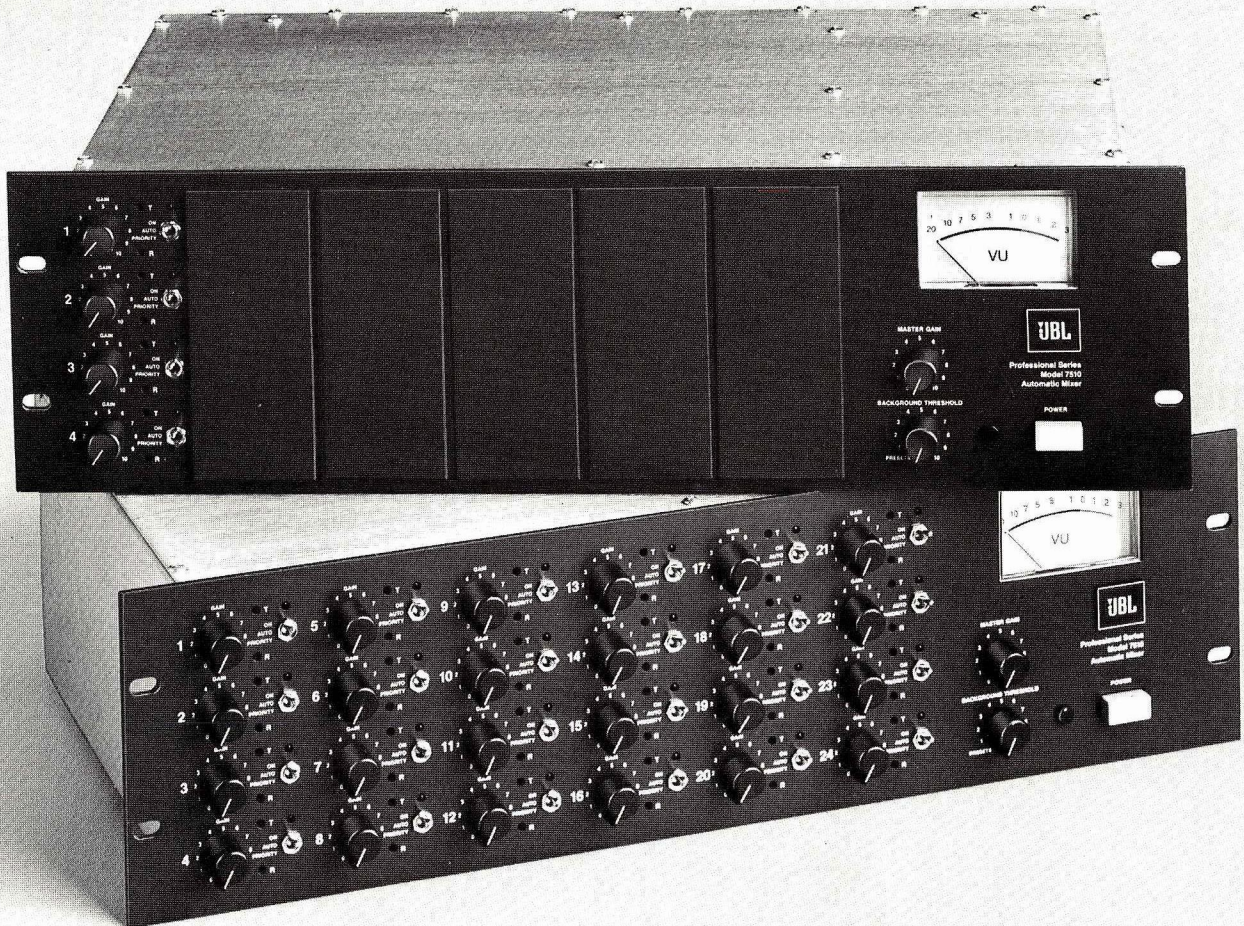


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Professional Series Installation and Technical Guide

Model 7510 Automatic Microphone Mixer



JBL

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WARNING TO PREVENT FIRE OR SHOCK HAZARD, DO NOT EXPOSE THIS APPLIANCE TO RAIN OR MOISTURE

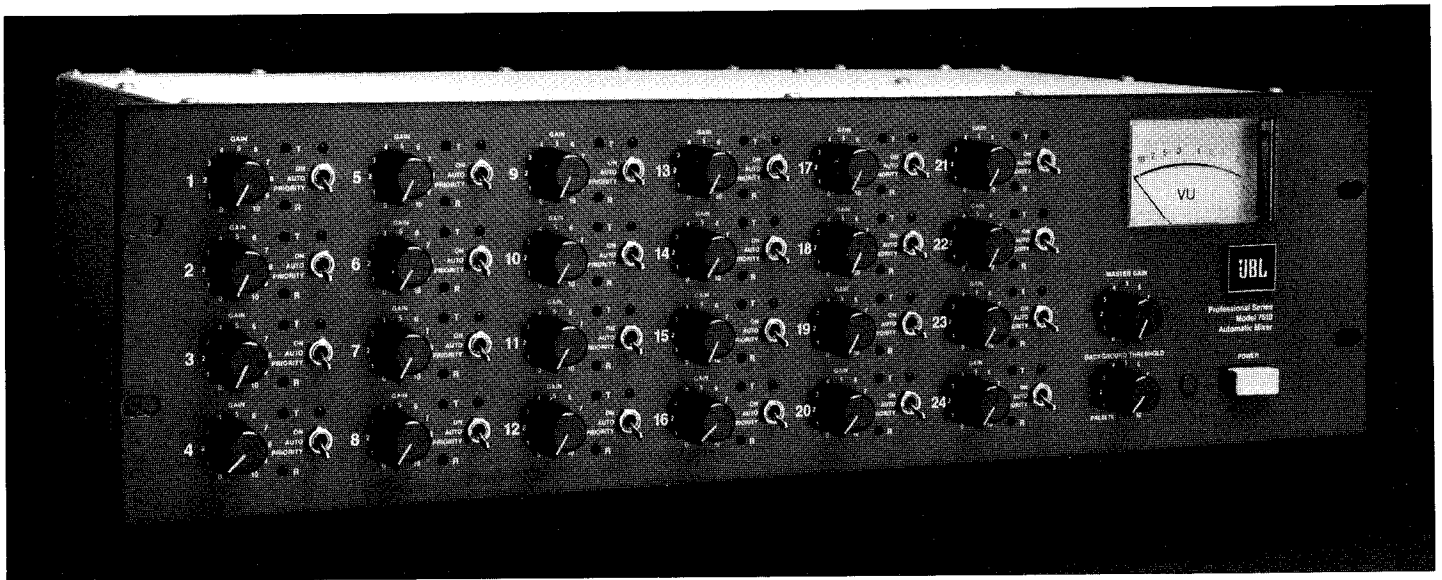


Figure 1. JBL Model 7510

INTRODUCTION

The JBL Model 7510 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 7510 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 7510 "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 7510 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 7510 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 7510'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 7510 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 7510 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 7510 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 4.) This prevents feedback from commencing as more and more mics are activated.

One key to the Model 7510'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 7510 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 7510'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.

The JBL 7510 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 7510 is an excellent tool for the sophisticated recording studio, theater, concert sound reinforcement company, or broadcaster.

How to Use This Manual

In order to make this manual more useful to the less technically oriented, we have provided straightforward explanations of each connector, control and procedure. The *indented type* is more technical and detailed information.

If your Model 7510 is already installed, see *page 5* for a review of all operating controls, and *page 6* for information on how to adjust the controls.

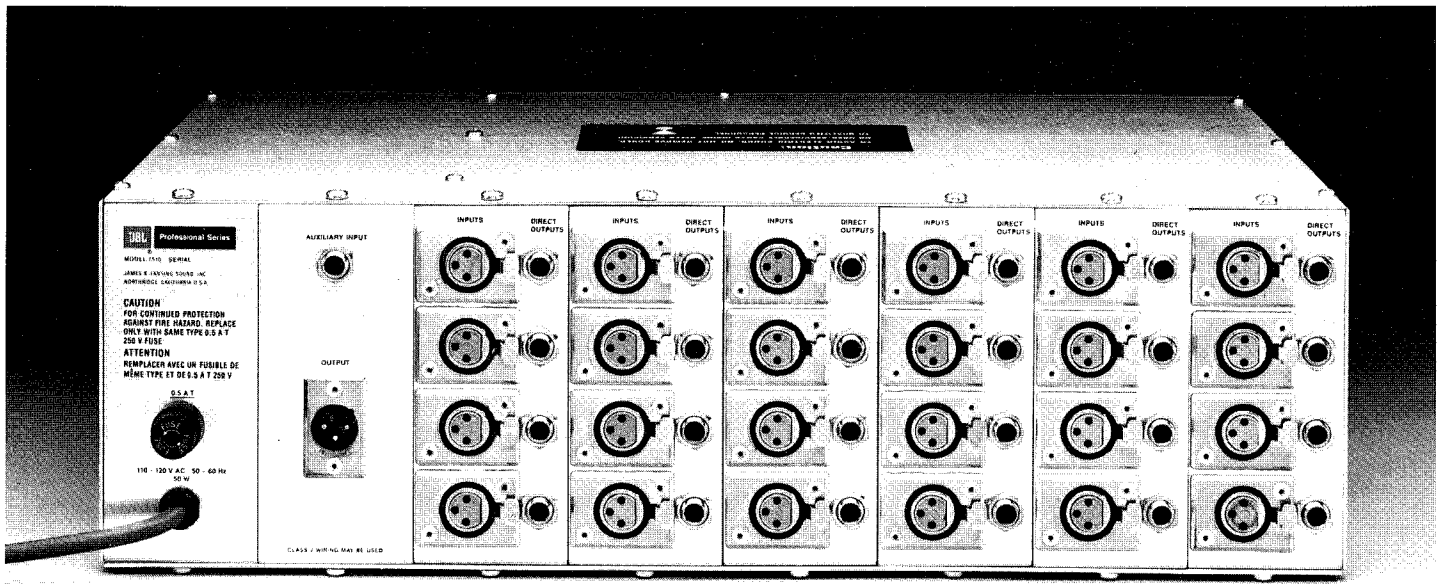


Figure 2. 7510 Rear Panel.

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 7510 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 7510 should be equipped with an XL-type connector (3-pin male).

The 7510 Mic Input connectors are female XL-type (Switchcraft D3F). These transformer-balanced inputs have a nominal input level of -50 dBv (0 dBv ref. 0.775 V RMS), with 25 dB of headroom. Having an actual input impedance of 1.35 k Ω at 1 kHz, the inputs are designed for use with low impedance (150 Ω) microphones. 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 in use. An internal Phantom On/Off 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

This stereo phone jack (wired for a mono signal plus an extra shield ground) carries the channel's microphone input signal after it has been amplified. The signal at this jack is affected 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 7510 serves as a multi-channel noise gate, or where special signal processing of the mic input is desired. The channel's Gain control also adjusts the level at this jack. When you plug a cable into the Direct Output jack, the channel's sound no longer contributes to the 7510's Main Output (although the digital circuitry continues to count it as an active mic when determining the output gain). The signal can be fed to the line-level input of any professional sound equipment, either low or high impedance. For optimum rejection of noise and elimination of ground loops, use a cord made with dual-conductor shielded cable and a stereo phone plug, wired as shown on page 8. It is permissible to use a cord consisting of a standard phone plug with single-conductor shielded cable.

The Direct Output is a 6 mm ($\frac{1}{4}$ in) tip-ring-sleeve phone jack, connected after the channel Gain control and before the booster amp.

While the jack has three conductors, it is unbalanced; the tip carries signal, the ring is signal common, and the sleeve is chassis ground, providing additional shielding against noise. The Direct Output has a 600 Ω source impedance, with a nominal level of 1.23 V (+4 dBm ref. 600 Ω for 0 VU). It can be used to feed an external signal processor, another mixing console, a tape machine, etc. The jack is normalised so that when no cable is plugged in, the signal flows straight from the input channel to the output summing amplifier.

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 7510'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 Aux 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. For optimum rejection of noise and elimination of ground loops, use a cord made with dual-conductor shielded cable and a stereo phone plug, wired as shown on page 8. It is permissible to use a cord consisting of a standard phone plug with single-conductor shielded cable.

The Aux Input is a 6 mm ($\frac{1}{4}$ inch) tip-ring-sleeve phone jack. While the jack has three conductors, it is unbalanced: the tip carries signal, the ring is signal common, and the sleeve is chassis ground, providing additional shielding against noise. Aux Input has a 10 k Ω load impedance, with a nominal input level of +4 dBv. It applies signal, via an isolation resistor, to the audio summing network, where it is mixed with any microphone input signals that 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 an XL-type male connector carrying a monophonic mix of all active microphone inputs plus any Aux Input signal. The output may be connected directly to a power amplifier (and the associated speaker system), or to another mixing console if the 7510

is being used as a submixer. The 7510 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 end of the cable that mates with the 7510 Main Output should be equipped with an XL-type connector (3-pin female).

The Main Output is a Switchcraft D3M connector. This transformer-balanced output has a nominal +4 dBm (ref. 600 Ω for 0 VU) level with 25 dB of headroom. The actual output impedance is 360 Ω at 1 kHz, and the output will drive 600 Ω or higher impedance circuits without special termination.

Power Cord and Fuse

In the U.S.A. and Canada, the 7510 is wired for connection to nominal 100-120 V AC, 50 or 60 Hz power mains. A 0.5 ampere 3AG type fuse protects the power supply, and should only be replaced with one of the same value. Export models are available wired for 200-240 V AC operation, and are equipped with 0.25 ampere fuses. Check the rear panel label to verify compatibility with the local mains voltage.

CAUTION: In the event of repeated fuse failure, check to see that the mains are within 10% of the nominal voltage. If the mains are within tolerance, have the 7510 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 5 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 7510 is equipped with less than a full complement of 6 modules (24 inputs).

