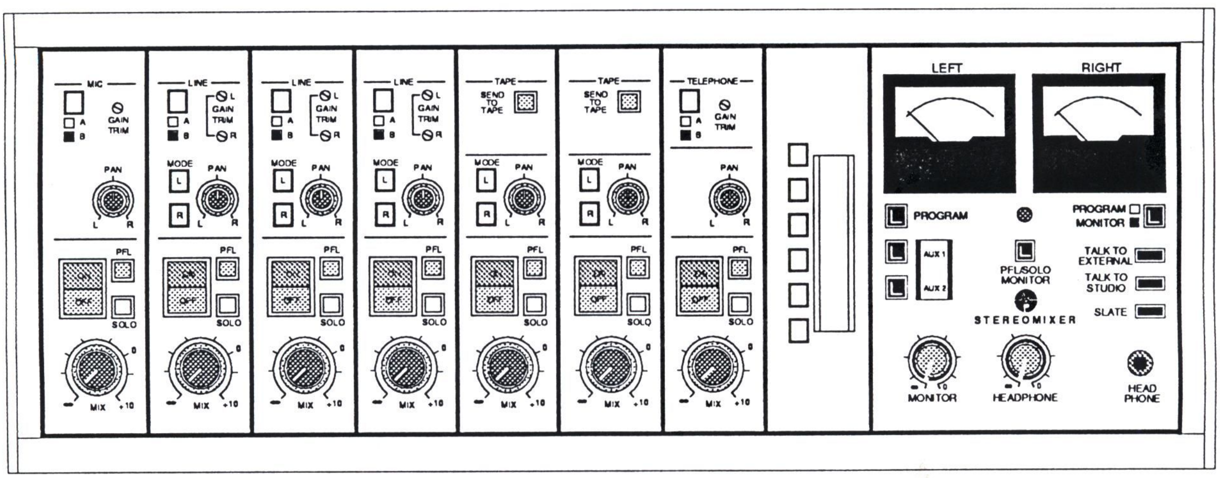
Stereomixer Control Console

Technical Manual



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This listing provides a reference of current pages of this document, and their revision numbers (i.e., A.1, A.2, etc.). When a revision to this document is received from PR&E, simply replace the old pages with the new ones, discard the old pages, and post the new status page in the front of this manual (**NOTE**: It may be desirable to retain replaced status pages in order to have a record of document changes). If deemed necessary by PR&E's Engineering Department, comment information relating to any change may also be included on this page.

Page No.	Revision	Comments
ALL	A	New Release (April 1989).
TOC	A.1	Added Chapter 7 to Table of Contents.
3, 4	A.1	Added Patch Send and Return specifications.
10, 11	A.1	Changed interface reference to Chapter 7. Expanded logic command descriptions.
110-129	A.1	Added chapter 7 ("INTERFACES") to existing manual.

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GENERAL INFORMATION

STEREOMIXER CONTROL CONSOLE
TECHNICAL MANUAL

1.0 GENERAL INFORMATION

This chapter contains an introduction to the STEREOMIXER Control Console Technical Manual, an overview of the STEREOMIXER's features, specifications and warranty information.

1.1 INTRODUCTION

Congratulations on your decision to join the growing ranks of Pacific Recorders & Engineering Corporation (PR&E) broadcasters. PR&E is in the business of supplying the finest audio systems to the world's leading broadcast facilities. Your decision to go with PR&E means that you expect more than simple working hardware. Please be assured that it is our strong desire to provide each of our customers with the kind of products, systems, documentation and support that we would specify if we were in your position.

We invite your comments and suggestions for improvement of this document, and of all our services. By constant attention to our customer's needs, we will continue to earn our reputation for excellence, and to refine our understanding of the requirements of the marketplace.

This manual is designed to provide the information required to understand, install, operate, and maintain the STEREOMIXER Control Console. It is assumed that the reader has a working knowledge of audio control consoles, systems, and installation practices. The STEREOMIXER is a sophisticated console with an extensive range of features and capabilities. To obtain the maximum benefit of the console's capabilities, it is strongly recommended that the Installation, Operation, and Equipment Description chapters of this manual be read thoroughly prior to installing the unit.

Each STEREOMIXER is specifically configured to the customer's requirements, thoroughly tested, and "burned-in" prior to packing for shipment. Should you encounter any difficulty during installation or initial operation, we recommend that you contact PR&E for assistance.

1.2 OVERVIEW

The STEREOMIXER is a compact full-featured audio mixing, routing and monitoring system incorporating many of the features of the larger BMX Series of broadcast consoles. Each STEREOMIXER is configured for a specific application by the installation of appropriate plug-in input or accessory modules. The STEREOMIXER comes equipped with a System Master Module and a plug-in regulated power supply. The power supply assembly is installed directly behind the System Master Module, and includes the System Master Module connector panel, which can be accessed from the rear of the mainframe.

The System Master Module contains the stereo mix bus amplifiers, output amplifiers, monitor selection system, slate and talkback system, and stereo VU meters with drivers. This module also provides patch points for stereo program buses, which are convenient for the connection of external signal processors.

The mainframe assembly can accommodate up to eight plug-in input or accessory modules. At present, there are four different input modules available for use in the STEREOMIXER mainframe - a Microphone Input Module, Stereo Line Input Module, Stereo Tape Recorder Input/Output Module, and Telephone Input/Mix-Minus Module. The Microphone Input Module can select either of two inputs, and is designed to accommodate a nominal input level range of -70 dBu to -30 dBu; the Stereo Line Input Module can select either of two inputs, and is designed to accommodate a nominal input level range of -30 dBu to +9 dBu; the Stereo Tape Recorder Input/Output Module serves the functions of being the input module for the playback of a reel, cartridge or cassette tape recorder, while providing an output switch and balanced line amplifiers to feed the input of the recorder; and the monaural Telephone Input/ Mix-Minus Module contains mixing, filtering and output amplifier circuitry, and can provide a special mix of the Program signal to a telco hybrid, less the input of the module itself (mix-minus).

There are also four accessory modules available for use in the STEREOMIXER mainframe - a Monaural Equalizer Module, Stereo Equalizer Module, Voice Processing Module, and Remote Line Selector. The Monaural Equalizer Module provides both equalization and filter facilities in one compact module; the Stereo Equalizer Module provides two separate, but coupled, three band equalizers; the Voice Processor Module provides a switch-insertable equalizer which covers the frequency range normally required for voice signal correction and/or enhancement, as well as a second section containing the expander, compressor, and de-esser systems; and the Remote Line Selector provides for the selection of up to six external stereo sources routed to a single input.

The input and accessory modules are described in detail in Chapters 3 and 4 of this document. Each module has a corresponding connector panel installed in the rear of the mainframe. Each connector panel is equipped with the type(s) of connector(s) and nomenclature appropriate to its module. For examples of the connector panels, and the correct wiring of each, see Section 2.7 of this document.

1.3 SPECIFICATIONS

Following is a list of specifications for the STEREOMIXER Control Console:

<u>MICROPHONE INPUTS</u>

Source Impedance	150 ohms.
Input Impedance	30K ohms minimum, balanced.
Input Level Range	Adjustable from -70 dBu to -35 dBu.
Input Headroom	Greater than 30 dB above nominal input.

HIGH LEVEL INPUTS

Source Impedance	600 ohms.
Input Impedance	Greater than 40K ohms, balanced.
Input Level Range:	
Line Input Module	Adjustable from -30 dBu to +9 dBu.
Telephone Module	Adjustable from -30 dBu to +9 dBu.
Tape I/O Module	Switchable to +4 dBu or +8 dBu.
External Inputs	Fixed at system operating level.
Input Headroom	Greater than 30 dB above nominal input.

MAIN OUTPUTS

Load Impedance	600 ohms minimum.
Source Impedance	80 ohms, balanced.
Nominal Output Levels:	
Program and Tape Outputs	+4 dBu to +8 dBu, adjustable.
Telephone Mix-Minus	0 dBu.
Maximum Output Levels:	
Program Outputs	+24 dBm, 600 ohm load.
Tape Outputs	+24 dBm, 600 ohm load.
Mix-Minus Outputs	+24 dBm, 600 ohm load.

MONITOR OUTPUTS

Main Outputs:	
Load Impedance	600 ohms or greater.
Source Impedance	40 ohms, unbalanced.
Output Level	0 dBu nominal, +20 dBu maximum.
Headphone Output:	
Load Impedance	300 ohms or greater.
Source Impedance	100 ohms.
Output Level	0 dBu nominal, +20 dBu maximum.

PATCH SENDS AND RETURNS

Patch Send Outputs	Nominal -10 dBu; unbalanced.
Patch Return Inputs	Nominal -10 dBu; 40K ohm balanced and bridging.

FREQUENCY RESPONSE

Mic Input to Program Output	+0 dB, -.7 dB, from 20 Hz to 20 kHz.
Line Input to Program Output	+0 dB, -.7 dB, from 20 Hz to 20 kHz.

NOISE

Microphone Input Amplifier	-126 dBu equivalent input noise, 150 ohm source, 20 kHz bandwidth.
Line Input Amplifier	-88 dBu equivalent input noise, 600 ohm source, 20 kHz bandwidth.

Output Noise with one microphone channel ON, fader at 0 dB, input sensitivity at -50 dBu.	75 dB below output, reference +8 dBu, 150 ohm source, 20 kHz bandwidth.
Output Noise with one line channel ON, fader at 0 dB, input sensitivity at +8 dBu.	83 dB below output, reference +8 dBu, 600 ohm source, 20 kHz bandwidth.
Output Noise with no input channels ON.	85 dB below output, reference +8 dBu, 20 kHz bandwidth.

DISTORTION, T.H.D.

Mic Input to Program Output	Less than 0.01%, 20 Hz to 20 kHz, -32 dBu input, +24 dBm output into a 600 ohm load, 80 kHz meter bandwidth.
Line Input to Program Output	Less than 0.01%, 20 Hz to 20 kHz, +24 dBu input, +24 dBm output into a 600 ohm load, 80 kHz meter bandwidth.

DISTORTION, I.M.D.

Mic Input to Program Output	Less than 0.01%, -32 dBu input, +24 dBm output into a 600 ohm load.
Line Input to Program Output	Less than 0.01%, +24 dBu input, +24 dBm output into a 600 ohm load.

SEPARATION

Program Outputs	Better than -70 dB at 1 kHz; Better than -60 dB at 20 kHz.

POWER REQUIREMENTS

Fully configured STEREOMIXER	50 watts @117 VAC, ±10%, 60 Hz.

NOTES:

A) These specifications are for the basic signal paths, per channel, with 600 ohm loads connected to the program outputs.

B) 0 dBu corresponds to an amplitude of 0.775 volts RMS regardless of the impedance of the circuit. It is the same voltage value as 0 dBm measured in a 600 ohm circuit. This enables convenient level measurement with meters calibrated for 600 ohm circuits.

C) Noise specifications are for a fully loaded STEREOMIXER. Noise specifications are based upon a 20 kHz measurement bandwidth; the use of a meter with 30 kHz bandwidth will result in a noise measurement increase of approximately 1.7 dB.

Pacific Recorders & Engineering Corporation reserves the right to change specifications without notice or obligation.

1.4 WARRANTY INFORMATION

This product carries a manufacturer's warranty which is subject to the following guidelines and limitations:

A) Except as expressly excluded hereinafter, Pacific Recorders & Engineering Corporation ("Seller") warrants equipment of its own manufacture against faulty workmanship or the use of defective materials for a period of one (1) year from date of shipment to Buyer. The liability of the Seller under this Warranty is limited to replacing, repairing or issuing credit (at the Seller's discretion) for any equipment, provided that Seller is promptly notified in writing within five (5) days upon discovery of such defects by Buyer, and Seller's examination of such equipment shall disclose to its satisfaction that such defects existed at the time shipment was originally made by seller, and Buyer returns the defective equipment to Seller's place of business in Carlsbad, California, packaging and transportage prepaid, with return packaging and transportage guaranteed.

B) Equipment furnished by Seller but manufactured by another shall be warranted only to the extent provided by the other manufacturer.

C) Thermal filament devices such as lamps and fuses are expressly excluded from this warranty.

D) The warranty period on equipment or parts repaired or replaced under warranty shall expire upon the expiration date of the original warranty.

E) This Warranty is void for equipment which has been subject to abuse, improper installation, improper operation, improper or omitted maintenance, alteration, accident, negligence (in use, storage, transportation or handling), operation not in accordance with Seller's operation and service instructions, or operation outside of the environmental conditions specified by Seller.

F) This Warranty is the only warranty made by Seller, and is in lieu of all other warranties, including merchantability and fitness for a particular purpose, whether expressed or implied, except as to title and to the expressed specifications contained in this manual. Seller's sole liability for any equipment failure or any breach of this Warranty is as set forth in subparagraph A) above; and Seller shall not be liable or responsible for any business loss or interruption, or other consequential damages of any nature whatsoever, resulting from any equipment failure or breach of this warranty.

INSTALLATION

2.0 INSTALLATION

This chapter provides instruction in the proper installation of the STEREOMIXER. Included are sections outlining general installation guidelines, cable preparation, mainframe configuration, grounding and shielding, power connection, patch point connection, audio and logic connection, remote control capabilities, and module internal option switches.

2.1 GENERAL GUIDELINES

The STEREOMIXER should be carefully unpacked and inspected for any shipping damage. If the inspection reveals any damage, file a claim with the delivering carrier. The packing material should be kept as evidence of mishandling, as well as to allow return of the equipment to the factory, if necessary.

Included with the STEREOMIXER are the tool and spare parts kits (as described in Chapter 5), the power cord, and a connector kit (PR&E #76-52), which contains all of the Molex connector housings and pins necessary to prepare the audio input/output and logic cables (as described in Section 2.2).

Signal, logic, and AC power connections are made to the connector panels located on the rear of the mainframe. The power supply module contains all of the System Master Module's connectors (reference Figure 2.2), and each input and accessory module has a corresponding connector interface panel located at the back of the console, directly behind the module.

The console is available in either a rack mount or a desktop version. The rack mount version requires four rack units of height (7 inches) in a standard rack width of 19 inches. The depth of the STEREOMIXER from the rack rails is 15-3/4 inches, however, it is recommended that the installation allow and additional two inches to accommodate rear panel connections and cable bends. The desktop version is installed by simply placing it on a table top. The optional oak end panels provide a five degree slope, making it easier for the user to see and operate the front panel controls.

Install legend strips by sliding them into the tops of the legend panels. Monitor legend strip dimensions are 0.6 inch wide by 1 inch high (15 mm by 25 mm); Remote Line Selector legend strip dimensions are 0.6 inch wide by 3.34 inches high (15 mm by 85 mm). Vertical switch spacing is 0.6 inch (15 mm).

Because the power supply is internal to the STEREOMIXER, adequate ventilation of the rear mounted regulator heat sinks must be provided for the proper dissipation of heat. This should not be a concern for the desktop version, but special attention should be taken to provide proper ventilation in rack mount applications. The supply is designed for convection cooling by use of a conservatively designed power transformer and a large regulator heatsink mounted on the rear of the power supply assembly.

<u>NOTE</u>: Care should be taken to avoid locating the console within six feet of any intense electromagnetic hum fields, such as those produced by large power transformers and motors. Likewise, cables to and from the console should be routed to achieve maximum practical distance from AC mains power wiring. Particular attention should be paid to some of the low-cost, supposedly "professional", power amplifiers which have appeared in the marketplace. In many cases the low cost has been partially achieved through

the use of small core power transformers operating at or on the edge of saturation. While these units may operate to their own specifications, the electromagnetic fields they radiate may impair the performance of the STEREOMIXER or neighboring equipment such as tape recorders, cassette decks and cartridge machines. The power transformer used in the STEREOMIXER is a toroidal type which exhibits a low radiated hum field, thereby reducing the need to isolate this console from sensitive magnetic equipment.

2.2 CABLE PREPARATION

Before beginning the installation, a plan should be drawn up showing how the system will be interconnected. All cables and connectors should be tagged with numbers and/or legends, and logged.

Only unspliced (preferably new) cables should be used in connecting the mainframe. Audio connections should be made with two-conductor stranded insulated foil shielded cable with drain wire. The cable used should be equivalent to Belden types 8451, 9451, or 8761.

Strip the cable insulation jacket and foil shield back about 1-1/2 inches, and sleeve the shield drain wire with heat-shrink tubing, leaving about 3/16 inch of the wire exposed. Then, strip the insulation of each signal wire back about 3/16 inch, and sleeve the shield (at cable ends) with heat-shrink tubing.

NOTE: It is very important to sleeve the shield drain wire and the shield (at cable ends) with heat-shrink tubing. This is the only means of assuring an installation according to recommended grounding procedures.

The Molex pins are designed so that the short tab "ears" are crimped onto the stripped wire to make the electrical connection, while the long "ears" are crimped over the insulated section of the wire to help support the connection (reference Figure 2.1).

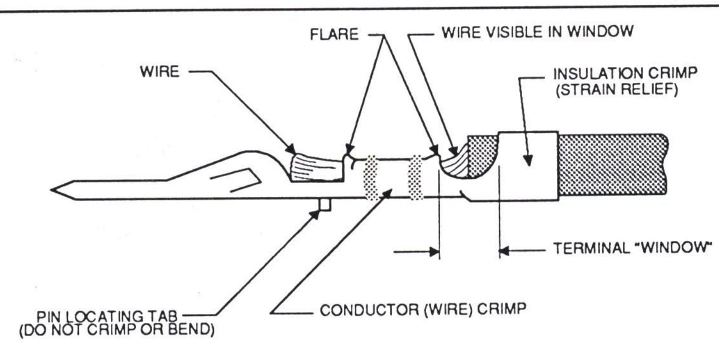


FIGURE 2.1 MOLEX CRIMP PIN

In order to crimp, insert the short ears of the Molex crimp pin into notch "B" of the crimping tool (PR&E #70-3), with the ears pointing toward the letter "B". Insert the wire into the terminal so that the stripped portion is between the short crimp ears, and the insulation is between the long crimp ears. Crimp the short ears.

Now place the long ears of the pin into tool notch "A", with the ears pointing toward the letter "A". Crimp the long ears over the insulated section of wire.

NOTE: When using Molex Crimping Tool #HTR-1719-C (PR&E #70-5), place a pin into slot "B" with the long ears on the "B" side of the tool and pointing toward the letter "B". Place the wire into the tool from the "B" side, and then crimp the pin.

Logic control cables should be fabricated in a similar manner using 22 gauge multiple conductor, non-shielded, jacketed cable. The number of conductors required will be determined by application.

NOTE: When connecting both "A" and "B" logic simultaneously, as described in Section 2.8, it will be necessary to use a "Y" cable. Prefabricated cables (PR&E #99-580) may be purchased from PR&E for a nominal charge, or may be constructed per the diagram contained in Section 6.23.

Once the pins are crimped, they may be inserted and locked into the nylon connector housing in accordance with the pin-out diagrams contained in section 2.7. A click can be felt indicating that the locking ears on the pin have set. If a pin is inserted in the wrong connector position, or it is desired to make a circuit change, use the connector pin extractor tool (PR&E #70-4) to release the pin and press it out of the connector housing.

2.3 MAINFRAME CONFIGURATION

The STEREOMIXER mainframe (reference Figure 2.2) is factory configured to each order by the

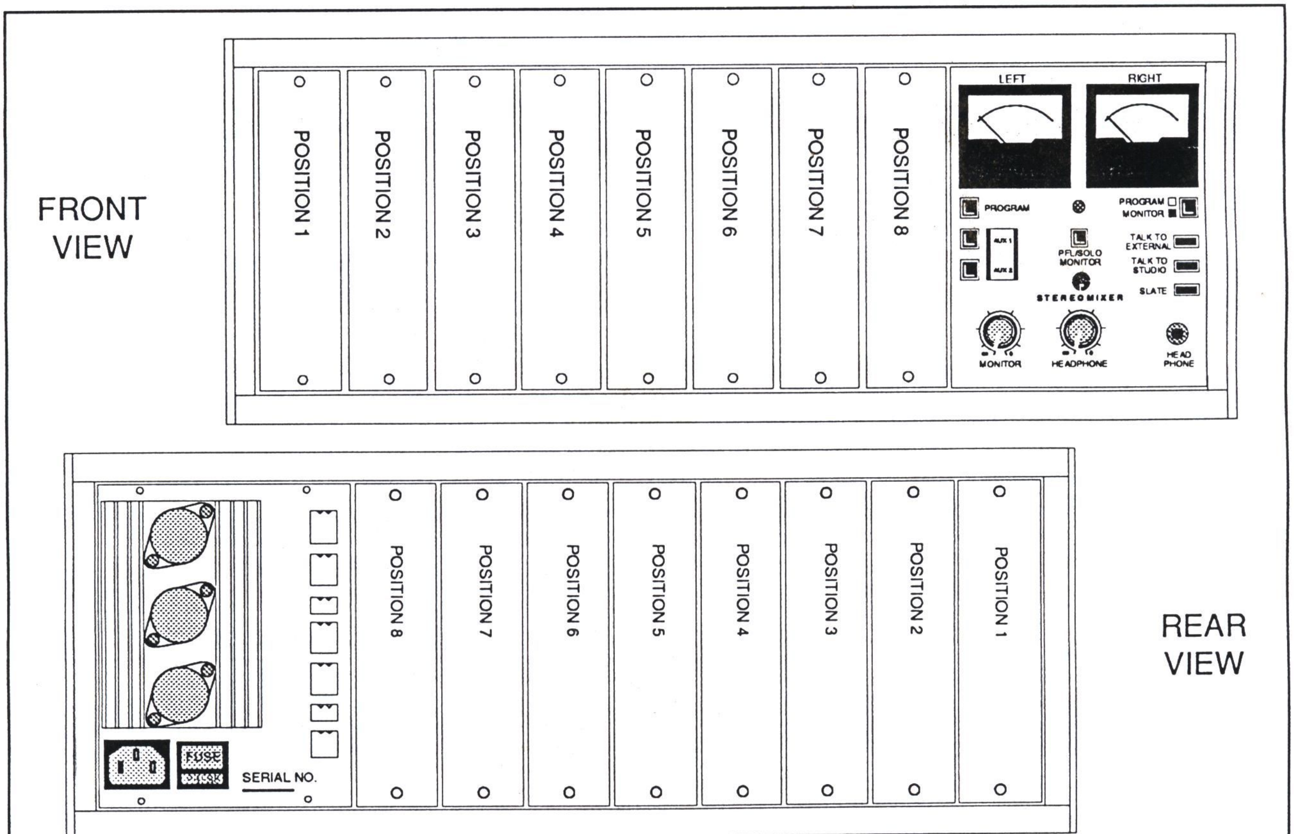


FIGURE 2.2 MAINFRAME CONFIGURATION

installation of input and accessory modules in customer specified locations. Each of the input and accessory modules installed in the front of the STEREOMIXER frame will have a corresponding connector interface module installed in the rear. Each of the connector interface panels contains the connector(s) required for audio and control of its dedicated input module.

The System Master Module can only be installed at the far right side of the console (front view), and the power supply assembly is installed immediately behind it.

NOTE: Special care should be taken when installing or removing the System Master Module and/or the power supply assembly, as their increased size and weight make them slightly more difficult to handle than the smaller input modules.

2.4 GROUNDING AND SHIELDING

Grounding in modern broadcast consoles is more critical than with older devices of more limited band-pass capabilities. Achieving low system ground impedance with a small piece of equipment is relatively easy. However, the problem becomes progressively more difficult as the system becomes larger. In designing the STEREOMIXER, much thought was given to system grounding requirements and the elimination of DC path ground loops.

The shield pins on each STEREOMIXER connector are connected directly to the mother board ground plane, which, in turn, is connected to the central STEREOMIXER ground point on the power supply assembly. The power cord assembly then provides STEREOMIXER to ground connection.

A preferred method of connecting the line shields in a system is to connect **both** ends of every shield to **all** affiliated equipment. However, this method is only satisfactory if every component shares a common earth ground. This can be accomplished using isolated ground receptacles tied to the station's "technical ground".

If isolated ground receptacles are not available when grounding the STEREOMIXER, observe the following guidelines:

A) Shields of cables connecting the console to auxiliary equipment should be connected at the console end only, and should not be terminated to the ground of the auxiliary equipment.

B) Ensure that the auxiliary equipment is connected to a "clean" ground by its power cord assembly, or by the addition of a separate ground wire connected between the chassis of the auxiliary equipment and the station's "technical ground".

NOTE: Buzz pickup is generally electrostatic, due to capacitive pickup between an audio line and a power line. When shielded lines are used this should be no problem, unless the audio lines are run in the same wire-way or area as a power line. Radio-frequency interference can also manifest itself as a buzz in the program audio. RF interference is minimized by the extensive RF bypassing and ground-plane techniques used in the STEREOMIXER, and the shielded lines external to the unit.

2.5 POWER CONNECTION

The STEREOMIXER is designed to operate from 117 VAC ±10%, 60 Hz, and requires a line fuse with a 1 ASB rating. Should the fuse ever need to be replaced, replace with the proper type only. The appropriate power cord (PR&E #30-13) should be installed between STEREOMIXER and the AC mains.

WARNING: Do not defeat the safety ground in any way. To do so may provide a potentially dangerous condition to the operator.

2.6 PATCH POINT CONNECTION

The STEREOMIXER System Master and Microphone Input Modules feature audio patch connectors. These connectors provide the connection point for external processing equipment such as equalizers, limiters, filters, etc., including those accessory modules which can be mounted in the STEREOMIXER mainframe.

Patch outputs are unbalanced, and designed to operate into low-impedance (600 ohm or higher) loads. The patch return is 40K ohm balanced and bridging.

The level at all patch points is -10 dBu nominal. This level was determined to provide optimum headroom within the console, as well as a good compatibility match with currently available processing equipment. See Section 2.7 for pin assignment.

2.7 AUDIO AND LOGIC CONNECTION

Good wiring practice calls for care in making each connection and in neatness of cable layout. Complete information relative to module input, output and logic terminations is contained in the connection reference drawings included in this section.

A standard connection theme is used throughout all PR&E console designs - the use of 3, 6, and 12 pin Molex connectors for audio wiring. This system of pin assignment takes advantage of the three pin per row design of the Molex connectors and, therefore, makes visual inspection of the finished wiring easier. As viewed from the rear of the console, the shields are always connected to the left pins, the low wires (black) to the center pins and the high wires (red) to the right pins. While this inspection will not indicate if a connector is in the correct position, it will verify proper shield and polarity connection.

Each STEREOMIXER input module offers a comprehensive set of logic functions, as outlined in Section 2.8. These functions are brought out from each of the module positions to the 15-pin Molex connectors labeled LOGIC, and are designed to drive interface relays and/or opto-isolators to control auxiliary equipment.

Fabrication of interface systems is straightforward. However, should the installer prefer to use prefabricated units, interface devices and cable assemblies are available from PR&E to control most professional grade tape decks, cartridge machines and turntables. Information on these interfaces is contained in Chapter 7.

NOTE: The logic functions are referenced to the console logic power supply, and, as such, should never be directly connected to auxiliary equipment control logic power supplies and/or grounds.

The STEREOMIXER uses "common" or "ground" switching for all of its control inputs, thus preventing the possibility of defective remote controls shorting out the logic power supply. Connection of the control circuitry requires an understanding of the logic nomenclature and symbols. These are outlined below:

Control Outputs (Commands):

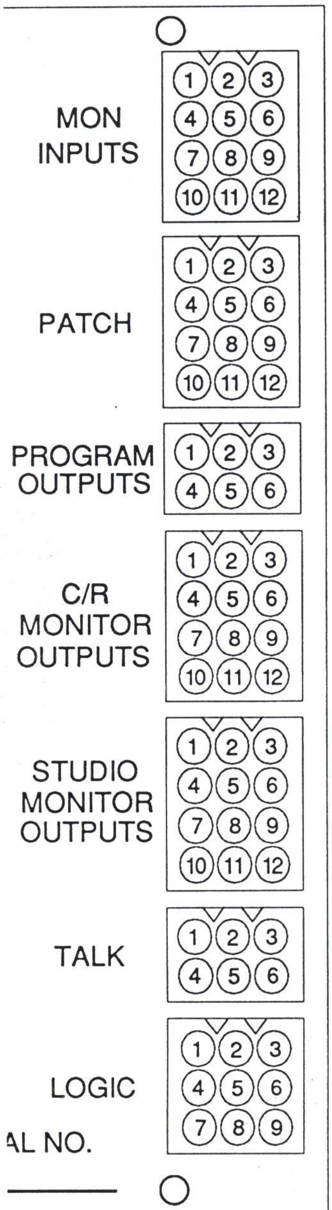
Tally (light)	Provides a +12 VDC continuous source when activated.
Pulse	Provides a +12 VDC pulse source when activated.
Sink	An open collector that provides a connection to Logic Common when activated.

Control Inputs:

Control	$\overline{\text{ON}}$	A line above the word indicates that the function is activated when a connection to Logic Common is made.

NOTE: When connecting Microphone Input, Stereo Line Input and Telephone Input/Mix-Minus Module logic, it may be desirable to have separate logic control for both "A" and "B" inputs. This can be accomplished by wiring the logic connector in a "Y" configuration using the "Y" cable defined in Section 6.23. See Section 2.8 for details.

2.7.1 System Master Module Connection



NOTE: These connectors are located on the rear panel of the Power Supply Assembly.

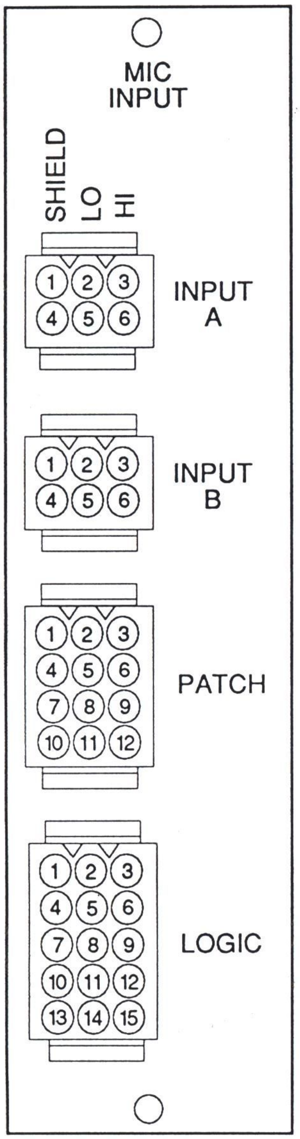
AUDIO WIRING				
		Pin Number		
Signal	Connector	Shield	Low	High
Monitor Input 1/Left	MON INPUTS	1	2	3
Monitor Input 1/Right	MON INPUTS	4	5	6
Monitor Input 2/Left	MON INPUTS	7	8	9
Monitor Input 2/Right	MON INPUTS	10	11	12
Pgm Patch Send/Left	PATCH	1	2	3
Pgm Patch Send/Right	PATCH	4	5	6
Pgm Patch Return/Left	PATCH	7	8	9
Pgm Patch Ret/Right	PATCH	10	11	12
Program Output/Left	PROGRAM OUT	1	2	3
Program Output/Right	PROGRAM OUT	4	5	6
C/R Monitor Out/Left	C/R MON OUT	1	2	3
C/R Monitor Out/Right	C/R MON OUT	4	5	6
Headphone Out/Left	C/R MON OUT	7	8	9
Headphone Out/Right	C/R MON OUT	10	11	12
Muting Output/Left	STU MON OUT	1	2	3
Muting Output/Right	STU MON OUT	4	5	6
Non-muting Out/Left	STU MON OUT	7	8	9
Non-muting Out/Right	STU MON OUT	10	11	12
Ext Talk to C/R Input	TALK	1	2	3
Talk to External Out	TALK	4	5	6

NOTE: The PATCH connector must be wired so that the Patch Sends are connected through to the Patch Returns. If no external processing or patch bay equipment is connected, a mating connector with jumpers from pins #2 to #8, #3 to #9, #5 to #11 and #6 to #12 must be installed.

LOGIC WIRING (Logic Connector)	
Function	Pin Number
Logic Common	1
Logic +12 VDC	2
EXT TALK TO CONTROL ROOM	3
DIM Control Room	4
MUTE Control Room	5
TALK TO EXTERNAL Command	6
TIMER RESET Command	7
CONTROL ROOM WARNING Tally	8
STUDIO WARNING Tally	9

NOTE: Consult Section 2.8.1 when connecting System Master Module remote controls.

2.7.2 Microphone Input Module Connection



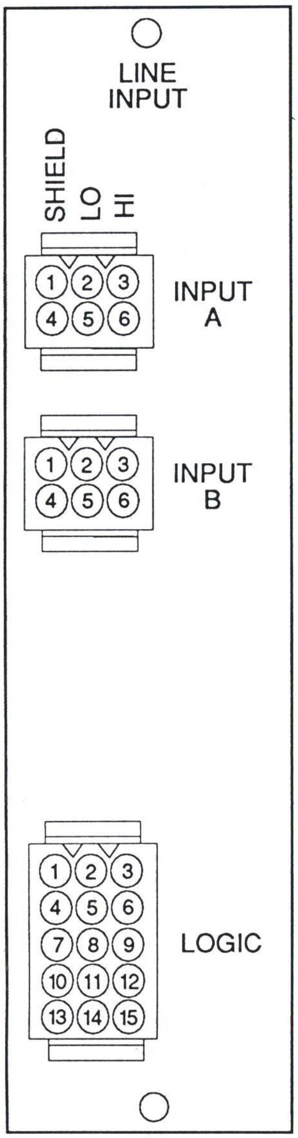
AUDIO WIRING				
		Pin Number		
Signal	**Connector**	**Shield**	**Low**	**High**
Microphone A	INPUT A	1	2	3
-No Connection-	INPUT A	4	5	6
Microphone B	INPUT B	1	2	3
-No Connection-	INPUT B	4	5	6
Patch Send	PATCH	1	2	3
-No Connection-	PATCH	4	5	6
Patch Return	PATCH	7	8	9
-No Connection-	PATCH	10	11	12

NOTE: The PATCH connector must be wired so that the Patch Send is connected through to the Patch Return. If no external processing or patch bay equipment is connected, a mating connector with jumpers from pins #2 to #8 and #3 to #9 must be installed.

LOGIC WIRING (Logic Connector)	
Function	**Pin Number**
Logic Common (Switched)	1
Logic +12 VDC	2
ON	3
OFF	4
ON Tally	5
OFF Tally	6
COUGH	7
TALK TO CONTROL ROOM	8
-No Connection-	9
-No Connection-	10
-No Connection-	11
-No Connection-	12
PFL/SOLO Inhibit (Privacy)	13
-No Connection-	14
"B" Input Select Command	15

NOTE: Consult Section 2.8.2 when connecting Microphone Input Module remote controls.

2.7.3 Stereo Line Input Module Connection

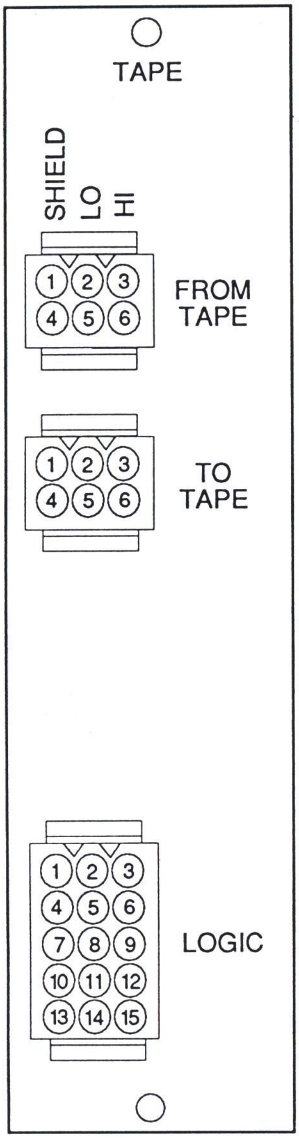


AUDIO WIRING				
		Pin Number		
Signal	**Connector**	**Shield**	**Low**	**High**
Input A/Left	INPUT A	1	2	3
Input A/Right	INPUT A	4	5	6
Input B/Left	INPUT B	1	2	3
Input B/Right	INPUT B	4	5	6

LOGIC WIRING (Logic Connector)	
Function	**Pin Number**
Logic Common (Switched)	1
Logic +12 VDC	2
$\overline{\text{ON}}$	3
$\overline{\text{OFF}}$	4
ON Tally	5
OFF Tally	6
$\overline{\text{READY}}$	7
$\overline{\text{AUDIO RESET TO OFF}}$	8
$\overline{\text{PFL}}$	9
START PULSE	10
STOP PULSE	11
PFL Tally	12
$\overline{\text{SOLO}}$	13
SOLO Tally	14
"B" Input Select Command	15

NOTE: Consult Section 2.8.3 when connecting Stereo Line Input Module remote controls.

2.7.4 Tape Recorder Input/Output Module Connection

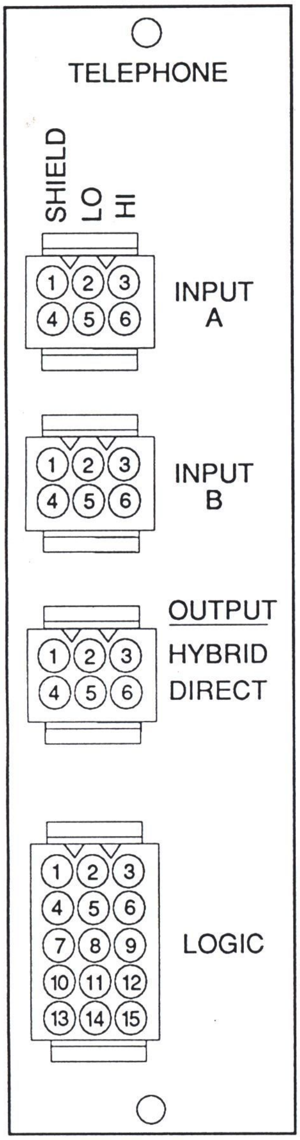


AUDIO WIRING				
		Pin Number		
Signal	Connector	Shield	Low	High
From Tape/Left	FROM TAPE	1	2	3
From Tape/Right	FROM TAPE	4	5	6
To Tape/Left	TO TAPE	1	2	3
To Tape/Right	TO TAPE	4	5	6

LOGIC WIRING (Logic Connector)	
Function	Pin Number
Logic Common	1
Logic +12 VDC	2
$\overline{\text{ON}}$	3
$\overline{\text{OFF}}$	4
ON Tally	5
OFF Tally	6
$\overline{\text{READY}}$	7
$\overline{\text{AUDIO RESET TO OFF}}$	8
$\overline{\text{PFL}}$	9
START PULSE	10
STOP PULSE	11
PFL Tally	12
$\overline{\text{SOLO}}$	13
SOLO Tally	14
-No Connection-	15

<u>NOTE</u>: Consult Section 2.8.4 when connecting Tape Recorder Input/Output Module remote controls.

2.7.5 Telephone Input/Mix-Minus Module Connection



AUDIO WIRING

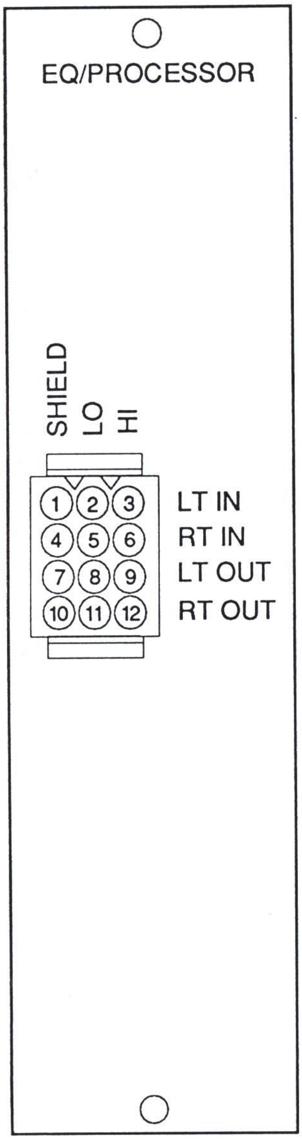
Signal	Connector	Pin Number: Shield	Pin Number: Low	Pin Number: High
Line A	INPUT A	1	2	3
-No Connection-	INPUT A	4	5	6
Line B	INPUT B	1	2	3
-No Connection-	INPUT B	4	5	6
Hybrid Output	OUTPUT HYB	1	2	3
Direct Output	OUTPUT DIR	4	5	6

LOGIC WIRING
(Logic Connector)

Function	Pin Number
Logic Common (Switched)	1
Logic +12 VDC	2
$\overline{\text{ON}}$	3
$\overline{\text{OFF}}$	4
ON Tally	5
OFF Tally	6
$\overline{\text{READY}}$	7
$\overline{\text{AUDIO RESET TO OFF}}$	8
-No Connection-	9
-No Connection-	10
-No Connection-	11
-No Connection-	12
-No Connection-	13
-No Connection-	14
"B" Input Select Command	15

NOTE: Consult Section 2.8.5 when connecting Telephone Input/Mix-Minus Module remote controls.

2.7.6 Accessory Module Connection

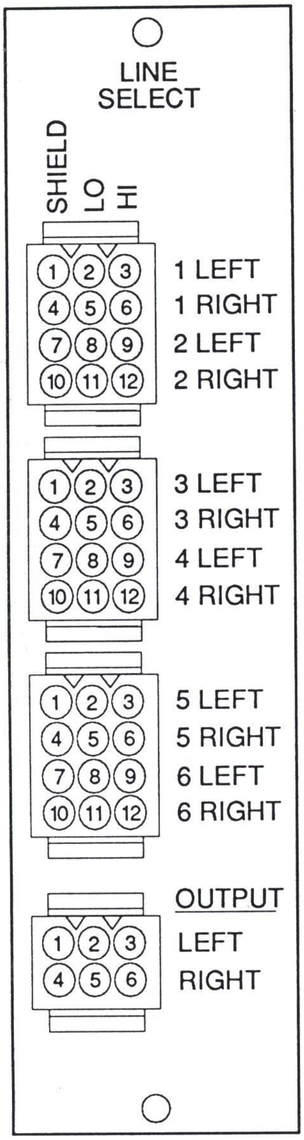


MONAURAL EQUALIZER AUDIO WIRING				
		Pin Number		
Signal	Connector	Shield	Low	High
Mono Equalizer Input	AUDIO	1	2	3
-No Connection-	AUDIO	4	5	6
Mono Equalizer Output	AUDIO	7	8	9
-No Connection-	AUDIO	10	11	12

STEREO EQUALIZER AUDIO WIRING				
		Pin Number		
Signal	Connector	Shield	Low	High
Stereo EQ Input/Left	AUDIO	1	2	3
Stereo EQ Input/Right	AUDIO	4	5	6
Stereo EQ Output/Left	AUDIO	7	8	9
Stereo EQ Output/Right	AUDIO	10	11	12

VOICE PROCESSOR AUDIO WIRING				
		Pin Number		
Signal	Connector	Shield	Low	High
Voice Processor Input	AUDIO	1	2	3
-No Connection-	AUDIO	4	5	6
Voice Processor Output	AUDIO	7	8	9
-No Connection-	AUDIO	10	11	12

2.7.7 Remote Line Selector Connection



AUDIO WIRING

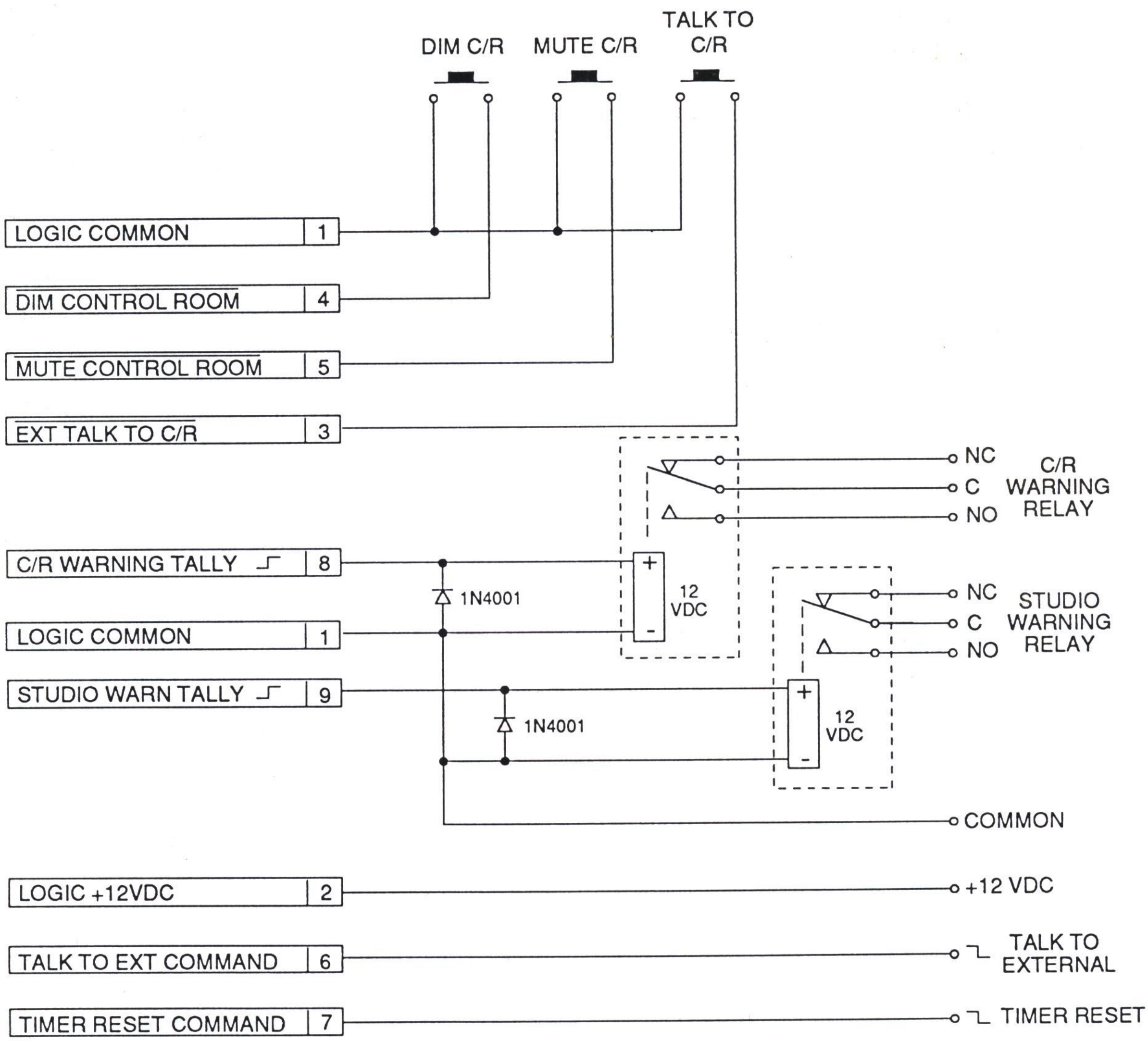
		Pin Number		
Signal	**Connector**	**Shield**	**Low**	**High**
Input 1/Left	1 LEFT	1	2	3
Input 1/Right	1 RIGHT	4	5	6
Input 2/Left	2 LEFT	7	8	9
Input 2/Right	2 RIGHT	10	11	12
Input 3/Left	3 LEFT	1	2	3
Input 3/Right	3 RIGHT	4	5	6
Input 4/Left	4 LEFT	7	8	9
Input 4/Right	4 RIGHT	10	11	12
Input 5/Left	5 LEFT	1	2	3
Input 5/Right	5 RIGHT	4	5	6
Input 6/Left	6 LEFT	7	8	9
Input 6/Right	6 RIGHT	10	11	12
Output/Left	OUTPUT LEFT	1	2	3
Output/Right	OUTPUT RIGHT	4	5	6

2.8 MODULE REMOTE CONTROL CAPABILITIES

This section outlines STEREOMIXER module remote controls. Included are schematics illustrating the System Master, Microphone Input, Stereo Line Input, Tape Recorder Input/Output, and Telephone Input/Mix-Minus Module remote control capabilities.

