Teletics Application Note

Radio to Teletics Interface

Rev 1.0 – December 2015

RADIO TO TELETICS INTERFACE

CONCEPT

The system described here allows the use of portable radios to access the public address amplifier as well as telephones on the Teletics system. The example shown here will demonstrate a simplex (receive and transmit frequencies are the same) radio system. Diagram A shows a block diagram of the equipment used in this system.

<u>Example A</u> – A person using a portable two way radio with a DTMF pad on it can transmit and press *. When the operator releases the ptt a dial tone will be heard. The operator presses the ptt and enters 99. This will allow the operator to make a public address announcement over the systems loudspeaker system. When finished the operator presses the # key while transmitting. This will stop the interconnect to the public address system. If the operator does not turn off the connection to the public address amplifier a timer will turn it off after a pre-set time.

<u>Example B</u> – A person using a portable two way radio with a DTMF pad on it can transmit and press *. When the operator releases the ptt dial tone will be heard. The operator presses the ptt and enters the number of the phone on the system to initiate a conversation. The operator can hang up the phone connection by transmitting and pressing the # key. If both parties quit talking for a pre-set time the phone connection will hang-up.

<u>Example C</u> – An operator on the Teletics system can contact someone via radio through a telephone at any remote unit. The operator goes off hook and dials the number for the remote phone that is attached to the radio interconnect equipment. The phone will ring and answer. The operator enters a code (example 12#) and now anything the operator says will be heard over the radios used in the system. When the operator quits talking it will be possible to hear any replies from the field radios. This connection can be stopped by simply hanging up. If both parties quit the system will “time out” and hang up.

EQUIPMENT RECOMMENDATIONS

Virtually any two way radio can be used. It is recommended that a digital or analog FM radio system be used, due to its constant volume characteristics. The base station radio used must have an accessory connector to provide the audio and keying connections to the radio interconnect equipment.

1 ea. Base Station Radio Kenwood TK-790.

1 ea. 12 volt power supply Astron RS-15 (110 VAC to 12 VDC power supply capable of 15 AMPS)

1 ea. Base Station Antenna (unity gain, mounting pole, transmission line and misc. connectors)

1 ea. Accessory cable (used to connect the base station radio to the radio interconnect equipment)

1 ea Radio Interconnect System either Midian STI-1 or STI-2 or Zetron Model 30 Worldpatch.

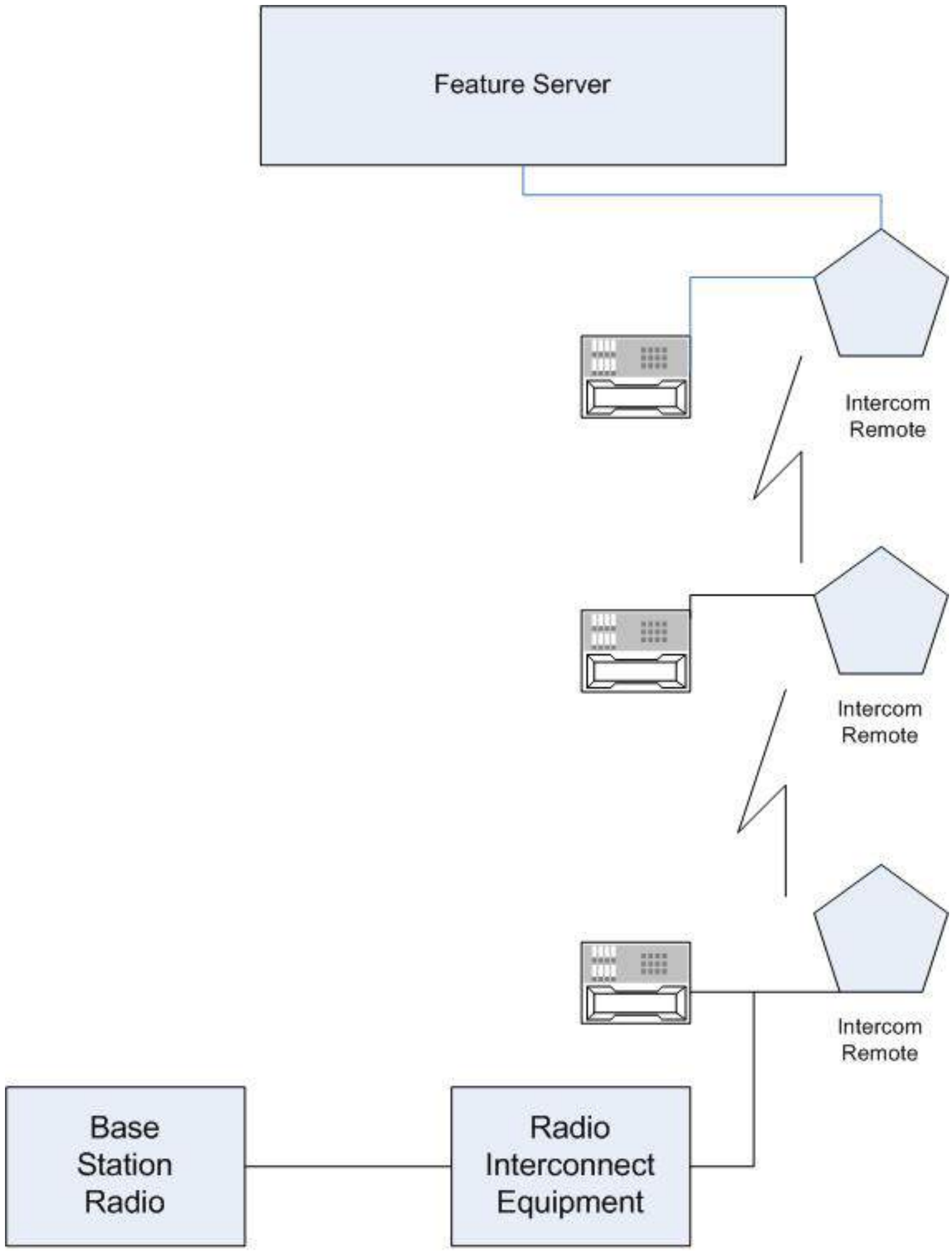

Feature Server
Intercom
Remote
Intercom
Remote
Intercom
Remote
Base
Station
Radio
Radio
Interconnect
Equipment

Diagram A

FCC LICENSING CONSIDERATIONS

Some larger companies may consider a unique license for their complete enterprise. A look at the internet will find a company that specializes in “General Pool Licenses”. However a generic “GMRS” license will work. These licenses are inexpensive and easy to get. Information can be found on the FCC internet site under the ULS service.

WIRING DIAGRAM

Diagram B will show a wiring diagram for the Kenwood TK-790 to the Midian STI-1 controller.

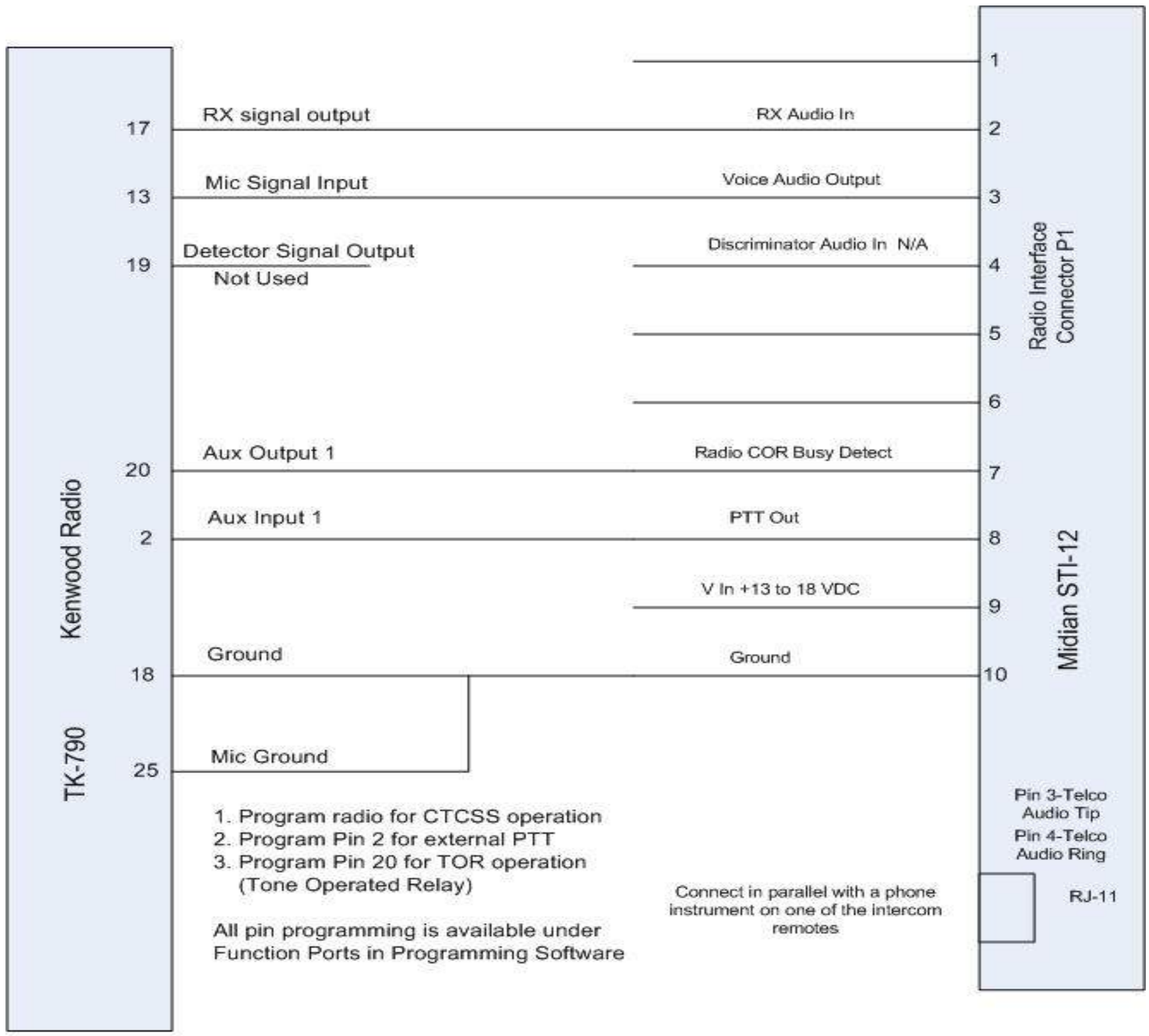


Diagram B

PROGRAMMING RECOMMENDATIONS

The STI radio telephone interconnect products are programmed using the KL-4F and KLF-4F-PC1 programming cables and the STI Programming Software (not the MPS). Please refer to the Midian STI manual shipped with the interconnect unit for software use and installation. The values listed here for program values are a good starting point. Values can be changed to customize the way the interconnect performs.

ANI Length: This specifies the number of digits used in the ANI. A value of “0” is recommended, the unit will work in the single user mode. If a value of four is used then the unit will work in a multi-user mode. In a single user mode of operation all speed dial entries are available for use. In a multi-user mode the speed dial entries can be assigned to different users if desired.

Up/Down Digit Relative to ANI Position: This field is only shown if the ANI length is set to 4 digits. This item allows the interconnect to know where to look for the Interconnect Up and Down Digits in relation to the incoming ANI digits. For example, if the selection is made for Leading and the Interconnect Up Digit = * and the Interconnect Down Digit = # then the sequence of entered digits would be:

*ANI = Bring the Interconnect Up with the following ANI

#ANI = Bring the Interconnect Down with the following ANI

And vice versa if the selection for Trailing is made:

ANI* = Bring the Interconnect Up with the following ANI

ANI# = Bring the Interconnect Down with the following ANI

Note: You may or may not need the ANI digits when bringing Down the Interconnect depending on the setting of Allow Disconnect with No ANI item located on the Subscribers Configuration Form.

Enable Direct Access: If checked the telephone caller (Teletics remote station phone) can get directly onto the radio system without a field radio answering or if a access number is entered into the field the caller must enter an access code prior to being given access to the radio system. If using the access number the caller will receive a tone prompt to indicate to the phone caller to enter the access code.

Tone Digit For Connect: This field sets up the DTMF tone that will cause the phone patch to take the phone line off hook. The selection range is 0-9, A-D, * and #. The * tone is recommended for use in connecting.

Tone Digit For Disconnect: This field sets up the DTMF tone that will cause the phone patch to take the phone line off hook. The selection range is 0-9, A-D, * and #. The # tone is recommended for use in disconnecting.

Connect/Disconnect Tone Minimum Time: This sets the time in 10 milliseconds increments that the tone must be present before the unit will accept the connect or the disconnect command (50 milliseconds to 2.55 seconds). Recommend 100 milliseconds.

Toll Override Access Management Code: This access code allows a Subscriber or System Operator to override any Toll Restrict Settings that may be in force. Note: Digits used for Interconnect Up and Down should not be used as part of the Toll Override Access Code.

Mobile to Telco DTMF Regenerator Delay: This is a delay from the time the unit brings the Telco off hook and before dialing begins. Since the unit does not detect if dial tone is present a value of 1 to 2 seconds is usually adequate for dial tone to become present before dialing commences.

Allow Over Air Speed Dial Reprogramming: This is a global enable/disable for allowing Subscribers to Program/Not Program Speed Dial locations respectively. This allows the System Operator to disable Over The Air Programming without having to visit each Subscriber Record.

Base/Repeater Type: Select between Simplex, Simplex with Sampling, Half-Duplex and Full-Duplex. Simplex, Simplex with Sampling are used when the RX and TX frequencies are the same. Simplex with Sampling gives the radio user a means to take control of the call if the phone caller is talking too long or is using obscene language. Half-Duplex is used when the RX and TX frequencies are different and the base station does not transmit and receive simultaneously. Full-Duplex is used when the RX and TX frequencies are different and the base station transmits and receives simultaneously.

Privacy Tone Select: If the unit is setup to operate in a Half Duplex mode, this parameter will instruct the unit to generate noise, block audio or allow audio to pass back over the transmitter when the portable in the field is talking.

Channel Busy Release Delay: This programs the amount of time that the interconnect will consider the radio channel to be busy after a radio busy condition has been detected and released. This applies to all types of busy detection inputs. The phone patch will not key-up the radio during the delay period when operating in Simplex modes therefore we recommend you keep it less than 1 second. This field will only affect Half Duplex and Full Duplex during this initial call setup until "ownership" of the channel is acquired.

Channel Busy Detection Source: This allows you to select which of the following sources will be used for radio busy detection; Carrier-Operated Relay (COR) via the COR input lead or the quieting filter via the discriminator input.

COR Logic Polarity: Select whether a Logic Low or Logic High is the active state of the radio's busy channel.

Simplex Telco VOX Sampling Interval: This sets the time in seconds the telephone caller may talk continuously. When this timer expires, a go-ahead beep is transmitted to the field radio and the radio user is given a small window of time to seize control of the conversation.

Simplex Telco VOX Sampling Width: This sets the amount of time in 100 msec increments the telephone caller is locked out, giving the radio an opportunity to seize control.

Simplex Go Ahead Courtesy Tone: When the unit is being operated in a Simplex mode, this checkbox will enable courtesy Go-Ahead Tones to indicate when the channel is clear so the other party can take a turn.

Morse Code ID String: This item is provided to allow the unit to issue a Station Identifier in Morse Code at a programmable time. This item will be ignored if no Characters are entered. Otherwise, the characters entered will be converted to the appropriate Morse Code equivalents before being transmitted.

Morse Code ID Output Delay Time: This item instructs the unit how often (0-60 minutes) to issue the CW-ID String. If the CW-ID String has no characters defined and the Delay Timer is greater than 0 Minutes then nothing will be transmitted. If the Delay Timer is set to 0 Minutes and the CW-ID String has characters defined nothing will be transmitted.

Morse Code ID Output Delay Time: This item instructs the unit how often (0-60 minutes) to issue the CW-ID String. If the CW-ID String has no characters defined and the Delay Timer is greater than 0 Minutes then nothing will be transmitted. If the Delay Timer is set to 0 Minutes and the CW-ID String has characters defined nothing will be transmitted.

Telco Hang-up Disconnect Time: This sets the amount of time that the unit will wait before seizing the phone line after disconnecting any previous call. Recommend 5 seconds.

Telco Abandoned Call Timeout: Once a connection is completed between the Telco caller and a radio user, this field is used to determine if the Telco caller is active by monitoring the VOX circuit. If no activity is detected on the Telco then a disconnect sequence will be sent to the radio user indicating the loss of Telco activity and the connection between the Telco and radio user will be disconnected.

Telco VOX Release Delay Time: The VOX circuit in the unit is a fast attack detector that triggers the Micro’s input when someone speaks. To prevent it from dropping out between words, the micro has a programmable release time. We recommend using at least 800ms. This time can be extended if the telephone caller’s voice drops out between words.

Number of Rings For Interconnect to Answer Multi-User: This controls how many times the telephone must ring before the unit will go off-hook and answer an incoming Telco call. If set to 2, the unit will go off-hook during the second ring.

Pulse Dialing Make/Break Ratio: If the unit is connected to a telephone line which accepts DTMF dialing, this field should be set to Tone Dialing. If DTMF is not supported, the unit can be made to do pulse dialing by selecting the appropriate make/break ratio. Currently supported make/break ratios are 40/60, 33/67, and 20/30milliseconds.

Pulse Dialing Inter-digit Gap Time: When using pulse dialing, this controls the amount of time to wait between dialing each digit.

Mobile Regenerated Dialed Digit On Time: This sets the amount of time the DTMF encoder generates a tone to the Telco. We recommend about 100 msec.

Mobile Regenerated Dialed Digit Off Time: This sets the amount of time the DTMF encoder stops generating atone between digits. We recommend about 100 msec.

Repeater/Base Station Key-Up Delay: This sets how long the unit will wait to send any tone dialing over the radio channel. This should take into consideration how long it takes the associated transmitter to key up, come up to full power, and how long it takes the receiving unit to wake up its carrier detect.

Courtesy Tone For Portable/Mobile Ringing: This feature is used to send courtesy ringing tones to a radio unit if needed. If a "0" is programmed into this field no courtesy tones are sent to the radio which allows the call to the radio unit to rely on its' decoder for ringing. Please Note theTimeout Delay For Portable/Mobile to Answer register will always override and disconnect when the timer has expired regardless of the setting of this register.

Timeout Delay For Portable/Mobile To Answer: This is used to provide an automatic disconnect if the radio unit that is being called does not answer within the specified period of time. It is not recommended that this ever beset to 0 except for purposes of testing. The recommended setting of this field should be between 5 to 10seconds.

Telco Received First Digit Detection Timeout: This allows the unit to look for the first incoming DTMF digit from the Telco. If the Telco has not entered a DTMF Digit within this time period the Error Tone is sent to theTelco followed by a Disconnect Tone and then the unit hangs up the Telco connection. The unit then goes back into an idle state and waits for further activity.

Telco Received Dialing Inter-digit Gap Delay: This item allows the unit to look for successive incoming DTMF digits from the Telco after the first digit has been entered. If the Telco has not entered a DTMF Digit within this inter-digit time period, the unit will then ascertain if it can continue to process the already collected information from the Telco or if a data entry error was made. If the unit can continue with the information already collected I will do so. However, if the unit detects an error with the information collected it will then generate the Error Tone message and/or a Disconnect Tone message after which the Telco will be disconnected. The unit will go back into an idle state and wait for further activity.

Mobile to Base Abandoned Call Timeout: This sets the amount of time before the phone patch disconnects its call to the telephone line if the mobile forgets to disconnect. The timer starts after the last transmission from the Radio.

