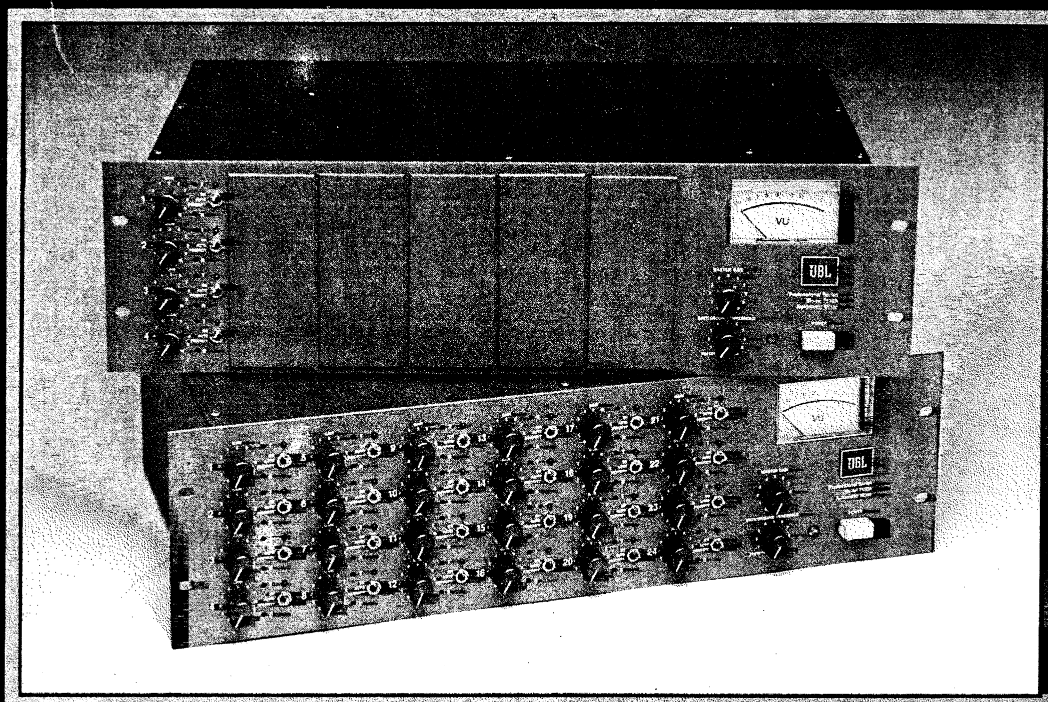
7510A AUTOMATIC MICROPHONE MIXER



PROFESSIONAL SERIES
INSTALLATION & TECHNICAL GUIDE

JBL

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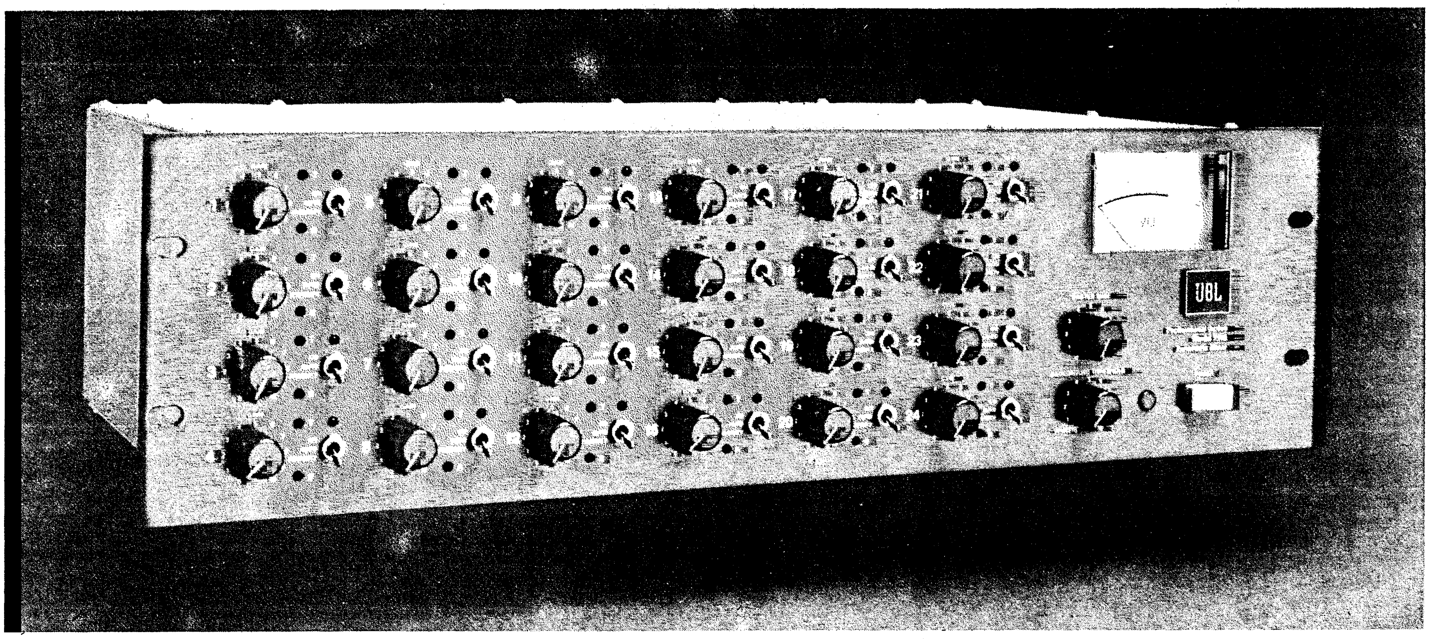


Figure 1. JBL Model 7510A Automatic Microphone Mixer.

1 INTRODUCTION

The JBL Model 7510A is a compact microphone mixer with a built-in "brain" whose unique capabilities are described below. The mixer is a modular, rack-mountable package with space for 6 plug-in cards, each card containing full electronics, controls and connections for 4 balanced microphone inputs. The front panel is fully labeled for the maximum of 24 input channels, although the 7510A comes standard with 4 inputs installed. The mainframe is fully wired, and unused input positions are covered by blank panels, making it easy to install additional 4-input cards at any time. The monophonic output section includes, among other items, a large VU meter, a Master Gain control, and a balanced +4 dBm at 0 VU output (Ref. 600 ohm) with an XL-type connector.

The 7510A "brain" is actually a combination of special digital and analog circuitry. A level sensor instantaneously switches On a channel when its microphone is in active use. The same circuits turn Off the channel when its mic is not in use. Individual channel mix levels are set manually. The mixer's "brain" tells the output how many mics are "live" (active), and the output then automatically adjusts itself to maintain essentially constant system level regardless of the number of live mics.

The "smart" circuits in the 7510A offer several powerful advantages over conventional mic mixers. For example, the automatic mic turn-Off/turn-On and output level correction together provide considerably more gain without danger of feedback (howling). No longer must an operator work feverishly to constantly adjust the controls (ride gain); the 7510A does it automatically and with greater speed and precision.

To ensure that you hear every word and musical note without chopping the beginning or end of a phrase, the 7510A's level sensing circuitry turns On extremely fast and has an adjustable release rate. In fact, the inputs respond quickly enough to capture the leading edge of a musical transient such as the beat of a loud drum. This fast attack time, in conjunction with the direct output jack on each channel, makes the 7510A an excellent device for use as a multi-channel noise gate in recording studio, broadcast, theatrical and concert sound reinforcement applications. By switching On only the active mic channels on a moment-to-moment basis, the 7510A not only eliminates a lot of extraneous background noise and leakage, it also reduces undesirable "off mic" sound which often plagues multi-microphone live performances.

The 7510A inputs can be set for any of three operating modes:

1) AUTOMATIC, where each channel turns On and Off automatically, and the output gain self-adjusts accordingly, as previously described.
2) PRIORITY, where using the microphone on any input(s) switched to Priority mode will mute all other inputs that are set to Automatic mode.
3) ON, where the input functions like a conventional mixer (i.e., it remains On at the set level whether or not the mic is being used).

In all three input modes, a unique digital attenuator automatically reduces the output gain such that the level drops 3 dB each time the number of active microphones is doubled. (Refer to Figure 16, *page* 18.) This prevents feedback from commencing as more and more mics are activated.

One key to the Model 7510A's success is the way it "knows" when a mic is active, and the way it turns On. This is accomplished by digital logic and control circuitry which compare those sounds present at all microphones (the system considers this to be background noise) to those sounds which are predominantly present at a specific microphone (the system considers this to be program material). The threshold for discrimination between background noise and program is adjustable, and once the "decision" is made that program is present, the channel turns On (so long as it is set to the appropriate mode). The turn-On itself is completely inaudible thanks to a zero-crossing detector and an ultra-fast rise time that eliminates clicks and pops.

Other features of the 7510A include built-in 48 V DC phantom power for condenser microphones, which can be switched Off for critical testing but otherwise may be left On at all times regardless of the type of mic in use. There is also an auxiliary input which can be used to return the sound from an external signal processor such as an equalizer or reverb; the "send" (signal source) to the processor can be taken from one or more of the 7510A's mic channels via their Direct Output jacks. An LED on each channel lets the operator know when the channel is active (switched On manually or automatically). A large VU meter provides an accurate indication of the mixer's output level.

Provision is made in the 7510A for DC logic output switches for each microphone input. These outputs can be used for local loudspeaker muting in complex installations, for actuating associated equipment, such as logging recorders, and being able to turn on TV cameras/telecommunications systems.

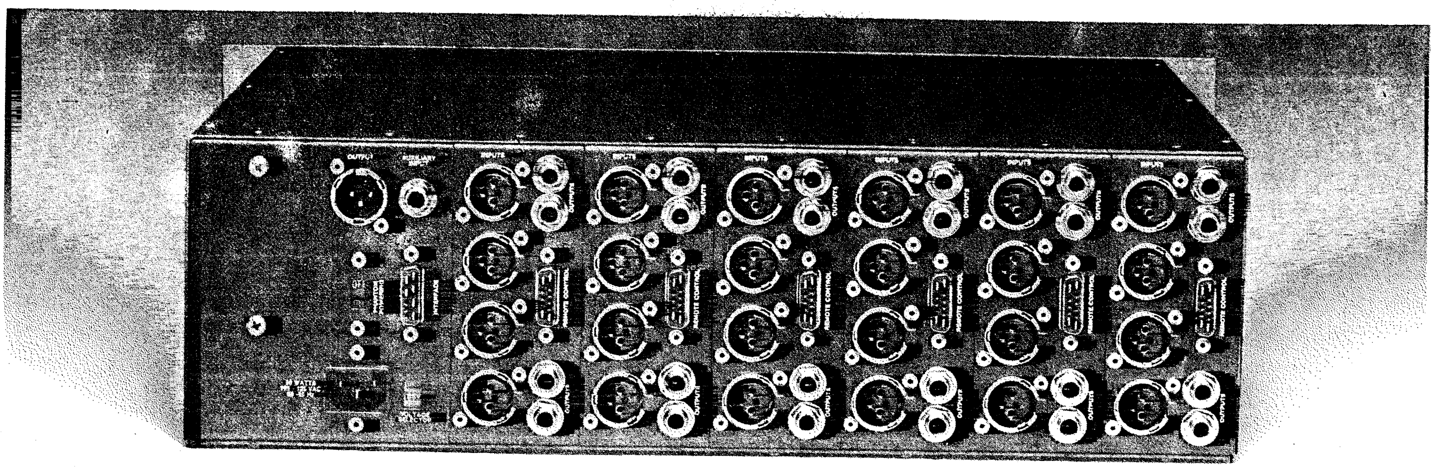


Figure 2. 7510A Rear Panel.

Multiple 7510A's can be coupled together for very large installations. In the interconnected mode, a group of 7510A's will behave as a single, larger unit.

The JBL 7510A is an ideal choice for the main mixer in board rooms, city/county council chambers, courtrooms, houses of worship, restaurants or small clubs, and other similar installations. In addition, the 7510A is an excellent tool for the sophisticated recording studio, theater, concert sound reinforcement company, or broadcaster.

HOW TO USE THIS MANUAL

If your Model 7510A is already installed, see *page* 4 for a review of all operating controls, and *page* 7 for information on how to adjust the controls.

2
BRIEF INSTALLATION AND OPERATING INSTRUCTIONS

REAR PANEL FEATURES

Input Modules

NOTE: The mic input and direct output connectors for 4 channels are housed on one input module. A blank aluminum rear panel occupies each vacant module position when the 7510A is equipped with less than a full complement of 6 modules (24 inputs).

Microphone Inputs

Virtually any low impedance professional microphone may be used, dynamic or condenser. The end of the microphone cable that mates with the 7510A must be equipped with an XL-type connector (3-pin male).

The 7510A Microphone Input connectors are 3 pin female XL-type. These electronically balanced inputs have a nominal input level adjustable from 2.5 mV (−50 dB) to 0.7 v (−1 dB)*. The input impedance is greater than 10K ohm, and the inputs may be used with source impedances from 0 ohm to 600 ohm. The input overload point is adjustable from 45 mV (−25 dB) to 6.13 v (+18 dB). Since the 48 V DC phantom power is applied across both signal carrying pins (2 and 3) via build-out resistors, the phantom power can remain ON when dynamic mics are used. An external phantom power switch is provided primarily so that the DC voltage will not interfere with signal generators or other critical test equipment when bench testing the mixer.

Direct Outputs

Each input has a multi-function direct output 6.3 mm (1/4 in.) phone jack, providing a normal through connection between the booster amplifier output and the summing buses. Insertion of a 6.3 mm (1/4 in.) 3-conductor phone plug into the jack breaks the normal through connection, and provides both a normal +4 dB unbalanced direct output and a normal unbalanced input to the summing buses, suitable for feeding a signal to and receiving a signal from an external signal processor. The signal at this jack is gated on and off by the channel's automatic turn-ON/turn-OFF circuitry. The Direct Output thus can be used to feed a larger mixing console in applications where the 7510A is used as a multi-channel noise gate, or where special signal processing of the microphone input is desired. The channel's Gain control also adjusts the level at this jack. When a cable is plugged into the Direct Output jack, the channel's sound no longer contributes to the 7510A's Main Output (although the digital circuitry continues to count it as an active microphone when determining the output gain). The signal can be fed to the line level input of any professional sound device, either low or high impedance.

The Direct Output is a 6.3 mm (1/4 in) 3-conductor phone jack, connected after the channel Gain control and the booster amplifier. If desired, the input channel signal can flow directly to the 7510A's Main Output and can also provide a Direct Output by connecting the tip and ring together on the 1/4" phone jack. The Direct Outputs can provide maximum output levels of +14 dBm (600 ohm load) or +20 dBv (10K ohm load).

Remote Control Connectors

Each group of four inputs has an associated 15-pin connector for providing external gain adjustment of 49 dB for the preamplifier stage of each of the four inputs. Proper setting of the gain provides the best combination of headroom versus noise between the low output of a dynamic microphone, the medium output of a condenser microphone, or the high output of a tape recorder or other line level device. The remote control connectors also provides TTL level on/off logic for each of the four channels. This on/off logic provides a suitable means for switching loudspeakers on or off or switching other external equipment, all by activating an associated input channel.

*NOTE: *Unless otherwise specified, 0dB is equivalent to 0.778 volts throughout this manual.*

The 15 pin connector is an AMP HD-20 female socket, and its mating connector is the AMP 205735-6, TRW DA-15P or equivalent. The unit is shipped with the appropriate pins strapped so that each input is set for high gain. Instructions for implementing other gain values, as well as implementing the logic outputs, are given in Section 4, DETAILED INSTALLATION INFORMATION.

Output Section

Auxiliary Input

This standard phone jack is intended primarily for use in conjunction with one or more of the Direct Outputs. Using the appropriate external equipment, sound from the Direct Output jack(s) can be equalized, limited, phase shifted, delayed, etc., then returned to the Aux Input so it can feed the 7510A's Main Output. The advantage to this arrangement, as opposed to feeding the amplifiers directly from the external signal processor or external mixer, is that the 7510's "brain" continues to adjust the overall system gain in accordance with the number of active microphones.

The Auxiliary Input is not generally recommended for injection of background music, although it can be used for this purpose. In this case, at least one mic input must be ON or the automatic gain control circuitry will drop the level of the Aux Input signal before it reaches the Main Output. To ensure the Aux Input will remain "live," turn down the Gain control of any channel and lock that channel ON manually, using its front panel Auto/Priority/On switch.

The Auxiliary Input is a 6.3 mm (1/4 in) 3-conductor phone jack. While the jack has three conductors, it is unbalanced; the tip carries the signal, the ring is audio ground, and the sleeve is chassis ground, providing additional shielding against noise. For optimum rejection of noise and elimination of ground loops, use a cord made with a dual conductor shielded cable and a stereo phone plug, wired as shown in Figure 5, *page* 9. It is permissible to use a cord consisting of a standard phone plug with single conductor shielded cable.

The Auxiliary Input has a 10K ohm load impedance, with a nominal input level of +4 dB. It applies signal, via an isolation resistor, to the audio summing network, where it is mixed with those input signals which have not been tapped at their direct outputs. The combined signals are then fed to the digital attenuator, Master Gain Control, and output amplifier.

Main Output

The Main Output is a 3 pin male XL-type connector carrying a monophonic mix of all active inputs plus an Auxiliary Input signal. The output may be connected directly to the input power amplifier or to a line input of another mixing console if the 7510A is being used as a submixer. The 7510A Master Gain control adjusts the overall Main Output level. Since the output is line level and low impedance, it can be fed to the line level input of any professional sound equipment, either low or high impedance. It is also capable of driving long cables without degradation of the sound.

The main output is symmetrical and floating, and has a nominal +4 dBm (ref. 600 ohm) level. Minimum rated output capability is +24 dBm. The actual output impedence is less than 0.5 ohm, and the output will drive 600 ohm or higher impedance circuits without special termination.

Interface Connector

Multiple 7510A units may be coupled together to function as a single unit by connecting them together via the appropriate 15-pin connector.

The 15-pin connector is the AMP HD-20, and the mating connector is the AMP 205735-6, TRW DA-15 P or equivalent. Implementation of this feature is discussed in Section 4, DETAILED INSTALLATION INFORMATION.

Fuse

The 7510A is externally switchable between 115 volts AC and 230 volts AC, 50/60 Hz. A 1.5 ampere 3 AG fuse protects the power supply and is mounted internally.

CAUTION: In the event of repeated fuse failure, check to see that the mains are within 10% of the normal voltage. If the mains are within tolerance have the 7510A checked by a qualified service technician. NEVER SUBSTITUTE A FUSE OF HIGHER VALUE.

FRONT PANEL FEATURES

Input Modules (Input Cards)

NOTE: Four channels of input signal processing are housed together in each input module. Each channel includes the following five front panel items: a Gain control, a Threshold trimmer, a Release Rate trimmer, a Mode switch, and a Channel-ON LED indicator. A blank cover panel with matching finish is installed over each vacant module position when the 7510A is equipped with less than a full complement of six modules (24 inputs).

Gain Control

This rotary control adjusts the relative amount of signal from the corresponding microphone to be mixed into the output. Lower the Gain setting if the sound is too loud or distorted, and raise the Gain setting if the input is too quiet or if it requires so high a setting of the Master Gain control that the noise level goes up.

The Gain Control is between the preamp and the level sensor. It affects not only the level fed to the audio mixing bus and output, but also the level at which the logic senses a signal is present. A minimum of 70 dB of Gain is available in the input module (from Microphone In to Direct Out).

Threshold Trimmer

This screwdriver-adjustable control sets the sensitivity of the circuitry which determines whether the microphone is active when the channel is in Auto or Priority mode. Rotating the Threshold control counterclockwise blocks the channel from turning On until louder sounds are present at the microphone. Ideally, the Threshold should be set low enough (turned clockwise) that the channel turns On whenever the microphone is being used, typical fully clockwise. Avoid too high a Threshold setting (i.e. too little sensitivity) as this prevents the channel from turning On as soon as it should (or from turning On at all in extreme instances of misadjustment.)

This potentiometer adjusts the Threshold of the channel's digital control circuit. Clockwise rotation lowers the threshold and counterclockwise rotation raises it. The circuitry "looks at" the signal after the Gain control.

If the Background Threshold control is turned maximum CCW until it clicks Off, only an individual channel's Threshold and Gain controls determine the sensed level at which it turns On (given that no Priority channel is on).

With the Background Threshold set to variable mode, the actual decision of a channel to turn On also depends on how much common background sound is present at *all* microphones (whether or not their channel is On). If the overall background sound (noise) level goes up, the input turn-On thresholds also go up. The background sound level is computed by the 7510A's background threshold control circuit, and is weighted or calibrated by the Background Threshold control. Thus, the input channel's set threshold becomes relative, riding up and down with the ambient sound level, rather than an absolute setting.

In addition, when the background threshold is properly set, any signal detected above the common signal at all inputs (approximately 3 dB) will activate that input.

The attack time is dependent on how soon a zero crossing occurs after the signal first exceeds the threshold level. The specified range of 10 milliseconds to 10 microseconds corresponds to a half cycle from 20 Hz to 20 kHz. Since the turn-On takes only 30 to 60 nanoseconds (30 to 60 trillionths of a second), it occurs within one half cycle of detection, and is completed while zero output voltage is present, and the whole process is totally inaudible.

Attack is so fast that the leading edge is not lost when a drummer strikes the snare drum with the drumstick. There is no audible difference in an

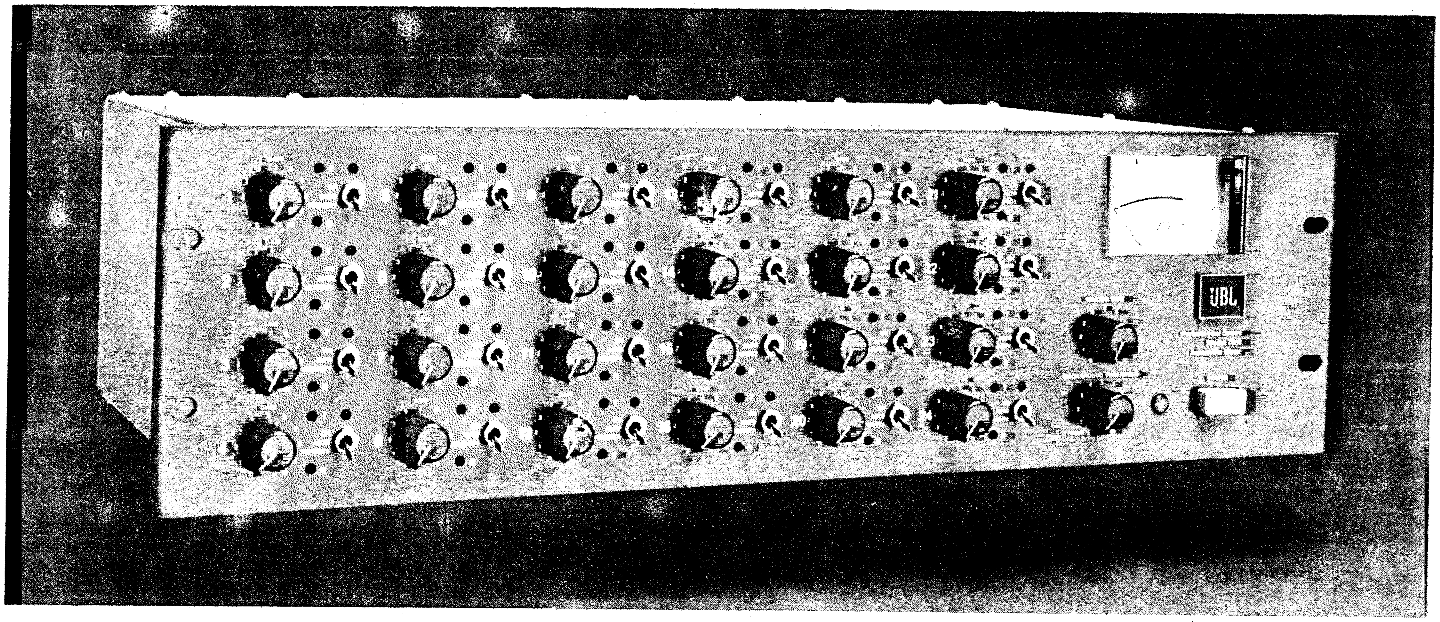


Figure 3. 7510A Front Panel.

A/B comparison between an input gating on and off, and an input being on all the time.

The preceding description assumes the channel is set to Auto or Priority mode and, if in Auto mode, that no other Priority-selected channels have an active microphone which would inhibit turn-On of those channels in Auto mode. If the channel is set to On mode, its Threshold setting is of no consequence, although the channel still contributes to the sensed background threshold.

Release Time Trimmer

This screwdriver-adjustable control sets the length of time a channel will stay On after the input level drops below the set threshold. Rotating the release trimmer clockwise lengthens the release time, preventing the channel from turning Off while a person takes a breath between sentences or a musician rests for one or more notes. Ideally, the Release Time should be set short enough that the channel turns Off immediately after the desired sound dies away. However, the Release Time must be sufficiently long that the unit does not cut off the end of words or notes; too short a Release Time will "chop up" the sound as the microphone turns On and Off excessively. During speech, the release time should be set for 2 to 3 seconds, so input does not shut off during short pauses.

The Release Time is adjustable from 100 milliseconds to 5 seconds. As with the Threshold, the Release setting has no effect if the channel is switched to On mode.

Mode Switch

This 3-position toggle switch selects any of three states for the input channel: On, Auto or Priority mode. Use On mode if the microphone is to be live at all times, as when setting up the basic mixing levels of the various channels. Use Auto mode if the microphone is to be live only when in use, and to automatically turn Off at other times. Use Priority mode if the microphone is to be On whenever it is in use, and it is to simultaneously turn Off any other microphone whose channels are in the Auto mode.

