Vega® a MARK IV company
Signaling Products

Instruction Manual

098-0301

Model C-550 Dial-up Tone-Remote Console
Model RP-250 Dial-up Tone-Remote Adapter

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Introduction

The Vega C-550/RP-250 dial-up tone remote system provides a reliable and inexpensive means of remotely controlling two-way radio base stations over standard Public Switched Telephone Network (PSTN) lines. The system consists of the Model C-550 console and the Model RP-250 remote station panel. Unlisted and dedicated telephone numbers are recommended at both the console and the base station to maintain near 100% line availability; this is not a requirement because the system is very secure against unauthorized use of the base-station transmitter. A three-part tone burst is required to initiate and maintain the base station in transmit. As shipped, anyone dialing the base-station number may listen to any audio present for about 40 seconds, but cannot transmit. If security against unauthorized listening is also desired, the base-station unit may be solder-bridge programmed to connect receiver audio to the telephone line only after decoding a PTT or MONITOR tone burst from a C-550 console.

The system is also secure against unauthorized use of the base-station telephone line by mobile operators, because the only telephone number that a mobile operator can reach is the C-550 console number.

Multiple C-550 consoles may be used with a single telephone line and a single RP-250 panel. However, like extension telephones, transmit and receive audio is attenuated when more than one console is off-hook at the same time. A maximum of two C-550 consoles per system is recommended, unless a switchboard or some other type of telephone line switching is used. The C-550 consoles are prepared for simultaneous parallel station operation by the inclusion of notch filters, which greatly attenuate PTT tone audio from one console, which would otherwise appear in the receive audio of the other console.

Auxiliary audio input and PTT control terminals are provided in the C-550 console, in order that audio from paging or other auxiliary equipment may be transmitted by the base station in the same manner as voice communications.

Console Originate

Upon lifting the C-550 console handset, the telephone number of the base station is automatically dialed about two seconds after dialtone is detected. The RP-250 at the base station automatically goes off-hook upon ring detection and answers with a beep after a 2-second billing delay. As shipped, monitor audio (CTCSS-disabled audio) is connected to the telephone line as soon as the beep tone is completed. The console operator may then listen to all base-station audio for as long as desired from the handset earpiece or, if the speaker switch is pressed, from the loudspeaker. If the base station has multiple users, the operator may momentarily depress the PTT switch on the handset, which will switch the base station to CTCSS (Continuous Tone Coded Squelch System) reception, and only the audio having the proper CTCSS frequency will be heard. PTT switch operation automatically resets the console for handset-only operation. This prevents speaker-to-microphone feedback. After releasing the PTT switch, the speaker may be enabled again at any time by pressing the speaker switch.

Monitoring of the base station for long periods is possible due to a "refresh base station timer" tone burst, which is automatically generated about every 25 seconds after off-hook or after the last PTT activity by the console operator.

When the console operator wishes to call a mobile station, operation is the same as with standard tone remote systems over a leased line, except for the delay between off-hook and the answer-back beep. Depressing the PTT switch generates a tone burst followed by a low-level PTT tone which switches the base station to transmit. The low-level PTT tone is notched out of the audio that is transmitted to the mobile stations. When the console operator releases the PTT switch, the base-station panel connects CTCSS audio from the mobile operator to the telephone line.

When communications between the console and mobile operators are completed and the console operator does not wish to monitor mobile-to-mobile transmissions, the base-station to telephone-line connection can be broken (i.e., the base-station telephone can be caused to go on-hook) in three different ways. The console operator can just hang up the handset and the base station will automatically go on hook in 15 to 40 seconds; or the console operator can press the monitor switch twice within a 1-second period before hanging up; or the mobile operator may transmit the "reset" code. In the last two cases, the base station will go on-hook immediately.

If the C-550 telephone line is not dedicated to base-station operation or if the C-550 rings from a wrong-number call, audio from any ordinary telephone can be heard, and simplex communications can take place, but a tone burst followed by an annoying low-level high-pitched (2174 Hz) sound will be heard by the caller upon PTT switch operation by the C-550 operator.

The C-550 wall power supply requires only 8 watts or less and should be plugged into a non-wallswitch-controlled outlet. If the C-550 is unpowered and the handset is off-hook, the telephone line is seized but no audio, including dialtone, telephone-company recordings, and howler sounds, can be heard. The C-550 handset should therefore

