ACCESS Stereo BRIC IP Codec

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About Comrex

Comrex has been building reliable, high quality broadcast equipment since 1961. Our products are used daily in every part of the world by networks, stations and program producers.

Every product we manufacture has been carefully designed to function flawlessly, under the harshest conditions, over many years of use. Each unit we ship has been individually and thoroughly tested. Most items are available off the shelf, either directly from Comrex or from our stocking dealers.

Comrex stands behind its products. We promise that if you call us for technical assistance, you will talk directly with someone who knows about the equipment and will do everything possible to help you.

Our toll free number in North America is 800-237-1776. Product information along with engineering notes and user reports are available on our website at www.comrex.com. Our email address is info@comrex.com.

Warranty and Disclaimer

All equipment manufactured by Comrex Corporation is warranted by Comrex against defects in material and workmanship for one year from the date of original purchase, as verified by the return of the Warranty Registration Card. During the warranty period, we will repair or, at our option, replace at no charge a product that proves to be defective, provided you obtain return authorization from Comrex and return the product, shipping prepaid, to Comrex Corporation, 19 Pine Road, Devens, MA 01434 USA. For return authorization, contact Comrex at 978-784-1776 or fax 978-784-1717.

This Warranty does not apply if the product has been damaged by accident or misuse or as the result of service or modification performed by anyone other than Comrex Corporation.

With the exception of the warranties set forth above, Comrex Corporation makes no other warranties, expressed or implied or statutory, including but not limited to warranties of merchantability and fitness for a particular purpose, which are hereby expressly disclaimed. In no event shall Comrex Corporation have any liability for indirect, consequential or punitive damages resulting from the use of this product.

SECTION 1 INTRODUCTION

Congratulations on purchasing the Comrex ACCESS codec. This product is the next step in the evolution of audio transportation over networks. For Comrex, this began in 1976 with the introduction of the Frequency Extender, followed by ISDN codecs in the early 1990s and POTs codecs in 1996. So we've been doing this for a long time.

The ACCESS product is the result of years of our research into the state of IP networks and audio coding algorithms. This has all been in the quest to do what we do best, which is to leverage existing, available services to the benefit of our core customers - radio remote broadcasters.

The heart of this product is called BRIC (Broadcast Reliable Internet Codec). While others have introduced hardware coined "IP Codecs," this is the first product we're aware of that dares to use the word Internet "with a capital I." Given the challenges the public Internet presents, it's no small boast to say that this product will perform over the majority of available connections.

BRIC represents a change that is both desirable and inevitable for remotes. It's inevitable because, as available connections move from old fashioned "circuit switched" style to newer "packet switched" style, technology like ISDN and POTS codecs will begin to work less and less often. The desirability stems from the new wireless networks that will make remote broadcasting more mobile, simpler and less expensive. BRIC technology has been engineered not only to be robust enough for the Internet, but usable in really challenging Internet environments like 802.11x Wi-Fi, Wi-Max, 3G cellular and satellite based Internet connections.

Those of us here who have been remote broadcasters have been wishing for a system like this for a long time. As former broadcasters turned designers, it's our hope that this kind of enabling technology will tickle the imagination of the user, enabling more creative and entertaining programming to be broadcast from more diverse and interesting locations. Please let us know about your unique ideas and adventures by dropping us a note at techies@comrex.com.

About BRIC

BRIC (Broadcast Reliable Internet Codec) is a breakthrough technology with hardware that will deliver audio over the public Internet in much the same way that ISDN and POTS codecs have performed in the past. BRIC consists of three pieces:

- Rackmount ACCESS codec (which you are using)
- Portable ACCESS codec
- BRIC Transversal Server

We will describe each piece independently:

1) Rackmount ACCESS codec — This product is designed for installation in a radio station's "remote rack" and is designed for "always on" operation. Hence the lack of a power switch. Also, it is envisioned that this product will be controlled entirely from a computer connected to the local LAN. There are no user controls on the ACCESS Rack (other than a recessed reset button) and the only indications are audio meters and a **Ready** light to indicate an incoming data stream. After initial configuration, all connection, status and diagnostics are available via the internal web server.

2) Portable ACCESS codec — This product is engineered to provide the most convenience for the remote broadcaster on the road. It combines small size, battery power, clip-on mixer and headphone drivers with an audio codec capable of remarkable quality on the public Internet.

3) BRIC Transversal Server — This server exists on the public Internet at a fixed address and performs several functions. Its use is optional but makes connections between ACCESS codecs much simpler and removes worries about dynamic IP's, NATs, and other concerns that can make peer-to-peer connection over the Internet difficult (especially over tightly controlled networks like 3G or Wi-Fi). The BRIC TS provides the following functions:

a) Communicates with all ACCESS codecs that are provisioned to work with it. It keeps a log of the IP address of every codec that wishes to be subscribed.

b) Maintains a "keep alive" channel to each codec subscribed, allowing transversal of firewall and Network Address Translators when receiving an incoming call.

c) Provides each subscribed ACCESS codec with a "Buddy List" of other users, their current status, and will facilitate connection to them if desired.

MORE ABOUT ACCESS RACK

ACCESS Rack incorporates all the features, algorithms and services of BRIC as defined in the previous sections. Its main function is to provide a robust, high quality, low-delay audio link in full-duplex over challenging IP networks like the public Internet. To this end, it provides an intuitive and attractive *User Interface* via web-browser. Using this interface, you can select operating modes, check audio levels, make and end connections, and check network statistics of any connections you make. While ACCESS is designed to handle most network challenges in its default configuration, advanced options are available to allow customization of parameters that have effect on link stability and delay.

But wait! There's more! ACCESS is also a POTS codec. It has a built-in modem which can be set to make calls over analog phone lines directly to other units. In this mode, ACCESS can communicate with other ACCESS devices, or with a range of previous generation POTS codec devices made by Comrex.

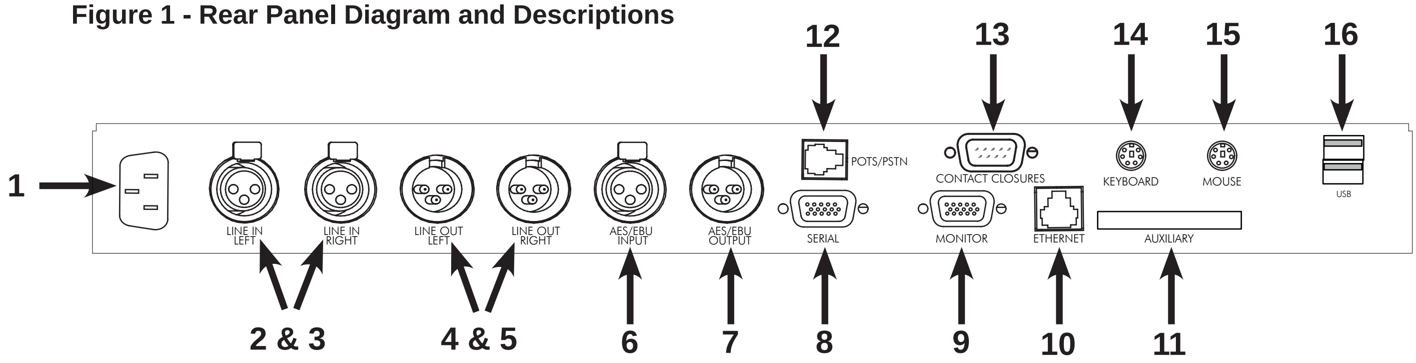
WHAT COMES WITH ACCESS RACK

The following items are shipped with a new ACCESS Rack:

(1) ACCESS Stereo BRIC IP Codec (Rackmount)
(1) 6' Ethernet cable
(1) 6' Telephone cable
(1) AC Power cord
(1) Operating manual
(1) Warranty card (Please fill out and return)

Section 2 Rack Diagrams and Installation

Figure 1 - Rear Panel Diagram and Descriptions



1) AC Input
This is an IEC connector for the main power. ACCESS works on worldwide AC power at 110-240VAC 50-60Hz, auto detecting.

2) and 3) Analog Audio Input
Apply balanced analog audio to be sent over the network here. Left channel is used for mono encoding modes. Level is set to 0dBu (0.775VRMS) nominal. Full scale input is +20dBu.

4) and 5) Analog Audio Output
Balanced analog audio is available at these ports. Level is set to 0dBu (0.775VRMS). Full scale output is +20dBu.

6) AES3 Digital Audio Input
Apply a stereo AES3 signal here. The AES3 input supports all standard sampling rates. When an AES3 signal is present on this connector, the analog input connectors are disabled.

7) AES3 Digital Audio Output
A 48KHz AES3 stereo signal is available here. AES3 output is available simultaneously with analog.

8) Serial Port
Asynchronous ancillary data is available here.

9) Monitor Output
Attach a VGA or better monitor here for obtaining IP addresses and accessing the *Config* program.

10) Ethernet Port
10/100BaseT Ethernet port for connection to your network.

11) Auxiliary CF Port
For future use.

12) Modem Telephone Connector
Attach an analog telephone line here for POTS codec compatibility.

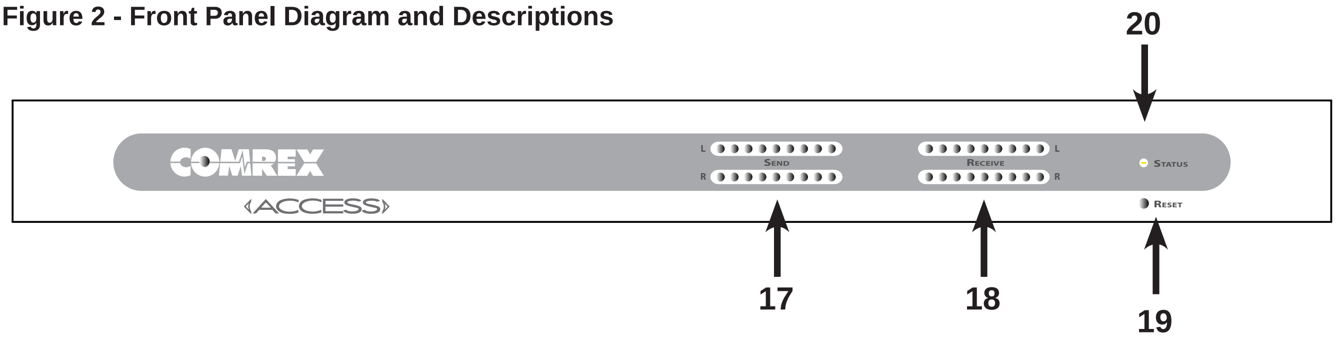
13) Contact Closure
Four sets of contact closure inputs and outputs are available on this port. These can be used to send signals to the far end of the link or to trigger remote control gear like automation equipment.

14) Keyboard Port
Attach a PS/2 style keyboard here for accessing the *Config* program.

15) Mouse Port
For future use.

16) USB Ports
For future use.

Figure 2 - Front Panel Diagram and Descriptions



17) SEND Peak meter that displays the level of audio being sent locally into the ACCESS, regardless of whether or not a connection is active. Proper level is indicated by peaks driving the **Yellow LEDs**, while avoiding lighting the **Red LEDs** (which indicates clipping).

18) RECEIVE Peak meter that displays the level of audio being sent remotely when a connection is active. Proper level is indicated by peaks driving the **Yellow LEDs**, while avoiding lighting the **Red LEDs** (which indicates clipping). Adjustments to this level must be made on the far end of the link.

19) RESET Recessed button to send ACCESS into hardware reset mode. Approximately 30 seconds are required to reboot when this is pressed.

20) STATUS Indicates when a valid incoming stream is present.

MONO VS. STEREO ACCESS uses its left channel input only for *Mono Modes*. Right channel is ignored. Output audio is available at both the left and right outputs in *Mono Mode*.

PINOUTS - AUDIO ACCESS audio connections are balanced professional level inputs and outputs.

Table 1 - XLR Pinout

Pin 1	Ground
Pin 2	Audio +
Pin 3	Audio –

Table 2 - AES3 Pinout

Pin 1	Ground
Pin 2	Data +
Pin 3	Data –

Pinouts - Contact Closure

Contact closures are available via the male 9-pin D connector on the back of the ACCESS Rack. Inputs are triggered by shorting the respective input to **Pin 5**. Outputs consist of an *open collector* circuit which, when inactive, will offer a high-impedance path to **Pin 5** and, when active, will offer a low impedance path to **Pin 5**. These outputs are capable of sinking up to 200mA at a voltage up to 12V. Do not switch AC mains power using these contacts.

Table 3 - Contact Closure Pinouts

Pin 1	Input #1
Pin 2	Input #2
Pin 3	Input #3
Pin 4	Input #4
Pin 5	Ground
Pin 6	Output #1
Pin 7	Output #2
Pin 8	Output #3
Pin 9	Output #4

Pinouts - Serial Port

The **Serial Port** is capable of transferring ancillary data to the far end of the connection. By default, the communication parameters are set for 9600bps, no handshaking, no parity, 8 data bits, one stop bit (9600,n,8,1). It is pinned on a 9-pin D female in DCE-style pinning. The port is designed to connect to a 9-pin PC serial port with a straight-through M-F cable. RS-232 levels are used.

