# Bridge-IT
# IP Codec User Manual


**Software Version: 2.18.xx**
**Manual Version: v.4.0_20190220**
**February, 2019**

# Table of Contents

# 1 How to Use the Documentation

## Overview of this User Manual

Use this manual to learn how to:

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

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

## Manual Conventions

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

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

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

## Typographic Conventions

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

## Help Button

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

# 2 Warnings and Safety Information

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

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

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

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

## Warranty and Disclaimer

This equipment manufactured by Tieline is warranted by Tieline against defects in material and workmanship for two years from the date of original purchase. During the warranty period, we will repair or, at our option, replace at no charge a product that proves to be defective, provided you obtain return authorization from Tieline and return the product, shipping prepaid, to Tieline. For return authorization, contact Tieline's US or Australian office (see http://www.tieline.com/Contact-Us).

This Warranty does not apply if the product has been damaged by accident or misuse or as the result of service or modification performed by anyone other than Tieline. With the exception of the warranties set forth above, Tieline makes no other warranties, expressed or implied or statutory, including but not limited to warranties of merchantability and fitness for a particular purpose, which are hereby expressly disclaimed. Use of this product is subject to Tieline's SOFTWARE LICENSE and WARRANTY conditions, which should be viewed at http://www.tieline.com/Support/Product-Warranty before using this product.

In no event will Tieline, its directors, officers, employees, agents, owners, consultants or advisers (its “Affiliates”), or authorized dealers or their respective Affiliates, be liable for incidental or consequential damages, or for loss, damage, or expense directly or indirectly arising from the use of any Product or the inability to use any Product either separately or in combination with other equipment or materials, or from any other cause.

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. Tieline takes no responsibility for any damage to equipment attached to the codec.

# 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 |
| **CSRF** | Cross-Site Request Forgery (CSRF) is an attack that forces a user to execute unwanted actions on a web application in which they are currently authenticated. |
| **DN** | Directory Number for ISDN |
| **DNS** | The Domain Name System (DNS) is used to assign domain names to IP addresses over the World-Wide Web |
| **Domain** | A group of computers or devices on a network which are administered with common rules and procedures. Devices sharing a common part of the IP address are said to be in the same domain |
| **DSCP** | The Differentiated Services Code Point is a field in an IP packet header for prioritizing data when traversing IP networks |
| **Failover** | Method of switching to an alternative backup audio stream if the primary connection is lost. |
| **Fuse-IP** | Tieline bonding of IP interfaces to aggregate data |
| **GUI** | Graphical User Interface |
| **ISP** | Internet Service Providers (ISPs) are companies that offer customers access to the internet |
| **IP** | Internet Protocol; used for sending data across packet-switched networks |
| **LAN** | Local Area Network; a group of computers and associated devices sharing a common communications link |
| **Latency** | Delay associated with IP networks and caused by algorithmic, transport and buffering delays |
| **LIO** | Logic Input/Output |
| **MIB** | A management information base (MIB) is a database used for managing the entities in a communications network. This term is associated with the Simple Network Management Protocol (SNMP). |
| **Multicast** | Efficient one to many streaming of IP audio using multicast IP addressing |
| **Multi-unicast** | A multi-unicast program (also known as multiple unicast) can transmit a single audio stream with common connection settings to a number of different destinations. |
| **MSN** | Multiple Subscriber Number for ISDN |
| **NAT** | Network Address Translation is 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. |
| **PAT** | Port Address Translation is 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 |
| **PSU** | Power Supply Unit (Bridge-IT XTRA only) |
| **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 |
| **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 is a common protocol which works with a myriad of other protocols to establish connections with other devices to provide interoperability |

| | |
|---|---|
| **SLA** | Service Level Agreements (SLAs) a contractual agreement between an ISP and a customer defining expected performance levels over a network |
| **SmartStream PLUS** | Tieline implementation of dual redundant IP streaming. |
| **SNMP** | Simple Network Management Protocol: Simple Network Management Protocol: a protocol used mostly in network management systems to monitor devices for conditions that warrant administrative attention. |
| **SSL** | Secure Sockets Layer is a security protocol for establishing encrypted links between a web server and a browser for online communication |
| **STL** | Studio-to-transmitter link for program audio feeds |
| **STS** | Studio-to-studio audio link |
| **STUN** | The STUN protocol (Simple Traversal of UDP through NATs) assists devices behind a NAT firewall or router with packet routing. A STUN client generates STUN requests and a STUN server, attached to the public internet, receives STUN requests and sends responses. |
| **TCP** | TCP protocol ensures reliable in-order delivery of data packets between a sender and a receiver |
| **TLS** | Transport Layer Security is an updated version of SSL. |
| **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** | User Datagram Protocol: the most commonly used protocol 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 |
| **Unicast** | Broadcasting of a single stream of data between two points |
| **VLAN** | Virtual Local Area Network: partitioning of a single layer-2 network to create multiple distinct broadcast domains |
| **WAN** | Wide Area Network; a computer network spanning regions and/or countries to connect separate LANs |
| **WheatNet-IP** | Network system that utilizes Internet Protocol to enable audio to be intelligently distributed to devices across scalable networks |

# 4 Introduction to the Codec

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


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

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

## Codec Features

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

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

✓ = included
**O** = optional

## Package Contents

Your codec is delivered with:

- Bridge-IT IP codec
- Multi-region plug pack 12 volt 1 Amp power supply
- 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

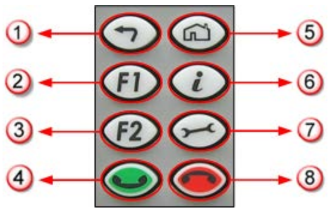
Bridge-IT has four arrow shaped navigation buttons for navigating codec menus and an **OK** button for selecting menu items.


## Dialing Keypad

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

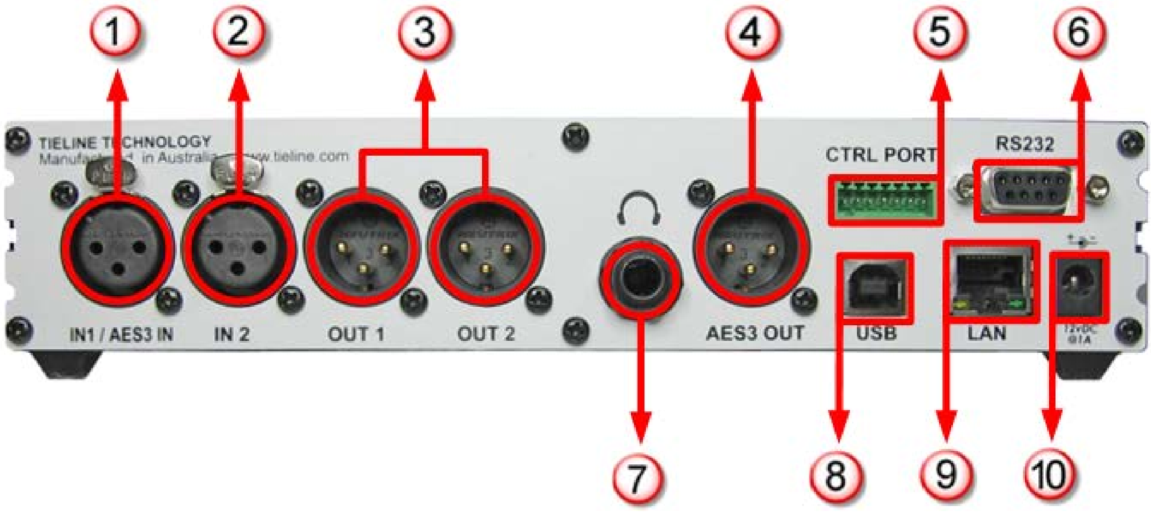
- Launch codec functions.
- Navigate menus.
- Dial and hang up connections.
- Configure contact details.

## 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 and relays |
| 3 | Function Button 2 | Press to open codec user functions |
| 4 | Connect Button | Press to dial IP connections |
| 5 | Home Button | Press to return to home screen |
| 6 | Information Button | Press to view a help menu on-screen |
| 7 | Settings Button | Press to configure codec settings |
| 8 | Hangup Button | Press to end a call |

# 6 Rear Panel Connections




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

## XLR Analog and Digital Inputs

Bridge-IT features two XLR microphone inputs.

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

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

## XLR Analog and AES3 Outputs

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

## Stereo Headphone Jack Output

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

## LAN Port

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

## Command & Control Interfaces

Bridge-IT features:

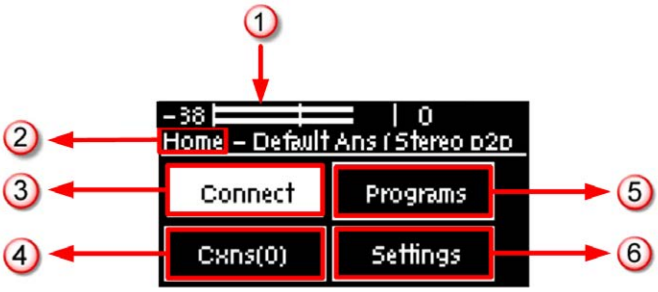
1. 2 relay inputs and 2 opto-isolated outputs for machine control via the **CTRL PORT**.
2. A nine pin **RS-232** connection for local and remote control of equipment at either end of the link
3. A **USB** 2.0 (slave) connection for codec web-GUI configuration.

## DC Power Input

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

# 7 Navigating Codec Menus

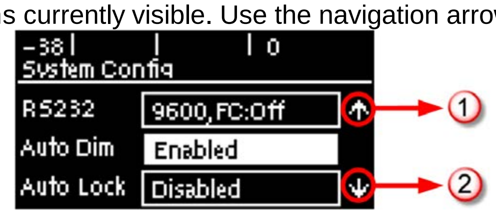
The codec has simple and intuitive menu navigation screens. All main codec menus can be launched from the **Home** screen and audio levels remain visible throughout all menus.




**Bridge-IT Home Screen**

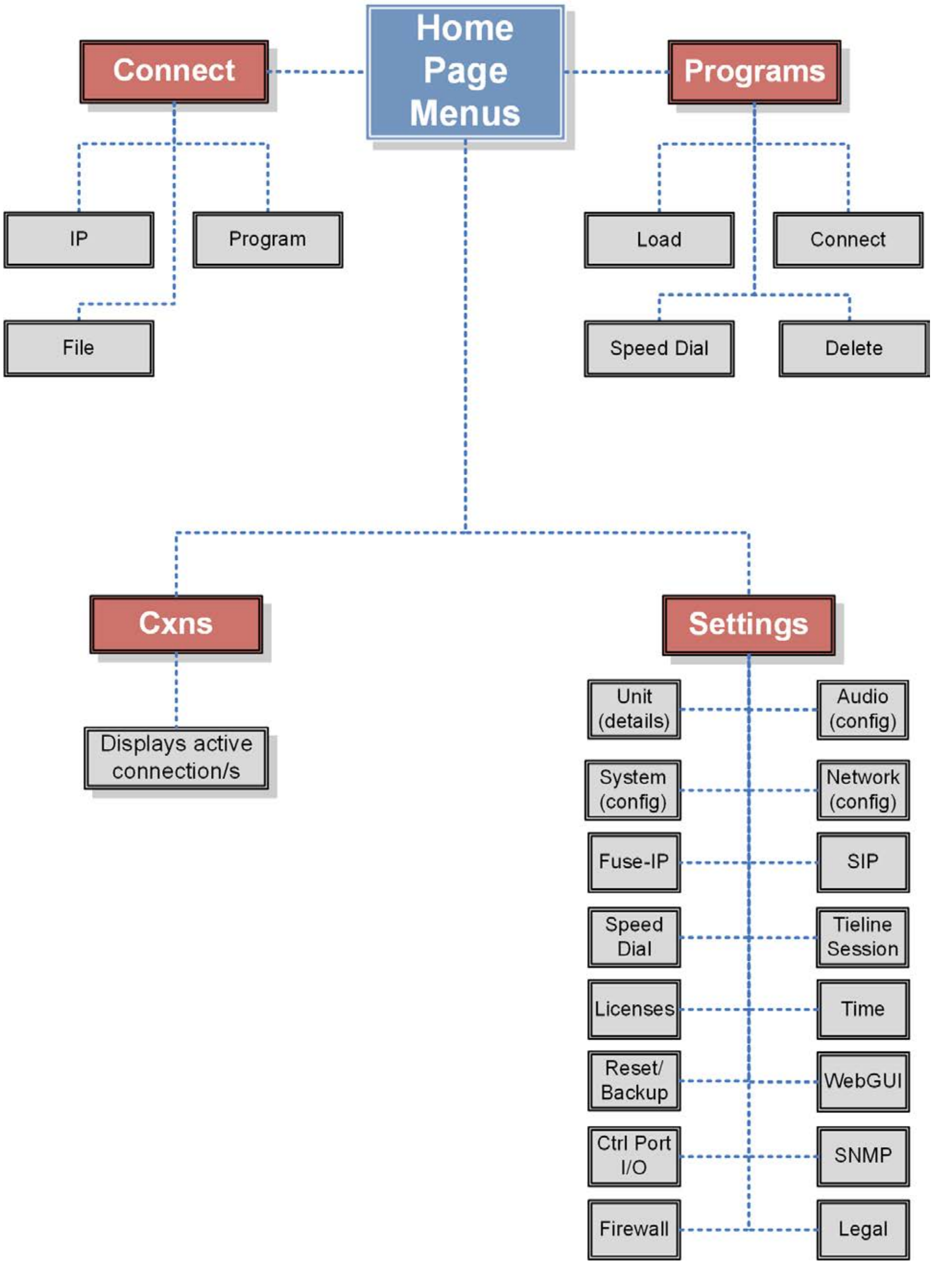
| | Features | Codec Home Screen Elements |
|---|---|---|
| 1 | PPM Meters | Left (top) and right channel audio levels |
| 2 | Screen Name | The name of the current screen being navigated |
| 3 | Connect | Select to dial & adjust connection settings |
| 4 | Cxns | Displays the number of current connections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

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

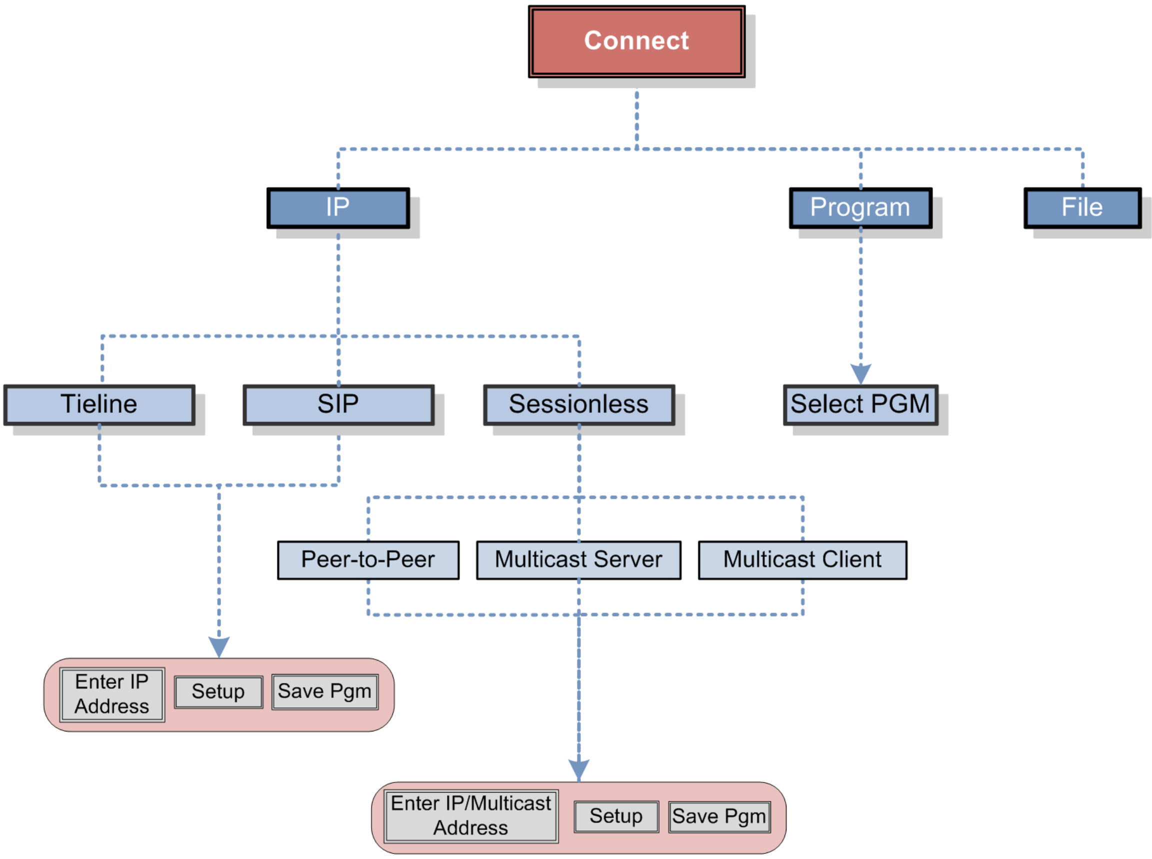



| | Features | Codec Home Screen Elements |
|---|---|---|
| 1 | Up Arrow | Arrow indicating menus can scroll upwards |
| 2 | Down Arrow | Arrow indicating menus can scroll downwards |

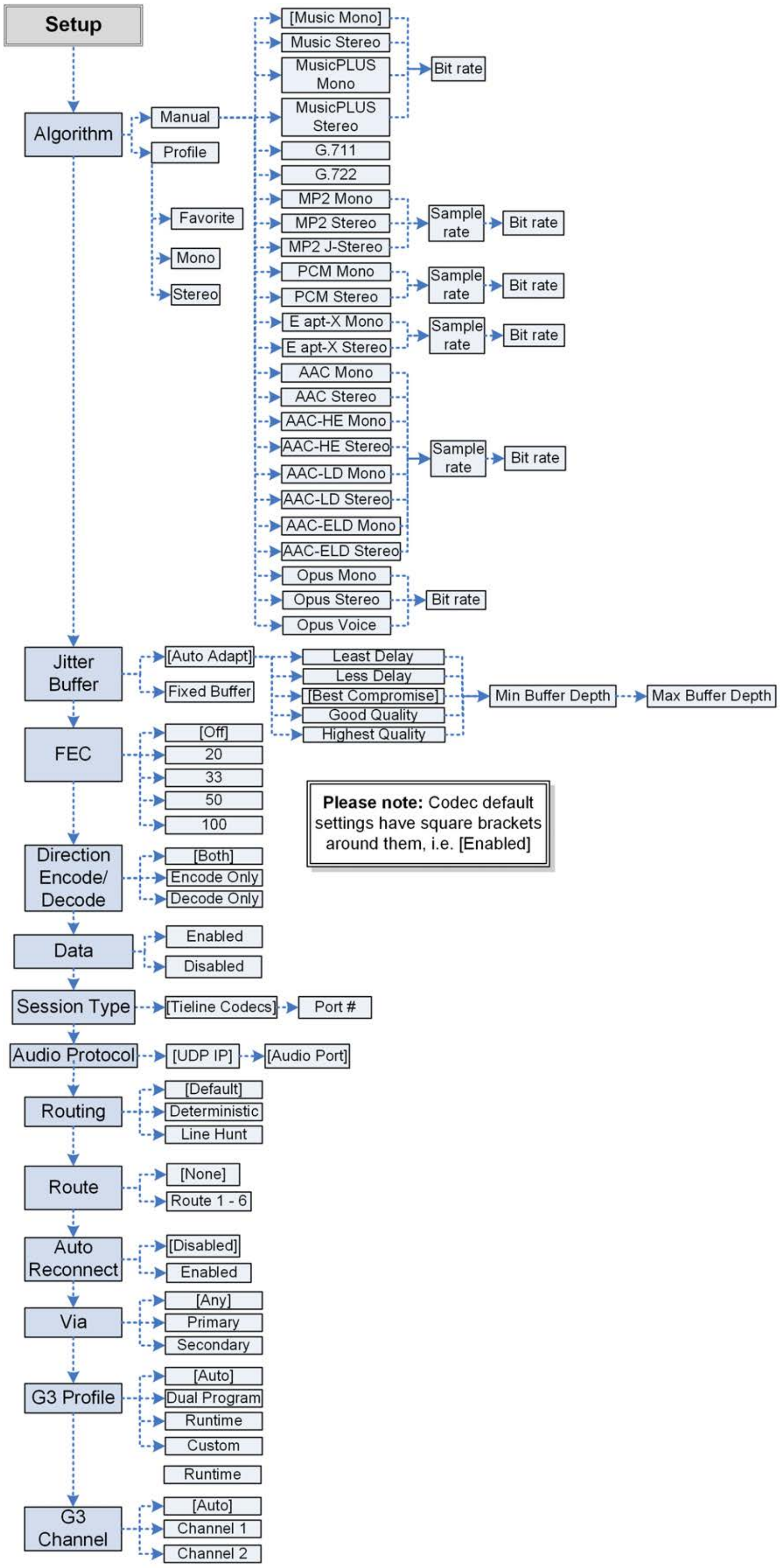
## Codec Menu Overview



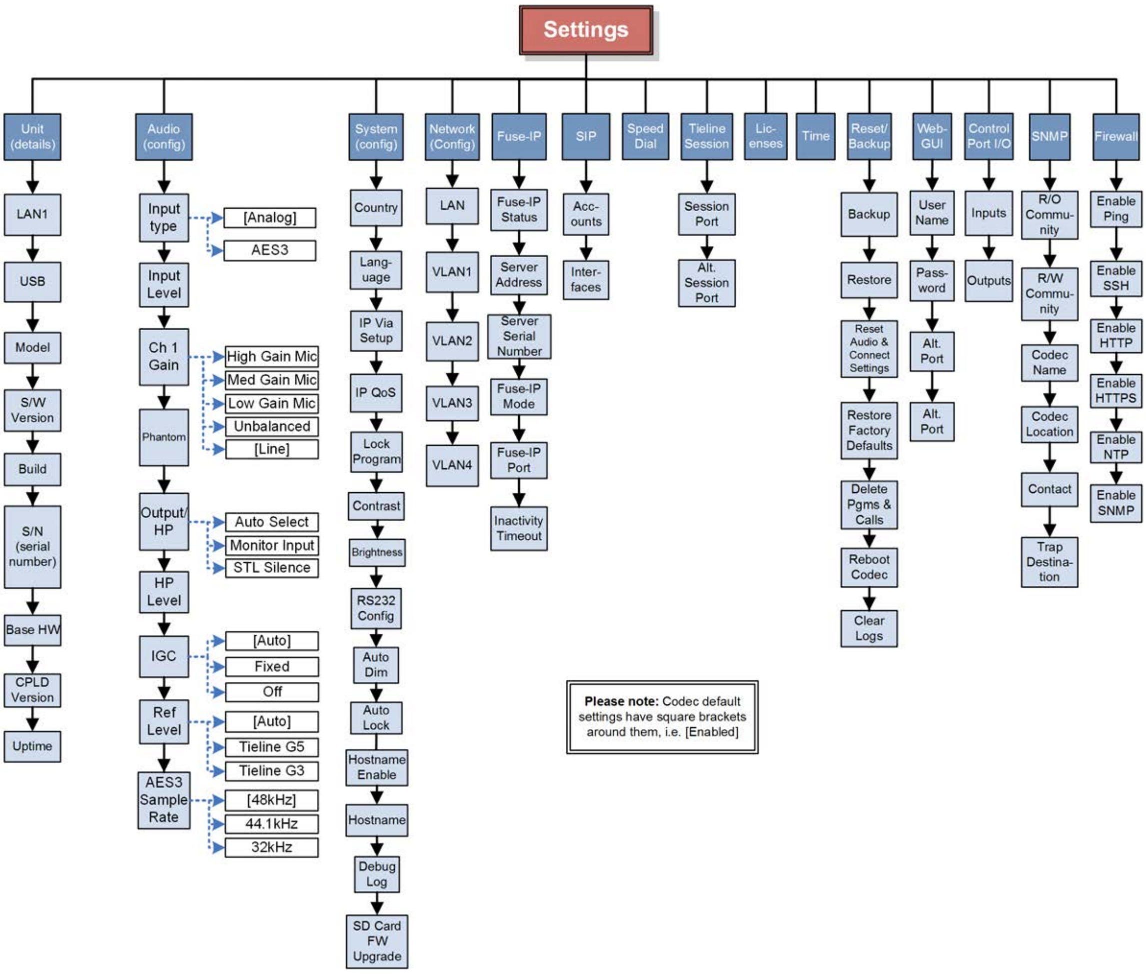
## Connect Menu



## IP Setup Menu Navigation



## Settings Menu



# 8 Adjusting Input/Meter Levels

By default, the **PPM METERS** on the front of the codec, or in the HTML5 Toolbox Web-GUI, use dBFS to express nominal operating, headroom and noise floor levels. The PPM meters display input audio by default when the codec is not connected and then switch to monitor decoded return program audio after making a connection.

## Mono and Stereo Audio Capabilities

The codec routes input 1 directly to the left output and input 2 directly to the right output when it is not connected.

To connect in stereo select a stereo algorithm in the **IP Setup** menu via **Connect > IP > Tieline > Setup**. To connect in mono select a mono algorithm in the **IP Setup** menu. In mono the codec mixes both inputs 1 and 2 together.

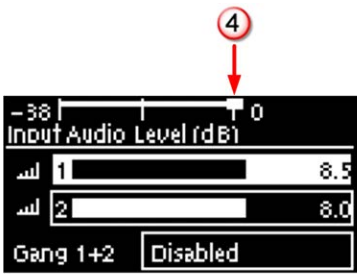
When a Bridge-IT codec answers a mono incoming call it feeds mono audio to both the left and right analog XLR outputs, as well as the left and right AES3 outputs. Note: It is not possible to mix channels 1 and 2 into dual mono outputs.

The codec feeds both analog and digital AES audio out at all times, whether the audio source is analog or digital.

## Adjusting Audio Meter Reference Scale Settings

The codec can also automatically adapt to different Tieline reference scales. A Tieline codec with proprietary Tieline session data enabled will automatically adjust the reference level to suit G5 and G3 codecs, or Report-IT, using the default **Auto** reference level setting. When connecting to a non-Tieline codec, or a Tieline codec without session data enabled, the codec will use the **Tieline G5** reference scale setting.

The default Tieline G5 audio reference scale displayed on the PPMs when you connect to a Tieline G5 codec is -38dBFS to 0dBFS (e.g. Merlin, Genie, Bridge-IT and ViA codec families). Using this reference scale audio peaks can safely reach 0dBFS without clipping, providing 18dB of headroom from the nominal 0vu point.




| | Features | Description |
|---|---|---|
| 1 | -38dBFS | PPM meter low point |
| 2 | -18dBFS | Nominal 0vu reference level at -18dBFS (+4dBu) |
| 3 | 0 dBFS | Peaks should not exceed 0dBFS when using analog audio to prevent clipping (Note: it is impossible to exceed 0dBFS when using digital audio) |
| 4 | PPM meter in clip | PPM indication displays a solid section at the right-hand end when audio is in danger of clipping |

Bridge-IT, Genie, Merlin and ViA codecs have more audio headroom than Tieline G3 audio codecs (Commander and i-Mix G3) and Tieline's Report-IT app, therefore the audio metering reference scale needs to be adjusted when connecting to one of these products. The comparison table below outlines the reference scales for G5 and G3 codecs, plus Report-IT in dBFS, as well as the equivalent dBU scale.

| | Reference Level | Description | dBu | dBFS |
|---|---|---|---|---|
| 1 | Tieline G5 | PPM meter low point | -16dBu | -38dBFS |
| | | Nominal 0vu reference level | +4dBu | -18dBFS |
| | | Level at which audio will clip/distort | +22dBu | 0dBFS |
| 2 | Tieline G3 | PPM meter low point | -11dBu | -29dBFS |
| | | Nominal 0vu reference level | +4dBu | -14dBFS |
| | | Level at which audio will clip/distort | +18dBu | 0dBFS |
| 3 | Report-IT | PPM meter low point | -9dBu | -23dBFS |
| | | Nominal 0vu reference level | +4dBu | -10dBFS |
| | | Level at which audio will clip/distort | +14dBu | 0dBFS |

## Configure Tieline G3 Codec Audio Reference Scales

Genie, Merlin, Bridge-IT and ViA codecs have more audio headroom than Tieline G3 audio codecs, therefore the audio metering reference scale needs to be adjusted when these codecs connect to a Commander or i-Mix G3 codec. The G3 metering scale is between -29dBFS and 0dBFS and audio levels should average around the nominal 0vu point at -14dBFS. Audio peaks should not exceed 0dBFS when using analog audio to prevent clipping.

1. Press the **SETTINGS** button.
2. Navigate to **Audio** and press .
3. Navigate to **Ref Level** and press .
4. Select **Tieline G3** and press .

**Important Note:** If your codec (Genie STL, Genie Distribution and Bridge-IT) supports sending multi-unicast connections and the **Auto** (default) reference level is selected, the first codec to which you connect configures the reference level used for all subsequent multi-unicast connections.

## Configure Report-IT Audio Reference Scales

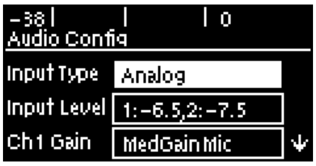
The **Report-IT** setting is used for compatibility when connecting using Tieline's Report-IT smartphone application. The Report-IT metering scale is between -23dBFS and 0dBFS and audio levels should average around the nominal 0vu point at -10dBFS. Audio peaks should not exceed 0dBFS when using analog audio to prevent clipping.

1. Press the **SETTINGS** button.
2. Navigate to **Audio** and press .
3. Navigate to **Ref Level** and press .
4. Select **Report-IT** and press .

## Channel 1 Mic/Line Level Audio Adjustment

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

1. Press the **SETTINGS** button.
2. Navigate to **Audio** and press .
3. Ensure **Input Type** is set to **Analog**.




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


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

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

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

## Quick Adjustment of Input Levels

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


| | Input Audio Features | Description |
|---|---|---|
| **1** | Channel On Symbol | Symbol indicates a channel is turned on |
| **2** | Channel Off Symbol | Symbol indicates a channel is turned off |
| **3** | Input 1 Level Control | Input 1 gain indication; adjust in 0.5db increments |
| **4** | Input 2 Level Control | Input 2 gain indication; adjust in 0.5db increments |
| **5** | Ch1/2 Gang Indication | Indicates whether ganging is enabled or disabled |

## Intelligent Gain Control (IGC)

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

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

### IGC Auto Level Recovery

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

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

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

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

## Ganging Audio Channels

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

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

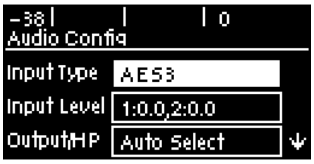
When channels 1 and 2 are ganged together:

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

# 9 Configuring AES3 Audio

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

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




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

**Important Note:** Input attenuation is available when the input type is **AES** digital, however there is no additional gain. If you switch back to the analog input setting after selecting AES3, the previous analog settings will be recovered.

## AES/EBU Sample Rate Conversion

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

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

## AES3 Audio Out & SD Card Sample Rate

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

## Adjusting the Codec Output Sample Rate

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

1. Press the **SETTINGS** button.
2. Navigate to **Audio** and press .
3. Navigate to **AES3 SR** and press .

4. Select a preferred sample rate then press 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** sample rate setting are the same.

# 10 Headphone/Output Monitoring

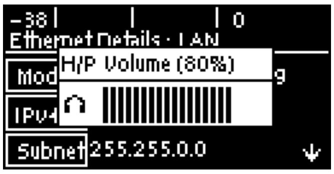
The 6.35mm (1/4") stereo headphone output on the codec can be used for monitoring audio inputs 1 and 2 and return link audio.

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

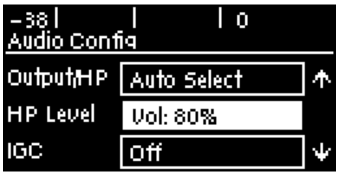
## Adjusting Headphone Output Levels

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

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



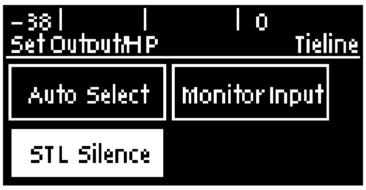

Headphone levels can also be adjusted by navigating to **SETTINGS** **> Audio** and using the down button to navigate to **HP Level** and press .




## Output/Headphone Monitoring Settings

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

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




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

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

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

### Auto Select Mode

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

### Monitor Input Mode

Select **Monitor Input** to configure the codec to always monitor input audio. In this mode local audio is looped to the output when the codec is not connected. This setting may be useful if an announcer wants to monitor their own voice and not return audio when connected.

### STL Silence Mode

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

# 11 Language Selection

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

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

# 12 About Program Dialing

## What Defines a Program?

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

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

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

## Using Programs to Dial between Two Tieline Codecs

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

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

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

## Creating Programs

Only the simplest peer-to-peer (point-to-point) programs can be created using the codec front panel. The HTML5 Toolbox Web-GUI contains a **Program Manager panel** with a wizard for configuring program settings and backup connections. Edit settings easily at the touch of a button and use existing programs as templates for creating other programs.

### Mono and Stereo Peer-to-Peer Programs

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

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

Ensure you configure all the correct connection settings when using the codec front panel, because these are stored as part of the program's profile when you first connect. They cannot be adjusted afterwards without using the editing features in the **Program Manager panel** within the HTML5 Toolbox Web-GUI.

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

## Multi-unicast Programs:

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

## Multicast Programs:

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

# 13 Getting Connected Quickly

## Preparing to Connect

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

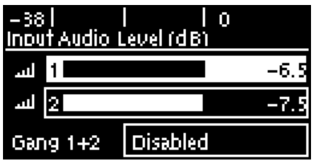
1. Attach the supplied 12 volt power supply to the codec.
2. Attach an RJ45 Ethernet cable to the **LAN** port on the rear panel of the codec.
3. Attach headphones to the 6.35mm (1/4") headphone jack on the rear panel of the codec.
4. Check that the correct country is selected in the codec.
   i. Press the **SETTINGS** button.
   ii. Navigate to **System** and press the button.
   iii. Navigate to **Country** and press the button.
   iv. Use the navigation buttons to select your country of operation.
5. Make sure you have the IP address of the codec you are dialing, or have used the HTML5 Toolbox web-GUI to load the programs you will be using to dial onto the codec. (see Configuring IP Addresses).

## 13.1 Quick Steps to Connect Bridge-IT

**Important Notes:**
- The following procedure will create a custom peer-to-peer connection program using the codec front panel keypad and navigation buttons. It instructs how to connect your codec over IP for the very first time without using the Toolbox web-GUI and your computer for configuration.
- See the HTML5 Toolbox Web-GUI Introduction for details on configuring connections remotely via a computer.
- See Installing the Codec at the Studio for valuable information about installing your codec, negotiating firewalls and port forwarding.
- See Tips for Creating Reliable IP Connections for a range of IP information to assist with setting up IP services for your codecs.
- See Testing IP Network Connections to learn how you can test and verify the reliability of your IP connection.

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

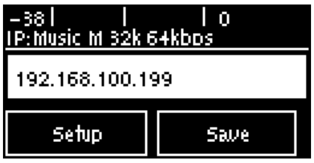
**Important Note:** 15 volt phantom power is not supplied to input 1 by default. To adjust this setting select **Settings**, then **Audio** and then **Phantom**. Press the **OK** button to toggle between **Disabled** and **Enabled**.



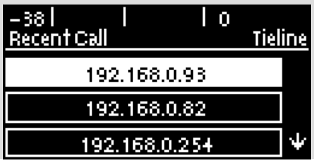

2. Press the **HOME** button to return to the **Home** screen, select **Connect** and press the button, then select **IP** and press the button.


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


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

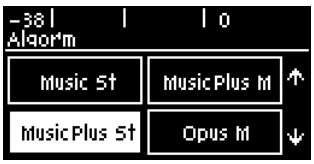



**Important Note:** The codec remembers recent IP addresses just like a cell-phone. To view these addresses just press the **OK** button when you select the **Connect IP** screen. The most recent addresses and programs are listed first and you can use the navigation buttons to scroll up and down. Press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

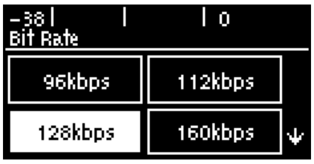



5. Press the down navigation button to select **Algor'm** (algorithm) and press OK.


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

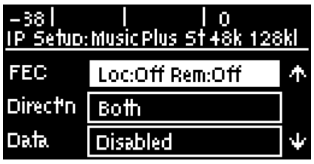



7. If you decide to manually configure the algorithm, use the navigation buttons to select your preferred sample rate (if displayed) and bit rate. Press OK after selecting each option.

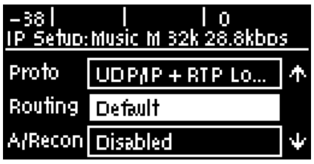



8. Press the down navigation button to select **Jitter** (jitter buffer) and press to select a different automatic jitter buffer setting for your connection. Alternatively you can enter a fixed jitter buffer value in milliseconds (maximum 5000 ms). The default **Auto, Best Compromise** setting is a good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


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

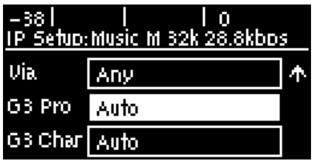



10. Navigate to **Routing** to configure routing options as per the table below:




| **Routing Type Options:** | |
|---|---|
| **Default** IP | No **Dial Route** or **Answer Route** is configured. An incoming call will be routed to an audio stream on a first-come, first-served basis in a multi-stream program. Note: By default IP streams are routed using audio ports. |
| **Deterministic** IP | Use of **Dial** and **Answer Routes** is not usually necessary over IP because dedicated ports or **Line Hunt** mode call answering is employed. However dial routes can be used over IP when a single stream on an answering codec answers using POTS and/or ISDN connections, as well as IP. This effectively creates an answering group using different transports. |
| **Line Hunt** IP | Create line hunt groups for multiple incoming callers on a first come, first served basis. This is ideal for separating groups of inputs and outputs between different studios or stations. See Line Hunt Call Answering for more information. |

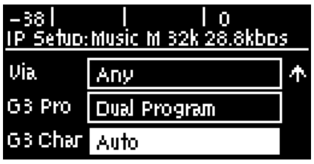
11. Navigate to **G3 Profile** to configure profile settings when dialing a Tieline G3 Codec.




**Important Note:** The G3 profile setting supports maintaining specific G3 codec settings when answering a call from a G5 codec.

1. **Auto**: The codec will dial the G3 codec and connect in mono or stereo.
2. **Dual Program**: This allows the codec to dial a G3 codec with a Dual Program profile loaded and support two simultaneous mono connections.
3. **Runtime**: The G3 codec will retain runtime settings when answering a call from a G5 codec.
4. **Custom**: The G3 codec will load a specified profile, e.g. profile 6, which is the first custom profile number.

12. Navigate to **G3 Channel** when connecting to a G3 codec in dual mono mode. This setting lets you configure which G3 channel (encoder) is used when the G3 codec receives a call from this codec. E.g. **Channel 1** will route the incoming stream to Encoder 1 on the G3 codec and **Channel 2** will route the incoming stream to Encoder 2 on the G3 codec.




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

**Important Note:** At this point you can navigate to **Save** on the **IP** dialing 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.

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


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

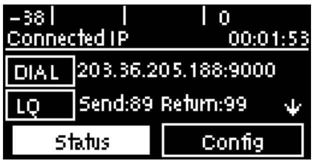
## 13.2 Monitoring IP Connections

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


To view more detailed connection information:

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

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



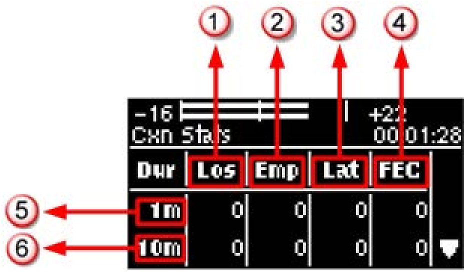

### Link Quality (LQ) Readings

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

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

### Viewing Connection Statistics

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



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

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

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

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

## 13.3 Load and Dial Custom Programs

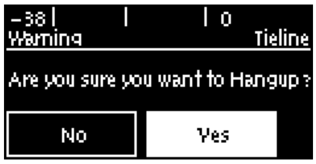
Custom programs stored on the codec are simple to load and dial from the codec front panel.

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

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

## 13.4 Disconnecting a Connection

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




## 13.5 Redialing a Connection

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

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


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

### Redialing Manually from the Connect IP Screen

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

## 13.6 Configuring Auto Reconnect

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

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

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

## 13.7 Speed Dialing Connections

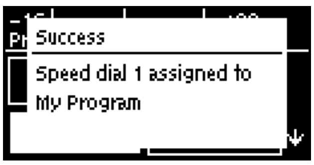
### Assigning Speed Dial Numbers

There are two methods of assigning speed dial numbers to saved programs:

#### Assigning Speed Dials via the Programs Menu

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

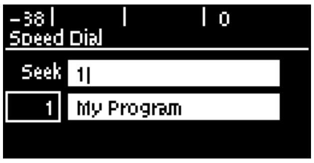

5. Navigate to the speed dial number you want to assign to the selected program and press the button.
6. A confirmation message will display the number assigned.




#### Assigning Speed Dials via the Speed Dial Menu

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Speed Dial** and press the button.

3. Navigate to an available speed dial number and press the button.
4. Navigate to a program in the **Program List** and press the button to select and assign a program.



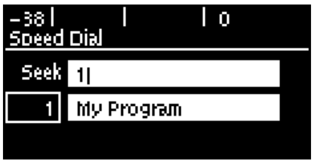

## Speed Dialing

There are two ways to speed dial.

**Option 1:** Press and hold a speed dial number on the **KEYPAD** for two seconds and then the codec will automatically dial when you release your finger from the keypad. Note: this is only available for buttons 0-9. Use the following procedure to dial speed dial number 10 or higher, or speed dial numbers 0-9 if preferred.

**Option 2:**

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




# 13.8 Dial/Disconnect Multiple Connections

## Multiple Connections within Programs

Multi-unicast programs allow you to simultaneously transmit a mono or stereo audio stream to up to 6 destination codecs. Multi-unicast programs can only be created using the HTML5 Toolbox web-GUI. There are two ways to simultaneously dial multiple IP audio stream connections multi-unicast programs:

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

## Dialing Multiple Connections via the Front Panel

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

It is also possible to redial the connection.

### Disconnect All Connections

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

### Disconnecting Individual Multi-unicast Connections

It is only possible to disconnect individual connections via the **Connections panel** in the dialing codec's HTML5 Toolbox web-GUI.

## 13.9 Creating a Multicast Server Program

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

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

### Prerequisites:

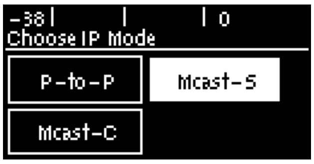
- Bridge-IT firmware v.1.01.00 r4219 or higher.
- ToolBox web-GUI v.1.2.2.3 or higher.
- A multi-unicast license installed in the server codec (Note: the multi-unicast license includes multicast server capability).

**Important Notes:**

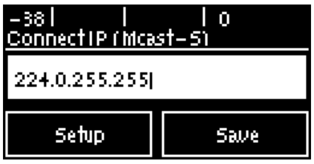
- You cannot edit a program when it is currently loaded in the codec.
- Ensure all connection related settings like the port, algorithm, bit rate (etc) match on both multicast server and client programs or they will not be able to join multicast streaming sessions.
- There is no jitter buffer setting in a multicast server program because it is an encode only program and never receives audio packets.
- You can lock a loaded custom program in a codec to ensure the currently loaded program cannot be unloaded by a codec dialing in with a different program type.
- Always dial the multicast server codec connection first before connecting multicast client codecs.
- Multicast client codecs will display return link quality (LQ) only. The **Return** reading represents the audio being downloaded from the network locally. Multicast server codecs do not display LQ readings.
- It is not possible for a G3 codec to receive multicast IP audio streams.
- To learn more about programs see the section titled "About Program Dialing."
- See HTML5 Toolbox web-GUI documentation for more detailed information about "Configuring Multicast Server Programs" or "Configuring Multicast Client Programs."

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

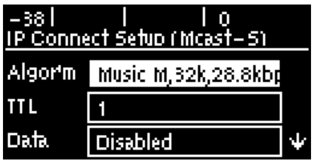

2. Select **M'cast-S** (Multicast Server) to configure a server codec program.




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

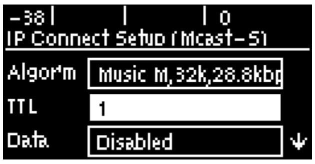



4. Press the down navigation button to select **Algor'm** (algorithm) and press .

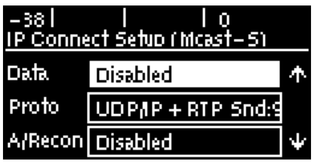



5. Use the navigation buttons to select an algorithm profile or manually select algorithm settings, then press .

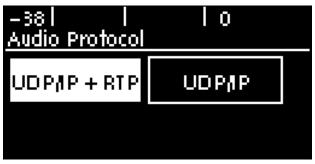
6. Navigate to and select **TTL** and enter the IP Time-to-Live value, then press . Please note: The TTL value you need to use is dependent upon your network infrastructure. Please consult your network administrator if you are unsure about how to configure this setting.



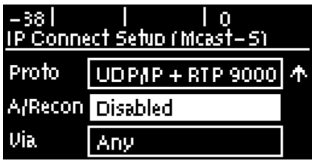

7. Select **Data** to enable or disable data in the audio stream. See RS232 Data Configuration for more information on data connections.




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

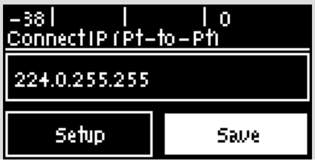


9. If required, enable **A/Recon** (auto reconnect) and use **Via** to specify which IP streaming interface is used to dial this connection, e.g. **Primary** (Ethernet port) or **VLAN** (if configured). Note: By default **Any** will select **Primary**.




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

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




### Connecting a Multicast Program

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

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

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

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

## 13.10 Creating a Multicast Client Program

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

### Prerequisites:

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

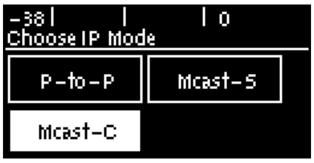
**Important Notes:**

- You cannot edit a program when it is currently loaded in the codec.

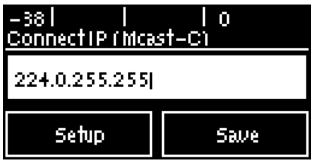
- Ensure all connection related settings like the port, algorithm, bit rate (etc) match on both multicast server and client programs or they will not be able to join multicast streaming sessions.
- The default UDP audio port is 9000 for a multicast client program configured via the codec front panel.
- You can lock a loaded custom program in a codec to ensure the currently loaded program cannot be unloaded by a codec dialing in with a different program type.
- Always dial the multicast server codec connection first before connecting multicast client codecs.
- Multicast client codecs will display return link quality (LQ) only. The **Return** reading represents the audio being downloaded from the network locally.
- It is not possible to connect to a G3 codec and receive multicast IP audio streams.
- To copy multicast client programs onto multiple codecs see Backup and Restore Functions.
- To learn more about programs see the section titled "About Program Dialing".
- See HTML5 Toolbox web-GUI documentation for more detailed information about "Configuring Multicast Client Programs."