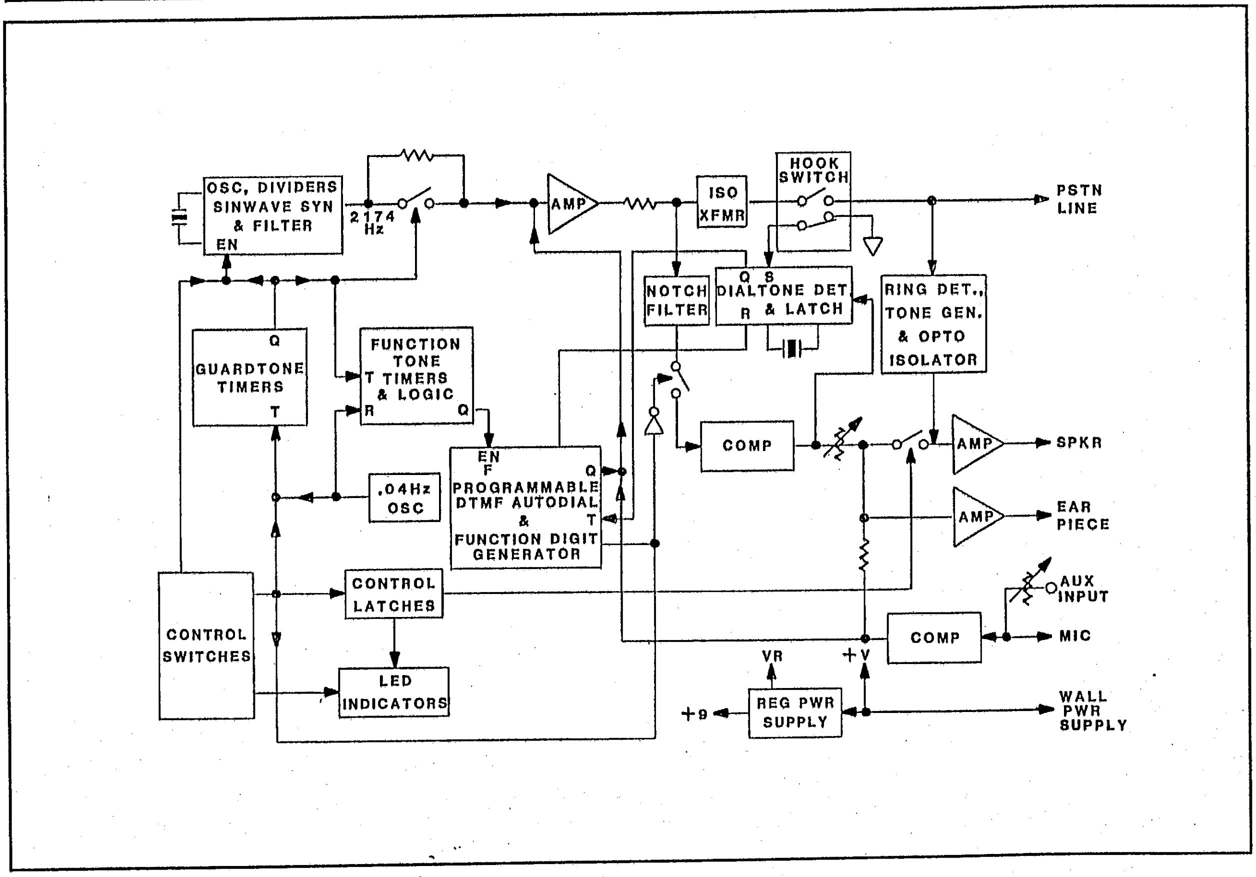


Figure 1. C-550 console block diagram.

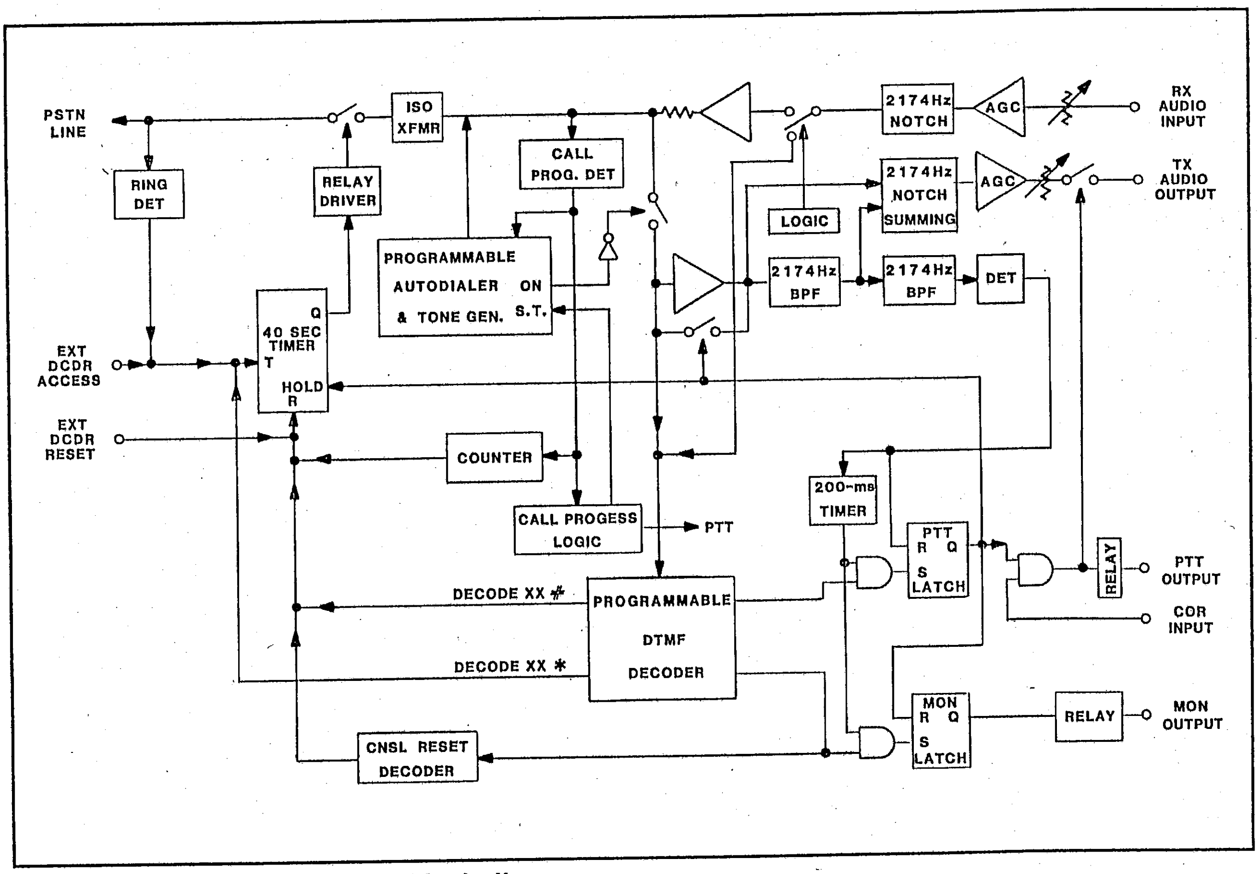


Figure 2. RP-250 station panel block diagram.

be placed on-hook when unattended and during power failures.

Mobile Originate – General

The RP-250 remote station panel provides internal programmable DTMF PSTN access/reset decoders and input terminals for external access/reset logic.

Use of the internal DTMF access/reset decoders requires the mobile stations to have means to transmit a three-digit DTMF access or reset message. A DTMF microphone is usually the most available and least expensive way to provide this capability.

The external decoder input terminals are operated by grounding or by a logic low from 8 to 28 V_{dc} logic. The external decoders may be extra CTCSS frequency decoders. In this case, the mobile stations are required to have radios with an operator-accessible switch for changing CTCSS frequencies, and the base station is required to have the additional CTCSS decoders.

If mobile-to-mobile communications are seldom, if ever, used, logic from the existing CTCSS decoder can often be used to operate the access decode input of the RP-250. In this case, no added equipment is required. However, if mobile-to-mobile transmissions are used with this method of operation, the panel will go off-hook and dial the console telephone number upon the termination of any first transmission. In addition, during "dead air" time (assuming proper input to the panel's COR input), ringback or other call-progress tones will be transmitted. Also, at the end of additional mobile transmissions, a short "off-hook" beep will be transmitted. The mobile operator has no means to terminate the call except by waiting for automatic reset after six rings or 40 seconds.

