TI-101
Telephone Interface
Manual

Contents

1. INTRODUCTION

The TI-101 telephone interface is intended to allow convenient, easy connection of audio equipment to telephone lines by fulfilling the following functions:

1.1 2 WIRE TO 4 WIRE CONVERSION

Most telephone systems are 2-wire bi-directional systems. That is to say, both parties in a conversation are carried on the same pair of wires. For most professional and industrial audio applications, it is more desirable to have the parties on two pairs so that one can adjust their level independently, avoid feedback due to multiple signal paths, and perform other signal processing functions.

1.2 LEVEL CONTROL

Send (host) level and return (caller) level are independently adjustable.

1.3 DYNAMIC RANGE CONTROL

The user may adjust a limiter on the send circuit and a compressor expander (to reduce telephone line noise) on the receive circuit.

1.4 EQUALIZATION OF RECEIVE SIGNAL

A boost or cut of 8 dB at 400 Hz and 2.5 Hz may be added to the caller's signal to increase intelligibility.

1.5 RECEIVE MUTE

The caller signal may be attenuated by a rear panel accessible contact closure. This is especially helpful for repressing the off-hook dial tone and dialing generated noises.

1.6 CONFERENCE LINK

Two TI-101's may be connected to separate telephone lines and then interconnected and patched to the user's mixer for three-way conversations.

2. FUNCTION DESCRIPTION

2.1 HYBRID (2 TO 4 WIRE CONVERSION)

Hybrids are necessary to separate the host from the caller line so that individual signal routing and processing may be applied to each one. To do so, the phone line that has a bi-directional two wire conversation between the caller and the host must have the host audio removed. This can effectively be done by taking the host audio at the source on its own two wire path and adding it to the bi-direction conversation audio path with the “two host lines” in opposite phase.Of course, the situation is really not quite this simple. The telephone line represents a complex impedance network (i.e. not just resistive, but capacitive and inductive as well), which phase shifts the send signal compared to the original. Phase correction must be applied to the send signal in order for it to be subtracted (nulled). The course null adjusts the hybrid for the variable resistive component of the telephone line; the low frequency (LF) and high frequency (HF) nulls adjust the phase correction network for the complex components of the line.

2.2 SEND EQUALIZATION

The FCC requires 18 dB of attenuation at 4 kHz and increasing attenuation at higher frequencies to avoid interference with telephone company equipment. This is accomplished by an elliptical low pass filter with a 3 kHz -3 dB point.

2.3 SEND LIMITING

The send signal is limited by two different methods.

The first is a voltage-controlled amplifier with user adjustable threshold, allowing less audible compression of the send dynamic range for greater intelligibility. The limit operation is indicated with the enabling of the “LIMIT” LED on the TI-101 front panel.

The second method available is hard limiting (clipping) with a fixed threshold of offering complete overdrive prevention of the phone line and hybrid circuit. Operation is indicated on the front panel when the “CLIP” LED is on.

2.4 RECEIVE COMPRESSION/EXPANSION

The caller signal input will trigger either the EXPANDER or the COMPRESSOR of this circuit depending on which side of the user set threshold the caller level falls on.

The VCA which accomplishes this operates at zero gain with no signal present. When the signal goes above the first threshold, the gain increases, and when the signal goes above a higher threshold still, the gain decreases. In this fashion background noise and other spurious low-level signals are passed at a lower level than the desired signal and loud signals are somewhat attenuated.

2.5 RECEIVE EQUALIZATION

A three-pole (18 dB per octave) 300 -3 kHz bandpass filter attenuates out-of-band signals, and improves the quality of the null. Gyrator circuits provide symmetrical boost and cut of 8 dB at 400 Hz and 2.5 kHz with a “Q” at full cut or boost of 9.