2.8.1 System Master Module Remote Control

The schematic below illustrates the full remote control capability of the System Master Module. A description of the System Master Module's two-way External Communication System is also included in this Section.



EXTERNAL COMMUNICATION SYSTEM

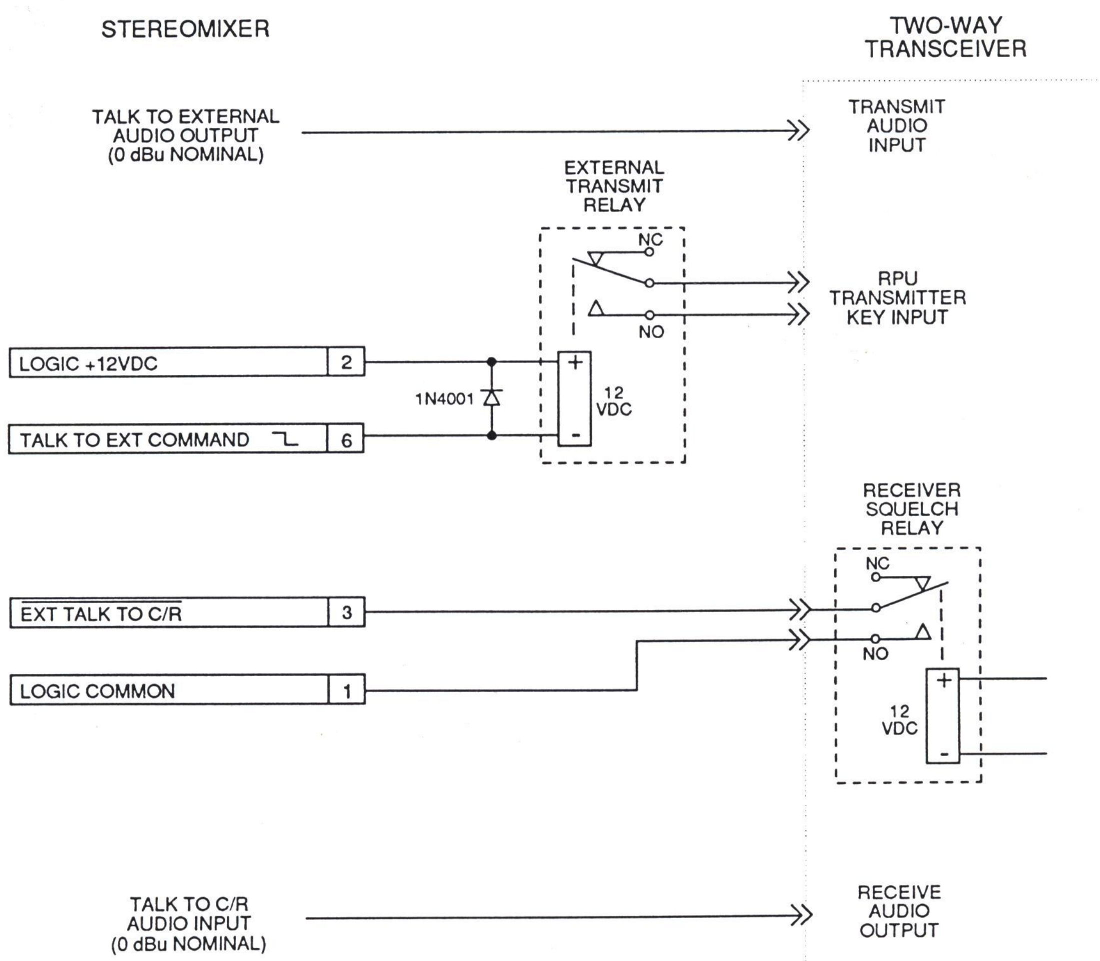
The Talk To External logic and audio outputs, coupled with the External Talk To Control Room logic and audio inputs, provide a convenient two-way communication system utilizing the System Master

Module electret microphone and the Control Room monitors. This facility is ideal for talking to and receiving a response from a news room, a call screener, or a producer's booth. Other functions include feeding an IFB system, with the Talk To External command keying the IFB, or communicating via a two-way transceiver, with the Talk To External command keying the transmitter.

The Talk To External command is an open collector to ground, which may be used for keying an external device or engaging a relay coil. This command is on as long as the System Master Module TALK TO EXTERNAL button is depressed. Depressing the TALK TO EXTERNAL button also feeds the preamplified output of the System Master Module electret microphone to the Talk To External audio output.

The External Talk To Control Room input is a switching input requiring closure to ground. When this input is pulled low by an external source, the Control Room monitors and headphones dim, and the Talk To Control Room audio input is turned on. Any audio appearing at this input, such as audio from a two-way transceiver, will be heard on the Control Room monitors and headphones.

The schematic below provides an example of a typical connection of this system to a two-way transceiver.

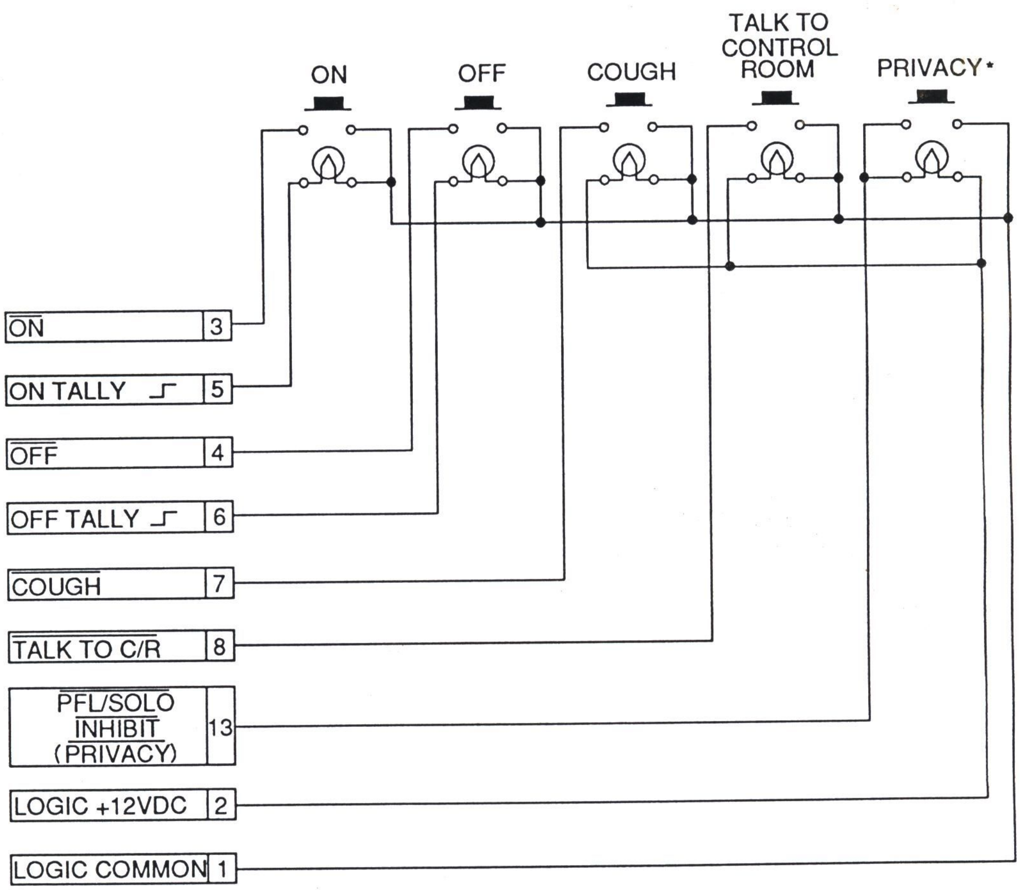


2.8.2 Microphone Input Module Remote Control

There are two methods of connecting Microphone Input Module remote controls - single input, which allows for the logic connection of one microphone (either "A" or "B"), or dual input, which allows for the simultaneous logic connection of two microphones (one "A" and one "B"). Both configurations are described below.

SINGLE INPUT CONFIGURATION

The single input configuration allows for the logic connection of a single microphone, either "A" or "B", as desired. Microphone logic can then be enabled/disabled by means of internal option switch S2. Setting S2-1 to the "ON" position enables input "A" logic, while setting S2-2 to the "ON" position enables input "B" logic, as described in Section 2.9.2. The schematic below illustrates the full remote control capability of the Microphone Input Module.



*Maintained action switch.

DUAL INPUT CONFIGURATION

In the dual input configuration, separate logic wiring for both "A" and "B" inputs is connected simultaneously, with the logic enabled/disabled by means of the module front panel "A/B" selector. This is accomplished by wiring the single logic connector in a "Y" configuration, using the A/B Logic "Y" Cable (PR&E #99-580). In this configuration, the single logic connector becomes two (one "A" and one "B"), with each fully capable of accommodating microphone logic connection per the above schematic.

Connect "A" input microphone logic to connector J1, and "B" input microphone logic to connector J2, as defined in Section 6.23.

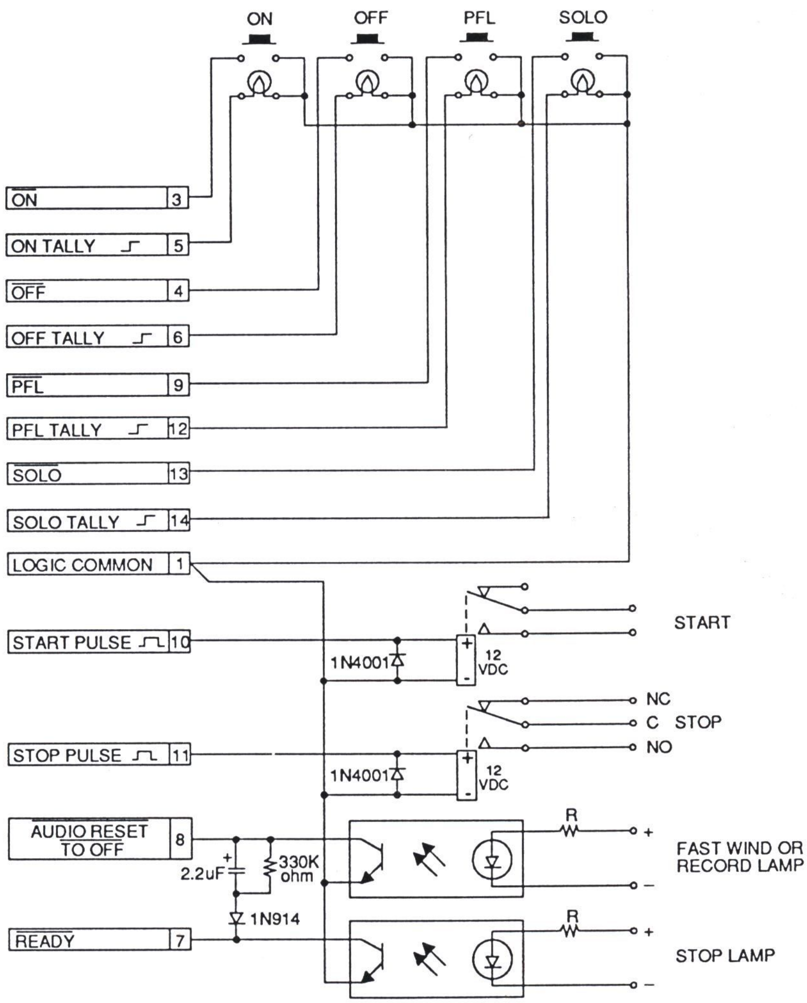
NOTE: In this configuration, internal option switch S2-1 must be set to the "ON" position and S2-2 must be set to the "OFF" position. This allows the front panel "A/B" selector to determine which input is logic enabled, "A" or "B".

2.8.3 Stereo Line Input Module Remote Control

There are two methods of connecting Stereo Line Input Module remote controls - single input, which allows for the logic connection of one source (either "A" or "B"), or dual input, which allows for the simultaneous logic connection of two sources (one "A" and one "B"). Both configurations are described below.

SINGLE INPUT CONFIGURATION

The single input configuration allows for the logic connection of a single source, either "A" or "B", as desired. Source logic can then be enabled/disabled by means of internal option switch S5. Setting S5-3 to the "ON" position enables input "A" logic, while setting S5-4 to the "ON" position enables input "B" logic, as described in Section 2.9.3. The schematic below illustrates the full remote control capability of the Stereo Line Input Module.



DUAL INPUT CONFIGURATION

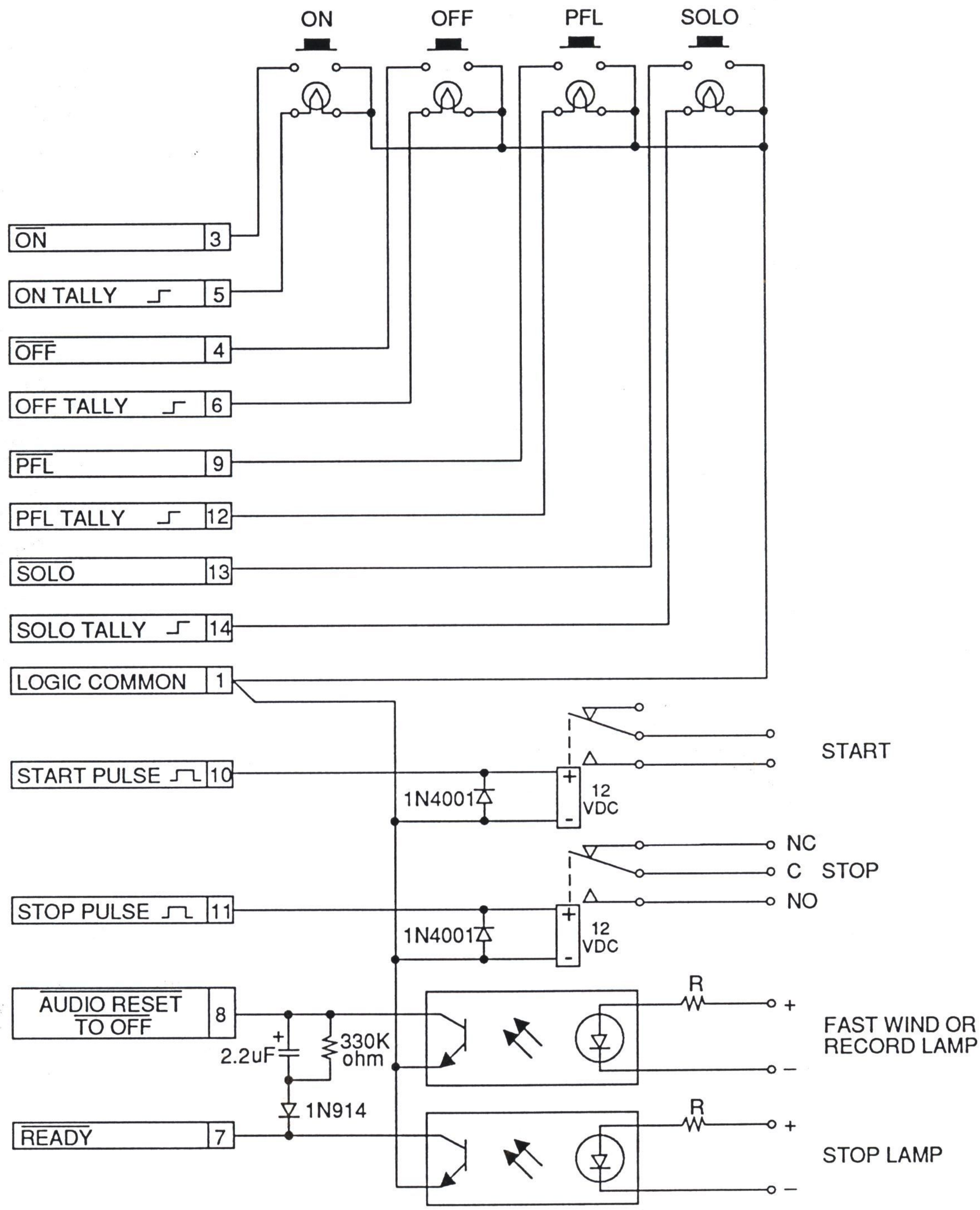
In the dual input configuration, separate logic wiring for both "A" and "B" inputs is connected simultaneously, with the logic enabled/disabled by means of the module front panel "A/B" selector. This is accomplished by wiring the single logic connector in a "Y" configuration, using the A/B Logic "Y" Cable (PR&E #99-580). In this configuration, the single logic connector becomes two (one "A" and one "B"), with each fully capable of accommodating source logic connection per the above schematic.

Connect "A" input source logic to connector J1, and "B" input source logic to connector J2, as defined in Section 6.23.

NOTE: In this configuration, internal option switch S5-3 must be set to the "ON" position and S5-4 must be set to the "OFF" position. This allows the front panel "A/B" selector to determine which input is logic enabled, "A" or "B".

2.8.4 Tape Recorder Input/Output Module Remote Control

The Tape Recorder Input/Output Module accommodates a single input. The schematic below illustrates the full remote control capability of this module.

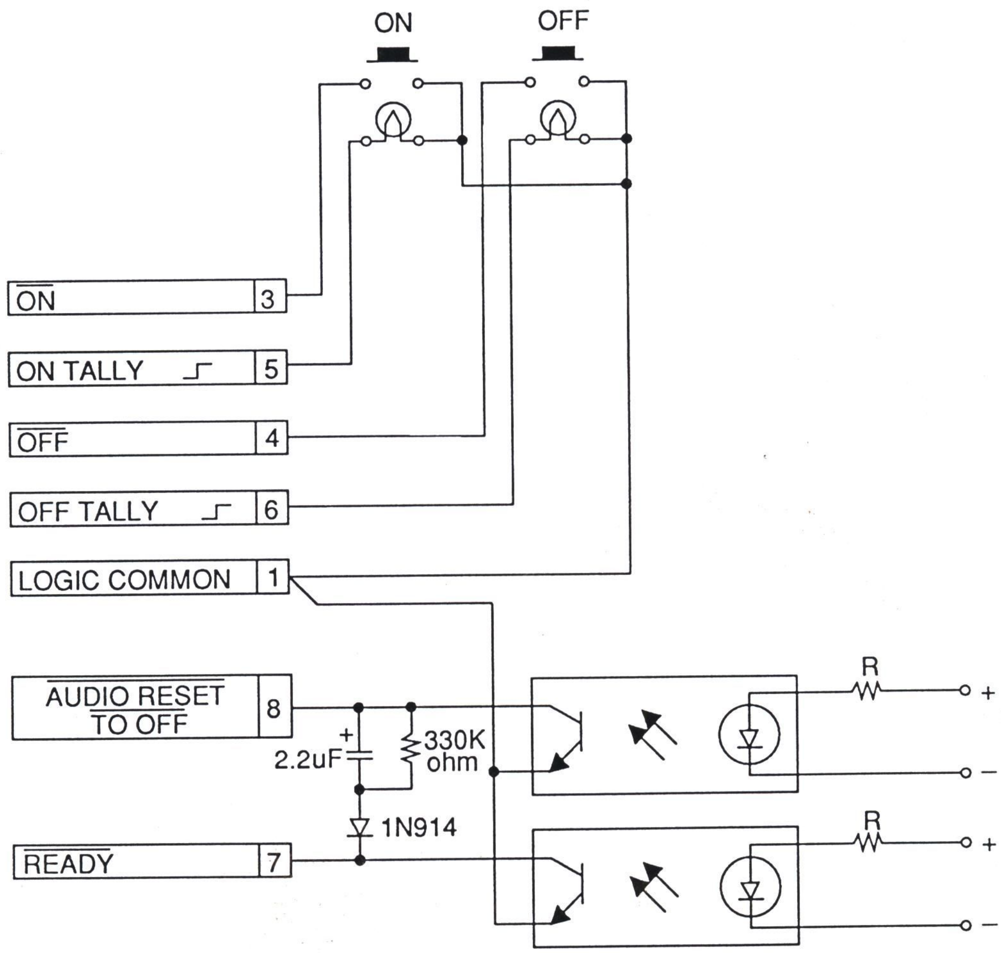


2.8.5 Telephone Input/Mix-Minus Module Remote Control

There are two methods of connecting Telephone Input/Mix-Minus Module remote controls - single input, which allows for the logic connection of one telco source (either "A" or "B"), or dual input, which allows for the simultaneous logic connection of two telco sources (one "A" and one "B"). Both configurations are described below.

SINGLE INPUT CONFIGURATION

The single input configuration allows for the logic connection of a single telco source, either "A" or "B", as desired. Telco source logic can then be enabled/disabled by means of internal option switch S5. Setting S5-3 to the "ON" position enables input "A" logic, while setting S5-4 to the "ON" position enables input "B" logic, as described in Section 2.9.5. The schematic below illustrates the full remote control capability of the Telephone Input/Mix-Minus Module.



DUAL INPUT CONFIGURATION

In the dual input configuration, separate logic wiring for both "A" and "B" inputs is connected simultaneously, with the logic enabled/disabled by means of the module front panel "A/B" selector. This is accomplished by wiring the single logic connector in a "Y" configuration, using the A/B Logic "Y" Cable (PR&E #99-580). In this configuration, the single logic connector becomes two (one "A" and one "B"), with each fully capable of accommodating telco source logic connection per the above schematic.

Connect "A" input telco source logic to connector J1, and "B" input telco source logic to connector J2, as defined in Section 6.23.

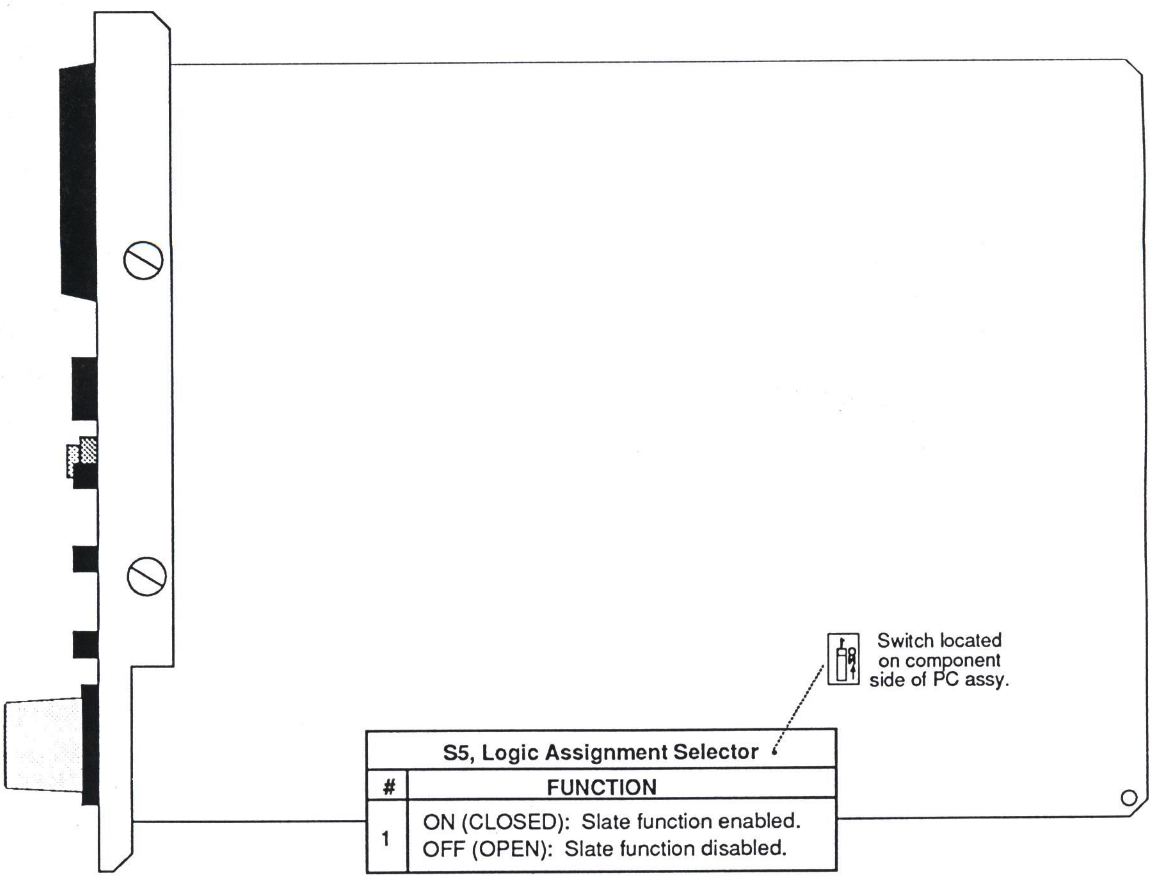
<u>NOTE</u>: In this configuration, internal option switch S5-3 must be set to the "ON" position and S5-4 must be set to the "OFF" position. This allows the front panel "A/B" selector to determine which input is logic enabled, "A" or "B".

2.9 MODULE INTERNAL OPTION SWITCHES

All STEREOMIXER modules except the Voice Processor and Remote Line Selector Modules are equipped with internal option switches, which can be set to enable or disable selected module functions. This section provides the location of these switches on the module printed circuit assemblies, and their function(s). Included are descriptions of the System Master, Microphone Input, Stereo Line Input, Tape Recorder Input/Output, Telephone Input/Mix-Minus, and Monaural and Stereo Equalizer Module option switches.

2.9.1 System Master Module Option Switch

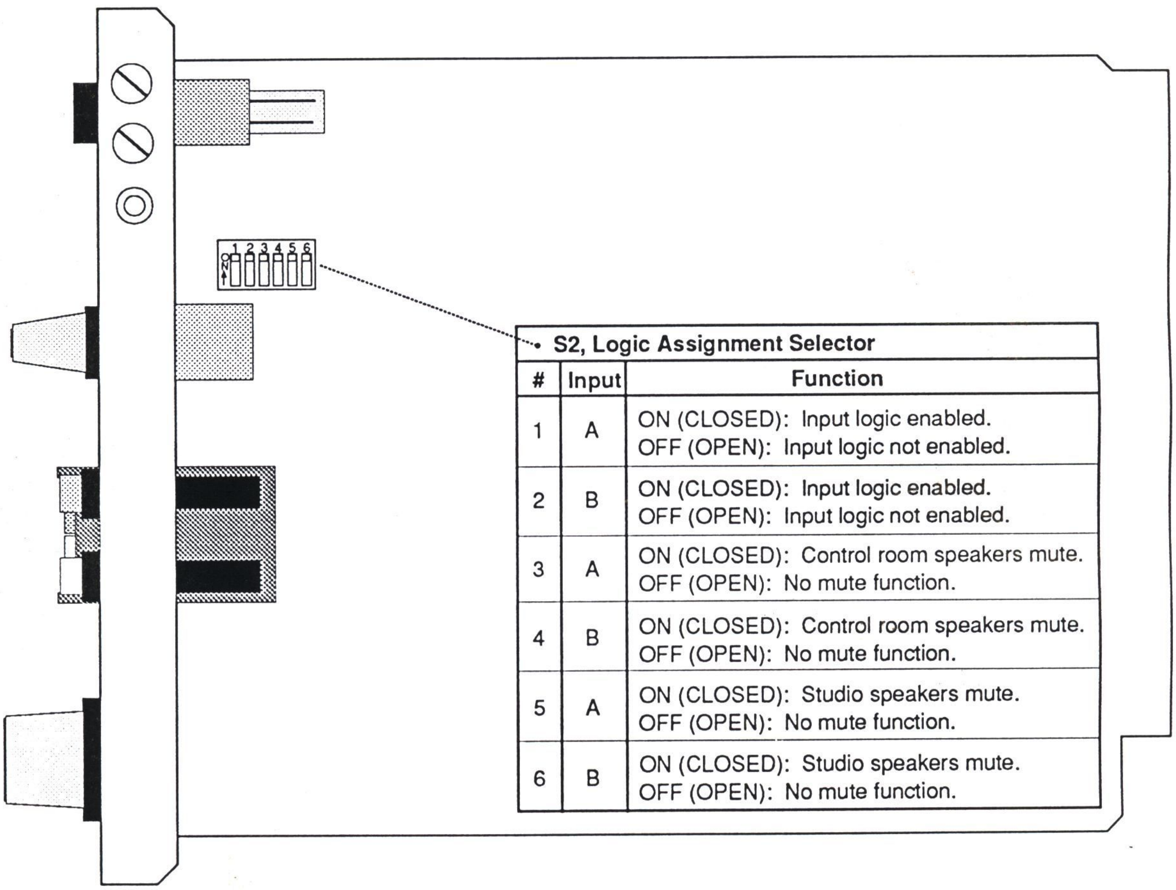
The slate function may be enabled/disabled by means of a one station DIP switch (S5) located on the circuit side of the System Master Module Satellite Board (PR&E #95-444).



S5, Logic Assignment Selector	
#	FUNCTION
1	ON (CLOSED): Slate function enabled. OFF (OPEN): Slate function disabled.

2.9.2 Microphone Input Module Option Switch

Input logic and monitor mute status is selectable for each input by means of a six station Dip switch (S2) located on the module PC board.



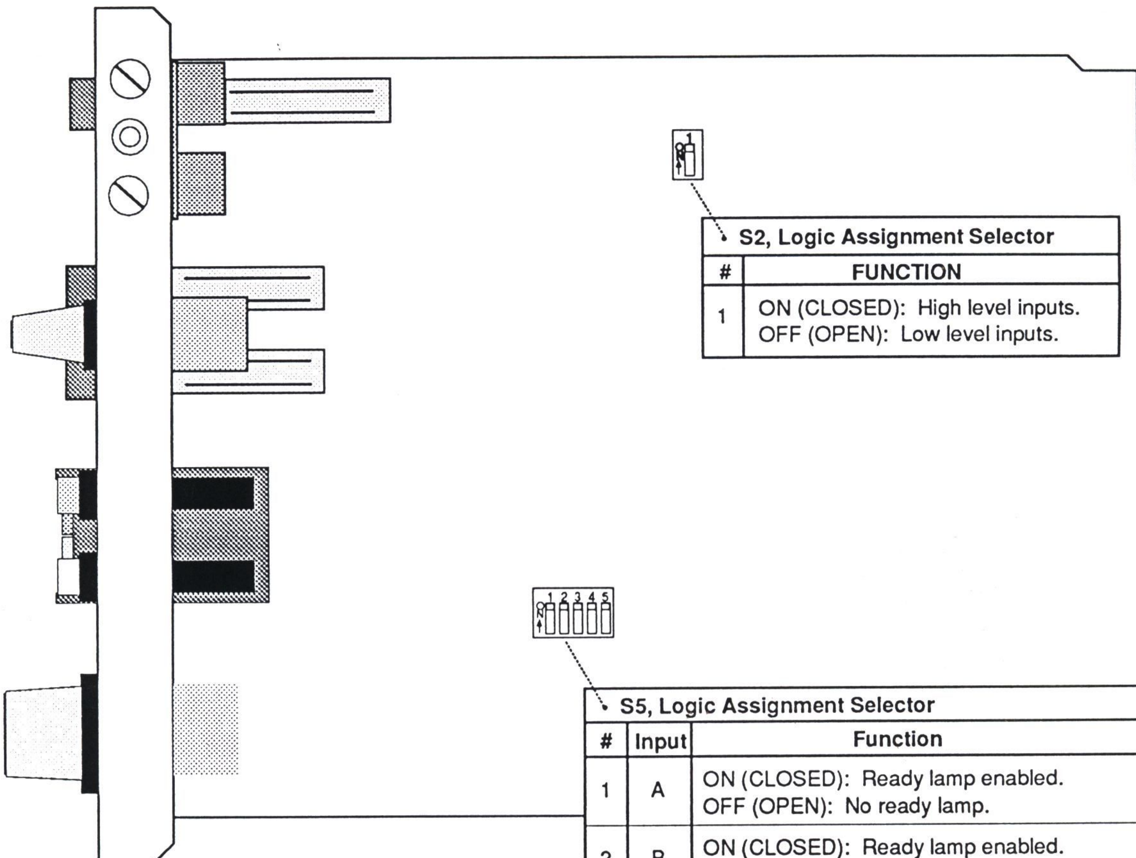
S2, Logic Assignment Selector

#	Input	Function
1	A	ON (CLOSED): Input logic enabled. OFF (OPEN): Input logic not enabled.
2	B	ON (CLOSED): Input logic enabled. OFF (OPEN): Input logic not enabled.
3	A	ON (CLOSED): Control room speakers mute. OFF (OPEN): No mute function.
4	B	ON (CLOSED): Control room speakers mute. OFF (OPEN): No mute function.
5	A	ON (CLOSED): Studio speakers mute. OFF (OPEN): No mute function.
6	B	ON (CLOSED): Studio speakers mute. OFF (OPEN): No mute function.

<u>NOTE</u>: When both "A" and "B" logic are connected (reference Section 2.8.2), switch S2-1 should be in the "ON" position and switch S2-2 should be in the "OFF" position.

2.9.3 Stereo Line Input Module Option Switches

Input logic, ready lamp and timer reset status is selectable for each input by means of a five station DIP switch (S5) located on the module PC board. This module is also equipped with an internal switchable attenuator (S1), allowing the preamp to accommodate input levels in the "high" (-10 dBu to +9 dBu; "ON") or "low" (-30 dBu to -9 dBu; "OFF") ranges.



S2, Logic Assignment Selector

#	FUNCTION
1	ON (CLOSED): High level inputs. OFF (OPEN): Low level inputs.

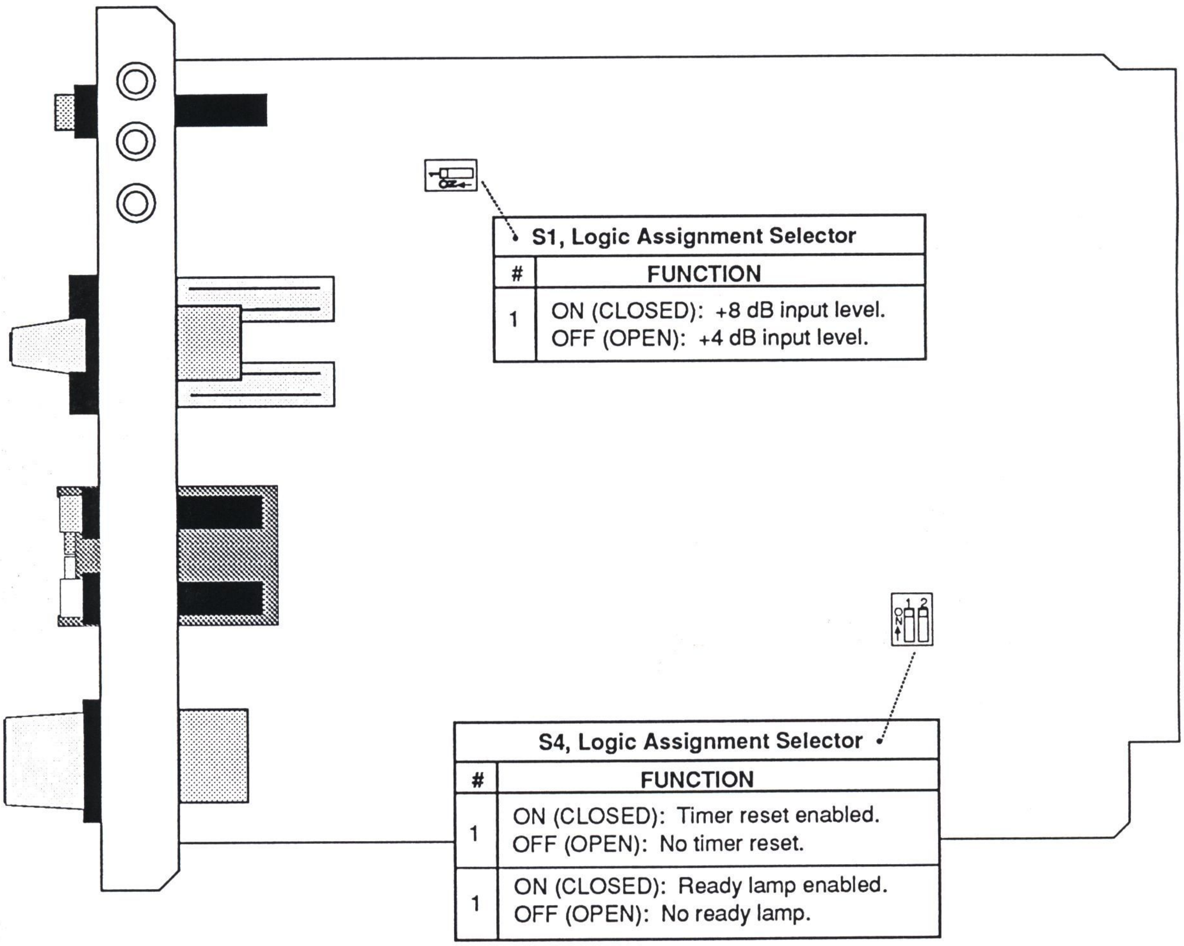
S5, Logic Assignment Selector

#	Input	Function
1	A	ON (CLOSED): Ready lamp enabled. OFF (OPEN): No ready lamp.
2	B	ON (CLOSED): Ready lamp enabled. OFF (OPEN): No ready lamp.
3	A	ON (CLOSED): Input logic enabled. OFF (OPEN): Input logic not enabled.
4	B	ON (CLOSED): Input logic enabled. OFF (OPEN): Input logic not enabled.
5	A/B	ON (CLOSED): Timer reset enabled. OFF (OPEN): No timer reset.

<u>NOTE</u>: When both "A" and "B" logic are connected (reference Section 2.8.3), switch S5-3 should be in the "ON" position and switch S5-4 should be in the "OFF" position.

2.9.4 Tape Recorder Input/Output Module Option Switches

Timer reset and ready lamp status is selectable by means of a two station DIP switch (S4) located on the module PC board. This module will also accommodate input levels of +8 dBu or +4 dBu, as selected by option switch S1.



S1, Logic Assignment Selector

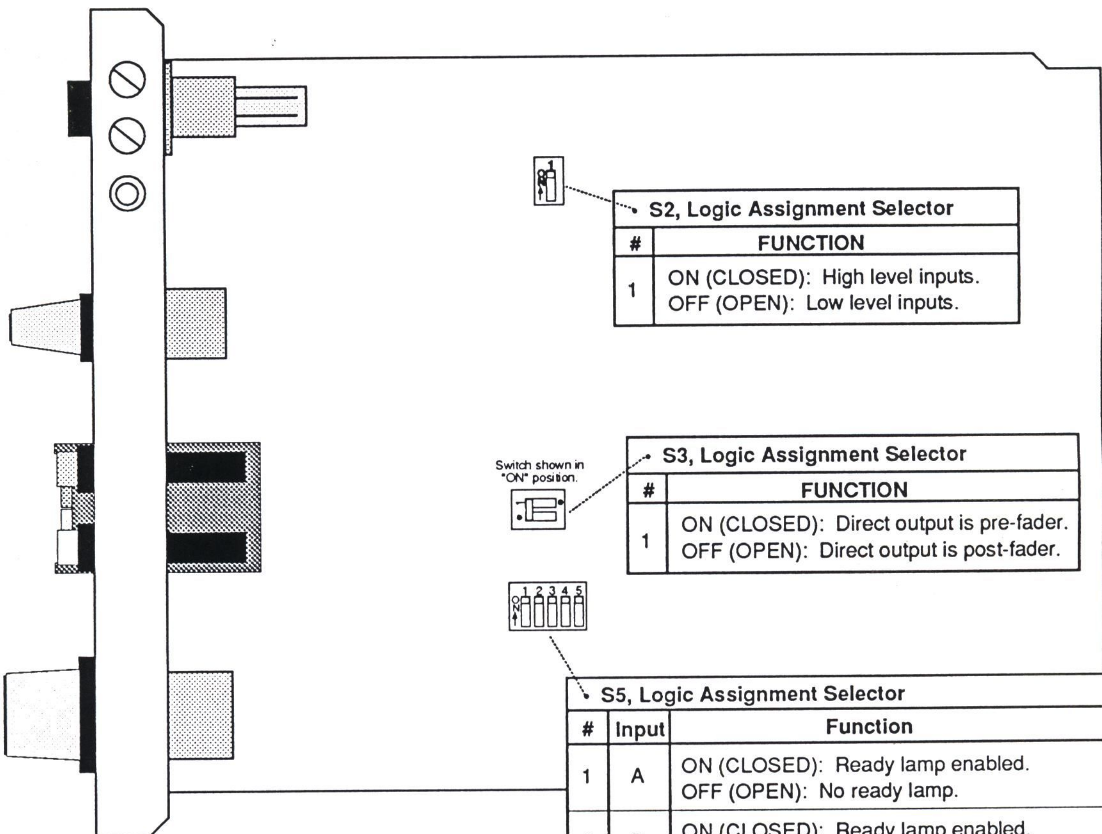
#	FUNCTION
1	ON (CLOSED): +8 dB input level. OFF (OPEN): +4 dB input level.

S4, Logic Assignment Selector

#	FUNCTION
1	ON (CLOSED): Timer reset enabled. OFF (OPEN): No timer reset.
1	ON (CLOSED): Ready lamp enabled. OFF (OPEN): No ready lamp.

2.9.5 Telephone Input/Mix-Minus Module Option Switches

Input logic, ready lamp and direct output status is selectable for each input by means of a five station DIP switch (S1) located on the module PC board. This module is also equipped with an internal switchable attenuator (S2), allowing the preamp to accommodate input levels in the "high" (-10 dBu to +9 dBu; ON) or "low" (-30 dBu to -9 dBu; OFF) ranges. Option switch S3 determines if the direct output signal is pre-fader or post-fader.



S2, Logic Assignment Selector

#	FUNCTION
1	ON (CLOSED): High level inputs. OFF (OPEN): Low level inputs.

S3, Logic Assignment Selector

#	FUNCTION
1	ON (CLOSED): Direct output is pre-fader. OFF (OPEN): Direct output is post-fader.

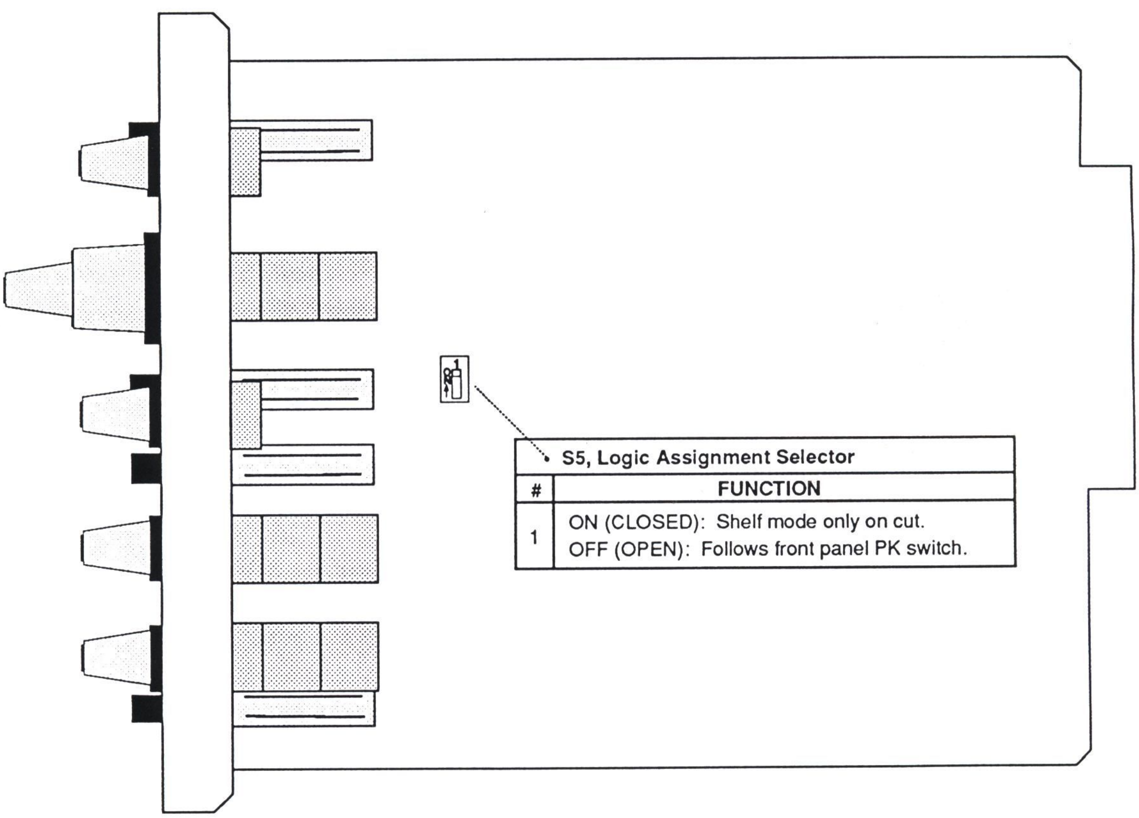
S5, Logic Assignment Selector

#	Input	Function
1	A	ON (CLOSED): Ready lamp enabled. OFF (OPEN): No ready lamp.
2	B	ON (CLOSED): Ready lamp enabled. OFF (OPEN): No ready lamp.
3	A	ON (CLOSED): Input logic enabled. OFF (OPEN): Input logic not enabled.
4	B	ON (CLOSED): Input logic enabled. OFF (OPEN): Input logic not enabled.
5	A/B	ON (CLOSED): Direct output follows input. OFF (OPEN): Direct output follows "ON".

<u>NOTE</u>: When both "A" and "B" logic are connected (reference Section 2.8.5), switch S5-3 should be in the "ON" position and switch S5-4 should be in the "OFF" position.

2.9.6 Monaural Equalizer Module Option Switch

The Monaural Equalizer Module is equipped with an internal option switch which allows the bass response to always be shelf mode on cut, regardless of the setting of the front panel peak/shelf switch. With this option switch (S5) set in the ON position, the front panel peak/shelf switch is only active in the boost mode. The purpose of this function is to recognize the realities of low frequency equalization of speech, music, or mixed program material. While it is very desirable to have both peak and shelf functions available in the boost mode, it is very rare to use the peak function in the cut mode. The option switch eliminates the need to be constantly changing the peak/shelf switch when shifting between boost and cut modes.

