SX200

SERIES

Symetrix

SX201

PARAMETRIC EQ/PREAMP

Features

- **EQ and notch filter**
- **Preamp input**
- **Balanced in/out**
- **Unbalanced in/out**

The Model SX201 Parametric Equalizer/Preamp is a high performance studio quality parametric equalizer/notch filter designed to handle both low level and line level inputs. Three fully parametric bands of equalization are provided, with +15dB boost and −30dB cut capability, allowing the SX201 to be used for both creative and corrective equalization. Overlapping frequency controls cover the entire audio range, from 16Hz to 20kHz. Bandwidth is continuously variable from .05 octave (for deep notch filtering), to 3.3 octaves (for smooth tone shaping). Distinguished by its superb sonic integrity, extremely low noise and low distortion performance, the SX201 delivers uncompromising quality in a remarkably compact package.

With separate line and preamp inputs, nearly any signal level may be used with the SX201. The line level input provides both balanced or unbalanced terminations, for levels ranging from a nominal −10 to +8. The preamp input is unbalanced, and is intended for use with low level signals, such as those from synthesizers, guitars, and electronic drums. The overall input level control allows the operator to set internal signal levels to match boost/cut conditions. To meet the demands of sensitive equalization and critical notch filtering, high headroom, high slew rate active devices are used throughout the SX201 to deliver the best possible dynamic range and superb audio quality. Separate balanced and unbalanced output line drivers, capable of +18dBm and +24dBm respectively, are provided.

SX202

DUAL MICROPHONE PREAMPLIFIER

Features

- **Exceptional sonic performance**
- **+48 volt phantom powering**
- **Input levels to +14dBv**
- **L, R, and L, + R outputs**

The Model SX202 Dual Microphone Preamplifier is an ultra clean two channel stereo/mono preamp, intended for use in the most critical digital and analogue recording situations. However, the SX202 is ideally suited for broadcast use, and for general purpose paging and public address applications.

Built around the same chip used in a well-known recording console widely acclaimed for its exceptional sound quality, the SX202 makes world-class performance affordable. Used in place of older preamp designs, the SX202 offers substantial sonic improvements with its solid stereo imaging (less than 10° phase shift at 20kHz), excellent transient handling (its positive and negative slew rates are symmetrical), very low noise (approaching the theoretical limit), and almost unmeasurable distortion (.007%).

Its unique combination of features and performance make the SX202 a very versatile product, designed to deliver superior performance in a wide variety of circumstances. Variable gain inputs, with 15dB pads, allow the SX202 to handle any input up to +14dBv. Switchable +48 volt phantom powering is included for professional condenser microphones. One channel is equipped with a polarity switch, to correct for improperly wired cables or unresolvable mic placement problems. In addition to the individual outputs from each preamp, a left + right output is included to provide a combined, mono feed. Balanced/unbalanced line drivers are on 3-conductor ¼″ (TRS) connectors, and are capable of +24dBm and +18dBm respectively.

Specifications

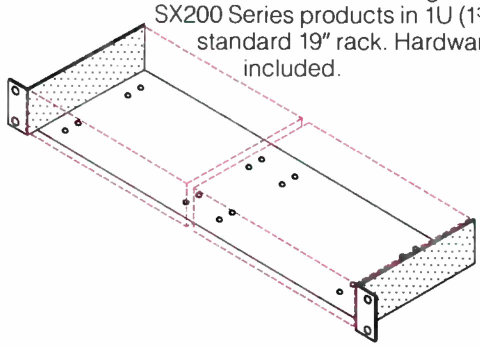
Specifications	SX201	SX202	SX203	SX204
Freq. Response +0dB, −1dB	20-20kHz	20-20kHz	300-3kHz[2]	20-20kHz
THD, 1kHz, 600 ohms				
0dBm	.01	.007	.2[2]	.01
+24dBm	.02	.02	.3	.02
Input Type[1]	A, B	A	C	A
Connector	¼″ TRS	XLR-3	RJ-11	¼″ TRS
Max. Level	+18dBv	+14dBv	N/A	+18dBv
Output Type[1]	A	A	A	A
Connector	¼″ TRS	¼″ TRS	¼″ TRS/TS	¼″ TRS
Max. Level	+24dBm	+24dBm	+24/+18	+24dBm
S/N Ratio				
−40 in/0 out		97dB		
0 in/max out	101dB	115dB	>80dB[2,4]	95dB
Gain	unity	50dB	23dB[3]	20dB

Note 1 – Type A: Balanced/unbalanced low impedance transformerless
Type B: Unbalanced
Type C: RJ-11 telephone line connection

