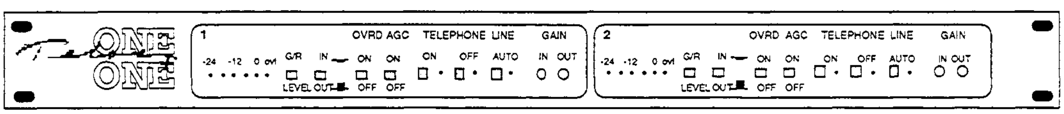
Telos ONE plus ONE

Dual Digital Telephone Interface



User's Manual

Version 3.0 – August, 1996

Telos Systems
2101 Superior Avenue
Cleveland, OH 44114
(216) 241-7225
FAX: (216) 241-4103
http://www.zephyr.com
email: support@zephyr.com

A personal note from Steve Church

This note was written for the very first Telos ONE manual. I have left it unchanged as it provides an interesting "snapshot" of Telos at the time. Now, we have been producing DSP hybrids for over 10 years; our fourth generation Telos 100 Delta is the world's premier digital hybrid telephone interface; and we are producing Zephyr™, our first ISDN transceiver.

Steve Church
May 1994

July 10, 1989

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 will like it.

Keep on keeping the GM happy...

Steve Church
President

User's Manual V2.0

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:

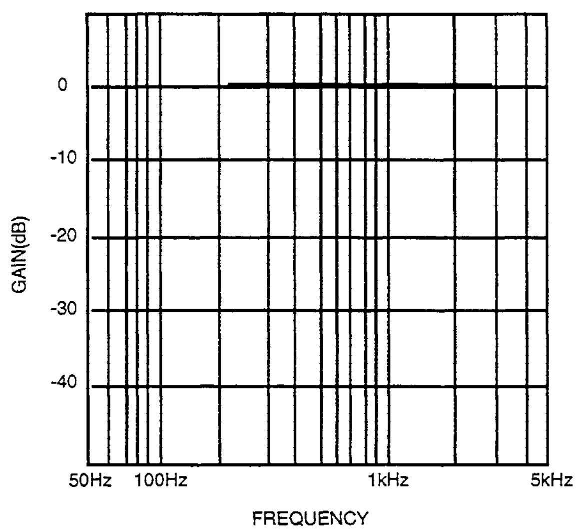
> *Switching.* The send and receive paths were separated by having only one talk direction active at a time. The common "speakerphones" use this approach. The disadvantage is that natural conversation is impossible, since the caller is cut-off when the announcer talks - and vice-versa.
>
> *Analog hybrids.*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

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 SOFTWARE OPTIONS

There are several Telos ONE software/hardware options for special applications. Options are subject to change without notice.

Version 1.X Normal Operation

This software serves the needs of the overwhelming majority of Telos ONE users and is ordinarily installed in units.

Version 4.X Four-Wire Intercom/Teleconferencing

This version is widely used with RTS, McCurdy, ClearCom, and other four-wire intercom systems. It has 6dB greater send level, ducking is modified to somewhat favor the receive audio, and input AGC is replaced with a limiter. This version can also be used to create a multi-line teleconferencing bridge.

Version 5.X "Dallas" Software

As far as we can tell, many of the worst phone line conditions in the US exist in the Dallas/Fort Worth and Miami/Fort Lauderdale areas. The "Dallas" software is optimized for very poor phone lines with widely varying levels. It can also be used by those who prefer more of a "speakerphone" switching effect; prefer a more aggressive AGC on the phone audio; or have feedback problems with open monitor speakers. We recommend that you try the Normal Operation software first. If its performance is not optimal, contact Telos to discuss whether the "Dallas" software will resolve your difficulties.

Version 6.X Intercom Interconnect

This software is used to interconnect two-wire intercom systems with four-wire intercom systems. The hybrid's telephone jack is used as the input for the two-wire intercom. The software keeps the telco line seized at all times in case of a power interruption. A minor hardware change blocks the DC on the intercom side, increases send-to-intercom level, and re-adjusts the analog hybrid to accommodate 200Ω intercom impedance. The interconnection is bandwidth limited to 300Hz to 3500Hz.

1.3 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 override 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 internal digital AGC.

Frequency Response (caller to output)

200 - 3400 Hz +-1 dB.

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

<.5% THD + N. 1 kHz; 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

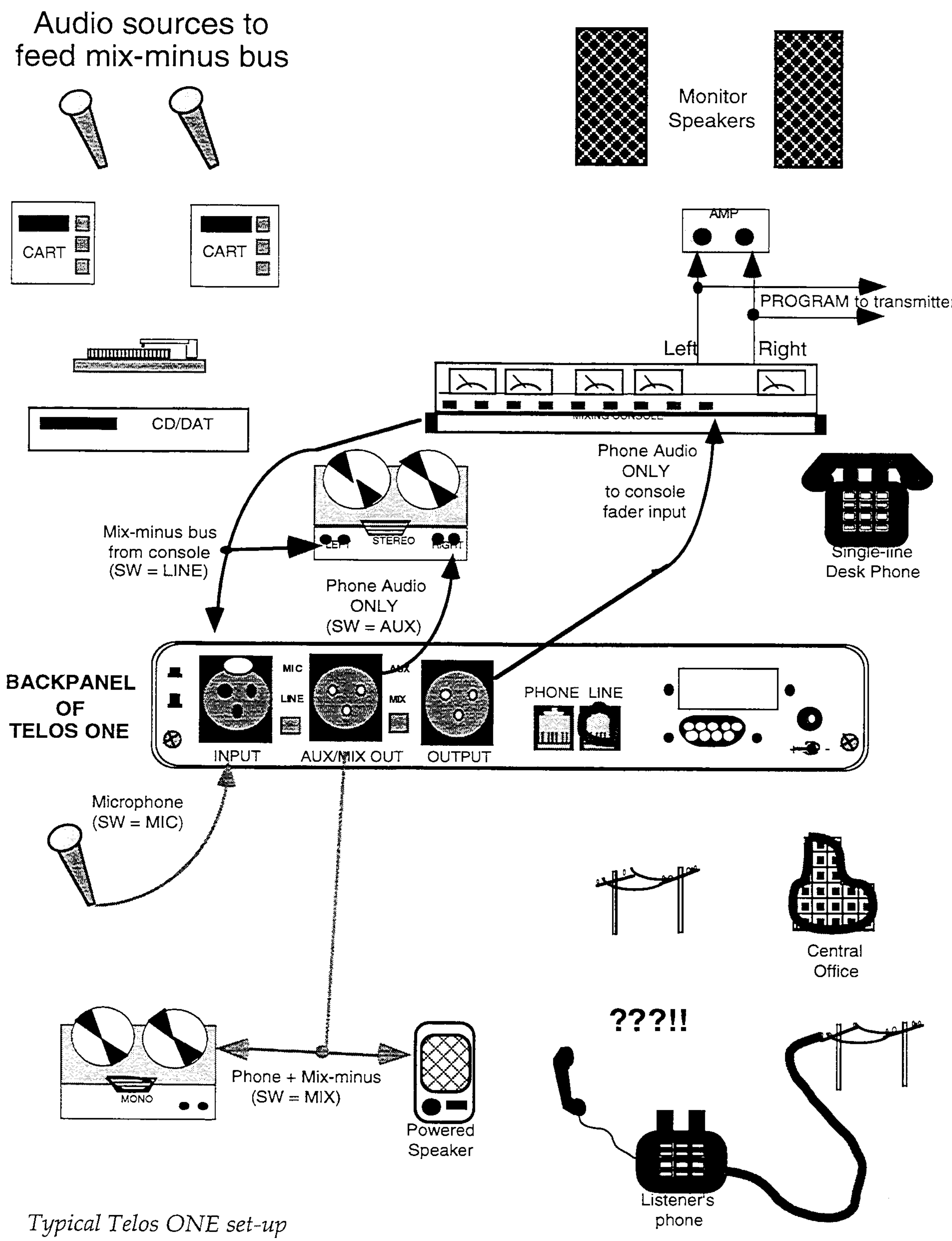
XLR male connector. Active differential. Output levels to +14 dBm depending upon caller telephone line level and adjustment of front panel level adjust. Will drive 600Ω.

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

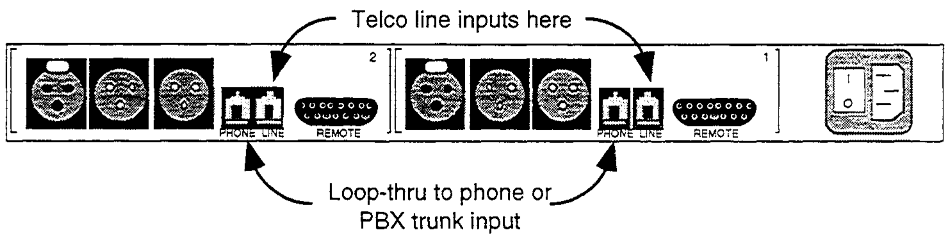


Typical Telos ONE set-up

2.1 CONNECTION TO THE TELEPHONE LINE

LINE and PHONE Modular Jacks

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



The LINE jack should be connected to the incoming central office telephone line using a modular cord (provided).

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

Both modular jacks use the center two pins (Red & Green) for the audio connection.

"A lead" Output

The "A lead" output provides a relay contact closure which may be used for any desired purpose. Typical application would be to hold up the line when user-devised connection schemes to multi-line phones are implemented.

The outer two pins (Black & Yellow) of both modular jacks provide the A lead output. This contact closure is available on pins 2 and 8 of the REMOTE connector as well.

Auto-answer Capability

Telos manufactures two kinds of plug-on PC boards which may be installed inside the hybrid to provide auto-answer/release. These cards can turn the hybrid on and off when appropriate signalling is provided. Call Telos or your Telos dealer for more details on the Standard Auto-answer card or Super Auto-answer card.

Multi-line Installation with Telos Interface Modules

Information on use of Telos multi-line control and interface modules is given in the

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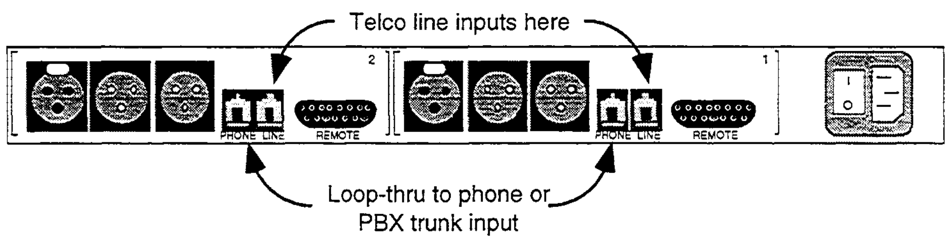

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Multi-line Installation with Telos Interface Modules

Information on use of Telos multi-line control and interface modules is given in the manual included with the appropriate modules. Telos manufactures a full line of line selection, interface, and control equipment. We have interface modules for connection to 1A2 key equipment, as well as for direct connection to telephone lines. Control surface options include table-top consoles and drop-in modules for many broadcast consoles.

"1A2" Key System Installation Without the Telos Modules

It is possible to connect the Telos ONE hybrid to 1A2 key phones without the Telos interface module. With this approach, the key phone is used as the line select device. An application note is available from us, if you wish to construct this kind of system.

The scheme is fairly involved because the 1A2 A leads must be sensed and turned into pulses which turn the hybrid on and off and provide a trigger for the mute/null mode upon selection of a new line. Ask about our Super Auto-answer board for this application.

The section on key systems in the *Telephone Q & A* may be helpful if you desire to construct your own interface.

Electronic Phone Systems

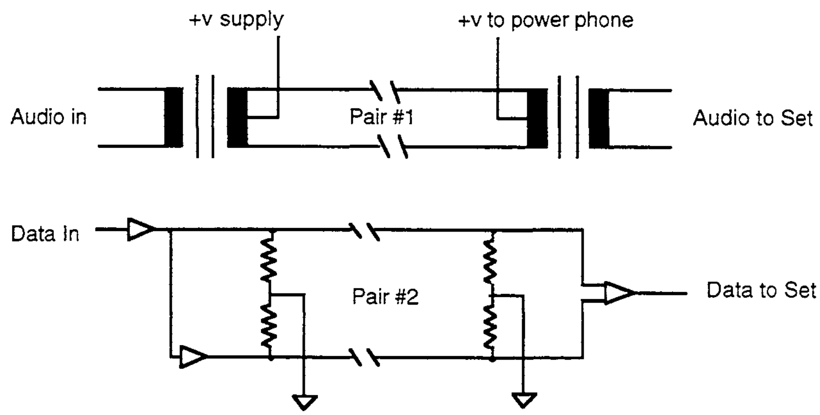
These have nearly completely taken over from older mechanical systems with the holdouts being broadcast studios and the financial brokerage business. The cable is smaller and cheaper and fancy features are easy to implement.

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.

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.

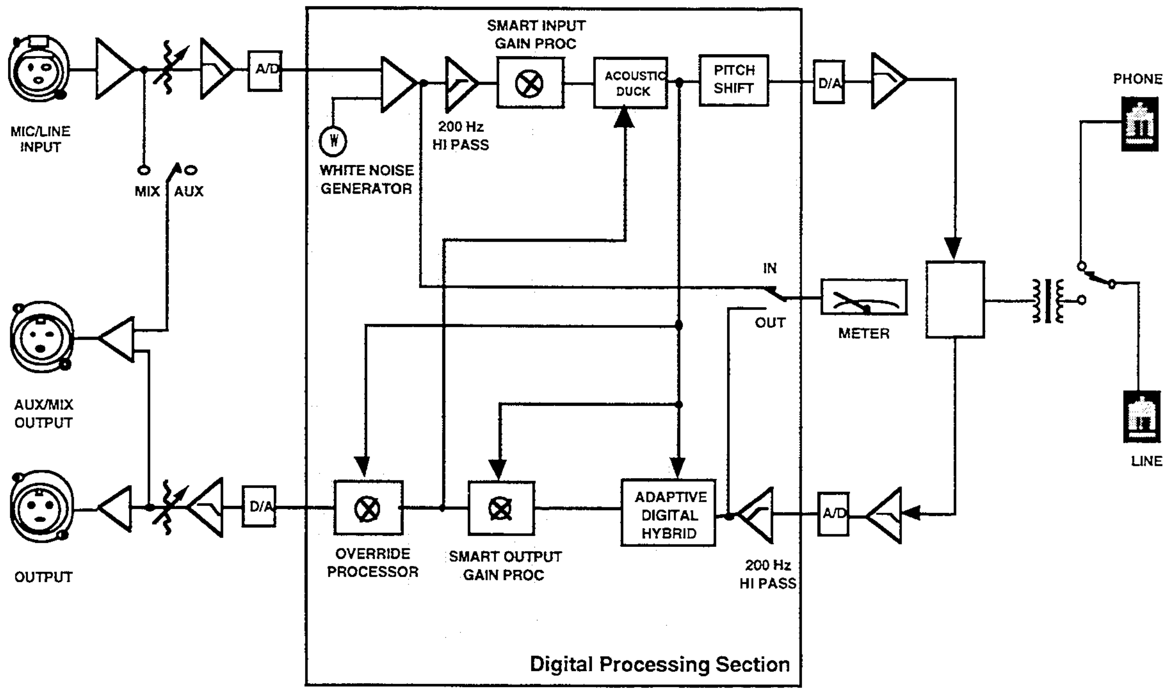
A problem remains, however, when calls need to be taken one after the other, as in a talk show situation. In most cases, the hybrid must be signalled each time a new line is selected so that adaption to the line can occur. It may be possible to derive an appropriate pulse from the phone by tacking-on 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.

2.2 AUDIO CONNECTIONS



Telos ONE audio block diagram

2.2.1 MIX-MINUS

The Telos ONE input 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. The sources that feed the mix-minus bus should be pre-fader, so that they are always feeding the hybrid and so the operators can't upset the send level to the hybrid.

Increasing the send level beyond a normal meter reading does not increase the level into the phone line, due to internal, digital limiting of the hybrid! Refer to Section 2.3.1 on how to set the send level to the hybrid.

There are a number of ways to create a mix-minus feed.