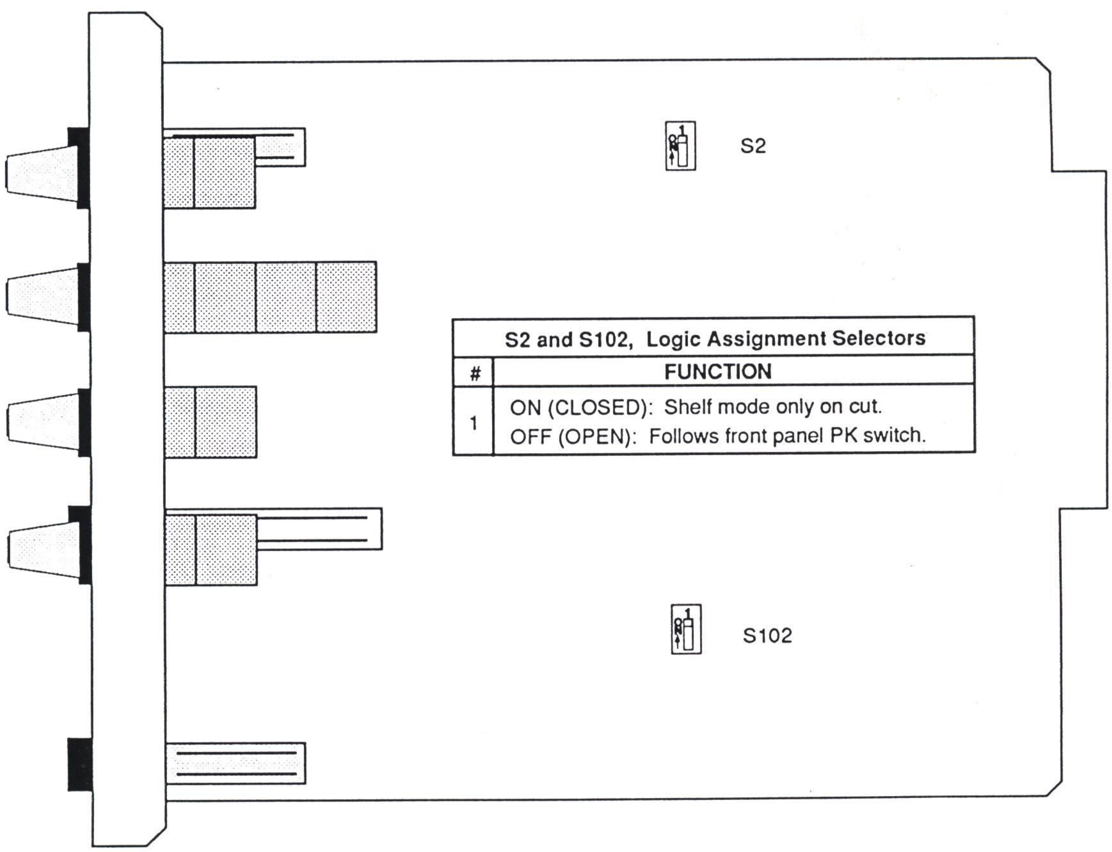


S5, Logic Assignment Selector	
#	FUNCTION
1	ON (CLOSED): Shelf mode only on cut. OFF (OPEN): Follows front panel PK switch.

2.9.7 Stereo Equalizer Module Option Switches

The Stereo Equalizer Module is equipped with internal option switches which allow the bass response to always be shelf mode on cut, regardless of the setting of the front panel peak/shelf switch. Unlike the Monaural Equalizer Module, this module requires that **two** internal switches, left (S2) and right (S102), be set to invoke this option. With these option switches set in the ON position, the front panel peak/shelf switch controls only the boost mode. The purpose of this function is to recognize the realities of low frequency equalization of speech, music, or mixed program material. While it is very desirable to have both peak and shelf functions available in the boost mode, it is very rare to use the peak function in the cut mode. The option switch eliminates the need to be constantly changing the peak/shelf switch when shifting between boost and cut modes.

<u>NOTE</u>: Ensure that both the left (S2) and the right (S102) option switches are set identically, to prevent differences in low frequency amplitude and phase response between the two channels.



S2 and S102, Logic Assignment Selectors	
#	FUNCTION
1	ON (CLOSED): Shelf mode only on cut. OFF (OPEN): Follows front panel PK switch.

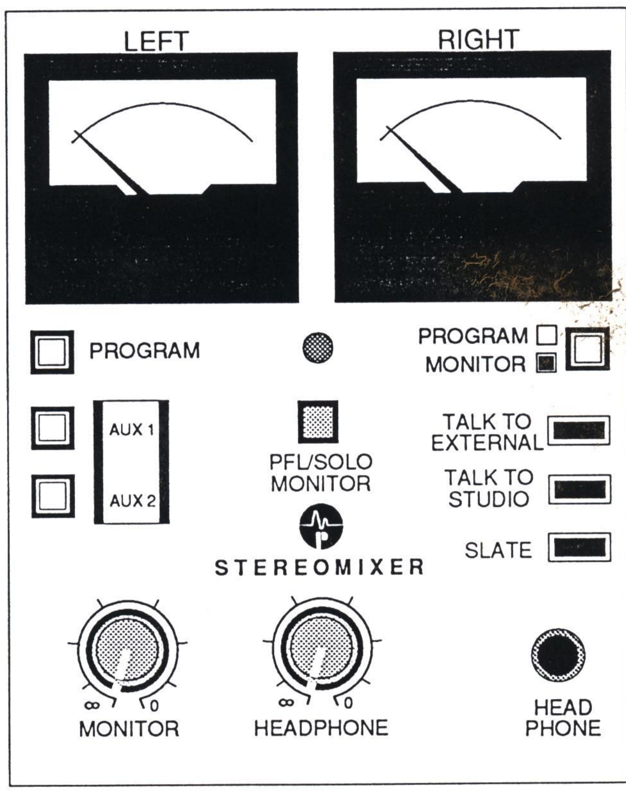
OPERATION

3.0 OPERATION

This chapter describes the operation of the System Master Module, and the four input and four accessory modules which can be inserted into the STEREOMIXER mainframe. Included are sections describing the Microphone Input Module, Stereo Line Input Module, Stereo Tape Recorder Input/Output Module, Telephone Input/Mix-Minus Module, Monaural Equalizer Module, Stereo Equalizer Module, Voice Processor Module, and Remote Line Selector. For a detailed description of module components, and their function, see Chapter 4 of this document.

3.1 SYSTEM MASTER MODULE

This section contains a figure illustrating the System Master Module's front panel controls, along with descriptions of the function(s) of each.



- The LEFT and RIGHT VU meters display the output levels of the stereo PROGRAM buses, or the stereo Monitor source.

- The built-in electret microphone (located in the center of the module, just below the VU meters) allows the operator to accomplish slate/talkback functions.

- The three-station Monitor switch (located just below the LEFT VU meter) selects the monitor source from PROGRAM and two auxiliary inputs, AUX 1 and AUX 2.

- The PFL/SOLO MONITOR indicator (located just below the electret microphone) illuminates whenever the signal from an input module is a PFL or SOLO source. When this is the case, the PFL or SOLO source temporarily overrides the monitor selection, thereby providing for input verification, level setting, and signal problem identification without disturbing the PROGRAM outputs.

- The alternate action METER button (located just below the RIGHT VU meter) selects the meter display mode to be PROGRAM, or to follow the MONITOR selection. In the "meter follows MONITOR" mode, the VU meters always display the level of the source being monitored, including PFL and SOLO. Metering SOLO provides the ability to preset an input level prior to its assignment to the PROGRAM bus.

- The TALK TO EXTERNAL button allows the console operator to talk to an external location through the built-in electret microphone.

- The TALK TO STUDIO button allows the console operator to talk to the Studio through the built-in electret microphone.

- The SLATE button sends a mix of the internal slate tone oscillator and the electret microphone to the PROGRAM bus. This function can be disabled by means of an internal option switch (reference Section 2.9.1).

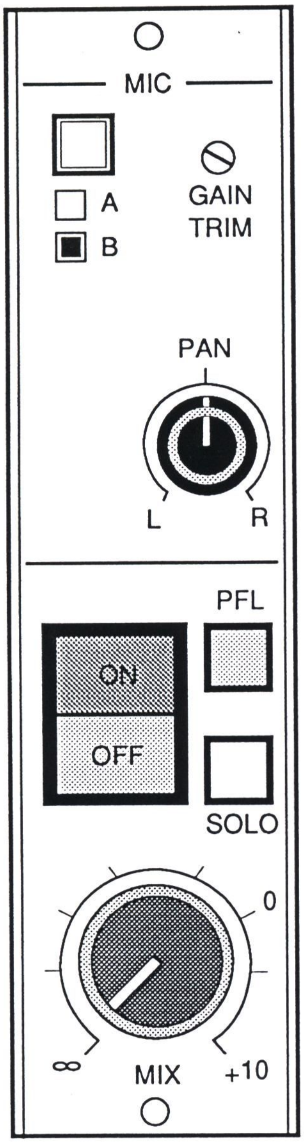
- The MONITOR level control adjusts the volume of the Control Room speakers.

- The HEADPHONE level control adjusts the volume of the console operator's headphones.

- The HEADPHONE jack provides convenient access for the console operator's headphones. The headphone output is also available on the rear connector panel (reference Section 2.7.1) for connection to a headphone panel built into the system furniture.

3.2 MICROPHONE INPUT MODULE

This section contains a figure illustrating the Microphone Input Module's front panel controls, along with descriptions of the function(s) of each.



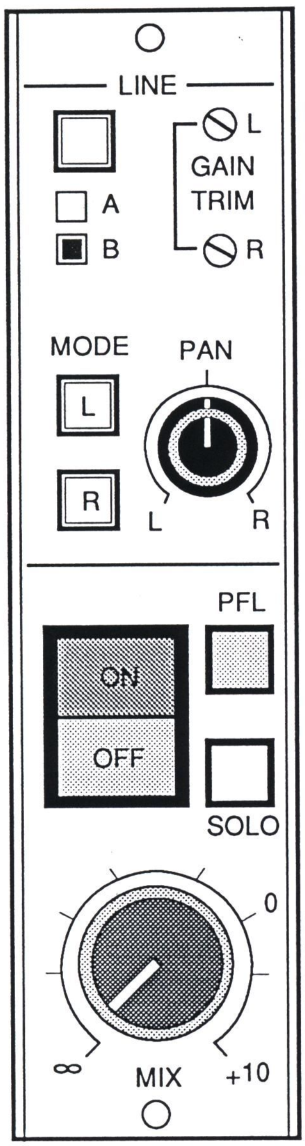
- The alternate action MIC button selects between microphones connected to the A and B inputs of the module.
- GAIN TRIM controls the microphone preamplifier gain for both of the two inputs.
- The PAN control positions the input signal in the left-right stereo image.
- The electronic alternate action ON/OFF button turns the module on and off and initiates the appropriate muting commands, as programmed during installation.
- The PFL button sends the pre-fader input signal into the mixer's automatic monitoring system, while the SOLO button sends the post-fader input signal. The PFL and SOLO functions temporarily override the normal monitor selection without disturbing the PROGRAM outputs.

  NOTE: When the PFL or SOLO button is pressed, any other input module(s) in the PFL or SOLO mode will automatically reset. To place more than one input module at a time in the PFL or SOLO mode (overriding the RESET command), hold down one PFL or SOLO button while pressing the PFL or SOLO button(s) on the additional module(s).

- The rotary fader is a stepless, infinite resolution control with a reference line at the 0 dB point. The reference is the nominal position for a properly adjusted input level to achieve "0" VU on the console meters.

3.3 STEREO LINE INPUT MODULE

This section contains a figure illustrating the Line Input Module's front panel controls, along with descriptions of the function(s) of each.



- The alternate action LINE button selects between the stereo sources connected to the A and B inputs of the module.

- The recessed L and R GAIN TRIM controls adjust the sensitivity of the channel's stereo input preamplifiers.

- The alternate action MODE buttons select the input signal mode of the module. With both buttons in the "out" position, the mode is stereo; pressing the L button alone sends the left source to both stereo buses; pressing the R button alone sends the right source to both stereo buses. Pressing both buttons sums the left and right input signals and sends the monaural result to both buses.

- The PAN control may be used to adjust the left-right balance of a stereo signal, and to position a monaural signal in the left-right stereo image.

- The PFL button sends the stereo pre-fader input signal into the STEREOMIXER's automatic monitoring system, while the SOLO button sends the post-fader input signal. The PFL and SOLO functions temporarily override the normal monitor selection without disturbing the PROGRAM outputs.

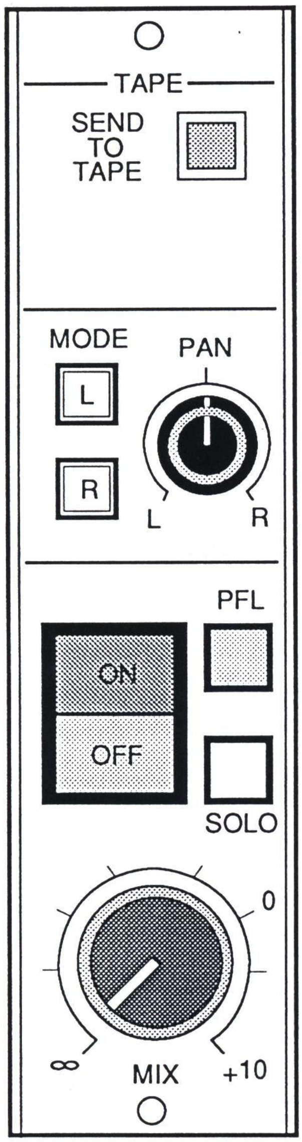
  NOTE: When the PFL or SOLO button is pressed, any other input module(s) in the PFL or SOLO mode will automatically reset. To place more than one input module at a time in the PFL or SOLO mode (overriding the RESET command), hold down one PFL or SOLO button while pressing the PFL or SOLO button(s) on the additional module(s).

- The electronic alternate action ON/OFF button turns the module on and off, and may optionally provide commands for the remote control of connected source equipment.

- The rotary fader is a stepless, infinite resolution control with a reference line at the 0 dB point. The reference is the nominal position for a properly adjusted input level to achieve "0" VU on the console meters.

3.4 TAPE RECORDER INPUT/OUTPUT MODULE

This section contains a figure illustrating the Stereo Tape Recorder Input/Output Module's front panel controls, along with descriptions of the function(s) of each.



- The electronic alternate action SEND TO TAPE button sends the stereo PROGRAM to the input of the tape recorder connected to this module. When this button is engaged, the module will automatically turn off to prevent the possibility of "round-robin" feedback through the tape recorder electronics.

- The alternate action MODE buttons select the input signal mode of the module. With both buttons in the "out" position, the mode is stereo; pressing the L button alone sends the left source to both stereo buses; pressing the R button alone sends the right source to both stereo buses. Pressing both buttons sums the left and right input signals and sends the monaural result to both buses.

- The PAN control may be used to adjust the left-right balance of a stereo signal, and to position a monaural signal in the left-right stereo image.

- The PFL button sends the stereo pre-fader input signal into the mixer's automatic monitoring system, while the SOLO button sends the post-fader input signal. The PFL and SOLO functions temporarily override the normal monitor selection without disturbing the PROGRAM outputs.

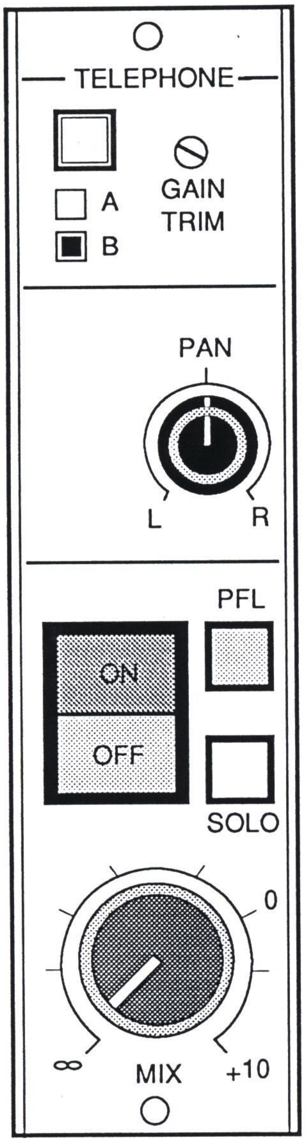
  NOTE: When the PFL or SOLO button is pressed, any other input module(s) in the PFL or SOLO mode will automatically reset. To place more than one input module at a time in the PFL or SOLO mode (overriding the RESET command), hold down one PFL or SOLO button while pressing the PFL or SOLO button(s) on the additional module(s).

- The electronic alternate action ON/OFF button turns the module on and off, and may optionally provide start and stop commands for the remote control of the connected tape recorder.

- The rotary fader is a stepless, infinite resolution control with a reference line at the 0 dB point. The reference is the nominal position for a properly adjusted input level to achieve "0" VU on the console meters.

3.5 TELEPHONE INPUT/MIX-MINUS MODULE

This section contains a figure illustrating the Telephone Input/Mix-Minus Module's front panel controls, along with descriptions of the function(s) of each.



- The alternate action TELEPHONE button selects between the monaural sources connected to the A and B inputs of the module.

- The GAIN TRIM control adjusts the sensitivity of the channel's input preamplifier.

- The PAN control may be used to position the signal in the left-right stereo image.

- The PFL button sends the pre-fader input signal into the mixer's automatic monitoring system, while the SOLO button sends the post-fader input signal. The PFL and SOLO functions temporarily override the normal monitor selection without disturbing the PROGRAM outputs.

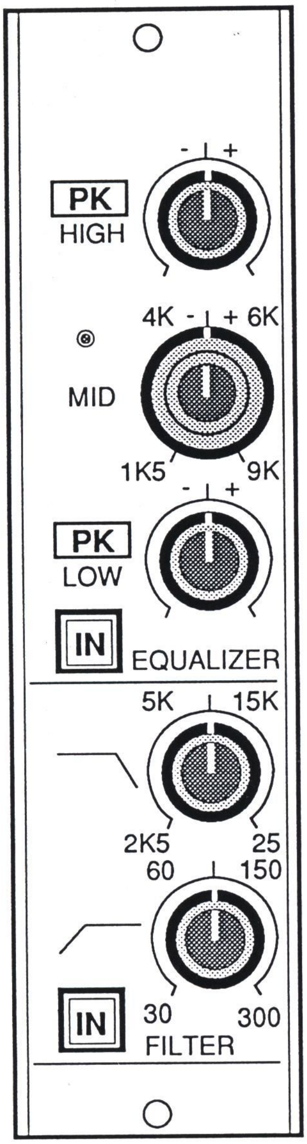
  NOTE: When the PFL or SOLO button is pressed, any other input module(s) in the PFL or SOLO mode will automatically reset. To place more than one input module at a time in the PFL or SOLO mode (overriding the RESET command), hold down one PFL or SOLO button while pressing the PFL or SOLO button(s) on the additional module(s).

- The electronic alternate action ON/OFF button turns the module on and off, and may optionally provide start and stop commands for the remote control of the connected tape recorder.

- The rotary fader is a stepless, infinite resolution control with a reference line at the 0 dB point. The reference is the nominal position for a properly adjusted input level to achieve "0" VU on the console meters.

3.6 MONAURAL EQUALIZER MODULE

This section contains a figure illustrating the Monaural Equalizer Module's front panel controls, along with descriptions of the function(s) of each.



- The HIGH control adjusts the amount of boost and cut of the high frequencies. The center frequency of operation is 9 kHz. The PK button switch changes the shape of the equalization curves from shelving to peaking.

- The outer knob of the concentric MID control tunes the operating frequency of this stage, while the inner knob adjusts the amount of boost and cut. The frequency tuning range is 1500 Hz to 9 kHz.

- The LOW control adjusts the amount of boost and cut of the low frequencies. The center frequency of operation is 80 Hz. The PK button switch changes the shape of the equalization curves from shelving to peaking. An internal option switch can set the cut mode as shelving in recognition of real-life equalizer applications (reference Section 2.9.6).

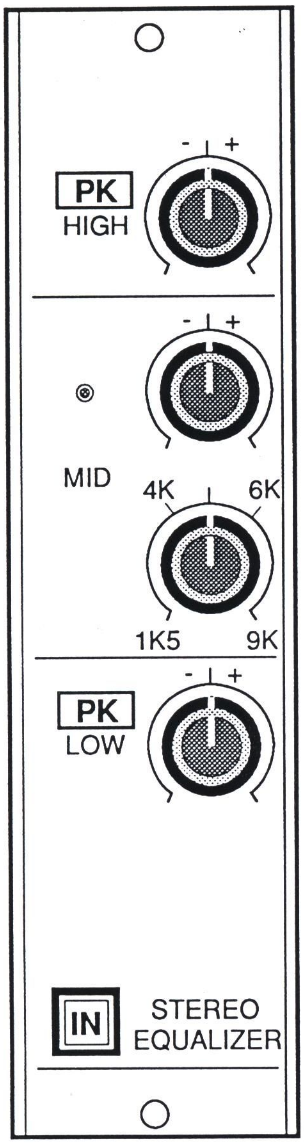
- The IN button switch inserts the three-band equalizer system into the signal path.

- The high and low pass control knobs set the cutoff frequencies of the two tuneable 18 dB/octave filters.

- The IN button switch inserts the filter system into the signal path.

3.7 STEREO EQUALIZER MODULE

This section contains a figure illustrating the Stereo Equalizer Module's front panel controls, along with descriptions of the function(s) of each.



- The HIGH control adjusts the amount of boost and cut of the high frequencies. The center frequency of operation is 9 kHz. The PK button switch changes the shape of the equalization curves from shelving to peaking.

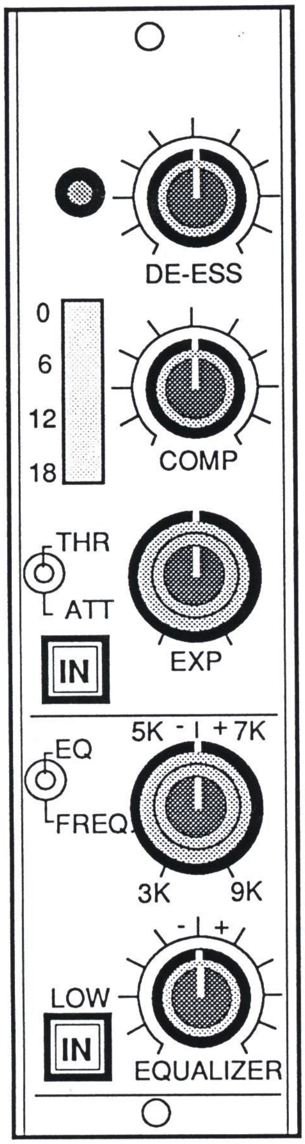
- The upper knob of the two MID controls tunes the center frequency of this stage, while the lower knob adjusts the amount of boost and cut. The frequency tuning range is 1500 Hz to 9 kHz. The "automatic parametric" feature of this circuit continuously increases the "Q" of the equalizer with increasing amounts of boost and cut.

- The LOW control adjusts the amount of boost and cut of the low frequencies. The center frequency of operation is 80 Hz. The PK button switch changes the shape of the equalization curves from shelving to peaking. An internal option switch can set the cut mode as shelving in recognition of real-life equalizer applications (reference Section 2.9.7).

- The IN button switch inserts the three-band equalizer system into the signal path.

3.8 VOICE PROCESSOR MODULE

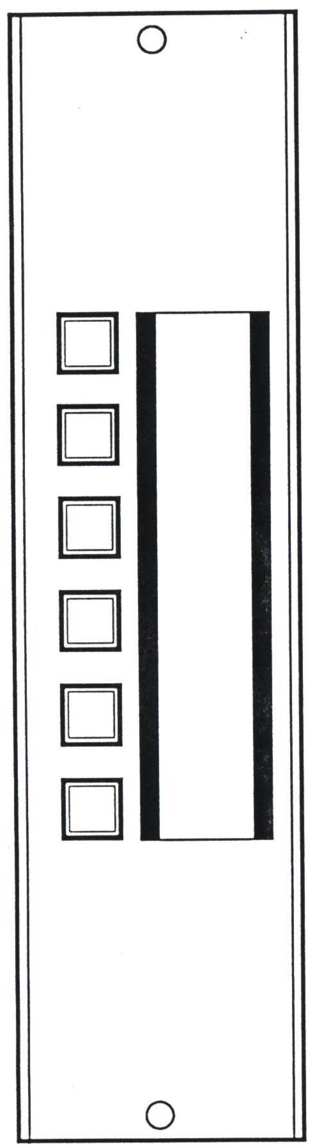
This section contains a figure illustrating the Voice Processor Module's front panel controls, along with descriptions of the function(s) of each.



- The DE-ESS potentiometer sets the threshold of operation for the de-essing control circuitry. The LED illuminates to indicate de-essing action.
- The COMPressor control sets the fixed gain value of the compressor VCA. Since the threshold of the compressor is fixed, the COMPressor control has the same effect as the traditional compressor input control.
- The LED dot/bar graph display is used to indicate both the amount of expander attenuation and the degree of compression. The expander attenuation is displayed as single illuminated segments (dots) while a bar graph is used to indicate the gain reduction of the compressor.
- The EXPander ATTenuation knob controls the amount of static attenuation and the EXPander THReshold knob adjusts the sensitivity of the expander control circuits.
- The IN button inserts the expander, compressor and de-esser into the audio signal path.
- The FREQuency sweep knob selects the frequency of operation and the EQualizer knob adjusts the degree of boost and cut of the high frequency equalizer.
- The low frequency knob adjusts the amount of boost and cut of the low frequency equalizer.
- The IN button inserts the equalizer into the circuit path.

3.9 REMOTE LINE SELECTOR

This section contains a figure illustrating the Remote Line Selector's front panel controls, along with descriptions of the function(s) of each.



- The six station switch may select from any of the six stereo inputs connected to the connector interface panel. The stereo output of the line selector is brought out to the connector interface panel for routing to console input positions, tape recorder inputs, etc.

NOTE: Legend strip dimensions are provided in Section 2.1.

EQUIPMENT DESCRIPTION

4.0 EQUIPMENT DESCRIPTION

This chapter describes the function(s) and circuitry of the various components incorporated in the STEREOMIXER and its associated modules. Included are sections describing the mainframe, power supply assembly, System Master Module, Microphone Input Module, Stereo Line Input Module, Stereo Tape Recorder Input/Output Module, Telephone Input/Mix-Minus Module, Monaural Equalizer Module, Stereo Equalizer Module, Voice Processor Module, and Remote Line Selector. For associated diagrams and schematics, see Chapter 6 of this document.

4.1 MAINFRAME

All of the STEREOMIXER's interconnections are accomplished through a single mother board. The input and accessory modules plug into edge connectors mounted on the front side of the mother board. On the rear of the mother board, shielded ribbon cables are used to attach the connector interface panels. The power supply plugs into an edge connector mounted on the rear of the mother board.

The mother board contains no components other than the connectors themselves, thereby isolating all of the active components to removable modules. This design approach minimizes system "down time" by providing the option of simply replacing any defective module with a "spare" module.

4.2 POWER SUPPLY ASSEMBLY

4.2.1 Function

The regulated power supply is a plug-in assembly which is installed in the rear of the mainframe, immediately behind the System Master Module. The supply uses a toroidal mains transformer to minimize the radiated magnetic field, and three integrated circuit, plug-in, adjustable regulators to provide the required bi-polar audio and mono-polar logic supply voltages. In addition, each of the STEREOMIXER's plug-in modules is equipped with on-board audio supply regulators to minimize power supply bus crosstalk.

The power supply assembly provides the power for all audio and logic circuits in the STEREOMIXER. The design employs a straightforward regulator design. The AC mains supply is protected against faults by a fuse located on the rear of the power supply assembly. The power supply is protected against load faults by power and current limiting internal to the IC regulators.

In addition, the audio and logic Molex connectors for the System Master Module are mounted on the power supply assembly.

4.2.2 Circuitry

The bipolar 20 volt, .75 amp, audio supply is configured by two identical +20 volt monopolar supplies, which are powered from two separate windings of power transformer T1. The first secondary of T1 (wires E12 and E13) is rectified by diode bridge BR2, filtered by capacitor C8, and regulated by series regulator U2. The various small components around the regulator are used to set the output voltage and improve the noise and transient response of the regulator. The second secondary of T1 (wires E14 and E15) is rectified by diode bridge BR3, filtered by capacitor C9, and regulated by series regulator U3. These high-current, low-noise regulators are of a plug-in integrated circuit design, and are mounted on the heat sink located on the outside of the power supply bracket.

The monopolar 13 volt, 2.4 amp logic supply uses the same circuit topology as the audio supply. The third secondary of T1 (wires E10 and E11) is rectified by diode bridge BR1, filtered by capacitor C7, and regulated by series regulator U1. The various small components around the regulator are used to set the output voltage and improve the noise and transient response of the regulator. This regulator, like the audio supply regulators, is also mounted on the heat sink located on the outside of the power supply bracket. A current-limited 12 volt output is available on the logic connector located on the rear of the power supply assembly. This output is provided by IC regulator U4 configured with current limit feedback via resistor R7.

The common point on these three supplies is the central grounding point for the entire system, and is distributed from this point as "Audio Common" and "Logic Common". This point is also tied to "Earth Ground" and "Chassis Ground".

4.3 SYSTEM MASTER MODULE

4.3.1 Function

The System Master Module contains the PROGRAM bus mixing amplifiers, output amplifiers, monitor system, slate and talkback systems, and stereo VU meters with drivers. It also provides a patch point for the stereo PROGRAM buses, which is a convenient point for the connection of external signal processing.

The monitor source for the Control Room monitor and headphone outputs is determined by the monitor selector on the front panel. The manual selection is overridden whenever PFL or SOLO has been activated on an input module. Depressing PFL on an input module sends a "pre-fader listen" signal to the System Master Module. This signal is the output of the input channel's preamp, taken before the fader and pan circuits, and enables the operator to verify an input signal without turning a module on or turning up the fader. PFL is similar in function to "broadcast CUE", except that PFL appears on the main monitor outputs (in stereo) instead of a separate cue output. Depressing SOLO on any input module sends the post-fader, post-pan signal to the System Master Module monitor circuits. The SOLO function provides a very convenient method of independently monitoring inputs for verifying sources and/or searching for a "problem" input in a mixing situation (i.e., noisy, intermittent, etc.).

The front panel headphone volume control affects the level at both the front panel headphone jack and the rear panel headphone connection. The module is equipped with a stereo headphone amplifier for the mixer operator. The monitor volume control affects the level of the rear panel C/R monitor output, which is intended to drive an amplifier and loudspeakers in the Control Room.

The front panel METER switch selects the VU meter operating modes. When the switch is in the PROGRAM position, the meters display the output levels of the PROGRAM amplifiers. When the switch is in the MONITOR position, the meters display the level of the source selected by the monitor system. The source is determined by the monitor switch on the front panel of the System Master Module, or by selection of PFL or SOLO on any of the input modules. By using this "meter follows monitor" function, the operator can depress the PFL or SOLO buttons on any input module and use the monitor and VU meters to preview and set levels without affecting the PROGRAM bus (using the SOLO function in this manner provides the functional equivalent of an AUDITION bus).

The System Master Module also includes provisions for communicating to another location via the Talk to External system. Pressing the TALK TO EXTERNAL button dims the Control Room monitors and sends the preamplified output of the built-in electret talkback microphone to the rear connector panel. An open collector transistor signal is available to interface with the logic of the talkback destination. For example, TALK TO EXTERNAL may be used to communicate with a two-way unit, producer's booth, newsroom or call screener. An external location or source may communicate with the mixer through the TALK TO CONTROL ROOM audio input. The TALK TO CONTROL ROOM logic command dims the Control Room monitors and illuminates the PFL/SOLO monitor tally on the front of the master module for the duration of the talkback command.

Pressing the SLATE button mutes the Control Room monitors and sends a mix of the talkback microphone and a 30 Hz oscillator to the PROGRAM summing amplifiers. The Slate function is normally used for audio marking during recording to simplify editing, however, since the mixer may also be employed as an "On Air" console, an internal DIP switch is provided to disable the Slate function entirely (reference Section 2.9.1).

Whenever a SLATE button is depressed, the monitors are immediately muted and the slate oscillator and console microphone audio are ramped up slowly after a brief delay. When the SLATE button is released, the mic audio and slate tone is ramped down immediately, and the monitors are un-muted after a brief delay.

There are two Studio monitor outputs: one muting and the other non-muting. The muting output is intended to drive an amplifier and loudspeakers in the Studio. The muting of this output is controlled by the selection of a DIP switch on the appropriate microphone input modules (reference Section 2.9.2). The non-muting output is intended to drive a headphone amplifier for the Studio headphones. Both of these outputs always monitor the console PROGRAM output. The TALK TO STUDIO function dims both the Control Room and Studio monitors, and sends the output of the console talkback microphone to the Studio monitor outputs.

4.3.2 Circuitry

<u>NOTE</u>: Each of these circuits has a relationship with the others as can best be appreciated by reviewing the functional block diagram located in Section 6.1 of this document.

<u>MAIN BOARD</u>

The circuitry is divided into two printed circuit assemblies mounted on opposite sides of the panel extrusion. The larger of these two assemblies is the Main Board, PR&E #95-421, which plugs into the mother board and contains the following circuitry:

AUDIO

NOTE: The stereo audio path consist of two identical sets of circuitry. For clarity and simplicity, only the left channel is described below.

The left PROGRAM bus input is summed by amplifier U1A, which operates at unity gain and has a nominal level of -10 dBu. The output provides an inverted mix of the PROGRAM bus to the left TELCO MIX BUS on the mother board. This signal is used by the telephone modules for their mix-minus outputs, and is also fed to inverting amplifier U1B, which operates at unity gain, and provides the PATCH SEND LEFT source.

The signal from PATCH RETURN LEFT is fed into unity gain differential amplifier U3A, whose output feeds the LEFT OUTPUT ADJustment pot R26. This control has enough range to adjust the output level to operate from +4 dBm to +8 dBm. The wiper of R26 feeds the non-inverting input of output amplifier U4A, which drives the high PROGRAM LEFT OUTput. The low PROGRAM LEFT OUTput is provided by unity inverter U4B, which derives it source from the output of U4A. U4A also drives the TAPE MIX BUS through attenuating resistors R51 and R148 at a level 22 dB below the level of the PROGRAM LEFT OUTput. The TAPE MIX BUS is used by the Tape modules for their SEND TO TAPE outputs.

The variable attenuating network comprised of R54, R55, R56, and R57 serve to feed a sample of the PROGRAM LEFT OUTput signal to the VU meter amplifiers. These amplifiers are located on the satellite card and are covered in the section describing that board. LEFT VU ADJustment pot R56 has enough range to attenuate signals in the +4 dBm to +8 dBm range to -10 dBu, which is the voltage required by the VU amplifiers.

The three section Monitor Switch selects between PROGRAM, AUX 1, and AUX 2. The output of the selector switch is fed to a variable attenuator comprised of R65, R66, R67, and R68. The attenuated signal is fed into differential amplifier U6A, which is operating with 10 dB of attenuation and an output level of -20 dBu. There is enough range on R67 to accommodate Monitor Input levels from +4 dBu to +8 dBu.

The output of U6A is fed through audio selection FET Q2 to inverting summing amplifier U8A, where it is amplified by 10 dB. FET Q2 is on whenever no PFL or SOLO buttons have been depressed. When PFL or SOLO has been selected on any input module, FET Q2 turns off and the input to U8A comes from the PFL/SOLO summing bus. The other input to U8A is from the TALK TO C/R INPUT. This nominal 0 dBu input is fed into differential amplifier U7A, which operates at 10 dB of attenuation. The -10 dBu output signal is fed through R94 and audio selection FET Q1 to the input of U8A. Q1 is on whenever the TALK TO C/R logic input has been activated. Whenever dimming FET Q5 is turned on, feedback resistor R102 is paralleled with R103, dropping the nominal gain of this stage by 12 dB.

The -10 dBu signal at the output of U8A drives the headphone amplifier located on the satellite card and the front panel MONITOR pot. The wiper of the MONITOR pot feeds C/R MONITOR output amplifier

U9A through R108 and audio muting FET Q7. Q7 turns off whenever the MUTE C/R logic has been activated. U9A is an inverting amplifier operating with 10 dB of gain and a nominal output level of 0 dBu.

LOGIC

When the PFL/SOLO ON BUS is pulled low by an input module, the PFL/SOLO MONITOR indicator lamp illuminates. Also, Q9 is turned off, which turns off audio selection FETs Q2 and Q3.

When the TALK TO C/R logic input is pulled low, the PFL/SOLO MONITOR indicator lamp illuminates and a high appearing at the output of U10B turns on Q14 which turns on audio selection FETs Q1 and Q4. Also, a high appears at the output of U10C, which turns on Q10 and dims the Control Room by turning on dimming FETs Q5 and Q6. Pulling the DIM C/R input low, either by an input module or an external logic command, will also dim the Control Room.

A MUTE C/R input low from the bus or an external logic command will cause the output of U11C to go high, charging C80 instantly and causing a low at the output of U11D. This will turn off Q15 and subsequently turn off muting FETs Q7 and Q8. The low at U11D will also turn on Q11, which drives the C/R WARNING TALLY output through short circuit protection regulator U12. When the MUTE C/R input returns high, there is a 100 millisecond delay before the Monitor un-mutes. This delay is caused by the discharge of C80 through R124, and prevents feedback through a microphone that has not fully turned off.

A low on the MUTE STUDIO bus causes the output of U11E to go high, charging C81 instantly and creating a low at the output of U11F. This low turns on Q12, which drives the STUDIO WARNING TALLY output through short circuit protection regulator U13. The output of U11F is also fed to the satellite board as the MUTE STUDIO DELAYED signal, where it mutes the Studio monitor output whenever it is low.

SATELLITE BOARD

The Satellite Board, PR&E #95-444, which connects to the Main Board via a shielded ribbon cable, contains the following circuitry:

AUDIO

NOTE: The stereo audio path consist of two identical sets of circuitry. For clarity and simplicity, only the left channel is described below.

The VU METER switch selects between the PROGRAM output and the monitor input to the HEADPHONE pot. The VU meter amplifier consists of dual integrated circuit amplifier U3 operating with cross connected feedback and a gain of +14 dB. The outputs of the two amplifiers drive the VU meter through the recommended source resistance of 3.57 kΩ. Light emitting diodes DS1 and DS2 provide overdrive protection to the meter movement.

The wiper of the HEADPHONE pot feeds inverting headphone amplifier function module A1, which is operating with 10 dB of gain and drives the HEADPHONE JACK and the C/R HEADPHONE Output through R36 at a nominal level of 0 dBu.

The low frequency slate tone is produced by a phase shift oscillator consisting of R22, C18, R23, C19, R24, and C20 operating as cascading low-pass filters to produce a phase shift at 30 Hz, followed by buffer amplifier U3A and inverting amplifier U3B. This oscillator is always free running to prevent startup delay or settling time. The level of the oscillator may be adjusted using SLATE OSCillator ADJustment pot R31.

The console microphone is an electret type, and is powered by a filtered +7 volts DC supply derived by CR1, R1, R2, and C1. The signal from the microphone is connected to preamplifier U1A, which operates with a gain of 46 dB. The output of the mic preamp is fed to TALKBACK MIC ADJustment pot R10 and SLATE MIC ADJustment pot R16.

The signals at the wipers of R16 and R31 are routed through FETs Q1 and Q2 and summed together by U2B, which is an inverting amplifier operating with 20 dB of gain, and whose output is fed to the PROGRAM left and right summing amplifiers on the main board. The wiper of R10, the TALKBACK MIC ADJustment pot, drives inverting amplifier U2A, operating at 20 dB of gain, and whose output drives the TALK TO EXTernal OUTPUT through the TALK TO EXTERNAL switch. The output of U2A is also connected to the TALK TO STUDIO switch.

The other connection to the TALK TO STUDIO switch is the PROGRAM LEFT signal coming from the main board. If the TALK TO STUDIO switch is in the normal (out) position, the PROGRAM LEFT audio is fed to inverting amplifier U4A through R42 and muting FET Q3. If the TALK TO STUDIO switch is in the in position, the PROGRAM LEFT audio is dimmed approximately 12 dB by R40 and mixed with the preamplified microphone signal. U4A operates with 16 dB of gain, and its output drives the STUDIO MONitor LEFT output at a nominal level of 0 dBu.

The TALK TO STUDIO switch also feeds inverting amplifier U5A. This amplifier operates at 16 dB of gain, and its output drives the STUDIO H/P LEFT output at a nominal level of 0 dBu.

LOGIC

When the SLATE button is depressed, the output of U6A goes high, turning on Q5 and muting the Control Room. Meanwhile, the output of U8B goes low and discharges C64 through R82 and, after this 300 millisecond delay, the output of U8C goes high and turns on Q6, which turns on Q1 and Q2 via a control voltage ramp created by R86 and C28. When the SLATE button is released, the 300 millisecond turn-on delay is bypassed by CR2. The SLATE function can be disabled by opening SLATE DISABLE DIP switch S5.

When the TALK TO EXTERNAL button is depressed, the output of U8D goes high and turns on Q7, which dims the Control Room. The Control Room is also dimmed when the TALK TO STUDIO button is depressed. The output of U8D also turns on Q8, which provides the TALK TO EXTernal logic command.

When the MUTE STUDIO DELAYED goes low, Q9 turns off, thereby turning off Q3 and Q4.

4.4 MICROPHONE INPUT MODULE

4.4.1 Function

The Microphone Input Module selects either of two inputs with the front panel switch, and is designed to accommodate a nominal input level range of -70 dBu to -30 dBu. The input preamplifier is an active differential design configured on a plug-in board. The preamplifier delivers a -10 dBu level to a patch connector located on the rear panel; the patch return is buffered by a balanced differential amplifier. The module is switched ON/OFF by a momentary button operating logic controlled switching. The module is equipped with a Penny & Giles conductive plastic fader and logic controlled PFL (pre-fader listen) and SOLO (after fader listen) monitor buttons. These two buttons actuate logic circuitry which clears any existing input monitor selection, and overrides the monitor selection at the System Master Module. Pressing two or more PFL/SOLO buttons simultaneously allows the operator to monitor more than one input source at a time. The PAN control is used to position the microphone signal in the stereo image.

The remote logic control connector is configured like larger PR&E consoles (i.e., BMX, AMX, ABX), therefore, it is possible to utilize standard remote control panels and cable assemblies. In addition to the remote control of ON and OFF, the module also provides remote COUGH, TALK TO CONTROL ROOM and PRIVACY functions. The COUGH feature temporarily mutes the feed to the PROGRAM bus when the module is ON. The TALK TO CONTROL ROOM feature allows a Studio located microphone to communicate with the mixer by temporarily overriding the existing control monitor selection. In addition, this feature mutes the feed to PROGRAM, dims both the Control Room and Studio monitors, and illuminates the PFL button on the associated microphone module and the PFL indicator on the System Master Module. Engaging the PRIVACY feature disables the microphone input module's PFL and SOLO functions.

Internal DIP switches are used to select the remote logic for the A or B inputs. The remote logic is only active when the corresponding input has been selected on the front panel. The logic will automatically reset to OFF if the input selector is switched while the module is ON.

<u>NOTE</u>: When both A and B input logic is connected, the input selector turns the selected input logic ON when switched while the module is ON (reference Section 2.8.2).

This module has two sets of internal DIP switches to control monitor muting functions for the Control Room or Studio monitors, depending on the location of the microphone (reference Section 2.9.2).

4.4.2 Circuitry

<u>AUDIO</u>

The input to the module is fed through the A/B selector switch to an active balanced differential microphone preamplifier. This amplifier is configured with three operational amplifiers and a matched transistor array, contained on plug-in function amplifier A1. The voltage gain range from module input to the output of the amplifier is +22 dB to +60 dB. Gain trim control R4 is provided to adjust the preamplifier gain, which is adjusted to provide a nominal -10 dBu level to the PATCH SEND output.

The PATCH RETURN drives balanced differential amplifier U1A operating at unity gain, the output of which drives MIX fader R13 at -10 dBu. The "in-hand" attenuation of R13 is 10 dB, which is recovered by fader buffer amplifier U1A to the -10 dBu internal system level and fed to the PAN pot R18. The left and right outputs of the PAN circuit are buffered and amplified 3 dB by dual amplifier U2 and routed to channel FETs Q1 and Q2 and to SOLO FETs Q7 and Q8. The channel FETs route the stereo signal to the left and right PROGRAM summing buses, and the SOLO FETs drive the PFL/SOLO summing buses.

PATCH RETURN amplifier U1A also drives PFL FETs Q5 and Q6, which route the pre-fader signal to the PFL/SOLO summing buses. FETs Q3 and Q4 serve to boost the gain of the PFL signal by 12 dB when the module is talking to the Control Room. This allows the talk signal to appear at normal volume while the Control Room monitors are dimmed.

LOGIC

The selection of the A or B input to the module controls LOGIC COMMON pin 1 on the remote logic connector. The setting of DIP switches S2-1 and S2-2 will determine whether or not pin 1 will be grounded. If for example, S2-1 is closed (Logic = A) and S2-2 is open, when the A/B selector is switched to the "A" position, pin 1 will be grounded, thereby enabling the remote functions. When the A/B selector is in the "B" position, pin 1 will not be grounded, and the remote functions will be disabled because any switches, lamps, relay interfaces, etc., will have no current return path to the logic connector. The A/B selector also feeds DIP switches S2-3, S2-4, S2-5, and S2-6, which determine whether the selection of the A or B input should mute the Control Room or the Studio when the module is turned on.

When the module is off, depressing the ON/OFF button on the module, or the ON button on the selected input's remote control panel, will cause bi-stable U20B pin 12 to go low, causing the output of U9B to go high and U8F to go low. This illuminates the on tally lamp in the ON/OFF button by turning on Q23. Regulator U14 is configured as a current limiting device to protect the transistor against short circuits. The remote ON TALLY is also turned on by transistor Q19 ,through short circuit protector U10. When the output of U8F goes low, C48 is discharged through R76, causing a 50 millisecond delay before the output of U8A goes high and saturates Q15, which turns on PROGRAM audio FETs Q1 and Q2.

When the module is on, depressing the ON/OFF button on the module, or the OFF button on the selected input's remote control panel, will toggle U20B pin 12 high. This high is inverted by U7D to turn on the off tally lamp in the ON/OFF button via Q23 and U14. The remote OFF TALLY is turned on via Q20 and U11. The high at U20B pin 12 forces the output of U9B low and U8F high, which causes the output of U8A to go low and turn off Q15 and, subsequently, Q1 and Q2. Time delay network R76 and C48 is bypassed by CR11 to eliminate any delay in turning the audio FETs off.

Operation of the A/B input selector button places a momentary low pulse at the input of U16A, which will key the channel logic to the "off" state to prevent any input switching transient from being routed into the console mix buses.

The remote control COUGH muting function is achieved by pulling the logic COUGH line low, which causes the output of U21F to go high and force U9B low and U8F high. This extinguishes the ON tally lights and mutes the channel audio output. When the COUGH line is released, the channel output is unmuted and the ON lamps illuminate.

Pressing the PFL button gives a high at the output of U21B, which toggles bi-stable U19B pin 13 high. When the PFL function is engaged, the high at U19B pin 13 forces the output of U17A low, and turns on the PFL audio FETs via the high at the output of U8C, which saturates Q13. The low at U17A turns on the PFL tally and forces the output of U16D high, which pulls the PFL/SOLO ON command bus low via Q17.

<u>NOTE</u>: The PFL/SOLO ON command bus is pulled low whenever PFL, SOLO, or TALK TO C/R functions are enabled.

If the PFL function is engaged on a Control Room microphone, the high at the output of U19B forces the output of U9A low and the output of U7A high, which pulls the MUTE C/R bus low via Q10 to prevent feedback. The low signal caused by the depression of the PFL button is inverted and routed by U18A to reset the SOLO bi-stable if the SOLO function was engaged.

Pressing the SOLO button gives a high at the output of U21B, which toggles bi-stable U19A pin 1 high and pin 2 low. When the SOLO function is engaged, the low at U19A pin 2 turns on the SOLO audio FETs via the high at the output of U8B, which saturates Q14. The low at U19A pin 2 turns on the SOLO tally and forces the output of U16D high, which pulls the PFL/SOLO ON command bus low via Q17. If the SOLO function is engaged on a Control Room microphone, the high at the output of U19A pin 1 forces the output of U9A low and the output of U7A high, which pulls the MUTE C/R bus low via Q10 to prevent feedback. When the SOLO function is engaged, the low at U19A pin 2 causes a high pulse at the output of U18B, which will reset PFL bi-stable U19B if the PFL function was engaged.

