SIM® SYSTEM II *Acoustic Test and Measurement System*

User Friendly Interface

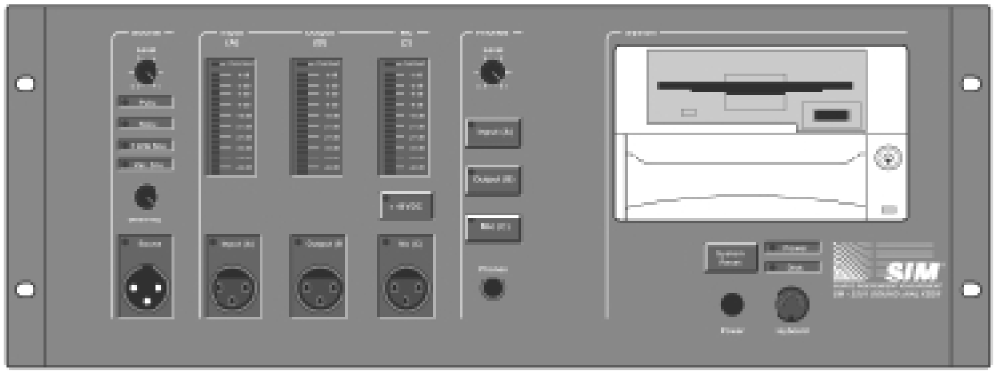
High-resolution color display

Source Independent Measurement

Full Bandwidth, log frequency axis

Integral Microphone Pre-amplifier

Laser Printer Support (b/w and color)



SIM System II is a powerful, compact instrumentation product line comprising the SIM-2201 Sound Analyzer, SIM-2403 Interface Network, and a selection of software options, microphones, and accessory cables. The instrument is optimized for making audio-frequency measurements of an acoustical system and applying precise electronic corrections to adjust the system response.

SIM System II implements Meyer Sound's Source Independent Measurement technique, a dual-channel method which accommodates statistically unpredictable excitation signals. Any excitation signal which encompasses the frequency range of interest (even intermittently) may be employed to obtain highly accurate measurements of acoustical or electronic systems. (For example, concert halls and loudspeaker systems may be characterized during a musical performance, using the program as the test signal.)

Housed in a four-space rack-mountable industrial chassis, the SIM-2201 Sound Analyzer performs 32-bit floating-point audio signal measurements with >100 dB dynamic range (actual input signal range is greater because of selectable gain). The instrument permits two-port measurements between any two of three front-panel inputs (one microphone with switchable phantom power, two isolated line level), and incorporates a rear-panel multi-pin interface for SIM-2201 Sound Analyzer Specifications automated measurements of two-channel systems. Optional hardware and software upgrades permit up to sixty-four analysis channel capacity.

Measurement data may be displayed as amplitude vs. time (Impulse Response), or amplitude and phase vs. frequency (Frequency Response). A single-channel Spectrum mode is provided, and frequency domain data are displayed with a logarithmic frequency axis. A Delay Finder function determines and internally compensates for propagation delays.

The SIM-2201 incorporates a front panel-controllable precision signal generator with low-distortion sine wave, pink noise and modulated, weighted pulse outputs; multi-segment level meters for each measurement input; a re-movable hard disk and DOS format 1.44 Mbyte floppy disk drive for data storage; high-resolution color monitor output; and dedicated rear-panel multi-pin system interface connectors. A headphone output is provided for aural monitoring of the measurement inputs. User-friendly software with pull-down menus streamlines operation, and measurement data may be exported to disk in ASCII format for post-processing or laser printed for presentation.

Superior engineering for the art and science of sound.

Meyer Sound

Physical Model

SIM System II is designed for not only characterizing, but also electronically correcting acoustical systems. Its architecture and nomenclature follow a physical model consisting of a loudspeaker in a room (object of measurement) with a measurement microphone, and a parametric equalizer (correction network) connected in series with the input signal. The excitation is assumed to be neither totally random nor predictable, though known test signals such as noise or stepped sine waves may be used.

The instrument's three input ports are connected respectively to the correction network input (A), the network output (B), and the measurement microphone (C).

