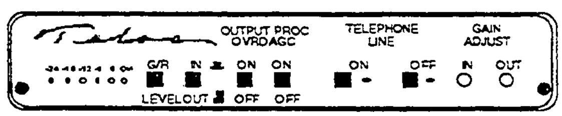
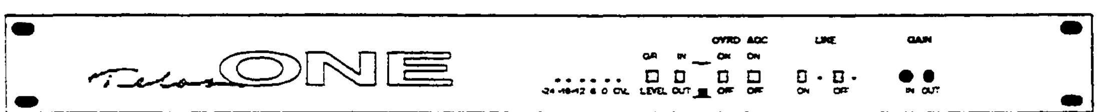
Telos ONE

Digital Hybrid Telephone Interface



Telos ONE Modem Case Version



Telos ONE Rack Mount Version

User's Manual

Version 3.2 – August 1996

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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 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

Telos ONE Digital Hybrid Telephone Interface

User's Manual V3.2 August 1996

SECTION 1

INTRODUCTION

1.1 A NOTE ON TELOS ONE VERSIONS

The Telos ONE comes in both a standard rack mounting chassis and a compact modem case. The rack mounting unit has a universal, input-switching power supply, while the modem version uses a voltage specific external plug transformer. With the exceptions of the form-factor and power supply, the two units are identical. Any illustration of the modem case version in this manual is valid for the operation of the rack mount version unless otherwise noted.

1.2 OVERVIEW

The Telos ONE Interface.

The Telos ONE interface brings the superb Telos digital hybrid performance to applications where cost is an important consideration. It embodies a state of the art approach to interfacing telephone lines for broadcast and teleconferencing use.

The Telos ONE is a *true digital* second generation telephone interface. The very fast and precise digital automatic nulling allows smooth, natural, simultaneous conversation without the usual speakerphone up-cutting effect or the audio distortion and feedback problems often experienced with hybrid-type interface devices.

Telephone connections are via standard modular jack, while audio input and output are connected via XLRs. One balanced input with provision for mic or line levels and two balanced outputs are provided. 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 digital hybrid telephone interface is to deliver to the console pure caller audio with as little of the send (announcer) audio as possible. 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.

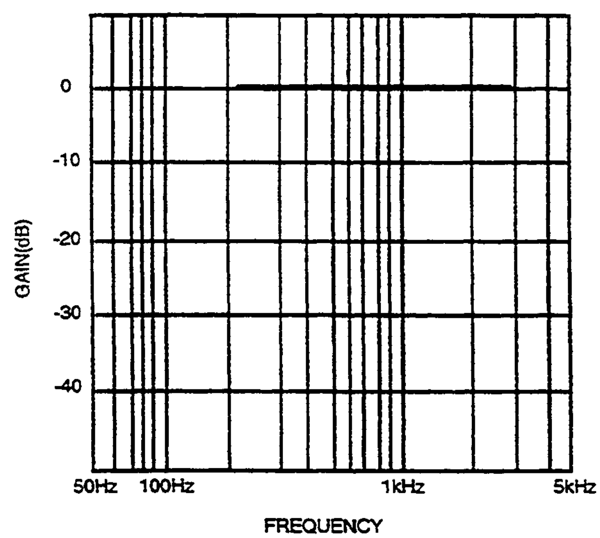
Analog hybrids These were, on most phone lines, very poor at removing the send signal from the caller's audio. This meant that the announcer's voice would become distorted as the phone audio was added to the mix.

The Telos ONE 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 announcer 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 4Hz 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 goes on the air. The caller hears a "noisy tone," but none of this tone is heard on the air 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

For multi-line systems, the unit may be interfaced directly to telephone lines or using a standard 1A2 key system by using add on modules made by Telos Systems. Since the Telos ONE system is modular, many configurations are possible to implement the desired number of lines and hybrids. Contact Telos Systems or your dealer or representative for details on current options.

1.3 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.

1.4 SPECIFICATIONS

System

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

Trans-hybrid Loss

>40dB 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 to >50dB.

Send Level to Phone Line

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

Frequency Response (caller to output)

200 - 3400Hz ±1dB.

Noise and Distortion (caller to output)

Distortion: Typical 0.4% THD+N, measured @1kHz at any level from -48dBm to -8dBm.
Signal-to-Noise: >72dB. Referred to 0dBm phone level.

Send Audio Input

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

Caller Audio Output

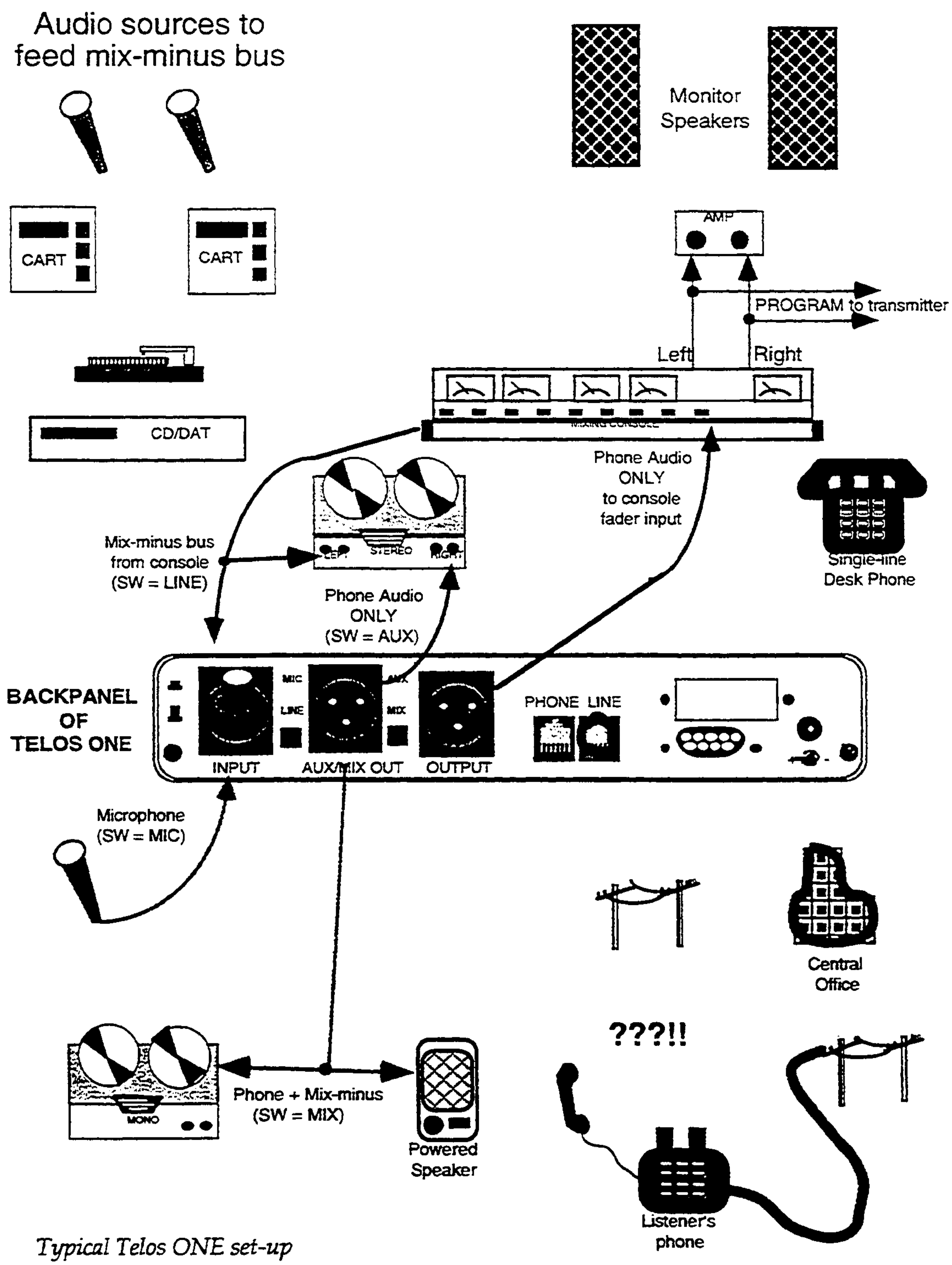
XLR male connector. Active differential. Output levels to +14dBm depending upon caller telephone line level and adjustment of front panel level adjust. Will drive 600Ω.

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: Unity gain. <0.04% THD; +12dBm clip point.

SECTION 2

INSTALLATION

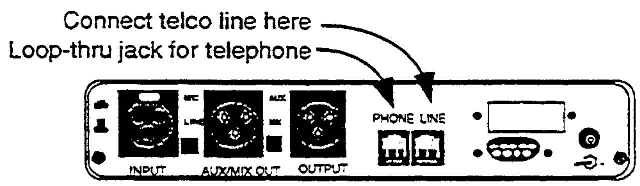


Typical Telos ONE set-up

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 styles of plug-in 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. Contact Telos or your Telos dealer for more details on the Basic Auto-Answer card or Super Auto-Answer board. Manuals for these accessories are found in the appendix of this manual.

Information on installation and use of the Telos ONE with Telos multi-line, multi-hybrid control and interface modules is in the manuals included with those modules.

"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 selection device. A special version of our Super Auto-Answer board makes this possible.

You may also build a system on your own, and an application note is available from us if you wish to do so. 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.

Electronic Phone Systems

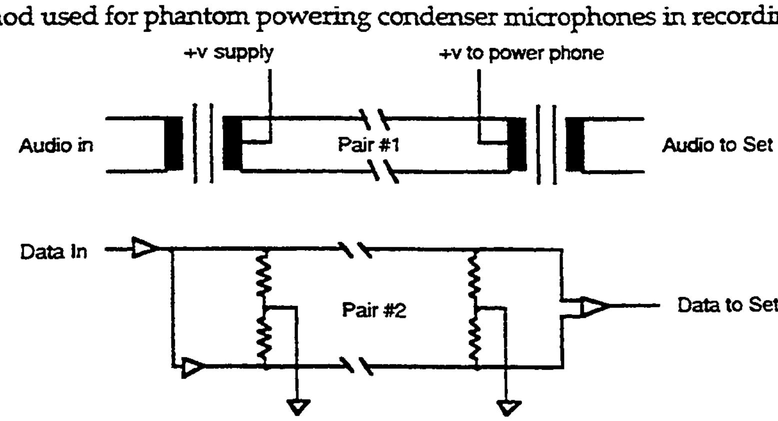
These have nearly completely taken over from older mechanical systems with the holdouts being broadcast studios and the financial brokerage business. The cable is smaller and cheaper and fancy features are easy to implement.

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 32kHz 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 news room or production studio or for occasional on-air use.

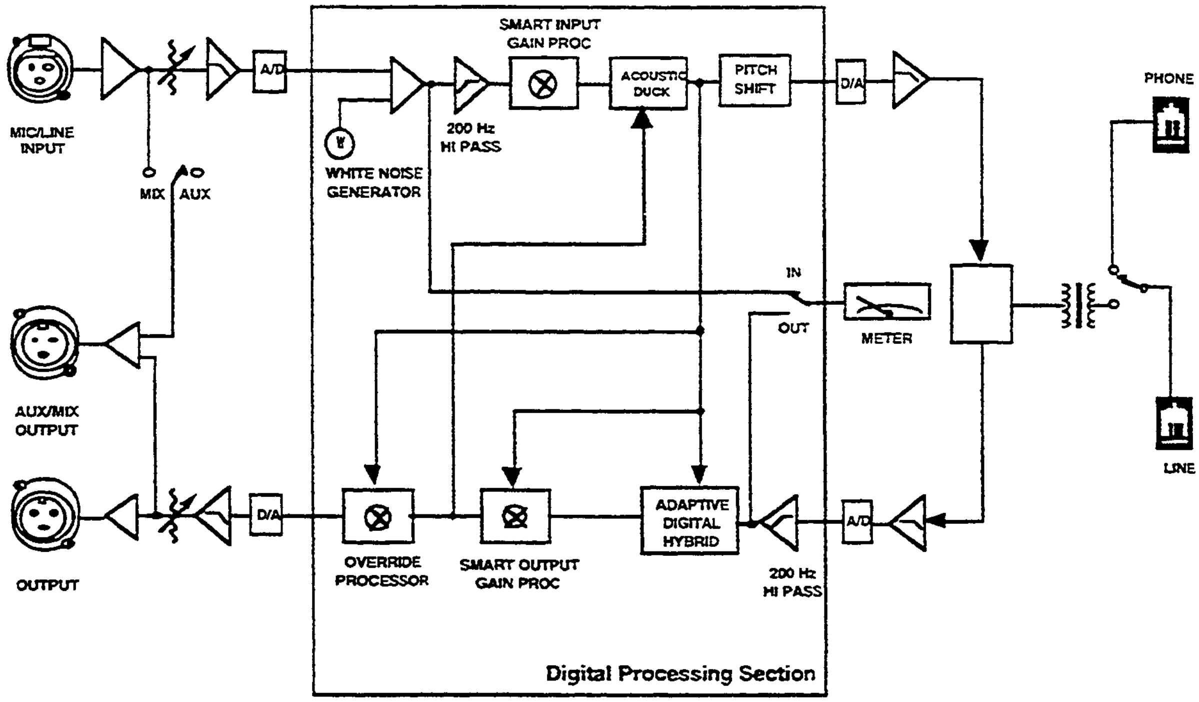
A problem remains, however, when calls need to be taken one after the other, as in a talk show situation. In most cases, the hybrid must be signalled each time a new line is selected so that adaption to the line can occur. It may be possible to derive an appropriate pulse from the phone by tacking on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