PTT Click Count For Call: This allows a radio subscriber to click the radio PTT x number of times to answer an incoming telephone call, disconnect from a telephone call, or to force a speed dial call on the telco if specified. Otherwise, an error tone will be issued to the radio Subscriber indicating that the feature is disabled. Note: The PTT Clicks for Speed Dial feature will only work if the subscriber has one Speed Dial Entry assigned as Off set Entry 000.

PTT Click Count Delay Time: This specifies how quick a subscriber must click the PTT on the radio x number of times to get the specified number to dial out on the telco. If the specified number is not defined then the subscriber will receive an error tone indicating that this feature is not enabled.

DTMF Dialed Digit To Mobile On Time: This sets the amount of time the DTMF encoder generates a tone to the mobile. We recommend this to be between 60 msec to 100 msec.

DTMF Dialed Digit To Mobile Off Time: This sets the amount of time the DTMF encoder stops generating a tone between digits. We recommend this to be between 40 msec to 100 msec.

Telco Busy Signal Wait Time For Auto Disconnect: The busy signal that results from calling a busy telephone number may prevent the radio from transmitting a disconnect ANI to the unit. To prevent the situation, the radio phone patch can detect the presence of a busy signal. This field controls how long after dialing that the unit will wait for a busy signal. If a busy signal is detected during this time, the unit will automatically disconnect the call.

Base/Mobile Received First Digit Detection Timeout: This item allows the unit to look for the first incoming DTMF digit from the field radio. If the field radio has not entered a DTMF Digit within this time period the Error Tone is sent to the field radio followed by a Disconnect Tone. The unit then goes back into an idle state and waits for further activity.

Base/Mobile Received Dialing Inter-digit Gap Delay: This item allows the unit to look for successive incoming DTMF digits from the radio after the first digit has been entered. If the radio has not entered a DTMF Digit within this inter-digit time period, the unit will then ascertain if it can continue to process the already collected information from the radio or if a data entry error was made. If the unit can continue with the information already collected it will do so. However, if the unit detects an error with the information collected it will then generate the Error Tone message and/or a Disconnect Tone message after which the radio will be disconnected. The unit will go back into an idle state and wait further activity.

Subscriber Quick Pick: This feature is not available in Single User Mode. In Multi-User mode, if ANI length on the System Configuration Form is greater than 0 digits, this list will provide a sorted pick list of ANI's with the subscriber name if a name was defined. By clicking on the arrow on the right hand side of the field, a list will be provided to select from the defined subscriber's in the database.

ANI ID: If ANI Length on the System Configuration Form is greater than 0 digits, this field allows the entry of unique ANI digits. If the ANI Length is specified to be 4 digits and the operator only enters in a 3 digit ANI, then the ANI will be left padded with 0's to force the 4 digit ANI size.

User Name: If ANI Length on the System Configuration Form is greater than 0digits, this field allows the entry of a User Name to be identified with the ANI that should have been entered. The user name can be up to 20characters in length.

Record Status: This is an information only field that indicates whether the currently selected subscriber record is Active (enabled) or Disabled.

Enable Subscriber Account: Check this box to enable the subscriber.

Active User Call Limit Reset: If this option is enabled, it will allow the subscriber to reset the call limit timeout timer for another full session of time. This command only works while a subscriber is connected to either an incoming or outgoing telephone call.

Speed Dial (OTAP) Enable: If this option is enabled, it will allow the subscriber to add and modify speed dial entries that are assigned to that subscriber over the air.

Transpond Acknowledgements: If this option is enabled, the field unit must transpond within 2 seconds after being called otherwise the Telco caller will be disconnected.

Call Limit: This item sets the time allowed per connection. 1 minute before the time expires, a warning tone will be sent to the radio letting the subscriber know of the impending disconnection. If the Call Limit Reset option is enabled for that subscriber and the subscriber enters the command to reset the timer a Go Ahead tone will be sent to the subscriber. Otherwise, an Error Tone will be sent to the subscriber.

Interconnect Disconnect Validation: DTMF ANI: If this item is not checked, then the unit will accept any valid incoming Down Digit command to bring the interconnect down. Otherwise, if the item is checked, then the unit will force the subscriber to issue the ANI and also the down digit before allowing the interconnect to be brought down.

Toll Restrict Profile: The programming screen contains the text "Profile 1." The profile names can be changed to be more descriptive by going to the Toll Restricts Profile Form. During add or edit mode of a subscriber record you can select a Toll Restrict Profile to be applied to the current subscriber. After selecting, you will see the parameters for the profile updated in the fields below and to the right of the selection field. The fields that show the profile are information fields only. To make changes to the profile you must change the profile under menu Config > Toll Restrict Profile.

Max Digits Allowed To Dial: This places a limit on the maximum number of digits that the radio subscriber can dial into the phone system.

Max Speed Dial Entries: This sets a limit on the maximum number of speed dial entries the currently selected subscriber can access.
Note: There are only 100 total available speed dial locations. So if you have 100 subscribers and wanted them to all have speed dial capability, the most you could assign to each subscriber would be 1. Please note that the if 99 entries is assigned to each subscriber and there are 100 subscribers in the system, the assignment of speed dial locations would be on a first come first serve basis.

Current Speed Dial Entries: This is an information only field that shows how many speed dial entries are currently in use by the selected subscriber.

Profile Name: This allows for a name to be assigned to the Toll Restrict Profile for easier identification. Some examples are "international", "Long Distance", "Local", and "Restrict 900."

Restrict Leading 0: When checked, any phone number beginning with a 0 will be considered invalid. You will not need to add any other entries to the restrict matrix unless specific prefix restrictions are desired.

Restrict Leading 1: When checked, any phone number beginning with a 1 will be considered invalid. You will not need to add any other entries to the restrict matrix unless specific prefix restrictions are desired.

Restricts: The columns labeled Digit 1, Digit 2, Digit 3 and Digit 4 are representative of the leading digits in a phone number. The rows labeled Entry 1, Entry 2, Entry 3 and Entry 4 are representative of the total possible verification checks against each phone number passed through the restrict checking filter. The valid entries for each position are any DTMF Digit and also the letter x. It is important to note that even though any DTMF digit may be used, it is highly recommended that the defined Up and Down digits used for the Interconnect not be used in this matrix. The letter x is used to indicate a don't care entry and is ignored during the verification checks. Note: It is recommended that the restrict digits are defined one digit more than are defined in the Override Matrix.

Overrides: In the programming software you will the Toll Restrict Profile Overrides matrix. The columns labeled Digit1, Digit 2, Digit 3 and Digit 4 are representative of the leading digits in a phone number. The rows labeled Entry1, Entry 2, Entry 3 and Entry 4 are representative of the total possible verification checks against each phone number passed through the overrides checking filter. The valid entries for each position are any DTMF Digit and also the letter x. It is important to note that even though any DTMF digit may be used, it is highly recommended that the defined Up and Down digits used for the Interconnect not be used in this matrix. The letter x is used to indicate a don't care entry and is ignored during the verification checks. Note: It is recommended that the override

Digits are defined one digit less than are defined in the Restrict Matrix.

LEVEL SETTING AND CALIBRATION

Proceed with the interconnect adjustments using the following steps. Refer to the interconnect schematic and printed circuit board layout for location of test points and adjustment potentiometers found in the user's manual shipped with the unit. As a note of caution, be sure that the antenna output of the radio transmitter is properly terminated because the interconnect will be automatically keyed at various points in the adjustment sequence below:

Be sure that the interconnect is properly connected to the radio. The telephone line must be connected to a dBm meter, which in turn must be connected to the interconnect via the RJ-11 female connector.

1. Plug the PC programmer into the programming jack on the interconnect. Go to the top of the screen on the menu bar. Click on Diagnostics and select Calibration.

2. Start by initiating a telephone call to the interconnect using an extra phone line or cell phone. When ringing is heard, click on the RECEIVE AUDIO option using your mouse. This will complete the phone line connection and turn on the audio path from the RX Input all the way through to IC15, the hybrid transformers T1 and T2, and the phone line via the RJ-11 connector. Using an FM signal generator or an FM service monitor with a 1KHz test tone set to maximum system deviation (ie.5 KHz for a wide band system or 2.5 for a narrow band system) and with enough signal to quiet the radio, adjust RECEIVE LEVEL FROM RADIO pot VR2 for 2.4V peak-to-peak sinewave or 800 MV RMS or Zero (0) dB or just below clipping on Pin 7 of IC9B while monitoring with an oscilloscope. Under a loaded condition (i.e., interconnect connected to Telephone line), set the dBm meter with a balanced input to a bridging position. Set the TX LEVEL TO TEL LINE pot VR7 to a level between -14 and +4 dBm so that a test tone is heard in the telephone earpiece at a desired volume.

Next, check the trans-hybrid isolation as part of this tune up procedure. Connect the scope to pin 1 of IC9A and observe that the 1kHz test tone is present at this point, although at low amplitude. This amplitude at this point must be reduced to the minimum possible using the balancing network. The balancing network consists of the balancing pot VR8 and a 4-position DIP-switch (SW2), which switches the balancing caps C70-C73 on and off. Use the DIP switch to select a combination that reduces crossover in the hybrid as seen on pin 1 ofIC9A. Each time you change DIP switch settings, fine-tune balancing pot VR8 to reduce the signal on pin 1 of IC9A. Repeat this process until the best combination is found. If this is not set properly, voice transmitted to the phone line from the mobile radio will be fed back into the interconnect transmit path via T1/T2 trans-hybrid. In simplex mode the audio fed back from the phone line may cause the telephone line VOX to respond and cause the interconnect to key up when the mobile interconnect speaks loudly and the anti-VOX feature is not turned on.

3. Next, select the TRANSMIT AUDIO using your mouse. After selecting this with your mouse, the
microprocessor will turn on the audio path from the telephone line hybrid all the way through to the radio via TX Input (P1-8 Voice Audio Out) IC11C. Use the voice network analyzer or other instrument to transmit a1kHz test tone at 0 dBm across the RJ11C and adjust the RX LEVEL FROM PHONE LINE pot VR6 so that 2.4 V peak-to-peak sine wave or 800 MV RMS or Zero (0) dB is seen at IC-9A Pin 1. Next set TX LEVEL TO RADIO pot VR5 so that the transmitter is modulating at the desired system deviation while monitoring with the service monitor. The TX path has now been calibrated. If you are unable to get enough modulation, it may be necessary to reduce the value of R23 from 27K. to perhaps 1K. for low impedance Mic circuits.

4. With the mouse, select the RADIO DTMF encode path. The microprocessor will now release the phone line, key the radio's PTT line, and generate a DTMF tone through Q3 into FET switches IC14A and IC13A and out the P1-3 Voice Audio Out line. Adjust DTMF LEVEL TO RADIO pot VR3 for 2/3 of system deviation using an FM service monitor. You may find it necessary to make further adjustments in order to fine-tune the operation of the interconnect. This may include increasing the RX LEVEL FROM PHONE LINE if the telephone caller's voice is dropping out. This can be determined by actually using the interconnect to make a call.

5. After calibrating the unit, select UNIT RESET under the diagnostic menu.

FEATURES

- Simplex and half-duplex interconnect for base stations, control stations, and repeaters
- Compatible with end-to-end telco lines and analog PBX extensions
- Remote-programmable from DTMF radio or Touch-Tone telephone
- Convenience features include Morse ID, toll restrictions, timers, DTMF regeneration, programmable ANI codes, courtesy tones
- Simplex audio control via VOX, sampling, or combined sampling with VOX
- Half-duplex operation with privacy mode
- Digital voice delay available for simplex
- Fits in small repeater enclosures such as Motorola Radius GR300, GR500, GR1225
- 19-inch rack-mount bracket available
- Connection diagrams for many radios
- Expert and responsive technical support
- Special, custom features available

Selective Calling Option:

- 100- and 1000-call two-tone paging
- DTMF paging
- Dial-up remote operation (* to key, # to unkey)

APO (Advanced Programming Operation):

- 50 autodials
- 4-click PTT autodial for non-DTMF mobiles
- Auxiliary FET output
- Direct-access security code
- Hookflash from mobile via *0
- Autodial access only
- Repeat courtesy tone
- Call-alert beeps
- Unkey to hear 2nd dial tone
- Ringout from connect button

INTRODUCTION

The Model 30 Worldpatch is a compact, telephone interconnect that allows a group of twoway radio users to place and receive telephone calls. It efficiently interfaces a telephone line (or analog PBX line) to a simplex base station, control station, or conventional repeater. It can also be used as a repeater maker to convert a pair of mobile radios, or a full-duplex base station, into a carrier-operated repeater.

The Model 30 is available both as a single-group interconnect as well as a selective-calling interconnect. The selective calling version provides 100-call and 1000-call two-tone and DTMF paging. For ordering convenience, the Model 30 Worldpatch is available in three different versions:

1. Basic with APO (901-9416) — recommended for half-duplex repeater operation, includes Advanced Programming Option (APO)

2. Basic with APO & Digital Voice Delay (901-9417) — as (1) above with digital voice delay for simplex operation on base stations or control stations

3. Basic with APO, Digital Voice Delay, and Selective Calling (901-9540) — as (2) above, but recommended for applications requiring selective calling on repeater or base stations

Any version can be changed into another by ordering a field upgrade kit. Other options include a 19-inch rack mount bracket, Deadbolt surge arrestor, and various interface cables for the connection to the radio.

An assortment of connection diagrams explains the hookup to the most popular radios. A responsive technical support department is also available to provide expert assistance over the telephone.

RADIO-TO-PHONE CALLS

To place a telephone call, a radio user keys up and enters the valid access code from the keypad of a DTMF microphone or handheld radio. Upon receiving dial tone, the user then dials into the public telephone network. Programmable toll restrictions can prevent long-distance dialing. Radio users can even be limited to dialing certain pre-programmed autodial numbers only (APO feature). Authorized personnel can bypass restrictions by entering the toll-restrict bypass code.

The Model 30 can also provide limited telephone access to radio users who don't have DTMF keypads on their radios via the four-click autodial feature (APO feature). By clicking the PTT button four times, a radio user can cause the Model 30 to activate autodial #1.

All calls are terminated when the radio user enters the disconnect code (separately programmable from the access code). The phone party may also disconnect a call by entering #0 on the telephone keypad. Conversation timers, busy tones, and dial tone can also terminate a call.

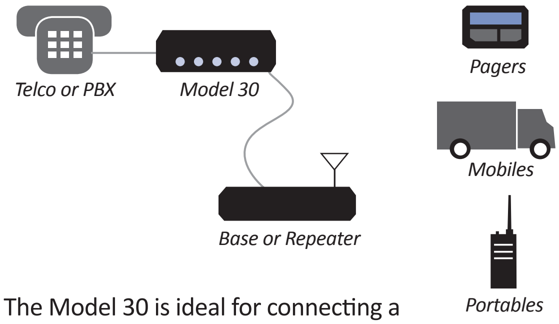
PHONE-TO-RADIO CALLS

If a telephone party (landline) wants to call a radio user, the phone party first dials the normal telephone number of the phone patch. This may be a standard seven-digit telephone number, or perhaps a 3-digit extension if the Model 30 is on a PBX system. After a number of rings before answer (programmable), the Model 30 picks up the line, keys up the transmitter, and broadcasts a ringing tone over the radio channel to alert the radio user(s) that a call is pending. The Model 30 can either ring once on the channel, or ring repetitively. When ringing repetitively, the Model 30 matches the ringing cadence of the local telephone system.

A radio user can answer the call by keying up and entering the valid access code. Or, the Model 30 can be programmed to allow radio users to answer a call by just keying up (COR to answer), which allows non- DTMF radios to answer calls.

For selective-calling versions, the sequence of events is somewhat different. After the Model 30 picks up the line, it prompts the phone party with a beep. The phone party then overdials the number of the desired radio (typically two or three digits) to be called. The Model 30 keys up the transmitter and broadcasts the corresponding paging tones, unsquelching the appropriate radio only. Other radios on the system are not disturbed.

The Model 30 also has a direct-channel access mode. This puts the phone party on the air immediately, without ringing tones or overdial, as soon as the phone party dials the telephone number of the phone patch. The phone party may then voice-hail directly over the channel to gain the attention of radio users. To prevent unauthorized access, the phone party can be required to first enter a direct-access security code (APO feature).



The Model 30 is ideal for connecting a standard telco line, or an anolog PBX extension to a radio base station, control station, or repeater.

BASE STATION INSTALLATIONS

When connected to a simplex base station or control station, a phone patch is responsible for changing the station from receive to transmit at the right moments. The Model 30 can use VOX (voiceoperated transmit), sampling, or various mixtures of the two, depending on the demands of local conditions.

In simplex installations, a Model 30 with the digital voice delay is recommended. This feature is a digital memory that delays the phone party's audio for a fraction of second (programmable 0.2-0.9 sec), allowing ample time for the transmitter to key up. The delay prevents clipping of the phone party's first syllable.

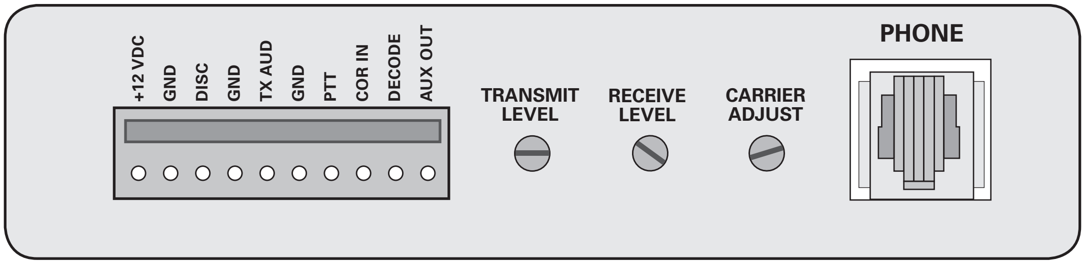
REPEATER INSTALLATIONS

The Model 30 Worldpatch can be added to an existing, full-duplex repeater to allow radio users to make and receive phone calls. Or, the Model 30 can be used to create a repeater out of a separate transmitter and receiver, such as a pair of low-cost mobiles, or a full-duplex base station.

To create a degree of privacy during interconnected calls, a half-duplex privacy mode can be enabled. When the radio user is talking to the phone party, the Model 30 will transmit a masking tone (instead of the radio user's audio) so eavesdropping mobiles will be discouraged from monitoring the conversation.

With selective-calling versions of the Model 30, radio users can be allowed to selectively call other two-way radios and pagers. After keying up and entering the sign-on code, the user enters a steering digit to indicate whether he wants to place a selective call or an interconnect call into the telephone system.

There are many features to assist with repeater conversations. A courtesy beep can be enabled, so every time a user finishes talking and releases the PTT button, the Model 30 transmits a quick beep to let the other user know to talk. The repeat hold timer can be set from 0 to 9 seconds to prevent other PL tones from interrupting a conversation. To alert conversing The Model 30 is ideal for connecting a standard telco line, or an analog PBX extension to a radio base station, control station, or repeater. mobiles that a call is coming in over the phone line, a call-alert beep can be transmitted at the same cadence as the telco ring cycle.



If it is desired to have encode/decode of a single PL tone on the repeater, it's best to enable that capability in the repeater itself. If it's desired to have encode/decode of several PL tones or DPL codes–that is, to divide the repeater into different work groups–it's best to use an interconnected tone panel like the Zetron Model 48jr Repeater Patch rather than the Model 30.

DIAL-UP REMOTE

The dial-up remote feature allows a phone caller to have positive control of a base station from afar. The phone party can dial the telephone number of the Model 30, enter a security code, and immediately begin listening to the radio channel without the radio users being notified. If desired, the phone party can press * to transmit, and release the transmitter by pressing #. The dial-up remote feature allows a phone party to physically control the transmitter without having to rely on a VOX circuit.