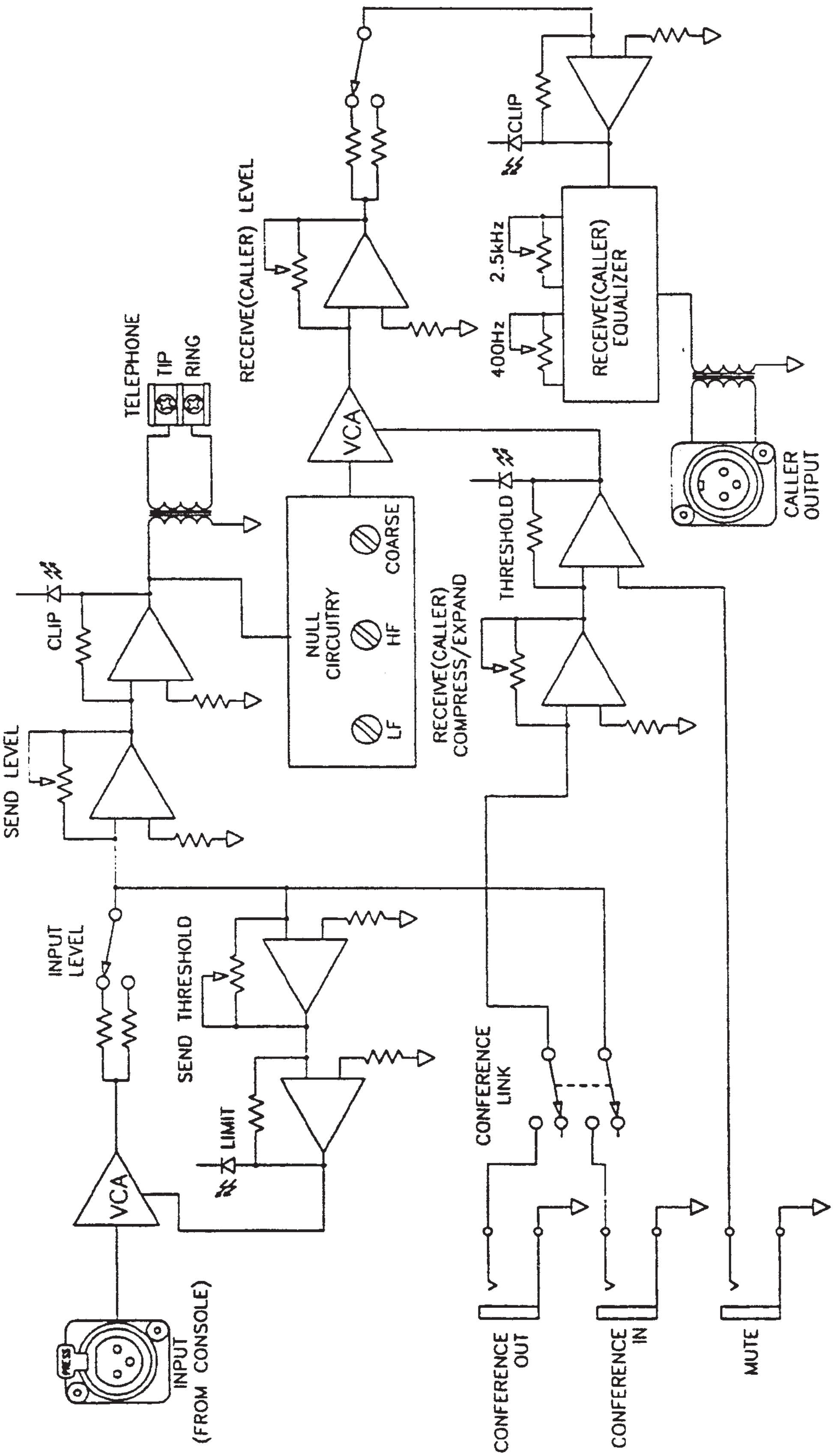
2.6 TELEPHONE LINE INTERFACE CIRCUIT

The telephone line is buffered by a transformer with good common mode rejection (another FCC requirement) and high DC breakdown voltage. In addition, this circuit includes a Metal Oxide Varistor to shunt AC or DC voltages of over 130 volts and a 2.2 µF 250V blocking capacitor to keep DC voltages off the transformer. Although the TI-101 is not type-approved by the FCC for direct connection to the telephone line, it is completely protected should this occur.

2.7 RECEIVE OUTPUT CIRCUIT

This consists of a current-boost amplifier (emitter follower) and transformer for DC isolation and common mode rejection.

3. TI-101 BLOCK DIAGRAM



4. CONNECTIONS TO THE TI-101

4.1 CALLER OUTPUT

The caller OUTPUT XLR connector feeds the caller's signal to your mixer. Pin 3 is high, pin 2 is low, and pin 1 is ground. For unbalanced operation, pin 2 may be grounded.

4.2 MUTE

A user supplied contact closure may be connected to this 1/4" jack. When contact is made from ring to tip, the caller's signal will be attenuated by a minimum of 20 dB.

4.3 CALLER OUTPUT LEVEL SWITCH

This switch sets the nominal output gain of the TI-101. Push the switch in for +8 dBm nominal and release the switch for -10 dBm nominal. The switch setting is, of course, dependent upon the requirements of the equipment being interfaced to the TI-101. In particular, professional mixing consoles will want to see +8 levels and semi-pro gear will want to see -10 levels.

4.4 CONFERENCE IN AND OUT

These connect to their opposite designated references on another TI-101 (that is to say "IN" to "OUT" and "OUT" to "IN"). The conference function is activated by the front panel "CONFERENCE LINK" switch. Refer to Section 7 for details.

4.5 INPUT (FROM CONSOLE)

The INPUT XLR connector accepts the send signal (the host) from your mixing console. This is the signal that will be sent to the caller on the other end of the phone line. Pin 3 is high, pin 2 is low, and pin 1 is ground. For unbalanced input signals, you should connect pin 2 to ground and apply the signal to pin 3.

NOTE.This signal should be only the talent's voice and not a full mix that includes the caller's signal (referred to as a "MIX-MINUS"). If you "loop" the caller's voice back down the line, you will cause an echo or perhaps an oscillation. Therefore, this input to the TI-101 should be fed from the mic pre-amp output patch or a separate console bus containing just the talent voice.

4.6 INPUT LEVEL SWITCH

The output level from your console or mixer can be matched to the input stage of the TI-101 with this switch. Press the switch in to the -10 dBm position if your mixer output level is nominally "low" (as the case with a semi-pro type of mixer). Alternately, use the +8 dBm positioning of the switch for professional mixers with high +4 or +8 dBm output bus levels.

4.7 TELEPHONE TIP-RING

Connections are made from the TI-101 to your phone line on this terminal strip through an FCC-approved coupler (QKT or other supplied by your telephone company). As mentioned above, the TI-101 is both transformer isolated and capacitively coupled to the phone line. Therefore, although “TIP” and “RING” are indicated, these connections may be reversed without any repercussions.

5. INSTALLATION

Connecting the TI-101 in parallel with the host phone line as described below is the least complex method to interface into your studio system.

Follow these steps to complete basic installation:

1) Split the host phone line in a "Y" with a user-supplied RJ-11 splitter. This modular phone jack splitter is available through Radio Shack (Part #279-357).