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


2. Select **Mcast C** (Multicast Client) to configure a client codec program.




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



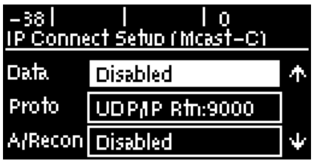

4. Press the down navigation button to select **Algor'm** (algorithm) and press .


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

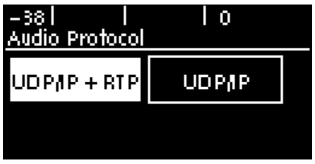
6. Click to configure your preferred Jitter Buffer setting. Select **Auto Adapt** and your preferred automatic jitter setting, or **Fixed Level**. For a fixed buffer setting enter the Jitter **Buffer Depth**, which has a maximum setting of 5000ms, then press .


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

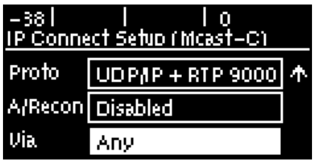
7. Select **Data** to enable or disable data in the audio stream. See RS232 Data Configuration for more information on data connections.




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

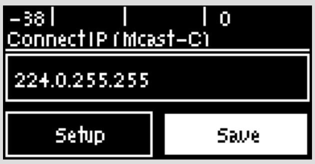



7. If required, enable **A/Recon** (auto reconnect) and navigate to **Via** to specify which IP streaming interface is used to dial this connection, e.g. **Primary** (Ethernet port) or **VLAN** (if configured). Note: By default **Any** will select **Primary**.




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

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



### Connecting a Multicast Client Program

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

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

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

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

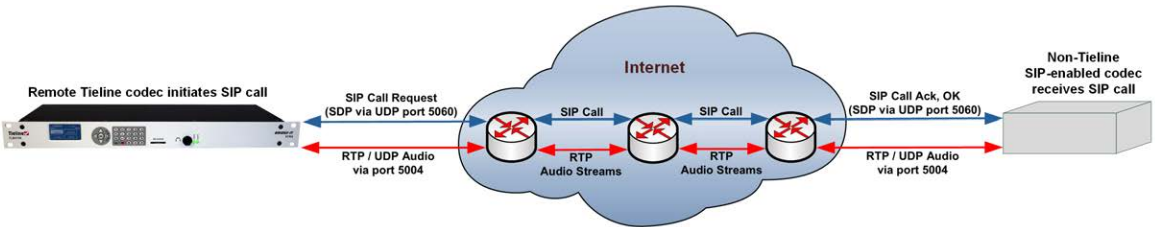
## 13.11 About SIP

SIP provides interoperability between different brands of codecs due to its standardized protocols for connecting different devices. The codec is fully EBU N/ACIP Tech 3326 compliant when connecting using SIP (Session Initiation Protocol) to other brands of IP codecs.

SIP is also a useful way of dialing another device and locating it easily. This task is usually performed by SIP servers, which communicate between SIP-compliant devices to set up a call. SIP connections can be made in two ways; registered or unregistered.

### Unregistered Peer-to-Peer SIP Connections

Codecs don't need to be registered to a SIP server to dial peer-to-peer SIP connections. An unregistered SIP peer-to-peer connection involves two codecs connecting to each other directly using an IP address, as you would for a standard Tieline IP call. The difference is that a Tieline IP call uses proprietary Tieline session data to negotiate call parameters (e.g. algorithm and bit rate) when a call is established, whereas a peer-to-peer SIP connection uses Session Description Protocol (SDP) for this purpose. SIP provides interoperability between different brands of codecs due to its standardized protocols for connecting dissimilar devices and is used when connecting Tieline codecs to non-Tieline devices.




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 and SDP is used for. The second stage is when data transfer occurs and this is left to the other protocols such as RTP/UDP to stream audio data. SDP works with a number of other protocols, to deliver the following functions when connecting devices over SIP:

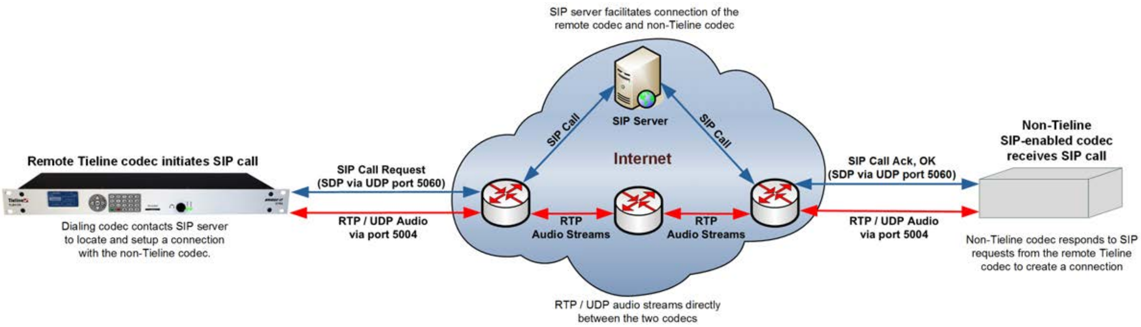
- Establish a codec's location.
- Determine the availability of a codec.
- Negotiate the features to be used during a call, e.g. the algorithm and bit rate.
- Provide call management of participants.

- Adjust session management features while a call is in progress (e.g. termination and transfer of calls).

All the mandatory EBU N/ACIP 3326 algorithms are supported in the codec, including G.711, G.722, MPEG-1 Layer 2 and 16 bit PCM, as well as optional algorithms including Opus, LC-AAC, AAC-LD, HE-AACv2 and aptX Enhanced.

## SIP Server Connections

The benefit of using a SIP server to connect is that any device can be 'discovered' via its SIP server registration. This is particularly useful if a codec is being used in multiple locations with IP addresses that are DHCP assigned. These DHCP addresses are unreliable and are not recommended for live broadcast connections. As long as your codec and the device you are dialing are both registered to a SIP server you can connect by simply dialing the destination SIP address.




Some SIP servers route RTP audio through the SIP server as well and Tieline recommends avoiding this type of server whenever possible. Otherwise you will be reliant on the SIP server for streaming broadcast audio packets and most servers are not designed for mission critical packet streaming.

To dial a codec via a SIP server requires:

1. Both devices to be registered with separate SIP accounts.
2. Both codecs configured to operate in SIP mode.
3. The IP address of the SIP server.
4. An IT administrator to open UDP port 5060 to enable SIP traffic, as well as UDP audio port 5004.

A SIP server administrator should be able to provide the following details to enable SIP registration of a device:

- Username
- Authorized User
- SIP address
- Domain
- Realm
- Registrar
- Registar port
- Outbound Proxy
- Proxy port

## Getting Started with SIP

To dial over SIP peer-to-peer without using a SIP server see Dialing SIP Connections. To dial over SIP using a SIP Server you will need to:

1. Register the codec to a SIP server using SIP account credentials.
2. Configure a SIP interface in the codec. Note: The **SIP1** or **SIP2** interface will include the proxy and port settings, as well as the selected IP interface used to make the connection, e.g. **LAN1**.

**Important Notes:**

- The codec supports dialing over SIP using a registered SIP server account, or peer-to-peer using one of the two SIP interfaces **SIP1** and **SIP 2**.
- SIP dialing is only supported over point-to-point connections, not multi-unicast connections.
- Some ISPs and/or cellular networks may block SIP traffic over UDP port 5060.
- Tieline G3 codecs do not support connections using algorithms like AAC, aptX Enhanced and Opus and will default to MPEG Layer 2 if an incoming call is configured to use these algorithms.
- Failover and SmartStream PLUS redundant streaming are not available with SIP connections.
- When connecting to a Tieline G3 codec using SIP you need to manually select the G3 audio reference level in the codec. To do this select **SETTINGS** **> Audio > Ref Level > Tieline G3**. In addition, configure 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]**

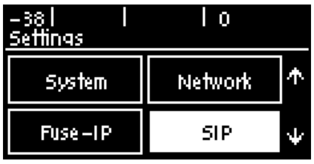
### 13.11.1 Configuring SIP Interfaces

The codec supports dialing over two SIP interfaces simultaneously.

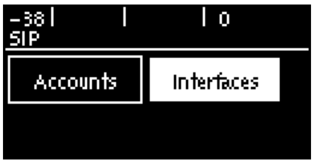
**Important Notes:**
1. SIP interfaces are disabled by default.
2. **SIP1** is configured to use **LAN** by default, which is mapped to the **Primary** Via interface by default.
3. **SIP2** is configured to use **VLAN1** by default, which is mapped to the **Secondary** Via interface by default.
4. **SIP1** and **SIP2** each need to use a separate IP interface when connecting, e.g. **LAN** or **VLAN1**.
5. **SIP1** and **SIP2** can each make multiple SIP calls, e.g. two calls can be made over **SIP1**, or two calls can be made over **SIP2**.
6. The settings for **SIP1** and **SIP2** cannot be edited if the interface is enabled.
7. Enter a public IP address in the **Public IP** menu if you want to dial over SIP from behind a firewall. Then configure port forwarding to route traffic to the codec's local IP address behind your firewall.

To configure **SIP1** or **SIP2**:

1. Press the the **SETTINGS** button, then navigate to **SIP** and press the button.




2. Navigate to **Interfaces** and press the button.

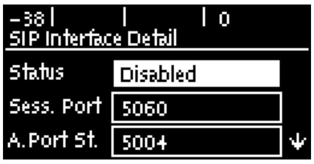



3. Select either **SIP1** or **SIP2** and press .

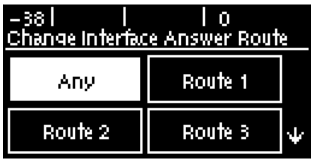

4. Select **Disable** to disable the interface before editing settings.


5. Navigate to **Edit** and press the button.


6. Navigate to each field in turn and press the button to edit SIP interface settings as required.




7. Select **Ans Route** and use the **Change Interface Answer Route** screen to route calls using this SIP interface to a specific audio stream. The route setting in this menu must correspond with the answering route configured in an audio stream within the loaded program. If the default value Any is used then a call will be routed to an audio stream on a first-come-first-served basis in a multi-stream program.




8. The SIP interface **Wizard** can also be used to enter SIP account information. Select **Wizard** to navigate through each screen in turn automatically.


9. Navigate to **Enable** in the **Modify SIP Interface** menu and press the button to enable the interface so that it can be used to make a SIP call.


## 13.11.2 Configuring SIP Accounts

### Getting Started

Up to 6 SIP accounts can be configured in the codec and registering codecs for SIP connectivity is simple. First, select the SIP server to which you will register your codec. On a LAN this may be your own server, or it could be one of the many internet servers available. When you register an account with a SIP server you will be provided with:

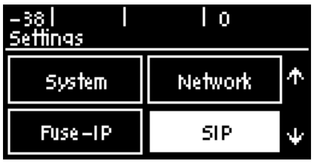
- The SIP registrar and Proxy Server details.
- An authorized user or username (often the same as a SIP number).
- A password.
- Domain details.
- Realm details (in most cases leave blank).

- Registration Timeout (this shouldn't need to be adjusted from the default setting).

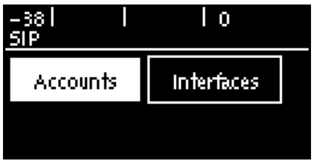
**Important Notes:**
- In most situations it is best to configure a SIP account when the codec is configured with a public IP address.
- Each SIP account can only be mapped to a single SIP interface, i.e. **SIP1** (**LAN1**) or **SIP 2** (**VLAN1**).
- Up to 6 SIP accounts can be added to the codec.
- It is also possible to add and register a SIP account to your codec using the HTML5 Toolbox Web-GUI.

## Adding a SIP Account

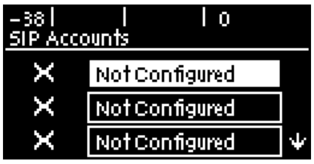
1. Press the the **SETTINGS** button, then navigate to **SIP** and press the button.




2. Navigate to **Accounts** and press the button.

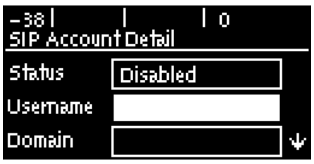



3. Select an account which is **Not Configured** and press .

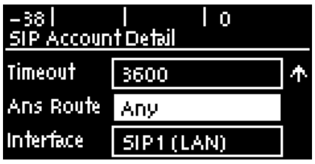


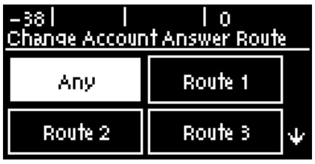

4. Select **Edit** to enter SIP server account details, then press .


5. Navigate to each field in turn and press the button to enter SIP account credentials.



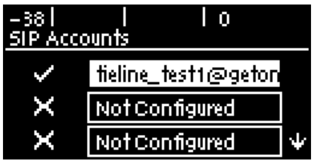

6. Select **Ans Route** and use the **Change Account Answer Route** screen to route incoming calls to this SIP account to a specific audio stream. The route setting in this menu must correspond with the answering route configured in an audio stream within the loaded program. If the default value Any is used then a call will be routed to an audio stream on a first-come-first-served basis in a multi-stream program.




7. After completing account configuration, select **Enable** in the **Modify SIP Account** menu and press the OK button to enable the SIP account and register it with the SIP server.

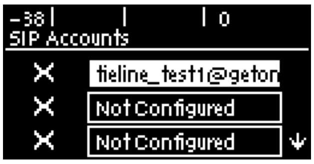
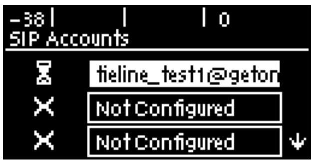
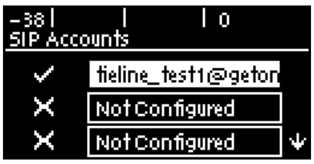

8. To confirm the account has been registered successfully, verify the account has a "tick" next to it in the **SIP Accounts** menu.




**Important Note:** Once enabled, the SIP account can be used when creating a new SIP connection. Configure this menu setting via **Connect > IP > SIP > Setup > Account**.

## Confirming Account Registration

There are three symbols displayed on the screen next to an account which indicate SIP account registration status.

| Symbol | Description |
|---|---|
|  | Cross symbol indicates the account is not yet registered. |
|  | Hourglass symbol indicates account registration is currently being attempted. |
|  | Tick symbol indicates the account is registered to a SIP server. |

## Troubleshooting SIP Registration

If a SIP account is not being registered please check the following:

1. Confirm all account registration information has been entered correctly.

2. Confirm the SIP **Interface** (**SIP1** or **SIP2**) configured in the account is enabled.
3. Verify that the **Via** selected in the **SIP1** or **SIP2** interface settings corresponds with the network interface being used by the codec to register the account. E.g. **LAN1** or **VLAN**.

## 13.12 Dialing SIP Connections

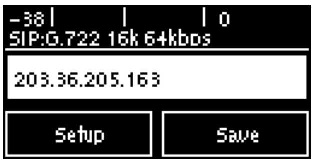
**Important Note:**
1. SIP interfaces are disabled by default.
2. **SIP1** is configured to use **LAN1** by default.
3. **SIP2** is configured to use **VLAN1** by default.
4. It is not possible to renegotiate the connection bit rate over a SIP connection.
5. When connecting to a Tieline G3 codec using SIP you need to manually select the G3 audio reference level. To do this select **SETTINGS** **> Audio > Ref Level > Tieline G3**. In addition, select the following on the G3 codec prior to dialing.
   - Select either a mono or stereo profile
   - Select **[Menu] > [Configuration] > [IP1 Setup] > [Session Type] > [SIP]**
   - Select **[Menu] > [Configuration] > [IP1 Setup] > [Algorithm] > [G711/G722 or MP2]**

### Dialing Peer-to-Peer SIP Connections

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

To make a peer-to-peer SIP call between codecs we recommend both codecs use public IP addresses. Find out the IP address of the codec being dialed and configure each codec with a compatible algorithm and sample rate etc. If the remote codec has a private IP address then it should be configured for port forwarding and should dial the public IP address at the studio.

1. To dial peer-to-peer press the **HOME** button to return to the **Home** screen, select **Connect > IP > SIP**.
2. Use the numeric **KEYPAD** to enter the IP address of the codec you want to dial, using the or buttons to enter the periods in the IP address and use the **RETURN** button to delete numbers already entered.




3. Then press the down navigation button to select **Setup** and press to adjust the algorithm, jitter buffer and encode/decode direction if required.
4. Press the **RETURN** button to navigate backwards to the **Connect SIP** screen.
5. Press the **CONNECT** button to make a connection.

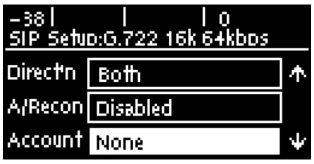
### Dialing SIP Addresses

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


## Dialing Using a SIP Account

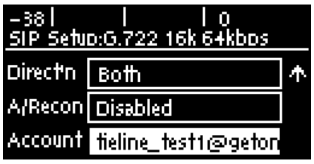
To dial using a SIP account you first need to add a SIP account and register it to the codec. See Configuring SIP Accounts for more info.

1. Press the **HOME** button to return to the **Home** screen, select **Connect > IP > SIP**.
2. Then press the down navigation button to select **Setup** and press .
3. Navigate down to **Account** and and press .




4. Navigate to a registered SIP account and press .


5. The SIP **Account** is now selected and the codec will attempt to establish the connection via a SIP server.




**Important Notes:**

- See Configuring SIP Accounts for instructions on entering SIP account details into the codec.
- If you don't save the program during configuration, a temporary program is created after you dial the SIP connection for the first time using the codec **KEYPAD**. The temporary program will appear in the recent calls list if you want to redial the program.

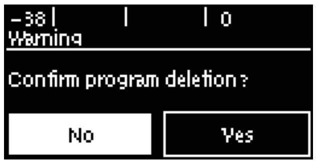

It is also possible to configure SIP programs using the Toolbox web-GUI.

## 13.13 Deleting Programs

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


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


5. Confirm program deletion and press the  button.




## 13.14 Selecting Algorithm Profiles

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

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

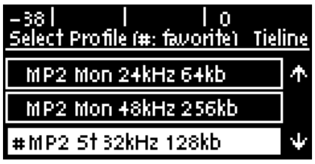
1. Press the **HOME**  button to return to the **Home** screen.
2. Use the navigation buttons to select **Connect** and press the  button.
3. Select **IP** and press the  button.
4. Select **Tieline** session mode and press the  button. Note: algorithm profiles are only available for Tieline session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

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

## Adding a Profile into the Favorite Menu

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

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




## Deleting a Profile from the Favorite Menu

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

## Available Profiles

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

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

## 13.15 Bridge-IT Backup Options

Tieline codecs feature highly advanced backup and redundancy options to maintain reliable audio codec streaming. These include the options outlined in the following table:

| Tieline Audio Codec Backup Features | | | |
|---|---|---|---|
| **Backup Option** | **Backup Behavior** | **Time Required to Respond** | **How to Enable** |
| **SmartStream PLUS** | Concurrent redundant IP packet stream sent; codec detects IP packet loss or delayed packets. | No fail over time delay - simultaneous dual redundant streaming. | Enabled in dialing codec program; configures local decoding, or remote decoding via session data. |
| **Fuse-IP** | Multiple IP interfaces are bonded and data is shared across interfaces (Note: Bridge-IT supports only one interface but can connect to remote codecs using Fuse-IP bonding). | No time delay if interface is lost; immediately adjusts to existing network data capacity | Configure multiple IP interfaces and create a Fuse-IP "Tunnel" prior to dialing destination codec |
| **On-demand (cold) Failover** | Codec detects loss of IP data or connection and redials the backup connection | User configurable detection parameters during program configuration. Fail over delay is equal to detection time and audio threshold specified, plus the time required to dial the alternative connection. | Dialing codec program monitors streaming and manages failover to a backup connection |
| **FEC (Tieline FEC or RFC 2733 compliant FEC)** | Decoding codec detects IP packet loss or delayed packets. Note: Only Tieline Music and Music PLUS can be used for Tieline FEC. | No failover time delay as packet replacement occurs in real-time. Note: RFC compliant FEC can be configured to send 100% FEC at a specified delay if desired. | • Dialing codec using Tieline session data configures local and remote FEC settings via session data transfer when connecting, or<br>• In sessionless mode FEC configuration is as per RFC compliant settings configured in the local and remote codecs. |
| **Auto Reconnect** | Codec will redial continuously to try and reconnect. | Immediately redials after loss of IP stream detected. | Enabled in dialing codec program. |

### SmartStream PLUS Redundant IP Streaming

Tieline's proprietary SmartStream PLUS IP technology ensures you're always on the air. There are three levels to SmartStream PLUS IP streaming.

1. The codec can stream simultaneous redundant data streams and deliver seamless redundancy by switching back and forth, without loss of audio, from the nominated primary data link to the backup link if one fails and then subsequently recovers. Use IP links from two different IP network providers for optimal redundancy over mission critical connections.

2. Second, when multiple redundant audio streams are sent, the decoding codec automatically reconstructs audio into a single stream on a first packet arrived basis, to minimize program latency and ensure audio integrity.
3. Third, SmartStream features automated jitter buffer management and Forward Error Correction (FEC) and these advanced network management tools deliver uncompromising audio quality, while dynamically responding to variable conditions over unmanaged IP networks like the internet.

These combined measures ensure Tieline is capable of offering a rock solid IP audio solution for distributing IP audio economically and efficiently across broadcast networks. See the procedures for configuring different programs using the web-GUI for more configuration details.

## Fuse-IP

Tieline's proprietary Fuse-IP data aggregation technology uses a point-to-point tunnel between two codecs to bond multiple IP interfaces (peers). See Configure Fuse-IP Bonding for more info.

## On-Demand Failover

On-demand failover requires configuration of a primary connection and an on-demand 'cold' backup connection. On-demand failover is activated when the dialing codec program detects the loss of the primary connection, or if audio streaming ceases. The backup connection is then dialed to replace the primary connection.

For details on configuring backup connections using failover see Configure Mono or Stereo Peer-to-Peer Programs.

## Forward Error Correction (FEC)

There are two modes of Forward Error Correction (FEC) available in the codec:

1. Tieline FEC.
2. RFC 2733 compliant FEC (Sessionless connections only).

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. Tieline FEC transmits a secondary stream of audio data packets over a single connection. RFC 2733 compliant FEC transmits audio packets over a separate connection.

For more info on FEC see Configuring Forward Error Correction.

## Auto Reconnect

Auto Reconnect is the simplest form of connection backup whereby the codec will redial a lost connection continuously until it is either:

- Re-established, or
- Dialing is manually stopped.

Auto reconnect can be enabled when configuring a codec program.

## SDHC File Backup

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

1. According to the silence threshold parameters configured for audio file backup, or
2. Immediately if a connection to another codec is lost.

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

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

## 13.16 SDHC Card File Playback

### Playing Audio from the SDHC Card

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

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

**Important Notes for SDHC Card File Playback:**
- A single partition FAT32 formatted SDHC Card is required (SD cards may be less reliable and are not recommended).
- The codec supports SDHC cards which have a physical capacity of up to 32GB.
- Create MP2 or MP3 files using a 32kHz, 44.1kHz or 48kHz sample rate.
- Ensure recordings used are not variable bit rate files.
- SDHC file audio is not sent to codec encoders and cannot be transmitted via an audio stream to another codec.
- File playback audio is sent directly to the codec outputs and therefore IGC is not available. When you create your MP2 or MP3 files ensure the audio levels match the audio reference level of your codec and that peaks average at the correct levels.
- If you create a single file name ensure you add the file extension, e.g. "test.mp3", or the file will not play back.
- If you create a directory name, all the files within the directory will be played back. We recommend you save all audio files as a playlist and link to this if you want them to play out sequentially. Please note that "M3U" is the playlist file format supported by the codec.

## 13.17 Lock or Unlock a Program in the Codec

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

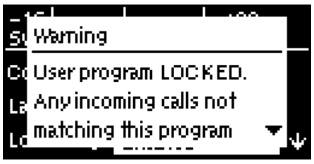
It is also possible to lock a loaded custom program in a codec to ensure a program with your preferred settings is not unloaded when a codec dials in. Incoming calls are generally down or up sampled to accommodate a locked program where possible. Scenarios in which you may wish to lock a program in the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

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

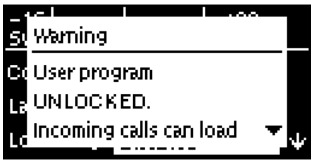
Incoming calls to an answering codec with a locked program can still specify different connection parameters such as algorithm preferences and bit rates via session data.

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


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




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




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

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

# 13.18 Locking the Front Panel

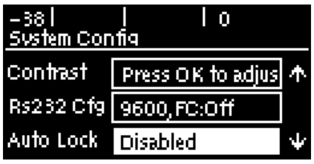
The codec features a front panel lock feature for tamper-proof operation. This feature is disabled by default.

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

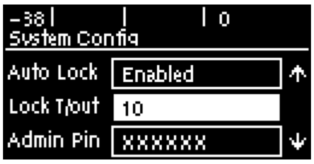
1. **Admin PIN**: Required to change codec connection or configuration settings accessed via the **SETTINGS** button. (Default PIN is: 456789)
2. **User PIN**: Required to use the codec front panel buttons and dial/hangup a connection (Default PIN is: 123456)

## Enabling the Front Panel Lock Feature

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




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




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

# 14 Connecting to the ToolBox Web-GUI

There are two graphical user interface (GUI) options for configuring and connecting Tieline G5 codecs:

1. HTML5 Toolbox Web-GUI: fully configure codec settings, create dialing programs and dial, hangup and monitor connections.
2. HTML5 Toolbox Quick Connect Web-GUI: designed for dialing and managing simple peer-to-peer connections; ideal for non-technical users.

### About the HTML5 Toolbox Web-GUI

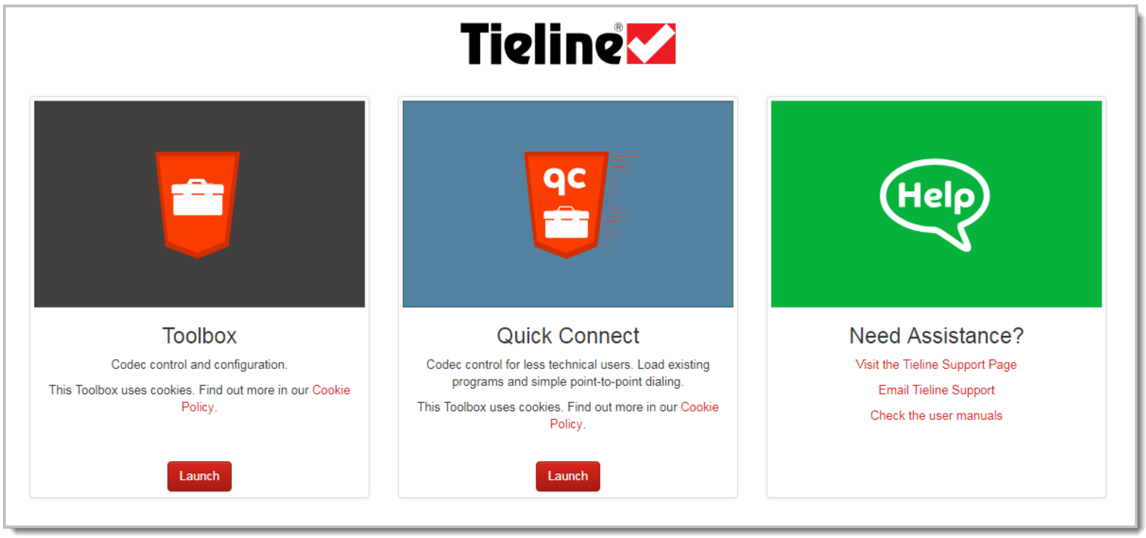
The HTML5 Toolbox Web-GUI improves the user experience with G5 codec command and control and runs seamlessly on modern browsers. It will run on Mac, Windows and Linux computers.

### About the HTML5 Toolbox Quick Connect

The HTML5 Toolbox Quick Connect Web-GUI has a reduced feature-set and allows non-technical users to load existing programs and dial via the **Quick Connect panel**. Users can dial a simple peer-to-peer connection over POTS, ISDN or IP. To enable the **Quick Connect panel** press the **SETTINGS** button, then navigate to **WebGUI > Quick Connect > Enabled**.

## 14.1 Opening the HTML5 Web-GUI & Login

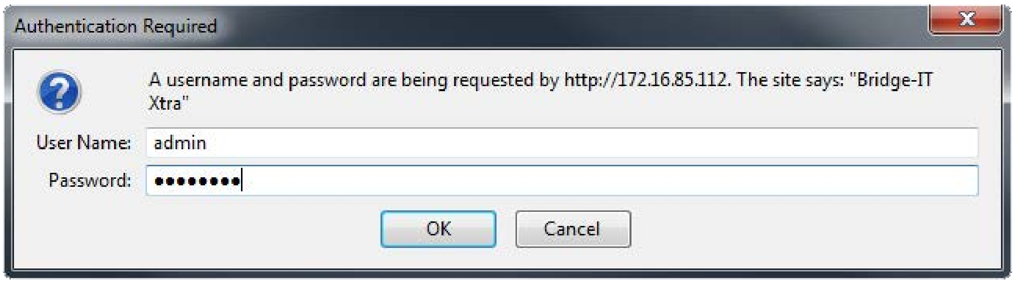
1. Attach an Ethernet cable to the **LAN** port on the codec.
2. Press the **SETTINGS** button and select **Unit** to display the IP address programmed into your codec.
3. Ensure your PC is connected to the same LAN.
4. Open your web browser and type the IP address of your codec into the address bar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




### Launching the HTML5 Toolbox Web-GUI

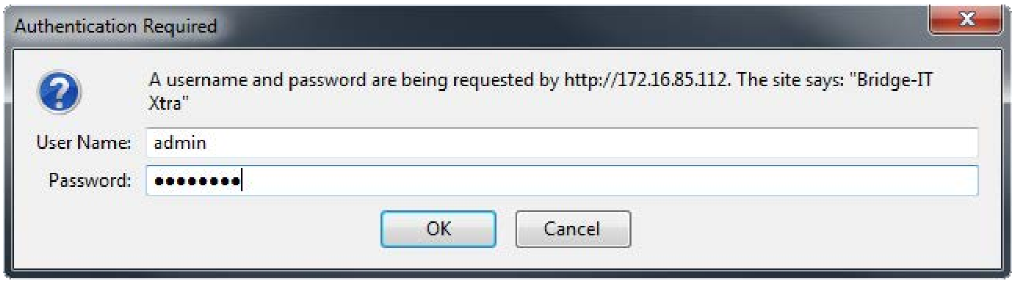
1. Click to launch the **HTML5 Toolbox Web-GUI**.

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




## Launching the HTML5 Toolbox Quick Connect

1. Click to launch the **HTML5 Toolbox Quick Connect** Web-GUI.
2. When you launch Toolbox for the first time an authentication dialog prompts you to enter the user name "**admin**" and password "**password**" to login, then click the **OK** button. Tieline **highly recommends** changing the password (see Changing the Default Password). This will provide better network security to maintain reliability during live broadcasts.




## Using the Web-GUI over the Internet

If your codec is connected over the internet via a public static IP address it is possible to connect and configure it from any PC which is also connected to the internet. If you have multiple browsers open on a PC for different codecs it is possible to customize the browser title for simple identification. To configure this:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **WebGUI** and press the button.
3. Navigate to **Browser** to enter the custom title.
4. Use the keypad to enter the title, then press the button to save the setting.

To configure this setting using the HTML5 Toolbox Web-GUI, click **Settings** at the top of the screen, then click **Options** to display the **Options panel**. Enter the **Browser Title** and then click **Save**.

## LAN Troubleshooting

### PC LAN Settings

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

1. Open Internet Explorer.
2. Click **Tools > Internet Options > Connections**.

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

## Port Selection

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

E.g. **192.168.0.176:8080**

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

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

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

## 14.2 Security and Changing the Default Password

Codecs connected to the internet can be accessed by anyone with knowledge of the codec's public IP address. In addition, search engines are widely available which can discover and expose unsecured 'internet connected devices'. Tieline recommends the following IP codec security precautions are followed as a bare minimum, to ensure your codec connections remain secure.

### Maintaining Codec Network Security

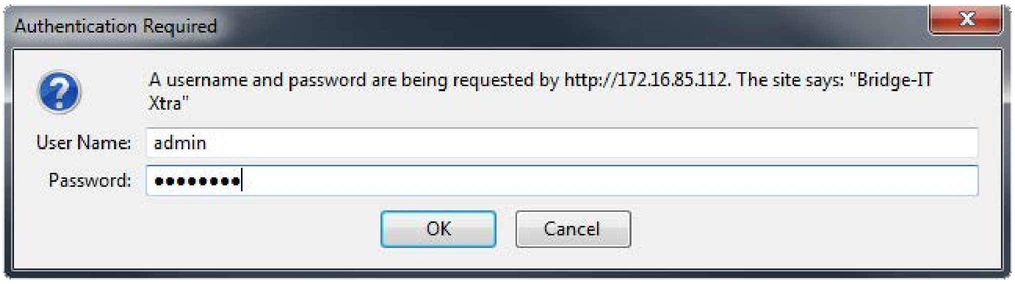
Adequate security is a major factor in ensuring your codecs and your broadcast network remain secure. There are several layers of security available in Tieline codecs to maintain secure connections. These include:

1. Immediately change the default password when you commission and install your codecs (see instructions which follow). Create a strong password which includes both capital and lower case letters, symbols and numbers (up to 15 characters can be entered). Password managers can be useful when managing multiple passwords within organizations.
2. Ensure your codec is behind a firewall and only open the TCP and UDP ports required to transmit session and audio data between your codecs. Using non-standard ports instead of Tieline default ports can also ensure the codec is more difficult to discover by nefarious characters.
3. Ports 80 and 8080 are commonly used to access the Tieline codec web server. You can add an additional layer of security by translating these ports on the WAN side of your network into non-standard port numbers. Adjust ports using the **Options panel** in the Toolbox HTML5 Web-GUI, or see Configuring TCP/UDP Ports.
4. By default SIP interfaces are disabled to avoid unwanted traffic. The **SIP Filter Lists panel** in the Toolbox HTML5 Web-GUI allows filtering of SIP URIs and User Agents to provide greater security when using SIP. See Configure SIP White and Blacklists for more information.
5. An SSL security certificate can be installed on each codec in your network to ensure it is a trusted device within your network. See Installing a Security Certificate for more information.
6. Firewall settings to enable or disable a range of firewall-related network services, or limit ping to only work in a local subnet. Adjust settings using the **Options panel** in the Toolbox HTML5 Web-GUI, or see Firewall Configuration.
7. Implementation of CSRF protection (Cross-Site Request Forgery). Enable and disable this setting using the **Options panel** in the Toolbox HTML5 Web-GU, or see Enabling CSRF Security for more info.

Be sure to document any port changes because this information will be required if you need to contact Tieline or other online support services.

### Changing the Default Password

The default password for the Toolbox Web-GUI is **password**. Enter this in the authentication dialog to use the Web-GUI initially and then Tieline highly recommends changing the default password to protect your codec from being tampered with during live broadcasts. Note: In the HTML5 Web-GUI authentication dialog it is necessary to enter **admin** as the **User Name**.




**Toolbox HTML5 Web-GUI Login Dialog on a Merlin Codec**

## Creating a New Password

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

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

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

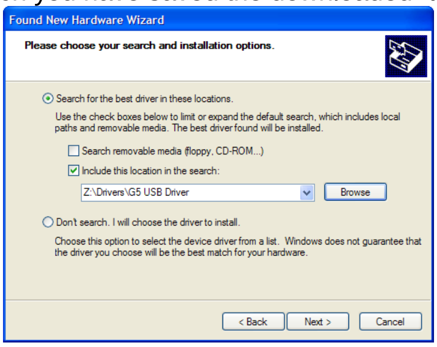
**Important Note:** The **Username** in the codec menu is permanently set to **admin** and cannot be changed; only the **Password** can be changed.

## 14.3 Installing USB Drivers

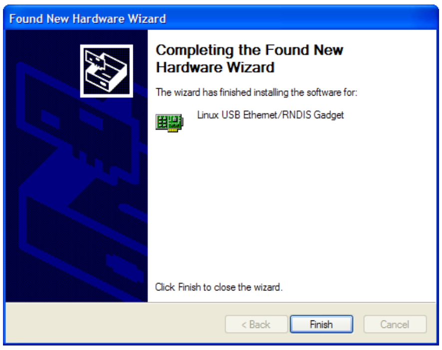
USB drivers need to be installed on your PC in order to connect successfully to the codec using the USB port. To install drivers:

1. Download the zipped USB driver ".inf" file from the Bridge-IT firmware download pages at http://www.tieline.com/Support/Latest-Firmware.
2. Unzip the file and save it to your PC.
3. Connect a USB cable between your PC and the Bridge-IT **USB** port on the rear panel of the codec.
4. Older PCs should detect that a new device has been attached and launch the **Found New Hardware Wizard**. Note: If the wizard doesn't launch automatically, please navigate to **Control Panel > System** in Windows to verify the status of the driver installation. Install the Microsoft RNDIS driver for the codec if required.
5. Select **Install from a list or specific location (Advanced)** and click **Next**.


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

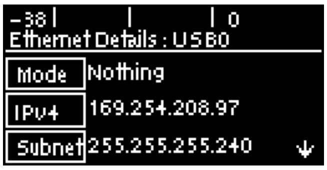



7. When the drivers have been installed click the **Finish** button.

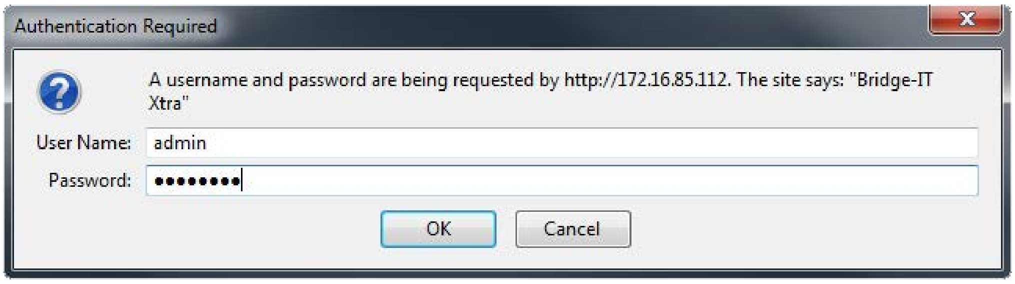


## 14.4 Launching the GUI over USB

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




4. Open your web browser and type the USB address of your codec into the address bar of your browser, i.e. **http://169.254.x.y** (the last 2 blocks of digits are the USB address details unique to your codec).
5. Refresh the browser and the web-GUI should launch automatically. Note: if it doesn't launch automatically please see Installing USB Drivers for more info.
6. When you launch Toolbox for the first time an authentication dialog prompts you to enter the user name "**admin**" and password "**password**" to login, then click the **OK** button. Tieline **highly recommends** changing the password (see Security and Changing the Default Password). This will provide better network security to maintain reliability during live broadcasts.




# 15 HTML5 Toolbox Web-GUI Introduction

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

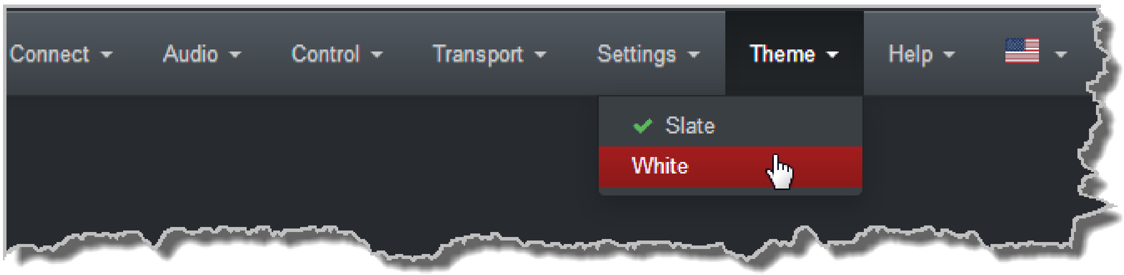

**HTML GUI Menu Bar for Opening Panels**

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

The green **Online** indication in the top left-hand corner of the Toolbox Web-GUI indicates it is online and can be used for codec control. A red **Offline** indication is displayed when the codec is unavailable. The **Upgrade** symbol is displayed when a new firmware version is available for the codec. Open the **Firmware panel** in the **Settings** menu to upgrade the codec with new firmware.

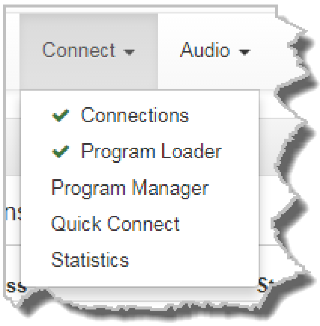
## Adjusting the Theme

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




## Opening a Panel & Adjusting Screen Position

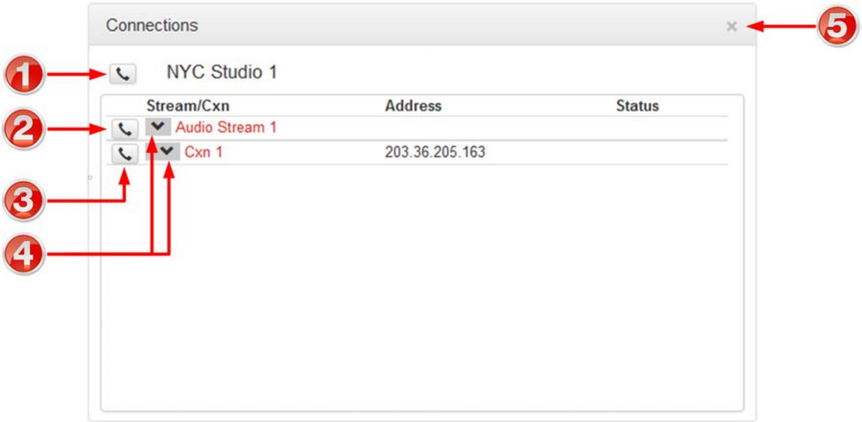
Click an item in the **Menu bar** to display available panel options, then click to select and open a panel. New panels automatically open in the top left of the screen. A green **Tick** adjacent to a panel name in the menu signifies it is already open in the web-GUI.