The PFL and SOLO buttons also drive U18D, whose output drives the PFL/SOLO CLEAR bus via Q16. This function is used to reset the PFL or SOLO functions on any other modules. When a low appears on the PFL/SOLO CLEAR bus, the output of U16C goes high, and triggers monostable U15B after a brief delay caused by R84 and C54. The low pulse appearing at U15 pin 9 is routed via U16B, U7C, U18A, and U18B to both the PFL and SOLO bi-stables in order to reset them. The reset pulse on a given module is disabled whenever the PFL/SOLO CLEAR bus command has originated from that module in that the PFL and SOLO buttons drive the two inputs of U18D, whose output is inverted by U8C which drives pin 13 of U15B. If either the PFL or SOLO button is depressed, U15B pin 13 will be driven high at the same time that the PFL/SOLO CLEAR bus is pulled low. This prevents the reset pulse on the monostable from resetting the CUE or SOLO bi-stables.

The PFL and SOLO functions are also reset whenever the module is turned on. When channel ON/OFF bi-stable U20B is set, a low appears at pin 12, which is coupled to U16C through a differentiating network consisting of C57 and R87. This creates a high pulse at the output of U16C, which triggers monostable U15B. The low pulse at the output of the monostable resets the PFL and SOLO bi-stables via U16B, U7C, U18A and U18B.

A remote control "privacy" mode is available which prevents the module from being able to enter the PFL or SOLO functions. This INHIBIT function pulls the input to U16B low forcing its output high and thereby holding the reset lines of the PFL and SOLO bi-stables high to lock them in the "off" mode.

For an input module to talk to the Control Room, the TALK TO C/R input is pulled low by an external "talk to Control Room" pushbutton switch. This will cause the output of U17B to go high if the currently selected input is a Studio microphone. This high actuates the DIM C/R command bus via transistor Q11, and turns on the PFL/SOLO ON bus by forcing the output of U17A low, which turns on the PFL tally and forces the output of U16D high and turns on Q17. The high output of U17B also forces the output of U9B low and U8F high, thereby extinguishing the ON tally and muting the channel audio FETs. The high at the output of U17B also turns on the PFL boost FETs by turning on Q13.

4.5 STEREO LINE INPUT MODULE

4.5.1 Function

The Stereo Line Input Module accepts either of two inputs by front panel selection. The input amplifiers are designed to accommodate the nominal input level range of -30 dBu to +9 dBu. The module is equipped with input MODE switching, which selects the operating signal mode. When both MODE buttons are in the "out" position, the channel mode is stereo. When the L button is depressed, the left input signal will feed both the left and right buses. Pressing the R button alone will cause the right input to feed both buses, while pressing both buttons will sum the left and right inputs and feed the composite signal to both the left and right buses. The button caps are the self-indicating type to alert the operator when any non-stereo mode is selected.

The mode switch output feeds the Penny & Giles conductive plastic stereo fader and the PFL system. The fader is buffered and the signal fed to a PAN/BALANCE circuit, which provides the balancing of a stereo source or the panning of a monaural source within the stereo image.

The assignment of the module to the stereo PROGRAM buses is controlled by the momentary ON/OFF switch on the front panel. Remote control of ON, OFF, PFL and SOLO is brought out to a logic connector located on the rear panel. Since the logic connector is configured like the larger BMX, AMX and ABX consoles, it is possible to utilize standard machine control interfaces and cable assemblies. The ON/OFF function may also provide start and stop pulses for the control of turntables, cartridge, tape, cassette and CD machines. An audio reset facility is provided which allows a cartridge machine or other device to turn the module off automatically at end of event. The module's OFF lamp may be interfaced with a source machine to tally a "ready" indication, such as cartridge inserted and available for play.

Internal DIP switches are used to select the remote logic for the A or B inputs (reference Section 2.9.3). The remote control logic is only active when the corresponding input has been selected on the front panel. The logic will automatically reset to OFF if the input selector is switched while the module is ON.

<u>NOTE</u>: When **both** A and B input logic is connected, the input selector turns the selected input logic ON when switched while the module is ON (reference Section 2.8.3).

The control logic also has provisions for resetting an external event timer. If the internal timer reset DIP switch is engaged, the module will output a 20 mS pulse to the timer reset bus whenever the module is turned ON. The reset signal is terminated on the System Master Module logic connector.

The Stereo Line Input module is also equipped with the PFL and SOLO facilities, as described for the Microphone Input module. In addition, both of these functions are available on the module's logic connector for remote control applications.

4.5.2 Circuitry

AUDIO

NOTE: The stereo audio path consist of two identical sets of circuitry. For clarity and simplicity, only the left channel is described below.

The input to the module is a balanced, symmetrical input, instrumentation amplifier. This amplifier is configured with three operational amplifiers, contained on the plug-in function module A1, operating with a cross-coupled input stage feedback gain trim control.

To accommodate a wide variety of input levels, this module is configured with a switchable attenuator, which is controlled by S2. The voltage gain range from module input to the output of A1 is -20 dB to +1 dB with S2 switched on, and -1 to +20 dB with S2 switched off. The gain trim control is adjusted to provide a level of -10 dB to the input of the MODE switch. The output of the MODE switch drives the MIX fader and PFL FETs Q5 and Q6, which feed the PFL/SOLO summing buses.

The MIX fader "in-hand" attenuation of 10 dB is recovered by fader buffer amplifier U1A to the -10 dBu internal system level, and routed to the PAN pot. The PAN pot is buffered by dual amplifier U2, which operates at a gain of 3 dB and whose outputs are connected to channel FETs Q1 and Q2, and to SOLO FETs Q3 and Q4. The channel FETs route the stereo signal to the left and right PROGRAM summing buses, and the SOLO FETs drive the PFL/SOLO summing buses.

LOGIC

The selection of the A or B input to the module controls the LOGIC COMMON pin 1 on the remote logic connector. The setting of DIP switches S5-3 and S5-4 will determine whether or not pin 1 will be grounded. If, for example, S5-3 is closed (Logic = A) and S5-4 is open, when the A/B selector is switched to the "A" position, pin 1 will be grounded, thereby enabling the remote functions. When the A/B selector is in the "B" position, pin 1 will not be grounded, and the remote functions will be disabled because any switches, lamps, relay interfaces, etc., will have no current return path to the logic connector.

When the module is off, depressing the ON/OFF button on the module, or the ON button on the selected input's remote control panel, will cause the bi-stable U19B pin 12 to go low, illuminating the on tally lamp in the ON/OFF button by turning on Q13. Regulator U13 is configured as a current-limiting device to protect the transistor against short circuits. The remote ON TALLY is also turned on by transistor Q22, through short circuit protector U24. The low at U19B is inverted by U20F, the output of which drives

START PULSE monostable U6B and turns on the PROGRAM audio FETs by saturating Q9. START PULSE monostable U6B is triggered whenever the module is turned on, causing a low pulse for 200 milliseconds at U6B pin 7, turning on transistor Q18 which drives the logic connector through short circuit protector U9. The low pulse at U6B pin 7 will trigger TIMER RESET monostable U7B if TIMER RESET DIP switch S5-5 is closed. The output of U7B pin 6 pulls the TIMER RESET command bus low by turning on Q16 for 20 milliseconds.

When the module is on, depressing the ON/OFF button on the module, or the OFF button on the selected input's remote control panel, will toggle U19B pin 12 high. This high is inverted by U18C to illuminate the off tally lamp in the ON/OFF button, via Q12 and U12. The remote OFF TALLY is turned on via Q21 and U23. The high at U21B pin 12 forces the output of U20F low, which turns off Q9 and, subsequently, Q1 and Q2. The high at U19B pin 12 triggers STOP PULSE monostable U7A, whose output, pin 9, turns on Q17 for 200 milliseconds. Whenever the ON/OFF bi-stable is reset to off, either by pulling low the remote AUDIO RESET line or toggling the module A/B input selector, the STOP PULSE monostable is disabled by the low appearing at U7A pin 13.

Operation of the A/B input selector button places a momentary low pulse at the input of U15D, which will key the channel logic to the "off" state to prevent any input switching transient from being routed into the console mix buses.

The module's OFF lamp may be remotely controlled by applying a low to the selected remote READY input. This is inverted by U20E and routed to U18C pin 5. The output of U18C will go low and illuminate the off tally if the module ON/OFF bi-stable is off and the READY input is low. This provides the facility of using the module's OFF lamp to tally the ready state of remote controlled equipment. If there is no connection to this logic function for a particular input, the OFF lamp will operate normally by closing DIP switch S5-1 for the A input or S5-2 for the B input.

Pressing the module or the selected input's remote control PFL button causes a high to appear at the output of U20B, toggling bi-stable U14B pin 13 high, which turns on the PFL audio FETs by saturating Q7. The low appearing at U14B pin 12 turns on the local and remote PFL tallies via Q11 and Q20, and forces the output of U15A high, pulling the PFL/SOLO ON command bus low via Q14. The low signal caused by the depression of the PFL button is inverted and routed by U18A to reset the SOLO bi-stable if the SOLO function was engaged.

Pressing the module or the selected input's remote control SOLO button causes a high at the output of U20A, which toggles bi-stable U14A pin 1 high, turning on the SOLO audio FETs via Q8. When the SOLO function is engaged, the low at U14A pin 2 illuminates the local and remote SOLO tallies via Q10 and Q19. When the SOLO function is engaged, the low at U14A pin 2 causes a high pulse at the output of U16B, which will reset PFL bi-stable U14B if the PFL function was engaged.

The PFL and SOLO buttons also drive U18A, whose output drives the PFL/SOLO CLEAR bus via Q15. This function is used to reset the PFL or SOLO functions on any other modules. When a low appears on the PFL/SOLO CLEAR bus, the output of U15B goes high and triggers monostable U6A after a brief delay caused by R80 and C50. The low pulse appearing at U15 pin 9 is inverted and routed via U17D, U16A, and U16B to both the PFL and SOLO bi-stables in order to reset them. The reset pulse on a given module is disabled whenever the PFL/SOLO CLEAR bus command has originated from that module, in that the PFL and SOLO buttons drive the two inputs of U18A, whose output is inverted by U17C, which drives pin 13 of U6A. If either the PFL or SOLO button is depressed, U6A pin 13 will be driven high at the same time that the PFL/SOLO CLEAR bus is pulled low. This prevents the reset pulse on the monostable from resetting the CUE or SOLO bi-stables.

The PFL and SOLO functions are also reset whenever the module is turned on. When channel ON/OFF bi-stable U19B is set, START PULSE monostable U6B is triggered, which drives the output of U15B high for 200 milliseconds. This, in turn, triggers monostable U6A, which delivers a low pulse to U15C which resets the PFL and SOLO bi-stables via U17D, U16A, and U16B.

4.6 TAPE RECORDER INPUT/OUTPUT MODULE

4.6.1 Function

The Tape Recorder Input/Output Module serves two functions. First, it is the input module for the playback of a reel, cartridge or cassette tape recorder. Second, it contains an output switch and balanced line amplifiers to feed the input of the recorder. The SEND TO TAPE button operates logic circuitry, which prevents the possibility of "round-robin" feedback via the tape by locking out the channel ON mode. The PFL and SOLO monitor functions are not locked out, and may be used to monitor the recorder output at any time.

The module has a dedicated input with a stereo pair of active balanced differential preamplifiers. The input gain of the module is not adjustable, but is switch selectable by an internal DIP switch to accommodate the standard operating levels of +4 dBu and +8 dBu (reference Section 2.9.4). The output of the preamplifiers feed the mode switch, stereo fader, and pan/balance controls identical to those of a Stereo Line Input Module.

The Tape Input/Output module also contains dedicated line amplifiers, which are used to feed the stereo inputs of the tape machine. The output level of these amplifiers is determined by the operating reference level of the System Master Module. The output amplifiers are a balanced push-pull configuration.

The module control logic is identical to the Stereo Line Input Module with the addition of the SEND TO TAPE function.

4.6.2 Circuitry

<u>AUDIO</u>

<u>NOTE</u>: The stereo audio path consist of two identical sets of circuitry. For clarity and simplicity, only the left channel is described below.

The input to the module is a balanced differential amplifier U1A operating with a gain (attenuation) of either -14 dB or -18 dB, depending on the setting of input attenuation DIP switch S1. When S1 is closed and the input level is +8 dBu, the output of U1A delivers a level of -10 dBu to the input of the MODE switch. When S1 is open, a +4 dBu input signal produces a level of -10 dBu to the MODE switch. The output of the MODE switch drives the MIX fader and PFL FETs Q5 and Q6, which feed the PFL/SOLO summing buses.

The MIX fader "in-hand" attenuation of 10 dB is recovered by fader buffer amplifier U2A to the -10 dBu internal system level, and routed to the PAN pot. The PAN pot is buffered by dual amplifier U3, which operates at a gain of 3 dB, and whose outputs are connected to channel FETs Q1 and Q2 and to SOLO FETs Q3 and Q4. The channel FETs route the stereo signal to the left and right PROGRAM summing buses, and the SOLO FETs drive the PFL/SOLO summing buses.

The signal appearing at the TO TAPE MIX BUS is derived from the PROGRAM output on the System Master Module, and operates 22 dB below the system operating level (+4 or +8 dBu). The signal is fed through R55 to SEND TO TAPE audio FETs Q7 and Q8 and, subsequently, to inverter U4A, which operates with a gain of +16 dB, and whose output serves as the low TO TAPE audio source. This output also feeds unity gain inverting amplifier U4B, the output of which serves as the high TO TAPE audio source. The level of the balanced TO TAPE output will be the same as the level of the PROGRAM output on the System Master Module, which is determined by the gain trim controls located inside that module.

LOGIC

When the module is off, depressing the ON/OFF button on the module, or the ON button on the remote control panel, will cause bi-stable U22B pin 12 to go low, illuminating the on tally lamp in the ON/OFF button by turning on Q16. Regulator U16 is configured as a current limiting device to protect the transistor against short circuits. The remote ON TALLY is also turned on by transistor Q26 through short circuit protector U28. The low at U22B is inverted by U23B, the output of which drives START PULSE monostable U9B, and turns on the PROGRAM audio FETs by saturating Q9. START PULSE monostable U9B is triggered whenever the module is turned on, causing a low pulse for 200 milliseconds at U9B pin 7, turning on transistor Q22, which drives the logic connector through short circuit protector U12. The low pulse at U9B pin 7 will trigger TIMER RESET monostable U10B if TIMER RESET DIP switch S4-1 is closed. The output of U10B pin 6 pulls the TIMER RESET command bus low by turning on Q20 for 20 milliseconds.

When the module is on, depressing the ON/OFF button on the module, or the OFF button on the remote control panel, will toggle U22B pin 12 high. This high is inverted by U21C to illuminate the off tally lamp in the ON/OFF button, via Q15 and U15. The remote OFF TALLY is turned on via Q25 and U27. The high at U22B pin 12 forces the output of U23B low, which turns off Q9 and, subsequently, Q1 and Q2. The high at U22B pin 12 triggers STOP PULSE monostable U10A, whose output, pin 9, turns on Q21 for 200 milliseconds. The ON/OFF bi-stable is reset to off whenever the remote AUDIO RESET line is pulled low. When this occurs, STOP PULSE monostable U10A is disabled by the high appearing at pin 13.

The module's OFF lamp may be remotely controlled by applying a low to the remote READY input. This is inverted by U23E and routed to U21C pin 5. The output of U21C will go low and illuminate the off tally if the module ON/OFF bi-stable is off and the READY input is low. This provides the facility of using the module's OFF lamp to tally the ready state of remote controlled equipment. If there is no remote connection to this logic function, the OFF lamp will operate normally by closing DIP switch S4-2.

Pressing the module or the remote control PFL button causes a high to appear at the output of U23B, toggling bi-stable U17B pin 13 high, which turns on the PFL audio FETs by saturating Q10. The low appearing at U17B pin 12 turns on the local and remote PFL tallies, via Q13 and Q24, and forces the output of U18A high, pulling the PFL/SOLO ON command bus low via Q18. The low signal caused by the depression of the PFL button is inverted and routed by U19A to reset the SOLO bi-stable if the SOLO function was engaged.

Pressing the module or the remote control SOLO button causes a high at the output of U23A, which toggles bi-stable U17A pin 1 high, turning on the SOLO audio FETs via Q11. When the SOLO function is engaged, the low at U17A pin 2 illuminates the local and remote SOLO tallies, via Q14 and Q23, and causes a high pulse at the output of U19B, which will reset PFL bi-stable U17B if the PFL function was engaged.

The PFL and SOLO buttons also drive U21A, whose output drives the PFL/SOLO CLEAR bus via Q19. This function is used to reset the PFL or SOLO functions on any other modules. When a low appears on the PFL/SOLO CLEAR bus, the output of U18B goes high and triggers monostable U9A, after a brief delay caused by R100 and C66. The low pulse appearing at U9 pin 9 is inverted and routed via U18C, U20D, U19A, and U19B to both the PFL and SOLO bi-stables in order to reset them. The reset pulse on a given module is disabled whenever the PFL/SOLO CLEAR bus command has originated from that module, in that the PFL and SOLO buttons drive the two inputs of U21A, whose output is inverted by U20C, which drives pin 13 of U9A. If either the PFL or SOLO button is depressed, U9A pin 13 will be driven high at the same time that the PFL/SOLO CLEAR bus is pulled low. This prevents the reset pulse on the monostable from resetting the CUE or SOLO bi-stables.

The PFL and SOLO functions are also reset whenever the module is turned on. When channel ON/OFF bi-stable U22B is set, START PULSE monostable U9B is triggered, which drives the output of U18B high for 200 milliseconds. This, in turn, triggers monostable U9A, which delivers a low pulse to U18C, which resets the PFL and SOLO bi-stables via U20D, U19A, and U19B.

Pressing the SEND TO TAPE button causes the output of U20A to go high, toggling bi-stable U22A pin 1 high, which turns on the SEND TO TAPE audio FETs by saturating Q12. When the function is engaged, the low at U22A pin 2 illuminates the SEND TO TAPE tally via Q17. The low at U22A pin 2 is also routed to the ON/OFF bi-stable via U21B, U20E, U19C, U20F, and U21D. Whenever the SEND TO TAPE function is engaged, the resulting high appearing at U22B pin 10 turns off the ON/OFF bi-stable and locks it in the off state until the SEND TO TAPE function is disengaged. By locking the ON/OFF bi-stable in the off mode while the SEND TO TAPE function is engaged, the possibility of feedback via the tape machine is eliminated.

4.7 TELEPHONE INPUT/MIX-MINUS MODULE

4.7.1 Function

The Telephone Input/Mix-Minus Module is a monaural input module which contains mixing, filtering, and output amplifier circuitry to provide a special mix of the PROGRAM signal back to a telco hybrid, less the input of the module itself (mix-minus). The STEREOMIXER will accommodate up to eight of these modules, therefore, it is theoretically possible to mix and control up to eight telephone inputs simultaneously, while providing unique mix-minus foldback signals to each of the callers. This is an extreme example, as most telephone interface requirements for talk shows, news feeds and contests are satisfied by one to four inputs.

The module has front panel selection of A and B (mono) inputs, with the input an active balanced differential preamplifier configured on a plug-in module. The input preamplifier will accommodate nominal input levels of -30 dBu to +9 dBu. The output of the preamplifier feeds fader and pan/balance controls identical to those of a Stereo Line Input Module.

The mix-minus output is a full time feed from the PROGRAM bus, independent of the ON/OFF status of the module. The module's input signal is electronically subtracted from the PROGRAM mix when the module is turned ON. The mix-minus signal is processed by a third order 300 Hz to 3 kHz bandpass filter to enhance hybrid operation, and delivered as a balanced 0 dBu nominal signal.

The module has an additional output labeled DIRECT, which is an output of the input signal selection of the module. The DIRECT output may be internally sourced from the pre or post fader signals, and may optionally follow the ON/OFF status of the module. The DIRECT output is intended for talk show use to feed a "caller speaker", and for contest use to record a caller while either off or onair. The DIRECT output is fed by an independent amplifier through an isolation resistor, and may be coupled to other telephone modules' DIRECT outputs to achieve a mix of the outputs.

Internal DIP switches are used to select the remote logic for the A or B inputs (reference Section 2.9.5). The remote control logic is only active when the corresponding input has been selected on the front panel. The logic will automatically reset to OFF if the input selector is switched while the module is ON.

NOTE: When both A and B input logic is connected, the input selector turns the selected input logic ON when switched while the module is ON (reference Section 2.8.5).

The module is equipped with PFL and SOLO facilities, which operate identically to the Stereo Line Input Module. Remote logic control inputs and outputs are provided for possible interface to hybrid or switching equipment.

4.7.2 Circuitry

<u>AUDIO</u>

The input to the module is a balanced, symmetrical input, instrumentation amplifier. This amplifier is configured with three operational amplifiers, contained on plug-in function module A1, operating with a cross-coupled input stage feedback gain trim control.

To accommodate a wide variety of input levels, this module is configured with a switchable attenuator, which is controlled by S2. The voltage gain range from module input to the output of A1 is -20 dB to

+1 dB with S2 switched on, and -1 to +20 dB with S2 switched off. The gain trim control is adjusted to provide a level of -10 dB to the input of the MIX fader and PFL FETs Q5 and Q6, which feed the PFL/SOLO summing buses.

The MIX fader "in-hand" attenuation of 10 dB is recovered by fader buffer amplifier U1A to the -10 dBu internal system level, and routed to the PAN pot. The PAN pot is buffered by dual amplifier U2 which operates at a gain of 3 dB, and whose outputs are connected to channel FETs Q1 and Q2 and to SOLO FETs Q3 and Q4. The channel FETs route the stereo signal to the left and right PROGRAM summing buses, and the SOLO FETs drive the PFL/SOLO summing buses.

The stereo signal at the outputs of U2 is also fed to dual summing amplifiers U5. The purpose of these amplifiers is to derive the "mix-minus" signal to be routed to the module's TO HYBRID output. This is accomplished by taking the summed PROGRAM signal from the System Master Module (TO TELCO MIX BUS), which includes the source audio from this module, and mixing in the module's source audio, out of phase. This effectively cancels the Telephone Module's audio from the rest of the PROGRAM signal being fed to the TO HYBRID output. The outputs of U2 feed left and right NULL potentiometers R30 and R33 through a time delay compensation network consisting of U28, C23, R31, and C24. The NULL LEFT and NULL RIGHT controls are adjusted by feeding a 1 kHz sine wave to the module input while the module is on and monitoring the TO HYBRID output for minimum 1 kHz level. Audio FETs Q7 and Q8 follow channel FETs Q1 and Q2 and are on when the module is on.

The outputs of U5 are summed by inverting summing amplifier U4B, which is operating at -2 dB of gain (attenuation), but the summing of the left and right PROGRAM signals gives an effective 6 dB increase in level, so the nominal output level of U4B is -6 dBu. U4A and the surrounding circuitry serves as a unity gain third-order hi-pass filter, with a crossover frequency of 300 Hz, while U3A is a 3 kHz low-pass filter, also third order and unity gain. The output of U3A serves as the low TO HYBRID output at a level of -6 dBu. This output also drives unity gain inverting amplifier U3B, the output of which serves as the high TO HYBRID output. The balanced output level of the TO HYBRID output is a nominal 0 dBu.

The output of input amplifier A1 feeds one contact of DIRECT PRE/POST switch S3, while the other feed to S3 comes from the output of fader buffer U1A. This switch allows the DIRECT output to be derived from either pre-fader or post-fader audio. The output of S3 feeds DIRECT FOLLOWS ON DIP switch S5. When S5 is open, audio FET Q20, which is turned on when the module is on, will determine whether or not audio will be present at the DIRECT output. When S5 is closed, the DIRECT output will be on, independent of the module status. The DIRECT output is fed from the output of non-inverting amplifier U1B, which operates with a gain of 10 dB, causing the nominal DIRECT output level to be 0 dBu.

LOGIC

The selection of the A or B input to the module controls LOGIC COMMON pin 1 on the remote logic connector. The setting of DIP switches S5-3 and S5-4 will determine whether or not pin 1 will be grounded. If for example, S5-3 is closed (Logic = A) and S5-4 is open, when the A/B selector is switched to the "A" position, pin 1 will be grounded, thereby enabling the remote functions. When the A/B selector is in the "B" position, pin 1 will not be grounded, and the remote functions will be disabled because any switches, lamps, relay interfaces, etc., will have no current return path to the logic connector.

When the module is off, depressing the ON/OFF button on the module, or the ON button on the selected input's remote control panel, will cause bi-stable U18B pin 12 to go low, illuminating the on tally lamp in the ON/OFF button by turning on Q16. Regulator U14 is configured as a current limiting device to protect the transistor against short circuits. The remote ON TALLY is also turned on by transistor Q17, through short circuit protector U15. The low at U18B is inverted by U6F, the output of which turns on the PROGRAM audio FETs by saturating Q9, and the DIRECT output FET by saturating Q21.

When the module is on, depressing the ON/OFF button on the module, or the OFF button on the selected input's remote control panel, will toggle U18B pin 12 high. This high is inverted by U10C to illuminate the off tally lamp in the ON/OFF button, via Q18 and U16. The remote OFF TALLY is turned on via Q19 and U17. The high at U18B pin 12 forces the output of U6F low, which turns off Q9 and Q21 and, subsequently, Q1, Q2, Q7, Q8, and Q20. The ON/OFF bi-stable may be reset to off by pulling the selected remote AUDIO RESET line low. This produces a high at pin 10 of U18B, via U10B, U9E, U12C, and U9F.

Operation of the A/B input selector button places a momentary low pulse at the input of U8D, which will key the channel logic to the "off" state to prevent any input switching transient from being routed into the console mix buses.

The module's OFF lamp may be remotely controlled by applying a low to the selected remote READY input. This is inverted by U6E and routed to U10C pin 5. The output of U10C will go low and illuminate the off tally if the module ON/OFF bi-stable is off and the READY input is low. This provides the facility of using the module's OFF lamp to tally the ready state of remote controlled equipment. If there is no connection to this logic function for a particular input, the OFF lamp will operate normally by closing DIP switch S5-1 for the A input or S5-2 for the B input.

Pressing the PFL button causes a high to appear at the output of U6B, toggling the bi-stable U7B pin 13 high, which turns on the PFL audio FETs by saturating Q10. The low appearing at U7B pin 12 turns on the PFL tally, via Q15 and U13, and forces the output of U8A high, pulling the PFL/SOLO ON command bus low via Q13. The low signal caused by the depression of the PFL button is inverted and routed by U12A to reset the SOLO bi-stable, if the SOLO function was engaged.

Pressing the SOLO button causes a high at the output of U6A, which toggles bi-stable U7A pin 1 high, turning on the SOLO audio FETs via Q11. When the SOLO function is engaged, the low at U7A pin 2 illuminates the SOLO tally via Q12 and Q20. When the SOLO function is engaged, the low at U7A pin 2 causes a high pulse at the output of U12B, which will reset PFL bi-stable U7B if the PFL function was engaged.

The PFL and SOLO buttons also drive U10A, whose output drives the PFL/SOLO CLEAR bus via Q14. This function is used to reset the PFL or SOLO functions on any other modules. When a low appears on the PFL/SOLO CLEAR bus, the output of U8B goes high and triggers monostable U11A, after a brief delay caused by R97 and C61. The low pulse appearing at U11 pin 9 is inverted and routed via U8C, U9D, U12A, and U12B to both the PFL and SOLO bi-stables in order to reset them. The reset pulse on a given module is disabled whenever the PFL/SOLO CLEAR bus command has originated from that module, in that the PFL and SOLO buttons drive the two inputs of U10A, whose output is inverted by U9C, which drives pin 13 of U11A. If either the PFL or SOLO button is depressed, U11A pin 13 will be driven high at the same time that the PFL/SOLO CLEAR bus is pulled low. This prevents the reset pulse on the monostable from resetting the CUE or SOLO bi-stables.

The PFL and SOLO functions are also reset whenever the module is turned on. When channel ON/OFF bi-stable U18B is set, monostable U11B is triggered, which drives the output of U8B high for 20 milliseconds. This, in turn, triggers monostable U11A, which delivers a low pulse to U8C, which resets the PFL and SOLO bi-stables via U9D, U12A, and U12B.

4.8 MONAURAL EQUALIZER MODULE

4.8.1 Function

The Monaural Equalizer Module provides both equalization and filter facilities in one compact module. This module is inserted into a Microphone Input Module's patch point by direct wiring, or through an external system patch field. The tunable filter section consists of high-pass and low-pass filters. The equalizer section, which may be switched in and out independently of the filter section, contains bass and treble equalizers, which are each independently switchable from peaking to shelving modes. The midrange equalizer is adjustable both with regard to the amount and to the frequency of equalization, and covers a mid-band frequency range of approximately 1500 Hz to 9 kHz.

4.8.2 Circuitry

A balanced differential amplifier (U1) is used for the input to the equalizer module. This amplifier operates at unity gain, and is intended for operation with signals at the console interstage nominal level of -10 dBu.

Capacitors C13, C14 and C16, along with the resistors R11, R13 and R15, are the frequency-determining components for the high-pass filter using U2 as the active element. The resistors are in series with the three-section potentiometer used for tuning. The remaining components around this stage are to keep the bandpass shape constant at the various cutoff frequencies. In its bandpass, this stage has an essentially flat amplitude response and unity gain. Capacitors C67, C68 and C69, operating with resistors R67, R68 and R69 are the frequency determining components for the low-pass filter, which uses U9 as the active element. The resistors are in series with the three section potentiometer used for tuning.

The bass and treble equalizers have been combined into a single stage. The signal is applied to R23, which is connected to the non-inverting input of amplifier U3A. Feedback is applied to the inverting input of the amplifier via R26.

Between R23 and R26 is a potentiometer (HIGH) whose wiper is connected to ground via capacitor C42. When the wiper is at the R23 end of the pot, the high frequency components of the signal are rolled off by the action of C42 to ground. When the wiper is at the R26 end of the pot, C42 reduces the feedback at high frequencies causing the gain at those frequencies to increase. R40 stops the bypass at a high frequency and, therefore, forms a "shelf" in the treble amplitude response. For the peak response characteristic, capacitor C42 is resonated by a synthesized inductor formed by U5 and its associated circuitry. When this inductor is entered into the circuit by operating switch S4, the series combination of C42 and the inductor form a resonant circuit, and the boost and cut are at the resonant frequency.

Also connected between R23 and R26 is a second potentiometer (LOW) whose wiper is connected to ground via a synthesized inductor formed by the circuitry around U4. When the wiper is at the R23 end of the pot, the pass components of the PROGRAM signal are rolled off by the action of the inductor bypassing the low frequencies to ground. When the wiper is at the R26 end of the pot, the inductor reduces the feedback at low frequencies, causing the gain at those frequencies to increase. The inductor is effectively in series with R32, which forms a shelf in the bass response. At each end of the bass potentiometer are capacitors which are shorted out in the shelving mode. Those capacitors are unshorted in the peaking mode (by the action of the LOW PK switch), in which case they work with the inductor to form a resonant circuit. The capacitor used in the low frequency cut mode may be shunted by internal option switch S5 to produce the "forced-shelf-on-cut" feature of the equalizer.

As with the bass and treble equalizer, the midrange equalizer is of unity gain. It is based on the circuitry around U3B. The wiper of the boost/cut control is connected to a relatively complex resonant circuit using U6 and both halves of U7. The circuitry is configured in a manner such that at the boost end of the amplitude control the input signal is bandpassed and applied to summer U3B, causing an increase in the overall amplitude response. At the cut end of the control the bandpass signal is applied as negative feedback, causing a dip in the amplitude response.

4.9 STEREO EQUALIZER MODULE

4.9.1 Function

The Stereo Equalizer Module consists of two identical sets of circuitry under the control of dual channel potentiometers and switches. It contains two separate, but coupled, three band equalizers. High and low frequency equalization is switchable between peaking and shelving modes. Midrange equalization is adjustable both with regard to the amount and to the frequency of equalization, and covers a mid-band frequency range of approximately 1500 Hz to 9 kHz.

This module may be inserted into the System Master Module's patch point for stereo equalization of the PROGRAM output.

4.9.2 Circuitry

NOTE: The stereo audio path consist of two identical sets of circuitry. For clarity and simplicity, only the left channel is described below.

A balanced differential amplifier (U1) is used for the input to the equalizer module. This amplifier operates at unity gain, and is intended for operation with input signals at the console interstage nominal level of -10 dBu.

The bass and treble equalizers have been combined into a single stage. The signal is applied to R12, which is connected to the non-inverting input of amplifier U2A. Feedback is applied to the inverting input of the amplifier via R16. Between R12 and R16 is a potentiometer (HIGH) whose wiper is connected to ground via capacitor C42. When the wiper is at the R12 end of the pot, the high frequency components of the signal are rolled off by the action of C42 bypass to ground. When the wiper is at the R16 end of the pot, C42 reduces the feedback at high frequencies, causing the gain at those frequencies to increase. R40 stops the bypass at a high frequency and, therefore, forms a "shelf" in the treble amplitude response. For the peak response characteristic, capacitor C42 is resonated by a synthesized inductor formed by U4 and its associated circuitry. When this inductor is entered into the circuit by operating switch S3, the series combination of C42 and the inductor form a resonant circuit so that the boost and cut are at the resonant frequency.

Also connected between R12 and R16 is a second potentiometer (LOW), whose wiper is connected to ground via a synthesized inductor formed by the circuitry around U3. When the wiper is at the R12 end of the pot, the low frequency components of the signal are rolled off by the action of the inductor bypass to ground. When the wiper is at the R16 end of the pot, the inductor reduces the feedback at low frequencies, causing the gain at those frequencies to increase. The inductor is effectively in series with R32, which forms a "shelf" in the bass response. At each end of the bass potentiometer are capacitors which are shorted out in the shelving mode. Those capacitors are unshorted in the peaking mode by the action of the LOW PK switch, in which case they form a resonant circuit with the inductor. The capacitor used in the low frequency cut mode may be shunted by internal option switch S2 to produce the "forced-shelf-on-cut" feature of the equalizer.

<u>NOTE</u>: When producing the "forced-shelf-on-cut" feature on the left channel (S2), insure that the right channel (S102) is set identically in order to prevent differences in low frequency amplitude and phase response between the two channels.

The midrange equalizer is a resonant type. This equalizer is adjustable both with regard to the degree and to the frequency of equalization. As with the bass and treble equalizers, the midrange equalizer is of unity gain, and is based on the circuitry around U2B. The wiper of the boost/cut control is connected to a relatively complex resonant circuit using U5 and both halves of U6. This resonant circuit is tunable from about 1500 Hz to about 9 kHz. The circuitry is configured in a manner such that at the boost end of the amplitude control the input signal is bandpassed and applied to summer U2B, causing an increase in the overall amplitude response. At the cut end of the control, the bandpass signal is applied as negative feedback, causing a dip in the amplitude response.

4.10 VOICE PROCESSOR MODULE

4.10.1 Function

The Voice Processor Module contains two major function sections. The first section is a switch-insertable equalizer, which covers the frequency range normally required for voice signal correction and/ or enhancement. The second section contains the expander, compresser, and de-esser systems. The expander threshold and attenuation are used to achieve noise reduction during pauses in speech, while the compressor provides signal "smoothing and density". The de-esser senses and operates only on the treble region, and provides adjustable control over excessive sibilance without the undesirable side effects of broadband designs. As with the Monaural Equalizer, this module is normally inserted into the patch point of a Microphone Module.

4.10.2 Circuitry

A balanced differential amplifier U1A is used for the input to the module. This amplifier operates at unity gain and is intended for operation with signals at the console interstage nominal level of -10 dBu.

The bass equalizer is formed by the circuitry around amplifier U2A and synthesized inductor U2B. This circuit operates at unity gain when the wiper of bass control R201 is set for flat response, mid-rotation. Rotating the control counterclockwise attenuates the bass components of the PROGRAM material into the input of U2A. Clockwise rotation attenuates the bass components of the feedback signal, increasing the gain of U2A at those frequencies. The inductor circuitry of U2B resonates with capacitor C12 in the boost mode, producing a peaking type response. The treble equalizer is also of unity gain, and is configured using the circuitry around U3A , U3B, U4A and U4B. The wiper of TREBLE control R204 is connected to a resonant circuit using U3B and both halves of dual amplifier U4. This resonant circuit is tunable from 3 kHz to 9 kHz using the two sections of frequency control R202/203. The entire bass and treble equalizer section is switched in or bypassed using EQUALIZER switch S1.

The output of the equalizer switch is routed to the expander/compressor section. This circuit has been designed specifically for producing a smooth, dense sound. The basic control element in the compressor is an integrated voltage-controlled amplifier (VCA) using U19 and buffer amplifier U5. The output of the VCA buffer is connected to the COMPRESSOR IN/OUT switch S3.

The VCA output signal is also routed to the compressor control system using U6A and U6B. This path includes frequency determining network R139, R52 and C40 to eliminate "pumping" or "ducking" due to strong bass content in PROGRAM material. The components around CR5, CR6, and CR7 are for the reduction of ripple on the AGC control bus. This technique contributes significantly to the excellent low frequency distortion performance of the compressor, while retaining an extremely fast recovery time.

The AGC voltage is applied to the precision OR circuit using U9B and U9C. The output of the OR is applied, along with a DC signal from the front panel COMPRESSION control R207, to U9D and then to the control port of the VCA U19. Trimmer R93 is used to null the distortion within the VCA circuit itself.

The audio signal which drives the input of the VCA is also routed to the expander side-chain control circuitry using U7 and U8. The output of this chain is also a DC signal which is applied to the precision OR circuit which controls the VCA. The sensitivity of the expander side-chain is determined by the setting of the front panel EXPANSION THRESHOLD control. The degree of gain reduction in the absence of audio is determined by the setting of the front panel EXPANSION ATTENUATION control.

The output of the OR circuit U9C and U9D, as applied to U9D, represents the compression portion of the gain control signal. It is metered by using display driver circuit U20 and front panel LED display DS-3. Trimmer R92 is the calibration control for the display. When the gain reduction voltage generated by the compressor circuitry is greater than the attenuation voltage from the expander circuitry, mode-switch comparator U9A operates switching transistor Q2. This changes the voltage on display driver U20 pin 9 so that the display is switched from a dot display (expansion) to a bar graph display (compression). In this manner, the same display is utilized to show both the expansion and compression modes.

The "O dB" LED is illuminated at all times. In a similar manner, the "18 dB" LED is dimly illuminated to provide a scale-length reference in darkened Control Rooms. The "18 dB" LED will be illuminated to a "normal" level when 18 dB of compression is used, and will be lit to a greater degree when 20 dB of compression is used. This is accomplished by connecting both the 18 dB and 20 dB driver outputs to the last LED segment. The display is calibrated by applying a low level signal (below the threshold of compression), setting the EXPANSION THRESHOLD control to minimum, setting 12 dB of gain reduction using the EXPANSION ATTENUATION control, and adjusting R92 to illuminate the "12" LED.

The audio output from the VCA is routed to the de-esser section. The audio path circuitry involves U10, U11, and U12. The output of VCA buffer amplifier U12A is applied to the control system using U14 and U15A. The sensitivity of this control loop is adjusted by front panel DE-ESS control R208. Treble content in the PROGRAM material above the threshold set by the DE-ESS control will produce a DC control voltage at the output U15A, which is applied to the control port of the VCA. The control voltage is monitored by the DE-ESS LED indicator, which is driven by U15B. The output of the de-essing circuitry is routed through internal level-match trimmer control R118 to the output buffer and to front panel IN/OUT switch S3. R105 is used to null the internal distortion of the de-esser VCA.

4.11 REMOTE LINE SELECTOR

4.11.1 Function

The Remote Line Selector provides for selection of up to six stereo input signals switched to one output. Typical applications include use as a line pre-selector ahead of an input module or the STEREOMIXER monitor system. The output of the selector is routed to a rear panel connector for user connection to the appropriate line or monitor input position.

4.11.2 Circuitry

This module consists of one six-station stereo selector switch fed from six stereo common remote inputs. The switch used is equipped with a mechanical lockout mechanism to prevent pushing two or more buttons simultaneously.

MAINTENANCE
AND ALIGNMENT

5.0 MAINTENANCE AND ALIGNMENT

This chapter contains sections describing routine maintenance, troubleshooting, level alignment, the tool kit, the spare parts kit, and replacement parts for the STEREOMIXER.

5.1 ROUTINE MAINTENANCE

Routine maintenance is usually limited to checking button switches for proper operation and keeping panel surfaces clean. The panel surfaces are finished with a baked polyurethane paint and may be cleaned with a weak solution of dishwashing detergent. The procedures for lamp replacement, button replacement, and collet knob removal are described below.

5.1.1 Lamp Replacement

NOTE: When replacing a lamp, replace it with an identical type. Do not substitute lamps of different voltage or current ratings.

The MONITOR, SEND TO TAPE, PFL and SOLO buttons are all EAO Series 19, and can be relamped by removing the lenscap by hand and using the rubber lamp removal tool (PR&E #70-23) to withdraw the miniature bi-pin lamp. Note the orientation of the square lenscap's tangs as it is removed. The lenscap must be reinstalled in the same orientation. Replace the 12 volt bi-pin lamp (PR&E #12-46), and carefully reinstall the lenscap.

The Module on/off control buttons are EAO series 31, and the lenscaps should only be removed using the tong-shaped lenscap removal tool (PR&E # 70-40) to grasp the top and bottom of the lenscap and pull it straight out of the button housing. Failure to use this tool will probably result in damage to the lenscap's molded retaining notches. The lamps should then be removed using the lamp removal tool (PR&E # 70-41). Replace the 14 volt lamp (PR&E #12-51), and carefully snap the lenscap back into the body of the switch.

5.1.2 Button Replacement

The square "winkeye" buttons (PR&E #25-769) can be removed by applying even pressure to the back of the button to dislodge it from the switch shaft. This can be accomplished using a two-prong device, such as a needlenose pliers. It may take considerable force to remove the button, so exercise caution. Carefully snap the replacement button onto the switch shaft.

5.1.3 Collet Knob Removal

The knobs used for the module's rotary controls are attached to the pot and switch shafts with collets instead of set screws. The machined brass collets provide the advantages of true alignment and concentricity with the axis of rotation, no set screws to score the shaft , the ability to clutch slip when excessive force is applied, and no holes in the side of the knob for the set screws. To remove a collet knob, carefully pry off the top cap of the knob using a thin blade or similar tool, then use the appropriate end of PR&E tool #70-44 to loosen the nut on the collet (as shown in Figure 5.1). Once the nut is loosened, the collet should release from the shaft. This nut should not need to be removed unless the collet refuses to release.

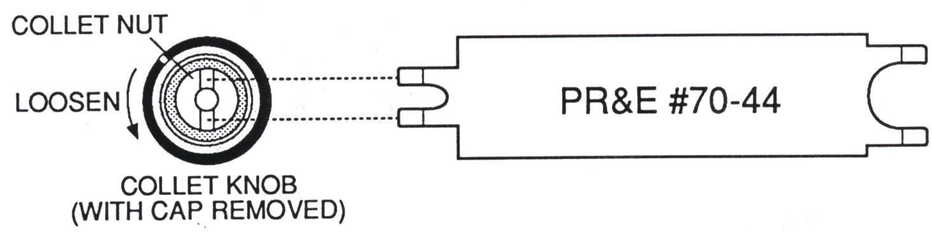


FIGURE 5.1 COLLET KNOB REMOVAL

5.2 TROUBLESHOOTING (MODULE REMOVAL)

The modular construction of the console greatly enhances troubleshooting since module substitution will usually isolate any problem.

To remove a module from the mainframe, remove the black button head retaining screws from the top and bottom of the module, and the silver button head screw from the module face. Screw the Module Pull/Extractor Tool (PR&E #70-43) into the hole vacated by the silver button head screw, and then pull the module straight out of the mainframe.

NOTE: An important feature when troubleshooting the STEREOMIXER is that modules may be removed or inserted with power supplied to the console.

Once the module at fault has been identified, it is recommended that the Equipment Description section for the module in question be read thoroughly prior to troubleshooting that module. Use the appropriate optional extender board (PR&E #99-441 for input and processor modules; PR&E #99-442 for the System Master Module), or standard bench service techniques to isolate the problem.

NOTE: Most of the analog and logic components are socketed for ease of replacement.

WARNING: The CMOS logic devices are susceptible to destruction from static discharge while being handled. It is recommended that considerable caution be exercised when working with these parts.

5.3 LEVEL ALIGNMENT

All consoles are set for +8 dBm output level operation unless another nominal output level reference is specifically requested. It is very important that the test levels are maintained exactly as specified to avoid a build-up of tolerance errors. The gain trim controls on the output amplifiers should not require trimming unless components which affect amplifier gain have been replaced.

NOTE: Always set the mixing fader at the 0 dB reference mark before adjusting the gain trim controls.

This will assure that the optimum gain structure (i.e. dynamic range, headroom and signal to noise ratio) has been maintained. Gain trim controls on the input modules should always be adjusted so that the input accommodates the signal level from the source equipment.

The STEREOMIXER amplifiers were level aligned at the time of factory test using the following procedures:

OUTPUT AMPLIFIERS

1 kHz test signal at -10 dBu applied to Master Module patch return; System Master Module internal gain trim controls adjusted to give +8 dBu output into 40K ohm bridging load. These controls can only be accessed with the use of an extender board for the System Master Module.

NOTE: The output amplifiers on the Tape Input/Output amplifiers have no adjustment because their level is determined by the level of the main output amplifiers. Once the system operating level has been set (on the System Master Module), the tape modules will operate at the same output level.

MICROPHONE INPUT MODULES

1 kHz test signal at -50 dBu to module input, fader set at 0 dB reference mark, module GAIN TRIM control adjusted to give +8 dBu console output into 40K ohm bridging load.

LINE INPUT MODULES

1 kHz test signal at +8 dBu to module input, fader set at 0 dB reference mark, internal attenuator switch "on", module GAIN TRIM control adjusted to give +8 dBu console output into 40K ohm bridging load.

TAPE INPUT/OUTPUT MODULES

Internal level DIP switch set in the "hi" position (it would be switched to the low position for a +4 dBm system operating level).

TELEPHONE INPUT/MIX-MINUS MODULES

1 kHz test signal at +8 dBu to module input, fader set at 0 dB reference mark, module GAIN TRIM control adjusted to give +8 dBu console output into 40K ohm bridging load.

VU METER AMPLIFIER CALIBRATION

1 kHz test signal at +8 dBu (unterminated) console output level, meter gain trim controls adjusted for 0 VU.

NOTE: These controls can only be accessed using the System Master Module extender board (PR&E #99-442).

SLATE, TALKBACK, AND AUXILIARY INPUT LEVELS

The slate tone level control is set for -6 VU.

The console talkback microphone gain control is set for a slate level of 0 VU using a voice speaking at a "normal" level. This level is important since it sets the microphone level for the talkback and voice slating functions, and should be readjusted as required upon completion of the installation.

The auxiliary input gain control is set fully counterclockwise.

NOTE: These controls can only be accessed using the System Master Module extender board (PR&E #99-442).

5.4 TOOL KIT

The following installation and servicing tool kit (PR&E #76-72) is provided with each console:

DESCRIPTION	PR&E#
MOLEX connector pin crimp tool	70-3
MOLEX connector pin extractor tool	70-4
MOLEX connector pin crimp tool (#HTR-1719-C)	70-5
EAO lamp removal tool for series 19 button switch	70-23
EAO nut wrench for series 19 button switch	70-38
EAO nut wrench for series 31 button switch	70-39
EAO lens removal tool for series 31 button switch	70-40
EAO lamp removal tool for series 31 button switch	70-41
PR&E module pull/extractor tool	70-43
Wrench for collet knobs	70-44

5.5 SPARE PARTS KIT

The following kit of spare parts (PR&E #76-51) is supplied with each STEREOMIXER mainframe at no charge. This initial kit is excluded from the limited warranty, and is only provided to support initial installation.