PBX APPLICATIONS

The Model 30 is often installed on an analog line of a PBX telephone system. This lets PBX users reach field radios simply by dialing the extension occupied by the Model 30. Several programmable items help support PBX applications. Radio users can generate hookflash signals into the PBX if necessary to control PBX features. A radio user generates a hookflash signal by pressing *0 during a call. Radio users can also unkey their radios and verify a second dial tone when requesting an outside line.

MANUAL OPERATORS

A parallel telephone may share the line with a Model 30. This gives a manual operator the opportunity to personally answer a call before the call is put onto the radio system. The Model 30 can be programmed to wait for up to 10 rings before answering the line.

After answering a parallel telephone, an on-site operator can use the connect button on the front panel of the Model 30 to manually connect the telephone party to the radio system. If the Model 30 has the APO feature set, then the connect button can even be used to transmit a ringing tone over the radio channel to alert the mobile users to take the call.

INSTALLATION & MOUNTING

The Model 30 is compact enough to fit in the industry's smallest repeater enclosures, including the Motorola Radius GR300, GR500, and GR1225. Radio-specific cables are available to make the interface as quick and easy as possible.

For proper installation in standard, 19-inch rackmount enclosures, an optional 19-inch rack mount bracket is available.

The phone line connection on the rear of the Model 30 is a simple RJ-11 phone jack. The radio interface is a 10-pin modular plug with crimp pins provided. Level settings for transmit, receive, and carrier adjust are accessible on the back panel.

All programming settings may be changed by remote control, without visiting the site. Programming takes place from a Touch-Tone telephone or keypad-equipped radio. Access to the program memory is protected by a programming access code. The installer may restore the unit to the factory default settings at any time.

STANDARD PROGRAMMABLE ITEMS

Mobile-to-Phone Calls:
ANI for access (1-9 digits)
ANI to disconnect (1-9 digits)
Validation for single digit ANI (y/n)
Mobile dialing timer (0-60 sec)
1st digit toll restricts (up to 4)
2nd digit toll restricts (up to 4)
Toll-restrict bypass code (1-8 digits)
Regenerated dialing (DTMF or pulse)
Busy tone disconnect (y/n)
Dial tone disconnect (0-9 sec)

Phone-to-Mobile Calls:
Rings before answer phone (1-10)
Ringing to radio (once or repetitive)
Mobile answer method (ANI or PTT)
Direct phone-to-radio access (y/n)
Dial-up remote (* = key, # = unkey)

Conversation Control:
Call limit timer (0-10 min)
Call timer reset via mobile * (y/n)
COR hold time (0.0-0.5 sec)
Mobile activity timer (0-60 sec)
Morse ID (0-8 digits)
Morse interval (10 min time or activity)
Phone courtesy tone (y/n)
Programming access code (5 digits)

Repeater Settings:
Half-duplex repeater operation (y/n)
Half-duplex privacy (y/n)
Repeat audio (y/n)
Repeat hold time (0-5 sec)

Simplex Station Settings:
VOX voice-operated transmit (y/n)
VOX with pre-key (y/n)
Sampling (y/n)
Sampling & VOX-extended interval (y/n)
VOX & sample between words (y/n)
VOX before dial tone (y/n)
VOX hold time (0.5-1.5 sec)
Sample rate (0.5-1.5 sec)
Sample window size (.01-sec steps)

A.P.O. PROGRAMMABLE FEATURES

Allow unkey to hear 2nd dialtone (y/n)
Autodial access only (y/n)
Autodials 0-49 (1-16 digits each)
Call-alert beeps (y/n)
Direct-access security code (1-9 digits)
FET output turn-on code (1-9 digits)
FET output turn-off code (1-9 digits)
Hookflash from mobile via *0 (y/n)
Repeat courtesy tone (y/n)
Ringout from connect button (y/n)
4-Click PTT sends autodial #1 (y/n)

SELECTIVE CALLING PROGRAMMABLE FEATURES

Paging Format Choices:
0. None
1. DTMF
2. 100-call 2-tone
3. 1000-call 2-tone
4. DTMF & 100-call two-tone
5. DTMF & 1000-call two-tone

Two-tone Paging Parameters:
Tone groups (Mot 1-6,10,11,A,B,Z; GE A-C)
Code plan (Mot B-Y,MT; GE X-Z)
Timing (Mot; GE std; NEC A-D,L,M)
Group or diagonal (replaces A or B)

DTMF Paging Parameters:
Digits entered by caller (1-8)
Strapped digits (1-5)
Strapped location (precede or follow)

Other:
Mobile-to-mobile paging (y/n)
Note: the digital voice delay is adjustable 0.1-2.0 seconds on the delay board.

SPECIFICATIONS

GENERAL	
Power:	11-16 VDC, 150 mA
Temperature:	0-65 degrees Celsius
Size:	5.9 x 7.4 x 1.7 inches
Weight:	2.5 lb
Data Retention:	Nonvolatile EEPROM, no batteries
Secondary Protection:	Telco high voltage clamps with protective fusing elements
TELEPHONE INTERFACE	
Connector:	RJ11-C modular jack for one end-to-end (B1) phone line
Incoming Call:	Ring detection on tip-ring pair. Programmable number of rings to answer
Call Answer:	Off-hook, tip-ring current draw
Call Disconnect:	Busy tone, dial tone, call limit, mobile activity timers
Approvals:	FCC Registration Part 68 Industry Canada
RADIO INTERFACE	
PTT:	FET pull to ground
COR:	External or internal carrier detector with squelch control
Tx Audio:	-40 to +6 dBm. Hi/lo selector 1Kohm output
Rx Audio:	-40 to +10 dBm. Hi/lo selector. 50Kohm input. 25 mV to 6 V P-P
Tone Validation:	Contact closure input from CTCSS decoder
Auxiliary Output:	FET pull to ground

ZETRON

ZETRON AMERICAS
PO Box 97004, Redmond, WA USA 98073-9704
(P) 1 425 820 6363 **(F)** 1 425 820 7031 **(E)** zetron@zetron.com
ZETRON EMEA
27-29 Campbell Court, Bramley, Hampshire RG26 5EG, United Kingdom
(P) +44 1256 880663 **(F)** +44 1256 880491 **(E)** uk@zetron.com
ZETRON AUSTRALASIA
PO Box 3045, Stafford Mail Centre, Stafford QLD 4053, Australia
(P) +61 7 3856 4888 **(F)** +61 7 3356 6877 **(E)** au@zetron.com

www.zetron.com

005-0339J May 2011

STI-1

Standard Radio-Telephone Interconnect (Phone Patch) – 19” Rack Mount

STI-2

Standard Radio-Telephone Interconnect (Phone Patch) – Desktop Box

Manual Revision: 2013-03-29

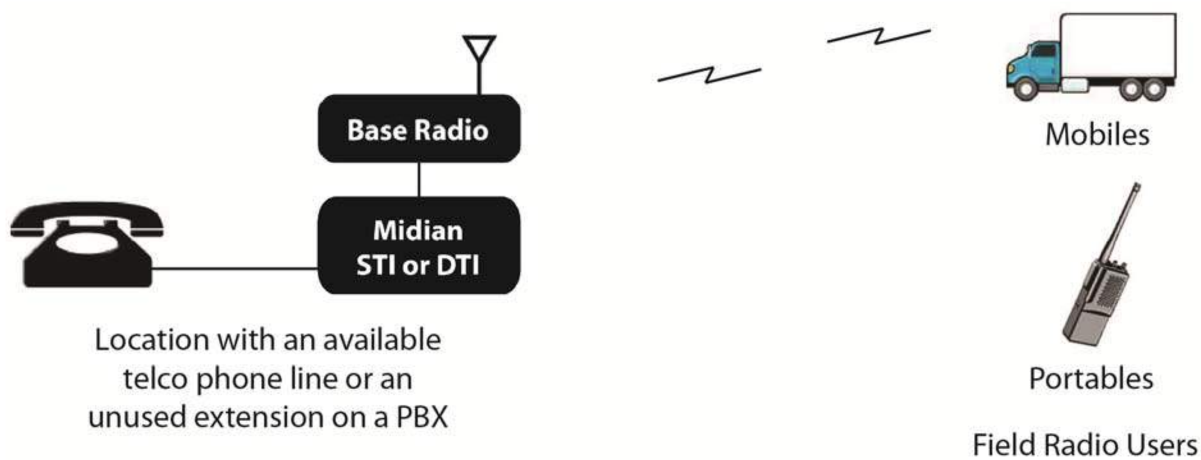
Covers Firmware Revisions:
STI: 2.1 and higher

Covers Hardware Revisions:
STI-1: E and higher
STI-2: G and higher

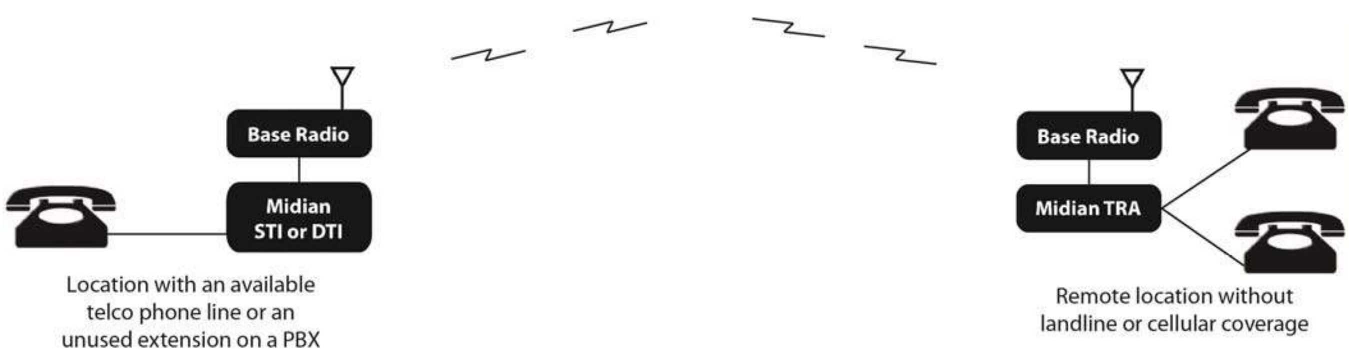
Radio-Telephone Interconnect (Phone Patch): Enables two-way radio users to make and receive telephone calls on the two-way radio.

Landline to Two-Way Radio Calls: Two-way radios may receive phone calls from a landline. The caller may enter a radio's unit ID to speak with a specific user in a multi-user system. In a single user system any field radio can answer the call using multiple PTT clicks or a DTMF connect digit.

Two-Way Radio to Landline Calls: Two-way radios without a DTMF keypad can have the interconnect dial a preprogrammed per-user telephone number by pressing the PTT switch multiple times. Two-way radios with a DTMF keypad can connect to the phone patch and then manual dial the desired telephone number.



Phone-Line Extender: When used in conjunction with Midian's TRA, the STI radio phone patch and TRA can extend phone line coverage using two-way radios into an area without landline or cellular coverage.



Operator (Manual) Control: Using the STI Option A an operator can be used to control the conversations between the phone line and field radios. This is ideal for HF SSB radio systems. For more information reference the STI Option A Manual.

SPECIFICATIONS

VOLTAGE/CURRENT

Operating Voltage	13-18 VDC
Operating Current	25 mA
LED Current	4 mA
Fuse Protection Auto Resettable	200mA
Power Control	Single pole toggle switch

INPUT FROM RADIO

RX Input Level	150 mV-2.5V p/p pot adjustable (52 mV RMS – 880 mV RMS)
RX Input Impedance	100KΩ
Discriminator Input level	pot adjustable
Discriminator Input Impedance	100KΩ
Programming Input	3-pin stereo jack
Frequency Range	300-3000
DTMF Decode Sinad Ratio	9 dB
Dynamic Range	30 dB
COR Input	0-5 V

OUTPUTS TO RADIO

Audio Output Tone Level	2/3 system deviation
Audio Output Impedance	<20KΩ
PTT Out	FET switch to ground

TELEPHONE LINE INPUT*

Input Level	-14 - +4 dB, pot adjustable
Input Impedance	600Ω balanced
Ring Detector	opto-isolator
Seizure Output	FET opto-isolator
FCC Registration #	MIDOT06BDTISTIDPT

MECHANICAL

19” Rack Dimension	19” W x 1.75” H x 1.4” D
Desktop Interconnect Dimension	6” w x 1.45” H x 7.6” D
Operating Temperature	-30° to +60°C

* This device complies with Part 68 and Part 15 of the FCC rules. Operation is subject to the following conditions:

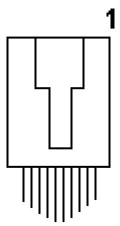
1. This device may not cause harmful interference.
2. This device must accept any interference received, including interference that may cause undesired operation.

INSTALLATION

Following are instructions for each connection that is required by the interconnect for proper operation. A qualified two-way radio technician should perform the installation and adjustment of the interconnect. Once the installation has been successfully completed, continue with the adjustment section of the manual. The adjustment procedures require some programming steps to be done concurrently. Therefore, frequent reference to the Programming section may be necessary. **Note:** Be sure to follow standard anti-static procedures when handling any of Midian's products.

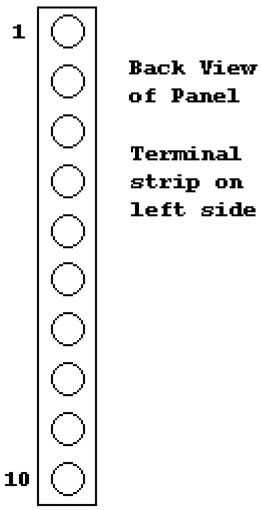
Wiring Information

STI-2 RJ-45 WIRING INFORMATION		
Wire Color	**Pin #**	**Description**
Gray	1	RX Audio In
Orange	2	Radio COR Busy Detect
Violet	3	Radio PTT
Yellow	4	Tone/Data Output
Green	5	Discriminator Audio Input
Red	6	VIN 13-18 Volts DC
Black	7	Ground
White	8	Voice Audio Output
Blue	9	Wiretap Switch
Brown	10	Wiretap Audio



RJ-45
Tab Facing Up

STI-1 INTERCONNECT STRIP WIRING INFORMATION	
Interconnect #	**Description**
1	Tone/Data Output
2	RX Audio In
3	Voice Audio Output
4	Discriminator Audio Input
5	Wiretap Switch
6	Wiretap Audio
7	Radio COR Busy Detect
8	Radio PTT
9	VIN 13 – 18 Volts DC
10	Ground



Radio Interface Connector P1

Voice Audio Output: (STI-1 = Pin 3 STI-2 = Pin 8): This connection supplies the transmitter with processed audio from the telephone line. This connection should be made to the transmitter's microphone audio input. The level is adjustable using adjustment pot VR5 (XMIT LEVEL TO RADIO).

RX Audio In: (STI-1 = Pin 2 STI-2 = Pin 1): This lead is connected to the radio's RX audio. This is used to pass voice and signaling into the interconnect. This input also feeds audio in to the high-pass noise squelch filter quieting detector circuit and the audio high-pass filter to remove the CTCSS/DCS squelch tones. The level is adjustable using pot VR2 (RECEIVE LEVEL FROM RADIO).

Radio COR Busy Detect: (STI-1 = Pin 7 STI-2 = Pin 2): The Carrier Operated Relay (COR) connects to the radio's squelch detect circuit. The COR connection should be made to a point in the radio receiver's squelch circuit that changes in DC voltage level when the squelch control is opened and closed or when a carrier comes through and breaks the squelch. Such a point usually exists at the output of the noise rectifier. The COR polarity can be either active high or active low. (See programming section.)

There is also a separate noise squelch busy detector that is derived from the discriminator input if no COR input is available. Any one of these may be used for detecting busy channels (See Programming Help Menu).

Discriminator Audio In: (STI-1 = Pin 4 STI-2 = Pin 5): This optional connection is supplied by unfiltered discriminator audio from the receiver to the interconnect's microprocessor for detecting high frequency noise via IC10B, IC14C, IC10C, IC10D, IC14B, IC12B and finally IC12A. It determines if the carrier is present for busy lockout determination. This connection must be made directly to the discriminator output of the associated receiver. The Disc Audio input must be connected at a point before de-emphasis. VR1 on IC10B sets the level.

PTT Out: (STI-1 = Pin 8 STI-2 = Pin 3): The PTT output is an open drain FET transistor with RF bypassing on its output lead. This simply pulls the push-to-talk input of the radio to ground to key up the radio when activated by the interconnect microprocessor.

Ground: (STI-1 = Pin 10 STI-2 = Pin 7): This connection is the main negative ground return between the radio and the interconnect.

Tone Data Output: (STI-1 = Pin 1 STI-2 = Pin 4): Not used in the STI.

VIN 13-18 Volts DC: (STI-1 = Pin 9 STI-2 = Pin 6): Power for the interconnect is derived from the radio. Due to the drop on the interconnect's 12-volt regulator input, the voltage should be at least 13 volts with a maximum of 18 VDC.

RJ-11 Connector for Telco Interface

Pin 3 - Telco Audio Tip

Pin 4 - Telco Audio Ring

This is the balanced 600-ohm telephone line interface. This interface has RF bypassing on the tip and ring to ground as well as spark suppressors DK4 and DK5 and a self-resetting poly-switch fuse (F-2). Line balancing is accomplished by selecting the right values of phasing capacitance using SW2 and adjusting VR8 for minimum trans-hybrid side tone return audio on the TX Audio during receive from the radio to Telco.

HARDWARE ALIGNMENT

Initial Radio Calibration

Using an FM service monitor, check the associated radio's frequency error to make sure that it is in spec with manufacturer's specifications.

Check the radio's modulation, bearing in mind that modulation is additive. Most radios are ***supposed*** to run ±5KCs of modulation. The CTCSS tone modulation is supposed to run about .75 KCs of deviation, thus leaving approximately 4 KCs of voice modulation for an additive total of 4.75 KCs.

Check the radio receiver's sensitivity to make sure it conforms to the particular manufacturer's specifications. Also check to make sure the receiver is on frequency.

Set the radio receiver's audio level. In the following adjustment section, we will be setting receive levels into the interconnect. If someone changes the radio's level significantly, it may overdrive the interconnect and cause distortion of the decode tones or voice audio.

If the RX input level to the interconnect is too high, the input receive level pot on the interconnect may have to be turned down so far that the adjustment becomes critical. Try to keep the level at the input at around 150 mV – 2.5 V Peak-to-Peak.

Calibration

Proceed with the interconnect adjustments using the following steps. Refer to the interconnect schematic and printed circuit board layout for location of test points and adjustment potentiometers. As a note of caution, be sure that the antenna output of the radio transmitter is properly terminated because the interconnect will be automatically keyed at various points in the adjustment sequence below:

Be sure that the interconnect is properly connected to the radio. The telephone line must be connected to a dBm meter, which in turn must be connected to the interconnect via the RJ-11 female connector.

1. Plug the PC programmer into the programming jack on the interconnect. Go to the top of the screen on the menu bar. Click on Diagnostics and select Calibration.

2. Start by initiating a telephone call to the interconnect using an extra phone line or cell phone. When ringing is heard, click on the RECEIVE AUDIO option using your mouse. This will complete the phone line connection and turn on the audio path from the RX Input all the way through to IC15, the hybrid transformers T1 and T2, and the phone line via the RJ-11 connector. Using an FM signal generator or an FM service monitor with a 1 KHz test tone set to maximum system deviation (ie. 5 KHz for a wide band system or 2.5 for a narrow band system) and with enough signal to quiet the radio, adjust RECEIVE LEVEL FROM RADIO pot VR2 for 2.4V peak-to-peak sine wave or 800 MV RMS or Zero (0) dB or just below clipping on Pin 7 of IC9B while monitoring with an oscilloscope. Under a loaded condition (i.e., interconnect connected to Telephone line), set the dBm meter with a balanced input to a bridging position. Set the TX LEVEL TO TEL LINE pot VR7 to a level between -14 and +4 dBm so that a test tone is heard in the telephone earpiece at a desired volume.

