

SA-3050A SPECTRUM ANALYZER

**Real Time, Third Octave Analyzer
with Memories and SPL Display**

Audio Control Industrial

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1. Introduction

The Audio Control Industrial SA-3050A is an affordable, measurement-grade, one-third octave real-time analyzer designed for audio signal analysis. Previous analyzer designs were either too costly for the average consumer or too inaccurate for serious use. The SA-3050A overcomes these inadequacies by combining a state-of-the-art microprocessor based design with modern electronic manufacturing techniques.

The SA-3050A offers the following features:

- 30 one-third octave bandwidth filters.
- Fourth-order filters conform to ANSI S1.11-1971 Class II, type E standards.
- Internal pink noise source.
- 9 X 30 large-format LED display with 1 dB Resolution.
- Six internal memories with battery backup.
- Frequency response averaging for up to six stored response curves.
- Peak-hold function.
- SPL scale with bargraph display or full-screen digital readout.
- 92 dB display range.
- Signal input from balanced microphone input, balanced phone jack or unbalanced BNC connector.
- Battery operation (optional).
- Printer port (optional).

1.1 Test Instruments vs Toys

There is a fundamental difference between many real-time analyzers currently available on the market today. That difference is one of calibration. An instrument that is calibrated is capable of making *measurements* based on some absolute reference (absolute measurement). An instrument that isn't calibrated can only make comparisons against a relative reference (relative measurement).

For example, someone asks you, "How hot is it today?" If you step outside for a moment and reply, "Oh, a bit hotter than yesterday," that's a relative measurement. On the other hand, if you step outside and consult a certified laboratory thermometer and say "85 degrees F," you've made a measurement against an absolute reference.

The SA-3050A was calibrated at the Audio Control factory to meet or exceed its published specifications. You can make a measurement with the SA-3050A and compare it with a measurement made by any other measurement-grade instrument and know that you are comparing apples with apples.

1.2 ANSI Filter Design

A real-time analyzer works by dividing the audio spectrum up into equal bandwidth parts using a set of calibrated (usually) bandpass filters. Then the output of each filter is displayed on some sort of level indicator: VU meter, LED column, Cathode Ray Tube (CRT), etc.

The American National Standards Institute (ANSI) oversees the establishment and maintenance of various engineering standards in conjunction with the International Standards Organization (ISO) in Europe. These two standards organizations have established standards for the performance of the bandpass filters used for acoustical measurement purposes.

For 1/3 octave analyzers, there are two classes of filters allowed by ANSI S1.11-1971. These are Class II, and Class III. Class I is reserved for octave bandwidth analyzers. Class III filters are the hardest to design and manufacture and are only found on the most expensive (read unaffordable) analyzers. Class II filters, are easier (not easy, just easier) to design and

manufacture than Class III filters. Any real-time analyzer intended for serious measurement work will use filters that at least meet Class II standards. The filters used in the SA-3050A analyzer meet or exceed the standards specified by ANSI S1.11-1971 for a Class II, Type E filter set.

Figures 1.1 and 1.2 show two different one-third octave filters. Figure 1.1 is representative of those found in most lower-cost analyzers. Figure 1.2 is an actual filter used in the SA-3050A. Note the difference in the response away from the center frequency. The inferior filter's broad response characteristic translates to potential measurement errors at frequencies removed from the band-center and undesirable interaction between adjacent frequency bands.

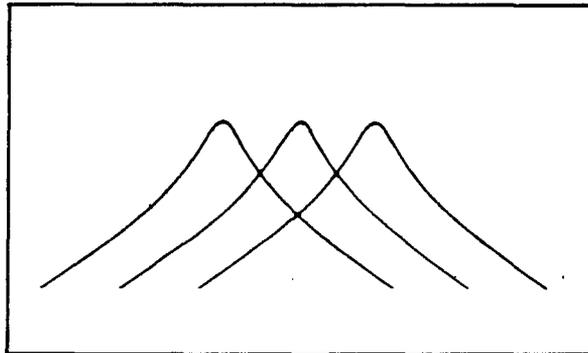


Figure 1.1. A non-measurement quality one-third octave filter.

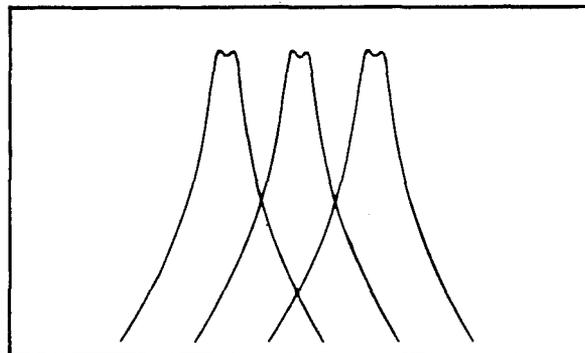


Figure 1.2. A measurement quality one-third octave filter.

1.3 Applications

The SA-3050A is well suited to many audio applications including but not limited to:

- Acoustical analysis for sound reinforcement, recording, or industrial noise measurement.
- Fast frequency response measurements on amplifiers, tape recorders, equalizers, etc.
- Tape recorder setup.

1.4 About This Manual

This manual describes the SA-3050A and a few of its audio applications. The manual is divided into several major sections, this section being the first.

Section 1 Introduction

An introduction to the SA-3050A and to this manual.

Section 2 Specifications

This section lists the SA-3050A specifications and describes the accessories available for it.

Section 3 SA-3050A Front and Rear Panel Features

Describes the physical features found on the front and rear panels of the SA-3050A.

Section 4 Acoustical Testing with the SA-3050A

This section has procedures for using the SA-3050A for Sound System equalization and analysis.

Section 5 SA-3050A Applications

Besides equalizing sound systems, the SA-3050A has a few other tricks. This section describes a few of them.

Section 6 Printer Operation

You can connect a printer to your SA-3050A. This section tells how.

Section 7 Battery Operation

The SA-3050A can operate from batteries. This section tells you all about it.

Section 8 Theory of Operation

For terminally curious technoids only, a description of what goes on inside the confines of the box.

Section 9 Service Information

This section describes the Audio Control warranty and tells how to obtain service or parts for the SA-3050A. We trust that you'll never need to use this section.

1.4.1 Notational Conventions

Within this manual, several different notation conventions are used to indicate various facets of the SA-3050A's features.

SMALL CAPS indicate a marked feature on the unit, like the **INPUT SENSITIVITY** control or the **PINK NOISE OUTPUT** connector. They are also used within procedures to identify controls and switches by function.

italics and **boldface**

are used for emphasis. Words printed in **boldface** convey more emphasis than those printed in *italics*.

1.4.2 Notes, Cautions, and Warnings

Some of the text in this manual is set apart by the headings Note, Caution, or Warning. These terms are used to denote varying degrees of awareness required by the user during installation, operation, or maintenance of the SA-3050A.

A **Note** conveys information that may be helpful to the user. A note is similar to an aside during a conversation.

A **Caution** indicates a potential danger to the instrument. An example of a Caution can be found in Section 3.1.

A **Warning** indicates a potential hazard to the operator. An example of a Warning can be found in Section 9.

2. Specifications

2.1 SA-3050A Specifications

Instrument Type	1/3 octave real-time spectrum analyzer
Inputs: Microphone: Connector Impedance Acoustical Signal Level Instrumentation: Connector Impedance Signal Level Balanced Bridging: Connector Impedance Signal Level Input Calibration: Instrumentation Balanced Bridging	Three Balanced low impedance with +12V phantom powering 3 pin XLR female (Neutrik D series) suitable for 150 ohm microphones, actual impedance is 5K ohms 44 dB SPL to 136 dB SPL High-impedance, unbalanced input BNC 1M ohm in parallel with 30 pF -56 to +36 dBu Balanced, high-impedance input 1/4 inch tip-ring-sleeve 10K ohms -56 to +36 dBu 0 dBu = 100 dB SPL 0 dBu = 100 dB SPL
Filters: Frequency Range Band Center Frequencies Center Frequency Accuracy Filter Characteristic	30 1/3 octave bands from 25 to 20 kHz on ISO preferred 1/3 octave center frequencies 25, 31.5, 40, 50, 63, 80, 100, 125, 160, 200, 250, 315, 400, 500, 630, 800, 1000, 1250, 1600, 2000, 2500, 3150, 4000, 5000, 6300, 8000, 10000, 12500, 16000 20000 Hz ± 2% of design center Fourth-order bandpass. Meets or exceeds S1.11-1971 Class II, Type E.
Display: Spectrum Display SPL Display Display Resolution Decay Time	9 X 30 LED matrix 9 LED bargraph 1, 2, 3, or 4 dB per LED Fast, Medium and Slow
Memory: Type Number	cmos with battery backup Six
Physical: Dimensions Shipping Weight	smaller than a breadbox (4.1 x 10 x 12.75, inches hwd) 10 pounds
Electrical: Power Requirements	Line Voltage: 108-125V ac, 60 Hz. Power Consumption: 15W

In the interest of continuing product development, prices and specifications of Audio Control Industrial products are subject to change without prior notice.

