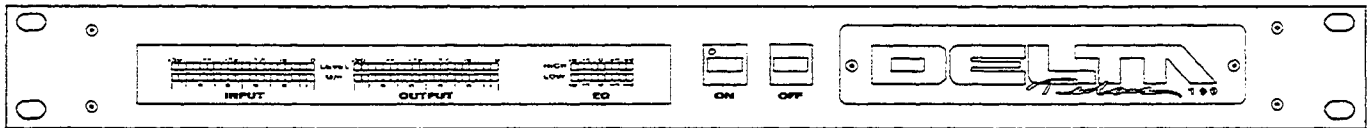
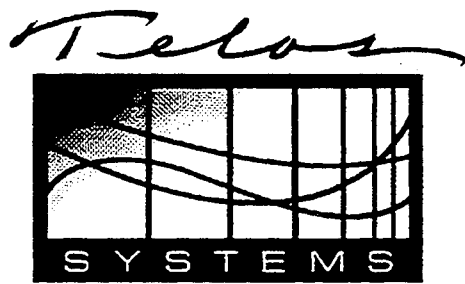


# ***Telos 100 Delta***

## **Digital Hybrid Telephone Interface**



***User's Manual***  
Version B – December, 1994



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## A personal note from Steve Church

October 1, 1992

You have in your possession a remarkable piece of technology. The Telos 100 Delta 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 in 1988 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. In 1989, we introduced the Telos ONE and offered a full DSP based hybrid at a more economical price point.

So here we are in the present, with perhaps the finest telephone hybrid available, the Telos 100 Delta. 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 nine years, and the near-incredible advances in digital audio and low-cost computing power have combined to allow the creation of a result even the digital dreamers would have been shy to predict a decade ago.

We trust you will like it.

Keep on keeping the GM happy...

Steve Church  
President

# User's Manual

## *Telos 100 Delta Digital Hybrid Telephone Interface*

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SECTION 1  
**INTRODUCTION**

## 1.1 OVERVIEW

### *Introduction*

Telos set out to create the ideal broadcast and teleconferencing telephone hybrid. The result of our efforts is the Telos 100 Delta, which establishes new standards for telephone hybrids.

Though it may seem easy, the job of making the output audio from a telephone line “pure” is quite difficult. Matters are complicated because both input and output audio signals are present on a telephone line at the same time. The output signal must be extracted from the telephone line by canceling the input audio, which, when done correctly, is an exacting process.

The process of canceling input audio involves compensating for the telephone line characteristics up to and including the telephone at the caller end. Because of this sensitivity, the circumstances of how and from where a call is placed will provide unpredictable line characteristics to the hybrid. In order to exactly cancel or null the input audio, multiple fine adjustments must be made that compensate for everything encountered on the phone line.

It has always been Telos’ approach to have these adjustments accomplished automatically under computer control using digital signal processing (DSP). Starting from the ground up, the 100 Delta was created to have more DSP power than previous Telos hybrids. This set the stage for a true digital, fourth generation telephone interface that improves upon a level of performance Telos users have come to enjoy while also expanding upon the features available.

### *Special Features*

With extra DSP power, the 100 Delta can address the creation of “pure” caller audio with greater accuracy. The result is an output which can be used for smooth, natural, simultaneous conversation. Once the output audio has been extracted, further DSP power is used for sophisticated audio processing to automatically compensate for telephone line variances and suppress feedback.

Among other innovations, Telos is introducing a totally new technology with the 100 Delta in the form of dynamic digital equalization. The caller audio is analyzed in three frequency bands and, if anything on the line, from the caller’s telephone handset down to the incoming phone line, causes undue tonal degradation, the high and low bands will adjust themselves relative to the middle band to compensate.



Previous Telos telephone hybrids have employed smart automatic gain control to normalize telephone level variances. The 100 Delta is no exception. This time, however, there is a twist to the implementation. The output audio is analyzed and controlled using a logarithmic scale. Using a logarithmic scale as opposed to a linear one provides consistent operation over a greater dynamic range, which is a key issue for automatic gain control where call-to-call levels may vary as much as 30dB. As a matter of fact, all dynamics processing in the 100 Delta is performed on a logarithmic scale.

When monitoring callers through open speakers, the 100 Delta suppresses feedback using two different methods: pitch shifting and duplex range limiting. A subtle, inaudible downward pitch shift is applied to the input audio before it is sent to the telephone line, suppressing the build-up of feedback.

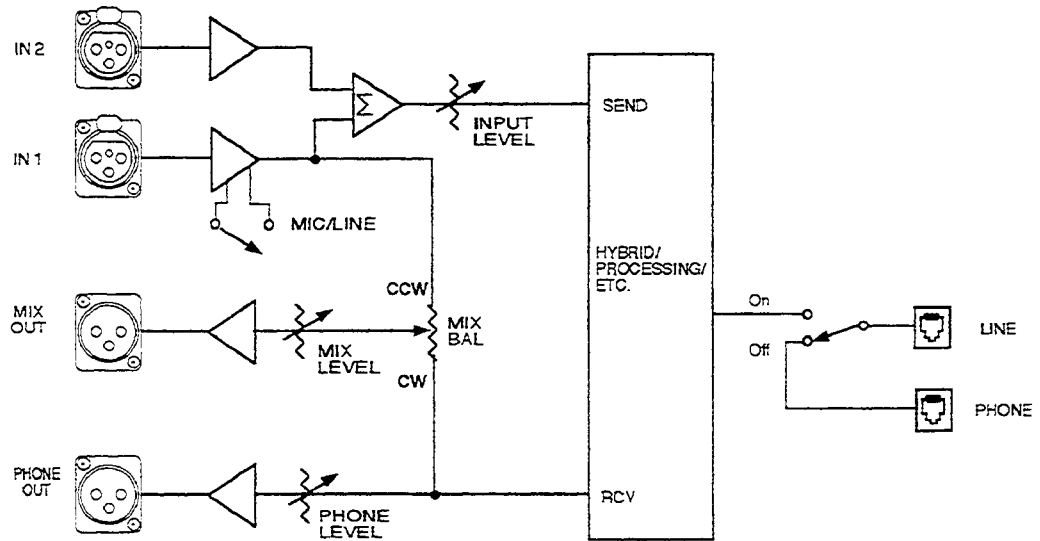
Duplex range limiting is an expanded version of the "override" function of previous Telos hybrids, but again with a twist in the implementation. The degree of two-way simultaneous (duplex) operation is continuously variable from full-duplex to half-duplex. The 100 Delta limits its inherent full-duplex operation by attenuating either the send or receive audio paths anywhere from 0dB to 40dB.

The twist in the 100 Delta's implementation is as follows: the user can program two separate duplex range states, and select either state using a remote control. This allows custom-tailored operation in an installation where the 100 Delta must sometimes have full duplex operation for optimum quality and at other times have half-duplex operation for feedback suppression.

### ***The Hardware***

All of the above features work together to create the best possible sounding caller audio. Of course, everything is accomplished strictly in the digital domain while the rest of the world is analog by nature. In order to be heard in an analog world, the 100 Delta employs analog-to-digital and digital-to-analog converters of the high quality one would find in a professional digital audio tape recorder.

Because the audio processing functions are all performed in the digital domain, the hardware design of the 100 Delta is quite simple. The hardware serves primarily to get the various signals into the digital domain where the processor does the work.



### **Multi-Line Systems**

For multi-line systems, the Telos 100 Delta may be interfaced to multiple central office lines with our Direct Interface Module. In addition, interface via a standard key system can be accomplished with our 1A2 Interface Module. Both modules have a variety of control surfaces available to enhance functionality. As part of a modular system, the 100 Delta offers many configurations to implement the desired number of lines and hybrids.

### **Reference**

For additional information on general hybrid operation and telephone interface systems refer to the *Telos Telephone Questions & Answers* available at no charge from Telos Systems.

**SECTION 2**  
**INSTALLATION**

## 2.1 GETTING STARTED

### 2.1.1 PHYSICAL CONSIDERATIONS

The Telos 100 Delta mounts in a single-unit high space in a standard 19" rack. No special attention need be paid to cooling or ventilation.

