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MODEL TI-101 TELEPHONE INTERFACE INSTRUCTION MANUAL

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1.0 INTRODUCTION

THE TI-101 TELEPHONE INTERFACE IS INTENDED TO ALLOW CONVENIENT, EASY CONNECTION OF AUDIO EQUIPMENT TO TELEPHONE LINES BY FULFILLING THE FOLLOWING FUNCTIONS:

4-WIRE TO 2-WIRE CONVERSION - MOST TELEPHONE SYSTEMS ARE 2-WIRE (BIDIRECTIONAL) SYSTEMS; THAT IS, BOTH PARTIES IN A CONVERSATION ARE CARRIED ON THE SAME PAIR OF WIRES. FOR MOST PROFESSIONAL AND INDUSTRIAL AUDIO APPLICATIONS ONE WOULD LIKE TO HAVE THE PARTIES ON TWO PAIRS, SO THAT ONE CAN ADJUST THEIR LEVELS INDEPENDENTLY, AVOID FEEDBACK DUE TO MULTIPLE SIGNAL PATHS, AND PERFORM OTHER SIGNAL PROCESSING FUNCTIONS.

LEVEL CONTROL - SEND (HOST) LEVEL AND RETURN (CALLER) LEVEL ARE INDEPENDENTLY ADJUSTABLE.

DYNAMIC RANGE CONTROL - THE USER MAY ADJUST A LIMITER ON THE SEND CIRCUIT AND A COMPRESSOR/EXPANDER (TO REDUCE TELEPHONE LINE NOISE) ON THE RETURN CIRCUIT.

EQUALIZATION OF RECEIVE SIGNAL - 8 DB OF BOOST OR CUT AT 400 AND 2.5K HZ MAY BE ADDED TO THE CALLER'S SIGNAL TO INCREASE

INTELLIGIBILITY.

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 FREQUENCIES AND NOISES.

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.

2.0 THEORY OF OPERATION (PLEASE REFER TO THE BLOCK CIRCUIT DIAGRAM AT THE BACK OF THIS MANUAL) THE USER-ADJUSTABLE FUNCTIONS ARE INDICATED, AND THE BASIC FLOW OF SIGNALS IS SHOWN.

HYBRID (2 TO 4-WIRE) CONVERSION: ON ONE SIDE OF RESISTOR "R" WE HAVE ONLY THE SIGNAL WHICH HAS BEEN APPLIED TO THE "SEND INPUT"; ON THE OTHER SIDE WE HAVE THAT PLUS THE CALLER SIGNAL FROM THE TELEPHONE LINE. BY SUBTRACTING THE FORMER FROM THE LATTER WE GET THE CALLER SIGNAL ALONE. 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 COARSE 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.

SEND EQUALIZATION: THE FCC REQUIRES 18 DB OF ATTENUATION AT 4KHZ AND INCREASING ATTENUATION AT HIGHER FREQUENCIES TO AVOID INTERFERENCE WITH TELEPHONE COMPNAY EQUIPMENT. THIS IS ACCOMPLISHED BY AN ELLYPTICAL LOW-PASS FILTER WITH A 3KHZ -3DB POINT.

SEND LIMITING: THE SEND SIGNAL IS LIMITED BY TWO DIFFERENT METHODS: A VOLTAGE-CONTROLLED AMPLIFIER WITH USER-ADJUSTABLE THRESHOLD, AND A HARD LIMITER (CLIPPER) WITH FIXED THRESHOLD. THE "LIMIT" LED INDICATES THE OPERATION OF THE FORMER; THE LATTER OPERATES WHEN THE "CLIP" LED IS ON. THE FORMER ALLOWS LESS AUDIBLE COMPRESSION OF THE SEND DYNAMIC RANGE FOR GREATER INTELLIGIBILITY, WHILE THE LATTER PREVENTS OVERDRIVING THE PHONE LINE AND HYBRID CIRCUIT.

