USER MANUAL

TIF951

Dual Telephone Interface for the CS9500, CS9600, and CS9700 Intercom Systems

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RTS™

9350-7154-000 Rev B 1/98

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SECTION 1: GENERAL DESCRIPTION

The TIF95 1 connects a pair of telephone dial tone lines to the **CS9500/CS9600/CS9700** series intercom systems. Each of the two telephone lines occupies an input/output port in the intercom matrix, and each channel occupies one key panel data address.

The TIF951 provides general purpose control and interfacing for telephone lines. Each telephone interface channel may be used to dial out from any key panel, auto answer incoming calls, or accept incoming calls to be answered by key panels.

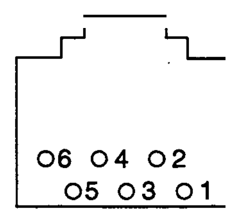
SECTION 2: INSTALLATION

2.0 Connecting the TIF951 to the intercom.

Each channel of the TIF95 1 connects to the matrix in exactly the same way as an intercom key panel. If your system utilizes the XCP954 cross connect panel, you may plug the TIF95 1 directly in **using** a 3 pair cable with DB9 connectors at each end.

If your intercom uses the XCP955 cross connect panel, you may use the DB9 to RJll adapters which are supplied in the accessory kit packaged with the TIF95 1. In the event you need to make your own adapter, the wiring is as follows (these adapters are readily available from companies which sell computer wiring accessories):

RJll	DB9	
1 (blue)	2	Data -
2 (yellow)	8	TIF input +
3 (green	4	TIF output +
4 (red)	5	TIF output -
5 (black)	7	TIF input -
6 (white)	1	Data +



RJ 11 SOCKET (FEMALE)

* Note, twist the wires, white with blue, black with yellow, red with green

2.1 DIP Switch Settings

The rear panel DIP switch contains switches to configure the most often changed options. These include auto answer mode, ring signal mode, password mode, intercom port address (key panel number), and baud rate.

2.1.1 Rear panel DIP Switch #1:

DIP switch 1 (rear panel) settings:

SW #		On (down)	Off (up)
1	Auto Answer	yes	no
2	Generate Ring Signal	yes	no
3	Password Required	yes	no
4	Address	1's	
5	Address	2's	
6	Address	4's	
7	Address	8's	
8	Baud Rate	76800	9600

DIP switch 1, address settings:

Address Settings: Rear DIP Switch, Switch # (down is on, up is *off)*

Port Number	4	5	6	7
1, 11, 21, etc	on	off	off	off
2, 12, 22, etc	off	on	off	off
3, 13, 23, etc	on	on	off	off
4, 14, 24, etc	off	off	on	off
5, 15, 25, etc	on	off	on	off
6, 16, 26, etc	off	on	on	off
7, 17, 27, etc	on	on	on	off
8, 18, 28, etc	off	off	off	on
9, 19, 29, etc	on	off	off	on
10, 20, 30, etc	off	on	off	on

AUTO ANSWER mode will set the unit to answer the phone automatically when it rings. The number of rings required before it answers is set on internal DIP switch **#2. If** set for manual answer, the line will ring until someone at a key panel answers the call.

GENERATE RING SIGNAL sets the unit so that when the phone line is ringing, certain designated key panels or circuits will receive an audible ring signal.

PASSWORD REQUIRED sets the unit so that when a call is automatically answered, it will require that the user enter a password via DTMF before the unit will allow communications. The actual password is set using internal DIP switch **#3.**

PORT ADDRESS tells the unit and the intercom which matrix inputs and outputs belong to the TIF. The settings are the same as for a key panel. The setting is the binary representation of the last digit of the matrix input/output number which that channel of the TIF is connected. See the address table above for the actual settings.

BAUD RATE sets the data rate with which the **TIF951** communicates with the intercom. The data rate for all **CS9500** intercom matrices is **9600 BPS** (switch ***off).*** The data rate for all **CS9600** and **CS9700** matrices is **76.8K BPS** (switch on).

2.1.2 Internal DIP switch #2:

DIP switch **#2,** which is located internally to the unit sets the number of rings before the unit auto answers. Please note that the ring count is approximate. This switch has no effect unless the rear panel **DIP** switch is set for auto answer mode. If the ring count is set for a high number (8), and auto answer is enabled, the unit may be used for mixed mode operation, where key panel operators normally answer the incoming call, but the line will auto answer in the event they are away from their panels.

DIP Switch #2 (internal):

1 Ring Count
2 Ring Count
3 Not used (future)
4 Not used (future)
5 Not used (future)
6 Not used (future)
7 Not used (future)
8 Not used (future)

DIP Switch #2, Ring Count settings

Rings	Internal DIP SW2, Switch # (down is on, up is *off*) 1	2
1	off	off
2	on	off
4	off	on
8	on	on

2.1.3 Internal DIP switch #3:

DIP switch #3, which is located internally to the unit selects the password. It has no effect unless password required has been enabled on DIP switch #1 (rear panel). When password required is enabled, the password must be entered via DTMF by the caller before he may communicate. This is to prevent unauthorized use of the intercom by callers. Switches 7 and 8 select the length of the password, from 1 digit to 4 digits. If set for 1 digit, only the first digit of the password is used, if set for 2 digits, then the first 2 digits are used etc.

DIP Switch #3 (internal):

1 Password select
2 Password select
3 Password select
4 Password select
5 Password select
6 Password select
7 Password length
8 Password length

DIP Switch #3, Password settings

Password	Internal DIP SW3, Switch # 1	2	3	4	5	6
4,7,8,8	off	off	off	off	off	off
7,7,7,7	on	off	off	off	off	off
4,6,8,7	off	on	off	off	off	off
1,0,5,8	on	on	off	off	off	off
1,4,8,4	off	off	on	off	off	off
7,0,3,3	on	off	on	off	off	off
5,9,0,7	off	on	on	off	off	off
0,9,3,5	on	on	on	off	off	off
3,7,8,0	off	off	off	on	off	off
1,4,5,0	on	off	off	on	off	off
6,9,2,7	off	on	off	on	off	off
8,3,0,3	on	on	off	on	off	off
8,3,3,6	off	off	on	on	off	off
6,0,8,0	on	off	on	on	off	off
2,9,5,7	off	on	on	on	off	off
5,8,5,1	on	on	on	on	off	off
9,5,9,9	off	off	off	off	on	off
8,2,0,6	on	off	off	off	on	off
4,7,4,0	off	on	off	off	on	off
4,5,7,3	on	on	off	off	on	off
8,8,3,0	off	off	on	off	on	off
0,6,2,0	on	off	on	off	on	off
3,3,3,9	off	on	on	off	on	off
9,8,5,0	on	on	on	off	on	off
7,3,5,6	off	off	off	on	on	off
9,1,4,6	on	off	off	on	on	off
9,9,9,1	off	on	off	on	on	off
3,8,8,1	on	on	off	on	on	off
4,2,4,0	off	off	on	on	on	off
1,0,6,3	on	off	on	on	on	off
8,6,3,2	off	on	on	on	on	off
4,2,3,4	on	on	on	on	on	off
0,8,5,1	off	off	off	off	off	on
0,6,7,4	on	off	off	off	off	on
0,0,1,5	off	on	off	off	off	on
6,2,9,4	on	on	off	off	off	on
9,9,5,4	off	off	on	off	off	on
1,0,7,9	on	off	on	off	off	on
9,0,3,0	off	on	on	off	off	on
0,1,6,6	on	on	on	off	off	on
9,5,5,6	off	off	off	on	off	on
8,0,5,4	on	off	off	on	off	on
6,2,9,3	off	on	off	on	off	on
6,6,1,1	on	on	off	on	off	on
6,3,6,7	off	off	on	on	off	on
1,5,2,9	on	off	on	on	off	on
2,7,8,6	off	on	on	on	off	on
8,3,1,3	on	on	on	on	off	on
1,6,5,6	off	off	off	off	off	on

DIP Switch #3, Password settings. continued

Password	Internal DIP SW3, Switch #					
	1	2	3	4	5	6
7,6,4,2	on	off	off	off	off	on
1,6,5,3	off	on	off	off	off	on
1,6,0,3	on	on	off	off	off	on
4,3,7,3	off	off	on	off	off	on
3,5,7,4	on	off	on	off	off	on
4,7,6,4	off	on	on	off	off	on
3,8,6,8	on	on	on	off	off	on
5,7,1,9	off	off	off	on	off	on
3,9,2,7	on	off	off	on	off	on
6,8,5,7	off	on	off	on	off	on
5,4,8,7	on	on	off	on	off	on
3,2,5,2	off	off	on	on	off	on
0,4,0,1	on	off	on	on	off	on
6,4,0,9	off	on	on	on	off	on
4,3,4,3	on	on	on	on	off	on

DIP Switch #3, Password Length
Internal DIP SW3 Switch #

Length	7	8
4	off	off
3	on	off
2	off	on
1	on	on

SECTION 3, OPERATION

3.1 OPERATION FROM A KEY PANEL

The TIF951 is operated from the intercom key panels, and from the dial pad on the telephone at the remote end of the line. Any key panel with a key pad may use a TIF951. All that is necessary is to program a key to talk to the TIF951, as if it were a key panel.

The alpha numeric display or tally LED for that key then provides information about the phone line. A solid display or non illumined LED indicates a line which is not in use. A slow flash indicates a line which is in use (off hook). A rapidly flashing display or LED indicates a line which is ringing. In addition, the alpha numeric display will display digits as they are dialed, and the LED will flash for each digit.

3.1.1 Programming a key to use the TIF951.

To use the TIF951, either to answer a call, or to call out, you first need to program a key to talk to the TIF951. This is accomplished in the same manner as programming a key to talk to a key panel. To program a key by port number, enter NUM-nnn-PGM-t, where NUM is the number 1 key, nnn is the port number of the TIF951 you want to use, and t is any talk key. In general, you will also need to use the listen key, so it should be assigned as either AF (auto follow), or AL (auto listen).

Note that the TIF951 only responds to commands which are sent via a point to point key assignment. If you wish to use the TIF951 primarily on a PL, you must add a point to point assignment as the L2 talk assignment on the talk key for any panels which are going to either answer the line, or dial out on the line.

3.1.2 Dialing a call:

Any key panel may dial calls on the TIF-951.

To dial a call on the TIF951:

1. Turn on the listen key for the line you wish to dial on. This will allow you to hear dial tone, and your DTMF dialing tones.
2. Enter dial mode by entering PHONE-PGM-T. **PHONE** is the 4 button on the keypad. PGM is the red PGM key on the key pad, and T is the talk key which is programmed to talk to the TIF951 you are dialing on. Leave the talk key in the up engaged position as you dial the number.
3. Dial the number. As you enter each digit, it will appear in the alpha display above the key you are dialing on. On panels with LED tally, the LED will flash on each digit. If the listen key is engaged, you will hear each DTMF tone as it is generated.
4. When you have completed dialing, momentarily disengage the talk key to exit dial mode. The alpha numeric display will revert to normal, and you may use the key and key pad in the normal manner.

   Note: The key pad is used in the usual way. Digits 0-9 generate the DTMF digits 0-9. PGM generates #, and CLR generates * (* and # are displayed for these keys). If the last number redial/speed dial option is present, it is necessary to press CLR twice if you wish to generate a *, as a single CLR is used to trigger the speed dial and redial features.

3.1.3 Hanging up:

The TIF951 will detect that the caller at the far end has hung up under most circumstances. It detects the hang up by either loop interrupt, battery reversal, or the presence of dial tone. Some telephone systems do not provide any of the above, so it will be necessary to force a hang up. In addition, if the call was placed to an auto answer device, it will be necessary to force a hang up when the *call* is complete.

Enter PHONE-CLR-t, where **PHONE** is the 4 button on the key pa4 0 is the 0 button, PGM is the red **PGM** button, and t is the talk key which is programmed to talk to the TIF951 which you wish to **hang** up. This will disconnect the line for which you struck the *talk* key.

Note that if the talk key is in the on position, you must turn off the key, then momentarily turn it on again to indicate which line you wish to disconnect. If the line is in dialing mode, then you must first exit dialing mode by **turning off** the key, then use PHONE-CLR-t to hang up.

3.1.4 Re-dialing the last number:

The TIF951 remembers the last number which it has dialed.

1. Enter dialing mode by following the instructions for dialing a call.
2. Enter CLR-0-0. The TIF will automatically redial the last number it dialed.
3. Momentarily release the talk key to exit dialing mode.

Note that the number is remembered on a phone line by phone line basis. When the last number redial command is issued, you will get the last number dialed on that TIF951 channel, regardless of which key panel dialed the number. For example, if you have a call to 818-566-6700 on channel 1 of the TIF, and a call to 201-891-6002 on the other channel, and you are disconnected, issuing the redial command will re-establish the calls on the same channels. The redial command may be issued from any key panel in the intercom, not just the key panel that originally dialed the calls.

3.1.5 Dialing a Speed Dial number (stored number):

The TIF951 has 32 internal memories for storing frequently used phone numbers. To dial one of these numbers:

1. Enter dial mode.
2. CLR-nn where CLR is the clear button on the key pad, and nn is two digits, which are the speed dial code.
3. Momentarily release the talk key to exit dialing mode.

3.1.6 Storing a speed dial number:

1. After dialing the number the usual way, enter the CLR-PGM-nn before you release the talk key to exit dialing mode.
2. Momentarily release the talk key to exit dialing mode.

   Note: To generate a pause during auto dial, enter *99. This is used for example if you need to enter a digit to get an outside line, and your phone system requires a pause before continuing to dial.

Each number may contain up to 30 digits. Note that the memory for the stored numbers is RAM in the TIF951. The numbers are unique to each TIF951 channel, and each TIF951 channel can have different numbers stored in it. If you plan to use the speed dial feature, it is advisable to supply the TIF951 with UPS power, to prevent loosing the speed dial information when the power fails.

3.1.7 Answering a call:

1. When a line is ringing, the alpha numeric display or LED above the talk key which is programmed for that line will flash rapidly.
2. To answer the call, first turn on the listen key, then press the talk key and speak into the microphone or headset.
3. If you have been programmed as a default station, your panel will "ring" when ever one of the lines rings. If you do not have a key already programmed, the ringing line will appear on your incoming call key (the key farthest to the right on the main panel). To answer, press the incoming call key and answer. You should copy the key to main key position, either just before or just after you answer, so you can turn on the listen key to hear the caller audio.

3.2 TIF951 SYSTEM SETUP TO RECEIVE CALLS

To the intercom system, the TIF951 is very much like a key panel. If the phone lines are to be used for outgoing calls only, then no programming in CSEDIT is necessary. If users are going to phone into the intercom from the outside, then the TIF951 needs to be configured to allow them to use the phone line in much the same way a local user uses his key panel.

Programming information for the phone line is entered into the CSEDIT software just as if the TIF951 were an ordinary key panel, by selecting "Keys" from the main menu, then selecting the TIF951 from the pick list of key panels. The TIF951 operates much the same way as a key panel, except that the "keys" are really the DTMF buttons on the users telephone.

32.1 Auto Answer Mode:

To use the TIF951 in auto answer mode, you must first enable auto answer mode on the rear panel DIP switch, toggle number 1. You may also wish to enable Password required, toggle number 3. In addition, you may select the number of rings before the unit answers (internal DIP switch #2), and the actual password (internal DIP switch #3).

When the caller dials into the TIF951, he will hear the line ring, then the unit will answer, and beep to request the pass word (if password required is enabled). The user then must enter the password. The unit will beep once to confirm a proper password. If the password is not correct, the unit will beep twice to allow another try.

Once the password has been entered, the TIF951 will establish communications on key #1 automatically. From CSEDIT, this will be talk and listen keys #1. If for example the user were a camera operator, it might be desirable to program the camera PL as talk and listen on talk and listen keys #1. If the caller were a reporter, you might program an IFB on listen key #1, but no talk on talk key #1.

Keys 2 to 7 may also be programmed. To use the other keys from the phone, just press the DTMF button for the key you wish to use. For example, if key #1 was the camera PL, and you have finished with the shot, you may press #1, which will toggle off key 1. If master control were programmed on key #2, you may then press 2 and call master control. Likewise, you might have an IFB programmed on listen 3, with no talk. If you press 3, you will hear the IFB. #4 could have an IFB talk on it, to allow a caller to speak on an IFB circuit.

Each DTMF button acts as if it were a push on/push off switch. When programming in

CSEDIT, just program the same key number as the number the user is going to press on the telephone to speak.

Talk keys 8 to 15 have a special purpose. If you are not using auto answer mode, but have set up the TIF951 to be manually answered, talk keys 8 to 15 will be programmed for the panels which are to receive the ring signal. They may also be toggled on and off from the phone by DTMF 8, so they may be used in auto answer mode as well. You may program only key 8, in which case it will behave the same as keys 1-7. You may also program additional panels, PL's, IFB's etc on keys 9-15, and they will be activated simultaneously by the 8 button on the phone.

3.2.2 Manual Answer Mode:

In manual answer mode, the line will ring until it is answered from a key panel. In general, you must designate panels which are to receive the "ring", so they can answer the line. When a line is manually answered, the caller does not have to enter a password, even if the password required switch is turned on. You may *mix* modes by enabling auto answer, but setting the ring count for 8 rings. If no user has answered the call by 8 rings, the TIF will then automatically answer the call, and if password required is also enabled, the call will be screened by requiring a password.

To use manual answer mode, you may choose to program keys 1 to 7 as above if you wish. When the phone is manually answered, key one will not be automatically activated, but the caller may activate any of the keys if he wishes.

You must also designate the panels which are going to ring when the line rings. Program these panels on keys 8 to 15, using both L1 and L2 if you have more than 8. It is generally not necessary to program the listen keys on these positions. When the line rings, the TIF951 will "call" these panels when the line is ringing. When the line rings, the TIF951 generates a ringer noise which is then transmitted to these panels. The panels will display the TIF951's alpha numeric in the incoming call window, and if a talk key has already been programmed on the panel, it's alpha numeric will flash rapidly.

3.3 Using the TIF951 from the telephone:

The TIF951 will behave differently depending on how it is programmed. It is up to the operator who programs the TIF951 to convey to the user what to expect. If the user is not familiar with the operation of the TIF951, it is best keep the operation as simple as possible, until they are familiar with its operation. For this reason, it is suggested that you not use password required unless you have had problems with nuisance calls in the past. If the TIF951 field user only requires one service, it is best to program that service on key 1, enable auto answer, and disable password required. The telephone user will then only have to dial the proper phone number to use the interface. As they become more familiar with it's operation, you can then begin to offer more options to the users, or begin to require a password.

In general, it is very easy to use, if the user has knows what to expect.

When calling in, if the unit is in auto answer mode, it will answer the call after the number of rings which have been selected. If password required is **not** enabled, the unit will indicate it is ready with a single beep. If password required is enabled, the TIF951 will prompt for a password with 2 beeps. The user will enter the password, and the unit will beep once if the password was correct, twice was wrong. The user is allowed 3 attempts to enter the password, after which the TIF951 will disconnect. In the event a user calls the TIF951 when the intercom system is either turned off or absent, the TIF951 will answer prompt with 3 beeps.

Once the password is entered, the TIF951 will enable talk and listen on key 1. This should be programmed ahead of time to what ever communications the caller generally needs first. If it is not desirable for the caller to be able to talk at this point, then only the listen key for key 1 should be programmed.

The caller may then either continue to use key 1, or he may select other keys with his **DTMF** pad. He may turn off key 1 by pressing **DTMF** 1, or may continue to just add other keys. At any time, the caller may turn off all keys without hanging up by pressing 0.

When the call is complete, the caller should enter *#, which will cause the TIF951 to disconnect. This is more reliable than waiting for the phone system to pass the disconnect information to the TIF951.

