

**BEFORE PROCEEDING WITH COMPLETE UNPACKING AND SETUP,  
READ THE SECTION ON UNPACKING AND INSPECTION**



**model 5547A  
model 5549A**

**1/3 OCTAVE EQUALIZERS**

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# SECTION I

## PREFACE

Thank you for purchasing this JBL/UREI product. We have prepared this instruction manual to enable you to achieve optimum utility and performance from your new equalizer. We encourage you to read and to make use of the material contained in this manual. We welcome your suggestions and comments on our products and on this manual.

This manual is dedicated to all the people who are interested in learning about the capabilities and limitations of our products in order to best use them. Learn, Enjoy and Share.

## UNPACKING AND INSPECTION

Your new JBL/UREI Equalizer was carefully packed at the factory, and the container was designed to protect the unit during shipment. Nevertheless, we recommend careful examination of the shipping carton and its contents for any sign of physical damage which may have occurred in transit.

If damage is evident, do not destroy any of the packing material or the carton, and immediately notify the carrier of a possible claim for damage. Damage claims must be made by you.

The shipping carton should contain:

The JBL/UREI Model 5547A One-Third Octave Graphic Equalizer,  
or the JBL/UREI Model 5549A One-Third Octave Room Equalizer

This Instruction Manual

An envelope containing Rack Mounting Hardware

## SETUP AND OPERATION-THE SHORT FORM

The following section contains an abbreviated installation procedure followed by setup and operation instructions. Detailed installation techniques and operational concerns follow in Section II. Section III is devoted to Applications and Section IV contains

**Maintenance and Warranty Information.****INSTALLATION**

1. Check that the AC Line Voltage at your installation is correct for the Equalizer you have. A label on the rear panel and a tag on the AC power cord identify the correct AC mains voltage for which your unit is wired. If your unit is not wired for the correct voltage, do not proceed any further with installation. See AC POWER in Section II for further details.

**WARNING: A SIGNIFICANT FIRE AND SHOCK HAZARD MAY EXIST IF THE EQUALIZER IS CONNECTED TO AN AC POWER SOURCE OTHER THAN THAT FOR WHICH IT IS RATED AND LABELLED.**

2. Rack mount your Equalizer with the enclosed rack mounting hardware. The Equalizer is equipped with rack ears which may be moved forward to allow the controls to be recessed and/or the optional security cover mounted flush with the front of the rack ears. For detailed instructions on moving the rack ears see RACK EAR PLACEMENT in Section II.

The Model 5547A/5549A Equalizers will operate satisfactorily over a range of ambient temperatures from 0° C to +50° C ( +32° F to +122° F ), and up to 80% non-condensing relative humidity.

If the unit is installed in an equipment rack, console or other area along with high heat producing equipment (such as power amplifiers), adequate ventilation should be provided to assure longest component life. Also, while internal circuitry susceptible to hum pickup is sufficiently shielded from moderate electromagnetic fields, avoid mounting the unit immediately adjacent to large power transformers, motors etc.

3. Connect audio input and output wiring. You have the choice of XL-style, phone plug or barrier strip. See Section II SIGNAL WIRING and SIGNAL CONNECTIONS for a complete description of the whys, wherefores and how-to's of wiring the connectors. Set the INPUT and OUTPUT level controls at the bottom of their range and all frequency controls at the "0" line. The three switches at the left end of the front panel labelled EQ, CUT FILTERS and HF SLOPE should all be in the out position.

4. Turn power on and the ON indicator will be illuminated. After a brief (3to 5 seconds) delay the bypass relay will engage, allowing

signal to pass through the unit. Apply program signal to the input. Monitor the output signal which should be passed through the Equalizer without gain change or frequency response alteration.

5. Depress the EQ switch. With no equalization "dialed in" there should be no difference in the audio. Check for proper operation of the one-third octave filters by moving several of the slide controls and listening to the result. Depress the switch labelled CUT FILTERS and turn the rotary knobs to hear the effect of low and high frequency bandwidth limitation. Switch the EQ Out and hear the signal return to flat frequency response. Note that the EQ button removes both the one-third octave filters and the end cut filters from the signal path. It is not necessary to have both buttons in the out position to remove all equalization.

6. The INPUT and OUTPUT controls may now be adjusted for optimum performance.

**IMPORTANT:** The Models 5547A and 5549A incorporate an unusually flexible input/output level control configuration to allow precise tailoring of system gain and headroom. In order to best utilize this capability it is important for you to understand the action of these controls.

The INPUT control changes the gain of the input amplifier from unity ( 0 dB ) at the bottom of its travel to +20 dB at the top. This allows low input signal levels to be raised to achieve better signal-to-noise ratio in the equalizer. The OUTPUT control changes the gain of the output stage. At the bottom of its travel the gain is unity -- at the top it is -20 dB. The net effect is that the equalizer will always be a unity gain device IF BOTH CONTROLS ARE MOVED TOGETHER (except for any gain or loss due to the equalization control settings).

To properly adjust the equalizer level controls, provide an "average" program signal to the equalizer and move both level controls together until the LED labelled "20" flashes regularly, the LED labelled "10" flashes on loud peaks and the LED labelled "0" almost never flashes. If Boost Equalization is used on the Model 5547A it may be necessary to lower the controls to allow for the extra gain in the passband of the filters. The LED display circuit monitors all necessary points inside the equalizer and indicates worst-case headroom.

## SECTION II

Section II of this manual provides additional detailed information on Installation, Setup and Operation to supplement Section I.

### SIGNAL CONNECTIONS

The JBL/UREI Model 5547A/5549A Equalizers provide exceptional flexibility to the user who may choose any of three connector types for input or output signal connection. A brief discussion of the reasons for using one connector type over another follows. In all cases good wiring practice must be followed using good quality two-conductor shielded cable. This will ensure maximum reliability and low noise operation.

#### BARRIER STRIP

The barrier strip is the preferred connector for all fixed installations. It is a highly reliable connector which requires only wire strippers and a screwdriver to use. Wires may optionally be terminated in #6 spade or ring lugs or a 6 position fanning strip. When using stranded wire take care that no individual strands work loose to short to adjacent connectors or to the chassis. Because of the high pressure exerted on the wire or lug by the screw terminal, the barrier strip connector is less subject to oxidation or corrosion than many other types.

#### XL CONNECTORS

This popular connector style allows quick connect and disconnect of cables to and from the equalizer which is of advantage in temporary or portable sound systems. Additional advantages include a locking connection using standard connectors and the ability to plug the two cable end connectors together to bypass a failed unit.

Disadvantages include the cost of the mating connectors and the time to wire them. Also because the connectors are not "gas-tight" there may be a tendency for the contacts to oxidize in some environments over time, so we recommend plugging and unplugging the connector at least once a year. This will remove any oxide build-up.

## PHONE JACKS

The third connector style that is available is the 6.3mm (1/4 inch) phone jack. The equalizer uses 3-conductor jacks to allow balanced operation, but 2-conductor plugs may be used if unbalanced wiring is desired.

Phone jacks are especially useful in systems where the majority of equipment is supplied with the same type connector.

## SIGNAL WIRING

Connections to and from other system components are simple and straightforward. Pin three of the XL-type connector is "hot" or in phase with the (+) terminal of the barrier strip, and pin two is in phase with the (-) terminal of the barrier strip. The tip of the 1/4" phone jack is in phase with the (+) terminal, and the ring is in phase with the (-) terminal.

Since all three connectors are wired in parallel, only one should be used at a given time (unless it is specifically desired to loop a signal through the input).

