

telos 10

digital telephone system

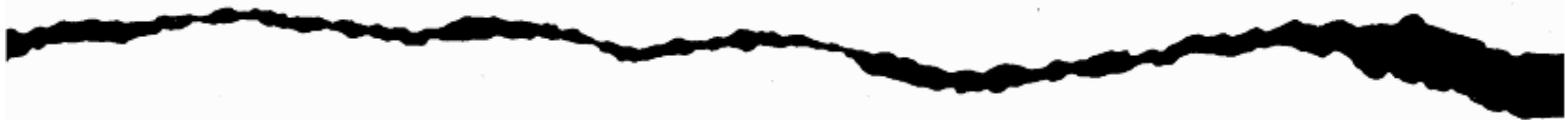


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July 3, 1987

A personal note:

The Telos 10 is not the product of a large company. It was designed by me to solve a problem I had encountered in my own work as a Chief Engineer.

Fourteen years ago, at my first station, the PD complained about the one-way-at-a-time operation of the speakerphone we were using to put calls on the air. He wondered if we could do better. Foolishly, I embarked on a quest to find some way to improve on the speakerphone arrangement.

The first thing we tried was a Johnson phone patch made for ham radio use. As you will discover, if you have not already, a simple hybrid of this sort works terribly when connected to most phone lines. When we hooked it into our studio system, it worked so poorly we thought it was defective. The announcers voice came out of the unit louder than most of the phone calls. It sounded terrible on the air. Not understanding at the time why it was so bad, I put the Johnson on the shelf and went back to a surreptitious connection to the speakerphone.

Ever since that experience, I had been looking for a better way. When digital audio processing technology came on the scene a couple of years ago, the possibility for vast improvement in hybrid performance was finally here. Because it incorporates this new technology, the Telos is the very best available. I am proud of it and use it myself. I wish I had had it 14 years ago!

Apparently, you agree - you bought one, and you're in good company: at this writing, over 250 Telos 10's are in daily use. The users list reads like a "who's who" among great radio stations and networks.

Having metamorphosed from a "garage" outfit, Telos Systems is now a "bone fide" company with a phone and an office and employees and everything! If you need to contact us for any reason, it's:

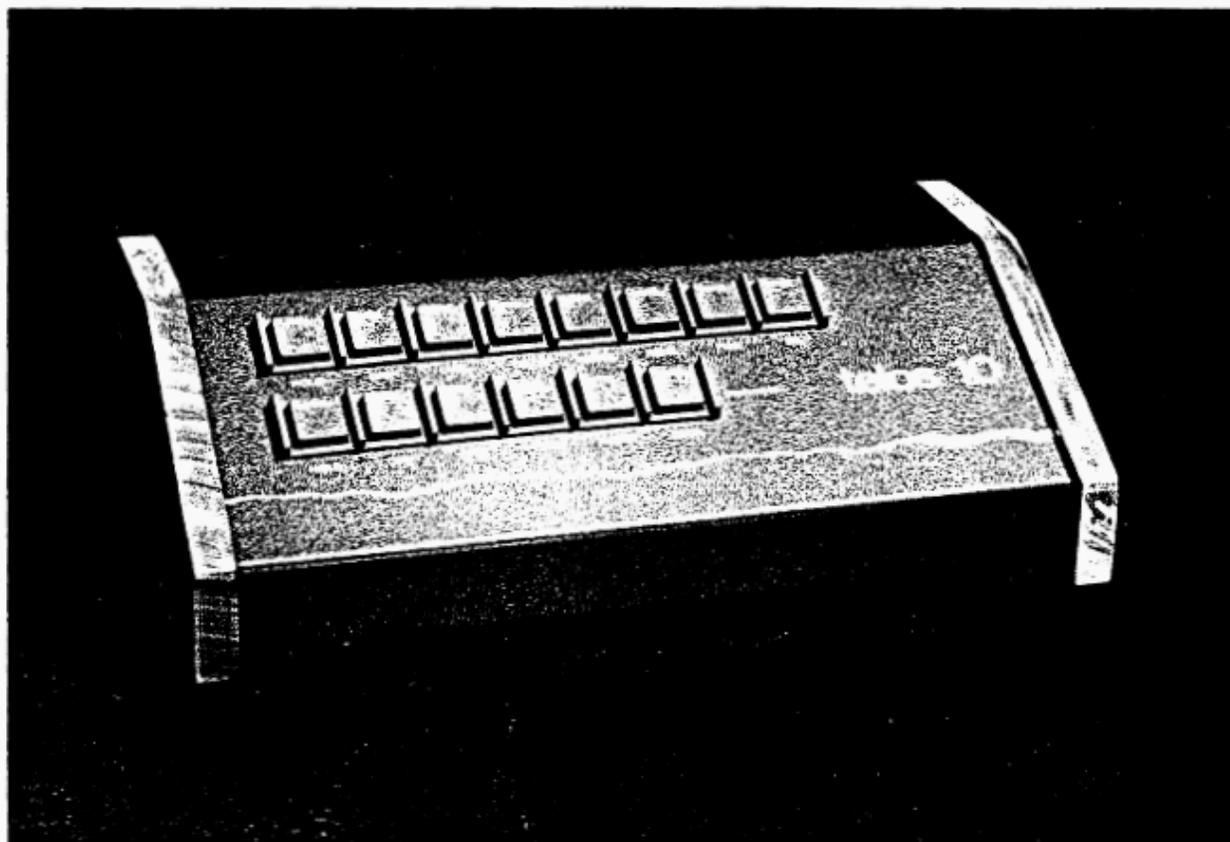
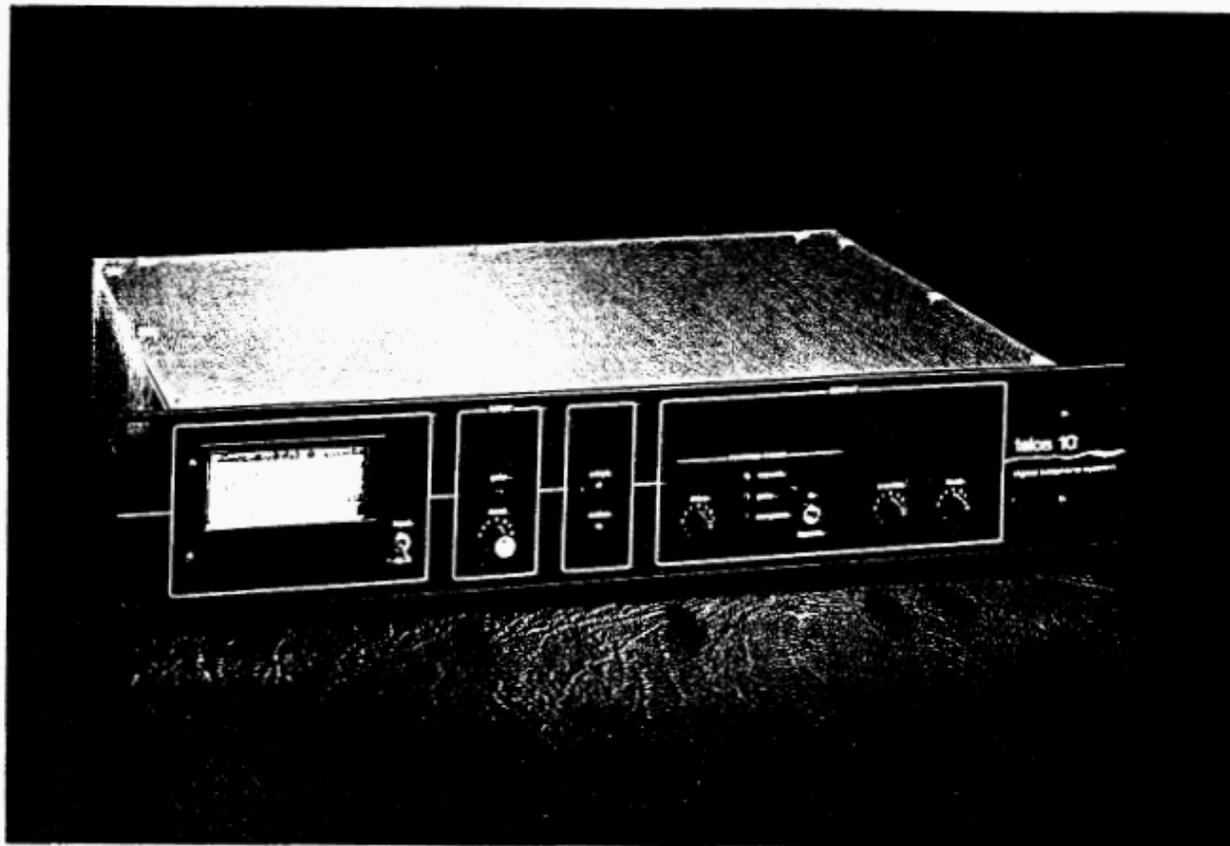
Telos Systems
(216) 241-7225

I welcome questions, suggestions and comments any time.

Of course, the Telos's very limited dealer network is also ready to assist. They usually know what plug goes where and can answer many questions on Telos installation and operation.

Keep on keeping the GM happy...

Steve Church



Section 1 - INTRODUCTION

Telos 10 digital telephone system

1.1 Introduction - Overview of device characteristics

The Telos 10 telephone interface represents the state-of-the-art among devices used to connect the switched telephone network for broadcast use.

The unit provides a complete system of audio and control functions to connect a maximum of ten telephone lines to a studio console. The main unit is a rack-mount 3 1/2" enclosure. Simple pushbutton contact closures provide remote line selection. Standard key phone line status indication outputs are provided, as well. The control panel may be constructed by you for integration into your station's studio scheme, or may be provided by us.

Telephone lines are controlled in two groups - a main group of 6 lines and a "conference" group of 4 lines. Transformer matched conferencing capability is provided between line groups, with "line-mash" parallel conferencing used within each of the two groups. No conference gain is provided, so multiple units are suggested when high-quality conferencing is desired.

A microprocessor handles the line-select and other control functions.

Superior hybrid performance is achieved by the application of digital signal processing technology to provide the central telephone interface audio functions. While not perfect, the digital signal processing approach results in the least possible leakage of send audio into the receive (telephone) output, thus allowing full simultaneous conversation on most phone lines - without announce mike quality degradation.

An AGC circuit is incorporated to ride gain on input audio. Input audio is also filtered to improve performance, and to satisfy FCC regulations regarding out-of-band energy.

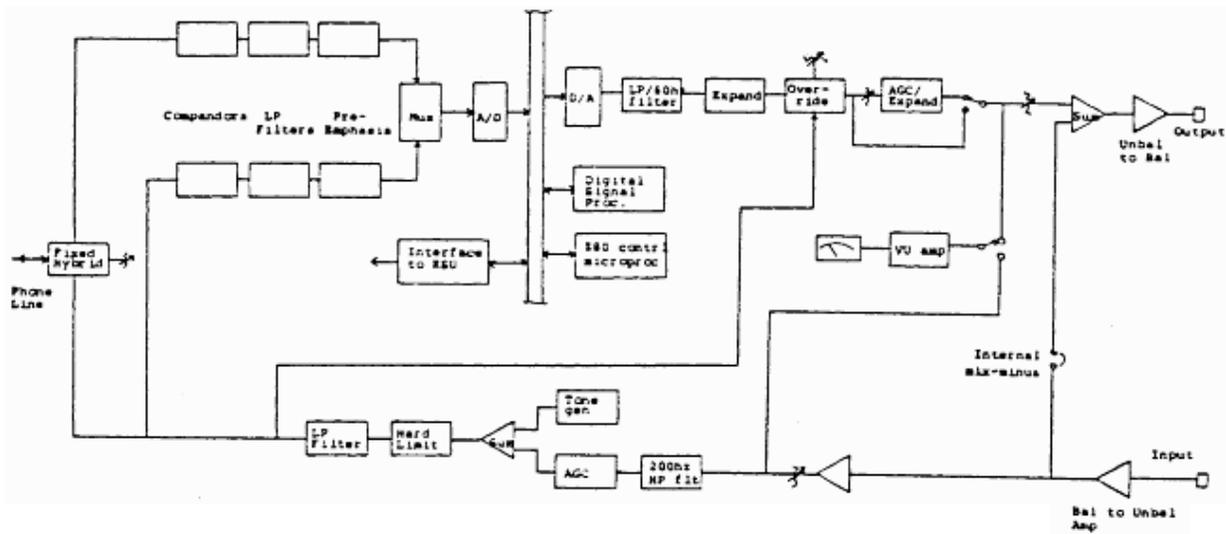
Telephone receive audio is filtered to improve quality. A notch filter at 50/60 HZ reduces hum, often present on telephone lines. A gated AGC/expander may be switched-in to help control gain and reduce noise. Finally, a "caller override" or "ducking" gain function allows dynamic reduction of caller volume when the announcer speaks.

Internal option jumpers allow the unit to be placed downstream of the console (that is, between the console output and the station processing input) with the telephone audio mixing function provided by the unit itself, or the station console can provide a "mix-minus" audio feed to the unit with the output appearing on a console fader as any other input in the usual fashion. The latter is preferred.

Telos 10 digital telephone system Introduction – Overview of device characteristics

Connection to telephone circuits is made via a standard 50 pin blue ribbon type connector. A "1A2" type Key Service Unit is used to interface between the Telos unit and the actual central office telephone lines. While newer business telephone systems are quickly replacing the old-style 1A2 key phones in the "front-offices" of many radio stations, the 1A2 type key systems remain the best choice for on-air studio application, since these systems provide a solid contact closure connection between studio equipment and the incoming central office phone lines for least degradation of telephone audio quality – as well as high reliability and relatively simple interfacing.

Audio and control connections use 25 pin "D" connectors. These are the same type used for computer serial data and modem links, so plugs and pre-wired cables are readily available.



Telos 10 Block Diagram

1.2 Introduction - The Switched Telephone Network

The phone pairs provided to us by the phone company are known officially as "subscriber loops."

On standard subscriber loops, the phone company only guarantees a frequency response of 300 - 3,000 Hz. In the not too distant past, when local calls were connected at the exchange by relay contacts, better frequency response was likely to be present on many conversations. However, the newer exchanges use digital switching techniques which cause all calls to be limited to about 3,400 Hz. Long distance calls usually go by bandlimited microwave or satellite links, so frequency response on them has never been much better than the promised 300-3,000.

A current-limited DC voltage and the conversation audio appear together on each phone pair. The DC leaves the exchange at 48 volts, and is limited to approximately 6 - 80 ma by a series resistor. The resistor's value is selected depending upon the resistance of the loop itself. A little known fact: a standard telephone incorporates a varistor which automatically adjusts voice level to compensate for various ^{loop} lengths.

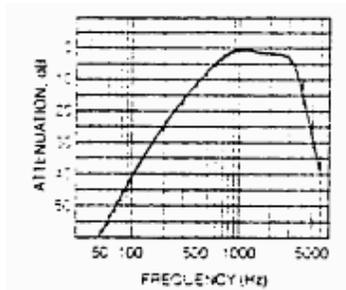
DC resistance of loops varies from 0 - 1,300 ohms.

The incoming audio level is usually around -18 dbm. Send audio is required to be limited to -9 dbm. Loop loss from the exchange to you is supposed to be kept below 8 db.

Ring voltage is 10-60 HZ AC at around 90 volts.

The telephone company uses a special weighting curve called "C-message weight" to determine the signal-to-noise on a phone line.

c msg curve



As you can see, this weighting curve has considerable low-frequency roll-off. This allows a line to have a lot of hum and other low frequency noise and still meet the phone company's noise spec. While this makes life easier for the phone people, it can be troublesome when we are using phones on the air.

Telos 10 digital telephone system Introduction to telephone network

If you are having noise problems, you might try to get the phone guys to switch their noise meter to the "flat" position, since their gear usually does have this option available. Probably years since they used it, but it's likely to be there. Incidentally, the C-msg curve was developed many years ago to simulate the frequency response of an old-style telephone earpiece!

The FCC rules for connection of equipment to phone company lines can be found in The Rules and Regulations of the Federal Communications Commission, Part 68: Connection of Terminal Equipment to the Telephone Network. The rules can be ordered from the government printing office, or can be found reprinted in the Code of Federal Regulations in the reference section of most larger libraries.

1.3 Introduction - Telephone Interfacing Background

In the beginning, there was only Ma Bell.

Since Ma was a very protective lady, if you wanted a two-way interface to the Miss Bell's network, you probably asked your local phone company to give you a speaker-phone, or, if you and they were more demanding and sophisticated, you may have requested a "hybrid transformer" arrangement. Some of us, on a lower budget, just paralleled an incoming phone pair or studio phone earpiece wires. Each of these approaches got the job done, sort of. Each had distinct disadvantages.

A number of home-brew projects were eventually undertaken by ambitious engineers to improve upon Bell's offerings and to provide needed functions (an example of the best of these projects was Mark Durenberger's system at WLOL, Minneapolis described in a BM/E article some years ago).

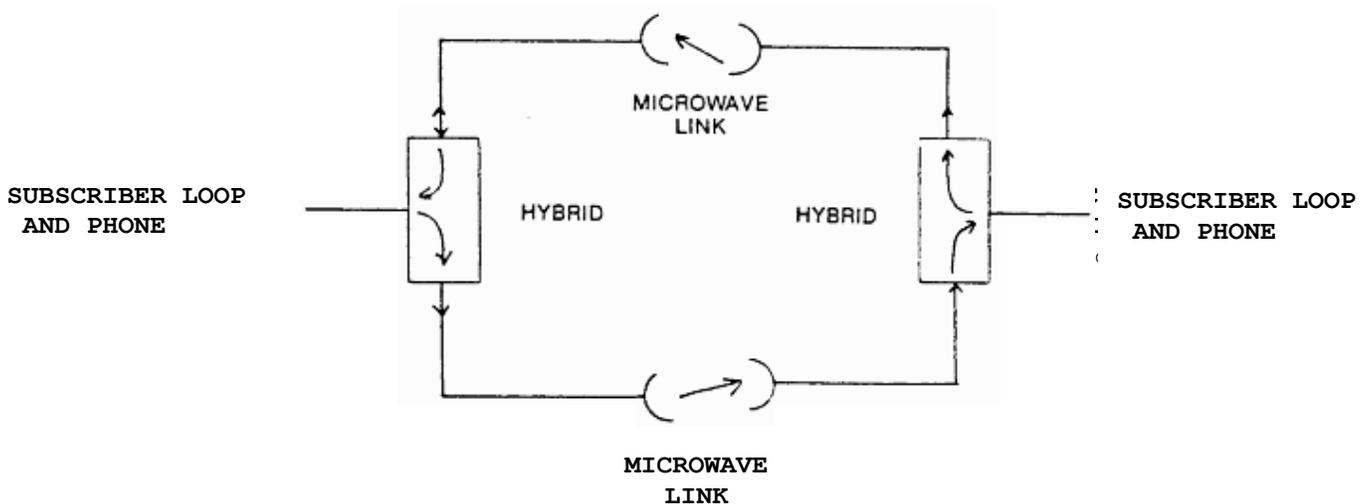
More recently, a number of manufacturers have introduced packaged systems to perform the interface function.

Now, with the passage of time, the introduction of new technology, and all of us wiser, we can explore new and better ways of making the broadcast-telephone connection.

Before we do this, lets take a detailed look at the various techniques that have gone before:

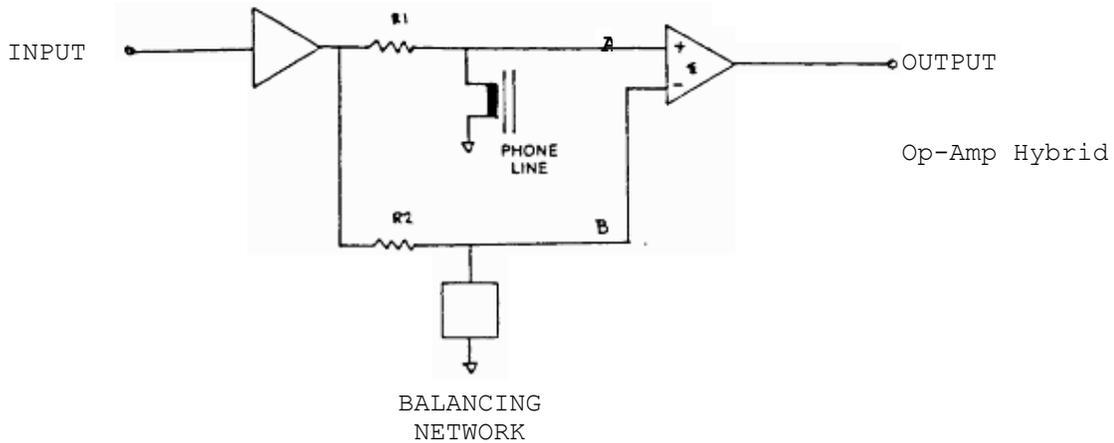
The hybrid approach -Hybrids were invented by the telephone company many years ago

in order to separate the send and receive signals from a standard phone pair. Long distance calls went by microwave, and microwave links go only one way. Thus, the need for some way to split the two signal directions.



Telos 10 digital telephone system Telephone interface background

The first hybrids were made from transformers with multiple windings. Nowadays, most hybrids are made with active components, op-amps or transistors, and are known as "active hybrids" (naturally). Both circuit types use the same principle, and achieve the same effect. Let's look at an op-amp version:



The first op-amp is simply a buffer. The second is used as a differential amplifier - the two inputs are added out of phase (or subtracted, if you prefer). If the phone line and the box labeled "balancing network" have identical characteristics, then the send signals at A and B will be identical, and no send audio will appear at the output of the second op-amp.

The balancing network is a circuit consisting of capacitance and resistance, and sometimes inductance, forming an impedance network. Depending on the hybrid's application, this circuit can be very simple, or be made of a large number of components, with a very complex impedance characteristic.

Notice that R1 and the phone line form a voltage divider, as does R2 and the balancing network. If you think of the phone line and balancing network as pure resistances, then it is apparent that the phone line and the balancing network must have the same value in order for the signals at A and B to have the same amplitude, and full cancellation to occur.

In the real world, the phone line is not purely resistive, but rather a complex impedance, causing both the amplitude and phase to vary at A as the send signal frequency varies.

Only when the impedance of the balancing network is the same as the phone line, and the signal at B is matched to that at A in both amplitude and phase, will full cancellation of the send signal be achieved. Otherwise, leakage results - the scourge of hybrid circuits.

Since the phone company's requirements were not generally too stringent,

Telos 10 digital telephone system Telephone interface background

they usually used (and still do) a simple network with compromise values of resistance and capacitance. The phone company's design goal is to get an average of about 12db rejection, with 6db acceptable on difficult lines. When better performance is required, modules with a number of R and C elements which can be switched in or out are employed, the switches being set to match the network to a particular line.

The amount of hybrid rejection is sometimes referred-to as "trans-hybrid loss". Trans-hybrid loss is NOT the loss from the phone line audio to the output, as one confused interface manufacturer seems to be convinced.

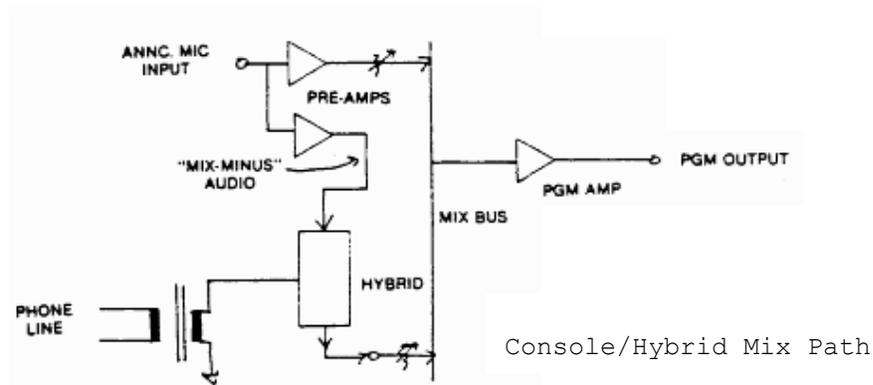
Incidentally, while we're talking terminology, a standard phone pair is called a "two-wire" circuit in telephone engineering jargon, and the hybrid has two ports - send and receive (in addition to the telephone line and balance network connections), requiring four wires to connect - thus, a hybrid is sometimes called a "two-to-four wire converter."

What use is a hybrid to us, as broadcasters? It depends.

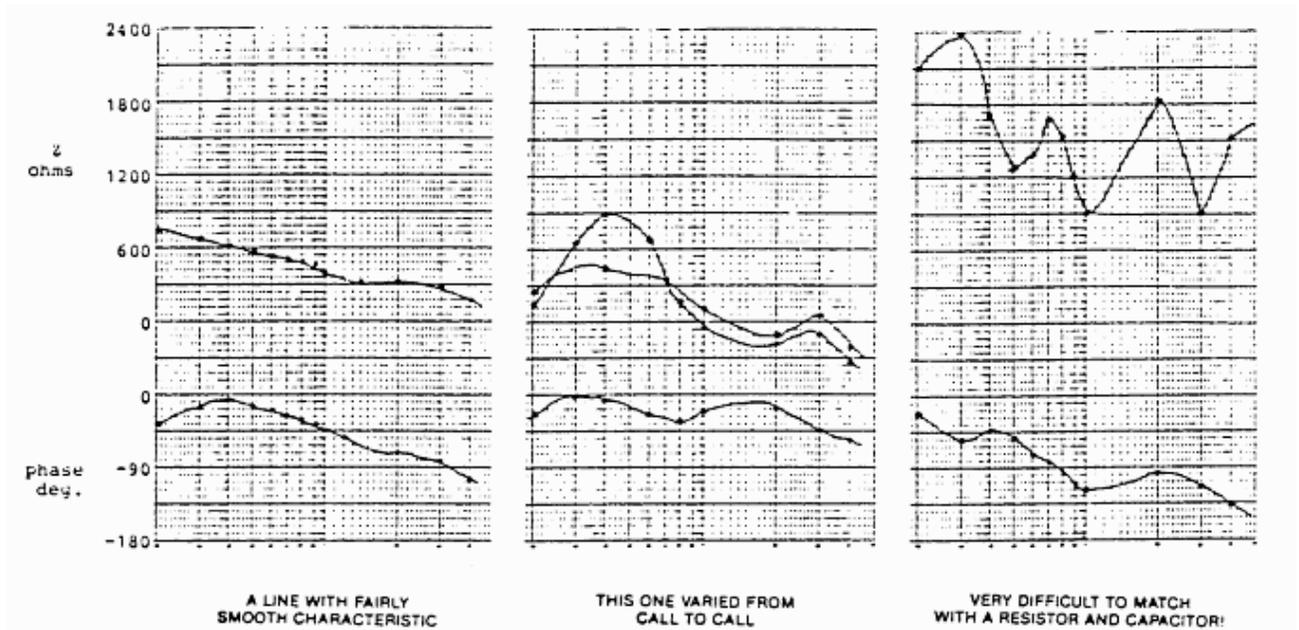
In broadcast application, the telephone audio, including hybrid leakage consisting of the local announcer audio phase-shifted because of the phone line reactance - is mixed with the original announcer audio.

