THD+N display screen contains all the generator toolbar icons at the bottom right. The IMD test requires special two tone output signals which are supplied from the SP395A's internal generator. In the IMD test function, the generator toolbar signal type and frequency selections are removed. The signal type for the generator is selected automatically with the test type selected, either DIN 250/8k or SMPTE 7kHz/60Hz. The metered test results are displayed the same.

THD+N Description

The Distortion Meter may be used in conjunction with the internal generator or with a separate low distortion generator. The SP395A's THD +N test is designed to be performed at 63, 125, 250, 500, 1000, or 2000 Hz. The THD+N test sums the energy in the harmonics (and residual noise) of the sine-wave audio input to the meter to compute the THD+N. The range is 0.02% to 50%. The display shows the numeric test value, and also has a bar-graph using a log-scale from 100% to 0.001% THD.

Note: You may use the SP395A's internal sine-wave generator or an external low distortion sine-wave audio generator along with the SP395A's distortion meter. Be sure to choose a generator frequency which matches the test frequency of the SP395A's Distortion Meter.

Distortion (THD) Meter Screen Description



THD +N test screen and description of control fields.

Distortion Meter Control Field Descriptions:

1. **Test Function** – Identifies Distortion Meter as the menu selection.
2. **Distortion Type** – Selects the type of distortion test either IMD or THD
3. **Test Frequency or Standard** – Selects the frequency of the THD+N distortion test. *Note: the generator low distortion sine-wave test frequency must agree with this frequency.* Selects the SMPTE IMD test (7kHz/60Hz) or DIN IMD test (8kHz/250Hz).

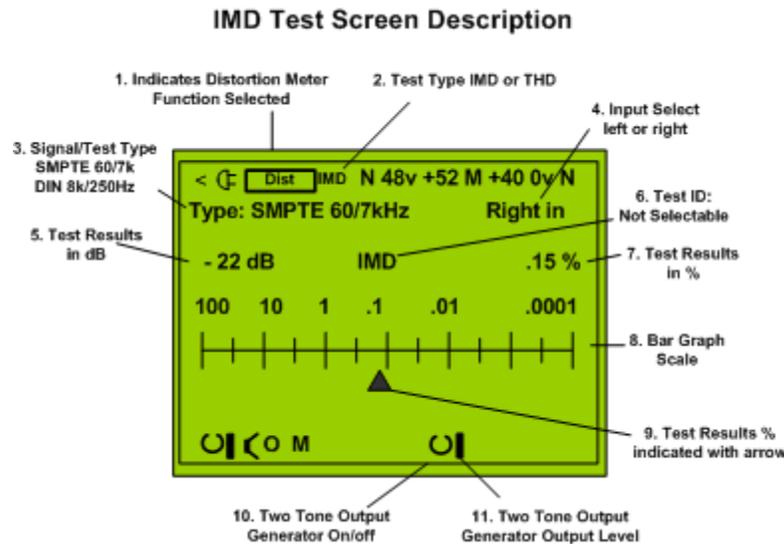
4. **Input Select** – Selects either the right or left input for the distortion test.
5. **Distortion Meter Test Readout** – Indicates distortion test results in dB.
6. **Test Function Identification** – Test Indicator only – this field is not selectable.
7. **Distortion Meter Readout** – Digital value indicating % of distortion for test.
8. **Measurement Scale** – Fixed scale used to graphically indicate distortion %. Not selectable or variable.
9. **Distortion Meter Indicator** – Arrow indicator that marks a distortion level on the visual scale indicating % distortion.
10. **Generator On/Off** – Turns generator output on or off. Active for all distortion tests.
11. **Generator Output Level** – Indicates and provides level adjustment for the generator output. Active for all distortion tests.
12. **Generator Waveform** – Selects generator output waveform. The waveform type should be a sine-wave for the THD + N test. This field does not appear in the IMD test as a special two tone test signal is generated and output by the generator.
13. **Generator Test Frequency** – Only shows on TDH+N test. Indicates the frequency output of the generator. The frequency should match the frequency of the THD+N test selection.

How to use the THD+N Distortion Meter

1. **Set up the Output:** Connect the SP295 output to the input of the device to be measured. Or use an external sine wave signal source at the correct frequency to match the distortion test. Typically THD+N testing is performed at 1 kHz.
2. **Set up the Input:** Connect the output of the device being tested to the SP395A's left input. Turn off the phantom power, set the input gain (auto-gain may be used), and select the Left In for the input on the distortion test field.
3. **Frequency:** Select the desired distortion test frequency within the THD+N test screen, typically 1 kHz. *Note - The distortion test frequency must match the generator frequency.*
4. **Generator On/Off & Level** – If using the SP395A generator, turn the generator on and adjust the output level. Be sure to check and/or adjust the input gain on the SP395A to avoid clipping distortion.
Note: Be sure you are using a sine-wave at the proper frequency.
5. **Read THD+N:** Read the distortion in %

IMD Description

Inter-Modulation Distortion (IMD) is another distortion measurement used in audio equipment testing. IMD is measured by applying a two-tone test signal to the device under test and measuring the distortion components resulting from this test signal. The SP395A supports both SMPTE IMD testing, using a 7 kHz/60 Hz composite waveform, and DIN IMD testing, using an 8 kHz/250 Hz composite waveform. This function is used very much like the THD+N function.



IMD Test Screen and descriptions of control fields.

Distortion Meter Control Field Descriptions:

- 1. Test Function** – Identifies Distortion Meter as the menu selection.
- 2. Distortion Type** – Selects the type of distortion test either IMD or THD
- 3. Test Frequency or Standard** – Selects the frequency of the THD+N distortion test. *Note: the generator low distortion sine-wave test frequency must agree with this frequency.* Selects the SMPTE IMD test (7kHz/60Hz) or DIN IMD test (8kHz/250Hz).
- 4. Input Select** – Selects either the right or left input for the distortion test.
- 5. Distortion Meter Test Readout** – Indicates distortion test results in dB.
- 6. Test Function Identification** – Test Indicator only – this field is not selectable.
- 7. Distortion Meter Readout** – Digital value indicating % of distortion for test.
- 8. Measurement Scale** – Fixed scale used to graphically indicate distortion %. Not selectable or variable.
- 9. Distortion Meter Indicator** – Arrow indicator that marks a distortion level on the visual scale indicating % distortion.
- 10. Generator On/Off** – Turns generator output on or off. Active for all distortion tests.
- 11. Generator Output Level** – Indicates and provides level adjustment for the generator output. Active for all distortion tests.

How to use the IMD Distortion Meter

- 1. Set up the Output:** Connect the SP395A output to the input of the device to be measured. Note that you must use the SP395A generator for the most accurate results. The generator automatically outputs the two-tone test signal selected within the IMD test type screen (60Hz/7kHz or 8kHz/250Hz).

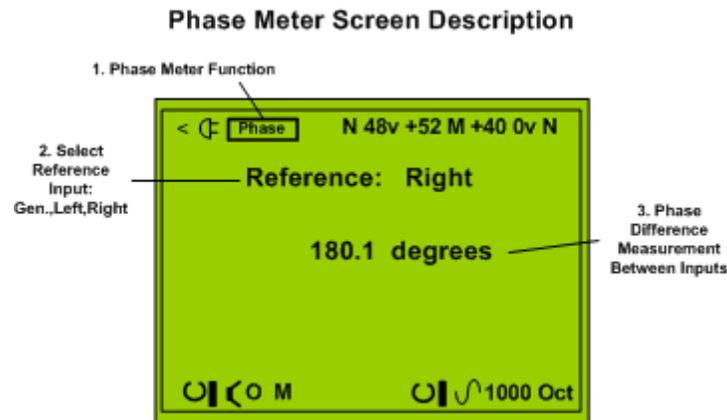
2. **Set up the Input:** Turn off the phantom power, set the input gain (auto-gain may be used), and connect the device output to the SP395A left input.
3. **Select Test Type** - Select the desired IMD test, either SMPTE or DIN.
4. **Turn Generator On** – Turn the generator on to apply the two tone test signal.
5. **Adjust Levels** - Adjust the output level of the generator for an appropriate level to test the unit under test. Be sure the correct input gain setting on the SP395A has been selected to avoid input signal clipping.
6. **Read Distortion** – Read the distortion dB, % or line graph indicators.

Phase Meter

The Phase Meter measures the phase difference in degrees between two input sine waves. The sine waves must be at the same frequency but may be at different levels. The sine waves must be stable in level and frequency for consistent measurement results.

Phase is determined by selecting a waveform as a reference. The phase difference between the reference and 2nd input waveform is displayed in degrees. The phase difference is displayed with a resolution of .1 degrees. You can use the Phase Meter to measure phase shift between channels of analog equipment, such as amps, equalizers or compressors.

Phase Meter Display and Control



Phase Meter Test Screen and field descriptions.

1. **Test Function** – Identifies Phase Meter as the current test function.
2. **Input Reference Select** – The Phase Meter always requires two sine wave inputs. This field selects the input or the internal generator sine wave as the reference waveform.
3. **Phase Measurement Readout** – Phase Difference Measurement in degrees

Phase Meter Operation

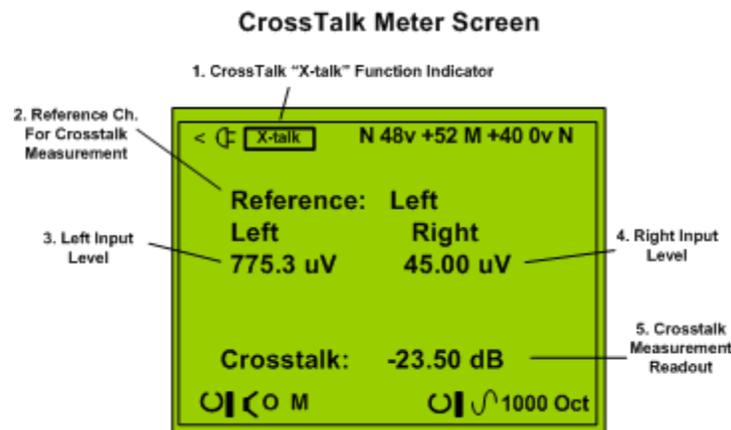
1. **Turn off phantom power.** Turn off phantom voltage to both inputs using the top Toolbar.
2. **Connect the input.** Connect two line level inputs to the left and right SP395A inputs. Note, that if you are using the SP395A generator (Gen) as the reference, route the SP395A balanced output back to one of the SP395A inputs.
3. **Turn on the signals.** If you are using the SP395A to create the test signal, turn on the sine wave generator in the bottom Toolbar. Otherwise, create the sine wave reference and test signals.
4. **Read the results.** After a short settling time, the SP395A displays the phase offset between the 2 inputs in degrees.

Crosstalk Meter

The Crosstalk Meter measures the amount of isolation that exists between two audio sources, such as a right and left audio channel. Crosstalk is further defined as the amount of signal that crosses or bleeds from one channel into the other. It is measured in dB.

Crosstalk is measured by applying signal to only one of the channels of a piece of audio equipment or a cable. The output channels from the equipment or cable output is applied to the left and right Crosstalk Meter inputs. The level difference in dB between the channel with signal, and channel without signal is the measured crosstalk. There should be little if any crosstalk, resulting in very large crosstalk dB values. For example if 1/1000 of the voltage leaks into the non-signal channel, the crosstalk is 60 dB.

Crosstalk Meter display and control



Test Function – Identifies CrossTalk Meter (Xtalk) as the current test function.

1. **Reference Channel** – Selects the reference waveform in which the CrossTalk Meter calculates the level difference (+dB, -dB).
2. **Left Input Level uV** – Displays the measured input level on the left channel input.
3. **Right Input Level uV** – Displays the measured input level on the right channel input.
4. **Crosstalk Measurement Value** – Displays the measured crosstalk level, in dB, between the selected reference and the other or second input.

Crosstalk Meter Operation

1. **Set up the inputs:** Select the CrossTalk Meter function. Turn off the phantom power to each input in the top Toolbar. Turn on auto-gain,
2. Connect two audio inputs, such as a left and right audio channel outputs from an audio device, system or cable. Note: Any two audio signals may be compared and a level difference measured as a crosstalk value.
3. **Set up the output:** Connect the SP395A output to only one channel input of the device or cable under test.
4. **Pick the reference channel:** Select a reference channel for the CrossTalk meter within the test screen. Select the channel that has the signal applied to it from the SP395A.
5. **Read the results:** Read and confirm input levels to the SP395A to confirm proper connection. Read the CrossTalk in dB.

Speakers Menu

The Speakers Test menu provides tests used to test speakers, sound distribution equipment and speaker cables. However, the tests are not limited to these applications as the tests can be used for other similar applications. The Speakers tests include:



Speaker Polarity
Speaker Distortion
Impedance Meter
Impedance Sweep
Cable Tester

The Speakers menu showing the available test functions.

Speaker Polarity

Correct loudspeaker polarity is critical for achieving proper integration of all the speakers in an audio system. The sound waveform reproduced by the loudspeakers should follow the polarity of the sound wave that energized the recording microphone. For example, the beat of a drum should move the loudspeaker cone outward on playback. A single out of phase driver in a loudspeaker can degrade the performance of the entire audio system.

Determining proper polarity by listening to the speaker output is very difficult and inaccurate, and observing cable polarity does not guarantee correct loudspeaker polarity.

The SP395A Speaker Polarity test provides a simple, reliable way of determining absolute polarity of all loudspeaker drivers. It applies pulses to the amplifier and loudspeakers and compares the polarity of the reproduced pulses picked up by the microphone to the polarity of the applied pulses.