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

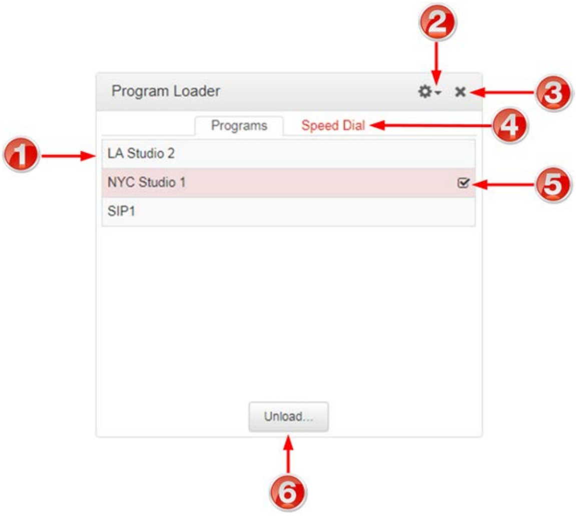
# Connect Panels: Load & Connect Programs & Manage Audio Streams

## Connections Panel



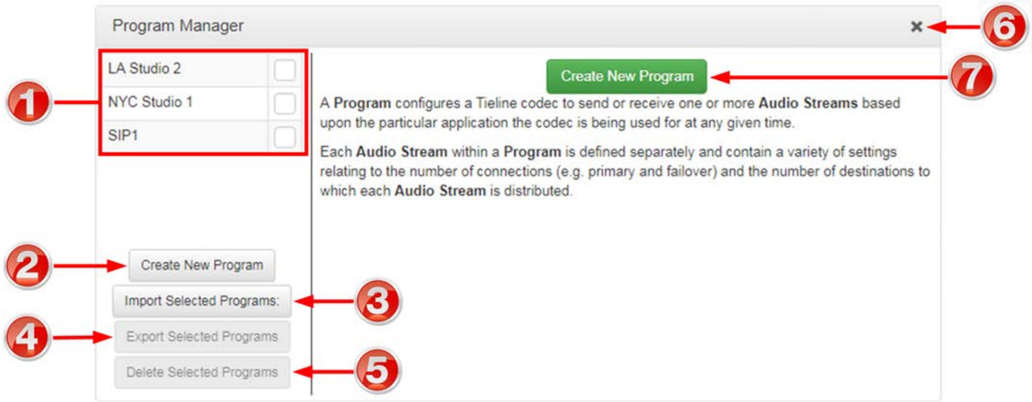

| | Feature | Description |
|---|---|---|
| 1 | Program **Connect/Disconnect** button | Click to connect/disconnect all audio streams in a program. |
| 2 | Audio Stream **Connect/Disconnect** button | Click to connect/disconnect all connections in an audio stream. |
| 3 | Connection **Connect/Disconnect**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

## Program Loader Panel



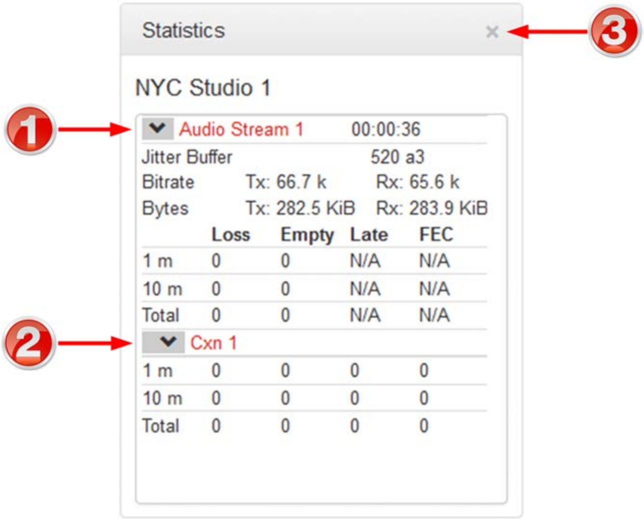

| | Feature | Description |
|---|---|---|
| **1** | Programs list | Lists all configured programs which have been added into the codec. Click to select a program before loading. |
| **2** | Listing options | Alphabetical listing options |
| **3** | **Close** button | Click to close the panel. |
| **4** | **Speed Dial** | Select and dial programs using preconfigured speed dials. |
| **5** | **Check-box symbol** | The **Check-box symbol** identifies the currently loaded program in the codec. |
| **6** | **Load** and **Unload** button | Click **Load** to load the currently selected program in the **Programs** list; the button changes automatically to **Unload** after loading, to allow unloading of a program when required. |

## Program Manager Panel




| | Feature | Description |
|---|---|---|
| 1 | Program list | The list of saved programs in the codec. |
| 2 | **Create New Program** button | Click to create a new program using the program wizard. |
| 3 | **Import Selected Programs** button | Click to select and import previously saved programs. |
| 4 | **Export Selected Programs** button | Click to export selected programs as a .zip file. |
| 5 | **Delete Selected Program** button | Click to delete all selected programs |
| 6 | **Close** button | Click to close the panel. |
| 7 | **Create New Program** | Click to create a new program using the Program Manager wizard. |

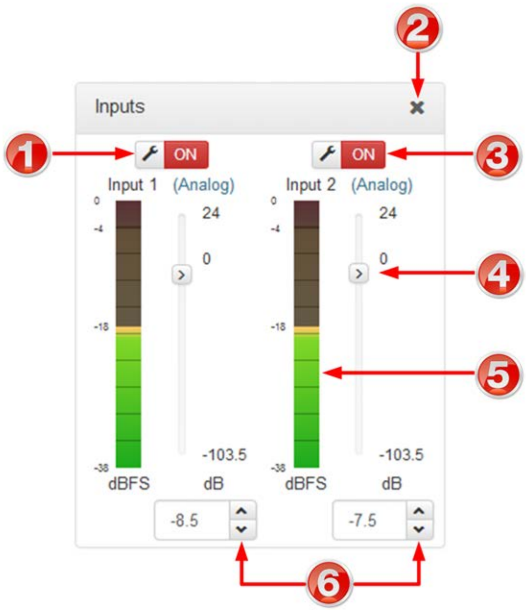
## Statistics Panel




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

# Audio Menu Panels

## Inputs Panel



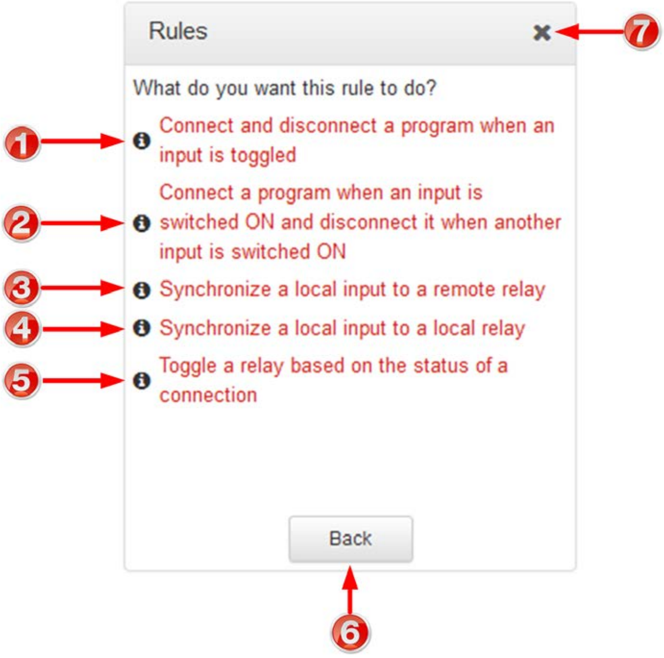

| | Feature | Description |
|---|---|---|
| 1 | **Settings** button | Click to adjust input **Type** and **Gain, IGC** settings and input **Name**. Note: input **Type** adjustment only available for input 1. |
| 2 | **Close** button | Click to close the panel. |
| 3 | **On/Off** button | Click to toggle an input on or off. |
| 4 | Input Sliders/Faders | Input gain control sliders/faders. |
| 5 | PPM meter | Input PPM meter. |
| 6 | Input gain adjustment | Click to adjust input gain in 0.5dB increments |

## PPMs Panel

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

## Control Menu

### Rules Panel

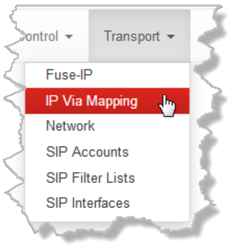



| | Rule | Description |
|---|---|---|
| 1 | Connect and disconnect a program when an input is toggled | Click to configure connection and disconnection by toggling an input. |
| 2 | Connect a program when an input is switched **ON**; Disconnect when another input is switched **ON** | Click to configure connection and disconnection after different relay inputs are switched **ON**. |
| 3 | Synchronize a local relay input with a remote relay output | Click to configure a local relay input to synchronize with the state of a remote relay output. |
| 4 | Synchronize a local relay input with a local relay output | Click to configure a local relay input to synchronize with the state of a local relay output. |
| 5 | Toggle a relay based on a connection's status | Click to configure a relay to toggle based on connection status. |
| 6 | **Back** / **Add New Rule** button | Click to add a new rule, or exit the rule creation function. |
| 7 | **Close** button | Click to close the panel. |

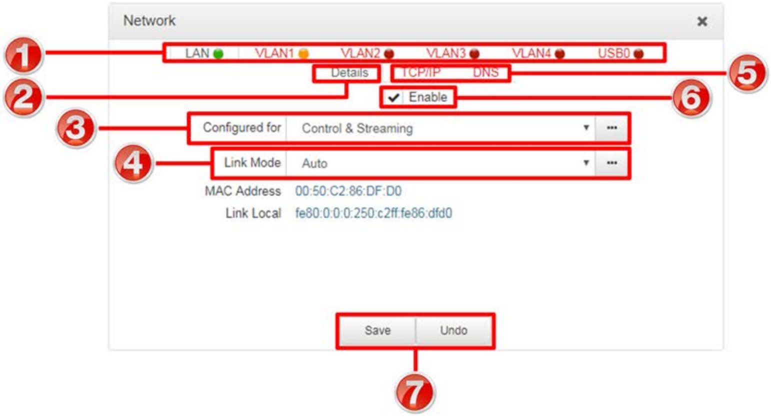
## Transport Panels

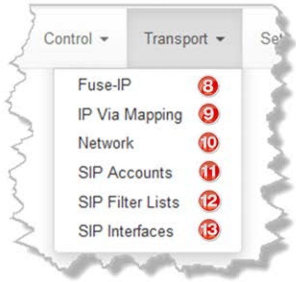
There are several **Transport** panels which can be opened in the Web-GUI. Each panel provides specific transport-related configuration settings and options. Click to select and open each panel.

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




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

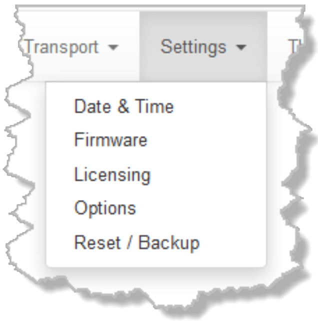



| | Feature | Description |
|---|---|---|
| 1 | Network | Click to select and edit, or view network configuration settings for the LAN, VLAN and USB interfaces. |
| 2 | Details tab | Display configuration options for a selected network interface, plus other device details. |
| 3 | Control/Streaming | Select **Control** and/or **Streaming** options for the selected interface. |
| 4 | **Link Mode** | Configure the Ethernet or VLAN link speed (10/100/1000/Auto) and whether an interface will operate in Full-Duplex or Half-Duplex mode. |
| 5 | **TCP/IP** and **DNS** tabs | Select the **TCP/IP** tab to configure **IPv4/IPv6** address details. Select the **DNS** tab to specify DNS addresses and domains to search. |
| 6 | **Enable check-box** | Select the check-box to enable an interface. |
| 7 | **Save/Undo** button | Click **Save** to store settings, or click **Undo** to revert to previously configured settings. |
| 8 | **Fuse-IP** | Click to open the **Fuse-IP panel** and configure Fuse-IP bonding |
| 9 | **IP Via Mapping** | Configure default **Primary**, **Secondary** and **Tertiary** interfaces. |
| 10 | **Network** | Click to open the **Network panel** and configure network settings. |
| 11 | **SIP Accounts** | Click to open the panel and edit SIP account settings. Up to 6 SIP accounts are supported. |
| 12 | **SIP Filter Lists** | Add trusted network codecs to the **URI Whitelist**. Add SIP URIs to the **URI Blacklist** and add user agents to the **User Agent Blacklist** to deny them access to the codec. |
| 13 | **SIP Interfaces** | Click to open the panel and configure port, proxy and **Via** settings for the **SIP1** and **SIP2** interfaces. The codec supports dialing over these SIP interfaces simultaneously. |

## Settings Panels

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

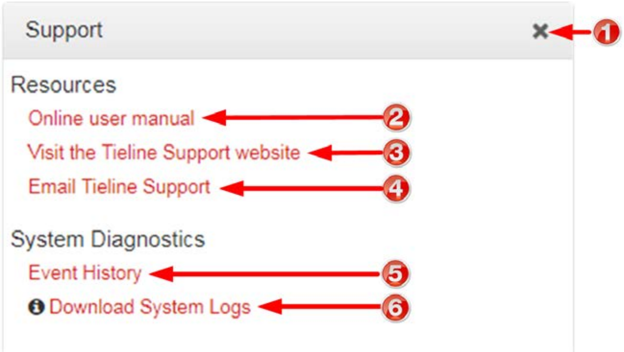



**Settings panels**


| | Feature | Description |
|---|---|---|
| 1 | **Options tab** | Click to open the panel; configure RS232 and QoS data settings, lock a loaded user Program and adjust Session Port settings, SNMP, the analog input PPM unit preference, reference level, Headphone Output Mode, Firewall, Hostname, AES Output Clock sample rate and enable/disable the Quick Connect HTML Web-GUI option. |
| 2 | **Licensing tab** | Click to open the panel; select a license file and install it in the codec. |
| 3 | **Firmware tab** | Click to open the panel; view software versions, download firmware and perform an upgrade. |
| 4 | **Reset / Backup** | Click to open the panel; reset codec default settings and perform backup/restore of codec programs and settings. |
| 5 | **Date and Time** | Click to open the panel view and sync the codec to NTP time. |

# Help Panels

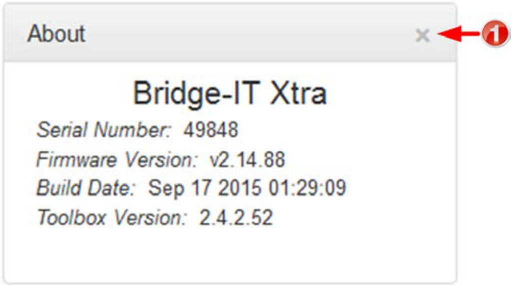
## Support Panel




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

## About Panel

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




| | Feature | Description |
|---|---|---|
| 1 | **Close** button | Click to close the panel. |

## Language Selection

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

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

# 16 HTML5 Toolbox Web-GUI Configuration

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

## 16.1 Using the HTML5 Toolbox Quick Connect Web-GUI

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

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

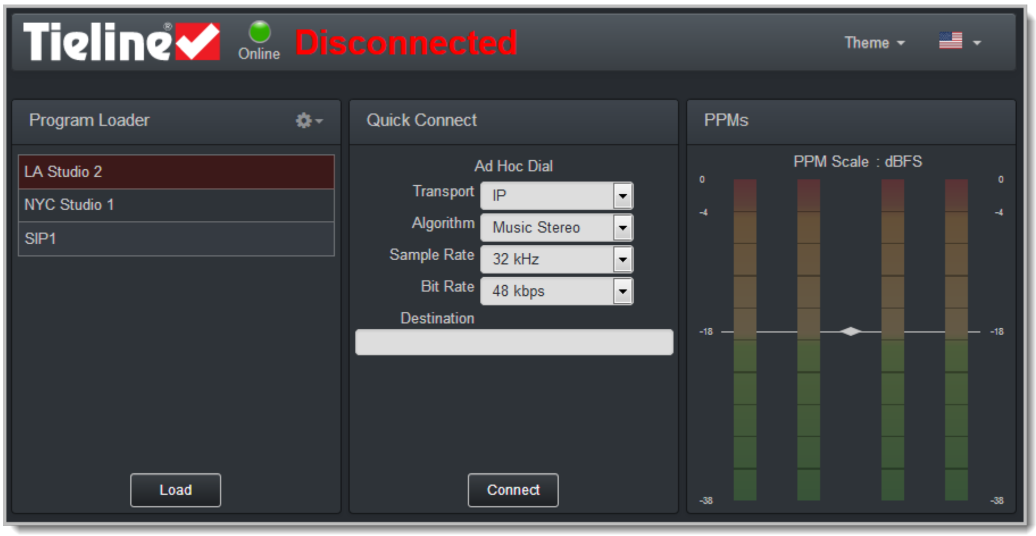
See Opening the HTML5 Web-GUI and Login for details about launching the standalone HTML5 Quick Connect Web-GUI. The **Quick Connect panel** can also be launched from the **Connect** menu in the HTML5 Toolbox Web-GUI.

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

To enable the **Quick Connect panel** press the **SETTINGS** button, then navigate to **WebGUI > Quick Connect (Q Cnct) > Enabled**.

### Launching the HTML5 Quick Connect Web-GUI

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

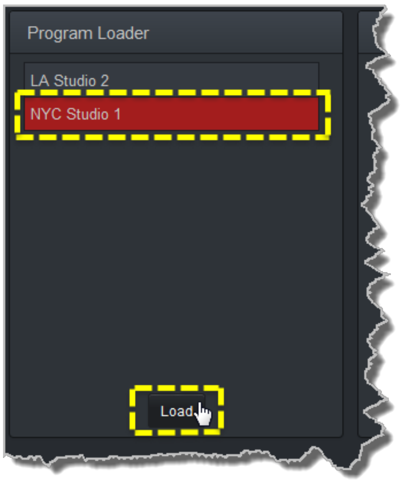



**Important Note:** To change the password using the codec front panel navigate to **Settings > WebGUI > Password** and press the button. Use the keypad to enter a new password and press the button to save the new setting.

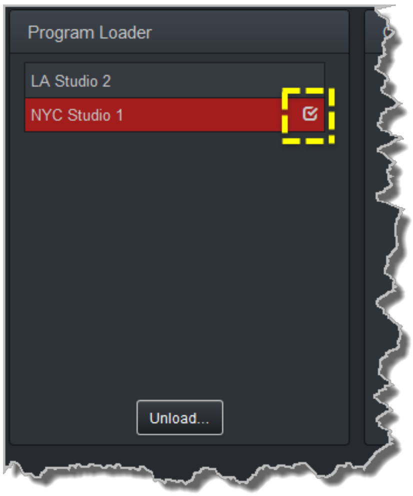
## Loading and Unloading an Existing Program

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

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



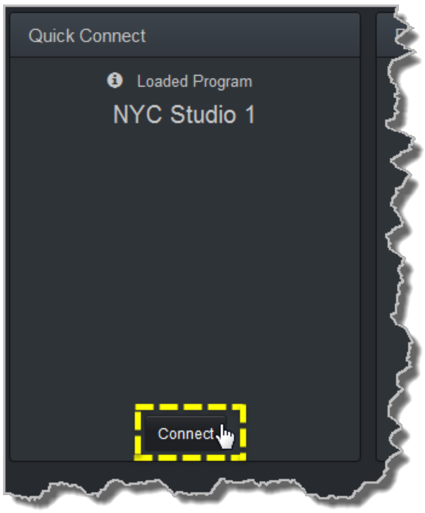

2. The **Check-box symbol** appears next to the program name to confirm it has been loaded and the **Load button** changes to an **Unload button**.




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

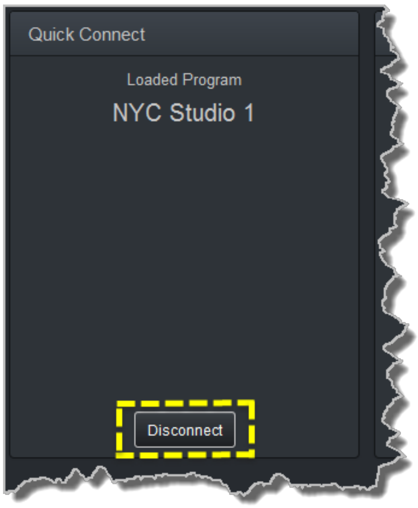
## Dial a Loaded Program

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

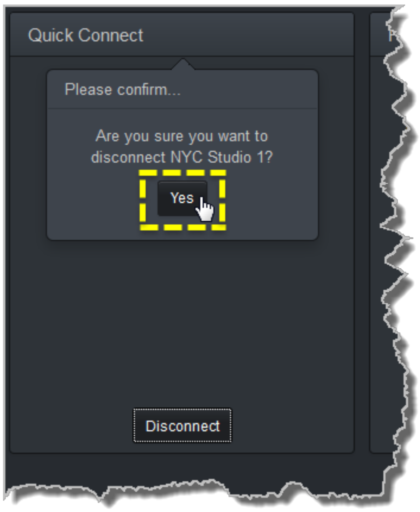



## Disconnect a Loaded Program

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




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

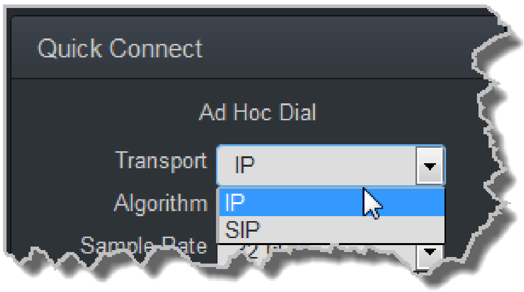


## Dial Peer-to-Peer over IP with Quick Connect

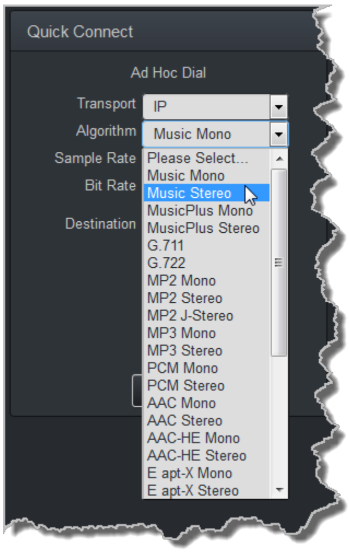
**Important Notes:**

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

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

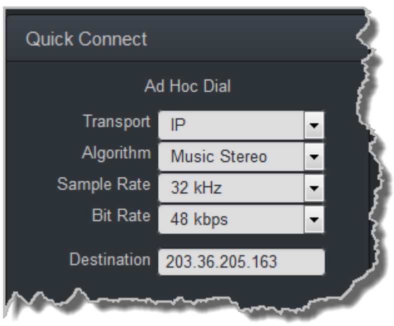



2. Click the drop-down **Algorithm** menu and select an algorithm.




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

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



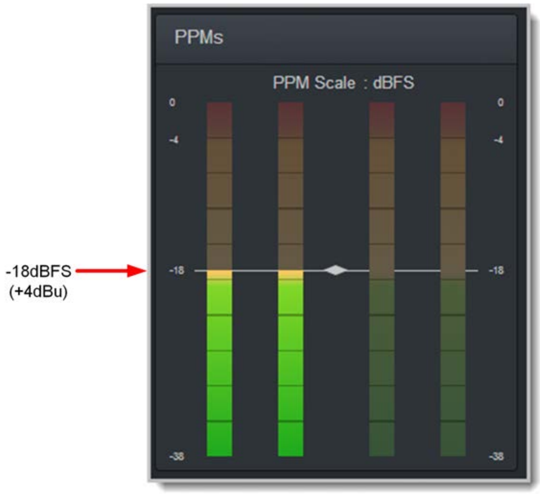

5. Click the **Connect button** to dial.

## Monitoring PPMs

Set audio levels so that audio peaks average at the nominal 0vu point indicated below on the PPM meters. By default, the **PPM METERS** on the front of the codec, or in the HTML5 Toolbox Web-GUI, use dBFS to express nominal operating, headroom and noise floor levels.

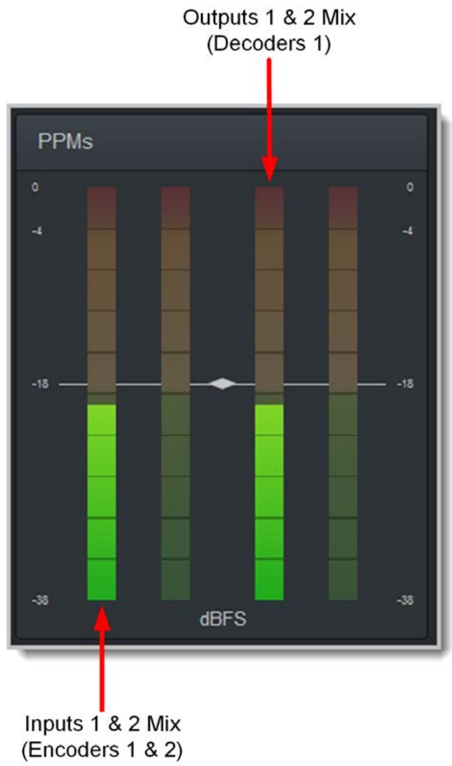
The codec can also automatically adapt to different Tieline reference scales. A Tieline codec with proprietary Tieline session data enabled will automatically adjust the reference level to suit G5 and G3 codecs, or Report-IT. When connecting to a non-Tieline codec, or a Tieline codec without session data enabled, the codec will use the **Tieline G5** reference scale setting.

The default Tieline G5 audio reference scale displayed on the PPMs is -38dBFS to 0dBFS (e.g. Merlin, Genie, Bridge-IT and ViA codec families). Using this reference scale audio peaks can safely reach 0dBFS without clipping, providing 18dB of headroom from the nominal 0vu point.

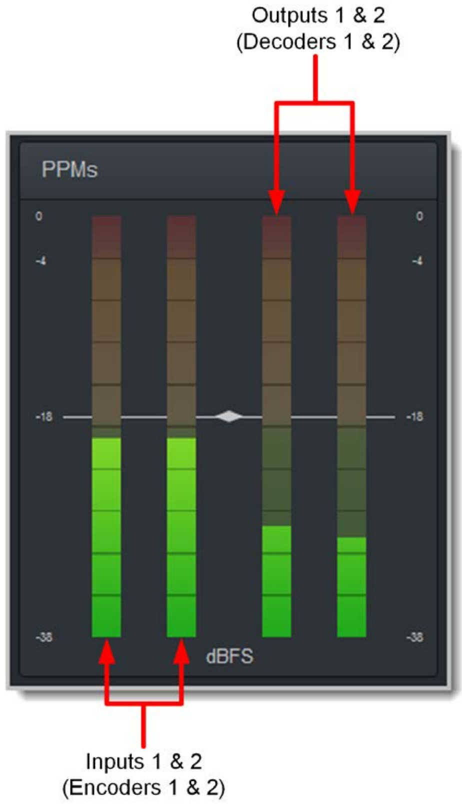



### Mono and Stereo PPM Metering

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




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



## 16.2 Configuring IP Settings

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

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

### IPv4 versus IPv6

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

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

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

### Configuring Ethernet Ports and VLANs

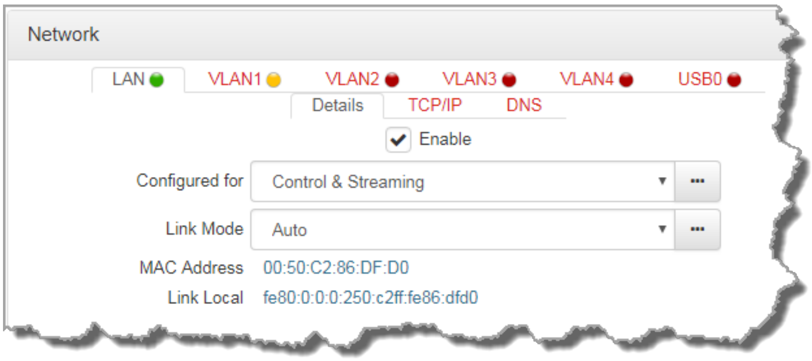
The codec features a physical Ethernet port and up to four additional VLAN interfaces.

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

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

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

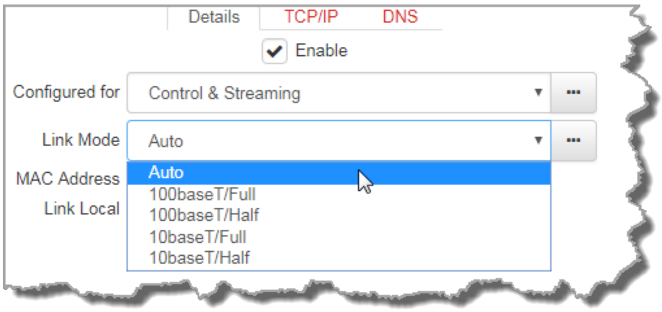
Select the **Details tab** in the **Network** panel to edit control and streaming settings. Select the **Enable** check-box to activate each interface. Note: **LAN** is enabled by default. An interface with a green status indication is enabled and available. An interface with red text and a yellow status indication is enabled but unavailable for some reason, e.g. VLAN networking is not configured correctly. An interface in red text with a red status indication is disabled.




## Configure Link Mode

It is possible to configure the Ethernet link speed (10/100/Auto) and whether each available interface operates in full-duplex or half-duplex modes.

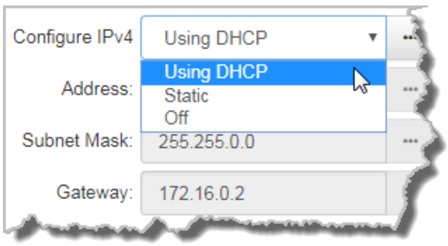
1. Click the drop-down **Link Mode** arrow and select the preferred setting.



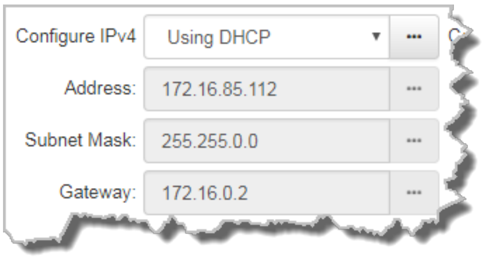

2. Click **Save** to store the new setting.

## IPv4 Address Configuration

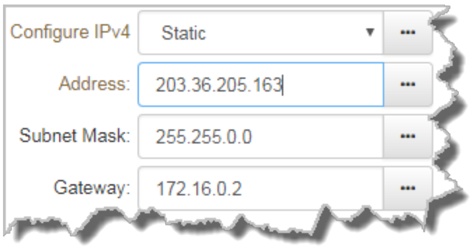
Click to select the **TCP/IP** tab in the **Network** panel to configure settings. The codec is capable of automatic DHCP address assignment, or manually configured static IPv4 address configuration via the drop-down **Configure IPv4** menu. If you want to ignore IPv4 settings select **Off**.




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




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




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

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

## IPv6 Address Configuration

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

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

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

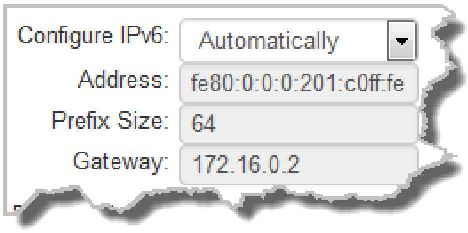
### Types of IPv6 Addresses

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

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

### Auto Address Assignment

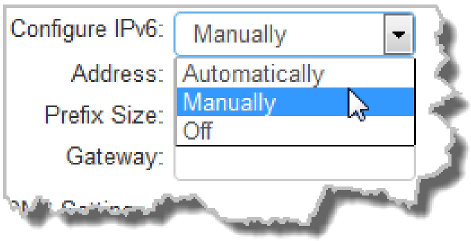
1. Click the **Edit** button in the **Network** panel to configure settings.
2. By default the codec is configured to connect to an IPv6 router which automatically allocates IPv6 address details, as displayed in the following example.




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

### Manual IPv6 Address Assignment

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

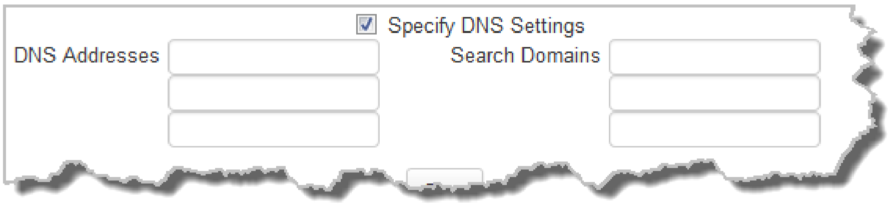



3. Click **Save** to store all configuration settings.

## Specifying DNS Settings

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

1. Select the **DNS** tab in the **Network panel** to configure settings.




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

## IP Via Mapping

When dialing over IP you can select the preferred interface to use when establishing a connection. By default **Any** is selected, which means the first available interface will be used to dial a connection. The default **Via** interfaces in order of use when available are:

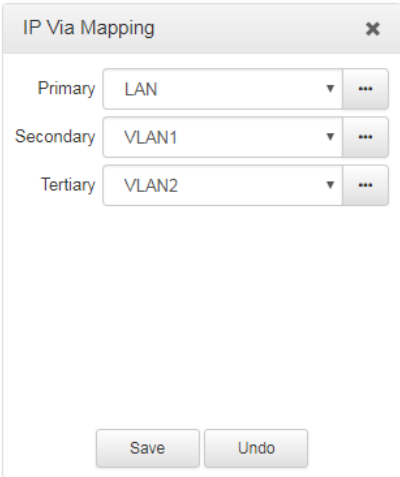
1. **LAN1** Ethernet port (default **Primary** Via interface)
2. **VLAN1** Ethernet port (default **Secondary** Via interface)
3. **VLAN2** (default **Tertiary** Via interface)

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

### Reconfigure Default Primary, Secondary and Tertiary Interfaces

It is possible to reconfigure the default **Primary** (LAN), **Secondary** (VLAN1) and **Tertiary** (VLAN2) interfaces in the codec. As an example, you may want to select **Primary** as the dialing interface in a program and then copy this program onto multiple codecs. However, the actual primary interface used at each location can vary for each codec. For one codec it may be an Ethernet port and for another it may be a VLAN interface. This allows you to configure site-specific settings to suit available network interfaces at different remote locations.

1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then click **IP Via Mapping** to open this panel.
2. Click the drop-down arrow for each interface to select the preferred default setting.



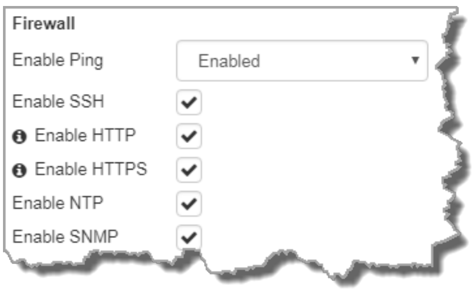

3. Click **Save** to store the configuration.

**Important Note:** Fuse-IP cannot be configured as a default Primary, Secondary or Tertiary **Via**.

## Configure Firewall Settings

The **Firewall** menu can be used to enable or disable a range of firewall-related network services, or limit ping to only work in a local subnet.

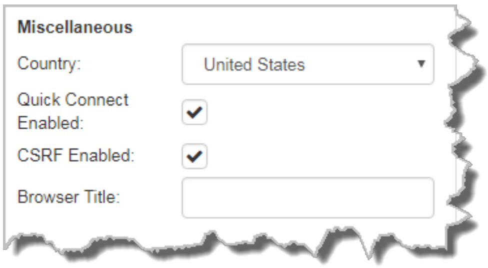
1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Options** to display the **Options panel**.
2. Adjust **Firewall** settings and click **Save** to store the new configuration settings.




## Configure Cross-Site Request Forgery

CSRF (Cross-Site Request Forgery) protection can be configured to protect the codec from CSRF attacks. To enable or disable this setting:

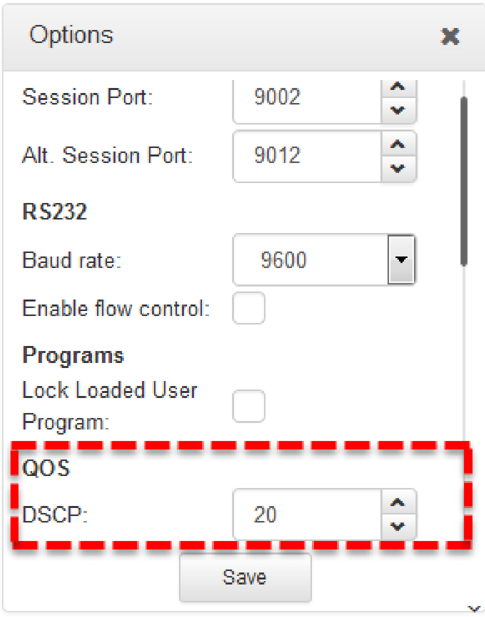
1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Options** to display the **Options panel**.
2. Select the **CSRF Enabled** check-box to enable this feature.




## Configuring QoS

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

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




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

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

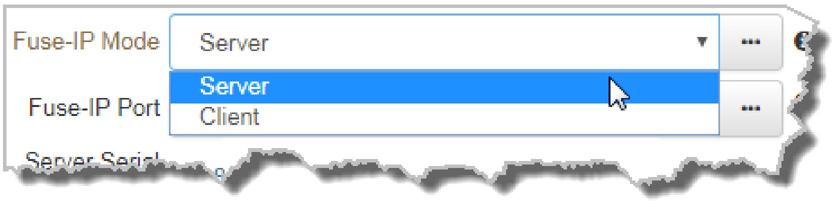
# 16.3 Configure Fuse-IP Bonding

Tieline's proprietary Fuse-IP data aggregation technology uses a point-to-point tunnel between two codecs to bond multiple IP interfaces (peers). Fuse-IP automatically distributes data over any two bonded interfaces. Bridge-IT only has a single LAN interface, however it can still connect as a Fuse-IP server or client. This means Bridge-IT can stream data between itself and another codec which is bonding over multiple IP interfaces.

For more details on setting up Bridge-IT for Fuse-IP using the codec front panel see Configuring Fuse-IP Bonding.

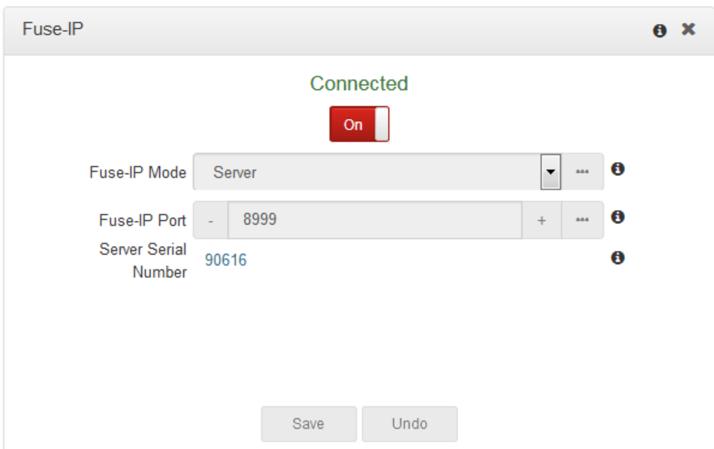
## Configuring a Fuse-IP Server at the Studio

1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then **Fuse-IP**.
2. Click the **Fuse-IP Mode** drop-down menu and select **Server** if the codec is not initiating the connection.

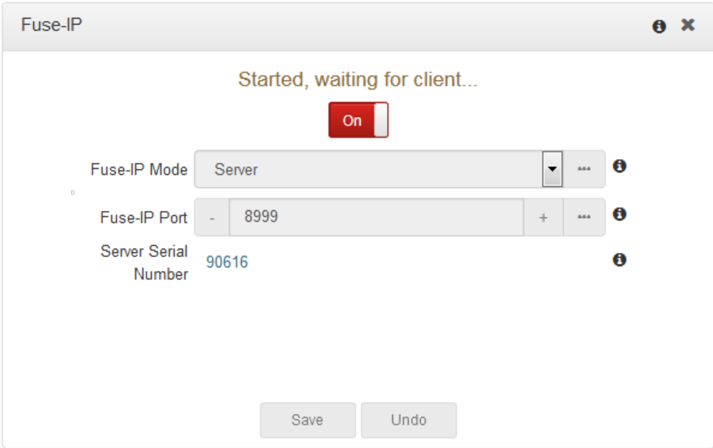



3. Leave the default **Fuse-IP Port** as **8999** in most situations unless this port is already in use, e.g. you have multiple codecs behind a firewall using Fuse-IP, therefore you need to allocate a different port for each Fuse-IP tunnel. Note: the port number on the client and server codecs must be the same.


4. Click the **Save** button to store all settings.
5. Click the **On/Off** button to create a Fuse-IP tunnel between the server and client codecs. Note: **Connected** should be displayed after a few seconds if both codecs are configured correctly and Fuse-IP is enabled on both codecs.

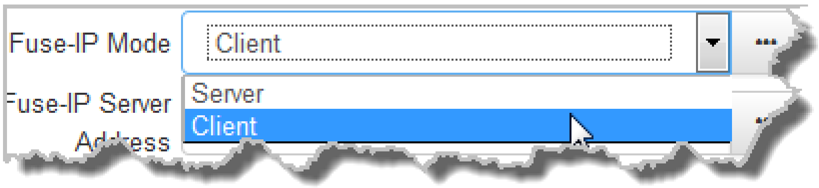



6. Double-check all settings on both the server and client codecs if the message **Started, dialing server...** persists after turning Fuse-IP **On**.

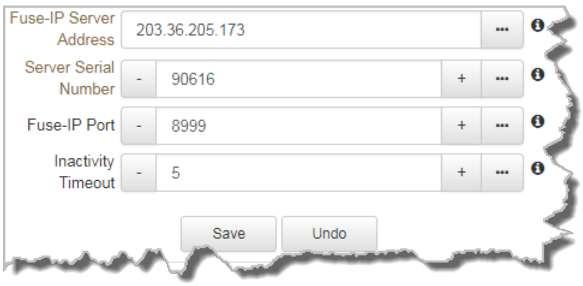



## Configuring a Fuse-IP Remote Client

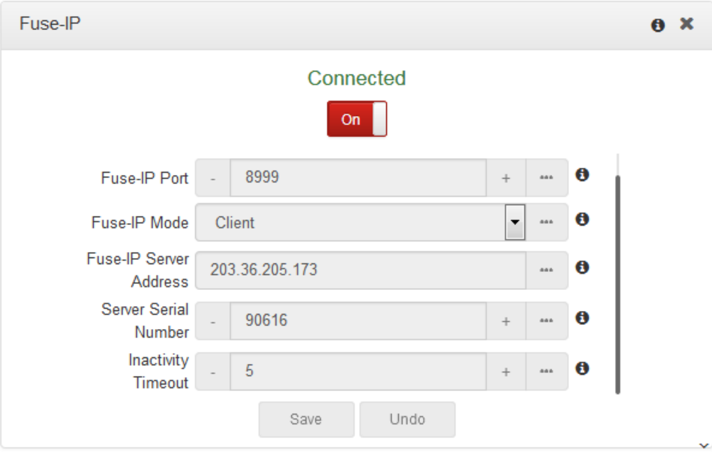
1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then **Fuse-IP**.
2. Click the **Fuse-IP Mode** drop-down menu and select **Client** as the Fuse-IP mode if you are dialing the studio codec. Note: the studio codec should be configured in server mode.



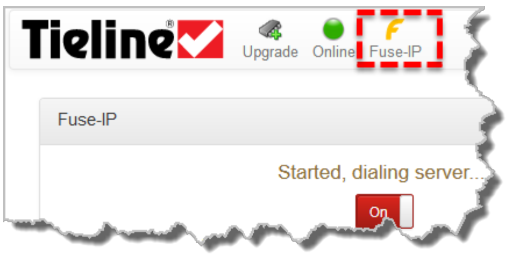

3. Enter the **Fuse-IP Server Address**, which is the public static IP address of the server codec at the studio. Then enter the server codec's serial number in the **Server Serial Number** text box. Leave the default **Fuse-IP Port** as **8999** in most situations unless this port is already in use, e.g. you have multiple codecs behind a firewall using Fuse-IP, therefore you need to allocate a different port for each Fuse-IP tunnel. Note: the port number on the client and server codecs must be the same. Configure the **Inactivity Timeout** if you want to turn the Fuse-IP tunnel off after a predetermined time period to save data, then click **Save**. Note: **Inactivity Timeout** can be configured from 0 to 1440 minutes. Enter **0** to disable the timeout.




4. Click the Fuse-IP **On/Off** button to create a Fuse-IP tunnel between the server and client codecs. Note: **Connected** should be displayed after a few seconds if both codecs are configured correctly and Fuse-IP is enabled on both codecs.



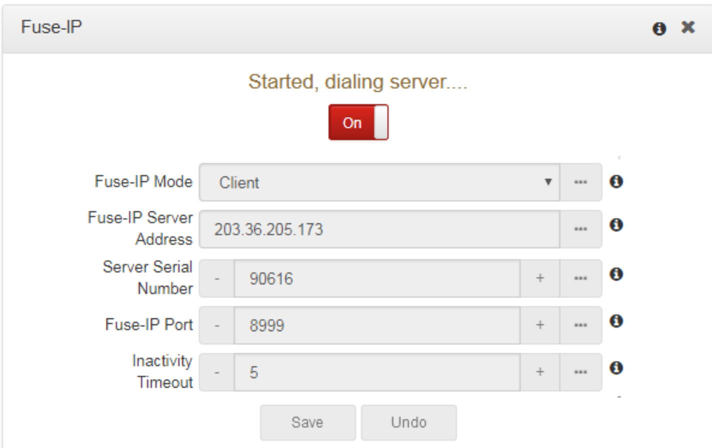

5. The status indicator is orange when Fuse-IP is enabled but no tunnel is created. It turns green when a tunnel is active.




**Fuse-IP Enabled & no tunnel created**


**Fuse-IP enabled & tunnel created**

6. Double-check all settings on both the server and client codecs if the message **Started, dialing server...** persists after turning Fuse-IP **On**.




**Important Notes:**

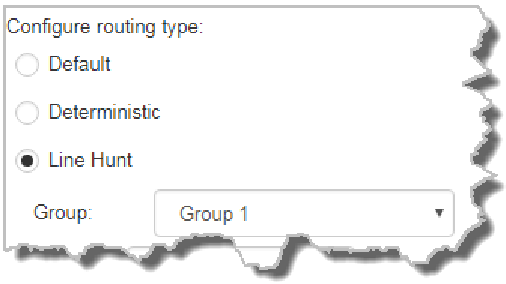
- Data is sent by the codec over the newly created 'tunnel' as soon as Fuse-IP is enabled, even if a connection has not been configured and dialed. Depending on the number of interfaces being used, the codecs may transmit and receive up to 24MB of data per hour at each end of the link.
- The codec remembers the Fuse-IP enabled/disabled state on power up.
- For additional stability it is recommended that a fixed jitter buffer is configured when streaming using Fuse-IP. The actual jitter buffer depth should account for the difference in delay between the interfaces and the maximum jitter experienced. To determine the

jitter over each link you can connect and stream audio over each interface separately and look at the jitter reading displayed on the **Connection Statistics** screen.
- Use a dotted quad IPv4 address when configuring the Fuse-IP Server Address.

## 16.4 Line Hunt Call Answering

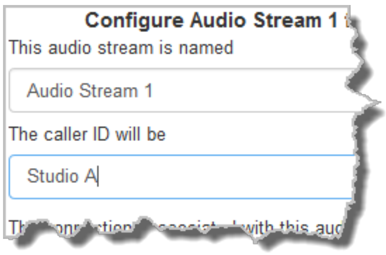
The codec supports line hunt call answering in multiple stream Tieline codecs, whereby line hunt groups can be assigned for multiple incoming callers on a first come, first served basis. This is ideal for separating groups of inputs and outputs between different studios or stations and this feature is available in Merlin, Genie and ViA codecs.

As an example, when dialing a codec using a multi-stream program with line hunt groups configured, select **Line Hunt** as the routing type. Then select the group to which the audio stream should be routed by the answering codec, e.g. **Group 1** in the following example.

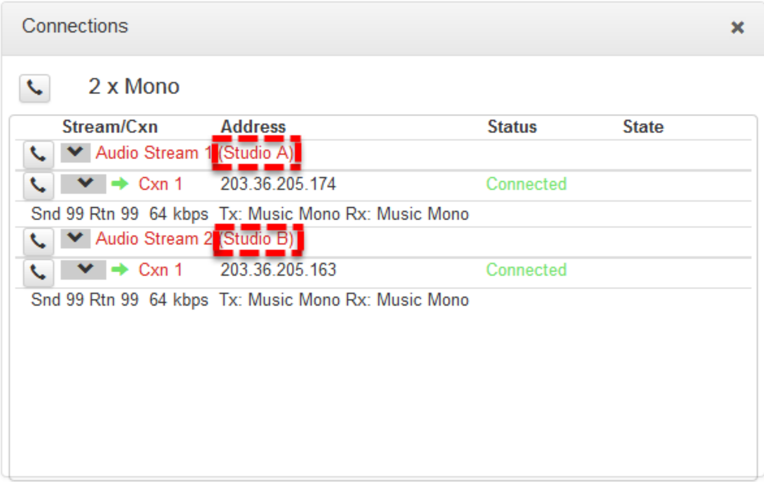



### Incoming Caller ID

Tieline codecs also support incoming caller IDs, so you can uniquely identify codecs or Report-IT users when they call in. This is particularly useful for identifying inbound callers when using line hunt answering mode.

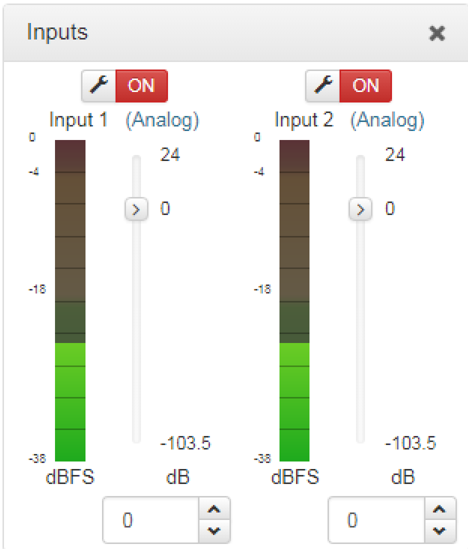



Any Tieline G5 codec dialing into a codec can display a designated **Caller ID** in the **Connection panel**. In the following example, **Studio A** and **Studio B** have called into a multi-stream codec using a specific caller ID, which is displayed next to the stream name.




## 16.5 Configuring Input/Output Settings

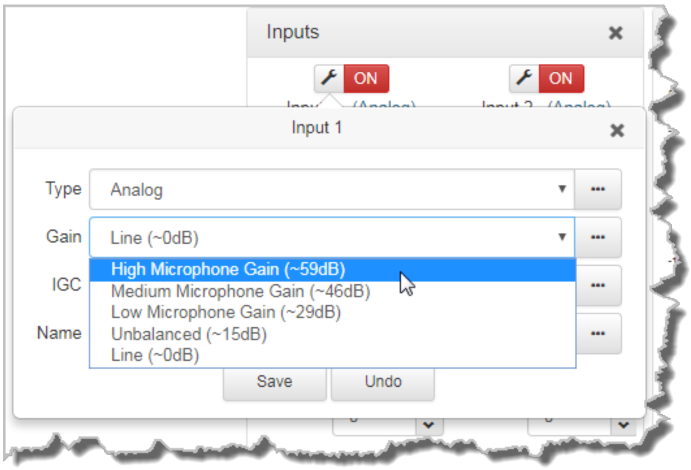
Open the HTML5 Toolbox Web-GUI and click **Audio** in the **Menu Bar**, then click **Inputs** to display the **Inputs panel**.