If the amount of the leakage is too great, and/or the phase shift too extreme, the announcer audio will suffer degradation. When this occurs, the announcer most often sounds "hollow" or "tinney" as the phase cancellation affects some frequencies more than others. Also, feedback can be a problem when it is desired to have the caller on a speaker in the studio.

Clearly, a hybrid will be useful only when leakage can be kept acceptably low.



Moving on, here are some graphs of impedance vs. frequency for some phone lines we have encountered:



Phone Line Graphs

While some of these are relatively smooth curves, others are pretty terrible. That last one is a real doozy. It is the shop phone line at a radio station in the Midwest where the Telos was designed.

The lines with the smooth curves have impedance characteristics which can be simulated with a simple resistor-capacitor combination. These lines would work very well with a hybrid type interface device, since a practical RC balance network would make the cancellation of send audio at the receive output port high enough to prevent announcer audio coloration. Of course, if the hybrid is to be switched among a number of lines, they would all have to have nearly the same curve. Another consideration: the line characteristic would have to be consistent from call to call.

While it would theoretically be possible to make a balance network to match the more difficult lines, practical considerations usually keep this approach from being used - the line impedances are often inconsistent from call to call, or the impedance characteristic required is too difficult to produce.

One manufacturer makes an interface unit with an automatically-adjusted resistor-capacitor balance network. This approach works well when the phone lines to be used have a smooth impedance curve, but poor performance will still result when the line impedance characteristic is non-linear and can't be adequately matched by a simple RC network.

Another way to improve hybrid performance on widely varying or difficult lines is to trade off single-frequency rejection for wide-band loss by making the phone line look less reactive to the hybrid with some series resistance. This is the approach used in a very popular integrated interface device.

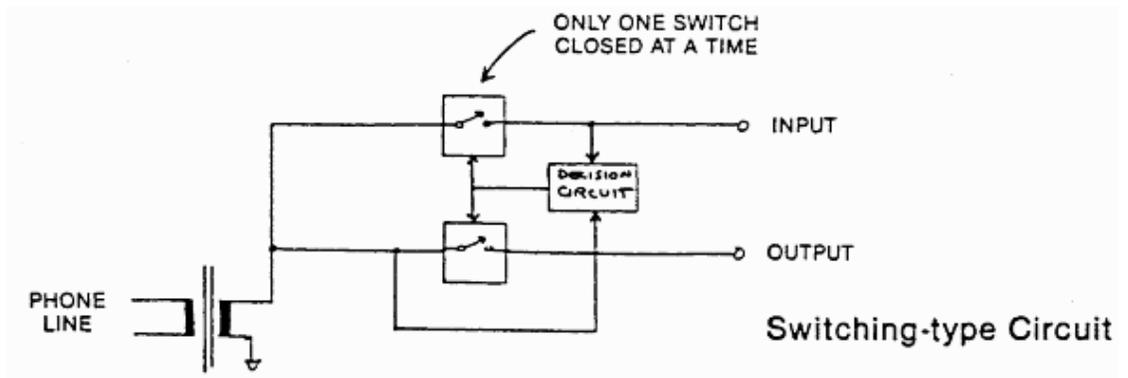
Before we leave the subject of hybrids, another point should be made:

THE TRUE TEST OF HYBRID PERFORMANCE IS DETERMINED BY MEASURING THE AMOUNT OF REJECTION ACROSS THE ENTIRE AUDIO FREQUENCY RANGE, PREFERABLY WITH WHITE NOISE AS A TEST SIGNAL AT THE SEND INPUT.

Any hybrid with an adjustable R and C balance network can claim high rejection at any given frequency, since both phase and amplitude at a single frequency can be tweaked for good cancellation. Voice is rarely a single-frequency source.

The Speaker-Phone Approach - This technique uses simple switching to keep the send audio from appearing at the receive output. Two electronic switches are used in such a way as to ensure that either the send or the receive path is closed at any given time - but, never both simultaneously.

Usually, a decision circuit compares the send and receive levels, with the direction of transmission being determined by the relative signal strength.



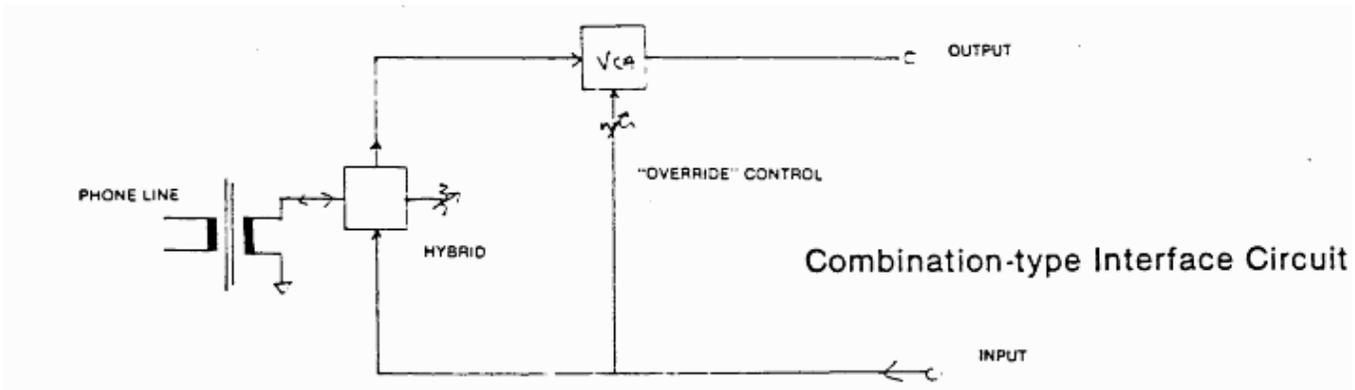
Telos 10 digital telephone system Telephone interface background

Sometimes Voltage Controlled Amplifiers (VCA's) are used in place of the switches to provide soft-switching rather than the absolute on-off of simple switches. The common telephone company-supplied speakerphones work this way.

The disadvantages of the switching technique result from the uni-directional nature of these systems. The primary problem is that the caller cannot be heard while the announcer is speaking. Also, noises in the studio can sometimes cause a caller to "disappear" momentarily - especially on weak calls.

Combination systems -

The best systems for broadcast use combine the hybrid and VCA-type switching techniques. Here is one such configuration:



The hybrid is used to produce as much send to receive isolation as can be achieved. The "override" or "caller-control" VCA causes the "dynamic" rejection to be greater than the hybrid alone can produce. When send audio is present, the VCA reduces the gain of the receive signal. Thus, leakage is reduced on a dynamic basis. However, it should be apparent that the level from the phone is also reduced when the announcer is speaking. A control pot in the VCA control signal path is often used to adjust the amount of receive ducking, allowing full duplex operation (when the hybrid alone produces sufficient rejection), or a speaker-phone-like effect whereby the caller is turned-off when the announcer speaks. As a practical matter, this control is usually set to provide the minimum amount of ducking which provides adequately low send-to-receive leakage.

The Telos uses the combination approach just described, with a high-performance digital signal processing hybrid, send and receive AGC, receive expansion, and carefully chosen filtering added to the basic configuration.

1.4 Introduction – Digital Signal Processing

Digital signal processing means, simply, the process of operating on continuous, analog signals which have been represented digitally.

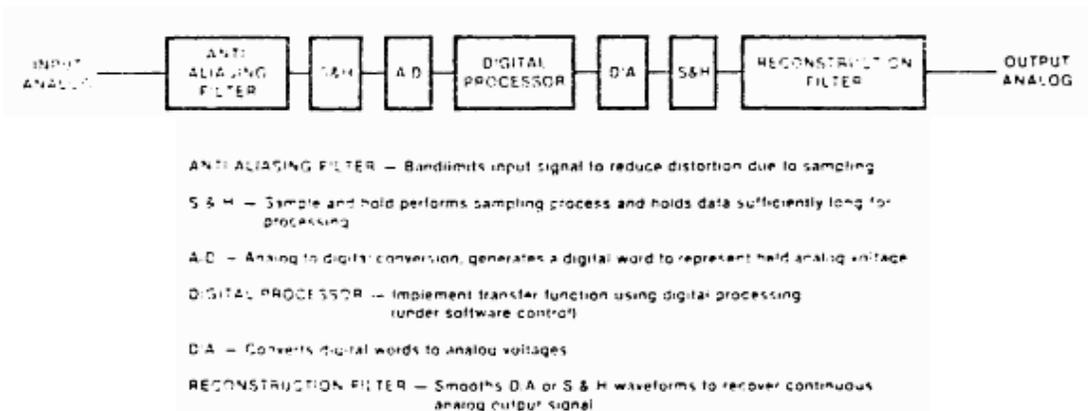
The advantages of conversion of analog signals to digital for storage and retrieval are well known, and the technology is now common – i.e. CD players, digital tape and satellite links. The next frontier is the processing of digitized signals. Digital processing offers a number of advantages:

- Digital accuracy is predictable and repeatable, so component selection and system tweaking are not needed.
- Stability is more easily achieved in digital systems.
- In most instances, signal degradation occurs only at the analog/digital interface. Thus, a more complex signal path is possible, since signal quality is not reduced as it passes through each stage.
- Software instructions determine circuit function. Improvements are possible without PC board modification.
- Digital makes possible complex processing that would be impractical in analog implementation. With regard to telephone interfacing, this feature is primary.

Let's look at a general-purpose digital signal processing system.

The block diagram below illustrates the basic components of a system using a digital signal processor:

The system is comprised of the following basic parts:



1) Anti-aliasing filter - This filter is used to bandlimit the incoming analog signal prior to sampling. If a signal too high in frequency, relative to the sample rate, is passed to the S/H, aliasing distortion results.

A very simple rule applies - the sampling rate must be at least twice as high as the highest signal frequency present at the output of the anti-aliasing filter. This rule is called the Nyquist theorem, and is fundamental to digital signal processing. If this rule is observed, the original analog signal will be preserved and correctly reproduced. In the real world, however, distortion may occur in the A-to-D or D-to-A converter.

2) Input sample and hold (S/H) - The S/H samples the signal at the system sampling rate. Each resulting sampled amplitude is held long enough for the A/D converter to accurately convert.

3) Analog-to-Digital converter. The A/D converts the sampled amplitude to a digital word.

4) Digital Processor - Performs mathematical operations on the digitized signal to simulate analog processes. Using the appropriate combinations of mathematical manipulations, it is possible to "simulate in real time" analog functions, such as gain control, filtering, and signal generation and detection.

While a general-purpose microcomputer (such as a Z80) is capable of this sort of calculation, it would be far too slow for real-time audio processing. Signal processor chips are optimized for speed, and have instruction sets especially suited to signal processing application.

5) Digital-to-Analog converter (D/A) and output S/H - converts the digitized samples back to analog.

6) Reconstruction filter - Sometimes called a "smoothing" filter, which describes what it does. This is a low-pass filter which, by removing the high-frequency components created by the sampling process, reconstructs the desired analog signal.

All of these elements are present in straightforward "textbook" form in the Telos system.

The automatic/adaptive hybrid balancing function is implemented in digital, while other functions, such as input and output
• AGC, are performed with typical analog circuits.

Section 2 - INSTALLATION

Telos 10 digital telephone system

2.1 Installation - Connection to the Telephone System

2.1.1 General

The Telos 10 is designed to work with a standard "1A2" key telephone system. This is the old-style system used nearly everywhere until electronic phones became common. These systems are sometimes called "400 series" since they usually use KSU-400 line cards.

If you already have this sort of system, it will only be necessary to connect the Telos 25-pair cable to the appropriate wires coming from the key service unit.

If you do not, at present, have such a system, you will either have to have one installed by your phone company or an interconnect firm - or, you can purchase the equipment and do it yourself. It is much less expensive to install a system yourself. A good place to purchase key equipment and phones is your local Graybar Electric branch, if there is one in your city. Also, some of the interconnect companies are willing to sell equipment outright, if you talk right to them. Believe it or not, some of them are worse than the phone company, though. Another good source of new and used key equipment at low cost is:

Consolidated Communications, Inc.
28 Calvert Street
Harrison, NY 10528
(914) 835-5000

Installation of phone equipment is really quite straightforward. You've met the guys from the phone company that do this work, haven't you? It's just a matter of connecting the incoming lines to the key service unit input, and wiring the output side to your phones and the Telos unit. All connections to the KSU are generally routed through "66" punch blocks, so that changes can be easily made. The instruction sheet packed with the KSU will probably have a suggested 66 block wiring sequence. It's a good idea to follow it, since using a standard scheme will make life easier for any phone people who have to deal with the system. The "standard 25-pair color code" describes the usual sequence of wire connections to a block.

2.1.2 "Electronic" phone systems

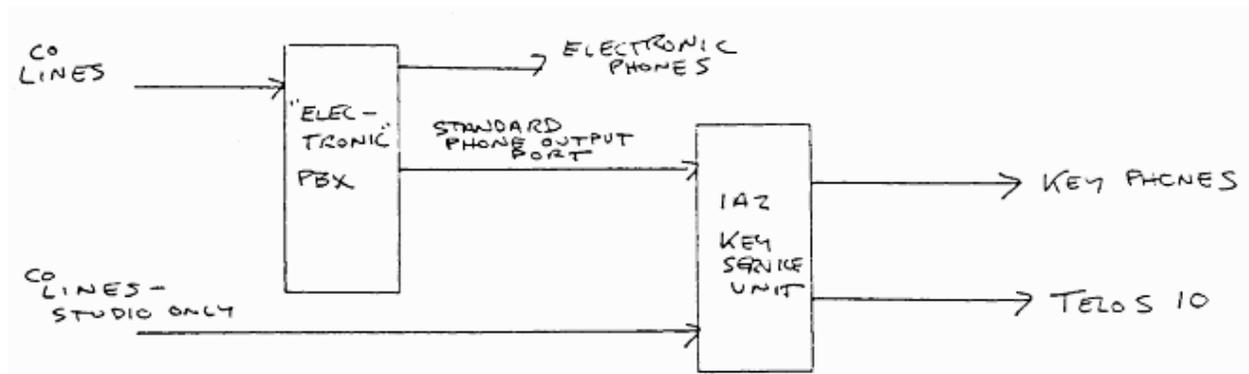
If you have an electronic system (like TIE, NEC, Rolm, Siemens, etc.), you are probably wondering how the 1A2-type system fits in. One option is to simply segregate the office and studio phone lines,

Telos 10 digital telephone system Connection to the telephone system

with the studios being served by a separate system with no connection to the electronic system.

However, most stations need to have some lines from each system in common. This can be accomplished. The electronic systems usually have provision for connection of a standard single line phone as an extension. Since these ports look just like Central Office incoming lines, they can be connected to an INPUT of the 1A2 key system like any other line. The key phone button would then access a PBX extension, rather than an outside line. Thus, all the PBX functions (transfer, pooled facility access, etc.) can be used.

2.1.3 Determining the connection scheme



So, now that you have a key system in place, the next step is to determine how it is wired to the point where you will make the interconnection.

Some common standard connection schemes are given in the table. If you are fortunate, your system will follow one of these standards.

Telos 10 digital telephone system

Connection to the telephone system

Standard 25 pair color-code:

6-wire fanout	5-wire fanout	Bell "Comkey" 4-wire fanout	Color
Line 1 Tip	Line 1 Tip	Line 1 Tip	white/blue
Line 1 Ring	Line 1 Ring	Line 1 Ring	blue/white
Line 1 A	Line 1 A	Line 1 Lamp	white/orange
'A' circuit common (gnd)	'A' circuit common (gnd)	Line 1 'A'	orange/white
Line 1 lamp ground	Line 1 lamp ground	Line 2 Tip	white/green
Line 1 lamp	Line 1 lamp	Line 2 Ring	green/white
Line 2 Tip	Line 2 Tip	Line 2 lamp	white/brown
Line 2 Ring	Line 2 Ring	Line 2 'A'	brown/white
Line 2 A	Line 2 A	Line 3 Tip	white/slate
'A' circuit common (gnd)	Usually line 9 'A'	Line 3 Ring	slate/white
Line 2 lamp ground	Line 2 lamp ground	Line 3 lamp	red/blue
Line 2 lamp	Line 2 lamp	Line 3 'A'	blue/red
Line 3 Tip	Line 3 Tip	Line 4 Tip	red/orange
Line 3 Ring	Line 3 Ring	Line 4 Ring	orange/red
Line 3 A	Line 3 A	Line 4 lamp	red/green
'A' circuit common (gnd)	Usually line 8 'A'	Line 4 'A'	green/red
Line 3 lamp ground	Line 3 lamp ground	Line 5 Tip	red/brown
Line 3 lamp	Line 3 lamp	Line 5 Ring	brown/red
Line 4 Tip	Line 4 Tip	Line 5 lamp	red/slate
Line 4 Ring	Line 4 Ring	Line 5 A	slate/red
Line 4 A	Line 4 A	Line 6 Tip	black/blue
'A' circuit common (gnd)	Usually line 7 'A'	Line 6 Ring	blue/black
Line 4 lamp ground	Line 4 lamp ground	Line 6 lamp	black/orange
Line 4 lamp	Line 4 lamp	Line 6 A	orange/black
Line 5 Tip	Line 5 Tip	Line 7 Tip	black/green
Line 5 Ring	Line 5 Ring	Line 7 Ring	green/black
Line 5 A	Line 5 A	Line 7 lamp	black/brown
'A' circuit common (gnd)	Usually line 6 'A'	Line 7 A	brown/black
Line 5 lamp ground	Line 5 lamp ground	1T 3	black/slate
Line 5 lamp	Line 5 lamp	1R 3	slate/black
Line 6 Tip	Line 6 Tip	1L 3	yellow/blue
Line 6 Ring	Line 6 Ring		blue/yellow
BL, AG, or spare	BL, AG, or spare		yellow/orange
SG, LK, or spare	SG, LK, or spare		orange/yellow
Line 6 lamp ground	Line 6 lamp ground		yellow/green
Line 6 lamp	Line 6 lamp	ground	green/yellow
Line 7 Tip	Line 7 Tip	ground	yellow/brown
Line 7 Ring	Line 7 Ring	ground	brown/yellow
B or B1	B or B1	ground	yellow/slate
R or R1	R or R1	ground	slate/yellow
Line 7 lamp ground	Line 7 lamp ground	10V	violet/blue
Line 7 lamp	Line 7 lamp		blue/violet
Line 8 Tip	Line 8 Tip		violet/orange
Line 8 Ring	Line 8 Ring		orange/violet
Line 9 lamp ground or T1	Line 9 lamp ground or T1	ground	violet/green
Line 9 lamp or R1	Line 9 lamp or R1		green/violet
Line 8 lamp ground	Line 8 lamp ground		violet/brown
Line 8 lamp	Line 8 lamp		brown/violet
Line 9 Tip	Line 9 Tip		violet/slate
Line 9 Ring	Line 9 Ring		slate/violet

Telos 10 digital telephone system Connection to the telephone system

To determine how your system is wired, here is what. to look for:

Tip/Ring – The name originated with the tips & rings of the patch cords telephone operators used to use to connect callers. The tip/ring pair is the actual telephone line connected thru from the incoming lines. They may be located with headphones or a telephone.

"A" leads – These tell the key system which lines are off-hook. Any time a line is selected on a phone and taken off-hook, a connection is made from that line's "A" lead to another wire called "A common." The "A" lead is at -24 volts, and "A common" is at ground potential, so when a line is selected, the "A" lead goes from -24 v to ground.

If the "A" lead connection is broken before the tip/ring is disconnected, the system puts the line on HOLD.

These wires may be located by using using a volt meter while selecting lines on a nearby phone.

There is usually only a single "A common" wire for all lines in a system.

Lamp leads- These light the lamps on the phone's line buttons. The lamps use 10 volts AC. The lamp circuit is completed through the "LG" lamp ground wires.

All of the LG wires are connected together at the KSU. Multiple wires are used to distribute the current to reduce the voltage drop when a number of lamps are lit.

In the great majority of systems, the lamp grounds and the "A" common are connected to ground, and are effectively the same.

In standard old-type "six-wire fanout systems," the A common is re-named "A1" and bussed to each line so that there are a full six leads devoted to each line – Tip/Ring, A/A1, and L/LG.

Telos 10 digital telephone system Connection to the telephone system

2.1.4 Making the Connection

Now that we know what's what, it shouldn't be too difficult to connect the Telos system.

One way is to locate a punch block which is used in your system as a central connect point. A cable with a 50-pin ribbon plug on one end and "pigtailed" on the other could then be "laid-down" on the block in the appropriate sequence. The connector would then plug into the Telos.

Another method is to provide your own punch block. This is probably the best way. You can simply unplug a nearby phone from the key system cable and insert the block. Connection to the Telos is then made as before, with another cable and plug. A number of companies offer the "66" blocks with pre-wired blue-ribbon type connectors mounted on each side.

There are, of course, many other schemes which may be used to make the Telos/phone equipment connection depending on the specific situation.

Cables, blocks, and other supplies are available from local interconnect phone companies, Graybar Electric branches, and specialty suppliers. Local companies can often make up cables with connectors to your specifications. You will certainly want to obtain the cables with connectors pre-installed, if you need them.

Incidentally, 3M makes a useful device for plugging wires onto a punch block. It's called an "ST" connector. You simply squeeze them onto the wires with pliers, and they push on over the punch block terminals. This makes it easy to change connections to re-arrange line appearances, or whatever, when you need to. The ST devices are available from telephone supply houses and some broadcast distributors - such as Allied, in Richmond, IN.

As this is written, a convenient source for telephone supplies are the Radio Shack Computer/Phone Center stores. They have 25-pair cables made up with plugs, blocks, junction devices, and many other accessories. They have the tool available for mass termination of the 50-pin telephone connectors, so they should be able to make-up custom cables. They even have a 1A2 compatible

- key service unit and telephones. The Shack's prices are very high compared to distributor prices, but you can't beat the convenience. The cables, connectors, etc. are usually not on display - you have to ask for them. They keep 'em in the back room.

Telos 10 digital telephone system

2.1.5 TELOS Phone Connector pin-out

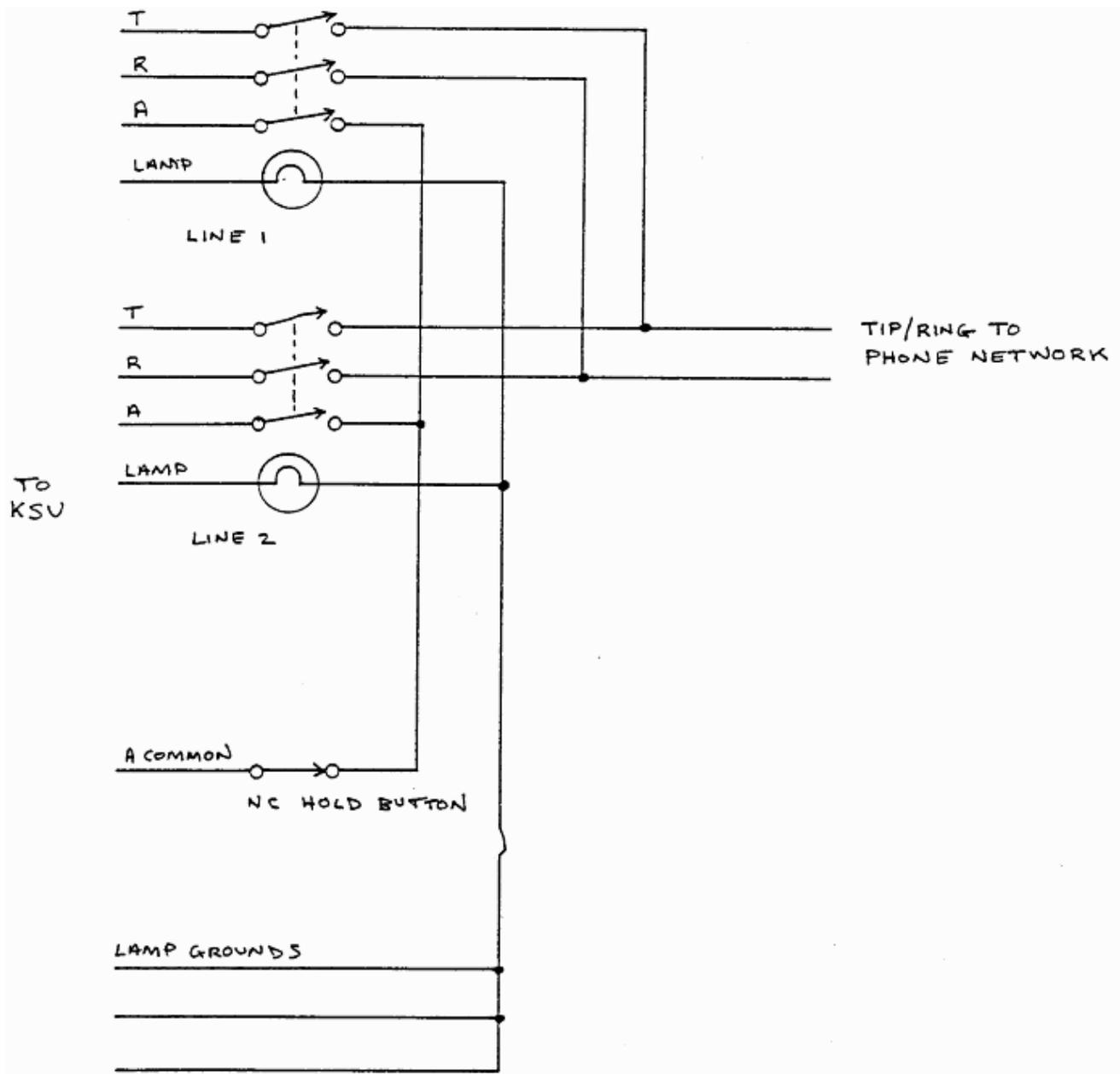
Wire Color	Pin #	Function	
white/blue	26	Line 1 RING	
blue/white	1	Line 1 TIP	
white/org	27	Line 1 lamp	
org/white	2	Line 1 "A"	
white/grn	28	Line 2 RING	
gm/white	3	Line 2 TIP	
white/brown	29	Line 2 lamp	
brown/white	4	Line 2 "A"	
white/slate	30	Line 3 RING	
slate/white	5	Line 3 TIP	
red/blue	31	Line 3 lamp	
blue/red	6	Line 3 "A"	
red/org	32	Line 4 RING	
org/red	7	Line 4 TIP	
red/grn	33	Line 4 lamp	
gm/red	8	Line 4 "A"	
red/brown	34	Line 5 RING	
brown/red	9	Line 5 TIP	
red/slate	35	Line 5 lamp	
slate/red	10	Line 5 "A"	
black/blue	36	Line c1 RING	
blue/black	11	Line c1 TIP	
black/org	37	Line c1 lamp *	
org/black	12	Line c1 "A"	
black/grn	38	Line c2 RING	
grn/black	13	Line c2 TIP	
black/brown	39	Line c2 lamp *	
brown/black	14	Line c2 "A"	
black/slate	40		
slate/black	15		
yellow/blue	41		
blue/yellow	16	"A" common	
yellow/org	42	line 1 RING relay	NC contacts
org/yellow	17	line 1 TIP relay	NC contacts
yellow/grn	43	lamp ground	
grn/yellow	18	lamp ground	
brown/yellow	44	lamp ground	
yellow/brown	19	lamp ground	
**yellow/slate	45	line 6 RING	
slate/yellow	20	line 6 TIP	
violet/blue	46	line 6 lamp	
blue/violet	21	line 6 "A"	
**violet/org	47	line c3 RING	
org/violet	22	line c3 TIP	
violet/grn	48	line c3 lamp	
gm/violet	23	line c3 "A"	