Frequency Response

In Frequency Response mode, the SIM-2201 displays amplitude, phase and coherence (or signal-to-noise) data for transfer function computations between any two of the three measurement inputs. The frequency response (amplitude and phase) transfer function is computed by dividing the cross-power spectrum by the auto-power spectrum of the reference channel.

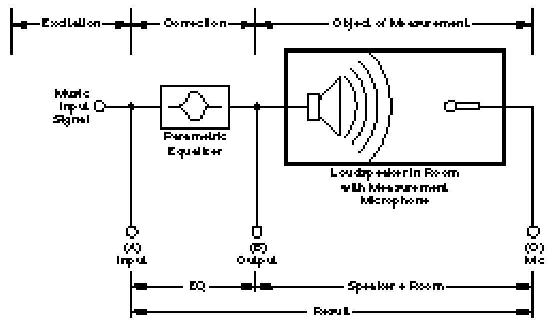
Three different frequency response measurements may be selected:

- **Room + Speaker** – the unequalized system response, measured by comparing the equalizer output and microphone;
- **EQ** – the equalizer response, measured across the equalizer from input to output;
- **Result** – the corrected system response, measured by comparing the equalizer input and microphone.

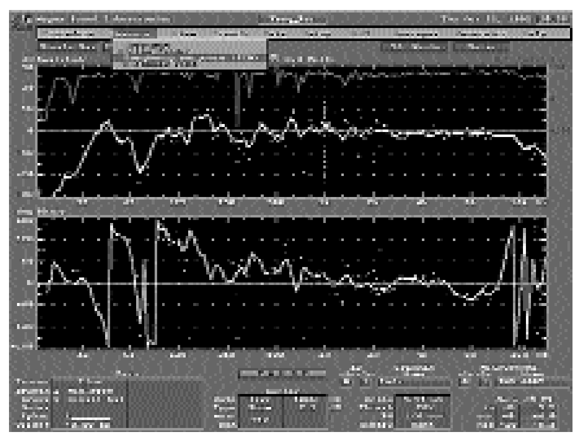
Frequency Response data are displayed in two windows, with amplitude and coherence (or signal-to-noise) in the upper window and phase in the lower. A selectable Group View displays all three of the above frequency responses at once (v. 2.3 only), and an inverse of the equalizer response is available for adjusting correction networks to match system response aberrations. A Zoom feature reveals fine details of the system response.

The maximum resolution in Frequency Response mode is $\frac{1}{24}$ octave ($\frac{1}{30}$ octave in v. 2.0 Lab Zoom mode). $\frac{1}{6}$ or $\frac{1}{3}$ octave smoothing may also be selected to facilitate observing general trends; when smoothing is employed, missing data points in the response curve are interpolated if they fall within the smoothing interval.

Measuring a loudspeaker in a room and correcting its response with parametric equalization

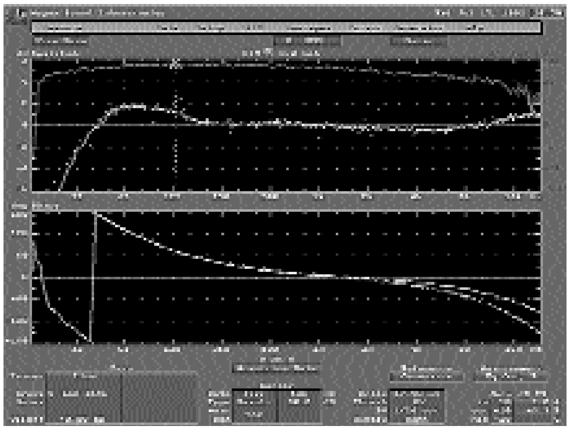


Frequency Response mode display – Amplitude Response with Signal-to-Noise (top window), and Phase Response (bottom window)



Frequency Response measurement of three parametric filter bands with equal Bandwidth and Gain settings