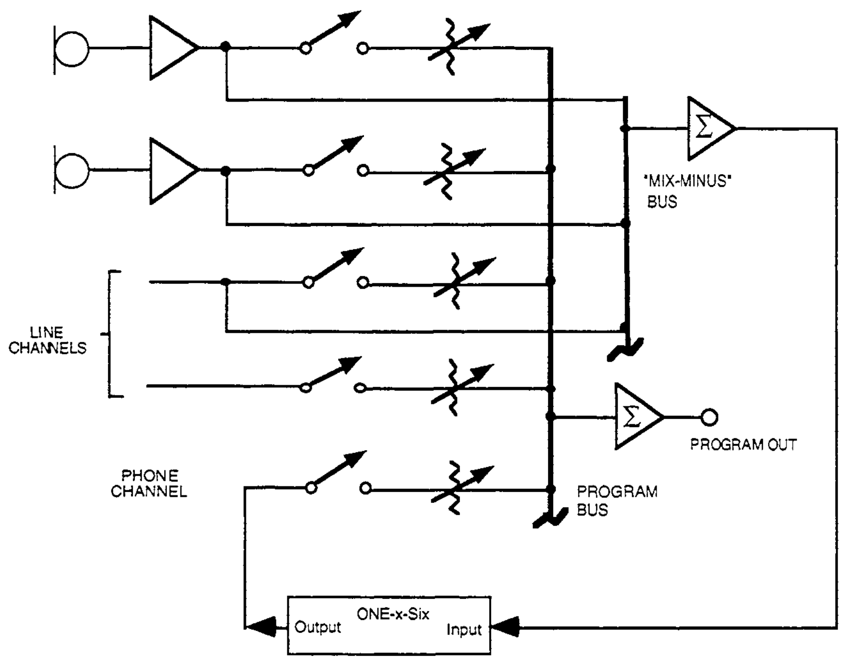
Simple Mix-Minus

For the simplest installation, you can just take the patch send or preamp output from the mic channel to feed the hybrid. This works well, but doesn't have much flexibility.

Another 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.

Sophisticated Mix-Minus

Most modern broadcast consoles provide 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. The diagram that follows illustrates in simplified form a portion of a broadcast console with a mix-minus bus.



Simplified mix-minus scheme

While on the subject of consoles and mix-minus, we'll digress here for a moment. Many hybrid installation problems are caused by an inadvertent signal path which creates a loop from the hybrid's output to its own input. Some console designs allow this to happen when certain control combinations are user-selected. This is the first place to look when strange or erratic performance is experienced. The ONE's front panel input meter should give you an immediate answer.

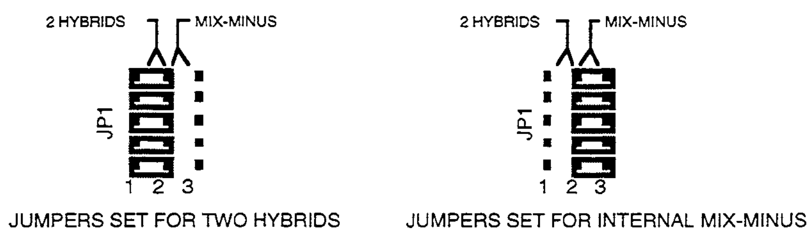
A Good Idea...

Here's a neat 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 input gets 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 with caller and announcer audio separated, which makes it easier to do production for contest squeals, etc. If you only have mono tape decks, use the second output of the hybrid in the MIX position. See Section 2.2.3 on the second output.

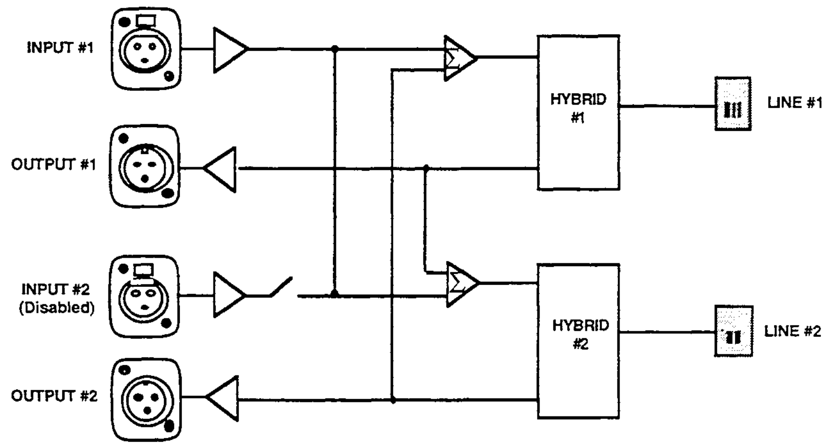
Internal Mix-Minus in the ONE plus ONE

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 two hybrid outputs are NOT summed together, so you should provide a fader for each hybrid.

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 XLR to the #2 hybrid is disabled. Both input trimmers on the front panel will set your mix-minus level for its corresponding hybrid. The cross-coupled hybrid audio is fixed at unity gain, relying on the output AGC to keep levels OK. The trimmers do NOT affect this level! The following block diagram and schematics should help make all of this clearer.



JP1 on rear panel connector board selects between two external mix-minuses or one external mix-minus cross-fed to both hybrids.

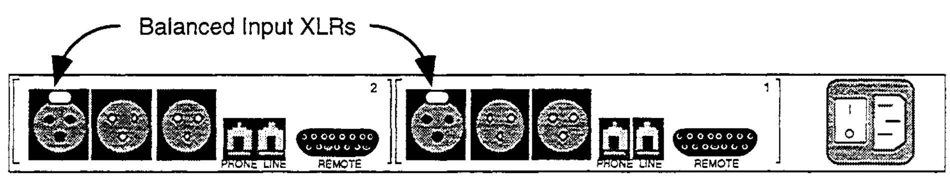


Internal Mix-Minus Scheme (JP1 in "MIX-MINUS" position)

2.2.2 INPUT AUDIO CONNECTIONS

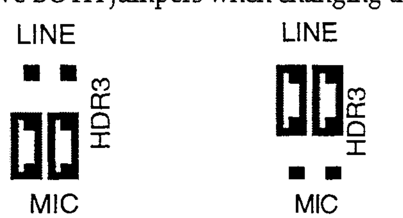
The input has the following characteristics:

- 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

2.2.3 OUTPUT AUDIO CONNECTIONS

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

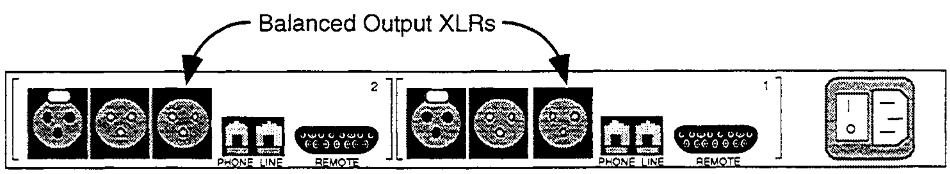
- Active balanced.
- Output level will vary from approximately -20 dBm to +10 dBm depending upon gain control adjustment, caller level and whether or not the AGC is engaged.
- Pin 1 is ground. Pins 2 and 3 are the balanced signal outputs.

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.*

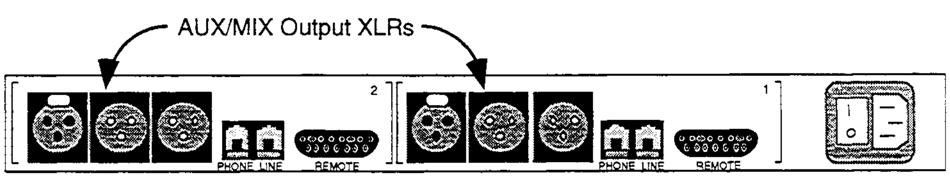
Main Output

Caller audio appears on each of the MAIN outputs.

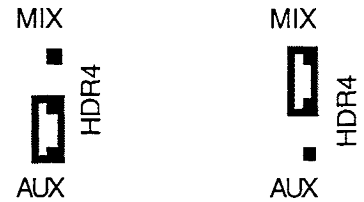


AUX/MIX Output

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



Internal jumpers on the hybrid circuit board, on HDR4, select which mode. Jumper pins 1 & 2 to get the AUX function, pins 2 & 3 for the MIX function.

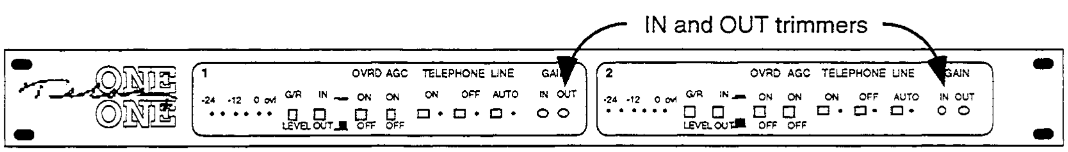


AUX FUNCTION MIX FUNCTION

When in the MIX mode, the input is passed to the mix output at unity gain. Note from the block diagram at the beginning of this section 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.

2.3 FRONT PANEL GAIN ADJUSTMENTS

There are two screwdriver-adjustable multi-turn trimmers recessed from the front panel, marked IN and OUT. These trimmers are used to set send (audio signal going down the phone line to the caller) and receive (audio output of the hybrid, i.e. caller audio) levels, respectively.



2.3.1 INPUT GAIN

The trim pot marked IN adjusts the audio level that is being sent to the hybrid's input XLR, which should be a mix-minus feed. (See Section 2.2.1 for a discussion of mix-minus). The unit comes from the factory with the trimmer fully clockwise–that is, all the way up! Follow these steps to adjust the IN trimmer:

1. Press the G/R - LEVEL button so it is in the OUT position; press the IN/OUT button so it is in the IN position. This makes the LED meter read INPUT LEVEL.
2. Select MIC or LINE level range by pressing the switch located to the right of the input XLR on the rear panel.
3. Feed some audio at a normal level into the mix-minus bus. This can be a microphone, a tone generator, or a cart or CD playing your favorite tunes.
4. Look at the meter and begin turning the IN trimmer counterclockwise with a small screwdriver until most of the green LEDs are lit and the red LED only comes on every once in a while during peaks in the audio. Adjusting

this level is just like setting the level to a tape deck. You wouldn't peg the meters on a tape deck, would you?

IMPORTANT NOTE: Increasing the send level beyond the "0" reading does NOT increase the level into the phone line. The level of the audio sent down the phone line is part of the digital signal processing and is set to provide the maximum level permitted. Due to the internal, digital limiting action of the hybrid excessive input level may cause unwanted aliasing distortion and poor adaption!

2.3.2 OUTPUT GAIN

The trim pot marked OUT adjusts the audio level that is being sent from the hybrid's output XLRs to your console. This trimmer changes the caller level at the OUTPUT and AUX/MIX XLRs for that hybrid the same amount. Once again, the unit is shipped with this trimmer fully clockwise– turned all the way up. Follow these steps to adjust the OUT trimmer:

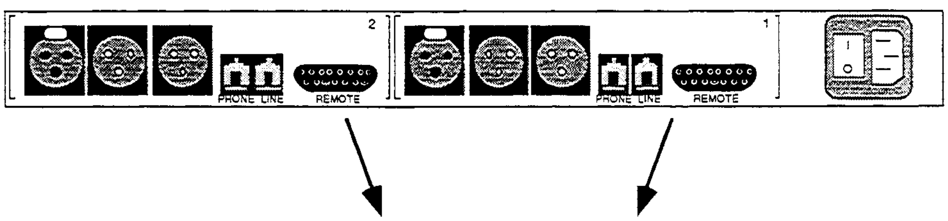
1. Press the G/R - LEVEL button so it is in the OUT position; press the IN/OUT button so it is in the OUT position. This makes the LED meter read OUTPUT LEVEL. The meter is reading caller audio level on the phone line *before* the output trimmer. A "0" reading corresponds to an approximately -15dBm phone level. This can tell you if your phone line level is acceptable. *The meter will not change as you adjust the output trimmer.*
2. Connect one of the hybrid's outputs to a line-level input on your mixer and set the mixer's meter to read that input's level.
3. Press the front panel "ON" button to seize the phone line. This will bring dial tone to that fader. Run the fader up to a normal gain level.
4. Look at the meter on your console and adjust the output trimmer on the Telos ONE with a small screwdriver so that the console's meter reads approximately +3dB. (Dial tone is typically a few dB hotter than most callers.) If you can't get enough level out of the hybrid, press the button on the front panel marked AGC so it is in the IN position and readjust the trimmer. Or, if your console fader has a gain adjust on its input, increase gain there.
5. Repeat step 4, only this time call a friend and have him read the Gettysburg Address or your favorite poetry while you adjust the trimmer. Try this several times on different lines to ensure you have good range with the big fader. Once again, don't peg the mixer's meters!

2.4 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 or not. Refer to the chart on the next page for pinouts and their associated functions

Without the SAA installed:
Hybrid OFF and ON control require momentary closures 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.

When the SAA PCB is installed:
The rear panel DB15 will be internally connected to the Super Auto-answer board's header. The on/off commands and other functions are passed to the hybrid from the SAA as required. Refer to its manual section for details on remote control operation.

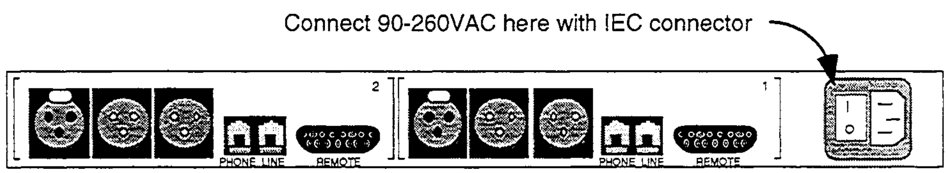


DB15 "REMOTE" Connector Rev. B		
Pin No.	**Without SAA**	**SAA Installed***
1	Front panel AUTO button	AUTO enable input
2	Hybrid OFF input	DROP control input
3	+5VDC	+5VDC
4	n/c	"D8" DTMF O/C output
5	n/c	"D4" DTMF O/C output
6	n/c	"D2" DTMF O/C output
7	n/c	"D1" DTMF O/C output
8	normally OPEN relay contact	SCL
9	Hybrid ON input	SEIZE control input
10	Digital GROUND	Digital GROUND
11	Relay WIPER contact	LINE MODE output
12	Front Panel AUTO LED	AUTO MODE output
13	n/c	DTMF Data Valid O/C output
14	n/c	n/c
15	normally CLOSED relay contact	SDA

*Refer to Super Auto-answer Manual for details

2.5 POWER INPUT

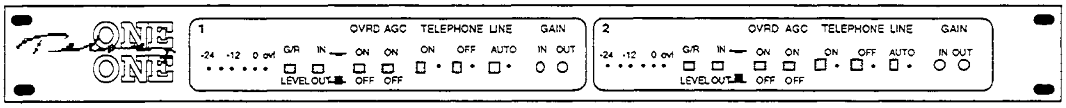
The ONE plus ONE has a built-in, universal-input power supply. The input is a standard IEC three-prong AC connector with a power on/off switch. The fuse is located on the internal power supply. For more details on the power supply, see Section 4 of this manual and the manufacturer's data sheet in the appendix.



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 *caller 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

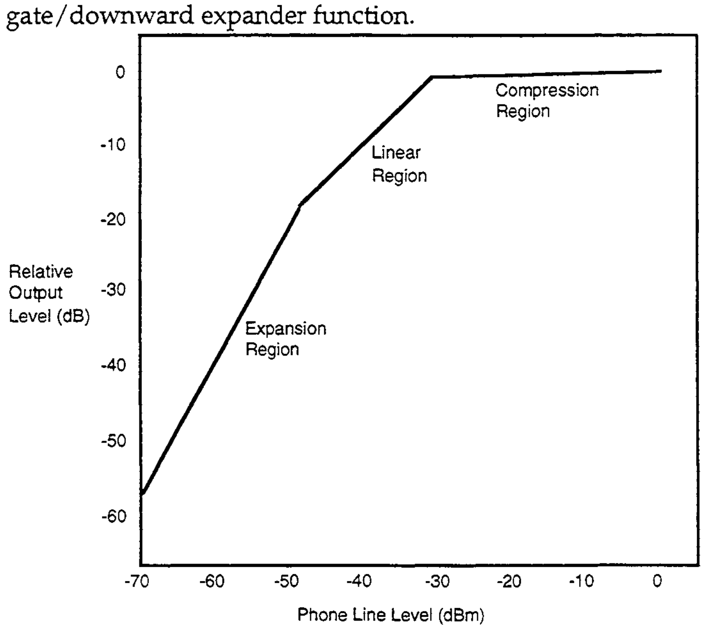
This function operates in the send audio path. The acoustic ducker works by reducing the send (announcer) signal dynamically when the caller speaks.

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.



This graph shows the relationship between telco line level and the Telos ONE's output level. The AGC maintains a constant output level when the phone line level varies 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

The meter mode pushbuttons select the desired function:

FUNCTION	GR/LEVEL SW	IN/OUT SW
• Input level	OUT	IN
• Input gain reduction	IN	IN
• Output level	OUT	OUT
• Output gain reduction	IN	OUT

The input level metering is after the input gain control and displays your mix-minus level to the hybrid.

The output level meter is placed before the output gain control. This is done so that the level may be adjusted to accommodate the equipment downstream of the hybrid without affecting the Telos' meter level. Note that when the AGC 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 "0" indication corresponds to a phone line level of approximately -15dBm.

When viewing input gain reduction the meter will only have one or two LEDs lit. As the meter moves to the left, it indicates input gain reduction; when it moves to the right, it shows input gain expansion. Input AGC is always active.

The output gain reduction meter is only active when the AGC button is pressed in, since this button turns the caller AGC on and off. Its display functions just like the input gain reduction meter.

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-20's 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.

