

MATRIX POTS/ISDN/WIRELESS CODEC

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SECTION 1.**INTRODUCTION**

The Comrex Matrix in its basic form is a digital audio transmission device for analog telephone lines. The Matrix allows wideband, broadcast quality audio over normal “toll quality” telephone circuits. Here’s a short list of the Matrix’s POTS Codec features:

MATRIX FEATURES

- ◆ Full duplex FM quality (15 kHz) audio bandwidth at connect rates of 24 kb/s and above.
- ◆ Full duplex audio scales from 12.7 kHz to 4.7 kHz at connect rates of 21.6 kb/s to 9.6 kb/s.
- ◆ Separate “Voice Mode” allows 7 kHz audio with a 300 BAUD ancillary data channel and extra forward error correction.
- ◆ Easy to use dialing and configuring interface.
- ◆ Available in portable or rackmount chassis.
- ◆ Automatically connects at highest possible rate supported by telephone connection.
- ◆ Monitors line quality and renegotiates should line degrade.
- ◆ Ability to “store and forward” 9 ³/₄ minutes of audio. A full fidelity feed may be recorded and sent over a low bitrate circuit to be played later in non-real time.
- ◆ Two tally closures (one on connection, one triggered by the user).
- ◆ Ability to work with a Comrex BlueBox, HotLine or Vector.
- ◆ Ability to store 19 telephone numbers and last number dialed in nonvolatile memory.
- ◆ Short coding delay (100 mS) allows seamless two-way conversation.
- ◆ Accessory slot allows insertion of optional modules, adapting the Matrix for use on other circuits such as ISDN and GSM.
- ◆ Optional battery runs Matrix for up to 7 hours.

PORTABLE UNIT INCLUDES

Two channel microphone mixer (1 channel mic/line selectable).
 Additional fixed level tape input.
 One headphone output (adjustable mix of send/receive audio).
 One balanced line level audio out (adjustable mix of send/receive audio).
 Rugged, lightweight plastic chassis with built-in handle.
 Accessory slot allows insertion of optional modules, adapting the Matrix for use on other circuits such as ISDN, GSM and allowing plain telephone calls on POTS lines or wireless phones equipped with headset jacks.

RACKMOUNT UNIT INCLUDES

Space efficient 1U rack.
 Internal universal AC power supply.
 Balanced line level audio input/output.
 Ability to dial on contact closure.
 Ability to remotely control contact closure.
 Accessory slot for ISDN adapter board
Future upgrade: Remote control of Matrix functions through TCP/IP protocol.

ABOUT THE ALGORITHM

The Matrix utilizes audio coding which is uniquely suited to the task of moving high quality audio along the narrow data channels provided by analog modems. The Matrix algorithm draws on principles from various previous coding methods yet doesn't incorporate the long processing delays (which equates to audio delay) that plague other coding schemes. While the Matrix algorithm does not provide true "transparency" to the reconstructed audio, much work has been done to place the artifacts that do exist in a range where they will be less irritating to the listeners.

*WHAT COMES WITH
A MATRIX?*

The following items are shipped with a new Matrix:

- (1) Comrex Matrix POTS codec (Portable or Rackmount)
- (1) AC Power module with AC cord (Rackmount unit has only AC cord)
- (1) RJ11C modular telephone cord
- (1) Operating manual
- (1) Warranty card (Please fill out and return)

UPGRADES

A substantial portion of the functions and features of the Matrix is software driven, allowing upgrades to these units by a simple exchange of EPROMs or reprogramming of existing EPROMs. Many software updates may be downloaded from the Comrex website and performed by connecting your Matrix to a com port on your PC. (See Section 11 for instructions.) We will keep our customers apprised of software updates and make these available at no charge. In order to assure that we will be able to contact you, please return the warranty card with complete address and contact information.

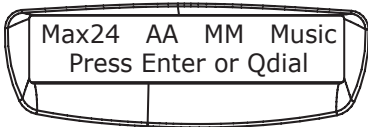
*OPERATIONAL
CONSIDERATIONS*

For use on POTS (Plain Old Telephone Service), two Comrex POTS codecs are required — one at each end of the telephone line. (Any combination of Matrix, BlueBox, HotLine or Vector will work.) POTS codecs require moderately high quality telephone lines on each end of the call and may have performance issues if the telephone connection has excessive noise, hum or crosstalk. As a general rule, avoid connecting through in-house PABX systems. A direct, outside line should be used for best performance. For further details on telephone quality issues, we suggest you read the "Modem Line Checklist" in Appendix A on page 72. Also, because the Matrix's algorithm creates a small coding delay, it is necessary to send a "Mix Minus" feed from the studio back to the remote site. For more on Mix-Minus, refer to Appendix B on page 77.

For wireless circuits the Matrix has two options. The first is the ability to improve the quality of "real-time" audio on GSM wireless service, please refer to Section 8. Matrix GSM Wireless Operation on pages 44-46. Secondly, the portable Matrix can send "unenhanced" audio through the headset jack of a wireless phone using the optional TelCell module. (See pages 32-34.)

SECTION 2. MATRIX MENU SELECTION TREE

MATRIX MAIN STATUS DISPLAY



Enter	Dial	Keypad	Dial Number	
		Ext Phone	Dial w/Ext Phone--<Enter>when Ringing ...connecting	
		Tel/Cell Module*	Phone Line--Phone Mode Active Enter = Off Hook	
			Cell Phone--Dial From Cellphone Cancel = Exit	
	Answ connecting.....			
	Test	Voice	33.6 - 9.6 kb/s	Loopback active
		Music	33.6 - 9.6 kb/s	Loopback active
		HotLine	33.6 - 12 kb/s	Loopback active
		GSM	9.6 or 14.4 kb/s	Loopback active
	Config	Pgm Qdial	Enter Qdial#: 01-19 Enter Phone Number	
		Codec	Voice Mode	Algorithm Selected
			Music Mode	Algorithm Selected
			GSM Mode	Algorithm Selected
		Max Rate	33.6 - 9.6 kb/s	Maximum Rate Stored
		More	Auto Ans	On
				Off
				6 Rings
			Dialing	Tone Dialing
				Pulse Dialing
			ModMon	Modem Monitor On
				Modem Monitor Off
		Levels		
		(Rack Matrix Only)	Input Level	0dBu, +4dBu, -10dBu
			Output Level	0dBu, +4dBu, -10dBu
	Store & Forward (See Page 9 for Menu Selection Tree)			
Qdial	Enter Qdial#: 01-19 (Note: For last number redial, enter 00)			

*If Tel/Cell Module does not appear, please refer to page 32 for details.

STORE & FORWARD MENU SELECTION TREE

Enter	<u>Dial No Audio Stored (if no audio cut is in memory)</u>		
	or	<u>Keypad ... Dial Number: Dialing</u>	
		<u>Ext Phone ... Dial with Ext. Phone <Enter> when Ringing</u>	
	Rec	<u>Erase All Audio Data 1=Yes Cancel=No (if audio cut is in memory)</u>	
	or	<u>Enter = Start Record Qdial = Pause</u>	
	Play	<u>No Audio Stored (if no audio cut is in memory)</u>	
	or	<u>Playing 0:01</u>	
	Setup	<u>Play/Rec</u>	<u>Play S/F Audio</u>
			<u>Rec S/F Audio</u>
		<u>Send/Rcv</u>	<u>Receive S/F Audio</u>
			<u>Send S/F Audio</u>
	<u>Exit S/F ... reverts back to Main POTS Status Display</u>		
<u>Qdial ... Enter Qdial #:00-19</u>			

PRIMARY AND SECONDARY USES OF FUNCTION KEYS.**HANGUP**

Disconnects call.

Secondary functions:

1. Backs out of any screens involving dialing functions.
2. Accesses the *HIDDEN MENU* (press key 3 times).

QDIALSpeed dials numbers prestored in memory, including *QDIAL 00* to redial the last number dialed.

Secondary functions:

1. If pressed once when line is connected, modem drops one connect speed.
2. If "double clicked" when line is connected, modem performs a complete renegotiation, without disconnecting the line.
3. Qdial=Pause in the Store & Forward mode during Record and Playback.
4. Qdial inserts two second pause in dialing string entries.

ENTER

Initiates functions via the menu screen.

Secondary function:

1. When line is connected, sends a momentary contact closure to far end.

CANCELBacks out of most menus. (see *HANGUP*)

Secondary function:

1. Backspaces during entry of dialing strings & programming of *QDIAL* numbers.

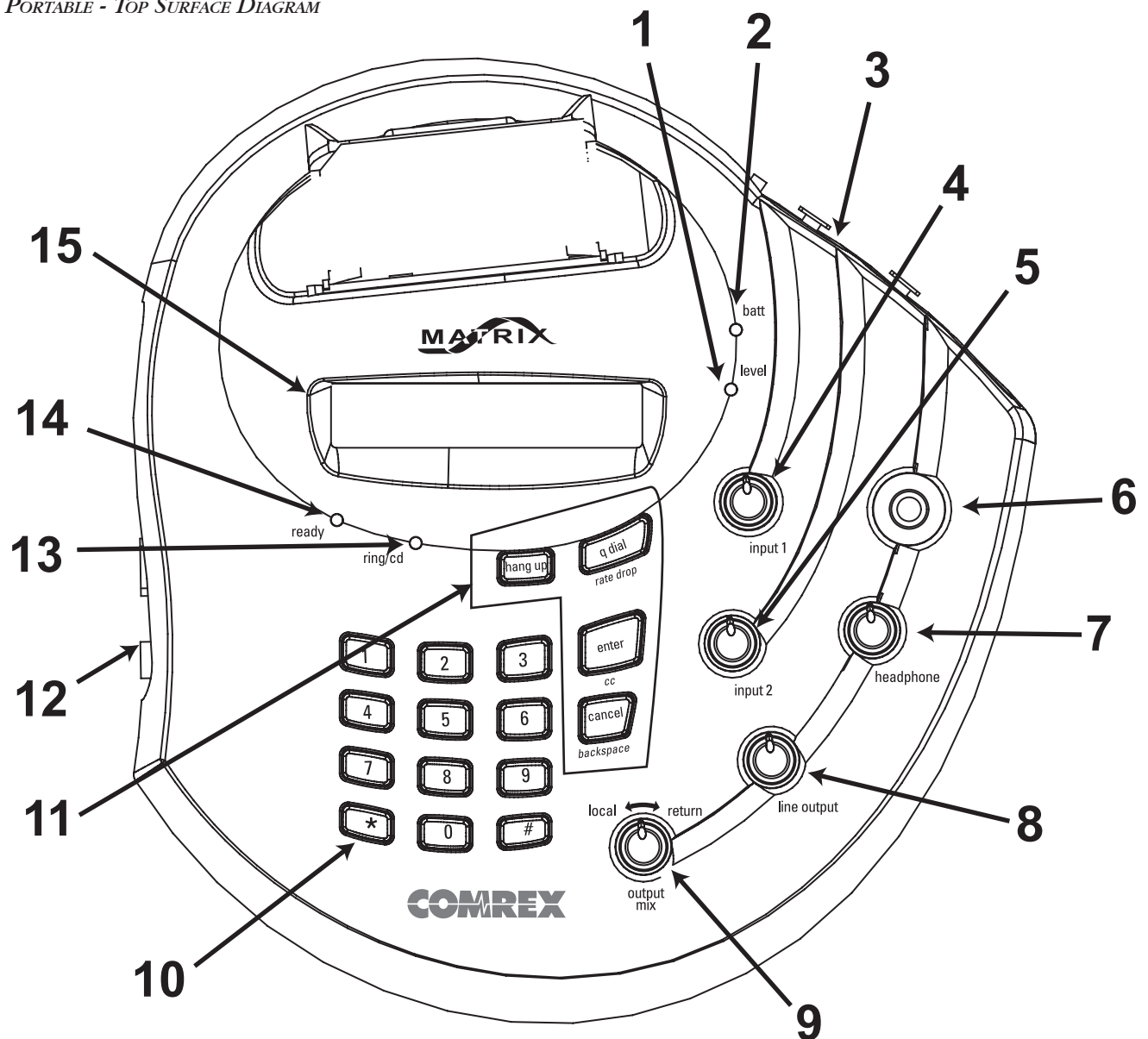
MATRIX "HIDDEN MENU"

(PRESS THE HANGUP KEY THREE TIMES FROM THE MAIN STATUS SCREEN)

Country	Country Code 00-20	
Reneg	Reneg/Retrain On	
	Retrain On	
	None	
Flush Modem	Erase All Settings? Enter = Continue	
More	ISDN (Rack Option Only)	Enable ISDN Menus? 1=Yes Cancel=No
	STL	1)Enable STL Mode
		2)Disable STL Mode
	Rev (Shows current software version)	
	Symbol Rate	1) A (Disable top rate)
		2) A&B (Disable top two rates)
		3) Enable ALL Rates
	DialTone	1) Require DialTone
		2) Ignore DialTone
	GSM14	1) Enable GSM 14.4
		2) Disable GSM 14.4

SECTION 3.

DIAGRAMS AND DESCRIPTIONS OF CONTROLS AND CONNECTORS

PORTABLE - TOP SURFACE DIAGRAM

1) BI-COLOR INPUT LEVEL INDICATOR

2) BATTERY FUNCTION LIGHT

3) REAR PANEL (SEE PAGE 14 FOR DETAILED VIEW)

4) INPUT LEVEL CONTROL 1

5) INPUT LEVEL CONTROL 2

6) HEADPHONE JACK

7) HEADPHONE LEVEL ADJUSTMENT

8) XLR LINE LEVEL OUTPUT ADJUSTMENT

9) OUTPUT MIX ADJUSTMENT

10) DIALING KEYPAD

11) FUNCTION KEYPAD

12) SIDE PANEL (SEE PAGE 15 FOR DETAILED VIEW)

13) RING/CARRIER DETECT INDICATOR

14) READY INDICATOR

15) BACKLIT LCD

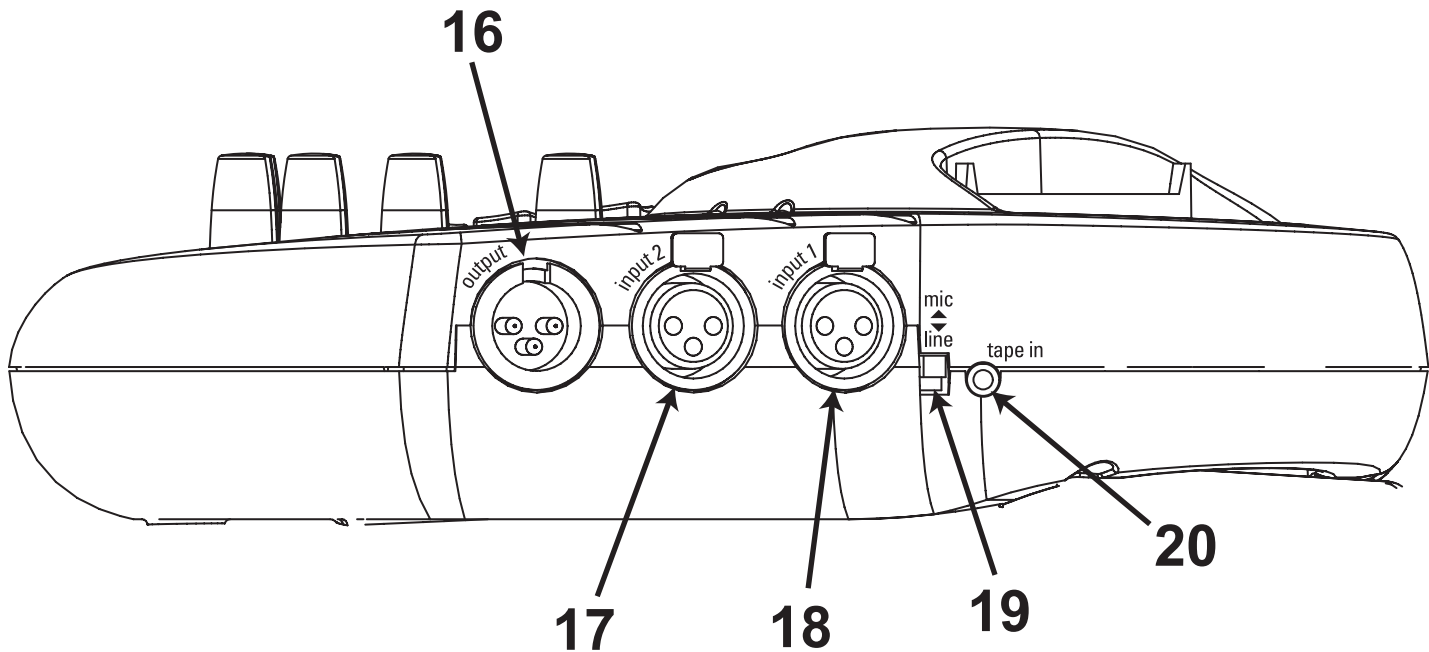
The rackmount and portable versions of the Matrix are described separately.

PORTABLE - DESCRIPTION OF TOP SURFACE

- | | | |
|----|---------------------------------------|---|
| 1 | <i>BI-COLOR INPUT LEVEL INDICATOR</i> | This provides an indication of the audio level being fed into the Matrix. It doesn't mirror the receive audio in any way. Adjust the input levels so that the green light is illuminated. This LED turns red to indicate when the limiter is engaged on program peaks. |
| 2 | <i>BATTERY FUNCTION LIGHT</i> | When this red LED is illuminated, the Matrix is operating on battery power. |
| 3 | <i>REAR PANEL</i> | Audio connectors are located on the rear panel. (See page 14 for details.) |
| 4 | <i>INPUT LEVEL CONTROL 1</i> | This knob controls the input audio level for audio channel 1. Adjust the level using this knob, making sure that the peak indicator light stays green, with only occasional red showing for program peaks. |
| 5 | <i>INPUT LEVEL CONTROL 2</i> | This knob controls the input audio level for audio channel 2. Adjust the level using this knob, making sure that the peak indicator light stays green, with only occasional red showing for program peaks. |
| 6 | <i>HEADPHONE JACK</i> | This is a 3-conductor 1/4 inch headphone jack suitable for most stereo headphones. |
| 7 | <i>HEADPHONE LEVEL ADJUSTMENT</i> | <p>This knob controls the headphone jack audio level. This includes all local and return audio. Adjust this knob until a comfortable listening level is achieved on the headphones.</p> <p><i>Note: A ratio of local and return audio levels may be set up using the Output Mix Adjustment. (See #9 below.)</i></p> |
| 8 | <i>XLR LINE LEVEL OUTPUT LEVEL</i> | <p>This knob controls the output audio level. This includes all local and return audio. Adjust this knob for desired output level.</p> <p><i>Note: A ratio of local and return audio levels may be set up using the Output Mix Adjustment. (See #9 below.)</i></p> |
| 9 | <i>OUTPUT MIX ADJUSTMENT</i> | This knob determines the mix/ratio of local versus return audio. Adjust this knob until a comfortable mix of local and return audio is achieved. Turning the knob counterclockwise gives more local audio. Turning the knob clockwise gives more return audio. |
| 10 | <i>DIALING KEYPAD</i> | Use this keypad to place calls, either in Matrix or POTS Mode. Numbers may also correspond to menu options. |
| 11 | <i>FUNCTION KEYPAD</i> | These keys provide "one touch" access for several major functions such as Qdial and hangup. Some of the keys also have secondary functions, noted under the key. |

- | | |
|--|---|
| <i>12 SIDE PANEL</i> | Power, phone lines and data connections are made on this panel. (See pages 15 & 16 for details.) |
| <i>13 RING/CARRIER DETECT (CD) INDICATOR</i> | When the Matrix receives an incoming call, this light will flash to indicate the line is ringing. When the modem inside the Matrix is connected to another compatible codec, this light will be on steadily. |
| <i>14 READY INDICATOR</i> | This is a signal from the digital audio decoder which signifies it has locked onto the incoming digital audio signal. This will usually light up a few seconds after connection and go out if a retrain is necessary or if the codec “loses sync” for any reason. A contact closure which follows this indication is available on the side panel. |
| <i>15 BACKLIT LCD</i> | Displays all status and menu options. |

