Gregg Laboratories

Technical Manual

teremix

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INTRODUCTION

The purpose of this manual is to fully acquaint the Telemix operator with the Telemix. This manual will describe what the Telemix can do and how to interface it to the telephone system. Please read this manual carefully before installation. It contains all that is needed to know about internal adjustments and external connections.

The Telemix is a fully-integrated studio telephone interface system. It contains its own power supplies, a two-channel interface for conferencing, a stereo headphone amplifier, balanced inputs and outputs, and a standard telephone connector termination.

AUDIO CONNECTIONS

The Telemix has a mono balanced bridging (20K) active input, accepting levels from -10dBm to +10dBm. It is shipped adjusted for a nominal Odbm, but the active input can be changed to accept any level, balanced or unbalanced. Feed into the rear panel connector any audio that must be transmitted through the telephone.

Audio ouput is 600 ohms balanced and completely adjustable with an internal control. As shipped, the Telemix will deliver a nominal Odbm with normal voice material, this level can be varied to a maximum output of +10dBm. For outputs below -30dBm, it is suggested that a loss pad be placed between the Telemix and the mixing console. The Telemix is designed to be fed back into the console via a remote or line fader, provided microphone audio only is supplied to callers.

AUDIO MIXING

If the audio path through the Telemix was though of as a "T", audio would appear at the three points of the "T". Send audio enters the "T" at the left end and is sent to the telephone through the bottom of the "T". The right side of the "T" is the output. Telephone audio enters the "T" from the bottom and comes out on the right. Also appearing at the output is a small amount of send audio. With proper adjustment of the internal electronic hybrid and judicious use of the "Caller-Override", the send audio appearing at the output is 15 to 30dB below that of the typical telephone call. The sophisticated switching, conditioning and bandpass filtering will alter the send audio as heard at the output of the Telemix. The filtering is necessary for compliance with registration requirements (FCC Part 68) and severely limits frequency response of send audio as heard by itself at the output. But when the Telemix is properly installed, send audio is fed to the console via its normal high-fidelity path at the same time it's feeding the Telemix. As long as the sources direct fader is open at the same time, the small amount of send audio returning from the Telemix is completely masked. The result is a truly simple yet transparent way to interface telephone audio with the console..

AUDIO CONTROL

There are several levels of control that can be exercised over the Telemix audio. Send audio can be muted before it mixes with the telephone. This is necessary when the Telemix is being used for incoming recordings only. This function is also used when the operator wishes to slave the Telemix send audio to the microphone logic, for example, when used as a hands-free telephone system. This send mute is normally always bypassed with an internal jumper, so it is necessary to move a jumper on an internal board if external control is desired.

The Telemix audio output is muted whenever the system is not in the "Air" mode. For a further discussion of this function, read on to the logic board description. It will suffice for the reader to know at this point that if the green "Air" button is not lit, audio will not appear at the Telemix output. This function can be remoted so it can be slaved to the station mic logic (anytime the mic is off, the Telemix functions as a hands-free answering device; when the mic is turned on, the Telemix goes to air).

REAR-PANEL CONNECTOR

The control options described above appear at pins 15 and 16 of the rear-panel "D" connector. Pins 1 and 2 are the headphone amplifier to these pins, via a 10 to 20dB loss pad. Pins 3 and 4 are the 20K balanced bridging input to the Telemix; pins 6 and 7 are the balanced output. The detailed layout appearing near the rear of this manual calls out all the pins but these comments will be appropriate: Pins 18 and 19 are the stereo unbalanced bridging inputs to the headphone switching and amplifier. For smooth transition between "Air" and "Off-Air" modes, the audio sent into these pins must be adjusted to match that obtained from the telephone when the Telemix is not in the "Air" mode, via external adjustment or an audio loss pad. Pins 20 thru 25 access the switch and lamp contacts of the two options switches which are labeled "Live" and "Swear". This last is usually connected to the dump function of a delay system; the other switch might be used for "delay-out". Other assignments can be made; by internal, this switch could be used as a "caller-wiper" so that a host can push down the caller without disconnecting him.

SEND LEVEL ADJUST

The unmarked control on the rear panel is the send level adjustment. It is set once, at installation, for correct audio level as heard by the caller. Do not succumb to the temptation to drive alot of audio into the telephone; this will simply degrade the trans-hybrid performance. Comments on processing of send and receive audio appear near the end of this manual.

OPENING THE BOX

There are a few reason to open the Telemix cover. The telephone needs to be plugged into the connector, the output level must be adjusted, and the hybrid will probably need an initial adjustment. The next few paragraphs detail how to carefully remove the cover to access the inside. Once inside the detailed circuit descriptions may be helpful for explanations of various elements. Please follow disassembly/assembly instructions carefully. It is most important to not the position of the headers and boards before unplugging anything.