33.1 DTMF codes

Once programmed as described in section **3.2,** the TIF951 may be operated via the **DTMF** Touch Tone™ key pad on the telephone. The DTMF keys have the following **functions:**

Normal Mode:

1 thru 7	Toggle on and off talk and listen #1 to #7. Note that initially, #1 will be enabled if the unit auto answered the line.
8	Toggle on and off talk and listen to the panels which ring when the line is ringing. This allows the caller to "recall" the panels without having to hang up and redial. Toggling this on will allow the callers voice to be heard from all the panels which normally ring.
9	Enters programming mode, to reassign keys.
0	Turn off all talk and listen keys. Since 1-8 are toggles, it is possible to forget which keys are "on" and which are "off". In this case, just press 0 to turn then all *off,* and start over.
*1 thru *7	Toggle on and off listen **1-7.** By pressing * before the key, you only change the listen. This allows you to listen to a circuit without talking to it, or to talk to a circuit without listening to it. Note that you will automatically listen and talk to #1 if the TIF951 auto answered the line.
*8	Toggle on and off listen for 8-15.
*#	Disconnect. This will cause the TIF951 to hang **up.** It is a good idea to do this before you hang up, as many phone systems take a long time to signal that the far end has hung up.

Programming Mode:	You may reprogram the talk and listen assignments on 1-7, just as you can on a key panel (if they are not restricted via CSEDIT). Note that the sequences are the same as the sequence you would use from a key panel, except that you must first enter programming mode by pressing 9.
1 nnn # K	Program a talk key to a point to point.
2 nn # K	Program a talk key to a PL.
01 nn # K	Program a talk key to a special list.
02 nn # K	Program a talk key to an IFB.
03 nn # K	Program a talk key to an ISO.
04 nn # K	Program a talk key to a Relay.
3 5 # K	Program a talk key to All Call (turns on the lower numbered talk keys)
1 nnn # *K	Program a listen key to a point to point.
2 nn # *K	Program a listen key to a PL
3 2 # *K	Program a listen key to Auto Follow
3 3 # *K	Program a listen key to Auto Mute
01 nn # *K	Program a listen key to a special list.
02 nn # *K	Program a listen key to an IFB.
03 nn # *K	Program a listen key to an ISO.
04 nn # *K	Program a listen key to a Relay.
*9	Exit programming mode
*0	Exit programing mode and turn off all talk & listen.
*#	Disconnect

Note: 0-9 are the number keys, * and # are the star and pound keys.

nnn is three digits for a key panel number

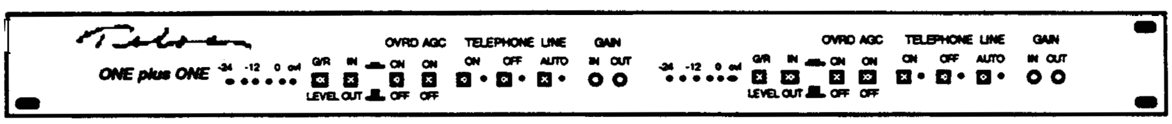
nn is two digits, for an IFB, PL, Relay, Special List, or ISO.

K is a key which you are programming, just press the digit (1-7).

*K is the * key followed by a digit (1-7). This is used to represent the listen key.

Telos *ONE plus ONE*
Dual Digital Telephone Interface

User's Manual
1.1



Telos

SYSTEMS

Telos Systems • 2101 Superior Avenue • Cleveland, OH 44114
(216) 241-7225 • FAX: (216) 241-4103

July 10,1989

A personal note:

You have in your possession a remarkable piece of technology. The Telos One does what would have been either impossible or impractical only a few years ago. It relies on ***digital signal processing,*** a concept **known** to theorists for years but only very recently available to us **as users.**

It is entertaining to read the **signal** processing textbooks written as recently as the mid 70's. The authors describe the state of the art: a few seconds of audio is collected and processed with **FORTRAN** programs running on million dollar mainframe computers. Only after minutes (or hours!) of expensive number crunching did the expectant researchers get to ***actually*** **hear** the brief audible result.

About the same time, **yours** truly began **his** first **radio** station job. Using phones on air was always a problem owing to the familiar shortcomings of speakerphones and hybrids. Thus began what was to become many **years** of tinkering with telephone interfacing. Nothing had worked - resulting in discouragement having become firmly rooted when, in **1983,** articles describing practical *real-time* DSP began to appear in obscure **journals.** This was made possible by the introduction of single-chip processors optimized for use in manipulating analog signals. They cost $350 - but I sensed that their availability signalled the beginning **of** a revolution. The next year and a half found me at work weekends and evenings learning the exciting new technology and experimenting with telephone interfacing approaches using it. By late **1984,** the now famous Telos 10 was the result.

It had the singular virtue among available interface devices that it ***actually worked.*** At last it was possible to carry on a natural on-air conversation without the common up-cutting or distortion difficulties. We put it on the **air** at WFBQ, Indianapolis and made a few for **friends.** Slowly, the word spread. Since I was happily employed and thus had no compelling interest in Telos' economic success, it was only amusing **to** observe the digital hybrid technology take the usual path of any new idea to eventual acceptance:

- *It was ignored*
- *It was accused (by other interface manufacturers) of being ineffective*
- *It was accepted by users with tolerance for risk and novelty*
- *It was accepted by large numbers of users*
- *It was co-opted and copied!*

For a couple of years, Telos Systems was operated as a sideline enterprise while I continued to work as a CE. However, as is now evident, increasing sales caused Telos to grow beyond its "garage" origins to become a bona-fide broadcast manufacturer, adding staff, an office, phones, and an occasional ad. I finally even had to quit my job!

Our research continued and resulted a little over a year ago in the second generation Telos 100 hybrid - which advanced the state of the art by significantly improving performance and taking advantage of DSP for the dynamic processing functions.

So here we are in the present. The work of the signal processing theorists for decades (actually, centuries Fourier was at it a few hundred years ago!), our continuous work over the course of the past six years, and the near-incredible advances in digital audio and low-cost computing power have combined to allow the creation of a result even the digital dreamers would have been shy to predict a decade ago.

We trust you wili like it.

Keep on keeping the GM happy.

Steve Church
President

User's Manual V1.l
Telos One plus ONE
Dual Digital Hybrid Telephone Interface

SECTION 1
INTRODUCTION

1.1 OVERVIEW

The Telos ONE plus ONE Dual Interface

The Telos **ONE** plus **ONE** consists of **two** Telos One digital hybrids in **one** 19 rack mount *enclosure.*

The Telos One interface embodies a state of the art approach to interfacing telephone lines for broadcast on-air, intercom interface, **or** teleconferencing use. The very fast and precise digital automatic nulling hybrid allows smooth, natural, simultaneous conversation without the **usual** speakerphone up-cutting effect or the audio distortion **and** feedback problems often experienced with poorer hybrid-type interface devices.

As well, a number of additional functions are accomplished in the digital domain in order to enhance "real-world" performance. Included are sophisticated automatic gain control in both the send and receive paths, a carefully-implemented override ducking system, and a pitch shifter for feedback reduction.

Telephone connections ***are*** via standard modular jacks, while audio input and output are connected via **XLRs.** Each hybrid **has** one balanced input with provision for mic or line levels and two balanced outputs. The second output may be switched to be either a second isolated output or a mix of the send and caller **signals.**

Purpose

The purpose of the Telos One broadcast telephone hybrid is to deliver **to** the receive output pure caller audio with **as** little **of** the send (announcer) audio **as** possible mixed-in. Until digital signal processing techniques were applied to the telephone interface problem, there were **two** choices:

> ***Switching.*** The send and receive paths were separated by having only one talk direction active at a time. The common "speakerphones" use this approach. The disadvantage **is** that natural conversation is impossible, since the caller is cut-off when the announcer talks - and vice-versa.

Analog **hybrids.** These were, on most phone lines, *very* poor at removing the send signal from the caller's audio. **This** meant that the announcer's voice would become distorted **as** the phone audio was added to the **mix.**

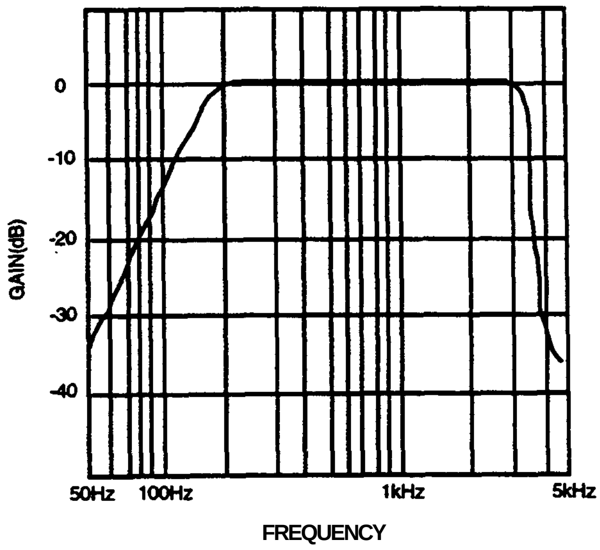
(A full discussion of hybrids and interface systems is included in the *Telos Telephone Q & A.)*

The Telos One is a *true digital* second generation telephone interface. It **uses** state-of-the-art digital techniques to perform the hybrid function - the subtraction of the **send** from the receive audio. The input and caller audio signals are converted to digital and operated on in **such** a way **as** to very effectively remove the send audio from the output while maintaining natural simultaneous full-duplex conversation. The digital approach assures consistently good trans-hybrid loss regardless of varying phone **line** impedance.

Special Features

The Telos One incorporates sophisticated audio processing **in** *the digital domain* for **gain** control and filtering.

- **A** digital high-pass filter is **used** to reduce hum and other low frequency interference. High frequency noise above the telephone frequency range is **also** attenuated.



- Smart Digital Automatic Gain control smooths input and output levels. A noise-gate/downward expander is provided on the receive path to reduce phone line noise during caller pauses.

- A switchable *override* function is provided to allow ducking of the caller while the announcer is speaking. The override function includes an *acoustic ducker* which dynamically reduces send audio when caller audio is present in order to reduce feedback and aid natural conversation.

- Unique to the Telos One is a special feedback reduction function using a pitch-shifting approach. The input (send) audio is shifted downward in frequency by 4 Hz to help prevent feedback build-up.

- Front panel metering is provided for input level, output level, and *gain* reduction.

Operation

When a call is initially established, a brief mute/adapt period provides an opportunity for the system to set up to the line before the call is passed to the output. The caller hears a "noisy tone," but none of this tone is heard at the output since the output is muted during this time. This has the incidental benefit of removing the line switching "clunk." Adaption to the telephone line characteristic continues as the conversation proceeds using voice as the driving signal.

The Hardware

The process of analog-to-digital and digital-to-analog conversion is critical to audio quality and hybrid performance. In the Telos One, IC converters called *CODECS* intended for telephone central office application are employed. The ICs in the Telos One are better than the usual telephone CODECS in that they use an oversampling and digital filtering technique for the anti-aliasing and reconstruction filters rather than the usual switched-capacitor filter approach. Thus noise and distortion are quite good.

Because the audio processing functions are performed in the digital domain, the hardware design of the Telos One is quite simple.

Multi-Line Systems

Telos makes interface modules for multi-line switching. The Direct Interface Module offers a convenient and flexible means to switch up to 10 telco lines with provision for program-on-hold, additional telephone sets for off-air conversation, etc. The 1A2 interface module is for use within a standard 1A2 key telephone environment. A number of options are available with regard to interface methods, control panels, etc. Please contact us for details.

In addition, the "Super" Auto-Answer PCB module may be configured so that the Telos One may be connected directly to 1A2-style telephones with the telephone itself being used as the line selection device.

In some cases, the Telos One may be interfaced to "electronic" telephone systems. There is a tutorial on these phone systems included later in this manual which should be helpful to those who wish to do this.

1.2 SPECIFICATIONS

System

True digital. **Second** generation Texas Instruments **TMS320C25** processor. **8** kHz sampling rate. Internal digital input and output **gain** processing, filtering.

Trans-hybrid Loss

>40 dB with pink noise or voice **as** test input. Test set-up **as specified** in our ***Telephone** Q&A. **All dynamic enhancement processing is** switched **off.*** With the ovemde and output expander **functions** switched-in, trans-hybrid **loss** is enhanced by approximately 12 dB.

Send Level to Phone Line

-10 dBm average level. Maintained by intemal digital AGC.

Frequency Response (callerto output)

200 - 3400 **Hz** +-1dB.

Noise and **Distortion** *(caller to* **output)**

Distortion: <.5% **THD + N.** 1kHz; ***caller*** levels from -48 to -8 dBm. Signal-to-Noise: >60 **dB.** Referred to **-18** dBm phone level. **>72 dB** ref ***to*** 0 dBm phone line level.

Send Audio Input

XLR female connector. Active balanced. Accommodates **-24** to +12 dBm levels in LINE mode; -68 ***to*** -35 dBm in **MIC** mode. Front panel screwdriver level adjust.

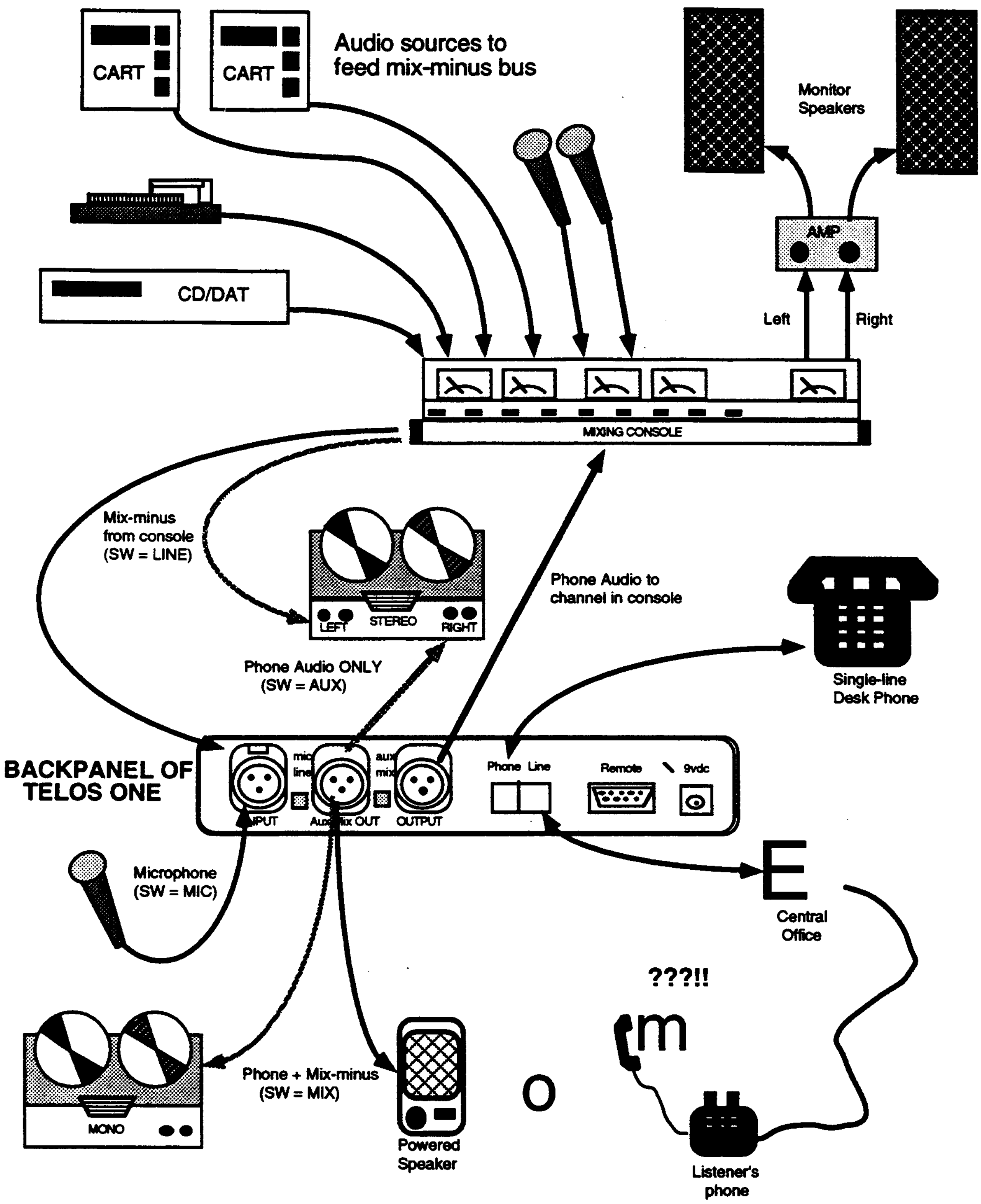
Caller ***Audio Output***

XLR male connector. Active differential. Output levels to +14 dBm depending upon caller telephone line level and adjustment of front panel level adjust. Will drive 600Ω.

Aux/Mix Output

XLR male connector. Active differential. In AUX mode, this output is **an** isolated second output. In **MIX** mode, **this** is a combined send and caller output. **INPUT to MIX** Output **specifications: Unity** gain; <.04% THD; +12 dBm clip point.

SECTION 2
INSTALLATION



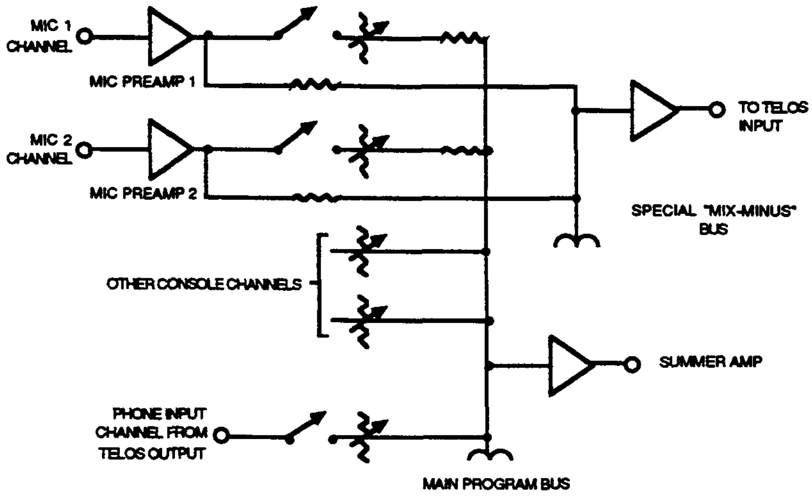
CART
CART
Audio sources to
feed mix-minus bus
Monitor
Speakers
AMP
CD/DAT
Left
Right
MIXING CONSOLE
Mix-minus
from console
(SW = LINE)
LEFT
STEREO
RIGHT
Phone Audio to
channel in console
Single-line
Desk Phone
Phone Audio ONLY
(SW = AUX)
BACKPANEL OF
TELOS ONE
mic
line
aux
mix
INPUT
Aux/Mix OUT
OUTPUT
Phone
Line
Remote
9vdc
Microphone
(SW = MIC)
Central
Office
???!!
MONO
Phone + Mix-minus
(SW = MIX)
Powered
Speaker
Listener's
phone

2.1 AUDIO

The drawing on the preceding page shows how the unit may be incorporated into an on-air studio environment.

2.1.1 Mix-minus

The Telos One-plus-One send inputs should be fed ***mix-minus*** audio. That is, the ***mix*** of all the sources you want to feed the phone ***minus*** the hybrid output itself.



Broadcast Consoles

Most modem broadcast consoles make some provision for mix-minus. The best allow selective feeds to the telephone system. This is useful since you sometimes want only one mic feeding the phone, sometimes you want three or four **mics** (during the morning show, for instance), and sometimes you want to feed cart **machines** when callers need to hear and react to contest effects, etc.

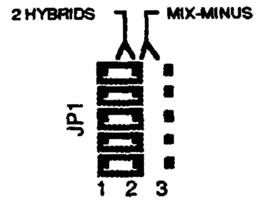
User-Provided Mix-Minus

For a simple installation, you can just take the patch send or pre-**amp** output from the mic channel to feed the phone. **This** works well, but doesn't have much flexibility. One approach which allows more control is to **use an** outboard mixer to combine sources **as** desired. All of the desired sources are paralleled into the on-air board and the mixer and the mixer's output feeds the hybrid.

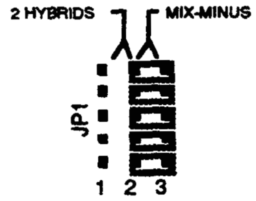
Internal Mix-Minus *in the* ONE *plus ONE*

The two hybrids in the One-plus-One unit may be configured so that a single mix-minus feed may be used for both hybrids, with each hybrid's output fed to the other's input *internally at* **unity gain.** Both hybrid **outputs still** function independently.

