Sanders Sound Systems

Preamplifier

OWNERS MANUAL

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INSTALLATION

LOCATION

Preamps usually are set on a shelf in your equipment rack. But sometimes shelf space is not available, so the Sanders Sound Systems Preamp was designed to be very thin to give you the option of setting it on top of another component.

You may place it on top of any component except those that get very hot, since the preamp's chassis is likely to interfere with the cooling of the hot component. For example, you shouldn't place the preamp on top of a hot power amplifier. You may place it on top of cool running power amplifiers, like Sanders Sound Systems' ESL amplifier. More likely you will place it on top of a component like a CD player or FM tuner.

INPUT CONNECTIONS

Signal input is made through gold plated RCA (unbalanced) or XLR (balanced) connectors. A total of six inputs are available. This is a "line stage" preamp, so although the inputs have specific labels, they are all at line level, are identical in function, and can be used with any line-level component.
The first 4 inputs (Balanced, CD, Tuner, Video) are "main" inputs. The last two (Monitor and Processor) are "loops" that have fixed level outputs associated with them. The "Monitor" loop is sometimes called a "Tape Monitor." It is designed to be used with an audio recorder (cassette, DAT, CD-R, mini-disk, etc.) where the currently selected input source is recorded. This output is unaffected by the overall system volume, so the recording level is not altered if you need to adjust the listening level. The "Processor" loop is identical to the "Monitor" loop and is designed to be used with a video processor — although it can be used with a second audio recorder if you prefer.

If you want to operate a turntable and play vinyl LPs, you will need a phono preamp. Sanders Sound Systems has a dedicated, matching phono preamp that can be connected to one of the inputs on this line-level preamp to properly play LPs. Please contact our factory for details.

OUTPUT CONNECTIONS

Both balanced and single-ended output connections are available. You may use both outputs simultaneously to drive two different amplifiers if you wish.

POWER CONNECTIONS

Be certain all associated equipment is turned off before plugging in the preamp. Insert the power cord into the AC LINE INPUT on the back panel and then connect it to an appropriate power source. Then you may turn on your other components.

The preamp is designed to be left on continually, so there is no power switch. By leaving the electronics on, they stay warm and are always in their optimum operating state and ready for immediate use. You can be confident of the finest sound at all times, and there is no need to let the unit "warm up" before starting a critical listening session.

Switching electronics on and off is abusive to them. It is much gentler and better for them to leave them on. They will last indefinitely when left on. The preamp uses just a few watts of power so power consumption is not an issue. All indicator lights are light-emitting diodes (LEDs), including the digital display, so they will never burn out or need replacement.

ELECTRICAL PROTECTION

Although not essential, it is a good idea to use a surge protector to prevent damage to your expensive equipment in the event of an electrical storm or other causes of abnormally high mains voltage. These are inexpensive and can be obtained from any hardware store.

Note that surge protectors do not insert anything in the power line, so will not adversely affect the sound of your audio system (they work by shunting excessive voltage to ground, so only operate when dangerously high voltages are present). You do not need to use an expensive line conditioner, but if you do, these normally will have surge protection built in so you do not need to use additional surge protection.

The AC line voltage of the amplifier is preset at the factory for your location. But it may easily be changed for use anywhere in the world. Contact the factory for instructions.

OPERATION

The Sanders Sound Systems Preamplifier is a very sophisticated, computer operated unit that has many functions. Despite its complexity, it appears very simple. Great care and thought were put into making it "user friendly" and ergonomically well-designed. All of the following functions are available from the front panel controls and by remote control. Each function is listed below with directions on how to operate it.

INPUT SELECTION

Press the appropriately labeled button on the front panel to select a source. When activated, a small, blue LED above the switch will light. The four "main" inputs (Balanced, CD, Tuner, Video) operate together so that selecting any one of these inputs switches out the others.