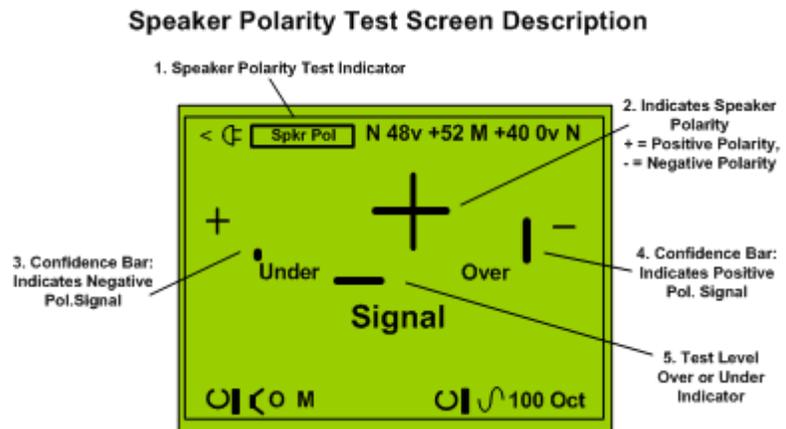
The polarity of all the loudspeaker drivers in an audio system can be quickly checked by driving all loudspeakers simultaneously and positioning the microphone closer to one loudspeaker driver at a time.

Note: Since the SP395A uses XLR pin 2 hot, if your system is wired pin 3 hot, you will likely get incorrect results! A simple way around this is to use an unbalanced output, either the RCA output or the 1/4" output with an unbalanced 1/4" plug.

Speaker Polarity Test screen and field descriptions.

Polarity Test Screen and Control Fields Description:

1. **Test Function** – Identifies Speaker Polarity (Spkr Pol.) as the selected function and measurement screen.
2. **Speaker Polarity Indicator** – Shows the test result or polarity of the speaker.
3. **Confidence Bars** – Indicates when a good level is being sensed by the SP395A and you can be confident in the measurement result.
4. **Test Level** – indicates when the level of the received signal is under or over the level required by the SP395A to perform the speaker polarity test.



Speaker Polarity Test Operation:

1. **Connect microphone.** Connect a measurement microphone to the left SP395A input, turn on phantom power (if required), and set the left input gain to an appropriate value (try +34 or +40).

Note: there is a chance that you will get incorrect results if you are using any untested microphone. Microphones supplied by TerraSonde are tested for correct polarity.

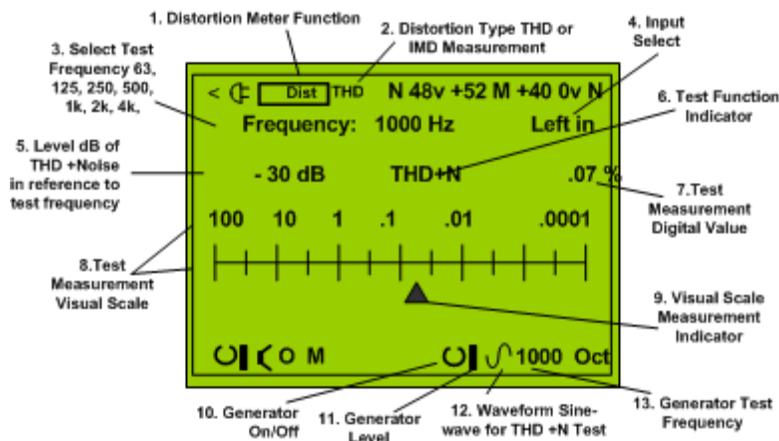
2. **Set up output.** Plug any SP395A audio output into the amplifier and speaker. Turn the level down at the amplifier!
3. **Generator On / Off.** Turn on the generator to send the polarity signal to the speaker. Make sure that only one speaker is playing, or that the microphone is at least 1 foot (0.5 meters) closer to one speaker than the other. Make sure that there is a direct audio path from the speaker driver to the SP395A's microphone.
4. **Position Microphone.** Point the microphone directly at the center of the loudspeaker driver. To best pick up the signal from just one driver, position the microphone directly in front of the driver, about 1-2 times the diameter of the driver away from the driver face (e.g. about 8-16 inches in front of an 8 inch woofer).
5. **Generator Level.** Carefully turn up the amplifier level until the Input Level bar graph shows a mid-scale reading. If required, change the Input gain field or adjust the system volume level to get a mid-scale reading. If the level is too high, reduce the input gain setting.
6. **Confidence Indicators.** Observe the + and – confidence meters to see that the SP395A is getting good information from the speaker. Only one bar should be tall. If both bars are tall or if both bars are low, then there may be excessive background noise in the room.
7. **Observe the display for speaker phase.** A plus sign indicates that the speaker output is the same polarity as the signal that is being applied to the system input. **Note: Some loudspeakers are intentionally wired with one driver out of phase in the enclosure to provide a smoother phase response at the crossover point of the crossover network.**

Speaker Distortion

Excessive speaker distortion can cause audible problems and may be a sign of speaker damage or impending speaker failure. The Speaker Distortion test uses FFT analysis to compare the level of the fundamental test frequency with its harmonics and noise. It generates a sine wave and sums the energy in the upper harmonics (and residual noise) to calculate the total harmonic distortion plus noise (THD+N). The range is 0.01% to 50%. The display shows the value numerically and also has a bar graph using a log-scale from 100% to 0.0001% THD.

Speaker distortion typically runs in the 1-5% range (with low levels of room noise). Use the Speaker Distortion test to check all speakers, looking for differences between left and right speakers and differences between speakers in matched sets. If one speaker has appreciably higher distortion than the others, it is probably defective or damaged. Several test frequencies are provided to properly test woofers, midrange and tweeter speaker types.

Distortion (THD) Meter Screen Description



Speaker Distortion display and description of control fields.

Speaker Distortion Display and Control

1. **Test Function** – Identifies Speaker Distortion (Spkr Dist) as the active test function.
2. **Test Frequency or Standard** – Selects the frequency of the THD+N distortion test. The generator low distortion sine-wave test frequency which is output to the audio system driving the speakers must agree with this frequency selection. Test distortion of full-range speakers over this full frequency range; test sub-woofers at 63 and 125 Hz; test satellite, bookshelf, and PA speakers at 125 Hz and above
3. **Input Select** – Selects either the right or left input for the distortion test. For the Speaker Distortion test, connect a microphone to the left channel input.
4. **Distortion Meter Test Readout** – Indicates distortion test results in dB.
5. **Test Function Identification** – Test Indicator only – this field is not selectable.
6. **Distortion Meter Readout** – Digital value indicating % of speaker distortion.
7. **Measurement Scale** – Fixed scale used to graphically indicate distortion %. Not selectable or variable.
8. **Distortion Meter Indicator** – Arrow indicator that marks a distortion level on the visual scale indicating % distortion. This graph displays the speaker distortion percentage displayed on an analog bar graph, ranging from 100% (left) to 0.0001% (right).

9. **Generator On/Off** – Turns generator output on or off.
10. **Generator Output Level** – Indicates and provides level adjustment for the generator output.
11. **Generator Waveform** – Selects generator output waveform. The waveform type should be a sine-wave for the THD + N test.
12. **Generator Test Frequency** – Indicates the frequency output of the generator. The frequency should match the frequency of the THD+N test selection.

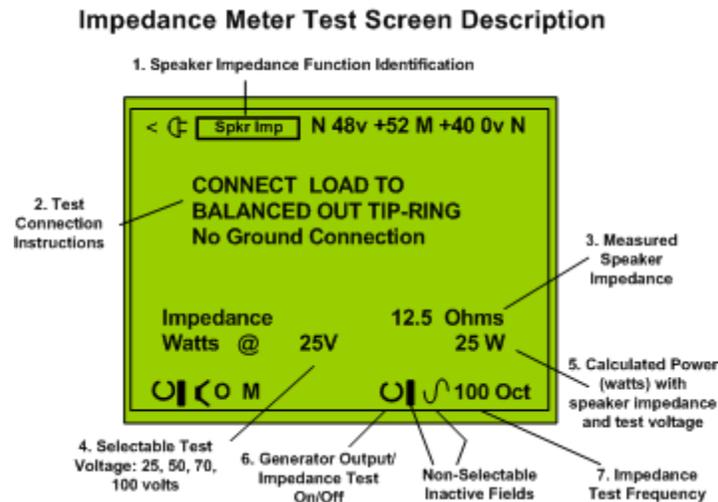
Speaker Distortion Test Operation

1. **Connect the SP395A's Output to the amplifier input.** The SP395A must be the signal source for the distortion test. Connect the output to the amplifier that is driving the speaker to be tested. Set the amplifier gain at 1/3 to 1/2 volume. (You will not hear an output from the speaker until the test is turned on).
2. **Select the desired Test Frequency.** Test full-range speakers at each frequency; sub-woofers at 63 and 125 Hz; satellite, bookshelf and PA speakers at 125 Hz and above.
3. **Set up the Input.** Turn off the phantom power, set the input gain (auto-gain may be used), and connect the device output to the SP395A's left input.
4. **Set the Output Level to minimum.** The speaker volume depends on the SP395A Output Level and the amplifier volume. Caution: Start with the output level at minimum to prevent speaker damage when the distortion test is turned on.
5. **Turn on the Speaker Distortion Test.** Click generator on/off field in the bottom Toolbar. (A line appears in the headphone symbol indicating the audio output is on)
6. Increase the Output Level. Increase the audio output level using the bottom Toolbar until a clear test tone is heard from the speaker. *Caution: If you do not begin to hear an output from the speaker as you increase the output level, check the connections between the SP395A and amplifier, the amplifier input and speaker selection switches, and the amplifier volume setting.*
7. **Read the speaker distortion.** The percent of speaker distortion is shown on the digital display and on the analog bar graph.
8. **Test the speaker at all appropriate test frequencies.**
9. Full-range speakers at each test frequency.
 - Sub-woofers at 63 and 125 Hz.
 - Satellite, bookshelf, and PA speakers at 125 Hz and above.
 - *Caution: Reduce the Output Level before changing the test frequency.*
10. **Test the remaining speakers in the audio system.**
Caution: Reduce the SP395A Output Level before switching speakers or amplifier inputs.

Impedance Meter

The Impedance Meter measures the AC opposition or “impedance” offered by a passive device, such as an audio speaker, to an audio frequency. It may also be used to measure the input impedance of audio devices at a specific frequency. It computes the power that would be required to drive a constant voltage speaker network, at a particular voltage. The output level and waveform are fixed in this function.

Impedance Meter Display and Control



Impedance Meter Display and description of control fields.

1. **Test Function** – Identifies Impedance Meter as the active test function.
2. **Test Connection Instructions** – Instruction on how to connect the balanced output to a load for measurement. Use the specialty Impedance Test Cable, Sencore #39G1171. This cable provides the proper connection from the ¼ inch mono (balanced) plug to clip connections which can be connected to a device or input to be tested.
3. **Test Impedance** - Measured impedance value determined by the Impedance Meter.
4. **System Voltage Selection** – Selects the distributed audio system voltage used by the Impedance Meter to calculate load power (watts). Use to calculate load power of an audio distribution system based upon the measured impedance and one voltage level.
5. **Calculated Power watts** – Computed from selected voltage and measured impedance values. Shows power (watts) required by the system to drive the system to full output power.
6. **Generator – Impedance Test On/Off** – Turns the Impedance Meter function on or off.
7. **Generator/Test Frequency** – Test frequency of the Impedance Meter.

Note: The Generator level and waveform selection fields are not selectable or variable in the Impedance Meter function.

Impedance Meter Operation

1. **Connect the special cable to device or system.** Use the specialty Impedance Test Cable, Sencore #39G1171. Plug this connector into the 1/4” TRS output (phone) jack, next to the male XLR output jack.
2. **Connect load.** Connect the test lead clips to the speaker, system input or other load to be measured.

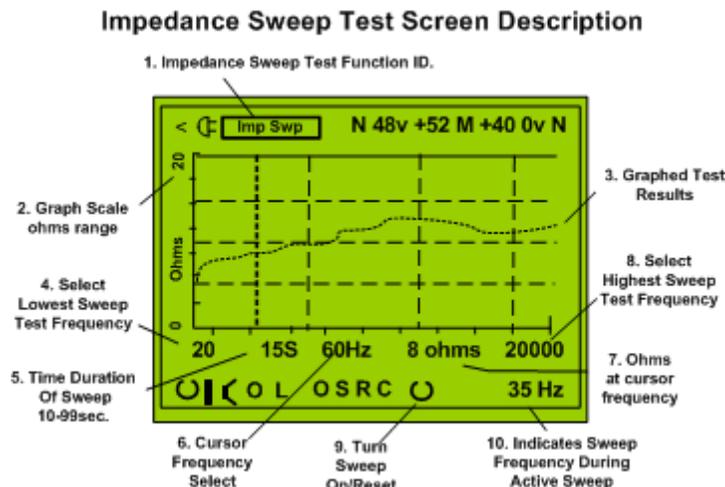
3. **Select frequency.** Select the frequency at which you wish to test the device impedance. This is selectable in the generator frequency field in the bottom Toolbar.
4. **Turn on the generator.** The generator output level is fixed for this test.
5. **Read impedance value.** Read the impedance in ohms.
6. **Select system voltage.** If desired, select the system voltage. The function will then compute the power required to drive the system to full output.

Impedance Sweep

The Impedance Meter measured the impedance of a load, such as an audio speaker, to a single test frequency. The Impedance Sweep test measures impedance as well but sweeps the applied test frequency through a user selected range. The resulting impedance through the range is measured and graphed on a display showing details of how the impedance changed with frequency. The graphed test data offers an impedance frequency response analysis.

The Impedance Sweep test plots the impedance of an externally connected load in each 1/3 or 1/12 octave band, as the signal generator is swept through the selected frequency range. The results are plotted graphically on the screen. The x-axis scale maximum may be varied from 20 to 8000 ohms full scale. The endpoints of the sweep can be set to any octave point. As the generator is swept in steps, the Impedance Sweep function computes the results for each of the 1/12 octave bands. You can set the starting and ending frequencies to any ISO standard 1/3 octave frequency, and the sweep time from 10 to 99 seconds.