STEP-BY-STEP DISASSEMBLY/ASSEMBLY INSTRUCTIONS

- 1. Unscrew the three front and three rear #6 Phillips-head screws on the bottom of the Telemix, then carefully turn the unit upright, holding it together. This frees the top cover from the bottom chassis
- 2. Very carefully lift the front of the unit up and back, watching various cables and wires to see how it opens.
- 3. The logic board on the bottom chassis contains the power supply, telephone interface, headphone amplifier and logic circuitry. A visual inspection should be made to see how cables are plugged in and to ensure their correct seating. It is preffered that the front cable that is attached from the control pot board to the logic board be left in place when the power is applied to the Telemix. Should it become necessary to unplug this cable when the power is on, we recommend that the cable not be left disconnected for more than five minutes at a time.
- 4. The relay board is one of the two upright circuit boards. It contains the 25-pair standard telephone plug (male). Carefully insert the pre-programmed 25-pair connector through the slot in the rear cover and fully seat the connector in the plug. This connector should be wired exactly as thoug it were a ten-button desk set because the Telemix appear just like an extention phone to the telco equipment.
- 5. The other vertical board contains the audio circuitry, described below. The multi-turn pot all by itself is the audio output level control; the pair of pots are the hybrid null. To adjust these controls it will be necessary to have service at the telco connector and audio fed in and out of the rear panel "D" connector. Being very careful of A.C wiring at all times, turn on the power and push the front panel Air switch so the green light turns on.
- 6. Monitor the audio output at pins 6 and 7 of the rear-panel "D" connector with a V.O.M. or Oscilliscope. From another telephone line, call up one of the lines that appear on the Telemix, punch up that line on the Telemix, and adjust the audio board output control for a desired level while the caller is holding down Touch Tone button #1 on the phone. (Factory set at +8dBm unterminated; which should normally correspond to approximately OdBm output with voice material, dependant on phone line quality.)

HYBRID NULL

- 7. Turn the front-panel "Caller-Override" control to full CCW (minimum).
- 8. Establish a typical connection using a phone line that appears on the Telemix and a second line on an adjacent telephone. Call one from the other.
- 9. Feed pink noise, or if unavailable a 1000Hz tone into the rear-panel "D" connector, pins 3 and 4, at the desired send level (or console OdB VU level).
- 10. Adjust the rear-panel send level control for normal level as heard in the telephone.
- 11. While watching the output audio with a VOM or oscilliscope, alternately adjust the two null controls on the audio board (close together), for a minimum reading. Continue this process until the hybrid is completely nulled. If the phone lines are average, the tone should null almost out of sight.
- 12. If using an oscillator, move the frequency back and forth to ensure there is not a "freak" null. With typical phone lines, there will be a slight dB change as the oscillator is moved up and down a couple of octaves from 1000Hz.

Once this null is found, it should not have to be readjusted.

This completes alignment of the audio board. At this point shut off power to the Telemix. If you wish to make the send-mute control options change mentioned earlier, refer to the mechanical drawing of the logic board just after the text in this manual. Move the vertical jumper just to the right of U3 into the horizontal position. Now whenever pin 15 on the rear-panel "D" connector is brought to ground the send audio is on; lifting pin 15 mutes it smoothly. If external logic is used to control this option, ensure the usual ground-loop rules are followed and be sure the external logic is capable of sinking a few mils current. (Most opto-isolators will be an ideal device.)

13. Make sure cables are plugged in correctly and firmly. Carefully replace the top cover and return the six Phillips-head screws and their washers to the holes in the bottom.

This completes the installation and adjustments. The Telemix is ready to go into operation, so a quick description of the front-panel controls is in order.

FRONT-PANEL CONTROLS

Control logic divides the lines into two groups, "Incoming" (5 lines) and "Conference" (2 lines). Lines are switched mutually-exclusively, just as it is done on a desk-set; punching up one line cancels any other in use. If it is desirable to hold two callers together, push two buttons in together and let go; they'll stay on-the-air talking to each other, until another line is selected or Off or Hold is requested. The two line groups are independent in terms of their logic. For this reason we suggest assigning dedicated unlisted telephone lines to conference 1 and 2. It is a lot easier to hold a newsmaker on the air while the incoming lines are switched around.

The Air switch is an alternate-action control. Live and Swear buttons have been discussed. The white Line swithces pick up their assigned telephone lines, and the Off and Hold buttons are obvious. The lamps within the Line swithces can be wired across the telco KSU lamp circuits to provide visual status of lines ringing, in hold, or on the air. The lamps inside the Off switches indicate "all clear".

CALLER-OVERRIDE

The front-panel caller-override control is one of the features which makes the Telemix unique. By varying the amount of caller-override, it is possible to adjust the balance between caller and talent from full two-way balanced-level all the way to a "quasi-speakerphone" effect where talent ducks caller. The appropriate adjustment point will depend on the type of sound the Telemix operator desires the show to have. Even in maximum override, timing is so fast and smooth that very little caller audio is upcut so the Telemix can never sound as bad as a speakerphone.

CONFERENCING

For maximum transfer of audio from caller to caller when conferencing, one call should be in the "Incoming" group and the other in the "Conferencing" group. The Telemix is designed for maximum power trnasger? from group to group, so best conferencing will happen when the newsmaker appears on a "Conference" line and the callers are switched around on the "Incoming" group. The Telemix has no bi-directional gain so that conferencing works best when the phone lines are of average to good quality and may not be totally predictable when the phone lines have alot of loss. But if the lines are good, as is the case in most cities where they are near the exchange, it is simple to tie two or more callers together. Be aware of the problems with Tip/Ring reversal when two buttons in the same group are pushed. Most Key Service Units (KSUs) solve this problem but should difficulty with conferencing be encountered, investigate this possiblity.

INTERCONNECT

Registration is being applied for. This involves certification by a testing authority and takes some time. Meanwhile there are several ways to make immediate connection to a telephone system. The first and most obvious is to plug the Telemix into a standard connector provided for a ten-button desk set. Other solutions involve the use of registered interface equipment. It will depend on the local telephone company. If the telephone company has any questions regarding registration, inform them that as soon as we receive a Registration number we will forward it to each Telemix customer for their records. But because this unit is in full compliance with Part 68 of the FCC Rules regarding interconnect equipment, there is no technical reason why immediate connection cannot be made. And if a KSU is purchased with the protected 400 cards, there will not be any legal interface problems since the KSU is the registered interface.