RECEIVE COMPRESSION/EXPANSION: THE CALLER SIGNAL ACTIVATES AN EXPANDER AT A LEVEL SET BY THE "COMPRESS/EXPAND THRESHOLD" CONTROL, AND AT A HIGHER LEVEL, IS COMPRESSED. THE VCA WHICH ACCOMPLISHES THIS OPERATES AT ZERO GAIN WITH NO SIGNAL PRESENT; WHEN THE SIGNAL GOES ABOVE THRESHOLD, THE GAIN INCREASES, AND WHEN THE SIGNAL GOES ABOVE A HIGHER THRESHOLD, THE GAIN AGAIN DECREASES. THUS BACKGROUND NOISE AND OTHER SPURIOUS LOW-LEVEL SIGNALS ARE PASSED AT A LOWER LEVEL THAN THE DESIRED SIGNAL, AND LOUD SIGNALS ARE SOMEWHAT ATTENUATED.

RECEIVE EQUALIZATION - A THREE-POLE (18DB PER OCTAVE) 300-3K HZ 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 AND 2.5K HZ, WITH A "Q" AT FULL CUT OR BOOST OF

2.5.

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 2.2UFD/250V BLOCKING CAPACITOR TO KEEP DC VOLTAGES OFF THE TRANSFORMER. THUS, ALTHOUGH THE TI-101 IS NOT TYPE-APPROVED BY THE FCC FOR DIRECT CONNECTION TO THE TELEPHONE LINE, IT IS COMLETELY PROTECTED SHOULD THIS OCCUR.

RECEIVE OUTPUT CIRCUIT: THIS CONSISTS OF A CURRENT-BOOST AMPLIFIER (EMITTER-FOLLOWER) AND TRANSFORMER FOR DC AND COMMON-MODE REJECTION.

3.0 CONNECTIONS TO THE TI-101

CALLER OUTPUT: THIS XLR CONNECTOR FEEDS THE CALLER'S SIGNAL BACK TO YOUR MIXER. PIN 3 IS HIGH, PIN 2 LOW, AND PIN 1 IS GROUND. FOR UNBALANCED OPERATION, PIN 2 MAY BE GROUNDED.

MUTE: THIS 1/4" CONNECTOR MAY BE CONNECTED TO A USER SUPPLIED CONTACT CLOSURE. WHEN CONTACT IS MADE FROM TIP TO RING THE CALLER SIGNAL OR ANY TELEPHONE SIGNAL WILL BE ATTENUATED AT LEAST 20DB.

CALLER OUTPUT LEVEL SWITCH: THIS SWITCH SETS THE NOMINAL OUTPUT GAIN OF THE TI-101. PUSH THE SWITCH IN FOR +8DBM NOMINAL AND RELEASE THE SWITCH FOR -10DBM 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.

CONFERENCE IN AND OUT: THESE CONNECT TO THEIR OPPOSITE NUMBER ON ANOTHER TI-101 (THAT IS TO SAY, "IN" TO "OUT" AND "OUT" TO "IN"). THE CONFERENCE FUNCTION IS ACTUATED BY THE FRONT PANEL "CONFERENCE LINK" SWITCH.

INPUT (FROM CONSOLE): THIS XLR CONNECTOR ACCEPTS THE SEND SIGNAL (THE HOST) FROM YOUR MIXING CONSOLE. THIS IS THE SIGNAL THAT WILL BE SENT TO YOUR 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. IMPORTANT NOTE: THIS SIGNAL SHOULD BE ONLY THE TALENT'S VOICE AND NOT A FULL MIX WHICH INCLUDES THE CALLER RETURN SIGNAL. IF YOU "LOOP" YOUR 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 A MIC PRE-AMP OUTPUT PATCH OR A SEPARATE CONSOLE BUSS CONTAINING JUST THE TALENT VOICE.

INPUT LEVEL SWITCH: THIS SWITCH MATCHESTHE OUTPUT LEVEL FROM YOU CONSOLE OR MIXER TO THE INPUT STAGE OF THE TI-101. PRESS THE SWITCH IN TO THE -10DBM POSITION IF YOUR MIXER OUTPUT LEVEL IS NOMINALLY"LOW" (AS MAY BE THE CASE WITH A SEMI-PRO TYPE OF MIXER). ALTERNATIVELY, USE THE +10DBM POSITIONING OF THE SWITCH FOR PROFESSIONALLY MIXERS WITH HIGH +4 OR +8DBM OUTPUT BUSS LEVELS.

TELEPHONE TIP-RING: THIS BINDING POST CONNECTS THE TI-101 TO YOUR PHONE LINE THROUGH A 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.

4.0 OPERATION OF THE TI-101

ONCE THE TI-101 HAS BEEN INSTALLED AS PER THE PRECEDING SECTION OF THIS MANUAL THE FRONT PANEL CONTROLS MAY BE ADJUSTED AS PER THE FOLLOWING INSTRUCTIONS.