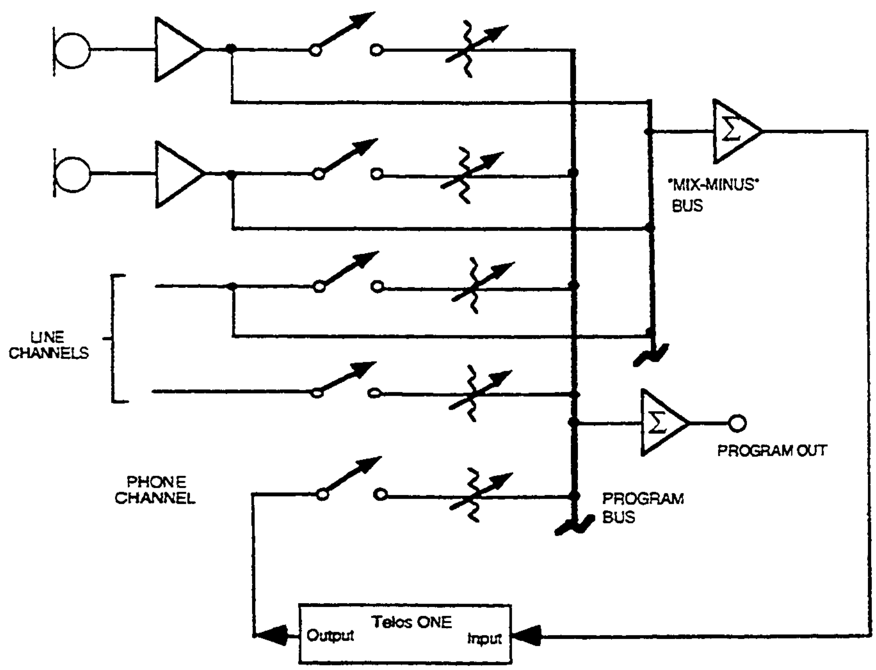
Simple Mix-Minus

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.

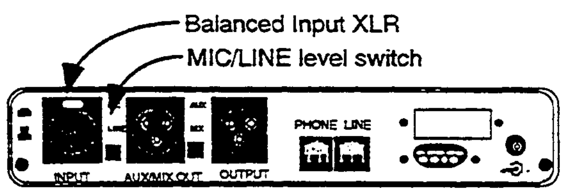
A Good Idea...

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.

2.2.2 INPUT AUDIO CONNECTION

The input has the following characteristics:

- Active balanced.
- Switch in LINE position: -24 to +12dBv level.
- Switch in MIC position: -68 to -35dBv level.
- 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 hot signal is connected to pin 3.

2.2.3 OUTPUT CONNECTIONS

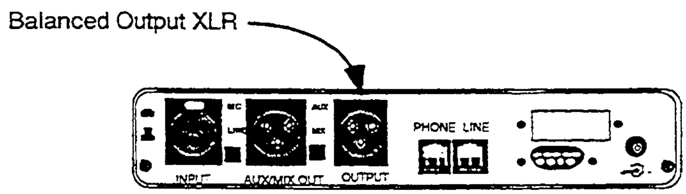
There are two separate and independent active differential audio outputs each with the following characteristics:

- Active balanced. If an unbalanced output is required, connect between ground and either of the hot pins. Do not ground the unused hot pin.
- Pin 1 is ground. Pins 2 and 3 are the balanced signal outputs.
- Output level will vary from approximately -20dBm to +10dBm depending upon gain control adjustment, caller level and whether or not the AGC is engaged.

Important note: The output level meter is before the gain control; it displays actual phone level before processing. Use the output trimmer on the front panel to adjust the level to your mixing board.

Main Output

Caller audio appears on the main output.

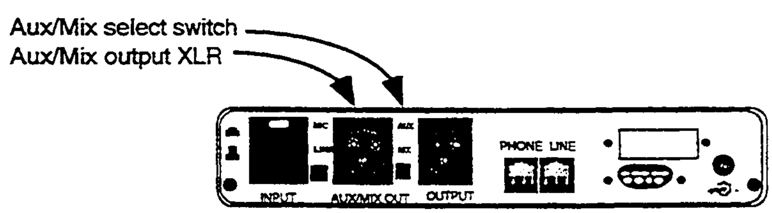


AUX/MIX Output

The AUX/MIX output is either a mix of the send and caller signals or an isolated second caller-only output, depending on the setting of the switch to the right of this male XLR.

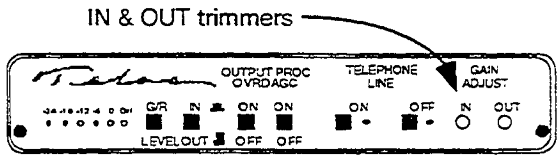
- Switch out = mixed caller and send signal is present.
- Switch in = caller-only audio is present.

The input is passed to the MIX output at unity gain. Note from the block diagram on page 13 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!

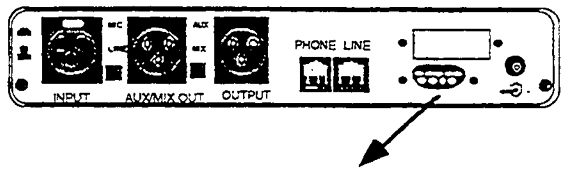
2.3.2 OUTPUT GAIN

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 both outputs 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

A female DB9-type connector on the rear panel provides access to control functions. In addition, parallel telephone connections are provided.



DB9 "REMOTE" Connector Rev. C	
Function	**Pin No.**
Hybrid "ON" input	1
"A" lead relay wiper	2
Open-collector output	3
+5VDC	4
Telco "LINE" tip	5
Ground	6
Hybrid "OFF" input	7
"A" lead N.O. contact	8
Telco "LINE" ring	9

Remote OFF/ON

OFF and ON control requires a momentary closure to ground. It is a standard TTL input pulled-up with a 2.2 KΩ resistor. Thus, it may be connected directly to switches or may be driven by an open collector or TTL-compatible logic output as desired.

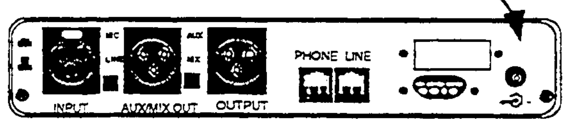
Telephone Line Connections via the Remote Connector

- The LINE connector is paralleled to pins 5 and 9. These may be used instead of the modular connector if desired.
- "A" control is provided on pins 2 and 8. "A" control is phone terminology for a simple relay closure which is active when a line is on. It may be used to control phone equipment or as a way to operate a remote indicator.
- The open collector output on pin 3 is the collector of the transistor that operates the telco and A-lead relays. It can sink up to 50mA and goes low whenever the line is seized. The relays act as a +5V pull-up.

2.5 POWER SUPPLY

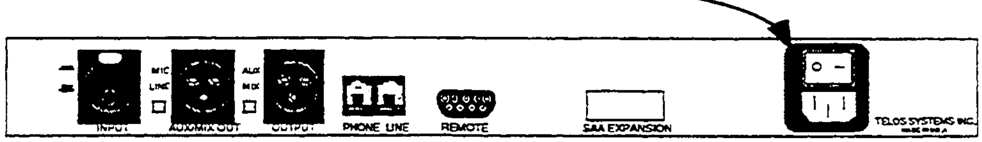
The Telos ONE modem case version uses a 9VDC/500ma external, plug-in power supply. This is a very common style wall transformer supply. It is widely available in most electronics stores, should replacement be necessary. The output jack is a 5.5mm O.D./2.1mm I.D. size, with ground on the sleeve, as shown on the silkscreen next to the jack on the rear panel of the unit.

Connect a 9VDC power supply here



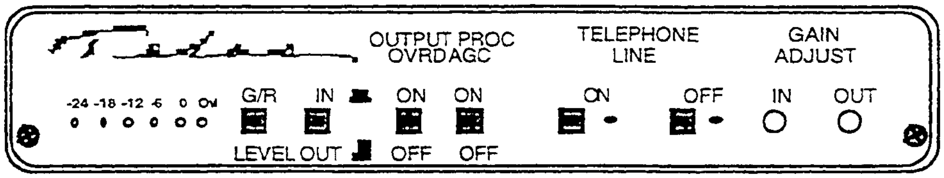
The Telos ONE rack mount version has a built-in, universal-input power supply. The input is a standard IEC three-prong AC connector with a power ON/OFF switch. The fuse is located on the internal power supply. For more details on the power supply, see Section 4 of this manual and the manufacturer's data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

Connect 90-260VAC here with IEC connector



SECTION 3
OPERATION

3.1 FRONT PANEL CONTROLS



ON/OFF Pushbuttons

When the ON button is pressed, the phone line is seized and the system sends a burst of white noise down the line, which allows the digital adaptive filters to adjust to the phone line's impedance. 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.

Note that the corresponding LED indicators next to the buttons follows the telco line status.

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 6dB 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

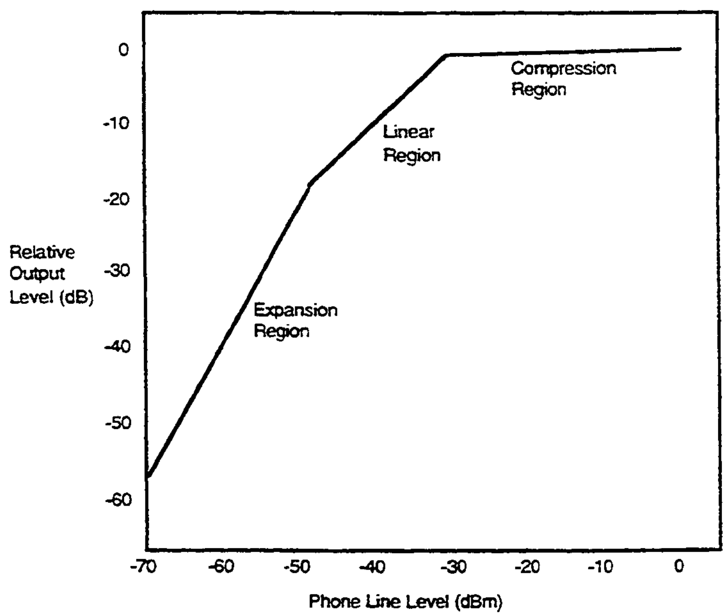
This function operates in the send audio path. The acoustic ducker works by reducing the send (announcer) signal dynamically when the caller speaks.

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 -30dBm and 0dBm. Below about -48dBm, 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 -16dBm. *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 SWITCH	IN/OUT SWITCH
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 ONE's 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.

4 Add equalization to the monitor path. Acoustic resonances usually cause pronounced peaks in the "feedback response" of a sound system. Since the largest peaks generally occur at just a few frequencies, reducing system gain at these frequencies with a graphic or notch EQ helps tremendously.

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

The foregoing is intended to help in those situations where you *must* have an open speaker. Whenever possible, it is best to use headphones to hear callers. When you have an open speaker, the on-air phone audio has both a direct and an acoustic path - from the speaker to the announce mic(s). Depending on the relative levels, phone audio quality may suffer.

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

SECTION 4

TECHNICAL DATA and TROUBLESHOOTING

4.1 OVERVIEW

Philosophy

In recent 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 Board (modem case version)

1 Remove front panel and bezel by unscrewing the two Phillips screws.
2 Slide the top cover forward.
3 Slide the bottom plate off while holding the rear panel to expose the solder side of the circuit board. The rear panel can remain attached to the PCB.

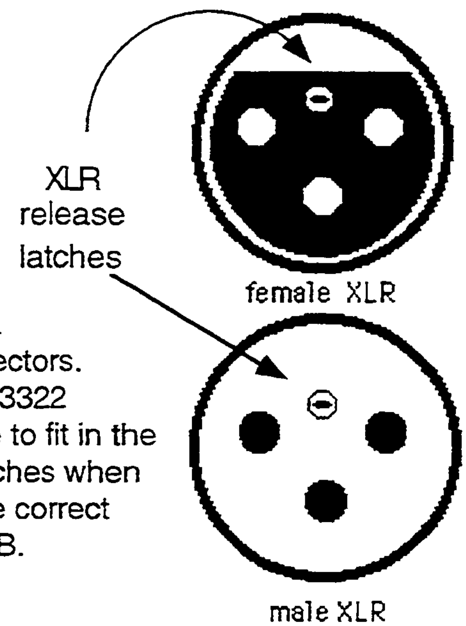
If the rear panel must be removed:

4 Unscrew the DB9 connector's standoffs using a 3/16" nut driver.

5 The three XLR connectors have retaining latches which have to be turned in order to be released. Do this as illustrated below.

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.



Access to the PC Board (rack mount version)

1 Remove the top cover first and then the front panel by unscrewing the Allen-head screws, using a 1/16″ Allen key.
2 Release the XLRs as shown above.
3 Remove the standoffs on the DB9 using a 3/16″ nut driver.
4 Remove the screws holding the circuit board in place.
The board should now slip out of its case by gently pushing on the XLRs and pulling the front of the circuit board.