The Mode Switch, in On position, overrides the level sensing and digital control circuitry, converting the channel to a conventional mixing input. Auto position causes the channel to sense the microphone level, channel Threshold and Background Threshold information, and decide whether or not to turn On. Auto mode turn-On will be inhibited, however, if any other channel (or channels) is both switched to Priority mode and activated by its signal sensing/threshold circuitry.

The microphone preamp and Gain control always feed signal to the digital control circuitry. The signal flow is turned On and Off by the control circuitry at the channel's booster amplifier. The actual turn-On is precisely timed so that it occurs at a zero-crossing (when the positive or negative going AC signal waveform crosses the zero voltage point). While a sudden and rapid increase of gain might cause a click or pop in a conventional gating circuit, there is no switching noise in the 7510A because gain increase occurs during an instant when the unit is amplifying zero volts.

Channel Attenuation

When turning an input channel On and Off, The 7510A does not actually turn the input audio signal completely off. The audio in each input channel is attenuated 25.5 dB when that input channel switches "off". Should you wish to have an input channel attenuate less than 25.5 dB contact factory for modification information.

Channel On Indicator

This red Light Emitting Diode (LED) is illuminated whenever the channel's microphone is live. If the LED is On, the channel's microphone is contributing sound to the Direct or Main output.

In the On mode, the LED is always illuminated. In priority mode it is illuminated only when the sound at the microphone produces an above-threshold signal. In Auto mode the LED is illuminated in the presence of an above-threshold signal provided no Priority channel has an active microphone. If a priority channel is on, channels in Auto mode will be gated off and their LED indicators will be extinguished.

Output Section

Master Gain Control

The Master Gain control adjusts the overall program level at the Main Output connector. A suggested initial setting is #5 on the control scale. During a performance, the setting can be increased, but with danger of possible howling or ringing; the control can always be turned down for overall program fades.

The Master Gain Control is a fader located between the digital output attenuator and the output driver amplifier. A maximum of 7 dB of gain is available in the output section, providing an overall maximum gain of 77 dB minimum (600 ohm load) from Microphone Input to Main Output.

While the Master Gain sets the 7510A's output level, the level is reduced progressively by the digital output attenuator as more input channels turn On to avoid feedback at the rate of 3 dB for each doubling of active inputs. (Refer to Figure 14, Section 6). The automatic output gain adjustment is equally useful in recording or broadcast, where it aids in preventing the average level from exceeding the saturation or 100% modulation point as more microphones are used, yet does not change the program dynamics in the same manner as a compressor/limiter.

VU Meter

The VU Meter displays the signal level fed to the 7510A Main output connector. In general, the Input and Master Gain controls should be adjusted so the meter points to the red zone only occasionally; a continuous meter indication in the red zone will probably be accompanied by unacceptable distortion. Most power amplifiers can be driven to full output by the 7510A, when its meter displays 0 VU, +4 dBm (ref. 600 ohm), or perhaps a few dB above 0 VU. Most of the time the amplifier should operate from 5 to 20 dB below that level. Thus, if a constant "red zone" VU level seems to be necessary, we suggest increasing the gain of the power amplifier or other equipment in the sound system, or checking for a bad cable, improper hookup or other problem.

The VU meter is isolated so it does not load the line or induce any distortion due to the meter diodes. The meter is calibrated so 0 VU corresponds to an output level of +4 dBm (ref. 600 ohm).

Background Threshold

The Background Threshold control is part of the circuit which alters the set Threshold of all microphone inputs in accordance with the overall background sound level (i.e. that sound which is sensed in common at all microphones). The circuit allows individual Threshold controls to be set relatively low without danger of loud background sounds causing one or more channels to turn On. When the environment is quiet, a relatively low sound level at a given microphone will turn the microphone's channel On. A signal as low as − 100 dB at the channel's input will activate that channel in a quiet environment. When the environment becomes noisier, the same microphone will require more level before it turns On. The Background Threshold control is used to set the ratio of background noise to the input threshold.

Ideally, the Background Threshold control should be set high enough so that no microphones turn On in the presence of the loudest expected *background* sound. However, the control should be set no higher than necessary as this will decrease the microphone's sensitivity to automatic turn-On in the presence of desired *program* sound.

Turning the Background Threshold fully counterclockwise switches Off the background level sensors, so background sounds do not affect the individual channel thresholds. This is done when you don't want the channel thresholds to "float" on the background noise level, as in recording or broadcast applications.

The Background Threshold control and its associated circuitry do not depend on the level at any one microphone. Instead, all microphones continuously apply signal to the threshold send mixing buss, even if the individual microphone channel is not On (i.e. if it is below the channel's threshold or if it is inhibited by a priority function). The buss, in turn, presents this instantaneous sum of the background sounds to the background threshold control circuit. The set Background Threshold Ratio is then made available to all input channels via the Threshold Receive buss, so it can serve as a comparison reference. Each channel's individual Threshold control thus sets the relative level at that microphone, above the established overall background level, necessary for the microphone's channel to turn on.

Power Switch And Indicator

An alternate action pushbutton switch turns On the AC power to the 7510A. This power switch is on the secondary side of the power transformer. The fuse, voltage switch, and power transformer are all located between the power connector and the power switch. The adjacent green panel LED glows when the power is On, and the VU meter illuminates.

3
RECOMMENDED SETUP ADJUSTMENT PROCEDURE

1. Set all input channels to auto mode.
2. Set all input channels gain controls to "0" (Maximum counterclockwise rotation).
3. Set input channels 1 and 2 front panel release time control for the fastest response (maximum counterclockwise rotation).
4. Set input channels 1 and 2 front panel threshold control for maximum (clockwise rotation).
5. Set master gain control at "5".
6. Set background threshold on "1" (maximum counterclockwise rotation, but not switched to preset).
7. Feed a 1000 Hz, −50 dBv signal from an external generator to input channel 1 and adjust gain control until the VU meter reads "0" VU.
8. Disconnect the signal from input channel 1 and connect it to input channel 2 and adjust gain control until the VU meter reads as follows for the number of in-use active microphones.

1 to 8 inputs set at −3VU	29 to 32 inputs set at −9VU
9 to 12 inputs set at −4.8VU	33 to 36 inputs set at −9.5VU
13 to 16 inputs set at −6VU	37 to 40 inputs set at −10VU
17 to 20 inputs set at −7VU	41 to 44 inputs set at −10.4VU
21 to 24 inputs set at −7.8VU	45 to 48 inputs set at −10.8VU
25 to 28 inputs set at −8.5VU	

(In-use active microphones mean microphones connected to a 7510A input and their gain controls adjusted to a normal in-use setting.)

9. While simultaneously feeding the 1000 Hz, −50 dBv signal to both input channels 1 and 2 (gain control still set as in steps 7 and 8 for given number of active inputs) turn background threshold control clockwise until the following:

For: 1 to 4 active inputs—until both inputs shut off
5 to 8 active inputs—until −3VU input shuts off
9 to 12 active inputs—until −4.8VU input shuts off
13 to 16 active inputs—until −6VU input shuts off
17 to 20 active inputs—until −7VU input shuts off
21 to 24 active inputs—until −7.8VU input shuts off
25 to 28 active inputs—until −8.5VU input shuts off
29 to 32 active inputs—until −9VU input shuts off
33 to 36 active inputs—until −9.5VU input shuts off
37 to 40 active inputs—until −10VU input shuts off
41 to 44 active inputs—until −10.4VU input shuts off
45 to 48 active inputs—until −10.8VU input shuts off

10. Any microphones to be used should be connected at this time. If no microphone is connected to a given channel, set it to Auto Mode, and set its Gain control to minimum.
11. Switch On all input channels and adjust their Gain controls for the desired mix. If feedback (ringing or howling) begins, lower the input Gain setting slightly. To fine tune for absolutely maximum gain before feedback, switch On and adjust only one channel at a time. If external equalization is used to increase potential gain before feedback, it should be accomplished at this time.
12. After all input Gain controls have been set, make sure all inputs are in Auto mode.
13. The Threshold should be set low enough (turned clockwise) that the channel turns On whenever the microphone is being used, typical fully clockwise.
14. Set the Release Time for each channel. There is no "ideal" setting, but it is easy to hear the effect of changing the setting when you use typical program material at the microphone under adjustment. Generally,

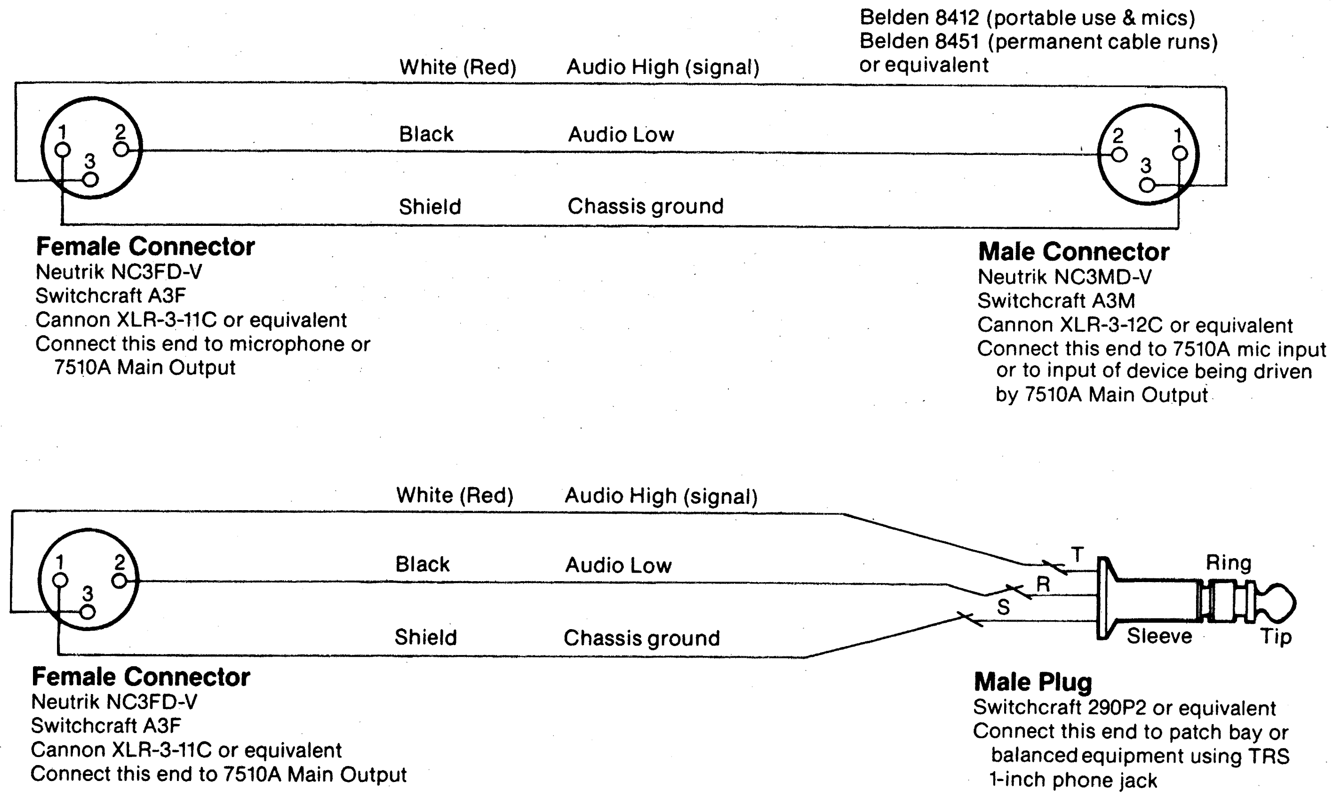


Figure 4. Cable wiring for 7510A's Mic Inputs and Main Output.

longer Release Times will be desirable for speech and shorter ones for music.

15. Set any desired channels to Priority or On mode.

WHAT TO DO IN THE EVENT OF A SETUP PROBLEM

If the beginning of words or musical notes tend to be cut off, or if one must talk or play more loudly than is desirable for the channel to turn on, a given input channel's threshold is too high (i.e. sensitivity is too low). Turn the Threshold trimmer clockwise until the symptom is alleviated.

If the end of words or musical notes tend to be cut off on a given input channel, its Release Time is too short. Turn the Release Time Trimmer clockwise until the symptom is alleviated.

If the unit cannot be made to work properly, all connections have been checked, and the preceding setup procedure has been accurately followed, there may be a malfunction in the electronics. See Section 8, MAINTENANCE/SERVICE, *page* 21.

4 DETAILED INSTALLATION INFORMATION

CABLES FOR THE CHANNEL INPUTS AND MAIN OUTPUT

The 7510A channel inputs are 3 pin XL-type connectors. These connectors are advantageous for several reasons: (1) they mate with the most popular types of professional mic cable connectors (2) they provide for balanced cables which are less susceptible to hum and buzz, and are necessary for the phantom mic power (3) they are locking connectors so the cables will not pull out accidentally, and (4) the ground pin is longer and mates first so there is less chance of a pop when the cable is plugged in. The Main Ouput Jack also is an XL-type, offering similar advantages, although phantom power is not a concern.

NOTE: The XL-type connectors in the JBL 7510A are wired to conform with U.S. practice, which dictates that pin 3 is high, pin 2 is low and pin 1 is the shield connection. Given a positive signal at pin 3 of an input, it will be in phase (same polarity) with the tip of the channel's Direct Output. Similarly, a positive signal at the tip of the Auxiliary Input will be in phase with pin 3 of the Main Output. Some sound equipment may be wired to the IEC standard, which sets pin 2 as audio high and pin 3 as audio low, leaving pin 1 for shield ground. In many instances, there will be no problem interconnecting such equipment with the 7510A as the 7510A maintains consistent input to main output polarity; if polarity reversal is objectionable, an extender adapter can be wired so that the wires between pins 2 and 3 at either end of the adapter cross over.

CABLES FOR THE DIRECT OUTPUTS

The 7510A Direct Outputs are unbalanced circuits that utilize stereo phone jacks (also known as three-conductor standard phone jacks or tip-ring-sleeve jacks). Each jack is used for a single channel of audio, not two channels.

While the jack has three conductors, it is unbalanced; the tip carries the signal, the ring carries the line level return to the mixing buss, and the sleeve is chassis ground.

The jack is normalled so that when no cable is plugged in, the signal flows straight from the input channel to the mixing buss. If desired, the input channel signal can flow directly to the 7510A's Main Output and simultaneously provide a Direct Output by connecting the tip and ring together on the phone jack.

CABLES FOR THE AUXILIARY INPUT

The 7510A Auxiliary Input is an unbalanced circuit that utilizes a stereo phone jack (also known as three-conductor standard phone jacks or tip-

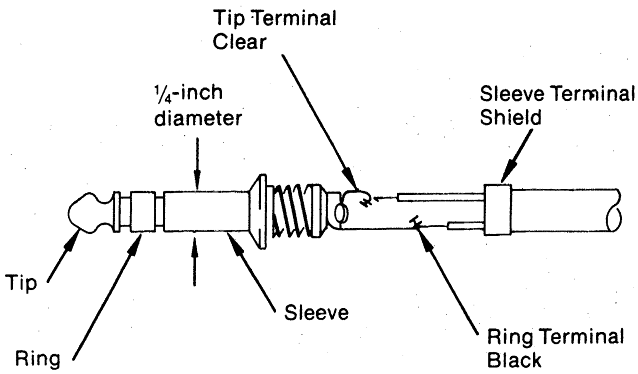


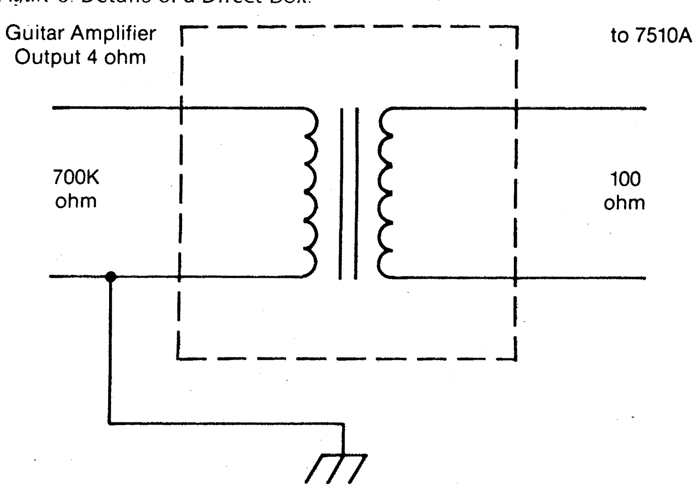
Figure 5. Cable wiring for the 7510A's Auxiliary Input.

ring-sleeve jacks). Like the direct output, each jack is used for a single channel of audio, not two channels. By wiring the jack's tip for the audio "high" conductor, its ring for the audio "low" conductor, or "common" and its sleeve for a separate shield ground (chassis ground), this unbalanced connection provides the opportunity for better shielding against hum and noise than a standard two-conductor (tip-shield) phone jack. Ideally, cable with two center conductors, plus a braided shield should be used, and wired as illustrated. This shield is cut (unused) at one end of the cable, avoiding an unwanted ground loop which can increase hum. If a cable with a tip-ring-sleeve phone plug is unavailable, a standard phone plug may be used; it will short the audio common to the chassis, but this is generally acceptable (the worst that would happen is some hum, which should go away when the proper connector is substituted.)

USING HIGH IMPEDANCE MICROPHONES

Although JBL does not recommend the use of high impedance microphones, they can work with the 7510A directly, provided their specifications allow them to be loaded with the 7510A's input impedance of 10K ohm. If the microphone requires a higher impedance than 10K ohm, then an auxiliary matching transformer will be necessary to avoid overloading the microphone. Such transformers are available from several vendors; some come in convenient in-line packages which resemble a large XL connector and actually contain a miniature transformer, an XL connector for the balanced, low impedance feed to the 7510A's input, and another connector (phone jack or equivalent) which accepts the cable from the microphone. If more than ten (10) feet of microphone cable is needed, locate the transformer close to the microphone and run the longer cable

Figure 6. Details of a Direct Box.



NOTE: Transformer reduces voltage by 38dB. The 7510A input set for 0 gain will allow a guitar amplifier up to 400 watts to be safely used.

on the low impedance side of the transformer. Bear in mind that the 7510A outputs can feed low or high impedance inputs without any special adapters or transformers.

USING HIGH LEVEL INPUT SOURCES WITH THE 7510A

The remote connector at the rear of each input module provides for individual gain adjustment of the pre-amplifier stage of each input for use with a high output level device. Figure 15, *page* 17, shows the pin configuration of the AMP HD-20 15 pin remote connector on the rear of the input module. Normally jumpers are provided for microphone use, setting the input pre-amplifier gain at 49 dB, but an appropriate resistor can be substituted if lower gain is desired. The following table shows gain and the required 5% ⅛ or ¼ watt resistor.

0dB	=	open	28dB	=	220 ohm
4dB	=	10K ohm	29dB	=	200 ohm
7dB	=	4.7K ohm	31dB	=	150 ohm
9dB	=	3.3K ohm	34dB	=	100 ohm
10dB	=	2.7K ohm	37dB	=	62 ohm
13dB	=	1.6K ohm	39dB	=	47 ohm
14dB	=	1.5K ohm	40dB	=	39 ohm
16dB	=	1.1K ohm	43dB	=	20 ohm
19dB	=	750 ohm	44dB	=	16 ohm
22dB	=	470 ohm	46dB	=	10 ohm
24dB	=	360 ohm	49dB	=	Jumper
25dB	=	330 ohm			

DIRECT BOXES

A "direct box" is an adapter which, as most often found, permits either (1) a guitar amplifier's speaker output to drive a microphone input, (2) a preamplified instrument's output to drive a microphone input, or (3) a very high impedance instrument pickup to drive a low to medium impedance microphone input. "Direct" refers to a direct-wired connection to the mixer's microphone input rather than the "indirect" approach of placing a microphone in front of the instrument amplifier's speaker. The direct hookup enables the musician to take advantage of the reverb, distortion, phasing, and other effects available at the amplifier, yet avoid the leakage and noise that might be a problem using the speaker and microphone approach. Direct boxes that come straight off the instrument can be used to obtain a cleaner sound. A variety of direct boxes are commercially available, some being switchable to perform both of the above functions.

It is important that a transformer be used for isolation any time the output of a guitar amplifier is to be fed to a 7510A. While external gain resistors can be used to reduce the gain of the input stage sufficiently so that a simple 15K - to - 150 ohm transformer can be used ahead of the input, as shown in Figure 6.

If, however, a standard direct box is to be used, the user must be aware that such boxes invariably have loss built into them ahead of the transformer, since most of these boxes are designed to reduce the signal to microphone level. In such a case, it would be best to leave the 7510A input gain set for typical microphone levels.

ROUTING CABLES

Cables found in a sound system can be divided into five general categories:

1. Low level cables (microphones/pickups) with nominal signal levels of from −60 to −30 dBm and maximum levels of about +4 dBm.
2. Line level cables (preamps/mixer outputs) with nominal signal levels of from −20 to +4 dBm, and maximum levels of about +24 dBm.
3. High level circuits (speaker lines) with maximum signal levels of above +30 dBm (24.5 volts).
4. AC power circuits (including lighting).
5. DC power circuits (including control signals).

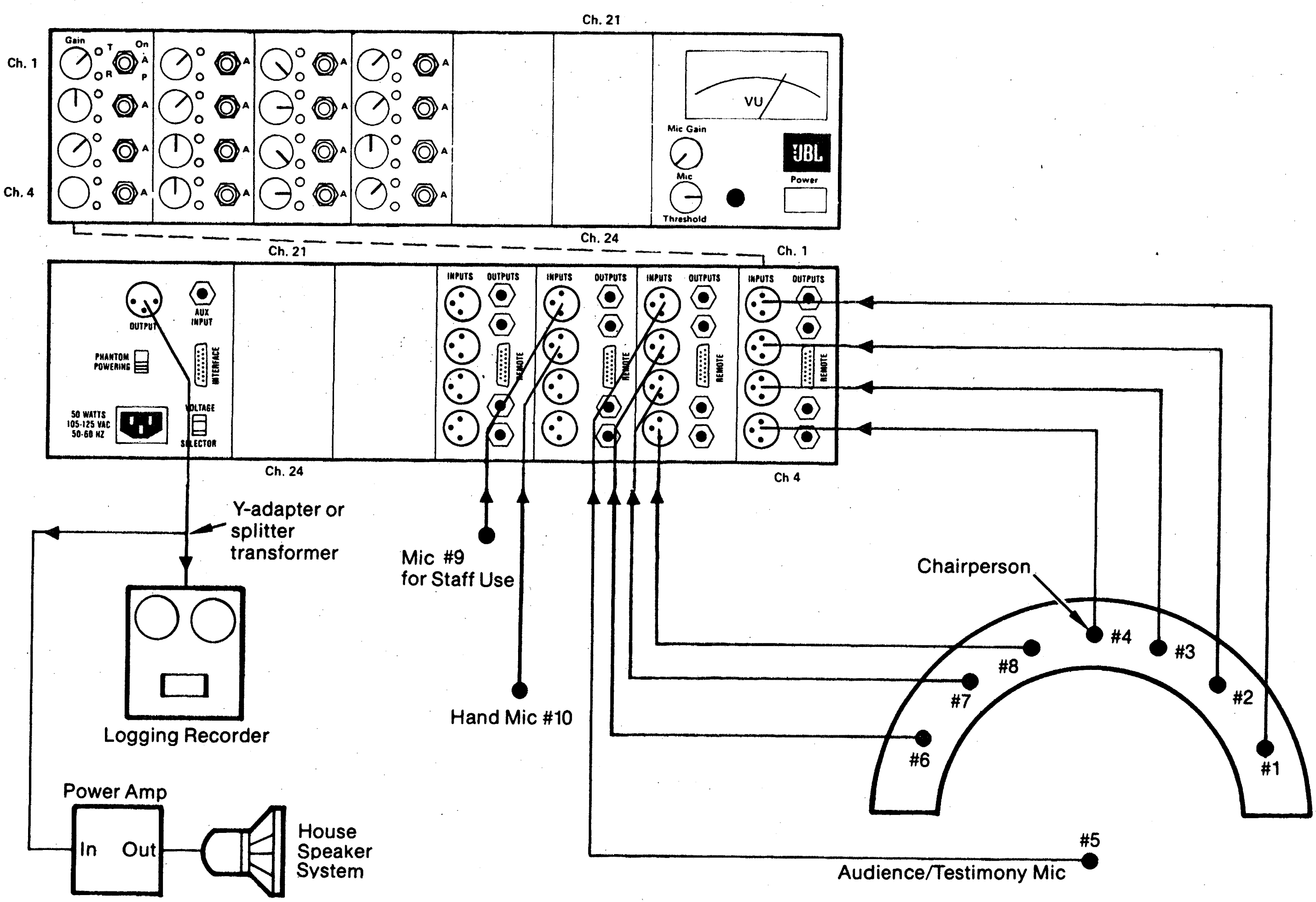


Figure 7. 7510A in a meeting Room Sound System.

To the greatest extent feasible, these five types of cables should be physically separated from each other; if they must run between the same places, keep them apart or loosely bundled, but never tightly strapped to each other. When the various types of cables must cross, cross at right angles. If using a "snake" (multi-channel shielded audio cable), don't mix microphone inputs and line outputs in the same cable. These practices help avoid unwanted crosstalk, including noise spikes from DC control lines into the audio, induced hum from AC lines into the audio, and feedback loops between high level outputs and the low level inputs to the same device.