FIRST ROTATE BOTH EQUALIZATION CONTROLS TO THEIR "12 O'CLOCK" POSITIONS. THEN ROTATE THE SEND LIMITER THRESHOLD POTENTIOMETER AND THE RECEIVE COMPRESSOR/EXPANDER THRESHOLD CONTROLS TO THEIR FULL CLOCKWISE POSITIONS. DOING SO WILL EFFECTIVELY CANCEL THESE FUNCTIONS FOR THE MOMENT. NEXT, ROTATE BOTH THE SEND AND RECEIVE LEVEL CONTROLS TO THEIR FULL COUNTERCLOCKWISE (OFF) POSITIONS. NOW CALL A PATIENT FRIEND OR ASSOCIATE ON THE TELEPHONE. MAKE SURE THAT YOUR TELEPHONE MOUTHPIECE IS NOW EITHER DISCONNECTED (UNSCREWED) OR REMOVED FROM THE SAME 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 NORMAL VOICE LEVEL ADVANCE THE TI-101'S SEND LEVEL CONTROL UNTIL THE CLIP LED ONLY OCCASIONALLY FLASHES. THIS WILL GIVE YOU MAXIMUM SEND LEVEL TO YOUR PHONE LINE. CHECK WITH YOUR FREIND AT THE OTHER END OF THE LINE AND VERIFY THAT YOU ARE BEING HEARD CLEARLY AND AT PROPER LEVEL.

NEXT, NOTE THE POSITIONING OF THE SEND LEVEL CONTROL AND THEN TURN THIS CONTROL DOWN TO IT'S FULL COUNTER-CLOCKWISE POSITION. THEN ADJUST YOUR CONSOLE OR MIXER SO THAT YOU CAN MONITOR THE RETURN FROM THE PHONE LINE (THE TI-101'S CALLER OUTPUT). HAVE YOUR FREIND 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 THE RECEIVE CLIP LIGHT FLASHES JUST OCCASIONALLY. YOU SHOULD NOW BE HEARING YOUR FRIENDS VOICE RETURNING THROUGH YOUR CONSOLE OR MIXER (REMEMBER TO KEEP YOUR TI-101 SEND LEVEL AT THE MAXIMUM COUNTER-CLOCKWISE (OFF) POSITION. YOU HAVE NOT YET ADJUSTED THE NULL, AND WITH THIS CONTROL UP YOU COULD POSSIBLY CAUSE FEEDBACK.). VERIFY THAT YOU ARE RECEIVING THE CALLERS' VOICE CLEARLY AND WITHOUT DISTORTION.

NEXT YOU WILL ADJUST THE NULL. ASK YOUR FRIEND TO PLACE HIS TELEPHONE RECEIVER IN A QUIET PLACE AND TO NOT MAKE ANY NOISE FOR A FEW MOMENTS. MEANWHILE SPEAK INTO YOUR MICROPHONE AND SLOWLY ADVANCE THE TI-101'S SEND LEVEL CONTROL. AS YOU DO SO 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" (LOW FREQUENCY AND HIGH FREQUENCY) POTENTIOMETERS ALSO. 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 BELOW.

ALTERNATIVE NULL ADJUSTMENT: WITH THE TELEPHONE CONNECTION MADE AS DESCRIBED ABOVE, INSTEAD OF CONNECTING THE OUTPUT OF YOUR MIXER OR CONSOLE TO THE TI-101 INSTEAD CONNECT AN AUDIO OSCILLATOR TO THE TI-101'S "INPUT FROM CONSOLE" CONNECTOR. ADJUST THE OSCILLATOR FOR AN OUTPUT FREQUENCY OF 2KHZ AT APPROXIMATELY 0 DBM. ADJUST THE COARSE NULL, THEN HF NULL UNTIL THE MINIMUM SIGNAL IS HEARD IN YOUR MONITORS. SET THE OSCILLATOR TO 600HZ, THEN ADJUST LF NULL FOR MINIMUM SIGNAL. DISCONNECT THE OSCILLATOR, CONNECT THE OUTPUT OF YOUR MIXER TO THE TI-101 AND THEN CONFIRM PROPER NULL ADJUSTMENT WITH THE VOICE METHOD. IN GENERAL, THE VOICE METHOD (OR USING WHATEVER SIGNAL WILL TYPICALLY BE APPLIED TO THE SEND) HAS PROVEN ITSELF TO BE VERY EFFECTIVE; THE OSCILLATOR METHOD SERVES TO PINPOINT TROUBLESOME FREQUENCIES.