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Tutorial

ELECTRONIC PHONE SYSTEMS

Electronic Phone Systems

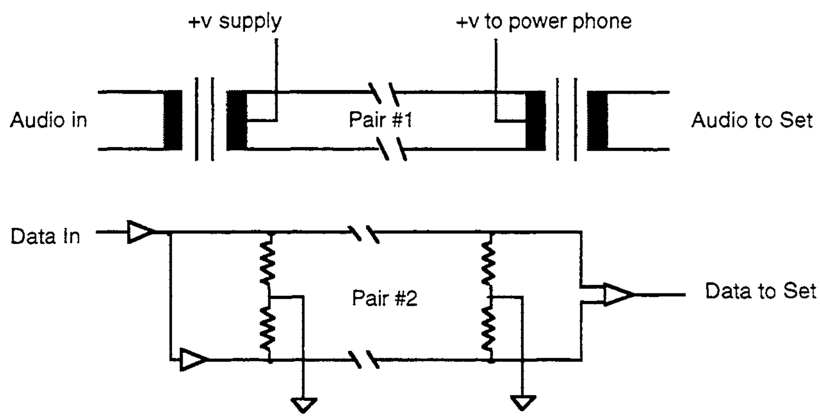
This tutorial introduction to electronic phone systems may be helpful if you desire to connect a Telos hybrid to such systems.

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.

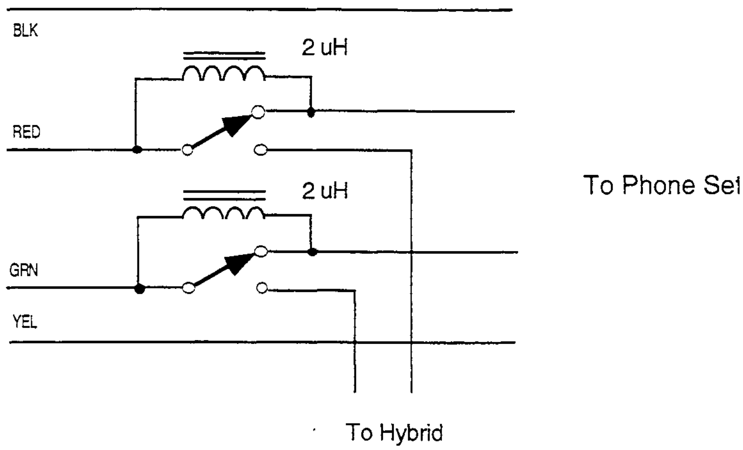
The most common approach used in new designs is a two pair, four-wire scheme. In this scheme, the talk and data are each balanced and each use one of the two pairs. The power is "phantom" applied between the two pairs in a way similar to the method used for phantom powering condenser microphones in recording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

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 units available for fast overnight shipping. In most cases, we will swap units with you at no cost. In the ten 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 happnin' inside the fancy box! So we tell you.

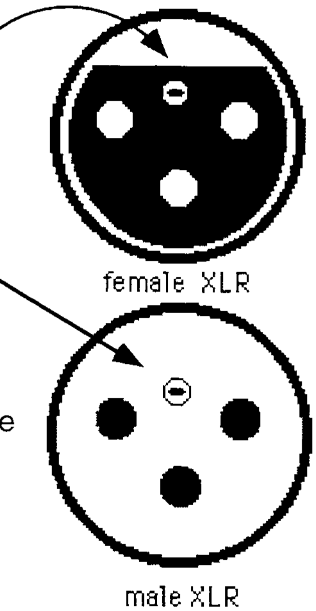
General Troubleshooting Information

Access to the PC Boards

1) Remove the top cover first by unscrewing the allen-head screws, using a 1/16" allen key.
2) To remove the rear panel board, release the XLRs as shown in the diagram. Remove the screws holding the circuit board in place. Disconnect all flat-ribbon cables. The board should now slip out of its case by gently pushing on the XLRs and pulling the front of the circuit board.
3) To remove the hybrid boards remove the screws holding the circuit board in place. Disconnect all flat-ribbon cables. The board should now slip out of its case by gently pulling on the rear of the circuit board.

NEUTRIK XLR REMOVAL

If, for any reason, you must remove the printed circuit board from its enclosure, you must first release the XLR connectors from their housings. To do this simply insert a small screwdriver in the holes in the connectors, shown at right. Turn the screwdriver about one eighth of a turn counterclockwise to release the connectors. (A small screwdriver such as the Xcelite R3322 or R3324 may need to be filed down some to fit in the slots.) Remember to retighten the XLR latches when replacing the circuit board. This will ensure correct support for the XLR connectors on the PCB.



CAUTION

The installation and servicing instructions in this manual are for use by qualified 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.

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 & I/O 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 anyof them are,then 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 5532's - *it is normal for these to run hot.*

4.4 POWER SUPPLY

The Telos ONE plus ONE unit has a built-in, universal AC-input switching supply which has a single +12VDC output (see the appendix for more information on the switching supply). The hybrid circuitry requires three power supply rails, each being generated on the PCB locally:

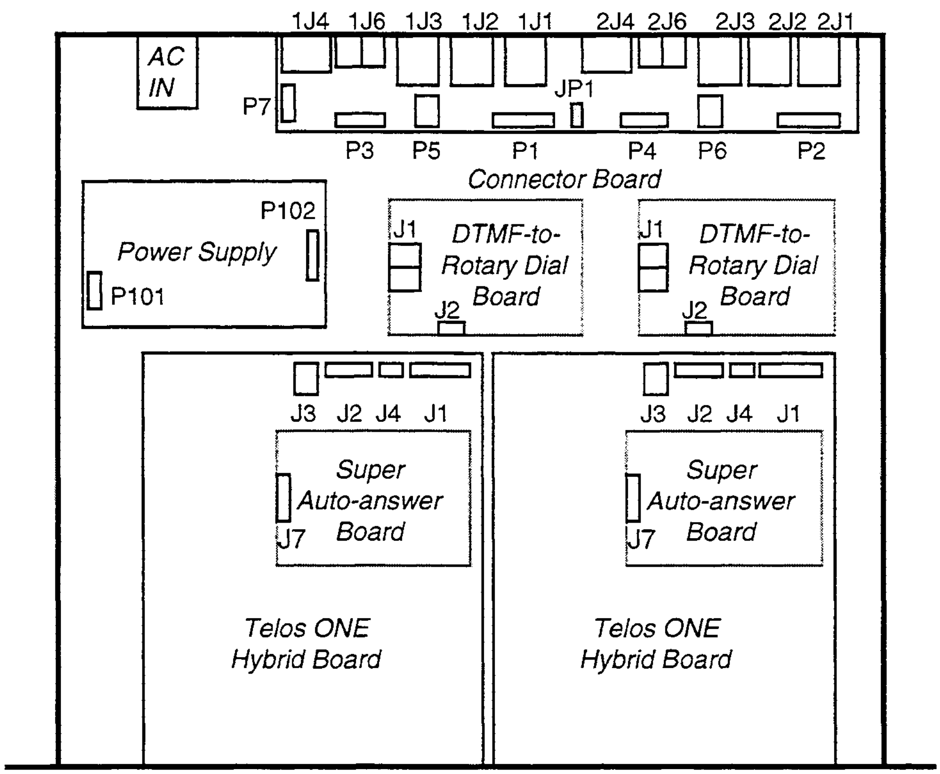
+5v DIGITAL: Powers the processor and other digital ICs. Regulated by three-terminal regulator VR3. A *gold cap*, C3, holds up the power voltage in the event of a brief line voltage dip.

+5v ANALOG: Powers 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

A stock Telos ONE plus ONE unit has four circuit boards inside: two Telos ONE hybrid boards, a rear panel connector board and the power supply. Optional circuit boards (the auto answer boards and the DTMF-to-rotary dial board) are shown in gray outlines.



Circuit board arrangement inside the Telos ONE plus ONE. Grayed out boxes indicate optional circuit boards.

Similarly sized connectors (represented by boxes in the diagram above) connect to each other. Here is a brief description of what signals are carried by each connector. Pinouts for the hybrid connectors follow the descriptions. Refer to the schematics for details on the other connectors.

HYBRID BOARD:

J1: 20-pin header. Has the audio signals to and from the rear panel XLRs (1J1,1J2,1J3,2J1,2J2,2J3), and brings +12VDC/ground from the power supply via the connector board's P7.
Connects to P1 or P2 on the connector board.

J2: 16-pin header. Routes hybrid control signals to and from 1J4 or 2J4 of the rear panel. (See the chart at the end of Sec. 2 for pinouts.)
Connects to P3 or P4 of the connector board.
Not used if Super Auto-answer card is installed.

J3: Female RJ11C. Has the tip/ring signals from 1J6 and 2J6.
Connects to either P5 or P6 of the connector board.
If a DTMF-to-Rotary Dial board is used, then J3 will connect to the frontmost RJ11 of J1 on the rotary board and the rotary board's rearmost RJ11 connects to P5 or P6 on the connector board. Said another way, the rotary board is in series with the phone line via its J1 connector.

J4: 6-pin header. Has send audio, the "hybrid ON" control signal and +12VDC/ground.
Connects to J2 of the DTMF-to-Rotary Dial board and therefore is only used when a DTMF-to-Rotary Dial board is installed.

CONNECTOR BOARD:

P1, P2: 20-pin header. Contains the audio signals of the rear-panel XLRs (1J1,1J2,1J3,2J1,2J2,2J3). Also has +12VDC/ground from P7.
Connects to J1 on the appropriate hybrid board.

P3, P4: 16-pin header. Can have either hybrid control signals or Super Auto-answer board signals, depending on which board is connected to it. Routes these signals to the DB15s (1J4/2J4) on the rear panel.

P5,P6: Female RJ11 jack. Has the telco tip/ring from the rear panel's dual RJ11 LINE and PHONE jack. Usually connects to J3 on the hybrid board but must connect to the DTMF-to-Rotary Dial board's rearmost RJ11 jack, if one is installed.

P7: 4-pin Molex. +12VDC and ground from power supply

1J1, 2J1: 3-pin female XLR. Input (send) audio to the hybrid.

1J2, 2J2: 3-pin male XLR. AUX/MIX output. See Sec. 2 to configure this output.

1J3, 2J3: 3-pin male XLR. Caller-only output.

1J4, 2J4: 15-pin female D-sub connector. Remote control functions for hybrid or Super Auto-answer board.

1J6, 2J6: Dual RJ11C connector. LINE connector for telco input; PHONE connector for tip/ring loop thru.

POWER SUPPLY:

P101: 4-pin Molex connector. AC input for the supply.
Connects to the rear panel IEC connector/on-off switch.
P102: 6-pin Molex connector. DC output of the power supply.
Connects to P7 on the connector board.

SUPER AUTO-ANSWER BOARD:

J7: 16-pin header. Has all the control signals for this board.
When installed will connect to P3/P4 on the connector board.

DTMF-TO-ROTARY BOARD:

J1: Dual female RJ11 connector. The rearmost RJ11 brings the tip/ring signals from the rear panel LINE and PHONE jacks to the pulse dial circuitry. The frontmost jack then passes the tip/ring signals to the hybrid board's J3.
J2: 6-pin header. Has the hybrid's ON control line, send audio for the tone detector and +12VDC/ground. Must connect to the hybrid's J4 connector.

HYBRID BOARD J1	
Pin No.	**Signal**
1	AUTO front panel button
2	AUTO front panel LED
3	+12VDC
4	+12VDC
5	Ground
6	Ground
7	XLR pin 2 OUT (-)
8	n/c
9	XLR pin 1 OUT AGround
10	XLR pin 3 OUT (+)
11	XLR pin 3 AUX/MIX (+)
12	XLR pin 2 AUX/MIX (-)
13	XLR pin 3 INPUT (+)
14	XLR pin 1 AUX/MIX AGround
15	XLR pin 1 INPUT AGround
16	XLR pin 2 INPUT (-)
17	Mix-Minus summing node
18	INPUT summing node
19	Output to mix-minus JP1
20	Input to mix-minus JP1

HYBRID BOARD J2	
Pin No.	**Signal**
1	n/c
2	AUTO input for SAA
3	hybrid ON control
4	hybrid OFF control
5	DGround
6	+5VDC
7	relay wiper
8	n/c
9	AUTO status from SAA
10	n/c
11	n/c
12	n/c
13	n/c
14	n/c
15	normally closed relay contact
16	normally open relay contact

HYBRID BOARD J3	
Pin No.	**Signal**
1	n/c
2	RING loop-thru
3	RING input
4	TIP input
5	TIP loop-thru
6	n/c

123456
typical pin numbering for RJ11C

HYBRID BOARD J4	
Pin No.	**Signal**
1	hybrid ON control
2	SEND audio
3	DGround
4	DGround
5	+12VDC
6	+12VDC

indicates pin 1

1 3 5
2 4 6

typical pin numbering for headers

SECTION 5
DRAWINGS

DRAWINGS:

Parts List
Hybrid Signal Flow Block Diagram
TMS320C25 Pinouts
MPU/power supply circuits schematic
Audio & Telco circuits schematic
Connector Board schematic

Telos ONE+ONE Hybrid Board

PARTS LIST

Designation	Description	Designation	Description
R1	100Ω	C1	2.2µF/25V tant
R2	100Ω	C2	2.2µF/25V tant
R3	10KΩ	C3	0.1 Farad Gold Cap
R4	1KΩ	C4	22pF mono
R5	560Ω	C5	2000pF mono
R6	10KΩ	C6	2000pF mono
R7	10KΩ	C7	0.01µF mono
R8	10KΩ	C8	2.2µF/25V tant
R9	100Ω	C9	2.2µF/25V tant
R10	33KΩ	C10	2.2µF/25V tant
R11	10KΩ	C11	100µF/35V electrolytic
R12	10KΩ	C12	1000µF/35V electrolytic
R13	1KΩ/1%	C13	10µF/25V tant
R14	1KΩ/1%	C14	0.1µF mono
R15	1KΩ/1%	C15	2.2µF/25V tant
R16	1KΩ/1%	C16	2.2µF/25V tant
R17	100KΩ/1%	C17	2.2µF/25V tant
R18	100KΩ/1%	C18	0.1µF mono
R19	1KΩ	C19	0.1µF mono
R20	2.4KΩ	C20	0.1µF mono
R21	10KΩ	C21	0.1µF mono
R22	49.9Ω/1%	C22	0.001µF mono
R23	49.9Ω/1%	C23	*for international use*
R24	10KΩ	C24	*for international use*
R25	56KΩ	C25	*for international use*
R26	1KΩ	C26	not stuffed
R27	15KΩ	RP1	2.2KΩ x10SIP
R28	49.9Ω/1%	RP2	330Ωx10 SIP
R29	49.9Ω/1%		
R30	10KΩ	RF1,2,3	RF filter: TDK Z7K51R1-05
R31	10KΩ		
R32	10KΩ	L1-5	Green LED LTL-1234A
R33	10KΩ	L6	Red LED LTL-1224A
R34	39KΩ	L7,8	Yellow LED LTL-1254A
R35	270KΩ		
R36	*for international use*	IND1	100µH toroid
R37	*for international use*		
R38	*for international use*		
R39	150KΩ	K1,2	5V relay: Omron G5V-2-H

Telos ONE+ONE Hybrid Board

PARTS LIST (con't.)