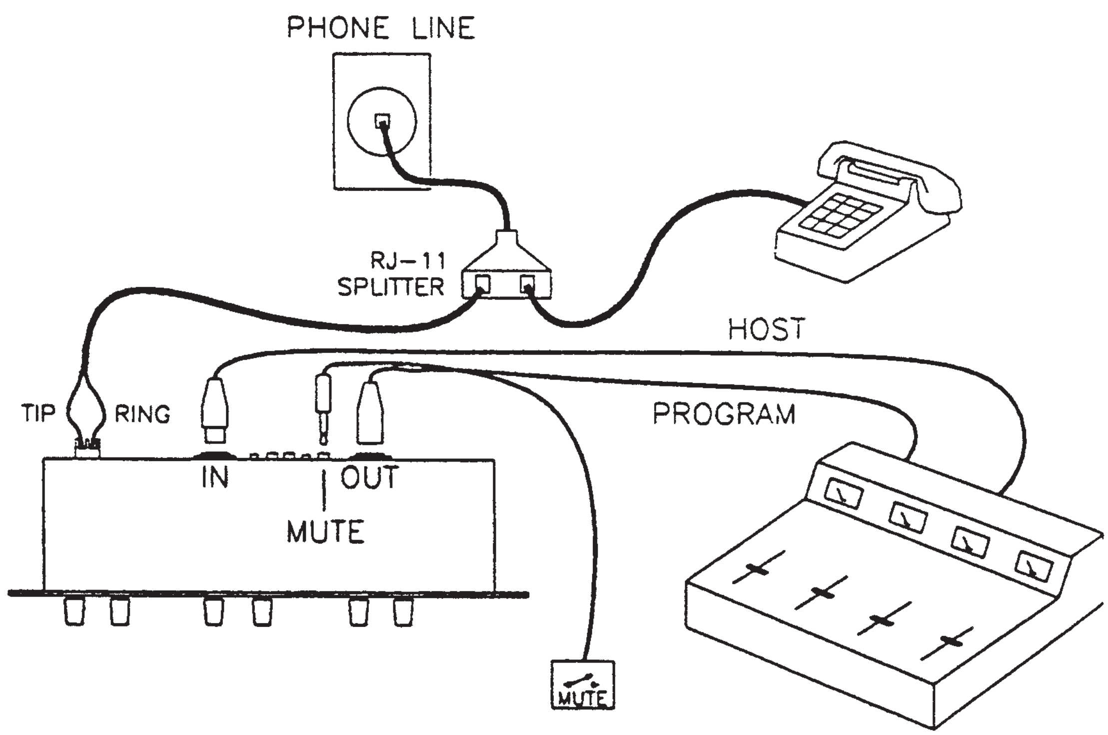
2) One side of the split will go to the studio phone.

3) The second side of the split will terminate at the terminal connector on the rear panel of the TI-101. Connection may be made using Radio Shack RJ-11, 12 foot extension with RJ-11 and spade lug ends (#279-364). Connect the GREEN and RED wires to the terminal connector.

4) A PROGRAM INPUT channel from the mixing console should connect to the OUTPUT XLR connector on the TI-101 rear panel.

5) The INPUT XLR connector on the TI-101 should receive HOST MIC audio by tapping off of the mic input channel at a pre-fader point, or by using a dedicated mix-minus bus from the console.

6) A remote contact closure connects to the MUTE jack on the rear panel.



6. FRONT PANEL SETUP OF THE TI-101

Once the TI-101 has been installed, the front panel controls may be adjusted using the following steps.

6.1 FOR STARTERS

First, rotate both equalization controls to their "12 o'clock" position. Then rotate the send limiter threshold control and the receive compressor/ expander threshold control to their full clockwise position. Doing so will effectively cancel these functions for the moment. Next, rotate the receive level control to its full counterclockwise (off) position.

6.2 SEND LEVEL ADJUST (HOST)

Now, call an assistant on the telephone. Make sure that your telephone mouthpiece is now either disconnected (unscrewed) or removed from the acoustic environment in which your microphone is located. Speak into your microphone and make the proper level adjustments on your console or mixer. Then, while speaking into the microphone at a normal voice level, advance the TI-101's send LEVEL control until the clip LED only occasionally flashes. This will give you the maximum send level to the phone line. Check with your assistant at the other end of the line and verify that you are being heard clearly and at a proper level.

6.3 RECEIVE LEVEL (CALLER)

Adjust your console or mixer so that you can monitor the return from the phone line (the TI-101's caller output). Have your assistant on the other end of the phone line speak to you at normal conversational volume. Turn up the receive level control on the TI-101 until a good receive signal is obtainable, using the 0 dB reference as a starting point. You should now be hearing your assistant's voice returning through your console or mixer clearly and without distortion.

6.4 HYBRID NULL ADJUST

Next you will adjust the null. Ask your assistant to place the telephone receiver in a quiet place and not to make any noise for a few moments. Speak into your microphone and adjust the front panel "COARSE NULL" potentiometer for the minimum signal (receive signal) in your monitors. Continue speaking into the microphone and adjust the "LF" and "HF" potentiometers. As you make these adjustments, be sure that the send or receive circuit clip lights are not coming on. The null degrades if either of these circuits are driven into clipping. The LIMIT and COMPRESS/ EXPAND THRESHOLD controls will be adjusted to help prevent this from happening as described in the next section.

6.5 SEND LIMITER

You are now ready to adjust the SEND LIMIT threshold control. With the telephone connection made, shout into your microphone. Make sure that you are not overloading the input to your mixer or clipping its output. As you shout, you should observe the send CLIP LED flashing. Adjust the send threshold control counterclockwise until the LED is no longer lit. You should now be at the optimum operating send level. Make sure no one changes these settings!