THEORY OF OPERATION: THE AUDIO BOARD

The design engineers of the Telemix have always believed the balanced level mixing method is the best approach to the telephone/studio interfacing. speakerphone idea provides good isolation, but it is a switch; it's not possible to hear both ends of a telephone conversation at the same time. Various balance-level schemes have been tried over the years with a limited degree of success, but to work at all properly, a great deal of finesse must be applied. Look at the audio board schematic in the rear of this manual. U1 is a differential input stage configured for a nominal +4dBm input. It feeds the rear-panel send pot through a first-order high-pass filter which helps shape response. U1 and U2 create a highly accurate band-pass filter which complies with FCC Rules Part 68 governing interconnect requirements. Q4 is the send mute discussed earlier. And at the far right U3, half of a dual OTA, which serves as a send noise-gate. During pauses in the send audio, this gate is biased down 10dB, in order that paper shuffling, room noise, etc. are removed from the mix. This is especially valuable conferencing. This circuit is commanded by U4 and its periphery, and the depth of the expander is decided by the value of R30.

U5 makes up the electronic hybrid. It is a differential amplifier which is adjusted for minimum output when a telephone line is connected and send audio is present. The mix coil is connected to J4-19 to help form one half of the differential ladder. The reactive components on the other side of the ladder are tuned to approximate the reactance of a typical telephone line. Both pots are adjusted alternately for the best null. It will be obvious that the more this half of the ladder duplicates phone line reactance, the better the null. We have chosen an approach which requires a minimum number of parts and yet provides a broad and predictable null.

Q3 functions as a receive audio mute and a click-mute. It's triggered by a timing chip on the logic board so that whenever a line relay is switching a telephone line in and out, receive audio is muted until the switch transient decays.

The other half of U3, the dual OTA, is the heart of the caller-override circuit and is driven by U6. R42 is the output level control and U7 the active output stage.

When the hybrid null is properly adjusted, assuming fair to good phone lines, audio at this output stage will consist primarily of telephone, with very little send audio.

It will be of interest to note that even in cases of poor hybrid rejection due to poor telephone lines, trans-hybrid loss will be greater by the amount of caller-override employed. For example, a setting of 10dB caller-override will dynamically reduce hybrid leakage by that amount whenever send audio is present.

LOGIC CIRCUIT DESCRIPTION

The Telemix control functions are implemented using the 74CXX group logic integrated circuits. The pin-outs are identical to those for the standard TTL gates, and the internal operation is also equivalent. However, the power supply and in/out voltages are different. The 74CXX IC's use a 15 volt power supply, with a logic low being represented by 4 volts or less, and logic high being 15 volts or greater. CMOS IC's have the advantage of being less affected by RF fields and power line glitches, and consuming less power from a supply which may be common to audio and logic circuits. While the 74CXX group is improved from standard 4000 series CMOS with regard to static sensitivity, care should be taken to prevent static discharge to the IC's during troubleshooting and parts replacement.

The power supply transformer and bridge rectifier deliver approximately +24 and -24 volts to the regulators, which are the common three terminal IC type. The unregulated +24 is used to power the line relays and indicator lamps for Air and Off pushbuttons, therefore conserving regulated power.

Each of the front-panel buttons is connected to a pull-up resistor and the input of either U15 or U16. U15 and U16 are Schmitt trigger gates, which buffer and invert the pushbutton inputs. The outputs of the Schmitt trigger gates are connected to the inputs of "D" type latches U1 and U2. These latches transfer the input state to the outputs upon a high to low transition on the clock input, pin 9.

When any button except Air is actuated, capacitors C1-C9 transmit a brief pulse to the appropriate inputs of U17. These positive going pulses are inverted to trigger the 556 timers (U4 and U5). The pulses are too short in duration to be seen on a meter, but may be observed on a oscilliscope as a very brief spike.

The timers outputs are inverted by U18 to drive the "D" latch clock inputs. The timer output duration is approximately 20mx. Therefore, 20ms after any button is pressed, the latches "take a picture" of their inputs. By doing this, contact debouncing and line selection memory is accomplished.

The Off button simply clocks the latches. Wit no other buttons pressed, the lines are switched off.

The Hold buttons trigger the other 556 timer sections, which are connected for a 300ms output pulse. Immediately, when the timer is triggered, a high appears on the appropriate input of U18. Therefore, U18's output provides a low to one of each of the two inputs on NAND gates U19 and U20. Consequently, the "A" leads, which are switched by Q1-Q7, are turned off. 300ms later, the latches, U1 and U2, are clocked, causing the line relays to drop. When the key system sees an "A" lead drop followed by a Tip/Ring drop, a hold condition is initiated.

U6 a 555 timer provides the line-select mute. When any button is actuated, this timer is triggered. Its output, of approximately 250ms duration, is used in the audio board to mute the system output briefly to prevent the line-select click from reaching the output.

One section of U3, a standard flip-flop, is used to provide the Air on-off latching function. The Air button is coupled through a section of U14 to the U3 clock input. Upon clocking, U3 changes state (toggles). Remote operation of the Air function is provided by using the direct set and clear inputs of the flip-flop. The scheme used allows the front panel Air button to be used regardless of the state of the remote input, and insures that the flip-flop can never get out of synch iwth the remote input. That is, the flip-flop is always set when the remote Air command is turned on, and always reset (cleared) when the remote command is turned off.

