BRIC-Link II

Product Manual

COMREX

BRIC-LINK IP CODEC

FIGURE LISTINGS

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.

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.

You can contact Comrex by phone at 978-784-1776. 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.

I. INTRODUCTION

The Comrex BRIC-Link is a low-cost, high-performance solution for audio-to-IP conversion. Leveraging many of the core technical aspects of Comrex's successful remote broadcast BRIC-Link product line, the BRIC-Link provides for an elegant way of moving Linear or compressed audio with very low delay. BRIC-Link may be used over a range of IP links, is very simple to use, and doesn't require the expense of more full-featured codecs. While it carries an entry-level cost, BRIC-Link maintains superb audio specifications and hardware reliability, making the system suitable for STLs and other mission-critical functions.

BRIC-Link is contained in a small desktop package. Two BRIC-Links may be installed to occupy 1U of rack space.

APPLICATIONS

BRIC-Link is uniquely suited to point-to-point "nailed up" high-quality audio links over a variety of data networks, like ISM band IP radios, T1s, satellite channels, WANs, and LANs. The robustness of the BRIC technology (Broadcast Reliable Internet Codec) used in the box allows the system to perform well on the public Internet as well (using AAC compression modes).

AUDIO CODING

For users concerned about delay and coding artifacts, the BRIC-Link offers a robust stereo or mono Linear mode that does not compress audio. In addition, unique to real-time audio codecs, BRIC-Link offers FLAC lossless compression, reducing network throughput by 30-40% with absolutely transparent coding and no tandem coding concerns. For situations where more reduced bandwidth is desired, the BRIC-Link offers AAC/HE-AAC modes as standard, allowing superb audio quality at dramatically reduced data rates. For compatibility with mobile phone and web apps, BRIC-Link also implements Opus audio compression, along with VoIP standards G.722 and G.711.

TRANSMISSION MODES AND DELAY

BRIC-Link is a true codec, offering a full-duplex stereo encoder and decoder in each box. When two-way transmission is not required, the reverse channel may be disabled. The BRIC technology incorporated includes a jitter buffer manager that automatically balances delay and stability, dynamically increasing and decreasing delay based on network performance. For networks where the QoS is known, these parameters may be set so that a consistent level of jitter buffer is maintained.

End-to-end coding delay in Linear modes is less than 25mS, and FLAC modes are less than 30mS. AAC modes incorporate around 100mS total end-to-end delay, and HE-AAC modes deliver around 220mS.

In addition to coding delay, network propagation and jitter buffers will add delay to any IP link and are network dependent.

ADDITIONAL FEATURES

BRIC-Link provides for four end-to-end contact closures to be delivered along with the audio stream in each direction. Alternately, the contact closure inputs may be configured to initiate connections. An ancillary data stream is available via RS232 along with the audio stream.

The system is capable of sending up to 3 one-way encode streams (using AAC or HE-AAC) to separate decoders (requiring additional bandwidth) and multicasting on capable networks.

Finally, BRIC-Link can act as a streaming audio server. In this mode, BRIC-Link is capable of delivering many HE-AAC streams that are compatible with computer-based media players like **WinAmp** and **VLC**.

II. BRIC-LINK DIAGRAMS AND INSTALLATION

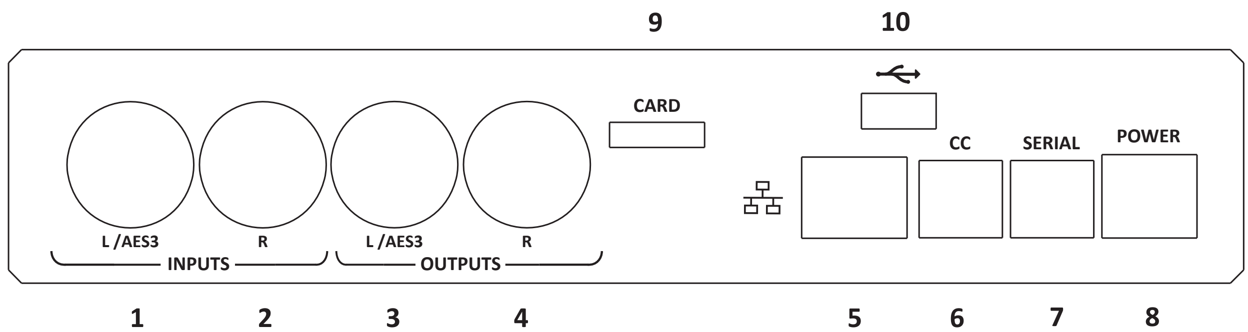

FIGURE 1 **REAR PANEL DIAGRAM AND DESCRIPTIONS**

1 **Left Audio/AES3 Input** - Accepts professional level, balanced analog audio, or if configured, AES3 stereo digital audio for input

2 **Right Audio Input** - Accepts professional level, balanced analog audio

3 **Left Audio/AES3 Output** - Delivers professional level, balanced analog audio, or if configured, AES3 stereo digital audio for output

4 **Right Audio Output** - Delivers professional level, balanced analog audio

5 **Ethernet 1000baseT** - Connection for network connections

6 **Contact Closures** - Provides for 4 contact closure triggers, and 4 open-drain style contact closure outputs

7 **RS232** - Provides serial data I/O across the IP link. Data rate is configurable

8 **Power** - 4 Pin connector for attachment of Comrex approved DC power adapter. Requires 24V DC @ 1A

9 **Card Slot** - For future use

10 **USB** - For future use

FIGURE 2 **FRONT PANEL DIAGRAM AND DESCRIPTIONS**

11 **Headphone Jack** - 3.5mm TRS output to monitor input audio or decoded audio (selectable via dip switch)

12 **Headphone Level Control** - Allows adjustment (Up/Down) of headphone output

13 **DIP Switches -** 8 individual DIP switches allow selection of certain operational parameters (see **DIP Switch Settings** on page 14)

14 **Left/Mono I/O Level Indicator -** Tri-color LED shows left channel input or output level

15 **Right I/O Level Indicator -** Tri-color LED shows right channel input or output level

16 **Ready/Status LED** Bi-color LED shows Ethernet carrier loss (Red) or valid connection state (Green)

17 **Reset Button -** Issues a hardware reset to the system when pressed momentarily.

PINOUTS - BALANCED AUDIO

Stereo professional level connections are available on the rear panel via XLR connectors for left and right input and output. These connectors are wired as follows:

1. Ground
2. Balanced Audio +
3. Balanced Audio -

PINOUTS - AES3

These connectors have a fixed level where a full scale signal represents +20dBu (22Vpp). A nominal input level of 0dBu (2.2Vpp) is recommended. When configured via the DIP switches, the left input and output become AES3 digital ports. The jacks connectors are wired as follows:

1. Ground
2. AES3 +
3. AES3 -

AES3 input connections can be at 32, 44.1 or 48kHz. The front panel DIP switches must be set appropriately. The output sample rate automatically locks to the input sampling rate. If AES3 input is unused, AES3 output is always 48kHz.

Because BRIC-Link can encode and/or decode in stereo and mono modes, it's important to understand how the audio inputs and outputs are handled in each mode.

INPUTS

In mono encode modes, BRIC-Link uses the left channel of the stereo input for delivery to the mono encoder.

OUTPUTS

In stereo decoder modes, left and right channels are delivered to the output connectors separately. In mono decoder modes, mono audio is delivered to both left and right output connectors.

PINOUTS - SERIAL PORT

The **Serial Port** is pinned to match serial connections on older Macintosh computers, so commercially available adapter cables should have the proper pinning.

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

PINOUTS - CONTACT CLOSURES

Contact closures are available via the 9-pin mini-DIN connector on the rear panel of the BRIC-Link. Inputs are triggered by shorting the respective input to Pin 9. Outputs consist of an open collector circuit which, when inactive, will offer a high-impedance path to Pin 9 and, when active, will offer a low impedance path to Pin 9. These outputs are capable of sinking up to 200mA at a voltage up to 12V.

Pin #	Function	Direction
1	RTS	To BRIC-Link
2	CTS	From BRIC-Link
3	TX Data	To BRIC-Link
4	Ground	
5	RX Data	From BRIC-Link
6		
7		
8	Ground	

DIP SWITCH SETTINGS

BRIC-Link has a set of eight DIP switches used for audio and indicator configuration

DIP Switch #	Function	Default (Down)
1	Analog/AES3 Input Select	Analog
2	Analog/AES3 Output Select	Analog
3	Audio Loopback*	Disabled
4	Level LEDs TX/RX Select	TX
5	Headphone out select TX/ RX	TX
6-8	Future Use	

* This function connects the send and receive audio together before it touches the encoder - the audio is digitized and converted back to analog, but not compressed or converted to a stream.

III. SETTING UP THE BRIC-LINK

ETHERNET

At a minimum, BRIC-Link needs a source of power, an audio connection, and a network connection. The external power supply delivers 24VDC at 1A.

The Ethernet connector is a standard 1000baseT. A normal patch cord, such as one used for a computer, should be connected here. In most ways, BRIC-Link will look like an ordinary computer to the network. In fact, BRIC-Link contains an embedded computer with a Linux-based operating system and a full network protocol stack. BRIC-Link 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 may have security concerns. If running over the public Internet, better performance is possible if BRIC-link has its own Internet connection. Often, it's worth the trouble to install a DSL line especially for BRIC-Link, especially if the cost is reasonable. Since there may be bandwidth, firewall, and security concerns with installing BRIC-Link on a managed LAN, it is recommended that your IT manager be consulted in these environments.

AUDIO INPUTS

Audio inputs should be applied and levels checked with Dip Switch 4 down. If the audio indicators are showing red, it indicates the level is approaching or reaching clipping stage. It is OK for audio levels to reach the yellow stage often.

SAMPLING RATES

When utilizing analog audio I/O, BRIC-Link assumes an audio sampling rate of 48kHz for all encoders. When utilizing AES3 I/O, the user has a choice of 32kHz, 44.1kHz, or 48kHz. The sampling rate changes have the following limitations:

1. Whenever utilizing the AES3 input, the AES3 output sampling rate clock will be locked to the input clock. This means it’s not possible to use a 32kHz input clock and a 48kHz output clock. If the output is switched to analog, the clock rate of the D/A converter remains locked to the input AES3 signal.
2. In AAC and HE-AAC modes, at AES3 rates other than 48kHz, the digital audio is sample-rate-converted to a stream based on 48kHz on the network.
3. In FLAC and Linear modes, the input AES3 sampling rate is the same sampling rate used by the network stream. If a different input sampling rate is used on each end of the link, the decoders on each side will sample-rate-convert the stream to the same rate as the local input audio.

IV. USING THE BRIC-LINK DEVICE MANAGER

Initial IP configuration is handled using the BRIC-Link **Device Manager** software, which is a Windows and MAC executable program. This program was provided on the disc with the BRIC-Link hardware, and can also be downloaded from the Comrex website.

In order to configure BRIC-Link, the **Device Manager** must be run on a computer located on the same physical LAN connection as the BRIC-Link hardware. If this is not possible, you may need to connect an Ethernet crossover cable between the BRIC-Link and the computer for configuration.

Once power is applied to BRIC-Link, you have five minutes to configure the IP settings. After five minutes, the power must be cycled on the hardware to make these changes.

As shipped from the factory, BRIC-Link is configured for **DHCP**, which means it will attempt to obtain an IP address from your network. Using the **Device Manager** software, you can change this to a **Static IP** with fixed Netmask, Gateway, and DNS Server settings.

As shown in **Figure 3**, running the **Device Manager** and clicking the **Scan for Devices** button will produce a list of all Comrex IP codecs found on the LAN.

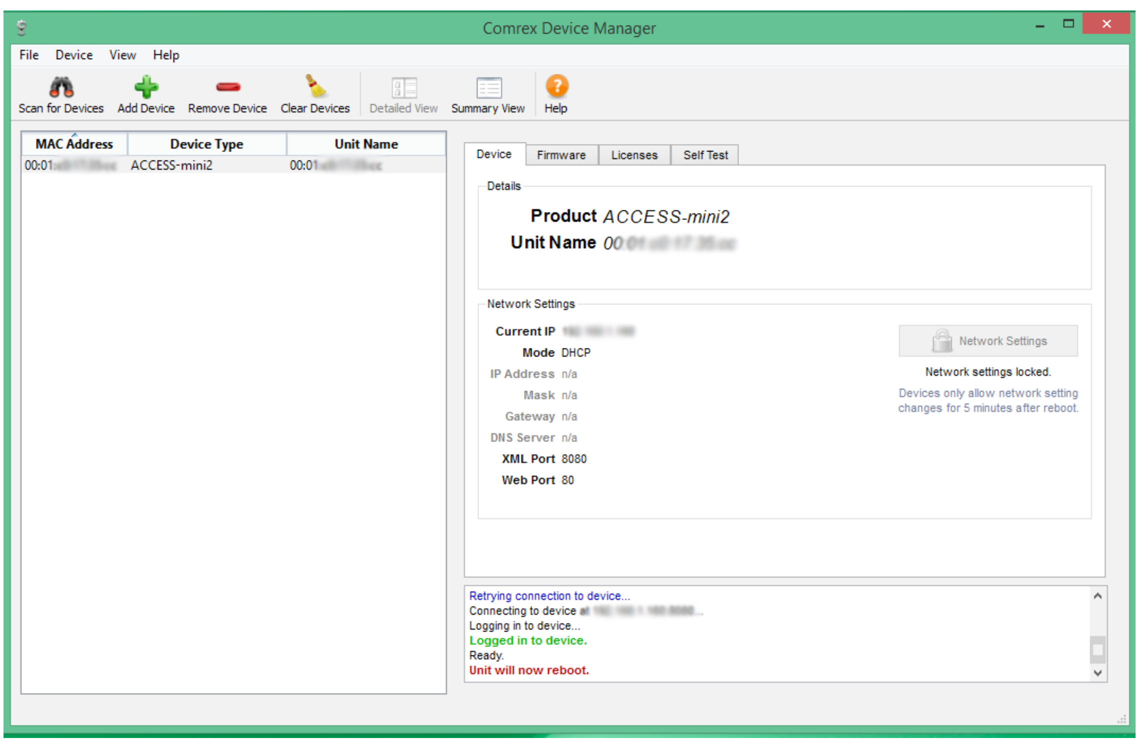

FIGURE 3 **DEVICE MANAGER**

Choosing the codec that appears in the left hand list, followed by pressing the **Network Settings** button, allows you to set the IP parameters of the codec, as shown in **Figure 4**.

