Bridge-IT IP Audio Codec
User Manual

Codec Firmware v.1.12.xx
Manual Version 1.3_20131211 En
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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 point-to-point, 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.
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.
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 muliticast 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.

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 point-to-point 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
- Cable Head End IP Audio Distribution
- Low-latency audio over IP bridging solutions
- Multiple codec installations (2 codecs fit in 1 x 19" rack unit)
- IP multicasts over compatible IP networks

Codec Features

- Compact design, with two codecs fitting into a 1RU mounting bracket.
- Suitable for point-to-point and multi-point IP connections over a variety of IP networks.
- Linear audio with suite of high quality broadcast algorithms as standard, plus optional LC-AAC and HE-AAC.
- Tieline's loss-tolerant MusicPLUS algorithm provides up to 20 kHz stereo audio quality with 20ms coding delay at bit-rates as low as 96kbps - making it ideal for today's IP and 3G networks.
- Tieline Music can deliver up to 15 kHz FM quality mono audio at bit-rates as low as 24Kbps, with only 20 milliseconds encoding delay.
- SmartStream software for automatically managing jitter buffering, forward error correction and packet repair.
- Full hardware front panel interface including navigation, LCD display, PPM metering and dialing key pad.
- Web-GUI for configuring codec functionality and RS232 data.
- Broadcast quality analog XLR inputs/outputs.
- XLR digital AES3 (AES/EBU) input/output.
- Simultaneous analog and digital AES3 (AES/EBU) audio outputs.

- Onboard SDHC card slot for automatic audio fail over.
- ¼” (6.35mm) stereo headphone output.
- 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.
- USB slave connection for codec configuration.
- Multilingual language options supported.

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

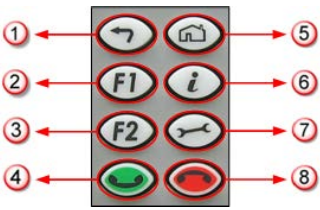
Bridge-IT has four arrow shaped navigation buttons for navigating codec menus and an **OK**  button for selecting menu items.

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

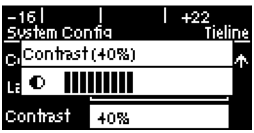


	Features	Operation Button Descriptions
1	Return Button	Press to navigate back through menus & delete characters
2	Function Button 1	Press to open codec user functions
3	Function Button 2	Press to open codec user functions
4	Dial Button	Press to dial IP connections
5	Home Button	Press to return to home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

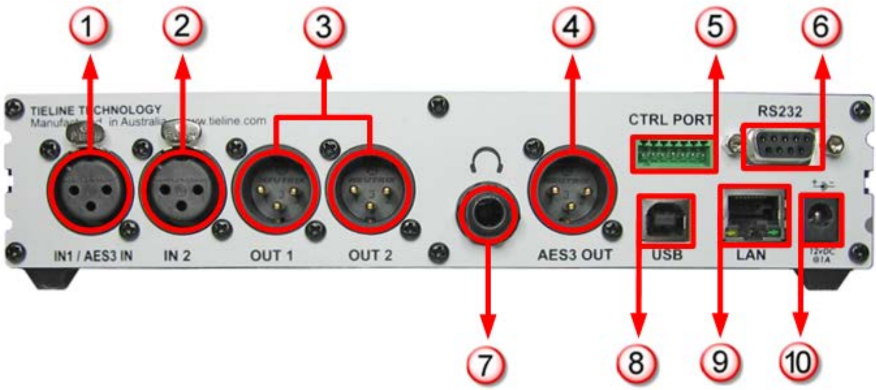
Adjusting LCD Screen Contrast Levels

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 **Config**, then **System**, and using the down button to navigate to **Contrast**.



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:

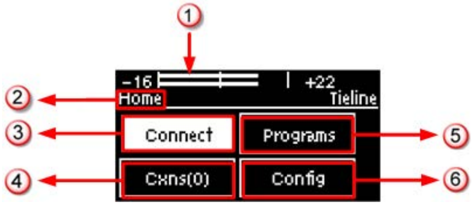
1. 2 relay inputs and 2 opto-isolated outputs for machine control via the **CTRL PORT**.
2. A nine pin **RS-232** connection for local and remote control of equipment at either end of the link
3. A **USB** 2.0 (slave) connection for codec web-GUI configuration.

DC Power Input

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

7 Navigating Codec Menus

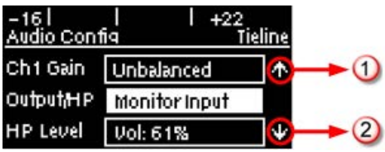
The codec has simple and intuitive menu navigation screens. All main codec menus can be launched from the **Home** screen and audio levels remain visible throughout all menus.



Bridge-IT Home Screen

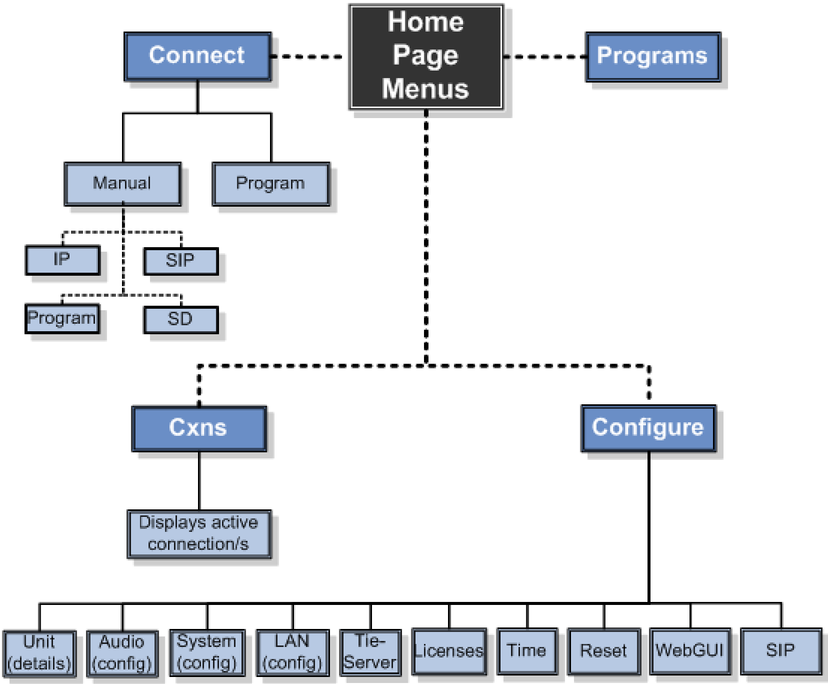
	Features	Codec Home Screen Elements
1	PPM Meters	Left (top) and right channel audio levels
2	Screen Name	The name of the current screen being navigated
3	Connect	Select to dial & adjust connection settings
4	Cxns	Displays the number of current connections
5	Programs	View and edit Program dialing configurations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

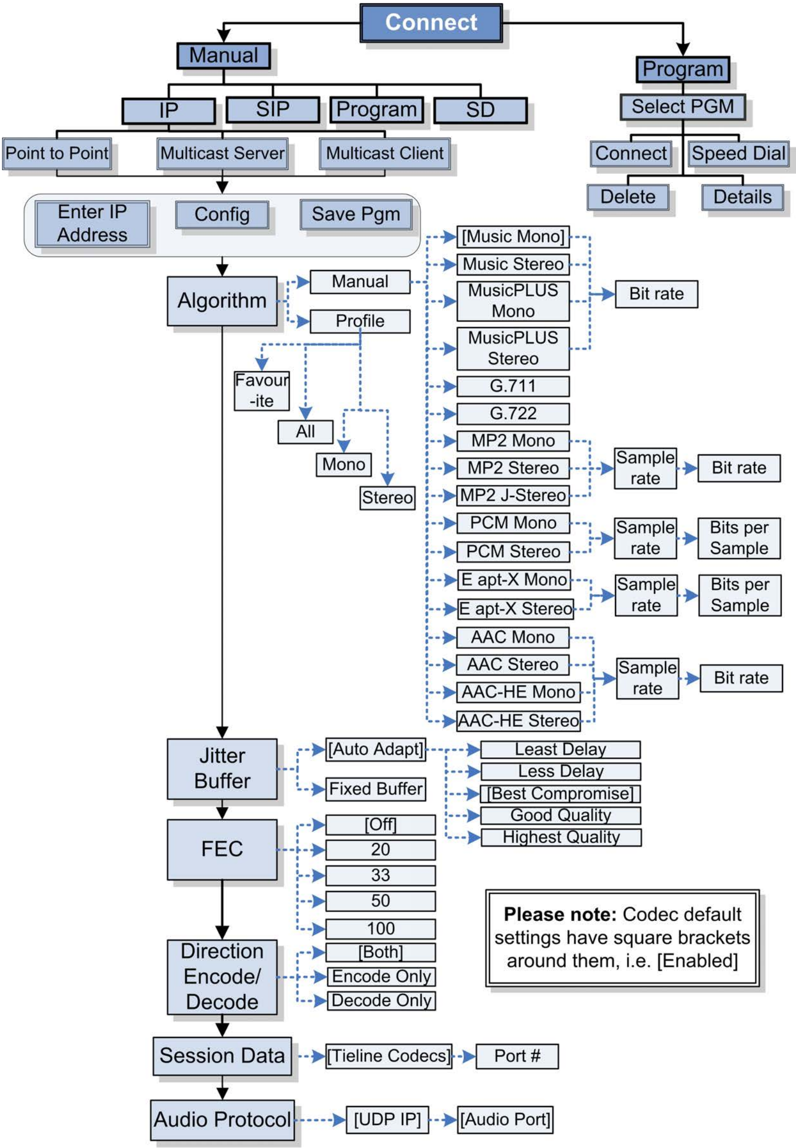


	Features	Codec Home Screen Elements
1	Up Arrow	Arrow indicating menus can scroll upwards
2	Down Arrow	Arrow indicating menus can scroll downwards

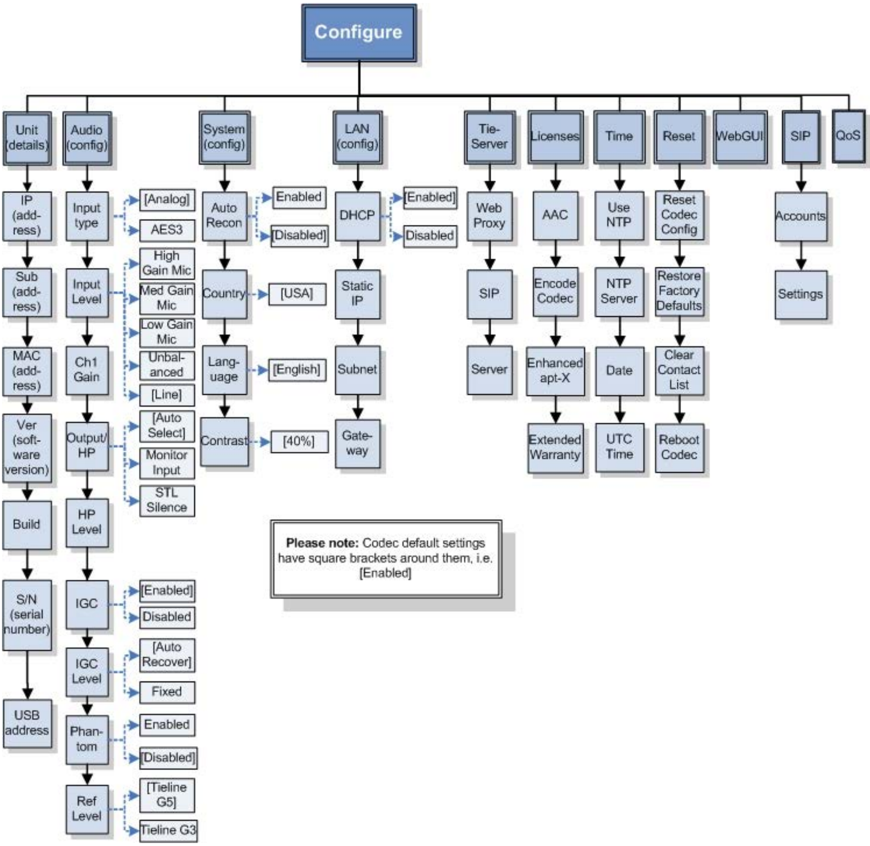
Codec Menu Overview



Connect Menu



Configure 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.

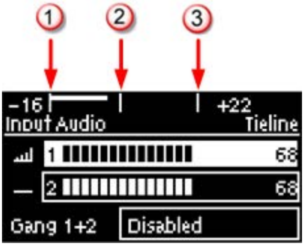
Mono and Stereo Audio Capabilities

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 Config** menu and connect audio to input 1 of the codec. Input audio is copied 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.

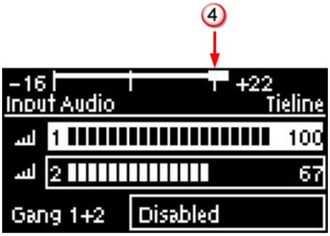
It doesn't matter whether you attach analog or digital audio to the codec, you still get both analog and digital audio out. The only difference occurs when you send mono analog audio using a mono algorithm setting. In this scenario the audio is not sent on both AES3 output channels like it is on the analog outputs.

Audio Metering when Connecting to Tieline Bridge-IT

Tieline codecs are automatically configured to connect to each other using the correct audio meter scales. If you are connecting two Bridge-IT codecs the audio scale displayed on the codec screen is between -16dBu and +22dBu. These 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



Some other brands of professional broadcast codecs use this audio scale. To configure this setting manually:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Config** and press the button.
3. Navigate to **Audio** and press .
4. Navigate to **Ref Level** and press .
5. Select **Tieline G5** and press .

Setting Audio Levels

When connected to stereo sources the top PPM meter is the left channel and the bottom PPM meter is the right channel. 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.

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 in the codec:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Config** and press the button.
3. Navigate to **Audio** and press .
4. 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 in the codec:

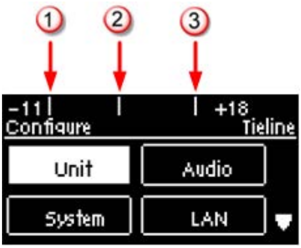
1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Config** and press the button.
3. Navigate to **Audio** and press .
4. Navigate to **IGC Level** and press to toggle between **Auto** and **Fixed**.

Configuring Audio Metering when Connecting Bridge-IT to Tieline G3 Codecs

The codec has more audio headroom than G3 audio codecs and metering needs to be adjusted when connecting between a Bridge-IT IP codec and a Tieline G3 codec. The G3 metering scale is is between -11dBu and +18dBu. Tieline codecs perform this metering adjustment automatically when they connect to each other. Other codecs don't and the audio scale adjustment will need to be configured manually if the default audio scale is not suitable:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Config** and press the OK button.
3. Navigate to **Audio** and press OK.
4. Navigate to **Ref Level** and press OK.
5. Select **Tieline G3** and press OK.

Audio levels should average around the nominal 0vu point and audio peaks should not exceed +16dbu as indicated by the PPM meter.



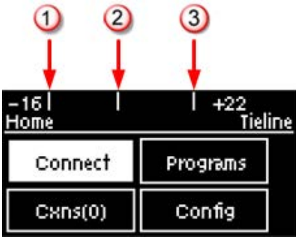
	Features	Description
1	-11dBu	PPM meter low point
2	+4dBu	Nominal 0vu reference level at +4dBu
3	+16dBu	+16 indication where audio will clip/distort

Audio Metering when Connecting to Other Codecs

This setting is mainly used for compatibility with other codecs. In particular it is used when connecting using G.711, G.722 and AAC algorithms.

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Config** and press the button.
3. Navigate to **Audio** and press .
4. Navigate to **Ref Level** and press .
5. Select **Other** and press .

Audio levels should average around the nominal 0vu point and audio peaks should not exceed +12dbu as indicated by the PPM meter.



	Features	Description
1	-9dBu	PPM meter low point
2	+4dBu	Nominal 0vu reference level at +4dBu
3	+12dBu	+12 indication where audio will clip/distort

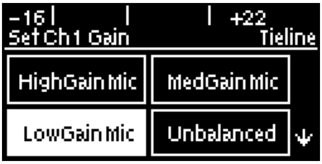
Channel 1 Mic/Line Level Audio Adjustment

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

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Config** and press the button.
3. Navigate to **Audio** and press .
4. Ensure **Input Type** is set to **Analog**.

5. Use the arrow-down button to highlight and select the **Ch 1 Gain** setting and press the button.

6. Use the navigation buttons to select the appropriate gain setting and press the OK button to save the setting.



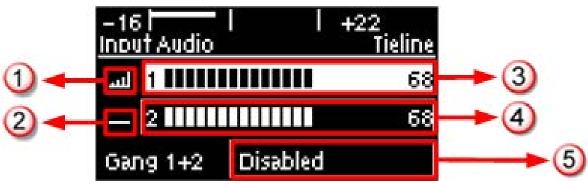
Important Note: 15 volt phantom power is not supplied to input 1 by default. To turn this on:

1. Select **Config,** then **Audio** and use the arrow-down ▽ button to highlight the **Phantom** setting.
2. Press the OK 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 Levels

1. Press the F1 button and the right ▷ arrow button to open the **Input Audio** level adjustment screen.
2. Press 1 on the numeric keypad to toggle channel 1 on and off and press 2 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

Ganging of Audio Channels

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

1. Press the F1 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 OK 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.

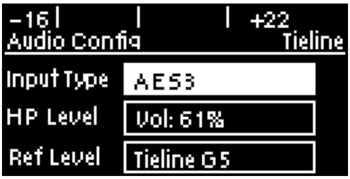
When channels 1 and 2 are ganged together:

- 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 XLR input or output is required for sending and receiving digital AES3 data. To configure the codec to accept AES3 data signals:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Config** and press the button.
3. Navigate to **Audio** and press .
4. 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

When you are not connected, you can adjust the AES3 output sample rate manually in the **Audio** menu via the **AES3 SR** setting. This will determine the output sample rate of the AES3 XLR output and the sample rate used by the standby SD card when a connection is lost. Note: the sample rate of the recording on the SD card called "fallback.mp3" file must match the sample rate of the **AES3 SR** setting in the **Audio** menu of the codec to playback successfully.

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 **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Config** and press the OK button.
3. Navigate to **Audio** and press OK.
4. Navigate to **AES3 SR** and press OK.
5. Select a preferred sample rate then press OK.

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** sampling rate setting are the same.

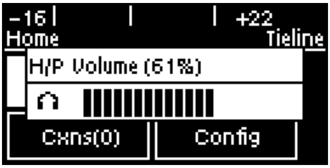
10 Headphone Monitoring

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

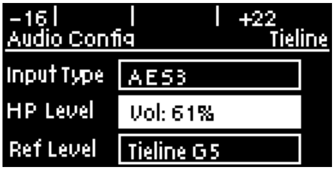
Important Note: When the codec makes a connection it will monitor the decoded return program audio link by default.

Adjusting Headphone Output Levels

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 pressing the **HOME** button, selecting **Config**, then **Audio**, and using the down button to navigate to **HP Level** and press .



Mono Connections

When a mono algorithm is selected a single channel of audio is sent to both of the program outputs. This provides input audio monitoring in both the left and right sides of the headphone output.

Stereo Connections

When a stereo algorithm is selected audio from channel one is sent to the left output and audio from channel two is sent to the right output.

Monitoring Input Audio and Return Program Audio

Auto Select is the default **Output/HP** monitoring setting in the codec. In this mode when the codec is not connected over IP the headphone output monitors input audio, which is sent to both the left and right audio outputs. Once the codec connects over IP it automatically switches to monitor decoded incoming audio. This is useful for confidence monitoring and receiving mix-minus IFB splits.

Two other audio monitoring options are also available:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Config** and press the button.
3. Navigate to **Audio** and press .
4. Navigate to **Output/HP** and press to select either **Monitor Input** or **STL Silence**.

Always Monitor Input Audio

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 program.

Important Note: Monitoring of input audio is not possible with AES3 connections once you have 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. Once the codec connects over IP it automatically switches to monitor 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 **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Config** and press the OK button.
3. Navigate to **System** and press OK.
4. Use the navigation buttons to select **Language** and press OK.
5. Select a language and press OK.

12 Getting Connected Quickly

The codec supports point-to-point, multi-unicast stereo connections to up to 6 end-points and multicast stereo connections over compatible IP networks.

How to Connect using 'Program' Dialing

The codec uses the concept of 'program' dialing to connect over point-to-point, multicast or multi-unicast connections. Essentially a program is like a supercharged 'contact' because it is 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 and dialing information for a range of point-to-point, multicast and multi-unicast connections and retrieve or edit them easily at the touch of a button. Simple point-to-point or multicast profiles can be created using the codec front panel, whereas multi-unicasts must be created with the **Connect panel** in the Toolbox web-GUI.

Point-to-Point Programs

New point-to-point programs can be created using the codec front panel keypad (see 10 Quick Steps to Connect Bridge-IT), or using the Toolbox web-GUI (see Configuring Point-to-Point Programs). If you know the IP address of the codec you want to connect to then all you have to do to connect is enter this into the codec, choose your preferred connection settings and then dial to connect.

Manually dialed point-to-point connections are automatically saved as programs - retaining all the dialing and configuration information configured into the codec. Ensure that you configure all the correct connection settings first as these are stored as part of the program's profile and once stored cannot be adjusted without using the **Connect panel** within the Toolbox web-GUI.

Multi-unicast Programs:

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

Multicast Programs:

Multicasts can be used to broadcast to multiple 'subscribers' who wish to connect to a multicast stream. 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. New programs can be created using either the codec front panel or the Toolbox web-GUI. (For front panel configuration see Creating Multicast Connections, or for web-GUI configuration see Configuring a Multicast Program).

Preparing to Connect

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

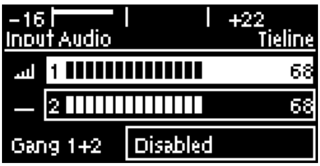
1. Attach the supplied 12 volt power supply to the codec.
2. Attach an RJ45 Ethernet cable to the **LAN** port on the rear panel of the codec.
3. Attach headphones to the 6.35mm (1/4") headphone jack on the rear panel of the codec.
4. Check that the correct country is selected in the codec.
   i. Press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

12.1 10 Quick Steps to Connect Bridge-IT

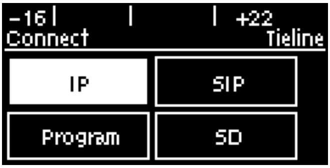
Important Note: The following information pertains to connecting Bridge-IT point-to-point for the very first time without using the Toolbox web-GUI to create custom programs. If you want to connect using the web-GUI, see Configuring Point-to-Point Programs or Configuring Multi-Unicast Programs for more information about creating programs. See Dialing Custom Programs for connection information when dialing with the Bridge-IT codec front panel controls.

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.

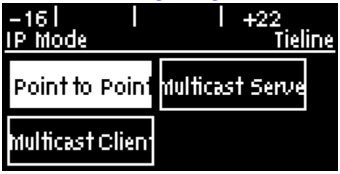
Important Note: 15 volt phantom power is not supplied to input 1 by default. To adjust this setting select **Config**, then **Audio** and then **Phantom**. Press the **OK** button to toggle between **Disabled** and **Enabled**.



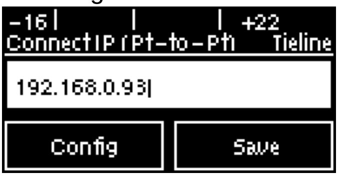
2. Press the **HOME** button to return to the **Home** screen, select **Connect** and press the button, then select **IP** and press the button.



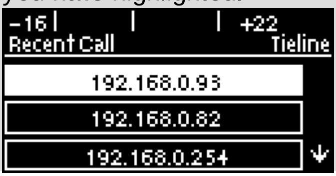
3. Select the **IP mode** you are using to connect, in this case **Point-to-Point**, then press the button (see also WebGUI [Configuration of Multi-Unicast Programs]() and [Configuring Multicast Programs]()).



4. Use the **RETURN** button to delete numbers already entered, then 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. Next, press the down navigation button to select **Config** and press .



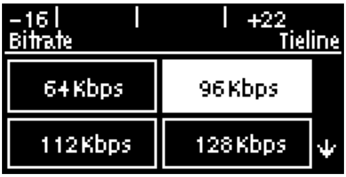
Important Note: The codec remembers recent IP addresses just like a cell-phone. To view these addresses just press the **OK** button when you select the **Connect IP** screen. The most recent addresses and programs are listed first and you can use the navigation buttons to scroll up and down. Press the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



4. Press the down navigation button to select **Alg** (algorithm) and press .

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

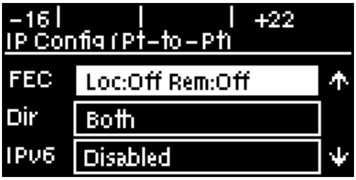
6. If you decide to manually configure the algorithm, use the navigation buttons to select your preferred algorithm sample rate (if displayed) and bit rate, pressing after each option is selected.



7. Press the down navigation button to select Jitt (jitter buffer delay) and press to select a different automatic jitter buffer setting for your connection, or to enter a fixed buffer setting in milliseconds (maximum 999 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 (forward error correction) 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 **DIAL** 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 F2 then **3** on the numeric **KEYPAD**; for lower bit-rates press F2 then **9**.

If you have difficulty connecting, please see [tips for creating reliable IP connections]() and Troubleshooting in this user manual. More support is available at [www.tieline.com/support]() or contact Tieline at [support@tieline.com]() for more assistance.

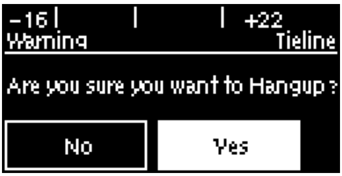
12.2 Dialing Custom Programs

Custom programs are simple to 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 **DIAL** button to make a connection.
4. The **Wait Connecting** screen appears during the connection process and then connection details are displayed.

12.3 Hanging up a Connection

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



12.4 Redialing a Connection

Pressing the **DIAL** button from any codec menu, except menus accessed via the **Connect** > **IP** screen, will let you redial previously dialed connections. When a manually dialed point-to-point or multi-unicast connection is dialed, it is stored as a 'program' in call history.

Manually dialed point-to-point and multi-unicast connections that have not been saved as programs will also be saved as programs - retaining all the dialing and configuration information programmed into the codec - just like a saved program. A program appears in the **Recent Program** redial screen with its saved name and manually dialed connections display IP addresses. A multi-unicast connection will display the first IP address dialed and the number of additional connections dialed.

	Screen Display	Description
1	Manual point-to-point 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 mode** and the codec assumes you want to dial a new manual connection. Pressing the **DIAL** button when the **Connect IP** screen is displayed will retrieve previously dialed IP addresses. Codec settings for a connection dialed from this screen will include the current settings in the **Config** menu, which can be accessed via this screen.

12.5 Configuring Auto Reconnect

Auto Reconnect is disabled in Bridge-IT IP codecs. When enabled, Auto Reconnect will ensure the dialing codec attempts to reconnect if audio is temporarily lost over an IP connection. This setting should only be configured on the dialing codec. To adjust the setting:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Config** and press the button.
3. Navigate to **System** and press .
4. Select **Auto Recon** and press to toggle between **Disabled** and **Enabled**.

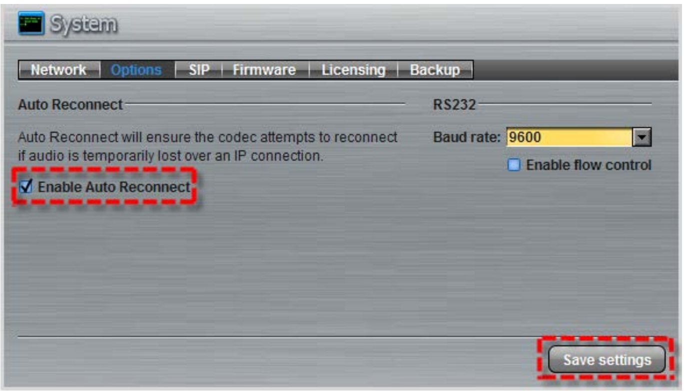
Important Notes:
- If **Auto Reconnect** is enabled, the dialing codec will attempt to connect to the remote codec when the connection is lost. This will continue until the connection has been hung up by the dialing codec.
- If a disconnection occurs during a multi-unicast connection, the dialing codec will attempt to reconnect to any lost connections without interrupting existing connections.

Enabling Auto Reconnect using the web-GUI

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

2. Click the **Options** button at the top of the **System panel**.

3. Click the **Enable Auto Reconnect** check-box to select this option, then click **Save settings**. Note: deselect the **Enable Auto Reconnect** check-box to disable Auto Reconnect.



12.6 Speed Dialing Connections

Important Note: It is necessary to configure a speed dial number into a program prior to 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 **DIAL** button to dial and connect.

12.7 Choosing Dialing Profiles

the codec has a number of preconfigured mono and stereo dialing profiles available. These can be used to configure the codec quickly without individually selecting algorithms and bit-rates etc. These profiles have been configured 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. Use the down navigation button to select **Config** and press the button.
5. Press the button to select **Alg**.
6. Use the right navigation button to select **Profile**.
7. Select the profile you want from the **Favorite**, **All**, **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	All	Displays all preconfigured mono and stereo profiles within the codec
3	Mono	Displays preconfigured mono profiles within the codec
4	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. Use the down navigation button to select **Config** and press the button.
5. Press the button to select **Alg**.
6. Use the right navigation button to select **Profile**.
7. Select the profile you want from the **All**, **Mono or Stereo** menus.
8. 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 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. Use the down navigation button to select **Config** and press the button.

5. Press the OK button to select **Alg**.
6. Use the right navigation button to select **Profile**.
7. Select the profile you want to delete from the **Favorite** menus.
8. 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 algorithm profiles are not available unless an AAC license has 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	Enhanced apt-X	Mono	32 (16 bit)	128
8	Enhanced apt-X	Mono	48 (24 bit)	288
9	Enhanced apt-X	Stereo	32 (16 bit)	256
10	Enhanced apt-X	Stereo	48 (24 bit)	576
11	G.711	Mono	8	64
12	G.722	Mono	16	64
13	MPEG 1 Layer 2	J-Stereo	32	128
14	MPEG 1 Layer 2	J-Stereo	48	192
15	MPEG 1 Layer 2	Mono	24	64
16	MPEG 1 Layer 2	Mono	48	256
17	MPEG 1 Layer 2	Stereo	32	128
18	MPEG 1 Layer 2	Stereo	48	256
19	Music	Mono	32	28.8
20	Music	Mono	32	48
21	Music	Stereo	32	64
22	Music	Stereo	32	96
23	MusicPLUS	Mono	48	48

24	MusicPLUS	Mono	48	96
25	MusicPLUS	Stereo	48	96
26	MusicPLUS	Stereo	48	128
27	MusicPLUS	Stereo	48	192
28	PCM Mono	Mono	48 (16bit)	768
29	PCM Stereo	Stereo	48 (16bit)	1,540

12.8 Creating Multicast Connections

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:

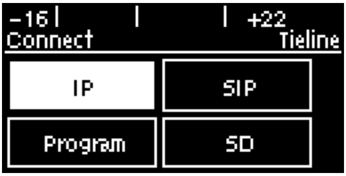
- Bridge-IT firmware v.1.01.00 r4219 or higher.
- G3 codec firmware v.1.6.56 or higher (if connecting to a G3 codec).
- ToolBox web-GUI v.1.2.2.3 or higher.
- A multi-unicast license installed in the dialing codec (Note: the Multi-Unicast license includes multicast server capability).

Important Notes:

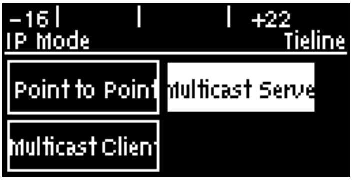
- When a connection is dialed Tieline codecs normally use session data to configure settings like the algorithm, connection bit-rate and sample rate setting etc. Multicast connections do not use session data and it is imperative that all codecs are configured with the same connection settings prior to 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 program because it never receives audio packets.

(See Toolbox web-GUI documentation for more detailed information about Configuring a Multicast Program)

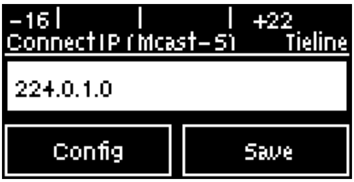
1. Press the **HOME** button to return to the **Home** screen, select **Connect,** then select **IP** and press the button.



2. Select the **IP mode** you are using to connect, in this case **Multicast Server** because we are configuring the broadcast server codec (or select **Multicast Client** when configuring codecs receiving multicast audio streams).



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. Next, press the down navigation button to select **Config** and press .



4. Press the down navigation button to select **Alg** (algorithm) and press .

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

6. 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 OK 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 OK to save the program. A confirmation message is displayed after the program is saved.

7. After you have created multicast server and client programs and loaded them onto your codecs you can dial multicast connections.

8. 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 **DIAL** button to make a connection. The **Wait Connecting** screen appears during the connection process.

9. Next select the multicast client program on each of the multicast client codecs and dial the multicast IP address to begin receiving multicast audio packets.

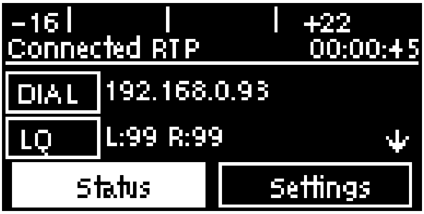
You can navigate to **Cxns** on the **Home** screen to view a codec's connection **Status**, then press to view connection statistics for IP packets being sent over the connection.

Important Notes:

- Always dial the multicast server codec connection first before connecting multicast client codecs.
- Multicast client codecs will display Local link quality (LQ) readings only. Multicast server codecs do not display LQ readings.
- The UDP audio port setting is 9000 for multicast connections and cannot be adjusted.
- Forward Error Correction (FEC) is not available for multicast connections.
- It is not possible to send auxiliary data using multicast connections.
- It is not possible to connect to a G3 codec and receive multicast IP audio streams.

12.9 Monitoring IP Connections

The **Connected RTP** screen displays information that assists in monitoring the performance of IP connections. This screen is displayed when an IP connection is dialed. The IP address that has been 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 and the amount of jitter buffer delay over the IP network.



You can also navigate to this screen from other menus:

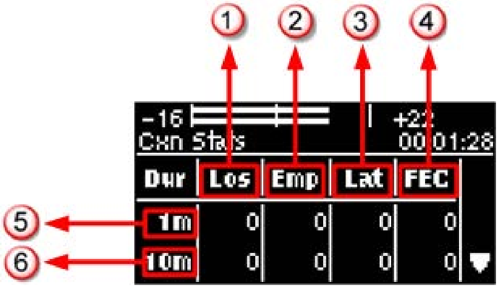
1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **CXNS** and press the button.

Link Quality (LQ) Readings

Local and Remote LQ numbers can also help you to determine if a problem is occurring at both ends of the connection or only one. For example, on an IP connection the "L" or "Local" reading represents the audio being downloaded from the network locally (i.e. audio data is being sent by the remote codec). Conversely, the "R" or "Remote" link quality reading represents the audio being downloaded by the remote codec (i.e. audio data is being sent by the local codec).

Viewing Connection Statistics

Press the button when viewing the **Connected RTP** screen to display the **Cxn Stats** (connection statistics) screen, which displays the performance of the codec in sending IP audio packets across the network. Analysis is assessed over 60 seconds and 10 minutes of connection time.



	Feature	Description
1	Lost Packets	Packets sent and 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 settings such as the jitter buffer, FEC or the connection bit rate are changed, it is best to assess a minute of recent connection performance in preference to 10 minutes of historical performance. 10 minutes of data will include historical data, which includes connection settings that may no longer be relevant. It is only when a connection is hung up that 'packet arrival history' is cleared.

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 dropouts using 3G cell 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.

Monitoring Multi-unicast Connections

The best way to monitor multi-unicast connections is to use the Toolbox web-GUI. The **Cxns** section on the **Home** screen also displays the status of all codec connections. It displays whether all multi-unicast connections have successfully connected. Use the navigation buttons to select **Cxns** and press OK to see more connection details.

12.10 SDHC Card Backup

The codec features an SD/SDHC card slot for automatic backup to MP2 or MP3 recordings if an IP connection is interrupted.

SDHC Card Requirements

1. A FAT32 formatted SDHC Card is required (SD cards may be less reliable and are not recommended).
2. Create an MP2 or MP3 file with the following properties:
   - File name "fallback.mp3"
   - 32kHz, 44.1kHz or 48kHz sample rate. IMPORTANT: the sample rate of the recording on the SD card file must match the sample rate of the **AES3 SR** setting in the **Audio** menu of the codec to playback successfully.
   - 128 Kbps constant bit rate
3. Copy the file into the root directory of the SD card.
4. Place the card into the slot on the front panel of the codec.

Important Note: When creating files please ensure audio levels match the audio scale of your codec connection and peaks average at the correct levels; IGC is only available on codec audio inputs and not SDHC card playback (see Adjusting Input Levels).

Using Backup

File playback commences automatically when a connection is lost, or the remote codec disconnects the connection. It will not occur if you hang up the connection locally. Backup will occur if:

1. The LAN cable is removed locally or there is a network problem.

2. Audio streaming stops for 30 seconds or more (i.e. no audio packets are being received).

The codec **Home** screen indicates SDHC card backup has occurred by displaying **(F)** in the **Cxns** display. The audio file will play continuously in loop mode until a new connection is created. SDHC card playback continues as you attempt to dial a new connection and only ceases after successful reconnection.

Important Note: 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 Connecting to the Web-GUI

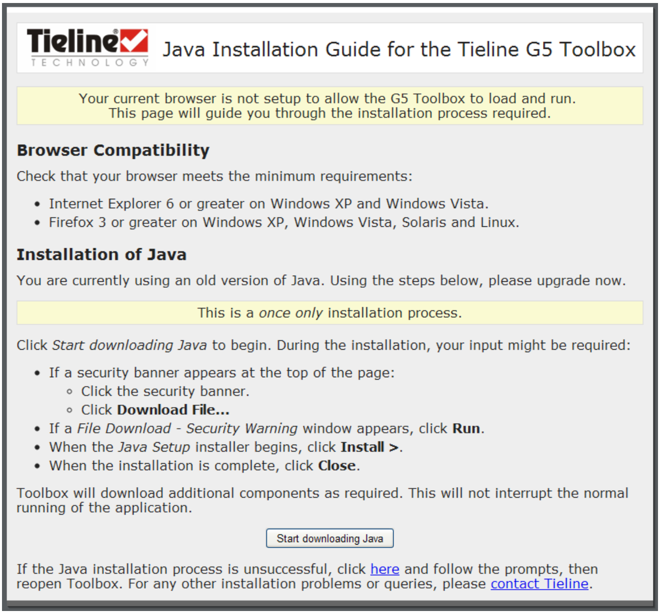
The codec can be configured using the Toolbox web-GUI and this can be launched using either an IP/LAN connection or a USB connection with the codec. Instructions for using the web-GUI are contained in the application itself from the **Help** panel. The Tieline Bridge-IT web-GUI application runs on:

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

See LAN Troubleshooting for more info about connecting the PC LAN connection and port settings.

Web-GUI Prerequisites

1. To use the 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 the following screen is displayed when attempting to launch the ToolBox web-GUI.



2. If you are connecting using the USB port on the codec you also need to install USB drivers, which can be downloaded from the Bridge-IT page via www.tieline.com/support.

13.1 Launching the GUI over a LAN

1. Attach an Ethernet cable to the **LAN** port on your codec.
2. From the codec **Home** screen select **Config** then **Unit** to display the IP address configured 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, i.e. **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 application 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 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 and have greater security

during live broadcasts.

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. In Windows XP or Vista:

1. Open Internet Explorer.
2. Click **Tools** and then click the **Connections** tab.
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 Bridge-IT.

1. Press the **HOME** button on the codec to return to the **Home** screen.
2. Use the navigation buttons to select **Config** 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 (Note: there is no character limit for passwords).
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.

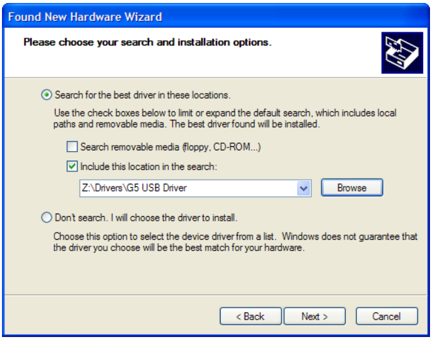
13.2 Installing USB Drivers

USB drivers need to be installed on your PC in order to connect successfully to the codec using the USB port. These can be downloaded from http://www.tieline.com/Support

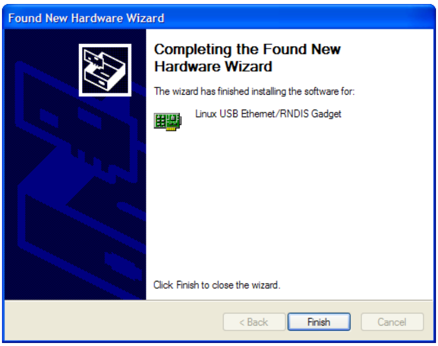
To install drivers:

1. Download the zipped USB driver ".inf" file from the Bridge-IT page link at http://www.tieline.com/Support.
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**.

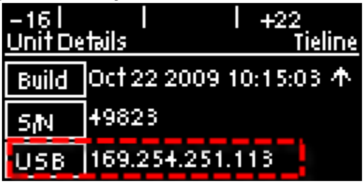


7. When the drivers have been installed click the **Finish** button.



13.3 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 select **Config** then **Unit** to display the **USB** address 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 application 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).

13.4 Changing the Default Password

The default password for the Toolbox web-GUI is **password**. This has to be entered to use the web-GUI and Tieline highly recommends changing the default password as soon as possible to protect your codec from being tampered with during live broadcasts.

Codecs that are 'visible' over the public internet can be accessed by anyone who connects to the codec's public IP address. Setting a strong password protects your equipment from being tampered with and jeopardizing live broadcasts.

Creating a New Password

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

1. Press the **HOME** button on the codec to return to the **Home** screen.
2. Use the navigation buttons to select **Config** and press the button.
3. Use the navigation button to navigate down to **WebGUI** and press the button.
4. Select **Password** and press .
5. 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).

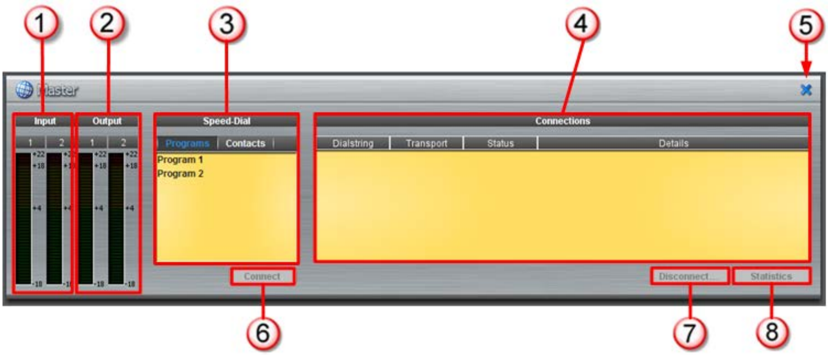
If you forget the password for the Toolbox web-GUI then you can always navigate to the **WebGUI Settings** screen to view the current password and change it.

14 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 configuration panels available within the 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.

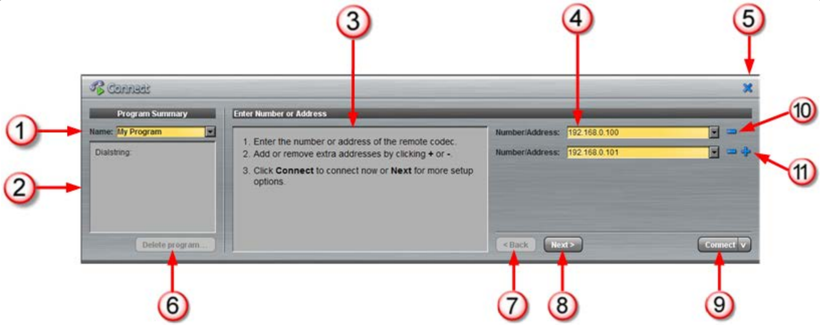
Master Panel for Viewing Contacts and Connections



	Feature	Description
1	Input PPMs	Displays PPM audio levels for inputs 1 and 2
2	Output PPMs	Displays PPM audio levels for outputs 1 and 2
3	Program and Contact list	Lists all contacts configured into the codec
4	Connections	Provides a summary of connection details

5	**Close** button	Click to close the **Master panel**
6	Connect button	Click to dial the selected contact
7	Disconnect button	Click to disconnect the selected connection
8	Statistics	Click to view the selected connection's statistics

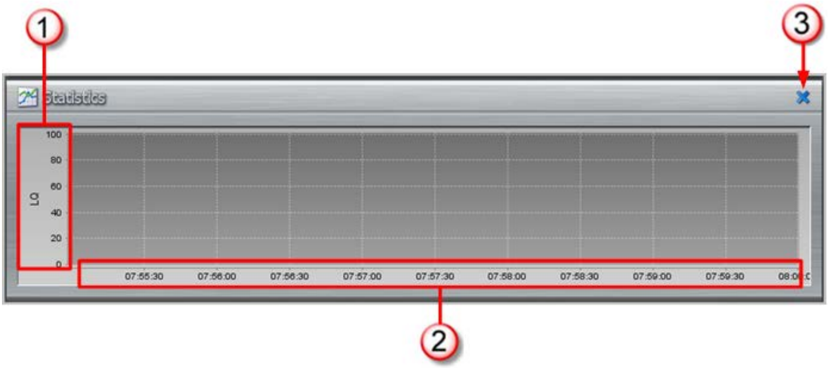
Connect Panel for Dialing IP Connections



	Feature	Description
1	Program **Name**	Displays the currently selected program. Click the drop-down arrow to display all available programs and select a new program.
2	Summary Pane	Details of the currently selected program are displayed.
3	Instruction Pane	Step-by-step instructions for configuring connections.
4	Dialing text box	The first of several configuration selections available in the **Connect panel**.
5	**Close** button	Click to close the **Connect panel**.
6	**Delete Program** button	Press to permanently delete the selected program.
7	**Back** button	Click to navigate to the previous screen.

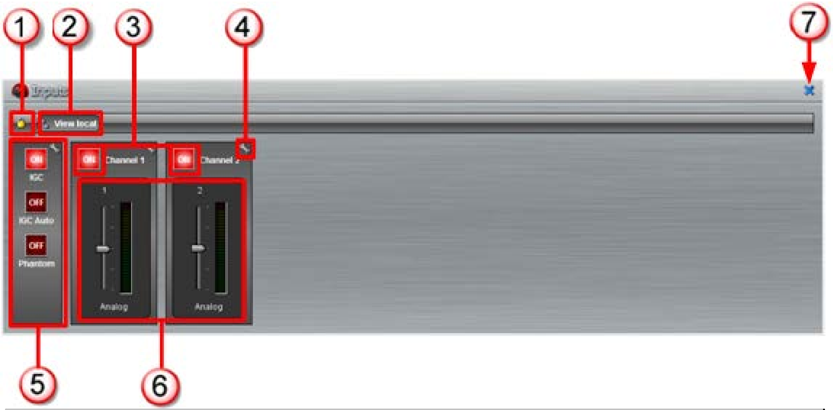
8	**Next** button	Click to navigate to the next screen.
9	**Connect** button	Click to dial a connection and click the arrow button to save a program.
10	**Delete connection** button	Click to delete the number/ address adjacent to the button.
11	**Add Connection** button	Click to add a new dialing text box for additional multi-unicast connections.

Statistics Panel for Viewing Link Quality



	Feature	Description
1	Link Quality (LQ)	Displays the IP link quality of a connection as a percentage, aim for over 80%
2	Timescale	Displays the timeframe for the link quality displayed
3	**Close** button	Click to close the **Statistics panel**

Inputs Panel for Adjusting Audio Input Levels

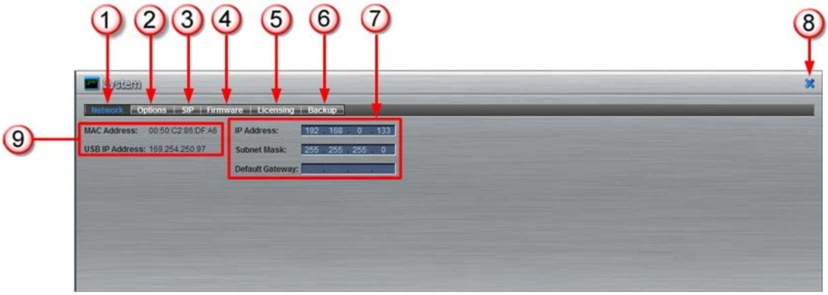


	Feature	Description
1	Lock Button	Click to lock all **Input** pane settings (greys out when locked)
2	View Button	Indicates which channels are currently displayed
3	Global Input Control Buttons	**IGC**, **IGC Auto** and **Phantom** power **ON**/**OFF** buttons
4	Configure symbol	Click to adjust input control and configuration settings
5	Channel **ON/OFF** Buttons	Click to turn each channel **ON** or **OFF**
6	Input Sliders/Faders	Input gain control sliders/faders
7	**Close** button	Click to close the **Inputs** pane

Rules Panel for Creating Relay Activation Rules

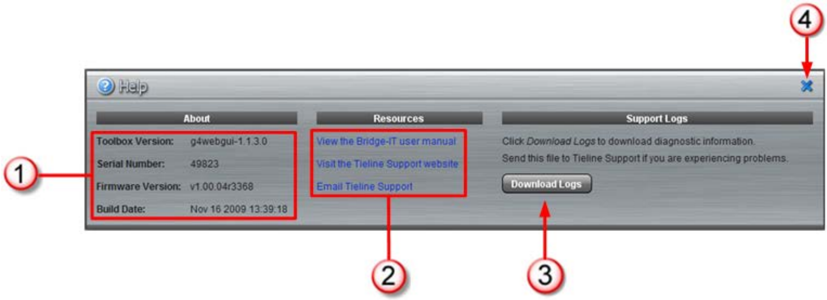
	Rule	Description
1	Connect/Disconnect a program by toggling a relay input	Click to configure Connection and Disconnection by toggling an input
2	Connect when an input is switched **ON**; Disconnect when another input is switched **ON**	Click to configure Connection and Disconnection after different relay 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 connection status	Click to configure a relay to toggle based on connection status

System Panel for Viewing Device Details and Upgrading Firmware



	Feature	Description
1	Network button	Click to view codec network configuration settings
2	Options button	Click to configure Auto Reconnect, RS232 and QoS data settings.
3	SIP button	Click to view SIP configuration settings
4	**Firmware** button	Click to view software versions and perform an upgrade
5	**Licensing** button	Click to select a license file and install it into the codec
6	Backup button	Click to reset codec default settings or save configuration files with codec and program settings
7	IP address details	Device IP address details
8	**Close** button	Click to close the **System panel**
9	Device address details	Device non-IP address details

Help Panel for Product Support



	Feature	Description
1	About	Details of the Toolbox web-GUI and codec firmware versions, as well as the 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	**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 display the new language selected.

14.1 Adjusting Input Settings

Codec input controls are available within the **Inputs panel** of the Toolbox web-GUI.

Configuring Input Channel Settings

Codec inputs are configured for analog high-gain mic level audio sources by default. Click the **Configure** symbol on each input to adjust input controls.

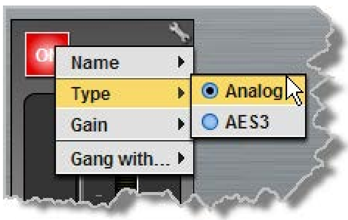
Renaming Input Channels:

1. Click the **Configure** symbol on the input channel you want to rename.

2. Select **Name** and click in the text box before entering a new name.
3. Click **Change Name** to confirm the name change.

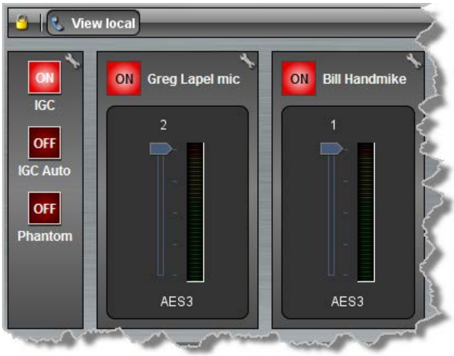
Selecting Analog and Digital Audio Sources:

1. Click the **Configure** symbol on channel 1.
2. Select **Type** and click to select either **Analog** or **AES3**.



3. The display changes to reflect 100% input levels; slider and input on/

off controls are locked on.



Choosing Input Gain Settings:

Input 2 on the codec is a line level input only and cannot be adjusted. Input 1 is also configured for a line level audio source by default and this setting can be adjusted for a mic-level or unbalanced source:

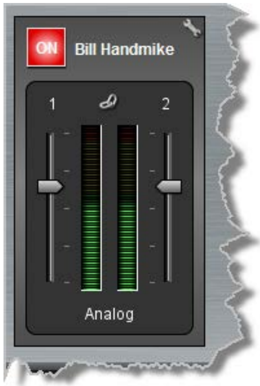
1. Click the **Configure** symbol on channel 1.
2. Select **Gain** and click to select the appropriate setting for your audio source.

Ganging Channels:

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

1. Click the **Configure** symbol on either channel.
2. Select **Gang** and click to either gang or ungang the two 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 adjust channel audio levels. Click the **Link** symbol again to resume ganging.

Setting Audio Levels

Audio levels on the **Input panel** should be set so that 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**. (Click here to find out more about audio input metering).

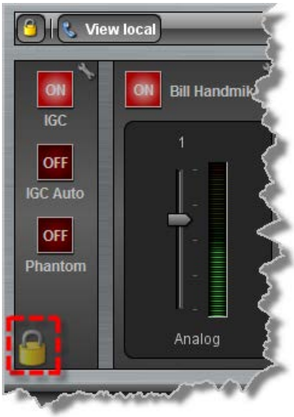
Other Input Controls

There are also input control **ON**/**OFF** buttons for **IGC** and **IGC Auto** on both codec inputs, as well as 15V **Phantom** power for input 1. Click to toggle between the on and off states. (Click here to find out more about IGC and Phantom)

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.

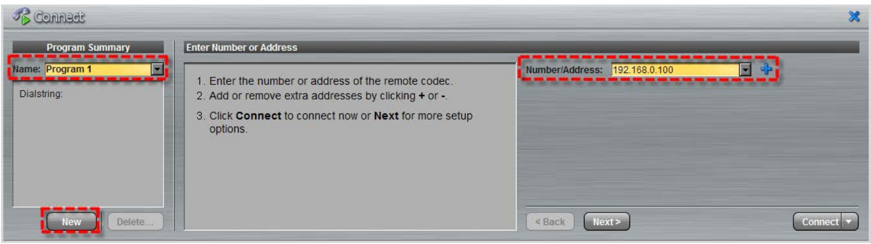


14.2 Configuring Point-to-Point Programs

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

2. Click **New** and then click in the **Name** text box to type a **Program** name.

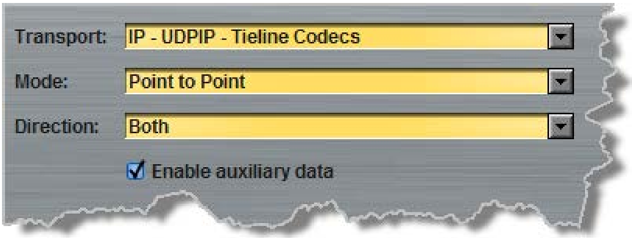
3. Click in the **Number/Address** text box and type the IP address of the codec you want to connect to, then click **Next**.



4. Click the drop-down arrows to adjust the:

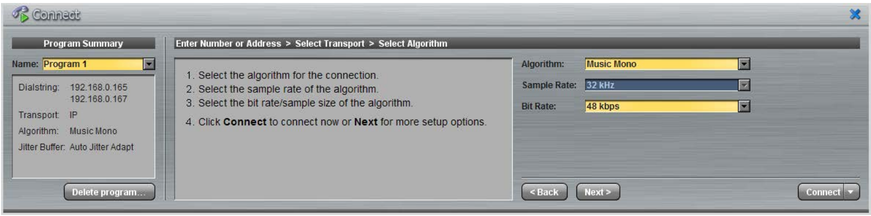
- **Transport** (IP or SIP connection).
- **Mode** of connection to Point-to-Point.
- Encode and decode directions; encode or decode only to save on data usage if bidirectional audio is not required.
- Auxiliary data setting (default is off); click the **Enable auxiliary data** check-box to enable/disable RS232 data and relay operations, then

click **Next**.

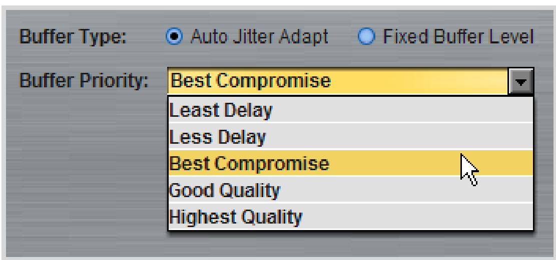


Important Notes on Auxiliary Data if Enabled: See RS232 Data Adjustments for detailed information on configuring RS232 data and see Enabling Relays and RS232 Data for more information on relay operations.

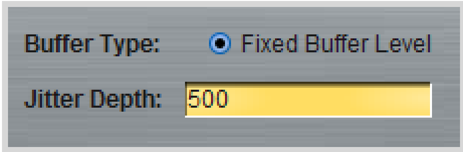
5. Click the drop-down arrows on the right-hand side of each text box to select the **Algorithm**, **Sample Rate** and **Bit-Rate** options.



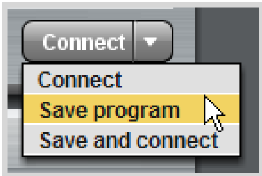
6. Click to select **Auto Jitter Adapt** and the preferred auto setting using the drop-down arrow for **Buffer Priority.**



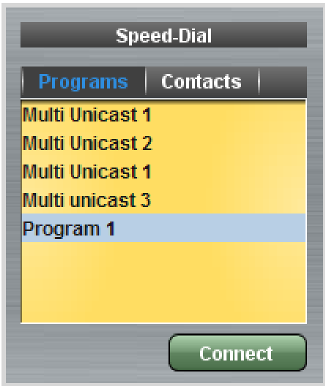
Alternatively, you can select a **Fixed Buffer Level** for the connection. Enter the **Jitter Depth**, which must be between 60ms and 1000ms (except PCM connections which support rates as low as 12ms).



7. Click the drop-down arrow adjacent to the **Connect** button to save the program, or save the program and connect immediately if you prefer.



8. When you save a program it becomes visible in the **Speed Dial** section of the **Master panel** under **Programs**.



9. Highlight the program you want to connect with, then click **Connect** to start connecting.

14.3 Configuring Multi-Unicast Programs

The codec supports up to 6 independent multi-unicast connections from a single codec. Multi-unicast connections can only be created using the ToolBox web-GUI and require software versions that support multi-unicasting.

Prerequisites:

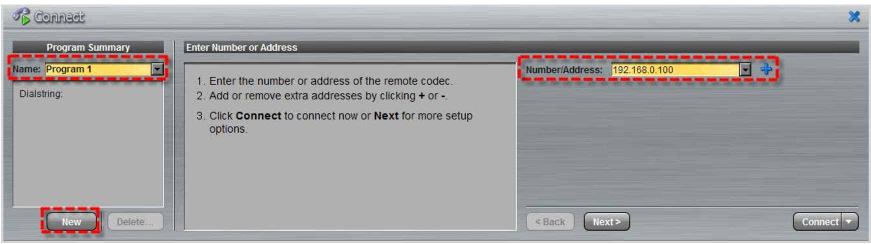
- Bridge-IT firmware v.1.01.00 r4219 or higher.
- G3 codec firmware v.1.6.56 or higher (if connecting to a G3 codec).
- ToolBox web-GUI v.1.2.2.3 or higher.
- A multi-unicast license installed in the dialing codec.

Creating a Multi-Unicast Program

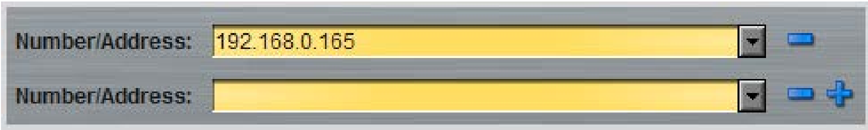
1. Open the web-GUI and click the **Connect**  button at the top of the screen to display the **Connect panel**.

2. Click **New** and then click in the **Name** text box to type a **Program** name.

3. Click in the **Number/Address** text box and type the IP address of the codec you want to connect to.



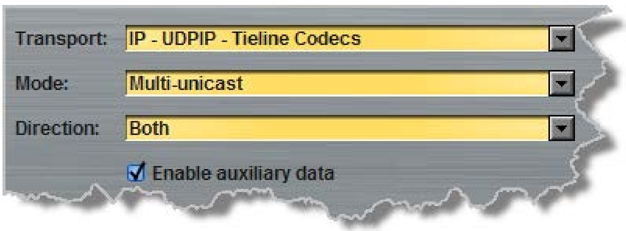
4. Click the blue plus ✚ symbol adjacent to the **Number/Address** text box to add another connection text box. Click the blue minus ▬ symbol to remove a multi-unicast connection text box.



5. Manually enter the IP address for each multi-unicast connection, or click the drop-down arrow on the right-hand side of each text box to select a **Contact** (see Managing Contacts), which has been configured into the codec. When all IP addresses are entered click **Next.**

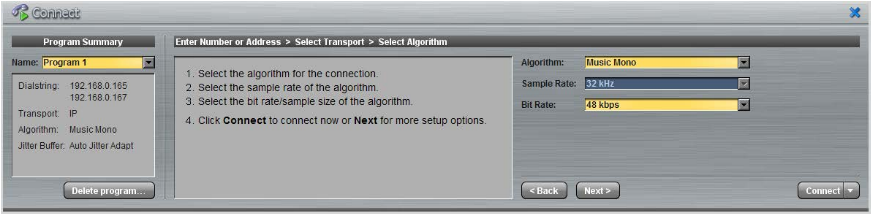
6. Click the drop-down arrow to adjust the:

- **Transport**; select **IP - UDPIP - Tieline Codecs** because SIP is only available with point-to-point connections.
- **Mode** of connection to **Multi-unicast**..
- Encode and decode directions; select decode only to save on data usage if bidirectional audio is not required on the first connection.
- Auxiliary data setting (default is off); click the **Enable auxiliary data** check-box to enable/disable sending RS232 and relay data on all multi-unicast connections (bidirectional data is only available on the first connection dialed when multi-unicasting), then click **Next**.

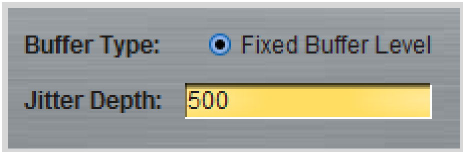


Important Note on Auxiliary Data if Enabled: See RS232 Data Adjustments for detailed information on configuring RS232 data and see Enabling Relays and RS232 Data for more information on relay operations.

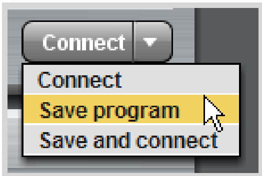
7. Click the drop-down arrows on the right-hand side of each text box to select the **Algorithm**, **Sample Rate** and **Bit-Rate** options.



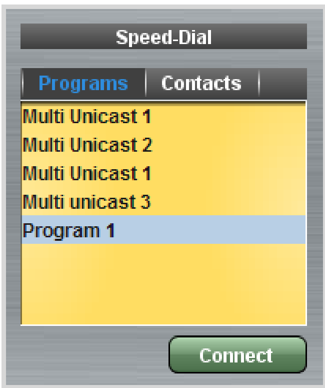
8. Enter the Jitter Buffer depth, which must be a minimum of 500ms for a multi-unicast connection and up to 1000ms.



9. Click the drop-down arrow adjacent to the **Connect** button to save the program, or save the program and connect immediately if you prefer.



10. When you save a program it becomes visible in the **Speed Dial** section of the **Master panel** under **Programs**.



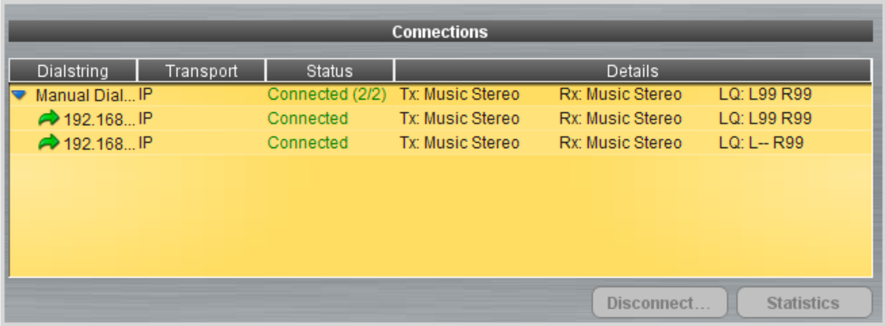
11. Highlight the program you want to connect with, then click **Connect** to start connecting.

Important Notes:

- It is possible to configure up to 6 multi-unicast connections over IP.
- The first connection in the program connection list is capable of sending bidirectional audio; all other connections are unidirectional.
- Available algorithms for multi-unicast connections include AAC, MPEG Layer 2, Tieline Music and Tieline MusicPLUS.
- Jitter buffering is preset at a minimum of 500 ms for multi-unicast connections.
- Renegotiation of connection bit-rates is not possible when connected.
- Forward Error Correction (FEC) is not available for multi-unicast connections.
- You need to ensure you have sufficient connection bandwidth at the local codec to support all the connections to which you are connecting.
- If you are dialing from Bridge-IT to both G3 and Bridge-IT codecs, be aware that 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.

14.4 Monitoring Multiple Unicast Connections

The **Master panel** in the Toolbox web-GUI is used to monitor connection details, as displayed in the following image. Click the blue arrow in the **Dialstring** column to expand and contract the connection details for a program that you dial, or the details of a manually dialed connection (as displayed below). The process is the same for point-to-point connections and multi-unicasts.



14.5 Configuring a Multicast Program

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.

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).

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 setting etc. Multicast connections do not

use session data and 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.

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.

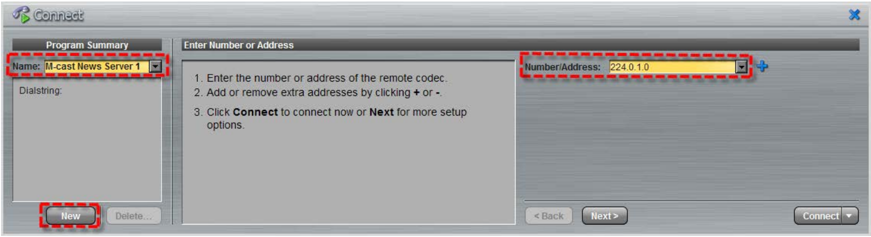
A multicast server codec sends audio packets only and a multicast client codec receives audio packets only. Codecs using the client program request multicast packets from the server codec, which are distributed by the multicast routers on the network.

Creating Multicast Programs

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

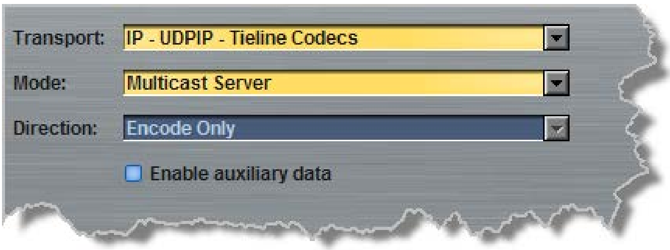
2. Click **New** and then click in the **Name** text box to type a **Program** name.

3. Click in the **Number/Address** text box and type the IP address of the multicast router you want to connect to, then click **Next**.

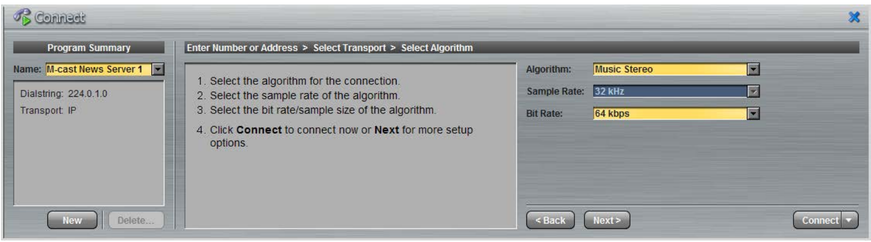


4. Click the drop-down arrows to adjust the:

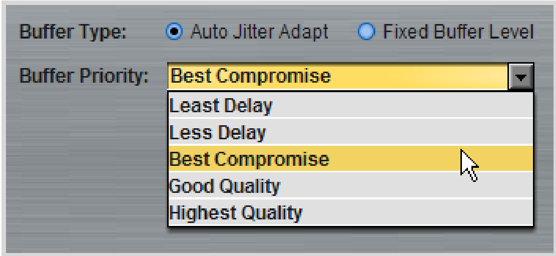
- **Transport** to **IP - UDPIP - Tieline Codecs**
- **Mode** of connection to **Multicast Server** when creating a multicast server program, or **Multicast Client** when creating a multicast client program.
- The encode and decode direction is configured automatically for encode (server) or decode (client) only.
- Auxiliary data is not available for multicast connections.



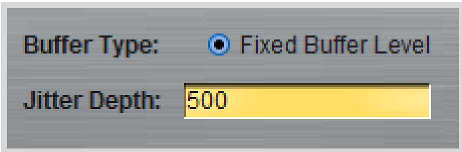
5. Click the drop-down arrows on the right-hand side of each text box to select the **Algorithm**, **Sample Rate** and **Bit-Rate** options.



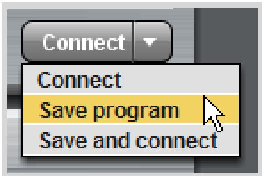
6. For multicast client programs click to select **Auto Jitter Adapt** and the preferred auto setting using the drop-down arrow for **Buffer Priority.**



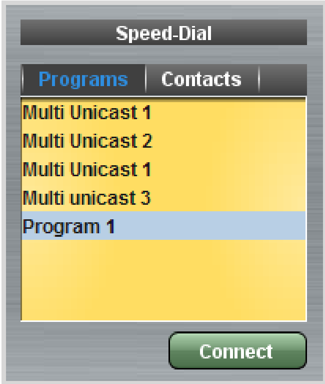
Alternatively, you can select a **Fixed Buffer Level** for the connection. Enter the **Jitter Depth**, which must be between 60ms and 1000ms (except PCM connections which support rates as low as 12ms).



7. Click the drop-down arrow adjacent to the **Connect** button to save the program, or save the program and connect immediately if you prefer.



8. When you save a program it becomes visible in the **Speed Dial** section of the **Master** pane under **Programs**. In the following image the server and client programs are listed.



9. To create a multicast connection, select the multicast server program you want to use on the server codec, then click **Connect**.

10. Next select the multicast client program on each of the multicast client codecs and click **Connect** to begin receiving multicast audio packets.

Important Notes:

- Always dial the multicast server codec connection first before connecting multicast client codecs.
- Multicast client codecs will display Local link quality (LQ) readings only. Multicast server codecs do not display LQ readings.
- The UDP audio port setting is 9000 for multicast connections and cannot be adjusted.
- Forward Error Correction (FEC) is not available for multicast connections.
- It is not possible to send auxiliary data using multicast connections.
- It is not possible to connect to a G3 codec and receive multicast IP audio streams.
- To copy programs onto multiple codecs see Save and Restore Configuration Files.

14.6 Configuring SIP Connections

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).

The default algorithm selected when connecting using SIP is G.711. Algorithms supported by the codec when connecting to compatible SIP devices include G.711, G.722, MPEG Layer 2, AAC-LC, AAC-HE and PCM 16 uncompressed RAW audio.

Important Notes:

- Each codec should be registered to a different SIP server account to avoid connection conflicts.
- SIP dialing is only supported over point-to-point 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.

Dialing Peer-to-Peer IP Connections

SIP can be used to make direct peer-to-peer calls from the codec 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 Bridge-IT codecs are detected automatically and require no special configuration beforehand.

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 unit being dialed.
- Configure each codec with a compatible algorithm and sample rate etc.
- Dial using **SIP** in the **Connect** menu.
- If the remote field codec is configured with 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**, then select **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 **Config** 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 **DIAL** button to make a connection.

Important Note: When dialing between Bridge-IT and a Tieline G3 codec using SIP you need to manually select the G3 audio reference level within the Bridge-IT **Audio** menu. 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 configure SIP account registration details into your Bridge-IT 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.

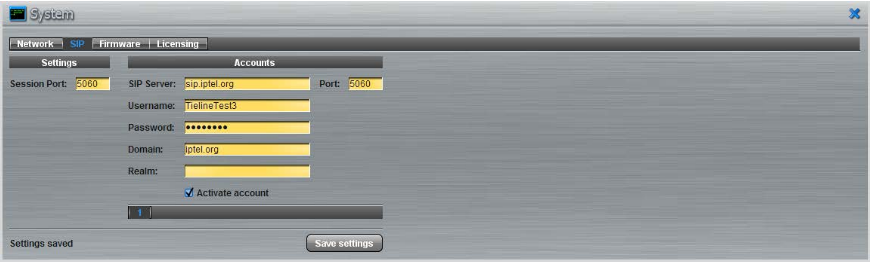
1. Connect your codec to a LAN connection with a public IP address, then login to the Toolbox web-GUI and click the **System** button to open the **System panel**.

2. Click the **SIP** button in the top-left corner of the **System panel**.

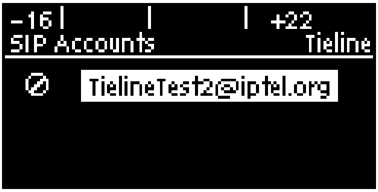
3. Enter the account details into the relevant text boxes.

4. Click to select **Activate Account** and click the **Save Settings**

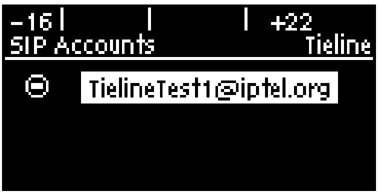
button to create the account in the codec. A confirmation message is displayed in the bottom-left corner of the **Settings panel** if the account details are saved successfully.



5. Enable SIP within the codec via **Config > SIP > Accounts > Select Account name > Active [Enabled].** After selecting **Enabled**, press the **RETURN** button to navigate backwards and make sure that the codec has been registered to the SIP server account by checking the registration symbol appears as it does in the following example.



6. If the codec symbol looks like the following example it has not been registered to the SIP server account correctly.



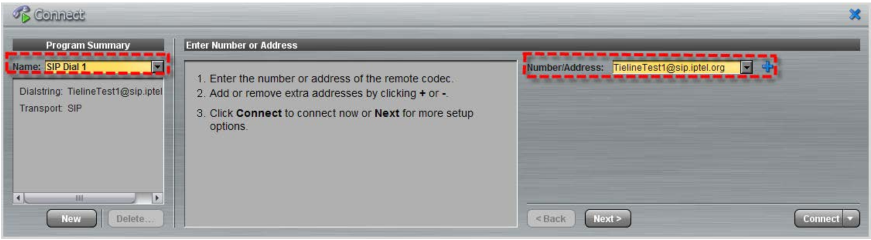
Important Notes: Some 3G service providers may block SIP traffic over UDP port 5060. It is possible to reconfigure this but we recommend that you contact our support desk at support@tieline.com before attempting this.

Dialing SIP Connections

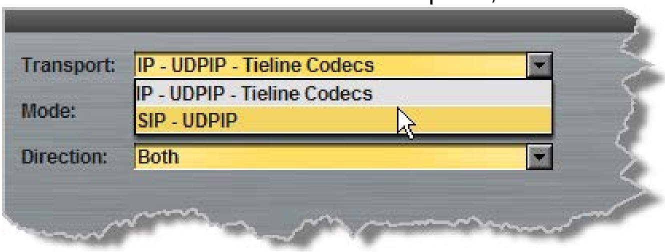
SIP connections are as simple to configure as a normal dialing program when using the Toolbox web-GUI. All settings are configured in the same way with two small differences; entering a SIP address and selecting SIP as the connection transport.

1. Click in the **Name** text box and type a **Program** name.

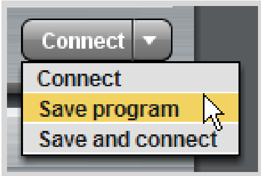
2. Click in the **Number/Address** text box and type the SIP address of the codec you want to connect to, then click **Next**. Note: it is not necessary to type "SIP:" before the address.



3. Click the drop-down arrows for **Transport** and select SIP and adjust the encode and decode directions if required, then click **Next**.



4. Configure the rest of the connection settings and click the drop-down arrow adjacent to the **Connect** button to save the program, or save the program and connect immediately if preferred.



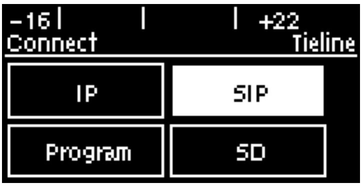
See [Configuring Point-to-Point Programs]() for more details on configuring programs.

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 to reregister it to your SIP server account. Otherwise the codec will not be able to reconnect.

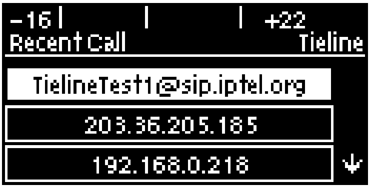
Important Notes: The default UDP audio port when using SIP is 5004 in Tieline codecs. To contact a codec that is behind a firewall or NAT-enabled router it is essential that all relevant ports are open and forwarded to the other device.

Dialing a SIP Address via the Codec Front Panel

1. Press the **HOME** button to return to the **Home** screen, select **Connect,** then select **SIP** and press the OK button.



2. Use the **KEYPAD** to enter the alphanumeric SIP address of the codec you want to dial, using 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 **DIAL** button to view the **Recent Call** screen and select the SIP address you want.



3. Press the down navigation button to select **Config** and press , then 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 **DIAL** button to make a connection.

14.7 Save & Restore Configuration Files

The web-GUI can be used to save configuration files containing:

- Profiles programmed to connect using different connection settings.
- All the settings that a user has adjusted to change the factory default settings (current runtime settings).

Programs are essentially connection profiles that may include:

- A Program Name.
- IP address dialing details for up to 6 connection end-points .
- Specific connection profile details pertaining to algorithm, FEC, jitter buffer and bit-rate settings etc.
- Support for up to six multi-unicast connections, one of which can be bidirectional.

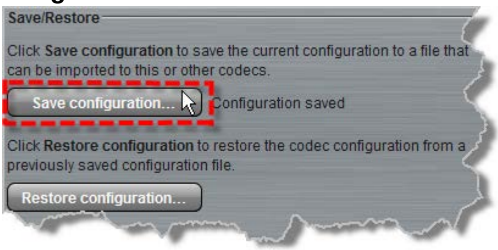
Saving Configuration Files

1. Open the web-GUI and click the **System** button at the top of the

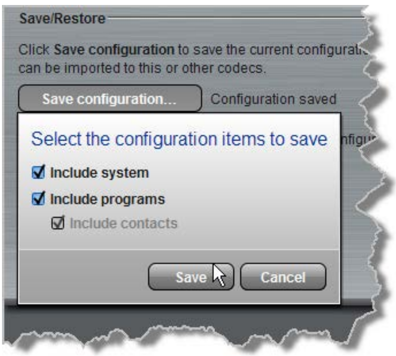
screen to display the **System panel**.

2. Click the **Backup** button at the top of the **System panel**.

3. Click **Save Configuration**.



4. Use your mouse-pointer to click and select the check boxes representing your configuration requirements.



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

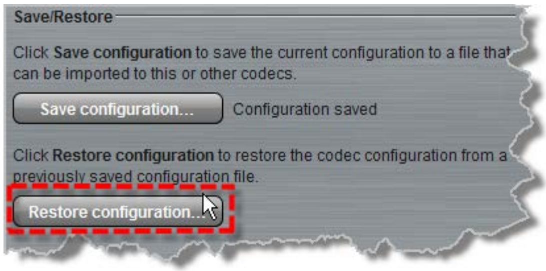
Restoring Configuration File Settings

1. Open the web-GUI and click the **System**  button at the top of the

screen to display the **System panel**.

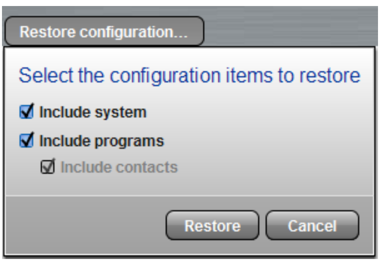
2. Click the **Backup** button at the top of the **System panel**.

3. Click **Restore Configuration**.



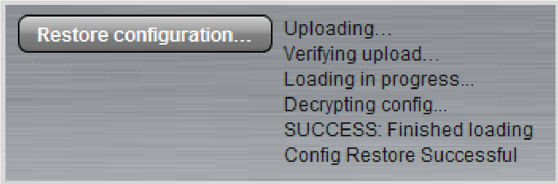
4. Navigate to the configuration file on your PC that you want to load, then click **Open**.

5. Use your mouse-pointer to click and select the check boxes representing your system restore requirements. 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 restore the configuration file settings;

confirmation of successful restoration is provided by the web-GUI.



14.8 Reset Default Settings

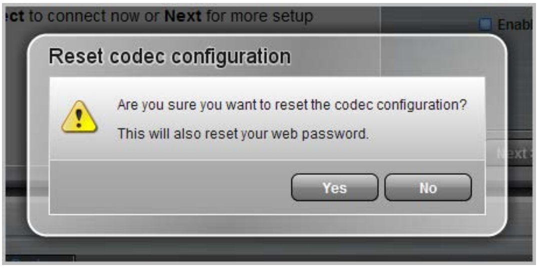
There are several settings which allow you to restore factory default settings within the codec.

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

2. Click the **Backup** button at the top of the **System panel**.

3. Click one of the four reset options available.

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



14.9 RS232 Data Adjustments

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 **Connect panel** wizard (see Configuring Point-to-Point Programs or Configuring Multi-Unicast Programs). Alternatively use the codec **Config** menu (see Enabling RS232 Data).

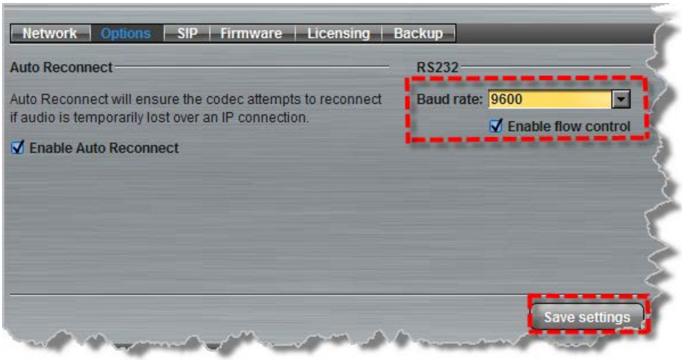
Setting RS232 Data Rates and Flow Control

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

2. Click the **Options** button at the top of the **System panel**.

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:

- The codec cannot send RS232 data or activate relays on IP-enabled Tieline G3 codecs.
- 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.

14.10 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. Use the down navigation button to select **Config** and press the button.
5. Navigate to **Data** and press to toggle between **Enabled** and **Disabled**

For more information please see Enabling Relays & RS232 Data.

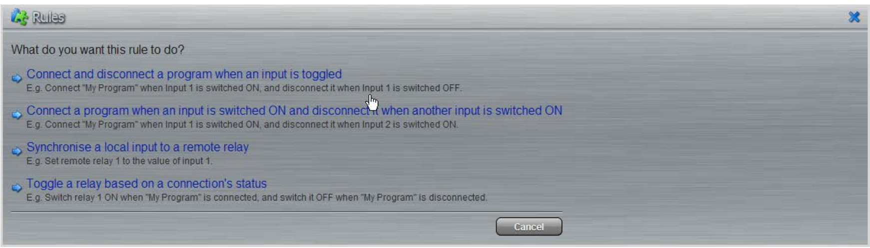
Configuration Rules

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.

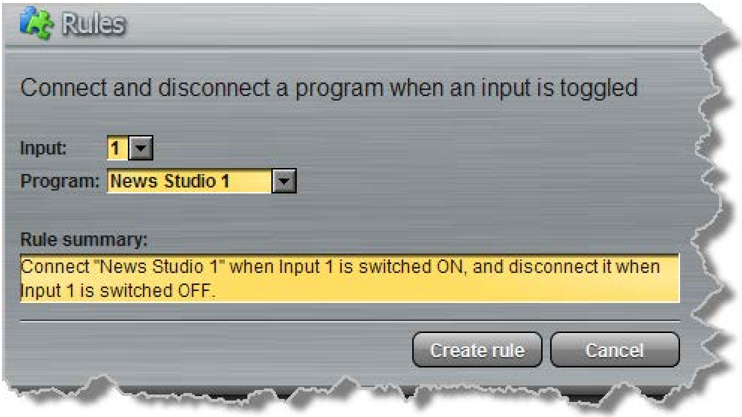


Rule 1: Toggle a Control Port Input to Connect and Disconnect a Program

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.

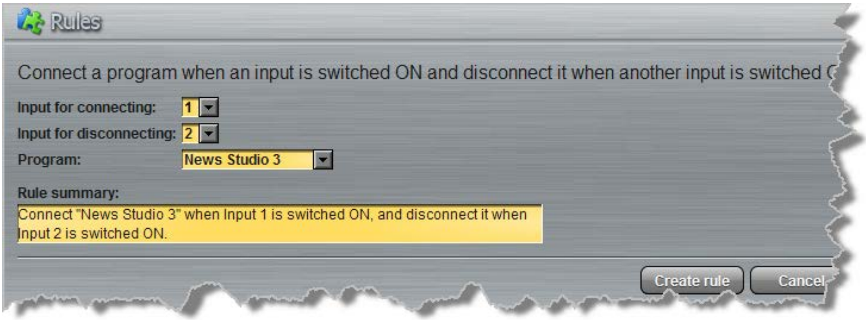
Rule 2: Switch Different Control Port Inputs On to Connect and Disconnect a Program

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.

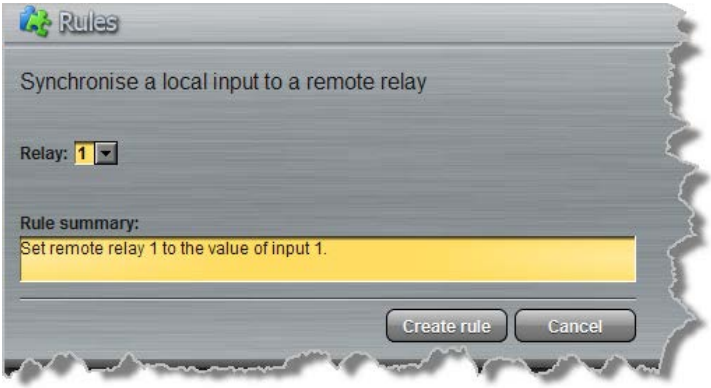
4. Check the **Rule summary** and click **Create Rule** to save the settings.

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.



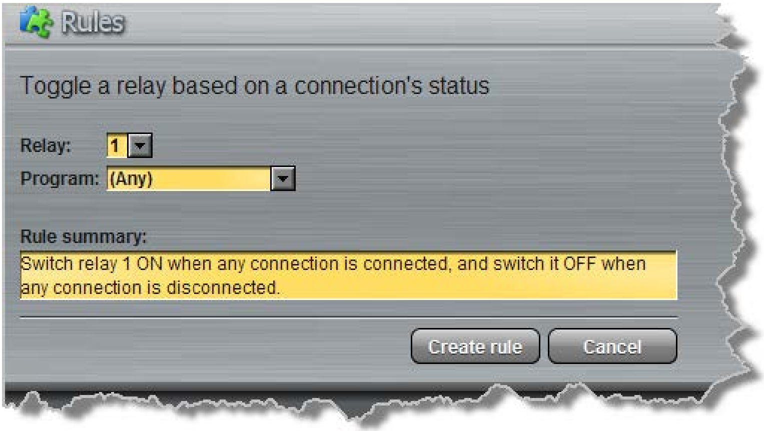
3. Check the **Rule summary** and click **Create Rule** to save the settings.

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.

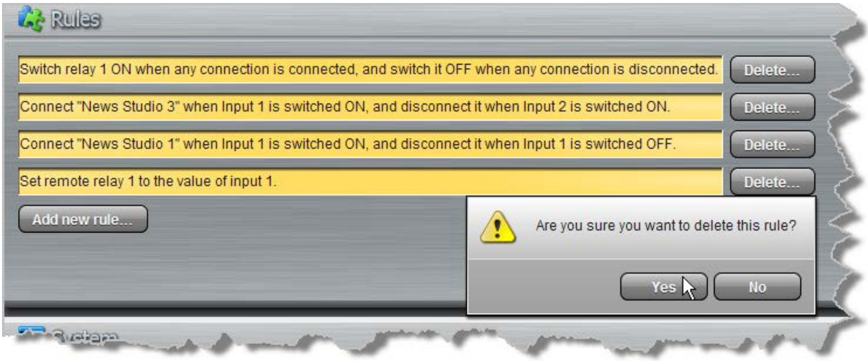
4. Check the **Rule summary** and click **Create Rule** to save the settings.

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.



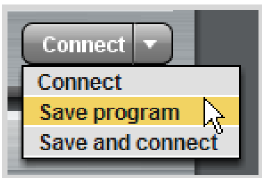
14.11 Editing Programs

1. Open the Toolbox web-GUI and click the **Connect**  button at the top of the screen to display the **Connect panel**.

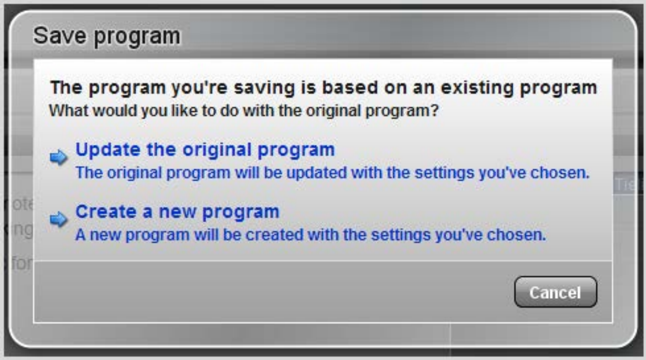
2. Click the drop-down arrow adjacent to the **Name** text box and select the program you want to edit.

3. Click **Next** to navigate through the connection wizard and adjust any settings as required.

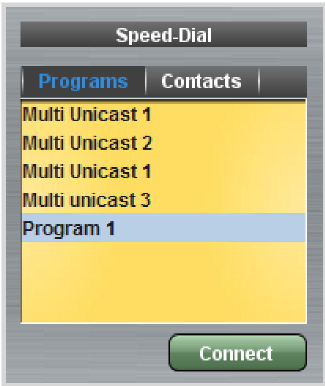
4. Click the drop-down arrow adjacent to the **Connect** button to save the edited program, or save the program and connect immediately if you prefer.



5. In the **Save Program** dialog click to confirm that you either want to overwrite the original program settings, or create a new program.



6. The edited program it becomes visible in the **Speed Dial** section of the **Master panel** under **Programs**.



7. Highlight the program you want to connect with, then click **Connect** to start connecting.

15 Routine Programming Tasks

The codec supports DHCP and static IP addressing for dialing over IP networks and also includes SmartStream technology for automatically managing connections over IP networks. This includes advanced Forward Error Correction (FEC) techniques designed to increase the stability of IP connections in the event that data packets are lost, as well as sophisticated automatic jitter buffering that simplifies management of network congestion over a range of IP network infrastructure.

15.1 Configuring IP Addresses

An IP address is a unique number that allows devices to communicate over IP networks and the internet using the internet Protocol standard and there are two types of IP addresses – public and private.

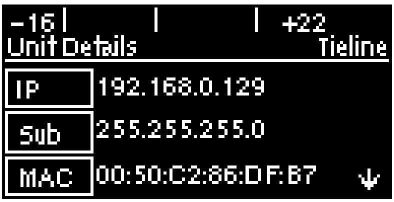
If you want to dial a codec with a public IP address you simply dial the IP address to connect. If you want to dial a codec with a private IP address you need to perform network address translation (NAT). NAT allows a single device, such as a broadband router, to act as an agent between the public internet 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.

The codec supports both DHCP (default) IP addressing and Static IP addresses for dialing IP connections.

(For more information on IP addresses and IP connections generally, please download the Tieline 3GIP Streaming Reference Manual from http://www.tieline.com/Transport-Support/IP-Support.)

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 **Config** and press the button.
3. Select **Unit** and press the button.
4. IP address details and other **Unit Details** are listed. Use the arrow up and down buttons to scroll and view all details listed.



Configuring a DHCP IP Address

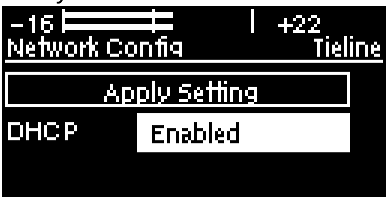
DHCP IP addresses are automatically assigned and can change each time you connect to your Internet Service Provider or to your own local area network (LAN). By default the codec is configured for DHCP-assigned IP addresses. To configure DHCP IP address settings:

1. Press the **Home** button to return to the home screen.
2. Use the navigation buttons on the front panel to select **Config** and press the button.
3. Select **LAN** and press the button.
4. Use the down navigation button to select **Disabled** and press the button to toggle to **Enabled**.
5. Use the up navigation button to scroll to the top of the menu and select **Apply Setting**, then press the button to configure DHCP IP addressing.

Configuring a Static IP Address

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

1. Press the **Home** button to return to the home screen.
2. Use the navigation buttons on the front panel to select **Config** and press the button.
3. Use the down navigation button to select **LAN** and press the button.
4. **DHCP** is enabled by default.



5. Use the down navigation button to select **Enabled** and press the button to toggle enabling and disabling of DHCP.
6. When **DHCP** is disabled, the **Static IP** address menu is revealed.

7. Use the up and down navigation buttons to select **Static IP**, press the button, and use the numeric **KEYPAD** to enter the IP address. Press the button to save changes. Note: use the or buttons to enter the periods in the IP address and use the **RETURN** button to delete numbers already entered.
8. Enter changes to the **Subnet** (Subnet Mask) or **Gateway** (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 up navigation button to scroll to the top of the menu and select **Apply Setting**, then press the button to save all changes.
10. Check the **Unit Details** menu to ensure the new static IP address has been entered correctly.

15.2 Creating Programs

Point-to-point programs can be created using the codec front panel keypad. When creating custom programs without the Toolbox web-GUI it is important to ensure that you configure all the connection settings first as these are stored as part of the configured profile. Once a program has been created it cannot be adjusted without using the **Connect panel** within the Toolbox web-GUI.

1. Press the **HOME** button to return to the **Home** screen.
2. Select **Connect,** then select **IP** and press the button.
3. 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.
4. Press the down navigation button to select **Config** and press to adjust or check any any connection parameters (see 10 Quick Steps to Connect Bridge-IT for more info).

5. Press the **RETURN** button to navigate backwards to the **Connect IP** screen that the IP address was entered into.
6. At this point navigate to **Save** on the **Connect IP** screen and press to save the settings as a custom program for subsequent recall and dialing. Give the program a name and press to save the program.
7. The program will now appear in the list of custom programs displayed via the **Programs** menu, which is accessed via the home screen.

15.3 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.

It comes standard with MPEG Layer 2, G.711 and G.722 algorithms, as well as Tieline Music and MusicPLUS. These two Tieline low-delay algorithms are optimized for wired and wireless IP connections.

Optional aptX® Enhanced, LC-AAC, HE-AAC v.1 and HE-AAC v.2 algorithms can also be purchased.

The codec 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 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

HE-AAC v.1 and HE-AAC v.2 are optimised for low bit-rate connections. 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.

Selection of HE-AAC v.1 and v.2 is automatically managed within the codec and only **AAC-HE** is displayed on the screen.

HE-AAC v.1 is used for mono connections and performs best at bit rates of 24Kbps per channel or higher. HE-AAC v.1 is 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.

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.

Configuring an Algorithm 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. Use the down navigation button to select **Config** and press the button.
5. Navigate to **Alg** and press .
6. Select the mono or stereo algorithm that you want to connect with and press .

How do I choose the 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.

Algor-ithm	Audio Band-width	Algor-ithmic Delay	IP bit-rate per channel	IP over-head	Recommended connection for on-air use
Linear (Uncom-pressed)	16/24 bit up to 24kHz	0ms	sample rate x bits per sample x no. channels	80Kbps	Extremely high quality uncompressed audio distribution and STLs
Tieline Music	Up to 15kHz	20ms	24 Kbps minimum	16Kbps	High quality low bit-rate remotes, STLs and audio distribution
Tieline Music-PLUS	Up to 22kHz	20ms	48 Kbps minimum	16Kbps	Very high quality low bit-rate remotes, STLs and audio distribution
G.711	3kHz	1ms	64Kbps minimum	80Kbps	Voice quality connections to other brands of audio codec
G.722	7kHz	1ms	64Kbps minimum	80Kbps	Voice quality connections to other brands of audio codec
MPEG Layer 2	Up to 22kHz	24 to 36ms	64Kbps minimum	8.5 - 13.3Kbps	Very high quality audio connections between Tieline or other brands of codec.

MPEG Layer 3	Up to 15kHz	100ms	64Kbps		High quality low bit-rate remotes, STLs and audio distribution
LC-AAC	Up to 15kHz	64ms	64Kbps	15Kbps	High quality low bit-rate remotes, STLs and audio distribution
HE-AAC v.1	Up to 15kHz	128ms	48Kbps	7.4Kbps	High quality low bit-rate remotes, STLs and audio distribution
HE-AAC v.2	Up to 15kHz	128ms	Minimum 16Kbps (Mono); 24Kbps (stereo)	7.4Kbps	DAB+ radio streaming and high quality low bit-rate remotes, STLs and audio distribution
aptX® Enhanced	10Hz-24kHz	2.5ms at 48kHz	128Kbps minimum (16bit; 32kHz) to 288Kbps (24bit;48kHz)	80Kbps	Very high quality STLs and audio distribution

Sampling Rates

When selecting linear 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 run at 48kHz sampling. G.711 and G.722 will always run at a 32kHz sampling rate (downsampled to 8kHz and 16kHz respectively).

15.4 Configuring the Jitter Buffer

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. Tieline's Jitter-buffer is smart because of its ability to:

- 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 configure 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 minimise delay.

A fixed jitter buffer is preferable when using 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.

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** (if an SD card is in the codec) and press the button
4. Use the down ▽ navigation button to select **Config** and press the button.
5. Navigate to **Jitt** and press .
6. Select **Auto Jitter** and press .
7. 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 pressing the Enter/Dial button while connected. 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. Use the down navigation button to select **Config** and press the button.
4. Navigate to **Jitt** and press .
5. Select **Fixed Buffer** and press .
6. Use the numeric **KEYPAD** to enter the fixed buffer value in milliseconds and press .

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 the use of either Tieline Music or Voice G3 algorithms. Do not use Raw (uncompressed) audio over highly contended DSL/ADSL connections without enough bandwidth to support the high connection bit-rates required.

15.5 Configuring Forward Error Correction

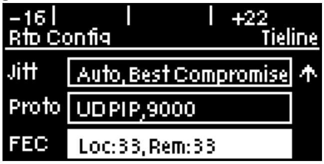
Forward Error Correction (FEC) is designed to increase the stability of UDP/IP connections in the event that data packets are lost. FEC works by sending a secondary stream of audio packets over a connection so that if your primary audio stream packets are lost or corrupted, then packets from the secondary 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 **L: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 OK button
4. Use the down ▽ navigation button to select **Config** and press the OK button.
5. Navigate to **FEC** and press OK.
6. Select the local codec FEC setting in the **Local FEC** screen and press OK.
7. Select the remote codec FEC setting in the **Remote FEC** screen and press OK.
8. 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 dropouts 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.

15.6 Configuring Encode/Decode Direction

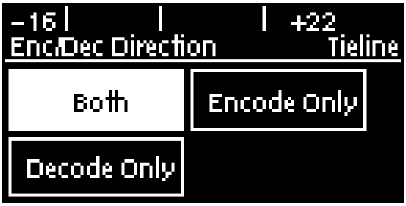
By default the codec is configured to encode and decode data. However, it is possible to encode or decode audio data only. This is useful for:

- 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. Use the down navigation button to select **Config** and press .
5. Navigate to **Dir** and press .
6. Select the encode or decode direction setting you want and press .



15.7 Enabling Relays & RS232 Data

Data sending capability is disabled in the codec 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 the **IP mode** you are using to connect, e.g. **Point-to-Point**, then press the button.
5. Use the down navigation button to select **Config** and press the button.
6. Navigate to **Data** and press to toggle between **Enabled** and **Disabled** (Note: default setting is **Disabled**).

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

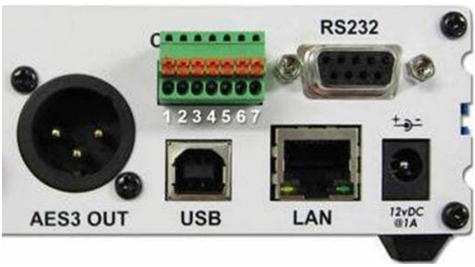
CMOS field effect transistors switch a low impedance path between the two pins when activated. These are opto-isolated and floating above ground. It is important to current-limit the source as damage will result where the current exceeds 100mA peak-to-peak. No more than 48 volts peak-to-peak should be used as a safety precaution. The resistance of the CMOS element is approximately 25 ohms in the ON state.

Relay Operation

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

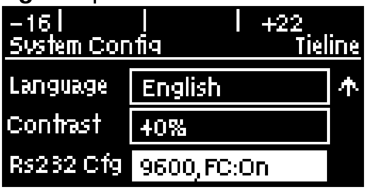


Important Note: For more information about how to configure relay operations with a PC using the Toolbox web-GUI, please see Creating Rules.

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 **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Config** and press the button.
3. Navigate to **System** and press .
4. Select **RS232 Cfg** and press .



5. Use the navigation buttons to select the correct baud rate.
6. Select **Enable** for flow control and press to save all settings.

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:

- The codec cannot send RS232 data or activate relays on Tieline G3 codecs.
- 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.

15.8 Configuring TCP/UDP Protocols

In TCP and UDP networks the codec port is the endpoint of your connection. Software network ports are doorways for systems to communicate with each other. For example, several codecs in your studio may use the same public static IP address. Unique port numbers can be used to route audio to each codec.

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, for both TCP and UDP connections, are outlined in the following table.

IP Connection	Session Data Port	Audio Port
IP1 connection	TCP 9002 (to send session data)	UDP 9000 (to send audio data)
Toolbox Web-GUI	TCP 80	
SIP	UDP 5060	UDP 5004

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.

Changing Codec Port Numbers

There are a several reasons why you may wish to adjust the port setting on your codec, including:

- Having to create a path through gateways and firewalls.
- Another IP device is already using a codec's port number.
- More than one studio codec is in use and each codec requires a different port number.

For a studio and remote codec to connect, they need to be configured with matching port numbers. If there is a need to change codec port settings, in most situations you should consult your organization's resident IT professional. To adjust these 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** (if an SD card is in the codec) and press the button
4. Use the down navigation button to select **Config** and press the button.
5. Navigate to either **Sess** (session protocol) or **Proto** (audio protocol) and press .
6. Select the session or audio protocol you want and and press .
7. Use the numeric **KEYPAD** add a new port number and press .

15.9 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

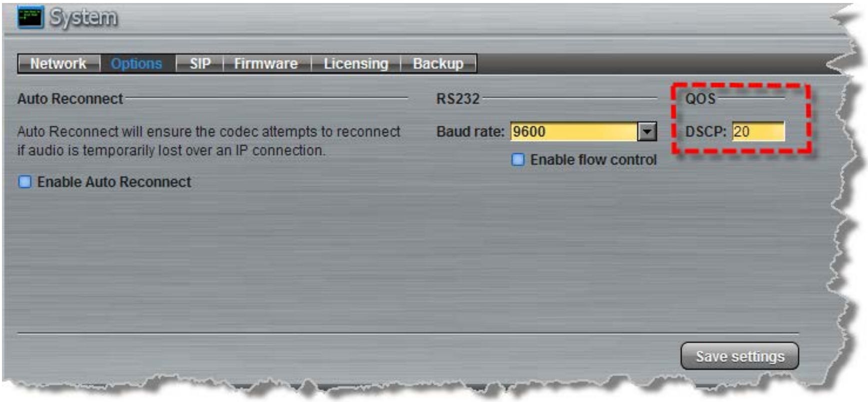
1. Press the **HOME** button to return to the **Home** screen.
2. Use the up and down navigation buttons to select **Config** and press the button.
3. Use the navigation buttons to select **QoS** and press the button.
4. Press the button and use the **RETURN** button to delete numbers already entered, then use the numeric **KEYPAD** to enter the new setting.
5. Press the button to save the new setting.

Configuring QoS Using the Web-GUI

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

2. Click the **Options** button at the top of the **System panel**.

3. Click in the **DSCP** field and enter the priority setting recommended by your IT administrator.



4. Click **Save settings**.

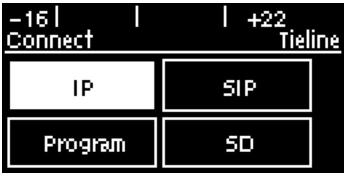
15.10 Configuring Time-to-Live

Time-to-Live (TTL) is a value you can program into the codec 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.

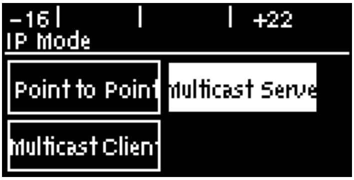
The time-to-Live setting is configurable and sets the maximum number of router hops allowable for multicast data packets. In most situations the default value of **1** is used, to ensure packets are sent through a single LAN router and not over multiple router hops and networks.

This value is only displayed in the codec when a codec is in **Multicast Server** mode. To adjust the setting:

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

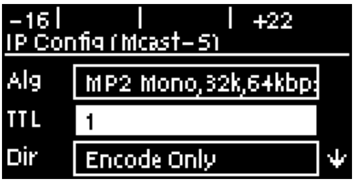


2. Press the right navigation button to select **Multicast Server** and press



3. Press the down navigation button to select **Config** 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.

15.11 Reset and Restore Factory Default Settings

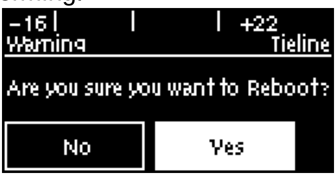
The codec offers several reset and reboot functions from within the **Reset Functions** menu. These options include:

	Function	Description
1	Reset Codec Config	This clears all codec settings but retains key settings that include LAN, language, LCD contrast and contact settings
2	Restore Factory Defaults	This clears all codec settings back to their factory defaults
3	Clear Programs & Recent Calls	This clears the list of programs and recent calls in the codec, buddy list contacts are retained
4	Reboot Codec	Reboots the codec

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 **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Config** and press the button.
3. Navigate to **Reset** and press the button.
4. Navigate to the preferred option from those available and press the button.

5. 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.

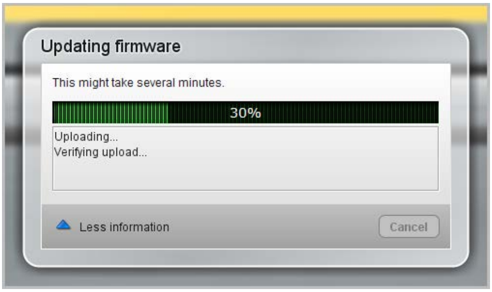
15.12 Upgrading Codec Firmware

Automatic Firmware Upgrades

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

1. Connect your codec to your PC using either a LAN or USB connection and open the web-GUI program (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.

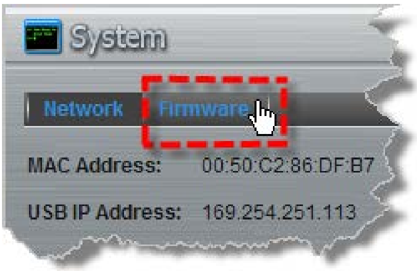
4. Click **More Information** in the **Updating firmware** dialog to display details of the upgrade process.



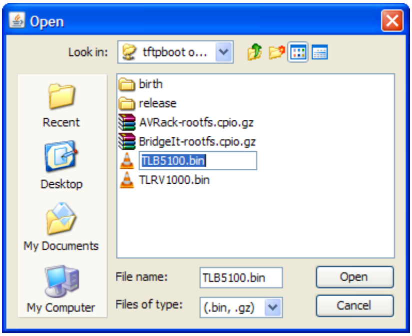
Manual Firmware Upgrades

It is possible to configure the web-GUI to allow codec firmware upgrades by selecting a file on a PC.

1. Click the **System**  button at the top of the web-GUI screen if the **System panel** is not displayed.
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.

15.13 Installing Software Licences

Bridge-IT XTRA codecs include all software and algorithm feature options when purchased and do not require a license upgrade. Bridge-IT Basic and Bridge-IT Pro codecs offer software feature upgrades managed by licenses

Optional features in Bridge-IT codecs include:

Bridge-IT Software Feature	Basic	Pro	XTRA
Multiple-unicast stereo to 6 endpoints; first connection is bidirectional* (license includes Multicast Server capability)		✓	✓
LC-AAC, HE-AAC v.1 and HE-AAC v.2 algorithms *		✓	✓
16 bit and 24 bit aptX® Enhanced algorithm *			✓

* Option available for purchase separately if required.

Contact Tieline at sales@tieline.com, or your favorite dealer, if you need to purchase a software license upgrade. When a Bridge-IT Basic or Bridge-IT Pro software license has been purchased there are two ways to perform an upgrade:

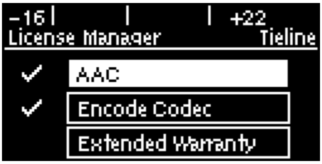
1. Automatically download the software license from TieServer and perform the upgrade.

2. Download the new software license file onto your PC and upgrade using the web-GUI.

Checking Installed Licenses

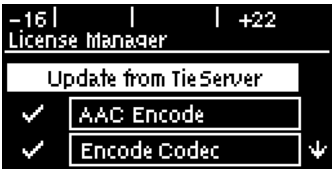
The codec **License Manager** is used to view which licenses are installed in each codec. To view the licenses installed in your codec:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Config** and press the button.
3. Use the navigation buttons to select **Licenses** and press the button.
4. 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.



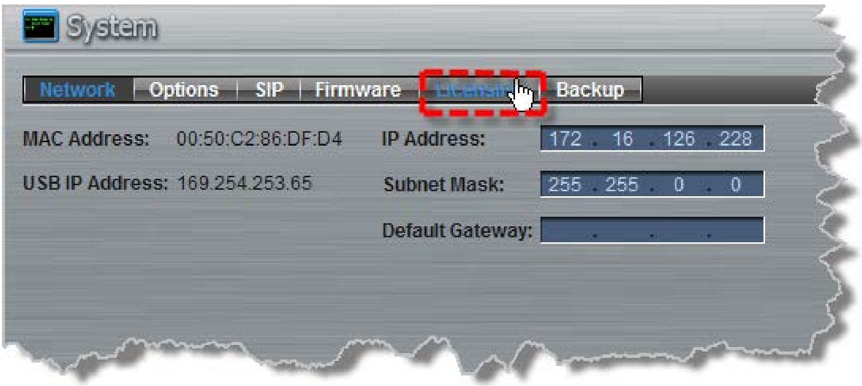
2. The codec will contact TieServer and automatically install all valid licenses.
3. The screen will indicate the update is in progress and then confirm it has been completed successfully.
4. Press the **RETURN** button a few times until you return to the **Home** screen.

5. Use the navigation buttons to select **Config** and press the OK button.
6. Navigate to **Reset** and press the OK button.
7. Navigate to **Reboot Codec** and press the OK button.
8. Select **Yes** and press the OK button to reboot the codec.

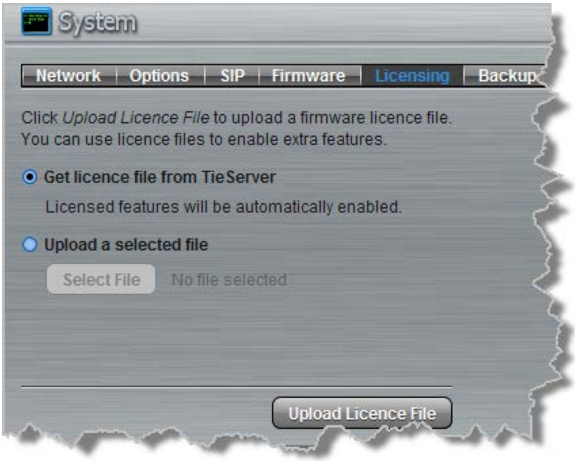
Perform an Automatic Software License Install using the web-GUI

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.

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 **System** button at the top of the web-GUI screen if the **System panel** is not displayed.
3. Click **Licensing** in the **System panel**.



4. Select the **Get license file from TieServer** button.



5. Click **Upload License File**.
6. After the upgrade is completed click **Finish** and the codec screen should display a confirmation message within a short period of time.

7. Follow the codec reset procedure to reboot the codec. 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 connect your codec to a PC and save the license file on this PC.

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 **System** button at the top of the web-GUI screen if the **System panel** is not displayed.
3. Click **Licensing** in the **System panel**.
4. Click the **Select File** button to open a dialog and navigate to the

".lcf" license file on your PC, then click the **Open** button.

8. Click the **Upload License File** button to upload the license file into the codec.
9. Click the **Finish** button.
10. Follow the codec reset procedure to reboot the codec. Note: do not reboot by removing the power cable from the codec.

16 Reference

The following sections contain reference and troubleshooting information.

16.1 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 L99R99. 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.	

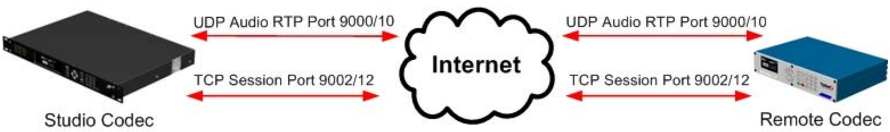
7	The ISP connection is not shared with other PCs or devices.	
8	UDP is being used as the audio transport protocol.	
9	No more than 80% of ISP connection bandwidth is being used.	
10	There are no wireless connections being used.	

16.2 Configuring Connection Protocols

Tieline Codec Session Data

When a connection is initiated between two Tieline codecs, by default Tieline session data is sent by the dialing codec to program the remote codec with compatible connection settings. This simplifies the operation of Tieline codecs and can eliminate the need to adjust them each time they are connected.

If you are connecting two Tieline codecs you would normally use the default **IP** setting in the **Connect** menu. The ports used for sending session data and audio using Tieline session data are displayed in the following image.



16.3 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.

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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>.

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This package is an SSL implementation written by Eric Young (eay@cryptsoft.com).

The implementation was written so as to conform with Netscapes SSL.

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16.4 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 B 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 B prescrites dans le Règlement sur le brouillage radioelectrique du ministère des Communications du Canada.

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17 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	Tieline Music, Tieline MusicPLUS, G.711, G.722, MPEG Layer 2
IP (Pro version only)	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

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