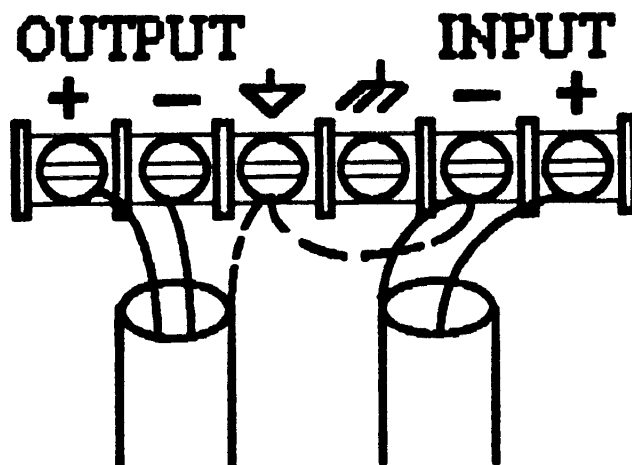
The equalizer will not unbalance floating or balanced output sources since the input circuits consist of balanced differential amplifiers. Balanced wiring is the preferred style, especially when running long lines, due to the ability of a balanced input to reject signals (such as hum fields) which are induced equally into both of the signal carrying conductors. Even if the previous piece of equipment has an unbalanced output it may be advantageous to wire it to the input of the equalizer as though it were balanced. This takes advantage of the ability of the equalizer input to reject common-mode noise.

To use an unbalanced source, wire the signal carrying conductor of the cable from that source to the (+) terminal of the barrier strip (XL-type pin 3, phone plug tip), and wire the shield to the (-) terminal of the barrier strip (XL-type pin 2, phone plug ring). The unused connector terminal, pin 1 (sleeve), may also be connected to shield ground if that is compatible with your system grounding. (See the section on Grounding and the illustration of barrier strip connections for more information about this.) Unbalanced connections may be simplified in this last case by using two-conductor standard phone plugs because they automatically connect the ring and sleeve together when inserted in the input and

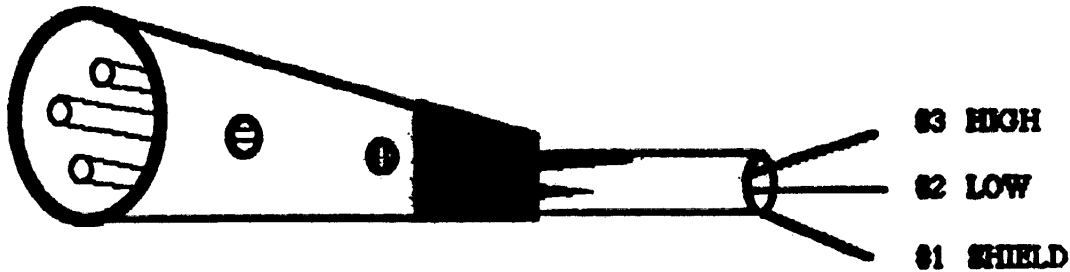
output jacks.

Output wiring is similar to input wiring. Because the output stage incorporates a transformer it is important to remember that connection must be made to both (+) and (-) terminals of the output. The output will drive any load of 600 ohms or greater and does not require a termination. If the equalizer is driving a long line (in excess of 30 meters [100 feet]) a 620 ohm termination resistor at the other end of the line from the equalizer, in combination with the use of balanced lines, will reduce the possibility of noise pickup in those lines.

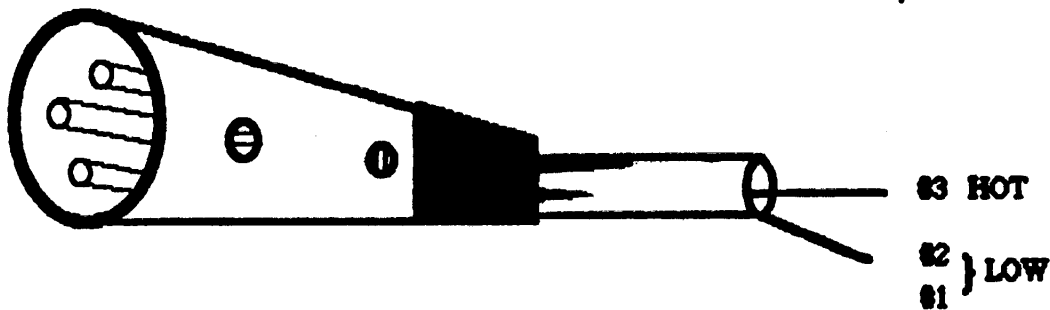
We recommend that two-conductor shielded cable be used even in an installation using unbalanced wiring. Do not depend on the shield wire itself to complete the signal connection. Stranded shield wires are more subject to breakage, especially in portable installations, than the more protected internal insulated wires. Using this wiring system the worst that would happen with a broken shield would be a rise in noise or hum due to the lack of shielding. If the ground connection were completely lost this would result either in loss of audio or a terrible loud hum.



BALANCED AND UNBALANCED  
BARRIER STRIP CONNECTIONS  
{GROUND CONNECTIONS DEPEND ON YOUR INSTALLATION}



BALANCED XL CONNECTION



UNBALANCED XL CONNECTION

### IMPEDANCE AND TERMINATION FOR 600 OHM LINES

In the USA the early history of the audio industry is very closely tied to the history and technology of the telephone industry. Much of the early equipment used for public address systems, recording, broadcast and reproduction of sound was either designed by or heavily influenced by the scientists and engineers at Bell Telephone Laboratories. The technology that they and others developed has had a lasting influence on the design and specification of all types of audio products. Among their contributions was the 600 ohm transmission line.

The 600 ohm line was developed because of a need for a standardized impedance for long distance transmission lines. Transmission and reception equipment using vacuum tubes, transformers, and passive equalization and mixing networks require known source and load impedances to achieve predictable results. Because much of the early professional audio equipment was



designed by telephone company people or used similar types of equipment, it is not surprising that the 600 ohm line became a standard in the professional audio industry.

Correct use of the 600 ohm transmission line requires a signal source with an exact 600 ohm source impedance, and a receiving device which also has a 600 ohm input impedance. If a device does not have the correct impedance, it must be modified until it does. In the case of a device with a lower than required source impedance this would require addition of a series buildout resistor (or two in the case of a balanced line). In the case of a higher impedance input a resistor across the input will suffice. Other variations require either a resistive network or transformer to match impedances.

Therefore, if the equalizer is to be used in a 600 ohm system attach a 620 ohm resistor across the input terminals of the equalizer to terminate the source and build out the output impedance with a 560 ohm resistor in series with the output (or one 270 ohm resistor in each leg of a balanced system.)

However, most modern audio systems do not require the use of 600 ohm transmission line practices. This is for two reasons. First most audio systems are relatively small (especially as compared to a telephone network) and the cabling between parts of the system is under more careful control, and second, modern audio electronic products are no longer designed to require the use of 600 ohm line. Matching of input and output impedances is no longer necessary because the output impedance of a device may be made very low and the input impedance very high. Multiple inputs may thus be connected in parallel to the same source with ease and no loss of signal level.

## AC POWER

All JBL/UREI equalizers shipped to destinations in the USA, Canada, and Japan are wired for 120V 60Hz operation. They will also operate properly at 100 VAC 50 Hz. Equalizers wired for operation on 220/240 VAC 50 Hz are identified as such on the packing box, a sticker on the rear panel and by a tag attached to the power cord.