Telos 10 digital telephone system

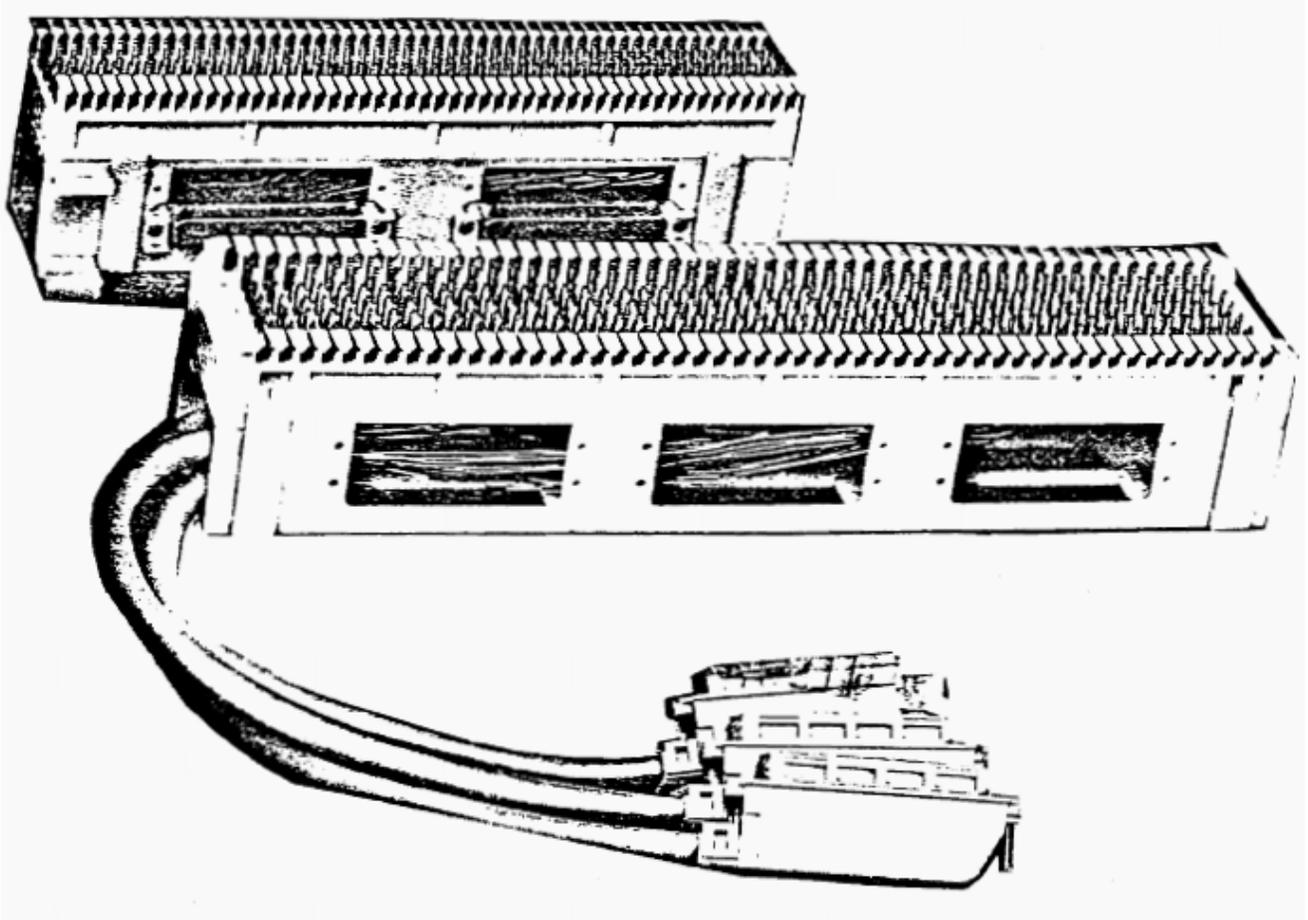
violet/brown	49	line	c4 RING
brown/violet	24	line	c4 TIP
violet/slate	50	line	c4 lamp
slate/violet	25	line	c4 "A"

* often confused - beware

** Sorry. . .PC board error: Pins 45 and 47 are reversed. Swap these when connecting.



Telephone Switching
Circuit



Pre-wired and connectorized "66" punch-blocks

Telos 10 Digital Telephone system

2.1.6 Single-line operation

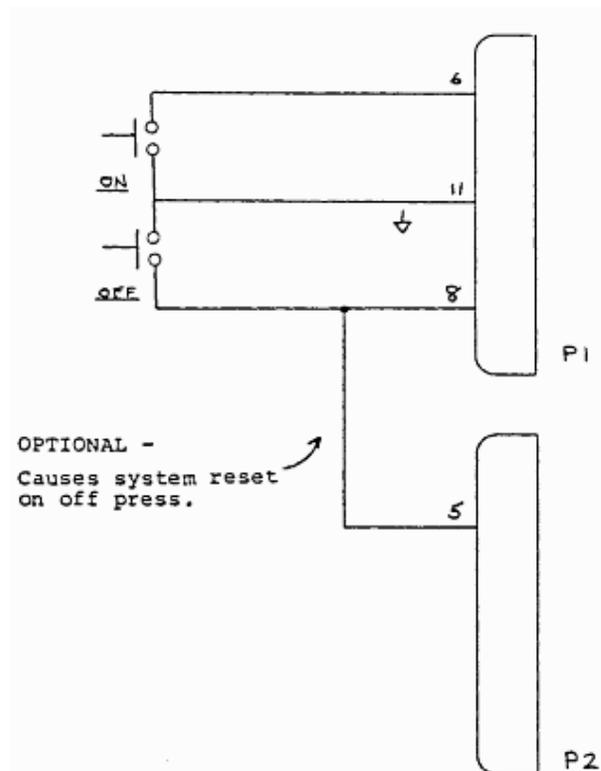
In simple single line operation, the Telos may be inserted between a phone line and a telephone, since the Normally Closed contacts of the line #1 relay are brought out to the phone connector. The phone may then be used normally until the Telos is activated. A simple two button on-off control would then be connected to the appropriate pins on the control connector.

If it is desired to use a multi-line phone as a select "front end", the tip and ring connections inside the phone may be broken between the switch bank and the network and routed through the Telos. Of course, the on-off control function will have to be provided. The on-off control could be provided from the studio console channel switching logic.

Single-line control connections

The main line #1 off and on connections are used. They are simple closures to ground.

Both main and conference off button inputs can be paralleled in order to cause a full system reset each time the off button is pressed.



2.2 Installation - Audio Connections

2.2.1 Mix-Minus

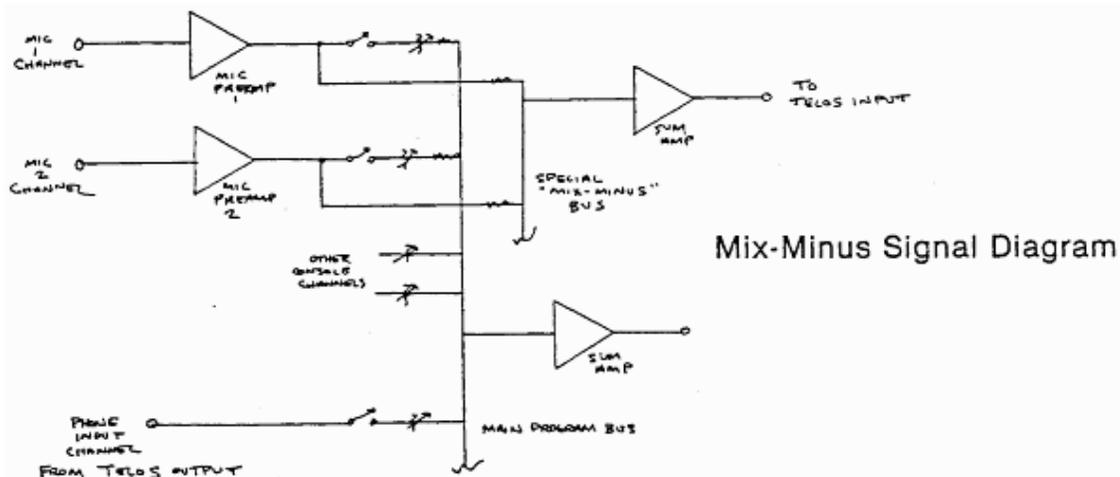
While it is possible to insert the Telos downstream - after the board main output and before the station audio processing - the best way is to obtain a "mix-minus" output from the board to feed the Telos input and route the Telos output into a board input. The advantages of the board-created mix-minus approach are:

- The operator has convenient control over phone level.
- The cue speaker can be used to provide hands-free answering capability.
- Reliability and audio quality of the station audio path is improved since station audio does not pass through the additional Telos stages.

Mix-minus, or "foldback", as it's sometimes called, is simply an audio mix of all sources you want to feed the phone - minus the telephone itself.

Inside the console, a new mix bus has to be created or assigned. This bus sums all the sources you want callers to hear. It must not, of course, include the telephone input. If it did, feedback would result.

Most newer consoles have built-in provision for mix-minus. For example, the Pacific Recorders BMX units have two "auxiliary" summing busses, one of which can be used to create a mix-minus. The summing resistors would then be added to the desired input modules, and a summing amplifier connected. Pacific Recorders makes a module which performs this function.



Telos 10 digital telephone system Audio Connections

With consoles which do not have internal mix-minus provisions, you will have to improvise. One option, incidentally, is to use an external mixer (such as the ubiquitous Shure M-67) to parallel the main and guest mike feeds to the board.

2.2.2 Inputs

The main and aux Telos inputs are internally summed, which may be useful in creating a mix-minus scheme.

Input levels of from approx. -30 to +4 dbm can be accommodated.

The Telos inputs are active balanced. If you feed the input from an unbalanced source, the "-" input side should be grounded, with the signal connected to the "+" side.

2.2.3 Outputs

The Telos output simply feeds a board input. The main and aux outputs are isolated, but both have the same phone audio present. The two outputs may be useful when you need to feed phone audio to another device, such as a studio tape machine.

In balanced mode, the output level is 0 dbm when the meter is at zero. The internal jumper plug increases meter sensitivity 10 db so that when the plug is in place, output level is -10 dbm at meter zero. When the output is taken unbalanced, the output level is 6 db lower.

```
INTERNAL VU JUMPERS  
[ ] -VU sensitivity incr. 10 db  
  o o }  
  o o } nothing  
  o o }
```

The outputs are active balanced (Uh, excuse me - "active differential, if you prefer). If you desire an unbalanced output, it should be taken from one side to ground. Do not short the unused output side.

In troublesome cases, you may have to add external transformers.

IMPORTANT NOTE:

Since there is usually some leakage audio present at the Telos output, you should try connecting the output with the phase both ways - almost always, one way will sound better than the other when listening to the combined announcer and phone audio. To find out which way is best, use a board mix with the phone audio potted up high and listen for least quality degradation.

2.2.3 Outputs (cont) - Internal "Mix-Minus"

As mentioned before, it is possible to create a mix-minus within the Telos unit. This would allow the unit to be used downstream of the console.

On the PC board near the phone transformer are jumper pins which allow this. The main input can be summed with the phone audio to produce the main output; the aux input can be summed with phone audio to produce the aux output. You will also have to add an appropriate value summing resistor to the R-134 and/or R-136 positions on the PC board. These are left off so you can set the "through" gain to meet your requirements.

A look at the audio schematic should make things clear.

Telos 10 digital telephone system

2.2.4 Audio Connections - P4, DE-25 rear panel connector

Pin 1	-	main input	+
2	-	main input	-
3	-	aux input	+
4	-	aux input	-
5			
6	-	aux output	+
7	-	aux output	-
8	-	main output	+
9	-	main output	-
10			
11			
12			
13		ground	
14		"	
15		"	
16		"	
17		closure to ground for call timer start (20ma max)	
18		Unassigned logic output	
19		ground	
20		"	
21		"	
22		"	
23		"	
24		"	
25		"	

Telos 10 digital telephone system

2.2.5 Feedback reduction

Sometimes, when the system is being used with an open speaker, trouble with feedback may occur. Generally, this happens only with poorer phone lines, or with very weak callers requiring lots of gain in the phone-to-speaker path.

Some suggestions for solution of this problem:

1. First, try repositioning the mikes and/or speaker. Of course, it also helps to use mikes and speakers that are directional. In typical broadcast studio application, EV RE-20's and Shure SM-7's have proven appropriate.
2. Add equalization to the monitor path. Acoustic resonances usually cause pronounced peaks in the "feedback response" of a sound system. Since the largest peaks generally occur at just a few frequencies, reducing system gain at these frequencies with a graphic or notch EQ helps tremendously. UREI makes a special EQ unit just for this purpose.
3. Use a pitch shifter on the SEND input to the Telos. Pitch shifters are very useful in this application. Good results have been obtained with inexpensive MXR brand units.

The foregoing is intended to help in those situations where you MUST have an open speaker. Whenever possible, it is best to use headphones to hear callers. When you have an open speaker, the on-air phone audio has both a direct and an acoustic path - from the speaker to the announce mike(s). Depending on the relative levels, phone audio quality may suffer. Also, callers hear themselves fed back at high volume by the same path - from the speaker to the announce mike, which also feeds the Telos input. This can be disconcerting.

The best thing in normal radio studio use is to have the announce mikes active to the phone system input even when the mike channel is switched off, and have the phone monitor speaker mute when the mike is turned-on. Thus, the system can be used as a speakerphone-like device to take calls off the air - but, when the call is to be used on-air and the announce mike is on, the speaker is muted.

Telos 10 digital telephone system

2.3 Installation - Control Connections

The two DB-25 rear panel control connectors, P1 & P2, are used to connect the control buttons and indicators.

Refer to the interface section schematic for connector pin-out, etc.

The buttons make simple closures to ground. The "OFF" lamps are 5v; the line lamps are 10 v.

Telos 10 digital telephone system

2.4 Installation – Conferencing with two units

To get high-quality conferencing, two hybrids are required.

Hybrid #1 is fed with the announce mike(s) and the output of hybrid #2. Hybrid #2 is also fed the announce mike audio and the output of hybrid #1. This way both callers get the announce audio and the other caller audio.

There are two ways to do this:

1 – Create TWO mix-minuses inside the station console. One for each hybrid. This is probably the best way since operator board gain-riding on the phone channels makes it more likely that each hybrid is getting the right level from the other despite varying incoming caller levels.

2 – Use the Telos AUX inputs and outputs. Hybrid #1's AUX input is fed from the #2 AUX output and vice-versa. Both hybrids are fed the same announce mike audio on the MAIN input. Each hybrid's output goes to it's own board input. Of course, system gains will have to be matched so that audio from the announce mike(s) matches that from the hybrids.

With poorer hybrids, a special kind of feedback called "singing" is often a problem when the hybrids are connected for conferencing in this way. The phone company, faces this problem when they apply bi-directional repeaters to standard two-wire phone lines. With the superior Telos trans-hybrid loss performance, considerable gain can be inserted in the caller-to-caller path before feedback is a problem. However, since the Telos units have both input and output AGC, and since some phone lines are very bad and some callers are very weak, feedback could occur. Experimentation with gain settings will usually produce the fix.

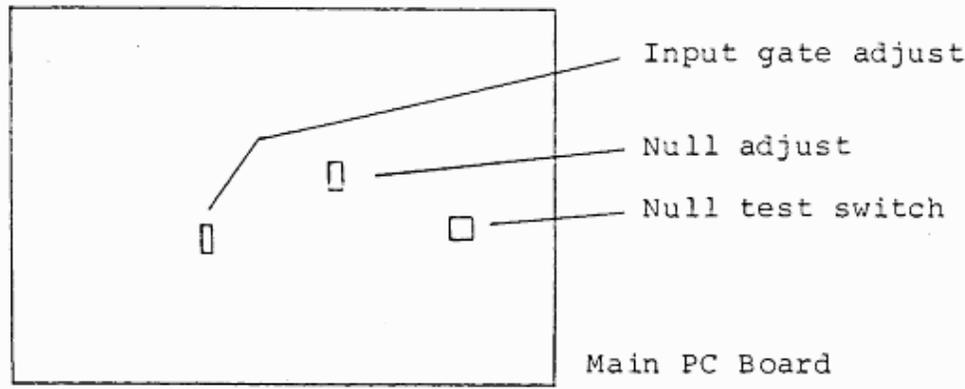
Control connections –If you need fewer than 10 phone lines total, divided 6 and 4 as with a single unit, just plug the P-1 control cable into the #1 hybrid and the P-2 cable into the #2 hybrid. All of the 6 main phone line control and indicator functions use P-1, while the 4 conference line functions use P-2.

A special version of the control software EPROM is required to allow each unit to work with the full 10 lines each.

Section 3 - OPERATION

3.1 Operation - Internal adjustments

There are two trimpots which may require adjustment:



3.1.1 Course Null

The course null pot adjusts the null on the simple active hybrid stage. This adjustment is uncritical, since the auto-adaptive balancing circuit can accommodate a wide range of conditions. The idea is to set the average, or central range, so that the auto-balancing section works optimally.

The DIP switch near the right edge of the circuit board should be switched to the NULL SET mode. This applies white noise to the hybrid input and switches the meter to a point after the active hybrid, but before the digital balancing section. The COARSE NULL pot should then be adjusted for a minimum indication on the meter. If a number of lines are to be used, the pot should be set to provide the best average null for all lines.

3.1.2 Input Gate

This adjustment establishes the input gate low threshold.

The actual gate threshold is adaptively varied to always be a few db below the long-term signal level. The INPUT GATE pot sets the lowest level the threshold is allowed to reach, and should be set so that gating always occurs during long periods of silence with normal room noise present at the input.

3.2 Operation - Front panel controls, indicators Input Level

Controls input level. Set so that VU meter has a normal indication in the input position.

Input Gate

Lights when input gate occurs. Actually, this circuit is both a gate and an expander, so, the LED indicates expansion as well.

Adapt LED

Shows when the digital balancing network is working. Useful to confirm that the digital signal processing section is OK.

Active LED

Lights when a line is selected. Useful to indicate operation of the Z80 and peripheral chips.

Compress/Expand Drive control

Sets the input level to the output compress/expand section.
Set higher to pick-up weaker callers; lower to reduce feed-thru (leakage).

Compress, Expand, and Gate LEDs

Indicates operation of the output compress/expand section.

Override control

Sets the amount of gain reduction, or "ducking," which is applied to the phone caller audio. Almost always, some override is useful - so, the "12 o'clock" position is normal. The center position corresponds to 6 to 10 db of ducking. This amount is barely noticeable.

Since advancing the override control provides improved "dynamic" hybrid performance, very poor phone lines may require more override ducking. This improved performance comes at the expense of full bi-directionality, of course. Unfortunately, with the present state of the art, this is the nature of things - very poor phone lines will require some sacrifice of full-duplex hybrid operation.

Output Level control

Sets output level, naturally.

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3.3 Operation - Line selection, etc

The line buttons are divided into two groups - "regular" (6 lines) and "conference" (4 lines). Generally, the conference buttons are used for unlisted lines to put "newsmakers" on the air. This way, the newsmakers can stay up while you select incoming calls on the regular bank.

Operation is very much like a multi-line desk set phone. To select a line, punch the appropriate button. Punching-up a line cancels all others active in the group. If you want to put two callers on the air from the same group, hold the one you've got while you punch the one you want. OFF and HOLD work as you would expect.

The buttons are illuminated straight through from the key service unit, so off-hook and hold conditions caused by any phones connected to the system are indicated.

Special Note:

On occasion, the unit may stop working for no apparent reason. This can happen when there is a brief AC power failure or surge. Just as a personal computer could get "locked-up" when there is a power problem, the Telos microprocessor gets confused. When this problem occurs, press the main and conference "OFF" buttons simultaneously. A complete system reset is performed.

Section 4 - Theory of Operation

4.0 System Description/Theory of Operation - Digital

4.1 Overview

Refer to the system block diagram.

The heart of the control section is a "textbook" microprocessor control system, using the common Z80. If you are unfamiliar with microprocessor systems, a visit to a bookstore may be a good idea. There are a good number of books available which explain microprocessor systems from the ground-up. A good one is "Build Your Own Z80 Computer" by Steve Ciarcia, published by Byte Books.

The system functions by executing the machine-level software programmed into the 2716 EPROM. This software is not considered to be proprietary, so, an assembler source listing is available by request.

When no line is selected, the microprocessor system is constantly polling the pushbutton input lines to see if a press is present. When a line button is pressed, the Z80 reads the digitized audio from the A/D, sends it to the signal processor chip, then retrieves it from the signal processor and sends it to the D/A. The signal processor operates as a peripheral to the Z80.

When a line is first selected, the noise generator is turned on and the audio is prevented from reaching the D/A. This happens for about 300ms. Two purposes are served. The "clunk" sound of the line engaging is muted, and the auto-adaptive balancing is accomplished before the call goes to air.

Filtering and input and output AGC are done with conventional analog circuits. In order to achieve top distortion and dynamic range performance from the digital section, a companding scheme is used. The two inputs to the digital section are compressed, and the digital section output is routed through an expander. A special-purpose companding chip is used to perform this function - the Signetics NE570. Also, pre and de-emphasis are applied.

Only one AID converter is used for the two inputs to the digital section. A FET switch IC - a DG201 - is used to switch the appropriate input to the A/D at the right time. The FET switch operates under the control of the digital section.

4.2 Digital Section

4.2.1 Circuit Description

Refer to the digital section block diagram, and the digital and interface section schematics.

The logic input and output to the Z80 is handled by U8 and U15
– the two 8255 peripheral I/O interface (PIO) chips.

The pushbutton inputs are buffered by the four 1498 chips. The capacitors slow the response of the button inputs so that transients are prevented from false triggering the inputs.

The various logic outputs are buffered by four 2003A peripheral driver chips. The 2003A's are Darlington transistors in a chip package. The 2003A's drive the relays directly, the "A" leads through the opto-isolators, and the front panel LEDS through current limiting resistors.

The system clock is provided by U1, a packaged oscillator module followed by U2 (7493), which divides the oscillator output to produce the required frequencies of 8mhz for the signal processor, 4mhz for the Z80, and 2mhz for the two 2912 filter chips.

Reset is provided by U11 (74LS14) and associated parts. Operation of this circuit is straightforward. When power is first applied, C1 begins charging through R5. Until the voltage reaches the 74LS14 threshold, the system is held in the reset mode. When the threshold level is reached at U11-13, the RES line goes low. The simultaneous pressing of the two line "OFF" buttons forces a system hardware reset by causing U11-4 to go low, thus discharging the capacitor.

U10 (74LS138) provides address decoding. The chip selects of the various peripheral devices are driven by the 'LS138.

4.2.2 Digital Section troubleshooting

A failure in the digital section may be such that it affects the operation of a certain function only, or it may cause the entire system to "crash."

If a control problem with some, but not all, lines develops -it is probably caused by a failure of a component in the pushbutton interface circuit or the relay or "A" lead driver circuits. The first step is to determine if the pushbutton signal is making it through the 1498's to the 8255 PIO chips. If it is not, the 1498 or related part is at fault. If the signal is making it to the PIO, check to see if the appropriate PIO output is functioning. All line select logic is active high, so you should see the PIO output go high when a line is selected. If the PIO output is OK, you have a bad ULN2003 driver chip, a bad relay, or, if an "A" lead problem, a bad opto-isolator. Note that, since the ULN2003's are really just transistors, a high level at the input causes a low at the corresponding output.

If the PIO chips are getting the right inputs, but producing incorrect outputs, or if the whole system "just doesn't work" -the problem is less easy to pin down.

Because microprocessor systems work using a bus approach, any chip on the bus could cause strange and unpredictable failure modes. Usually, you will find either the Z80 or the EPROM is bad. Sometimes, a peripheral chip on the bus may short a data or address line to either +5 or ground. A scope on these lines while chips are removed from sockets will often point up this sort of problem.

Of course, you will want to make sure the +5 power supply and the reset signal are OK. Incidentally, a bar over a logic line name means that that signal is active LOW. For example, the RESET signal is active low at the Z80.

The worst kind of problem to track down is a failure of the audio functions of the digital section with the control functions still working. If you have an audio absent or audio distorted problem, and you are sure it's in the digital section, there are a number of chips which could be the cause. The A/D and D/A chips are suspect, as is the '7720 signal processor. Also, the PIO output port which runs the FET switch and the 74LS138 or 74LS14 could be faulty. The PIO port and the '138 and '14 chip operation can be verified with a scope.

It is possible for the Z80 or EPROM to cause an audio problem, but it would be very unlikely to this without having an effect on the control functions.

The front panel LED's may be of some aid in tracking down problems. The "active" LED is controlled by the Z80, EPROM, and PIO chips. The "adapt" LED is driven by the signal processor chip.

A general strategy for digital section failures, when a 'scope check doesn't point up anything obvious, is to substitute the readily available chips, and see what changes. For example, the Z80 is cheap and commonly available - so it may be a good idea to pop another one in before you do anything else. Same goes for the 8255 PIO chips, and the EPROM, if you have a spare.

4.3 Audio Section

4.3.1 Circuit Description

Refer to the audio section block diagram and schematic.

U38 is a simple active differential input stage. The main and aux inputs are summed in u37 and fed to the input level pot.

The send high-pass filter, a fifth order Chebychev type, removes audio below 200 HZ. Note that C34 and R92 form a high pass time constant, and are part of the overall filter. Thus, the response measured at U37-1 will have a rise at low frequencies.