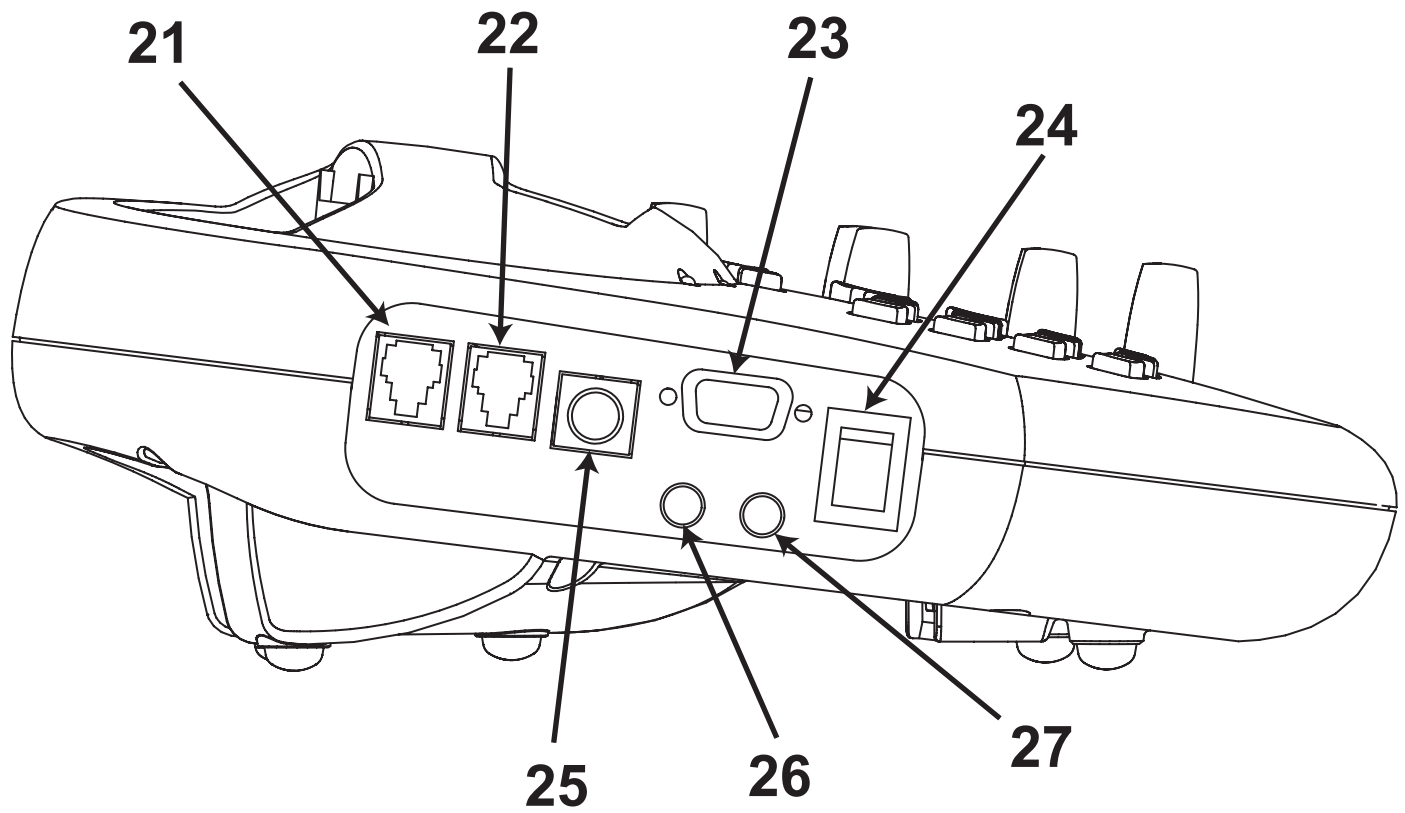
PORTABLE - REAR PANEL DIAGRAM & DESCRIPTION



- 16) *XLR LINE LEVEL OUTPUT*
- 17) *INPUT 2 MICROPHONE CONNECTOR*
- 18) *INPUT 1 Mic/LINE CONNECTOR*
- 19) *INPUT 1 Mic/LINE SWITCH*
- 20) *TAPE INPUT -10 dBu MINI JACK*

- 16 *XLR LINE LEVEL OUTPUT* This 3-pin male XLR output provides a mix of codec return and local audio from the Matrix. It is a balanced, line level audio feed, adjustable via the *LINE OUTPUT* and *OUTPUT MIX* knobs on the top surface, shown in #8 and #9 of the top surface diagram on page 11. This output is suitable for connection to a mixer, PA system or additional headphones.
- 17 *INPUT 2 MICROPHONE CONNECTOR* This 3-pin female XLR connector is designed to accept a balanced, microphone level audio feed. The input level for this channel is adjustable via the knob shown in #5 of the Top Surface diagram on page 11.
- 18 *INPUT 1 Mic/LINE CONNECTOR* This 3-pin female XLR connector is designed to accept a balanced, microphone or line level audio feed. This input level is adjustable via the top surface control shown in #4 of the Top Surface diagram. The level is selectable between nominal mic or line level. (See #19 below.)
- 19 *INPUT 1 Mic/LINE SWITCH* This switch determines whether audio input 1 is configured to receive a microphone level (-70 to -40 dBu) or line level (-10 to +10 dBu) audio input.
- 20 *TAPE INPUT* This 1/8 inch mini tape input is designed to accept an unbalanced, -10 dBu fixed audio feed, allowing for easy connection to a minidisc or DAT player.

PORTABLE - SIDE PANEL DIAGRAM

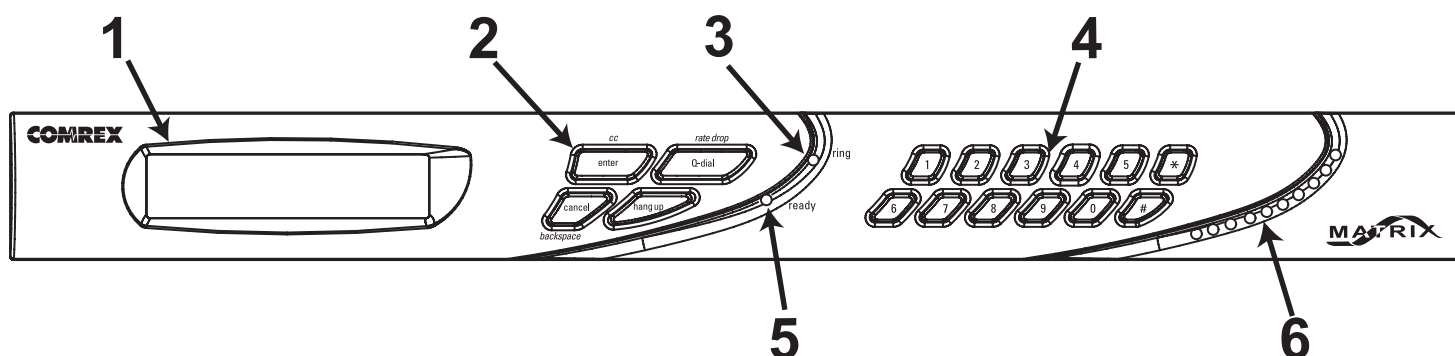


- 21) TEL LINE JACK*
- 22) TEL SET JACK*
- 23) MULTI-PURPOSE PORT*
- 24) POWER ON/OFF SWITCH*
- 25) POWER CONNECTOR*
- 26) CONTACT CLOSURE*
- 27) READY CLOSURE*

PORTABLE - DESCRIPTION OF SIDE PANEL

- | | | |
|----|---------------------------|---|
| 21 | <i>TEL LINE JACK</i> | This RJ11 jack is for attaching the telephone line from the outside world to the Matrix. Be careful what you plug into here. Some PABXs provide currents which can damage the Matrix. |
| 22 | <i>TEL SET JACK</i> | This RJ11 jack is for attaching an external telephone to the Matrix. This can be useful for dialing complicated numbers with lots of pauses and special codes. It can also serve to provide an audible “ring” when an incoming call is received. |
| 23 | <i>MULTIPURPOSE PORT</i> | Female DB-9 connector through which flash upgrades can be performed (see page 54 for more details). |
| 24 | <i>POWER SWITCH</i> | 0 = off, 1 = on. These markings are screened on the rocker switch. |
| 25 | <i>POWER CONNECTOR</i> | This is a 5-conductor DIN type connector designed to mate with the Matrix power supply module. Be sure to attach only the power supply provided from Comrex here. |
| 26 | <i>CC RECEIVE CLOSURE</i> | This mini jack provides a momentary dry contact closure which triggers closed when the Matrix is on-line and the far end sends a <i>CC</i> command via the keypad. |
| 27 | <i>READY CLOSURE</i> | This mini jack provides a latching dry contact closure when the digital audio decoder achieves synchronization with an incoming digital audio signal. This contact is closed when the green <i>READY</i> light below the LCD display is illuminated and is released when the <i>READY</i> light goes out. |

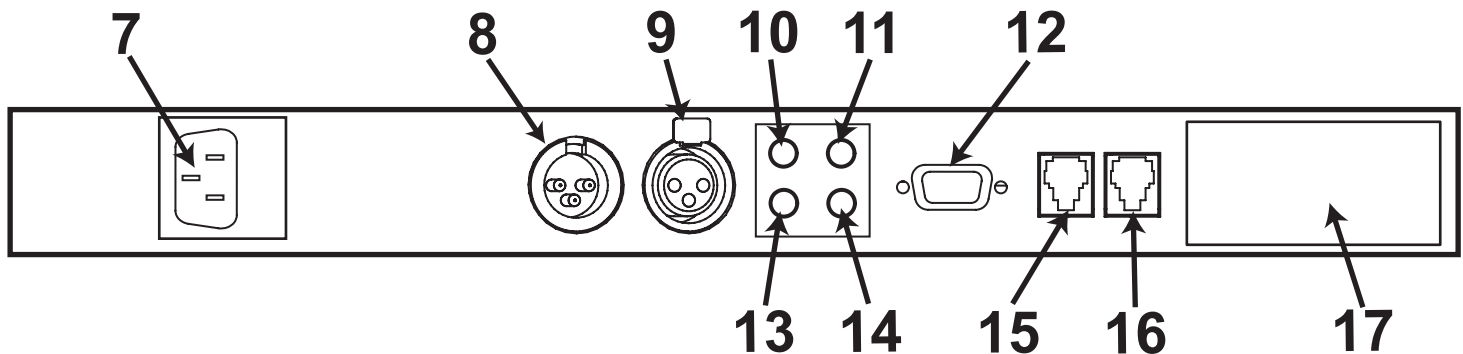
RACKMOUNT - FRONT PANEL DIAGRAM & DESCRIPTION



- 1) BACKLIT LCD
- 2) FUNCTION KEYPAD
- 3) RING/CARRIER DETECT (CD) INDICATOR
- 4) DIALING KEYPAD
- 5) READY INDICATOR
- 6) VU METER

1	BACKLIT LCD	Displays all status and menu options.
2	FUNCTION KEYPAD	These keys provide “one touch” access for several major functions such as Qdial and hangup. Some of the keys also have secondary functions which are noted by the key. (See page 9 for more details.)
3	RING/CARRIER DETECT (CD) INDICATOR	When the Matrix receives an incoming call, this light will flash to indicate the line is ringing. When the modem inside the Matrix is connected to another compatible codec, this light will be on steadily.
4	DIALING KEYPAD	Use this keypad to place calls, either in Matrix or POTS Mode. Numbers may also correspond to menu options.
5	READY INDICATOR	This is a signal from the digital audio decoder which signifies it has locked onto the incoming digital audio bitstream. This will usually light up a few seconds after connection and go out if a retrain is necessary or if the codec “loses sync” for any reason. A contact closure which follows this indication is available on the rear panel.
6	VU METER	This provides an indication of the audio level being fed into the Matrix. It doesn’t mirror the receive audio in any way. Most users set input level so the VU peaks just over the “0” mark, which is the yellow LED on the meter.

RACKMOUNT - REAR PANEL DIAGRAM & DESCRIPTION



- 7) *IEC 320 POWER CONNECTOR/FUSE OUTLET*
- 8) *OUTPUT CONNECTOR*
- 9) *INPUT CONNECTOR*
- 10) *AUX CLOSURE*
- 11) *READY CONTACT CLOSURE*
- 12) *MULTIPURPOSE PORT*
- 13) *CC SEND CONTROL*
- 14) *CC RECEIVE CLOSURE*
- 15) *TEL LINE JACK*
- 16) *TEL SET JACK*
- 17) *ACCESSORY OPTION BACK PLATE*

- | | |
|--|---|
| 7 <i>IEC 320 POWER
CONNECTOR/FUSE OUTLET</i> | This is an IEC connector for the main power. The Matrix works on any world-wide AC power from 100-240VAC. |
| 8 <i>OUTPUT CONNECTOR</i> | This is a 3-pin male XLR connector with balanced, line level output audio. The rackmount Matrix does not mix any local audio into this output. The only audio heard here will be coming from the far end. The level of this output is selectable via the menu system. |
| 9 <i>INPUT CONNECTOR</i> | This is a 3-pin female XLR connector designed to accept balanced, line level audio. This audio sent into the encoder on the Matrix is reflected on the front panel VU meter. The nominal level of this input is selectable via the menu system. |
| 10 <i>AUX CLOSURE</i> | This mini jack input provides the ability to dial the phone number stored in the <i>QDIAL 01</i> position when the contact is closed. (See page 49 for more details on using the Matrix as a backup to your STL.) The <i>AUX CLOSURE</i> may also be used for future features, including select remote control functions. |

RACKMOUNT - DESCRIPTION OF REAR PANEL CONTROLS AND CONNECTORS

- | | | |
|----|--|--|
| 11 | <i>READY CONTACT
CLOSURE</i> | This mini jack provides a latching dry contact closure when the digital audio decoder achieves synchronization with an incoming digital audio signal. This contact is closed when the green <i>READY</i> light below the LCD display is illuminated and is released when the <i>READY</i> light goes out. |
| 12 | <i>MULTIPURPOSE PORT</i> | Flash upgrades may be performed through this port. (See page 54 for more details.) Future uses of this port include connection for a TCP/IP remote controller for Matrix functions. |
| 13 | <i>CC SEND CONTROL</i> | This mini jack allows the user remote control of the <i>CC</i> button on the front keypad. When the Matrix is on-line, a closure here will send a signal to close the <i>CC</i> receive relay at the far end. An asterisk will appear in the upper right corner of the far end Matrix, indicating that the signal has been received. |
| 14 | <i>CC RECEIVE CLOSURE</i> | This mini jack is a momentary dry contact closure which triggers closed when the Matrix is on-line and the far end sends the <i>CC</i> command via the keypad. |
| 15 | <i>TEL LINE JACK</i> | This RJ11 jack is for attaching the telephone line from the outside world to the Matrix. Be careful what you plug into here. Some PABXs provide currents which can damage the Matrix. |
| 16 | <i>TEL SET JACK</i> | This RJ11 jack is for attaching an external telephone to the Matrix. This can be useful for dialing complicated numbers with lots of pauses and special codes. It can also serve to provide an audible “ring” when an incoming call is received. |
| 17 | <i>ACCESSORY OPTION
BACK PLATE</i> | This plate can be removed if the optional ISDN board is installed. Connectors for ISDN will be placed in the voids behind this plate. |

SECTION 4. SETTING UP THE MATRIX

POWER CONNECTIONS

Rackmount — Attach the supplied IEC power cable to the Matrix power inlet. Attach the other end to AC power. The Matrix display should light up, and you should see the introductory screen replaced shortly by the *MAIN STATUS DISPLAY*. See “Power Up Sequence” on page 24.

Portable — Attach the external power supply to the connector on the side panel marked “power.” Attach the provided AC power cord to the power supply, and connect to AC power. Turn the Matrix on using the side panel power switch (0 = off and 1 = on). The Matrix display should light up. You will then see the introductory screen, followed shortly by the *MAIN STATUS DISPLAY*. See “Power Up Sequence” on page 24.

TELEPHONE CONNECTIONS

Rackmount and Portable — Attach the supplied telephone (RJ11) cable to the jack on the rear/side panel of the Matrix labeled *TEL LINE*. Connect the other end of the telephone cable to the telephone jack provided by the phone company. Note: If you plan to use a PABX connection, you may damage the Matrix by plugging it into the jack. If you’re unsure about this, better to try a cheap telephone first before plugging in the Matrix. You can also use a small device called a “modem saver” to test the line. These are available through computer supply outlets and catalogs for testing phone lines before plugging in laptop computers. Usually, any jack which will support a fax machine or computer modem can be used with the Matrix, although use on PABXs often results in less than optimal performance. Under rare circumstances, you may wish to connect a telephone to the jack on the Matrix labeled *TEL SET*. This will allow you to hear the telephone line ring when an incoming call comes — or just to use the phone line for other purposes when not on-air. Also, an external telephone may be used to dial the Matrix when the dialing functions of the Matrix won’t fit the bill (very long, complex dialing strings with lots of digits and prompts). But usually you’ll leave this jack unconnected.

AUDIO CONNECTIONS - RACKMOUNT

Rackmount Input — Connect your send audio to the jack on the rear panel labeled *INPUT*. This jack is designed to accept line level, balanced audio. If you will be delivering unbalanced audio to the Matrix (not recommended), see the section on page 23 called “About Unbalanced Connections.” The nominal input level to the Matrix is 0dBu (.775V) but can be changed to -10 and +4 dBu via the menu. A +8 dBu level can also be selected via an internal jumper. (See page 70.) Beyond that, you will need to adjust the level of the device feeding the Matrix to achieve an appropriate level. Once the audio is connected correctly, you will see the level reflected on the front panel LED VU meter.

To change the input level to the Matrix, press the *ENTER* key from the *MAIN STATUS DISPLAY*. Next press “4” for *CONFIGURE* to enter the configuration options. Select “4” for *MORE* options, then “4” for *LEVELS*. Next select “1” for *INPUT LEVEL*. You can make your choice between the three nominal levels, “1” for 0 *dBu*, “2” for +4 *dBu* or “3” for -10 *dBu*. Whichever level you select will be stored in the Matrix until it is changed, even after being powered off. Once the correct level is selected, watch the VU meter for appropriate deflection. Most users like to see a VU meter peak occasionally over the “0” mark, which is the yellow LED on this meter. If you cannot achieve a good input level using the available selections, you will need to adjust the level of the device feeding the Matrix.

Rackmount Output — Attach your balanced output audio connector to the male XLR on the rear panel labeled *OUTPUT*. If you require unbalanced audio from the Matrix, see the section on page 23 called “About Unbalanced Connections.” Like the input level, the output level is factory configured for 0*dBu* (.775V) nominal but can be set via the menu for -10, 0 or +4 *dBu*. To do this, press the *ENTER* key from the *MAIN STATUS DISPLAY*. Next press “4” for *CONFIGURE* to enter the configuration options. Select “4” for *MORE* options, then “4” for *LEVELS*. Next select “2” for *OUTPUT LEVEL*. You can make your choice between the three nominal levels, “1” for 0 *dBu*, “2” for +4 *dBu* or “3” for -10 *dBu*. Your selection will remain until it is changed, even after powered off. A +8*dBu* level is also available via internal jumpers. See page 70.

There is an easy way to check the output level of the Matrix, even before placing a call. First, set up your input level so it has audio at the correct level. From the *MAIN STATUS DISPLAY*, press *ENTER* to access the *MAIN MENU*. Now select “3” for *TEST*. You can make your choice between several available algorithms, such as “2” for *MUSIC*. Next you may select any loopback test here, such as “3” for 28.8 kb/s. The *READY* light on the front panel is on now, and you can monitor the audio you are sending through the output port. Assuming the end to which you’re connected has set their input level correctly, this is the approximate level you will receive on the output. You may adjust the input stage of the device being fed from the Matrix based on this level.

*AUDIO CONNECTIONS -
PORTABLE*

Portable Input 1 — As you look at the unit from the rear, Input 1 is the 3-pin female XLR connector farthest to the right. This input can be switched between microphone or line level. To get started, select whether you will be feeding mic or line level audio into channel 1. If you are connecting a microphone, the switch to the right of the channel 1 input connector must be up. Otherwise, set the switch down and feed a balanced line level into the connector. If your source is unbalanced, read the section on page 23 “About Unbalanced Connections.” Now feed some typical audio into Input 1, and adjust the level using the Input 1 knob on the front panel. Watch the level indicator for appropriate deflection. The level indicator is a bi-color LED. When a sufficient level of audio is fed to the unit, the LED will be green. The level indicator will turn red to indicate peaks. Adjust the input level so that the LED flashes red only occasionally. If you cannot get the LED to flash green, or if it is steadily on red, check to see that the Mic/Line switch is set correctly.

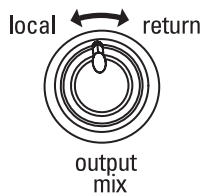
Portable Input 2 — Input 2 is located directly to the left of Input 1 when viewing the portable unit from the rear. This is a microphone level input. Connect your microphone to this XLR, and adjust your input level via the Input 2 level control on the top surface. The input level will be reflected on the bi-color LED. As described under Input 1, when a sufficient level of audio is fed to the unit, the LED will be green. The level indicator will turn red to indicate peaks. Adjust the input level so that the LED flashes red only occasionally.

If you cannot achieve a good input level, make sure that you are feeding the correct level. If that doesn't work, you will need to try a different microphone or cable. These inputs do not support microphones which require phantom power.

Headphone Output 1 — You must have stereo headphones to use with the Matrix. Plug the headphones into the headphone jack. Feed some audio into one of the inputs. Adjust the Headphone knob for a comfortable listening level. The output mix may be adjusted to hear more local audio (audio fed to inputs) or return audio (audio fed from other end). Please refer to “Adjusting the Output Mix” on page 23 for details.

XLR Line Level Output — There is one line level audio output on the Matrix portable, labeled *OUTPUT*. This can feed ancillary equipment such as an external mixer, PA system, extra headphones or recording device (DAT or tape recorder). Use the *LINE OUTPUT* knob, on the top surface of the Matrix for level adjustment. Like the headphone output, this is an adjustable mix of local and return audio. Please refer to “Adjusting the Output Mix” on page 23 for details.

ADJUSTING THE OUTPUT MIX (MATRIX PORTABLE)



The portable Matrix outputs (XLR output and headphone output) feed the user a mix of two different audio sources. The *LOCAL* portion contains all the microphones or other sources connected to the Matrix inputs, and the *RETURN* portion contains all the audio being sent from the other end of the link. The portable Matrix mixes the two sources and feeds the output XLR and the headphones (both left and right) with the same mix. The ratio of local and return audio may be adjusted using the *OUTPUT MIX* adjustment knob. Turning the knob counterclockwise gives more local audio. Turning the knob clockwise gives more return audio. Adjust this knob until the desired mix of local and return audio is achieved. There is only one *OUTPUT MIX* adjustment knob which determines the mix for both the XLR output and the headphone output.