2.2 Accessory Information

Table 2.1 lists the accessories available for the SA-3050A. The paragraphs following the table describe them.

Table 2.1. SA-3050A Accessories

Model	Description
RM-10	Heavy steel 19 inch rack mount adaptor for the SA-3050. Requires 3 (5.25 inches) rack spaces. Includes handles and ventilation slots.
BP-10	Rechargeable internal battery system for SA-3050. 4.5 to 5 hours battery life.
PI-10	Parallel printer output adapter for IBM/Epson compatible printers.
SC-10	Soft carrying case. Includes space for analyzer, mike, and cables

IBM is a registered trademark of International Business Machines Corporation.

Epson is a registered trademark of Epson America, Inc.

2.2.1 Rechargeable Batteries

The rechargeable battery option allows the SA-3050A to be used independent of AC mains power. The batteries install within the instrument and provide up to 4.5 to 5 hours of operation on a full charge. The batteries are recharged by plugging the instrument into a source of AC mains power and leaving the front panel POWER switch set to the OFF position for a period of 9 to 10 hours (twice the discharge time).

2.2.2 Printer Output

The printer output option provides a Centronics parallel interface for an Epson compatible printer (not provided). The printer output option consists of an EPROM (Eraseable, Read-Only-Memory) and an interface cable and connector. The analyzer-to-printer cable is not provided. The printed output is a "form" with suitable blank spaces for recording the time and place of the measurement and other data, followed by the a printed representation of the real-time spectrum display on the analyzer.

2.2.3 Carrying Case

The optional carrying case protects the SA-3050A from the ravages of portable operation.

3. Operation

Using the SA-3050A is simple and straightforward. This section describes the physical features of the unit and presents a basic procedure for using the SA-3050A.

3.1 SA-3050A Front and Rear Panel Features

The following paragraphs describe the front and rear panel features of the SA-3050A. Each item is keyed to the drawing found in Figures 3.1 and 3.2.

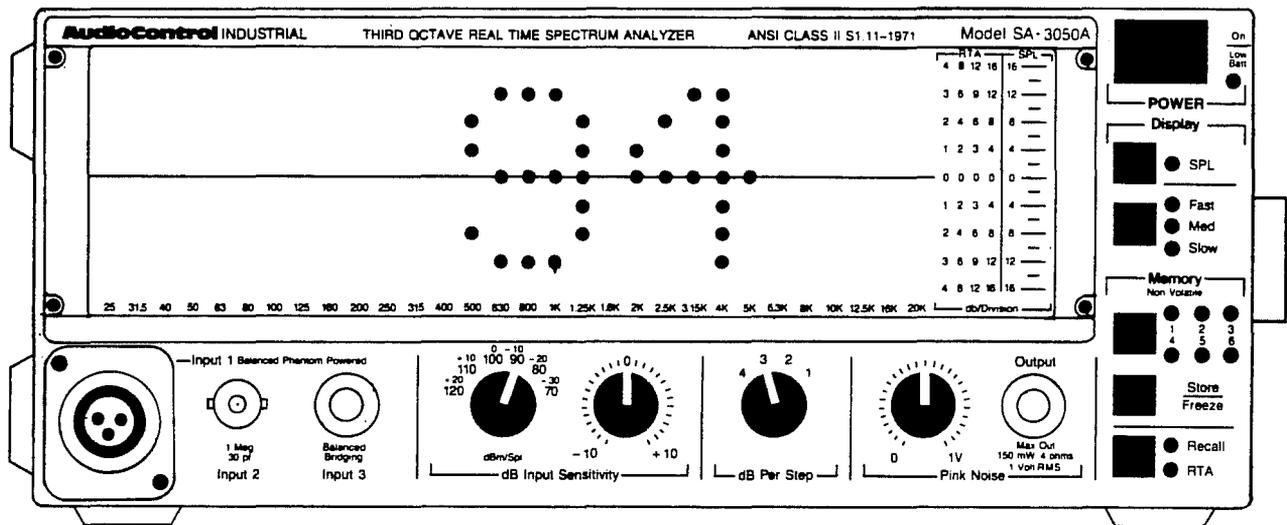


Figure 3.1. SA-3050A Front Panel.

3.1.1 Front Panel

LED DISPLAY The left portion of the display shows the one-third octave energy content of the input signal from 25 Hz to 20 kHz. Each column within the display represents one one-third octave band. The band centers are marked on the bottom of the screen. The right-hand end of the display indicates the setting of the dB PER STEP switch, and consequently the amount of change in the input signal that each led on the left-hand side of the display represents. The far right portion of the display indicates the sound pressure level (SPL), relative to the settings of the dB INPUT SENSITIVITY switch and control.

POWER switch A three-position momentary-contact switch that turns the SA-3050A on or off (one direction is on, the other is off).

LOW BATT led The LOW BATT indicator illuminates during normal operation of the SA-3050A. When operating the SA-3050 on the optional internal rechargeable batteries, a flashing LOW BATT light indicates that the batteries need recharging. When the light extinguishes, the battery supply has discharged to the point that further operation of the SA-3050A is not possible. The SA-3050A has a voltage sense circuit that prevents operation of the instrument below a certain critical voltage. This prevents total battery discharge, which shortens the battery life. To continue operation, the SA-3050A may be operated from the AC power line. The batteries may be recharged by plugging the SA-3050A into a suitable source of AC power, with the front panel POWER switch set to the OFF position for a period of 6 to 10 hours. The minimum recharge time from the voltage-cutout state is around 2 hours, which will then operate the instrument for around one hour. Completely recharging the batteries from a discharged state (instrument dead) requires 9 to 10 hours

(2 times the discharge time).

SPL Momentarily pressing the SPL push-button switch turns the SPL display in the right side of the display window on and off. Pressing and holding the SPL push-button activates the full-screen digital SPL display. In this mode, the display directly indicates the SPL at the microphone. Momentarily pressing the SPL switch again restores normal analyzer operation. The associated LED indicator indicates the status of the SPL switch. The SPL bargraph display scaling is always 4 dB/step, regardless of the setting of the dB PER STEP switch. The SPL switch works regardless of the setting of the RECALL/RTA switch. This means that you can switch the SPL bargraph on and off even with a stored display.

display speed The display speed push-button switch is located underneath the SPL switch and to the left of the FAST, MED, and SLOW LEDs. The lighted LED to the right of the switch, indicates the decay time of the display. Press the button to change the decay time of the display. The FAST setting is optimized for looking at transients, the MED setting is useful for program monitoring, and the SLOW setting works well for measurements using pink noise. The setting of the switch also affects the ballistics of the SPL bargraph.

MEMORY The SA-3050A can store up to six different frequency response curves (including the SPL display) in its internal non-volatile memory. An internal back-up battery maintains the contents of the memory for periods of up to one year, even with the AC power supply disconnected. This feature allows "remembering" a standard curve that may be recalled each time that the SA-3050A is used.

Storing a Response Curve

- Press the STORE/FREEZE button. The display is now frozen and ready to be stored.
- Press the MEMORY button to select a memory for use. Pressing the MEMORY button again steps through the six memories in sequence. The display shows the contents of each memory as it is selected.

The RECALL and RTA LEDs are not illuminated. This indicates that the SA-3050A is ready to store the display.

- Pressing the STORE/FREEZE button stores the contents of the display in the selected memory. The associated LED indicators display which memory is currently active.
- The yellow RECALL LED illuminates to indicate that you are now in recall mode. You can recall any other memory for comparison purposes by selecting that memory by repeatedly pressing the MEMORY button.
- Press the RECALL/RTA button to return to RTA mode.

Recalling a Response Curve

- Press the RECALL/RTA button until the RECALL LED is illuminated.
- Press the MEMORY pushbutton to successively view the contents of the six memories.
- Pressing the RECALL/RTA button again returns the SA-3050A to the RTA mode.

Averaging Several Readings

The averaging mode allows up to six previously stored readings to be averaged together. The averaged result is stored in memory number six, replacing its contents.

- Acquire and store up to six response curves.
- Press and hold the STORE/FREEZE button for two seconds. The yellow LED representing memory six is now flashing, indicating the initiation of the averaging mode.
- Select the first memory for averaging using the MEMORY button. Touch the STORE/FREEZE button once.
- Select the next memory for averaging using the MEMORY button. Touch the STORE/FREEZE button once. Repeat this step for each additional memory to be averaged, up to six.
- After touching the STORE/FREEZE button for the last memory location to be averaged, press and hold the STORE/FREEZE button for two seconds. The average is displayed on the screen and is written into memory location six.
- You can weight a particular curve by entering it into the average more than once. (When a memory location is used for averaging, its contents are *not* lost.)
- The contents of memory location six are lost when the average is calculated. The result of the averaging operation is overwritten into memory location six.