The "send gated AGC" circuit uses an LM13600 gain control element. An important point to note is that the control voltage cannot be seen at pin 1, as you might expect. This is because the 13600 control pin is a CURRENT input, not voltage. Thus, to observe the control voltage, look at U22-7. As input level is increased, the voltage at 022-7 goes more negative. CR7 rectifies the audio. R14 sets attack time, R10 sets release time. The FET switch is open during gating, holding the AGC control voltage constant. The 022 section following C7 is a buffer, the next U22 section is a summer. The summer is used to provide expansion during gating. When a gated condition exists, the voltage at U22-8 is high. R15, C6, and R11 form a voltage divider and time constant. The result is a "floating expander" - that is, the expansion changes gain a fixed amount relative to the "platform" established by the AGC. The gate circuit is unique in that the gate threshold varies depending on the long-term average signal level at the input. The pot, thru CR9, sets the lowest possible threshold level.

The four diodes form a hard limiter to protect the phone line as required by FCC rules. R20 and C8 are a simple low pass filter to prevent the harmonic content produced by clipping from reaching the U25 low-pass filter.

U36 is the active hybrid and phone line driver.

The hybrid input and output are required by the digital section. Both follow identical parallel paths consisting of a compander, anti-alias filter, and pre-emphasis.

The compander function is provided by special compander IC's -the NE570. Pin 16 is a rectifier output, which feeds C37 (C38). These capacitors are the compander time-constant-determining parts. The two 028 sections are "analog OR gates." That is, whichever input is higher (at pin 10 or pin 12) determines the output voltage at the top of R56. This voltage is used to

Telos 10 digital telephone system

control the gain reduction identically in all of the three NE570 sections. The audio-path input and output coupling capacitors are needed because the 570's are operated from a single-ended supply.

Pre-emphasis is used to improve system noise performance.

018 is a FET switch which operates under control of the digital section to select the appropriate input for the A/D converter. The A/D is presented alternately with the hybrid-in and the hybrid-out signals.

The D/A current output is converted to a voltage by 026 and is filtered by 027 to reconstruct a continuous signal.

De-emphasis is provided by R48 and C3. Adjustment of R48 value results in a change in high-end frequency response. The value is selected to provide a slight high-end boost.

The expander uses the same control voltage as the input 570 stages.

The first half of 032 is used for the "override" or "ducking" function. Three 026 sections rectify, filter, and amplify the system input audio to provide a control voltage at U26-14.

The second 032 section is used for the output compress/expand circuit. The expander has a very fast time-constant, while the AGC is fairly slow. The 033-5,6,7 section is a high-gain amplifier which feeds the rectifier diode. The rectified signal goes to C49 and R112, which set the time-constant. The control voltage is buffered by 033-8,9,10, which is also half of a "negative analog OR gate" circuit - the other half being U35-1,2,3. Whichever circuit - the AGC or the expander - has the most negative voltage is the one which determines the control voltage. The control voltage may be observed at the junction of CR22, CR28, R62, and R11. The more negative this voltage, the more the gain is reduced in 032. The gate circuit is the same "adaptive-floating" type used in the input AGC section. The gate threshold is set to the long term average level.

The output sections are the common active differential type. The first stage can also be used as a summer to produce an internal mix-minus.

The little rectangles labeled "RFI" are pi-filter RFI filters.

The noise generator is turned-on under control of the digital section. During the 300 ms mute period, the noise is turned-on. The noise allows the Hybrid to adjust to each line before the call goes to air, and serves as an on-air warning to

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the caller. The noise is produced by an IC noise generator. A 555 produces a single frequency tone, which is also mixed-in.

4.3.2 Audio Section - Troubleshooting

Troubleshooting should be relatively simple in the audio section. A signal level chart is provided on the next page. The circled numbers on the schematic are "test points" which correspond to the indicated points on the chart.

Simple tone and oscilloscope tracing should quickly determine the location of any problem.

The only potentially difficult problems might be caused by the closed-loop nature of the two AGC circuits. Thus, when you have a problem in the AGC sections, it might be helpful to remove the FET switch, 018 from its socket. This opens the loop. The time-constant capacitors can then be fed with an external voltage to see if the gain control is working. During development, it was noticed that the LM13600 was somewhat more likely to fail than other parts. It usually gets very hot when it goes.

Signal Level Chart - "send" sections

Apply 800 HZ at 0 dbm, balanced, to the input. Adjust the input level control for a 0 VU indication, with the meter switch in the input position. System must be active so that the input AGC can ungate.

<u>Point</u>	<u>Signal level</u>
1 (U38-1)	<u>2v P-P</u>
2 (U37-8)	<u>8v P-P</u>
3 (U37-7)	<u>1v P-P</u>
4 (U37-1)	<u>2.5v P-P</u>
5 (U37-14)	<u>20v P-P</u>
6 (U22-8)	<u>-8v DC</u>
7 (U21-8)	<u>.4v P-P</u>
8 (U22-7)	<u>-10v DC</u>
9 (U22-14)	<u>+ .2v DC</u>
10 (U22-1)	<u>1v P-P</u>
11 (U25-4)	<u>1v P-P</u>
12 (U36-1)	<u>1.4v P-P</u>
16 (U31-16)	<u>+1.4v DC</u> (varies with input change)
17 (U31-7)	<u>.7v P-P</u> on 7.5vDC
20 (U25-16)	<u>1v P-P</u>
22 (U28-1)	<u>2.5v P-P</u>
"Override" circuit:	
28 (U26-8)	<u>3.5v DC</u> (varies with input change)
29 (U26-14)	<u>-10v DC</u> (ovr cont at 12 o'clock)

Telos 10 digital telephone system Troubleshooting

Signal Level Chart - "receive" sections

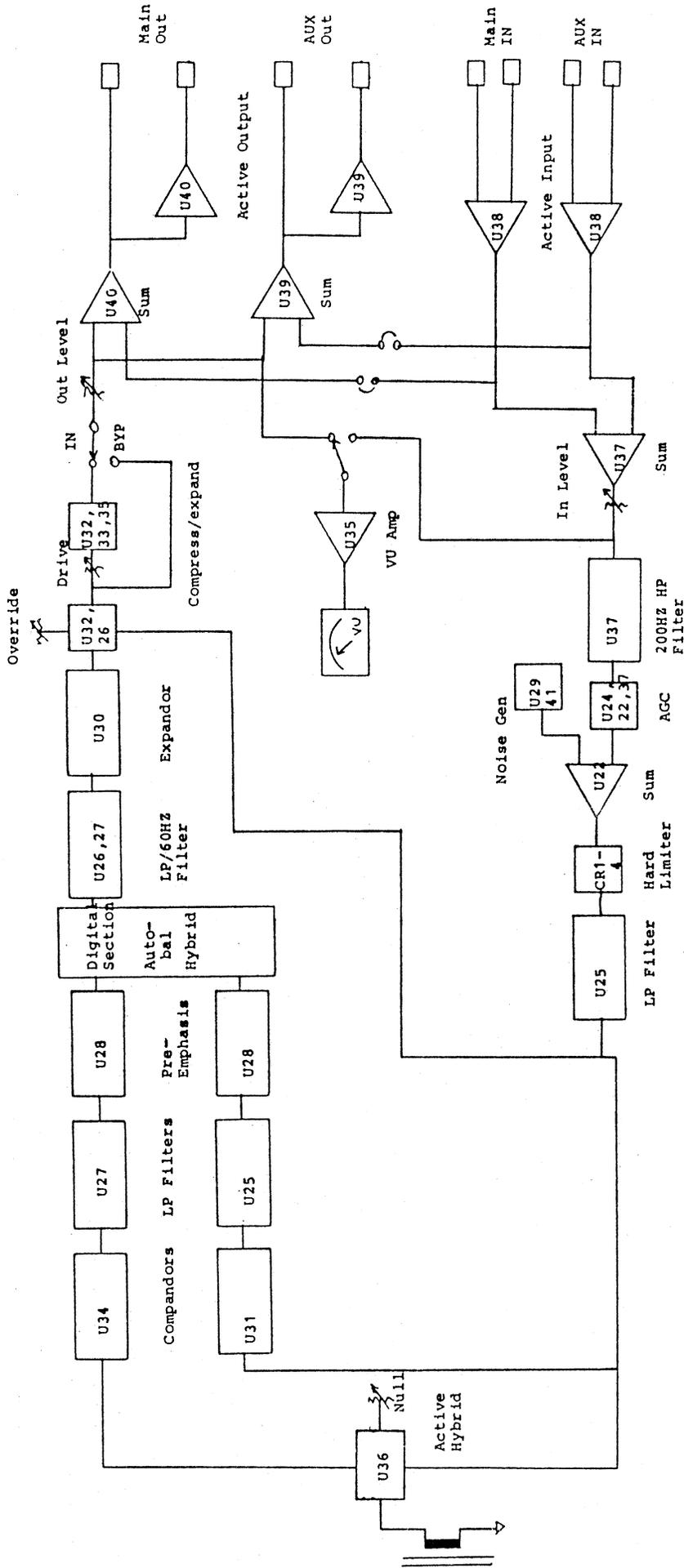
Apply 800 HZ at -20 dbm (balanced) to a telephone line input. Select this line, so system is active.

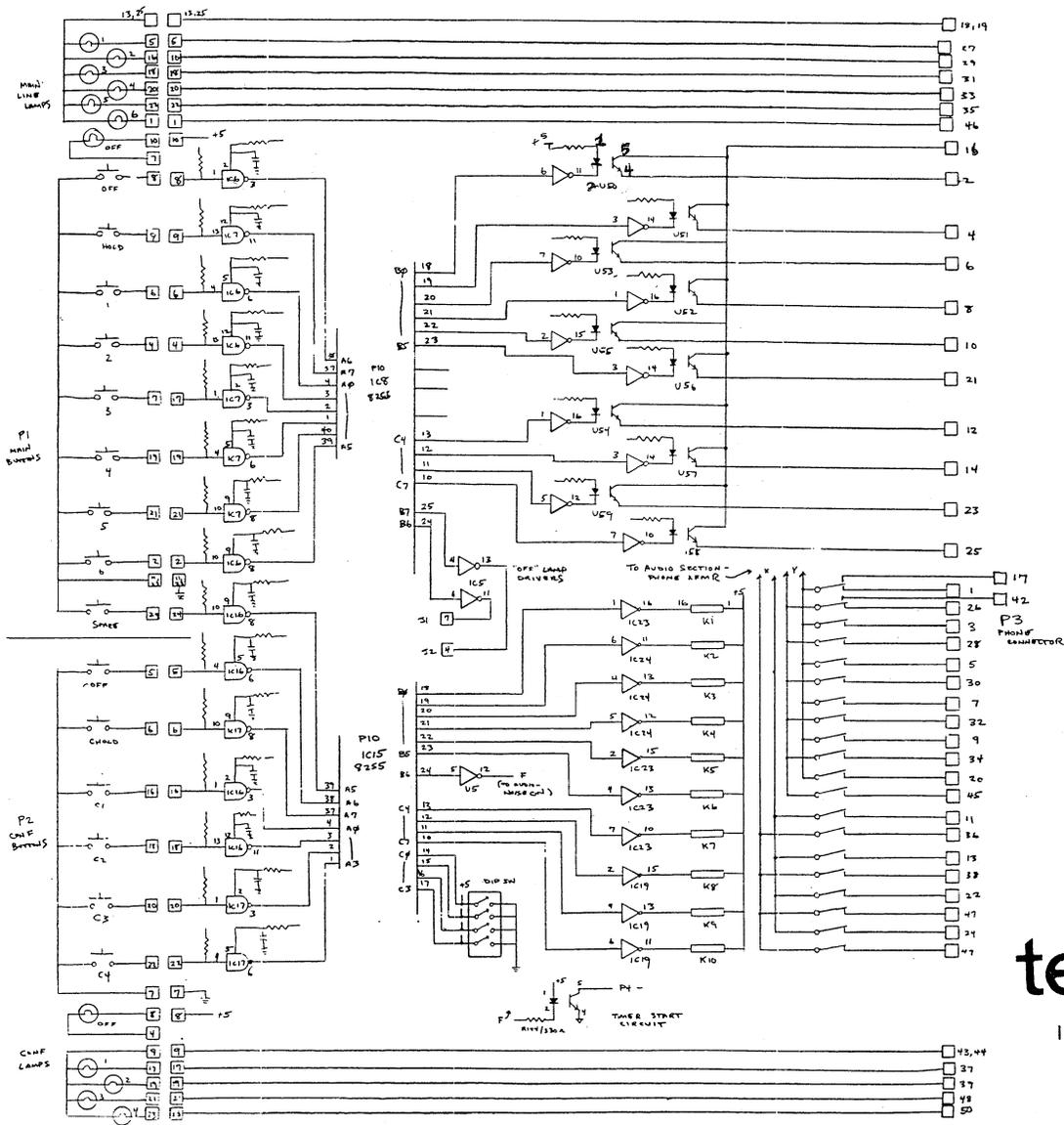
Set COMP/EXP DRIVE to 12 o'clock, COMP/EXP switch to IN, and adjust OUTPUT LEVEL to 0 VU with meter selected to output.

Turn OVERRIDE control all the way down (counter-clockwise).

Point	Signal level	Point	Signal level
13 (U36-7)	<u>.4v P-P</u>	30 (U32-9)	<u>2v P-P</u>
14 (U34-16)	<u>+1.3v DC</u>	31 (U35-14)	<u>4v P-P</u>
15 (U34-7)	<u>.6v P-P ON 7.5VDC</u>	32 (U33-7)	<u>30v P-P (CLIPPED)</u>
18 (U34-1)	<u>+1.2v DC</u>	33 (U33-4)	<u>-14v DC</u>
19 (U27-16)	<u>.8v P-P</u>	34 (R62)	<u>-10v DC (APRISK)</u>
21 (U28-7)	<u>2v P-P</u>	35 (U40-1)	<u>1v P-P</u>
23 (U26-1)	<u>1v P-P (dig sin)</u>	36 (U39-1)	<u>1v P-P</u>
24 (U27-4)	<u>1v P-P</u>	37 (U40-7)	<u>1v P-P</u>
25 (U30-7)	<u>.2v P-P (on 6vdc)</u>	38 (U39-7)	<u>1v P-P</u>
26 (U32-8)	<u>.4v DC</u>		
27 (U33-1)	<u>3v P-P</u>		

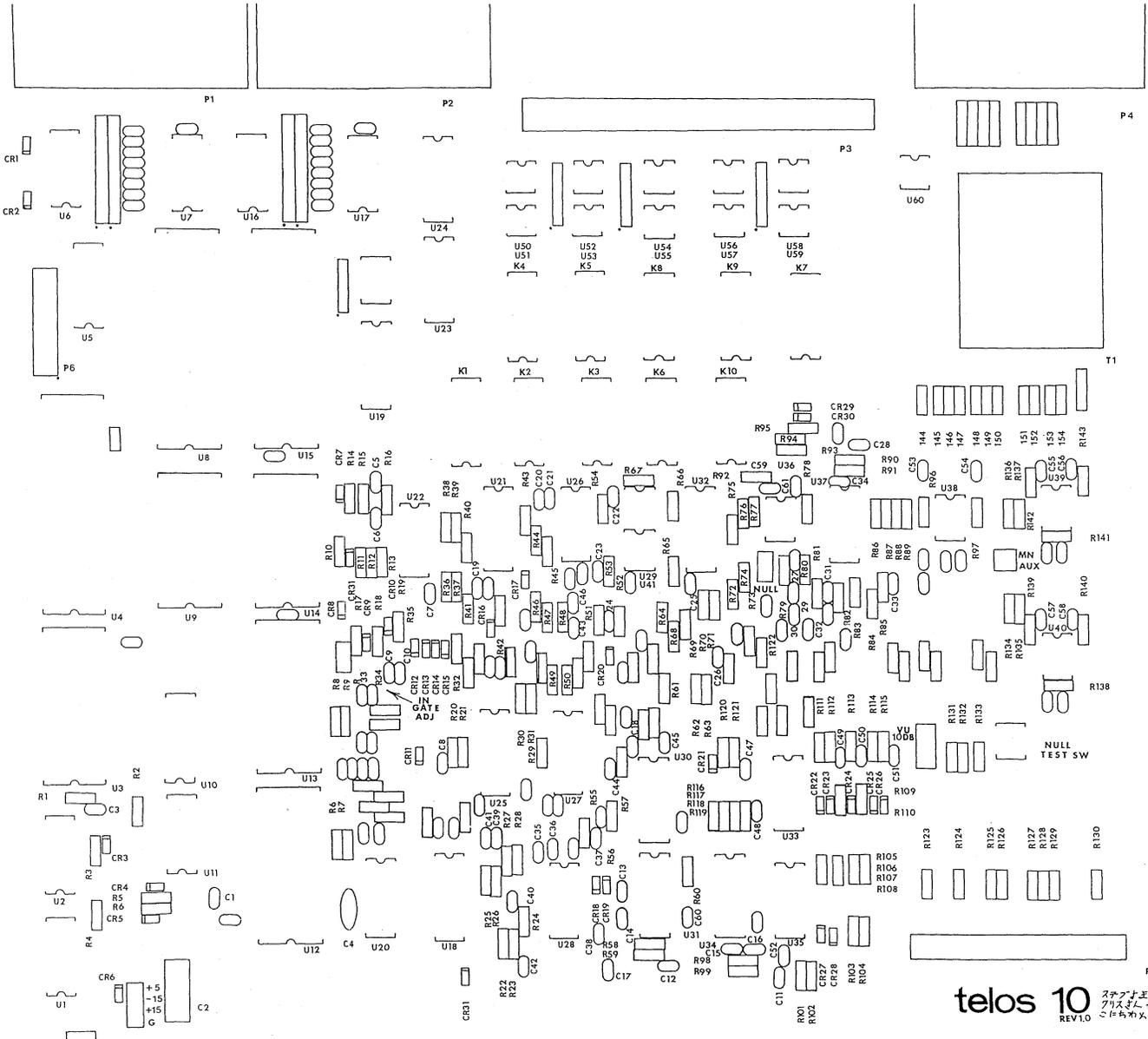
Section 5 - Drawings



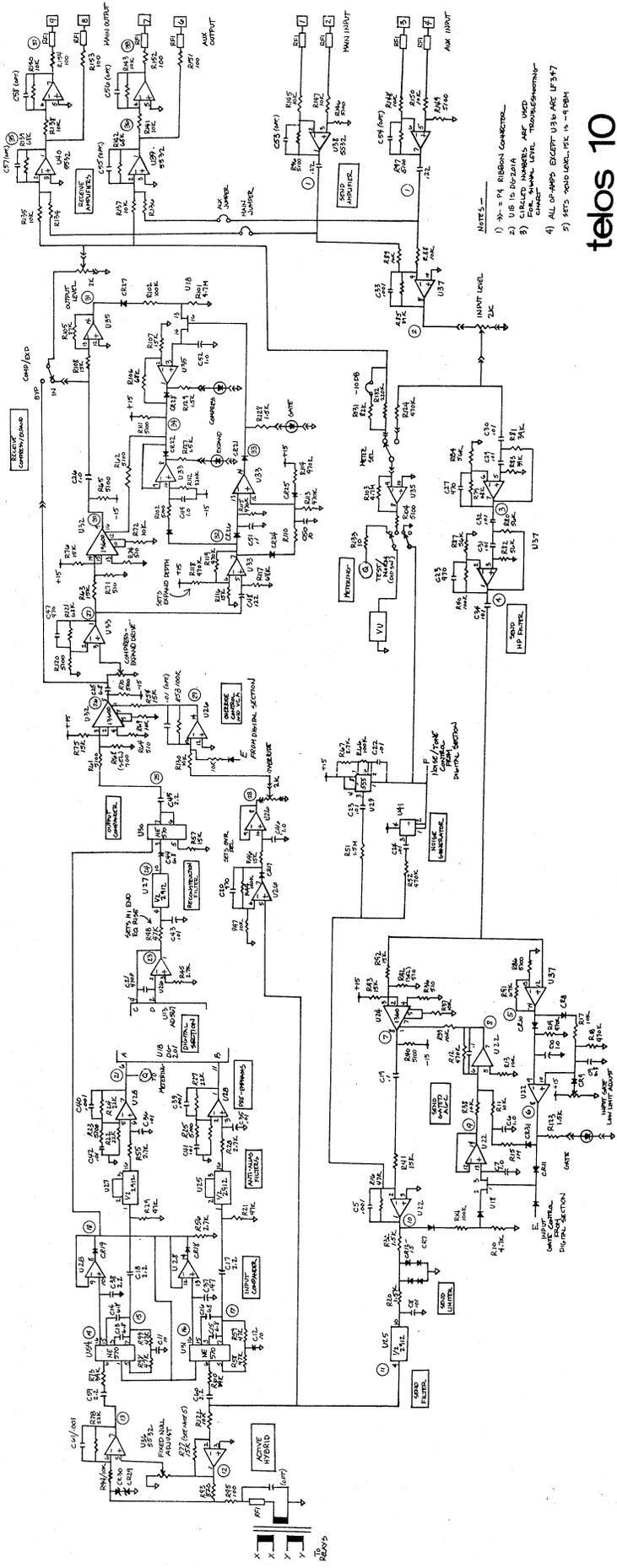


telos 10

Interface Schematic



telos 10 ステアリングシステム
 REV1.0 グリスマン社
 コムウェアの本部



- NOTES—
- 1) 20 = 14 RIBBON CONNECTOR
 - 2) 20 = 14 RIBBON CONNECTOR
 - 3) CIRCLED NUMBERS ARE USED FOR SIGNAL LEVEL TRANSFORMATIONS
 - 4) ALL CAPS EXCEPT US3 ARE US347
 - 5) R25 PHOTO CELL, PL. 10-11528

telos 10

digital telephone system
Audio Schematic

Section 6 - Parts List

This section will not be the usual "laundry list" of all of the parts used in the Telos. Rather, information will be given to help obtain the more unusual parts for those who choose to do component-level troubleshooting.

The following parts distributors are suggested sources for many components:

Digi-Key Corporation
P.O. Box 667
Thief River Falls, MN 56701
(800) 344-4539

Very fast service. Will accept telephone orders. Open accounts for radio stations or charge cards. Good source for caps and resistors, ribbon cables and connectors, National and Signetics IC's, Omron relays.

Jameco Electronics
1355 Shoreway Road
Belmont, CA 94002
(415) 592-8097

Relatively fast service. Telephone orders. Open accounts for radio stations or charge cards. Best for digital IC's.

Newark Electronics
See phone book for local branch or call home office at (312) 638-4411.

Large distributor. Higher than average prices but still welcomes small orders from radio stations.

Arrow, Hamilton-Avnet, Pioneer, etc.

These distributors have branches in many major cities and are oriented to serving multi-megabuck companies. They will take smaller orders and will set up open accounts for radio stations but they usually prefer not to. They are, however, the only source for some specialized parts.

Now, on to the parts....

Main Unit Chassis

VU meter	Sifam	AL29WF	Selco Sales Company (213) 921-0681
11 mm knob	Sifam	S110-125	
Knob cap	Sifam	C112	
Min Pots -2.5K	Allen Bdly	WA2G056S202UA	Newark
or	Clarostat	392JA2.5K	Newark
Switches	C-H, JBT	MPC-123	Jameco, Newark
Ribbon cable plugs/headers	Ansley, Amphenol, others		Radio Shack has cable and 25 pin "D" type con- nectors. Digikey is good source. Also, Ansley, Amphenol ribbon cable distributors.
Pwr conn/fuse	Corcom	6J4	Newark, Pioneer
Power Supply	Power One	HTAA-16W-A	Arrow, others

Main PC board

Rt angl DB25	AMP	206584-1	
	Cinch	TB-25S-6	
50 pin con/fem	AMP		AMP, Harrisburg, PA
5v relays	Omron	G2V-2-US 5vdc	Digi-Key #Z300-ND

These relays are made by a number of other companies as well - PB, Midland, Aromat, ITT, Fujitsu and others. Should be able to cross the Omron number. As of this writing, Radio Shack stocks an equivalent type.

Integrated circuits:

Z80A	Be sure to get the "A" suffix - denotes 4mhz speed.
2716 EPROM	Since this is a special programmed part, it is available only direct.
P8255	Made by Intel, National, others. Available from Jameco, Digi-Key.
uPD7720	Special programmed part available only direct. Handle it carefully - it is very expensive!
AD574AJD	Analog Devices, Harris, Micro-Networks, Hybrid Systems
AD567JD	Analog Devices.
AD583	Analog Devices or Harris Al-2425-5. Analog Devices has a number of local sales offices -call them in Norwood, MA at (617) 329-4700. Harris is in Melbourne, FL. Hybrid Systems is in Billerica, MA.

DG201ACJ	Siliconix. Don't use other brands. From Siliconix distributors. For local distr. call Siliconix in Santa Clara, CA.
2912A	Use only Intel part for 027. Intel or Harris Ok for U25. From Intel or Harris distributor. For local Intel distr. call them in Santa Clara, CA.
MC1498AN ULN2003A 74LS93 74 LS 138 74LS14 LM1 3600 NE570 LF 347 NE5532 NE5738	These are all standard IC's available from many sources including Digi-key, Jameco and many others.
ICM7555	(low power cmos 555 - Radio Shack has 'em)
4N33	Opto-Isolator. Available everywhere. Standard part.
16mhz osc	Jameco # OSCI6.000

Pushbutton Control Unit

Pushbuttons	Microswitch	AML22CBH3BA-001	
Button tops	Microswitch	AML52C10K	
Center barriers	Microswitch	AML71SCB	
End barriers	Microswitch	AML71SEB	

Available by special order through any Microswitch distributor. The -001 suffix on the pushbutton part number denotes a special version that is quiet when pushed. This number will not be in distributor listings, but Microswitch will know what to do with the number. Microswitch is in Freeport, IL.



INTERNATIONAL SERIES
DC POWER SUPPLIES

DRAWING NO. 51281 REV. D

SPECIFICATIONS AND APPLICATION DATA

INTERNATIONAL SERIES DESCRIPTION

The INTERNATIONAL SERIES is a high reliability line of open-frame power supplies designed to operate from the wide range of AC power sources found worldwide.

This feature greatly simplifies your inventory and service considerations by allowing the use of one standard power supply regardless of destination.

Additionally, these models are designed to meet many domestic and European regulatory agency requirements.

If you plan to distribute your products worldwide, obtaining necessary agency approvals can be greatly simplified by specifying POWER-ONE, INC. INTERNATIONAL SERIES.