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

### Adjusting Analog and Digital Audio Levels

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



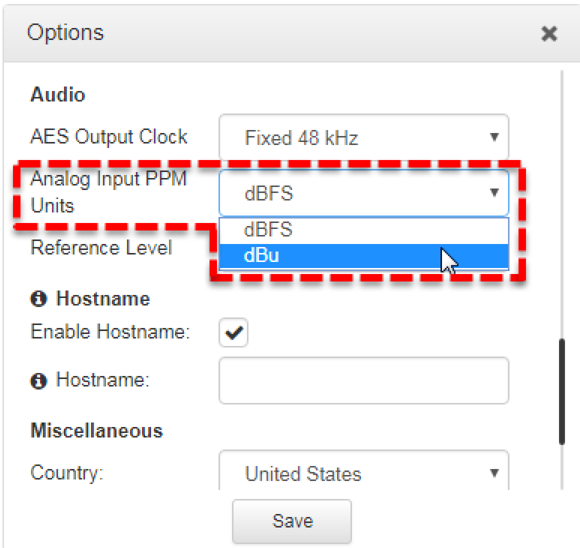

To adjust input audio levels, click on the input slider and drag it to the desired input gain level. Alternatively, click the arrows below a PPM meter to incrementally increase or decrease the input level in 0.5dB steps. Input levels on the **Input panel** should be set to ensure audio peaks average at the first yellow indications on the PPM meters, which represents nominal 0 VU at -18dBFS. Audio levels should also be verified using the meters in the **PPMs panel**.

Input attenuation is available when the input type is **AES** digital, however there is no additional gain.

## Changing the Input PPM Meter Units from dBFS to dBU

It is also possible to switch the analog input PPM meter unit of measurement from dBFS (default) to dBU:

1. Open the HTML5 Toolbox Web-GUI and click **Settings** in the **Menu Bar**, then click **Options** to display the **Options panel**.
2. Click the **Analog Inputs PPM Units** drop-down menu and select **dBU**.



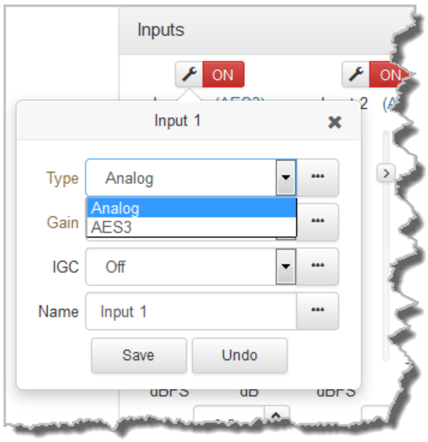

3. Click **Save** to change the setting.

# Configuring Input Settings

## Selecting Analog and Digital Audio Sources

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

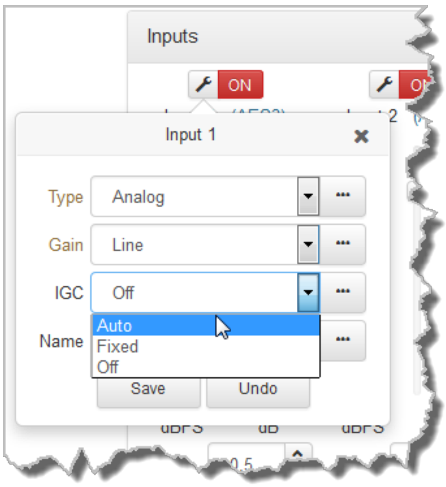
1. Click the **Input Settings** symbol.
2. Select **Type** and click to select either **Analog** or **AES3**.




3. Click **Save** to confirm the new setting.
4. See Configuring AES3 Audio for more information about the digital inputs and outputs.

## Adjusting IGC

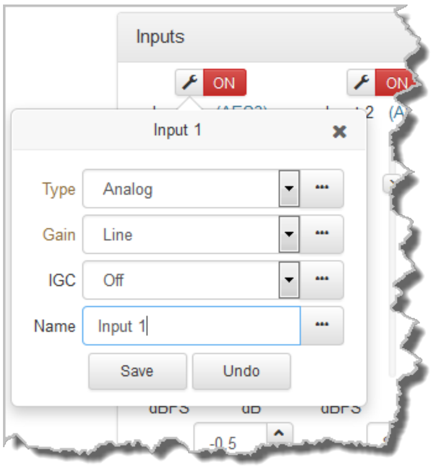
1. Click the **Input Settings** symbol.
2. Select **IGC** (Intelligent Gain Control) and then **Auto**, **Fixed** or **Off** as required.




3. Click **Save** to confirm the new setting.

## Renaming Inputs

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

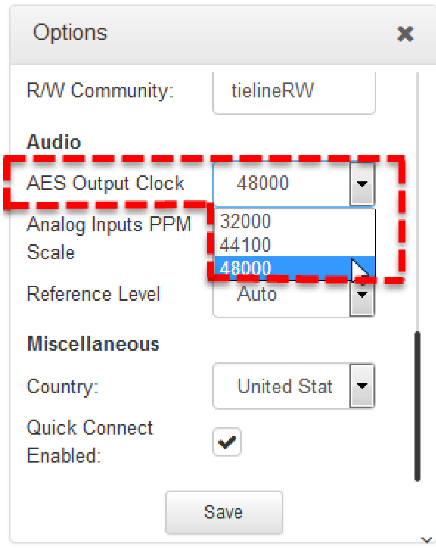



3. Click **Save** to confirm the name change.

## AES3 Output Sample Rate Configuration

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

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



## 16.6 Configure Mono or Stereo Peer-to-Peer Programs

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

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

This section contains instructions for:

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

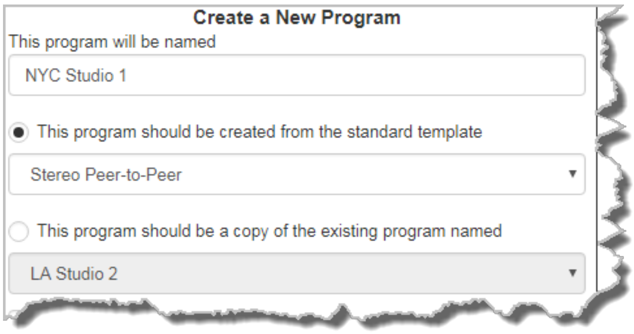
For more information about programs and audio streams within programs see the section titled About Program Dialing. Note: The following instructions will display how to configure a dial and answer program, with SD card file playback. If you want the codec to either dial or answer only, select the preferred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

### Configuring Peer-to-Peer Programs: Dialing

**Important Notes:** Before you start program configuration please note:

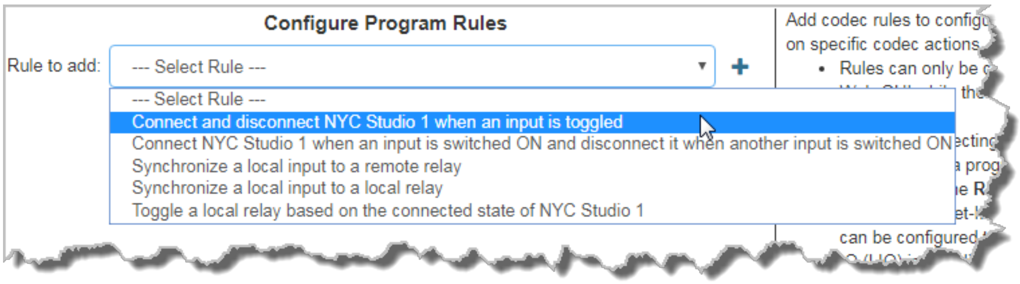
- You cannot edit a program when it is currently loaded in the codec.
- You can lock a loaded custom program in a codec to ensure the currently loaded program cannot be unloaded by a codec dialing in with a different type of program..
- Some drop-down menus and settings may be greyed out intentionally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

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click the **New Program** button to open the wizard and:
   - Click in the text box to name the new program.
   - Select **Mono/Stereo Peer-to-Peer,** or if you want to use an existing program as a template, select this option. Then click **Next**.




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

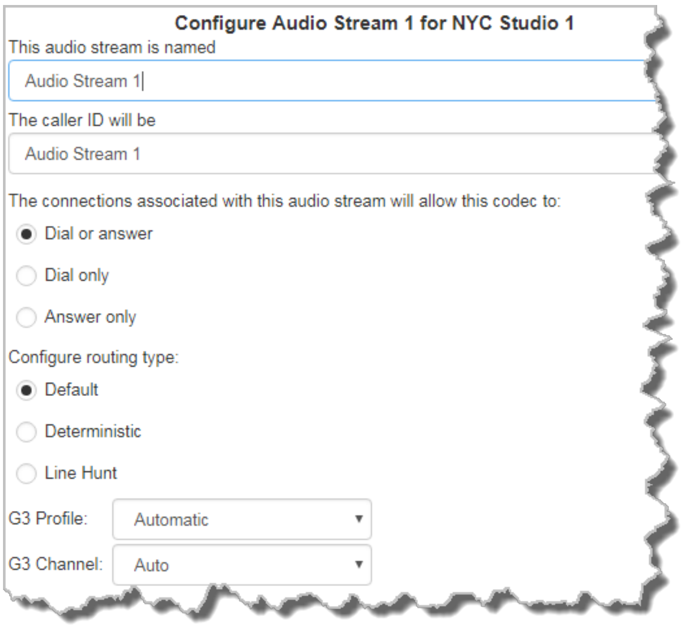
3. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** to add a new rule and click the **Minus symbol** to remove a rule.




**Important Notes for Rules:**

- Bridge-IT XTRA has 4 physical **CONTROL PORT GPIOs**; 7 Tieline and WheatNet virtual inputs (1-7); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network).
- Bridge-IT has 2 physical **CONTROL PORT GPIOs**; 5 Tieline and WheatNet virtual inputs (1-5); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network).
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- Tieline WheatNet-IP codecs require Wheatstone Razor firmware version 1.4.22 or later to support WheatNet LIOs. In addition, the WheatNet-IP codec must have the **WNet Enable LIO** checkbox selected in the **Options panel** of the HTML5 Toolbox Web-GUI.
- Relay reflection is not available for SIP and Multicast Client programs.
- Connection-related rules are not displayed in **Answer only** audio streams.
- Program level rules intended to activate dialing are not valid in **Answer only** programs or audio streams.
- For more details about rules see Creating Rules.

4. Enter a name for the **Audio Stream**, add a **caller ID** and configure the codec to dial, answer or dial and answer. Then click **Next**. Note: The caller ID is used to identify calls.



| Routing Type Options: | |
|---|---|
| **Default** IP | No **Dial Route** or **Answer Route** is configured. An incoming call will be routed to an audio stream on a first-come, first-served basis in a multi-stream program. Note: By default IP streams are routed using audio ports. |
| **Deterministic** IP | Use of **Dial** and **Answer Routes** is not usually necessary over IP because dedicated ports or **Line Hunt** mode call answering is employed. However dial routes can be used over IP when a single stream on an answering codec answers using POTS and/or ISDN connections, as well as IP. This effectively creates an answering group using different transports. |
| **Line Hunt** IP | Create line hunt groups for multiple incoming callers on a first come, first served basis. This is ideal for separating groups of inputs and outputs between different studios or stations. See Line Hunt Call Answering for more information. |

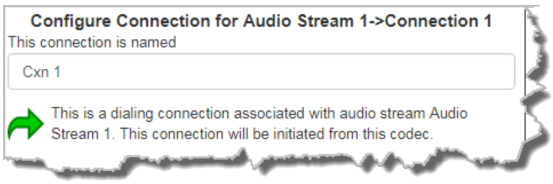
**Important Notes on G3 Profile Settings:** The G3 profile setting supports maintaining specific G3 codec settings when answering a call from a G5 codec.

1. **Auto**: The codec will dial the G3 codec and connect in mono or stereo.
2. **Dual Program**: This allows the codec to dial a G3 codec with a Dual Program profile loaded and support two simultaneous mono connections.
3. **Runtime**: The G3 codec will retain runtime settings when answering a call from a G5 codec.
4. **Custom**: The G3 codec will load a specified profile, e.g. profile 6, which is the first custom profile number.

**Important Notes on G3 Channel Settings**:
This setting is for compatibility with the **Dual Mono** profile in Tieline Commander G3 and i-Mix G3 codecs. It is designed to configure routing of the audio stream to a specific G3 codec channel consistently.

1. **Auto** (default): The answering codec will route incoming calls on a first come first served basis.
2. **Channel 1**: The answering codec will always route incoming calls to codec **Channel 1** (left output).
3. **Channel 2**: The answering codec will always route incoming calls to codec **Channel 2** (right output).

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



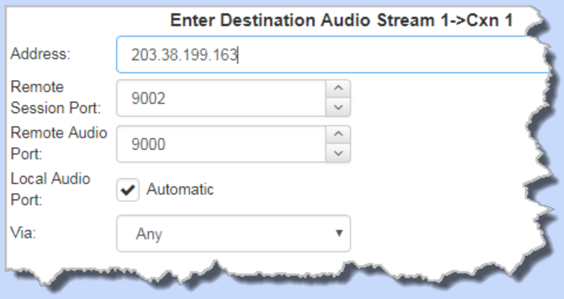

6. Follow the instructions on the right-hand side of the panel to configure the transport settings for the connection, then click **Next**.


**Important Notes:**

- If you select **Sessionless** as the **Session Protocol** select **UDP/IP +RTP** for RFC-compliant IP streaming.
- See RS232 Data Configuration for detailed information on RS232 data and see Enabling Relays and RS232 Data for more information on relay operations.

7. Configure destination codec dialing and encoding settings:

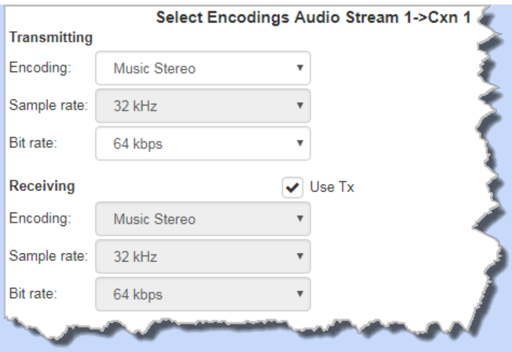
IP Configure the IP address, ports, and then specify which streaming interface is used to dial this connection, e.g. **Primary** (port **LAN1**) or **VLAN** if configured. **Session Port 9002** and **Remote Audio Port 9000** are used by default. VLANs can be used to configure separate control and streaming interfaces if required. Note: By default **Any** will select **LAN** if it is available.




**Important Note:** The **Remote Audio Port** is the codec port at the remote end of the link to which you are sending audio. The **Local Audio Port** is used by the local codec to receive audio from the remote codec. When **Tieline Codecs** is the **Session Protocol** selected (using Tieline session data), the default port value for the **Local Audio Port** is **Automatic**. Note: **Automatic** indicates that the codec will arbitrarily allocate the local port value and send this information to the codec to which you are dialing. Click to deselect the **Automatic** check-box and change this setting. When you select **Sessionless** as the **Session Protocol**, the **Session Port** is not configurable and you can manually configure the **Remote Audio Port** and **Local Audio Port**.

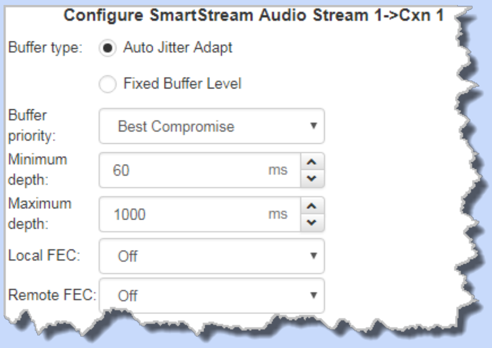
Click **Save Program** to save the program with the default algorithm, jitter and FEC settings which are physically entered in the codec. Alternatively, click **Next** to specify individual algorithm, jitter buffer and FEC settings, and configure SmartStream PLUS for this audio stream (recommended).

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



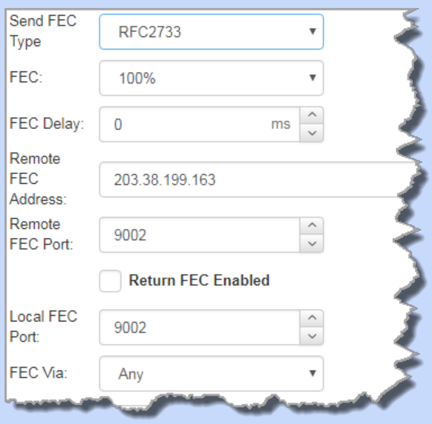

For IP connections click to configure:

- **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**. It is also possible to configure the **Minimum depth** and **Maximum depth** of jitter over the connection. See Configuring the Jitter Buffer for more details.
- Alternatively, select a **Fixed Buffer Level** and enter the **Jitter Depth**, which must be between 12ms and 5000ms depending on the algorithm you select.
- **Local** and **Remote FEC** settings if required.



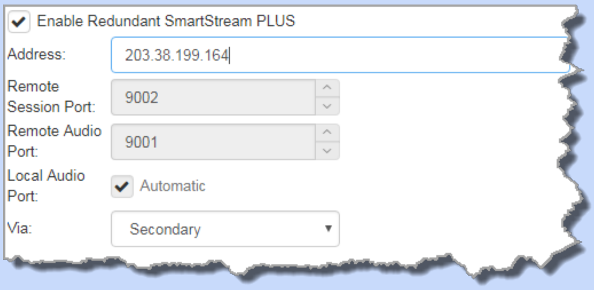

**Important Notes:**

- If you select **Sessionless** as the **Session Protocol** then RFC-compliant FEC is available. Configuration instructions are displayed in the right-hand pane.
- **FEC Delay** is only available when the **FEC** percentage is **100%**. This is designed to delay the sending of FEC packets for a predetermined period after the primary audio stream's packets are sent. This will increase the likelihood that the FEC packets will take an alternate route to the primary stream's packets. This means that if primary audio stream packets are not received at the remote codec, there is a good chance that FEC packets taking an alternate route will be received and replace them. When a **FEC** percentage lower than **100%** is configured, FEC packets are automatically delayed based on the ratio of primary packets to FEC packets sent at the selected setting.

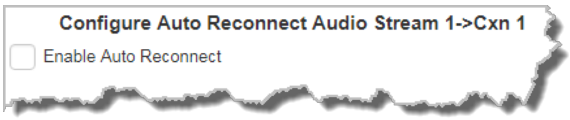


Click the check-box to select **Enable Redundant SmartStream PLUS** and configure dual Ethernet SmartStream IP streaming (To learn more visit http://www.tieline.com/Transports/SmartStream-IP). Alternatively, click **Next** to configure **Auto Reconnect** or a failover connection, whereby the alternative connection is dialed if the primary connection fails.

By default, primary IP streaming is via the **LAN** port. To achieve the maximum level of redundancy select **Secondary** to configure redundant streaming from **VLAN1** (or select another **Via**). The redundant stream uses **Remote Audio Port 9001** by default and the **Local Audio Port** allocated is **Automatic**. Note: **Automatic** indicates that the codec will arbitrarily allocate the local port value and send this information to the codec to which you are dialing.




8. Click **Next** and select the **Enable Auto Reconnect** check-box at this point in the wizard to enable this setting. Then click **Next** to configure answering settings.

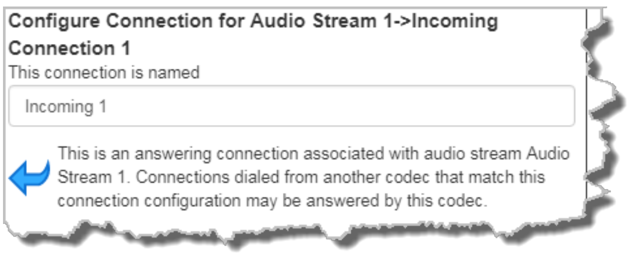



**Important Note:** When **Auto Reconnect** is enabled, the dialing codec will continue to attempt a connection with the remote codec until **Disconnect** is pressed either on the dialing codec's keypad, or in the Web-GUI.

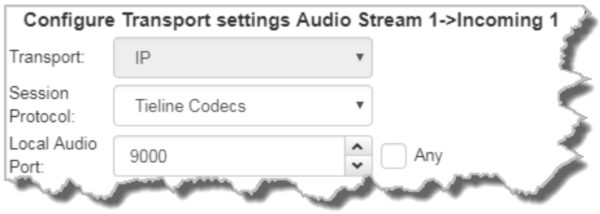
## Configuring the Codec to Answer Connections

If you are configuring the codec to answer an incoming audio stream connection:

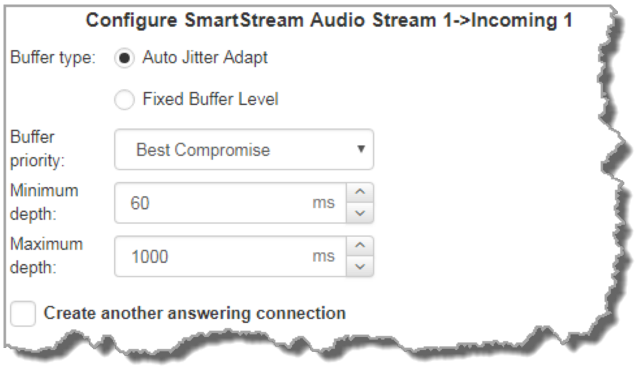
1. Enter a name for the answering connection and click **Next**.




2. Configure the IP transport settings, including the **Session Protocol** and **Local Audio Port**.




3. Click **Next** to specify jitter buffer settings, or create another answering connection. Click to configure:
   - **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**. It is also possible to configure the **Minimum depth** and **Maximum depth** of jitter over the connection. See Configuring the Jitter Buffer for more details.
   - Alternatively, select a **Fixed Buffer Level** and enter the **Jitter Depth**, which must be between 12ms and 5000ms depending on the algorithm you select.



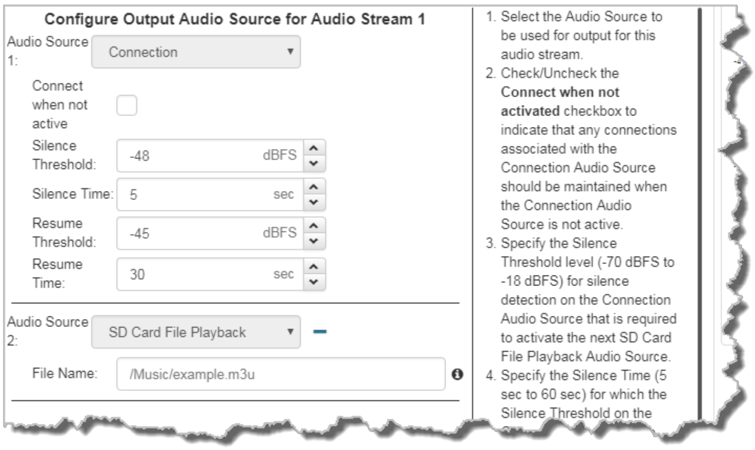

3. After configuring all settings there are 3 options:

   i. If you want to create another answering connection, select the check-box for **Create another answering connection** and continue through the wizard.
   ii. Click **Save Program** to save the program at this point.
   iii. Click **Next** to configure file playback using **Output Audio Source** options.

## Configuring SD Card File Playback Options

1. Click **Next** to configure SD card file playback using **Output Audio Source** configuration options to maintain program audio at transmitter sites.


2. Click the blue **Plus symbol** ➕ to add the **SD Card File Backup Output Audio Source**. Click the **Minus symbol** ➖ to remove it. Follow the instructions on the right-hand side of the panel to configure silence and resume threshold parameters.




3. After configuring **Output Audio Source** options you can:

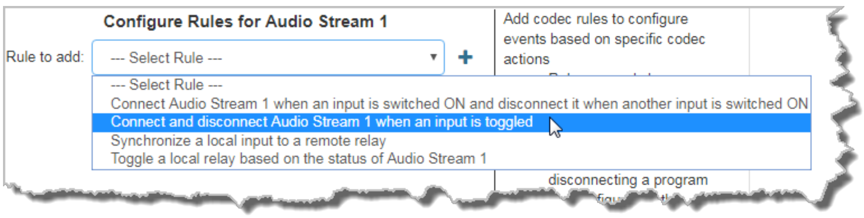
   i. Click **Save Program** to save the program at this point.
   ii. Click **Next** to configure rules options.

**Important Notes for File Playback:**

- A single partition FAT32 formatted SDHC Card is required (SD cards may be less reliable and are not recommended).
- The codec supports SDHC cards which have a physical capacity of up to 32GB.
- Create MP2 or MP3 files using a 32kHz, 44.1kHz or 48kHz sample rate.
- Ensure recordings used are not variable bit rate files.
- SDHC file audio is not sent to codec encoders and cannot be transmitted via an audio stream to another codec.
- File playback audio is sent directly to the codec outputs and therefore IGC is not available. When you create your MP2 or MP3 files ensure the audio levels match the audio reference level of your codec and that peaks average at the correct levels.
- If you create a single file name ensure you add the file extension, e.g. "test.mp3", or the file will not play back.
- If you create a directory name, all the files within the directory will be played back. We recommend you save all audio files as a playlist and link to this if you want them to play out sequentially. Please note that "M3U" is the playlist file format supported by the codec.

## Configuring Rules

1. To configure new stream level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** ➕ to add a new rule and click the **Minus symbol** ➖ to remove a rule.




**Important Note:** Program level rules intended to activate dialing are not valid in **Answer only** programs or audio streams.

2. Click **Save Program** to save the program.
3. Click **Finish** to exit the wizard.

4. The newly created program will be displayed in the left pane within the **Program Manager panel** and in the **Program Loader panel**. Select and connect audio streams in a program using the **Connections panel**, or dial the program manually using the codec front panel.

## 16.7 Configuring Multicast Client Programs

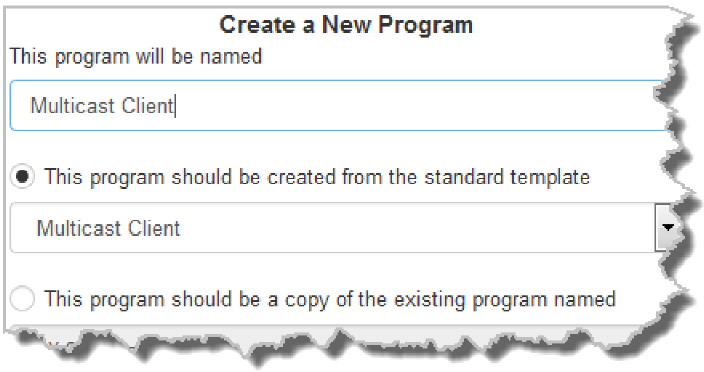
**Important Notes:** Before you commence program configuration please note:

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

- Use firmware higher than 2.8.xx in the Bridge-IT, Genie and Merlin families of codecs to enable auxiliary data.
  - It is not possible to connect to a G3 codec and receive multicast IP audio streams.
  - To copy multicast client programs onto multiple codecs see Save and Restore Configuration Files.

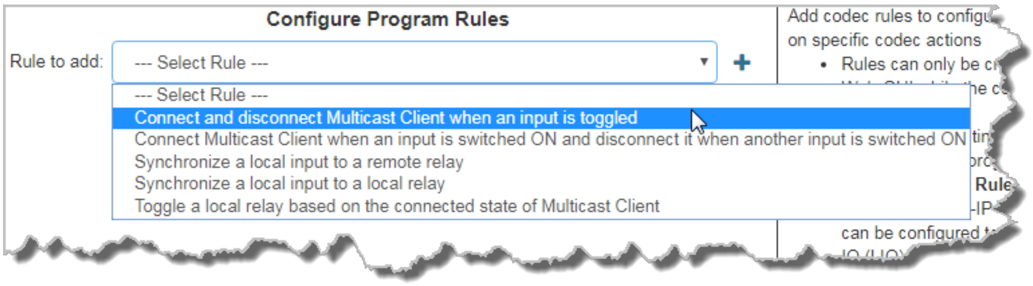
### Configuring Multicast Client Programs

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click the **New Program** button to open the wizard and:
   - Click in the text box to name the new program.
   - Select **Multicast Client** to configure a multicast program, or if you want to use an existing program as a template, select this option. Then click **Next**.



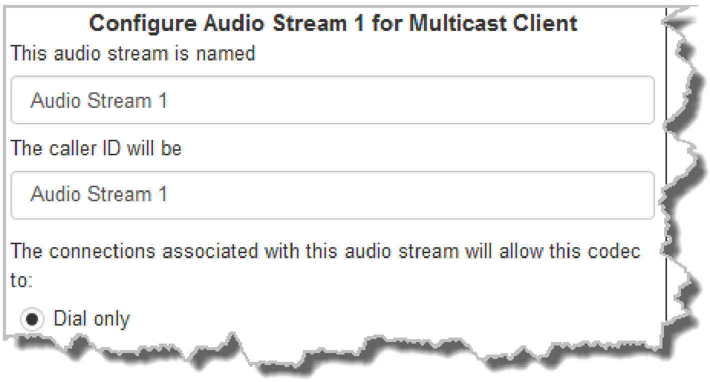

**Important Notes:** When you decide to use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

3. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** to add a new rule and click the **Minus symbol** to remove a rule.

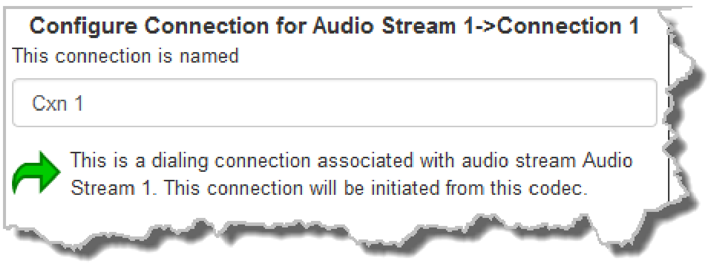



**Important Notes for Rules:**
- Bridge-IT XTRA has 4 physical **CONTROL PORT GPIOs**; 7 Tieline and WheatNet virtual inputs (1-7); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network).
- Bridge-IT has 2 physical **CONTROL PORT GPIOs**; 5 Tieline and WheatNet virtual inputs (1-5); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network).
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- Tieline WheatNet-IP codecs require Wheatstone Razor firmware version 1.4.22 or later to support WheatNet LIOs. In addition, the WheatNet-IP codec must have the **WNet Enable LIO** checkbox selected in the **Options panel** of the HTML5 Toolbox Web-GUI.
- Relay reflection is not available for SIP and Multicast Client programs.
- Connection-related rules are not displayed in **Answer only** audio streams.
- Program level rules intended to activate dialing are not valid in **Answer only** programs or audio streams.
- For more details about rules see Creating Rules.

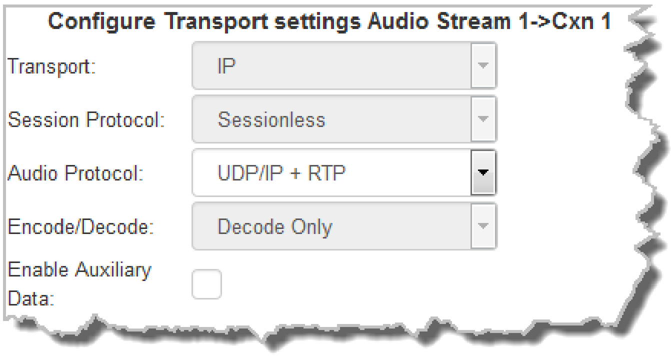
4. Enter a name for the **Audio Stream**, then click **Next**.




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

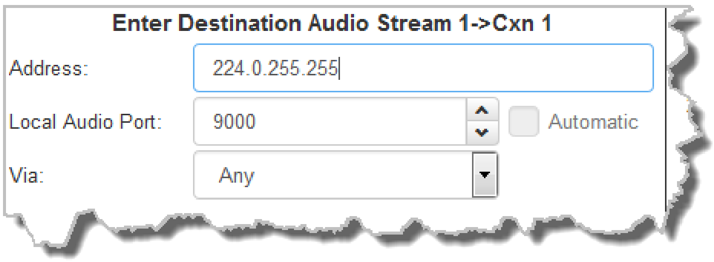


6. Follow the instructions on the right-hand side of the panel to configure the transport settings for the connection, then click **Next**. Note: select **UDP/IP +RTP** for RFC compliant streaming. If auxiliary data is enabled the audio stream will not be RFC-compliant.

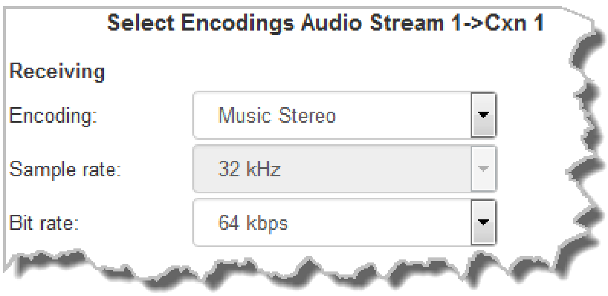



**Important Notes:**
- The encode and decode direction is configured automatically for **Encode Only** (server program) or **Decode Only** (client program). This setting is configured when you select either **Multicast Server** or **Multicast Client** when you first create the program in the wizard.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

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

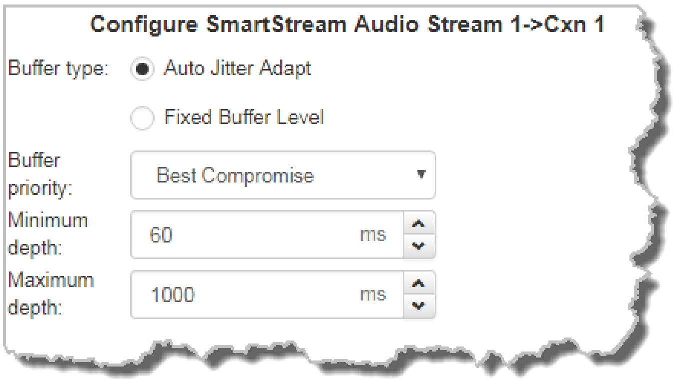



8. Click the drop-down arrows on the right-hand side of each text box to select the **Encoding**, **Sample rate**, **Bit rate** or **Sample size** options. Click **Next** to continue.



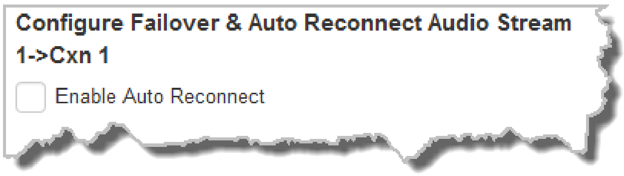

9. Click to configure:
   - **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**. It is also possible to configure the **Minimum depth** and **Maximum depth** of jitter over the connection. See Configuring the Jitter Buffer for more details.
   - Alternatively, select a **Fixed Buffer Level** and enter the **Jitter Depth**, which must be between 12ms and 5000ms depending on the algorithm you select.

Next, select **Return FEC Enabled** to enable RFC2733 compliant Forward Error Correction, which is available because multicast connections are sessionless. Follow the instructions in the right-hand pane. When configuration is complete click **Next**.




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

10. Click the **Enable Auto Reconnect** check-box to turn this feature on, then click **Next**.

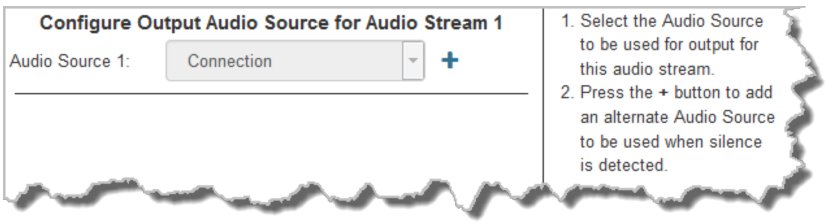



11. After configuring all settings there are 2 options:

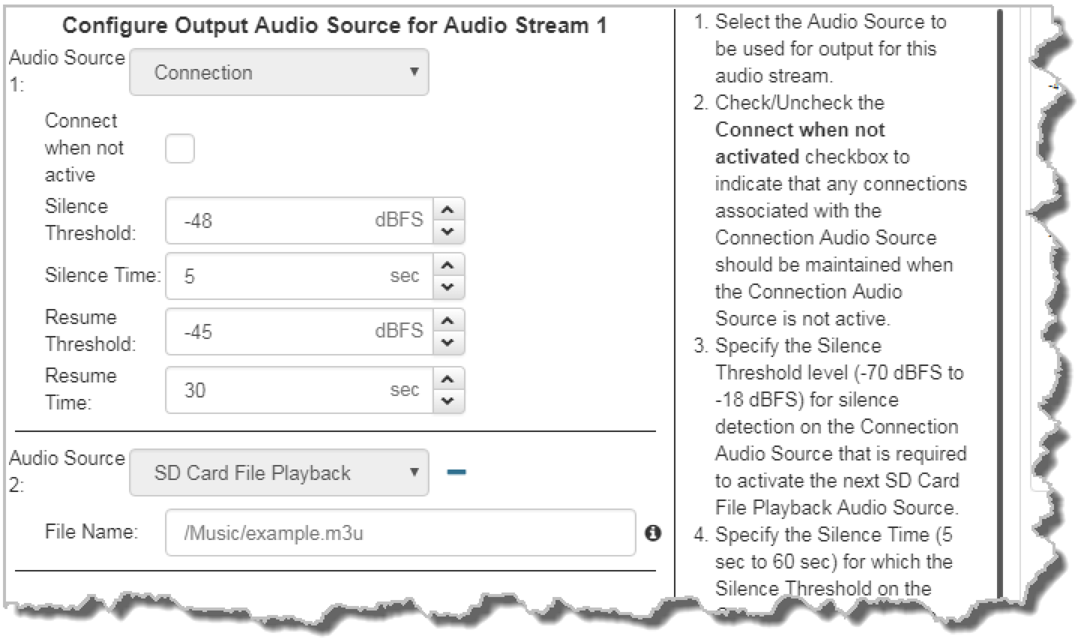
- Click **Save Program** to save the program at this point.
- Click **Next** to configure file playback using **Output Audio Source** options.

## Configuring SD Card File Playback Options

1. Click **Next** to configure SD card file playback using **Output Audio Source** configuration options to maintain program audio at transmitter sites.




2. Click the blue **Plus symbol** to add **SD Card File Playback** as an **Output Audio Source**. Click the **Minus symbol** to remove it. Follow the instructions on the right-hand side of the panel to configure silence and resume threshold parameters.




3. After configuring **Output Audio Source** options you can:

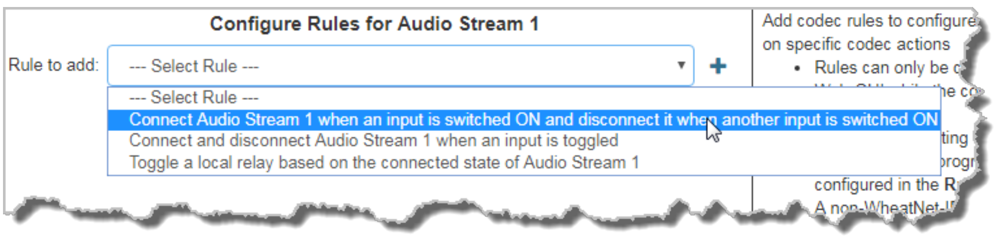
   i. Click **Save Program** to save the program at this point.
   ii. Click **Next** to configure rules options.

**Important Notes for File Playback:**

- A single partition FAT32 formatted SDHC Card is required (SD cards may be less reliable and are not recommended).
- The codec supports SDHC cards which have a physical capacity of up to 32GB.
- Create MP2 or MP3 files using a 32kHz, 44.1kHz or 48kHz sample rate.
- Ensure recordings used are not variable bit rate files.
- SDHC file audio is not sent to codec encoders and cannot be transmitted via an audio stream to another codec.
- File playback audio is sent directly to the codec outputs and therefore IGC is not available. When you create your MP2 or MP3 files ensure the audio levels match the audio reference level of your codec and that peaks average at the correct levels.
- If you create a single file name ensure you add the file extension, e.g. "test.mp3", or the file will not play back.
- If you create a directory name, all the files within the directory will be played back. We recommend you save all audio files as a playlist and link to this if you want them to play out sequentially. Please note that "M3U" is the playlist file format supported by the codec.

## Configuring Rules

1. To configure new stream level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** ➕ to add a new rule and click the **Minus symbol** ➖ to remove a rule.



**Important Note:** Program level rules intended to activate dialing are not valid in **Answer only** programs or audio streams.

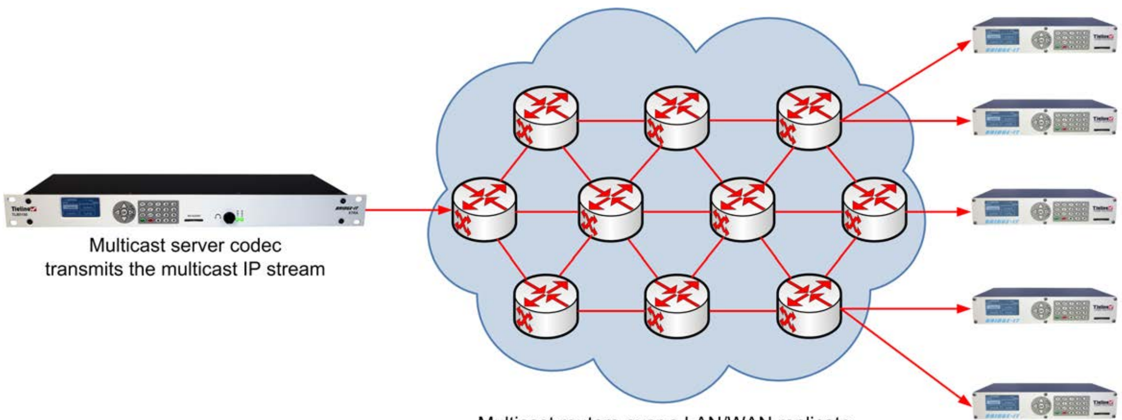
2. Click **Save Program** to complete configuration of the program.
3. Click **Finish** to exit the wizard.
4. Configure multicast server and multicast client programs and load all codecs with the appropriate program. Select and connect audio streams in a program using the **Connections panel**, or dial the program manually using the codec front panel. Dial the multicast server program connection first and then connect multicast client codec programs to begin receiving multicast streams.

# 16.8 Configuring Multicast Server Programs

## How Multicasting Works

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

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




**Important Notes:**
- When a connection is dialed Tieline codecs normally use session data to configure settings like the algorithm, connection bit rate and sample rate etc. Multicast connections are sessionless and do not use Tieline session data. As a result, it is imperative that all codecs are configured with the same connection settings prior to connecting, or they will not be able to join multicast streaming sessions.
- Automatic or fixed jitter buffer settings can be adjusted on individual client codecs as required. There is no jitter buffer setting on the server codec because it never receives audio packets.

## Prerequisites:

- Bridge-IT firmware v.1.01.00 r4219 or higher.
- G3 codec firmware v.1.6.56 or higher (if connecting to a G3 codec).
- ToolBox web-GUI v.1.2.2.3 or higher.

- A multi-unicast license installed in the dialing codec (Note: the Multi-Unicast license includes multicast server capability).
- Use firmware higher than 2.8.xx in the Bridge-IT, Genie and Merlin families of codecs to transmit auxiliary data.

## Multicast Server versus Multicast Client Programs

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

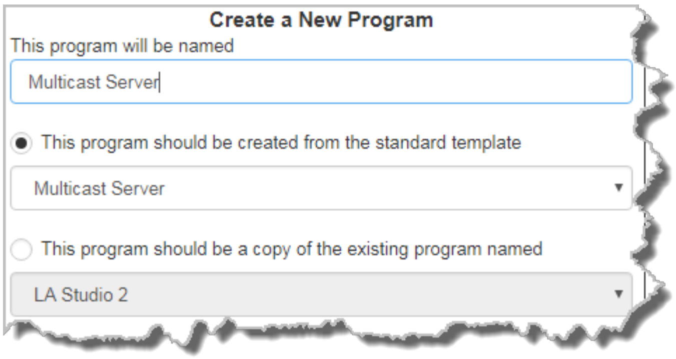
- A multicast server program is used by the broadcasting codec to send multicast IP packets to multicast routers on a network.
- A multicast client program is used by codecs to receive multicast IP audio packets.

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

## Creating Multicast Server Programs

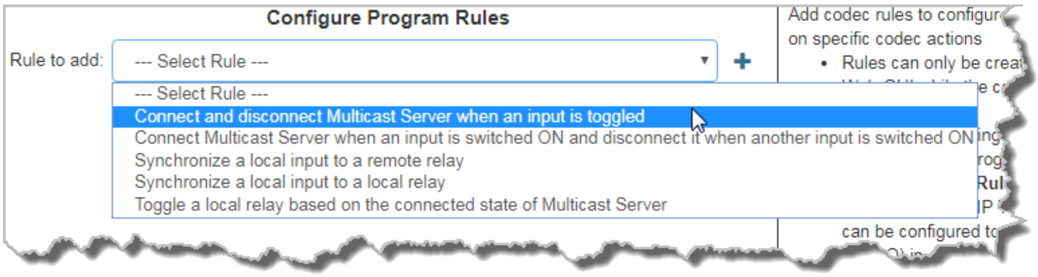
**Important Notes:** Before you start program configuration please note:
  - Ensure all connection related settings like the port, algorithm, bit rate (etc) match on both multicast server and client programs or they will not connect successfully.
  - You cannot edit a program when it is currently loaded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

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click the **New Program** button to open the wizard and:
    - Click in the text box to name the new program.
    - Select **Multicast Server** or if you want to use an existing program as a template, select this option. Then click **Next**.




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

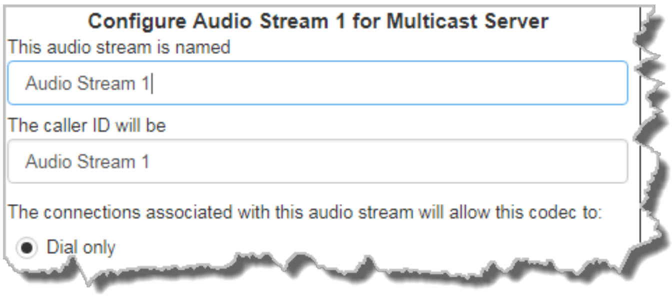
3. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** ➕ to add a new rule and click the **Minus symbol** ➖ to remove a rule.