Table 4 - Serial Port Pinouts

Pin #	Function	Direction
1	CD	Unused
2	RX Data	From ACCESS
3	TX Data	To ACCESS
4	DTR	To ACCESS
5	Ground	
6	DSR	From ACCESS
7	RTS	To ACCESS
8	CTS	From ACCESS
9	RI	Unused

SECTION 3 SETTING UP ACCESS

HOOKING UP

At a minimum, ACCESS will need an audio connection and a network connection. Levels of all analog audio I/O is 0dBu (0.775V) nominal. This level will provide 20dB headroom before the clipping point. Input audio is reflected on the front panel LED based peak meters. Clipping is indicated by the **Red LED** on these meters.

ABOUT NETWORK CONNECTIONS

ACCESS needs a network connection to be useful. On ACCESS Rack, the network connection is made via a standard 10/100baseT Ethernet connection on an RJ-45 connector.

In most ways, ACCESS will look like an ordinary computer to this network. In fact, ACCESS contains an embedded computer with a Linux-based operating system and a full network protocol stack.

ACCESS is perfectly capable of working over most LANs, but there may be situations where a LAN is heavily firewalled, subject to overloaded traffic conditions, or have security concerns. Better performance is possible if ACCESS has its own Internet connection. Often, it's worth the trouble to install a DSL line especially for ACCESS, especially if the cost is reasonable.

Since there may be bandwidth, firewall, and security concerns with installing ACCESS on a managed LAN, it is recommended that your IT manager be consulted in these environments. The details that follow assume a working knowledge of IT topics and network configuration.

SETTING UP ACCESS NETWORK CONNECTIONS

We recommend putting ACCESS on a LAN and scoping out its functions before use. To do this, ACCESS must be given an IP address. This is the Internet location where you can connect to ACCESS through a web browser. It will also be the address used when another ACCESS is connecting to it.

Every device on an IP network must have a unique IP address. This is a number between 0 and 4,294,967,295, which is the range of values that can be represented by 32 binary bits. For simplicity, we break this 32-bit value into four eight-bit values and represent each as a decimal number (between 0-255) separated by dots. For example, the Comrex test IP number is 70.22.155.133.

A device with a public Internet connection can either have a public IP address (which is directly accessible by the Internet) or a private IP address, which is directly accessible only by the LAN on which it is connected.

Figure 3 shows connection of an ACCESS directly to the Internet using a public IP address. Figure 4 shows connection to a subnet (or LAN) using a private IP address, with a gateway router separating the LAN from the public Internet.

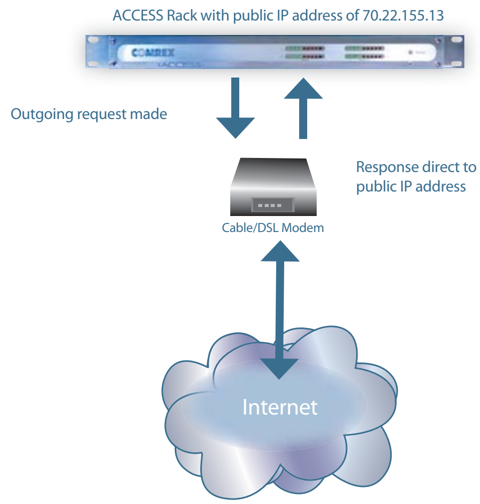


Figure 3 - Direct connection to Internet

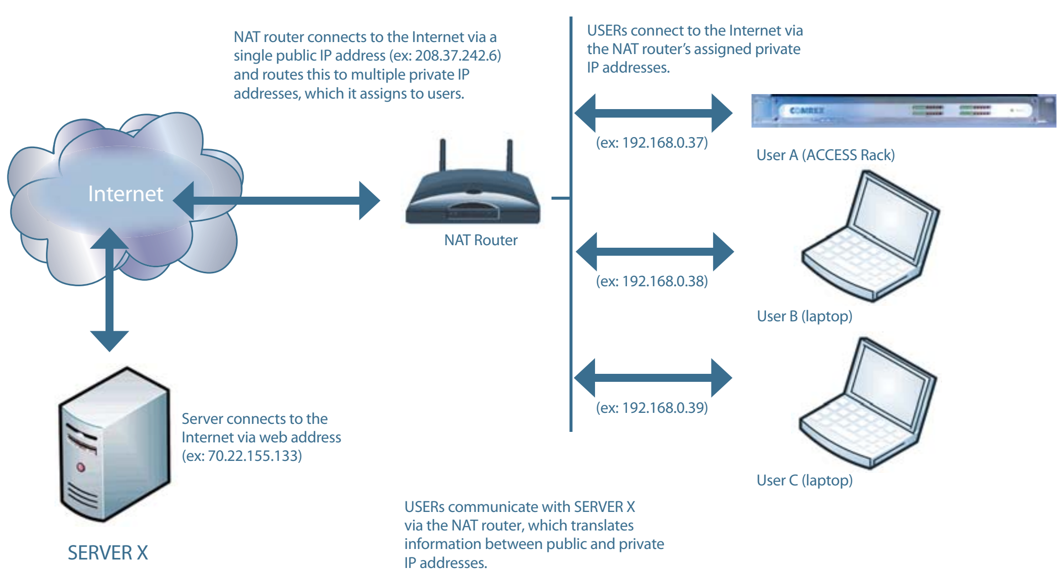


Figure 4 - Connection to Internet via subnet (or LAN)

To have the ability to make connections universally, in advance of the delivery of BRIC Transversal Server (BRIC-TS), one of the ACCESS in the link should be connected to a public IP address. This can be achieved several ways:

1) ACCESS can be the only device connected directly to its Internet link or it can share an Internet link that provides more than one IP address.
2) ACCESS can be connected behind a NAT router, which can be programmed to provide public Internet access to it through port forwarding.

But for now we'll assume you have a way to set up at least one end of your ACCESS link with a public IP. In a radio remote environment, this should probably be the studio end, since you will often have much less control on the remote side.

DYNAMIC VS. STATIC ADDRESSING

ACCESS can be set to its own, fixed IP address (referred to as *Static* in Internet-speak) or can obtain its address from the network (known as *Dynamic* or DHCP). This concept is entirely independent from the *Public vs. Private* concept. Public and private addresses can each be dynamic or static.

Dynamic (DHCP) — ACCESS is set by default to DHCP addressing, meaning that it looks to your network for assignment of an IP address. If your network has a DHCP server and this is the way you intend to use it, you don't need to alter any settings in the config program. You will, however, need to know what address is being assigned to ACCESS by the network. This is easily done by attaching a computer monitor to the VGA port on ACCESS before applying power. After ACCESS boots, it will display the current IP address on the monitor. Note: DHCP addresses can change over time, so you may need to recheck the address if you are having trouble connecting.

Finally, there's one other way to determine the IP address of ACCESS. If you're unable to put a computer monitor on the system, you can infer the IP address by what's displayed on the front panel LEDs for a few seconds during the boot process.

DHCP servers typically assign IP address in a standard format. This is because they must choose addresses that are not in use by the Internet at large. They will likely choose an address at one of 3 distinct ranges:

192.168.x.x   172.16.x.x   10.x.x.x

Also, on 192.168 style and 172.16 style subnets, the third entry will typically be a single digit (often 0 or 1). You can usually find out the DHCP assignment style by querying a Windows computer on the same LAN using Run->Cmd->ipconfig. If you know your DHCP server assigns addresses using one of the first two formats (or you know the DHCP assignment range on a 10.x.x.x network) you can usually derive the true IP address by the front panel LEDs. They will display a "coded" version of the IP address assigned for a few seconds during boot just before the ACCESS enters operational mode. During this time, the **Ready** light on the front panel flashes quickly, and the **Level LEDs** display the last 4 digits of the IP address. This is best shown by example:

Assume you are using a Linksys router on your network that has a built-in DHCP server. You may be aware that by default this router assigns IP addresses using the range 192.168.1.2-255. Let's assume that when connected, the ACCESS is assigned an IP address of 192.168.1.7. The **LEDs** will display the last four decimals of this address (including zeros) so during boot you will see the following code:

L input will display 1 LED
R input will display 0 LEDs
L output will display 0 LEDs
R output will display 7 LEDs

You can now assume that your ACCESS has the address of 192.168.**1.007**

Static IP — Setting a Static IP requires that you enter the config program. You will need to enter the following information into the ACCESS:

- **IP address of the ACCESS** – make sure this has been provided by your ISP or that nobody else on your LAN is using this address.
- **Subnet Mask** – A series of numbers that indicate the range of your LAN addresses. If in doubt, try 255.255.255.0.
- **Gateway Address** – The address of the Internet gateway on your account. If in doubt, try the first three number of your IP address with the last digit of 1 (e.g. xxx.xxx.xxx.1).
- **DNS address** – Not currently used by ACCESS but may be utilized for future features. Enter it if you know it.

RUNNING CONFIG

Running *Config* requires that you attach a PS/2 style keyboard and video monitor to the appropriate jacks on the rear panel of ACCESS Rack. Remove and re-apply power after connection of a keyboard. After the system boots (and you see the Linux status messages stop scrolling) you will see the **Main IP Address** screen, as shown in Figure 5A.

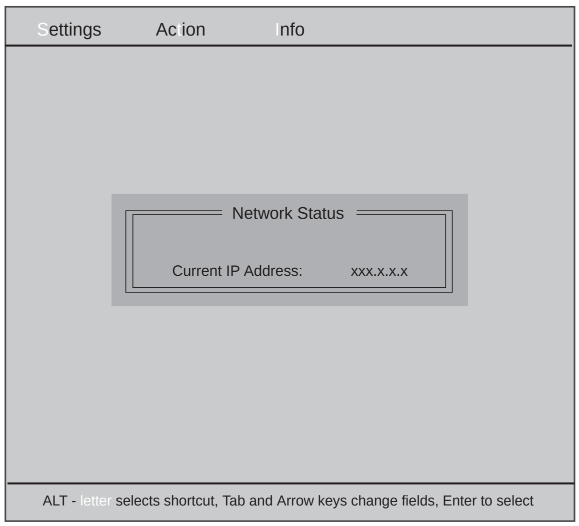


Figure 5A - Main IP Address Screen

Enter the *Config* program by selecting **Alt-S** on the keyboard. Then scroll down to **IP Networking** (see Figure 5B) and press the **Enter** key.

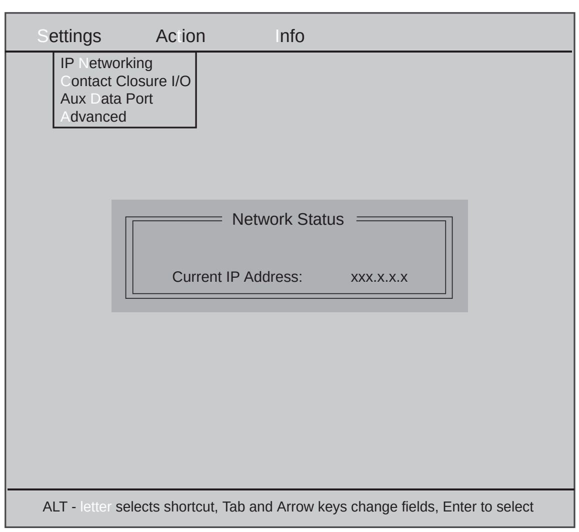


Figure 5B - Main IP Address Screen with pull down Settings Menu

The **IP Configure** screen is shown in Figure 6. Press the **Tab** key to scroll between the options. First, tab to the top field to select **Static IP** addressing, then tab down to enter your fixed IP Address, your Netmask, and your Gateway information. If you know your DNS info you may enter it, or leave this field blank. Selecting **OK** will store the changes and return to the **Main IP Address** display.

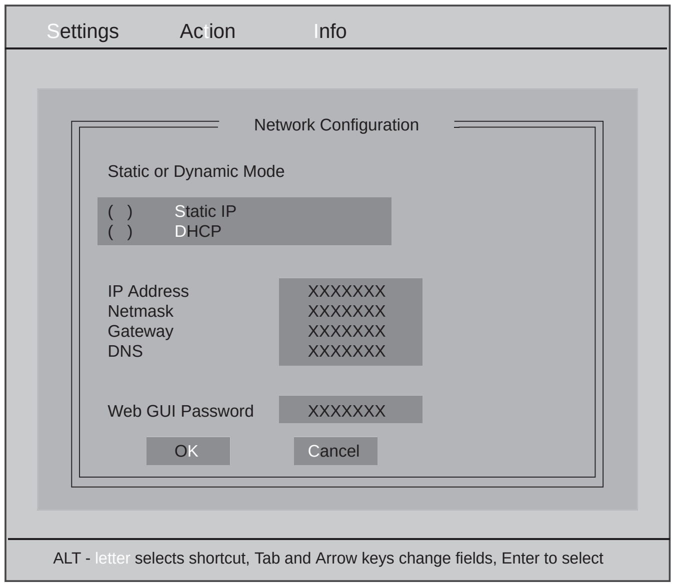


Figure 6 - IP Configure Screen

The **Aux Data Port Configuration** screen, as shown in Figure 7, allows for set-up of asynchronous ancillary data available via the serial port.