The "Monitor" button toggles between your recorder and whatever main input is selected. If the recorder has a monitoring function (where the recording can be played back during recording), you can set your recorder for playback and by pressing the "Monitor" button, you can switch between the input source and the recorder's playback. This way you can "monitor" and compare the quality of your recording to the source. If your recorder does not have a simultaneous playback feature, then you will hear the source when the "Monitor" button is pressed. Generally, high quality tape recorders (including DAT, cassette, and analog open reel) offer simultaneous playback and record, while this feature is not available on disk recorders.

The "Processor" button will select the video processor loop and will override all others, including the monitor loop. This loop acts exactly like the tape monitor loop and you can use it to operate a second recorder rather than a video processor if you wish.

MONO

Use this switch to toggle between monaural and stereo. The light indicates the mono position. This function is useful for checking to see that your speakers are in phase (a solid center image should be present in mono) and for checking system left/right balance.

Tip: For a speaker to image well, you must be equidistant from them. Using a measuring tape; measure from your listening chair to each speaker. Try to get within a quarter inch of being equidistant. The difference in the quality of the image may surprise you.

VOLUME

This knob adjusts the output level of the preamp in precise 1 dB steps across a range of 100 dB. This control rotates continually — it is not limited to the 280° rotation of conventional rotary controls. The control is detented so you can feel each 1 dB step. The digital readout shows the output level in 1 dB increments between the numbers "00" and "99."

This volume control is not a conventional potentiometer — it is an optical device that controls a microprocessor. Unlike conventional stepped resistor ladder attenuators, this microprocessor has no transient clicks or pops as you change the volume. Like a resistor ladder, the level of each channel is closely matched so the left/right balance will remain constant as you adjust the level up and down. The tolerance is within 0.1%, which is vastly better than even the best conventional, dual stereo potentiometers.

BALANCE

The volume knob is used to control several functions in addition to overall volume. Pressing the knob momentarily, (like a large push-button) will cause the preamp to change modes. In the window next to the number display are three indicator diodes, "Main", "Left", and "Right." In the normal "Main" mode (indicated by a blue light), the knob functions to control the overall system volume. Pressing the volume knob momentarily will switch to the next position ("Left"), and a color coded amber light will appear under "Left", and the "Main" blue light will go out. Simultaneously, the number display will show "00", which means zero attenuation (or maximum volume).

At this point, you may reduce the level of the left channel by turning the volume knob counter- clockwise as you normally would to reduce the overall volume. As you do, the numeric display will show you the number of dB you have reduced the left channel. Once the level has been reduced, you may, of course, turn the level back up to the maximum of "00" or anywhere in between. Reducing the level of the left channel will shift the image toward the right.

Pressing the volume knob again will switch to the right channel, the color coded red indicator light will appear under "Right", and the numeric display will again show "00" (or the previously set level for this channel). Adjust the level as desired. Pressing the volume knob again will return to the "Main" control.

In summary, press the volume knob to select the channel towards which the sound is shifted, then reduce the level until the image is centered. Then press the volume knob to return to the "Main" control. It's simpler to do than to describe.

SYSTEM GAIN

Sanders Sound Systems' unique balance control makes it possible to adjust the overall system gain. For example, let's assume you want to reduce the gain by 10 dB. Simply press the volume knob once, which will switch you to the left channel. Reduce the gain by 10 dB with the volume knob. Press the volume knob again, which will switch you to the right channel. Reduce the gain on that channel by 10 dB. Press the volume knob again to return to "Main."

The channels will be in perfect balance, but their levels will be reduced by exactly 10 dB. As proof of this, you will see that for a given acoustic output level, you will have to turn up the volume so the number indicator reads 10 dB higher.

Why would you want to do this? There are many reasons. For example, if you have relatively delicate speakers and an extremely powerful amplifier, you can overdrive the speakers and damage them. By reducing the gain, you can limit the output to any safe level you wish.

Another reason is for comfort. It is easy to accidently set the volume too high and get a very loud "blast" of music you didn't expect. By setting the gain so that the maximum loudness level is an indicated 99 dB, the level will never be excessive.