ABOUT UNBALANCED CONNECTIONS

It is best when using professional audio gear to use balanced audio connections. Sometimes, however, it is unavoidable to connect to consumer equipment with unbalanced connections. If you must use unbalanced audio, here's how to connect to the Matrix:

Matrix inputs — When driving the Matrix from an unbalanced source, connect pins 1 and 3 together on the Matrix input XLR. This should connect to the ground signal on your unbalanced device. Connect pin 2 on the Matrix XLR to the unbalanced device's audio + connection.

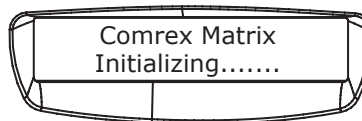
Matrix outputs — The Matrix uses active balanced outputs designed to drive unbalanced sources directly. This means there won't be any level difference when feeding an unbalanced source. On the Matrix output XLR, connect pins 1 and 3 together, and tie this to ground on your unbalanced device. Connect pin 2 on the Matrix XLR to + audio side on the unbalanced device.

SECTION 5. CONFIGURING THE MATRIX

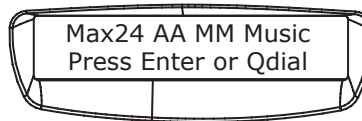
POWER UP SEQUENCE

When the Matrix is connected to power and turned on, the initializing menu appears for approximately two seconds. The Matrix then shows the *MAIN STATUS DISPLAY*. The main setup options are displayed in the top portion of the screen. As shipped from the factory, the Matrix *MAIN STATUS DISPLAY* will be the same as the screen shown below. Again, these selections will remain in nonvolatile memory until changed.

INITIALIZING SCREEN



MAIN STATUS DISPLAY

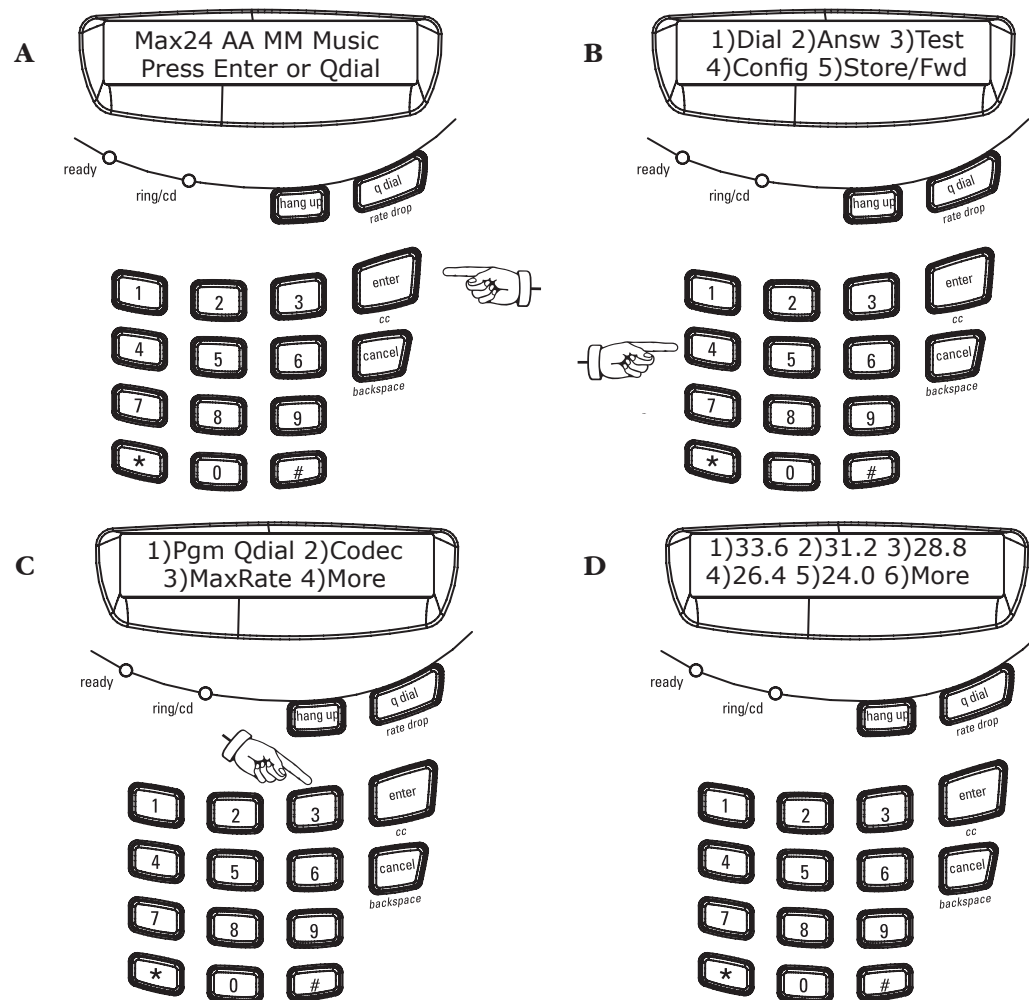


SETTING THE MAXRATE

The MaxRate option allows you to limit the maximum modem connect rate. The MaxRate factory default setting of a new Matrix is 24 kb/s. If the Matrix is allowed to negotiate on an unrestricted basis, it will find the highest possible connect rate. Since telephone line connections can vary over time due to crosstalk, burst errors and other changing parameters, the initial connect rate may not be sustainable. It may help to set a MaxRate at a level or two below the maximum unrestricted rate to provide a “guard band” of sorts against the noise and corruption which may cause errors on the line. The Matrix (or Vector or BlueBox or HotLine) with the lowest MaxRate setting will determine the maximum connect rate.

To set the MaxRate:

- A. Press *ENTER* from the *MAIN STATUS DISPLAY*.
- B. Then press “4” for *CONFIGURE*.
- C. Then “3” for *MAXRATE*.
- D. Using the keypad, press the number which corresponds to the MaxRate setting you want.



PROGRAMMING QDIAL

Up to 19 numbers may be pre-programmed into the Matrix and then recalled through the Quick Dial or *QDIAL* button on the *MAIN STATUS DISPLAY*. Pressing *QDIAL 00* will redial the last number dialed. These numbers will remain in memory even when power is turned off. The number programmed into the *QDIAL 01* location may be used in the STL Backup Mode. (See page 58 for more information on STL Backup Mode.)

To program a Qdial number into memory:

- A. Press *ENTER* in the *MAIN STATUS DISPLAY*.
- B. Then press “4” for *CONFIGURE*.
- C. Then “1” for *PGM QDIAL* (Program Quick Dial).
- D. The screen will display “Enter Qdial #:01-19.” Enter a number from 01 to 19 to correspond to the Qdial position you wish to use.
- E. Then key in the number to dial, including any country, area or other codes you wish (up to 44 digits). To insert a two second pause, press the *QDIAL* key. A comma will appear to indicate the pause. Multiple pauses may be inserted. The *CANCEL* key backspaces for corrections.
- F. When the number is entered correctly, press the *ENTER* key. Pressing the *HANGUP* key will exit this menu without saving your entry.

MODEM MONITOR

You may choose to hear the modems negotiate through the audio port. This is found in the *CONFIGURE MENU*. The factory default is for the Modem Monitor to be “on,” allowing you to hear the modem “chat” familiar to computer users. This includes dial tone, pulse or tone digits, ringing and all those funny “blurps” that happen at the beginning of a modem connection. Under some circumstances, you may not wish to hear these things. If, for example, the Matrix in the studio were “on-air” all the time, the listeners would hear these sounds.

To select a modem monitor option:

- A. Press *ENTER* from the *MAIN STATUS DISPLAY*.
- B. Then select “4” for *CONFIGURE*.
- C. Now select “4” for *MORE*.
- D. Next press “3” for *MODMON* (Modem Monitor).
- E. Choose “1” for *MODEM MONITOR ON* or “2” for *MODEM MONITOR OFF*.

AUTO ANSWER

To choose how/if you want the Matrix to answer incoming calls:

- A. Press *ENTER* in the *MAIN STATUS DISPLAY*.
- B. Then select “4” for *CONFIGURE*.
- C. Now select “4” for *MORE*.
- D. Next press “1” for *AUTO ANS* (Auto Answer).
- E. Choose “1” for *ON*, “2” for *OFF* OR “3” for *6 RINGS*
(The Matrix will answer after 6 rings.)

*TONE/PULSE DIALING***To select tone or pulse dialing:**

- A. Press *ENTER* from the *MAIN STATUS DISPLAY*.
- B. Then select “4” for *CONFIGURE*.
- C. Now select “4” for *MORE*.
- D. Next press “2” for *DIALING*.
- E. Choose “1” for *TONE DIALING* or “2” for *PULSE DIALING*.

The Matrix will remember your selection until you change it, even when powered down. The Matrix factory default setting is for *TONE* dialing.

AUDIO CODING SELECTION

Music Mode — This mode, optimized to give the widest frequency response possible, is the factory default setting. Your Matrix will automatically operate in this mode unless it calls or is called by a Matrix, BlueBox or Vector set to a less demanding mode or by a HotLine. If connecting to a HotLine, the Matrix will automatically switch to HotLine Mode. If connecting to a Matrix, BlueBox or Vector set to Voice Mode, the Matrix will automatically default to the more conservative Voice Mode. If you are having difficulty connecting in Music Mode, double check that both codecs are set to Music Mode.

Voice Mode — Under normal operation, the Matrix is designed to provide maximum bandwidth based on the connect rate. Some users, however, may prefer to forego very high bandwidth in favor of a more reliable connection as well as to provide an ancillary data channel. Voice Mode locks the highest bandwidth at 7 kHz, and any leftover data rate is used to provide extra error correction, along with a 300 BAUD ancillary data channel. Voice Mode will (within limits) provide for a more reliable audio link. For more on the ancillary data connection, see page 47. If one Matrix is set to Voice Mode and the other is not, the connection will default to Voice Mode in both directions.

GSM Mode — This mode allows for 7 kHz audio transmission over a 9600 baud GSM connection. This mode is only applicable when you are using the Matrix on a POTS line as the receiving end of a GSM call. The receiving Matrix cannot automatically adapt between GSM Mode and normal POTS codec, therefore GSM Mode must be enabled for proper GSM operation.

To set the Matrix for Music, Voice Mode or GSM Mode:

- A. Press the *ENTER* key from the *MAIN STATUS DISPLAY*.
- B. Then select “4” for *CONFIGURE*.
- C. Now select “2” for *CODEC*.
- D. Choose “1” for *VOICE MODE*, “2” for *MUSIC MODE* OR “3” for *GSM MODE*.

After a selection is made, the screen will display *ALGORITHM SELECTED*. The Matrix will remain in this mode until you change it. You can now *CANCEL* back to the *MAIN STATUS DISPLAY*.

SECTION 6.

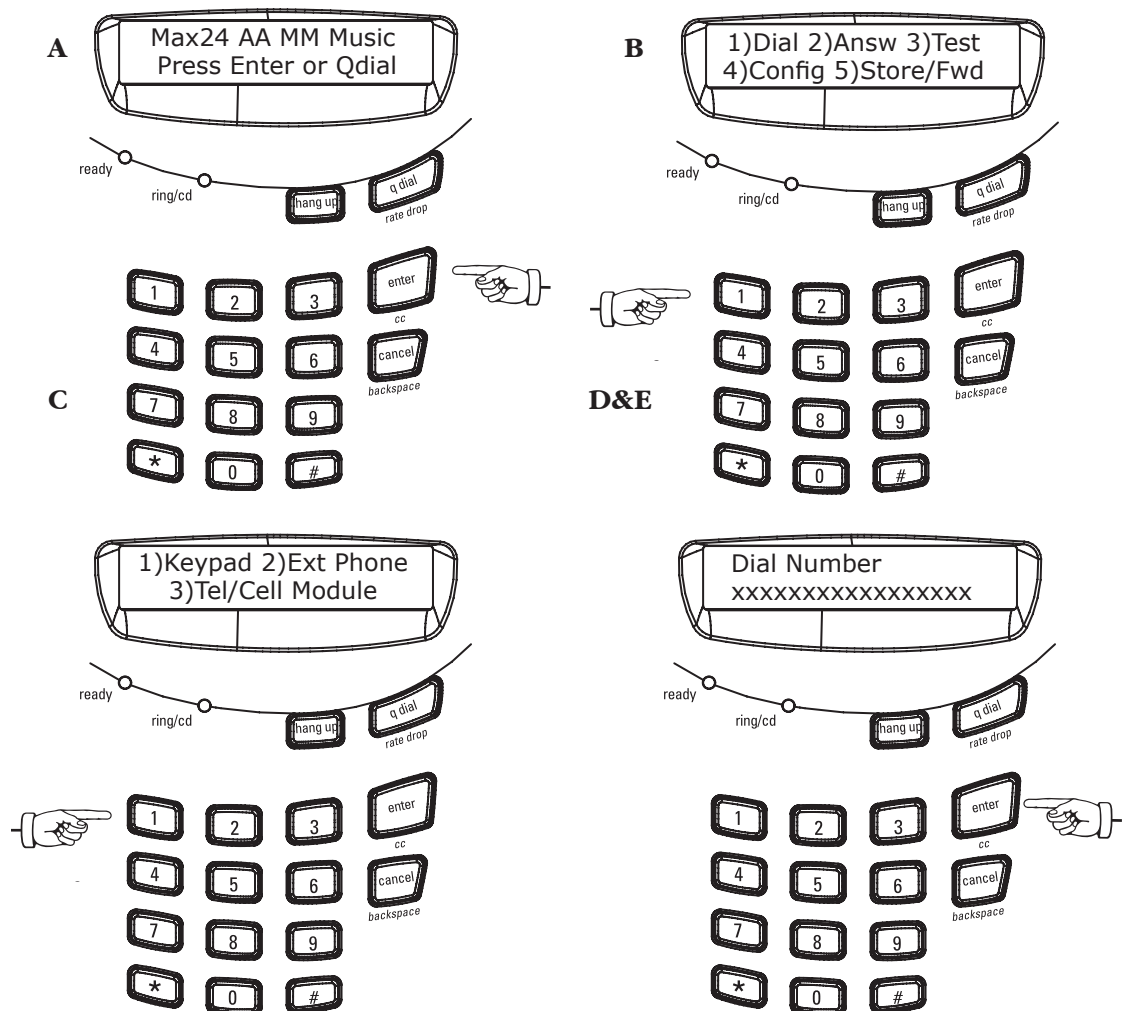
PLACING A CALL FROM THE MATRIX

So now it's time to use your Matrix. We will assume that all the proper connections have been made, and you're ready to place a call. There are several ways to dial from the Matrix. The most common method is to dial from the built-in keypad.

DIALING FROM THE KEYPAD

To place a call from the keypad:

- A. Press *ENTER* from the *MAIN STATUS DISPLAY*.
- B. Then press "1" for *DIAL*.
- C. Then press "1" for *KEYPAD*.
- D. Now key in the number you wish to dial. The *CANCEL* key will back space, and the *HANGUP* key will take you back to the *MAIN STATUS DISPLAY*. To insert pauses and disable call waiting, see page 29.
- E. Then press *ENTER* to dial the number. (Pressing the *HANGUP* key disconnects the call.)



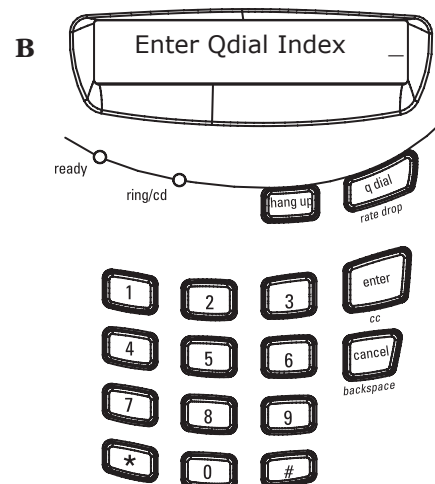
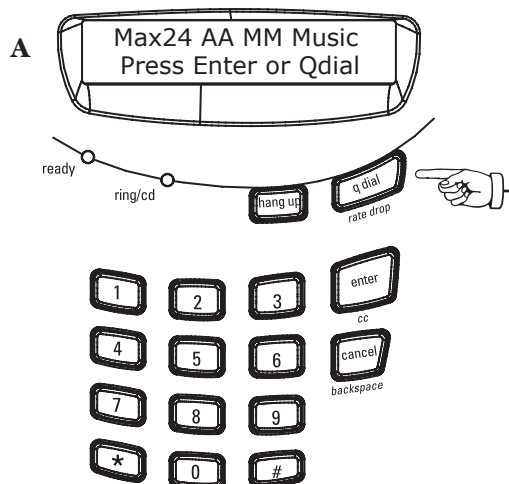
If you wish to use the Comrex Matrix test line number, call 978-784-1700. You may enter up to 44 digits in a dialing string. A two second pause may be entered anywhere and used multiple times in the dialing sequence by pushing the *QDIAL* softkey. This will insert a comma to indicate the position of the pause. This pause may be needed to dial through certain PABXs or to place credit card calls. Also, you may wish to enter other characters, such as “*70” which is typically used at the beginning of a number to disable call waiting.

To dial the number, press *ENTER* after you have keyed in the digits. Note: The phone line doesn’t go “off hook” until the *ENTER* key is pressed (much like a fax machine). The far end unit will either auto answer or will need to be answered manually, depending on its setup. Once the far end has answered, the Matrixes will negotiate and a “connect” message will be displayed on both units, including the rate at which the connection was accomplished. Within a second or two, the *READY* light on each Matrix should illuminate. At this point, you will have wideband audio (response dependent on connect rate and selected mode) in both directions.

DIALING WITH *QDIAL*

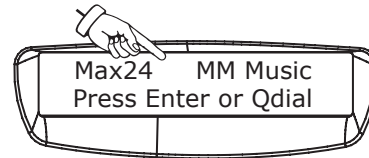
To dial a pre-programmed Qdial number:

- A. Press *QDIAL*.
- B. Then press the two-digit location (01-19) where the number is stored. Pressing *QDIAL 00* will redial the last number dialed. (See page 26 for instructions on programming numbers into Qdial.) All Qdial numbers, including last number redial are stored in non-volatile memory.



*ANSWERING A CALL
MANUALLY*

First, check on the *MAIN STATUS DISPLAY* that auto answer is off. If AA is seen on the display, your auto answer is on.



If you need to reset this, follow the instructions on page 26. You can connect a telephone set to the *TEL SET* jack on the rear panel to hear the line ring. Otherwise, you will need to wait for the yellow *RING/CD* light on the Matrix to begin flashing to indicate an incoming call. To answer a call when the line is ringing:

- A. Press the *ENTER* key.
- B. Then press “2” for *ANSW* (Answer Call).

The Matrix will answer the incoming call and negotiate a connect speed. The call can be disconnected by pressing the *HANGUP* key.

*DIALING FROM AN EXTERNAL
PHONE*

In complex dialing situations, such as when an operator may be involved or when the dialing string is longer than 44 digits, you can dial the Matrix using an external telephone set. Connect an external single line telephone set to the modular jack on the rear panel of the Matrix labeled *TEL SET*.

- A. Press *ENTER* from the *MAIN STATUS DISPLAY*.
- B. Then press “1” for *DIAL*.
- C. Then “2” for *EXT PHONE* (External Phone).
- D. Pick up the external telephone handset and listen for a dial tone. Dial the number, and press *ENTER* at the first ring and hangup the telephone handset.

When you press *ENTER*, this engages the modem. You need to hangup the external telephone handset immediately after pressing *ENTER* because the modem negotiation process is most reliably done without an external handset off-hook across the line. Note: When the external telephone option is used, the number you are dialing does not appear on the display. You can disconnect the call normally on either end of the line by pressing the *HANGUP* key.

LAST NUMBER REDIAL

The last number dialed is automatically stored in the *QDIAL 00* location. To redial the last number:

- A. Press the *QDIAL* key from the *MAIN STATUS DISPLAY*.
- B. Press the “0” key two times.

The last number you dialed from the Matrix will be re-dialed. This number will remain in memory after the Matrix is powered down. Numbers dialed through an external phone will not be kept in memory.

HANGING UP THE CALL

When you wish to disconnect your call, simply press the *HANGUP* key. After several seconds, the call will be cleared and the *MAIN STATUS DISPLAY* will return. If the other end initiates the disconnect, your unit will also disconnect and revert to the *MAIN STATUS DISPLAY*. Note: It can take about five seconds between the time you terminate the call to the time the Matrix releases the line for the next call.

DROPPING THE CONNECT RATE DURING A CALL

If a telephone line degrades during a broadcast, it is possible to lower the connect rate on the line without disconnecting the call. The lower connect rate will help to provide a “guard band” against the errors on the line. During the renegotiation period, the audio will be muted for about 2.5 seconds. To lower the connect rate, simply press the *QDIAL* key.

By pressing the *QDIAL* key while on-line, you’re instructing the Matrix to drop one rate lower than that at which you’re actually connected. This rate is now “locked” as the *MAXRATE* for this call (e.g. if the Matrix suffers a renegotiation, it won’t choose a connect rate above the one to which you’ve dropped).

Another exclusive Matrix feature is the ability to do a full retrain at will. This is accomplished by pressing the *QDIAL* key twice rather than once (like a “double mouse click”). This will cause the Matrix to do a full negotiation (as it does at the beginning of the call) on the connection. We highly recommend the use of this feature on prolonged broadcasts (longer than one hour) to “reset” all the parameters of the modem to the changing parameters of the phone line. It’s much more desirable to force the Matrix to retrain during your program break than to have it trigger its own retrain during your program due to changing phone line conditions. Audio in both directions will be muted for about 10 seconds during the retrain sequence.