5 APPLICATIONS

MEETING ROOM SOUND SYSTEM

In a board room, city or county council chamber, public hearing room, courtroom, or other similar situation, there is generally a desk or table around which is placed one microphone per person. One or two additional microphones may be provided for testimony. The audio mixing system typically has two functions: (1) to feed a sound reinforcement system so that everybody can hear the person speaking, and (2) to feed a tape recorder for documentation of the proceedings.

A common requirement in this application is the need for at least one of the microphones to take priority over all others—usually the microphone used by the person who is in charge of the meeting or hearing. The 7510A will automatically mix the sound as required, preventing background coughing or whispering from being amplified, yet ensuring that direct comments are heard.

By setting the chairperson's microphone input in Priority mode and all other inputs in Auto mode, that person has instant override of other discussion. Sometimes, as in a hearing, it may be desirable to enforce decorum by setting all inputs in Priority mode except the testimony microphone or microphones, which are set in Auto mode. In a city council meeting, for example, any councilperson could speak simultaneously with all others, but someone from the audience giving testimony would not be heard until all councilpersons were silent. By activating a switch connected between Pin 9 and Pin 10 of the interface connector on the output module the chairperson can turn off all microphones in the auto mode, but not any microphones in the Priority or On modes.

In most of these situations, the 7510A's Main Output would be fed to both the sound reinforcement amplifier/speaker system and the tape recorder by using either a "Y" adapter cable or a splitter transformer. In more sophisticated systems, as where a stereo recording is made or where sound must be distributed to remote speakers, the direct outputs could be fed to another mixing board. A mono output from that outboard mixer would then be returned to the 7510A Auxiliary Input so the unit's digitally attenuated Main Output would continue to control feedback in the "local" reinforcement system. The outboard mixer's other outputs could feed remote speakers and/or a multitrack recorder.

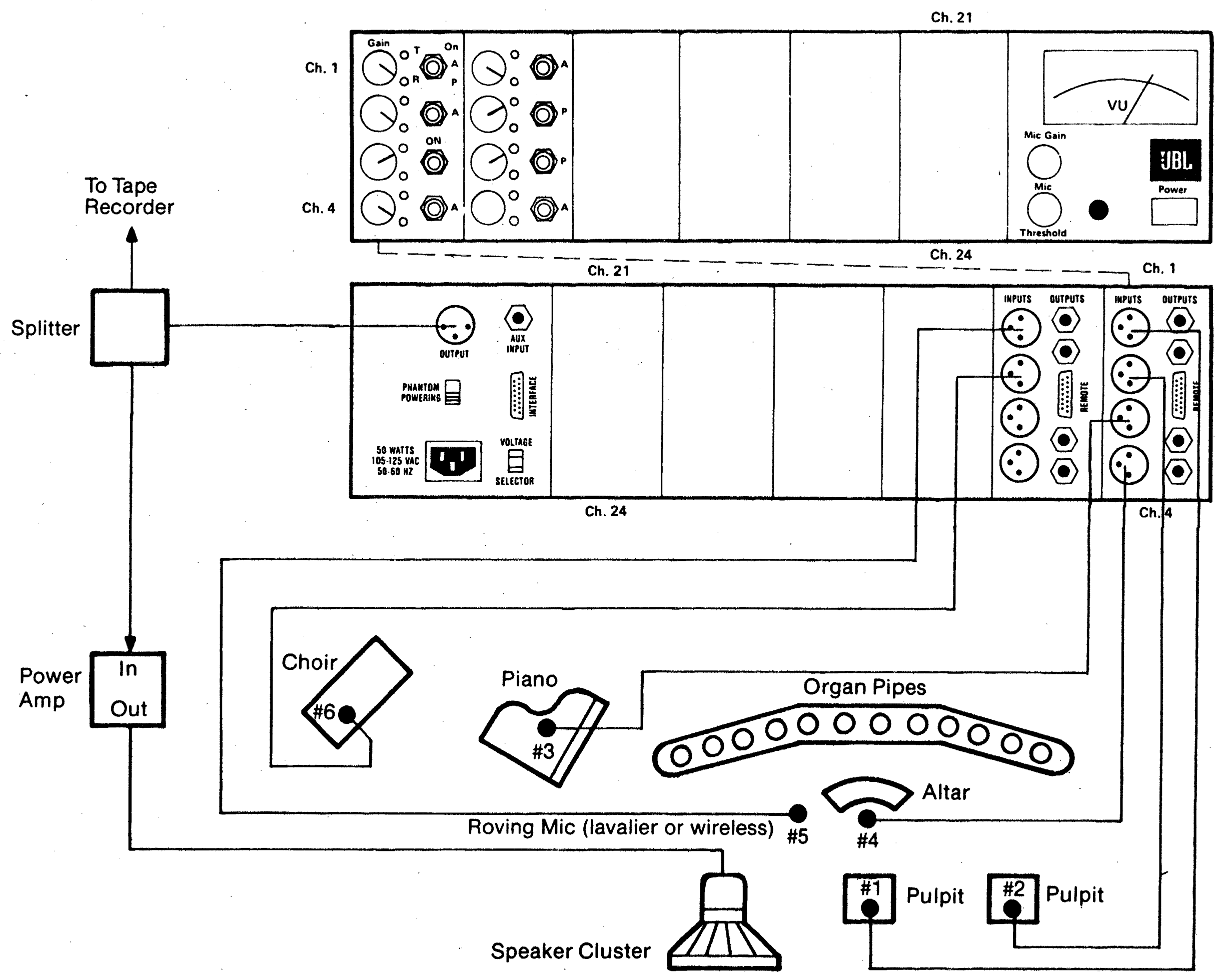


Figure 8. 7510A in a Hall of Worship.

CHURCH OR SYNAGOGUE SOUND SYSTEM

In any house of worship, the typical sound system must provide ample reinforcement so the congregation can hear the spiritual leader. The 7510A is an excellent choice here for several reasons. Because the mixer automatically turns Off unused microphones, it avoids the "tinny" or "hollow" characteristics that otherwise would detract from the sound quality. At the same time, the automatic output gain reduction avoids feedback or "howling." Additionally, it's all automatic, reliable, and 100% unattended.

By setting the roving microphone (either a lavalier or wireless microphone) in the priority mode and all other inputs in the Auto mode, that person has instant override over all other microphones. This prevents a choir microphone or any other microphone from turning on and amplifying someone coughing or other unwanted noise during the sermon. This also prevents the pulpit or other microphones in the Auto mode from being activated should the person with the roving microphone be in the area of these microphones, causing a sound quality change. If possible, only one microphone should be used to pick up the choir. If this is not possible, the background Threshold Control must be set at a lower (counterclockwise) setting than normal, thus allowing the additional microphones to all be activated simultaneously when the entire choir sings.

TELEVISION QUIZ SHOW AUDIO

When mixing sound for a TV quiz show, there are several requirements. Typically, there are several contestant/guest microphones, plus an emcee (Master of Ceremonies) microphone and an announcer microphone. The mixed audio must be fed not only to the VTR (Video Tape Recorder), but also to audience monitor speakers. The 7510A is useful in three ways in this application: (1) its gated input circuitry keeps inactive microphones from contributing background noise, mumbles, and other extraneous "off microphone" sounds to the mix, (2) its priority system allows the announcer and/or emcee to override the other microphones, and (3) its automatic gain reduction circuitry can be used to prevent feedback from developing in the audience monitor speakers.

The 7510A is not meant to replace the conventional audio mixing board in this instance, but to augment its capability, ease the demands on the person doing the mixing, and improve the overall sound quality.

The 7510A Direct Outputs are all fed to the main audio board in the control room, where the sound can be appropriately equalized, sub-grouped, and otherwise processed. A monitor mix is then returned to the 7510A's Auxiliary Input for feed to the monitor amplifiers and speakers, utilizing the 7510A's digital attenuator to provide maximum gain before feedback to the audience. The audience mix may not be the same as that fed to the VTR, in terms of EQ and overall content. Still, the main audio board feeds a mix derived from the same 7510A Direct Outputs probably with limiting and compression added to the VCR (or, in the less common instance of a live show, to the transmission point).

RADIO SHOW AUDIO

The 7510A can be used as the sole mixer for remote productions where the crew is small and the tasks many. However, it is of equal value in the

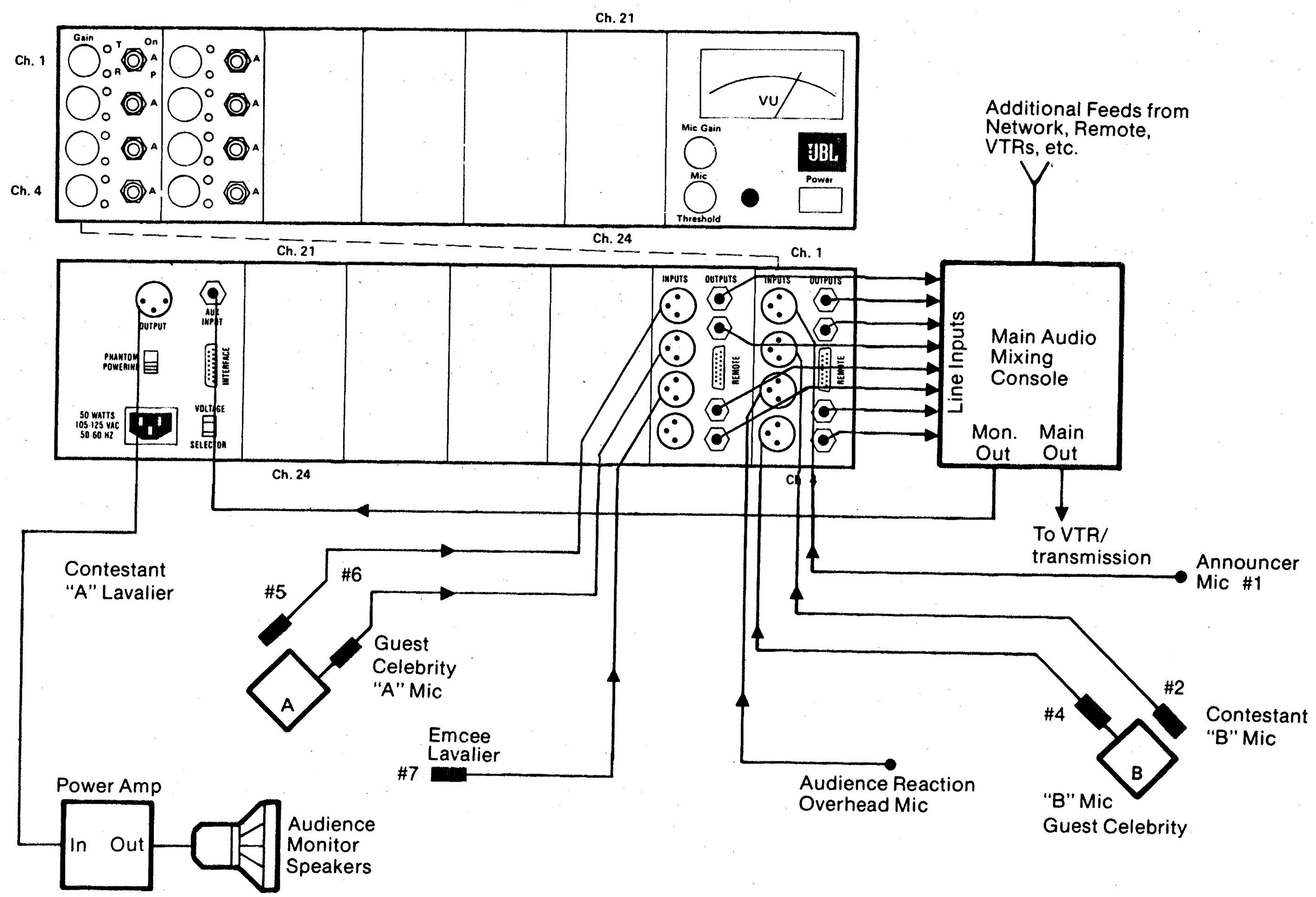


Figure 9. Mixing the Sound for a TV Quiz Show.

studio because it can clean up the sound of interviews, discussions, and other multiple microphone conversation productions. The Background Threshold would generally be turned On. The input channel Threshold controls should be set low enough to ensure that the channel will turn On when required (typically full clockwise). This not only relieves the mixer of "riding gain," it also takes pressure off performers and guests because they need not be overly concerned about making distracting off-mic noises.

The 7510A Main Output can be fed directly to the transmission point, TELCO (telephone company) inteıface, or recorder. It is more likely, however, that the Direct Outputs would be fed to another mixing board, where EQ, reverb, compression, limiting, etc. may be added as required, and the program can be distributed and mixed with recorded commercial sports, news and so forth.

If the host is also the engineer, as sometimes is the case with automated stations, the 7510A's priority feature can be used to aid in smoother transitions between live studio productions and automated programming. For example, consider a situation where the automated cartridge carousel (cart machine) is set to play the program just before and immediately following an in-studio talk show interview.

The 7510A is placed in the studio rather than the control room, and the talk show microphones are plugged into it. These channels are set to Auto mode. The 7510A main output is fed into the main audio board, which relays the audio to the transmission point. The cart machine's output is split to feed not only the main audio board, but also to feed one input on the 7510A, and its channel on the 7510A is set to Priority mode.

Before the show, the host/engineer goes into the control room, brings down the direct cart feed to the board, and brings up the cart that is routed through the 7510A. As the cartridge is playing the introduction to the show, the host walks into the studio. The instant the cart machine shuts down, as programmed, the host and the guest microphones become live. When the host wraps up the show at the correct time, the cart machine plays another tape through the 7510A, thus killing the studio microphones and simultaneously feeding the taped program to the transmission point. The host/engineer then walks into the control room, and at an appropriate pause in the program, fades out the 7510A and fades in the cart machine's direct feed to the main audio board.

RECORDING STUDIO NOISE GATE

In the recording studio the 7510A may be used as an unattended submixer, although acoustic feedback is not a problem here and the digital attenuation is not needed. However, there is a much more valuable studio application for the 7510A—that of a multi-channel noise gate. Almost any vocal or instrument microphone can be gated with the 7510A; the application depends on the microphone setup and the overall mix.

For example, suppose a drum set is being recorded with between 4 and 8 close microphones. The microphones on the kick drum may be picking up a large amount of sound from the toms. This indirect tom sound, when mixed down to stereo, may be acoustically out-of-phase with the direct sound from the microphone at the tom toms. The result is a muddiness or lack of clarity that cannot be cured with any amount of post-production processing. Similar leakage (crosstalk) problems are likely to occur with the other drum microphones, as well. The solution is to have a microphone "live" only during the brief time its related drum or percussion instrument is sounding a note. This process is called "gating," and only a

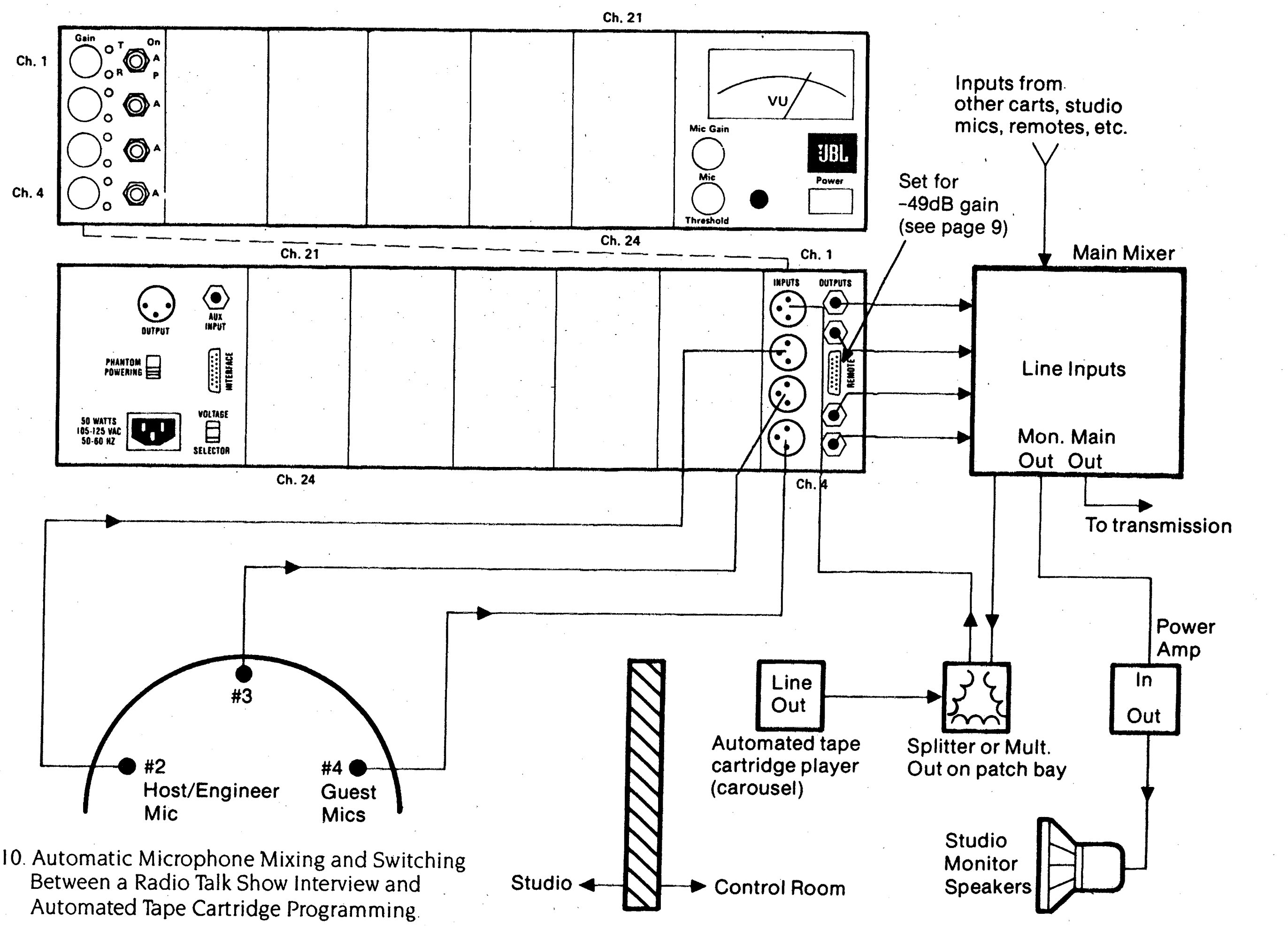


Figure 10. Automatic Microphone Mixing and Switching Between a Radio Talk Show Interview and Automated Tape Cartridge Programming

very fast, high-quality gate will do a creditable job. The JBL 7510A is such a unit.

The drum microphones are connected to the 7510A channel inputs, whose individual Gains are adjusted to yield maximum level without distortion. The Background Threshold is turned on and set as per normal setup procedure and the individual channel Threshold Control should be set low enough to ensure that the channel will turn On when required (typically full clockwise). The Direct Output of each channel is connected to a line input on the recording board. Channel-to-channel level adjustments, EQ, reverb, and other signal processing can be done on the main board. In this instance, the 7510A Main Output is not used. Where tracks are at a premium, however, the 7510A's individual channel Gains can be set for the desired mix and the Main Output can be fed to the recording board as a complete mono mix of the drums. Since only two or three microphones are likely to be live at any given instant, the digital attenuator's gain reduction should not significantly alter the dynamics.

CONCERT SOUND REINFORCEMENT NOISE GATE

In live musical performances, the sound levels on stage can be extremely high, because of amplified instruments and powerful stage monitor speakers. There may be from 24 to 48 open microphones or more, all picking up this high ambient background sound level. The result is a combination of ills ranging from a "thin" or "muddy" off-microphone sound, to a substantial decrease in the available system gain before feedback. During the performance, there is no feasible way for the soundman to manually shut down those microphones which are not actually being used from one moment to the next, but this can be done easily with the 7510A. The setup is essentially the same as that described in the previous description of a recording studio application.

Using the 7510A, gate Off inactive stage microphones to benefit both stage monitor mixing and house mixing. The 7510A Direct Outputs can each be fed to a splitter transformer (or to a simple "Y" adapter if both mixers have actual line input impedance of over 1K ohm). While the 7510A digital output attenuator is not being used, there is typically an increase of from 5 to 10 dB usuable gain before feedback simply because fewer microphones are live at any given moment. Of course, such a setup must be tested thoroughly in a sound check with the maximum number of microphones being used in order to ensure that feedback will not occur.

For a smaller scale production where there is a main audio board for the house and not separate stage monitor mixer, the 7510A's Direct Outputs can be fed to the main board for mixing, signal processing, and fed to the house speakers. Locating the 7510A at the stage is advantageous because it provides low impedance line-level feeds back to the main board, reducing susceptibility to hum and noise. A mono submix of those microphones can be done on one of the main audio board's busses and feed back to the 7510A Auxiliary Input. The 7510A Main Output can then be fed to the monitor amplifier and stage monitor speaker(s). In this instance, the 7510A digital output attenuator does help prevent feedback. What's more, the stage monitor level can be readjusted by the performers during a show, something difficult to achieve when the levels are being set at the board in the back of the house. They simply reset the 7510A Master Gain; so long as the Input Gain controls are untouched, the house levels will be unaffected.

SMALL CLUB SOUND REINFORCEMENT MIXER

In a small night club/cabaret situation, it is rare to have a soundman. Often the bartender, headwaiter, or club host doubles as the sound mixer,

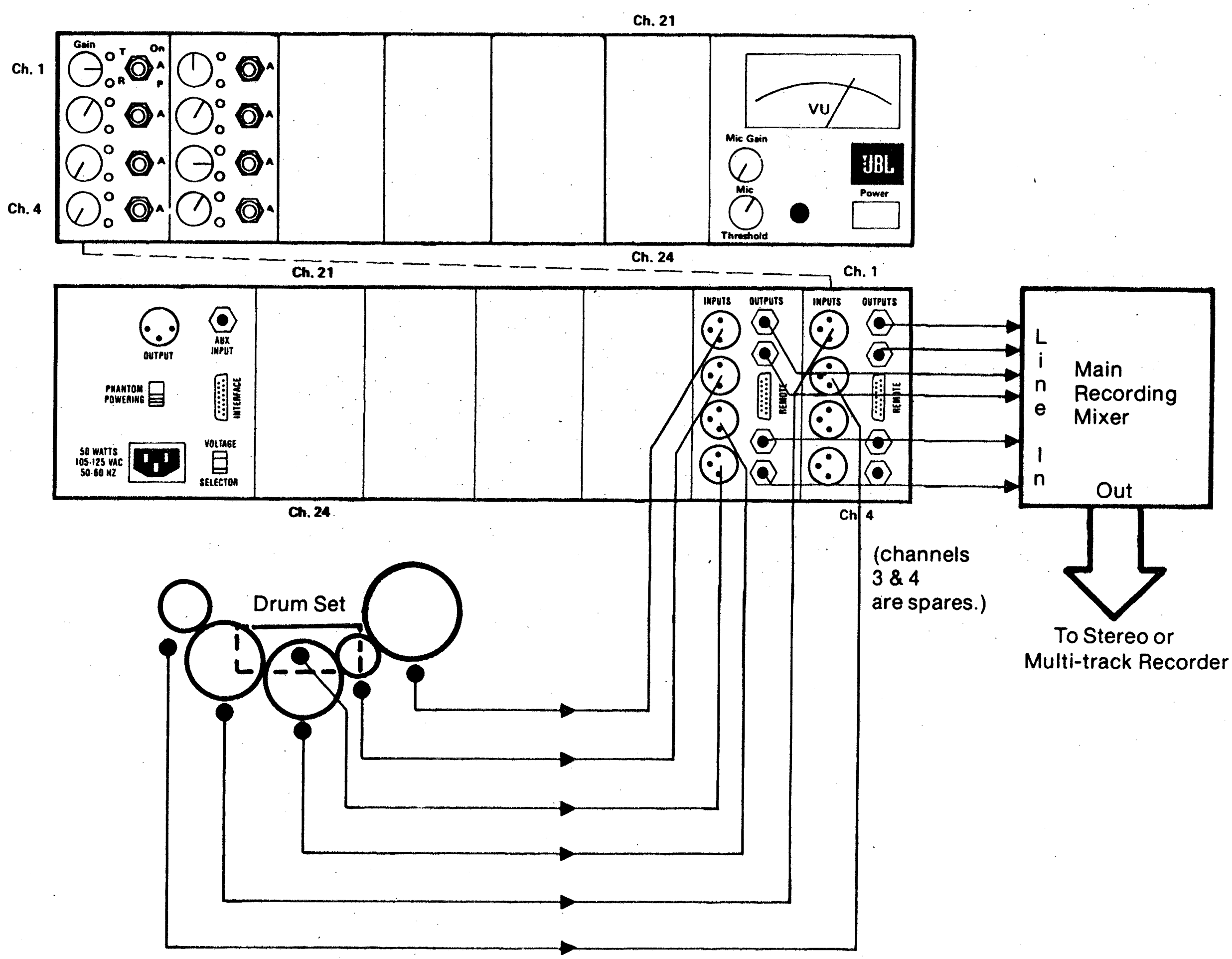


Figure 11. Using the 7510A as a Noise Gate.

which is obviously not the ideal approach for obtaining the best sound. The 7510A can do wonders in this application. The channel Gain controls for the stage microphones can be balanced on a day-to-day basis to suit the performer's requirements, and Master Gain can be used as the overall sound system installer to a reasonable average, avoiding the need for the operator to be extensively trained or even knowledgeable about audio.