The other section of U3 is used to provide remote control of the send audio on-off. If it is desired to have the audio feed to the phone remotely controlled, the small jumper on the logic board near U3 must be moved to connect U3 pin 10 to the proper U14 gate input. When this function is desired, normally it is connected to the studio monitor mute system. Therefore, when the announce microphone is turned off, feed from it into the phone is also switched off.

Actuation of the Air button and line buttons control the headphone split function and the audio output on-off through U21 as set forth below:

AIR ON	&	LINE SELECTED	=	AUDIO ON	. &	HEADPHONE NOT SPLIT
AIR ON	&	LINE NOT SELECTED	=	AUDIO OFF	&	HEADPHONE NOT SPLIT
AIR OFF	&	LINE SELECTED	=	AUDIO OFF	&	HEADPHONE SPLIT
AIR OFF	&	LINE NOT SELECTED	=	AUDIO OFF	&	HEADPHONE NOT SPLIT

The various gates of U21 are interconnected to implement the above table.

The audio switching for the headphone audio is accomplished by JFET gate U10. When the headphone is not split, U10 simply passes the monitor audio through from the inputs to the amplifier. When the headphone is split, the balance control is switched in, and the telephone audio is switched into the right channel of the amplifier. If desired, the relative gains may be adjusted by changing the appropriate resistors.

The line button lamps are powered directly from the telephone key system lamp leads.

The headphone amplifier is a standard application of a common amplifier IC, the National LM 378. If it is desired to use an external amplifier, the 378 may be removed from its socket, and jumpers placed to connect the input through to the output. The external amplifier would then be connected directly to the headphone gain pot wiper.

U13 provides a momentary short to ground to start a clock-timer.

U7 is an inverting current driver. When all lines are low on the outputs of U1 and U2, the Off lamps are lit.

U8 is simply a relay driver consisting of darlington transistors. When a logic high is present on an input, the associated output goes low, switching the line relay on.

IDEAS AND COMMENTS UP TO NOW

It hasn't been noted yet that the lamps in the Line switches can be wired across any signalling circuit. It's simply handiest to put them across the telephone lamp circuits coming out of the KSU. The Telemix has been shipped with all lamp circuits in parrallel with the KSU.

The lamps, if used, draw quite a few mils of current. It's important that the common lamp ground be distributed over several wires if the run from the Key Service Unit is more than a few dozen feet. Provisions for seperate lamp grounds are included with wiring diagrams furnished in this manual. If it is found that the lamps are dimming when several lines are on, the lamp grounds may be tied together at a convenient point close to the Telemix.

THE TELEPHONE SYSTEM

The Telemix is designed to work with a standard 400 series key telephone system. If the system that the Telemix is being interfaced to is an electronic system which does not provide the Tip/Ring, "A", and lamp leads on a multi-conductor cable, the Telemix will not directly interface. One solution is to remove the studio lines from the electronic system and install a key system for these lines exclusively. In most areas, the KSU can be purchased from a private interconnect company. In some cases it is possible to have a private interconnect company cross-connect an electronic and a key system so that the studio lines may appear on both with all Hold functions intact, etc. This of course assumes the contractor has provided both systems. If the key system is rented from the telephone company, they may be able to add extensions from the electronic system into the key system, so that all station lines are available to the studio. The reverse, studio lines appearing on the electronic system, however, is probably not possible. The least expensive way to get a standard key system is to buy the equipment from a telephone supplier. Interconnection to phone compnay lines is accomplished via a standard connector known as a RJ21X, which is a standard 50-pin telephone connector wired with the incoming telephone lines in s specified order.

The cross-connect sheet provided in this manual indicates the cable color-code designation of the Telemix as well as some standard telephone company protocols for connection to a key system.

The following is a guide to the key system wire functions and how to determine which is connected where.

TIP/RING

These are the actual audio pairs for each line. They may be located using a standard telephone or headphones.

"A" CIRCUITS

These tell the key system which telephone is off the hook. Anytime a line is selected on a phone and taken off the hook, a connection is made from that lines "A" lead to "A" common. This causes the -24 volts dc from the key system to go to 0 volts. If the "A" lead connection is broken before the Tip/Ring is disconnected, the system puts the line on Hold. Therefore, these wires may be located using a volt-meter while selecting lines on a nearby phone. There is a single "A" common lead for all lines in a system.

"L" LAMP/"LG LAMPGROUND

These light the lamps on the telephone line which has been selected on the line buttons. When a line is off the hook, 10 vac is present between these wires. The "LG" wires are connected together at the key service unit. For small systems, only one connection may be required. It is best, however, to connect to as many "LG" leads as possible so as to distribute the current among many wires to reduce the voltage drop when a number of lines are lit.

In the great majority of systems, the "LG" lamp ground leads, and the "A" common lead are connected to ground, and are in effect the same thing.

In standard six-wire systems, the "A" common is renamed "A1" and is bussed to each line. Therefore, there are a full six leads devoted to each line: T/R, A/A1, and L/LG. This scheme is never used in systems with ten-button phones, unless each phone is serviced by two or more 25-pair cables.

CONNECTION TO THE SYSTEM

The connection to the telephone system is made via the 50 pin ribbon connector within the Telemix. There are a number of options as to how the unit may be interfaced.