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 4kHz and the ultimate sampling rate is 8kHz.

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 320C25s 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 +5V power supply drops below 4.5V.

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 40MHz clock oscillator module. The 40MHz output is fed directly to the 320C25. The 'HC390, U9 divides the 40MHz to 2MHz 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 8kHz 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 40MHz 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 putting out data.

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

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

4.3 AUDIO SECTION

4.3.1 THEORY OF OPERATION

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

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

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

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

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

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

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

4.3.2 TROUBLESHOOTING THE AUDIO SECTION

Using a scope for signal tracing should do the trick.

All chips should run cool except for non-Signetics brand 5532's - *it is normal for these to run hot.*

4.4 POWER SUPPLY

The Telos ONE modem-style case uses a wall-mount AC-to-DC transformer to supply +9VDC to the unit. The Telos ONE rack mount 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.

SECTION 5

DRAWINGS

DRAWINGS

Parts List
Signal Flow Block Diagram
TMS320C25 Pinouts
MPU and Power Supply Schematics
Audio and Telco Section Schematics

Telos ONE

PARTS LIST

Designation	Description
R1	100Ω
R2	100Ω
R3	10KΩ
R4	1KΩ
R5	560Ω
R6	10KΩ
R7	10KΩ
R8	10KΩ
R9	100Ω
R10	33KΩ
R11	10KΩ
R12	10KΩ
R13	1KΩ/1%
R14	1KΩ/1%
R15	1KΩ/1%
R16	1KΩ/1%
R17	100KΩ/1%
R18	100KΩ/1%
R19	1KΩ
R20	2.4KΩ
R21	10KΩ
R22	49.9Ω/1%
R23	49.9Ω/1%
R24	10KΩ
R25	56KΩ
R26	1KΩ
R27	15KΩ
R28	49.9Ω/1%
R29	49.9Ω/1%
R30	10KΩ
R31	10KΩ
R32	10KΩ
R33	10KΩ
R34	39KΩ
R35	270KΩ
D1	1N4730 3.9v zener
D2	1N4730 3.9v zener
D3	1N5818
D4	1N4004
D5	1N4148
D6	1N4148

Designation	Description
C1	2.2µF/25V tant
C2	2.2µF/25V tant
C3	0.1 Farad Gold Cap
C4	22pF mono
C5	2200pF mono
C6	2200pF mono
C7	0.01µF mono
C8	4.7µF/25V tant
C9	4.7µF/25V tant
C10	2.2µF/25V tant
C11	100µF/35V electrolytic
C12	1000µF/35V electrolytic
C13	10µF/25V tant
C14	0.1µF mono
C15	4.7µF/25V tant
C16	2.2µF/25V tant
C17	2.2µF/25V tant
C18	0.1µF mono
C19	0.1µF mono
C20	0.1µF mono
C21	0.1µF mono
C22	0.001µF mono
IND1	100µH toroid
Q1	2N2222
RP1	2.2K SIP
RP2	330Ω SIP
RF1,2,3	RF filter: TDK ZJK51R1-05
L1-5	Green LED LTL-1234A
L6	Red LED LTL-1224A
L7,8	Yellow LED LTL-1254A
K1,2	5V relay: Omron G5V-2-H
VR1	LM7905CT -5V reg
VR2,3	LM7805 +5V reg

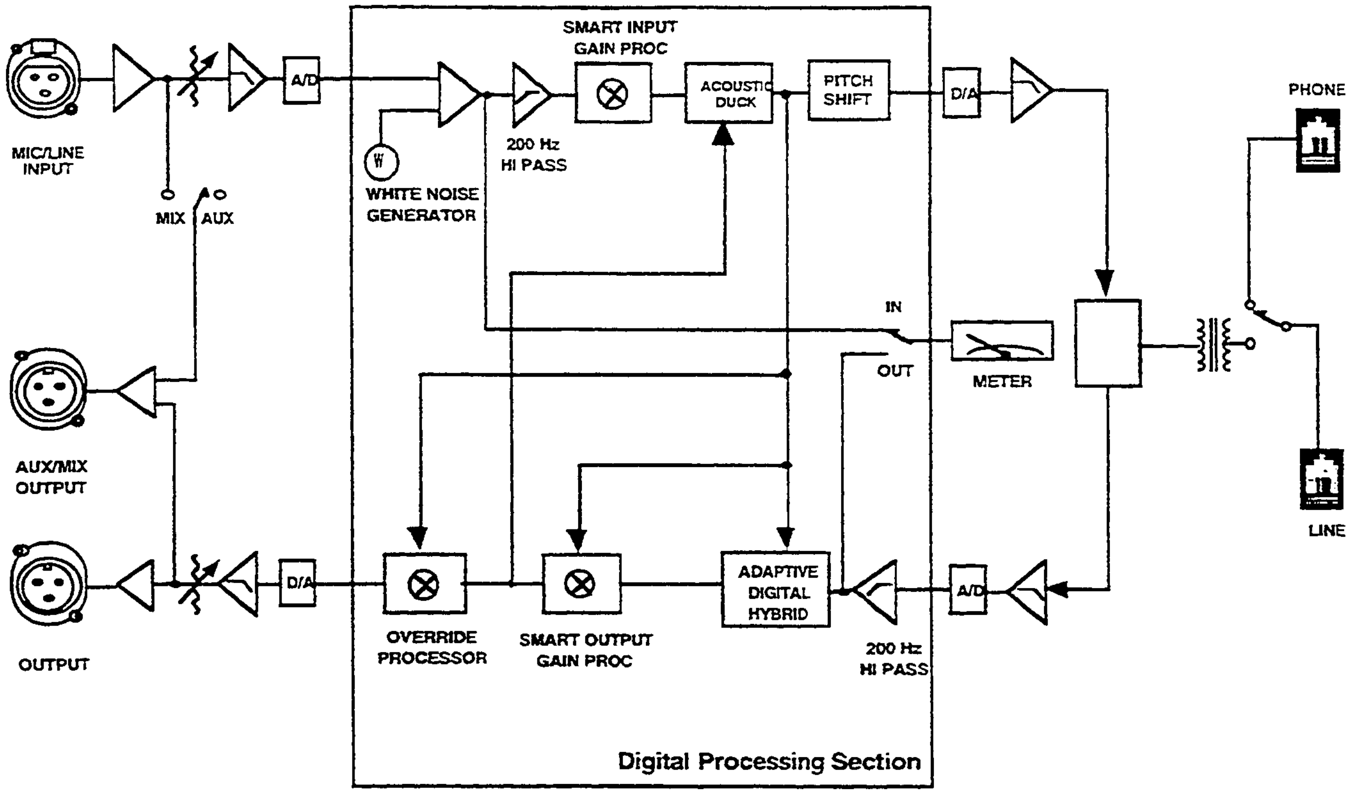
Telos ONE

PARTS LIST (con't.)

Designation	Description	Designation	Description
U1	TI 320C25 MPU	T1	Prem SPT-195 xfmr
U2,3	27C292-3JL EPROM		
U4	74AC244	J1	XLR female: Neutrik NC3FDHBAG
U5	74HCT374	J2,3	XLR male: Neutrik NC3MDHBAG
U6	74ACT138	J4	DB9 female
U7	DS1232 watchdog/reset	J5	2.1mm coax power connector
U8	74HC590	J6	Dual modular phone receptacle
U9	74HC390		
U10	Saronix NCH080C-40MHz osc	Z1	250V MOV
U11	74HCT04		
U12,13	AMD7901CPC CODEC	P1,2	10K rt angle multiturn trimpot
U14	2912A		(Bournes 3266-X-103)
U15,16,17	NE5532 dual op-amp		
U18	MAX636 switching inverter	Heatsink	TO-220: AAVID 5073B
U1S	68 pin PGA socket		
U2,3S	24 pin IC socket		
SW1-4	Alt action switch		
SW5,6	Mom switch		
SW7	Two pos DIP sw		
SW8,9	Alt action switch		

Telos ONE Modem Case
External Power Supply 9VDC @ 0.5A
(with 5.5/2.1mm plug)

Telos ONE Rack Mount
Internal Power Supply: Autec UPS20-5002
12VDC @ 1.0A

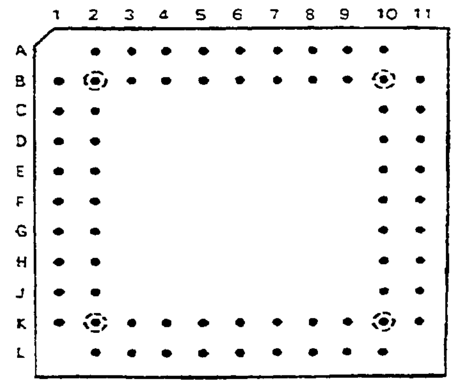


Telos ONE
Signal Flow Block Diagram

PIN ASSIGNMENTS

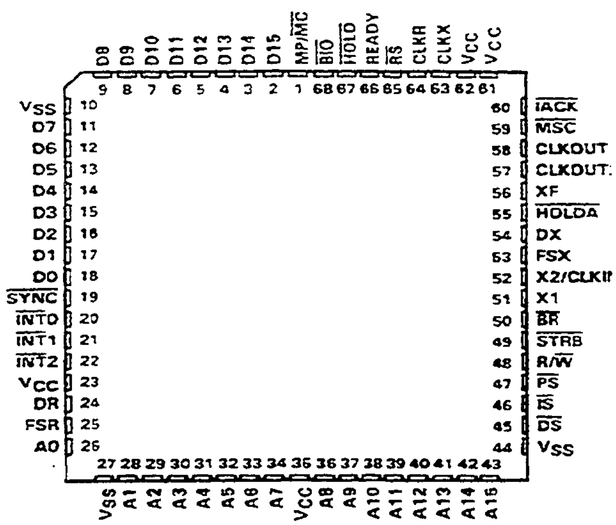
PIN	FUNCTION	PIN	FUNCTION	PIN	FUNCTION
A2	D8	C11	CLKOUT1	J10	$\overline{PS}$
A3	D10	D1	D4	J11	$\overline{IS}$
A4	D12	D2	D3	K1	A0
A5	D14	D10	CLKOUT2	K2	A1
A6	MP/$\overline{MC}$	D11	XF	K3	A3
A7	$\overline{HOLD}$	E1	D2	K4	A5
A8	$\overline{RS}$	E2	D1	K5	A7
A9	CLKX	E10	$\overline{HOLDA}$	K6	A8
A10	V_{CC}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

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



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

TEXAS INSTRUMENTS



Telos ONE
TMS320C25 pinouts

SECTION 6

APPENDIX

APPENDIX

Warranty and Application Note
Rack Mount Version Power Supply Data Sheet
Basic Auto-Answer Board Manual
Super Auto-Answer Board Manual

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.

AUTEC POWER SYSTEMS

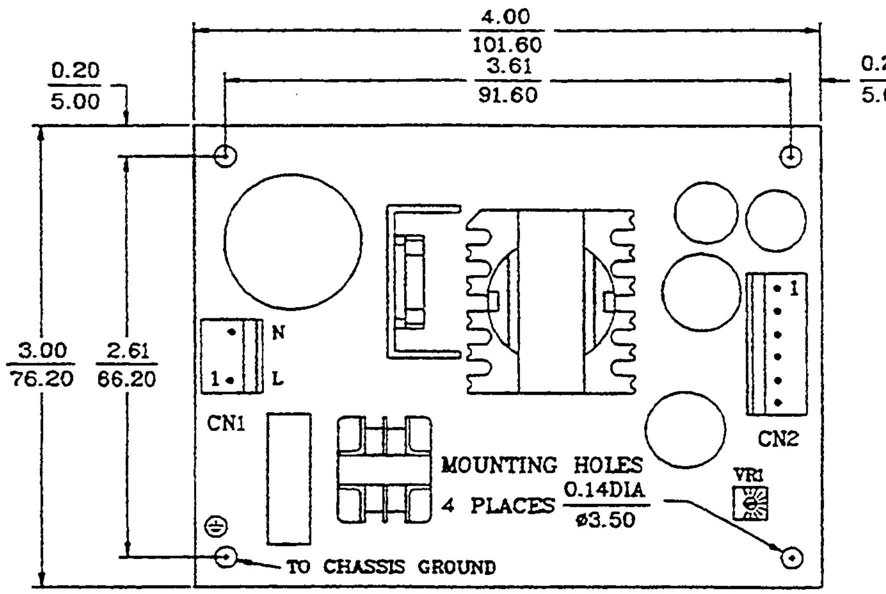
WORLD CLASS POWER HAS NO BOUNDARIES

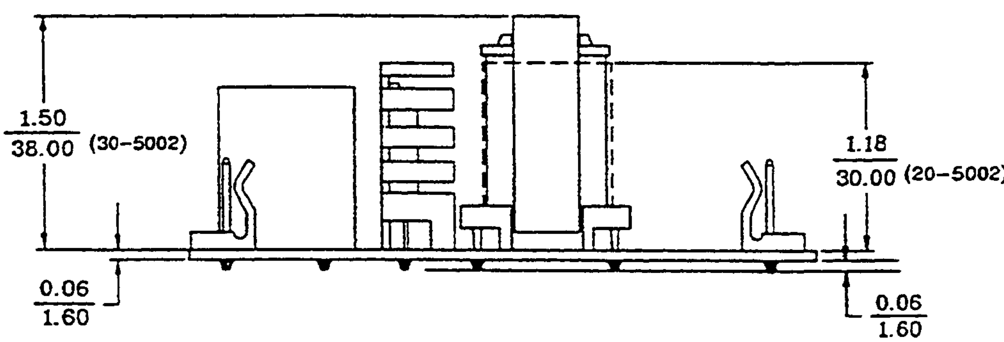
20 & 30 WATT
SWITCHING POWER SUPPLIES

3 X 4 SERIES

FEATURES:

- FCC CLASS 'B' COMPLIANCE
- UL 1950 APPROVED NO. E133148
- CSA 1402C LEVEL 3 NO. LR89164
- TUV APPROVED TO IEC950 FILE NO. R98290
- 90 - 264 VAC (CONTINUOUS) UNIVERSAL INPUT
- LOW MINIMUM LOAD
- EXTRA SMALL SIZE FOR TIGHT MECHANICAL REQUIREMENTS.