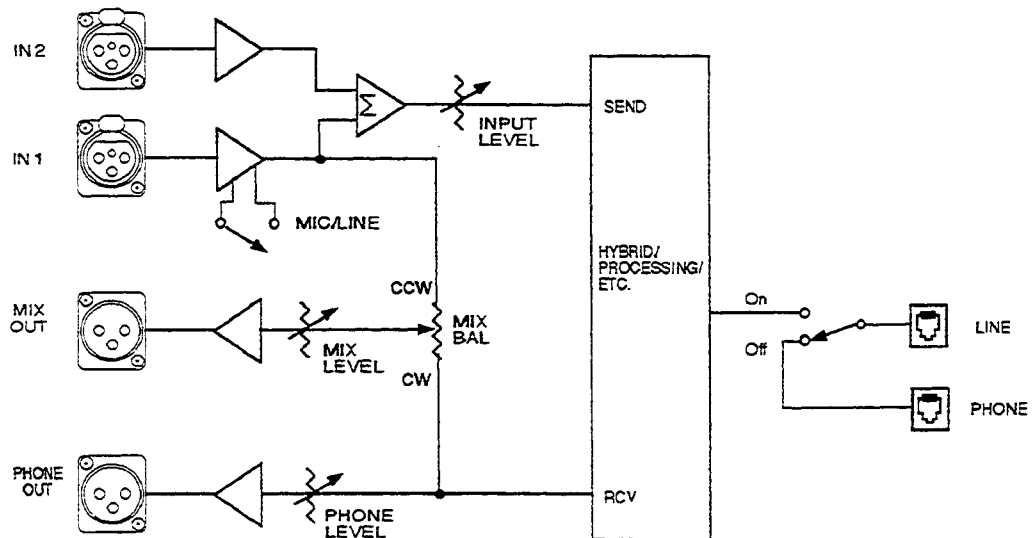
The telephone interface transformer is located near the right rear corner of the unit and should not be placed in the immediate vicinity of any source of significant magnetic interference

### 2.1.2 AUDIO CONNECTIONS

The 100 Delta audio sections were designed to permit integration into many differing studio environments. To this end, we've included:

- Two independent inputs, one of which is switchable to serve either microphone or line levels.
- Two outputs, one of which can be used to obtain a mixed caller and announcer signal.
- Provision for AES/EBU digital I/O (future).

The diagram below shows the relationship between the various signal connections and controls.



## Input Audio Connections



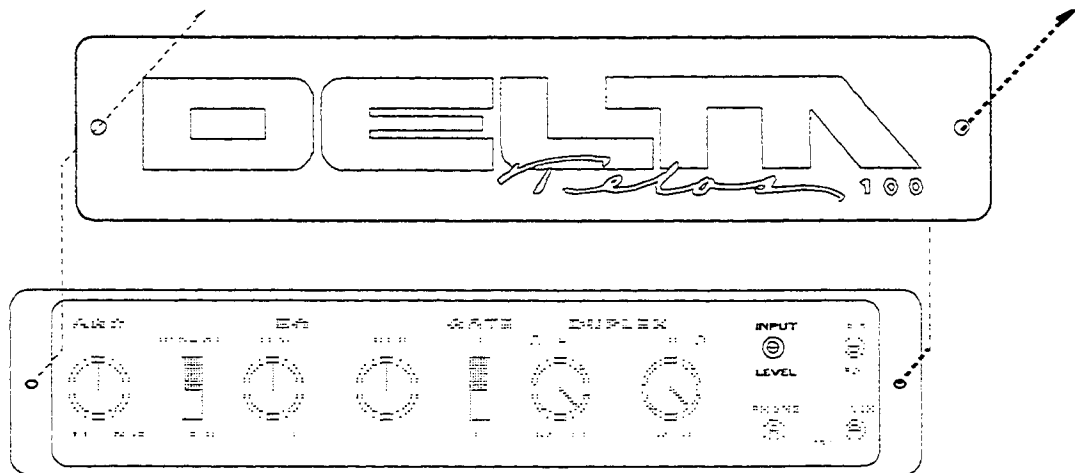
Each input has the following characteristics:

- Active balanced with “instrumentation amp” topology.
- -20 to +18dBv level.
- Switch in MIC position: -70 to -38dBv level (IN 1 only).
- 2K $\Omega$  impedance.
- Pin 1 is ground and pins 2 & 3 are the balanced audio inputs.

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

Signals presented to IN 1 may be either at microphone or line level. The rear panel switch selects the appropriate mode: in for mic/out for line. IN 2 permits only line level.

The precise input signal range is determined by the setting of the front panel INPUT LEVEL trimmer. Since the two inputs are summed prior to gain adjustment, the input level trimmer affects both inputs. Some method for control of one or both levels externally may be required to achieve a proper relationship when both are used.



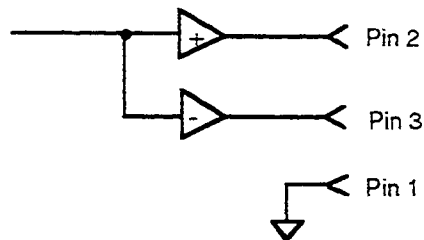
## Output Audio Connections



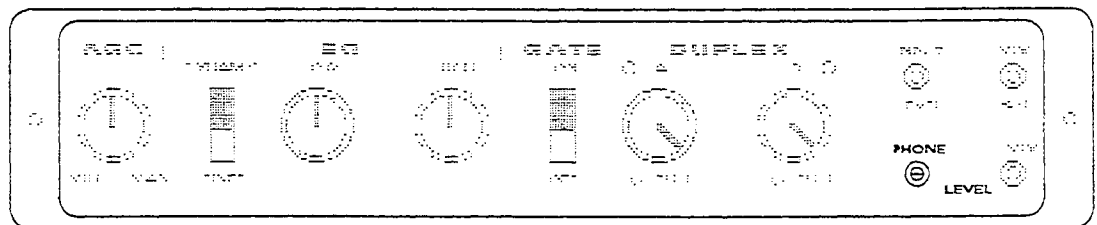
There are two separate and independent audio outputs, each with the following characteristics:

- Active balanced.
- Output level varies from approximately -20dBm to +4dBm depending upon gain control adjustment, caller level and whether or not the AGC is engaged.
- Pin 1 is ground. Pins 2 and 3 are the balanced signal outputs.

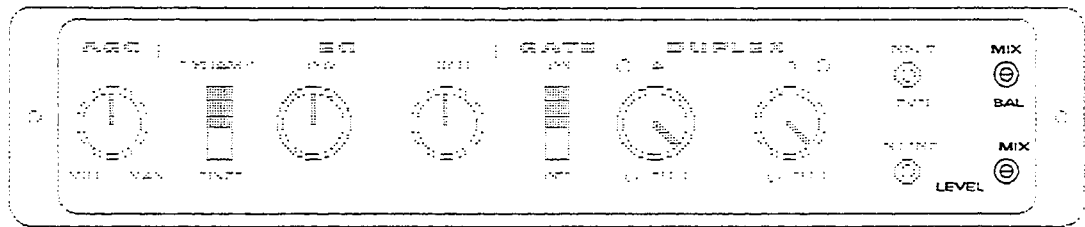
As with any active balanced output with this (very common) topology, if an unbalanced output is required, connect between ground and either of the output pins. Do not ground the unused pin.



The hybrid output consisting of caller audio only appears at the PHONE OUT connector. The level at this output is adjusted with the front panel PHONE LEVEL trimmer.



Audio at MIX OUT may be used as a second isolated caller output or can be a mix of both the caller audio from the hybrid and the announcer send audio. The ratio of the two signals is set by the front panel MIX BAL trimmer. The further clockwise this control is adjusted, the more the caller will predominate in the mix. Therefore, the fully clockwise position is used when it is desired to use the MIX OUT for a second caller only output. The level at this output is adjusted with the front panel MIX LEVEL trimmer.



### 2.1.3 MIX-MINUS

The 100 Delta 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. There are a number of ways to accomplish this.

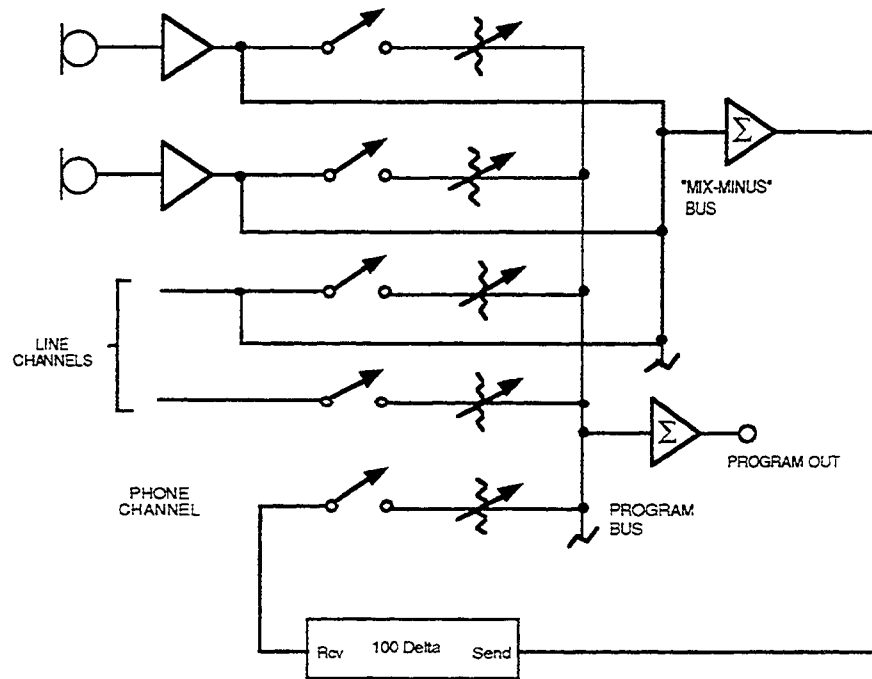
#### ***Simple Mix-Minus***

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

One approach which allows more control is to use an outboard mixer to combine sources as desired. All of the desired sources are paralleled into the on-air board and the mixer and the mixer's output feeds the hybrid.