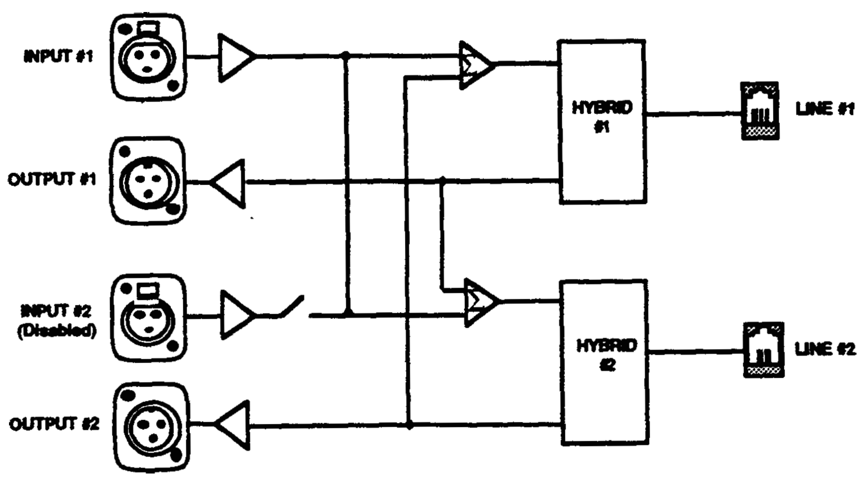
The internal mix-minus cross-couples each **hybrid's** receive audio to the other's input and sums the user-provided **mix-minus** to the inputs. The user mix-minus must have NO hybrid audio **mixed** in it. Five jumper pins are **located** on the connector board, marked JP1. Move **all** five jumpers from the "2 **HYBRIDS**" position to the "MIX/MINUS" position. Then feed your mix-minus to the **#1** hybrid input. The input to the #2 hybrid is disabled. Both input trimmers on the front panel function normally. The cross-coupled hybrid audio is **fixed** at **unity** gain, **relying** on the **output** AGC to keep levels OK. The block diagram below and schematics should help to make all of this clearer.



JUMPERS SET FOR TWO HYBRIDS



JUMPERS SET FOR INTERNAL MIX-MINUS



Internal Mix-Minus Scheme

A Good Idea ...

Here's a **useful** scheme for stations which do a lot of taping of calls for later play on the air. The **mix-minus** goes into the left channel of the studio tape machine, while the right channel gets fed **from** the hybrid output. The result is a two-track tape with the announcer and caller audio separated. When you play back on the air, you set the console input to mono and adjust the relative balance as desired. You also have a tape which is easier to do production from for contest squeals, etc.

2.1.2 INPUT AUDIO CONNECTION

The input has the following characteristics:

- Active balanced.
- Approximately 2KΩ impedance.
- Pin 1 is ground and pins **2** & **3** are the balanced audio inputs.

Unbalanced sources **may** be used by connecting pins **1** & **2** to the source ground while the signal hot is connected to pin **3.**

There **are** internal jumpers on the hybrid board to select line or mic level for the inputs. ***On*** HDR3 jumper pins **1** & **2** for line level; jumper pins **2** & **3** for mic level. Be sure to move BOTH jumpers when **changing** this function.

MIC LEVEL LINE LEVEL

- With **jumpers set** to LINE, input level is **-24** to **+12** dBv
- With **jumpers set** to **MIC,** input level is -68 to **-35** dBv

2.1.3 OUTPUT AUDIO CONNECTIONS

There are two separate and independent active differential audio outputs each with the following characteristics:

- Active balanced.

- Output level will vary from approximately -20 dBm to +10 dBm depending upon gain control adjustment, caller level and whether or not the AGC is engaged.
- **Pin 1 is ground. Pins** 2 and **3 are** the balanced signal **outputs.**

If an unbalanced output is required, connect between ground **and** either of the hot pins. Do not ground the unused hot pin.

Note that the output level meter is before the gain control.

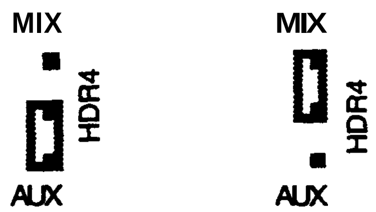
Main Output

Caller audio appears on each of the MAIN outputs.

AUX/MIX Output

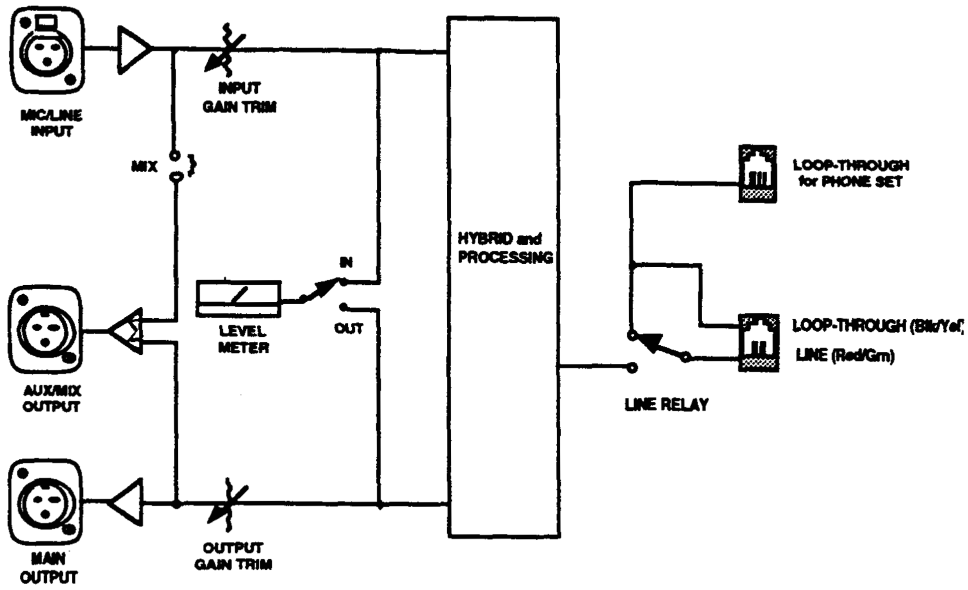
The AUX/MIX output **is** either an extra isolated AUXilliary output or a MIX of the send and caller signals.

Internal jumpers on the hybrid circuit board, on HDR4, select which of the **modes** is to be enabled. Jumper **pins 1** & 2 **to** get the AUX **function,** pins *2* & **3** for the **MIX function.**



AUX FUNCTION MIX **FUNCTION**

When in **the MIX** mode, the input is passed to the ***mix*** output at **unity** gain. Note from the block diagram below that the pass-through gain is ***not*** affected by the input or output **gain** controls. However, since the output gain control ***does*** affect the caller level in the **mixed** output, it *can* be **used** to adjust the balance between the send and caller signals.



Audio Signal Flow (Each Hybrid)

2.2 CONNECTION TO THE TELEPHONE LINE

LINE and PHONE Modular Jacks

Phone connections are made via the standard **modular** jacks on the rear panel.

Each LINE jack is to be connected to **an incoming** central office telephone line. The **two** center pins **(red** and green) are the Telco connection, while the two **outer** pins (black and yellow) **are** a loop through connection which pass the phone line **through** when the hybrid is not active.

Each **PHONE** jack provides another loop-through connection which **also passes** the phone line when the hybrid is not active. It is normally used for connection **of a** desk *set* phone.

2.3 REMOTE CONTROL

Female DB15-type connectors on the rear panel provide access to control functions. **This** connector's functionality and pin-out depends upon whether the "Super" Auto-Answer PCB module is installed. (When the **SAA PCB** is installed, the rear panel DB15 is internally connected to it, and **the** on/off commands and other functions are **passed** to the hybrid from the **SAA as** required. When the **SAA** is not installed, **the** rear panel **DB15 goes** directly to the hybrid.

Without the SAA installed, **OFF** and **ON** control requires a momentary closure *to* ground. It is a standard TTL input pulled-up with a **2.2** KΩ resistor. Thus, it may be connected directly to switches or may be driven by **an** open collector or TTL-compatible logic output **as** desired.

If the "Super"Auto-answer board is installed, refer to **its** manual section for details on remote control operation.

REMOTE CONTROL PINOUT/FUNCTION		
	IF PLUGGED TO SUPER AA PCB	IF DIRECT TO HYBRID PCB
1	**AUTO control** input	AUTO button
2	DROP control input	OFF (paralled **from** switch)
3	+5VDC	**+5VDC**
4	D8 DTMF output	n/c
5	D4 DTMF output	n/c
6	**D2** DTMF output	n/c
7	D I DTMF output	n/c
8	SCL	**N.O.** A-lead contact
9	SEIZE control input	**ON** (paralled **from** switch)
10	Digital Ground	Digital Ground
11	**tine Mode** output	A-lead wiper contact
12	Auto **Mode** output	AUTO **LED**
13	DTMF Data **Valid**	n/c
14	n/c	n/c
15	**SDA**	N.C A-lead contact

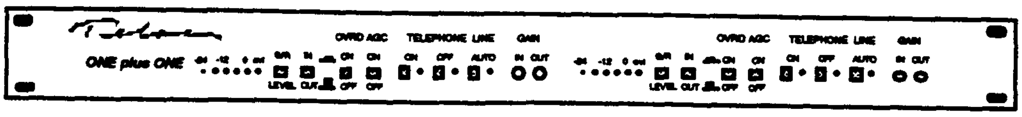
2.4 POWER INPUT

The **ONE** plus ONE **uses** a universal input switching power supply which accepts **AC** input at either **50/60 Hz, from** 90 to 260 volts without any user selection being required.

The line fuse is located on the supply **PCB.**

SECTION 3
OPERATION

3.1 FRONT PANEL CONTROLS



ON/OFF/AUTO Pushbuttons

When the ON button is pressed, the phone line is ***seized*** and the system sends a burst **of** white noise down the line, allowing the hybrid to adapt to the phone line prior to the **start of** conversation. During **this** time, the outputs ***are*** muted.

At the conclusion of the mute/adapt period, the output is enabled and the conversation **may** proceed.

When the **OFF** button is pressed, the phone line is released.

The **AUTO** button is a momentary button that, when a "Super" Auto-answer board is installed, toggles the unit in and out **of** the auto answer/release mode. Its accompanying **LED** indicator illuminates when in **auto** mode. (If the "Super" Auto-answer board is not installed, this button and **LED** is brought to the rear panel remote connector and may be used for any desired external **function.**)

Output Processing: Override

This pushbutton engages two independent **functions:** the ***culler ducking*** and the ***acoustic ducking*** function. In most broadcast applications, you will find that engaging the override function results in better overall performance.

Caller Ducking

This function operates in the caller audio path. When active, there is approximately ***6*** dB **of** ducking applied to the caller audio when **the** announcer speaks. However, very little change in caller level will be noticed due to the very high speed operation **of** the duck function and the masking provided by the presence **of** the *send* audio.

The purpose of this function is twofold

- ***Aesthetic Preference.*** Many air talents prefer the effect of having some control over the caller when they speak.

- *Improvement of "dynamic" Trans-hybrid Loss.*

Acoustic Ducking

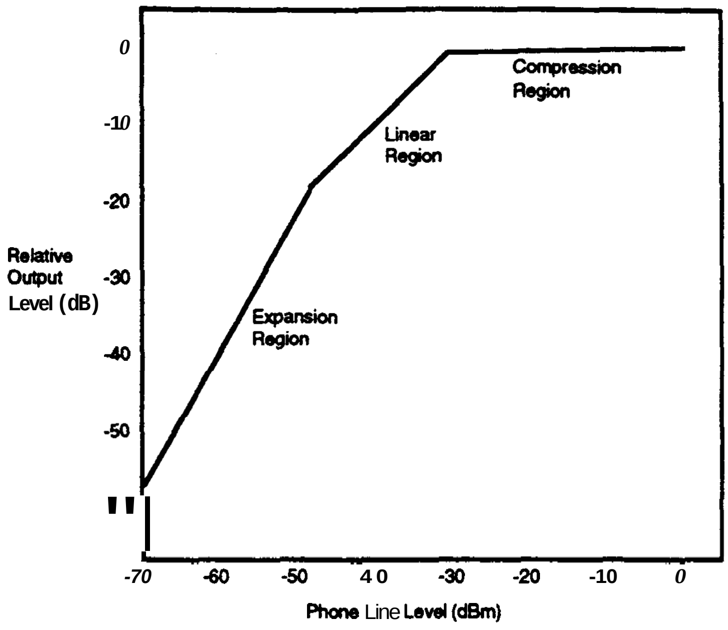
This function operates in the send audio path. The acoustic ducker works by reducing the send (announcer) signal dynamically when the caller speaks.

The acoustic ducker is a used to prevent feedback when open speakers are being used to monitor callers. *Also,* since an open monitor causes callers to hear themselves fed back via the acoustic path from the speaker to the announce mic, this function has the additional benefit of allowing the caller to perceive a more natural sounding conversation.

The time constant is very fast, so the effect of the acoustic ducking usually is not be noticed by the caller. The only disadvantage might be that the announcer may be less able to break in on a caller who is insisting upon shouting on.

Output Processing: AGC

The telephone dynamic gain processing is enabled with this pushbutton. The output processing includes both an automatic gain control and a smart noise gate/downward expander function.



As can be seen in the graph **which** shows the relationship between telco line level and the Telos One output level, the AGC **maintains** a constant output level when the phone **line** level vanes between approximately -30 and *0* dBm. Below approximately -48 dBm, a gentle **noise** gate/downward expander operates to reduce residual hybrid leakage and phone line noise.

• According to AT&T statistical studies **of** the USA telephone network, the average level found on standard phone lines **is -16** dBm. *In our experience high-volume "choke network" lines generally have poorer levels.*

3.2 METERING AND LEVEL ADJUSTMENT

The meter mode pushbuttons select the desired **function:**

- Input level
- Input gain **reduction**
- Output level
- **Output** gain **reduction**

• The input level metering is after the input gain control. After selecting **MIC** or **LINE** level, adjust the trimmer marked "IN" to **get** a *"0"* reading **on** the input level meter (G/R **-LEVEL** button out, IN-OUT button in) while audio is present on the **INPUT.** **Turning this** trimmer up much past the "0" LED *will not* increase the send level to the phone line due to the digital AGC! Overdriving the input may result in distortion and very **poor** hybrid performance!

• The output level meter is placed *before* the output gain control. This is *so* that the level may be adjusted to equipment downstream The Telos' output level meter (G/R **-LEVEL** button out, IN-OUT button out) shows caller level. Adjust the trimmer marked **"OUT"** on the hybrid for a normal (0 VU) reading on your equipment while audio is present on the phone line. Dial tone is OK for **this,** but keep in mind that dial tone level is about *6* dB hotter than voice.

Note: When the AGC button is switched out, the meter reads the telco line level and thus may be used to determine if there is a phone line level problem. The meter's 0dB indication corresponds to a phone line level of approximately **-15** dBm.

3.3 FEEDBACK CONTROL

Sometimes, even with the exceptional trans-hybrid **loss** produced by the digital process, trouble with feedback may occur when the system is being used with an open speaker. 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. Enable the Telos **OVERRIDE** function.

2. When mic processing is being used, connect the hybrid input in such a way that it gets the ***unprocessed*** mic signal. The problem here is that the mic processing combines with the internal Telos input AGC to increase **gain** in the feedback path when no announcer audio is present. Depending on the mic processor, the feedback **margin** could be reduced by many dB. The Telos internal AGC **has** a smart adaptive gate to prevent inappropriate **gain** increase, but it is thwarted by additional processing. If it is not possible to wire around the processing, **try** to ***set*** the mic processing gate function so that the **gain** is not "sucked-up" during pauses. You might **also** try reducing the input level to the hybrid. The send level will still be OK, since the AGC **has** considerable range, but the system won't

have as much room to reach for gain. In some difficult cases, it may be desirable to disengage the Telos output AGC function since it could reduce feedback margin in the same way input AGC does.

3. Try repositioning the mics and/or speaker. Of course, it also helps to use mics and speakers that are directional. In typical broadcast studio application. EV RE-20s and Shure SM-7's have proven appropriate.

4. 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.

5. If necessary, soften acoustic reflections in your studio by adding curtains or wall treatment.

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 mic(s). Depending on the relative levels, phone audio quality may suffer.

Generally, the best scheme is to have the phone monitor speaker mute when the mic is turned on. If you have the announce mics active to the phone system input even when the mic channel is switched off, the system can still be used like a speakerphone when taking calls off the air. When the call is to be used on air, the announce mike is on, so the speaker is muted.

SECTION 4

TECHNICAL DATA and TROUBLESHOOTING

4.1 OVERVIEW

Philosophy

In the past few years, the nature of broadcast engineering has changed considerably. At many stations, the engineering staff has been reduced in size and new responsibilities have been added. At the same time, equipment has gotten more complicated and specialized. Thus, many practitioners of the broadcast electronic arts are forced to become **systems** engineers, emphasizing equipment application rather than component-level trouble-shooting.

This is probably a positive development since it really would be impossible for a station engineer to fully understand the internal nuances of all the wonderful new high-tech stuff that is now available to improve station operations! Also, as equipment becomes more sophisticated and specialized, stocking spare parts for every eventuality has become difficult.

Thus, we don't really expect that much component-level troubleshooting will occur. So, to support you when you need help, we keep spare boards available for fast overnight shipping. In most cases, we will swap boards with you at no cost. In the five years since we introduced the Telos 10, we have yet to charge for a routine repair.

However, despite the comments above, we do provide full schematics and component level troubleshooting information in case you have the need or desire to tackle a repair (or modification) yourself. Another reason we provide the information is to satisfy your curiosity. If you are like me, you probably just *have* to know what's happenin' inside the fancy box. So we tell you.

General Troubleshooting Information

CAUTION

The installation and servicing instructions in this manual are for use by qualiied personnel only. To avoid electric shock do not perform any servicing other than that contained in the Operating Instructions unless you are qualified to do so. Refer all servicing to qualified service personnel.

Access to the **PC** *Boards*

1) The top cover is removed with a **1/16"** allen key. This **will** expose all the circuit boards. Refer to the chart to identify the various PC boards.

2) The power supply and **AC** receptacle are covered by a **small** piece of clear plexiglass. **Do** not remove **this** protective cover or place **your** fingers **under** it **unless** the ac plug is disconnected **from** the unit! When the cover **is** removed the power **supply has** exposed live **surfaces!**

If the hybrid PC board is to be removed, continue **as** follows:

3) Unscrew the four Phillips screws located near ***each*** comer of the board. **If** a "Super" Auto-answer board is installed, unscrew its mounting hardware and gently lift it off the headers. **A** mounting **bushing** must be removed, too.

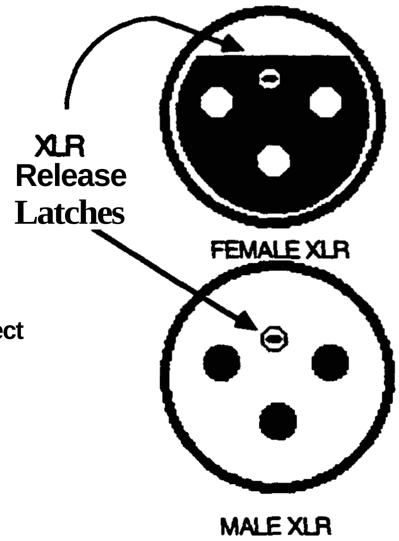
If the rear panel is to be removed:

4) Unscrew the **DB15** connector retaining stand-offs.

5) The **six** XLR **connectors** have retaining screws **which** have to be turned in order to be released. Do **this as** illustrated below.

XLR CONNECTOR RELEASE:

Insert a small screwdriver into the holes in the **connectors, shown at right. Turn the screwdriver** *about* **one eighth of a** turn **counter clockwise to release the connectors. A small screwdriver such as the Xcelite R3322 or R3324 may** need **to be filed down some to** fit **the slots. Remember to retighten the XLR latches when replacing** the **Telos ONE PCB. This will ensure correct support for the XLR connectors on the PCB.**



Desoldering

While we socket the **ICs** that have the greatest potential for failure, most of the Telos One **ICs** are soldered in. That's because most of the time the socket is more likely to cause trouble than the IC. This is of no consolation when one of the soldered **ICs** appears to have failed. When you need to replace a soldered-in chip, the right tool is essential. We use a vacuum desoldering system made by Pace (the **MBT-100)** and **highly** recommend it. Cost is about $450 - worth it if you do much PC board troubleshooting work. The only other real alternative is to clip the leads from the top and remove the solder from the holes with solder-wick. We've not had much luck with the non-heated, manual vacuum desoldering devices such as the one sold by Radio Shack We do not recommend that newly soldered connections be defluxed.