6.6 COMPRESS/EXPAND THRESHOLD

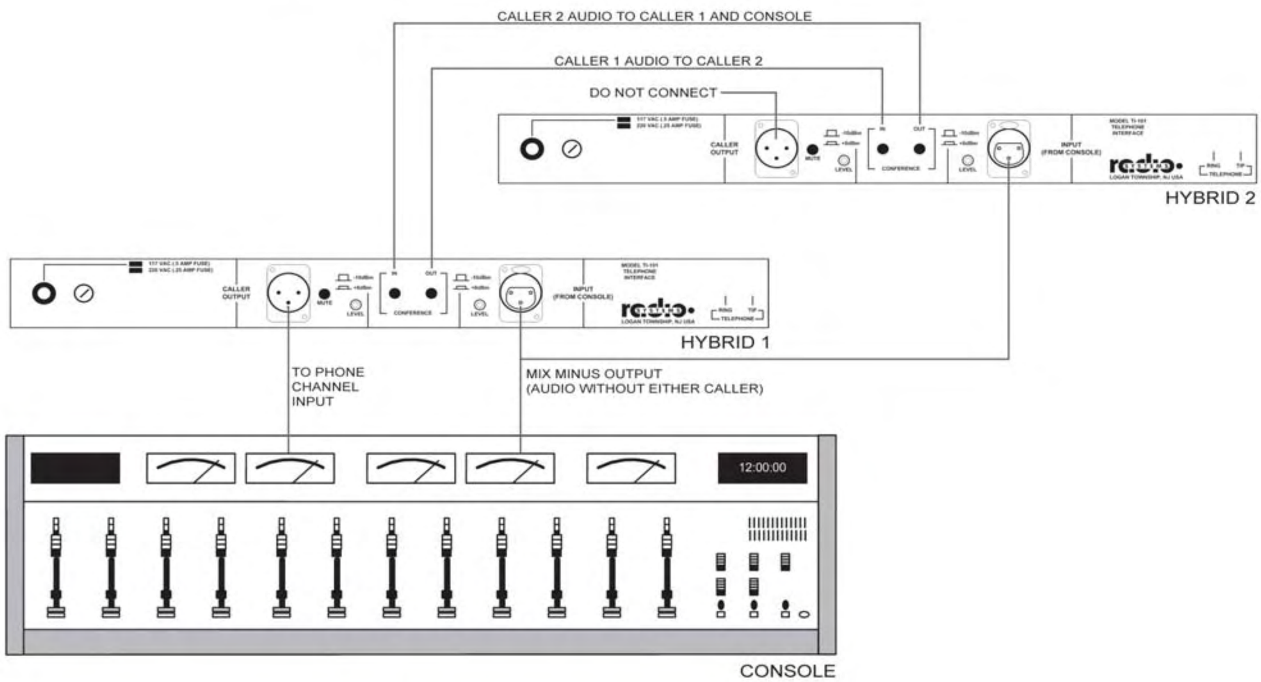
Now, ask your assistant to shout back at you. Adjust the COMPRESS/EXPAND THRESHOLD control in the counterclockwise direction. You will notice an increase in the volume level as you adjust this control. Compensate for this by decreasing the RECEIVE LEVEL control slightly. If you continue to adjust the COMPRESS/EXPAND THRESHOLD in the counterclockwise direction, you will start to notice the effects of the expander. Any of the noise on the phone line will be attenuated during the pauses in the conversation. Adjust the settings of this threshold control until you get the desired effect. Remember, that if you set this threshold control too low, then you run the risk of cutting off your caller if he/she speaks weakly. A little experimentation with different callers should help determine the optimum setting.

6.7 EQUALIZATION

Finally, have your assistant read something. As he does, adjust the equalization section of the TI-101. In general, some boost of high frequency (2.5 kHz) is always helpful for improving intelligibility. However, adjust both controls to suit your taste.

7. USING THE CONFERENCE MODE

CONFERENCE FEATURE
USE OF 2 TI-101'S TO CONFERENCE 2 CALLERS SIMULTANEOUSLY



NOTES: WIRE CONSOLE MIX MINUS TO INPUTS OF BOTH TI-101'S IN PARALLEL (MIX MINUS MUST NOT CONTAIN EITHER CALLER)

CONNECT BOTH CONFERENCE LINK LINES AS SHOWN

CONFERENCE LINK BUTTONS ON BOTH TI-101 FRONT PANELS MUST BE DEPRESSED

BOTH CALLERS MAY BE NULLED SEPARATELY

DO NOT CONNECT OUTPUT OF SECOND TI-101

8. SERVICE INFORMATION/SCHEMATICS

Radio Systems will service any of its products, no matter when it was manufactured or what condition it's in. However, no goods will be accepted without a Return Authorization Number. If we don't know its coming, we won't be prepared to make the necessary repairs.

Before sending anything to Radio Systems, call for an R/A number. Just ask, we'll gladly give you one. Call 856/467-8000 weekdays 8:30 a.m. to 6:00 p.m. EST.

8.1 IN WARRANTY REPAIRS

The TI-101 Telephone interface is covered by a limited warranty for a period of one year from the date of purchase. The Limited Warranty statement supplied with your unit spells out all the legal details and the generalities that follow are not intended to modify that warranty statement.

Contact Radio Systems at 856-467-8000 for a return authorization number.

Pack all items carefully and ship pre-paid, via UPS insured, to:

Radio Systems

601 Heron Drive

Logan Township, NJ 08085

Attn: R.A.#_____________

Enclose a note which includes your name, company, phone number, the serial number, return address (no box numbers), and a complete description of the problem.

8.2 OUT OF WARRANTY REPAIRS

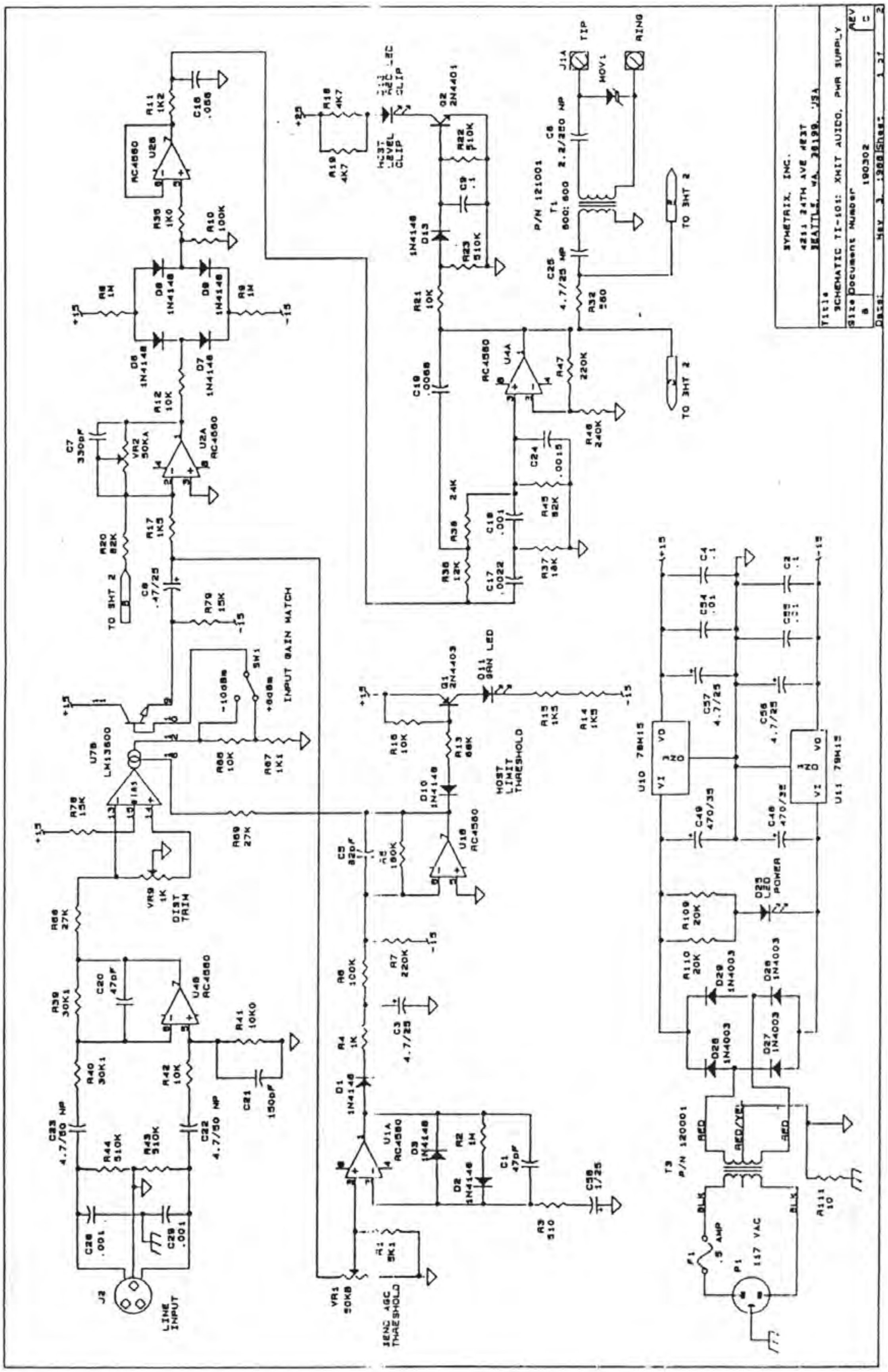
We'll gladly service any Radio Systems product at any time. If the warranty period is passed, you'll be billed for all necessary parts, labor, packaging materials, as well as any applicable freight charges.