Signal-to-Noise



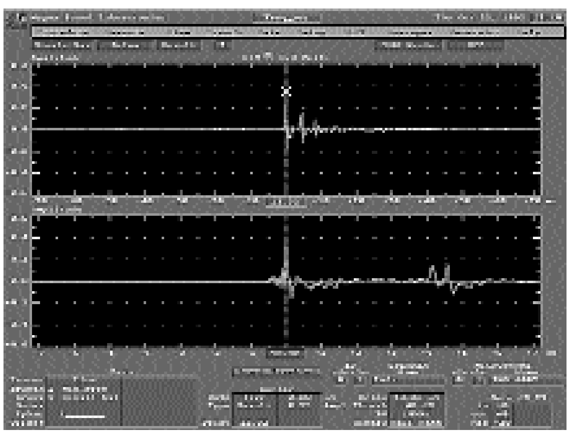
Frequency Response measurement of an analog audio tape recorder illustrating S/N+D by frequency (red trace, upper window)

SIM System II provides for display of dynamic signal-to-noise + distortion at each frequency, as an alternative to coherence.

The signal-to-noise of a measurement may be calculated as the coherence divided by one minus the coherence. Where the coherence is 1 (no contamination), signal-to-noise is theoretically infinite (>100 dB, the dynamic range of the instrument). At 0.5 coherence, where the noise equals the signal, signal-to-noise is 0 dB.

Note that nonlinearities in the system being measured may convert some of the input signal to energy at other frequencies. These distortion components are treated as noise by the coherence function, so they will be included in the signal-to-noise.

Delay Finder

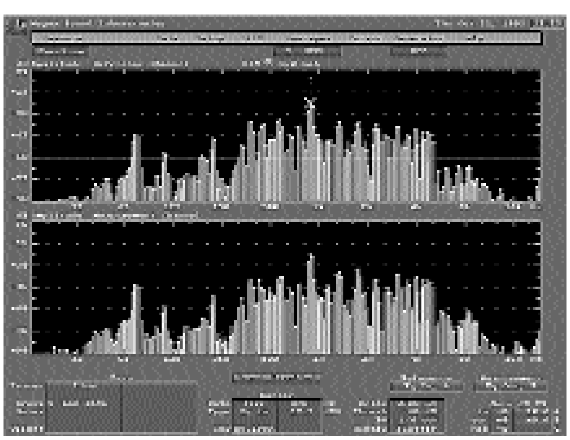


Delay Finder mode display of loudspeaker impulse response with boundary reflection; ±70 msec. (upper trace) and ±7 msec. (lower trace)

In Delay Finder mode, the SIM-2201 Sound Analyzer displays a windowed impulse response of the system under test, permitting accurate identification of reflections. The SIM-2201 calculates the propagation delay between the reference and measurement channels, and sets an internal delay to compensate for any time offset and synchronize the channels. The function is accurate to within approximately $\frac{1}{8}$” (for sound in air at STP) with propagation distances up to 1000 feet.

An associated External Delays Procedure determines and displays the time offset between two measured systems, and may be used in setting delay lines to synchronize physically separated systems.

Spectrum



Spectrum mode display with Flat top window shape and Peak Finder cursor

Spectrum mode employs single-channel computation to display the spectral content of a measurement input with $\frac{1}{6}$th octave resolution. Amplitude flatness is ±0.1 dB.

Optimized for sine wave stimulus, Spectrum mode is used primarily for distortion analysis, and can resolve up to nine harmonics. Distortion components are summed and displayed as a percentage relative to the amplitude at the cursor position.

A Peak Hold function, in which the cursor finds and remains on the spectral line with the highest instantaneous amplitude, is provided.