#### ***Sophisticated Mix-Minus***

Most modern broadcast consoles make some provision for mix-minus. The best allow selective feeds to the telephone system. This is useful since you sometimes want only one mic feeding the phone, sometimes you want three or four mics (during the morning show, for instance), and sometimes you want to feed cart machines when callers need to hear and react to contest effects, etc. The diagram on the next page, illustrating a very simplified portion of a broadcast console with a mix-minus buss, shows the concept in simplified form.

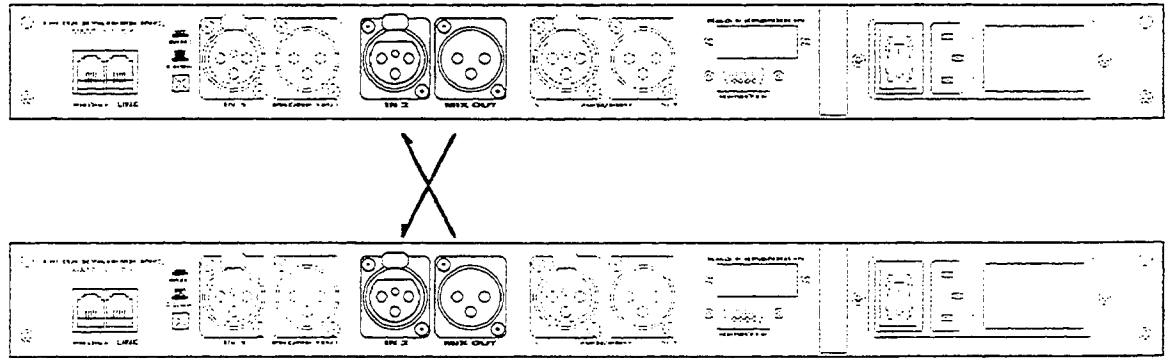


While on the subject of consoles and mix-minus, we'll digress here for a moment. Many hybrid installation problems are caused by an inadvertent signal path which creates a loop from the hybrid's output to its own input. Some console designs allow this to happen when certain control combinations are user-selected. This is the first place to look when strange or erratic performance is experienced. The 100 Delta's front panel input meter should give you an immediate answer.

### ***Using the Delta's Multiple Inputs & Outputs for Conferencing***

A "local" mix-minus interconnecting two hybrids may be created by using the multiple audio I/Os available in the 100 Delta.





In the example above, the upper hybrid's MIX OUT feeds the lower hybrid's IN 2, and vice-versa. Since IN 2 does not appear at MIX OUT, feedback is prevented. The MIX OUT on the upper hybrid consists of both that hybrid's output and the audio presented to IN 1 on that hybrid. Likewise, the MIX OUT on the lower hybrid consists of both that hybrid's output and the audio presented to IN 1 on that hybrid. Therefore, either or both IN 1 inputs can be used as inputs to the complete hybrid system. Each MIX BAL trim pot determines how much of that hybrid's IN 1 signal is passed to the other hybrid. These should be adjusted for a correct balance by applying audio and observing each hybrid's input level meter.

There are a number of ways to take advantage of this local mix-minus arrangement depending upon the particulars of your installation. It is possible to accommodate consoles with one or two mix-minuses and to have either one console fader for both hybrids or one for each.

Contemplating this kind of approach? Get a large sheet of paper, refer to the 100 Delta audio flow block diagram at the beginning of this section and sketch out your plan to confirm that all of the audio gets where it needs to go. Be sure that no hybrid output gets connected to its own input! If you get stuck, give us a call.

### **A Good Idea...**

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

## 2.2 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 the provided modular cord.

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.

### *Auto-Answer Capability*

Telos manufactures two plug-on PC board modules which may be installed inside the hybrid to provide this function.

#### *Basic Auto-Answer*

With this card, the hybrid will automatically switch ON when a line rings and will switch OFF when a loop current drop is detected. This loop current drop, referred to as CPC (Calling Party Control), is present on most phone lines to indicate that the other end has hung-up.

#### *Super Auto-Answer*

This is a sophisticated microprocessor-based sub-system which performs a number of functions:

- Remote control of auto-answer function.
- Selectable number of rings for answer.
- Disconnect on either drop or reversal of loop current.

- “Failsafe” disconnect upon dial tone detection (DIP switch selectable).
- Integral DTMF decoder.

The software-based control logic permits configuration for operation on non-standard and non-USA lines as well. When the Super Auto-Answer PCB is installed in the 100 Delta, a second (DB-15) remote connector is added to the rear panel permitting remote control of the auto-answer functions. These are described in the manual which accompanies the module.

### ***Multi-Line Systems***

Information on use of Telos multi-line control and interface modules is given in the manual included with the appropriate modules. Telos manufactures a full range of line selection, interface, and control equipment. We have interface modules for connection to 1A2 key equipment, as well as for direct connection to telephone lines. Control surface options include table-top consoles and drop-in modules for many studio audio consoles and turrets. We also have a PC-based communications and control system for talk shows with a call screener and/or producer.

### ***“1A2” Key System Installation Without the 1A2 Interface Module***

It is possible to connect the Telos One hybrid to 1A2 key phones without the Telos interface module. With this approach, the key phone is used as the line select device. A special version of the Super Auto-Answer PCB senses the presence of loop current when a button is pressed on the phone and activates the hybrid. Appropriate signals are generated to cause a hybrid re-null when new lines are selected and to turn the hybrid off when the caller disconnects.

### ***Electronic Phone Systems***

These have nearly completely taken over from older mechanical systems. The cable is smaller and cheaper and fancy features are easy to implement, but they present significant challenges to use in the broadcast environment. For this reason, we recommend that your on-air lines be segregated from the office phone system. Nevertheless, Telos’ *Electronic Phone Tutorial* has some information which may be useful if you intend to persevere in an effort to connect to these kinds of phone systems directly.

## 2.3 POWER

The 100 Delta has a “universal” input switching power supply. The unit is immediately operable on any source from 85 - 250vac without changing any jumpers or switches. The AC power is connected to the unified IEC connector/power switch/fuse with the supplied cord set.



A fuse is installed adjacent to the IEC power connector as follows:

115vac	1 A
230vac	5 A

(Note: There is another fuse inside the unit, mounted on the power supply PCB. Since it is rated for 2A, it should not ever blow.)

## 2.4 REMOTE CONTROL

The female DB9-type connector on the rear panel provides access to control functions.



**DB9 REMOTE Connector**  
(Default Mode)

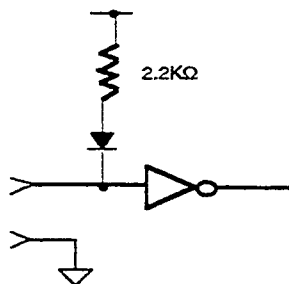
Function	Pin #	Direction
Ground	6	
ON (closure to ground)	1	Input
OFF/MUTE* (closure to ground)	7	Input
ON Status output (low true)	3	Output
Continuous ON (Ground = on)	9	Input
Duplex A/B select (Ground = B)	5	Input

\*Becomes mute input only when Continuous On is active.

**Remote OFF/ON Control**

The OFF and ON inputs require a momentary closure to ground. These inputs are buffered and pulled up with a 2.2KΩ resistor in series with a diode. They may be connected directly to switches or may be driven by an open collector or TTL-compatible logic output. They may also be sourced by up to 24vdc if necessary.

The input circuit is shown below.



**Continuous ON Control**

Another way to accomplish on/off control is with the Continuous ON input. When this input is grounded, the hybrid turns on. The hybrid turns off when the input is ungrounded.

When Continuous ON is active, Pin 7 (normally the OFF control input) becomes an audio mute function. This is a send mute - receive audio remains present.

### **Alternate "Pin 9 = MUTE" Mode**

There is an option available for applications which do not require the Continuous On input capability but need a send mute function. Again, this is a send mute - receive audio remains active. This alternate mode is selected using the "Utility Mode" functions described in the *Operation* section.