Remember, you must call for an R/A number before you send the unit to Radio Systems.

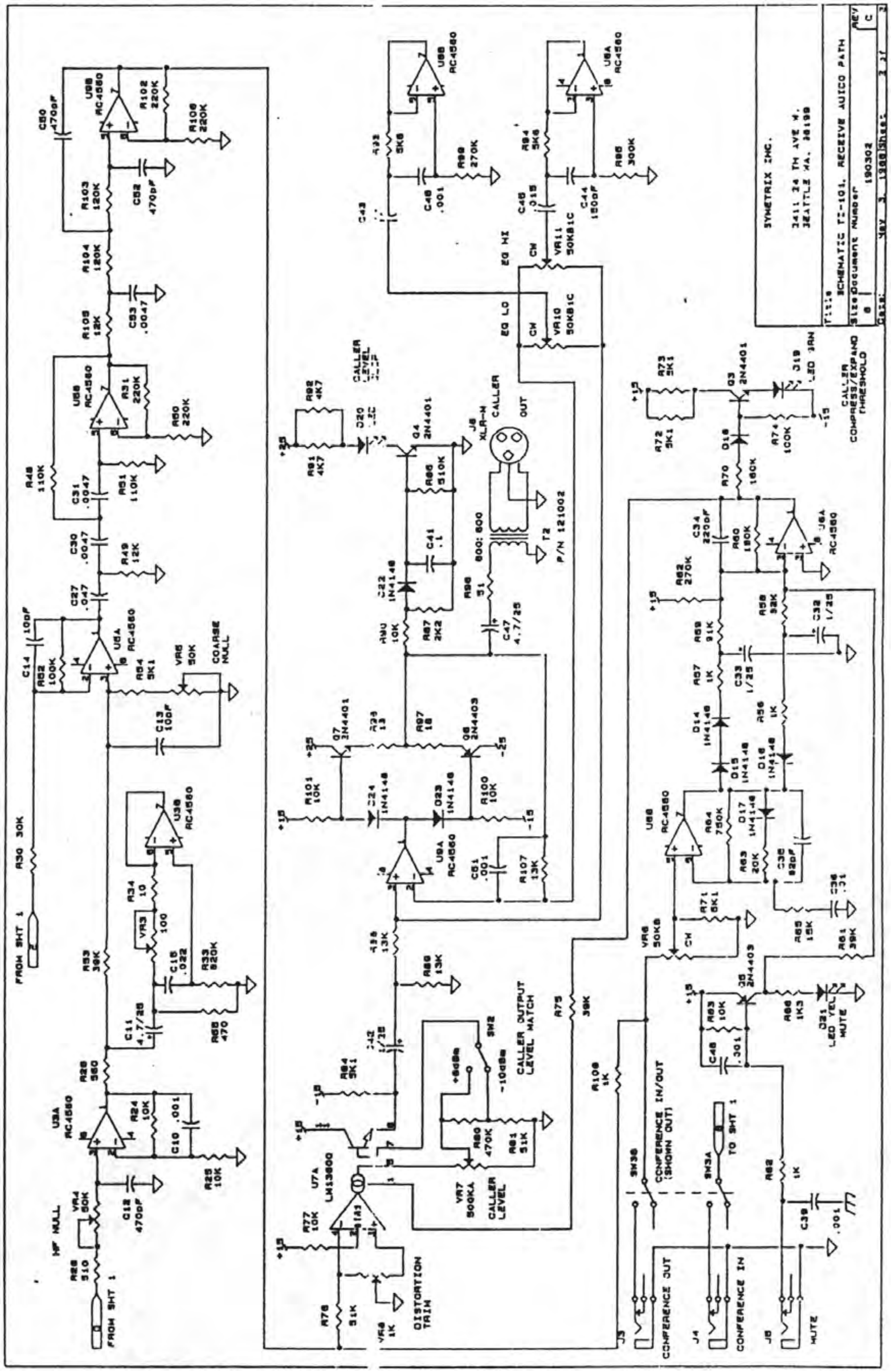
8.3 SCHEMATIC DIAGRAMS

The schematic diagrams are to be used by qualified service technicians only. No license to use this information for anything other than normal repairs is implied to given by the inclusion of proprietary information in the schematic diagrams.