DESCRIPTION:

This family of 20 and 30 watt switchers were designed for use in zero footprint external memory expansion peripherals. Compatible applications are half height 5-¼" and 3-½" hard disk drives, floppy disk drives, DAT tape drives and streaming tape drives. An ultra-compact 3" by 4" footprint eases integration in even the most space constricted mechanical constraints. 90 - 264 VAC continuous universal input eliminates costly input select accessories and associated labor. Optional AC & DC cable assemblies are available and custom enclosures can be quoted. Please consult an Applications Engineer for further details.

AUTEC POWER SYSTEMS

69 Moreland Road
Simi Valley, CA 93065

Tel : (805)522-0888
Fax: (805)522-8777

GENERAL SPECIFICATIONS:

INPUT VOLTAGE: 90-264 VAC, 47-440 Hz

INRUSH CURRENT: 20A @ 115 VAC, 40A @ 230 VAC

HOLD-UP TIME: 20 Ms, minimum, see note 4

DC OUTPUTS: See OUTPUTS table below. Output 1 is adjustable 5%. All other otputs track Output number 1.

SWITCHING FREQUENCY: 40 Khz

EFFICIENCY: 70% Minimum

CONDUCTED EMI: FCC Class 'B'

TEMPERATURE COEFFICIENT: 0.02% / degree Celcius

OVERLOAD: Provided within 150% of total power

HI-POT ISOLATION: 3750 VAC input to output for 1 minute.

FUSING: 2A, 5TT for 115/230 VAC

OVERVOLTAGE PROTECTION: Output 1: output level exceeding 115 - 135% causes crowbar, automatic recovery.

TEMPERATURE: Operating: 0° to 50° C at rated power. Storage: -55° C to 85° C

OUTPUT POWER DERATING: Derate each output linearly to 50% rated output at 70°C.

HUMIDITY: 5% to 95% RH non-condensing

ALTITUDE: 10,000 ft., 40,000 ft. non-operating

OUTPUT TABLE:

MODEL	OUTPUT VOLTAGE	LOAD			TOLERANCE	RIPPLE AND NOISE[5]	REGULATION	
		MIN.	RATED[2]	PEAK[1]	±		LINE	LOAD[3]
UPS20-5002 20W / 35W PK	5.0V 12.0V	0.2A 0.0A	1.60A 1.00A	2.00A 2.00A	1.50% 5.00%	50 mV 100 mV	1.0% 1.0%	2.0% 3.0%
UPS30-5002 30W / 45W PK	5.0V 12.0V	0.2A 0.0A	2.10A 1.80A	 3.00A	1.50% 5.00%	50 mV 100 mV	1.0% 1.0%	2.0% 3.0%

CN2 DC OUTPUT PINOUT AND CONNECTOR DATA

1	2	3	4	5	6	MOLEX OR EQUIVALENT CONNECTOR	 MATING
+5V	+5V	COM	COM	+12V	+12V	09-65-2068	09-50-1061

CN1 AC INPUT PINOUT & CONNECTOR DATA

1	2	3	CONNECTOR	MATING
LINE	N/C	NEUTRAL	09-65-2038	09-50-1031

NOTES:

1. Each output can provide up to its Peak current for 30 seconds. The sum wattage of all outputs, inclusive of Peak currents, must not exceed 30 watts (UPS20-5002) and 45 watts (UPS30-5002). Each output must never exceed its Peak rating.
2. Each output can provide up to its rated load, provided the total continuous wattage does not exceed its rated wattage.
3. Load regulation is measured with the output to be measured set at 100% rated load and all other outputs at 60% of their rated load. The load on the output being measured is varied from 100% - 60% rated load.
4. Hold-up time is measured at 120 VAC input and 80% of rated load.
5. Ripple & Noise are measured using a 12" twisted pair conductor terminated with a 47 uF capacitor.
6. All specifications are typical at nominal line, full load and 25°C after a 1 minute warm-up period unless otherwise noted.
7. Specifications are subject to change without notice.

BASIC AUTO-ANSWER BOARD MANUAL

Purpose

The Basic 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 Basic Auto-Answer is installed by plugging it into the header connectors on the Telos ONE board. Remove the small jumper plugs on HDR1 first.

The Basic Auto-Answer board should be removed and the jumpers replaced if 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 on the first ring and hang-up upon loop drop.

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.

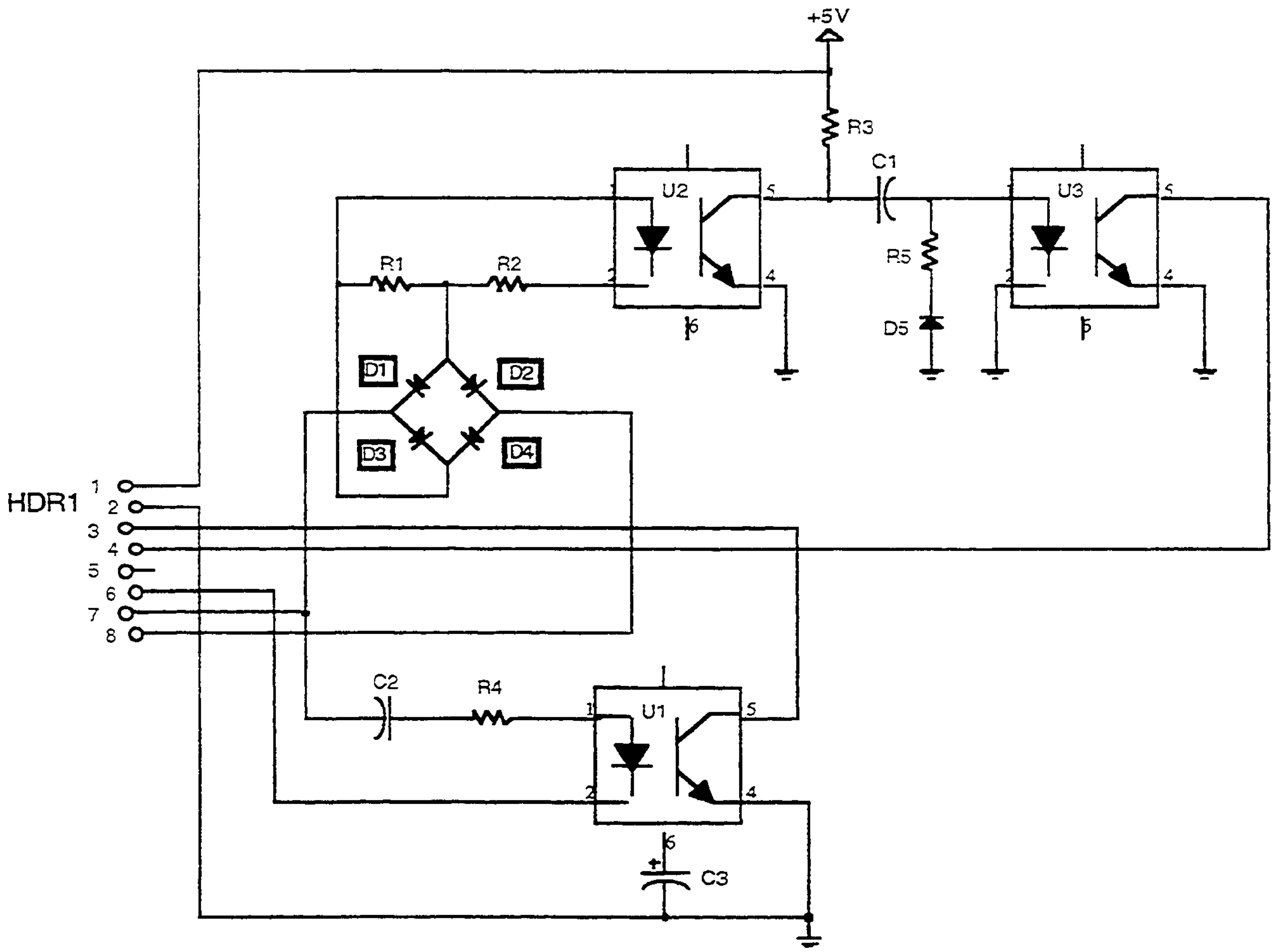
The remaining components form the loop current detector. When loop current is present, U2's transistor conducts and C1 is discharged. When loop current goes away, C1 charges through the diode in U3 thus causing a low at HR1-1 for the length of the R3/C1 time constant. Upon restoration of loop current, 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.

Telos Basic Auto-Answer Board

PARTS LIST

Designation	Description
U1,2,3	opto-isolator 4N33
D1,2,3,4	1N4004 dioide
D5	1N4148 diode
C1	6.8µF @ 25V
C2	0.47µF @ 200V
C3	3.3µF @ 25V

Designation	Description
R1	100Ω
R2	220Ω
R3	2.2KΩ
R4	2.2KΩ
R5	470KΩ



Telos basic AutoAns
Rev 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

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

USER'S MANUAL REVISION B2

YOUR NOTES HERE

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 modem case version

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

2) *FOR UNITS REV B. OR EARLIER:* On the bottom of the PCB, solder a small piece of wire from U14-4 to HDR1-5. The jumper wire allows the auto-answer board to have access to the hybrid-separated telephone receive signal for DTMF detection.

3) 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.)

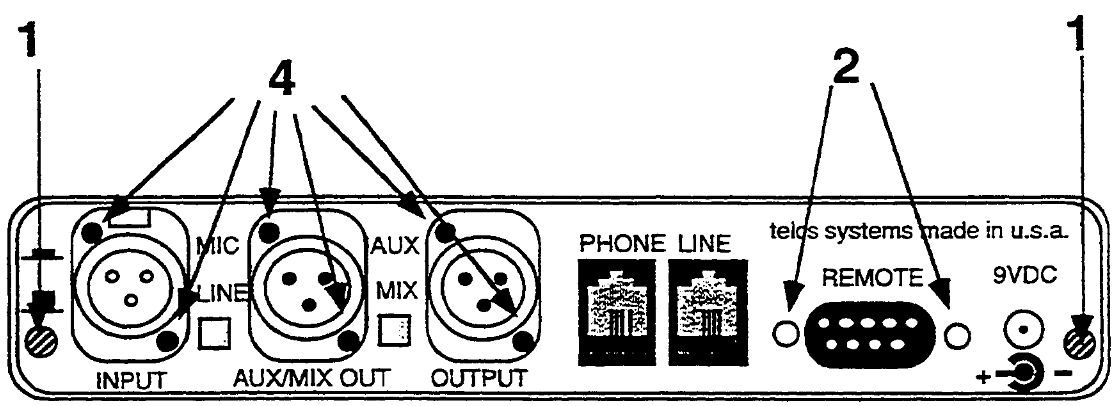
4) Replace rear panel with the new one, mount the DB15 remote connector, and plug the ribbon cable into the PCB socket.

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

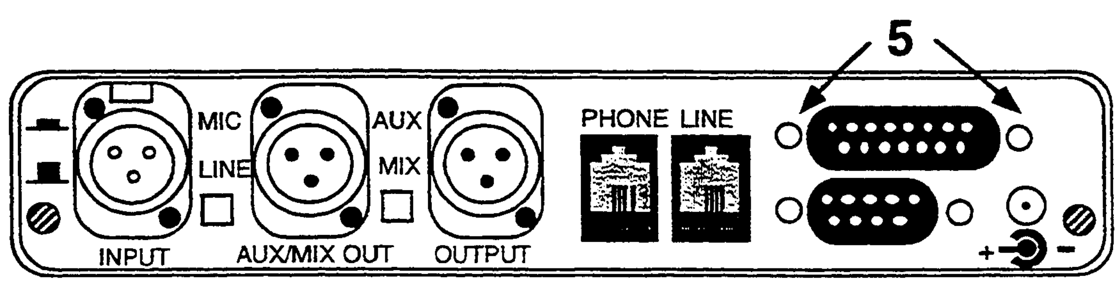
INSTALLING THE NEW REAR PANEL FOR THE TELOS ONE

1) Remove the two Phillips screws on either side of the rear panel; do the same to the front panel.
2) Remove the two stand-offs holding the 9-pin D-sub on.
3) Release the XLRs from their shells (refer to Telos ONE manual under "Technical Data and Troubleshooting"). The rear panel should now slip off. Slide the black cover of the hybrid off, as well.
4) Using a 1/16" allen key, remove the XLR shells and mount them on the new rear panel.
5) Using the provided hardware, mount the new 15-pin D-sub cable assembly onto the new rear panel.
6) If the Super Auto-answer board has not been installed, carefully mount it on the headers as shown in the Super Auto-answer manual and attach the new ribbon cable to the 16-pin connector. Be sure to select which special functions will be needed on the circuit board 4-position DIP switch. Refer to the Super Auto-answer manual for details.
7) Carefully slide the black cover back on. Fit the bezel and rear panel onto the hybrid and retighten all fasteners in reverse order (XLRs, stand-offs, Phillips screws).Put the front panel plate on last, carefully lining up the switches and LEDs (don't force it!).