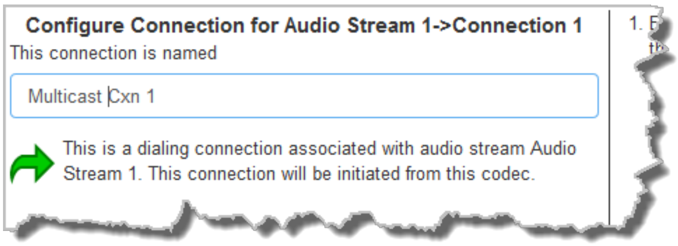

**Important Notes for Rules:**

- Bridge-IT XTRA has 4 physical **CONTROL PORT GPIOs**; 7 Tieline and WheatNet virtual inputs (1-7); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network).
- Bridge-IT has 2 physical **CONTROL PORT GPIOs**; 5 Tieline and WheatNet virtual inputs (1-5); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network).
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- Tieline WheatNet-IP codecs require Wheatstone Razor firmware version 1.4.22 or later to support WheatNet LIOs. In addition, the WheatNet-IP codec must have the **WNet Enable LIO** checkbox selected in the **Options panel** of the HTML5 Toolbox Web-GUI.
- Relay reflection is not available for SIP and Multicast Client programs.
- Connection-related rules are not displayed in **Answer only** audio streams.
- Program level rules intended to activate dialing are not valid in **Answer only** programs or audio streams.
- For more details about rules see Creating Rules.

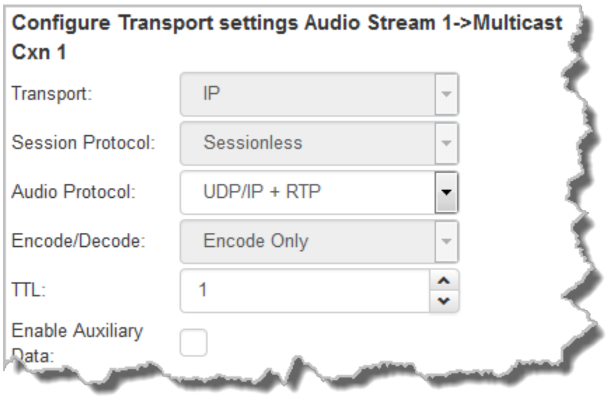
4. Enter a name for the **Audio Stream**, then click **Next**.




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



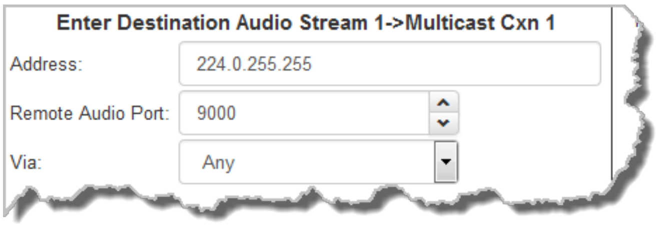

6. Follow the instructions on the right-hand side of the panel to configure the transport settings for the connection, then click **Next**. Note: select **UDP/IP +RTP** for RFC-compliant streaming. If auxiliary data is enabled the audio stream will not be RFC-compliant.



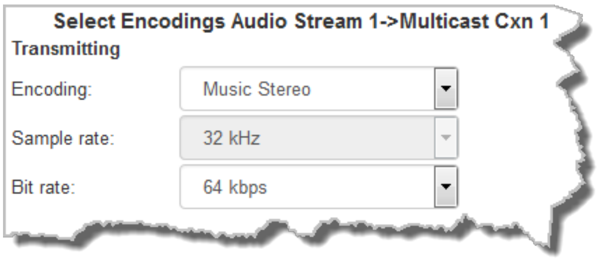

**Important Notes:**

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

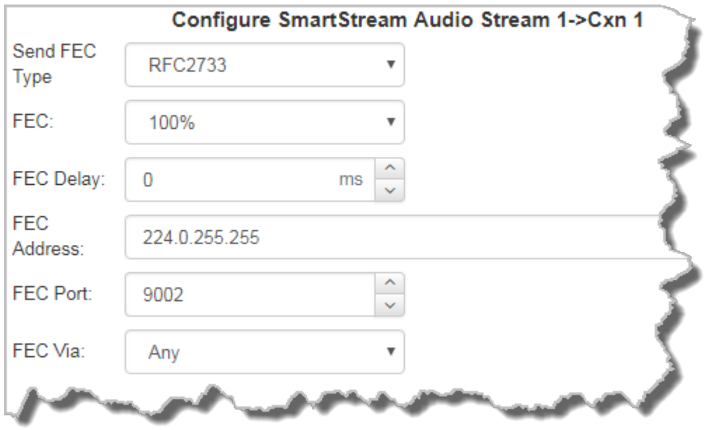
7. Configure the multicast IP address and **Remote Audio Port** (the same multicast address and port must be used for both the server and client programs), then specify which IP streaming interface is used to dial this connection, e.g. **Primary** (**LAN / ETHERNET** port and default setting) or **VLAN**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



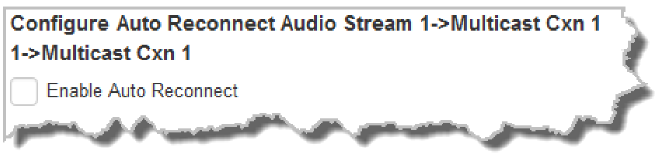

8. Click the drop-down arrows on the right-hand side of each text box to select the **Encoding**, **Sample rate**, **Bit rate** or **Sample size** options. Click **Next** to continue.




9. Click the drop-down arrow for **Send FEC Type** to enable RFC-compliant FEC. Configuration instructions are displayed in the right-hand **Program Manager panel** pane. **FEC Delay** is only available when the **FEC** percentage is **100%**. This is designed to delay the sending of FEC packets for a predetermined period after the primary audio stream's packets are sent. This will increase the likelihood that the FEC packets will take an alternate route to the primary stream's packets. This means that if primary audio stream packets are not received at the remote codec, there is a good chance that FEC packets taking an alternate route will be received and replace them. When a **FEC** percentage lower than **100%** is configured, FEC packets are automatically delayed based on the ratio of primary packets to FEC packets sent at the selected setting.



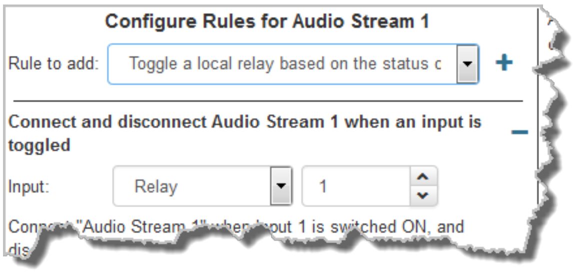

10. Click **Next** to select configure **Enable Auto Reconnect** if required.




11. Next you can either:

    i. Click **Next** to configure **Rules** options.
    ii. Click **Save Program**.

12. To configure new stream-level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** to add a new rule and click the **Minus symbol** to remove a rule.




13. After configuring all streams in the multicast server program click **Save Program** to save the program settings, then click **Finish**.

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

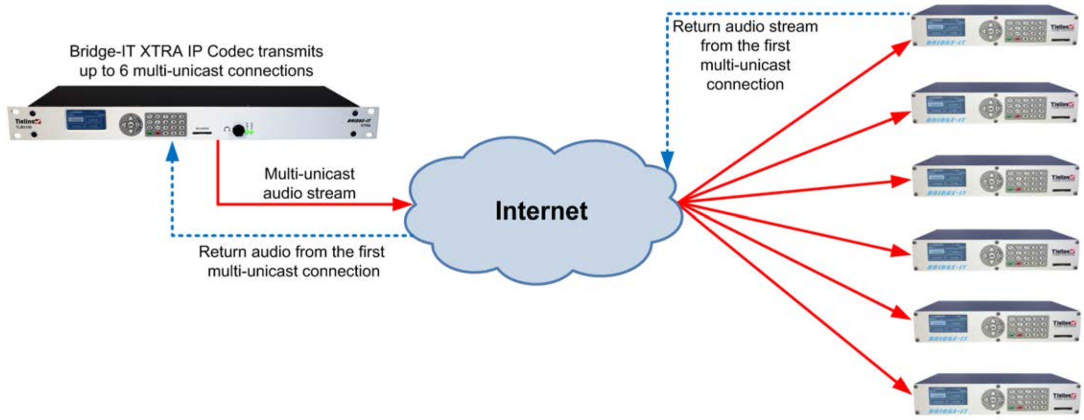
14. Configure multicast server and multicast client programs and load all codecs with the appropriate program. Dial the multicast server program connection first and then connect multicast client codec programs to begin receiving multicast audio packets. Select and connect audio streams in a program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

## 16.9 Configuring Multi-Unicast Dialing Programs

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

### Prerequisites:

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

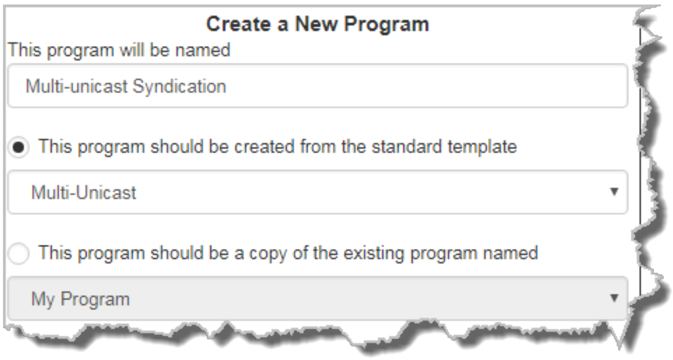



**Important Notes:** Before you start program configuration please note:
- You cannot edit a program when it is currently loaded in the codec.
- You can lock a loaded custom program in a codec to ensure the currently loaded program cannot be unloaded by a codec dialing in with a different program type.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

### Creating a Multi-Unicast Program

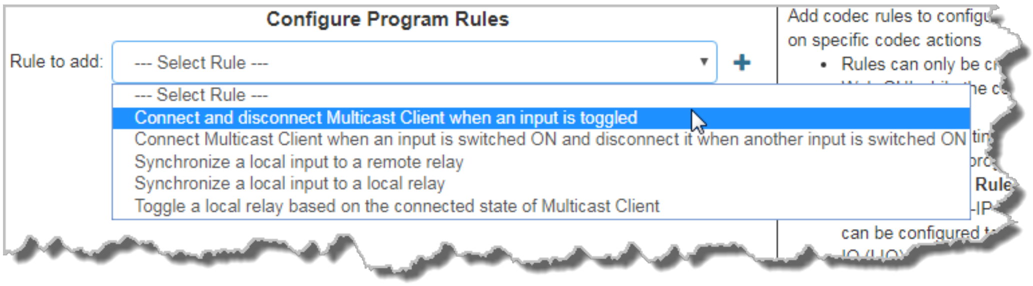
1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.

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




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

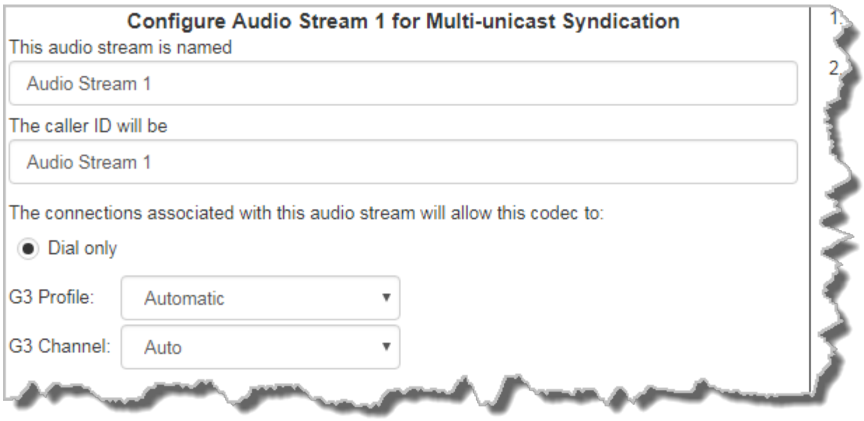
3. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** to add a new rule and click the **Minus symbol** to remove a rule.




**Important Notes for Rules:**

- Bridge-IT XTRA has 4 physical **CONTROL PORT GPIOs**; 7 Tieline and WheatNet virtual inputs (1-7); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network).
- Bridge-IT has 2 physical **CONTROL PORT GPIOs**; 5 Tieline and WheatNet virtual inputs (1-5); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network).
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- Tieline WheatNet-IP codecs require Wheatstone Razor firmware version 1.4.22 or later to support WheatNet LIOs. In addition, the WheatNet-IP codec must have the **WNet Enable LIO** checkbox selected in the **Options panel** of the HTML5 Toolbox Web-GUI.
- Relay reflection is not available for SIP and Multicast Client programs.
- Connection-related rules are not displayed in **Answer only** audio streams.
- Program level rules intended to activate dialing are not valid in **Answer only** programs or audio streams.
- For more details about rules see Creating Rules.

4. Enter a name for the **Audio Stream** and add a **caller ID**. Then click **Next**. Note: The caller ID is used to identify calls.




| Routing Type Options: | |
|---|---|
| **Default** IP | No **Dial Route** or **Answer Route** is configured. An incoming call will be routed to an audio stream on a first-come, first-served basis in a multi-stream program. Note: By default IP streams are routed using audio ports. |
| **Deterministic** IP | Use of **Dial** and **Answer Routes** is not usually necessary over IP because dedicated ports or **Line Hunt** mode call answering is employed. However dial routes can be used over IP when a single stream on an answering codec answers using POTS and/or ISDN connections, as well as IP. This effectively creates an answering group using different transports. |
| **Line Hunt** IP | Create line hunt groups for multiple incoming callers on a first come, first served basis. This is ideal for separating groups of inputs and outputs between different studios or stations. See Line Hunt Call Answering for more information. |

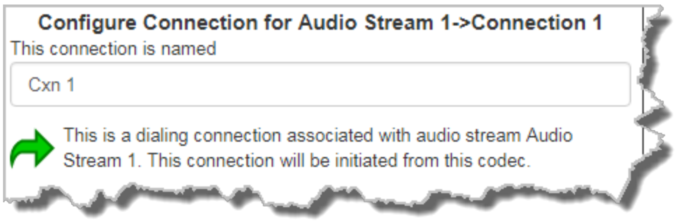
**Important Notes on G3 Profile Settings:** The G3 profile setting supports maintaining specific G3 codec settings when answering a call from a G5 codec.

1. **Auto**: The codec will dial the G3 codec and connect in mono or stereo.
2. **Dual Program**: This allows the codec to dial a G3 codec with a Dual Program profile loaded and support two simultaneous mono connections.
3. **Runtime**: The G3 codec will retain runtime settings when answering a call from a G5 codec.
4. **Custom**: The G3 codec will load a specified profile, e.g. profile 6, which is the first custom profile number.

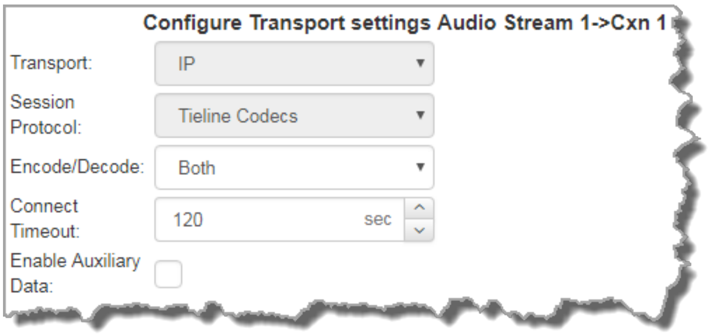
**Important Notes on G3 Channel Settings**:
This setting is for compatibility with the **Dual Mono** profile in Tieline Commander G3 and i-Mix G3 codecs. It is designed to configure routing of the audio stream to a specific G3 codec channel consistently.

1. **Auto** (default): The answering codec will route incoming calls on a first come first served basis.
2. **Channel 1**: The answering codec will always route incoming calls to codec **Channel 1** (left output).
3. **Channel 2**: The answering codec will always route incoming calls to codec **Channel 2** (right output).

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

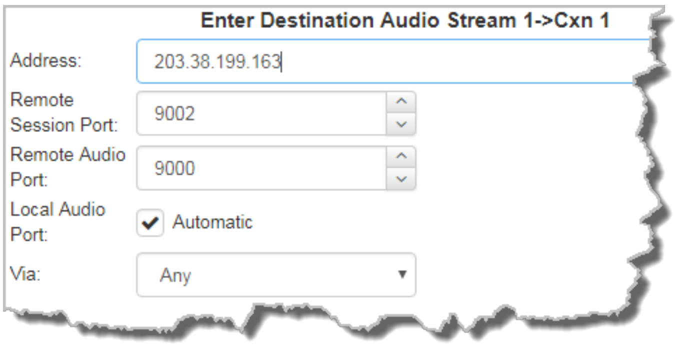



6. Follow the instructions on the right-hand side of the panel to configure the transport settings for the connection, then click **Next**. Note: only the first connection dialed in a multi-unicast program can encode and decode audio. All other connections are unidirectional and encode only.




**Important Note:** See RS232 Data Configuration for detailed information on RS232 data. Only in-band auxiliary data using the Music or MusicPLUS algorithms is possible when connecting to G3 Commander and i-Mix codecs.

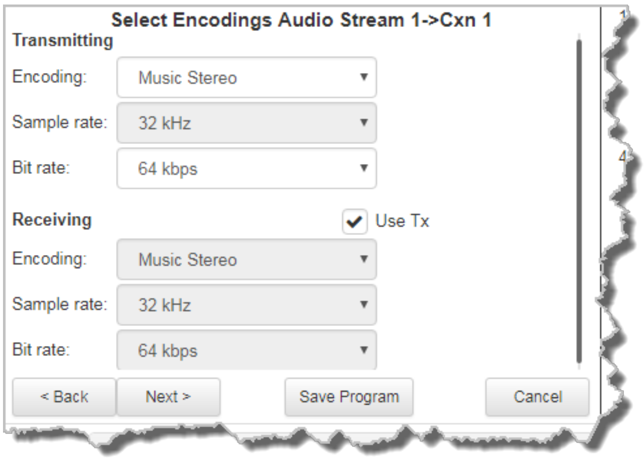
7. Configure the IP address, ports, and then specify which streaming interface is used to dial this connection, e.g. **Primary** (**LAN / ETHERNET** port and default setting) or **VLAN** if configured. Note: By default **Any** will select **Primary**.




**Important Notes:** The **Remote Audio Port** is the codec port at the remote end of the link to which you are sending audio. The **Local Audio Port** is the port used by the local codec to receive audio from the remote codec. When **Tieline Codecs** is the **Session Protocol** selected (using Tieline session data), the default port value for the **Local Session Port** and **Local Audio Port** is **Automatic**. Note: **Automatic** indicates that the codec will arbitrarily allocate the local port value and send this information to the codec to which you are dialing. Click to deselect the **Automatic** check-box and change this setting.

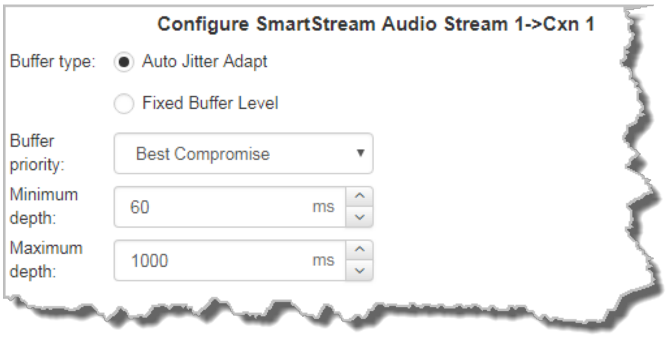
Click **Save Program** to save the program with the default algorithm, jitter and FEC settings which are physically entered in the codec. Alternatively, click **Next** to specify individual algorithm, jitter buffer, FEC and auto reconnect settings for this audio stream (recommended).

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

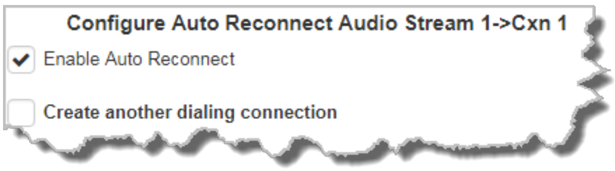



Click to configure:

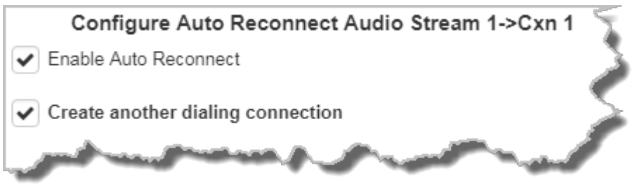
- **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**. It is also possible to configure the **Minimum depth** and **Maximum depth** of jitter over the connection. See Configuring the Jitter Buffer for more details.
- Alternatively, select a **Fixed Buffer Level** and enter the **Jitter Depth**, which must be between 12ms and 5000ms depending on the algorithm you select.



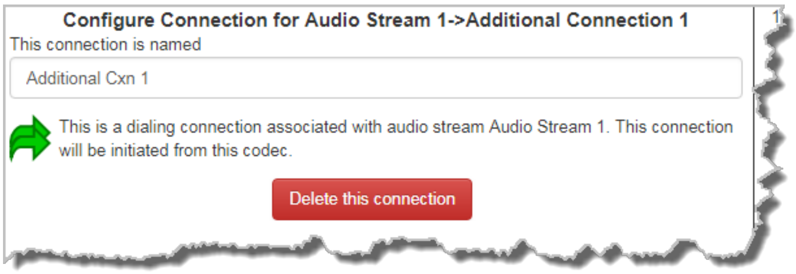

9. Select the **Enable Auto Reconnect** check-box if you want to enable this feature. This is enabled by default.



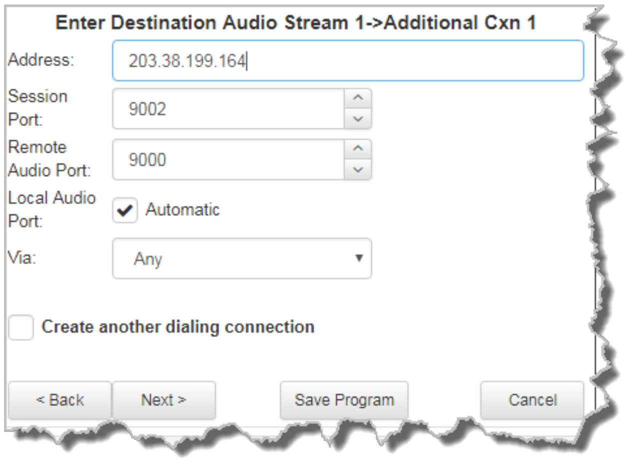

10. Select the **Create another dialing connection** check-box to configure a new connection for an additional endpoint, then click **Next**.




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



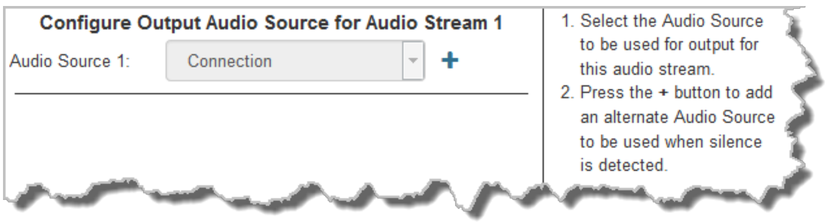

12. Enter destination settings for the dialing connection.



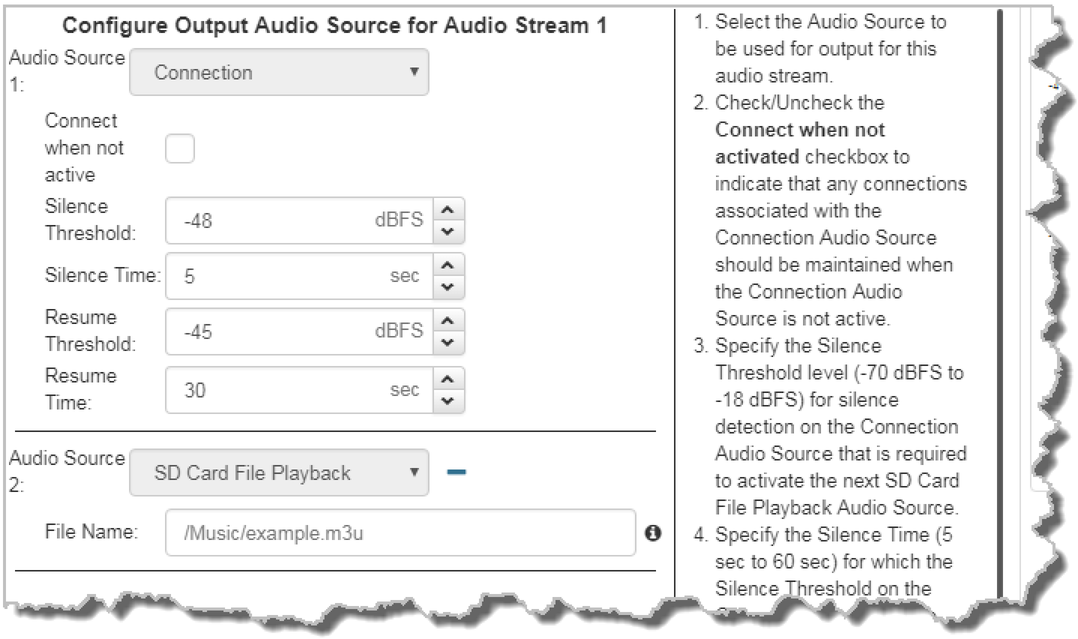

13. There are three options at this point:
    - Select **Create another dialing connection** to dial another endpoint.
    - Click **Next** to configure the **Output Audio Source** for the audio stream.
    - Click **Save Program** to save the program at this point.

## Configuring SD Card File Playback Options

1. Click **Next** to configure SD card file playback using **Output Audio Source** configuration options to maintain program audio at transmitter sites.




2. Click the blue **Plus symbol** ➕ to add **SD Card File Playback** as an **Output Audio Source**. Click the **Minus symbol** ➖ to remove it. Follow the instructions on the right-hand side of the panel to configure silence and resume threshold parameters.




3. After configuring **Output Audio Source** options you can:

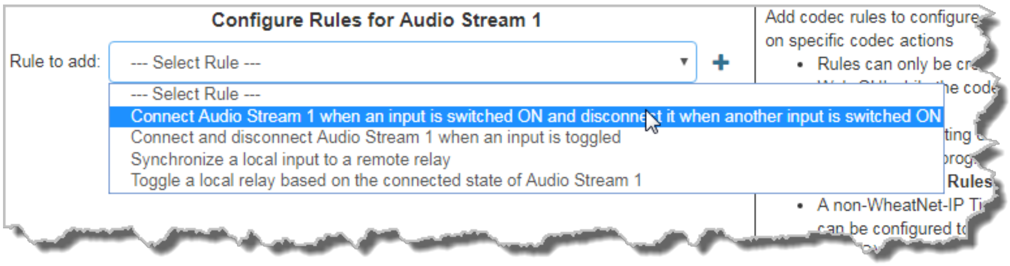
   i. Click **Save Program** to save the program at this point.
   ii. Click **Next** to configure rules options.

**Important Notes for File Playback:**

- A single partition FAT32 formatted SDHC Card is required (SD cards may be less reliable and are not recommended).
- The codec supports SDHC cards which have a physical capacity of up to 32GB.
- Create MP2 or MP3 files using a 32kHz, 44.1kHz or 48kHz sample rate.
- Ensure recordings used are not variable bit rate files.
- SDHC file audio is not sent to codec encoders and cannot be transmitted via an audio stream to another codec.
- File playback audio is sent directly to the codec outputs and therefore IGC is not available. When you create your MP2 or MP3 files ensure the audio levels match the audio reference level of your codec and that peaks average at the correct levels.
- If you create a single file name ensure you add the file extension, e.g. "test.mp3", or the file will not play back.
- If you create a directory name, all the files within the directory will be played back. We recommend you save all audio files as a playlist and link to this if you want them to play out sequentially. Please note that "M3U" is the playlist file format supported by the codec.

## Configuring Rules

1. To configure new stream level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** to add a new rule and click the **Minus symbol** to remove a rule.




2. Click **Save Program** to complete configuration of the program.
3. Click **Finish** to exit the wizard.
4. The newly created program will be displayed in the left pane within the **Program Manager panel** and in the **Program Loader panel**. Select and connect audio streams in a program using the **Connections panel**, or dial the program manually using the codec front panel. Note: When you initially dial using a multi-unicast program all audio streams are dialed. It is then possible to connect and disconnect audio streams individually.
5.

## 16.10 Configuring SIP

The codec is fully EBU N/ACIP Tech 3326 compliant when connecting using SIP (Session Initiation Protocol) to other brands of IP codecs. For more background on SIP connections and the differences between registered and unregistered peer-to-peer SIP connections see About SIP.

To configure the codec to dial over SIP using a SIP Server you will need to:

1. Register the codec to a SIP server using SIP account credentials.
2. Configure a SIP interface in the codec. Note: This **SIP1** or **SIP2** interface will include the proxy and port settings, as well as the selected IP interface used to make the connection, e.g. **LAN** or **VLAN**.
3. Create a SIP program using the front panel via the **Home screen**, or create a SIP program using the HTML5 Toolbox Web-GUI.

**Important Notes:**

- The codec supports dialing over SIP using a registered SIP server account, or peer-to-peer using one of the two SIP interfaces **SIP1** and **SIP 2**.
- SIP dialing is only supported over point-to-point connections.
- Some ISPs and/or cellular networks may block SIP traffic over UDP port 5060.
- Tieline G3 codecs do not support connections using algorithms like AAC, aptX Enhanced and Opus and will default to MPEG Layer 2 if an incoming call is configured to use these algorithms.
- Failover is not available with SIP; SmartStream PLUS redundant streaming is not available with SIP or sessionless IP..
- For detailed information about connecting with other brands of codec using SIP click http://www.tieline.com/interoperability
- When connecting to a Tieline G3 codec using SIP you need to manually select the G3 audio reference level. To do this select **Home screen > Settings > Audio > Reference Level > Tieline G3**. In addition, select the following on the G3 codec prior to dialing.
  - Select either a mono or stereo profile
  - Select **[Menu] > [Configuration] > [IP1 Setup] > [Session Type] > [SIP]**
  - Select **[Menu] > [Configuration] > [IP1 Setup] > [Algorithm] > [G711/G722 or MP2]**

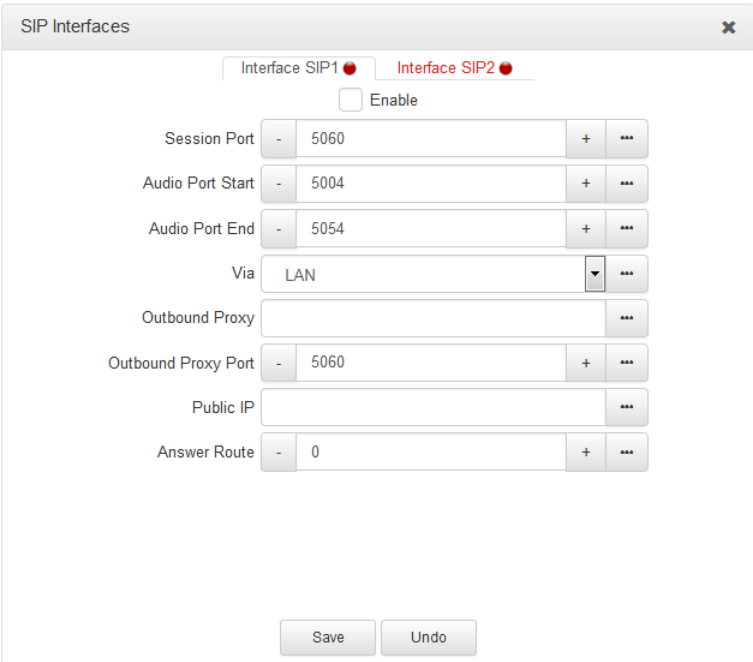
### 16.10.1 Configuring SIP Interfaces

The codec supports dialing over two SIP interfaces simultaneously.

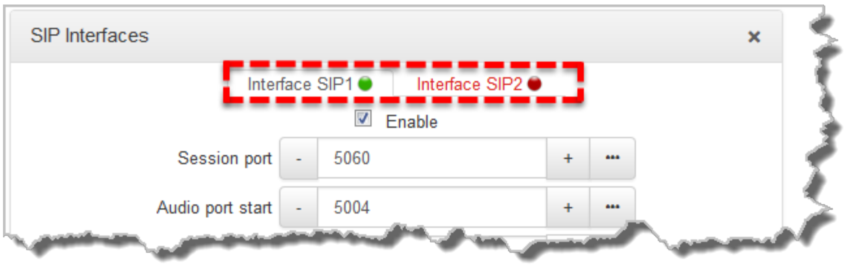
**Important Notes:**
1. SIP interfaces are disabled by default.
2. **SIP1** is configured to use **LAN** by default, which is mapped to the **Primary** Via interface by default.
3. **SIP2** is configured to use **VLAN1** by default, which is mapped to the **Secondary** Via interface by default.
4. **SIP1** and **SIP2** each need to use a separate IP interface when connecting, e.g. **LAN** or **VLAN1**.
5. **SIP1** and **SIP2** can each make multiple SIP calls, e.g. two calls can be made over **SIP1**, or two calls can be made over **SIP2**.
6. The settings for **SIP1** and **SIP2** cannot be edited if the interface is enabled.
7. Enter a public IP address in the **Public IP** menu if you want to dial over SIP from behind a firewall. Then configure port forwarding to route traffic to the codec's local IP address behind your firewall. Note: Do not enter a **Public IP** address if STUN is configured. They cannot be used together because both will attempt to use a public IP address over SIP. STUN settings are prioritized and used if both are configured.

To configure **SIP1** or **SIP2**:

1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then click **SIP Interfaces** to view and configure SIP interface settings.
2. Default SIP settings are configured and select **Interface SIP1** or **Interface SIP2** to adjust each interface. Note: Ensure each interface uses a unique "Via" IP interface because they can't share one, e.g.**LAN**.




3. Select the **Enable** check-box and then click the **Save** button to confirm settings.
4. The SIP interface indicator is green when an interface is enabled and red when it is disabled.




## 16.10.2 Configuring SIP Accounts

Up to 6 SIP accounts can be configured in the codec and registering codecs for SIP connectivity is simple. First, select the SIP server to which you will register your codec. 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:

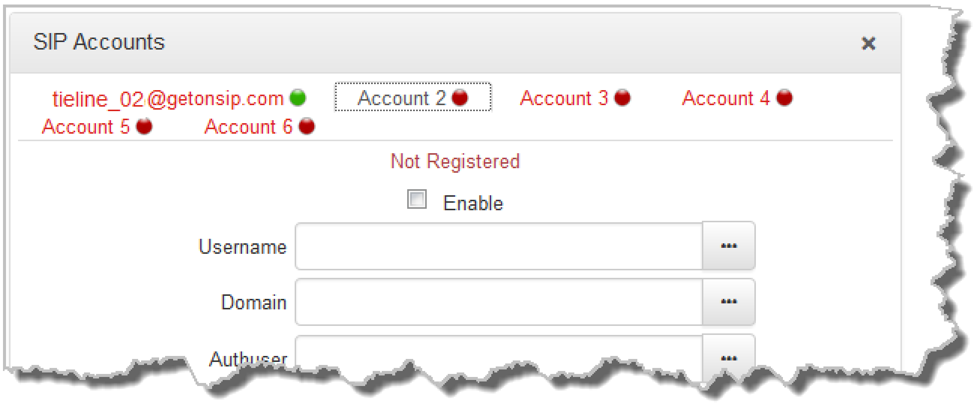
- Username
- Authorized User
- SIP address
- Domain
- Realm
- Registrar
- Registar port
- Outbound Proxy
- Proxy port
- Password
- Registration Timeout (this shouldn't need to be adjusted from the default setting).

**Important Notes:**
- In most situations it is best to configure a SIP account when the codec is configured with a public IP address.
- Each SIP account can only be mapped to a single SIP interface, i.e. **SIP1** or **SIP 2**.
- Up to 6 SIP accounts can be added to the codec.
- To configure a SIP Account using the codec **LCD SCREEN** see Configuring SIP Accounts.

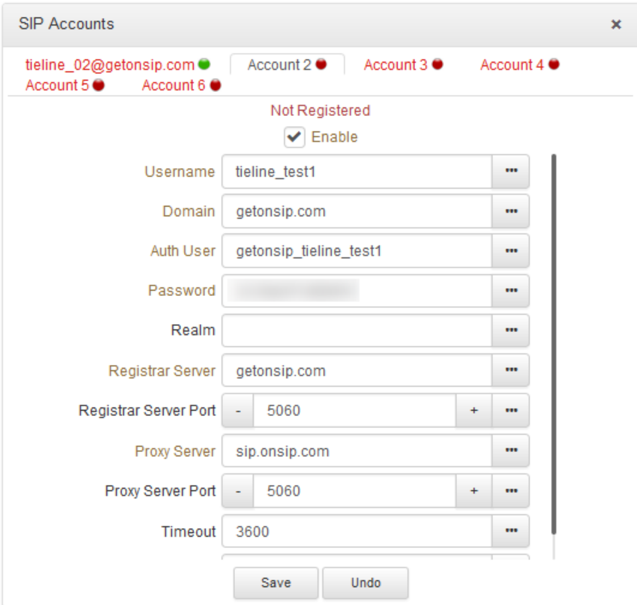
### Adding a SIP Account

Enter SIP account details and register the account in your codec. Once configured, the codec will contact the SIP server automatically to acknowledge its presence over a wide area network when connected to a public IP address.

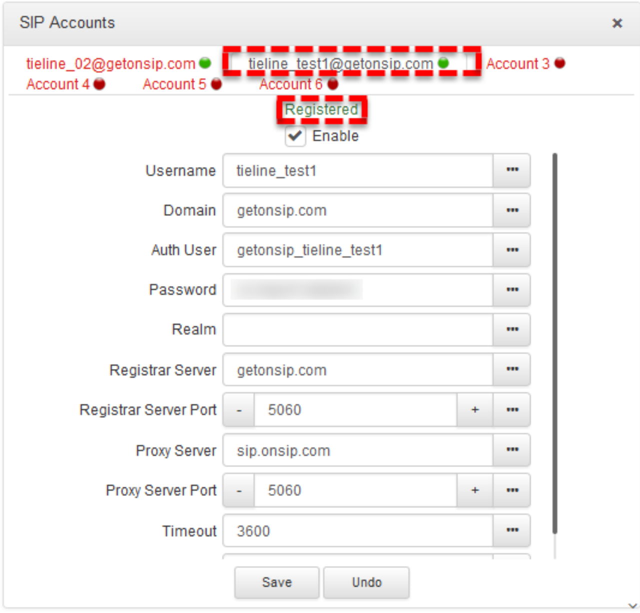
1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then click **SIP Accounts** to view and configure SIP account settings.
2. Click to select one of the unused **Accounts** at the top of the **SIP Accounts panel**.



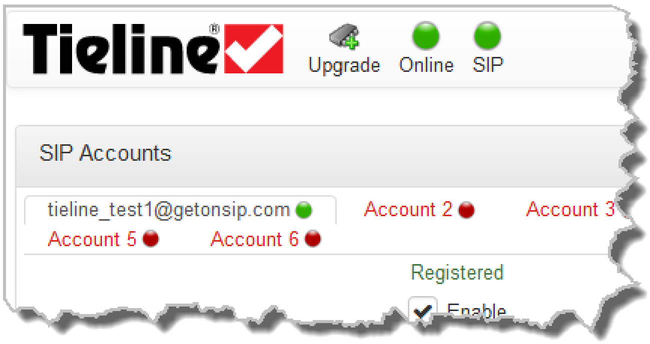

3. Enter the SIP account details into the relevant text boxes, including the registration **Timeout** (which shouldn't need to be adjusted from the default setting). Also ensure a SIP **Interface** is selected (e.g. **SIP1** or **SIP2**.) The SIP interface contains settings related to ports and the selected **Via** interface, e.g.**LAN**. See [Configuring SIP Interfaces](#) for more details.




4. Click the **Enable** check-box at the top of the panel and then click the **Save** button to register the codec to the server.
5. If an account is registered successfully, the account registration indicator changes from red to green, and **Not Registered** (above the **Enable** check-box) becomes **Registered**.




6. In the Toolbox Web-GUI the red **SIP** indicator adjacent to the codec **Online** indicator also changes to green when an account is currently registered in the codec and ready to be used when dialing over SIP.




7. Once enabled, the SIP account can be selected when creating a new SIP connection.

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

## Troubleshooting SIP Registration

If a SIP account is not registering successfully please check the following:

1. Confirm all account registration information has been entered correctly.
2. Confirm the SIP interface (**SIP1** or **SIP2**) configured as the **Via** in the account is enabled.
3. Verify that the **Via** selection in the **SIP1** or **SIP2** interface settings corresponds with the network interface being used by the codec to register the account. E.g. **LAN1** or a **VLAN**.

### 16.10.3 Configure SIP White and Blacklists

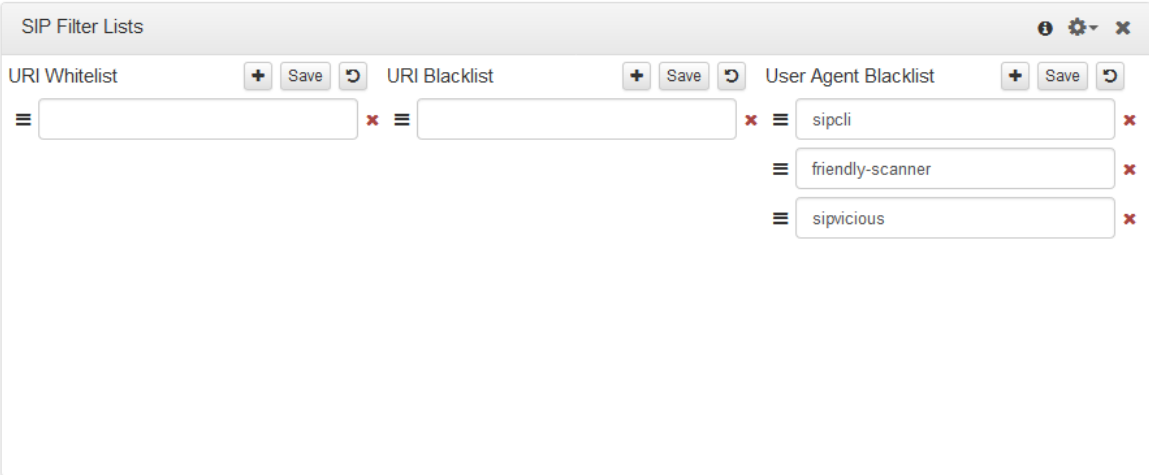
The **SIP Filter Lists panel** allows filtering of SIP URIs and User Agents to provide greater security for codec connections. For example, add trusted network codecs to the **URI Whitelist** in this panel and only codecs using these SIP URIs will be able to connect. This is like saying, "if you have the key you can open the door" and is perhaps the easiest way to filter outside access to your codec's "front door".

It is also possible to add SIP URIs to the **URI Blacklist** and add user agents to the **User Agent Blacklist** to deny them access to the codec. These blacklists also filter unwanted traffic and increase the likelihood of rejecting unwanted traffic. Note: If an incoming SIP caller is not on the URI Whitelist it will be scanned using the **URI Blacklist**. If there is no match it will be scanned using the **User Agent Blacklist**. A connection will be established if there is no match on either Blacklist.

**Important Note:** To only allow a predefined list of codecs to connect, add them to the **URI Whitelist** and add a wildcard (asterisk) * to the **URI Blacklist**: all incoming calls will be blocked except for codecs in the Whitelist.

#### Filter URIs and User Agents

1. Open the HTML5 Toolbox Web-GUI and click **Transport** in the **Menu Bar**, then click **SIP Filter Lists** to launch the **SIP Filter Lists panel**.

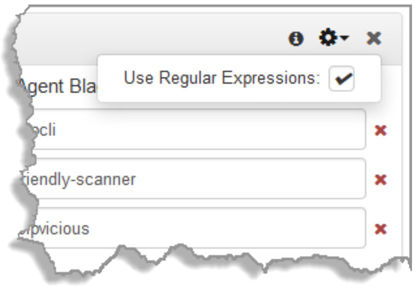



2. Click the **Plus symbol** for **URI Whitelist**, **URI Blacklist** or **User Agent Blacklist** to add a new item to the list.
3. Enter the new item in the text box, click to select the check-box and then click **Save** to store the new setting.
4. Click the **Undo symbol** to undo editing and click and drag the **List symbol** ≡ to shift the position of whitelist and blacklist items.

**Important Note:** From firmware v2.16.xx and later releases the user agent in a Tieline G5 codec is configured as "Tieline <ProductName> <Firmware Version>". E.g. Tieline Bridge-IT v2.18.68

#### Using Regular Expressions

To filter using regular expressions in the **SIP Filter Lists panel**, click the **Options symbol** in the top right-hand corner of the panel and then click to select the **Use Regular Expressions** check-box.




**Important Note:** When regular expressions are used, the $ ^ to represent the beginning and end of a match is not supported, because searches implicitly try to match anywhere in the line.

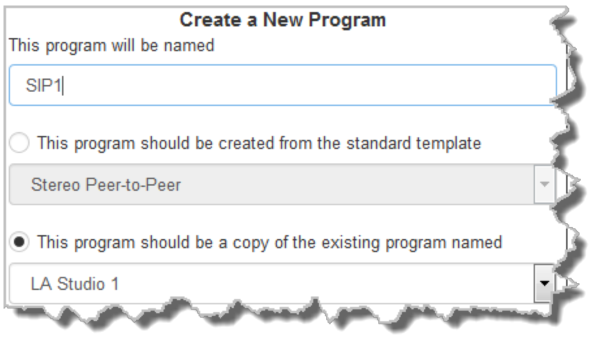
## 16.11 Configure Peer-to-Peer SIP Programs

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

**Important Notes:** Before you start program configuration please note:

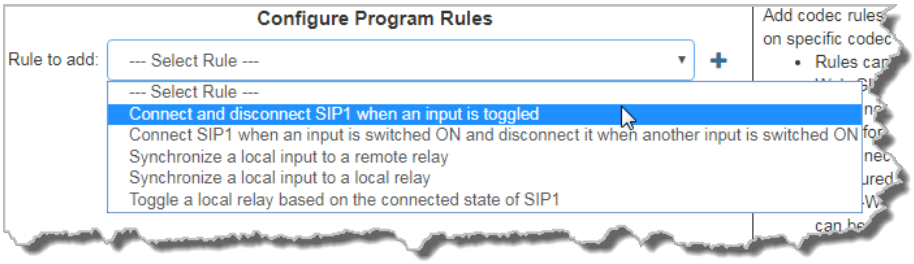
- You cannot edit a program when it is currently loaded in the codec.
- Some drop-down menus and settings may be greyed out intentionally depending on features available.
- Failover is not available with SIP; SmartStream PLUS redundant streaming is not available with SIP or sessionless IP.
- To learn more about programs see the section titled About Program Dialing.Remember to lock an answering program in a codec when answering multiple SIP calls.
- Ensure the appropriate TCP and UDP audio ports are open in your firewall to allow SIP audio streams to connect. See Installing the Codec at the Studio for more information.

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click the **New Program** button to open the wizard and:
   - Click in the **Program Name** text box to name the new program.
   - Select **Mono/Stereo Peer-to-Peer,** or if you want to use an existing program as a template, select this option. Then click **Next**.




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

3. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** to add a new rule and click the **Minus symbol** to remove a rule.

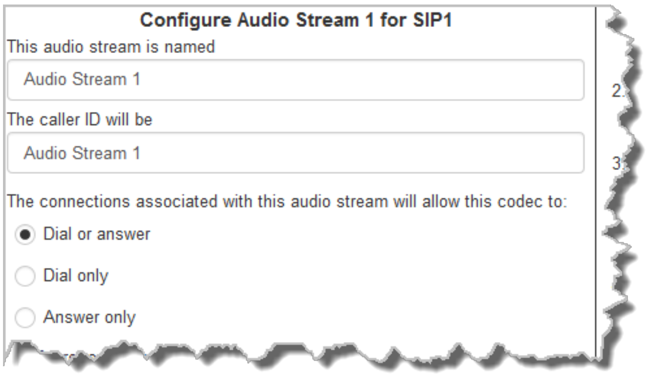



**Important Notes for Rules:**

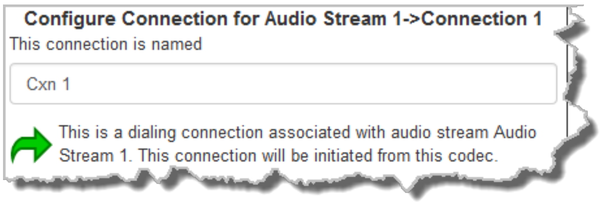
- Bridge-IT XTRA has 4 physical **CONTROL PORT GPIOs**; 7 Tieline and WheatNet virtual inputs (1-7); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network).
- Bridge-IT has 2 physical **CONTROL PORT GPIOs**; 5 Tieline and WheatNet virtual inputs (1-5); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network).
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- Tieline WheatNet-IP codecs require Wheatstone Razor firmware version 1.4.22 or later to support WheatNet LIOs. In addition, the WheatNet-IP codec must have the **WNet Enable LIO** checkbox selected in the **Options panel** of the HTML5 Toolbox Web-GUI.
- Relay reflection is not available for SIP and Multicast Client programs.
- Connection-related rules are not displayed in **Answer only** audio streams.
- Program level rules intended to activate dialing are not valid in **Answer only** programs or audio streams.
- For more details about rules see Creating Rules.

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

Note: The following example will display how to configure a dial and answer program. If you want the codec to either dial or answer only, select the option and the wizard will automatically display screens to allow you to configure the codec correctly. Please note that caller ID, dial routes and G3 profile or G3 channel information can not be used for SIP connections because Tieline session data is replaced by SDP for SIP connections.



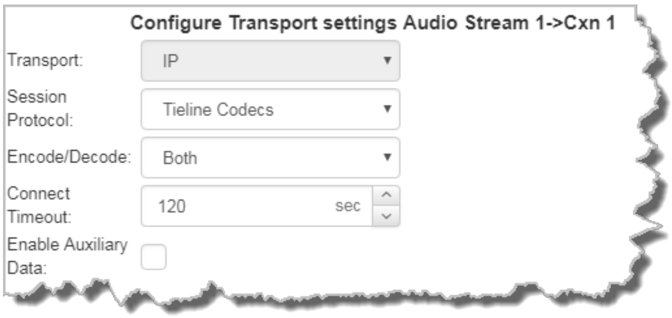

5. This audio stream connection in the wizard will allow the codec to dial. Enter the name of the connection in the text box, then click **Next**.



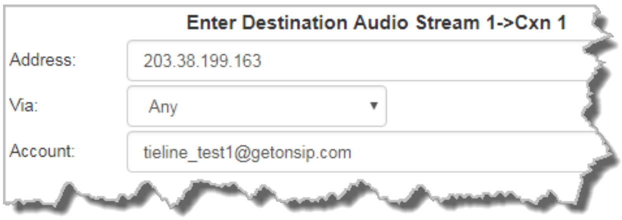

6. Follow the instructions on the right-hand side of the panel to configure the transport settings for the connection: Ensure that you select:

- **IP** as the **Transport**.
- **SIP** from the **Session Protocol** menu option.

Then click **Next**.



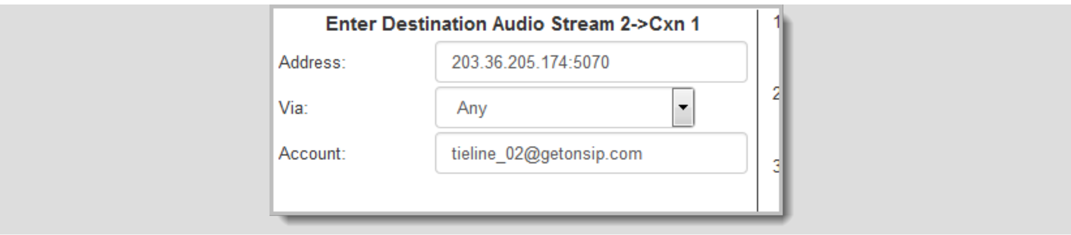

7. Configure the destination codec **Address** if you are dialing peer-to-peer, then specify the network interface used to dial the connection, e.g. **Primary** (Ethernet port **LAN**). Enter the name of a registered SIP account if you are using a SIP server to establish a connection. If you wish to dial from one of the codec's registered accounts, then enter the account name in the Account field using the format accountname@sipserverdomain, e.g. tieline_test1@getonsip.com. In this configuration the specified account's interface will be used rather than the specified **Via**, e.g. if the account is using **SIP2** and **SIP2** is using a VLAN then the call will proceed using the VLAN. If you do not wish to use an account for the dial then leave the **Account** field blank and select the required interface. Note: the interface must be associated with either **SIP1** or **SIP2** for the call to be able to proceed.



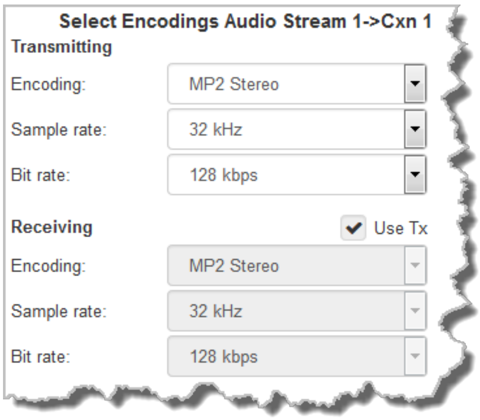

At this point you can click **Save Program** and save the program with default algorithm and jitter settings. Alternatively, click **Next** to confirm and specify algorithm and jitter settings for this connection and configure backup audio settings (recommended).

**Important Notes:**

- The default UDP audio port when using SIP for a peer-to-peer connection is 5004 in Tieline codecs. To contact a codec that is behind a firewall or NAT-enabled router, it is essential that this and all other relevant ports are open and forwarded to the other device.
- Tieline codecs automatically add "**sip:**" to the address you enter in the **Address** field when dialing, so it's not necessary to add this.
- Enter the IP address or SIP URI, then a full colon and the session port number to change the session port from the default setting 5060.

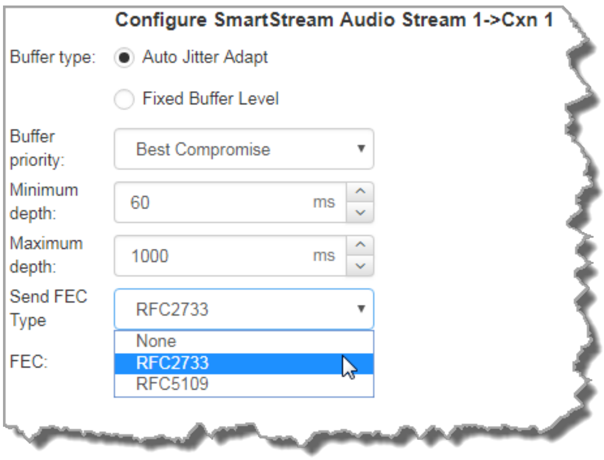



8. Click the drop-down arrows on the right-hand side of each active drop-down menu to adjust the **Encoding, Sample rate** or **Bit rate** parameters. Click **Next** to continue.

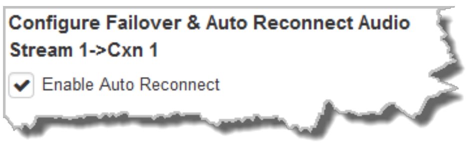



9. Click to configure:

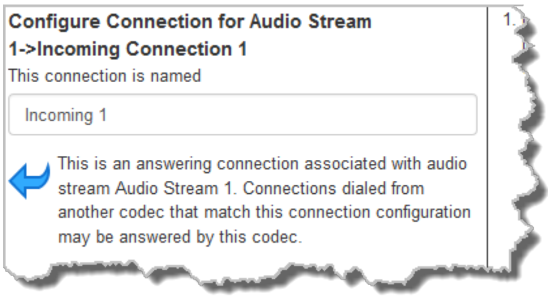
- **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**. It is also possible to configure the **Minimum depth** and **Maximum depth** of jitter over the connection. See Configuring the Jitter Buffer for more details.
- Alternatively, select a **Fixed Buffer Level** and enter the **Jitter Depth**, which must be between 12ms and 5000ms depending on the algorithm you select.
- RFC-compliant FEC can also be configured if required and the percentage is configurable.



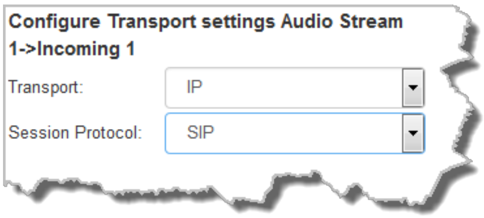

10. Click **Next** to select the check-box if you want to **Enable Auto Reconnect**.



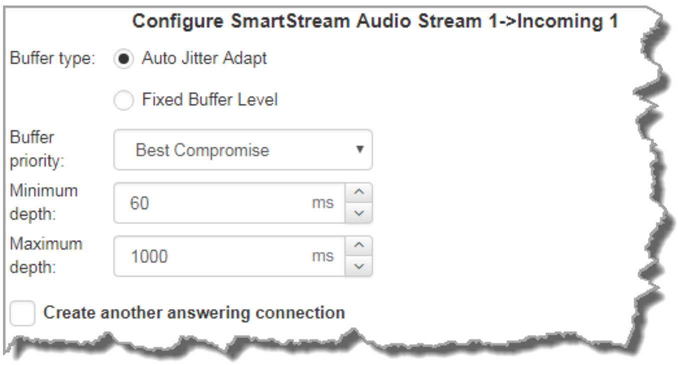

11. Click **Next** to name the answering connection for when calls are received by the codec.




12. Click **Next** to configure the **Session Protocol** as **SIP** for the answering connection to receive a SIP call.




13. Click **Next** to configure the jitter buffer settings for the answering connection.

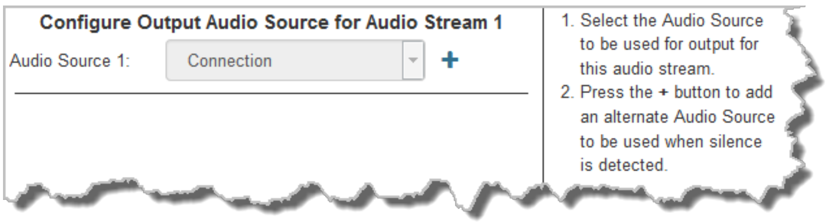



14. After configuring all settings there are 3 options:

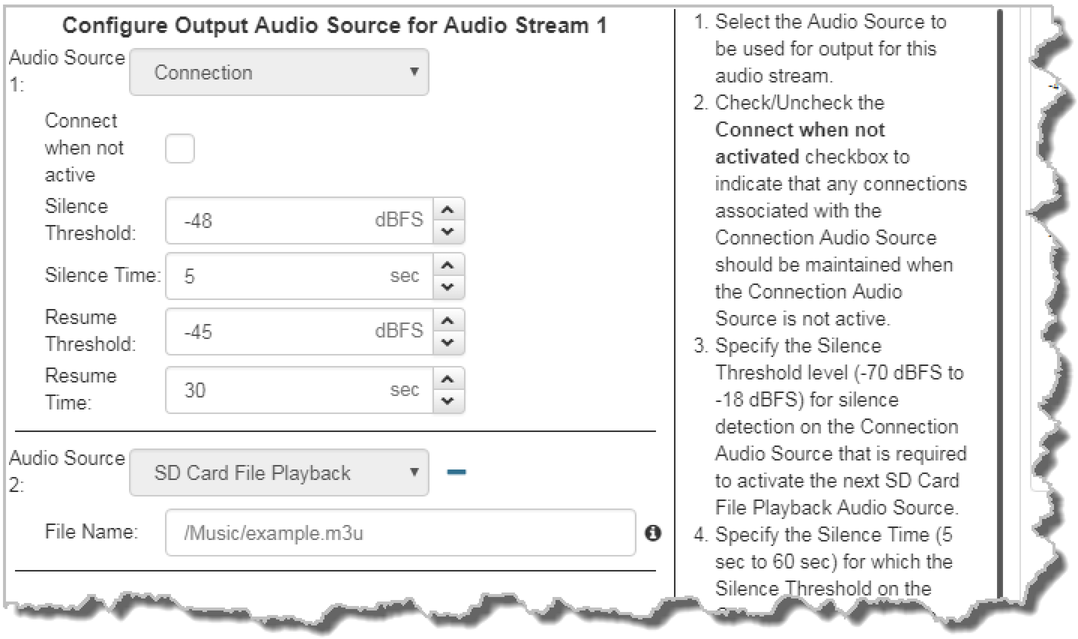
    i. If you want to create another answering connection, select the check-box for **Create another answering connection** and continue through the wizard.
    ii. Click **Save Program** to save the program at this point.
    iii. Click **Next** to configure **Output Audio Source** options.

## Configuring SD Card File Playback Options

1. Click **Next** to configure SD card file playback using **Output Audio Source** configuration options to maintain program audio at transmitter sites.




2. Click the blue **Plus symbol** to add the **SD Card File Backup Output Audio Source**. Click the **Minus symbol** to remove it. Follow the instructions on the right-hand side of the panel to configure silence and resume threshold parameters.




3. After configuring **Output Audio Source** options you can:

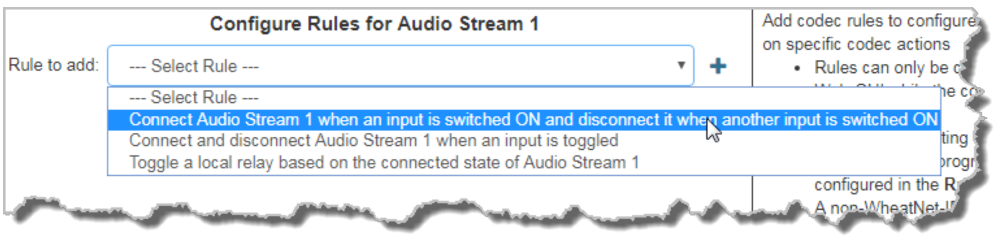
   i. Click **Save Program** to save the program at this point.
   ii. Click **Next** to configure rules options.

**Important Notes for File Playback:**

- A single partition FAT32 formatted SDHC Card is required (SD cards may be less reliable and are not recommended).
- The codec supports SDHC cards which have a physical capacity of up to 32GB.
- Create MP2 or MP3 files using a 32kHz, 44.1kHz or 48kHz sample rate.
- Ensure recordings used are not variable bit rate files.
- SDHC file audio is not sent to codec encoders and cannot be transmitted via an audio stream to another codec.
- File playback audio is sent directly to the codec outputs and therefore IGC is not available. When you create your MP2 or MP3 files ensure the audio levels match the audio reference level of your codec and that peaks average at the correct levels.
- If you create a single file name ensure you add the file extension, e.g. "test.mp3", or the file will not play back.
- If you create a directory name, all the files within the directory will be played back. We recommend you save all audio files as a playlist and link to this if you want them to play out sequentially. Please note that "M3U" is the playlist file format supported by the codec.

## Configuring Rules

1. To configure new rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** to add a new rule and click the **Minus symbol** to remove a rule.



**Important Note:** Program level rules intended to activate dialing are not valid in **Answer only** programs or audio streams.

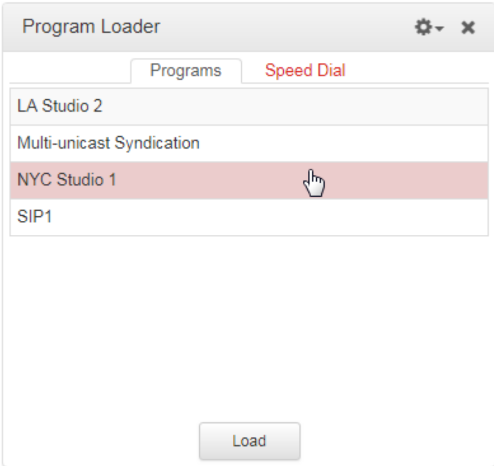
2. Click **Save Program** to save the program.
3. Click **Finish** to exit the wizard.
4. The newly created program will be displayed in the left pane within the **Program Manager panel** and in the **Program Loader panel**. Select and connect audio streams in a program using the **Connections panel**, or dial the program manually using the codec front panel.

# 16.12 Load, Unload and Dial a Program

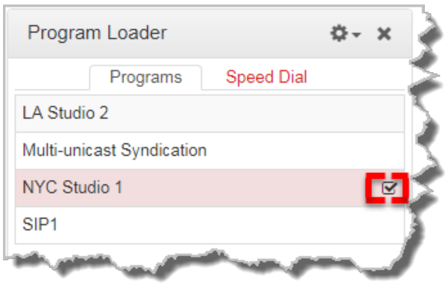
To load and dial a program it is necessary to open the **Program Loader panel** and **Connections panel**. Use the **Program Loader panel** to load a program and then manage connecting and disconnecting using the **Connections panel**.

## Loading a New Program

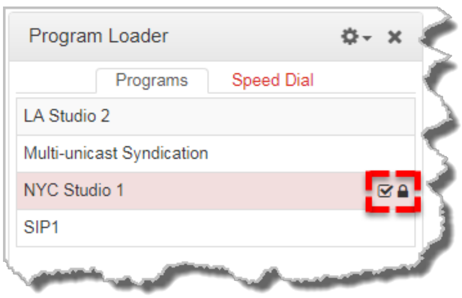
1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then click **Program Loader** to open the **Program Loader panel**.
2. Click to select a program in the **Program Loader panel** and then click **Load** to load the program in the codec.




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



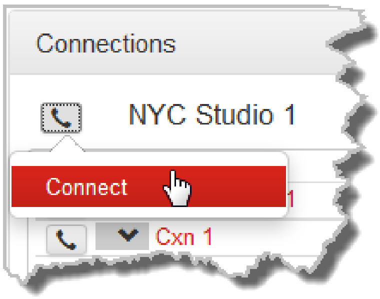

If **Lock Loaded User Program** has been configured in the **Options panel**, a black **Padlock** symbol appears next to the program name in the **Program Loader panel**, to indicate a program is locked in the codec.



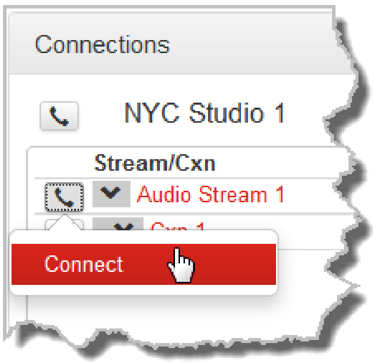
## Connecting a Program

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

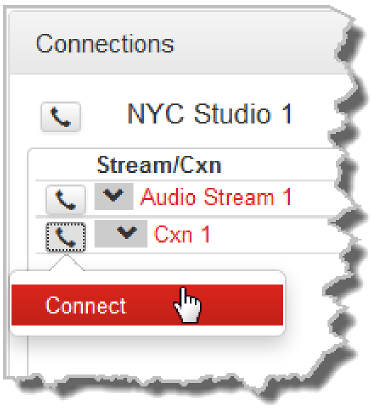
1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then click **Connections** to open the **Connections panel**.
2. Click the program **Connect/Disconnect** symbol and then click **Connect**; this connects all active audio streams and connections associated with the program.




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



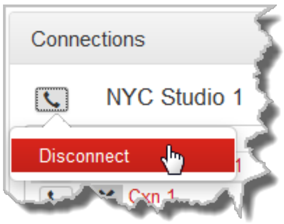

4. Click the connection **Connect/Disconnect** symbol and then click **Connect**; this connects an individual audio stream connection.



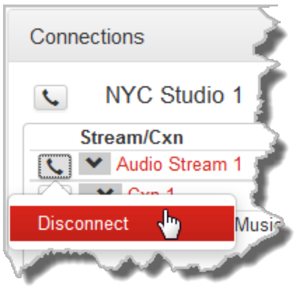
## Disconnect a Program

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

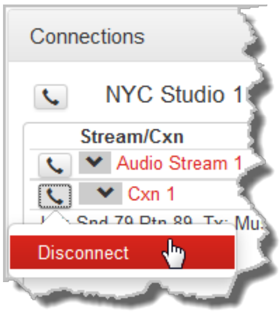
1. Click the program **Connect/Disconnect** symbol and then click **Disconnect** to disconnect a program; this includes all audio streams and connections associated with the program.




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

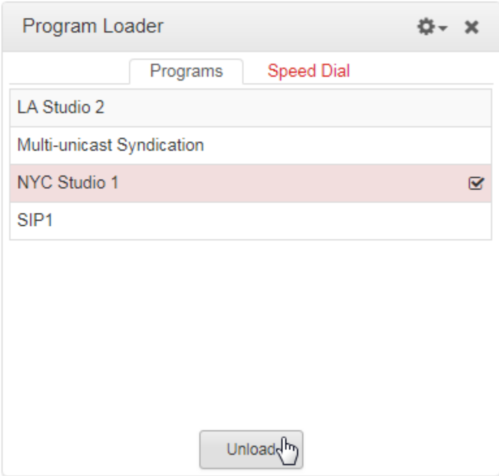



3. Click the connection **Connect/Disconnect** symbol to disconnect an individual audio stream connection.



### Unloading a Program

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



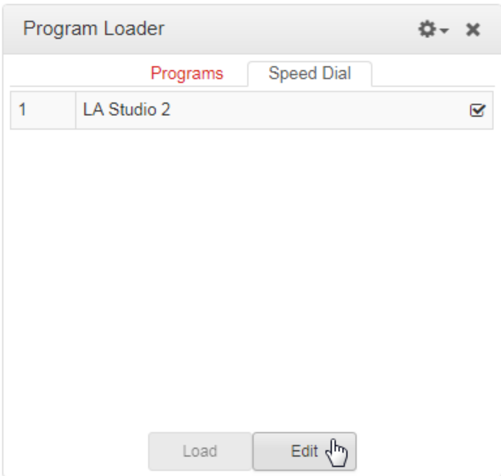

## 16.13 Configure Speed Dialing

The Toolbox HTML5 web-GUI supports dialing programs using preconfigured speed dials. This is configured and managed using the **Program Loader** panel.

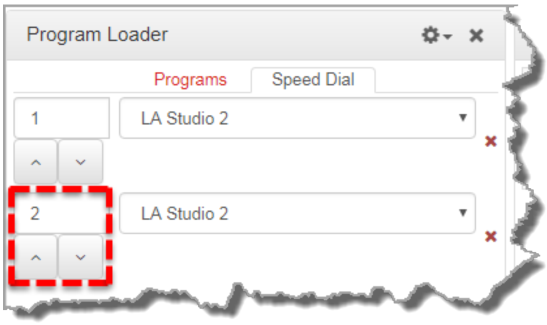
### Create a New Speed Dial

To configure speed dialing:

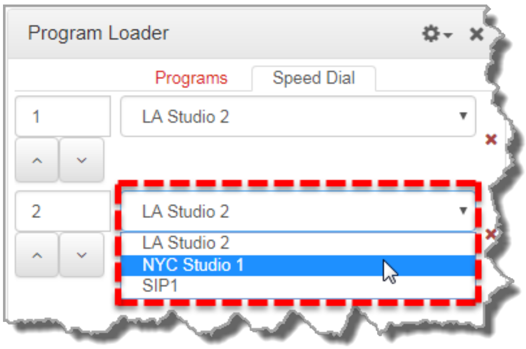
1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then click **Program Loader** to open the **Program Loader panel**.
2. Click **Speed Dial** and then **Edit**.




3. Click **New** to create a new speed dial and then click the drop-down arrows on the left side of the panel to select the speed dial number.




4. Click the drop-down arrow on the right side of the panel to select the program to associate with the speed dial number.

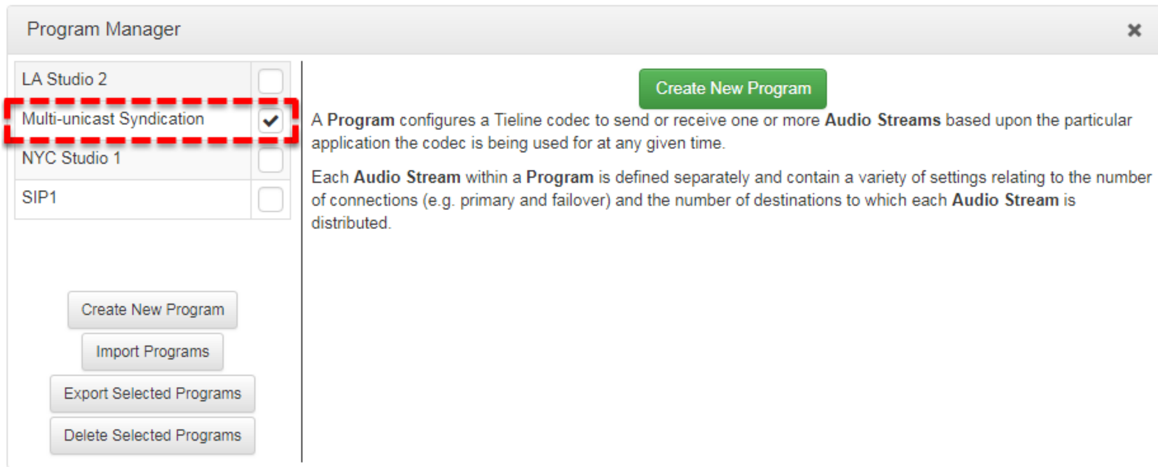



5. Click **Save** to store newly configured Speed Dials.

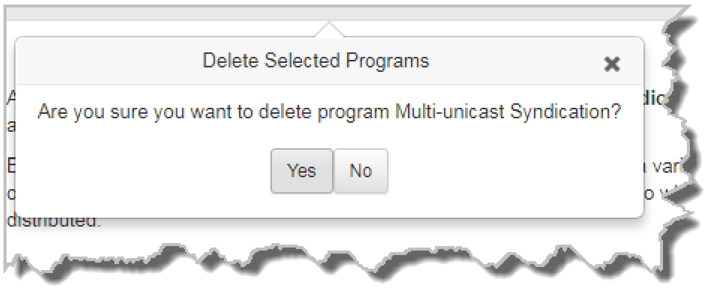
**Important Note:** To view speed dial configurations using the front panel of the codec select **SETTINGS** **> Speed Dial** and press the button.

## 16.14 Delete a Program

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then click **Program Manager** to open the **Program Manager panel**.
2. Click to select the check-box for each program to be deleted. Note: multiple programs can be selected and deleted simultaneously.



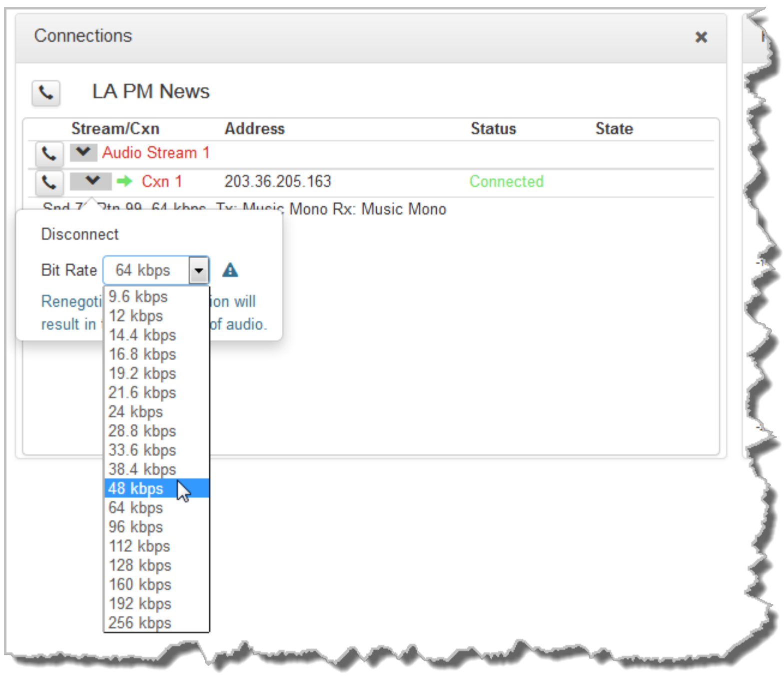

3. Click the **Delete Selected Programs** button.
4. Click **Yes** in the **Delete Selected Programs** confirmation dialog to delete all selected programs.




**Important Notes:** Any program that is currently loaded or listed in the **Sheduler Events panel** cannot be deleted.

## 16.15 Adjusting the Connection Bit Rate

1. Open the HTML5 Toolbox Web-GUI and click **Connect**, then select **Connections** to open the **Connections panel**.
2. Click **Connect/Disconnect** symbol for an active connection and then click the drop-down **Bit Rate** arrow to select and renegotiate a new bit-rate.

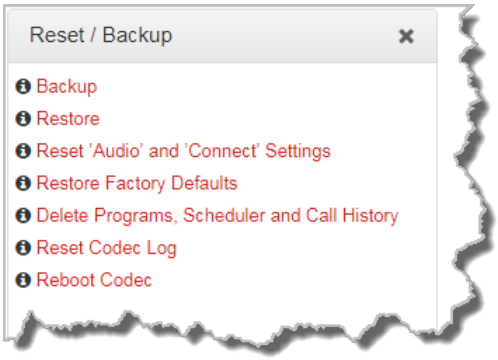



**Important Notes:** It is not possible to renegotiate the connection bit rate of a SIP connection.

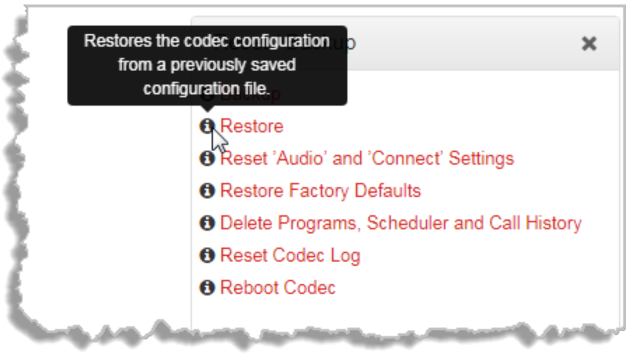
## 16.16 Reset Factory Default Settings

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

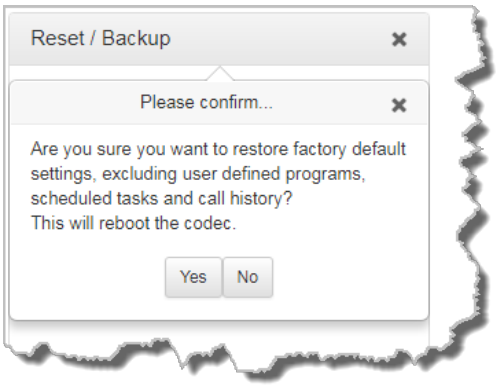
1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Reset / Backup** to display the **Reset / Backup panel**.




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




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



## 16.17 Backup and Restore Functions

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

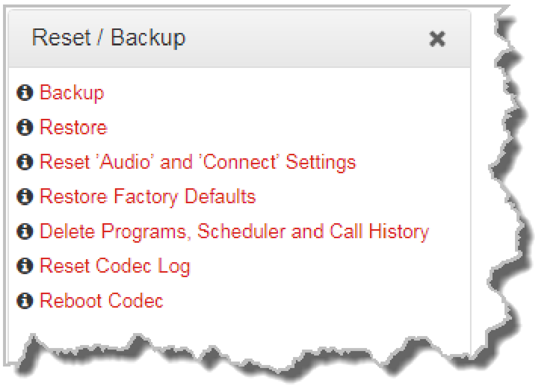
- Programs containing a variety of connection settings.
- All system settings that have been adjusted to change the factory default codec settings (current runtime settings).

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

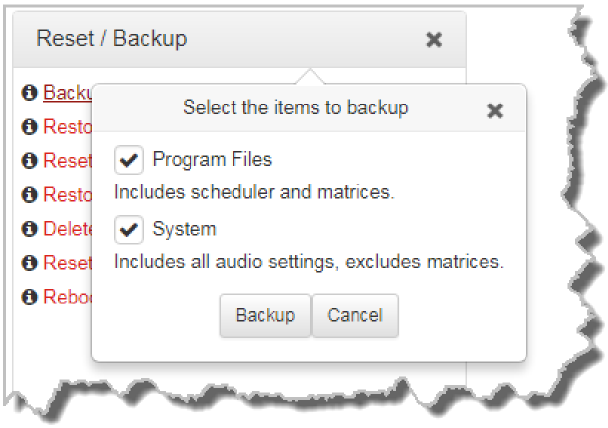
- Program, audio stream and connection names.
- IP address, port, algorithm, jitter buffer, FEC and bit rate settings (etc.) for audio stream connections.

### Creating Backup Files

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




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

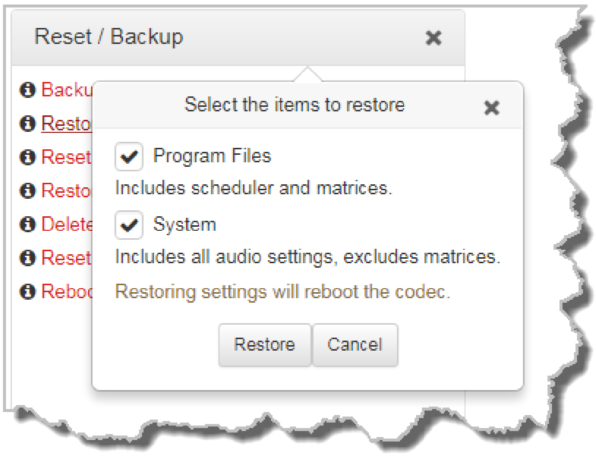



4. Select a location on your PC to save the configuration file. Note: You may need to "allow" your browser to display the pop-up dialog.

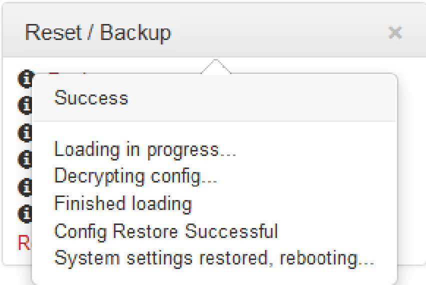
### Restoring Configuration File Settings

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

2. Click **Restore**.
3. Click to select the check-boxes and confirm your restore settings. For example, you could select the **Program Files** check-box and deselect the**System** check-box if you are only copying programs onto codecs.




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



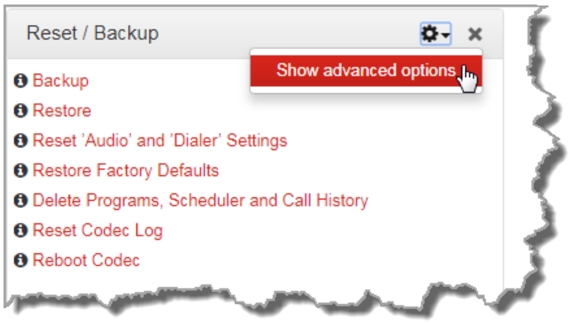

Note: The codec will automatically reboot when restoring system settings.

## Advanced Settings: XML Config

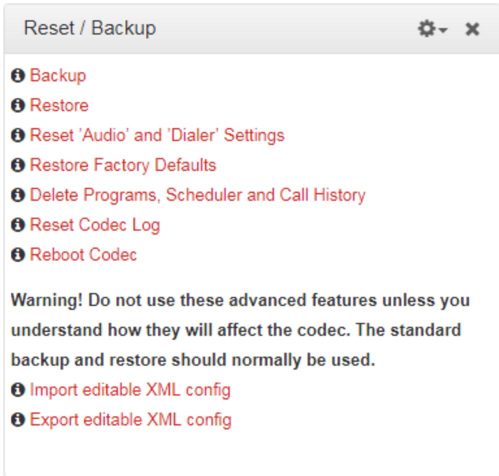
**Caution:** DO NOT use advanced XML configuration features unless you fully understand how they will affect the codec. The standard backup and restore function should normally be used. Damage to the codec may occur if this feature is used without fully understanding how it will affect the codec.

XML Config is a highly advanced feature which should only be performed by suitably qualified personnel. To import or export XML config files:

1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Reset/Backup** to display the **Reset/Backup panel**.
2. Click the **Options symbol** to view **Show Advanced Options**.




3. Click to select **Import/Export editable XML config** as required.

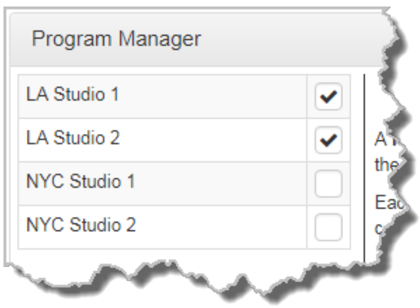



# 16.18 Import and Export Programs

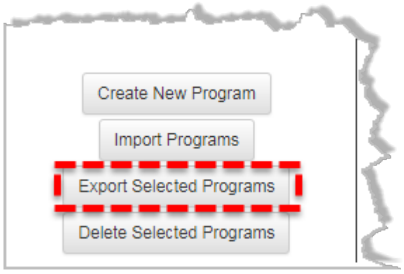
It is possible to import and export individual programs via the **Program Manager panel**.

## Exporting Programs

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click to select the check-box for the program or programs you wish to export.



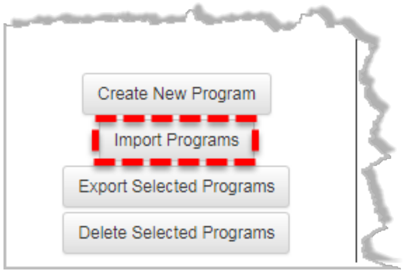

3. Select **Export Selected Programs** in the bottom-left corner of the **Program Manager panel**.




4. Navigate to a file folder and save the program .zip file.
5. Click **Save** to save the program file.

### Importing Programs

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Select **Import Programs** in the bottom-left corner of the **Program Manager panel**.

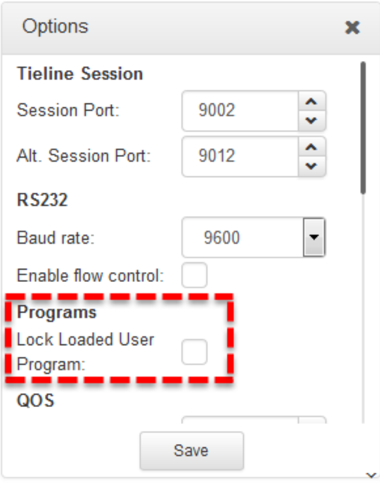



3. Navigate to the file folder containing the program .zip file to be imported.
4. Click to select the .zip file and click **Open** to import it.

## 16.19 Lock or Unlock Programs

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

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




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

**Important Note:**

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

# 16.20 HTML5 Software License Installation

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

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

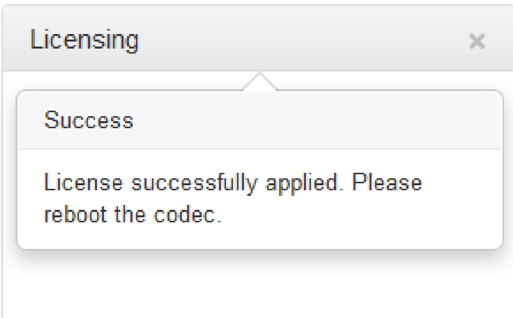
1. Open the HTML5 Toolbox web-GUI in a browser on your PC by typing either the IP address of the codec (LAN connection), or the USB address of the codec (USB connection) into the address bar.
2. Ensure you have unloaded any currently loaded program in the codec via the **Program Loader panel**.
3. Click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


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

## Download a License File and Install Manually

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

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




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

# 16.21 Configuring SNMP in the Codec

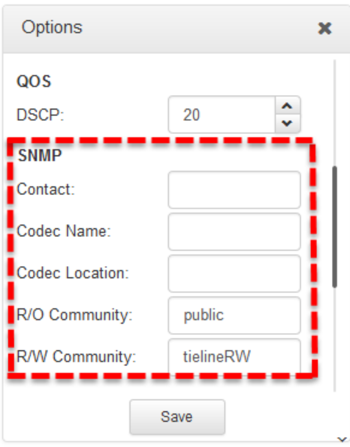
The codec supports Simple Network Management Protocol (SNMP ) for managing devices on IP networks.

## Description of SNMP Settings in the Codec

| Features | Operation Button Descriptions |
|---|---|
| **Codec Name** | A user-specified alphanumeric identifier which may be used by third-party SNMP software to identify a device. The device name corresponds to the ".iso.org.dod.internet.mgmt.mib-2.system.sysName" SNMP attribute and is completely independent of DNS, NIS, WINS or other device naming and identification schemes, though convention is to use the device's fully-qualified domain name. |
| **Codec Location** | A user-specified alphanumeric string which may be used by third-party SNMP software to identify a device. Device location corresponds to the ".iso.org.dod.internet.mgmt.mib-2.system.sysLocation" SNMP attribute. |
| **Contact** | A text identifier for the contact person for this managed node, together with information on how to contact this person. |
| **R/O Community** | SNMP provides two types of access, namely Read-Only access and Read-Write access. The R/O Community identifier allows Read Only level access. |
| **R/W Community** | The R/W Community identifier allows Read/Write level access. |

## Configuring SNMP Settings in the Codec

1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Options** to display the **Options panel**.
2. Click the **Edit** button to configure settings.
3. Click in the text boxes to enter SNMP configuration settings.




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

## MIB Files for SNMP Configuration

Management Information Base (MIB) files are required for SNMP applications to interact with your Tieline codec and interpret SNMP data. The codec supports SNMPv1 and SNMPv2 MIB protocols. The required MIB files can be downloaded from the codec using the following link in a PC web browser connected to the same network as your codec:

- http://<YOUR_CODEC_ADDRESS>/mibs/tieline-mibs.zip

Save the .zip file to your PC and import the contents into the MIB browser you use to manage SNMP-enabled network devices.

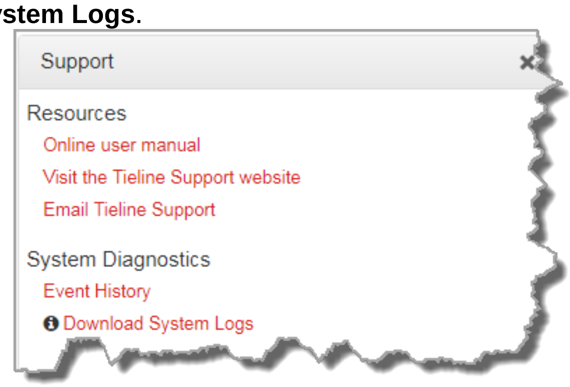
**Important Note:** The codec supports the attributes specified in the MIB-II standard. Please verify that your SNMP software contains the required files as specified in RFC 1213.

## 16.22 Download Logs

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

### Procedure for Sending Logs to Tieline

1. Open the HTML5 Toolbox Web-GUI and click **Help** in the **Menu Bar**, then click **Support**.
2. Click **Download System Logs**.




3. Save the file to your computer and then send the .zip file to Tieline support at support@tieline.com

### Download Event Logs

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

1. Open the HTML5 Toolbox Web-GUI and click **Help** in the **Menu Bar**, then click **Support**.
2. Click **Event History** to view the event log in a new web-browser window.

### Clearing Logs

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

## 16.23 RS232 Data Configuration

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

| Algorithm Selected | IP Transport |
|---|---|
| **Tieline Music and MusicPLUS** | • In-band RPTP data is enabled automatically<br>• Synchronized out-of-band data can be enabled and disabled as required<br>• Using out-of-band data with rules between G5 codecs employing relay reflection minimizes latency<br>• These algorithms must be used when connected to G3 codecs as they don't support out-of-band data |
| **All other algorithms** | • No in-band data available; synchronized out-of-band data can be enabled and disabled |

Select **Enable Auxiliary Data** when creating a program in the **Program Manager panel** to enable out-of-band data and activate rules employing relay reflection over a connection. This will allow the codec to connect to external devices and send RS232-compatible data via the serial port on the rear panel. Alternatively, enable auxiliary data using the **Setup** menu (see Enabling RS232 Data).

### Setting RS232 Data Rates and Flow Control

1. Open the HTML5 Toolbox Web-GUI and click **Settings** in the **Menu Bar**, then click **Options** to display the **Options panel**.
2. Click the **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.
3. Click to select the **Enable flow control** check box and enable flow control, then click **Save** to store the new settings.


**Important Notes:**

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

## 16.24 Creating Rules

Codec 'rules' configure events based on specific codec actions. A range of default rules are preprogrammed into the codec to facilitate activation of the most common events required by broadcast engineers. Typically rules are based on a change in the state of a physical **CONTROL PORT GPIO**, or a WheatNet-IP logic IO, or a codec program being connected or disconnected. There are three categories of rules:

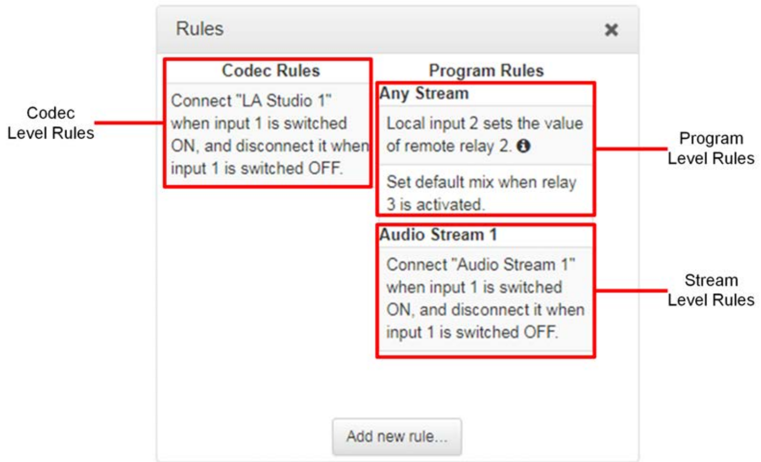
1. Codec level rules: Rules based on programs or codec hardware and software I/O states, e.g. Connect or disconnect a program when an input is toggled, or synchronize a local input to a remote relay.
2. Program level rules: Rules based on codec behaviors at the program level, e.g. Connect and disconnect program A when an input is toggled, or synchronize a local input to a remote relay.
3. Stream level rules: Rules based on codec behaviors at the stream level, e.g. Synchronize a local input to a remote relay.