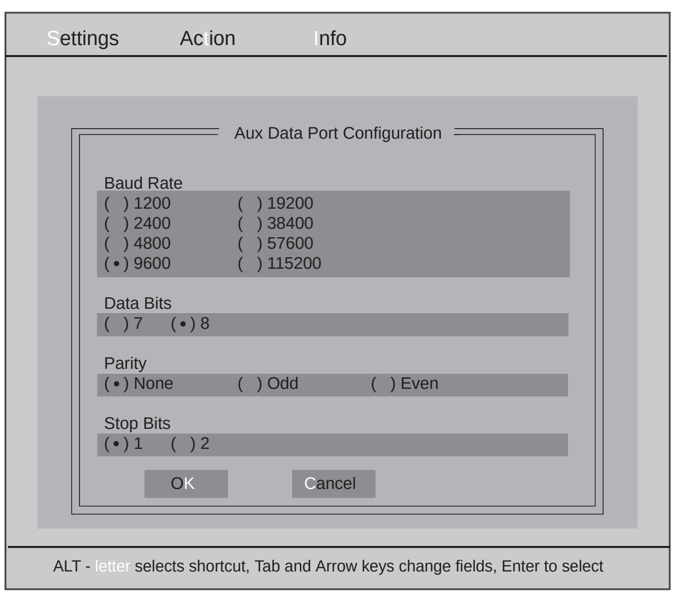


Figure 7 - Aux Data Port Configuration Screen

There are four input and output contact closures available via the contact closure port. By default, these closures are set for normal end-to-end operation. When closed, they can be used to send signal to the far end of the connection or to trigger automated equipment. The **Contact Closure I/O Configuration** screen (Figure 8) provides the ability to change the #4 contact closure from normal end-to-end signal to auto connect to a specified address on input. Alternately, the #4 output contact closure can be set to close only upon connection.

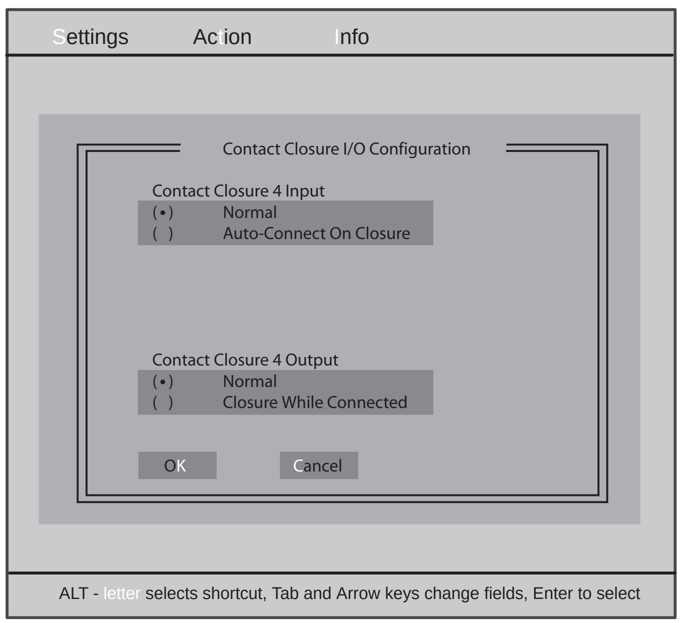


Figure 8 - Contact Closure I/O Configuration Screen

When ACCESS is configured to auto connect on closure you must also select a destination address from the peer list in the **Connections Tab.** For more details on the **Connections Tab** please refer to page 19. The **Auto-Connect On Contact Closure** box in the **Change Remote Settings** option (also found in the **Connections Tab**) must be selected in order for the auto connect feature to work. Please refer to the *CHANGE REMOTE SETTINGS* section 8 for more information.

SECTION 4 GAINING ACCESS TO ACCESS

ACCESS USER INTERFACE

Once your IP settings are configured and ACCESS has cleanly booted on your LAN, it's time to take a look at the *ACCESS User Interface*. This is done by pointing a web browser on your LAN to the ACCESS IP address. To do this, simply type the address into the URL bar of your browser. You will need Internet Explorer 6 or higher or Mozilla Firefox 1.0 or higher with Macromedia Flash plug-in 7 or higher. Opera 8.5 works well also. If you experience trouble connecting to ACCESS, be sure you have the latest Flash Plug-in installed by right-clicking your mouse in the main browser window and selecting "about Macromedia Flash". This will take you to the Macromedia website where you can download the latest free plug-in.

Once you are connected to ACCESS, a login screen will appear (see Figure 9). Key in any user name along with the default password (comrex, case sensitive) to get to the ***Main User Interface*** display. This display is optimized for full-screen mode (F11 on most browsers) on a 1024x768 display.

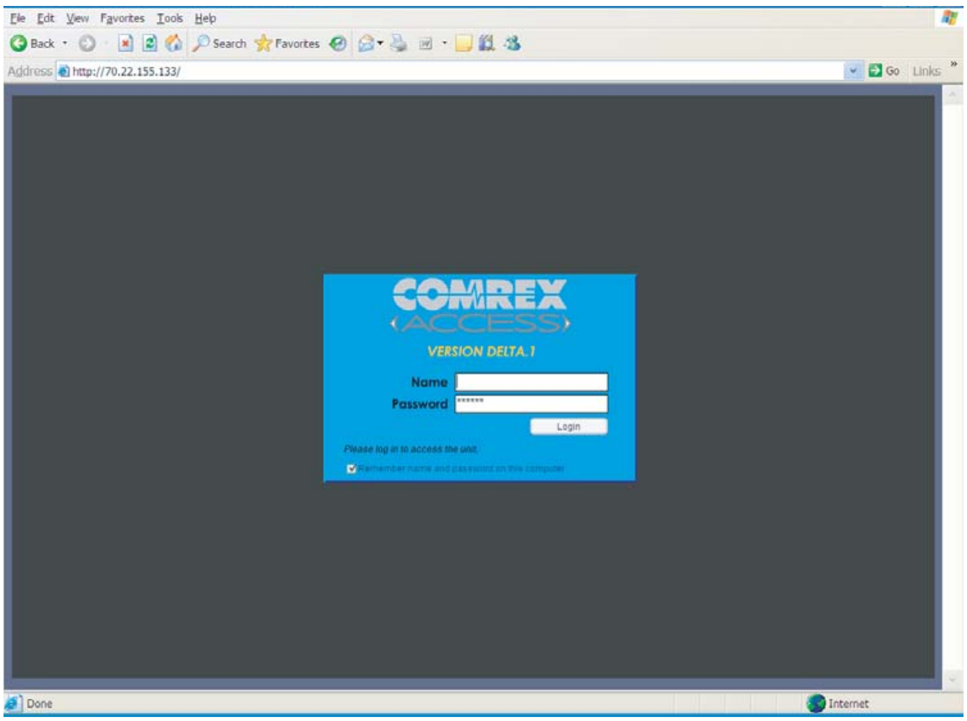


Figure 9 - ACCESS Login Screen

There are three main parts to the *ACCESS User Interface* screen:

1) Encoder Mode and Main Audio Meter — This displays the current setting of the ACCESS encoder. The level meters are defaulted to off to conserve bandwidth and client CPU, but when these are enabled this top bar emulates the front panel of ACCESS.
2) Tabs — Use these tabs to control and obtain status of ACCESS. They are described in detail in the next four sections.
3) Chat Window — Allows for a chat utility between any users that are logged into that particular ACCESS web interface. In addition, when ACCESS is connected to a remote user, chat text will appear from any users logged into the remote web interface.

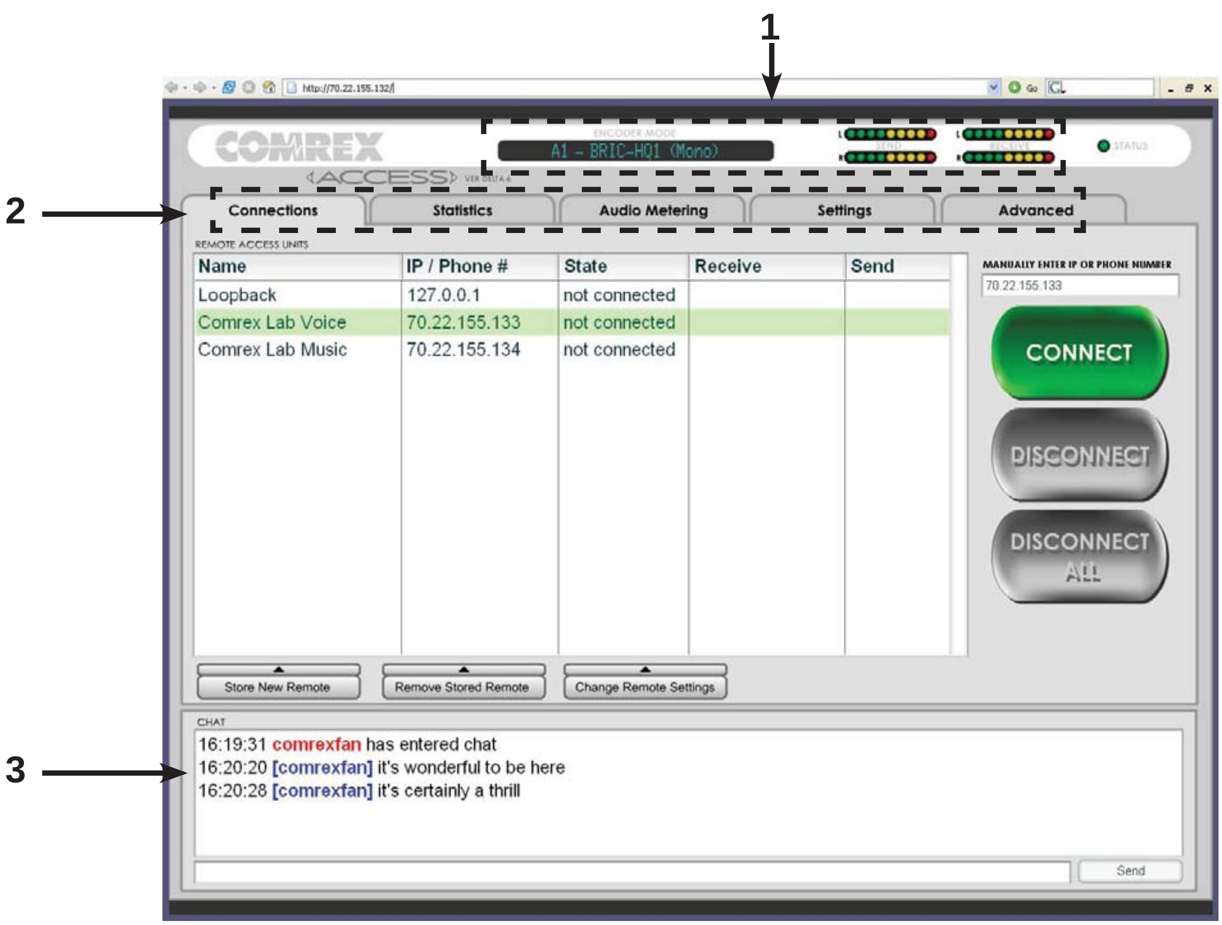


Figure 10 - ACCESS User Interface Screen

Connections Tab

The **Connections Tab** is the default setting for the *User Interface* (as shown in Figure 10). In this tab you can program and save the names and addresses of any remote units you connect to. This allows custom programming of policy parameters for each remote and allows point-and-click connect and disconnect. To add a remote ACCESS to the list, simply click **Store Remote Address** in the lower section. An input box will appear allowing you to enter a user name (which can be anything) and the IP address of the unit. You may remove any stored value simply by highlighting and clicking **Remove Remote Address**. Stored remote addresses are saved to system memory, where they will remain through power cycles.

The **Connection Tab** will also display **IP** and **Status** information of a remote ACCESS when it has initiated a connection to you. Their information will only appear while the connection is active.

By default, three users appear on the list. You may use any of these to test different encoder modes.

1) Loopback — Allows for connection between encoder and decoder in the same system.
2) Comrex Lab Voice — Allows testing back to the Comrex headquarters in Massachusetts, USA.
3) Comrex Lab Music — This additional user provides a music feed from the Comrex lab.

STATISTICS TAB

The information in the **Statistics Tab** either can be updated instantly or the system can provide a moving average of information over an interval of one second. The **Statistics Tab** defaults to **Moving Average** but can be changed to **Instantaneous** using the radio button in the upper right corner, shown as #1 in Figure 11.