The AC Power Cord uses International standard wire colors. Should it ever be necessary to replace the power plug the wire color code is: Brown=Line; Blue=Neutral; Green/Yellow=Ground.

## GROUNDING

For safe operation this equalizer must be connected to a good mechanical ground. This provides a current path for any voltage which might appear on the chassis due to a severe electrical fault in the equalizer. Without this path the equalizer might be a shock hazard. In addition, a good quality ground on the chassis provides shielding from external fields and minimizes radiation of internal fields to the outside world. To comply with safety regulations in many localities, and to protect our customers, we provide this product with a ground connection through a three-wire power cord.

In many situations this will present no problem. But there are instances where a hum or buzz will be introduced due to a phenomenon known as a ground loop. This results when there is a significant potential between the audio ground of the previous piece of equipment and the mechanical ground to which the equalizer has been connected.

If this is the case, the first attempt at a solution should be to remove the strap on the rear panel barrier strip which connects audio ground and chassis. Removal of this strap may have a significant effect on reducing the hum. Audio ground will then be referenced from the signal source and the chassis ground will be separate but still connected to mechanical ground for shock protection.

In some instances the voltage difference between the grounds will be so great that a direct connection to mechanical ground is not possible without hum in the equalizer output. Use of an isolation transformer in the input signal line may allow the signal to be connected while maintaining ground isolation. Check for this using a 3 prong to 2 prong AC adaptor between the power cord and the power outlet, temporarily ungrounding the equalizer. Try the equalizer both with and without the ground strap on the barrier strip. Determine which connection works best. Remember, however, that for safety you must still have a connection to chassis ground. This is normally made through a properly grounded third pin connection.

## HEADROOM CONTROL

Signal-to-noise ratio is an important consideration in an audio system. But all too frequently much of the performance capability

of a given piece of equipment may be thrown away during actual use. This is because the program signal must be kept below the maximum signal handling capability of the unit to prevent clipping. The difference between the program level and the maximum level is called headroom.

Different systems may need different amounts of headroom because of program material variations. For example, a large amount of headroom is normally required in a live recording studio situation where program level may change rapidly over a short period of time, but where compression or limiting is not yet appropriate (or even desirable). Conversely, an audio system which plays primarily pre-recorded program material with much more tightly controlled dynamic range may operate well with less headroom.

In many systems it is possible to determine the amount of headroom necessary for good operation. If, in such a situation, it were possible to optimize the signal level for each piece of equipment in the audio chain, it would be possible to make a trade-off of any excess headroom for improved signal-to-noise ratio. Unfortunately, this is not always easy to do.

An equalizer in a sound system should be operated at as high a signal level as possible to keep the signal-to-noise ratio high, but with 10-20dB of headroom to account for the crest factor in the program material. The JBL/UREI equalizers are equipped with a novel and extremely convenient method for accomplishing headroom adjustment. To understand how this works refer to FIGURE 1 which shows the noise/headroom performance of a typical equalizer with -90 dBm output noise and maximum output of +20 dBm. (Note that the noise is generated within the equalizer and not by the preceding equipment.) In FIGURE 1A the equalizer is being driven by a signal level of +4 dBu (Ref. 0dBu = 0.775 V). The signal-to-noise is 94 dB ( $90+4=94$ ) and the headroom is 16 dB ( $+20-\{+4\}=16$ ).

In FIGURE 1B the same equalizer is being driven by a signal level of -10 dBu. The signal-to-noise has degraded by 14 dB to -80 dB ( $90+\{-10\}=80$ ) and the headroom has increased by 14 dB to 30 dB ( $+20-\{-10\}=30$ ). Occasionally 30 dB of headroom is appropriate but in many situations it is excessive and we would prefer to trade off excess headroom for better noise performance.

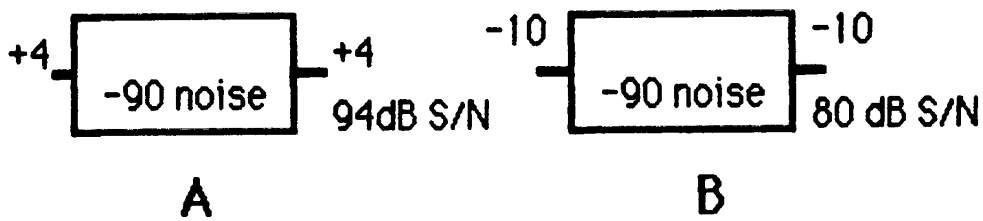


FIGURE 1 TYPICAL ACTIVE EQUALIZER

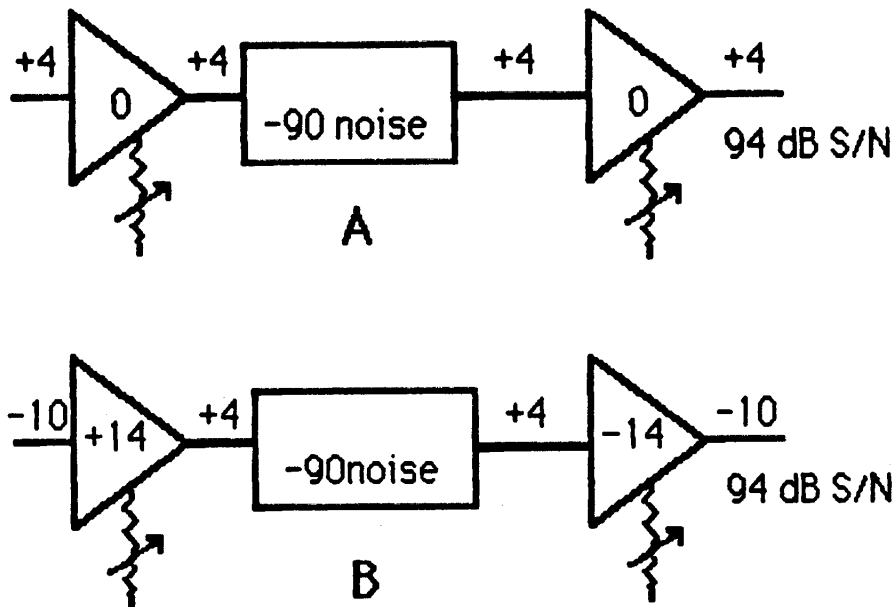
FIGURE 2. JBL/UREI EQUALIZER WITH  
IMPROVED SIGNAL TO NOISE

Figure 2 shows what happens in the JBL/UREI headroom circuit for different levels. Note that a variable gain amplifier is inserted before and after the equalizer. In the case of a +4 dBu input signal the gains of the two amplifiers are set to unity and as shown in FIGURE 2A the noise and headroom numbers are unchanged from the example in FIGURE 1A. In the case of the -10 dBu signal however, the gain of the input amplifier may be raised by 14 dB and the gain of the output amplifier reduced by the same amount. The signal level actually seen by the equalizer section has now increased back to the +4 dBu level and the signal to noise and headroom numbers at the output of the equalizer have been restored to their previous values as in FIGURE 2A. The attenuation in the output amplifier then returns the signal to the original -10

**dBu level. In actual practice the output attenuator further improves the noise output from the device as a whole because it not only attenuates the signal from the equalizer section, but attenuates the noise output from the equalizer section as well. This continues until the noise floor of the output amplifier is reached.**