DESCRIPTION	QTY	PR&E#
Diodes		
1N4001	2	11-7
1N914	2	11-13
Integrated Circuits		
Dual Monostable, 4538	1	21-76
Dual D flip-flop, 4013	2	21-43
Hex Inverter, 4584	2	21-46
Op Amp, Dual, LF353	2	20-32
Op Amp, Dual, 8-pin, 5532	2	20-53

DESCRIPTION	QTY	PR&E#
Op Amp, DIP, 5534	2	20-28
Quad, 2-input, Nor, 4001	2	21-61
Quad, 2-input, Nand, 4011	2	21-45
Lamps		
Bi-pin, 12 V, 0.7 watt, 1099BPE	2	12-46
14 V, .08 amp, 386	2	12-51
Transistors		
FET, J-174	2	9-5
MJE-181	1	7-1
MJE-171	1	8-1
NPN, PN2222 or 2N2222	2	7-25
PNP, MPS-A63	2	8-4
Voltage Regulators		
Variable, 1.5A, Pos., LM317T	1	20-49
Variable, 1.5A, Neg., LM337T	1	20-59
Variable, LM350K Steel	1	20-30

This kit should provide sufficient support spares for the initial operating period, however, it is recommended that this kit be replenished and kept on hand for service use. The components used are, wherever possible, standard items of general availability. However, should difficulty be encountered locating any of these items, PR&E maintains a stock of replacement parts.

In applications where any system "down-time" is unacceptable, it is recommended that the following modules be kept on hand as spares:

DESCRIPTION	PR&E#
Microphone Input Module	99-427
Power Supply Assembly	99-422
Stereo Line Input Module	99-424
Stereo Tape Input/Output Module	99-430
System Master Module	99-421
Telephone Input/Mix-Minus Module	99-439

5.6 REPLACEMENT PARTS

Most of the components used are, wherever possible, standard items of general availability. However, should difficulty be encountered locating any of the items, PR&E maintains a stock of replacement parts. The power supply transformer, Schadow button switches, Penny & Giles rotary faders, engraved button caps, and VU meters are all manufactured or modified to custom design specifications, and are, therefore, available only from PR&E.

Following is a partial list of parts and assemblies used in the STEREOMIXER and its associated modules, and the PR&E part number for easy reference:

DESCRIPTION	PR&E#
Bridge Rectifier, 4A, 200V	11-38
Button Caps, "Winkeye"	
Square, Blk-Org	25-769
Square, Blk-Org, "IN"	25-769-8
Square, Blk-Org, "L"	25-769-9
Square, Blk-Org, "R"	25-769-10
Capacitors	
Ceramic Disc, .001µF, 1KV	62-1
Electrolytic, Low Leak, 1µF, 50V	60-45
Electrolytic, Low Leak, 10µF, 16V	60-48
Electrolytic, Low Leak, 10µF, 25V	60-67
Electrolytic, 10µF, 25V, NP	60-84
Electrolytic, 22µF, 16V	60-55
Electrolytic, 22µF, 25V	60-76
Electrolytic, 22µF, 25V, NP	60-85
Electrolytic, 100µF, 16V, NP	60-82
Electrolytic, 100µF, 25V	60-52
Electrolytic, 220µF, 16V	60-44
Electrolytic, 4700µF, 50V	60-94
Metalized Polyester, Stacked Film, .01µF, 400V	63-4
Metalized Polyester, Stacked Film, .015µF, 100V	63-1
Metalized Polyester, Stacked Film, .022µF, 250V	63-9
Metalized Polyester, Stacked Film, .047µF, 250V	63-6
Metalized Polyester, Stacked Film, .1µF, 100V	63-7
Metalized Polyester, .15µF, 100V, Radial	63-20
Metalized Polyester, Stacked Film, .47µF, 100V	63-10
Metalized Polyester, Stacked Film, .68µF, 100V	63-21
Metalized Polyester, Stacked Film, 1.5µF, 100V	63-19
Metalized Polyester, Stacked Film, 2.2µF, 100V	63-12
Metalized Polyester, Stacked Film, 6800pF, 400V	63-22
Monolythic, .01µF, 50V	62-4
Monolythic, .1µF, 50V	62-5
Polypropylene, 4700pF, 160V, Axial	64-23
Tantalum, .47µF, 35V	65-15
Tantalum, 10.0µF, 25V	65-5
Tantalum, 10µF, 20V	65-6
Caps	
Black (for S110 knob)	32-121
Black (for S210 knob)	32-141
Blue (for S110 knob)	32-123
Blue (for S210 knob)	32-143
Orange (for S150 knob)	32-136
Red (for S110 knob)	32-125
Red (for S210 knob)	32-139

DESCRIPTION	PR&E#
Yellow (for S110 knob)	32-126
Yellow (for S210 knob)	32-144
Connector Interface Panels	
Line Input	99-425
Microphone Input	99-428
Remote Line Selector	99-434
Tape Input/Output	99-431
Telephone Input/Output	99-440
Cover, Insulating, for TO-3, Black	23-20
Diodes	
1N4001	11-7
1N914B	11-13
Faders	
Rotary, Stereo, P&G	24-129
Rotary, Mono, P&G	24-130
Function Modules	
Instrumentation Input	95-123
Microphone Preamplifier	95-427
Output Amplifier	95-119-1
Fuse, Slo-Blo, 3AG, 2AMP, 250V	30-24
Fuse Holder	30-711
Headphone Jack, P.C. Mounted	17-54
Insulators	
For TO-3, Silpad	31-2
For TO-220AB, Silpad	31-3
Integrated Circuits	
Driver Display Dot/Bar 3914	20-65
Dual D Flip-Flop, 4013	21-43
Dual Monostable, 4538	21-76
Hex, Schmitt Inverter, 4584	21-46
Inverting Modulator 2151 DBX	20-63
Op Amp, DIP, 5534	20-28
Op Amp, Dual, LF353	20-32
Op Amp, Dual, 8-pin DIP, 5532	20-53
Quad, 2 -input Nand, Schmitt Trigger, 4093	21-4
Quad, 2-input, Nand, 4011	21-45
Quad, 2-input NOR, 4001	21-61
Triple, 3-input NOR, 4025	21-95
Wide Band Width Quad JFET Input Op Amp LF347N	20-64
Knobs	
11mm, 1/8" Shaft, Grey	32-120
15mm, 1/4" Shaft, Grey	32-129
21mm, 1/4" Shaft, Grey	32-138

DESCRIPTION	**PR&E#**
Lamps	
T1, Bi-Pin, 12V, 0.7W OL1099BPE 12V .7W	12-46
14V, .08A, Midget Groove	12-51
LED, Array, Red	12-59
LED, Red	12-53
Legend Strip, 1.02" Long	80-222-16
Lenscaps	
Square, Red, EAO Series 19	25-115
Square, Yellow, EAO Series 19	25-116
Square, White, EAO Series 19	25-118
Red and Yellow, EAO Series 41	25-151
Meter Assembly, SIFAM R-14	90-658
Microphone Assembly, 3-Pin Wafer	90-637
Modules	
Microphone Input Module	99-427
Monaural Equalizer Module	99-415
Power Supply Assembly	99-422
Stereo Equalizer Module	99-416
Stereo Line Input Module	99-424
System Master Module	99-421
Tape Input/Output Module	99-430
Telephone Input/Mix-Minus Module	99-439
Voice Processor Module	99-296
MOLEX Connector Kit	76-52
Power Cord	30-13
Potentiometers	
Concentric Triple, Linear	24-105
Concentric Dual	24-118
Dual, 10K, Linear	24-109
Quad, 10K, Log, CCW	24-110
Quad, 25K, Linear	24-99
Single, 1K, Audio Taper, CCW	24-18
Single, 10K, Linear	24-102
Single, 10K, Audio Taper, CCW	24-20
Single, 10K, Long, CCW	24-98
Triple, Log, CCW	24-103
Triple, Log, CCW	24-104
Screw, Binder Head, #6-32 X 1/4", Nylon	37-111
Shorting Plug Assembly, Male, 12-pin, 4 Jumper	
Socket, Transistor, TO-3, for .040 Dia. Pins	16-305
Spare Parts Kit	76-51
Switches	
1-Station, 4-Pole, Push-Push	25-775
1-Station, Pushbutton, Momentary, 4PDT	25-784

DESCRIPTION	PR&E#
1-Station, 2-Pole, Momentary	25-797
3-Station, 4-Pole, Interlock, Cut SL, LT Springs	25-802
6-Station, 4-Pole, Interlock/Lockout, LT Springs	25-803
SPDT, MOM, Dual Lamp, EAO Series 41	25-150
SPST, MOM, Square, EAO Series 19	25-114
Technical Manual	75-21
Tool Kit	76-72
Transformer, Power, Toroidal, 117V/60Hz	48-115
Transistors	
FET, J-174	9-5
MJE-171	8-1
MJE-181	7-1
NPN, MPS-6560	7-11
NPN, PN2222	7-25
PNP, MPS-A63	8-4
Transistor/Diode Assemblies	
Q-CR, MJE171 and 1N914B	90-119-1
CR, MJE181 and 1N914B	
Trim-Pots	
Single-Turn, Cermet, 1K ohm, 72XRIK	24-53
Single-Turn, 10K ohm, 72PR10K	24-54
Single-Turn, Cermet, 10K ohm, 72XR10K	24-57
Voltage Regulators	
Variable, 1.5 A, Pos., LM317T	20-49
Variable, 1.5 A, Neg., LM337T	20-59
Variable, LM350K Steel	20-30
Variable, 100 mA, Pos., TL317C	20-72

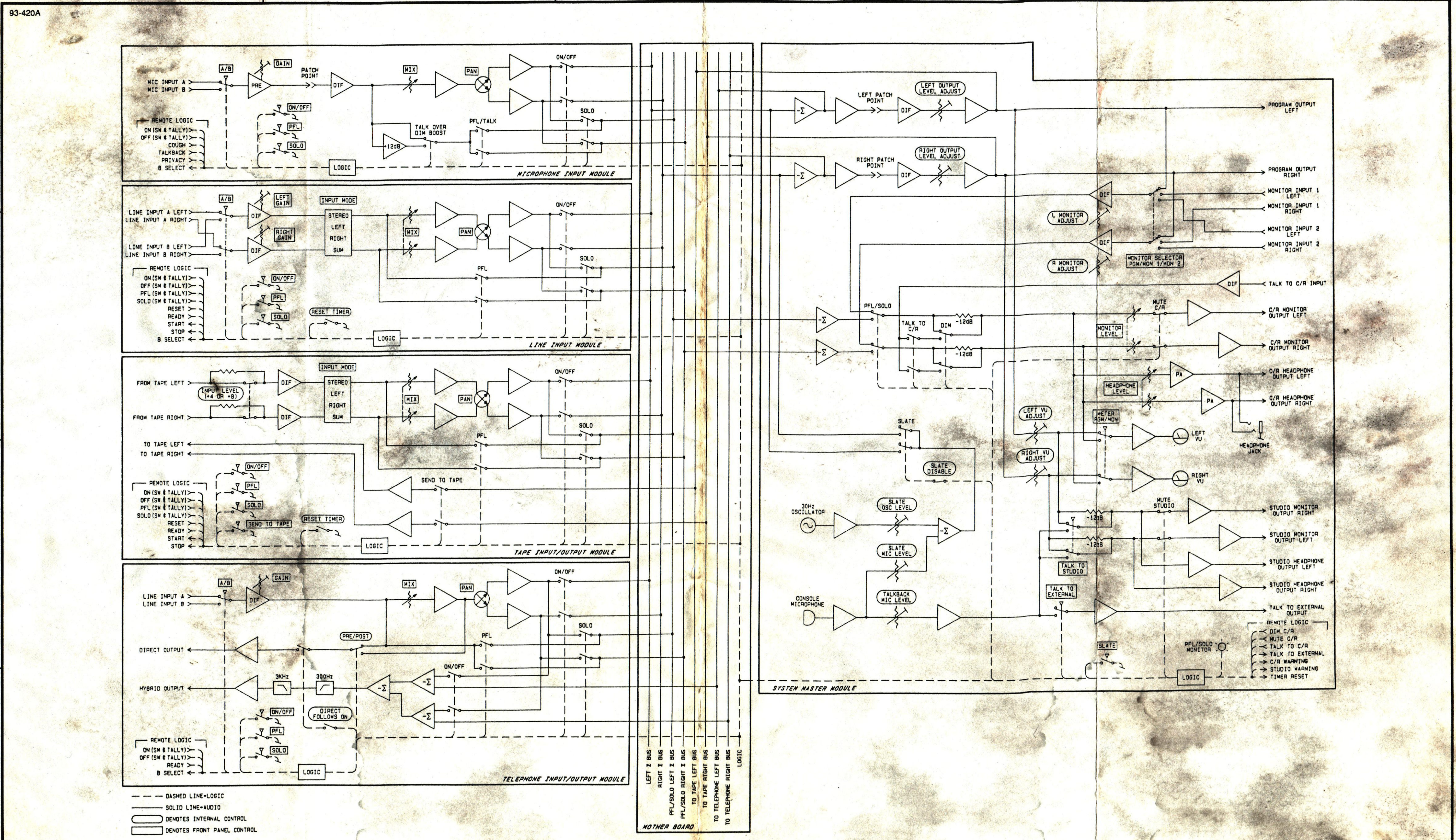
DRAWINGS AND SCHEMATICS

6.0 DRAWINGS AND SCHEMATICS

This chapter is made up of the following drawings and schematics:

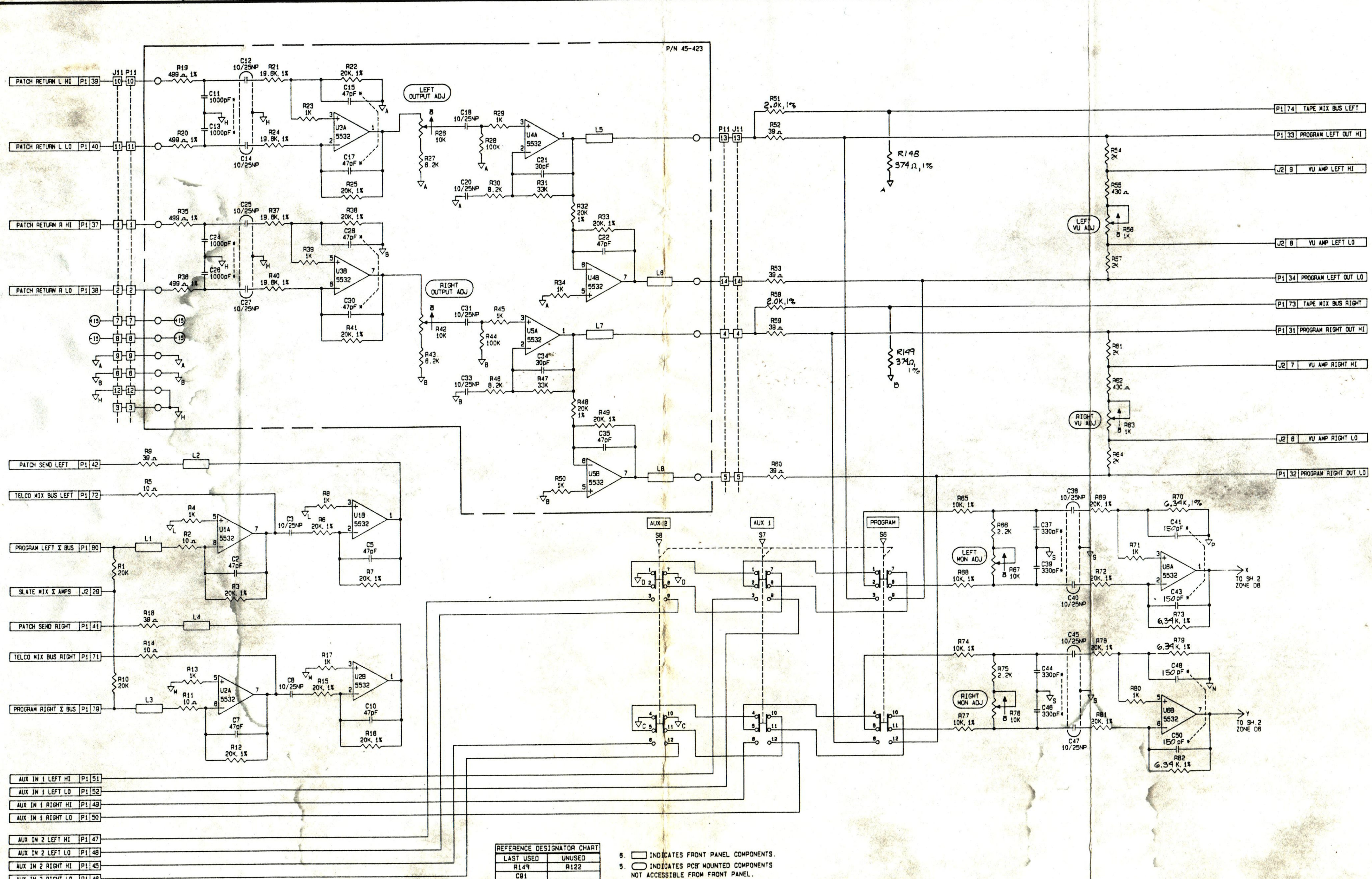
<u>NOTE</u>: On PR&E schematics, capacitor values are in microfarads (µF), unless otherwise specified, and resistors are 1/4 watt, 5%, unless otherwise specified.

93-420A



PACIFIC RECORDERS & ENGINEERING CORPORATION

6.1 FUNCTIONAL BLOCK DIAGRAM



REFERENCE DESIGNATOR CHART	
LAST USED	UNUSED
R149	R122
C81	
U18	
L14	
CR8	
Q15	
J13	J3-10
S8	S1-5

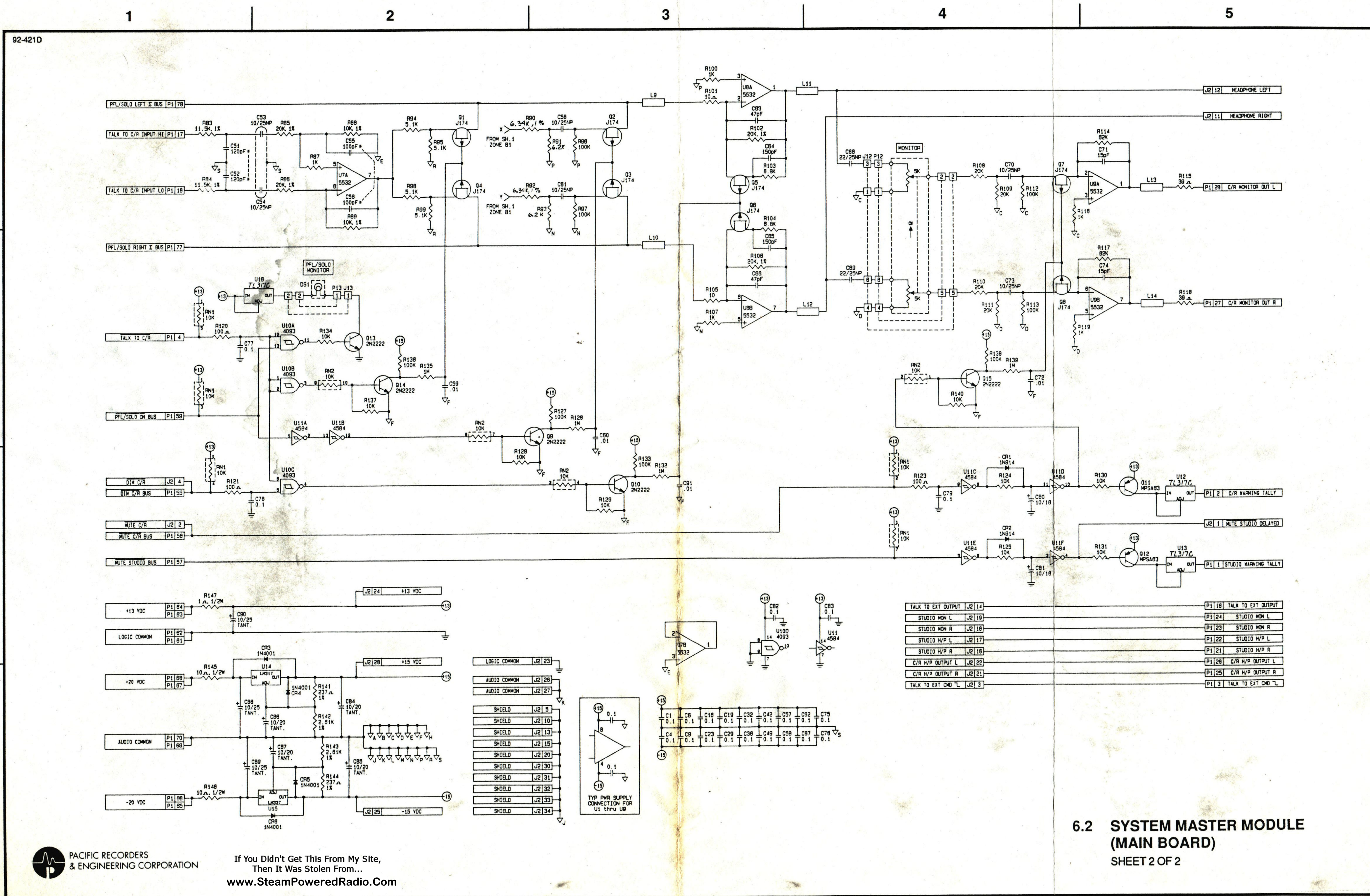
GROUND LEGEND	
⏚	AUDIO COMMON
⏚	LOGIC COMMON

6. ▭ INDICATES FRONT PANEL COMPONENTS.
5. ⬭ INDICATES PCB MOUNTED COMPONENTS NOT ACCESSIBLE FROM FRONT PANEL.
4. *---* INDICATES MATCHED PAIR.
3. NP=NONPOLARIZED.
2. ALL CAPACITORS ARE MEASURED IN MICROFARADS.
1. ALL RESISTORS ARE 1/4W, 5%.
NOTES: UNLESS OTHERWISE SPECIFIED.

6.2 SYSTEM MASTER MODULE (MAIN BOARD)

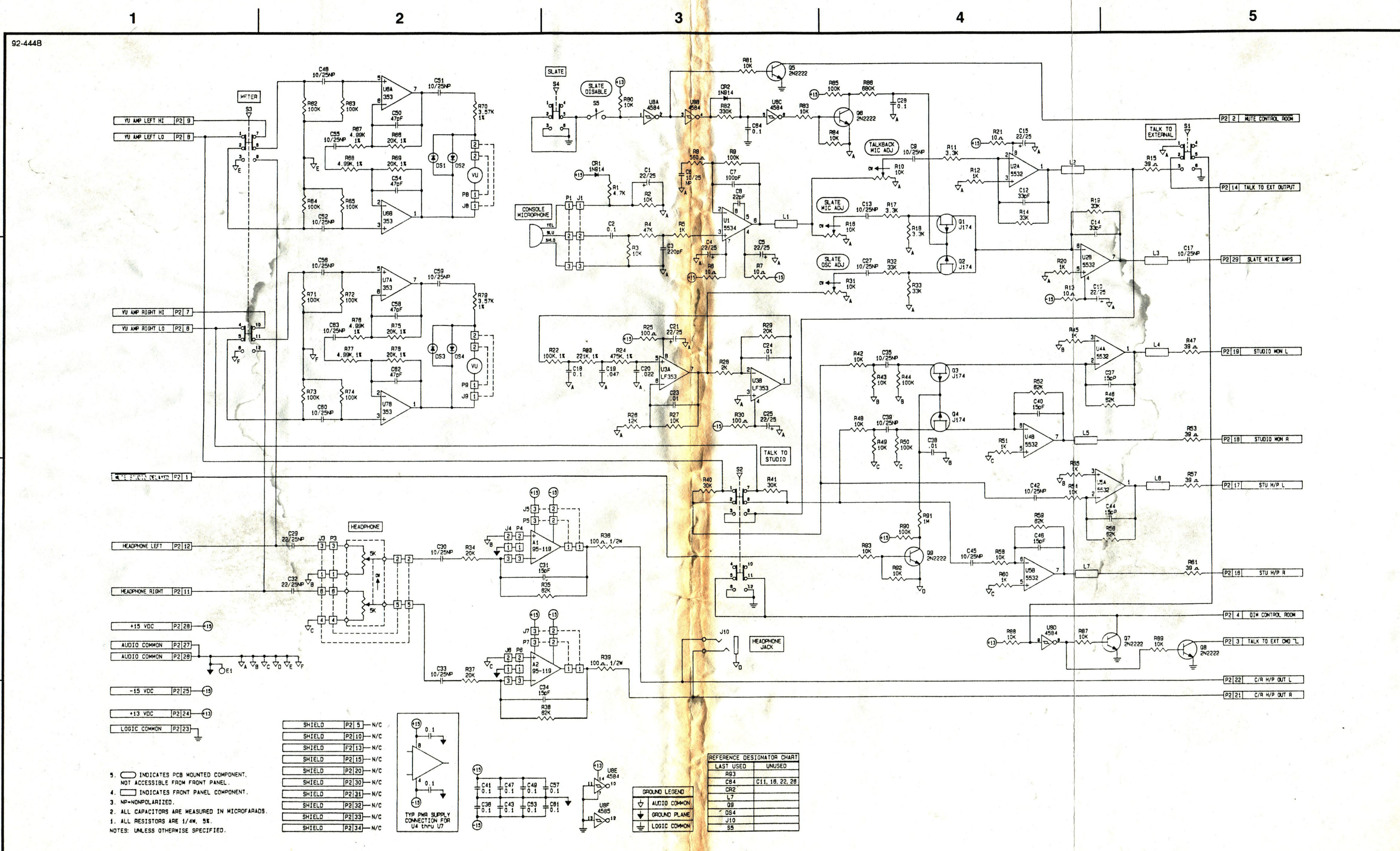
SHEET 1 OF 2

PACIFIC RECORDERS & ENGINEERING CORPORATION



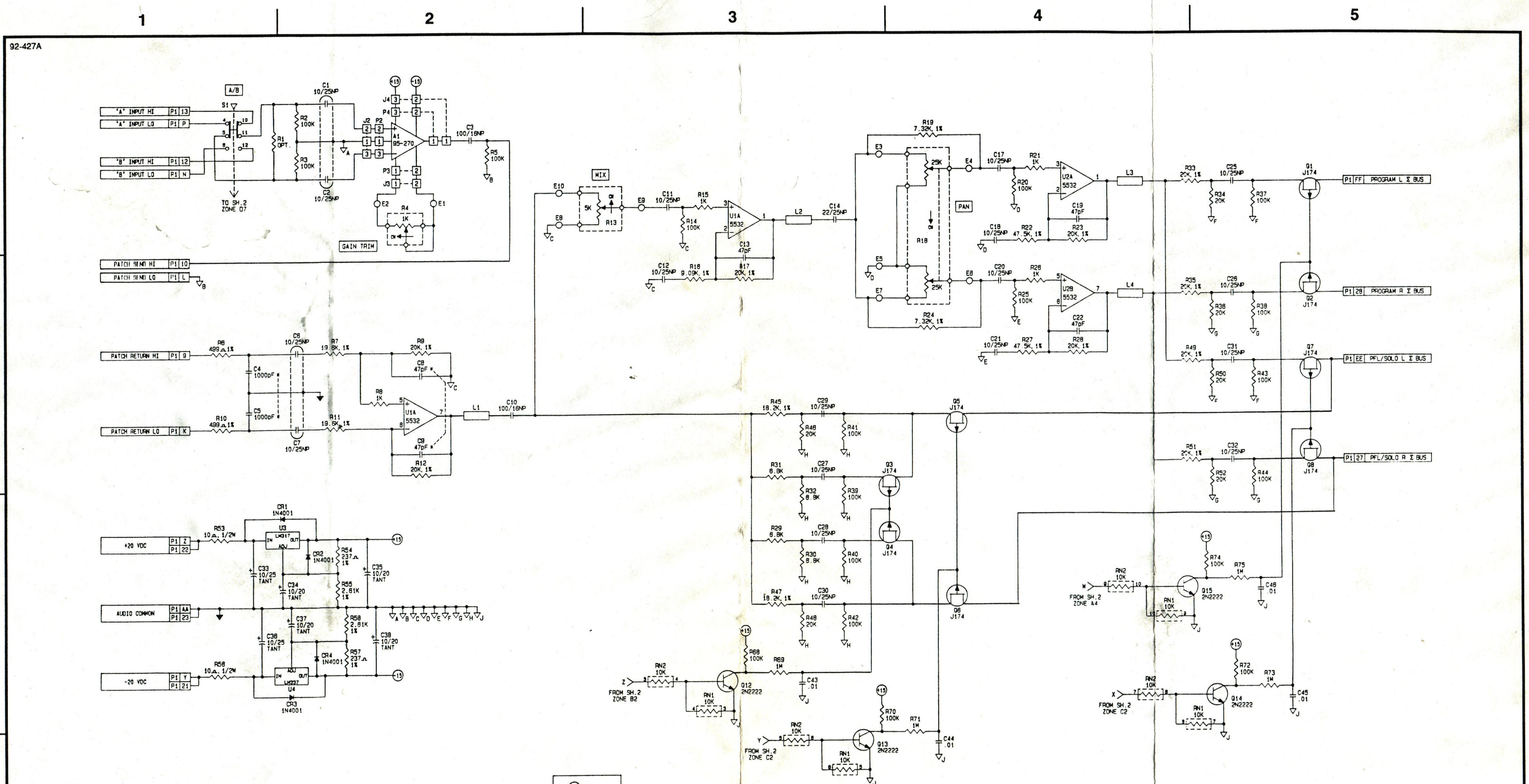
6.2 SYSTEM MASTER MODULE (MAIN BOARD)

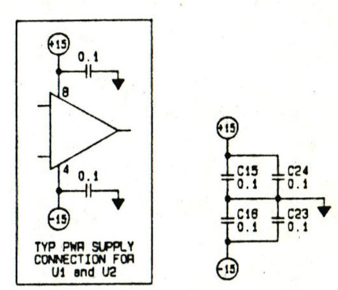
SHEET 2 OF 2



PACIFIC RECORDERS & ENGINEERING CORPORATION

6.3 SYSTEM MASTER MODULE (SATELLITE BOARD)





D

6. ▭ INDICATES FRONT PANEL COMPONENTS.
5. ⬭ INDICATES PCB MOUNTED COMPONENTS NOT ACCESSIBLE FROM FRONT PANEL.
4. *---* INDICATES MATCHED PAIR.
3. NP=NONPOLARIZED.
2. ALL CAPACITORS ARE MEASURED IN MICROFARADS.
1. ALL RESISTORS ARE 1/4W, 5%.
NOTES: UNLESS OTHERWISE SPECIFIED.

REFERENCE DESIGNATOR CHART	
LAST USED	UNUSED
R89	
C71	
U21	
CR16	
L4	
Q23	
S5	
RN5	
E20	
J4	

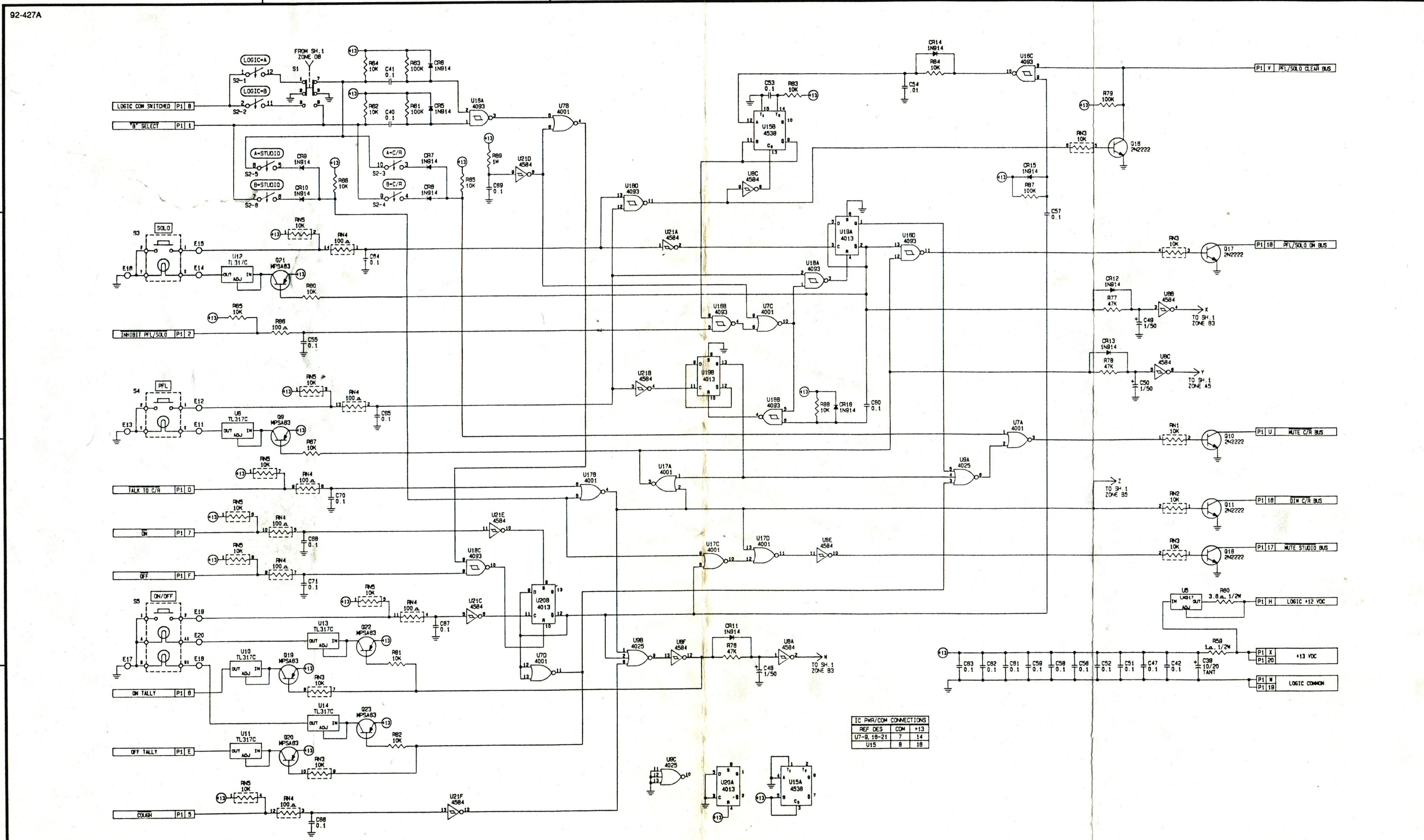
GROUND LEGEND	
▽	AUDIO COMMON
▼	GROUND PLANE
⏚	LOGIC COMMON
⏊	CHASSIS GND

PACIFIC RECORDERS
& ENGINEERING CORPORATION

6.4 MICROPHONE INPUT MODULE

SHEET 1 OF 2

92-427A



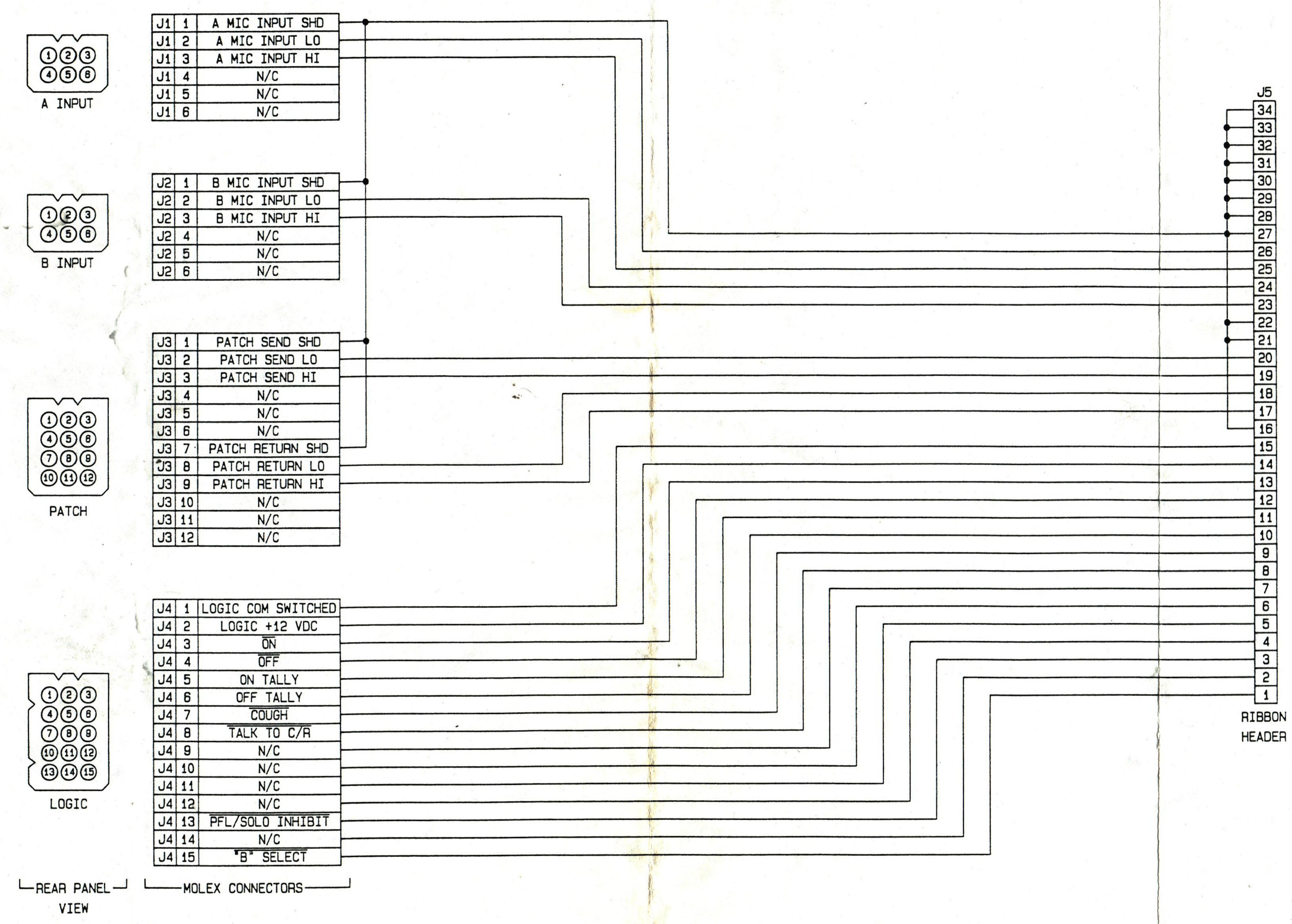
IC PWR/COM CONNECTIONS		
REF DES	COM	+13
U7-8, 18-21	7	14
U15	8	16

6.4 MICROPHONE INPUT MODULE

SHEET 2 OF 2

PACIFIC RECORDERS & ENGINEERING CORPORATION

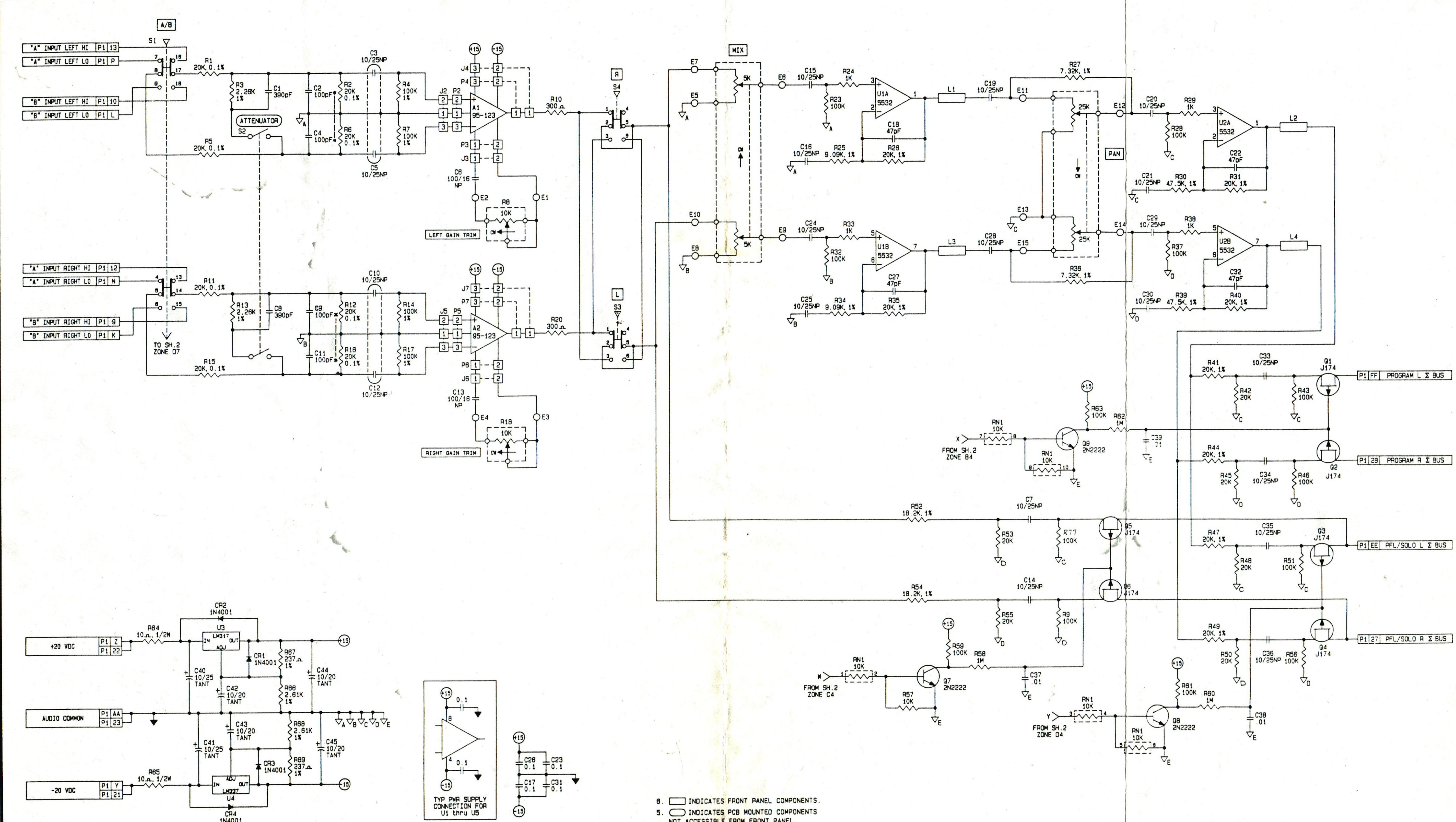
92-428A



PACIFIC RECORDERS & ENGINEERING CORPORATION

6.5 MICROPHONE INPUT CONNECTOR INTERFACE

92-424E



6. ▭ INDICATES FRONT PANEL COMPONENTS.
5. ⬭ INDICATES PCB MOUNTED COMPONENTS NOT ACCESSIBLE FROM FRONT PANEL.
4. *---* INDICATES MATCHED PAIR.
3. NP=NONPOLARIZED.
2. ALL CAPACITORS ARE MEASURED IN MICROFARADS.
1. ALL RESISTORS ARE 1/4W, 5%.
NOTES: UNLESS OTHERWISE SPECIFIED.