Peak-Hold

The Peak Hold mode allows you to accumulate the highest peak level in any of the SA-3050A's thirty display bands. This display may be stored in any memory location, averaged, etc.

- Press and hold the RTA/Recall button for two seconds. The red RTA LED is now flashing, indicating peak hold mode.
- The display now indicates the overall peak level for each of the thirty bands.
- Operating the dB/STEP switch clears the display and restarts peak-hold mode.

Clearing the Internal Memories

- Press and hold the STORE/FREEZE button while turning on the SA-3050A.

PINK NOISE

The internal pink noise generator is an accurate, digital, laboratory-grade test source. The pink noise output is accurate throughout the measurement range to within ± 0.25 dB. The maximum output provided at the 1/4" tip-ring-sleeve phone jack is 1 volt into a 600 ohm load (unbalanced, ring and sleeve grounded), or 150 mW into a 4 ohm load. The signal level at this connector is controlled by the knob to its immediate left. The pink noise generator has sufficient output to drive virtually any speaker directly or passive crossover.

dB PER STEP This switch sets the resolution of the spectrum analyzer portion of the display. The setting represents the value of each LED in the display. Thus a setting of 1 dB per step causes each LED in the display to represent a 1 dB change. At this setting, the overall range of the display is 9 dB. At the 4 dB per step setting, the LEDs in the display represent a change of 4 dB, with a 32 dB overall display range.

When using the SA-3050A for sound system equalization, start with the 4 dB/STEP setting and progressively advance the setting of the dB PER STEP switch as the equalization process progresses. For program monitoring, the 4 dB position works well because it displays the widest dynamic range.

Stored response curves are always stored at 1 dB per step resolution, regardless of the setting of the front panel switch. You can select whatever resolution you wish for a stored display and know that it is accurate. The same holds true for displays dumped to the printer output.

dB INPUT SENSITIVITY

The dB INPUT SENSITIVITY control and switch selects the reference level of the curve shown in the display window. The control to the right of the input selector switch is a fine adjustment and alters the range selected by the switch over a range of ± 10 dB. The normal setting for this control is the detented (click-stopped) setting. The reference level of the display is shown on the display window by the light grey horizontal line. The actual reference level of the reference line corresponds to the dB level setting of the dBm/SPL switch.

For example, the dBm/SPL switch (coarse adjustment) is set to 90 dB. The light grey line on the display window represents a sound pressure level of 90 dB (if the INPUT control (fine adjustment) is set at the detented setting).

INPUT 3 A 1/4 inch tip-ring-sleeve phone jack used for connecting to balanced and unbalanced sources. Connect unbalanced sources by using a tip-sleeve (2 conductor) plug inserted into this jack.

This input has an input impedance of 10 Kohms and is suitable for signal levels from -56 dBu to +36 dBu. An input signal of 0 dBu represents 100 dB SPL.

INPUT 2 This is an instrumentation style (BNC) connector with an input impedance of 1 Mohm in parallel with 30 pF. This input is suitable for signal levels from -56 dBu to +36 dBu. An input signal of 0 dBu represents 100 dB SPL.

The particular combination of resistance and capacitance allows connecting a standard oscilloscope probe to this input.

INPUT 1 This is a standard +12 volt, phantom-powered; balanced microphone input. The phantom power supply is intended for the Audio Control CM-10 measurement microphone or an AKG C460B/CK62-ULS condenser microphone. Ordinary dynamic microphones may also be connected directly to this input providing that they have a balanced output. Unbalanced microphones may not be used. This input is suitable for acoustical levels at the microphone ranging from 36 dB SPL to 136 dB SPL. The SPL display is only accurate if the Audio Control CM-10 microphone is used, or if the external microphone matches the sensitivity of the CM-10 microphone.

CAUTION

A microphone with an unbalanced output may be damaged if connected to this input. Audio Control Industrial assumes no responsibility for microphones damaged in this manner. Connect microphones with 1/4 inch phone plug connectors to the BALANCED BRIDGING input connector.

3.1.2 Rear Panel

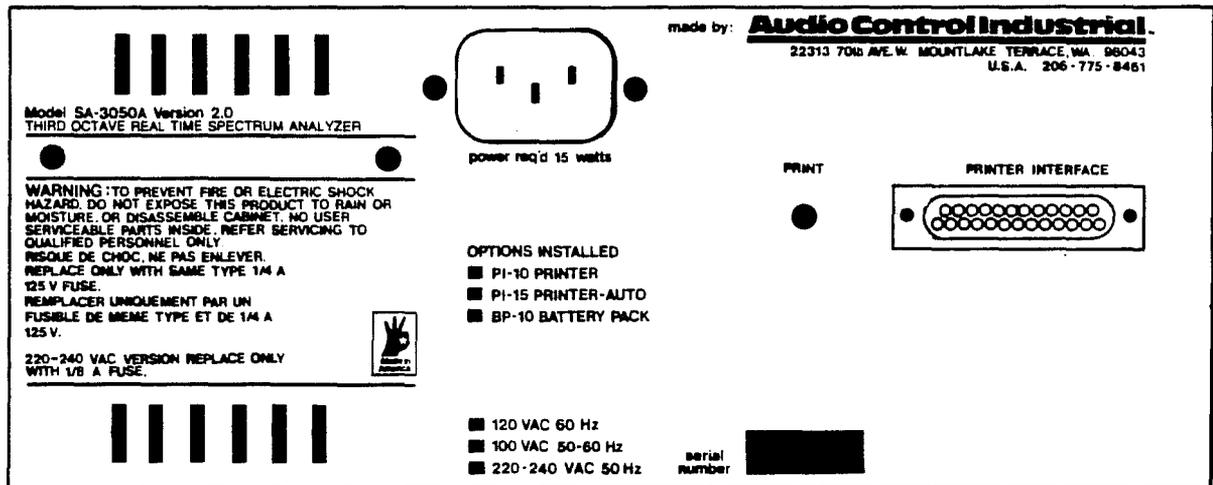


Figure 3.2. SA-3050A Rear Panel.

AC power input receptacle

An IEC 3-prong receptacle for AC power input to the instrument. The SA-3050A can operate using one of the following power sources:

1. 120 V ac, 60 Hz
2. 220-240 V ac, 50 Hz
3. 100 V ac, 50-60 Hz

The SA-3050A is factory supplied to operate at one of the above listed ac power supply configurations. The supplied configuration is marked on the rear panel of the unit. Consult the Audio Control factory for power supply conversion instructions.

PRINTER INTERFACE (optional)

This is a DB-25F connector used for connection to an external (not supplied) printer. The connector is wired for a Centronics compatible, parallel interface printer. This connector is present only if the SA-3050A is equipped with the printer interface option. A description of the connector pin connections can be found in Section 6.3.

PRINT (optional) A momentary-contact switch used to print the contents of the display on the external printer. This switch is present only if the SA-3050A is equipped with the printer interface option.

3.2 Operating Instructions

The following paragraphs describe the fundamentals of using the SA-3050A. Additional operating information can be found earlier in this section and in Section 4.

3.2.1 Operating Procedure

Here's how to use the SA-3050A.

1. Connect the SA-3050A to a suitable source of AC power. If your instrument is equipped with the battery option, the batteries must be charged before use. The correct AC power configuration is marked on the rear panel of the instrument.
2. Connect the input signal source to the appropriate input on the front panel of the instrument. Refer to Section 3.1 if you aren't sure.
3. If you are using the internal pink noise source, connect the PINK NOISE output of the analyzer to the amplifier or sound system line input. The PINK NOISE output has sufficient power capacity to directly drive a loudspeaker to modest levels.
4. Set the dB INPUT SENSITIVITY switch so that signal peaks reach the upper 1/3 of the display. Adjust the dB INPUT SENSITIVITY control as needed. If the dB INPUT SENSITIVITY control is centered, then the grey line dividing the display represents the SPL setting of the dB INPUT SENSITIVITY switch.
5. Select the display resolution using the dB PER STEP switch. The switch setting represents the value of each LED in the display. A scale is printed on the right-hand side of the display window.
6. Select the decay time of the display by pressing the pushbutton switch adjacent to the FAST MED or SLOW LEDs. The FAST setting is suitable for viewing transients. The MED setting is suited to signal monitoring and the SLOW setting works well for measurements.
7. You can determine the sound pressure level at the microphone by pressing the SPL pushbutton until the SPL LED illuminates. The SPL bargraph is always 4 dB/step, regardless of the setting of the dB PER STEP switch.

3.2.2 Storing a reading

The SA-3050A has six memories that can store the entire display (including the SPL bargraph). Displays are stored at maximum resolution (1 dB/step), regardless of the setting of the dB PER STEP switch. This procedure assumes that the current display reading is about to be stored.