Impedance Sweep Display and Control



Impedance Sweep Display and a description of its control fields.

1. **Test Function** – Identifies Impedance Sweep as the active test function.
2. **Impedance Scale** – This field sets the ohms full-scale range that is represented by the vertical scale. The full scale value can be set to **20, 50, 100, 200, 500, 1000, 2000, 4000,** or **8000** ohms. To change the full-scale value, use the control knob to highlight the field and click to underline it. Then rotate the control knob to select the desired value, and click to accept the value.
3. **Graphed Measurement Data** – Measurement data of impedance vs. frequency.
4. **Minimum Test Frequency** – Sets the minimum frequency used in impedance sweep. Sets the low end frequency of the graph.

5. **Sweep Interval** – Selects the time of the Impedance Sweep from 10 to 99 seconds. Select the field and vary the control knob to increase or decrease the sweep time interval.
6. **Cursor Frequency** – Indicates the frequency at the position of the graph cursor. Select the field and vary the control knob to move and position the cursor to the desired frequency step.
7. **Cursor Ohms Measurement** – Measured value of ohms at the frequency selected by the cursor.
8. **Maximum Test Frequency** – Sets the maximum frequency used in the impedance sweep test. Set the high end frequency of the graphed data.
9. **Generator/Impedance Sweep On/Off** – Turns on or off the Impedance Sweep function.
10. Indicates the audio frequency steps as the Impedance Sweep test is performed.

Impedance Sweep Operation

1. **Connect the special cable to device or system.** Use the specialty Impedance Test Cable, Sencore #39G1171. Plug this connector into the 1/4" TRS output (phone) jack, next to the male XLR output jack.
 - (Disconnect the load from any power amplifier for best results and safe operation.)
2. **Set the sweep endpoints.** Using the fields on the bottom of the graph, set the sweep start and stop frequencies. The lowest frequency is 20Hz; the highest frequency is 20 kHz.
3. **Start / Stop** Start the test by turning on the generator. Wait for the sweep to finish. You can stop a test by turning off the generator while the test is running.
4. **View the results.** You can change the graph window by using the field on the upper left of the graph window. This field sets the top Ohms value for the graph.
5. **Cursor** Use the cursor to display the exact impedance of any point on the screen.

Memories

The results of the Impedance Sweep test can be stored in any of the 40 non-volatile memories, using the icons on the bottom toolbar. To access the memories, follow these steps:

- **Select a memory number** Use the memory number field to select a memory number. When the memory number field is selected, the type of memory stored (or “---“ if empty) will appear on the bottom of the graph. The memory designation for the impedance sweep is “Imp”.
- **Store memory.** Click on the S (Store) icon to store the memory to the current memory number.
- **Recall memory.** To recall a memory, click the R (Recall) icon. If the memory is the wrong type, an error message will appear momentarily.

After recalling a memory, you can use the graph control fields and the cursor to read out individual data values. These memories can be transferred to the PC/Mac using the Interface Utility provided on the SP395A software CD.

Real-time Interface

The Impedance Sweep, when used with the TerraLink PC software, allows you to see the results in real time on the screen of a PC or Mac computer. See the Terralink software section of this manual for more information.

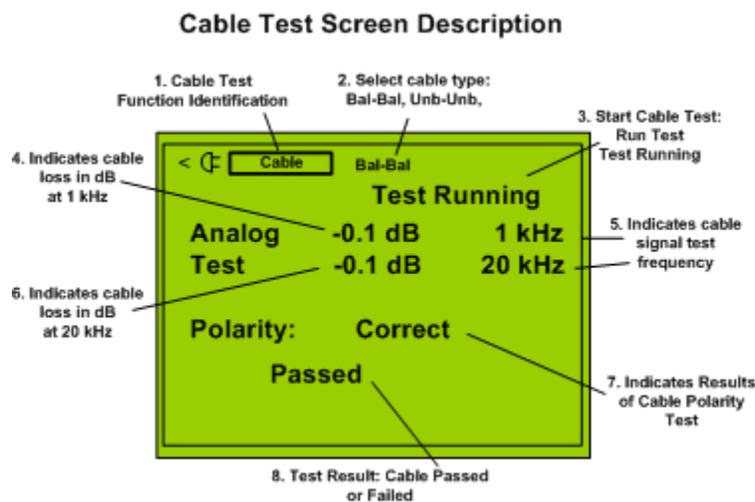
Cable Tester

The SP395A Cable Tester verifies the wiring and integrity of audio cables. It provides connection to test cables with XLR balanced or ¼ phono unbalanced connectors. Simply connect one end of the cable being tested to the appropriate input connector on the left side of the SP395A ("L" input). Connect the other cable end to the corresponding output connector on the right side of the SP395A. The SP395A input/output connectors are clearly marked "Cable Test."

The SP395A performs both an analog transmission loss test and a polarity test of the cable. The SP395A performs an analog transmission test of the balanced or unbalanced cable. This test alternately passes a 1 kHz and a 20 kHz signal through the cable and displays the dB loss at each frequency. Good cables should have less than 0.5 dB loss at both frequencies. Poor cables often show a difference in loss between the two test frequencies.

Use the Cable Tester to verify the wiring and functioning of microphone cables, patch cables, or guitar cables. If the cable is good and has the proper wired polarity, the test readout indicates "Passed" and the internal speaker beeps momentarily. If the cable is open or the polarity reversed, the test readout indicates "Failed" and the internal speaker buzzes momentarily. Note that the speaker beep only checks the results of the cable polarity tests.

Cable Test Display and Control Fields



Cable Test display and a description of its test fields.

- 1. Test Function** – Identifies Cable Test (Cable) as the active test function.
- 2. Cable Type** - Selects the type of audio cable that you are testing. Select one of the following types:
 - **Unbal > Unbal** - unbalanced-to-unbalanced cables
 - **Bal > Bal** - balanced-to-balanced
- 3. Cable Test Status** – Starts the cable test. Indicates "Test Running" when the cable test is being performed. Indicates "Run Test" when the test is ready to be performed or when the cable test is complete.
- 4. Analog Test Results 1 kHz-** This shows the results of the 1 kHz transmission test. The numbers are the dB attenuation loss through the cable. Better cables show readings closer to 0, indicating no signal attenuation. For most audio cables, the 1 kHz levels should be less than 0.5 dB.

5. **Transmission Loss Test Frequency** – Indicates the frequency used to test the dB loss of the cable.
6. **Analog Test Results 20 kHz-** This shows the results of the 20 kHz transmission test. The numbers are the dB attenuation loss through the cable. Better cables show readings closer to 0, indicating no signal attenuation. For most audio cables, the 20 kHz levels should be less than 0.5 dB. Unequal response or a large difference in dB loss values between 1kHz and 20 kHz indicates a bad cable or a faulty solder connection.
7. **Polarity Test Result** – Briefly indicates results of polarity test as “Correct” or “Incorrect.”
8. **Cable Test Analysis** – Indicates result of cable polarity test indicating “Passed” or “Failed.”

Cable Tester Operation

1. **Select Cable Test:** Select the Cable Tester listing in the SP395A sub-menu.
2. **Select Cable Type:** Select the cable type from the sub-menu. Microphone cables are BAL >BAL, while guitar cables are UNBAL>UNBAL.

Note: The output and input connectors on the SP395A side panels are clearly marked for the cable test. Be sure to use the connectors that are labeled “Cable Test”.

3. **Start the test.** Highlight and click the Test Status field and click to begin the test. The field displays "Test Running" while the test is in progress and "Run Test" when completed
4. **Analog Test Result** Next, an analog audio test is performed, running 1kHz and then 20kHz through the cable. The drop in transmission levels are shown for each frequency. A good patch cable will have typical values of 0.0dB to -0.3 dB for both frequencies. Note: Longer audio cables will have additional attenuation.
5. **Polarity Test** After the analog test, a polarity test is run. This test checks the cable for improper wiring. During the test a “Correct” or “Incorrect” indication is briefly displayed. After the test runs, the word “Passed” or “Failed” appears, and the speaker beeps if the cable is good, or buzzes momentarily if it is bad.

Note: The SP395A uses XLR pin 2 hot. If your system is wired pin 3 hot, you will get wrong results. A simple way around this is to use an unbalanced output, either the RCA output or the ¼" output with an unbalanced ¼" plug.

TOOLS

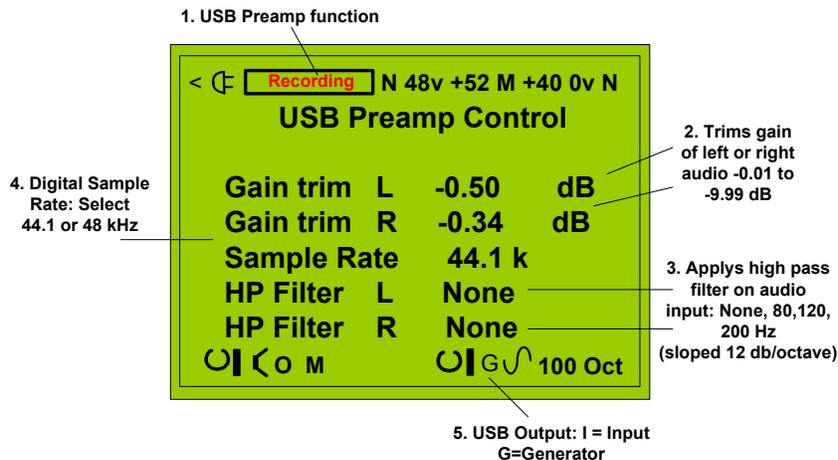
The Tools menu of the SP395A incorporates a full-function two-way USB audio interface. This allows routing the audio from the mic/line inputs or the generator output of the SP395A to a computer through a USB port connection. It further provides receiving USB audio from a computer, and routing it to the main output of the SP395A.



The Tools menu further provides selection of two automated tests that may be performed when using the SP395A with the Sencore DAG5161 generator. The SP395A provides serial control commands to the generator which controls the generator's signal output to coincide with a series of SP395A acoustic tests. The series of test results are stored in memory locations for download to a PC.

USB Preamp Display & Control

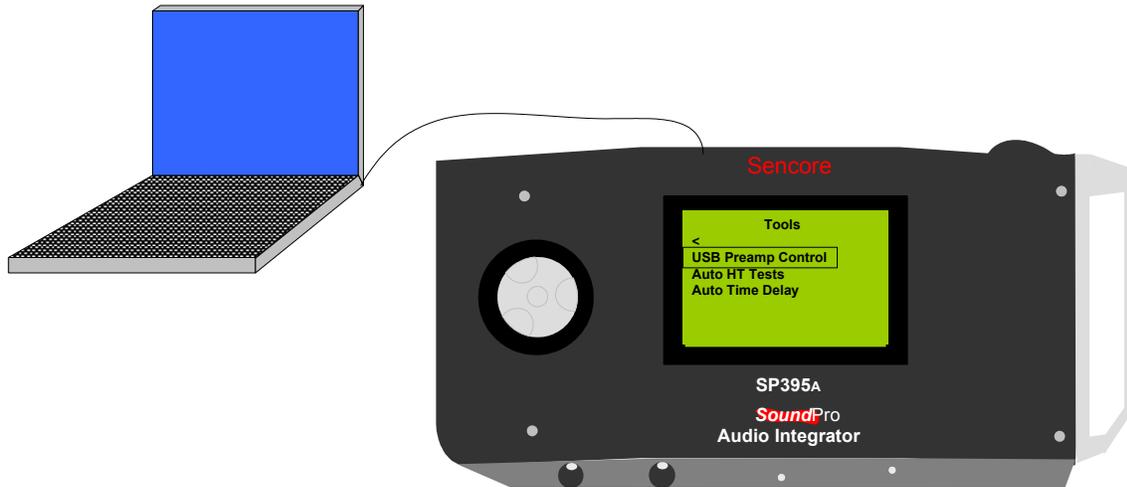
USB Preamp Screen Description



- USB Preamp Function** – Indicates USB Preamp display screen with control fields.
- Left & Right Audio USB Level Attenuator Control** – Decreases the level of the left and right USB audio.
- Left & Right Channel High Pass Filter Select** – Selects None, 80 Hz, 120 Hz or 200 Hz high pass filter in the USB audio signal path.
- Digital Sample Rate:** Selects digital audio sample rate 44.1 or 48 kHz.
- Digital Output Control:** Selects the signal source applied to the SB output. I = Input, which means the input to the SP395A is routed to the digital outputs. G = Generator, in which the generator signal is applied to the digital outputs

USB Preamp Operation

The SP395A may be used as a calibrated USB microphone preamp. Since the audio signal is split before being routed to the DSP for processing, you can use the Audio Integrator functions and the USB preamp at the same time. For example, you can use the SP395A as a sound card for a PC-based analysis program, like SmaartLive, while you record the A-weighted sound level, using the Sound Study Graph.



The USB Audio Interface permits digital audio between a computer and the SP395A.

To use the SP395A as a USB preamp:

1. **Connect microphone(s)** Connect one or two microphones to the SP395A's inputs.
2. **Select USB Preamp Control** in the Utilities menu. **Phantom Power** Enable 48V phantom power to mics (Top Tool Bar).
3. **Set Input Gain** Adjust the input preamp gain settings to an appropriate level (Top Tool Bar). You can use the "A" Auto-gain setting function, but, realize that when the gain changes, you may hear an audible click.
4. **Connect USB cable** Plug a standard USB cable from the host PC to the Consultant. Since the PC is the host, and the SP395A is the peripheral, the square end of the cable goes into the SP395A and the flat end goes into the PC.
5. **Set the Generator Digital Output** or I/G field to "I" (Bottom Right Toolbar).
6. **Set the Sample Rate.** The SP395A USB output runs only at 48k. Set the Sample Rate field to 48k. Set your computer's sample rate to 48k (48000). (SmartLive: Select Options/Devices menu – set sample rate to 48k)
7. **Computer Setup** No special software driver is required, as the SP395A USB section appears as a standard USB audio codec.
8. **Set Gain Trim.** Use the Gain Trim controls to adjust the level to the USB port in 0.01 dB steps as desired.