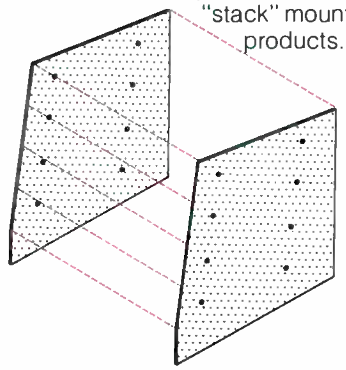
Note 2 – Telephone line limitations affect these specifications
Note 3 – Gain available, telephone line in to line level out
Note 4 – 600 Ω termination, unity gain
N/A – Not Applicable

SX200 Series Accessories

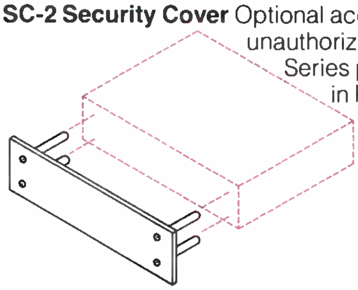
RM-2 Rack Mount Optional accessory shelf provides flush or recessed mounting for two SX200 Series products in 1U (1¾″) of a standard 19″ rack. Hardware included.



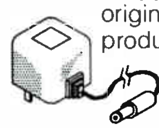
SR-4 Stack Rack Optional accessory rack provides "stack" mounting for four SX200 Series products. Hardware included.



SC-2 Security Cover Optional accessory cover prevents unauthorized adjustment of SX200 Series products when mounted in RM-2 or SR-4. Hardware included.



PS-2 Power Supply Replacement for power supplies originally included with SX200 Series, products sold in North America. Input 120VAC, output 16 volts, AC, 10VA.



Symetrix

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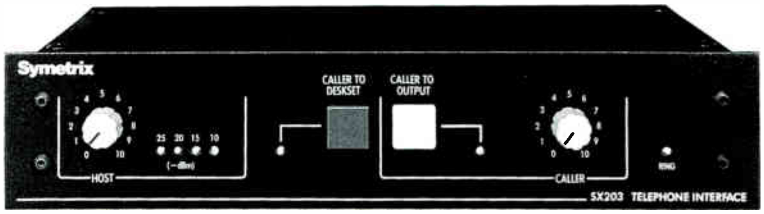
Signal processing at its best

Printed in the USA

SX203

TELEPHONE INTERFACE



Features

- **No balancing adjustment**
- **Very easy to use**
- **Positive tracking half-duplex**
- **Modular phone connectors**

The Model SX203 Telephone Interface is a single line coupling device that provides high quality audio transfer between telephone lines and professional audio equipment. Very fast hook up and ease of operation make the SX203 ideal for radio and television talk shows and news gathering, recording studios, teleconferencing, and rental systems.

With Symetrix' positive tracking half-duplex control circuitry, the SX203 responds very naturally, yet requires absolutely no time consuming null balance. Just plug a telephone line into the rear panel modular jack, and it's ready to work. An extra modular jack is provided for a standard telephone desk set, so calls may be originated. The SX203's performance is predictably excellent with any phone line, and with any kind of telephone equipment, so even the most inexperienced user always gets great results.

By virtue of the SX203's positive tracking half-duplex technique, only four control functions are required. Provided are separate controls for the caller level and the host level, and two push button switches to route the caller to either the desk set or to the SX203's balanced audio output.

Telephone line and desk set connections are made via RJ-11 modular jacks. The audio input is balanced, mic or line level (switchable, nominal −50dBm or 0dBm) on a ¼″ 3-conductor TRS connector. The audio output is available balanced at line level (nominal +4dBm) on a ¼″ 3-conductor (TRS) connector, unbalanced at line level (nominal +4dBm) on a ¼″ 2-conductor (TS) connector, and balanced at mic level (nominal −50dBm) on a ¼″ 3-conductor (TRS) connector.

SX204

HEADPHONE AMPLIFIER

Features

- **Extra headroom for Hi-Z phones**
- **Stereo or mono operation**
- **Four separate channels**
- **Individual level controls**

The Model SX204 Headphone Amplifier is a 1-in 4-out stereo device, specially designed to drive multiple headphones of any impedance. The SX204 is intended for monitoring use in recording studios, radio and television stations, and for multiple headphone distribution systems in churches, theaters and meeting rooms.

Although most headphone amps will drive low impedance headphones well enough, many cannot deliver sufficient voltage swing to drive high impedance headphones. Symetrix' proprietary high voltage converter technology solves this common problem, giving the SX204 the output voltage capability of a big, high-powered amplifier. As a result, the SX204's four independent stereo amplifiers can drive even high impedance headphones to maximum levels without distortion. Of course, there's plenty of power for low impedance headphones.