6.6 STEREO LINE INPUT MODULE

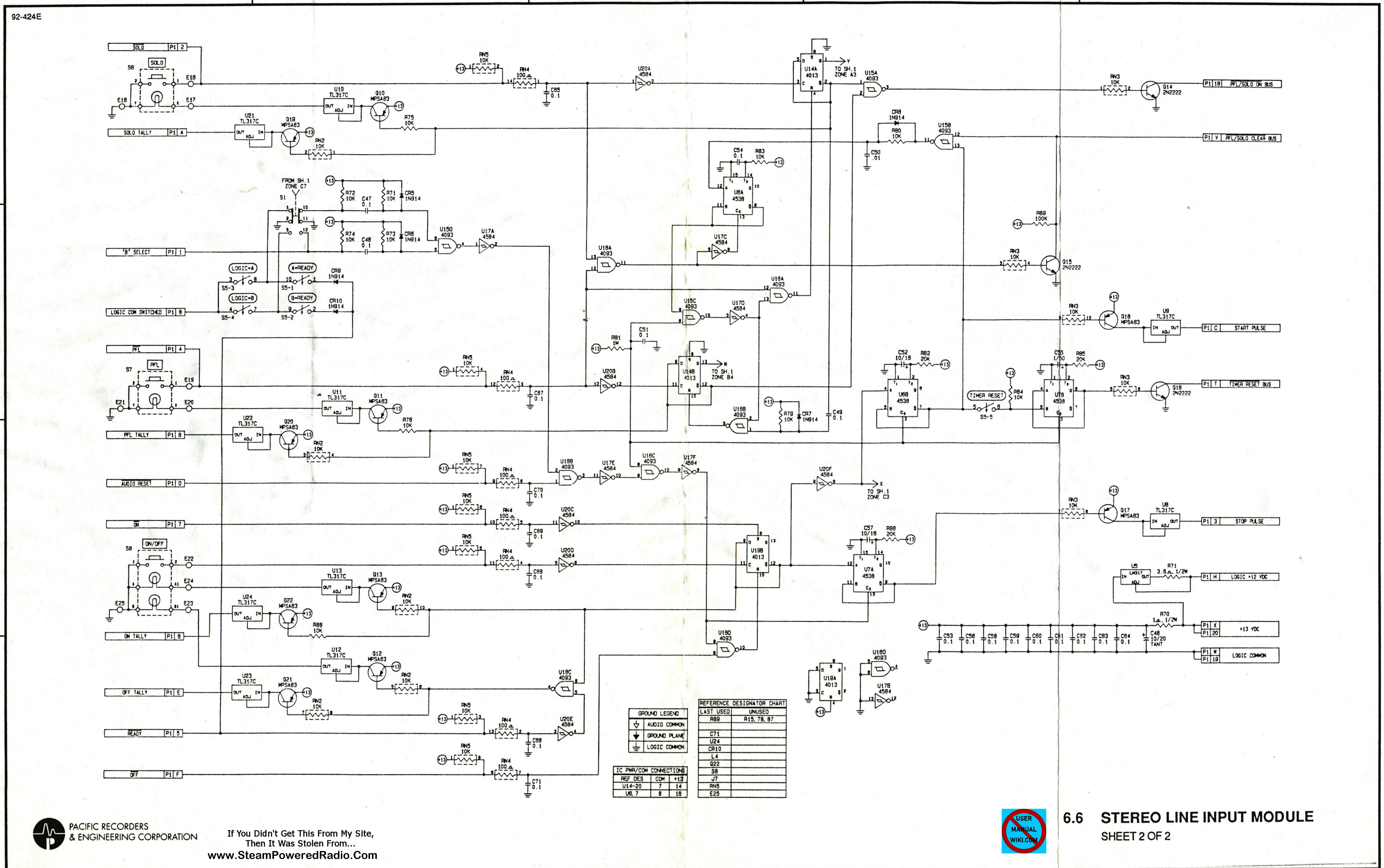
SHEET 1 OF 2

PACIFIC RECORDERS & ENGINEERING CORPORATION

1 2 3 4 5

A B C D

92-424E



GROUND LEGEND	
▽	AUDIO COMMON
▼	GROUND PLANE
⏚	LOGIC COMMON

IC PWR/COM CONNECTIONS		
REF DES	COM	+13
U14-20	7	14
U6, 7	8	16

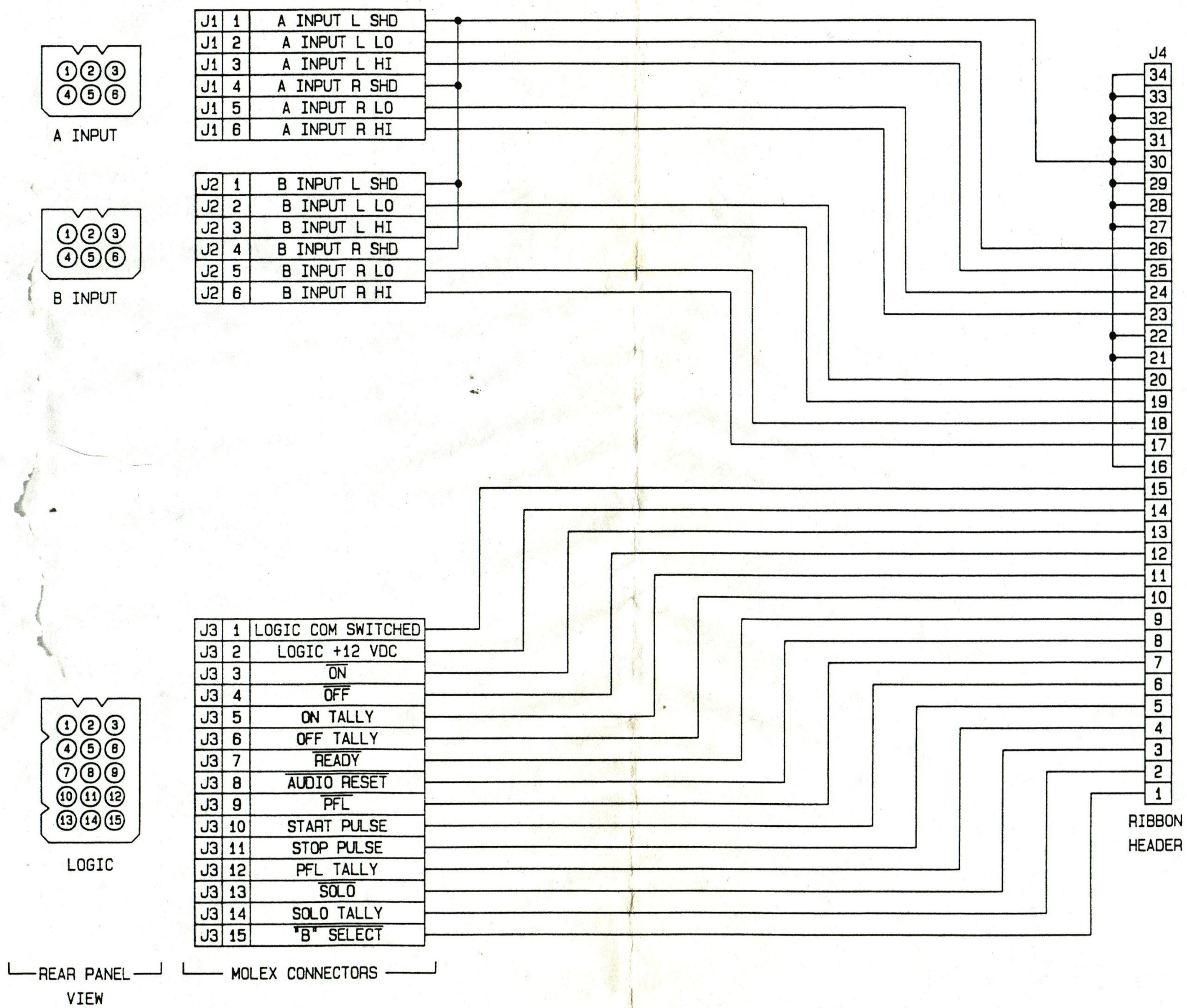
REFERENCE DESIGNATOR CHART	
LAST USED	UNUSED
R89	R15, 78, 87
C71	
U24	
CR10	
L4	
Q22	
S8	
J7	
RN5	
E25	

PACIFIC RECORDERS & ENGINEERING CORPORATION

6.6 STEREO LINE INPUT MODULE

SHEET 2 OF 2

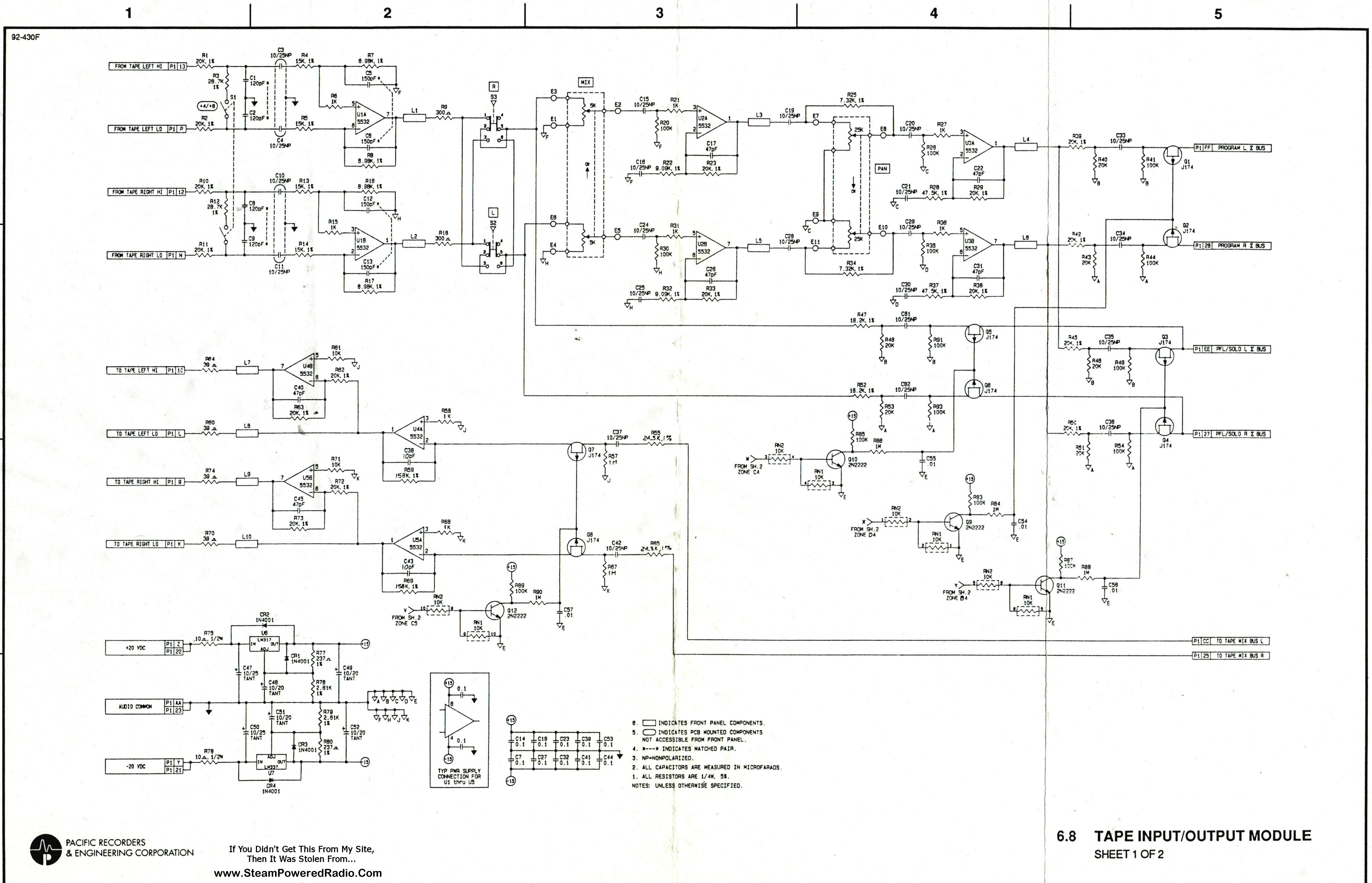
92-425A



PACIFIC RECORDERS & ENGINEERING CORPORATION

6.7 STEREO LINE INPUT CONNECTOR INTERFACE

92-430F

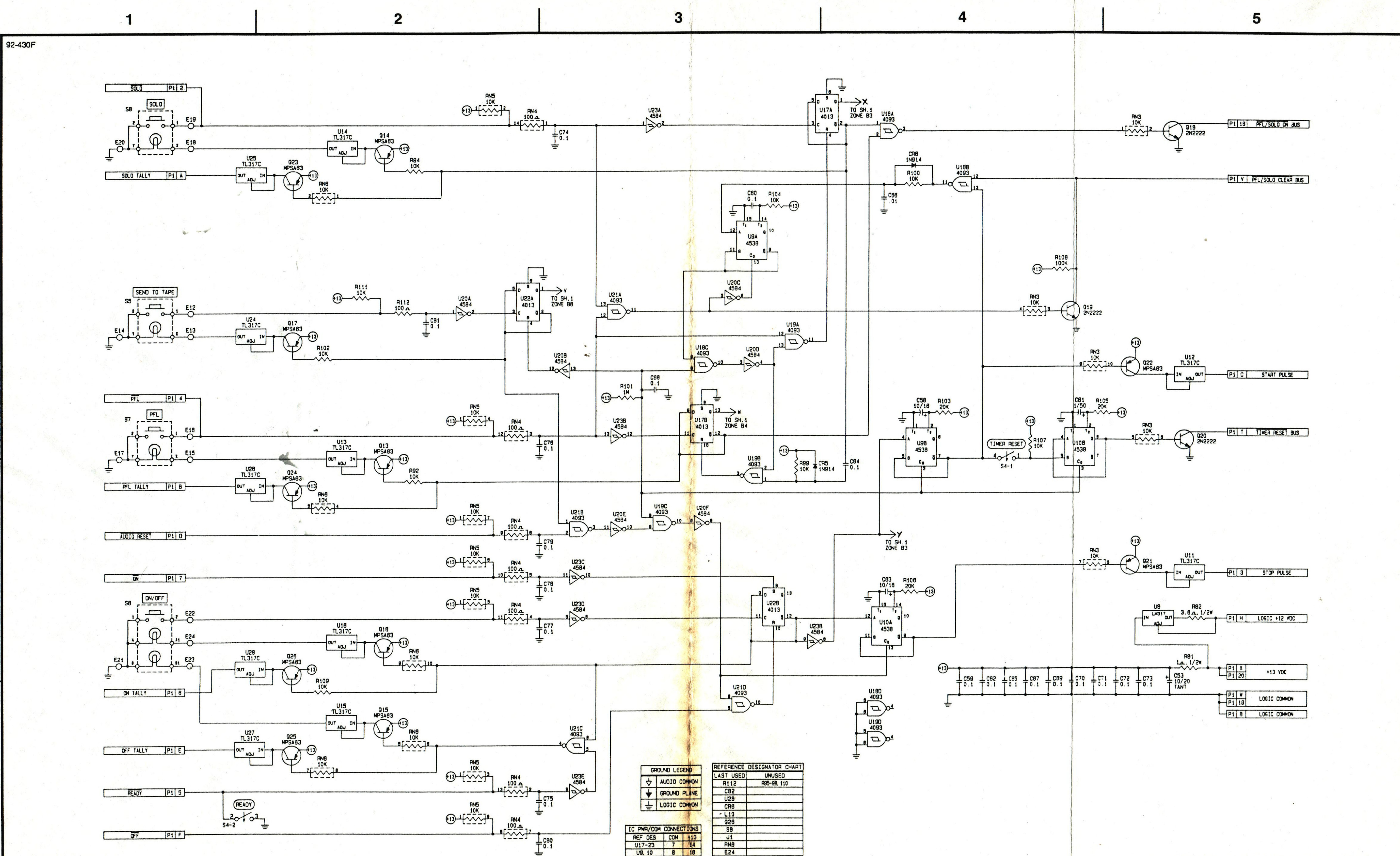


PACIFIC RECORDERS & ENGINEERING CORPORATION

6.8 TAPE INPUT/OUTPUT MODULE

SHEET 1 OF 2

92-430F



GROUND LEGEND	
⏚	AUDIO COMMON
⏚	GROUND PLANE
⏚	LOGIC COMMON

IC PWR/COM CONNECTIONS		
REF DES	COM	+13
U17-23	7	14
U8, 10	8	16

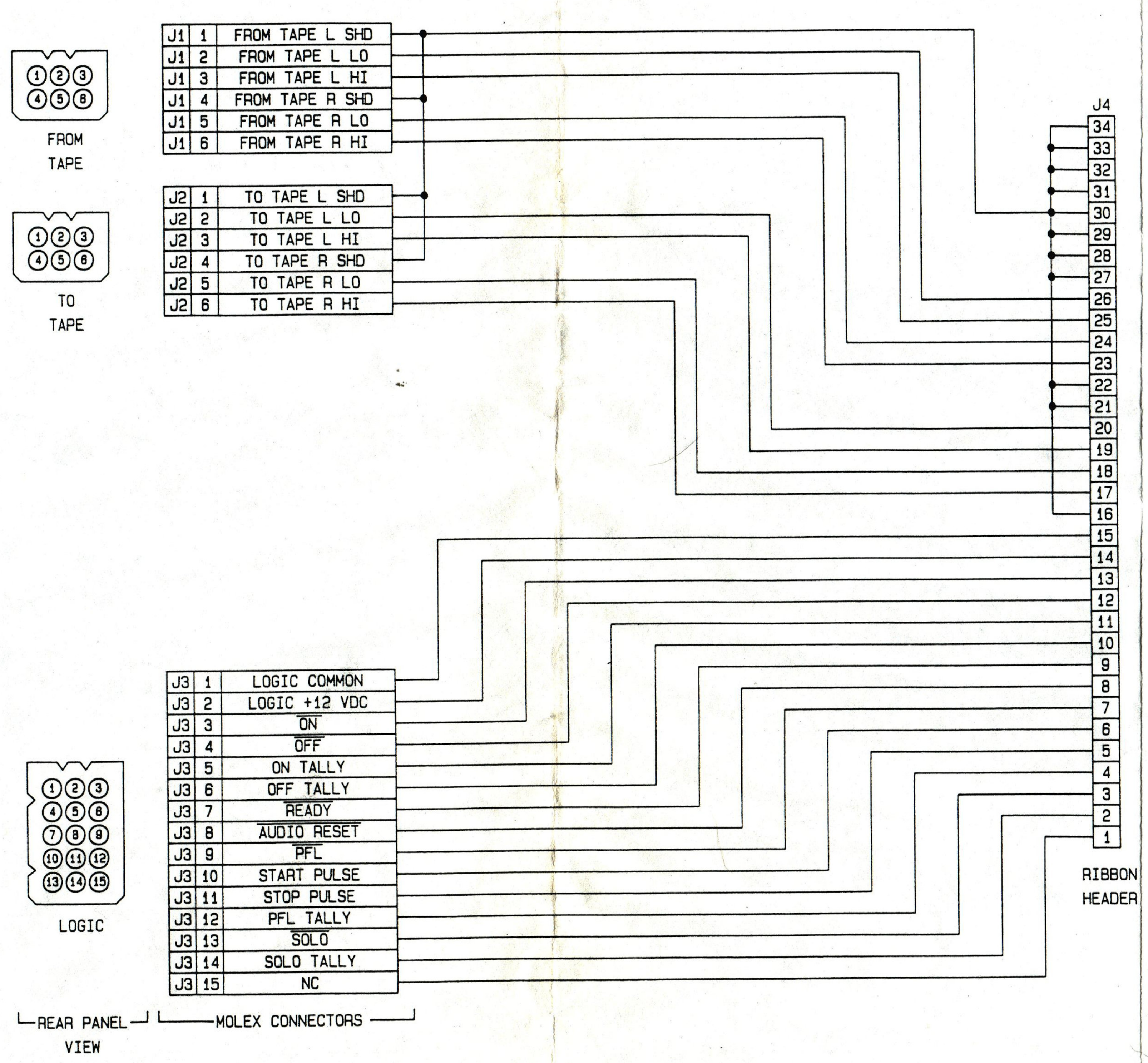
REFERENCE DESIGNATOR CHART	
LAST USED	UNUSED
R112	R95-98, 110
C82	
U28	
CR6	
L10	
Q26	
S8	
J1	
RN8	
E24	

PACIFIC RECORDERS & ENGINEERING CORPORATION

6.8 TAPE INPUT/OUTPUT MODULE

SHEET 2 OF 2

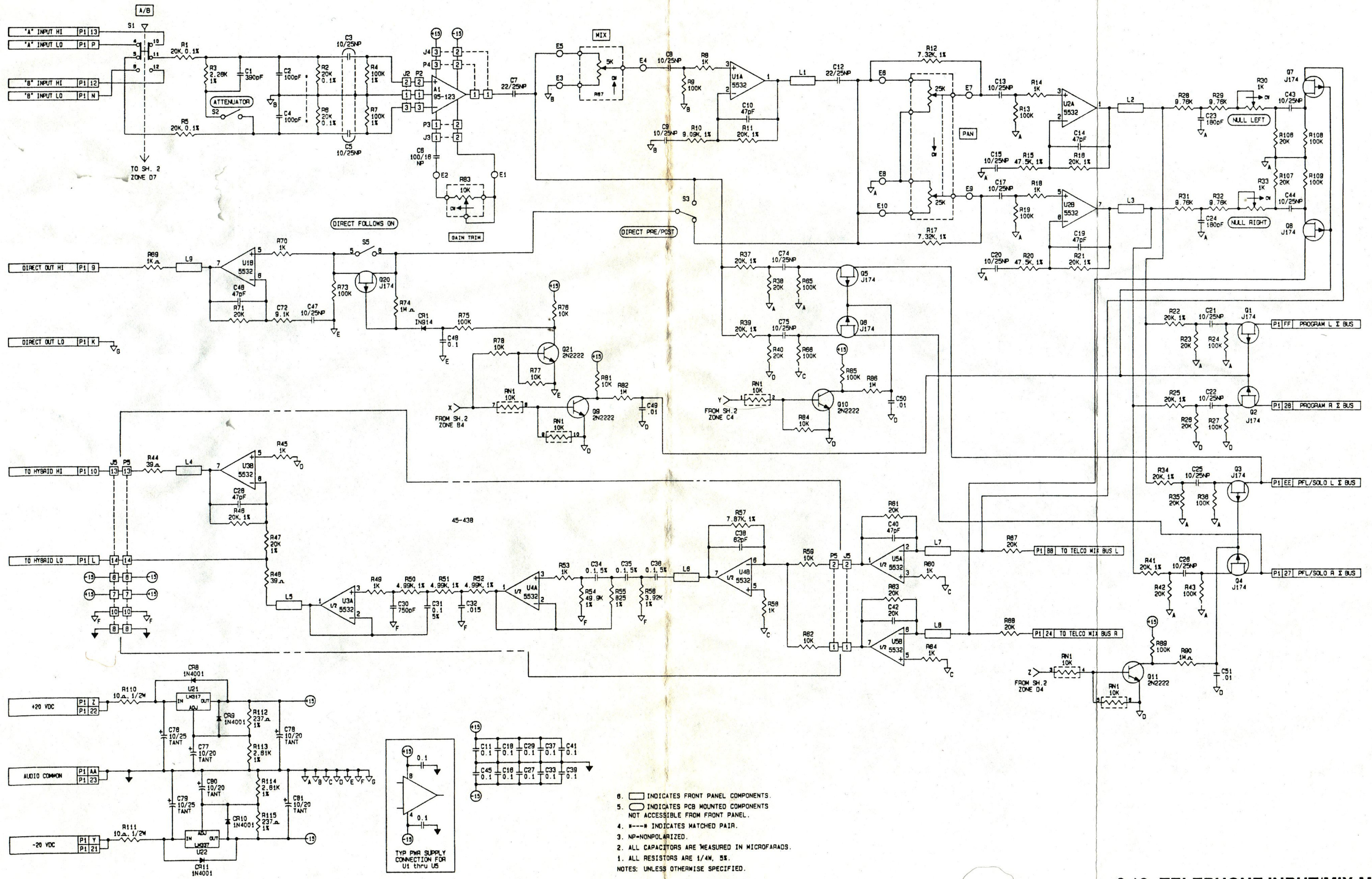
92-431A



PACIFIC RECORDERS & ENGINEERING CORPORATION

6.9 TAPE INPUT/OUTPUT CONNECTOR INTERFACE

92-439A

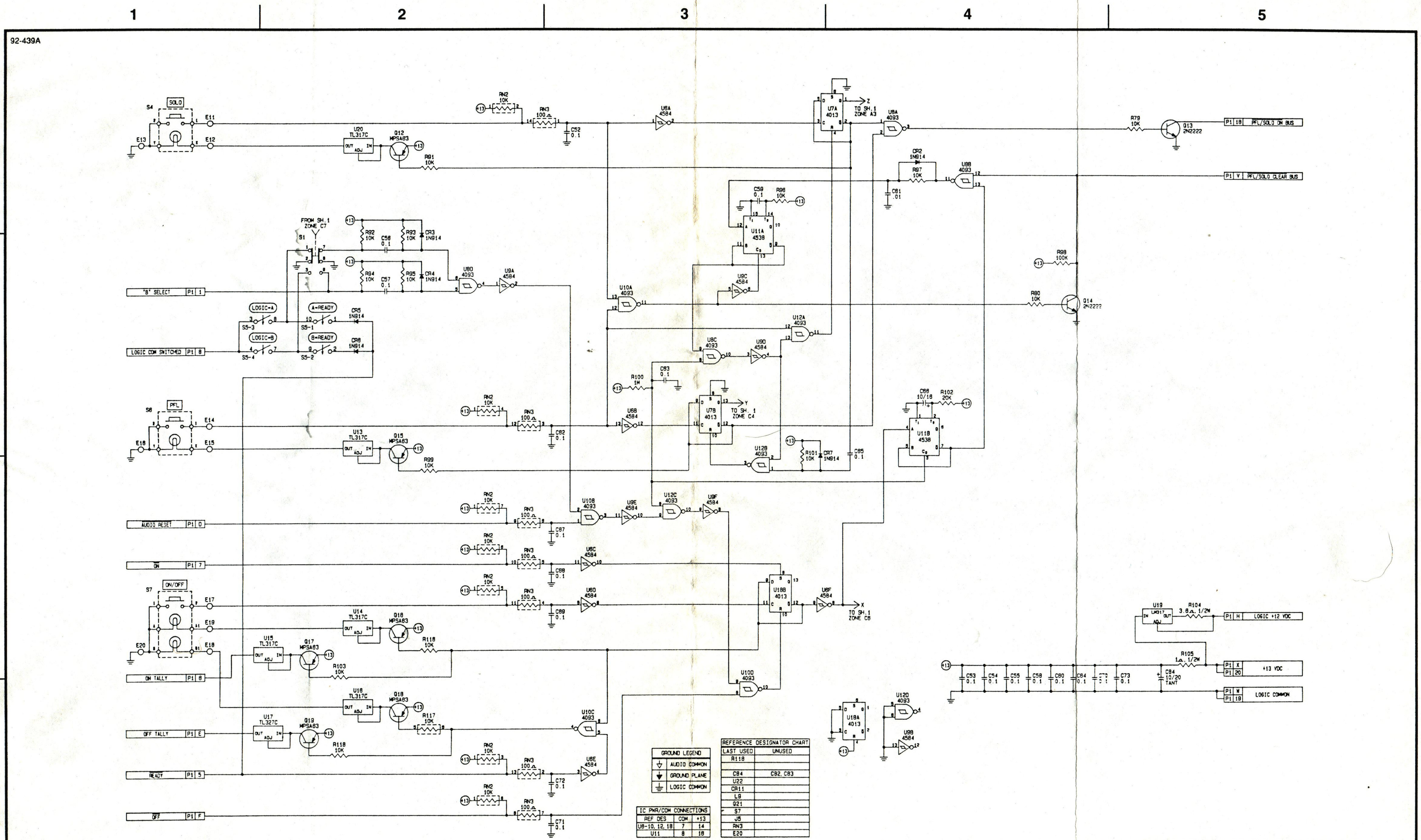


6. ▭ INDICATES FRONT PANEL COMPONENTS.
5. ⬭ INDICATES PCB MOUNTED COMPONENTS NOT ACCESSIBLE FROM FRONT PANEL.
4. #---# INDICATES MATCHED PAIR.
3. NP=NONPOLARIZED.
2. ALL CAPACITORS ARE MEASURED IN MICROFARADS.
1. ALL RESISTORS ARE 1/4W, 5%.
NOTES: UNLESS OTHERWISE SPECIFIED.

6.10 TELEPHONE INPUT/MIX-MINUS MODULE

SHEET 1 OF 2

PACIFIC RECORDERS & ENGINEERING CORPORATION



GROUND LEGEND	
	AUDIO COMMON
	GROUND PLANE
	LOGIC COMMON

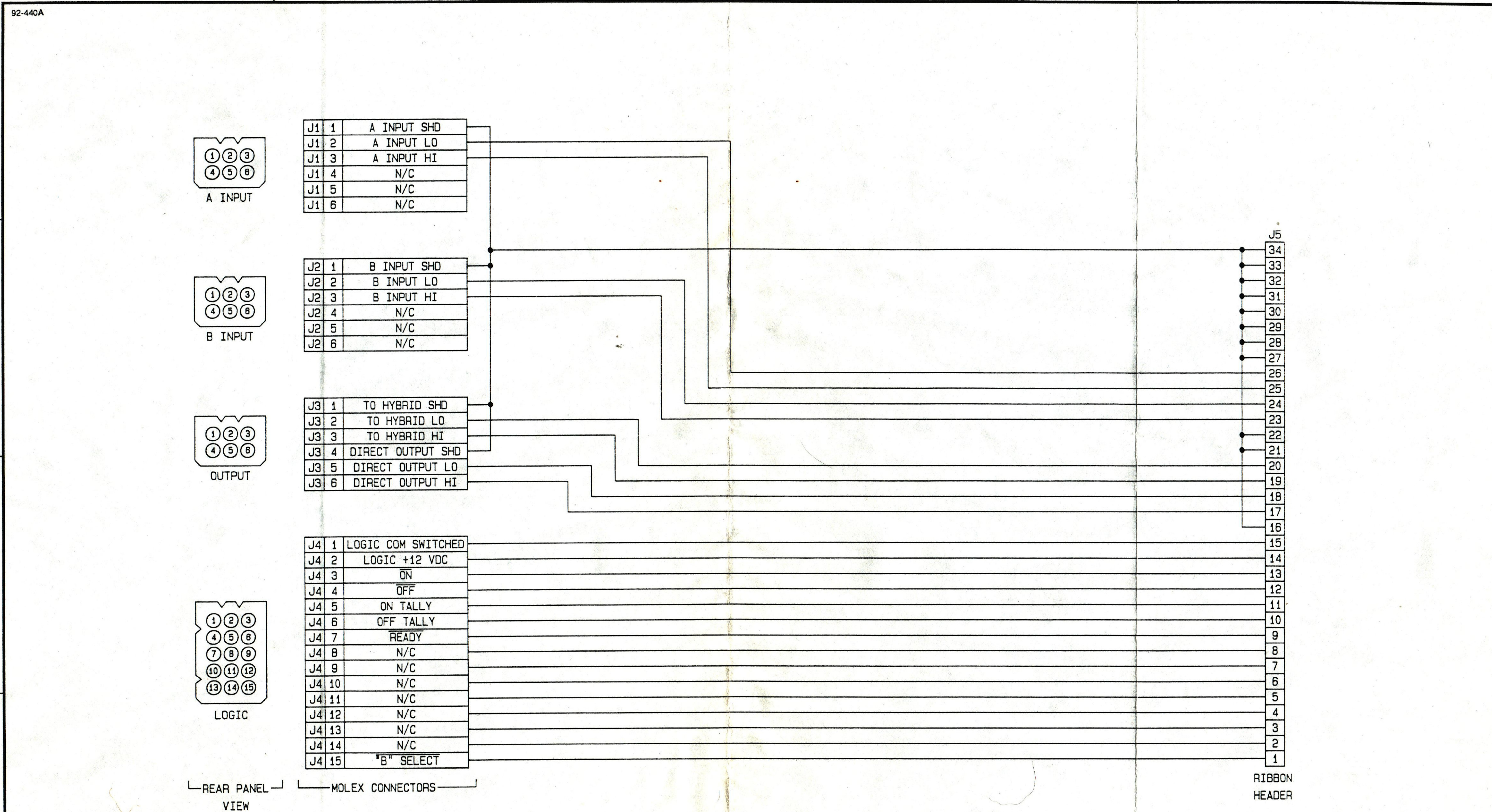
IC PWR/COM CONNECTIONS		
REF DES	COM	+13
U8-10, 12, 18	7	14
U11	8	16

REFERENCE DESIGNATOR CHART	
LAST USED	UNUSED
R118	
C84	C82, C83
U22	
CR11	
L9	
Q21	
S7	
J5	
RN3	
E20	

6.10 TELEPHONE INPUT/MIX-MINUS MODULE

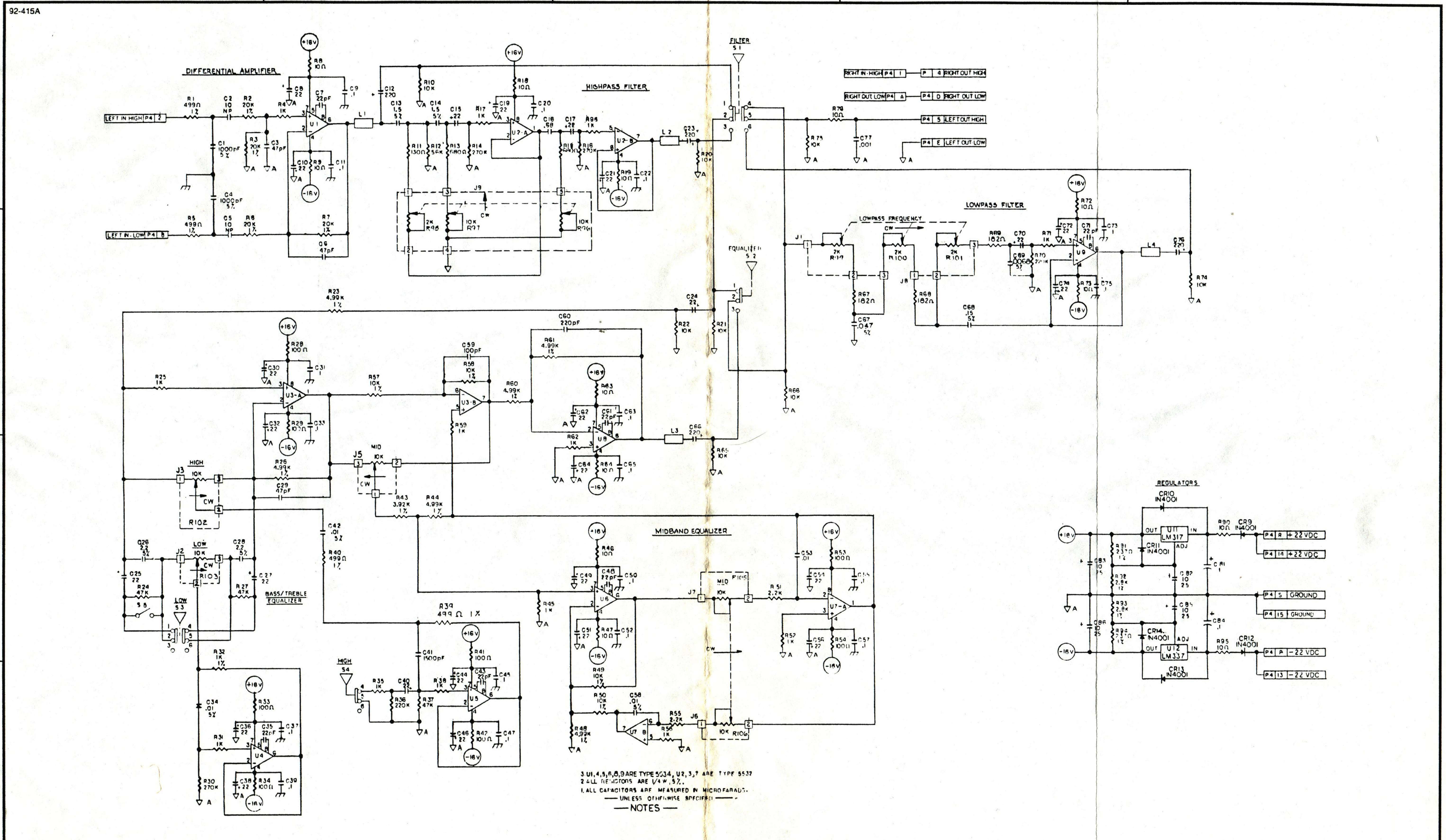
SHEET 2 OF 2

PACIFIC RECORDERS & ENGINEERING CORPORATION



PACIFIC RECORDERS & ENGINEERING CORPORATION

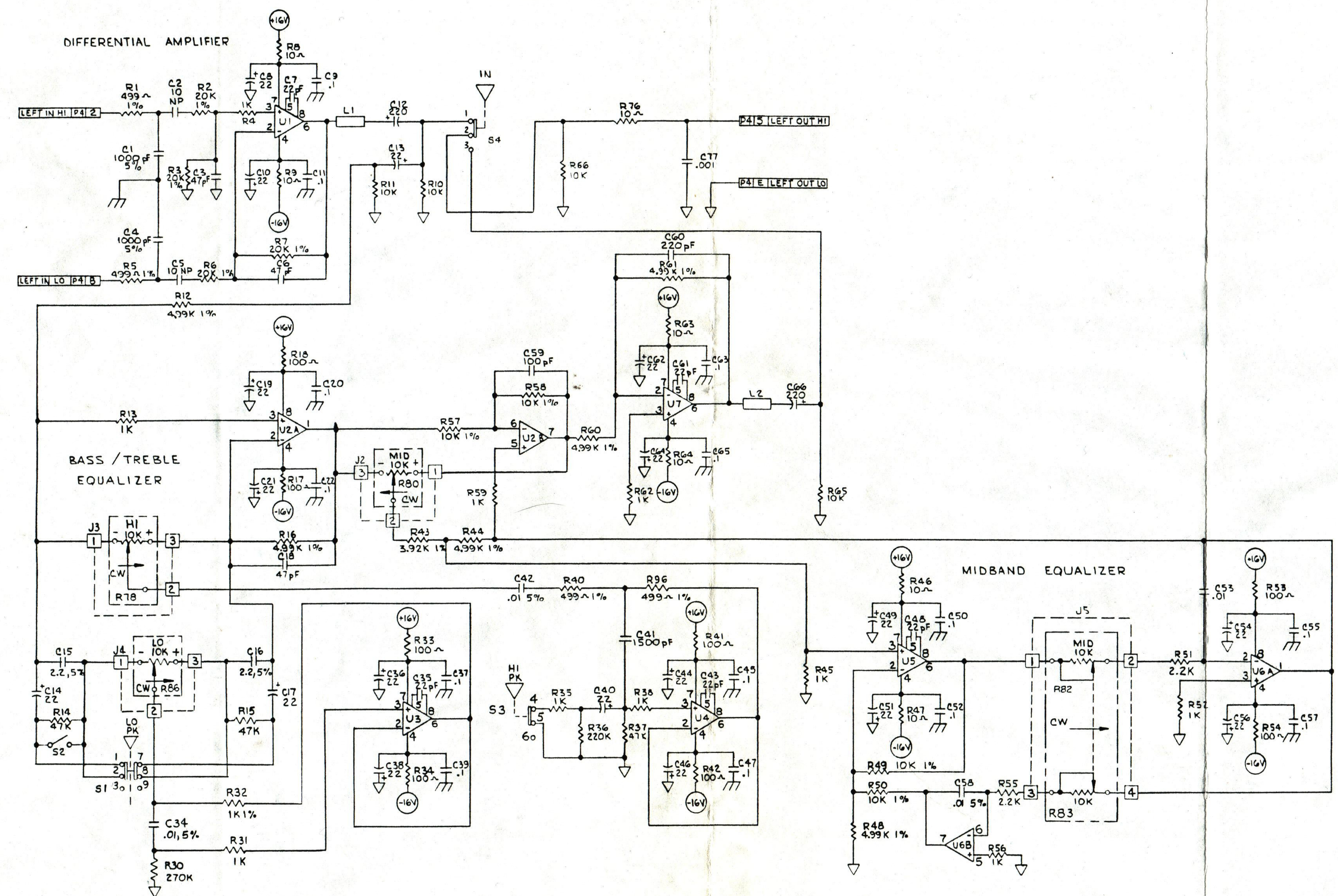
6.11 TELEPHONE INPUT/MIX-MINUS CONNECTOR INTERFACE



PACIFIC RECORDERS & ENGINEERING CORPORATION

6.12 MONAURAL EQUALIZER MODULE

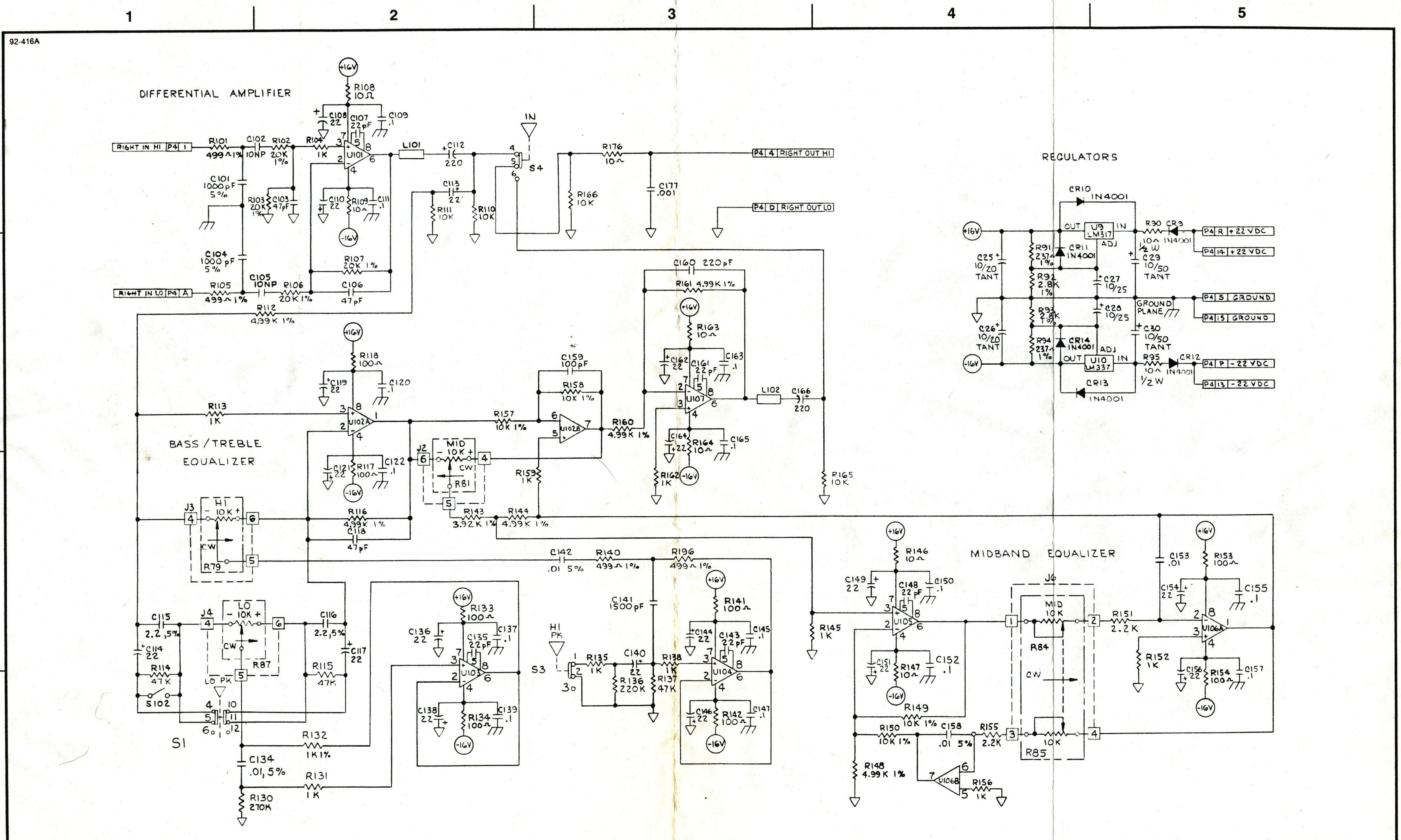
92-416A



6.13 STEREO EQUALIZER MODULE

SHEET 1 OF 2

PACIFIC RECORDERS & ENGINEERING CORPORATION

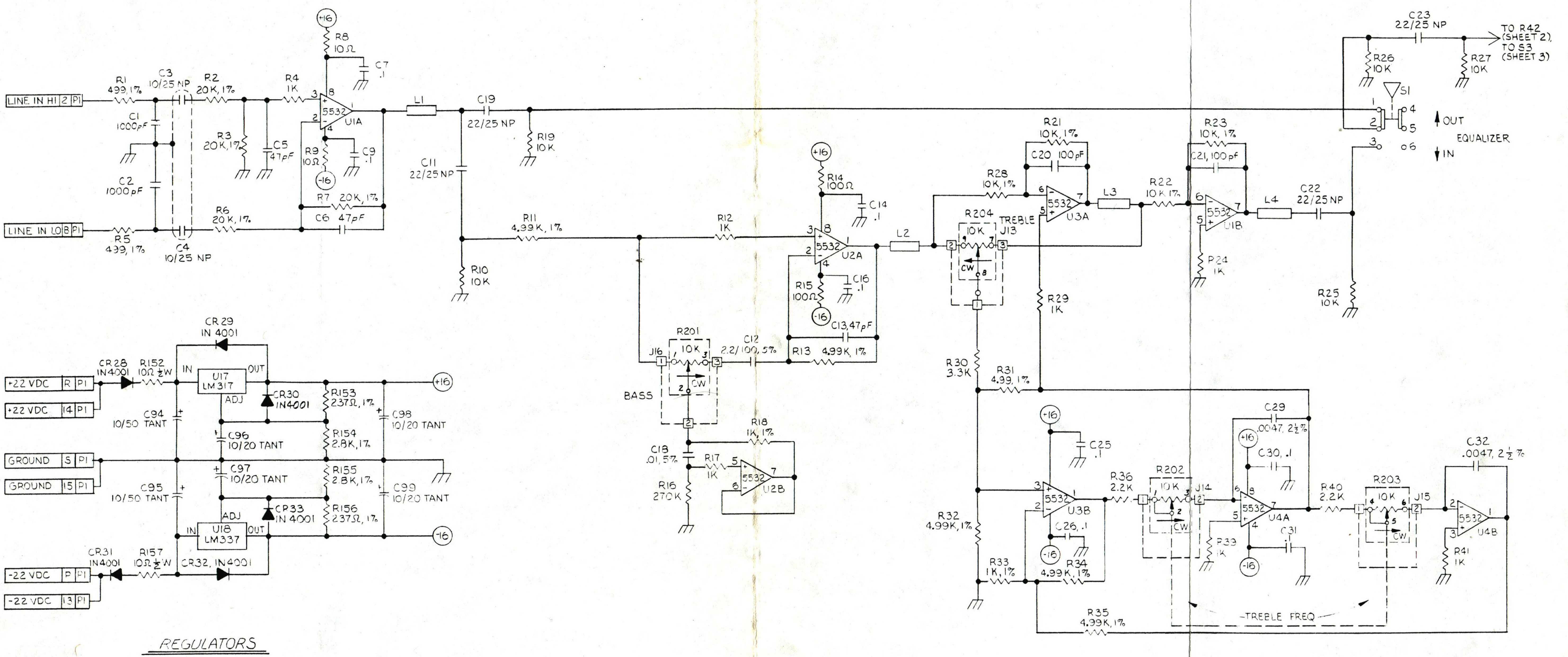


6.13 STEREO EQUALIZER MODULE

SHEET 2 OF 2

PACIFIC RECORDERS & ENGINEERING CORPORATION

92-233D



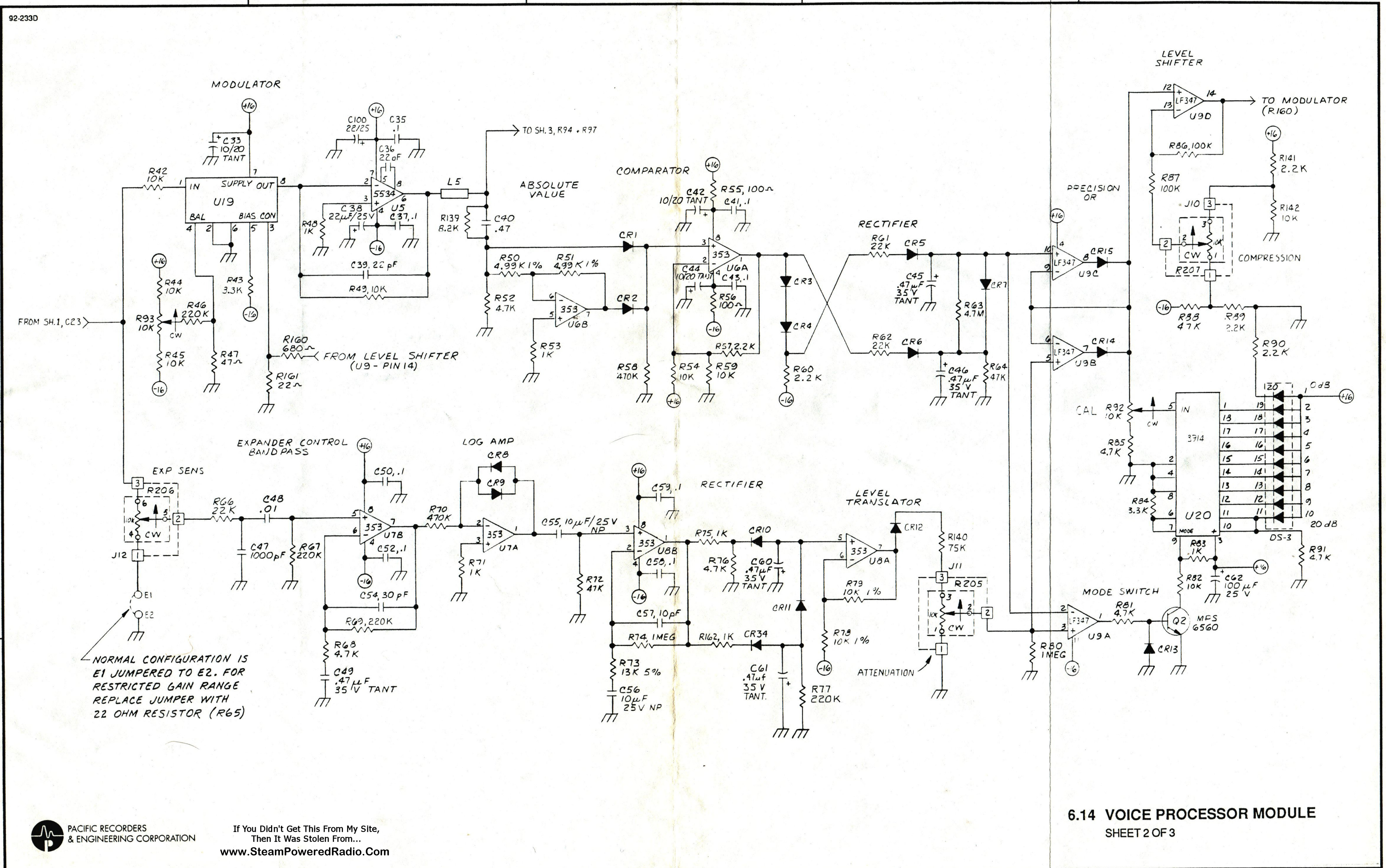
5. R201, 207, 208 ARE SINGLE ELEMENT POTS.
R205, 206 IS DUAL ELEMENT POT ADJUSTED WITH SINGLE SHAFT.
R202, 203, 204 IS TRIPLE ELEMENT POT;
R202, 203 ADJUSTED BY INNER SHAFT;
R204 ADJUSTED BY CONCENTRIC OUTER SHAFT.
4. NP = NON POLARIZED
3. ALL DIODES ARE IN914B
2. ALL CAPACITORS ARE MEASURED IN MICROFARADS
1. ALL RESISTORS ARE ¼W, 5%