SmaartLive Setup Notes: Select the Options/Devices menu – Select "USB Audio CODEX" for both Wave In and Wave Out. Turn "Meas Sig" button ON. Click the "Spectrum" button. Turn "Smaart

Windows XP note: Most operating systems allow setting the SP395A to either 44.1 or 48 kHz sample rate. Windows XP, however, allows using only the 48 kHz sample rate setting.

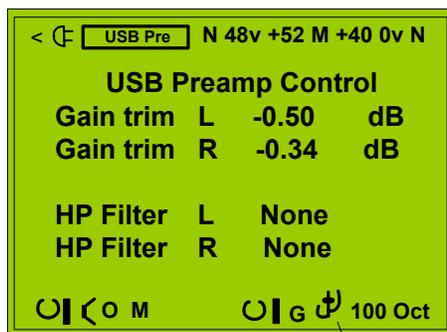
Any audio that enters the SP395A's inputs is routed to the computer over the USB audio link. You can still use the SP395A normally, running audio analysis or tests, while the digital audio is being routed to the computer. You can even change functions without losing the USB connection to the PC.

USB Output Operation

You can route audio from the computer to the SP395A's main outputs through the USB connection. For example, you can send a test signal from a PC program to an audio playback system.

To route USB audio from computer to SP395A's outputs:

1. **Connect USB Cable** As in the steps above, connect a standard USB cable from the SP395A to the computer.
2. **Computer Setup** No special software driver is required, as the SP395A USB section appears as a standard USB audio codec output. You may, however, have to adjust the computer's sample rate to 48 kHz to match the SP395A.
3. **Select USB Output** On the right end of the bottom toolbar, change the generator waveform field to select the USB icon. The left channel of the computer's USB audio output is now routed directly to the SP395A's main (mono) generator output.
4. The computer USB audio now appears at the unit's main outputs. You can use the XLR, 1/4" phone, or RCA phono connector to route the PC's left USB audio channel to an amplifier input.



USB Icon

Select the generator USB icon in the waveform field on the bottom toolbar.

Note: In this mode, the SP395A output level is completely controlled from the computer, not from the SP395A. Use the normal computer output level controls to set the SP395A output level.

Note: Only the left PC USB audio channel is available. The USB signal is available only on the main analog SP395A outputs, and is not available in the monitor section, so you cannot hear USB audio in the headphones.

Auto HT Tests

The Auto HT Tests function provides automated measurements of a home theater when combined with the Sencore DAG5161. The SP395A is connected via a serial connection cable to the DAG5161 for automated test signal selection by the SP395A during the auto tests. A series of automated tests are performed by the SP395A and captured to memory locations. The test data contained in the memory locations may be viewed to easily indicate the A/V receiver adjustments needed for proper level and delay adjustment. The memory contained test data may be downloaded to a PC using the SoundPro Report Software for report creation.

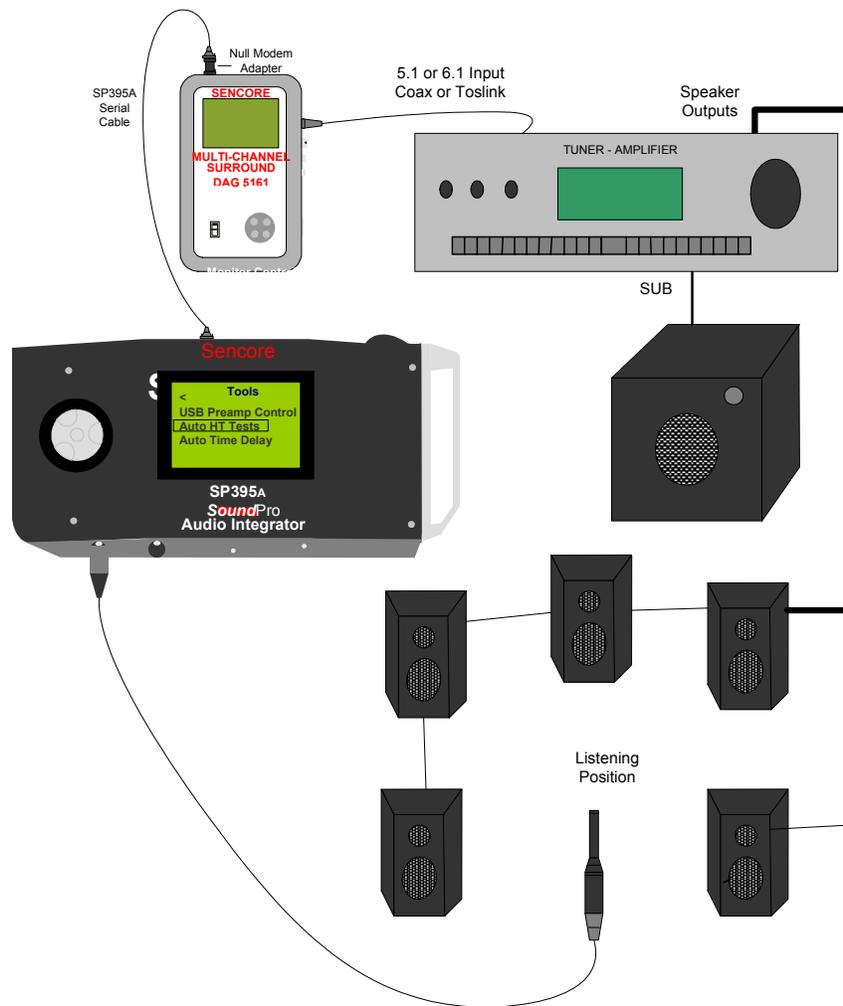
Auto HT Tests Setup

The Auto HT Tests fully automate the tedious process of selecting signals and manually analyzing SPL levels, polarity and delay times from each speaker. In addition the Auto HT Tests document early reflections, system frequency response and ambient noise.

The SP395A's Auto HT Tests and Auto Time Delay tests within the Tools menu differ from all other instrument functions in that the RS232 Serial jack is used to automatically control the Sencore DAG5161 generator. Serial commands from the SP395A select the appropriate signals to correspond to the SP395A's acoustic analyzing tests. This permits the SoundPro to simply increment through a series of system measurements for fast hands-off testing.

The setup for analyzing a home theater sound system is shown in the nearby figure. Connect the Coaxial S/PDIF or Toslink output from the DAG5161 to the Digital Input of the A/V receiver. Connect the SP395A's Serial Cable (Sencore #13G124, straight-through cable) from the SP395A's RS232 Serial Jack through a null modem adapter to the RS232 jack of the DAG5161. ***You must include a null modem adapter in the serial connection when using any straight through serial cable.***

Auto HT Tests Setup



The Auto HT Tests Setup for analyzing a home theater system.

Caution: The communications link between the SP395A and DAG5161 is very sensitive to RF noise. Use the cable and adapters supplied with the SP395A and DAG5161. Others may work, but problems or errors could result in physical damage to system loudspeakers.

Position the measurement microphone on a stand in the listening position and connect the mic cable to the "L" input of the SP395A. All the system speakers should be connected properly to the amplifier or A/V receiver. Select the Auto HT Tests or Auto Time Delay Tests within the Tools menu of the SP395A.

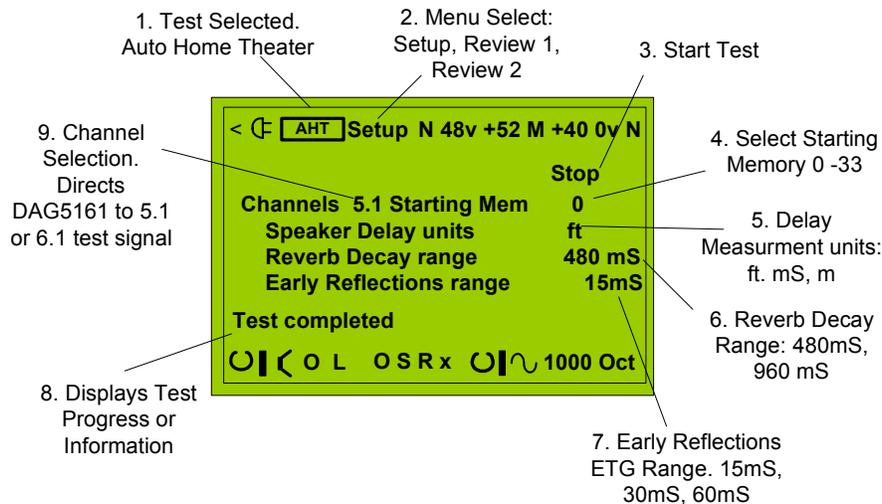
Caution: Before starting the Auto HT Tests or Auto Time Delay Test, you should first verify that the A/V receiver is properly configured and can safely tolerate DAG5161 generator signal switching. Some A/V receivers output speaker pops or bursts as the digital signal output from the generator turns on and off during signal switching. You should manually apply a test signal from the DAG with a low level volume setting and set the receiver for operation with the 5.1 or 6.1 test input. Manually select a few different signals within the 5.1 or 6.1 menu and start the generator output while monitoring the sound system for excessive speaker pops or bursts. Do not perform either the Auto HT Tests or Auto Time Delay Test if you feel the pops could be threatening to the speakers.

All A/V receivers have adjustments within their setup menu permitting fine adjustment of speaker output level and time/distance delay. These adjustments may be set using the data results of the Auto HT Tests. Adjustments insure that the speaker levels are balanced (Equal SPL measurements) and that the sound arrival time to the listening position (Delay in ft. or mS) from each speaker is approximately the same. Before performing the HT Auto Tests you should set the A/V receiver's speaker level settings to 0 dB and the distance/time settings to the approximate distance from the speaker to the listening position. Each foot of distance is approximately 1 mS.

Auto HT Tests Setup:

1. Connect a S/PDIF or Toslink cable from the output of DAG5161 to the Digital Input of the A/V receiver.
2. Connect the SP395A's Serial Cable (Sencore #13G124, straight-through cable) from the SP395A's RS232 Serial Jack through a null modem adapter to the RS232 jack of the DAG5161. ***You must include a null modem adapter in the serial connection when using any straight through serial cable.***
3. Position the measurement microphone on a stand in the listening position and connect a cable from the measurement mic to the "L" input of the SP395A.
4. Be sure the speakers are properly connected to the A/V receiver. Reduce the volume setting of the A/V receiver. Set the A/V receiver to properly receive and output the Dolby 5.1 or DTS 6.1 input signal.
5. In the A/V receiver's setup menu, set the speaker level settings to 0 dB and the distance/time settings to the approximate distance from the speaker to the listening position. For each foot of distance figure 1 mS of sound travel time.
6. Select the 5.1 or 6.1 Surround Signal on the DAG5161 to match the sound system being tested. Output several different 5.1 or 6.1 test signals from the DAG5161 generator and confirm normal A/V receiver output. **Note: If excessive speaker pop or burst are heard when switching generator signals, do not perform the Auto HT Tests, as speaker damage may occur.**
7. Select the "500Hz-2kHz Pink L" test signal on the DAG5161 generator. Set the volume level on the A/V receiver for approximately 75 dB SPL at the listening position or center of the listening area. *You are now ready to perform the Auto HT Tests or Auto Time Delay Tests within the SP395A Tools Menu.*

Auto HT Tests Setup Display and Control Fields



Auto HT Tests screen and description of control fields.

1. **Test Function** - Identifies the Auto HT Tests as the active function.
2. **Setup/Review Menu Select:** This field selects between three menu screens (Setup, Review1, and Review2). Setup is used to change the settings for the test, as well as to initiate the tests. Review1 and Review2 contain the results of previously run tests.
3. **Start Test** - This field is used to start the auto test. Initially it is set to “Stop”, meaning that the test is not running. To start the Auto HT Test, select this field and click the control knob to toggle its state to “Start.” Once the test starts running, it cannot be stopped; nor can the other settings be changed.
4. **Starting Memory** -This field selects the starting or “Base” memory location used by the Auto HT tests to store test data. The auto tests use seven sequential memory locations, starting at the location specified. The memory locations are used as follows:
 - Base – The numeric results from the first three AHT tests, including speaker level, polarity, and time synchronization.
 - Base+1 –Energy Time Graph for reverberation evaluation
 - Base+2 –Energy Time Graph for left channel early reflection evaluation
 - Base+3 –Energy Time Graph for right channel early reflection evaluation
 - Base+4 –RTA for ambient noise analysis
 - Base+5 –RTA-Low for subwoofer equalization evaluation
 - Base+6 –RTA for main channel equalization evaluation
5. **Speaker Delay Measurement Units** -This control field toggles the delay units between feet (ft), milliseconds (ms), and meters (m) for the ETG test functions. For convenience you may set this to match the settings used by the AV receiver to simplify speaker time synchronization.
6. **Reverb Decay Range** - This field selects the range to be used during the Reverb Decay test; the values are 960 ms and 480 ms. Use 480 for most home theater rooms. Use 960 only when the reverberation decay time is greater than 4 seconds (a fairly large room). The default for this parameter can be changed with the “Save Settings” option.

7. **Early Reflections Range** -This toggles between 15/30/60 ms ranges to be used for the Early Reflection tests. This setting is the range of the difference between the direct sound path and the reflected path. This will generally be set to 15. If the reflection path difference is longer, use a higher setting. The default for this parameter can be changed with the “Save Settings” option.
8. **Information/Test Progress:** This field is not selectable. This line shows information regarding the test and indicates when the auto tests are completed.
9. **Channel Selection** - This toggles the mode between 6.1 (DTS-ES), and 5.1 (Dolby AC3). This selection determines what signal is applied by the Sencore DAG5161 and adds a CS (Center Surround) test to the auto tests.