Digital Signal Processing

Because the Telos ***One*** hybrid makes use of digital signal processing for functions traditionally done in analog, the hardware design of the hybrid is relatively uncomplicated and straightforward. In many ways, the hardware is a "textbook" implementation of a general-purpose processing system.

As in any DSP system, the input **signals** are passed through anti-aliasing low-pass filters to remove signal components above the Nyquist frequency. In this *case*, the Nyquist frequency is **4 kHz** and the ultimate sampling rate is ***8*** kHz.

After **A/D** conversion, the signals are presented to the TMS320C25 DSP processor, where software performs the hybrid and processing **functions.**

Then, the signals are converted back to analog and filtered to "re-construct" the desired analog audio.

Notation

Whenever a slash (/) is used after a signal designation in the text or on the schematics, an active low is signified.

4.2 DIGITAL SECTION

4.2.1 THEORY OF OPERATION

The Processor and Bus

(Refer to the Processor & 1/0 Logic Schematic in the DRAWINGS section of this manual.)

The **TMS320C25** is a specialized high-speed processor intended for signal processing applications. Despite its unique properties, it operates much like any other microcomputer from a hardware standpoint.

Program store is provided by the two high-speed EPROMS (**U2** & **U3**). These connect to the bus and are selected directly by the **320C25's assertion** of STRB/, PS/(Program Select/), and R/(W/).

U6, an AC138, provides the chip select signals decoded in the usual microprocessor way by expanding the lower address **lines.**

CS7/ is used to trigger **a** watchdog timer, U7, at regular intervals. **If** processor operation should fail, the watchdog reacts by asserting RES/ to the processor, thus restarting it. The watchdog also provides a reliable reset when the +5 V power supply drops below **4.5** V.

U5 is an output port used for the meter as well **as** for the control input to the CODECS **U12** and **13.**

U4 is an input **port** which is used to communicate button **status** to the processor.

The CODECS have logic outputs. One **of** these is used to operate the line and "**A**" relays as well as the ON and OFF LEDS. **Q1** provides current drive to the relay, while the appropriate **U11** sections drive the LEDS.

CODEC Interface

The CODECS are interfaced serially to the **320C25** through its on-board serial port. Each CODEC is programmed to occupy a time slot on the serial bus.

Clock and Timing

The timing chain starts with a 40 MHz clock oscillator module. The 40 MHz output is fed directly to the 320C25. The HC390, U9 divides the 40 MHz to 2 MHz in order to generate the CLKR and CLKX signals for U1 and the CODECS. This is the data clock.

The HC393 further divides the signal to 8 kHz in order to generate the Frame-Sync input to the DSP and CODECS.

4.2.2 TROUBLESHOOTING THE DIGITAL SECTION

Check the power supply.

Check the 40 MHz oscillator output and the divided-down clock signals to see that they are OK.

Unlike TTL, the CMOS logic ICs used in the Telos One Hybrid should have a nice almost rail-to-rail output.

Make sure that all of the required signals are getting to the CODECS and that the CODECS are outputting data.

None of the logic section ICs should get hot, so if any is, you've found the problem. On rare occasions, a CMOS chip may latch up and get hot, but recover and work normally when power is removed for awhile and restored.

Check the 16 data bus lines to see if any are shorted. They should all exhibit lots of activity, as should the lower address bits.

4.3 AUDIO SECTION

4.3.1 THEORY OF OPERATION

(Refer to the Block Diagram and the audio section schematic in the DRAWINGS section of this manual.)

The audio section is simple and straightforward. Everything should be self-evident from the schematic.

U15, a 5532 op-amp, provides an active hybrid function. One section drives the phone line while the other is configured as a differential amplifier in order to subtract some of the send audio before the digital process completes the job. RF3 is a special pi-filter network used to remove RF interference.

The audio input section uses one half of U16, another 5532 op-amp. This is a standard active differential configuration. **SW8** changes gain for either mic or line level inputs levels. RF1 and RF2 provide RFI filtering.

The other half of U16 is used in a circuit which provides the output gain control and a single pole of low-pass filtering for de-emphasis. (The pre-emphasis is a digital function provided within CODEC U13).

U17, another 5532 op-amp, provides the balanced output for the AUX/MIX output port. One section operates as a summing amplifier; one of the summer inputs is always connected to the hybrid output while the other is switched by SW9 to either ground or the input audio signal.

U14 is a special purpose IC which has two low-pass filter sections as well as a balanced 600Ω line driver. The U14 line driver section provides the main audio output.

4.3.2 TROUBLESHOOTING THE AUDIO SECTION

Using a scope for signal tracing should do the trick. All chips should run cool except for non-Signetics brand 5532s - *it is normal for these to run hot.*

4.4 POWER SUPPLY

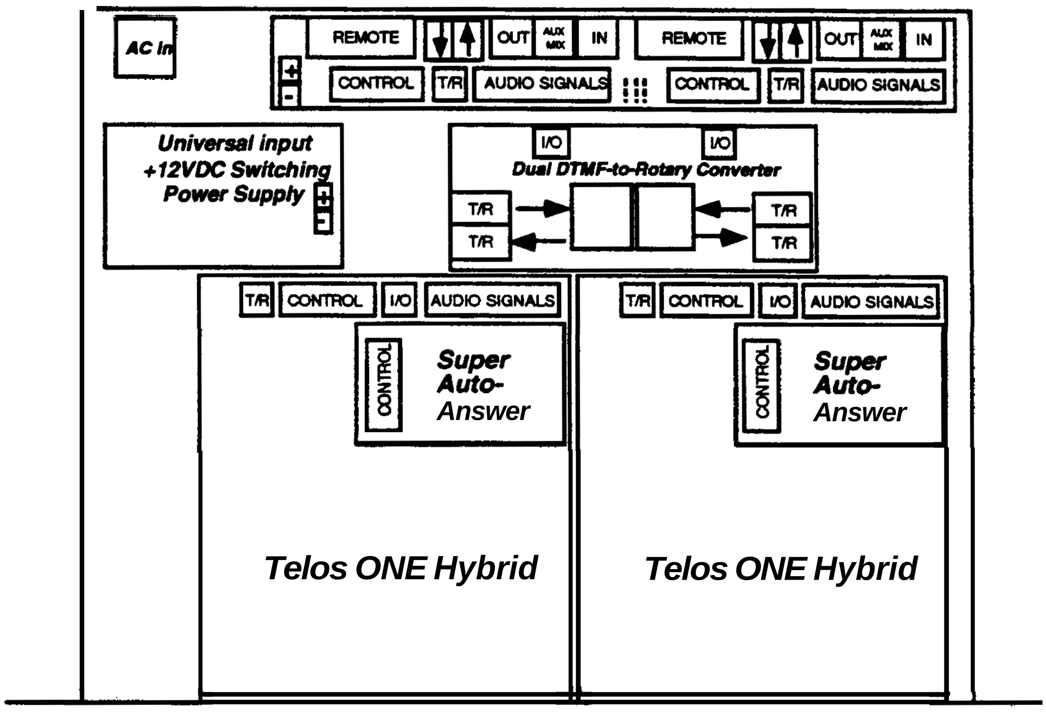
The Telos ONE plus ONE uses a 40W +12VDC universal input power supply. Refer to the appendix for manufacturer's data sheet. Each circuit board that requires power locally regulates the +12VDC down to the required voltage. Each board has its own regulators:

+5v DIGITAL Powers the processor and other digital ICs. Regulated by three-terminal regulator VR3. A *gold cup,* C3, holds up the power voltage in the event of a brief line voltage dip.

+5v ANALOG Powers all the op-amps and the CODECS. Regulated by VR2.

-5v ANALOG: Powers op-amps **and** the CODECS. **Uses** U18, **a** switching inverter, in order to generate a negative voltage which is then regulated **by** VR1.

4.5 INTERNAL CONNECTIONS



Telos ONE plus ONE circuit board and connector identification diagram

T/R

The jacks marked "T/R" are the "tip and **ring" of** the telephone line. **A** four-conductor cable routes the **main** telco **and** loop-through **signals** from the rear panel connector board **to** the hybrid. **If** the DTMF-to-Rotary converter board **is used,** the connector board cable **goes to** the rearmost RJ11 jack on the converter board; the other RJ11 jack then **goes** to the hybrid. That is, the converter board **goes** in series with the phone line.

CONTROL

The connectors marked **CONTROL** route various control signals from the hybrid to the rear panel 15-pin D-sub connector. This cable runs from the connector board to the hybrid. If a "Super" auto-answer card is being used, the **CONTROL** cable on the connector board is plugged to the "Super" auto-answer board's connector.

I/O

This connector routes send audio to the DTMF-to-Rotary converter board (for DTMF detection), the ON pin of the auto-answer card (to mute the hybrid during rotary dialing), +12VDC and ground. Not used if the converter board is absent.

AUDIO SIGNALS

Route the audio input, output, aux/mix, mix-minus signals, +12VDC and ground **from** the connector **boards XLRs** and power supply connector.

+/-

This is the power supply connector. +12VDC and ground are brought to the rear panel connector board, which, in **turn,** distributes these signals to both hybrids via their respective AUDIO SIGNALS connector.

J2 to P3,P4 pin #'s	REMOTE CONTROL PINOUT/FUNCTION		
		IF PLUGGED TO SUPER AA PCB	IF DIRECT TO HYBRID PCB
2	1	AUTO control input	AUTO button
4	2	DROP control input	OFF (paralled from switch)
6	**3**	+5VDC	+5VDC
8	4	D8 DTMF output	n/c
10	5	D4 DTMF output	n/c
12	**6**	D2 DTMF output	n/c
14	**7**	D I DTMF output	n/c
16	8	SCL	n/c
3	**9**	SEIZE control input	ON (paralled from switch)
5	10	Digital Ground	Digital Ground
7	11	Line **Mode** output	n/c
9	12	Auto **Mode** output	AUTO LED
11	13	DTMF Data Valid	n/c
13	14	n/c	n/c
15	**15**	SDA	n/c

AUDIO SIGNALS (J1 to P1.P2)

	'AUTO" button	11	aux/mix XLR pin **3**
	'AUTO' LED indicator	12	aux/mix XLR pin 2
	+12VDC	13	input XLR pin **3**
	+12VDC	14	aux/mix XLR pin 1
	GND	15	input XLR pin 1
	GND	16	input XLR pin 2
	output XLR pin 2	17	extra input from JP1
	GND	**18**	input to 'IN LVL" pot
	output XLR pin 1	19	**extra output to JP1**
10	output XLR pin **3**	20	**output of U16 pin 7**

***I/O* SIGNALS to decoder board (J4 to J3,J4)**

1	P3 pin **3** 'ON' signal from Super Auto answer **board**
2	P1 pin 18 INput audio for DTMF converter chip
3	GND
4	GND
5	+12VDC
6	+12VDC

Telos ONE plus ONE
Connector Cross-Connect and Function Directory

Telos One **plus** *One*

HYBRID PARTS LIST

Designation	Description	Designation	Description
R1	100R	R33	1OKR
R2	100R	R34	39KR
R3	1OKR	R35	270KΩ
R4	1KR	C1	2.2µF/25 tant
R5	560Ω	c2	2.2µF/25 tant
R6	1OKR	C3	0.1 Farad Gold Cap
R7	1OKR	C4	22pF mono
R8	1OKR	C5	2000pF mono
R9	100R	C6	2000pF mono
R10	33KR	C7	.01µF mono
R11	1OKR	C8	2.2µF/25 tant
R12	1OKR	C9	2.2µF/25 tant
R13	1KΩ/1Yo	C10	2.2µF/25 tant
R14	1KR/1Yo	C11	100µF/35 electrolytic
R15	1KR/1Yo	c12	1000µF/35 electrolytic
R16	1KR/1%	C13	10µF/25 tant
R17	100KΩ/1%	C14	0.1µF mono
R18	100KΩ/1Yo	C15	2.2µF/25 tant
R19	1KR	C16	2.2µF/25 tant
R20	2.4KΩ	C17	2.2µF/25 tant
R21	1OKR	C18	01µF mono
R22	49.9Ω/1%	C19	0.1µF mono
R23	49.9Ω/1%	c20	0.1µF mono
R24	1OKR	c 21	0.1µF mono
R25	56KR	D I	1N4730 3.9v zener
R26	1KR	D2	1N4730 3.9v zener
R27	15KR	D3	1N5818
R28	49.9Ω/1%	D4	1N4004
R29	49.9Ω/1%	IND1	100uH toroid
R30	1OKR	Q1	2N2222
R31	1OKR	RP1	2.2KΩ SIP
R32	1OKR	RP2	330Ω SIP

HYBRID PARTS LIST continued...

Designation	Description
U1	TI 320C25
U2,3	27C292-3JL EPROM
u4	74AC244
u5	74HCT374
U6	74ACT138
u7	DS1232 watchdog/reset
U8	74HC590
U9	74HC390
U10	Saronix NCH080C-40Mhz osc
U11	74HCT04
U12,13	AMD7901CPC CODEC
U14	2912A
U15,16,17	5532 op-amp
U18	MAX636 switching inverter
T1	Prem #SPT-195 xfmr
K1,2	Omron 5v DIP relay

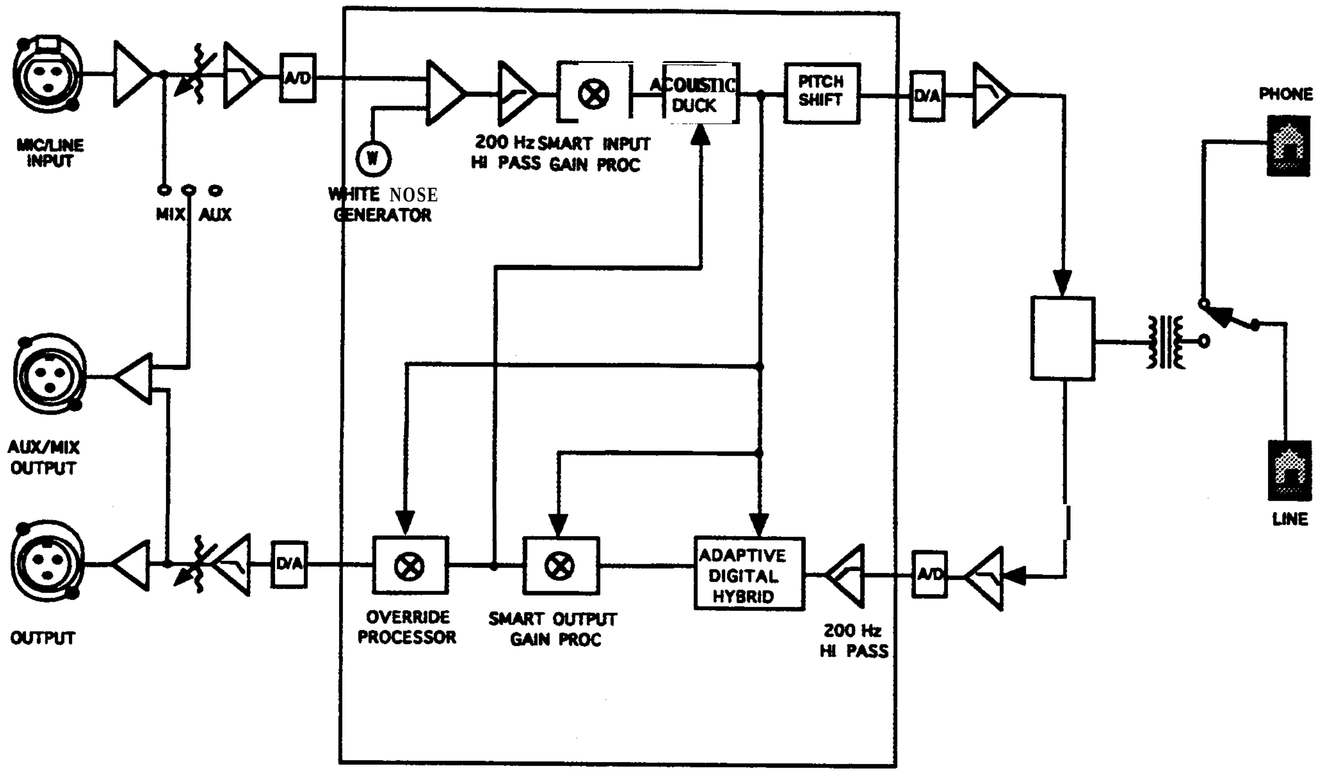
Designation	Description
J1	Neutrik NC3FDHBAG XLR f
J2,3	Neutrik NC3MDHBAG XLR
J4	DB15 female
J3	modular phone receptacle
VR1	7905 -5v reg
VR2,3	7805 +5v reg
Z1	150v ZNR
P1,2	10K rt angle multiturn tri
SW1,2,3,4	Alt action switch
SW5,6,8	Mom switch
sw7	Two pos DIP sw
RF1,2,3	TDK Z7K51R1-05 RF filte
L1,2,3,4,5,8	Green LED LTL-1234A
L6,7	Red LED LTL-1224A
L9	Yellow LED LTL-1254A

SECTION 5

DRAWINGS

DRAWINGS:

Hybrid Signal Flow Block Diagram
TMS320C25 Pinouts
Hybrid Digital Circuits Schematic
Connector Board Schematic
Hybrid Input and Output Audio Schematic
Hybrid Phone and Control Schematic



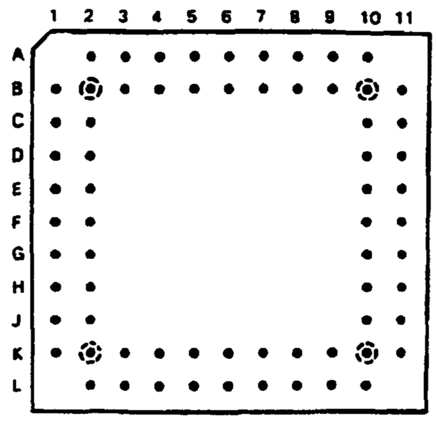
Digital Processing Section

Telos ONE PLUS ONE
Signal Flow Block Diagram

PIN ASSIGNMENTS

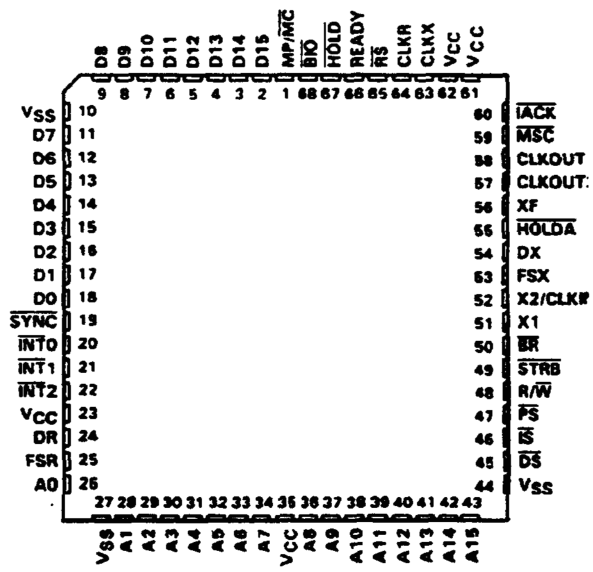
PIN	FUNCTION	PIN	FUNCTION	PIN	FUNCTION
A2	D8	C11	CLKOUT1	J10	$\overline{PS}$
A3	D10	D1	D4	J11	$\overline{IS}$
A4	D12	D2	D3	K1	A0
A5	D14	D10	CLKOUT2	K2	A1
A6	MP/$\overline{MC}$	D11	XF	K3	A3
A7	$\overline{HOLD}$	E1	D2	K4	A5
A8	$\overline{RS}$	E2	D1	K5	A7
A9	CLKX	E10	$\overline{HOLDA}$	K6	A8
A10	V_{CC}	E11	DX	K7	A10
B1	V_{SS}	F1	D0	K8	A12
B2	D7	F2	$\overline{SYNC}$	K9	A14
B3	D9	F10	FSX	K10	$\overline{DS}$
B4	D11	F11	X2/CLKIN	K11	V_{SS}
B5	D13	G1	$\overline{INT0}$	L2	V_{SS}
B6	D15	G2	$\overline{INT1}$	L3	A2
B7	$\overline{BIO}$	G10	X1	L4	A4
B8	READY	G11	$\overline{BR}$	L5	A6
B9	CLKR	H1	$\overline{INT2}$	L6	V_{CC}
B10	V_{CC}	H2	V_{CC}	L7	A9
B11	$\overline{IACK}$	H10	$\overline{STRB}$	L8	A11
C1	D6	H11	R/$\overline{W}$	L9	A13
C2	D5	J1	DR	L10	A15
C10	$\overline{MSC}$	J2	FSR		