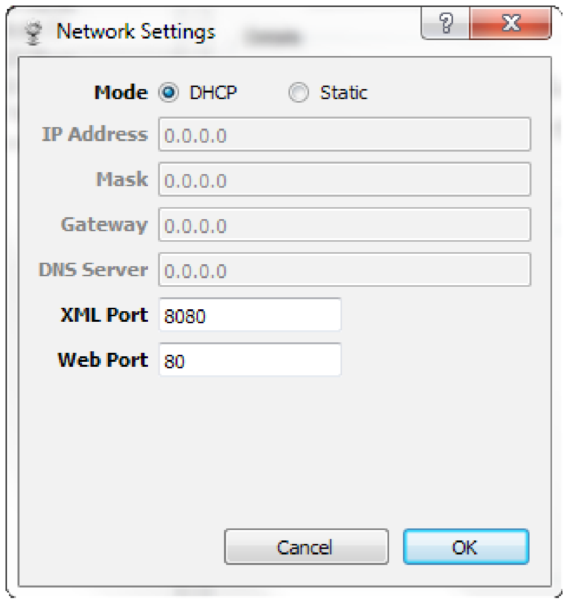

FIGURE 4 **IP CONFIGURATION**

Once you know the IP address (or have changed it) using the **Device Manager**, the rest of the setup and operation of BRIC-Link is done via the built-in **Web-based Interface.**

V. CONTROLLING THE BRIC-LINK VIA THE WEB-BASED INTERFACE

Once your IP settings are configured and BRIC-Link has cleanly booted on your LAN, it's time to take a look at the BRIC-Link **Web-based Interface**. This is done by pointing a web browser on your LAN to the BRIC-Link 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 Firefo**x 1.0 or higher with **Adobe Flash Plug-in 7** or higher. **Opera 8.5** works well also. If you experience trouble connecting to the BRIC-Link, be sure you have the latest Flash Plug-in installed by right-clicking your mouse in the main browser window and selecting "about **Adobe Flash**." This will take you to the **Adobe** website where you can download the latest free plug-in.

LOGIN

Once you are connected to BRIC-Link, a login screen will appear (see **Figure 5**). Key in any username 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.

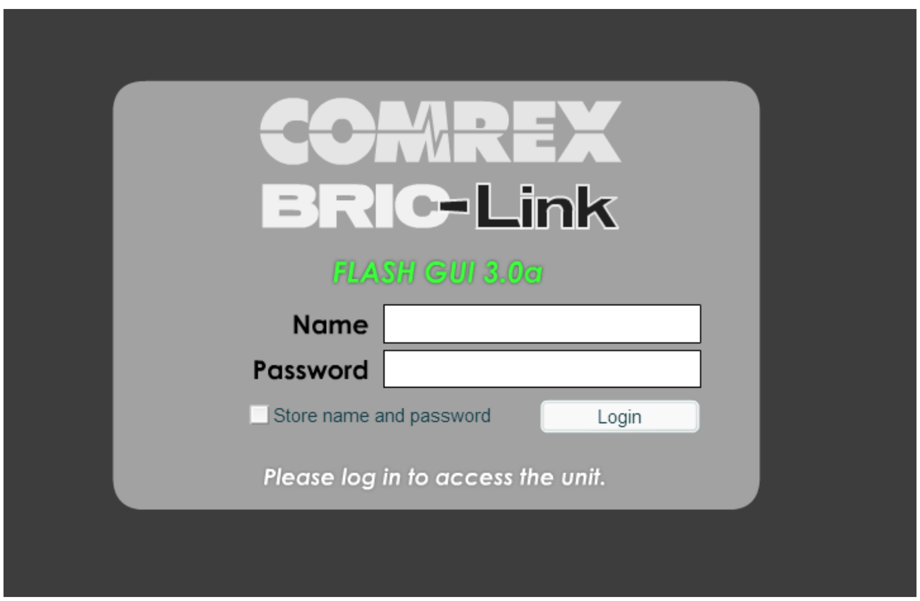

FIGURE 5 **WEB LOGIN**

There are three main parts to the BRIC-Link Web-based Interface screen:

1 **Main Audio Meter** - The level meters are defaulted to off to conserve bandwidth and client CPU, but when these are enabled this top bar gives an indication of audio levels.

2 **Tabs** - Use these tabs to control and obtain status of BRIC-Link. 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 BRIC-Link web interface. In addition, when BRIC-Link is connected to a remote user, chat text will appear from any users logged into the remote web interface.

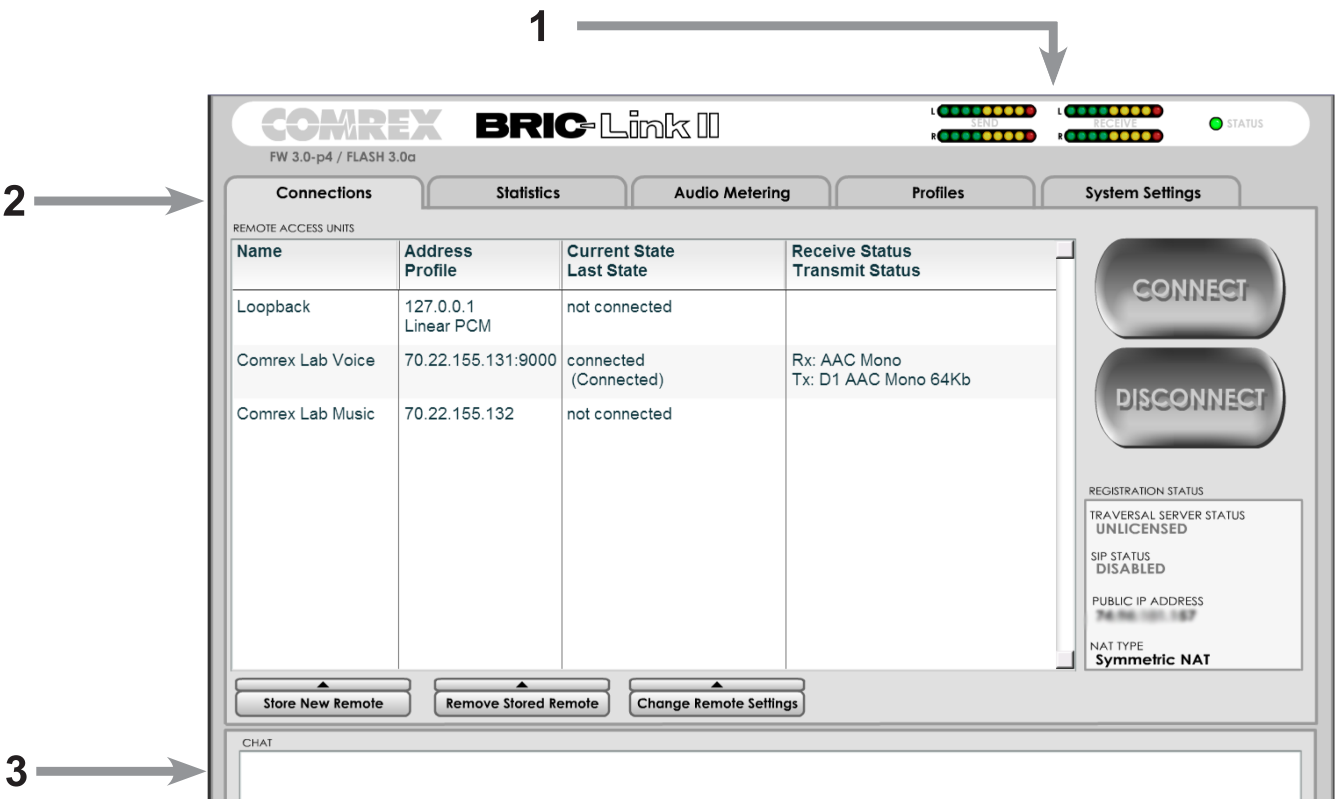

FIGURE 6 **CONNECTIONS TAB**

CONNECTIONS TAB

The **Connections** tab is the default setting for the Web-based Interface as shown in **Figure 6**. 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 BRIC-Link to the list, simply click **Store New Remote** 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 will also need to choose a profile to use when connections to that remote are initiated. To get started, simply choose one of the default profiles provided (we'll show you how to build your own later). You may remove any stored value simply by highlighting and clicking **Remove Stored Remote**. Stored remote addresses are saved to system memory, where they will remain through power cycles.

The **Connections** tab will also display **IP** and **Status** information of a remote BRIC-Link 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** - This user provides a talk feed from the Comrex headquarters in Massachusetts, USA for testing your connection.
3. **Comrex Lab Music** - This user provides a music feed from the Comrex headquarters in Massachusetts, USA for testing your connection.

STATISTICS TAB

The **Channel Statistics** field (**1** in **Figure 9**) delivers information on the total number of bits entering or leaving the BRIC-Link (including multiple connections if applicable), IP , UDP and RTP packet headers and coded audio.

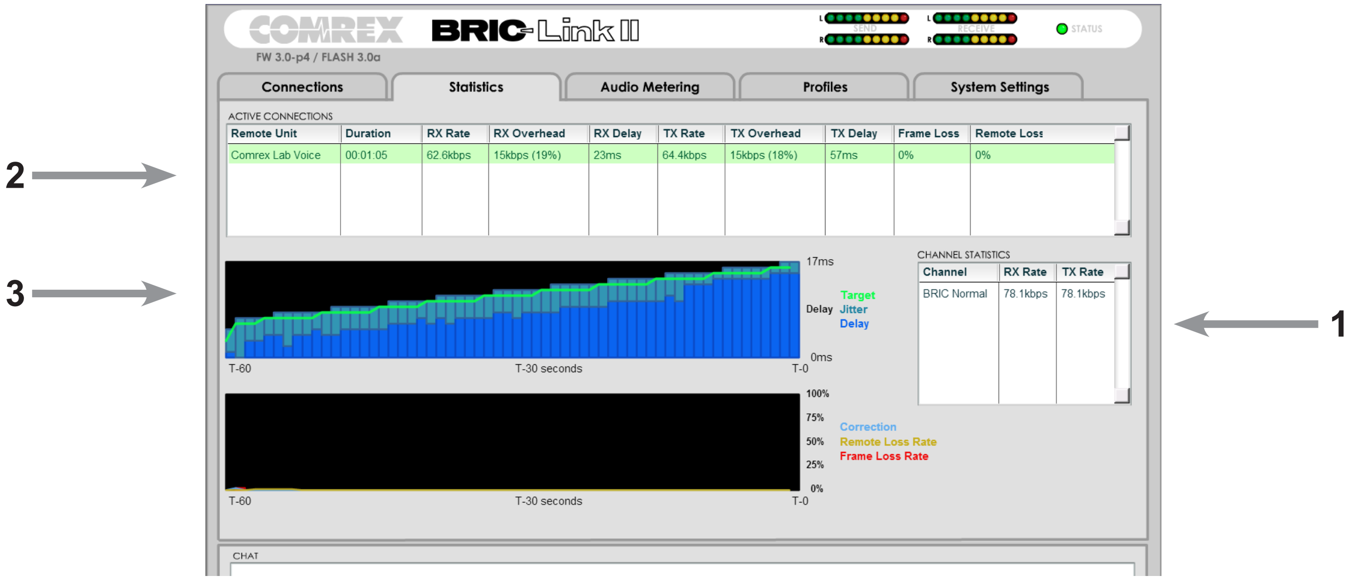

FIGURE 7 **STATISTICS TAB**

The **Active Connections** box (**2** in **Figure 7**) breaks this information down further. Because BRIC-Link 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. **Frame Loss** is also listed as an individual figure for lost and late packets.

Finally, a **Max 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.

Graphical representations of **Jitter Buffer Manager** activity and Frame Loss are also displayed (**3** in **Figure 7**). The light blue area in the upper graph represents the jitter values over time. The work of the **Buffer Manager** is shown by the green line, which is the target buffer delay that the system is trying to achieve, based on measurements done over the jitter window.

The lower graph displays a real time and historical representation of frame loss. If the decoder does not receive packets in time, the chart will show a red line indicating percentage of lost packets over the one second interval.

AUDIO METERING TAB

The **Audio Metering** tab, as shown in **Figure 8**, 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 BRIC-Link 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 BRIC-Link is connected to a data constrained network (e.g. wireless), it is strongly recommended that these meters be **Off**, especially if the **Web-based 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.