DB9 REMOTE Connector  
(Alternate "Pin 9=MUTE" Mode)

Function	Pin #	Direction
Ground	6	
ON (closure to ground)	1	Input
OFF (closure to ground)	7	Input
ON Status output (low true)	3	Output
MUTE (Ground = muted)	9	Input
Duplex A/B select (Ground = B)	5	Input

### **Selection of Duplex Mode**

A unique and useful feature in the 100 Delta is the provision for two selections of *duplex depth*. While the audio aspects of the duplex function are described in detail in the *Operation* section of the manual, we'll cover it here briefly to provide context for the understanding of the remote control of the duplex function.

The term *duplex* refers to the degree to which the hybrid attenuates the inactive audio path. With the front panel DUPLEX control set to FULL, there is never any attenuation and the system behaves as a "pure" hybrid with both the send and caller paths being active at all time simultaneously. With the control set to half duplex, the system behaves like a "speakerphone," with the caller getting fully cut-off when the announcer speaks and vice-versa.

The 100 Delta is equipped with a means for setting two duplex depths with remote control selection of which one is active. In a typical radio studio situation the modes may be selected by a logic connection to the audio console which informs the hybrid whether or not the phone input channel is in an ON or OFF state. When the phone input channel is on, it is presumed that we are working live to air

and thus a more fully duplex condition is to be selected. Conversely, when the channel is OFF, we assume that we are recording the callers and thus we want to have operation closer to half duplex.

Since the remote input for this function can accept either a closure to ground or a voltage, almost all consoles with logic outputs may be connected without any intervening interface. Note that this is required to be a continuous logic signal - active all the while a channel is on. Polarity, however, is irrelevant since the two controls for duplex depth are identical. The LEDs near each front panel DUPLEX control knob indicate which state is active - useful for troubleshooting hook-up problems.

## 2.5 SOFTWARE VERSIONS AND FEATURES

Most every feature in the 100 Delta is the result of it's software programming. As such, the Delta's feature set can be changed or completely re-defined simply by changing the software within the Delta. This allows for two important advantages over a "hardware" approach:

1. Feature improvements/additions are more easily implemented, and
2. Upgradability. Established users can also benefit from current and future improvements.

It has always been Telos' policy to provide free software updates to all of it's users- all the more reason to send in those registration forms!

One of the software features available to all delta users as of v1.8 is the "alternate software" option. In Deltas with software versions 1.8 or above, two completely different hybrid programs reside in the Delta's memory at the same time. This allows for further "personalization" of the hybrid, depending on which hybrid program is used.

Upon powerup, one of the two programs is chosen based on a non-volatile "soft-switch" setting. Refer to the *Operation* section for more information about the utility mode, and soft switch settings.

For specific information regarding the version-dependant features of your 100 Delta, refer to the *software version/feature addendum* included with your hybrid.

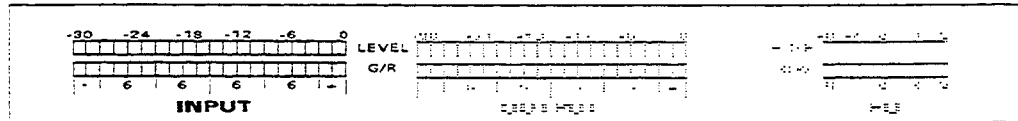
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SECTION 3  
**OPERATION**

### 3.1 METERING

#### *Input Level and Gain Reduction*



The INPUT LEVEL meter is used to set the level for the audio which is input to the hybrid and sent to the caller. It is functionally after the input level trim pot, but before the send AGC.

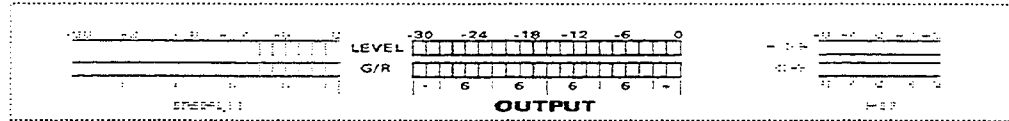
Set for proper send level by adjusting the input level trim pot while observing this meter. The red LED segments should be illuminated only occasionally upon peaks.

Unlike some digital systems - RDATA machines, for instance - the 100 Delta hard digital clip point is 6dB above the 0dB meter indication. Since the green/red junction is at -10dB on the meter, there is 16dB headroom from this point. This headroom, combined with the metering system's fast attack/slow decay characteristics create an "analog VU-meter-like" quality.

*Please note that, due to the action of the automatic gain control, increasing the input level past the green/red junction will not increase the level sent to callers. Furthermore, if the clip point is exceeded, nasty effects are likely to be noticed.*

The INPUT G/R meter shows the activity of the system's send audio dynamics processing functions. Both the AGC and the duplex ducking operating on the send audio are indicated. Thus, the meter indication will reflect the send input level and the setting of the DUPLEX control. The meter markings, incidentally, are relative - the numbers are not referenced to a 0dB.

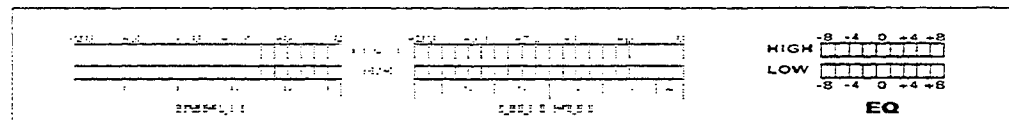
## Output Level and Gain Reduction



The OUTPUT LEVEL meter indicates the caller audio level. Because it is functionally post-AGC, it displays a more consistent level than that which arrives from the telco line. The meter is, however, before the output level trim pots and so does not register any adjustments made with them.

The OUTPUT G/R meter shows the output processing activity. As with the input G/R meter, it reflects both the activity of the AGC and the duplex ducking on the caller receive path. It does not, however, show the effects of the gate function.

## Equalization



The high and low EQ meters indicate the status of the caller audio frequency equalization functions, calibrated in dB. When the dynamic EQ mode is selected, these will vary according to the frequency content of the incoming calls. In the fixed EQ mode, these will simply show the constant boost or loss as determined by the setting of the EQ controls.

### 3.2 ON/OFF CONTROL

When the ON button is pressed, the phone line is seized and the system sends a timed signal to the phone line for alerting and adaption. During this time, the LED in the button illuminates, and the outputs are muted. At the conclusion of the mute/adapt period, the outputs are enabled and the conversation may proceed.

Pressing the OFF button releases the phone line. *When the 100 Delta is in the off*

*condition, the phone line is passed through to the PHONE modular jack.*

### **3.3 SEND AUDIO PROCESSING**

Three subsystems affect the send input audio as it passes to the telephone line. All are active continuously, and no operator control is necessary or required.

- A highpass filter.
- An Automatic Gain Control for dynamics processing.
- A pitch-shifter for feedback reduction.

#### ***Highpass Filter***

The highpass filter improves hybrid performance and enhances send audio intelligibility by removing unnecessary low-frequency information from the wide-band audio likely to be fed to the 100 Delta's input.

#### ***Automatic Gain Control***

The AGC makes the audio fed to the caller more consistent and provides a limiting function to prevent the phone line from being overdriven. This AGC is always active, as it is required to meet regulatory requirements regarding audio levels fed to telco lines. This is a well-crafted feed-forward topology digital AGC with near inaudible "artifacts" below the limit threshold.

#### ***Pitch-Shifter***

The pitch-shifter inserts a small, unnoticeable frequency change into the send path for the purpose of prevention of feedback build-up when the 100 Delta is used in an installation where there is acoustic coupling between the output and input (as would be the case in a radio studio environment where a cue speaker is used for live call monitoring).

## 3.4 RECEIVE AUDIO PROCESSING

These subsystems are designed to enhance the sound in the phone line-to-output path. All are intended to extract the maximum possible audio quality from real-world telco lines.

- A lowpass filter.
- A highpass filter.
- An Automatic Gain Control.
- A "downward expander" gate.
- A switchable fixed/dynamic equalizer.

### ***Lowpass Filter***

The lowpass filter, with its breakpoint at 3.4kHz, removes high-frequency phone line noise and helps to remove the "reconstruction" noise often present on phone lines with poor quality CODECs. (This issue is discussed fully in the *Telos Telephone Q&A*).

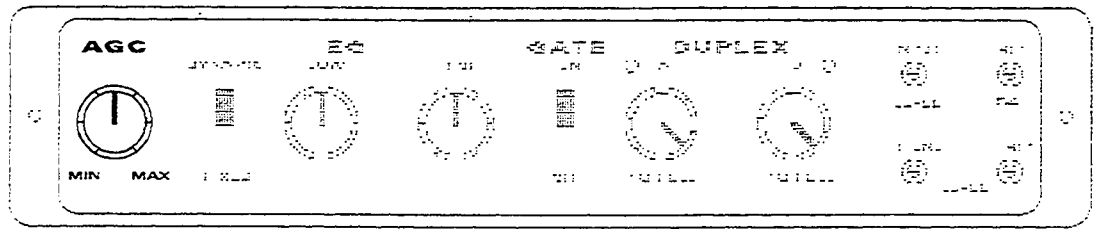
### ***Highpass Filter***

The highpass filter removes hum and other unwanted low-frequency noise from the caller audio.