1. Press the STORE/FREEZE pushbutton. Both the RECALL and the RTA LEDs should be extinguished.
2. Press the MEMORY pushbutton repeatedly to select a storage location. The display shows the contents of the currently selected memory. Pressing any other pushbutton switch than this one cancels the storage operation.
3. Press the STORE/FREEZE pushbutton again to save the display contents prior to step 1 of this procedure.
4. The instrument is now in recall mode. You may recall any other memory by pressing the MEMORY pushbutton to select another memory location for display.
5. Return to the RTA mode by pressing the RECALL/RTA pushbutton.

3.2.3 Recalling from memory

Here's how to recall a stored display from memory.

1. Press the RECALL/RTA pushbutton switch until the RECALL LED is illuminated.
2. Press the MEMORY pushbutton switch repeatedly to select the desired memory location. The display shows the contents of each memory location as the MEMORY switch steps through each of the memories.
3. Return to RTA mode by pressing the RECALL/RTA pushbutton switch until the RTA LED is illuminated.

3.2.4 Freezing the display

You can freeze the display momentarily, without storing it in one of the memories. Press the STORE/FREEZE pushbutton. The display will remain frozen for 15 seconds, or until any pushbutton switch except the MEMORY switch is pressed.

3.2.5 Averaging the Contents of the Memory

1. Acquire and store up to six different response curves.
2. Press and hold the STORE/FREEZE button until the yellow LED for memory location six begins flashing. Release the STORE/FREEZE
3. Select the first memory location to be averaged.
4. Press the STORE/FREEZE button momentarily.
5. Select the next memory location to be averaged. Press the STORE/FREEZE button momentarily. Repeat this step until all desired memory locations have been accumulated.

You can weight the average in favor of one curve by entering it into the average more than once. Just select it again and momentarily press the STORE/FREEZE button again.

6. Press and hold the STORE/FREEZE button for at least 2 seconds. The computed average is shown on the display and stored in memory location six.

3.2.6 Digital SPL Display

The SA-3050A can display the SPL at the CM-10 microphone digitally on the display screen. The instrument samples the SPL 20 times per second and displays the median of the samples taken. The display is updated about every 1.2 seconds. In digital SPL mode, all of the memory buttons are disabled, however the response time switches are still functional.

3.2.7 Peak-Hold Display

Peak-hold mode allows displaying non-recurring events, or accumulating a spectral history of the input signal. In this mode, the SA-3050A stores and holds the highest peak in any of the thirty bands shown on the display. If the input signal exceeds this peak level, then the SA-3050A updates that band on the display. The peak-hold display can be stored and recalled like any other display.

1. Press and hold the RECALL/RTA button until the RTA LED starts flashing.
2. The SA-3050A is now in peak-hold mode.

3.2.8 Peak-Hold Digital SPL

The digital SPL display may also be operated in peak-hold mode.

1. Press and hold the SPL button until the instrument enters digital SPL mode.
2. Press and hold the RECALL/RTA button until the SPL LED starts flashing.
3. The SA-3050A is now in digital SPL peak-hold mode.

4. Acoustical Testing with the SA-3050A

Perhaps the most important use of the SA-3050A is as an acoustical measuring tool: e.g. equalizing sound systems. This section of the manual discusses the process and rationale behind using a real-time-analyzer and an equalizer as an integral part of any sound system.

4.1 Overview (or what we're trying to accomplish)

The primary reason for using the analyzer with an equalizer is to do what our ears can't do reliably: adjust a sound system to a known reference condition. The reason that we can't do this "by ear" is our rather poor long term auditory memory. In other words, our hearing process is *very* adaptive. Our hearing adjusts to compensate for varying conditions, sound pressure levels, and tonal balances. When presented with a change, our sense of hearing can discern that change very readily. This is our short-term auditory memory. On the other hand, our ears are not very good at remembering things over a long period of time, especially when there are unrelated sounds between the periods of recall.

Another problem is threshold shift caused by exposure to loud sounds. Without getting into a lot of detail (don't worry, we will later), the threshold of hearing (the SPL where you can sense the presence of a sound) shifts upward when the ear is exposed to loud sound. The recovery time can be measured in hours to days, depending on the intensity and duration of the exposure. For this discussion, it's just another problem, or source of error.

For these reasons, and many more, a real-time-analyzer with a calibrated microphone is one of the most popular methods of measuring the frequency response of a sound system within an acoustical space.

In a nutshell, a pink noise source excites the space, a microphone picks up the acoustical signal and converts it to an electrical signal. The real-time-analyzer (RTA) breaks the signal up into equal octave-percentage bands, and displays the signal level of each band individually. The display could be VU meters, a CRT (video monitor or oscilloscope), or a LED bargraph. The SA-3050A uses the latter method.

In the real world, very few things are perfect, and expensive sound systems are no exception. Traditionally loudspeakers are measured in an anechoic chamber (*an* - not having, *echoic* - echoes). An anechoic chamber is a large room, with the floor, walls, and ceiling lined with acoustically absorbent material. Any sound emitted within the chamber is absorbed.

Being inside an anechoic chamber is a strange experience. First, you are standing on a suspended floor made of steel cables. If you are afraid of heights, this is just a good beginning. Since the walls absorb all sound, it is eerily quiet inside. You can actually hear your blood coursing through your blood vessels.

At any rate, an anechoic chamber is hardly the same as an average living room. However perfectly the loudspeaker measured within the chamber, putting it into an average living room changes the whole ball game. First, the floor, walls, and ceiling are anything but totally absorbent. This allows the room to have resonances, standing waves, and a sound of its own. The room *interacts* with our perfect loudspeaker to produce an imperfect system.

Before we go *any* further, you should note that equalization is a tool. Correctly applied it can do wonders. Some situations are beyond help. No amount of equalization will overcome the problems caused by reflections, standing waves, and excessive resonance. There is only one cure for an acoustical disaster: explosives.

There is more to equalizing a sound system than simply adjusting the equalizer until the analyzer display is a straight line. It's important to know which defects you can improve on, and which ones you're wasting your time on. If this weren't so, we could train monkeys to do this, and a fair number of system installers would be looking for work.

4.2 Basic Procedure

Equalizing any sound system is basically a five-part process:

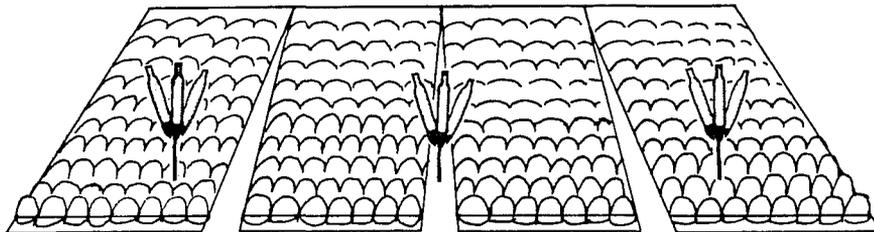
1. Listen
2. Measure
3. Balance speaker components and equalize
4. Listen
5. Trim equalization settings

The following procedure describes the basics of equalizing a sound system installed in a real-world acoustical space.

NOTE

If the sound system is biamplified, or if the speakers have level controls (like midrange and/or tweeter level controls), the first thing that you should do is to set these controls for the best overall response curve.

1. Connect a pink noise source to the sound system. This could be the output of the SA-3050As pink noise generator, a tape or a CD with prerecorded pink noise on it.
2. Place the microphone at the listening position. It helps to position the microphone away from any reflecting surfaces. Orient the microphone as shown in Figure 4.1.
3. Turn on the sound system. Increase the pink-noise level until the SPL is at least 10 dB over the ambient noise level. If the response curve looks terrible (huge peaks and/or valleys), try moving the microphone slightly. If the peaks or valleys go away, the problem was probably the microphone position.
4. Store the resulting response curve in one of the SA-3050As memories.
5. Repeat this process (position mike, store response) several times.
6. Recall the stored curves and derive a curve that is an average of all of them. You can derive the average by recalling each stored curve and visually estimating a composite curve that represents the average between the highest and lowest readings for each band. The rationale behind averaging is to help separate out response flaws that are caused by the microphone position versus those flaws that are caused by the acoustical environment and its interaction with the sound system.
7. Now adjust the equalizer to make the *average* response curve flat. (level out the peaks, and smooth the overall curve)
8. Recheck your work by checking the response through the microphone positions that you used to derive the average curve. Repeat the equalization process if necessary. Your overall goal in this process is **smooth** response, not necessarily **flat** response.
9. Insert any high-frequency rolloff or bass boost needed for contouring or system voicing.



Use multiple locations

Figure 4.1. Positioning the microphone.