Note: Better networks can support higher quality settings

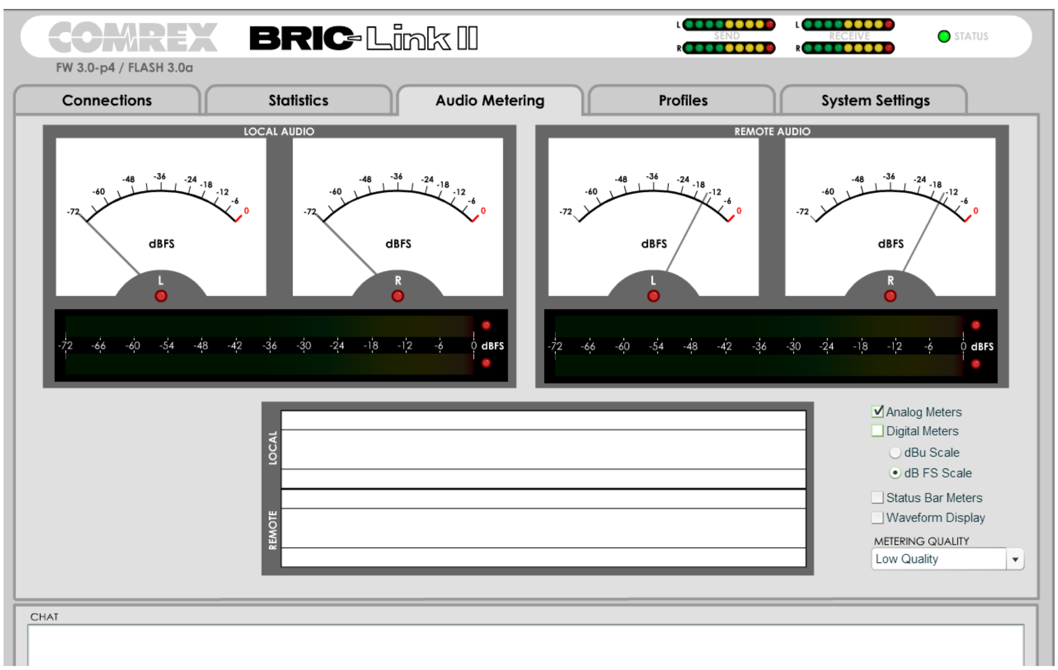

FIGURE 8 **AUDIO METERING TAB**

PROFILES TAB

BRIC-Link provides a powerful set of controls to determine how it connects. The **Profiles** tab allows you to define one or more profiles to assign to outgoing remote connections. It's often not necessary to define any profiles since BRIC-Link ships with a set of default profiles that cover most users, but this tab allows you to build custom profiles to allow for different encoders in each direction and special options for jitter buffer management. Keep in mind that these profiles are useful only for connections initiated from the local BRIC-Link. Incoming connections are defined by the BRIC-Link at the other end.

Profile creation is segmented into commonly used and advanced options. In order to simplify the interface, **Advanced Options** are normally hidden from the user.

Remember, building a profile doesn't change how any remotes are connected until that profile is assigned to a remote on the **Connections** tab. Once a profile is defined, it will be available on the **Connections** tab to be assigned to any defined connection.

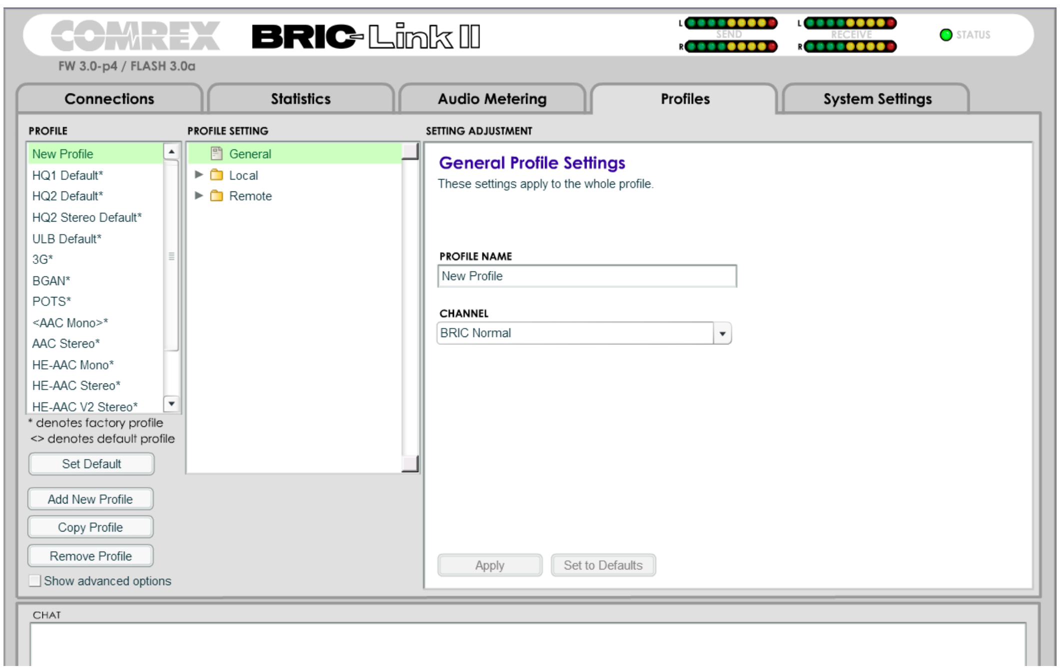

FIGURE 9 **PROFILES TAB**

BUILDING A PROFILE

To build a new profile, select **Add New Profile** (**1** in **Figure 10**) and a new profile appears on the list labeled **New Profile**. Select it and you'll see the first set of options available in the **General Profile Settings** category (**2** in **Figure 10**). Here you can rename the profile to something that will help you remember it. Under the Channel category (**3** in **Figure 10**), you can select whether this is a UDP IP connection (BRIC normal) or one of the other connection modes offered by BRIC-Link. These other choices are defined in the Advanced Topics section. Note: It's important to define the channel of a profile before moving on to other options, since the choices in the subsequent sections will vary in this choice. Make sure to press **Apply** in order to confirm your selection.

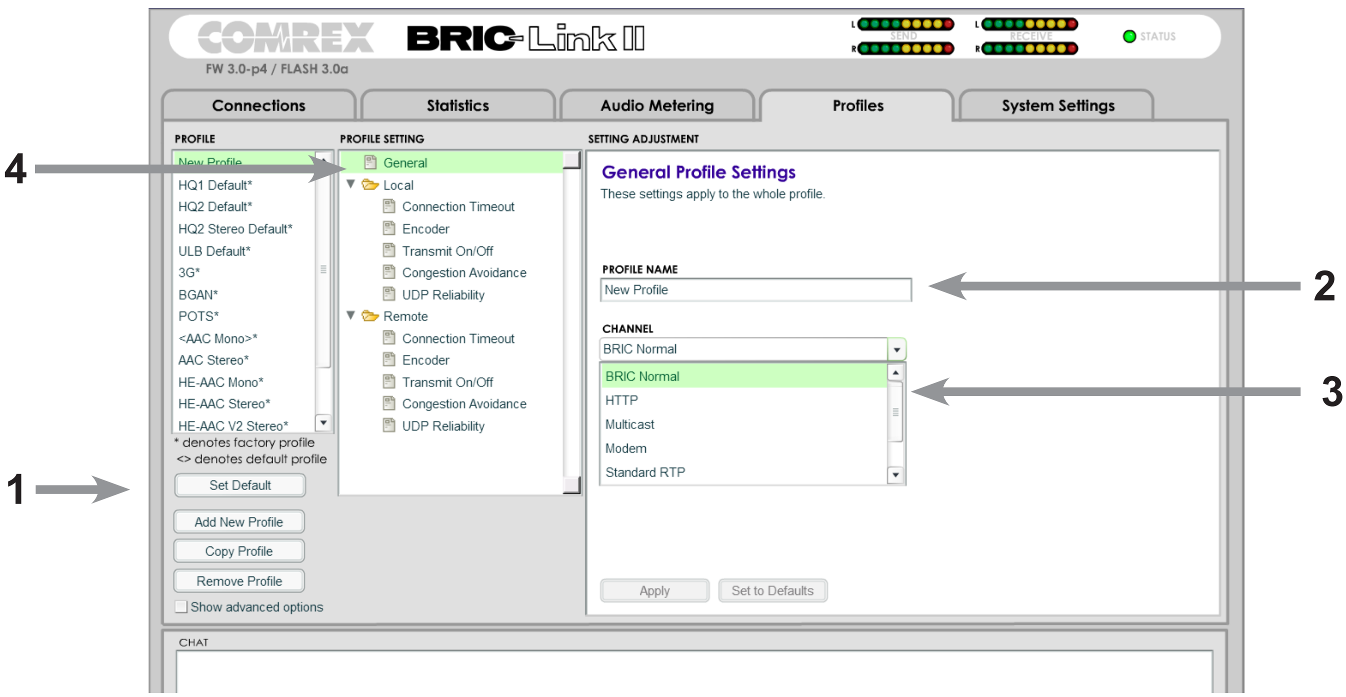

FIGURE 10 **GENERAL PROFILE SETTINGS**

LOCAL & REMOTE SETTINGS

You'll be presented with two categories of options: **Local** and **Remote** (**4** in **Figure 10**). You'll use the **Local** section to determine how your BRIC-Link behaves, and the **Remote** section will determine how the BRIC-Link on the far end behaves. Each category lists identical options, so we'll cover only the **Local Settings:**

Connection Timeout - Under normal circumstances, a connection will be terminated on one end and the other end will drop the connection in turn. However, if a network failure occurs or a connection is ended abruptly (e.g. killing power to a BRIC-Link), the system will drop the connection after a predetermined time. The default is 60 seconds, but this can be shortened or lengthened here. If an indefinite connection is required, see **Section VIII Operating BRIC-Link in a 24/7 Environment** for additional information.

Encoder - It's not necessary to define any decoder types when using BRIC-Link because they automatically adapt to the incoming stream. Using this menu, you can select the encoder used to send audio from this BRIC-Link (local) as well as the encoder used to return audio to this BRIC-Link (remote). The default value of the remote encoder is to follow the local encoder (i.e., it will send exactly the same codec mode it receives). This is defined as **Follow Mode** in the remote encoder selection table. See **About the Algorithms** section for more info on selecting encoders.

Transmit On/Off –-This option determines whether the selected encoder (local or remote) is actually sending any data. By default, all encoders are turned on, but there may be circumstances where one-way operation is desired (e.g. multi-streaming, as described in **Section IX**). Turning off the local encoder disables outgoing audio streaming, and disabling the remote encoder disables incoming audio streaming.

BRUTE RELIABILITY SETTINGS

Two options are available to help transmissions that are suffering from poor network performance. There are encoder treatment options, which are applied to the "local" encoder, the "remote" encoder, or both. BRUTE options require 2.7 or higher software on both ends of the link.

Congestion Avoidance - Enabling this option allows the encoder to dynamically change the number of frames per packet sent, thereby reducing total data requirements. In addition, in most encode modes, enabling congestion avoidance provides the system a license to step down to a lower encode data rate if desired. This will happen automatically and with no audio interruption. Step down congestion avoidance is not enabled in the Linear PCM mode.

UDP Reliability – UDP, the Internet protocol used by BRIC Normal connections, does not have any inherent error correction capability. UDP reliability adds an intelligent algorithm that requests packet resends only when appropriate. UDP reliability can be useful on some wireless connections that have unsatisfactory performance due to packet loss.

ADVANCED PROFILE OPTIONS

The options available in the default mode should provide good performance for most users, but in some circumstances it may become important to fine tune some of the more obscure parameters that make BRIC-Link work. By clicking the **Advanced Options** box in the lower left of the **Profile Settings** screen, the following **Advanced Options** will be available:

ADVANCED CHANNEL

In addition to BRIC Normal, BRIC-Link provides the ability to set up several other channel types. The Advanced menu gives the option to use a different channel rather than the normal UDP/RTP created in BRIC Normal mode. Some explanation:

Internet IP packets come in two flavors: TCP and UDP. Most web browsing, email and other computer-based functions travel over the TCP protocol, which inherently assures retransmission if a packet is lost, and is therefore reliable. UDP is optimized for real-time applications, and does not offer any guarantee of packet delivery. Retransmission typically causes extra delay in an IP network, and BRIC-Link is optimized to conceal an occasional lost packet, so it makes more sense for BRIC-Link to use UDP for transmission under most circumstances. But there are occasions where a network will treat UDP packets poorly. Some examples are:

- Networks with high packet loss (rather than jitter)
- Networks with very high security firewalls
- Networks trying to discourage the use of VoIP functions

In these circumstances it makes more sense to enable a TCP channel. The result will usually be a more robust audio channel with a delay several magnitudes higher than an equivalent UDP channel. Channel overhead is also raised so you will utilize a higher network bandwidth.

In addition to TCP, there are several other advanced channel modes:

HTTP - BRIC-Link has the ability to act as a streaming server, delivering AAC and HE-AAC to compatible PC based media players. Normally in this mode, connections are requested on an incoming basis so no outgoing profile setup is required. But BRIC-Link also has the ability to initiate a stream to a Shoutcast-compatible server in order to distribute the stream to users. Only in this instance should a profile be set for HTTP.

Multicast - Should only be used to initiate IP Multicast connections (not for use on the Internet). See Section 13 for more on Multicast connections.

Standard RTP - This setting is used in the unusual scenario where the network is viable in only one direction. Standard RTP has the ability to send and receive streams without any status information being relayed between the codecs.

ADVANCED CHANNEL OPTIONS

When designating **Local** and **Remote** options for a normal BRIC or TCP channel, several new categories will appear. Some of them address the encoder and some address the decoder.

Most of the **Advanced Encoder** options alter the relationship between frames and packets. In this context, a frame is the smallest chunk of encoded audio that can be extracted from the encoder. For the lowest possible delay, this frame is wrapped into its own packet and sent into the network.

ADVANCED ENCODER OPTIONS

The following advanced options affect the encoder:

Frames per Packet - This function allows the encoder to wait for “**X**” number of frames to exist before sending a packet. This option differs from FEC because each frame is only sent once. Setting this value to a number higher than one can reduce network usage, at the expense of delay. This is because packet overhead bits like IP and UDP headers are sent less often.