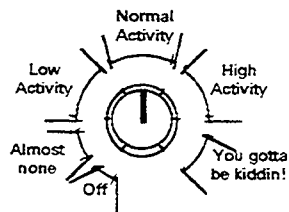
### ***Automatic Gain Control***

The caller AGC serves to improve the level consistency of the phone audio. Nominal levels can vary as much as 30db between calls due to telephone line conditions. The 100 Delta uses an advanced digital logarithmic (dB linear) compressor with a feed-forward topology. This type of compressor provides a sound which is independent of the drive level, important for this application. The "smart" response of the compressor normalizes levels while retaining the natural dynamics of the caller's voice.

An additional feature is the cross-coupling of the 100 Delta's AGC to other sections so that it can reliably distinguish real caller audio from hybrid leakage. This permits fairly aggressive gain control for bringing up low caller levels, while preserving the hybrid's excellent leakage performance.



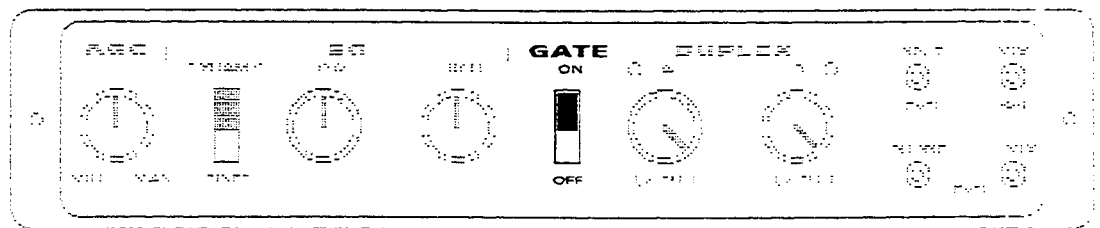
Adjustment of the AGC knob causes a number of parameters to change simultaneously within the AGC function: attack and release time constants, thresholds, and ratio. The AGC will become more effective at normalizing levels from widely disparate calls as the knob is advanced clockwise toward MAX, but with a more noticeable (and possibly desirable) compressed sound. In the full counter-clockwise MIN position, the AGC is completely disabled.



*Note: When multiple hybrids are used for conferencing, it is best to use the least possible AGC to prevent feedback from having excessive gain around the loop - which includes the send and caller AGCs in each hybrid. A setting in or below the "low activity" range is suggested. A more aggressive setting may be used when higher levels of duplex ducking are employed.*

### **Downward Expander Gate**

This function is activated when the GATE switch is in the ON position.



The downward expander serves two functions:

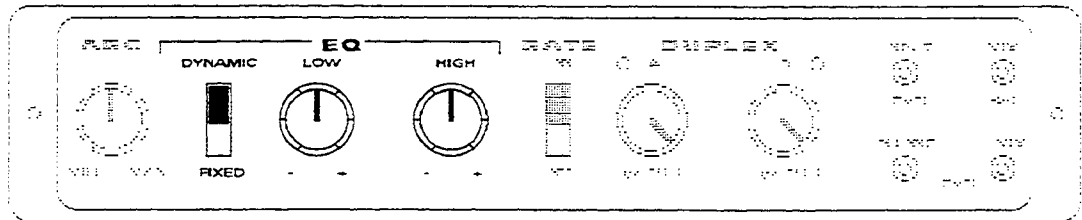
- Reduces low-level phone line noise when the caller is not speaking.
- Aids the reduction of any residual hybrid leakage that might be present.

The threshold is set to about -48dbm telco line level, but varies adaptively under certain conditions.

### **Switchable Fixed/Dynamic Equalizer**

Phone audio spectral balance varies widely. Many factors impact the end-to-end frequency response: The length of the loop at each end, the presence of load coils, the quality of the caller's phone, and many other influences. (We've measured the response on a number of calls - the results were eye-opening!) Thus, equalization is required to obtain good audio from phone lines.

The equalizer in the 100 Delta may be used either in a fixed mode or a dynamically adaptive mode.



#### **Fixed mode**

When fixed mode is selected, the function works as a basic EQ, allowing boost or cut at both the high and low extremes of the 300 to 3400Hz phone audio spectrum. The adjustment of the knob is reflected in the EQ HIGH and LOW LED meters which indicate directly in dB.

#### **Dynamic mode**

The dynamic EQ is one of the unique features of the 100 Delta. When this mode is selected, the proper value of high and low frequency boost or cut is automatically determined and set. We've chosen the frequency breakpoints, time constants, and other characteristics to optimize the tonal quality of telephone lines.

In this mode, the EQ LOW and HIGH controls set the "target" or desired ultimate

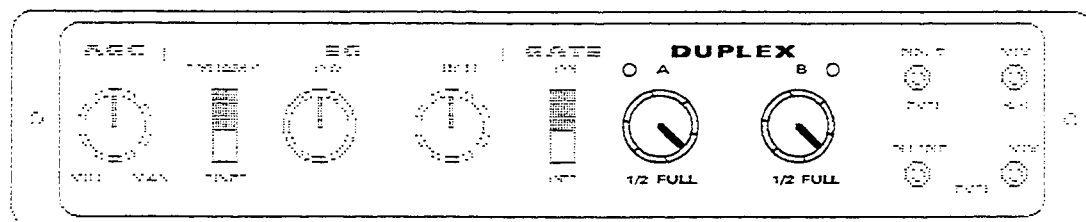
EQ effect. When the knobs are in the “straight up” or zero position, the system will attempt to compensate for bad phone line response by adjusting to what we’ve determined to be a normal or typical spectral mix. *As the knobs are moved from the center position, the system will compensate to normal and then add the additional amount of EQ as set by the control.*

### 3.5 DUPLEX “DUCKING” SUBSYSTEM

The purpose of this subsystem is threefold:

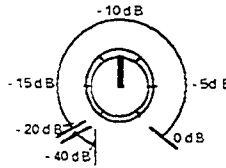
- Allows the announcer to “override” the caller by causing ducking to occur in the receive audio path when he speaks. This is often desired as an aesthetic effect.
- Permit open loudspeaker monitoring of callers without feedback or problems with caller hearing himself via the acoustic coupling from the speaker to the microphone.
- Dynamically enhance trans-hybrid loss to reduce leakage when necessary.

The duplex function inserts a controlled loss, or ducking, into the audio path which is not active at any given moment. When the caller is speaking, loss is inserted into the announcer send path; conversely, when the announcer is speaking, the caller gain is reduced. The effect is somewhat “seesaw-like” with the total loss always being held constant.



The amount of the loss is adjustable using the DUPLEX controls. With the knob counter-clockwise, toward half duplex, the system inserts more loss and thus does not permit as much simultaneity of conversation. Clockwise, toward FULL, the system permits the caller and announcer to speak more concurrently.





The 100 Delta has a novel feature which permits two different settings for duplex ducking - with remote control selection of which one is active. The LED near each knob tells you which is active.

In most every broadcast application these controls will be set at some point between the two extremes. As a rule, a setting for less duplex (more "ducking" attenuation) will be appropriate when open loudspeaker monitoring of calls is desired. This helps the prevention of feedback and is less disturbing to the caller - since in the open speaker and full-duplex condition, the caller is able to hear himself returning via the acoustic coupling of the speaker to the microphone. On the other hand, when headphones are being used to listen to callers, feedback and caller audio return are not a problem, and a more full-duplex operating mode is usually desired.

In many radio stations, there are programs (a "Morning Zoo," for example) where all monitoring of callers is via headphones and therefore a higher degree of duplex is permissible. Fortunately, this condition coincides with a desire to have more of a duplex nature - the talent are more likely to have contests or "bits" where it is necessary to have audio fed to the caller without the caller being ducked off. Later in the day, the phone system may be used primarily to take request calls and the like onto tape for later replay. In this situation, the talent will probably want to listen to callers using the cue speaker and therefore a less duplex operation mode is preferred (more ducking attenuation).

In a typical radio studio situation the modes may be selected by a logic connection to the audio console which informs the hybrid whether or not the phone input channel is in an ON or OFF state. When the phone input channel is on, it is presumed that we are working live to air and thus a more fully duplex condition is to be selected. Conversely, when the channel is OFF, we assume that we are recording the callers and thus we want to have operation closer to half duplex.

In the case where it is not possible to get an appropriate control signal and you need a compromise setting, try around -10db. This amount of loss is just past the noticeable threshold and has the effect of "smoothing" the on-air interaction between caller and announcer and serves also to provide reasonable feedback margin when live loudspeaker monitoring is required.

Note that the A knob is the active one in the default condition where no remote signal is connected for this function.