Auto HT Tests

There are eight SP395A’s Auto HT tests. These are performed in the sequence listed and summarized below.

1. Loudspeaker Reference Level for each of the 5.1 or 6.1 channels (six or seven tests). The SP395A is automatically set to “Sound Level Meter, Slow, C-Weighted.” The DAG sequences through the test signals listed in the chart. SPL measurement results are tabulated to SP395A memory (Base). Test results may be view in Review 1 screen upon conclusion of all 8 auto tests.

Dolby AC-3 5.1

500Hz - 2kHz Pink L
500Hz - 2kHz Pink C
500Hz - 2kHz Pink R
500Hz - 2kHz Pink RS
500Hz - 2kHz Pink LS
40Hz - 80Hz Pink LFE

DTS 6.1

500Hz - 2kHz Pink L
500Hz - 2kHz Pink C
500Hz - 2kHz Pink R
500Hz - 2kHz Pink RS
500Hz - 2kHz Pink CS
500Hz - 2kHz Pink LS
40Hz - 80Hz Pink LFE

2. Loudspeaker Polarity test for the 5 or 6 main channels (five or six tests). The SP395A is automatically set to “Speaker Polarity”, and the DAG generates and sequences through the test signals listed in the chart. Speaker Polarity results are tabulated to memory (Base). Test results may be view in Review 1 screen upon conclusion of all 8 auto tests.

Dolby AC-3 5.1

Polarity L
Polarity C
Polarity R
Polarity RS
Polarity LS

DTS 6.1

Polarity L
Polarity C
Polarity R
Polarity RS
Polarity CS
Polarity LS

3. Loudspeaker Time Synchronization (one test). The SP395A is automatically set to “Auto Time Delay” mode, and the DAG generates the “Auto Time Delay” test signal sequence. The results are tabulated to a memory location (Base). The test results may be viewed in the Review 2 menu.

- *These first three tests are typically the only ones that need to be repeated for adjustments on the receiver / speakers. The rest are for more in-depth analysis of the theater room, i.e. acoustical treatments, speaker placement, etc.*
- 4. Reverb Decay Time (one test). The SP395A is automatically set to “Energy Time Graph”, A-Weighting. The DAG generates the “Impulse C” signal and the ETG measurement is captured to memory (Base/Starting Memory # +1).
- 5. Early Reflection Analysis for left & right (two tests). The SP395A is automatically set to “Energy Time Graph”, A-Weighting for sequencing through both tests. The DAG will generate “Impulse L”, and then “Impulse R” test signals. ETG results for each speaker are captured and recorded to memory (Left = Starting Memory # + 2, Right = Starting Memory # + 3)
- 6. Ambient noise analysis (one test). The SP395A is automatically set to “Real-Time Analyzer”, Full, 1 Octave, Flat. The DAG does not generate any signals for this test. *Remain quiet during this test so the SP395A properly analyzes only room noises.* The RTA is captured and recorded to memory (Starting Memory # +4)
- 7. Subwoofer Frequency Response (one test). The SP395A is automatically set set to “Real-Time Analyzer”, 1/3 Octave, Flat. The DAG generates the “20Hz – 220Hz Pink LFE” signal. The RTA measurement is captured and recorded to memory (Starting Mem # +5)
- 8. System Frequency Response (one test). The SP395A is automatically set to “Real-Time Analyzer”, Full, 1/3 Octave, Flat. The DAG generates the “20Hz – 20KHz Pink All” signal. The RTA is captured and recorded to memory (Starting Memory # + 6)

Once started, the eight automated tests sequence automatically. You can view the SP395A and DAG5161 changing test screens and hear the sound system outputting the various test signals. The eight tests require about five minutes for the 5.1 and about six minutes for the 6.1 tests. There is no way to run each automated test individually. You can stop the tests at a desired point in the test sequence by power cycling both the SP295C and DAG5161. The minimum number of times you typically run the test sequence is twice, once to capture a “before” snapshot of system performance, and once to verify the adjustments you make to the system

Auto HT Tests Operation

1. **Select Auto HT Tests:** With proper setup complete, (see the “Auto HT Tests Setup” section of this manual) select the Auto HT Tests within the Tools menu of the SP395A.
2. **Select Setup Settings:**
 - A. Select the 5.1 or 6.1 test signal in the Channel Selection field to match the sound system being tested.
 - B. Select a Starting Memory.

Hint: For ease in creating a Pre-Calibration/Post-Calibration report with the Sencore Report Software, choose 0 for a starting memory for the first non-calibrated Auto HT Tests. After you have made receiver adjustments, choose 10 as a starting memory for

the second post calibration Auto HT Tests. Memories 0-6 then contain the Pre-calibration data while memory locations 10-16 contain the post calibration test data.

- C. Select the Speaker Delay Units to agree with the A/V Receiver's adjustment units, either ft. mS or m.
 - D. Set the Reverb Decay range to 480 mS. Use 960mS only in a large room with a long reverberation decay time.
 - E. Set the Early Reflection range. 15 mS is appropriate for most home theaters. Use 30 mS to view reflections out to 30 mS in duration when desired.
3. **Start the Auto HT Tests:** Highlight the "Stop" field and click the control knob to start the tests. Observe and hear the tests being automatically performed. Be quiet and stay out of the path between the speakers and microphone as to not impact the test results.

Note: You can stop the auto tests at any time by turning off the power of the SP395A and DAG5161 as indicated in the Test Progress/Information field. However, doing so may impact the test data captured to memory. If you wait to recycle power until after Test 3, the Speaker Time Synchronization test, the base memory holds the results of the first 3 tests and may be reviewed in the Reveiw1 and Review 2 menus.

4. **View Test Results:** Upon the conclusion of the Auto HT Tests, view the test data for tests 1-3 by selecting the Review 1 and Review 2 menus in the Menu Select field. These are indicated by "Rev 1" and "Rev 2." Balanced levels are indicated with nearly the same SPL values in the Rev 1 display menu. Polarities should all be the same. In the Rev 2 screen the delay times should be near 0 reflecting an arrival time nearly equal to the left front channel speaker used as a reference. Within the A/V receiver setup, make adjustment to individual channels to balance channel levels and distance/time. Repeat the Auto HT Tests as needed.

Review 1 Test Results

< [AHT] Rev 1 N 48v +52 M +40 0v N			
L	72.0 dB	+	Pol
C	71.9 dB	+	Pol
R	69.9 dB	+	Pol
RS	73.4 dB	+	Pol
LS	73.2 dB	+	Pol
LFE	75.2 dB		
O L O S R x O 1000 Oct			

Review 2 Test Results

< [AHT] Rev 1 N 48v +52 M +40 0v N	
L	Ref
C	5.2 ft
R	6.9 ft
RS	2.2 ft
LS	1.8 ft
O L O S R x O 1000 Oct	

Auto HT Tests review 1 and review 2 test screens indicate the results of the previous performed Auto HT Tests.

To review the test results of Auto HT Tests 4 and 5, select the ETG function of the SP395A. To review the test results of the Auto HT Tests 6-7, select the RTA function of the SP395A. Highlight the memory # field and choose the memory number in which the test data was stored. Click on the "R" field to recall and view the captured memory showing you the ETG measurement captured during the auto test.

Test 4 Reverb Decay Time (ETG): Starting Memory # +1
Test 5 Early Reflection (ETG) Left = Starting Memory # + 2, Right = Starting Memory # + 3)
Test 6 Noise (RTA): Base/Starting Memory # +4
Test 7 Subwoofer Frequency Response (RTA): Starting Memory # +5
Test 8 System Frequency Response (RTA): Starting Memory # + 6)

Auto HT Tests Conditions

A few room and speaker conditions may at times alter the measurement results of the Auto HT Tests. Here are some common conditions and recommendations.

- When the Auto HT Tests are running it is important to have very little room noises. Noises can easily be mistaken for sound wave fronts used for speaker delay and polarity testing. Keep ambient noises to a minimum.
- When the Auto HT Tests are running it is important to have clear unobstructed paths for sound waves from the speaker to the microphone at the listening position. Remove obstructions and do not walk around in the room during testing.
- A room that is excessively “live” or has very reflective wall surfaces can alter polarity measurements. This is especially true when using dipolar surround channel speakers, as rear wall reflections combine with direct sound wave fronts. Test individual drivers manually with the Speaker Polarity test holding the mic. in front of the speakers at a distance of 1-2 times the diameter.
- The Auto HT Speaker Polarity test is designed to work at SPL levels of 75 dB. If levels are initially low, the polarity tests may be inconclusive.

Memory Transfer –RS95 Reporting

The Auto HT Tests create data in memories that may be used to create a professional report documenting the performance of a home theater. The data can be transferred into the Sencore Report Software RS95.

When it is desired to create a Pre-Calibration/Post-Calibration report with the Sencore Report Software, it is recommended that you choose 0 for a starting memory on the SP395A for the first non-calibrated Auto HT Tests. After you have made receiver adjustments, choose 10 as a starting memory for the second post calibration Auto HT Tests. Memories 0-6 then contain the Pre-calibration data while memory locations 10-16 contain the post calibration test data.

To transfer memory data to the PC and RS95 software connect a straight through serial cable from the SP395A RS232 Serial jack to the PC’s RS232 Serial jack. You may use the SP395A’s Serial cable used to connect to the DAG during the Auto HT Tests, but the null modem adapter should be removed. Place the SP395A in a serial transmit mode by selecting the “PC Interface/About function within the Utility Functions menu. You can jump to this function by selecting the “x” field in the bottom center memory section of the Auto HT Tests display menu.

Note: You need to have RS95 revision 3.00.07 or greater to be compatible with the SP395A Auto HT Tests.

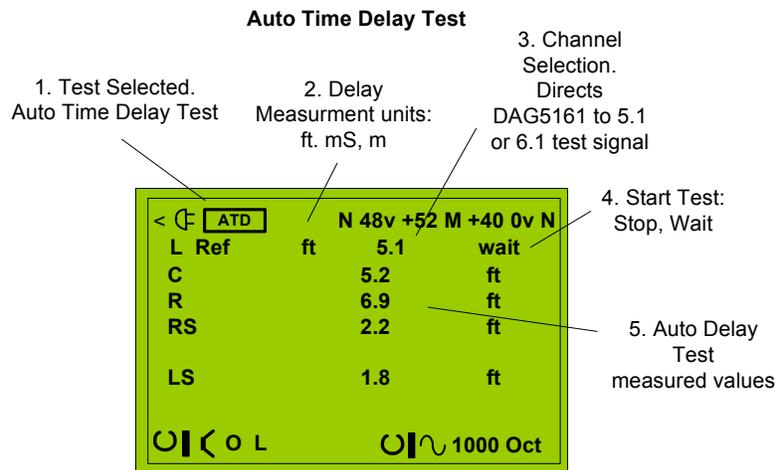
Auto Time Delay

The Auto Time Delay test is the same test as test 3 of the Auto HT Tests. The test requires the use of the Sencore DAG5161 and the same setup as used for the Auto HT tests. Please see the Auto HT Tests Setup section of this manual for setup instructions.

The Auto Time Delay test measures the time delay from each speaker to the listening position in a home theater surround sound system. Measurement results are indicated compared to the distance/ time of the left front channel speaker which is used as a reference. Therefore, distance/time measurements of all the other loudspeaker channels in the surround system indicate the difference in distance/time compared to the left front channel speaker. A properly aligned multi-channel surround system should have equal distance/times from all speakers to the listening position. A properly calibrated system should show nearly 0 on all speaker channels on the Auto Time Delay Test.

The Auto Time Delay test commands the DAG5161 generator to output an automated time delay test signal. This signal is a series of sound pulses, directed to each of the main five or six speaker channels. The pulses are spaced 100 mS apart. The measurement mic picks up the sound waves and the SP395A uses the arrival times to calculate the time between the pulses to determine speaker distance/times.

If you are having difficulty getting the speaker delays to zero out using repeated Auto HT Tests, switching to the Auto Time Delay test can save time. The Auto Time Delay test provides a continuous test with measurement updates.



Auto Time Delay Test screen and control.

1. **Test Function** - Identifies the Auto Time Delay Tests as the active function.
2. **Speaker Delay Measurement Units** -This control field toggles the delay units between feet (ft), milliseconds (ms), and meters (m) for the ETG test functions. For convenience you may set this to match the settings used by the AV receiver to simplify speaker time synchronization.

3. **Channel Selection** - This toggles the mode between 6.1 (DTS-ES), and 5.1 (Dolby AC3). This selection determines what signal is applied by the Sencore DAG5161 and adds a CS (Center Surround) test to the auto tests.
4. **Start Test** - This field is used to start the test. Initially it is set to “Stop”, meaning that the test is not running. To start the Auto Time Delay Test, select this field and click the control knob to toggle its state to “Wait.” Once the test starts running, it can be stopped by clicking this field.
5. **Measurement Values** – Displays the measurement values or results of the Auto Time Delay test.

Auto Time Delay Test Operation

The Auto Delay Test is not fully automated as in the Auto HT Tests. You must select the DAG5161 “Auto Delay Tests” signal within either the 5.1 or 6.1 signal selection. Then turn the DAG5161 output on or enable it to apply the test signal to the sound system.

1. **Select Auto Time Delay Test:** With proper setup complete, see the “Auto HT Tests Setup” section of this manual, select the Auto time Delay Test within the Tools menu of the SP395A. The Auto Time Delay Test screen will appear.
2. **Select Setup Settings:**
 - A. Select the 5.1 or 6.1 test signal in the Channel Selection field to match the sound system being tested.
 - B. Select the Speaker Delay Units to agree with the A/V Receiver’s adjustment units, either ft., mS, or m.
3. **Setup DAG5161:** Select the 5.1 or 6.1 Multi-Channel signal on the DAG5161. Select the “Auto Time Delay” test signal clip. Highlight and click the knob to start or enable the generator output. You should here the click, click, click or pulses sequencing from each speaker repeating after several seconds delay.
4. **Perform Auto Time Delay Test:** In the Auto Time Delay test menu highlight the start field. It may indicate “WAIT” or “STOP.” Click on this field to start measuring. The SP395A triggers on the first pulse and calculates the delays relative to the first reference pulse from the left front speaker.