Log Statistics - This function is used in factory diagnostics and should be left disabled unless instructed by Comrex support.

UDP Reliability Max Retransmissions – This parameter allows you to set an upper limit on how much additional bandwidth is utilized by the BRUTE UDP reliability layer. The default setting is 100, which allows the error correction layer to use the same amount of bandwidth as the audio stream. As an example, if your audio stream is consuming 80 kb/s of network bandwidth, and UDP Max Retransmissions is set at 50%, up to 40kb/s additional network bandwidth may be used for error correction.

Nagle Algorithm – Nagle is applicable to TCP transmission only. When Nagle is enabled, encoder packets are sometimes buffered and concatenated into larger packets, depending on the network. It can be used to lower overhead on TCP networks, but adding delay.

ADVANCED DECODER OPTIONS

Advanced Decoder options have to do with how the jitter buffer manager performs. This is the algorithm that determines, based on network performance, how much delay to install in front of the decoder to achieve uninterrupted audio. It does this by creating a statistical analysis of the amount of jitter experienced over a fixed interval of time (the window) and making a judgment based on other parameters like the decoder's resiliency to errors. This is actually a very complex decision-making process involving many variables, and most of the time the default parameters should work well. The **Advanced Decoder** options are a means to override these defaults, and changing them should be done with care.

The following advanced options affect the **Decoder**:

Retransmit Squelch - These options are used to determine how the buffer manager reacts to typical data dropouts like those seen on wireless networks. Some explanation:

Many wireless networks have their own layer of data protection riding on top of any other data layer, providing packet retransmissions in the event of signal fade. The symptom from the network standpoint is that data will come to a stop for some period of time while the signal is faded, and the network will buffer all packets during this time. Once the wireless link is restored, all the buffered packets will appear to the decoder as if they were simply very late. In essence, the protection layer will "fight" the buffer manager. The effect will be that the buffer manager will expand the buffer, increasing delay dramatically without any benefit.

The **Retransmit Squelch** allows the decoder to detect these events and avoid having the buffer manager react. The squelch has several user adjustable parameters with good default settings. These should normally be left where they are, but there may be unusual circumstances where they should be changed.

Retransmit Squelch Trigger - Determines the amount of time the decoder must experience 100% packet loss before the **Retransmit Squelch** function is triggered. Default is one second.

Retransmit Squelch Max - The longest period of data loss during which the squelch function is active (default is two seconds). During the squelch period, the buffer manager ignored the relative jitter experienced and does not adjust buffer size to compensate.

Jitter Window - This parameter defines the amount of time (in minutes) that historical network performance is analyzed in order to make the rest of the calculations. As an example, if the **Jitter Window** is set to the default of five minutes, and if a dramatic network event happens and the buffer manager reacts (perhaps by increasing the buffer), the event will be included in the manager's calculations for the next five minutes. If the network experiences improved performance over this period, the manager may choose to wind the buffer back down after the five minutes has passed.

Loss Cushion - Packets may arrive at the decoder displaying a range of statistical properties. They may arrive in reasonably good timing and in order, or half may arrive quickly with the other half delayed significantly. In some cases, most of the packets arrive in a timely manner, but a small percentage of them may be extremely late. In this case, it's usually preferable to allow these late packets to be left out, and keep the delay lower. The decoder error concealment does a very good job of hiding these losses. The **Loss Cushion** parameter instructs the buffer manager to ignore a certain percentage of late packets in its calculation. The default value is 5%. Applications that are not at all delay sensitive may wish to reduce this value to zero, while extremely delay sensitive applications may prefer to have this closer to 25%.

Delay Cushion - The jitter buffer manager usually works very hard to keep absolute delay to a minimum. Some applications are not delay sensitive and would rather not have the manager working that hard. The **Delay Cushion** setting is a way to instruct the manager not to attempt to drive the delay below a certain value. E.g., if the delay cushion is set to 500mS, this amount of fixed delay will be added to the buffer. If the jitter manager needs to increase the buffer it will do so, but will not fall below the ½ second level.

Delay Limit - The inverse of the **Delay Cushion**, this parameter instructs the manager not to wind the buffer out beyond a certain delay value, regardless of how many packets are lost. This is useful in applications where staying below a certain delay figure is essential, but use of the delay limit can result in very poor performance if the network jitter dramatically exceeds the limit.

Fixed Delay - This option simply sets the **Delay Cushion** and **Delay Limit** at a similar value, so that the delay buffer is defined to the chosen value and will not increase or decrease significantly.
Buffer Management On/Off – This option is available only as a troubleshooting tool. Turning the manager off will result in eventual failure, since the manager is required to compensate for clock skew between the encoder and decoder.

SYSTEM SETTINGS TAB

The **System Settings** tab defines parameters that are not specific to a particular remote connection. Examples are how incoming calls are handled, codec name and how the contact closures are assigned. The **System Settings** tab is shown in **Figure 11.**

The **Systems Settings** tab has several categories: **System Settings**, **Aux Serial Settings**, **Security Settings**, **BRIC Normal Settings**, and **N/ACIP SIP Settings**. As with the **Profiles** tab, basic options are shown by default. Less used options are hidden until the **Show Advanced Options** box is clicked.

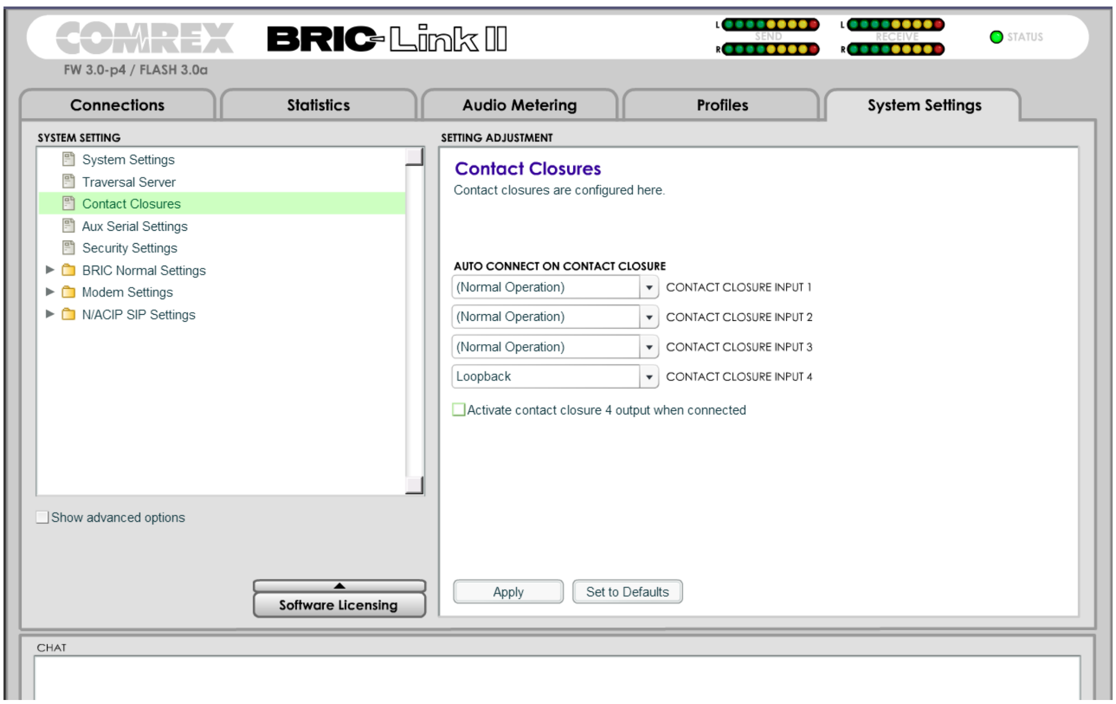

FIGURE 11 **SYSTEM SETTINGS TAB**

Unit Name – Users are encouraged to name their codecs here. The default name of a codec is the unique MAC address of the Ethernet port. By changing this to something familiar and unique (e.g. roving reporter, weather guy, etc.) this name is reflected in several places:

1. In the browser used to show the remote control page
2. In Comrex provided utility software such as **Remote Control** and **Device Manager**
3. In Switchboard TS Buddy lists

Always Connect to Remote - This field is available to designate a remote for always on operation. This is useful in “nailed up” environments, where a signal is required across the link 24 hours a day. To assign an always on remote, simply pull down the menu and select which remote to designate as **Always On**. A connection will be made and sustained to the chosen remote. Remote connections must be created in the **Connections** tab before they can be assigned to this function.

The next four fields define auto connect rules for remotes to be triggered by the four external triggers available on the rear panel of the BRIC-Link. Note: These inputs are shared with the end-to-end contact closure signals, so if a remote is designated as Auto Connect on a closure, that closure signal is sacrificed in the direction from this BRIC-Link.

To assign a remote connection to a contact closure, simply pull down the menu box next to the desired closure and select the proper remote. A connection attempt will be made whenever the contact is triggered, and will disconnect whenever the contact is released.

CC Connect Status - The last **System Setting** (**Contact Closure Input 4** - see **Figure 11**) alters the performance of output contact closure #4. Under normal circumstances the signal indicates a trigger of the corresponding contact closure input on the far end of the connection. If this box is selected, that function is no longer available, and the signal follows the BRIC-Link front panel **Ready** light. This signal will be valid (closed) when a valid connection is present, and invalid (open) when no connection is present.

AUX SERIAL SETTINGS

This allows you to set the parameters of the auxiliary serial data port provided on the BRIC-Link. This port is always active during an IP connection and allows serial data transfer along the same path used for the audio data. It does not remove any audio data; the serial data is added to the packets and bandwidth is increased to support the additional data. For this reason, heavy use of serial data can affect overall codec performance. Settings are available for **BAUD RATE, DATA BITS, PARITY, STOP BITS** and **FLOW CONTROL**. Most users will leave the defaults of **9600, 8, 1, None for Flow Control** and **None** for **Parity**.

SECURITY SETTINGS

Connection Password – This allows you to define a password that must be attached to all incoming connections before they are accepted. Units placing outgoing connections to you must know this password and apply it to their outgoing stream. Leaving the field blank will disable this function.

GUI Password - This allows you to define a password for the web page screen and firmware updater. The default password is **comrex** (lower case). You can disable the remote control and firmware updating functionality completely by disabling the **Remote Control** option.

Enable Remote SSH Access - This provides the ability for Comrex support to connect to this unit using the SSH protocol in order to troubleshoot. We recommend leaving this option enabled, since SSH access requires a key value that is not disclosed by Comrex, generic SSH requests are rejected.

BRIC NORMAL SETTINGS

Accept Incoming Connections - This determines if this BRIC-Link is to used for incoming normal IP connections. If this function is not enabled, BRIC-Link will only support outgoing calls using BRIC Normal Mode.

N/ACIP SIP SETTINGS

Accept Incoming Connections - This determines whether incoming calls are accepted in N/ACIP SIP format (used for compatibility with other manufacturers who follow this protocol).

ADVANCED SYSTEM SETTINGS

When the **Advanced System Settings** box is checked, a few additional options are enabled.

BRIC NORMAL SETTINGS

IP Port - This option allows you to define the incoming UDP port: the number to be used for incoming IP connections. The default is 9000. Note that since most BRIC-Link codecs attempt a connection on this port number, changing it can mean the BRIC-Link in the field must dial specifically to your new port number in order to connect. An outgoing call must be made to a specific port number in the form of **IPADDRESS:PORT** e.g. dialing port **5004** on the Comrex test line is formatted **70.22.155.131:5004**

STANDARD RTP SETTINGS

These settings offer several modes that allow compatibility with specific IP coding devices. For complete details, please review the **IP Compatibility** appendix of this manual.

N/ACIP SIP SETTINGS

IP Port – The port used by the SIP negotiation channel when using N/ACIP SIP Mode. If this port is changed, it's likely to break compatibility with other manufacturer's codecs

RTP IP Port – The port used for audio transfer during N/ACIP SIP mode. Since this port info is transferred during the negotiation process, it can be changed without breaking compatibility. Note that RTSP data is always sent and received on the port one address higher than this.

Public IP Override – Enable this in an environment where ports have been forwarded through a router to the BRIC-Link, and a N/ACIP SIP connection is desired. The SIP protocol is assuming no ports are forwarded and may have trouble connecting without this function enabled.

TCP SETTINGS

BRIC-Link performs best when using UDP for connections but there are some rare circumstances when the system may need to be switched over to TCP operation. This advanced option defines how incoming TCP calls are handled.

Outgoing calls are defined as TCP when their profile is configured. BRIC-Link normally listens for incoming calls on both TCP and UDP ports, and chooses the first to arrive. If a TCP call is detected, BRIC-Link will attempt to use the same TCP link to transmit in the reverse direction.

Accept Incoming Connections - This allows you to turn TCP Auto Answer on and off. Disabling this function means only outgoing TCP calls can be established.

IP Port - You have the option of setting the incoming TCP port number, which can be different than the UDP port number.

Note: Warnings given above about changing port numbers — calls with mismatched port numbers will fail

VI. MAKING CONNECTIONS ON THE BRIC-LINK

CREATING A REMOTE CONNECTION

So now it's time to make a connection on BRIC-Link. We will assume that the network and audio connections have been made. Before you can establish an outgoing connection on BRIC-Link, you must enter the info about remote connection into the **Connections** tab. This acts like a phone book, saving the names and numbers of everyone to whom you connect.

As shown in **Figure 12**, BRIC-Link comes pre-programmed with three connections. Loopback is chosen when you wish to test BRIC-Link by connecting the local encoder and decoder together. The other two entries are connections to Comrex in Massachusetts, and these may be used for your testing (when they're not busy with other users). We maintain two CD players on these codecs, feeding voice and music audio respectively.