Designation	Description
U1	TI 320C25 MPU
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	D2912A filter
U15,16,17	NE5532 dual op-amp
U18	MAX636 switching inverter
U1S	68 pin PGA socket
U2S,3S	24 pin IC socket
U12S,13S	28-pin IC socket
U14S	16-pin IC socket
SW1,2,3,4	Alt action switch
SW5,6	Mom switch
SW7	Two pos DIP sw
SW8,9	Alt action switch

Designation	Description
T1	Prem SPT-195 xfmr
Q1	2N2222
J1	20-pin 0.1" header w/shroud
J2	16-pin 0.1" header w/shroud
J3	Mod. phone receptacle
J4	6-pin 0.1" header w/shroud
HDR1	2x4 pin 0.1" header
HDR2	1x4 pin 0.1" header
HDR3	2x3 pin 0.1" header
HDR4	1x3 pin 0.1" header
P1,2	10K rt angle multiturn trimpot (Bournes 3266-X-103)
Heatsink	TO-220: AAVID 5073B
D1	1N4730 3.9v zener
D2	1N4730 3.9v zener
D3	1N5818
D4	1N4004
D5	1N4148
D6	1N4148
Z1	250V MOV
VR1	LM7905CT -5v reg
VR2,3	LM7805 +5v reg

Telos ONE+ONE Connector Board

PARTS LIST

Designation	Description
P1,P2	20-pin 0.1" header w/shroud
P3,P4	16-pin 0.1" header w/shroud
P5,P6	Mod. phone receptacle
P7	4-pin 0.156" Molex header

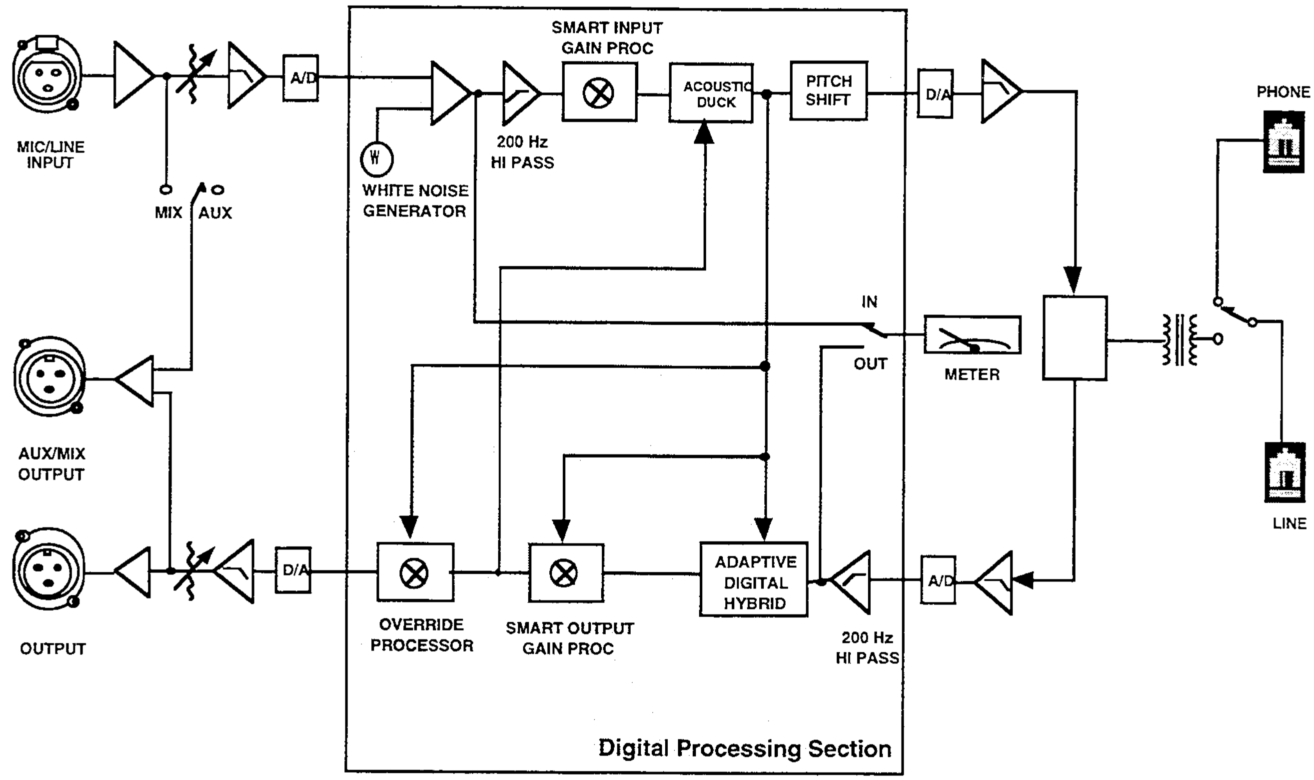
Designation	Description
1J1,2J1	XLR female
1J2,1J3,2J1,2J3	XLR male
1J4,2J4	DB15 female
1J6,2J6	Dual mod phone receptacle

ITEMS INCLUDED WITH UNIT:

qty 2 6' Modular Phone Cord
qty 2 DB15 male w/ hoods
Small screwdriver, 1/16" allen key
Manual, Enclosure
Carton/Packing Foam

MISCELLANEOUS PARTS:

qty 2 20-pin flat ribbon ass'y
qty 2 16-pin flat ribbon ass'y
qty 2 RJ11 cable
qty 1 4-cond. Molex cable ass'y
qty 1 AC receptacle w/on-off sw
Internal Power Supply: Autec UPS40-2121



Telos SYSTEMS

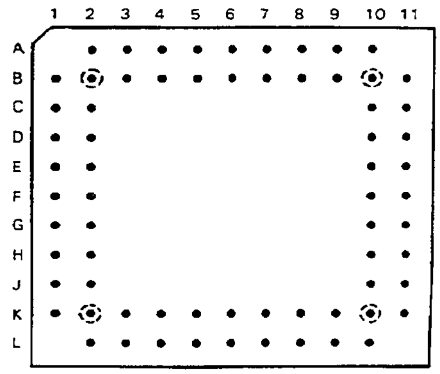
Telos ONE

Signal Flow Block Diagram

PIN ASSIGNMENTS

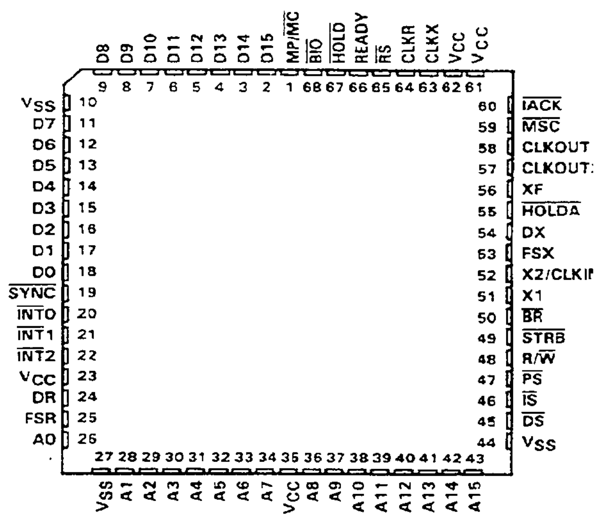
PIN	FUNCTION	PIN	FUNCTION	PIN	FUNCTION
A2	D8	C11	CLKOUT1	J10	$\overline{\mathrm{PS}}$
A3	D10	D1	D4	J11	$\overline{\mathrm{IS}}$
A4	D12	D2	D3	K1	A0
A5	D14	D10	CLKOUT2	K2	A1
A6	MP/$\overline{\mathrm{MC}}$	D11	XF	K3	A3
A7	$\overline{\mathrm{HOLD}}$	E1	D2	K4	A5
A8	$\overline{\mathrm{RS}}$	E2	D1	K5	A7
A9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

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

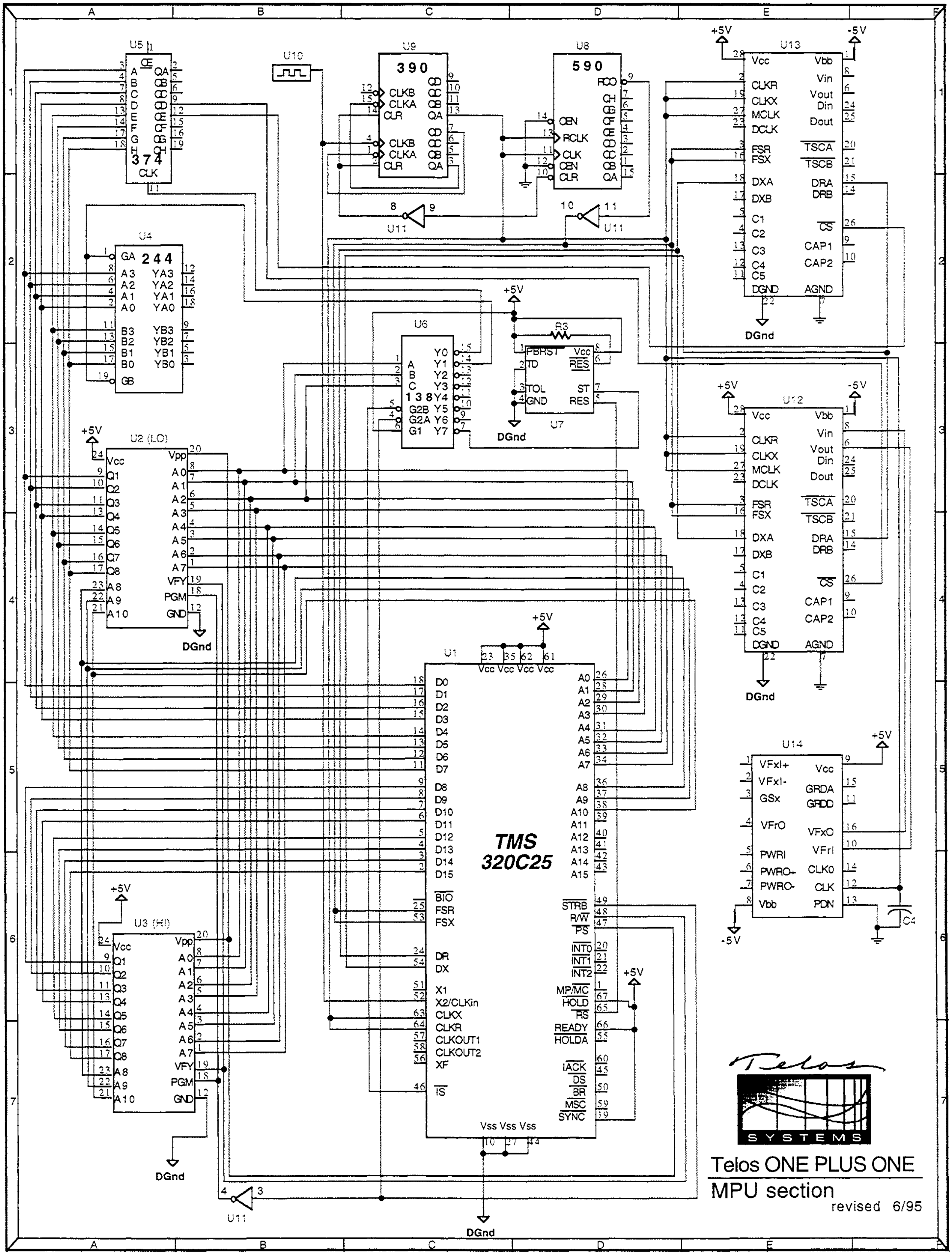


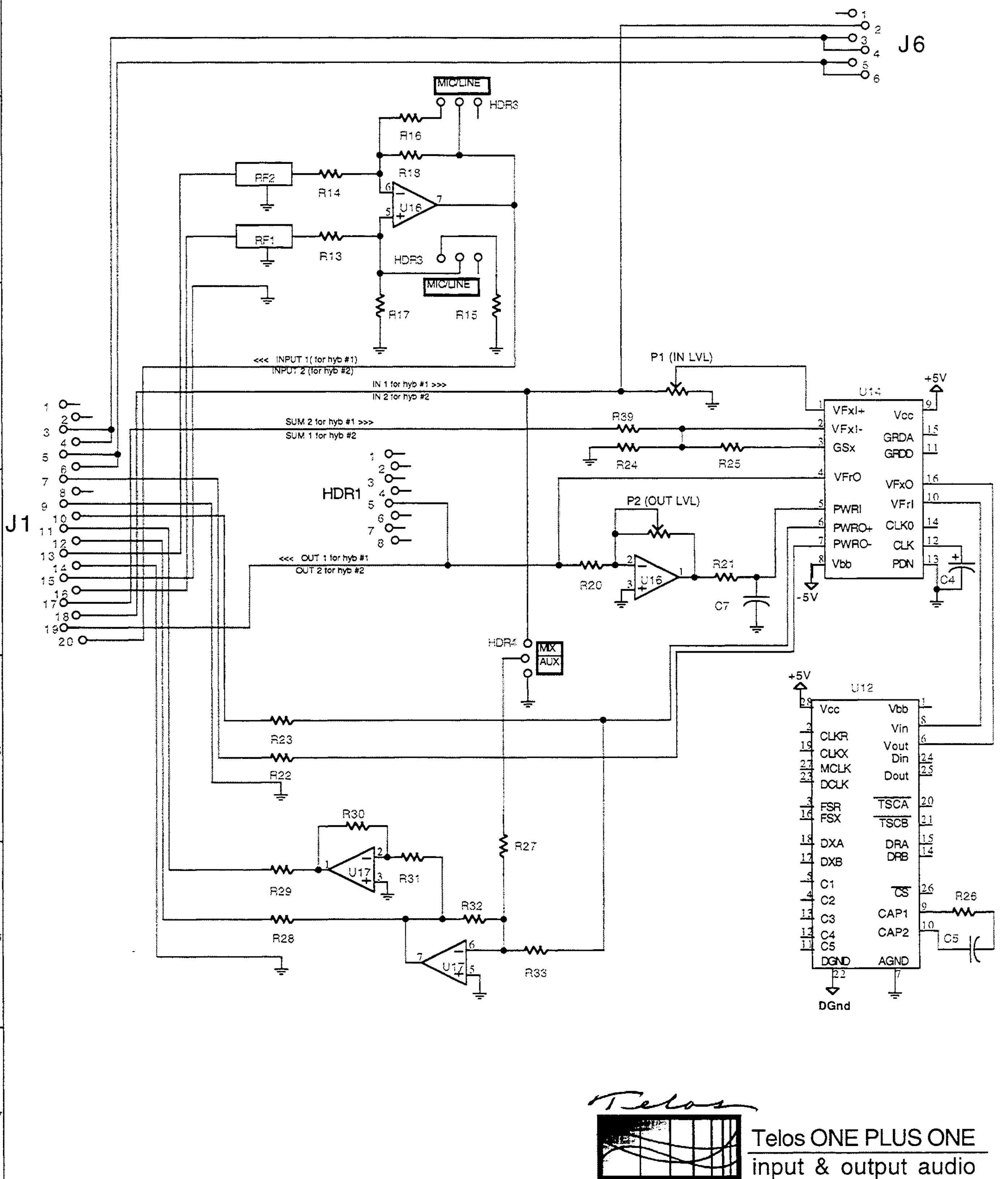
† See Pin Assignments Table (Page 1) and Pin Nomenclature Table (Page 2) for location and description of all pins.

TEXAS INSTRUMENTS



Telos ONE
TMS320C25 pinouts

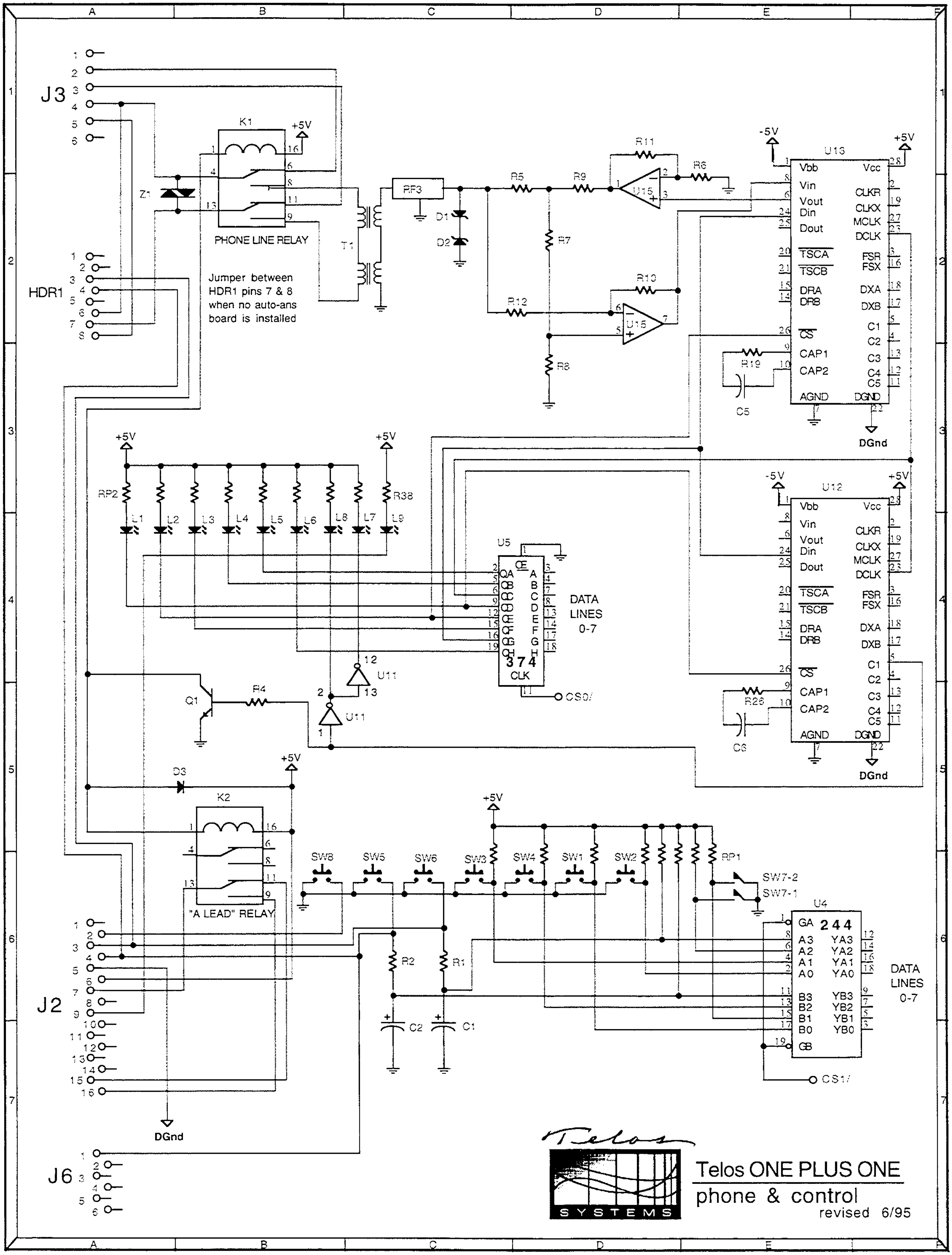

U5
374
U10
U9
390
U8
590
U13
U4
244
U6
138
U7
R3
U2 (LO)
U12
U1
TMS
320C25
U14
C4
U3 (HI)
U11
+5V
-5V
DGnd
Telos ONE PLUS ONE
MPU section
revised 6/95