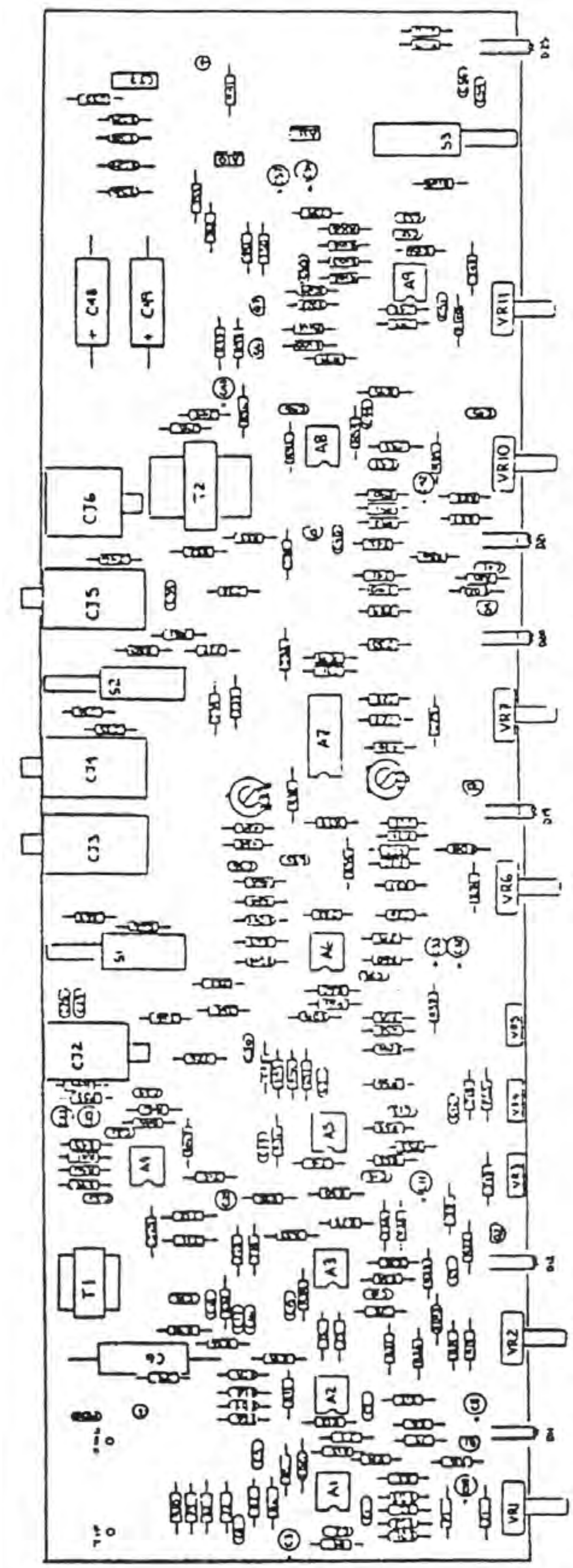
SCHEMATICS


SYMETRIX, INC.
SCHEMATIC TI-101: XMIT AUDIO, PWR SUPPLY
LINE INPUT
INPUT GAIN MATCH
HOST LIMIT THRESHOLD
TO SHT 2
117 VAC
TIP
RING

SCHEMATICS, CONT.



PARTS LAYOUT



PARTS LIST

REF-DES.............	DESC.........................	PART#	QTY.	UM
	TI-101 PCB DETAIL	9882	1	EA
C10;C28;C29;C39;C46;C51	CAP .001 UF CDISK 100V	1103	6	EA
C12;C50;C52	CAP 470PF DISK -	9913	3	EA
C13;C14	CAP 10PF CDISK	5862	2	EA
C15	CAP .022UF FILM 100V 10%	9322	1	EA
C16	CAP FILM .056 UF 50V	8920	1	EA
C17	CAP .0022UF +/-50V FILM	5863	1	EA
C18	CAP .001UF FILM 5% 50V	5861	1	EA
C19	CAP .0068UF FILM	9915	1	EA
C1;C20	CAP 47PF DISC +-10% 500V	5742	2	EA
C21;C44	CAP 150PF DISK	9910	2	EA
C22;C23;C25	CAP 4.7UF 35V NP	9917	3	EA
C24	CAP .0015UF 50V FILM	9083'	1	EA
C27	CAP .047 UF 50V FILM	8837	1	EA
C2;C4;C9;C40;C41;C43	CAP .1UF CDISK	9045	6	EA
C30;C31;C53	CAP .0047UF FILM	7587	3	EA
C32;C33;C42;C58	CAP 1UF TANTALUM 16V	9359	4	EA
C34	CAP 220PF DISK	9911	1	EA
C36;C54;C55	CAP .01 UF 50V DISC	2741	3	EA
C3;C47;C56;C57	CAP 4.7 UF 50V ELEC	8868	4	EA
C45	CAP .015UF FILM	9558	1	EA
C48;C49	CAP 470UF 35V AXIAL	9914	2	EA
C5;C35	CAP 82PF DISK	9909	2	EA
C6	CAP 2.2 UF 250V MYLAR AXIAL	9918	1	EA
C7	CAP 330PF DISK	9912	1	EA
C8	CAP .47UF TANTALUM	9916	1	EA
CJ2	CONNECTOR NC3FDH-B FEMALE	8402	1	EA
CJ3;CJ4;CJ5	1/4 IN JACK PCB MT	9884	3	EA
CJ6	CONNECTOR NC3MDH-B MALE	8401	1	EA
D11;D19	LED GREEN 3MM	9924	2	EA
D12;D20;D25	LED RED 3MM	9922	3	EA
D1;D2;D3;D6;D7;D8;D9;D10;D13;,D14;D15;D16;D17;D18;D22;D23;,D24	DIODE 1N4148	1012	17	EA
D21	LED YELLOW 3MM	9923	1	EA
D26;D27;D28;D29	DIODE 1N4004	1020	4	EA
J1-J57	RES 0.0 OHM 1/4 W	7133	57	EA
J7	HEADER 3 PIN .156	9883	1	EA
MOV1	SURGE ABSORBER	9921	1	EA
Q1;Q5;Q6	TRANSISTOR 2N4403	6119	3	EA
Q2;Q3;Q4;Q7	TRANSISTOR 2N4401	6118	4	EA
R103;R104	RES 120K 1/4W 5%	9903	2	EA
R11	RES 1.2K 1/4W 5%	5724	1	EA
R12;R16;R21;R24;R25;R66;R77;,R83;R90;R100;R101	RES 10K 1/4W 5%	1017	11	EA
R13	RES 68K 1/4W 5%	1123	1	EA
R14;R15;R17	RES 1.5K 1/4W 5%	2760	3	EA
R18;R19;R91;R92	RES 4.7K 1/4W 5%	1121	4	EA
R1;R54;R71;R72;R73;R84	RES 5.1K 1/4W 5%	9896	6	EA
R20;R45;R58	RES 82K 1/4W 5%	9900	3	EA
R22;R23;R43;R44;R85	RES 510K 1/4W 5%	9907	5	EA
R26;R32	RES 560 OHM 1/4W 5%	9895	2	EA
R2;R8;R9	RES 1 MEG 1/4W 5%	1047	3	EA