NOTES: UNLESS OTHERWISE SPECIFIED

PACIFIC RECORDERS & ENGINEERING CORPORATION

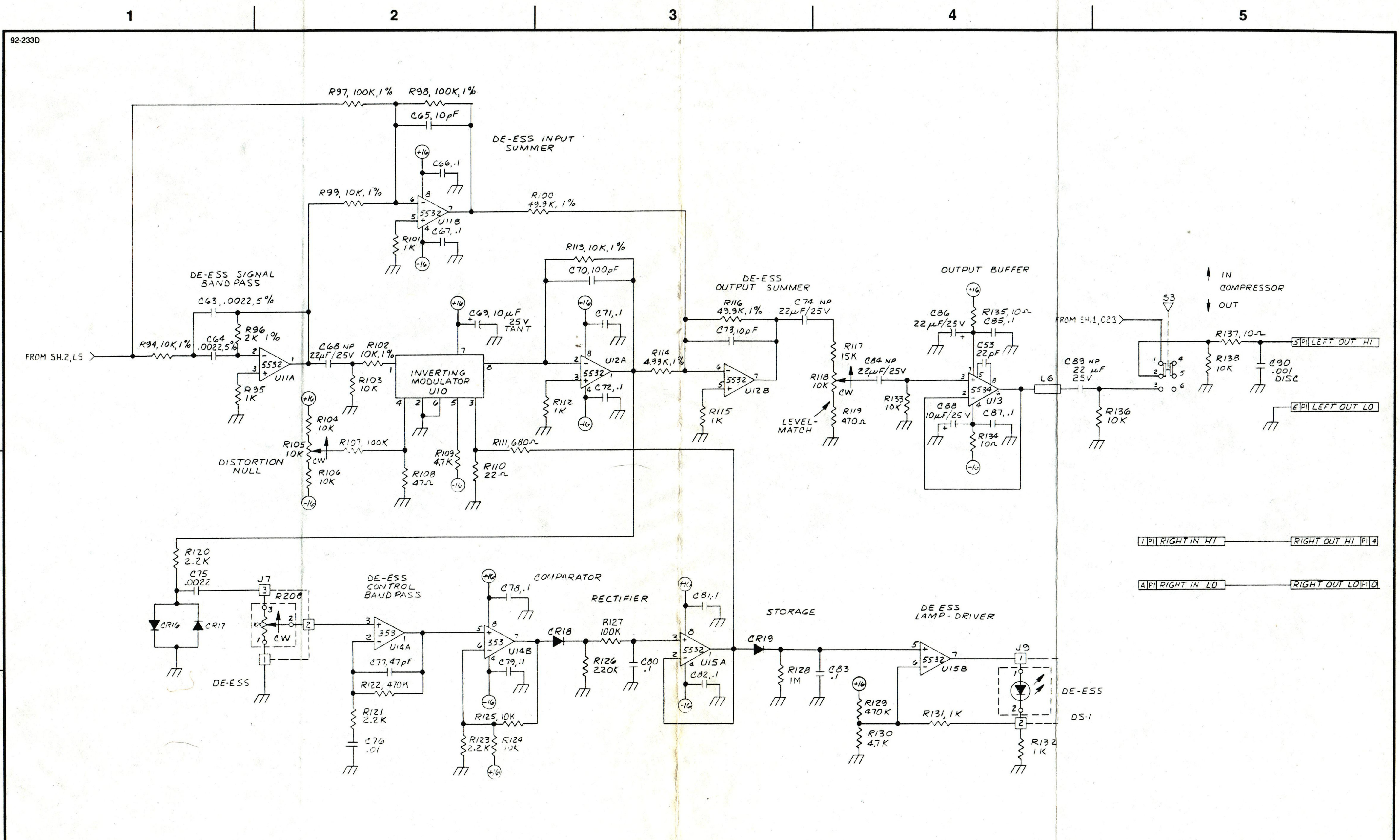
6.14 VOICE PROCESSOR MODULE

SHEET 1 OF 3



6.14 VOICE PROCESSOR MODULE

SHEET 2 OF 3

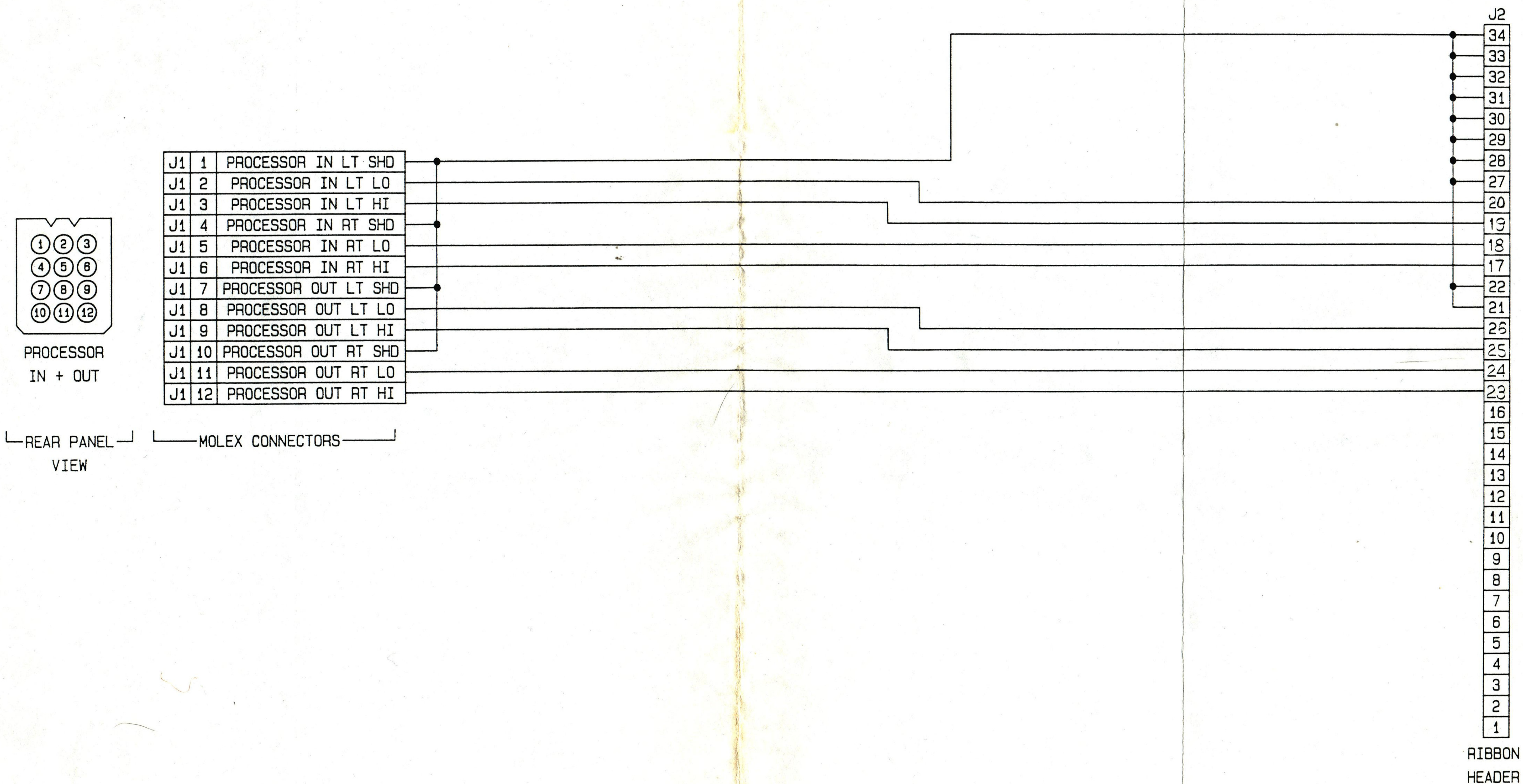


PACIFIC RECORDERS & ENGINEERING CORPORATION

6.14 VOICE PROCESSOR MODULE

SHEET 3 OF 3

92-446B



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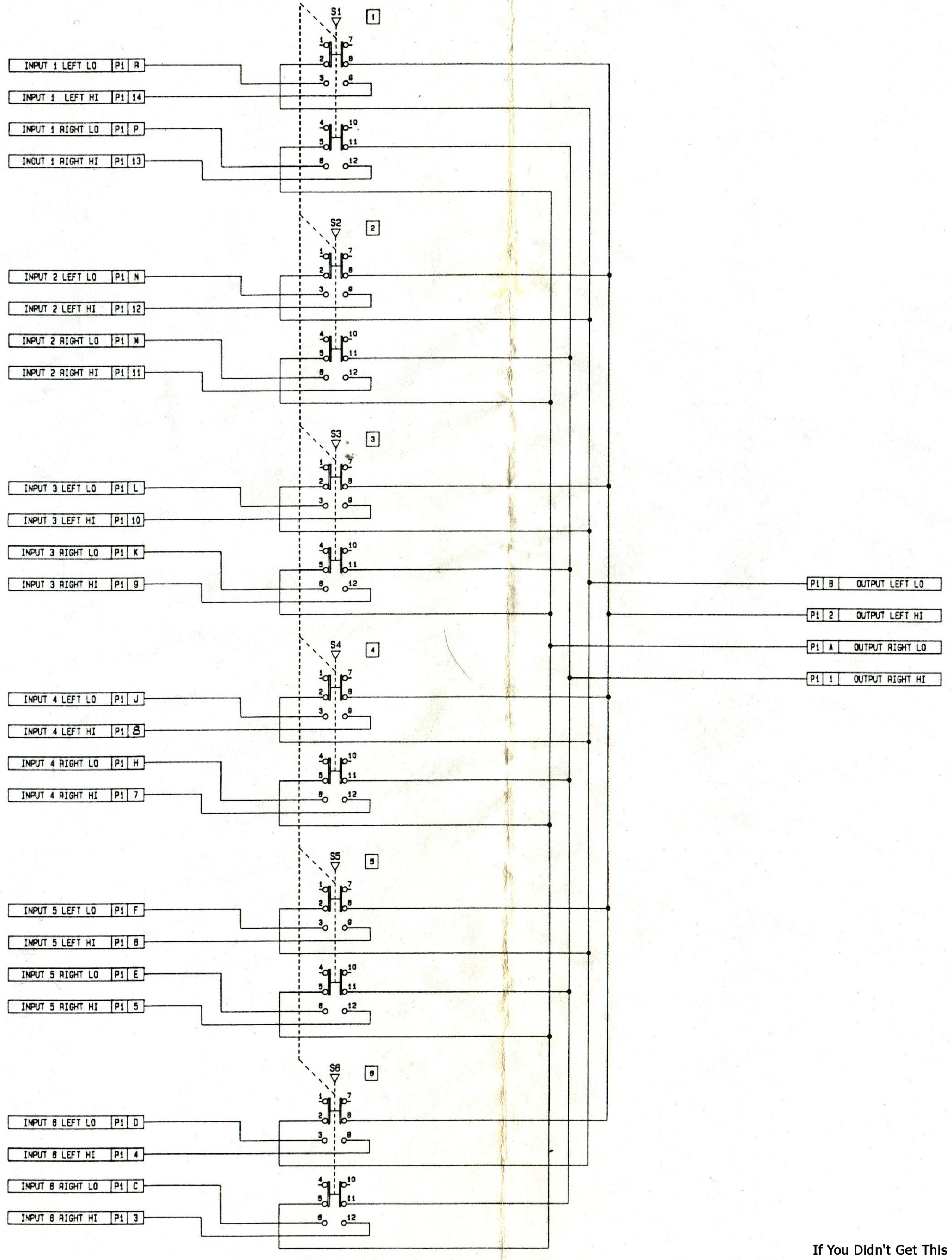
6.15 EQUALIZER/PROCESSOR CONNECTOR INTERFACE

92-433A



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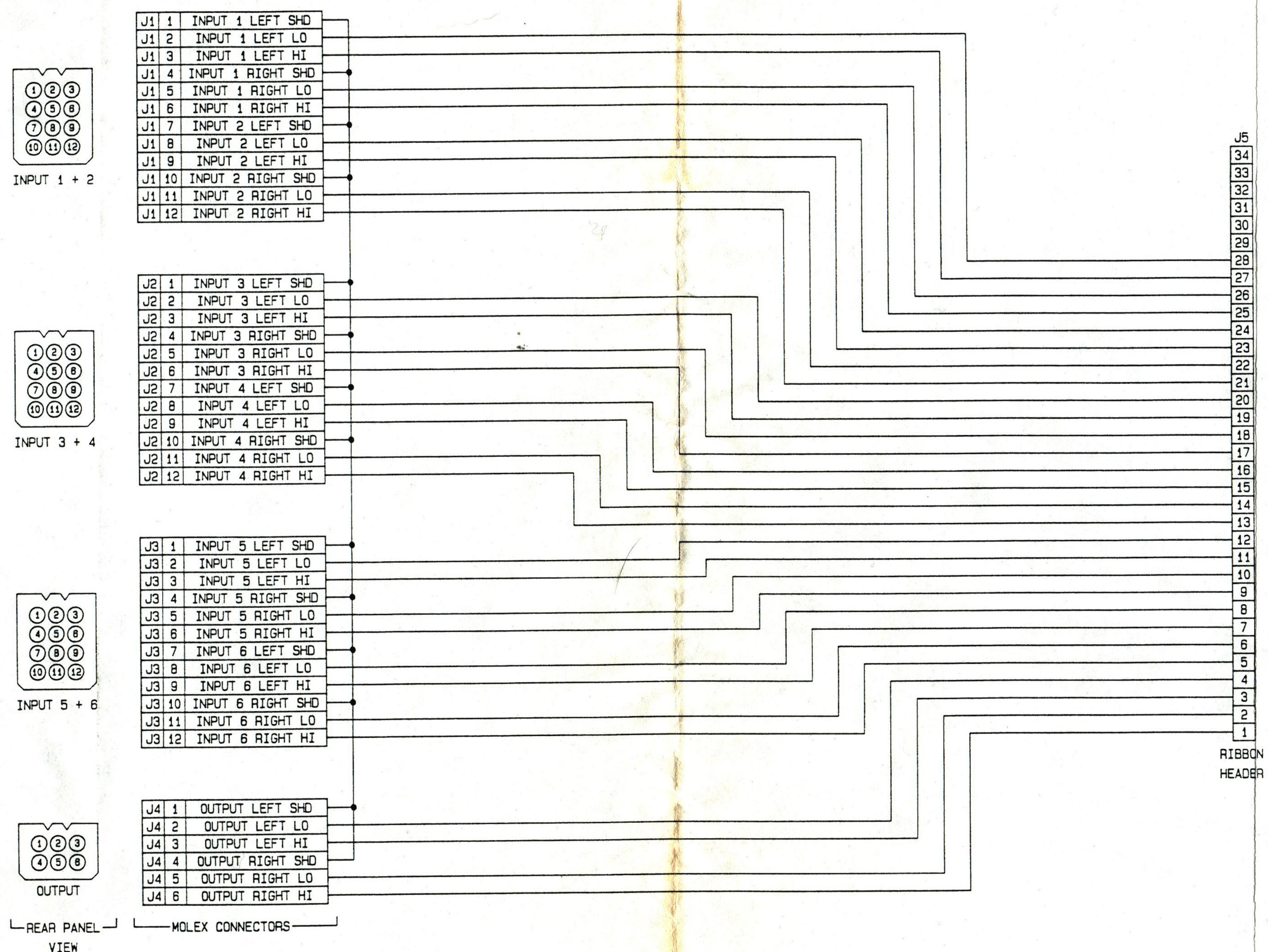
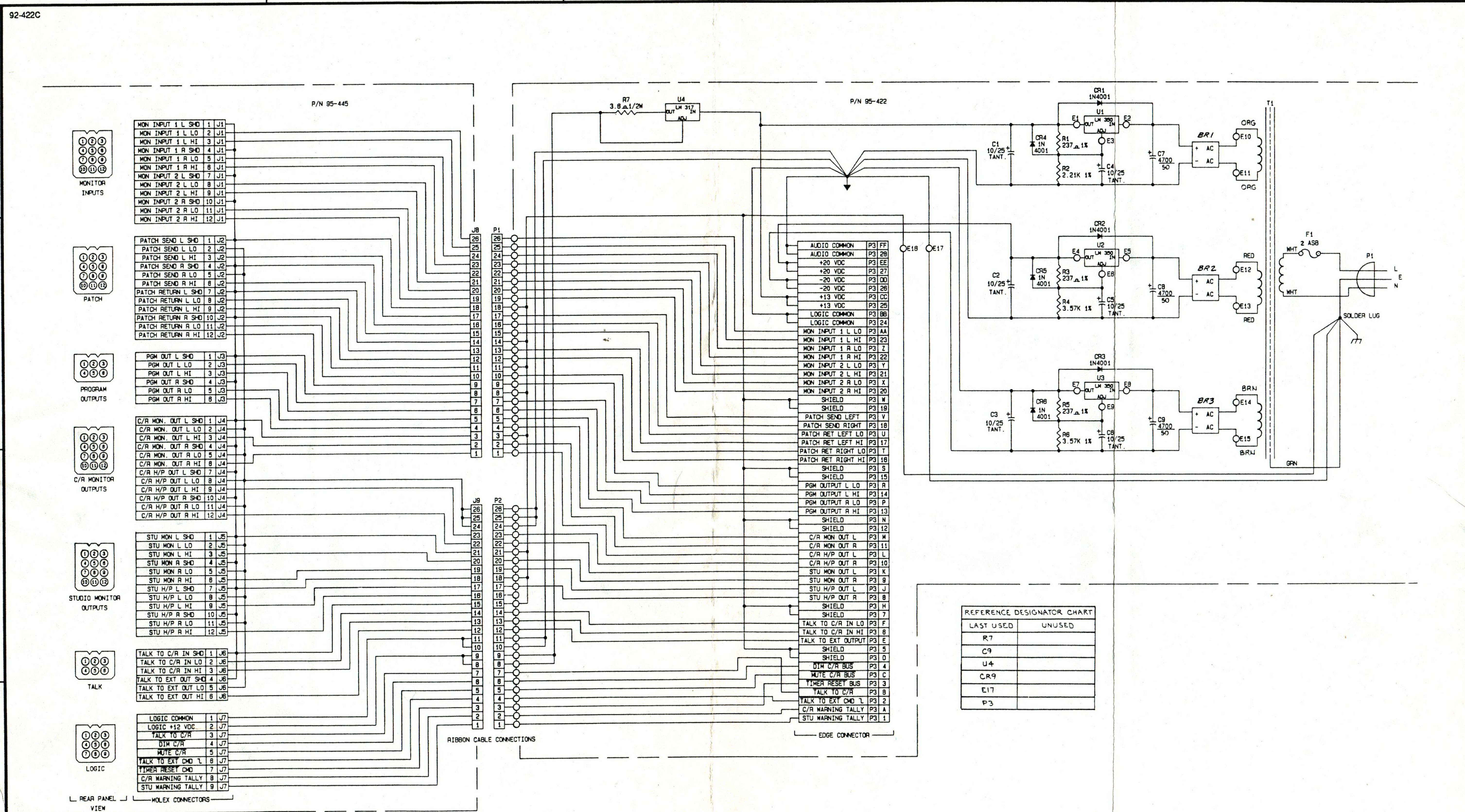
6.16 REMOTE LINE SELECTOR MODULE

92-434A



PACIFIC RECORDERS & ENGINEERING CORPORATION

6.17 REMOTE LINE SELECTOR CONNECTOR INTERFACE

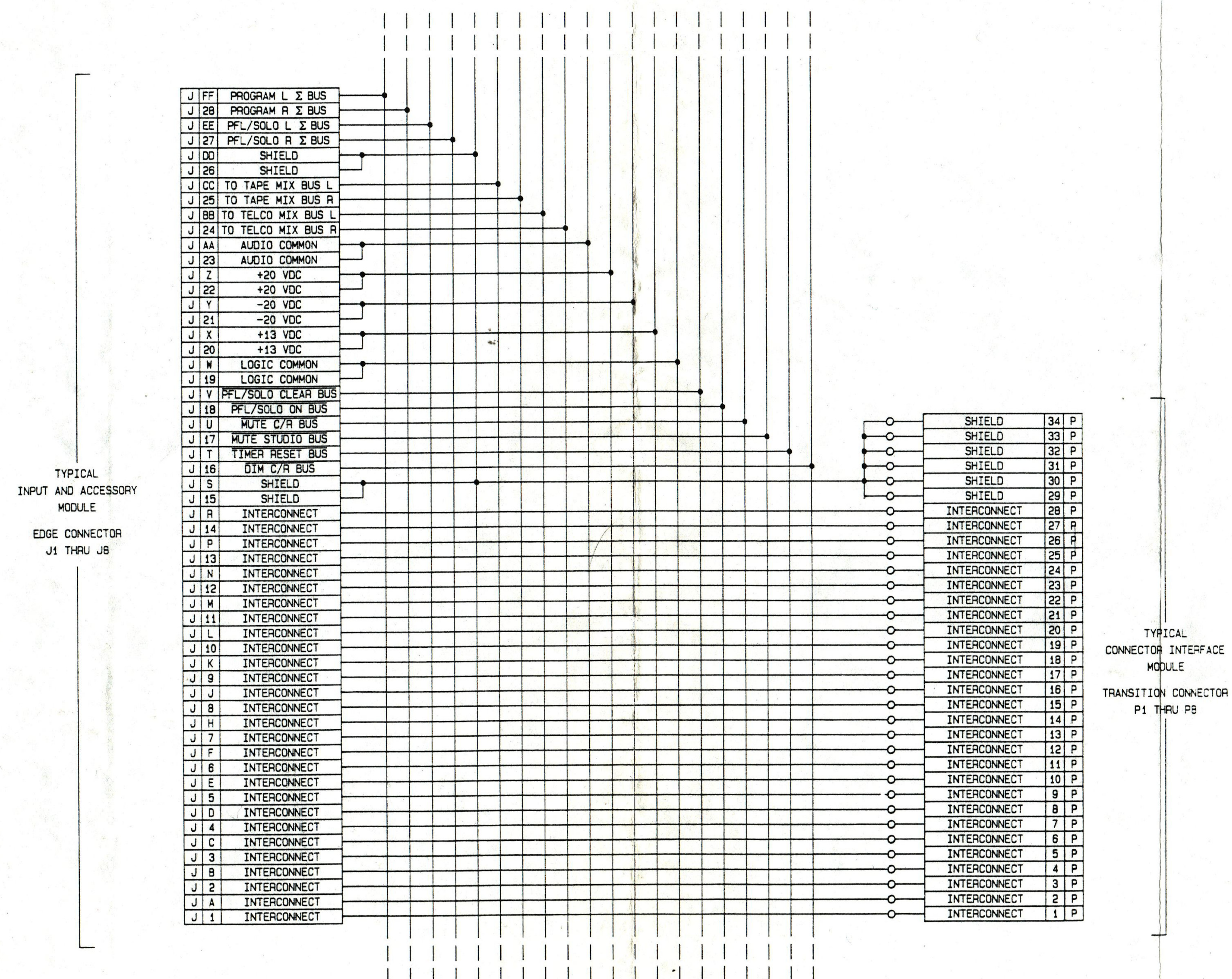


REFERENCE DESIGNATOR CHART	
LAST USED	UNUSED
R7	
C9	
U4	
CR9	
E17	
P3	

PACIFIC RECORDERS & ENGINEERING CORPORATION

6.18 POWER SUPPLY ASSEMBLY

92-420B



PACIFIC RECORDERS
& ENGINEERING CORPORATION

6.19 MAINFRAME MOTHER BOARD

SHEET 1 OF 2

1 2 3 4 5

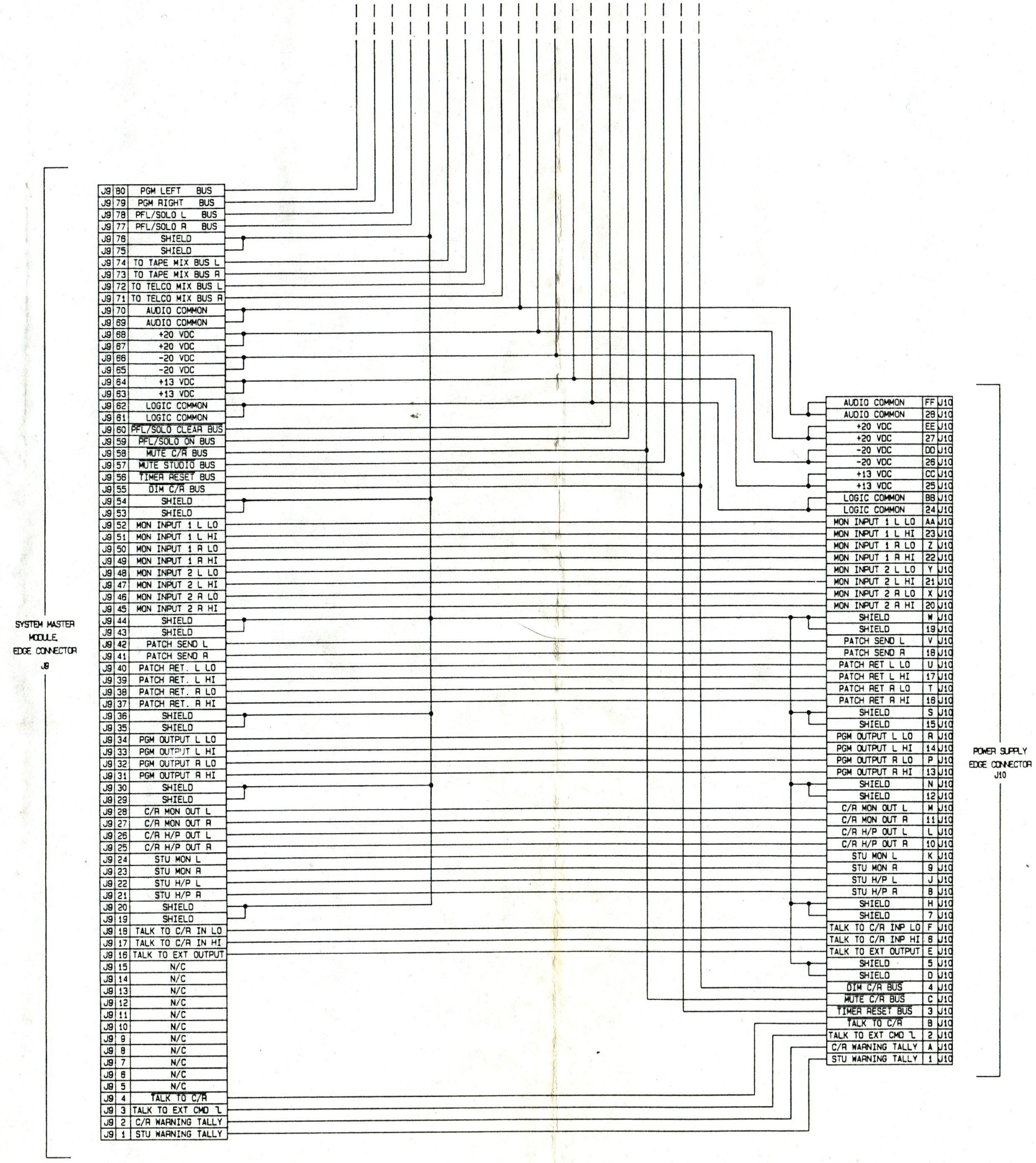
92-420B

A

B

C

D

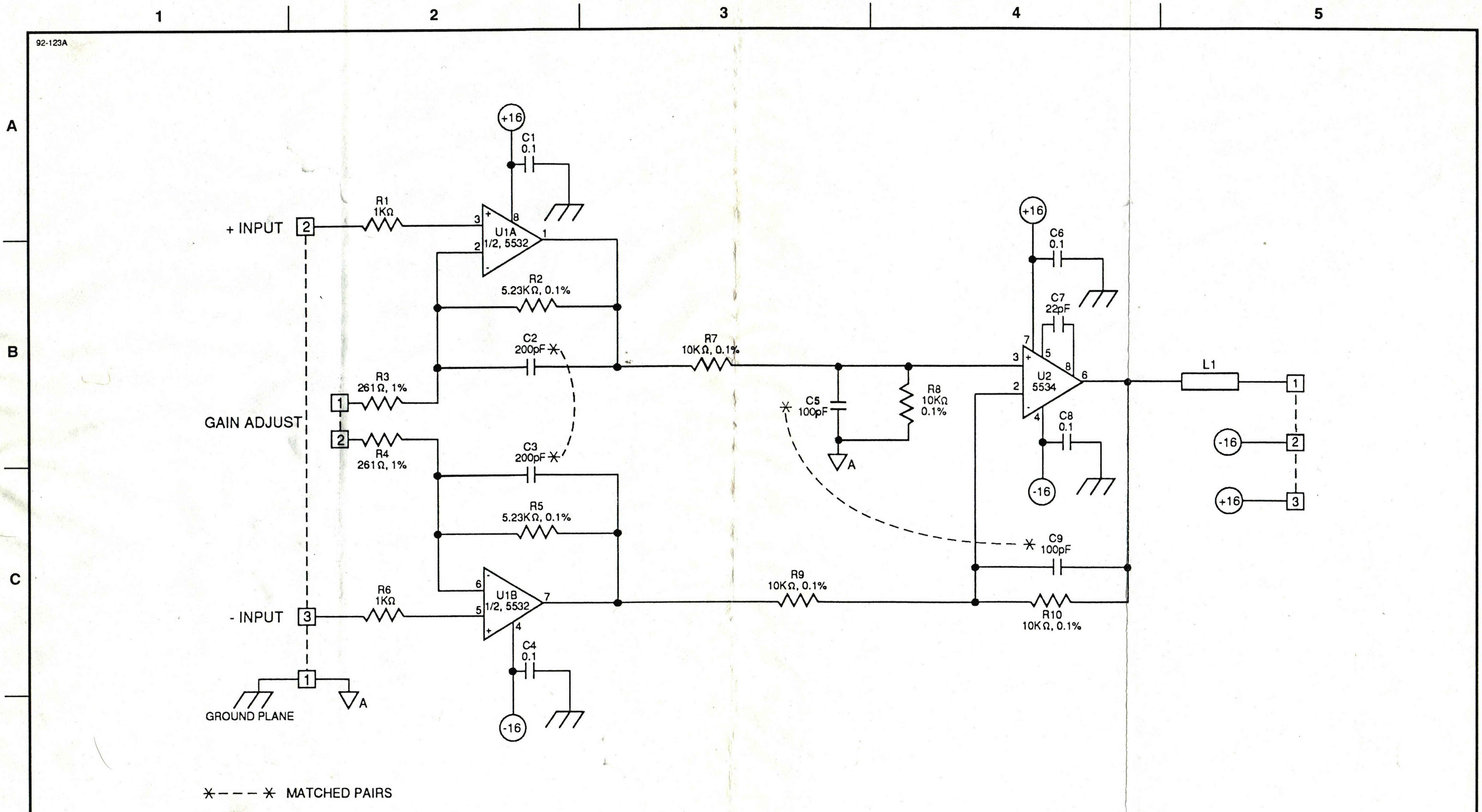


PACIFIC RECORDERS & ENGINEERING CORPORATION

6.19 MAINFRAME MOTHER BOARD

SHEET 2 OF 2

92-123A



PACIFIC RECORDERS
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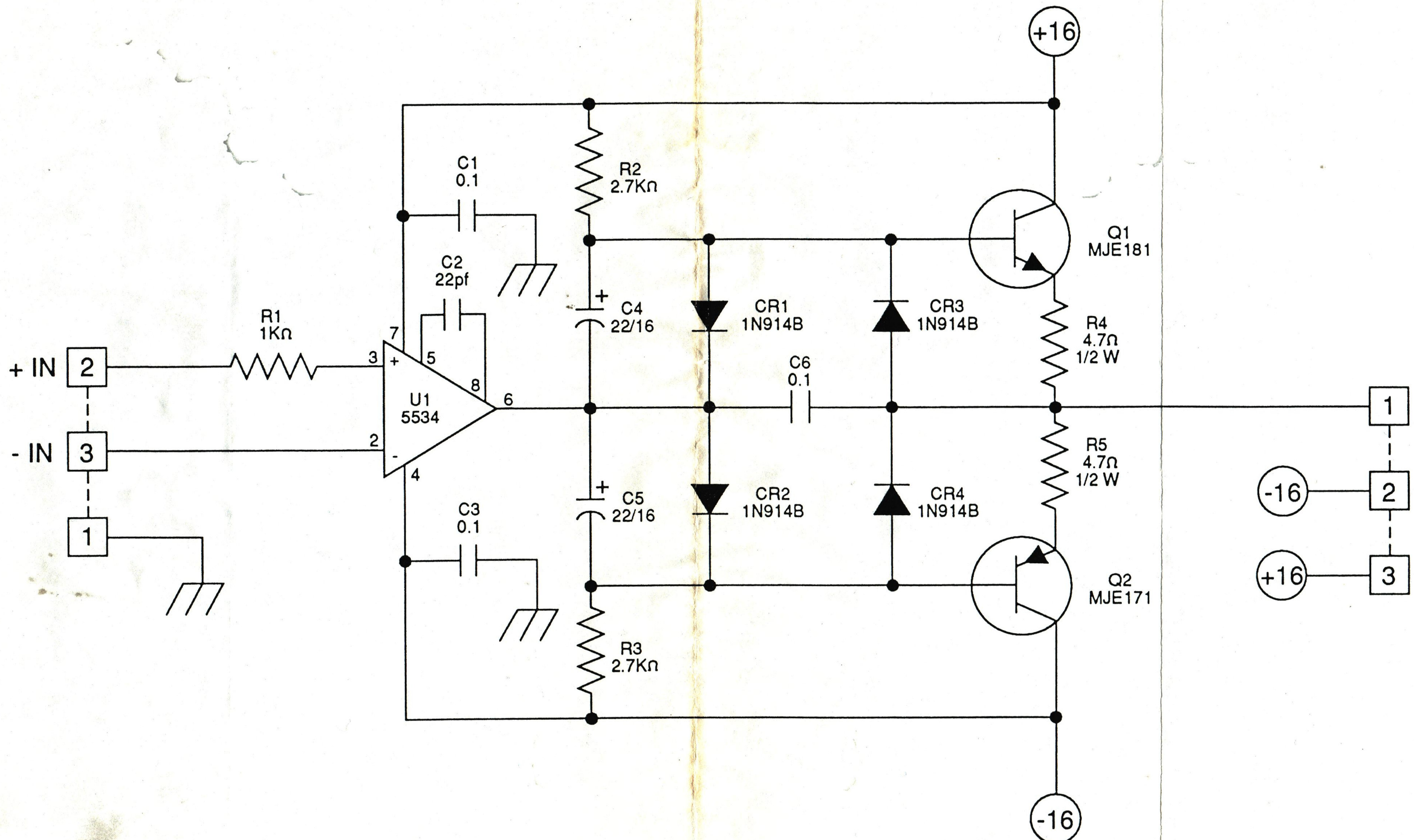
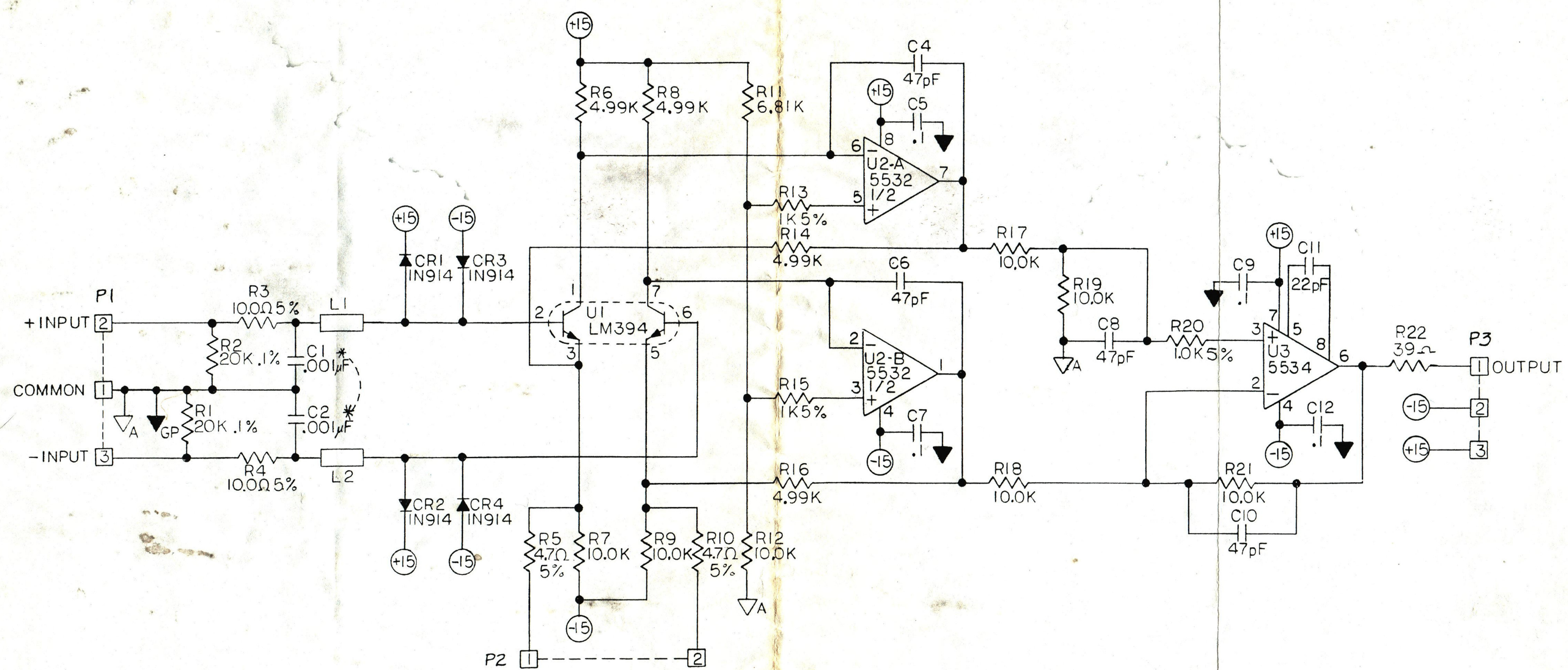
6.20 INSTRUMENTATION AMPLIFIER FUNCTION MODULE



PACIFIC RECORDERS & ENGINEERING CORPORATION

6.21 OUTPUT AMPLIFIER FUNCTION MODULE

92-270B



* MATCHED PAIRS

1. ALL RESISTORS 1/8 W ±1%

NOTES: UNLESS OTHERWISE SPECIFIED

PACIFIC RECORDERS
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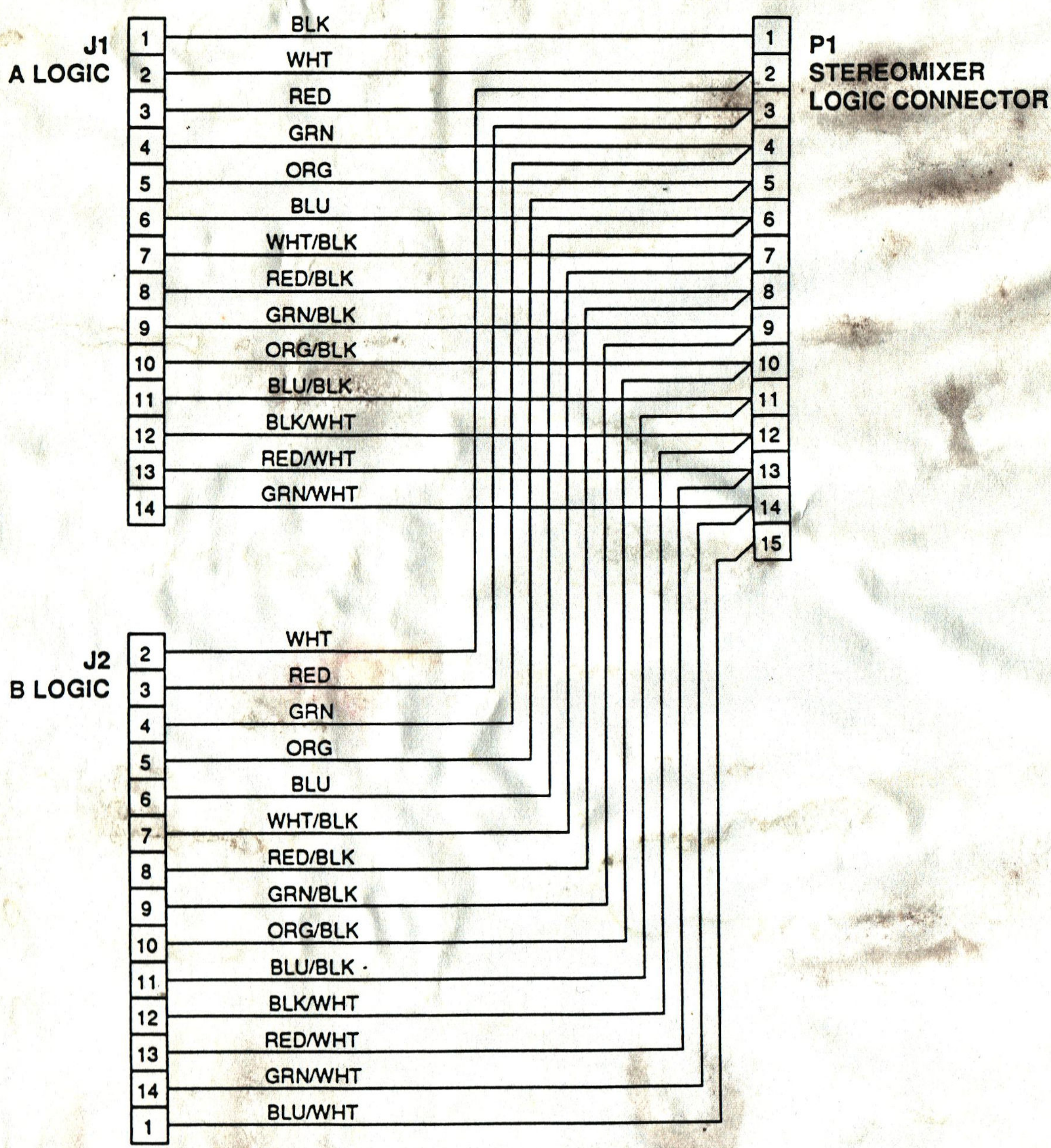
6.22 MICROPHONE PREAMPLIFIER FUNCTION MODULE

P1: HOUSING: 15 PIN MALE MOLEX #03-06-2152 (PR&E #15-607)
TERMINALS: MALE MOLEX #02-06-2103 (PR&E #15-3)

J1,J2: HOUSING: 15 PIN FEMALE MOLEX #03-06-1152 (PR&E #15-712)
TERMINALS: FEMALE MOLEX #02-06-1103 (PR&E #15-2)

CABLE TYPE: 15 COND., BELDEN #8458



NOTE: THE PR&E PART NUMBER FOR THIS ASSEMBLY IS 99-580.

6.23 STEREOMIXER A/B LOGIC "Y" CABLE

PACIFIC RECORDERS
& ENGINEERING CORPORATION

LOGIC INTERFACES

7.0 LOGIC INTERFACES

PR&E recommends the use of logic interface electronics between a console and any piece of auxiliary equipment to protect the console from hazardous voltages, and to prevent ground loops, such as those which occur when control circuitry is referenced to a different ground or power supply.

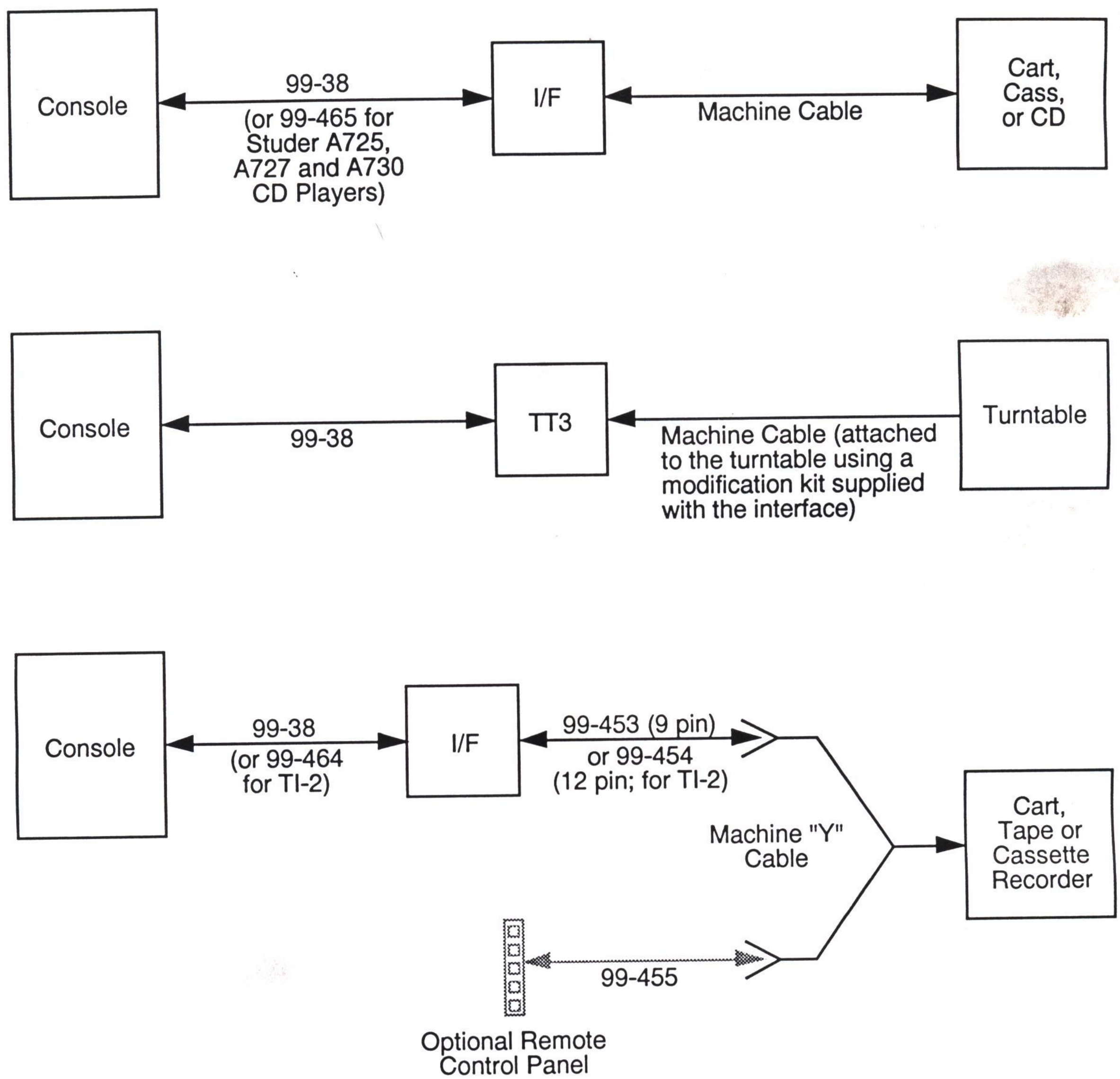
A variety of different interfaces is manufactured by PR&E to accommodate most types of cart, tape and cassette machines, turntables, and compact disc players. Cable assemblies connecting the console to the interface, and the interface to the auxiliary equipment are also produced by PR&E. For complete interface information, including all standard machine cable assemblies and machine "Y" cables, contact PR&E.