   Next, check the trans-hybrid isolation as part of this tune up procedure. Connect the scope to pin 1 of IC9A and observe that the 1kHz test tone is present at this point, although at low amplitude. This amplitude at this point must be reduced to the minimum possible using the balancing network. The balancing network consists of the balancing pot VR8 and a 4-position DIP-switch (SW2), which switches the balancing caps C70-C73 on and off. Use the DIP switch to select a combination that reduces crossover in the hybrid as seen on pin 1 of IC9A. Each time you change DIP switch settings, fine-tune balancing pot VR8 to reduce the signal on pin 1 of IC9A. Repeat this process until the best combination is found. If this is not set properly, voice transmitted to the phone line from the mobile radio will be fed back into the interconnect transmit path via T1/T2 trans-hybrid. In simplex mode the audio fed back from the phone line may cause the telephone line VOX to respond and cause the interconnect to key up when the mobile interconnect speaks loudly and the anti-VOX feature is not turned on.

With the mouse, select the TELEPHONE DTMF encode path. The microprocessor will now turn on IC8 pins 7 and 8 and the phone line will remain seized (off hook). It will also turn on IC13B and generate a DTMF tone that is fed through to the hybrid transformer to the phone line via the RJ-11 connector. Set DTMF LEVEL TO TEL LINE pot VR4 for –15.0 to 0 dBm to the telephone line using a dbm meter in bridging mode.

3. Next, select the TRANSMIT AUDIO using your mouse. After selecting this with your mouse, the microprocessor will turn on the audio path from the telephone line hybrid all the way through to the radio via TX Input (P1-8 Voice Audio Out) IC11C. Use the voice network analyzer or other instrument to transmit a 1kHz test tone at 0 dBm across the RJ11C and adjust the RX LEVEL FROM PHONE LINE pot VR6 so that 2.4 V peak-to-peak sine wave or 800 MV RMS or Zero (0) dB is seen at IC-9A Pin 1. Next set TX LEVEL TO RADIO pot VR5 so that the transmitter is modulating at the desired system deviation while monitoring with the service monitor. The TX path has now been calibrated. If you are unable to get enough modulation, it may be necessary to reduce the value of R23 from 27KΩ to perhaps 1KΩ for low impedance Mic circuits.

4. With the mouse, select the RADIO DTMF encode path. The microprocessor will now release the phone line, key the radio's PTT line, and generate a DTMF tone through Q3 into FET switches IC14A and IC13A and out the P1-3 Voice Audio Out line. Adjust DTMF LEVEL TO RADIO pot VR3 for 2/3 of system deviation using an FM service monitor.

   You may find it necessary to make further adjustments in order to fine-tune the operation of the interconnect. This may include increasing the RX LEVEL FROM PHONE LINE if the telephone caller's voice is dropping out. This can be determined by actually using the interconnect to make a call.

5. After calibrating the unit, select UNIT RESET under the diagnostic menu.

CONTROLS & INDICATORS

Power Switch: The power switch is on the front right side of the STI-1 and on the front left side of the STI-2. The down position is off and the up position is on.

Programming Jack: There is a programming input jack (3.5 mm stereo connector) on the top left front of the STI-1 (when swung open) or on the front of the STI-2. This allows a PC to program all of the features for these products.

There are 7 indicator LED's:

STI-1 INDICATOR LEDS

Power LED – Shows presence of voltage (13-18 VDC)	Green
Carrier - Shows busy activity on channel.	Green
Transmit - Shows VOX/PTT activity from Telco	Red
DTMF Decode - Shows presence of incoming DTMF from the radio or telephone line	Red
Tone – Not used	Green
Ring Detector – Indicates incoming ring from the telephone line	Red
Hook Output – Shown when the phone patch seizes the line	Yellow

STI-2 INDICATOR LEDS

Power LED - Shows presence of voltage (13-18 VDC)	Red
Carrier - Shows busy activity on channel	Red
Transmit - Shows VOX/PTT activity from Telco	Red
DTMF Decode - Shows presence of incoming DTMF from the radio or telephone line	Red
Tone – Not used	Red
Ring Detector – Indicates incoming ring from the telephone line	Red
Hook Output – Shown when the phone patch seizes the line	Red

PRODUCT PROGRAMMING

The STI radio-telephone interconnect products are programmed using the KL-4F and KL-4F-PC1 programming cables and the STI Programming Software (not the MPS).

Software Installation:

Insert the CD into the CD-ROM drive and a browser will auto-start. If the browser does not auto-start, click on the Start button, select Run and run the setup.exe program for the appropriate drive. Select Miscellaneou Software from the navigation bar and then click on the STI link to start the installation process. Be certain that the "Install KL-4 USB Driver" box is checked during the installation process.

NOTE: Please make sure any programs that are running have been closed prior to installing this software. Also, even though it is not required, it is good practice to reboot the system after installation of new software.

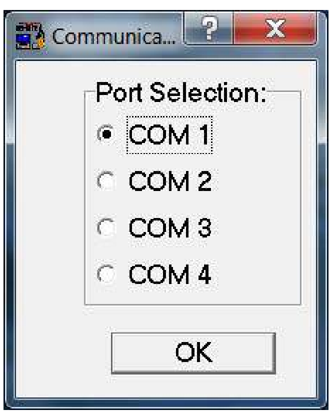
Online Help:

You can access online help at any time within the program when the "?" mark icon is located in the upper right hand corner of the current window. To utilize the online help left-click on the mouse while pointing at the "?" mark icon. This will change the pointer to a pointer with a question mark. Place the pointer with the question mark over the field that you need help with and left-click the mouse. This will display the online help for that field or by pressing F1 while the is highlighted. In future versions this will be changed to right click on labels when the mouse cursor changes from an arrow to an arrow with a question mark.

Load an Existing Configuration File:

The Configuration File contains subscriber information and system parameters for the interconnect. On the Main Screen of the program, located on the menu bar you will see the word File. With the mouse pointing to the word File, left click the mouse. This will open a submenu and you will see the word Load in the submenu. With the mouse pointing to the word Load, left click the mouse. The action that was just performed should open a new window with a title of Open. With the mouse pointing to the name of the file that you wish to load, left click the mouse twice. This will automatically select the file to open and load the configuration file into the program.

Setting the Communications Port:



On the Main Screen of the program, located on the menu bar you will see the word "Config." With the mouse pointing to the word "Config," left click the mouse. This will open a submenu and you will see the word "Communications" in the submenu. With the mouse pointing to the word "Communications" another submenu will popup to the right displaying selections COM1 – COM4. With the mouse pointing to the COM port that you wish to use, left click the mouse. If the COM port is available for use, then the words "Programming" and "Diagnostics", which are located on the menu bar on the Main Screen, will be enabled for access. However, if the COM port is not available for use then the words "Programming" and "Diagnostics" will remain disabled. If a higher number COM port is being used the ini file located where the application was installed must be edited. For Windows 7 and higher the virtual storage section must be located to make the changes.

Read from the interconnect:

On the Main Screen of the program, located on the menu bar you will see the word "Programming." With the mouse pointing to the word "Programming," left click the mouse. This will open a submenu and you will see the words "Read Unit" in the submenu. With the mouse pointing to the "Read Unit," left click the mouse. The action that was just performed should open a new window with a title of "Reading Unit's System Parameters." If the communication port has been setup correctly and the programmer has been plugged into the phone patch and the pc, you should see the word "RETRIEVING" being displayed in the middle of the window. Upon successful retrieval of the data from the interconnect, you will see the word "SUCCESS" displayed in the window. If you see any other message, check the COM port selection, make sure the programmer is properly attached and there is power to the interconnect.

Write all to the interconnect:

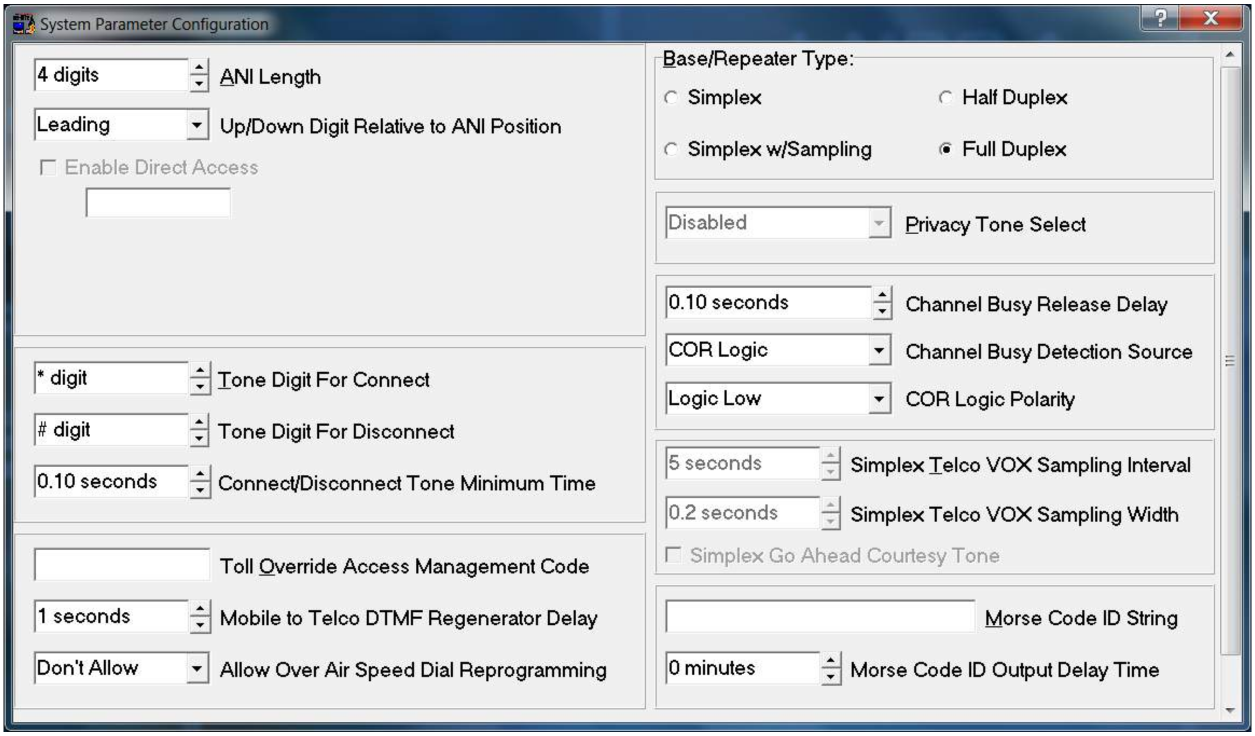
On the Main Screen of the program, located on the menu bar you will see the word "Programming." With the mouse pointing to the word "Programming," left click the mouse. This will open a submenu and you will see the word "Write" in the submenu. With the mouse pointing to the word "Write," left click the mouse. The action that was just performed should open a new window with a title of "Sending Unit Parameters." If the communication port has been setup correctly and the programmer has been plugged into the phone patch and the pc, you should see the word "WRITING" being displayed in the middle of the window. Upon successful writing of the data to the interconnect, you will see the word "SUCCESS" displayed in the window. If you see any other message, check the COM port selection, make sure the programmer is properly attached and there is power to the interconnect.

Verify the interconnect:

On the Main Screen of the program, located on the menu bar you will see the word "Programming." With the mouse pointing to the word "Programming," left click the mouse. This will open a submenu and you will see the word "Verify" in the submenu. With the mouse pointing to the word "Verify," left click the mouse. The action that was just performed should open a new window with a title of "Verifying Unit's System Parameters." If the communication port has been setup correctly and the KL-4F and KL-4F-PC1 have been plugged into the interconnect and the pc, you should see the word "RETRIEVING" being displayed in the middle of the window. Upon successful reading of the data and comparison to the configuration currently loaded into the program, you will see the phrase "UNIT MATCHES" displayed in the window. If the data that was retrieved from the interconnect does not match the configuration currently loaded into the program you will see the phrase "UNIT DOES NOT MATCH." If you see any other messages, check the COM port selection, make sure the programmer is properly attached and there is power to the interconnect.

Write updates to the Interconnect:

If after reading the device or performing at least a write all to the device and a change to the parameters is made, a write updates can be performed rather than having to write all data again.



ANI Length: This specifies the number of digits used in the ANI. If a value of 0 is used, the unit will work in a single user mode. If a value of 4 is used then the unit will work in a multi-user mode. In a single user mode of operation all speed dial entries are available for use. In a multi-user mode the speed dial entries can be assigned to different users if desired.

Up/Down Digit Relative to ANI Position: This field is only shown if the ANI length is set to 4 digits. This item allows the interconnect to know where to look for the Interconnect Up and Down Digits in relation to the incoming ANI digits. For example, if the selection is made for Leading and the Interconnect Up Digit = * and the Interconnect Down Digit = # then the sequence of entered digits would be:

*ANI = Bring the Interconnect Up with the following ANI
#ANI = Bring the Interconnect Down with the following ANI

and vice versa if the selection for Trailing is made:

ANI* = Bring the Interconnect Up with the following ANI
ANI# = Bring the Interconnect Down with the following ANI

NOTE: You may or may not need the ANI digits when bringing Down the Interconnect depending on the setting of Allow Disconnect with No ANI item located on the Subscribers Configuration Form.

Enable Direct Access: If checked the telephone caller can get directly onto the radio system without a field radio answering or if an access number is entered into the field the caller must enter an access code prior to being given access to the radio system. If using the access number the caller will receive a tone prompt to indicate to the phone caller to enter the access code. Note: This feature may not be allowed by all national communications authorities. Check with your local authorities prior to use.

Tone Digit For Connect: This field sets the DTMF tone that will cause the radio phone patch to take the phone line off hook. The selection range is 0-9, A-D, * and #. The * tone is typically used for connecting.

Tone Digit For Disconnect: This field sets the DTMF tone that will cause the radio phone patch to put the phone line on hook. The selection range is 0-9, A-D, * and #. The # tone is typically used for disconnecting.

Connect/Disconnect Tone Minimum Time: This sets the time in 10 millisecond increments that the tone must be present before the unit will accept the connect or the disconnect command (50 msec to 2.55 sec)

Toll Override Access Management Code: This access code allows a Subscriber or System Operator to override any Toll Restrict Settings that may be in force. Note: Digits used for Interconnect Up and Down should not be used as part of the Toll Override Access Code.

Mobile to Telco DTMF Regenerator Delay: This is a delay from the time the unit brings the Telco off hook and before dialing begins. Since the unit does not detect if dial tone is present a value of 1 to 2 seconds is usually adequate for dial tone to become present before dialing commences.

Allow Over Air Speed Dial Reprogramming: This is a global enable/disable for allowing Subscribers to Program/Not Program Speed Dial locations respectively. This allows the System Operator to disable Over The Air Programming without having to visit each Subscriber Record.

Base/Repeater Type: Select between Simplex, Simplex with Sampling, Half-Duplex and Full-Duplex. Simplex and Simplex with Sampling are used when the RX and TX frequencies are the same. Simplex with Sampling gives the radio user a means to take control of the call if the phone caller is talking too long or is using obscene language. Half-Duplex is used when the RX and TX frequencies are different and the base station does not transmit and receive simultaneously. Full-Duplex is used when the RX and TX frequencies are different and the base station transmits and receives simultaneously.

Privacy Tone Select: If the unit is setup to operate in a Half Duplex mode, this parameter will instruct the unit to generate noise, block audio or allow audio to pass back over the transmitter when the portable in the field is talking.

Channel Busy Release Delay: This programs the amount of time that the interconnect will consider the radio channel to be busy after a radio busy condition has been detected and released. This applies to all types of busy detection inputs. The phone patch will not key-up the radio during the delay period when operating in Simplex modes therefore we recommend you keep it less than 1 second. This field will only affect Half Duplex and Full Duplex during this initial call setup until “ownership” of the channel is acquired.

Channel Busy Detection Source: This allows you to select which of the following sources will be used for radio busy detection; Carrier-Operated Relay (COR) via the COR input lead or the quieting filter via the discriminator input.

COR Logic Polarity: Select whether a Logic Low or Logic High is the active state of the radio’s busy channel.

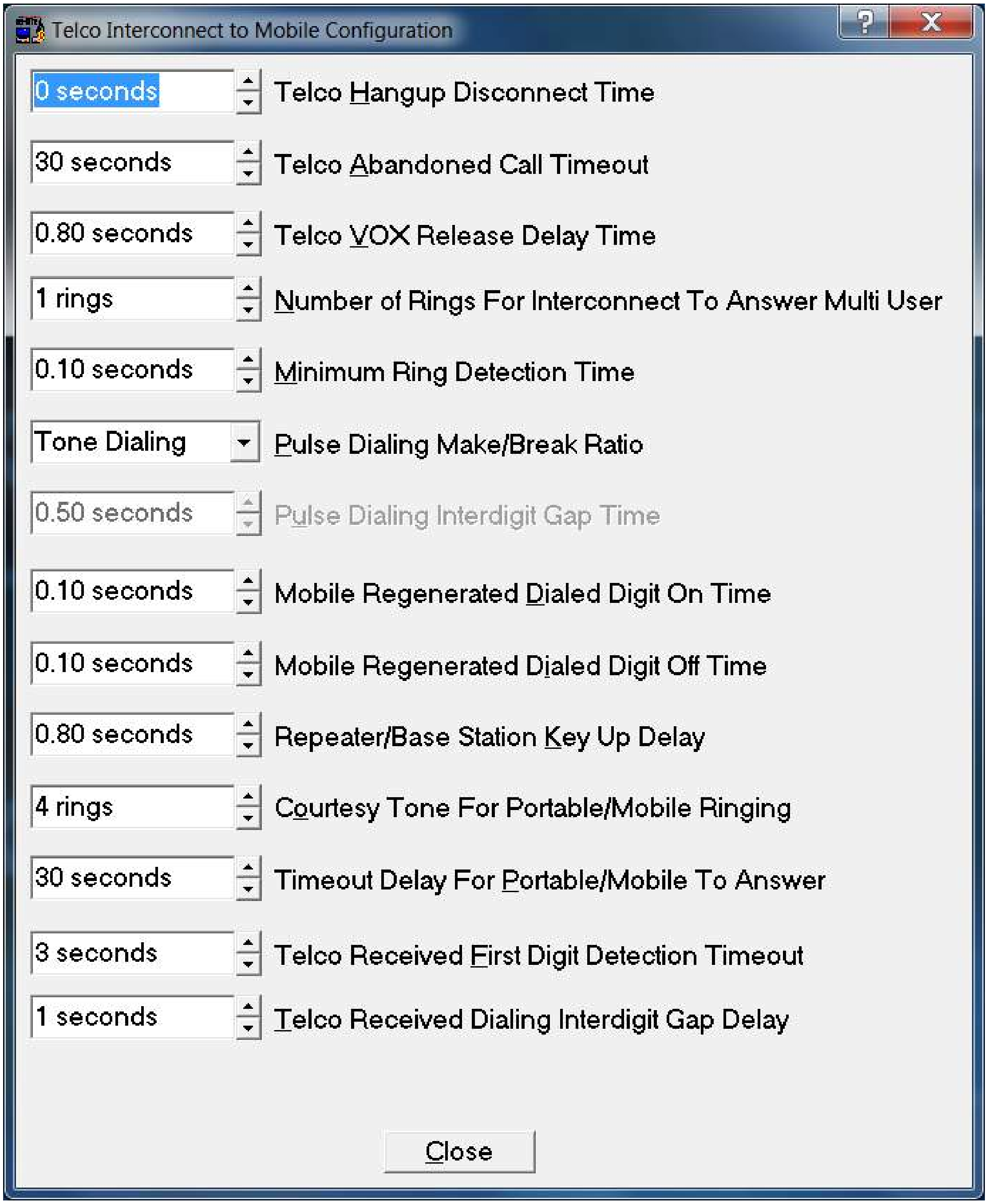
Simplex Telco VOX Sampling Interval: This sets the time in seconds the telephone caller may talk continuously. When this timer expires, a go-ahead beep is transmitted to the field radio and the radio user is given a small window of time to seize control of the conversation.