If you are using a low value for the duplex ducking and if your caller levels are routinely low and/or your phone lines are very bad (with regard to impedance), you may have one of these problems:

- When the gain is advanced on the studio console to compensate for the low level, the hybrid leakage may be more evident and could cause the announcer to sound distorted.
- If you are using live loudspeaker monitoring, feedback could happen.
- If you are using multiple hybrids for conferencing, you may have echo or feedback.

If you have any of these conditions, adjust more toward half duplex. (Also, see the next section, *Feedback Control*.)

### 3.6 FEEDBACK CONTROL

We've taken a number of steps in the design of the 100 Delta to prevent feedback from being a problem. However, when you have the combination of the full-duplex operation that a hybrid such as the 100 Delta permits and strong coupling in the acoustic path between a microphone and a loudspeaker used for live caller monitoring, it is possible in some extreme cases that feedback trouble could occur. This will probably happen only with very poor phone lines, or with very weak callers requiring lots of gain in the phone-to-speaker path.

Some suggestions for solution of this problem:

- Adjust the 100 Delta's DUPLEX controls more toward 1/2.
- 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 reduce the Telos output AGC action since that could reduce feedback margin in the same way input AGC does.

- Try repositioning the mics and/or speaker. Of course, it also helps to use mics and speakers that are directional. In typical broadcast studio application, cardioid mics such as the EV RE-20 and Shure SM-7 have proven appropriate.
- 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.
- If necessary, soften acoustic reflections in your studio by adding curtains or wall treatment.

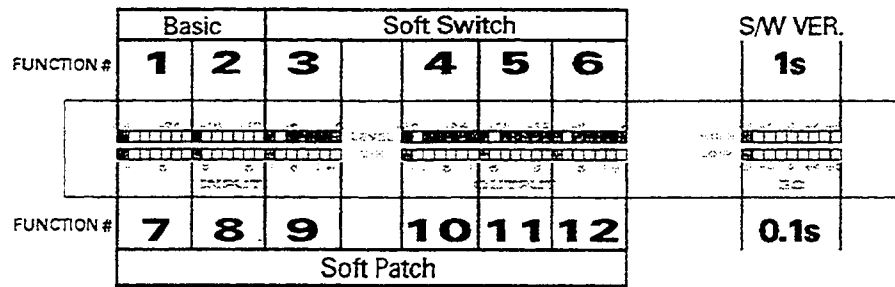
The foregoing is intended to help in those situations where you must have an open speaker. Whenever possible, it is best to use headphones to hear callers. When you have an open speaker, the on-air phone audio has both a direct and an acoustic path - from the speaker to the announce mic(s). Depending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

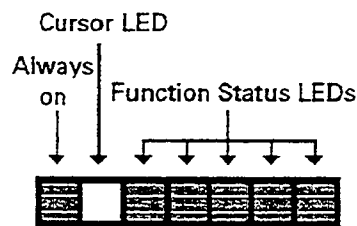
### **3.7 UTILITY MODE**

The utility mode is used to set certain permanent operational modes and to put the unit into various test and debug states. Enter the utility mode and make the desired changes by performing the following actions sequentially:

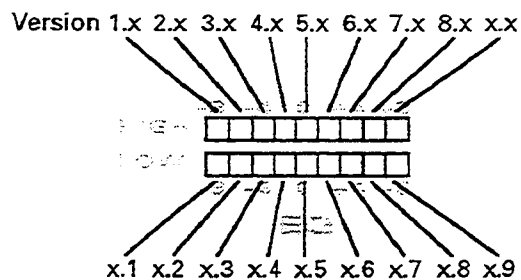
1. Switch the unit off
2. Hold both the ON and OFF buttons down simultaneously, while...
3. Switching the unit back on.



The front panel LEDs which are normally level meters are now function status indicators.



The EQ meter displays the current active software version.



The ON and OFF buttons become special function buttons: the ON button selects the desired function; the OFF button toggles the status of each function.

4. Select what you want to change by pressing ON to move the "cursor" to the desired function. Each time the ON button is pressed, the cursor moves to the next function in sequence (the cursor will wrap back to #1 from #12). The cursor is the flashing LED adjacent to the function status LEDs.
5. Once the cursor is at the function you wish to modify, press the OFF button to toggle that function's state.
6. Repeat steps #4 and #5 as required. After the desired changes are made, the cursor is moved to function #1, SAVE/EXIT and the OFF button then is pressed to store the changes and return the unit to normal operation.

## **Basic Functions**

Basic functions 1 and 2, described below, take effect immediately upon pressing of the OFF button.

### **Function 1: SAVE/EXIT**

When OFF is pressed, all changes are permanently stored in the 100 Delta's non-volatile memory and the unit returns to normal operating state. (Changes are preserved even when no power is applied to the unit.)

### **Function 2: FLASH UPDATE**

Not yet implemented

## **“Soft-Switch” Functions**

Functions 3 through 6, described below, are “soft switch” functions which work very much like dip switches. They set certain modes which accommodate unusual operating requirements. Once set, they remain permanently so - even if power is removed. Pressing the OFF button toggles between off and on modes.

### **Function 3: REMOTE PIN 9 = CONT ON**

When this function is on, pin 9 on the remote connector can be used for a continuous on control input. When the function is off, pin 9 becomes a send mute input. (See Installation section for details.)

Default: ON.

### **Function 4: ADAPT NOISE ACTIVE**

When the function is on, the hybrid sends the adaption noise/tone to the telephone line during the brief mute period just after the hybrid is turned on. This should be on for all normal applications.

Default: ON.

### **Function 5: ALT SOFTWARE ON**

The 100 Delta stores two separate hybrid programs in memory at all times. Upon powerup, one of these two programs is chosen for use (either the main or the alternate software) based on the setting of this function.

When the function is on, the alternate software is used.

Default: ON.

### **Function 6: HYB OFF AT POWER-UP**

When this function is on, the hybrid comes up in the off mode upon power-up. This function is for special applications which require the hybrid to be continuously on.

Default: ON.

## “Soft Patch” Functions

Functions 7 through 12, described below, are “soft patch” functions which switch various audio paths for test and set-up purposes. Unlike the functions described above, pressing the OFF button toggles through three states:

ON > ON with normal level meter operation > OFF

Function 7: OSC->XLR OUT

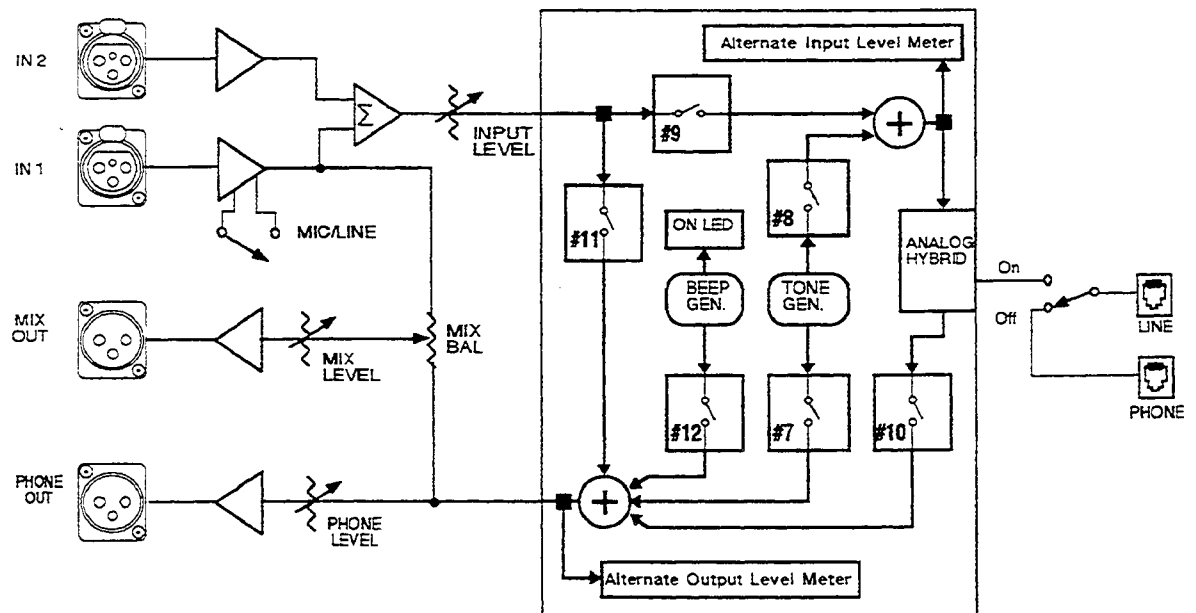
Function 8: OSC ->RJ11

Function 9: XLRin ->RJ11

Function 10: RJ11 ->XLRout

Function 11: XLRin->XLRout

Function 12: BEEP ->XLRout



Just as in the “soft switch” functions, any combination of “soft patches” may be saved by selecting the save/exit function. This can be useful for establishing a “quick” way to generate a test tone, for example. After enabling function 7 (OSC->XLR OUT) and saving it, simply entering the utility mode will activate the test tone.