For the safety of both your equipment and your family, you may want to prevent teenage children from abusing your audio equipment (and their ears) in your absence. Just reduce the gain so that the system will only play at a moderate level, even when turned up fully.

Tip: Adjust the input levels (as described in the next section) before making overall gain adjustments. That way the overall gain will be correct for all inputs.

INPUT LEVEL CONTROLS

The level of each input can be adjusted so that you can match the levels of all the components in your system. This way, when you switch from one component to another, you won't get "blasted" with one and have to turn up the volume on another.

Adjusting an input is very similar to adjusting the system balance. To adjust an input, press and hold its selector button while you press the volume knob — then release them both. The number display will switch to "00" and the affected input light will flash, showing that you are in the "adjustment" mode. Now, using the volume knob, you can reduce the level just like you did with the balance function.

Note that this adjustment automatically modifies both the left/right source channels simultaneously. You do not have to adjust the left and right channels individually like the balance control.

To switch back to "Main", you may press any button, including the volume knob. Or you can simply do nothing and the function will "time out" in fifteen seconds and return to "Main" by itself.

Tip: Leave the quietest component at "00", and reduce the others as needed to match it. Make a note of the levels you set so that you can easily refer to them at a later time.

MUTE

A mute function is available on the remote control transmitter. Mute reduces the level by 20 dB rather than eliminating it, so that you can still hear the program material in the background. When muted, the number display will flash.

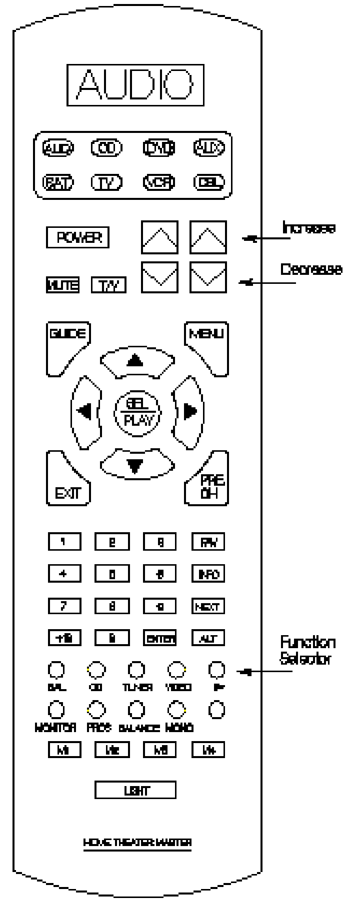
You can adjust the volume while the signal is muted. The number display will change to indicate the new setting and you will hear the muted level change. When you "un-mute" (by pressing the mute button again), the preamp will operate at the new level you selected.

REMOTE CONTROL

Most of us have a pile of conventional remote control transmitters we have to pick through every time we want to control any component in our systems. Rather than to add to "remote clutter", Sanders Sound Systems has supplied you with the very sophisticated *Home Theater Master* remote. This unit is a computer-controlled transmitter, into which you can put the IR codes from ***all*** your remotes. You can do this using pre-programmed codes, or by using its "learning" feature. You can then control all the components in your audio/video system with this single remote and eliminate the pile of others. This is a truly amazing unit and you will be richly rewarded by reading its booklet.

Sanders Sound Systems has installed the batteries and pre-programmed the remote so it is ready to use. If the remote has lost power for a long period of time, or if you change the function of the preamp control buttons and wish to revert back to the pre-programmed configuration, simply enter the code #148 in the usual manner as described on the back of the remote and in the booklet.

The information below describes pre-programmed operation:



Select AUD to operate the preamp controls. AUDIO will appear in the window.

Volume control and mute buttons operate in the normal manner.

Input selection is by direct access and is done by pressing the appropriate button in this row.

Additional controls include selection of Tape or Video processor loops, balance, and mono options from this row.