The table below provides a listing of the various interfaces described in this chapter, as well as their part numbers and a general description of their use.

Interface	PR&E P/N	For Use With
TCI-2A	99-165	PR&E Tomcat and Micromax cart machines
CI-2	99-125-0	Cart machines, CD players and cassette machines w/12-28 V lamp tallies
CI-2 (5V)	99-125-1	Cart machines, CD players and cassette machines w/5 V lamp tallies
TI-2	99-111	Reel to reel tape machines w/12-28 V lamp tallies
TT-3	99-110	Technics SP-10 and SP-15 turntables
WL-2	99-143	Studio and Control Room warning lights
Dual Relay	99-419	General purpose switching, mute contacts
Cassette Skimmer	99-410	Cassette recorders
Timer Reset	99-409	External timers

Section 7.10 contains schematics of these interfaces, and diagrams of the three console to interface cables: PR&E #99-464 (for use with the TI-2 Tape Interface); PR&E #99-465 (for use with the CI-2 Cart Interface when interfacing Studer A725, A727 and A730 compact disc players); and PR&E #99-38 (for use with all other interfaces).

Figure 7.1 (on the following page) provides an overview of the logic interface and cable system.



NOTES:
A) The addition of the optional cart or tape recorder remote control panel is made possible by the installation of the machine "Y" cable.
B) Pin assignment for the PR&E #99-38, #99-464 and #99-465 console to interface cables is defined in Section 7.10.

FIGURE 7.1
LOGIC INTERFACE AND CABLE SYSTEM OVERVIEW

7.1 TCI-2A CARTRIDGE INTERFACE

The TCI-2A Cartridge Interface (PR&E #99-165) was specifically designed for use with PR&E Tomcat and Micromax cart machines. Each TCI-2A can accommodate up to three reproducers or recorder/ reproducers.

NOTE: If you already have TCI-2 interfaces for Tomcat, they must be modified slightly to work with both Tomcat and Micromax cart machines, as follows:

1. Replace resistors R1, R3, R5, R7 and R9 with 750 ohm, 1/2 watt resistors (PR&E #3-751).

2. On the component side of the PC board, cut the trace going to J4 pin 4.

3. Add an insulated jumper wire between J4 pin 1 and the trace that previously went to J4 pin 4 (which was cut in step 2).

4. Verify that resistor R14 is a 10M ohm, 1/4 watt resistor (PR&E #2-106).

CIRCUITRY

The TCI-2A uses opto-isolators to eliminate potential audio/logic ground loops. The LED inputs of opto-isolators U1 thru U5 are connected to the cart machine tally outputs through current limiting resistors R1, R3, R5, R7 and R9. The collector of opto-isolator U1 is connected to the AUDIO RESET line on console connector J1. When the FWD LIGHT tally is present at cart connector J4, current flows through U1's LED, turning on its output transistor, and pulling the AUDIO RESET line to console LOGIC COMMON. When the STOP LIGHT tally is present at cart connector J4, U2 will operate in the same manner, pulling the READY line to console LOGIC COMMON. Capacitor C2 will pulse the AUDIO RESET line low at the "rising edge" of the STOP LIGHT tally, thus providing a console AUDIO RESET when the cart machine stops.

Opto-isolators U3, U4 and U5 provide open collector outputs referenced to an isolated COMMON on connector J3. These opto-isolators function in the manner described above, in conjunction with the remaining cart tallies: PLAY LIGHT, SEC LIGHT and TER LIGHT.

NOTE: These outputs can be used to provide automatic "cart sequencing" by connecting the appropriate cart cue line on connector J3 to the console ON input on J2 of the "next" cart machine interface. The isolated COMMON on connector J3 needs to connect to LOGIC COMMON as well. If needed, a switch can be installed in series with the connection to disable the cart sequence. The J3 outputs from several cart machines may also be paralleled together to form an "end of message" warning bus. This bus can then be buffered, as needed, to drive external warning lights. Contact PR&E's Customer Service Department for more information.

Opto-isolators U6 and U7 are connected through current limiting resistors R11 and R13 to the console START and STOP pulses on J1, respectively. The open collector outputs of U6 and U7 are connected to the cart START and STOP inputs of connector J4, respectively. The resistor and capacitor on each opto-isolator (i.e., C1/R2 on U1, etc.) slows down the switching time on the opto-isolator's output transistor.

7.2 CI-2 CARTRIDGE INTERFACE (+12 TO +28 VOLT LOGIC)

The CI-2 Cartridge Interface (PR&E #99-125-0) was designed for use with those commercial cart machines which use +12 to +28 volt logic. The CI-2 uses dry relay contacts for start and stop functions, and also serves as a capable logic interface for CD players and cassette machines. The CI-2 also routes machine status indicator feedback to the console module "OFF" lamp, mimicking machine READY, PAUSE or STOP indicators.

Each CI-2 can accommodate up to three machines.

<u>CIRCUITRY</u>

When a cartridge is inserted, current is fed to isolator circuit CR9-12, R2, R4, C2 and U2, forcing the NOT READY line low and momentarily pulsing NOT AUDIO reset low (ensuring module turn-off). A START pulse input via CR1 will turn on relay K1, starting the cartridge machine. Current fed to isolator circuit CR5-8, R1, R3, C1 and U1 will force the NOT AUDIO RESET line low. A STOP pulse input via CR2 will turn on relay K2, stopping the cartridge machine.

7.3 CI-2 CARTRIDGE INTERFACE (+5 VOLT LOGIC)

The CI-2 Cartridge Interface (PR&E #99-125-1) was designed for use with those commercial cart machines which use +5 volt logic feedback. Its function is identical to the CI-2 Interface for +12 to +28 Volt Logic.

<u>CIRCUITRY</u>

The circuitry for the +5 volt version of the CI-2 Cartridge Interface functions identically to that of the +12 to +28 volt version (reference Section 8.4.2), except that resistors R3 and R4 are 100 ohm, 1/4 watt, instead of 1K ohm, 1/2 watt.

7.4 TI-2 TAPE INTERFACE

The TI-2 Tape Interface (PR&E #99-111) was designed for use with those commercial reel to reel tape machines which use +12 to +28 volt logic. Each TI-2 accommodates one machine. Use console to interface cable #99-464 with this interface.

<u>**NOTE**</u>: This interface may also be used with cartridge recorders if the replay lockout function is desired. In this case, it will be necessary to use a custom machine "Y" cable.

<u>CIRCUITRY</u>

When the tape machine is stopped, current is fed to STOP LAMP isolator circuit R1, R2, R3, CR1, C1 and U1, forcing the NOT STOP LAMP line low, which, via U4C and pulse driver circuit R4, R20, CR8, C10 and Q1, momentarily holds the NOT AUDIO RESET low (ensuring module turn-off).

When the tape machine is in the record mode, current is fed to RECORD LAMP isolator circuit R9, R10, C3, CR3 and U3, holding the NOT AUDIO RESET low.

The NOT REPLAY line, strapped low by jumpering pins 14, 15 of J1 (i.e., reel to reel mode) via R11, R12, C4, U5B and U5C, holds resistor U6B such that the START ENABLE line is always high. When the STOP LAMP and START ENABLE lines are both high, via U5D, R14 and Q2, the NOT READY output will go low. With the NOT START ENABLE line low, a START pulse input via R16, R17, C6, U4E and U4D will trigger monostable circuit R18, C7, CR4 and U6A, which momentarily turns on driver circuit R19 and Q3 and TAPE START relay K1 (starting the tape machine). A STOP pulse input via CR6 turns on TAPE STOP relay K2 (stopping the tape machine).

Either the NOT STOP LAMP line's low state or current fed to PLAY LAMP isolator circuit R6, R7, R8, CR2, C2 and U2 forces the NOT RUN line low, which, via U5A and U4A, asserts the CART line high (i.e., there is a cartridge in place).

When used for cartridge machine operation, the NOT REPLAY line is left unstrapped to the high state. Without a cartridge loaded, the CART line is low, which, via U5B and U5C, presets U6B such that the NOT START ENABLE line is low. After insertion of a cartridge (CART line is high), a start pulse input will, as in reel to reel operation, start the cartridge machine. The NOT RUN line will then go low, which, via U4B, will clock the low state on the data input of register U6B, asserting the NOT START ENABLE line high. When the cartridge machine recues and stops, the RUN line will go low and the NOT START ENABLE line remains high, which inhibits further start pulses from triggering monostable U6A, and prevents restarting the cartridge machine. Either the CART line going low (by removing the cartridge) via U5B, or a STOP pulse input, via R13, C5 and U4F via U5C, will again preset register U6B, asserting the NOT START ENABLE line low.

7.5 TT-3 TURNTABLE INTERFACE

The TT-3 Turntable Interface (PR&E #99-110) was designed for use with Technics SP-10 and SP-15 turntables. Each TT-3 can accommodate up to two turntables.

<u>CIRCUITRY</u>

When the turntable is stopped, no current will be fed to isolator circuit R1, R2, C1, C2 and U1, allowing the NOT RUN line to go high, which, via driver R3, R12, and Q1, holds the NOT READY output low (i.e., turntable is ready) and momentarily pulses, via C8, the NOT AUDIO RESET line low (ensuring module turn-off). With the NOT RUN line's high state fed to U2D, a START pulse, via R4, R5, C3, U2D and U2B, will cause a CONTROL pulse output via isolator circuit R8, R9, Q2, U3, R10, R11 and C5, starting the turntable. With the RUN line's high state fed to U2C (i.e., the turntable is running), a STOP pulse, via R6, R7, C4 U2C and U2B will similarly cause a CONTROL pulse output, stopping the turntable.

7.6 WL-2 WARNING LIGHT INTERFACE

The WL-2 Warning Light Interface (PR&E #99-143) is an opto-isolated switched AC relay designed to illuminate external Control Room and/or Studio warning lights. This interface has two inputs (STUDIO and CONTROL ROOM), which are activated by 12 volt DC command outputs from the console's Control Room and Studio Monitor Module LOGIC connectors. Each input has a corresponding AC output.

NOTE: The WL-2 inputs are polarity sensitive, and will not work unless connected correctly.

CIRCUITRY

NOTE: The WL-2 contains two independent relays. For clarity and simplicity, only relay K1 is described below.

A voltage between 3 VDC and 32 VDC applied to input connector J1 energizes solid state relay K1. Relay K1 then switches the "hot" lead (black) of the 115 VAC input to output connector J2. Switching transitions occur during the zero crossing point of the AC source to minimize noise transients.

7.7 DUAL RELAY INTERFACE

The Dual Relay Interface (PR&E #99-419) consists of two separately controlled relays, which can be used for audio or control circuitry.

CIRCUITRY

NOTE: The Dual Relay Interface contains two independent 4PDT relays. For clarity and simplicity, only relay A is described below.

With no control voltage applied, relay A will have continuity between each WIPER terminal and its respective N.C. terminal. With 12 VDC applied to terminals COM and +12 V, each WIPER will have continuity to its respective N.O. terminal.

NOTE: Relays A and B each draw .036 amps when energized with contacts rated at 2 amps at 30 VDC.

7.8 CASSETTE SKIMMER INTERFACE

The Cassette Skimmer Interface (PR&E #99-410) is used to create cassette recordings of selected console microphone inputs. One to four Microphone Input Modules may be connected to the skimmer, which will toggle up to two cassette tape machines from the pause mode into the record mode the instant a connected mic module is turned "ON". When the mic module is turned "OFF", the cassette tape machines toggle from the record mode into the pause mode. In this way, a recording of a personality or personalities may be created for later review.

NOTE: When the mic module is turned "OFF", there is a one-second delay in the cassette machine toggling from the record mode to the pause mode. This allows for the recording of the voice-to-music transition.

Another method of connecting the cassette skimmer is to connect it via Control Room Monitor Module WARNING COMMAND. When this method is used, the cassette machine will be placed in record whenever the Control Room speakers mute.

CIRCUITRY

Power is applied to the interface through input connectors J1 through J4 and diodes CR2, CR4, CR6 and CR8. This arrangement allows any or all of the four inputs to be used. The ON TALLY voltage is routed through diodes CR1, CR3, CR5 and CR7 to the "fast attack/slow release" circuit comprised of R1, R2, CR9, R3 and C5.

When ON TALLY voltage is applied, C5 is charged quickly through CR9 and R3. When ON TALLY voltage is removed, C5 is discharged slowly through R2 and R1. Schmitt inverter U1 squares up and inverts this signal and sends it to monostables that generate the PLAY and PAUSE pulses for each cassette recorder. U2 generates pulses for Cassette 1 and U5 generates pulses for Cassette 2.

U2A and U5A are configured to trigger on the "falling edge" of the inverted ON TALLY voltage, which indicates the beginning of cassette skimming and cassette RECORD. U2B and U5B are configured to trigger on the "rising edge" of the inverted ON TALLY voltage, which causes the cassette to PAUSE.

Grounding the SKIM DISABLE input on J5 disables the PLAY and PAUSE pulses for Cassette 1, and grounding the SKIM DISABLE input on J7 disables these pulses for Cassette 2.

Each of the four monostables generates a 0.1 second pulse governed by the time constant of C7/R6, C8/R9, C10/R14 and C11/R17. Each monostable is connected to an opto-isolator to eliminate any audio or logic ground loops. Opto-isolators U3, U4, U6 and U7 are driven by transistors Q1, Q2, Q3 and Q4.

7.9 TIMER RESET INTERFACE

The Timer Reset Interface (PR&E #99-409) was designed to interface a cart machine with a stand-alone timer. The cart machine play lamp tallies are connected to the interface, which provides timer START and RESET commands. Up to three cart machines may be connected to a single interface.

CIRCUITRY

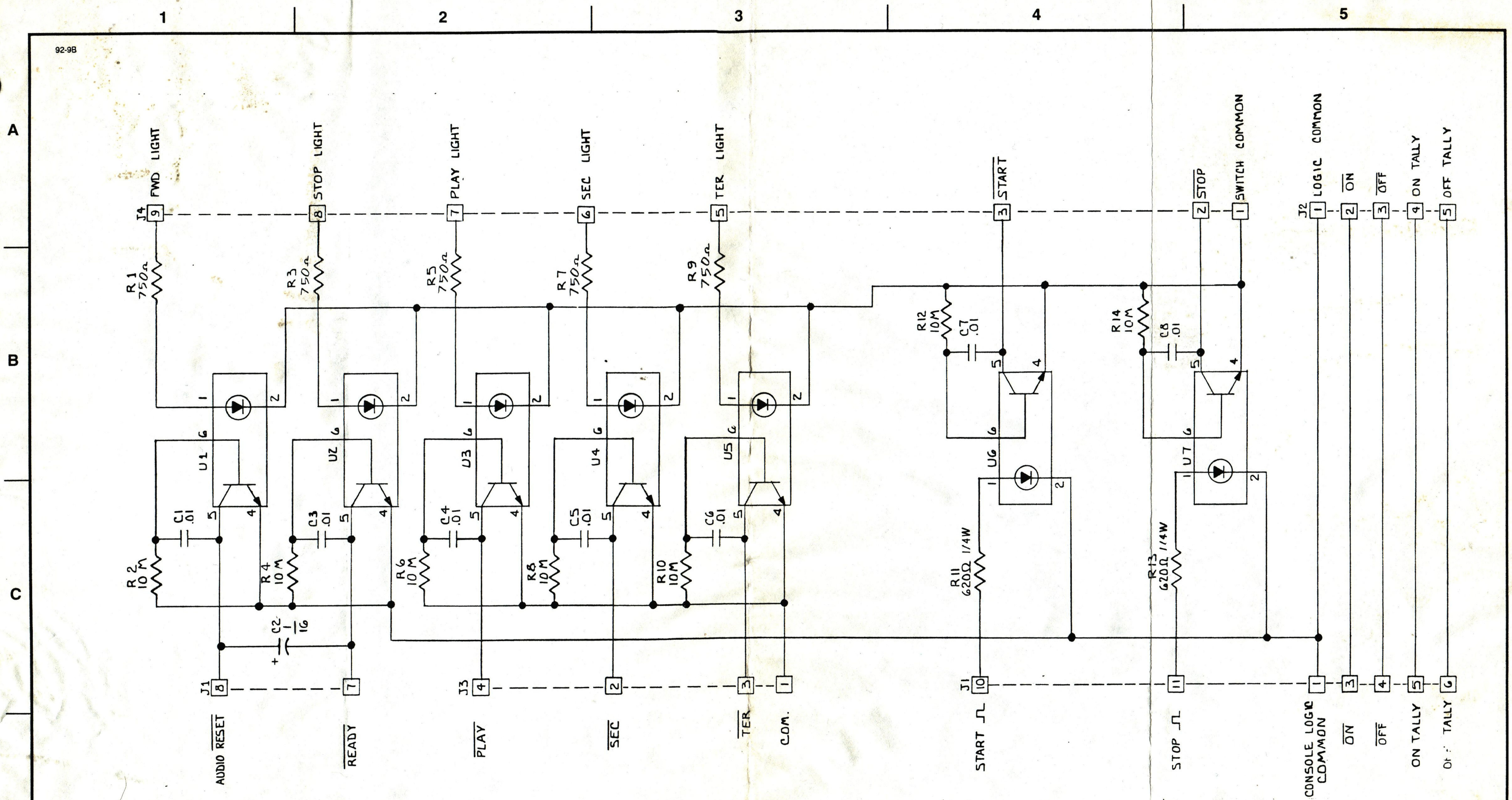
When a positive voltage is applied to pin 2 of connector J1 (CART 1), CR1 will be forward biased, allowing current to flow through the diode of opto-isolator U1. The use of an opto-isolator eliminates any audio/logic ground loops within the system. The current is maintained at 18 mA by the constant current circuit comprised of Q1, CR2, CR3, R1 and R2. This current is consistent over an input voltage range of 4 VDC to 30 VDC. CR2 is thermally coupled to Q1 to reduce current change due to thermal rise. The open-collector output of U1 is pulled-up to +V by R4 when no PLAY LAMP voltage is present. When PLAY LAMP voltage is applied, the collector of U1 is pulled to timer common causing the cathodes of CR11 and CR12 to be pulled low until C2 charges up through the pull-up resistors inside the timer. This action causes the TIMER START and TIMER RESET lines to be pulsed low, which will start and reset the timer on the "rising edge" of the PLAY LAMP voltage. Each of the three inputs, CART 1, CART 2 and CART 3, function as described above, with each able to start and reset the timer independently. CR10 guarantees that C2, C4 and C6 are discharged when no PLAY LAMP voltages are present. All connections to the timer are made through J4, with the timer providing +V for the pulse circuitry.

7.10 INTERFACE SCHEMATICS AND CABLE DIAGRAMS

This section contains schematics for the various interfaces and diagrams for the three console to interface cables described in this document, as follows:

NOTE: On PR&E schematics, capacitor values are in microfarads (μF), unless otherwise specified, and resistors are 1/4 watt, 5%, unless otherwise specified.

92-9B



NOTE: U1 THRU U7 ARE TYPE MCT-2.

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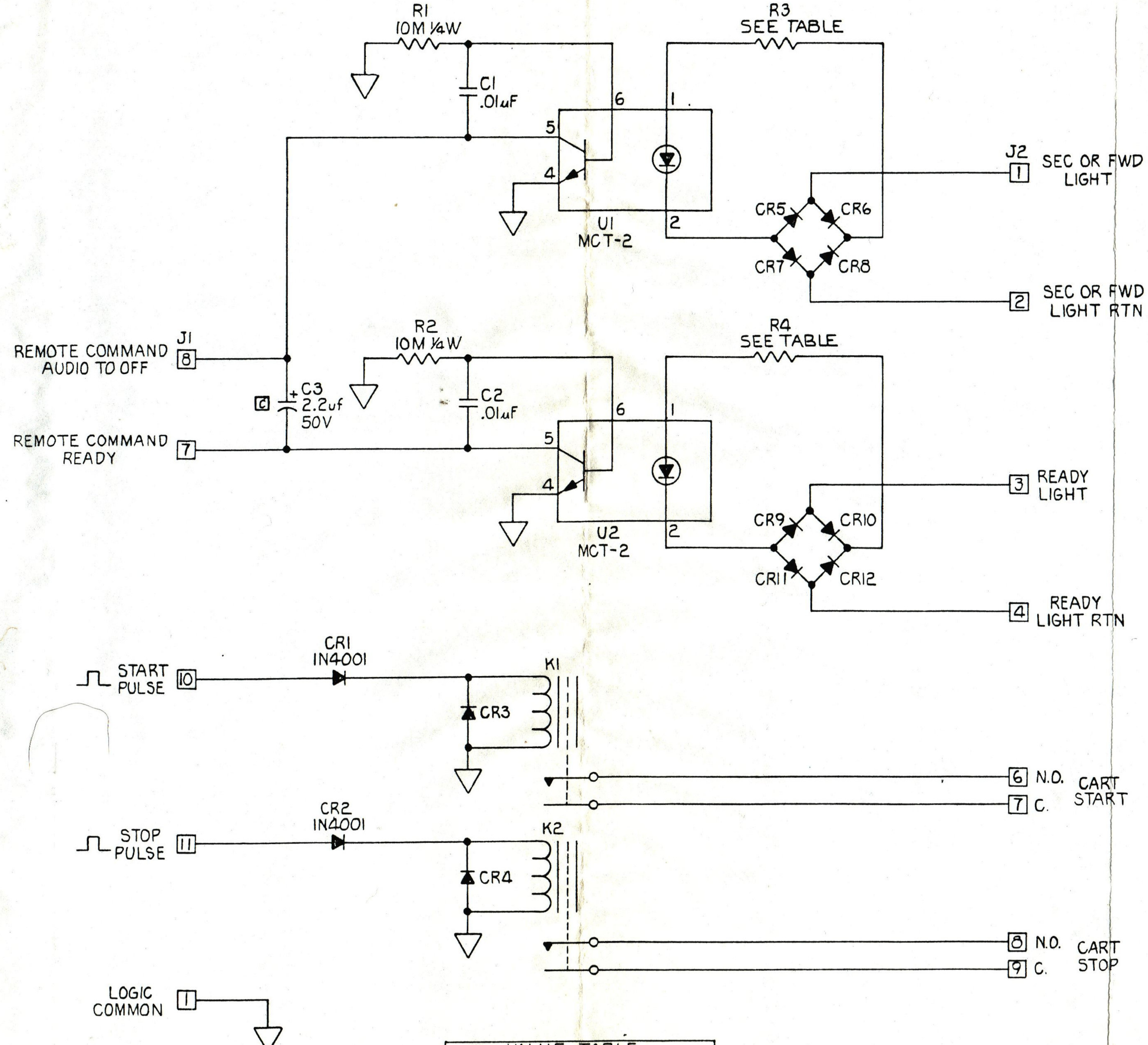
7.10.1 TCI-2A CARTRIDGE INTERFACE

90-170C



VALUE TABLE		
DASH NO.	-0	-1
R3	1K 1/2 W	100Ω 1/4W
R4	1K 1/2 W	100Ω 1/4W

NOTE: CR3-CR12 ARE 1N914B.

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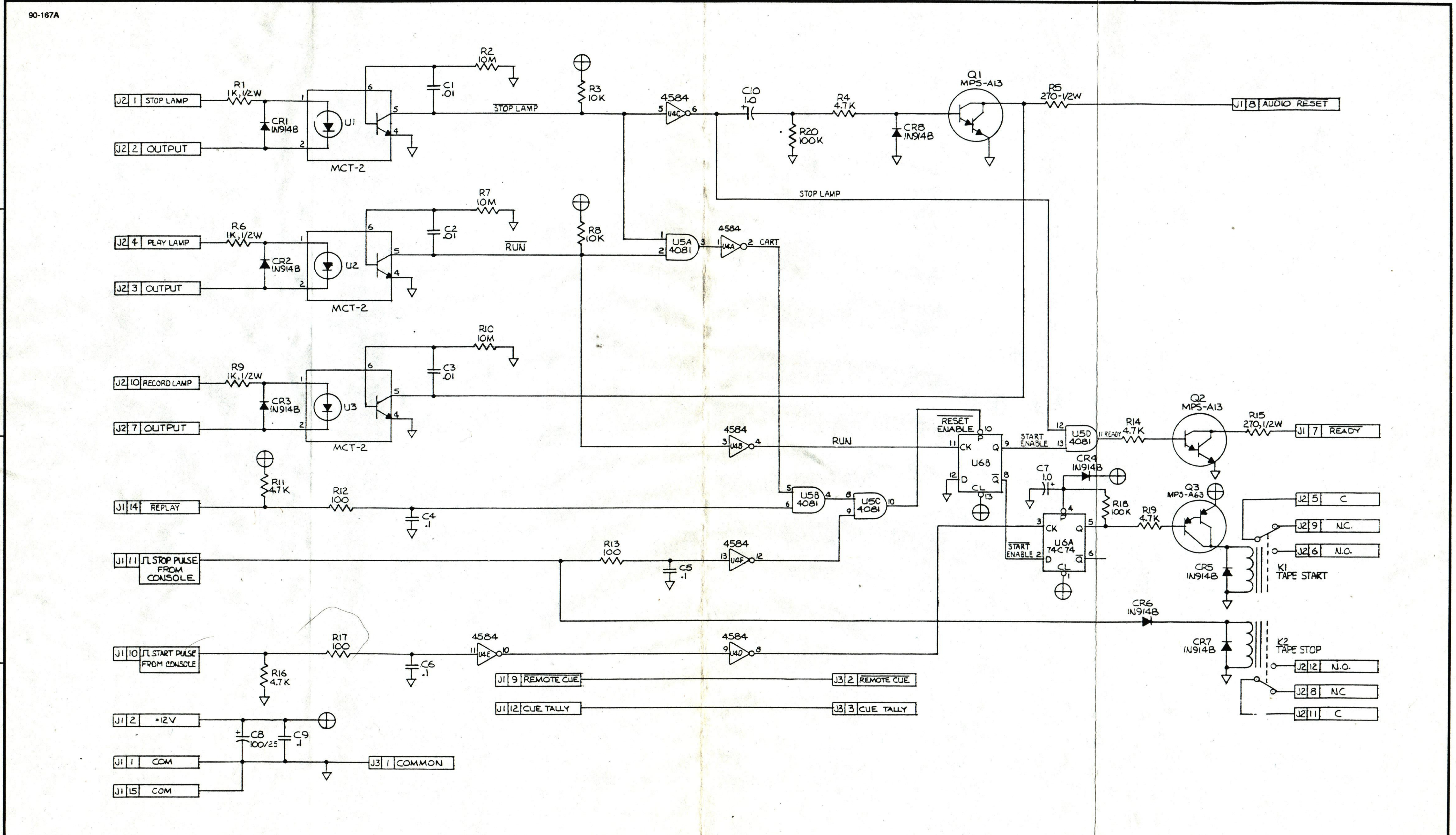
7.10.2 CI-2 CARTRIDGE INTERFACE



NOTE: FOR REPLAY, INSTALL JUMPER BETWEEN J1-14 AND J1-15.

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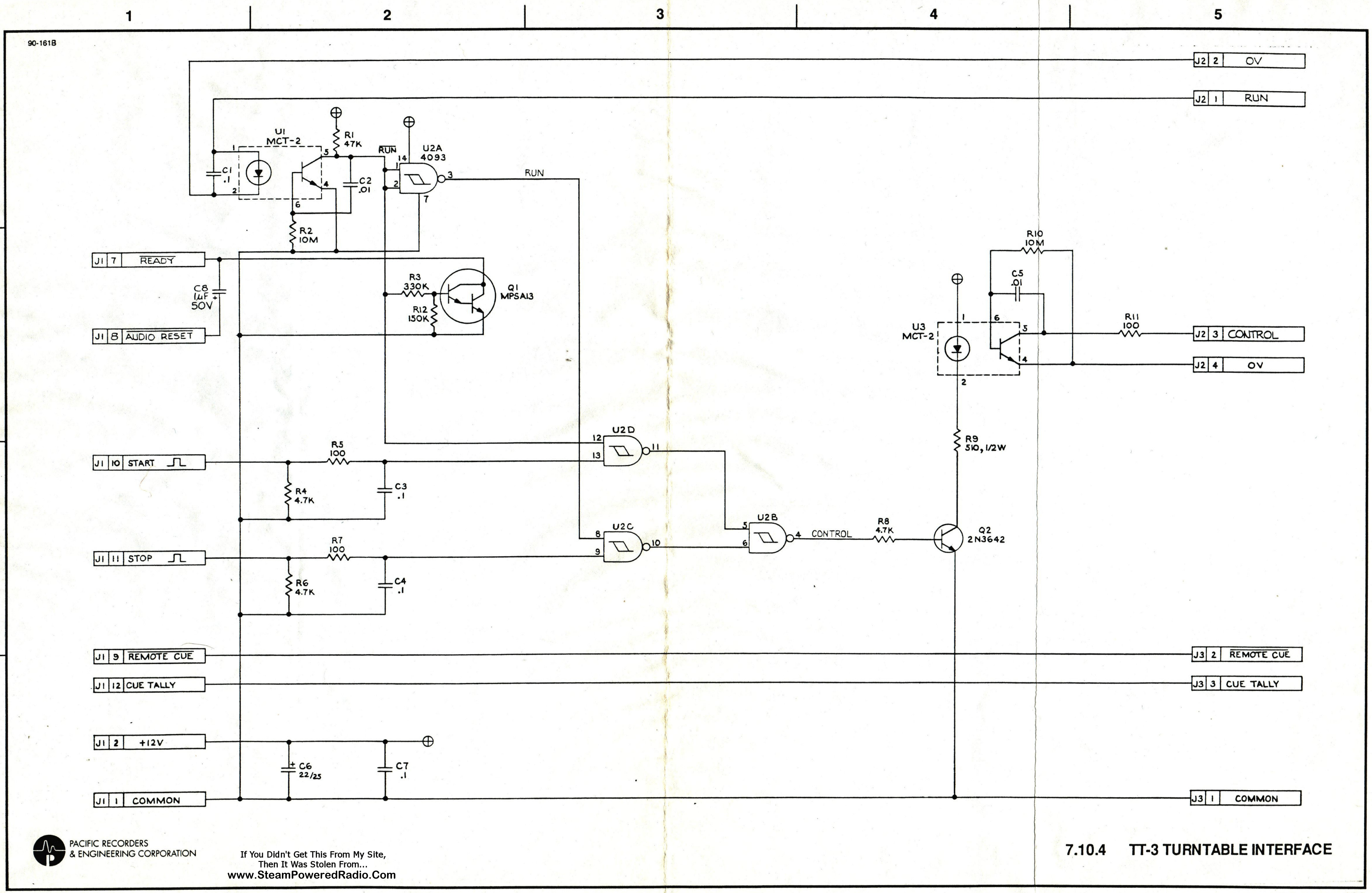
7.10.3 TI-2 TAPE INTERFACE



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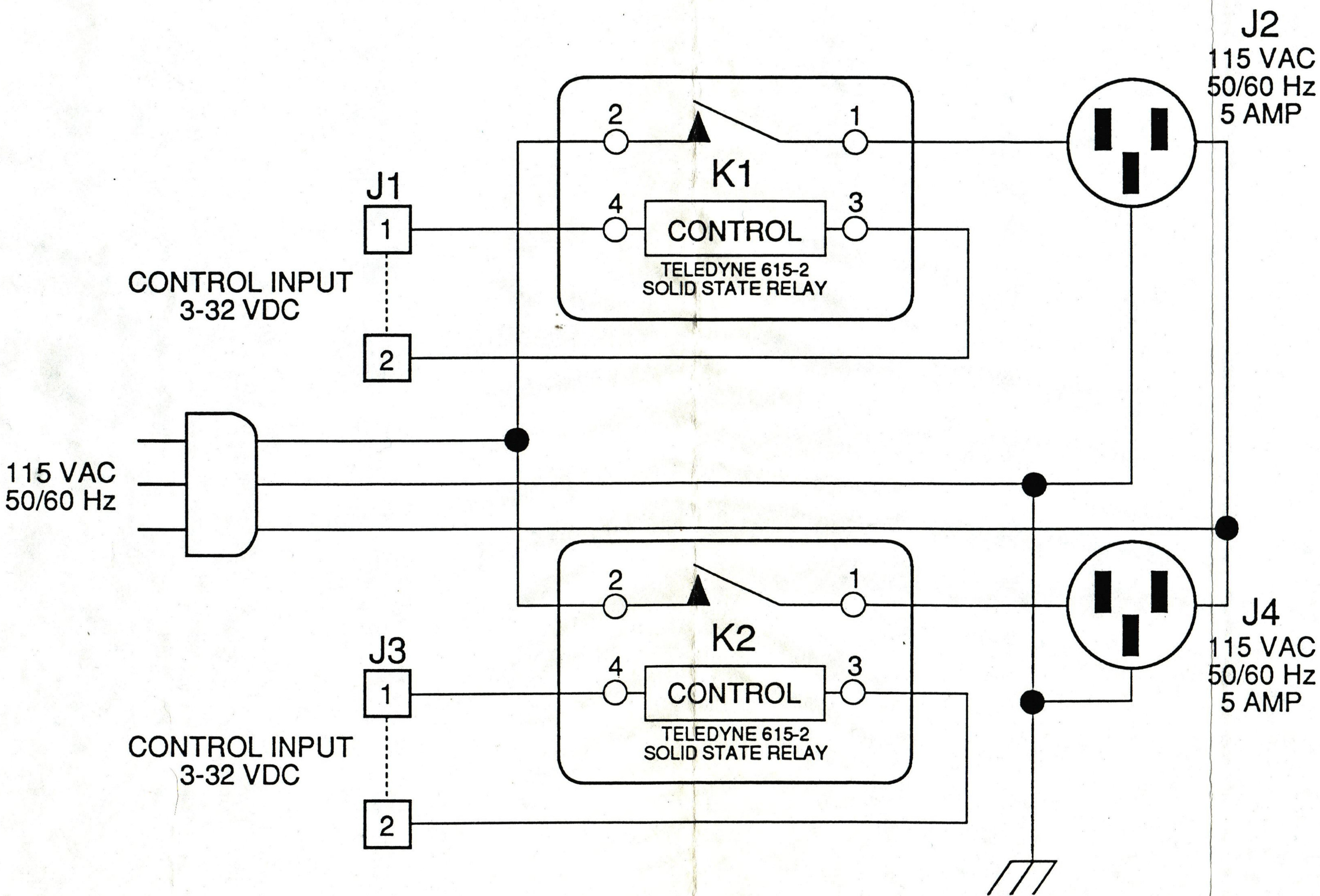
7.10.4 TT-3 TURNTABLE INTERFACE

99-143D



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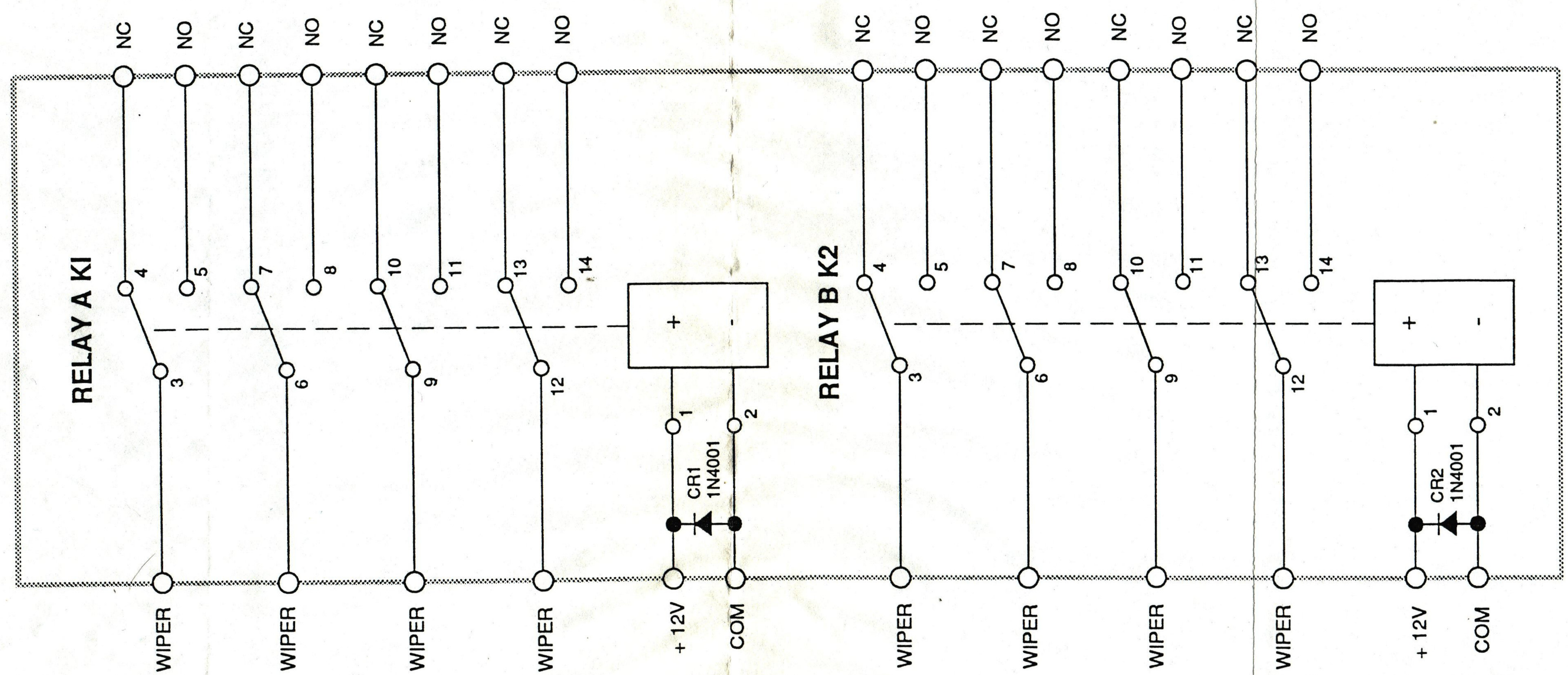
7.10.5 **WL-2 WARNING LIGHT INTERFACE**

92-419A



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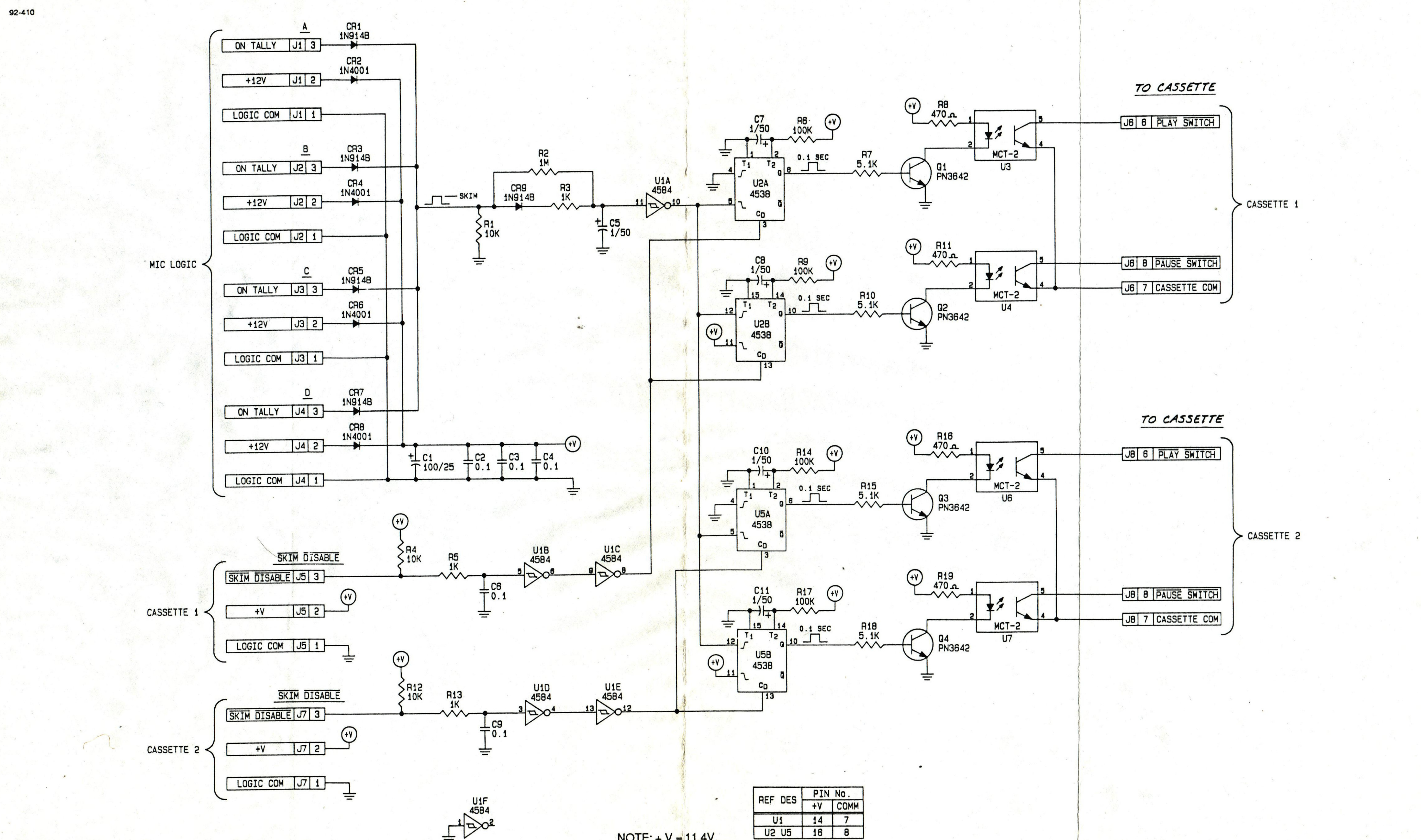
7.10.6 DUAL RELAY INTERFACE



REF DES	PIN No. +V	PIN No. COMM
U1	14	7
U2 U5	16	8

NOTE: + V = 11.4V.

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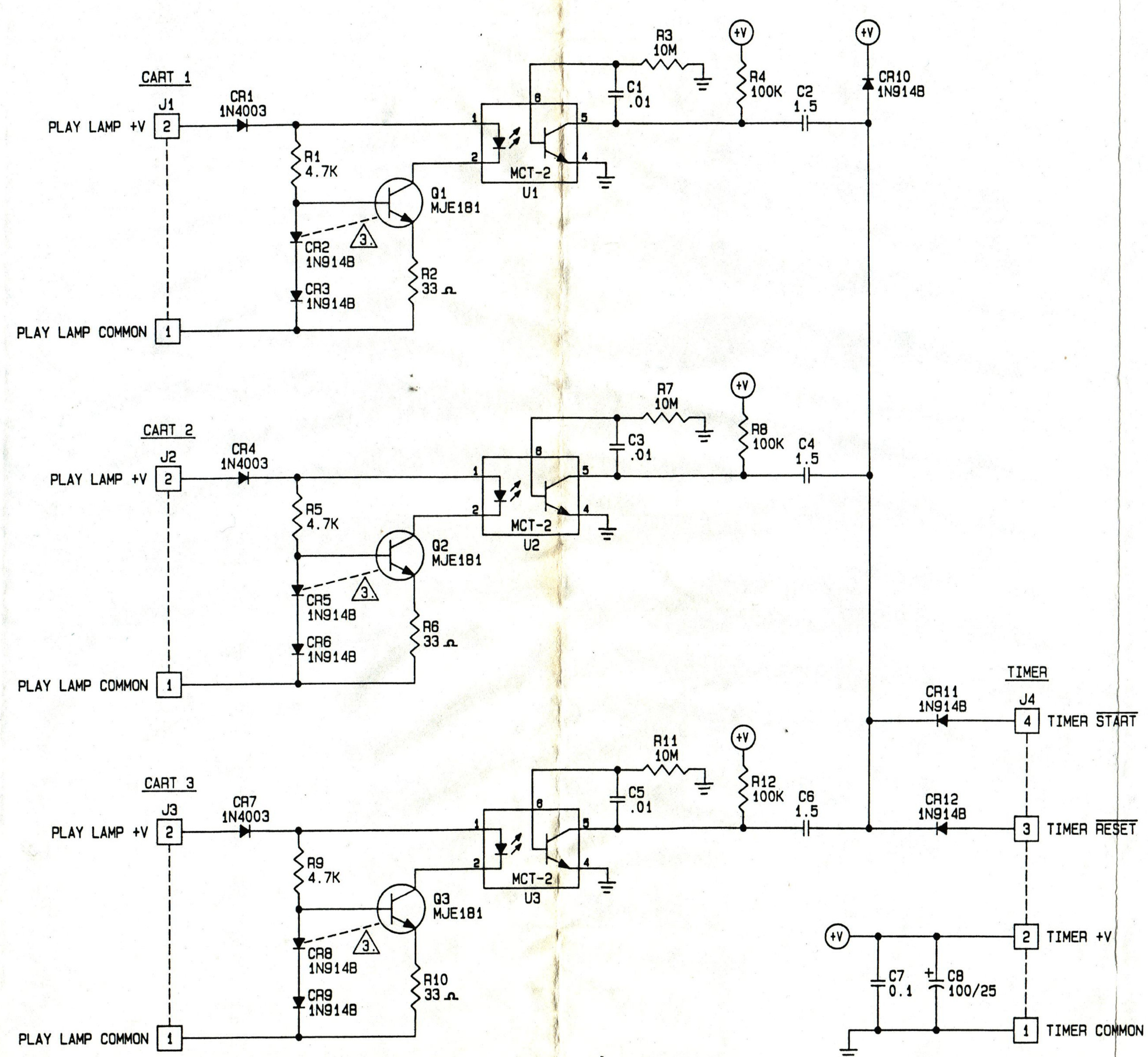
7.10.7 CASSETTE SKIMMER INTERFACE

92-409



NOTE: 3 THERMALLY COUPLED COMPONENTS.

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7.10.8 TIMER RESET INTERFACE

P1, P2: HOUSING: 15-PIN MALE MOLEX #03-06-2152 (PR&E #15-607)
TERMINALS: MALE MOLEX #02-06-2103 (PR&E #15-3)

CABLE TYPE: BELDEN #8458, 15-CONDUCTOR

P1	P2	WIRE COLOR
1	1	BLK
2	2	WHT
3	3	RED
4	4	GRN
5	5	ORG
6	6	BLU
7	7	WHT/BLK
8	8	RED/BLK
9	9	GRN/BLK
10	10	ORG/BLK
11	11	BLU/BLK
12	12	BLK/WHT
13	13	RED/WHT
14	14	GRN/WHT
15	15	BLU/WHT

P1, P2: HOUSING: 15-PIN MALE MOLEX #03-06-2152 (PR&E #15-607)
TERMINALS: MALE MOLEX #02-06-2103 (PR&E #15-3)

CABLE TYPE: BELDEN #8458, 15-CONDUCTOR

P1*	P2*	WIRE COLOR
1	1	BLK
2	2	WHT
3	3	RED
4	4	GRN
5	5	ORG
6	6	BLU
7	7	WHT/BLK
8	8	RED/BLK
9	9	GRN/BLK
10	10	ORG/BLK
11	11	BLU/BLK
12	12	BLK/WHT
13	13	RED/WHT

JUMPER		
P1 14	P1 15	WHT JUMPER

***NOTE**: P1 is the interface connector; P2 is the console connector.

P1, P2: HOUSING: 15-PIN MALE MOLEX #03-06-2152 (PR&E #15-607)
TERMINALS: MALE MOLEX #02-06-2103 (PR&E #15-3)

CABLE TYPE: BELDEN #8458, 4-CONDUCTOR

P1 IS CONNECTED AT THE CONSOLE END

P1	P2	WIRE COLOR
1	1	BLK
5	10	WHT
7	7	RED
8	8	GRN

7.10.11 #99-465 Console to CI-2 Interface Cable (for Studer A725, A727 and A730 CD Players)