Mobile Originate

Using the DTMF microphone method as an example, the mobile-originate sequence is as follows. The mobile operator monitors the radio channel, and, if idle, transmits a three-digit DTMF access code (two programmable digits, last digit "*"). The three-digit DTMF message is decoded by the RP-250 panel at the base station, which triggers a 40-second timer and seizes the telephone line. The telephone company (telco) detects the off-hook and connects a dialtone generator to the line. The RP-250 panel detects the dialtone and triggers the autodialer which has been programmed to dial the telephone number at the C-550 console. Telco decodes the autodialed number, connects a ringback tone generator to the base-station line, and connects a ringer-voltage generator to the console line.

The ringback tone on the base-station line is detected by the RP-250 panel, which switches the base station to transmit and supplies a tone signal to the modulator for the duration of the ringback signal. The mobile operator hears the transmitted tone signal, which has the same duration characteristics (2 seconds on, 4 seconds off) as the telco's ringback signal. In the meantime, the C-550 console is ringing.

When the C-550 console operator goes off-hook, ringing at the console and ringback at the base station stops. If, as recommended, the telco line at the console is dedicated to base-station operation, the console operator knows that a mobile station is calling, presses the PTT switch, and answers the mobile's call. Operation from now on is the same as a console-operator originate call.

When the mobile operator dials the access code, the console operator may be monitoring the channel and the telco line is already in use. In this case, the RP-250 panel will briefly switch the base station to transmit and generate a beep tone which will be heard by both the mobile and console operators.

If the C-550 console's line is busy when autodialed by the RP-250 panel at the base station, the base station will transmit a tone having the same duration characteristics as the telco's busy signal (1/2 second on, 1/2 second off). After six or seven on-off cycles of the busy signal, transmission stops and the RP-250 panel goes on-hook (releases the telco line).

If no dialtone is detected after access decode and line seize (all of the telco's lines are busy), the RP-250 panel will wait for dialtone to appear before autodialing. If all of the telco's lines remain busy for some period of time, the telco will connect a busy signal to the line. This will not be detected by the RP-250 panel which at the moment is programmed to detect dialtone, and the unit will remain off-hook for about 40 seconds from the initial line seize. The mobile operator will hear nothing if no dialtone occurs, and may think that he is out of range and may retransmit the access code. Anytime the access message is decoded and the RP-250 panel is already off-hook (for more than 2 seconds), an "off-hook" beep tone will be transmitted to the mobile operator.

When a wrong number is reached due to telco or programmer's error, voice from the wrong number may cause short tone-modulated transmissions (corresponding to words or syllables over 150 ms in length), but no voice will be transmitted. If transmission does occur, the RP-250 panel will terminate the call by going on-hook after six on-off transmitted tone transitions. If less than six tone-modulation transitions occurred during the wrong number, the RP-250 panel will go on-hook in

DTMF DIGIT	S1 THRU S5 SW POS 5 / 1	6 / 2	7 / 3	8 / 4
DIGIT OFF	0	0	0	0
1	0	0	0	1
2	0	0	1	0
3	0	0	1	1
4	0	1	0	0
5	0	1	0	1
6	0	1	1	0
7	0	1	1	1
8	1	0	0	0
9	1	0	0	1
0	1	0	1	0
*	1	0	1	1
#	1	1	0	0
A	1	1	0	1
B	1	1	1	0
C	1	1	1	1

1=ON, 0=OFF

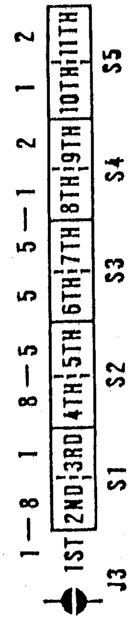


Chart 1. Autodial programming

about 40 seconds from the time of access decode.

The mobile operator may terminate the call at any time by transmitting the reset code ("#" third digit), except during an automatic tone transmission from the base station. (If the base station is a repeater type with the ability to transmit and receive simultaneously and with proper RP-250-to-radio interconnections, a mobile reset is possible at any time.)

Setup, Adjustments, and Programming

C-550 Disassembly

Access to autodial programming switches and auxiliary input terminals is obtained by loosening two screws on the bottom of the console and "folding" the case forward. This opens up the entire unit for setup. Make sure that the wall power supply is unplugged during disassembly and assembly to prevent accidental short circuits.

C-550 Autodial Programming

Any PSTN telephone number up to 11 digits may be programmed, except that the first digit of an 11-digit number must be the DTMF digit "1".

To program a standard seven-digit telephone number, refer to Chart 1 or to the programming label on the bottom of the unit; move all switches on S1 and the first four switches on S2 to their off positions and program the seven desired digits on the last four switches of S2 and on S3, S4, and S5. To program a standard 10-digit long-distance telephone number, use all programming switches starting with the first four switches of S1 and, if in your area the digit "1" must precede a long-distance number, close J3.

Dialing takes place at about seven digits per second with one digit pausing after the first and fourth digits of an 11-digit programming field.

Because this 11-digit field is scanned for programming, dialing requires about 2 seconds regardless of the number of digits.

C-550 Console Adjustments

Other than the volume control on the front panel and the auxiliary input level control on the PC board, no customer adjustments are required, and, with the exception of the factory-preset 2174-Hz notch potentiometers R6 and R12, no others are present.

If the auxiliary audio input terminals TB1-3 and TB1-2 are used, input a normal auxiliary audio level and adjust R53 for about 1 $V_{p\text{-}p}$ or 350 mV_{rms} at TP1. This is about 3.5 dB above the threshold of compression. Increased level at TP1 does not increase line output level and may cause excessive noise or distortion.

RP-250 Autodial Programming

Refer to Chart 1 or the programming label inside the RP-250 cover and to the C-550 programming instructions. Program the C-550 telephone number using S1 through S5 and, if long distance, J3.

RP-250 DTMF Access/Reset Code Programming

The first two digits of the three-digit access/reset code are programmed by S6. The third digit is a fixed "*" for access and "#" for reset. Refer to Chart 1 or to the programming label for the switch positions for the desired DTMF digits. Program the first DTMF digit on the first four switches of S6 and the second DTMF digit on the last four switches of S6. The only difference between autodial programming and access/reset programming is that 0000 programming on S6 is DTMF character "D" while 0000 programming on

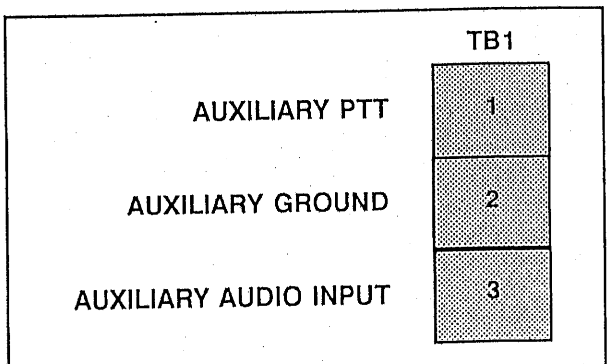


Figure 3. C-550 console terminals for auxiliary connections.