Individual channel Threshold controls can be preset to a relatively low level (typically fully clockwise, maximum sensitivity) ensuring that the channel will turn On whether used for quiet female vocalist's microphone or a brass player. To prevent unwanted "keying" of the microphones by loud audiences, or leakage from off-microphone instruments, the Background Threshold is turned On and adjusted per the normal setup procedure.

The Main Output will feed the power amplifier and speaker system, which is likely to include graphic equalization. Stage microphones can be set to Auto mode, and the host/announcer microphone can be set to Priority mode with a high channel Threshold. In this way, no fader adjustment is needed to introduce a group; just pick up the house microphone, speak with a firm voice, and all the stage microphones will be silenced.

This basic system can be enhanced as desired. An FM tuner can be connected to one of the 7510A input channels, which is set to Priority mode. When the channel Gain is up, the channel will turn On and silence the stage microphones, as is desirable during breaks or when no group is playing. When the stage microphones are to be used, just turn down the tuner channel's Gain. Alternately, the tuner might be located at the bar. If the 7510A "tuner" channel gain is left up, the bartender can remotely turn Off the stage microphones and bring up the background music by turning up the tuner volume.

If reverb, channel EQ, or other effects are desired, the 7510A Direct Outputs can be connected to another mixing board, graphic equalizer, reverb, etc. and the external processor's output can be brought back into the 7510A's Auxiliary Input or each individual channels mixing buss.

MORE ABOUT THE PRIORITY SYSTEM

The 7510A, while it makes an excellent noise gate, is not set up to be a keyable noise gate. That is, there is no "key" input on a channel that will turn the channel On when a second input to the channel rises above a set threshold level. Instead, a 7510A channel turns On and Off only by sensing the level at its input connector. There is a way to have one channel affect another: use the Priority system. The results are not the same as with a keyable noise gate, but in some instances they may be more desirable.

For example, let's go back to the recording studio application. Perhaps leakage is not so much of a problem, but a "tight," well synchronized drum sound is critical. You may find that it is advantageous to silence the kick drum whenever the tom-tom is struck. It's easy. Plug the mircophones for these drums into the 7510A, set the kick channel to Auto mode and the tom (or as many tom microphones as you wish) to Priority mode. If you want to conserve on channel usage in the recording board and don't mind using the same EQ and signal assignment, you can feed the 7510A Main Output to the recording console; otherwise use the Direct Outputs of these channels. You'll want to be sure the tom channels are set for a short release time so that the kick can come on the moment the tom's sound

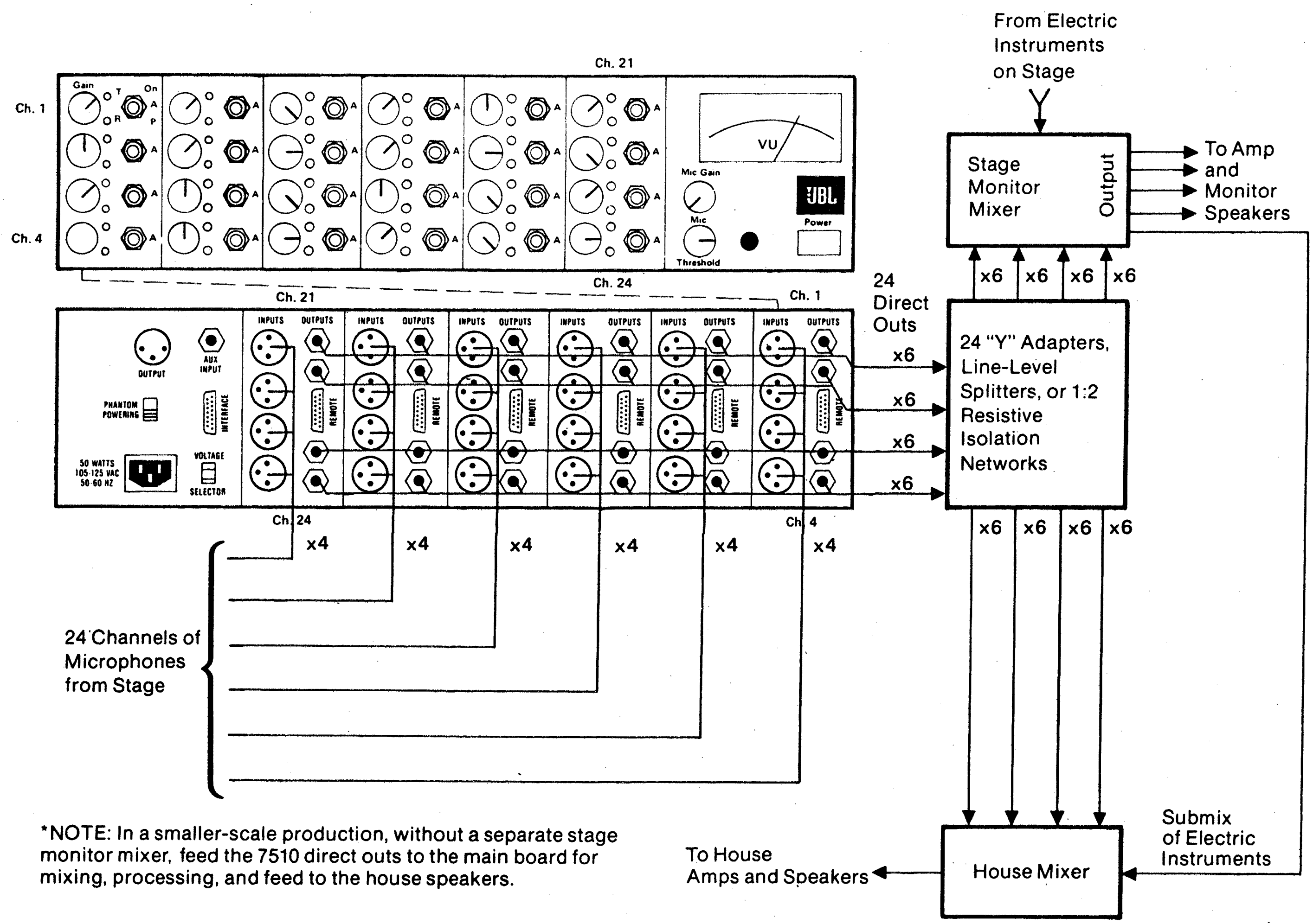


Figure 12. Concert sound Reinforcement Noise Gate.

dies away. Other microphones are either set on On mode, or are not run through this 7510A.

The various applications have pointed to many uses for the Auto and Priority modes, but not the On mode. On is always used for setup, but it is a functional mode in many applications. For instance, where On mode is useful is where you don't want a channel to turn Off when a Priority channel turns On — nor do you want the channel in question to be a Priority channel itself. In the TV show application, for instance, the emcee microphone could be set to On so that when the announcer is speaking the contestants are silenced, but the emcee can converse with the announcer or the contestants.

6 MORE INFORMATION ON THE 7510A AND HOW IT WORKS

ABOUT NOISE GATES

The traditional noise gate is essentially an electronic switch that lets signal flow through it (turns On) only when the signal is above a set level (the Threshold). By setting the threshold appropriately, background noises and "leakage" from nearby sound sources are excluded, and only the desired sound, identified because it is louder than the noise, is allowed to pass. Traditional noise gates are useful to clean up individual tracks in a recording situation, but have been less than useful in live sound reinforcement because the sound from stage monitors, instrument amps, etc., causes the gates to turn On even if the microphone is not supposed to be live. Another drawback of conventional noise gates is that sudden noise spikes can turn the channel On.

The 7510A, while it acts like a noise gate, is hardly conventional. Because of its background threshold circuitry, the 7510A is able to raise and lower the threshold in accordance with the overall ambient sound level, making it possible to take advantage of the noise gate in live performance applications. Moreover, because the 7510A can react to the presence of above-threshold sound very rapidly, and switches On at a zero crossing, the noise gate action itself is virtually inaudible. Once the channel is on, the sound flows through state-of-the-art audio amplifier stages, and is therefore free of distortion and modulation noise, unlike some VCA-attenuated or FET-switched designs.

NOTE: The signal is not actually switched on and off; only the gain is changed (raised) 25.5 dB.

ABOUT PHASE CANCELLATIONS AND COMB FILTERS

When a given sound source is fed to two different microphones that are located at different distances from the source, the electrical output from those microphones may be out of phase from each other. When such out of phase signals are mixed together, they may reinforce (add together) or cancel (subtract from each other). Since the effective phase difference depends on frequency as well as relative distance from the sound source, these reinforcements and cancellations will change across the audible spectrum. The resulting frequency response graph has many sharp dips (notches), and may look like a drawing of a comb. Hence, the term "comb filter" is used to describe this effect.

Comb filters (or multiple microphones which pick up the same sound

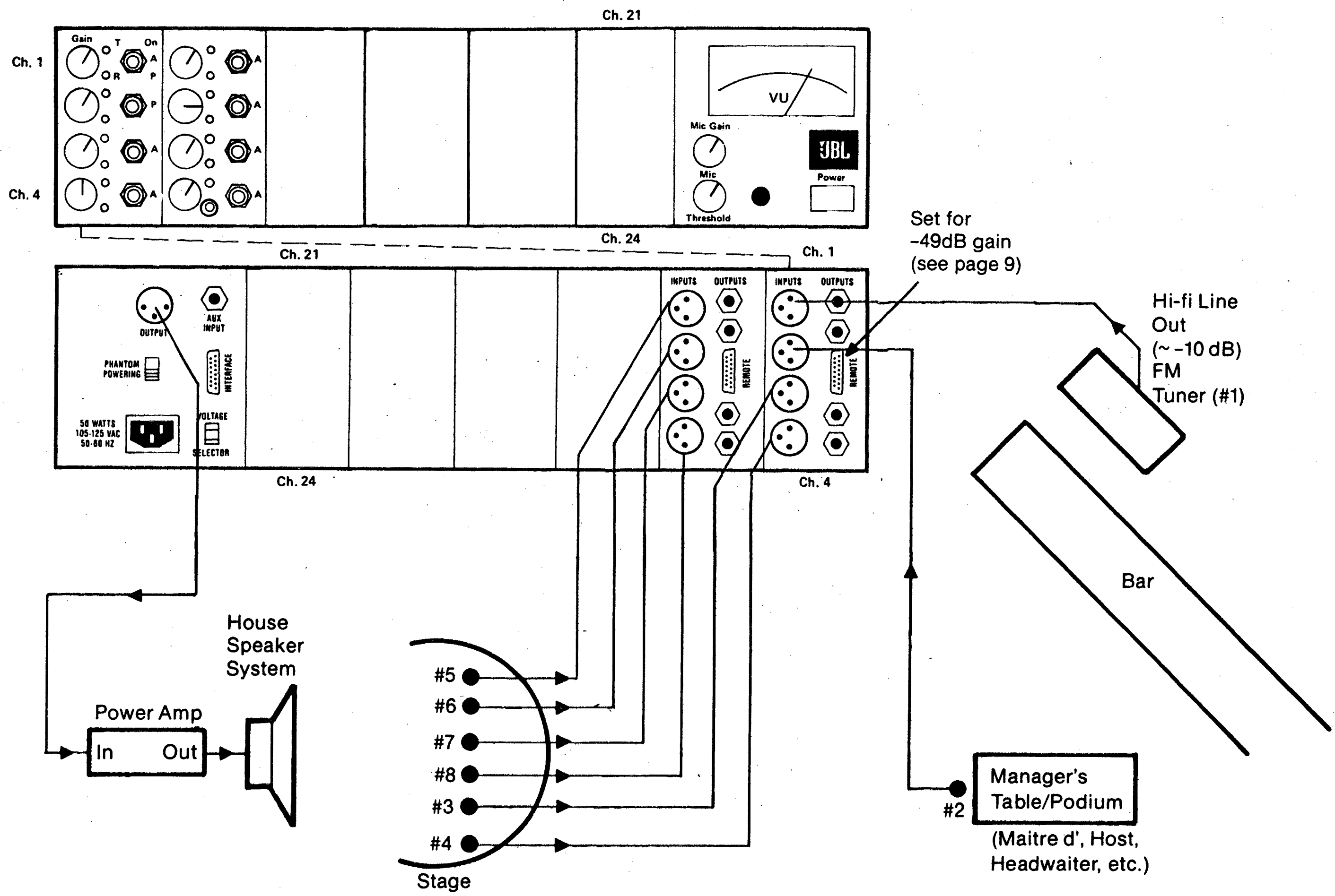


Figure 13. Small Club Sound System.

and inadvertently create a comb filter) can significantly color the sound. The hollow, thin, or "swishy" effect known as "off microphone sound" is a comb filter effect.

Because the 7510A, in Auto mode, keeps a channel Off until its microphone is directly in use, it does not create comb filters as does a conventional mixer, where many microphones will be open even if they are not active.

GAIN AVAILABLE BEFORE FEEDBACK

Acoustic feedback occurs whenever the electro-acoustic gain of the sound system exceeds unity. Simply stated, if the sound reaching the microphone from the speakers is higher than the sound reaching the microphone from the original souce, the sound system will "howl". The idea is to get as much amplification of the voice or instrument as possible without causing "howl", or the ringing that occurs at the verge of howling.

If a sound system had perfectly flat frequency response, the gain could be increased to the point where the whole system would oscillate at all frequencies. This never happens, however. Instead, any sound reinforcement system will have more gain at some frequencies than at others, even if only a dB or two. These peaks may be caused by a resonance in the speaker system or microphone, excess acoustic reflectivity at a given frequency, a specific phase relationship, etc. Thus, when the gain is increased, the system will first begin to feed back at one or two specific frequencies—the frequencies with more system gain.

More overall system gain can be obtained before feedback occurs if the sound system is equalized to smooth response. 1/3 octave, 1/6 octave or even narrower notch filters can be used here. Acoustic treatment and microphone placement are also important factors. But there is one more factor—the number of "open" or "live" microphones. Normally, the available gain before feedback drops by 3 dB every time the number of live microphones in a system is doubled. With fewer microphones live, one can obtain more usable gain (more amplification) from the remaining microphones.

The problem with keeping fewer microphones live in a complex sound system is that the microphones which are actually being used will change from moment to moment. It is impractical, if not impossible, for a soundman to keep up with the microphone usage, and turn down the overall system gain proportionately to the number of microphones in use. Fortunately, the 7510A can keep track of the number of live microphones, and, on an instantaneous basis, the digital output attenuator adjusts the system gain exactly as required. What's more, if a given microphone is not actually needed, the channel turns Off, and the digital attenuator increases the system gain. In practical terms, the system gain available before feedback will increase from 6 to 10 dB when the 7510A is substituted for a conventional microphone mixer.

HOW THE DIGITAL OUTPUT ATTENUATOR WORKS

When a 7510A input channel is turned On, it not only feeds an audio signal to the direct output and/or the audio mixing bus and summing amplifier, it also feeds a digital signal to the digital mixing bus and summing amplifier. The digital summing amplifier thus "knows" how many channels are live from moment to moment. This information is fed to the controlling input of the patented digital attenuator circuit, an eight (8) bit microcomputer of sorts. The attenuator is programmed to reduce an audio amplifier's gain such that for every doubling of the number of live microphones, the gain drops 3 dB. The actual gain reduction required is logarithmically scaled, and is incremented for each additional live microphone (See the table below). The Master Gain control is a conventional fader which follows the digital output attenuator.

Figure 14. How the digital output attenuator reduces the gain in response to the number of active mics. (An active mic may be defined by a channel whose LED indicator is illuminated.)

ACTIVE MICS	OUTPUT GAIN REDUCTION	ACTIVE MICS	OUTPUT GAIN REDUCTION	ACTIVE MICS	OUTPUT GAIN REDUCTION
1	0 dB	9	−9.5 dB	17	−12.3 dB
2	−3 dB	10	−10 dB	18	−12.6 dB
3	−4.8 dB	11	−10.4 dB	19	−12.8 dB
4	−6 dB	12	−10.8 dB	20	−13.0 dB
5	−7 dB	13	−11.1 dB	21	−13.2 dB
6	−7.8 dB	14	−11.5 dB	22	−13.4 dB
7	−8.5 dB	15	−11.8 dB	23	−13.6 dB
8	−9 dB	16	−12 dB	24	−13.8 dB

USE OF THE 7510A'S LOGIC OUTPUTS

In Figure 15, we show the remote control pin configuration for the logic outputs for each of the input gating functions of the 7510A. When an input is activated, an active switching function takes place internally. Up to ±5 volts can be switched with maximum current of 10 milliampers.

In Figure 16, we show four ways the active switching function can be implemented. Two of these circuits are used to activate relays, one is used to drive Series 7400 TTL devices, and one would activate an external LED.

Note carefully that this function of the input module *does not* provide an output voltage; it merely provides a switching action for an external voltage source.

As an example of how we might use this feature, consider a board room installation with a dense array of ceiling loudspeakers. When a given microphone is activated the logic output for that microphone can be used to mute loudspeakers directly overhead, allowing the system to be operated at higher than normal gains.

Another application: Assume that a participant on a panel discussion wanted to record or videotape *only* his own comments. The logic output for his microphone can be used to start and stop the recorder.

In a teleconferencing application, video cameras could be switched, allowing the video to follow the audio. If more than one person was speaking, an overall camera could be activated.

INTERFACING TWO OR MORE 7510A'S

Two or more 7510A's can be connected to act as a single, larger unit by connecting their 15-pin interface outputs with the mating equipment and hardware, see Figure 17, *page* 17. When connected in this way, the digital attenuator, priority system, background threshold and audio will operate as a single 7510A.

A typical application would be in a large hotel or convention complex with many rooms, normally partitioned off with dividers, but which can be opened up into larger spaces. A single 7510A would be assigned to each room. When the spaces are coupled, the respective 7510A's would be coupled as well to handle the larger number of microphones which would be required for the larger space.

Another application would be large meeting rooms or a legislative room requiring more than twenty-four (24) microphones.

Contact the factory for additional information regarding interfacing several 7510A's to operate as a single unit.

HOW AUTOMATIC DIGITAL ATTENUATION DIFFERS FROM COMPRESSION OR LIMITING

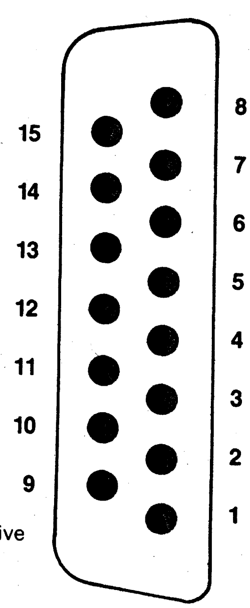
The digital output attenuator is used to maintain a constant system gain, thus preventing feedback. More live microphones lead to more system gain, hence the attenuator responds by lowering the gain. Fewer live microphones permit more gain to be used.

A compressor doesn't care how many microphones are live; it simply holds the output level from exceeding the set threshold by very much, and it does this according to a set compression ratio. A limiter effectively doesn't let the level get above threshold at all. In either case, more sound level (not more open microphones) is met with more attenuation. The result is that a compressor/limiter then affects the program dynamics

Figure 15. 7510A Remote Control Connector.

PIN NUMBER

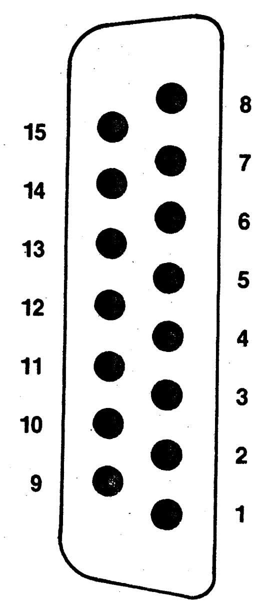
1.–2. Channel C (3) Audio Gain (150M)
3. Channel C (3) Active Switch
4. Channel A (1) Active Switch
5. Channel D (4) Active Switch
6. Channel B (2) Active Switch
7.–8. Channel B (2) Audio Gain (150M)
9.–10. Channel D (4) Audio Gain (150M)
11.–13. Ground
14.–15. Channel A (1) Audio Gain (150M)



NOTES:
1. Audio Gain controls gain of Model 150M Active Input Module, 1st stage.
2. Active Switch capable of ±5 volts at 10 ma maximum, switches to pins 11, 12, & 13.
3. Switch circuits are open until an input channel is activated.

Figure 17. 7510A Interface Connector.

Pin 1 From Background Threshold Buss (Pin 7 of U8)
Pin 2 To Background Threshold Buss (R33)
Pin 3 To Background Threshold Buss (R34)
Pin 4 From Background Threshold Buss (Pin 14 of U7)
Pin 5 To Attenuator Buss (R25)
Pin 6 From Attenuator Buss (Pin 8 of U7)
Pin 7 From Attenuator Buss (Pin 1 of U7)
Pin 8 To Attenuator Buss (R24)
Pin 9 Priority Buss (Pin 13 of U9)
Pin 10 Ground
Pin 11 Ground
Pin 12 From Audio Mix Buss (R9)
Pin 13 To Audo Mix Buss (R3)
Pin 14 From Audio Mix Buss (R10)
Pin 15 To Audio Mix Buss (R4)

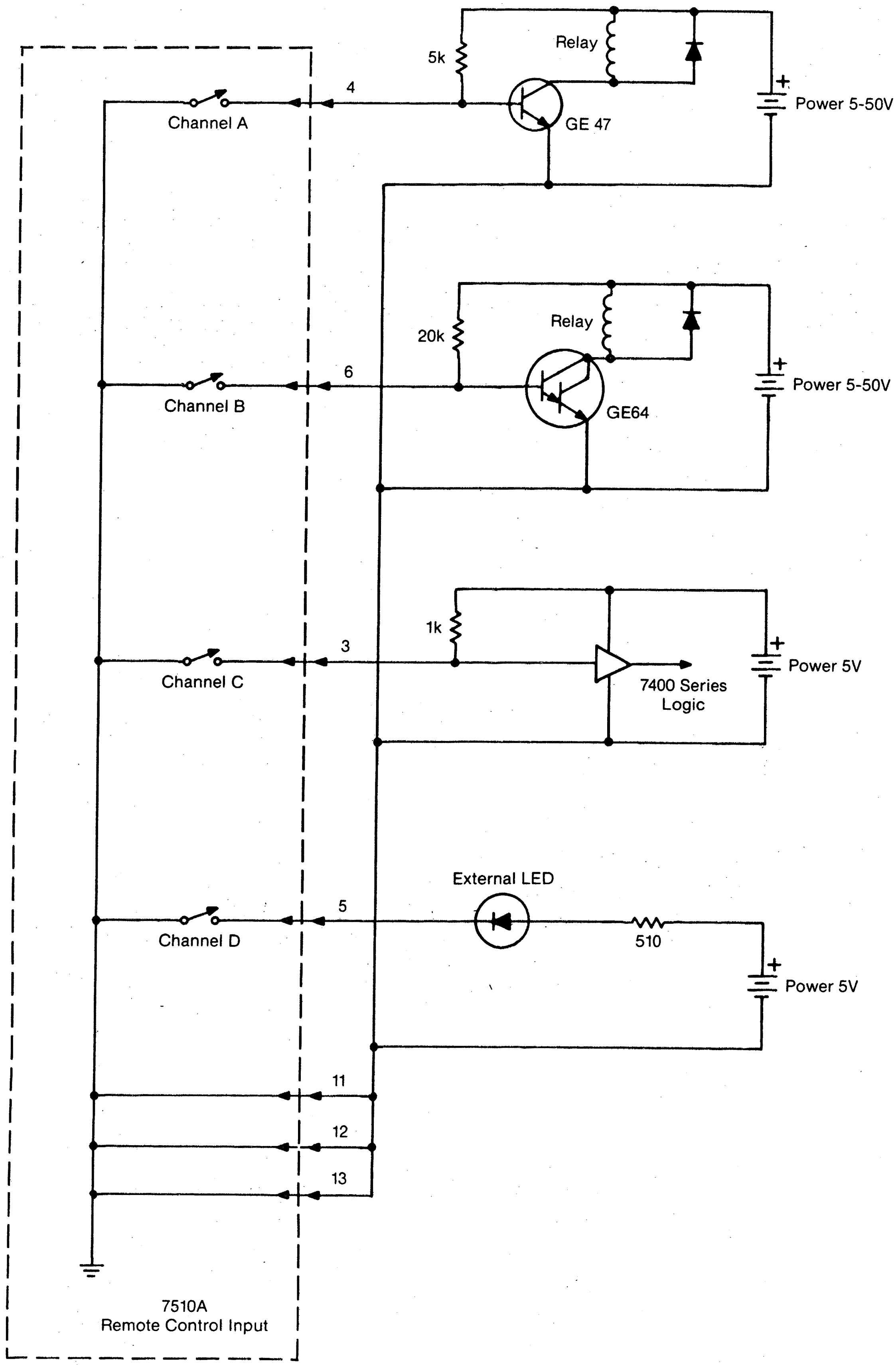


The following pins must be jumpered together for single unit operation:

Pin 1 to **Pin 2** | **Pin 5** to **Pin 6** | **Pin 12** to **Pin 13**
Pin 3 to **Pin 4** | **Pin 7** to **Pin 8** | **Pin 14** to **Pin 15**

NOTE: Connecting Pin 9 to Pin 10 switches the entire system to Priority. (Mutes all microphones in the audio mode.)

Figure 16. Circuits for Implementing the Logic Switching Function of the 7510A.



regardless of how many microphones are being used. The 7510A does not affect program dynamics so long as any given number of microphones are in use.

Another contrast between compression and the 7510A digital output attenuator can be seen when the background noise level is considered. With compressors, if microphones are being used, the overall gain can actually increase, and can increase the audible background noise and leakage. With the 7510A, if more microphones are being used, those microphone channels not in use are turned OFF, thus reducing the noise and leakage.