VOLTAGE/CURRENT RATING CHART

MODEL	+5V	+12V	+15V	+24V	+28V	-5V	-12V	-15V	CASE
SINGLE OUTPUT									
HB5-3/OVP-A	3.0								B
HB12-1.7-A		1.7							B
HB15-1.5-A			1.5						B
HB24-1.2-A				1.2					B
HB28-1-A					1.0				B
HC5-6/OVP-A	6.0								C
HC12-3.4-A		3.4							C
HC15-3-A			3.0						C
HC24-2.4-A				2.4					C
HC28-2-A					2.0				C
HNS-9/OVP-A	9.0								N
HN12-5.1-A		5.1							N
HN15-4.5-A			4.5						N
HN24-3.6-A				3.6					N
HN28-3-A					3.0				N
HD5-12/OVP-A	12.0								D
HD12-6.8-A		6.8							D
HD15-6-A			6.0						D
HD24-4.8-A				4.8					D
HD28-4-A					4.0				D
HE5-18/OVP-A	18.0								E
HE12-10.2-A		10.2							E
HE15-9-A			9.0						E
HE24-7.2-A				7.2					E
HE28-6-A					6.0				E
DUAL OUTPUTS									
HAA15-0.8-A			1.0 or 0.8			±0.4 or 1.0 or 0.8			AA
HBB15-1.5-A			1.7 or 1.5			±0.7 or 1.7 or 1.5			BB
HCC15-3-A			3.4 or 3.0			±0.4 or 1.0 or 1.0			CC
HDD15-5-A			5.0 or 5.0			±0.7 or 1.7 or 1.5			E
HAA512-A	2.0	*0.5	*0.5						AA
HBB512-A	3.0	*1.25	*1.25						BB
HCC512-A	6.0	*2.5	*2.5						CC
TRIPLE OUTPUTS									
HTAA-16W-A	2.0	*0.4	*0.4			±0.4 or ±0.4 or ±0.4			AA
HBAA-40W-A	3.0	1.0 or 0.8				±0.4 or 1.0 or 0.8			BAA
HCAA-60W-A	6.0	1.0 or 1.0				±0.4 or 1.0 or 1.0			D
HCB8-75W-A	6.0	1.7 or 1.5				±0.7 or 1.7 or 1.5			CBB
CP131-A	8.0	1.7 or 1.5				±0.7 or 1.7 or 1.5			131
HD88-105W-A	12.0	1.7 or 1.5				±0.7 or 1.7 or 1.5			DBB
DISC DRIVES									
MODEL	+5V	+12V	+24V	-5V	-12V	-15V			
CP379-A	6.0		3.5 (3.0 PK)	1.2	1.2				131
CP384-A	9.0		2.0 (8.0 PK)	1.2	1.2				131
CP323-A	2.0	4.0							N
CP205-A	1.0		1.5 (1.7 PK)	0.5					BAA
CP206-A	2.5		3.0 (3.4 PK)	0.5					CBB
CP162-A	3.0		5.0 (6.0 PK)	0.6					131

*9V to 15V Floating Output
+Requires jumper on PCB for -5V
Underlined output indicates no remote sense

FEATURES

- VDE transformer construction
- 100/120/220/230-240 VAC
- Remote sense - most outputs
- OVP on 5V outputs
- +.05% regulation
- Industry standard size
- Foldback/current limit
- Full-rated to 50°C
- I.C. burned-in to MIL-883 Lev B
- 2 hour burn-in period
- Two year warranty
- UL recognized
- CSA certified
- Chassis notched for AC input
- Input accepts .110 x .032 fast-on or solder connection

SPECIFICATIONS

- AC INPUT:** 100/120/220/230-240VAC** +10%, -13% 47-63 HZ
(Derate output current 10% for 50 Hz operation.)
See AC connection table under APPLICATION NOTES for jumper information. Fuse information is next to outline and mounting drawings.
- DC OUTPUT:** See Voltage/Current Rating Chart. Adjustment range +5% minimum.
- LINE REGULATION:** ±.05% for a 10% line change
- LOAD REGULATION:** ±.05% for a 50% load change
- OUTPUT RIPPLE:** 2V to 15V unit: 5.0mV PK-PK maximum
20V to 200V unit: .02% PK-PK maximum
- TRANSIENT RESPONSE:** 50µseconds for 50% load change
- SHORT CIRCUIT AND OVERLOAD PROTECTION:** Automatic current limit/foldback
- OVERVOLTAGE PROTECTION:** Built-in on all 5V outputs. Set at 6.2V ± .4V. Other models use optional overvoltage protection
- REMOTE SENSING:** Provided on most models, open sense lead protection built-in
- STABILITY:** ±0.3% for 24 hour period after 1 hour warm-up
- TEMPERATURE RATING:** 0°C to 50°C full-rated, derated linearly to 40% at 70°C
- TEMPERATURE COEFFICIENT:** ±.03%/°C maximum
- EFFICIENCY:** 5V unit: 45%; 12V and 15V units: 55% (typical)
20V and 24V units: 60%
- VIBRATION:** Per MIL-STD-810C, Method 514, Procedure X
- SHOCK:** Per MIL-STD-810C, Method 516, Procedure V
- **Tolerance for 230VAC operation is +15%, -10%

WARRANTY

POWER-ONE, INC. warrants each power supply of its manufacture that does not perform to published specifications, as a result of defective materials or workmanship, for a period of two (2) full years from the date of original delivery.

POWER-ONE, INC. assumes no liabilities for consequential damages of any kind through the use or misuse of its products by the purchaser or others. No other obligations or liabilities are expressed or implied.

PRODUCTS RETURNED FOR REPAIR

- Please follow this procedure when returning products for servicing:
- Contact Power-One's Customer Service Department for authorization to return products.
POWER-ONE, INC. PHONE: (805) 484-2906
740 Calle Plano (805) 987-3391
Camarillo, CA 93010 (800) 235-5943
USA TWX: 910-336-1297
 - A Returned Material Authorization (RMA) will be issued and must appear on all shipping documents and containers.
 - Products must be returned freight pre-paid.
- Products returned freight collect or without a RMA number will be rejected and returned freight collect.

REMOTE SENSE

Remote sense terminals may be used to compensate for output line losses and provide for a remote point of regulation. Figure 1 shows the proper termination for a power supply with remote sensing.

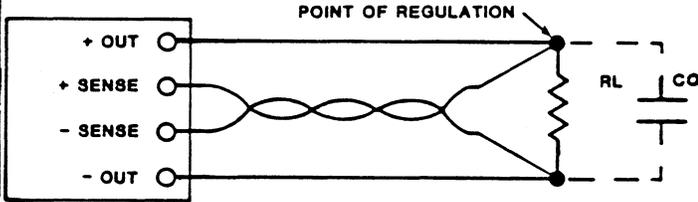


FIGURE 1

Load lines must be sized to prevent an excessive voltage drop from the output to the load. Since the point of regulation is at the load, the power supply must compensate for line losses. Excessive load line losses may affect current limiting, AC line dropout point and OVP margin (if applicable).

Leads should be sized to drop no more than 0.5V - the less the better. Use of a twisted pair or shielded pair for the sense lines is recommended for noise immunity. In problem applications, the use of a small AC decoupling capacitor (.1 to 10uFd) across the sense terminals is highly recommended. In some applications there may be a tendency for the power supply to oscillate due to the additional phase shift caused by the series resistance and inductance in the load leads. The addition of capacitor Co will reduce output impedance and provide stability. The recommended value of Co is 100uFd per ampere or 50uFd per foot and can be the sum of the distributed decoupling capacitors found in most systems.

All Power-One supplies have open sense lead protection to protect the load from an overvoltage condition if the sense leads are removed. There is no need to strap the sense terminals to the output terminals in the local sense mode.

OVERVOLTAGE PROTECTION (OVP)

An overvoltage protection circuit, commonly referred to as a crowbar, is used to prevent damage to voltage sensitive loads such as TTL logic. Trip point of the OVP is usually set at 115% - 135% of the output voltage. The OVP will short the output terminals upon sensing a fault condition. The primary fuse of the supply will blow if the supply is not foldback current limited. Nuisance tripping of the OVP is a common problem. Noise from input line spikes or load noise can cause an OVP to fire. The INTERNATIONAL SERIES has OVP noise filtering to prevent nuisance tripping and reduced transformer interwinding capacitance to minimize input line susceptibility.

COMMON-MODE LATCH UP

In certain instances dual power supplies can exhibit a problem known as common-mode latch up. This occurs when the positive supply comes up first and forces a reverse bias condition on the negative supply. The negative supply latches up in a current limit condition. Power-One has incorporated a unique anti-latch circuit into every dual power supply in the INTERNATIONAL SERIES which will minimize this problem.

EMI/RFI

These linear power supplies have inherently low conducted and radiated noise levels. For most system applications they will meet the requirements of FCC Docket 20780 for Class A equipment and VDE 0871 for Class A equipment without additional noise filtering. For special applications consult factory.

PRINTED CIRCUIT BOARDS

POWER-ONE, INC.'s printed circuit board fabrication facility is recognized under the component program by Underwriters Laboratories. All boards meet UL 94V-0 flammability specification.

SAFETY SPECIFICATIONS

The INTERNATIONAL SERIES power supplies were designed to meet or exceed requirements for the following specifications: IEC 380, IEC 435, VDE 0730 Part 2, VDE 0804, ECMA-57, CEE 10 Part 2P, UL 1012, CSA 22.2 No. 143, CSA 22.2 No. 154. Specifically, field terminal to terminal spacing is 5.25 mm with 9.0 mm creepage to other metal, leakage current is less than 5.0uA and dielectric withstanding voltages are 3750 VAC input to chassis, 3750 VAC input to output and 300 VDC output to chassis.

GROUNDING

Grounding considerations in designing a power distribution system are often overlooked but can have a significant impact on overall system performance. A single point system ground should be employed where possible to eliminate ground loops and improve regulation.

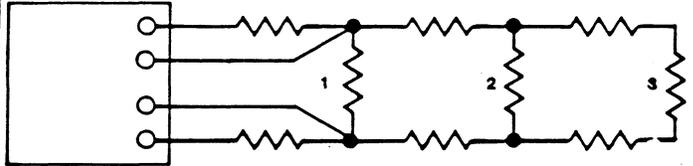


FIGURE 2

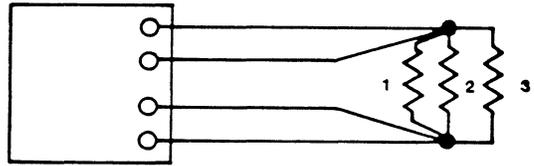


FIGURE 3

Figure 2 shows a simple but undesirable connection scheme. Regulation at loads 2 and 3 becomes progressively worse due to voltage drops in the finite wire resistance between loads. Figure 3 shows an improved connection system in which regulation is maintained at all three loads because wire losses are not cumulative.

AC INPUT CONSIDERATIONS

Almost all power supplies use a capacitive input filter that draws current only at the peaks of the AC input voltage. The peak to RMS ratio can be very high, typically 3 to 1. When a supply is turned on, the input capacitor has a very low impedance and draws an initially high surge current until it charges to its nominal voltage. The input surge current can be as high as 20 times the rated input current and lasts for several cycles of the AC input.

AC CONNECTION AND FUSING *

The five wire input to the INTERNATIONAL SERIES provides four voltage ranges: 100/120/220/230-240** +10%, -13%. See chassis AC connection table (Figure 4) for the jumpering requirements. For convenience the jumper sequence from the Hi-Vol series is retained. Extended low line tolerance provides additional drop out margin in areas where line voltages are marginal. Inputs must be fused.

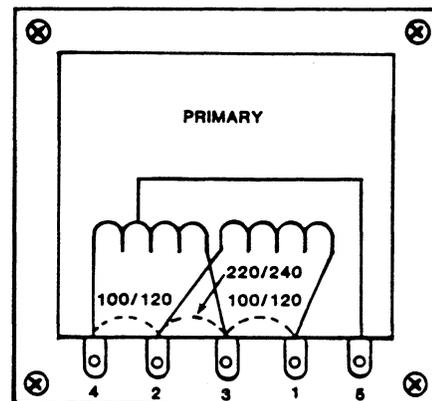


FIGURE 4

AC INPUT, 47-63 HZ				
FOR USE AT	100 VAC	120 VAC	220 VAC	230/240 VAC
JUMPER	1&3 2&4	1&3 2&4	2&3	2&3
APPLY AC	1&5	4&1	1&5	4&1

*NOTE: Use 700°C. iron for soldering input connections. Varnish acts as flux and is solder stripable.
**Tolerance for 230VAC operation is +15%, -10%.

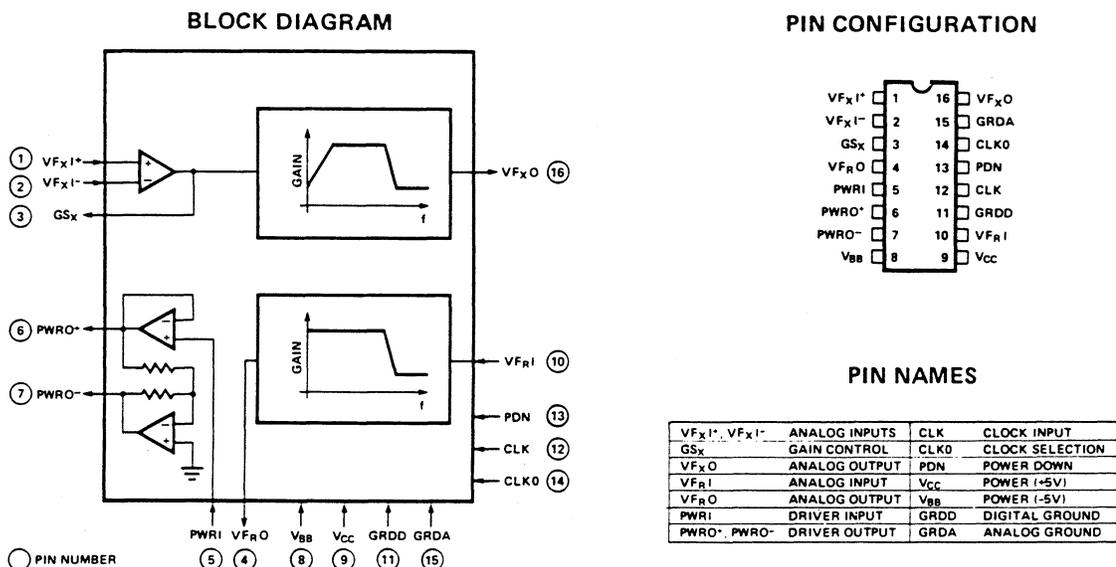
2912A PCM TRANSMIT/RECEIVE FILTER

- **Low Power Consumption:**
50mW Typical without Power Amplifiers
80mW Typical with Power Amplifiers
0.4mW Typical Standby
- **Low Idle Channel Noise:**
3 dBnc0 Typical, Receive
6 dBnc0 Typical, Transmit
- **Excellent Power Supply Rejection:**
45dB Typical on V_{BB} @ 1kHz
40dB Typical on V_{BB} @ 20kHz
- **High Pass Filter Rejects Low Frequency Noise:**
23dB @ 60Hz
28dB @ 50Hz
58dB @ 16Hz, Typical
- **Adjustable Gain in Both Directions**
- **Fully Compatible with the Industry Standard Intel 2912**
- **D3/D4 and CCITT G712 Compatible**
- **Common Mode Op Amp Input Rejection 75dB Typical**
- **Direct Interface to the Intel 2910A/2911A PCM Coders Including Stand-By, Power Down Mode**
- **Direct Interface with Transformer or Electronics Hybrids**
- **Fabricated with Reliable N-Channel MOS Process**

The Intel 2912A 2nd generation PCM line filter is a fully integrated monolithic device containing the two filters of a PCM line or trunk termination. It has improved key parameters of power consumption, idle channel noise, and power supply rejection. A single part exceeds both AT&T D3/D4 and CCITT transmission specs, exceeds digital Class 5 central office switching system stringent specifications, and is fully compatible with the 2912. The primary application for the 2912A is in telephone systems for transmission, switching, or remote concentration.

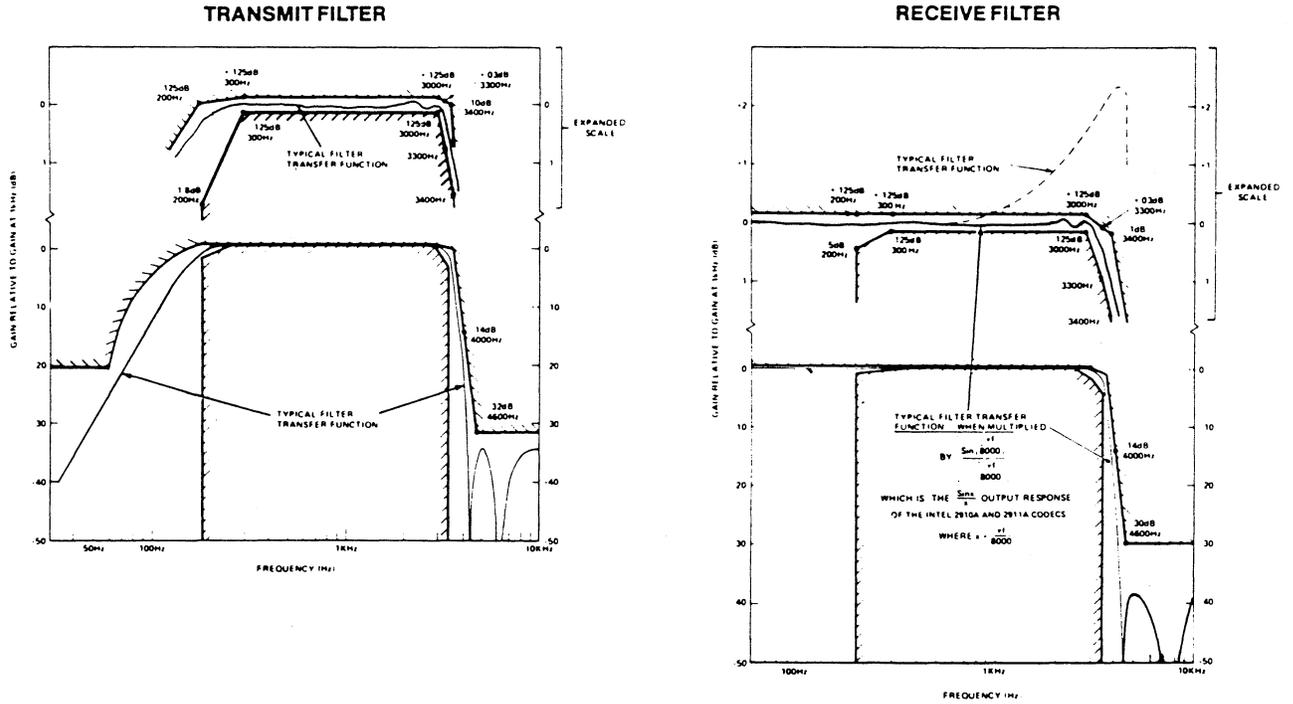
An advanced version of the switched capacitor technique used for the 2912 is used to implement the transmit and receive passband filter sections of the 2912A. The device is fabricated using Intel's reliable two layer polysilicon gate NMOS technology. (See Intel Reliability Report RR-24 on the 2910A, 2911A, and 2912.) The combination of advances in the switched capacitor techniques first used on the 2912 and the NMOS technology results in a monolithic 2912A filter which is packaged in a standard 16 pin DIP.

For additional specs, operating characteristics, and applications information please see the 2912 data sheet.



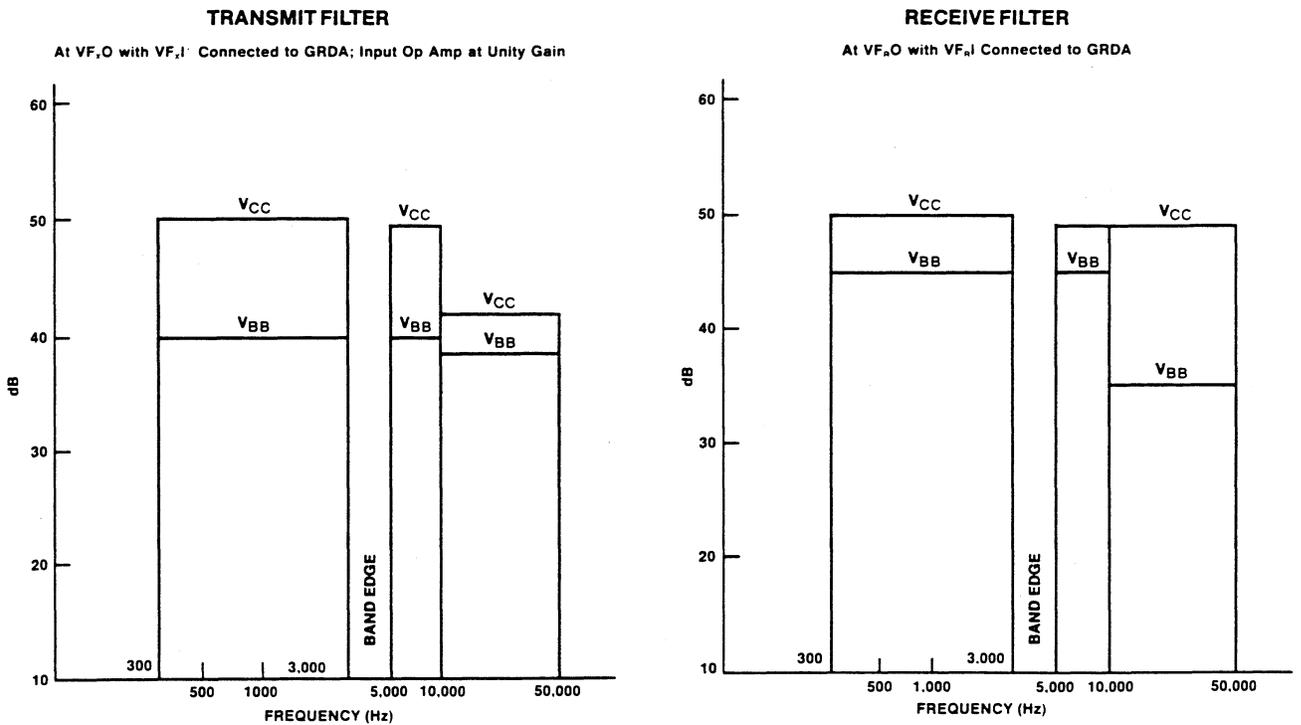
AT&T is a registered trademark of American Telephone and Telegraph Corporation.

TRANSFER CHARACTERISTICS



NOTES
 1 TYPICAL TRANSFER FUNCTION OF THE RECEIVE FILTER AS A SEPARATE COMPONENT
 2 TYPICAL TRANSFER FUNCTION OF THE RECEIVE FILTER DRIVEN BY THE SAMPLE AND HOLD OUTPUT OF THE INTEL 2910A AND 2911A CODECS THE COMBINED FILTER/CODEC RESPONSE MEETS THE STATED SPECIFICATIONS

POWER SUPPLY REJECTION
 AVERAGE VALUES OVER 3 RANGES



ABSOLUTE MAXIMUM RATINGS*

Temperature Under Bias -10°C to +80°C
 Storage Temperature -65°C to +150°C
 Supply Voltage with Respect
 to V_{BB} -0.3V to +14.0V
 All Input and Output Voltages
 with Respect to V_{BB} -0.3V to +14.0V
 All Output Currents ±50mA
 Power Dissipation 1 Watt

***COMMENT:**

Stresses above those listed under "Absolute Maximum Rating" may cause permanent damage to the device. This is a stress rating only and functional operation of the device at these or at any other condition above those indicated in the operational sections of this specification is not implied. Exposure to absolute maximum rating conditions for extended periods may affect device reliability.

D.C. AND OPERATING CHARACTERISTICS

T_A = 25°C, V_{CC} = +5V, V_{BB} = -5V, GRDA = 0V, GRDD = 0V

POWER DISSIPATION

Symbol	Parameter	Limits			Unit	Test Conditions
		Min.	Typ.	Max.		
I _{CC0}	V _{CC} Standby Current		40		μA	PDN = V _{IH MIN}
I _{BB0}	V _{BB} Standby Current		40		μA	PDN = V _{IH MIN}
I _{CC1}	V _{CC} Operating Current, Power Amplifiers Inactive		5		mA	PWRI = V _{BB}
I _{BB1}	V _{BB} Operating Current, Power Amplifiers Inactive		5		mA	PWRI = V _{BB}
I _{CC2}	V _{CC} Operating Current		8		mA	
I _{BB2}	V _{BB} Operating Current		8		mA	

ANALOG INTERFACE, TRANSMIT FILTER INPUT STAGE

CMRR	Common Mode Rejection, V _{F_XI} ⁺ , V _{F_XI} ⁻		75		dB	-2.2V < V _{IN} < 2.2V, 0dBmO
A _{VOL}	DC Open Loop Voltage Gain, G _{S_X}		6000			

A.C. CHARACTERISTICS

T_A = 25°C, V_{CC} = +5V, V_{BB} = -5V, GRDA = 0V, GRDD = 0V

TRANSMIT FILTER CHARACTERISTICS

Symbol	Parameter	Min.	Typ.	Max.	Units	Test Conditions
N _{CX1}	Total C Message Noise at Output, V _{F_XO}		6		dBrnc0 [1]	Gain Setting Op Amp at Unity Gain
N _{CX2}	Total C Message Noise at Output, V _{F_XO}		9		dBrnc0 [1]	Gain Setting Op Amp at 20dB Gain

RECEIVE FILTER CHARACTERISTICS

N _{CR}	Total C Message Noise at Output, V _{F_RO}		3		dBrnc0 [1]	V _{F_RO} Output or PWRO ⁺ and PWRO ⁻ Connected with Unity Gain
-----------------	--	--	---	--	------------	---

NOTES:

1. A noise measurement of 15dBrnc into a 600Ω load at the 2912A device is equivalent to 9dBrnc0.

DESCRIPTION

The NE570/571 is a versatile low cost dual gain control circuit in which either channel may be used as a dynamic range compressor or expander. Each channel has a full wave rectifier to detect the average value of the signal; a linearized, temperature compensated variable gain cell; and an operational amplifier.

The NE570/571 is well suited for use in telephone subscriber and trunk carrier systems, communications systems and hi-fi audio systems.

FEATURES

- Complete compressor and expander in 1 IC
- Temperature compensated
- Greater than 110dB dynamic range
- Operates down to 6Vdc
- System levels adjustable with external components
- Distortion may be trimmed out

CIRCUIT DESCRIPTION

The NE570/571 compandor building blocks, as shown in the block diagram, are a full wave rectifier, a variable gain cell, an operational amplifier and a bias system. The arrangement of these blocks in the IC result in a circuit which can perform well with few external components, yet can be adapted to many diverse applications.