PARTS LIST (CONT.)

REF-DES.............	DESC...........................	PART#	QTY.	UM
R30	RES 30K 1/4W 5%	3907	1	EA
R33	RES 820K 1/4W 5%	9908	1	EA
R34;R111	RES 10 OHM 1/4W 5%	1019	2	EA
R36;R49;R105	RES 12K 1/4W 5%	9897	3	EA
R37	RES 18K 1/4W 5%-	9898	1	EA
R38	RES 24K 1/4W 5%	9899	1	EA
R39;R40	RES 30.1K 1/4W 1%	9573	2	EA
R3;R28	RES 510 OHM 1/4W 5%	9894	2	EA
R41;R42	RES 10K 1/4W 1%	2816	2	EA
R46	RES 240K 1/4W 5%	9906	1	EA
R48;R51	RES 110K 1/4W 5%	9902	2	EA
R4;R35;R56;R57;R82;R108	RES 1K 1/4W 1%	5872	6	EA
R53;R61;R75	RES 39K 1/4W 5%	1053	3	EA
R55	RES 470 OHM 1/4W 5%	1030	1	EA
R59	RES 91K 1/4W 5%	9901	1	EA
R5;R60	RES 180K 1/4W 5%	9905	2	EA
R62;R99	RES 270K 1/4W 5%	2738	2	EA
R63;R109;R110	RES 20K 1/4W 5%	1031	3	EA
R64	RES 750K 1/4W 1%	9057	1	EA
R65;R78;R79	RES 15K 1/4W 5%	8288	3	EA
R67	RES 1.1K 1/4W 1%	9625	1	EA
R68;R69	RES 27K 1/4W 5%	8799	2	EA
R6;R10;R52;R74	RES 100K 1/4W 1%	5876	4	EA
R70	RES 160K 1/4W 5%	9904	1	EA
R76;R81	RES 51K 1/4W 1%	9164	2	EA
R7;R31;R47;R50;R102;R106	RES 220K 1/4W 5%	2755	6	EA
R80	RES 470K 1/4W 5%	1122	1	EA
R86	RES 1.3K 1/4W 1%	9565	1	EA
R87	RES 2.2K 1/4W 5%	1016	1	EA
R88;R89;R107	RES 13K 1/4W 5%	8912	3	EA
R93;R94	RES 5.6K 1/4W 5%	1117	2	EA
R95	RES 300K 1/4W 5%	8909	1	EA
R96	RES 51 OHM 1/4W 5%	5728	1	EA
R97;R98	RES 18 OHM 1/4W 5%	9893	2	EA
REF;U1;U2;U3;U4;U5;U6;U8;U9	SOCKET 8 PIN DIP	1011	8	EA
REF;U7	SOCKET 16 PIN DIP	1039	1	EA
SW1;SW2;SW3	SWITCH SINGLE STATION "F"	9542	3	EA
T1	TRANSFORMER TELEPHONE	9925	1	EA
T2	TRANSFORMER T-34X	2842	1	EA
U10	VR L7815CV	3680	1	EA
U11	IC 79M15	9920	1	EA
U1;U2;U3;U4;U5;U6;U8;U9	IC 5532	1010	8	EA
U7	IC LM13600	9927	1	EA
VR10;VR11	POT 50K CNTR DETENT W/HDWRE	9887	2	EA
VR1;VR2;VR6	POT 50K PNL MT W/HARDWARE	9885	3	EA
VR3	POT 100 OHM VERT.	9889	1	EA
VR4;VR5	POT 50K VERT	9890	2	EA
VR7	POT 500K PLN MT W/HARDWARE	9888	1	EA
VR8;VR9	POT 1K TRIMMER	9892	2	EA

9. TI-101 SPECIFICATIONS

Input impedance	16.7K ohms (Electronically balanced)
Output Impedance	>600 ohms (Transformer Balanced)
Telephone Port Impedance	560 ohms (Transformer Isolated)
Nominal Input and Output Level Ranges	Switchable between -10 dBm and +8 dBm
CMRR	>4 dB at Input @ 1 kHz
Maximum Input Level	+21 dBm
Maximum Output Level	+20 dBm
Typical THD	.1%
Controls	Send Level, send limit, receive level receive compress/expand 400 Hz and 2.5 Hz. Equalization, conference link, coarse, low frequency and high frequency null adjust
Visual Indicators	LED's for indication of send, clip, send limit, receive, clip, receive compress/ expand, receive mute and power on
Frequency response	300 Hz to 3k Hz +/-3 dB (measured from telephone port to output port)
Typical Transhybrid loss	20 dB over the specified frequency band width
Connectors	3 pin XLR type for input and output ports, barrier terminal strip for telephone tip and ring, 1/4" phone jacks for external mute and conference interconnect cables
Physical Size	1 3/4" high, 19" wide, 6" deep (4.45 x 48.3 x 15.2 cm)
Shipping Weight	5 lbs (11.0 kg)
Power Requirements	60 Hz, 120 VAC standard, 50 Hz 220 VAC, upon request
Construction	All plated steel chassis

10. THEORY OF OPERATION

10.1 DERIVING A MIX MINUS SIGNAL

Mix-minus refers to one of two almost identical bus mixes. The first bus mix, which can be thought of as the main mix, contains all line inputs as its summed output. If a second bus mix contains all but one of the main mix inputs, and is identical to it in every other respect, it is known as a mix-minus.