BEFORE



AFTER



Telos ONE rack-mount case:

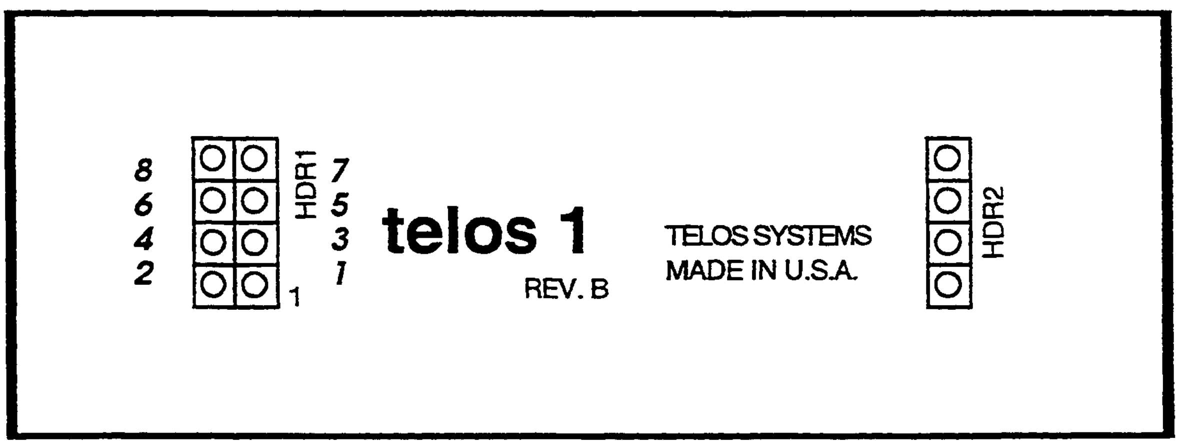
1) Remove the Telos ONE's top cover using a 1/16" Allen key. Then remove the front panel by removing the four black screws on either end of the front panel. The release the rear panel XLRs as described in the hybrid manual. Pushing gently on the XLRs while pulling on the front of the board will remove the board from the case.

2) *FOR UNITS REV B. OR EARLIER:* On the bottom of the PCB, solder a small piece of wire from U14-4 to HDR1-5. The jumper wire allows the auto-answer board to have access to the hybrid-separated telephone receive signal for DTMF detection.

3) 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.)

4) Mount the DB15 remote connector on the rear panel cutout and plug the ribbon cable into the right-angle PCB header. Discard the cutout insert.

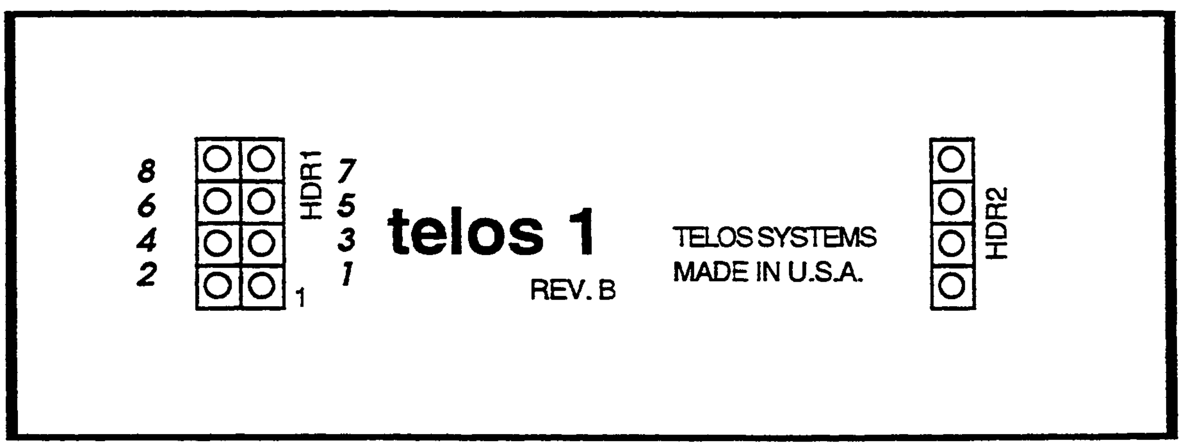
5) Set the auto-answer dip switches for the desired options (described in the operation section next) before re-assembling unit.



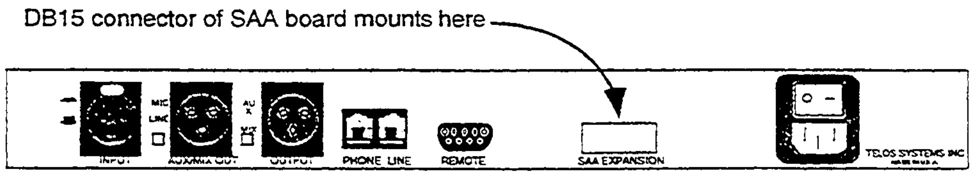
HDR1 & HDR2 locations and pinout

Telos ONE rack-mount case:

1) Remove the Telos ONE's top cover using a 1/16" Allen key. Then remove the front panel by removing the four black screws on either end of the front panel. The release the rear panel XLRs as described in the hybrid manual. Pushing gently on the XLRs while pulling on the front of the board will remove the board from the case.
2) *FOR UNITS REV B. OR EARLIER:* On the bottom of the PCB, solder a small piece of wire from U14-4 to HDR1-5. The jumper wire allows the auto-answer board to have access to the hybrid-separated telephone receive signal for DTMF detection.
3) 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.)
4) Mount the DB15 remote connector on the rear panel cutout and plug the ribbon cable into the right-angle PCB header. Discard the cutout insert.
5) Set the auto-answer dip switches for the desired options (described in the operation section next) before re-assembling unit.



HDR1 & HDR2 locations and pinout

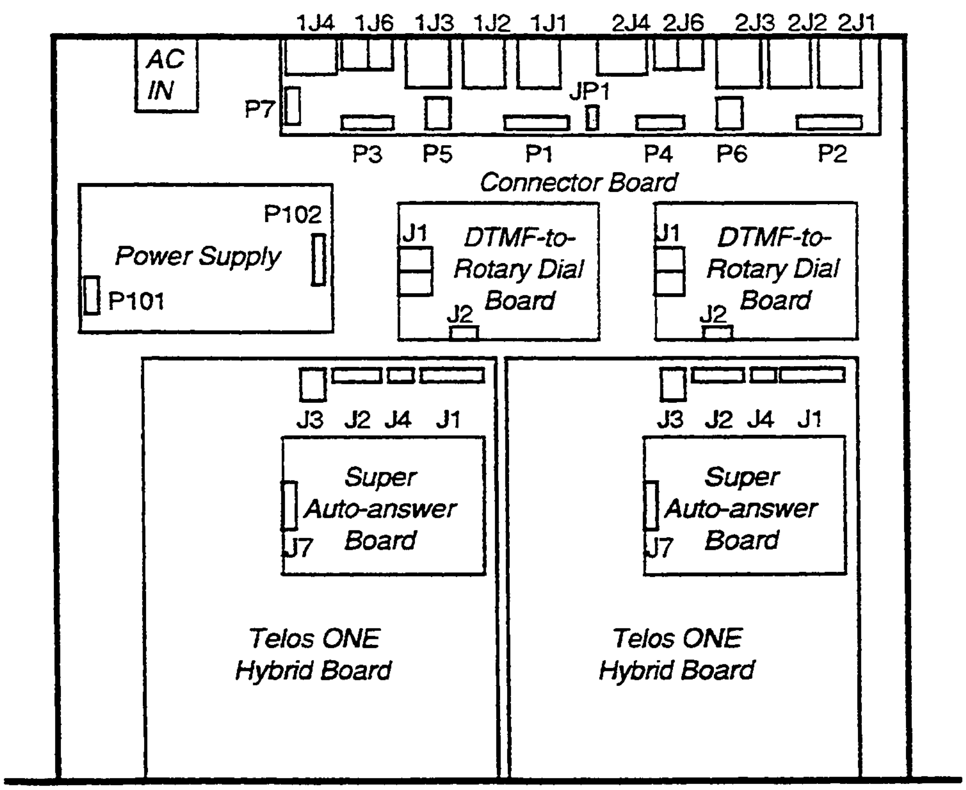


The Super Auto-Answer board's DB15 connector mounts through a cutout on the rear panel.

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.

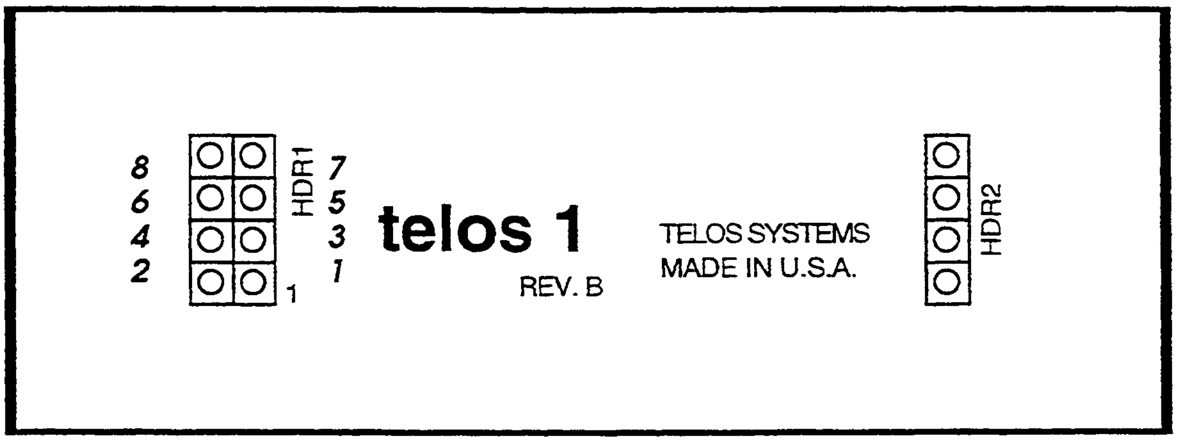
Telos ONE plus ONE

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.)

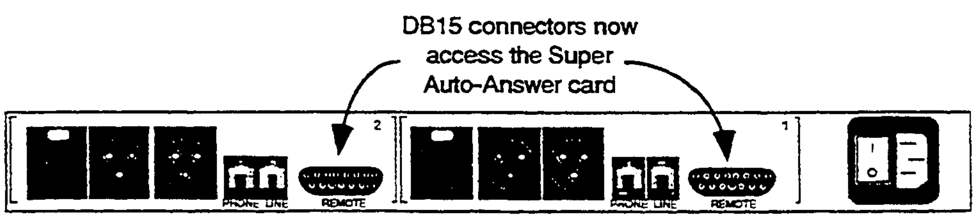


HDR1 & HDR2 locations and pinout

3) 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.