7 SPECIFICATIONS

Maximum Gain

Input Module	Adjustable, 15 dB to 64 dB (600 ohm load) Adjustable, 21 dB to 70 dB (10 kilohm load)
Output Module	7 dB (600 ohm or higher load)
Overall System	77 dB (600 ohm or higher load)

Main Output Characteristics

Actual Impedance	Less than 0.5 ohm 20 Hz to 20 kHz
Load Impedance	For 600 ohm or higher loads
Maximum Output Level	+24 dBm
Common Mode Output Impedance	10 kilohm

Direct Output Characteristics

Actual Impedance	600 ohm unbalanced
Load Impedance	For 600 ohm or higher loads
Maximum Output Level	+14 dBm (600 ohm load) +20 dBv (10 kilohm load)

Input Characteristics

Actual Impedance	Greater than 10 kilohm (20 Hz to 20 kHz unbalanced)
Source Impedance	0 ohm to 600 ohm
Nominal Input Level	Adjustable, 2.5 mV RMS (−50 dBv) to 0.7 V RMS (−1 dBv)
Input Overload	Adjustable, 45 mV RMS (−25 dBv) to 6.13 V RMS (+18 dBv)

Aux Input Characteristics

Actual Impedance	10 kilohm (± 5%) unbalanced
Source Impedance	0 ohm to 10 kilohm
Nominal Input Level	+4 dBv typical
Input Overload	7.75 V RMS (+20 dBv)

Frequency Response

Input Module	+0, −0.5 dB 20 Hz to 20 kHz
Output Module	+0, −0.5 dB 20 Hz to 20 kHz
Overall System	+0, −0.5 dB 20 Hz to 20 kHz

Total Harmonic Distortion

Input to Direct Output	0.02% maximum 20 Hz to 20 kHz at +20 dBv
Input to Main Output	0.02% maximum 20 Hz to 20 kHz at +24 dBm

Equivalent Input Noise

−130 dBv (0 dBv RE, 0.775 V RMS, 50 ohm source)

Automatic Mix Function

Input Attack Time	10 ms to 10 µs (½ waveform from 20 Hz to 20 kHz after signal exceeds set threshold)
Input Rise Time	30-60 ns (tolerance for turn-on once attack occurs)
Input Release Time	100 ms to 5 seconds, adjustable
Channel Attenuation	25.5 dB
Feedback Prevention	3 dB for each doubling of inputs

Controls

Channel Gain	Audio taper potentiometer
Master Gain	Audio taper potentiometer
Channel Threshold	Screw-adjustable linear taper potentiometer
Background Threshold	Audio taper potentiometer with CCW switch
Release Time	Screw-adjustable linear taper potentiometer
Mode Switch	3 position toggle switch
Power Switch	Push ON/Push OFF
Phantom Power Switch	Slide Switch
Voltage Selector Switch	Slide Switch

Indicators

Channel ON	Red LED
Power ON	Green LED
Output Level	VU meter; 0 VU = +4 dBm output

Power Supply

115 volts AC switchable to 230 volts AC, ± 10%, 50/60 Hz— Internal fuse protection

Connectors

Input	Neutrik NC3FD-V (Female 3 pin panel socket); mating connector—Neutrik NC3MC or Switchcraft A3M
Direct Output	Switchcraft 113BPC (1/4 inch phone jack); mating connector—Switchcraft 267 (3 conductor, ring-tip-sleeve)
Main Output	Neutrik NC3MD-V (male 3 pin panel socket); mating connector—Neutrik NC3FC or Switchcraft A3F
Remote Control	AMP HD-20 (15 pin female panel socket); mating connector—AMP 205735-6 or TRW DA-15 P
Interface	AMP HD-20 (15 pin female panel socket); mating connector—AMP 205735-6 or TRW DA-15 P
AC Power	Panel components 8843.ZP. 30. 60; Mating AC line cord—Switchcraft P-2392

Dimensions

133 mm high X 483 mm wide X 283 mm deep,
5 1/4 in. high X 19 in. wide X 11 1/8 in. deep

Weight

Mainframe with 1 input Module (4 channels) and Output Module	Net 6.4 kg (14 lb, 3 oz) Shipping 8.2 kg (18 lb, 1 oz)
1 input Module (4 channels)	Net 0.8 kg (1 lb, 11 oz) Shipping 1.3 kg (2 lb, 14 oz)
1 output Module	Net 2 kg (4 lb, 7 oz) Shipping 2.7 kg (5 lb, 14 oz)

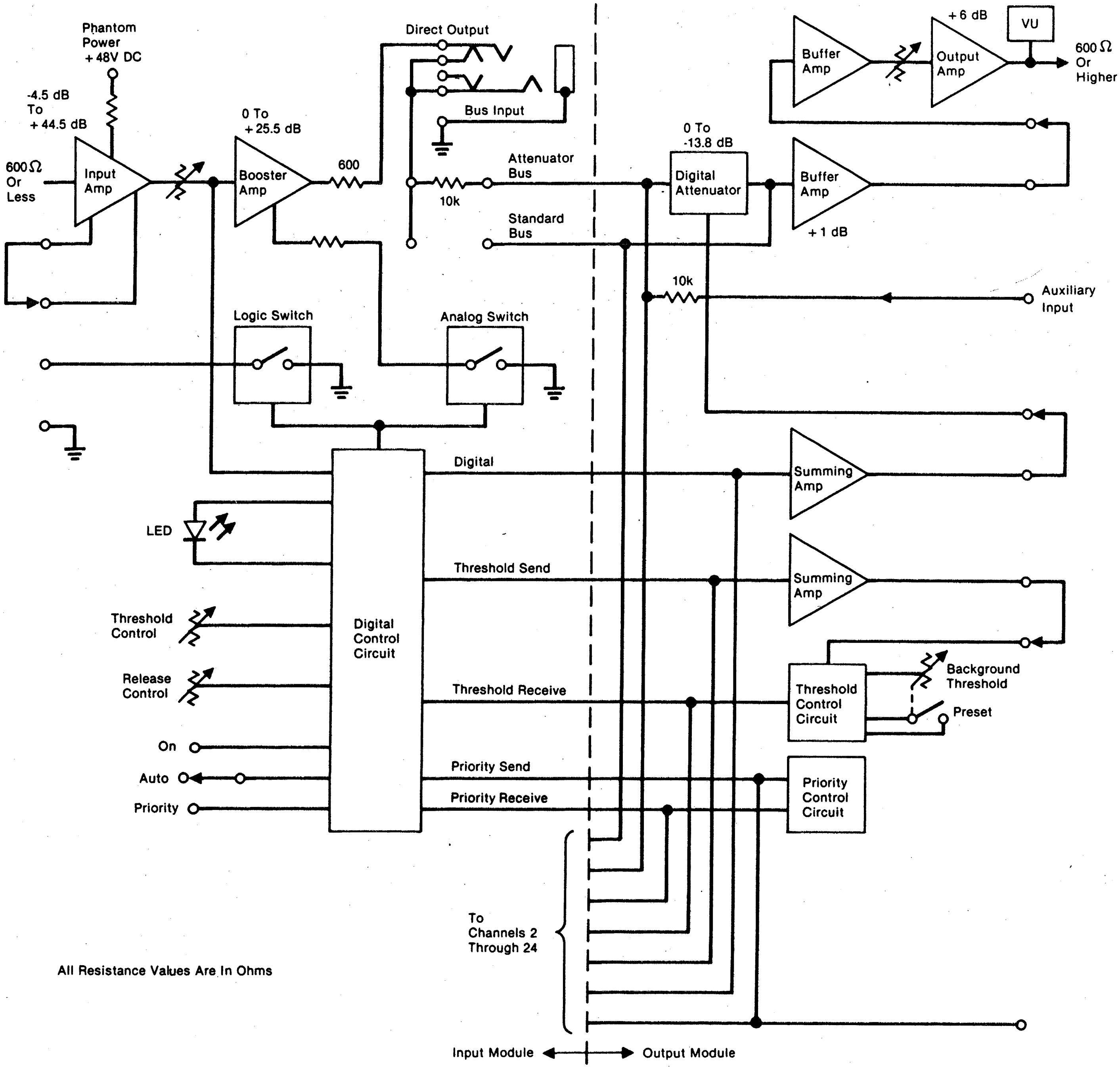
Mounting

Occupies 3 EIA standard rack spaces

Panel Finish

Textured semi-gloss baked enamel, dark grey; white nomenclature

Figure 18.



All Resistance Values Are In Ohms

8 MAINTENANCE/SERVICE

TROUBLESHOOTING CHART

Symptom	Recommendation
The beginning of words or musical notes tends to be cut off on a given input channel in Auto mode.	Set the channel's Threshold for more sensitivity (turn screw CW).
You must talk or play very loudly in order to turn the channel On in Auto mode.	Channel Threshold may be too high (turn CW). If symptom noticed on several channels, Background Threshold may be set too high.
The end of words or musical notes tends to be cut off on a given input channel in Auto mode.	Channel Release time may be too fast (turn screw CW).
Channels in Auto or Priority mode turn On spontaneously when their microphones are not directly in use.	Background Threshold set too low (turn CW).
One or more channels appear to be On, yet their signals do not appear at the Main Output	The signals are exiting via the Direct Outputs and are not returned to the Auxiliary Input, or the Direct Out phone jacks' normalling switches are not making contact internally.
All input channels appear to work normally, as indicated by the LEDs, but the Main Output is dead.	Check to insure Master Gain is set properly. If so, check the Direct Output jacks for signal. If OK, the output module requires service.
Sound at Main Output is distorted regardless of Master Gain setting.	Channel Gain settings are all too high. Output module is defective (also see below). Possible oscillation/ground loop (also see below).
Sound from a given input channel is very distorted at any setting of channel Gain and Master Gain, whether monitored at the Main Output or the Direct Output.	Input signal is too high; adjust the preamplifier Gain — replace input module.
Sound system goes into feedback during live performance/show/meeting/ etc., even though setup proved howl-free.	The presence of more people changed the room acoustics and/or some-one turned up the amplifier gain or moved the microphones.
Fuse blows repeatedly as soon as 7510A is turned On, or shortly thereafter, whether or not the unit is connected to other sound equip.	Mains voltage is too low or high—check specifications and measure the mains voltage; or fuses are of incorrect value; or 7510A power supply is defective.
Fuse blows only when 7510A is connected to other sound equipment. May be associated with distorted sound prior to fuse failure.	Possible ground loops or feedback paths due to improper cable routing, (causing oscillation), or very high voltage applied to 7510A.

*Never use a fuse rated higher than the specified current, which depends on the mains voltage for which the unit is wired.

WARNING THIS SECTION OF THE MANUAL CONTAINS SERVICE INSTRUCTIONS FOR USE BY QUALIFIED SERVICE PERSONNEL ONLY.

The sophisticated logic circuitry in the JBL 7510A is extremely reliable, comprised entirely of solid state integrated circuits. Keeping a spare input module handy is a moderately priced insurance policy. Should a problem develop, the offending module can be returned for factory repair, and the spare module can be installed, immediately restoring normal operation.

In the event of a problem in the power supply/output section of the 7510A, contact the JBL factory for a return authorization or the name of the nearest authorized 7510A service facility.

No regular maintenance is required for the 7510A. The painted surfaces may be cleaned using a sponge dampened with mild detergent and water solution, abrasives and strong detergents should be avoided. The meter face may be cleaned with a soft cloth and a plastic cleaner/polish such as Mirro-Glaze.

9 INSTALLATION OF 7510-02 INPUT MODULES

The 7510A is supplied with 4 input channels as standard equipment, (i.e., one input module) and can be expanded in increments of 4 input channels up to a maximum of 24 inputs. Whenever one or more input modules are added to the 7510A, two procedures are required: (a) Physical installation of the module or modules in the mainframe, and (b) readjustment of the internal trimmer (RT2) for the low level background threshold circuitry. (If a defective module is being replaced and input capacity is not being expanded, the internal trimmer (RT2) requires the same adjustment as required when adding additional modules.)

CAUTION: Whenever installing a module, make sure the AC power is first turned OFF and the AC power cord is unplugged from the mains.

PHYSICAL INSTALLATION:

1. Unscrew the 16—#6-32 machine screws from the 7510A top cover and lift cover from the unit.

2. Remove the one #6-32 machine screw which holds the bottom of the rear filler plate and save for installation of the new module. The plate may be discarded unless future removal of the input module is contemplated.

3. Remove the two nuts and associated washers which secure the front cover plate to the front panel, and withdraw the cover plate. Notice that the 7510A front panel has already been labeled to accommodate the new input module. The cover plate, nuts and washers may be discarded unless future removal of the input module is contemplated.

4. Temporarily remove the 4 outer nuts from the toggle switches and the 4 knobs from the gain controls that have been installed on the new input module so the bushings can be inserted through the 7510A front panel. Lower the new input module into the chassis, sliding it forward so the gain controls, switch shafts and LED's protrude from their respective front panel holes. (Note: Make sure LED leads DO NOT touch front panel). Install the 4 nuts on the toggle switches to secure the new input module to the 7510A front panel. (Note: DO NOT overtighten toggle switch nuts, as the toggle switch bushings will turn and short against the printed circuit board, causing the input module to operate incorrectly). Then, using the one #6-32 machine screw saved from the rear filler plate, secure the back of the new input module to the chassis bottom cover.

5. Attach the flat ribbon cable to the new input module.

6. Install the gain control knobs, making sure the knobs are aligned to the front panel markings at maximum CW and CCW rotation.

INSERTION OF A NEW INPUT MODULE

One filler plate occupies the rear panel space and one cover panel occupies the front panel space for each available input module position. The filler plate and cover panel must be removed to make room for the module to be added.

ADJUSTMENT OF LOW LEVEL BACKGROUND THRESHOLD:

Whenever the 7510A input capacity is expanded with one or more new input modules or an existing input module is changed, the low level background threshold trimmer (RT2) on the 7510A output module must be recalibrated as follows:

Note: While they affect the same circuitry, the external background threshold control and the internal threshold trimmer (RT2) perform different functions, so re-setting the external control cannot substitute for this trim adjustment procedure.

1. Set all input channels to auto mode.

2. Set all input channel gain controls to "0" (maximum counterclockwise rotation).

3. Set input channels 1 & 2 front panel release time control for the fastest response (maximum counterclockwise rotation).

4. Set input channels 1 and 2 front panel threshold control for maximum threshold (maximum clockwise rotation).

5. Set master gain control at "5".

6. Set background threshold on "1" (maximum counterclockwise rotation but not switched to preset).

7. Feed a 1000 Hz, − 50 dBv signal from an external generator to input channel 1 and adjust its gain control until the VU meter reads "0" VU.

8. Disconnect signal from input channel 1 and connect it to input channel 2. Adjust its gain control until the VU meter reads − 3 VU.

9. While simultaneously feeding the 1000 Hz, − 50 dBv signal to both input channels 1 and 2 (gain controls still set as in steps 7 and 8) turn the external background Threshold control clockwise until both inputs shut off.

10. Remove input signal from both input channels 1 and 2 and turn both gain controls to "0" (maximum counterclockwise rotation).

11. Using a high impedance digital voltmeter, connect its positive lead (+) to Pin 14 or IC8 (LM324N) and its negative lead (−) to chassis ground. Adjust RT2 for a reading of + 5mV. Note: IC8 is located directly above and towards the rear from trimpots RT1 and RT2 on the 7510A Output Module. Pin 14 of IC8 is the top pin toward the VU meter.

12. Should a high impedance digital voltmeter not be available, the low level background threshold in step #11 may be set by turning RT2 to the setting just above (clockwise) the point where all LED's shut off. Should this procedure be used, all front panel threshold controls must be set at maximum threshold (maximum clockwise rotation) and all input channel gain controls at minimum gain (maximum counterclockwise rotation).

ADJUSTMENT OF PRESET THRESHOLD RT1:

1. Set all input channels to auto mode.

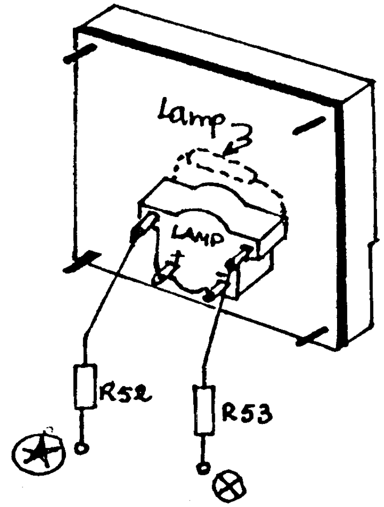
2. Set all input channel gain controls to "0" (maximum counterclockwise rotation).

3. Set the Background Threshold to its switched preset position (maximum counterclockwise rotation).

4. Using a high impedance digital voltmeter, connect its positive lead (+) to Pin 14 of IC8 (LM324N) and its negative lead (−) to chassis ground. Adjust RT1 for a reading of .3 volts.

NOTE: Installation of 7510-02 Input Module in a JBL 7510 Automatic Mixer, instead of a 7510A Automatic Mixer, requires changing resistor R29 on the 7510-02 Input Module 1000 ohms (Brown, Black, Red) to 2200 ohms (Red, Red, Red). Resistor R29 is the third resistor on the top, from the ribbon connector (S1), toward the front panel.

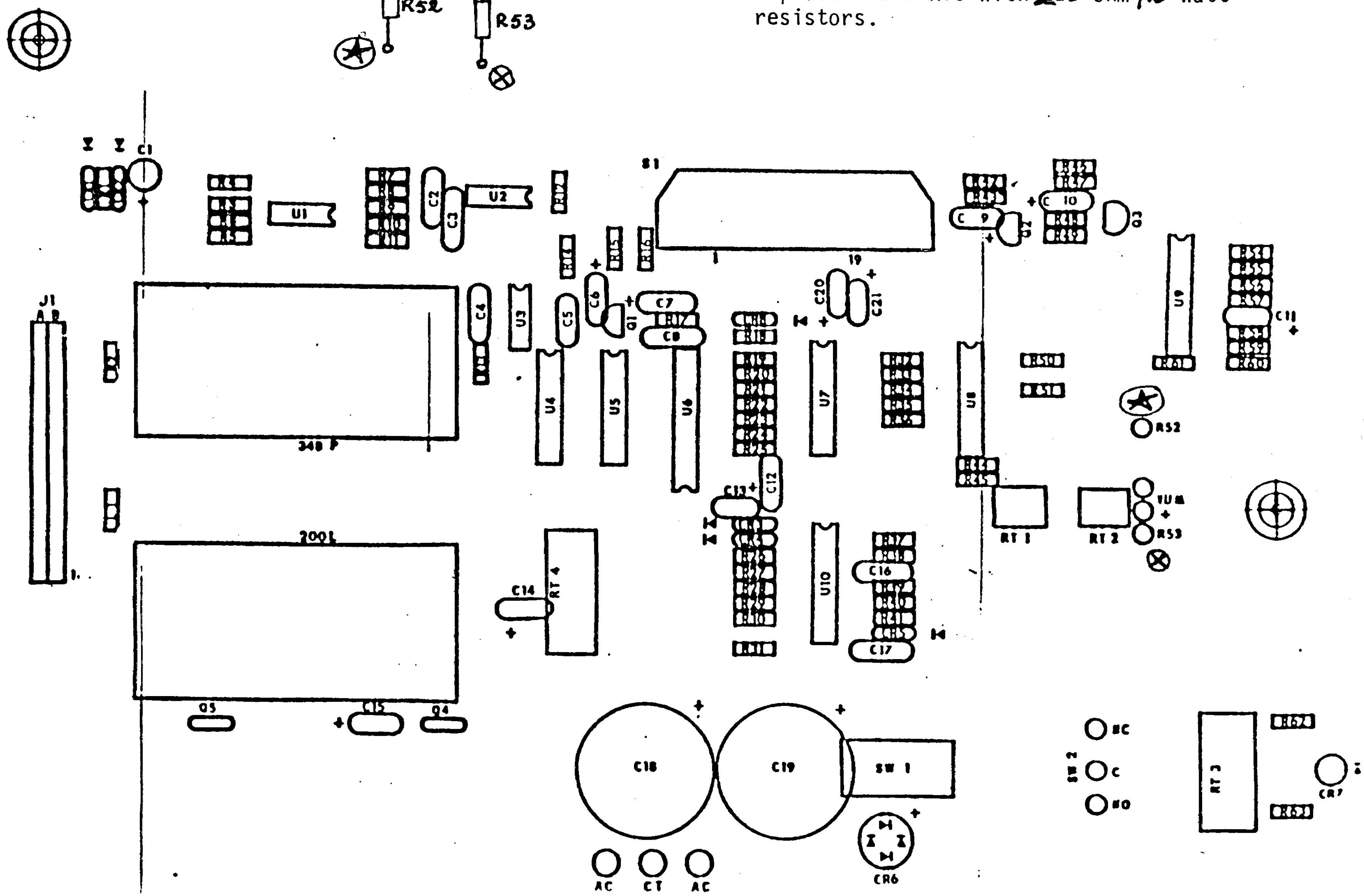
7510A/B METER LAMP REPLACEMENT



1. Remove cover from meter by firmly gripping from buttom to top and giving a gentle tug.

2. Cut the leads off the old lamp from their studs. Wrap and solder leads of new lamp (JBO-140) to their studs, replace cover.

3. Remove R52 &R53, (150 ohm .5 watt) and (150 ohm 1 watt), respectively, in 7510B. Replace R52 & R53 with 200 ohm 1.0 watt resistors.

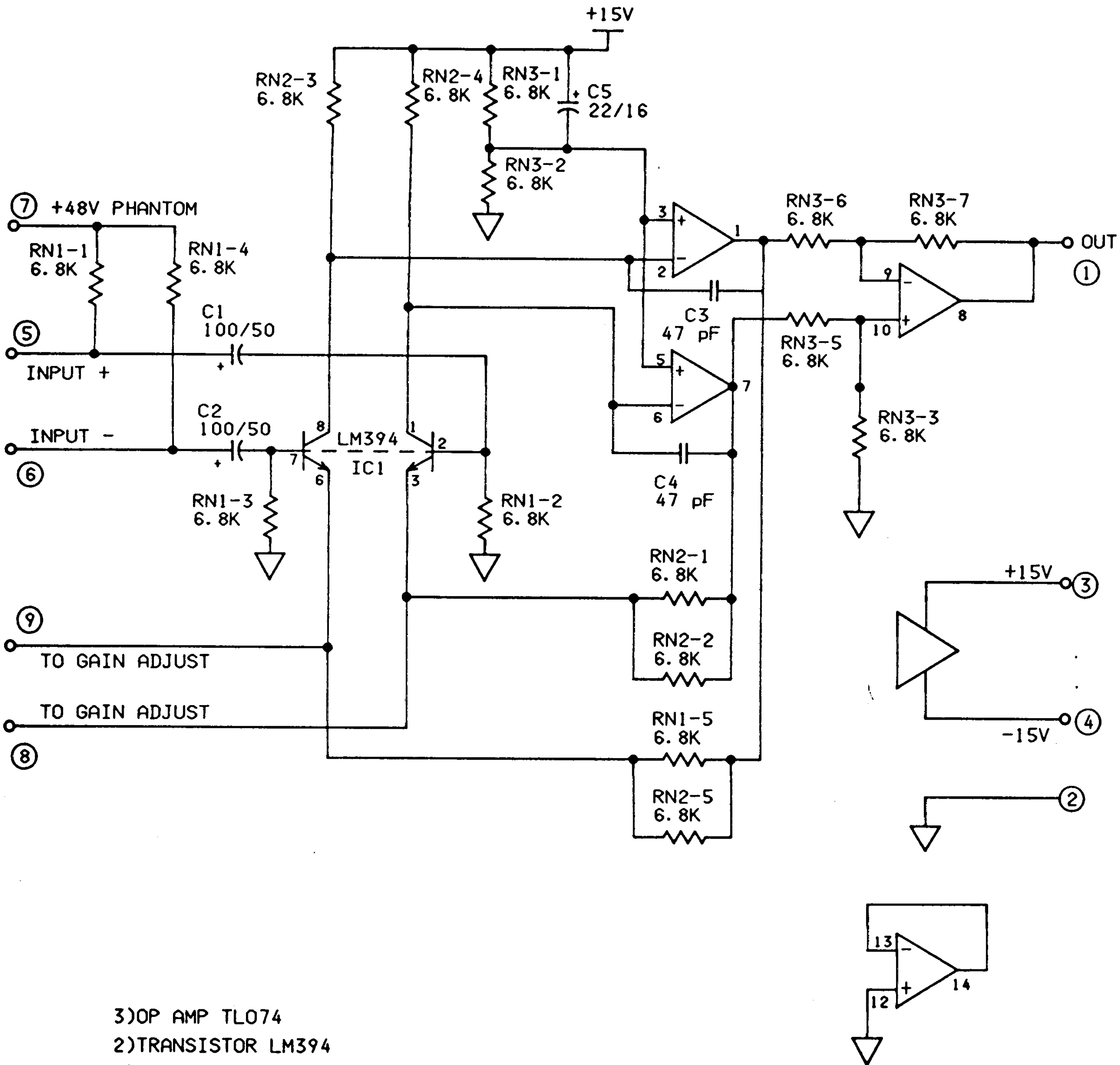


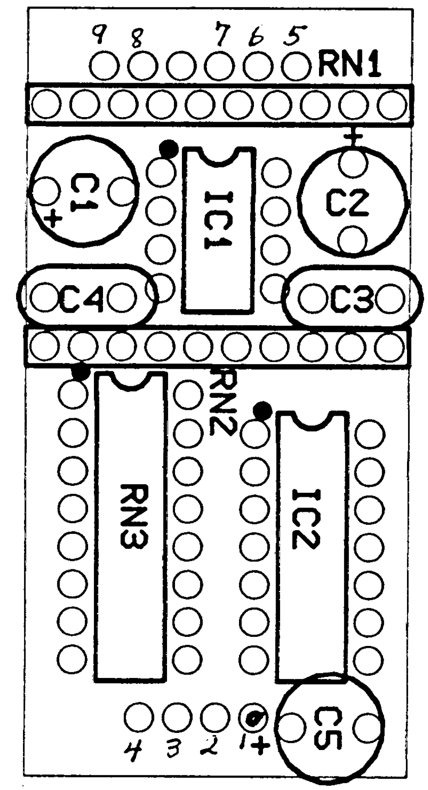
7510A Retrofit
Assembly pin cutting instructions

The model 7510A Automatic Microphone mixer has been fitted with input modules and output modules designed for the model 7510B. To ensure correct assembly the following instructions must be followed.