68-PIN GB
PIN GRID ARRAY CERAMIC PACKAGE†
(TOP VIEW)

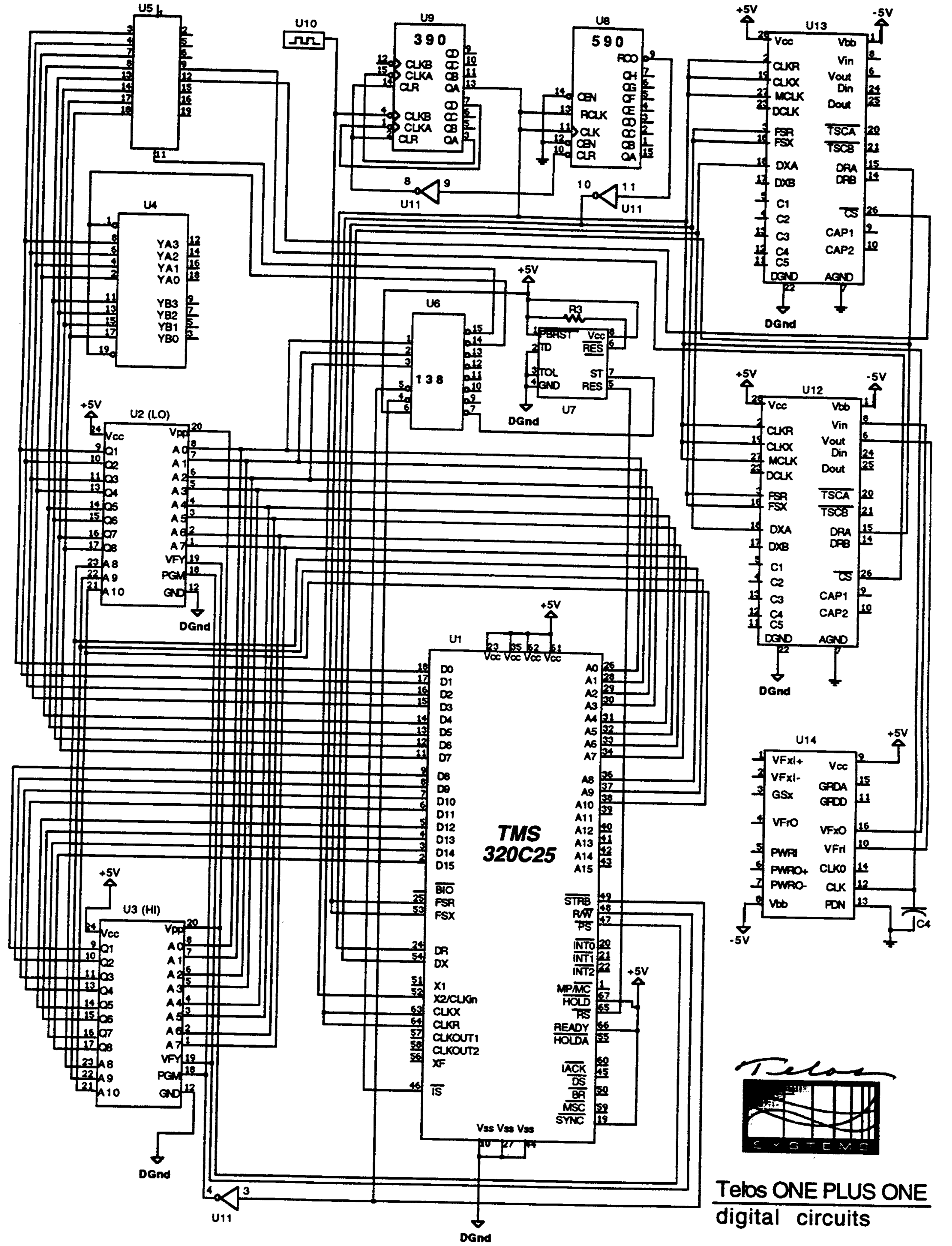


† See Pin Assignments Table (Page 1) and Pin Nomenclature Table (Page 2) for location and description of all pins.

TEXAS INSTRUMENTS

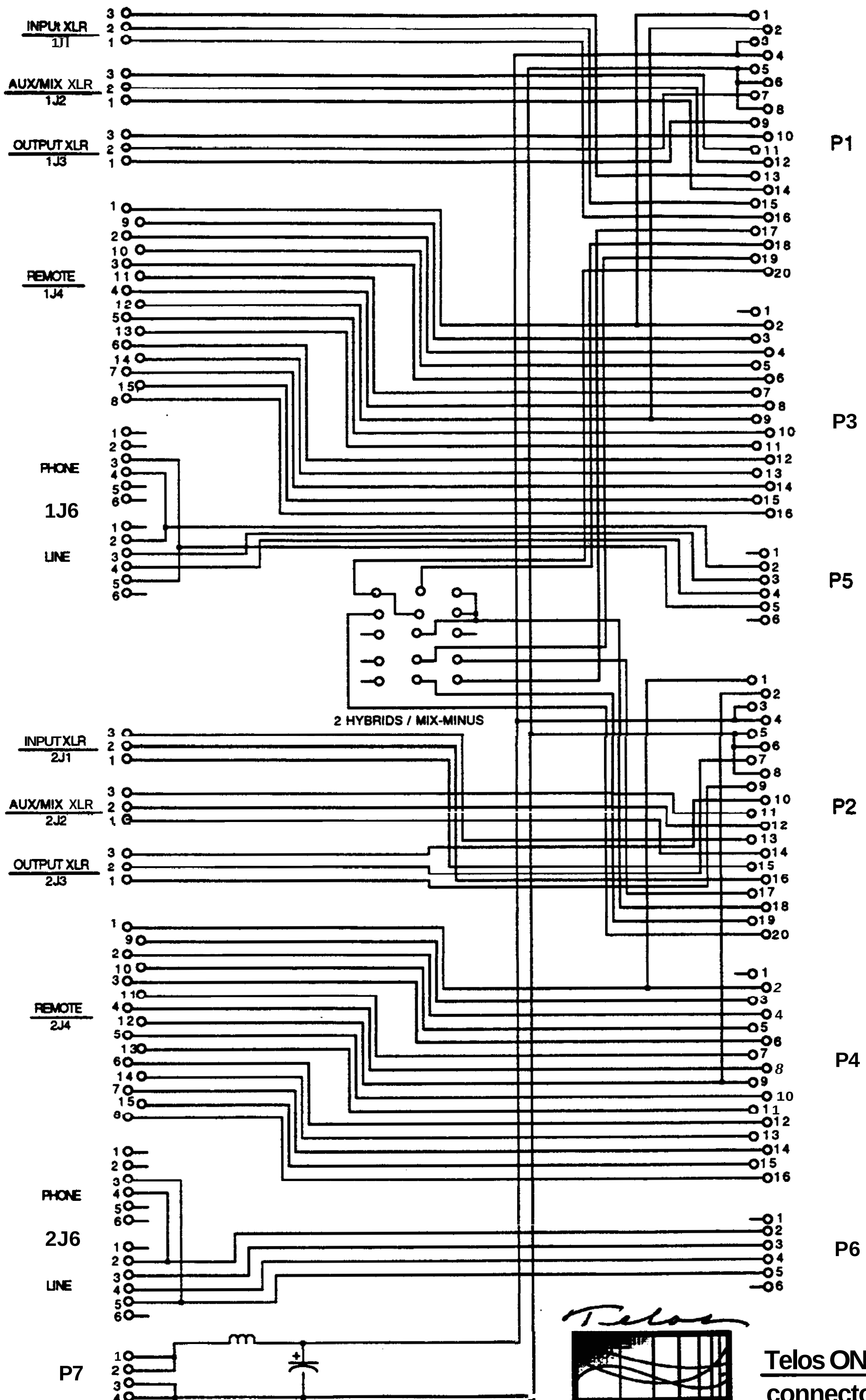


Telos ONE PLUS ONE
TMS320C25 Pinouts

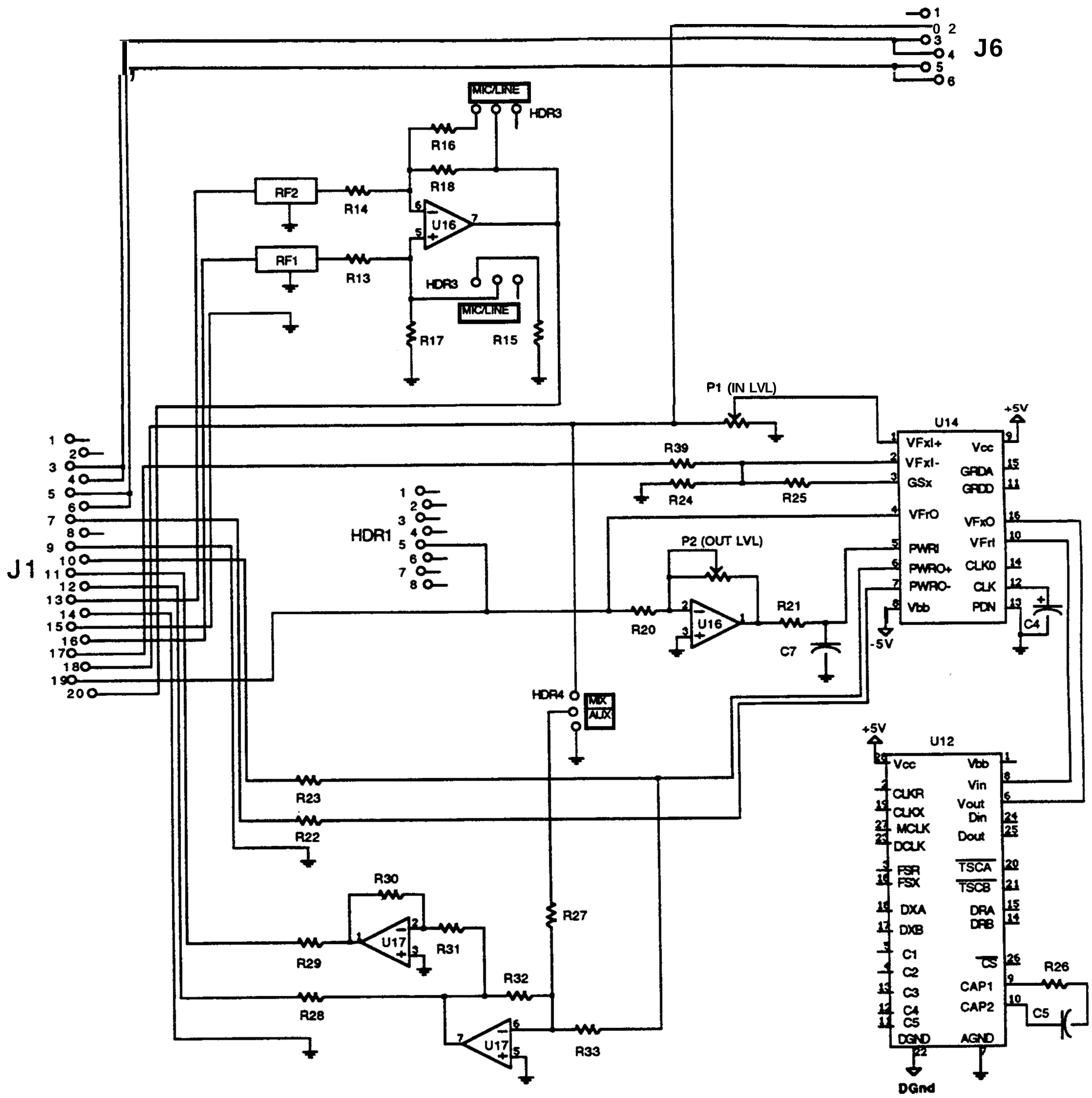


Telos ONE PLUS ONE

digital circuits

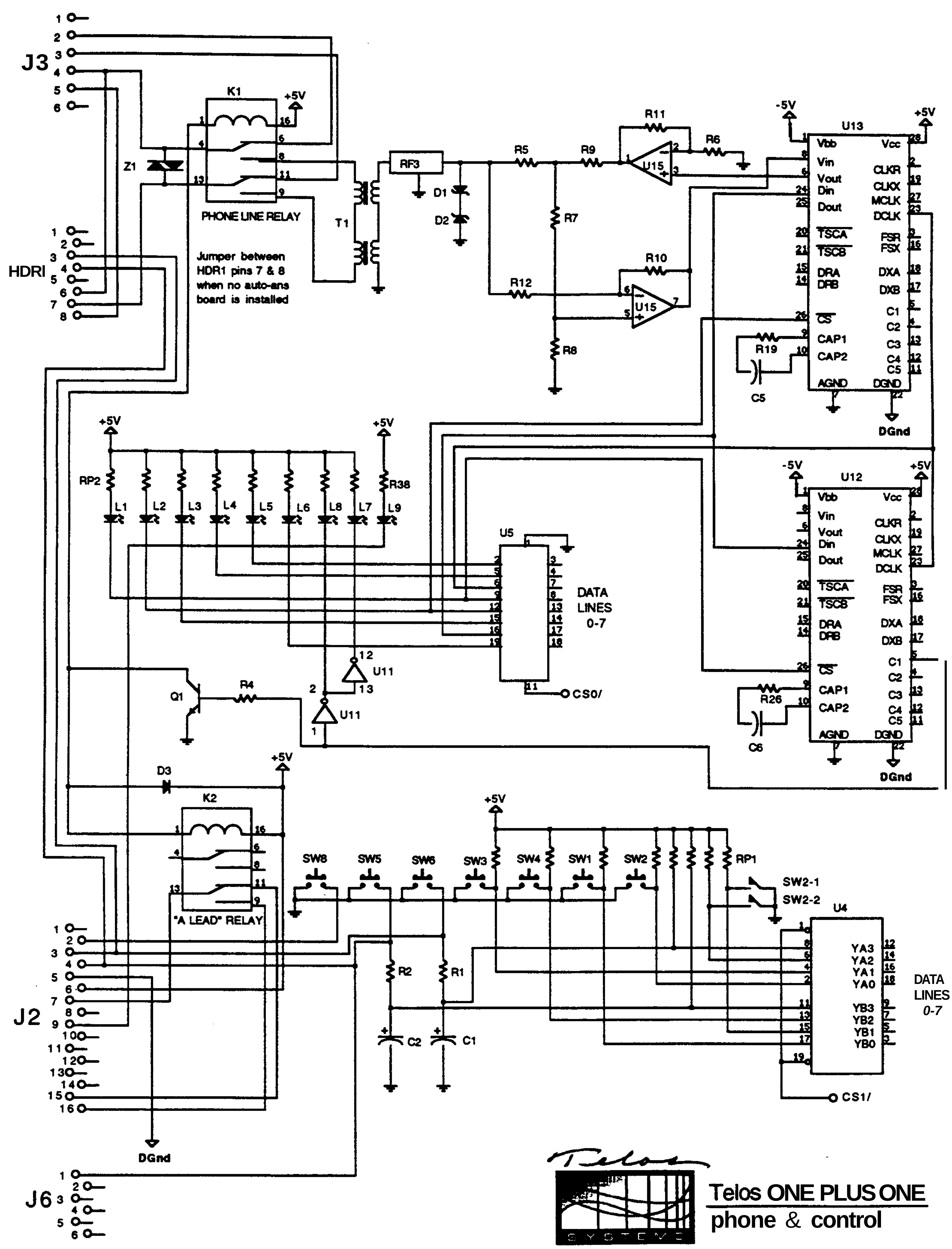


Telos ONE PLUS ONE
connector board



SYSTEMS

Telos ONE PLUS ONE
input & output audio


J3
K1
+5V
Z1
PHONE LINE RELAY
RF3
D1
D2
T1
R5
R9
R11
R6
U15
R7
R10
R12
R8
-5V
U13
Vbb
Vcc
Vin
Vout
Din
Dout
CLKR
CLKX
MCLK
DCLK
TSCA
TSCB
FSR
FSX
DRA
DRB
DXA
DXB
C1
C2
C3
C4
C5
CS
CAP1
CAP2
R19
C5
AGND
DGND
DGnd
HDRI
Jumper between HDR1 pins 7 & 8 when no auto-ans board is installed
RP2
L1
L2
L3
L4
L5
L6
L8
L7
L9
R38
U5
DATA LINES 0-7
CS0/
U11
Q1
R4
U12
R26
C6
D3
K2
"A LEAD" RELAY
SW8
SW5
SW6
SW3
SW4
SW1
SW2
RP1
SW2-1
SW2-2
U4
YA3
YA2
YA1
YA0
YB3
YB2
YB1
YB0
R2
R1
C2
C1
CS1/
J2
J6
Telos ONE PLUS ONE
phone & control

SECTION 6

APPENDIX

TELOS ONE HYBRID: ALTERNATE SOFTWARE VERSIONS

The Telos One is a "true" digital-signal-processing system. That means that most system characteristics are fixed by the software which is contained within two EPROM chips. This provides us with the opportunity to change the hybrid's audio operation to accommodate special or unique circumstances by merely swapping the EPROMs.

A number of special software versions have become more-or-less standard. This are described below. There are, in addition, a few more which have been developed for unusual applications.

If you think you need special software, either the versions listed below or something **really** different, please contact Telos.

Note: If we know or think that your hybrid is to be used for intercom interface applications, we automatically install version 4.X software.

In each case below, there are two components to the version number. The number before the decimal point is the series number and defines the basic characteristic, while the portion after the decimal point changes as improvements or minor changes are implemented.

1.X: Normal Telos ONE operation

This is the software normally shipped with Telos Ones. It implements the operating characteristics described earlier in this manual.

4.0: Bell Atlantic Teleconference Version

Has limiter on *send* audio instead of AGC. Symmetrical ducking with about 14 dB on each path. Best for making multi-line conference bridges.

4.x: "ABC-TV Version" - For Interfacing 4-Wire Intercoms

Same as 4.0 but provides 6 dB more send level. *Also,* ducking modified somewhat - favors receive. Widely used with McCurdy, Clearcom and other 4-wire intercom systems. Permits multiple lines to be conferenced without feedback, etc. Telos has considerable experience in this area. Please call with your special requests and unusual applications.

5.X: The "Dallas" Version.

Developed in response to the very bad phone line conditions which exist in the Dallas/Fort Worth and Miami/Fort Lauderdale areas. For some reason, these parts of the country have exceptionally bad phone service. Levels vary over a 30 - 40 dB range and line impedances are particularly difficult for the hybrid to accommodate. This version adds more ducking in both directions, increases the AGC action, lowers the receive expansion threshold, and alters the adaptive hybrid parameters to effect better trade-offs for bad phone lines.

Outside of Dallas, this software is for people who:

1) Prefer a more of "speakerphone" switching effect.
2) Prefer a more aggressive AGC action on the phone audio.
3) Need to accommodate very poor phone lines with widely varying levels.
4) Have open monitor speakers and are having trouble with feedback

When the phone line impedance is very complex. The Telos ONE's *analog* hybrid then leaks too much during *send* audio, causing the *digital* hybrid to halt adaption. Version 5.x software can tolerate more leakage, due to a change in one of the adaption parameters. Secondly, a "see-saw" ducking effect has been added. That is, the ducking *occurs* on the send path during receive and on the receive path during send. The level of the ducking is about 15 dB, yet it has a very fast recovery time and is biased more toward the receive side. The full-duplex operation is not affected, just the level of the ducking (about 6 dB with standard software). The expansion threshold level has been reduced so lower-level callers don't get expanded down. All this, along with Telos' digital pitch shifter and digital AGC helps the hybrid adapt better to poor phone lines.