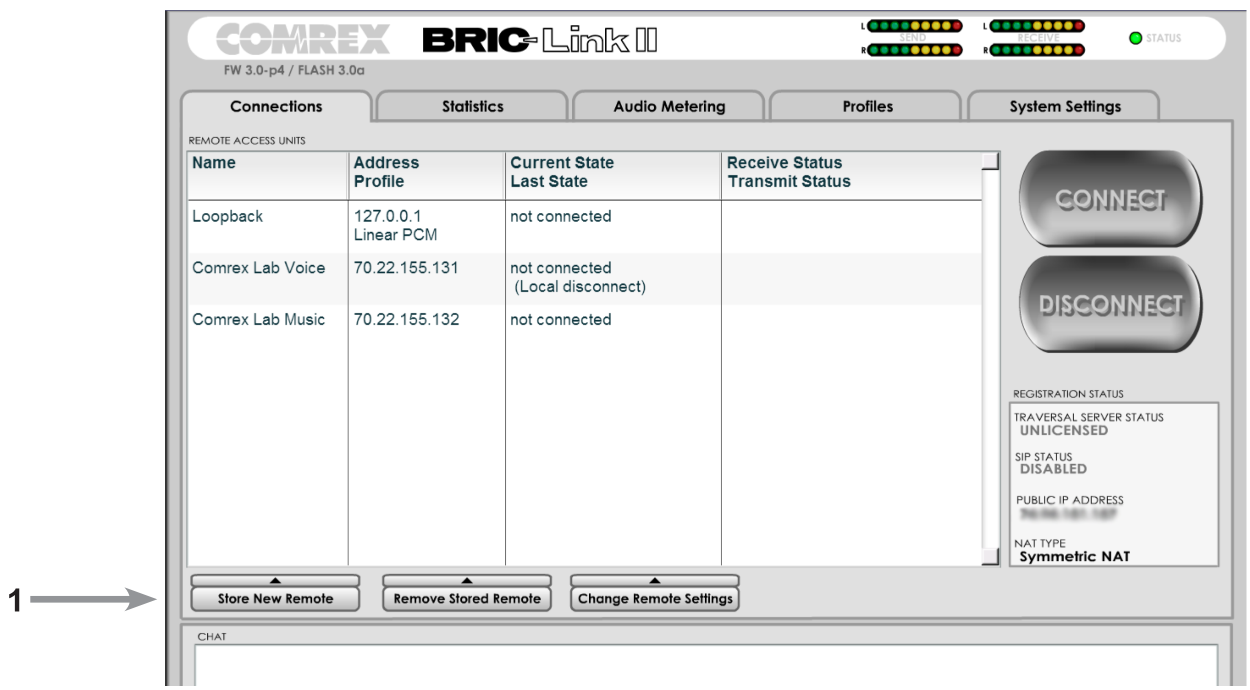

FIGURE 12 **CONNECTIONS TAB**

To create your own outgoing connection, click **Store New Remote** (**1** in **Figure 12**) to get the entry pop-up. Choose a name for the remote, (e.g.. **WXYZ**), followed by the IP address or phone number of the remote. The next field is optional. If the remote has password filtering enabled for incoming calls, you will need to enter that password into the next field (case sensitive) in order to make a connection to it. If no password is required, leave this blank.

Finally, you will need to choose a profile to use when making these connections. BRIC-Link includes several common default profiles to choose from, each of which enable a simple full-duplex link using one of the available algorithms. If you wish for a more complex feature set when making this connection, you will need to click over to the **Profiles** tab and set up a specific profile using your custom parameters. Custom options can include one-way transmission, different encoders in each direction, specialized packet arrangement, etc. Once defined on the **Profiles** tab, the new profiles will be available in the **Profile Select** window and they can be assigned to a remote connection.

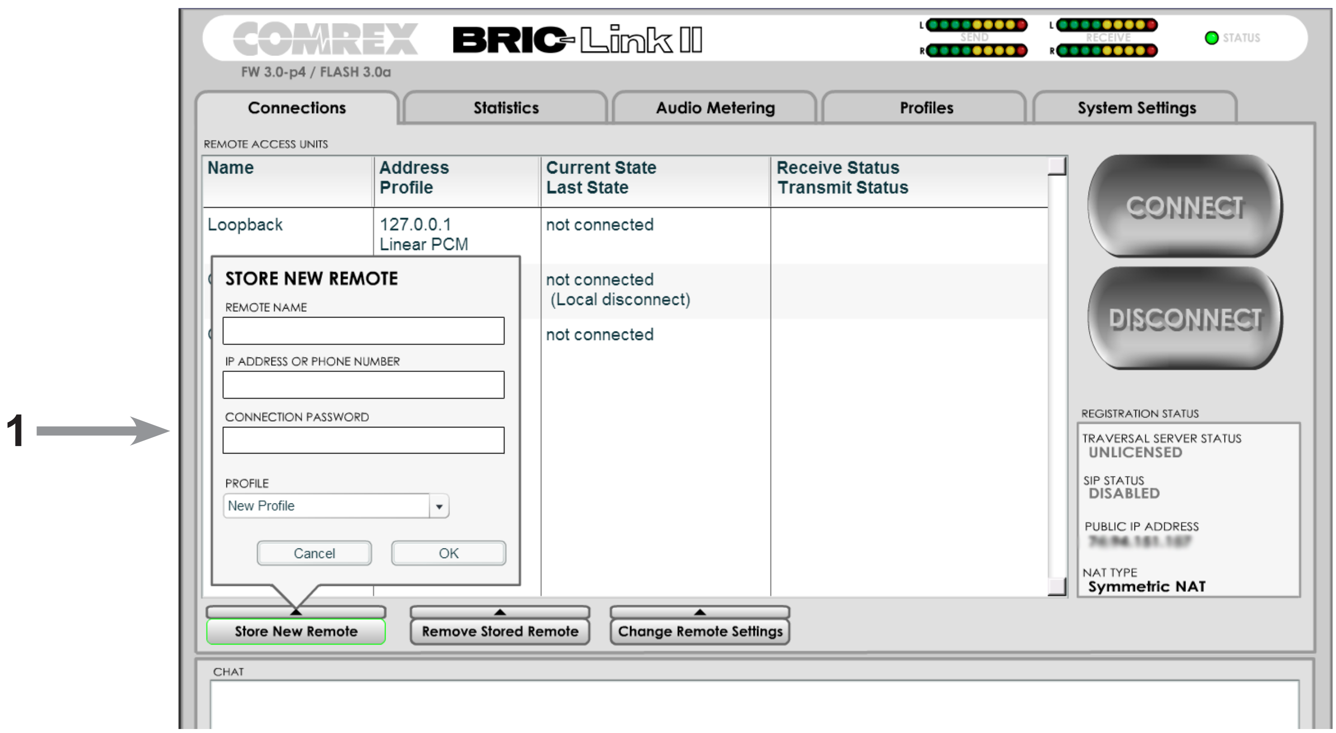

FIGURE 13 **STORE NEW REMOTE**

CONNECTING

Once your remote connection entry is correct, it's simply a matter of pointing and clicking to connect and disconnect a remote. When a connection is attempted, the **Current State** value in the connection table will change to reflect the progress of the connection. If the connection fails, the reason for failure will be shown in the **Last State** category. If it succeeds, the encoder and decoder mode will be reflected in the **Transmit Status** and **Receive Status** columns.

DISCONNECTING

Disconnecting is just as simple. Highlight the desired connection and click **DISCONNECT** to end the connection.

PASSWORD FILTERING

The **CONNECTION PASSWORD** function can be used to filter incoming connections. Using this function, attempted incoming connections will be rejected if they do not know the proper case-sensitive password. For outgoing connections, the password is entered when the remote connection is created on the **Store New Remote** menu. For incoming connections, the password is set on the **System Settings** tab.

There is no way to retrieve a forgotten password; it must simply be changed in each BRIC-Link.

VII. OPERATING BRIC-LINK IN A 24/7 ENVIRONMENT

BRIC-Link can be easily set up for "always on" operation. It will be helpful to describe a little bit about the BRIC-Link data transfer protocol before describing how to set the system up.

In BRIC Normal mode, the default mode of operation, BRIC-Link transfers all its audio data via the UDP protocol. This is in contrast to most web-based connections like browsing and email, which use the TCP protocol. UDP, unlike TCP, is not "connection oriented", (i.e., no virtual connection actually exists in this protocol layer between the devices). In UDP, the transmitter simply launches packets into the network with the correct address, hoping the network will make its best effort to deliver the packets in a timely fashion. If a packet is delayed or lost, no error message is sent and no packets are retransmitted. It is up to the receiver to cover up any lost data, if it can. This allows the Internet to deliver packets with the smallest amount of overhead and delay. Since there is no intelligent connection built between the codecs, there isn't actually any connection to break in the event of network failure. The encoder simply launches packets into the network, regardless of whether they arrive or not. If the network fails and is later restored, the packets stream will be restored to the decoder.

For most applications like remote broadcasting, it's useful to simulate a connection-oriented stream, so BRIC-Link uses a low-bandwidth sub channel to deliver information back to the encoder about overall connection status. It does this in its "application layer", rather than the "transport layer" where UDP exists. By default, it monitors the health of a connection and if no data is detected as received by the decoder for 60 seconds (this is a user adjustable timeout), it "tears down" this connection and goes back to idle state. This can give an indication to the user that the network has failed and it's time to look at the problem.

The good thing about having the connection protocol in the application layer is that its use is optional. For 24/7 operation, there's no advantage to having the connection end if no data is received for a timeout interval.

So to set BRIC-Link for 24/7 operation, several parameters are changed:

1. The timeout value is set to infinity; the connection will never be torn down regardless of data status.
2. BRIC-Link is configured to re-establish the connection in the event of a power-up.
3. The local **Disconnect** control is disabled. The **Disconnect** function on the receiving side is still enabled, but will result in an immediate reconnection by the initiating side.

In the **System Settings** tab, the field labeled **ALWAYS CONNECT TO REMOTE** offers a pull-down menu of all available connections. Setting this value to one of your pre-defined connections results in configuring the unit for 24/7 operation to that remote. No configuration is necessary on the remote side.

BRIC-Link has another option for persistent connections. When building a remote entry a field is available for backup options, one of those options is Keep Retrying This Remote mode. In a similar fashion, using this mode will allow the unit to disregard the timeout value and keep a persistent connection. The difference is that the Disconnect function still works and the connection will not be reinitiated on a power-up. This mode is meant for users who are making temporary connections, but do not want the system to time out and disconnect in the event of network failure.

BRIC-Link has the capability of automatically making a backup IP connection in the event of failure of the main connection. This is called Fallback, and is an option chosen after defining a new Remote connection.

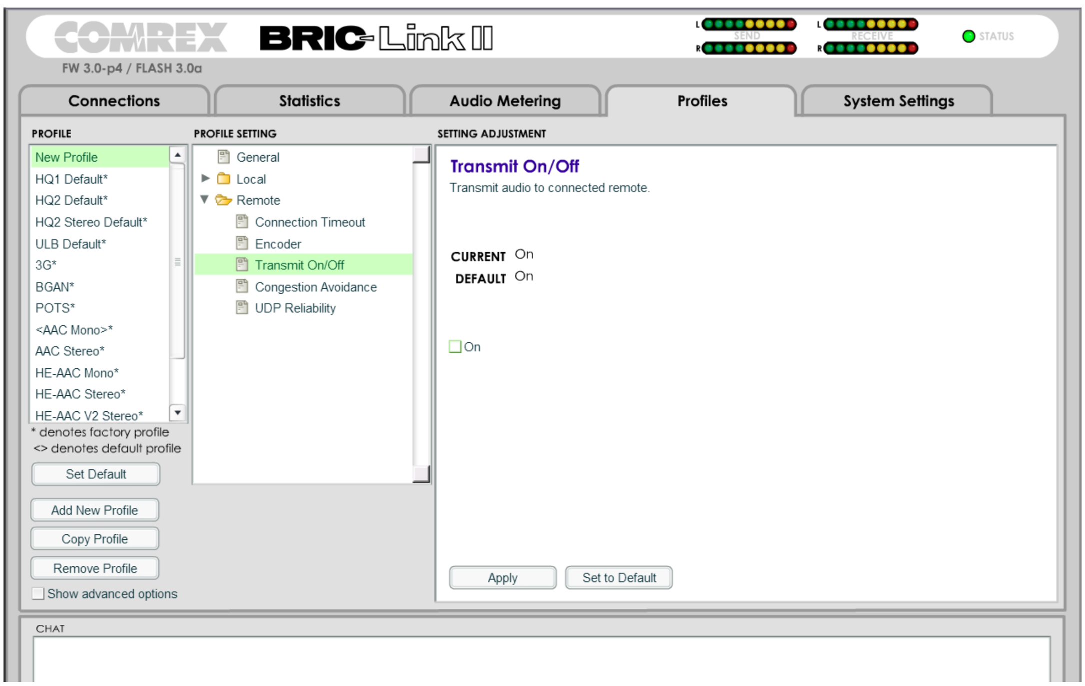

FIGURE 14 **BACKUP REMOTE**

As shown in **Figure 14**, highlight an existing connection (This will be your primary connection) and choose **Change Remote Settings**. In the pop up window, a pull-down box is available to allow you to choose a fallback connection from the list of existing remotes.

After connection, if data is stopped on the primary connection for the length of the timeout value (set in the connection's profile), a connection will be attempted and maintained to the fallback remote.