**Now, of course, all of this gain adjustment could be performed external to the equalizer itself, but that would require additional amplifiers, pads and wiring with the attendant increase in circuit complexity and possible reduction in system reliability. It is much more convenient and cost-effective to include the facility right in the equalizer. Two adjacent front panel linear slide pots serve as gain controls for the input and output stages with unity to +20dB of gain and unity to -20dB of attenuation, respectively. If the controls are moved together the gain through the equalizer does not apparently change to the outside world, but it can make a significant difference to the noise of the system. Of course the controls may be moved separately (that's why we provided two controls instead of one) and in some instances it may be necessary to do so.**

**Traditionally the adjustment of controls affecting headroom has either been a hit-or-miss proposition or one that required test equipment and time. Neither method is optimum. We have therefore included a simple, but effective method for setting the headroom controls. A peak reading LED display of available headroom is positioned next to the headroom adjustment controls on the front panel. This display, calibrated in 10 dB steps from 30 dB to 0 dB (clipping), gives the operator immediate visible indication of peak signal level and allows the adjustment of headroom in less time than it takes to explain or read.**

### **HEADROOM INDICATOR**

**Signal levels are monitored throughout the equalizer. The amount of headroom available before clipping is indicated by four light emitting diodes (LEDs) labelled "30", "20", "10", and "0dB". The LED(s) flashes ON when peak signals exceed the indicated level in any part of the circuit. They remain ON long enough to be seen even if the signal is only a brief transient. To avoid audible distortion the signal must be lowered if the "0 dB" LED flashes frequently.**

**A listening test may show that the "0 dB" LED flashes before noticeable distortion is audible. This may be due to high amplitude, short duration transients. You may never hear the resulting**

overload, depending on program material, listening conditions, and yes, even your own ears. Therefore the overload LED should not be ignored.

### END CUT FILTERS

Although there are some who demand that the frequency response of their audio system should extend "from DC to Channel Five," restriction of frequency bandwidth on a selective basis is not only reasonable, but often necessary. The human body is not capable of sensing acoustic input over that wide a bandwidth even if the system transducers were able to reproduce it. There are numerous situations where the restriction of bandwidth will improve the sound - not degrade it. For example acoustic leakage from one instrument in a studio to the microphone of another instrument. Also, restriction of the passband of a loudspeaker system may improve intelligibility or increase power handling capability by restricting out-of-band energy.

As a matter of fact, there are some occasions where the only equalization needed is some judicious roll-off of the high and/or low frequencies. In the recording studio environment this means that the equalizer may serve double duty. The end cut filters of these equalizers are 12 dB/octave and are tunable through a wide range. In addition the High Cut filter (low-pass) may be switched to 6 dB/octave to allow more gentle tailoring of a House Curve. This last innovation was pioneered by UREI and has since been copied by others because of its utility.

### BYPASS

The Models 5547A and 5549A incorporate three different methods of circuit bypass for the two different purposes of bypass--circuit or AC mains power failure, EQ in-out audition and End Cut bypass. Bypass on circuit failure ensures that program material will continue uninterrupted even if the equalizer fails during the program. This is accomplished via a relay with bifurcated gold contacts which connects the output terminals directly to the input terminals on failure of the AC supply voltage. If a failure occurs inside the equalizer but without an AC supply failure (for example: an op-amp going bad with resultant noise output), the power switch may be turned off and the signal will bypass the equalizer. The relay circuit has a short turn-on delay to prevent any possible power-on transients generated within the equalizer from being coupled to the succeeding equipment.

If there is any DC voltage on the output of the preceding equipment, it will be coupled through the relay and an audible pop or click may be heard when the relay engages. This is not a fault within the equalizer, and should it occur the appropriate corrective action is to remove the DC voltage.

The second type of bypass incorporated is for auditioning the effect of equalization. This circuit, activated by the front panel pushbutton labelled EQ, bypasses only the filter sections while retaining input and output amplifier buffers and gain control. The buffers are kept in circuit so that the interface to preceding and succeeding equipment is retained with no chance for gain shift or ground loop to occur as EQ is switched in and out. Some other equalizers provide only a hard wire bypass of the entire unit for this purpose which requires that the wiring be made compatible for both modes. This may not always be convenient to do. The EQ Bypass circuit allows the effect of any change in equalization to be easily heard without distracting level changes.

The third bypass circuit is just for the End Cut Filters. The End Cut Filters may be completely removed from the signal path so that they have absolutely no effect on program material. Many equalizers only allow the filters to be tuned to the widest bandwidth position to "remove" the filters from the circuit. While this allows most of the amplitude variation to be eliminated, some phase shift will still occur at the extremes of the passband.

## RACK EAR PLACEMENT

When shipped, the Model 5547A/5549A rack ears will be in the rearward position. The chassis has six threaded holes for placing the rack ears in either of two positions. The forward position is used when the security cover is used and flush mounting is required. When in place, the front surface of the security cover will be flush with the front surface of the rack ears. To move the rack ears use the following procedure:

Remove the four screws holding each rack ear to the chassis. Note that there are six holes, and that the four rear holes were used. Reattach the rack ear using the four front holes. We recommend the use of Loctite® Type 222, 242, Permalok®LH 113 or similar threadlocking adhesive on the screw threads.

## **SECURITY COVER**

In some installations it may be necessary to safeguard the equalizer control settings from deliberate or accidental mis-adjustment. In some environments the controls may benefit from protection against entry of dirt and dust.

The Model SC5 Security Cover offers protection for all operating controls of 3-1/2 in (89 mm) panel height JBL-UREI Electronic Products. The Security Cover is a five-sided box of smoked-gray plexiglass. It attaches to the front panel of the unit with two 6-32 x 2 in. Phillips head machine screws. Control positions and display status may be viewed through the cover after installation.

As noted earlier, the equalizer may be rack mounted in two different ways: flush mounted with the panel and rack ears in the same plane, or recess mounted with the rack ears extending forward so that the controls do not extend beyond the front panel of the rack ears. With the unit flush mounted the Security Cover will extend approximately 1-1/4 in. (32 mm) in front of the front surface of the rack ears. With the unit recess mounted the Security Cover is flush with the rack ears.

Actual installation of the Security Cover is simply a matter of placing it against the front panel of the unit, pushing the screws through the holes in the cover and screwing them into the two holes provided on the front panel. Note that the screws in the Security Cover match up with the holes in the front panel in only one orientation. Take care not to scratch the front panel of the unit with the screws.

The Security Cover may be cleaned with any mild, non-abrasive cleaner and a clean cloth.



## SECTION III

### APPLICATION- GENERAL

There are two general classes of application for one-third octave equalizers - creative and corrective. In this section we will describe some of the important concerns which face the user in his/her attempt to achieve his/her ends. We believe that the following material will be of interest and assistance.

This section should not be considered as a complete guide to equalization, but rather as an initial stimulant to correct use and experimentation with the equipment. We believe that it is of primary importance for the user to know and understand the equipment he/she works with and its capabilities.

As in any other field, "practice is the key to success." Just as it is not reasonable to expect to get beautiful music out of an instrument the first time you play it, it is also not reasonable to expect to immediately master the techniques necessary to adjust a one-third graphic equalizer for desired results the first time you work with it.

The remainder of this section first covers areas of common concern in the use of all equalizers, followed by specific creative and corrective equalization concerns and methods.

### SPECTRAL DISTRIBUTION IN THE AUDIO SIGNAL

It may seem obvious, but we'll point it out anyway; a major advantage of a graphic equalizer is to be able to see at a glance what the instrument is doing. The general shape of the filter response is apparent immediately from the positions of the slide controls. This information, together with the knowledge of the spectral distribution of the program material, is very advantageous in arriving at a desired frequency response.