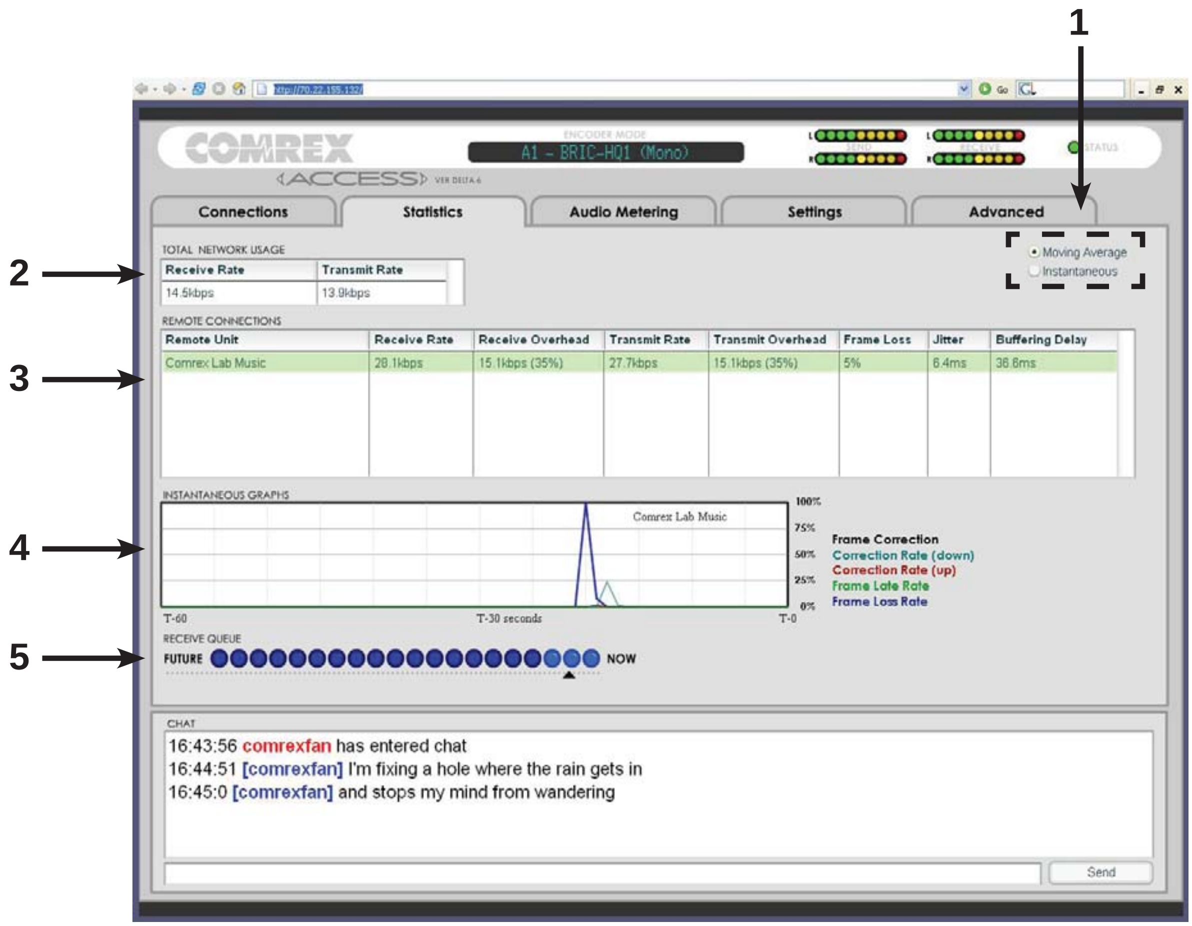


Figure 11 - Statistics Tab in the User Interface

The **Total Network Usage** field (#2 in Figure 11) delivers information on the total number of bits entering or leaving the ACCESS (including multiple connections if applicable), IP, UDP and RTP packet headers and coded audio.

The **Remote Connections** box (#3 in Figure 11) breaks this information down further. Because ACCESS is capable of more than one simultaneous connection (in some modes), each connection is listed independently. The raw **Receive Rate** and **Transmit Rate** are listed, along with an indication of how much overhead is required for the various IP headers on each packet. (See *About IP Audio Packets* section on page 24 for more information.) **Frame Loss** is also listed as an individual figure for lost and late packets. Finally, a **Jitter** figure is calculated to give an indication of the time difference between earliest and latest received packets, followed by an indication of how much delay is being added to the decoder to compensate for jitter.

A graphical representation of network quality vs. time is available in the **Instantaneous Graph** (#4 in Figure 11). The **Frame Loss** field provides a scrolling indication of the network quality by graphing the number of lost or late packets experienced by the decoder every second. You will also see graphs indicating correction of the audio buffer to compensate for a changed jitter buffer size. It's quite normal to see activity on both graphs, so don't panic if you see some spikes. ACCESS does a good job of covering up any network deficiencies while keeping the delay as low as possible.

Receive Queue and **Lag Cursor** fields (#5 in Figure 11) graphically show the actual size of the **Receive Jitter Buffer**. Packets can be considered as entering from the left side of this chart and leaving from the right. If the network experiences congestion, the graph will show the jitter buffer filling up to the left, and once the congestion is removed, the buffer will slowly wind down. Lost packets are shown when the buffer empties and the far right indication glows red.

The **Statistics Tab** provides feedback on settings made in the **Advanced Tab**. The meaning of items in the statistics tab is explained more fully in that section.

AUDIO METERING TAB

The **Audio Metering Tab**, as shown in Figure 12, provides a representation of **Input** and **Output** audio levels in several formats. Each of these meters (including the top section meters, which are always visible) may be turned **On** and **Off** individually. All audio meters are defaulted to **Off** when ACCESS is first enabled. This is because transfer of audio level information consumes bandwidth on the local network, as well as CPU cycles on the client computer. Whenever ACCESS is connected to a data constrained network (e.g. wireless), it is strongly recommended that these meters be **Off**, especially if the **User Interface** on the constrained network will also be accessed via the wireless network (e.g. from the studio end). The bandwidth requirements to drive the meters may affect performance of the audio codec.

The **Metering Quality** option (which is defaulted to low) adjusts how often the meters are updated—better networks can support higher quality settings. Refer to #1 in Figure 12.

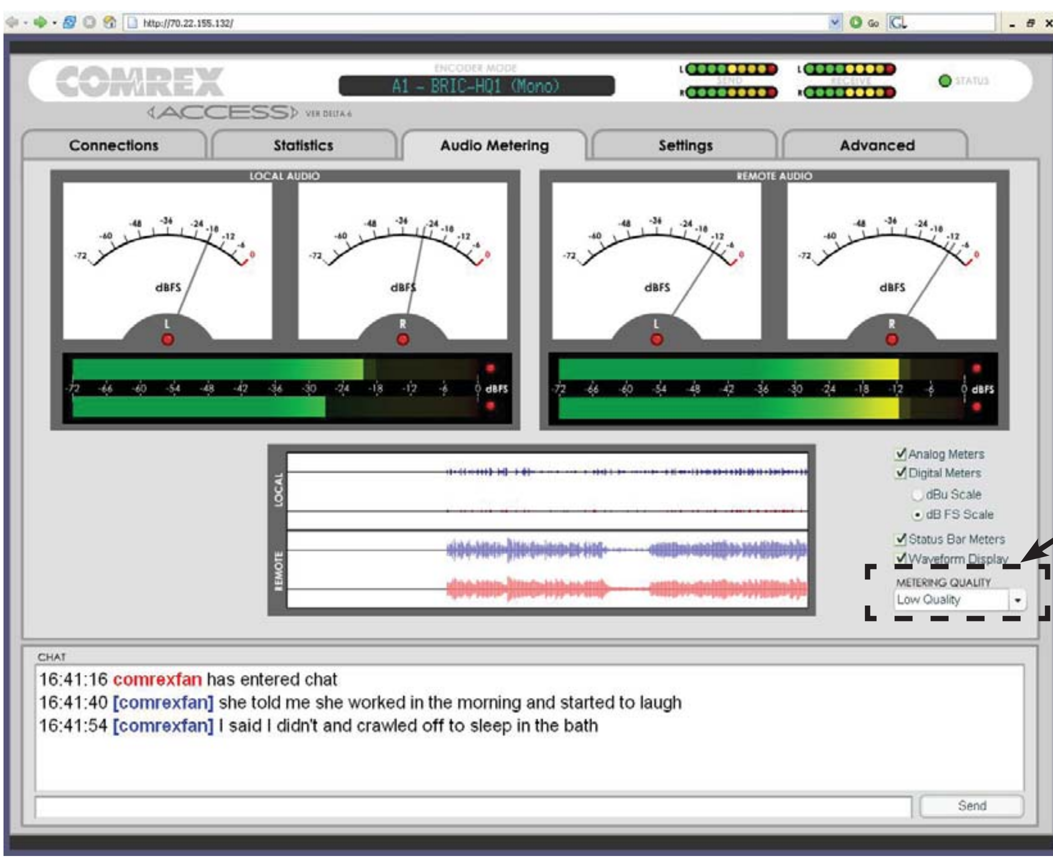


Figure 12 - Audio Metering Tab in the User Interface

Settings Tab

The **Settings Tab** is primarily used to choose which encoder will be used for outgoing connections. Each selection has a short description of the encoder's best use, along with its advantages and disadvantages. This brief description is located in the **Encoder Parameters** box, #1 in Figure 13. For a complete description of the different *Encoder Modes*, see *About The Algorithms* on pages 32-34. After selecting an encoder, be sure to click the **Change Encoder** button to verify your change (#2 in Figure 13).

Current firmware does not support the use of multiple decoders, so this option on the right hand side of the Settings Tab is currently greyed-out. The lower right section provides two options to control the built-in modem used for POTS codec calls (#3 in Figure 13). You can enable/disable the auto-answer function of the modem by clicking the box labeled **Automatically Answer Incoming Calls**. You can also pre-determine the maximum connect rate at which the modem will connect. See *Section 6 POTS Codec Connections* for more information.

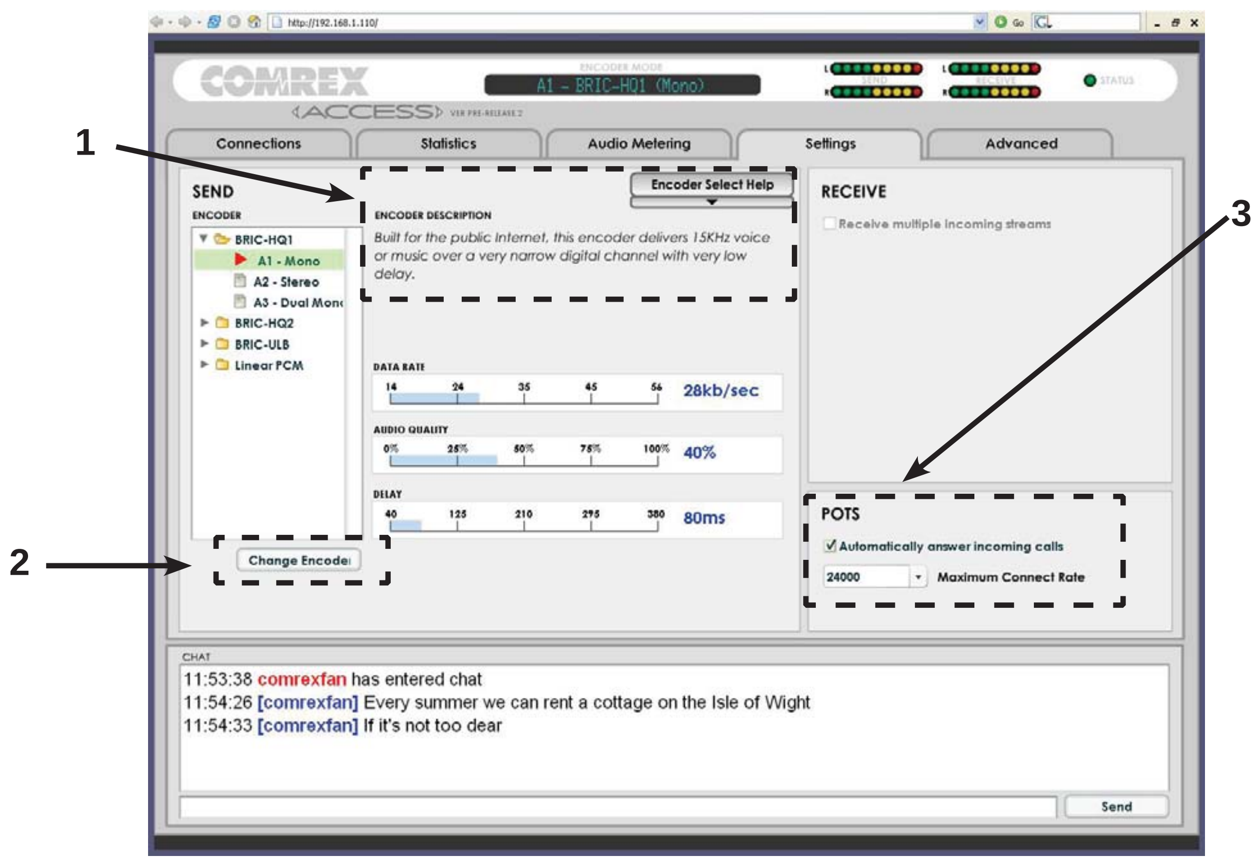


Figure 13 - Settings Tab in the User Interface

A LITTLE IP AUDIO PACKET BACKGROUND……

We'll be talking about the **Advanced Tab** next, but first here's some background on IP audio packets:

The ACCESS encoder works by converting audio to a digital stream and then running this stream through a digital compression encoder which reduces the amount of data in the stream. The output of the encoder is segmented into audio frames. Each frame is essentially a snapshot of the input audio at discrete intervals. This interval is typically around 1/50th of a second (20mS). The stream is then converted into IP packets for transmission over the Internet.

All data sent or received over the Internet must be in the form of IP packets. Each packet is sent individually with headers attached. By default, each audio frame forms the payload for one IP packet. On ACCESS, two headers are added to each packet. The first header describes the type of payload and the destination address. The second header includes a timestamp to allow the decoder to deal with the packet in the correct order. These headers are fairly long, and in some modes their size rivals the entire payload. We define all non-audio headers added to the packets as overhead. Each packet then enters the Internet, and the headers are read by the various routing hardware. The packets are then sent to their destination by one of many possible routings. Each packet sent can take a different routing to its destination, and some may be delayed due to network congestion.