There is also a box in the **Change Remote Settings** tab labeled **Automatically fall forward**. If this box is checked, the system will constantly attempt to reconnect the primary remote while connected to the fallback remote. If connection is successful, the connection to the fallback will be terminated.

VIII. ABOUT THE ALGORITHMS

When building a profile, there are several choices of encoders to use for each direction of the link. Here's a description of the choices:

AAC

This algorithm is a highly regarded standard for compressing audio to critical listening standards. It has been judged to produce "near transparent" audio at a coding rate of 128 kbps stereo. The standard is a collaborative of several audio companies best efforts, and has become popular as the default audio codec of the **Apple™ iTunes™ program**. AAC should be considered the highest quality codec in BRIC-Link - Enhancements like HE-AAC and attempt to maintain a similar quality and reduced bandwidth and delay.

HE-AAC

This is a newer version of AAC defined for increased efficiency. The goal of the algorithm is to produce AAC comparable quality at a lower bit rate. It does this by encoding lower frequencies to AAC, and higher frequencies using Spectral Band Replication (SBR), a technique that partially synthesizes these high frequencies. HE-AAC is trademarked by other companies as AACPlus™. HE-AAC (and close derivatives) are often used as the main audio codec for digital radio and satellite networks.

HE-AACV2

This algorithm further increases the efficiency of HE-AAC by adding intensity stereo coding. This results in a lower bit rate for stereo signals. We also cluster a very reduced rate HE-AAC mono into this category, although technically it does not contain v2 coding.

LINEAR PCM

This encoder does not compress audio at all. It uses a 48 kHz sampling rate (using analog inputs or 48kHz AES3) 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 804 kbps while Stereo (Dual Mono) Mode requires a network bandwidth over 1.56 Mb/s.

In Linear PCM, if the input AES3 sampling rate is 32kHz or 44.1kHz, the network stream will also run at this rate and required bandwidth will be lower.

FLAC

This encoder compresses the audio data using a lossless algorithm. This means that the audio extracted from the decoder is identical to the audio input to the encoder, with no coding artifacts. FLAC typically removes 30-40% of the network data compared to Linear PCM, but the actual data rate is variable and is based on the complexity of the coded audio.

Using FLAC over Linear PCM typically results in a slightly higher (5ms) overall delay.

G.711

G.711 (µ-law and a-law) - These are the coding algorithms used by standard digital POTS calls, and provide about 3kHz (telephone quality) audio. µ-law is utilized in North America, while a-law is prevalent in Europe. These algorithms are provided for compatibility with SIP-style VoIP phones, but don't provide much benefit over standard telephony in audio terms.

G.722

G.722 - This is a well known 7kHz (medium fidelity) algorithm used in some VoIP telephones and codecs. It is provided for compatibility purposes, but is not considered a superior algorithm for audio codecs.

OPUS

Opus - A newer offering that combines low delay and low network utilization. Opus is included primarily for compatibility with softphone apps, and Internet connections using WebRTC (see Technotes about WebRTC on the Comrex website). Special CBR modes are offered for compatibility with Tieline products - avoid these in other applications.

IX. MULTI-STREAMING

BRIC-Link supports the ability to run one encoder per box, but this single encoder stream may be sent to up to three 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 Multicast, which is described in the next section.

Each BRIC-Link 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 15 shows an BRIC-Link multi-stream arrangement. BRIC-Link A is the multi-streamer, with BRIC-Link B, C and D listening to the same audio. In order to set up a multi-stream scenario, you will need to know how to turn BRIC-Link encoders **Off**. This must be done by building a profile with either the **Local** or **Return Transmitter** mode set to **Off**, as shown in **Figure 16**.

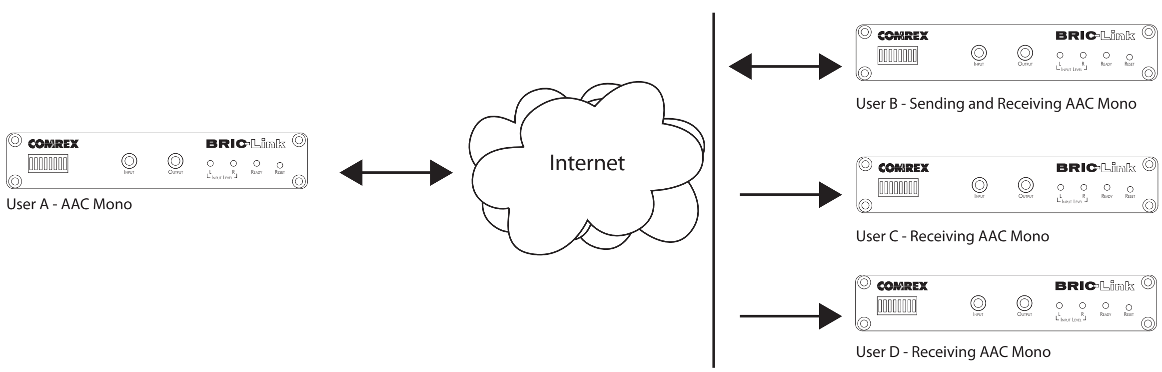

FIGURE 15 **MULTI-STREAMING ARRANGEMENT**

We'll give two examples of multi-steaming scenarios. The first is an environment where the BRIC-Link that is serving the multi-stream initiates the calls, and in the second the serving BRIC-Link accepts all its incoming connections.

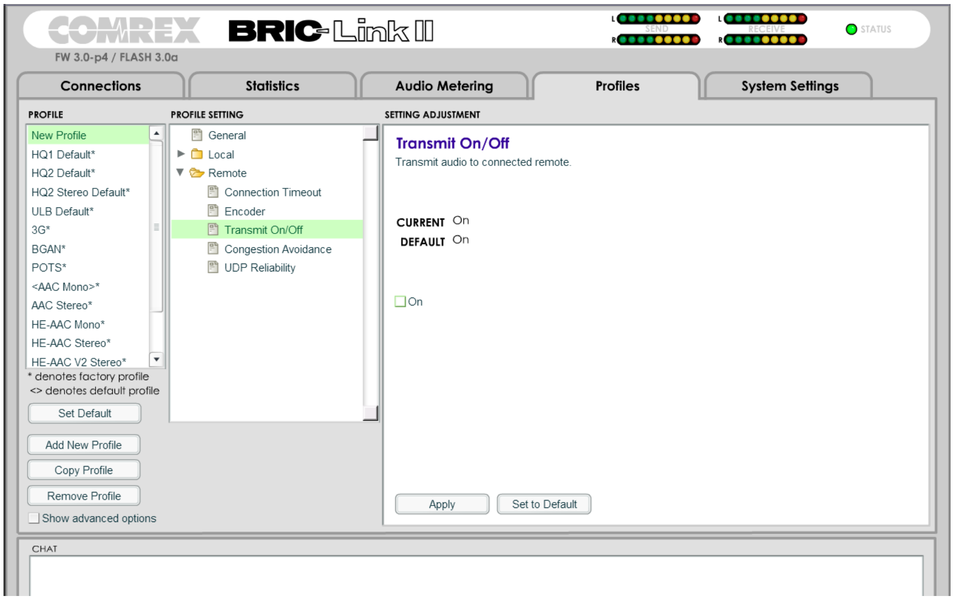

FIGURE 16 **TRANSMIT ON/OFF**

In the "multi-streamer as caller" model, two different profiles will be built on BRIC-Link A. The first profile, labeled "Multi-Duplex", will be defined as a normal, full-duplex BRIC-Link connection. The encoder to be used will be selected in the **Local Encoder** section, and the stream desired in return will be defined in the **Remote Encoder** section.

The second profile is called "Multi-Simplex" and in this profile the **Remote Transmitter** is turned **Off**. Most other selections in this profile are irrelevant.

User A will define remote connections for BRIC-Link B, C, and D. He will assign the "Multi-Duplex" profile to BRIC-Link B, and "Multi-Simplex" profile to the others. He will then establish a connection with BRIC-Link B first, followed by C and D.

In model number 2 where the serving BRIC-Link accepts all incoming connections, all the profiles are built on the **Remote Receivers**. BRIC-Link B will use a simple profile by defining the encoders in each direction, and assign it to BRIC-Link A. BRIC-Link C and D will each define a profile with their **Local Encoders** turned **Off**, and assign them to A. BRIC-Link B should connect first. When C and D connect, they will hear the same stream as B, regardless of how their **Remote Encoders** are set in their profiles.

In a multi-streaming environment, the first man wins. For example, the first connection made between units will determine the encoders used for all others. After the first full-duplex connection is made, all other attempts at full-duplex connections to either end will be rejected.

X. IP MULTICAST

IP Multicast is an efficient way of delivering BRIC-Link digital audio streams to multiple locations. This involves relying on the network to distribute the stream to the locations that require it, rather than creating an independent stream for each user.

IP Multicast requires the use of an IP Multicast-capable network. The commercial Internet, with few exceptions, is not capable of supporting IP Multicast. Some private LANs and WANs are IP Multicast capable.

IP Multicast supports only a single direction stream. An IP Multicast encoder can not receive input streams.

In this manual, we assume that IP Multicast users will be familiar with the basic concepts of setup and operation of the network, so we will focus on how to configure BRIC-Link for Multicast mode.

MULTICAST PROFILES

To set any remotes to Multicast, you must first create a profile for either a Multicast Sender or a Multicast Receiver on the **Profiles** tab.

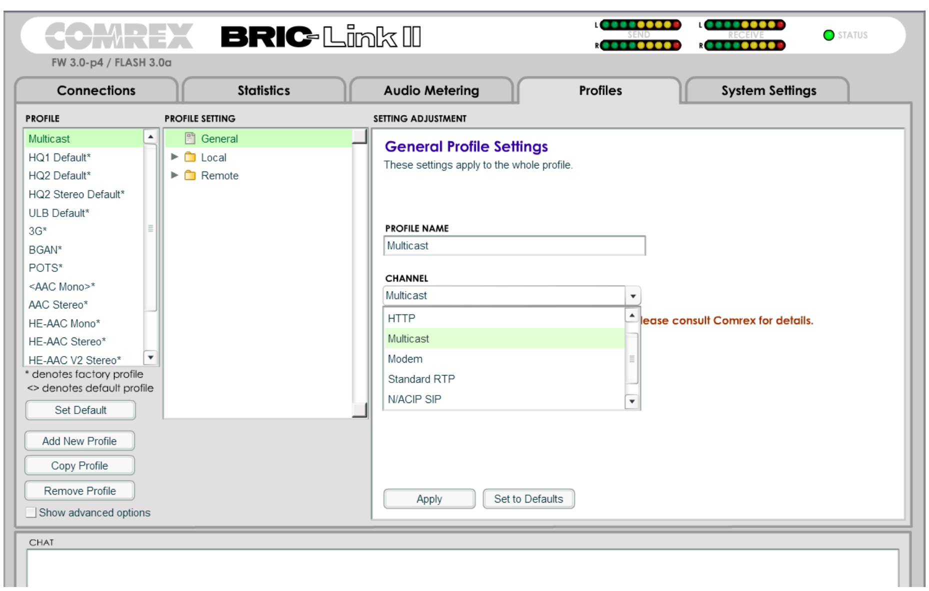

FIGURE 17 **MULTICAST SETTINGS**

As shown in **Figure 17**, when you define a new profile, you have the option to choose **Multicast** as the profile type. Multicast profiles have fewer options than other profile types, and some of the available options will have no effect (e.g. setting an encoder type on a Multicast receiver has no effect). The important settings for Multicast are:

- **Sender/Receiver** - Determines whether this particular BRIC-Link is designed to generate the IP Multicast stream (send) or decode one (receive).
- **Encoder Type** - Determines the type of stream to be used by the Multicast encoder - not relevant for decoders.

In addition to the basic options for IP Multicast profiles, clicking the **Advanced** box will allow setting of the same **Advanced Options** available for Normal BRIC (Unicast) profiles. See the **Profiles** tab section for more information.

SETTING UP A MULTICAST REMOTE

All Multicast connections are outgoing connections - A Multicast Sender must initiate an outgoing stream, and a Multicast Receiver must initiate an incoming one. These remotes are configured within a special address range known as a Multicast Block, typically **224.0.0.0** to **239.255.255.255**. To establish a Multicast connection, simply define a remote as having an address within the IP Multicast Block, use an IP Multicast profile, and press **Connect**.

TIME-TO-LIVE

Time-to-Live (TTL) is a variable set by Multicast encoders to determine how long a packet is processed before it is dropped by the network. The default value of TTL in BRIC-Link is 0, which limits its use to within a LAN environment. TTL may be manually changed on a Multicast Sender remote by configuring the IP address followed by a “/”, followed by the TTL value. An example remote Multicast encoder could be set for the address 224.0.2.4/255, which would signify an address with the Multicast Block with a TTL of 255 (which is the max value available).

CHANGING PORT NUMBERS FOR MULTICAST

The default port of UDP 9000 may also be changed on Multicast remotes. The port number is assigned in the usual way, directly after the IP address, preceded by “:”, followed by the TTL. As an example, the IP address of a Multicast Sender on port 443 with a TTL of 100 would read:

224.0.2.4:443/100

XI. STREAMING SERVER FUNCTION

BRIC-Link has the ability to act as a streaming server, delivering AAC and HE-AAC to compatible PC based media players. Currently tested media players include **WinAmp, VLC** and **Windows Media Player 12** and up.

By default, streaming server functionality is turned off. To enable it, go to the **System Settings** tab of the User Interface and choose **HTTP Settings** option. Under the first option, set **Accept Incoming Connections** to **Enabled**. This will allow outside users to initiate "pull" connections to your codec.

The default port to serve streams from is **8000**. If you want to change this, it can be done in the HTTP settings under **IP Port**. Note you will need to reference the specific server port in the URL you provide to listeners (see below).