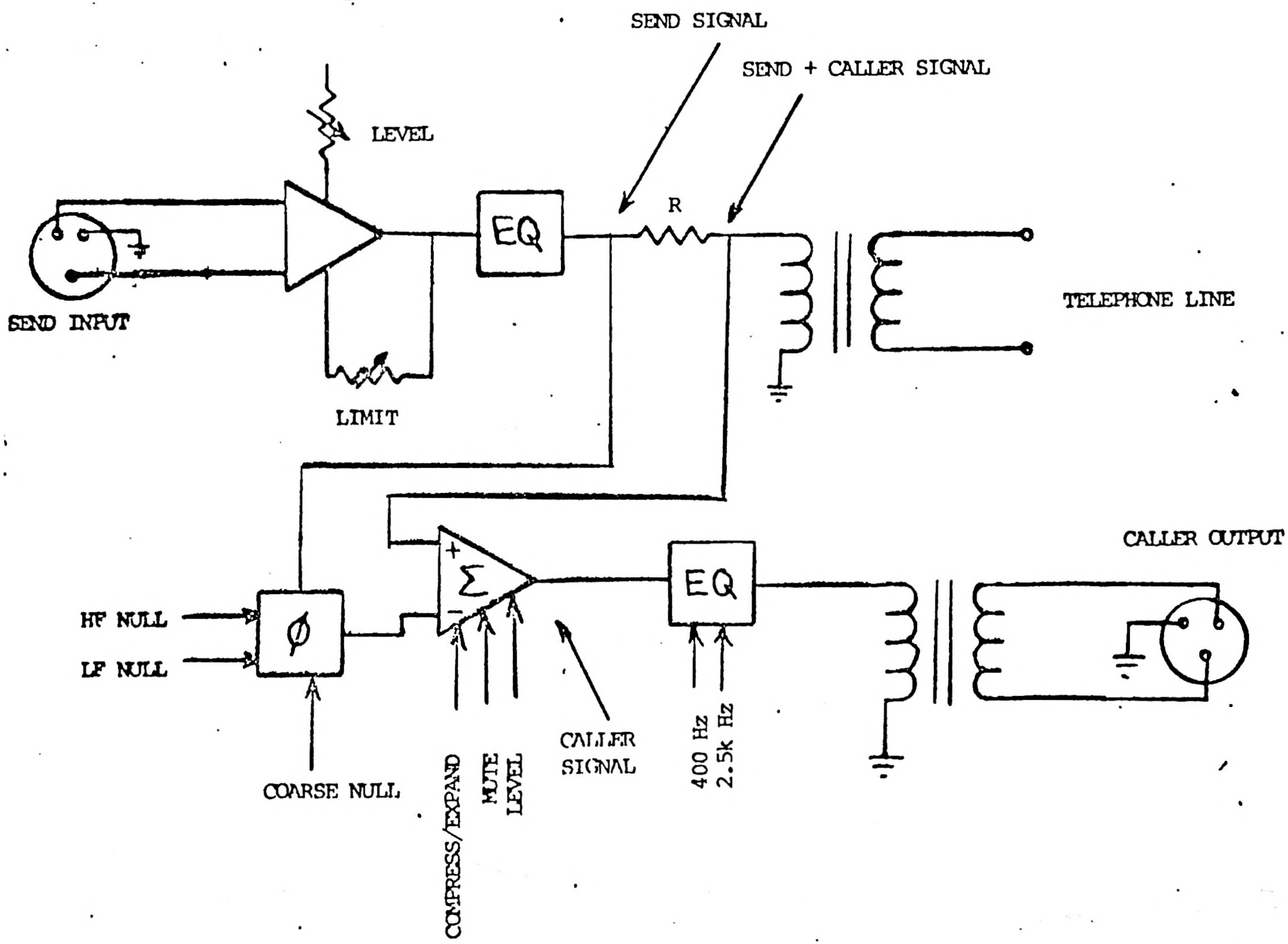
YOU ARE NOW READY TO ADJUST THE "SEND LIMIT" THRESHOLD CONTROL. WITH TELEPHONE CONNECTION MADE WITH YOUR FRIENDLY ASSISTANT AS DESCRIBED ABOVE, SHOUT INTO YOUR MICROPHONE. MAKE SURE THAT YOU'RE NOT OVERLOADING THE INPUT TO YOUR MIXER OR CLIPPING ITS' OUTPUT. AS YOU SHOUT YOU SHOULD BE OBSERVING THE SEND "CLIP" LED FLASHING. ADJUST THE SEND THRESHOLD CONTROL COUNTERCLOCKWISE UNTIL THE LED NO LONGER LIGHTS. YOU SHOULD NOW BE AT THE OPTIMUM OPERATING SEND LEVEL. MAKE SURE NO DUMBKOPFS OR INCOMPETANTS MESS WITH THESE SETTINGS.