6.X: "RTS Version"

This is the preferred version for connecting RTS or other 2-wire intercom system to the Telos ONE phone jack. Stays on continuously. A change in the adapt threshold allows send level to be increased with a resistor for RTS operation.

WARRANTY and Application Caution

The Telos One is warranted to be free from defects in material and workmanship for a period of 365 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 One to a place designated by the seller, transportation prepaid, seller at its option, will either repair or replace the part or the Telos One, anti 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 sole right and remedy. In no event shall the seller or it's 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 One Interface is intended to be used with FCC registered protective interface devices.

AUTEC POWER SYSTEMS
9301-101 Jordan Avenue, Chatsworth, CA 91311
Tel: (818) 341-6123
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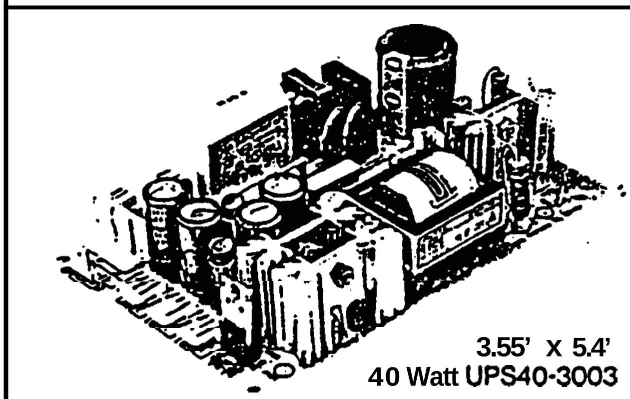


3.55' x 5.4'
40 Watt UPS40-3003

SINGLE OUTPUT MODELS

MODEL NUMBER	PWR	OUTPUT 1	DIMENSIONS (INCHES)	MOUNTING HOLES
UPS40-2051	40	5V @ 8.0A	3.00 x 5.00 x 1.25	4.55" x 2.55"
UPS40-2091	40	9V @ 4.4A	3.00 x 5.00 x 1.25	4.55" x 2.55"
UPS40-2121	40	12V @ 3.5A	3.00 x 5.00 x 1.25	4.55" x 2.55"
UPS40-2151	40	15V @ 2.5A	3.00 x 5.00 x 1.25	4.55" x 2.55"
UPS40-2241	40	24V @ 1.5A	3.00 x 5.00 x 1.25	4.55" x 2.55"
UPS65-1051	65	5V @ 13.0	3.50 x 6.00 x 1.54	4.80" x 3.15"
UPS65-1121	65	12V @ 5.5A	3.50 x 6.00 x 1.54	4.80" x 3.15"
UPS65-1151	65	15V @ 4.5A	3.50 x 6.00 x 1.54	4.80" x 3.15"
UPS65-1241	65	24V @ 2.8A	3.50 x 6.00 x 1.54	4.80" x 3.15"

E133148
UL1950

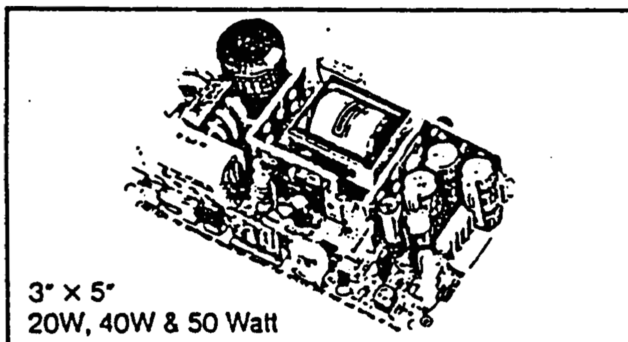
LR89164
1402C
LEVEL 3

TÜV Rheinland
A98290
IEC950

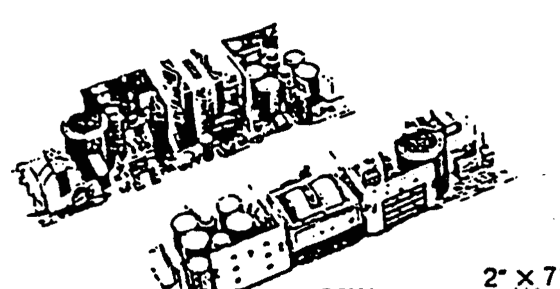
VDE EN60950
APPROVAL
PENDING ON
65 WATT MODELS

20 WATT THRU 50 WATT DUAL & TRIPLE OUTPUT MODELS

MODEL NUMBER	PWR	OUTPUT 1	OUTPUT 2 (Peak)	OUTPUT 3	DIMENSIONS (INCHES)	MOUNTING HOLES
UPS20-2002	20	5V @ 1.0A	12V @ 1.5A (3.0A)		3.00 x 5.00 x 1.20	4.55" x 2.55"
UPS20-5002	20	5V @ 1.0A	12V @ 1.6A (2.0A)		3.00 x 4.00 x 1.20	3.56" x 2.62"
UPS25-1002	25	5V @ 1.0A	12V @ 1.5A (3.0A)		2.00 x 8.00 x 1.00	6.60" x 1.60"
UPS30-4003	30	5V @ 1.5A	12V @ 1.5A (3.0A)	-12V @ 0.3A	5.12 x 2.76 x 1.30	4.72" x 2.36"
UPS30-5002	30	5V @ 2.1A	12V @ 1.8A (3.0A)		3.00 x 4.00 x 1.50	3.56" x 2.62"
UPS40-1002	40	5V @ 3.0A	12V @ 2.0A (4.5A)		2.00 x 7.00 x 1.20	6.60" x 1.60"
UPS40-2002	40	5V @ 3.0A	12V @ 2.0A (4.5A)		3.00 x 5.00 x 1.20	4.55" x 2.55"
UPS40-2003	40	5V @ 3.04	12V @ 2W (4.5A)	-12V @ 0.3A	3.0 x 5.W x 1.20	4.55" x 2.55"
UPS40-3003	40	5V @ 3.0.4	12V @ 2.0A (4.5A)	-12V @ 0.3A	5.40 x 3.55 x 124	4.82" x 3.15"
UPS50-1002	50	5V @ 3.0A	12V @ 3.0A (5.0A)		2.00 x 7.00 x 1.57	6.60" x 1.60"
UPS51-2002	50	5V e 4.w	12V @ 3.0A (5.0A)		3.00 x 5.00 x 1.20	4.55' x 2.55"



3" x 5"
20W, 40W & 50 Watt



2" x 7"
30W, 40W & 50 Watt

Accessory PCB Module

DTMF-TO-ROTARY CONVERTER BOARD

DTMF-TO-ROTARY CONVERTER BOARD

PURPOSE

The DTMF-to-Rotary Dial converter board takes DTMF (dual tone-multifrequency signals, or touchtones®) and converts them into rotary dial (decadic) make/break pulses.

THEORY OF OPERATION

The phone line is passed through this board, which has a loop-current detector and a make/break relay. Another relay is used to take the hybrid off the phone line during pulse dialing.

The tones are fed from the input of the hybrid (send) to a DTMF receiver chip, which, in turn, drives the pulse dialer IC. An LED on the board indicates on-hook, off-hook (OK to dial) and time out. The pulse dialer chip is buffered from the relays with an open-collector output device. Another signal is taken to the hybrid's ON pin to turn the hybrid on during pulse dialing and to hold the hybrid in the mute state. This prevents the "clicks and clunks" of the pulse dialer relay from being audible. The hybrid's telco line is fed into a phantom talk battery during dialing, to keep the "Super" Auto-answer line detection circuitry from releasing the line by turning the hybrid off.

There are two independent decoder circuits per card. Some ICs are shared by the two circuits, however.

When the DTMF-to-Rotary converter board is being used, the tones are received from the hybrid send audio. That means that the send gain trimmer affects this level and the LED meter indicates it. If the tones are too hot (input meter OVL LED is lit during the presence of tones), the input trimmer must be turned down or the tone level reduced externally. If the tones are being clipped, the decoder board may not properly recognize the tones.

Special Note Regarding Phone Lines

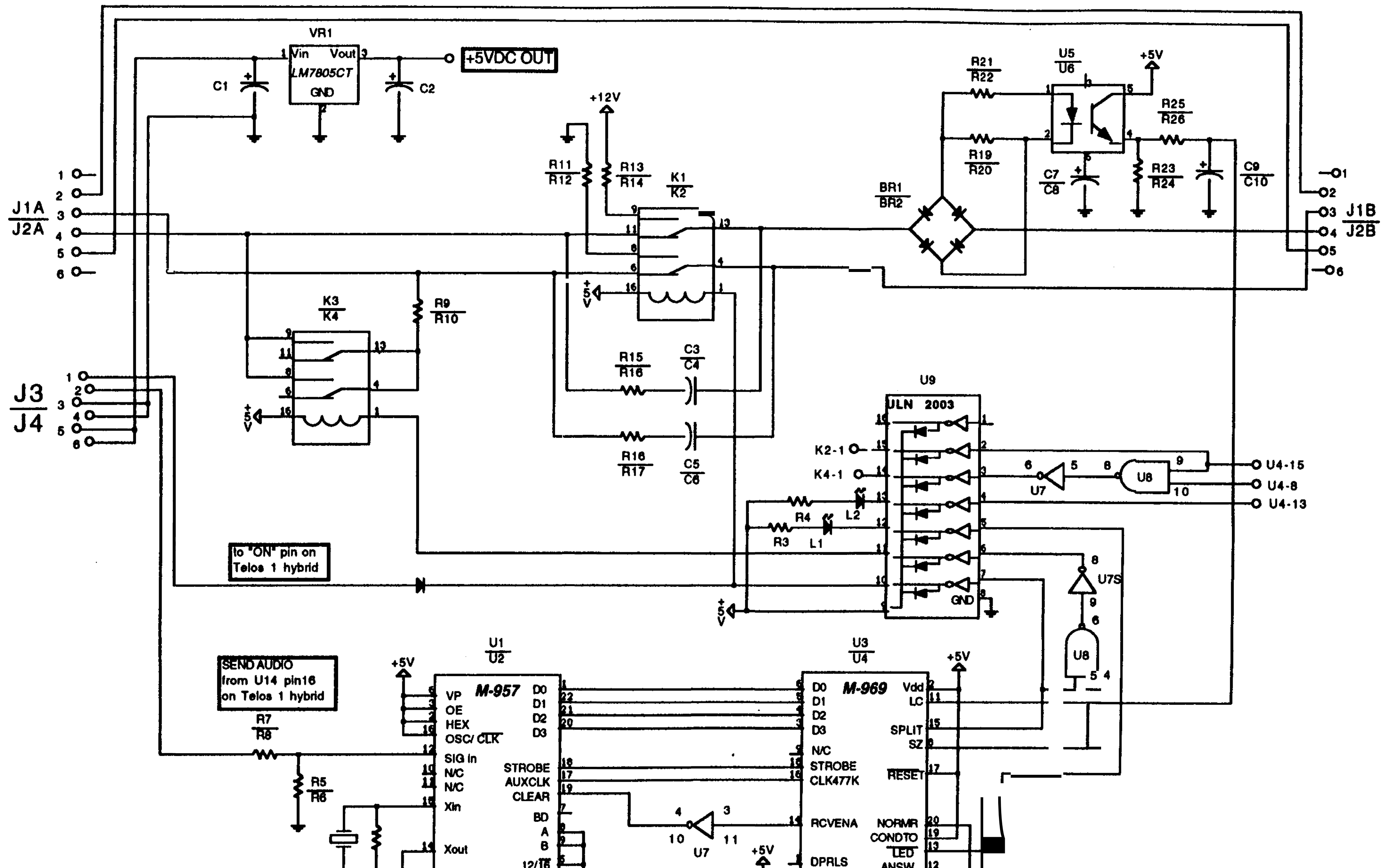
With the DTMF-to-Rotary converter board installed, the unit must be connected to a "pulse dial only" line. If the phone line *can* detect DTMF, the unit will not dial properly, because some of the

tone will make it to the central office equipment *before* the pulse dialer begins. Then the pulse dialer will *repeat* the last digit dialed.

DTMF-to-Rotary Converter **Board**

PARTS LIST

Designation	Description	Designation	Description
U1,2	DTMF decoder (Teltone M-957)	C1,2	2.2µF @ 25V
U3,4	DTMF-to-pulse converter (Teltone M-969)		C3,4,5,6
	270pF @ 1000		
U5,6	Optoisolator 4N33	C7,8	not stuffed
u7	hex inverter 74LS14	C9,10	not stuffed
U8	Quad NAND gate 74LS00	X1,2	3.58MHz crystal
U9	Octal open collector driver ULN2003		K1,2,3,4
	DPDT relay		
R1,2	1MΩ 1/4W 5%	VR1	+5V regulator LM7805
R3,4	330R 1/4W 5%	BR1,2	bridge rectifier
R5,6,7,8	150Ω 1/4W 5%	L1,2	green LED
R9,10	330R 2W 5%	J1,2	dual RJ11 jack
R11,12,13,14	75R 2W 5%	J3,4	6-pin 0.1" connector
R15,16,17,18	1MΩ 1/4W 5%	SU3,4	20-pin socket BU200Z
R19,20	47R 1/4W 5%	SU5,6	6-pin socket BU060Z
R21,22	150R 1/8W 5%	SU7,8	14-pin socket BU140Z
R23,24	910Ω 1/4W 5%	SU9	16-pin socket BU160Z
R25,26	220R 1/4W 5%		



NOTES:
Two identical circuits per card;
Top component numbers for hybrid #1;
bottom component numbers for hybrid#2.
Component values on parts list.
Component numbers screened on PCB.
VR1, U7, U8 & U9 shared by both circuits.

Telos ONE PLUS ONE
DTMF-to-rotary
converter board

Accessory PCB Module

BASIC AUTO-ANSWER BOARD

BASIC AUTO-ANSWER BOARD

Purpose

The auto-answer board is used when automatic answering and hang-up of the Telos One hybrid is desired. The circuit **turns** the hybrid on in response to ringing voltage on the connected phone line, and **turns** the hybrid off when a break in loop current is detected. The loop current interruption, often referred to **as** *CPC,* or Calling Party Control, is present on most telephone lines. However, some central office equipment or PBXs may not provide it and other detection methods will have to be used in that case.

Installation

The board is installed by plugging it into the header connectors on the Telos One board. Remove **the small** jumper plugs on HDR1 first.

The auto-answer board should be removed and the jumpers replaced if use with other than Central Ofice-type lines with talk battery is expected, since the diodes in the loop detect section will cause severe audio distortion if not biased by talk battery.

Operation

Not much excitement here. With the board installed, the hybrid will automatically answer and hang-up.

Circuit Description

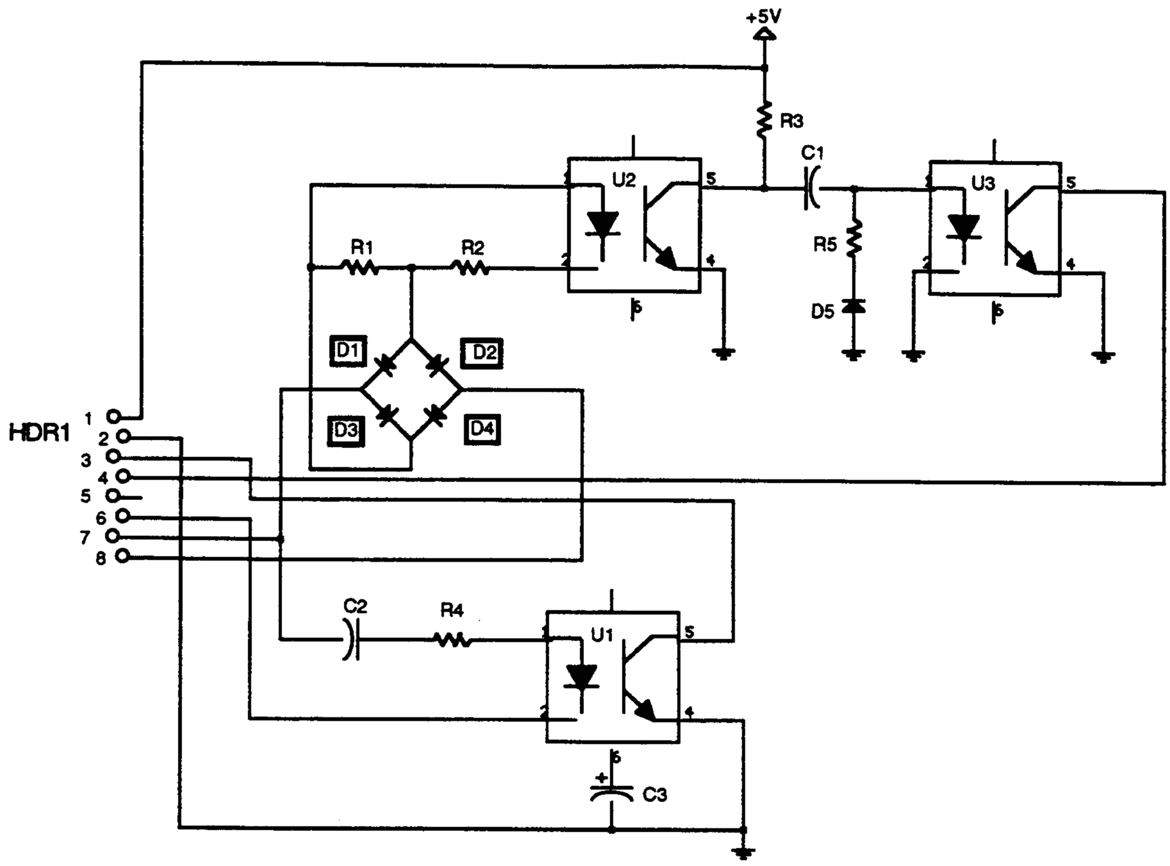
U1, R4, and C2 **form** the ring detect section. C2 blocks DC ***so*** that only the **AC ring** voltage will trigger U1. When Ul's diode gets current, its transistor pulls HR1-2 low **turning** the hybrid on. C3 prevents false tripping.

The remaining components **form** the loop current detector. When loop current is present, U2's transistor conducts **and** C1 is discharged. When loop current **goes** away, C1 **charges** through the diode in **U3** thus causing a low at HR1-1 for the length **of** the R3/C1 time constant. Upon restoration of loop current, the capacitor discharges through R5 and D5. The C1/R5 time constant prevents falsing during initial answer by preventing C1 discharge on brief periods of loop current presence.

Basic Auto-answer Board

PARTS LIST

Designation	Description	Designation	Description
U1,2,3	opto-isolator 4N33	R1	100Ω
D1,2,3,4	1N4004 dioide	R2	220Ω
D5	1N4148 diode	R3	2.2KΩ
C1	6.8µF @ 25V	R4	2.2KΩ
c2	0.47µF @ 200V	R5	4.7MΩ
c3	3.3µF @ 25V		



Telos basic AutoAns
Rev 3.1

Accessory PCB Module

"SUPER" AUTO-ANSWER BOARD

INTRODUCTION

GENERAL DESCRIPTION

The "Super" Auto-Answer board is a plug-in PCB module for the Telos One which provides reliable auto-answer and disconnect as well as other functions:

* Remote control for seize, drop and auto functions
* Status indicators for off-hook condition and auto mode
* Open collector outputs from an integral DTMF decoder
* Selectable answer on first ring or after three rings
* Disconnect on drop or reversal of talk battery (CPC)
* Disconnect on dial tone detection (selectable enable/disable)
* Auto mode enabled by remote button or dip switch position

Since the unit's operation is controlled by a microprocessor, the functionality may be changed by reprogramming the EPROM-based processor. For instance, future versions may be configured with an I2C serial bus interconnect, so that multiple units could communicate with a master controller. The software-based control logic permits configuration for operation on non-standard (or non-USA) phone lines as well.

ANSWERING FUNCTION

The auto-answer board responds to the standard USA ringing signal of 90 VAC at 20 Hz. The system is protected from the false detection sometimes caused by dial-pulses on adjacent lines, or other causes, by a software function which counts AC cycles and integrates them over time.