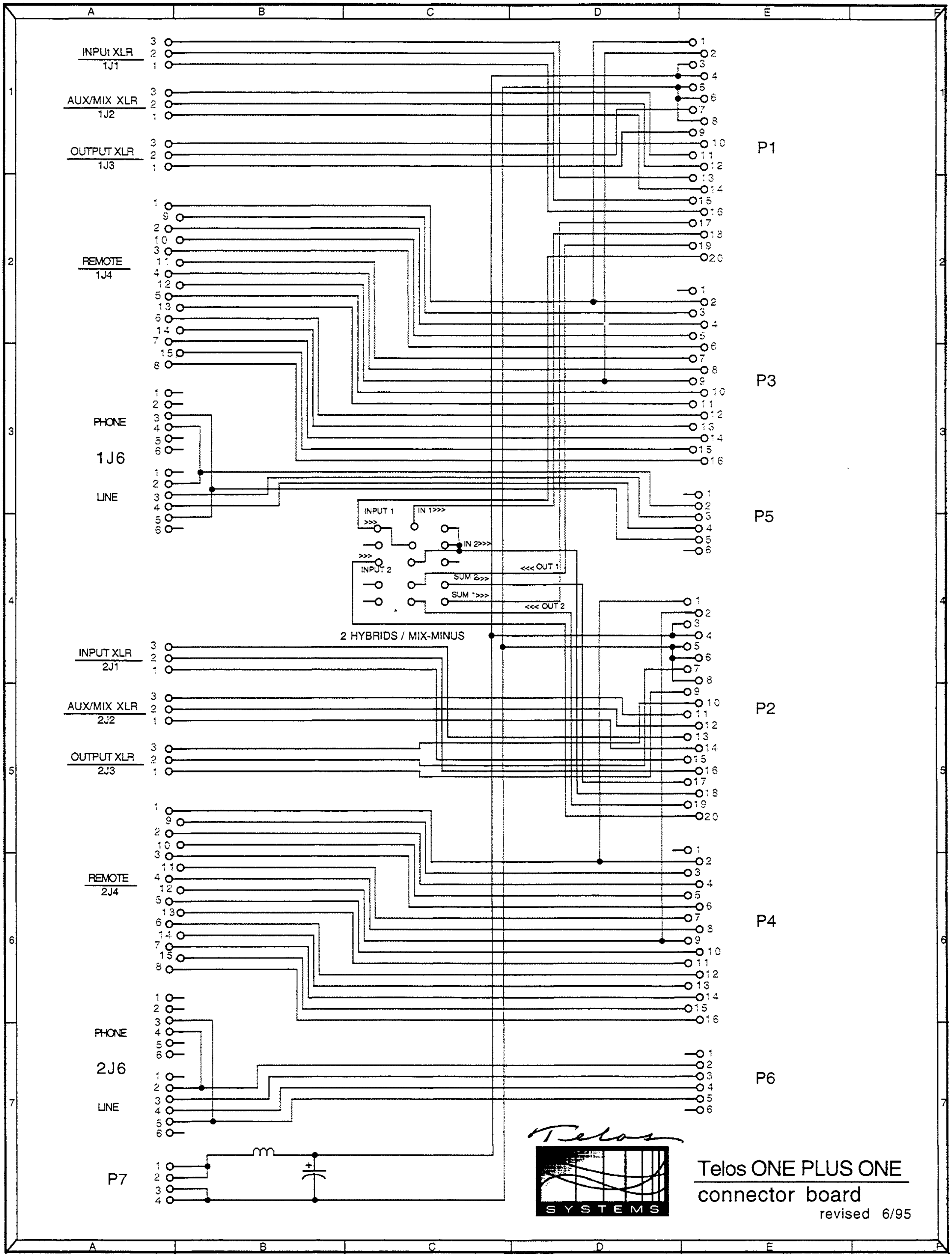


Telos ONE PLUS ONE

input & output audio

revised 6/95


J3
HDR1
Jumper between HDR1 pins 7 & 8 when no auto-ans board is installed
K1
PHONE LINE RELAY
Z1
T1
RF3
D1
D2
R5
R9
R11
R6
R7
R8
R10
R12
U15
U13
U12
Vbb
Vcc
Vin
Vout
Din
Dout
CLKR
CLKX
MCLK
DCLK
TSCA
TSCB
FSR
FSX
DRA
DRB
DXA
DXB
CS
CAP1
CAP2
AGND
DGND
C1
C2
C3
C4
C5
R19
R26
C6
DGnd
+5V
-5V
RP2
R38
L1
L2
L3
L4
L5
L6
L8
L7
L9
U5
374
CLK
CS0/
DATA LINES 0-7
Q1
R4
U11
D3
K2
"A LEAD" RELAY
SW8
SW5
SW6
SW3
SW4
SW1
SW2
RP1
SW7-2
SW7-1
U4
244
GA
GB
A3
A2
A1
A0
B3
B2
B1
B0
YA3
YA2
YA1
YA0
YB3
YB2
YB1
YB0
CS1/
R2
R1
C2
C1
J2
J6
Telos
SYSTEMS
Telos ONE PLUS ONE
phone & control
revised 6/95



INPUt XLR
1J1
AUX/MIX XLR
1J2
OUTPUT XLR
1J3
REMOTE
1J4
PHONE
1J6
LINE
P1
P3
P5
INPUT 1
IN 1>>>
IN 2>>>
INPUT 2
<<< OUT 1
SUM 2>>>
SUM 1>>>
<<< OUT 2
2 HYBRIDS / MIX-MINUS
INPUT XLR
2J1
AUX/MIX XLR
2J2
OUTPUT XLR
2J3
REMOTE
2J4
PHONE
2J6
LINE
P2
P4
P6
P7
SYSTEMS
Telos ONE PLUS ONE
connector board
revised 6/95

SECTION 6

APPENDIX

WARRANTY and APPLICATION CAUTION

This Warranty covers "the Products," which are defined as the various audio equipment, parts, software and accessories manufactured, sold and/or distributed by TLS Corporation, d/b/a Telos Systems (hereinafter "Telos Systems").

With the exception of software-only items, the Products are warranted to be free from defects in material and workmanship for a period of one year from the date of receipt by the end-user. Software-only items are warranted to be free from defects in material and workmanship for a period of 90 days from the date of receipt by the end-user.

The terms and conditions of Telos Systems' warranty in effect at the time of shipment shall apply.

In order to invoke this Warranty, notice of a warranty claim must be received by Telos Systems within the above-stated warranty period and warranty coverage must be authorized by Telos Systems. Notice of a warranty claim may be made orally by telephoning Telos Systems at +1 (216) 241-7225 or in writing sent by facsimile to +1 (216) 241-4103. If Telos Systems authorizes the performance of warranty service and if Telos Systems will be performing the warranty service, the defective Product must be delivered, shipping prepaid, to: Telos Systems, 2101 Superior Avenue, Cleveland, Ohio 44114, USA. If Telos Systems authorizes the performance of warranty service and if it authorizes another entity to perform that warranty service, the Product must be delivered, shipping prepaid, to that entity, whose address will be provided by Telos Systems.

Telos Systems (or its designee) at its option will either repair or replace the Product and such action shall be the full extent of Telos Systems' obligation, and buyer's sole remedy, under this Warranty.

After the Product is repaired or replaced, Telos Systems (or its designee) will return it to the party that sent the Product and Telos Systems will pay for the cost of shipping.

Telos Systems will have no responsibility under this Warranty for any Products subject to: Acts of God, including (without limitation) lightning; improper installation or misuse, including (without limitation) the failure to use telephone and power line surge protection devices; accident; neglect or damage.

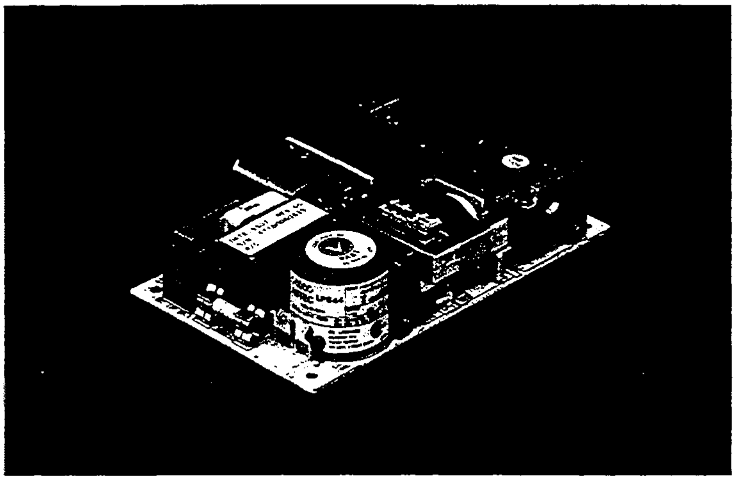
Telos Systems' dealers are not authorized to assume for Telos Systems any additional obligations or liabilities in connection with the dealers' sale of the Products.

EXCEPT FOR THE ABOVE-STATED WARRANTY, TELOS SYSTEMS MAKES NO WARRANTIES, EXPRESS OR IMPLIED (INCLUDING IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE).

In no event will Telos Systems, its employees, agents or authorized dealers be liable for incidental or consequential damages, or for loss, damage, or expense directly or indirectly arising from the use of any Product or the inability to use any Product either separately or in combination with other equipment or materials, or from any other cause.

Telos products are to be used with registered protective interface devices which satisfy regulatory requirements in their country of use.

LPS 40
40 WATTS



LPS Series
40 Watts • Single Output

The LPS Series of power supplies is an AC/DC universal input, single output design offering the latest in high technology performance. This rugged PCB design measures only 3" x 5" and features Class B EMI, high efficiency, and very high reliability. The LPS Series is ideal for telecommunications and computer peripheral applications, test and industrial equipment, medical instrumentation, and business machines.

SPECIAL FEATURES

- Universal input
- High efficiency
- Remote sense
- Built-in EMI filter
- Low output ripple
- Adjustable output
- Overvoltage protection
- Overload protection
- Enclosure kit available (see page 27 or PowerFAX Doc. no 1025)

ENVIRONMENTAL

Operating temperature: 0° to 50°C ambient; derate each output at 2.5% per degree from 50° to 70°C

Electromagnetic susceptibility: designed to meet IEC 801,-2, -3, -4, -5, -6, Level 3

Humidity: Operating; non-condensing 5% to 95%

Vibration: Three orthogonal axes, sweep at 1 oct/min, 5 min. dwell at four major resonances 0.75G peak 5 Hz to 500 Hz, operational

Storage temperature: -40° to 85°C

Temperature coefficient: ± .04% per degree C

MTBF demonstrated: > 550,000 hours at full load and 25°C ambient conditions

ELECTRICAL SPECIFICATIONS

Input

Input range....... 85 VAC to 264 VAC; 120 to 370 VDC
Frequency .... 47 - 440 Hz
Inrush current..... < 18 A peak @ 115 VAC; < 36 A peak @ 230 VAC cold start @ 25°C
Input current...... 1 A max. (RMS) @ 115 VAC
Efficiency .... 70% typical at full load
EMI filter......... FCC Class B conducted, CISPR 22 Class B conducted, EN55022 class B conducted and VDE 0878 PT3 class B conducted
Safety ground leakage current..... < 0.5mA @ 50/60Hz, 264 VAC input

Output

Maximum power... 40 W for convection; 55 W with 30 CFM forced air
Adjustment range.. -5, +10% minimum
Hold-up time...... 20 ms at 40 watt load and 115 VAC nominal line
Overload protection .... Short circuit protection on all outputs. Case overload protected @ 110% to 145% above peak rating
Overvoltage protection .... 5V output: 5.7 to 6.7 VDC. Other outputs 10% to 25% above nominal output

SAFETY

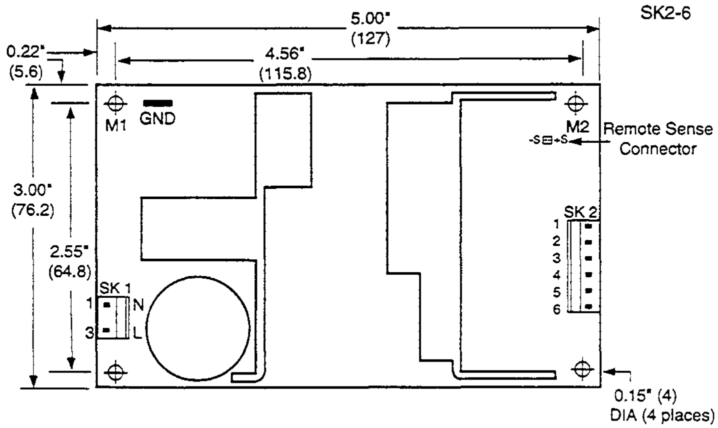
VDE	0805/EN60950 (IEC950)	11774-3336-1241 (LC# 84936)
UL	UL1950	E132002
CSA	CSA 22.2-234 Level 3	LR53982C
NEMKO	EN 60950/EMKO-TUE	P94100375 (74-sec) 203
BABT	EN60950/BS7002	PS/604781
CB	Certificate and report	1119,1125,1126,1127
CE	Mark	

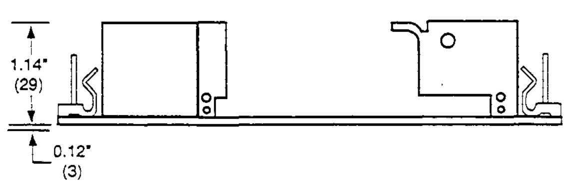
ORDERING INFORMATION

Model Number	Output Voltage	Minimum Load	Maximum Load with Convection Cooling	Maximum Load with 30 CFM Forced Air	Peak Load[1]	Regulation[2]	Ripple P/P (PARD)[3]
LPS41	3.3V	0A	8A	11A	12A	± 2%	33 mV
LPS42	5 V	0 A	8 A	11 A	12 A	± 2%	50 mV
LPS43	12 V	0 A	3.3 A	4.5 A	5 A	± 2%	120 mV
LPS44	15 V	0 A	2.6 A	3.6 A	4 A	± 2%	150 mV
LPS45	24 V	0 A	1.6 A	2.3 A	2.5 A	± 2%	240 mV
LPS48	48V	0A	0.9A	1.2A	1.3A	± 2%	480 mV

1. Peak current lasting < 30 seconds with a maximum 10% duty cycle.
2. At 25°C including initial tolerance, line voltage, load currents and output voltages adjusted to factory settings.
3. Peak-to-peak with 20 MHz bandwidth and 10 μF in parallel with a 0.1 μF capacitor at rated line voltage and load ranges.