Output Amplifier Module
Assemble board number 15975 (part number 10-15797) pins 1, 8, 10 must be cut, for pin reference check schematic (drawing number 16030).

Mic Preamp Assemble
Board number 15407 (part number 10-15785) pin between 7 and 8 should be cut, for pin reference check schematic (drawing number 15993).





PCB#12-15407

ASSY#10-15785

3)OP AMP TL074

2)TRANSISTOR LM394

1)ALL RESISTORS 1/8W IN R PACKS

NOTES:UNLESS OTHERWISE SPECIFIED

UREI	SCHEMATIC MIC PREAMP
DO-239	
7510B-03	C 15993 D
SHEET	1 of 1
DRAWN	GM 3-9-88

IMPORTANT NOTE

This model 7510A Automatic Microphone Mixer has been fitted with the output amplifier module desighed for the Model 7510B. To ensure stable operation the 7510-02 Input Assembly has been modified with the addition of four capacitors on the underside of the PC board near the Gain controls.

If you should ever install additional 7510-02 Input Assemblies, or change the assembly which is already in the unit, these capacitors need to be added to the new Assembly. The capacitors are .0033 microFarad. Use the original 7510-02 Input Assembly as a guide fo location of the parts.

If you have any questions or problems associated with this modification, contact UREI Customer Service.

JBL Incorporated, 8500 Balboa Boulevard, P.O. Box 2200,
Northridge, California 91329 U.S.A.

INSTALLATION OF 7510B-STYLE OUTPUT MODULE IN 7510A

GENERAL

The output module for the 7510B may be installed in the output assembly of a 7510A to replace a defective JED 200 output module. Because of the construction differences between the 7510A and the 7510B, and between runs of 7510As, there are some variations in the ways in which the replacement must be done. Please read and understand this instruction sheet fully before beginning the job.

MODULE REMOVAL/INSTALL

Unplug the AC power cord from the 7510A and remove the unit from the rack. Remove the output assembly from the 7510A. This requires removing the top cover, two screws holding the output rear panel to the bottom cover, two screws in the bottom cover into the voltage regulator heatsink, the two knobs on the front panel and, depending on when your output assembly was built, a screw holding the toroid power transformer to the side panel.

Unplug the ribbon cable connector where it plugs into the output assembly. Remove the output assembly from the 7510A.

Unsolder the output module from the printed circuit board. Do this very carefully, using a solder vacuum or solder wick to avoid lifting PC pads and/or pulling the copper out of the inside of the holes. Remove the module.

The new module has been designed with an extra buffer amplifier to provide an auxiliary output for feeding a tape recorder. The added amplifier requires extra input and output pins which are not usable on a 7510A, therefore these pins have been cut off at the factory. The module will now plug directly into the holes left by the previous module. Place the module into position. If you have accidentally pulled out the copper from within any of the plated thru holes where there is a trace on both sides of the PCB, you must achieve continuity between both sides during the soldering operation. This is most easily done by elevating the module slightly, rather than letting it sit fully down on the PCB. This will allow the pin to be heated from below and solder applied on each side to connect the pin both top and bottom.

OUTPUT TRANSFORMER MOUNTING

The new output circuit incorporates an output transformer to provide a fully floating output. The transformer must be mounted to the chassis of the 7510A. Early production units used a power transformer mounted in a metal can attached to the rear panel of the output assembly, or a toroid attached to a metal bracket which was attached to the rear panel. If your unit is of this type, proceed as follows:

Place the output assembly back into the mainframe of the 7510A temporarily. Place the output transformer against the inside-side of the 7510A, with the wire leads to the rear and one mounting hole above the other. Place the transformer about as far forward and toward the top as you can get it. Mark the top hole location. Remove the transformer and output assembly and drill a hole for a 6-32 or 8-32 mounting screw. Using the transformer as a guide on the outside of the mainframe, locate, mark and drill the second mounting hole. Be very careful not to get drill chips into any of the circuitry, to deburr the holes and clean up any bits of metal from the inside of the chassis following drilling.

If the power transformer in your unit is a toroid, mounted with a single screw to the side panel, the holes for the original power transformer still exist on the rear panel. There are screws and nuts in them. Mount the transformer using the top screw. One screw should be sufficient for most fixed installations. A second hole may be drilled if you believe your installation needs it.

WIRING THE OUTPUT TRANSFORMER

The wires from the transformer are terminated in a 4-conductor polarized connector which plugs into a header at the rear of the new output module. The wire must be routed between the main PC board of the output assembly and the connector board at the rear. Unscrew the screw through the rightangle bracket which joins these two boards at the top of the assembly, slide the transformer wires between the two boards, reinstall the screw and plug the connector into the mating connector on the output module.

Reinstall the output assembly into the mainframe. Make sure the power LED slides into the green diffusor on the front panel. (If the output transformer needs to be attached to the side panel, do it now). Reinstall the screws holding the output assembly in place.

INPUT MODULE MODIFICATIONS

Install capacitors on the input assemblies according to the directions and drawing in the attached Important Note.

Inspect your work, and put the unit back together. The job is done. None of the work that you have done should require any re-adjustments to any control settings.

97-0033

PROCEDURE FOR REPLACING 348P MODULE WITH NEW UREI PHANTOM MODULE IN 7510A

1. Remove 348P module. Try very hard not to pull out the "plate-thru's" inside the holes of the double-sided board. Several of the holes are connected on both top and bottom side and are used as vias for the connections.

2. Remove the zener diode and inductor shown on Figure 1. Replace the inductor with an insulated wire jumper.

3. Install the new phantom power supply assembly. If you have accidentally pulled any of the "plate-thru's" do not fully seat the module. Leave a little room under the 5 pin end so that you can solder the pins to both top and bottom sides of the board to complete the via connection.

4. Install an insulated jumper wire as shown in Figure 2. The route drawn is approximate. Slight deviations should not be a problem.

5. Route the two wires from the phantom supply board under the shaft of the gain control, twist the wires together and solder one to each of the two back terminals of the power switch.

6. Inspect your work. Before reinstalling the output assembly, power it up and check for 48 VDC $\pm$ 1 V output at the location marked in Figure 2.

97-0034

FIGURE 2

INSTALL INSULATED JUMPER WIRE AS SHOWN

348P REPLACEMENT

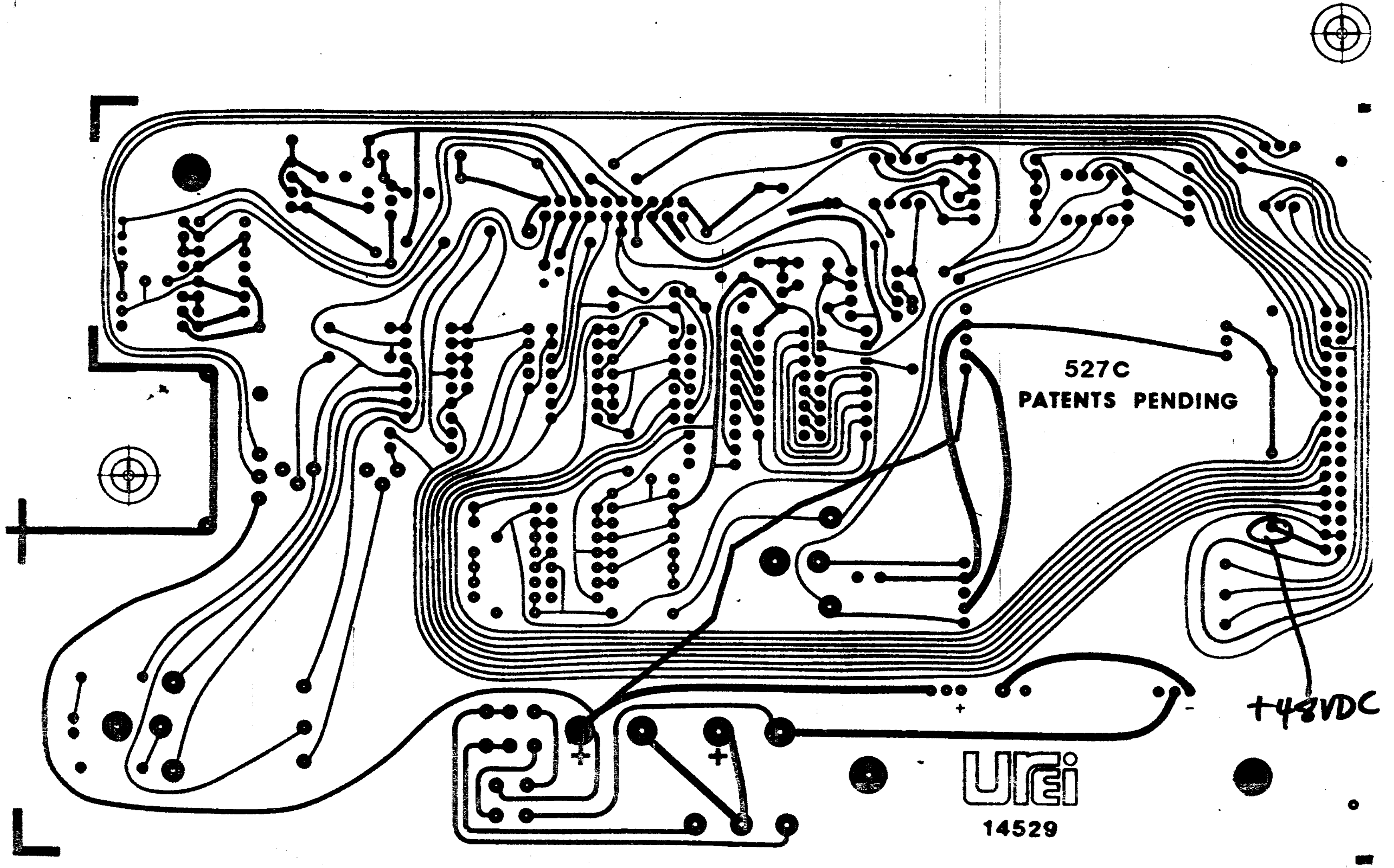
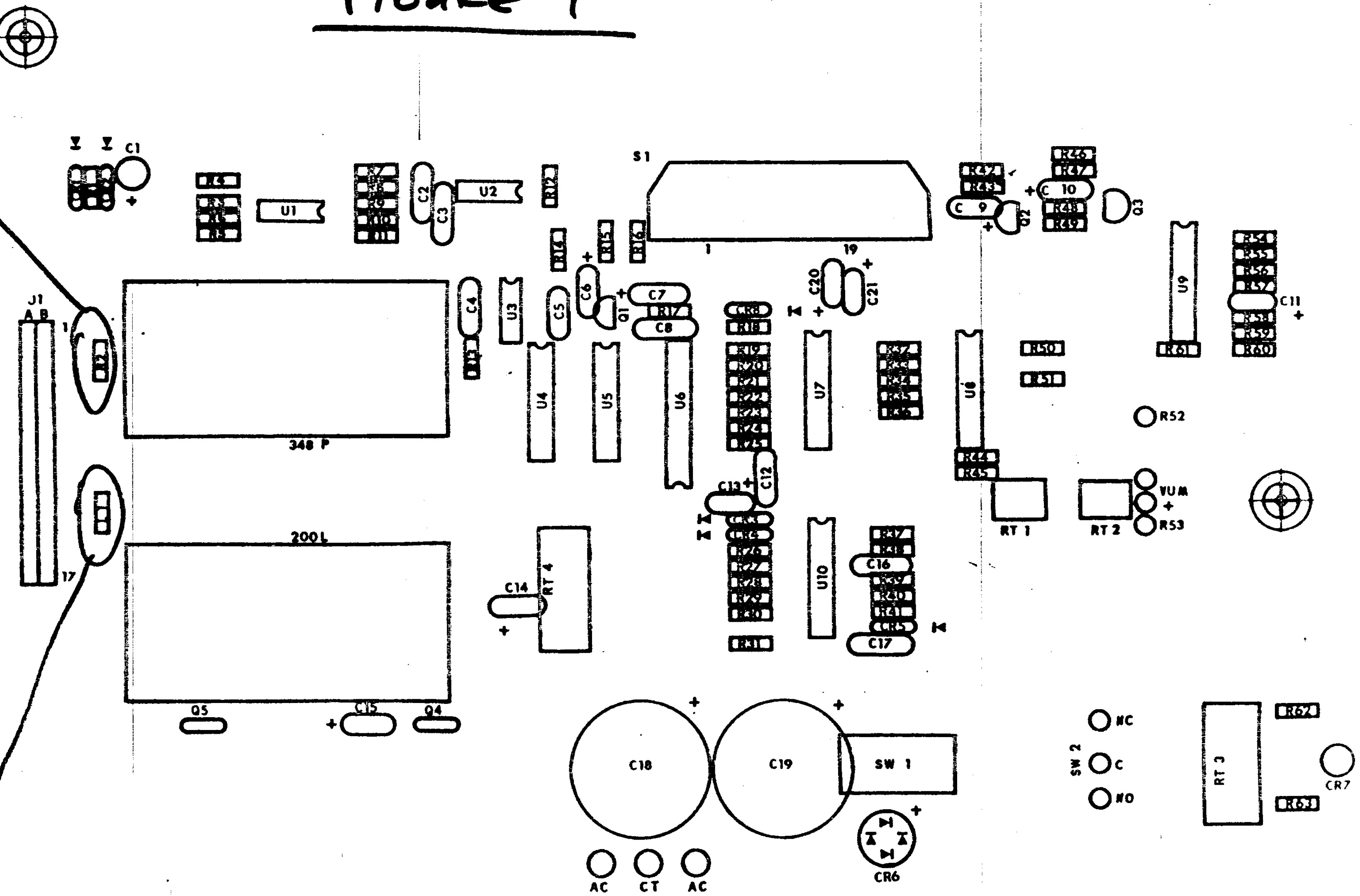


FIGURE 1

348P REPLACEMENT

REMOVE ZENER DIODE



REMOVE INDUCTOR (LOOKS LIKE 2W RESISTOR) REPLACE WITH INSULATED JUMPER WIRE

IMPORTANT NOTE

This output assembly for the 7510A Automatic Microphone Mixer includes the output amplifier module designed for the 7510B. To ensure stable operation with 7510-02 input modules, four capacitors must be installed on the underside of each 7510-02.

The accompanying drawing shows the circuit traces on the underside of the 7510-02. Install a .0033 microFarad capacitor at each of the four locations shown.

If additional input modules are installed at a later date, the same modification must be done to them. We recommend placing a copy of this note inside the 7510A housing to alert future service personnel to this need.

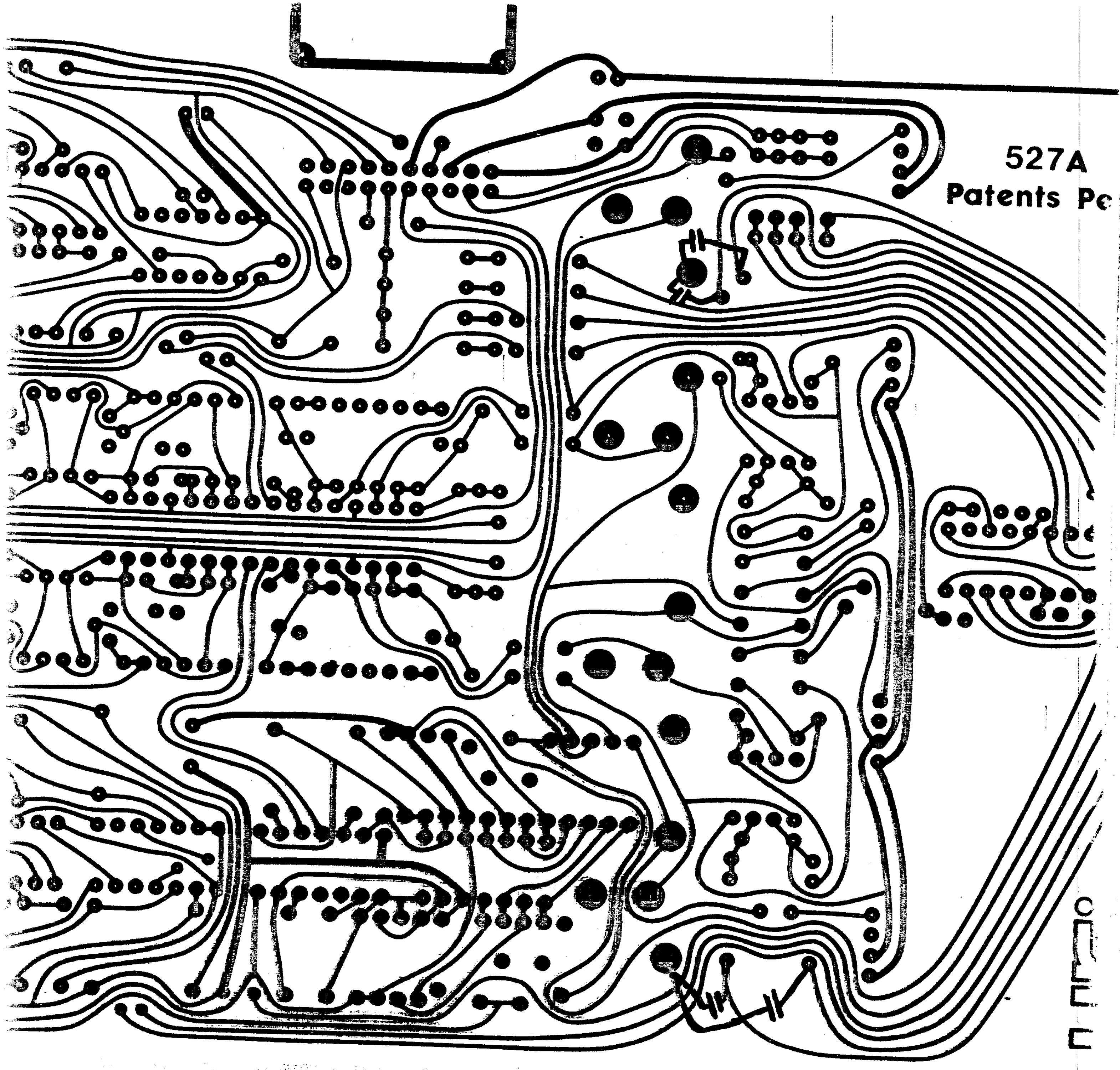
If you have any questions or problems with this modification, contact UREI Customer Service.

97-0035

Universal Audio Professional Products

TELETRONIX INFORMATION SYSTEMS

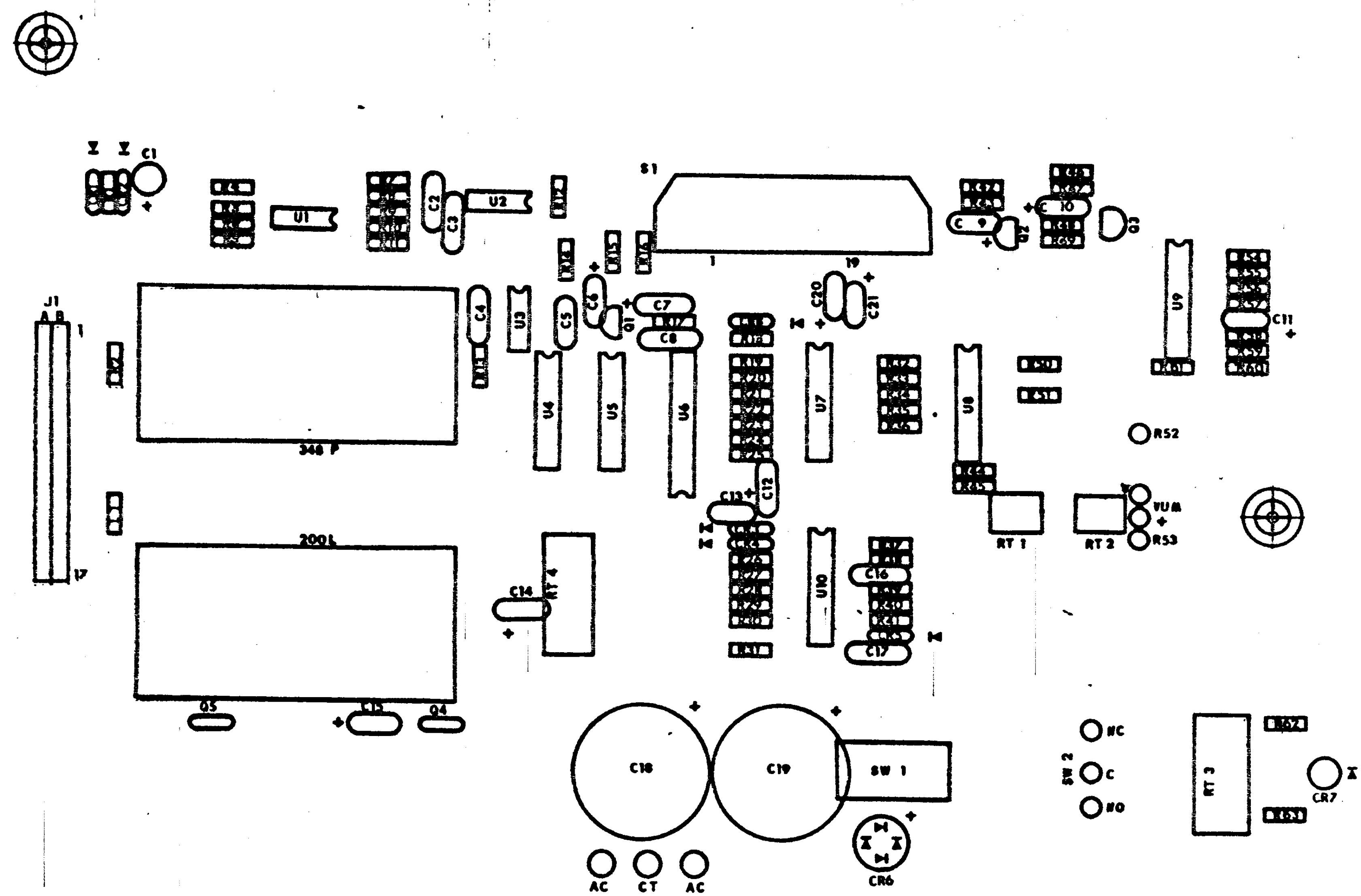
8460 SAN FERNANDO ROAD • SUN VALLEY, CA 91352-3292 • (818) 767-1000


527A
Patents Pe

INSTALLATION OF NEW INPUT MODULES IN JBL 7510A

The input modules UREI #10-15980 (PCB #15407) may directly replace the IED input modules used on the 7510A. Unsolder and remove the old module and solder in the replacement. Use a solder vacuum or solder-wick*, and be extremely careful as you remove the old module. There are several connections made from the component side of the board and, in some cases, the holes are also used as VIAs from one side of the board to the other. Try not to pull out the plating inside the holes as you remove the old module. If you do, the best method to re-connect the circuits is to install the new module so that it is not fully seated to the main PCB. Then, apply solder to the pins on both the solder side and component side where necessary to complete circuits.