EXPANSION OPTIONS

SIM System II is designed to accommodate varying levels of complexity in measurement and equalization requirements. Three versions are available.

v. 2.0

SIM System II v. 2.0 comprises the SIM-2201 Sound Analyzer with single DSP engine, and system software v. 2.0. Optimized for laboratory operation and automated measurements of two-channel systems, v. 2.0 is upwardly expandable to v. 2.3s or 2.3m.

v. 2.3s

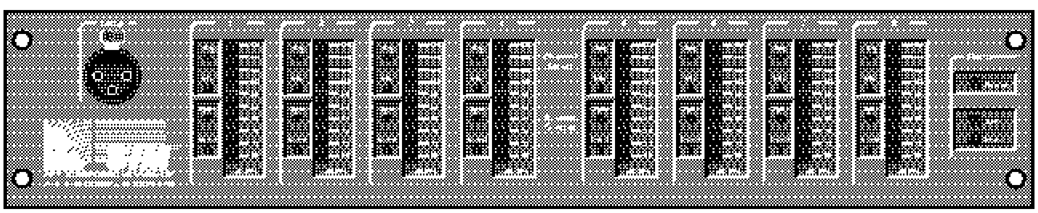
SIM System II v. 2.3s employs two additional DSP engines, v. 2.3s software and an accessory Stereo Interface Snake. It allows for simultaneous display of three live frequency response functions. This version is optimized for speed in analyzing systems with up to two channels.

v. 2.3m

SIM System II v. 2.3m employs system software v. 2.3, two additional DSP engines, interface card, accessory cables and up to eight SIM-2403 Interfaces.

At maximum capacity, SIM System II v. 2.3m accommodates sixty-four microphones and displays simultaneous real-time measurements of the room, equalizer and equalized system response for any of sixty-four separate measurement branches. Switching and selective muting of branches is controlled from the SIM-2201 Analyzer. With the ability to utilize multiple measurement microphones, accurately determine and compensate for multiple propagation delays, and efficiently manage large amounts of data, SIM System II v. 2.3m affords prodigious power for the most complex acoustical analysis tasks.

SIM-2403 INTERFACE NETWORK



The SIM-2403 Interface Network is a switching unit that may be controlled remotely from the SIM-2201 Sound Analyzer (v. 2.3m only). Incorporating internally calibratable microphone pre-amplifiers with phantom power and dB SPL multiple-segment meters, each SIM-2403 provides access and control of up to eight distinct microphones and correction networks. The unit may be connected in a Mute mode, allowing muting and level control of individual channels, or a Safe mode when muting is not desirable. Front-panel LEDs indicate muting functions and status.

Measurement Inputs	**Channel Capacity**	24 (8 microphone, 16 line level)
	Impedance	10k ohms (A or B) balanced ISO™ Input, 20k ohms (Mic)
	Phantom Power (Mic only)	48 VDC ±2 V
	Maximum Input Level	24 Vpk balanced (A or B), 1.0 Vpk (Mic)
	Mic Sensitivity Range	2.5 mv/pa to 12.5 mv/pa (internally adjustable)
	Frequency Response	±0.2 dB 20 Hz to 20 kHz
Output Channels	**Type**	300 ohms balanced, >24 Vpk swing capability
	Frequency Response	±0.2 dB 20 Hz to 20 kHz, -6 dB 5 Hz and 100 kHz
	Attenuation Range	0 dB to -12 dB (front-panel screwdriver adjustment)
	Power	100/120/220/240 VAC (switchable), 25 watts
	Shipping Weight	44 lbs (20 kg) including cables and gooseneck lamp
Meters[1]		20 segment LED dB SPL in 3 dB steps, VU ballistics, 0.18 sec time constant, crest factor 4
Notes		1. Maximum meter reading is 120 dB SPL with 12 dB of headroom (132 dB SPL overload point).

SIM® SYSTEM II MEASUREMENT MODES

SIM (Source Independent Measurement) is a two-channel acoustical analysis method in which the excitation signal may be independent of-that is, not generated or determined by-the measurement system. It enables highly accurate frequency response measurements where it is inconvenient to employ calibrated test signals, or where an excitation already exists.

Conventional FFT analyzers have been used to make such two-port measurements, but they are subject to significant errors when the excitation is statistically unpredictable, especially when the output is contaminated with noise. Averaging can help, but with conventional methods, some error term always remains in the result.

SIM System II overcomes these limitations with new algorithms that substantially eliminate errors.

Signal Thresholding

The SIM-2201 Analyzer permits establishing an amplitude threshold value for the measurement input signal. When the signal exceeds the threshold at a given frequency, the transfer function is computed for that spectral line. When the signal is below threshold, it is ignored. Signals that overload the instrument's inputs are automatically rejected.

In this fashion, the instrument builds a valid frequency response over time. If two or more samples are acquired at a given frequency during the measurement period, they may be averaged. Signal thresholding enhances the signal-to-noise of the measurement. It is useful when the excitation contains a given frequency only sporadically, is otherwise sparse in content, or varies widely in level.

Coherence Blanking

Coherence is the output power of the system under test that is due to the excitation signal, divided by the system's total output power. Its value varies between 1 (no contamination) and (100% contamination).

For each frequency data bin, the SIM-2201 Analyzer establishes a preset coherence threshold that is tied to the number of averages employed. Where the coherence drops below the threshold, the frequency response traces for that bin are blanked from the display. The SIM-2201 therefore does not display data of questionable accuracy, simplifying interpretation of the data.

Coherence blanking enhances measurement accuracy in the presence of substantial output signal contamination. It is useful for rejecting output noise, reverberation, or other effects that may bias the measurement.

Constant-Q Transform

The SIM-2201 utilizes a short measurement window at high frequencies and progressively longer windows for each successive lower octave. Since the excitation is not correlated to the measurement window, reflections that extend from one window into the next are averaged out due to uncorrelated arrival.

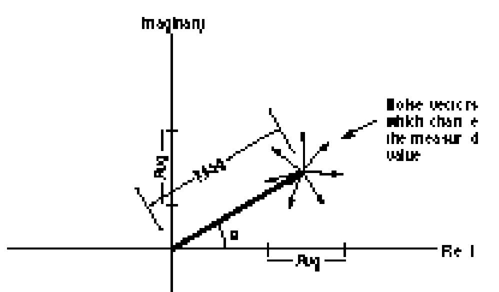
The use of this near constant-Q transform allows measurement and correction of near-field reflections while properly rejecting reverberation, and displays frequency-domain data with equal resolution per octave.

Vector Averaging

Vector averaging is the statistically correct way to remove both periodic and random output noise contamination. It is employed whenever a time-invariant system is to be tested in a noisy environment.

Often viewed as magnitude and phase, audio signals may also be converted into a vector containing real and imaginary components at each frequency. A contaminated signal thus may be visualized as the actual vector with noise vectors surrounding it. In vector averaging, the real and imaginary parts are linearly averaged simultaneously. Since the noise vectors are uncorrelated to the excitation, they statistically reach a value of zero, yielding the actual magnitude and phase value for that frequency.

Vector averaging yields the most accurate estimate of the system response as long as the frequency response is stable with time. RMS averaging is also available for occasions where the system frequency response is time-variant (for example, when making outdoor measurements under windy conditions).



Vector representation of an audio test signal with noise contamination

Features

- Equal resolution per octave
- Adjustable amplitude threshold
- Vector or RMS averaging
- "Smart" coherence blanking
- SIM-2403 Interface Network
- Software upgrades
- Additional DSP cards
- Multi-conductor interface cables

Benefits

- ±0.1 db amplitude accuracy
- Flexible excitation options
- Unbiased frequency response estimate
- Enhanced signal-to-noise
- Rejects periodic and random noise
- Rejects long-term reverberation