On occasion, phone line parameters are such that the modems can’t support the change to a lower rate. This is because, along with changing the data rate, the modem may be required to change filters, symbol rates and other parameters that you don’t see. If the line won’t support the parameters, you may see the unit break into renegotiation, retrain or (on rare occasion) disconnect the call. The fix here is to set the *MAXRATE* setting to the rate at which you need to work and redial the call.

*PLAIN TELEPHONE MODE vs.
TELCELL MODE*

Earlier Matrix differ from current production models in that they have the built in ability to place a “plain telephone” call when a modem connection cannot be established or if the far end does not have a Matrix, Vector, BlueBox or HotLine. This function is enabled in current production models through the addition of the optional TelCell module. Not only does the TelCell module add the ability to send and receive “plain telephone” calls but also provides a connection to the hands-free port of many mobile phones. Follow the steps listed below to determine which method applies to your unit.

A. Press the *ENTER* key from the *MAIN STATUS DISPLAY*.

B. Then press “1” for *DIAL*.

C. The display will read one of the following:

- *PLAIN PHONE MODE* — Refer to Appendix E “Plain Telephone Mode” on page 82.
- *TEL/CELL MODULE* — See the next section “Using the TelCell Module.”
- No third option — Your Matrix is a rackmount which does not support *PLAIN TELEPHONE MODE* or the *TELCELL MODULE*.

*USING THE TELCELL MODULE
(PORTABLE MATRIX ONLY)*

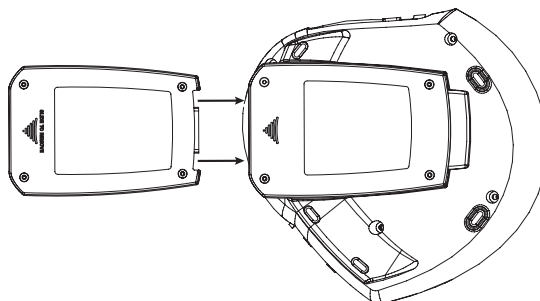
The Matrix portable has an optional module available that allows operation on phone lines and mobile phones, without any quality enhancement. This allows the Matrix to be used as a “field mixer,” connecting to studios without Comrex codecs or in cases where phone lines can’t support codec transmissions.

Note: The rackmount Matrix does not operate in Plain Telephone Mode; therefore, if you place calls from a field Matrix using the TelCell Module, you will need to dial into a separate coupler or hybrid at the studio

The TelCell module is inserted into the accessory port under the portable Matrix before the Matrix is powered up. The Matrix will detect its presence and allow you to enter *PLAIN TELEPHONE MODE* or *CELLULAR HANDS-FREE MODE*. When using the TelCell module, the phone line is connected to the telephone jack on the module rather than the jack on the Matrix itself.

Normal POTS codec operation is possible with the TelCell module installed, so it’s OK to leave the module connected to the Matrix when you’re not using it. You will, however, need to connect the phone line back to the Matrix to operate in normal POTS mode.

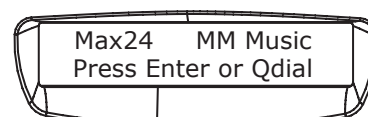
Diagram of TelCell Module insertion into the base of the field Matrix.



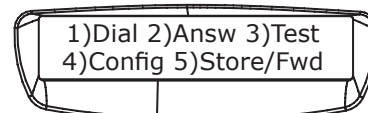
Note: Portable Matrix with serial numbers greater than 0539 require the TelCell module to enter PLAIN TELEPHONE MODE. Earlier Matrix have PLAIN TELEPHONE MODE built in. If you have an earlier version and would also like the ability to send/receive unenhanced audio over a mobile phone, your Matrix will require a modification to work with the TelCell module. Please contact Comrex customer service for details.

PLAIN TELEPHONE MODE OPERATION

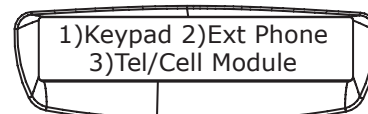
Press the *ENTER* key from the *MAIN STATUS DISPLAY* to access the main menu.



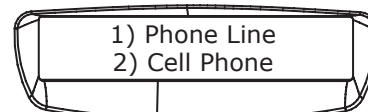
Select 1) *DIAL*.



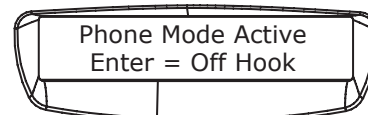
Select 3) *TELCELL MODULE*.



Select 1) *PHONE LINE*.



The Matrix will display:



When you press the *ENTER* key, the line will go off-hook, and you will hear dial tone at the *LINE OUT* and *HEADPHONE* ports. You can now place a call using the Matrix keypad.

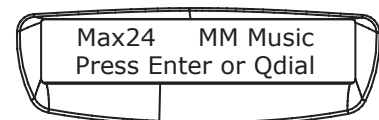
The audio from the Matrix inputs is sent to the phone line, and you will hear everything being sent back on the phone line. Press the *HANGUP* key to disconnect your call.

The *PLAIN TELEPHONE MODE* uses a simple telephone hybrid arrangement that isolates the send and receive audio enough to allow communication, but doesn't provide the isolation of a digital hybrid. Anything you send into the Matrix will appear at the Matrix output at a reduced level, so true "four wire" operation (as in the normal POTS codec mode) isn't possible.

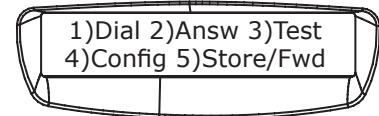
CELLULAR HANDS-FREE MODE OPERATION

The *CELLULAR HANDS-FREE MODE* allows you to connect send and receive audio into the hands-free port of most mobile phones. See below for more about cables and phones. Once the wiring is complete, enter the *CELLULAR HANDS-FREE MODE* as follows:

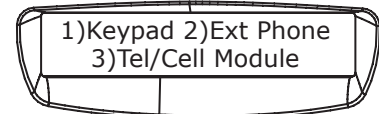
Press the *ENTER* key from the *MAIN STATUS DISPLAY* to access the main menu.



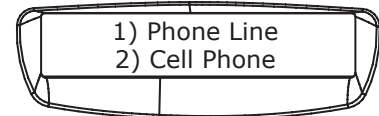
Select *1)DIAL*.



Select *3)TELCELL MODULE*.



Select *2)CELL PHONE* to put the Matrix into the proper mode where send and receive audio are routed via the *CELL* jack on the TelCell module.



You will need to dial the call from the mobile phone keypad, but when the far end answers, you will be able to converse through the Matrix. Press the *CANCEL* key to exit *CELLULAR HANDS-FREE MODE*.

TELCELL MODULE CON- NECTIONS FOR HANDS-FREE CELLPHONE PORTS

The jack labeled *CELL* on the TelCell Module will need to be connected to the hands-free port of your mobile phone. This is done differently for different makes of phones. We've included the cable required for most Motorola phones, which have a 2.5 mm jack built in for hands-free operation. Cellular Accessory.com provides hands free audio adapters for a wide range of other cellphone manufacturers which convert a universal 2.5 mm handsfree jack to the required cellular phone model audio connection. These adapters can then be used with the cable provided.

The web address is: www.cellularaccessory.com/audioadapter.html, and the phone number is 1-800-2-BUY-PCS.

SECTION 7.

USING STORE AND FORWARD

ABOUT STORE AND FORWARD

The Matrix includes a *STORE AND FORWARD* feature which allows the user to record a high quality audio feed to be sent on a low speed circuit for later, non-real time broadcast. This is useful for drop-ins from a poor phone line or cellular circuit that cannot sustain an adequate data rate for codec use. Up to 9 minutes and 45 seconds of 15 kHz audio can be stored in the Matrix in nonvolatile memory. The audio will be stored until it is erased, even if the unit is powered off. *STORE AND FORWARD* (S/F) can only be used with Matrix codecs at both ends of the link (no Hotlines or Vectors).

Operation will require the following steps:

On the remote end of the link:

- 1) Set up the Matrix to record the S/F cut and to send it.
- 2) Feed the live or recorded audio into the Matrix audio input.
- 3) Initiate the call to the studio in order to send the S/F cut.

On the studio end of the link:

- 1) Set up the Matrix to receive the S/F cut.
- 2) Let the Matrix answer the call and receive the audio cut.
- 3) When complete, play the S/F audio cut.

In general, one Matrix will always be designated as the “recorder” and “sender” (typically the field unit) and the other Matrix will always be designated as the “receiver” and “player” (typically the studio unit). By default, portable units are configured for “record” and “send” and rack units are configured for “receive” and “play” – although this configuration may be changed on both units. See page 42-43 for setup instructions, if you wish to change the default settings.

ENTERING STORE AND FORWARD MODE

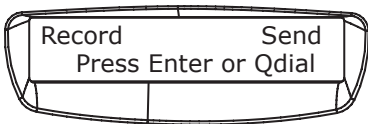
To enter Store and Forward mode, press the *ENTER* key from the *MAIN STATUS DISPLAY*. Next, select “5” for STORE/FWD.

Note: Your Matrix will not operate as normal real-time POTS or ISDN codec until you manually exit this mode. Also, Matrix units at both ends of the line must be manually set for Store and Forward mode in order for the S/F function to operate. A studio Matrix cannot automatically detect whether a call is a normal POTS codec call or a S/F call.

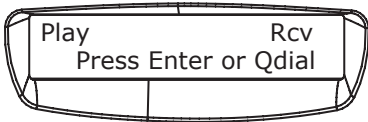
*STORE AND FORWARD STATUS
DISPLAY DEFAULT SETTINGS*

Once S/F mode is engaged, you will see the *MAIN STATUS DISPLAY*. The default settings are *RECORD* and *SEND* for the portable Matrix in the field and *PLAY* and *RCV* for the Matrix rack in the studio. Instructions for changing these default settings can be found on page 42-43.

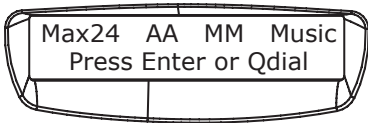
On the Matrix portable, the *S/F STATUS DISPLAY* looks like this:



On the Matrix rack, the *S/F STATUS DISPLAY* looks like this:



To exit S/F mode, press the *ENTER* key from the status display. You will be taken back to the *MAIN STATUS DISPLAY*:



*EXITING STORE AND
FORWARD MODE*

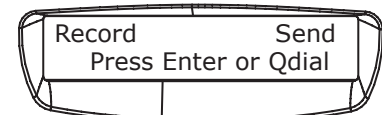
Choosing option 5 will return the Matrix to normal POTS codec operation. This is the only way to exit S/F mode, since the Matrix will remain in S/F mode, if powered off in this mode.

Note: You will choose option 4 to exit S/F on a Matrix which is set to PLAY, since option 2)REC (record) does not appear on a Matrix set to PLAY.

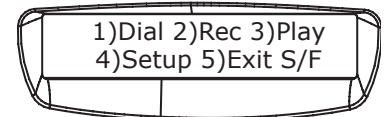
*STORE AND FORWARD
OPERATION - RECORD,
PLAYBACK AND SEND*

First, check the *S/F STATUS DISPLAY* to see that your Matrix has been set up for *RECORD*. This is the default setting for portable Matrix units.

- A. Press *ENTER* from the *S/F STATUS DISPLAY*.

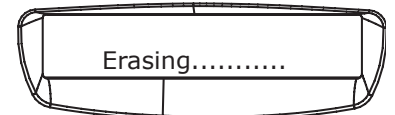
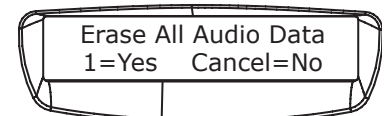


- B. Press *2)REC*.

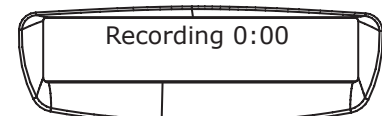
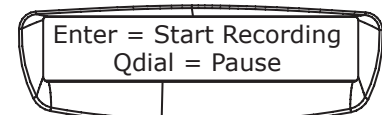


If audio is already stored in the Matrix, you will be asked if you want to erase all audio. If audio is not stored, you will skip directly to D.
(Note: If the *2)REC* option does not appear in the menu, follow the setup procedures on page 42-43 to set the Matrix for record.)

- C. Press *1* for *YES* or *CANCEL* for *NO*.



- D. Press *ENTER* to begin recording.
A timer will be displayed.

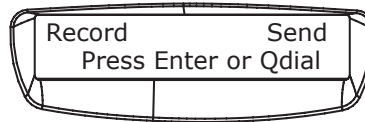


Pressing the *QDIAL* key during recording allows a pause, and pressing the *ENTER* key resumes recording.

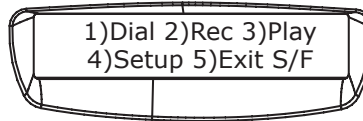
- E. Press *CANCEL* to end the recording and exit to previous menu.

*PLAYING BACK
RECORDED AUDIO*

- A. Press *ENTER* from the *S/F STATUS DISPLAY*.



- B. Press *3)PLAY*.



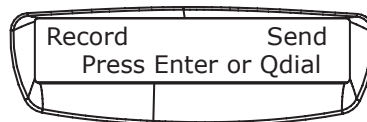
- C. The screen will read *PLAYING*. The audio clip will play and the *READY* light will illuminate. A timer in the upper right corner will count down for the duration of the recording.
- D. Pressing the *QDIAL* key pauses playback, and pressing the *ENTER* key resumes playback.
- E. Pressing *CANCEL* ends the playback and exits to previous menu.

*SENDING A STORE AND
FORWARD AUDIO CUT*

Here, it is assumed that the field unit will dial the call to the station. (This isn't necessarily required, i.e. the studio can just as easily dial the field. If this is the case, the instructions are the same. Simply do the dialing from the studio end. As long as the *SEND/RCV* settings are correct, audio will be sent in the correct direction.)

DIALING USING QDIAL

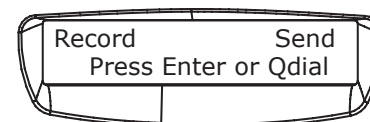
- A. Press the *QDIAL* button from the *S/F STATUS DISPLAY*.



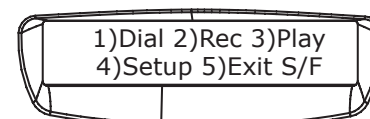
- B. Input the two digit index (01-19) of the programmed number you wish to dial. (See page 26 for information on how to program these numbers. *QDIAL* locations are shared with POTS codec mode).
- C. The unit will automatically dial the programmed number, initiate the upload procedure and disconnect the call when done.

*DIALING MANUALLY FROM
KEYPAD*

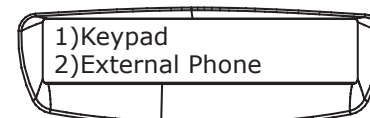
- A. Press *ENTER* from the *S/F STATUS DISPLAY*.



- B. Select *1)DIAL*.



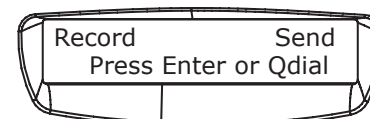
- C. Select *1)KEYPAD*.



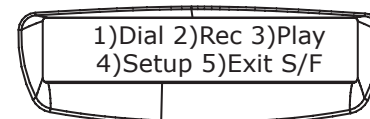
- D. Enter the number you wish to dial using the keypad. The *CANCEL* key backs over any errors, and the *HANGUP* key aborts the call. Press *ENTER* when finished.
- E. The unit will dial the call, initiate the upload procedure and disconnect when the audio cut has been sent.

*DIALING WITH AN EXTERNAL
PHONE*

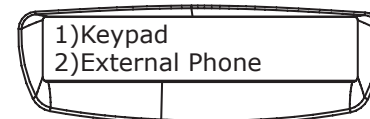
- A. Press *ENTER* from the *S/F STATUS DISPLAY*.



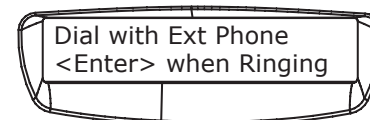
- B. Select *1)DIAL*.



- C. Select *2)EXTERNAL PHONE*.



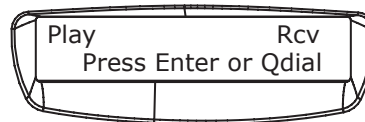
- D. The Matrix will then display:



- E. Dial the call using an external telephone connected to the *TEL SET* jack on the side of the Matrix. At the 1st ring, press the *ENTER* key and hang up the telephone.
- F. The Matrix will initiate the upload procedure and disconnect when the audio cut has been sent.

RECEIVING A STORE AND FORWARD AUDIO CUT

The factory default setting for a rackmount Matrix is to receive a *STORE AND FORWARD* feed, and the *S/F STATUS DISPLAY* will appear like this.

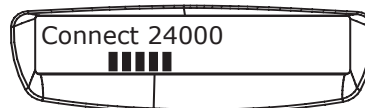


As long as the Matrix is set to receive and there are no stored audio cuts, the Matrix will automatically answer an incoming call and initiate the download procedure. (To erase a stored audio cut, see page 41.) After a new S/F audio cut is received this way, the Matrix will not revert back to the status display but will instead enter the “ready to play” mode described below.

Note: A Matrix does not have to answer the call in order to receive a S/F cut. This means that dialing can be initiated from either the studio or the remote end. As long as the SEND/Rcv settings are correct, the audio will be sent in the right direction.

SENDING/RECEIVING STATUS

While the Matrix is uploading or downloading, the screen will show the connect rate and a line of segments, each segment corresponding to approximately 30 seconds of audio. These segments will count up for an upload and down for a download.

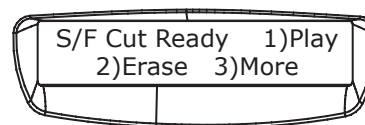


If the line is disconnected or the transfer is not completed, the screen will say Upload Failed! or Download Failed! These screens will show for about 3 seconds. The unit will freeze if uploading to (or downloading from) a unit that isn’t set to do the opposite, and the *HANGUP* key will have to be pressed on one end.

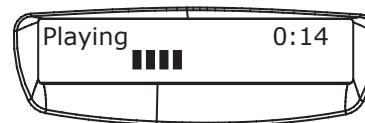
*PLAYING A RECEIVED STORE
AND FORWARD AUDIO CUT*

When a new S/F audio cut has been received, the Matrix will enter “ready to play” mode. Since the Matrix can only store one audio cut at a time, the options here are limited. You basically have the choices of playing the cut, erasing the cut, or reconfiguring the Matrix

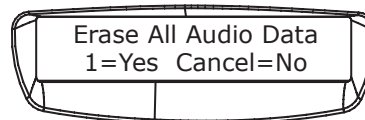
When in “ready to play” mode, the Matrix display looks like:



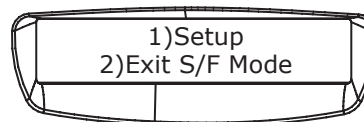
- A** 1) *PLAY* will immediately start to play the cut. Audio will appear on the Matrix output connectors. The *QDIAL* key functions as a “pause” button, and the *CANCEL* key will abort the playback.



- B** 2) *ERASE* gives the options of erasing the current S/F audio cut.



- C** 3) *MORE* displays the following options:



- D** 1) *SETUP* provides the setup options that are described in the Setup for Store and Forward section on page 42-43.

- E** 2) *EXIT S/F MODE* sets the Matrix back to the normal POTS codec mode.

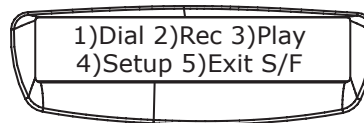
To recap, once a Matrix is locked into “ready to play” mode, there are three ways to exit this mode:

- 1) Erase the audio cut.
or 2) Change the setup so the Matrix is no longer in “receive” mode.
or 3) Exit the S/F mode.

SETUP FOR STORE AND FORWARD

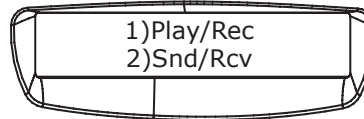
Setup for *STORE AND FORWARD* needs only be done once because the Matrix will remember the settings. For many users, setup will not even be necessary since the default settings will be correct (Record/Send for portable, Receive/Play for rackmount.)

If you wish to change the *STORE AND FORWARD* setup parameters, press *ENTER* from the main *S/F STATUS DISPLAY* and the S/F menu will appear:



**Note: REC (record) does not appear on a Matrix that has been set to PLAY.*

Press *4)SETUP*, and the setup menu appears:



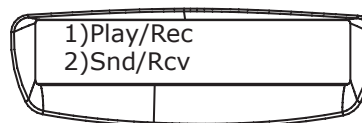
It's important to understand the meaning of the available setup options:

1) *PLAY/REC* – This shows whether the Matrix is configured to be the unit that records the S/F audio in the field, or whether it plays the S/F audio in the studio.

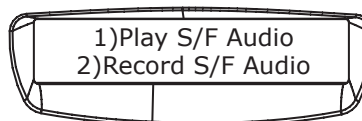
The *PLAY/REC* selection determines how the operational menus are structured in the following ways:

- A) A Matrix configured to “record” will have the *REC* option available in the main S/F menu.
- B) A Matrix configured to “play” will not have the *REC* option available in the main S/F menu.
- C) A Matrix configured to “play” will have a special “ready to play” menu appear after a S/F cut is loaded into its flash memory.

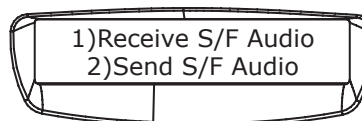
2) *SND/RCV* – This shows whether the Matrix is configured to send (upload) audio from the field or receive (download) audio at the studio. This is independent from the *PLAY/REC* selection and also has no effect on which Matrix dials and which answers the call.