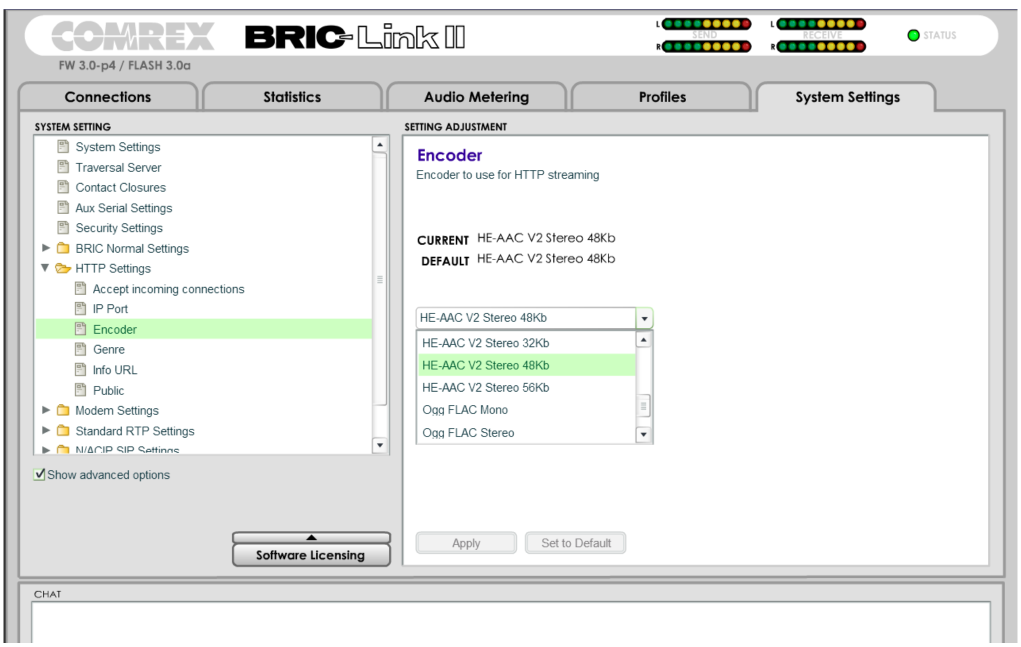

FIGURE 18 **HTTP STREAMING ENCODER**

Next you will need to choose an encoder for use by the streaming server. Only the encoder choices that are compatible with the players listed are shown in this menu. Choices span between a mono audio feed at 18kb/s, up to a stereo feed at 128kb/s. Keep in mind, multiple streams will require this bandwidth along with around 25% overhead for each stream.

The **Genre, Info URL** and **Public** options may be set for anything, or left alone. These options, if applied, will be embedded into the stream.

DECODING A BRIC-LINK STREAM

To decode a stream, open one of the supported players and find the option to open a URL-based stream.

In **Winamp** and **VLC**, input the address of the BRIC-Link in the following format:

http://192.168.0.75:8000
(using the actual IP address, and the actual port if not changed from default **8000**)

In Windows media, input the address like this:

http://192.168.1.75:8000/stream.asx
(using the actual IP address, of course)

SIMULTANEOUSLY CONNECTING BRIC-LINKS AND STREAMING

BRIC-Link can stream while connected to another BRIC-Link in Normal mode. If the BRIC connection is using an AAC algorithm supported by players, when a stream is requested it will be delivered using the same encoder as the BRIC connection, regardless of the HTTP settings. If the BRIC-Link encoder is Linear or FLAC, the stream request will be rejected.

XII. MAKING N/ACIP SIP COMPATIBLE CONNECTIONS

Comrex codecs (and many other brands) have a set of protocols that allow easy IP connections between units. In general, when connecting between Comrex hardware, it's best to use these proprietary modes to take the most advantage of the features of the product.

However, many users are concerned about getting "locked in" to a certain codec brand. Because of this, an international committee was formed by the European Broadcast Union called N/ACIP to hammer out a common protocol to interconnect codec brands. This committee resulted in the establishment of EBU3326, a technical document describing how best to achieve this goal.

EBU3326 by and large establishes a set of features each codec should support, then leaves most of the heavy lifting to other, previously established standards like SIP (IETF RFC 3261). Topics not covered (yet) by EBU3326 include things like carrying ancillary data and contact closures from end-to-end, codec remote control and monitoring, and complex NAT traversal, which at this point are still left to the individual manufacturer's discretion. So if these topics are important to your application, it's best to stick to a single codec vendor and their proprietary protocols.

MORE ABOUT EBU3326

The Tech 3326 document defines several mandatory encoding algorithms, and the transport layer that could be used on them for compatibility. But the most complex part of the standard was the decision on how to arrange Session Initialization, which is the handshake that takes place at the start of an IP codec call. The most commonly used protocol is called SIP, which is used extensively by VoIP phones and therefore was a logical choice. SIP carries the advantage of making BRIC-Link compatible with a range of other non-broadcast products, like VoIP hardware, software, and even mobile phone apps.

EBU3326 IN BRIC-LINK

BRIC-Link does not fully comply with EBU3326, as it does not feature the mandatory MPEG Layer II codec. Aside from this, BRIC-Link has been tested to be compatible with several other manufacturer's devices using encoders supported by both products. When using **N/ACIP SIP Compatible** mode (this is what how the user interface describes EBU3326), ancillary data, contact closures, Switchboard TS, Multi-streaming and Multicasting are not supported. Outgoing call profiles built with the NACIP/SIP channel may lack some advanced options, and can not be set for different encoders in each direction (i.e. N/ACIP SIP calls are always symmetrical).

N/ACIP SIP MODES

A function of placing a SIP-style call is the ability to register with a SIP server. This is a server that exists somewhere on the network, usually maintained by a service provider. Several free servers exist that can offer registration like **GetOnsip.com**.

BRIC-Link allows N/ACIP SIP calls to be placed or received with or without registration on a SIP server. If registration is not enabled, connections are made directly to the compatible device by dialing its IP address, just like in **BRIC Normal** mode.

UNREGISTERED MODE

Placing a call in **Unregistered N/ACIP SIP** mode is simple--just build a profile, but instead of choosing **BRIC Normal** channel, choose **NACIP/SIP**. This will make sure the call is initiated on the proper ports and with the proper signaling. The majority of system settings relating to N/ACIP SIP relate to **Registered** mode.

REGISTERED MODE

Registering with a SIP server in **N/ACIP SIP** mode can have some advantages. When using a SIP server:

- The server can be used to help make connections between codecs through routers
- The remote codec can be dialed by its SIP URI instead of IP address
- The SIP server can be used to find codecs on dynamic IP addresses

SIP SERVERS

A SIP server exists in a domain. This domain is represented by a web-style URL like **sipphone.com** or **iptel.org**. A SIP server or proxy generally handles IP connections within its domain.

SIP URIS

The SIP server assigns a fixed alphanumeric name to each subscribed account. For example, an Iptel user may be assigned the user name comrex_user. URIs consist of a SIP user name, followed by a domain, delineated with the @ symbol, like an email address. Our Iptel user's URI would be comrex_user@iptel.org. Comrex devices do not use the designation "sip:" before a SIP address.

If a connection is to be made exclusively within a domain, the domain name can be left off. As an example, to make a call to this codec from another Iptel registered codec, the dialing string can simply be comrex_user (with the domain being assumed).

REGISTERING WITH A SERVER

At a minimum, you will need the following information when registering ACCESS with a SIP server:

1. The Internet address of your SIP proxy/server (e.g. **proxy01.sipphone.com**)
2. The user name on the SIP account (this is usually the dialing address)
3. The password on the SIP account

Figure 19 shows where this information can be applied in the systems setting section. You will also need to enable the **Use SIP Proxy** option in that menu.

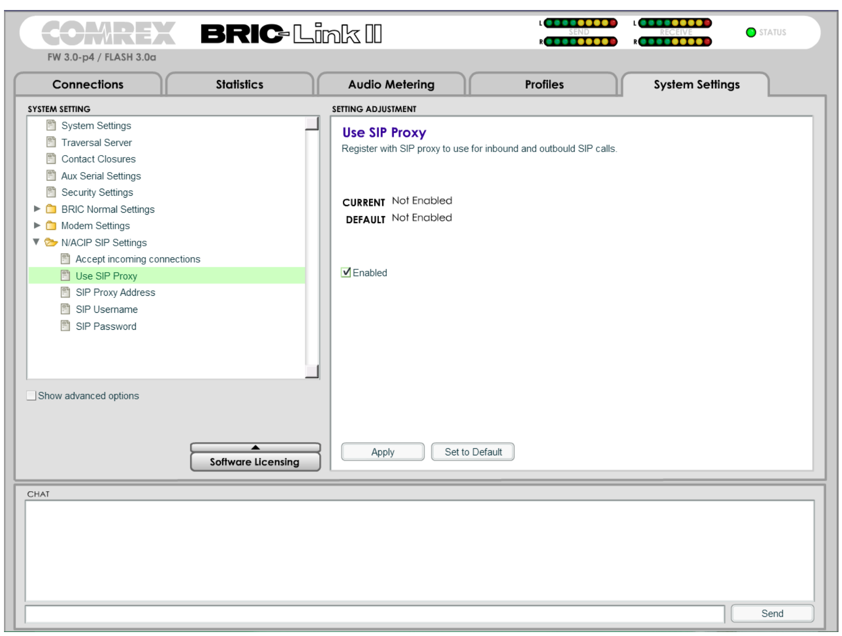

FIGURE 19 **N/ACIP SIP SETTINGS**

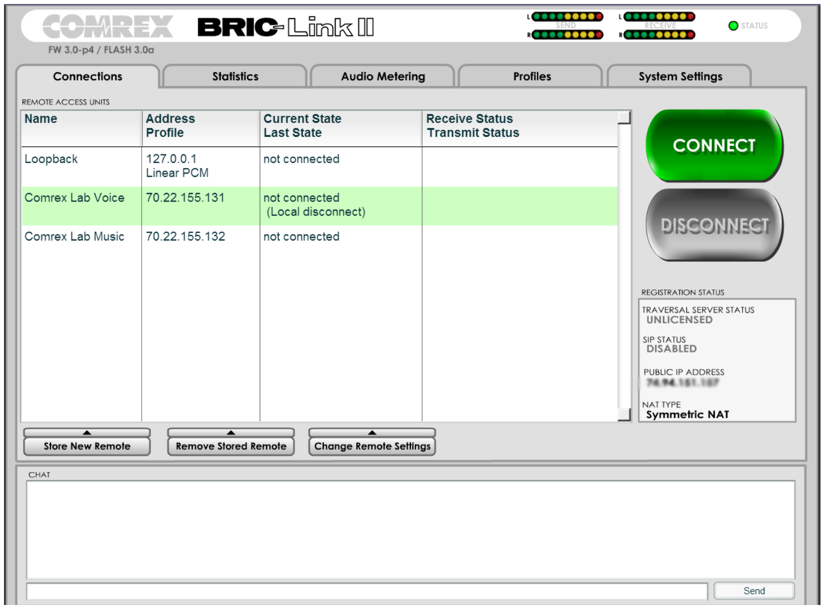

FIGURE 20 **SIP STATUS**

Once this information is correctly entered, a new field appears in the "Registration Status" box located on the **Connections** tab (see **Figure 20**).

The status will reflect the progress of the registration process. When complete, this will display **Online**. If the box does not display **Online** after a short time, it means that registration likely failed. It's best to go back and carefully check the registration info. It might also be useful to be sure the registration information is valid by configuring a VoIP phone or softphone with it and see if that registers.

SIP registration can be very simple with some servers, and others can require more advanced settings. There are several advanced settings available for use with SIP and they are described in the **Advanced Topic** sections.

MAKING SIP REGISTERED CALLS

When registered, calls made using a N/ACIP SIP profile behave differently than normal. The address field, regardless of whether it is a SIP URI or an IP address, is forwarded to the server. No connection attempt is made until the server responds.

If the server accepts the address, the call will be attempted. If not, an error message will appear in the status line. Reasons for call rejection by a server are many. Some examples:

- The server does not support direct connection to IP addresses (if the address is in this format)
- The server does not recognize the address
- The server does not forward calls beyond its own domain
- The server does not support the chosen codec
- The called device does not support the chosen codec
- The address is a POTS telephone number, and POTS interworking is not supported
- The address is a POTS telephone number, and no credit is available (most services charge for this)

ADVANCED N/ACIP TOPICS

The basic entries provided will allow support for the vast majority of N/ACIP SIP based applications. However, there are inevitably situations where the defaults don't work. We've provided some advanced options that can help. As always, these options are located in the Systems Settings and can be made visible by selecting the **Advanced** box.

1. **IP Port** - Universally, SIP connections are supposed to use UDP port **5060** to negotiate calls between devices (and between servers and devices). Note this is only the negotiation channel - actual audio data is passed on the RTP ports. Changing this port number will change which incoming ports are used to initiate connections and to which ports connection requests are sent. Obviously, the change must be made on both devices, and this change will essentially make your codec incompatible with industry-standard VoIP devices.
2. **RTP Port** - This is one of two port numbers used for audio data transfer (the port number directly above this is used as well). Because this port number is negotiated at the beginning of a call (over the IP port), this port may be changed without breaking compatibility. Note that many SIP standard devices use port **5004** for this

function. Due to the negotiation, it is not important that these numbers match on each end. Changing this port to **5004** can actually have an adverse effect, since **5004** is the default port for other services on Comrex codecs.

3 **Public IP Override** - See the next section, **SIP Troubleshooting**, for more information on this option.

4 **Use STUN Server** - See the next section, **SIP Troubleshooting**, for more information on this option.