Now that you have an idea of the basic procedure, here are a few hints:

- If the response curve really looks horrible, experiment with the microphone position. You may have just picked out a position that happened to be at a room node, or inside of a standing wave. A few inches should tell you.
- The first thing that you should attack with the equalizer is the highest peak showing on the display. When you've pulled this down to the level of the surrounding terrain, find the next highest peak and pull it down. Repeat this process as necessary. Adjust the output gain control of the equalizer to make up for equalizer's loss.

If the peak straddles two equalizer controls, then split the difference between them. Unless the system and/or space are really poor, you won't need much equalization to smooth things out.

- Don't try to equalize out dips in the overall response curve, unless they are quite shallow (less than 3 dB). Anything more than this is a decided **waste** of power (3 dB = 2X power). Fortunately for us, our hearing is more sensitive to presence rather than absence. Thus, we'll hear the peaks in the response long before we notice the absence caused by the dips.
- Once you've gotten the overall curve flattened out, listen to the system with music and speech. You'll probably want to put in some controlled high-frequency rolloff...say 1 to 3 dB per octave starting somewhere between 1 and 8 kHz.
- If you're equalizing an autosound system, you'll probably need to introduce some carefully contoured bass boost to overcome the low-frequency ambient noise level in the car, and to help compensate for the Fletcher-Munson effect.
- If the system is installed in a car, drive around at varying speeds to assess the effect of engine and road noise on the performance of the sound system. You may need to trim the overall equalization to suit.
- Listen to the system for an extended period of time, at varying volume levels. If the overall system equalization curve isn't right, listening fatigue will let you know that you haven't finished yet. It may take several tries to get it *just right*.

4.3 Body Effects

When measuring, it's important to keep any reflecting surfaces away from the microphone. This includes your body if you are hand-holding the microphone. If you aren't careful, the reflection can cause response errors because of the multiple paths into the microphone. Ideally, the microphone should be suspended in free space. In reality three feet should be adequate spacing.

In an automobile, the three foot rule may turn into science fiction. On the other hand, that's the environment that the system is going to be listened to in. In this case, the rule goes out the window. You may notice more variation in the response as you move nearer to the windows. That is just another good reason for using a multiplicity of measurement points, and taking an average of all of them.

4.4 Sound Pressure Level and Hearing Loss

Although it's great to have a sound system that will achieve levels of 130+ dB SPL, it's also great to be able to hear it for a while. It's a proven fact that one of the causes of hearing loss is prolonged exposure to excessive sound levels.

Research has shown that prolonged exposure to average levels under 90 dB SPL will not cause harm to our hearing. As the level exceeds 90 dB SPL, the capacity for damage increases, and the damage becomes more and more permanent. The amount of exposure that our ears can withstand without damage is related to intensity and time. We can tolerate 105 dB SPL for less time than we can 95 dB SPL.

The reason is the way that our ears respond to stimulus outside of the normal range of sounds in our environment. For example, if you enter a building where there is loud music playing, it may seem loud to you at first, but after 20 minutes or so, the music will seem less loud. That is because your hearing has acclimated itself to the new ambient environment. Unless you're conscious of this, you may not notice the change until you leave the building and go outside. Suddenly everything seems very quiet. You have just experienced what is known as *threshold*

shift. After a while, your threshold will return to a level that is close (but not quite) to your original threshold. This is one mechanism of permanent hearing loss.

WARNING

Audio Control Industrial assumes no liability or responsibility for hearing loss incurred directly or indirectly by the use of the SA-3050A.

5. SA-3050A Applications

The SA-3050A has many applications. Some of these include:

- Sound System Equalization
- Monitor System Feedback Control
- Home Stereo Equalization and Setup
- Tape Machine Alignment
- Sound and Music Monitoring
- Crossover Testing

5.1 Sound System Equalization

Sound reinforcement systems, large or small benefit even more from carefully applied equalization. While only dynamite can really cure a really bad case of poor acoustics, equalization comes right after loudspeaker array design when it comes to getting the most out of any speaker system in any room. If you set up in different venues night after night, the SA-3050A can be the impartial judge that helps you make your sound consistent, night after night.

The SA-3050A can work wonders on a home stereo. Most home stereos can benefit from a bit of carefully applied equalization. Such equalization can go a long ways towards enhancing your listening pleasure. The SA-3050A can also be used to verify the performance of various portions of the electronics in your system.

Whether you're equalizing a home stereo, or a 100,000 watt sound system, the basic technique is the same:

1. Listen
2. Measure
3. Balance speaker components and equalize
4. Listen
5. Trim equalization settings

5.1.1 Hi-Fi Equalization

Before attempting any corrective equalization, use the SA-3050A to make a preliminary assessment of the situation. Use several microphone positions, finally drawing a curve that represents the average of a representative sample of the positions. We recommend no less than 3 different positions in a one to three foot cube around your listening position.

Next, adjust the various level controls on the speakers themselves to smooth out the mid- and high-frequency response. You can combat peaky bass response by trying various elevations above the floor. Conversely, you can aid anemic bass response by using floor or corner placement. The name of the game here is to get things as "right" as possible *without* equalization.

Now its time to use the equalizer. Remember what we discussed in Section 4. It's important to remember to **not** try to equalize the dips in the overall response curves unless they are quite shallow. Deeper dips may be due to a door, window, or other surface acting as a diaphragmatic absorber (the acoustical equivalent of a black hole). If that is the case, then structural modification of the offending surface is the only real cure. If you get really desperate, there's always explosives.

If you're using an octave-band equalizer, you've probably noticed that the analyzer has a few more bands than the equalizer does. With octave-band equalizers and octave-band analyzers, the important thing that is missing is the response characteristic *in between* adjacent bands. The equalizer's controls interact with each other, allowing you to affect the response in the sidebands. If the controls didn't, the equalizer wouldn't be very useful, let alone musical. The SA-3050A shows you what is going on in-between the bands of the equalizer and the effects of the controls away from their band-centers. Because of the interaction, you can still deal with

problems that aren't conveniently centered on a particular control by splitting the difference between two adjacent controls.

Regardless of which equalizer you're using, a overall guiding principle is: "**Less is Better.**" With good components to start with, all that should be needed is gentle, gradual shaping. Avoid large amounts of boost or cut as well as sharp discontinuities in the overall curve.

Once you're finished, you'll want to add a controlled amount of high-frequency rolloff. Why? Because "ruler-flat" speakers sound unnaturally bright to most ears. Ordinarily we hear most sounds at a distance. This causes high-frequency loss because of the friction between the sound wave and the air, as well as other losses due to absorption caused by walls and other surfaces. As a starting point, try 1 to 3 dB per octave, starting somewhere between 1 and 8 kHz. Experiment; let your ears be your guide and the SA-3050A be your compass.

5.1.2 Sound System Equalization

Equalizing a large sound system isn't much different from equalizing a stereo system. Granted, the scale of things is quite different, but the process is more or less the same.

One thing that you need to keep in mind when equalizing a system in a large space is that the equalized response curve represents the average response taken at many points within the room. This alone can make the equalization process quite tricky. The aiming of the components of the array, their mechanical alignment, and their directional characteristics all contribute. Its easy to make the system curve look right at one point, but how about the other seats in the room?

One source of problems is picking a microphone position that is too close to the loudspeakers. If the loudspeaker has flat power response throughout its coverage pattern, then you are safe. If not then what happens is that you end up equalizing the direct field response, which is fine at that location but since the power response isn't flat, you get a lumpy response curve at other locations. On the other hand, if you equalize outside of the direct field (in the reverberant field) and successfully flatten things out then the direct field will be lumpy. The solution is constant coverage loudspeakers that maintain their directional characteristics over a wide frequency range.

5.1.3 Monitor Speaker Equalization

Monitors in this context are the monitors used in a recording studio for monitoring and evaluating the program being recorded. Equalization in this context is especially critical because any response aberrations in the control room monitor speakers has an inverse relationship on the material being monitored once the material is removed from the room and played on other equipment.

Consider: Your playback system has a midrange bump in it around 3kHz, and a fairly flat high-end. You're mixing some material down using these speakers as a reference. When the producer takes the material out of the studio and listens to it that night at home, he notes that the vocals didn't seem quite as present as they were at the studio and that the drums seem a bit lifeless.

What has happened is that the 3k bump caused you pull the vocals down in the mix (since most vocal energy is around 3k) and the flat high-end resulted in not having enough treble energy content in the final mix. This occurs partially from a level judgement error (too much top end in the monitors, not enough on tape) and perhaps from an equalization error (it sounded fine in the monitors, why EQ it?).

If the speakers have a bass bump, then your end-product could be bass shy. As you can see, the speakers really do have an inverse effect on your end-product.