Simplex Telco VOX Sampling Width: This sets the amount of time in 100 msec increments the telephone caller is locked out, giving the radio an opportunity to seize control.

Simplex Go Ahead Courtesy Tone: When the unit is being operated in a Simplex mode, this checkbox will enable courtesy Go-Ahead Tones to indicate when the channel is clear so the other party can take a turn.

Morse Code ID String: This item is provided to allow the unit to issue a Station Identifier in Morse Code at a programmable time. This item will be ignored if no Characters are entered. Otherwise, the characters entered will be converted to the appropriate Morse Code equivalents before being transmitted.

Morse Code ID Output Delay Time: This item instructs the unit how often (0-60 minutes) to issue the CW-ID String. If the CW-ID String has no characters defined and the Delay Timer is greater than 0 Minutes then nothing will be transmitted. If the Delay Timer is set to 0 Minutes and the CW-ID String has characters defined nothing will be transmitted.



Telco Hangup Disconnect Time: This sets the amount of time that the unit will wait before seizing the phone line after disconnecting any previous call.

Telco Abandoned Call Timeout: Once a connection is completed between the Telco caller and a radio user, this field is used to determine if the Telco caller is active by monitoring the VOX circuit. If no activity is detected on the Telco then a disconnect sequence will be sent to the radio user indicating the loss of Telco activity and the connection between the Telco and radio user will be disconnected.

Telco VOX Release Delay Time: The VOX circuit in the unit is a fast attack detector that triggers the micro's input when someone speaks. To prevent it from dropping out between words, the micro has a programmable release time. We recommend using at least 800ms. This time can be extended if the telephone caller's voice drops out between words.

Number of Rings For Interconnect to Answer Multi-User: This controls how many times the telephone must ring before the unit will go off-hook and answer an incoming Telco call. If set to 2, the unit will go off-hook during the second ring.

Minimum Ring Detection Time: This controls how long a ringing signal from Telco must be present in order to register one ring. A value of 0.8 to 1.0 second should prove adequate for most Telco systems.

Pulse Dialing Make/Break Ratio: If the unit is connected to a telephone line which accepts DTMF dialing, this field should be set to Tone Dialing. If DTMF is not supported, the unit can be made to do pulse dialing by selecting the appropriate make/break ratio. Currently supported make/break ratios are 40/60, 33/67, and 20/30 milliseconds.

Pulse Dialing Inter-digit Gap Time: When using pulse dialing, this controls the amount of time to wait between dialing each digit.

Mobile Regenerated Dialed Digit On Time: This sets the amount of time the DTMF encoder generates a tone to the Telco. We recommend about 100 msec.

Mobile Regenerated Dialed Digit Off Time: This sets the amount of time the DTMF encoder stops generating a tone between digits. We recommend about 100 msec.

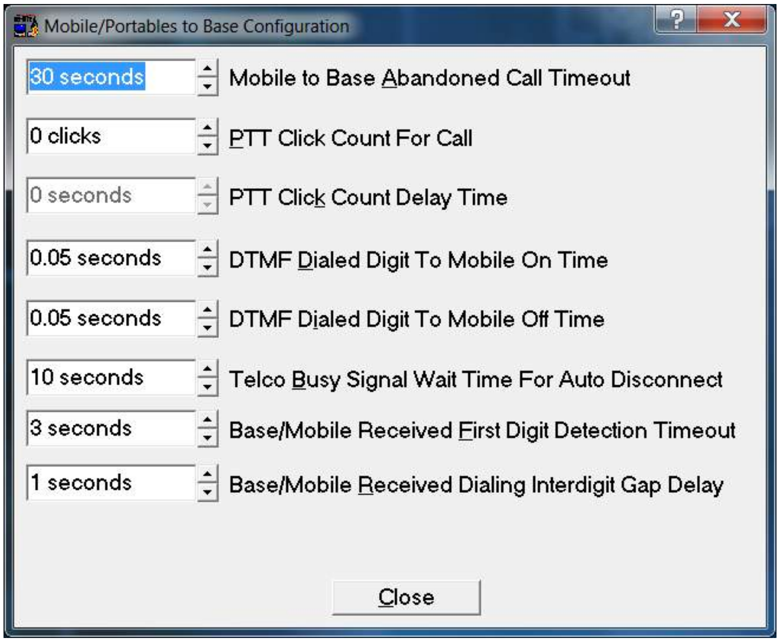
Repeater/Base Station Key-Up Delay: This sets how long the unit will wait to send any tone dialing over the radio channel. This should take into consideration how long it takes the associated transmitter to key up, come up to full power, and how long it takes the receiving unit to wake up its carrier detect.

Courtesy Tone For Portable/Mobile Ringing: This feature is used to send courtesy ringing tones to a radio unit if needed. If a 0 is programmed into this field no courtesy tones are sent to the radio which allows the call to the radio unit to rely on its' decoder for ringing. Please Note the Timeout Delay For Portable/Mobile to Answer register will always override and disconnect when the timer has expired regardless of the setting of this register.

Timeout Delay For Portable/Mobile To Answer: This is used to provide an automatic disconnect if the radio unit that is being called does not answer within the specified period of time. It is not recommended that this ever be set to 0 except for purposes of testing. The recommended setting of this field should be between 5 to 10 seconds.

Telco Received First Digit Detection Timeout: This allows the unit to look for the first incoming DTMF digit from the Telco. If the Telco has not entered a DTMF Digit within this time period the Error Tone is sent to the Telco followed by a Disconnect Tone and then the unit hangs up the Telco connection. The unit then goes back into an idle state and waits for further activity.

Telco Received Dialing Inter-digit Gap Delay: This item allows the unit to look for successive incoming DTMF digits from the Telco after the first digit has been entered. If the Telco has not entered a DTMF Digit within this inter-digit time period, the unit will then ascertain if it can continue to process the already collected information from the Telco or if a data entry error was made. If the unit can continue with the information already collected it will do so. However, if the unit detects an error with the information collected it will then generate the Error Tone message and/or a Disconnect Tone message after which the Telco will be disconnected. The unit will go back into an idle state and wait for further activity.



Mobile to Base Abandoned Call Timeout: This sets the amount of time before the phone patch disconnects its call to the telephone line if the mobile forgets to disconnect. The timer starts after the last transmission from the Radio.

PTT Click Count For Call: This allows a radio subscriber to click the radio PTT x number of times to answer an incoming telephone call, disconnect from a telephone call, or to force a speed dial call on the telco if specified. Otherwise, an error tone will be issued to the radio Subscriber indicating that the feature is disabled. Note: The PTT Clicks for Speed Dial feature will only work if the subscriber has one Speed Dial Entry assigned as Offset Entry 000.

PTT Click Count Delay Time: This specifies how quick a subscriber must click the PTT on the radio x number of times to get the specified number to dial out on the telco. If the specified number is not defined then the subscriber will receive an error tone indicating that this feature is not enabled.

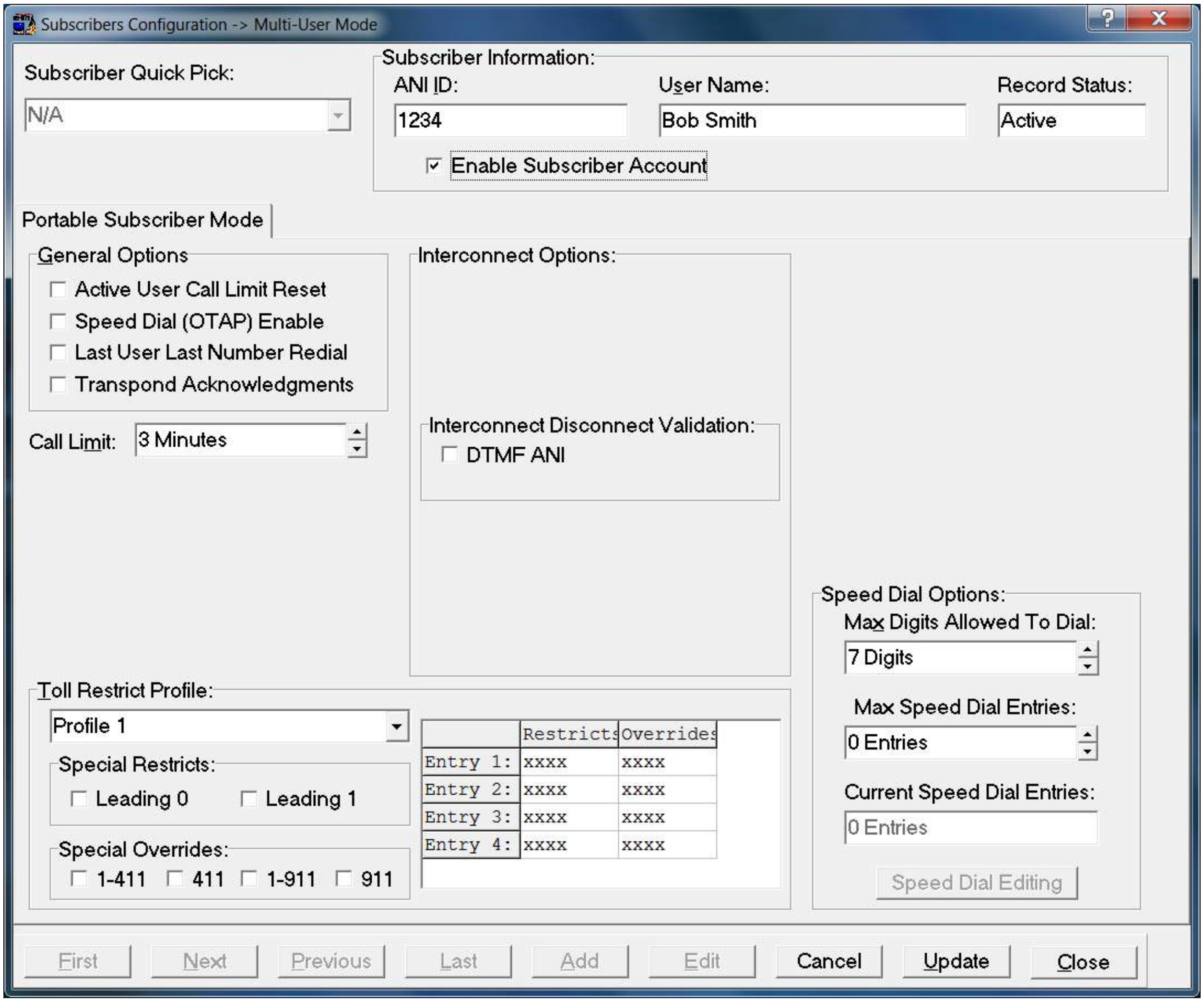
DTMF Dialed Digit To Mobile On Time: This sets the amount of time the DTMF encoder generates a tone to the mobile. We recommend this to be between 60 msec to 100 msec.

DTMF Dialed Digit To Mobile Off Time: This sets the amount of time the DTMF encoder stops generating a tone between digits. We recommend this to be between 40 msec to 100 msec.

Telco Busy Signal Wait Time For Auto Disconnect: The busy signal that results from calling a busy telephone number may prevent the radio from transmitting a disconnect ANI to the unit. To prevent the situation, the radio phone patch can detect the presence of a busy signal. This field controls how long after dialing that the unit will wait for a busy signal. If a busy signal is detected during this time, the unit will automatically disconnect the call.

Base/Mobile Received First Digit Detection Timeout: This item allows the unit to look for the first incoming DTMF digit from the field radio. If the field radio has not entered a DTMF Digit within this time period the Error Tone is sent to the field radio followed by a Disconnect Tone. The unit then goes back into an idle state and waits for further activity.

Base/Mobile Received Dialing Inter-digit Gap Delay: This item allows the unit to look for successive incoming DTMF digits from the radio after the first digit has been entered. If the radio has not entered a DTMF Digit within this inter-digit time period, the unit will then ascertain if it can continue to process the already collected information from the radio or if a data entry error was made. If the unit can continue with the information already collected it will do so. However, if the unit detects an error with the information collected it will then generate the Error Tone message and/or a Disconnect Tone message after which the radio will be disconnected. The unit will go back into an idle state and wait for further activity.



Subscriber Quick Pick: This feature is not available in Single User mode. In Multi-User mode, if ANI Length on the System Configuration Form is greater than 0 digits, this list will provide a sorted pick list of ANI's with the subscriber name if a name was defined. By clicking on the arrow on the right hand side of the field, a list will be provided to select from the defined subscribers in the database.

ANI ID: If ANI Length on the System Configuration Form is greater than 0 digits, this field allows the entry of unique ANI digits. If the ANI Length is specified to be 4 digits and the operator only enters in a 3 digit ANI, then the ANI will be left padded with 0's to force the 4 digit ANI size.

User Name: If ANI Length on the System Configuration Form is greater than 0 digits, this field allows the entry of a User Name to be identified with the ANI that should have been entered. The user name can be up to 20 characters in length.

Record Status: This is an information only field that indicates whether the currently selected subscriber record is Active (enabled) or Disabled.

Enable Subscriber Account: Check this box to enable the subscriber.

Active User Call Limit Reset: If this option is enabled, it will allow the subscriber to reset the call limit timeout timer for another full session of time. This command only works while a subscriber is connected to either an incoming or outgoing telephone call.

Speed Dial (OTAP) Enable: If this option is enabled, it will allow the subscriber to add and modify speed dial entries that are assigned to that subscriber over the air

Last User Last Number Redial: If this option is enabled, it will allow the currently selected subscriber to redial the last number that was entered by them.

Transpond Acknowledgements: If this option is enabled, the field unit must transpond within 2 seconds after being called otherwise the Telco caller will be disconnected.

Call Limit: This item sets the time allowed per connection. 1 minute before the time expires, a warning tone will be sent to the radio letting the subscriber know of the impending disconnection. If the Call Limit Reset option is enabled for that subscriber and the subscriber enters the command to reset the timer a Go Ahead tone will be sent to the subscriber. Otherwise, an Error Tone will be sent to the subscriber.

Interconnect Disconnect Validation: DTMF ANI: If this item is not checked, then the unit will accept any valid incoming Down Digit command to bring the interconnect down. Otherwise, if the item is checked, then the unit will force the subscriber to issue the ANI and also the down digit before allowing the interconnect to be brought down.

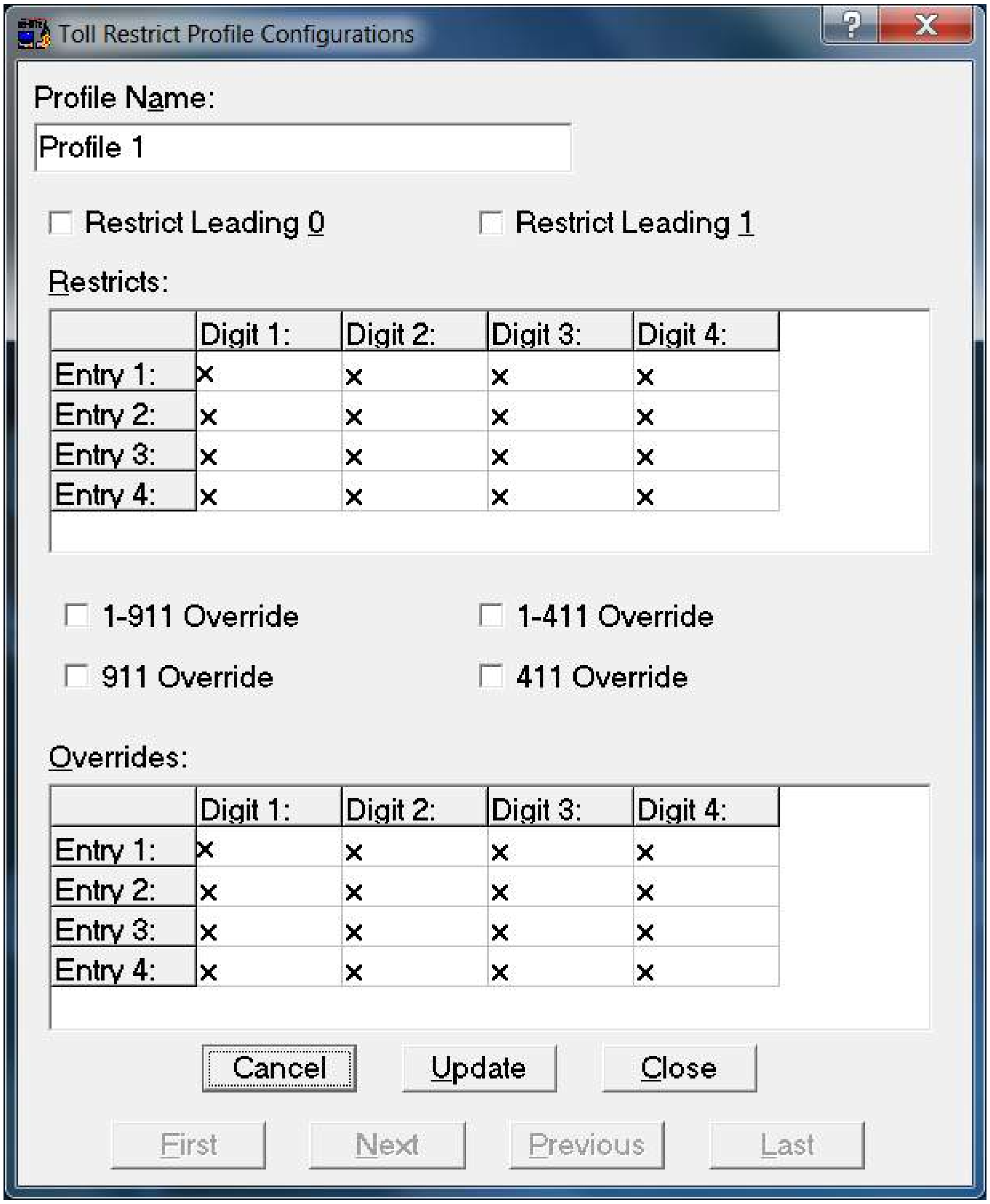
Toll Restrict Profile: The programming screen contains the text "Profile 1." The profile names can be changed to be more descriptive by going to the Toll Restricts Profile Form. During add or edit mode of a subscriber record you can select a Toll Restrict Profile to be applied to the current subscriber. After selecting, you will see the parameters for the profile updated in the fields below and to the right of the selection field. The fields that show the profile are information fields only. To make changes to the profile you must change the profile under menu Config > Toll Restrict Profile.

Max Digits Allowed To Dial: This places a limit on the maximum number of digits that the radio subscriber can dial into the phone system.

Max Speed Dial Entries: This sets a limit on the maximum number of speed dial entries the currently selected subscriber can access.

Note: There are only 100 total available speed dial locations. So if you have 100 subscribers and wanted them to all have speed dial capability, the most you could assign to each subscriber would be 1. Please note that the if 99 entries is assigned to each subscriber and there are 100 subscribers in the system, the assignment of speed dial locations would be on a first come first serve basis.

Current Speed Dial Entries: This is an information only field that shows how many speed dial entries are currently in use by the selected subscriber.