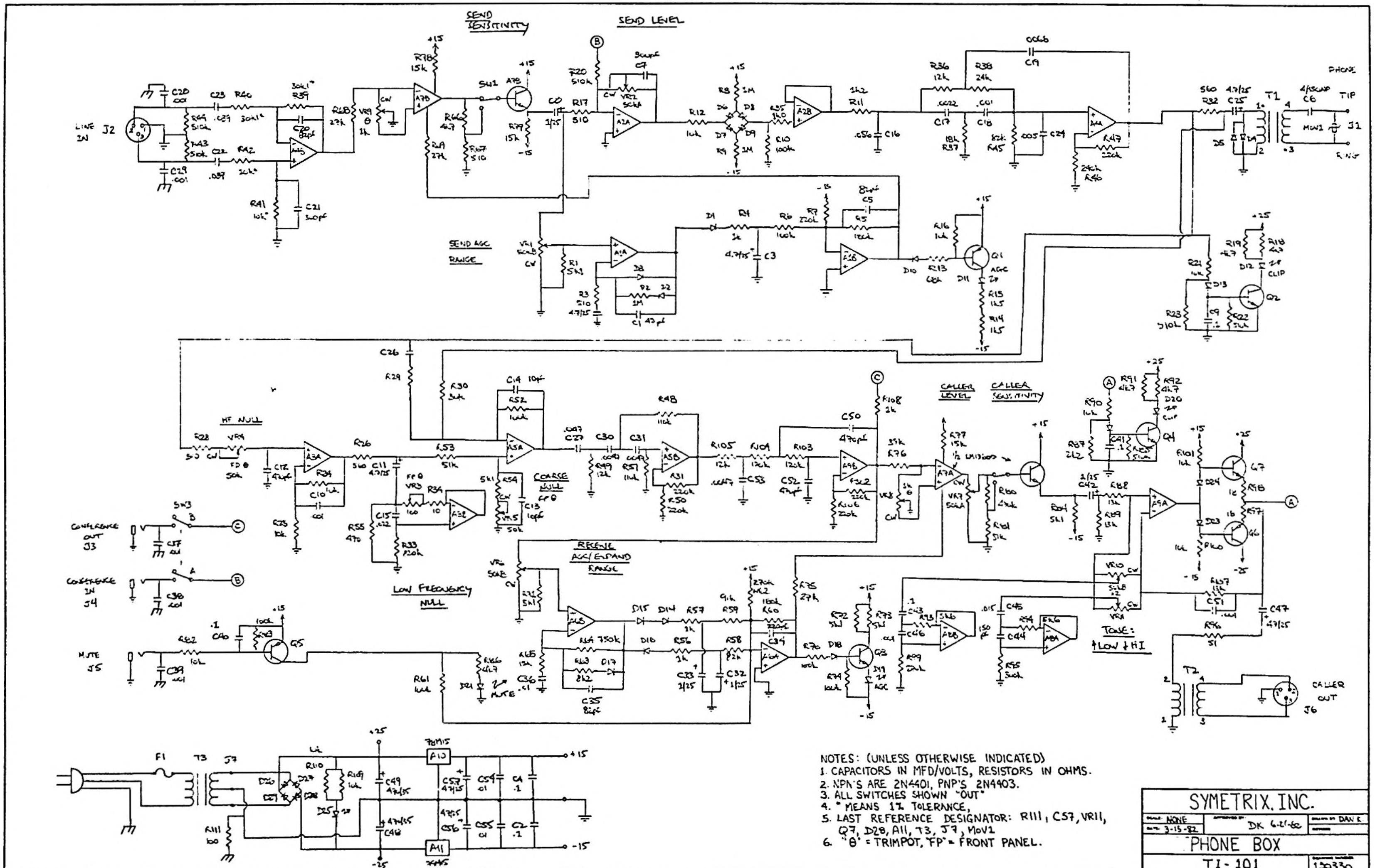
NOW ASK YOUR ASSISTANT ON THE OTHER END OF THE PHONE LINE TO SHOUT BACK AT YOU. ADJUST THE COMPRESS/EXPAND THRESHOLD CONTROL IN THE COUNTER-CLOCKWISE DIRECTION. YOU WILL NOTICE AN INCREASE IN VOLUME LEVEL AS YOU ADJUST THIS CONTROL. COMPENSATE FOR THIS BY DECREASEING THE RECEIVE LEVEL CONTROL SLIGHTLY. IF YOU CONTINUE TO ADJUST THE COMPRESS/EXPAND THRESHOLD IN THE COUNTERCLOCKWISE DIRECTION YOU WILL START TO NOTICE THE EFFECTS OF THE EXPANDER. ANY NOISE ON THE PHONE LINE WILL BE ATTENUATED DURING THE PAUSES IN THE CONVERSATION. PLAY AROUND WITH THE SETTING OF THIS THRESHOLD CONTROL UNTIL YOU GET THE DESIRED EFFECT. REMEMBER THAT IF YOU SET THIS THRESHOLD TOO LOW THEN YOU RUN THE RISK OF CUTTING OFF YOUR CALLER IF HE OR SHE SPEAKS VERY WEAKLY. A LITTLE EXPERIMENTATION WITH DIFFERENT CALLERS SHOULD HELP YOU TO DETERMINE THE OPTIMUM SETTING.