SIM-2201 SOUND ANALYZER SPECIFICATIONS

Freq. Range		8 Hz to 22 kHz
Screen Update Rate		<0.7 sec (Frequency Response or Spectrum Mode) 0.2 sec to 1.3 sec (Delay Finder Mode)[1]
Frequency Resolution	**Frequency Response Mode (Transfer Function)**	1/3rd, 1/6th, 1/24th octave (selectable)
	Spectrum Mode (Single Channel)	1/6th octave (fixed)
Averaging of Data	**Frequency Response Mode**	2 to 16 Vector or RMS, sliding, performed on a per-frequency-bin basis
	Delay Finder Mode	1 to 999 Vector, linear accumulator
	Delay Finder Accuracy	±10 µsec time, ±0.5 dB amplitude (lower trace)
Amplitude Accuracy[2]	**Absolute**	±0.1 dB 20 Hz to 20 kHz, Input (A) and Output (B); ±0.15 dB 20 Hz to 20 kHz, Mic (C)
	-1 dB Points	8 Hz and 22.5 kHz
	Channel Matching	0.1 dB Input (A) to Output (B), 10 Hz to 20 kHz, ±1 degree phase matching 0.4 dB Input (A) or Output (B) to Mic (C), 20 Hz to 20 kHz
Signal Alanysis Linearity		THD <0.025% with full scale signal
Minimum Resolvable Signal	**Input (A), Output (B)**	<-90 dBV Spectrum Mode (<105 dBV midband), <-70 dBV Frequency Response Mode
	Mic (C)	<-115 dBV Spectrum Mode, <75 dBV Frequency Response Mode
Maximum Measurement Signal Range		<1 µVrms (Mic channel) to >28 Vpk (A or B Input)
Gain Range		-20 to +50 dB relative gain, switched in eight 10 dB steps
Measurement Inputs	**Impedance**	10k ohms (A or B) balanced ISO™ Input, 5k ohms unbalanced; 8k ohms (Mic), balanced
	Phantom Power (Mic only)	48 VDC ±2 V, switchable on/off
	Maximum Input Signal	26.0 dBV Input (A), Output (B), -4.0 dBV Mic (C)
A/D Converters[3]	**Type**	16 bit Sigma Delta, 64 times oversampling, 3rd order noise shaping for 94 dB dynamic range
	Anti-aliasing filters	30-pole FIR digital and 5th order linear phase analog
Signal Generator	**Output**	300 ohms balanced, 24 Vpk maximum output level
	Sine Wave	16 Hz to 22 kHz tunable in two ranges, <0.01% THD 100 Hz to 22 kHz, <0.1% THD at 16 Hz
	Noise	Pink spectrum ±1.5 dB, 20 Hz to 20 kHz
	Pulse (Frequency Weighted)	Average repetition rate 1.25 sec, time modulated
Physical	**Storage**	1.44 Mbyte floppy disk drive, 120 Mbyte removable hard drive, 16 Mbyte internal RAM
	FRONT PANEL: Controls	Source Level control, Sine Frequency, Phantom Power switch, Phones Level control, Phones source select switches (3), System Reset switch
	Indicators	Power, Disk, Generator source waveforms on/off, Phantom power on/off, Measurement inputs on/off, Phones sources on/off, Measurement input levels (20 segment LED bar displays), Input Overload
	Connectors	Input (A), Output (B) and Mic (C), 3-pin XLR female; Source, 3-pin XLR male; Phones, 3 circuit monophonic 1/4-inch phone jack, 32 ohms minimum impedance; Keyboard, 5-pin DIN
	Rear Panel Connectors	Audio Interface, 39-pin Whirlwind W-1 female; Control Bus, D-25 female; Video Output, D-15 high density male; Serial Interface, D-9 male; RS-232 Serial Interface, D-25 male; Parallel Interface, D-25 female; Power, IEC receptacle
	Power	90-250 VAC switchable in two ranges, 50/60 Hz, 300 watts
	Dimensions	19" W x 7" H x 24" D
	Shipping Weight	60 lbs (27 kg) including keyboard, cables, software and manual, without video monitor
Notes		1. Screen update rate in Delay Find Mode is dependent on record length. 2. Frequency Response and Spectrum Modes 3. Three simultaneous channels

Meyer Sound Laboratories has devoted itself to designing, manufacturing, and refining components that deliver superb sonic reproduction. Every part of every component is designed and built to exacting specifications and undergoes rigorous, comprehensive testing in the laboratories.

Research remains an integral, driving force behind all production. Meyer strives for sound quality that is predictable and neutral over an extended lifetime and across an extended range.

LISTED 3K59 COMMERCIAL AUDIO SYSTEM

SIM - 04.102.083.01D

Meyer Sound Laboratories, Inc.
2832 San Pablo Avenue
Berkeley, CA 94702
tel: 510.486.1166
fax: 510.486.8356
e-mail: techsupport@meyersound.com
http: www.meyersound.com