DISCONNECT FUNCTION

Disconnect may result from any of the following:

* Loopcurrent interruption
* Loopcurrent reversal
* Detection of dial tone (selectable)

Loop-current interruption occurs on most telco lines when the calling party hangs up. It is sometimes referred to as CPC, or *Calling Party Control,* since the *calling* party controls your equipment when he hangs up. The CPC interruption was probably never intentional, having been a by-product of early mechanically-switched relay-controlled exchanges. Thus, some phone lines do not provide this function or they provide it unreliably. However, with the proliferation of answering machines which rely upon CPC, most central office equipment now has this capability designed in. In some cases, though, it is necessary to specifically request this feature from the phone company on a per-line basis.

Loop-current reversal, on the other hand, has long been a phone company signalling method. First used between the telco's own central offices, loop-reversal was later employed to communicate with some large premises PBX systems. Thus, lines which are set up for for PBX use, or originate at central offices with large concentrations of business customers, sometimes use this method. (However, the preferred and more modem situation for PBX control is to use either "ground-start lines" or "E&M signalling." A digression from matters relevant here. Both are discussed in the *Telos Telephone Q&A* - which you should have, since it comes packed with every Telos hybrid.)

As mentioned above, while most exchanges do provide CPC, there are some that don't reliably provide it, or provide it after a variable time delay... and most PBX's don't generate it. For this reason, this "super" auto-answer board has a "failsafe" dial tone detector. Every (USA) telco central office eventually returns dial tone to its lines when the calling party hangs up. Thus, we use the presence of dial tone to cause a disconnect when the loop-current detection methods fail.

The auto-answer board uses a sophisticated software-implemented statistical approach to ensure that the line is never inappropriately disconnected. Four tones are continuously monitored: 350, 440, 480, & 620 Hz. When the two tones (350 & 440 Hz) which comprise dial tone are detected as simultaneously present, a one-second detection "window" is opened. Only if, during this window, both dial tones are present for more than 80% of the time and both of the other tones are present for less than 20% of the time is a valid dial tone condition decided. In this way, false "talk-off" from noise, applause, or other spectrally-rich audio is prevented.

REGARDING DETECTION OF DIAL AND DTMF TONES IN SYSTEMS WITH CONFERENCING CAPABILITY

The auto-answer board is connected so as to receive the hybrids separated telco *receive* audio signal. Were this not the case, and the detector was merely connected across the phone line, there would be a major problem when multiple lines are used together in a conference. Why? *Because the tones would be conveyed to each line in use (through the switching matrix) from every other line, causing all of the detectors to respond to the tones from all of the other lines as well as its own!* Fortunately, the **Telos** One's excellent trans-hybrid loss keeps this from happening. Aren't you glad?

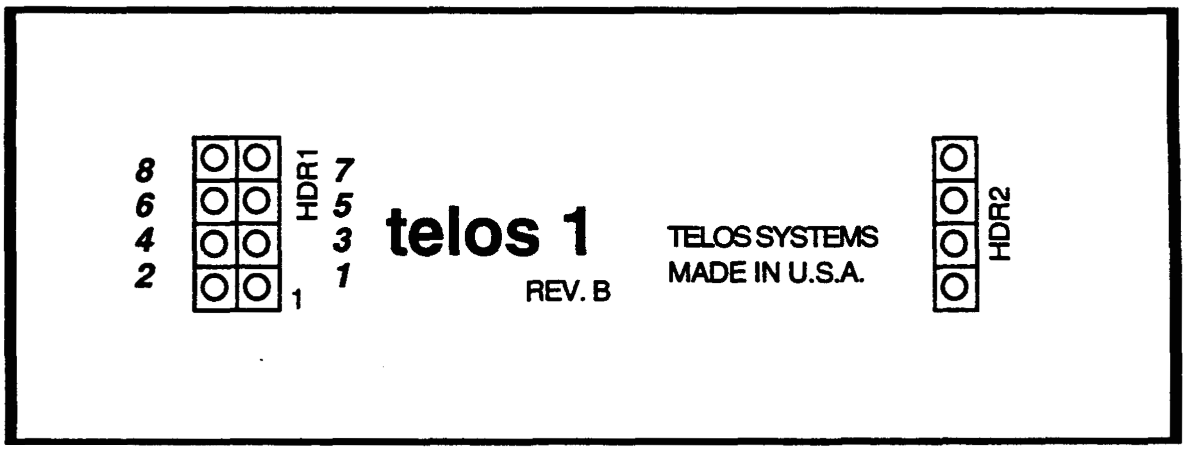
INSTALLATION

Installation consists of removing a jumper plug from a header connector on the Telos One and plugging in the auto-answer PCB.

The auto-answer board should be removed and the jumpers replaced if use with other than Central' Ofice-type lines with talk battery is expected, since the diodes in the loop detect section will cause severe audio distortion if not biased by talk battery.

1) Remove the Telos One front panel by removing **the two** black screws on either end of the front panel. Pull the top **case** forward to remove it. Remove the **two** ***similar*** screws on the back panel. Hold the rear bezel and gently pull the rear panel and **main** PCB out of the bottom case.

2) On the bottom of the PCB, solder a **small** piece of wire from U14-4 to HDR1-5. (The header is numbered 1,3,5,7 on one side and 2,4,6,8 on the other.) ***This modification need not be done on units with serial numbers greater than 08-100-00901 - Reo. C or later PCBs.*** The jumper wire allows the auto-answer board to have access to the hybrid-separated telephone receive signal.



HDRI & HDR2 locations and pinout

3) The auto answer board mounts on HDR1 and HDR2. First, remove the jumper plug which should be installed across HDR1-7 and HDR1-8. Line up the ***two*** headers **with** the **two** PCB connectors and push down on the board. Make sure both headers ***are*** picking up **all** the pins!

4) Replace rear panel with the new one, mount the DE15 remote connector, **and** plug the ribbon cable into the PCB socket.

5) Set the auto-answer dip switches for the desired options (described in the operation section next) before reassembling unit.

OPERATION

SOFTWARE VERSION AA V1.1

DIP **SWITCH OPTIONS**

Some user-selectable options are provided via a 4-position DIP switch located in one corner of the auto-answer PCB. **Functions** are **as** follows:

Dip #1: Auto mode permanent enable. Intended for when no remote control of auto enable/disable is available. When on, auto-answer mode is always enabled. The remote AUTO input **has** no effect and the remote Drop/Seize inputs will not disable the auto mode.

Dip #2: Future use.

Dip #3: Dial tone detect enable/disable. When on and the unit is in auto mode, will cause a disconnect when **dial** tone is present for ≈ 1 second, or longer. When this switch **is** off, only an interruption or reversal of loop current will cause a disconnect: dial tone **will** have no effect.

Dip #4: The numbersf-rings selector. When on (**and** the system is in auto mode) the unit will answer on the third **ring.** Otherwise, the answer ***occurs*** on the first ring.

REMOTE **CONTROL**

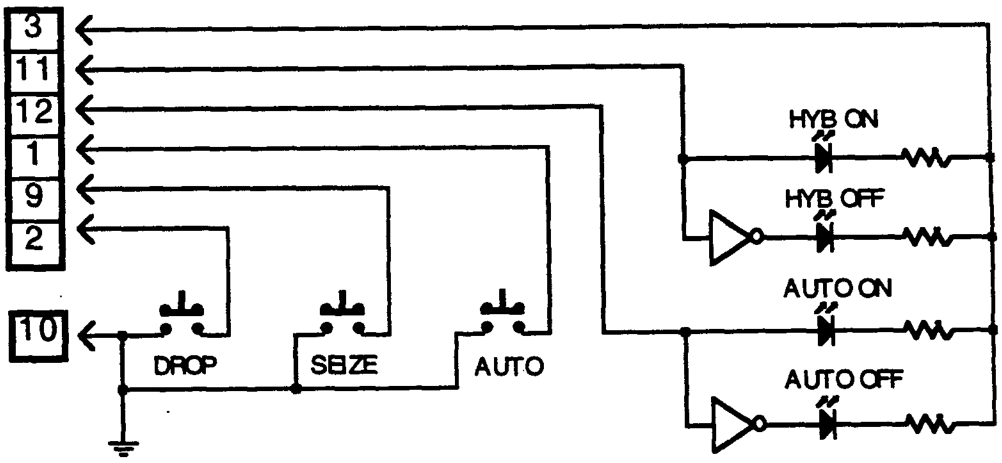
Remote control of the hybrids ON/OFF **functions, as** well **as** the auto mode, are available on the rear panel DB15 connector. Remote buttons are momentary and ground-common. (That is, they are normally pulled high through a resistor and taken to ground to activate.) Refer to the connector pin-out table for assignments.

When the **SEIZE** input is brought to ground, the auto board **turns** the hybrid on and disables the auto mode (except if DIP #1 is on). When the DROP input is brought to ground the hybrid will turn off **and** disable the auto mode (again, except if DIP #1 is on). The auto mode is disabled upon press of either **of** these buttons ***so*** that the unit doesn't hang up **from** the dial tone that would be present on the line when the **unit** is activated in order to originate a call.

The AUTO input also requires a momentary pushbutton, but is made to be alternate action in operation. Each time the button is pressed, the auto mode will toggle to the opposite state. Power-up is to the "no auto" mode (unless DIP #1 is active).

The LINE STATUS output will be made low as long as the hybrid is active. An external pull-up resistor to the +5V pin is required to drive an LED or similar indicator, since this output is driven by an open collector device. Should someone press the hybrids front panel ON button, the LINE STATUS output will flash a few times before staying on. This feature is intended to alert anyone watching the remote panel that someone else has activated a unit locally. Neither the front panel ON or OFF button will affect the auto mode.

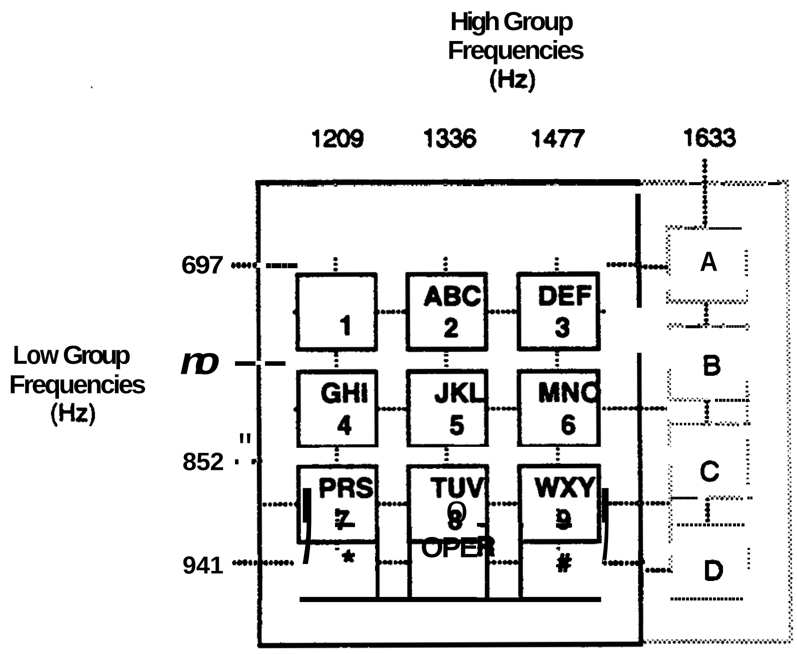
The AUTO MODE status output is also an open collector output. It is made low when the system is in the auto mode.



Schematic for remote control. Pin numbers are for the 15-pin D-connector. The inverter should be a 74LS14. The pull-up resistors' value depends on the type of LED used. Check the manufacturer's data sheets for details.

DTMF TONE DECODER

The auto-answer board brings out to the connector five signals from a DTMF (TouchTone®) decoder: four DTMF DATA bits and a DATA VALID signal. The tones are decoded and presented according to the chart. These outputs are pulled up (weakly) on the auto board and thus may be used as inputs to standard TTL and/or may be "wiresred" with other boards for special applications.



Frequencies and Buttons for DTMF Generation/Detection

DB15 PINOUTS (REMOTE)	
1	AUTO control input
2	DROP control input
3	+5VDC
4	D8 DTMF output
5	D4 DTMF output
6	D2 DTMF output
7	D I DTMF output
8	SCL
9	SEIZE control input
10	Ground
11	Line Mode output
12	Auto Mode output
13	DTMF Data Valid
14	n/c
15	SDA

DTMF OUTPUT CODES

Digit	D8	D4	D2	D1
1	1	1	1	0
2	1	1	0	1
3	1	1	0	0
4	1	0	1	1
5	1	0	1	0
6	1	0	0	1
7	1	0	0	0
8	0	1	1	1
9	0	1	1	0
0	0	1	0	1
*	0	1	0	0
#	0	0	1	1
A	0	0	1	0
B	0	0	0	1
C	0	0	0	0
D	1	1	1	1

Charts showing the hexadecimal codes for the touchtone decoder and pinouts for the 75-pin remote control connector.

SOFTWARE VERSION AA V1.2

This version of software functions similarly to **AA** V1.1 with the following differences:

1)The SEIZE input no longer disables the **AUTO function** when pressed.

2) The SEIZE input can **also** be used **as** a MAINTAINED input. That **is**, if the SEIZE input pin is held LO (to ground) for more than about 2.5 seconds, the hybrid will turn on (obviously) but will **turn off** (release the line) when the SEIZE input returns to a logic HI (+5VDC).

3) The DROP input will not disable the **AUTO** function while the SEIZE input is held LO, but will **turn** the hybrid off still. If the SEIZE input is used for momentary action, however, the DROP input will disable the auto functions just like **AA** V1.1.

This means that the operator can still enable and disable the auto function from a momentary LO signal on the **AUTO** pin while the SEIZE input is being held **LO.** If the DROP input **is** brought LO while the SEIZE input is being held LO, the hybrid will **turn** off but the auto function cannot be toggled anymore **via** the **AUTO** input. The auto function will remain in the state it was in when the DROP pin went LO. This is done to signal the operator that the device that initially turned the hybrid on via a maintained **closure has** not released the **SEIZE** input yet. The ON/OFF and **AUTO** outputs will still function normally. *All* other **functions** of AA V1.1 have been retained.

SOFTWARE VERSION AA V2.0

This software version is designed for **use** with multiline 1A2 key system telephones. It will turn the hybrid on whenever an active line button is pressed on the key phone and turn the hybrid off when the line button is popped up, either by hitting another button half way, pressing an unused button (i.e. no phone line connected for that button), **hanging** up the handset or placing a line on hold.

Circuit modifications: The **ring** detector optoisolator, **U3,** should be moved to the voltage detector circuit, **U4.** R2 should be **33KΩ,** C5 must be removed, and the DTMF receiver chip, U7, should not be **stuffed. Call** Telos if you'd like the **DTMF** decoder option restored to your unit.

Key phone modification: One key phone must be modified to get T/R audio from the selected line to the hybrid. Inside the phone, a RED and a **GREEN** wire **go from** the switch bank to the network under the touchtone pad, the RED wire to terminal ***6***, the GREEN wire to terminal 8. Remove these wires from the terminal block and run them to the hybrid's **LINE** modular jack. Now take another pair of **wires** and run them from the hybrids **PHONE** modular jack BACK TO THE **TERMINAL** BLOCK pins ***6*** & 8. Essentially, we're putting the hybrid IN SERIES with the phone line that has 'been selected. That is, when the hybrid is in its **OFF** mode the phone line is looped through the hybrid back to the phone's network. When the hybrid turns ON, a relay closes, moving the T/R from the phone and into the hybrid.

DIP SWITCH OPTIONS

DIP #1: When on, enables the **AUTO** function; when **off** the **AUTO** function can be toggled on & off via the **AUTO** button input provided on the connector, J7-1.

DIP #2: Not supported.

DIP #3: Dial tone detect enable/disable. When on and the unit is in auto mode, will cause a disconnect when dial tone is present for ≈ **1** second, or longer. Same **as V1.1. You** might want **this** option active if your talents tend to select inactive lines!

DIP #4: Was for number of **rings** selector. Disabled in software, since **this** pin is now looking for DC voltage to answer, not cycles of AC **ring** voltage.

All other remote control functions (remote seize, remote drop, line status, auto status, auto on/off) remain the same. It is recommended that the key phone handset be **OFF-HOOK** so the key phone's buttons flash at the correct rates. If the talent wants to release a line without taking another caller, he can hit the hook switch to pop the button **up!** If the phone must be **used** normally, the hybrid must be turned off manually. One way to do **this** is to rewire the disconnect button to function **as** the autoanswer board's DROP signal. (The disconnect button is an optional button on some key phones; it's used to disconnect a line without **having** to **pop** the button up.) If your phone doesn't have the disconnect button already, add a momentary button across **pins** J7-2 & J7-10 and mount it either on the phone or near the phone.

We recommend turning on **DIP** #1 and then installing the board in your unit. Of course, you *can* take advantage of the other options by connecting some extra momentary switches and some **LED** indicators. This way the operators can toggle between **AUTO** and NO **AUTO,** turn the hybrid on and **off** manually and have indicators of the hybrids status **as** well. Refer to **this** manual's **OPERATION** section for a circuit you *can* build up that will do **this.**

If touchtones are needed to dial out and must be heard on the air, another touchtone pad should be fed into the hybrids mix-minus scheme. The touchtone pad in the key phone needs the -48V from the phone line to operate, so it would need its own power supply. Another scheme could use some **type** of relay circuit that would reconnect **ONLY** the touchtone pad in the key phone (**NOT** the network!) **to** the line selected **ONLY** while dialing . We don't want the tone pad hanging on the line all the time, because it will load down **the** line. Since the tone pad is **also** powered from the phone line, it will only be active while it is connected to the phone line through the relay. Some stations use prerecorded carts **with** tones on them, **so** for instance, the mayor's *unlisted* home phone doesn't **go** out on the air!

SOFTWARE VERSION AA V3.1

This version of **software** is very similar to the AA V1.1 software, but is intended for use on the telephone system **of** France. Here are the major differences:

1) **Ring** detection is based on the French **ringing** frequency **of 50Hz**.

2) Dial tone release has been changed to differentiate between the various reorder tones applied on the French telephone network. First, the French phones **use only** a *MHz* tone (instead of the 440Hz + 350Hz used in the **US).** More precisely, a steady M H z tone will not turn the unit off. Rather, a 440Hz tone that is pulsed at a rate of **1/2** Hz **±10%(the** French reorder tone) will turn the hybrid **off.** Moreover, a 440Hz tone that is on for 1.5 sec. and off for 3.5 sec. **±10%** (another **type** of reorder tone) **will** turn the hybrid off, **as** well.

All **DIP** switch options are identical to **AA** V1.1. **as** well **as all** the I/O signals on the 15-pin connector, including remote on/off/auto, their respective indicator outputs, and the decoded **DTMF.**

"SUPER" AUTO ANSWER/RELEASE ADDENDUM #2a (supercedes Addendum 2)

SOFTWARE VERSION V5.2

This version of software is intended for use on the Italian telephone system. The Italian phone system **uses** a non-standard reorder tone (425Hz) that is pulsed at a rate **of 200 msec.** on/ **200** msec. **off. Ring** frequency is 50 ***Hz.***

HARDWARE CONSIDERATIONS: In order to recognize this tone the hardware must be changed. First, the crystal frequency is changed to make the call progress detector chip detect **425Hz.** Because of the short duration of the tone, this chip has a difficult time detecting a valid tone. So, **an** extra op-amp section is added to the hardware to ensure a high enough level is presented to the chip's input. To avoid line release from music and other sources the talk-off portion of the code **has** been modified to detect ***four*** consecutive tone sequences. Should the levels not be consistently high enough, the unit may take more than four beeps to turn off. The DTMF decoder will not operate at all because its reference frequency **has** been changed (to accommodate the call progress detector chip).

SOFTWARE CONSIDERATIONS: This version of software can be used with the Telos DTMF-to-Rotary Dial converter board. When a DTMF tone is present on the ***send*** input of the hybrid, the converter board will hold the hybrid's "on" pin low. This is done so that the hybrids output is muted during pulse **dialing.** Furthermore, the Super Auto-answer boards software must not **try** to tum the hybrid off because of T/R reversal or loop drop, which could happen during pulse **dialing.** *(See* the schematic of the converter board for details.) This software version addresses this problem by looking at the "on" pin while in the "off-hook" mode and compensates for the pulse dialing by adding a time-out period after the **"on"** pin has been released. It then **goes** back to the current detectors to reset its "when-to-tum-off sequence.