Gain Control

This rotary control adjusts the relative amount of signal from the corresponding microphone to be mixed into the output program. 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 71 dB of Gain is available in the input module (from Mic In to Direct Out).

Threshold Trimmer

This screwdriver-adjustable control sets the sensitivity of the circuitry which determines whether the mic 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 mic. Ideally, the Threshold should be set low enough (turned clockwise) that the channel turns On whenever the mic is being used, but not so low that background noises turn it On. 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 7510'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.

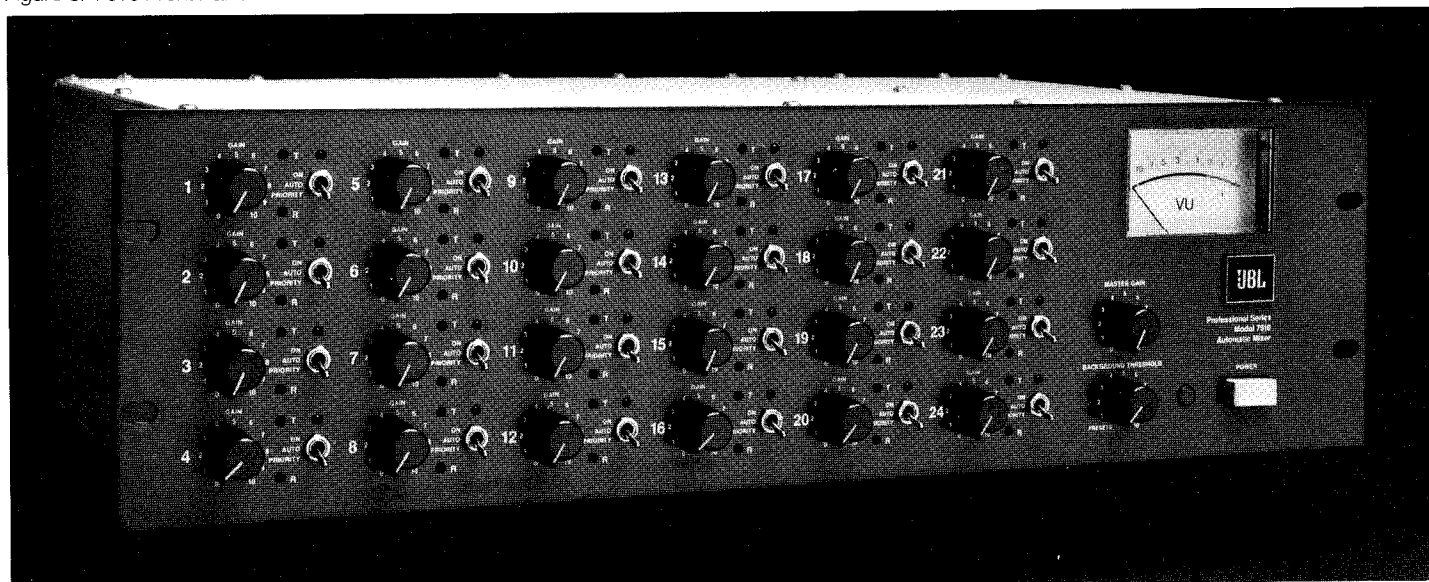
The attack time is dependent on how soon a zero crossing occurs after the signal first exceeds the threshold level. The specified range of 10 milliseconds to 10 microseconds corresponds to a half cycle at from 20 Hz to 20 kHz. Since the turn-On takes only 30 to 60 nanoseconds (30 to 60 trillionths of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

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 mic 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

Figure 3. 7510 Front Panel.



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 mic turns On and Off excessively.

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 you want the microphone to be live at all times, as when setting up the basic mixing levels of the various channels. Use Auto mode if you want the microphone to be live only when it is in use, and to automatically turn Off at other times. Use Priority mode if you want the microphone to be On whenever it is in use, and you want also want it simultaneously to turn Off any other microphones whose channels are in 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 mic 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 7510 because gain increase occurs during an instant when the unit is amplifying zero volts.

Channel On Indicator

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

In On mode, the LED is always illuminated. In Priority mode it is illuminated only when the sound at the mic 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 mic.

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 at the danger of possible howling or ringing; it 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 18 dB of gain is available in the output section, providing an overall maximum gain of 80 dB minimum (600 Ω load) from Mic Input to Main Output. While Master Gain sets the 7510's output level, to avoid feedback the level is reduced progressively by the digital output attenuator as more input channels turn On, at the rate of -3 dB for each doubling of active inputs. (Refer to *Figure 16, page 16*.) 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 mics are used, yet it does not interfere with dynamic range like a compressor/limiter.

VU Meter

The VU meter displays the signal level fed to the 7510 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. Almost all power amplifiers

can be driven to full output by the 7510 when its meter displays 0 VU, +4 dBm (ref. 600 Ω), or perhaps a few dB above 0 VU. Most of the time you'll want the amplifier operating from 5 to 20 dB below that level. Thus, if a constant “red zone” VU level seems to be necessary, we suggest you turn up the gain of the power amplifier or other equipment in the sound system, or look for a bad cable, improper hookup, or other problem.

The VU meter is located on the primary side of the output transformer. It is amplifier-buffered 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 Ω).

Background Threshold

The Background Threshold control is part of the circuit which alters the set Threshold of all mic inputs in accordance with the overall background sound level (i.e., that sound which is sensed in common at all mics). 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 mic may turn the mic's channel On. When the environment becomes noisier, that same mic will require more level before it turns ON. The Background Threshold control is used to determine the maximum amount the input threshold will rise when the background sound level goes up.

Ideally, the Background Threshold control should be set high enough so that no mics 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 mic's sensitivity to automatic turn-On in the presence of desired *program* sound.

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

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 bus, even if the individual mic channel is not On (i.e., if it is below the channel's threshold or if it is inhibited by a priority function). The bus in turn presents this instantaneous sum of the background sounds to the background threshold control circuit. The set Background Threshold is then made available to all input channels via the Threshold Receive bus, so it can serve as a comparison reference. Each channel's individual Threshold control thus sets the relative level at that mic, above the established overall background level, necessary for the mic's channel to turn On.

Power Switch and Indicator

An alternate action pushbutton switch turns On the AC power to the 7510. The adjacent green panel light glows when the power is On, and the VU meter illuminates.

RECOMMENDED SETUP ADJUSTMENT PROCEDURE

NOTE: Any microphones to be used should be connected at this time. If no mic is connected to a given channel, set it to Auto mode, and turn its Gain control all the way down to zero.

1. Set the Master Gain control at #5 on the control scale.
2. Switch ON all input channels and adjust their Gain controls for the desired mix. If feedback (ringing or howling) begins, lower the input Gain settings slightly. If fine-tuning for absolutely maximum gain before feedback, switch On and adjust only one channel at a time.
3. After all input Gain controls have been set, make sure all the inputs are in Auto mode.
4. Turn down the Background Threshold control to almost all the way (counterclockwise, setting it at the point just before it clicks to “off” position).

5. Speak, sing or play an instrument into each mic just as will be done during the actual performance or other live mixing situation. Set that channel's Threshold control just high enough (that is, a low threshold for high sensitivity) that the mic turns On reliably. Too high a control setting (too much sensitivity) will cause random triggering by background noise.
6. Temporarily set all Release Time controls fully counterclockwise for fast release to aid in accurate adjustment of the Master Threshold control in Step 7 below.
7. Temporarily set the Background Threshold at #10. Produce the loudest of typical *background* sounds that is expected to occur and, simultaneously, lower the Background Threshold setting so one or more mics begin to turn On. (Remember, background sound is off-mic sound, sound you don't want activating any mics). Then gradually increase the setting until the last mic just turns Off. (If no mics turn On in the presence of the background sound, even with the Background Threshold set at its lowest position before it clicks Off, then leave it set at the maximum sensitivity position, full CCW).
8. 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 mic under adjustment. Generally a longer Release Time will be desirable for speech and a shorter one for music.
9. 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 tends to be cut off on a given input channel, its Threshold is too high (i.e., sensitivity is too low). Turn the Threshold screw clockwise until the symptom is alleviated.

If you have to talk or play too loudly for the channel to turn On at all, either the channel Threshold is set too high or the Background Threshold is set too high. If most channels seem OK, it is probably a channel Threshold misadjustment. If several channels are too insensitive, or if turning the channel Threshold to maximum sensitivity still doesn't help, then lower the Background Threshold.

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

If you cannot make the unit work properly and you have checked all connections and followed the preceding setup procedure, there may be a malfunction in the electronics. See *page 19* of this manual.

DETAILED INSTALLATION INFORMATION

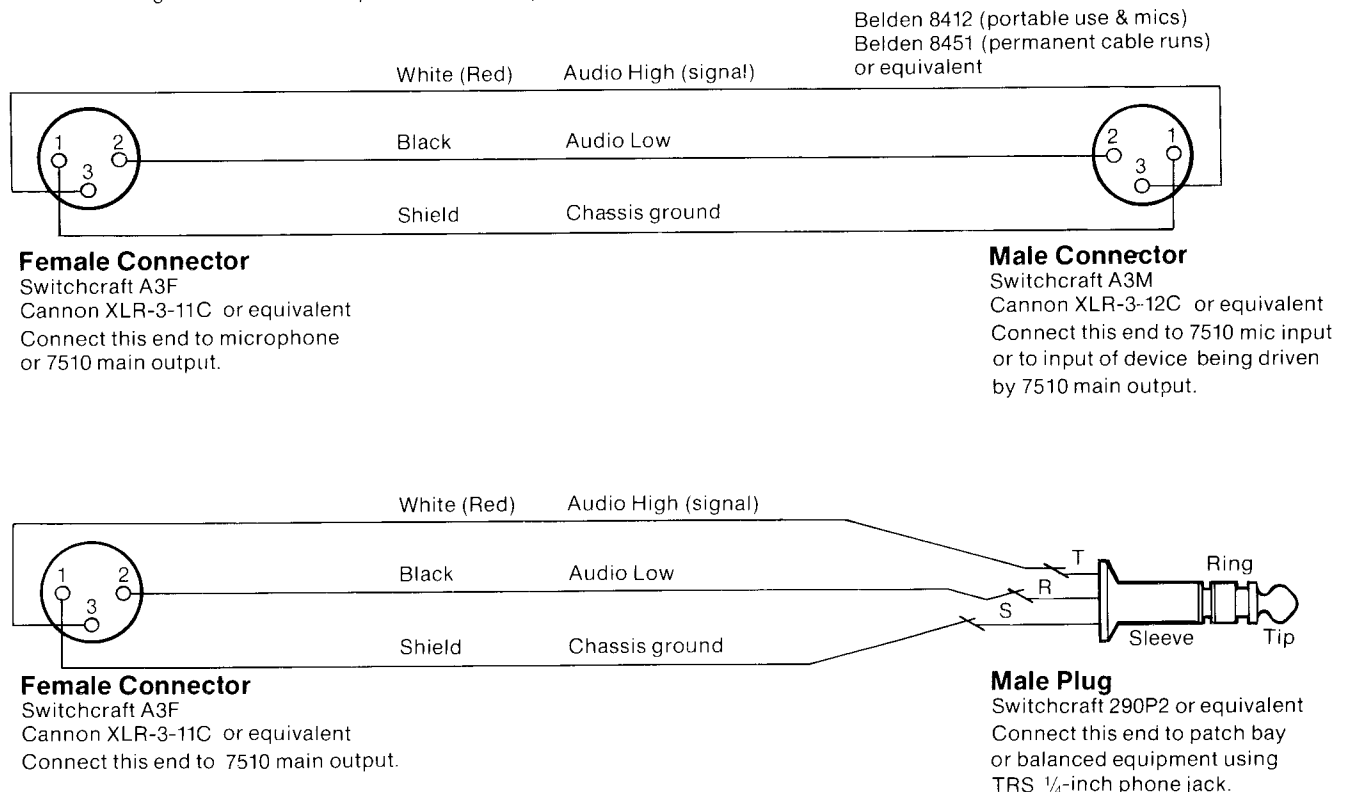
Cables for the Channel Inputs and Main Output

The 7510 channel inputs are XL-type connectors (Switchcraft D3). These connectors are advantageous for several reasons: (1) they mate with the most popular type of professional mic cable connector, (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 Output jack, also an XL-type, offers similar advantages although phantom power is not a concern, and the jack is "floating" rather than "balanced."

The distinction between "balanced" and "floating" is simply that the signal-carrying conductors in a floating circuit have no ground reference due to transformer-provided isolation, whereas the signal-carrying conductors in a balanced circuit have equal potential difference from ground—provided in the 7510 input circuit by the phantom power build-out resistors that tie the transformer primary to the power supply. (Some balanced circuits use no transformer, but instead use a differential amplifier.)

NOTE: The XL-type connectors in the JBL 7510 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 Aux Input will be in phase with pin 3 of the Main Output. Some sound equipment may be wired to the DIN (European)/JIS (Japanese) 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 7510; if phase-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.

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



Cables for the Direct Outputs and Aux Input

The 7510 Direct Outputs and Aux Input 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. By wiring the jack's tip for the audio "high" conductor, its ring for the audio "low" 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-sleeve) phone jack. Ideally, cable with two center conductors plus a braided shield should be used, and wired as illustrated. The shield is cut (unused) at one end of the cable, avoiding an unwanted ground loop which can actually 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 you might hear some hum, which should go away when the proper cable is substituted).

Using High Impedance Microphones

If a high impedance microphone is to be used with the 7510, 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 7510's input, and another connector (phone jack or equivalent) which accepts the cable from the mic. If more than 10 feet of mic cable is needed, locate the transformer close to the mic, and run the longer cable on the low impedance side of the transformer. Bear in mind that the 7510 outputs can feed low or high impedance inputs without any special adapters or transformers.