FINALLY, HAVE YOUR ASSITANT READ A MAGAZINE OR OTHER CONVENIENT SOURCE OF THE PRINTED WORD. AS HE OR SHE DOES SO YOU CAN EXPERIMENT WITH THE EQUALIZATION SECTION OF THE TI-101. IN GENERAL SOME BOOST OF THE HIGH FREQUENCY (2.5KHZ) IS ALWAYS HELPFUL FOR IMPROVING INTELLIGIBILITY. HOWEVER, ADJUST BOTH CONTROLS TO SUIT YOUR TASTE.

BEFORE YOU HANG UP, BE SURE TO THANK YOUR ASSISTANT FOR HIS OR HER PATIENCE. SOMETIMES THESE ADJUSTMENTS TAKE A LITTLE LONGER THAN YOU MIGHT EXPECT.

TI-101 BLOCK DIAGRAM





NOTES: (UNLESS OTHERWISE INDICATED)

1. CAPACITORS IN MFD/VOLTS, RESISTORS IN OHMS.
2. NPN'S ARE 2N4401, PNP'S 2N4403.
3. ALL SWITCHES SHOWN "OUT"
4. * MEANS 1% TOLERANCE,
5. LAST REFERENCE DESIGNATOR: R111, C57, VR11, Q7, D28, A11, T3, J7, MOV1
6. "θ" = TRIMPOT, "FP" = FRONT PANEL.

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TI-101 APPLICATION NOTE 1

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

1. "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 conditions. 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:

Microtran part no. T4415, T5415, T7410, T8410

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 a 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 will shift the impedance of the line such that the TI-101 cannot effectively null.

2. "What should the signal be that I apply to the "Send Input" of the TI-101?

This signal should be everything that you want the caller (the person at 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 that 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.

3. "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 a different impedance),
(c) in general, anything has been changed about the telephone connection which would affect 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 level settings on 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 speakers 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 send signal is being clipped. This degrades the null, and is indicated by the flashing of the "Clip" LED on the send section of the TI-101.

4. "How can I get more "receive output" level from the TI-101?"

(1) Change the rear-panel level switch from "-10 dBm" to "+8 dBm";
(2) Decrease (turn counter-clockwise) the "receive expand-compress threshold" control.
(3) Apply gain in the unit following the TI-101.

5. "What telephone loads 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 impedance. This is pretty much a "try it and see if it works" situation, given Ma Bell's unpredictability of line routing. The test load we use here at Symetrix is pictured below, to give an indication of the passive load for which the TI-101 is designed.