We can arbitrarily divide the audible frequency spectrum into four sections: bass, lower midrange, upper midrange, and high frequencies. When we are concerned with wide bandwidth program material, we should remember that each of these frequency bands is important for its contribution to a full and balanced sound. Lack of bass frequencies will make the sound thin, while too much will

make it boom. The lower midrange, which provides much of the rich and mellow texture to music, may make the program material sound muffled if too abundant. The upper midrange is necessary for clarity and articulation, but gives the program a "canned" quality if overused. The high frequencies in excess create a harsh sound, but if balanced, give the audio program a spaciousness which it would not otherwise have.

### EQ SYSTEMS AND LIMITERS

The use of a compressor or limiter in the audio chain is frequently dictated by the system's headroom restraint, the dynamic range of program material, or aesthetic reasons. The question then arises, "Do I put the limiter before or after the equalizer?" The answer is that generally the equalizer should be ahead of the limiter in the chain.

The equalizer, by its very nature, changes the frequency response and harmonic relationship of the input signal. Frequencies that were greatest in amplitude at the input may not be so at the output. Conversely, some frequencies may be at a high level at the output due to the equalizer's action. The limiter, which is actuated by the level of the signal, should see the equalized or balanced signal.

The use of the equalizer prior to the limiter offers a further advantage, i.e., the program material may be equalized to optimize the limiter output by adjusting the input signal to the limiter. As an example, if a recording has a large dynamic range and an excess of low frequencies, the equalizer would be able to reduce the level of the low frequencies fed to the limiter, thereby reducing the amount of limiting needed. It also enables the user to select a faster attack and release time on the limiter if needed for peak limiting. The excessive low frequencies would have restricted the speed at which the limiter could operate and still achieve the desired low distortion.

There are a few instances where the equalizer might be placed after the limiter. This might be in a system where extreme uncontrollable dynamics are expected. The high input levels might otherwise overload the equalizer. However, these cases are rare, and the exceptionally wide dynamic range of the 5547A and 5549A makes this situation even more unlikely.

**BEFORE EQUALIZATION**

Equalization is only one of the solutions to the problems faced by designers or users of audio systems. Applied correctly, it is a powerful tool; but if it is assumed to be a cure-all without careful consideration of other possible alternatives, results may be disappointing. If the sound system is not thoroughly checked out before equalization procedures are attempted, much time may be wasted. Therefore, we feel that it is important to mention a few of the tests which should be made prior to the start of corrective room equalization. The list is not intended to be exhaustive, rather to point out that unless you know what the sound system IS doing, you may find it difficult (if not impossible) to make it do what it SHOULD do.

**A PARTIAL CHECK-LIST OF PROCEDURES BEFORE EQUALIZATION:**

1. At normal operating levels, check the frequency response and the signal-to-noise and distortion characteristics of the individual components of the system and the system as a whole.
2. Check polarity of signal lines from all inputs through to all outputs.
3. Check loudspeakers to see that all units function, that there are no rattles due to loose screws or units poorly attached to walls. Check phasing of all loudspeakers, even of the several drivers within one cabinet. Check loudspeaker aiming.
4. Check phasing of all microphones and microphone cables.
5. Check that all electronic components of the system are connected correctly, that correct signal levels are observed, that loads are where they should be, and not where they should not be. Check the system for oscillations, R.F. interference, and hum.
6. Make sure you have a good system ground, and that shielding and grounding are done in accordance with good practices.
7. Make sure that the AC electrical power to the system is of good quality, adequately protected, and that the wire size is sufficiently large to avoid excessive voltage drop when large amounts of power are drawn.

## LOW CUT AND HIGH CUT FILTERS AFTER EQUALIZATION

Examination of the filter's frequency response characteristic after equalization adjustments have been made may show that the band ends (low and high frequency extremes) could overdrive the amplifier or speakers (or the recorder in a studio situation) if the program material contains energy at these frequencies, i.e. microphone pops, etc. Adjust the tunable low cut and high cut filters until the resulting system response curve loses its "bathtub" shape and the house curve rolls off smoothly at each end.

## HIGH CUT FILTER CHARACTERISTIC, 12 dB or 6 dB/OCTAVE

This switch-selectable feature was added to change the high frequency rolloff quickly without readjusting the individual 1/3-octave controls. The 12 dB/octave setting provides a rather sharp cutoff point, while the initial attenuation of the 6 dB/octave filter is more gradual. This is very useful in situations where frequent changes of high frequency response are required, as in control and monitor rooms, to adjust to different program material or producers' taste. The filter skirts intersect at the continuously variable -3 dB point.

## DOCUMENTATION

The very nature of a graphic equalizer permits an almost infinite number of control settings. For later duplication of a certain sound or a corrective response curve it is necessary to document the position of the controls (Frequency and End Cut). This is made convenient by the graphic layout of the Equalizer front panel and its calibrated markings.

## CREATIVE EQUALIZATION

### GENERAL

The one-third octave graphic equalizer is obviously one of the many powerful tools in the repertoire of the creative audio mixer and its use is fairly well known. It offers the ability to shape the audio spectrum in an almost unlimited way while simultaneously presenting a front panel display of control settings that makes it very easy to understand at a glance what has been done to the EQ (and what yet needs to be done). The goal is the achievement of a

sound that is not only balanced in tone, but remains musical. For creative tasks we recommend the Model 5547A Graphic Equalizer. With a Boost/Cut range of  $\pm 12$  dB, 30 Bands and separate End Cut filters it incorporates both the high performance standards and the high degree of control flexibility demanded of a "studio quality" audio product.

### ADDITIONAL HINTS

The Model 5547A Graphic Equalizer is a powerful tool for the audio engineer to modify, adjust, alter, enhance, destroy or reclaim the audio signal. The extreme result would be a sound that never existed before. In such instances, there are very few rules for what to do and what not to do with the equalizer controls...in creative equalization whatever works is right.

If this is your initial use of a graphic equalizer, we strongly suggest that you experiment with the effect that various controls have on the sound of your program material. It is good to begin the work in a controlled environment where there is enough time to experiment, and the freedom to do it "wrong" without offending someone else. Doing it the wrong way means to know what does not work! Start out with simple program material of various types and try to make it sound "better." Different program material will require different control settings.

Extreme filter settings are generally to be avoided. Of course special effects and corrective equalization of program material which must be saved may require such settings. The important thing is to avoid correcting one problem while creating another.

Whenever a big notch in the frequency response of the filter is selected, it can cause important parts of the program material to be removed. It is possible to make adjacent notes on the musical scale differ considerably in level. If that's what is desired, fine. However, if you are trying to correct for problems which would better be dealt with by choice of microphone, microphone positioning, acoustic control, etc., you may run into trouble. The one-third octave graphic equalizer (or any signal processing device) may be a lifesaver in one context and a dead weight in another.

### OBJECTIVE MEASUREMENTS VS. SUBJECTIVE HEARING

Since the hearing process is subjective, each individual may have

a different opinion of what constitutes "good sound" or "bad sound". Until science has found all the answers about the phenomenon of hearing, we will have to use the available tools to modify sound with a certain imprecision. However, this lack of ability to quantify the relationship which exists between the subjective response of the listener, and the objective measurements of amplitude-vs-frequency response, phase, distortion, etc., need not be a barrier to our efforts to "improve" the sounds that we hear.

We have a most sophisticated quality assessment mechanism which is specifically suited to do this job...the human ear. Therefore, the educated listener with good hearing will probably always be the standard of reference to decide if a sound has been accurately reproduced. However, accurate faithful reproduction is not always the goal, and in many circumstances it may not even be possible or desirable.