There are three ways to create rules in the HTML5 Toolbox Web-GUI:

1. **Rules panel**: Configure codec level rules related to programs and/or hardware and software I/O states.
2. **Program Manager panel**: Configure program level rules early in the **Program Manager panel** wizard.
3. **Program Manager panel**: Configure stream level rules for each audio stream as you proceed through the **Program Manager panel** wizard.

**Important Notes:**
- Rules can only be created with the Web-GUI while the codec is disconnected.
- Program and stream level rules configured in the **Program Manager panel** are only active when the program is loaded.
- Bridge-IT codecs only support a single mono or stereo IP audio stream and therefore Program level rules and Stream level rules are essentially identical.

Following is a summary of how codec, program and stream level rules are displayed in the **Rules panel** when configured.




### Enabling Data

**Data** is disabled by default and must be enabled to allow contact closure operation and transmission of RS232 data. Select **Enable Auxiliary Data** when creating a program in the **Program Manager panel** to enable RS232 data and activate rules employing relay reflection over a connection. It is also possible to enable data using the front panel of the codec:

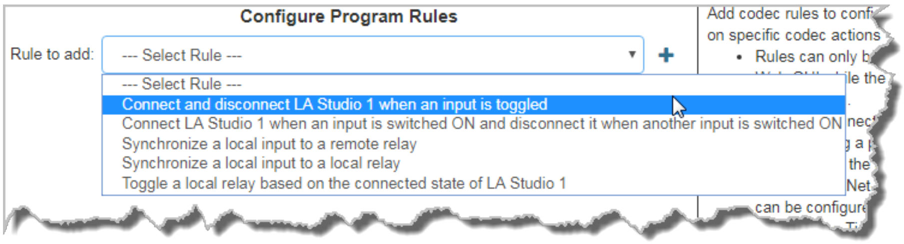
1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select your preferred **IP Mode** and press the button.
5. Use the down navigation button to select **Setup**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

## Configure Rules with the Program Manager Panel

To configure program or stream level rules follow the instructions in this user manual for setting up connections.

### Program Level Rules

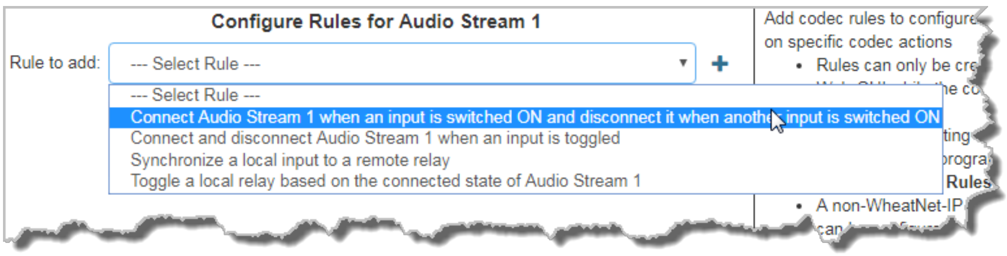
In the **Program Manager panel** wizard use the **Configure Program Rules** screen to configure program level rules. The rules available are displayed in the following image.




Note: Rules intended to activate dialing will not be valid in **Answer only** programs or audio streams.

### Stream Level Rules

In the **Program Manager panel** wizard use the **Configure Rules for Audio Stream** screen later in the wizard to configure stream level rules. The rules available are displayed in the following image.

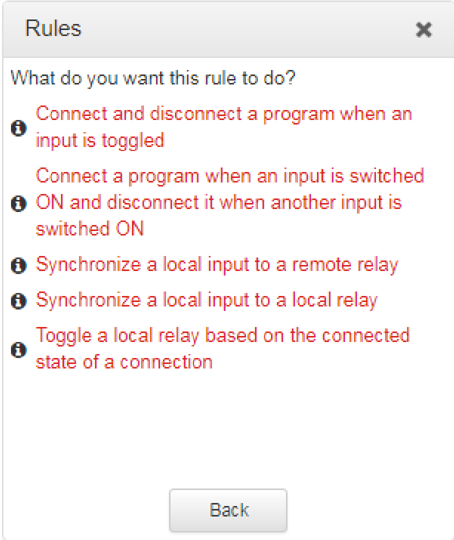



Note: A subset of filtered rules will be displayed for an **Answer only** audio stream connections.

## Configure Rules using the Rules Panel

Use the **Rules panel** to configure codec level rules related to programs and/or hardware and software I/O states.

1. Open the Toolbox HTML5 Web-GUI and click **Rules** in the **Menu Bar** to display the **Rules panel**.
2. Click **Add New Rule**.
3. Click to select the appropriate rule for your requirements. See the Rules panel section in Using the Toolbox HTML5 Web-GUI for an explanation of the action each rule can perform.

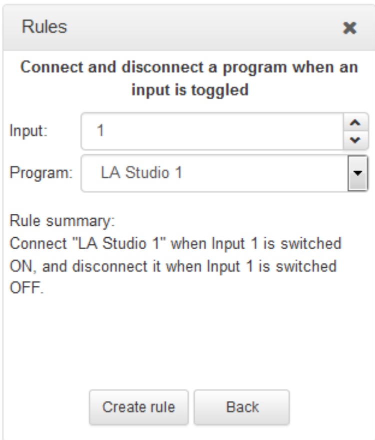



Note: When rules have been configured previously they are displayed when the **Rules panel** is opened.

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

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

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



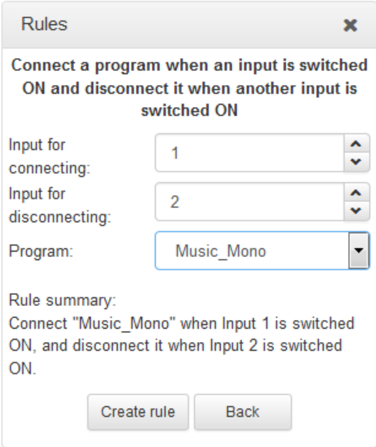

4. Check the **Rule Summary** and click **Create Rule** to save the settings.

## Rule 2: Use Control Port Inputs to Connect and Disconnect a Program

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

1. Click the second rule in the **Rules panel** titled **Connect a program when an input is switched ON and disconnect it when another input is switched ON**.
2. Click the drop-down arrows to select the control port input used to connect and the alternative input for disconnecting.

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

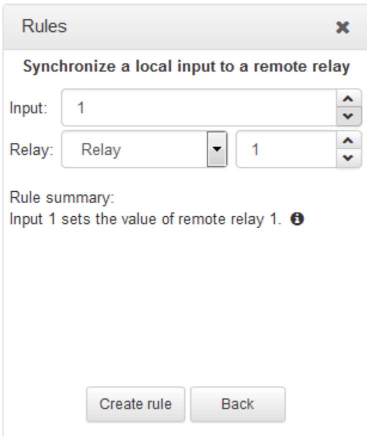



4. Check the **Rule summary** and click **Create Rule** to save the settings.

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

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

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

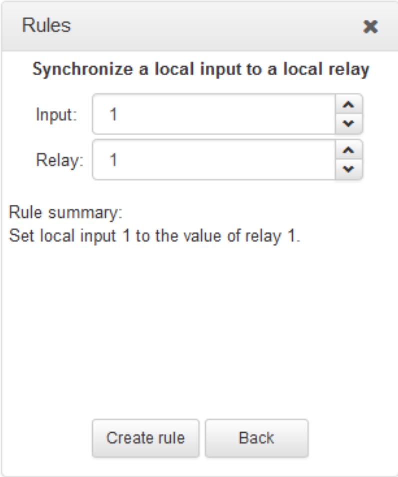



3. Check the **Rule summary** and click **Create Rule** to save the settings.

## Rule 4: Synchronize a Local Input with a Local Relay Output

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

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

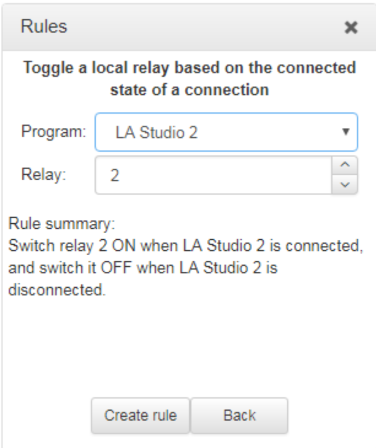



3. Check the **Rule summary** and click **Create Rule** to save the settings.

## Rule 5: Toggle a Relay Output with each Change in Connection Status

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

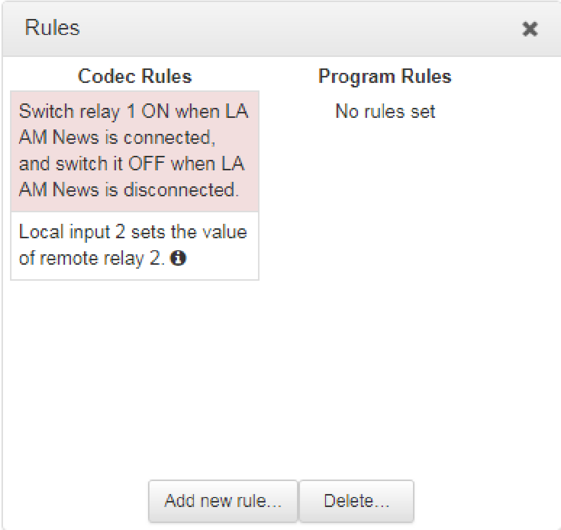
1. Click the fourth rule in the **Rules panel** titled **Toggle a relay based on the connected state of a connection**.
2. Click the drop-down **Relay** arrow and select the relay output you want to toggle.
3. Click the drop-down **Program** arrow to select a specific program which will affect the relay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




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

## Deleting Rules

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

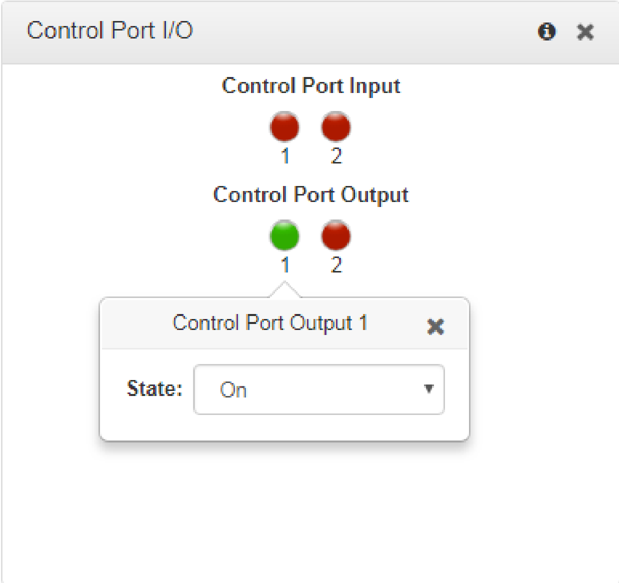



4. Click **Yes** in the confirmation dialog.

## 16.25 Monitoring Control Port I/O Status

It is possible to monitor the status of the two control port inputs and two opto-isolated outputs available using the **CONTROL PORT IN/OUT** connector. To monitor status:

1. Open the Toolbox HTML5 Web-GUI and click **Control** in the **Menu Bar**, then click **Control Port I/O** to display this panel.
2. Click on an output to change the state from **Off** to **On**, then click **Save**. Note: Input states cannot be changed.



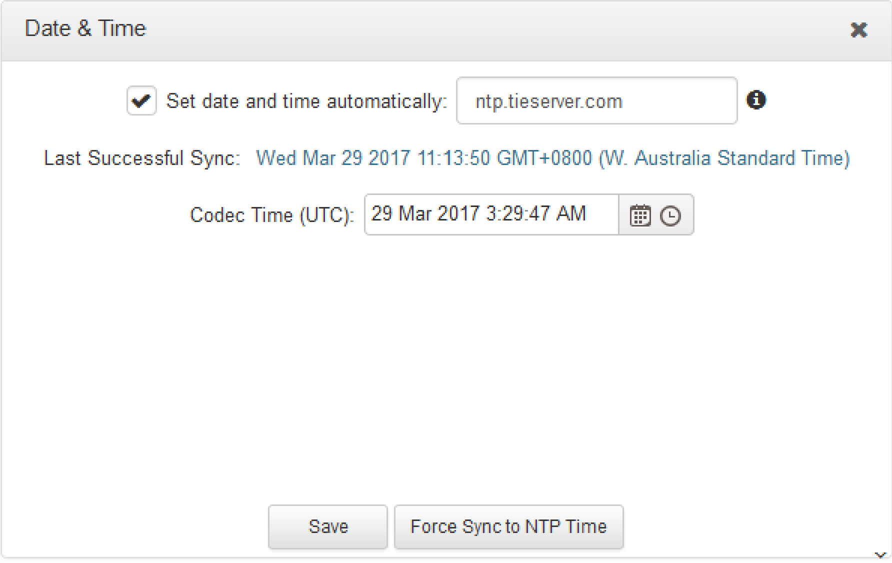

**Important Note:** Tieline and WheatNet virtual logic inputs 5-7 can be activated by pressing the **F1** button and keypad numbers **1-3** on the codec.

## 16.26 Adjusting Codec Time and Date

Network Time Protocol (NTP) is a networking protocol for clock synchronization between computer systems over packet-switched, variable-latency data networks. By default **Use NTP** time is enabled in the codec and it will synchronize with **ntp.tieserver.com**. When the codec is attached to a network it will automatically ping the selected NTP server every two hours and update the time in the codec. It is also possible to manually synchronize the time. Note: The codec will not ping a server while connected.

To manually synchronize time settings in the codec:

1. Open the HTML5 Toolbox Web-GUI and click **Settings** in the **Menu Bar**, then click **Date & Time** to display the **Date & Time panel**.
2. Click **Force Sync to NTP Time** to manually synchronize the codec to NTP time.




**Important Notes:**
- It may take more than one attempt to **Force Sync to NTP Time**.
- When NTP address settings are configured and enabled, the codec will immediately jump to the new time when it synchronizes with the server.

## 16.27 Upgrading Codec Firmware

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

### New Firmware Notifications

By default the HTML5 Web-GUI integrates with TieServer to automatically update users when a firmware upgrade is available.

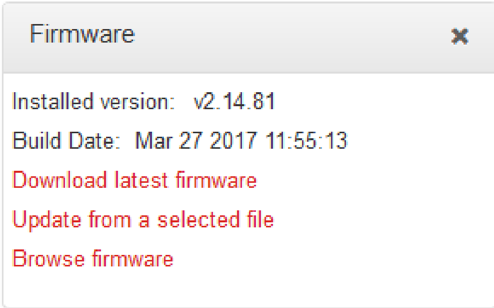
1. Connect the codec to a PC using either a LAN or USB connection and open the HTML5 Toolbox Web-GUI (See Opening the HTML5 Web GUI).
2. If new software is available the **Upgrade** symbol appears in the top-left of the screen.


3. Click **Settings** in the **Menu Bar**, then click **Firmware** to display the **Firmware panel** to perform the firmware upgrade.

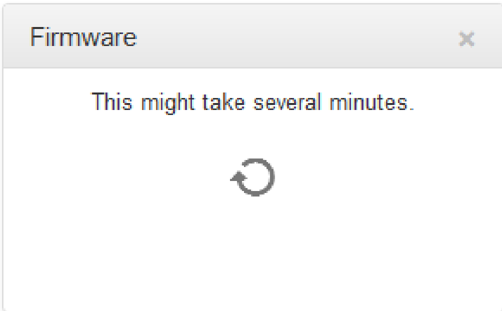
### Firmware Upgrades

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

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




2. Click **Browse firmware** to search for the latest firmware and download it to your computer. Note: the **Download latest firmware** link is only visible if newer firmware is available.
3. Once the firmware has been saved locally, click **Update from a selected file** in the **Firmware panel**.
4. Select the **.bin** file you are using to perform the upgrade and click **Open** to start the upgrade. **IMPORTANT**: The codec will reboot automatically after the firmware upgrade. DO NOT remove power or reboot the codec before the update has completed and the codec has rebooted itself.




**Important Note:** We recommend clearing your browser cache after the upgrade is complete when using the HTML5 Toolbox web-browser GUI to control codec functions. The short cuts for this are:

- Google Chrome: shift+Ctrl+delete
- Mozilla Firefox: Ctrl+shift+delete
- Internet Explorer: Ctrl+shift+delete
- Safari: Ctrl+alt+e

# 17 Front Panel Configuration Tasks

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

## 17.1 Configuring IP via the Front Panel

### Checking IP Address Details in the Codec

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


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

### Ethernet and VLAN Configuration Options

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

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

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

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


### Configure an IPv4 DHCP Address

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

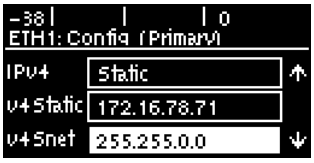
1. Press the **SETTINGS** button.
2. Select **Network** and press the button.
3. Use the down navigation button to select **LAN** or a **VLAN** interface.

4. Select **Config** and then **Usage** and the the appropriate control and/or streaming mode for the connection. Next, press the button.
5. Select **IPv4** and press the button.
6. Select **DHCP** and press the button.
7. Use the up navigation button to scroll to the top of the menu and select **Apply Setting**, then press the button to confirm the new settings.

## Configure a Static IPv4 Address

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

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




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

## IPv6 Address Assignment

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

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

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

To adjust this setting:

1. Press the **SETTINGS** button.
2. Select **Network** and press the button.
3. Use the down navigation button to select **LAN** or a **VLAN** interface.
4. Select **Config** and then **IPv6** and press the button.
5. Select **Auto**, **Manual** or **Off** and press the button.

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

### Manual IPv6 Address Assignment

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

## DNS Server

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

1. Press the **SETTINGS** button.
2. Use the navigation buttons on the front panel to select **Network** and press the button.
3. Use the down navigation button to select **LAN** or a **VLAN** interface and press the button.
4. Select **Config** and press the button.
5. Use the down navigation button to scroll to **DNS**.
6. Press the button to toggle between **Auto** and **Manual**.
7. Enter DNS Address and Domain details as required.

## Link Mode Configuration

It is possible to configure the Ethernet link speed (10/100/Auto) and whether each available interface operates in Full-Duplex or Half-Duplex modes.

1. Press the **SETTINGS** button.
2. Select **Network** and press the button.
3. Use the down navigation button to select **LAN** or a **VLAN** interface, then press the button.
4. Select **Config** and press the button.
5. Use the down navigation button to scroll to **Link Mode**.
6. Press the button to select a preferred setting. Note: Default setting is **Auto**.

## VLAN ID (VLAN configuration only)

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

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

## VLAN Priority (VLAN configuration only)

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

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

### VLAN Interface (VLAN configuration only)

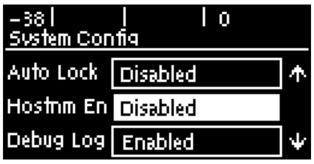
This setting applies the VLAN settings to a physical Ethernet port in the codec. Only one physical Ethernet port is available which cannot be reconfigured.

## 17.2 Configuring a Hostname

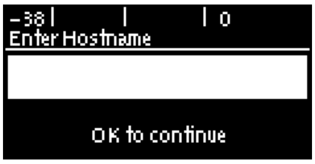
It is possible to assign a hostname to the codec to provide a flexible way of identifying the codec on a network.

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **System** and press the button.


3. Navigate down to **Hostname Enable** and press the button to enable this feature.




4. Navigate to **Hostname** and press the button to display the **Enter Hostname** screen and enter the hostname. Next, navigate to **Enter** and press the button to save the settings.


**Important Notes:**
- Modifying hostname settings requires a codec restart before they take effect
- In the **Hostname** only enter the characters a-z, A-Z, 0-9 and - and the first or last character cannot be a hyphen/dash.

## 17.3 IP Via Mapping

When dialing over IP you can select the preferred interface to use when establishing a connection. By default **Any** is selected, which means the first available interface will be used to dial a connection. The default **Via** interfaces in order of use when available are:

1. **LAN1** Ethernet port (default **Primary** Via interface)
2. **VLAN1** Ethernet port (default **Secondary** Via interface)
3. **VLAN2** (default **Tertiary** Via interface)

**Important Note:**

- If an interface is not available it is not listed in the **Via** interface selection screen. E.g. **Fuse-IP**.
- If you select Fuse-IP as the **Via** interface, a link to the Fuse-IP configuration menu appears above the **Via** menu.


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

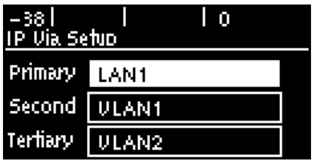
### Reconfigure Default Primary, Secondary and Tertiary Interfaces

It is possible to reconfigure the default **Primary** (LAN), **Secondary** (VLAN1) and **Tertiary** (VLAN2) interfaces in the codec. As an example, you may want to select **Primary** as the dialing interface in a program and then copy this program onto multiple codecs. However, the actual primary interface used at each location can vary for each codec. For one codec it may be an Ethernet port and for another it may be a VLAN interface. This allows you to configure site-specific settings to suit available network interfaces at different remote locations.

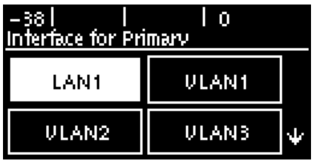
1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **System** and press the button.


3. Navigate to **IP Via Set** and press the button.

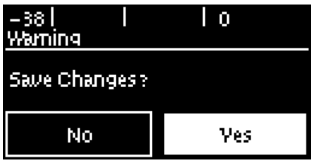

4. Navigate to the **Primary**, **Second** or **Tertiary** IP Via interface setting and press the button.




5. Select an alternative default interface and press the OK button.




6. Press the **Return button** to navigate out of the menu, then navigate to **Yes** in the **Warning** dialog and press the OK button to save all changes.




**Important Note:** Fuse-IP cannot be configured as a default Primary, Secondary or Tertiary **Via**.

## 17.4 Configuring Fuse-IP Bonding

Fuse-IP is a proprietary Tieline IP bonding technology which aggregates data by bonding multiple IP interfaces (peers) and establishing a “tunnel” between two Tieline codecs. A streaming connection can be established after the tunnel is created and Fuse-IP automatically distributes data over any two bonded interfaces.

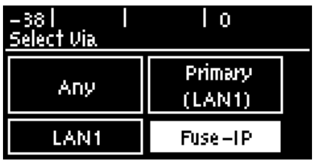
Bridge-IT only has a single LAN interface, however it can still connect as a fuse Fuse-IP server or client. This means Bridge-IT can stream data between itself and another codec which is bonding over multiple IP interfaces.

There are several benefits in using Fuse-IP to aggregate data from multiple IP interfaces, including:

- The ability to create more stable connections with higher overall data bandwidth.
- Greater choice of encoding algorithms because of higher available bandwidth.
- Redundancy in case one IP connection is lost.

### How does Fuse-IP work?

Fuse-IP is another **Via** interface you can use to dial, similar to selecting a LAN port or Wi-Fi.




Fuse-IP requires one codec to be a server and the other codec is the client. Normally the remote codec is configured as the client and the studio codec is the server, because it's generally easier to dial static IP addresses configured at the studio than interfaces at the remote site. Like

SmartStream PLUS redundant streaming, you can use two IP interfaces at the studio for additional redundancy.

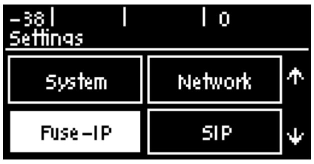
### Prerequisites

Version 2.16.xx firmware is required to use Fuse-IP. Before configuring Fuse-IP you need:

- The IP address (or addresses) for the codec acting as the server at the studio.
- The serial number of the server codec to which you are connecting using Fuse-IP.

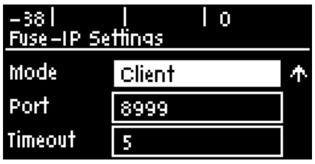
## Configuring a Fuse-IP Server at the Studio

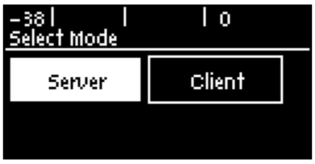
1. Press the **SETTINGS** button.
2. Use the down navigation button to select **Fuse-IP** and press the button.



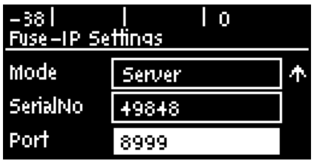

**Important Note:** Ensure Fuse-IP is disabled prior to configuration. Navigate to **Settings > Fuse-IP > Status** and press the button to disable Fuse-IP if it is enabled.

3. Navigate down to Fuse-IP **Mode** and press the button to select **Server** if the codec is at the studio and not initiating the connection. Note: the server codec serial number will be displayed and needs to be entered into the Fuse-IP client codec.



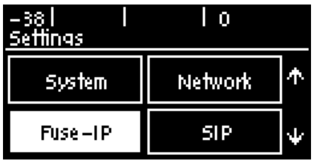

4. Leave the default Fuse-IP **Port** as **8999** in most situations unless this port is already in use, e.g. you have multiple codecs behind a firewall using Fuse-IP, therefore you need to allocate a different port for each Fuse-IP tunnel. Note: the port number on the client and server codecs must be the same.




5. The codec is now configured to connect with a client codec over a **Fuse-IP** tunnel. Navigate up to **Status** and then press the button to create a Fuse-IP tunnel between the server and client codecs.

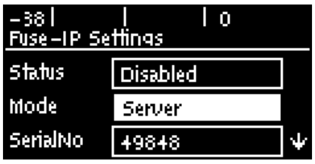
## Configuring a Fuse-IP Remote Client

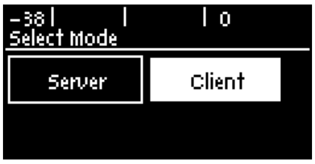
1. Press the **SETTINGS** button.
2. Select **Fuse-IP** and press the button.



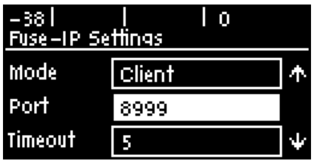

**Important Note:** Ensure Fuse-IP is disabled prior to configuration. Navigate to **Fuse-IP Status** and press the button to disable Fuse-IP if it is enabled.

3. Navigate to Fuse-IP **Mode** and press the button to select **Client** if it is a remote codec.

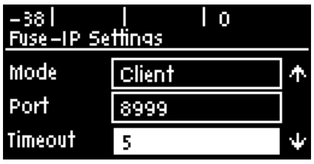


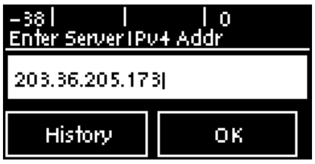

4. Leave the default Fuse-IP **Port** as **8999** in most situations unless this port is already in use, e.g. you have multiple codecs behind a firewall using Fuse-IP, therefore you need to allocate a different port for each Fuse-IP tunnel. Note: the port number on the client and server codecs must be the same.

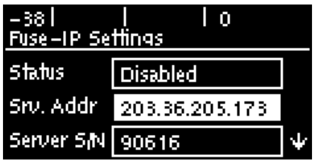



5. Navigate to inactivity **Timeout** and press the button if you want to adjust the predetermined time period for turning the Fuse-IP tunnel off. Adjust the setting and press the button to store the new setting. Note: **Inactivity Timeout** can be configured from 0 to 1440 minutes. Enter **0** to disable the timeout.

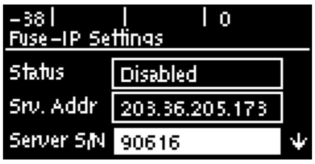



6. Navigate to **Srv. Address** (server address) and press the button to enter a public static IP address associated with the bonded interfaces at the other codec, then press the button. Note: if the bonded interfaces have private addresses behind a firewall then port forwarding needs to be configured. See Installing the Codec at the Studio for more details on port forwarding.

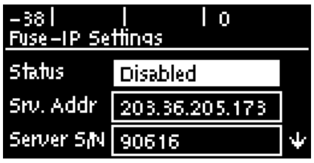


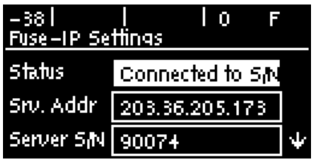




7. Navigate to **Server S/N** (serial number) and press the button to enter the serial number of the server codec to which you are connecting, then press the button.

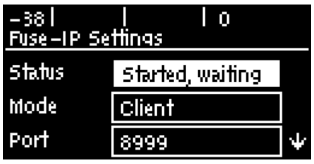




8. Navigate up to Fuse-IP **Status** and then press the button to create a Fuse-IP tunnel between the server and client codecs. Remember Fuse-IP must be enabled on both codecs. The Fuse-IP F symbol is displayed in the top right-hand corner of the screen to confirm two codecs have created an IP tunnel.

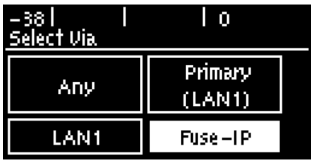



Please note: double-check all settings on both the server and client codecs if the message **Started, waiting** persists after starting Fuse-IP.




9. Select **Fuse-IP** as the **Via** interface with which to connect when creating a program using the front panel codec menus (see IP Via Mapping), or the HTML5 Toolbox Web-GUI **Program Manager panel**.




**Important Notes:**

- Data is sent by the codec over the newly created 'tunnel' as soon as Fuse-IP is enabled, even if a connection has not been configured and dialed. Depending on the number of interfaces being used, codecs may transmit and receive up to 24MB of data per hour at each end of the link.
- The codec remembers the Fuse-IP enabled/disabled state on power up.
- For additional stability it is recommended that a fixed jitter buffer is configured when streaming using Fuse-IP. The actual jitter buffer depth should account for the difference in delay between the interfaces and the maximum jitter experienced. To determine the jitter over each link you can connect and stream audio over each interface separately and look at the jitter reading displayed on the **Connection Statistics** screen.
- Use a dotted quad IPv4 address when configuring the Fuse-IP Server Address.
- For more details on setting up Bridge-IT for Fuse-IP using the codec HTML5 Web-GUI see Configure Fuse-IP Bonding.

# 17.5 Selecting an Algorithm

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

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

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

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

✔ Included
* Option available for purchase separately if required.

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

## Overview of Tieline Algorithms

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

## Overview of AAC Algorithms

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

### AAC-LC

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

### AAC-HE

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

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

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

### AAC-LD

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

### AAC-ELD

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

### AAC-ELD v 2

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

## Overview of aptX® Enhanced Audio Coding

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

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

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

## Overview of Opus Algorithm

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

There are three Opus algorithm configurations available:

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

## Configuring an Algorithm in the Codec

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

## How do I choose the right algorithm?

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

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

**Important Notes:** Music and MusicPLUS algorithms cannot be used over SIP connections. Use MP2 algorithms at 64kbps mono or 128kbps stereo for high quality connections when using SIP, or use G.711 and G.722 if required. Tieline G3 codecs do not support connections using AAC, aptX Enhanced and Opus algorithms and will default to MPEG Layer 2 if an incoming connection is configured to use these algorithms.

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

| Algorithm | Audio Band-width | Algor-ithmic Delay | IP bit rate per channel | IP over-head per connection | Audio Quality and Features | Recommended applications for on-air use |
|---|---|---|---|---|---|---|
| PCM/Linear (Uncom-pressed) | 16/24 bit up to 24kHz | 0ms | sample rate x bits per sample x no. channels | 80kbps | • Full bandwidth, perfect audio quality for voice and music<br>• No error concealment/correction or artefacts | • Extremely high quality PCM linear uncompressed audio for STLs and audio distribution.<br>• Ideal for fiber or high bandwidth links. |
| Tieline Music | Up to 15kHz | 20ms | 24 kbps minimum | 16kbps | • High quality voice and music<br>• Very low delay at low bit rates | • Great for live voice or music remotes as well as STLs and audio distribution with limited connection bandwidth (e.g. 3G wireless)<br>• Suitable when bidirectional communication between announcers is required |
| Tieline Music-PLUS | Up to 22kHz | 20ms | 48 kbps minimum (Optimised for 64kbps per audio channel) | 16kbps | • Very high quality voice and music<br>• Very low delay at low to moderate bit-rates | • Very high quality, very low delay STLs and audio distribution<br>• Remote connections able to achieve 48kbps for each audio channel<br>• Suitable when bidirectional communication between announcers is required |
| G.711 | 3kHz | 1ms | 64kbps minimum | 80kbps | • Low quality 3kHz POTS phone quality audio<br>• Very low delay at moderate bit rates | • Highly compatible with other brands of audio codec<br>• Low quality and used generally for compatibility |
| G.722 | 7kHz | 1ms | 64kbps minimum | 80kbps | • Good quality 7kHz voice<br>• Better quality than a standard POTS phone call<br>• Very low delay at moderate bit rates | • Highly compatible with other brands of audio codec<br>• Good voice quality audio for remotes and other voice quality applications |
| MPEG Layer 2 | Up to 22kHz | 24 to 36ms | 64kbps minimum | 8.5 - 13.3kbps | • Very high quality voice and music<br>• Low to moderate delay at moderate to high bit rates | • Highly compatible with other brands of audio codec<br>• Very high quality audio for remotes, STLs and audio distribution |
| LC-AAC | Up to 15kHz | 64ms | 64kbps | 15kbps | • High quality voice and music at lowest bit rate; better quality at higher bit rates<br>• Moderate delay at moderate to high bit rates | • Voice or music remotes as well as STLs and audio distribution where some delay is tolerable<br>• Tieline Music or MusicPLUS deliver lower delay |
| HE-AAC v.1 | Up to 15kHz | 128ms | 48kbps | 7.4kbps | • High quality voice and music at the lowest bit rate; better quality at higher bit rates<br>• Low to Moderate bit rates<br>• High delay | • Live voice or music remotes as well as STLs and audio distribution with limited connection bandwidth<br>• Use when bidirectional communication between announcers is not required |

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

## Algorithm Selection Guide

| Algorithm | Very Low Delay | Moder- ate to High Delay | Excell- ent Perfor- mance at Low Bit rates | Preferr- ed for Live Remotes | Preferred for STLs and Audio Distribu- tion | Highly Compat- ible with other Codecs |
|---|---|---|---|---|---|---|
| Linear/PCM | ✓ | | | | ✓ | ✓ |
| Opus | ✓ | | ✓ | ✓ | | |
| Tieline Music | ✓ | | ✓ | ✓ | | |
| Tieline MusicPLUS | ✓ | | ✓ | ✓ | ✓ | |

| | | | | | | |
|---|---|---|---|---|---|---|
| apt-X Enhanced | ✓ | | | | ✓ | |
| LC-AAC | | ✓ | | | ✓ | |
| HE-AACv1 | | ✓ | | | ✓ | |
| HE-AACv2 | | ✓ | ✓ | ✓ * | | |
| AAC-LD | ✓ | | | ✓ | ✓ | |
| AAC-ELD | ✓ | | ✓ | ✓ | | |
| AAC-ELDv2 | ✓ | | ✓ | ✓ | | |
| MPEG Layer 2 | ✓ | | | | ✓ | ✓ |
| G.722 | ✓ | | | | | ✓ |
| G.711 | ✓ | | | | | ✓ |

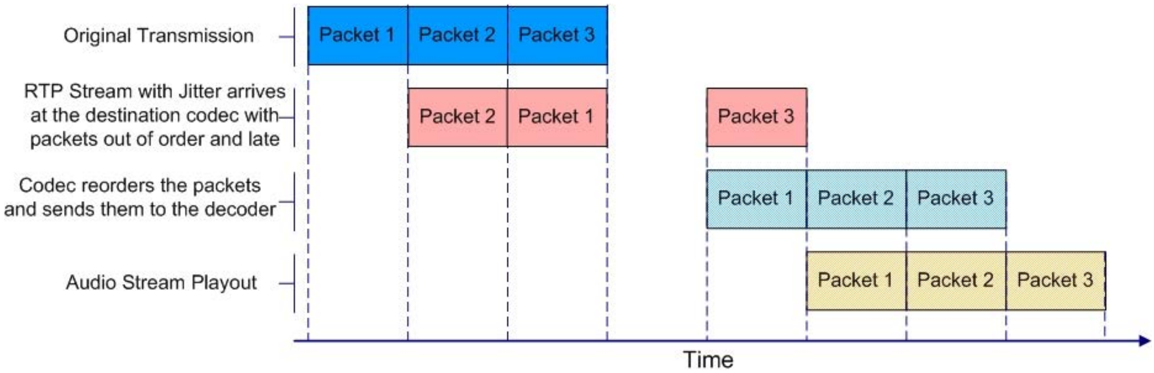
***** Use with caution for remotes due to high delay; not suitable when bidirectional communications is required.

## Sampling Rates

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

# 17.6 Configuring the Jitter Buffer

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




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

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

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

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

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

## Tieline 'Auto Jitter Buffer' Settings

The following automatic jitter buffer settings range from the most aggressive "Least Delay" setting, which endeavors to minimize delay as much as possible while adapting to prevailing network conditions, to the "Highest Quality" setting, which is the most conservative delay setting to minimize packet loss when packet latency is not as critical. Best compromise is the most popular setting used by most broadcasters as it achieves an excellent balance that minimizes latency for bidirectional communications, as well as minimizing packet loss over most wired and wireless IP networks.

**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 high level. This setting is the most aggressive in adapting to prevailing conditions, so the jitter buffer may vary more quickly than with the other settings. It is not recommended in situations where jitter variation is significant, or occurs in bursts. (E.g. cellular/multi-user wireless networks). It is best for stable and reliable links such as dedicated or lightly-loaded WAN/LANs.

**Less Delay**: This setting lies between "Best Compromise" and "Least Delay". It may assist in reducing latency over a connection without incurring packet loss.

**Best Compromise:** This default setting is the midpoint between the jitter buffer settings applicable for "Highest Quality" and "Least Delay." It is designed to provide the safest level of good audio quality without introducing too much latency. In most situations it will deliver very high quality and low delay to support live bidirectional communications over cellular and wireless networks.

**Good Quality:** This settings lies between "Best Compromise" and "Highest Quality." It may assist in achieving higher quality connections without incurring extreme delays in transmission or significant packet loss.

**Highest Quality:** This setting is the most conservative in terms of adapting to prevailing network conditions to reduce delay. The jitter-buffer will remain higher for a longer period after a jitter spike is detected – just in case there are more spikes to follow. This setting is best to use where audio quality is the most important factor and delay is not as critical. Unless delay is irrelevant, this setting is not recommended over peaky jitter networks (e.g. cellular networks) and is best used on more stable networks where fluctuating jitter bursts are not common.

## Jitter Depth

The jitter **Depth** setting allows you to select predetermined minimum and maximum jitter settings within the auto jitter buffer's minimum and maximum jitter limitations. The default setting of **60 to 1000ms** is a good starting point for most networks. It may be necessary to increase the maximum auto jitter latency setting for networks experiencing higher packet latency, or the minimum depth depending on the reliability of the network.

## Which Algorithms can use Automatic Jitter Buffering?

The following table provides an overview of which algorithms are capable of using the automatic jitter buffer feature over SIP and non-SIP connections.

| Algorithm | Tieline Session Data Connections | SIP Connections |
|---|---|---|
| PCM Linear (Uncompressed) | ✘ | ✘ |
| Tieline Music | ✔ | ✘ |
| Tieline MusicPLUS | ✔ | ✘ |
| G.711 | ✘ | ✔ |
| G.722 | ✘ | ✔ |
| MPEG Layer 2 | ✔ | ✔ |
| LC-AAC | ✔ | ✔ |
| HE-AAC v.1 | ✔ | ✔ |
| HE-AAC v.2 | ✔ | ✔ |
| AAC-LD | ✘ | ✘ |
| AAC-ELD | ✘ | ✘ |
| Opus | ✔ | ✔ |
| aptX Enhanced | ✘ | ✘ |

## Configuring Automatic Jitter Buffering (Default Setting)

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

2. Use the navigation buttons on the front panel to select **Connect** and press the OK button.
3. Select **IP** and press the OK button.
4. Select your preferred **IP Session** mode and press the OK button.
5. Use the down navigation button to select **Setup** and press the OK button.
6. Navigate to **Jitter** and press OK.
7. Select **Auto Adapt** and press OK.
8. Select your preferred jitter buffer setting and press OK.

## How to get the Best Jitter Buffer Results

When configuring automatic jitter buffer settings, establish the IP connection for a while before 'going live', to let the codec evaluate the prevailing network conditions. The initial jitter buffer setting when a codec connects is 500ms and it is kept at this level for the first minute of connection (as long as observed delay values are lower than this point).

After the initial connection period the jitter buffer is adapted to suit the current network conditions and is usually reduced. Establish a connection for at least 5 minutes prior to broadcasting, so that the codec has been provided with enough jitter history to ensure a reliable connection.

There are five states or stages that jitter buffer may display and these can be observed in the connection status screen by selecting **HOME** **> Cxns** button while connected and use the down and up navigation buttons to scroll through connection statistics.. The first four stages are observed in "auto" jitter buffer mode.

1. **Stabilization period (a1):** A few seconds at the start of a connection where no action is taken at all while the establishment of a stable connection means analysis of jitter data is not valid.
2. **Stage 2 (a2):** A compatibility check to ensure the RTP connection is compliant and RTP clocks are synchronized enough to perform jitter analysis.
3. **Stage 3 (a3):** If the compatibility check is successful, this is the analysis hold-off period. During a minute, the jitter buffer is held at a safe, fixed value of 500ms while enough history is recorded to start jitter buffer adaptation.
4. **Stage 4 "live" (A)**: This is where the codec determines it is safe enough to start broadcasting using the auto-jitter buffer level. We recommend running the codec for a few more minutes to obtain a more comprehensive history of the connection's characteristics.
5. **Fixed (F):** This state is displayed if the jitter buffer is fixed.

**Important Note:** The jitter buffer setting in the codec can only be adjusted when a connection is off-line. Automatic jitter buffering is disabled for a PCM (linear uncompressed) audio connection.

## Configure the Jitter Buffer on the Answering Codec

Session data is exchanged when a Tieline codec dials another Tieline codec and this configures the answering codec algorithm, bit-rate and jitter buffer etc. Depending on the capability of your network infrastructure, you may want to independently configure the jitter buffer settings on the answering codec. To do this:

1. Create a new answering program on the answering codec.
2. Configure preferred jitter buffer settings in this answering program.
3. Lock the answering program in the codec.

This ensures the answering program jitter buffer settings will be used when answering a call from a Tieline codec.

## Fixing Jitter Buffer Settings

The default jitter-buffer setting in Tieline codecs is 500 milliseconds. This is a very reliable setting that will work for just about all connections. However, this is quite a long delay and we recommend that when you set up an IP connection you test how low you can set the jitter-buffer in your codec.

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select your preferred **IP Session** mode and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Navigate to **Jitter** and press the button.
7. Select **Fixed Buffer** and press the button.
8. Use the numeric **KEYPAD** to enter the fixed buffer value in milliseconds and press the button.

If you change the jitter buffer setting in a codec it will only adjust to the new level when link quality is high (e.g. above 70%). This is done to ensure audio quality is not compromised. When manually configuring the jitter-buffer delay in a codec it is necessary to think carefully about the type of connection you will be using. Following is a table displaying rule of thumb settings for configuring jitter-buffer delays into your codec.

| Connection | Jitter-Buffer Recommendation |
|---|---|
| **Private LAN** | 60 milliseconds |
| **Local** | 100 - 200 milliseconds |
| **National** | 100 - 300 milliseconds |
| **International** | 100 – 400 milliseconds |
| **Wireless Network** | 250 - 750 milliseconds |
| **Satellite IP** | 500 - 999 milliseconds |

**Important Note:** The preceding table assumes Tieline Music is the algorithm in use. Do not use PCM (uncompressed) audio over highly contended DSL/ADSL connections without enough bandwidth to support the high connection bit-rates required.

## Relationship between the 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.

## 17.7 Configuring Forward Error Correction

There are two modes of Forward Error Correction (FEC) available in the codec:

1. Tieline FEC.
2. RFC 2733 compliant FEC (Sessionless connections only).

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. Tieline FEC transmits a secondary stream of audio data packets over a single connection. RFC 2733 compliant FEC transmits audio packets over a separate connection.

Both the local and remote codec FEC settings can be configured in the codec before dialing. FEC should only be used if link quality displayed on the codec is below **S:99 R:99**, as it is of no benefit otherwise.

The FEC settings are outlined in the following table with their bit rate ratios.

| FEC Setting | Bit rate Ratios | Connection Use | Tieline FEC | RFC 2733 FEC |
|---|---|---|---|---|
| **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. | Yes | Yes |
| **50%** | Additional data is sent by FEC in a ratio of 2:1. | Recommended for international & national connections | Yes | Yes |
| **33%** | Additional data is sent by FEC in a ratio of 3:1. | Recommended for national and local connections. | Yes | Yes |
| **25%** | Additional data is sent by FEC in a ratio of 4:1. | Recommended for national and local connections. | No | Yes |
| **20%** | Additional data is sent by FEC in a ratio of 5:1. | Recommended for local and LAN connections. | Yes | Yes |
| **10%(Highest delay)** | Additional data is sent by FEC in a ratio of 10:1. | Recommended for local and LAN connections. | No | Yes |
| **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 with connections that aren't shared & have quality of service (QoS). | Yes | Yes |

## Configuring Tieline FEC in the Codec

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select **Tieline** and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Navigate to **FEC** and press .
7. Select the local codec FEC setting in the **Local FEC** screen and press .
8. Select the remote codec FEC setting in the **Remote FEC** screen and press .

**Important Note:** Tieline FEC can only be configured for use with the Music and MusicPLUS algorithms.

## Configuring RFC 2733 FEC in the Codec

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select **Sessionless** and press the button.
5. Select peer-to-peer (**P-to-P**) and press the button.
6. Use the down navigation button to select **Setup** and press the button.
7. Navigate to **FEC** and press .
8. Select **Enable** and press .
9. Select the **FEC Percentage** and press .
10. Select the **FEC Delay** used by the codec and press .

**Important Notes:**

- The **FEC Delay** configured should take into account the packet arrival (jitter buffer) strategy at the remote codec. For example, if the maximum jitter buffer at the remote codec is 1000 ms, the FEC Delay setting should be lower, to ensure there is enough time for FEC packets to arrive and replace lost packets prior to audio playout.
- By default, the codec will use the audio stream IP address as the remote FEC IP address as well. This can be adjusted in the **Program Manager panel** in the HTML5 Toolbox web-GUI.
- The default local and remote UDP audio FEC ports are 9002.
- Any of the available algorithms can be selected when configuring RFC 2733 FEC in the codec.

## How does FEC work?

If you enter a FEC setting of 20% and you are losing one packet in every five sent, the lost packet will be replaced by FEC to maintain the quality of the connection. If you are losing more packets than this, say one in three, it will be necessary to increase the FEC setting to 33% to compensate.

**Note**: There is an inverse relationship between FEC settings and the jitter-buffer millisecond setting that you use for IP connections.

So why not use 100% FEC every time? The answer is because you need twice the bit rate to achieve full redundancy and depending on the link conditions, this could potentially cause more drop-outs because of network congestion than it fixes. Here is a simple rule to remember: Your maximum uplink speed is all the bandwidth you have to play with. As a rule of thumb, try not to exceed more than 80% of your maximum bandwidth. If your link is shared, be even more conservative.