On the decoder, incoming packets are received and stripped of their headers. Using the information from the timestamp, a stream of frames is organized into the correct order and placed into a buffer called a receive queue. This buffer is required due to the random arrival times of each packet. The receive queue grows and shrinks dynamically and automatically. It makes decisions based on the difference between early and late packets, as well as information gleaned from the network about congestion at the encoder.

Advanced Tab

For ACCESS to work correctly it must send data from the encoder in such a way that it does not overload the outgoing network. It also needs to be relatively immune from congestion caused by devices that may be utilizing the same network. Further, it must maintain the decoder receive queue at a place where packet loss is kept to an acceptable level, yet overall delay is minimized.

The **Advanced Tab** contains adjustments to the parameters that allow ACCESS to find the correct balance between audio delay and stability.

The following advanced settings affect the ACCESS encoder:

- **Error Correction Level** — This setting allows for previous audio frames to be reproduced in the current packet. In this way, if a packet is lost or late the audio data lost can be restored. FEC has a large impact on outgoing data rate, and shouldn't be used on constrained networks. The FEC level determines how many of the previous audio frames are included in the current packet.

- **Frames Per Packet** — Allows the inclusion of multiple audio frames into a single packet. Since only one set of headers is required for this compound packet, overhead is reduced and therefore overall data rate is reduced. Larger packet size means higher latency. Packet size is measured in audio frames.

The following advanced setting affect the ACCESS decoder:

- **Delay Cushion** — Instructs the **Receive Queue** to start at a certain size and not to drop below that value at any time. Higher **Delay Cushions** means higher latency. The **Delay Cushion** is measured in milliseconds.

- **Window** — This setting determines how long of a window (in seconds) is analyzed by the buffer management algorithm in determining whether to adjust the receive queue size. Lower values mean the buffer will adjust faster, while large values will slow buffer adjustments.

- **Loss Cushion** —This setting determines a threshold where the buffer correction algorithm will act. Lower values lead to more aggressive correction, while higher values produce lower delay.

Changes to the **Advanced Tab** settings can have dramatic impact on digital bandwidth, stability and delay. For most uses, it's best to leave the **Advanced Tab** settings at the factory defaults, as shown in Figure 14. The default setting may be reset any time by pressing the **Reset To Default** button. When non-default **Advanced Tab** settings are chosen, a warning icon will appear next to the **Change Remote Settings** box in the **Connections Tab**.

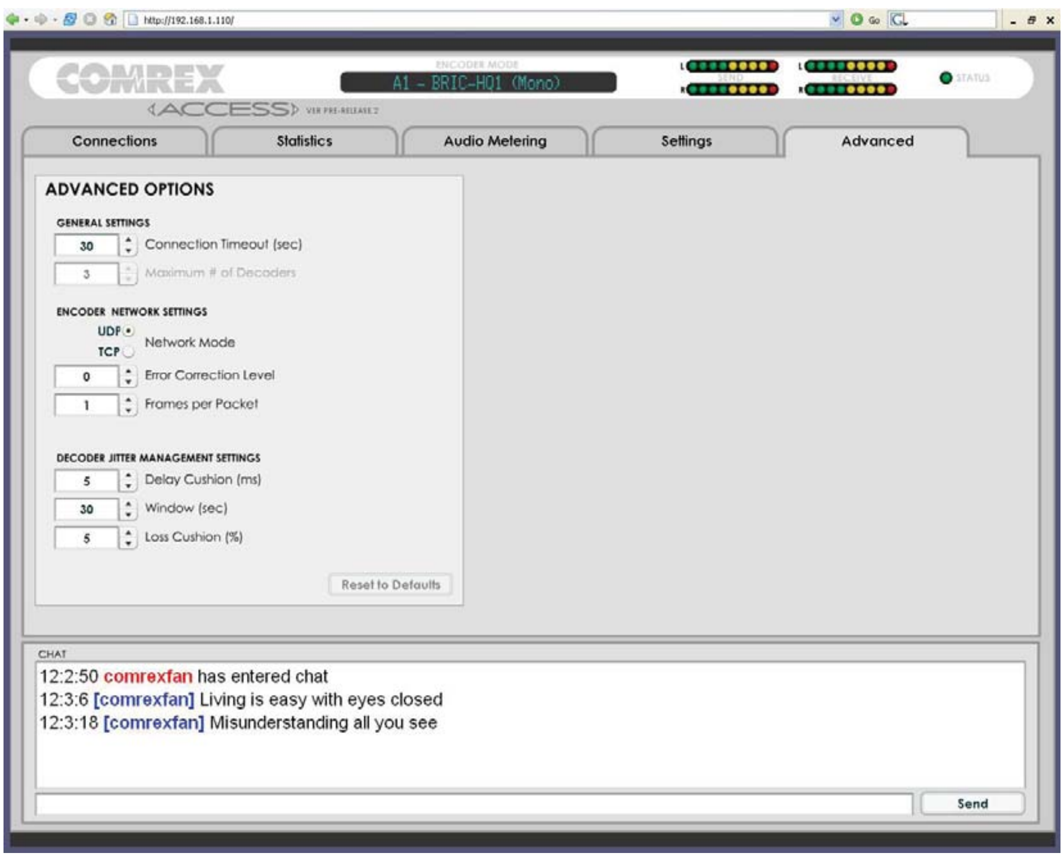


Figure 14 - Advanced Tab in the User Interface

SECTION 5 MAKING CONNECTIONS

CONNECTING

So now it's time to make a connection on ACCESS. We will assume that the proper network and audio connections have been made. There are two ways to make a connection on ACCESS, both of which are done from the **Connections Tab** in the ***User Interface***. The most common and simplest method is done by highlighting the remote user you wish to connect to and click **Connect**. When a connection is made (whether incoming or outgoing) the state message will change to **Connected** and status messages will appear in the **Send** and **Receive** columns, indicating the presence of valid outgoing and incoming ACCESS packets. Alternately, you may "hot-connect" by keying in the remote ACCESS IP address in the field labeled **Manually Connect to Remote IP Address**. In this case, as with incoming calls, the remote IP address will appear only during the connection, and will be removed upon disconnection.

When an ACCESS connection is made, several things occur:

1) Audio is transferred in one or both directions, based on the settings in the **Change Remote Settings** box for that connection. For additional information, please refer to the *CHANGE REMOTE SETTINGS* in section 8.

2) The chat box in the user interface includes any users connected to the remote ACCESS user interface.

3) ACCESS will deliver its four contact closure input signals to the remote ACCESS contact closure outputs.

4) ACCESS will deliver any serial data applied at the serial input to the serial output on the remote ACCESS.

DISCONNECTING

To disconnect any ACCESS connection, highlight it on the user list and click **Disconnect**. **Disconnect ALL** may be used when there are several connections active (See *SECTION 9 MULTI-STREAMING* for additional information.)

TROUBLESHOOTING

If your connection fails, it could be due to several factors:

1) The remote user may be off-line or suffering network congestion or failure.

2) The remote user may be behind a NAT or firewall preventing incoming connections.

3) The remote user may be connected to someone else. In this case, one-way connection may still be possible if you configure this connection to **Remote Only Sends Audio** using the **Change Remote Settings** box (in essence, engaging the multi-streaming option as described in section 9).

Note for users of Comrex ACCESS “test” lines: These addresses may experience high volume at times. If you are having trouble connecting, it may be that they are in use.

MORE ABOUT CONTACT CLOSURES AND SERIAL DATA

ACCESS uses a separate data channel to transfer chat, contact closures, and serial data. Be aware that this channel impacts the overall network utilization. For example, if your network is constrained, heavy use of ancillary data will impact audio performance.

SECTION 6 POTS (PLAIN OLD TELEPHONE SERVICE) CODEC CONNECTIONS

ACCESS is capable of connections over modem links. This mode emulates the function of Comrex POTS codecs, which have been used for years to deliver high quality audio over normal, dial-up telephone lines. This mode provides for a point-to-point connection between the codecs i.e. no internet access is used, and the call is placed directly from one ACCESS (or legacy codec) to the other.

In the current firmware, ACCESS is capable of connecting over dial-up phone lines to:

ACCESS Codecs
Comrex Matrix Codecs
Comrex BlueBox Codecs
Comrex Vector Codecs

Note: Backward compatibility to Hotline codecs is not supported.

POTS CODEC SET-UP FOR ACCESS COMPATIBILITY

The legacy codecs (Matrix, Vector or BlueBox) must be configured for operation in *Music Mode*, which will allow full-fidelity (up to 15KHz) connections. *Voice Mode* is not supported by ACCESS. Contact closures and ancillary data supported by legacy codecs are not supported by ACCESS.

ACCESS can only connect over its modem if no other IP-based connections are currently ongoing. There are no encoding algorithm choices when working in POTS. The encoder selected on the **Settings Tab** is ignored.

USING ACCESS WITH POTS

To use ACCESS on POTS, a normal, analog telephone line must be connected to the rear panel telephone line RJ-11 connector. If possible, try to obtain a true telephone company grade line, rather than an extension from your digital phone system. Under no circumstances should the raw extension from a digital phone system be attached to this port—you will likely damage ACCESS, your phone system, or both.

To initiate calls from ACCESS, simply create a remote connection with a telephone number as an address, rather than an IP address, in the **Connections Tab**. You may “hot-connect” by keying the phone number in the **Manual Enter** box. The remote POTS codec phone number may also be saved in a list, allowing for “point and click” connect and disconnect. To add a remote POTS codec to the list, select **Store New Remote**. Next, an input box will appear. Simply key in the user name and phone number. Either way, ACCESS will understand this to be a modem-based connection and, when selected, will change to the **Modem Status** display in Figure 15.

As long as no IP connections are present and the *Auto Answer* option is enabled on the **Settings Tab**, ACCESS will automatically answer incoming calls and switch to the **Modem Status** display.

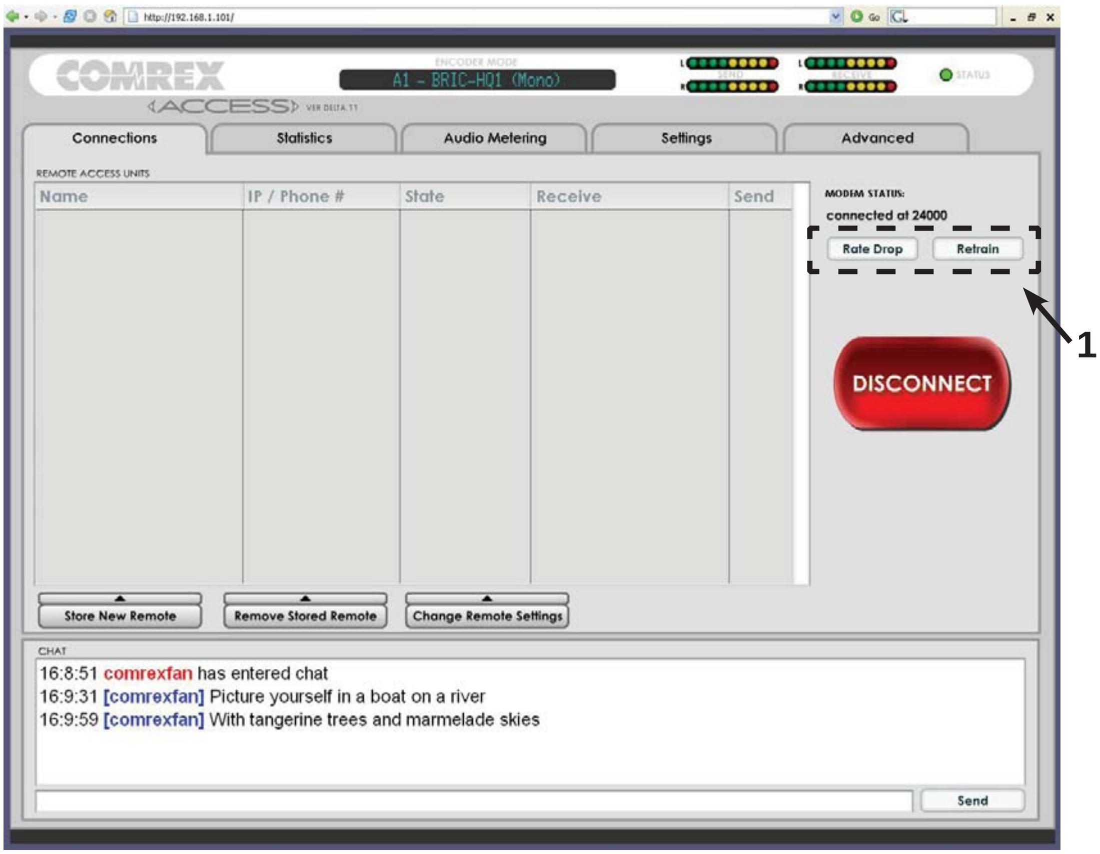


Figure 15 - Modem Status Display in the User Interface

RATE DROP VS. RETRAIN

The **Modem Status** display contains two user controls, **Rate Drop** and **Retrain** (#1 in Figure 15). These controls are similar in function to those provided on POTS codecs. ACCESS will initially connect at the best data rate supported by the telephone line, and will display that connect rate on the **Modem Status** page. You can force the system to drop to the next lowest connect rate at any time by clicking the **Rate Drop** button at any time. Audio transfer will be interrupted momentarily while the units negotiate the new connect rate. Alternately, you can force the system to initiate the entire training sequence again (the “chat” sounds heard at the beginning of a call) by clicking the **Retrain** button. You will lose audio for a longer time (approx 7 seconds) but the modems will completely re-equalize the connection and return audio when finished.