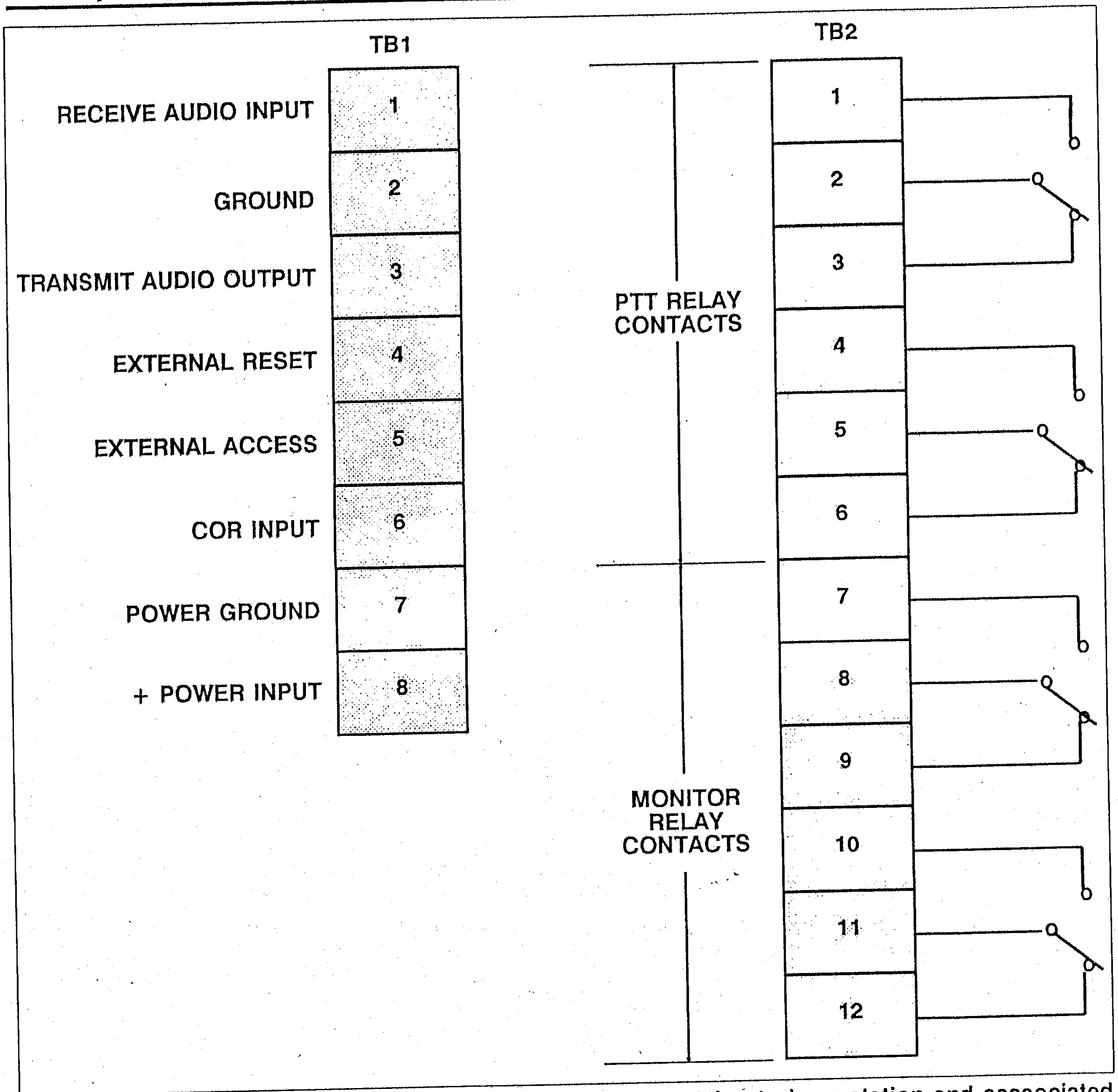


Figure 4. RP-250 remote-station-adapter terminals for connecting to base station and asssociated equipment.

S1 through S5 is DTMF off. The three-digit access and/or reset codes may be part of a longer string of DTMF digits if desired. For example, the access code may be added to a DTMF ANI message and still be decoded. If automatic DTMF message generators are used instead of a manual DTMF microphone, the digit rate must be 10 digits/s or less.

RP-250 Adjustments

Only the RX audio input and the TX audio output require adjustment upon installation. These controls may be adjusted from the front of the panel by removing only the top cover, but must be turned counter-clockwise to increase level when adjusted from this direction. A nonmetallic or insulated screwdriver should be used to avoid accidental short circuits. The TX output control R53 should be adjusted for proper transmitter deviation with voice audio rather than a sinewave, because the TX compressor maintains its output at an average power level and, for a given power level, typical voice peak-to-peak is 2.4 times that of a sinewave. The compressor between the RX input and the telephone line is of the same type as the TX compressor, and RX level should also be adjusted using typical voice audio. Adjust R51 for about 1 $V_{p\text{-}p}$ of typical voice audio at TP1.

RP-250 Solder-Bridge Programming

As shipped, upon ring detection, the RP-250 panel connects base-station audio, if any, to the telephone line immediately after the answer beep.

The C-550 console operator therefore is not required to operate any controls to monitor base-station audio for activity. However, if the base-station telephone number is listed or has become common knowledge, anyone dialing the base-station number from anywhere can listen to base-station audio, if any, for up to 40 seconds and tie up the base-station telephone line. To change RP-250 operation so that a monitor or PTT tone burst from a C-550 console is required before base-station audio is connected to the telephone line, open J1 and J8 and close J2.

Sensitivity of the RP-250 as shipped is for typical C-550 to RP-250 telephone-line losses of from 7 to 17 dB. If higher telephone-line losses are typical, close J6. Do not close J6 unnecessarily, because the increased sensitivity also increases sensitivity to line noise.

J4 has been provided to defeat the auto-answer function, which is necessary only on certain multiple RP-250 installations on a PSTN line.

C-550 Console Theory of Operation

C-550 Autodialer—General

Upon lifting the handset from the cradle (going off-hook), telephone-company (telco) line is connected to transformer T1 by the hookswitch, causing direct-current flow (loop current) through the telco line. Telco senses the loop current and switches a dial-tone generator to the line.