With the SX204, use headphones of any impedance, any efficiency, because each listener has control over the volume in his own headphones. Along with the individual level controls, an overall input level control is provided. A stereo/mono switch allows the left and right headphone outputs to be fed from just one side, and combines the left and right into mono when a stereo input signal is used. Balanced or unbalanced inputs are on 3-conductor ¼″ (TRS) connectors. Headphone outputs are on 3-conductor (stereo) ¼″ connectors.

Symetrix

TI-101
Telephone Interface

Data Sheet

The Symetrix TI-101 Telephone Interface is the most practical, yet economical device on the market today designed *specifically* for the connection of professional audio equipment to telephone lines in broadcast and production operations.
The TI-101 employs a carefully engineered electronic hybrid circuit which creates a maximum trans-hybrid loss, yielding effective isolation between your studio's send to the telephone line, and your caller return signal. Your net result is clear, intelligible telephone audio which helps you keep pace with today's rapid improvements in AM, FM, and TV audio.

The TI-101 provides you with not just some, but all of the features you need in a telephone interface. At the same time the unit is simple to connect, simple to adjust, and simple to operate. An output buss or mic pre-amp feed from your console connects to the TI-101 and sends your studio signal down the telephone line. The TI-101 allows you to optimize your send level for maximum intelligibility. At the same time you can adjust the unit's built-in send limiter to prevent overdrive of the telephone line. The return signal from the TI-101 to your console is just the caller's voice. The DJ or host voice has been effectively nulled out.

Impedance and level matching to and from the telephone line and your console are done correctly by the TI-101. In addition, you can bring the caller's voice up on your studio monitor speakers without feedback. There's no need to wear headphones if you don't want to.

One of the most important features of the TI-101 is that it is a true hybrid. This means that there is no objectionable "gating" as in "speakerphone" type phone boxes. You get natural, *two-way* conversation between host and caller.

Features
Caller equalization. A two band equalizer with 8dB of boost and cut at 400 Hz and 2.5k Hz brightens up the caller and enhances intelligibility.

Send limiter. The send signal passes through a limiter with user adjustable threshold level. When properly adjusted this limiter maximizes host level, prevents overdrive of the phone line, and helps improve the send and receive circuit isolation.

Receive compressor/expander. This circuit also has a user adjustable threshold level. Above threshold the compressor rides gain on the caller, helping to maximize level, and most importantly keeping things under control during caller-host "shouting matches." Below threshold a noise reducing expander takes over and is especially valuable for keeping long distance noise down during caller pauses.

Caller mute. A user provided remote contact closure mutes the caller instantly without clicks or pops.

LED clip indicators are provided for simple optimization of send and receive levels. When setting levels there's no guesswork involved. The user simply increases the level controls until the LEDs flash on and then backs off on the setting slightly for the optimum operating point.

Conference linking. Two TI-101's may be selectively linked together for simultaneous conferencing between two incoming telephone lines and the host.

Level compatibility. Back-panel gain switches permit the TI-101 to operate with virtually any professional mixer or console.

Bandpass filtering. A sophisticated elliptical filter is provided on the TI-101's send section which prevents studio generated signals outside of the telephone passband from interfering with telephone company signaling frequencies. On the receive section, sharp "Chebychev" filters limit the passband from 300 to 3k Hz preventing spurious signals on the telephone line from interfering with your broadcast audio.

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Professional Audio Products

TI-101
Telephone Interface

Input impedance		16.7K ohms (electronically balanced)
Output load impedance		> 600 ohms (transformer balanced)
Telephone port impedance		560 ohms (transformer isolated)
Nominal input and output level ranges		back panel switchable between −10dBm and +8dBm
Maximum output level		+20dBm
Maximum input level		+21dBm
Typical THD		.1%
Controls	send level, send limit, receive level, receive compress/expand, 400 Hz and 2.5k Hz equalization, conference	link, coarse, low frequency, and high frequency null adjust.
Visual indicators	LED's for indication of send clip, send limit, receive clip, receive compress/	expand, receive mute, and power on
Frequency response		(measured from telephone port to output port) 300 Hz to 3k Hz, ±3dB.
Typical transhybrid loss		20dB over the specified frequency bandwidth
Connectors	3 pin "XLR" type for input and output ports, dual banana posts for telephone tip and ring, ¼" phone jax for external	mute and conference interconnect cables.
Physical size		1¾" high, 19" wide, 6" deep (4.45 48.3 x 15.2 cm)
Shipping weight		7 lbs (15.4 kg)
Power requirements	60 Hz, 120 VAC, standard	50 Hz, 220 VAC, upon request
Construction	Aluminum front panel, plated steel chassis, all connectors pc mounted for	maximum reliability

In the interest of continuous product improvement and development, Symetrix, Inc., reserves the right to change or modify any of the above specifications or features, whenever, in our opinion, such a change produces an advantage mutual to our customers and ourselves.

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