AA V5.2 supports all DIP switch options (identical to **AA V1.1.)** **as** well **as** most all the I/O signals on the 15-pin connector, including remote on/off/auto, and their respective indicator outputs. The decoded DTMF outputs are disabled and should not be used.

Super Auto-answer Board

PARTS LIST

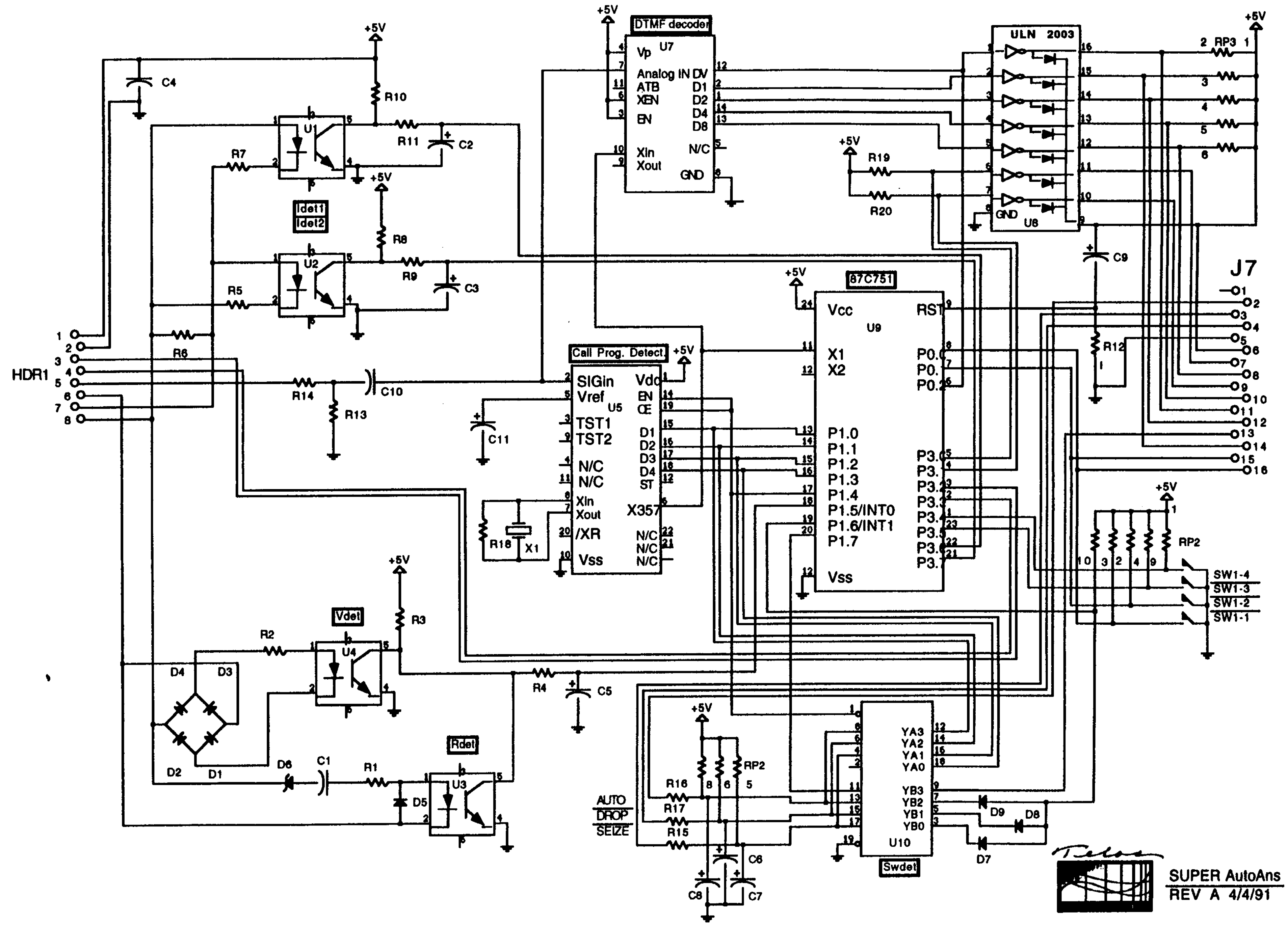
Designation	Description
u1-4	Optoisolator 4N33
U5	Call Prog Detect SSI75T982-CP
u7	DTMF Decoder SSI75T204-IP
U8	Octal open coll. driver ULN2003
U9	Microprocessor 87C751-1N24
U10	Octal buffer 74LS244
RP1	10KΩ SIP 4306R-101-103
RP2	10KΩ SIP 4310R-101-103
R1	2.2KΩ 1/4W 5%
R2	47KΩ 1/4W 5%
R3	100KΩ 1/4W 5%
R4	910Ω 1/4W 5%
R5,7	1.8KΩ 1/4W 5%
R6	47R 1/4W 5%
R8,10,13,14	10KΩ 1/4W 5%
R9,11	1KΩ 1/4W 5%
R12	8.2KΩ 1/4W 5%
R15,16,17	100R 1/4W 5%
R18	10MR 1/4W 5%
R19,20	1KΩ 1/4W 5%
C1	0.47µF mono
C2,3,6,7,8	2.2µF tant. 9 25V
C4	0.1µF mono
C5	0.33µF tant. @25V
C9:	10.0µF 025V
C10	0.0022µF mono
C11	1.0µF 025V
D I-5	diode 1N4004
D6	Zener diode 1N5246
D7,8,9	signal diode 1N4148
x1	3.58MHz crystal

Designation	Description
SU1-4	6-pin socket
SU5	22-pim socket (400mil)
SU6	8-pin socket
su7	14-pin socket
SU8	16-pin socket
SU9	24-pin socket (300mil)
SU10	20-pin socket
SW1	4-position DIP SW

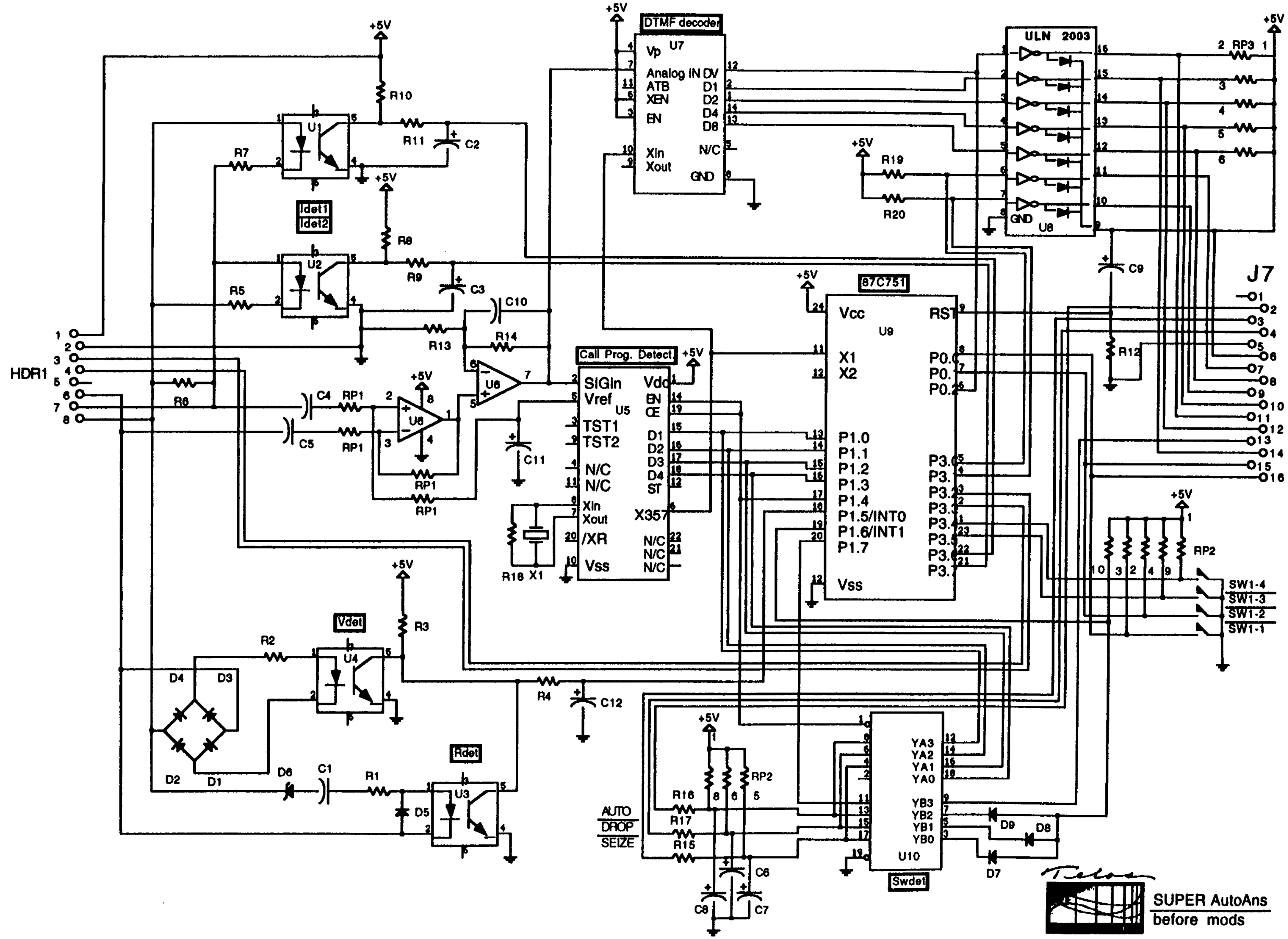
16-PIN RT ANGLE HEADER 0.1" spacing
8-PIN FEMALE HEADER 0.1" spacing
4-PIN FEMALE HEADER 0.1" spacing
15-pin female FRC D-shell ass'y

NOTES:

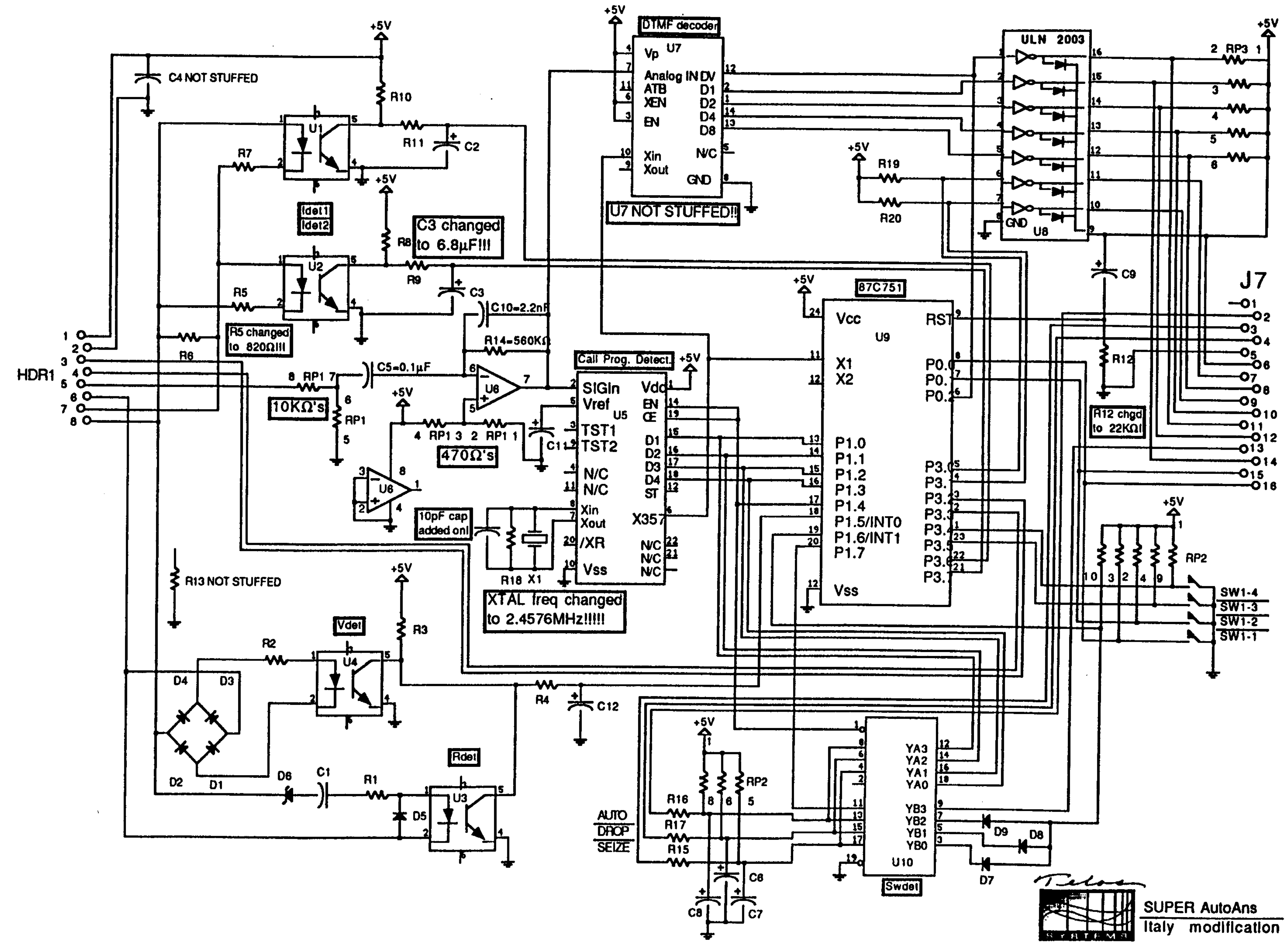
-All sockets machined pin, selective gold
-MPU must be programmed; available from Telos.
Refer to text for software options.



DTMF decoder
U7
Call Prog. Detect
U5
87C751
U9
ULN 2003
U8
J7
HDR1
Idet1
Idet2
Vdet
Rdet
Swdet
U10
AUTO
DROP
SEIZE
SW1-1
SW1-2
SW1-3
SW1-4
SUPER AutoAns
REV A 4/4/91



SUPER AutoAns
before mods



DTMF decoder
U7 NOT STUFFED!!
ULN 2003
C4 NOT STUFFED
C3 changed to 6.8µF!!!
R5 changed to 820Ω!!!
R12 chgd to 22KΩ!
87C751
Call Prog. Detect.
10KΩ's
470Ω's
10pF cap added on!
XTAL freq changed to 2.4576MHz!!!!!
R13 NOT STUFFED
HDR1
J7
AUTO
DROP
SEIZE
Swdet
Rdet
Vdet
Idet1
Idet2
SUPER AutoAns
Italy modification

Tutorial

ELECTRONIC PHONE SYSTEMS

Electronic Phone Systems

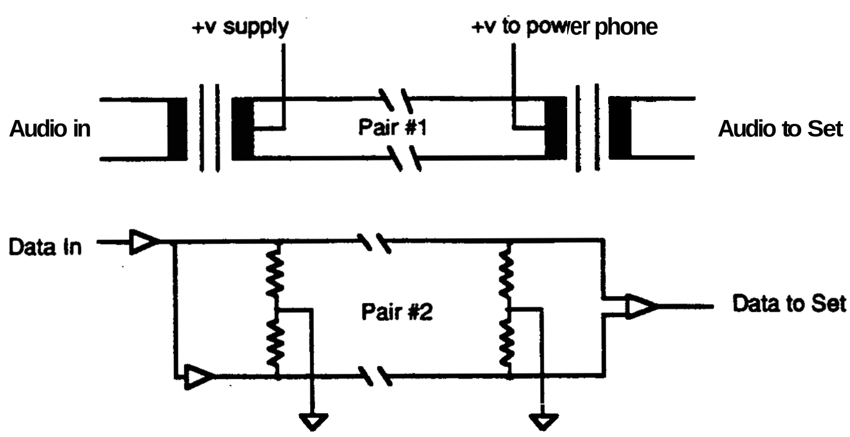
This tutorial introduction to electronic phone systems may be helpful if you desire to connect a Telos hybrid to such systems.

The cable from the phone sets to the "common equipment" must convey:

- Power to operate the phone
- A two-way data path
- The speech audio path

The early electronic phones used a separate pair for each of the three functions, and thus required six wires. The AT&T Horizon was an example of this approach. Another is the popular Merlin system which uses four pairs.

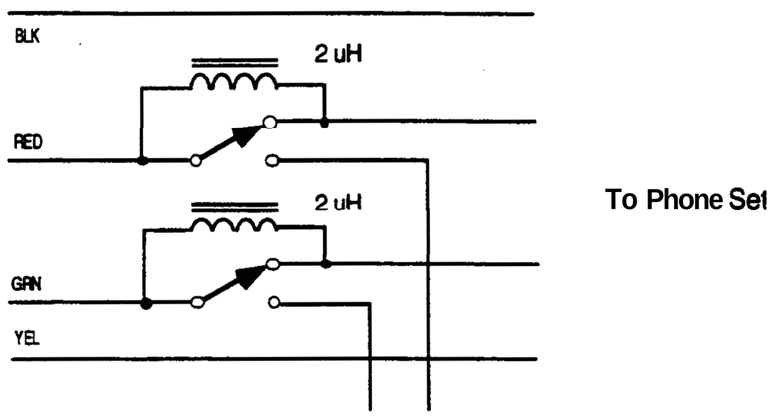
The most common approach used in new designs is a two pair, four-wire scheme. In this scheme, the talk and data are each balanced and each use one of the two pairs. The power is "phantom" applied between the two pairs in a way similar to the method used for phantom powering condenser microphones in recording studios.



Electronic Phone Scheme Using Two Pairs - Note that this diagram shows data flow in only one direction for simplicity.

A transformer is generally used at each end of the audio pair so that the phantom power may be added. The data pair often uses resistors to obtain a "center tap" rather than transformers since the data signal has DC components that would not pass through a transformer.

Usually, in the two pair approach, the center two wires on the modular plug are the audio path.



One method for interfacing to some electronic phones. The chokes pass the DC voltage to the phone set when the hybrid is active.

The Mitel "Superset" phones use a unique scheme that requires only one pair for all three functions. How do they do it? The data is amplitude shift modulated onto a 32 kHz carrier "over voice" and then the combined voice and data are AC coupled across the DC power voltage.

The most advanced systems use a pure digital bit stream for both voice and data. The phone set contains the CODEC for conversion to and from the analog and digital domains. The pure digital approach is used in the AT&T System 85 and the digital version of the NEC NEAX 2400 as well as in the new Northern Telecom Norstar system.

Since most electronic phone systems use an analog talk path, a hybrid can be connected to the system by inserting it in the analog path by breaking the connection from the phone switching equipment to the instrument. Usually, the talk audio is found on the center two wires of the line going to the phone. Looping this through the hybrid gets the audio connected.

The operator is responsible for turning the hybrid on and off upon initiation of each call. The hybrid's on and off remote may be connected to the console's on/off logic outputs to perform this function. This scheme works well for one-call-at-a-time situations as you would have in a newsroom or production studio or for occasional on-air use.

A problem remains, however, when calls need to be taken one after the other, as in a talk show situation. In most cases, the hybrid must be signalled each time a new line is selected so that adaption to the line can occur. It may be possible to derive an appropriate pulse from the phone by tacking-on an external circuit of some kind which would trigger the hybrid's remote on/off function. Remember too, the hybrid should be switched off or its input audio removed when no line is active. Otherwise, the hybrid will proceed to adjust itself to the high impedance presented to it and may take a few seconds to recover from this condition when a call is subsequently taken. Your kindly phone supplier may be able to help. If you have any luck along these lines, please let us know so that we can produce an application note for other users.

With phone systems that have a digital talk path, the situation with the on/off control is as above with the additional complication that you need to find a way to get at the converted-to-analog signal inside the phone set.

Perhaps some poking around with a scope or headphones may be useful in this endeavor. Again, maybe your phone provider company might help. Please let us know what you find!

A universal way to obtain an audio signal from any phone is to use an adapter which plugs into the handset jack. These should be available from your local telephone supplier. If you have problems finding an adapter locally, we may be able to help.