Using High Level Input Sources with the 7510

If a high level source is to be connected to a 7510 channel input, it will be necessary to drop the signal level to avoid saturating the input transformer and overdriving the input amplifier, both of which will cause

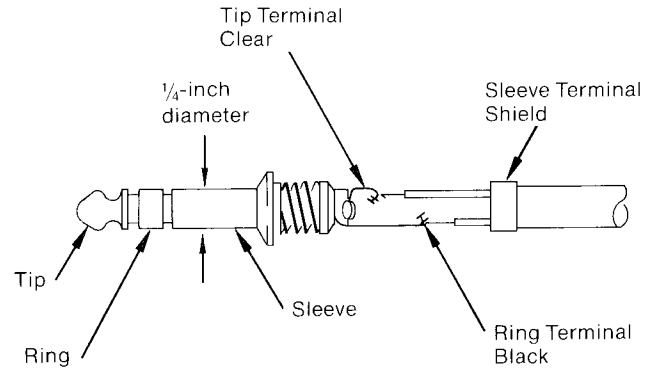
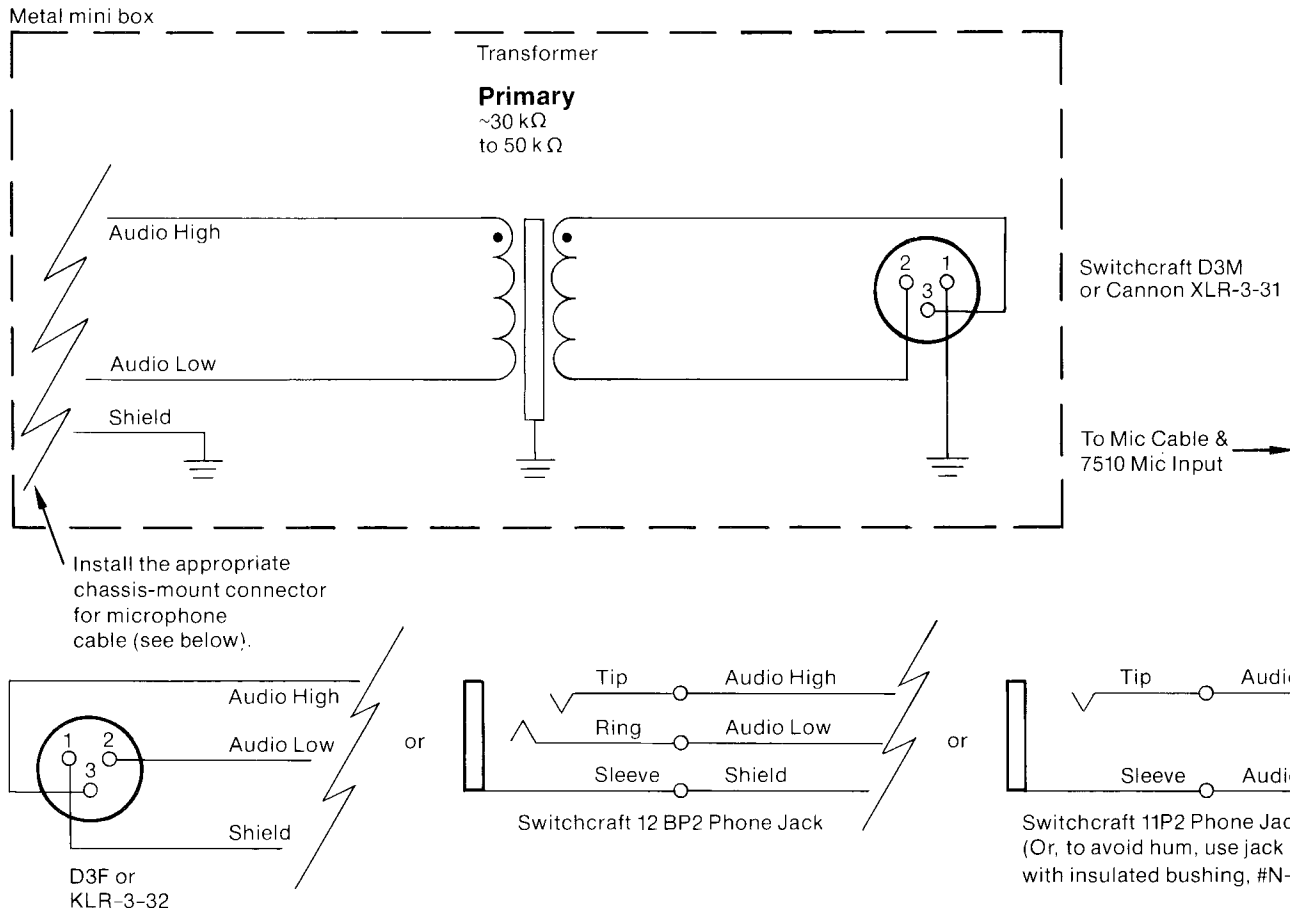


Figure 5. Cable wiring for the 7510's Direct Outputs and Aux Input.

unacceptable distortion. (High level sources include such items as a tape recorder, an FM tuner, a preamplified musical instrument like a synthesizer or organ, or most mixer outputs.) The level can be dropped by installing an attenuation pad ahead of the 7510 input. If the high level signal source also has a high impedance output, you will need both the pad and the matching transformer previously described. Be sure to install the pad ahead of the matching transformer to avoid saturating its core with excess levels. (See note in indented text below regarding DC blocking capacitors.) These same principles apply when connecting the 7510's Main Output or any of its Direct Outputs to a high sensitivity input (i.e., a mic level or low line level input).

Various types of pads are available, the most common of which are "T-pads" and "H-pads." T-pads unbalance a floating or balanced circuit but work well in unbalanced circuits. H-pads are best for balanced or floating circuits, but should not be used in an unbalanced circuit. Regardless of the type of pad, it should be installed near the input of the device it feeds, with as short a length of cable as possible on the low-level side of the pad. This practice ensures the best possible signal-to-noise ratio. Commercially manufactured pads are available for a variety of circuit impedances and attenuation values.

Figure 6. Use of a matching transformer for high impedance microphones.



Note that the circuit impedance *does* matter, and a pad which is specified for use in a circuit of a particular impedance range will not provide the specified attenuation in a circuit of substantially different impedance. Fortunately, with the 7510 the pads you are most likely to use will be those intended for insertion in front of a low impedance mic input, a pad type widely available.

To determine how much attenuation is necessary, look at the specifications of the device to be connected to the 7510 input, and find that device's nominal output level in dBm (or dBv referenced to 0.775 volts). Then subtract 50 from that number since -50 dBm is the nominal input sensitivity of the 7510. (If the device is rated in dBv referenced to 1 volt, subtract another 2 dB from the number of dB to correct it to the 0.775 volt dBv/dBm reference.) The resulting number is the number of dB of attenuation you want. Minor adjustments ± 5 dB can be made with the 7510 channel Gain control or the external device's output level control, so the pad need not be of the precisely calculated value. For most applications, either a 20 dB, 30 dB, 40 dB or 50 dB pad will do the job. If a pad is used to match a high impedance unbalanced device to the 7510, and no transformer is added, then a pair of DC blocking capacitors must be inserted in the "high" and "low" sides of the audio line between the 7510 input and the pad. Use capacitors rated for at least 100 microfarads at 48 volts (or more). The capacitors serve to isolate the signal source from the mixers phantom power supply.

You may prefer to make your own pads. For advice, consult any good audio engineering handbook, such as the *Audio Cyclopedia* by Howard M. Tremaine, published by Howard W. Sams, or *Sound System Engineering* by Don and Carolyn Davis, also published by Sams.

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 mic input, (2) a preamplified instrument's output to drive a mic input, or (3) a very high impedance instrument pickup to drive a low to medium impedance mic input. "Direct" refers to a direct-wired connection to the mixer's mic input rather than the "indirect" approach of placing a microphone in front of the instrument amp'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.

Both functions (1) and (2) above are accomplished by using an attenuation pad and a transformer, the pad to reduce the level and the transformer to correctly match the impedance and terminate the mixer's input. In function (3), the pad may not be necessary, but the transformer must present a very high impedance to the pickup so that it does not overload the pickup and degrade the transient and high frequency response of the instrument. An alternative type of direct box

for function (3) is made by using a very high impedance preamplifier, plus an isolation transformer; such units are called "active" direct boxes. When connecting a power amplifier (guitar amp speaker output) to a mic input, the proper amount of attenuation is essential. This is not always easy to calculate since the transformer can introduce as much as 20 dB of signal loss.

A typical transformer for use in a speaker-output type direct box would have a 15 k Ω to 30 k Ω primary winding and a 150 Ω secondary winding. The high impedance primary will not draw significant power from the amplifier, and thus will not burn out even if a pad is inadvertently omitted. The low impedance secondary is properly terminated by the mixer's mic input, ensuring proper frequency response and minimum ringing. A 15k to 150 Ω transformer yields a voltage reduction of 100:1, or 40 dB of loss. The pad required ahead of that transformer will depend on the power rating of the amplifier (actually on its voltage fed to the speakers), but can be calculated very roughly as follows. Given the 7510's nominal input level of -50 dBm, and the matching transformer providing a 40 dB loss, one must drop the amp output to -10 dBv before it goes into the transformer.* A 1 watt amp is +30 dBm, 10 watts is +40 dBm, 20 watts +43 dBm, 40 watts +46 dBm, 60 watts +49 dBm, and 100 watts +50 dBm, and 200 watts +53 dBm. Remember, however, that we are concerned with the nominal level, not the maximum. Assuming the nominal level is about 10 dB below maximum, a 200 watt amp would have a nominal level of about +43 dBm, and a 100 watt amp a level of about +39 dBm —both very close to +40 dBm. Subtracting the desired -10 dBm level from +40, we find that a 50 dB pad will do the job for virtually any moderate to high power guitar amplifier. Since the speaker output is unbalanced, a T-pad should be used.

*The term "dBm" describes is a power ratio referenced to 1 milliwatt, which just happens to be 0.775 volts RMS across a 600 Ω load. The term "dBv" also describes a power ratio, but is referenced to the specific voltage of 0.775 volts RMS. Thus, only at 600 Ω will a given signal level have identical dBm and dBv values; as the impedance changes, the dBm and dBv values will differ. In this discussion, for simplicity, we have overlooked the changes in voltage values that occur when converting between 8 Ω or lower impedance speaker outputs and the 2.5 k Ω 7510 mixer input impedance.

Figure 8. General schematic of a direct box for feeding a mic input from a guitar amp output.

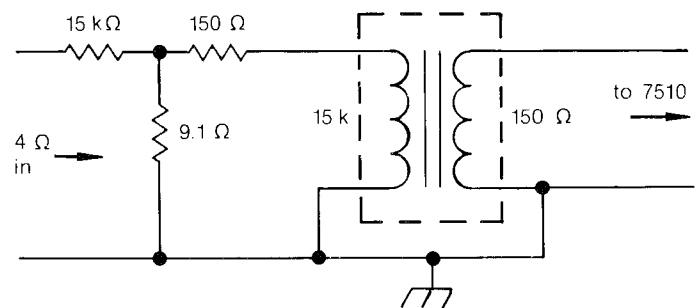
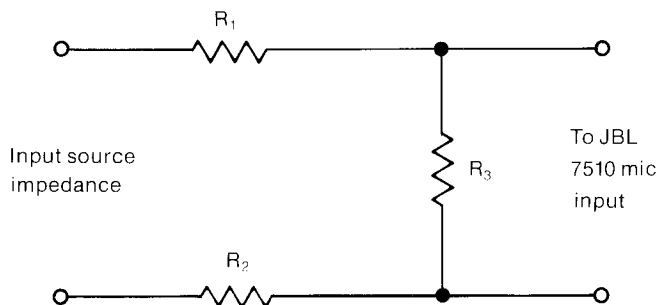


Figure 7. Schematics for most common "H" pads for use in low Z (600 Ω) and high Z (10 k Ω) lines.



AMOUNT OF LOSS IN PAD	INPUT SOURCE IMPEDANCE (OUTPUT IMPEDANCE OF DEVICE FEEDING 7510)	
	500 TO 600 Ω	APPROX. 10 k
-20 dB	$R_1 = 220 \Omega$ $R_2 = 50 \Omega$	Not possible to construct a 20 dB pad in this case.
-30 dB	$R_1 = 250 \Omega$ $R_2 = 15 \Omega$	Not possible to construct a 30 dB pad in this case.
-40 dB	$R_1 = 250 \Omega$ $R_2 = 5 \Omega$	$R_1 = 5 \text{ k} \Omega$ $R_2 = 100 \Omega$
-50 dB	$R_1 = 300 \Omega$ $R = 2 \Omega$	$R_1 = 5 \text{ k} \Omega$ $R = 30 \Omega$

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).

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 mic inputs and line outputs in the same cable. These practices help avoid unwanted crosstalk, induced 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.