PLAY/RECORD SETUP*STORE & FORWARD SETUP MENU*

Select *1)PLAY/REC* from the S/F setup menu to determine whether the Matrix is a recorder or a player. Then select whether you want to *1)PLAY S/F AUDIO* or you want to *2)RECORD S/F AUDIO*.

*SEND/RECEIVE SETUP*

Select *2)SND/RCV* from the S/F setup menu to determine whether the Matrix will be sending the S/F audio cut or receiving it. To transfer S/F audio, one Matrix must be set to *SEND* and the other to *RECEIVE (RCV)*. The menu selection looks like this:



The Send/Receive selection does not determine which end initiates the call and which end answers. This determination is made when the call is placed.

SECTION 8.

MATRIX GSM WIRELESS OPERATION

ABOUT THE GSM MODULE

When Comrex introduced the Matrix, the concept was to build the world's most versatile audio codec, one able to work over a variety of present and future digital networks. We certainly had our own ideas about what types of digital networks would gain wide acceptance, but since one can never be sure about these things, we left the ability for the product to expand into the unknown by including an accessory port for future upgrades.

Wireless audio transmission (with enhanced quality) has been our number one demand for years, and we've tried lots of techniques for providing it. The Matrix was designed with the thought of connecting external phones to the "multi-purpose" serial port and transmitting audio over the serial connection. This did work, but we found it to be cumbersome and complex, and the low data channel available on the wireless networks really stressed our existing audio coding algorithms to their limits. And phone models kept changing, with different models providing different connection issues, especially when used as data modems.

The GSM Module overcomes these limitations in two ways. First, by providing our own GSM phone built into the system, we can be sure the hardware is the very best suited to the task. The GSM modem within the module is a vastly superior design to handheld phones for this purpose. Secondly, we've enabled the Matrix to use a new coding algorithm in GSM Mode. This algorithm is uniquely suited to coding voice at the ridiculously low bitrates available on the GSM data channel. With these enhancements, we feel the application has finally "arrived" and become much more useful to many broadcasters.

When adapting an existing infrastructure (like a cellular network) for strange uses (like audio codecs), there will always be some technical limitations, and some of these limitations remain, even with the GSM Module. While the voice codec is the absolute best available at this data rate, it is not transparent and artifacts do remain. Also, while the codec system itself introduces only a small audio delay ($\sim 1/8$ second), the GSM network typically adds substantial delay, to the tune of $1/2$ second or more in each direction. Finally, any wireless codec will be susceptible to the same problems of interference, congestion, and signal strength as a normal cellphone.

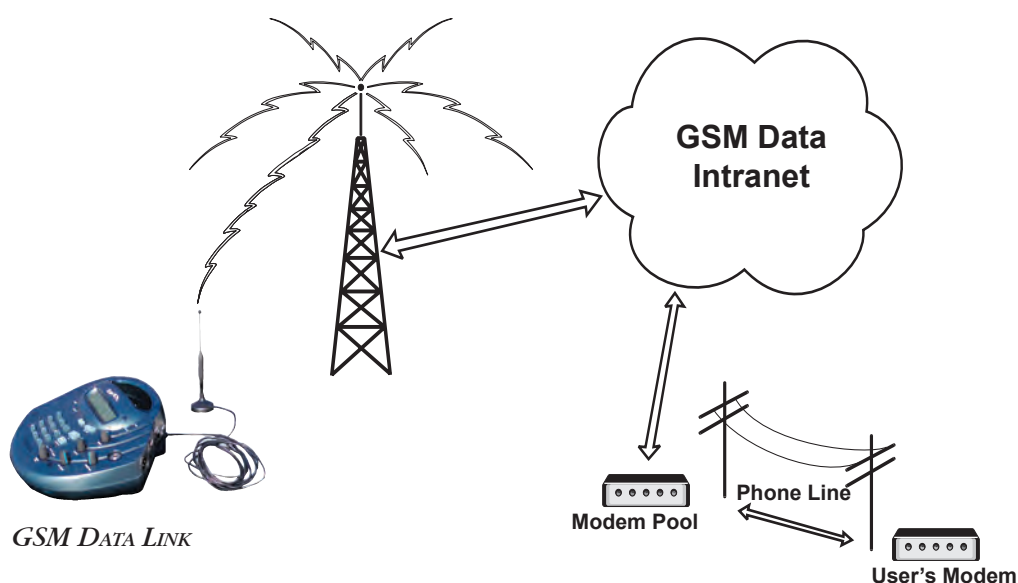
Our hope is you will understand these limitations and choose the appropriate applications for the GSM Module. We at Comrex will be hard at work on the next Matrix "accessory." Stay Tuned!

ABOUT GSM

Second generation digital cellular networks come in several flavors. The most common network available internationally is known as GSM (Global System for Mobile Communication). There is also a large deployment of AMPS (analog), TDMA, and CDMA networks, especially in North America. Our testing has determined GSM to be the best network for transmission of real-time audio from both a data availability and stability standpoint. GSM also has the advantage of subscription via Smart Card (SIM) so a GSM account can easily be moved from phone to phone (or phone to codec).

ABOUT GSM DATA

When you use the Matrix GSM Module, you are actually “piggybacking” on the digital stream normally used for wireless voice calls. The difference is that instead of converting the wireless data to analog and routing it down a POTS line, the GSM network is feeding this data into a normal analog modem. This modem resides somewhere in the GSM network, possibly thousands of miles away from both ends of the link. The modem then places a call to the destination and negotiates a data link over the phone line. As shown in the diagram below, there are actually three “daisy chained” digital links: The first is the wireless link between the GSM Module and the tower, the second is the circuit from the digital tower to the network modem pool, and the last is the normal modem connection between the modem pool and the POTS side of the call.



This type of connection is called Circuit Switched Data (CSD), because an actual physical circuit is established and kept open throughout the call, much like a normal phone call is.

*OTHER TYPES OF WIRELESS
DATA*

GSM (and most other types of digital wireless service) have newer generations of data offerings. In the GSM world, the most common is known as GPRS (usually some spiffy marketing name is applied to it). GPRS boasts data speeds much higher than those available via CSD. But GPRS, being designed primarily for web traffic, has serious deficiencies where real-time audio is desired. GPRS is a web-based connection, meaning that rather than dialing a remote point; you are connected directly to the carrier's web server. Any connections to remote locations must be virtual and packet-based along the public Internet, with all the congestion and delays associated with that. Also, deployed GPRS systems typically provide a wider download speed than upload, and remote broadcasting relies on the upload path in a wireless environment. So for now, "state of the art" wireless broadcasting will rely on the older, thinner, circuit switched data connections provided by GSM. The same limitations apply to high-speed offerings by CDMA systems, typically known as "1X" networks.

GSM MODULE OPERATION

Please refer to the "Matrix GSM Module" manual for more information and operating instructions.

SECTION 9.

ADDITIONAL FUNCTIONS

TEST

The *TEST* function allows the user to test the encoder and decoder portion of the Matrix and to check out audio levels by “looping” the input audio back through to the output. It also gives the user the ability to listen to audio at various data rates. If you are not familiar with the audio connections on the Matrix, read more about these on pages 20-23 of this manual.

To initiate the *TEST* function, first feed audio into the Matrix at a typical level, and adjust the appropriate *INPUT* control until the Matrix Level Indicator or VU Meter reads correctly. (For portable, see pages 22 & 23 and for rackmount, see pages 20 & 21.) Connect the Matrix output to your monitoring facility. On the Portable Matrix, turn the *OUTPUT MIX* knob all the way to the right (clockwise) so that no local (input) audio is heard.

To enable the *TEST* function:

- A. Press *ENTER* from the *MAIN STATUS DISPLAY*.
- B. Next press “3” for *TEST*.
- C. Select the coding type you wish to use: “1” for Voice, “2” for Music, “3” for HotLine or “4” for GSM.
- D. Select a data rate by pressing the corresponding number on the keypad.
- D. When the *READY* LED comes on, adjust the *HEADPHONE* or the *OUTPUT* level control until you have a comfortable level in your headphone and/or monitor.
- E. To exit *TEST*, press the *CANCEL* key.

ANCILLARY DATA

The Matrix provides a 300 bps asynchronous data channel when connected in Voice Mode at a data rate of 16.8 kb/s and above. 300 bps was chosen as the ancillary data rate because it allows us to steal only one byte from each audio frame, having little impact on audio quality. This speed is perfectly adequate for sending and receiving text characters but will prove quite lacking in applications requiring graphics or any other “high-overhead” data transmissions. If your applications require higher data throughput, it may be best to use another telephone connection with a normal modem.

Ancillary data is provided on the Matrix through the 9-pin “D” connector on the rear panel labeled *ACCESSORY PORT*. Pinouts for the *ACCESSORY PORT* are shown in the specifications section on pages 69 of this manual and should mate correctly with a 9-pin PC style serial port. Use a “straight through” data cable. A “null modem” cable will not work. More information on setting up an ancillary data connection is outlined in the “About Ancillary Data” on page 64 and “Troubleshooting the Ancillary Data Channel” on page 66.

CONTACT CLOSURES

Two contact closures are available on the Matrix.

The latching contact closure labeled *READY*, available through the $\frac{1}{8}$ " mini jack on the rear panel of the Matrix, will remain closed as long as the green *READY* light is illuminated on the front panel .. i.e. as long as the call is connected.

There is also a momentary "dry closure" available through the $\frac{1}{8}$ " mini jack on the rear panel of the Matrix, marked *CC*. To activate this closure, press the *ENTER* key when two Matrixes are on line.

While this key is pressed, two things will occur:

- 1) A contact closure will be made silently across the *CC* terminal on the remote end.
- 2) An asterisk will be present on the LCD of the unit on the other end.

Use this feature if you wish to provide an indication to the other Matrix user at any time during the audio transfer or to activate a device remotely through the far end Matrix. The contact closures will have no effect on audio quality.

Rackmount Note: A mini jack is available on the Rackmount Matrix which allows the user remote control of the *CC* button on the front keypad. When the Matrix is on-line, a closure here will send a signal to the other end and close the *CC* receive relay and place an asterisk on the LCD of the remote Matrix. There is also a mini jack labeled *AUX* which may be used to trigger the STL backup function.

MATRIX CONTROL AND DATA SIGNALS

The Matrix provides contact closures which can be used to trigger automated equipment on the far end of the audio link. This type of signal can be useful to the user but has its limitations. Because of the uncertain nature of telephone lines, it is possible that the contact closures can be corrupted if errors are present on the telephone line. If your requirements are for control over on-air automation systems, this can cause problems. We would recommend under these circumstances that you use a different phone line and attach a normal computer modem, which will provide higher speed as well as a higher probability of an error free link. Another approach would be to use external logic circuitry that requires the closure to present for at least 1 second - something that random errors would be unlikely to create.

STL BACKUP MODE

The Matrix has a special feature that allows the unit to automatically initiate and maintain a POTS codec call. When engaged, the Matrix will dial the number stored in *QDIAL 01*, and redial the call should the line drop for any reason. STL Backup mode works slightly differently in the Matrix Rack than in the Matrix Portable. Essentially, the Matrix Rack STL Backup dialing is triggered by the Aux input closure on the rear panel. On the Matrix portable, the stored number is dialed immediately upon entering STL Backup mode.

STL Backup mode is engaged on both units from the *HIDDEN MENU*. (See *HIDDEN MENU* section on page 50.) This menu is hidden to prevent casual users from accessing it and setting parameters that will cause the Matrix to malfunction.

*STL BACKUP MODE FOR
MATRIX RACKMOUNT*

On the Matrix Rackmount, you may still access other menus once STL Mode is engaged. But the unit must be left at the “*STL READY TO DIAL*” for STL mode to trigger correctly. When a closure is made between the pins on the Aux connector, the Matrix Rackmount will dial the phone number stored in *QDIAL 01*. If the number is busy or doesn’t connect, it will continue to try to dial until the Aux connector is opened. Once connection is made, the Matrix Rackmount will monitor the Aux connector’s status. If the connector opens, the Matrix Rack will drop the call and go back to waiting for the next closure. To disengage STL Backup Mode on the Matrix Rackmount, open the Aux connector so the unit will drop the call and not try to redial. Enter the *HIDDEN MENU* again, choose STL, and select option 2 “Disengage STL Mode.” Normal operation will resume.

*STL BACKUP MODE FOR
MATRIX PORTABLE*

On the Matrix Portable, once STL Backup mode is engaged, it is not possible to access the configuration menu system. For this reason, it’s important to configure all parameters and Qdial locations before entering this mode. No external trigger is provided on the Matrix Portable. Once engaged, the Matrix Portable will drop directly into dial mode as soon as the *HIDDEN MENU* is exited. Disengaging STL mode on the portable can only be done during the few seconds the Matrix is getting ready to dial the STL number. The display will read:

STL Ready to Dial
Cancel = Exit

Press the *CANCEL* key at this point to disengage STL mode. If the portable Matrix is on-line, press the *HANGUP* key to end the call, and then press the *CANCEL* key when the above screen re-appears.

SECTION 10.

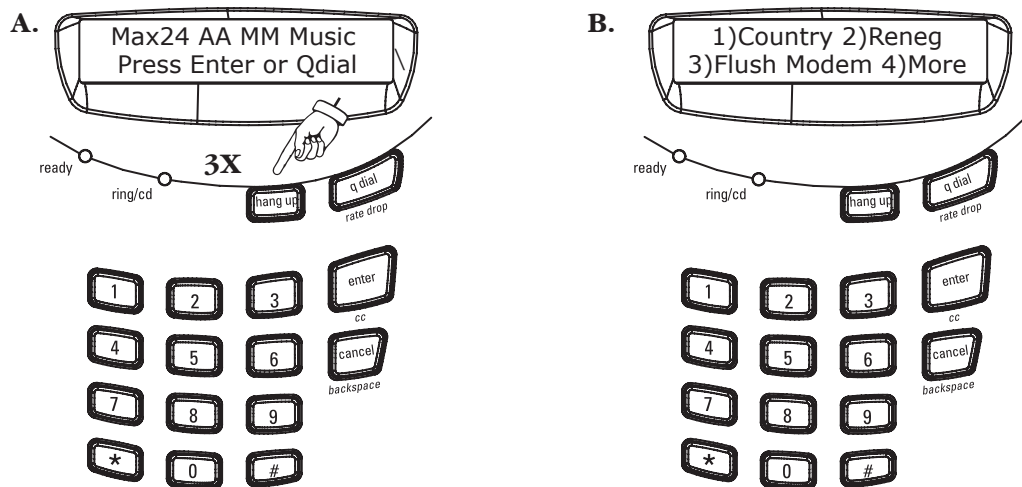
HIDDEN MENU

The Matrix has several reliability enhancing features built-in. Some of these, such as “Rate Drop,” “Retrain” and “MaxRate,” are designed right into the main configuration and operation menus. Some others are hidden within the “Hidden Menu” to prevent access by casual users. This is because these configurations (in some situations) may cause as many problems as they prevent. But they are included because, in our experience, having access to these modes has saved an occasional remote broadcast.

The important thing to remember about these modes is that once they are selected, they will stay that way until they are reset via the Hidden Menu, or when a “Modem Flush” is performed. If you ever suspect that the Matrix has had one of these (unwanted) modes enabled, it may be valuable to issue a “Modem Flush” to clear out any unwanted configuration information. (See page 61 for details.)

To Access the Hidden Menu:

- A. From the *MAIN STATUS DISPLAY*, press the *HANGUP* key 3 times.
- B. The *HIDDEN MENU* choices appear on the screen.



REVISION INFORMATION

To show the current software version contained in the Matrix:

- A. Enter the *HIDDEN MENU* as described above.
- B. Select "4" for *More*.
- C. Select "3" for *REV*, and the current software version will show on the screen for approximately 4 seconds.

MODEM COUNTRY CODES

00.....NORTH AMERICA
 01.....AUSTRIA
 02.....BELGIUM
 03.....DENMARK
 04.....FINLAND
 05.....FRANCE
 06.....GERMANY
 07.....IRELAND
 08.....ITALY
 09.....SINGAPORE
 10.....NETHERLANDS
 11.....NORWAY
 12.....PORTUGAL
 13.....SPAIN
 14.....SWEDEN
 15.....SWITZERLAND
 16.....UK
 17.....GREECE
 18.....AUSTRALIA
 19.....JAPAN
 20.....CTR21

The Matrix's modem can be user configured for operation in twenty different countries. Once configured, the Matrix will remain in this setting until the country parameter is changed. Country configuration is confirmed after selecting *COUNTRY* from the *HIDDEN MENU*, where the Matrix's display will show the current configuration (*CONFIGURED FOR NORTH AMERICA*, etc.). Country configurations change dialing parameters, such as DTMF level and pulse and ring cadence, as well as call progress tone detection, including dial and busy tones. The factory default setting is for North America.

To change the country configuration:

- A. Enter the *HIDDEN MENU* as described on the previous page.
- B. Select "1" for *COUNTRY*. Current country configuration will appear momentarily.
- C. Enter the appropriate two digit number from the chart on the left. The Matrix will go through an initializing process and confirm the country selection on the display screen before returning to the *MAIN STATUS DISPLAY*.

Note: The country parameter only needs to be changed when dialing from a country, not to it. Some experimentation may yield a listed code which works well in countries not listed here. Also, dialing from an external phone is a good method to avoid issues with international phone systems altogether.

RENEGOTIATION & RETRAIN OPTIONS

In most cases, this option will only be used at the field end. We have found some locations (sports venues in particular) are quite unkind to modem data. Often, EMI will affect the venue phone lines, and cause POTS codecs to renegotiate or retrain at particularly inconvenient times. Renegotiation and retrain are two tools the Matrix uses to evaluate telephone line conditions. When the Matrix renegotiates, it's quickly determining whether a lower data rate is necessary due to short bursts of noise or errors on the phone line. Retrain is a "full negotiation sequence" of the connection, resetting and adapting the phone line parameters of the modem to the changing parameters on the phone line. There are three options for controlling the renegotiation and retrain functions:

1) *RENEG/RETRAIN ON* — This is the factory default option, allowing for the Matrix to monitor the telephone line quality and renegotiate or retrain should the telephone line degrade. To minimize the number of errors during an audio feed, the user can set a *MAXRATE* and/or force a *RATE DROP* (For more on *RENEG/RETRAIN*, *MAXRATE* and *RATE DROP* see page 31 & 56-57).

2) *RETRAIN ON* — By selecting this option, you're instructing the Matrix to "reset" all the phone line modem parameters automatically when it senses lots of errors on the telephone line. Again, please refer to page 31 & 56-57 for further details.

RENEGOTIATION & RETRAIN
(CONT.)

3) *NONE* — This option instructs the Matrix not to renegotiate or retrain automatically when errors occur on the telephone line. The result will be that the return audio coming back from the studio may become unusable, but in one-way program transmission applications it doesn't make sense to renegotiate or retrain on errors in the direction you're not using. Note that if errors are prolonged and substantial, the Matrix will probably drop the line eventually due to insufficient carrier between the modems. But we have found, particularly in sports venues, that short bursts of telephone line noise are present when the game gets exciting, due to PA, lights and scoreboard changes. In this circumstance, the "Disable Renegotiations & Retrains" setting may improve things. This setting is particularly dangerous to leave engaged for the poor sucker on the next remote. For this reason you will see an asterisk (*) after the codec mode on the *MAIN STATUS DISPLAY*, if it is engaged.

To Select Renegotiation and Retrain Options:

- A. Enter the *HIDDEN MENU* (press the *HANGUP* key 3 times).
- B. Select "2" for *RENEG* (Renegotiation Menu).
- C. Enter "1" for *RENEG/RETRAIN ON* or "2" for *RETRAIN ON* or "3" for *NONE*.

FLUSH MODEM

This is a way to reset the Matrix to factory default mode. It clears out the Matrix NV Ram, which deletes all configuration settings, including *MAXRATE* and *QDIAL* entries. It's like getting the Matrix new out of the box. To flush your modem, access the *HIDDEN MENU* by pressing the *HANGUP* key three times. Then select *FLUSH MODEM*, and confirm your selection. After a few seconds, you will see the normal menu screen. **The Matrix must be powered off for a few seconds after the modem flush.** When you turn it back on, the Matrix will be "factory fresh."

***Note: After a modem flush, it is important to re-enter your country modem code via the hidden menu (see "Modem Country Codes" on page 51). This is because the modem maintains the old country settings during a modem flush, but the NV Ram in the Matrix has been wiped out, which contains the info as to what country to display at power up. So to synchronize the two, re-enter your country information after the modem flush.*

STL BACKUP

The STL Backup function is described in detail on page 49. Both portable and rackmount units are set for this mode in the same manner:

To Enable or Disable the STL Backup Mode:

- A. Enter the *HIDDEN MENU* as described at the beginning of this section.
- B. Select "4" for *MORE*.
- C. Select "2" for *STL*.
- D. Select either "1" to *ENABLE STL MODE* or "2" to *DISABLE STL MODE*.

DIAL TONE

Usually, it is advantageous to have the Matrix wait to sense dial tone on the line before initiating a call. This means if your line is unplugged or out of service, you'll know pretty quickly. Also, some phone lines have delayed dial tone and won't accept dialing digits until a few seconds after the line is picked up. But you may find yourself on a phone line that delivers dial tone that isn't sensed by the Matrix. This may happen in countries that aren't provided for in the Matrix country code options or when used in conjunction with some PBXs (yes, we know, we told you not to use them). So make this setting change if you encounter these lines, but be sure to change it back before moving the Matrix to another location.