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SECTION 4  
**TECHNICAL DATA and  
TROUBLESHOOTING**

## 4.1 OVERVIEW

### *Philosophy*

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

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

Thus, we don't really expect that much component-level troubleshooting will occur. So, to support you when you need help, we keep spare boards available for fast overnight shipping. In most cases, we will swap boards with you at no cost.

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 us, you probably just *have* to know what's happening inside the fancy box. So we tell you.

#### **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.

## General Troubleshooting Information

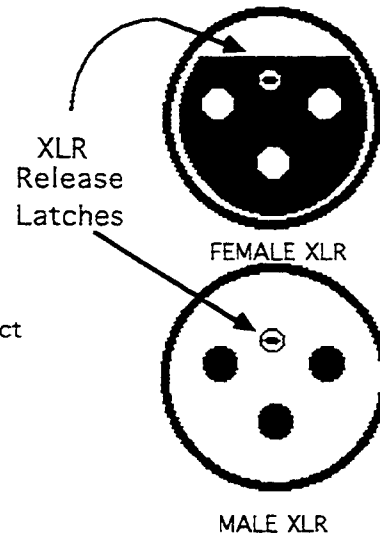
### Removal of the Audio PCB

1. Unscrew the DB9 connector retaining screws.
2. The XLR connectors have retaining screws which have to be turned in order to be released. Do this as illustrated below.

#### XLR CONNECTOR RELEASE:

Insert a small screwdriver into the holes in the connectors, shown at right. Turn the screwdriver about one eighth of a turn counter clockwise to release the connectors. A small screwdriver such as the Xcelite R3322 or R3324 may need to be filed down some to fit the slots.

Remember to retighten the XLR latches when replacing the Telos ONE PCB. This will ensure correct support for the XLR connectors on the PCB.



#### Desoldering

While we socket the IC that have the greatest potential for failure, some of the 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.

#### Notation

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

## ***Digital Signal Processing***

Because the 100 Delta 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.

The analog signals are passed to the digital processing section through high-quality A-to-D converters which contain integral 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 Motorola DSP56001 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.

## **4.2 DIGITAL SECTION**

### **4.2.1 THEORY OF OPERATION**

#### ***The Processor***

There are two independent processors on the CPU board: a Signetics 80C550 and a Motorola DSP56001. The 80C550 is responsible for running the LED display, monitoring the remote plug, switches, knobs, and performing a "watchdog" function over the DSP56001. All digital audio processing occurs within the DSP56001. Although the 80C550 is connected to the DSP56001 through its address and data busses, the two devices are running *asynchronously* with respect to each other. This is possible due to the dedicated parallel port interface internal to the DSP56001. Data is sent between processors using "handshaking" techniques.

#### ***A/D/A Interface***

The analog-to-digital and digital-to-analog converter chips (located on the analog board) are interfaced serially to the 56001 through its on-chip serial port. The serial clock is generated from the 56001's master clock, but frame-by frame signaling is generated in the converter chips.

## ***Clock and Timing***

A free-running 32MHz clock, U12, drives the DSP56001. All timing signals for the DSP-related parts U9-U11 are derived from this clock. Slower clock frequencies are created by dividing the 32MHz down in U13 which drive the 80C550 as well as the audio converter chips.

### **4.2.2 TROUBLESHOOTING THE DIGITAL SECTION**

Check the power supply for the digital 5v output.

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

Verify the main clock signals: 32MHz for the DSP56001, 8MHz for the 80C550, and 2MHz for the audio converter chips. All these signals are present on U13.

Check the data buss 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 audio section schematic in the *Drawings* section.)

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

U7, 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.

### **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-Signetix brand 5532's - *it is normal for these to run hot.*

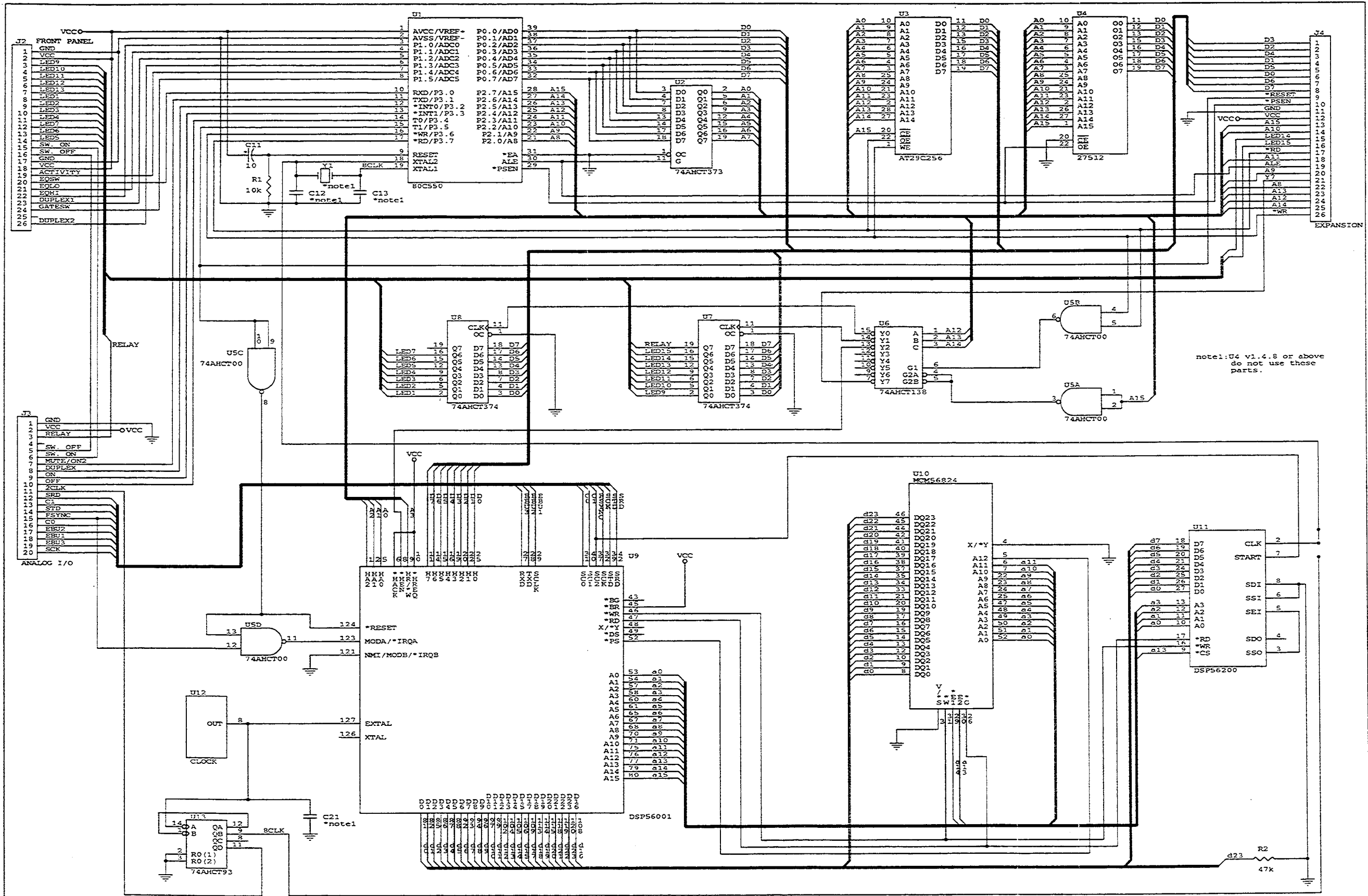
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SECTION 5  
**DRAWINGS**

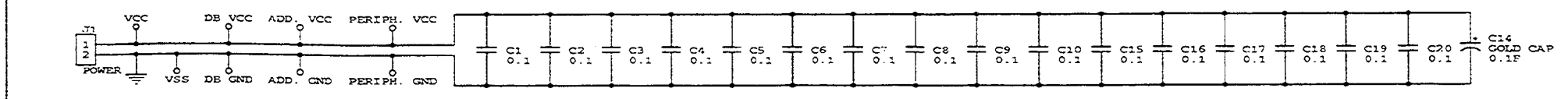
## **List of Drawings:**

- 1. Digital Schematic***
- 2. Audio Schematic***
- 3. Front Panel Schematic***
- 4. Control Panel Schematic***