The simplest is to plug the connector which comes from the key system directly into the Telemix connector. This method will work only when the key system connector is wired to the same standard as the Telemix, and it is desirable to have the lines appear on the Telemix in the same sequence as on present telephones. In most applications, a standard telephone should still be used for dialing outgoing calls. This would require a "bridging adapter" to split the key system line into two feeds; one for the desk phone, and one for the Telemix. Such adapters are available from telephone supply houses and local interconnect phone companies.

Another method which could be used is to locate a "66" punch-block which is used in the telephone system as a central connect point. A cable may then be obtained which has a female 50-pin connector on one end and pigtails on the other. The connector plugs into the Telemix, and the pigtails are connected in the appropriate sequence to the block.

Another method is to provide a punch block. This can be accomplished by unplugging the desk phone from the key system cable and inserting a punch-block between the two connectors. A number of manufactures make "66" punch-blocks with ribbon connectors mounted on the sides. A variation on this approach would be to wire a standard block with two cables, one having a female ribbon connector, and the other having a male on the ends. This would accomplish the same function as the connectorized block.

The Telemix will work without a KSU on any single or multiline system, with or without flashing lamps and hold.

If the key system being interfaced to the Telemix, has five or less lines on it and it's desirable to have lines 4 or 5 easily moved to conference 1 and 2, or there is an unusual situation which calls for a unique approach to connection of the Telemix, please contact the Product Support Department of Gregg Laboratories for assistance.

FINAL SUGGESTIONS FOR INTERFACING THE TELEMIX WITH MULTI-LINE KEY SYSTEMS

The telephone company will provide a standard 50-pin connector or plug wired to the customers specifications which will plug into the Telemix. Order a USOC RJ23X. The phone company business office should recognize this USOC number. They will need further information which will be furnished below.

First, remember that a connector (female) is needed on the end of the cable they furnish. They will want to know how the connector is to be wired. The normal array comes with Tip, Ring, and A circuitry, but not the lamp wiring. This will in no way hinder the operation of the Telemix but should the use of the lamps be required for lamp indicators on the Telemix, request lamp leads be furnished as well when specifing the wiring layout. The enclosed description on Bell's USOC RJ23X has a "position" table which essentially is a wiring diagram of the plug they furnish. Information below shows how to lay this out so as to connect with the Telemix.

Suppose the phone numbers and intended assignments are as follows:

555-0000 555-0001	Telemix line 1	AUTHOR'S NOTE: Please remember that if
555-0002	2	this USOC is ordered, the telephone company computer may spit it out without
555-0003	Δ	the registration number. Please refer
555-0004	5	to the comments on page 6 and note that
853-1400	Conf 1	a fundamental our page of and note such
853-1401	Conf 2	The state of the s

The information supplied to the telephone company would be as follows, assuming inclusion of lamp wiring.

TELCO INFORMATION		"USOC	RJ23X	pinout"		
LINE	<u>T</u>	<u>R</u>	<u>A</u>	<u>A1</u>	LG	<u>L</u>
Line 1 - 555-0000 Line 2 - 555-0001 Line 3 - 555-0002 Line 4 - 555-0003 Line 5 - 555-0004 Line 6 - 853-1400 Line 7 - 853-1401	26 29 32 35 38 41 44	1 4 7 10 13 16	27 30 33 36 39 14	2	28 31 34 37 40 43	3 6 9 12 15 18 21

Note that the A leads are separate but the A1 lead (ground) is only connected once. This is often standard wiring practice.

Refer the installer to a "standard" 10-button keyset if he has any questions about wiring fanout.

In cases where a standard 10-button desk set is already in service, it will only be necessary to furnish a bridging adaptor at the point where the set plugs into its service cable, and then add a short male-female extension cord between the bridging adaptor and the Telemix.

This wiring layout applies to conventional Key Service Units and will be modified to accept Comkey or other special-service systems.

And in cases where a speakerphone is employed, minor rewiring will be needed.

AUDIO PROCESSING

FCC Rules and Regulations require a band-limited signal of a definite maximum RMS level be the only thing fed into the telephone network. General requirements are that maximum transmit level, over a three-second period, not exceed -9dBm at any frequency; that no excessive signal power be inserted at 2600 Hz; that the send audio be down nearly 30dB at 4000 Hz, but if it's notched there, a much gentler slope can be used up to 12Khz. The Telemix provides the required send audio specifications unless the send audio is fed at to high of a level. At that point, diode-clipping starts.

In the return circuit, some equalization is often used to brighten the telephone audio. Do a little peaking-e.q. at 250 Hz and 3500 Hz. But put that equalizer after a noise-gate for maximum effectiveness.

PARTS LIST

The only parts that should ever need to be replaced are the lamps in the switches. Pull the switch caps and pull out on the lever to pop out the bulb. The Line switches have an AML91LA93 14V bulb in them, the other switches have an AML91LA85 28V bulb. If anything else needs replacing, please contact Gregg Laboratories.

WARNING

Gregg Laboratories assumes no liability or fault for damage to person, property, equipment, goods, merchandise, profits, goodwill, or reputation arising out of any defect in or failure of this product.

SUBMINIATURE 'D' CONNECTOR PIN ASSIGNMENTS

Please refer to page 2 in this manual where this connector is discussed.