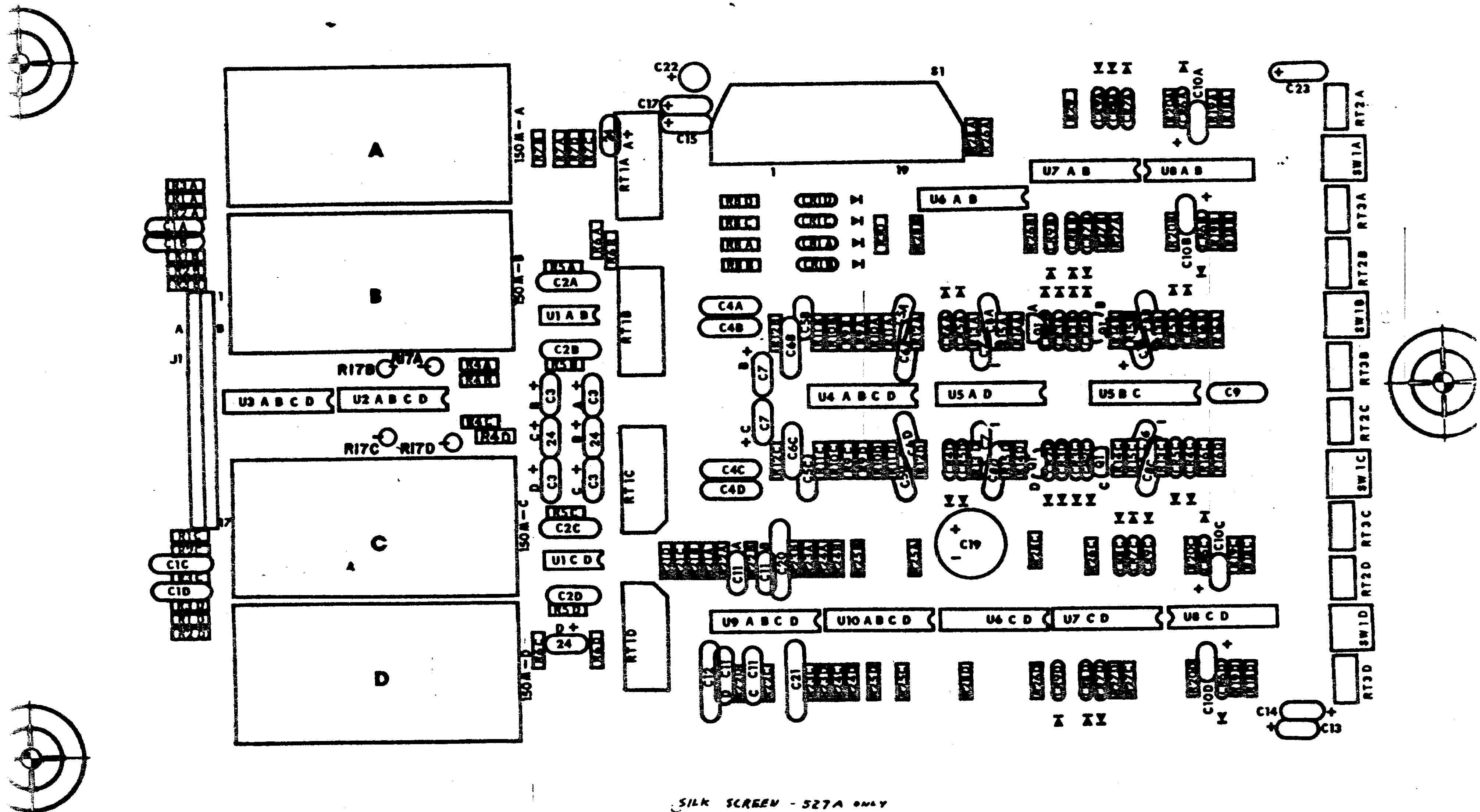
The new preamp has 2-3dB more gain than the old one. This should not generally be a problem but, if you want to make them exact, change R1A-R1D from 10 ohm to 27 ohm. These resistors are located between the modules and the back panel.



SILK SCREEN 537C ONLY
COMPONENT SIDE
7510A 6/13/83

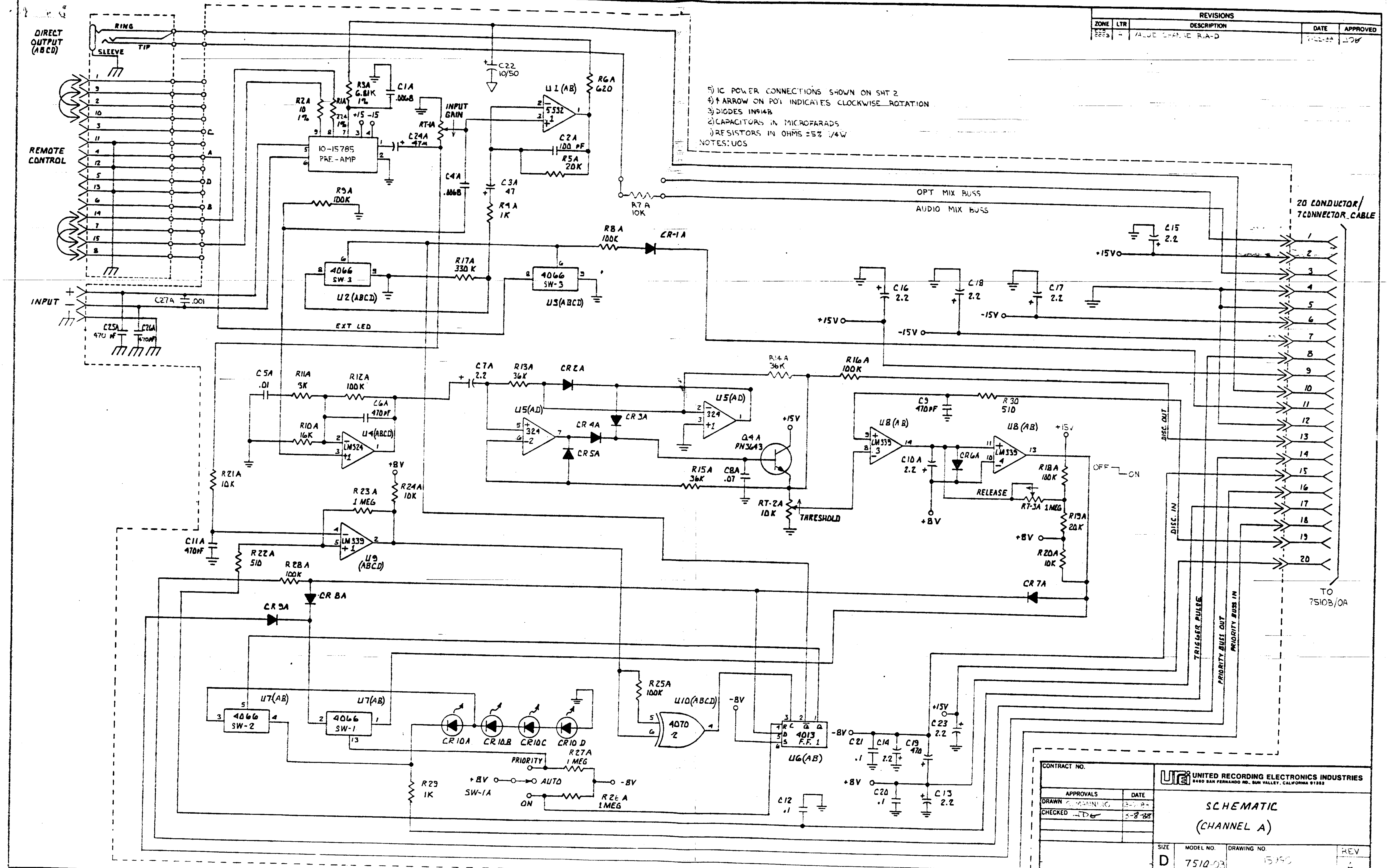
1 50% neg.

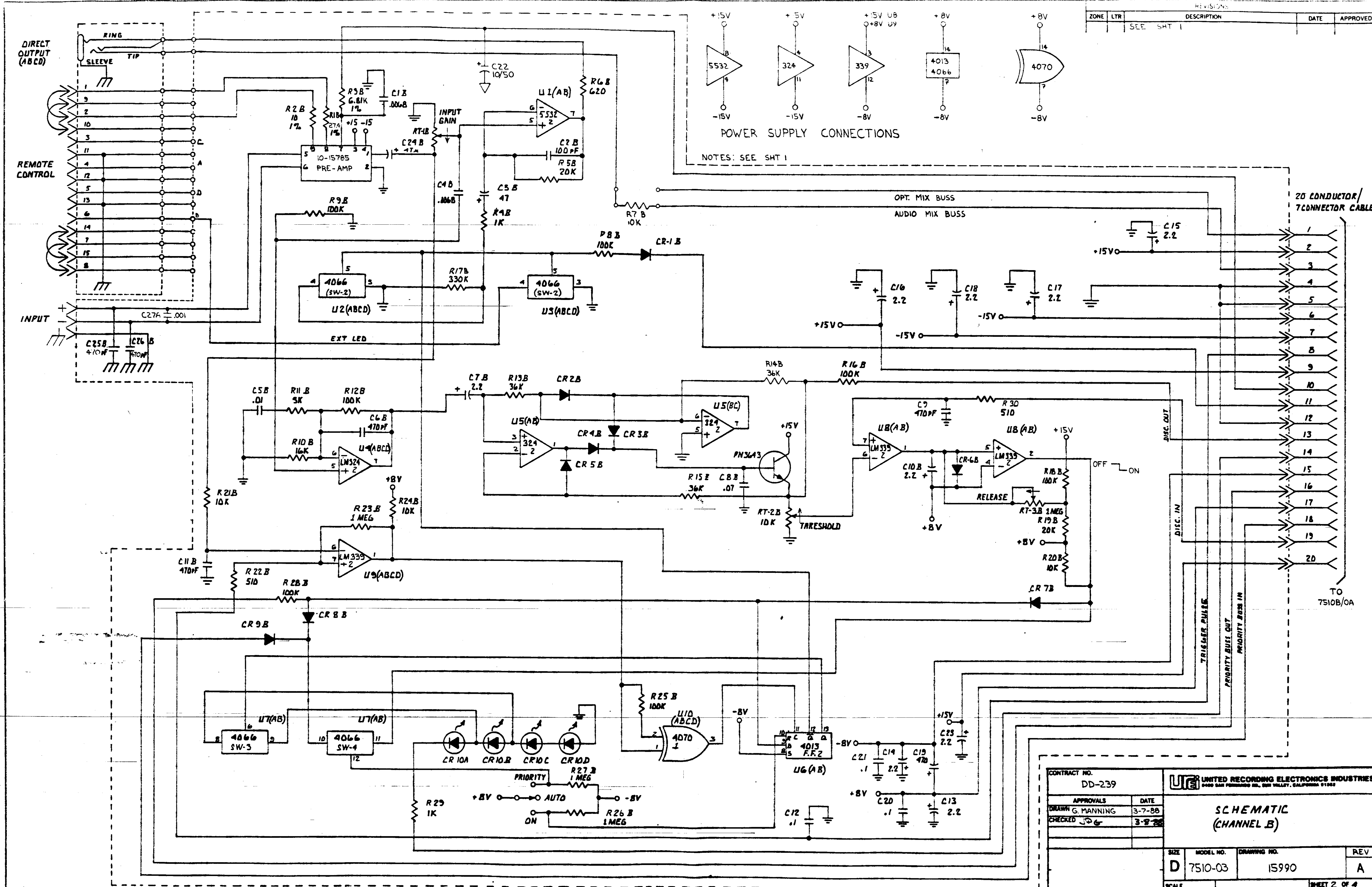
UREI A/W 14529 REV.C SHT. 4



SILK SCREEN - 527A ONLY
COMPONENT SIDE

UREI A/W 14532 SHT. 4 REV. B


REVISIONS
ZONE | LTR | DESCRIPTION | DATE | APPROVED
A | VALUE CHANGE R1A-D
5) IC POWER CONNECTIONS SHOWN ON SHT 2
4) ↑ ARROW ON POT INDICATES CLOCKWISE ROTATION
3) DIODES IN914B
2) CAPACITORS IN MICROFARADS
1) RESISTORS IN OHMS ±5% 1/4W
NOTES: UOS
DIRECT OUTPUT (ABCD)
RING
TIP
SLEEVE
REMOTE CONTROL
INPUT
C27A .001
C25A 470 pF
C26A 470pF
10-15785 PRE-AMP
R2A 10 1%
R1A 274 1%
R3A 6.81K 1%
C1A .0068
+15 -15
INPUT GAIN
RT-1A
C24A 47M
C4A .0068
C22 10/50
U1 (AB) 5532
R6A 620
C2A 100 pF
R5A 20K
C3A 47
R4A 1K
R9A 100K
R7A 10K
OPT MIX BUSS
AUDIO MIX BUSS
20 CONDUCTOR/ 7CONNECTOR CABLE
R8A 100K
CR-1A
4066 SW-3
U2 (ABCD)
R17A 330K
U3 (ABCD)
EXT LED
C15 2.2
+15V
C16 2.2
C18 2.2
C17 2.2
-15V
C5A .01
R11A 3K
R12A 100K
C6A 470pF
R10A 16K
U4 (ABCD) LM324
C7A 2.2
R13A 36K
CR2A
U5 (AD) 324
CR3A
CR4A
CR5A
R14A 36K
R16A 100K
Q4A PN3643
R15A 36K
C8A .07
RT-2A 10K
THRESHOLD
C9 470pF
R30 510
U8 (AB) LM339
C10A 2.2
CR6A
RELEASE
RT-3A 1MEG
R18A 100K
R19A 20K
R20A 10K
+8V
OFF ON
DISC. OUT
DISC. IN
TRIGGER PULSE
PRIORITY BUSS OUT
PRIORITY BUSS IN
TO 7510B/0A
R21A 10K
R23A 1 MEG
R24A 10K
C11A 470pF
U9 (ABCD) LM339
R22A 510
R28A 100K
CR8A
CR9A
CR7A
U7 (AB) 4066 SW-2
U7 (AB) 4066 SW-1
CR10A CR10B CR10C CR10D
R25A 100K
U10 (ABCD) 4070
-8V
U6 (AB) 4013 F.F. 1
C21 .1
C14 2.2
C19 470
C23 2.2
C20 .1
C13 2.2
C12 .1
PRIORITY
R27A 1 MEG
R26A 1 MEG
R29 1K
SW-1A
AUTO
ON
CONTRACT NO.
APPROVALS
DRAWN
CHECKED
DATE
3-8-88
UREI UNITED RECORDING ELECTRONICS INDUSTRIES
SCHEMATIC (CHANNEL A)
SIZE D
MODEL NO. 7510-03
DRAWING NO.
REV



ZONE	LTR	DESCRIPTION	DATE	APPROVED
		SEE SHT 1		

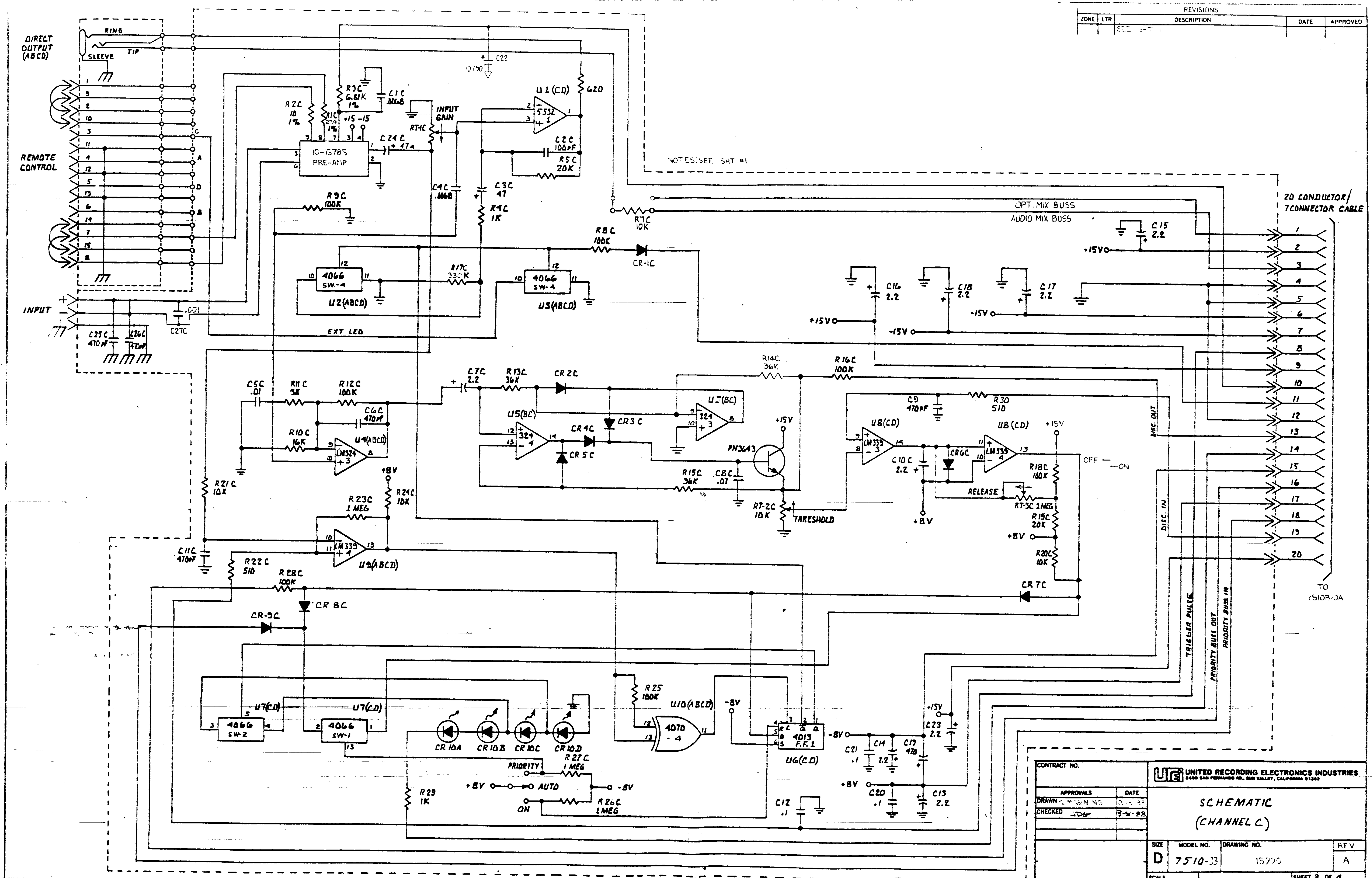
CONTRACT NO.	DD-239
DRAWN	G. MANNING, 3-7-88
CHECKED	3-8-88

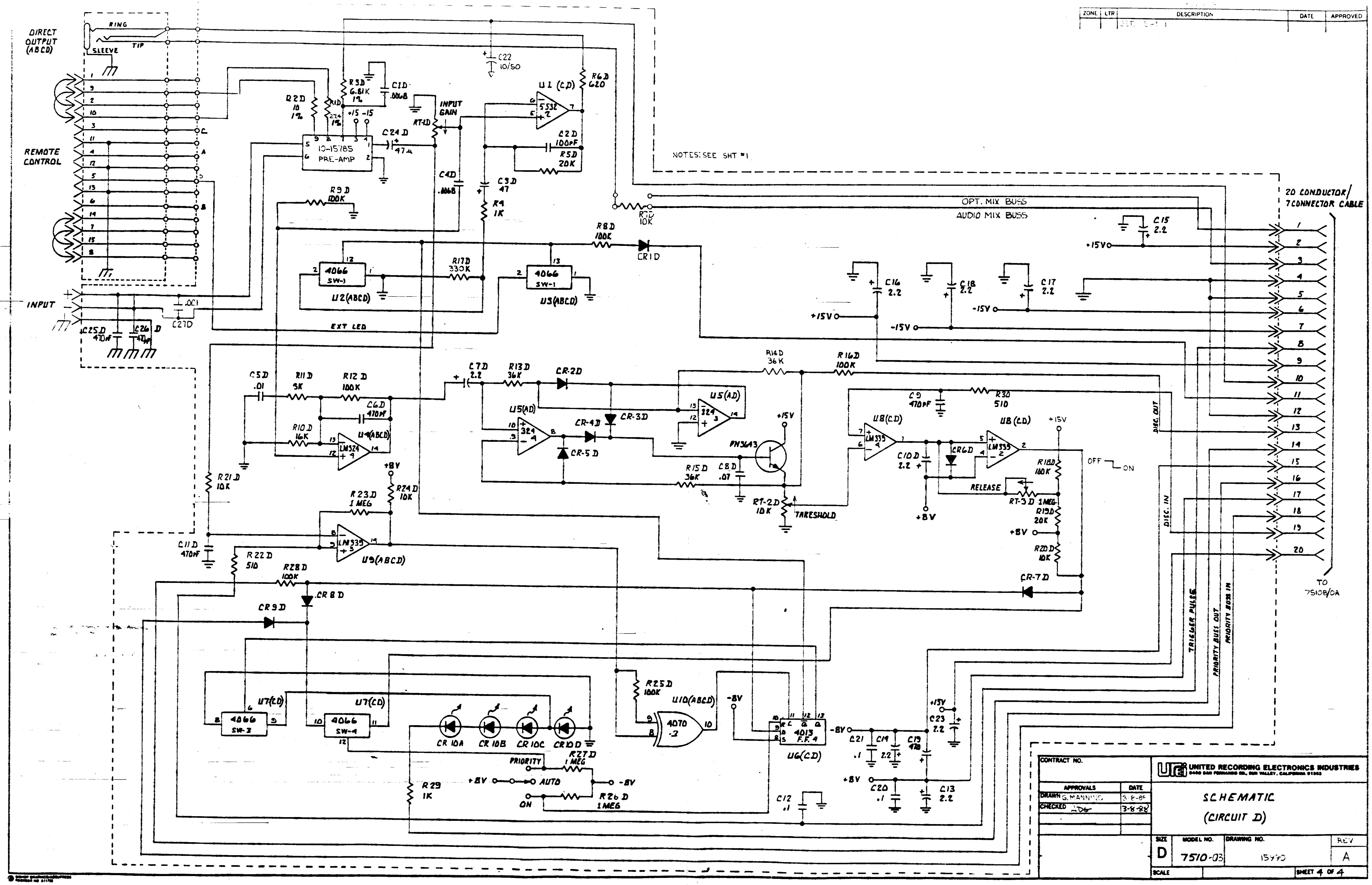
UNITED RECORDING ELECTRONICS INDUSTRIES

SCHEMATIC (CHANNEL B)

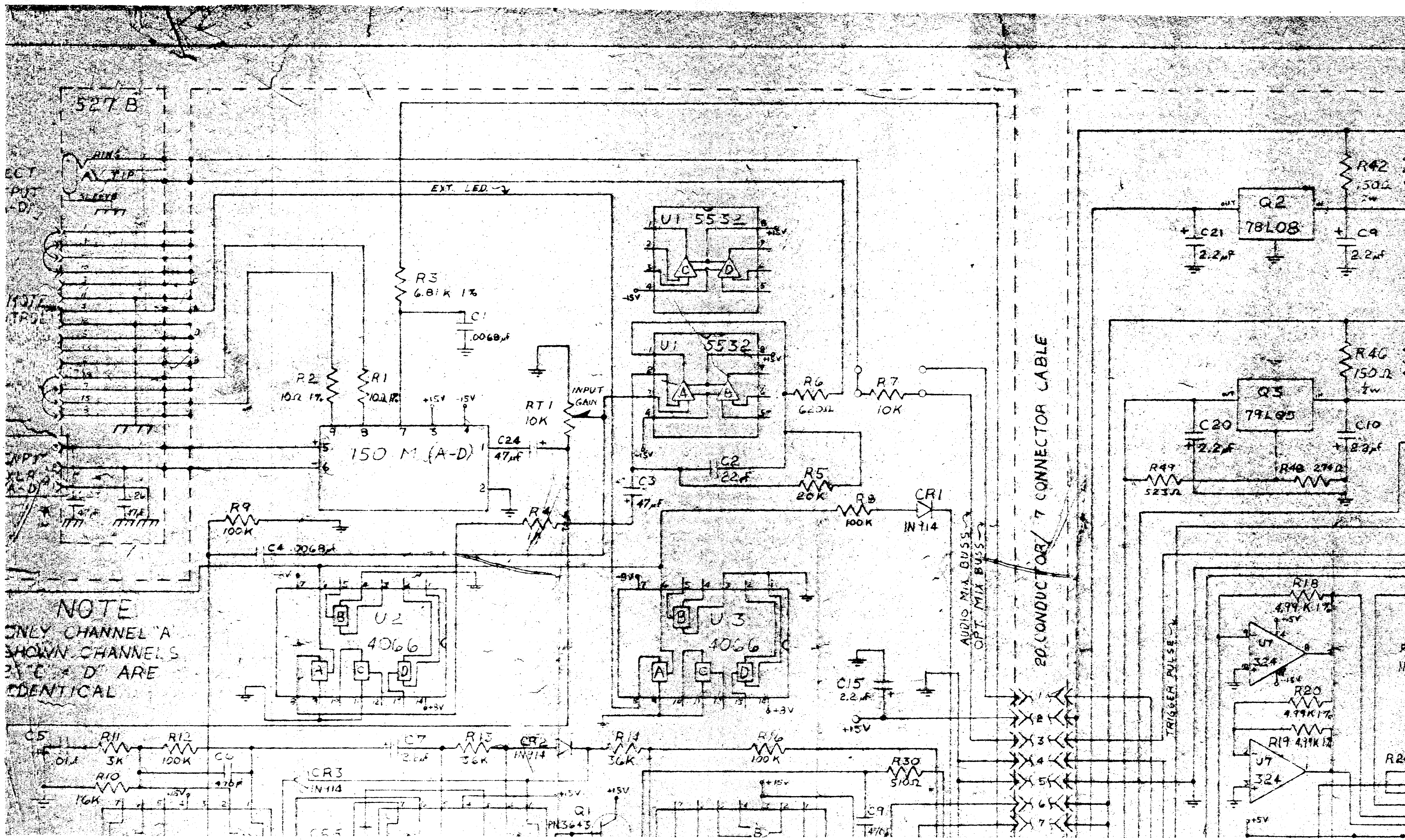
SIZE	MODEL NO.	DRAWING NO.	REV
D	7510-03	15990	A