Careful equalization of your monitor system using the SA-3050A and a good equalizer can give your studio the ability to make tapes that sound good **anywhere**. The process and pitfalls are very similar to equalizing a home hi-fi system. Additionally you should be aware of the following:

- Your microphone positions should be approximately where the engineer sits, at approximately head level. It's not a bad idea to use several positions at approximately ear level, but along the length of the console. Make sure that the response at the producer's chair is very similar to that at the engineer's chair.
- Make the overall response curve **smooth** rather than **flat**.
- Add some controlled rolloff at the high frequencies to compensate for the rolled-off high-frequency response of most home systems.
- Pay attention to the response in the midrange of the audio spectrum, between 300 Hz and 5KHz. Make sure that it is smooth and reasonably flat.
- Do flatten peaks.
- Do remove shallow, wide bandwidth holes.
- Do not remove large holes of any bandwidth.
- Beware of adding large amounts of boost at low frequencies. Unless you have infinite amplifier power and speakers capable of moving infinitely far, this is an invitation to amplifier clipping or speaker destruction.
- Remember that 3 dB of boost requires twice the amplifier power at that frequency.

5.1.4 Foldback Monitor Equalization

Foldback monitor speakers are a difficult proposition at best. The SA-3050A can help by telling you exactly what frequency is ringing when the monitors feed back. Even if you only have an octave band equalizer, the one-third octave display of the SA-3050A tells you if the offending frequency is right on a band center, or if it is in-between.

There are two schools of thought presently. We'll simply present them both. School A probably evolved from those folks that didn't have a real-time analyzer. School A is popularly known as feedback tuning. School B involves using pink noise to flatten out the frequency response, followed by feedback tuning.

Regardless of which school you use, remember that when a person steps in front of the microphone, their face acts as a reflector and reflects energy from the monitor speaker back into the microphone. This will change your tuning on the monitor equalizer.

5.1.4.1 Feedback tuning

What could be simpler? Make the sound system feed back, find the feedback frequency on the equalizer and notch it out slightly. Repeat as necessary. One-third octave equalizers make this a tricky proposition to say the least.

Before the SA-3050A, the process of finding the feedback frequency on the equalizer was the hard part. Some of the techniques used are: guess-and-by-gosh, frequency counter, zero-beat with an oscillator, relate the feedback pitch to the nearest musical pitch then convert musical pitch to numerical frequency (easier than it sounds).

With the SA-3050A connected across the mixer output (you can use the microphone instead), bring the system slowly into feedback. Watch the display. When the system starts feeding back, reduce the gain. The offending frequency is the last one to decay off of the screen. This even works during performance. Now find the frequency on your equalizer and dial in some cut at that frequency. Once more, bring the system slowly into feedback. Repeat the squeal and notch technique until two or more frequencies feed back simultaneously.

5.1.4.2 Pink noise equalization and feedback tuning

The procedure for School B is very similar, except you first use pink noise and the SA-3050A's calibrated microphone to equalize the monitors. Then revert to School A's technique to find out where the system wants to squeal.

5.1.5 Component Checkout

You can use the SA-3050A to check out the frequency response of any component that you can get pink noise into or out of. Of course, signal processors, preamplifiers and amplifiers can all use the pink noise output of the SA-3050A. Testing RIAA preamplifiers requires a precision inverse RIAA network. A schematic for such a network can be found in Figure 5.1.

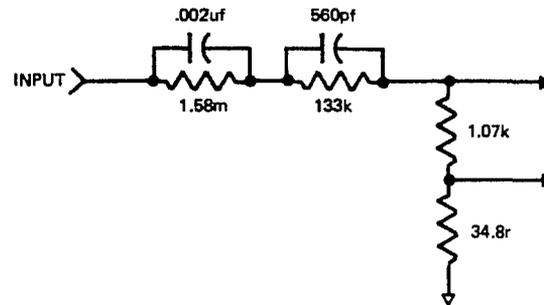


Figure 5.1. An inverse RIAA network.

Signal sources require a pink noise source, compatible with their input format. For instance, a turntable requires a pink noise record, a tape machine requires a pink noise tape. Pink noise sources are available for all common high-fi sources, including CD players.

The diagrams in Figures 5.2 to 5.7 show how to wire everything. The 1 dB/step setting gives maximum resolution. Most purely electronic devices should be able to easily exceed the ± 1 dB resolution of the SA-3050A display. Most tape machines (especially cassette machines) will not.

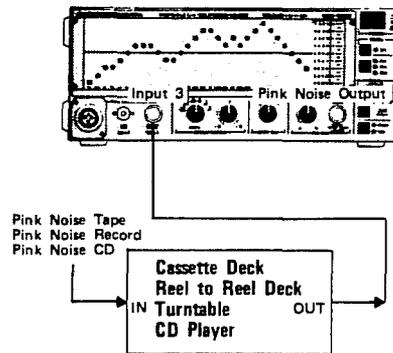


Figure 5.2. Checking a signal source.

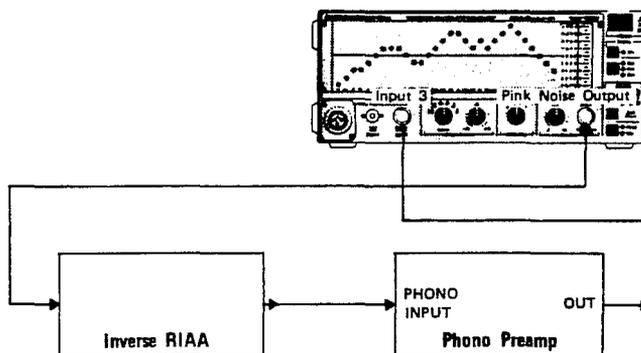


Figure 5.3. Checking a phono preamp.

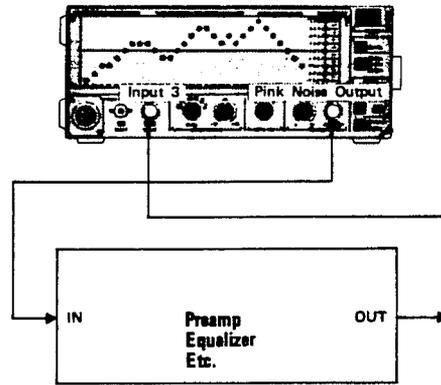


Figure 5.4. Checking a signal processor or preamp.

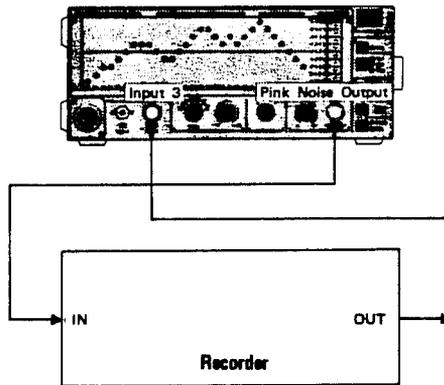


Figure 5.5. Checking a tape recorder.

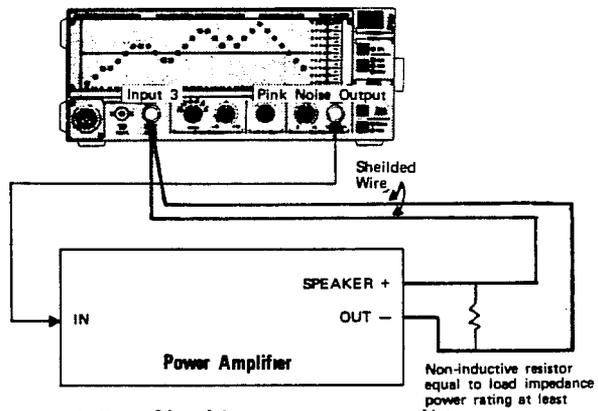


Figure 5.6. Checking a power amplifier.

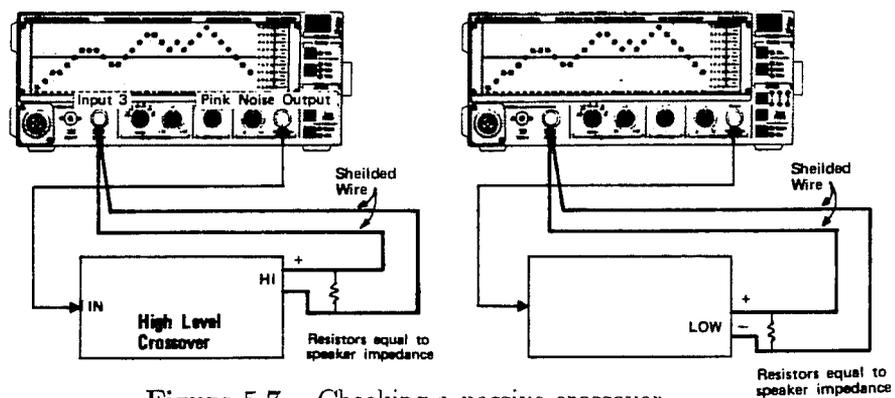


Figure 5.7. Checking a passive crossover.