DRAWINGS





PIN ASSIGNMENTS

Connector	LPS41	LPS42	LPS43	LPS44	LPS45	LPS48
SK1-1	Neutral	Neutral	Neutral	Neutral	Neutral	Natural
SK1-3	Line	Line	Line	Line	Line	Line
SK2-1	+3.3V	+5 V	+12 V	+15 V	+24 V	+48V
SK2-2	+3.3V	+5 V	+12 V	+15 V	+24 V	+48V
SK2-3	+3.3V	+5 V	+12 V	+15 V	+24 V	+48V
SK2-4	Common	Common	Common	Common	Common	Common
SK2-5	Common	Common	Common	Common	Common	Common
SK2-6	Common	Common	Common	Common	Common	Common

MATING CONNECTORS

AC Input: Molex 09-50-8031 (USA)
09-91-0300 (UK)
PINS: 08-58-0111

DC Outputs: Molex 09-50-8061 (USA)
09-91-0600 (UK)
PINS: 08-58-0111

Remote Sense: 22-01-2025

NOTES

1. Specifications subject to change without notice.
2. All dimensions are in inches and (mm), tolerance is ± .01".
3. Mounting holes M1 and M2 should be grounded for EMI purposes.
4. Mounting hole M1 is safety ground connection.
5. Specifications are for convection rating unless otherwise stated.
6. Warranty: 1 year
7. Weight: 0.5 lb./0.23 kg

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	1MΩ1/4W 5%	VR1	+5V regulator LM7805
R3,4	330Ω 1/4W 5%	BR1,2	bridge rectifier
R5,6,7,8	150Ω 1/4W 5%	L1,2	green LED
R9,10	330Ω 2W 5%	J1,2	dual RJ11 jack
R11,12,13,14	75Ω 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	47Ω 1/4W 5%	SU5,6	6-pin socket BU060Z
R21,22	150Ω 1/8W 5%	SU7,8	14-pin socketBU140Z
R23,24	910Ω 1/4W 5%	SU9	16-pin socket BU160Z
R25,26	220Ω 1/4W 5%		

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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 Office-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

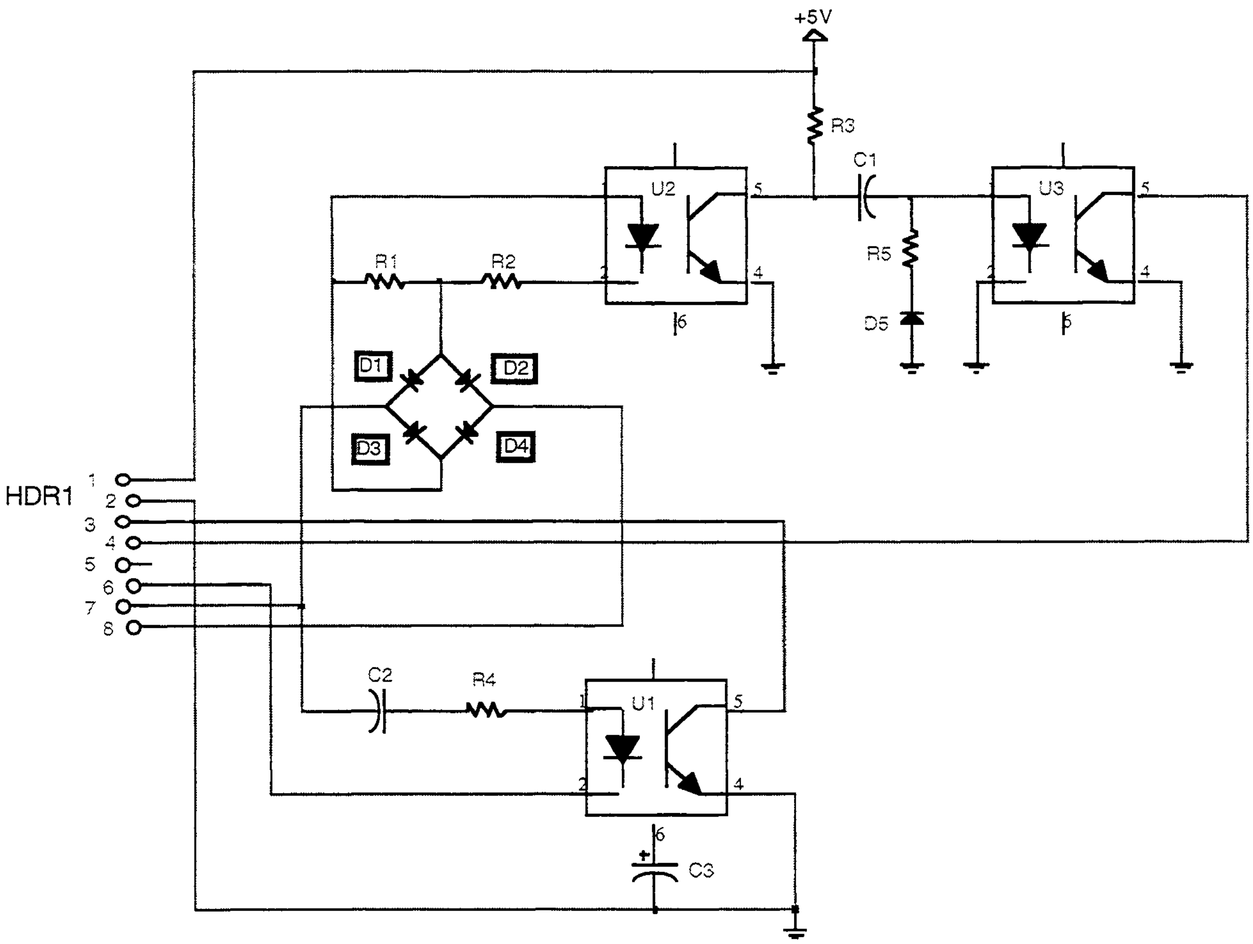
U1, R4, and C2 form the ring detect section. C2 blocks DC so that only the AC ring voltage will trigger U1. When U1's diode gets current, its transistor pulls HR1-2 low turning the hybrid on. C3 prevents false tripping.

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
Re v 3.1

Telos

Super Auto-Answer Board

for use with:
Telos ONE (modem and rack mount versions)
Telos ONE plus ONE
Telos 100 Delta

User's Manual

Version B2 – August 1996

USER'S MANUAL REVISION B1

1.0 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. 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 90VAC at 20Hz. 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:

- Loop-current interruption
- Loop-current 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 PBX use, or originate at central offices with large concentrations of business customers, sometimes use this method. (However, if we may digress, the preferred and more modern situation for PBX control is to use either "ground-start lines" or "E&M signalling." A digression from matters relevant here.)

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 PBXs 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, & 620Hz. When the two tones (350 & 440Hz) 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 hybrid's 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 excellent trans-hybrid loss of Telos' hybrids keep this from happening. Aren't you glad?

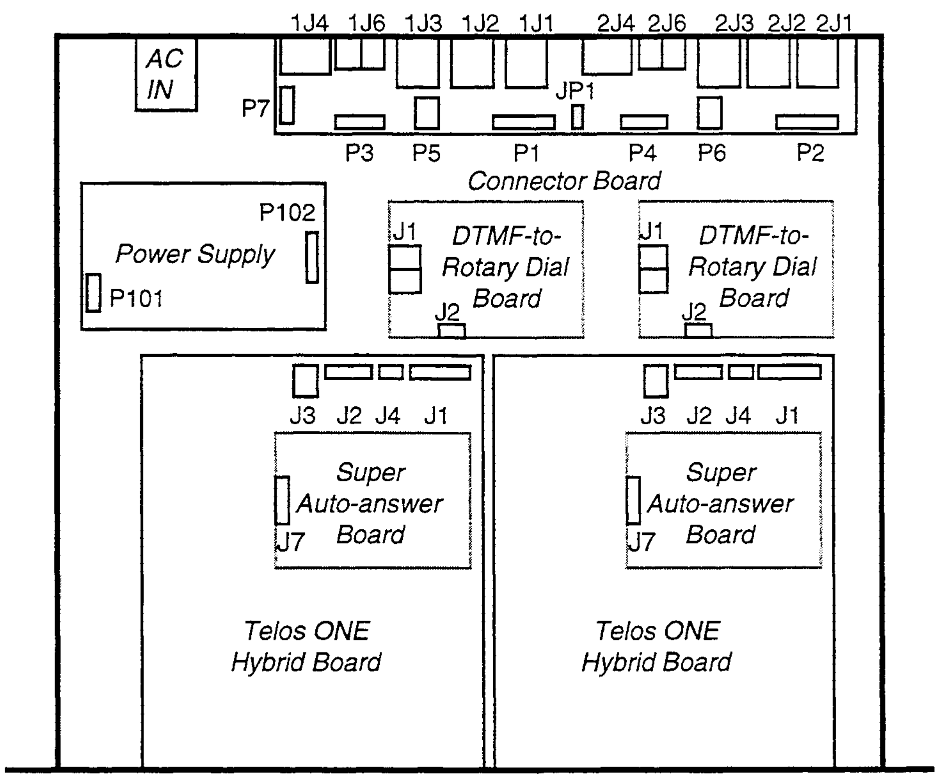
2.0 INSTALLATION

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

The auto-answer board should be removed and the jumpers replaced if use with other than Central Office-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.

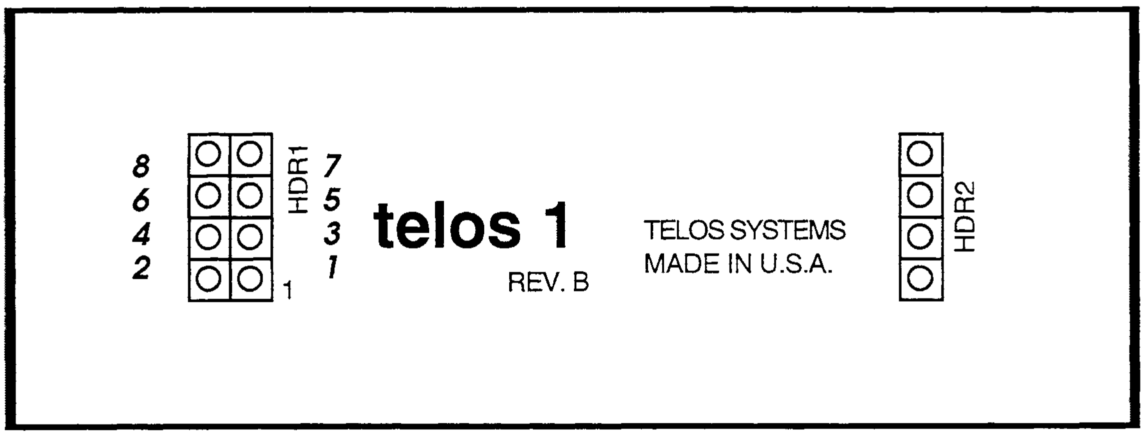
Telos ONE plus ONE

1) Remove the Telos ONE plus ONE's top cover to expose the circuit boards. J2 on each hybrid board connects to the rear panel board's P3 & P4. Lift the ribbon cables out of J2 on each hybrid board.



Circuit board arrangement inside the Telos ONE plus ONE. Grayed out boxes indicate optional circuit boards.

2) The auto answer board mounts on HDR1 and HDR2 near the rear of the unit. 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! Then, tighten the Super Auto-Answer board to the hybrid board where the nylon standoff lines up. (Some units have the nylon standoff already mounted to the hybrid board; others have the standoff mounted to the Super Auto-Answer board, in which case the hybrid boards will have to be removed to install the nylon standoff.)

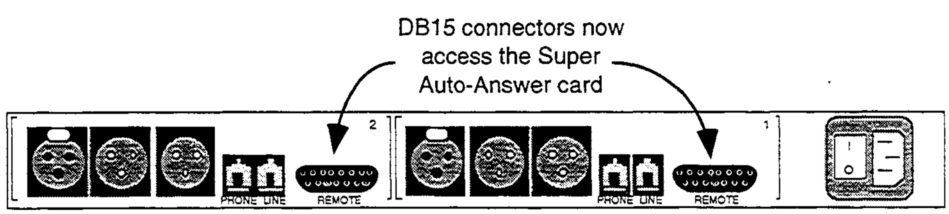


HDR1 & HDR2 locations and pinout

4) Once the Super Auto-answer card(s) are in place, reconnect the ribbon cable from P3/P4 tot he 16-pin header on the Super Auto-answer card.

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

Of course, Telos can install the Super Auto-Answer board for you. Simply call Telos for a Return Authorization Number. Turnaround is usually less than a week.



3.0 OPERATION

3.1 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 number-of-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

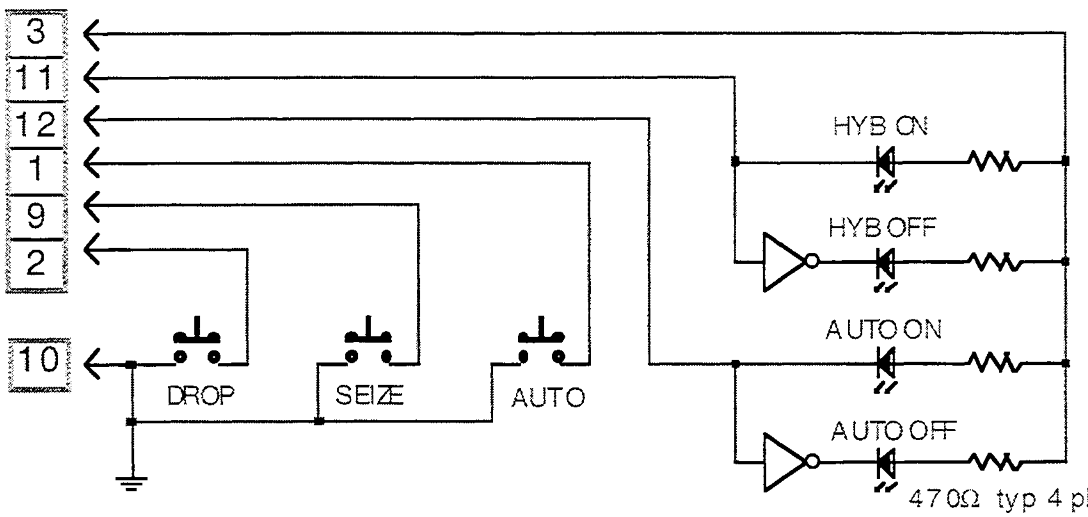
Remote control of the hybrid's ON/OFF functions, as well as the auto mode, are available on the rear panel DB15 connector. Remote buttons are momentary and ground-common. (That is, they are normally pulled high through a resistor and taken to ground to activate.) Refer to the connector pin-out table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

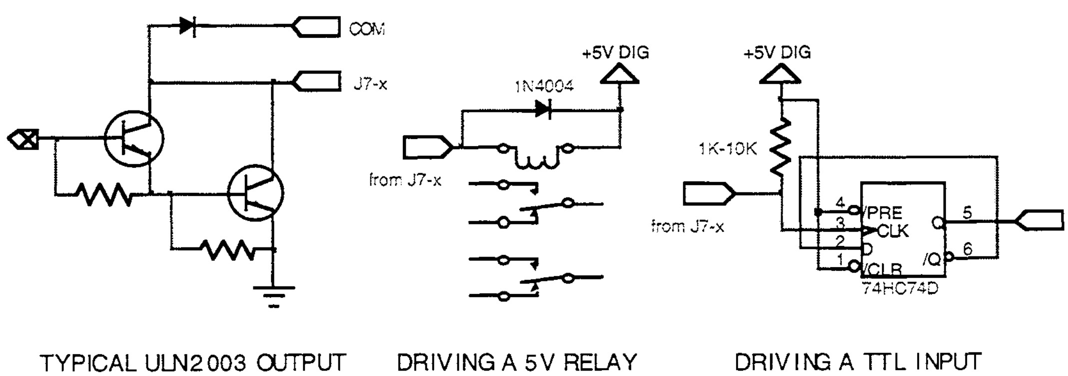
The AUTO input also requires a momentary pushbutton, but is made to be alternate action in operation. Each time the button is pressed, the auto mode will toggle to the opposite state. Power-up is to the "no auto" mode (unless DIP #1 is active).

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 hybrid's 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.



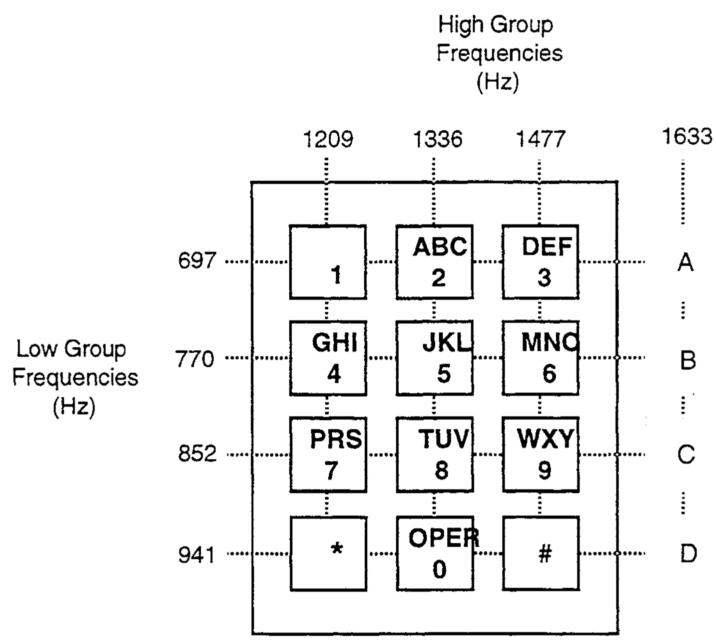
Schematic for remote control. Pin numbers are for the 15-pin D-connector. The inverter should be a 74LS14. The pull-up resistors' value depends on the type of LED used. Check the manufacturer's data sheets for details.



The open-collector outputs for LINE STATUS and AUTO MODE require pull-up resistors to +5VDC. The DTMF decoded outputs already have weak internal pull-ups.

3.2 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 "wire-OR'd" with other boards for special applications.



Frequencies and Buttons for DTMF Generation/Detection

DTMF DECODE TABLE				
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

DB15 PINOUT FUNCTIONS	
Pin no.	Function
1	AUTO control input
2	DROP control input
3	+5VDC
4	D8 DTMF output
5	D4 DTMF output
6	D2 DTMF output
7	D1 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

Charts showing the hexadecimal codes for the touchtone decoder and pin outs for the 15-pin remote control connector.