To take full advantage of all the features available from the *Home Theater Master* (including how to operate all your other components with it), read the comprehensive booklet included elsewhere in the packaging. In particular, it will be very convenient if you set up "punch-through" so that you can operate the volume control on the preamp from any other component selection. This way, when operating another component (like a CD player) you won't have to repeatedly switch the remote back and forth between CD and AUD to control both the CD transport and Preamp's volume — you will be able to control both from the CD position. Alternatively, you can press the CD selection button twice, and all the preamp controls (not just the volume) will work while using your CD player.

CARE AND CLEANING

If you wish to clean your preamplifier, use diluted, ammonia based cleaner. Window cleaners like "Windex" also work well. **Do not use any abrasive cleaners or chemical solvents like "Ajax", acetone, or paint thinners.**

Use particular care not to damage the aluminum chassis. Aluminum is a medium hardness metal and although it is anodized, it can be easily scratched by the careless use of tools during installation, or by rough handling.

The unit may overheat and the finish may fade if exposed to direct, unfiltered sunlight or intense heat for prolonged periods.

Save your box and packing materials. They will be very helpful for moving or if you need to ship the unit for any reason.

DESIGN PHILOSOPHY

The circuitry used in the Sanders Sound Systems Preamplifier is the result of an advanced and complete design process combining innovation and proven fundamentals. This process avoids both the limitations of total adherence to convention and the flaws resulting from inappropriate applications of clever circuit gimmicks.

Impedance isolation and matching is derived from FETs (Field Effect Transistors). While careful design can yield good results from any device type, FETs consistently have the edge in voltage gain, low-noise, low interaction, and interface applications.

FETs are inherently transconductance devices, meaning that an input voltage controls an output current. Unlike conventional transistors, FETs have extremely high input impedance (about 10 megΩ — similar to vacuum tubes). In other words, the FET “senses” the audio signal without drawing current from the source. This eliminates complex interactions with the source, allows maximum performance from each system element, and greatly reduces the chance of cable characteristics altering the sound. The absence of input current in FETs allows high bias currents for linearity and speed without sacrificing DC parameters.

Noise is kept low by multiple paralleling of input devices, careful selection of circuit impedances, and pre-screening of devices. The Class A complimentary followers used to drive the preamp output are of such speed, linearity, and low output impedance that no feedback correction is required or used. The advantage of this is that the circuit's perfect stability and transient response are preserved into a wide range of difficult and unpredictable loads. Variation in sound, which could occur through interactions with interconnect cables and other system elements are thus avoided.

Until recently, perfect volume controls (attenuators) did not exist. This is because conventional stereo potentiometers have serious channel mis-tracking (20% typically), become noisy with age, wear out, and have poor resolution of level — particularly when operated by a motor as required for remote operation. In an attempt to solve these problems, many manufacturers have been using “switched attenuators”, which are discrete, resistor ladders built on rotary switches. But while these eliminate channel mis-tracking, they introduce new problems, including limited resolution, stepping transients, and cannot be operated remotely. And they still wear out.

They limit resolution because the most contacts available on rotary switches are thirty-one. This requires two or three dB steps in order to get enough range out of the attenuator, which isn't fine enough resolution for most listeners. Switched attenuators also introduce switching transients (a “click” or “pop”) each time they change positions.

This noise comes from two sources. The first is mechanical noise from the switch itself as its ball-detent mechanism moves from detent to detent. But a more troublesome source is the voltage difference caused by the change in the musical wave-form during the time it takes the switch to move from one position to the next. The greater the voltage difference, the louder the transient.

Sanders Sound Systems solves these problems by using an exotic, precision, resistor-ladder built on an integrated circuit that produces one-hundred, one-dB steps. The switching is controlled by semiconductors that switch much faster than any mechanical switch and unlike a mechanical switch, are silent in operation. A micro-processor controls the semiconductor switches and waits to move to the next step in the ladder until the music wave-form crosses the zero point. As a result, there is absolutely no switching noise and the channel tracking is maintained at better than 0.1%.