CONNECTOR PIN NUMBER	ASSIGNMENT
17	Headphone Amplitter out 1Send Audio InNo ConnectionTelemix Audio Out +Telemix Audio Out24vdc +24vdcGroundGroundGroundGroundNo connectionSend Mute Enable (connect to ground)Air Mode Enable/Disable (triggered when grounded)No connectionHeadphone amplifier In Left ProgramHeadphone amplifier In Right ProgramLive Switch LampLive/Swear switches lamp commonSwear switch lampSwear switch normally openLive/Swear switches common
Input level and impedanceOutput level and impedance	
Frequency Response	Complies with Part 68, FCC Rules and Regulations, and with
Trnas-hybrid loss through the system Hum and noise Bi-directional gain Loss between the two conference trunks Headphone amplifier power External input level to headphone amp External headphone amplifier input impedance	None OdBLess than 3dB, worst caseNMT 2 watts/channel into 8 ohmsDepends on level in the telephone (adjust)

STANDARD TELEPHONE COLOR CODES

TELEMIX	CIRCUIT
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Line 1 TIP Line 1 RING Line 1 'A' circuit 'A' common (ground) Lamp common ground* Line 1 lamp L

Line 2 TIP Line 2 RING Line 2 'A' circuit Line 2 lamp L

Line 3 TIP Line 3 RING Line 3 'A' circuit Line 3 lamp L

Line 4 TIP Line 4 RING Line 4 'A' circuit Line 4 lamp L

Line 5 TIP Line 5 RING Line 5 'A' circuit Line 5 lamp L

Conference 1 TIP (line 6) Conference 1 RING Conference 1 'A' circuit Conference 1 lamp L

Conference 2 TIP (line 7) Conference 2 RING Conference 2 'A' circuit Conference 2 lamp L

10-BUTTON SETS

white/blue blue/white white/orange orange/white white/green green/white

white/brown brown/white white/slate blue/red

red/orange orange/red red/green brown/red

red/slate slate/red black/blue orange/black

black/green green/black black/brown slate/black

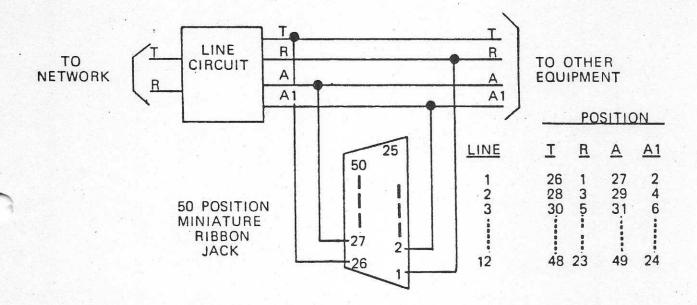
yellow/blue blue/yellow brown/black green/yellow

yellow/brown brown/yellow blue/black blue/violet

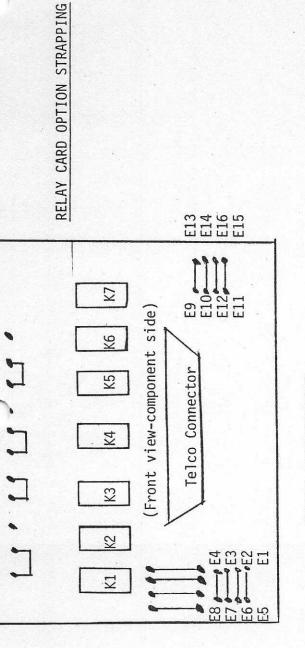
*Lamp ground are common to this wire. All other lamp grounds are unused, and may be used for special services. If lamp intensities are a problem, more suggestions are available by call Gregg Laboratories.

Standard 6-button desk sets may use the last 10 pair for special service applications (speakerphone, etc.) and will need to be removed from the cable feeding the Telemix. By moving jumpers inside. Information on this is available from the factory.

In a 20-button set, two 25-pair cables will be used. The first connector covers lines 1 thru 9, the second 25-pair connector lines 10 thru 19.