### INPUT LEVEL

If the equalization controls are in the flat position or attenuated, no special care is necessary with regard to input signal level other than keeping it below +20 dB (Ref. 0.775 V rms). However, if any of the controls are adjusted for boost, then it must be remembered that at the frequencies affected by that control the gain of the amplifier is greater than unity, and the maximum input signal level which is allowable at those frequencies is reduced by the amount of boost at those frequencies.

For example, if the 400 Hz control is set for 5 dB of boost, an input signal level of +15 dB at 400 Hz would result in an output signal level of +20 dB. Because +20 dB is the maximum capability of the filter amplifier, the maximum input level permissible at 400 Hz without overload has become +15 dB. At moderate amounts of boost, this should not present a problem, but if, for example, all controls are in full boost, some frequencies will be boosted as much as 15 to 16 dB due to the combining effect of adjacent filters. This would bring the maximum input level down to +4 or +5 dB, and under such circumstances, the input level should be reduced to a safe level to avoid overload.

## **CORRECTIVE EQUALIZATION**

### **GENERAL**

Corrective equalization of audio systems is used to compensate for response anomalies occurring in the system and its environment, and to adjust the system response curve to some desired shape. In some systems this means adjusting for best overall flatness of response through the entire audio range, and in other instances some different response shape is desirable.

Three important characteristics of audio systems may be improved by judicious application of corrective equalization techniques. They are:

#### **1. DISTORTION OF FREQUENCY RESPONSE BALANCE**

Sources of sound which may be amplified through a sound system start out with some specific frequency relationships. If in being processed through an audio system this balance is disturbed by variation in frequency response, then truly the sound has been distorted. For example, a clarinet has a specific characteristic with regard to the levels of various harmonics compared to the fundamental tone. If these signals are processed through an audio system with non-linear frequency response, then the resultant sound is simply not the distinctive sound which that clarinet originally produced.

#### **2. SUSCEPTIBILITY TO FEEDBACK**

In public address and sound reinforcement systems, the maximum acoustic gain that may be obtained for a microphone in the vicinity of a loudspeaker which is part of that system, will be determined by the positive feedback loop created when sound from the loudspeaker enters the microphone, reinforcing the signal level until the system goes into oscillation. This positive feedback problem is aggravated if some frequencies in the audio range are reproduced at a higher level than the rest of the frequency band. If the level of these frequencies which are being produced at an exaggerated level can be reduced, then the sound system gain may be increased to some degree without feedback.

### 3. INTELLIGIBILITY

For an audience in an auditorium to understand a lecturer, or for a nightclub audience to hear the words which a vocalist is singing, there are two important requirements:

a. The sound level of the source must be sufficiently loud as to be heard clearly and not masked by other sounds which may be present in the listening environment.

b. The sound which the listener hears must be intelligible to him. It should not be distorted or garbled. There are a number of causes for poor intelligibility in a sound system. Among them are: amplifier or speaker distortion, frequency band masking, improper balance of direct and reverberant fields, and inadequate projection of the higher frequencies which are necessary for the recognition of words. The frequency band from approximately 1 kHz to 5 kHz is very important to the recognition of sibilant and consonant sounds. If the sound system has inadequate output in this frequency band, or if other bands have considerably greater level which "cover up" the sounds in the intelligibility band, the words may simply not be understood. Equalization is very helpful to correct some of these problems.

#### SOME PROBLEMS WITH CORRECTIVE ROOM EQ

As stated earlier, the process of corrective equalization of sound systems is not a cure-all. There are some rooms which, because of very long reverberation times and poor acoustic configuration, cannot be significantly improved. There are a few others in which the acoustics and the sound system are already so well balanced that there is little need for correction. It is for the majority of situations which lie between unsolvable and relatively perfect that corrective equalization is intended.

Many of the problems that people have when performing corrective equalization stem from a tendency to "overkill." It has been found in practice that it is not desirable to attempt to equalize to the "last dB" all minor anomalies in the response of an acoustical system, as this often results in an artificial or processed sound. Also, it sometimes requires extreme filter settings with introduction of unwanted phase shift, which may cause more problems than the



EQ solves. In general, if the final results are within a 3 dB envelope of the desired theoretical values, the equalization should be considered successful.

Attention should also be given to the fact that different frequencies are attenuated differently when traveling through air. As the distance between the sound source and the listening position (or position of the measuring microphone) increases, it is natural for high frequencies to be attenuated more. Therefore, if equalization is adjusted to create a perfectly "flat" response, particularly throughout a large room, the resulting sound will be excessively bright. A house curve which produces a natural response in sound systems is subjective and a matter of personal preference, as well as a function of the primary use of the location. However, a flat response up to about 3 kHz and then an attenuation of 3 dB per octave above 3 kHz is frequently used in sound reinforcement. In recording studios and monitoring rooms, the response is usually flat to higher frequencies.

Widely disparate settings of adjacent controls should be avoided if at all possible. A control panel in which adjacent adjustment knobs are 8 to 10 dB apart may look interesting, but the resultant sound will not be natural. It is good to remember that the trained human ear is still the ultimate judge of any equalization.

The remainder of this section is concerned with a general description of the corrective equalization process and some suggestions for successful application of the equalizer are given. Techniques are continuously being improved upon. Therefore, this discussion should be considered only as a starting point for those who are new to the field, and as a general review for those who are already proficient in the methods used.

Those interested in more complete information on the general subject of sound system design and corrective equalization are encouraged to study specialized literature\* and/or participate in seminars dedicated to audio acoustics, etc.

An excellent series of continuing seminars are conducted in various cities by Don and Carolyn Davis. For further information, and a schedule of their regular seminars, contact:

Synergetic Audio Concepts  
P.O. Box 1239  
Bedford, Indiana 47421

Phone: 812-275-3853

**\*SOUND SYSTEM ENGINEERING, by Don and Carolyn Davis, Howard W. Sams, Inc., Indianapolis, Indiana 46268, © 1975.**

### **BOOST-CUT VS CUT ONLY**

In one form or another, equalization of sound systems has been around since the thirties. However, it was not until Dr. Paul Boner's work in the early sixties that sound system equalization came of age. Later in that decade Altec introduced the hardware that began to make equalization a common practice among sound contractors. These early devices were passive losser networks capable of Cut-only action. As active devices became available they initially imitated the Cut-only action of the passive units. But soon manufacturers produced units which could Boost as well as Cut. The question then arose: "Which do I use - Boost or Cut?"

The answer is that for room equalization Cut is best, but that with a knowledge of the limitations in its use, some small amount of Boost may be acceptable. There are several reasons for the choice of Cut-only EQ. First, of course, is the realization that when performing the equalization there is a good deal of difficulty in determining which frequency response anomalies are amenable to correction. The natural tendency is to "fill in the holes" of the frequency response with Boost equalization. Unfortunately, if the cause of the dip is not amenable to correction, then no amount of Boost EQ will help. In addition, the effect of Boost EQ is more easily heard by the ear and sounds less natural than an equalization curve arrived at by Cut EQ. One individual has described the effect of Boost equalization as similar to looking out over an empty field dotted with telephone poles. Cut EQ is then similar to the same open field except that the poles have been replaced by telephone pole size/shape holes. The poles are clearly visible, but the holes are not! The analogy is sound. The ear is much more sensitive to the effects of Boost EQ than to the effects of Cut EQ.