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 mics may be provided for testimony. The audio mixing system typically has two functions: (1) to feed a sound reinforcement system so

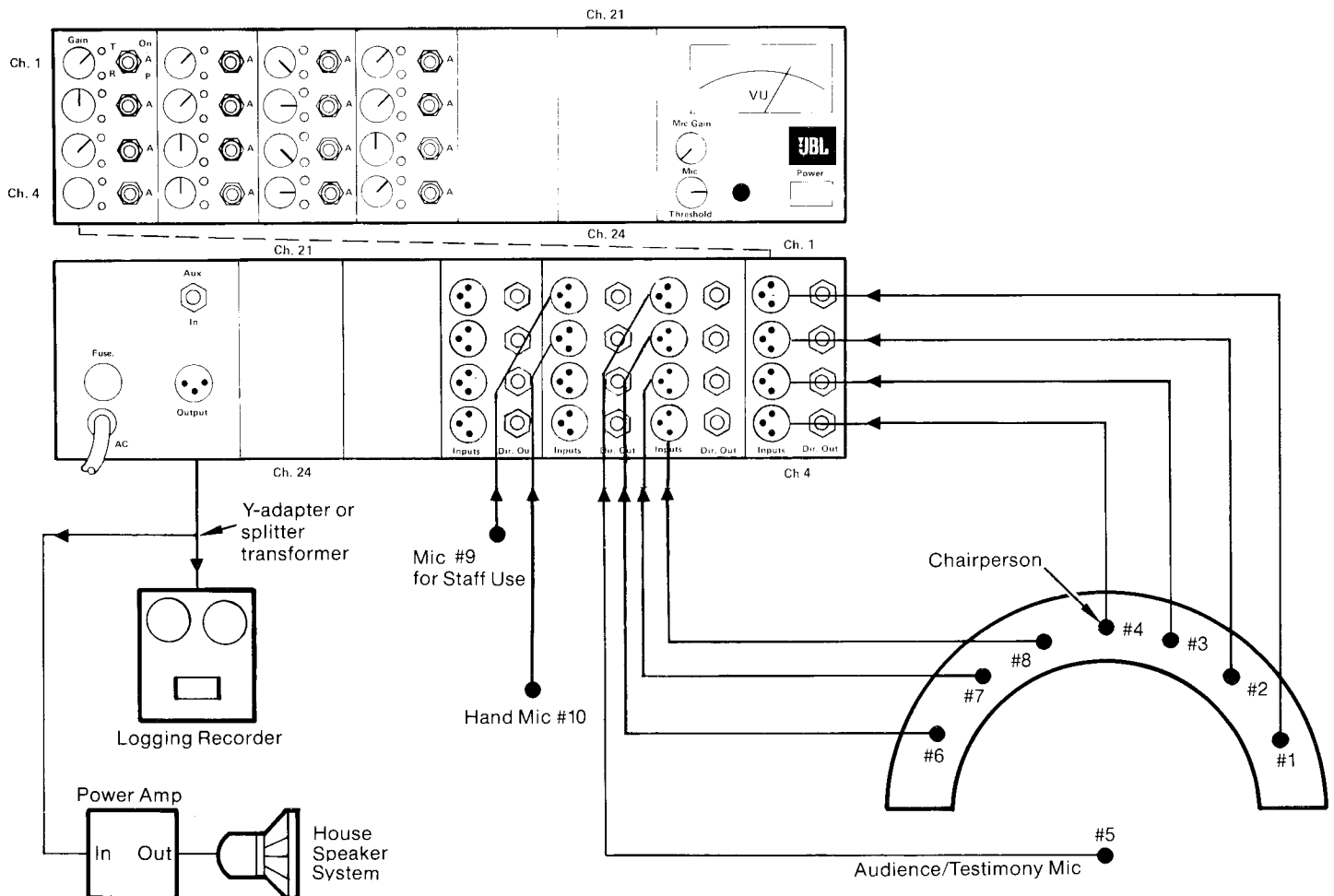
that everybody can hear the person speaking, and (2) to feed a tape recorder for documentation of the proceedings.

A common requirement in this application is the need for at least one of the microphones to take priority over all others—usually the mic used by the person who is in charge of the meeting or hearing. The 7510 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 mic 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 mic or mics, 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. The Background Threshold should be set so that if the audience becomes loud, it cannot turn On the council's Priority mics; were this to happen, the chairperson would have the least mic gain (due to the digital attenuator sensing that all mics are On) when the maximum gain is really needed on just that one mic.

In most of these situations, the 7510'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 7510 Aux Input so the unit's digitally attenuated Main Output would continue to fight feedback in the "local" reinforcement system. The outboard mixer's other outputs could feed remote speakers and/or a multitrack recorder.

Figure 9. 7510 in a meeting room sound system.



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 7510 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." Together, these features let you hold the microphone further away than would otherwise be practical. And you don't need an operator to obtain these advantages . . . it's all automatic, reliable, and 100% unattended.

Decorum is generally good in this environment, and the background sound levels only moderate at the pulpit and altar, so the temptation might be to set all inputs to Auto mode and turn off the Background Threshold.

Sometimes another approach is needed. For example, suppose the choir is located in an adjoining vestibule, with an acoustically transparent screen masking it from the congregation. The organ might be very much closer to the pulpit and/or altar, and considerably louder than the voice at that main mic.

In this instance, it might be best to set all choir mic inputs in Priority mode, and set the altar and pulpit mics in Auto mode. Also, set the Background Threshold so that the pulpit and altar mics do not turn On when the choir is singing and the organ is playing. (Surprised the pulpit mic is not the Priority mic? Read on.)

Were the altar or pulpit mic set to Priority and the choir mics in Auto mode, consider what would happen as soon as the organ played: the choir would be turned off, and the organ would be amplified by the

nearby altar/pulpit mic. Remember, when the choir is singing there is generally no need to have a "live" altar/pulpit mic. Also, when the choir is not singing, their channels' Thresholds are set so that the normal prayer, coughing, etc., do not turn ON the mics. This arrangement thus avoids unwanted amplification of already loud sound, and it prevents the feedback or "hollow" sound of having all the mics live when they are not actually in use.

With the 7510 you can still use a tape recorder or an RIAA preamplified record player; simply add the appropriate attenuation pad between that device and the 7510 input.

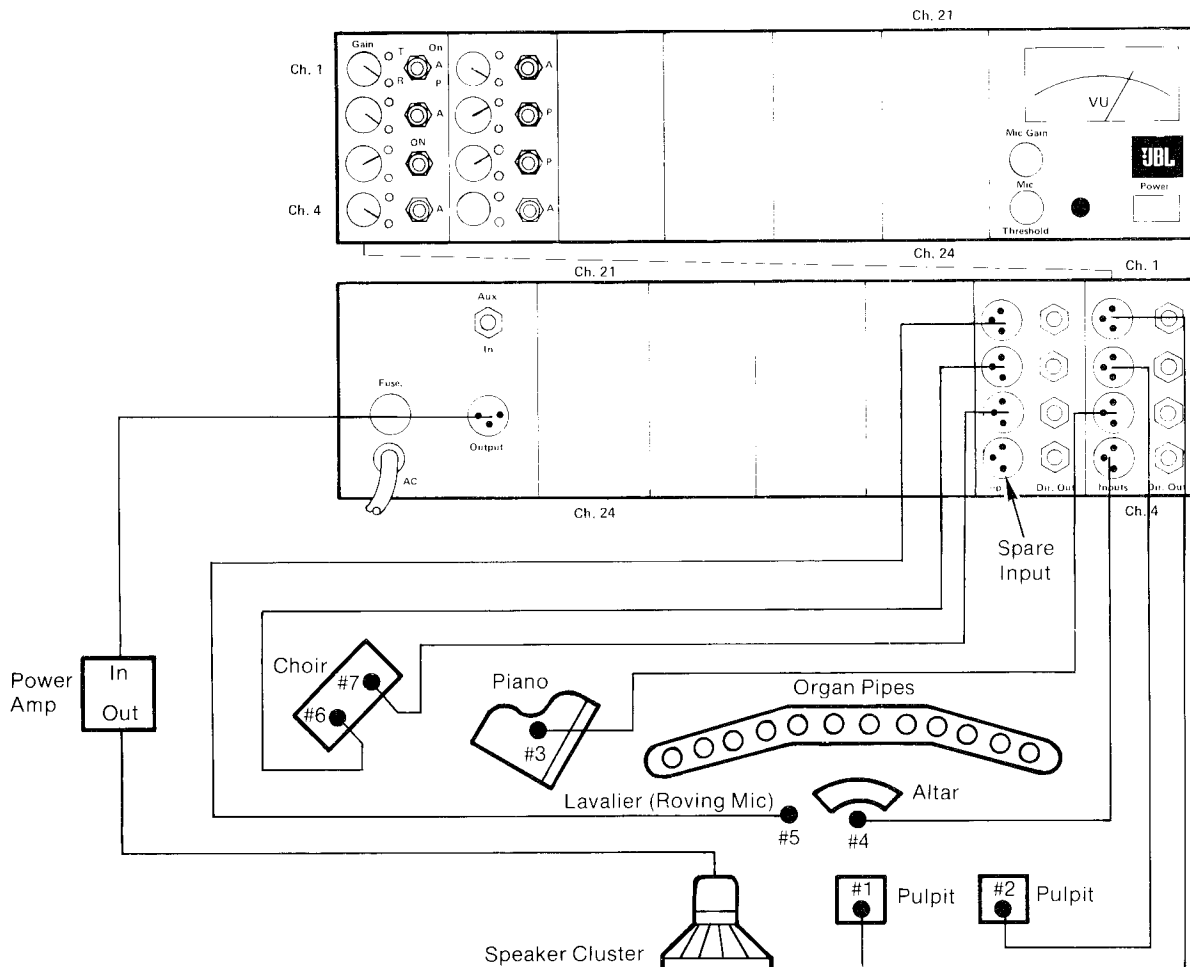
Television Quiz Show Audio

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

The 7510 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 7510's Direct Outputs are all fed to the main audio board in the control room, where the sound can be appropriately equalized, sub-

Figure 10. 7510 in a hall of worship.



grouped, and otherwise processed. A monitor mix is then returned to the 7510's Auxiliary Input for feed to the monitor amplifiers and speakers. 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 7510 Direct Outs, probably with limiting and compression added, to the VCR (or, in the less common instance of a live show, to the transmission point).

It may not be desirable to place the emcee mic channel in Priority mode and the contestant channel(s) in Auto mode, since one would have difficulty hearing the contestant's delighted screams as the emcee pronounces the winnings. Still, the emcee mic should have some preeminence. Here, the 7510 can be "tricked" into giving relatively more level to the emcee mic in comparison to any contestant who might be talking at the same time. Place all channels in Auto mode, except perhaps the announcer, who can be put in Priority mode if the producer wants him to be able to override all the other audio. The trick is to feed the emcee mic through a 1:2 splitter transformer, and feed it simultaneously to 2 of the 7510's input channels.* Set the emcee channels for equal Gain, and so that together they yield the desired output mix level. When a contestant mic is live and the emcee begins to talk, the digital attenuator will drop the contestant's level to half that of the emcee, providing the desired differential. When the contestant alone is talking, his or her mic will provide the mixer's full-level output.

* If a splitter is unavailable, the emcee mic can be plugged into one channel. That channel's Direct Output can then be connected to an attenuation pad and plugged into another input channel. One can use a 50 dB pad and set both the channels Gain controls equally. Alternately, a slightly better signal-to-noise ratio may be obtained by using a 30 or 40 dB pad, taking more gain in the mic input, and less gain in the channel fed by the pad.

For a greater preeminence of the announcer mic, split it to feed three 7510 input channels, thus giving it more "influence" in the digital attenuator than is given to the contestant mic or mics. (Too much noise may result if the Direct Out/Pad method is used for three channels).

Radio Show Audio

The 7510 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 studio because it can clean up the sound of interviews, discussions, and other multiple microphone conversational productions. The Background Threshold would generally be turned Off. The input channel Threshold controls can be set high enough that rustling papers, quiet coughs, and soft whispers do not activate the mic, yet direct speech does. 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.

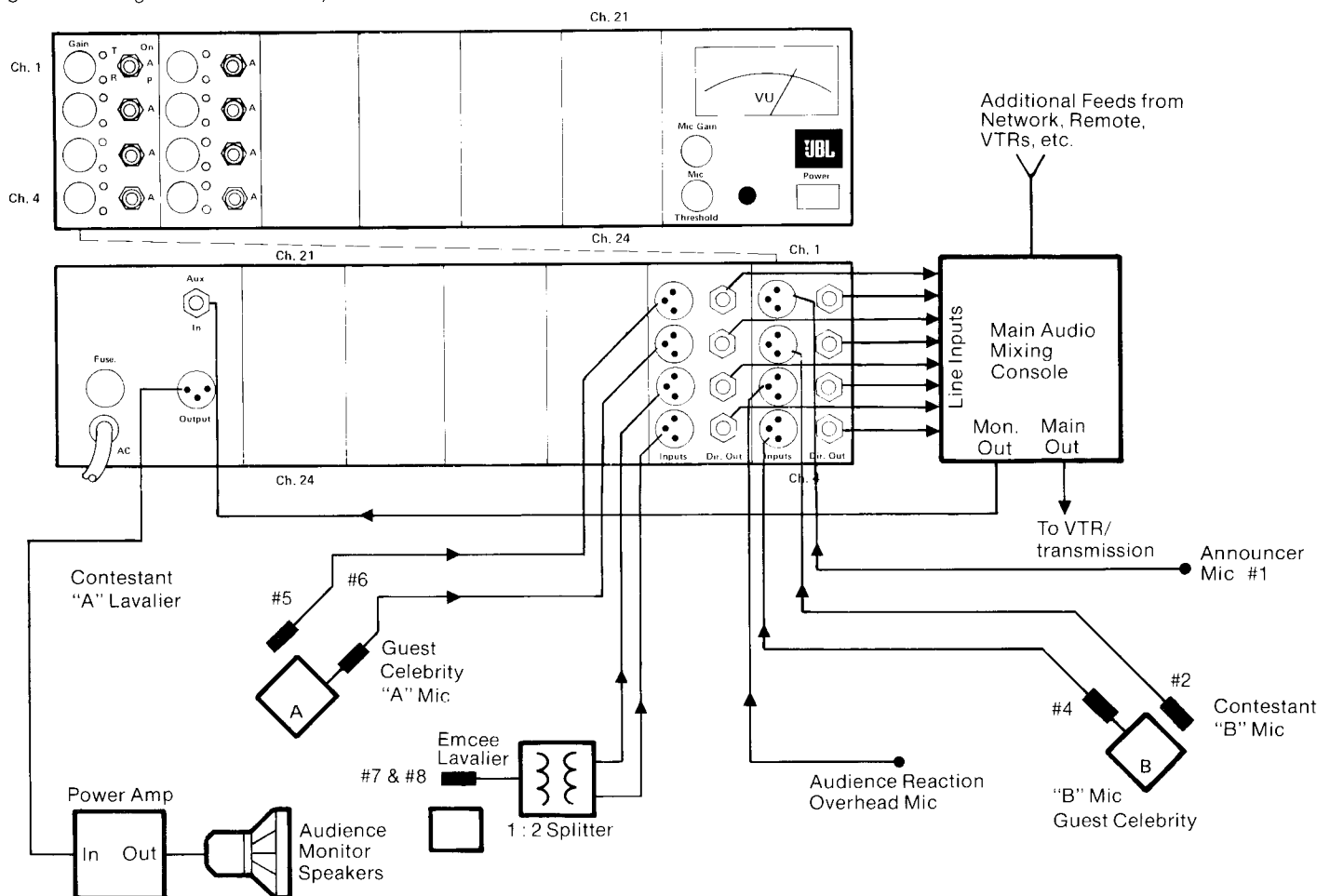
The 7510 Main Output can be fed directly to the transmission point, TELCO (telephone company) interface, or recorder. It is more likely, however, that the Direct Outputs would be fed to another mixing board, where EQ, reverb, compression, limiting, etc. may be added as required, and the program can be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