4) 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

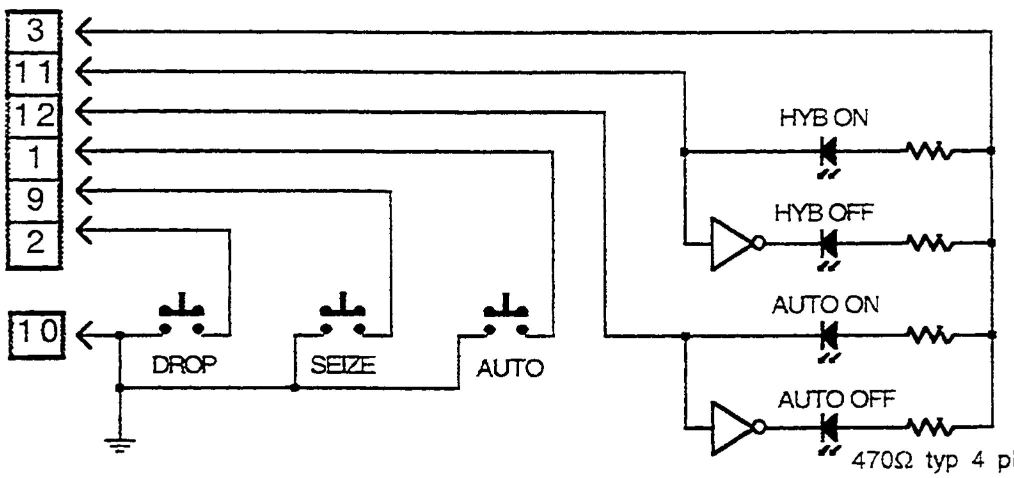
Remote control of the hybrid's ON/OFF functions, as well as the auto mode, are available on the rear panel DB15 connector. Remote buttons are momentary and ground-common. (That is, they are normally pulled high through a resistor and taken to ground to activate.) Refer to the connector pin-out table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

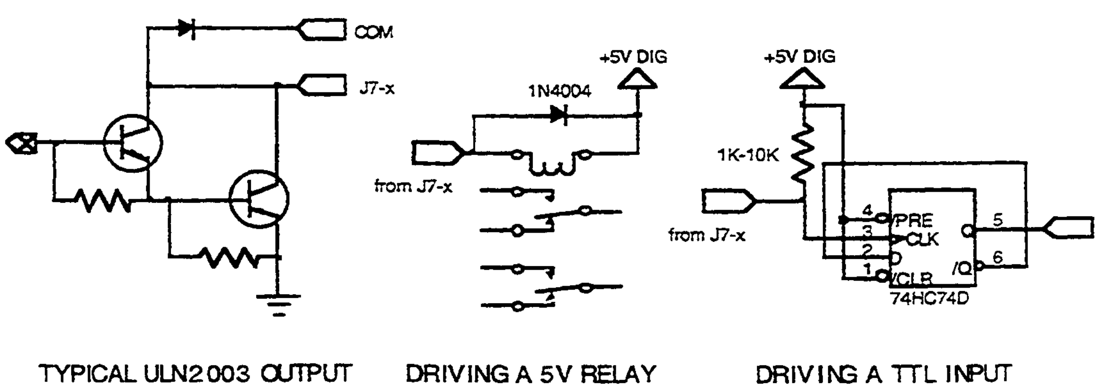
The AUTO input also requires a momentary pushbutton, but is made to be alternate action in operation. Each time the button is pressed, the auto mode will toggle to the opposite state. Power-up is to the "no auto" mode (unless DIP #1 is active).

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.



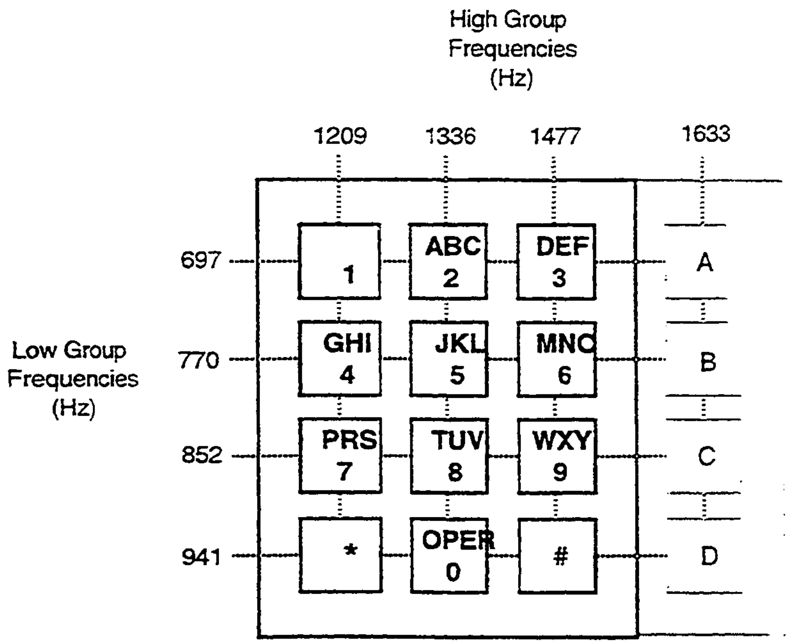
Schematic for remote control. Pin numbers are for the 15-pin D-connector. The inverter should be a 74LS14. The pull-up resistors' value depends on the type of LED used. Check the manufacturer's data sheets for details.



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.

These software versions have been optimized to work in conjunction with ClearCom Matrix Plus digital intercom systems, where AA V1.3 is for domestic use and AA V5.3 is for use in Italy. Hybrid control is extended to the intercom system through remote status and control lines. The status line is programmable to send line status, ring cadence, or auto-answer occurrences, and the control line will accept hybrid ON (high or logic 1) and OFF (low or logic 0) commands. These lines are connected to the intercom system as shown in Figure 1. For proper operation, the interface cable must include a 10kΩ pull-up resistor on the status line output and a 720Ω pull-down resistor on the control line input.

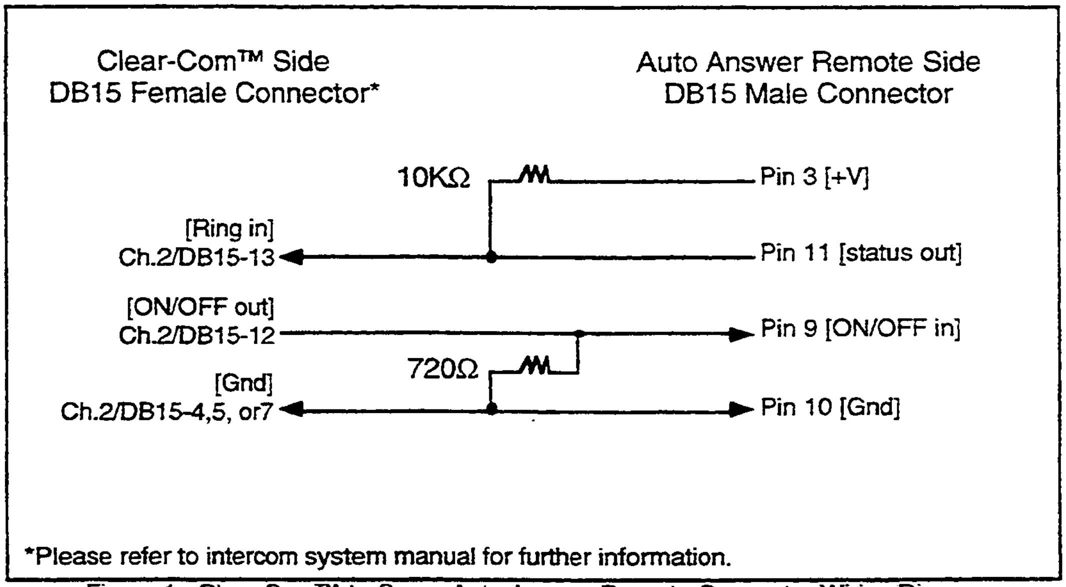


Figure 1. Clear-Com™ to Super Auto Answer Remote Connector Wiring Diagram

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	OFF	•no delay from remote off->hybrid off
1	ON	•12sec delay from remote off->hybrid off
2	OFF	•ON/OFF remote input is active
2	ON	•ON/OFF remote input is disabled
3	OFF	•disable auto answer/disconnect
3	ON	•enable auto answer/disconnect
4	OFF	•basic remote status (see text)
4	ON	•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 V1.42 (65)

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

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

- Dial tone release has been changed to release the phone line whenever a tone cadence of 425Hz, 0.75 sec. on, 0.75 sec off is detected (the Singapore dial tone). 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.

All DIP switch options are identical to AA V1.42 as well as all the I/O signals on the 15-pin connector, including remote on/off/auto, their respective outputs, and the decoded DTMF. Because the Call Progress Detector chip (U5) cannot detect 425 Hz, its crystal frequency reference must be changed to 2.4576 MHz. The original 3.579 MHz crystal must be reconnected to the MPU at U9 pins 10 & 11 (along with 22pF capacitors on each leg to ground) for the processor and DTMF detector to function properly. R13 must be changed to 20KΩ for more reliable detection of disconnect tones, as well. Refer to the schematic "SuperAutoAns Rev A (61) 425Hz detect" for details.

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

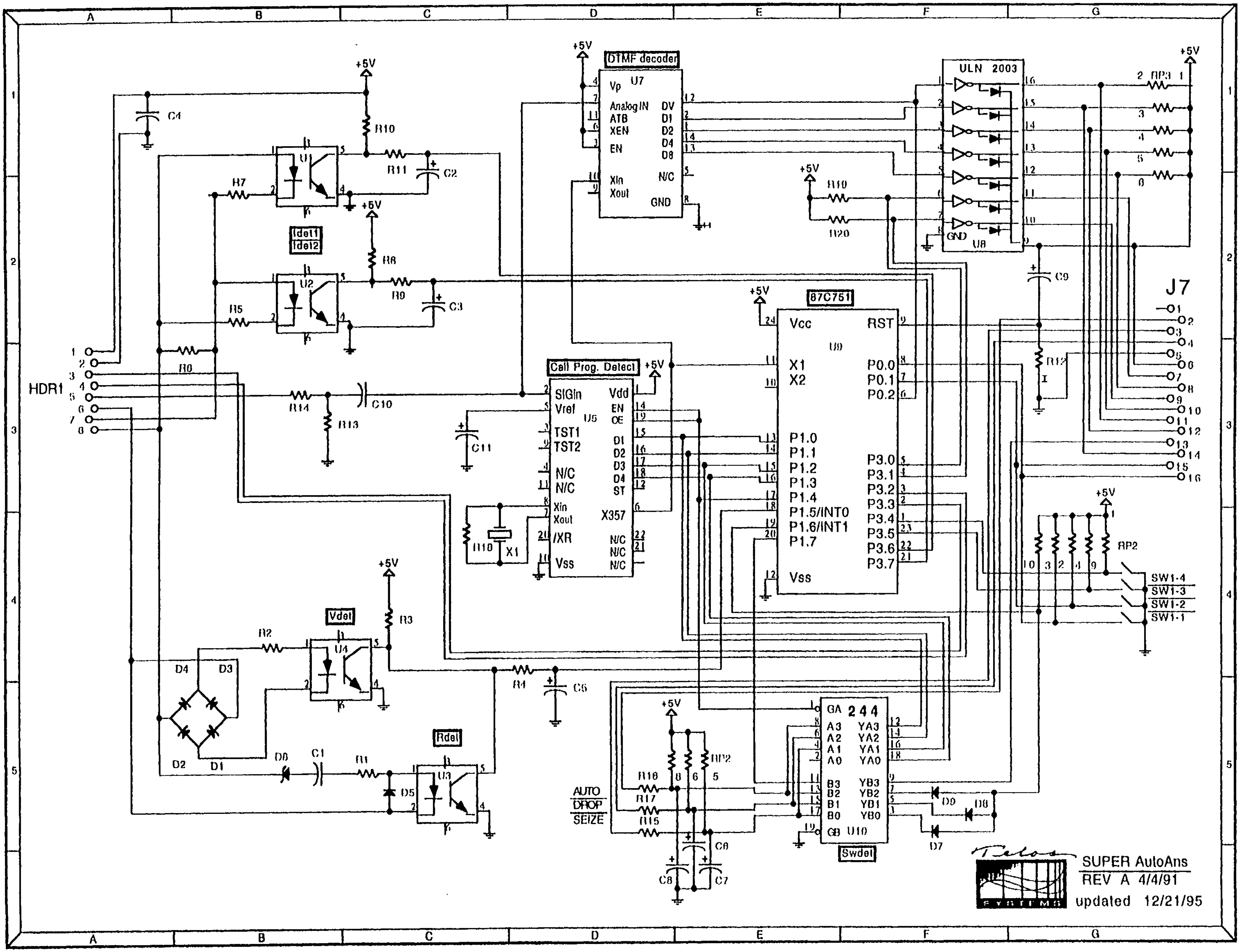
PARTS LIST

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	33KΩ 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