The full wave rectifier rectifies the input current which flows from the rectifier input, to an internal summing node which is biased at V_{REF} . The rectified current is averaged on an external filter capacitor tied to the C_{RECT} terminal, and the average value of the input current controls the gain of the variable gain cell. The gain will thus be proportional to the average value of the input signal for capacitively coupled voltage inputs as shown in the following equation. Note that for capacitively coupled inputs there is no offset voltage capable of producing a gain error. The only error will come from the bias current of the rectifier (supplied internally) which is less than $.1\mu A$.

$$G \propto \frac{|V_{IN} - V_{REF}|_{ave}}{R_1}$$

or

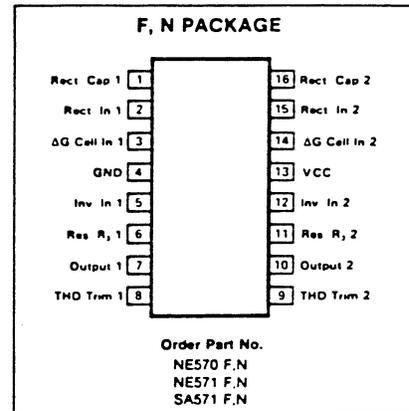
$$G \propto \frac{|V_{IN}|_{ave}}{R_1}$$

The speed with which gain changes to follow changes in input signal levels is determined by the rectifier filter capacitor. A small capacitor will yield rapid response but will not fully filter low frequency signals. Any ripple on the gain control signal will modulate the signal passing through the variable gain cell. In an expander or com-

APPLICATIONS

- Telephone trunk compandor—570
- Telephone subscriber compandor—571
- High level limiter
- Low level expander—noise gate
- Dynamic noise reduction systems
- Voltage controlled amplifier
- Dynamic filters

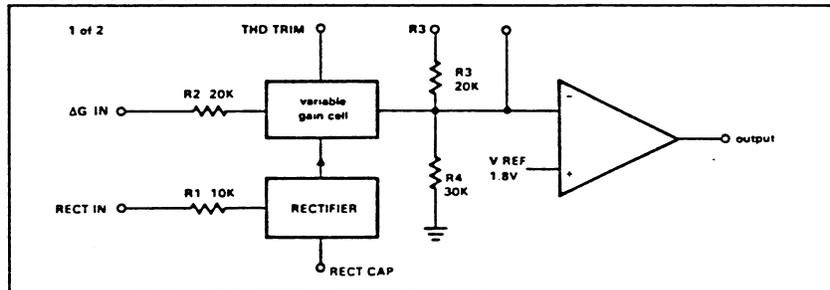
PIN CONFIGURATION



ABSOLUTE MAXIMUM RATINGS

PARAMETER	RATING	UNIT
Positive supply	24	Vdc
570	18	
571		
T_A Operating temperature range	0 to 70	$^{\circ}C$
NE	-40 to +85	$^{\circ}C$
SA		
P_D Power dissipation	400	mW

BLOCK DIAGRAM



pressor application, this would lead to third harmonic distortion, so there is a tradeoff to be made between fast attack and decay times, and distortion. For step changes in amplitude, the change in gain with time is shown by this equation.

$$G(t) = (G_{initial} - G_{final}) e^{-t/\tau} + G_{final}; \tau = 10K \times C_{RECT}$$

The variable gain cell is a current in, current out device with the ratio I_{OUT}/I_{IN} controlled by the rectifier. I_{IN} is the current which flows from the ΔG input to an internal summing node biased at V_{REF} . The following equation applies for capacitively coupled inputs. The output current, I_{OUT} , is fed to the summing node of the op amp.

$$I_{IN} = \frac{V_{IN} - V_{REF}}{R_2} = \frac{V_{IN}}{R_2}$$

A compensation scheme built into the ΔG cell compensates for temperature, and cancels out odd harmonic distortion. The only distortion which remains is even harmonics, and they exist only because of internal offset voltages. The THD trim terminal provides a means for nulling the internal offsets for low distortion operation.

The operational amplifier (which is internally compensated) has the non-inverting input tied to V_{REF} , and the inverting input connected to the ΔG cell output as well as brought out externally. A resistor, R_3 , is brought out from the summing node and allows compressor or expander gain to be determined only by internal components.

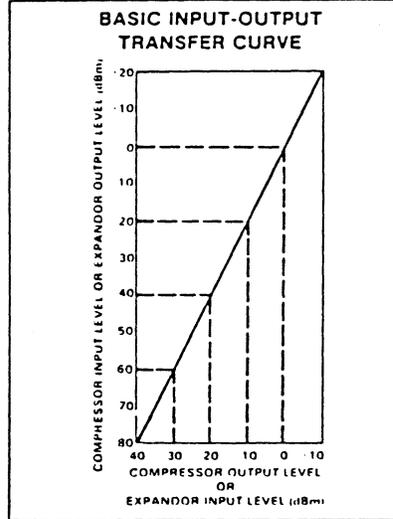


The output stage is capable of $\pm 20\text{mA}$ output current. This allows a -13dBm (3.5V rms) output into a 300Ω load which, with a series resistor and proper transformer, can result in $+13\text{dBm}$ with a 600Ω output impedance.

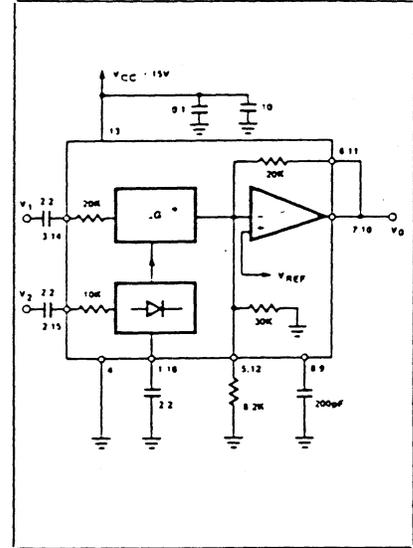
A band gap reference provides the reference voltage for all summing nodes, a regulated supply voltage for the rectifier and ΔG cell, and a bias current for the ΔG cell. The low tempo of this type of reference provides very stable biasing over a wide temperature range.

The typical performance characteristics illustration shows the basic input-output transfer curve for basic compressor or expander circuits.

TYPICAL PERFORMANCE CHARACTERISTICS



TYPICAL TEST CIRCUIT



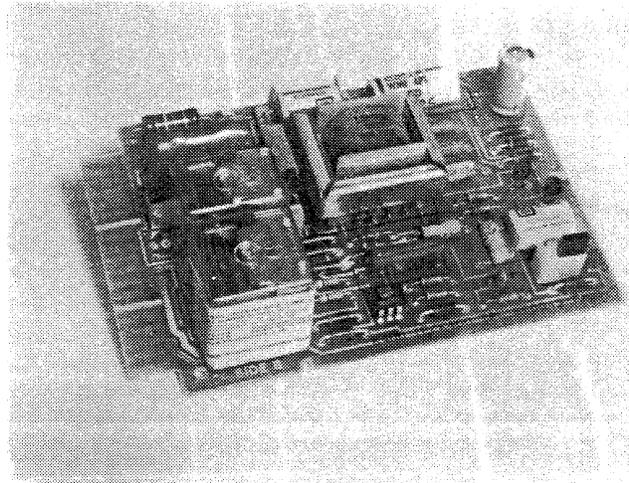
DC ELECTRICAL CHARACTERISTICS T_A = 25°C, V_{CC} = 15V¹

PARAMETER	TEST CONDITIONS	NE570			NE/SA571 ⁶			UNIT
		Min	Typ	Max	Min	Typ	Max	
V _{CC} Supply voltage	No signal	6	3.2	24	6	3.2	18	V
I _{CC} Supply current		± 20		4.0			4.8	mA
Output current capability		± 20						mA
Output slew rate	Untrimmed		± 5			± 5		V/us
Gain cell distortion ²		Trimmed		.3	1.0		.1	2.0
Resistor tolerance	Untrimmed		± 5	± 15				%
Internal reference voltage		Trimmed	1.7	1.8	1.9	1.65	1.8	1.95
Output dc shift ³	No signal, 20Hz-20kHz		± 20	± 50		± 30	± 100	mV
Expander output noise				20				μV
Unity gain level	-40°C < T < 70°C	-1	0	+1	-1.5	0	+1.5	dB
Gain change ^{2,4}		0°C < T < 70°C		± 1	± 2		± 1	± 4
Reference drift ⁴	-40°C < T < 70°C		+2, -25	10, -40		+2, -25	+20, -50	mV
	0°C < T < 70°C		± 5	± 10		± 5	± 20	
Resistor drift ⁴	-40°C < T < 70°C		+8, -0					%
	0°C < T < 70°C		+1, -0					
Tracking error ⁵ , input V ₁ = 0dBm	Rectifier input, V ₂ =							dB
	-6dBm		± 2			± 2		
	-10dBm		-2	-2, -4		+2	-2, +5	
	-20dBm		+2	-3, -6		+2	-4, +7	
	-30dBm		+2	-5, -1		+2	-1, +1.5	
	-40dBm		+2, -4			+2, -4		

NOTES

- 1 Except where indicated, the 571 specifications are identical to the 570
- 2 Measured at 0dBm, 1kHz
- 3 Expander ac input change from no signal to 0dBm
- 4 Relative to value at T_A = 25°C
- 5 Relative to 0dBm
- 6 Electrical characteristics for the SA571 only are specified over -40 to +85°C temperature range

K400TPL INTERFACE LINE CARD (B60049-1000)



1.00 GENERAL

1.01 The K400TPL INTERFACE LINE CARD is a Key Telephone Unit (KTU) plug-in CO/PBX line circuit with interface circuitry. The K400TPL permits direct connection to be made to a telephone company (telco) line facility. The interface circuitry maintains longitudinal balance on a telephone line while providing direct current (DC) isolation. Each card has a light emitting diode (LED) as a card status indicator to facilitate servicing the Key Telephone System (KTS).

1.02 The K400TPL is equipped with three jumper block/plug arrangements to provide the following options:

- a. short time out (5-10 seconds),
- b. interrupted ringing,
- c. auxiliary common audible signal, and
- d. music-on-hold.

NOTE: MUSIC-ON-HOLD REQUIRES ADDITIONAL EQUIPMENT.

1.03 The short time out option reduces the time out of a card from 25-35 seconds to 5-10 seconds when a calling party abandons a call prior to it being answered.

1.04 The K400TPL *does not* contain any power failure transfer circuitry. However, the Power Failure Transfer Card (B60052-0000) can be used in conjunction with a K400TPL Line Card to transfer a CO line to an emergency line, or an emergency telephone, in the event of local power failure. The Power Failure Transfer Card is not automatically shipped with the Interface Line Card; it must be ordered separately.

1.05 The K400TPL is registered with the FCC under the following number:

AS293P-67342-VP-E
Ringer Equivalence 0.4B

The VP Classification gives the K400TPL Interface Line Card fully protected status. Any key service unit requiring 400-type line cards can be used behind the K400TPL Line Card. Therefore, the K400TPL Interface Line Card *replaces* the K400TP and K400TP/TPL line cards because of its *less restrictive* classification.

2.00 LIMITATIONS

2.01 The K400TPL is interchangeable with the K400E line card except for the following features:

- a. The K400TPL line card *does not* provide a steady illuminated lamp on *hold* (X Option).
- b. The card *remains on hold* a held party abandons the call.
- c. Central Office (CO) ringing *cannot* be bridged across the station TIP/RING leads.
- d. If the A-lead is not grounded prior to bridging the station TIP and RING leads during troubleshooting, the Interface Line Card *will* go on continuous hold.

3.00 SPECIFICATIONS

3.01 The TIP/RING bridged impedance allowance is approximately ten times the minimum allowance.

3.02 The response time to a CO ringing signal is 100 to 200 milliseconds to avoid false ringing.

3.03 The normal time out of the card, when a calling party abandons a call before it is answered, is 25-35 seconds, or 5-10 seconds with the short time out option.

3.04 The K400TPL Line Card is environmentally tested to ensure reliability. Each unit shall be able to withstand ten repeated 24-hour tests consisting of two high cycles of 4.5 hours @ 55°C and 95% relative humidity without condensation, and one low cycle for 4.0 hours @ -10°C. Each cycle must return to a reference temperature of +18°C.

4.00 ORDERING

4.01 The card is ordered as follows:

K400TPL INTERFACE LINE CARD
Part Number B60049-1000

4.02 The power failure transfer unit is ordered as follows:

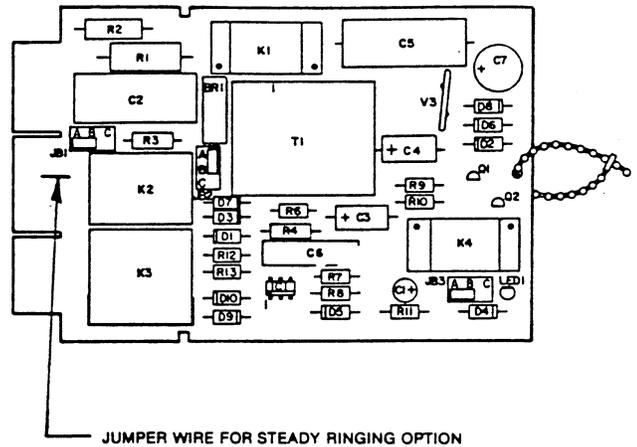
POWER FAILURE TRANSFER CARD
Part Number B60052-0000

JUMPER BLOCK ARRANGEMENT

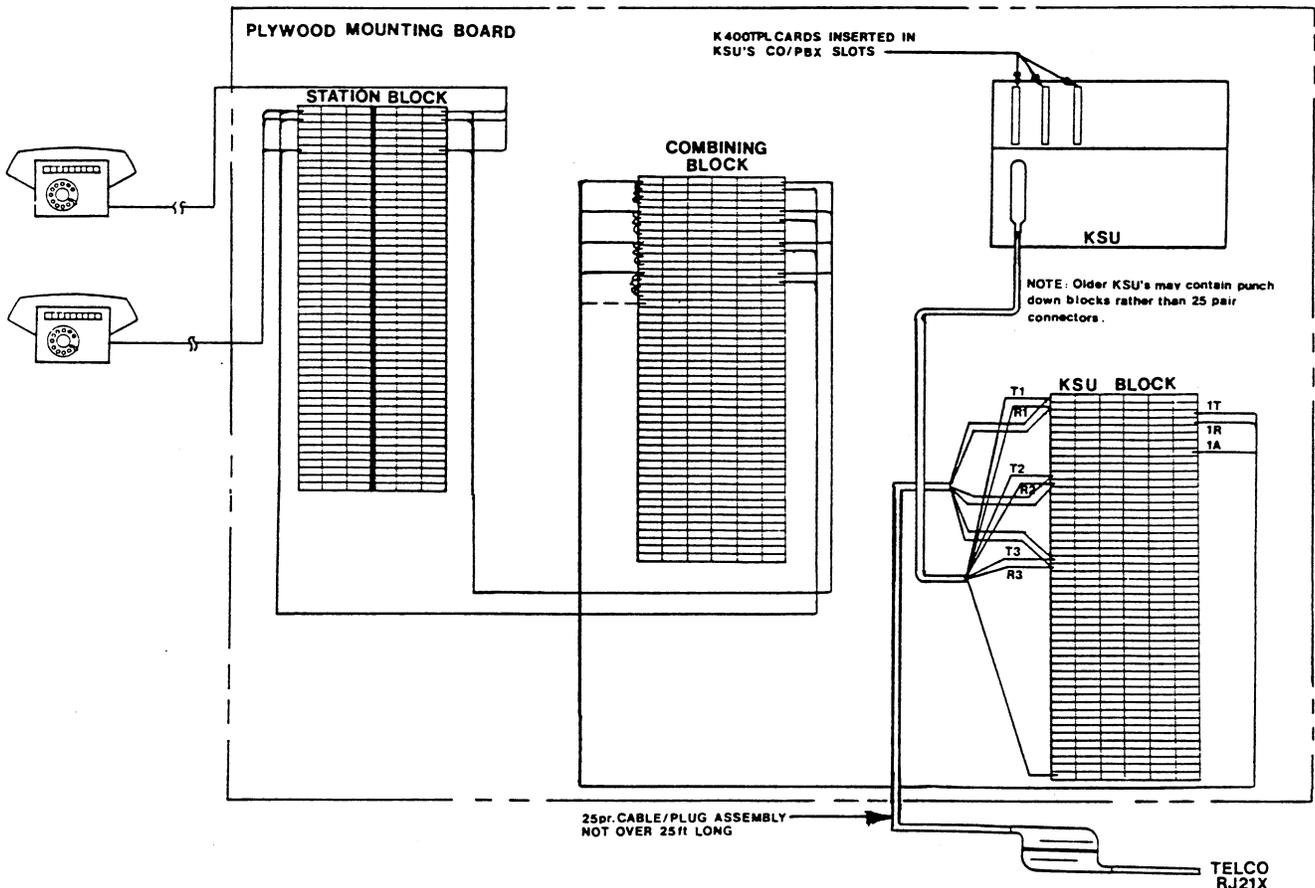
OPTION	JB1	JB2	JB3	COMMENTS
Short Time Out (Z)			AB	Factory Standard
Long Time Out			BC	
Standard "HOLD" (No Music)		AB		Factory Standard
Music-On-Hold (M)		BC		SEE NOTE 2
Auxillary Common Audible Signal Control (V)	BC			
Interrupted Ringing (W)	AB			Factory Standard
Steady Ringing (T)	Remove Plug			SEE NOTE 1

NOTES:

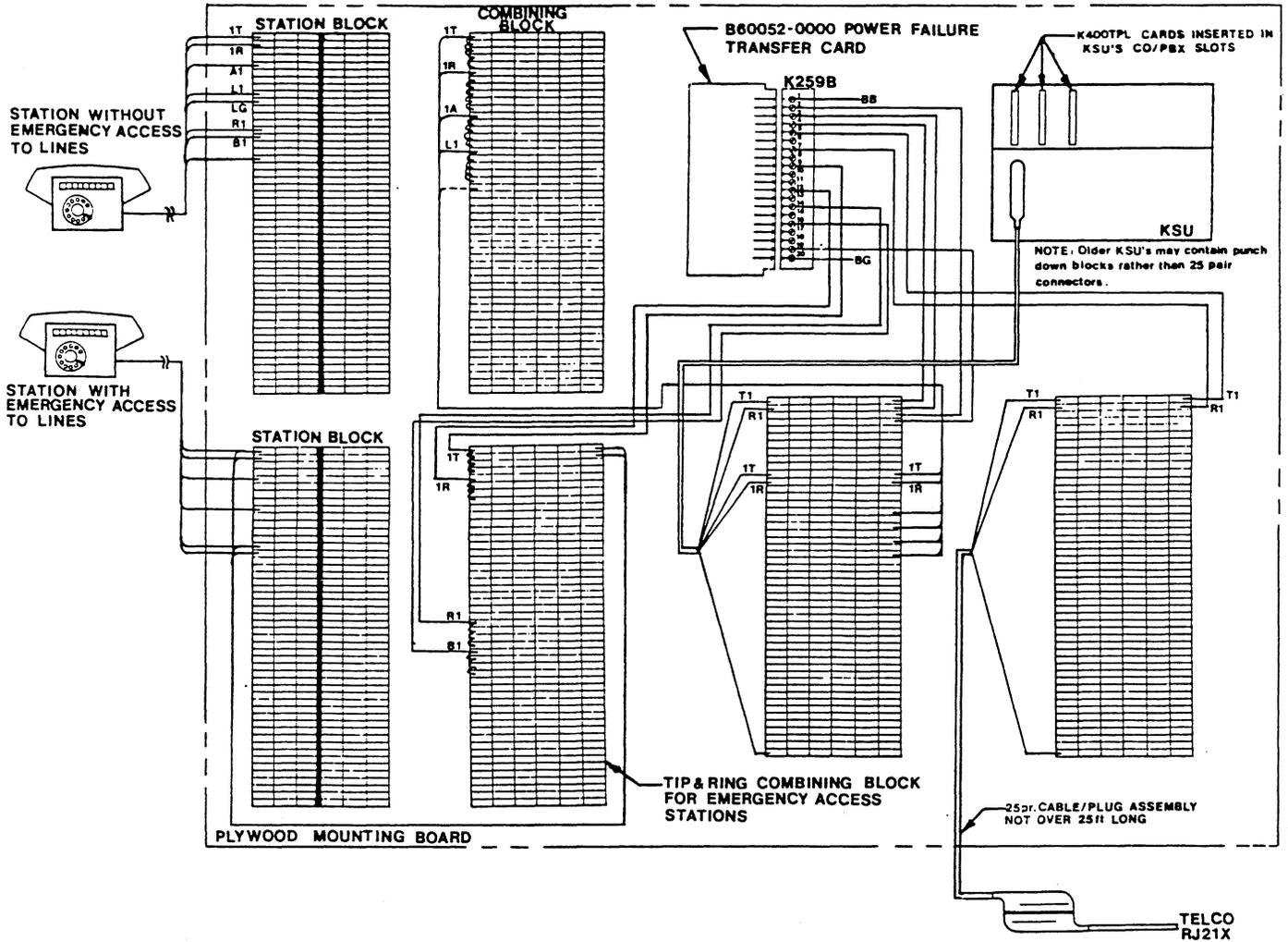
1. For Steady Ringing Option, Solder a Jumper Wire on the PC Board Assembly as Indicated.
2. For MUSIC-ON-HOLD option, wire a K403A output between Tab 12 (CT) and Tab 18 (MOH).



TYPICAL INSTALLATION
K400TPL Interface Line Card



TYPICAL INSTALLATION
K400TPL & Power Failure Transfer



TYPICAL ITT KSU BLOCK ARRANGEMENTS

LINE	FUNCTION	COLOR CODE	TERMINAL OR PIN NUMBER			
			K501, K501A	K512, K512A	K36A	K76A
1	T1	WHT-BLU	Block C, Pos. 25	26	26	PLUG P1-26
	R1	BLU-WHT	26	1	1	PLUG P1-1
	1T	WHT-ORN	Block A, Pos. 1	27	27	PLUG P1-27
	1R	ORN-WHT	2	2	2	PLUG P1-2
	1A	WHT-GRN	3	28	28	PLUG P1-28
	A1	GRN-WHT	4	3	3	PLUG P1-25
	LG	WHT-BRN	5	29	29	PLUG P2-25
	1L	BRN-WHT	6	4	4	PLUG P1-3
	B	WHT-SLT	Block B, Pos. 19	30	30	PLUG P4-47
	R	SLT-WHT	20	5	5	PLUG P1-29
2	T2	RED-BLU	Block C, Pos. 27	31	31	PLUG P1-4
	R2	BLU-RED	28	6	6	PLUG P1-30
	2T	RED-ORN	Block A, Pos. 7	32	32	PLUG P1-5
	2R	ORN-RED	8	7	7	PLUG P1-31
	2A	RED-GRN	9	33	33	PLUG P1-6
	A1	GRN-RED	10	8	8	PLUG P1-25
	LG	RED-BRN	11	34	34	PLUG P3-50
	2L	BRN-RED	12	9	9	PLUG P1-32
	B	RED-SLT	Block B, Pos. 21	35	35	PLUG P4-47
	R	SLT-RED	22	10	10	PLUG P1-7
3	T3	BLK-BLU	Block C, Pos. 29	36	36	PLUG P1-33
	R3	BLU-BLK	30	11	11	PLUG P1-8
	3T	BLK-ORN	Block A, Pos. 13	37	37	PLUG P1-34
	3R	ORN-BLK	14	12	12	PLUG P1-9
	3A	BLK-GRN	15	38	38	PLUG P1-35
	A1	GRN-BLK	16	13	13	PLUG P1-25
	LG	BLK-BRN	17	39	39	PLUG P3-25
	3L	BRN-BLK	18	14	14	PLUG P1-10
	B	BLK-SLT	Block B, Pos. 23	40	40	PLUG P4-47
	R	SLT-BLK	24	15	15	PLUG P1-36
4	T4	YEL-BLU	Block C, Pos. 31	41	41	PLUG P1-11
	R4	BLU-YEL	32	16	16	PLUG P1-37
	4T	YEL-ORN	Block A, Pos. 19	42	42	PLUG P1-12
	4R	ORN-YEL	20	17	17	PLUG P1-38
	4A	YEL-GRN	21	43	43	PLUG P1-13
	A1	GRN-YEL	22	18	18	PLUG P1-25
	LG	YEL-BRN	23	44	44	PLUG P4-27
	4L	BRN-YEL	24	19	19	PLUG P1-39
	B	YEL-SLT	Block B, Pos. 25	45	45	PLUG P4-47
	R	SLT-YEL	26	20	20	PLUG P1-14
5	T5	VIO-BLU	Block C, Pos. 33	46	46	PLUG P1-40
	R5	BLU-VIO	34	21	21	PLUG P1-15
	5T	VIO-ORN	Block A, Pos. 25	47	47	PLUG P1-41
	5R	ORN-VIO	26	22	22	PLUG P1-16
	5A	VIO-GRN	27	48	48	PLUG P1-42
	A1	GRN-VIO	28	23	23	PLUG P1-25
	LG	VIO-BRN	29	49	49	PLUG P4-28
	5L	BRN-VIO	30	24	24	PLUG P1-17
	B	VIO-SLT	Block B, Pos. 27	50	50	PLUG P4-47
	R	SLT-VIO	28	25	25	PLUG P1-43

6.00 CIRCUIT DESCRIPTION

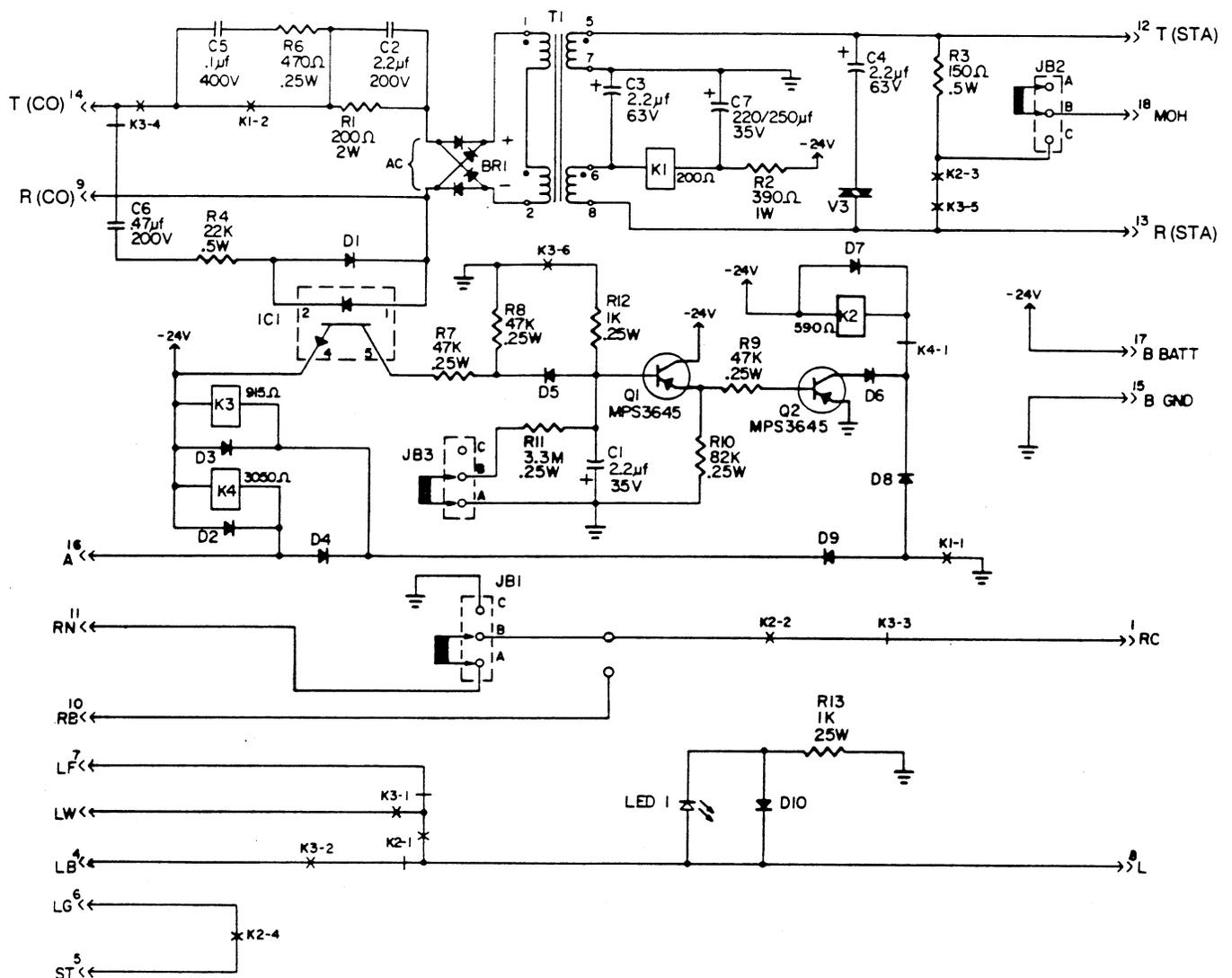
6.01 AC ringing voltage from the Central Office (CO) is applied to the CO TIP/RING (Tabs 14 and 9). Current flows via contact K3-4, C6, R4, and the LED portion of photo coupler IC-1 during negative cycles of the applied AC ringing voltage. During the positive cycles of ring voltage, current flows via contact K3-4, C6, R4, and D1, protecting the LED against reverse voltages. During the half cycles of ringing when current flows through the LED portion of IC-1, the phototransistor section is activated and applies -24V DC to RING/HOLD circuit. The RING/HOLD circuit activates Relay K2. Contact K2-2 then connects the ringing voltages or signals to the RINGER LEAD (Tab 1). Voltage from GND (Tab 11) may be selected by jumper block/plug JB1. If steady ringing is required, then solder a jumper wire as shown in the diagram to route the ring battery at Tab 10 to K2-2. Contact K2-1 connects the LAMP FLASH VOLTAGE (Tab 7) to the LAMP LEAD (Tab 8) via inactive contact K3-1. Contact K2-4 connects LG (Tab 6) to the

interrupter motor start [ST] (Tab 5), thus starting the key system interrupter.