Profile Name: This allows for a name to be assigned to the Toll Restrict Profile for easier identification. Some examples are “International”, “Long Distance”, “Local” and “Restrict 900”.

Restrict Leading 0: When checked, any phone number beginning with a 0 will be considered invalid. You will not need to add any other entries to the restrict matrix unless specific prefix restrictions are desired.

Restrict Leading 1: When checked, any phone number beginning with a 1 will be considered invalid. You will not need to add any other entries to the restrict matrix unless specific prefix restrictions are desired.

Restricts: The columns labeled Digit 1, Digit 2, Digit 3 and Digit 4 are representative of the leading digits in a phone number. The rows labeled Entry 1, Entry 2, Entry 3 and Entry 4 are representative of the total possible verification checks against each phone number passed through the restrict checking filter. The valid entries for each position are any DTMF Digit and also the letter x. It is important to note that even though any DTMF digit may be used, it is highly recommended that the defined Up and Down digits used for the Interconnect not be used in this matrix. The letter x is used to indicate a don’t care entry and is ignored during the verification checks. Note: It is recommended that the restrict digits are defined one digit more than are defined in the Override Matrix.

Overrides: From the above figure, you see the Toll Restrict Profile Overrides matrix. The columns labeled Digit 1, Digit 2, Digit 3 and Digit 4 are representative of the leading digits in a phone number. The rows labeled Entry 1, Entry 2, Entry 3 and Entry 4 are representative of the total possible verification checks against each phone number passed through the overrides checking filter. The valid entries for each position are any DTMF Digit and also the letter x. It is important to note that even though any DTMF digit may be used, it is highly recommended that the defined Up and Down digits used for the Interconnect not be used in this matrix. The letter x is used to indicate a don't care entry and is ignored during the verification checks. Note: It is recommended that the override digits are defined one digit less than are defined in the Restrict Matrix.

Mobile to Landline Calling

There are several ways to call a telephone number in the public switched telephone network from a portable or mobile two-way radio. One method employs using multiple PTT clicks to dial a dedicated per subscriber speed dial on a multi-user system. This requires an ANI with quick multiple PTT clicks to activate the speed dial for the ANI verified subscriber. On a single user system no ANI is required. Multiple clicks following ANI can also bring the system down for multi-user setups.

The other method requires the radio subscriber to use a DTMF keypad. A “Connect” tone (usually a *) is transmitted, usually with ANI, for verification to get line seizure and dial tone. After sending the connect digit release the PTT and listen for the dial tone presence before continuing or the unit will issue an error sequence and disconnect the call. Then the subscriber dials the desired number with his DTMF keypad. The phone patch provides various levels of toll restrictions. A “Disconnect” tone (usually a #) is transmitted, usually with ANI to terminate the call. The caller will send a connect ANI plus dialing sequence to enter the phone patch and the unit will verify the ANI and dial the digits into the phone system using DTMF or pulse dialing. After ringing occurs, the radio subscriber may send additional DTMF digits to activate voice mails or other types of subscriber services. In order to do this the user may have to hold the Push-to-Talk button down on the radio while dialing the digits.

Landline to Mobile

To make a call to a radio from a telephone line, the phone line user must dial the appropriate phone number to reach the interconnect. The unit will detect ringing for the programmed number of times (usually 1 or 2 rings) and then generate a go-ahead tone so that the calling party can dial the mobile extension number in a multi-user system. In a single user system ringing is passed over the air. After the interconnect sends the DTMF over the air to the radio it can ring from its own decoder or the interconnect can transmit a courtesy ring tone. The user may then send multiple clicks to answer the call. When using a base station in simplex mode the interconnect will key up using a programmable VOX timer to key the radio. There is also a simplex sampling mode available to allow the mobile to take control of the interconnect in case of a long-winded or cursing landline caller. There is a full duplex mode when using a repeater with duplex mobiles. There is a half duplex mode when using a repeater with simplex radios. In half duplex when the radio is transmitting to the Telco trans-hybrid side tone audio from the interconnect to the mobile can be passed, muted, or jammed with a tone.

When using the direct connect feature and the interconnect answers the caller will automatically get access to the radio system or will hear a tone prompt to indicate to the caller to enter the access code before being granted access to the radio system. Midian recommends using an access code, to prevent accidental access to the radio system by non-authorized callers. Once access has been granted the caller can talk over the radio system or selectively call a specific unit using the DTMF signaling of the telephone.

STI Manual Control By Dispatcher

The STI-1 and STI-2 can have the STI Option A added for use in HF SSB Systems. HF SSB systems do not offer a COR indication and DTMF signaling is unreliable. In order to use the STI in an HF system the STI Option A must be used. This enables the dispatcher to connect field radios to the phone or phone callers to the field radios. For more information on this option, please reference the STI Option A manual.

Power Requirements

The unit receives its 13–18 VDC from the radio via the P1-9. The unit incorporates a 12V regulator, IC4, to power the Telephone Audio Line Driver, IC15A and B. This 12 volt regulator requires a minimum of 13 volts to properly regulate. There is also a separate analog pseudo-ground bias source, R75/76 and C60, to put a 6V analog pseudo-ground reference on IC15. All of the other audio and logic circuitry utilize a 5V regulator, IC3. IC3 also has a low voltage reset to protect the micro and EEPROM during power glitches. Both regulators are controlled by the on-off switch and are protected by an auto-resettable fuse (F1), surge protector DK1, and RF bypass cap C11. D4 provides polarity protection by opening the fuse if power is reversed. There is also a 2.5V analog pseudo-ground reference for the audio circuitry. IC10A and its associated components R19, R20, C16, 17, 18, 19, 20, & 21, generate this reference.

DTMF Decode Audio Path

The DTMF decoder receives audio from either the Telephone line or the radio. During an incoming land line call, the microprocessor will steer the DTMF audio from the Telephone line via T1, adjustment pot VR6 (REC LEVEL FROM TELCO) and IC9A to FET switch U13C. It is then fed from Pin 4 of IC13C to input coupling cap C1 and gain resistors R2 and R3 of the DTMF decoder chip IC2. The DTMF decoder then presents binary information and strobe to the microprocessor for decoding and processing.

DTMF and voice audio from the radio receiver comes into the RX input and is RF bypassed by C22. It is then presented to IC9B for an adjustable gain controlled by adjustment pot VR2 (REC LEVEL FROM RADIO). DTMF and voice audio out of Pin 7 of IC9B is fed to a hi-pass filter to remove CTCSS/DCS tone squelch via IC9C & IC9D. The output of the hi-pass feeds back to an input on IC13C, Pin 3, where it is again steered back to the decoder circuit as shown in the paragraph above.

DTMF Encode Audio Path

DTMF is generated in IC2 and output on Pin 8 where it is fed into the base of Q3, an emitter-follower that has a FET switch that can steer two different pot-adjustable emitter resistors for transmitting DTMF audio either to the Telephone switch or to the radio transmitter modulator.

DTMF adjustment pot VR4 (DTMF LEVEL TO TELCO) feeds over to IC13B Pin 1, where it’s audio is steered to the RX audio driver circuit, IC15A & B, to feed into the Telephone line. DTMF adjustment pot VR3 (DTMF LEVEL TO RADIO) feeds over to IC13A Pin 13, where the microprocessor steers its audio into the TX audio circuit, IC11C. The output of this circuit feeds the radio transmitter modulator and has an RF bypass cap, C25.

Radio Receive Audio Path

The receive audio from the radio to the Telephone line is applied across RF bypass cap C22. It is then fed into adjustable gain stage IC9B. Pot VR2 (RECEIVE LEVEL FROM RADIO) sets the input level to the DTMF decoder and the high-pass filter IC9C and IC9D. The high-pass filter removes the low frequency CTCSS/DCS if the radio does not already do so. Filtered audio is then fed into Pin 2 of IC13B, where the microprocessor switches it out on Pin 15 of IC13B to pot VR7 (TX LEVEL TO TEL LINE). VR7 can then be set to the appropriate level to drive the Telephone line tip and ring. IC15A and B is a differential line driver that drives Pins 1 & 10 of T2. It has its own 6V analog pseudo-ground reference, R75/76 and C80, on both of the non-inverting inputs. The phone line hybrid T1/T2 employs a surge protection device, DK5 & DK4, and two RF bypass caps to ground C68 and C69.

Radio Transmit Audio Path

The transmit audio path from the Telephone line to the transmitter modulator originates on T1 pin 10. DK2 across the winding of T1 acts as a surge suppresser. Audio is coupled into the hybrid winding of T1 and across pot VR6 (RECEIVE LEVEL FROM TELCO). The audio is fed into gain stage IC9A and coupled over to IC13A, Pin 12. The microprocessor steers the audio through IC13A to Pin 14. Pot VR5 (TX LEVEL TO RADIO) passes the audio through gain stage IC11C, to the TX output via coupling resistor R23 and capacitor C28. Resistor R23 can be lowered if necessary to drive lower impedance Mic inputs. C25 is an RF bypass cap.

Telco Line VOX

The unit is designed to be used in duplex or simplex. In simplex, the radio transmitter must be able to key up automatically when the phone line party desires to talk. Audio from the Telephone line on T1 is fed through U9A to Telco line circuit VOX IC11B and 11A. This is a fast attack circuit through Q5 discharging C40. The output of IC11A is fed into Pin 39/40 of the U1, so it can control the push-to-talk transistor Q1 which provides a logic low to the PTT line of the radio. There is also a PTT LED indicator to show when the radio is being keyed. The release time is programmable in software by the VOX release delay time register.

Busy Lockout Methods

Radio VOX: There is an identical radio anti-VOX circuit that receives audio from two different paths. These paths are selected by the microprocessor, which controls IC14B and also IC14C. If the micro has been programmed for radio VOX busy lockout, then receive audio from IC9B goes into the high-pass filters IC9C and IC9D and into Pin 1 of IC14B, where it is routed to the radio VOX circuit IC12B/A. When voice audio from this path is detected, the microprocessor considers the channel to be busy. There is a busy LED that illuminates when a busy condition is detected.

Hi-pass Filter: If the microprocessor is programmed to use a squelch-quieting busy lockout method, then audio from IC9B is routed into Pin 3 of IC14C or optionally, Pin 5 of IC14C, if the discriminator audio input is used via IC10B. In either case, high frequency squelch noise (above 5.5 KC) is filtered by IC10C and IC10D and fed into Pin 2 of IC14B for detection by the radio busy detection VOX circuit. In this case, if there is no squelch noise due to quieting due to carrier signal, the microprocessor will recognize a busy condition.

COR: There is also available a carrier operated relay input lead that turns on Q6 to tell the microprocessor that the radio's own squelch circuit has detected a signal and therefore provides a busy indication to the interconnect.

Ring Detector

Opto-isolator IC8 pins 5 and 6 are used to detect ringing on the Telco line. This causes Pin 3 of IC8 to go low during the presence of inbound ringing, turning on Ring LED 2 and driving the micro to detect an incoming call.

Line Seizure (Off Hook)

IC8 pins 7 and 8 are used to seize the phone line. This opto-isolator is also used for pulse dialing when DTMF is not available. LED3 indicates Off-Hook seizure or pulse dialing.

Microprocessors and EEPROMs

U1 is a Motorola 8-bit microprocessor and is the controller for the interconnect. It is read-write programmable via the programming interface jack, using a stereo 3.5mm jack. There are two EEPROMS, IC5 and IC6. Q6 & Q7 share a common emitter resistor to feed TX and RX audio to the recorder.

Hybrid Transformer

T1 and T2 is a hybrid transformer for interfacing to the phone line. DIP switch SW2 is used to select phasing caps C70, 71, 72, & 73 for nulling the hybrid. Fine adjustment is accomplished by adjusting balance pot VR8. During receive calibration test a scope can be connected to pin 1 of IC9 and pot VR8 adjusted for minimum trans-hybrid return signal when the hybrid is seizing the phone line and passing radio receive audio to the telephone line.

MIDIAN CONTACT INFORMATION

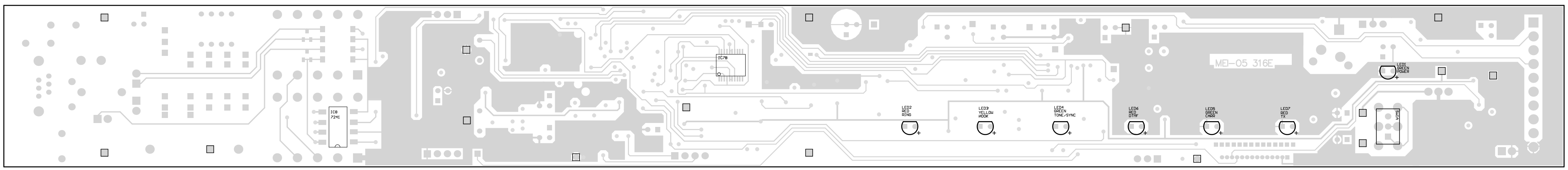
MIDIAN ELECTRONICS, INC.
2302 East 22nd Street
Tucson, Arizona 85713 USA

Toll-Free: 1-800-MIDIANS
Main: 520-884-7981
E-mail: sales@midians.com
Web: www.midians.com

- *This page intentionally left blank* -

ALSO SEE LOWER RIGHT CORNER
P2
SCRAMBLER
POWER
SCR/MODE
COR
OFF HOOK
PTT IN
GND
PTT OUT
BUSY FROM PARALLEL DPT UNIT
COR IN
DTMF RX SWITCH
TEL LINE VOX
TELCO 4 WIRE TO 2 WIRE INTERFACE
LIGHTNING BARRIER
TX LEVEL TO TEL LINE
HOOK
RING
OBSOLETE MODEM
FUTURE HANGUP
FUTURE FLASH
TO PHONE LINE
P4
BEAD
60V@150mA
VOICE, DTMF, 2/5 TONE, 1200 POCSAG
TX OUT
TX LEVEL TO RADIO
TONE AUDIO
VOICE AUDIO
RADIO TX SWITCH
DTMF LEVEL TO RADIO
DTMF TX SWITCH
RX LEVEL FROM RADIO
RX IN
QUIETING FILTER
NOISE QUIETING
VOICE AUDIO
VOX SOURCE SWITCH
VOX OR NOISE BUSY DETECTION
HI-PASS FILTER CTCSS REMOVAL
RECEIVE LEVEL FROM DISC.
DISC IN
LO-PASS FILTER TO CTCSS DETECTOR
CTCSS/DCS DECODE
DTMF LEVEL TO TEL LINE
TX RX MIX FOR WIRE TAP
WIRE TAP AUDIO
TAP AUDIO
RECEIVE LEVEL FROM TEL LINE
SEE NOTE 1
TONE GENERATING DAC 2/5-TONE, CTCSS, DTS, POCSAG
EXT RS-232 SERIAL INPUT FOR EXT MODEM OPTION
MODEM OPTION
PROGRAM
ALSO SEE UPPER LEFT CORNER
P2
SCRAMBLER
TX IN
TX OUT
RX OUT
RX IN
TONE OUT TO FLAT AUDIO FOR POCSAG CTCSS/DCS
TONE OUT
TONE DIGITAL MOD
2/5-TONE, CTCSS, DTS, POCSAG
JUMP P5:1 TO P5:2 - TONE TO NORMAL MIC INPUT
+VIN[13-18V]
GND
PSEUDO GND REF
2.4V
NOTE
* = NOT INSTALLED
1 = REMOVE C61 WHEN SCRAMBLER INSTALLED
MIDIAN ELECTRONICS, INC.
DTI-1/STI-1
REV F-1
DOCUMENT NAME CP
DATE:2000-01-01
DWN BY:DML
APPR
SCHEMATIC
DESIGN:CJS
REV:2013-05-28
COPYRIGHT © 2013
SHEET 1 of 1
PROJECT NUMBER 7316

- This page intentionally left blank -



* = NOT INSTALLED

MIDIAN ELECTRONICS, INC.		
DATE:2000-01-01	DWG BY:DML	APPR
DESIGN:CJS	REV:2013-05-28	

DTI-1/STI-1	REV F-1	DOCUMENT NAME CP
TOP	SHEET 1 of 8	PROJECT NUMBER 7316
COPYRIGHT©2013		

* = NOT INSTALLED

MIDIAN ELECTRONICS, INC.		
DATE:2000-01-01	DWG BY:DML	APPR
DESIGN:CJS	REV:2013-05-28	

DTI-1/STI-1	REV F-1	DOCUMENT NAME CP
TOP	SHEET 2 of 8	PROJECT NUMBER 7316
COPYRIGHT©2013		

* = NOT INSTALLED

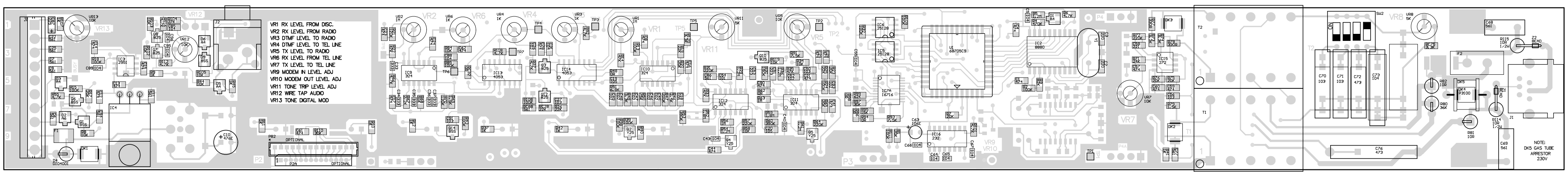
MIDIAN ELECTRONICS, INC.		
DATE:2000-01-01	DWG BY:DML	APPR
DESIGN:CJS	REV:2013-05-28	

DTI-1/STI-1	REV F-1	DOCUMENT NAME CP
TOP	SHEET 3 of 8	PROJECT NUMBER 7316
COPYRIGHT©2013		

* = NOT INSTALLED

MIDIAN ELECTRONICS, INC.		
DATE:2000-01-01	DWG BY:DML	APPR
DESIGN:CJS	REV:2013-05-28	

DTI-1/STI-1	REV F-1	DOCUMENT NAME CP
TOP	SHEET 4 of 8	PROJECT NUMBER 7316
COPYRIGHT©2013		



* = NOT INSTALLED

MIDIAN ELECTRONICS, INC.			DTI-1/STI-1	REV F-1	DOCUMENT NAME CP
DATE:2000-01-01	DWG BY:DML	APPR	BOTTOM	SHEET 5 of 8	PROJECT NUMBER 7316
DESIGN:CJS	REV:2013-05-28		COPYRIGHT©2013		