If the host is also the engineer, as sometimes is the case with automated stations, the 7510'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 7510 is placed in the studio rather than the control room, and the talk show mics are plugged into it. These channels are set to Auto mode. The 7510 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 7510. The cart feed is first dropped to mic level with a 40 or 50 dB pad, and its channel on the 7510 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 feed that is routed through the 7510. As the cartridge is playing the introduction to

Figure 11. Mixing the sound for a TV quiz show.



the show, the host walks into the studio. The instant the cart machine shuts down, as programmed, the host and guest mics become live. When the host wraps up the show at the correct time, the cart machine plays another tape through the 7510, thus killing the studio mics 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 7510 and fades in the cart machine's direct feed to the main audio board.

Recording Studio Noise Gate

In the recording studio the 7510 might 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 7510—that of a multi-channel noise gate. Almost any vocal or instrumental mic can be gated with the 7510; the application depends on the mic setup and the overall mix.

For example, suppose a drum set is being recorded with between 4 and 8 close mics. The mic 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 mic 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 mics, too. The solution is to have a mic be "live" only during the brief time its related drum or percussion instrument is sounding a note. This process is called "gating," and only a very fast, high-quality gate will do a creditable job. The JBL 7510 is such a unit.

The drum mics are connected to the 7510 channel inputs, whose individual Gains are adjusted to yield maximum level without distortion. The Background Threshold is turned Off, and the channel Thresholds are set so the mic is switched On only by the drum it covers, not by adjacent drums. 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 7510's Main Output is not used. Where tracks are at a premium, however, the 7510'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 mics are likely to be live at any given instant, the digital attenuator's gain reduction should not significantly alter the dynamics.

There are situations where the 7510 might not yield the desired result. Consider, for instance, a session where the lead vocalist also plays keyboards, and the nearby brass is leaking into the vocal mic. On the surface, it might seem a good idea to gate the vocal mic and eliminate the leakage. However, if the leakage is very strong, then when the vocal mic does turn On, there will suddenly be a different brass sound due to the introduction of random brass leakage. If the leakage were minor and the studio were fairly "dead," gating the vocal mic could be advantageous.

Concert Sound Reinforcement Noise Gate

In live musical performances, the sound levels on stage can be extremely high, thanks to amplified instruments and powerful stage monitor speakers. There also may be from 24 to 48 open mics, 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-mic 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 mics which are not actually being used from one moment to the next, but this can be done easily with the 7510. The setup is essentially the same as that described in the previous description of a recording studio application.

Using the 7510 to gate Off inactive stage mics will benefit both stage monitor mixing and house mixing. The 7510's Direct Outputs can each be fed to a splitter transformer (or to a simple "Y" adapter if both mixers have actual line input impedances of over 1 kΩ). While the 7510's digital

Figure 12. Automatic mic mixing and switchover between a radio talk show interview and automated tape cartridge programming.

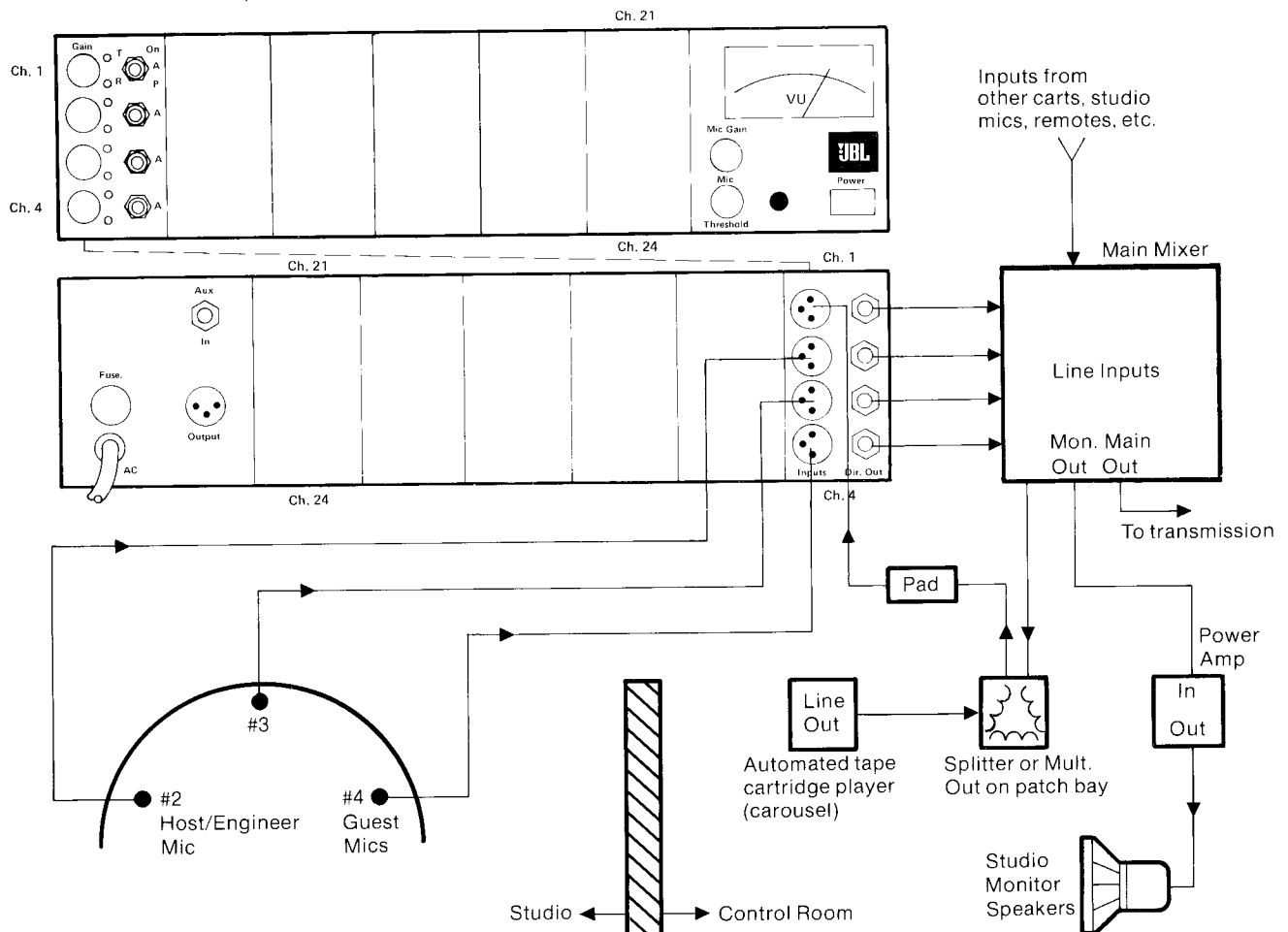


Figure 13. Using the 7510 as a studio noise gate.

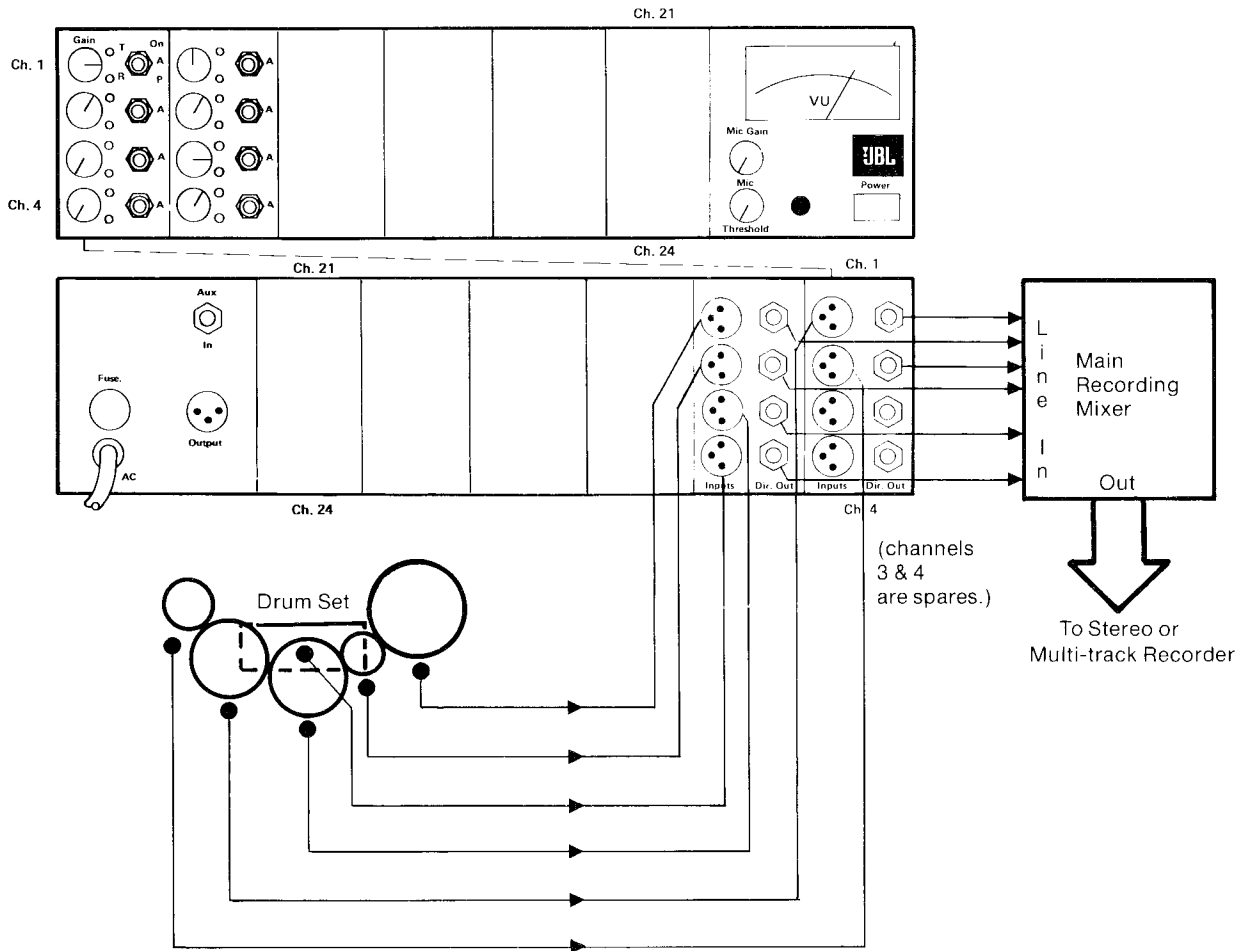
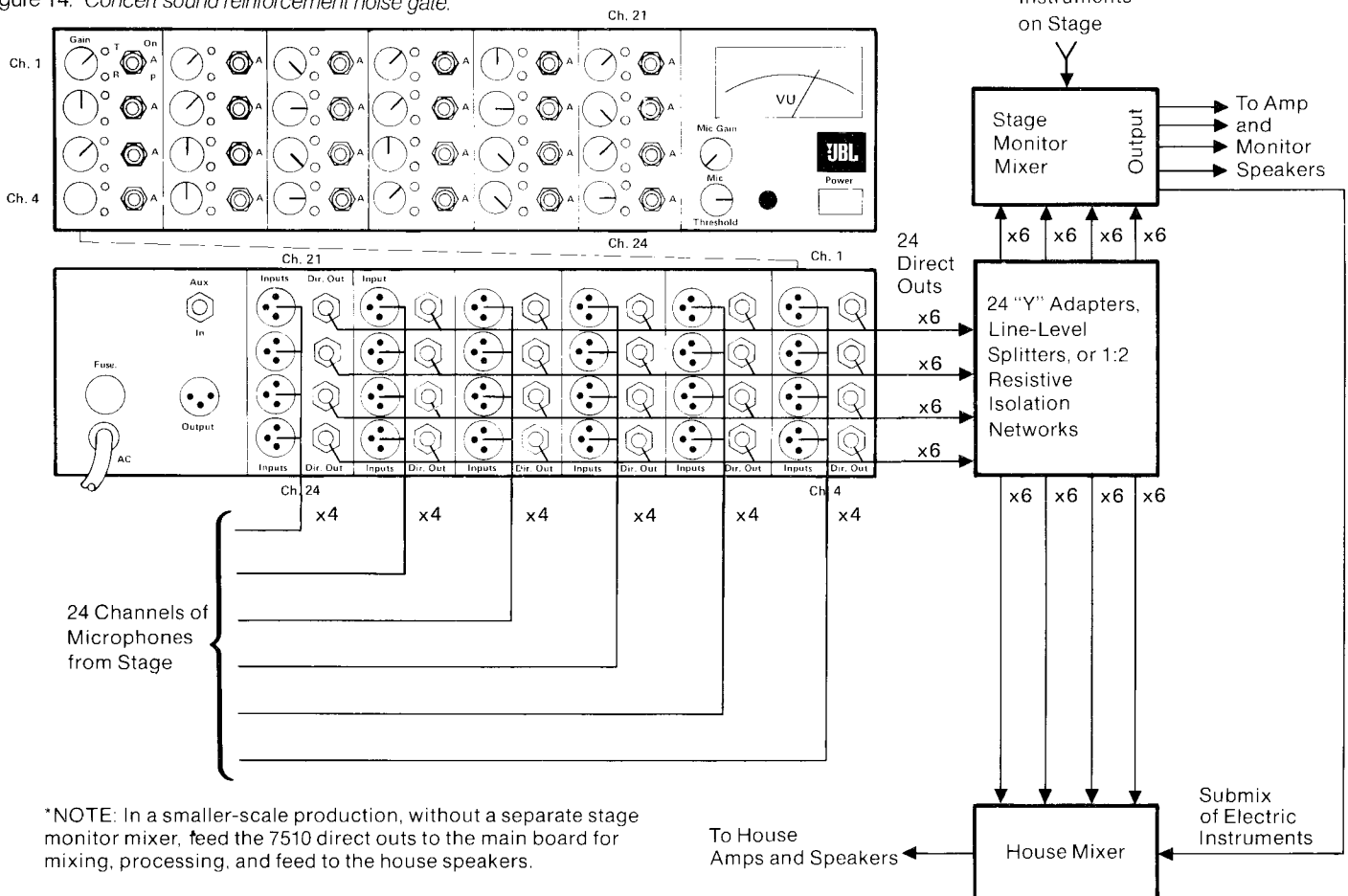


Figure 14. Concert sound reinforcement noise gate.