Utility Function

The SP395A Utility Function menu has the following options:

Audio Scope
Save Settings
Setup & Calibration
PC Interface/About



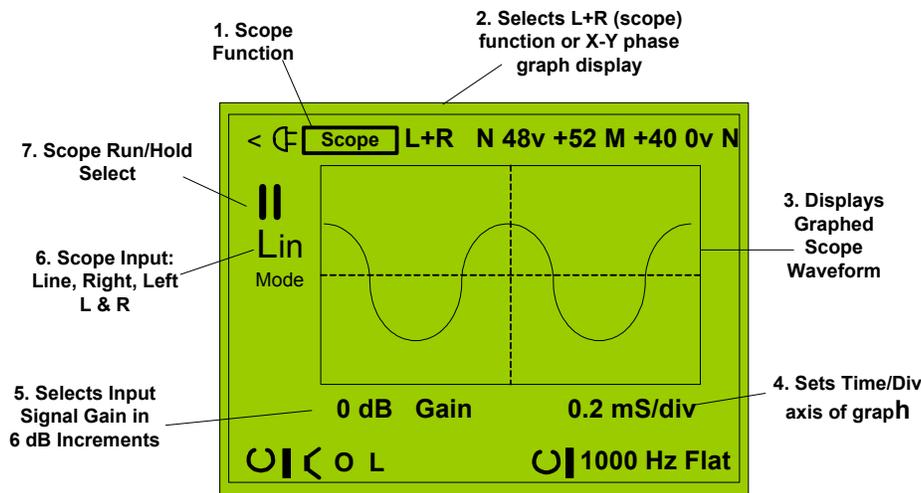
Audio Scope

The Audio Scope works with either or both of the inputs. The Audio scope is an audio-bandwidth digital oscilloscope. It has a sample rate of 48 kHz, meaning that it is useful for viewing signals up to 10 kHz. It provides AC only input coupling to 3 Hz for viewing AC only component of the waveform.

Use the Audio Scope to look for audio waveforms, identify audio distortion, clipping, polarity problems and the like. It provides a good graphical representation of the audio wave shape.

The Audio Scope auto-triggers on repeating waveforms with a trigger level set to $1/256^{\text{th}}$ of the selected scale. The display is refreshed approximately every $\frac{1}{2}$ second.

Audio Scope Screen Description



The Audio scope is an audio-bandwidth digital oscilloscope.

- 1. Scope Function:** Indicates the Audio Scope function as the active test.
- 2. Scope Mode:** This sub-menu field selects L + R or X-Y modes. The L+R mode is an oscilloscope mode for viewing audio waveforms graphed voltage vs. time. The X-Y mode is a graphical phase comparator of two audio inputs of the same frequency. The X-Y (phase) requires both a left and right input from a stereo source, to observe program material phase. For example, an audio signal generator and the output from an analog tape machine when adjusting the azimuth of a tape head.
- 3. Graphed waveform:** Graphed audio waveform or phase display for viewing.

4. **Horizontal Scale:** Select the desired horizontal (time) scale using this field. The time scale has 8 divisions from the left to the right of the scale. The time selections range from .1 to 10 mS. This is the time for one of the scale divisions.
5. **Gain** Sets the gain to add to the input signal, in increments of 6dB. This is useful to see a very low input or noise signal on the scope. The range is 0 to 96 dB. The gain scale reflects the scale of the digital 16-bit input stage. If you see what looks like clipping on the maximum vertical scale as the signal runs into the top of the display, then you are over-driving the inputs. Select a lower gain setting.
6. **Display Mode:** Selects the display to show the right, left or sum of both the left and right input waveforms. The table below lists the input modes and shows the resultant display on the scope screen and the bar graph level meter

Input Mode	Scope Signal Displayed	Bar Graph Level Meter
L	Left Line Input	Left input signal, dBu
R	Right Line Input	Right input signal, dBu
L&R	Both inputs, overlaid, triggered on the left input	Sum of left and right signals, dBu

7. **Scope Run/Hold** - Click this field to turn the Audio Scope on and off. Turning the scope off freezes the waveform that is currently displayed. (Changing the time/div or amplitude/div while the display is in "Hld" will not change the display).

How to Setup the Audio Scope- Waveform Viewing

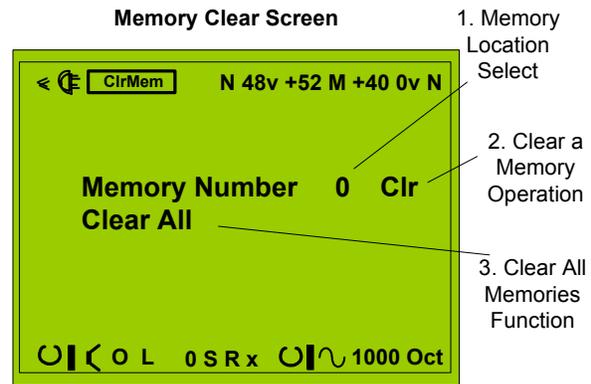
1. **Connect the input signal.** Plug the audio source(s) into the L or R inputs on the right side of the SP395A. If **NOT** using microphones turn off the phantom power by changing the 48V to 0V in the top Tool Bar phantom voltage field.
2. **Select the Utilities function of the main menu.** Move the cursor to *Utilities* and click the knob. **Select the Audio Scope function.** Move the cursor to highlight *Audio Scope* and click the knob.
3. **Select the mode.** Select the scope mode or *L +R* mode.
4. **Select the input:** Select Left (Lin) or Right (Rin) to select the input.
5. **Select RUN:** Be sure Run/Hold selection is in the RUN (I I) mode.
6. **Adjust the display amplitude** for a display that has sufficient height, but is not clipped. Selecting a Gain setting that is too low causes the waveform display to be clipped.
7. **Select an mS/Div setting:** Select a time setting to see 1-2 waveforms.
8. **View the input signal.** (Selecting "Hold" freezes the display)

Clear Memories

The SP395A's internal memories containing test data can be cleared individually or all cleared with a single command within the Clear Memory function or display screen. You can select individual memory locations or numbers to clear or use the "Clear All" to erase or clear all unit memories of test data.

Clear Memories Control

- 1. Memory Select** – selects the memory to be cleared by the "Clr" field.
- 2. Clr** – clears the selected memory when this field is highlighted and the control knob is pressed or clicked.
- 3. Clear All** – clears all memories of test data when this field is selected (highlighted) and the knob is pressed or clicked.



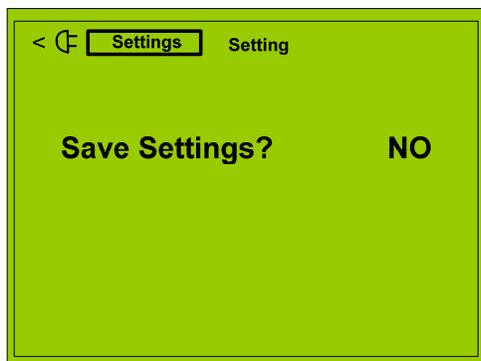
Save Settings

The Audio Integrator stores its current setup with a "Save Settings" function. This function is used to save a snapshot of all the current Audio Integrator settings to non-volatile memory. Once saved, all the current settings are recalled on the next and subsequent times that the power is turned on.

Fields saved include: All Toolbar settings, including phantom power on/off, input gain, auto-ranging, monitor section settings, and the generator settings; most function parameters, including filter selections and modes. Anytime you want to update the saved settings, run this function again.

How to save current settings

- **Select the Utilities menu.** Highlight the Save Settings "NO" function.
- **Set Save Settings to Y.** Change the N to a Y.
- **Exit to menu.** Click the back arrow to exit the function. The settings will be saved in non-volatile memory and will be restored the next time that you turn on the SP395A.



The Save Settings function stores your favorite unit setup.

Setup & Calibration

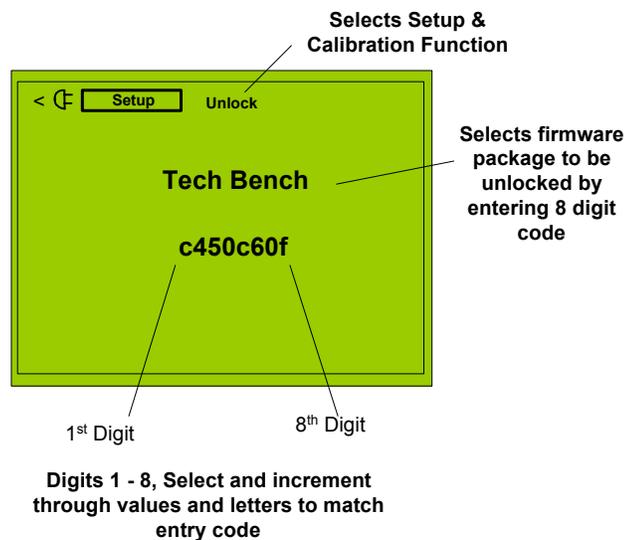
The Setup and Calibration utility has several options: Unlock, General, Input calibration, Output calibration, SPL calibration, and Counter Calibration. Each Audio Integrator is fully tested and calibrated before it leaves the factory. All calibration constants are stored in non-volatile memory. Accurate operation depends on this calibration. For this reason, only the General Setup, Unlock and SPL Calibration menus are covered here. Contact the Sencore Service Department if you feel your unit needs re-calibration.

Setup – Unlock Firmware

Many of the Audio Integrator functions are part of optional firmware packages. To activate these additional measurement capabilities, you need to purchase a firmware package from Sencore. Sencore issues you an unlock code to enter into the SP395A. Contact the Sencore sales department directly to purchase a firmware package. Once a package is unlocked, it will remain available on that Integrator.

When you receive the unlock code for a firmware purchase, you enter the code into the instrument using the Utilities function of the Setup & Calibration.

Firmware “Unlock” Setup Menu



Selecting “Unlock” within the Setup & Calibration submenu is used to enter unlock codes activating test functions which are part of firmware purchases.

1. Selects Setup or Calibration Function: Selects Unlock, General, Input, Output, SPL, or Counter options for setup and/or calibration.
2. Selects the firmware package for entry of 8-digit unlock code.
3. Selects each digit for incrementing through values 0-9 or a-f for unlock code entry.

How to unlock a firmware package

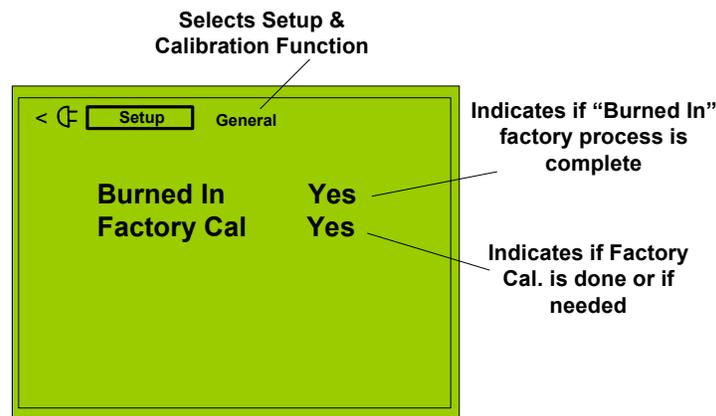
1. **Select the Unlock sub-menu.** This is a sub-menu of Setup & Calibration which is under UTILITIES in the main menu.
2. **Select the package to unlock.** Use the cursor to select and then click on the name of the package to get to the one that you wish to unlock.

3. **Enter code.** The code is an 8-character hexadecimal value. It contains the numbers 0-9 and the letters A-F. One at a time, select the character position, beneath the name of the package, and click and turn the encoder to change the value to match the character in your unlock code. Do this for all 8 characters.
4. **Exit function.** When you are done, exit the function to store the values. The function should now be unlocked.

Setup - General

This menu has fields for miscellaneous data values that are stored in the SoundPro. Each is described below. Since you can alter the contents of the calibration values stored in the SoundPro, be careful with these data fields! ***If you are unsure of what to do, contact Sencore technical support before altering any setting.***

General Setup & Calibration Screen



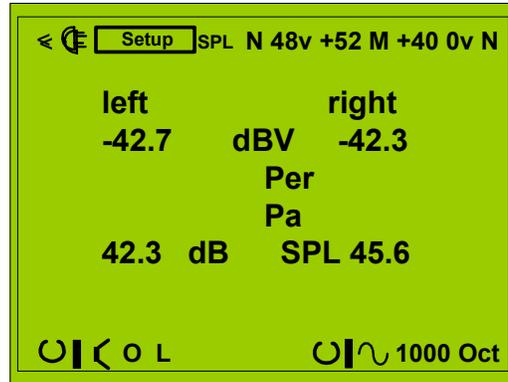
The General screen provides information regarding Burn-In, Factory Calibration and Restoring Calibration.

1. **Burn-in** - This field indicates whether or not the SoundPro has been through the burn-in process.
When a SoundPro is first powered up in manufacture, it enters a burn-in and self-calibrate state. Several constants are stored at this time. The unit may be forced through the burn-in process again by setting the burn-in data field on the General Setup menu to No, exiting the menu, and powering the unit off and back on. The burn-in procedure takes one hour to complete. Note that this may affect the main calibration.
2. **Factory Calibration** - This field indicates whether or not the SoundPro has been through the factory calibration process. The factory calibration process requires a special calibration fixture. *Contact Sencore technical support before changing this field.*

SPL Calibration

The Audio Integrator has been fully factory calibrated and should not require in-the-field calibration. However, you can check and recalibrate the microphone SPL calibration with a known accurate SPL calibrator.