5 **SIP Proxy Keepalive** - Only applies to **Registered** mode. This variable determines how often the codec "phones home" if registered with a SIP server. It's important that the codec periodically "ping" the server, so the server can find the codec for incoming calls. It can be adjusted primarily to compensate for firewall routers that have shorter or longer binding timings, i.e., the router may have a tendency to "forget" that the codec is ready to accept incoming calls and block them.

6 **SIP Domain** - [only applies to **Registered** mode]. This is the name of the network controlled by the SIP server. This parameter must be passed by the codec to the server. Under most circumstances, this is the same as the server/proxy address, and if this field is not populated, that is the default. If, for some reason, the domain is different than the server/proxy address, then this field is used.

SIP TROUBLESHOOTING

In a nutshell, SIP establishes a communication channel from the calling device to the called device (or server) on port 5060. All handshaking takes place over this channel, and a separate pair of channels is opened between the devices: one to handle the audio and the other to handle call control. The original communication channel is terminated once the handshaking is complete. Note that firewalls must have all three ports open to allow calls to be established correctly. Also, port forwarding may be required to accept calls if your codec is behind a router.

The main area where SIP complicates matters is in how an audio channel gets established once the handshake channel is defined. In the common sense world, the call would be initiated to the destination IP address, then the called codec would extract the source IP address from the incoming data and return a channel to that address. In fact, that's how the default mode of Comrex codecs work, and it works well.

But SIP includes a separate "forward address" or "return address" field, and requires that a codec negotiating a call send to that address only. This is important in the case of having an intermediate server. And this works fine as long as each codec knows what its public IP address is.

OUTGOING CALL ISSUES

A unit making an outgoing call must populate the ”return address” field. But any codec sitting behind a router has a private IP address, and has no idea what the public address is. So, naturally, it will put its private IP address (e.g. **192.168.x.x** style) address into that “return address” field. The called codec will dutifully attempt to connect to that address and undoubtedly fail, since that can’t be reached from the Internet at large.

INCOMING CALL ISSUES

Incoming calls to codecs behind routers are complicated by the fact that ports on the router must be forwarded to the codec. In the case of SIP, this must be three discrete ports (For Comrex codecs these are UDP 5060, 5014 and 5015)<6014 and 6015 with 3.0 firmware>. And since even the “forward address” is negotiated in SIP, the incoming unit is likely to populate the “forward address” field with its private address as well.

SOLUTIONS

Many times the “return address” field issue is fixed by the SIP server (in **Registered** mode) and no compensation measures are necessary. Often, in fact, the server insists on acting as a “proxy” and handles all the traffic itself--outgoing and incoming streams are relayed directly by the server, solving any router issues.

In point-to-point connections, this isn’t possible. All is not lost here, since we can find some hacks to make this work. The first place to look is your router, since many modern routers are aware of this issue and have taken steps to relieve the pain. If your router supports a SIP Application Layer Gateway (ALG), then enabling this option can fix the issue. Essentially, the router will get smart enough to read your SIP handshake, find the outgoing address field, and replace it with your public IP. This is a pretty slick solution, but there may be environments when you are not aware whether this option is supported on your router, or have the ability to enable it. So on to solution two:

STUNNING SUCCESS

Another technique for working around the SIP-Router issue is by using a protocol called STUN. This can be enabled in Comrex codecs in the **Advanced N/ACIP SIP** options and essentially allows for the codec to learn what its public IP address is. It does this by contacting a STUN server out on the Internet (the default one is maintained by Comrex) and simply asking. If this option is enabled, the codec itself will handle the address switching.

Be aware of the dreaded “battling workarounds” issue. In our simple description, we left out the fact that ports are being translated by the router as well as IP addresses. If the ALG-enabled router receives an unexpected result in the SIP address field (as it might if using STUN), it may not translate ports as expected, and it’s likely that the call will fail. When in doubt, the best technique is to try a SIP call with STUN turned off, and if the return channel fails, try enabling STUN.

FIX OF LAST RESORT

Finally, there’s a brute-force option available on Comrex Codecs when STUN ports are blocked by a firewall, or it can’t be used for some other reason. Under **Advanced System Settings**, a field is available called **Public IP Override**. Any address put into that field will be pasted into the address SIP field. So if you know what your public IP address is (you can obtain it from many websites via a browser) you can manually paste it here. Keep in mind, this is often subject to change over time (and obviously if you use a different network) so it’s important to remember this change has been made on your codec.

XIII. LICENSE AND WARRANTY DISCLOSURES FOR COMREX BRIC-LINK

LICENSES

MPEG-4 audio coding technology licensed by Fraunhofer IIS
http://www.iis.fraunhofer.de/amm/

Fraunhofer Institut
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BRIC-Link uses proprietary and open-source software programs. Some of the open-source programs are licensed under the Gnu Public License (GPL). For more information on GPL see http://www.gnu.org.

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Import code in **keyimport.c** is modified from PuTTY's import.c, licensed as follows:

PuTTY is copyright 1997-2003 Simon Tatham

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Libpcap

tcpdump

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XIV. CONFORMITY AND REGULATORY INFORMATION

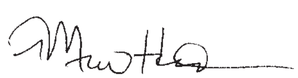
SUPPLIERS' DECLARATION OF CONFORMITY

Place of Issue: Devens, Massachusetts

Date of Issue: April 2, 2009

Equipment: Comrex BRIC-Link

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.



Thomas O. Hartnett, Vice President, Comrex Corporation

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 BRIC-Link
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 (2006/95/EEC)
EN 60950-1: 2001

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

Signed: ______________________

Date: 02 April 2009

XV. APPENDIX A - IP COMPATIBILITY

The BRIC-Link is capable of encoding and decoding a choice of three different types of non-BRIC-Link streams: Standard RTP, **Luci Live** and **Zephyr Xstream**. The choice is exclusive i.e. you must set the BRIC-Link specifically for the type of stream you wish to be compatible with, and you will remain incompatible with the other two types until you change it. This setting has no effect on normal BRIC-Link functions, which continue to operate as before.

1. **Luci Live** - This PDA/PC-based software allows real-time streaming over IP links. As of version 1.2, **Luci Live** includes AAC and HE-AAC in addition to the default MP2 algorithm. BRIC-Link can communicate with **Luci Live** only in Luci's AAC modes. Note: The free demo available from Luci does not incorporate the AAC functions; you must have a licensed and registered copy to use AAC.

To communicate with a **Luci Live** device:

a. **Initial Setup** - This will define all Standard RTP connections to be Luci Compatible

b. **BRIC-Link** - On the **System Settings** tab, open the **Standard RTP Settings** option and choose **RTP Compatibility Mode**. On the pull-down box, choose **Luci Live**.

c. **Incoming Connections** - **Luci Live** sends either an AAC or HE-AAC stream to the BRIC-Link on UDP port 5004. These streams will be automatically decoded. By default, a return channel of AAC 56kb/s mono is returned to the **Luci Live** product. The return channel may be altered to any Luci-compatible mode in the **Systems Setting** section.

d. **Outgoing Connections** - Build a profile using the **Profile Manager** on the BRIC-Link and select a **Channel Mode** of **Standard RTP**. Then choose a Luci-compatible encoder for the outgoing call. The Luci software will control what type of stream, if any, is returned to the BRIC-Link.

2 **Zephyr Xstream** - Xstream Firmware version 3.2.0 and higher support an “RTP Push” function that is compatible with BRIC-Link in some modes. BRIC-Link is not currently compatible with the Xstream’s HTTP and SIP streaming functions. There are several limitations imposed by the Xstream when using the RTP Push function:

a On the Xstream, only AAC and MP3 coding are available in this mode, and BRIC-Link is only compatible with the AAC mode.

b The Xstream uses downsampling in modes below 96Kb/s, which is not supported by BRIC-Link.

c In order for an Xstream to decode an BRIC-Link stream, the default decoder setting must be changed from <Auto> to <AAC> in the codec menu of the Xstream.

To communicate with a **Zephyr Xstream**:

a **Initial Setup** - This will define all Standard RTP connections to be Xstream Compatible.

b **BRIC-Link** - On the **System Settings** tab, open the **Standard RTP Settings** option and choose **RTP Compatibility Mode**. On the pull-down box, select **Zephyr Xstream**.

c **Incoming Connections** - **Zephyr Xstream** sends an AAC stream to the BRIC-Link on UDP port 9150. These streams will be automatically decoded. By default, a return channel of AAC 96kb/s mono is returned to the Xstream. The return channel may be altered to any Xstream-compatible mode in the **Systems Setting** section.

d Outgoing Connections - Build a profile using the **Profile Manager** on the BRIC-Link and select a **Channel Mode** of **Standard RTP**. Then choose an Xstream-compatible encoder for the outgoing call. The Xstream will control what type of stream, if any, is returned to the BRIC-Link.

3 **Standard RTP** - This mode is set to receive a basic, unformatted AAC stream within a standard RTP/UDP structure. At present, this mode does not offer compatibility with other industry devices.

XVI. APPENDIX B - USING BRIC-LINK ON UNIDIRECTIONAL NETWORKS

Under most circumstances, BRIC-Link and ACCESS require an IP path in both directions for successful connections, even when audio is being sent only one-way. For networks that provide data only in one direction, it is possible to use **Standard RTP** mode to establish and maintain these links. This section describes how to set that up.

The following setting applies to both codecs in the link (encoder and decoder):

The codec has several compatibility modes under the **Standard RTP** channel mode. The units default to a mode that is compatible with the **Luci Live** PC-based encoder. This must be changed on both codecs.

1. On the BRIC-Link or ACCESS Rack, enter the **Web-based User Interface** and choose the **System Settings** tab. On the ACCESS Portable choose **Configure** > **System Settings**
2. Find the **Advanced** tick-box and check it
3. Find **Standard RTP Settings** and choose to edit the **RTP Compatibility mode**
4. Change this setting to **Standard** and click **Apply** (or **Save** on ACCESS Portable)

DECODE SIDE SETTINGS ONLY

Also under **Advanced Standard RTP Settings**, find the **Return Channel Enable** entry. Disable the return channel and click **Apply** (or **Save** on ACCESS Portable). This will make sure that no channel will be set up in the direction to the encoder.

ENCODE SIDE SETTINGS ONLY

Obviously, connections of this type must be established from the encoding side of the link. So you'll need to build a new Profile that uses the **Standard RTP** channel mode under the Profile Editor. Choose your outgoing encoder along with any other special attributes in the profile editor. Name the Profile something descriptive like "Simplex".

Next, create your outgoing remote entry in the address book. Apply the new profile to that entry. Any connection made with that entry will connect in a unidirectional fashion.

FULL-TIME OR TRIGGERED CONNECTIONS

A remote entry using a unidirectional profile can still utilize the tools required for automatic connection.

To set up a connection to be “always active” (i.e. reconnect in the case of power outage or network failure), choose that connection on the **System Settings** tab as the **Always Connect To** location.

To trigger the connection when an external contact is closed, choose the connection under one of the **Contact Closure** settings on the **System Settings** tab.

XVII. INFORMATION FOR IT MANAGERS

The purpose of this appendix is to describe all open ports and services available on the Comrex BRIC-Link. If a service is not mentioned here, it is disabled by default.

The Comrex BRIC-Link is a device designed to move real-time, wideband audio over IP networks. The main network interface is 1000baseT Ethernet.

The device contains an optimized version of the **2.6 Linux kernel**. The IP parameters are set by a PC on the local LAN using a proprietary broadcast UDP protocol. Comrex provides a Java-based application to perform this function on the local PC. After five minutes of operation, this function is disabled.

Updates to the system are provided by a custom on-line updater utility. This update process is password protected and done via XML over TCP port **8080**. In addition to the password protection, the update data itself must have a valid cryptographic signature from Comrex, or else it is rejected. In order for the unit to be factory updated, TCP port **8080** must be forwarded to the device. Alternately, updates can be initiated from any local PC using the Comrex supplied java based update utility.

The BRIC-Link codec delivers an RTP/UDP stream from source port **9000** to destination port **9000** by default. By default it listens for incoming RTP/UDP streams on port **9000**. To use the default mode, only UDP **9000** needs to be forwarded to the device.

Alternately, the device can be configured to deliver a similar TCP-based stream on TCP port **9000**. By default, the device listens for incoming TCP streams on TCP **9000**. This function may be disabled. The source port of TCP streams is ephemeral, and, if an incoming stream is detected, one will be returned to the ephemeral port.

The device also supports transmitting and receiving UDP multicast streams, using UDP port **9002** unless another port is specified by the user. This is not enabled by default, and a configuration must be explicitly created and connected on both ends for this function. Multicast streams are inherently unidirectional, and the configured port must be forwarded to the device on the receiving end. The multicast TTL value defaults to 1 (for in-network multicasting), but may be changed to any valid TTL by the user.

Outgoing ports and incoming ports may be altered via the user interface.

For compatibility with other industry devices, the BRIC-Link also listens for incoming streams (and can place outgoing streams) on UDP **5004** and **9150**. These ports may be changed via the user interface, and this function may be disabled.

By default, the device serves as an SSH host on TCP port **22**. Only SSH clients with an authorized DSA key can access SSH services on the device. Other forms of authentication are disabled. This key is kept confidentially by Comrex for factory diagnostics only. SSH services may be disabled completely via the user interface.

Under normal operation, the device is controlled by a networked computer via a web page served from the device on the standard HTTP port **80** (TCP). This page requires Adobe Flash player on the browser; and the flash plugin establishes a TCP connection back to the device on the XML port **8080**. Both of these ports are required for the remote UI to function, and the port assignments are configurable. These services may be disabled by the user interface, but this will disable both the remote GUI and the on-line updater.

The device will respond to standard ICMP requests.