*NOTE: In a smaller-scale production, without a separate stage monitor mixer, feed the 7510 direct outs to the main board for mixing, processing, and feed to the house speakers.

output attenuator is not being used, there is typically an increase of from 5 to 10 dB usable gain before feedback simply because fewer mics are live at any given moment. Of course, such a setup must be tested thoroughly in a sound check with the maximum number of mics 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 no separate stage monitor mixer, the 7510's Direct Outputs can be fed to the main board for mixing, signal processing, and feed to the house speakers. Locating the 7510 at the stage is advantageous because it provides low impedance line-level feeds back to the main board, avoiding susceptibility to hum and noise. A mono submix of those mics can be done on one of the main audio board's busses and fed back to the 7510's Aux Input. The 7510's Main Output can then be fed to the monitor amplifier and stage monitor speaker(s). In this instance, the 7510's 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 7510 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, which is obviously not the ideal approach for obtaining the best sound. The 7510 can do wonders in this application. The channel Gain controls for the stage mics 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 volume control. Other settings can be preset by the 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, ensuring that the channel will turn ON whether used for quiet female vocalist's mic or a brass player. To prevent unwanted "keying" of the mics by loud audiences, or leakage from off-mic instruments, the

Background Threshold is turned On and adjusted per the normal setup procedure; when in doubt, assume the background sound will be louder and set Background Threshold at the higher of the estimated range of background levels.

The Main Output will feed the power amplifier and speaker system, which is likely to include graphic equalization. Stage mics can be set to Auto mode, and the host/announcer mic 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 mic, speak directly into in a firm voice, and all the stage mics will be silenced.

This basic system can be enhanced as desired. An FM tuner can be run through a 40 dB pad and connected to one of the 7510's input channels, which is set to Priority mode. When the channel Gain is up, the channel will turn ON and silence the stage mics, as is desirable during breaks or when no group is playing. When the stage mics are to be used, just turn down the tuner channel's Gain. Alternately, the tuner might be located at the bar. If the 7510's "tuner" channel gain is left up, the bartender can remotely turn OFF the stage mics and bring up the background music by turning up the tuner volume.

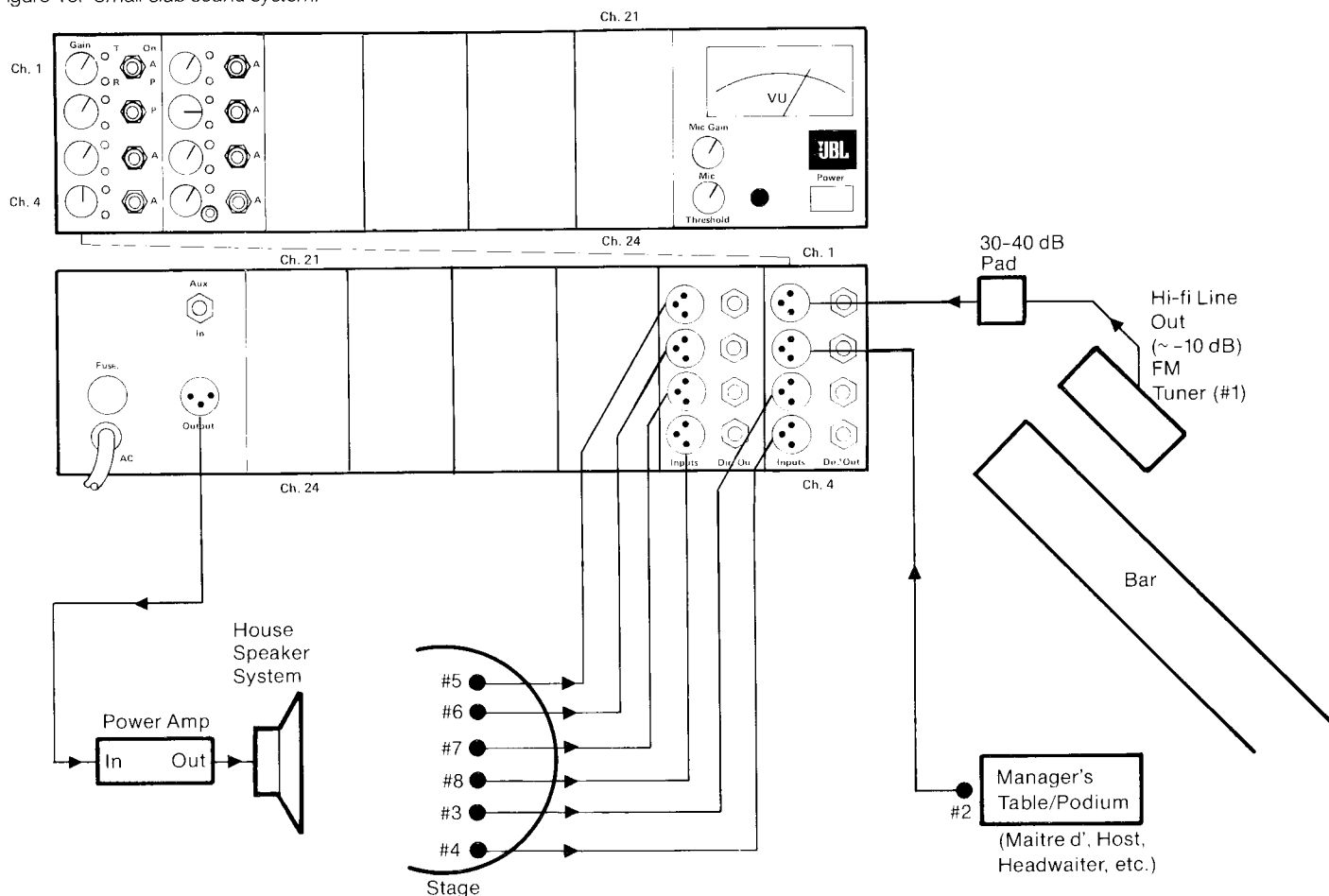
If reverb, channel EQ, or other effects are desired, the 7510 Direct Outputs can be connected to another mixing board, graphic equalizer, reverb, etc., and the external processor's output can be brought back into the 7510's Aux Input.

More About the Priority System

The 7510, 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 7510 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 very interesting.

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

Figure 15. Small club sound system.



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 mics for these drums into the 7510, set the kick channel to Auto mode and the tom (or as many tom mics 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 7510 Main Output to the recording console; otherwise use the Direct Outputs of these two 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 dies away. Other mics are either set on ON mode, or are not run through this 7510.

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, in the small club sound system application, suppose the lead vocalist walks through the audience and speaks with patrons. Sometimes their voices are at pretty low levels, and, if the Background Threshold circuit has raised the overall threshold due to the high ambient noise, the patron may not reliably actuate a channel set to Auto mode, so On mode makes more sense.

Another 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 mic could be set to On so that when the announcer is speaking the contestants are silenced, but the emcee can converse with the announcer.

MORE INFORMATION ON THE 7510 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 7510, while it acts like a noise gate, is hardly conventional. Because of its background threshold circuitry, the 7510 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 7510 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.

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 will be out-of-phase from one another. 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 mics which pick up the same sound and inadvertently create a comb filter) can significantly color the sound. The hollow, thin, or "swishy" effect known as "off mic sound" is a comb filter effect.

Because the 7510, in Auto mode, keeps a channel OFF until its mic is directly in use, it does not create comb filters as does a conventional mixer, where many mics 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 source, 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, you could increase the gain and then 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 you turn up the gain, 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 its 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. Keep fewer mics live, and you can get more usable gain (more amplification) from the remaining mics.

The problem with keeping fewer mics live in a complex sound system is that the mics 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 mic usage, and to turn down the overall system gain proportionately to the number of mics in use. Fortunately, the 7510 can keep track of the number of live mics, and on an instantaneous basis the digital output attenuator adjusts the system gain exactly as required. What's more, if a given mic 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 7510 is substituted for a conventional microphone mixer.

How the Digital Output Attenuator Works

When a 7510 input channel is turned On, it not only feeds an audio signal to the audio mixing bus and summing amp, it also feeds a digital signal to the digital mixing bus and summing amp. The digital summing amp 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, a 16-bit high-resolution microcomputer of sorts. The attenuator is programmed to reduce an audio amplifier's gain such that for every doubling of the number of live mics, the gain drops 3 dB. The actual gain reduction required is logarithmically scaled, and is incremented for each additional live mic. (See the table below.) The Master Gain control is a conventional fader which follows the digital output attenuator.

Figure 16. 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

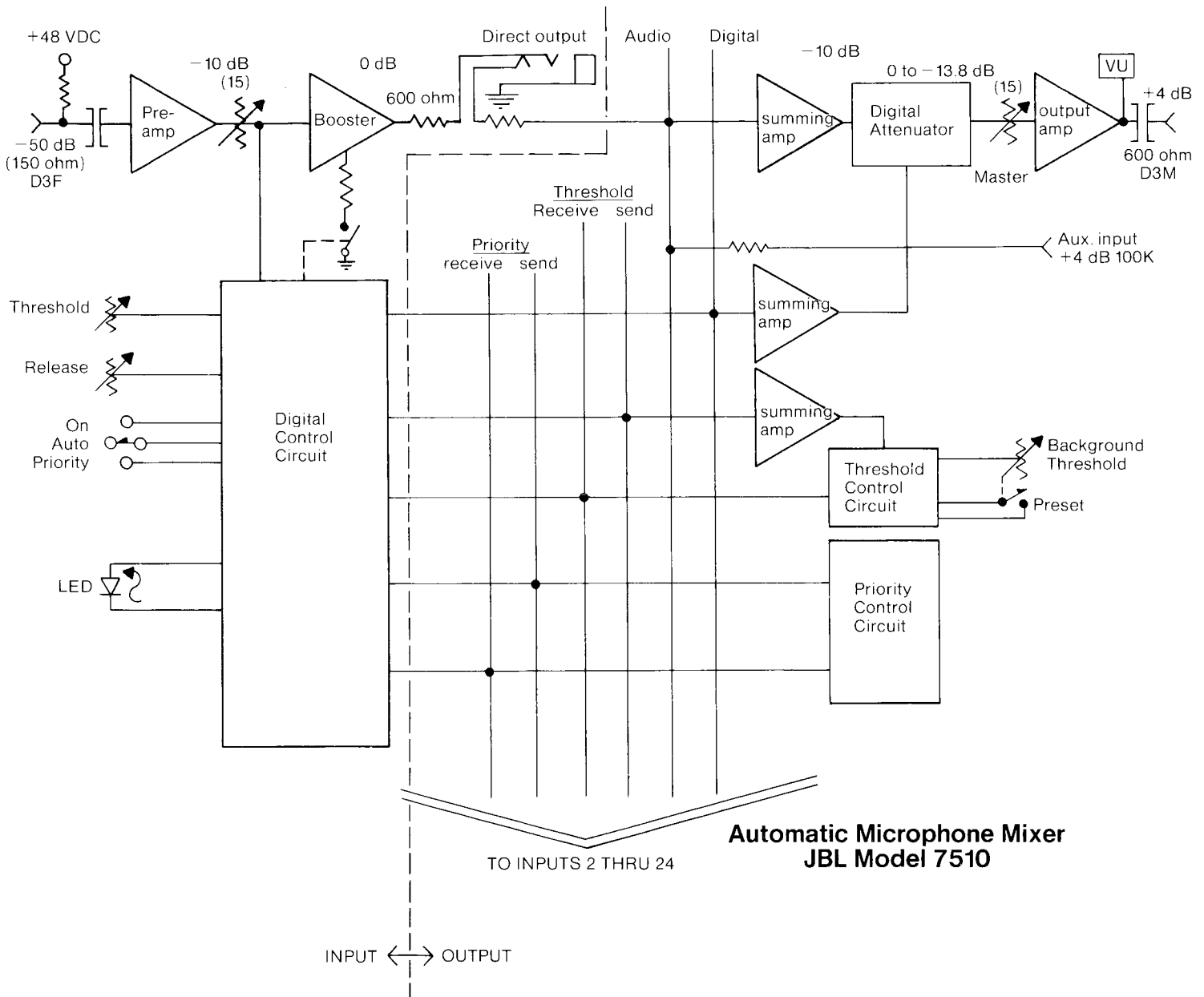
How Automatic Digital Attenuation Differs from Compression or Limiting

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

A compressor doesn't "care" how many mics 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 doesn't let the level get above threshold at all. In either case, more sound level (not more open mics) is met with more attenuation. The result is that a compressor/limiter can be set so a system can't go into feedback, but it then affects the program dynamics regardless of how many mics are being used. The 7510 does not affect program dynamics so long as any given number of mics are On.