To Require or Ignore Dial Tone:

- A. Enter the *HIDDEN MENU* as described at the beginning of this section.
- B. Select "4" for *MORE*.
- C. Select "5" for *DIAL TONE*.
- D. Select either "1" to *REQUIRE DIAL TONE* or "2" to *IGNORE DIAL TONE*.

SYMBOL RATE

The telephone line interface in the Matrix is via a V34 (33.6 kb/s) modem. V34 is a very complex modem specification but in a nutshell, provides for negotiation, equalization and rate selection between compatible equipment. One of the parameters chosen during the negotiation phase is one of five possible carrier "symbol rates." Note that this choice is related to, but not the same as, the data rate chosen. We have found that on some phone lines, especially when connecting to older POTS codecs like the Vector and HotLine (which use a different but compatible modem), it is helpful to disable the ability to connect at the highest symbol rate (we call this Symbol Rate A). So the Matrix disables this symbol rate by default. If you are using Matrix in pairs, and not mixing other Comrex codecs in, you may find higher data rate connections by re-enabling Symbol Rate A via the *HIDDEN MENU*. Another option exists in the menu to disable the two highest symbol rates (A & B). This may help on particularly unusual telephone connections, but should be done on both codecs to connect reliably.

To Disable or Enable Symbol Rates:

- A. Enter the *HIDDEN MENU* as described at the beginning of this section.
- B. Select "4" for *MORE*.
- C. Select "4" for *SYMBOL RATE*.
- D. Select either "1" to *DISABLE A* (the top symbol rate) or "2" to *DISABLE A&B* (The top two symbol rates) or "3" to *ENABLE ALL RATES*.

ISDN (MATRIX RACK ONLY)

Please refer to page 12 of the "Matrix ISDN Module" manual for details.

GSM14

This option is for use outside of North America. Certain European Networks support 14.4 kb/s GSM data transmission. Both codecs must have this option enabled for proper operation.

SECTION 11.

UPGRADING THE MATRIX

Upgrades may become available for the Matrix as new features are added or new software is written. A list of available upgrades can be obtained from Comrex Tech Support or our website at www.comrex.com.

*HOW TO UPGRADE THE
MATRIX CONTROL PROCESSOR*

The Matrix gives the user the ability to update the firmware that drives the main control processor. This procedure, also known as “flashing” the Matrix, requires that you have access to:

- 1) A PC running a terminal emulation program capable of Xmodem uploading. (Windows 98 Hyperterminal and Procomm 95 have been tested.)
- 2) A 9-pin male to 9-pin female straight through serial cable or a 25-pin female to 9-pin male “modem” cable with the pin-outs on page 69.
- 3) A Matrix control upgrade file (MatrXXX.bin where XXX is a three digit number). This file can be downloaded from the Comrex website or can also be emailed to you by our tech support staff.

Note: Once the Flash memory in the Matrix is erased, you won’t be able to use the Matrix until the upgrade procedure is complete. Therefore, don’t go past step 6 below if you’re unsure of either your upgrade file or your terminal software.

To Flash Upgrade the Matrix:

- 1) Run the emulation program, and set it to communicate directly to the desired serial port (usually com1 or com2).
- 2) Set the serial port parameters as follows:
Baud Rate – 57600 (57.6 Kbaud)
No parity, 8 data bits, 1 Stop bit (n,8,1)
No handshake required (Software handshake MUST be off; hardware handshake shouldn’t matter.)
- 3) If your emulation software has an option to preset the file transfer protocol, select Xmodem.
- 4) Connect the serial cable to the Matrix *MULTI-PURPOSE* port. Connect the other end the to the proper serial port on your computer
- 5) Turn on the Matrix. Quickly, while the “Initializing” display is on, press the *HANGUP* key 3 times. The display will then say “Flash Upgrade Mode.”

- 6) In about 5 seconds, the following words should appear on your computer display:

COMREX MATRIX
FLASH UPGRADE MODE
ALL CONTROL SOFTWARE WILL BE DESTROYED
ARE YOU SURE? (Y/N)

If these words don't appear, try pressing the *ENTER* key on the computer. If you still don't see the text, turn off the Matrix and recheck all connections and settings. No change has been made to the Matrix firmware at this point.

- 7) If you are certain you have the correct upgrade file, and you are certain your program supports Xmodem, press the Y key on your computer. You will then see:

ERASING...

And then:

BEGIN XMODEM TRANSFER

- 8) Using your terminal software, send (or upload) the file using the Xmodem protocol. Don't use Xmodem-CRC or Xmodem-1K, simply Xmodem. There's no rush to do this, as the Matrix will wait until you begin sending the file. Also, there may be a ten second delay before the file transfer actually starts. If there is an error or power failure during this process, simply reboot the Matrix and enter flash upgrade mode again. Your Matrix will still be able to enter flash upgrade mode, but it won't work in its regular mode until the upgrade procedure is completed successfully.

When the upgrade has successfully been completed, the following message will be displayed on your computer monitor:

UPGRADE COMPLETE!

On completion, the Matrix will resume normal operation, showing the *MAIN STATUS DISPLAY*.

SECTION 12.**HOW TO ACHIEVE OPTIMUM MATRIX PERFORMANCE**

For those who have used digital telephone technology (like ISDN), good data performance is taken for granted. This is because the information transmitted is kept in digital form from end to end and is, therefore, somewhat impervious to the nasty effects of telephone carriers. When using analog modems, however (as in the Matrix), data is converted to analog tones which are sent through the telephone channel. These tones are much more likely to be affected by the limitations of the telephone channel.

Moving high speed data along analog telephone lines can best be described as a “balancing act.” It requires sophisticated modem technology which senses the quality of each telephone connection and then dynamically adapts all kinds of parameters in short order to “clean up” the connection. These parameters are set and locked during the “negotiation sequence” which takes place shortly after the Matrixes are connected together. This is the hash sound you hear when the modem monitor is enabled. During this sequence, the modems also measure bandwidth and noise level on the phone connection to determine the highest data rate which may be supported.

This adaptation is usually done quite well, and the modems can usually be trusted to optimize for a reliable data rate and a solid connection. If the adaptation to the phone call is less than perfect, errors will result. Errors manifest themselves in the Matrix audio as occasional clicks, dropouts or frame repeats (almost like a “cd skipping” sound). Errors may occur more on long distance calls than local ones and are often a problem when connected to in-house phone systems (another reason to bypass them). A much more detailed discussion of issues encountered on telephone line modem connections is found in Appendix A, “Modem Line Check List,” on pages 72-76.

To address this potential problem, the Matrix has features which allow the user to minimize the number of errors during an audio feed. Each has its advantages and often can be used in conjunction with each other to fix any error problems which might occur. Either forcing a *DROP RATE* (page 31), or full renegotiation of the connection data rate are the first and most obvious steps to take when experiencing errors during program transmission. We also recommend that you look at the actual line you will be using and set a *MAXRATE*, particularly if the connection is to be made regularly.

The best maximum rate setting can be determined by dialing the connection with no MaxRate selected and seeing the rate at which the modem negotiates. If the connection appears to be error free, you may wish to select the next rate down as a MaxRate just to provide some insurance. If you experience errors, it may be best to set it a couple of notches below the unrestricted connect rate. (See page 24-25 for further details on setting the MaxRate.)

The Matrix has built-in error correction in the Voice Mode. Since the audio bandwidth in Voice Mode is limited to 7 kHz, we have been able to use the leftover data rate for error correction. If errors become objectionable in Music Mode, you may wish to sacrifice some audio bandwidth in favor of a more reliable connection. (See “Audio Coding Selection” on page 27 for instructions on how to select Voice Mode.)

Note: Under most circumstances, the options described above should normally not be necessary. The Matrix contains the most sophisticated modem technology available and will negotiate the highest reliable speed. No modem can, however, predict what will happen to a telephone connection in the future. Rate Drop, MaxRate and Voice Mode settings are provided as tools to enhance the reliability of the phone network. Please understand that POTS coding has its inherent limitations and risks, and that the Matrix was designed specifically to minimize these problems. These problems exist in the phone network, however, not in the Matrix.

SECTION 13.

SOME OPERATING TIPS - (STUFF YOU REALLY NEED TO KNOW!)

1. Use direct, outside telephone lines for best performance.
2. A line used by a fax machine usually provides this direct access. (Be sure to disconnect the fax machine before connecting the Matrix!)
3. Check to see that there are no extensions or modems on the line you are using — or at least arrange that no one uses these during your broadcast.
4. If there is call-waiting on your line, disable it by entering “*70” in front of the number you are dialing.
5. If possible, try the Matrix out at the remote site before your actual broadcast, at about the same time of day that you plan to use it. This will give you a good idea of expected connect rates and possible line problems.
6. At minimum, connect a few minutes before airtime to assess the connection quality. Setting a MaxRate (see pages 24-25 & 56-57) based on your findings is highly recommended.
7. Make sure that a “mix-minus” feed is available from the studio to the remote site so that the remote talent will not hear themselves “echoed back.” See Appendix B on pages 77-78 for more information on mix-minus.
8. If operation starts to degrade after a long period of connection, it may be that the phone line parameters have changed over time, and modems should be given the opportunity to renegotiate for these new parameters. The easiest way to do this is press the *QDIAL* key twice. This triggers the full modem handshake process, without disconnecting the line. Audio will be muted for approximately 10 seconds during this process. You can also disconnect the call and redial. It is not a bad idea to have the modems re-handshake every hour or so, during program breaks. This will provide the Matrix with a “fresh start” on negotiating the best connection, since the line evaluation only occurs during the initial handshake.

OPERATING TIPS (CONT.)

If you experience low connection rates or errors, try the following:

Low Connection Rate (or no connection at all):

- A. Try redialing. If a good connection is found, keep that line up.
- B. Dial from the other end.
- C. If the call is long distance, try forcing to another carrier.
(See next section.)
- D. Verify that you have a direct, outside connection and that there is nothing else connected to that line.

Errors:

- A. Set a MaxRate before your feed. (See pages 24-25 & 56-57.)
- B. Renegotiate to a lower connect rate with *DROP RATE*.
(See page 31 & 56.)
- C. Change to Voice Mode for additional error correction.
(See pages 27 & 57.)
- D. Refer to the "Modem Line Check List" in Appendix A.
(See pages 72-76.)

*USING LONG DISTANCE
ACCESS CODES*

In our field tests with the Matrix, Vector, BlueBox and HotLine POTS codecs, we have seen a wide variety in the quality of connections, particularly in long distance calls. This makes sense, because there is no way of predicting how a call will be routed by the telephone companies involved. We looked for meaningful statistical results from our trials which would show an advantage in using a particular long distance carrier and did not come up with anything concrete. What we did find, however, is that it may help to be able to force a different routing, if your particular connection is not good. To find out which long distance access has been selected for a given line, dial 0-0. The above said, we have found, in general, that you "get what you pay for." We have seen some highly discounted long distance consolidators that seem to provide consistently low data connections. If your station is using "Joe's Long Distance Phone Service" you may well find much better connect rates with a mainstream carrier.

Here is a list of some commonly used long distance carriers and their access codes. To force a particular long distance carrier, simply dial the access code number, followed by the telephone number you would normally dial. For example, to dial Comrex Corporation's main number (978 784-1776) via AT&T, you would enter 101028819787841776.

AT&T	1010288	Sprint	1010333
MCI	1010222	Frontier	1010444

*INTERNATIONAL AC POWER
CORDS AND TELEPHONE
CONNECTORS*

AC adapter cords are available which will connect the standard IEC 320 Inlet on the in-line portable Matrix switching supply and the built in rackmount switching supply to a wide range of international power receptacles. These may be purchased from:

Panel Components Corp.	Tel: 800-662-2290 (USA)
P.O. Box 115	641-673-5000 (INT)
Oskaloosa, IA 52577-0115	Fax: 800-645-5360 (USA)
	641-673-5100 (INT)

One solution for international telephone connections — look for the fax machine. Fax machines provide direct line connections and use standard RJ11 modular jacks, so you can simply unplug the fax machine and plug into the Matrix. It may be necessary to change the modem country code (see below).

*DIALING THE MATRIX
INTERNATIONALLY*

MODEM COUNTRY CODES

00.....NORTH AMERICA
01.....AUSTRIA
02.....BELGIUM
03.....DENMARK
04.....FINLAND
05.....FRANCE
06.....GERMANY
07.....IRELAND
08.....ITALY
09.....SINGAPORE
10.....NETHERLANDS
11.....NORWAY
12.....PORTUGAL
13.....SPAIN
14.....SWEDEN
15.....SWITZERLAND
16.....UK
17.....GREECE
18.....AUSTRALIA
19.....JAPAN
20.....CTR21

The Matrix's modem can be user configured for operation in twenty different countries. Once configured, the Matrix will remain in this setting until the country parameter is changed. Country configuration is confirmed after selecting *COUNTRY* from the *HIDDEN MENU*, where the Matrix's display will show the current configuration (*CONFIGURED FOR NORTH AMERICA*, etc.). Country configurations change dialing parameters, such as DTMF level and pulse and ring cadence, as well as call progress tone detection, including dial and busy tones. The factory default setting is for North America. The country parameter only needs to be changed when dialing from a country, not to it. Some experimentation may yield a listed code which works well in countries not listed here. Also, dialing from an external phone is a good method to avoid issues with international phone systems altogether.

To change the country configuration:

- Enter the *HIDDEN MENU*, press the *HANGUP* key three times.
- Select "1" for *COUNTRY*. Current country configuration will appear momentarily.
- Enter the appropriate two digit number from the chart on the left. The Matrix will go through an initializing process and confirm the country selection on the display screen before returning to the *MAIN STATUS DISPLAY*.

Never heard of a country called CTR21? Neither have we. Actually it's a modem specification designed to work ubiquitously in many European countries. Go ahead and try it in Europe, and if it doesn't work, curse those European standards bodies (like we often do) and then go back to the correct country setting. The term "European Standard" is still held by many to be an oxymoron.

*COMPUTER MODEMS VS.
MATRIX MODEM*

Some users attempt to compare the connect rate message they receive when using their computer modem to that which is produced by the Matrix. Often, they find the Matrix message reports a dramatically lower connect rate than their computer. This is because most computer modems default to reporting the speed between the computer and the modem, rather than the actual speed between the modems. Most computer modems can be programmed to report the actual connect speed using the following procedure:

Using Procomm[™], Windows[™] terminal, or some other communications program, get the attention of your modem:

Type:

AT

The modem will respond:

OK

Now Type:

ATW1

The modem should again respond:

OK

If you want to make your modem always work this way type:

AT&W

And the modem will respond:

OK

Now the modem should report the actual connect speed, which is more likely to compare to Matrix speeds on similar circuits. Note: The initialization strings of some programs (like internet access or on-line service programs) may delete these settings in their initialization strings. You may need to edit the initialization string in your program by removing any reference to Wx in the string, where x is any number. Don't remove it if the W is preceded by a symbol (such as & or !).

SECTION 14.**TECHNICAL DETAILS***TROUBLESHOOTING THE
MATRIX*

Because of the delicate nature of moving audio data over telephone lines, it is likely that you will experience problems at some point with establishing a reliable connection using the Matrix. There are dozens of factors that can affect the success or failure of a POTS codec call, some within the user's control and some not. Comrex has support personnel to help troubleshoot problems which may occur, but please use this section first to "run down" the most common issues when using the Matrix.

1) Are you on an in-house phone system?

There are only so many times we can say this... an in house phone system is almost always a bad idea with the Matrix. If at all humanly possible, use the Matrix on a direct telephone company line. If you call for support, and you are using an in-house phone system, the first request you are likely to receive is to move the Matrix to a direct, "Ma Bell" line.

2) Have you checked your audio quality going into the Matrix?

Often problems with noise or distortion in audio are added before the Matrix, but since the Matrix is the most mysterious link in the chain, it is assumed to fail first.

Check your audio in and out of the Matrix locally, by listening to the "local" program output. Any noise or distortion present in the Matrix audio input will be present on this output. Next, enter the *TEST* Mode (see page 47), and then select a loopback rate similar to the rate at which you have been able to establish a connection. This will allow you to monitor audio after it has gone through the encoder and decoder section. (On the Portable unit, turn the *OUTPUT MIX* knob clockwise as far as it can go, so that no local audio is heard.) Due to the extremely high compression of the Matrix, some artifacts due to the coding are unavoidable, and these artifacts seem more apparent listening through headphones than through a studio monitor. Also, check that your input level is correct (the level indicator on the Matrix Portable should be green, flashing red on program peaks only, or the VU meter on the Matrix Rackmount should peak occasionally over the "0" mark, which is the yellow LED. You may also wish to experiment between the Music and Voice Modes (see page 27), as some audio material may sound better in one mode than the other.

3) Have you disconnected all other devices from your lines?

Be absolutely certain that other equipment, such as fax machines, modems, etc. are removed from your telephone line on each end, not simply disabled. Even "on-hook," some devices can affect the Matrix performance. Also, their removal will prevent anyone from inadvertently "picking up" the line during audio transfer and causing the Matrix to drop out.

4) Have you tried redialing and/or changing lines or carriers?

Some telephone connections simply won't support the movement of data at the rates required by the Matrix. If you have trouble establishing a reliable connection even with a rate drop, try redialing several times.

The telephone systems often route calls differently each time, and you may pick up a cleaner circuit on a different call. The problem may also be that your local loop runs a very long distance, is subject to crosstalk, or is "loaded" by the phone company, causing the modem in the Matrix (or any modems, for that matter) to perform poorly. If possible, try a different line. Finally, we find a change in long distance carriers (assuming you are dialing long distance) can make a world of difference. Try dialing the access code of a different carrier if you have difficulty. See page 59 for some access codes. In general, we find better connections with the larger carriers (avoid "Joe's phone company" if possible).

Phone line performance can be affected by factors such as time of day, weather and geographic location. Once you have used the Matrix, you will probably gain a better understanding of how it works on different lines and what can go wrong.

5) Using the Matrix with a Comrex HotLine or Vector:

Current information on achieving the best performance when connecting to our other POTS codecs may be found in Appendix D on page 81.

6) Have you read the following?:

Section 12 - How to Achieve Optimum Matrix Performance
(pages 56-57)

Section 13 - Some Operating Tips (pages 58-61)

Appendix A - Modem Line Checklist (pages 72-76)

The information contained on these pages is CRITICAL for successful use of a POTS Codec.

If you call Comrex for support, we will likely attempt to connect to each of your Matrixes from our office. This way, by connecting from a known good line and carrier, we can often at least isolate which end of the connection is causing the difficulty. To save your time and ours, please be sure you have run through the above checklist before calling for support.

ABOUT ANCILLARY DATA

The Matrix provides an ancillary data channel, when set in Voice Mode, to send low speed data along the same digital telephone channel used for the Matrix audio. The vital information you need to know is that this channel has the following parameters:

300 Baud; 8 bits; no parity; 1 stop bit

Now, we will describe a few concepts:

The Matrix ancillary data channel is asynchronous. This is the most common format for information exchange between computers. An asynchronous data link simply provides a “pipe” which passes bits back and forth between the devices. What you do with this data is dependent completely on the computers and software used in the connection.

It is the nature of asynchronous data that we define a “baud rate.” This is the rate at which individual bits run along the asynchronous link. In some asynchronous systems, however, a continuous stream of data at the specified baud rate will overload the system. This is because the baud rate specifies only the speed on the pipe feeding into the system. Further down, the pipe may narrow and less throughput is possible. Asynchronous communication allows flow control where the sending device will be triggered by the network when enough capacity is available to send more information. In the Matrix, the “pipe” has the same throughput all the way across, and flow control is not necessary.

Flow control works because asynchronous links do not need to send information at all times. When there are no characters for a computer to send to a modem, for example, the asynchronous link is idle. When the computer has information to send, it will usually format this information into one or more bytes, attach a start and stop bit, (so the receiving system knows where the byte begins and ends) send it off and again make the line idle. In the most common application for ancillary data (two terminals with operators sending text characters back and forth), the asynchronous link is active only a small fraction of the time. During a file download, however, the link will most likely be constantly active, with one byte being sent immediately after the next (unless flow control is active).

Here is a description of a typical ancillary data hookup:

At least one Matrix must be configured for Voice Mode, and the connect rate on the line must be at least 16.8 kb/s. Each operator will attach a PC com port to the ancillary data connector on the Matrix (via a straight-through 9-pin cable) and load a terminal emulation program such as Windows™ terminal or Procomm™. Each will set the correct com port in software and the communications parameters to 300 baud, 8 bits, no parity, 1 stop bit. Flow control should be turned off.

In most ways, the link will resemble a normal modem connection. When a key is typed on one terminal, the ASCII byte corresponding to that character will be sent out the computer com port to the Matrix. The Matrix will embed this character into the data it is sending, and the Matrix on the far end will extract this character from the incoming data. It will then send the character to the other computer com port, and it will appear on the other display. Of course, since the channel is full duplex, this exchange may be happening in both directions simultaneously.

A few common options in terminal programs will ease communication:

a) Local echo — Engage this if you wish to see what you are typing on your own display. The Matrix cannot echo the characters you send back to your display, only send them to the other end. You must configure your communications program to do this.

b) Cr-> CR-LF — When you type a carriage return (enter key), the CR character is all that is usually sent through the Matrix data link. Your program will likely interpret this correctly and send the cursor back to the beginning of the same line. Your program can probably be set to interpret the CR character as both a CR and LF (line feed) sending the cursor to the beginning of the next line. Your program can also usually be altered to send both characters when the “enter” key is pressed. Usually only one of the above options is required.

c) Other options — The software setup of your communications program may require additional parameters. Keep in mind that the Matrix data channel is simply a “pipe.” What bytes are sent and how they are interpreted is completely dependent on the software being used and how that software is configured.