Once ACCESS has dropped to a lower rate, either by rate drop or retrain from either end, there is no way to force it to connect at a higher rate. If you want ACCESS to try again for a higher connect rate, you will need to disconnect the call and dial again.

In POTS mode, ACCESS uses a synchronous modem protocol. For this reason, there will be no IP statistics displayed on the **Statistics Tab** during any POTS-based connection.

Troubleshooting POTS Connection

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. Here's a short list of rules to follow for POTS codec connections:

1. Use the POTS codec on a direct telephone company line and avoid in-house phone systems. A line used by a fax machine usually provides this direct access. (Be sure to disconnect the fax machine before connecting the codec!)

2. 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 POTS codec 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 on the POTS codec, based on your findings, is highly recommended. MaxRate usually should be set at a level or two below the maximum unrestricted rate. This will provide a "guard band" of sorts against noise and corruption which may cause errors on the line.

7. If operation starts to degrade after a long period of connection, it may be that the phone line parameters have changed. These parameters are affected by factors such as time of day, weather and geographic location. The modems should be given the opportunity to renegotiate for these new parameters.

8. If you experience low connection rates or errors, try redialing. If that does not help, dial from the other end. If the call is long distance, try forcing the call to another carrier. If a good connection if found, keep that line up.

SECTION 7 ABOUT THE ALGORITHMS

ACCESS contains six different types of encoders and decoders for use on networks

BRIC-HQ1
(HIGH QUALITY 1)

This encoder/decoder provides 15 kHz voice/music transmission with extremely low delay and low network utilization. It supports mono, stereo, and dual-mono. Here are some details of *BRIC-HQ1*:

- **Low Delay** — *BRIC-HQ1* uses a 20mS audio frame, with an overall encode/decode time of around 80mS. This makes BRIC-HQ1 a good choice for real-time, interactive applications.
- **Low Digital Bandwidth** — *BRIC-HQ1* has a data rate of around 28 kbps for mono, and 56 kbps for dual mono allowing it to travel over medium-to-low speed networks.
- **Voice/Music Capable** — *BRIC-HQ1* is designed as a voice codec, but does a respectable job at encoding music.
- **Dual Mono Mode** — Supports the encoding of two independent audio channels, such as a dual-language broadcast. These two channels will be multiplexed into a single outgoing stream.
- **Stereo Mode** — This mode uses matrixing to deliver stereo audio at less than double the bandwidth.

BRIC-HQ2
(HIGH QUALITY 2)

This encoder/decoder provides high fidelity (12 or 15 kHz) mono or stereo transmission at low data rates and reasonable delay. Here are some details of *BRIC-HQ2*:

- **Medium Delay** — *BRIC-HQ2* uses a 64 or 80mS audio frame, with an overall encode/decode time around 360 mS. Interactive applications are possible using *BRIC-HQ2* in the forward direction and *BRIC-ULB* or *BRIC-HQ1* in the reverse.
- **Low Digital Bandwidth** — *BRIC-HQ2* encodes at 24 kbps for a full bandwidth mono signal. Stereo signals occupy 30 kbps. Dual mono is not supported in *BRIC-HQ2*.
- **Voice/Music Agnostic** — *BRIC-HQ2* utilizes a blend of different audio coding techniques, so it does a good job of encoding non-voice audio.
- **Mono/Stereo** — *BRIC-HQ2* has stereo modes which utilize a parametric stereo effect; so it is not possible to send independent audio over the L&R channels. The channels must be a related stereo image. Use *BRIC-HQ1* or *AAC-LD* when *Dual Mono* is required.
- **Audio Bandwidth** — *BRIC-HQ2* default modes utilize a 32 kHz sampling rate to deliver 15 kHz audio fidelity. BRIC-HQ2 12K modes utilize a 26 kHz sampling rate to achieve a 12 kHz audio fidelity. Because the data rate is the same between the two modes, *BRIC-HQ2 12K* can be considered to sacrifice some slight high end fidelity in exchange for overall lower audio coding artifacts.

BRIC-ULB (Ultra Low Bitrate)

This encoder/decoder provides 7 kHz voice audio transmission with extremely low delay and extremely low network utilization. Due to its low digital bandwidth, it is considered to be the most robust mode for use on constrained networks. Here are some of the details of *BRIC-ULB*:

• **Low Delay —** *BRIC-ULB* uses a 20mS audio frame, with an overall encode/decode time of around 80mS. This makes BRIC-ULB a good choice for real-time, interactive applications.
• **Low Digital Bandwidth —** *BRIC-ULB* has a data rate of around 12 kbps, allowing it to travel over very low speed networks. Also, since BRIC-ULB is so efficient, error correction may be added in many situations without congesting the network.
• **Vocoder —** *BRIC-ULB* relies on an voice based vocoder, which is the same principal utilized in many digital mobile phones. The difference is that while mobile phone vocoders typically provide about 3 kHz audio bandwidth, *BRIC-ULB* delivers more than twice that fidelity, providing a much more listenable and less fatiguing sound. *BRIC-ULB* is optimized for human voices. It does a respectable job of encoding background noise and crowds, but music tends to suffer rather dramatically on *BRIC-ULB*.
• **Mono** — Only a single audio channel is supported on *BRIC-ULB*.
• **Dynamic Data Rate** — The *BRIC-ULB* encoder adapts its outgoing audio frame size based on the complexity of the incoming audio.

Linear PCM

This encoder does not compress audio at all. It uses a 48 kHz sampling rate and simply applies small frames of linear audio to IP packets. This mode is only useful on high bandwidth LAN or managed WAN environments. *Mono Mode* requires a network capacity of 768 kbps while *Stereo Mode* requires a network bandwidth over 1.5 Mb/s.

HE-AAC (Optional Upgrade)

This algorithm is the standardized version of an algorithm previously known as AAC+. It utilizes a moderate digital bitrate (46-64 kbps) and produces near-transparent audio quality. It’s actually a combination of AAC (Advanced Audio Coding) which is the result of collaboration between several of the best audio coding companies worldwide and SBR (Spectral Band Replication) which further reduces data rate by replicating high frequency bands. *HE-AAC* has the highest delay figure of any ACCESS algorithm (~600mS), so it is best used in situations where interactivity isn’t important. *HE-AAC* (and close derivatives) are often used as the main audio codec for digital radio and satellite networks.

AAC-LD
(OPTIONAL UPGRADE)

This algorithm is an extension of AAC developed by the FhG IIS, who are the contributors to AAC and primary inventors of the MP3 algorithm. It's quality is superior to MP3 at similar bitrates (64-128 kbps) but it exhibits very low delay (100mS). This choice is best when reasonable network throughput is assured, near-transparent audio is required and interactivity is needed.

SECTION 8 CHANGE REMOTE SETTINGS

Because ACCESS has so many modes and functions, determining exactly what will happen when they are mixed can be confusing, especially in a multi-streaming environment (see *SECTION 9 MULTI-STREAMING*). Here we outline the policies the ACCESS follows when making or receiving connections of different types.

ACCESS by nature treats incoming and outgoing connection differently. An outgoing connection is that which is initiated by the user interface of the local ACCESS. An incoming connection is defined as not being initiated from the local codec (although an outgoing connection will usually be created in response to the incoming stream).

The outgoing ACCESS is the master and has control over how the connection is made in both directions. By default, an incoming ACCESS delivers a reverse channel of the same algorithm and mode as the outgoing ACCESS. Once ACCESS has chosen an encoder (regardless of whether the encoder was chosen manually by the user or automatically in response to an incoming connection), ACCESS will remain with that encoder until all connections are dropped. Additional connections, whether incoming or outgoing, will be served by the stream from the original encoder.

Note: ACCESS decoders will automatically adapt to whatever incoming stream they receive, so no setup is ever required for the decoder. All settings affect encoders in each direction only.

If you are making outgoing connections, you can alter the policy used when connecting to a particular unit using the **Change Remote Settings** option in the **Connection Tab**. This topic can be a bit confusing, so bear with us as we'll describe the concepts then give some examples.

We'll refer to each of our remote codecs listed in our **Connections Tab** as peers. Figure 16A shows a typical list of peers on a **Connections Tab**. As mentioned, the default behavior of each peer is to return an audio channel of the same mode and algorithm as the one you've delivered to it. To change this behavior for a particular peer, select it (green) and click the **Change Remote Settings** dialog box. You are presented with a list of options that will allow you to change the behavior of connections to that particular peer. These settings will remain with that peer until they are changed or the peer is deleted.

Whenever a peer has had its **Change Remote Settings** altered from the default, it will glow yellow in the **Peer List** as a warning.

As shown as #1 in Figure 16A, there are three options that define whether a full-duplex or one-way connection is established with that peer. The options are:

1) **Remote Sends and Receives Audio** (default) — Full duplex connection.
2) **Remote Only Sends Audio** — Connection only in the direction to your ACCESS*.
3) **Remote Only Receives Audio** — Connection only in the direction from your ACCESS*.

** These settings become important in the multi-streaming application in the next section.*

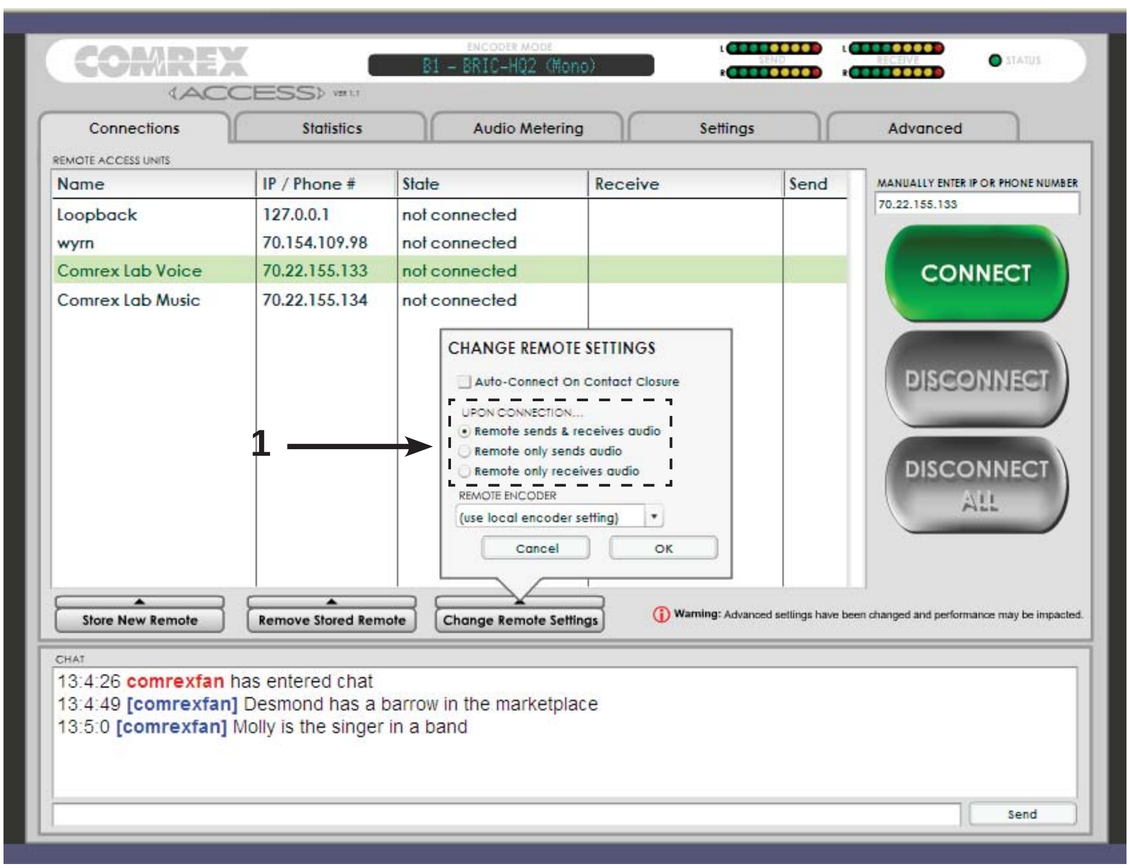


Figure 16A - Change Remote Settings Dialog Box

In the event that you have selected that a peer send audio to you, you can also choose which algorithm for that peer to use as its encoder. Do this by selecting from the **Remote Encoder** drop-down list, shown as #2 in Figure 16B. The options are:

1) **Use Local Encoder Setting** (default) — The remote peer sends back the same type of audio it receives from you.
2) **Do Not Change** — The peer will send whatever has been selected as its own encoder in its own user interface.
3) **Individual Encoder Selection** — You can choose for this peer to always send you a certain type of audio regardless of how it's set or what you're sending it.