Another contrast between compression and the 7510 digital output attenuator can be seen when you consider the background noise level. With most compressors, if fewer mics are being used, the overall gain is actually increased, thus increasing the audible background noise and leakage. With the 7510, if fewer mics are being used, those mic channels are turned OFF, thus reducing the noise and leakage.

Figure 17. JBL Model 7510 Automatic Microphone Mixer signal flow/block diagram.



SPECIFICATIONS

Maximum Gain

Input Module	65 dB minimum (600 Ω load) 71 dB minimum (10 k Ω load)
Output Module	9 dB minimum (600 Ω load) 12 dB minimum (10 k Ω load)
Overall System	80 dB minimum (600 Ω load) 83 dB minimum (10 k Ω load)

Main Output Characteristics

Actual Impedance	360 Ω @ 1 kHz, transformer isolated (floating)
Load Impedance	For 600 Ω or higher loads
Maximum Output Level	+23 dBm minimum (ref. 600 Ω)

Direct Output Characteristics

Actual Impedance	600 Ω , unbalanced
Load Impedance	For 600 Ω or higher loads
Maximum Output Level	+15 dBm minimum (ref. 600 Ω)

Mic Input Characteristics

Actual Impedance	1.35 k Ω typical at 1 kHz, transformer balanced
Source Impedance	Optimized for 150 Ω microphones
Nominal Input Level	2.5 mV RMS minimum (-50 dBv)
Input Overload	45 mV RMS minimum (-25 dBv)

Aux Input Characteristics

Actual Impedance	10 k Ω \pm 5%, unbalanced
Nominal Input Level	+4 dBv (1.23 V RMS) typical
Input Overload	11 V RMS minimum

Frequency Response

Input Modules	20 Hz - 20 kHz, +0, -1 dB
Output Modules	20 Hz - 20 kHz, +0, -1 dB
Overall System	20 Hz - 20 kHz, +0, -1 dB

Total Harmonic Distortion

Mic In to Direct Out	0.2% maximum, 35 Hz - 20 kHz
Mic In to Main Out	0.2% maximum, 35 Hz - 20 kHz

Equivalent Input Noise

-130 dBv minimum
(20 Hz - 20 kHz, 0 dBv re: 0.775 V RMS, 150 Ω source)

Automatic Mix Functions

Input Attack Time	10 ms to 10 μ s ($\frac{1}{2}$ waveform from 20 Hz to 20 kHz after signal exceeds set threshold)
Input Rise Time	30 - 60 ns typical (tolerance for turn-on once attack occurs)
Input Release Time	100 ms to 5 s, adjustable, typical
Channel Attenuation	26 dB, \pm 1 dB, adjustable
Feedback Prevention	3 dB typical 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
Release Time	Screw-adjustable linear taper potentiometer
Mode Switch	3-position toggle switch
Power Switch	Push ON/Push OFF

Indicators

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

Phantom Power

48 V DC typical on mic inputs

Power Supply

100-120 V AC, 50/60 Hz,
0.5 A fuse
Other mains voltages available outside the U.S.A.

Connectors

Mic Inputs	D3F (female 3-pin panel socket)
Main Output	D3M (male 3-pin panel socket)
Direct Outputs	6.3 mm ($\frac{1}{4}$ in) phone jacks (3-circuit)
Aux Input	6.3 mm ($\frac{1}{4}$ in) phone jack (3-circuit)

Dimensions

483 mm x 133 mm x 292 mm deep
19 in x 5 $\frac{1}{4}$ in x 11 $\frac{1}{2}$ in deep

Weight

Mainframe & 1 Input Module (4 Channels)	Net, 6.1 kg (13 $\frac{1}{4}$ lb) Shipping, 6.9 kg (15 $\frac{1}{8}$ lb)
1 Input Module Alone	Net, 0.7 kg (1 $\frac{1}{2}$ lb) Shipping, 0.9 kg (2 lb)

Mounting

Occupies 3 EIA standard rack spaces

Panel Finish

Semi-gloss baked enamel, dark gray; white nomenclature

NOTE: All dBv ref. 0.775 V

MAINTENANCE/SERVICE

Troubleshooting Chart

SYMPTOM	RECOMMENDATION
You hear a click regularly when any input channel automatically turns On.	DC offset adjustment is probably required. Refer to qualified service personnel.
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 mics are not directly in use.	Channel threshold may be too low (turn screw CCW), or Background Threshold may be too low.
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 Aux Input, or the Direct Out phone jacks's normaling 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 ensure 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 level is too high—use attenuation pad. Source device has high output impedance—use matching transformer. Channel is defective—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 someone turned up the amplifier gain or moved the microphones.
Fuse blows repeatedly as soon as 7510 is turned On, or shortly thereafter, whether or not the unit is connected to other sound equipment.	Mains voltage is too low or high—check specifications and measure the mains voltage; or fuses are of incorrect value; or 7510 power supply is defective.*
Fuse blows only when 7510 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 7510.*

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

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There are only two internal adjustments on the 7510. A DC offset adjustment for the zero-crossing point eliminates any switching noise that might develop in the digital attenuator. The second adjustment sets the lower level limit of the background threshold circuitry, and must be reset only when the input capability is expanded by one or more modules, as explained in the preceding Section of this manual.

The sophisticated logic circuitry in the JBL 7510 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 7510, contact the JBL factory for a return authorization or the name of the nearest authorized 7510 service facility.

No regular maintenance is required for the 7510. The painted surfaces may be cleaned using a sponge dampened with mild detergent and water solution; solvents, 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.

INSTALLING INPUT MODULES

The 7510 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 7510, two procedures are required, (a)

physical installation of the module or modules in the mainframe, and (b) readjustment of the internal trimmer for the low level background threshold circuitry. (If a defective module is being replaced and input capacity is not being expanded, the internal trimmer need not be readjusted.)

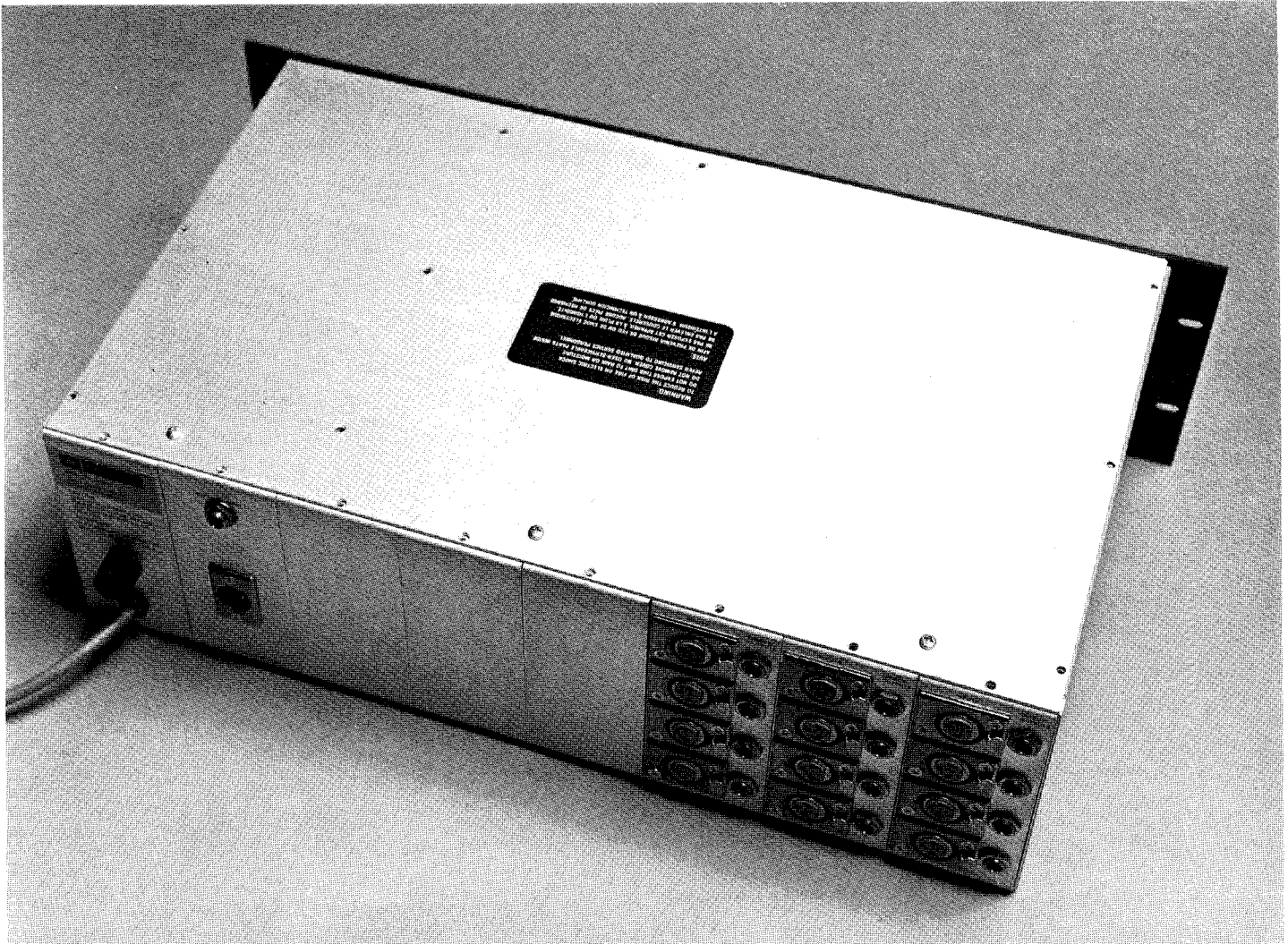
CAUTION: Whenever installing a module, make sure the AC power is first turned OFF and the AC power cord is unplugged from the mains.

Top Cover Removal and Replacement

It is necessary to remove the 7510 top cover in order to install an input module. Unscrew the 10 #6-32 machine screws and 8 #8-32 machine screws from the top cover. There are three other screws on the top cover, just inboard from the rear panel, that hold a metal strip which aids in fitting the cover to the unit. These should be loosened one turn; do not remove them. Lift the cover from the unit, pulling it slightly towards the front of the unit.

When re-installing the cover, be sure to place the proper type of screw in each location. Since there is some play in the output section heat sink/bracket, we recommend you first install the 3 #6-32 cover screws that mate with the heat sink, then insert and partially tighten the rest of the screws. When all screws have been started in their holes, tighten them. Be sure not to overtighten any screws as this may strip the threads in the module frames.

Figure 18. Top Cover Screw Locations.

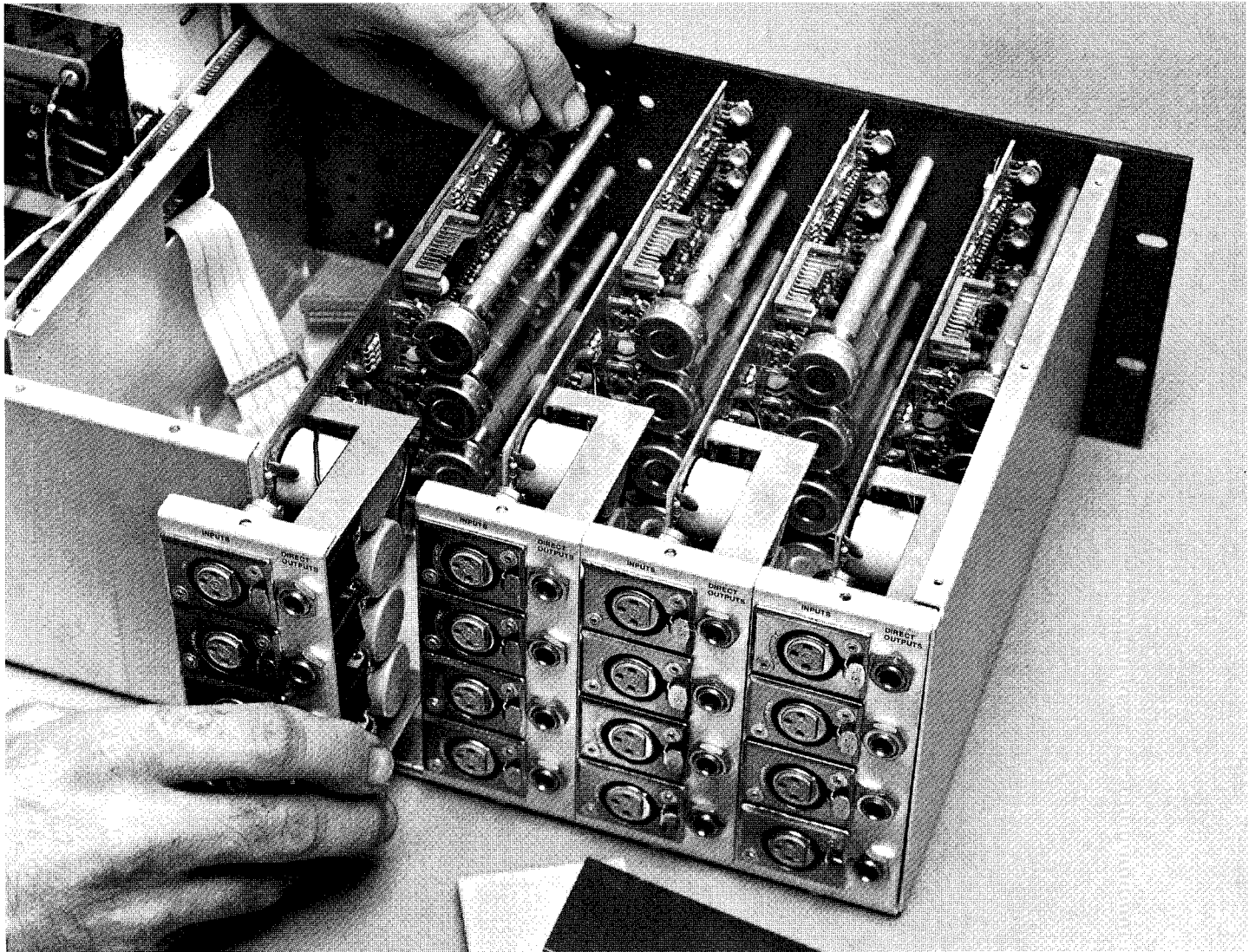


Insertion of a New Input Module

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.