6.02 The called party answers the telephone by taking the handset off-hook, either before or after depressing the line button associated with the incoming call. With the telephone off-hook and the button depressed, a sequence of two events occurs.

6.03 First, GND is connected to the A-lead (Tab 16) via the A-lead contact in the line button of the telephone. The presence of GROUND (Tab 16) activates Relay K3. The RING/ HOLD circuit now deactivates K2. Ringing signals, or voltages, are removed from Tab 1 via the contacts K2-2 and K3-3. LAMP BATTERY (Tab 4) is connected to the LAMP LEAD (Tab 8); and LAMP FLASH (Tab 7) is removed via contacts K3-2 and K2-1. Contact K3-4 removes the CO TIP/RING connection (Tabs 14 and 9) from the ring detector circuit and connects them to de-energized contact K1-2. Contact K2-4 opens, thus stopping the interrupter motor.

SCHEMATIC DIAGRAM



6.04 The second event is the connection of station TIP and RING (Tabs 12 and 13) to the subset. Current flows from GROUND through windings 5-7 of T1, the off-hook subset, windings 6-8 of T1, the coil of relay K1, and R2 to -24V DC. K1 now activates. Contact K1-2 completes the CO TIP/RING path to the isolation transformer T1 circuit. CO current now flows from CO TIP (Tab 14), through contact K3-4, contact K1-2, R1, and transformer windings 1-2 via bridge rectifier BR1 to CO RING (Tab 9). This DC current flow trips CO ringing. C2 provides an AC bypass around R1. Voice signals are coupled via T1 between the CO line and the called party.

6.05 When the subset hold button is depressed, the A-lead contact (wired to Tab 16) is broken before the TIP and RING contacts. When the GROUND is removed from Tab 16, K2 operates. R3 is connected across T(STA) and R(STA) via contacts K3-5 and K2-3 to provide a current path to keep relay K1 operated when the subset TIP and RING connection is broken. K1 maintains the CO TIP/RING connection via contact K1-2. Contact K2-1 removes the LAMP LEAD (Tab 8) from the LAMP BATTERY INPUT (Tab 4), and substitutes the LAMP WINK SIGNAL (Tab 2) via energized contact K3-1.

6.06 When the subset line button is depressed to go off hold, relay K2 is deactivated. Contact K2-1 removes LAMP WINK (Tab 2) and re-establishes LAMP BATTERY (Tab 4) to the LAMP LEAD (Tab 8). The K2-3 contact removes resistor R3 from the station TIP/RING circuit. Since the TIP/RING line button contacts have closed, the current path that keeps relay K1 operating remains unbroken.

6.07 When a line button is depressed to initiate a call, the A-lead is grounded operating relay K3 via Tab 16. Contact K3-2 connects

LAMP BATTERY (Tab 4) to the LAMP LEAD (Tab 8) for a steady visual signal on all associated subsets. Contact K3-4 provides one of the connections necessary to draw central office current via CO TIP/RING (Tabs 14 and 9). When the subset line button connects station TIP and RING (Tabs 12 and 13) to the subset, relay K1 operates via R2 and the windings of T1. Contact K1-2 completes the CO DC circuit through T1 winding 1-2 and R1 via the CO TIP/RING (Tabs 14 and 9). Dial tone is now sent from the CO.

6.08 During dialing, the subset TIP/RING circuit is pulsed off/on by the action of dial contacts. Relay K1 follows this off/on pulsing and repeats this action to the CO via contact K1-2. R6 and C5 suppress noise and arcing during the pulsing of contact K1-2.

NOTE: IF THE SUBSET IS EQUIPPED WITH A TONE DIAL PAD, THE GENERATED TONES ARE COUPLED VIA T1 TO THE CO DECODING CIRCUITS.

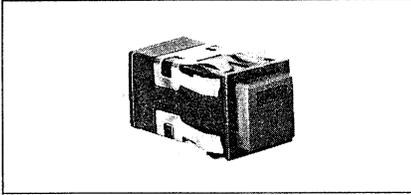
6.09 MUSIC-ON-HOLD (MOH) is provided by connecting a music source between Tabs 12 and 16. The music signal can be connected to R3 via the jumper block/plug JB2. When the circuit is placed on hold, contacts K2-3 and K3-5 connect the music signal to T1 which couples the signal to the telephone line.

6.10 An LED is provided to give visual status indication to service/repair personnel. D10 protects the LED during negative half cycles of the incoming 10V AC lamp signals, and R13 limits the LED current.

6.11 D3 and D7 suppress inductive voltages when relays K3 and K2 are de-energized.

6.12 Bridge rectifier BR1 polarizes CO line current.

LED DISPLAY



LEDs are not replaceable.

FEATURES

- Identical to AML21 switches, except furnished with high efficiency LED display.
- Rectangular LED's are flush with button surface, providing wide angle indication.
- Buttons with LED "window" ordered separately.
- Optional diode protection for LED's.
- 5 thru 24 VDC LED devices have an internal resistor to maintain current at nominal 20 mA.
- UL recognized, CSA certified.

Electrical Data	Page 19
Buttons	Page 43, 44
Lamps	Page 59
Accessories	Page 57, 58
Mounting Dimensions	Page 60, 63



AML22 ORDER GUIDE

AML22 C	B	B	2	AA									
Housing Type	Bezel Color	LED Color/ Voltage	Terminal Type	Circuitry Codes (Each pole has double-throw)									
Standard Bezel: AML22C Square 1 LED AML22H Square 1 High-Profile LED (For use with AML52-A buttons) Full Guard Bezel: AML22J Square 1 LED AML22K Square 1 High-Profile LED (For use with AML52-A buttons)	B Black	Red B V* C 5V D 10V E 15V F 24V Yellow H V* J 5V K 10V L 15V M 24V Green R V* S 5V T 10V W 15V X 24V	2 .110 × .020 (Solder or Quick-Connect) 3 .025 × .025 (Printed Ckt., or Push-On) 8 .110 × .020 With Diode Protection for LED 9 .025 × .025 With Diode Protection for LED	<table border="1"> <tr> <td rowspan="2">Silver Contacts</td> <td>Mom. Action AA 1-Pole AC 2-Pole CC 4-Pole</td> </tr> <tr> <td>Alt. Action AB 1-Pole AD 2-Pole CD 4-Pole</td> </tr> <tr> <td rowspan="2">Gold Contacts</td> <td>Mom. Action BA 1-Pole BC 2-Pole DC 4-Pole</td> </tr> <tr> <td>Alt. Action BB 1-Pole BD 2-Pole DD 4-Pole</td> </tr> <tr> <td rowspan="2">Gold-Plated Silver Contacts</td> <td>Mom. Action EA 1-Pole EC 2-Pole</td> </tr> <tr> <td>Alt. Action EB 1-Pole ED 2-Pole</td> </tr> </table>	Silver Contacts	Mom. Action AA 1-Pole AC 2-Pole CC 4-Pole	Alt. Action AB 1-Pole AD 2-Pole CD 4-Pole	Gold Contacts	Mom. Action BA 1-Pole BC 2-Pole DC 4-Pole	Alt. Action BB 1-Pole BD 2-Pole DD 4-Pole	Gold-Plated Silver Contacts	Mom. Action EA 1-Pole EC 2-Pole	Alt. Action EB 1-Pole ED 2-Pole
Silver Contacts	Mom. Action AA 1-Pole AC 2-Pole CC 4-Pole												
	Alt. Action AB 1-Pole AD 2-Pole CD 4-Pole												
Gold Contacts	Mom. Action BA 1-Pole BC 2-Pole DC 4-Pole												
	Alt. Action BB 1-Pole BD 2-Pole DD 4-Pole												
Gold-Plated Silver Contacts	Mom. Action EA 1-Pole EC 2-Pole												
	Alt. Action EB 1-Pole ED 2-Pole												

* See LED information for devices without current limiting resistor, page 59.

Example: AML22CBB2AA

Square pushbutton switch housing with one LED, black bezel; red LED (without resistor); .110 × .020 termination; momentary action, 1-pole, double-throw; silver contacts.

NOTE: Order switch and lens separate will reduce lead time.

Paul Evans

CONTACT ARRANGEMENT

1, 2 or 4 poles: Form C



Turning Technology Into Switching and Sensing Solutions

Through the combined strength of Honeywell's technology centers and MICRO SWITCH's own sophisticated laboratories and engineering expertise, we offer you the industry's broadest line of switches and sensors.

As the worldwide leader in quality switching and sensing solutions, let us put our technology to work for you today.

Helping You Compete

MICRO SWITCH can help you compete by providing:

- Products that perform to specification
- Industry-acclaimed reliability
- Quality you can count on
- Application assistance and support, worldwide
- Fast, dependable service.

Warranty/Remedy

Seller warrants its products to be free from defects in design, material and workmanship under normal use and service. Seller will repair or replace without charge any such products it finds to be so defective on its return to Seller within 18 months after date of shipment by Seller.

The foregoing is in lieu of all other expressed or implied warranties (except of title), including those of merchantability and fitness for a particular purpose. The foregoing is also purchaser's sole remedy and is in lieu of all other guarantees, obligations, or liabilities or any consequential incidental, or punitive damages attributable to negligence or strict liability, all by way of example.

While we provide application assistance, personally and through our literature, it is up to the customer to determine the suitability of the product in the application.

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Section 7 - Appendix

Telos 10
Application Note #1
3/16/85

INTERFACE TO ALLOW USE OF MULTI-LINE PHONE FOR LINE-SELECT

In some installations it may be desirable to use a standard multi-line phone for line selection. Need for this may arise when the Telos 10 line capacity is insufficient, or when operators are more comfortable using the phone in the usual fashion rather than a separate button control panel.

With this approach, only the Telos line #1 relay is used.

A special interface is required to provide the Telos with ON and OFF control. The Telos must be told when a line is selected so that it can be activated and the auto-nulling can occur. When moving directly from line to line, the Telos needs to know this so that it can mute and re-null each time. Also, when no lines are selected, the Telos must be turned-off so that it does not attempt to null to an infinite impedance.

The interface circuit shown in the schematic will accomplish these functions. In addition, it allows normal use of the phone handset. In operation, it works like this: when the handset is in the cradle and a line button is pushed, the Telos takes the call; when the handset is off hook, the Telos will not activate, and the phone can be used in the usual way.

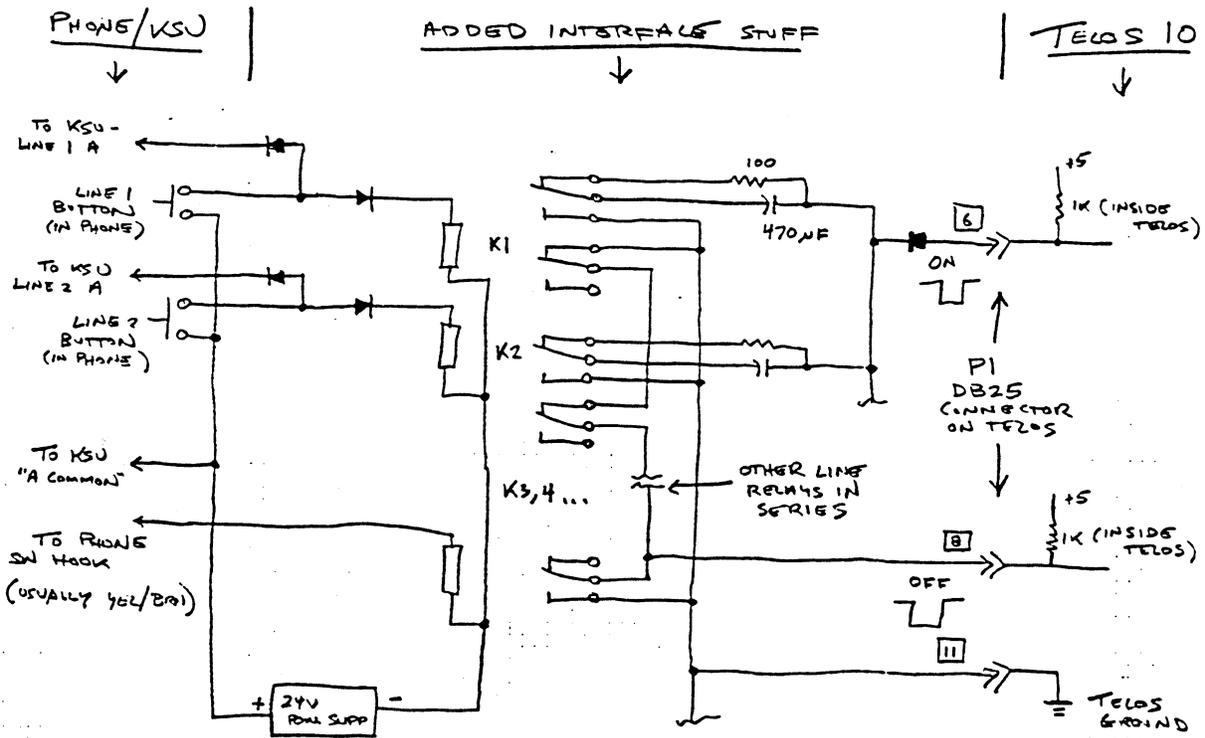
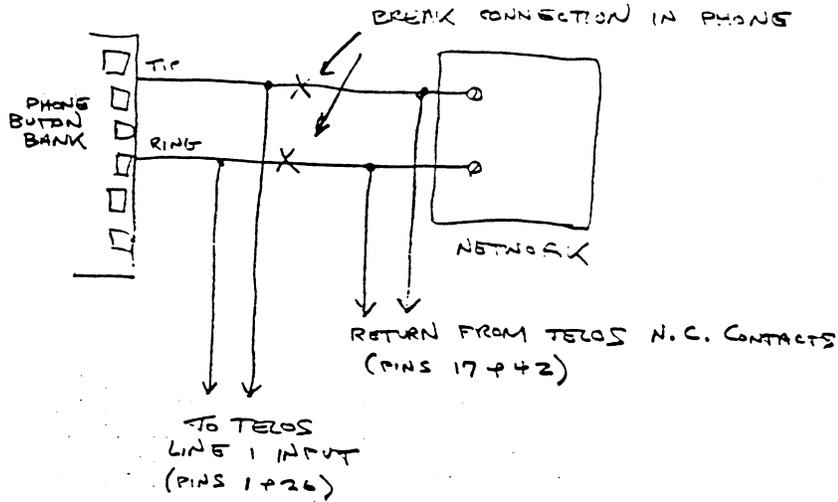
Circuit function is straightforward "bells and buzzers." When a phone line button is pressed, the corresponding relay in the interface operates. The two diodes keep the relays from operating when a line button is pressed in another room. The Normally Open relay contacts ground the capacitor, which brings the Telos input low until the capacitor charges (through the Telos' internal pull-up 1K resistor), causing the Telos to turn on.

When the line is released, the capacitor is discharged through the 100 ohm resistor. Also, the normally closed contacts re-close, causing the Telos to turn off. All of the Line relay normally closed contacts are in series so that the Telos off function occurs when all relays are released.

The lower-most relay on the schematic operates when the phone is off-hook. This prevents the Telos from turning on so that the phone handset can be used in the usual way.

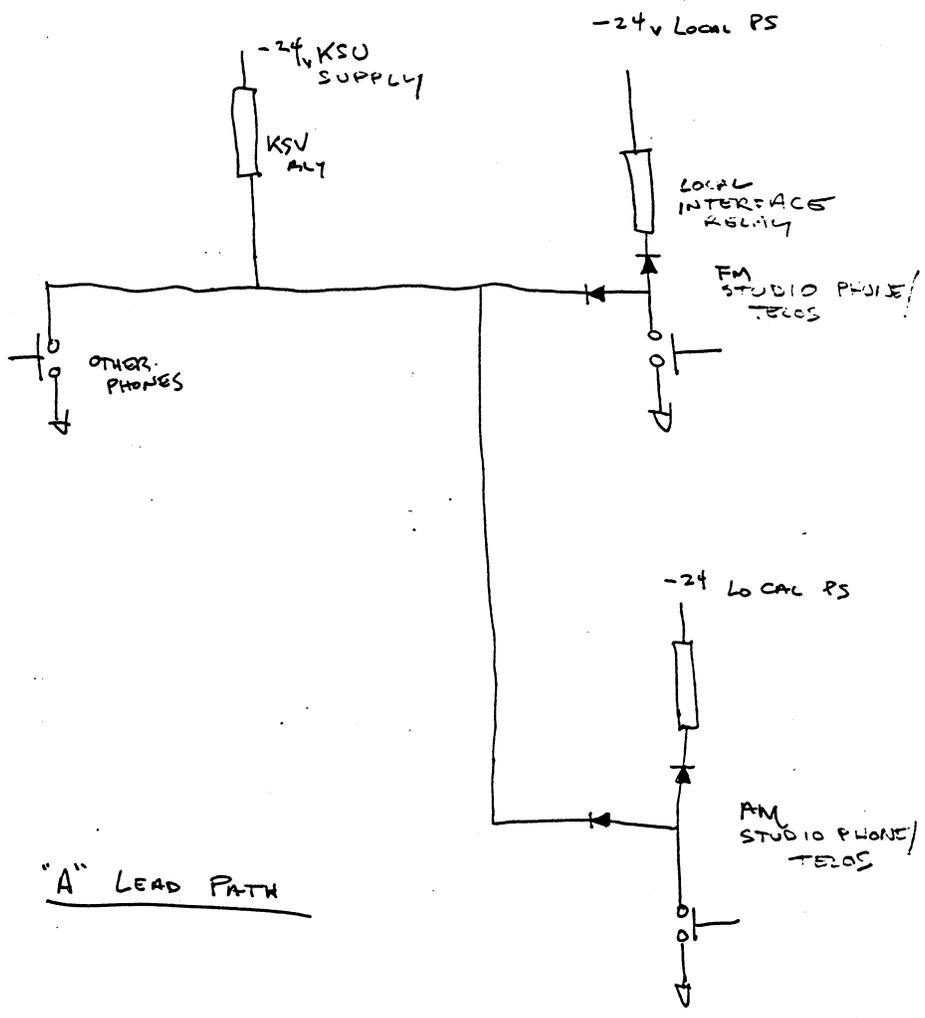
This scheme can be used even when a standard speakerphone is connected to the phone.

Thanks to Bob Helbush at WKLS, Atlanta for contributing to the design of the interface!

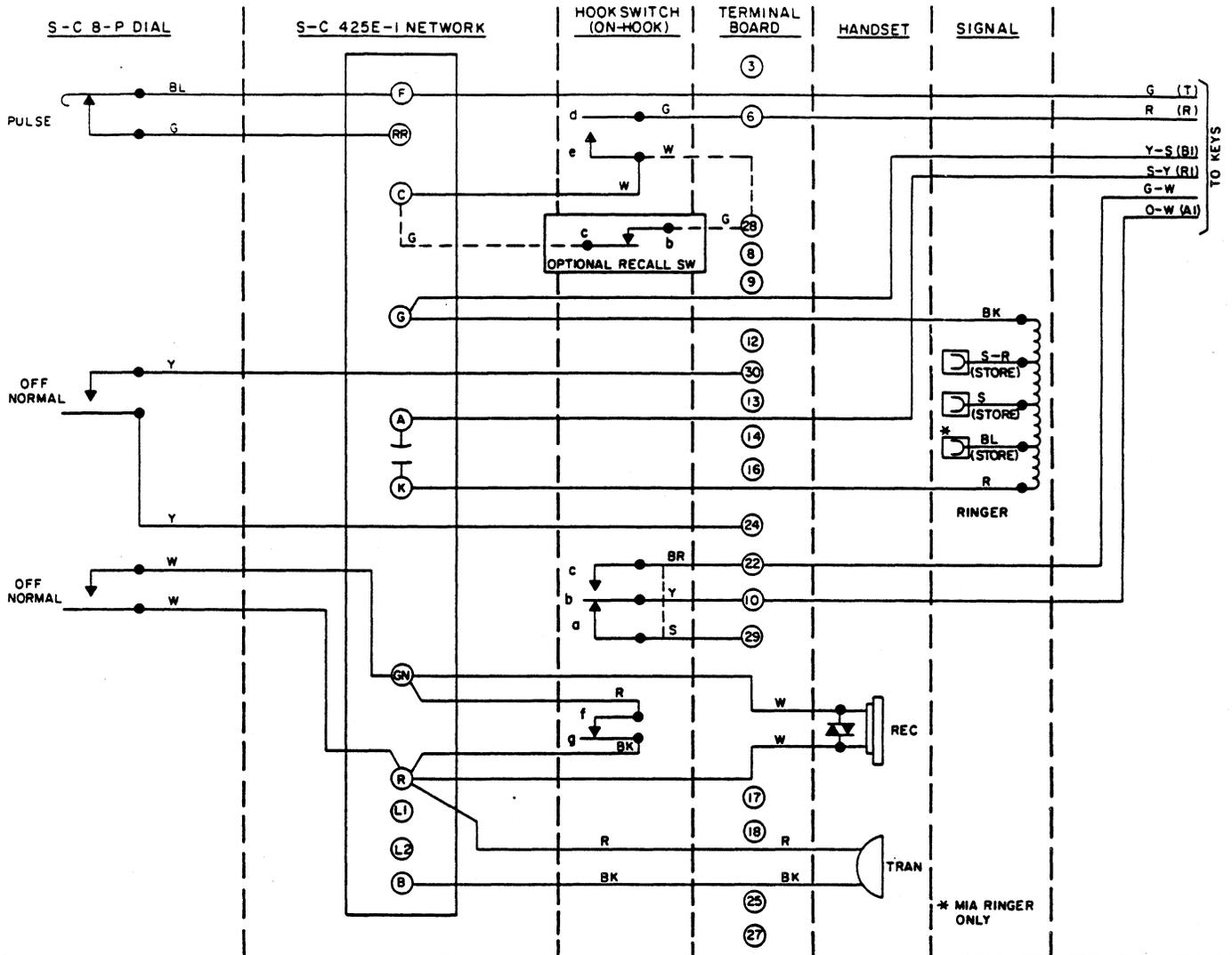


INTERFACE TO ALLOW LINE SELECTION ON PHONE ...

- 1) DUPLICATE DIODES, RELAY, RESISTOR, & CAPACITOR FOR EACH LINE YOU WANT TO USE.
- 2) ALL RELAYS 24VDC.