5.2 Tape Machine Alignment

A real-time analyzer makes child's play out of aligning a tape machine. The concept is simple: record pink noise, play back pink noise, watch on analyzer, tweek until flat.

Of course, it's never quite **that** simple. The problem is one of slow speed recording, and has been with us for quite some time. Simply stated, as the recording speed falls, so does high-frequency headroom. In simple terms, this means that you can't record a 10 kHz signal on tape at the same levels that you would record a 1 kHz signal.

Why? At slower speeds, additional record equalization (treble boost) is needed to overcome the losses in the recording process. The crossover point (point at which you must reduce the level to preserve flat frequency response) occurs at around 7.5 IPS. You can see that at 1.875 IPS (cassette speed), you're going to have to reduce the level considerably. Another loss factor is one of self-erasure. Higher frequencies tend to look more like a part of the bias/erase signal than they do part of the record signal. If you get the levels high enough, they'll erase themselves.

These reasons are precisely why you see record-play frequency response tests in magazines conducted at -20 dB levels. Purely and simply...the machine can't put a high-frequency signal on tape at the same level as a 1 kHz signal unless you drop the level.

When testing tape equipment with the SA-3050A, if the overall response isn't flat, try reducing the level in 5 dB steps. Most cassette machines need to be tested around -15 or -20 dB (below the "zero" level on their meter).

If you have an adjustable bias, three-head tape deck, the SA-3050A makes child's play out of setting the bias.

1. Connect the pink noise output to the deck's input.
2. Put in a cassette, set the record levels, go to recording.
3. Monitor the deck's playback output with the SA-3050A set at slow speed.
4. Adjust the bias for flattest overall response.

It's advisable to start with the bias control at minimum, then increase it until you see a maximum occur in the mid-band (500 Hz - 2 kHz), then continue increasing the bias until the high-frequency range (8 kHz up) smooths out. It's important to end up **past** the mid-band peak rather than before it.

If you have a two-head deck, then there's no easy way to do this short of recording a short stretch of pink noise, then backing the tape up, then playing it back and looking at the playback output on the SA-3050A.

5.3 Sound and Music Monitoring

Many recording studios use a real-time analyzer as a music monitor. More than a light show, the analyzer tells the engineer many things about its input signal. Besides instantaneous amplitude, the display shows the *spectral balance* of the input signal. This is useful in deciding how to equalize a certain instrument, or in trying to figure out how another engineer equalized a certain instrument.

Likewise, if the mix is starting to sound cluttered, a glance at the RTA can tell if the problem is energy build-up in some portion of the frequency spectrum. Conversely, the RTA can identify a portion of the spectrum that is perhaps under-used. The engineer may elect to equalize an instrument to place the dominant portion of its energy content into that region. This helps to make the instrument heard, while not adding clutter to the mix. Of course, the equalization picked should favor the instrument. You might look at the problem as picking the instrument to fit the equalization. It may seem like the chicken before the egg, but it works!

5.4 Crossover Testing

Our last application is crossover testing. Actually, this is just an extension of the setups used for component checking. The setup is shown in Figure 5.7. You'll need the following items:

1. Two load resistors, resistance equal to the impedance rating of the crossover, 1 Watt power rating is sufficient. You can also use the speakers that will be connected to the crossover as a load. If you're checking an active crossover, you don't need the load resistors. Use the setup shown in Figure 5.8.
2. The SA-3050A.

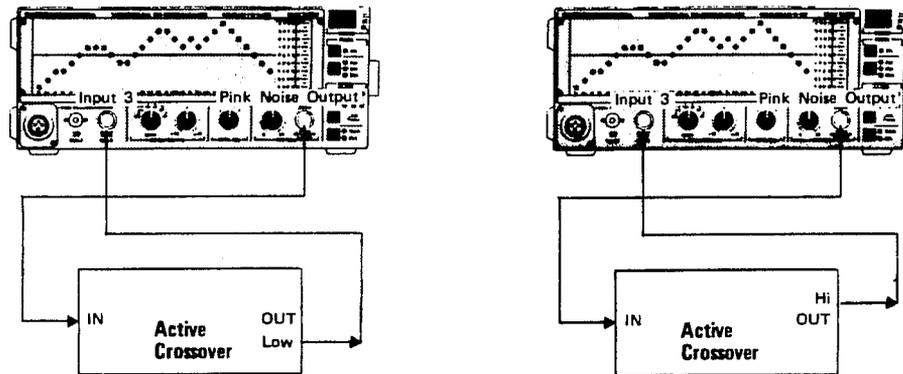


Figure 5.8. Checking an active crossover.

It really doesn't matter which output you test first. Set the pink noise output of the SA-3050A at maximum. Set the SA-3050A for a display reading about 3 LEDs below the top of the display. Use the 1 dB/step setting. Count three LEDs down from the flat portion of the line shown on the display. Move to the right or left until you intersect the falling portion of the line shown on the display. Now move straight down to read the approximate crossover frequency. Store this reading in one of the memories.

Now repeat the operation for the other crossover output. Store this reading also. Now toggle between the two memories. The two curves should look reasonably symmetrical (mirror images of each other). Beware however, some crossover designs are deliberately asymmetrical.

If you're using load resistors for testing, note that the load that is normally connected to the crossover looks only vaguely like a resistor. It's a good idea to test crossovers both ways: using a dummy load resistor (ideal conditions) and using the actual loudspeaker loads (real-world conditions).

6. Printer Operation

An optional printer interface allows making hard-copy print-outs of the SA-3050A's display. The printer interface must be installed at time-of-order.

The printer output is a "form" with space provided for logging the time and date of the job, as well as any other data. You can also print the response-graph portion of the form, without the logging spaces.

6.1 Printers Supported

The SA-3050A printer interface is designed to connect to any Centronics-type¹ parallel interface compatible printer.

6.2 Printing

Printing the display is easy.

1. Store and recall the response curve that you wish to print. The freeze mode will also work.
2. Make sure that the printer is loaded with paper, connected to a power source, turned on, and on-line.
3. Press the rear-panel PRINT button. Pressing the PRINT button twice, quickly, prints only the response-graph information.

6.3 Connector Pin Designations

The printer interface connector uses a DB25F connector. The connector wiring is the same as that used by the IBM PC.² The connector pin designations are shown in Figure 6.1.

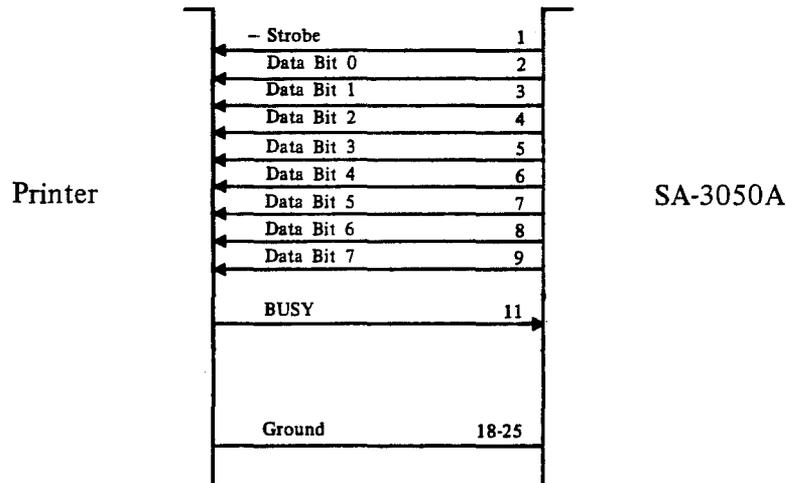


Figure 6.1. Parallel Interface Signals and Pin Assignments.

1. Centronics is a trademark of Centronics Data Computer Corporation.
2. IBM is a trademark of International Business Machines Corporation.

7. Battery Operation

An optional battery package allows the SA-3050A to be operated without connection to the AC mains supply. The battery supply has sufficient capacity to operate the SA-3050A continuously for 4.5 to 5 hours. The battery package uses four 2 volt, 5 ampere-hour Gates Cyclon sealed lead-acid batteries.

An integral battery charger operates when the SA-3050A is plugged into the AC mains supply. The instrument must be turned off to fully charge the battery.

If the SA-3050A is plugged into the AC mains supply while it is operating on battery power, the instrument may appear to malfunction. Proper operation may be restored by turning the SA-3050A off and on again. This is not harmful.

7.1 Charging the Battery

1. Plug the SA-3050A into a suitable source of AC power.
2. Make sure that the SA-3050A is turned off.
3. Leave the SA-3050A for a period of 6 to 10 hours. (twice the discharge time)
4. The battery charger automatically adjusts the rate of charge as the battery reaches full charge. Once this occurs, the battery charger switches to trickle-charge mode to maintain the battery at full charge.
5. Operating the SA-3050A on AC power will very, very slowly charge the battery.