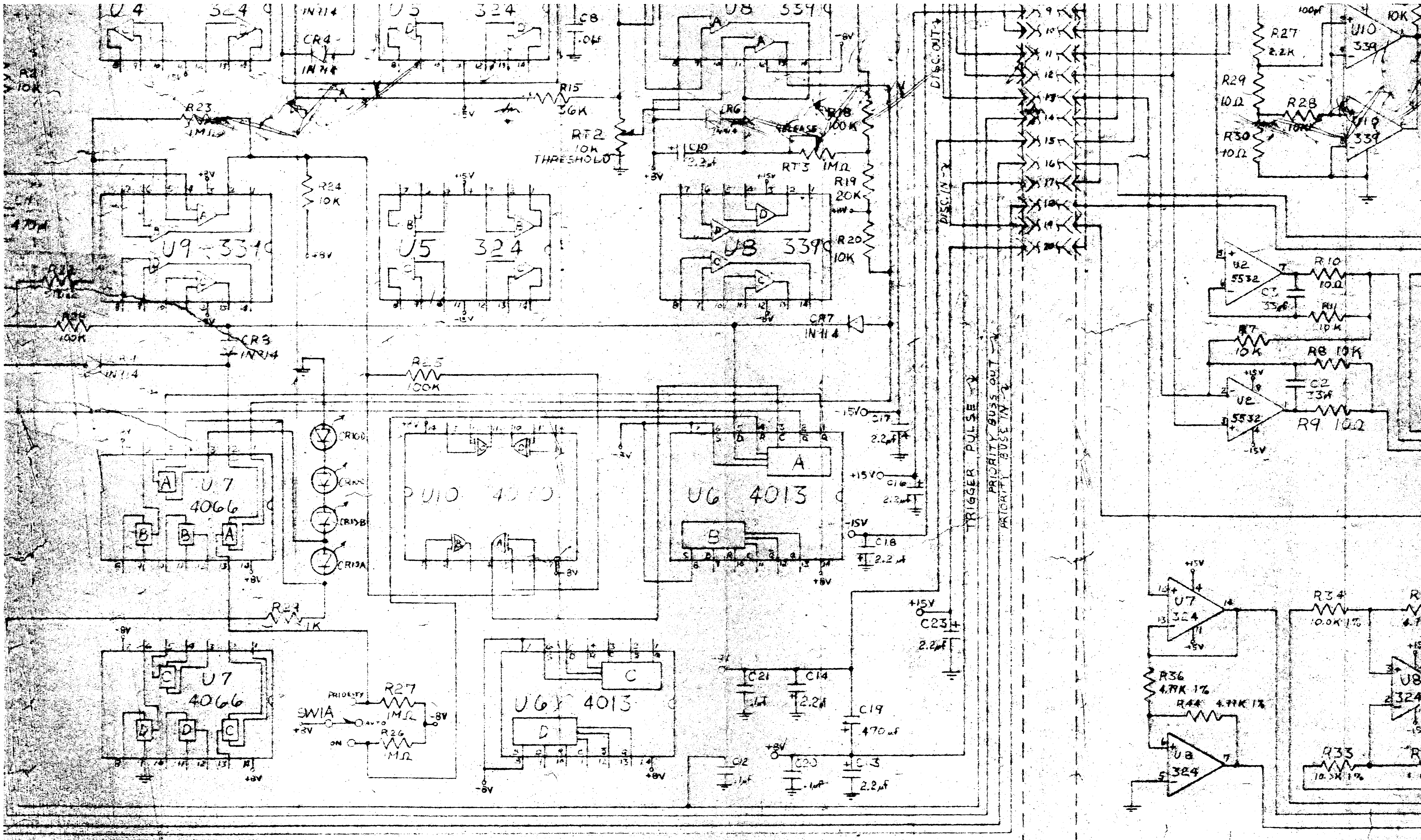
SHEET 2 OF 4


REVISIONS
ZONE
LTR
DESCRIPTION
DATE
APPROVED
SEE SHT 1
DIRECT OUTPUT (ABCD)
RING
TIP
SLEEVE
REMOTE CONTROL
INPUT
C25C 470pF
C26C 470pF
C27C .001
EXT LED
R2C 10 1%
R1C 274 1%
R3C 6.81K 1%
C1C .0068
10-13785 PRE-AMP
+15 -15
C24C 47
INPUT GAIN
RT-1C
C22 10/150
U1(CD) 5532
620
C2C 100pF
R5C 20K
C3C 47
C4C .0068
R4C 1K
R9C 100K
NOTES: SEE SHT #1
R7C 10K
OPT. MIX BUSS
AUDIO MIX BUSS
20 CONDUCTOR/ 7 CONNECTOR CABLE
R8C 100K
CR-1C
4066 SW-4
U2(ABCD)
R17C 330K
U3(ABCD)
C15 2.2
+15V
C16 2.2
C18 2.2
C17 2.2
-15V
C5C .01
R11C 5K
R12C 100K
C6C 470pF
R10C 16K
U4(ABCD) LM324
C7C 2.2
R13C 36K
CR2C
U5(BC) 324
CR3C
CR4C
CR5C
R14C 36K
R16C 100K
PN3643
R15C 36K
C8C .07
RT-2C 10K
THRESHOLD
C9 470pF
R30 510
U8(CD) LM339
C10C 2.2
CR6C
OFF ON
R18C 100K
RELEASE
RT-3C 1MEG
R19C 20K
R20C 10K
+8V
DISC. OUT
DISC. IN
R21C 10K
R23C 1 MEG
R24C 10K
U9(ABCD) LM339
C11C 470pF
R22C 510
R28C 100K
CR 8C
CR-9C
CR 7C
TO 7510B/10A
TRIGGER PULSE
PRIORITY BUSS OUT
PRIORITY BUSS IN
U7(CD)
4066 SW-2
4066 SW-1
CR10A
CR10B
CR10C
CR10D
PRIORITY
R27C 1 MEG
AUTO
ON
R26C 1 MEG
R29 1K
R25 100K
U10(ABCD) 4070 -4
-8V
4013 F.F. 1
U6(CD)
C21 .1
C14 2.2
C19 47
C23 2.2
C20 .1
C13 2.2
C12 .1
CONTRACT NO.
APPROVALS
DATE
DRAWN
CHECKED
3-4-78
UREI UNITED RECORDING ELECTRONICS INDUSTRIES
SCHEMATIC (CHANNEL C)
SIZE D
MODEL NO. 7510-03
DRAWING NO. 15770
REV A
SCALE
SHEET 3 OF 4


DIRECT OUTPUT (ABCD)
RING
TIP
SLEEVE
REMOTE CONTROL
INPUT
C22 10/50
R3D 6.81K 1%
C1D .0068
R2D 10 1%
R1D 2.74 1%
+15 -15
10-15785 PRE-AMP
INPUT GAIN
RT-1D
C24D 47μ
U1 (CD)
5532
R6D 620
C2D 100pF
R5D 20K
C4D .0068
C3D 47
R4 1K
R9D 100K
NOTES: SEE SHT #1
OPT. MIX BUSS
AUDIO MIX BUSS
R7D 10K
R8D 100K
CR1D
4066 SW-1
U2(ABCD)
R17D 330K
U3(ABCD)
EXT LED
C25D 470pF
C26D 470pF
C27D .001
20 CONDUCTOR/ 7CONNECTOR CABLE
TO 7510B/DA
C15 2.2
+15V
C16 2.2
C18 2.2
C17 2.2
-15V
C5D .01
R11D 5K
R12D 100K
C6D 470pF
R10D 16K
LM324
U4(ABCD)
C7D 2.2
R13D 36K
CR-2D
CR-3D
CR-4D
CR-5D
U5(AD)
324
R14D 36K
R16D 100K
PN3643
R15D 36K
C8D .07
RT-2D 10K
THRESHOLD
C9 470pF
R30 510
U8(CD)
LM339
C10D 2.2
CR6D
RELEASE
RT-3D 1MEG
R18D 100K
R19D 20K
R20D 10K
OFF ON
DISC. OUT
DISC. IN
+8V
R21D 10K
R23D 1 MEG
R24D 10K
C11D 470pF
U9(ABCD)
R22D 510
R28D 100K
CR8D
CR9D
CR-7D
TRIGGER PULSE
PRIORITY BUSS OUT
PRIORITY BUSS IN
U7(CD)
4066 SW-3
4066 SW-4
CR10A
CR10B
CR10C
CR10D
R25D 100K
U10(ABCD)
4070 -3
-8V
4013 F.F. 4
U6(CD)
C21 .1
C14 2.2
C19 470
C23 2.2
C20 .1
C13 2.2
C12 .1
R29 1K
PRIORITY
AUTO
ON
R27D 1 MEG
R26D 1 MEG
ZONE
LTR
DESCRIPTION
DATE
APPROVED
CONTRACT NO.
UREI UNITED RECORDING ELECTRONICS INDUSTRIES
APPROVALS
DATE
DRAWN G. MANNING 3-8-85
CHECKED 3-8-85
SCHEMATIC
(CIRCUIT D)
SIZE D
MODEL NO. 7510-03
DRAWING NO. 15990
REV A
SCALE
SHEET 4 OF 4

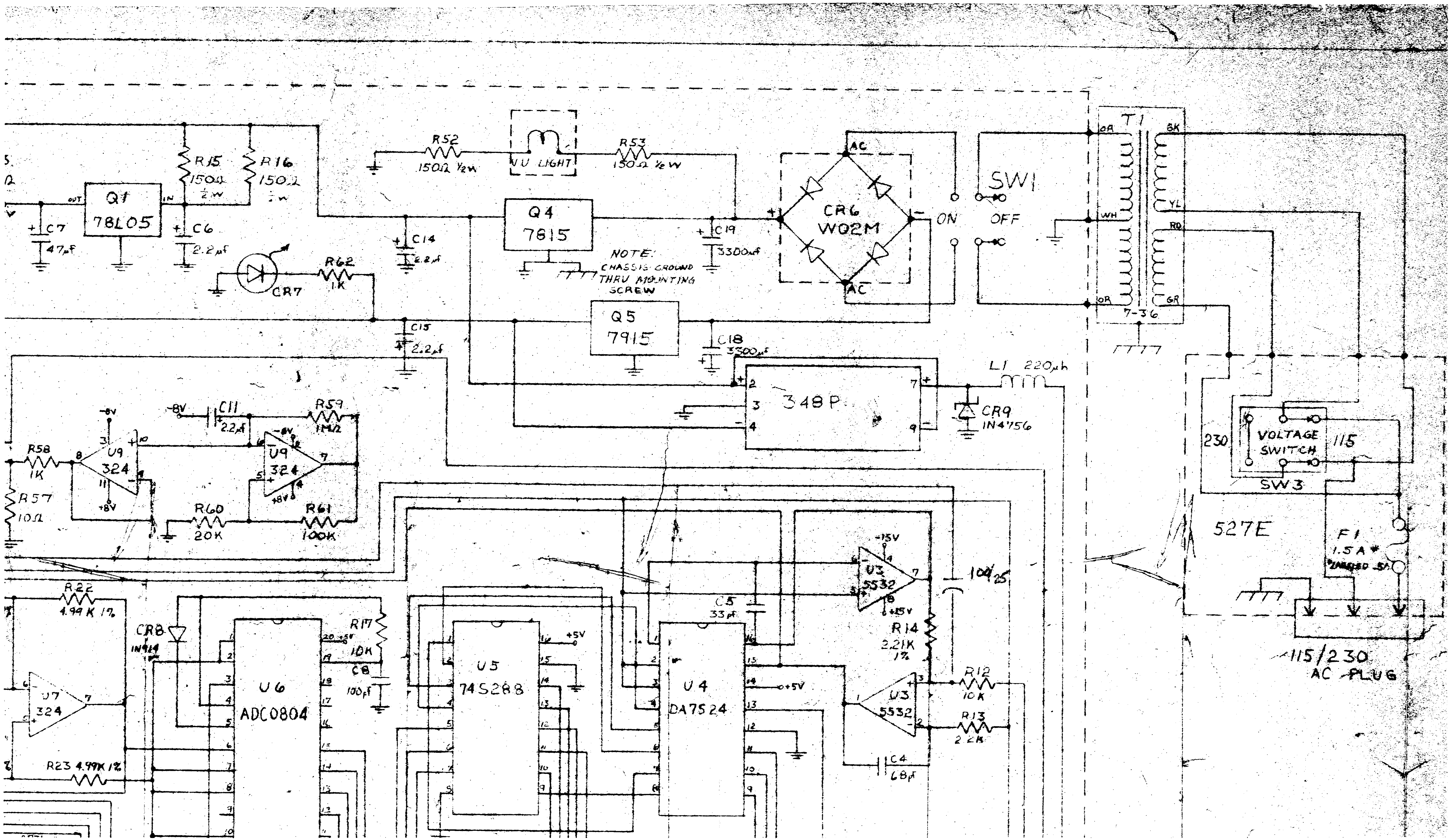
7510A SCHEM. 1 OF 4 (TOP LEFT)

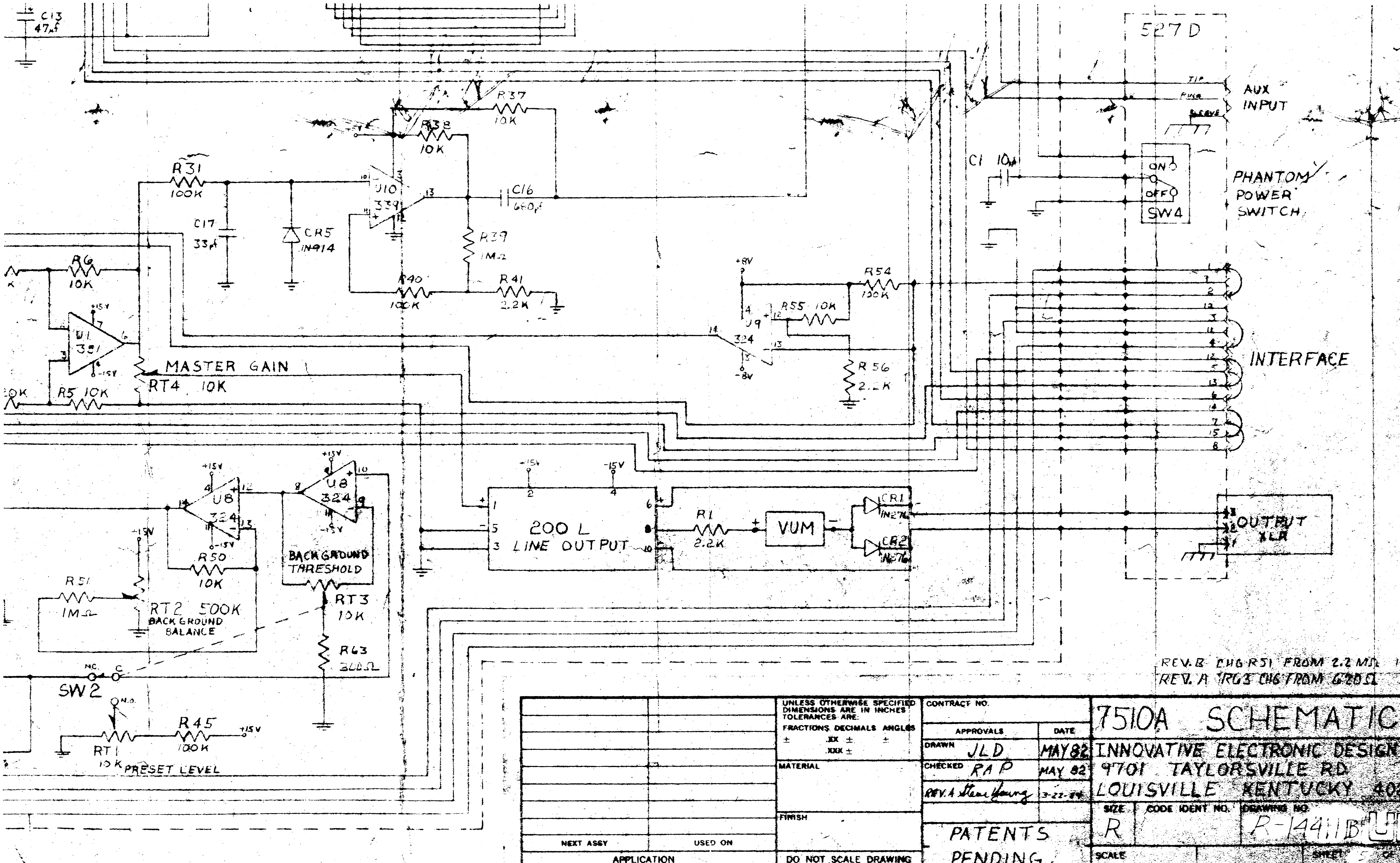




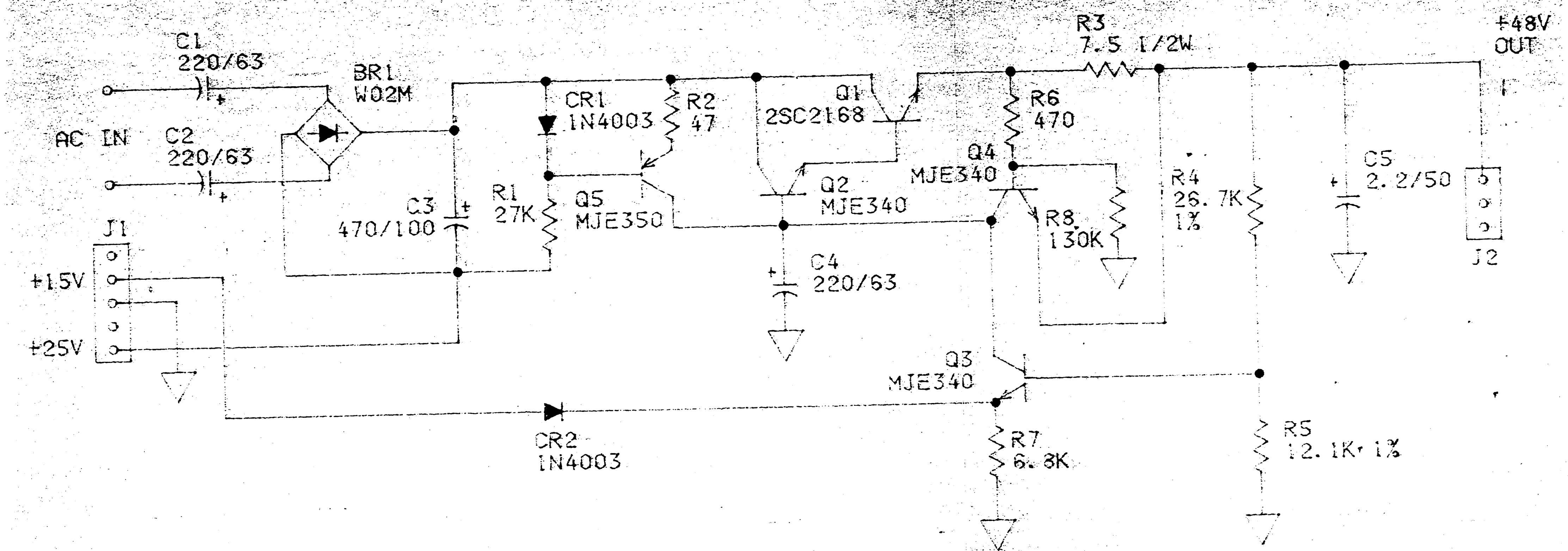
7510A SCHEM. 2 OF 4 (527A BOTTOM LEFT)

7510A SCHEMATIC 3 OF 4 (TOP RIGHT)





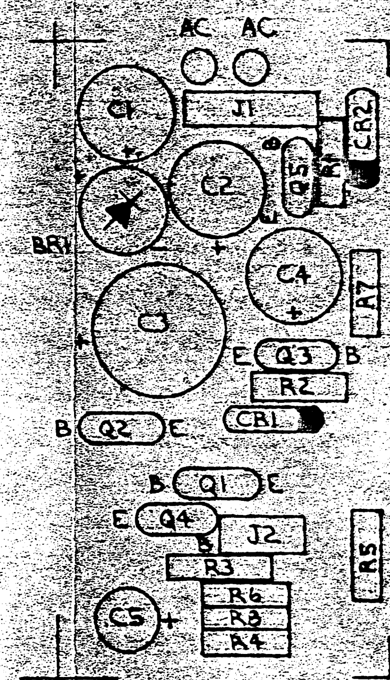
7510 A SCHEM 4 OF 4 (BOTTOM RIGHT)



NOTES: UNLESS OTHERWISE SPECIFIED

1) ALL RESISTORS IN OHMS ±5% 1/4W

2) ALL CAPACITORS IN MICROFARADS



PCB #12-15794
ASSY #10-15979

SCHEMATIC
PHANTOM POWER SUPPLY

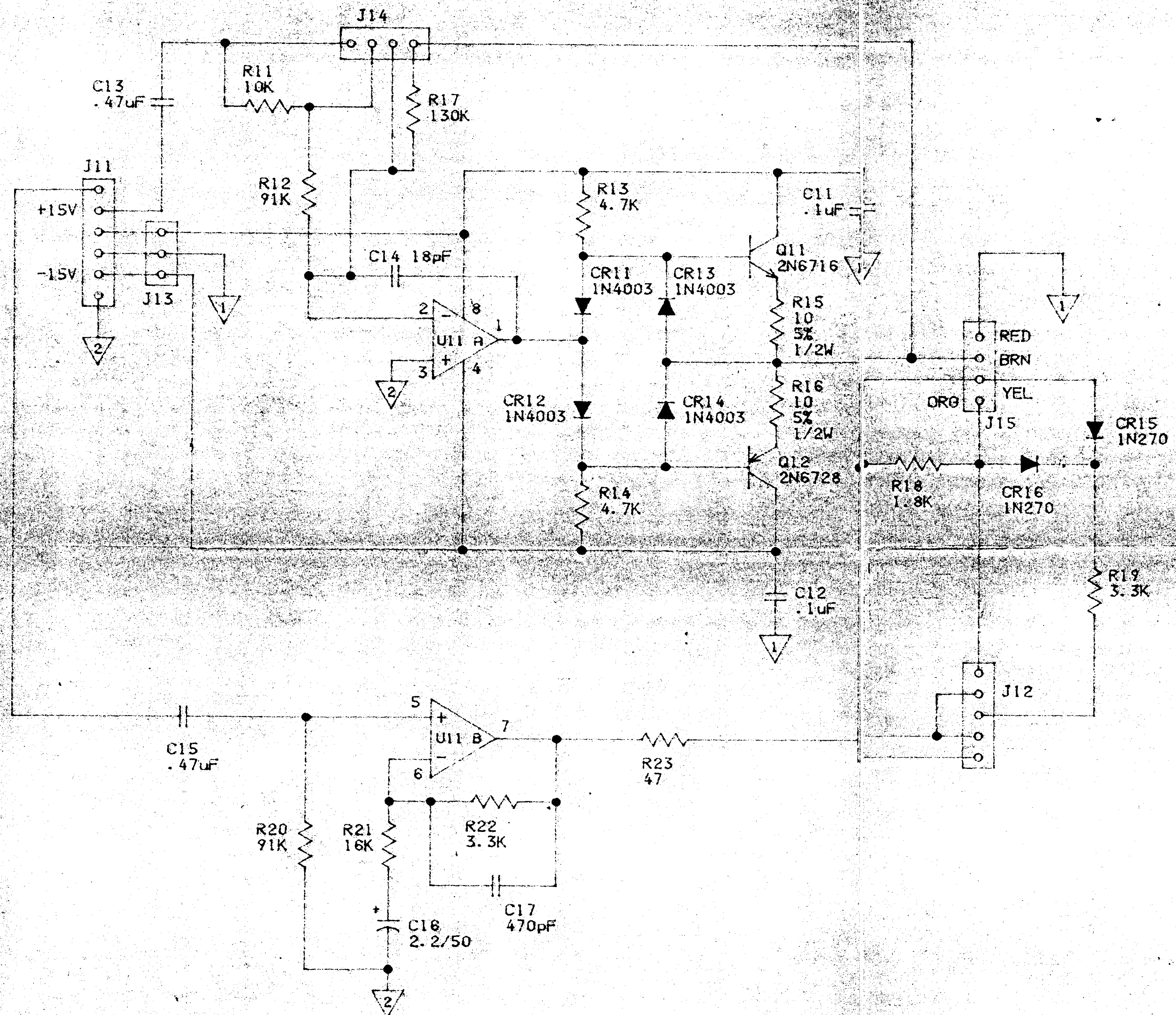
DD-239

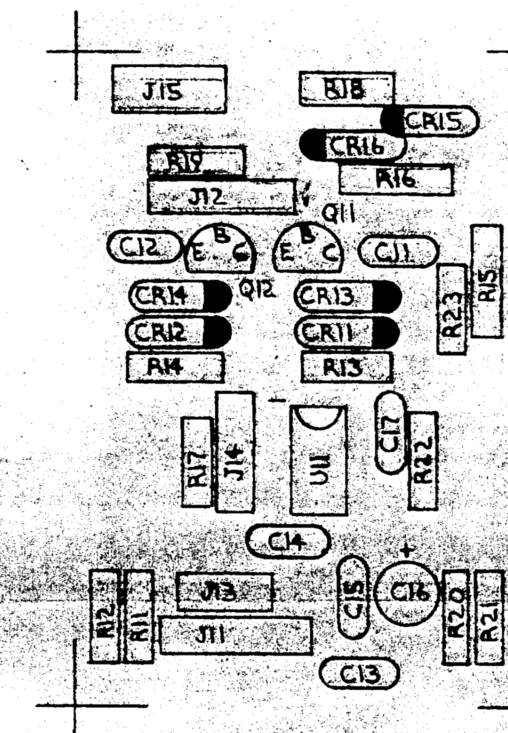
7510-03

C

15992

GM 3-9-88





PCB # 12-15975
ASSY # 10-15797

3)U11 IS TYPE 5532
2)CAPACITORS IN MICROFARADS
1)ALL RESISTORS IN OHMS ±5% 1/4W
NOTES:UNLESS OTHERWISE SPECIFIED

REV	CHANGE DOCUMENT	CHANGE DESCRIPTION	DATE	APVD

UREI	TITLE: SCHEMATIC OUTPUT AMPLIFIER
JOB NO: PD-239	
MODEL: 7510F03	SIZE: C · DRAWING NO: 15991
TOLERANCES:	SCALE: — · SHEET 1 OF 1
	DRAWN: GM 3/9/88 · CHECK: