Bridge-IT
IP Codec User Manual

Software Version: 2.14.88
Manual Version: v.2.2_20151125
November, 2015

Table of Contents

1 How to Use the Documentation

Overview of this User Manual

Use this manual to learn how to:

- Connect the codec to an IP network and configure peer-to-peer, multicast or multi-unicast connections.
- Configure the codec over a LAN or USB cable.
- Adjust audio and other settings within the codec.
- Configure automatic SDHC card backup.

Please read Getting Connected Quickly for an overview of how to configure the codec using 'programs' to store connection settings.

Manual Conventions

Warnings: Instructions that, if ignored, could result in death or serious personal injury caused by incorrect operation of the equipment. These must be observed for safe operation.

Cautions: Instructions warning against potential hazards, or to detail practices that must be observed for safe operation and to prevent damage to equipment or personnel.

Important Note: Information you should know to connect and operate your codec successfully.

Typographic Conventions

- Codec software elements are in Arial bold, e.g. **Contacts**
- Codec hardware elements are in bold Capitals, e.g. **KEYPAD**

Help Button

Press the (information/help) button on the codec when navigating codec menus to display a help dialog on the LCD screen suggesting actions which can be performed from the current menu item.

2 Warnings and Safety Information

THUNDERSTORM AND LIGHTNING WARNING:
DO NOT USE Tieline codecs during thunderstorms and lightning. You may suffer an injury using a Tieline codec, or any device connected to a LAN connection during a thunderstorm. This can lead to personal injury and in extreme cases may be fatal. Protective devices can be fitted to lines, however, due to the extremely high voltages and energy levels involved in lightning strikes, these devices may not offer protection to users, the Tieline codec and equipment connected to the codec.

Secondary strikes can occur. These secondary strikes are induced by lightning strikes and also produce dangerously high currents and energy levels. You only need to be near an object struck by lightning to lead to personal injury or damage to equipment. e.g. if located near a lighting tower at a sports facility, water features and drains on golf courses you may be affected by these secondary strikes.

Damage to personnel and Tieline codecs may occur during thunderstorm, even if the codec is turned off but is connected to the system or the power.

ANY DAMAGE TO A TIELINE PRODUCT CAUSED BY LIGHTNING or an ELECTRICAL STORM WILL VOID THE WARRANTY. Use of this product is subject to Tieline's SOFTWARE LICENSE and WARRANTY conditions, which should be viewed at www.tieline.com/support before using this product.

Disclaimer

Whilst every effort has been made to ensure the accuracy of this manual we are not responsible for any errors or omissions within it. The product specifications and descriptions within this manual will be subject to improvements and modifications over time without notice, as changes to software and hardware are implemented.

3 Glossary of Terms

AES/EBU	Digital audio standard used to carry digital audio signals between devices.
AES3	Official term for the audio standard referred to often as AES/EBU.
DNS	The Domain Name System (DNS) is used to assign domain names to IP addresses over the World-Wide Web.
DSCP	The Differentiated Services Code Point is a field in an IP packet header for prioritising data when traversing IP networks
Fail over	Method of switching to an alternative audio stream if the primary connection is lost.
GUI	Acronym for Graphic User Interface
ISP	Internet Service Providers (ISPs) are companies that offer customers access to the internet
IP	Internet Protocol; used for sending data across packet-switched networks.
LAN	Local Area Network; a group of computers and associated devices sharing a common communications link
Latency	Delay associated with IP networks and caused by algorithmic, transport and buffering delays.
Multicast	Efficient one to many streaming of IP audio using multicast IP addressing.
Multi-unicast	A multi-unicast program (also known as multiple unicast) can transmit a single audio stream with common connection settings to a number of different destinations.
Network Address Translation (NAT)	A system for forwarding data packets to different private IP network addresses that reside behind a single public IP address.
Packet	A formatted unit of data carried over packet-switched networks.
Port Address Translation (PAT)	Related to NAT; a feature of a network device that allows IP packets to be routed to specific ports of devices communicating between public and private IP networks.
QoS (Quality of Service)	Priority given to different users or data flows across managed IP networks. This generally requires a Service Level Agreement (SLA) with a Telco or ISP.
Redundancy	Choosing an alternative audio stream to use if a primary audio connection is lost.
RTP	A standardized packet format for sending audio and video data streams and ensures consistency in the delivery order of voice data packets.
SDP	SDP defines the type of audio coding used within an RTP media stream. It works with a number of other protocols to establishes a device's location, determines its availability, negotiates call features and participants and adjusts session management features.
SIP	SIP works with a myriad of other protocols to establish connections with other devices. It is used to find call participants and devices and is the method used by most broadcast codecs to connect to competing brands of codec for interoperability.
SLA	Service Level Agreements (SLAs) a contractual agreement between an ISP and a customer defining expected performance levels over a network
STL	Studio to transmitter link for program audio feeds.
TCP	TCP protocol ensures reliable in-order delivery of data packets between a sender and a receiver. Its two functions include controlling the transmission rate of data and ensuring reliable transmission occurs. Generally not well-suited to streaming live audio because buffering (latency) is employed to ensure data packets are received in order
TTL	Time-to-Live is the setting used in multicast servers to ensure data packets

	have a finite life and don't cause congestion over networks.
UDP	The protocol most commonly used for sending internet audio and video streams. UDP packets include information which allows them to travel independently of previous or future packets in a data stream. In general, UDP is a much faster and more efficient method of sending audio over IP.
Unicast	Broadcasting of a single stream of data between two points.
WAN	Wide Area Network; a computer network spanning regions and/or countries to connect separate LANs

4 Introduction to the Codec

Welcome to Tieline's Bridge-IT, the ultimate low-cost, high-performance, stereo IP audio codec solution for broadcast and professional applications. Capable of both peer-to-peer or multi-point connections, Bridge-IT transports audio streams reliably, simply and effectively over IP data networks such as wired and wireless LANs, WANs, the internet, satellite IP, Wi-MAX and Wi-Fi.

Bridge-IT is perfect for a large range of broadcast and professional applications that include:

- Studio-to-Transmitter Link (STL) applications
- Stereo multi-unicast IP audio distribution (stereo to up to 6 endpoints)
- Simple remote broadcast links
- IP multicasts over compatible IP networks
- Low-latency audio over IP bridging solutions
- Multiple codec installations (2 codecs fit in 1 x 19" rack unit)

Codec Features

The following table outlines the features available in Bridge-IT.

Bridge-IT Features	
Peer-to-peer mono and stereo IP audio	✓
SmartStream IP software for automatic management of IP connection streaming	✓
High quality low-delay linear audio	✓
G.711 G.722, MPEG Layer 2, plus low-delay Opus, Tieline Music and MusicPLUS algorithms	✓
Simultaneous analog and digital XLR AES/EBU outputs	✓
Automatic SD/SDHC card connection failover	✓
Web-GUI for remote control and configuration	✓
EBU N/ACIP Tech 3326 compatibility over IP	✓
2 relay inputs and 2 opto-isolated outputs plus RS-232 for local and remote control of equipment at either end of your codec link	✓
Multi-unicast and send 6 stereo connections, one with bidirectional audio	✓
AAC-LD, AAC-ELD, LC-AAC, HE-AAC v.1 and HE-AAC v.2 algorithms	✓
TieServer for automatic firmware upgrade notification	✓
16 bit and 24 bit aptX® Enhanced algorithm	**O**

✓ = included
O = optional

Package Contents

Your codec is delivered with:

- Bridge-IT IP codec

* Multi-region plug pack 12 volt 1 Amp power supply
* Phoenix 7-way connector for control port activation

If any of the parts are incorrect, missing, or damaged, contact Tieline or your nearest authorised dealer.

5 Front Panel Controls

Bridge-IT features a full hardware front panel interface with menu navigation buttons, an LCD display with PPM metering and a dialing keypad.

The codec also features an SDHC card slot, which can be used for automatic program audio backup, audio playout and in-store audio recordings.

Navigation Buttons

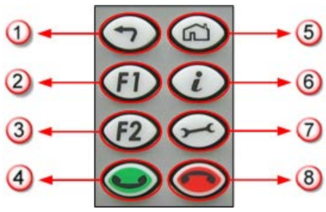
Bridge-IT has four arrow shaped navigation buttons for navigating codec menus and an **OK** button for selecting menu items.

Dialing Keypad

The keypad has alpha-numeric buttons and operation buttons used to:

- Launch codec functions.
- Navigate menus.
- Dial and hang up connections.
- Configure contact details.

Operation Button Descriptions



	Features	Operation Button Descriptions
1	Return Button	Press to navigate back through menus & delete characters
2	Function Button 1	Press to open codec user functions
3	Function Button 2	Press to open codec user functions
4	Connect Button	Press to dial IP connections
5	Home Button	Press to return to home screen
6	Information Button	Press to view a help menu on-screen
7	Settings Button	Press to configure codec settings
8	Hangup Button	Press to end a call

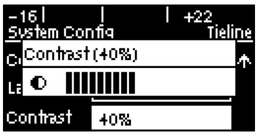
Adjusting LCD Screen Contrast Levels

1. Press and hold the button and then press the arrow up button to display the **Contrast** adjustment screen.
2. Use the left and right arrow buttons to adjust the LCD screen contrast until viewing is

optimised.

3. Press when you have finished.

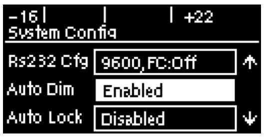
Contrast can also be adjusted by pressing the **HOME** button, selecting **Settings**, then **System**, and using the down ▽ button to navigate to **Contrast**.



Enabling and Disabling LCD Screen Auto Dim Mode

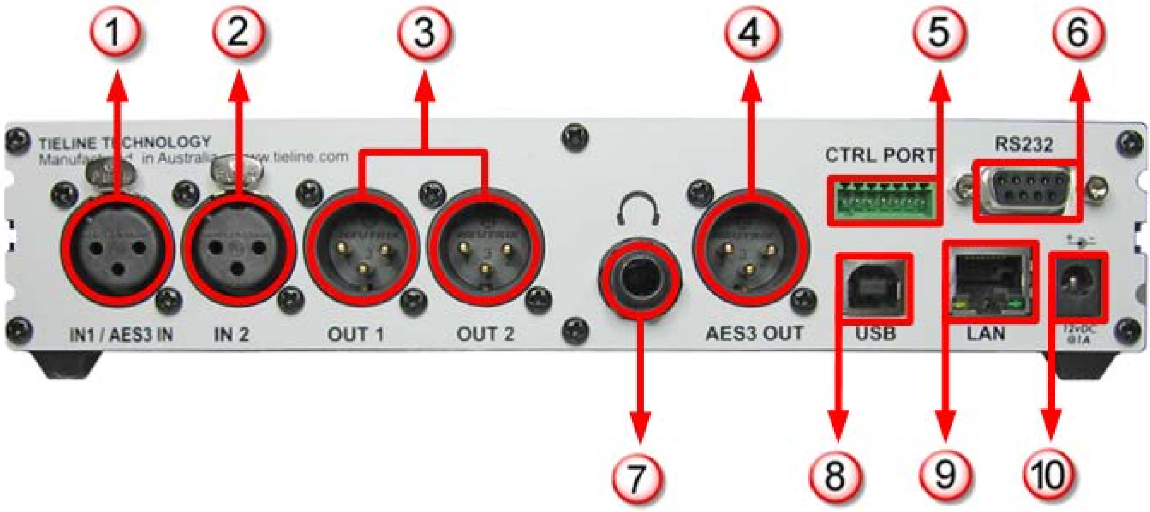
By default the codec LCD screen has **Auto Dim** mode enabled. This dims the intensity of the display 30 secs after inactivity and is designed to maximise the working life of the screen. Disable this mode if you want the screen to be illuminated at all times.

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Settings** and press the button.
3. Navigate to **System** and press the button.
4. Navigate to **Auto Dim** and press the button to toggle between **Enabled** and **Disabled**.



Important Note: The default **Auto Dim** time-out is reduced from 30 seconds to 10 seconds when the **Auto Lock** function is enabled (to lock the front panel controls). Disabling **Auto Dim** mode will override all time-out periods and the LCD will remain fully illuminated at all times.

6 Rear Panel Connections



	Features	Rear Panel Connection Descriptions
1	IN1/AES3 IN	Balanced Female XLR Mic/Line input 1 or AES3 (AES/EBU) input for mono and stereo AES3 sources
2	IN 2	Balanced XLR Line input 2
3	Out 1/Out2	Balanced Male XLR Analog Line Outputs 1 and 2
4	AES3 OUT	AES3 (AES/EBU) output via male XLR for mono and stereo AES3 signals
5	CTRL PORT	2 inputs/2 opto-isolated relay outs
6	RS232	RS-232 (DB9) connection for serial device control
7	Headphone Output	Stereo 6.35 mm (¼") headphone out
8	USB 2.0 port	USB 2.0 for codec configuration
9	LAN	RJ-45 port for Ethernet 10/100 BaseT network connection
10	12v DC input	2.1mm 12V DC power supply connection

XLR Analog and Digital Inputs

Bridge-IT features two XLR microphone inputs.

Input 1 is a balanced mic/line input with the ability to connect high, medium and low gain mics, as well as an unbalanced source. It has switchable phantom power of 15 volts that is turned off by default and can also be used as an AES3 (AES/EBU) digital input. This input accepts both mono and stereo digital AES3 signals. Input 2 is a line input only.

Important Note: Channel 2 input gain can only be adjusted using the **Input Audio** screen on the codec (See: Adjusting Input Levels)

XLR Analog and AES3 Outputs

Bridge-IT features two balanced XLR analog audio outputs and a digital XLR AES3 (AES/EBU) audio output. Both the analog and digital outputs can be used simultaneously and the AES3 output can send both mono and stereo signals via the single XLR output.

Stereo Headphone Jack Output

Bridge-IT has a 6.35mm (1/4") stereo headphone output jack for monitoring audio inputs 1 and 2 and return link audio. Channel 1 is mapped directly to the left headphone output and channel 2 is mapped directly to the right headphone output. When listening to return link audio channel 1 is mapped directly to the left headphone output and channel 2 is mapped directly to the right headphone output. (See Headphone Monitoring)

LAN Port

The codec features a RJ-45 port for Ethernet 10/100 BaseT network connections.

Command & Control Interfaces

Bridge-IT features:

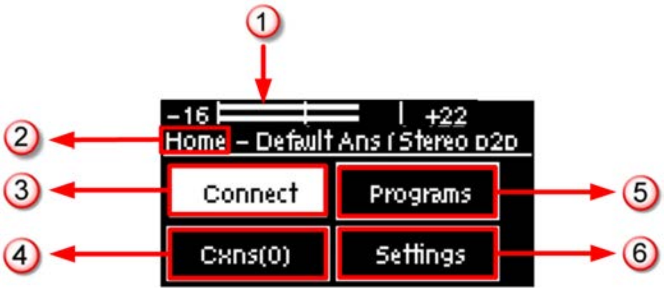
1. 2 relay inputs and 2 opto-isolated outputs for machine control via the **CTRL PORT**.
2. A nine pin **RS-232** connection for local and remote control of equipment at either end of the link
3. A **USB** 2.0 (slave) connection for codec web-GUI configuration.

DC Power Input

The codec is powered by a 12 volt DC power supply using a standard polarised DC plug.

7 Navigating Codec Menus

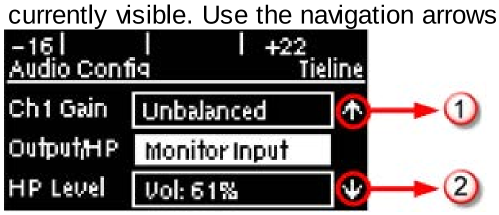
The codec has simple and intuitive menu navigation screens. All main codec menus can be launched from the **Home** screen and audio levels remain visible throughout all menus.



Bridge-IT Home Screen

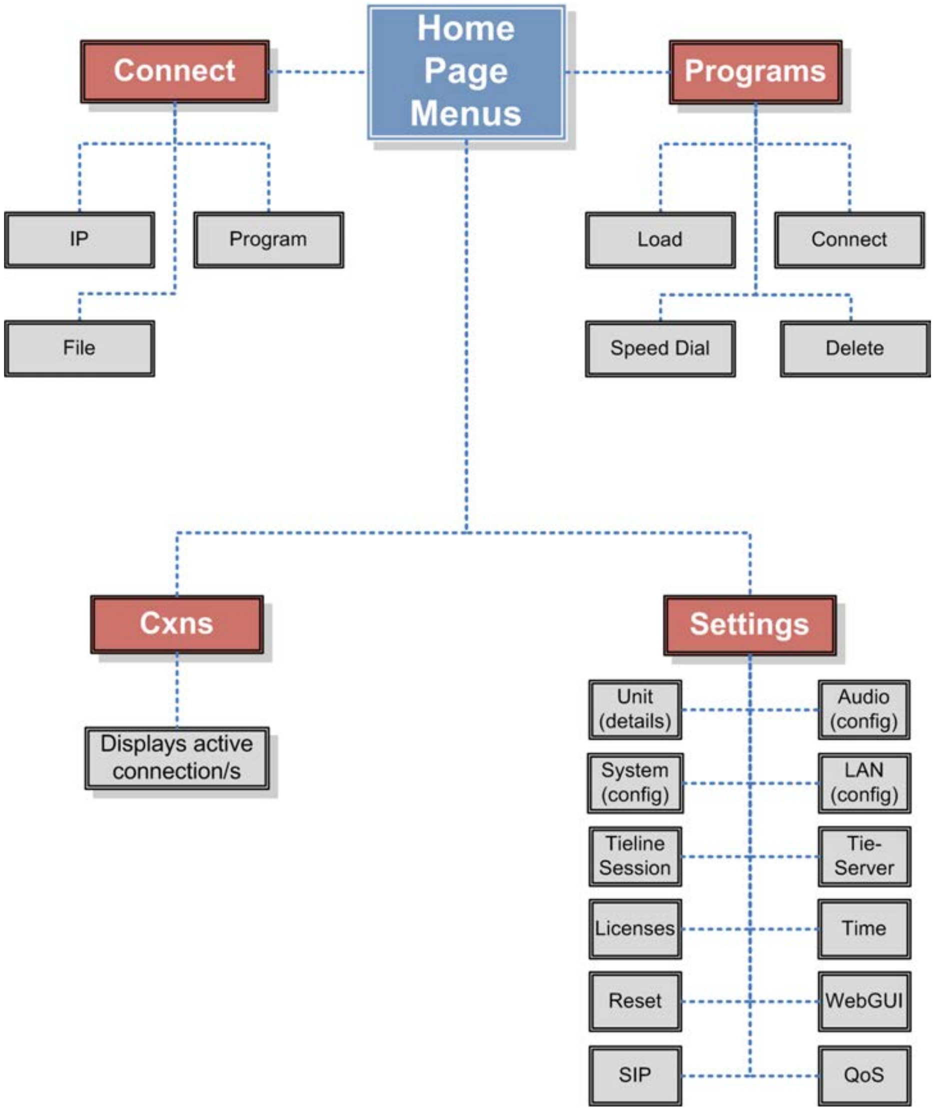
	Features	Codec Home Screen Elements
1	PPM Meters	Left (top) and right channel audio levels
2	Screen Name	The name of the current screen being navigated
3	Connect	Select to dial & adjust connection settings
4	Cxns	Displays the number of current connections
5	Programs	View and edit Program dialing configurations
6	Settings	Select to configure codec settings

Press the **RETURN** button to navigate backwards through menus, or press the **HOME** button to return to the **Home** screen from any menu. If a full menu cannot be viewed on the codec screen then arrows on the right hand side of the screen indicate that the current menu has items below and/or above the items currently visible. Use the navigation arrows to scroll up and down.

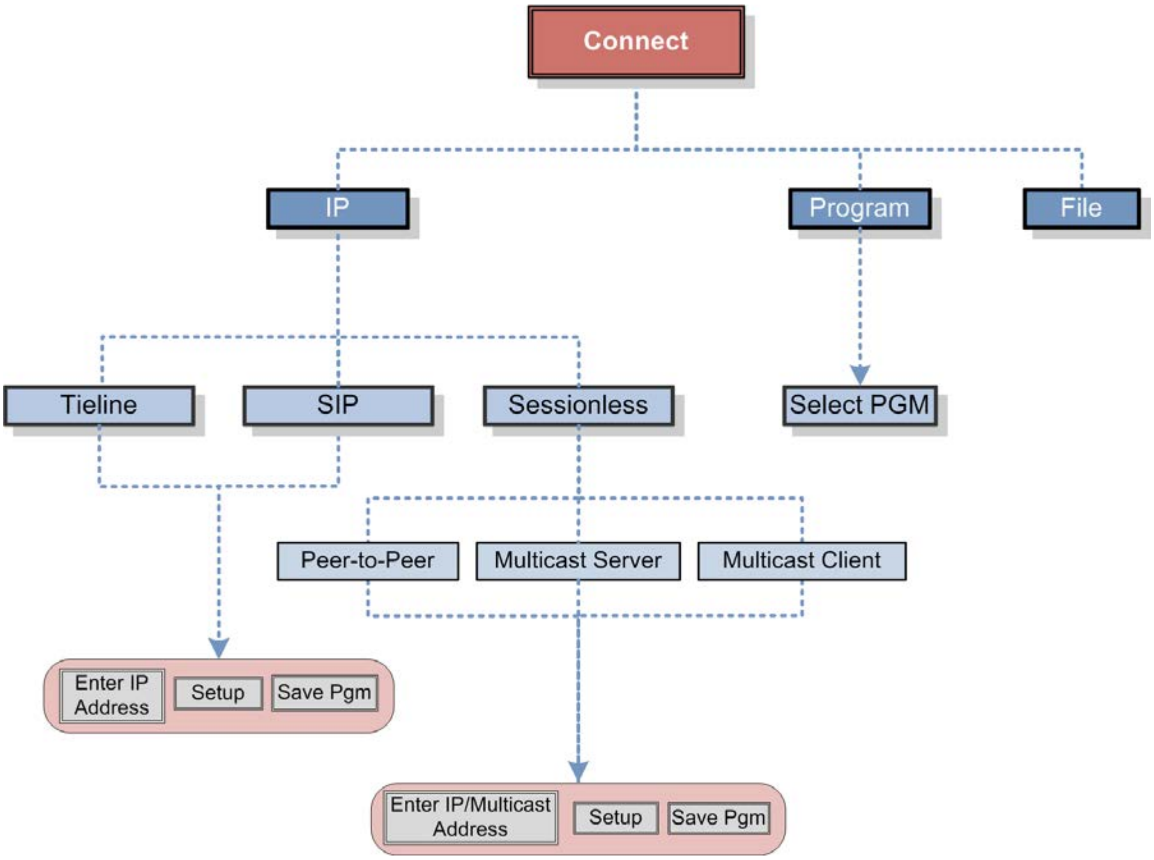


	Features	Codec Home Screen Elements
1	Up Arrow	Arrow indicating menus can scroll upwards
2	Down Arrow	Arrow indicating menus can scroll downwards

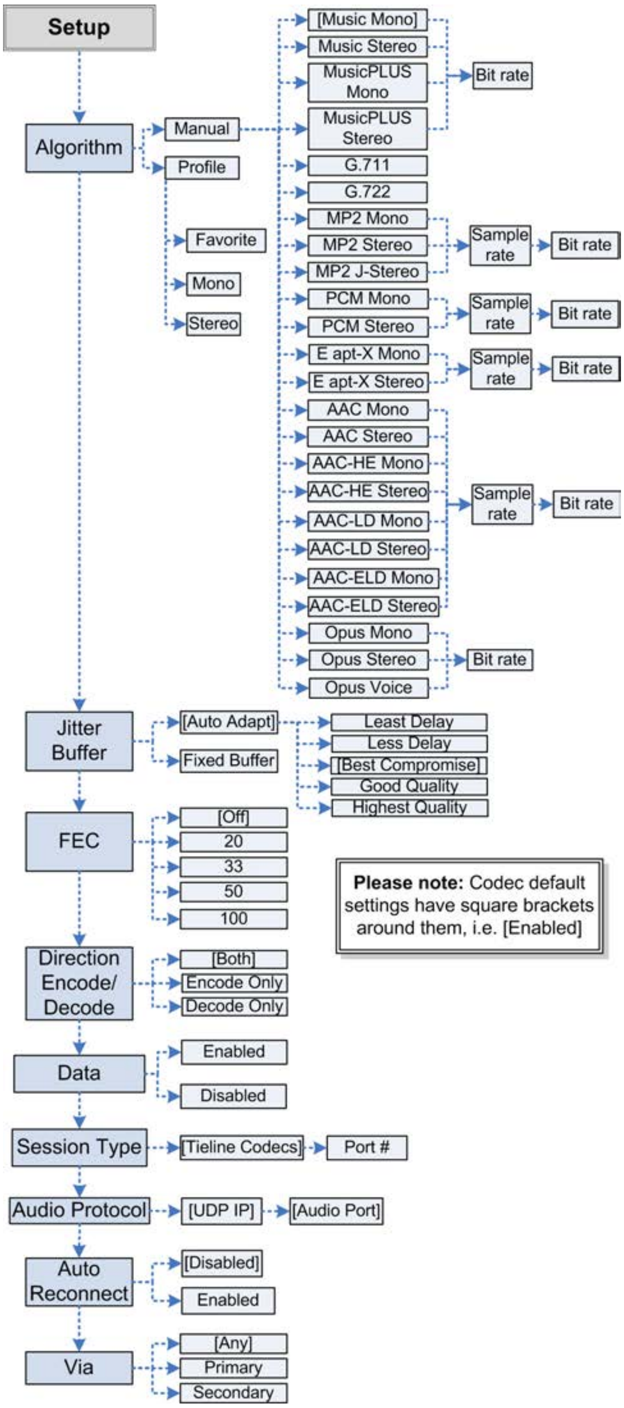
Codec Menu Overview



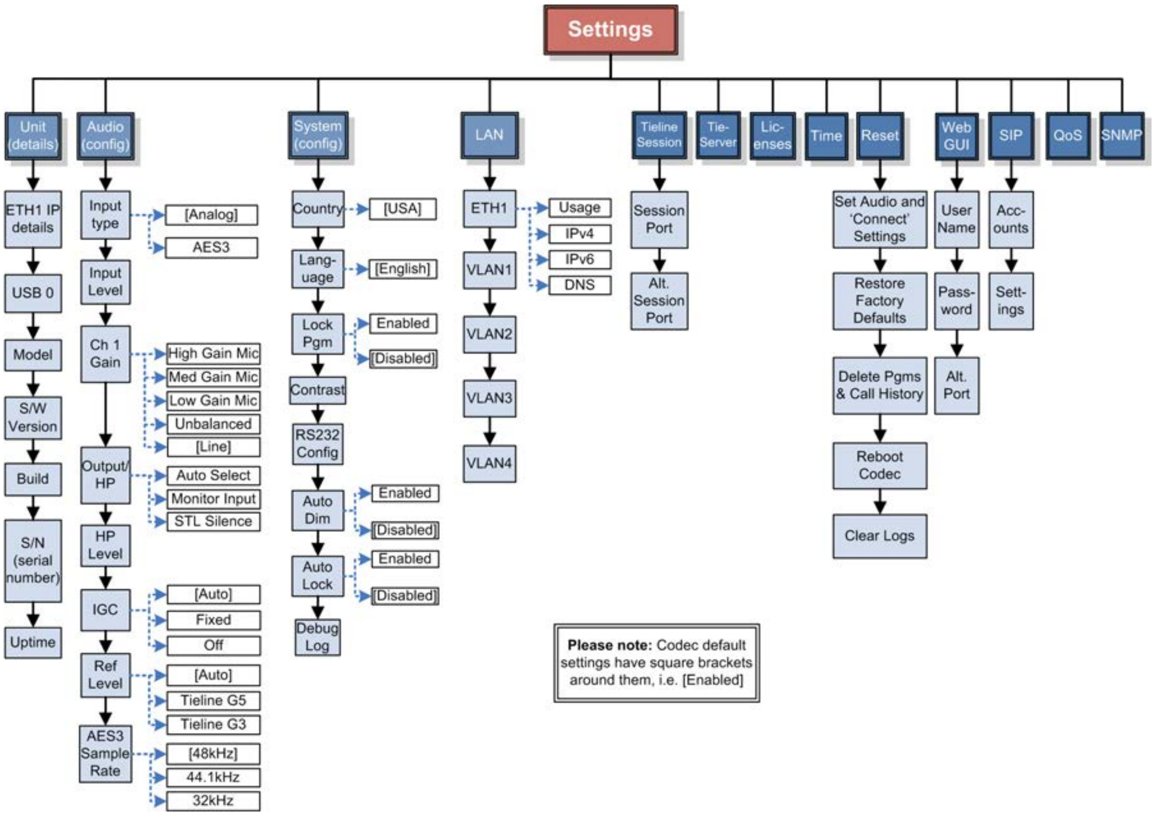
Connect Menu



IP Setup Menu Navigation



Settings Menu



8 Adjusting Input/Meter Levels

The codec uses dBu to express nominal operating level, headroom and noise floor levels. The PPM meters display input audio by default when the codec is not connected and they then switch to monitor decoded return program audio after making a connection.

Mono and Stereo Audio Capabilities

The codec sends input 1 directly to the left output and input 2 directly to the right output. When sending mono analog audio select a mono algorithm in the **IP Connect Setup** menu and connect audio to input 1 of the codec. Input audio is replicated and sent to both channel 1 and 2 analog XLR outputs in this mode. The AES3 outputs are directly mapped to both the analog and digital inputs, therefore if a mono analog profile is selected, only channel one will have audio on it over AES3. Note: It is not possible to mix channels 1 and 2 into dual mono outputs.

The codec will provide both analog and digital audio out at all times and this is not dependent on whether your audio source is analog or digital. The only point to note is that when you configure a mono analog connection the codec will only send audio on one of the AES3 outputs, but it will send audio on both the left and right channels of the analog outputs.

Adjusting Audio Meter Reference Scale Settings

When connected to stereo sources the top PPM meter is the left channel and the bottom PPM meter is the right channel. The codec is configured by default to automatically connect to other Tieline codecs using the correct audio reference meter scales. The audio reference level settings in the codec are:

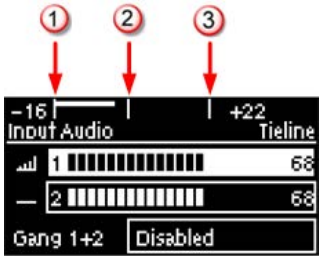
	Reference Setting	Description
1	Auto (default)	When connecting to a Tieline codec with session data enabled the codec will automatically adjust the reference level for G5 and G3 codecs. When connecting to a non-Tieline codec, or a Tieline codec without session data enabled, the codec will use the **Tieline G5** setting.
2	Tieline G5	The audio reference scale is -16dBu to +22dBu
3	Tieline G3	The audio reference scale is -11dBu and +18dBu

To configure this setting manually:

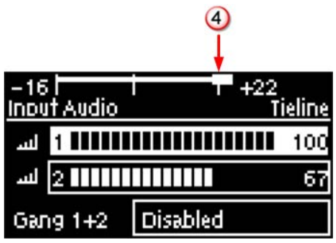
1. Press the **SETTINGS** button.
2. Navigate to **Audio** and press .
3. Navigate to **Ref Level** and press .
4. Select the preferred setting and press .

Audio Metering when Connecting to Tieline G5 Codecs

The **Tieline G5** audio reference scale displayed on the codec screen is -16dBu to +22dBu when you connect to a codec in Tieline's Merlin, Genie or Bridge-IT IP codec families. Set audio levels so that audio peaks average at the nominal 0vu point. This represents a program level of +4 dBu leaving the codec. Audio peaks can safely reach +22 dBu without clipping, providing 18dBu of headroom from the nominal 0vu point. The default PPM audio meter indications are as follows.

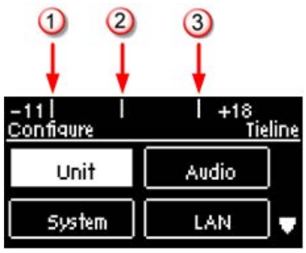


	Features	Description
1	-16dBu	PPM meter low point
2	+4dBu	Nominal 0vu reference level at +4dBu
3	+20dBu	+20dBu indication that should not be exceeded to prevent clipping at +22dBu
4	PPM meter in clip	PPM indication displays a solid section at the right-hand end when audio is in danger of clipping



Audio Metering when Connecting to Tieline G3 Codecs

New generation Genie, Merlin and Bridge-IT IP codecs have more audio headroom than Tieline G3 audio codecs, therefore metering needs to be adjusted when connecting to a Commander or i-Mix G3 codec. The G3 metering scale is between -11dBu and +18dBu and audio levels should average around the nominal 0vu point. Audio peaks should not exceed +16dbu as indicated on the PPM meter.



	Features	Description
1	-11dBu	PPM meter low point
2	+4dBu	Nominal 0vu reference level at +4dBu
3	+16dBu	+16 indication where audio will clip/distort

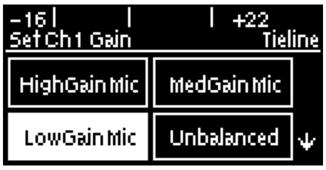
Channel 1 Mic/Line Level Audio Adjustment

The default input level setting in the codec for channel 1 is line level. To adjust this setting for a mic-level or unbalanced source:

1. Press the **SETTINGS** button.
2. Navigate to **Audio** and press .

3. Ensure **Input Type** is set to **Analog.**

4. Use the arrow-down ▽ button to highlight and select the **Ch 1 Gain** setting and press the button.
5. Use the navigation buttons to select the appropriate gain setting and press the button to save the setting.



Important Note: 15 volt phantom power is not supplied to input 1 by default. To turn this on:

1. Select **Settings > Audio** and use the arrow-down ▽ button to highlight the **Phantom** setting.
2. Press the button to toggle between **Enabled** and **Disabled**.

Channel 2 is a line input only and gain can only be adjusted using the **Input Audio** screen on the codec.

Quick Adjustment of Input Levels

1. Press the button and the right arrow button to open the **Input Audio** level adjustment screen.
2. Press on the numeric keypad to toggle channel 1 on and off and press to toggle channel 2 on and off.
3. Use the up △ and down ▽ arrow buttons to navigate to the channel you want to adjust. Note: A channel is highlighted when selected.
4. Use the left and right arrow buttons to adjust the input levels up or down.
5. Press the **RETURN** button to exit the screen.

	Input Audio Features	Description
1	Channel On Symbol	Symbol indicates a channel is turned on
2	Channel Off Symbol	Symbol indicates a channel is turned off
3	Input 1 Level Control	Ch 1 level indication with percentage of gain indicated, i.e. **68.**
4	Input 2 Level Control	Ch 1 level indication with percentage of gain indicated, i.e. **68**
5	Ch1/2 Gang Indication	Indicates whether ganging is enabled or disabled

Intelligent Gain Control (IGC)

When the broadcast action really starts to heat up, the codec's inbuilt DSP limiter automatically takes care of any instantaneous audio peaks that occur in demanding broadcast situations. IGC (Intelligent Gain Control) is enabled by default and is activated at +20 dBu (G5 audio scale) and +14dBu (G3 audio scale) to prevent audio clipping. IGC automatically adjusts high audio input levels downwards until they are acceptable. If IGC auto level recovery (IGC Level) is not enabled, the input level will remain at the adjusted point until the input gain is manually adjusted again by the user. If IGC is active in the codec it is indicated in the PPM meter section. To adjust this setting:

1. Press the **SETTINGS** button.
2. Navigate to **Audio** and press .
3. Navigate to **IGC** and press to toggle between **Enabled** and **Disabled**.

IGC Auto Level Recovery

IGC Level works with **IGC** to detect when incoming audio levels have reduced sufficiently. There are two settings; **Auto** and **Fixed**.

If the **IGC Level** setting is **Auto** then the codec will return input levels to the gain setting prior to **IGC** being activated. The codec takes just 250 milliseconds to detect audio levels have returned to normal (after **IGC Level** has been initiated) and will then return the levels to the previous setting within half a second. This response is linear.

If the setting is **Fixed** then audio levels will remain lower and not return to the original setting. To adjust this setting:

1. Press the **SETTINGS** button.
2. Navigate to **Audio** and press .
3. Navigate to **IGC Level** and press to toggle between **Auto** and **Fixed**.

Ganging Audio Channels

Ganging allows you to adjust the audio level of both inputs simultaneously.

1. Press the button and the right arrow button to open the **Input Audio** level adjustment screen.
2. Use the up and down arrow buttons to navigate to and select **Gang 1 + 2 Enabled** or **Disabled.**
3. Press the button to select **Enabled**.
4. Use the up and down arrow buttons to highlight and select the audio channels.
5. Use the left and right arrow buttons to adjust the levels for both inputs up or down simultaneously.
6. Press the **RETURN** button to exit the screen.

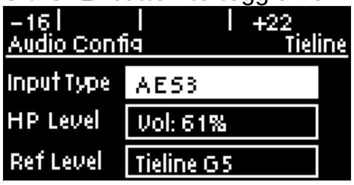
When channels 1 and 2 are ganged together:

- Both channels highlight together when selected.
- You can adjust the audio of both channels simultaneously.
- The gain setting for both channels is automatically set to match the gain level of the lowest of the two channels when ganging is configured.
- If one channel is turned on when ganging is configured then the other one will be turned on automatically.

9 Configuring AES3 Audio

If your input source is AES3 (AES/EBU) format use the **IN1/AES3 IN** input on the rear panel of the codec. This is a balanced 110 ohm female XLR input and can operate effectively over distances of up to 100 meters. The input accepts both mono and stereo AES3 signals as only a single 3 pin XLR input or output is required for sending and receiving two channels of AES3 data. To configure the codec to accept AES3 data signals:

1. Press the **SETTINGS** button.
2. Navigate to **Audio** and press .
3. Select **Input Type** and press the button to toggle from **Analog** to **AES/EBU**.



The 3 pin male XLR AES3 output on the rear panel is labeled **AES3 OUT**. It is capable of sending both mono and stereo AES3 signals.

Important Note: Input levels are set at 100% automatically for AES3 connections. If you switch back to the analog input setting after selecting AES3, the previous analog settings will be recovered.

AES/EBU Sample Rate Conversion

The codec implements an Asynchronous Sample Rate Converter (ASRC) to convert the sample rate of the AES3 input to the sample rate set in the codec. The codec sample rate is determined by the selected algorithm. For example, if you select the Music algorithm, the sample rate will be set to 32kHz when the codec is connected.

The codec accepts AES3 input sample rates of 32kHz, 44.1kHz or 48kHz. The output sample rate is determined by the algorithm used when connected.

AES3 Audio Out & SD Card Sample Rate

1. When you are not connected, you can adjust the AES3 output sample rate manually in the **Audio** menu via the **AES3 SR** setting.
2. If file playback occurs before a connection is initiated, the AES3 output sample rate will match the audio file sample rate.
3. For best performance, the SD card file sample rate should match the **AES3 SR** setting in the codec and the algorithm sample rate. If they don't match, the codec will re-sample the SD card file audio to match the connection sample rate, and this will be used by the AES3 output. For example, if you are streaming audio using Tieline Music (32kHz sampling), file playback will be re-sampled to 32kHz if the SD card file is 44.1kHz or 48kHz. In this example the AES3 output sample rate will be 32kHz.

Adjusting the Codec Output Sample Rate

As there is no external reference clock for the codec it is necessary to set the output sample rate of the codec when you are not connected. The **AES3 SR** setting in the **Audio** menu will configure audio outputs and audio monitoring, as well as SD card playback at this sample rate. The default setting is 48kHz sampling:

1. Press the **SETTINGS** button.
2. Navigate to **Audio** and press .
3. Navigate to **AES3 SR** and press .
4. Select a preferred sample rate then press .

Tieline normally recommends selecting **STL Silence Mode** for audio monitoring if using AES3. If using **Auto Select** make sure the algorithm sample rate and the **AES3 SR** sample rate setting are the same.

10 Headphone/Output Monitoring

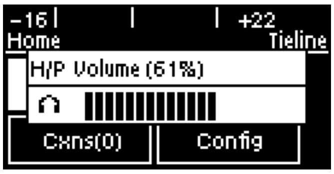
The 6.35mm (1/4") stereo headphone output on the codec can be used for monitoring audio inputs 1 and 2 and return link audio.

Important Note: When the codec makes a connection it will monitor the decoded return audio link by default.

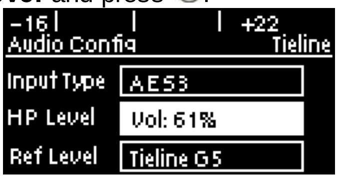
Adjusting Headphone Output Levels

When using analog or digital inputs you will see input audio on the PPMs and hear it in the headphones.

1. Press and hold the button and then press the right arrow button to display the **H/P Volume** adjustment screen.
2. Use the left or right navigation buttons to adjust the volume levels up or down. The screen displays level adjustments in real-time.
3. Press when you have finished.



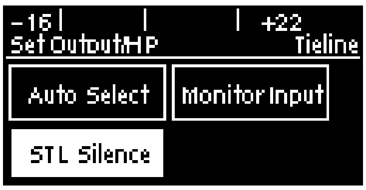
Headphone levels can also be adjusted by navigating to **SETTINGS** **> Audio** and using the down button to navigate to **HP Level** and press .



Output/Headphone Monitoring Settings

There are three **Output/HP** monitoring configurations available in the codec. The default setting is **Auto Select** and to adjust this setting:

1. Press the **SETTINGS** button.
2. Navigate to **Audio** and press .
3. Navigate to **Output/HP** and press to select either **Monitor Input** or **STL Silence**.



The table below displays how audio is routed to the codec outputs/headphones based on the configuration selected and the current codec state (IP streaming or idle). Please note:

- The same audio is always routed to the headphone and XLR outputs.
- Output 1 corresponds to headphone left and output 2 corresponds headphone right.
- The same audio is routed to the analog and AES XLR outputs.
- When inputs are routed to the outputs, input 1 is sent to output 1 and input 2 is sent to output 2.

	Codec Connection State		
Mode	**Idle**	**Streaming (Mono)**	**Streaming (Stereo)**
Auto Select (default)	Inputs	Same decoded audio on all outputs	Channel 1 decoded on output 1, and channel 2 decoded on output 2
STL Silence	No audio	Same decoded audio on all outputs	Channel 1 decoded on output 1, and channel 2 decoded on output 2
Monitor Input	Inputs	Inputs	Inputs

Auto Select Mode

Auto Select is the default **Output/HP** monitoring setting in the codec. Use this setting if you want to monitor the inputs when not connected and return audio when connected.

Monitor Input Mode

Select **Monitor Input** to configure the codec to always monitor input audio. This may be useful if an announcer wants to monitor their own voice and not return audio when connected.

STL Silence Mode

1. In **STL Silence** mode input audio is not monitored on the PPMs or via the headphone output before the codec is connected. It is necessary to check input audio levels using **Auto Select** or **Monitor Input** modes prior to connecting.
2. When the codec connects in **STL Silence** mode it automatically monitors decoded incoming audio.
3. If the connection is lost for any reason then silence is enabled, ensuring input audio cannot be misconstrued as return program audio for STL connections.
4. If the connection is subsequently restored the codec will again monitor decoded incoming audio.

11 Language Selection

English is the default language in the codec. To adjust this setting:

1. Press the **SETTINGS** button.
2. Navigate to **System** and press .
3. Use the navigation buttons to select **Language** and press .
4. Select a language and press .

12 About Program Dialing

What Defines a Program?

The codec uses the concept of 'program' dialing to connect using peer-to-peer, multicast or multi-unicast connections. A **Program** configures a Tieline codec to send or receive an **Audio Stream**. The attributes of each audio stream and associated connections are embodied within a program when it is created, including the configuration, dialing and answering parameters. Essentially a program is like a connection profile with:

- A Program Name.
- IP address dialing details for up to 6 connection end-points or a multicast IP address.
- Specific connection profile details pertaining to algorithm, FEC, jitter buffer and bit-rate settings etc.

Custom programs allow you to store connection settings for a range of peer-to-peer, multicast and multi-unicast connections and retrieve or edit them easily at the touch of a button. Simple peer-to-peer or multicast profiles can be created using the codec front panel, whereas multi-unicasts must be created with the **Programs panel** in the Toolbox web-GUI.

Using Programs to Dial between Two Tieline Codecs

Tieline Bridge-IT and Bridge-IT XTRA codecs operate similarly to Tieline G3 codecs. By default, Tieline codecs send proprietary session data when connecting to each other in order to establish, manage and terminate connections. When a connection between two codecs is established:

1. The dialing codec sends information about how the codec receiving the call should be configured.
2. Once the codec receiving session data from the dialing codec has received information successfully, it sends an acknowledgement to the dialing codec and streaming can commence.

If you configure a stereo program on the dialing codec using a particular algorithm and bit rate etc., these settings will be configured on the answering codec when it connects. It is also possible to lock a loaded program in a codec to ensure the currently loaded program cannot be unloaded by a codec dialing in with different program settings.

Creating Programs

Simple peer-to-peer (point-to-point) programs can be created using the codec front panel. The Toolbox web-GUI contains a **Programs panel** with a wizard for configuring program settings and backup connections. Edit settings easily at the touch of a button and use existing programs as templates for creating other programs.

Mono and Stereo Peer-to-Peer Programs

New peer-to-peer programs can be created using the codec front panel keypad (see 10 Quick Steps to Connect). If you know the IP address of the codec you want to dial then all you need to do is enter this into the codec, choose your preferred connection settings and then press **CONNECT** .

Front panel configured programs are automatically saved as **Recent Programs** which retain all the audio stream dialing and configuration information. These **Recent Programs** are displayed when you press the **CONNECT**  button from within any menu except the **IP Mode** or **SIP Mode** screens, or the **Connect IP** or **Connect SIP** screens.

Ensure you configure all the correct connection settings when using the codec front panel, because these are stored as part of the program's profile when you first connect. They cannot be adjusted afterwards without using the editing features in the **Program panel** within the Toolbox web-GUI.

Important Note: When configuring a connection use the **Save** function in the **Connect IP** and **Connect SIP** screens to save programs permanently to the codec's **Programs** menu. Otherwise they are stored to the **Recent Programs** list and will be overwritten after several calls have been made.

Multi-unicast Programs:

Multi-unicast programs can contain dialing information for up to 6 connection end-points. They are useful for distributing audio to several studios and can be created using the **Programs panel** in the Toolbox web-GUI. (See Configuring Multi-Unicast Programs for configuration details). Once multi-unicast connections have been created they can be dialed via the codec keypad without using the web-GUI.

Multicast Programs:

Multicasts can be used to broadcast to 'subscribers' who wish to connect to a multicast stream. Multicast server transmissions are sent using a dedicated IP multicast address that looks similar to a regular IP address and multicast (client) subscribers request transmissions from this address. New programs can be created using either the codec front panel or the **Programs panel** in the Toolbox web-GUI. (For web-GUI configuration see Configuring a Multicast Server Program or Configuring a Multicast Client Program).

13 Getting Connected Quickly

Preparing to Connect

Before attempting a new connection please connect and adjust the following:

1. Attach the supplied 12 volt power supply to the codec.
2. Attach an RJ45 Ethernet cable to the **LAN** port on the rear panel of the codec.
3. Attach headphones to the 6.35mm (1/4") headphone jack on the rear panel of the codec.
4. Check that the correct country is selected in the codec.
   i. Press the **SETTINGS** button.
   ii. Navigate to **System** and press the button.
   iii. Navigate to **Country** and press the button.
   iv. Use the navigation buttons to select your country of operation.
5. Make sure you have the IP address of the codec you are dialing, or have used the Toolbox web-GUI to load the programs you will be using to dial onto the codec. (see Configuring IP Addresses).

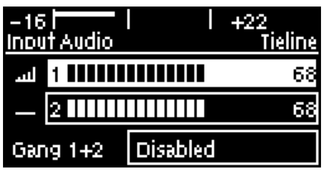
13.1 10 Quick Steps to Connect Bridge-IT

Important Notes:

- The following procedure will create a custom peer-to-peer connection program using the codec front panel keypad and navigation buttons. It instructs how to connect your codec over IP for the very first time without using the Toolbox web-GUI and your computer for configuration.
- See the Java Toolbox Web-GUI Introduction for details on configuring connections remotely via a computer. Creation of programs is not currently supported in the HTML5 Toolbox Web-GUI.
- See Installing the Codec at the Studio for valuable information about installing your codec, negotiating firewalls and port forwarding.
- See Tips for Creating Reliable IP Connections for a range of IP information to assist with setting up IP services for your codecs.
- See Testing IP Network Connections to learn how you can test and verify the reliability of your IP connection.

1. Press the button and right navigation button to open the **Input Audio Level** adjustment screen and adjust audio levels.
   - Press on the numeric **KEYPAD** to toggle channel 1 on and off and press to toggle channel 2 on and off.
   - Use the up and down navigation buttons to select **Gang 1 + 2** and press the button to toggle ganging on/off.
   - Use the up and down navigation buttons to select a single channel, or ganged channels. Note: A channel is highlighted when selected.
   - Use the left and right navigation buttons to adjust the input levels up or down.

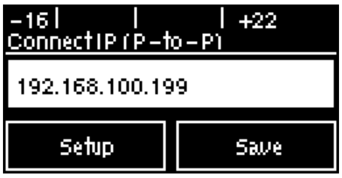
Important Note: 15 volt phantom power is not supplied to input 1 by default. To adjust this setting select **Settings**, then **Audio** and then **Phantom**. Press the **OK** button to toggle between **Disabled** and **Enabled**.



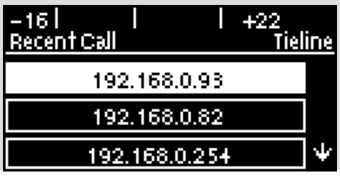
2. Press the **HOME** button to return to the **Home** screen, select **Connect** and press the button, then select **IP** and press the button.

3. Select your preferred **IP Session** mode. In this **Peer-to-Peer** connection example we have selected **Tieline**, which uses Tieline session data, then press the button. Note: Select **SIP** or **Sessionless** if these connections are required.

4. Use the numeric **KEYPAD** to enter the IP address of the codec you want to dial, using the or buttons to enter the periods in the IP address. Use the **RETURN** button to delete numbers already entered. Next, press the down navigation button to select **Setup** and press .



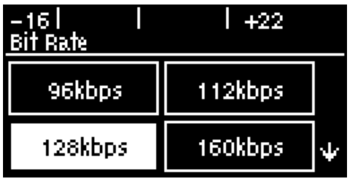
Important Note: The codec remembers recent IP addresses just like a cell-phone. To view these addresses just press the **OK** button when you select the **Connect IP** screen. The most recent addresses and programs are listed first and you can use the navigation buttons to scroll up and down. Press the **OK** button to select the address you have highlighted.



4. Press the down navigation button to select **Algor'm** (algorithm) and press .

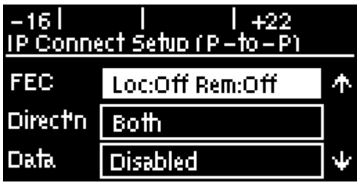
5. Use the navigation buttons to select a preconfigured algorithm profile, or manually enter algorithm settings, then press .

6. If you decide to manually configure the algorithm, use the navigation buttons to select your preferred sample rate (if displayed) and bit rate. Press after selecting each option.



7. Press the down navigation button to select **Jitter** (jitter buffer) and press to select a different automatic jitter buffer setting for your connection. Alternatively you can enter a fixed jitter buffer value in milliseconds (maximum 5000 ms). The default **Auto, Best Compromise** setting is a good starting point for most internet connections.

8. Press the down navigation button to select **FEC** and press to view selection options. Use the navigation buttons to choose the FEC percentage you want to use and press .



9. When configuration is complete press the **RETURN** button to navigate backwards to the **Connect IP** screen that the IP address was entered into.

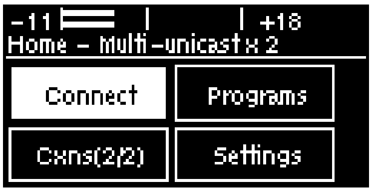
Important Note: At this point you can navigate to **Save** on the **Connect IP** screen and press to save the settings as a custom program for subsequent recall and dialing. Use the numeric **KEYPAD** to give the program a name and press to save the program. A confirmation message is displayed after the program is saved.

10. Press the **CONNECT** button to make a connection. The **Wait Connecting** screen appears during the connection process.

After successfully connecting the codec will display connection details. Use the down navigation button to view connection **Status** and press to view connection statistics for IP packets being sent over the connection. To negotiate higher bit-rates press then **3** on the numeric **KEYPAD**; for lower bit-rates press then **9**.

13.2 Monitoring IP Connections

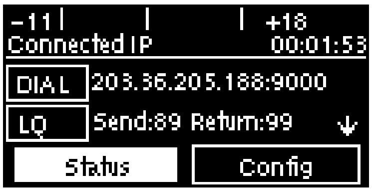
The number of active audio streams and connections is displayed on the **Home** screen via **Cxns**. In the following image the program has two connections configured and both are active, expressed as **2/2**. If only one was active the bracketed number would display **1/2**.



To view more detailed connection information:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Cxns** and press the button.

The **Connected IP** screen displays all connections. The IP address dialed and the **LQ** (link quality) is displayed on the screen and you can use the down navigation button to view the algorithm being used, the connection bit rate, total bytes used and the jitter buffer latency over IP network connections.



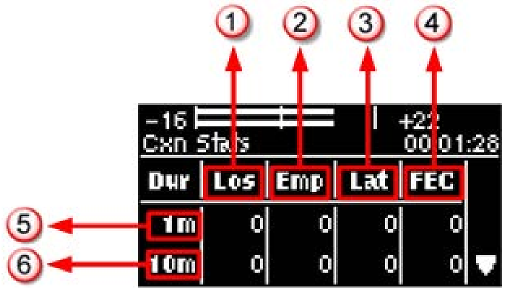
Link Quality (LQ) Readings

Send and return LQ numbers can also help to determine if a problem is occurring at either end of a connection. For example, on an IP connection the **Return** reading represents the audio being downloaded from the network locally (i.e. audio data is being sent by the remote codec). Conversely, the **Send** link quality reading represents the audio data being sent by the local codec (i.e. being downloaded by the remote codec). To ensure a stable connection, try to maintain a reliable reading of 80 or higher for both the **Send** and **Return** LQ reading.

Important Note:
- The **Return** link quality reading is the same as the Local (**L**) setting displayed on a G3 codec.
- The **Send** link quality reading is the same as the Remote (**R**) setting displayed on a G3 codec.

Viewing Connection Statistics

Navigate to **Status** in the **Connected IP** screen and press the button to display the **Cxn Stats** (connection statistics) screen. This displays the performance of the codec in sending IP audio packets across the network. Analysis is historic and assessed over 60 seconds and 10 minutes of connection time.



	Feature	Description
1	Lost Packets	Packets sent that failed to arrive
2	Empty (Jitter Buffer)	Indicates how often the jitter buffer ‘reservoir’ empties causing loss of audio
3	Late Packets	The number of packets that arrive late, i.e. after audio play out
4	FEC Packets	Indicates the number of forward error correction (FEC) packets that have been sent if it is enabled in the codec
5	1 minute	Statistics listed for the last minute of network activity
6	10 minutes	Statistics for the last 10 minutes of network activity

Important Note: If the jitter buffer, FEC or the connection bit rate is changed, we recommend assessing a minute of recent connection performance in preference to 10 minutes of historical connection performance. 10 minutes of data will include connection settings which may no longer be relevant. ‘Packet arrival history’ is cleared when you hang up a connection.

Following is an analysis of possible causes and solutions for the packet analysis statistics displayed on the screen.

Packet Analysis	Displays	Possible Causes	Possible Solutions
Loss	Packets sent and that failed to arrive.	• LAN/WAN congestion • Unreliable ISPs • Unreliable networks • Inferior IP hardware	• Renegotiate connection bit rate downwards • If link quality good add or increase FEC as required • Assess ISP's QOS if very bad performance
Empty	Indicates how often the jitter buffer 'reservoir' empties causing loss of audio.	• High number of packets being lost or arriving late • Signal drop-outs using cell-phone networks • Renegotiation causes the jitter buffer reservoir to empty	• Once could be an anomaly – assess lost & late packets • If many lost packets network is unreliable – renegotiate bit rate and /or FEC down • If many late packets increase jitter buffer
Late	The number of packets that arrive late and after audio play out.	• Network congestion • Jitter Buffer depth is too low	• Auto-jitter buffer will adjust automatically • For manual jitter buffer settings increase jitter buffer depth 50-100 ms & reassess (if only a few packets arrive late over time, audio repairs will be automatic and may not require buffer changes).
FECd	Indicates the number of FEC repaired packets if FEC active.	• Packets have been lost or corrupted over the network	• Assess audio quality & the number of FEC repairs – if many packets are being 'lost' perhaps reduce FEC &/ or renegotiate bit rate down.

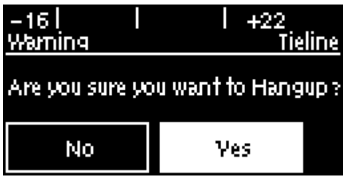
13.3 Load and Dial Custom Programs

Custom programs stored on the codec are simple to load and dial from the codec front panel.

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Programs** and press the button.
3. Use the up and down navigation buttons to select the program you want to connect with, then press the **CONNECT** button to dial.
4. The **Wait Connecting** screen appears during the connection process and then connection details are displayed.

13.4 Disconnecting a Connection

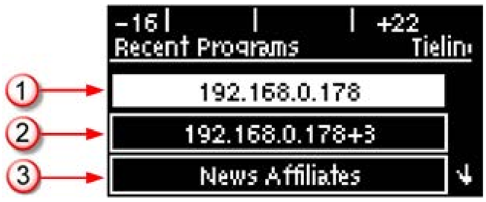
1. Press the red **DISCONNECT** button on the numeric **KEYPAD** at any time to hangup a connection.
2. Use the right navigation button to select **Yes** and press the **DISCONNECT** button or the button to confirm the connection hangup.



13.5 Redialing a Connection

Press the **CONNECT** button from any codec menu to redial previous connections (except menus accessed via the **Connect** > **IP** screen).

Manually dialed connections are saved as programs - retaining all the dialing and configuration information programmed into the codec. A program is identified in the **Recent Program** redial screen using either a previously entered name, or by an IP address (manually dialed connections). A multi-unicast connection will display the first IP address dialed and the number of additional connections dialed.



	Screen Display	Description
1	Manual peer-to-peer connection	Displays the IP address of an IP connection to a single end-point
2	Manual multi-unicast connection (via web-GUI)	Displays the IP address of first codec dialed and the number of additional codec connections
3	Program name (via web-GUI)	Displays the name of a program configured by the Toolbox web-GUI

Redialing Manually from the Connect IP Screen

From the **Home** screen select **Connect > IP > [Select an IP Session mode]** and the codec will assume you want to dial a new ad hoc manual connection. Press the **CONNECT** button when the **Connect IP** (or **Connect SIP**) screen is displayed to retrieve previously dialed IP addresses. Codec settings for any connection dialed from the **IP Connect** (or **Connect SIP**) screen will include the current settings in the **Setup** menu.

13.6 Configuring Auto Reconnect

Auto Reconnect is disabled by default. When enabled the dialing codec attempts to reconnect if data is temporarily lost over an IP connection. To adjust the setting:

1. Press the **HOME** button to return to the **Home** screen, select **Connect,** then select **IP** and press the button.
2. Select the **IP Session** mode you are using to connect.
3. Select **Setup** and press .
4. Navigate to **A/Recon** and press to toggle between **Enabled** and **Disabled**.

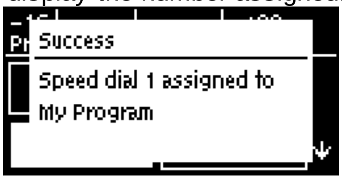
Important Note: When **Auto Reconnect** is enabled, the dialing codec will continue to attempt a connection with the remote codec until **Disconnect** is pressed either on the dialing codec's keypad, or in the web-GUI. This setting should only be configured on the dialing codec.

13.7 Speed Dialing Connections

Assigning Speed Dial Numbers

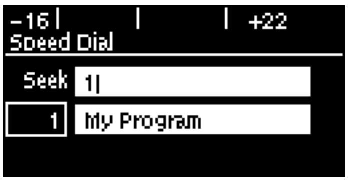
1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Programs** and press the button.
3. Navigate to the program you want to assign a speed dial number, then press the button.
4. Navigate to **Speed Dial** and press the button.

5. Navigate to the speed dial number you want to assign to the selected program and press the button.
6. A confirmation message will display the number assigned.



Speed Dialing

1. Press the **HOME** button to return to the **Home** screen.
2. Use the numeric **KEYPAD** to enter the speed dial number.
3. When the **Speed Dial** screen appears, press the button or the **CONNECT** button to connect.



13.8 Dial/Disconnect Multiple Connections

Multiple Connections within Programs

Multi-unicast programs allow you to simultaneously transmit a mono or stereo audio stream to up to 6 destination codecs. Multi-unicast programs can only be created using the Toolbox web-GUI. There are two ways to simultaneously dial multiple IP audio stream connections multi-unicast programs:

1. Load the program into the codec via the front panel and dial.
2. Connect to the codec using the Toolbox web-GUI and use the **Master panel** to load the program and connect.

Dialing Multiple Connections via the Front Panel

1. Press the **HOME** button to return to the **Home** screen.

2. Use the navigation buttons to select **Programs** and press the button.
3. Use the up and down navigation buttons to select the multi-unicast program you want to connect with, then press the **CONNECT** button to make a connection.
4. The **Wait Connecting** screen appears briefly and then the **Home** screen is displayed.

It is also possible to redial the connection.

Disconnect All Connections

1. Press the red **DISCONNECT** button on the numeric **KEYPAD** at any time to hangup all connections.
2. Use the right navigation button to select **Yes** and press the **DISCONNECT** button or the button to confirm the disconnection.

Disconnecting Individual Multi-unicast Connections

It is only possible to disconnect individual connections via the **Master panel** in the dialing codec's Toolbox web-GUI.

13.9 Creating a Multicast Server Program

Two different types of multicast programs need to be created when multicasting:

- A multicast server program is used by the broadcasting codec to send multicast IP packets to multicast routers on a network.
- A multicast client program is used by codecs to receive multicast IP audio packets.

Prerequisites:

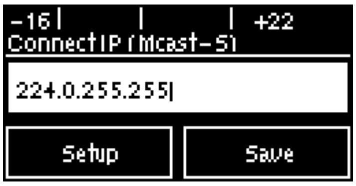
- Bridge-IT firmware v.1.01.00 r4219 or higher.
- ToolBox web-GUI v.1.2.2.3 or higher.
- A multi-unicast license installed in the dialing codec (Note: the multi-unicast license includes multicast server capability).

Important Notes:
- You cannot edit a program when it is currently loaded in the codec.
- Ensure all connection related settings like the port, algorithm, bit rate (etc) match on both multicast server and client programs or they will not be able to join multicast streaming sessions.
- There is no jitter buffer setting in a multicast server program because it is an encode only program and never receives audio packets.
- You can lock a loaded custom program in a codec to ensure the currently loaded program cannot be unloaded by a codec dialing in with a different program type.
- Always dial the multicast server codec connection first before connecting multicast client codecs.
- Multicast client codecs will display return link quality (LQ) only. The **Return** reading represents the audio being downloaded from the network locally. Multicast server codecs do not display LQ readings.
- Forward Error Correction (FEC) is not available for multicast connections.
- It is not possible for a G3 codec to receive multicast IP audio streams.
- To learn more about programs see the section titled "About Program Dialing".
- See Toolbox web-GUI documentation for more detailed information about "Configuring Multicast Server Programs" or "Configuring Multicast Client Programs"

1. Press the **HOME** button to return to the **Home** screen, select **Connect** > **IP > Sessionless** and press the button.

2. Select **M'cast Server** to configure a server codec program.

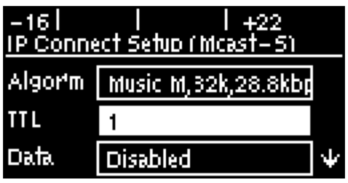
3. Use the **RETURN** button to delete any numbers already entered, then use the numeric **KEYPAD** to enter the multicast IP address you want to dial, using the or buttons to enter the periods in the IP address. Note: The same multicast address and audio port must be used for both the server and client programs. Next, press the down navigation button to select **Setup** and press .



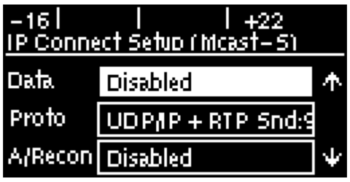
4. Press the down navigation button to select **Algor'm** (algorithm) and press .

5. Use the navigation buttons to select an algorithm profile or manually choose algorithm settings, then press .

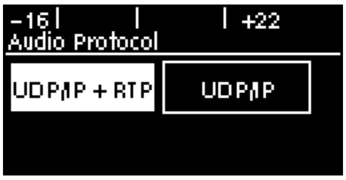
6. Navigate to and select **TTL** and enter the IP Time-to-Live value, then press . Please note: The TTL value you need to use is dependent upon your network infrastructure. Please consult your network administrator if you are unsure about how to configure this setting.



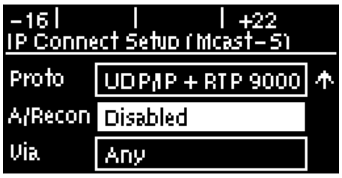
7. Select **Data** to enable or disable data in the audio stream. See RS232 Data Configuration for more information on data connections.



8. Select **Proto** (protocol) to select the audio protocol and adjust the **Remote Audio Port**. Select **UDP/IP +RTP** for RFC-compliant IP streaming. Press to save settings.

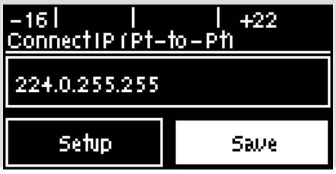


9. If required, enable **A/Recon** (auto reconnect) and use **Via** to specify which IP streaming interface is used to dial this connection, e.g. **Primary** (Ethernet port) or **VLAN** (if configured). Note: By default **Any** will select **Primary**.



10. Press the **RETURN** button when configuration is complete to navigate backwards to the **Connect IP** screen that the multicast IP address was entered into.

Important Note: At this point you can navigate to **Save** on the **Connect IP** screen and press to save the settings as a custom program for recall and dialing. Use the numeric **KEYPAD** to give the program a name and press to save the program. A confirmation message is displayed after the program is saved.



Connecting a Multicast Program

1. After you have created multicast server and client programs on your codecs you can dial multicast connections. First select the multicast server program you want to use on the server codec:

   a. Press the **HOME** button to return to the **Home** screen.
   b. Use the navigation buttons to select **Programs** and press the button.
   c. Use the up and down navigation buttons to select the multicast server program you want to connect with, then press the button to load the program.
   d. Press the **CONNECT** button to make a connection.

2. Select and load the multicast client program on each of the multicast client codecs and dial the multicast IP address to begin receiving multicast audio packets.

Navigate to **Cxns** on the **Home** screen to view a codec's connection **Status**, then press to view connection statistics for IP packets being sent over the connection.

13.10 Creating a Multicast Client Program

Use the procedure which follows to configure a multicast client program and allow the codec to receive multicast IP audio packets.

Prerequisites:

- Bridge-IT firmware v.1.01.00 r4219 or higher.
- ToolBox web-GUI v.1.2.2.3 or higher.
- A multi-unicast license installed in the dialing codec (Note: the Multi-Unicast license includes multicast server capability).

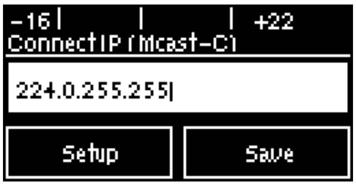
Important Notes:
- You cannot edit a program when it is currently loaded in the codec.
- Ensure all connection related settings like the port, algorithm, bit rate (etc) match on both multicast server and client programs or they will not be able to join multicast streaming sessions.
- The default UDP audio port is 9000 for a multicast client program configured via the codec front panel.
- You can lock a loaded custom program in a codec to ensure the currently loaded program cannot be unloaded by a codec dialing in with a different program type.
- Always dial the multicast server codec connection first before connecting multicast client codecs.
- Multicast client codecs will display return link quality (LQ) only. The **Return** reading represents the audio being downloaded from the network locally.
- Forward Error Correction (FEC) is not available for multicast connections.
- It is not possible to connect to a G3 codec and receive multicast IP audio streams.
- To copy multicast client programs onto multiple codecs see Backup and Restore Functions.
- To learn more about programs see the section titled "About Program Dialing".
- See Toolbox web-GUI documentation for more detailed information about "Configuring Multicast Client Programs"

1. Press the **HOME** button to return to the **Home** screen, select **Connect** > **IP > Sessionless** and press the button.

2. Select **M'cast Client** to configure a client codec program.

3. Use the **RETURN** button to delete any numbers already entered, then use the numeric

KEYPAD to enter the multicast IP address you want to dial, using the or buttons to enter the periods in the IP address. The same multicast address and audio port must be used for both the server and client programs. Next, press the down navigation button to select **Setup** and press .



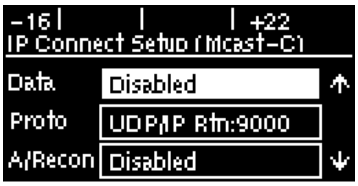
4. Press the down navigation button to select **Algor'm** (algorithm) and press .

5. Use the navigation buttons to select an algorithm profile or manually choose algorithm settings, then press .

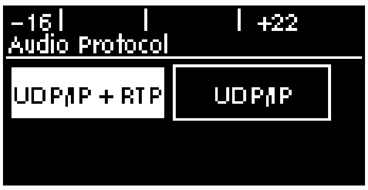
6. Click to configure your preferred Jitter Buffer setting. Select **Auto Adapt** and your preferred automatic jitter setting, or **Fixed Level**. For a fixed buffer setting enter the Jitter **Buffer Depth**, which has a maximum setting of 5000ms, then press .

Important Notes: Automatic or fixed jitter buffer settings can be adjusted on individual client codecs as required. There is no jitter buffer setting on the server codec because it never receives audio packets.

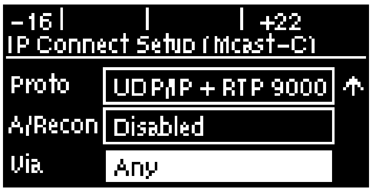
7. Select **Data** to enable or disable data in the audio stream. See RS232 Data Configuration for more information on data connections.



8. Select **Proto** (protocol) to select the audio protocol and adjust the **Local Audio Port**. Select **UDP/IP +RTP** for RFC compliant IP streaming. Press to save settings.

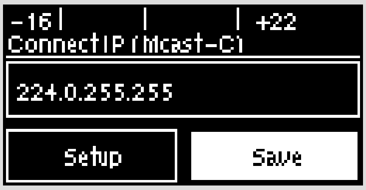


7. If required, enable **A/Recon** (auto reconnect) and navigate to **Via** to specify which IP streaming interface is used to dial this connection, e.g. **Primary** (Ethernet port) or **VLAN** (if configured). Note: By default **Any** will select **Primary**.



8. Press the **RETURN** button when configuration is complete to navigate backwards to the **Connect IP** screen that the multicast IP address was entered into.

Important Note: At this point you can navigate to **Save** on the **Connect IP** screen and press to save the settings as a custom program for subsequent recall and dialing. Use the numeric **KEYPAD** to give the program a name and press to save the program. A confirmation message is displayed after the program is saved.



Connecting a Multicast Client Program

1. After you have created multicast server and client programs on your codecs you can dial multicast connections. First select the multicast server program you want to use on the server codec and dial to connect.

2. Select and load the multicast client program on each of the multicast client codecs and dial the multicast IP address to begin receiving multicast audio packets.

   a. Press the **HOME** button to return to the **Home** screen.
   b. Use the navigation buttons to select **Programs** and press the button.
   c. Use the up and down navigation buttons to select the multicast client program you want to connect with, then press the button to load the program.
   d. Press the **CONNECT** button to make a connection.

Navigate to **Cxns** on the **Home** screen to view a codec's connection **Status**, then press to view connection statistics for IP packets being received over the connection.

13.11 Dialing SIP Peer-to-Peer

Important Notes:
- The codec is fully EBU N/ACIP Tech 3326 compliant when connecting using SIP (Session Initiation Protocol) to other brands of IP codecs.
- Both codecs connecting over SIP need to be configured
- SIP dialing is only supported over peer-to-peer connections, not multi-unicast connections.
- Tieline G3 codecs do not support AAC and will default to MPEG Layer 2 if a Bridge-IT

codec configured for AAC attempts to connect.
- Some Telcos and ISPs may block SIP traffic over UDP port 5060.
- When connecting to a Tieline G3 codec using SIP you need to manually select the G3 audio reference level in the codec. To do this select **SETTINGS** **> Audio > Ref Level > Tieline G3**. In addition, select the following on the G3 codec prior to dialing:
  - Select either a mono or stereo profile
  - Select **[Menu] > [Configuration] > [IP1 Setup] > [Session Type] > [SIP]**
  - Select **[Menu] > [Configuration] > [IP1 Setup] > [Algorithm] > [G711/G722 or MP2]**

Dialing Peer-to-Peer SIP IP Connections

SIP can be used to make direct peer-to-peer calls to different brands of IP codecs with public IP addresses, or between two codecs over a LAN which do not pass through firewalls. Peer-to-peer SIP calls are usually used to connect to other brands of codecs and perform call and session management tasks. Peer-to-peer SIP calls between two codecs are detected automatically and require no special configuration.

To make a peer-to -peer call between codecs we recommend both codecs use public IP addresses:

- Find out the IP address of the remote codec being dialed.
- Program each codec with a compatible algorithm and sample rate etc.
- Dial using **SIP** within the **Connect** menu.
- If the remote codec has a private IP address then it should be configured for port forwarding and should dial the public IP address at the studio (see Configuring TCP/UDP Protocols for more details on port forwarding).

1. To dial peer-to-peer press the **HOME** button to return to the **Home** screen, select **Connect > IP > SIP**.
2. Use the numeric **KEYPAD** to enter the IP address of the codec you want to dial, using the or buttons to enter the periods in the IP address and use the **RETURN** button to delete numbers already entered.
3. Then press the down navigation button to select **Setup** and press to adjust the algorithm, jitter buffer and encode/decode direction if required.
4. Press the **RETURN** button to navigate backwards to the **Connect SIP** screen.
5. Press the **CONNECT** button to make a connection.

13.12 Dialing SIP Addresses

Dialing a SIP Address via the Codec Front Panel

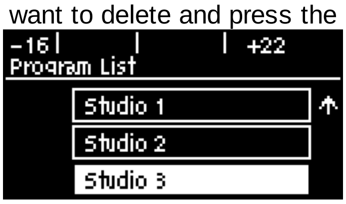
1. Press the **HOME** button to return to the **Home** screen, select **Connect > IP > SIP**.
2. Use the **KEYPAD** to enter any combination of alphabetic and numeric characters in the SIP address of the codec you want to dial. Use the or buttons to enter the periods in the SIP address and use the **RETURN** button to delete any numbers already entered. Alternatively, if you have dialed the SIP address previously, press the **RETURN** button to view the **Recent Call** screen and select the SIP address you want.

3. Press the down navigation button to select **Setup** and press , then adjust the algorithm, jitter buffer, encode/decode direction, port and auto reconnect settings if required.
4. Press the **RETURN** button to navigate backwards to the **Connect SIP** screen and select **Save** to name and save the program.
5. Press the **CONNECT** button to make a connection.

Important Notes:
- See Configuring SIP Settings for instructions on entering SIP account details into the codec using the Toolbox web-GUI. If your codec is registered with same SIP registrar as the destination codec then you only need to enter the SIP user name to dial successfully.
- Each codec should be registered to a different SIP server account to avoid connection conflicts.
- If you don't save the program during configuration, a temporary program is created after you dial the SIP connection for the first time using the codec **KEYPAD**. The temporary program will appear in the recent calls list if you want to redial the program.

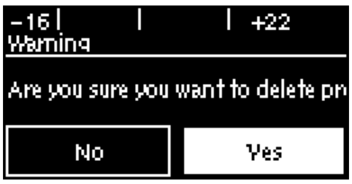
13.13 Deleting Programs

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Programs** and press the button.
3. Navigate to the program you want to delete and press the button.



4. Navigate to **Delete** and press the button.

5. Confirm program deletion and press the button.



13.14 Selecting Algorithm Profiles

The codec has a number of preconfigured mono and stereo dialing profiles available. These can be used to configure the codec quickly with the most popular settings that provide high quality connections using each available algorithm.

If you are unsure about which algorithm to use, see Selecting an Algorithm for more details on each algorithm available in the codec.

1. Press the **HOME** button to return to the **Home** screen.

2. Use the navigation buttons to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select **Tieline** session mode and press the button. Note: algorithm profiles are only available for Tieline session connections.
5. Use the down navigation button to select **Setup** and press the button.
6. Press the button to select **Algor'm**.
7. Use the right navigation button to select **Profile**.
8. Select the profile you want from the **Favorite**, **Mono or Stereo** menus.

	Features	Codec Home Screen Elements
1	Favorite	Displays a list of favorite profiles that have been selected manually within the codec by users
2	Mono	Displays preconfigured mono profiles within the codec
3	Stereo	Displays preconfigured stereo profiles within the codec

Adding a Profile into the Favorite Menu

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select your preferred **IP Session** mode and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Press the button to select **Algor'm**.
7. Use the right navigation button to select **Profile**.
8. Select the profile you want from the **All**, **Mono or Stereo** menus.
9. Press the hatch button to add the profile into the **Favorite** menu.

Profiles that have been added into the **Favorite** menu are identified by the hatch (pound) symbol next to their name after they have been selected.

Deleting a Profile from the Favorite Menu

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select your preferred **IP Session** mode and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Press the button to select **Algor'm**.

7. Use the right navigation button to select **Profile**.
8. Select the profile you want to delete from the **Favorite** menus.
9. Press the hatch button to delete the selected profile from the favorite menu.

Available Profiles

The following profiles are pre-configured in all Bridge-IT codecs. Note: AAC and aptX Enhanced algorithm profiles are not available unless valid licenses have been installed in the codec.

	Profiles			
	Algorithm	**Mono/Stereo**	**Sample Rate (kHz)**	**Bit-rate (Kbps)**
1	AAC	Mono	48	64
2	AAC	Stereo	48	128
3	AAC	Stereo	48	256
4	HE-AAC	Mono	32	16
5	HE-AAC	Stereo	32	32
6	HE-AAC	Stereo	32	48
7	AAC-LD	Mono	32	48
8	AAC-LD	Stereo	32	64
9	AAC-ELD	Mono	32	24
10	AAC-ELDv2	Stereo	32	48
11	Enhanced apt-X	Mono	32 (16 bit)	128
12	Enhanced apt-X	Mono	48 (24 bit)	288
13	Enhanced apt-X	Stereo	32 (16 bit)	256
14	Enhanced apt-X	Stereo	48 (24 bit)	576
15	G.711	Mono	8	64
16	G.722	Mono	16	64
17	MPEG 1 Layer 2	J-Stereo	32	128
18	MPEG 1 Layer 2	J-Stereo	48	192
19	MPEG 1 Layer 2	Mono	24	64
20	MPEG 1 Layer 2	Mono	48	256
21	MPEG 1 Layer 2	Stereo	32	128
22	MPEG 1 Layer 2	Stereo	48	256
23	Music	Mono	32	28.8
24	Music	Mono	32	48
25	Music	Stereo	32	64
26	Music	Stereo	32	96
27	MusicPLUS	Mono	48	48
28	MusicPLUS	Mono	48	96
29	MusicPLUS	Stereo	48	96
30	MusicPLUS	Stereo	48	128
31	MusicPLUS	Stereo	48	192
32	PCM Mono	Mono	48 (16bit)	768
33	PCM Stereo	Stereo	48 (16bit)	1,540

13.15 SDHC Card Backup

SDHC File Backup

The codec features an SD/SDHC card slot for automatic backup to MP2 or MP3 recordings if an IP connection is interrupted. Backup connections are configured using the web-GUI and this is outlined in Configuring Mono or Stereo Peer-to-Peer Programs. USB file backup is automatic and occurs:

1. If encoded audio streaming from a remote codec is lost for a time period predetermined within the web-GUI (default 30 seconds). Note: Loss of audio is measured by whether data packets can be received or decoded. If audio is muted at the remote codec and it continues to send audio packets containing 'silence', the receiving codec will not activate USB file backup.
2. Immediately if a connection to another codec is lost.

Backup will occur according to the silence threshold parameters configured for audio file backup. After SD/SDHC file backup is activated the audio file plays continuously in loop mode until a backup connection is dialed and connects, or the primary connection is restored. The codec **Home** screen indicates failover to the backup SD/SDHC card has occurred by displaying **(F)** in the **Cxns** display. Playback continues during reconnection attempts and ceases when a connection is restored.

Important Notes:
- File playback will occur automatically if the silence threshold parameters are breached; if the codec is not connected for any reason file playback will commence. To stop file playback open the **Master panel** in the web-GUI, click to select the file playback connection, then click **Disconnect**.
- The USB drive can be inserted or removed at any time as long as the codec is not already playing audio in failover mode. Avoid removing the card while audio is playing or it will result in poor audio quality. If it is removed accidentally you must reboot the codec to ensure backup audio will continue to operate reliably.

13.16 SDHC Card File Playback

Playing Audio from the SDHC Card

SDHC card files can be played back using the codec front panel controls.

1. Press the **HOME** button to return to the **Home** screen.
2. Select **Connect,** then select **File** and press the button.
3. Use the navigation buttons to select a file.
4. Press the button or the **CONNECT** button to play the selected file.
5. Press the red **DISCONNECT** button on the numeric **KEYPAD** to stop file playback.

Important Notes for SDHC Card File Playback:
- A FAT32 formatted SDHC Card is required (SD cards may be less reliable and are not recommended).
- Create MP2 or MP3 files using a 32kHz, 44.1kHz or 48kHz sample rate.
- Ensure recordings used are not variable bit rate files.
- SDHC file audio is not sent to codec encoders and cannot be transmitted via an audio stream to another codec.
- File playback audio is sent directly to the codec outputs and therefore IGC is not available. When you create your MP2 or MP3 files ensure the audio levels match the audio reference level of your codec and that peaks average at the correct levels.
- If you create a single file name ensure you add the file extension, e.g. "test.mp3", or the file will not play back.
- If you create a directory name, all the files within the directory will be played back. We

recommend you save all audio files as a playlist and link to this if you want them to play out sequentially. Please note that "M3U" is the playlist file format supported by the codec.

13.17 Lock or Unlock a Program in the Codec

By default Tieline codecs will attempt to answer a call from another codec if possible. For example, if a mono program is loaded in the codec and a stereo incoming call is detected, the codec will adjust and load a compatible answering program.

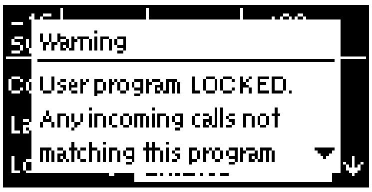
It is also possible to lock a loaded custom program in a codec to ensure a program with your preferred settings is not unloaded when a codec dials in. Incoming calls are generally down or up sampled to accommodate a locked program where possible. Scenarios in which you may wish to lock a program in the codec include:

1. Locking a dialing program to ensure the codec only dials and never answers an incoming call.
2. Locking an answering program to ensure an incoming codec call is not allowed to:
   - Unload the current codec program, e.g. mono or stereo.
   - Change the preferred local site settings like the jitter buffer and FEC configuration etc.

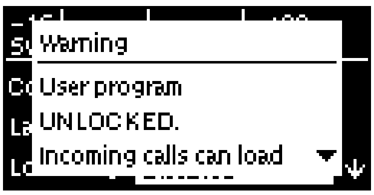
Incoming calls to an answering codec with a locked program can still specify different connection parameters such as algorithm preferences and bit rates via session data.

1. Press the **HOME** button to return to the **Home** screen.
2. Select **Settings** and press .
3. Navigate to **System** and press .
4. Navigate to **Lock Pgm** (lock program) and press to toggle between **Enabled** and **Disabled**.

5. When program lock is **Enabled** a warning message confirms program status.



6. When program lock is **Disabled** a warning message confirms incoming calls may load any supported factory program.



7. Press the **RETURN** button to exit the warning message.

Important Note: It is only possible to lock custom programs in a codec. If **Lock Program** is enabled and you load a new custom program in the codec, **Lock Program** remains enabled and locks the most recently loaded custom program.

13.18 Locking the Front Panel

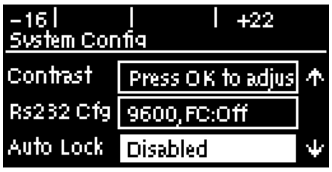
The codec features a front panel lock feature for tamper-proof operation. This feature is disabled by default.

There are two levels of panel lock and each requires a user to enter a PIN to access different features:

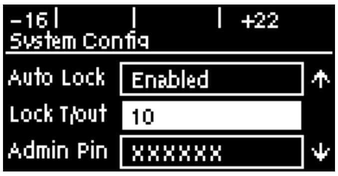
1. **Admin PIN**: Required to change codec connection or configuration settings accessed via the **SETTINGS** button. (Default PIN is: 456789)
2. **User PIN**: Required to use the codec front panel buttons and dial/hangup a connection (Default PIN is: 123456)

Enabling the Front Panel Lock Feature

1. Press the **SETTINGS** button.
2. Navigate to **System** and press .
3. Navigate to **Auto Lock** and press to toggle from **Disabled** to **Enabled**.



4. Navigate down to the panel **Lock Timeout** field and press to enter the desired time-out period in seconds. Note: The time-out period is the time in seconds before the codec front panel is relocked after being used.



5. If you want to change the default **Admin PIN** or **User PIN**, navigate to each in turn and press to enter a new PIN.

14 Connecting to the ToolBox Web-GUI

There are three graphical user interface (GUI) options for configuring Tieline G5 codecs:

1. Java Toolbox Web-GUI: codecs can be fully configured including program creation, dial and hangup, command and control.
2. HTML5 Toolbox Web-GUI: most codec settings can be configured, dial and hangup existing programs only, plus codec command and control.
3. HTML5 Toolbox Quick Connect Web-GUI: designed for simple peer-to-peer connections and non-technical users.

About the Java Toolbox Web-GUI

Codecs can be fully configured using the Java ToolBox Web-GUI and this can be launched using an IP/LAN connection with the codec. Instructions for using the Java web-GUI are contained in the application itself from the **Help panel** and additional information is available at http://www.tieline.com/support/toolbox. The Tieline Java Toolbox Web-GUI application runs on:

- Internet Explorer 6 or greater on Windows® XP, Windows Vista ® and Windows 7 ®.
- Firefox® 3 or greater on Windows® XP, Windows Vista ® and Windows 7 ®, Solaris™ and Linux®.

Java Toolbox Web-GUI Prerequisites

1. To use the Java ToolBox Web-GUI you will need to download the latest version of Java™ by visiting http://www.java.com. The Web-GUI will prompt you to do this if Java is not installed and you attempt to launch the Java ToolBox Web-GUI.
2. After updating to the latest version of Java you need to refresh your browser.

About the HTML5 Toolbox Web-GUI

The HTML5 Toolbox Web-GUI was developed to improve the user experience with G5 codec command and control. With some major web-browsers moving away from Java compatibility, Tieline has delivered an HTML5 configuration option which runs seamlessly on modern browsers.

The HTML5 Toolbox Web-GUI will run on computers and tablets, as well as iOS and Android smartphones, which expands the range of devices engineers can now use for configuration. In addition, many users have previously experienced connectivity issues due to regular Java updates designed to mitigate exposure to security vulnerabilities. By using the HTML5 Toolbox these issues will be avoided.

Most codec settings can be configured using the HTML5 Toolbox Web-GUI, including:

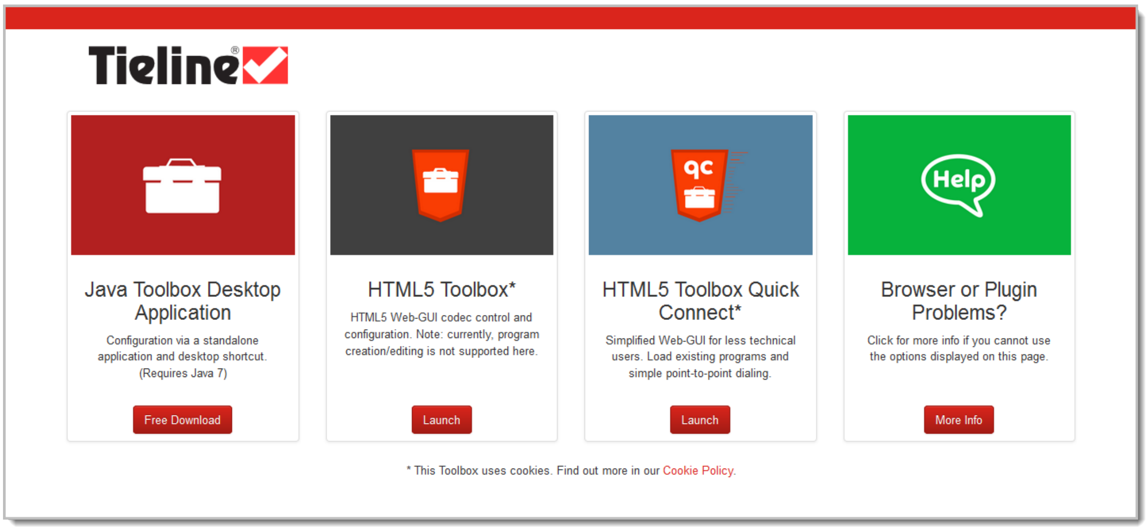
- Dial, monitor and hangup existing programs only (currently program creation wizard is not available).
- Extensive command and control of codec settings.

About the HTML5 Toolbox Quick Connect

The HTML5 Toolbox Quick Connect Web-GUI has a reduced feature-set and allows non-technical users to load existing programs and dial via the Quick Connect panel. Users can dial a simple peer-to-peer connection over IP or SIP.

14.1 Opening the Java or HTML5 Web-GUI & Login

1. Attach an Ethernet cable to the **ETH1** port on the codec.
2. Press the **SETTINGS** button and select **Unit** to display the IP address programmed into your codec.
3. Ensure your PC is connected to the same LAN.
4. Open your web browser and type the IP address of your codec into the address bar of your browser, e.g. **http://192.168.0.xxx** (the last digits are the private address details unique to your codec over a private LAN).
5. Refresh the browser and the Web-GUI landing page will display the various command and control options.



Launching the Java Toolbox Web-GUI

1. Click to launch the **Java ToolBox Desktop Application** (this is recommended in preference to launching the Java Toolbox Browser Applet). Note: When you launch for the first time the application will download and launch the desktop Toolbox application that will allow you to configure your codec. A desktop short-cut will also be created.

Desktop Icon

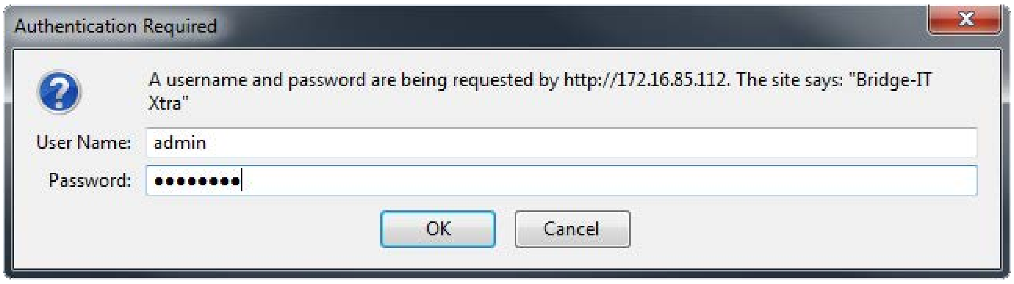
2. When you launch Toolbox an authentication dialog prompts you to enter a password to login. The first time you log in you can enter the default setting "**password**" and click the **OK** button. Tieline highly recommends you click the hyperlink in the login dialog or visit Changing the Default Password to change the password. This will provide better network security to maintain reliability during live broadcasts.

Important Note: If you update Java software or clear the Java cache on your computer you will need to repeat the preceding steps. If you have trouble launching the Web-GUI in a browser, type http://<insert codec IP address>.htm directly in your browser.

Launching the HTML5 Toolbox Web-GUI

1. Click to launch the **HTML5 Toolbox Web-GUI**.

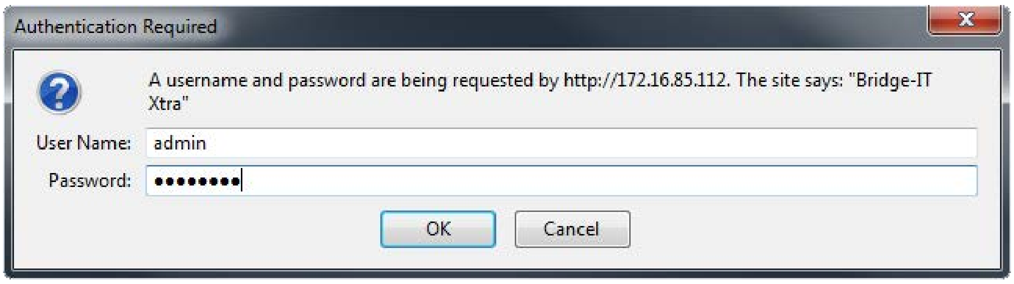
2. When you launch Toolbox for the first time an authentication dialog prompts you to enter the user name "**admin**" and password "**password**" to login, then click the **OK** button. Tieline highly recommends you change the password (see Changing the Default Password). This will provide better network security to maintain reliability during live broadcasts.



Launching the HTML5 Toolbox Quick Connect

1. Click to launch the **HTML5 Toolbox Quick Connect** Web-GUI.

2. When you launch Toolbox for the first time an authentication dialog prompts you to enter the user name "**admin**" and password "**password**" to login, then click the **OK** button. Tieline highly recommends you change the password (see Changing the Default Password). This will provide better network security to maintain reliability during live broadcasts.



Using the Web-GUI over the Internet

If your codec is connected over the internet via a public static IP address it is possible to connect and configure it from any PC which is also connected to the internet.

LAN Troubleshooting

PC LAN Settings

Check the LAN settings on your PC if it is connected to a LAN and is having trouble opening the Toolbox Web-GUI in a web-browser.

1. Open Internet Explorer.
2. Click **Tools > Internet Options > Connections**.
3. Click the **LAN settings** button.
4. If the PC is using a proxy server over the LAN you may need to select the **Bypass proxy server for local addresses** option box.
5. If you still can't connect, click the **Advanced** button in the **LAN Settings** dialog and ask your IT administrator to assist you with entering the IP address of the codec into the **Exceptions** pane of the **Proxy Settings** dialog.

Port Selection

By default port 80 is used by your PC to communicate with the codec and launch the web-GUI. If port 80 cannot be used across your network for some reason, type the IP address of your codec into your browser with a full colon and the port number 8080.

E.g. **192.168.0.176:8080**

It is also possible to specify a different port for connecting the Toolbox web-GUI to your codec.

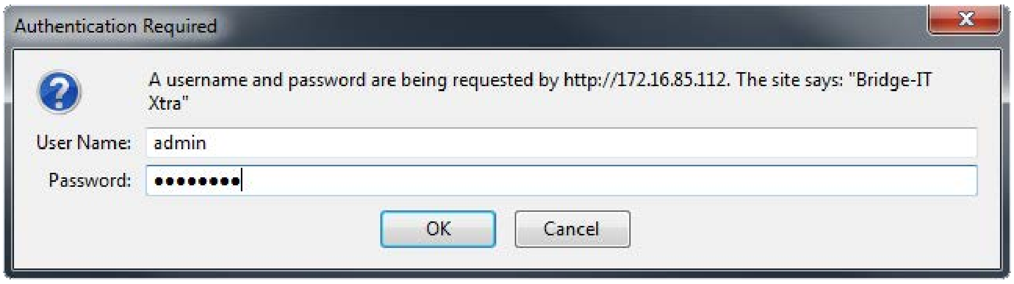
1. Press the **HOME** button on the codec to return to the **Home** screen.
2. Use the navigation buttons to select **Settings** and press the button.
3. Use the navigation button to navigate down to **WebGUI** and press the button.
4. Select **Alt. Port** and press .
5. Use the **KEYPAD** to enter a new port number and press the button to save the new setting.
6. Type the IP address of your codec into your browser with a full colon and then the new port number.

Important Note: Any new port specified must be within the range 2000 to 65535 inclusive.

14.2 Changing the Default Password

The default password for the Toolbox Web-GUI is **password**. Enter this in the authentication dialog to use the Web-GUI initially and then Tieline highly recommends changing the default password to protect your codec from being tampered with during live broadcasts. Note: In the HTML5 Web-GUI authentication dialog it is necessary to enter **admin** as the **User Name**. This field is not visible in the Java Web-GUI authentication dialog.

Toolbox Java Web-GUI Login Dialog



Toolbox HTML5 Web-GUI Login Dialog on a Merlin Codec

Caution: Codecs connected to the internet can be accessed by anyone with knowledge of the codec's public IP address. Setting a strong password protects your equipment from being tampered with during live broadcasts.

Creating a New Password

The authentication login password can be changed at any time using the codec keypad and LCD screen. Note that passwords are case sensitive:

1. Press the **SETTINGS** button.
2. Use the navigation button to select **WebGUI** and press the button.
3. Select **Password** and press .
4. Use the **KEYPAD** to enter a new password and press the button to save the new setting (Note: there is no character limit for passwords).

If you forget the password for the Toolbox web-GUI then you can always press the **SETTINGS** button on the codec and navigate to **WebGUI** to view the current password and change it if required.

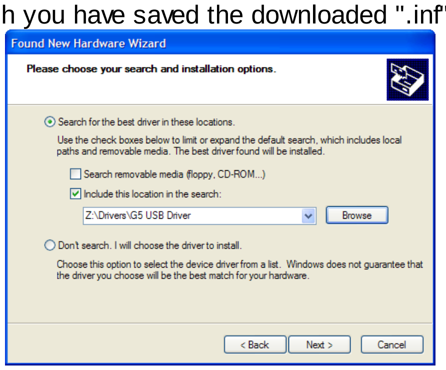
Important Note: The **Username** in the codec menu is permanently set to **admin** and cannot be changed; only the **Password** can be changed.

14.3 Installing USB Drivers

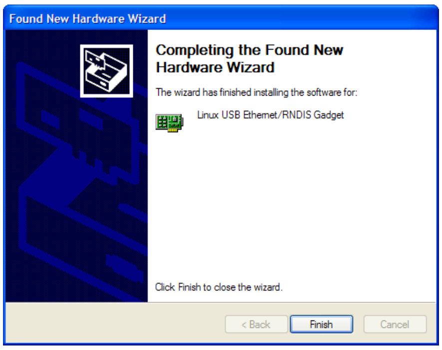
USB drivers need to be installed on your PC in order to connect successfully to the codec using the USB port. To install drivers:

1. Download the zipped USB driver ".inf" file from the Bridge-IT firmware download pages at http://www.tieline.com/Support/Latest-Firmware.
2. Unzip the file and save it to your PC.
3. Connect a USB cable between your PC and the Bridge-IT **USB** port on the rear panel of the codec.
4. The PC should detect that a new device has been attached and launch the **Found New Hardware Wizard**.
5. Select **Install from a list or specific location (Advanced)** and click **Next**.

6. Select the folder in which you have saved the downloaded ".inf" file and click **Next**.

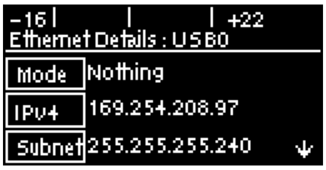


7. When the drivers have been installed click the **Finish** button.



14.4 Launching the GUI over USB

1. Install USB drivers into your PC.
2. Connect a USB cable between your PC and the **USB** port on the rear panel of the codec.
3. From the codec **Home** screen navigate to **Settings > Unit > USB0** and press the OK button to display the **USB** address details configured into your codec.



4. Open your web browser and type the USB address of your codec into the address bar of your browser, i.e. **http://169.254.x.y** (the last 2 blocks of digits are the USB address details unique to your codec).
5. Refresh the browser and the web-GUI should launch automatically.
6. When you open the ToolBox web-GUI an authentication dialog prompts you to enter a password to login. The first time you log in enter the default setting "password" and click the **OK** button. (See Changing the Default Password for instructions on changing the default password to increase your network security).

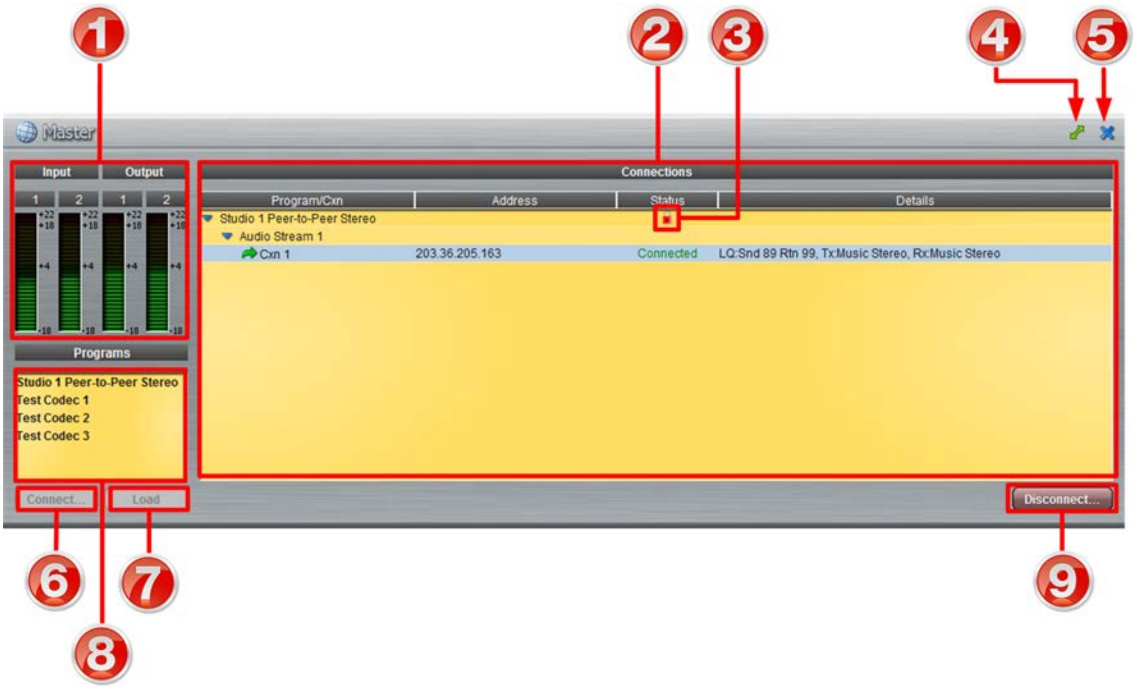
15 Java Toolbox Web-GUI Introduction

Important Note: The default password for the web-GUI is **password**. This has to be entered to initially use the Toolbox web-GUI. For additional security Tieline highly recommends changing the authentication login password using the codec keypad and screen (see Changing the Default Password for more info).

The following sections provide an overview of the different programming panels available within the codec's Toolbox web-GUI. Navigate with the mouse pointer to a symbol at the top of the web-GUI screen and click to open the panel selected. When a panel is opened in the web-GUI, the text below the symbol at the top of the screen is highlighted (see **Master** in the following image).

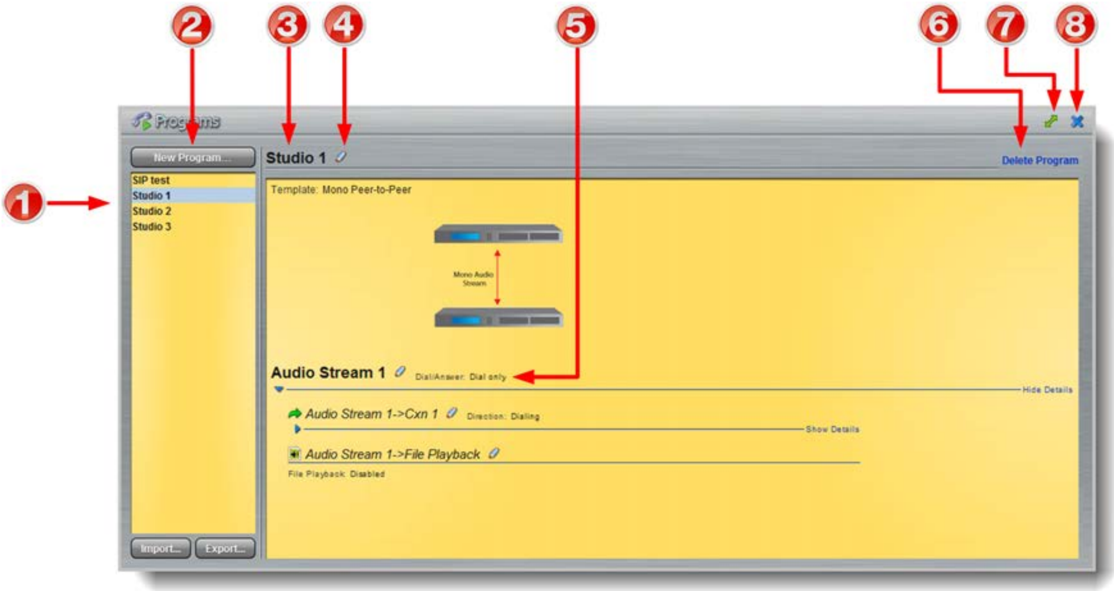
Web-GUI Symbols for Opening Panels

Master Panel to Load Programs and Connect Audio Streams



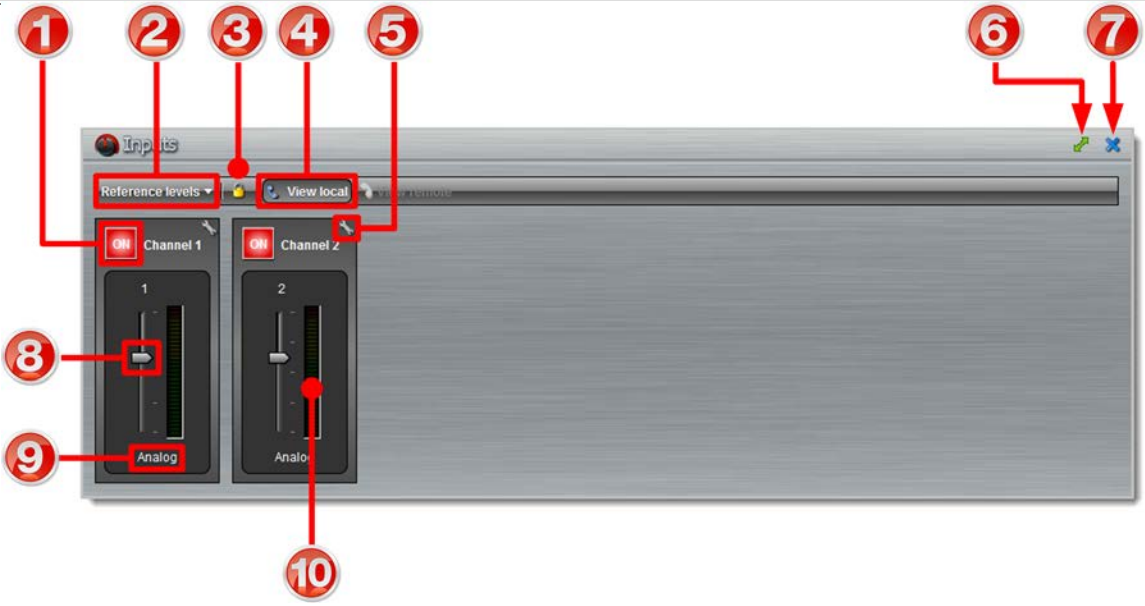
	Feature	Description
1	Input/Output PPMs	4 PPM meters to display audio levels for inputs and outputs
2	**Connections**	Provides a summary of audio stream and connection details audio streams
3	Locked program	Symbol indicates the currently loaded program is locked
4	Maximize/Minimize	Click to maximize a panel to view it in full-screen mode, or click to minimize back to the default panel size
5	**Close** button	Click to close the **Master panel**
6	**Connect** button	Click **Connect** to connect all audio streams configured within the currently selected program in the **Programs** list; this button also loads the program currently selected in the **Programs** list
7	**Load** button	Click to **Load** the codec with the program currently selected in the **Programs** list
8	Programs list	Lists all configured programs which have been added into the codec. Click to select a program before loading or connecting
9	**Disconnect** button	Click to disconnect the currently selected audio stream or a specific connection. Note: this button becomes a **Connect** or **Unload** button when all audio streams are disconnected.

Programs Panel for Connection Configuration



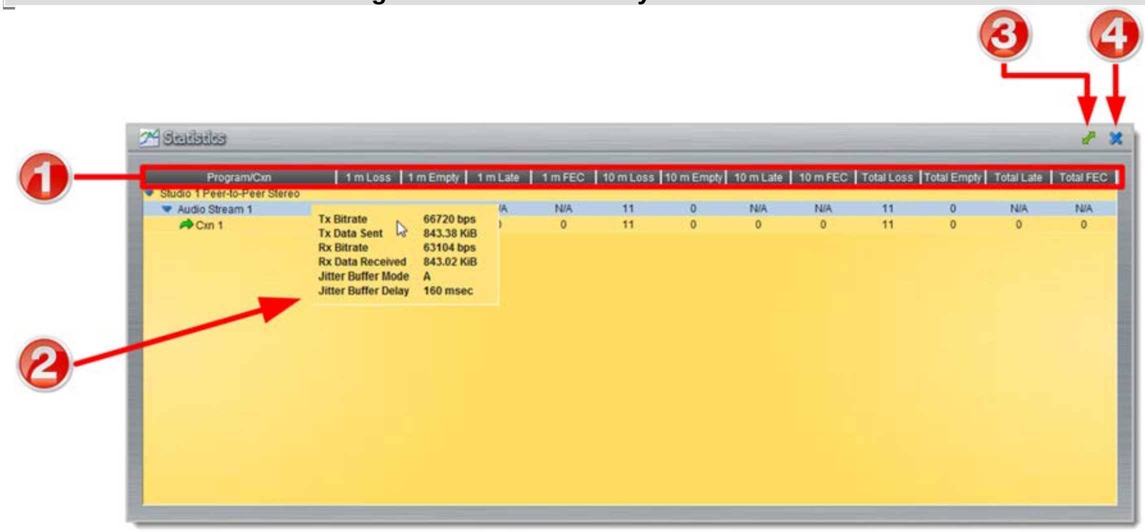
	Feature	Description
1	Programs List	Displays all programs in the codec
2	**New Program** button	Click to add a new program.
3	Program Name	The name of the currently selected program in the panel.
4	**Edit Name**	Click to edit the name of the currently selected program.
5	Audio Stream overview	Click the blue arrows to expand audio stream and connection information; click the **Edit** symbol to adjust program settings. This panel displays the program wizard when creating a new program.
6	**Delete Program**	Click to delete the currently selected program (Note: Ensure the program is not loaded or the delete function will not work).
7	Maximize/Minimize	Click to maximize a panel to view it in full-screen mode, or click to minimize back to the default panel size
8	**Close** button	Click to close the **Connect panel**.

Inputs Panel for Adjusting Input Levels



	Feature	Description
1	Channel **ON/OFF** Buttons	Click to turn each channel **ON** or **OFF**
2	**Reference levels** menu	Click the drop-down arrow to select the codec input reference level (default setting **Auto**)
3	Lock Button	Click to lock all **Input** panel settings (greys out when locked)
4	**View local**	Click to view local codec inputs (default)
5	**Settings** button	Click to adjust input **Name**, **Type**, **IGC** and **Ganging**
6	Maximize/Minimize	Click to maximize a panel to view it in full-screen mode, or click to minimize back to the default panel size
7	**Close** button	Click to close the panel
8	Input Sliders/Faders	Input gain control sliders/faders
9	Analog/AES3 Indication	Indicates whether the codec input is configured for analog or digital audio sources
10	Input PPM meter	Input PPM meter

Statistics Panel for Monitoring Connection Stability

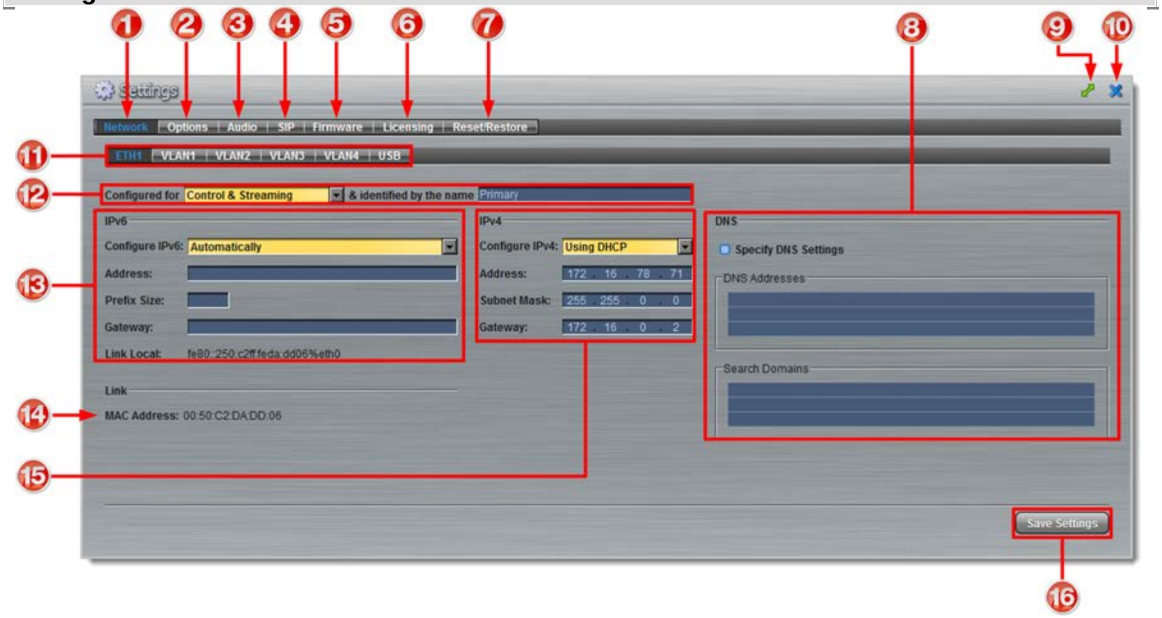


	Feature	Description
1	Headings	Headings for the various packet arrival statistics available
2	Connection Statistics	Right-click to view audio stream bit-rate and jitter buffer statistics
3	Maximize/Minimize	Click to maximize a panel to view it in full-screen mode, or click to minimize back to the default panel size
4	Close button	Click to close the panel

Rules Panel for Creating Relay Activation Rules

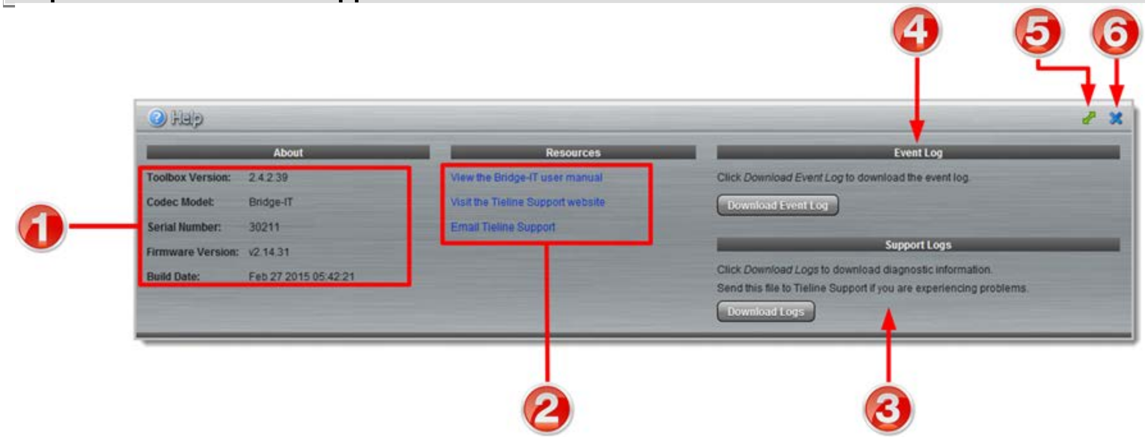
	Rule	Description
1	Connect/Disconnect a program by toggling a relay input	Click to program Connection and Disconnection by toggling an input
2	Connect when an input is switched **ON**; Disconnect when another input is switched **ON**	Click to program Connection and Disconnection after different relay inputs are switched **ON**
3	Synchronise a local relay input with a remote relay output	Click to program a local relay input to Synchronise with the state of a remote relay output
4	Toggle a relay based on connection status	Click to program a relay to toggle based on connection status
5	Maximize/Minimize	Click to maximize a panel to view it in full-screen mode, or click to minimize back to the default panel size
6	**Close** button	Click to close the **Rules** panel

Settings Panel



	Feature	Description
1	**Network tab**	Click to edit or view codec network configuration settings
2	**Options tab**	Click to configure RS232 and QoS data settings, lock a loaded user Program and Adjust Session Port settings
3	**Audio tab**	Click to configure the AES Output Clock sample rate
4	**SIP tab**	Click to edit or view SIP configuration settings
5	**Firmware tab**	Click to view software versions and perform an upgrade
6	**Licensing tab**	Click to select a license file and install it into the codec
7	**Reset/Restore tab**	Click to reset codec default settings and perform backup/restore of codec programs and settings
8	**DNS** Pane	Activate to specify DNS addresses and domains to search.
9	Maximize/Minimize	Click to maximize a panel to view it in full-screen mode, or click to minimize back to the default panel size
10	**Close** button	Click to close the panel
11	Network Interface	Select a network interface for configuration options
12	Network Interface Identifier	Control and streaming configuration options for each network interface, e.g. Ethernet Port or LAN.
13	**IPv6** details	IPv6 addressing details and configuration
14	**MAC Address**	Device MAC address
15	**IPv4** details	IPv4 addressing details and configuration
16	**Save Settings** button	Saves all configuration settings

Help Panel for Product Support



	Feature	Description
1	**About**	Details about the Toolbox web-GUI and codec firmware version, as well as the codec type and codec serial number
2	**Resources**	Links to open the user manual in a new browser, or view support information
3	**Support Logs**	Click to download diagnostic information that can be sent to Tieline support
4	**Event Logs**	Click to download user-viewable event logs
5	Maximize/Minimize	Click to maximize a panel to view it in full-screen mode, or click to minimize back to the default panel size
6	**Close** button	Click to close the **Help panel**

Language Selection

The Toolbox web-GUI offers language support for several languages.

1. Click on the language drop-down menu arrow in the top right-hand corner of the web-GUI page.

2. Select your language of choice.
3. Click to refresh your web-browser and download the user manual for the language selected.

16 Java Toolbox Web-GUI Codec Configuration

The following sections describe how to configure your codec using the Java Toolbox web-GUI.

16.1 Configuring IP Settings

Click the **Settings** symbol to open the **Settings** panel and click the **Network** button to view Ethernet and VLAN interface settings in the web-GUI.

Important Note: For assistance with configuration of IPv4 or IPv6 network connections contact your IT Administrator.

IPv4 versus IPv6

An IP address is a unique address to identify a device on a TCP/IP network. Your codec uses dual IP protocol stacks to allow your codec to work on both IPv4 and IPv6 networks. Your Tieline codec supports both DHCP (default) IP addressing and static IP addresses for dialing IPv4 connection endpoints.

If you want to dial a codec with a public IP address you simply dial the IP address to connect. If you want to dial a codec with a private IP address you need to perform network address translation (NAT). NAT allows a single device, such as a broadband router, to act as an agent between the public internet and a local private LAN. Usually this will be set up at the studio end so you can dial into the studio from the remote codec.

Support for IPv6 connections allows you to use IPv6 infrastructure to connect to other codecs globally.

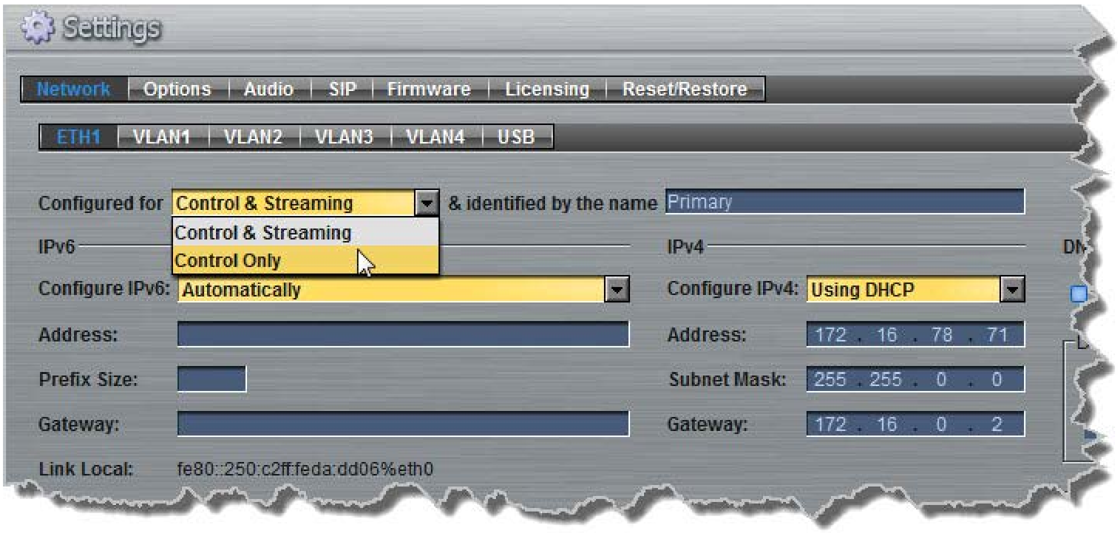
Configuring Ethernet Ports and VLANs

The codec features a physical Ethernet port and up to four additional VLAN interfaces.

VLAN interfaces have features similar to the Ethernet port; however, your network administrator will need to configure VLAN support throughout your network for them to be supported in your codec.

As an example, VLANs can be used to separate codec Control and Streaming functions if required. Ethernet and VLAN interfaces can be configured for:

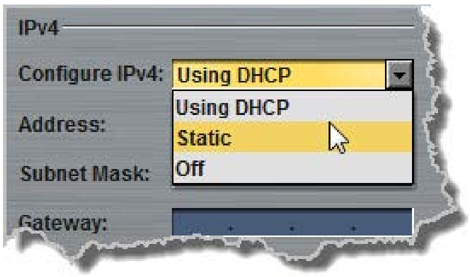
- Control Only: codec control and command only from the Ethernet port.
- Control and Streaming: stream audio and control and command the codec via the Ethernet port.
- Streaming audio: stream audio only from an Ethernet port (**VLANs** only).
- Nothing: Disable the Ethernet port from streaming audio and codec command and control (**VLANs** only).



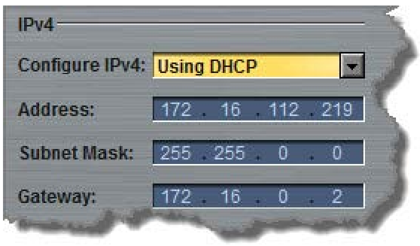
The name entered into the right-hand text box, e.g. **Primary** or **Secondary**, is an interface identifier used when configuring new programs via the **Programs panel**.

IPv4 Address Configuration

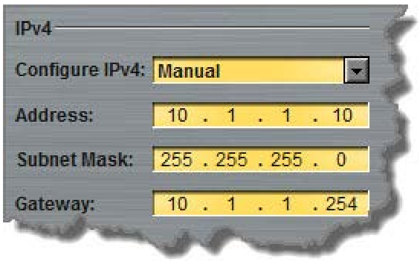
The codec is capable of automatic DHCP address assignment, or manually configured static IPv4 address configuration via the drop-down **Configure IPv4** menu. If you want to ignore IPv4 settings select **Off**.



DHCP IP addresses are automatically assigned and can change each time you connect to your Internet Service Provider or to your own local area network (LAN). By default the codec is configured for DHCP-assigned IP addresses.



Static IP addresses are fixed addresses that are recommended for studio installations, so that IP address dialing remains the same over time for incoming codec connections.



Click **Save Settings** to store all configuration settings.

Note: The **Subnet Mask** is used by the TCP/IP protocol to determine whether a host is on the local subnet or on a remote network. The default **Gateway** is the router linking the codec's subnet to other networks. See your IT administrator for more details.

IPv6 Address Configuration

An IPv6 address is represented by 8 groups of 16-bit hexadecimal values separated by colons (:). The drop-down **Configure IPv6** menu provides three address configuration options:

1. Auto: An address is automatically assigned to the codec when you connect the codec to an IPv6 router. This process is similar to how an IPv4 DHCP address is assigned.
2. Manual: Select to enter static IPv6 address details.
3. Off: Select to ignore IPv6 address details.

Important Note: Select **Off** in the drop-down **Configure IPv6** menu if you are not using IPv6 to connect to another device. This ensures your codec will attempt to connect using IPv4 at all times.

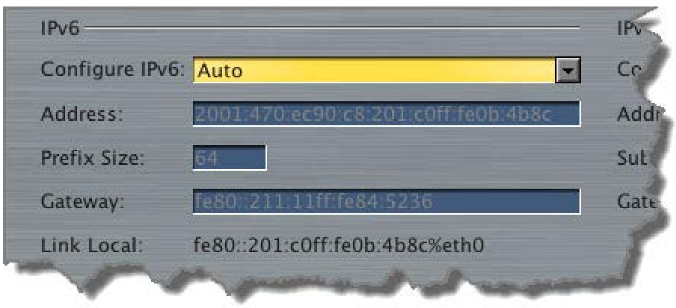
Types of IPv6 Addresses

There are two types of addresses displayed in the IPv6 section:

1. IPv6 address (normally global): A router-allocated IP address with 'global' visibility, details of which are displayed in the **Address, Prefix** and **Gateway** text boxes.
2. Link Local Address: A local address which can only be used to connect to another device directly over a LAN. This address is allocated by the codec internally based on MAC address details.

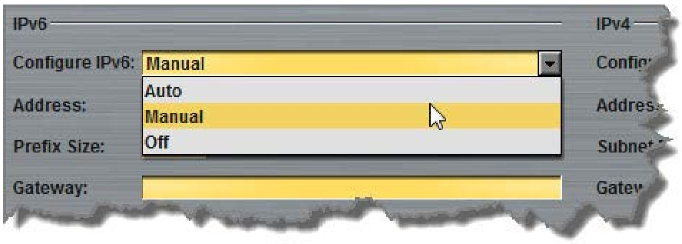
Auto Address Assignment

By default the codec is programmed for connecting to an IPv6 router which automatically allocates IPv6 address details, as displayed in the following example.



Manual IPv6 Address Assignment

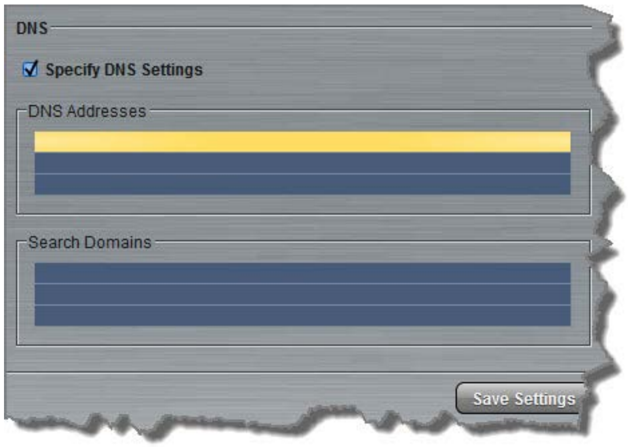
To configure IPv6 address details into the codec manually, select **Manual** and enter details into the **Address, Prefix** and **Gateway** text boxes.



Click **Save Settings** to store all configuration settings.

Specifying DNS Settings

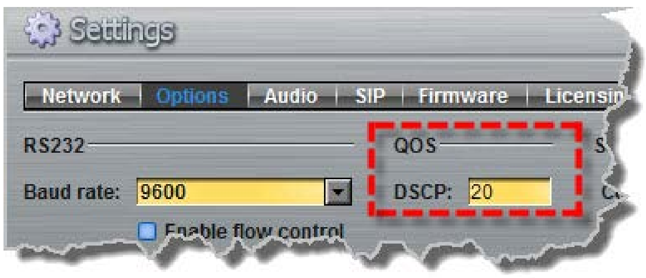
It is possible to specify Domain Name Server (DNS) settings to allow easy look up of codecs within the specified **DNS Addresses** or **Domains**.



The codec can be configured to tag IP data packets sent across a network by entering a value into the Differentiated Services Code Point (**DSCP**) field within the header of data packets transmitted over the network.

Configuring QoS

1. Open the web-GUI and click the **Settings** symbol at the top of the screen to display the **Settings panel**.
2. Click the **Options** button at the top of the **Settings panel**.
3. Click in the **QoS** text box and enter the new value.



4. Click the **Save Settings** button at the bottom of the panel to save the new setting.

Important Note: Check with your IT administrator before changing this setting. By default the codec is programmed for Assured Forwarding and more details about DSCP are available on Wikipedia at http://en.wikipedia.org/wiki/Dscp. For more information on configuring QoS see Configuring QoS for Broadcasts in this manual.

16.2 Configuring Input/Output Settings

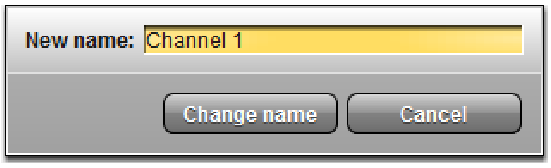
Click the **Inputs** button to view input controls available within the Toolbox web-GUI.

Important Note: 15 volt phantom power can only be supplied on analog input 1; this is disabled by default.

Configuring Input Channel Settings

Renaming Input Channels:

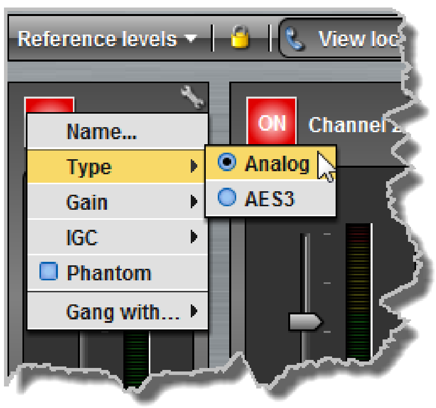
1. Click the **Input Settings** symbol on the input channel you want to rename.
2. Select **Name** and click in the text box to edit or enter a new name.
3. Click **Change Name** to confirm the name change.



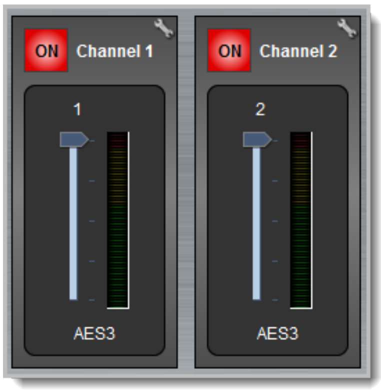
Selecting Analog and Digital Audio Sources:

Codec inputs are configured for analog high-gain mic level audio sources by default.

1. Click the **Input Settings** symbol.
2. Select **Type** and click to select either **Analog** or **AES3**.



3. When you select AES3, the display changes to reflect 100% input levels; slider and input on/off controls are locked on.

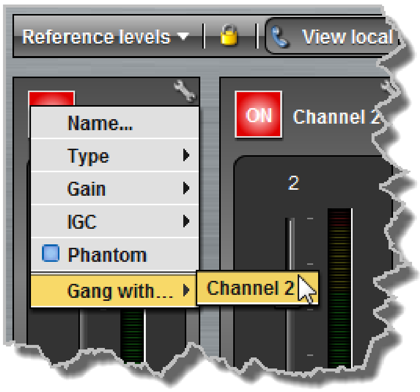


Important Note: Input levels can only be adjusted on analog inputs. See Configuring AES3 Audio for more information about the digital inputs and outputs.

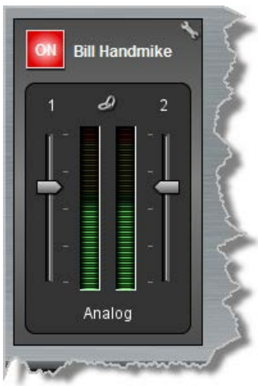
Ganging Channels:

Ganging is useful because it allows you to adjust the audio level of both inputs simultaneously.

1. Click the **Input Settings** symbol on either channel.
2. Select **Gang** and click to either gang or ungang channels.



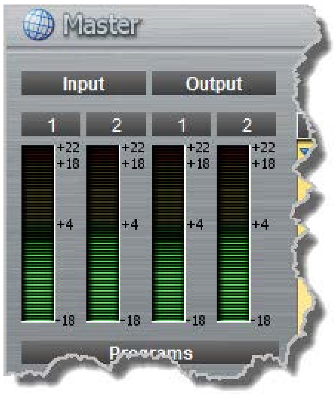
3. When ganged, the two channel sliders move in sync with each other when dragged using a mouse-pointer.



4. Click the **Link** symbol to temporarily disable the ganging function and fine-tune channel audio levels. Click the **Link** symbol again to resume ganging.

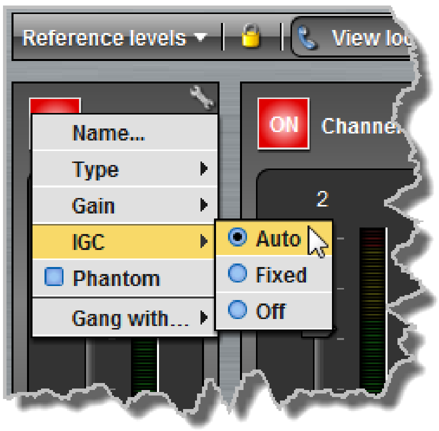
Setting Analog Audio Levels

Audio levels on the **Input panel** should be set to ensure audio peaks average at the first yellow indications on the PPM meters, which represents +4dBu. These levels should also be checked against the **Input PPM Meters** on the **Master panel**. When the codec is not connected the **Master panel** will display the input audio going to the outputs (default setting).



Other Input Controls

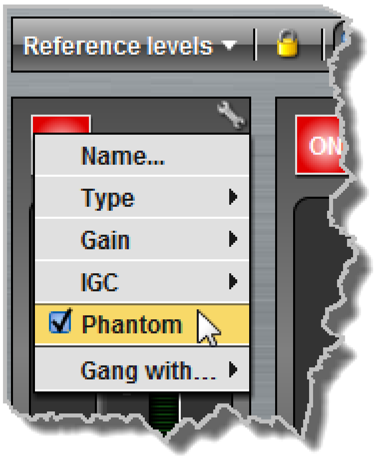
Adjust the **IGC** (Intelligent Gain Control) input settings to **Auto**, **Fixed** or **Off** as required.



Channel 1 Phantom Power

Input 1 can be configured to supply 15 volt phantom power if required; this is disabled by default.

1. Click the **Input Settings** symbol on channel 1.
2. Click to select the **Phantom** check box.



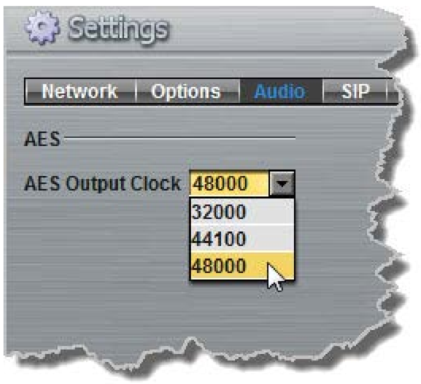
Locking Input Settings

1. Click the **Lock** symbol to lock all **Input panel** settings.
2. When locked, the **Input panel** is greyed out and the lock symbol appears in the bottom-left corner. Note: the lock function does not affect the codec front panel controls.

AES3 Output Sample Rate Configuration

The AES3 output sample rate can be configured using the Toolbox web-GUI.

1. Click the **Settings** symbol at the top of the screen to open the **Settings panel**.
2. Click the **Audio tab** and use the drop-down menu to select your preferred **AES Output Clock** setting, then click **Save Settings**.



16.3 Configuring Mono or Stereo Peer-to-Peer Programs

The **Programs panel** incorporates a wizard to configure a new program and all audio stream settings. Before you configure a new codec program consider if:

- You want your codec to be capable of dialing and answering, dialing only or answering only.
- A backup connection is required.

This section contains instructions for:

1. Configuring Peer-to-Peer Programs: Dialing and Auto reconnect
2. Configuring Answering Connections

For more information about programs and audio streams within programs see the section titled About Program Dialing. Note: The following connection setup instructions will display how to configure a dial and answer program, with a backup connection. If you want the codec to either dial or answer only, select the option and the wizard will automatically display relevant screens to allow you to configure the codec correctly.

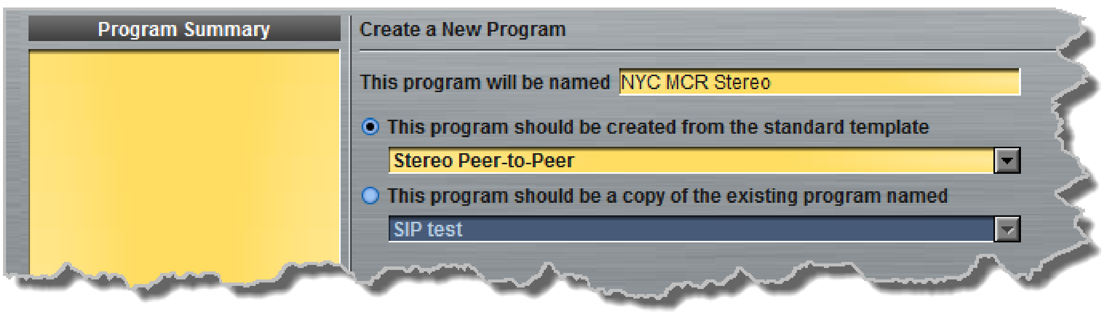
Configuring Peer-to-Peer Programs: Dialing

Important Notes: Before you start program configuration please note:

- You cannot edit a program when it is currently loaded in the codec.
- You can lock a loaded custom program in a codec to ensure the currently loaded program type cannot be unloaded by a codec dialing in with a different program type.
- Some drop-down menus and settings may be greyed out intentionally depending on features available.
- It is possible to save a program at several points throughout the program wizard and use default settings to save configuration time.
- To learn more about programs see the section titled About Program Dialing.

1. Open the web-GUI and click the **Programs** symbol at the top of the screen to display the **Programs panel**.

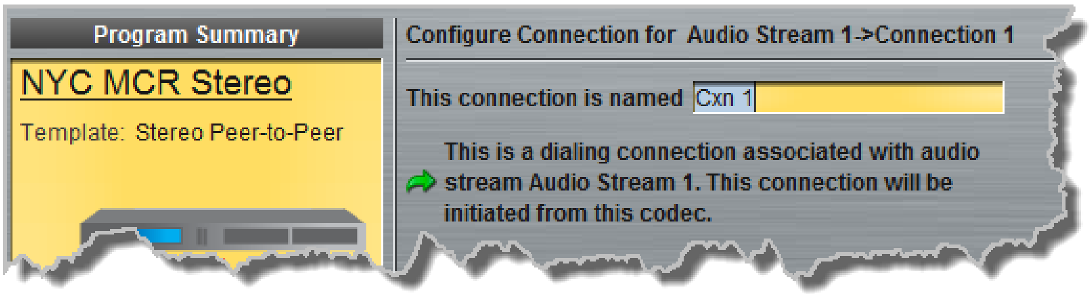
2. Click the **New Program** button to open the wizard and:
   - Click in the text box to name the new program.
   - Select **Mono/Stereo Peer-to-Peer,** or if you want to use an existing program as a template, select this option. Then click **Next**.



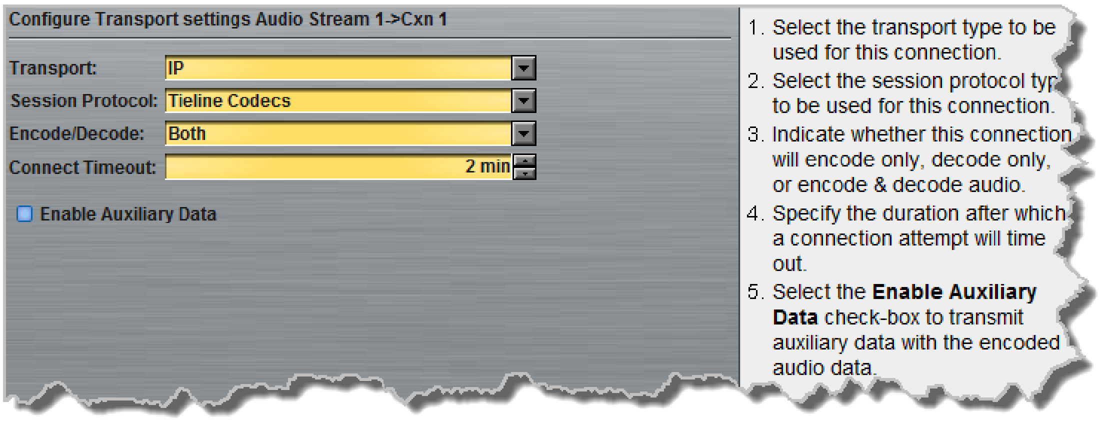
Important Notes: When you decide to use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

3. Enter a name for the **Audio Stream** and configure the codec to dial, answer or dial and answer. Then click **Next**.

4. This audio stream connection in the wizard will allow the codec to dial. Enter the name of the connection in the text box, then click **Next**.

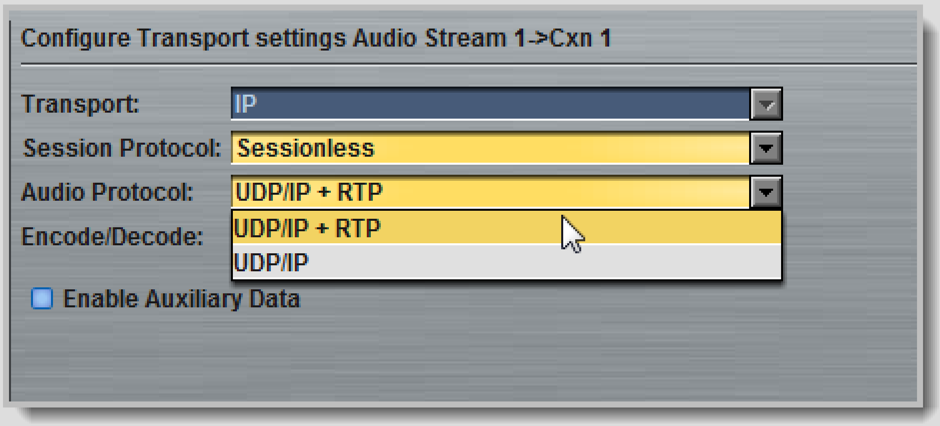


5. Follow the instructions on the right-hand side of the panel to configure the transport settings for the connection, then click **Next**.



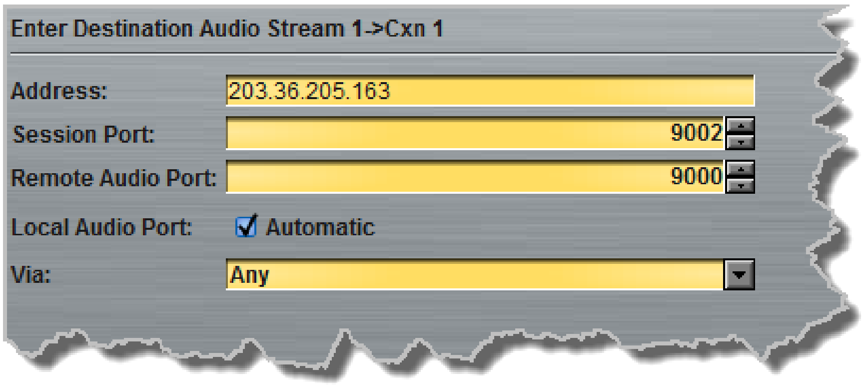
Important Notes:

- If you select **Sessionless** as the **Session Protocol** select **UDP/IP +RTP** for RFC-compliant IP streaming.



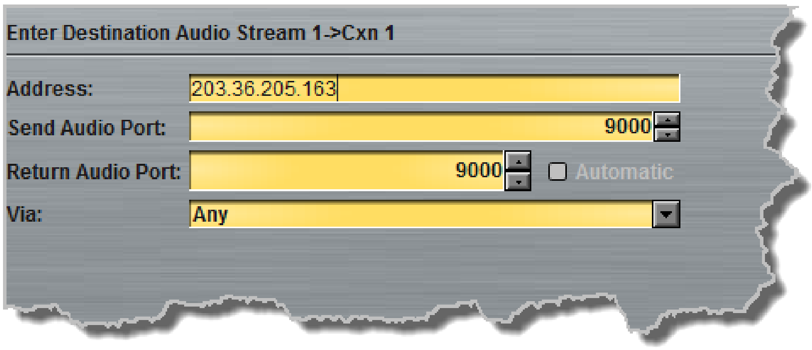
- See RS232 Data Configuration for detailed information on RS232 data and see "Enabling Relays and RS232 Data" for more information on relay operations.

6. Configure the IP address, ports, and then specify which streaming interface is used to dial this connection, e.g. **Primary** (**LAN / ETHERNET** port) or **VLAN** if configured. VLANs can be used to configure separate control and streaming interfaces if required. Note: By default **Any** will select **Primary** as the streaming interface.



Important Notes: The **Remote Audio Port** is the codec port at the remote end of the link to which you are sending audio. The **Local Audio Port** is the port used by the local codec to receive audio from the remote codec. When **Tieline Codecs** is the **Session Protocol** selected (using Tieline sesssion data), the **Local Audio Port** is automatically configured as UDP audio port 9000 by default. This is the default audio port used by all Tieline IP codecs for the first audio stream connection. Click to deselect the **Automatic** check-box to change this setting.

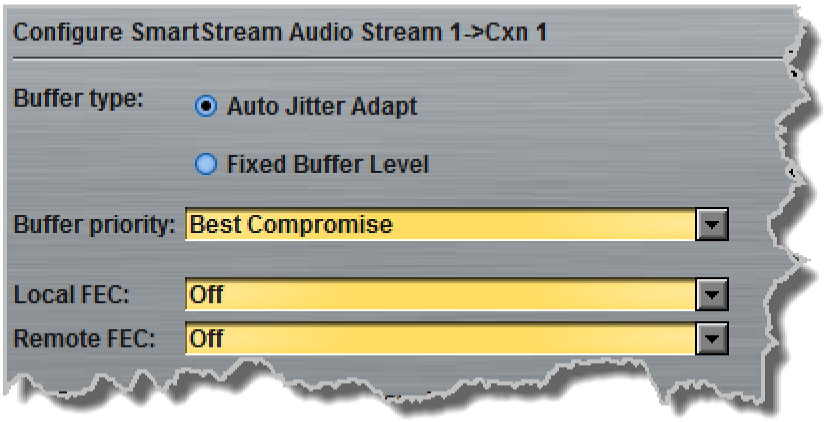
When you select **Sessionless** as the **Session Protocol** the **Session Port** is not configurable and you can manually configure the **Remote Audio Port** and **Local Audio Port**.



7. Click **Save Program** to save the program with the default algorithm, jitter and FEC settings which are physically entered in the codec. Alternatively, click **Next** to specify individual algorithm, jitter buffer and FEC settings for this audio stream (recommended).

8. Click the drop-down arrows on the right-hand side of each text box to adjust the **Encoding**, **Sample rate** and **Bit rate** options, then click **Next**.

9. For IP connections click to configure:
- Auto Jitter Adapt and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**, or
- **Fixed Buffer Level** and enter the **Jitter Depth** (5000ms maximum).
- **Local** and **Remote FEC** settings if required.



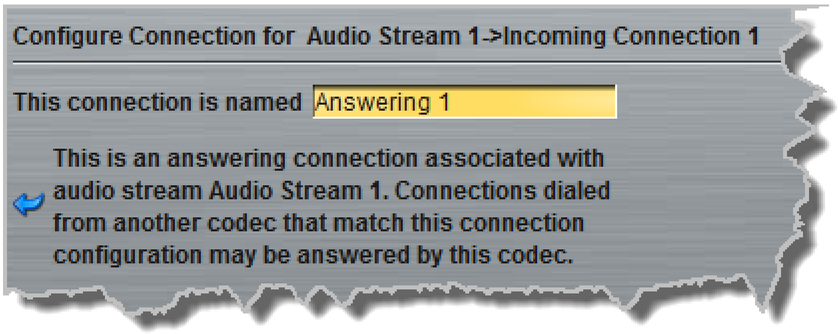
10. Click **Save Program** or click **Next** to **Enable Auto Reconnect**.

11. Click **Save Program** to complete configuration of a dialing only program, or click **Next** to configure an answering connection

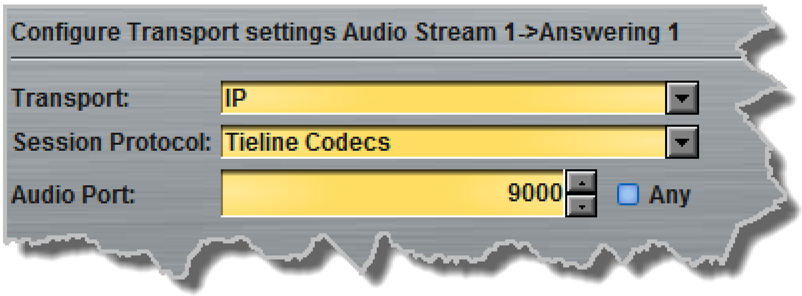
Configuring Answering Connections

The codec is capable of being configured to use specific settings when answering calls. This useful to configure the answering codec with appropriate jitter buffer and FEC settings to suit the prevailing conditions at the codec's location:

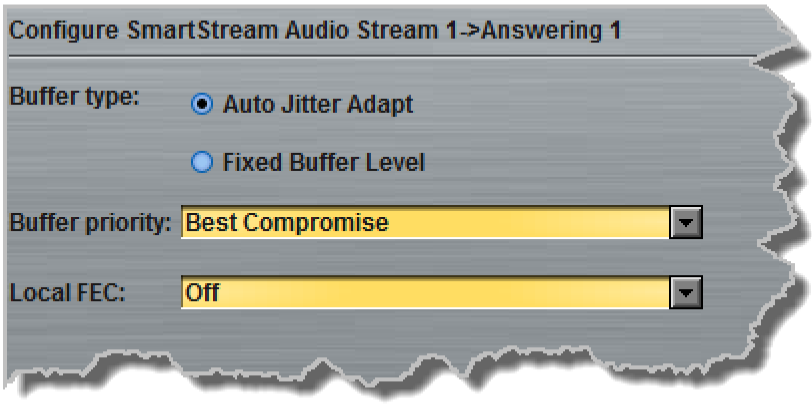
1. Enter a name for the answering connection and click **Next**.



2. Follow the instructions on the right-hand side of the panel to configure the **Transport**, **Session Protocol** and **Audio Port** settings for the connection, then click **Next**.



3. Click to configure jitter buffer and FEC settings:
   - **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**, or
   - **Fixed Buffer Level** and enter the **Jitter Depth**, which must be between 12ms and 5000ms depending on the algorithm you select.
   - **Local** and **Remote FEC** settings if required.



4. After configuring these settings there are 3 options:

   i. If you want to create another answering connection, select the check-box for **Create another answering connection** at the bottom of the panel and continue through the wizard.
   ii. Click **Next** and select **Enable File Playback on silence detection**.
   iii. Click **Save Program** to save the program at this point.

   When you save the program it will be confirmed by the following message.

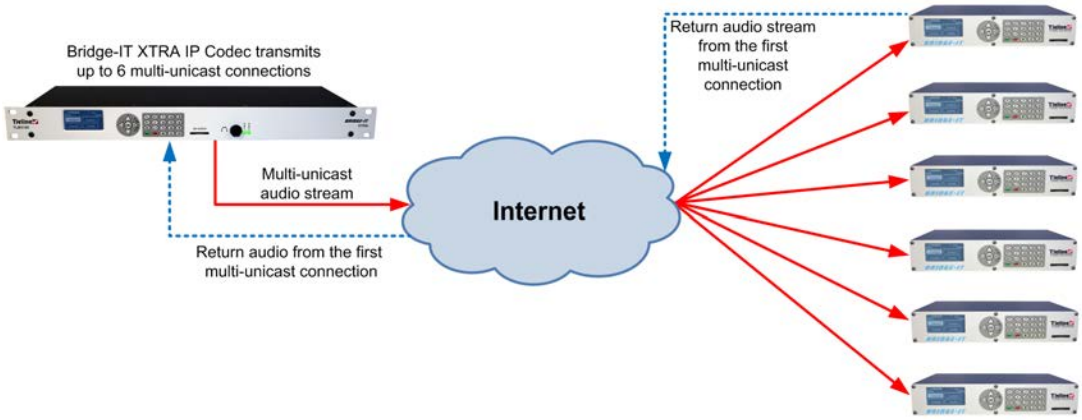
5. Click **Finish** and the newly created program will be displayed in the left pane within the **Programs panel** and in the **Master panel**. Select and connect audio streams in a program using the **Master panel**, or dial the program manually using the codec front panel.

16.4 Configuring Multi-Unicast Dialing Programs

The codec can transmit a mono or stereo multi-unicast audio stream to a maximum of 6 endpoints in total. The first connection in a multi-unicast program is capable of bidirectional audio. Multi-unicast connections can only be created using the ToolBox web-GUI and require a software license which supports multi-unicasting.

Prerequisites:

- Bridge-IT firmware v.1.01.00 r4219 or higher.
- G3 codec firmware v.1.6.56 or higher (if connecting to a G3 codec).
- ToolBox web-GUI v.1.2.2.3 or higher.
- A multi-unicast license installed in the dialing codec.

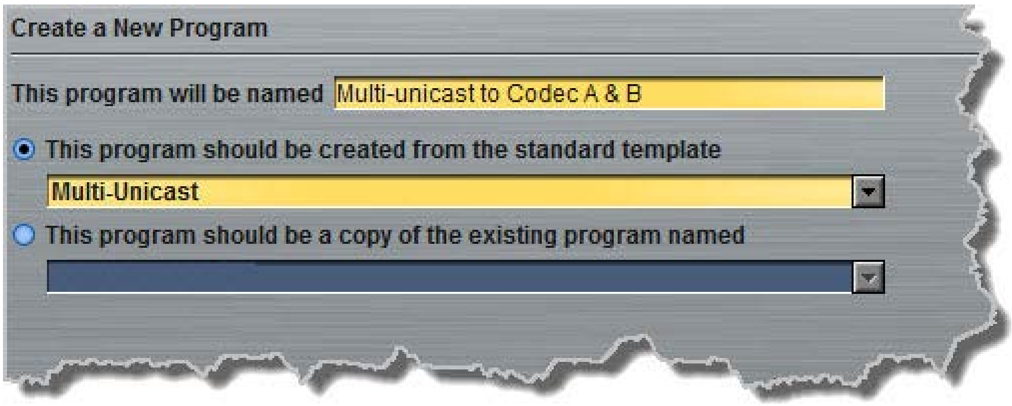


Important Notes: Before you start program configuration please note:
- You cannot edit a program when it is currently loaded in the codec.
- You can lock a loaded custom program in a codec to ensure the currently loaded program cannot be unloaded by a codec dialing in with a different program type.
- If you are dialing to both G3 and Bridge-IT, Merlin or Genie codecs, by default the Audio Reference Level will be configured for the compatibility of the codec dialed first. I.e. if you dial a G3 codec first then the G3 Audio Reference Level will be configured for all connections.
- Connections are dial only for multi-unicast programs.
- Some drop-down menus and settings may be greyed out intentionally depending on features available.
- It is possible to save a program at several points throughout the program wizard and use default settings to save configuration time. The first connection in each multi-unicast stream determines default settings (e.g. algorithm, sample rate, bit rate), for all subsequent connections in that stream, except for dialing settings.
- Select any algorithm for multi-unicast connections except aptX Enhanced and PCM.
- Bidirectional audio is only available on the first connection dialed.
- Forward Error Correction (FEC) is not available on multi-unicast connections.
- Renegotiation of connection bit rates is not possible when connected.
- Ensure you have sufficient connection bandwidth at the local codec to support all the connections to which you are connecting.
- To learn more about programs see the section titled About Program Dialing.

Creating a Multi-Unicast Program

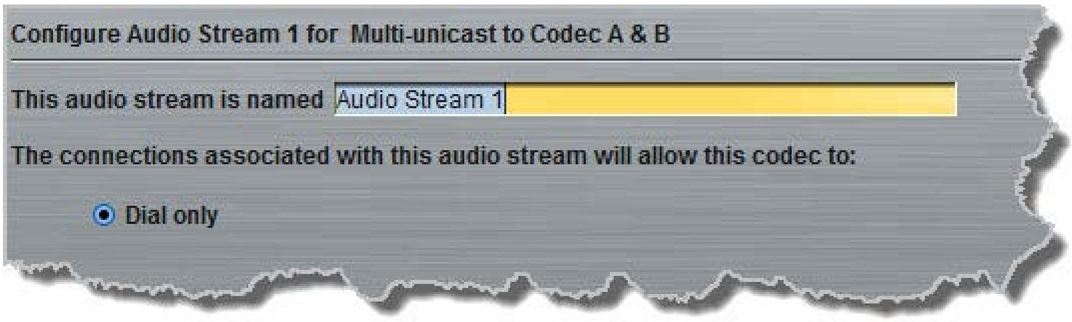
1. Open the web-GUI and click the **Programs** symbol at the top of the screen to display the **Programs panel**.

2. Click the **New Program** symbol to open the wizard and then:
   - Click in the text box to name the new program.
   - Select **Multi-unicast**, or if you want to use an existing program as a template, select this option. Then click **Next**. Note: In this example we are creating a Multi-unicast program to dial 2 end points.

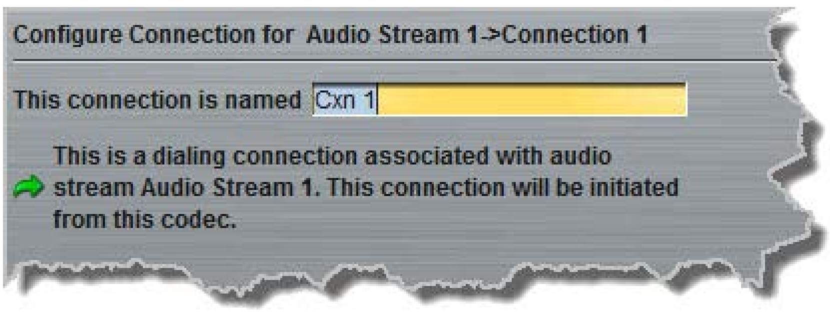


Important Note: When you decide to use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

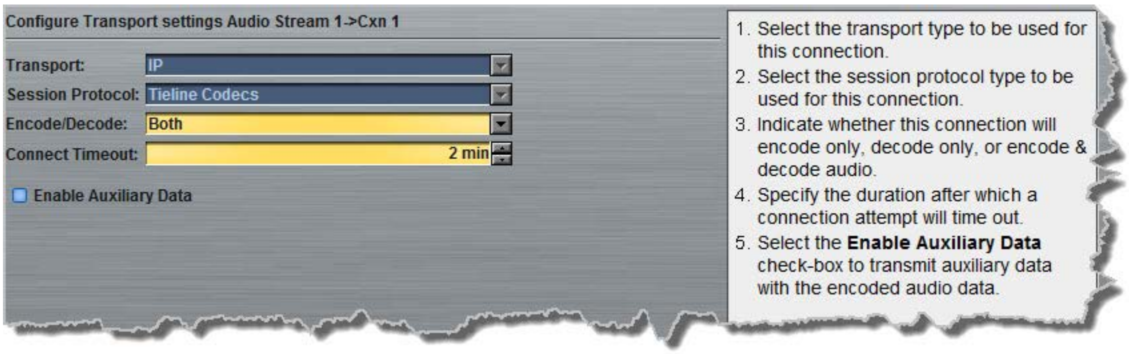
3. Enter a name for the **Audio Stream** and then click **Next**.



4. This audio stream connection in the wizard will allow the codec to dial. Enter the name of the connection in the text box, then click **Next**.

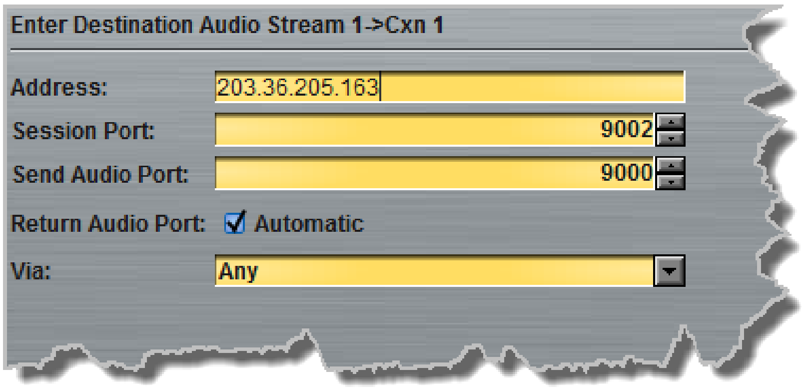


5. Follow the instructions on the right-hand side of the panel to configure the transport settings for the connection, then click **Next**. Note: only the first connection dialed in a multi-unicast program can encode and decode audio. All other connections are unidirectional and encode only.



Important Note: Auxiliary IP data connections are not possible between the codec and G3 Commander and i-Mix codecs over multi-unicast connections. See RS232 Data Configuration for detailed information on RS232 data and see "Enabling Relays and RS232 Data" for more information on relay operations.

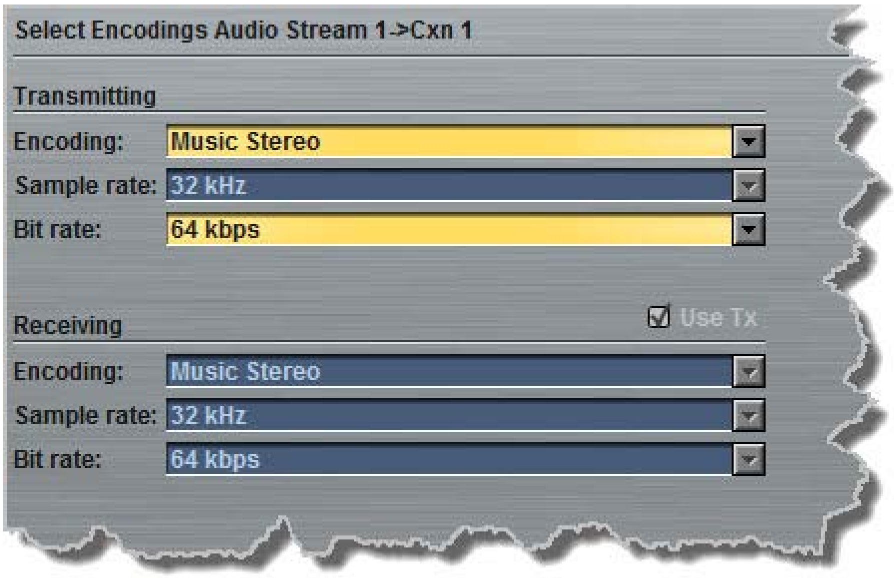
6. Configure the IP address, ports, and then specify which streaming interface is used to dial this connection, e.g. **Primary** (**LAN / ETHERNET** port and default setting) or **VLAN** if configured. Note: By default **Any** will select **Primary**.



Important Notes: The **Remote Audio Port** is the codec port at the remote end of the link to which you are sending audio. The **Local Audio Port** is the port used by the local codec to receive audio from the remote codec. When **Tieline Codecs** is the **Session Protocol** selected (using Tieline sesssion data), the **Local Audio Port** is automatically configured as UDP audio port 9000 by default. This is the default audio port used by all Tieline IP codecs for the first audio stream connection. Click to deselect the **Automatic** check-box to change this setting.

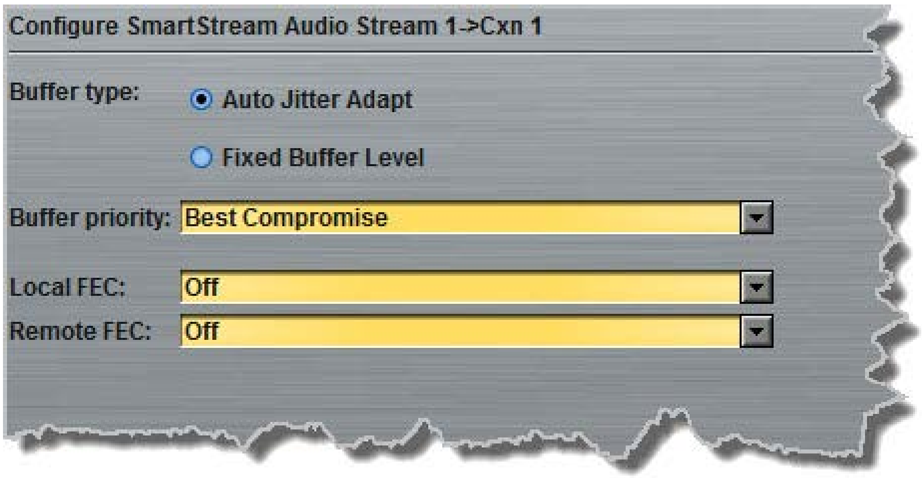
Click **Save Program** to save the program with the default algorithm, jitter and FEC settings which are physically entered in the codec. Alternatively, click **Next** to specify individual algorithm, jitter buffer, FEC and auto reconnect settings for this audio stream (recommended).

7. Click the drop-down arrows on the right-hand side of each text box to adjust the **Encoding**, **Sample rate** or **Bit rate** options.



Click to configure:

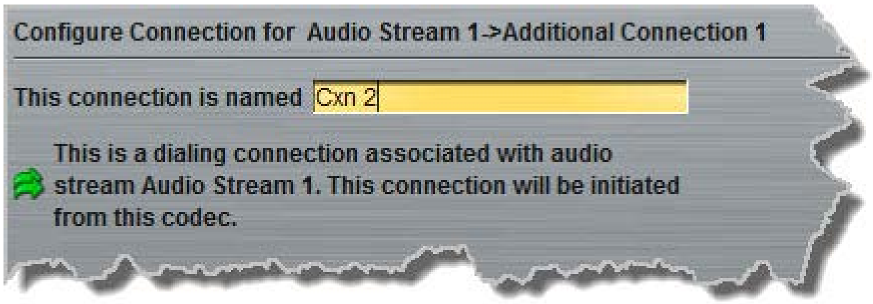
- **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**, or
- **Fixed Buffer Level** and enter the **Jitter Depth** (5000ms maximum).
- **Local** and **Remote FEC** settings if required.



8. Select the **Enable Auto Reconnect** check-box if you want to enable this feature. This is enabled by default.

9. Select the **Create another dialing connection** check-box to configure a new connection for an additional endpoint, then click **Next**.

10.Enter the name of the second connection in the text box, then click **Next**.



11.Continue through the program wizard and configure all multi-unicast connections in a similar manner.

12.After configuring all connections there are 2 options:

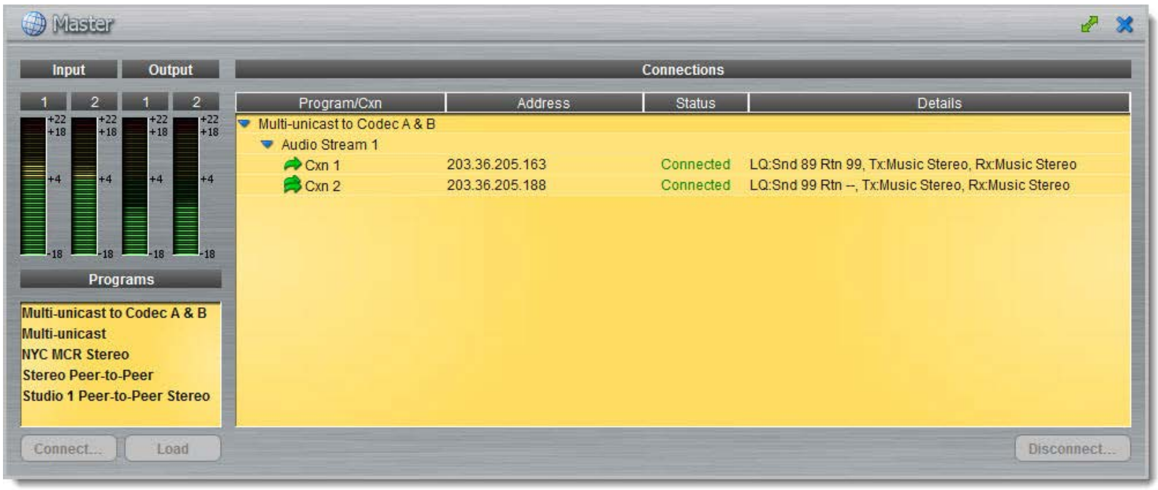
i. Click **Next** and select **Enable File Playback on silence detection**.
ii. Click **Save Program** to save the program.

When you save the program it will be confirmed by the following message.

13.Click **Finish** and the newly created program will be displayed in the left pane within the **Programs panel** and in the **Master panel**. Select and connect audio streams in a program using the **Master panel**, or dial the program manually using the codec front panel.

Monitoring Multi-Unicast Programs

The **Master panel** in the ToolBox web-GUI is used to monitor connection details and view PPMs, as displayed in the following image. Click the blue arrow in the **Program/Cxn** column to expand and minimize connection details (as displayed below).

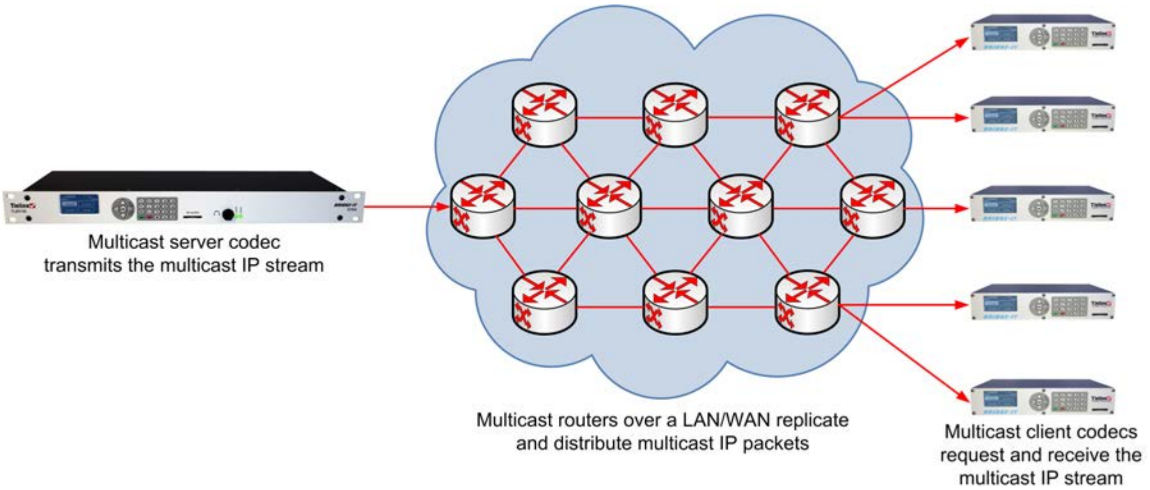


16.5 Configuring Multicast Server Programs

How Multicasting Works

Multicast transmissions are sent using a dedicated IP multicast address that looks similar to a regular IP address and multicast subscribers request transmissions from this address. This unique address allows multicast routers to identify multicast requests from a group of codecs interested in a particular transmission and packets are replicated depending on demand. This can create large demands on network bandwidth if the multicast group is significant in size.

As a result, only small sections of the internet are multicast enabled and many internet service providers (ISPs) block multicast traffic over wide area networks. This restricts most multicast broadcasts to private local area networks. Some ISPs provide quality of service (QoS) priority to multicast streams for an increased service charge. You need to check with your ISP to find out what multicast services, if any, are available over WANs.



Important Notes:
- When a connection is dialed Tieline codecs normally use session data to configure settings like the algorithm, connection bit rate and sample rate etc. Multicast connections are sessionless and do not use Tieline session data. As a result, it is imperative that all codecs are configured with the same connection settings prior to connecting, or they will not be able to join multicast streaming sessions.
- Automatic or fixed jitter buffer settings can be adjusted on individual client codecs as

required. There is no jitter buffer setting on the server codec because it never receives audio packets.

Prerequisites:

- Bridge-IT firmware v.1.01.00 r4219 or higher.
- G3 codec firmware v.1.6.56 or higher (if connecting to a G3 codec).
- ToolBox web-GUI v.1.2.2.3 or higher.
- A multi-unicast license installed in the dialing codec (Note: the Multi-Unicast license includes multicast server capability).
- Use firmware higher than 2.8.xx in the Bridge-IT, Genie and Merlin families of codecs to transmit auxiliary data.

Multicast Server versus Multicast Client Programs

Two different types of multicast programs need to be created when multicasting:

- A multicast server program is used by the broadcasting codec to send multicast IP packets to multicast routers on a network.
- A multicast client program is used by codecs to receive multicast IP audio packets.

A multicast server codec sends audio packets only and a multicast client codec receives audio packets only. Codecs using the client program request multicast packets (sent from the server codec), which are distributed by multicast routers.

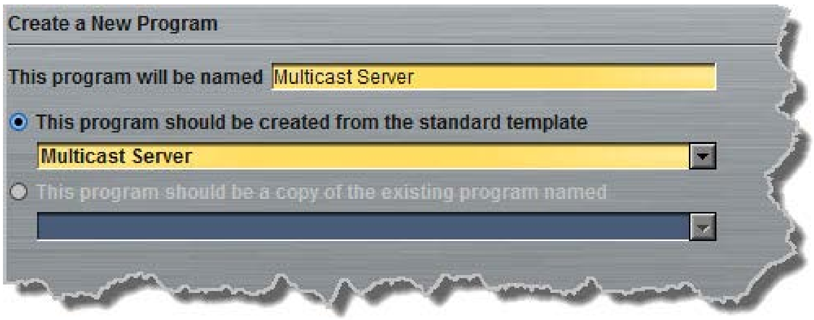
Creating Multicast Server Programs

Important Notes: Before you start program configuration please note:
- Ensure all connection related settings like the port, algorithm, bit rate (etc) match on both multicast server and client programs or they will not connect successfully.
- You cannot edit a program when it is currently loaded in the codec.
- You can lock a loaded custom program in a codec to ensure the currently loaded program cannot be unloaded by a codec dialing in with a different program type.
- Some drop-down menus and settings may be greyed out intentionally depending on features available.
- It is possible to save a program at several points throughout the program wizard and use default settings to save configuration time.
- To learn more about programs see the section titled "About Program Dialing".
- Always dial the multicast server codec connection first before connecting multicast client codecs.
- Multicast client codecs will display return link quality (LQ) only. The **Return** reading represents the audio being downloaded from the network locally. Multicast server codecs do not display LQ readings.
- The default UDP audio port setting is 9000 for a multicast and the client and server port settings must match.
- Forward Error Correction (FEC) is not available for multicast connections.
- It is not possible to connect to a G3 codec and receive multicast IP audio streams.
- To copy multicast client programs onto multiple codecs see Backup and Restore Configuration Files.

1. Open the web-GUI and click the **Programs** symbol at the top of the screen to display the **Programs panel**.

2. Click the **New Program** button to open the wizard and:
   - Click in the text box to name the new program.
   - Select **Multicast Server** or if you want to use an existing program as a template, select

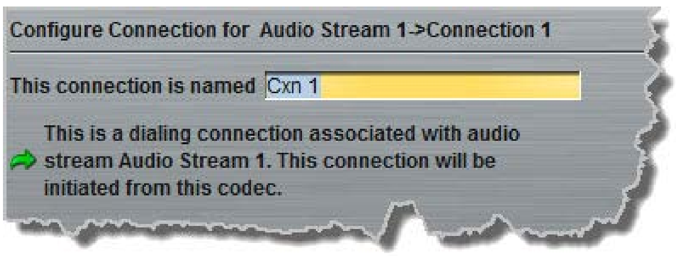
this option. Then click **Next**.



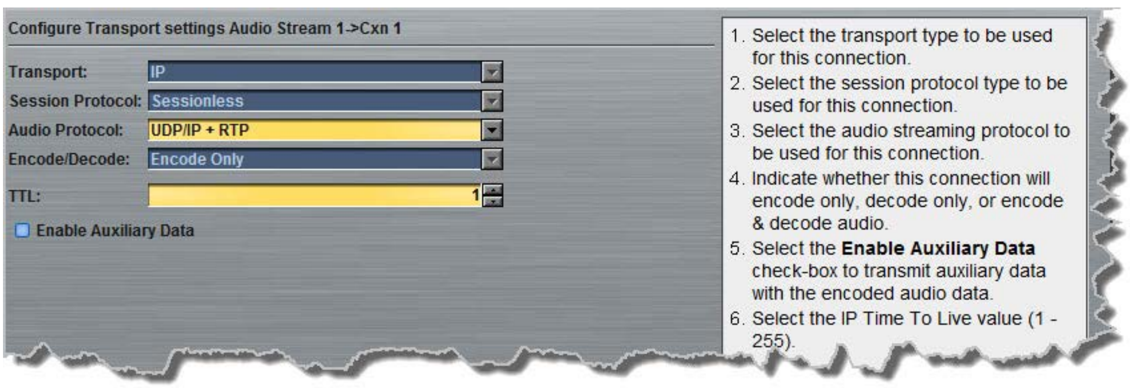
Important Notes: When you decide to use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

3. Enter a name for the **Audio Stream**, then click **Next**.

4. This audio stream connection in the wizard will allow the codec to dial. Enter the name of the connection in the text box, then click **Next**.



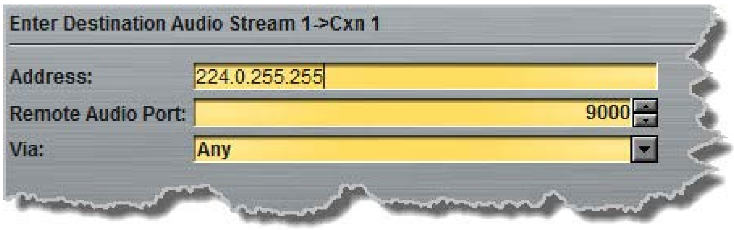
5. Follow the instructions on the right-hand side of the panel to configure the transport settings for the connection, then click **Next**. Note: select **UDP/IP +RTP** for RFC-compliant streaming. If auxiliary data is enabled the audio stream will not be RFC-compliant.



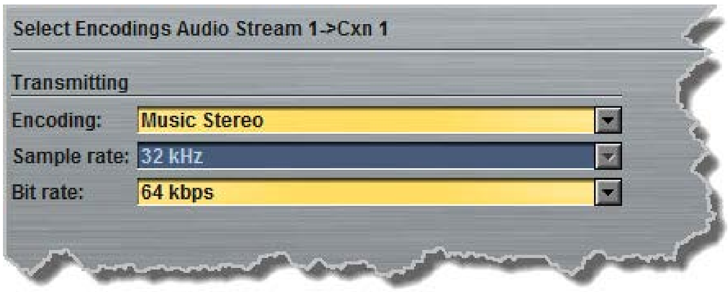
Important Notes:

- The encode and decode direction is configured automatically for **Encode Only** (server program) or **Decode Only** (client program). This setting is configured when you select either **Multicast Server** or **Multicast Client** when you first create the program in the wizard.
- The TTL value you need to use is dependent upon your network infrastructure. Please consult your network administrator if you are unsure about how to configure this setting.
- Use firmware higher than 2.8.xx in the Bridge-IT, Genie and Merlin families of codecs to enable auxiliary data.

6. Configure the multicast IP address and **Remote Audio Port** (the same multicast address and port must be used for both the server and client programs), then specify which IP streaming interface is used to dial this connection, e.g. **Primary** (**LAN / ETHERNET** port and default setting) or **VLAN** if configured. Note: By default **Any** will select **Primary**.



7. Click the drop-down arrows on the right-hand side of each text box to select the **Encoding**, **Sample rate**, **Bit rate** or **Sample size** options. Click **Next** to continue.



8. Click to select **Enable Auto Reconnect** to enable automatic reconnection, then click **Save Program**.

Important Note: There is no jitter buffer setting on the server codec because it never receives audio packets.

9. Configure multicast server and multicast client programs and load all codecs with the appropriate program. Dial the multicast server program connection first and then connect multicast client codec programs to begin receiving multicast audio packets. Select and connect audio streams in a program using the **Master panel**, or dial the program manually using the codec front panel.

16.6 Configuring Multicast Client Programs

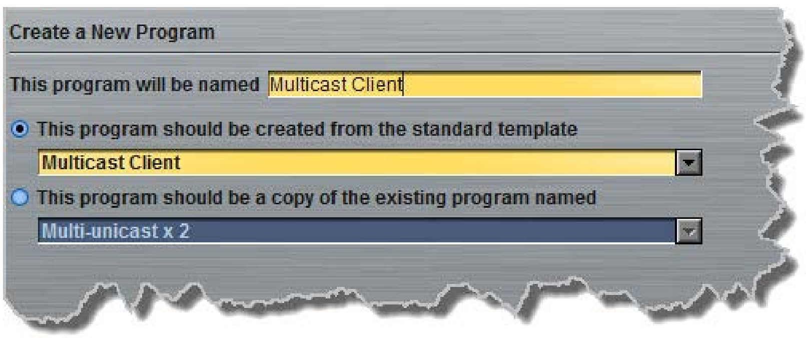
Important Notes: Before you commence program configuration please note:

- Ensure all connection related settings like the port, algorithm, bit rate (etc) match on both multicast server and client programs or they will not connect successfully.
- You cannot edit a program when it is currently loaded in the codec.
- You can lock a loaded custom program in a codec to ensure the currently loaded program cannot be unloaded by a codec dialing in with a different program type.
- Some drop-down menus and settings may be greyed out intentionally depending on features available.
- It is possible to save a program at several points throughout the program wizard and use default settings to save configuration time.
- To learn more about programs see the section titled About Program Dialing.
- Always dial the multicast server codec connection first before connecting multicast client codecs.
- Multicast client codecs will display return link quality (LQ) only. The **Return** reading represents the audio being downloaded from the network locally. Multicast server codecs do not display LQ readings.
- The default UDP audio port setting is 9000 and the client and server port settings must match.
- Forward Error Correction (FEC) is not available for multicast connections.
- Use firmware higher than 2.8.xx in the Bridge-IT, Genie and Merlin families of codecs to enable auxiliary data.
- It is not possible to connect to a G3 codec and receive multicast IP audio streams.
- To copy multicast client programs onto multiple codecs see Save and Restore Configuration Files.

Configuring Multicast Client Programs

1. Open the web-GUI and click the **Programs** symbol at the top of the screen to display the **Programs panel**.

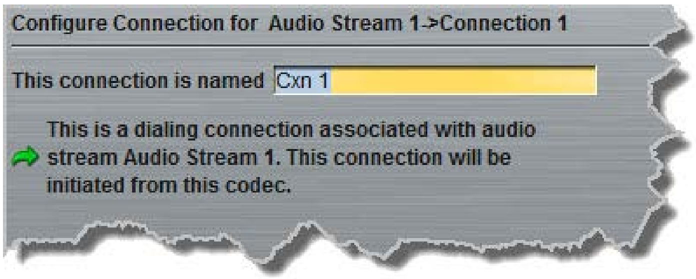
2. Click the **New Program** button to open the wizard and:
   - Click in the text box to name the new program.
   - Select **Multicast Client** to configure a multicast program, or if you want to use an existing program as a template, select this option. Then click **Next**.



Important Notes: When you decide to use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

3. Enter a name for the **Audio Stream**, then click **Next**.

4. This audio stream connection in the wizard will allow the codec to dial. Enter the name of the connection in the text box, then click **Next**.

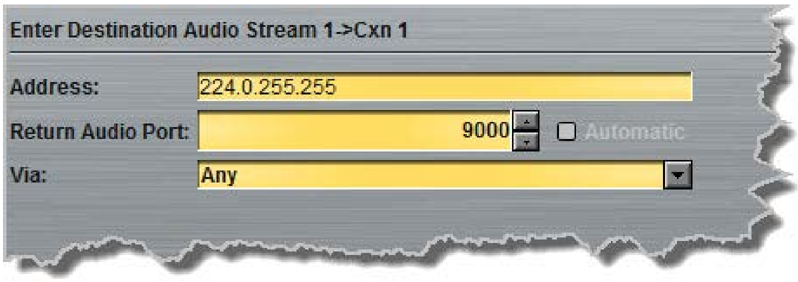


5. Follow the instructions on the right-hand side of the panel to configure the transport settings for the connection, then click **Next**. Note: select **UDP/IP +RTP** for RFC compliant streaming. If auxiliary data is enabled the audio stream will not be RFC-compliant.

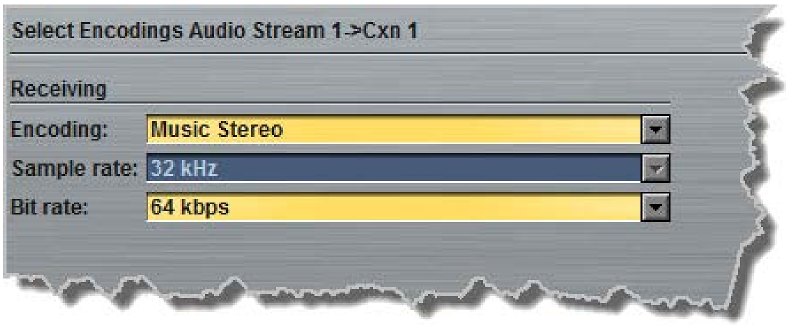
Important Notes:

- The encode and decode direction is configured automatically for **Encode Only** (server program) or **Decode Only** (client program). This setting is configured when you select either **Multicast Server** or **Multicast Client** when you first create the program in the wizard.
- Use firmware higher than 2.8.xx in the Bridge-IT, Genie and Merlin families of codecs to enable auxiliary data.

6. Configure the multicast IP address and audio port (the same multicast address and port must be used for both the server and client programs), then specify which IP streaming interface is used to dial this connection, e.g. **Primary** (**LAN / ETHERNET** port and default setting) or **VLAN** if configured. Note: By default **Any** will select **Primary**.



7. Click the drop-down arrows on the right-hand side of each text box to select the **Encoding**, **Sample rate**, **Bit rate** or **Sample size** options. Click **Next** to continue.



8. Click to configure:
   - **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer Priority**, or
   - **Fixed Buffer Level** and enter the **Jitter Depth** (5000ms maximum).

Important Notes: Automatic or fixed jitter buffer settings can be adjusted on individual client codecs as required. There is no jitter buffer setting on the server codec because it never receives audio packets.

9. Click to select **Enable Auto Reconnect** to enable automatic reconnection, then either:
   i. Click **Next** to select **Enable File Playback on silence detection**, or
   ii. Click **Save Program** to save the program at this point.

When you save the program it will be confirmed by the following message.

10.Click **Finish** and the newly created program will be displayed in the left pane within the **Programs panel** and in the **Master panel**.

11.Configure multicast server and multicast client programs and load all codecs with the appropriate program. Select and connect audio streams in a program using the **Master panel**, or dial the program manually using the codec front panel. Dial the multicast server program connection first and then connect multicast client codec programs to begin receiving multicast audio packets.

16.7 Configuring SIP Settings

The codec is fully EBU N/ACIP Tech 3326 compliant when connecting using SIP (Session Initiation Protocol) to other brands of IP codecs.

About SIP

SIP provides superior interoperability between different brands of codecs due to its standardized protocols for connecting devices and is intended to be used when connecting Tieline codecs to non-Tieline devices. Devices primarily use SIP to dial another device's SIP address and find its location with a minimum of fuss. This task is usually performed by SIP servers, which communicate between SIP-compliant devices to set up a call.

When connecting two devices, SDP performs similar tasks to Tieline's proprietary session data, which is used to configure all non-SIP IP connections. There are two very distinct parts to a call when dialing over IP. The initial stage is the call setup stage and this is what SIP is used for. The second stage is when data transference occurs and this is left to the other protocols used by a device (i.e. using UDP to send audio data).

All the mandatory EBU N/ACIP 3326 algorithms are supported (G.711, G.722, MPEG-1 Layer 2 and 16 bit PCM), as well as optional algorithms including LC- AAC, HE-AAC and aptX Enhanced. The default algorithm selected when connecting using SIP is G.711.

Important Notes:
- Each codec should be registered to a different SIP server account to avoid connection conflicts.
- SIP dialing is only supported over peer-to-peer connections, not multi-unicast connections.
- Tieline G3 codecs do not support AAC and will default to MPEG Layer 2 if a Bridge-IT codec configured for AAC attempts to connect.
- Some Telcos and ISPs may block SIP traffic over UDP port 5060.
- SIP account registration can only be configured via the Ethernet port and not a VLAN.
- When connecting to a Tieline G3 codec using SIP you need to manually select the G3 audio reference level in the codec. To do this select **SETTINGS** **> Audio > Ref Level > Tieline G3**. In addition, select the following on the G3 codec prior to dialing.
  - Select either a mono or stereo profile
  - Select **[Menu] > [Configuration] > [IP1 Setup] > [Session Type] > [SIP]**
  - Select **[Menu] > [Configuration] > [IP1 Setup] > [Algorithm] > [G711/G722 or MP2]**

SIP Server Connections: Getting Started

Registering codecs for SIP connectivity is simple. First, choose the SIP server that you wish to register your codec with. On a LAN this may be your own server, or it could be one of the many internet servers available. We recommend that you use your own SIP server and configure it to use G.711, G.722, MP2 and AAC algorithms. This is because most internet SIP servers are for VoIP phones and are only configured for G.711 and GSM algorithms.

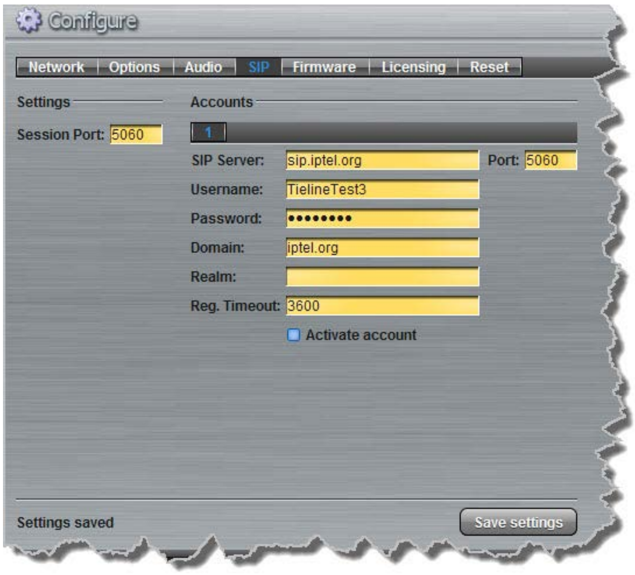
When you register an account with a SIP server you will be provided with:

- The SIP server IP address.
- A username (often the same as a SIP number).
- A password.
- Domain details.
- Realm details (sometimes).

Configure the Codec for SIP using the Web-GUI

Use the Toolbox web-GUI to program SIP account registration details into your codec. Once these details have been entered into the codec, each time it is connected to a public IP address it will contact the SIP server automatically to acknowledge its presence over a wide area network.

1. Connect your codec to a LAN connection with a public IP address, then login to the Toolbox web-GUI and click the **Settings** symbol at the top of the screen to display the **Settings panel**.
2. Click the **SIP** button.
3. Enter the account details into the relevant text boxes.
4. Enter the **Registration Timeout** (this shouldn't need to be adjusted from the default setting).
5. Click to select **Activate Account** and click the **Save Settings** button to create the account in the codec. **Settings Saved** is displayed in the bottom-left corner of the **Settings panel** if the account details are saved successfully.



5. Navigate to **SETTINGS** **> SIP > Accounts** to verify that the account has been registered to the SIP server. The registration symbol appears when it is activated successfully.

Important Notes: Some ISPs may block SIP traffic over UDP port 5060.

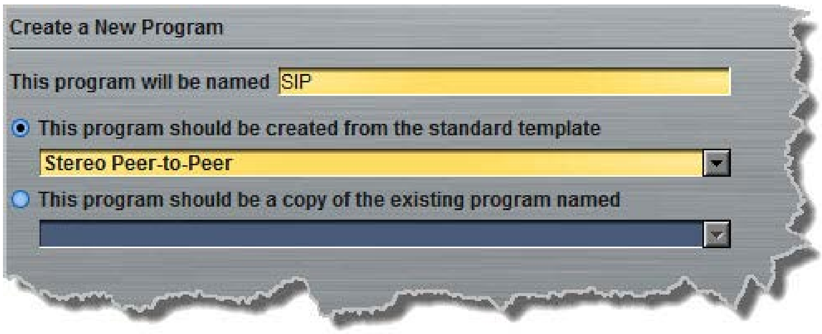
16.8 Configuring SIP Programs

SIP programs are like a normal IP program to configure, with two small differences; entering a SIP address and selecting SIP as the **Session Protocol**.

Important Notes: Before you start program configuration please note:
- SIP can only operate using the **LAN / ETHERNET** port on the rear panel of the codec.
- You cannot edit a program when it is currently loaded in the codec.
- Some drop-down menus and settings may be greyed out intentionally depending on features available.
- To learn more about programs see the section titled About Program Dialing.

1. Open the web-GUI and click the **Programs** symbol at the top of the screen to display the **Programs panel**.

2. Click the **New Program** button to open the wizard and:
   - Click in the **Program Name** text box to name the new program.
   - Select **Mono/Stereo Peer to Peer,** or if you want to use an existing program as a template, select this option. Then click **Next**.

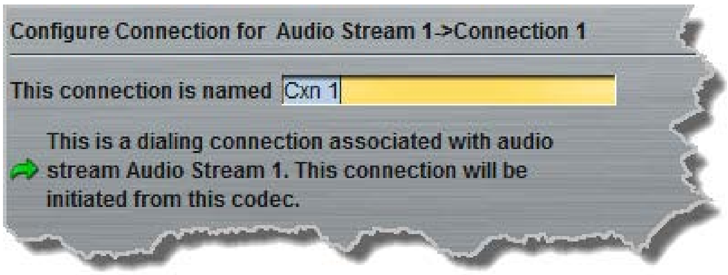


Important Notes: When you choose to use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

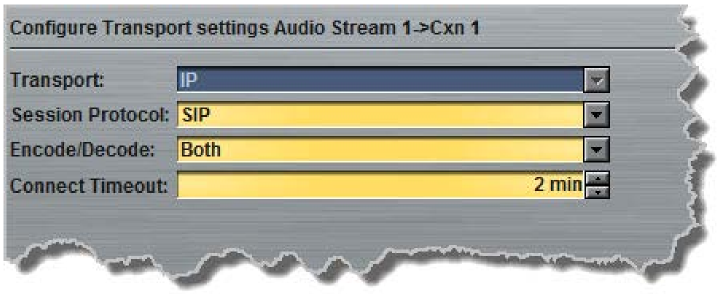
3. Enter a name for the **Audio Stream** and configure the codec to dial, answer or dial and answer. Then click **Next**.

Note: The following example will display how to configure a dial and answer program. If you want the codec to either dial or answer only, select the option and the wizard will automatically display screens to allow you to configure the codec correctly.

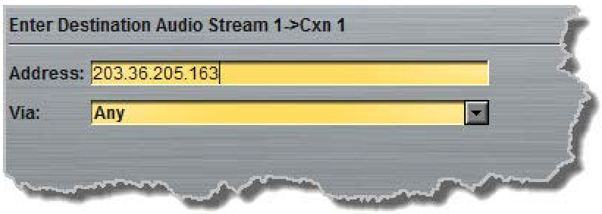
4. This audio stream connection in the wizard will allow the codec to dial. Enter the name of the connection in the text box, then click **Next**.



5. Follow the instructions on the right-hand side of the panel to configure the transport settings for the connection: Ensure that you select **SIP** as the **Session Protocol** menu. Then click **Next**.



6. Configure the destination codec **Address** and **Audio Port**, then the network interface used to dial the connection, e.g. **Primary** (**LAN / ETHERNET** port and default setting) or **VLAN** if configured. Note: By default **Any** will select **Primary**.



At this point you can click **Save Program** and save the program with the default jitter and FEC settings in the codec. Alternatively, click **Next** to specify individual algorithm, jitter buffer and FEC settings for this connection and configure backup audio for this audio stream (recommended).

Important Notes:
- If your codec is registered with same SIP registrar as the destination codec then you only need to enter the SIP user name to dial successfully.
- The default UDP audio port when using SIP is 5004 in Tieline codecs. To contact a codec that is behind a firewall or NAT-enabled router, it is essential that this and all other relevant ports are open and forwarded to the other device.

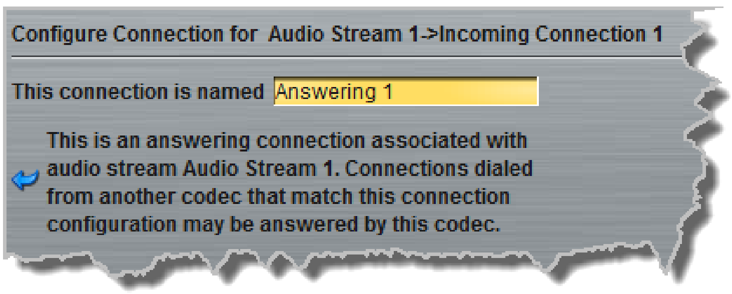
7. Click the drop-down arrows on the right-hand side of each active drop-down menu to adjust the **Encoding, Sample rate** or **Bit rate** parameters. Click Next to continue.

8. Click to configure:

- **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer Priority**, or
- **Fixed Buffer Level** and enter the **Jitter Depth** (5000ms maximum).

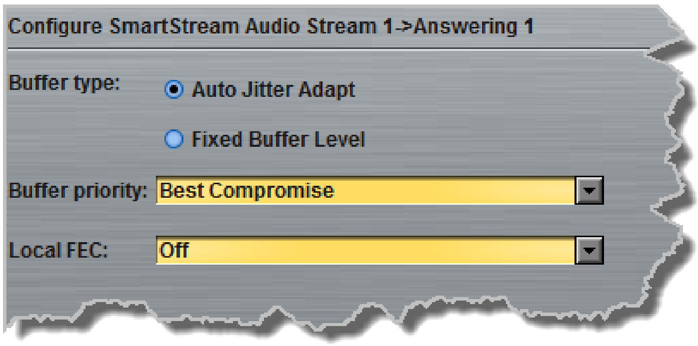
9. Click to select **Enable Auto Reconnect** to enable automatic reconnection, then click **Next**.

10. Enter a name for the answering connection and click **Next**.



11. Follow the instructions on the right-hand side of the panel to configure the **Transport**, **Session Protocol** and **Audio Port** (preconfigured for SIP) settings for the connection, then click **Next**.

12.Click to configure jitter buffer and FEC settings:
- **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**, or
- **Fixed Buffer Level** and enter the **Jitter Depth** (5000ms maximum).



13.After configuring these settings there are 3 options:

i. If you want to create another answering connection, select the check-box for **Create another answering connection** at the bottom of the panel and continue through the wizard.
ii. Click **Next** and select **Enable File Playback on silence detection**.
iii. Click **Save Program** to save the program at this point.

When you save the program it will be confirmed by the following message.

5. Click **Finish** and the newly created program will be displayed in the left pane within the **Programs panel** and in the **Master panel**. Select and connect audio streams in a program using the **Master panel**, or dial the program manually using the codec front panel.

Caution: If the codec LAN cable is disconnected and the IP address changes when dialing in SIP mode, you will need to reboot the codec, otherwise the codec will not be able to reconnect.

16.9 Dial and Disconnect a Program

Connecting a Program

1. Click to select the program you want to load from the **Programs** list in the **Master panel**.

2. Click **Connect** to load the program and connect all audio streams.

Disconnecting a Program

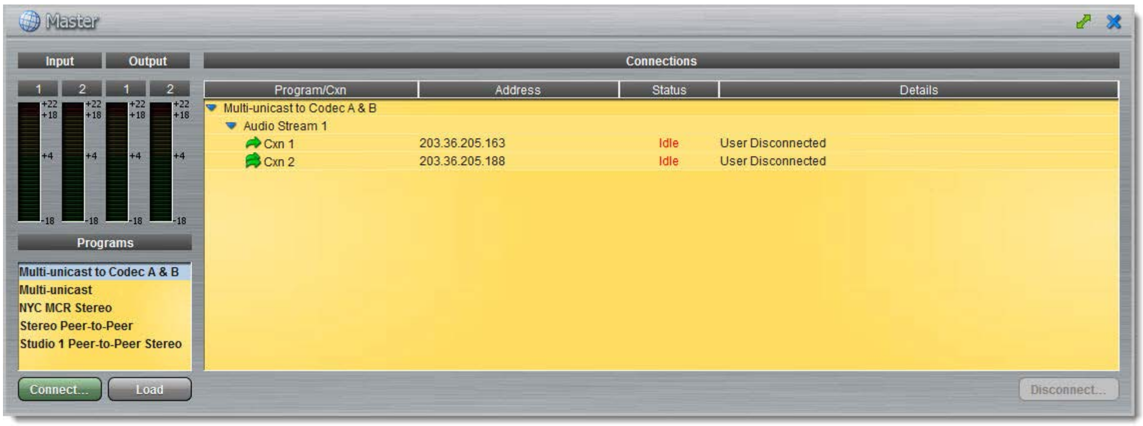
1. Click to highlight the audio stream in the **Connections** pane of the **Master panel**.

2. Click **Disconnect** to end the connection.

16.10 Dial and Disconnect Multi-unicast Connections

Dial Multiple Connections within a Program

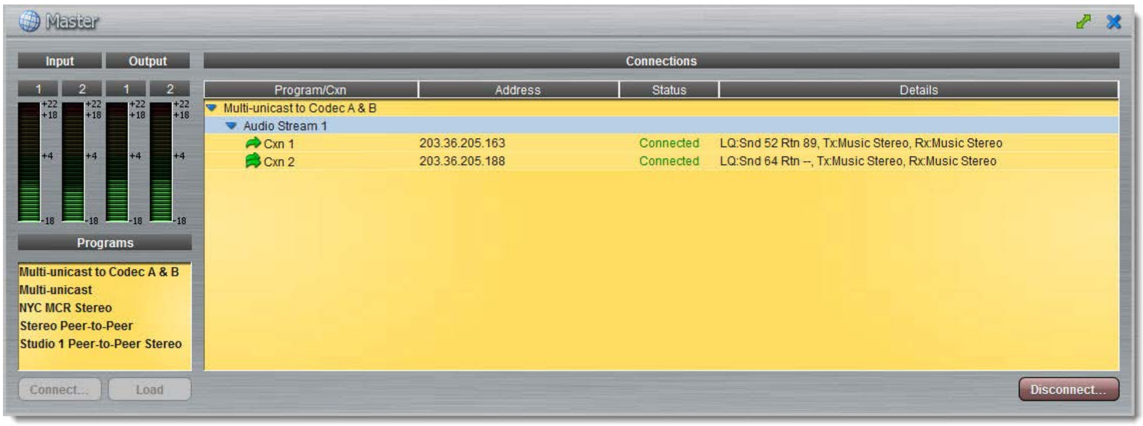
1. Click to select the program you want to load from the **Programs** list.



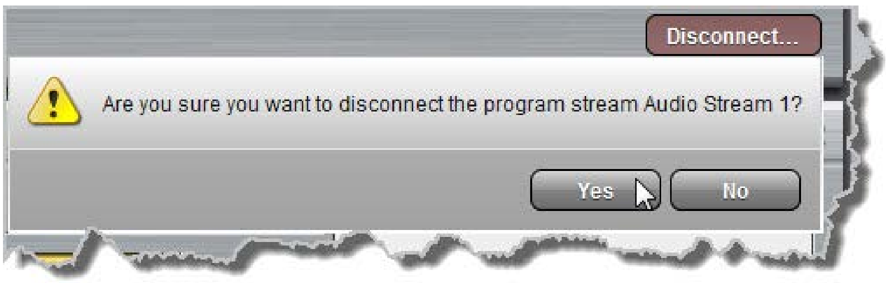
2. Click **Connect** to load the program and connect all connections.

Disconnect All Audio Stream Connections

1. Click to select the program in the **Connections** pane, e.g. **Multi-unicast to Codec A & B** in the following example.

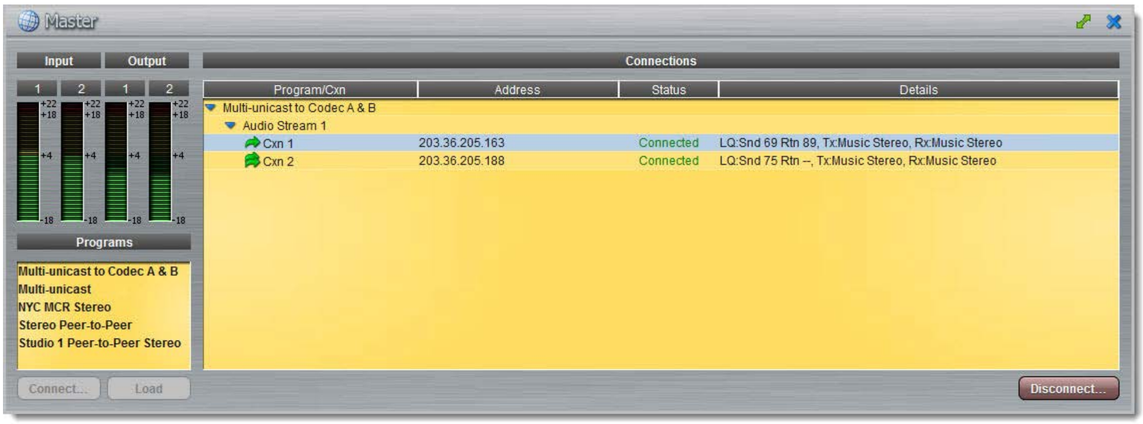


2. Click the **Disconnect** button.
3. Click **Yes** in the confirmation dialog to disconnect all audio stream connections.



Disconnect a Single Audio Stream Connection

1. Click to select the audio stream connection you want to disconnect.



2. Click the **Disconnect** button.
3. Click **Yes** in the confirmation dialog to disconnect all audio stream connections.

16.11 Lock or Unlock Programs

It is possible to lock a loaded custom program in a codec to ensure the currently loaded program type, e.g. mono, cannot be unloaded by a codec dialing in with a different program type, e.g. stereo. For example, if you require the codec at the studio to always connect in mono, simply load and lock a mono program in the codec.

1. Open the web-GUI and click the **Settings** symbol at the top of the screen to display the **Settings panel**.
2. Click the **Options** button at the top of the **Settings panel**.
3. Click the **Lock Loaded User Program** check-box to lock or unlock a user program in the codec.

4. Click **Save Settings** at the bottom of the panel to save the new configuration.

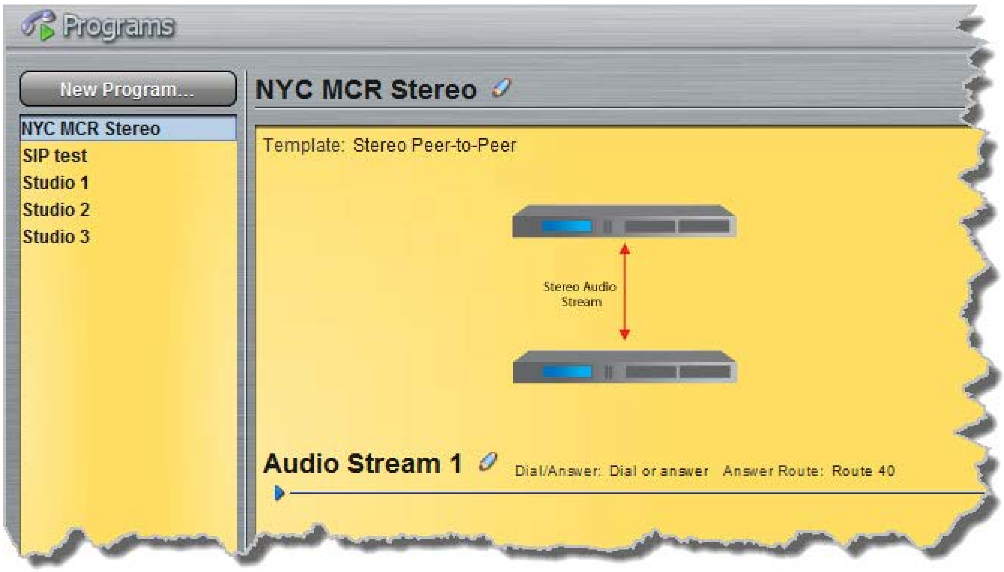
Important Note:
- It is only possible to lock custom programs in a codec.
- If **Lock Program** is enabled and you load a new custom program in the codec, **Lock Program** remains enabled and locks the most recently loaded custom program.
- A red **Padlock** symbol appears in the **Status** section of the **Master panel** to indicate a program is locked in the codec.

16.12 View/Edit/Delete Programs

Important Notes: You cannot edit or delete a program when it is currently loaded in the codec. Ensure you have unloaded a program prior to editing the current configuration.

To view configuration settings for an existing program, or edit settings:

1. Open the web-GUI and click the **Programs** symbol at the top of the screen to display the **Programs panel**.
2. Click to select a program in the left-hand pane.



3. Click the blue arrow to expand audio stream information and click the **Edit** symbol to adjust program settings.

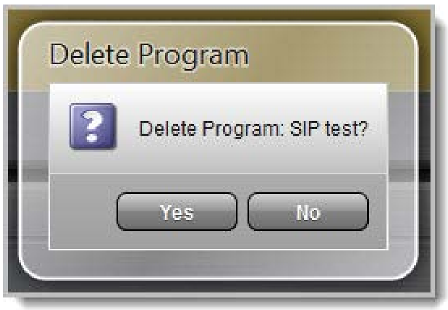
The program wizard will open at the relevant point to facilitate editing of connection parameters. Click **Save Program** to store settings.

Deleting Programs

There are two ways to delete a program.

1. Ensure the program you want to delete is not currently loaded in the codec.
2. Click to select a program listed on the left hand side of the **Programs panel** and then right-click to display menu options.

3. Select and click **Delete Program**.
4. Click **Yes** in the confirmation dialog.



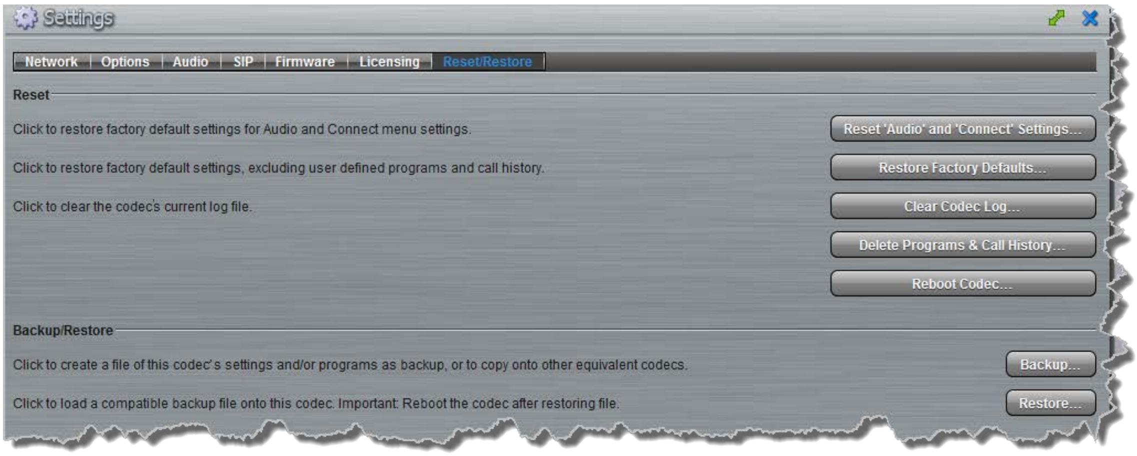
5. Alternatively, click **Delete Program** next to the program name in the top-right corner of the **Programs panel**.

16.13 Reset Factory Default Settings

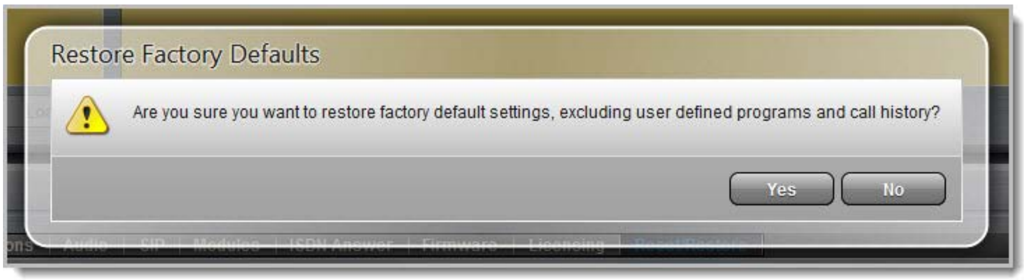
There are several options which allow you to restore factory default settings within the codec. See Reset and Restore Factory Defaults for more details on each option.

1. Open the web-GUI and click the **Settings**  symbol at the top of the screen to display the **Settings panel**.
2. Click the **Reset/Restore** button at the top of the **Settings panel**.

3. Click one of the reset options available.



4. A confirmation dialog appears for each option, click **Yes** to proceed or **No** to cancel the reset function.



16.14 Backup and Restore Functions

The Toolbox web-GUI can be used to backup and restore codec settings, including:

- Programs containing a variety of connection settings.
- All system settings that have been adjusted to change the factory default codec settings (current runtime settings).

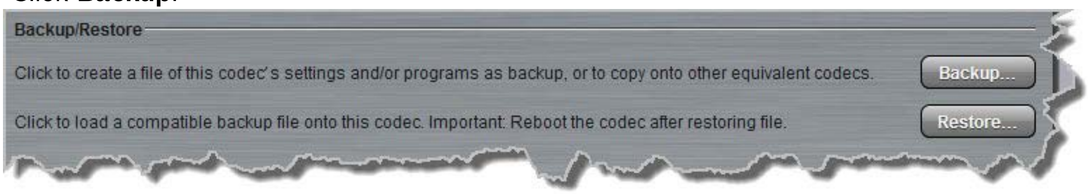
Files can also be used to copy configurations onto other similar codecs. Programs are essentially connection profiles that may include:

- Program, audio stream and connection names.
- IP address, port, algorithm, jitter buffer, FEC and bit rate settings (etc.) for audio stream connections.

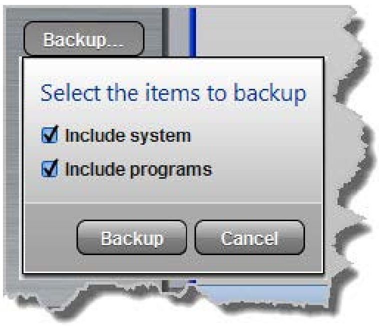
Creating Backup Files

1. Open the web-GUI and click the **Settings** symbol at the top of the screen to display the **Settings panel**.
2. Click the **Reset/Restore** button at the top of the **Settings panel**.

3. Click **Backup**.



4. Use your mouse-pointer to click and select the check boxes to confirm your backup requirements, then click **Backup**.



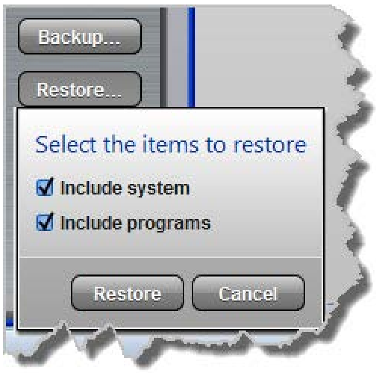
5. Click **Save** and select a location on your PC to save the configuration file.

Restoring Configuration File Settings

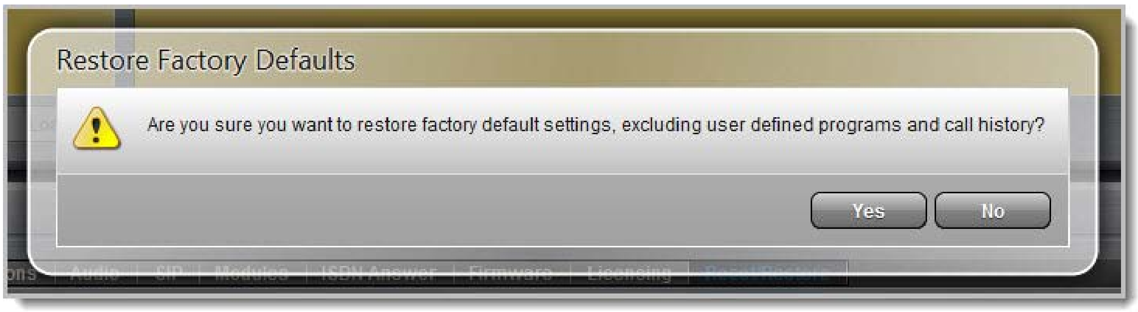
1. Open the web-GUI and click the **Settings** symbol at the top of the screen to display the **Settings panel**.

2. Click the **Reset/Restore** button at the top of the **Settings panel**.

3. Click **Restore**.
4. Navigate to the configuration file on your PC that you want to load, then click **Open**.
5. Use your mouse-pointer to click and select the check boxes for restoring items. For example, you could select the **Include programs** check-box and deselect the **Include system** check-box if you are only copying programs onto codecs.



6. Click **Restore** to copy the configuration file settings onto the codec; confirmation of successful file restoration is provided.



7. The codec will reboot automatically to ensure the restored configuration is loaded successfully.

16.15 Web-GUI Software License Installation

Prior to installation you will need connect your codec to a PC and ensure you are connected to the internet. You should also check that you have received notification by email from Tieline that your new license file is ready to download from TieServer.

Perform an Automatic Software License Install using the Java Toolbox Web-GUI

1. Open the Java Toolbox web-GUI in a browser on your PC by typing either the IP address of the codec (LAN connection), or the USB address of the codec (USB connection) into the address bar.
2. Click the **Settings** button at the top of the web-GUI screen to open the **Settings panel**.
3. Click **Licensing** in the **System panel**.

4. Select the **Get license file from TieServer** button.
5. Click the **Upload License File** button.
6. After the upgrade is completed click **Finish** and the codec screen should display a confirmation message within a short period of time.

7. Reboot the codec via **Settings > Reset > Reboot Codec** and press the button. Note: do not reboot by removing the power cable from the codec.

Download a License File and Install Manually

Prior to installing any new software license you will need to connect your codec to a PC and save the license file on this computer.

1. Open the Toolbox web-GUI in a browser on your PC by typing either the IP address of the codec (LAN connection), or the USB address of the codec (USB connection) into the address bar.
2. Click the **Settings** button at the top of the web-GUI screen to open the **Settings panel**.
3. Click **Licensing** in the **System panel**.
4. Click **Upload a selected file**.

5. Click the **Select File** button to open a dialog and navigate to the ".lcf" license file on your PC, then click the **Open** button.
6. Click the **Upload License File** button to upload the license file into the codec.
7. Click the **Finish** button.
8. Reboot the codec via **Settings > Reset > Reboot Codec** and press the OK button. Note: do not reboot by removing the power cable from the codec.

16.16 Download Logs

The codec is capable of providing diagnostic information via user logs, which can either be sent to Tieline support, or downloaded for user diagnostics.

Procedure for Sending Logs to Tieline

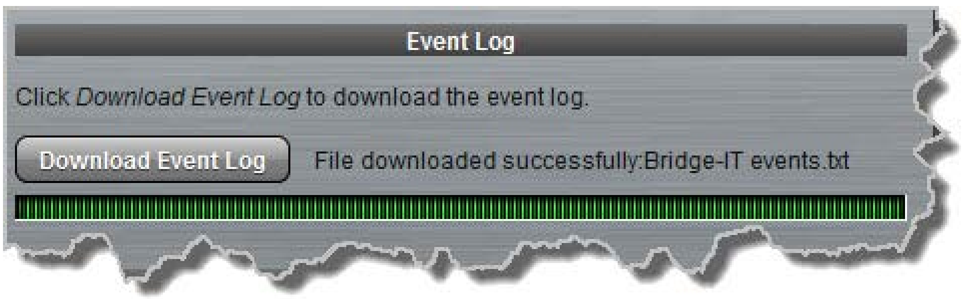
1. Open the web-GUI and click the **Help** symbol at the top of the screen to display the **Help panel**.
2. Click **Download Logs**.

3. Save the file to your computer and then send it as a .zip file to Tieline support via support@tieline.com

Download Event Logs

Event logs can be downloaded from the codec and viewed using any text editor, e.g. Microsoft® Word.

1. Open the web-GUI and click the **Help** symbol at the top of the screen to display the **Help panel**.
2. Click **Download Event Log** and select a location to save the log file.



Clearing Logs

This option should only be used if instructed to by Tieline support staff. To clear all event and other logs in the codec via the front panel see the Reset and Restore Factory Default Settings section of this manual, or see Reset Factory Default Settings to clear recent log history using the web-GUI.

16.17 RS232 Data Configuration

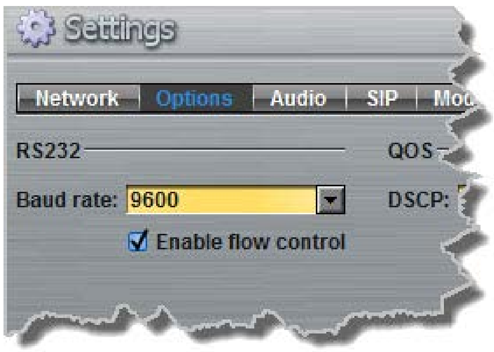
The codec supports both in-band and out-of-band data depending on the algorithm you are using. RPTP data is automatically enabled when using the Tieline Music or MusicPLUS algorithms over IP. It is also possible to enable synchronized out-of-band data using any algorithm.

Algorithm Selected	IP
Tieline Music and MusicPLUS	• In-band RPTP data enabled automatically • Synchronized out-of-band data can be enabled and disabled
All other algorithms	• Synchronized out-of-band data can be enabled and disabled

The codec can be connected to external devices and send RS232-compatible data via the serial port on the rear panel of the codec. To enable RS232 data within a connection, select **Enable Auxiliary Data** when creating a program in the **Programs panel** wizard. Alternatively, select using the codec **Setup** menu (see Enabling RS232 Data)).

Setting RS232 Data Rates and Flow Control

1. Open the web-GUI and click the **Settings** symbol at the top of the screen to display the **Settings panel**.
2. Click the **Options** button.
3. Click the **Baud rate** drop-down menu arrow to select the serial port baud rate which matches the baud rate of the external device connected to the RS232 port on the codec.
4. Click to select the **Enable flow control** check box and enable flow control, then click **Save settings**.



Important Notes:
- When connecting to G3 codecs over IP only in-band data is available via the Music and MusicPLUS algorithms.
- Use firmware higher than 2.8.xx in the Bridge-IT, Genie and Merlin families of codecs to enable auxiliary data over multicast connections.
- It is important to enable serial port flow control as it regulates the flow of data through the serial port. If disabled, data will flow unregulated and some may be lost.
- Ensure you configure the serial port baud rate to match the setting of the external device to which you are connecting. Ideally the settings on both codecs should match, or you could have data overflow issues.
- Only the dialing codec needs to be configured to send RS232 data. Session data sent from the dialing codec will configure all other compatible codecs (non-G3) when you connect.

- RS232 data can be sent from the dialing codec to all endpoints of a multi-unicast connection if your codec is capable of these connections. Note: Bidirectional RS232 data is only available on the first connection dialed when multi-unicasting.

16.18 Creating Rules

The **Rules panel** in the Toolbox web-GUI is used to configure actions which are dependent upon changes to GPIO control port states or connection events. Rules can only be created with the web-GUI while the codec is disconnected. Note: **Data** must be enabled in the **Connection** menu to enable contact closure operation and RS232 data. This is disabled by default. **Data** can be enabled in the codec as follows:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select your preferred **IP Session** mode (**Tieline** or **Sessionless**).
5. Use the down navigation button to select **Setup** and press the button.
6. Navigate to **Data** and press to toggle between **Enabled** and **Disabled**.

For more information please see "Enabling Relays & RS232 Data" in this user manual.

Configuring Rules

Default rules have been configured into the codec to facilitate programming the most common events required by broadcast engineers. To view rules options:

1. Click the **Rules** button at the top of the web-GUI screen to open the **Rules panel**.

2. Click **Add New Rule**.

3. Click to select the appropriate rule for your requirements. See the Web-GUI Introduction section for explanations of what actions each rule can perform.

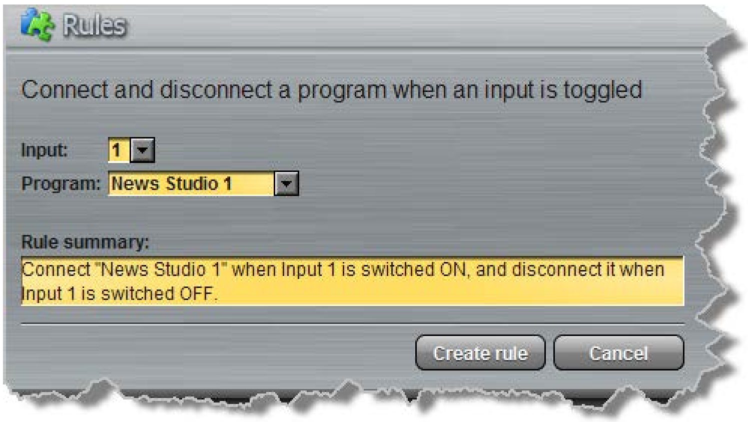
Any previously configured rules are displayed when the **Rules panel** is first opened.

Rule 1: Toggle a Control Port Input to Connect and Disconnect a Program

This rule is used to connect and disconnect a selected program when a control port input is toggled.

1. Click the first rule in the **Rules panel**.

2. Click the drop-down **Input** arrow and select the control port input which will trigger program connection and disconnection.



3. Click the drop-down **Program** arrow to select the program to be connected.

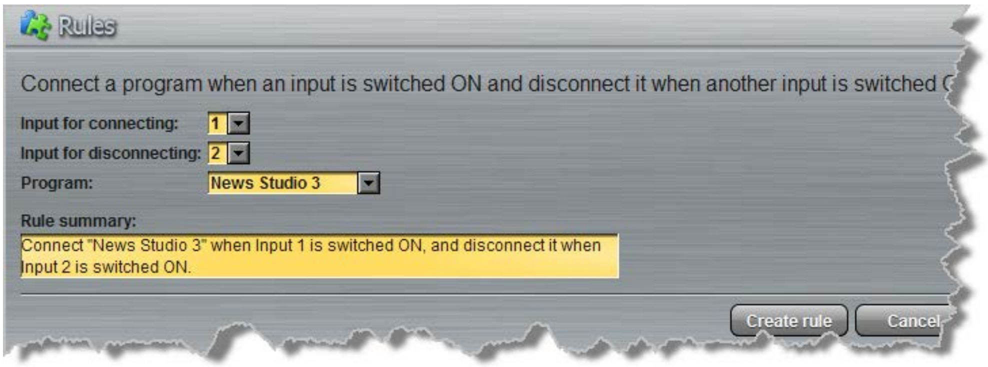
4. Check the **Rule Summary** and click **Create Rule** to save the settings.

Rule 2: Switch Different Control Port Inputs On to Connect and Disconnect a Program

This rule is used to connect and disconnect a selected program when different codec control port inputs are turned on.

1. Click the second rule in the **Rules panel**.

2. Click the drop-down arrows to select the control port input for connecting and the alternative one for disconnecting.



3. Click the drop-down **Program** arrow to select an individual program which will be connected and disconnected by the change in the control port input states.

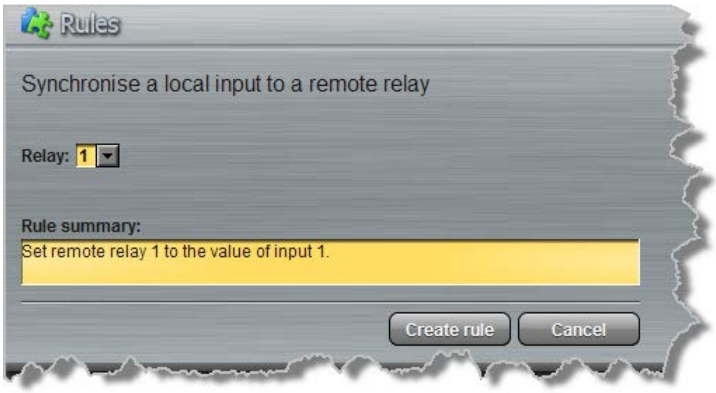
4. Check the **Rule summary** and click **Create Rule** to save the settings.

Rule 3: Synchronise Local Control Port Input Status with a Remote Relay Output

Use this rule allow a local codec's control port input to change the state of a remote relay output.

1. Click the third rule in the **Rules panel**.

2. Click the drop-down arrow to select the local control port input used to control a remote relay output.



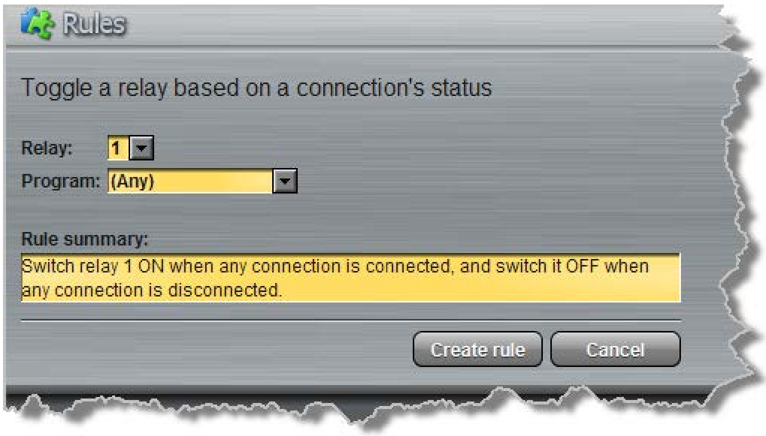
3. Check the **Rule summary** and click **Create Rule** to save the settings.

Rule 4: Toggle a Relay Output with each Change in Connection Status

This rule is used to toggle a codec's control port relay output each time a program connects and disconnects.

1. Click the fourth rule in the **Rules panel**.

2. Click the drop-down **Relay** arrow and select the relay output you want to toggle.

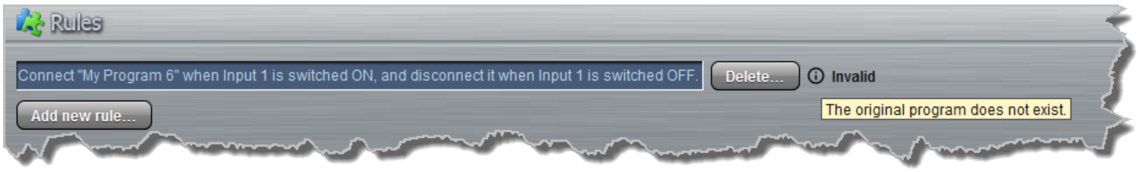


3. Click the drop-down **Program** arrow to select a specific program which will affect the relay toggle function, or use the default setting whereby any program will toggle the relay output.

4. Check the **Rule summary** and click **Create Rule** to save the settings.

Invalid Rules

Any rule in the **Rules panel** related to a deleted program is invalid. An **Invalid** error warning will be displayed when this type of rule exists.

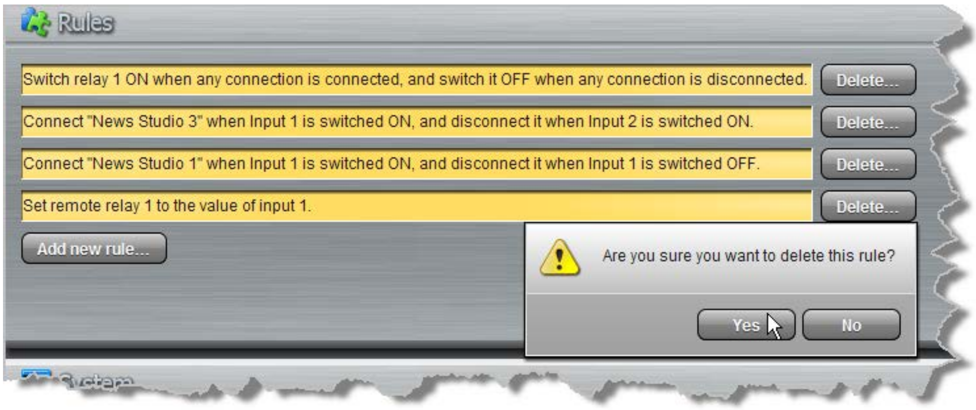


Deleting Rules

1. Click the **Rules**  button at the top of the web-GUI screen to open the **Rules panel**.

2. Click the **Delete** button next to the rule you want to delete.

3. Click **Yes** in the confirmation dialog.



16.19 Upgrading Codec Firmware

To download the latest codec firmware visit http://www.tieline.com/Support/Latest-Firmware.

Manual Firmware Upgrades

The following procedure explains how to perform codec firmware upgrades with a downloaded firmware file saved to your PC.

1. Click the **Settings**  button at the top of the web-GUI screen.
2. Click **Firmware** in the **Systems panel**.

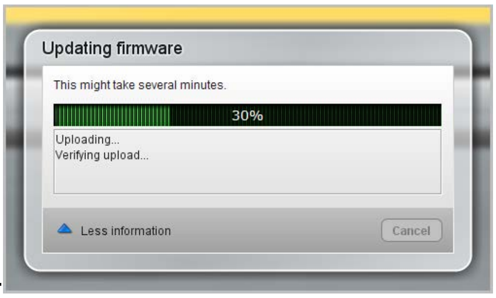
3. Click **Update from a selected file** and click the **Select File** button.
4. Select the **.bin** file you are using to perform the upgrade and click **Open**.
5. Press the **Update Firmware** button to commence the upgrade.

Automatic Firmware Upgrades

By default the web-GUI application integrates with TieServer to automatically update users when a firmware upgrade is available.

1. Connect your codec to your PC using either a LAN or USB connection and open the web-GUI (See Connecting to the Web GUI)
2. If new software is available the **Update** symbol appears in the top-left hand side of the screen.
3. Position your mouse-pointer over the **Update** symbol and click the update dialog when it appears to download the new software.

4. Click **More Information** in the **Updating firmware** dialog to display details of the upgrade process.

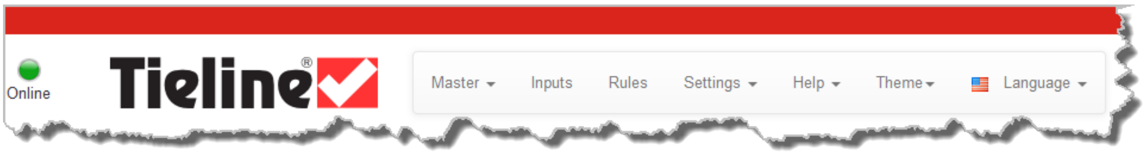


+

Important Note: Firmware upgrade files are very large and it is usually much quicker to download the file to your PC first and then upgrade the codec manually using the Manual Firmware Upgrade procedure.

17 HTML5 Toolbox Web-GUI Introduction

The following sections provide an overview of the different configuration panels available within the codec's HTML5 Toolbox Web-GUI. Navigate with the mouse pointer to the **Menu bar** at the top of the Web-GUI screen and click to select and open each panel in turn.

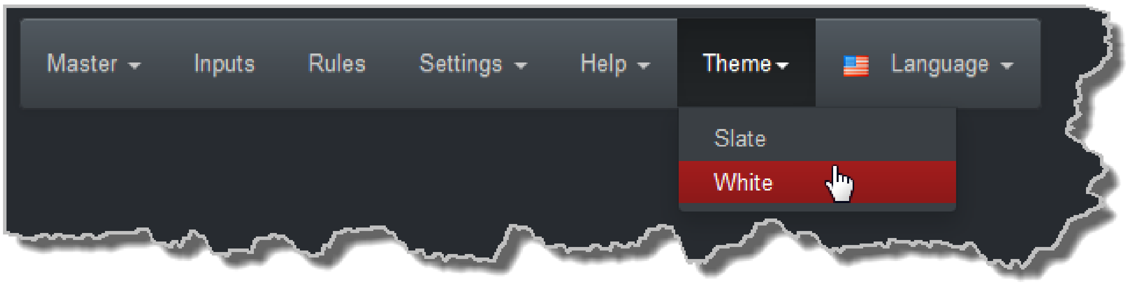


HTML GUI Menu Bar for Opening Panels

When you first open the HTML5 Toolbox Web-GUI the **Program Loader panel**, **Connections panel** and **PPMs panel** are loaded by default. If you retain cookies in your browser, any panels opened previously in the Web-GUI are automatically populated when you log in next. The default panel view is displayed on login if cookies have been cleared.

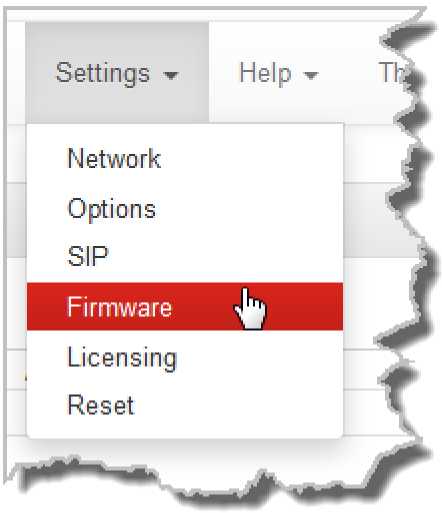
Adjusting the Theme

To adjust the **Theme** or 'skin' of the HTML5 Toolbox Web-GUI, navigate to the **Menu bar** at the top of the screen and click **Theme**, then click to select your preferred option. Note: this manual uses the **White** theme for most images.



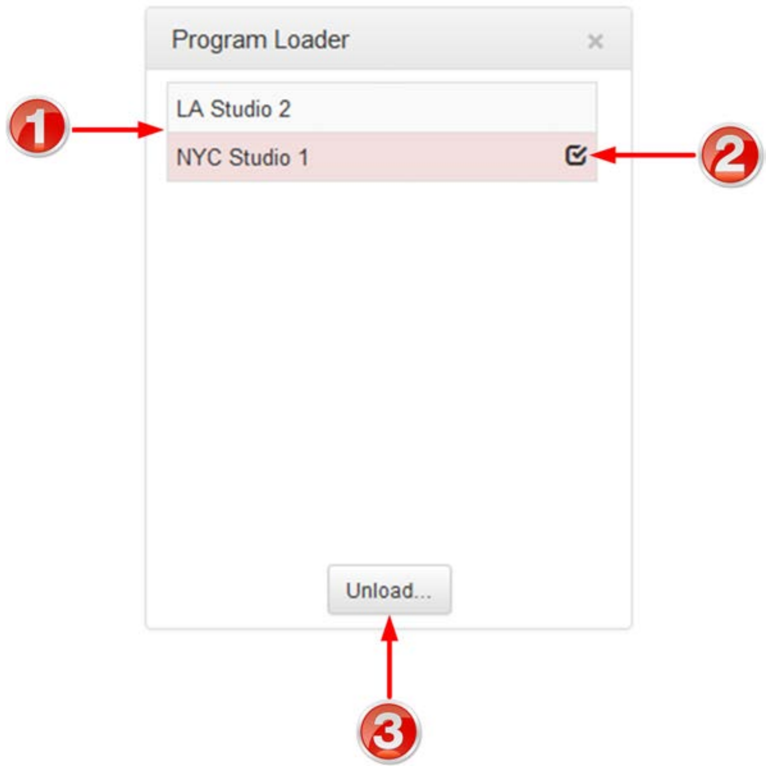
Opening a Panel & Adjusting Screen Position

Click an item in the **Menu bar** to display available panel options, then click to select and open a panel. New panels automatically open in the top left of the screen.

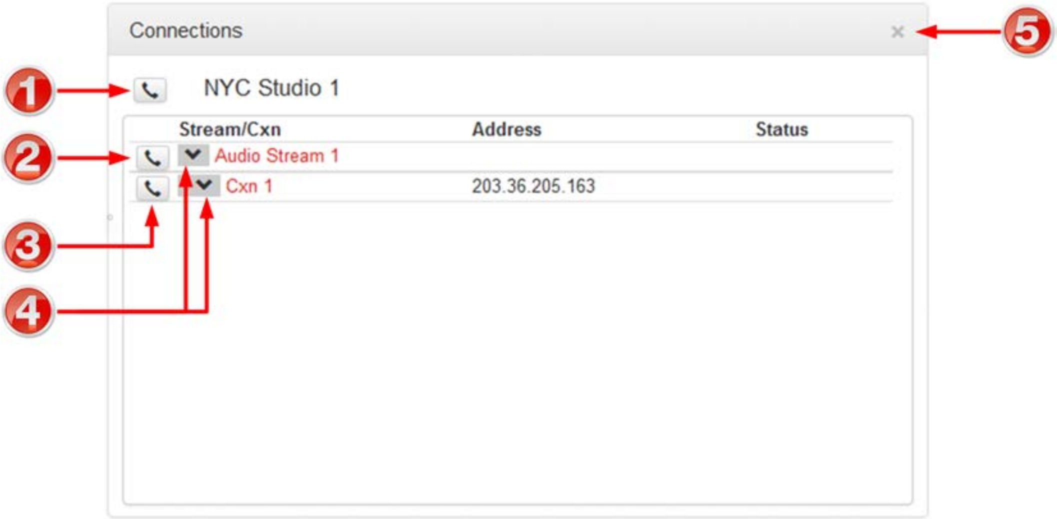


Position the mouse pointer over a panel's **Title bar** and click and drag to move a panel and reposition it in a preferred screen position.

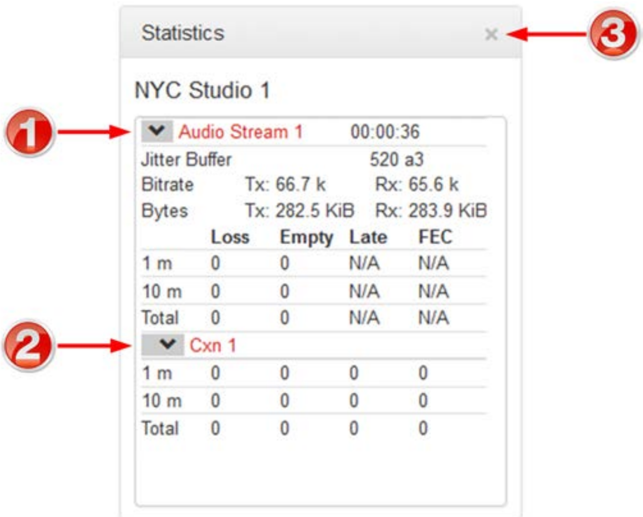
Master Panels: Load Programs & Manage Audio Streams



Program Loader Panel		
	Feature	**Description**
1	Programs list	Lists all configured programs which have been added into the codec. Click to select a program before loading.
2	**Close** button	Click to close the panel.
3	**Check-box symbol**	The **Check-box symbol** identifies the currently loaded program in the codec.
4	**Load** and **Unload** button	Click **Load** to load the currently selected program in the **Programs** list; the button changes automatically to **Unload** after loading, to allow unloading of a program when required.



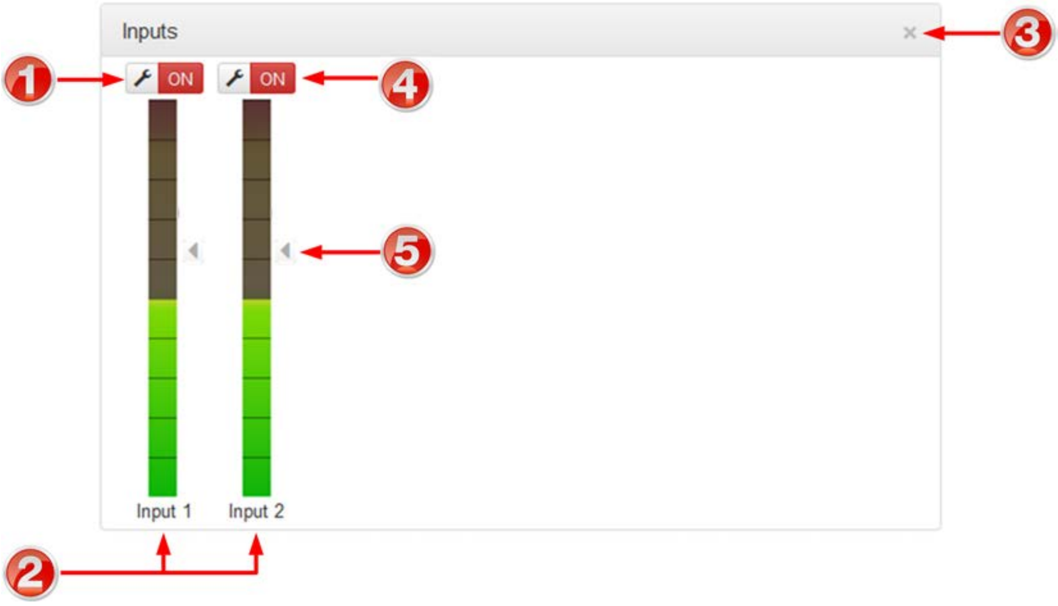
Connections Panel		
	Feature	Description
1	Program **Connect/ Disconnect** button	Click to connect/disconnect all audio streams in a program.
2	Audio Stream **Connect/Disconnect** button	Click to connect/disconnect all connections in an audio stream.
3	Connection **Connect/ Disconnect** button	Click to connect/disconnect an individual connection.
4	Show/Hide Arrow	Click to show/hide audio stream and connection details.
5	**Close** button	Click to close the panel.



Statistics Panel		
	Feature	Description
1	Show/Hide Arrow	Click to show/hide audio stream statistics.
2	Show/Hide Arrow	Click to show/hide individual connection statistics.
3	**Close** button	Click to close the panel.

PPMs Panel		
	Feature	Description
1	PPM Meters	4 PPM meters.
2	**Close** button	Click to close the panel.

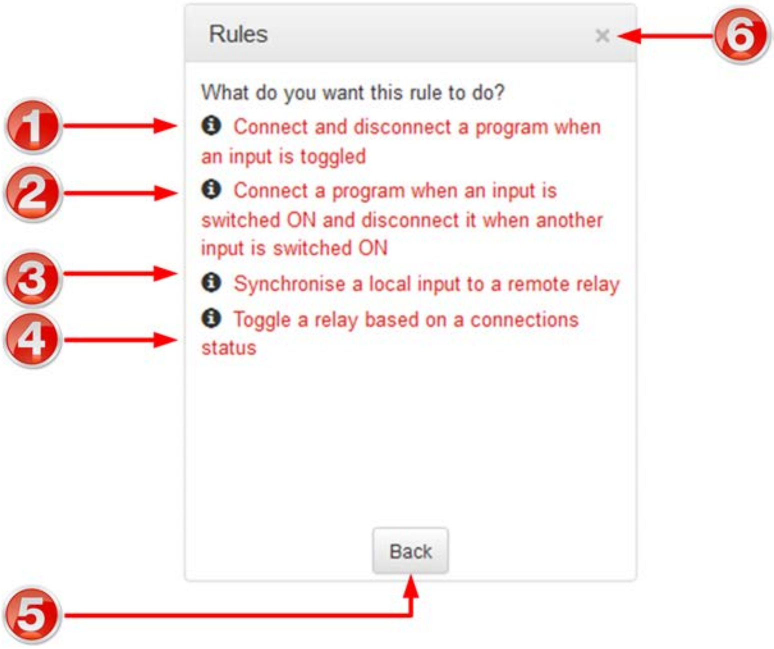
Inputs Panel for Input Adjustments



Important Note: Tieline codecs have different input configurations, therefore the image shown may not reflect the number of inputs displayed in your codec Web-GUI.

	Feature	Description
1	**Settings** button	Click to adjust input **Name**, **Type** and **IGC**.
2	Input PPM meter	Input PPM meter.
3	**Close** button	Click to close the panel.
4	**On/Off** button	Click to toggle an input on or off.
5	Input Sliders/Faders	Input gain control sliders/faders.

Rules Panel for Creating Relay Activation Rules

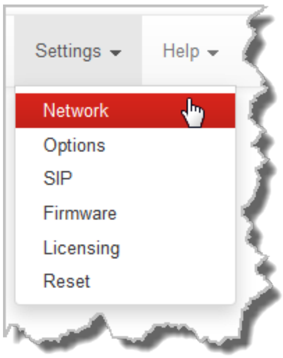


	Rule	Description
1	Connect and disconnect a program when an input is toggled	Click to configure connection and disconnection by toggling an input.
2	Connect when an input is switched **ON**; Disconnect when another input is switched **ON**	Click to configure connection and disconnection after different relay inputs are switched **ON**.
3	Synchronise a local relay input with a remote relay output	Click to configure a local relay input to synchronise with the state of a remote relay output.
4	Toggle a relay based on a connection's status	Click to configure a relay to toggle based on connection status.
5	**Back** / **Add New Rule** button	Click to add a new rule, or exit the rule creation function.
6	**Close** button	Click to close the panel.

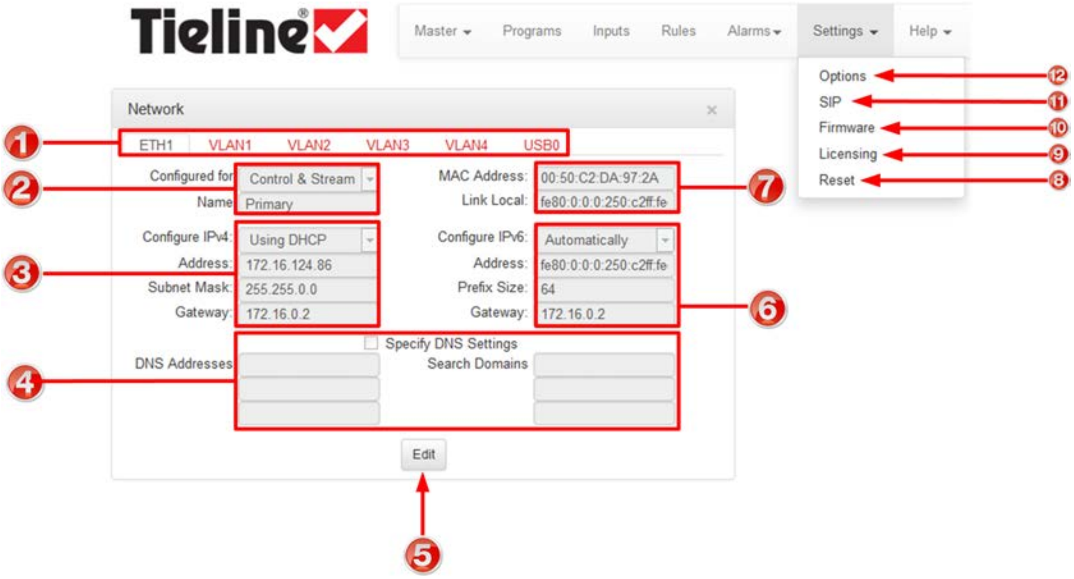
Settings Panels

There are 6 **Settings** panels which can be opened in the Web-GUI. Each panel provides specific codec configuration settings and options. Click to select and open each panel.

As an example, the **Network panel** is displayed with network interface configuration options. A brief description of the other panel options is also provided.



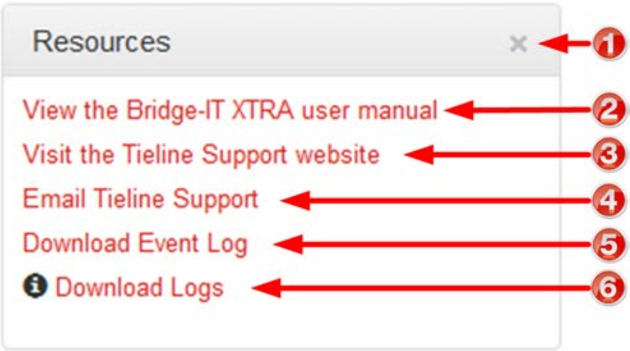
Settings panels



	Feature	Description
1	**Network tab**	Click to select and edit, or view network configuration settings for each Ethernet, VLAN and USB interface.
2	Network Interface	Control and streaming configuration options for each network interface.
3	IPv4 details	IPv4 address details and configuration.
4	DNS details	Select the check-box and specify DNS addresses and domains to search.
5	**Edit** / **Save** button	Click to edit Network settings, or save configured settings.
6	IPv6 details	IPv6 address details and configuration.
7	**MAC Address / Link Local**	Click to open the panel and view the device MAC address and IPv6 local network address created by the codec.
8	**Reset**	Click to open the panel; reset codec default settings and perform backup/restore of codec programs and settings.
9	**Licensing tab**	Click to open the panel; select a license file and install it in the codec.
10	**Firmware tab**	Click to open the panel; view software versions and perform an upgrade.
11	**SIP tab**	Click to open the panel and edit or view SIP configuration

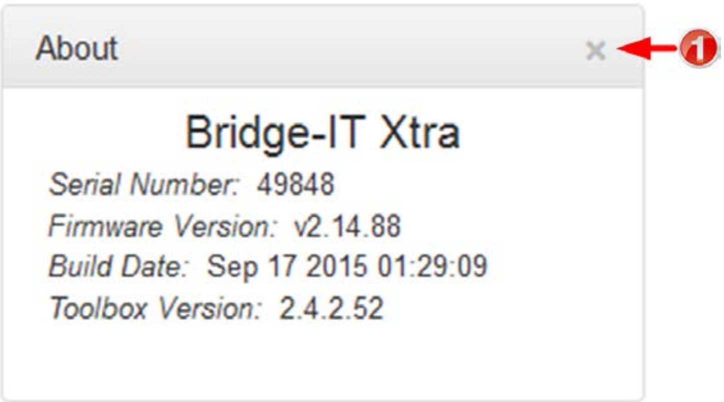
		settings.
12	**Options tab**	Click to open the panel; configure RS232 and QoS data settings, lock a loaded user Program and adjust Session Port settings and SNMP. Also configure the AES Output Clock sample rate.

Help Panels



Resources Panel		
	Feature	**Description**
1	**Close** button	Click to close the panel.
2	User manual link	Click to open the codec user manual in a new browser, or view support information (Note: the codec name displayed will vary by product type)
3	Support website link	Click to visit the support page on the Tieline website.
4	Email Support	Click to email Tieline support.
5	Event Logs	Click to download user-viewable event logs
6	Support Logs	Click to download diagnostic information that can be sent to Tieline support

The **About panel** provides details of the codec Toolbox and firmware version, as well as the codec serial number. Note: the codec name displayed will vary by product type.



About Panel		
	Feature	**Description**
1	**Close** button	Click to close the panel.

Language Selection

The HTML5 Toolbox Web-GUI offers language support for several languages.

1. Click on the **Language** drop-down menu arrow in the top right-hand corner of the Web-GUI page.
2. Select the preferred language to display.

18 HTML5 Toolbox Web-GUI Configuration

The following sections describe how to configure your codec using the HTML5 Toolbox web-GUI.

18.1 Using the HTML5 Toolbox Quick Connect Web-GUI

The HTML5 Quick Connect Web-GUI is designed for simple peer-to-peer connections and non-technical users. It has a reduced feature-set and allows users to:

1. Load existing programs in a codec via the **Program Loader panel** and then dial via the **Quick Connect panel**.
2. Use the **Quick Connect panel** to create and dial a simple peer-to-peer connection using IP/SIP.

Important Note: Simple peer-to-peer connections are not saved as programs with unique names. Details of the last ad hoc dial are retained in the **Quick Connect panel**, even after a program is loaded and unloaded using the **Program Loader panel**.

Launching the HTML5 Quick Connect Web-GUI

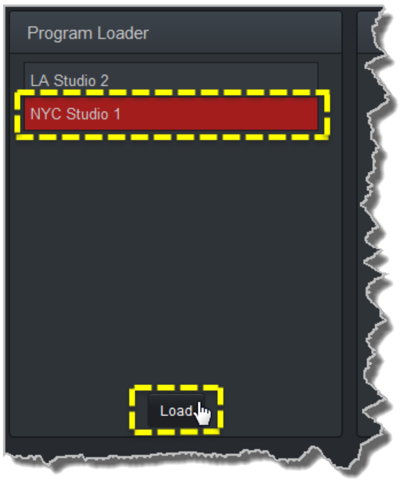
1. Type the codec IP address in your web-browser.
2. Click to launch the HTML5 Toolbox Quick Connect Web-GUI.
3. Enter the authentication **Password** for the codec and click **OK**.
4. The panels in the Quick Connect Web-GUI will automatically be displayed.

Important Note: To change the password using the codec front panel navigate to **Settings > WebGUI > Password** and press the button. Use the keypad to enter a new password and press the button to save the new setting.

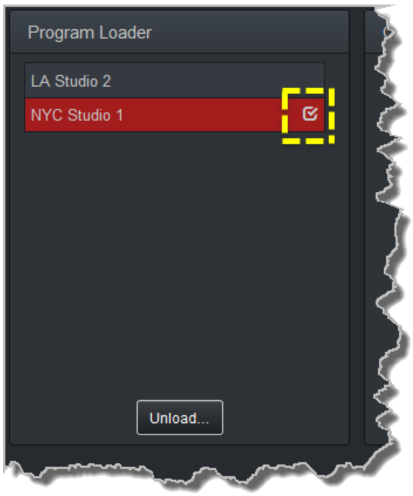
Loading and Unloading an Existing Program

If programs are saved in the codec they are displayed in the **Program Loader panel**.

1. Click to select a program in the **Program Loader panel** and click the **Load button** to load it in the codec.



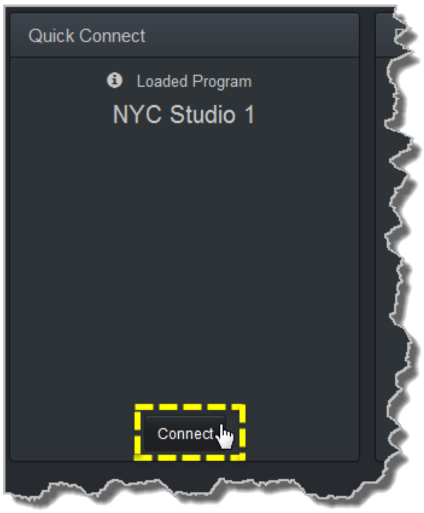
2. The **Check-box symbol** appears next to the program name to confirm it has been loaded and the **Load button** changes to an **Unload button**.



To unload a program click the **Unload button**.

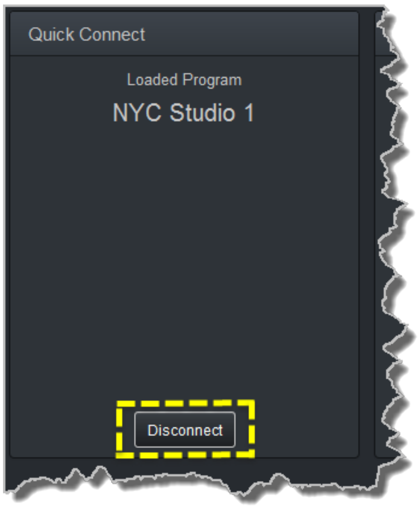
Dial a Loaded Program

1. Click the **Connect button** in the **Quick Connect panel** to dial a loaded program. Note: After connecting, the **Connect button** changes to a **Disconnect button**.

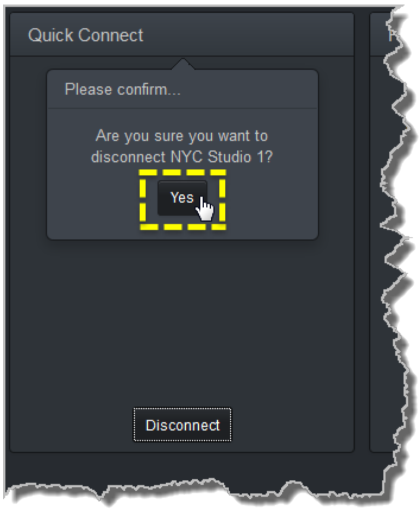


Disconnect a Loaded Program

1. Click the **Disconnect button** in the **Quick Connect** panel.



2. Click **Yes** in the confirmation dialog to disconnect the connection.

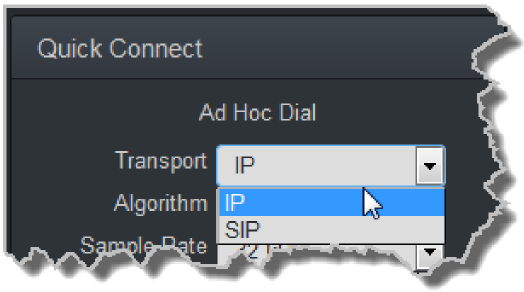


Dial Peer-to-Peer over IP with Quick Connect

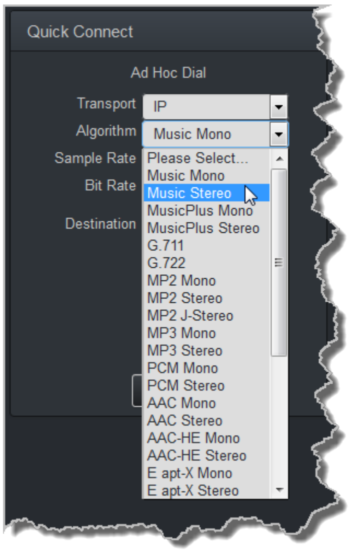
Important Notes:

- Click the **Unload button** in the **Program Loader panel** if a program is currently loaded.
- The transcriptor algorithm is for closed captioning and not normal broadcast configurations.

1. Click the drop-down **Transport** menu arrow in the **Quick Connect panel** and select **IP**.

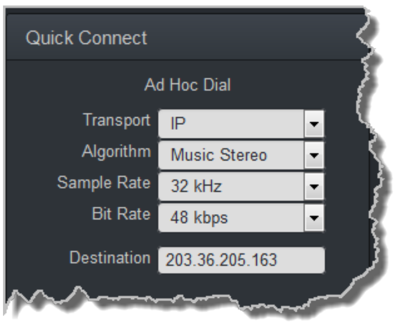


2. Click the drop-down **Algorithm** menu and select an algorithm.



3. Click the select the appropriate **Sample Rate** and **Bit Rate** for the connection. Note: If only one sample rate is available this will be automatically selected.

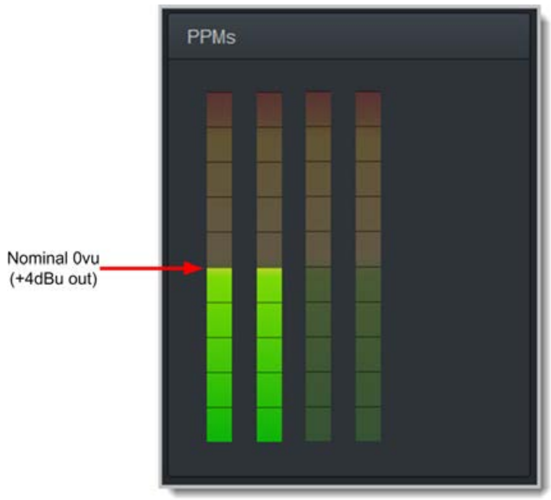
4. Click in the **Destination** text box and enter the IP address of the destination codec.



5. Click the **Connect button** to dial.

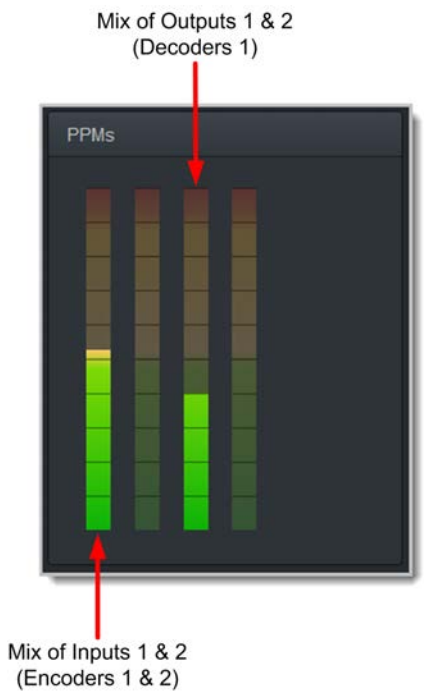
Monitoring PPMs

Set audio levels so that audio peaks average at the nominal 0vu point indicated below on the PPM meters. This represents a program level of +4 dBu leaving the codec. Audio peaks can safely reach +22 dBu without clipping, providing 18dBu of headroom from the nominal 0vu point. Note: the audio metering reference scale is automatically adjusted by default when a Merlin codec connects to a Commander G3 codec. The G3 metering scale is between -11dBu and +18dBu.

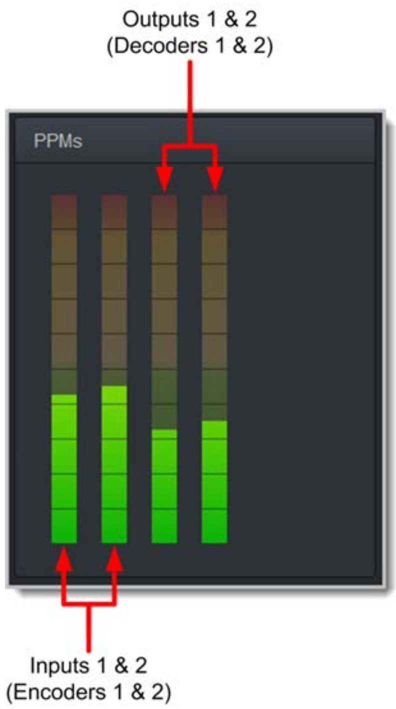


Mono and Stereo PPM Metering

When connected with a mono program the codec will display a mix of inputs 1 and 2 on **PPM1**. **PPM 3** displays the level of return audio.



When connecting with a stereo program, the codec displays audio on **PPM1** and **2** for inputs 1 and 2 and **PPM 3** and **4** for the return audio.



18.2 Configuring IP Settings

Open the HTML5 Toolbox Web-GUI and click **Settings** and then click **Network** to view and configure Ethernet and VLAN interface settings in the Web-GUI.

Important Note: For assistance with configuration of IPv4 or IPv6 network connections contact your IT Administrator.

IPv4 versus IPv6

An IP address is a unique address to identify a device on a TCP/IP network. Your codec uses dual IP protocol stacks to allow your codec to work on both IPv4 and IPv6 networks. Tieline codecs support both DHCP (default) IP addressing and static IP addresses for dialing IPv4 connection endpoints.

If you want to dial a codec with a public IP address you simply dial the IP address to connect. If you want to dial a codec with a private IP address you need to perform network address translation (NAT). NAT allows a single device, such as a broadband router, to act as an agent between the public internet and a local private LAN. Usually this will be set up at the studio end so you can dial into the studio from the remote codec.

Support for IPv6 connections allows you to use IPv6 infrastructure to connect to other codecs globally.

Configuring Ethernet Ports and VLANs

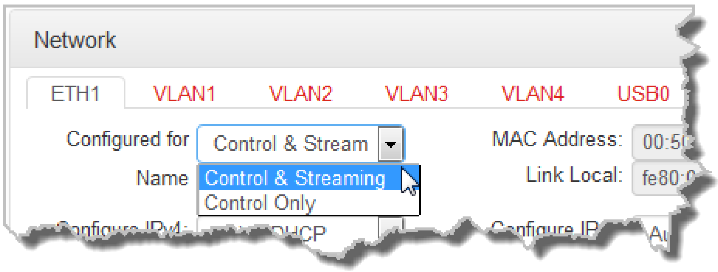
The codec features a physical Ethernet port and up to four additional VLAN interfaces.

VLAN interfaces have features similar to physical Ethernet interfaces. However, your network administrator will need to configure VLAN support throughout your network for them to be supported in your codec.

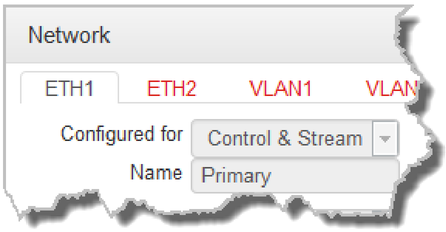
As an example, VLANs can be used to separate codec Control and Streaming functions if required.

Ethernet and VLAN interfaces can be configured for:

- Control Only: codec control and command only from the Ethernet port.
- Control and Streaming: stream audio and control and command the codec via the Ethernet port.
- Streaming audio: stream audio only from an Ethernet port (**VLANs** only).
- Nothing: Disable the Ethernet port from streaming audio and codec command and control (**VLANs** only).

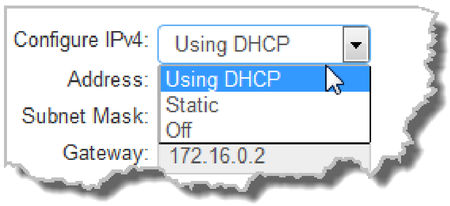


The **Name** text box, e.g. **Primary** or **Secondary**, is an interface identifier used when configuring new programs via the **Programs panel**.

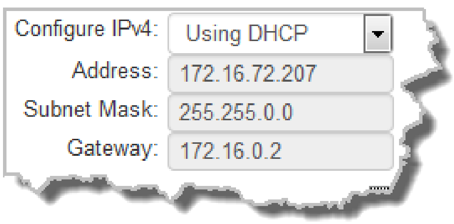


IPv4 Address Configuration

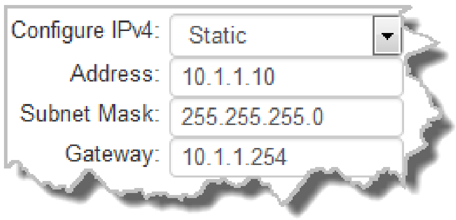
Click the **Edit** button in the **Network** panel to configure settings. The codec is capable of automatic DHCP address assignment, or manually configured static IPv4 address configuration via the drop-down **Configure IPv4** menu. If you want to ignore IPv4 settings select **Off**.



DHCP IP addresses are automatically assigned and can change each time you connect to your Internet Service Provider, or to your own local area network (LAN). By default the codec is configured for DHCP-assigned IP addresses.



Static IP addresses are fixed addresses that are recommended for studio installations, so that IP address dialing remains the same over time for incoming codec connections.



Click **Save** to store all configuration settings.

Note: The **Subnet Mask** is used by the TCP/IP protocol to determine whether a host is on the local subnet or on a remote network. The default **Gateway** is the router linking the codec's subnet to other networks. See your IT administrator for more details.

IPv6 Address Configuration

An IPv6 address is represented by 8 groups of 16-bit hexadecimal values separated by colons (:). The drop-down **Configure IPv6** menu provides three address configuration options:

1. **Automatically**: An address is automatically assigned to the codec when you connect the codec to an IPv6 router. This process is similar to how an IPv4 DHCP address is assigned.
2. **Manually**: Select to enter static IPv6 address details.
3. **Off**: Select to ignore IPv6 address details.

Important Note: Select **Off** in the drop-down **Configure IPv6** menu if you are not using IPv6 to connect to another device. This ensures your codec will attempt to connect using IPv4 at all times.

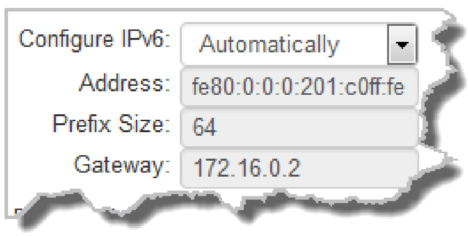
Types of IPv6 Addresses

There are two types of addresses displayed in the IPv6 section:

1. IPv6 address (normally global): A router-allocated IP address with 'global' visibility, details of which are displayed in the **Address, Prefix size** and **Gateway** text boxes.
2. Link Local: A local address which can only be used to connect to another device directly over a LAN. This address is allocated by the codec internally based on MAC address details.

Auto Address Assignment

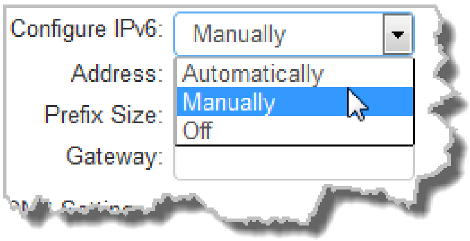
1. Click the **Edit** button in the **Network** panel to configure settings.
2. By default the codec is configured to connect to an IPv6 router which automatically allocates IPv6 address details, as displayed in the following example.



3. Click **Save** to store all configuration settings.

Manual IPv6 Address Assignment

1. Click the **Edit** button in the **Network** panel to configure settings.
2. To configure IPv6 address details into the codec manually, select **Manually** and enter details into the **Address**, **Prefix** and **Gateway** text boxes.

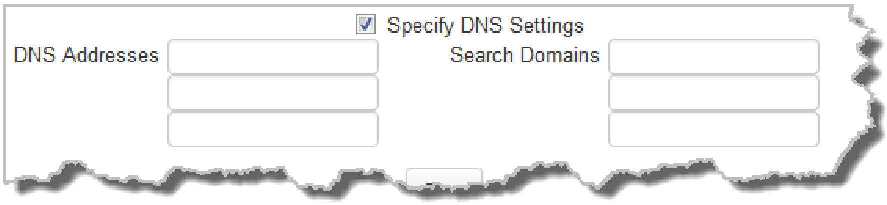


3. Click **Save** to store all configuration settings.

Specifying DNS Settings

It is possible to specify Domain Name Server (DNS) settings to allow easy look up of codecs within the specified **DNS Addresses** or **Domains**.

1. Click the **Edit** button in the **Network panel** to configure settings.

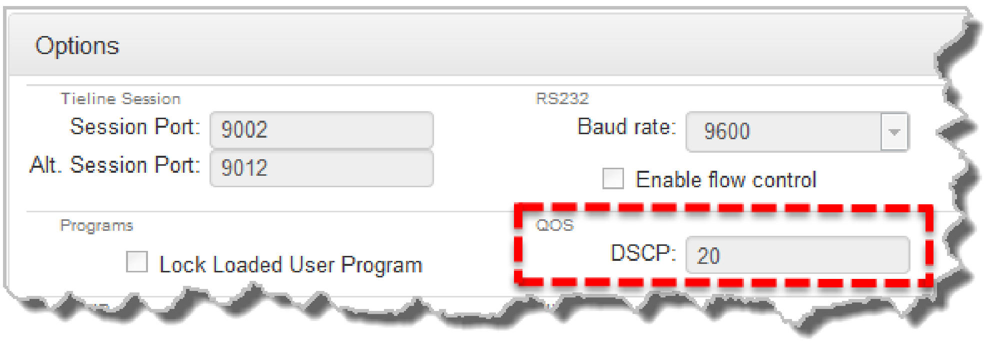


2. Click **Save** to store all configuration settings.

The codec can be configured to tag IP data packets sent across a network by entering a value into the Differentiated Services Code Point (**DSCP**) field within the header of data packets transmitted over the network.

Configuring QoS

1. Open the HTML5 Toolbox web-GUI in a browser on your PC.
2. Click **Settings** at the top of the screen and then click **Options** to display the **Options panel**.
3. Click the **Edit** button at the bottom of the panel.
4. Click in the **QoS DSCP** text box and enter the preferred value.



5. Click the **Save** button at the bottom of the panel to save the new setting.

Important Note: Check with your IT administrator before changing this setting. By default the codec is programmed for Assured Forwarding and more details about DSCP are available on Wikipedia at http://en.wikipedia.org/wiki/Dscp. For more information on configuring QoS see Configuring QoS for Broadcasts in this manual.

18.3 Configuring Input/Output Settings

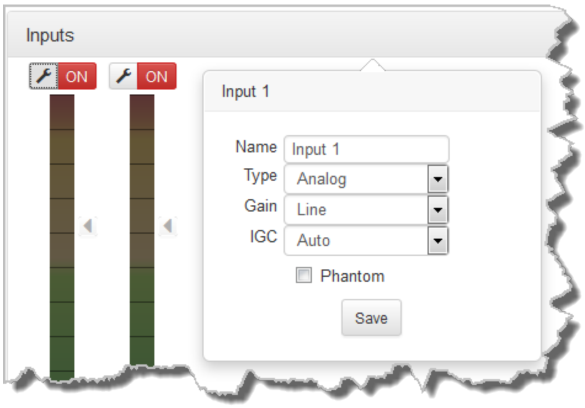
Open the HTML5 Toolbox Web-GUI and click **Inputs** to display the **Inputs panel**.

Important Note: 15 volt phantom power can only be supplied on analog input 1; this is disabled by default.

Configuring Input Channel Settings

Renaming Input Channels:

1. Click the **Input Settings** symbol on the input you want to rename.
2. Click in the **Name** text box to enter a new name, or edit an existing name.

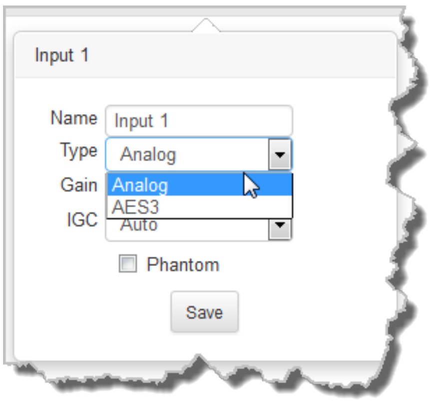


3. Click **Save** to confirm the name change.

Selecting Analog and Digital Audio Sources:

Codec inputs are configured for analog line level audio sources by default.

1. Click the **Input Settings** symbol.
2. Select **Type** and click to select either **Analog** or **AES3**.

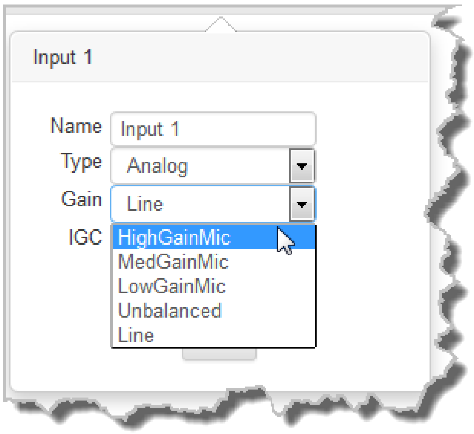


3. When you select AES3 each input is automatically configured for 100% input levels;

input level and input on/off controls are removed and unable to be adjusted. See Configuring AES3 Audio for more information about the digital inputs and outputs.

Adjusting Analog Audio Levels

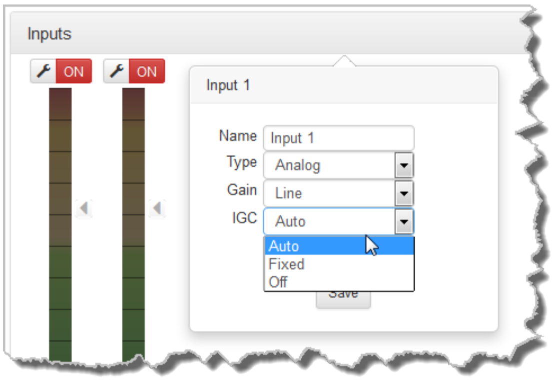
Gain on **Input 1** can be configured for mic or line level sources. **Input 2** accepts line level only.



Audio levels on the **Input panel** should be set to ensure audio peaks average at the first yellow indications on the PPM meters, which represents +4dBu. These levels should also be checked against the **Input PPM Meters** on the **Master panel**.

Other Input Controls

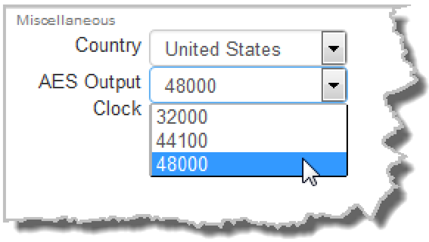
Adjust the **IGC** (Intelligent Gain Control) input settings to **Auto**, **Fixed** or **Off** as required.



AES3 Output Sample Rate Configuration

The AES3 output sample rate can be configured using the HTML5 Toolbox Web-GUI.

1. Open the HTML5 Toolbox Web-GUI and click **Settings**, then click **Options** to open the **Options panel**.
2. Click the **Edit** button to configure settings.
3. Click the **AES Output Clock** drop-down menu to select your preferred **AES Output Clock** setting, then click **Save**.



18.4 Configure SIP Settings

The codec is fully EBU N/ACIP Tech 3326 compliant when connecting using SIP (Session Initiation Protocol) to other brands of IP codecs.

About SIP

SIP provides superior interoperability between different brands of codecs due to its standardized protocols for connecting devices and is intended to be used when connecting Tieline codecs to non-Tieline devices. Devices primarily use SIP to dial another device's SIP address and find its location with a minimum of fuss. This task is usually performed by SIP servers, which communicate between SIP-compliant devices to set up a call.

When connecting two devices, SDP performs similar tasks to Tieline's proprietary session data, which is used to configure all non-SIP IP connections. There are two very distinct parts to a call when dialing over IP. The initial stage is the call setup stage and this is what SIP is used for. The second stage is when data transference occurs and this is left to the other protocols used by a device (i.e. using UDP to send audio data).

All the mandatory EBU N/ACIP 3326 algorithms are supported (G.711, G.722, MPEG-1 Layer 2 and 16 bit PCM), as well as optional algorithms including LC- AAC, HE-AAC and aptX Enhanced. The default algorithm selected when connecting using SIP is G.711.

Important Notes:
- Each codec should be registered to a different SIP server account to avoid connection conflicts.
- SIP account registration can only be configured via Ethernet port 1.
- SIP dialing is only supported over point-to-point connections, not multi-unicast connections.
- Tieline G3 codecs do not support connections using AAC and will default to MPEG Layer 2 if an incoming call is programmed to use this algorithm.
- Failover and SmartStream PLUS redundant streaming are not available with SIP connections.
- When connecting to a Tieline G3 codec using SIP you need to manually select the G3 audio reference level in the codec. To do this select **SETTINGS** **> Audio > Ref Level > Tieline G3**. In addition, select the following on the G3 codec prior to dialing.
  - Select either a mono or stereo profile
  - Select **[Menu] > [Configuration] > [IP1 Setup] > [Session Type] > [SIP]**
  - Select **[Menu] > [Configuration] > [IP1 Setup] > [Algorithm] > [G711/G722 or MP2]**

SIP Server Connections: Getting Started

Registering codecs for SIP connectivity is simple. First, choose the SIP server that you wish to register your codec with. On a LAN this may be your own server, or it could be one of the many internet servers available. We recommend that you use your own SIP server and configure it to use

G.711, G.722, MP2 and AAC algorithms. This is because most internet SIP servers are for VoIP phones and are only configured for G.711 and GSM algorithms.

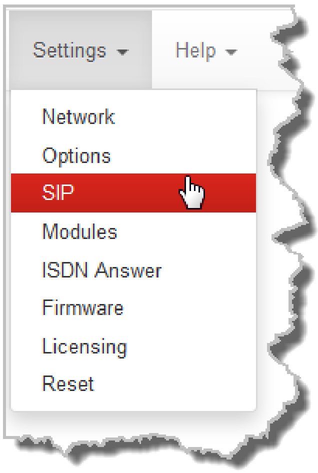
When you register an account with a SIP server you will be provided with:

- The SIP server IP address.
- A username (often the same as a SIP number).
- A password.
- Domain details.
- Realm details (sometimes).

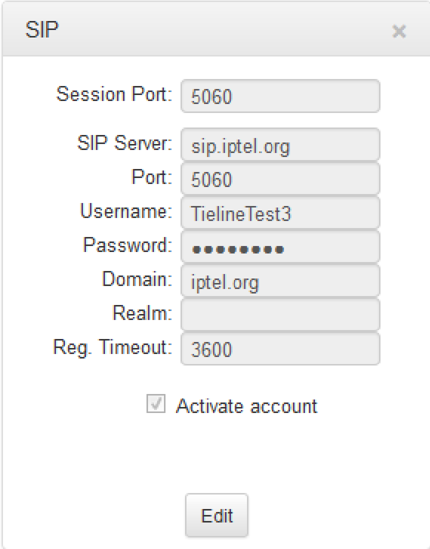
Configure the Codec for SIP using the Web-GUI

Use the HTML5 Toolbox Web-GUI to configure SIP account registration details in your codec. Once these details have been entered into the codec, each time it is connected to a public IP address it will contact the SIP server automatically to acknowledge its presence over a wide area network.

1. Connect your codec to a LAN connection with a public IP address, then login to the HTML5 Toolbox Web-GUI.
2. Click **Settings** at the top of the screen and then click **SIP** to display the **SIP panel**.



3. Click the **Edit** button to configure settings.
4. Enter the account details into the relevant text boxes.
5. Enter the **Registration Timeout** (this shouldn't need to be adjusted from the default setting).
6. Click to select the **Activate Account** check-box and click **Save** to create the account in the codec.



7. Navigate to **SETTINGS** > **SIP > Accounts** to verify that the account has been registered to the SIP server. The registration symbol appears when the account has been activated successfully.

Important Notes: Some ISPs may block SIP traffic over UDP port 5060.

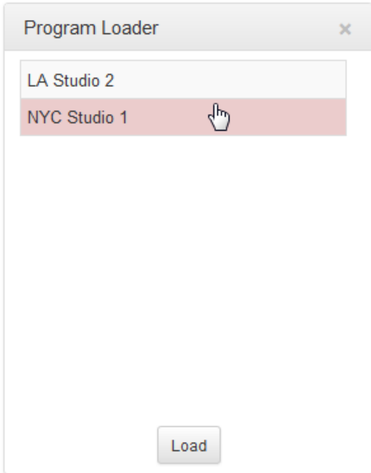
18.5 Load, Unload and Dial a Program

To load and dial a program it is necessary to open the **Program Loader panel** and **Connections panel**. Use the **Program Loader panel** to load a program and then manage connecting and disconnecting using the **Connections panel**.

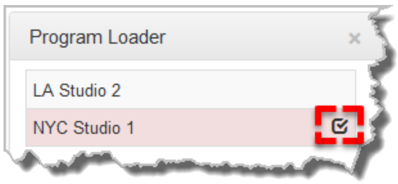
Important Notes: The HTML5 Toolbox Web-GUI currently does not support the creation of new programs. Use the Java Toolbox Web-GUI to create a new program.

Loading a New Program

1. Open the HTML5 Toolbox Web-GUI and click **Master** and then click **Program Loader** to open the **Program Loader panel**.
2. Click to select a program in the **Program Loader panel** and then click **Load** to load the program in the codec.



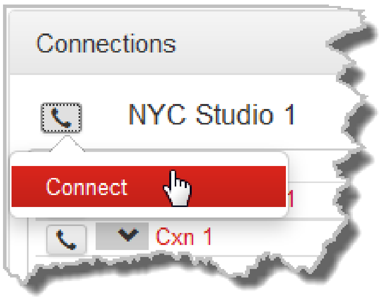
Note: the currently loaded program has the check-box symbol displayed next to its name.



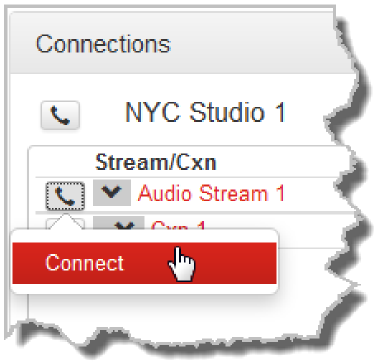
Connecting a Program

To connect audio streams and connections within an existing program there are three options:

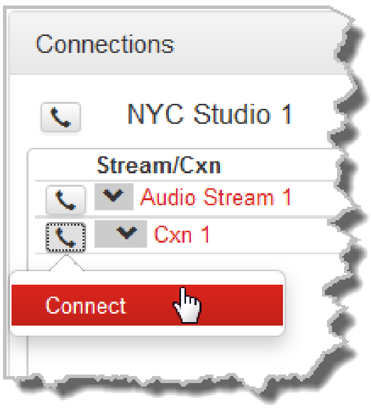
1. Click the program **Connect/Disconnect** symbol and then click **Connect**; this connects all active audio streams and connections associated with the program.



2. Click the audio stream **Connect/Disconnect** symbol and then click **Connect**; this connects all connections associated with this audio stream.



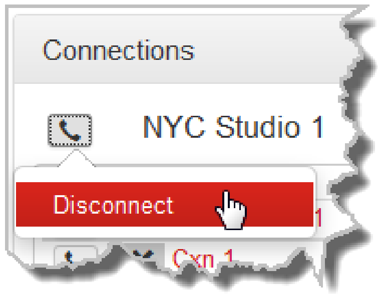
3. Click the connection **Connect/Disconnect** symbol and then click **Connect**; this connects an individual audio stream connection.



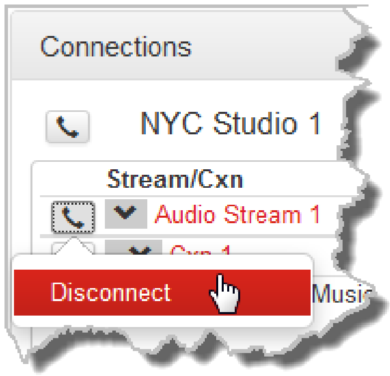
Disconnect a Program

To disconnect audio streams and connections within an existing program there are three options:

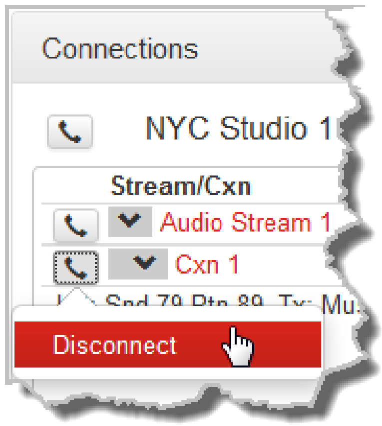
1. Click the program **Connect/Disconnect**  symbol and then click **Disconnect** to disconnect a program; this includes all audio streams and connections associated with the program.



2. Click the audio stream **Connect/Disconnect**  symbol and then click **Disconnect** to disconnect an individual audio stream and all associated connections.

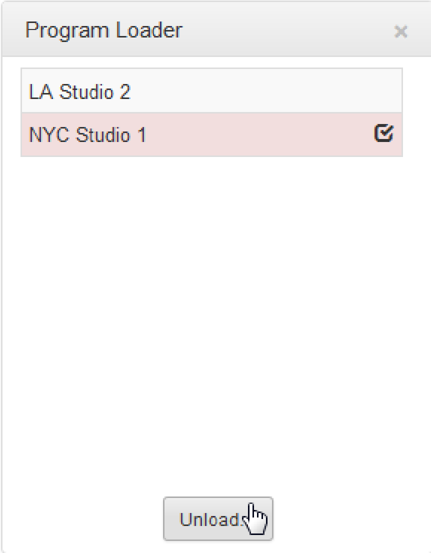


3. Click the connection **Connect/Disconnect**  symbol to disconnect an individual audio stream connection.



Unloading a Program

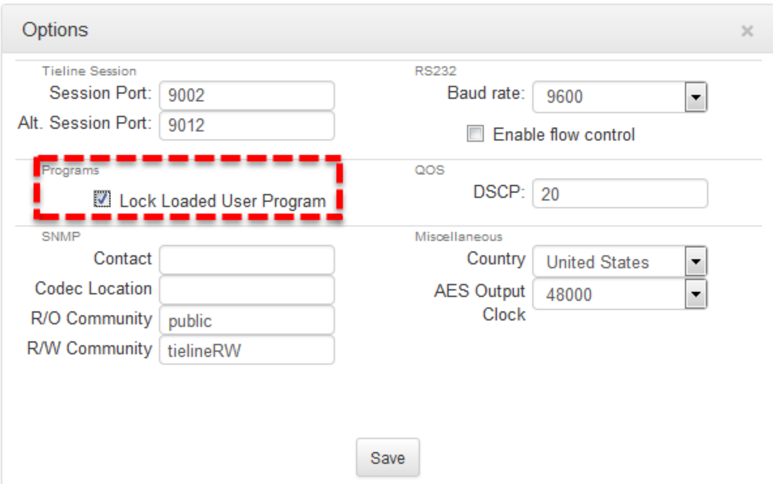
1. Click to select the loaded program in the **Program Loader panel** and then click **Unload** to unload the program in the codec.



18.6 Lock or Unlock Programs

It is possible to lock a loaded custom program in a codec to ensure the currently loaded program type, e.g. mono, cannot be unloaded by a codec dialing in with a different program type, e.g. stereo. For example, if you require the codec at the studio to always connect in mono, simply load and lock a mono program in the codec. Generally programs will be up or down-mixed by the answering codec to match the loaded program type. In some situations incompatible program types will be rejected.

1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Options** to display the **Options panel**.
2. Click the **Edit** button to configure settings.
3. Click the **Lock Loaded User Program** check-box to lock or unlock a user program in the codec.



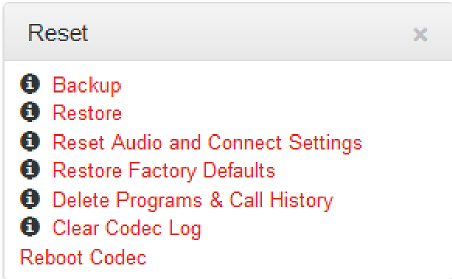
4. Click **Save** to store the new configuration.

Important Note:
- A black **Padlock** symbol appears next to the program name in the **Connections panel** and in the **Program Loader panel**, to indicate a program is locked in the codec.
- It is only possible to lock custom programs in a codec.
- If **Lock Program** is enabled and you load a new custom program in the codec, **Lock Program** remains enabled and locks the most recently loaded custom program.

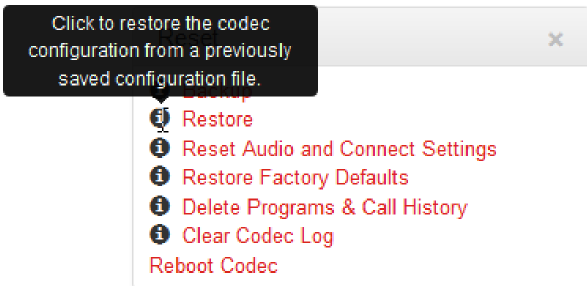
18.7 Reset Factory Default Settings

There are several options which allow you to restore factory default settings within the codec. See Reset and Restore Factory Defaults for more details on each option.

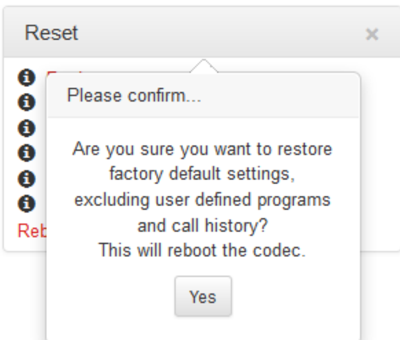
1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Reset** to display the **Reset panel**.



2. Click one of the available reset options to adjust codec settings, or reboot the codec. Note: Hover with the mouse pointer over the **Information** symbol to view a tool-tip for each reset option.



4. A confirmation dialog appears for each option; click **Yes** to proceed.



18.8 Backup and Restore Functions

The HTML5 Toolbox Web-GUI can be used to backup and restore codec settings, including:

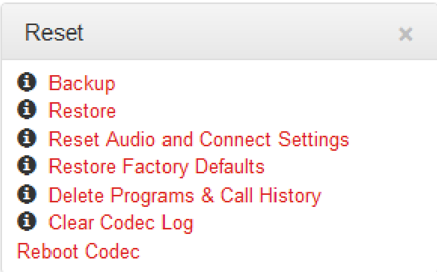
- Programs containing a variety of connection settings.
- All system settings that have been adjusted to change the factory default codec settings (current runtime settings).

Files can also be used to copy configurations onto other similar codecs. Programs are essentially connection profiles that may include:

- Program, audio stream and connection names.
- IP address, port, algorithm, jitter buffer, FEC and bit rate settings (etc.) for audio stream connections.

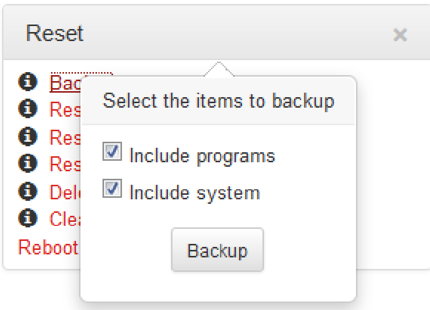
Creating Backup Files

1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Reset** to display the **Reset panel**.



2. Click **Backup**.

3. Click to select the check-boxes to confirm your backup requirements, then click **Backup**.



4. Select a location on your PC to save the configuration file. Note: You may need to "allow" your browser to display the pop-up dialog.

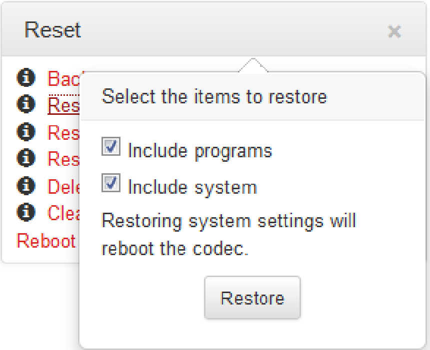
Restoring Configuration File Settings

1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Reset** to display the **Reset panel**.

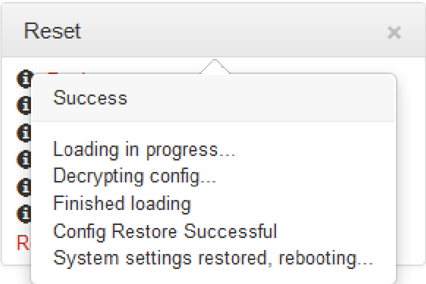
2. Click **Restore**.

3. Click to select the check-boxes and confirm your restore settings. For example, you could select the **Include programs** check-box and deselect the **Include system** check-box if you are only

copying programs onto codecs.



4. Click **Restore** and select the .tgz file you want to load onto the codec. A **Success** dialog confirms the files have been restored.



Note: The codec will automatically reboot if you restore system settings.

18.9 HTML5 Software License Installation

Prior to installation you will need connect your codec to a PC and ensure you are connected to the internet. You should also check that you have received notification by email from Tieline that your new license file is ready to download from TieServer.

Perform an Automatic Software License Install with the HTML5 Toolbox Web-GUI

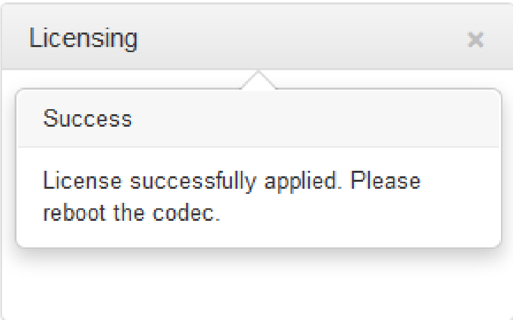
1. Open the HTML5 Toolbox web-GUI in a browser on your PC by typing either the IP address of the codec (LAN connection), or the USB address of the codec (USB connection) into the address bar.
2. Ensure you have unloaded any currently loaded program in the codec via the **Program Loader panel**.
3. Click **Settings** at the top of the screen and then click **Licensing** to display the **Licensing panel**.
4. Click **Get license file from TieServer**.
5. A **Success** dialog in the web-GUI **Licensing panel** confirms when installation is complete and the codec screen should display a confirmation message.

6. Reboot the codec via **Settings > Reset > Reboot Codec** and press the OK button. Note: do not reboot by removing the power cable from the codec.

Download a License File and Install Manually

Prior to installing any new software license you will need to connect your codec to a PC and save the license file on this computer.

1. Open the HTML5 Toolbox web-GUI in a browser on your PC.
2. Click **Settings** at the top of the screen and then click **Licensing** to display the **Licensing panel**.
3. Click **Upload a selected file**.
4. Navigate to the ".lcf" license file on your PC, then click the **Open** button to commence license installation.
5. A **Success** dialog in the web-GUI **Licensing panel** confirms when installation is complete and the codec screen should display a confirmation message.



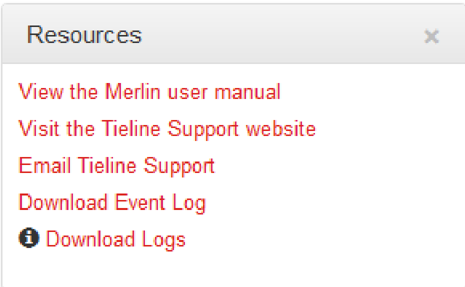
6. Reboot the codec via **Settings > Reset > Reboot Codec** and press the OK button. Note: do not reboot by removing the power cable from the codec.

18.10 Download Logs

The codec is capable of providing diagnostic information via user logs, which can either be sent to Tieline support, or downloaded for user diagnostics.

Procedure for Sending Logs to Tieline

1. Open the HTML5 Toolbox Web-GUI and click **Settings** in the **Menu Bar**, then click **Help** to display the **Help panel**.
2. Click **Download Logs**.

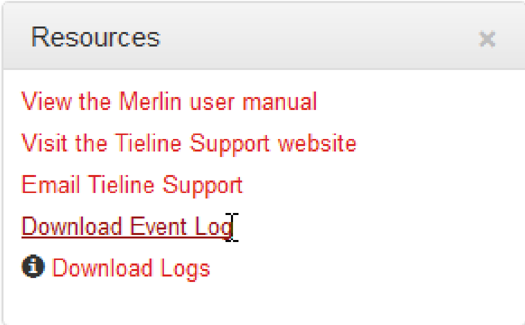


3. Save the file to your computer and then send it as a .zip file to Tieline support via support@tieline.com

Download Event Logs

Event logs can be downloaded from the codec and viewed in your browser.

1. Open the HTML5 Toolbox Web-GUI and click **Settings** in the **Menu Bar**, then click **Help** to display the **Help panel**.
2. Click **Download Event Log** to view the log in a new web-browser window.



Clearing Logs

This option should only be used if instructed to by Tieline support staff. To clear all event and other logs in the codec via the front panel, see the Reset and Restore Factory Default Settings section of this manual, or see Reset Factory Default Settings to clear recent log history using the Web-GUI.

18.11 RS232 Data Configuration

The codec supports both in-band and out-of-band data depending on the connection transport and algorithm you are using. RPTP data is automatically enabled when using the Tieline Music or MusicPLUS algorithms over any transport. Over IP it is also possible to enable synchronized out-of-band data using any algorithm.

Algorithm Selected	IP Transport
Tieline Music and MusicPLUS	• In-band RPTP data enabled automatically • Synchronized out-of-band data can be enabled and disabled
All other algorithms	• Synchronized out-of-band data can be enabled and disabled

The codec can be connected to external devices and send RS232-compatible data via the serial port on the rear panel of the codec. To enable RS232 data within a connection, select **Enable Auxiliary Data** when creating a program in the **Programs panel** wizard.

Setting RS232 Data Rates and Flow Control

1. Open the HTML5 Toolbox Web-GUI and click **Settings** in the **Menu Bar**, then click **Options** to display the **Options panel**.
2. Click the **Edit** button to configure settings.
3. Click the **Baud rate** drop-down menu arrow to select the serial port baud rate which matches the baud rate of the external device connected to the RS232 port on the codec.
4. Click to select the **Enable flow control** check box and enable flow control, then click **Save** to store the new settings.

Important Notes:

- When connecting over IP to G3 codecs only in-band data is available via the Music and MusicPLUS algorithms.
- Use firmware higher than 2.8.xx in the Bridge-IT, Genie and Merlin families of codecs to enable auxiliary data over multicast connections.
- It is important to enable serial port flow control as it regulates the flow of data through the serial port. If disabled, data will flow unregulated and some may be lost.
- Ensure you configure the serial port baud rate to match the setting of the external device to which you are connecting. Ideally the settings on both codecs should match, or you could have data overflow issues.
- Only the dialing codec needs to be configured to send RS232 data. Session data sent from the dialing codec will configure all other compatible codecs (non-G3) when you connect.
- RS232 data can be sent from the dialing codec to all endpoints of a multi-unicast or multicast connection if your codec is capable of these connections. Note: Bidirectional RS232 data is only available on the first connection dialed when multi-unicasting.

18.12 Creating Rules

The **Rules panel** in the Toolbox Web-GUI is used to program events for specific codec actions. Typically these 'rules' are based on a change in the state of a GPIO control port or the codec being connected or disconnected. Rules can only be created with the Web-GUI while the codec is disconnected.

Important Note: Data transmission is disabled by default. **Data** must be enabled in the **Connection** menu to enable contact closure operation and RS232 data.

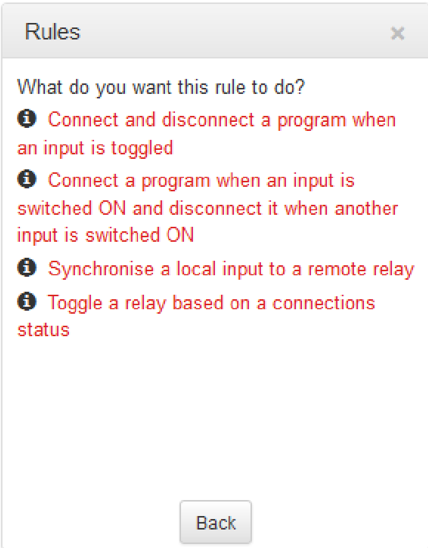
1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select your preferred **IP Mode** and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Navigate to **Data** and press to toggle between **Enabled** and **Disabled**.

Configuring Rules

Default rules have been preprogrammed into the codec to facilitate programming the most common events required by broadcast engineers. To view rules options:

1. Open the Toolbox HTML5 Web-GUI and click **Rules** in the **Menu Bar** to display the **Rules panel**.
2. Click **Add New Rule**.

3. Click to select the appropriate rule for your requirements. See the Rules panel section in [Using the Toolbox HTML5 Web-GUI] for an explanation of the action each rule can perform.

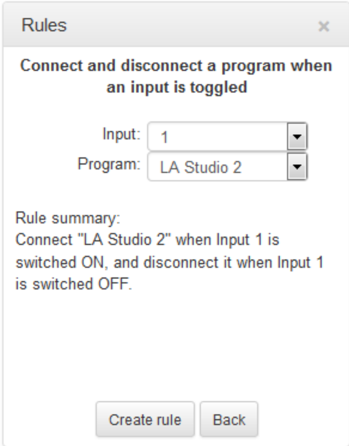


Note: When rules have been configured previously they are displayed when the **Rules panel** is opened.

Rule 1: Toggle a Control Port Input to Connect and Disconnect a Program

This rule is used to connect and disconnect a selected program when a control port input is toggled.

1. Click the first rule in the **Rules panel** titled **Connect and disconnect a program when an input is toggled**.
2. Click the drop-down **Input** arrow and select the control port input which will trigger program connection and disconnection.
3. Click the drop-down **Program** arrow to select the program to be connected.

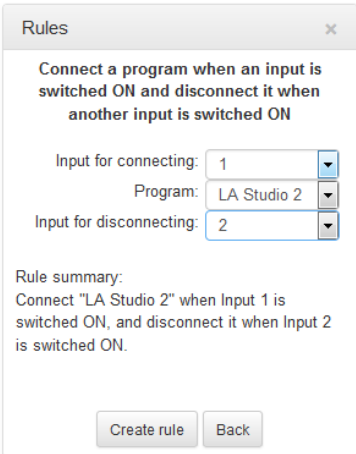


4. Check the **Rule Summary** and click **Create Rule** to save the settings.

Rule 2: Switch Different Control Port Inputs On to Connect and Disconnect a Program

This rule is used to connect and disconnect a selected program when different codec control port inputs are turned on.

1. Click the second rule in the **Rules panel** titled **Connect a program when an input is switched ON and disconnect it when another input is switched ON**.
2. Click the drop-down arrows to select the control port input used to connect and the alternative input for disconnecting.
3. Click the drop-down **Program** arrow to select an individual program which will be connected and disconnected by the change in the control port input states.

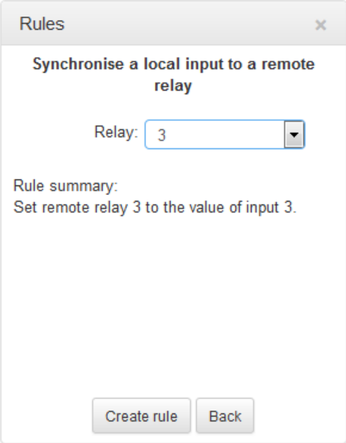


4. Check the **Rule summary** and click **Create Rule** to save the settings.

Rule 3: Synchronise Local Control Port Input Status with a Remote Relay Output

Use this rule allow a local codec's control port input to change the state of a remote relay output.

1. Click the third rule in the **Rules panel** titled **Synchronise a local input to a remote relay**.
2. Click the drop-down arrow to select the local control port input used to control a remote relay output.

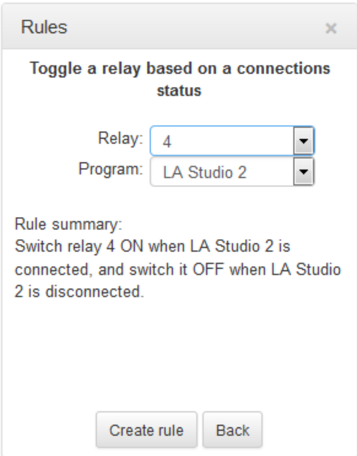


3. Check the **Rule summary** and click **Create Rule** to save the settings.

Rule 4: Toggle a Relay Output with each Change in Connection Status

This rule is used to toggle a codec's control port relay output each time a program connects and disconnects.

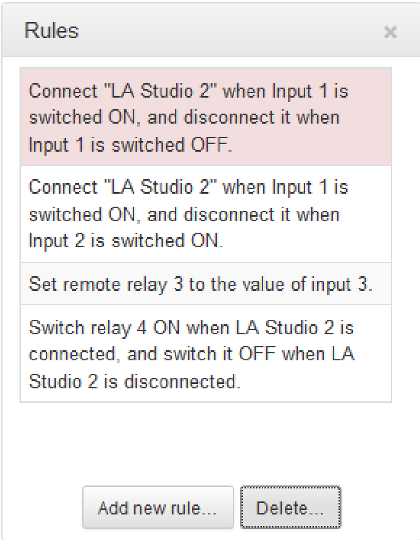
1. Click the fourth rule in the **Rules panel** titled **Toggle a relay based on a connection's status**.
2. Click the drop-down **Relay** arrow and select the relay output you want to toggle.
3. Click the drop-down **Program** arrow to select a specific program which will affect the relay toggle function, or use the default setting whereby any program will toggle the relay output.



4. Check the **Rule summary** and click **Create Rule** to save the settings.

Deleting Rules

1. Open the Toolbox HTML5 Web-GUI and click **Rules** in the **Menu Bar** to display the **Rules panel**.
2. Click to select the rule you want to delete.
3. Click the **Delete** button.



4. Click **Yes** in the confirmation dialog.

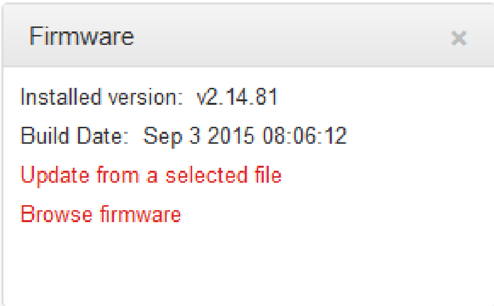
18.13 Upgrading Codec Firmware

To download the latest codec firmware visit http://www.tieline.com/Support/Latest-Firmware.

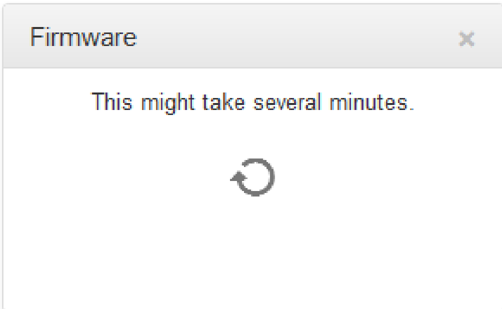
Firmware Upgrades

The following procedure explains how to perform codec firmware upgrades with a downloaded firmware file saved to your PC.

1. Open the Toolbox HTML5 Web-GUI and click **Settings** in the **Menu Bar**, then click **Firmware** to display the **Firmware panel**.



2. Click **Browse firmware** to search for the firmware for your codec and download it to your computer.
3. Once the firmware has been saved, click **Update from a selected file** in the **Firmware panel**.
4. Select the **.bin** file you are using to perform the upgrade and click **Open** to start the upgrade.



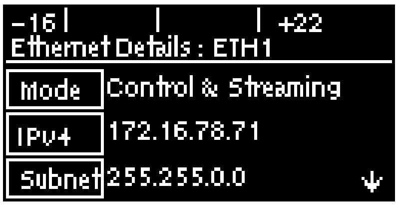
19 Front Panel Configuration Tasks

The following sections explain how to configure codec settings using the front panel **LCD** screen and **KEYPAD**.

19.1 Configuring IP via the Front Panel

Checking IP Address Details in the Codec

1. Press the **Home** button to return to the home screen.
2. Use the navigation buttons on the front panel to select **Settings** and press the button.
3. Select **Unit** and press the button.
4. Select **ETH1** and press the button.
5. IP address details and other relevant information is listed. Use the arrow up △ and down ▽ buttons to scroll and view all details listed.



Important Note: See the Configuring IP Connections sections for more details about IP connections. For assistance with configuration of IPv4 or IPv6 network connections contact your IT Administrator.

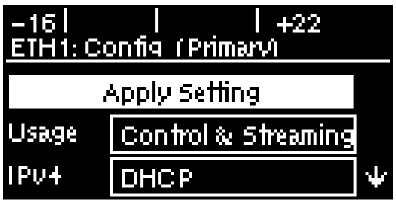
Ethernet and VLAN Configuration Options

The codec features a Ethernet port and up to four additional VLAN interfaces.

VLAN interfaces have features similar to physical Ethernet ports. However, your network administrator will need to configure VLAN support throughout your network for VLANs to be supported in your codec.

As an example, when only one physical Ethernet interface is available, VLANs can be used to separate codec Control and Streaming functions if required.

Following are a range of Ethernet and VLAN settings which can be configured in the **LAN** menu. After completing configuration ensure you navigate to **Apply Setting** and press the button to apply the new settings.



Configure an IPv4 DHCP Address

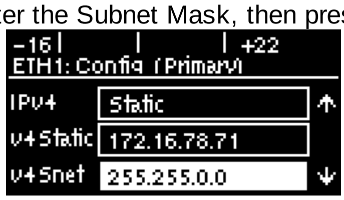
By default the codec is programmed for DHCP-assigned IP addresses. DHCP IP addresses are automatically assigned and can change each time you connect to your Internet Service Provider or by a router on your local area network (LAN).

1. Press the **SETTINGS** button.
2. Select **LAN** and press the button.
3. Use the down navigation button to select **ETH1** or a **VLAN** interface.
4. Select **Usage** and choose the appropriate control and/or streaming mode for the connection, then press the button.
5. Select **IPv4** and press the button.
6. Select **DHCP** and press the button.
7. Use the up navigation button to scroll to the top of the menu and select **Apply Setting**, then press the button to confirm the new settings.

Configure a Static IPv4 Address

Static IP addresses are fixed addresses which are recommended for studio installations. Using a static IP address ensures remote codecs can connect reliably using the same IP address over time.

1. Press the **SETTINGS** button.
2. Select **LAN** and press the button.
3. Use the down navigation button to select **ETH1** or a **VLAN** interface.
4. Select **Usage** and choose the appropriate control and/or streaming mode for the connection, then press the button.
5. Select **IPv4** and press the button.
6. Select **Static** and press the button.
7. Navigate to **v4 Static** and enter the IP address, then press the button.
8. Navigate to **v4 Snet** and enter the Subnet Mask, then press the button.



9. Navigate to **v4 Gway** and enter the Gateway details, then press the button.
10. Use the up navigation button to scroll to the top of the menu and select **Apply Setting**, then press the button to confirm the new settings.
11. Check the **Unit Details** menu to ensure the new static IP address has been entered correctly.

IPv6 Address Assignment

There are three IPv6 settings available for the Ethernet port and any VLANs which are configured.

1. Auto: An address is automatically assigned to the codec when you connect the codec to an IPv6 router. This process is similar to how an IPv4 DHCP address is assigned.
2. Manual: Select to manually enter IPv6 address details.
3. Off: Select to ignore IPv6 address details.

Important Note: Select **Off** if you are not using IPv6 to connect to another device. This ensures your codec will attempt to connect using IPv4 at all times.

To adjust this setting:

1. Press the **SETTINGS** button.
2. Select **LAN** and press the button.
3. Use the down navigation button to select **ETH1** or a **VLAN** interface.

4. Select **IPv6** and press the button.
5. Select **Auto**, **Manual** or **Off** and press the button.

By default the codec is configured to allow the codec to automatically receive IPv6 address information from an IPv6 enabled router.

Manual IPv6 Address Assignment

Select **Manual** mode using the previous procedure and enter information into the **v6 Static** (Address)**, v6 Prefix** and IPv6 **Gway** fields in the codec to manually configure address details.

DNS Server

It is possible to specify Domain Name Server (DNS) settings to allow easy look up of codecs within the specified **DNS Addresses** or **Domains** section within the web-GUI. This feature can be turned on or off in the LAN codec menu.

1. Press the **SETTINGS** button.
2. Use the navigation buttons on the front panel to select **LAN** and press the button.
3. Use the down navigation button to select **ETH1** or a **VLAN** interface.
4. Use the down navigation button to scroll to **Auto DNS**.
5. Press the button to toggle between **Yes** and **No**.

VLAN ID (VLAN configuration only)

The **VLAN ID** is encapsulated in IP packets to facilitate routing throughout your network.

1. Press the **SETTINGS** button.
2. Use the navigation buttons on the front panel to select **LAN** and press the button.
3. Use the down navigation button to select a **VLAN** interface.
4. Select **Usage** and press the button.
5. Select the mode of operation for this VLAN (e.g. Control & Streaming, Streaming only, Control Only) and press the button.
6. Use the down navigation button to scroll to **VLAN ID**.
7. Press the button to enter a number between 1-4094 inclusive.
8. Press the button to confirm this setting.

VLAN Priority (VLAN configuration only)

The **VLAN Priority** setting represents a prioritization scheme for forwarding data packets throughout Virtual Local Area Networks.

1. Press the **SETTINGS** button.
2. Use the navigation buttons on the front panel to select **LAN** and press the button.
3. Use the down navigation button to select a **VLAN** interface.
4. Select **Usage** and press the button.
5. Select the mode of operation for this VLAN (e.g. Control & Streaming, Streaming only, Control Only) and press the button.
6. Use the down navigation button to scroll to **Priority**.
7. Press the button to enter a number from 0 to 7 inclusive.
8. Press the button to confirm this setting.

VLAN Interface (VLAN configuration only)

This setting applies the VLAN settings to a physical Ethernet port in the codec. Only one physical Ethernet port is available which cannot be reconfigured.

19.2 Selecting an Algorithm

The codec offers a range of high quality algorithm options as well as 16 Bit 22kHz linear audio at less than 12 ms encode delay for high quality, uncompressed audio.

All Bridge-IT and Bridge-IT XTRA codecs include Opus, MPEG Layer 2, G.711 and G.722 algorithms, as well as the AAC suite of algorithms and Tieline Music and MusicPLUS as standard. Music and Music PLUS are optimized for wired and wireless IP connections.

aptX® Enhanced is included in Bridge-IT XTRA and can be purchased separately as a license upgrade in Bridge-IT codecs.

Bridge-IT Algorithm Encode License Options	Bridge-IT	Bridge-IT XTRA
AAC-LD, AAC-ELD, LC-AAC, HE-AAC v.1 and HE-AAC v.2 algorithms	✔	✔
16 bit and 24 bit aptX® Enhanced algorithm	*	✔

✔ Included
* Option available for purchase separately if required.

Note: Bridge-IT has a range of default connection profiles that make it very simple to easily program your codec to connect using all available algorithms. See Choosing Dialing Profiles for more details.

Overview of Tieline Algorithms

1. The Tieline Music algorithm is optimized for audio bit rates as low as 19.2kbps with only a 20 millisecond encode delay. It offers 15 kHz mono from 24Kbps to 48Kbps.
2. Tieline MusicPLUS delivers up to 20 kHz mono from 48kbps upwards. It can also deliver up to 20 kHz stereo from 96kbps upwards, offering huge savings on your IP data bills and outstanding audio quality.

Overview of AAC Algorithms

Important Notes: AAC algorithms are only available in Bridge-IT if the AAC license has been purchased and uploaded into the Bridge-IT codec. For more information see Installing Software Licenses.

AAC-LC

LC-AAC is optimised for audio bit-rates of 64Kbps per channel or higher using a sample rate of 48kHz. Tieline recommends using LC-AAC instead of HE-AAC if bandwidth of 64Kbps or higher per channel is available, to optimise audio quality. If lower than 64Kbps is available, consider using HE-AAC, Tieline Music or Tieline MusicPLUS.

AAC-HE

Codecs include both HE-AAC v.1 and HE-AAC v.2, which are optimized for low bit rate connections. Selection of HE-AAC v.1 and v.2 is automatically managed within the codec, so only **AAC-HE** is displayed on the screen. When used for mono connections, HE-AAC v.1 performs best at bit rates of 24kbps per channel or higher. HE-AAC v.1 is also used for stereo connections when audio connection bandwidth is 48kbps or higher.

HE-AAC v.2 is used for stereo connections when audio connection bandwidth is below 48kbps and is capable of delivering 15kHz quality stereo audio at audio bit rates as low as 24kbps.

A sample rate of 32kHz is used in the codec's default profiles to achieve ultra-low bit-rate connections, but this is adjustable to 44.1kHz or 48kHz if required.

AAC-LD

AAC-LD (Low Delay AAC), AAC-ELD (Enhanced Low Delay AAC) and AAC-ELDv 2 are optimized for low latency real-time communication. AAC-LD is suited to bit rates of 96kbps or higher for stereo audio.

AAC-ELD

AAC-ELD is optimised for high quality stereo connections from 48 - 96kbps and performs better at these bit rates when compared with AAC-LD.

AAC-ELD v 2

For stereo connections below 48kbps AAC-ELD v2 will deliver better performance than AAC-ELD down to 24kbps.

Overview of aptX® Enhanced Audio Coding

Important Notes: aptX® Enhanced is only available if the aptX® Enhanced license has been purchased and uploaded into the codec. For more information see Installing Software Licenses.

aptX® Enhanced audio coding is used by thousands of radio stations to deliver very low delay audio for studio to transmitter links, audio distribution and remote broadcasts. It delivers outstanding audio quality with exceptionally low delay across a range of IP networks. It is ideal for high quality studio-to-transmitter links and audio distribution.

32kHz, 44.1kHz or 48kHz sampling rates are available at either 16 bit or 24 bits per sample. aptX® Enhanced has a minimum connection bit-rate of 128Kbps per channel and offers 10Hz to 24kHz frequency response. 24 bit, 48kHz aptX® Enhanced at the maximum bit-rate of 576Kbps delivers >120dB of dynamic range.

Overview of Opus Algorithm

Opus is a highly versatile open source audio coding algorithm. It incorporates technology from the well-known SILK and CELT codecs to create a low latency speech and audio codec. It is a variable bit rate algorithm ideal for live broadcast situations because of its capacity to deliver high quality, real-time Audio over IP (AoIP) at low bit rates. Visit http://www.opus-codec.org for more info.

There are three Opus algorithm configurations available:

Algorithm	Recommended connection for on-air use
Opus Voice	High quality low bit rate remotes (9.6kbps -64kbps)
Opus Mono	Very high quality mono remotes, STLs and audio distribution (48kbps -128kbps)
Opus Stereo	Very high quality stereo remotes, STLs and audio distribution (64kbps -256kbps)

Configuring an Algorithm in the Codec

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select your preferred **IP Session** mode and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Navigate to **Algor'm** and press .
7. Select the mono or stereo algorithm that you want to connect with and press .

How do I choose the right algorithm?

The algorithm you select will not only affect the quality of the broadcast but it will also contribute to the amount of latency or delay introduced. For example, if MP2 algorithms are used, program delays will be much longer than when using Tieline Music or MusicPLUS algorithms. This is due to the additional inherent encoding delays involved when using MP2 algorithms. This can be a major consideration for live applications that integrate remote-crosses into a broadcast. The algorithm you choose to connect with will also depend upon:

- The codecs you are connecting to (Tieline versus non-Tieline)
- Whether you are creating multi-unicast connections.
- Whether you are connecting using SIP or not.
- The uplink bandwidth capability of your broadband connection.

Important Notes: Music and MusicPLUS algorithms cannot be used over SIP connections. Use MP2 algorithms at 64kbps mono or 128kbps stereo for high quality connections when using SIP, or use G.711 and G.722 if required. Tieline G3 codecs do not contain AAC algorithms so choose one of the other available algorithms if you are connecting to a G3 codec.

It can be a good idea to listen to the quality of your program signal using each algorithm and to see how it sounds when it is sent at different connection bit-rates (as well as different FEC and jitter-buffer millisecond settings). This will assist you to determine what the best algorithm setting is for the connection you are setting up. Please see the following table for details on the connection requirements of the different algorithms available.

Algorithm	Audio Band-width	Algor-ithmic Delay	IP bit rate per channel	IP over-head per connection	Audio Quality and Features	Recommended applications for on-air use
PCM/Linear (Uncom-pressed)	16/24 bit up to 24kHz	0ms	sample rate x bits per sample x no. channels	80kbps	• Full bandwidth, perfect audio quality for voice and music • No error concealment/ correction or artefacts	• Extremely high quality uncompressed audio for STLs and audio distribution. • Ideal for fiber or high bandwidth links.
Tieline Music	Up to 15kHz	20ms	24 kbps minimum	16kbps	• High quality voice and music • Very low delay at low bit rates	• Great for live voice or music remotes as well as STLs and audio distribution with limited connection bandwidth (e.g. 3G wireless) • Suitable when bidirectional communication between announcers is required
Tieline Music-PLUS	Up to 22kHz	20ms	48 kbps minimum (Optimised for 64kbps per audio channel)	16kbps	• Very high quality voice and music • Very low delay at low to moderate bit-rates	• Very high quality, very low delay STLs and audio distribution • Remote connections able to achieve 48kbps for each audio channel • Suitable when bidirectional communication between announcers is required
G.711	3kHz	1ms	64kbps minimum	80kbps	• Low quality 3kHz POTS phone quality audio • Very low delay at moderate bit rates	• Highly compatible with other brands of audio codec • Low quality and used generally for compatibility
G.722	7kHz	1ms	64kbps minimum	80kbps	• Good quality 7kHz voice • Better quality than a standard POTS phone call • Very low delay at moderate bit rates	• Highly compatible with other brands of audio codec • Good voice quality audio for remotes and other voice quality applications
MPEG Layer 2	Up to 22kHz	24 to 36ms	64kbps minimum	8.5 - 13.3kbps	• Very high quality voice and music • Low to moderate delay at moderate to high bit rates	• Highly compatible with other brands of audio codec • Very high quality audio for remotes, STLs and audio distribution
LC-AAC	Up to 15kHz	64ms	64kbps	15kbps	• High quality voice and music at lowest bit rate; better quality at higher bit rates • Moderate delay at moderate to high bit rates	• Voice or music remotes as well as STLs and audio distribution where some delay is tolerable • Tieline Music or MusicPLUS deliver lower delay

HE-AAC v.1	Up to 15kHz	128ms	48kbps	7.4kbps	• High quality voice and music at the low est bit rate; better quality at higher bit rates • Low to Moderate bit rates • High delay	• Live voice or music remotes as w ell as STLs and audio distribution w ith limited connection bandw idth • Use w hen bidirectional communication betw een announcers is not required
HE-AAC v.2	Up to 15kHz	128ms	Minimum 16kbps (Mono); 24kbps (stereo)	7.4kbps	• High quality voice and music • Low bit rates • High delay	• Used for DAB+ radio streaming • Ideal for low bit rate remotes • Use w hen bidirectional communication betw een announcers is not required
AAC-LD	Up to 20kHz	20ms at 48kHz	48kbps minimum	30kbps	• Very high quality voice and music • Very low delay at low to moderate bit rates	• Very high quality, very low delay STLs and audio distribution • Remote connections able to achieve 48kbps for each audio channel requiring • Suitable w hen bidirectional communication betw een announcers is required
AAC-ELD	Up to 20kHz	15-30ms	24 kbps minimum	15-30kbps	• Very high quality voice and music • Very low delay at low bit rates	• Great for live voice or music remotes • Suitable w hen bidirectional communication betw een announcers is required
AAC-ELDv.2	Up to 20kHz	35ms	Pending release	Pending release	• High quality voice and music • Low delay at low bit rates	• Great for live voice or music remotes w here limited connection bandw idth is available • Suitable w hen bidirectional communication betw een announcers is required
aptX Enhanced	10Hz-24kHz	2.5ms at 48kHz	128kbps minimum (16bit; 32kHz) to 288kbps (24bit;48kHz)	80kbps	• Very high quality voice and music • Extremely low delay at high bit rates • Highly cascade resilient	• Ideal for STLs and audio distribution w here high connection bandw idth is available and very low delay is highly desirable. • Resilient w ith multiple encodes/decodes w hen required
Opus	4Hz-20kHz	20ms	9.6-256kbps	16kbps	• Very high quality voice and music • Very low delay at low bit rates	• "Opus Voice" is ideal for high quality, and low delay voice quality remotes at extremely low bit rates. • "Opus Mono" and "Opus Stereo" are perfect for high fidelity remotes, STLs and

						audio distribution at higher bit rates

Algorithm Selection Guide

Algorithm	Very Low Delay	Moder-ate to High Delay	Excell-ent Perfor-mance at Low Bit rates	Preferr-ed for Live Remotes	Preferred for STLs and Audio Distribu-tion	Highly Compat-ible with other Codecs
Linear/PCM	✓				✓	✓
Opus	✓		✓	✓		
Tieline Music	✓		✓	✓		
Tieline MusicPLUS	✓		✓	✓	✓	
apt-X Enhanced	✓				✓	
LC-AAC		✓			✓	
HE-AACv1		✓			✓	
HE-AACv2		✓	✓	✓ *		
AAC-LD	✓			✓	✓	
AAC-ELD	✓		✓	✓		
AAC-ELDv2	✓		✓	✓		
MPEG Layer 2	✓				✓	✓
G.722	✓					✓
G.711	✓					✓

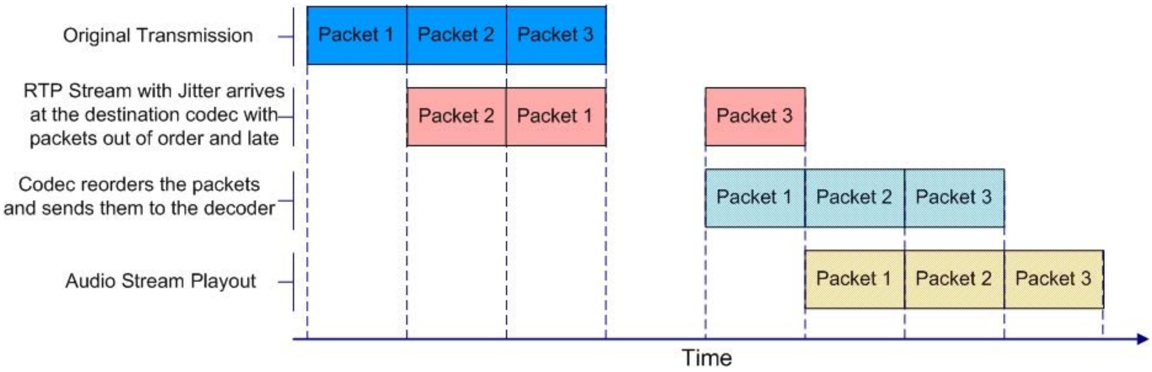
***** Use with caution for remotes due to high delay; not suitable when bidirectional communications is required.

Sampling Rates

When selecting linear (PCM) uncompressed audio or AAC, MPEG and aptX® Enhanced algorithms, it is possible to select different either 32kHz, 44.1kHz and 48kHz sample rates as required. Tieline Music runs at 32kHz sampling and MusicPLUS runs at 48kHz sampling. G.711 and G.722 will always run at a 32kHz sampling rate (downsampled to 8kHz and 16kHz respectively).

19.3 Configuring the Jitter Buffer

Jitter, (also known as latency or delay), is the amount of time it takes for a packet of data to get from one point to another. A jitter buffer is a temporary storage buffer used to capture incoming data packets. It is used in packet-based networks to ensure the continuity of audio streams by smoothing out packet arrival times during periods of network congestion. Data packets travel independently and arrival times can vary greatly depending on network congestion and the type of network used, i.e. LAN versus wireless networks. The concept of jitter buffering is displayed visually in the following image.



Jitter buffer management is encompassed within Tieline's SmartStream IP technology which can:

- Remove duplicate packets.
- Re-order packets if they arrive out-of-order.
- Repair the stream in the event of packet loss (error concealment).
- Manage delay dynamically based on current network congestion.
- Manage forward error correction (FEC).

Tieline codecs can be used to program either a fixed or automatic jitter buffer and the setting you use depends on the IP network you are connecting over. Over LANs, WANs and wireless networks the automatic jitter buffer generally works well. It adapts automatically to the prevailing IP network conditions to provide continuity of audio streaming and minimizes delay.

A fixed jitter buffer is preferable over satellite connections to ensure continuity of signals.

CAUTION: If a Tieline codec connects to a device that is using non-compliant RTP streams then the last fixed setting entered into the codec will be enabled (default is 500ms). Non-compliant devices include some other brands of codec, web streams and other devices.

Tieline ‘Auto Jitter Buffer’ Settings

Least Delay: This setting attempts to reduce the jitter buffer to the lowest possible point, while still trying to capture the majority of data packets and keep audio quality at a reasonable level. This setting is the most aggressive in its adaptation to prevailing conditions, so jitter buffer may vary more quickly than with the other settings. It is not recommended in situations where jitter variation is significant and/or peaky. (E.g. 3G/multi-user wireless networks). It is best for stable and reliable links such as dedicated or lightly-loaded WAN/LANs.

Highest Quality: This setting is the most conservative in terms of adapting down to reduce delay. The jitter-buffer setting will actually stay high for a longer period after a jitter spike is detected – just in case there are more spikes to follow. This setting is best used where audio quality is most highly desired and delay is not so critical. Unless delay is irrelevant, this setting is also not recommended over peaky jitter networks (such as 3G) and is best used on more stable networks where large jitter peaks are not as common.

Best Compromise: This (default) setting is literally the midpoint between the jitter buffer levels that would have been chosen for the Highest Quality and Least Delay settings. It is designed to provide the safest level of good audio quality without introducing too much extra delay.

Good Quality and **Less Delay**: These two settings lie between the mid-point setting of Best Compromise and two settings Highest Quality and Least Delay. They indicate a slight preference and may assist in achieving better performance from a connection without incurring extreme delays in transmission or packet loss.

Which Algorithms can use Automatic Jitter Buffering?

The following table provides an overview of which algorithms are capable of using the automatic jitter buffer feature over SIP and non-SIP connections.

Algorithm	Non-SIP Connections	SIP Connections
Linear (Uncompressed)	✗	✗
Tieline Music	✓	✗
Tieline MusicPLUS	✓	✗
G.711	✗	✓
G.722	✗	✓
MPEG Layer 2	✓	✓
LC-AAC	✓	✓
HE-AAC v.1	✓	✓
HE-AAC v.2	✓	✓
AAC-LD	✓	✓
AAC-ELD	✓	✓
Opus	✓	✓
aptX Enhanced	✗	✗

Configuring Automatic Jitter Buffering (Default Setting)

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select your preferred **IP Session** mode and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Navigate to **Jitter** and press .
7. Select **Auto Adapt** and press .
8. Select your preferred jitter buffer setting and press .

How to get the Best Jitter Buffer Results

When configuring automatic jitter buffer settings, establish the IP connection for a while before ‘going live’, to let the codec evaluate the prevailing network conditions. The initial jitter buffer setting when a codec connects is 500ms and it is kept at this level for the first minute of connection (as long as observed delay values are lower than this point).

After the initial connection period the jitter buffer is adapted to suit the current network conditions and is usually reduced. Establish a connection for at least 5 minutes prior to broadcasting, so that the codec has been provided with enough jitter history to ensure a reliable connection.

There are five states or stages that jitter buffer may display and these can be observed in the connection status screen by selecting **HOME** **> Cxns** button while connected and use the down and up navigation buttons to scroll through connection statistics.. The first four stages are observed in "auto" jitter buffer mode.

1. **Stabilization period (a1):** A few seconds at the start of a connection where no action is taken at all while the establishment of a stable connection means analysis of jitter data is not valid.
2. **Stage 2 (a2):** A compatibility check to ensure the RTP connection is compliant and RTP clocks are synchronized enough to perform jitter analysis.
3. **Stage 3 (a3):** If the compatibility check is successful, this is the analysis hold-off period. During a minute, the jitter buffer is held at a safe, fixed value of 500ms while enough history is recorded to start jitter buffer adaptation.
4. **Stage 4 "live" (A)**: This is where the codec determines it is safe enough to start broadcasting using the auto-jitter buffer level. We recommend running the codec for a few more minutes to obtain a more comprehensive history of the connection's characteristics.
5. **Fixed (F):** This state is displayed if the jitter buffer is fixed.

Important Note: The jitter buffer setting in the codec can only be adjusted when a connection is off-line. Automatic jitter buffering is disabled for a Raw (linear uncompressed) audio connection.

Auto Jitter Buffer and Forward Error Correction (FEC)

If forward error correction is configured then additional data packets are sent over a connection to replace any data packets lost. There is no need to modify jitter buffer settings if you are sending FEC data, only if you are receiving FEC data.

The jitter buffer depth on the receive codec needs to be increased if forward error correction is employed. We recommend you add 100ms to the jitter buffer on a codec receiving FEC at a setting of 20% and 20ms at a setting of 100%.

Tieline's auto jitter buffer detects the amount of FEC that is being used and automatically compensates to increase the codec jitter buffer if FEC is being used.

Fixing Jitter Buffer Settings

The default jitter-buffer setting in Tieline codecs is 500 milliseconds. This is a very reliable setting that will work for just about all connections. However, this is quite a long delay and we recommend that when you set up an IP connection you test how low you can set the jitter-buffer in your codec.

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select your preferred **IP Session** mode and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Navigate to **Jitter** and press the button.
7. Select **Fixed Buffer** and press the button.
8. Use the numeric **KEYPAD** to enter the fixed buffer value in milliseconds and press the button.

If you change the jitter buffer setting in a codec it will only adjust to the new level when link quality is high (e.g. above 70%). This is done to ensure audio quality is not compromised. When manually configuring the jitter-buffer delay in a codec it is necessary to think carefully about the type of connection you will be using. Following is a table displaying rule of thumb settings for configuring jitter-buffer delays into your codec.

Connection	Jitter-Buffer Recommendation
Private LAN	60 milliseconds
Local	100 - 200 milliseconds
National	100 - 300 milliseconds
International	100 – 400 milliseconds
Wireless Network	250 - 750 milliseconds
Satellite IP	500 - 999 milliseconds

Important Note: The preceding table assumes Tieline Music is the algorithm in use. Do not use PCM (uncompressed) audio over highly contended DSL/ADSL connections without enough bandwidth to support the high connection bit-rates required.

19.4 Configuring Forward Error Correction

Forward Error Correction (FEC) is designed to increase the stability of UDP/IP connections in the event that data packets are lost. FEC works by sending a secondary stream of audio packets over a connection so that if your primary audio stream packets are lost or corrupted, then packets from the secondary stream can be substituted to replace them. The amount of FEC required depends on the number of data packets lost over the IP connection.

Both the local and remote codec FEC settings can be configured in the codec before dialing. These settings can also be changed 'on the run' while the codecs are connected. FEC should only be used if link quality displayed on the codec is below **S:99 R:99**, as it is of no benefit otherwise.

Configuring FEC into the Codec

1. Press the **HOME** button to return to the **Home** screen.
2. **Use the navigation buttons on the front panel to select Connect and press the button.**
3. Select **IP** and press the button.
4. Select **Tieline** and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Navigate to **FEC** and press .
7. Select the local codec FEC setting in the **Local FEC** screen and press .
8. Select the remote codec FEC setting in the **Remote FEC** screen and press .
9. Check that the settings are correct in the **RTP Settings** screen.

The four FEC settings in Tieline codecs are outlined in the following table with their bit rate ratios.

FEC Setting	Bit rate Ratios	Connection Use
100% (Lowest delay)	A simultaneous dual-redundant stream (1:1 ratio) is sent from the codec. Twice the connection bit rate is required to operate the codec using the 100% setting. E.g. if your connection is 14,400Kbps, you will require an additional 14,400 Kbps of bandwidth to allow for the FEC data stream.	Recommended to be used over wireless and international connections.
50%	Additional data is sent by FEC in a ratio of 2:1.	Recommended for international & national connections
33%	Additional data is sent by FEC in a ratio of 3:1.	Recommended for national and local connections.
20% (Highest delay)	Additional data is sent by FEC in a ratio of 5:1.	Recommended for local and LAN connections.
Off	FEC is off in the codec and the connection bandwidth is equal to the connection bit rate setting in the codec.	Recommended for wired LAN connections & managed T1 & E1 connections for STLs that have connections that aren't shared & have quality of service (QoS).

Important Note: FEC can only be configured for use with the Music and MusicPLUS algorithms.

How does FEC work?

If you enter a FEC setting of 20% and you are losing one packet in every five sent, the lost packet will be replaced by FEC to maintain the quality of the connection. If you are losing more packets than this, say one in three, it will be necessary to increase the FEC setting to 33% to compensate.

Note: There is an inverse relationship between FEC settings and the jitter-buffer millisecond setting that you use for IP connections.

So why not use 100% FEC every time? The answer is because you need twice the bit rate to achieve full redundancy and depending on the link conditions, this could potentially cause more drop-outs because of network congestion than it fixes. Here is a simple rule to remember: Your maximum uplink speed is all the bandwidth you have to play with. As a rule of thumb, try not to exceed more than 80% of your maximum bandwidth. If your link is shared, be even more conservative.

You should also consider the remote end too. What is their maximum upload speed? Is the connection shared at either end? Your bit rates, FEC settings and buffer rates must be pre-configured at both ends before you connect, so it's always better to set your connection speed and balance your FEC according to the available uplink bandwidth at each end for best performance.

As an example, if you want 15 kHz mono (using the Tieline Music Algorithm) you will need at least a 24kbps connection for audio. Adding 100% FEC will add another 24kbps making your bit rate 48kbps plus some overhead of around 10kbps is required. If you're on a 64kbps uplink, you should consider reducing your FEC to minimise the likelihood of exceeding your bandwidth capacity.

Here is another example, if you want 15 kHz stereo, you need at least 56kbps for the audio. 100%

FEC requires at least 112kbps and 50% FEC requires at least 84kbps. If your uplink speed is 256kbps and you're on a shared connection, then choosing a lower FEC setting of 20%-33% may give you better results.

19.5 Configuring Encode/Decode Direction

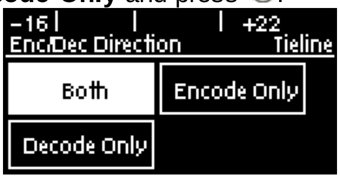
By default the codec is configured to encode and decode data. However, it is possible to encode or decode audio data only. This is useful for:

- Conserving connection bandwidth when unidirectional data streaming is required.
- Lowering data costs.
- Increasing overall connection reliability.

Configure the transmitting codec to encode only, and configure the receive codec to decode only when using this feature.

To adjust this setting:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the button
3. Select **IP** and press .
4. Select your preferred **IP Session** mode and press the button.
5. Use the down navigation button to select **Setup** and press .
6. Navigate to **Dir** and press .
7. Select **Encode Only** or **Decode Only** and press .



19.6 Enabling Relays & RS232 Data

The codec supports both in-band and out-of-band data depending on the algorithm you are using. (See RS232 Data Configuration for more details). RPTP data is automatically enabled when using the Tieline Music or MusicPLUS algorithms over IP. It is also possible to enable synchronized out-of-band data using any algorithm via the **RS232** data port on the codec.

Data must be enabled to activate contact closure operation and RS232 data.

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select **Tieline** (or **Sessionless**) as the **IP Session** mode and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Navigate to **Data** and press to toggle between **Enabled** and **Disabled** (Note: default setting is **Disabled**).

Configuring Control Port Contact Closure Operation

The **Rules panel** on the web-GUI can be used to configure switch inputs and relay outputs. See the section titled Creating Rules for more information.

About Relays

The codec has two CMOS solid state relays for the control of equipment, consisting of two relay closures and two opto-isolated outputs.

Inputs

The input signal is referenced to chassis ground, i.e. the ground reference terminal on the terminal block is connected the chassis. The input device is a high impedance CMOS device with a 330 ohm pull-up resistor to +5 volts.

Operation is as simple as joining the input pin to the ground terminal. This can be via a remote relay contact or the open circuit collector of a transistor or FET. DO NOT feed voltages into the inputs.

Outputs

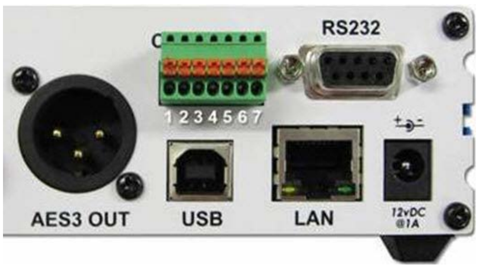
CMOS field effect transistors switch a low impedance path between the two pins when activated. These are opto-isolated and floating above ground. It is important to current-limit the source as damage will result where the current exceeds 100mA peak-to-peak. No more than 48 volts peak-to-peak should be used as a safety precaution. The resistance of the CMOS element is approximately 25 ohms in the ON state.

Relay Operation and Pinouts

A closing contact across Input 1 or 2 (pins 5 or 6) to Ground (pin 7) will provide a closing contact on the remote codec Output 1 (pins 1 and 2) or Output 2 (pins 3 and 4).

For multi-unicast connections to multiple codecs, a contact closure will appear on each of the compatible (non-G3) remote codecs' corresponding contacts. I.e. Input 1 shorted, Output 1 contacts on all codecs closed.

Pins	Pin Function
1	Output 1
2	Output 1
3	Output 2
4	Output 2
5	Input 1
6	Input 2
7	Ground



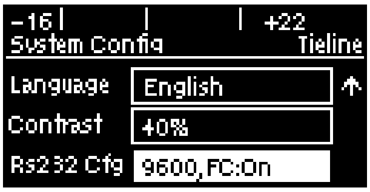
Important Note: For more information about how to configure relay operations with a PC using the Toolbox web-GUI, please see Creating Rules.

Configuring the Codec to Send RS232 Data

Once Data is enabled, the codec can also be connected to external devices and send RS232-

compatible data via the serial port on the rear panel of the codec.

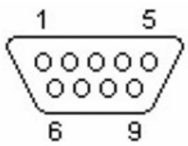
1. Press the **SETTINGS** button.
2. Navigate to **System** and press .
3. Select **RS232 Cfg** and press .



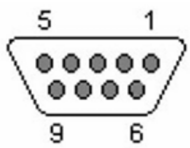
4. Use the navigation buttons to select the correct baud rate.
5. Select **Enable** for flow control and press to save all settings.

RS232 Pin-outs and Data Connections

Pin	INTERFACE Female DB9 (RS232) DCE	DATA Male DB9 (RS232) DTE
1	No Connection	No connection
2	TX Data	RX Data
3	RX Data	TX Data
4	No connection	No connection
5	Signal Ground	Signal Ground
6	No Connection	No connection
7	CTS	RTS
8	RTS	CTS
9	No connection	No connection



DB9 Male Connector Pins



DB9 Female Connector Pins

Important Notes:

- Codecs using Bridge-IT firmware lower than v2.8.xx cannot activate relays on Tieline G3 codecs or send RS232 data to them.
- It is important that you enable serial port flow control within the codec. Flow control regulates the flow of data through the serial port. If disabled, data will flow unregulated and some may be lost.
- Ensure you match the serial port baud rate to match the rate of the external device you are connecting to. Ideally the settings on both codecs should match, or you could have data overflow issues.
- Only the dialing codec needs to be configured to send RS232 data. Session data sent from the dialing codec will configure all other compatible codecs (non-G3) you connect with.
- RS232 data can be sent from the dialing codec to all end-points of a multi-unicast connection. Note: Bidirectional RS232 data is only available on the first connection dialed when multi-unicasting.

19.7 Configuring TCP/UDP Ports

Tieline Codec Default Port Settings

By default, the codec uses a TCP session port to send session data and a UDP port to send audio. The session port uses the TCP protocol because it is more likely to get through firewalls – ensuring critical session data (including dial, connect and hang-up data) will be received reliably.

The default session and audio port settings in Tieline codecs, for both TCP and UDP connections, are outlined in the Installing the Codec at the Studio section of the manual. This section also contains useful information for configuring port forwarding and troubleshooting IP connections.

Changing Codec Port Numbers

Reasons for adjusting the port setting on your codec include:

- Having to create a path through gateways and firewalls.
- Another IP device is using the codec's default port number.
- You are sharing a single IP address with multiple codecs and each codec requires a different port number to perform port forwarding.

Configuring the Session and Audio Port Numbers used when Dialing a Program

Codecs require matching port numbers to connect successfully. When you create a program the session and audio ports can be adjusted from the defaults as required. Note: If there is a need to change codec port settings please consult your organization's resident IT professional. To adjust port settings:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select an **IP Session** mode.
5. Use the down ▽ navigation button to select **Setup** and press the button.
6. Navigate to either **Session** (session protocol) or **Proto** (audio protocol) and press .
7. Select the session or audio ports you want to adjust and press .
8. Use the numeric **KEYPAD** add a new port number and press .

Configuring the Tieline Session Ports when Answering

To adjust the local Tieline session data port used by your codec to answer incoming calls:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Settings** and press the button.
3. Select **Tieline Session** and press .
4. Navigate to **Port** or **Alt. Port** (alternative session port) and press .
5. Adjust the setting and press the button to store the new configuration.

Audio Port Settings for Tieline Session Data and Sessionless IP Calls

The codec supports sessionless IP streaming, whereby the codec does not send Tieline session data when attempting to connect. When using this mode you need to configure the "send" audio port (codec port at the remote end of the link to which you are sending audio) and "return" audio port

(port used by the local codec to receive audio from the remote codec).

It is also possible to configure the remote and local audio ports for a codec using Tieline session data to establish IP connections. This may be required because some firewalls require symmetric port configuration.

Sessionless Audio Port Configuration

When you select **Sessionless** as the **Session Protocol**:

- The default value for both the **Send** and **Return** (audio) **Ports** is 9000
- The range of values for the audio ports is 2000 to 65535
- The audio port values can be set independently
- Both audio ports can always be configured, i.e. there is no dependency on encode/decode direction

"Tieline Codec" Port Configuration

If using the **Tieline Codec** setting for call establishment (i.e. Tieline session data is enabled), you can also change the default audio ports if required.

- The default value for the **Send** (audio) **Port** is 9000
- The range of values for the **Send Port** is 2000 to 65535
- The default port value for the **Return** (audio) **Port** is **Automatic**. Note: **Automatic** indicates that the codec will allocate the return port value and send this information to the codec to which you are dialing
- The range of values for the **Return Port** is 2000 to 65535

Sessionless Multicast Connections

For a sessionless multicast server connection:

- Only the **Send Port** is available
- The default value for the port is 9000
- The range of values for the port is 2000 to 65535

For a sessionless multicast client connection:

- Only the **Return Port** is available
- The default value for the port is 9000
- The range of values for the port is 2000 to 65535

19.8 Configuring QoS for Broadcasts

It is possible for IP networks to differentiate between and prioritise data packets being transmitted through routers across networks. This is useful because in modern data networks many different IP services like email, voice, web pages, video and streaming music coexist within the same network infrastructure.

Prioritising IP Data Packets when Broadcasting

Broadcast IP audio data packets can be configured for expedited or assured forwarding (Quality of Service or QoS) when traversing different networks. Routers can also be configured to ignore these forwarding priorities so they are not assured across all networks.

The codec can be configured to prioritise IP data packets sent across a network by entering a value into the Differentiated Services Code Point (DSCP) field within the header of data packets transmitted by the codec over the network. Check with your IT administrator before changing this setting. By default the codec is configured for Assured Forwarding and more details about DSCP are available on Wikipedia at http://en.wikipedia.org/wiki/Dscp.

Configuring Bridge-IT for QoS

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **QoS** and press the button.
3. Press the button and use the **RETURN** button to delete numbers already entered, then use the numeric **KEYPAD** to enter the new setting.
4. Press the button to save the new setting.

See Configuring IP Settings for instructions on configuring this setting using the HTML5 Toolbox web-GUI.

19.9 Configuring Data Packet Time-to-Live

Time-to-Live (TTL) is a value you can configure to set a finite life for data packets sent by the codec. This avoids situations where packets can keep circulating through routers causing network congestion.

The Time-to-Live setting is configurable and sets the maximum number of router hops allowable for multicast data packets. In most situations the default value of **1** is used, to ensure packets are sent through a single LAN router and not over multiple router hops and networks.

This setting is only used in **Multicast Server** mode.

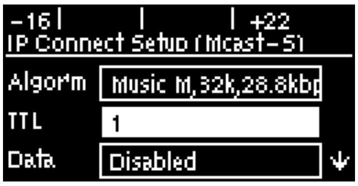
1. Press the **HOME** button to return to the **Home** screen, select **Connect,** then select **IP** and press the button.

2. Select **Sessionless** and press .

3. Press the right navigation button to select **Multicast Server** and press .

3. Press the down navigation button to select **Setup** and press .

4. Press the down navigation button to select **TTL** and press .



5. Press the **RETURN** button to delete the current setting and use the numeric **KEYPAD** to enter a new value. Press to confirm the new setting.

19.10 Reset and Restore Factory Default Settings

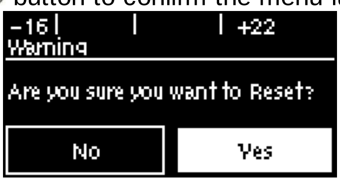
There are several options in the **Reset** menu which allow you to restore factory default settings within the codec.

	Function	Description
1	Reset Audio and 'Connect' Settings	Click to restore factory default settings for Audio and Connect menu settings
2	Restore Factory Defaults	Click to restore factory default settings, excluding user defined programs and call history
3	Delete Programs & Call History	Deletes custom programs and recent calls in the codec
4	Reboot Codec	Click to restart the codec
5	Clear Logs	Deletes codec event and log history. Note: This should only be performed if instructed to by Tieline support staff.

Important Note: After restoring factory defaults, always reboot the codec using the **Reboot Codec** function, not by removing power from the codec.

1. Press the **SETTINGS** button.
2. Navigate to **Reset** and press the button.
3. Navigate to the preferred option from those available and press the button.

4. Select **Yes** and press the button to confirm the menu function that you are performing.



Reset and Restore Factory Defaults using the Web-GUI

The web-GUI can also be used to reset and restore factory defaults. See Reset Factory Default Settings for in the web-GUI section of this manual for more details.

19.11 Installing Software Licences

Bridge-IT XTRA codecs include all software and algorithm feature options when purchased. Other Bridge-IT codecs can be upgraded to include aptX® Enhanced encoding via an additional license purchase and upgrade.

Contact Tieline at sales@tieline.com, or your favorite dealer, if you need to purchase a software license upgrade. When a software license has been purchased there are two ways to perform an upgrade:

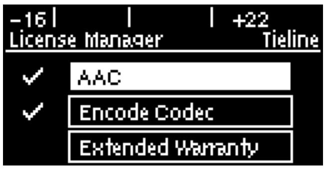
1. Automatically download the software license from TieServer and perform the upgrade.
2. Download the new software license file onto your PC and upgrade using the web-GUI.

To install a software license using the Toolbox Java or HTML5 web-GUI options see Web-GUI Software License Installation.

Checking Installed Licenses

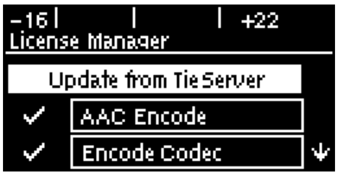
The codec **License Manager** is used to view which licenses are installed in each codec. To view the licenses installed in your codec:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Licenses** and press the button.
3. A list of all possible licenses is displayed and all licenses that have been installed have a tick next to them.



Update and Install Licenses from the Codec

1. Navigate to **Update from TieServer** in the **License Manager** screen and press the button.



2. The codec will contact TieServer and automatically install all valid licenses.
3. The screen will indicate the update is in progress and then confirm it has been completed successfully.
4. Press the **RETURN** button a few times until you return to the **Home** screen.
5. Use the navigation buttons to select **Settings** and press the button.

6. Navigate to **Reset** and press the button.
7. Navigate to **Reboot Codec** and press the button.
8. Select **Yes** and press the button to reboot the codec.

20 Reference

The following sections contain reference and troubleshooting information.

20.1 Installing the Codec at the Studio

Studio IP Streaming Setup for Tieline Audio Codecs

The following instructions are intended to help you configure your internet connection and Tieline codecs at the studio to enable incoming calls over the internet from a remote Tieline codec. It is assumed that you have a basic understanding of your IP network and how to configure IP devices. If you have limited IT network knowledge, we recommend you engage the services of an IT professional to install the public IP address and perform the Network Address Translation (NAT) and port forwarding between the public internet and your private Local Area Network (LAN) at the studio.

Prerequisites

The following procedures are valid for:

- All firmware versions in the Genie and Merlin codec families.
- All Bridge-IT Basic and Pro and Bridge-IT XTRA codecs with firmware release v.2.x or higher.
- All Commander G3 and i-Mix G3 codecs.