To some extent Boost equalization may be used to smooth the amplitude response. The amount that may be used is determined by several factors including the type of system, the type of program material, the frequency at which the Boost EQ would be used and the sensitivity of the listeners to the effects of the equalization. Generally, more critical program material, better listening environments, and more sensitive listeners are less able to tolerate Boost equalization. In addition, the effects of Boost equalization are more easily heard (to their detriment), when the frequencies being

boosted are in the middle of the audio frequency band. The effect of excess Boost will be heard as artificial and the filters may ring. The Model 5549A Room Equalizer is recommended for the correction of room response anomalies. With 15 dB of Cut available at each of 30 bands of one-third octave equalization and separate End Cut filters it provides the range of control necessary to deal with the wide range of situations found in both fixed and portable sound systems of all types.

### MEASUREMENT TECHNIQUES

Several different methods are in use to measure the frequency response characteristics of a sound system in its acoustic environment:

1. Pink noise, measured in 1/3-octave bands and displayed on a real time spectrum analyzer or suitable LED-display.
2. Swept sine waves, measured on an audio voltmeter or plotted on a graphic recorder.
3. Swept sine waves, frequency modulated with a constant percentage bandwidth (Warble Tone) generated and plotted on a chart or strip recorder.
4. Time delay spectrometry (TDS) is a more recent development with the capability of analyzing not only frequency versus amplitude, but also with respect to time.

Basically, all of the measurement methods are similar in technique -- differing only in hardware. They all involve generating a known quantity of audio signal, transmitting it through the sound system, and reading the results through a calibrated microphone and readout device. The results will closely resemble each other if comparative measurements are made, presuming correct application of the various methods.

The microphone is located in the listening area, and its position is changed to see if there are response variations in different locations. The response variations are then averaged together to make what is called the RAW HOUSE CURVE. Corrective equalization procedures are based on this curve. Many problems associated with level balance between multiple speakers in large areas, acoustic deficiencies, and speaker aiming and locating faults will show up on the curves taken in various locations in the listening area.



### RAW HOUSE CURVE

For good results, these faults should be corrected by re-balancing of levels and re-aiming or relocating speakers -- not by equalization. In systems which use electronic crossovers or multi-amplifier configurations, the balance should be checked and adjusted between low and high frequencies and between speakers covering different parts of the listening area.

### ADJUSTMENT OF CORRECTIVE EQUALIZERS

Room equalization is performed as a series of approximations of control settings. Looking at the sample RAW HOUSE CURVE we can infer that if an inverse electrical response curve were placed in the audio chain that the result would then be a flat frequency response.

### OCTAVE OR 1/3-OCTAVE EQUALIZERS?

The octave graphic or 1/3-octave graphic equalizers are tools with

which this inverse curve is approximated. Because there are combining effects between adjacent filters, it is necessary to make settings on the equalizer and then to re-check the room response to see if the adjustments were correct. Additional adjustments and measurements may be made until the desired results are obtained.

It is important to understand the difference between these equalization filters. If the device contains filters which are one octave wide (such as the single channel UREI Model 533 or the dual channel Model 535), the user is able to adjust for broad spectrum anomalies, which in some applications is absolutely sufficient. However, the 1/3-octave equalizer, with its 30 controls plus end cut filters, enables much more precise adjustment to correct the frequency response in the system.

### FEEDBACK SUPPRESSION

An improvement of a system's tendency to feedback should be attempted after the general equalization of the frequency response is performed. SLOWLY advance the gain until the feedback frequency becomes detectable and stabilize the feedback to a constant, comfortable level. (Caution: feedback is not only annoying to the ear, but it is also dangerous to unprotected amplifiers and loudspeakers!). The control which reduces the feedback with a minimum of additional attenuation is closest to the actual feedback frequency. Experimentation by trial and error, and the resulting experience, is the best method.

Advancing the gain further will cause the next feedback frequency to occur. This procedure may be repeated as often as it seems practical. Again, extreme filter settings should be avoided.

The result will be a higher amplifier gain setting than was possible before feedback suppression. Remember: an improvement of 3 dB is equal to twice the previously available power. To prevent ringing, it is best to adjust the gain at least 3 dB below the threshold of feedback.

NOTE: Where the budget allows the expense, we suggest using a dedicated instrument such as the UREI Model 562 Feedback Suppressor. It is specifically designed to "tune-out" a very narrow band of frequencies with continuously variable filters without noticeably altering the program material.

## SECTION IV

### MAINTENANCE

The Models 5547A and 5549A are all solid state, ruggedly constructed and use the finest components. As such they should provide years of trouble free use with normal care. All parts are conservatively rated for their application.

**NO SPECIAL PREVENTIVE MAINTENANCE IS REQUIRED, AND THERE ARE NO INTERNAL SERVICE ADJUSTMENTS.**

The metal and plastic surfaces of the equalizer may be cleaned with a damp cloth. In case of heavy dirt, a non-abrasive household cleaner such as Formula 409 or Fantastik may be used. **DO NOT SPRAY THE CLEANER DIRECTLY ONTO THE FRONT OF THE EQUALIZER AS IT MAY DESTROY THE LUBRICANTS USED IN THE SWITCHES AND CONTROLS!** Spray onto a cloth and then use the cloth to clean the equalizer.

### REPAIRS AND WARRANTY

This product is warranted by the manufacturer to the original purchaser against defects in material and workmanship for a period of one year from the date of purchase. We suggest that you retain a copy of your dated sales receipt for proof of warranty status should that be necessary.

If you wish to return the unit directly to the factory, please call or write to the Customer Service Department at the Service address listed on the title page of this manual for a Return Authorization Number. All products returned to the factory must be accompanied by a Return Authorization Number, and must be shipped prepaid. COD shipments will not be accepted.

For prompt service, ship the unit to the factory with the RA number marked on the shipping label. Be sure that it is well packed in a sturdy carton, with shock absorbing material such as styrofoam pellets or "bubble-pack" surrounding the unit. Pay particular attention to protecting the controls and switches and make sure that the unit cannot drift around in the shipping box. Shipping damage caused by inadequate packing is not covered by the JBL/UREI warranty. Tape a note to the top of the unit describing the

problem, include your name and a phone number where we may contact you if necessary, and give us instructions for returning the product. We will pay return shipping costs on any repair covered under the terms of this warranty.

Field repairs are not normally authorized during the warranty period, and repair attempts by unqualified personnel may invalidate the warranty.