7.2 Getting the Most From the Battery Supply.

The SA-3050A uses a sealed, lead-acid storage battery. These batteries are not of the Nickel-Cadmium type commonly used for rechargeable applications. Lead-acid batteries offer some significant advantages over Ni-Cad batteries.

- Given proper charging and discharge depth, the batteries should deliver 1000 to 2000 charge/discharge cycles.
- No Ni-Cad memory effects due to constant charging. You can operate the SA-3050A continuously from AC line power without causing the battery pack to get "lazy" (Ni-Cads are notorious for this).
- Higher energy density. The lead-acid design delivers more energy per unit volume than other rechargeable battery systems.
- Lead-acid batteries can be charged in less time than Ni-Cads.
- Wide temperature range: -65 degrees Celsius to +65 degrees Celsius.

Getting maximum performance from your battery pack is simple. Follow the following suggestions:

- Lead-acid batteries do not operate well when deeply discharged. Operate the SA-3050A until the low-battery indicator begins flashing. When the indicator begins flashing, you have 10 to 20 minutes of operation left. Cease operation and recharge the battery when the indicator extinguishes. A low-voltage cutout prevents operation of the SA-3050A once the battery voltage reaches this point.
- Store the unit with the batteries fully charged.

NOTE

The low-voltage cutout could make the instrument appear to be totally dead. If the SA-3050A appears dead, plug the instrument into a source of AC power. Ensure that the power switch is set to the OFF position. Leave the instrument plugged in for at least two hours.

If this "revives" the instrument, you can resume operation of the SA-3050A using AC line power. Be sure to charge the batteries for another six to eight hours. If not, consult the factory for additional information.

Plugging the SA-3050A into AC line power when the unit is operating on battery power may cause the microprocessor to "hang." If this occurs, turn the unit off, then back on again.

Store the unit with the batteries fully charged.

8. Theory of Operation

This section of the SA-3050A discusses the theory of operation to a functional block diagram level. This section is intended to familiarize you with the major circuit blocks that make up the SA-3050As circuitry.

The circuitry of the SA-3050A is contained on two printed circuit boards (PCBs). The analog circuitry and part of the digital circuitry is contained on the main PCB, while the microprocessor and its related circuitry is contained on a second PCB.

The discussion begins at the input connectors to the SA-3050A.

8.1 Input Circuitry

Microphone level signals are applied to the preamplifier circuit. The preamplifier consists of three op-amps connected as an instrumentation amplifier. This converts the balanced input signal into an unbalanced signal, and supplies enough gain to bring the input signal up to the SA-3050As standard operating level. The overall gain of the preamplifier circuit is set by the dB INPUT SENSITIVITY switch. Simplex (phantom) powering for 12 V condenser microphones is also provided at the microphone input connector.

Line level signals applied to the 1/4 inch tip-ring-sleeve phone jack are first attenuated to microphone level, and then sent to the preamplifier circuit.

Signals applied to the BNC connector are applied to an op-amp which provides any voltage gain needed as well as impedance buffering. The overall gain is set by the dB INPUT SENSITIVITY switch.

The output signals from both preamplifiers are then summed together. The gain of the summing amplifier is also set by the dB INPUT SENSITIVITY switch. The last stage of the input circuit is another variable gain stage. The gain of this stage is controlled by the dB INPUT SENSITIVITY control. Last, the signal is scaled to the correct level by the dB PER STEP switch.

8.2 One-Third Octave Filters

The output of the input circuit drives the inputs of thirty one-third octave filters, and the SPL circuit. Since the SPL circuit is simpler, we'll discuss it first.

The input to the SPL circuit is a broadband signal. It is rectified, filtered, and applied to one input of the multiplexer.

Each of the thirty one-third octave filters is the same, except for parts values. Each of the filters is made up of two cascaded 2 pole bandpass filters. The Q (bandwidth) and center frequency of each filter is precisely set by precision resistors and capacitors. The center frequencies of the two filters are split on each side of the ANSI specified center frequencies. The proper combination of filter Q and the offset center frequencies combines to give an overall curve that meets the ANSI specification for a one-third octave, Class II, type E filter.

Each of the thirty filter outputs is then rectified, filtered, and sent on to the multiplexer.

8.3 Signal Multiplexing and Analog to Digital Conversion

The multiplexer scans each filter output, in sequence, for a fraction of a second. Each filter output is scanned for the same period of time. Another multiplexer input reads the output of the SPL circuit discussed earlier.

The output of the multiplexer contains the sequential output of each of the thirty analyzer filters and the SPL circuit output. This signal is then sent to the A-D converter.

The A-D converter converts the output of the multiplexer into a digital word whose value represents the value of its input signal. Since the output of the multiplexer contains 31 different values, the A-D converter repeats the conversion process 31 times, once for each of the multiplexer inputs.

8.4 Microprocessor Circuitry

The microprocessor (μ P) controls the timing of various events within the SA-3050A, writes information into its memory, and retrieves that information when needed. In addition, the μ P reads the settings of various switches and maintains the status of the various LED indicators on the front panel. The μ P program is stored in a 64K EPROM (Electrically Programmable Read Only Memory). A 16K RAM (Random Access Memory) provides memory for the 6 display memories, and for scratch-pad memory used during instrument operation.

Periodically, the microprocessor instructs the A-D converter to make another conversion and then place the result on the internal data bus. The μ P reads this data, and uses it to drive the display and/or memory. Once the μ P has the data from the A-D converter, it performs some math on it (decibel conversion), sends it to the display, and stores it in the temporary display memory.

When told to store a reading, the μ P transfers the contents of the temporary display memory to another RAM location for safekeeping. On recall, the μ P does the opposite, moving the data from RAM to the temporary display memory.

8.5 LED Display

The SA-3050A display gets its data from the μ P bus. The display is row and column multiplexed to conserve power. Eight-bit latches latch the row and column data for each column in the display. The intersection of a selected row and selected column results in an illuminated led at that position of the selected column.

8.6 Pink Noise Generator

The pink noise generator uses a series of shift registers, connected as a pseudo-random sequence generator. The output of this generator looks (and is, for all practical purposes) like white noise. The recycle time of the generator is long enough that the pattern repeats once in hours of cycles.

The output of the noise generator drives the pinking filter. The pinking filter applies a precise 3 dB/octave rolloff to its input signal. The resultant output is pink noise.

The output of the pink noise generator is a small power amplifier IC. The output amplifier has sufficient output current capability to deliver 150 mW into a speaker-impedance load.

9. Service Information

WARNING

There are no user-serviceable parts inside the SA-3050A. Lethal voltages are present inside the case. Refer all servicing to qualified service personnel.

The SA-3050A is covered by a one year limited warranty. Repair and parts service is available at the address given below.

9.1 Warranty

If the unit fails, here is what we will do:

1. Repair or replace the unit (our option). If the unit is under warranty, there is no charge for this service. If the unit is out of warranty, there is a nominal time/materials charge made for the repair. We reserve the right to return out-of-warranty units COD.
2. Return the unit to you, shipping prepaid, via UPS. Other shipping arrangements are at the owner's expense.

Here is what you must do:

1. **IMPORTANT:** Complete, sign and return the warranty registration card. You have 15 days from the date of sale in which to do this. Aside from knowing who buys our products, we'll use the information to keep you informed of any software updates for your SA-3050A.
2. Keep the sales receipt for the unit as proof of purchase. The date on this receipt establishes the starting date for the warranty.
3. Write or call the factory for a Return Authorization number.
4. Return the SA-3050A in its original shipping carton to us or our designated agent, freight prepaid. If you don't have one, we'll be glad to send you one. If you're in a hurry, put the unit into a large plastic bag. Now find an oversized carton (four inches clearance around the unit) and pack the unit using plastic "peanuts", plastic foam, or "bubble pack". If you ship the unit this way, we are not responsible for *any* shipping damage incurred.
5. Include a concise statement of the problem, and any special conditions necessary to duplicate it. Be sure to include your particulars (name, address, city, state, zip, and a **daytime phone number**), and a **copy** of the sales receipt.
6. Take reasonable care of the unit, as stated in the warranty and in this manual. Don't modify it (all bets are off and your warranty is void).

In case of dispute, the legal warranty included with the unit shall supercede this plain-text version.

9.2 Factory Service

Parts or repair service is available at the address and phone number given below:

Audio Control Industrial
Attn: Service Department
22313 70th Ave. W.
Mountlake Terrace WA, 98043
(206) 775-8461

Legalese

This is the only warranty given by Audio Control Industrial. This warranty gives you specific legal rights, and you may also have rights which vary from state to state. Promises of how well your Audio Control Industrial product will work are not implied by this warranty. Other than what we've covered in this warranty, we have no obligation, express or implied. Also, we will not be obligated for direct or indirect consequential damage to your system caused by hooking up the Audio Control Industrial Product.

Failure to send in the properly completed warranty card negates any service claims.