SPL Setup & Calibration Screen



The SPL Setup & Calibration feature permits microphone calibration.

Calibrating a microphone using an SPL calibrator

If you have access to an SPL calibrator you do not need the calibration value, nor do you need to know the units. Connect the microphone to the calibrator and change the Cal value until you read the correct value on the screen.

1. **Connect microphone** - Connect the microphone(s) and turn on phantom power, if required.
2. **Turn on the calibrator** - Attach the calibrator to the microphone and turn it on.
3. **Set gain** - Set the input gain to the highest value that does not result in a clip indication (Clipping results in blinking gain field in the top tool bar)
4. **Calibration Values** - Adjust the Cal fields until the value shown on the screen matches the calibrator output level.
5. **Exit function** - Exit the function to store the values into non-volatile memory.

When you click on the “<” to return to the menus, the new calibration constants are stored in non-volatile memory.

Calibrating a Microphone – Entering Calibration Constants

A second method of calibrating a microphone is to enter the microphone’s known calibration constants into the SPL calibration screen on the SP395A. When purchasing a microphone, such as the Sencore Precision Low Noise Microphone ANSI Type 1 (#SP-PMIC1), the mic is supplied with the calibration constants values. These are values typically specified in units of dBV/Pascal (dBV/Pa).

The calibration constant values may also be specified in dBV/ubar. If so, you need to convert these values to dBV/Pascal for entry into the SP395A SPL calibration display screen. To convert dBV/ubar to dBV/Pascal, add 20 dB to the dBV/ubar value. For example: If you are provided a reading of -60 dBV/ubar, adding 20 dB results in -40 dBV/Pascal.

$$\text{Formula: dBV/ubar} + 20 \text{ dB} = \text{dBV/Pascal}$$

Note: If the specification is rated in mV/Pascal you will need to convert it to a dBV/Pascal value. Use the following formula.

$$\text{dBV} = 20 \times \log (\text{mV}/1000)$$

To calibrate the microphone by entering calibration constants:

1. Select Setup & Calibration under the Utilities menu.
2. Select the “SPL” setup or calibration function.
3. Select the left or right input calibration field. *Note for most measurements the left input is used for microphone pickup and sound/acoustic measurements.*
4. Enter the dBV/Pascal value into the calibration field.
5. Exit the Setup & Calibration display menu by selecting the “<” icon. The new calibration constants values are stored into non-volatile memory upon exiting the setup menu.

PC Interface/About

This screen provides unit information and technical support contact information about your Audio Integrator. It shows the serial number, tech support phone number and web page address. It displays the version number and date of the installed firmware.

PC Interface/About Screen



The PC Interface/About screen provides unit and manufacturer information plus allows RTA, ETG graph data to a computer

This display screen also allows you to transfer memory data, captured RTA and ETG screens, to a computer for storage, analysis or printing. The data is in a comma delimited, ASCII format and includes headers that identify the data.

To transfer data:

1. **Connect the supplied serial adapter cable** between the Serial Interface port on the Integrator and an open serial port on the host computer. This connects the PC serial port to the serial port on the SP395A.
2. **Start the application program** such the *SoundPro Audio Report* software to transfer memories from the Integrator to the PC program.

Computer Interface

The SP395A Audio Integrator communicates to a PC or Mac computer using RS232 serial communication format through the DB-9 serial connector. It does not pass serial data through the USB port. A standard straight-through serial cable is used.

SP395A offers a range of capabilities when it is interfaced to a computer and the computer is loaded with associated software. You can transfer instrument data memories to a computer for printing, upgrade the unit operating firmware, display the test results of some functions in real time on a computer screen or, control the SP395A for automated testing. Here is summary of the SP395A's computer/software automated capabilities.

Interface Software Utility: Transfer test results held in unit memory data to a PC. Memory functions include the RTA, SPL, FFT, Energy-Time Graph, Amplitude Sweep and Impedance Sweep tests.

Upgrader Software: Download firmware revisions and upgrade the SP395A operating firmware.

TerraLink Software: See real time test results on the computer screen for most or all test functions.

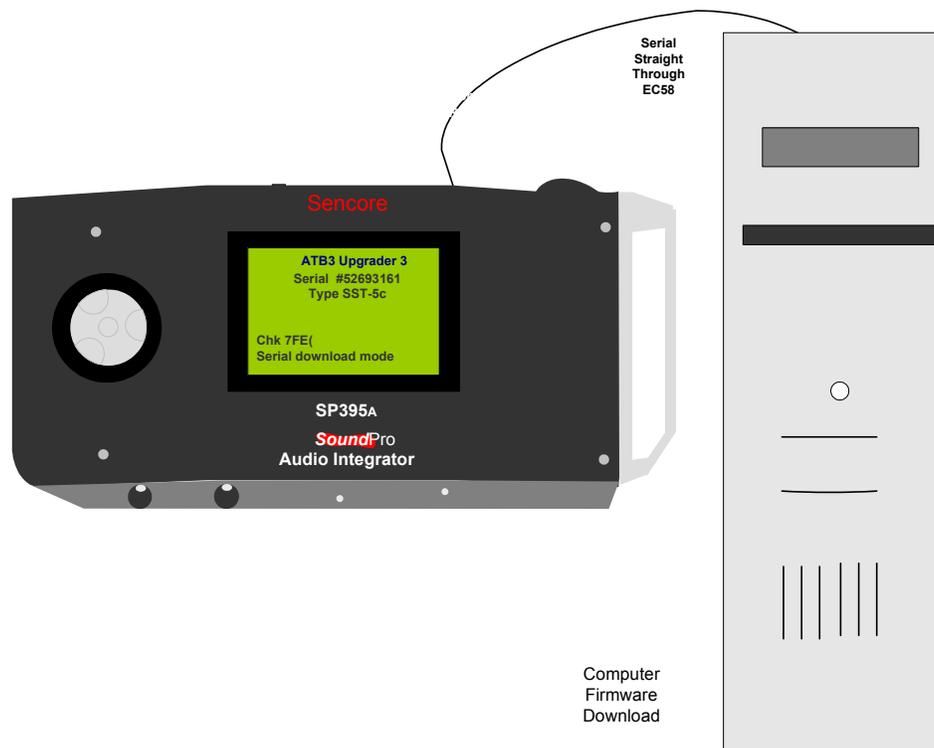
Uploading memories to a Computer

The SP395A can be interfaced to computer software or special applications programs for the purpose of moving memory data captured by the SP395A's test functions to a PC. Any test function in which memory data can be captured can be exported. This includes the SPL, RTA, FFT, Energy-Time Graph, Amplitude Sweep and Impedance Sweep tests. The memory data can be used by the computer program for analysis, storage, report generation or printing. The data is in a comma delimited, ASCII format and includes headers that identify the data. Please call Sencore to inquire about software availability.

Upgrading Firmware in the SP395A

The SP395A Audio Integrator is firmware upgradeable in the field. As product improvements and factory upgrades to the operating firmware program become available, the new program can be loaded into the instrument via a computer and serial interface connection.

Firmware upgrade files will be posted on the Sencore web site under the SP395A product page. They will be identified by version number. The current firmware versions in the instrument can be found in the PC Interface/About submenu. It is listed on the 3rd line down from the top of the screen as a version number. (example: Version 1.67 11/30/07) To determine the firmware version, select the Utilities in the main menu and PC Interface/About in the submenu.



The SP395A firmware can be updated in the field using a PC along with Upgrader Software.

The SP395A Audio Integrator firmware may be field-upgraded to new firmware by using the serial port on the instrument and a PC which meets the minimal requirements.

Minimum requirements for upgrading an SP395A include:

- A PC running Windows 98 or 2000, or a Mac running system 8.6 or later, but before OS X, unless running in Classic mode.
- A serial port on the PC, or a serial port on the Mac (USB is not supported directly).
- Standard serial cable.
- The SP395A Upgrader software (a PC or Mac program).
- The new version SP395A firmware or software program (to be downloaded into the SP395A).
- Unique software unlock code.

The Upgrader software is available as a free download from the Sencore web site. The current SP395A software is also available for downloading. To get your unique unlock code; use the

request form on the Sencore web site. You will need to have your SP395A serial number and the version of the software that you wish to install, and Sencore will give you your unlock code. If you have not registered your SP395A, you will be required to do so at this time. Note that you may be required to provide proof of purchase to obtain a free software upgrade.

Go to the Sencore web site, www.sencore.com, go to the SP395A product page and locate the download firmware section. Follow the instructions there. You will have to download the Upgrader program if you have not done so previously, and the new software upgrade.

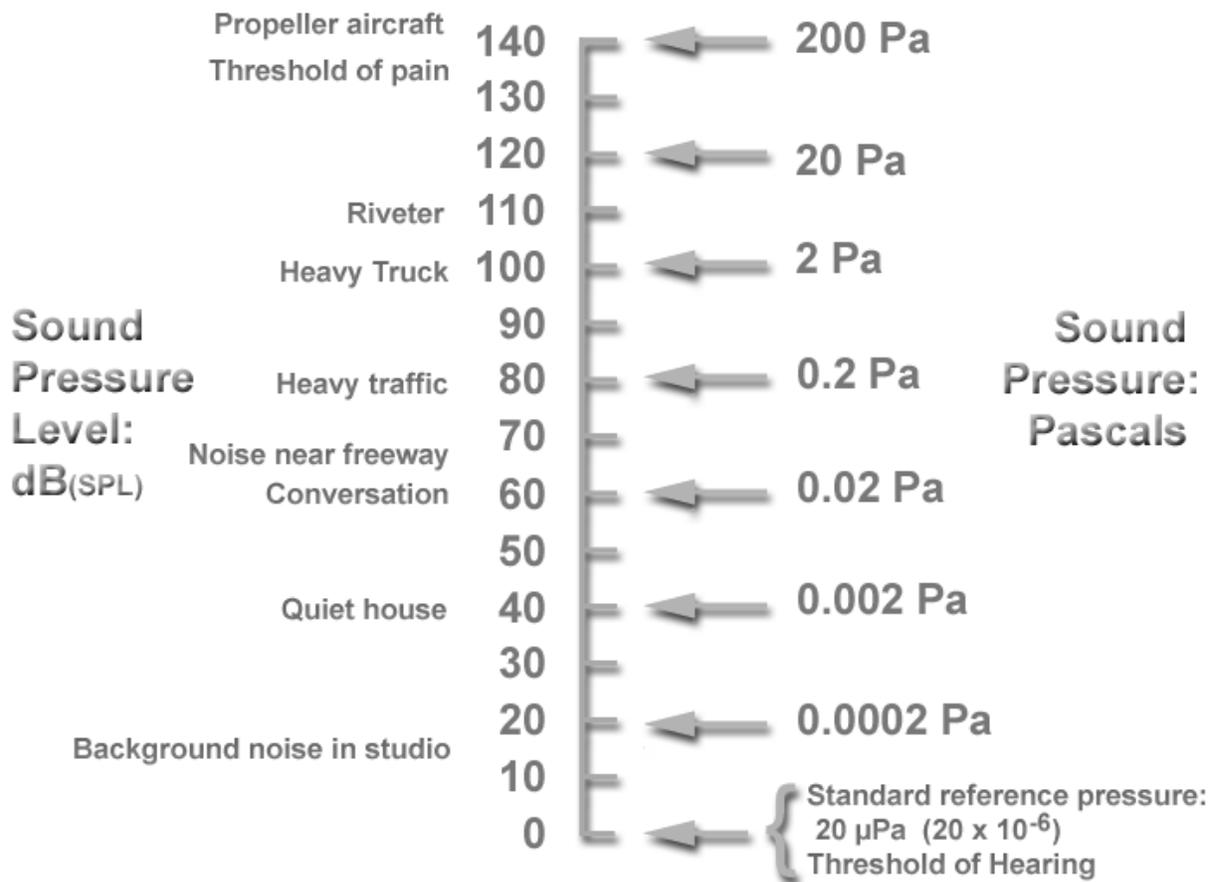
The program in the SP395A that communicates with the PC (called the Monitor) is stored in protected, non-volatile memory. Barring physical damage or abuse, it cannot be erased. This means that even if a faulty program is somehow loaded into the SP395A, the Monitor program will be safe, and a new, corrected program can still be downloaded to make a functional unit. In fact, several different programs can be stored on the PC hard disk, and loaded in whenever desired. The unlock code works with a specific SP395A and upgrade, so be sure to keep the unlock code safe.

TerraLink Software: Real-Time Computer Interface

TerraLink has the ability to communicate with a SP395A in real time. This means that for the functions for which this feature is enabled, you can see on your computer screen the results of the function as they are occurring. The functions currently enabled for real-time display are the RTA, the SPL meter, the FFT, the Energy-Time Graph, and the amplitude and impedance sweeps functions. TerraLink runs on both Windows and Mac computers, and communicates using a standard RS-232 serial port. TerraLink also allows plotting memories directly and report generation. See the Sencore web site for more information on TerraLink.