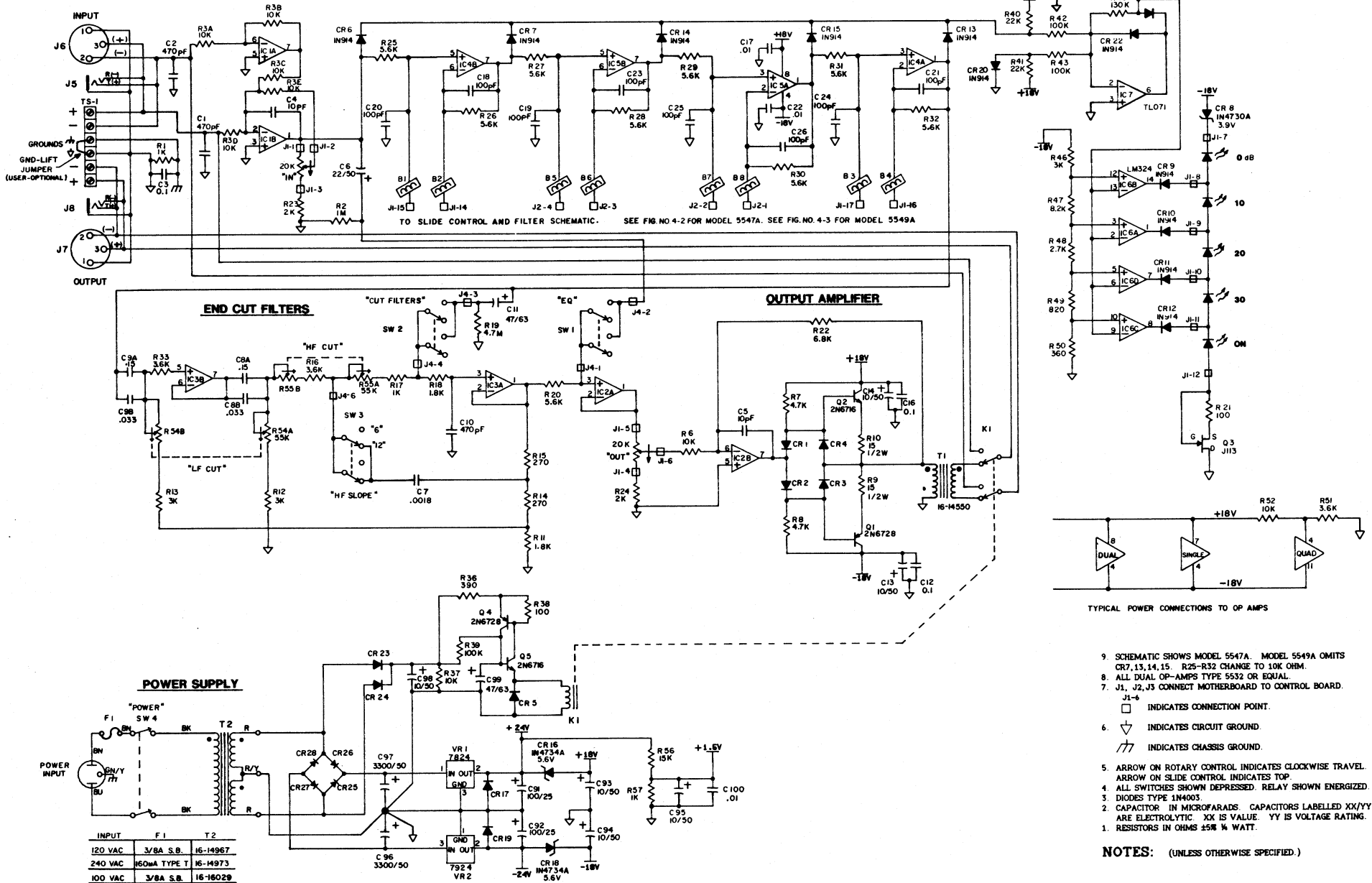
### THE MODELS 5547 AND 5549

This manual was written to cover the JBL/UREI Models 5547A and 5549A. They were preceded by the Models 5547 and 5549. The earlier units had an external calculator-style power transformer which connected to the equalizer chassis by a 7 pin DIN connector. In all operational and performance characteristics the older units are otherwise equal and the instructions in this manual are applicable.

**INPUT AMPLIFIER**

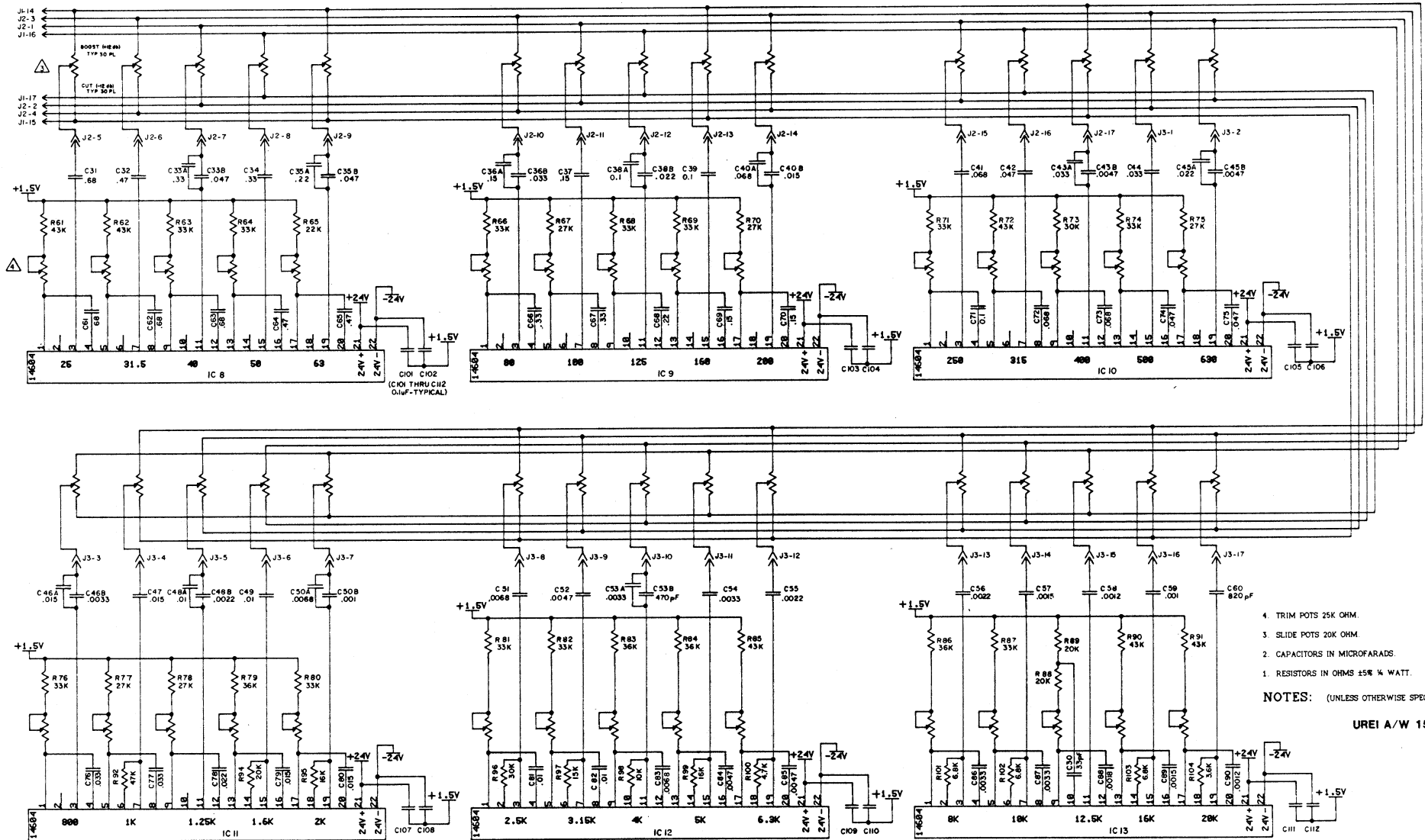
**SUMMING AMPLIFIERS**

**HEADROOM DETECTOR / DISPLAY**





MODEL 5547A SLIDE CONTROLS AND FILTERS



- 4. TRIM POTS 25K OHM.
  - 3. SLIDE POTS 20K OHM.
  - 2. CAPACITORS IN MICROFARADS.
  - 1. RESISTORS IN OHMS ±5% ¼ WATT.
- NOTES: (UNLESS OTHERWISE SPECIFIED)

UREI A/W 15361

Figure 4-2