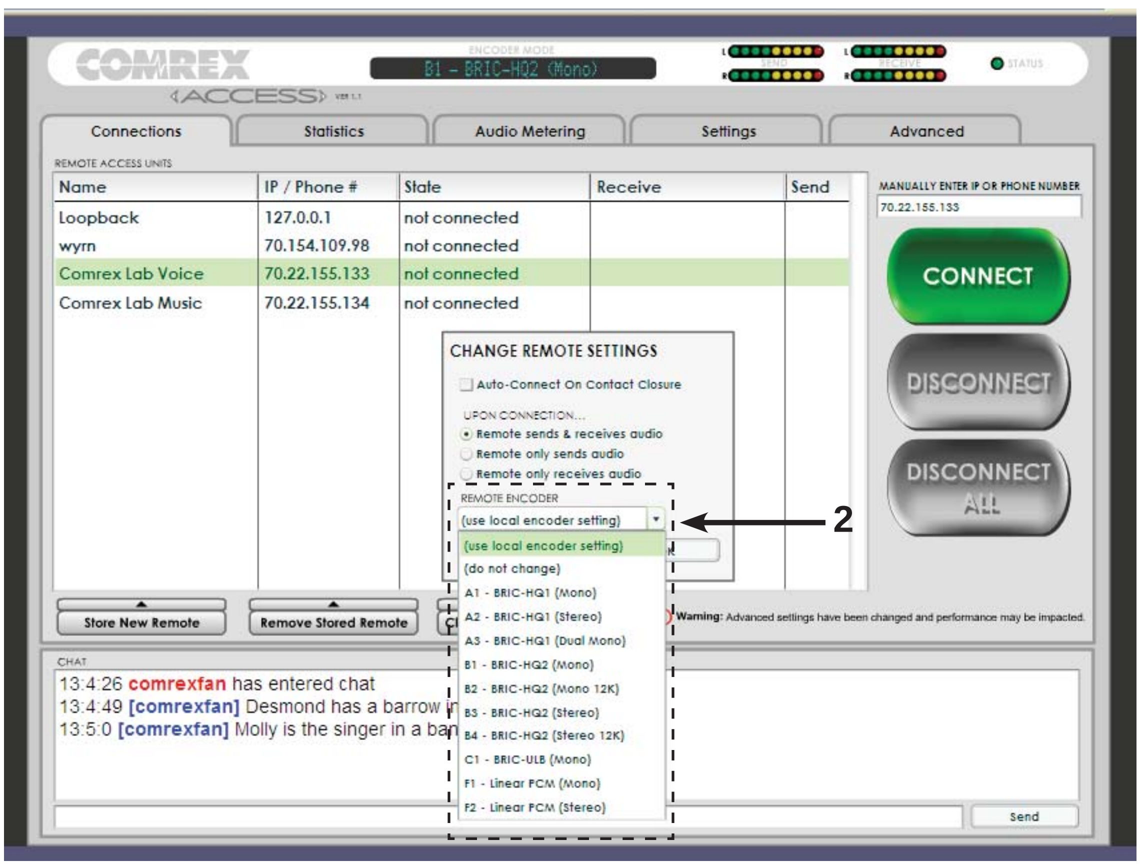


Figure 16B - Change Remote Settings Dialog Box with Remote Encoder List

ACCESS can be configured to make and maintain a connection. Four steps must be taken to fully enable this feature:

1) Highlight the remote user you wish to auto connect to from the list of peers in the **Connections Tab**. The connection will be made using the peer's default settings.

2) From the **Change Remote Settings** tab, click on the box next to the **Auto-Connect On Contact Closure** (see #3 in Figure 16C). An asterisk will appear next to the peer that you selected in step 1, shown as #4 in Figure 16C. Note: Only one peer at a time may be set for **Auto-Connect On Contact Closure.**

3) Set the #4 input contact closure for **Auto-Connect On Closure**. This is done via the **Contact Closure I/O Configuration** screen in the *Config* program. To gain access to the *Config* program attach a monitor and keyboard to the ACCESS rear panel. Next, press **ALT-S** to enter the pull-down menu, press **ALT-C** to select **Contact Closure I/O**. With this change, contact closure #4 will no longer be available as an end-to-end signal.

4) Make the closure on contact #4 input. Pinouts can be found on page 9.

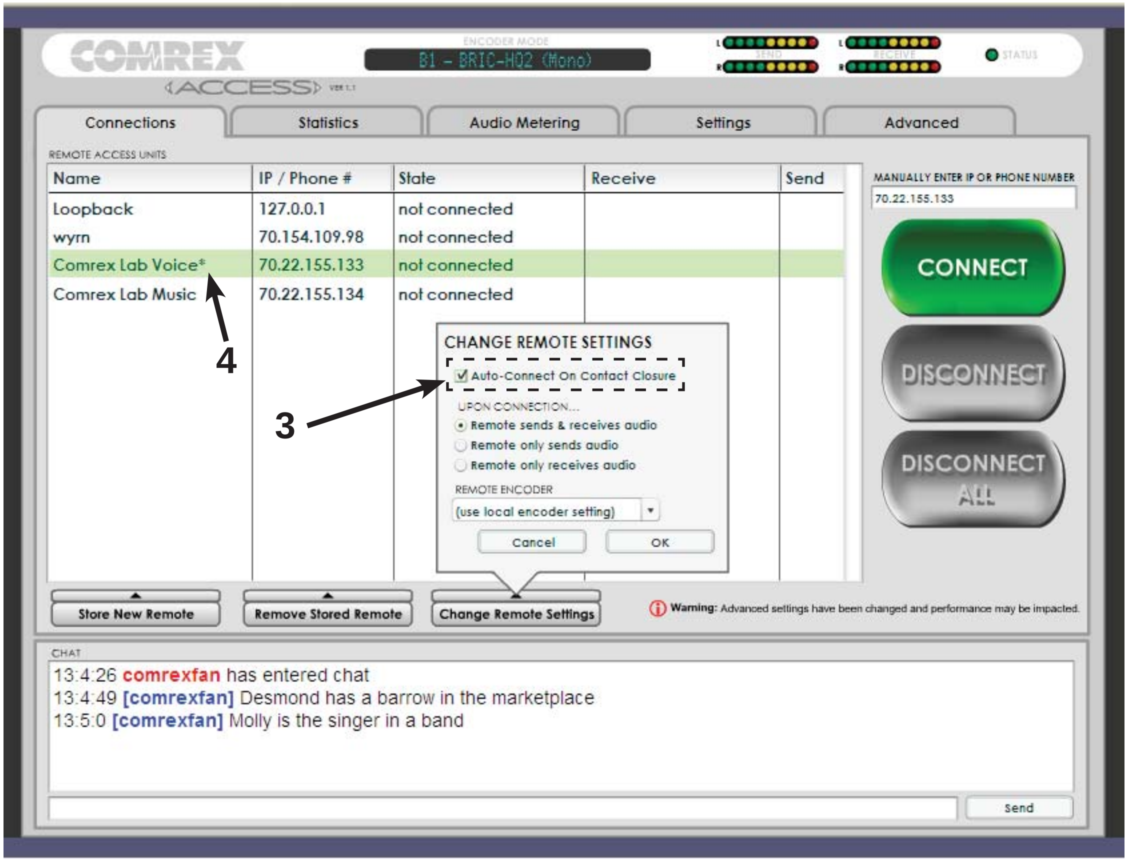


Figure 16C - Change Remote Settings Dialog Box with Auto-Connect Enabled

Here are those examples we promised:

1) ACCESS A has ACCESS B in its list. User A selects ACCESS B and chooses **Remote Only Sends Audio** from the **Change Remote Settings** box. ACCESS A connects to ACCESS B. ACCESS A hears audio from ACCESS B. ACCESS B hears nothing. The connection is ended, and ACCESS B connects to ACCESS A. Audio is heard in each direction (because ACCESS B is running the show).

2) ACCESS A has ACCESS B in its list. User A has gone to the **Settings Tab** and selected *BRIC-ULB* its outgoing encoder. User A then selects ACCESS B in his list and chooses **Remote Sends and Receives Audio** (default) from the **Change Remote Settings** box. User A also selects *BRIC-HQ2 Stereo 15K* from the **Remote Encoder** list. Whenever User A connects to ACCESS B, ACCESS B will send *BRIC-HQ2 Stereo 15K* to ACCESS A. ACCESS A will send *BRIC-ULB*. If after disconnect User C connects to ACCESS B using his default settings, ACCESS C will get back the same as it sends (because ACCESS C is now master).

3) Regardless of what settings User A has changed on ACCESS A, ACCESS B will follow its own policies on outgoing connections made to ACCESS A and ACCESS C.

Clear enough? Didn’t think so..let’s mix things up a bit more - see next section.

SECTION 9 MULTI-STREAMING

ACCESS supports the ability to run one encoder per box. But this single encoder stream may be sent to up to nine destinations simultaneously. We call this capability multi-streaming, since the encoder creates a separate but identical outgoing stream to each decoder. Note: Your Internet connection must be able to support these streams. For example, if your encoder runs at 35 kbps network utilization, sending to two locations will require 70 kbps upload speed from your network.

Multi-streaming should not be confused with IP Multi-cast, which will be supported in future ACCESS firmware.

Each ACCESS can also run only one decoder. So it's important that in a multi-stream environment, a maximum of one stream is sent in the reverse direction. This means that users interested in hearing a multi-stream must turn off their encoders.

This can be a bit confusing because multi-streams can be initiated from either end of the link.

Figure 17 shows an ACCESS multi-stream arrangement. ACCESS A is the multi-streamer, with ACCESS B, C, and D listening to the same audio. In this scenario, ACCESS A has made all the outgoing connections. User A has set his ACCESS for *BRIC-HQ1 Mono*. Before making any connections, User A determined he wanted to hear audio back from ACCESS B. He left the **Change Remote Settings** for ACCESS B to the default **- Remote Sends and Receives Audio**. He next selected ACCESS C and D from the list and set the **Change Remote Settings** to **Remote Only Receives Audio**. When he makes his three outgoing connections, he'll hear back only from ACCESS B in *BRIC-HQ1 Mono*. He'll hear nothing from ACCESS C or D, but they'll receive him in *BRIC-HQ1 Mono* .

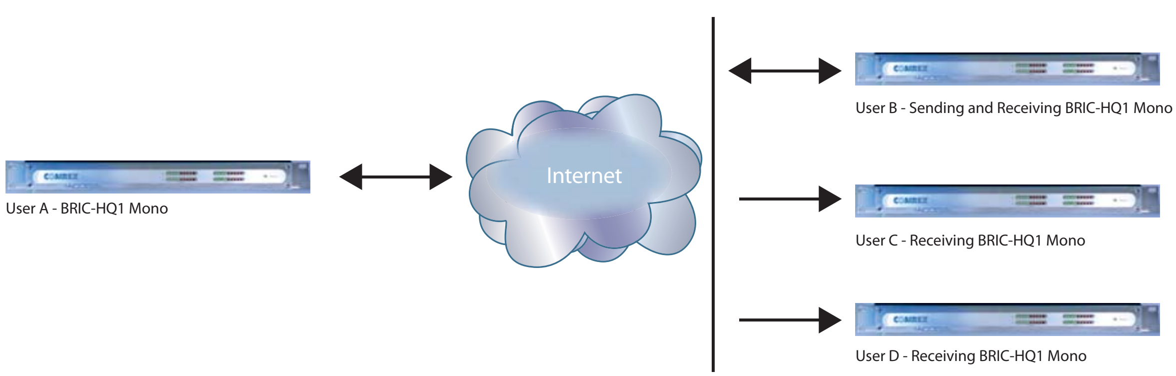


Figure 17 - Multi-Streaming Arrangement

Alternately, ACCESS A can act as an inbound streaming server. Using this technique, it's the responsibility of the receivers to set their policies not to send any audio to ACCESS A. ACCESS A will decode the first reverse channel it receives (and follow the encoder policy as dictated by the first connection). In this scenario ACCESS B has connected first in full-duplex and established a connection using *BRIC-HQ1 Mono*. Subsequent listeners will not be allowed to connect until they set a policy for ACCESS A as **Remote Only Sends Audio**. ACCESS C and D will hear the same *BRIC-HQ1 Mono* encode stream that was set up by ACCESS B, even if ACCESS B subsequently disconnects.

SECTION 10 ADVANCED TOPICS

This section discusses some frequently asked questions (and possible solutions) encountered when setting up, configuring, troubleshooting and achieving optimum ACCESS performance.

Q: How do I choose which encoding algorithm to use?

A: ACCESS offers a very wide range of encoding algorithms. To some this may seem daunting. Here's a short guide and comparison chart on how to choose what's best for your application:

1) *Do I have lots and lots of bandwidth?* If you're running on an entirely unconstrained network like a campus LAN or local Wi-Fi, *Mono* or *Stereo Linear Mode* will offer the highest audio quality with lowest delay. If you're hitting the public Internet at any point in the link, however, avoid *Linear Mode*.

2) *Do I require interactivity?* If you need to chat back and forth across the link, choose one of our low delay algorithms like *BRIC-ULB*, *BRIC-HQ1*, or *AAC-LD*. The deciding factor between these algorithms is digital bandwidth — *BRIC-ULB* uses very little, *BRIC-HQ1* uses more, and *AAC-LD* (optional upgrade) requires a relatively fat pipe.