* = NOT INSTALLED

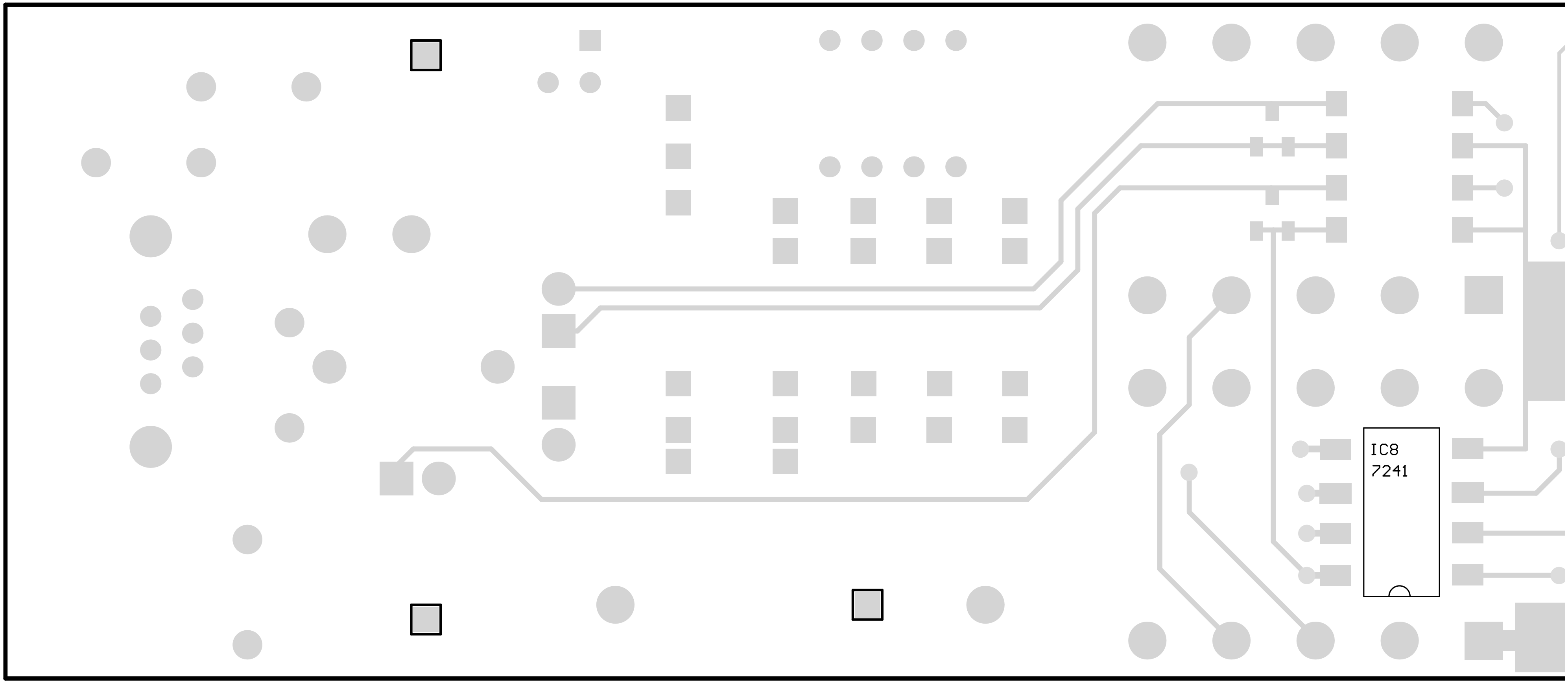
MIDIAN ELECTRONICS, INC.			DTI-1/STI-1	REV F-1	DOCUMENT NAME CP
DATE:2000-01-01	DWG BY:DML	APPR	BOTTOM	SHEET 6 of 8	PROJECT NUMBER 7316
DESIGN:CJS	REV:2013-05-28		COPYRIGHT©2013		

* = NOT INSTALLED

MIDIAN ELECTRONICS, INC.			DTI-1/STI-1	REV F-1	DOCUMENT NAME CP
DATE:2000-01-01	DWG BY:DML	APPR	BOTTOM	SHEET 7 of 8	PROJECT NUMBER 7316
DESIGN:CJS	REV:2013-05-28		COPYRIGHT©2013		

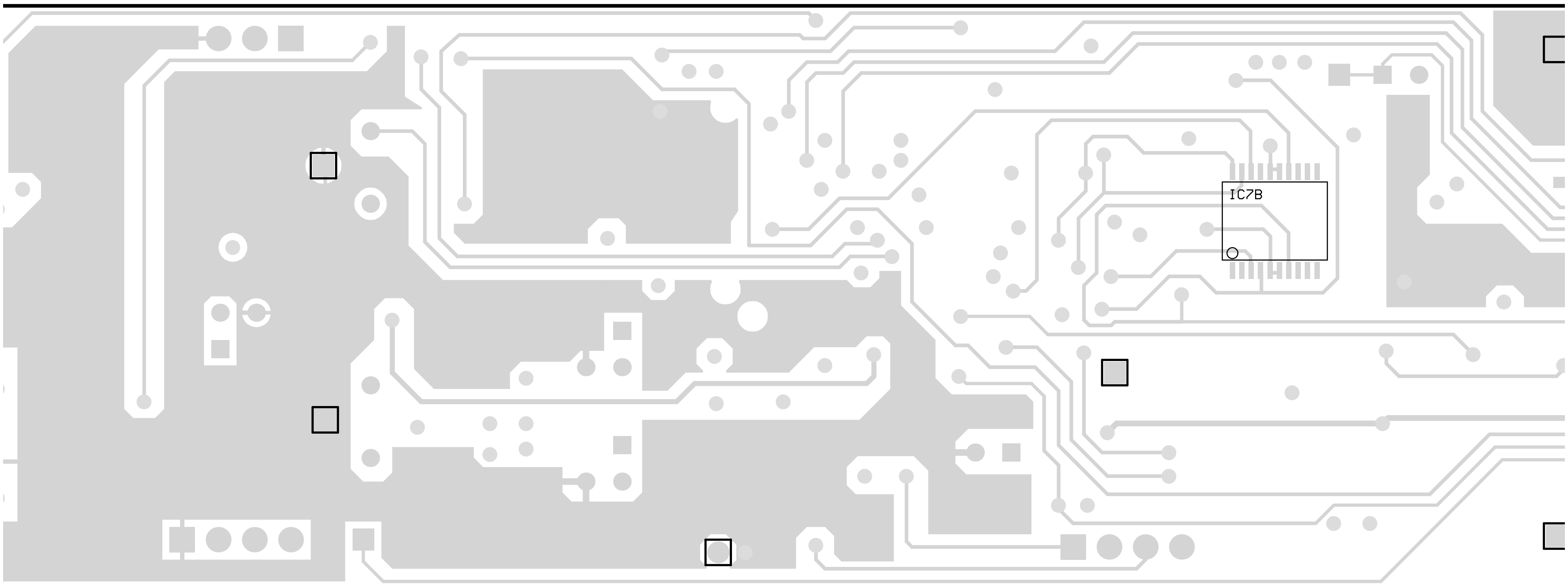
* = NOT INSTALLED

MIDIAN ELECTRONICS, INC.			DTI-1/STI-1	REV F-1	DOCUMENT NAME CP
DATE:2000-01-01	DWG BY:DML	APPR	BOTTOM	SHEET 8 of 8	PROJECT NUMBER 7316
DESIGN:CJS	REV:2013-05-28		COPYRIGHT©2013		



* = NOT INSTALLED

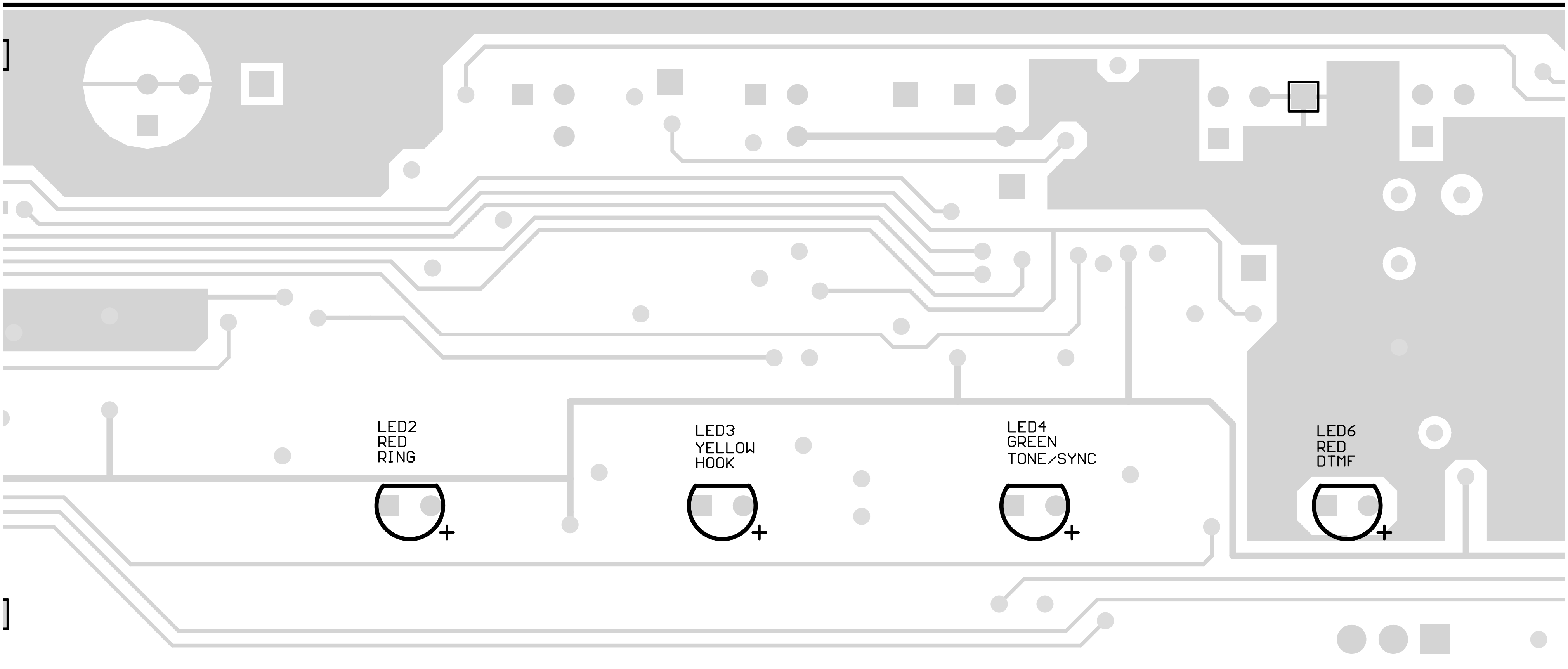
MIDIAN ELECTRONICS, INC.			DTI-1/STI-1	REV F-1	DOCUMENT NAME CP
DATE:2000-01-01	DWG BY:DML	APPR	TOP	SHEET 1 of 8	PROJECT NUMBER 7316
DESIGN:CJS	REV:2013-05-28		COPYRIGHT © 2013		



* = NOT INSTALLED

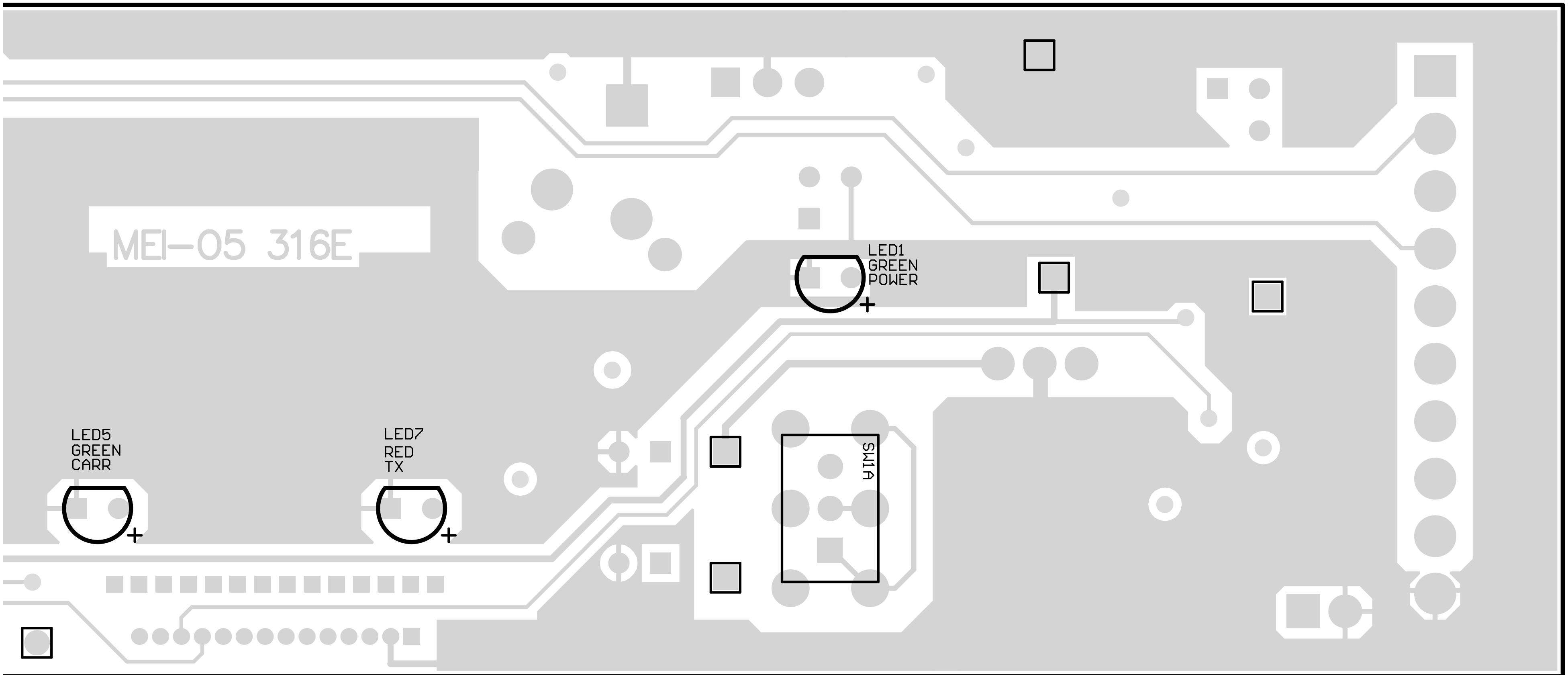
MIDIAN ELECTRONICS, INC.		
DATE:2000-01-01	DWG BY:DML	APPR
DESIGN:CJS	REV:2013-05-28	

DTI-1/STI-1	REV F-1	DOCUMENT NAME CP
TOP	SHEET 2 of 8	PROJECT NUMBER 7316
COPYRIGHT © 2013		



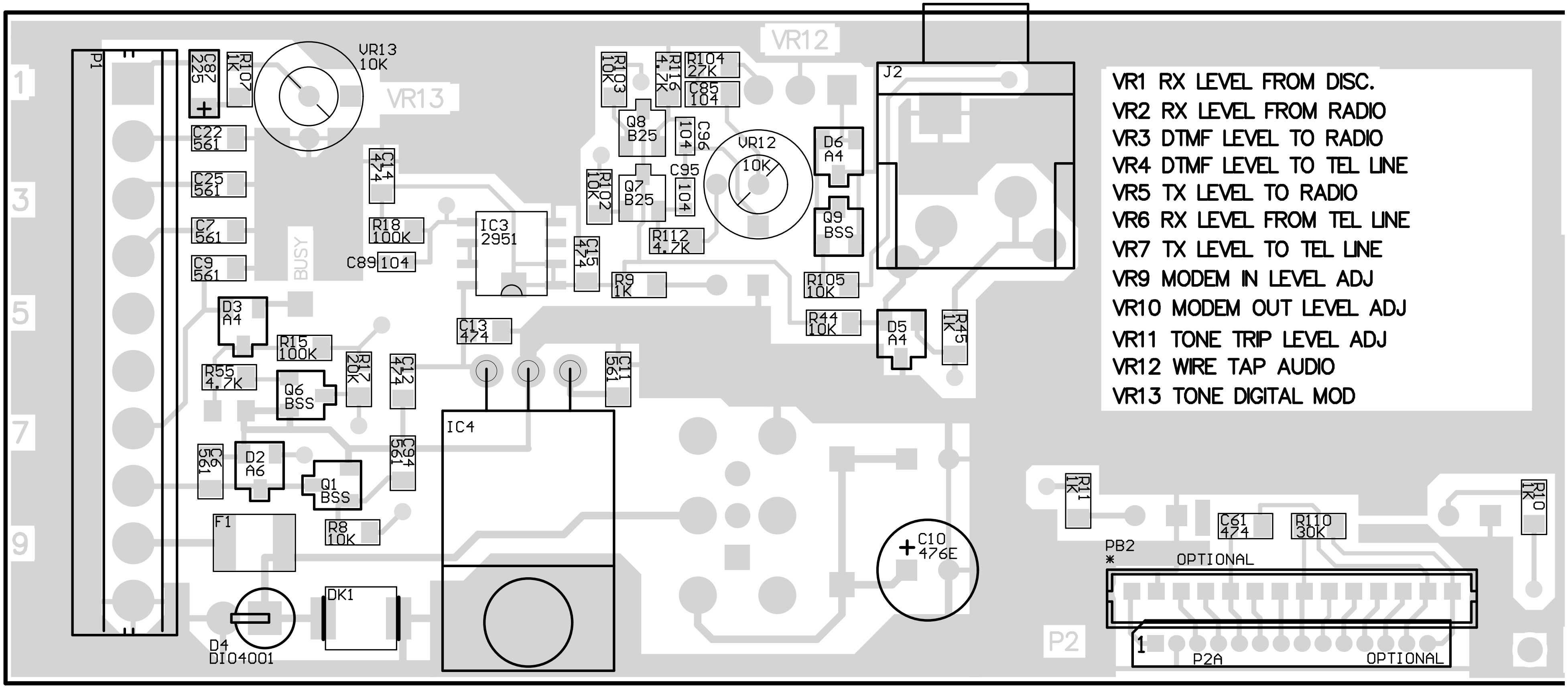
* = NOT INSTALLED

MIDIAN ELECTRONICS, INC.			DTI-1/STI-1	REV F-1	DOCUMENT NAME CP
DATE:2000-01-01	DWG BY:DML	APPR	TOP	SHEET 3 of 8	PROJECT NUMBER 7316
DESIGN:CJS	REV:2013-05-28		COPYRIGHT © 2013		



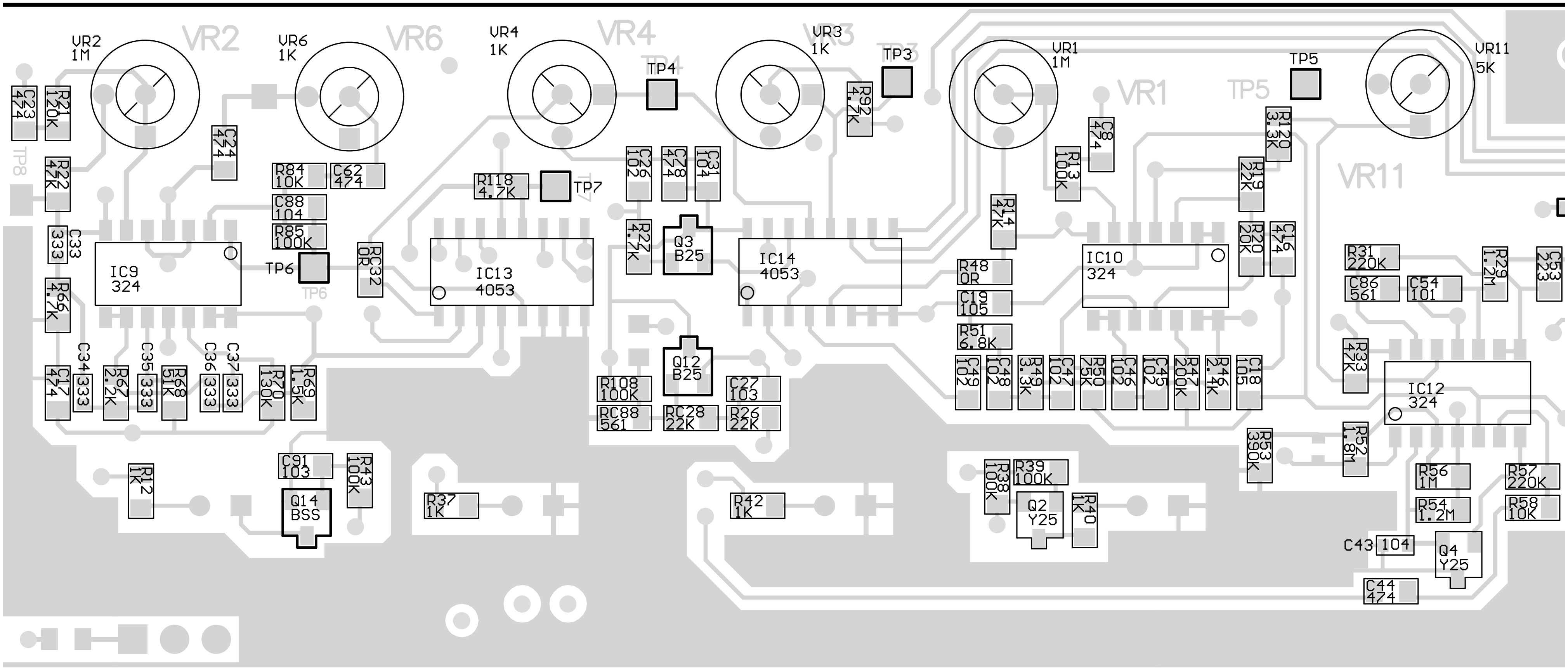
* = NOT INSTALLED

MIDIAN ELECTRONICS, INC.			DTI-1/STI-1	REV F-1	DOCUMENT NAME CP
DATE:2000-01-01	DWG BY:DML	APPR	TOP	SHEET 4 of 8	PROJECT NUMBER 7316
DESIGN:CJS	REV:2013-05-28		COPYRIGHT © 2013		