APPENDIX A - dB SPL Scale

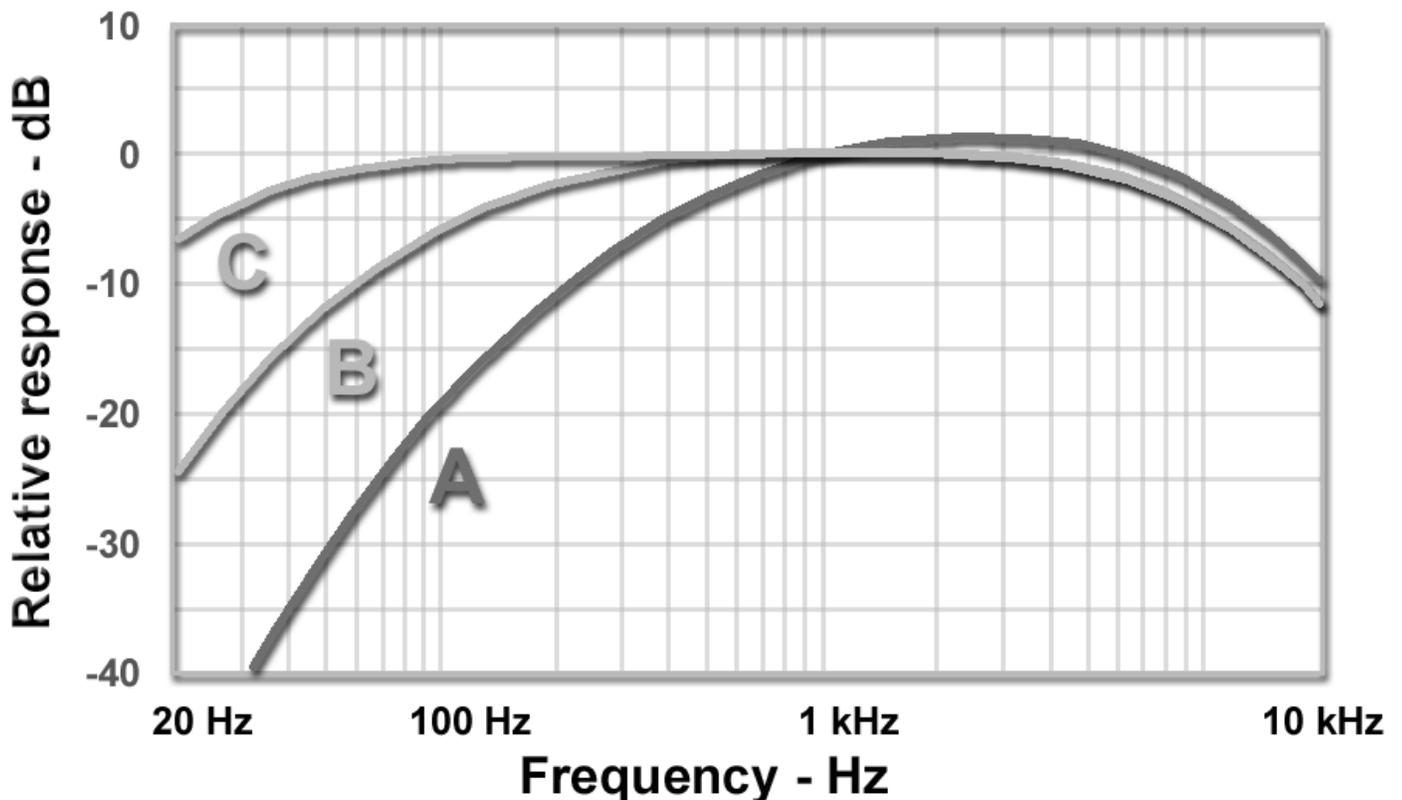
The 0 dB SPL reference level is set at 20 μ Pa (pressure equal to the threshold of hearing at 1 kHz). This places the upper threshold of pain near 135 dB SPL. Unless otherwise noted, SPL readings use this reference and omit the "SPL" notation. For example, a sound level expressed as 60 dB is actually 60 dB SPL, referenced to 20 μ Pa.



APPENDIX B – ANSI Weighting Curves

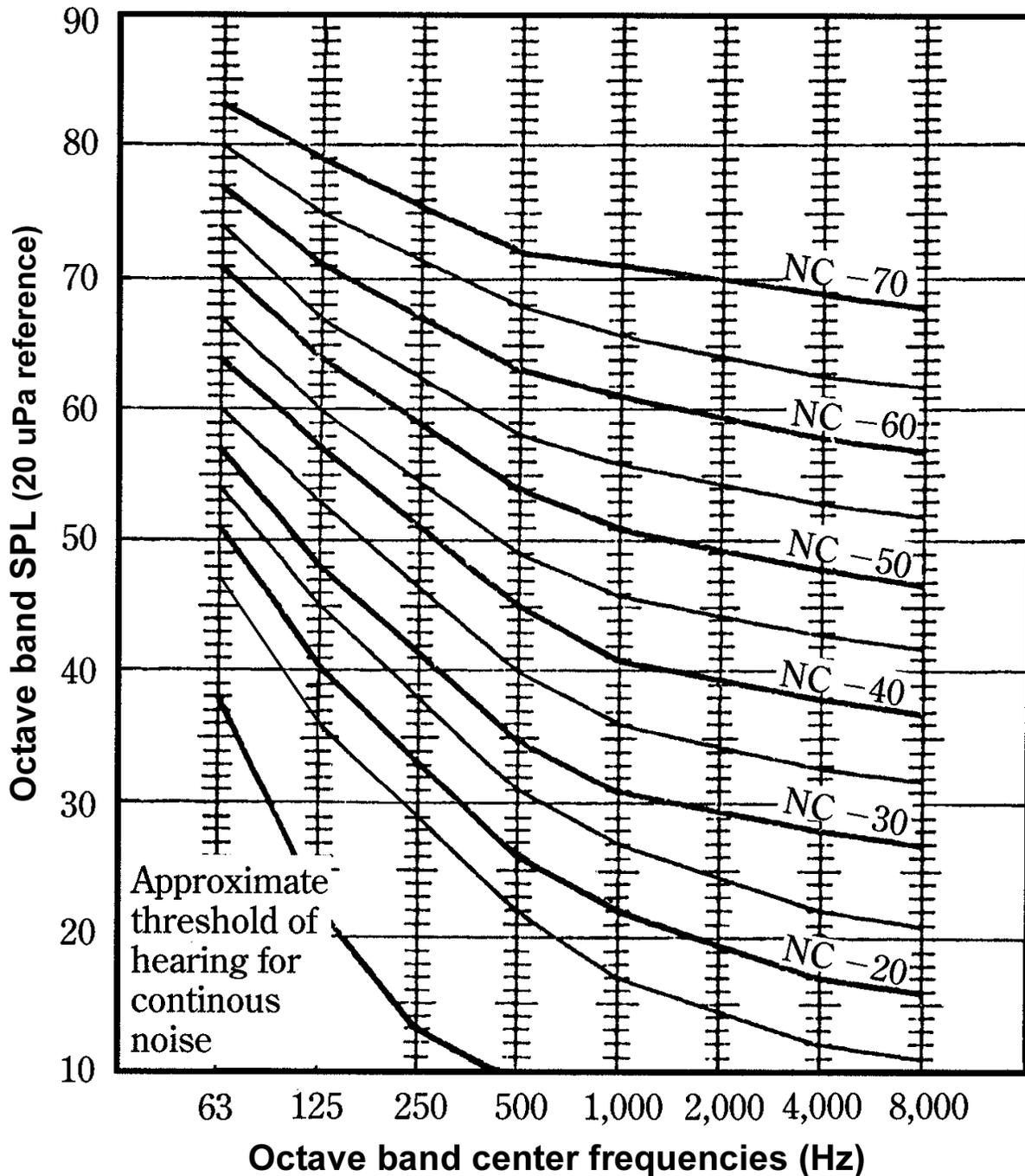
SPL meters convert sound pressure variations into an electrical signal that changes in amplitude, according to the sound pressure. The display shows the sound pressure level with a flat or linear response to all audio frequencies. Our ears however, do not have the same sensitivity for all frequencies. Instead, the sensitivity of our ears changes with sound level and frequency. In order for SPL readings to more closely correspond with what we hear, electronic filters are added to the SPL measurement circuitry. These filters, or weighting networks as they are commonly called, filter the measurements to correspond to the various sensitivities of our ears.

Three standard ANSI (American National Standards Institute) weighting curves are used in SPL meters. They are simply called A, B, and C weighting. The A-weighted curve approximates the 40 phons equal loudness curve, and is used when measuring SPLs of 20-55 dB. The B curve closely follows the 70 phons curve, and is used when measuring SPL levels of about 55-85 dB. The C curve, which closely follows the 100 phons curve, is used when measuring SPLs from about 85-140 dB. Weighted SPL readings are often referenced by the corresponding weighting curve; dB(A), dB(B) or dB(C).



APPENDIX C – ANSI Balanced Noise-Criterion Curves

Simple SPL measurements do not represent the background noise our ears hear. Because our ears are less sensitive to low frequency noise than to high frequency noise, noise levels at higher frequencies must be given more weight than levels at lower frequencies. The result is a set of curves called Noise Criterion or NC curves. The curves are arbitrarily numbered for easy reference. The Noise Criteria (NC) rating of a room is the number of the lowest curve that is not exceeded by any of the noise in the room. To determine an NC rating, the noise levels at 8 different octave band frequencies are measured and compared to the Noise Criteria curves.



Appendix D: Serial Protocol for Accessing SP395A Memories

This document details the serial protocol for retrieving the contents of the SP395A memories via the serial ports.

Serial hook-up and protocol

The SP395A connects to the computer's serial port with a straight-through serial cable. The standard SP395A serial cable may be used for this purpose.

The RS232 protocol is 19200 baud, no parity, 8-bit word length, and 1 stop bit.

Transmit Memory

All SP395A or previous SP395A versions after 2.91 transmit 231 bytes of data. Allow enough space in the serial buffer to hold the full 231 bytes.

To cause the SP395A to transmit a memory, send it the following ASCII serial data: Xnnn

Where nnn is a 3-character memory number, with leading 0's. For example, to obtain memory number 24, send X024

Note that there is no terminator (line feed or CR) for this data string.

The SP395A will immediately transmit 231 bytes of data for the requested memory number. The data sent is as follows:

Xnnn repeat of requested memory number

Then, 128 sets of byte pairs:

low byte low 8 bits of word

high byte high 8 bits of word

chksum 8-bit checksum. This is the sum, truncated to 8 bits, of the previous 230 bytes transmitted.

The receiver should verify the checksum and then assemble the bytes into 256 16-bit words.

Memory Data Format

The data transmitted from the SP395A has the following format, as shown in an example C struct:

```
struct memstr
{
char id[4]; /* Xnnn – memory number sent */
union
{
short idata[120]; /* int form of data block */
char cdata[240]; /* byte form of data block */
} data;
short tags[7]; /* extra data sent */
short memtype; /* memory type field */
unsigned char chksum;
};
```

The possible memory types are:

```
#define MEMTYPE_UNUSED 0
#define MEMTYPE_RTA 1
#define MEMTYPE_RTA_LOW 2
#define MEMTYPE_FREQRESP 3
#define MEMTYPE_IMP 4
```

```
#define MEMTYPE_ETC 5
#define MEMTYPE_NOISE 6
#define MEMTYPE_TDA 7
#define MEMTYPE_TDA_PHASE 8
```

The data stored in the data block and extra data sent vary by memory type:

RTA, RTA LOW

data block stores 120 1/12 octave integer RTA magnitude value data points in 10 * dB.
No data stored in extra data.

FREQRESP

data block stores 120 magnitude value data points in 10 * dB.
No data stored in extra data.

IMP

data block stores 120 impedance value data points in 10 * Ohms.
No data stored in extra data.

ETC

data block stores 120 magnitude value data points in 10 * dB.
First extra field stores the window index number, corresponding to the time window selected. 0 ==

15 ms, etc.

Second extra field stores the octave band filter in effect:

no filter 0

static char pm ec14[] = {" 31"}; 1

static char pm ec15[] = {" 63"}; 2

static char pm ec16[] = {" 125"}; 3

static char pm ec17[] = {" 250"}; 4

static char pm ec18[] = {" 500"}; 5

static char pm ec19[] = {"1000"}; 6

static char pm ec20[] = {"2000"}; 7

static char pm ec21[] = {"4000"}; 8

static char pm ec22[] = {"8000"}; 9

NOISE

data block stores 120 magnitude value data points in 10 * dB.

First extra field holds time index, 0 = min time.

Second extra field holds input source, 0 = MicL

Third extra field holds filter,

None 0

A 1

B 2

C 3

static char pm ec14[] = {" 31"}; 4

static char pm ec15[] = {" 63"}; 5

static char pm ec16[] = {" 125"}; 6

static char pm ec17[] = {" 250"}; 7

static char pm ec18[] = {" 500"}; 8

static char pm ec19[] = {"1000"}; 9

static char pm ec20[] = {"2000"}; 10

static char pm ec21[] = {"4000"}; 11

static char pm ec22[] = {"8000"}; 12

Fourth extra field holds mode, 0 = AVG, 1 = PEAK

TDA

tbd

WARRANTY AND SERVICE INFORMATION

WARRANTY

Your SP395A Audio Integrator has been built to the highest quality standards in the industry. Each unit has been tested, aged under power for at least 24 hours and the every function and range was retested to insure it met all published specifications. Your instrument is fully protected with a 1-year warranty.

SERVICE

Save the original shipping carton and packing material for reuse if you ever need to ship your generator, or return it to the Sencore factory for repair.

1. If you are returning the unit to Sencore for service, call the service department at the number below and enclose the following information: owner's address, billing information, purchase order (if applicable), name and phone number of contact person, description of problem and reason for return. No return authorization is required.
2. Completely enclose the unit inside a plastic bag to protect its finish and prevent foreign material from getting inside.
3. Cushion the unit equally on all sides with a minimum of 3 inches of padding material. Pack the padding tightly enough to prevent the unit from shifting during shipment.
4. Seal all seams on the container with strapping tape.
5. Send the packed unit to the following address (we recommend shipping via United Parcel Service):

SENCORE FACTORY SERVICE

**3200 Sencore Drive
Sioux Falls, SD 57107**

Most service repairs are completed within 72 hours. If you need to ask about your unit, the Sencore Service Department phone number is:
1-800-SENCORE (736-2673) or FAX (605) 339-7032

Fill in for your records:

Purchase date: _____

Serial Number (on opening power-on display) _____

Software version (in Utilities ABOUT menu) _____

SENCORE

3200 Sencore Drive
Sioux Falls, SD 57107
Call 1-800-SENCORE (736-2673)
www.sencore.com

*Innovatively designed
with your time in mind.*