The enabled dial-tone detector U20 detects the dial-tone signal and, after about two seconds of delay, enables the autodial circuit and disables the dial-tone detector by resetting the dial-tone latch. The autodial circuit scans the programming field, and, when programming is detected at any digit position, the programmed DTMF digit is generated by U18 and outputed to the telco line. Telco detects DTMF, disconnects the dial-tone generator, decodes the autodialed number, and applies a ringing voltage to the dialed number at the base station and a ringback signal to the C-550 console. Upon detecting loop current at the base station (the RP-250 has gone "off hook"), telco disconnects the ringing voltage and the ringback signal generator and provides a two-way audio signal path between the RP-250 and the C-550.

Autodial Logic

The dial-tone latch U19-3,4,9,8 is set through C48 and CR9 from one set of hookswitch contacts at J7,8 whenever the handset is placed on hook. Upon going off hook to originate a call to the base station, the dial-tone detector is enabled at U20-3 by the set dial-tone latch. Dial-tone audio signal is conducted to the dial-tone detector audio input at U20-5 through J2-2, J10, J9, T1, R35, U5-14, R8, U1-14, C29, C20, R22, U2-10, R76, and C53.

U20 detects the dial-tone signal and V20-4 goes high and, in about two seconds, charges C50 to the threshold of U19-1, which causes U19-6 to go high and enable the autodial sequence.

At idle, the autodial counter U17 is latched at the last autodial position plus one, causing U17-9 to be high. When the autodial circuit is enabled by a high at U19-1, both U16 and U17 counters are reset to their zero count positions, causing U17-9 to go low. The low at U17-9 resets the autodial latch through CR8, which then disables the dial-tone detector at U20-3. Capacitor C50 is quickly discharged through CR10 by the low at U20-4, which then removes the reset logic at U16-15 and U17-15 through U19-1,2,5,6.

When U17-9 went low at the start of the autodial sequence, U14-13 went high through R69 and U19-11,10, enabling the clock-pulse generator U14-11, R63, and C47. The high at U14-13 causes U14-11 to go low because U14-12 is high from a fully charged C47. U14-11 remains low for about 150 ms until C47 is discharged by R63 to the threshold of U14-12. U14-11 then goes high and, due to the Schmitt input hysteresis characteristics, now oscillates at about 7 Hz. The first low-to-high transition at U14-11 clocks the high at U16-3 to U16-2, which, if J3 is closed, causes U18-9 to go high through CR5.

A high at U18-9,10,11 or 12 (data inputs) also causes a high at U14-1 through U15-13, and, if U14-2 is also high from the clock being high, U14-3 goes low, U13-4 goes high, and U18-2 goes high, enabling DTMF output at U18-14. The DTMF digit generated is determined by the data at U18-9,10,11,12. In the above case where only U18-9 is high, DTMF "1" is generated.

The clock output at U14-11 remains high for about 70 ms and then goes low for about 70 ms, disabling U18 at pin 2 through U14-3 and U13-3,4. U14-11 goes high again, clocking U16 from pin 2 to pin 4. Pin 4 has no circuit connections and is a 140-ms "pause" position of the counter. The low-to-high transition at U14-11 clocks U16 from pin 4 to pin 7, which provides data to U18 if any of the S1-1,2,3,4 switches are on, and enables DTMF output for the 70-ms period when the clock is high.

The process continues, with another pause at counter position 6, until the counter is clocked to position 7 (pin 6). A high at U16-6 is directly connected to U16-13, which inhibits U16's clock input. U16 is now latched at position 7 (pin 6), but the U16-6 high has also enabled U17 counting at U17-13 through U14-8,10. The clock-line high and the high-to-low transition at U17-13 clocks U17 from pin 3 to pin 2, enabling DTMF if any of S3-1,2,3,4 switches are on.

The sequence continues until U17 is clocked to pin 9 (position 8), which disables the clock at U14-13 through R69 and U19-11,10. Autodialing stops and the counter is latched as it was before autodial trigger.

The U18 to line audio path is from U18-14 through R67, C38, U8-2, U8-1, R39, T1, and the hookswitch contacts.

When U18 is enabled, the average DC level at U18-14 would normally step from 0 to 4.5 volts, but reverse DC logic and R68 maintains a near-constant DC average voltage at the junction of R67 and R68.

C-550 Transmit Logic and Tone Bursts

The 3.58-Mhz crystal at U20-1,2 is oscillating continuously. This frequency is applied to U11-9 through U19-13,12. U11 is a programmable divider wired to divide by 206, causing the output at U11-1 to be 17.376 kHz. U7 is another programmable divider used as a 16-step-per-cycle sinewave synthesizer. U7 also divides by 8 in the process, causing output at C36 to be 2172 Hz if the preset enable at U7-10 is low. (At idle, U7-10 is high, presetting U7-6 high and U7-4 low, so that the bias applied to U5-6 is 4.5 V). C37, U5-6,7, and associated components comprise an active low-pass filter which attenuates the steps on the synthesized waveform to a low-distortion sinewave at the U5-7 filter output.

Depression of the PTT switch triggers timer U6A at U6-5 through the debounce network R44 and C35, and U10-12,11. Once triggered, the timer output at U6-7 will go low for about 75 ms, enabling the sinewave synthesizer at U7-10 through U10-1,3,6,4. The analog gate U3-3,4 is enabled to conduct at U3-5 through U10-1,3 and U13-11,10. The 75-ms 2172-Hz audio signal path is from U5-7, through U3-3,4, through R33, U8-2,1, and through R39 and T1 to the line at a level of about –4 dBm.

Upon timeout of U6A, U6-7 goes high, disabling analog gate U3-3,4 and triggering timer U6B at U6-12. U6-10 goes high for about 47 ms, enabling DTMF character "C" generation from U18 through DN1. Upon U6B timeout and for as long as the PTT switch remains depressed, the sinewave synthesizer remains enabled at U7-10 through U10-5,4. The 2172-Hz sinewave path is now through R19, which causes the line output level to be –29 dBm.

Tone-burst generation from monitor-switch operation is the same as from PTT operation, except that U9A and U9B are used for the timers and U9B enables the DTMF character "B" through DN2.