* = NOT INSTALLED

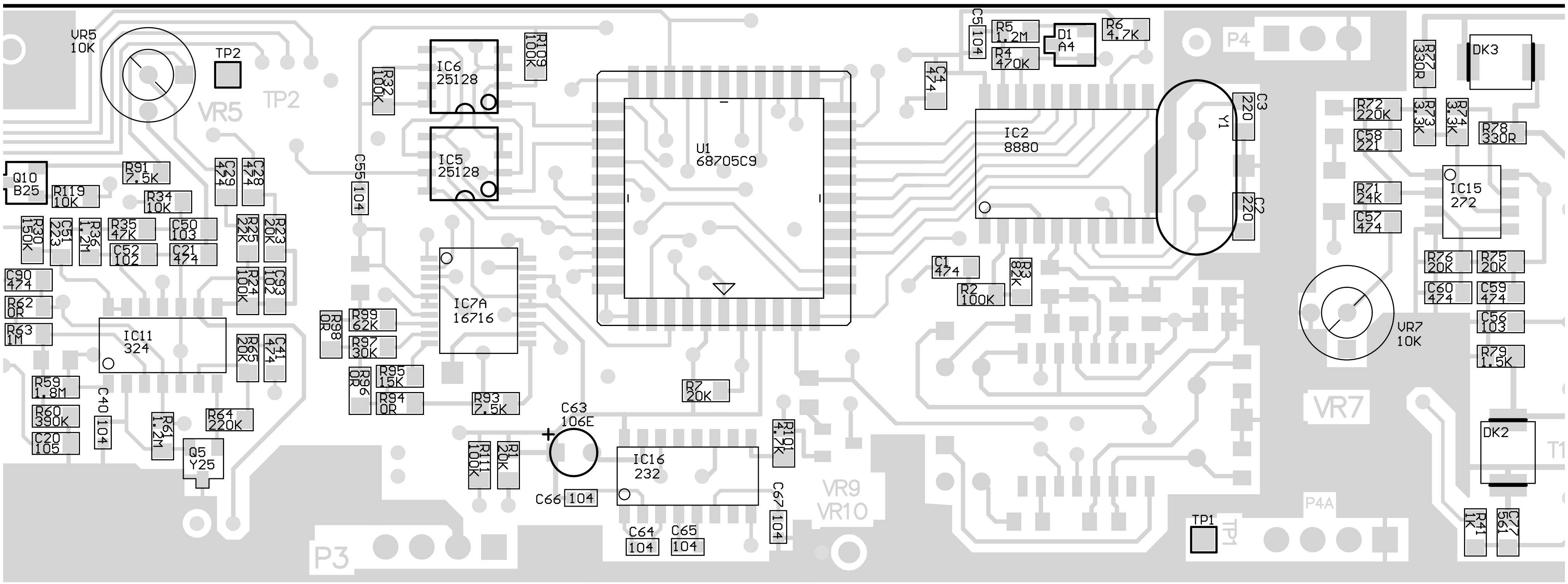
MIDIAN ELECTRONICS, INC.			DTI-1/STI-1	REV F-1	DOCUMENT NAME CP
DATE:2000-01-01	DWG BY:DML	APPR	BOTTOM	SHEET 5 of 8	PROJECT NUMBER 7316
DESIGN:CJS	REV:2013-05-28		COPYRIGHT © 2013		



* = NOT INSTALLED

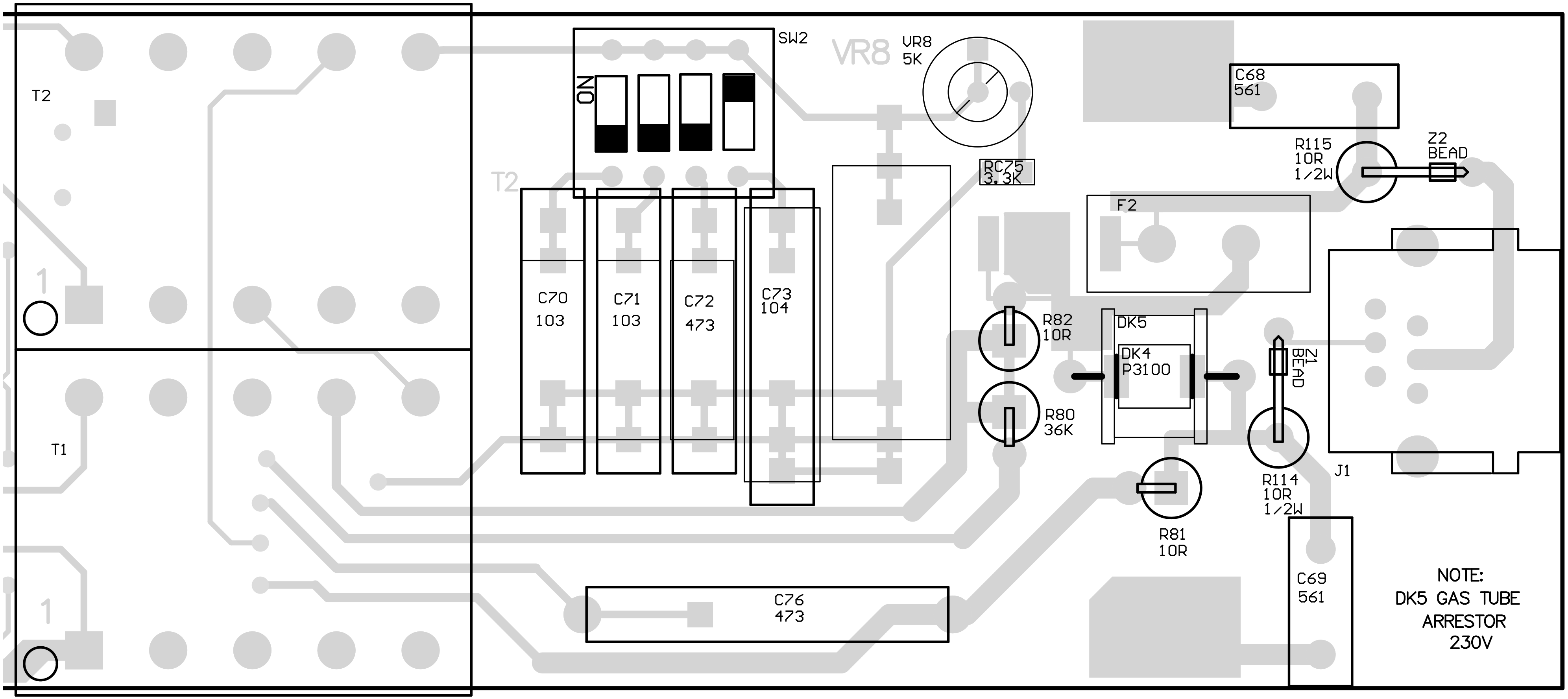
MIDIAN ELECTRONICS, INC.		
DATE:2000-01-01	DWG BY:DML	APPR
DESIGN:CJS	REV:2013-05-28	

DTI-1/STI-1	REV F-1	DOCUMENT NAME CP
BOTTOM	SHEET 6 of 8	PROJECT NUMBER 7316



* = NOT INSTALLED

MIDIAN ELECTRONICS, INC.			DTI-1/STI-1	REV F-1	DOCUMENT NAME CP
DATE:2000-01-01	DWG BY:DML	APPR	BOTTOM	SHEET 7 of 8	PROJECT NUMBER 7316
DESIGN:CJS	REV:2013-05-28		COPYRIGHT © 2013		

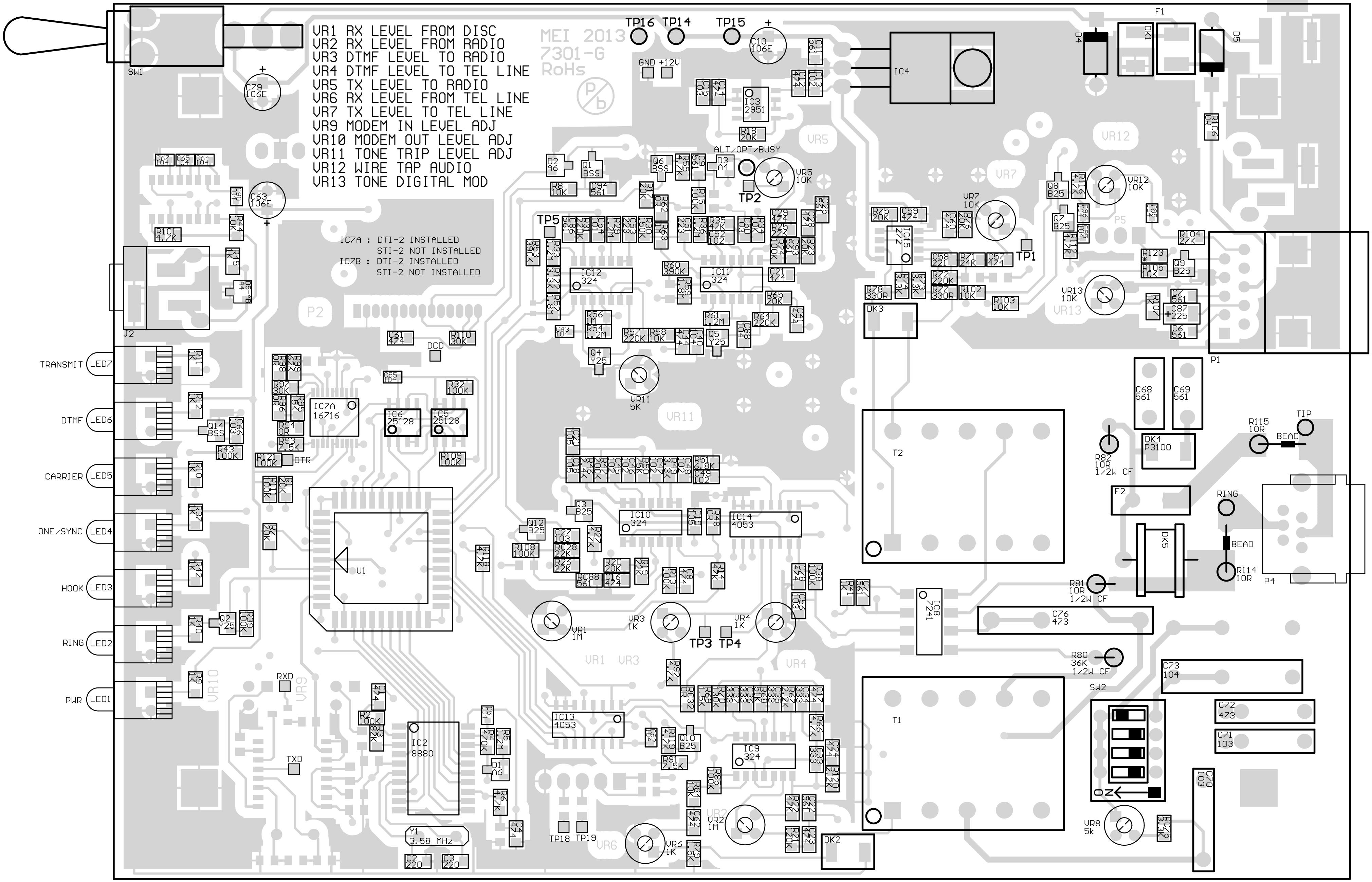


* = NOT INSTALLED

MIDIAN ELECTRONICS, INC.			DTI-1/STI-1	REV F-1	DOCUMENT NAME CP
DATE:2000-01-01	DWG BY:DML	APPR	BOTTOM	SHEET 8 of 8	PROJECT NUMBER 7316
DESIGN:CJS	REV:2013-05-28		COPYRIGHT © 2013		

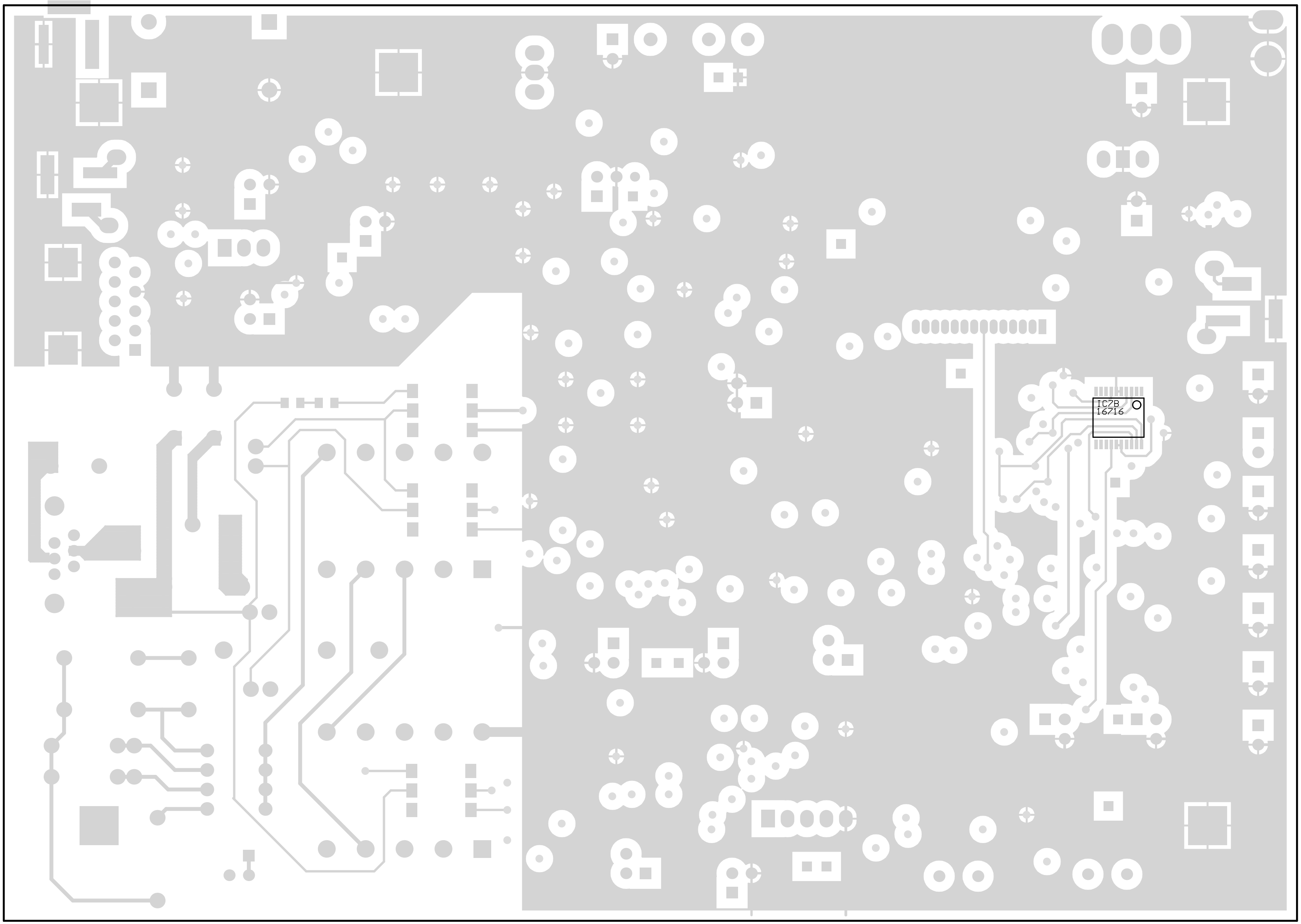
P2 SCRAMBLER
ALSO SEE LOWER RIGHT CORNER
DTMF ENCODE DECODE
DTMF RX SWITCH
TEL LINE VOX
DTMF LEVEL TO RADIO
DTMF TX SWITCH
TONE AUDIO
VOICE AUDIO
RADIO TX SWITCH
TX LEVEL TO RADIO
VOICE, DTMF, 2/5 TONE, 1200 POCSAG
BUSY FROM PARALLEL DPT UNIT
RX LEVEL FROM RADIO
QUIETING FILTER
NOISE QUIETING
VOX SOURCE SWITCH
VOX OR NOISE BUSY DETECTION
HI-PASS FILTER CTCSS REMOVAL
RECEIVE LEVEL FROM DISC.
LO-PASS FILTER TO CTCSS DETECTOR
CTCSSS/DCS DECODE
2/5-TONE, CTCSS, DTS, POCSAG
TONE OUT TO FLAT AUDIO FOR POCSAG CTCSS/DCS
TONE DIGITAL MOD
JUMP P5:1 TO P5:2 - TONE TO NORMAL MIC INPUT
TONE GENERATING DAC 2/5-TONE, CTCSS, DTS, POCSAG
TELCO 4 WIRE TO 2 WIRE INTERFACE
LIGHTNING BARRIER
TX LEVEL TO TEL LINE
OBSOLETE MODEM
HOOK
RING
FUTURE HANGUP
FUTURE FLASH
TO PHONE LINE P4
MODEM OPTION
DTMF LEVEL TO TEL LINE
TX RX MIX FOR WIRE TAP
WIRE TAP AUDIO
RECEIVE LEVEL FROM TEL LINE
SEE NOTE 1
PROGRAM
EXT RS-232 SERIAL INPUT FOR EXT MODEM OPTION
PSEUDO GND REF
NOTE
* = NOT INSTALLED
1 = REMOVE C61 WHEN SCRAMBLER INSTALLED
MIDIAN ELECTRONICS, INC.
DTI-2/STI-2
SCHEMATIC
DATE:1998-12-01
DWN BY:DML
APPR
DESIGN:CJS
REV:2013-05-28
COPYRIGHT © 2013
REV G-1
DOCUMENT NAME CP
SHEET 1 of 1
PROJECT NUMBER 7301

- *This page intentionally left blank* -



✱ = NOT INSTALLED

MIDIAN ELECTRONICS, INC.			DTI-2/STI-2	REV G-1	DOCUMENT NAME CP
DATE:1998-12-01	DWG BY:DML	APPR	TOP	SHEET 1 of 2	PROJECT NUMBER 7301
DESIGN:CJS	REV:2013-05-28		COPYRIGHT © 2013		



✱ = NOT INSTALLED

MIDIAN ELECTRONICS, INC.			DTI-2/STI-2	REV G-1	DOCUMENT NAME CP
DATE:1998-12-01	DWG BY:DML	APPR	BOTTOM		
DESIGN:CJS	REV:2013-05-28			SHEET 2 of 2	PROJECT NUMBER 7301