Note: The ancillary data functions on both the Matrix and Vector are identical, and these units will inter-operate in this mode. Also, because of the uncertain nature of telephone lines, it is possible that the ancillary data can be corrupted if errors are present on the telephone line. If your requirements are for control over on-air automation systems, this can cause problems. We would recommend under these circumstances that you use a different phone line and attach a normal computer modem, which will provide higher speed as well as a higher probability of an error free link.

*TROUBLESHOOTING THE
ANCILLARY DATA CHANNEL*

The loopback testing facility in the Matrix can aid in troubleshooting ancillary data connections. It is often difficult to find a fault in a problem like this, since the trouble could be at either end of the link. By loopback testing the ancillary data channel on each end, at least you can easily find which end has the trouble. To loopback test ancillary data:

- 1) Connect a computer serial port to the Matrix.
- 2) Run a program on the computer with basic serial communications functions, such as Procomm™, Windows Terminal™ or Win95 Hyperterminal™.
- 3) Set the communications parameters as follows:
 - 300 baud
 - 8 bits
 - no parity
 - 1 stop bit
 - echo off (half-duplex mode in Hyperterminal)
- 4) Make sure the software has the serial port chosen which is connected to the Matrix.
- 5) Verify that your Matrix has been set to Voice Mode.
- 6) Type some characters on your computer, and verify that you do NOT see them on your screen.
- 7) Go into the *TEST* menu on the Matrix (see page 47), and select a loopback rate of 16.8 kb/s or higher. The *READY* light should come on.
- 8) Type on your computer keyboard. You should see your characters echoed back to you on the screen.

If this test is successful, you have sent data down the cable to the Matrix, into the encoder, looped through to the decoder and back to your computer screen.

If not successful, try the following:

- 1) Type "AT" (Enter). If you get a response "OK" then you are probably talking to a modem inside your computer, configured for the chosen com setting. Try a different configuration.
- 2) Try a different com port. Sometimes these ports "burn out," if connected wrong previously.
- 3) Remove any adapters, "dongles" or other connectors on the port.
- 4) Make sure you are using a "straight through" 9-pin to 9-pin cable. Do not use a null modem cable.

SECTION 15.**SPECIFICATIONS***PORTABLE**Connections*

Audio In: 3-pin XLR female (2)

Tape In: 1/8" 2-conductor mini-jack (1)

Audio Out: 3-pin XLR male (1)

Headphone Out: 1/4" stereo phone jack (1)

Tel Line & Tel Set: 6-pin RJ11C modular jacks (2)

Contact Closures (Ready & CC): 1/8" 2-conductor mini-jacks (2)

Multipurpose Port: 9-pin "D" type connector (1)

*Levels**Audio Input*

Mic Input Levels: -70 to -40 dBu @ 2K ohms

Line Input Levels: -10 to +10 dBu @ 20K ohms

Tape Input: -10 dBu (fixed) @ 10 K ohms

Audio Output

Line Level Out: +12 dBu max @ 100 ohms

Headphone Out: .5 watt @ 100 ohms

Telephone Line Out: -9 dBm @ 600 ohms

Power: External supply; 15V, 2 Amp; 100 to 240 VAC 50/60 Hz

Power Connection: 5-pin mini DIN Connector

Size: 9.5" W x 10" D x 3.5" H

Weight: Net: 2.5 lbs; Shipping: 7 lbs

Nominal Coding Delay: 100 mS

RACKMOUNT

Connections

- Audio In: 3-pin XLR female (1)
- Audio Out: 3-pin XLR male (1)
- Tel Line & Tel Set: 6-pin RJ11C modular jacks (2)
- Contact Closures "Ready" "CC" "AUX" "CC input":
1/8" 2-conductor mini-jacks (4)
- Multipurpose Port: 9-pin "D" type connector (1)

Levels

- Audio Input Levels: -10, 0, +4 dBu (via jumpers to +8) @ 20K ohms
- Audio Output Levels: -10, 0, +4 dBu (via jumpers to +8) @ 100 ohms
- Telephone Line Out: -9 dBm @ 600 ohms

- Power: Internal supply 110 to 240 VAC 50/60 Hz
- Fuse: 315 mA 5x20mm SLO BLO
- Size: 1U Rack 19" W x 8" D x 1.75" H
- Weight: Net: 5.5 lbs; Shipping: 9 lbs

Nominal Coding Delay: 100 mS

AUDIO FREQUENCY RESPONSE
(PORTABLE & RACKMOUNT)

At all connect rates and in all modes, the low frequency response is 20 Hz. The high end response is listed below.

Connect rate	Music Mode	Voice Mode	HotLine Mode
9.6 kb/s	4.7 kHz	4.7 kHz	n/a
12 kb/s	6 kHz	6 kHz	4.5 kHz
14.4 kb/s	7 kHz	7 kHz	5.4 kHz
16.8 kb/s	7 kHz	7 kHz	5.6 kHz
19.2 kb/s	11.2 kHz	7 kHz	5.8 kHz
21.6 kb/s	12.7 kHz	7 kHz	6.5 kHz
24 kb/s	14 kHz	7 kHz	7 kHz
26.4 kb/s	14 kHz	7 kHz	8 kHz
28.8 kb/s	14 kHz	7 kHz	8.6 kHz
31.2 kb/s	14 kHz	7 kHz	9.3 kHz
33.6 kb/s	14 kHz	7 kHz	10 kHz

PINOUTS

Tel Line:

Physical: RJ11C 6-pin Modular Jack

Pin 3: Tip

Pin 4: Ring

Tel Set:

Physical: RJ11C 6-pin Modular Jack

Pin 3: Ring

Pin 4: Tip

Audio Inputs:

Physical: 3-pin Female XLR

Pin 1: Ground

Pin 2: + Audio In

Pin 3: - Audio In

Audio Outputs:

Physical: 3-pin Male XLR

Pin 1: Ground

Pin 2: + Audio Out

Pin 3: - Audio Out

Headphone Output (portable only):

Physical: 3-conductor 1/4" Phone Jack

Tip: L Audio Out +

Ring: R Audio Out +

Sleeve: Ground

Contact Closures and Remote Controls:

Physical: 2-conductor 1/8" mini jack

Contact closures are floating

Input controls have sleeve connected to Ground

Power (portable):

Physical: 5-pin mini DIN type connector

Inner core: +15V

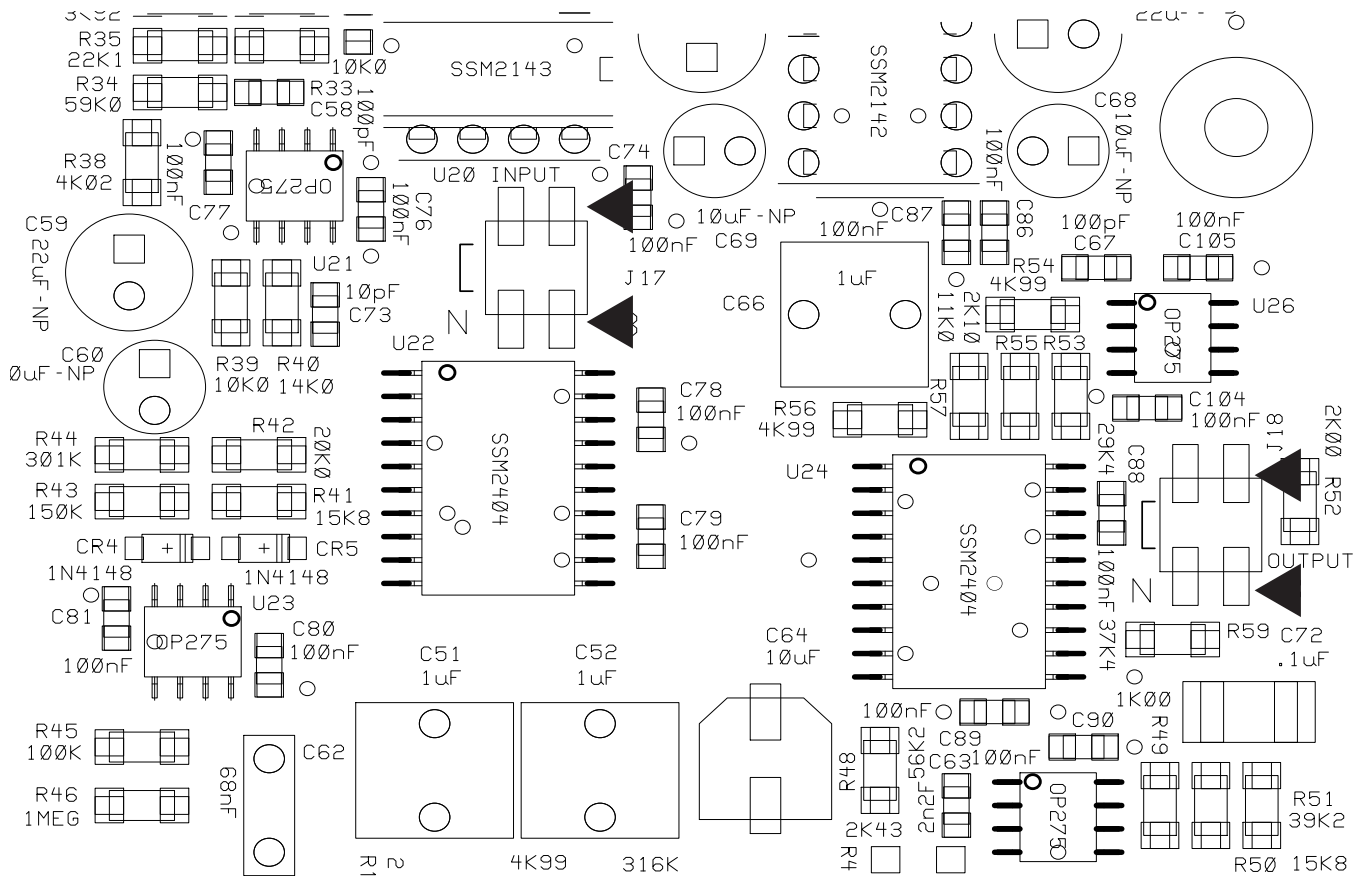
Accessory Port:

Physical: 9-pin "D" type female (pinned as DCE)

Pin#	Function	Direction
1	CD	From Matrix
2	RXD	From Matrix
3	TXD	To Matrix
4	DTR	
5	GND	
6	DSR	From Matrix
7	RTS	From Matrix
8	CTS	To Matrix
9	RI	No Connection

MATRIX RACKMOUNT MAIN BOARD +8 dBu INPUT & OUTPUT JUMPERS

Rackmount audio input/output level jumpers — On the Matrix rackmount only, there are jumpers available to set the input and output levels to +8dBu. Moving these jumpers disables the ability to control levels via the Matrix menu. These jumpers are shown as J17 (input) and J18 (output). Factory default is for these jumpers to be in “normal” mode, where levels are controlled via the menu system.



FCC REQUIREMENTS

1) This equipment complies with Part 68 of the FCC rules. On the bottom of the Matrix is a label that contains, among other information, the FCC registration number and ringer equivalence number (REN) for this equipment. The USOC jack required is an RJ11C. If requested, this information must be provided to the telephone company.

2) The REN is used to determine the quantity of devices which may be connected to the telephone line. Excessive REN's on the telephone line may result in the devices not ringing in response to an incoming call. In most, but not all areas, the sum of the REN's should not exceed five (5.0). To be certain of the number of devices that may be connected to the line, as determined by the total REN's, contact the telephone company to determine the maximum REN for the calling area.

3) If the Matrix caused harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. But if advance notice isn't practical, the telephone company will notify the customer as soon as possible. Also, you will be advised of your rights to file a complaint with the FCC if you believe it is necessary.

4) The telephone company may make changes in its facilities, equipment, operation or procedures that could affect the operation of the equipment. If this happens, the telephone company will provide advance notice in order for you to make the necessary modifications in order to maintain uninterrupted service.

5) If trouble is experienced with the Matrix, please contact Comrex Corporation at the address below for repair and warranty information. If the trouble is causing harm to the telephone network, the telephone company may request you remove the equipment from the network until the problem is resolved.

Comrex Corporation
19 Pine Road
Devens, MA 01432
Tel: 978-784-1776

6) The Matrix cannot be used on public coin service provided by the telephone company. Connection to Party Line Service is subject to state tariffs. Contact the state public utility commission, public service commission or corporation commission for information.

APPENDIX A

MODEM LINE CHECK LIST

This Comrex Engineering Note discusses problems (and possible solutions) encountered in using modems on dial telephone lines. Such circuits are sometimes called POTS (plain old telephone service) or PSTN (public switched telephone network) lines. It is not applicable to ISDN, SW-56 or DDS lines or to equalized program circuits.

A LITTLE BACKGROUND...

The Comrex POTS codecs use internal high-speed modems to send audio in digital format over POTS lines. These modems follow the international V.34+ standard and are very similar to modems used in personal computers. Some of you may remember when 300 bps and 1200 bps modems were considered very fast. Modern V.34+ modems communicate at rates up to 33,600 bps on high quality phone lines. The faster rate is made possible by having the modems make a detailed analysis of technical parameters like frequency response, noise and time delay at the beginning of the call and at any time they are required to renegotiate the connection. The beeps and “hash” you hear at the beginning of the modem call take about 10 seconds and are used to set the connect speed and a number of internal parameters.

If the telephone line is fairly “quiet,” the negotiation at the start of the call may be all you need to insure reliable operation for the duration of a broadcast. You should realize, though, that the digital data moving between Matrixes, Vectors, BlueBoxes or HotLines must be very error-free, because the audio is being decoded in real time. If much of the data is damaged in transit, you will have gaps or strange noises in the audio. You can eliminate small problems with error correction (standard with the Matrix and selectable on the Hotline), but there is no time to fix large errors by re-sending data that was missed. The usual remedy is to lower the data rate, as the slower speeds are more tolerant of errors. On the other hand, you would prefer to have the data rate set as high as practical, as this gives better audio quality. The battle between these approaches starts many users thinking about the “health” of their phone lines.

In most cases, you will have better results with a direct line to the telephone company than through a business phone system. Modems work best if the call is converted once to digital form and once back to analog. You may get additional analog to digital conversions by running the call through your phone system. This will limit the maximum speed that the modems can achieve. The major exception to the direct line “rule” is in truly large business phone systems where the call is sent to the phone company digitally over a high-capacity trunk like a T1 system.

One more note on in-house telephone systems: Modular jacks are used on all kinds of phone systems. If you see multi-line phones with lots of lights and buttons, be very cautious about plugging a Hotline, BlueBox, Vector or Matrix into that jack!

The voltages and currents present may “fry” your modem. If you need to borrow a line from a business, try to use a fax or modem line. You might want to bring along a simple tester like the IBM “Modem Saver” or even a cheap phone. It is a lot cheaper to blow up a hardware store phone than a POTS codec!

If you are having trouble getting good modem connections, we strongly suggest you start your troubleshooting at the studio end. If you have a problem there, it will affect all of your POTS codec broadcasts.

CHECKLIST

❑ Make a test call to the Comrex POTS codec test number: 978-784-1700. Note the modem connect speed. Listen to the audio for 5-10 minutes. If you notice errors, try using the *DROP RATE* key to step down one speed at a time until the audio is stable. You may want to try more than one call, and see if there is any difference. If you are happy with the performance, your problems may be at the other end of your broadcasts. Try the same test there.

❑ If your POTS codecs work fine on long-distance calls, but have problems on calls within your area code, the real culprit may be the manner in which the phone company routes calls between those offices. You may be “getting stuck” with older microwave circuits, or extra analog/digital conversions at the boundaries of different phone companies. Your ability to solve this problem is limited. It is worth complaining about, but there may be no practical solution. In any case, if you are happy with long distance calls from a particular site, the local part of your telephone circuit is OK! The rest of the information in this checklist may help, but your real problems are elsewhere.

❑ Since natural events (like thunderstorms) and unnatural events (like interns who plug your expensive POTS codec into electronic key phone systems) can damage the modem or interface circuitry in your Matrix, you should try to weed out a bad unit. Make calls from each codec to our test number, using the same phone line, if possible. If one unit behaves oddly, call Comrex Tech Support.

❑ Start at the back of the codec. Try another modular phone cord, even if you have been using the one we supplied. You might also look at the telephone wall jack to see if the pins are straight and the cable connections are OK. You’ll feel mighty foolish if you spend lots of time with the phone company, only to find that this is the problem!

❑ Next, plug a phone into the line jack, and make a call to a cooperative person on another (hopefully clean) line. Listen carefully for hum, buzz, clicks or crosstalk from other phone lines. Try to do this at a time of day when other phones at that site are in frequent use. If you can hear such noises, you will have a target to aim for. Then try the same call with the phone connected at the demarcation point where the phone lines are terminated by the phone company. Be sure that your other internal wiring is disconnected when you

make the call. If the line is clean, the trouble may be in your inside wiring. If the line is still noisy, the problem is either with the phone company or the phone wiring at the other end.

❑ Check out your inside wiring. How does the line get from the telephone company demarc point to your codec? You may want to route a separate line away from other lines or electrical circuits. Make sure there are no cable staples that go through the cable! What kind of wire is used? We strongly suggest twisted pair phone cables — if you are using “quad wire,” you may be picking up extra noise. If you are buying cable, consider using Category 3 (or higher) computer network cable.

❑ Are there other phones, modems, fax machines or devices on your line? We suggest that your POTS codec be the only thing on the line. Even if the other equipment is turned off, the extra circuitry on the line may limit your modem performance. If you need to share the line with other equipment, consider putting in a simple switch to isolate the codec.

❑ Are you located at an AM transmitter site? If so, we strongly suggest using a telephone line RF filter with good attenuation in the AM broadcast frequency range. In really stubborn cases, you may need to filter the line at the telephone company demarc point and at the back of the codec.

❑ Assuming that the inside wiring checks out OK at each end, it probably is time to start talking with the phone company. Just a caution, though — they are now being deluged with calls from Internet users who feel that their modem performance is being affected by their phone lines. It is probably not in your best interest to stress the word modem in your conversations with the telephone company. They will probably just give you a terse dissertation on how they are not required to support modems above the 2400 bps level.

❑ If there is audible noise on the outside line, go right ahead and report it. If the problem is weather related (such as with rainy weather), make the call when the line is actually acting up. They will do an initial test through their computer system. If they see something wrong, they will dispatch a repair person to deal with it. The problem will likely be some imbalance between the line and ground, may be due to a bad cable or a bad line protection device. If they don't see a significant problem by looking through their computer system, you may have to do some talking to get them to dispatch someone. Be persuasive, but don't try to dazzle them with your technical competence or assertiveness training.

❑ If a repair person visits you, make the most of the situation. Don't just send them to the closet at the end of the hall. Clean it out in advance so they can get in there without danger. Stay with them. Gently explain your problem. Offer coffee. Even if they don't find the problem, you may wind up with a better friend the next time. Try to get them to check the line for "balance," even if you are a little shaky as to what that means. See if you can find out how the line gets to you from the telephone company office. In order for this conversation to make sense, we suggest that you read the following:

PHONE LINE 101

The simplest phone line is just a pair of copper wires running from the telephone company office to your location. Part of the line may run above ground suspended on phone poles, and part may be underground, either directly buried or in conduit pipes. Both are subject to weather-related problems. Water can enter improperly sealed cables and cause partial connections between your wires and ground or other wires in the cable. Small animals can chew at cables and damage the insulation on the wires, with the same result. Lightning strikes can damage protective devices on the lines, resulting in partial connections from the wires to ground.

Even the simple phone line is actually fairly complicated. If the line is over 3 miles in length, the phone company will insert "loading coils" which help equalize the frequency response of the line. These loading coils don't introduce any extra noise, but they will limit modem performance to 21,600 bps or so. A nastier thing is the addition of "bridge taps" on the line. These are simply other pairs of wires connected across your line. They may run down other streets and end on poles, in manholes or inside other buildings. Nothing is connected at the other end, but the extra capacitance of that cable will cause peaks and valleys in the frequency response of your line. Bridge taps may also introduce extra noise or crosstalk as well. A "worst case" scenario would be for you to be assigned a cable pair that was previously used at another location and still has inside wiring connected in that building.

As telephone use has expanded, phone companies have found it impractical to keep running more copper lines, particularly in urban and large suburban areas. This has led to the increased use of "SLIC" (Subscriber Line Interface Concentrator) cabinets connected by conventional copper wire to homes and businesses in a neighborhood. You have seen these cabinets (usually green or gray) at street corners. The equipment in the cabinet converts the local call into digital format so that individual pairs are not necessary all the way back to the telephone company office.

There are two basic types of SLICs: integrated and nonintegrated. An integrated SLIC is connected by fiber-optic cable or copper wire to the telephone office. It is a high capacity digital device that is actually part of the phone company switching equipment. Your call undergoes an analog to digital conversion in the cabinet, and stays in that form all the way through the telco switch. An integrated SLIC is very modem-friendly, as it shortens that analog part of your phone line, avoiding loading coils and extra bridge taps. A nonintegrated SLIC is not tied directly to the switch, but requires a companion unit at the telco office that will introduce another set of digital/analog conversions, and that will limit modem performance. It may be better than an excessively long copper line, but it will not be as good as an integrated SLIC ... AND NOW BACK TO THE CHECKLIST!

❑ If the phone company believes that you have a bad cable pair, they will probably switch you to another pair. This is a good opportunity to request (don't demand) the shortest possible routing for the line, hopefully without loading coils and with a minimum number of bridge taps. You can also plead not to be put on a nonintegrated SLIC, as that may worsen the situation.