3.3 ALTERNATE SOFTWARE VERSIONS

Since the Super Auto-Answer board uses a programmable microcontroller, only the software must be changed to alter the board's functions, such as dial tone detection, ring cadences and/or frequency, number of rings to answer on and so forth. Here is a listing of current software versions available. If you have a software version not mentioned here or you have a special or unique application, please call Telos for assistance. Since we are constantly developing new software, we include the latest memos for these at the end of this manual.

OBSOLETE VERSIONS

As of this writing, the following versions have been superceded:
V1.4, V1.41, V1.50, V1.50(61), V3.1, V5.0, V5.1.

AA V1.16

This version is the same as AA V1.1 except that, when DIP#4 is closed, the unit will answer on the sixth ring, instead of the third ring.

AA V1.18

This version is the same as AA V1.1 except that, when DIP#4 is closed, the unit will answer on the eighth ring, instead of the third ring.

AA V1.1 (31)

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

- Ring detection is based on the Netherlands ringing frequency of 25Hz.

- Dial tone release has been changed to detect a reorder tone of 433Hz cadenced at 350ms intervals (i.e. 50% duty cycle).

- Hardware Modifications: To make the Call Progress Tone Decoder chip detect 433Hz its crystal reference frequency must be changed to 2.5MHz. However, the DTMF decoder IC still needs a reference frequency of 3.5795MHz, so an additional crystal must be added to the circuit board on pins 9 & 10 of U7. Refer to the schematic "SuperAutoAns (31) 433Hz detect" for details.

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.

AA V1.2

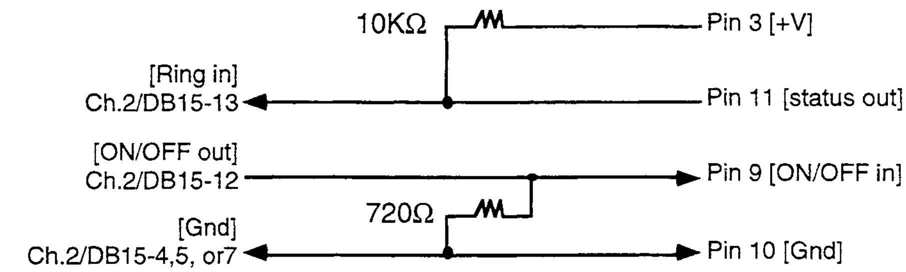
This version of software functions similarly to AA V1.1 with the following differences:

- The SEIZE input no longer disables the AUTO function when pressed.

- 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).

- 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.



*Please refer to intercom system manual for further information.

Figure 1. Clear-Com™ to Super Auto Answer Remote Connector Wiring Diagram

Configuration

The four dip switch functions have been redefined to accommodate intercom system requirements as follows:

DIP #1 "Remote OFF" delay. When on, remote OFF commands are delayed by twelve seconds before being performed. This causes short OFF commands (under 12 seconds) to be ignored allowing the intercom system some "transferring" flexibility. When switch #1 is off, remote OFF commands will be performed immediately.

DIP #2 Enables or disables the intercom system remote control line. When ON, the intercom system control line will be ignored. This switch should be turned OFF to allow remote control of the hybrid functions.

DIP #3 Selects the auto-answer mode. In the on position the hybrid will turn ON with an incoming call, and turn OFF with loop drop or dial/reorder tones. The remote status line will also pulse low (≈1 second) whenever the hybrid is turned ON. When switch #3 is off, the auto answer/disconnect functions are disabled, and ring cadence will be passed (active low) onto the remote status line.

DIP #4 Selects whether or not line status is reflected on the remote status line. If on, the remote status line will go low when the hybrid is off-hook (ON) and high when the hybrid is on-hook (OFF). These dip switch settings are summarized in Table 1.

#	ON/OFF	FUNCTION
1 1	OFF ON	•no delay from remote off->hybrid off •12sec delay from remote off->hybrid off
2 2	OFF ON	•ON/OFF remote input is active •ON/OFF remote input is disabled
3 3	OFF ON	•disable auto answer/disconnect •enable auto answer/disconnect
4 4	OFF ON	•basic remote status (see text) •hybrid on/off status appears on remote out

Table 1. Dip Switch Options

Operation

The hybrid's front panel control, auto answer/disconnect control, and remote control each have an established "priority level," which permits all control sources to be active without conflict. Front panel control has high priority, auto answer/disconnect has medium priority, and remote control has low priority. If the hybrid is activated from the front panel ON button, both remote and auto control are disabled. Once the hybrid is turned off, also from the front panel, remote and auto control are restored. The auto answer/disconnect function can be overridden by front panel control, but not from remote control. Finally, when the remote control is enabled (SW #2=OFF), the telephone hybrid can be turned on with a logic 1 (high) signal applied to the control line input. A logic 0 (low) signal will conversely turn the hybrid off (depending on SW #1 with or without a delay). This control is subject to front panel or auto answer priority as outlined above, which take precedence.

AA V1.42 (supercedes AA V1.4)

This is the software found in the Telex TIF951 unit. It "talks" to the special rear panel Telex processor board, which communicates with the Telex intercom mainframe. The software has the following change: A previously-unused output from the processor (P1.7) goes low whenever the phone line has ring voltage. The Super Auto-Answer merely tells the Telex board that the line is ringing; the Telex board decides when to answer the line. This signal is called "RINGMODE".The number of rings to answer on is set by DIP switch #2, internal to the TIF951 unit. The following chart shows how to set these DIP switches.

No. of Rings	DIP 2-1	DIP 2-2
1	off	off
2	on	off
4	off	on
8	on	on

SAA DIP Switch Options

DIP #1 Not supported.

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. 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 Not supported.

The Telex processor uses the SEIZE, DROP and AUTO inputs as expected. All DTMF signals are brought out the Telex rear board, as well, which use these for password identification, to set up intercom channels and so forth. Refer to the TIF951 manual for more details.

AA V1.42 (61)

This version of software is very similar to the AA V1.42 software, but is intended for use on the telephone system of Australia. Here are the major differences:

- Dial tone release has been changed to differentiate between the various reorder tones applied on the Australian telephone network: 425 Hz only, pulsed at 370 msec on, 370 msec off or 2.5 sec on, 4 sec off. Either three valid, consecutive 370 msec bursts or one 2.5 sec burst of 425 Hz will cause the SAA to release the phone line.

A hardware modification is required to the circuit board so the Call Progress

Detector chip (U5) can detect the non-standard 425Hz signal. Refer to the schematic "SuperAutoAns Rev A (61) 425Hz detect" for details.

AA V1.42 (33)

This version of software is very similar to the AA V1.42 software, but is intended for use on the telephone system of France. Here are the major differences:

• Ring detection is based on the French ringing frequency of 50Hz.

• 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 440Hz tone (instead of the 440Hz + 350Hz used in the US). More precisely, a steady 440Hz tone will not turn the unit off. Rather, a 440Hz tone that is pulsed at a rate of 2 Hz ±10% at 50% duty cycle (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.

No hardware changes are required.

AA V1.42 (886)

This version of software is very similar to the AA V1.42 software, but is intended for use on the telephone system of Taiwan. This version also works well on Northern Telecom PBX analog (2500) ports, as these PBXs apply the same tone cadence. Here are the major differences:

• Ring detection is based on the Taiwan ringing frequency of 50Hz.

• Dial tone release has been changed to release the phone line whenever a tone cadence of 480Hz + 620Hz, 0.25 sec. on, 0.25 sec off is detected. Three consecutive valid tone bursts must be detected for the unit to release the line. DIP SW #3 must be turned on for the SAA board to look for these tones and the unit must be have its auto-answer function activated.

No hardware changes are required.

AA V1.51 (supercedes AA V1.50)

This software version was developed so that the Telos ONE can communicate with Drake intercoms. DIP switch options are the same as in AA V1.1. The normal Drake interface handshaking is summarized below:

- "Seize" input is a maintained closure - low to seize the phone line, high to release; however, this input can still be used as a momentary input, if desired.
- "Auto" input is also maintained - low for auto-answer/disconnect enable, high for disable;
- Dial tone detect window is now 12 seconds, to allow for dialling out; still activated by DIP#3. The unit must have "Auto" enabled for dial tone detect to release the line.
- A previously undocumented output is now available at pin 14 of the DB15 connector. This TTL output will go low if the dial tone detect, current detects or front panel "OFF" button releases the phone line while the "Seize" input is still being held low to indicate to the Drake system to release its "Seize" input. If the unit is NOT in auto mode AND this line is not low already it will blink to indicate the line is ringing.
- "Drop" input now acts like a "system reset" and requires a momentary pulse to ground. It will release the phone line, briefly disable the auto function and make pin 14 (from #4) go low as long as "Drop" is held low.

DRAKE recommends the following hardware changes, as well:

a) Replace U10 with a 74HC244
b) Remove C6, C7, C8 and D9
c) Cut trace between U10 pin 8 and U10 pin 13.
d) Add these jumpers: U10 pin 8 to U10 pin 20; U10 pin 7 to U9 pin 8.

AA V1.51 (61) (supercedes AA V1.50(61))

This software is idetical to AA V1.51, except that this software has been modified for use in Australia. Refer to the section on AA V1.42(61) for details on the Australian requirements. Hardware must undergo the same modifications described for AA V1.42(61), as well as the usual Drake modifications.

AA V1.51 (33)

This software is idetical to AA V1.51, except that this software has been modified for use in France. Refer to the section on AA V1.42(33) for details on the Australian requirements. No hardware are needed, except for the usual Drake modifications.

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 hybrid's 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.

> *NOTE: Not all key phones have the same terminal numbers. A scope or butt set may be helpful to locate the signals. Be sure the hybrid is inserted in series with the common tip/ring AFTER the hook switch.*

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 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 auto-answer 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 hybrid's status as well. Refer to this manual's OPERATION section for a circuit you can build up that will do this.

If Touch Tones are needed to dial out and must be heard on the air, another touchtone pad should be fed into the hybrid's 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!

AA V3.11 (supercedes 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:

- Ring detection is based on the French ringing frequency of 50Hz.

- 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 440Hz tone (instead of the 440Hz + 350Hz used in the US). More precisely, a steady 440Hz tone will not turn the unit off. Rather, a 440Hz tone that is pulsed at a rate of 1/2Hz ±10%(the French reorder tone) will turn the hybrid off. Moreover, a 440Hz tone that is on for 1.5 seconds and off for 3.5 seconds ±10% (another type of reorder tone) will turn the hybrid off, as well.

- The previous version, AA V3.1, sometimes would not pulse the Telos 100 Delta long enough for it to adapt properly. The hybrid on/off pulses have been changed so that, instead of making one long pulse, two shorter pulses are created.

DIP Switch Options

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.

AA V5.2 (supercedes AA V5.0 & V5.1)

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 200ms on/ 200ms. off. Ring frequency is 50Hz.

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). Refer to the schematics "SuperAutoAns before mods" and "SuperAutoAns Italy Modification" for details.

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 hybrid's output is muted during pulse dialing. Furthermore, the Super Auto-answer board's software must not try to turn 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-turn-off" sequence.

DIP Switch Options

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.

AA V5.3

Software to work with ClearCom intercom systems installed in Italy. Refer to the sections on AA V1.3 and AA V5.2 for details.