SIMPLIFIED SCHEMATIC - JACK RJ23X FIGURE 21

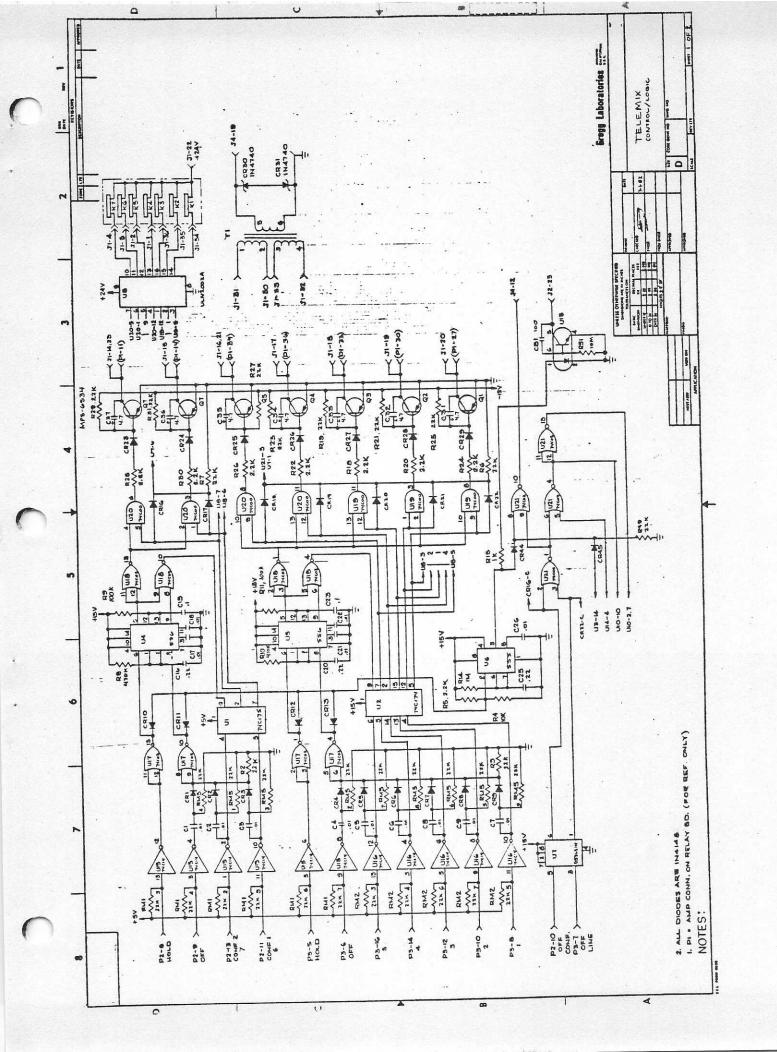


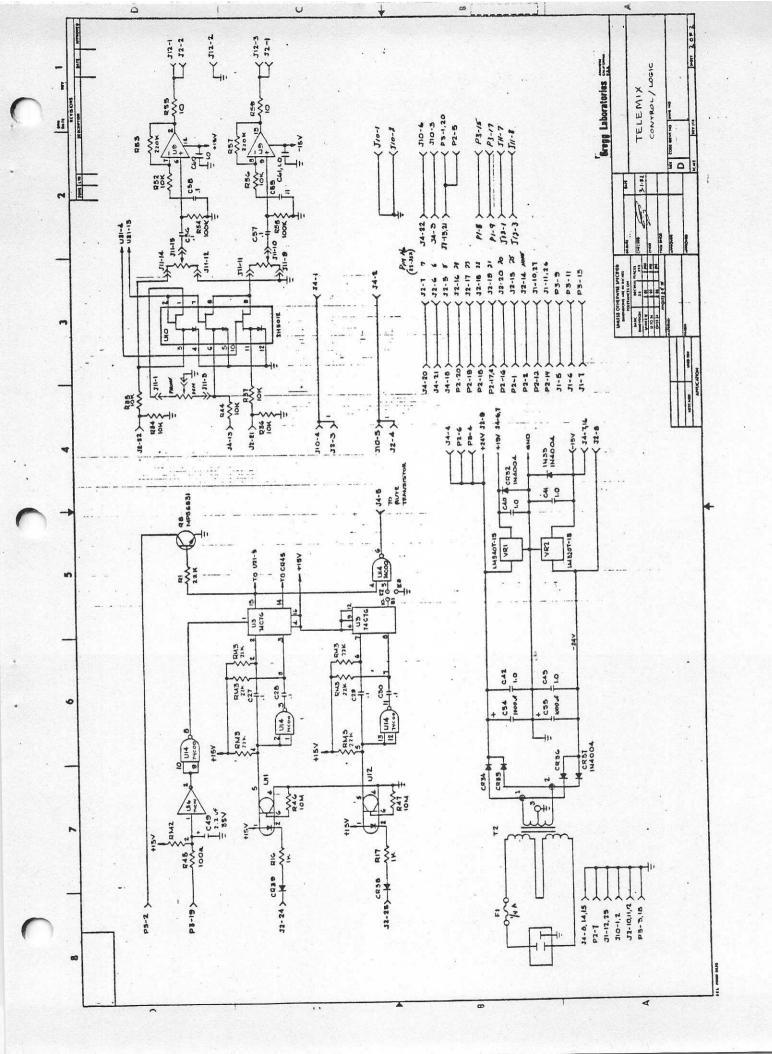
JUMPER DESIGNATIONS

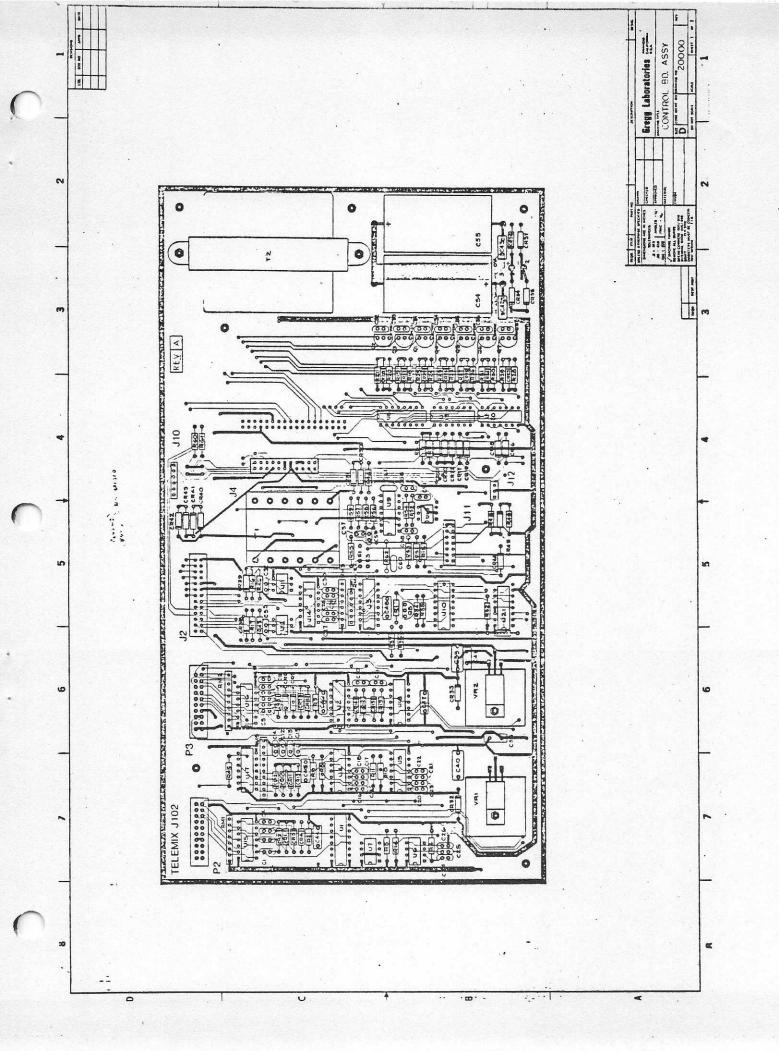
(5-line)Line 5 to line 6 option	E5-E1 E6-E3 E8-E4	E9-E13 E10-E16 E11-E15	Pull jumpers E17-22 only *Line 5 Tip/Ring paralleled at cable end with Telemix line 6 wiring (see wiring chart)
(5-line) lines 4%5 changed to 6%7	E5-E3 (Line 6 lamp) E6-E4 (Line 7 lamp)	E9-E16 E10-E15	Pull any jumpers on E17-28 *Tip & Ring for lines 4%5 parralleled at cable end with Telemix line 6 wiring. (see wiring chart for colors and pin numbers)
(9-line) standard option	E1-E5 Lamp E3-E7 Circuit E4-E8	E9-E13 Hold E10-E14 Circuit E11-E15 ("A")	E18-E19 E21-E22 Ring E27-E28