U12 is used to trigger a –4-dBm, 75-ms, 2172-Hz tone burst every 25 seconds after off-hook or after PTT-switch operation. During on-hook, U14-4 is high due to the low at U14-6. Likewise, during PTT, U14-4 is high due to the low at U14-5. The high at U14-4 causes a low at U12-3,13 through U13-13,12, holding the timers in a reset condition. Upon off-hook or termination of PTT, U14-4 goes low, causing U12-3,13 to go high, removing timer reset, and U10-9 to go high, which causes U10-10 to go high, because U10-8 is also high from the untriggered Q of U12B at U12-9. The high at U10-10 triggers the 25-second timer U12A at U12-4 through R54, which pulse-discharges C41 and causes timer Q output at U12-7 to go low. C41 slowly charges through R58, reaching the threshold point in about 25 seconds, causing timer timeout.

Upon U12A timeout, U12-7 goes high, triggering timer U12B at U12-12, causing U12-9, U10-8, U9-3,13, U6-13, and U10-13 to go low for 100 ms. Timers U9A, U9B, and U6B are held in reset for 100 ms, but timer U6A is triggered by the high-to-low transition at U10-13. A 75-ms pulse from U6A causes –4-dBm, 2172-Hz line output in the same manner as from PTT operation, but U6B is held in reset upon U6A timeout and, therefore, is not triggered. U12A is triggered again upon timeout of the U12B 100-ms timer through U10-8,10 and R54. The net result is a 75-ms, –4-dBm, 2172-Hz line output every 25 seconds. This tone burst is used by the RP-250 at the base station to refresh its 40-second timer and thereby prevent timeout up to indefinitely.

C-550 Transmit Voice and Auxiliary Audio

Voice signals from the handset microphone and auxiliary audio from TB1-3 are applied to audio buffer U5-3,2,1, which drives the transmit compressor at U2-6 through C21, C16, and R10. The maximum gain of this compressor is set by R11, which causes C17 to become slightly charged, which then controls the amount of internal inverse feedback. Audio signal present at U2-2 is internally full-wave rectified and applied to C17 whenever the rectified signal voltage exceeds the existing bias caused by R11. Any increase in voltage at C17 causes a decrease in the compressor gain and therefore maintains a near-constant output at U2-7 for input levels above threshold. RN1-1,2,3,4 and C11 provide a DC but not audio-frequency feedback path which maintains the average DC voltage at U2-7.

During PTT, U3-1,2 is enabled at U3-13 by the lows at U10-12,11 through R64, U13-5,6, and R61. The audio signal path is then from U2-7 through C12, U3-1,2, R32, U5-9,8, R38, U8-2,1, R39, T1, and the hookswitch contacts to the line at a level of about –10 dBm.

Another transmit audio signal path exists from U5-8 through R40 to the junction of C7 and R50. This path provides transmit audio at attenuated volume-control-adjustable level to the handset earpiece through R50, U8-6,7, C23, and R24. This sidetone in the earpiece makes talking into the handset microphone seem more natural (because standard telephones have sidetone, and lack of sidetone might appear that power is off and no transmission is occurring).

Receive Audio

The incoming audio path is through the hookswitch contacts, T1, R35, to U5-13,14. From U5-14, two signal paths exist to U1-13—a direct path through R8 and another path through R9 to the U1 tunable 2172-Hz bandpass filter, R6, and R7. R12 and R6 are factory-adjusted to cause 180° and equal-amplitude 2172-Hz audio at U1-14. The result is a 45-dB or greater notch at 2172 Hz in the audio at U1-14. This eliminates the annoying low-level 2172-Hz PTT tone that would otherwise be present when a parallel C-550 console is transmitting.

U3-6,9 and R18 shunts R16 in the U1-13,14 feedback loop. This shunt greatly attenuates the receive audio path during PTT, autodial, and tone-burst generation. PTT enables U3-6 (which almost disables the receive audio path) through R44, U10-12,11, R31, and Q1. U3-6 is also enabled by a high at U15-1, caused by guard tone, autodial enable, 100-ms timer U12B, and function-tone DTMF enable.

Audio signals from U1-14 are applied to the receive compressor at U2-11 through C29, C20, and R22. Operation of the receive compressor is identical to that of the transmit compressor, and a near-constant audio output signal is developed at U2-10 at all input signal levels above threshold.

The audio signal path from U2-10 is through C8, the front-panel volume control R101, C7, R50, U8-6,7, C23, and R24 to the handset earpiece.

U3-10,11 blocks the audio path to the speaker unless the speaker latch U13-1,2,9,8 has been set by speaker-switch operation, which enables the U3-10,11 path at U3-12. The speaker latch is reset by PTT or on-hook through U14-5,6,4, U13-13,12, and CR-1.

When the speaker latch is set, the audio path to the speaker is through R17, U3-10,11, C6, U4-2,8, C26, and E1.

Ring Detector

Ringing voltage from the telco line is applied to ringer IC U23, which, upon ringing-voltage detection, outputs ringing audio through C55 and R79 to the optical isolator at U22-1,2. Ringing audio signals are coupled to U22-5, through R84, C62, and C6 to the input of power-amplifier U4, and then to the speaker. Ringing tone to the speaker path is unaffected by the volume control.

Power Supply

The external wall power supply connects to J5, supplying unregulated voltage through CR13 and R81 to power-amplifier U4. Regulated 9-V power for everything else appears at the output of regulator U21. The 4.5-V reference voltage derived from the regulated 9 V appears at C3, to bias most of the opamps.

RP-250 Adapter Theory of Operation

General—Autodialer and Call Progress

Upon DTMF ACCESS decode or upon a logic-low input to the external access terminal, the telco line is connected to transformer T1 by relay contacts, causing direct current flow (loop current) through the telco line. Telco senses the direct current flow and switches a dial-tone generator to the line.

The enabled call-progress detector U27 detects the dial-tone signal and, after about 2 seconds of delay, triggers the autodial circuit, which then disables the call-progress detector. The autodial circuit scans the programming field and, when programming is detected at any digit position, the programmed DTMF digit is generated by U19 and applied to the telco line. Telco detects the DTMF and disconnects the dial-tone generator, decodes the autodialed number, applies a ringing voltage to the dialed number at the C-550 console, and applies a ringback signal to the RP-250 line.

The RP-250 call-progress detector, which was enabled at autodial completion, detects the ringback signal, and switches the base station to transmit, modulated by a tone signal which lasts for the duration of the ringback signal on the telco line.

When the C-550 operator goes off-hook, telco detects the loop current, disconnects the ringing voltage and ringback signal, and provides a two-way audio signal path between the C-550 and the RP-250.