In broadcasting situations, the mix-minus signal usually contains all but the receive audio (the caller). 'This might consist of a host and a guest along with a tape deck or two in a studio, and a guest calling in on the phone. In the studio, they are both hearing a full audio mix, but the caller who is also on the air is hearing a mix which contains all but his own voice.

10.2 WHY HAVE A MIX-MINUS SIGNAL?

Utilizing a mix-minus as a monitoring send for the remote location allows the remote source to hear all aspects of the mix without sending the source's output back down the line to itself. By doing this, two problems of considerable importance can easily be avoided. The first is feedback. This situation is very much like speaking into a microphone directly in front of the monitoring speaker. Without using a mix-minus approach, the caller source signal would almost inevitably form a regenerative audio loop resulting in a "howl". The second problem is echo in the circumstance that the call is satellite delivered. This is due to a delay which is inherent to the use of a satellite link in communications. Without a mix-minus, the caller will hear a slap back repeat of his/her own voice.

The following examples provide some of the interfacing methods for a variety of system configurations to derive and apply a mix-minus signal.

10.3 USING A SMALL ADD-ON MIXER

Any small mixer with mic inputs can be used to derive a mix-minus signal. The outboard mixer can be used as a sub-mixer for all the mics in the booth. If the mixing console used with the TI-101 doesn't have multiple bus capability, this technique must be used if programming is to include guests (and, therefore, the need for more than one studio mic).

All microphones used during a phone show are routed through the small mixer. Since the output from this mixer contains only signals from the studio mics and no caller audio, it is, by definition, a mix-minus signal. This output is fed to the TI-101 host audio input and to a line input of the air console. The TI-101's receive audio output is run to another line input of the air console, so the studio level and the caller level can be controlled separately.

10.4 USING THE CUE BUS OF AN EXISTING MIXING CONSOLE

Where there are enough extra line inputs available on the existing air console, use the following technique:

The input used for the studio mic is switched to the cue (or audition) bus. Assuming the caller is not put on cue, the cue bus output will be a mix-minus signal because it contains only the signals from the studio microphones. It can, therefore, be used to feed the host audio input on the TI-101.

In addition, the cue bus output is routed to the air bus through a separate line input on the console, so the studio mic can be heard on the air. To complete this technique, the TI-101's caller audio output is returned to the console through yet another line input to the air bus only, providing separate level controls for both the studio host and the caller.

11. COMMON QUESTIONS AND ANSWERS

Below are listed some of the most often asked questions about connecting the TI-101:

How do I seize the telephone line if I don't have a telephone set connected in parallel with the TI-101?

The telephone company uses the DC current flowing between tip and ring to sense off-hook condition. Since the TI-101 is AC coupled to the line, it will not seize the line. Basically, a substitute for the coil in the telephone set must be provided. This can be a holding coil such as the following:

Radio Systems Part No. 10170

Triad-Utrad Part No. TY-350P

These are available from many electronic distributors. In a pinch, one side of a signal transformer, either primary or secondary, with the other side unterminated, can be used. It should be approximately 1-2 Henries inductance 180 Ohms DC resistance. Also, a holding coil salvaged from an old telephone set could be used. The primary considerations are (1) that the coil hold the line when connected across tip and ring of the line; and (2) that a satisfactory low frequency null is achieved with the coil connected. In no case should a resistor be used to hold the line, since this would shift the impedance of the line such that the TI-101 cannot effectively null.

What should the signal be that I apply to the SEND INPUT of the TI-101?

The signal should be everything you want the caller (the person on the other end of the phone line) to hear except:

1) The caller's voice which has come from the "RECEIVE OUTPUT" of the TI-101. Including this would give the caller an objectionable echo of his own voice. This means you should not use the total mix from the output of your mixing board, since this contains the caller's voice. Rather, assign all mic signals to another bus, or patch off the mic preamp output to derive the send input signal.

2) Other callers' voices if the "conference" feature of the TI-101 is to be used. These are to be provided through the "CONFERENCE IN" and "CONFERENCE OUT" jacks on the back of the TI-101. Also, the input signal should be line level, nominally -10 or +8 dBm, depending on the setting of the rear panel "LEVEL" switch.

What should I do if I still get feedback (or too much of the local talent/send input signal from my monitor speakers after I null the TI-101?

This can be caused by several conditions:

1) A telephone mouthpiece connected to the line in the same acoustic space as the monitors. In general, all signals feeding the telephone line must go through the TI-101.

2) The null adjustment being incorrect. This can be because:
   a) The telephone line has been disconnected
   b) The TI-101 is no longer connected to the line for which it was nulled (i.e. it's been connected to a line with different impedance)

c) In general, anything has been changed about the telephone connection that would effect its impedance.

3) The gain from the microphone to the monitor speaker through the TI-101 is too high. Reasons for this include:

a) Too high send or receive levels in the TI-101, or too high receive EQ settings.

b) Too much EQ or too high level settings anywhere else in the mic to monitor signal chain.

c) Omni-directional microphones, microphones or monitor speaker with peaks in their frequency response. Note: the above three situations, like any other feedback, can sometimes be cured by the judicious application of narrow-band EQ of the monitor (not main signal).

4) The telephone and signal is being clipped. This degrades the null and is indicated by the flashing of the CLIP LED on the send sections of the TI-101.

How can I get more RECEIVE/OUTPUT level from the TI-101?

1) Change the rear panel switch from -10 dBm to +8 dBm

2) Decrease (turn counterclockwise) the RECEIVE EXPAND/COMPRESS THRESHOLD control.

3) Apply gain in the unit following the TI-101

What telephone load is the TI-101 designed to be connected to?

For effective nulling, the TI-101 should be connected to the same line for which it was nulled or possibly to sequentially numbered lines on the same exchange. This would rule out connecting it after rotary line selectors or key sets if the lines to which they are connected are of different

For Assy:

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