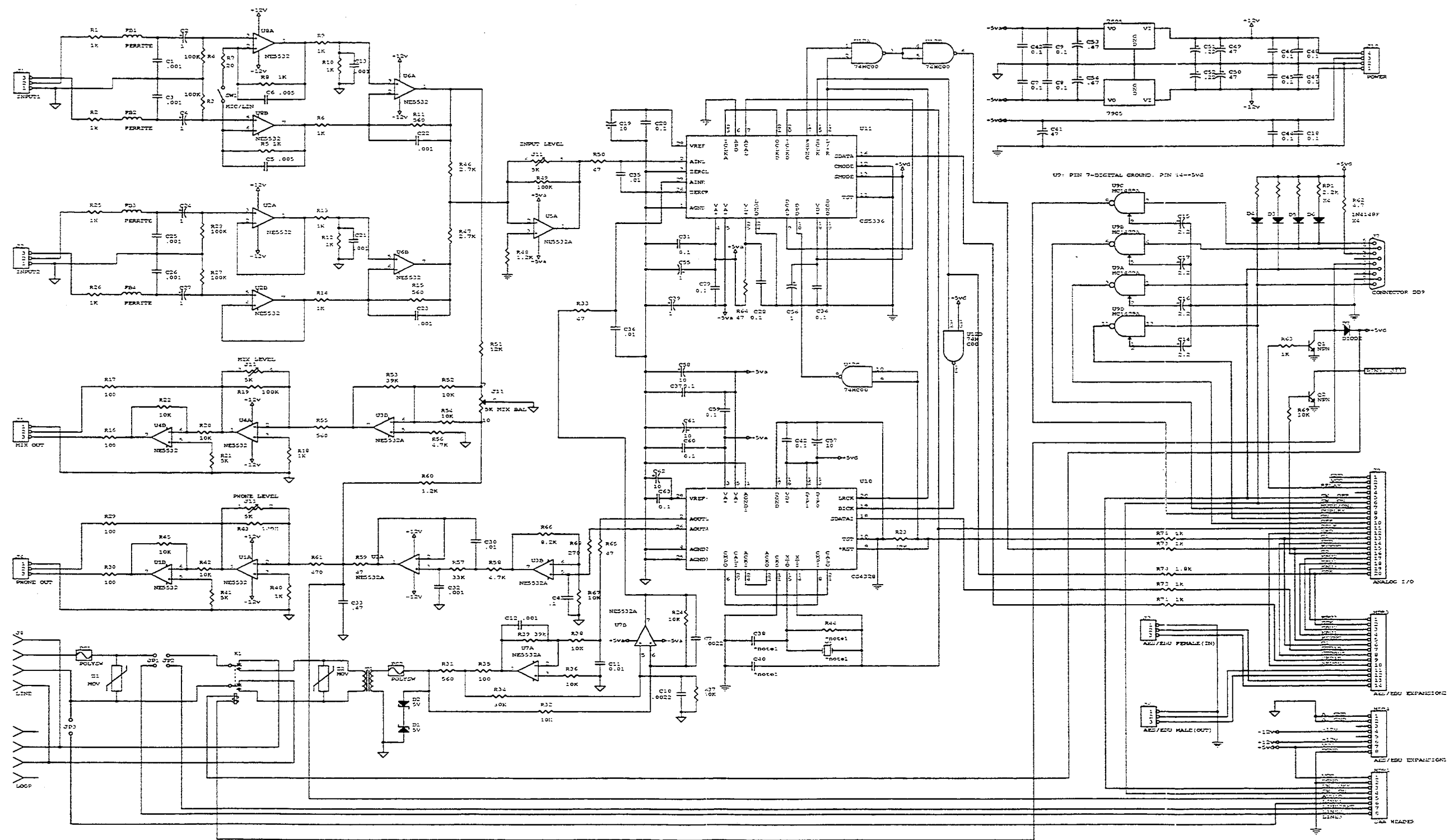


note1: U4 v1.4.8 or above do not use these parts.



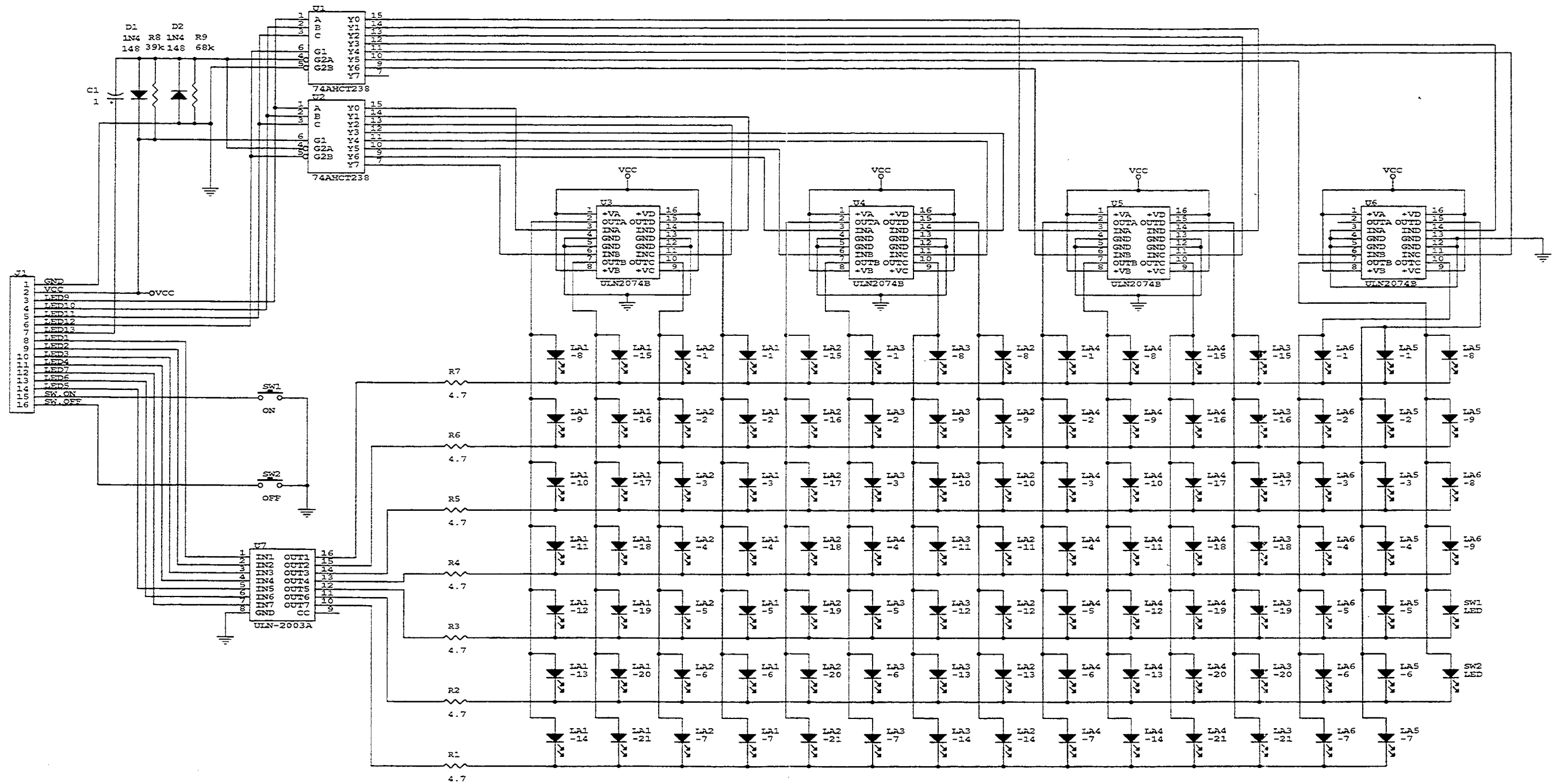
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 Size: Document Number: D\_CPU  
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note: not used

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AUTO DDATE 100 DELTA	
DOCUMENT NUMBER	
10	1



DISPLAY BOARD LED LAYOUT: TOP VIEW

LA1

G	G	G	G	G	G	G	G	G	G	G	R	R	R	R	R	R	R	R	R	
1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21
G	G	G	G	G	G	G	G	G	G	G	R	R	R	R	R	R	R	R	R	R
1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21

LA2

LA3

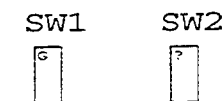
G	G	G	G	G	G	G	G	G	G	G	R	R	R	R	R	R	R	R	R	R
1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21
G	G	G	G	G	G	G	G	G	G	G	R	R	R	R	R	R	R	R	R	R
1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21

LA4

LA5

R	R	R	R	R	R	R	R	R	R	R	R	R	R	R	R	R	R	R	R	R
1	2	3	4	5	6	7	8	9												
R	R	R	R	R	R	R	R	R	R	R	R	R	R	R	R	R	R	R	R	R
1	2	3	4	5	6	7	8	9												

LA6



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Title: DISPLAY BOARD, 100DELTA

Size	Document Number	REV
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Date: December 7, 1994 Sheet 1 of 1