RP-250 Autodialer and Call-Progress Logic

Upon DTMF ACCESS decode or upon external access input at TB1-5, the 40-second timer U17 is triggered at U17-5, and the MOBILE ACCESS latch U6 is set at U6-15. U17-6 goes high, energizing line-relay K1 through U12-5,12, which connects the telco line to transformer T1. Setting ACCESS latch U6 causes U6-1 to go high, which disables RX audio at U24-5 through U33-8,9 and U32-6,4. The high at U6-1 also enables the call-progress detector at U27-3 through U35-12,11, U28-5,6, and R84. Dial-tone audio signal is applied to U27 through T1, R82, and C58.

Detection of the dial-tone signal causes U27-4 to go high, Q4-D to go low, Q3 to go open-circuit, and C44 to charge through R73. C44 charges to

Information Required To Be Supplied to the Customer

User's Responsibility

The user is required to notify the telephone company of the connection or disconnection of the device, the make, model number, FCC registration number, and ringer equivalence and the particular line to which it is to be made. If the proper jack(s) are not available, the user must order the type of jack(s) to be used from the telephone company.

Information for the Remote Console

Manufacturer: VEGA, A MARK IV COMPANY
Model Number: C-550
Registration Number: BFD9EG-19272-OT-E
Ringer Equivalence: 1.0 B
Jack(s) Which May Be Used: RJ11C or RJ11W

Information for the Station Panel

Manufacturer: VEGA, A MARK IV COMPANY
Model Number: RP-250
Registration Number: BFD9EG-19271-OT-E
Ringer Equivalence: 0.41 B
Jack(s) Which May Be Used: RJ11C or RJ11W

It is prohibited to connect this equipment to pay-telephone or party lines.

Telephone Company Rights and Responsibilities

Under certain circumstances, the telephone company may discontinue service if the device causes harm to the telephone network. In this case, the telephone company shall:

1. Promptly notify the customer of discontinuance.
2. Afford the customer the opportunity to correct the situation which caused discontinuance.
3. Inform the customer of his rights to bring a complaint to the FCC concerning the discontinuance.

The telephone company may make changes in its facilities and services which may affect the operation of the user's equipment. However, the user shall be given adequate notice in writing to allow the user to maintain uninterrupted service.

In case of trouble with this unit, return the unit to the manufacturer (Vega) for repair, or have the manufacturer or his representative repair it in place. Do not attempt to repair the unit as this will violate the FCC Rules and may cause danger to persons or to the telephone network.

FCC Requirements

This equipment complies with Part 68 of the FCC Rules. On the bottom of the C-550 console and on the back of the RP-250 station panel is a label that contains, among other information, the FCC Registration Number and Ringer Equivalence Number (REN) for this equipment. If requested, this information must be given to the telephone company.

The REN is useful to determine the quantity of devices you may connect to your telephone line and still have all of those devices ring when your telephone number is called. In most, but not all, areas, the sum of the RENs of all devices connected to one line should not exceed 5.0. To be certain of the number of devices you may connect to your line, as determined by the REN, you should contact your local telephone company to determine the maximum REN for your calling area.

If your telephone equipment causes harm to the telephone network, the telephone company may discontinue your service temporarily. If possible, they will notify you in advance. But if advance notice isn't practical, you will be notified as soon as possible. You will be informed of your right to file a complaint with the FCC.

Your telephone company may make changes in its facilities, equipment, operations, or procedures that could affect the proper functioning of your equipment. If they do, you will be notified in advance to give you an opportunity to maintain uninterrupted telephone service.

If you experience trouble with this telephone equipment, please contact your telephone company for information on obtaining service or repairs. The telephone company may ask that you disconnect this equipment from the network until the problem has been corrected or until you are sure that the equipment is not malfunctioning.

This equipment may not be used on coin service provided by the telephone company. Connection to party lines is subject to state tariffs.

C-550 CONSOLE SPECIFICATIONS

Input and Output Impedance: 600 Ω nominal

Line Output Level: Less than −9 dBm average in any 3-s time interval

PTT Tone-Burst Format

- **Guard Tone:** 75 ms ± 20%, 2174 Hz ± 0.02%, −4 dBm ± 1 dB
- **Function Tone:** 47 ms ± 20%, −4 dBm ± 1 dB
- **PTT Tone:** Continuous 2174 Hz ± 0.02%, −29 dBm ± 1 dB

Monitor Tone-Burst Format

- **Guard Tone:** 75 ms ± 20%, 2174 Hz ± 0.02%, −4 dBm ± 1 dB
- **Function Tone:** 47 ms ± 20%, −4 dBm ± 1 dB

Line Output Level from Handset Microphone or Auxiliary Input: −10.3 dBm ± 1 dB (about 2.2 $V_{p\text{-}p}$ from typical voice)

Autodial

- **Digits:** 1 to 11, 10 DIP-switch programmable
- **Digit Rate:** 7 digits/s nominal
- **DTMF Duty Factor:** 50% nominal
- **DTMF Output Level:** −4 dBm ± 1 dB

Audio Compression, Receive, and Transmit Audio: Less than 3 dB change in output for 20 dB change in input above threshold

Distortion: 2% THD maximum at full compression

Audio Frequency Response: ±1.5 dB, 300 to 3000 Hz, except at transmit notch frequency

Hum and Noise: 50 dB below operating signal levels, minimum

Speaker: 4 in, 8 Ω, heavy-duty

Amplifier Power: 800 mW at 10% THD

Handset Earpiece Level: Volume-control adjustable

Sidetone Level: About 25 dB below receive level

Telephone Line Interface: Modular cord

Power Requirements: 105 to 130 V_{ac}, 60 Hz, 8 W

Operating Temperature Range: 0 to +50°C

Notch Filter: 45 dB typical attenuation of a parallel console's PTT tone

Miscellaneous: Crystal-controlled tone frequencies; electret handset microphone element; auxiliary audio and PTT input terminals; speaker mute upon PTT

RP-250 PANEL SPECIFICATIONS

Line Input and Output Impedance: 600 Ω nominal

Receive Audio Input Impedance: 15 kΩ nominal

Transmit Audio Output Impedance

- **Receive or Idle:** 22 kΩ typical
- **During Transmit:** 250 to 500 Ω

Receive Input Sensitivity at Threshold of Compression: 280 mV_{rms} to 6 V_{rms}, adjustable

Transmit Output Level: −40 to −6 dBm (50 $mV_{p\text{-}p}$ to 2.6 $V_{p\text{-}p}$ typical voice) into a 600-Ω load, adjustable. Tone output levels are 6 dB below voice

Telephone Line Loss

- **Low Sensitivity:** 7 dB minimum to 17 dB maximum (as shipped)
- **High Sensitivity:** 17 dB minimum to 27 dB maximum, solder-bridge programmable