❑ If the phone company thinks your line is OK, you may want to try an "end run" around the regular people. The telco folks have higher-level customer service people who can advocate for you within the company. If your monthly bills are substantial enough, you can request that such a person be assigned to your account. If that fails, you might contact the person in charge of media relations in your area. They want to stay friendly with radio and TV stations, and may be able to connect you with an influential "inside person." As always, be friendly and diplomatic. Mugs and T-shirts might be an idea for people who seem to be working on your behalf. It wouldn't hurt to have your call letters on their desk!

❑ Attention "tech types": If you used to order equalized phone lines for remotes, and you still remember which cable pairs the telephone company used for the lines to your studio, you might try requesting one of those pairs for your POTS codec line. Equalized lines are usually free of loading coils and bridge taps and are tested more rigorously for noise and balance.

❑ If you are absolutely stuck with a lousy phone line to your studio, you might want to try creating a "perfect" phone line from an ISDN line (assuming that is available). All you will need is a computer-type external terminal adapter with at least one POTS port. That port will have ringing and dial tone just like a regular phone line and will be all-digital to the phone company. You will need to program the TA with a computer, entering switch type, SPIDs and LDNs. After that, no computer connection is necessary.

APPENDIX B

ABOUT MIX-MINUS

Even the simplest remotes are a two-way process. The remote site must send its audio to the studio and receive a return feed to monitor the programming. This return feed may be done over a radio station's regular transmitter (with an AM or FM radio at the remote), a special radio link or a telephone circuit. This feed may just go to headphones at the remote and also be put on speakers for the local audience.

The problem comes when there is a time delay in getting audio to and/or from the studio. In this case, the remote talent hears a delayed version of their voice in the headphones and may find this very distracting. Even a remote done with simple equipment or a frequency extender on plain phone lines may have this problem on a long-distance call. All remotes using ISDN, Switched-56 and POTS codecs will have delays each way as signals are processed from analog to digital, compressed, uncompressed and converted back to analog audio. Some digital compression schemes, such as G.722, result in shorter delay times, but there will still be a "reverb" effect in headphones at the remote site, if their audio is sent back from the studio. In any of these cases, it may not be possible for the remote people to listen to an off-air or program channel feed.

The solution is **mix-minus**. A mix-minus feed has a mix of all of the programming on the radio station (or network) **minus** the audio from the remote. In other words, the station or network doesn't send the remote audio back to the remote. At the remote end, this mix-minus feed is converted back to an "air monitor" by mixing in the local audio from the remote. This is easily done on the Matrix, with the *OUTPUT MIX* knob.

For radio stations, in addition to fixing the time delay problem, using a mix-minus feed has two other advantages. First, if the station uses a 6-7 second delay to allow editing of phone calls, pre-delay audio can be sent to the remote site. Second, if there is a PA system at the remote, they will be able to run the speaker levels higher with the mix-minus audio. This is because the remote microphone audio is not running through the station's audio processing, and the levels stay under the control of the remote operator.

The simplest way to do one mix-minus feed in a typical radio studio is to use the Audition or second program channel. On many audio consoles, each fader's output may be sent to both Program and Audition. If your board will allow those feeds simultaneously, just set all of the modules to Program and Audition, **with the exception of the one carrying the remote audio. Set that one to Program only.** The Audition channel will then be a mix of everything on the console except the remote. That will be your mix-minus, and it should be sent to the remote site. One caution — make sure that audio is being sent to and from any telephone modules you may have in the console. They may have been designed to work with only one channel at a time — either Program or Audition, but not both. If so, you will have to check with your "tech guy" or the

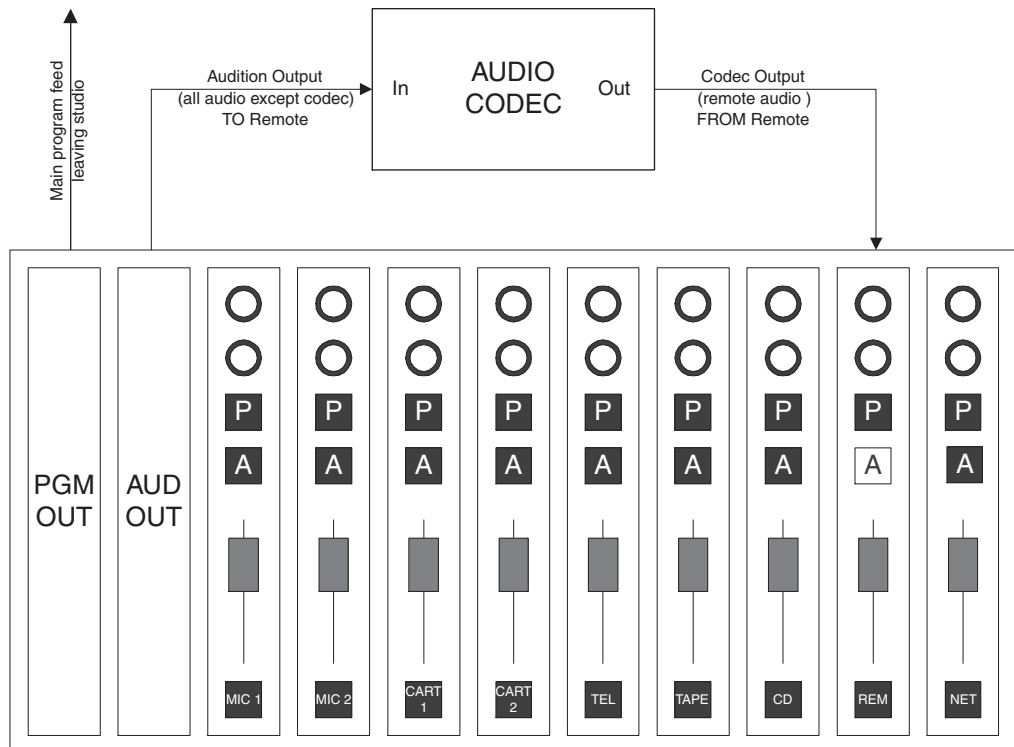
board manufacturer for advice. If you use multiple audio codecs, you should investigate the Comrex Mix-Minus Bridge. This will allow you to expand one Program/Audition setup to handle five codecs or other remote audio devices. It also provides IFB (talkback) to remote sites.

If you are using a Portable Matrix at the studio, check the *OUTPUT MIX* knob. If this knob is turned counterclockwise, audio from the studio board will be sent back to the board, and funny effects will result.

*"I'M USING MIX-MINUS AND
I STILL HEAR AN ECHO!"*

If you are doing a call-in talk show on the road, the remote people may complain of hearing an echo when a caller is put on the air. With the telephone pot down, everything is OK. The culprit is the telephone hybrid being used to put callers on the air. Some of the remote audio is "leaking" through the hybrid and mixing with the caller audio. Modern digital hybrids do a much better job of preventing this than the older units that had to be manually "tweaked" for each call. If you are using a digital hybrid and having this problem, dig out the manual and redo the hybrid's initial setup.

Some stations do not feed regular audio back to the remote, particularly for sports broadcasts. Instead, they send a continuous feed of the board operator's microphone out to the game. In this case, the board op must remember to wear headphones and keep the studio speakers *off*. His microphone would pick up the remote audio from the speakers, and the crew at the game would hear themselves in delayed form.



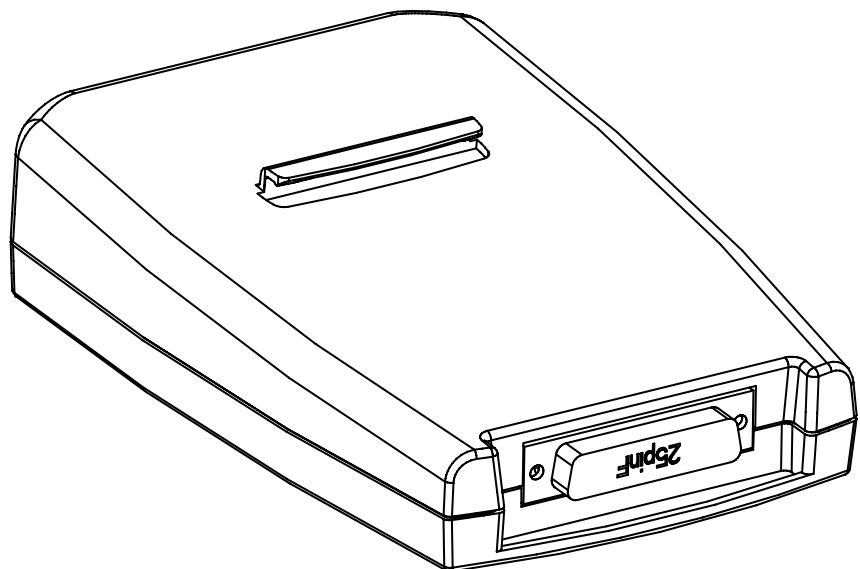
APPENDIX C

ACCESSORIES

The Matrix was designed to allow users to take advantage of virtually every suitable circuit that is available, with flexibility to add modules as new technologies become viable. Optional accessory modules may be inserted into a channel in the bottom surface of the base unit. The modules connect to the base unit via a DB-25 connector. The fluid design of the Matrix with its accessory slot and connector allows virtually infinite upgrade potential as new circuits become available.

Current portable accessory options include:

- Optional ISDN Module to allow operation on ISDN using Layer III or G.722. (An optional ISDN board is also available for the Rackmount Matrix.)
- Optional GSM Module includes a GSM wireless phone and software designed to transmit 7 kHz audio over a 9600 baud GSM connection.
- Optional TelCell Module includes a hybrid allowing the user to deliver unenhanced audio on plain telephone lines as well as allowing the user to connect the Matrix to the headset jack on a wireless/mobile phone.
- An optional battery kit is available, including NiMH battery, charger and mounting clip.



ISDN MODULE

The ISDN module transforms the Matrix into a lightweight yet powerful mono ISDN codec with integrated terminal adapter. The codec engine includes 3 different algorithms. ISO/MPEG Layer III and G.722 are implemented for maximum industry compatibility, and Turbo G.722 yields 15 kHz, very low delay audio transmission. The terminal adapter is designed to work with virtually all international ISDN switch protocols and is available with either an “S/T” or a “U” interface.

GSM MODULE

With the GSM Wireless Module, you can broadcast quality audio from anywhere, without being tied to a phone line. This module incorporates a GSM wireless phone which is operated via the keypad of the Matrix. A new, enhanced voice algorithm boosts frequency response to 7 kHz on bit rates as low as 9600 baud. Finally, broadcasters have a suitable solution to deliver audio live from press conferences, parking lots, or nearly anywhere a phone line is not available. This module may be used in any portable Matrix. Existing units may require upgrading.

TELCELL MODULE

The TelCell module connects to the Matrix Portable Codec adding the ability to send and receive audio through the handsfree port of many mobile phones or standard telephone lines. Although this module does not add any audio enhancement to the wireless or land line phone call, it makes a useful backup tool. Adding professional microphones will often improve audio quality. Calls can be placed to regular telephone hybrids with this module (no Matrix is required on the receiving end of the call).

THE BATTERY KIT

The battery kit includes a NiMH (Nickel Metal Hydride) battery with mounting bracket and external charger. The battery may be used as a primary power source to run the unit for a maximum of 7 hours, or it can be used as an optional backup should AC power be interrupted during a broadcast.

APPENDIX D

USING THE COMREX MATRIX WITH A HOTLINE OR VECTOR

The Matrix is compatible with the Comrex HotLine and the Vector, automatically sensing which algorithm is in use.

Throughout the extensive field testing we did during the development of the Matrix, we found issues on some phone calls at higher than average connect speeds to or from a HotLine or Vector. Typically, the modems would continue to negotiate at the start of the call. While redialing the connection frequently cured this; we found that the better approach was to upgrade the software in the HotLine and Vector to help match the newer modem in the Matrix.

This new control software is identified as Revision J for the HotLine and Revision 7 for the Vector. If you will be using the Matrix with these codecs, you will get the best performance if their software is updated. Recent HotLine and Vector software shows its version on the screen:

HotLine — Watch the screen during boot-up. If the software is Rev. F or higher, you will see the information just before the *MAIN MENU* appears.

Vector — From the *STATUS DISPLAY*, press the *MENU* key. If the software is Rev. 6 or higher, you will see the information in the lower left corner of the *MAIN MENU*.

In both cases, if you do not see an indication of the version, the unit has older software.

The HotLine and Vector upgrades can be done in the field or here at Comrex. For a field upgrade, you only need the new chips, some simple tools, our instructions and a “PLCC Extractor,” available at Radio Shack or similar electronic suppliers.

For more information on the upgrades, call us at 978-784-1776, and we’ll be happy to help you.

Also, for the present, we recommend that you turn off Error Correction on a HotLine being used with a Matrix.

APPENDIX E

PLAIN TELEPHONE MODE

USING THE MATRIX IN PLAIN TELEPHONE MODE

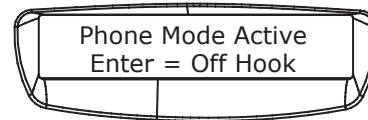
Note: This section applies to Matrix Portable units with serial numbers 540 and below that have NOT received the modification for TelCell Module compatibility and Matrix Rackmount units with serial numbers 400 and below.

This mode converts the Matrix into a plain old telephone. You can speak and listen through the microphones and headphones connected to the Matrix. If the Matrix is unable to make a modem connection on a telephone line or if the far end does not have a Matrix or HotLine, it is possible to use the Matrix mixing facilities and Plain Telephone Mode to send program. Naturally, the program quality will not match the POTS codec quality.

To enter the *PLAIN TELEPHONE MODE*:

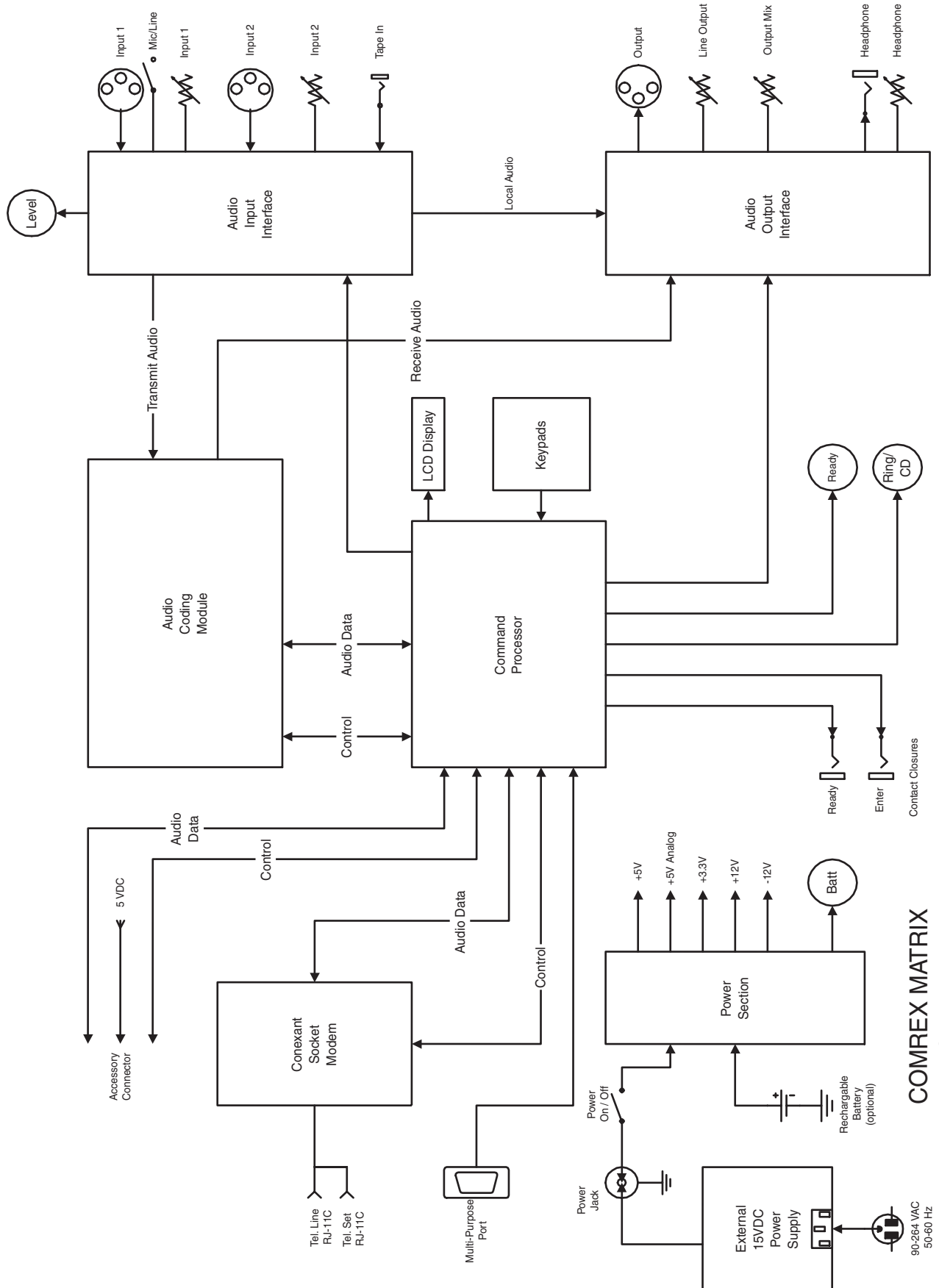
- A. Press *ENTER* from the *MAIN STATUS DISPLAY*.
- B. Then press “1” for *DIAL*.
- C. Then “3” for *PLAIN PHONE MODE*.

- D. The display will read —

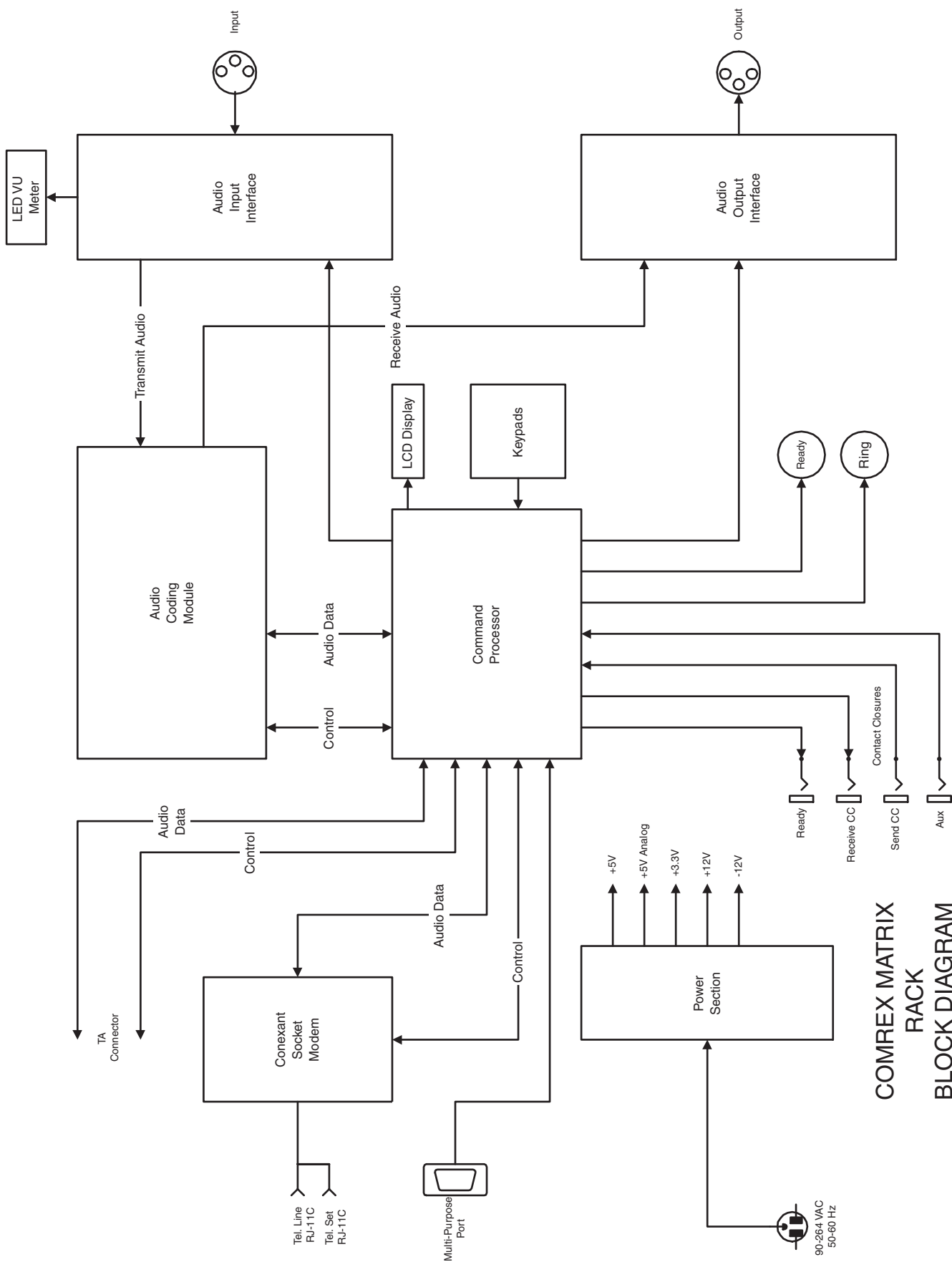


- E. Press *ENTER* to get a dial tone.
- F. You may now dial another telephone, telephone coupler, or hybrid.

The auto answer function is disabled in *PLAIN TELEPHONE MODE*.



COMREX MATRIX
PORTABLE
BLOCK DIAGRAM



COMREX MATRIX
RACK
BLOCK DIAGRAM