Telos Super Auto-Answer Board

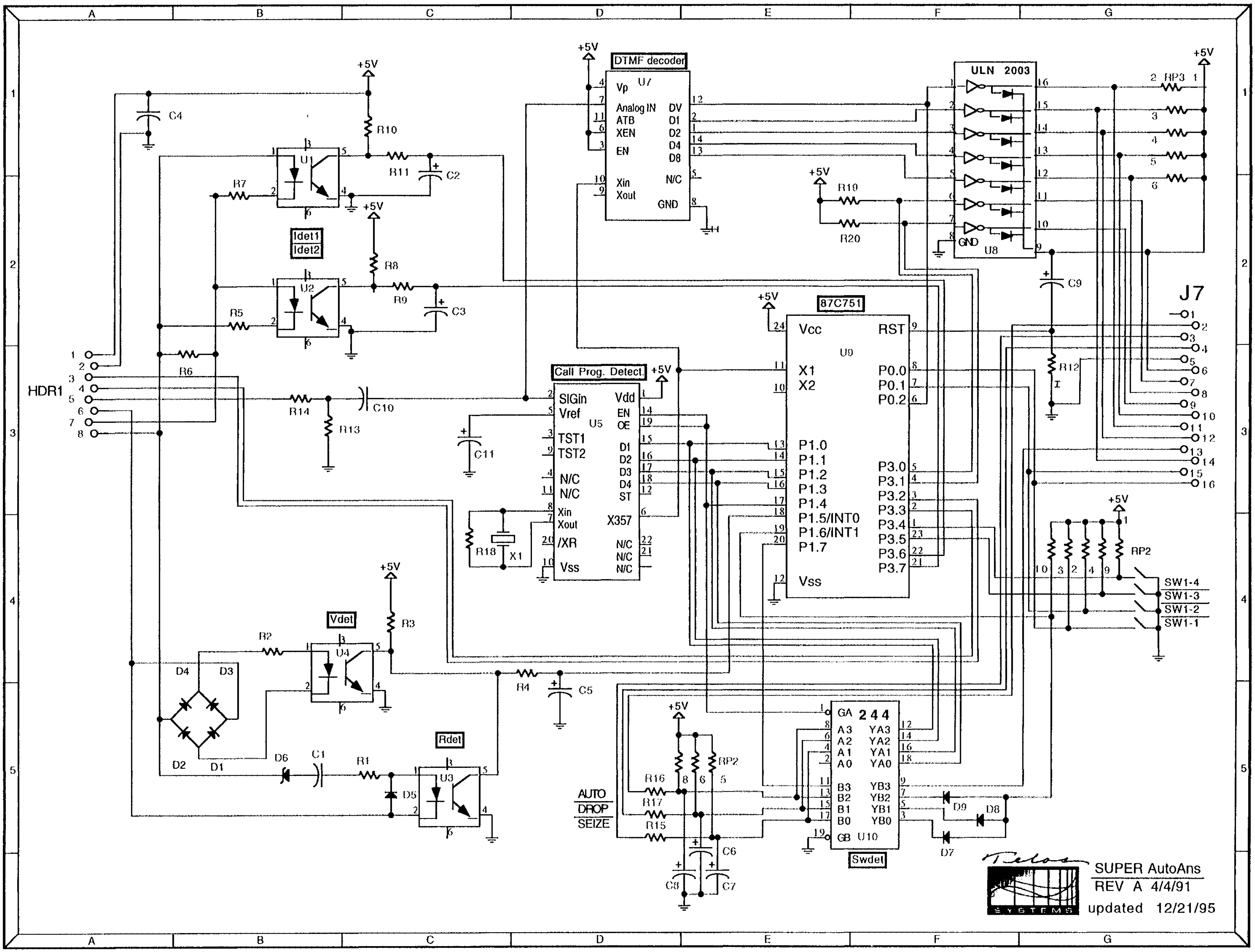
PARTS LIST

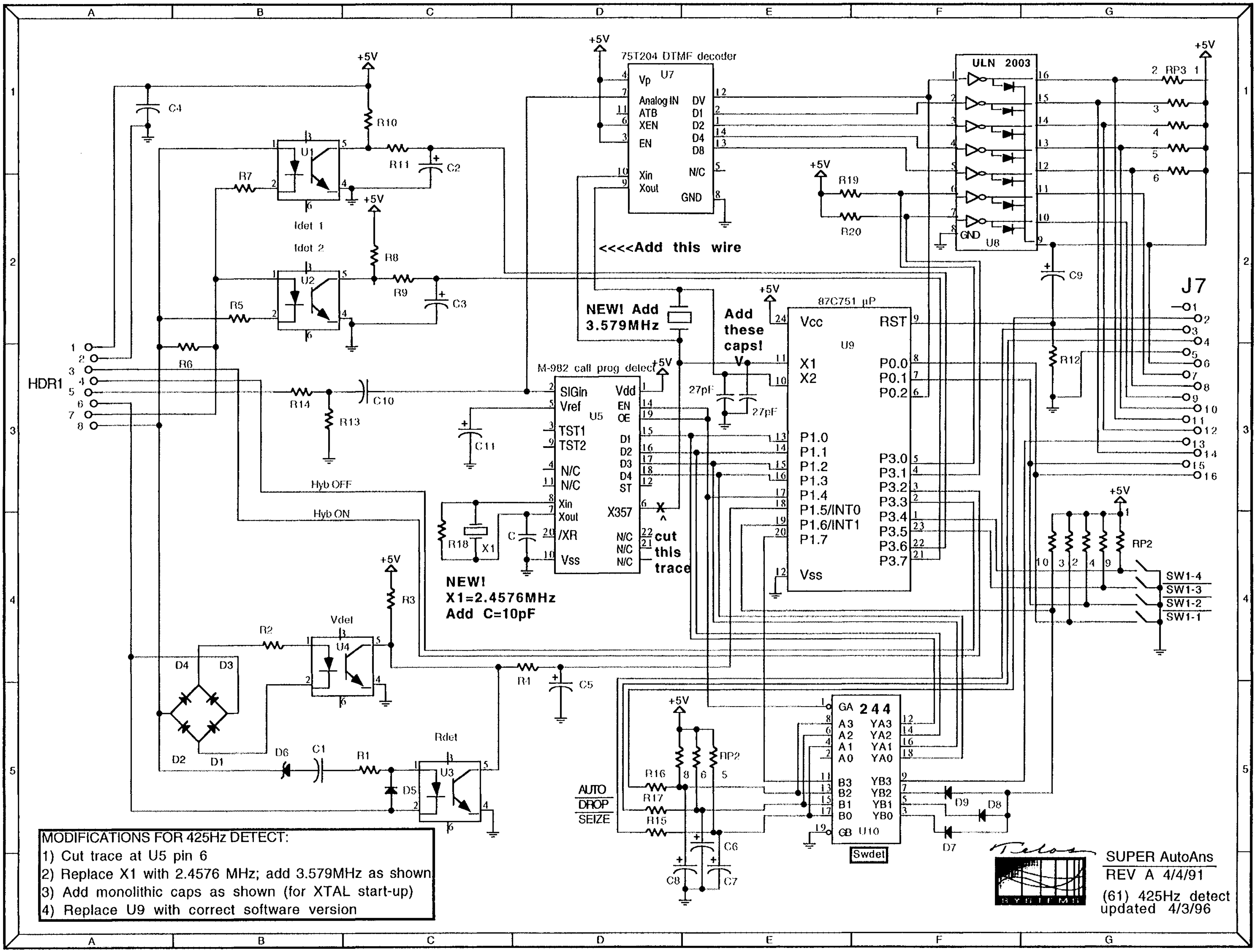
Designation	Description
U1-4	Optoisolator 4N33
U5	Call Prog Detect SSI75T982-CP
U7	DTMF Decoder SSI75T204-IP
U8	Octal open coll. driver ULN2003
U9	Microprocessor 87C751-1N24
U10	Octal buffer 74LS244
RP1	10KΩ SIP 4306R-101-103
RP2	10KΩ SIP 4310R-101-103
R1	2.2KΩ 1/4W 5%
R2	47KΩ 1/4W 5%
R3	100KΩ 1/4W 5%
R4	910Ω 1/4W 5%
R5,7	1KΩ 1/4W 5%
R6	47Ω 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	100Ω 1/4W 5%
R18	10MΩ1/4W 5%
R19,20	1KΩ 1/4W 5%
C1	0.47µF mono
C2,3,6,7,8	2.2µF tant. @ 25V
C4	0.1µF mono
C5	0.33µF tant. @25V
C9:	10.0µF @25V
C10	0.0022µF mono
C11	1.0µF @25V
D1-5	diode 1N4004
D6	Zener diode 1N5246
D7,8,9	signal diode 1N4148
X1	3.58MHz crystal

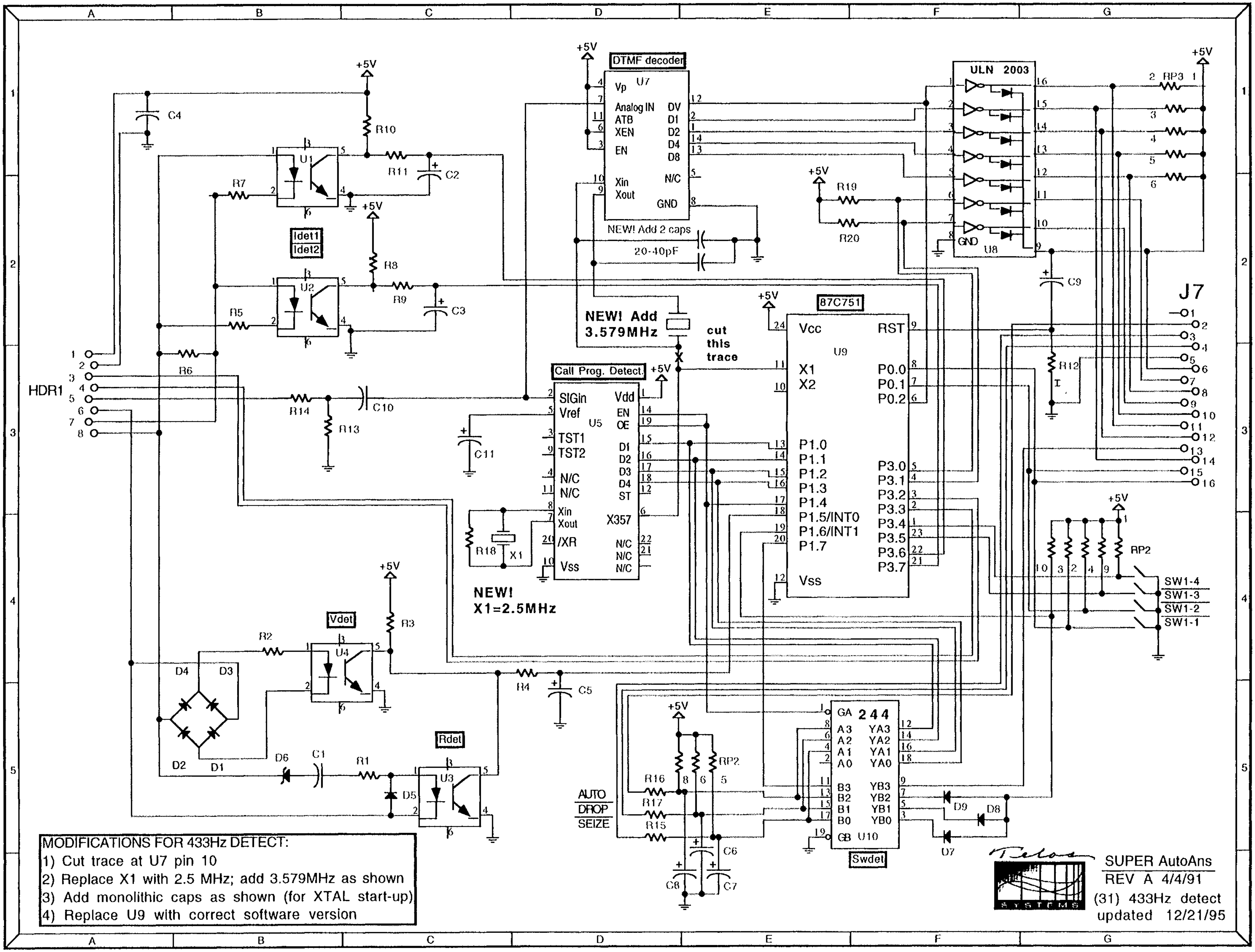
Designation	Description
SU1-4	6-pin socket
SU5	22-pin socket (400mil)
SU6	8-pin socket
SU7	14-pin socket
SU8	16-pin socket
SU9	24-pin socket (300mil)
SU10	20-pin socket
SW1	4-position DIP SW

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.

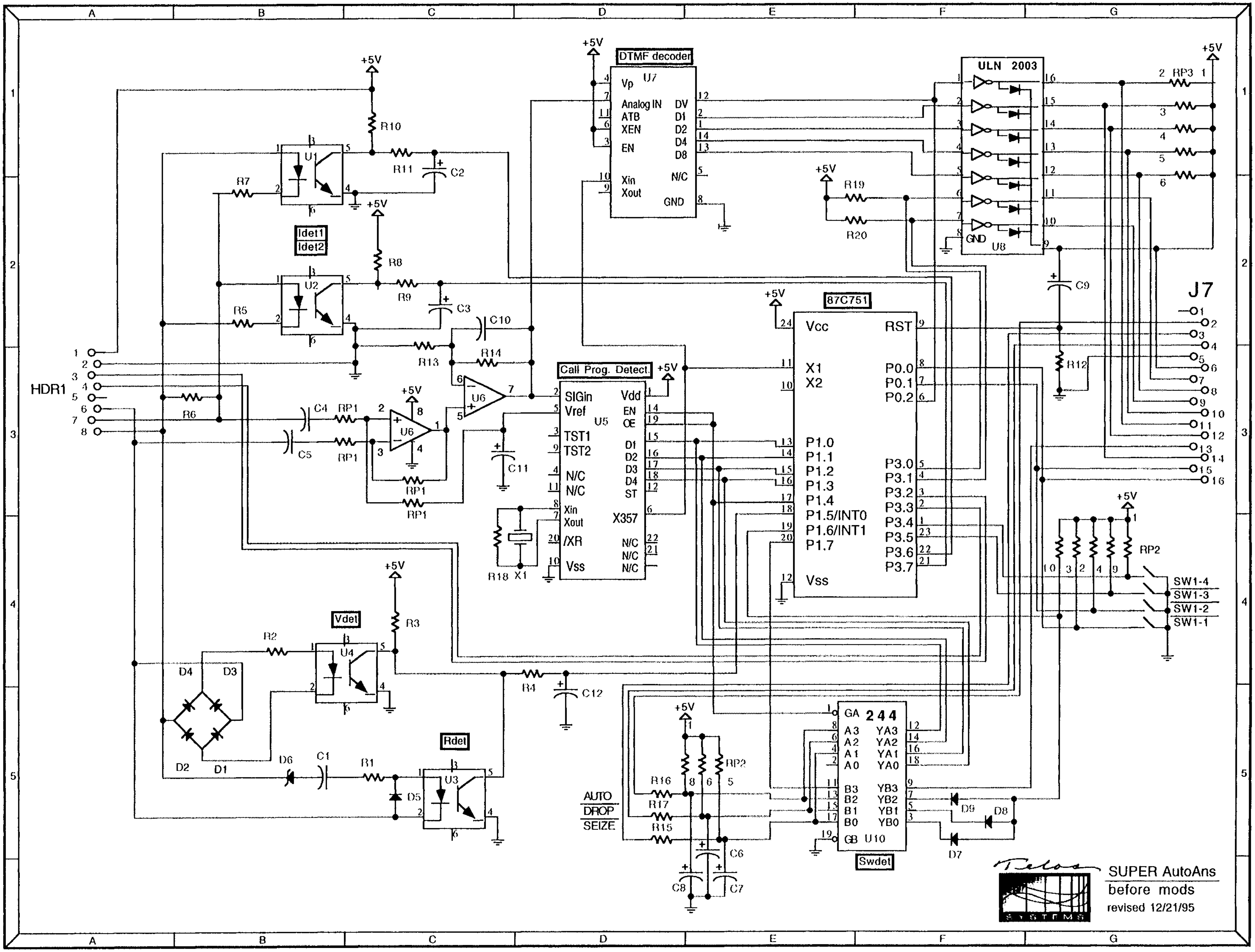

DTMF decoder
U7
ULN 2003
U8
87C751
U9
Call Prog. Detect.
U5
244
U10
Swdet
Idet1
Idet2
Vdet
Rdet
HDR1
J7
AUTO
DROP
SEIZE
SW1-4
SW1-3
SW1-2
SW1-1
SUPER AutoAns
REV A 4/4/91
updated 12/21/95

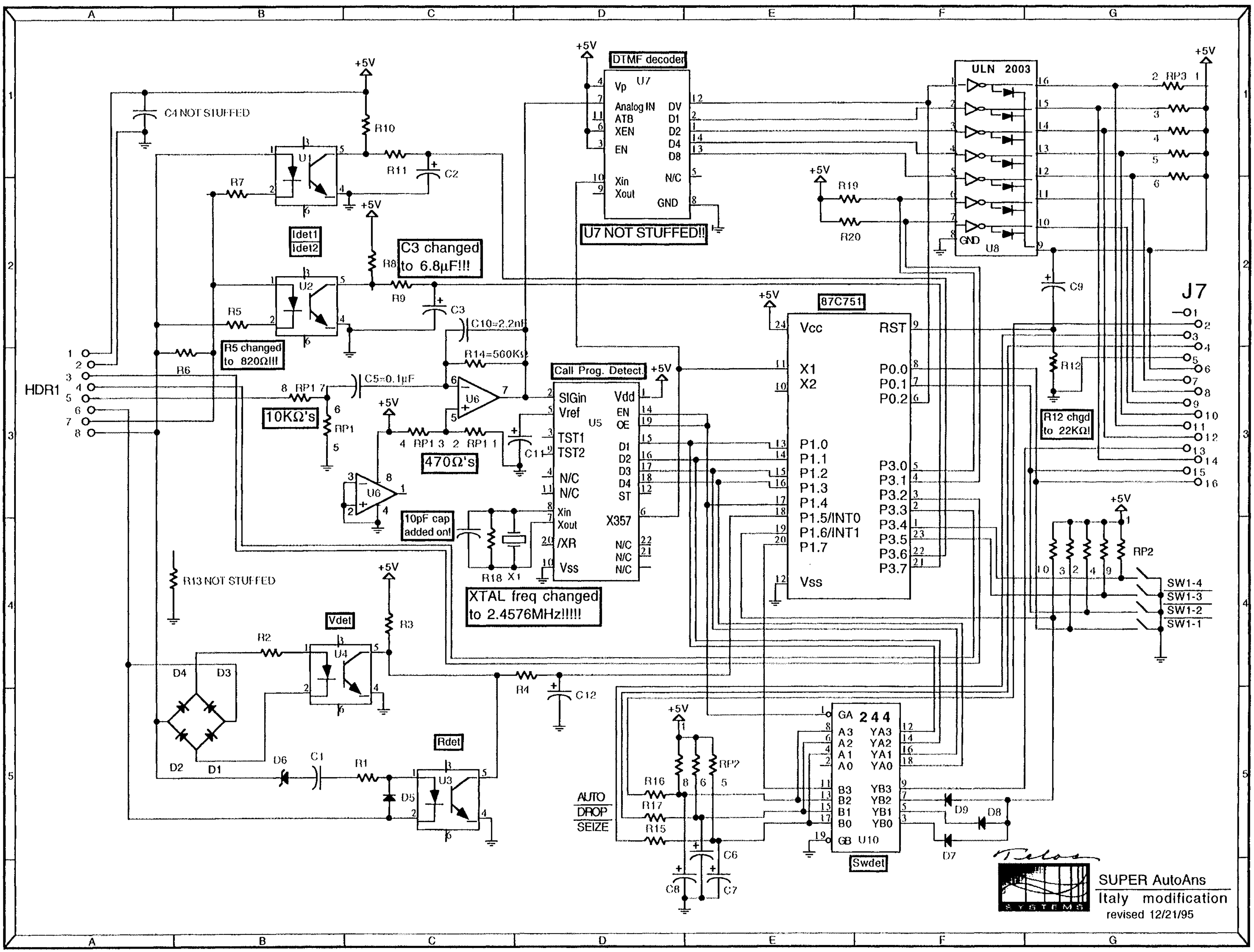

75T204 DTMF decoder
U7
M-982 call prog detect
U5
87C751 µP
U9
ULN 2003
U8
244
U10
Swdet
J7
HDR1
Idet 1
Idet 2
Vdet
Rdet
Hyb OFF
Hyb ON
AUTO
DROP
SEIZE
SW1-4
SW1-3
SW1-2
SW1-1
<<<<Add this wire
NEW! Add 3.579MHz
Add these caps!
27pF
27pF
cut this trace
NEW!
X1=2.4576MHz
Add C=10pF
MODIFICATIONS FOR 425Hz DETECT:
1) Cut trace at U5 pin 6
2) Replace X1 with 2.4576 MHz; add 3.579MHz as shown
3) Add monolithic caps as shown (for XTAL start-up)
4) Replace U9 with correct software version
SUPER AutoAns
REV A 4/4/91
(61) 425Hz detect
updated 4/3/96



MODIFICATIONS FOR 433Hz DETECT:
1) Cut trace at U7 pin 10
2) Replace X1 with 2.5 MHz; add 3.579MHz as shown
3) Add monolithic caps as shown (for XTAL start-up)
4) Replace U9 with correct software version

SUPER AutoAns
REV A 4/4/91
(31) 433Hz detect
updated 12/21/95


DTMF decoder
U7
ULN 2003
U8
87C751
Call Prog. Detect.
U5
X357
U6
Idet1
Idet2
U1
U2
Vdet
U4
Rdet
U3
HDR1
J7
244
U10
Swdet
SW1-4
SW1-3
SW1-2
SW1-1
AUTO
DROP
SEIZE
SUPER AutoAns
before mods
revised 12/21/95


DTMF decoder
U7 NOT STUFFED!!
ULN 2003
C4 NOT STUFFED
Idet1
Idet2
C3 changed to 6.8µF!!!
R5 changed to 820Ω!!!
HDR1
10KΩ's
470Ω's
C10=2.2nF
R14=560KΩ
C5=0.1µF
Call Prog. Detect.
X357
10pF cap added on!
XTAL freq changed to 2.4576MHz!!!!!
87C751
R12 chgd to 22KΩ!
J7
R13 NOT STUFFED
Vdet
Rdet
SW1-4
SW1-3
SW1-2
SW1-1
AUTO
DROP
SEIZE
244
Swdet
SUPER AutoAns
Italy modification
revised 12/21/95