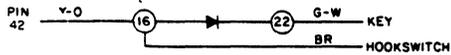
S-C 10-BUTTON SERIES TELEPHONE



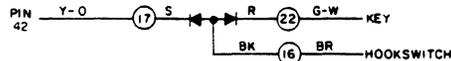
S-C 10-BUTTON SERIES TELEPHONE (LESS PRIVACY)
INCLUDES 0830-/2830-/2854- 42- 46

NOTES:

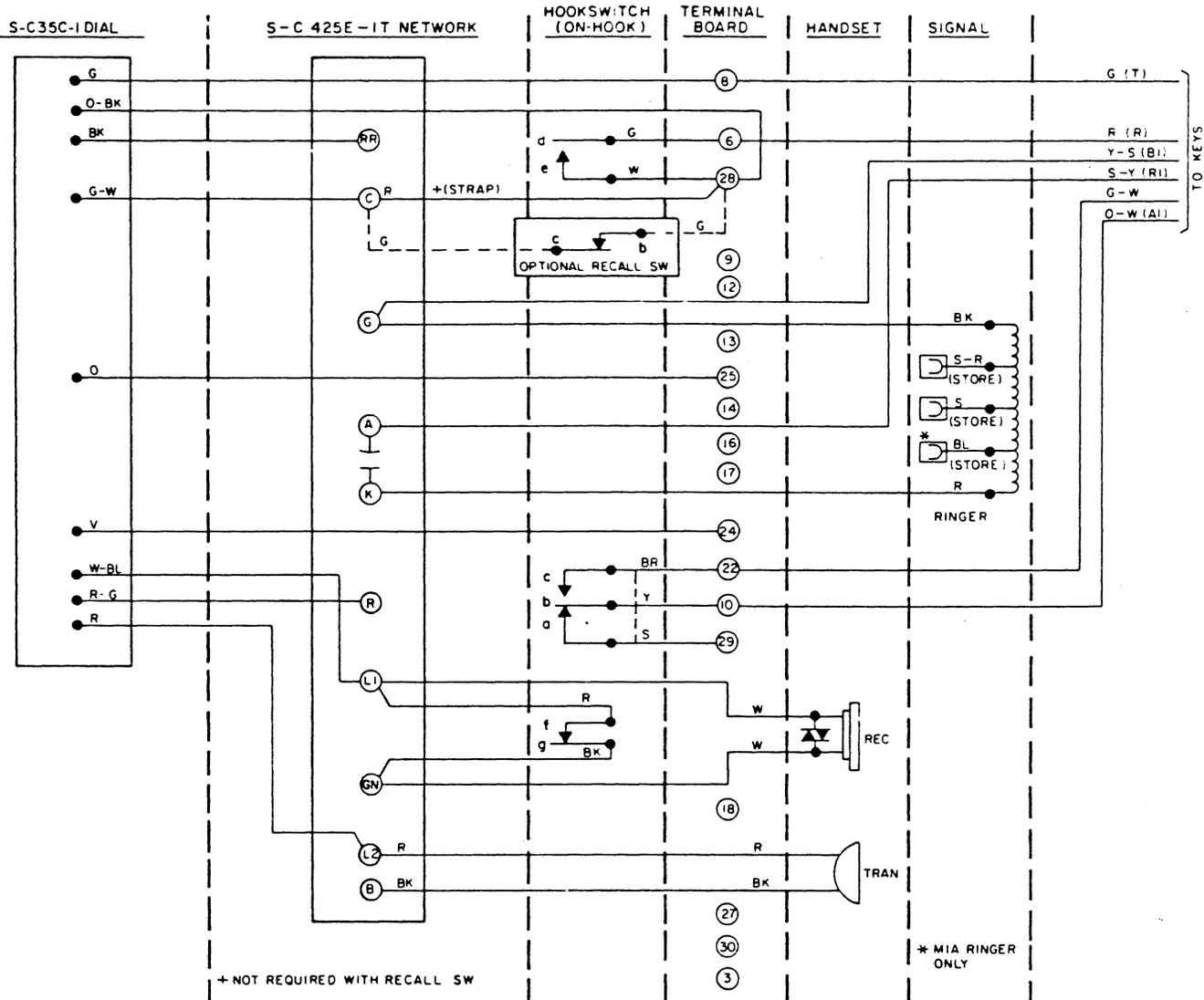
- CONNECT AUDIBLE SIGNAL (USING STANDARD RINGING VOLTAGE) AS SHOWN IN WIRING DIAGRAM. EXCEPTIONS: WHEN CAPACITOR IS NOT REQUIRED, WIRE ONE SIDE OF AUDIBLE SIGNAL TO TERMINAL "A" INSTEAD OF TERMINAL "K" (EXAMPLES: FREQUENCY RINGERS, TONE RINGERS, Q-20 BUZZERS).
- OPTIONS: DUE TO A LIMITED NUMBER OF LINE CORD LEADS, ALL OPTIONAL FEATURES CANNOT BE USED ON ANY ONE SET.
 - (A) AN 8-24 VAC BUZZER (S-C Q-20, 703015-729 CAN BE MOUNTED INTERNALLY AND CONNECTED TO RINGER LEADS OR TO SPARE (Y-O AND O-Y) PAIR AS REQUIRED.
 - (B) WIRING FOR USE WITH BUSY STATION NUMBER DISPLAY.
24 V LAMPS (24 VDC POWER): MOVE BRN HOOKEWITCH LEAD FROM 22 TO 16. CONNECT Y-O LINE CORD LEAD TO 16 AND ADD DIODE 202852-138 AS SHOWN:
 - (C) TO CONVERT LINE KEYS FROM "PICK-UP" TO "SIGNAL" FUNCTION, MOVE ASSOCIATED SIGNAL LEAD (O-W, S-W, G-R, BL-BK, BK-BR, BR-BK, BK-BL, R-G OR W-S) FROM FACTORY WIRED TERMINALS TO TERMINAL 26. CONNECT O-Y LINE CORD LEAD TO TERMINAL 26.
 - (D) THE USE OF FEATURES SUCH AS EXTERNAL SPEAKERPHONE, ETC., WILL REQUIRE AN EXTERNAL ADAPTER CABLE OR SACRIFICING LINE KEYS FOR SPARE LEADS.
- TO DISABLE BUTTON RESTORATION, TURN SCREW IN HOOKEWITCH LINKAGE APPROXIMATELY FIVE (5) TURNS IN A COUNTER-CLOCKWISE DIRECTION. READJUST SPRING TENSION ON HOOKEWITCH (IF REQUIRED) SO THAT CRADLE IS FULLY DEPRESS-ED BY WEIGHT OF HANDSET AND RETURNS TO FULL UP POSITION WHEN HANDSET IS REMOVED.
- FOR ADDITIONAL INSTALLATION AND CONNECTION INFORMATION ON THIS TELEPHONE SET, REFER TO SCP 81-269-72.



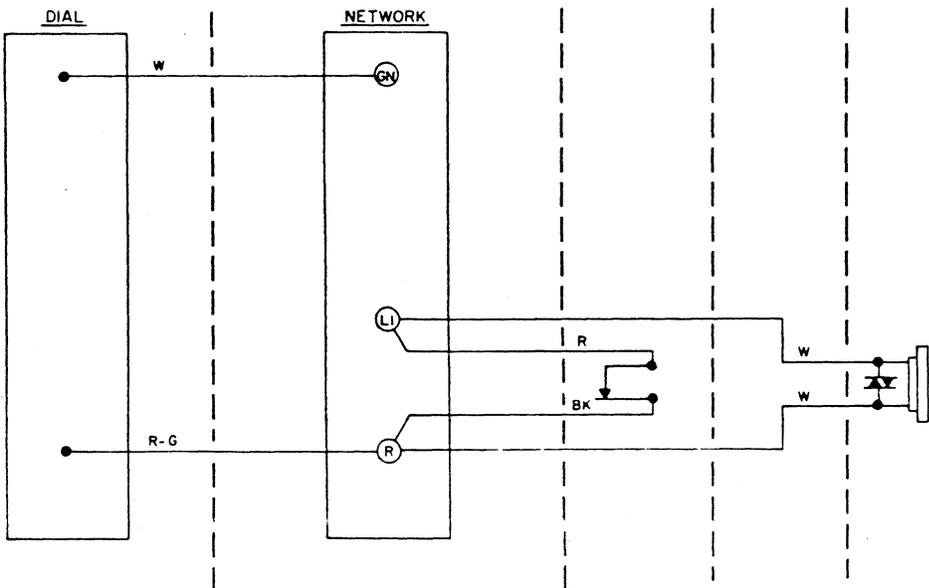
10 V LAMPS (18 VAC POWER): MOVE BRN HOOKEWITCH LEAD FROM 22 TO 16. CONNECT Y-O LINE CORD LEAD TO 17 AND ADD DIODE ASSEMBLY 206012-254 AS SHOWN:



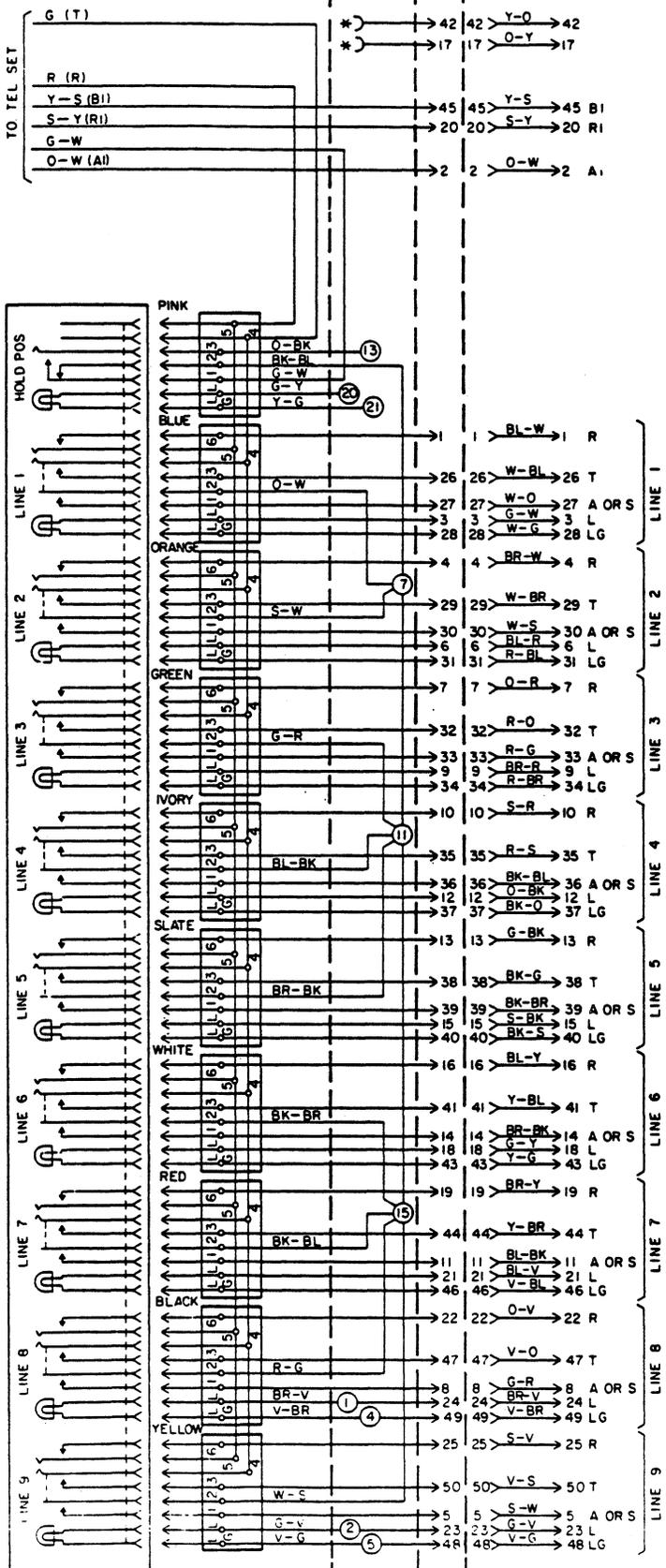
S-C 10-BUTTON SERIES TELEPHONE



PHONES EQUIPPED WITH 35C3AP DIALS-CONNECT AS ABOVE EXCEPT AS SHOWN BELOW



KEY & CONNECTOR BLOCKS TERMINAL BOARD TEL CONN LINE CORD



* TAPED AND STORED

TELOS 10

AP Note #2

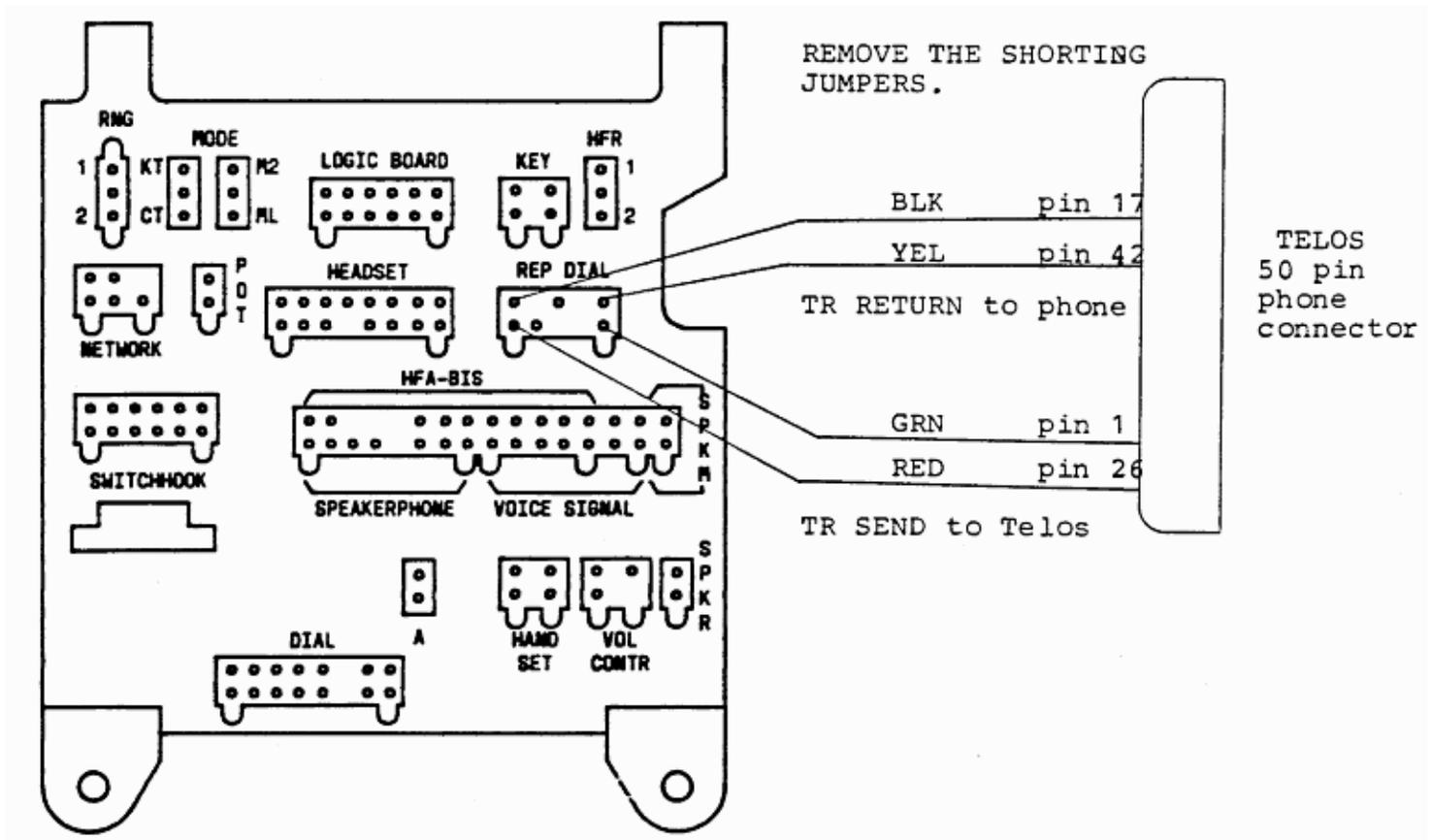
1/22/85

Using the unit with Horizon phones

The Telos hybrid is used as a "single-line" device with the Horizon MET set phone being used as the selector device.

The switched TIP and RING audio is routed out of the phone to the Telos unit. The normally-closed relay contacts in the Telos return TIP and RING to the phone when the hybrid is not active.

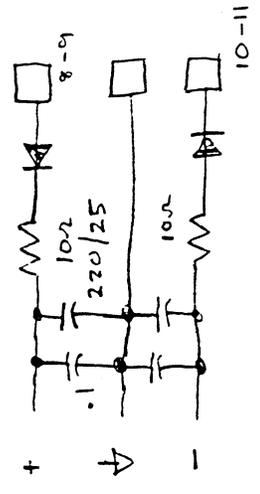
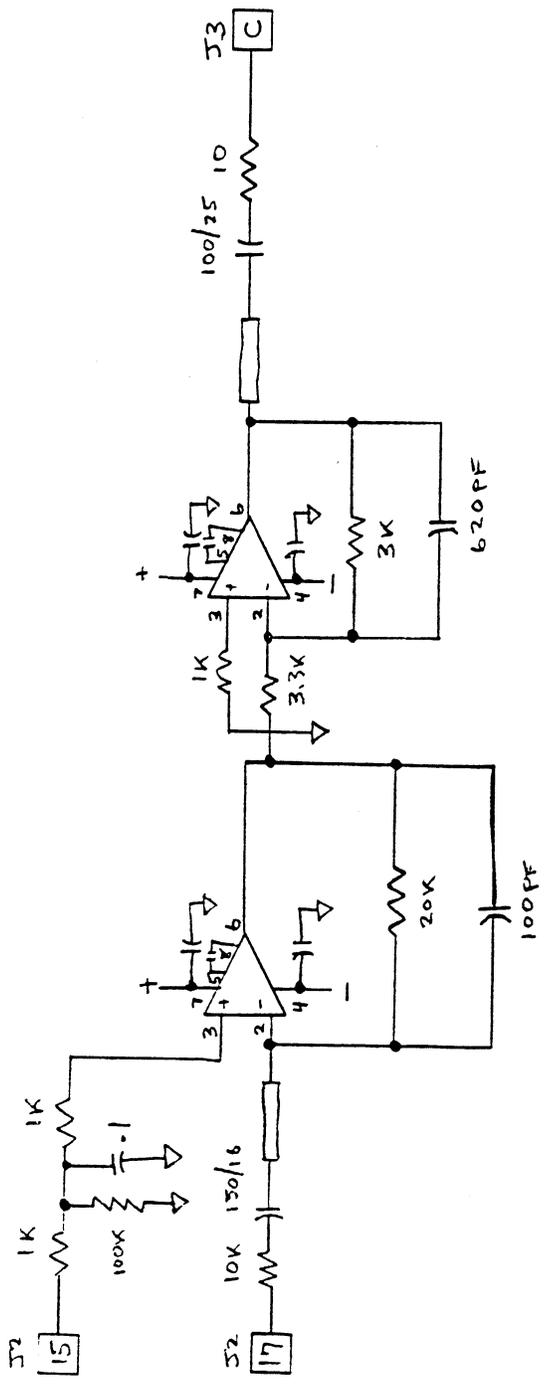
Wire the Telos phone connector to the MET set per the diagram:



Fortunately, the Horizon phones give us access to the TIP and RING. It is intended for a Rep Dialer, but it works fine for our purposes.

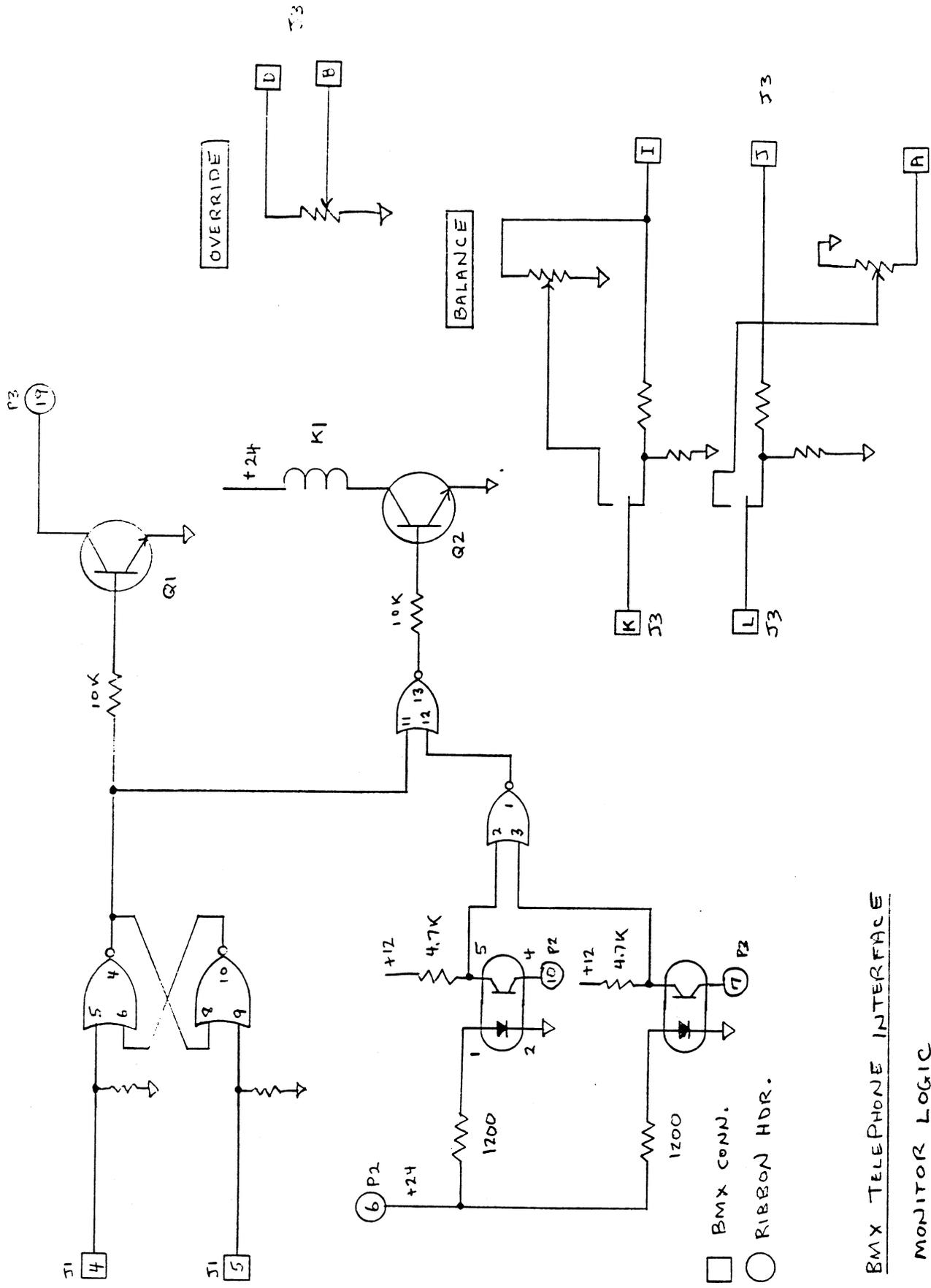
There are two limitations with this approach:

- 1) The "click mute" function doesn't work when switching line-to-line hot.
- 2) If your phone lines have widely differing impedance curves, hot line-to-line switching wont produce optimum mulling, since the Telos has no way of knowing when a new line is selected. In this situation, you will have to turn the Telos on and off each time you select a new line on the phone.



BMX TELEPHONE INTERFACE

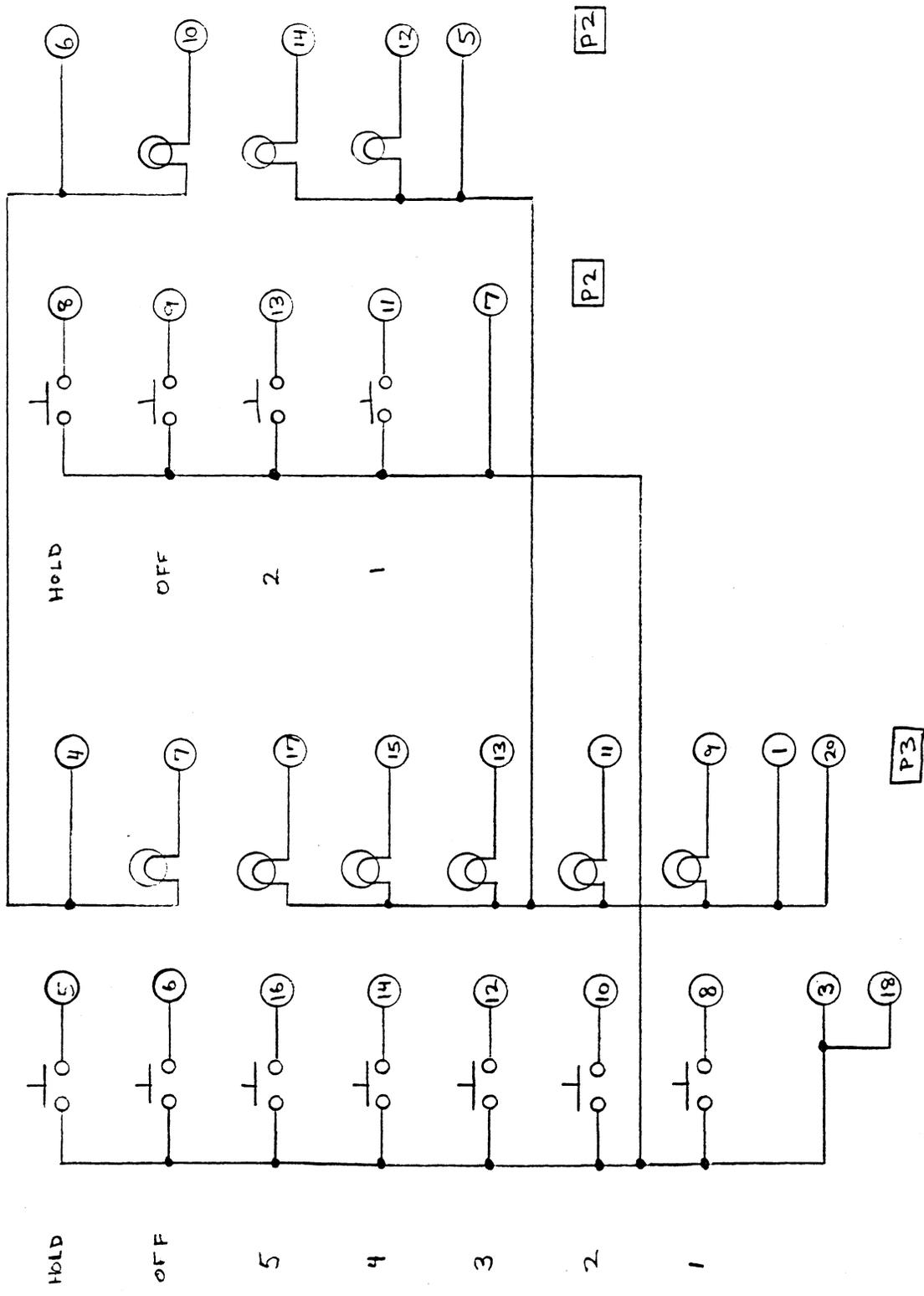
AUDIO SUMMING AMIP
7.5-PT



- BMX CONN.
- RIBBON HDR.

BMX TELEPHONE INTERFACE

MONITOR LOGIC
7.5-PT



[P3]

BMY TELEPHONE INTERFACE

PUSHBUTTON CONNECTIONS
7.5 - PT

WARRANTY and Application Caution

The Telos 10 is warranted to be free from defects in material and workmanship for a period of 120 days. Written notice of claim must be received by seller within the warranty period. In the event of a defect during the warranty period, if customer returns the defective part or the Telos 10 to a place designated by the seller, transportation prepaid, seller at its option, will either repair or replace the part or the Telos 10, and such action by seller shall be the full extent of seller's obligation hereunder. Seller will pay the transportation charges to return the part or unit to the customer. Of course, the warranty is void if the unit is subject to misuse, accident, neglect or damage.

No other warranties express or implied, all of which are specifically excluded, including, but not limited to, the warranties of merchantability or fitness for a particular purpose, shall be applicable to any equipment sold hereunder, and the foregoing shall constitute the customer's sole right and remedy. In no event shall the seller or its agents be liable for incidental or consequential damages, or for loss, damage, or expense directly or indirectly arising from use of the products, or any inability to use them either separate or in combination with other equipment or materials, or from any other cause.

The Telos 10 is intended to be used with FCC registered protective interface devices. A registered protected Key Service Unit meets this requirement.