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Figure 19. *Installing a New Input Module.*



Remove the single #6-32 Phillips-head screw which holds the bottom of the rear filler plate, and set the screw aside. The plate may be discarded unless future removal of the input module is contemplated.

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 7510 front panel has already been labeled to accommodate the new input module. The plate, nuts and washers may be discarded unless future removal of the input module is contemplated.

Temporarily remove the nuts and knobs that are installed on the new module so the bushings can be inserted through the 7510 front panel. Lower the new input module into the chassis, sliding it forward to the control and switch shafts and LEDs protrude from their respective front panel holes. Install the nuts to secure the new module's toggle switches to the 7510 front panel. Then, using the screw saved from the rear filler plate, secure the back of the new module to the chassis bottom cover. Attach the flat ribbon cable to the module. Finally, install the Gain control knobs, making sure the knobs are aligned to the front panel markings at maximum CW or CCW rotation.

When expanding input capacity, before returning the 7510 to normal operation, we recommend that the low level background threshold trimmer be recalibrated in accordance with the procedure on *page 22*.

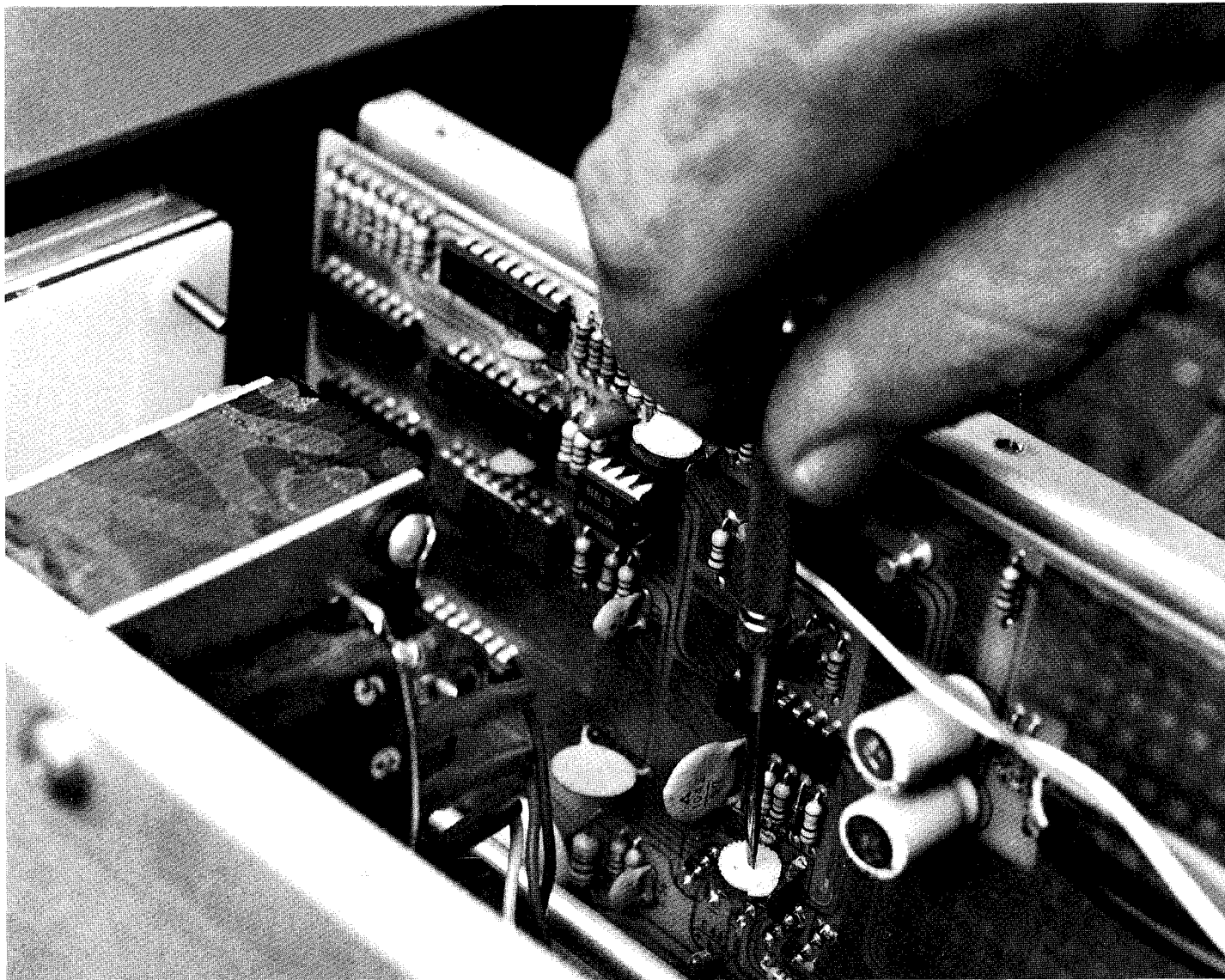
NOTE: In an "emergency" rush situation, or when replacing a malfunctioning input module rather than expanding the input capacity, the top cover could now be re-installed, and the unit returned to normal operation.

Replacing a Defective Input Module

Remove the chassis top cover as previously described. Then remove the screw which secures the back of the module to the chassis bottom cover. Remove the Gain control knobs from the module. Remove the dress nuts from the toggle switches, disconnect the flat ribbon cable, and withdraw the module from the chassis. Follow these steps in reverse order to install the replacement input module.

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Figure 20. Location of low level background threshold trimmer, RT-10.



Calibration of the Low Level Background Threshold

Whenever the 7510 input capacity is expanded with one or more new input modules, the low level background threshold trimmer must be recalibrated as follows.

NOTE: While they affect the same circuitry, the external Background Threshold control and the internal threshold trimmer perform different functions, so resetting the external control cannot substitute for this trim adjustment procedure. (The external control sets the high level threshold, while the internal trimmer sets the low level threshold.)

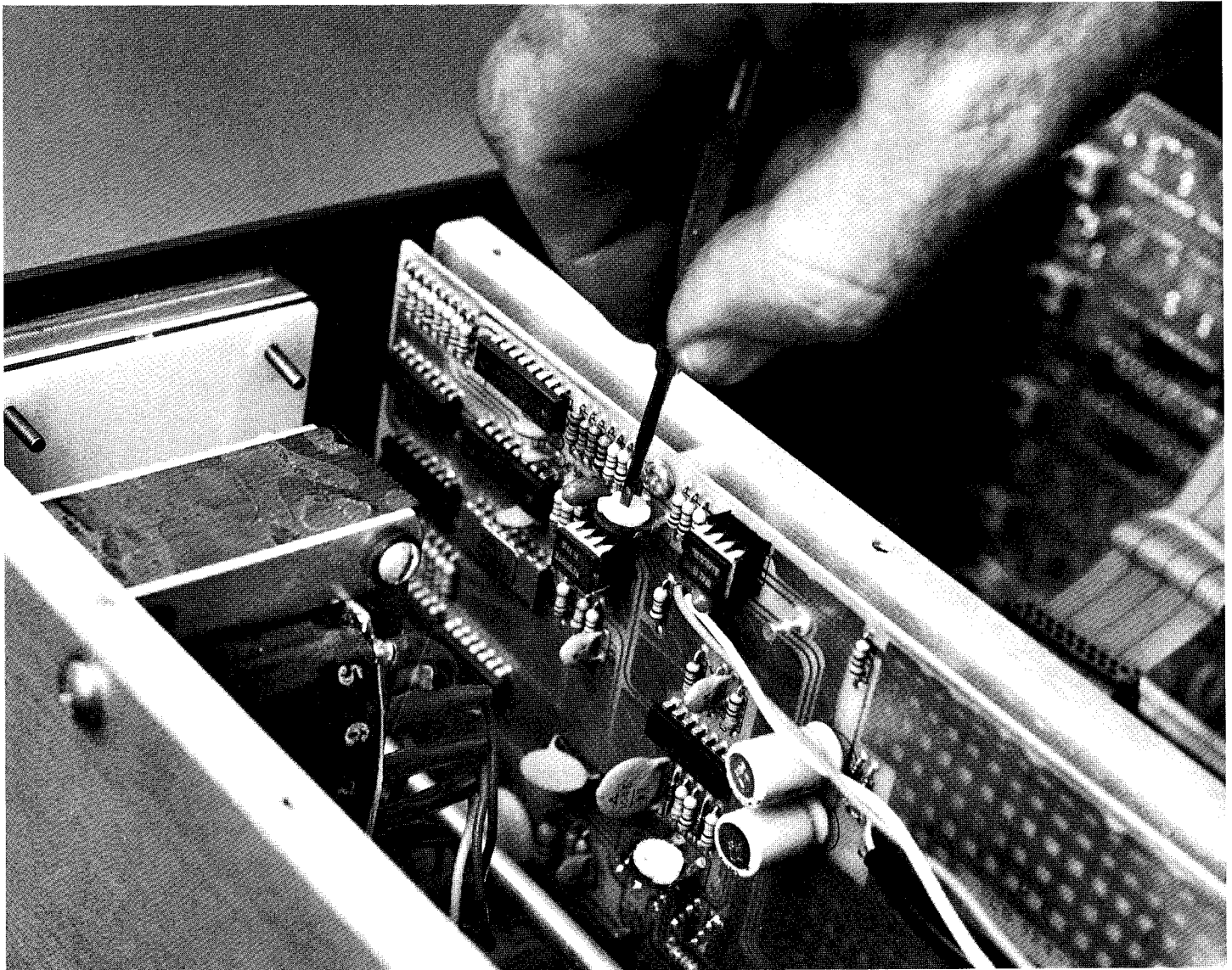
1. Disconnect all mics.
2. Set all input channels to Auto mode.
3. Place the top cover over the chassis (but do not secure with screws). Plug the unit into the AC mains, turn On the power, and let the 7510 warm up for 15 minutes. Some mic-On LEDs may be illuminated, even though no mics are connected.

NOTE: In the following steps, the Release Time and Master Threshold controls will be temporarily readjusted. If your 7510 is critically set up for a given mixing application, you may wish to write down these settings so the unit can be quickly returned to normal operation.

4. Set all front panel Release Time controls for the fastest response (maximum counterclockwise [CCW] rotation).
5. Set the front panel Background Threshold control at #6.
6. The internal trimmer is located at the center of the 7510 output module, and is labeled "RT-10." Turn the trimmer clockwise until any illuminated mic-ON LEDs just turn OFF. If no LEDs are illuminated, rotate the trimmer CCW until one does turn ON, then rotate the trimmer clockwise (CW) until the LEDs turn OFF.
7. a) Rotate the Background Threshold control back to #0, then slowly from #0 to #10, and observe to make sure no LED's turn On.
b) If an LED should turn On, stop turning the Background Threshold control at the turn-on point, and increase the RT-10 trimmer setting (CW rotation) until the LED turns Off.
c) If step (b) above was necessary, then repeat steps (a) and (b) until no more mic-ON LEDs turn ON when Background Threshold is rotated between #0 and #10.
8. This completes the trimmer calibration. Turn OFF AC power, re-install the top cover, connect all microphone cables, set up and adjust the 7510 in the normal fashion.

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Figure 21. Location of DC offset trimmer RT-9.



DC Offset Adjustment

If there is a pop or click at the Main Output when an input channel turns On or Off, as is caused by an above-threshold signal when the channel is in Auto mode, the digital output attenuator's DC offset voltage trimmer requires adjustment. This adjustment may become necessary as a result of component aging, or when certain components in the output module are replaced. The procedure is as follows:

1. Unscrew the top cover (see *Installing Input Modules*, page 20).
2. Leave the top cover in place, and turn On the 7510 power. Warm up the unit for 15 minutes to fully stabilize all components.
3. Set all input channels to Auto mode, and turn the channel Gain controls all the way down so that no mics turn On.
4. Locate the dc offset trimmer on the output module, close to the front panel and near the top of the mixer. The trimmer is labeled "RT-9."
5. Connect the positive (+) lead from a high impedance digital voltmeter to the test point adjacent to trimmer RT-9, and connect the negative (-) lead to the 7510 chassis. Measure and note the voltage (initially, use the 100 mV range).
6. Switch one input channel ON, and again measure the voltage across the test point and chassis ground. The two measurements should be within 10 mV of each other. If not, adjust RT-9 so the voltage change between one input switched On and all inputs switched Off in Auto mode is as small as possible (about 1 mV).
7. Disconnect the voltmeter leads, reinstall the top cover, and return the 7510 to normal operation.



Professional Division

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