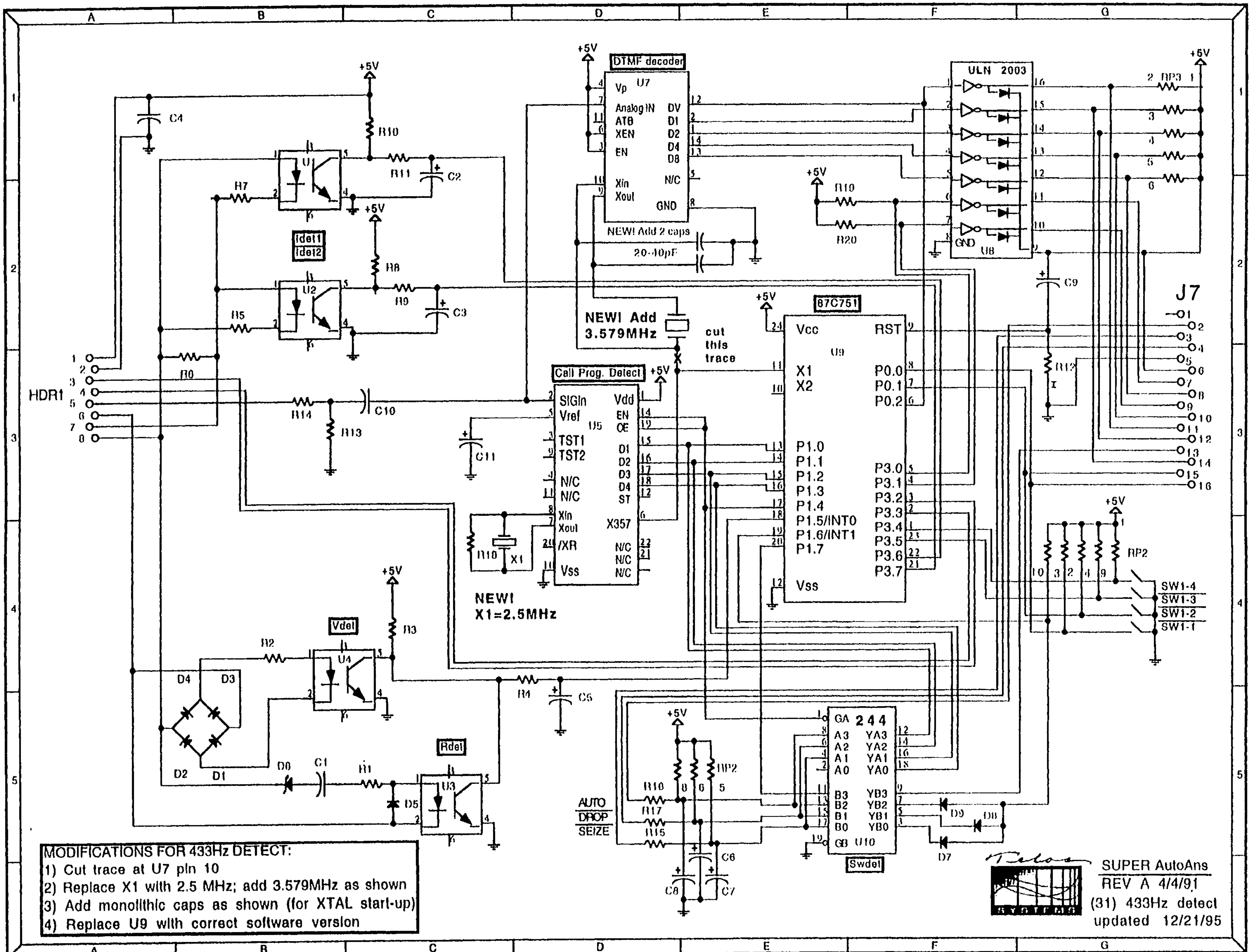
Designation	Description
SU1-4	6-pin socket
SU5	22-pin socket (400mil)
SU6	8-pin socket
SU7	14-pin socket
SU8	16-pin socket
SU9	24-pin socket (300mil)
SU10	20-pin socket
SW1	4-position DIP SW

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.



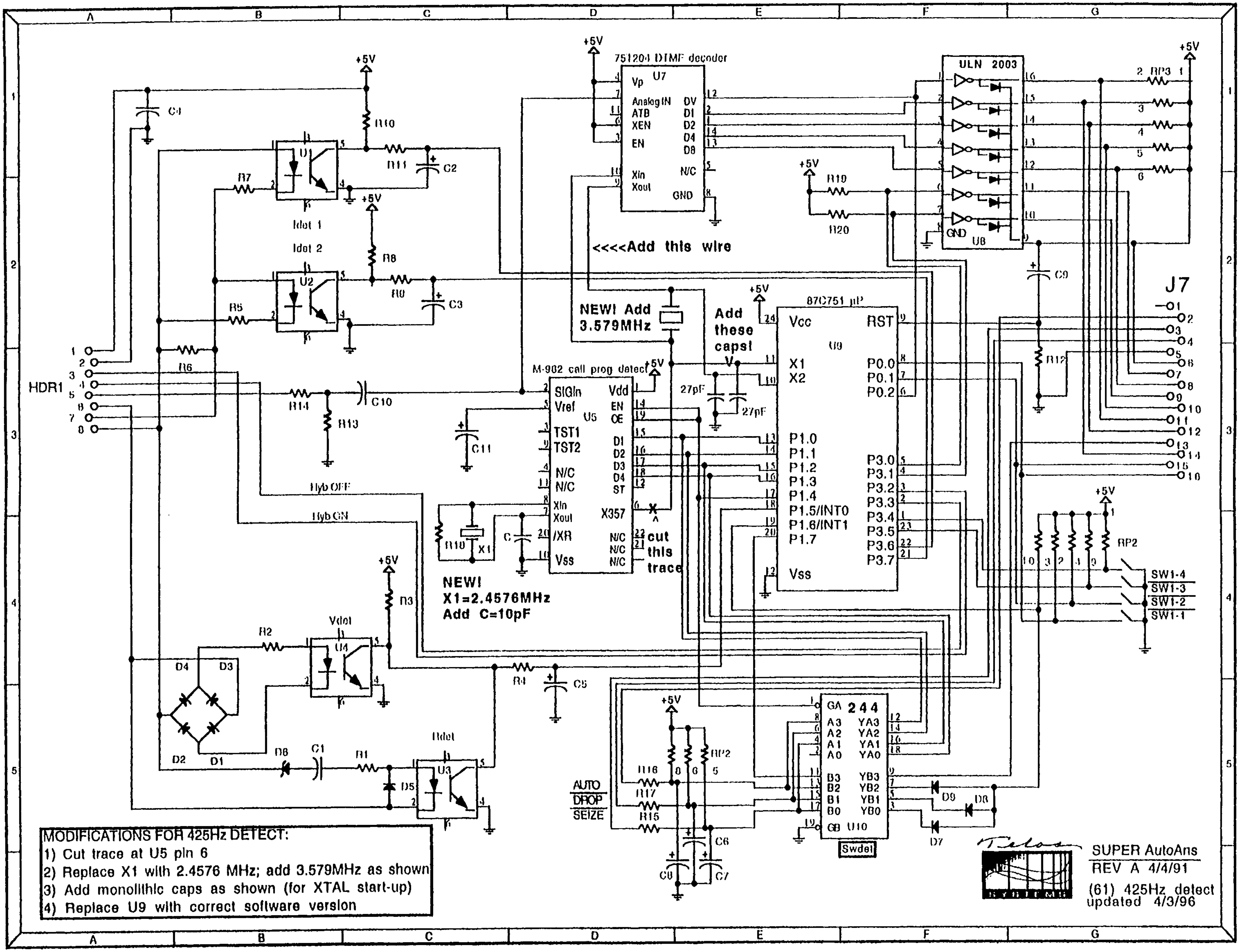
DTMF decoder
U7
Call Prog. Detect
U5
87C751
U9
ULN 2003
U8
244
U10
Swdet
Idet1
Idet2
Vdet
Rdet
HDR1
J7
SW1-4
SW1-3
SW1-2
SW1-1
AUTO
DROP
SEIZE
SUPER AutoAns
REV A 4/4/91
updated 12/21/95



MODIFICATIONS FOR 433Hz DETECT:

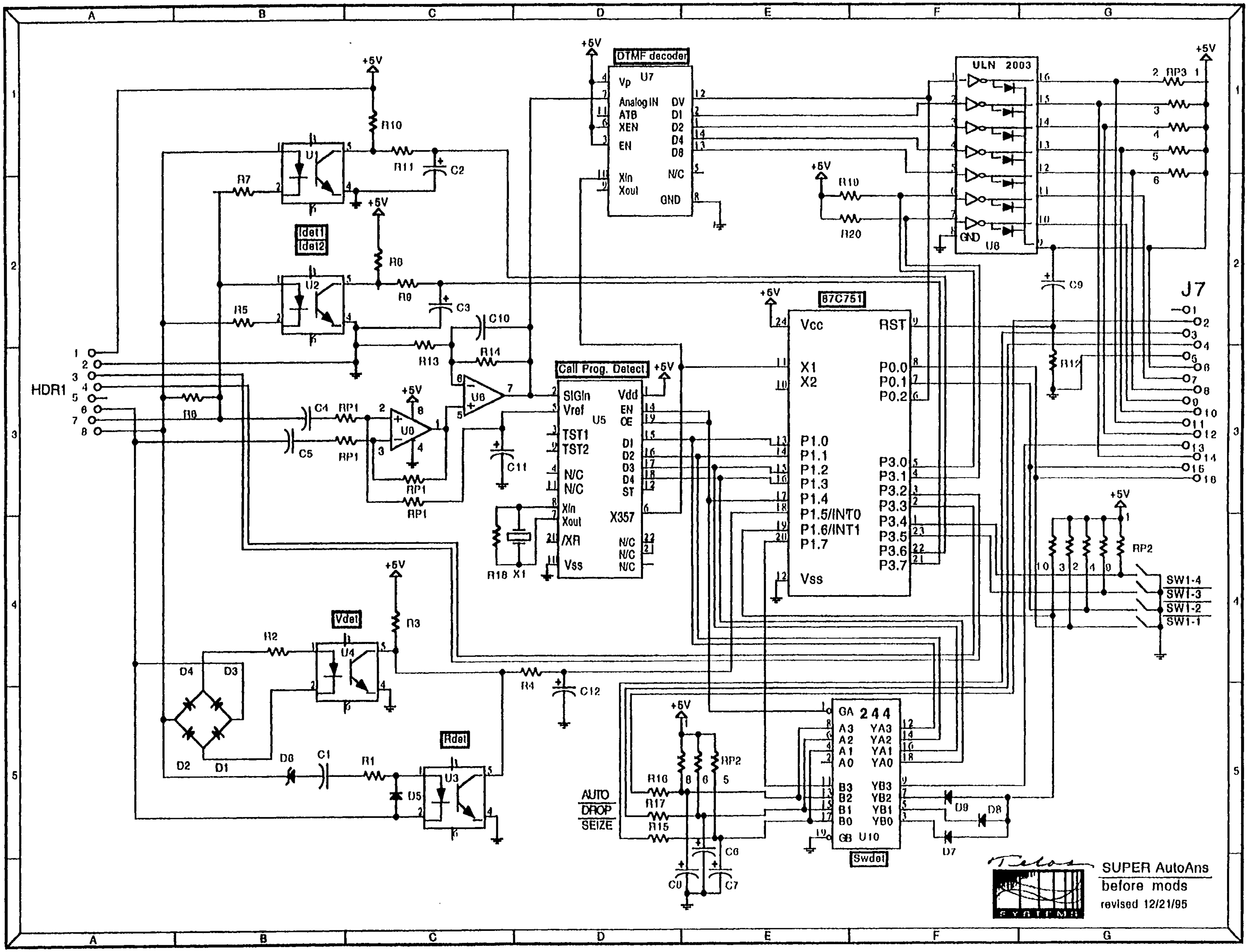
1) Cut trace at U7 pin 10
2) Replace X1 with 2.5 MHz; add 3.579MHz as shown
3) Add monolithic caps as shown (for XTAL start-up)
4) Replace U9 with correct software version

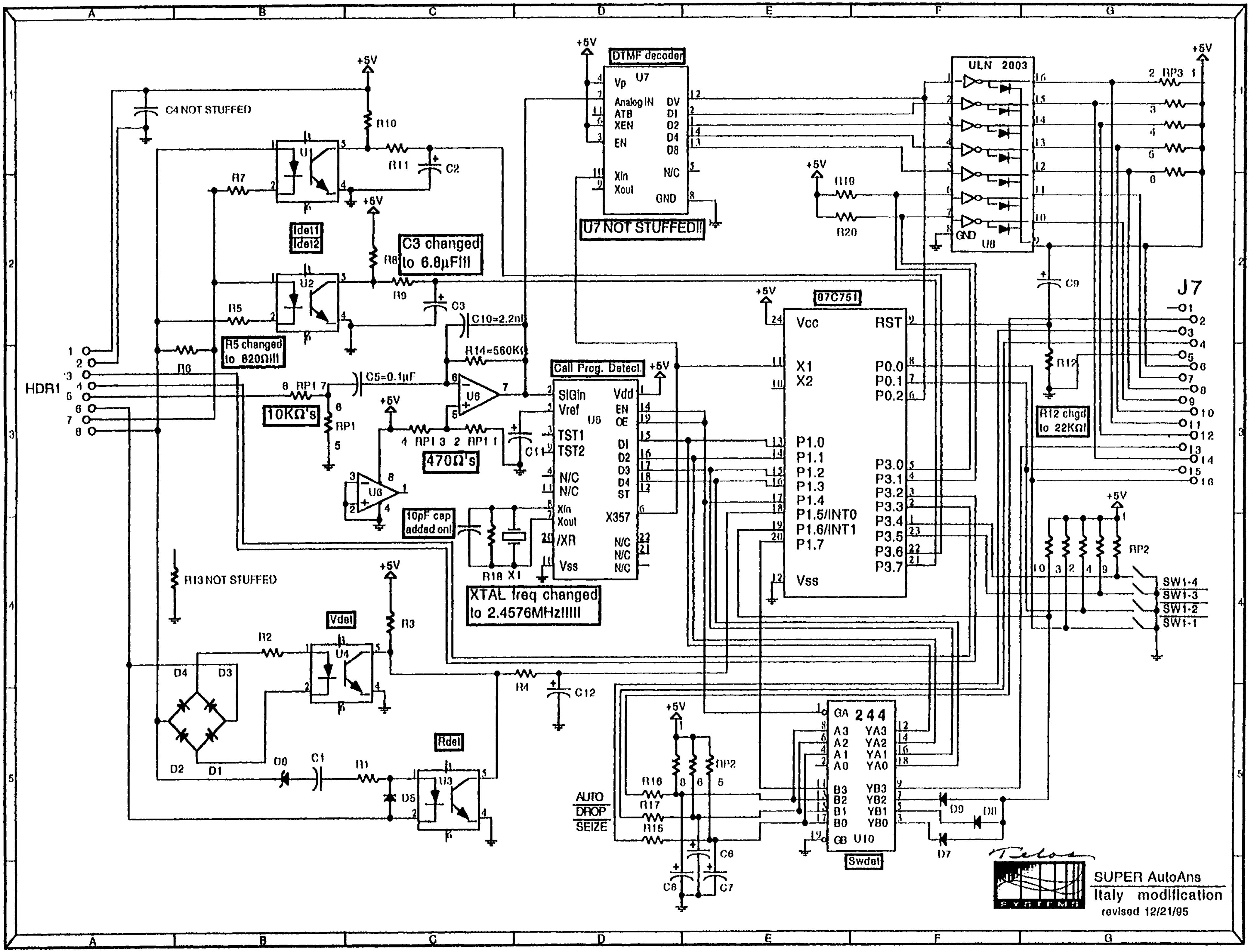
SUPER AutoAns
REV A 4/4/91
(31) 433Hz detect
updated 12/21/95