The requirements of a power supply for flawless audio reproduction are straightforward but important. Our supplies take a very direct approach to high performance.

First, a top quality, shielded, toriod transformer with plenty of reserve current capability is used. The shielding eliminates strong magnetic fields which could induce hum into sensitive circuitry. About 20,000 μF of capacitance with very low ESR and inductance provides good passive filtering. A reference voltage is developed by delivering a constant current to zener diodes. The resulting voltage is heavily filtered and delivered to each stage through independent, Class A followers, which completely decouple the stages. The resulting non-reactive, low-impedance over an extremely wide bandwidth, yields a perfect power source for the individual circuits. The simplicity and absolute stability of our power supplies removes the chance of unpredictable interactions that may occur with the more elaborate, high-feedback circuitry often used by other manufacturers.

Most companies in the upper end of the audio industry use either a sheet metal or formed aluminum chassis. By contrast, the FET Phono Preamplifier's chassis is all anodized, machined aluminum. The advantage of this over a conventional chassis is that a machined chassis is more precise, which makes it possible to work at tighter tolerances and use PC mounted parts more easily. Moreover, such a design allows easier servicing either for repair or for future upgrading. From the standpoint of appearance, a machined surface can be contoured in a much more precise manner and gives the final product a more seamless appearance.

PARTS QUALITY

FINISHES — All exterior and interior metal parts are anodized. While paint may be more impact-resistant, the anodized surface is more resistant to solvents and prevents corrosion. And our anodized parts are enhanced by either graining or bead-blasting the surface.

CIRCUIT BOARDS — Circuit boards are fiberglass epoxy with gold plating over a tin/nickel barrier. This gold layer will not corrode, while the barrier plate prevents the gold from migrating to the lower copper layer and detracting from its appearance.

RESISTORS — All are high reliability, flame-proof, metal film, 1% tolerance.

CAPACITORS — All capacitors have been eliminated where possible on the basis that "no cap is better than the best cap." Where they must be used, they are high quality, film caps. The only electrolytics used are in the power supply where large numbers provide enormous filtering capacitance for the supply.

SEMICONDUCTORS — Very high quality dual FETs are used for impedance matching and were selected for their superb noise performance and precision matching. They are operated as pure Class A divices. The remaining semiconductors are also of very high quality, each possessing parameters ideally suited for the specific application.

CONNECTORS — RCA connectors are CNC machined brass, gold plated, and insulated with Teflon®. The balanced XLR connectors are Neutriks from Switzerland. These are black chrome finished with gold contacts.

WIRE — All signal wire has been eliminated wherever possible. Where wire is used, it is silver plated copper, 141 strand, 18 gauge with silicone insulation.

SPECIFICATIONS

CIRCUIT SPECIFICATIONS

INPUTS:	Six inputs at line level, including tape and video processor loops. One balanced using XLR connector.
OUTPUTS:	Four outputs at line level. Two fixed level outputs (monitor and processor loops) and two controlled by electronic attenuator, (one balanced and one single-ended). All outputs may be used simultaneously.
FREQUENCY RESPONSE:	-3 dB at 5 Hz and 200 kHz
NOISE:	Greater than 100 dB below 1 volt reference
DISTORTION:	Less than 0.01% from 10 Hz to 40 KHz @ 5 volts peak into 600Ω or higher, shunted by 1000 pF or less.
OUTPUT IMPEDANCE:	50Ω, non-reactive, balanced or single-ended
INPUT IMPEDANCE:	47kΩ balanced or single-ended
GAIN:	20 dB
MAXIMUM OUTPUT:	10 volts peak
CROSSTALK:	Better than 70 dB @ 20 KHz

POWER SUPPLY

	Independently regulated with shielded toroidal transformer and 20,000 µF of capacitance
POWER CONSUMPTION:	10 watts

DIMENSIONS

HEIGHT:	1¾"	(4.5 cm)
WIDTH:	17"	(44 cm)
DEPTH:	9¾"	(25 cm)
WEIGHT:	14 pounds shipping	