Getting Started at the Studio

To perform a typical codec installation at the studio you will need to:

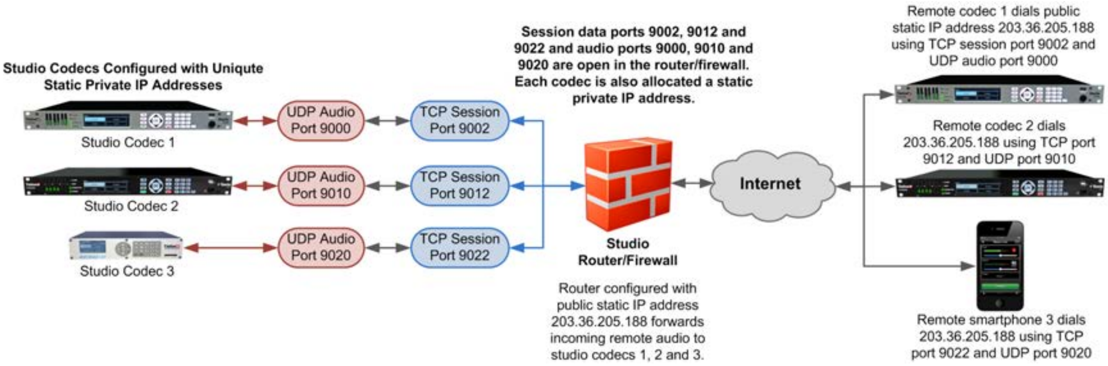
1. Contact your Internet Service Provider and organize a dedicated high speed broadband connection at the studio for your codec with a public static IP address. Do not share this connection with other devices.
2. Install your codec at the studio and attach an active RJ-45 LAN cable to the “LAN” or “Ethernet” port on the rear of the codec. Please note:
   - The green LED underneath the “LAN” or "Ethernet" port will illuminate and the orange LED will flash steadily if you are connected to an active LAN connection.
   - The Genie and Merlin families of IP codecs support two simultaneous Ethernet connections.
3. If you are connecting a single codec to a router without a firewall you can enter the public IP address, Subnet Mask and Gateway directly into the codec and your work is done. Note: your Telco should be able to provide this information.
4. Alternatively, if you are connected to a router with a firewall, configure Network Address Translation (NAT) in your router. NAT is performed between the public internet and your private Local Area Network (LAN) by your router. Your remote codec sends IP data packets to the studio router's public static IP address and the router performs NAT, which forwards these data packets to the private IP address allocated by the router to your codec. As part of this process we recommended you:
   - Connect to your router using a web-browser.
   - Configure it to allocate a static private IP address for each codec.

Important Note: The IP address may change if the codec is allocated a DHCP IP address by the router and it loses power or is temporarily disconnected from the LAN. This will cause problems for remote codecs attempting to dial and connect.

5. Ensure your router's firewall is configured with the relevant TCP and UDP IP ports open to allow data traffic between your codec and the remote codec. The process is fairly simple if you use the following procedure:

a. Connect to your router using a web-browser.
b. Navigate to http://portforward.com/english/applications/port_forwarding/Tieline-G5/default.htm (Note: when configuring a Commander or i-Mix G3 codec at the studio use http://portforward.com/english/applications/port_forwarding/Tieline-G3/default.htm)
c. Click to select your router manufacturer from the list.
d. Next, click to select your router model from the list.
e. Follow the instructions to complete port forwarding

6. Visit www.portforward.com and download the port checking application to verify your router's ports are open.
7. Configure the static IP address in your codec using the instructions in the next section. To allow multiple codecs to share a single public static IP address behind a firewall and route the calls correctly, your codecs and the firewall need to be configured similarly to the example diagram which follows. Ensure the port, IP address, Subnet Mask and Gateway settings in your codecs match those configured in your router.



Port Forwarding to 3 Studio Codecs Sharing a Public Static IP Address

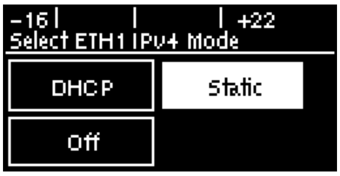
Important Note:
- The most common studio configuration issue is a firewall which blocks the incoming and/or outgoing TCP and UDP ports, or not configuring NAT and port forwarding correctly. The following table lists the firewall ports you need to open for each model of Tieline codec if they are dialing your router at the studio. If the remote codec is also connected to a LAN with a firewall you may also need to open the ports at the remote end of the link to connect successfully.
- Some firewalls require symmetric port configuration. The codec supports configuration of the "send" audio port (codec port at the remote end of the link to which you are sending audio) and "return" audio port (port used by the local codec to receive audio from the remote codec).

Firewall Ports					
Commander G3/i-Mix G3		Bridge-IT/Bridge-IT XTRA		Merlin and Genie Codec Families	
TCP	UDP	TCP	UDP	TCP	UDP
IP1 Session Port: 9002	IP1 Audio Port: 9000	Session Port (Sess): 9002	Audio (Proto): 9000	Session Port: 9002	Audio Port Stream 1: 9000
IP2 Session Port: 9012	IP2 Audio Port: 9010	Web-GUI: 80	SIP Session: 5060	Alternative Session: 9012	Audio Port Stream 2: 9010
Toolbox Software: 5550	Toolbox Software: 5550	Alternative Web-GUI: 8080	SIP Audio: 5004	Web-GUI: 80	Audio Port Stream 3: 9020
	SIP Session: 5060			Alternative Web-GUI: 8080	Audio Port Stream 4: 9030
	SIP Audio: 5004				Audio Port Stream 5: 9040
					Audio Port Stream 6: 9050
					SIP Session: 5060
					SIP Audio: 5004, 5006, 5008, 5010, 5012, 5014

Configuring a Static Public or Private IP Address in Bridge-IT (v.2.x firmware)

To enter a static IP Address into the codec for NAT:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Settings** and press .
3. Use the down navigation button to select **LAN** and press .
4. Select **Eth1** and navigate to **IPv4** mode and press .
5. DHCP is enabled by default. Select **Static** and press .



6. The Static IP address menu is revealed after DHCP is disabled. Use the navigation buttons to select **v4 Static** and press .
7. Use the numeric **KEYPAD** to enter the IP address and press to store the setting. Note: use the * or # buttons to enter the periods in the IP address and use the **RETURN** button to delete any numbers already entered.
8. Enter changes to the **v4 Subnet** (Subnet Mask) or **v4 Gway** (Default Gateway) in the same way if they are required (check with your network administrator for these settings).
9. After all changes have been made use the navigation buttons to scroll to the top of the menu and select **Apply Setting**, then press the OK button to save all changes.
10. From the **Home** screen select **Settings > Unit > Eth1** in the codec menus to ensure the new static IP address has been entered correctly.

Configuring a Static IP Address in Commander G3 and i-Mix G3 Codecs

To set up a static IP address in Commander G3 and i-Mix G3 codecs select **Menu > Configuration > Advanced > LAN settings > IP Setup > Setup > Static > IP Address > [enter IP address] > press OK > Subnet Mask [enter Subnet Mask] > press OK > Gateway [enter Gateway] > press OK > reboot the codec**.

Record IP Address Details

IPv4 Static IP Address	
IP Address	. . .
Subnet Mask	. . .
Default Gateway	. . .
IPv6 Mode: Manual	**(Bridge-IT, Genie and Merlin codecs only)**
IP Address	: : : : : : :
IPv6 Prefix Size	
IPv6 Gateway	: : : : : : :

Getting Connected

Once the studio codec is configured you are now ready to receive an incoming call from the remote codec over the internet. Always dial from the field codec to the studio codec over the internet unless the remote codec is assigned a public static IP address and you know this address.

If you dial the studio using a cell-phone data network at the remote site you will not normally experience any firewall or port blocking issues at the remote end of the link using default Tieline ports.

Troubleshooting: How to Determine Where Firewall Port Blocking is Occurring

If you find you are unable to either send or receive audio between the studio and remote codecs you can use Tieline's Link Quality reading to diagnose where ports are being blocked. LQ can be displayed on the front LCD screen of Tieline's Bridge-IT, Merlin and Genie codecs by selecting **Cxns**, then select the connection you want to view and press the **OK** button. LQ readings are also displayed on the home screen of all Commander and i-Mix G3 codecs.

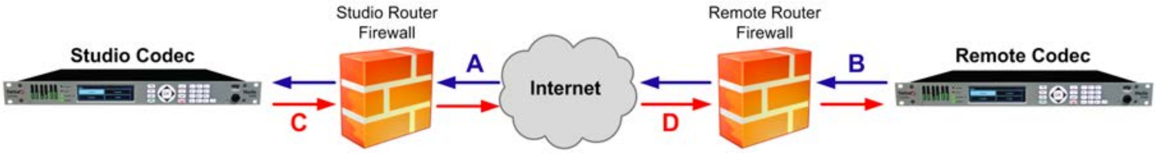
Link Quality (LQ) Readings

Send and Return LQ numbers help you to determine if a problem is occurring at either end of a connection. For example, on an IP connection the Return LQ reading represents the audio being downloaded from the network locally (i.e. audio data is being sent by the remote codec). Conversely, the Send LQ reading represents the audio data being sent by the local codec (i.e. being downloaded by the remote codec). To ensure a stable connection, try to maintain a reliable reading of 80 or higher for both the **Send** and **Return** LQ reading.

Important Note:

- The **Return** link quality reading is the same as the Local (**L**) setting displayed on a G3 codec.
- The **Send** link quality reading is the same as the Remote (**R**) setting displayed on a G3 codec.

Diagnosing Port Blocking via the Studio Codec LQ
If the studio codec **Return** LQ reading is **01** then incoming audio from the remote codec is being blocked by a firewall at either point A or B in the following diagram. If the studio codec **Send** LQ reading is **01** then outgoing audio from the studio is being blocked by a firewall at either point C or D in the following diagram.



Diagnosing Port Blocking via the Remote Codec LQ
If you attach your Tieline codec at the remote site to a LAN with access to the internet you can often dial and connect to the studio without any problem. It is less likely that a firewall will block outgoing TCP and UDP ports. However, if there is a firewall at the remote site it may block incoming data packets from the studio.

The principle is the same at the remote codec for diagnosing blocked ports. If the remote codec **Return** LQ reading is 01 then incoming audio from the studio codec is being blocked by a firewall at either point C or D in the preceding diagram. If the remote codec **Send** LQ reading is 01 then the outgoing audio from the remote codec is being blocked by a firewall at either point A or B in the preceding diagram.

Troubleshooting TCP Port Blocking

Error messages on the codec screen can help to diagnose TCP port blocking.

1. **"Connection Refused"** usually means that the firewall is configured correctly but the codec is not using the expected port. For example, the firewall is set up to forward via port 9002 but codec is 'listening' to port 10,000. “Connection Refused” is not normally shown if the firewall is not configured correctly because a firewall will by design silently drop any forwarding requests to ports that it doesn’t have open (see next point). Note: "Connection Refused" will also be displayed if the Commander G3 or i-Mix G3 codec you are calling is already connected.
2. **“Connection Timeout”** can mean one of two things:
   - The firewall is not configured correctly and the attempted codec connection is being silently dropped, e.g. a remote codec is dialing to port 9002 but the studio firewall port forwarding is not configured.
   - The UDP port is not port forwarded correctly. Tieline codecs send test data during connection establishment to make sure that the audio path is configured correctly; if this process fails then it will also result in a “Connection Timeout”.

How do I determine which end is blocking data flow?

Tieline test codec firewalls have the default Tieline TCP and UDP ports open. You can dial into these test codecs (or other codecs you know are configured correctly) from your recently configured studio and remote codecs and use the LQ readings to diagnose whether your studio or remote codec firewall is blocking your data packets. If one codec connects ok and the other one doesn't, then you will know which end is likely to be causing the problem. As an example:

1. Dial from site 1 to a Tieline test codec.
2. Dial from site 2 to Tieline test codec.

If both of these connect successfully then the “outbound” TCP path for session data is OK, and the inbound UDP audio path is OK.

3. Dial to site 1 from a codec you know is configured correctly.
4. Dial to site 2 from a codec you know is configured correctly.

If either of these calls fail then TCP and/or UDP inbound data is being blocked on the failed connection (see "Troubleshooting TCP Port Blocking" above).

Testing your Codec

- Visit http://www.tieline.com/Support/Test-lines for a list of test IP codec addresses you can use to verify your codec is configured correctly.
- See Testing IP Network Connections for more IP test information.

Learning More About IP Networks

For more IP network information please see the section titled Understanding IP Networks which discusses:

- Private versus public IP addresses.
- Static versus DHCP assigned IP addresses.
- Network Address Translation (NAT), port forwarding and firewalls.

20.2 Understanding IP Networks

Types of IP Addresses Available

	Type of IP Address	How the IP Address is Allocated	Description
Public	Static Public IP Address	Internet Service Providers (ISPs)	ISP's allocate a static public IP address to allow network devices to communicate with each other over the internet. It works like a public telephone number and will allow your remote codec to call your studio codec over the Internet.
	Dynamically Assigned Public IP Address	Internet Service Providers (ISPs)	ISP's usually allocate dynamically (automatically) assigned public IP addresses to allow network devices to communicate with each other over the Internet. (Not recommended for studio installations because each time you connect to your ISP the IP address can change).
Private	Dynamically Assigned Private IP Address	DHCP Server/ Router on your own private LAN network.	A DHCP server-allocated IP address that is automatically assigned to a device on a LAN to allow it to communicate with other devices and the internet. This address can change each time a device connects.
	Static Private IP Address	LAN Administrator	A network administrator-allocated static address which is programmed into a device to allow it to connect to a LAN. Often a security measure to only allow access to devices approved by a network administrator.

Obtaining Public IP Addresses

To send audio streams over the public internet you need to use a public IP address assigned to you by your ISP (Internet Service Provider).

A public IP address is like your public telephone number and allows you to be contacted over the internet in much the same way people dial your public telephone number. They come in two forms; dynamic (DHCP) and static. Most ISPs assign a dynamic public IP address by default, which can often change without you knowing. This is suitable for a quick demo of your Tieline codec, but for a permanent installation you will need to request a permanent static public IP address.

Once the Static Public IP address is assigned to your internet connection (router) at the studio you need to create a link between the public IP address and your codec's private IP address on the LAN. This is called Network Address Translation.

Depending upon how your network is configured, it may also be possible to simply connect your Tieline codec directly into your ADSL modem/router and receive a public address from the router.

Private LAN IP Addresses

By default your Tieline codec will normally be automatically assigned a private IP address when you connect it to a typical router over a LAN.

Private IP Addresses are associated with LANs and normally reside behind a firewall and are not visible to the internet. They are generally in the ranges: 10.0.0.1 – 10.255.255.255, 169.254.0.0 –

169.254.255.255, 172.16.0.0 – 172.31.255.255 and 192.168.0.0 – 192.168.255.255 and are assigned by network DHCP servers and routers.

These IP Addresses are generally assigned for a predefined period (known as a lease) by your network's DHCP server or router. This IP address will generally expire after the lease period. DHCP assigned IP Addresses may also change if the device is disconnected for lengthy periods or if power to the device is turned off and back on. As a result, it is advised that you make this IP address permanent by assigning it as a Static DHCP IP Address. This will ensure you are able to always forward incoming audio packets to your codec using the same private IP address at the studio using port forwarding (see the section on port forwarding for more details). Consult your Network Administrator if you are unsure how to do this.

Network Address Translation (NAT)

Network Address Translation (NAT) is a method of connecting multiple devices to the internet using one public IP address.

The best way to explain NAT is to use the example of a phone system at an office that has one public telephone number and multiple extensions. This type of telephone system allows people to call you on a single public telephone number and performs the translation and routing of the public number to a particular private extension. Similarly, in order to receive an IP call from a remote codec over the public internet, the same network address translation principle applies. NAT and port forwarding allows a single device, such as a broadband router, to act as an agent between the public internet and a local private LAN.

The relationship between public and private IP addresses and NAT is displayed in the following diagram and the following section explains port forwarding configuration in more detail.

Port Forwarding: Tieline TCP and UDP Port Settings

For your Tieline Codec to communicate over the public internet an IP Address alone is not sufficient. In TCP/IP and UDP networks the codec port is the endpoint of your connection. Ports are doorways for IP devices to communicate with each other. Picture a house and imagine the front door is the entry point represented by a public or private IP address. Then you want to get to several codecs in different rooms of the same house and ports represent the doors to each of those rooms. In principle this is how port addressing works.

For example, several codecs may dial into your studio using the same public static IP address. In this situation it is necessary to configure codec 'programs' with audio streams using different audio ports for discretely routing each incoming and outgoing audio stream. By doing this your studio's network routers know where IP packets for each audio stream should be routed, i.e. to which codec and respective audio outputs.

When data packets are received from remote codecs at a particular public IP address, port information is translated from data packets to ensure the correct packets are sent to the correct studio codecs. This process is performed by PAT (Port Address Translation), which is a feature of NAT (Network Address Translation) devices.

Tieline codecs use TCP ports for setting up the communication session and UDP ports for streaming audio. While TCP ports are generally open, UDP ports are generally blocked by network devices which contain firewalls and will stop you delivering your audio. Depending on the codecs you are using, you need to configure your firewall to allow TCP and UDP protocols to pass through the ports listed in the table below.

20.3 Tips for Creating Reliable IP Connections

The following 10 tips are provided to help obtain the best possible IP connection between two codecs, without paying for Quality of Service (QoS).

1. Always use the best quality Internet Service Provider (ISP). Tier 1 service providers are best as their infrastructure actually makes up the internet 'backbone'. Wikipedia lists the major service providers that make up the internet backbone at: http://en.wikipedia.org/wiki/internet_backbone. In Australia Telstra is equivalent to one of these service providers.

2. You will get the best quality connection if both the local (studio) and remote codecs use the same ISP. This can substantially increase reliability, audio bandwidth and reduce audio delay. Using the same service provider nationally can give better results than using different local service providers. This is especially true if one of the service providers is a cheap, low-end domestic service provider, which buys its bandwidth from other ISPs. Second and third tier providers sublease bandwidth from first tier providers and can result in connection reliability issues due to multiple switch hops. We also highly recommend using First Tier ISPs if connecting two codecs in different countries.

3. Sign up for a business plan that provides better performance than domestic or residential plans. Business plans typically have a fixed data limit per month with an additional cost for data beyond that limit. In addition, Service Level Agreements (SLA) will often provide better support and response times in the event of a connection failure. Domestic plans are often speed-limited or "shaped" when usage exceeds a predefined limit. These plans are cheap but they are dangerous for streaming broadcast audio.

4. Ensure that the speed of the connection for both codecs is adequate for the job. The minimum upload speed recommended is 256 kbps for a studio codec and 64 kbps for a field unit connection.

5. Use good quality equipment to connect your codecs to the internet. (Tieline successfully uses Cisco® switching and routing equipment.):

- If you are using a DSL or ADSL connection make sure you purchase a high quality modem that can easily meet your speed requirements. This is especially important if you are over 4 kms from an exchange.
- If you have multiple codecs connected to a local area network (LAN) please ensure that your network infrastructure is designed for media streaming and not domestic usage. Tieline has tested several cheap 8-port switches that lose more packets between local computers than an international IP connection between Australia and the USA!
- If using a wireless connection ensure that the antenna signal strength received is strong. The type of antenna used and the amount of output gain also affects connection quality.

Important Note: You should be able to stream audio between two codecs on your LAN and get 'link quality' readings of S99R99. If you see anything less than this then you should get a network engineer to investigate the issue.

6. Once your internet connection is installed at the studio check that the connection performance is approximately what you ordered and are paying for. A connection can perform below advertised bit rates if:

- There is an error in ISP configuration;
- There is an error in modem configuration;
- There is a poor quality line between the studio and the exchange;
- There are too may phones or faxes connected to the phone line; or
- Line filters have been connected incorrectly.

You can test your internet connection speed by connecting a PC to the internet and using http://

www.speedtest.net/index.php. If the bandwidth detected is low then something is wrong. Get it fixed before going live!

7. Use a dedicated DSL/ADSL line for your codecs. Do not share a link with PCs or company networks. The only exception to this rule is if an organisation has network equipment and engineers that can implement and manage quality of service (QoS) on its network.

8. Use UDP as the preferred audio transport protocol. TCP generally results in lower bit rates and random drop-outs of audio over the internet. Only use TCP if UDP is blocked by firewalls and you are unable to connect.

9. When using UDP ensure the total bit rate (audio bit rate plus header bit rate) is no more than 80% of the ISP connection rate. IP headers require around 20 kbps in addition to the audio bit rate. For example, with a 64 kbps connection the audio bit rate should be (64-20) x 0.8 = 31.2 kbps or lower. For TCP we suggest a limit of 50% or less.

10. Wireless IP connections can easily become congested and result in packet loss and audio drop-outs. It is very difficult to guarantee connection quality when there is no way of knowing how many people are sharing the same wireless connection.

Important Note: Be careful when using cell-phone connections at special events where thousands of people have mobile phones. This can result in poor quality connections and audio drop-outs if cell-phone base stations are overloaded.

IP Connection Checklist

Complete the following check list and aim for a score of at least 8 out of 10 before going live.

Number	Check	Result
1	Using a reputable Tier1 ISP that's part of internet backbone.	
2	The same ISP is being used for both codec connections.	
3	The ISP Plan is a Business Plan or equivalent.	
4	The ISP connection speed is adequate.	
5	Equipment is high quality and suitable for media streaming.	
6	The ISP connection speed has been tested and is suitable.	
7	The ISP connection is not shared with other PCs or devices.	
8	UDP is being used as the audio transport protocol.	
9	No more than 80% of ISP connection bandwidth is being used.	
10	There are no wireless connections being used.	

20.4 Testing IP Network Connections

There are a few very simple tools that you can use to test whether a codec can be reached over an IP network.

- Visit http://www.speedtest.net/ to test the upload and download speed of your IP connections and identify your public IP address.
- Visit www.portforward.com and download the port checking application to verify your router's ports are open. Note: Using a port scanner to test a codec will be unsuccessful if you try to scan and

the port is already in use, i.e. the codec is connected.

- Visit www.subnetonline.com and use an online port scanner to check for open and closed TCP ports. This site also has numerous other software tools, including an online ping web-tool for IPv4, plus TraceRoute and TracePath software tools.

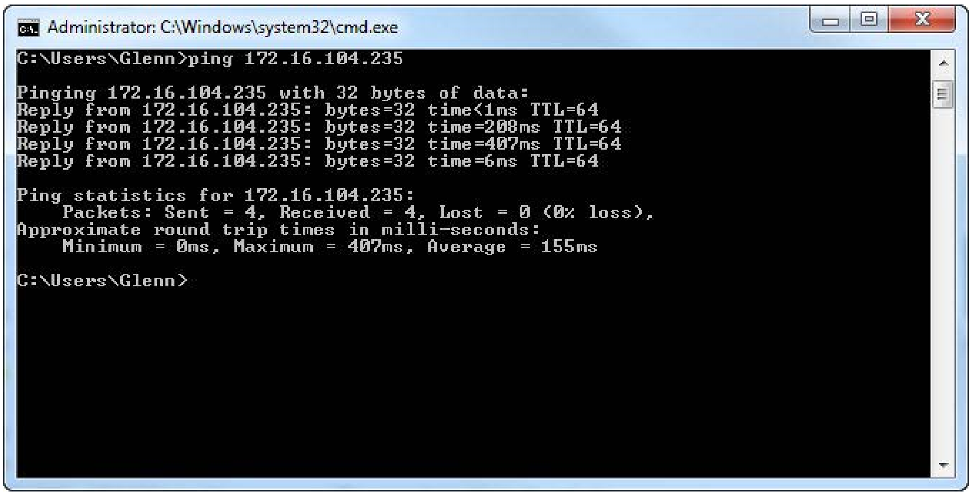
Ping the Codec

A ping test can be used to test whether it is possible to reach a codec or any device over an IP network. A ping test measures:

- The round-trip time of packets.
- Any packet loss.

There are two types of ping tests:

1. **Short test:** sends 4 packets and delivers statistics.
   i. Point to the **start** menu on your PC and click once.
   ii. In the search text box type **Run** and press **Enter.**
   iii. Type **CMD** in the **Run dialog** text box and click **OK**.
   iv. Type **ping** and the IP address of the codec you are pinging (i.e. **ping 192.168.0.159**) and press the **Enter** key on your keyboard.
   v. The round trip time of the packets is displayed, as well as any packet loss.



2. **Long test:** sends packets continuously until stopped.
   i. Point to the **start** menu on your PC and click once.
   ii. In the search text box type **Run** and press **Enter.**
   iii. Type **CMD** in the **Run dialog** text box and click **OK**.
   iv. Type **ping**, the IP address of the codec you are pinging, and then **-t** (i.e. **ping 203.36.205.163 -t**) and press the **Enter** key on your keyboard.
   v. Let the test run for several minutes and then press **CTRL C**.
   vi. The round trip time of the packets is displayed, as well as any packet loss for the period of time that the test occurred.

Trace the Route of IP Packets

Another utility available on your PC is traceroute. This tool can be used to determine the route and number of hops that data packets are taking to their destination (codec). This is useful because the more routers that packets traverse, the more latency your connection will have, and the less reliable it will be.

i. Point to the **start** menu on your PC and click once.
ii. In the search text box type **Run** and press **Enter.**
iii. Type **CMD** in the **Run dialog** text box and click **OK**.
iv. Type **tracert**, the IP address of the codec you are contacting (i.e. **tracert 203.36.205.163**) and press the **Enter** key on your keyboard.

20.5 Software Licences

This product uses a combination of proprietary and open-source software programs. Some of the software included in this product contains copyrighted software that is licensed under various open-source licenses (e.g. GNU General Public License v2, GNU Lesser GPL v2.1). A detailed list of open source licenses used in this product is included in the user manual. This can be downloaded from the Help Panel in the Web Browser Interface or from the Tieline website <http://tieline.com>. You may request a copy for the open source software on DVD by contacting our support team on +61 (0)8 9249 6688. Tieline Pty Ltd will charge a small handling fee for distribution of this software.

Some of the open source software of this product is based on the works of the Gentoo project and is not directed, managed, sold or supported by Gentoo Foundation, Inc. The Gentoo name is a trademark of Gentoo Foundation, Inc.

Open Source GPL compatible Licenses:

o Some of the open-source software in the product is licensed under GPL version 3. A copy of the license can be obtained at http://www.gnu.org/licenses/gpl.html.

o Some of the open-source software in the product is licensed under GPL version 2. A copy of the license can be obtained at http://www.gnu.org/licenses/old-licenses/gpl-2.0.html.

o Some of the open-source software in the product is licensed under LGPL version 3. A copy of the license can be obtained at http://www.gnu.org/licenses/lgpl.html.

o Some of the open-source software in the product is licensed under LGPL version 2.1. A copy of the license can be obtained at http://www.gnu.org/licenses/old-licenses/lgpl-2.1.html.

Open Source BSD style Licenses:

- bind:

o Portions: Copyright (c) 1987, 1990, 1993, 1994 The Regents of the University of California. All rights reserved. Additional clause - All advertising materials mentioning features or use of this software must display the following acknowledgment: This product includes software developed by the University of California, Berkeley and its contributors.

o Portions: Copyright (c) 2004 Masarykova universita (Masaryk University, Brno, Czech Republic) All rights reserved.

o Portions: Copyright (c) 1997 - 2003 Kungliga Tekniska Högskolan (Royal Institute of Technology, Stockholm, Sweden). All rights reserved.

o Portions (2 clause BSD license, 3rd clause removed): Copyright (c) 1998 Doug Rabson. All rights reserved.

o Portions: Copyright ((c)) 2002, Rice University. All rights reserved.

o Portions: Copyright 2000 Aaron D. Gifford. All rights reserved.

o Portions (2 clause BSD license, 3rd clause removed): Copyright (c) 1998 Doug Rabson. Copyright (c) 2001 Jake Burkholder. All rights reserved.

o Portions: Copyright (C) 1995, 1996, 1997, and 1998 WIDE Project. All rights reserved.

o Portions: Copyright (c) 2000-2002 Japan Network Information Center. All rights reserved.

o idnkit: Copyright (c) 2000-2002 Japan Network Information Center. All rights reserved.

o zkt: Copyright (c) 2005 - 2008, Holger Zuleger HZnet. All rights reserved.

- dhcpcd - 2 clause BSD license, clause 3 removed

o Copyright (c) 2006-2011 Roy Marples <roy@marples.name>

- eventlog

o Copyright (c) 2003 BalaBit IT Ltd.

- file - 2 clause BSD license, clause 3 removed

o Copyright (c) Ian F. Darwin 1986, 1987, 1989, 1990, 1991, 1992, 1994, 1995.

o Software written by Ian F. Darwin and others;

o maintained 1994- Christos Zoulas.

o This software is not subject to any export provision of the United States

Department of Commerce, and may be exported to any country or planet.

- glibc

o Code incorporated from 4.4 BSD: Copyright (C) 1991 Regents of the University of California. All rights reserved.

o Sun RPC support (from rpcsrc-4.0): Copyright (c) 2010, Oracle America, Inc.

- htop

o Copyright (c) 2004-2006 The Trustees of Indiana University and Indiana University Research and Technology Corporation. All rights reserved.

o Copyright (c) 2004-2005 The Regents of the University of California. All rights reserved.

o Copyright (c) 2007 Cisco Systems, Inc. All rights reserved.

o Portions: Copyright (c) 2004-2005 The University of Tennessee and The University of Tennessee Research Foundation. All rights reserved

o Portions: Copyright (c) 2004-2005 High Performance Computing Center Stuttgart, University of Stuttgart. All rights reserved.

o Portions: Copyright (c) 2006, 2007 Advanced Micro Devices, Inc. All rights reserved.

- less - 2 clause BSD license, clause 3 removed

o Copyright (C) 1984-2011 Mark Nudelman

- libpcre

o Basic Library Functions: Copyright (c) 1997-2010 University of Cambridge. All rights reserved.

o C++ Wrapper Functions: Copyright (c) 2007-2010, Google Inc. All rights reserved

- libuuid

o Copyright (c) 1996, 1997, 1998, 1999, 2007. Theodore Ts'o.

- lighttpd

o Copyright (c) 2004, Jan Kneschke, incremental. All rights reserved.

- net-snmp

o Copyright 1989, 1991, 1992 by Carnegie Mellon University. All rights reserved.

o Derivative Work - 1996, 1998-2000

o Copyright 1996, 1998-2000 The Regents of the University of California. All rights reserved.

o Copyright (c) 2001-2003, Networks Associates Technology, Inc. All rights reserved.

o Portions of this code are copyright (c) 2001-2003, Cambridge Broadband Ltd. All rights reserved.

o Copyright © 2003 Sun Microsystems, Inc., 4150 Network Circle, Santa Clara,

California 95054, U.S.A. All rights reserved.

o Copyright (c) 2003-2010, Sparta, Inc. All rights reserved.

o Copyright (c) 2004, Cisco, Inc and Information Network. Center of Beijing University of Posts and Telecommunications. All rights reserved.

o Copyright (c) Fabasoft R&D Software GmbH & Co KG, 2003. oss@fabasoft.com. Author: Bernhard Penz <bernhard.penz@fabasoft.com>

o Copyright (c) 2007 Apple Inc. All rights reserved.

o Copyright (c) 2009, ScienceLogic, LLC. All rights reserved.

- openrc - 2 clause BSD license, clause 3 removed

o Copyright (c) 2007-2009 Roy Marples <roy@marples.name>

- OpenSSH

o Copyright (c) 1995 Tatu Ylonen <ylo@cs.hut.fi>, Espoo, Finland. All rights reserved.

o 32-bit CRC compensation attack detector: Copyright (c) 1998 CORE SDI S.A., Buenos Aires, Argentina. All rights reserved.

o ssh-keyscan: Copyright 1995, 1996 by David Mazieres <dm@lcs.mit.edu>.

o One component of OpenSSH source code: Copyright (c) 1983, 1990, 1992, 1993, 1995. The Regents of the University of California. All rights reserved.

o Remaining components under 2 clause BSD (clause 3 removed) Copyright holders: Markus Friedl, Theo de Raadt, Niels Provos, Dug Song, Aaron Campbell, Damien Miller, Kevin Steves, Daniel Kouril, Wesley Griffin, Per Allansson, Nils Nordman, Simon Wilkinson

o Parts of portable version under 2 clause BSD (clause 3 removed) Copyright holders: Ben Lindstrom, Tim Rice, Andre Lucas, Chris Adams, Corinna Vinschen, Cray Inc., Denis Parker, Gert Doering, Jakob Schlyter, Jason Downs, Juha Yrjölä, Michael Stone, Networks Associates Technology, Inc., Solar Designer, Todd C. Miller, Wayne Schroeder, William Jones, Darren Tucker, Sun Microsystems, The SCO Group, Daniel Walsh, Red Hat, Inc.

o Parts of openbsd-compat: Copyright holders: Todd C. Miller, Theo de Raadt, Damien Miller, Eric P. Allman, The Regents of the University of California, Constantin S. Svintsoff.

- OpenSSL: crypto/blowfish, crypto/des

o Copyright (C) 1995-1997 Eric Young (eay@cryptsoft.com).

o Clause 3: All advertising materials mentioning features or use of this software must display the following acknowledgement: This product includes software developed by Eric Young (eay@cryptsoft.com).

- strace:

o Copyright (c) 1991, 1992 Paul Kranenburg <pk@cs.few.eur.nl>

o Copyright (c) 1993 Branko Lankester <branko@hacktic.nl>.

o Copyright (c) 1993 Ulrich Pegelow <pegelow@moorea.uni-muenster.de>.

o Copyright (c) 1995, 1996 Michael Elizabeth Chastain <mec@duracef.shout.net>.

o Copyright (c) 1993, 1994, 1995, 1996 Rick Sladkey <jrs@world.std.com>.

o Copyright (C) 1998-2001 Wichert Akkerman <wakkerma@deephackmode.org>..

• util-linux: text-utils

Redistribution and use in source and binary forms, with or without modification, are permitted provided that the following conditions are met:

1. Redistributions of source code must retain the above copyright notice, this list of conditions and the following disclaimer.

2. Redistributions in binary form must reproduce the above copyright notice, this list of conditions and the following disclaimer in the documentation and/or other materials provided with the distribution.

3. Neither the name of the <ORGANIZATION> nor the names of its contributors may be used to endorse or promote products derived from this software without specific prior written permission.

THIS SOFTWARE IS PROVIDED BY THE COPYRIGHT HOLDERS AND CONTRIBUTORS "AS IS" AND ANY EXPRESS OR IMPLIED WARRANTIES, INCLUDING, BUT NOT LIMITED TO, THE IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE ARE DISCLAIMED. IN NO EVENT SHALL THE COPYRIGHT OWNER OR CONTRIBUTORS BE LIABLE FOR ANY DIRECT, INDIRECT, INCIDENTAL, SPECIAL,EXEMPLARY, OR CONSEQUENTIAL DAMAGES (INCLUDING, BUT NOT LIMITED TO, PROCUREMENT OF SUBSTITUTE GOODS OR SERVICES; LOSS OF USE, DATA, OR PROFITS; OR BUSINESS INTERRUPTION) HOWEVER CAUSED AND ON ANY THEORY OF LIABILITY, WHETHER IN CONTRACT, STRICT LIABILITY, OR TORT (INCLUDING NEGLIGENCE OR OTHERWISE) ARISING IN ANY WAY OUT OF THE USE OF THIS SOFTWARE, EVEN IF ADVISED OF THE POSSIBILITY OF SUCH DAMAGE.

Open Source MIT style Licenses:

• glibc: DNS resolver taken from BIND 4.9.5

• ncurses

o install-sh : 1994 X Consortium

• OpenSSH

o Portions of code under MIT-style license to the copyright holders: Free Software Foundation, Inc.

Permission is hereby granted, free of charge, to any person obtaining a copy of this software and associated documentation files (the "Software"), to deal in the Software without restriction, including without limitation the rights to use, copy, modify, merge, publish, distribute, sublicense, and/or sell copies of the Software, and to permit persons to whom the Software is furnished to do so, subject to the following conditions:

The above copyright notice and this permission notice shall be included in all copies or substantial portions of the Software.

THE SOFTWARE IS PROVIDED "AS IS", WITHOUT WARRANTY OF ANY KIND, EXPRESS OR IMPLIED, INCLUDING BUT NOT LIMITED TO THE WARRANTIES OF MERCHANTABILITY,FITNESS FOR A PARTICULAR PURPOSE AND NONINFRINGEMENT. IN NO EVENT SHALL THE AUTHORS OR COPYRIGHT HOLDERS BE LIABLE FOR ANY CLAIM, DAMAGES OR OTHER LIABILITY, WHETHER IN AN ACTION OF CONTRACT, TORT OR OTHERWISE, ARISING FROM,OUT OF OR IN CONNECTION WITH THE SOFTWARE OR THE USE OR OTHER DEALINGS IN THE SOFTWARE.

Open Source ISC style Licenses:

• bind

o Portions: Copyright (C) 1996-2001 Nominum, Inc.

o Portions: Copyright (C) 1995-2000 by Network Associates, Inc.

o Portions: Copyright (C) 2002 Stichting NLnet, Netherlands, stichting@nlnet.nl.

o Dynamically Loadable Zones (DLZ) contributer: Rob Butler.

o Portions: Copyright (c) 1993 by Digital Equipment Corporation.

O Portions: Copyright (c) 1999-2000 by Nortel Networks Corporation.

O Portions: Copyright (C) 2004 Nominet, Ltd.

O Portions: Copyright RSA Security Inc.

O Portions: Copyright (c) 1996, David Mazieres <dm@uun.org>, Copyright (c) 2008, Damien Miller <djm@openbsd.org>

- expat

o Copyright (c) 1998, 1999, 2000 Thai Open Source Software Center Ltd and Clark Cooper.

o Copyright (c) 2001, 2002, 2003, 2004, 2005, 2006 Expat maintainers.

- libffi

o Copyright (c) 1996-2011 Anthony Green, Red Hat, Inc and others.

- OpenSSH

o Portions of code under ISC-style license to the copyright holders: Internet Software Consortium, Todd C. Miller, Reyk Floeter, Chad Mynhier.

- popt

o Copyright (c) 1998 Red Hat Software.

- vixie-cron

O Copyright 1988,1990,1993 by Paul Vixie. All rights reserved.

o Copyright (C) 2004-2011 Internet Systems Consortium, Inc. ("ISC")

o Copyright (C) 1997,2000 by Internet Software Consortium, Inc.

THE SOFTWARE IS PROVIDED "AS IS" AND THE COPYRIGHT HOLDERS AND CONTRIBUTORS DISCLAIM ALL WARRANTIES WITH REGARD TO THIS SOFTWARE INCLUDING ALL IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS. IN NO EVENT SHALL THE COPYRIGHT HOLDERS AND CONTRIBUTORS BE LIABLE FOR ANY SPECIAL, DIRECT, INDIRECT, OR CONSEQUENTIAL DAMAGES OR ANY DAMAGES WHATSOEVER RESULTING FROM LOSS OF USE, DATA OR PROFITS, WHETHER IN AN ACTION OF CONTRACT, NEGLIGENCE OR OTHER TORTIOUS ACTION, ARISING OUT OF OR IN CONNECTION WITH THE USE OR PERFORMANCE OF THIS SOFTWARE.

Open Source UCB License:

- util-linux

This product includes software developed by the University of California, Berkeley and its contributors.

Copyright (c) 1989 The Regents of the University of California.

All rights reserved.

THIS SOFTWARE IS PROVIDED BY THE REGENTS AND CONTRIBUTORS ``AS IS'' AND ANY EXPRESS OR IMPLIED WARRANTIES, INCLUDING, BUT NOT LIMITED TO, THE IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE ARE DISCLAIMED. IN NO EVENT SHALL THE REGENTS OR CONTRIBUTORS BE LIABLE FOR ANY DIRECT, INDIRECT, INCIDENTAL, SPECIAL, EXEMPLARY, OR CONSEQUENTIAL DAMAGES (INCLUDING, BUT NOT LIMITED TO, PROCUREMENT OF SUBSTITUTE GOODS OR SERVICES; LOSS OF USE, DATA, OR PROFITS; OR BUSINESS INTERRUPTION) HOWEVER CAUSED AND ON ANY THEORY OF LIABILITY, WHETHER IN

CONTRACT, STRICT LIABILITY, OR TORT (INCLUDING NEGLIGENCE OR OTHERWISE) ARISING IN ANY WAY OUT OF THE USE OF THIS SOFTWARE, EVEN IF ADVISED OF THE POSSIBILITY OF SUCH DAMAGE.

Open Source OpenSSL License:

- OpenSSL

o "This product includes software developed by the OpenSSL Project for use in the OpenSSL Toolkit (http://www.openssl.org/)"

o "This product includes cryptographic software written by Eric Young (eay@cryptsoft.com)"

Copyright (c) 1998-2011 The OpenSSL Project. All rights reserved.

Redistribution and use in source and binary forms, with or without modification, are permitted provided that the following conditions are met:

1. Redistributions of source code must retain the above copyright notice, this list of conditions and the following disclaimer.

2. Redistributions in binary form must reproduce the above copyright notice, this list of conditions and the following disclaimer in the documentation and/or other materials provided with the distribution.

3. All advertising materials mentioning features or use of this software must display the following acknowledgment: "This product includes software developed by the OpenSSL Project for use in the OpenSSL Toolkit. (http://www.openssl.org/)"

4. The names "OpenSSL Toolkit" and "OpenSSL Project" must not be used to endorse or promote products derived from this software without prior written permission. For written permission, please contact openssl-core@openssl.org.

5. Products derived from this software may not be called "OpenSSL" nor may "OpenSSL" appear in their names without prior written permission of the OpenSSL Project.

6. Redistributions of any form whatsoever must retain the following acknowledgment: "This product includes software developed by the OpenSSL Project for use in the OpenSSL Toolkit (http://www.openssl.org/)"

THIS SOFTWARE IS PROVIDED BY THE OpenSSL PROJECT ``AS IS'' AND ANY EXPRESSED OR IMPLIED WARRANTIES, INCLUDING, BUT NOT LIMITED TO, THE IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE ARE DISCLAIMED. IN NO EVENT SHALL THE OpenSSL PROJECT OR ITS CONTRIBUTORS BE LIABLE FOR ANY DIRECT, INDIRECT, INCIDENTAL, SPECIAL, EXEMPLARY, OR CONSEQUENTIAL DAMAGES (INCLUDING, BUT NOT LIMITED TO, PROCUREMENT OF SUBSTITUTE GOODS OR SERVICES; LOSS OF USE, DATA, OR PROFITS; OR BUSINESS INTERRUPTION) HOWEVER CAUSED AND ON ANY THEORY OF LIABILITY, WHETHER IN CONTRACT, STRICT LIABILITY, OR TORT (INCLUDING NEGLIGENCE OR OTHERWISE) ARISING IN ANY WAY OUT OF THE USE OF THIS SOFTWARE, EVEN IF ADVISED OF THE POSSIBILITY OF SUCH DAMAGE.

Original SSLeay License:

This product includes cryptographic software written by Eric Young (eay@cryptsoft.com). This product includes software written by Tim Hudson (tjh@cryptsoft.com).

Copyright (C) 1995-1998 Eric Young (eay@cryptsoft.com)

All rights reserved.

This package is an SSL implementation written by Eric Young (eay@cryptsoft.com).

The implementation was written so as to conform with Netscapes SSL.

This library is free for commercial and non-commercial use as long as the following conditions are aheared to. The following conditions apply to all code found in this distribution, be it the RC4, RSA, lhash, DES, etc., code; not just the SSL code. The SSL documentation included with this distribution is covered by the same copyright terms except that the holder is Tim Hudson (tjh@cryptsoft.com).

Copyright remains Eric Young's, and as such any Copyright notices in the code are not to be removed.

If this package is used in a product, Eric Young should be given attribution as the author of the parts of the library used. This can be in the form of a textual message at program startup or in documentation (online or textual) provided with the package.

Redistribution and use in source and binary forms, with or without modification, are permitted provided that the following conditions are met:

1. Redistributions of source code must retain the copyright notice, this list of conditions and the following disclaimer.

2. Redistributions in binary form must reproduce the above copyright notice, this list of conditions and the following disclaimer in the and/or other materials provided with the distribution.

3. All advertising materials mentioning features or use of this software must display the following acknowledgement: "This product includes cryptographic software written by Eric Young (eay@cryptsoft.com)". The word 'cryptographic' can be left out if the rouines from the library being used are not cryptographic related :-).

4. If you include any Windows specific code (or a derivative thereof) from the apps directory (application code) you must include an acknowledgement: "This product includes software written by Tim Hudson (tjh@cryptsoft.com)"

THIS SOFTWARE IS PROVIDED BY ERIC YOUNG ``AS IS'' AND ANY EXPRESS OR IMPLIED WARRANTIES, INCLUDING, BUT NOT LIMITED TO, THE IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE ARE DISCLAIMED. IN NO EVENT SHALL THE AUTHOR OR CONTRIBUTORS BE LIABLE FOR ANY DIRECT, INDIRECT, INCIDENTAL, SPECIAL, EXEMPLARY, OR CONSEQUENTIAL DAMAGES (INCLUDING, BUT NOT LIMITED TO, PROCUREMENT OF SUBSTITUTE GOODS OR SERVICES; LOSS OF USE, DATA, OR PROFITS; OR BUSINESS INTERRUPTION) HOWEVER CAUSED AND ON ANY THEORY OF LIABILITY, WHETHER IN CONTRACT, STRICT LIABILITY, OR TORT (INCLUDING NEGLIGENCE OR OTHERWISE) ARISING IN ANY WAY OUT OF THE USE OF THIS SOFTWARE, EVEN IF ADVISED OF THE POSSIBILITY OF SUCH DAMAGE.

Open Source netperf License:

Copyright (C) 1993 Hewlett-Packard Company

ALL RIGHTS RESERVED.

THE SOFTWARE AND DOCUMENTATION IS PROVIDED "AS IS". HEWLETT-PACKARD COMPANY DOES NOT WARRANT THAT THE USE, REPRODUCTION, MODIFICATION OR DISTRIBUTION OF THE SOFTWARE OR DOCUMENTATION WILL NOT INFRINGE A THIRD PARTY'S INTELLECTUAL PROPERTY RIGHTS. HP DOES NOT WARRANT THAT THE SOFTWARE OR DOCUMENTATION IS ERROR FREE. HP DISCLAIMS ALL WARRANTIES, EXPRESS AND IMPLIED, WITH REGARD TO THE SOFTWARE AND THE DOCUMENTATION. HP SPECIFICALLY DISCLAIMS ALL WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE. HEWLETT-PACKARD COMPANY WILL NOT IN ANY EVENT BE LIABLE FOR ANY DIRECT, INDIRECT, SPECIAL, INCIDENTAL OR CONSEQUENTIAL DAMAGES (INCLUDING LOST PROFITS) RELATED TO ANY USE, REPRODUCTION, MODIFICATION, OR DISTRIBUTION OF THE SOFTWARE OR DOCUMENTATION.

Open Source Info-ZIP license:

20.6 Compliances and Certifications

FCC Compliance Notice

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 may cause harmful interference, in which case the user will be required to correct the interference at his/her own expense. There is no guarantee, however, that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment to an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio or TV technician for help.

CAUTION:
Changes or modifications to this unit not expressly approved by the party responsible for compliance could void the user's authority to operate this equipment.

Declaration of Conformity

The Tieline Bridge-IT IP codec meets the requirements of directives for CE and C-Tick certifications. Technical documentation required by the conformity assessment procedure is kept at the head office of Tieline Technology; 1/25 Irvine Drive, Malaga, Western Australia 6090.

EN 55 022 Statement

This is to certify that Tieline Bridge-IT is shielded against the generation of radio interference in accordance with the application of EN 55 022: 2006 Class A. Technical documentation required by the conformity assessment procedure is kept at the head office of Tieline Technology; 1/25 Irvine Drive, Malaga, Western Australia 6090.

Canadian Department of Communications Radio Interference Regulations

This digital apparatus (Tieline Bridge-IT) does not exceed the Class A limits for radio-noise emissions from digital apparatus as set out in the Radio Interference Regulations of the Canadian Department of Communications.

Règlement sur le brouillage radioélectrique du ministère des Communications

Cet appareil numérique (Tieline Bridge-IT) respecte les limites de bruits radioélectriques visant les appareils numériques de classe A prescrites dans le Règlement sur le brouillage radioelectrique du ministère des Communications du Canada.

Safety of Electrical and Electronic Products and Components

The IECEE CB Scheme is an international system for mutual acceptance of CB test reports and certificates covering the safety of electrical and electronic products and components. The IEC CB is a multilateral scheme among participating countries and certification organisations, based on the use of international (IEC) standards.

This product has been tested by an independent certifying company and has been certified to comply with IEC 60950-1(ed.2);am1

20.7 Trademarks and Credit Notices

1. Windows is a registered trademark of Microsoft Corporation in the United States and/or other countries.
2. Windows XP and Vista are either trademarks or registered trademarks of Microsoft Corporation in the United States and/or other countries.
3. Firefox is a registered trademark of Mozilla Corporation in the United States and/or other countries.
4. Solaris is a trademark of Sun Microsystems Inc. in the United States and/or other countries.
5. Linux is the registered trademark of Linus Torvalds in the U.S. and other countries.
6. Google™ is the registered trademark of Google Inc in the United States and/or other countries.
7. iPod® is a registered trademark of Apple Inc., registered in the U.S. and other countries.
8. Java is a trade mark Sun Microsystems Inc. in the United States and/or other countries.
9. Other product names mentioned within this document may be trademarks or registered trademarks, or a trade name of their respective owner.

21 Specifications

Input/Output Specifications	
Analog Audio Inputs	2 x Female XLR (Channel 1 mic/line; channel 2 line only)
Analog Audio Outputs	2 x Male XLR
AES3 In	1 x female XLR (Channel 1 in; shared with Ch1 analog input)
AES3 Out	1 x male XLR
Headphones	1 x 6.35mm (1/4") Jack on rear panel
Control Ports In/Out	Two relay inputs and two opto-isolated outputs for machine control via Phoenix connector
Audio Input Impedance	High Impedance > 5K ohm
Output Impedance	<50 ohm Balanced
Clipping Level	+22dBu (input and outputs)
24 bit A/D & D/A Converters	
Frequency Response	20Hz to 20kHz
Total Harmonic Distortion	<0.0039% at +16dBu, or -88dBu unweighted
Signal To Noise Ratio	>90dB at +22dBu, unweighted
Sample Frequencies	
IP Sample Frequencies	16kHz, 32kHz, 44.1kHz, 48kHz
Algorithms	
IP	Opus, Tieline Music, Tieline MusicPLUS, G.711, G.722, MPEG Layer 2
IP (Pro version only)	AAC-LD, AAC-ELD, AAC-LC, AAC-HE, AAC-HEv2
IP (Pro version only)	16/24 bit Enhanced apt-X
IP (uncompressed)	Linear PCM16
Data and Control Interfaces	
USB	USB 2.0 (Type B) host port on the rear panel
LAN	10/100 base T RJ45 connector
Serial	RS232 up to 115kbps with or without CTS/RTS flow control via female DB9 connector, can be used as a proprietary data channel
Front Panel Interfaces	
Display	128 x 64 monochrome LCD
SD/SDHC Card Slot	Supports SDHC Flash Cards
Keypad	20 button keypad
Navigation	5 button keypad
General	
Dimensions	8.5” x 5.9” x 1.75” (216mm x 150mm x 44mm)
Weight	2.07lb/940g
Power Consumption	12V DC, 400mA
Operating Temperature	0°C to 50°C (32°F to 122°F)
Humidity Operating Range	20% ≤RH ≤70% (0 to 35°C), non-condensing
Internal Battery	Panasonic CR2032, 3V coin type 20mm

Index

- A -

- B -

- C -

- Q -

- R -

- S -

- T -

- U -

- V -

- W -