S/N Tolerance

- **DTMF Access, Reset, Function Tone Decode:** 10 dB

Line-Noise Tolerance, PTT or Monitor Decode: −22 dBm at 7 dB line loss to −42 dBm at 27 dB line loss

Line-Noise Tolerance, False TX Hold of 800 ms or Longer, 1 in 10,000 Probability

- **Low Sensitivity:** −36 dBm, 3 kHz bandwidth white noise
- **High Sensitivity:** −46 dBm, 3 kHz bandwidth white noise

RX Input Noise Tolerance

- **False DTMF Access or Reset Decode:** Less than one false in 1000 hours of typical radio audio estimated
- **False TX Initiate:** 0 dBm, 3 kHz white noise, 1 in 10,000 probability with properly adjusted RX input control

Receive and Transmit Audio Compression: Less than 3 dB change in output for 20 dB change in input above threshold

Distortion: 2% THD maximum at full compression

Audio Frequency Response: ±1.5 dB, 300 to 3000 Hz except at transmit tone notch frequency

Hum and Noise: 60 dB below operating levels, minimum

Autodial

- **Digits:** 1 to 11, 10 DIP-switch programmable
- **Digit Rate:** 7 digits/s nominal
- **DTMF Duty Factor:** 50% nominal
- **DTMF Output Level:** −4 dBm ± 1 dB

Line Output Level

- **Voice:** −10 dBm ± 1 dB (about 2.2 $V_{p\text{-}p}$ typical voice)
- **Tones to Line:** −16 dBm ± 1 dB
- **Autodial DTMF:** −4 dBm ± 1 dB

Logic Hold after Power Loss: 200 ms minimum

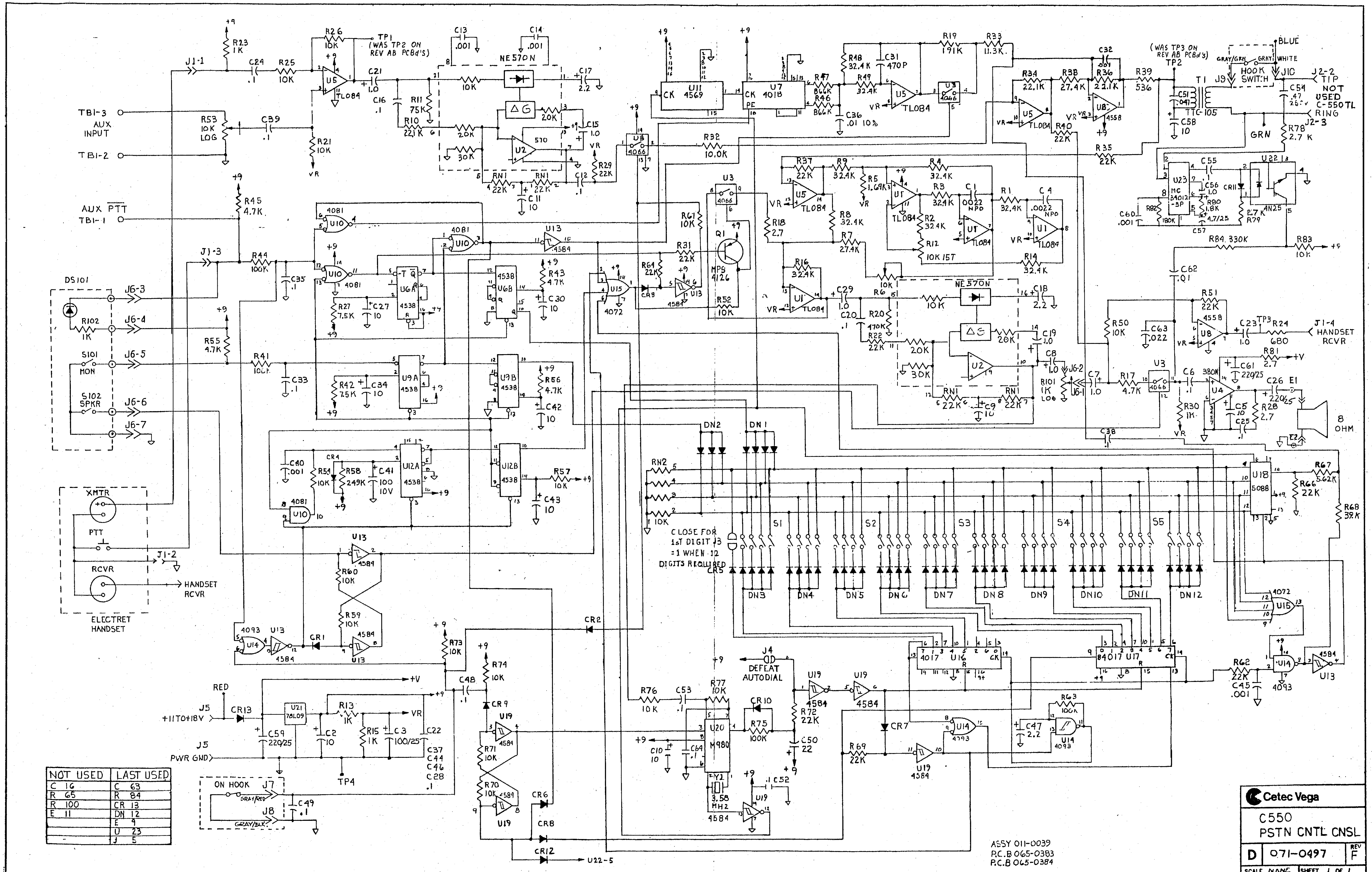
Telephone Line Interface: Modular cord

Power Requirements: 11.5 to 16.5 V_{dc}, semi-regulated; 140 mA maximum at 12 V_{dc} input; 100 mA maximum at idle and 12 V_{dc} input, factory programming; 70 mA maximum at idle, 12 V_{dc} input and alternate-mode programming; provision for optional 105 to 130 V_{ac}, 8 W wall power supply

Operating Temperature Range: −30 to +60°C

Size: 16.6 W, 6.56 D, 1.72 H inches; 42.2 W, 16.7 D, 4.37 H centimeters

Miscellaneous: Crystal-controlled tone frequencies and decoders; inhibit PTT/COR input terminals; external access/reset decoder input terminals



NOT USED	LAST USED
C 16	C 63
R 65	R 84
R 100	CR 13
E 11	DN 12
	E 4
	U 23
	J 5

Cetec Vega

C550
PSTN CNTL CNSL

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SCALE NONE SHEET 1 OF 1