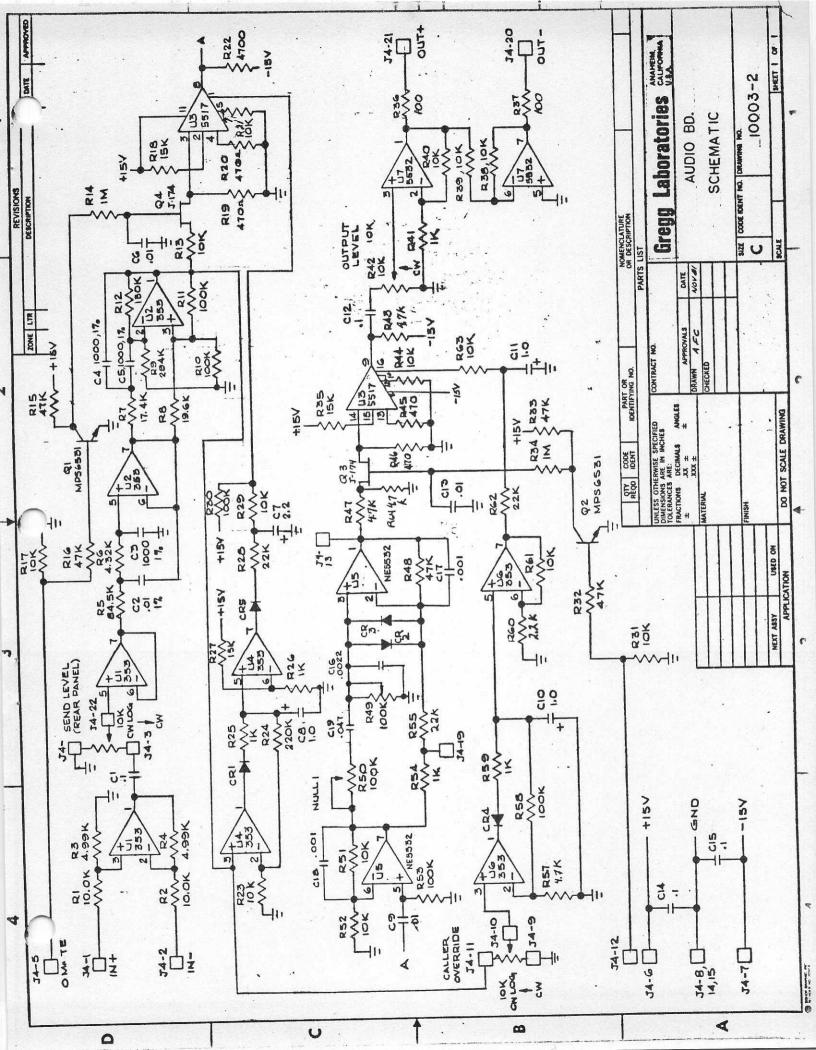
^{*} Any other services on these wires must be removed (usually "A" grounds), or may use other unused pairs. Other options available on request from from Gregg Laboratories Product Support Department.

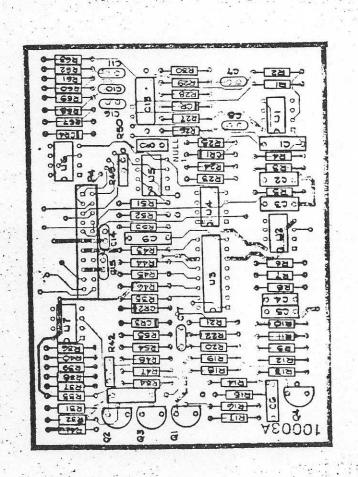
	no line 5, or #5 as intercom)		7-28 11el at cable end p/Ring wiring	
	Line 4 to Line 6 option (no line 5, E5-E3	E9-E16 E11-E15	Pull any jumpers on E17-28 * Line 4 Tip/Ring parallel at cable with Telemix line 6 Tip/Ring wiring (see wiring chart)	
0	(5-1ine)			











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