3) *Is audio quality the paramount concern? HE-AAC* (optional upgrade) is the best choice for applications that need excellent audio quality. If delay is also a concern, consider *AAC-LD* (optional upgrade).

4) *Am I running on a constrained network?* If your Internet pipe is subject to being throttled, use *BRIC-ULB* for mono voice audio and *BRIC-HQ2* for stereo voice or music. These algorithms offer the absolute highest quality in exchange for extremely low network bandwidth.

5) *Do I need to deliver two unrelated audio signals to the same location? BRIC-HQ1* and *AAC-LD* (optional upgrade) offer *Dual Mono* options that allow uncorrelated signals (such as dual language broadcasts) to be combined to a single outgoing stream. Note: It isn't possible to send one stream to location A and one to location B. However, it is possible to send the combined stream to locations A and B and have them tap only their respective channels (although this can be a confusing solution subject to operator error).

Comparison Chart for ACCESS Settings

Required Bitrate	Coding Delay	Audio Bandwidth	Setting	Mode	Notes
			BRIC HQ1 sends good quality audio over narrow digital channels with low delay.		
28 kb/s	80 ms	15 Khz	**1A**	**Mono**	compatible with earlier Comrex POTS codecs on standard phone line connections
42 kb/s	80 ms	15 Khz	**1B**	**Stereo**	
56 kb/s	80 ms	15 Khz	**1C**	**Dual Mono**	allows independent programming to be sent on L&R channels
			BRIC HQ2 sends excellent quality audio over narrow digital channels with moderate delay.		
24 kb/s	320 ms	15 KHz	**2A**	**Mono**	
24 kb/s	340 ms	12 KHz	**2B**	**Mono**	reduced bandwidth with fewer coding artifacts
30 kb/s	320 ms	15 KHz	**2C**	**Stereo**	
30 kb/s	340 ms	12 KHz	**2D**	**Stereo**	reduced bandwidth with fewer coding artifacts
			BRIC ULB for "worst case" networks - delivers 7 KHz voice at ultra low bitrates with low delay (Not recommended for music)		
12 kb/s	80 ms	7 KHz	**3A**	**Default**	lowest bitrate of any BRIC algorithm
			HE-AAC provides near transparent audio at low data rates - for situations where latency is not important.		
32 kbs	600 ms	20 KHz	**4A**	**Mono**	
48 kb/s	600 ms	20 KHz	**4B**	**Stereo**	
64 kb/s	600 ms	20 KHz	**4C**	**Dual Mono**	allows independent programming to be sent on L&R channels
			AAC-LD requires higher data rates but provides near transparent voice or music with low delay.		
64 kb/s	80 ms	20 KHz	**5A**	**Mono**	
96 kb/s	80 ms	20 KHz	**5B**	**Stereo**	
128 kb/s	80 ms	20 KHz	**5C**	**Dual Mono**	allows independent programming to be sent on L&R channels
			Linear delivers transparent audio with no compression and very low delay - for use on high throughput networks.		
768 kb/s	40 ms	20 KHz	**6A**	**Mono**	
1536 kb/s	40 ms	20 KHz	**6B**	**Stereo**	

BRIC and HE-AAC are suitable for the Public Internet.
AAC-LD requires reasonably good QOS (Quality of Service).
Linear requires managed networks with extremely high throughput.

Q: Can I make ACCESS maintain an IP connection regardless of network status?

A: Yes. This can be done two ways:

1) Because ACCESS uses an Internet protocol known as UDP, there isn't actually an Internet "connection" going on at all. UDP simply sends data into the network regardless of what's going on further down in the network. ACCESS does communicate with the far end unit via a separate "heartbeat" signal and will terminate the connection if that heartbeat isn't received for 30 seconds. In the case where this isn't desirable, the timeout option can be defeated completely in the **Advanced Tab** by setting the **Connection Timeout** variable to zero.

2) ACCESS can also be configured to make and maintain a connection whenever an external contact closure is present. First, enter the *Config* program by attaching a monitor and keyboard to the ACCESS rear panel. Press **ALT-S** to access the pull-down menu, then select **Contact Closure I/O**. In this menu, under **Contact Closure 4 Input**, select **Auto Connect on Closure**. To choose which address is to be connected to when the contact is closed, it is necessary to enter the web-based interface as described in section 4. As described more fully in section 8, you will choose a peer for automatic connection using the **Change Remote Settings** options for that peer.

These settings will change the way ACCESS works in the following ways:

a) **Contact Closure #4** will no longer be available as an end-to-end signal.

b) When **Contact Closure #4 Input** is closed, ACCESS will make a connection automatically to the address specified (shown by an asterisk next to the peer selected). It will maintain that connection the entire time the closure is maintained, and will end the connection when the closure is opened.

c) ACCESS will use the default settings of the destination address.

Q: Can I get a remote indication that ACCESS is connected to someone?

A: Yes. Using the *Config* program, you can re-assign contact **Closure Output #4** to trigger whenever the ACCESS front panel **Ready** light is lit, indicating a valid incoming connection. Get to the *Config* program by attaching a monitor and keyboard to the ACCESS rear panel. Press **ALT-S** to access the pull-down menu, then select **Contact Closure I/O**. In this menu, under **Contact Closure 4 Output**, select **Closure While Connected**. This will change the way ACCESS works in the following ways:

a) **Contact Closure #4** will no longer be available as an end-to-end signal.
b) Whenever ACCESS detects a valid incoming stream, it will trigger **CC #4** and maintain it until all valid connections stop.

Q: What steps should I take when I'm having connection problems with ACCESS?

A: There are several steps you can take to determine that cause of poor IP connection using ACCESS. The first step is to determine whether the problem is occurring in one direction or both. If in only one direction, take a look at network usage patterns on the local end of each ACCESS. If someone else on your LAN is downloading large files on the decoder side (or uploading large files on the encoder side) this may cause some performance issues. You may need to ask them to temporarily cease activity, or investigate a network router solution that will offer ACCESS priority over other traffic. Next, take a look at the **Status Tab** on the ACCESS that is decoding the faulty audio. Take a look at the jitter figure for your incoming connection. If this number is varying dramatically (good networks keep this figure below 50mS) then you may need to increase the **Delay Cushion** setting on the **Advanced Tab**. Although it will increase your audio time delay, you may find increasing the cushion by 100-300mS or more will result in more stable connections, since the jitter buffer manager will no longer attempt to reduce delay by making the buffer smaller than the cushion.

Q: How can I optimize settings for EVDO, UMTS, or other wireless access?

A: Changes for these networks are best done on the wireless transmit side of the link. Since there is typically already a substantial delay in these networks, it's often not a priority to keep ACCESS delay to the absolute minimum. Using the **Advanced Tab** on the ACCESS on the wireless end, increase the **Frames per Packet** setting to 4. This will reduce overall bandwidth and enhance reliability on many networks. You may also need to increase **Delay Cushion** on the non-wireless decode side as described in the previous answer.

Q: I'm paying for my network bandwidth by the megabyte. How can I conserve?

A: Set both ACCESS to *BRIC-ULB*, which uses by far the lowest amount of data. On both ends, go to the **Advanced Tab** and set **Frames per Packet** to at least 4. This will decrease overhead. Finally, if you don't require audio in both directions, disable the return channel using the **Change Remote Settings** option in the **Connections Tab**. As a guide, an ACCESS set this way will average about 8 minutes of talk time per megabyte in each direction.

Q: My IT guy wants to know what protocols and ports are being used. Can you enlighten him?

A: Sure. The ACCESS user interface uses TCP ports 80 and 8080. Unless you intend to "drive" ACCESS from the outside world, you don't need to worry about opening or forwarding these ports through firewalls. The audio streaming app uses UDP port 9000. This may need to be opened on some firewalls or port forwarded through NATs in order to accept incoming calls. Note: This is UDP only. A common mistake is to open TCP instead.

Q: My IT guy is concerned about security and wants to know what services are open on this box.

A: As mentioned, we're serving an HTML/XML page on the well known server ports, 80 and 8080. Our streaming application is UDP/RTP on port 9000. SSH is enabled by default but requires a passkey. You can disable it completely in the config program (with a monitor and keyboard on the ACCESS) by selecting settings->advanced and clicking off the box labeled enable remote access (SSH). Leaving SSH enabled will help if Comrex support needs to interface with your ACCESS.

Q: That pesky IT guy is at it again. He wants to send pings and traceroutes from the ACCESS and check logs under Linux. Help?

A: A limited user account is available under the Linux OS to perform these functions. To get there, go to the *Config* program (with a monitor and keyboard on the ACCESS) and select Alt-S->Actions->Command Prompt. You'll get directly to a shell prompt with the ability to run these commands.

Q: Can I change the ports used for the User Interface and/or Streaming?

A: The **User Interface** service ports cannot be changed in current firmware. Contact Comrex support for instructions on how to change the application port, but be aware this will impact compatibility with other ACCESS in the field.

Q: How can I change modem parameters like dial-tone-detect and ring cadence detection?

A: Contact Comrex for more info on this.

Q: I notice on the Advanced Tab that I can change my streaming from UDP to TCP. Should I?

A: Not if you want the best overall performance. ACCESS is optimized in terms of data rate, stability and delay to use UDP. TCP mode increases overhead and delay, and is included only for environments where UDP is hopelessly blocked by a firewall. ACCESS decoders do listen to both TCP and UDP ports and choose whichever arrives first. If an ACCESS gets an incoming TCP connection, it will establish TCP in the other direction automatically. One other note for use with TCP — most of the information presented on the **Statistics Tab** is generated by the UDP functionality, so you won't see much here using TCP.

SECTION 11 CONFORMITY AND REGULATORY INFORMATION

SUPPLIERS' DECLARATION OF CONFORMITY

Place of Issue: Devens, Massachusetts

Date of Issue: January 23, 2006

Comrex Corporation, located at 19 Pine Road, Devens, MA in the United States of America hereby certifies that the Comrex ACCESS Rack bearing identification number US:DXDMD01BACCRK complies with the Federal Communications Commission's ("FCC") Rules and Regulations 47 CFR Part 68, and the Administrative Council on Terminal Attachments ("ACTA")-adopted technical criteria TIA/EIA/IS-968, Telecommunications – Telephone Terminal Equipment – Technical Requirements for Connection of Terminal Equipment To the Telephone Network, July 2001.

Thomas O. Hartnett, Vice President, Comrex Corporation

Note: This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

EC DECLARATION OF CONFORMITY FOR R&TTE DIRECTIVE

We:

Manufacturer's Name: Comrex Corporation

Manufacturer's Address: 19 Pine Road
Devens, MA 01434

hereby declare on our sole responsibility that the product:

Comrex ACCESS Rack
Digital Audio Codec

to which this declaration relates is in conformity with the essential requirements and other relevant requirements of the R&TTE Directive (1999/5/EC). This product is compliant with the following standards and other normative documents:

European EMC Directive (89/336/EEC)
EN 55022:1998/A1:2000, Class A Conducted and Radiated Emissions
EN55024: 1998/A1:2001/A2:2003 (Immunity, ITE Equipment)
Low Voltage Directive (72/23/EEC)
EN 60950-1: 2001

Information regarding configuration of this equipment for operation on the telephone networks of the EC countries may be found in the Comrex ACCESS Rack product manual.

Contact person: Thomas O. Hartnett, V.P., Engineering

Signed: ______________________

Date: 23 January 2006

U.S. and Canadian Regulatory Information for the ACCESS Rack

This equipment complies with Part 68 of the FCC rules and the requirements adopted by the ACTA, as well as the applicable Industry Canada technical specifications. On the bottom of this equipment is a label that contains, among other information, a product identifier in the format US: DXDMD01BACCRK. If requested, this number must be provided to a U.S. telephone company.

Telephone line connections to the Comrex ACCESS Rack are made via an RJ11C jack. A plug and jack used to connect this equipment to the premises wiring and telephone network must comply with the applicable FCC Part 68 rules and requirements adopted by the ACTA. A compliant telephone cord and modular plug is provided with this product. It is designed to be connected to a compatible modular jack that is also compliant. See installation instructions for details.

The REN is used to determine the number of devices that may be connected to a telephone line. Excessive RENs on a telephone line may result in the devices not ringing in response to an incoming call. The sum of RENs should not exceed five (5.0). To be certain of the number of devices that may be connected to a line, as determined by the total RENs, contact the local telephone company. The REN for the Comrex ACCESS Rack is 0.1, and is shown as the digits represented by ## in the product identifier US: DXDMD###ACCRK.

If the Comrex ACCESS Rack causes 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 right to file a complaint with the FCC if you believe it is necessary.

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

If trouble is experienced with the Comrex ACCESS Rack, please contact Comrex Corporation at 978-784-1776 for repair or warranty information. If the equipment is causing harm to the telephone network, the telephone company may request that you disconnect the equipment until the problem is solved.

No user serviceable parts are contained in this product. If damage or malfunction occurs, contact Comrex Corporation for instructions on its repair or return.

Connection to party line service is subject to state tariffs. Contact the state public utility commission, public service commission or corporation commission for information. This equipment cannot be used on telephone company provided coin service.

If you have specially wired alarm equipment connected to the telephone line, ensure the installation of the Comrex ACCESS Rack does not disable your alarm equipment. If you have questions about what will disable alarm equipment, consult your telephone company or a qualified installer.