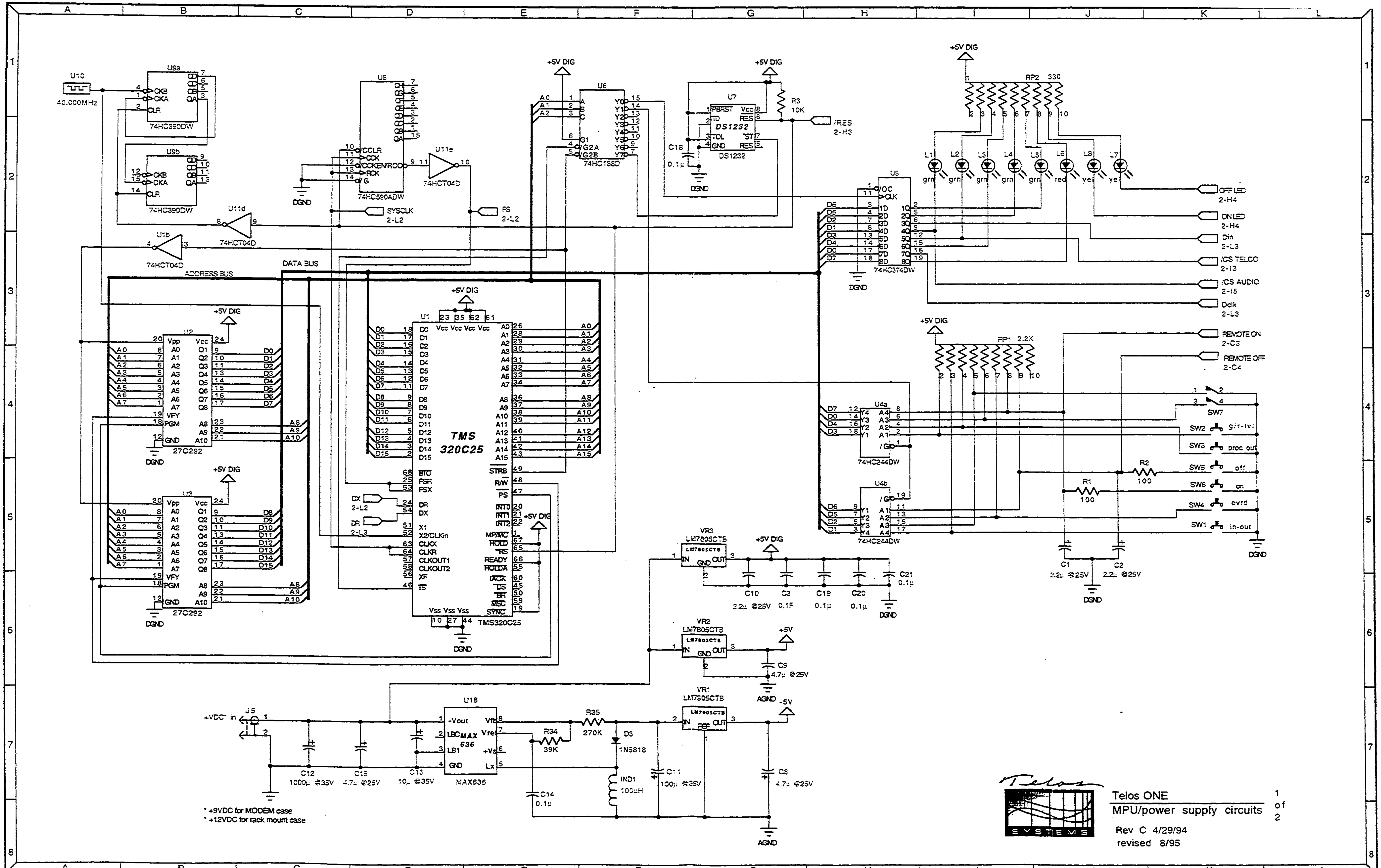


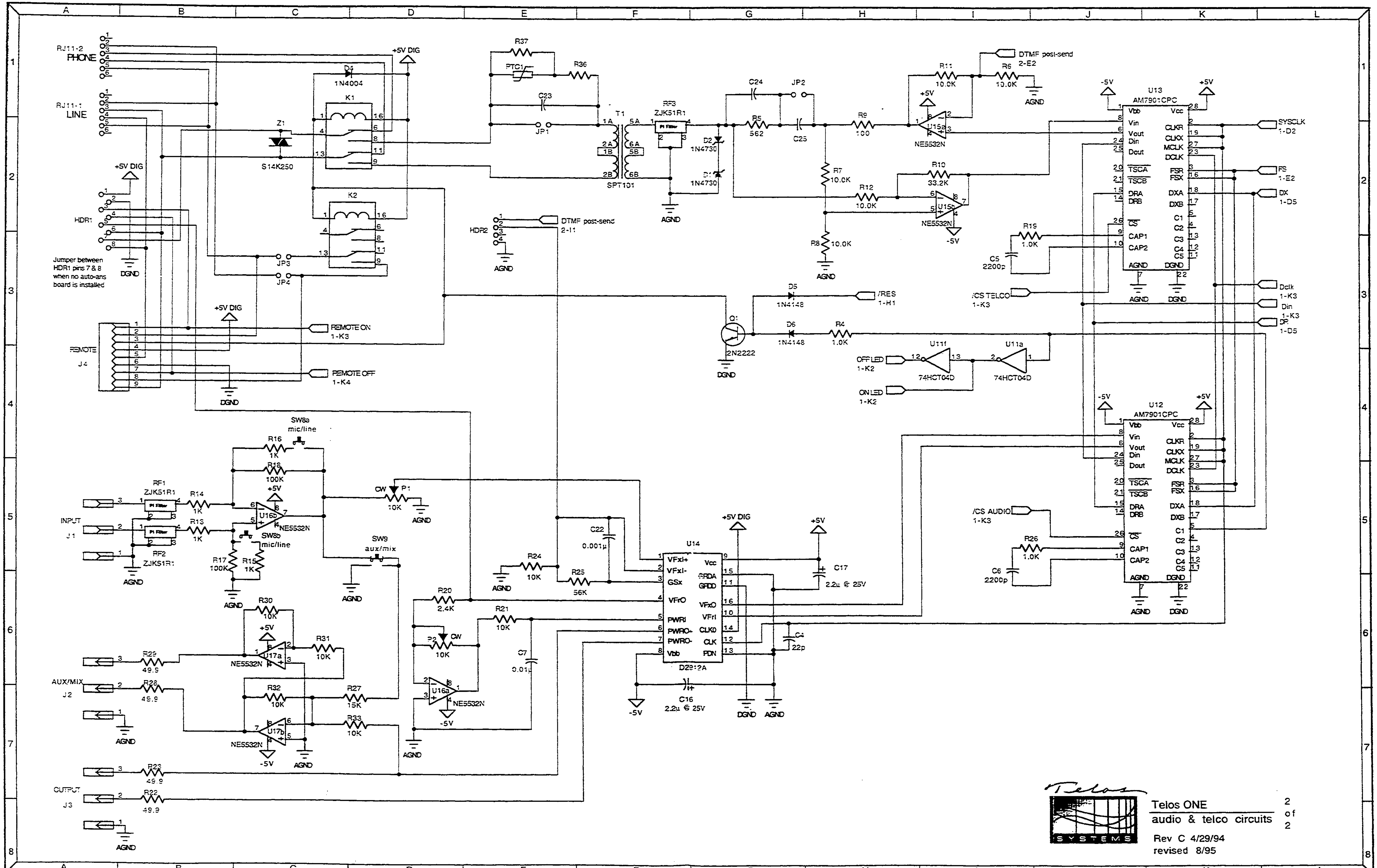
MODIFICATIONS FOR 425Hz DETECT:
1) Cut trace at U5 pin 6
2) Replace X1 with 2.4576 MHz; add 3.579MHz as shown
3) Add monolithic caps as shown (for XTAL start-up)
4) Replace U9 with correct software version

SUPER AutoAns
REV A 4/4/91
(61) 425Hz detect
updated 4/3/96


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


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


U10
40.000MHz
U9a
74HC390DW
U9b
74HC390DW
U6
74HC590ADW
U11e
74HCT04D
U11d
74HCT04D
U1b
74HCT04D
SYSCLK
2-L2
FS
2-L2
+5V DIG
74HC138D
U7
DS1232
R3
10K
/RES
2-H3
C18
0.1µ
DGND
RP2 330
L1 L2 L3 L4 L5 L6 L8 L7
grn grn grn grn grn red yel yel
U5
74HC374DW
OFF LED
2-H4
ON LED
2-H4
Din
2-L3
/CS TELCO
2-I3
/CS AUDIO
2-I5
Dclk
2-L3
REMOTE ON
2-C3
REMOTE OFF
2-C4
RP1 2.2K
SW7
U4a
74HC244DW
U4b
74HC244DW
SW2 g/r-lvl
SW3 proc out
SW5 off
SW6 on
SW4 ovrd
SW1 in-out
R2
100
R1
100
C1
2.2µ @25V
C2
2.2µ @25V
DATA BUS
ADDRESS BUS
U2
27C292
U3
27C292
U1
TMS 320C25
TMS320C25
DX
2-L2
DR
2-L3
VR3
LM7805CTB
C10
2.2µ @25V
C3
0.1F
C19
0.1µ
C20
0.1µ
C21
0.1µ
VR2
LM7805CTB
+5V
C9
4.7µ @25V
AGND
VR1
LM7905CTB
-5V
C8
4.7µ @25V
U18
MAX636
R35
270K
R34
39K
D3
1N5818
C11
100µ @35V
IND1
100µH
C14
0.1µ
J5
+VDC* in
C12
1000µ @35V
C15
4.7µ @25V
C13
10µ @35V
* +9VDC for MODEM case
* +12VDC for rack mount case
Telos SYSTEMS
Telos ONE
MPU/power supply circuits
1 of 2
Rev C 4/29/94
revised 8/95



RJ11-2 PHONE
RJ11-1 LINE
+5V DIG
D1 1N4004
K1
K2
Z1
S14K250
HDR1
Jumper between HDR1 pins 7 & 8 when no auto-ans board is installed
DGND
JP3
JP4
REMOTE J4
REMOTE ON 1-K3
REMOTE OFF 1-K4
R37
PTC1
R36
C23
JP1
T1
SPT101
RF3 ZJK51R1
D2 1N4730
D1 1N4730
AGND
HDR2
DTMF post-send 2-I1
C24
JP2
R5 562
C25
R9 100
R7 10.0K
R8 10.0K
R11 10.0K
R6 10.0K
DTMF post-send 2-E2
U15a
NE5532N
R10 33.2K
R12 10.0K
U15b
-5V
+5V
U13 AM7901CPC
SYSCLK 1-D2
FS 1-E2
DX 1-D5
R19 1.0K
C5 2200p
/CS TELCO 1-K3
Dclk 1-K3
Din 1-K3
DR 1-D5
D5 1N4148
/RES 1-H1
Q1 2N2222
D6 1N4148
R4 1.0K
U11f
U11a
74HCT04D
OFF LED 1-K2
ON LED 1-K2
U12 AM7901CPC
/CS AUDIO 1-K3
R26 1.0K
C6 2200p
SW8a mic/line
R16 1K
R18 100K
RF1 ZJK51R1
R14 1K
R13 1K
U16b
NE5532N
SW8b mic/line
INPUT J1
RF2 ZJK51R1
R17 100K
R15 1K
P1 10K
CW
SW9 aux/mix
C22 0.001µ
R24 10K
R25 56K
U14
D2912A
C17 2.2µ @ 25V
C4 22p
C16 2.2µ @ 25V
R20 2.4K
R21 10K
P2 10K
C7 0.01µ
U16a
R30 10K
R31 10K
U17a
R29 49.9
R28 49.9
AUX/MIX J2
R32 10K
R27 15K
R33 10K
U17b
R23 49.9
R22 49.9
OUTPUT J3
Telos
SYSTEMS
Telos ONE
audio & telco circuits
Rev C 4/29/94
revised 8/95
2 of 2