You should also consider the remote end too. What is their maximum upload speed? Is the connection shared at either end? Your bit rates, FEC settings and buffer rates must be pre-configured at both ends before you connect, so it's always better to set your connection speed and balance your FEC according to the available uplink bandwidth at each end for best performance.

As an example, if you want 15 kHz mono (using the Tieline Music Algorithm) you will need at least a 24kbps connection for audio. Adding 100% FEC will add another 24kbps making your bit rate 48kbps plus some overhead of around 10kbps is required. If you're on a 64kbps uplink, you should consider reducing your FEC to minimize 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.

### Conserving Bandwidth with FEC

There is a trade-off between the quality and the reliability of an IP connection – particularly when FEC is activated on your codecs. However, it is possible in certain situations to configure different FEC settings on each codec to match connection bandwidth capabilities at either end of the link, conserve bandwidth and create more stable IP connections.

For example, if your broadcast is a one-way broadcast from a remote site, i.e. you are not using the return path from the studio, or only using it for communications purposes, it is possible to reduce or turn off FEC at the studio codec. This effectively reduces the bandwidth required over the return link and increases the overall bandwidth available for the incoming broadcast signal from the remote site.

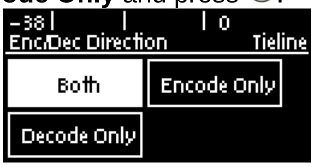
## 17.8 Configuring Encode/Decode Direction

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.

To use this feature configure the transmitting codec to encode only and the receive codec to decode only. To adjust this setting:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the button
3. Select **IP** and press .
4. Select your preferred **IP Session** mode and press the button.
5. Use the down navigation button to select **Setup** and press .
6. Navigate to **Dir** and press .
7. Select **Encode Only** or **Decode Only** and press .



## 17.9 Enabling Relays & RS232 Data

Data must be enabled to activate contact closure operation and RS232 data. The codec supports both in-band and out-of-band data depending on the algorithm you are using. In-band RPTP data is automatically enabled when using the Tieline Music or MusicPLUS algorithms over IP. It is also possible to enable synchronized out-of-band data using any algorithm via the **RS232** data port on the codec.

| Algorithm Selected | IP |
|---|---|
| **Tieline Music and MusicPLUS** | • In-band RPTP data is enabled automatically<br>• Synchronized out-of-band data can be enabled and disabled as required<br>• Using out-of-band data with rules between G5 codecs employing relay reflection minimizes latency<br>• These algorithms must be used when connected to G3 codecs as they don't support out-of-band data |
| **All other algorithms** | • No in-band data available; synchronized out-of-band data can be enabled and disabled |

Select **Enable Auxiliary Data** when creating a program in the **Program Manager panel** to enable out-of-band RS232 data and activate rules employing relay reflection over a connection. This will allow the codec to connect to external devices and send RS232-compatible data via the serial port on the rear panel. Alternatively, enable auxiliary data using the **Setup** menu as follows:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select **Tieline** (or **Sessionless** and then **Peer-to-Peer**) and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Navigate to **Data** and press to toggle between **Enabled** and **Disabled** (Note: default setting is **Disabled**).

### Configuring Control Port Contact Closure Operation

The **Rules panel** in the Web-GUI can be used to configure switch inputs and relay outputs. Codec 'rules' configure events based on specific codec actions. Typically rules are based on a change in the state of a physical **CONTROL PORT** GPIO, or a WheatNet-IP logic IO, or a codec program being connected or disconnected. Rules can only be created with the Web-GUI while the codec is disconnected. There are three ways to create rules in the HTML5 Toolbox Web-GUI:

1. **Rules panel**: Configure codec level rules related to programs and/or hardware and software I/O states.
2. **Program Manager panel**: Configure program level rules early in the **Program Manager panel** wizard.
3. **Program Manager panel**: Configure stream level rules for an audio stream when proceeding through the **Program Manager panel** wizard..

**Important Notes:** A non-WheatNet-IP Tieline codec can be configured to trigger a logic IO in a Tieline Genie Distribution and Merlin PLUS WheatNet-IP codec, as well as physical **CONTROL PORT** GPIOs. The codec has:

- 2 physical **CONTROL PORT** GPIOs.
- 5 Tieline and WheatNet virtual inputs (1-5). Note: Virtual inputs 3-5 can be activated using the F1 button and keypad numbers 1-3 on the codec.
- 64 Tieline virtual logic outputs. Note: Tieline logic IOs (LIOs) are only supported between Genie, Merlin, ViA and Bridge-IT IP codecs.

- 64 virtual WheatNet logic outputs available in Genie Distribution and Merlin PLUS WheatNet-IP codecs. These allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network.

## About Relays

Bridge-IT has two CMOS solid state relays for the control of equipment, consisting of two relay closures and two opto-isolated outputs.

### Inputs

The input signal is referenced to chassis ground, i.e. the ground reference terminal on the terminal block is connected the chassis. The input device is a high impedance CMOS device with a 330 ohm pull-up resistor to +5 volts.

Operation is as simple as joining the input pin to the ground terminal. This can be via a remote relay contact or the open circuit collector of a transistor or FET. DO NOT feed voltages into the inputs.

### Outputs

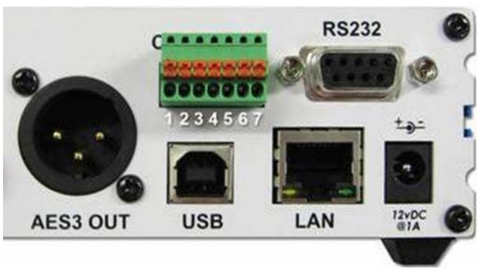
CMOS field effect transistors switch a low impedance path between the two pins when activated. These are opto-isolated and floating above ground. It is important to current-limit the source as damage will result where the current exceeds 100mA peak-to-peak. No more than 48 volts peak-to-peak should be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

## Relay Operation and Pinouts

A closing contact across Input 1 or 2 (pins 5 or 6) to Ground (pin 7) will provide a closing contact on the remote codec Output 1 (pins 1 and 2) or Output 2 (pins 3 and 4).

For multi-unicast connections to multiple codecs, a contact closure will appear on each of the compatible (non-G3) remote codecs' corresponding contacts. I.e. Input 1 shorted, Output 1 contacts on all codecs closed.

| Pins | Pin Function |
|---|---|
| 1 | Output 1 |
| 2 | Output 1 |
| 3 | Output 2 |
| 4 | Output 2 |
| 5 | Input 1 |
| 6 | Input 2 |
| 7 | Ground |

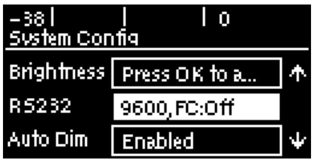



**Important Note:** See Creating Rules for more information about how to configure relay operations with a PC using the Toolbox web-GUI.

## Configuring the Codec to Send RS232 Data

Once Data is enabled, the codec can also be connected to external devices and send RS232-compatible data via the serial port on the rear panel of the codec.

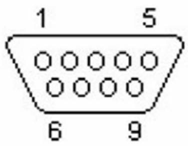
1. Press the **SETTINGS** button.
2. Navigate to **System** and press .
3. Select **RS232** and press .



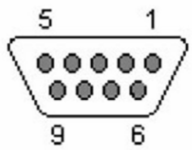

4. Use the navigation buttons to select the correct baud rate.
5. Select **Enable** for flow control and press to save all settings.

## RS232 Pin-outs and Data Connections

| Pin | INTERFACE Female DB9 (RS232) DCE | DATA Male DB9 (RS232) DTE |
|---|---|---|
| 1 | No Connection | No connection |
| 2 | TX Data | RX Data |
| 3 | RX Data | TX Data |
| 4 | No connection | No connection |
| 5 | Signal Ground | Signal Ground |
| 6 | No Connection | No connection |
| 7 | CTS | RTS |
| 8 | RTS | CTS |
| 9 | No connection | No connection |




**DB9 Male Connector Pins**




**DB9 Female Connector Pins**

**Important Notes:**

- When connecting to G3 codecs only in-band data is available via the Music and MusicPLUS algorithms.
- Codecs using Bridge-IT firmware lower than v2.8.xx cannot activate relays on Tieline G3 codecs or send RS232 data to them.
- It is important that you enable serial port flow control within the codec. Flow control regulates the flow of data through the serial port. If disabled, data will flow unregulated and some may be lost.
- Ensure you match the serial port baud rate to match the rate of the external device you are connecting to. Ideally the settings on both codecs should match, or you could have data overflow issues.
- Only the dialing codec needs to be configured to send RS232 data. Session data sent from the dialing codec will configure all other compatible codecs (non-G3) you connect with.

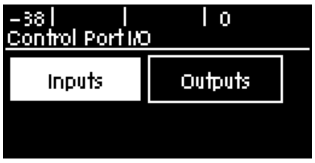
- RS232 data can be sent from the dialing codec to all end-points of a multi-unicast connection. Note: Bidirectional RS232 data is only available on the first connection dialed when multi-unicasting.

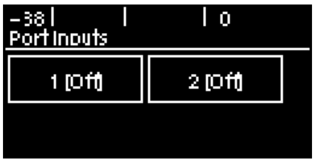
## 17.10 Monitor Control Port I/O Status

It is possible to monitor the status of the two relay inputs and two opto-isolated outputs available using the DB15 **CONTROL PORT IN / OUT** connector. To monitor status:

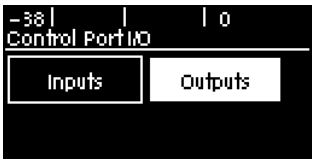
1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Control Port I/O** and press the button.

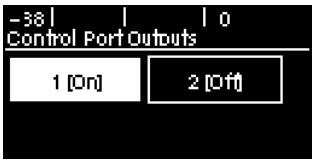

3. Select **Inputs** and then press the button to view input status.




4. Select **Outputs** and then press the button to view output status. Select an output and press the button to toggle the output state from **Off** to **On**. Note: Input states cannot be changed in this way.




## 17.11 Configuring TCP/UDP Ports

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 port forward 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 most likely to get through firewalls – ensuring critical session data (including dial, connect and hang-up data) will be received reliably.

The default session and audio port settings in Tieline codecs, for both TCP and UDP connections, are outlined in the Installing the Codec at the Studio section of the manual. This section also contains useful information for configuring port forwarding and troubleshooting IP connections.

### Changing Codec Port Numbers

Reasons for adjusting the port setting on your codec include:

- Having to create a path through gateways and firewalls.
- Another IP device is using the codec's default port number.
- You are sharing a single IP address with multiple codecs and each codec requires a different port number to perform port forwarding.

## Configuring the Session and Audio Port Numbers used when Dialing a Program

Codecs require matching port numbers to connect successfully. When you create a program the session and audio ports can be adjusted from the defaults as required. Note: If there is a need to change codec port settings please consult your organization's resident IT professional. To adjust port settings:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select an **IP Session** mode.
5. Use the down navigation button to select **Setup** and press the button.
6. Navigate to either **Session** (session protocol) or **Proto** (audio protocol) and press .
7. Select the session or audio ports you want to adjust and press .
8. Use the numeric **KEYPAD** add a new port number and press .

## Configuring Tieline Session Ports when Answering

To adjust the local Tieline session data port used by your codec to answer incoming calls:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Settings** and press the button.
3. Select **Tieline Session** and press .
4. Navigate to **Port** or **Alt. Port** (alternative session port) and press .
5. Adjust the setting and press the button to store the new configuration.

## Audio Port Settings for Tieline Session Data and Sessionless IP Calls

The codec supports sessionless IP streaming, whereby the codec does not send Tieline session data when attempting to connect. When using this mode you need to configure the "send" audio port (codec port at the remote end of the link to which you are sending audio) and "return" audio port (port used by the local codec to receive audio from the remote codec).

It is also possible to configure the remote and local audio ports for a codec using Tieline session data to establish IP connections. This may be required because some firewalls require symmetric port configuration.

### Sessionless Audio Port Configuration

When you select **Sessionless** as the **Session Protocol**:

- The default value for both the **Send** and **Return** (audio) **Ports** is 9000
- The range of values for the audio ports is 2000 to 65535
- The audio port values can be set independently
- Both audio ports can always be configured, i.e. there is no dependency on encode/decode direction

### "Tieline Codec" Port Configuration

If using the **Tieline Codec** setting for call establishment (i.e. Tieline session data is enabled), you can also change the default audio ports if required.

- The default value for the **Send** (audio) **Port** is 9000
- The range of values for the **Send Port** is 2000 to 65535
- The default port value for the **Return** (audio) **Port** is **Automatic**. Note: **Automatic** indicates that the codec will allocate the return port value and send this information to the codec to which you are dialing
- The range of values for the **Return Port** is 2000 to 65535

### Sessionless Multicast Connections

For a sessionless multicast server connection:

- Only the **Send Port** is available
- The default value for the port is 9000
- The range of values for the port is 2000 to 65535

For a sessionless multicast client connection:

- Only the **Return Port** is available
- The default value for the port is 9000
- The range of values for the port is 2000 to 65535

## 17.12 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.

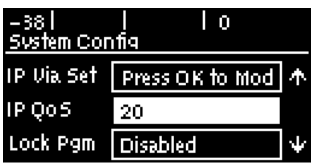
### Prioritizing IP Data Packets when Broadcasting

Broadcast IP audio data packets can be configured for expedited or assured forwarding (Quality of Service or QoS) when traversing different networks. Routers can also be configured to ignore these forwarding priorities so they are not assured across all networks.

The codec can be configured to prioritise IP data packets sent across a network by entering a value into the Differentiated Services Code Point (DSCP) field within the header of data packets transmitted by the codec over the network. Check with your IT administrator before changing this setting. By default the codec is configured for Assured Forwarding and more details about DSCP are available on Wikipedia at http://en.wikipedia.org/wiki/Dscp.

### Configuring Bridge-IT for QoS

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **System** and press the button.
3. Navigate to **IP QoS**, then press the button and use the **RETURN** button to delete the DSCP value already entered, then use the numeric **KEYPAD** to enter the new setting.



4. Press the button to save the new setting.

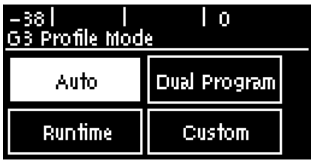
**Important Note:** To ensure the continuous and regular flow of tagged data packets along the path from point-to-point, all routers and switching equipment must allow the QoS DSCP setting. Any bandwidth partitioning schemes should partition over a small interval to ensure the codec jitter buffer does not empty and audio remains continuous.

See Configuring IP Settings for instructions on configuring this setting using the HTML5 Toolbox web-GUI.

## 17.13 Tieline G3 Profile Compatibility

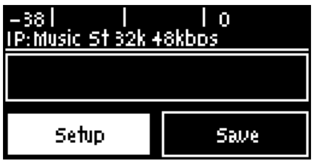
The codec is able to support dialing to Tieline Commander and i-Mix G3 codecs in various modes:

1. **Auto**: the codec will dial the G3 codec and connect in mono, stereo.
2. **Dual Program**: This allows the codec to dial a G3 codec with a Dual Program profile loaded and support two simultaneous mono connections.
3. **Runtime**: The G3 codec will retain runtime settings when answering a call from a G5 codec.
4. **Custom**: The G3 codec will load a profile specified, e.g. profile 6, which is the first custom profile number.



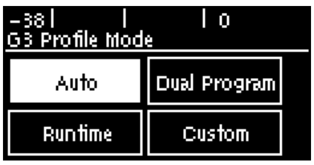

### G3 Profile Configuration

1. Press the **HOME** button to return to the **Home** screen, select **Connect > IP > Tieline** and press the button.
2. Press the down navigation button to select **Setup** and press .




3. Navigate down to **G3 Pro** and press .


4. Select the preferred compatibility mode and press .



## 17.14 Configuring Data Packet Time-to-Live

Time-to-Live (TTL) is a value you can configure to set a finite life for data packets sent by the codec. This avoids situations where packets can keep circulating through routers causing network congestion.

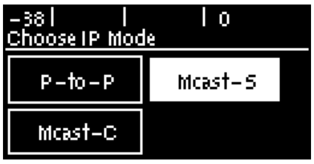
The Time-to-Live setting is configurable and sets the maximum number of router hops allowable for multicast data packets. In most situations the default value of **1** is used, to ensure packets are sent through a single LAN router and not over multiple router hops and networks.

This setting is only used in **Multicast Server** mode.

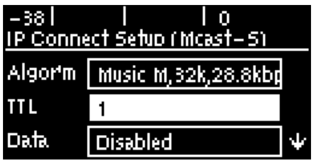
1. Press the **HOME** button to return to the **Home** screen, select **Connect,** then select **IP** and press the button.


2. Select **Sessionless** and press .


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




4. Press the down navigation button to select **Setup** and press .
5. Press the down navigation button to select **TTL** and press .




6. 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.

## 17.15 Reset and Restore Factory Default Settings

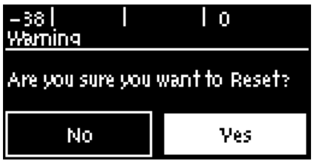
There are several options in the **Reset/Backup** menu which allow you to restore factory default settings within the codec.

| | Function | Description |
|---|---|---|
| 1 | Backup | Select to backup custom Program and/or System data. |
| 2 | Restore | Select to restore custom Program and/or System data. |
| 3 | Reset Audio and 'Connect' Settings | Select to restore factory default settings for Audio and Connect menu settings |
| 4 | Restore Factory Defaults | Select to restore factory default settings, excluding user defined programs and call history |
| 5 | Delete Programs & Call History | Deletes custom programs and recent calls in the codec; speed dial contacts are retained |
| 6 | Reboot Codec | Select to restart the codec |
| 7 | Reset Codec Log | Deletes codec event and log history. Note: This should only be performed if instructed to by Tieline support staff. |

**Important Note:** After restoring factory defaults, always reboot the codec using the **Reboot Codec** function, not by removing power from the codec.

1. Press the **SETTINGS** button.
2. Navigate to **Reset/Backup** and press the button.
3. Navigate to the preferred option from those available and press the button.


4. Select **Yes** and press the button to confirm the menu function that you are performing.




### Reset and Restore Factory Defaults using the Web-GUI

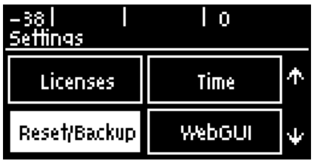
See Reset Factory Default Settings to use the HTML5 Toolbox Web-GUI to reset and restore factory defaults.

# 17.16 System Backup and Restore

The **Reset / Backup** menu allows users to backup and restore Program information and/or system data. Backup or restore data using an SDHC card inserted into the SD slot on the front panel of the codec. Note: A single partition FAT32 formatted SDHC Card is required (SD cards may be less reliable and are not recommended).

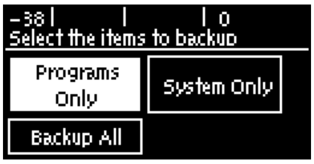
## Creating an SD Card Backup File

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Reset/Backup** and press the button.




3. Insert a single partition FAT 32 formatted SDHC card into the SD card slot on the front panel of the codec.
4. Select **Backup** and press the button.

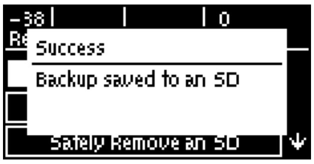

5. Select the preferred backup option and press the button.



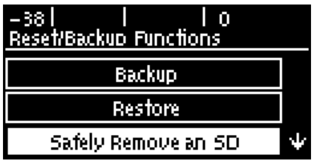

6. Edit the file name and press the button.


7. Select a directory in which to save the backup.tgz file and press to save it.


8. A confirmation dialog is displayed when the file has been saved successfully.




9. Select **Safely Remove an SD** before removing the SD card from the codec.

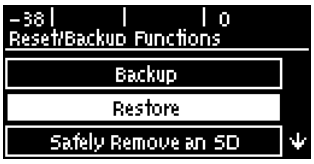



10. A confirmation dialog appears when it is safe to remove the SD card from the codec.

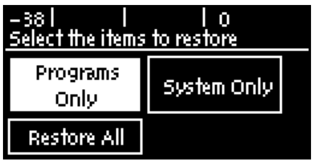

## Restoring Data from an SD Backup File

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Reset/Backup** and press the button.

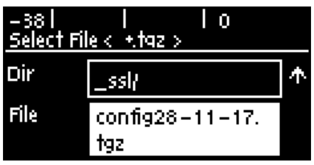

3. Insert a single partition FAT 32 formatted SDHC card into the SD card slot on the front panel of the codec.
4. Select **Restore** and press the button.



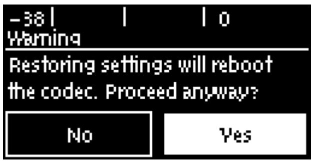

5. Select the preferred restore option and press the button.



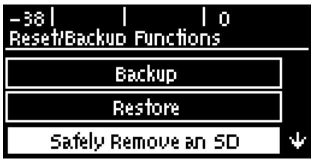

6. Select the file to restore from the SD card and press the button.



7. Select **Yes** to perform the system restore. Note: The codec will automatically reboot after settings have been restored.




8. Select **Safely Remove an SD card** before removing the SD card from the codec.



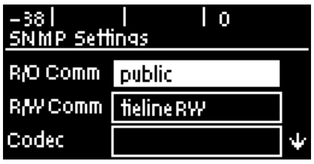

9. A confirmation dialog appears when it is safe to remove the SD card.


## 17.17 Configuring SNMP Settings

The codec supports Simple Network Management Protocol (SNMP ) for managing devices on IP networks. To configure SNMP settings:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **SNMP** and press the button.


3. Navigate to each setting in turn and press the button to adjust and save each new setting.




**Important Note:** For more information on SNMP codec settings and downloading MIB files, see Configuring SNMP in the Codec.

## 17.18 Adjusting the LCD Screen Display

### Adjusting LCD Screen Contrast Levels

1. Press and hold the button and then press the arrow up button to display the **Contrast** adjustment screen.

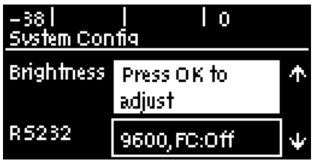
2. Use the left and right arrow buttons to adjust the **LCD SCREEN** contrast until viewing is optimized.
3. Press when you have finished.


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

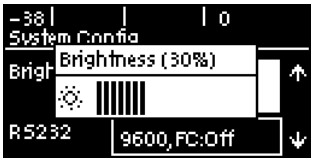
## Adjusting LCD Screen Brightness

To adjust **LCD SCREEN** brightness:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **System** and press the button.
3. Navigate to **Brightness** and press the button.



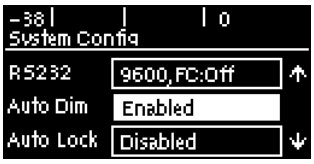

4. Use the left and right arrow buttons to adjust the brightness level up and down.




## LCD Screen Auto Dim Mode

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

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



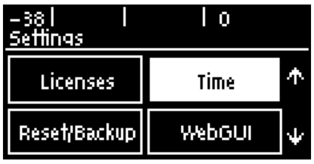

**Important Note:** The default **Auto Dim** time-out is reduced from 30 seconds to 10 seconds when the **Auto Lock** function is enabled (to lock the front panel controls). Disabling **Auto Dim** mode will override all time-out periods and the LCD will remain fully illuminated at all times.

## 17.19 Adjusting Time Settings

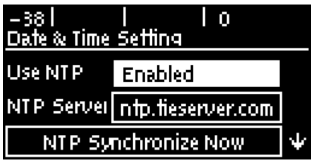
By default **Use NTP** time is enabled in the codec. When the codec is attached to a network it will automatically ping the selected NTP server every two hours and update the time in the codec. It is also possible to manually synchronize the time. Note: The codec will not ping a server while connected.

To adjust time settings in the codec:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Time** and press the button.




3. Navigate to **Use NTP** and press the button to enable or disable this feature.




4. Navigate to **NTP Synchronize Now** and press the button to synchronize the codec time with the designated **NTP Server**.

## 17.20 Installing Software Licences

Bridge-IT XTRA codecs include all software and algorithm feature options when purchased. Bridge-IT codecs can be upgraded by purchasing and installing licenses with new features. Options include a multi-unicast license (includes multicast capability), AAC Encode license, or aptX® Enhanced encoding license. Contact Tieline at sales@tieline.com, or your favorite dealer, if you need to purchase a software license upgrade. When a software license has been purchased there are two ways to perform an upgrade:

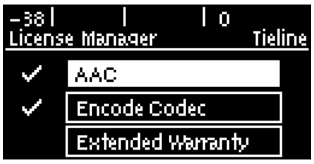
1. Automatically download the software license from TieServer and perform the upgrade.
2. Download the new software license file onto your PC and upgrade using the web-GUI.

To install a software license using the HTML5 Toolbox web-GUI see Web-GUI Software License Installation.

### Checking Installed Licenses

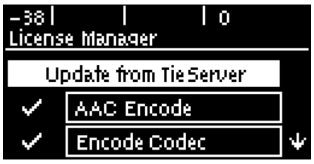
The codec **License Manager** is used to view which licenses are installed in each codec. To view the licenses installed in your codec:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Licenses** and press the button.
3. A list of all possible licenses is displayed and all licenses that have been installed have a tick next to them.




## Update and Install Licenses from the Codec

1. Navigate to **Update from TieServer** in the **License Manager** screen and press the button.




2. The codec will contact TieServer and automatically install all valid licenses.
3. The screen will indicate the update is in progress and then confirm it has been completed successfully.
4. Press the **RETURN** button a few times until you return to the **Home** screen.
5. Use the navigation buttons to select **Settings** and press the button.
6. Navigate to **Reset** and press the button.
7. Navigate to **Reboot Codec** and press the button.
8. Select **Yes** and press the button to reboot the codec.

## 17.21 Upgrading Firmware via SD Card

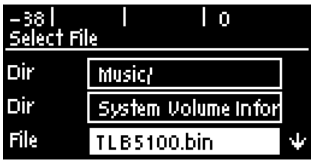
To download the latest codec firmware visit http://www.tieline.com/Support/Latest-Firmware. Copy the firmware file onto an SDHC card and then use the following procedure to perform a firmware upgrade.

**Important Note for SDHC Card:** A single partition FAT32 formatted SDHC card must be used to perform the firmware upgrade.

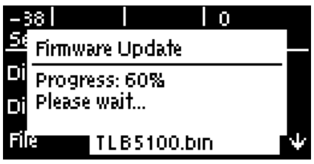
1. Insert an SDHC card with the latest firmware into the SD card slot on the front panel of the codec.
2. Press the **SETTINGS** button.
3. Use the navigation buttons to select **System** and press the button.


4. Navigate down to **Firmware update from an SD** and press the button. Note: it can take a few seconds for the USB drive to be detected.


5. Navigate to the firmware file after the SDHC card has been detected, then press the button.




6. The codec will automatically reboot after the upgrade is complete.




7. To safely remove the SD card, press the **SETTINGS** button and select **Reset/Backup > Safely Remove an SD** before removing the SD card from the codec.

**Important Note:** We recommend clearing your browser cache after the upgrade is complete when using the HTML5 Toolbox web-browser GUI to control codec functions. The short cuts for this are:

- Google Chrome: shift+Ctrl+delete
- Mozilla Firefox: Ctrl+shift+delete
- Internet Explorer: Ctrl+shift+delete
- Safari: Ctrl+alt+e

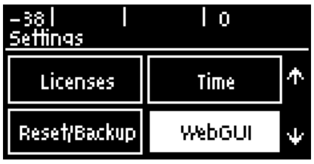
# 17.22 Installing a Security Certificate

Tieline codecs support the installation of TLS/SSL (hereafter referred to as SSL) security certificates to deliver an additional layer of security when connecting to IP networks. The digital SSL security certificate authenticates the codec and provides more secure encrypted HTTPS browser connections. The codec supports installing a private key as well as an intermediate and SSL certificate.

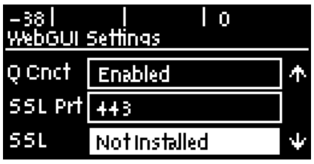
## Certificate Installation

To install certificates purchased from a reputable vendor:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Web-GUI** and press the button.




3. Navigate down to **SSL** and press the button.




4. Select **Configure SSL** and press the button.


5. Ensure the Private Key, digital SSL Certificate and Intermediate Certificate (if required), are loaded onto an SDHC card and then insert it into the SD card slot on the front panel of the codec. Note: A single partition FAT32 formatted SDHC card must be used


6. Select **Private Key** and navigate to the correct directory and .key (Private Key) file to install from the SDHC card and press the button.

7. Select **Certificate** and navigate to the SSL Certificate (.crt) file on the SDHC card and press the button.

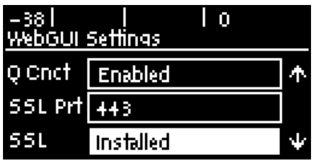

8. If an Intermediate Certificate has been supplied, select **Intermediate** and navigate to the Intermediate Certificate (.crt) file on the SDHC card and press the button.


9. After adding the private key, SSL certificate, and intermediate certificate (if supplied), navigate up to **Install** and press the button .


10. Select **Yes** to confirm installation of the certificates.


11. A dialog confirms the certificates have been installed correctly.


12. The **SSL** menu also confirms the files are successfully installed.

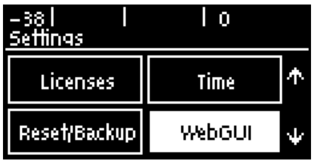



13. To safely remove the SD card, press the **SETTINGS** button and select **Reset/Backup > Safely Remove an SD** before removing the SD card from the codec.
14. To access a codec via the HTML5 Toolbox Web-GUI in a browser after installing SSL security certificates ensure you type "https://" before the codec IP address. For example, https://172.16.0.100.

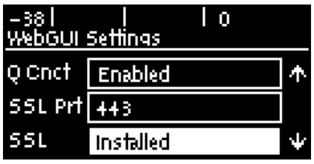
## Remove SSL Security Certificates

To remove installed SSL security certificates from a codec:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Web-GUI** and press the button.

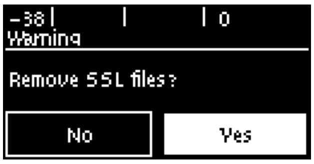



3. Navigate down to **SSL** and press the button.




4. Select **Remove SSL** and press the button.


5. Confirm removal of the SSL files.




6. A dialog confirms the certificates have been removed succesfully.


## Changing the Default SSL Port

The codec uses the standard TCP port 443 for SSL communications. The port number can be adjusted by navigating to **SETTINGS** **> Web-GUI > SSL Port**.

## Troubleshooting Certificate Installation

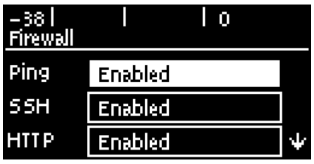
If you type "https://" before the codec IP address and can't open the Toolbox web-GUI, first uninstall the certificates and then reinstall them. Also double-check you are installing the correct certificates. If you continue to have issues, contact your certificate vendor to ensure the certificate is valid.

## 17.23 Firewall Configuration

This menu can be used to enable or disable a range of firewall-related network services, or limit ping to only work in a local subnet.

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Firewall** and press the button.

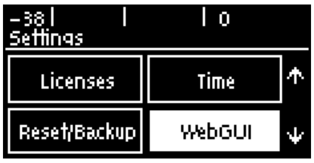

3. Select **Ping**, **SSH**, **HTTP**, **HTTPS**, **NTP** and **SNMP** firewall options.



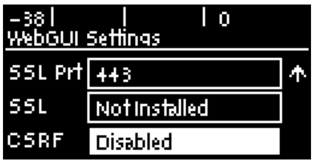

## 17.24 Enabling CSRF Security

CSRF (Cross-Site Request Forgery) protection can be configured to protect the codec from CSRF attacks. To enable or disable this setting:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Web-GUI** and press the button.




3. Navigate down to **CSRF** and press the button to toggle between **Enabled** and **Disabled**.




# 18 Reference

The following sections contain reference and troubleshooting information.

## 18.1 Installing the Codec at the Studio

### Studio IP Streaming Setup for Tieline Audio Codecs

The following instructions are intended to help you configure your internet connection and Tieline codecs at the studio to enable incoming calls over the internet from a remote Tieline codec. It is assumed that you have a basic understanding of your IP network and how to configure IP devices.

If you have limited IT network knowledge, we recommend you engage the services of an IT professional to install the public IP address and perform the Network Address Translation (NAT) and port forwarding between the public internet and your private Local Area Network (LAN) at the studio.

## Prerequisites

The following procedures are valid for:

- All firmware versions in the Genie and Merlin codec families.
- All Bridge-IT Basic and Pro and Bridge-IT XTRA codecs with firmware release v.2.x or higher.
- All Commander G3 and i-Mix G3 codecs.

## Getting Started at the Studio

To perform a typical codec installation at the studio you will need to:

1. Contact your Internet Service Provider and organize a dedicated high speed broadband connection at the studio for your codec with a public static IP address. Do not share this connection with other devices.
2. Install your codec at the studio and attach an active RJ-45 LAN cable to the “LAN” or “Ethernet” port on the rear of the codec. Please note:
   - The green LED underneath the “LAN” or "Ethernet" port will illuminate and the orange LED will flash steadily if you are connected to an active LAN connection.
   - The Genie and Merlin families of IP codecs support two simultaneous Ethernet connections.
3. If you are connecting a single codec to a router without a firewall you can enter the public IP address, Subnet Mask and Gateway directly into the codec and your work is done. Note: your Telco should be able to provide this information.
4. Alternatively, if you are connected to a router with a firewall, configure Network Address Translation (NAT) in your router. NAT is performed between the public internet and your private Local Area Network (LAN) by your router. Your remote codec sends IP data packets to the studio router's public static IP address and the router performs NAT, which forwards these data packets to the private IP address allocated by the router to your codec. As part of this process we recommended you:
   - Connect to your router using a web-browser.
   - Configure it to allocate a static private IP address for each codec.

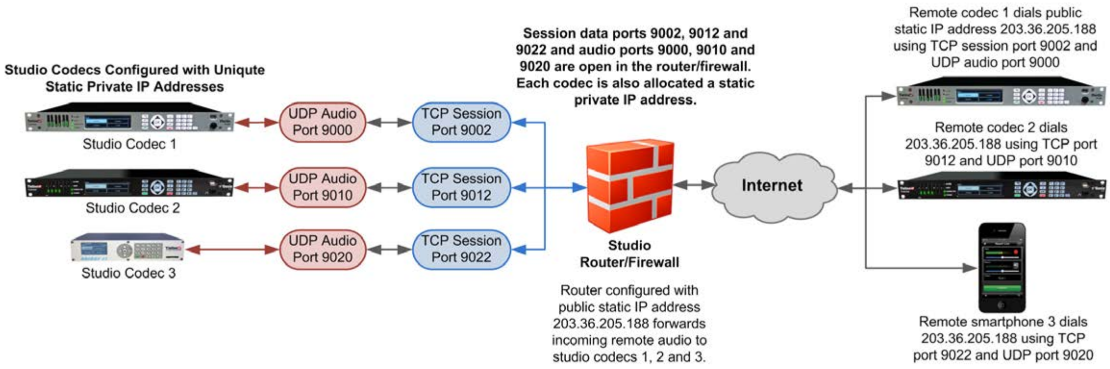
**Important Note:** The IP address may change if the codec is allocated a DHCP IP address by the router and it loses power or is temporarily disconnected from the LAN. This will cause problems for remote codecs attempting to dial and connect.

5. Ensure your router's firewall is configured with the relevant TCP and UDP IP ports open to allow data traffic between your codec and the remote codec. The process is fairly simple if you use the following procedure:

   a. Connect to your router using a web-browser.
   b. Navigate to http://portforward.com/english/applications/port_forwarding/Tieline-G5/default.htm (Note: when configuring a Commander or i-Mix G3 codec at the studio use http://portforward.com/english/applications/port_forwarding/Tieline-G3/default.htm )
   c. Click to select your router manufacturer from the list.
   d. Next, click to select your router model from the list.
   e. Follow the instructions to complete port forwarding

6. Visit www.portforward.com and download the port checking application to verify your router's ports are open.
7. Configure the static IP address in your codec using the instructions in the next section. To allow multiple codecs to share a single public static IP address behind a firewall and route

the calls correctly, your codecs and the firewall need to be configured similarly to the example diagram which follows. Ensure the port, IP address, Subnet Mask and Gateway settings in your codecs match those configured in your router.




**Port Forwarding to 3 Studio Codecs Sharing a Public Static IP Address**

**Important Note:**

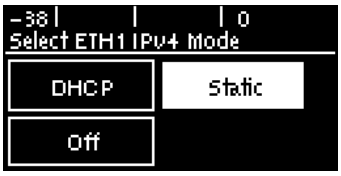
- The most common studio configuration issue is a firewall which blocks the incoming and/or outgoing TCP and UDP ports, or not configuring NAT and port forwarding correctly. The following table lists the firewall ports you need to open for each model of Tieline codec if they are dialing your router at the studio. If the remote codec is also connected to a LAN with a firewall you may also need to open the ports at the remote end of the link to connect successfully.
- Some firewalls require symmetric port configuration. The codec supports configuration of the "send" audio port (codec port at the remote end of the link to which you are sending audio) and "return" audio port (port used by the local codec to receive audio from the remote codec).

| Firewall Ports | | | | | | | | |
|---|---|---|---|---|---|---|---|---|
| Commander G3 / i-Mix G3 | | Bridge-IT / Bridge-IT XTRA | | Merlin and Genie Codec Families | | ViA Codec | | Cloud Codec Controller |
| TCP | UDP | TCP | UDP | TCP | UDP | TCP | UDP | TCP |
| IP1 Session Port: 9002 | IP1 Audio Port: 9000 | Session Port (Sess): 9002 | Audio (Proto): 9000 | Session Port: 9002 | Audio Port Stream 1: 9000 | Session Port: 9002 | Audio Port Stream 1: 9000 | HTTP 80 |
| IP2 Session Port: 9012 | IP2 Audio Port: 9010 | Web-GUI: 80 | SIP Session: 5060 | Alternative Session: 9012 | Audio Port Stream 2: 9010 | Alternative Session: 9012 | Audio Port Stream 2: 9010 | HTTPS 443 |
| Toolbox Software: 5550 | Toolbox Software: 5550 | Alternative Session: 9012 | SIP Audio: 5004 | Web-GUI: 80 | Audio Port Stream 3: 9020 | Web-GUI: 80 | SIP Session: 5060 | |
| | SIP Session: 5060 | Alternative Web-GUI: 8080 | Fuse-IP 8999 | Alternative Web-GUI: 8080 | Audio Port Stream 4: 9030 | Alternative Web-GUI: 8080 | SIP Audio: 5004-5054 | |
| | SIP Audio: 5004 | TLS/SSL 443 | | TLS/SSL 443 | Audio Port Stream 5: 9040 | TLS/SSL 443 | Fuse-IP 8999 | |
| | | | | | Audio Port Stream 6: 9050 | | | |
| | | | | | SIP Session: 5060 | | | |
| | | | | | SIP Audio: 5004-5054 | | | |
| | | | | | Fuse-IP 8999 | | | |

## Configuring a Static Public or Private IP Address in Bridge-IT (v.2.x firmware)

To enter a static IP Address into the codec for NAT:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Settings** and press .
3. Use the down navigation button to select **Network** and press .
4. Select **LAN** and navigate to **IPv4** mode and press .
5. DHCP is enabled by default. Select **Static** and press .




6. The Static IP address menu is revealed after DHCP is disabled. Use the navigation buttons to select **v4 Static** and press .

7. Use the numeric **KEYPAD** to enter the IP address and press OK to store the setting. Note: use the * or # buttons to enter the periods in the IP address and use the **RETURN** button to delete any numbers already entered.
8. Enter changes to the **v4 Subnet** (Subnet Mask) or **v4 Gway** (Default Gateway) in the same way if they are required (check with your network administrator for these settings).
9. After all changes have been made use the navigation buttons to scroll to the top of the menu and select **Apply Setting**, then press the OK button to save all changes.
10. From the **Home** screen select **Settings > Unit > Eth1** in the codec menus to ensure the new static IP address has been entered correctly.

## Configuring a Static IP Address in Commander G3 and i-Mix G3 Codecs

To set up a static IP address in Commander G3 and i-Mix G3 codecs select **Menu > Configuration > Advanced > LAN settings > IP Setup > Setup > Static > IP Address > [enter IP address] > press OK > Subnet Mask [enter Subnet Mask] > press OK > Gateway [enter Gateway] > press OK > reboot the codec**.

### Record IP Address Details

| **IPv4 Static IP Address** | |
|---|---|
| IP Address | . . . |
| Subnet Mask | . . . |
| Default Gateway | . . . |
| **IPv6 Mode: Manual** | **(Bridge-IT, Genie and Merlin codecs only)** |
| IP Address | : : : : : : : |
| IPv6 Prefix Size | |
| IPv6 Gateway | : : : : : : : |

## Getting Connected

Once the studio codec is configured you are now ready to receive an incoming call from the remote codec over the internet. Always dial from the field codec to the studio codec over the internet unless the remote codec is assigned a public static IP address and you know this address.

If you dial the studio using a cell-phone data network at the remote site you will not normally experience any firewall or port blocking issues at the remote end of the link using default Tieline ports.

## Troubleshooting: How to Determine Where Firewall Port Blocking is Occurring

If you find you are unable to either send or receive audio between the studio and remote codecs you can use Tieline's Link Quality reading to diagnose where ports are being blocked. LQ can be displayed on the front LCD screen of Tieline's Bridge-IT, Merlin and Genie codecs by selecting **Cxns**, then select the connection you want to view and press the **OK** button. LQ readings are also displayed on the home screen of all Commander and i-Mix G3 codecs.

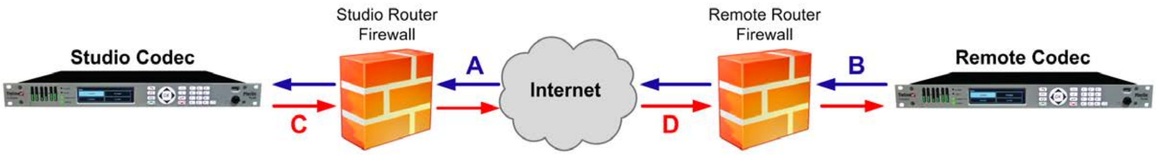
### Link Quality (LQ) Readings

Send and Return LQ numbers help you to determine if a problem is occurring at either end of a connection. For example, on an IP connection the Return LQ reading represents the audio being downloaded from the network locally (i.e. audio data is being sent by the remote codec). Conversely, the Send LQ reading represents the audio data being sent by the local codec (i.e.

being downloaded by the remote codec). To ensure a stable connection, try to maintain a reliable reading of 80 or higher for both the **Send** and **Return** LQ reading.

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

**Diagnosing Port Blocking via the Studio Codec LQ**
If the studio codec **Return** LQ reading is **01** then incoming audio from the remote codec is being blocked by a firewall at either point A or B in the following diagram. If the studio codec **Send** LQ reading is **01** then outgoing audio from the studio is being blocked by a firewall at either point C or D in the following diagram.




**Diagnosing Port Blocking via the Remote Codec LQ**
If you attach your Tieline codec at the remote site to a LAN with access to the internet you can often dial and connect to the studio without any problem. It is less likely that a firewall will block outgoing TCP and UDP ports. However, if there is a firewall at the remote site it may block incoming data packets from the studio.

The principle is the same at the remote codec for diagnosing blocked ports. If the remote codec **Return** LQ reading is 01 then incoming audio from the studio codec is being blocked by a firewall at either point C or D in the preceding diagram. If the remote codec **Send** LQ reading is 01 then the outgoing audio from the remote codec is being blocked by a firewall at either point A or B in the preceding diagram.

## Troubleshooting TCP Port Blocking

Error messages on the codec screen can help to diagnose TCP port blocking.

1. **"Connection Refused"** usually means that the firewall is configured correctly but the codec is not using the expected port. For example, the firewall is set up to forward via port 9002 but codec is 'listening' to port 10,000. “Connection Refused” is not normally shown if the firewall is not configured correctly because a firewall will by design silently drop any forwarding requests to ports that it doesn’t have open (see next point). Note: "Connection Refused" will also be displayed if the Commander G3 or i-Mix G3 codec you are calling is already connected.
2. **“Connection Timeout”** can mean one of two things:
   - The firewall is not configured correctly and the attempted codec connection is being silently dropped, e.g. a remote codec is dialing to port 9002 but the studio firewall port forwarding is not configured.
   - The UDP port is not port forwarded correctly. Tieline codecs send test data during connection establishment to make sure that the audio path is configured correctly; if this process fails then it will also result in a “Connection Timeout”.

## How do I determine which end is blocking data flow?

Tieline test codec firewalls have the default Tieline TCP and UDP ports open. You can dial into these test codecs (or other codecs you know are configured correctly) from your recently configured studio and remote codecs and use the LQ readings to diagnose whether your studio or remote codec firewall is blocking your data packets. If one codec connects ok and the other one doesn't, then you will know which end is likely to be causing the problem. As an example:

1. Dial from site 1 to a Tieline test codec.
2. Dial from site 2 to Tieline test codec.

If both of these connect successfully then the “outbound” TCP path for session data is OK, and the inbound UDP audio path is OK.

3. Dial to site 1 from a codec you know is configured correctly.
4. Dial to site 2 from a codec you know is configured correctly.

If either of these calls fail then TCP and/or UDP inbound data is being blocked on the failed connection (see "Troubleshooting TCP Port Blocking" above).

### Testing your Codec

- Visit http://www.tieline.com/Support/Test-lines for a list of test IP codec addresses you can use to verify your codec is configured correctly.
- See Testing IP Network Connections for more IP test information.

### Learning More About IP Networks

For more IP network information please see the section titled Understanding IP Networks which discusses:

- Private versus public IP addresses.
- Static versus DHCP assigned IP addresses.
- Network Address Translation (NAT), port forwarding and firewalls.

## 18.2 Understanding IP Networks

### Types of IP Addresses Available

| | Type of IP Address | How the IP Address is Allocated | Description |
|---|---|---|---|
| **Public** | Static Public IP Address | Internet Service Providers (ISPs) | ISP’s allocate a static public IP address to allow network devices to communicate with each other over the internet. It works like a public telephone number and will allow your remote codec to call your studio codec over the Internet. |
| | Dynamically Assigned Public IP Address | Internet Service Providers (ISPs) | ISP’s usually allocate dynamically (automatically) assigned public IP addresses to allow network devices to communicate with each other over the Internet. (Not recommended for studio installations because each time you connect to your ISP the IP address can change). |
| **Private** | Dynamically Assigned Private IP Address | DHCP Server/Router on your own private LAN network. | A DHCP server-allocated IP address that is automatically assigned to a device on a LAN to allow it to communicate with other devices and the internet. This address can change each time a device connects. |
| | Static Private IP Address | LAN Administrator | A network administrator-allocated static address which is programmed into a device to |

| | | | |
|---|---|---|---|
| | | | allow it to connect to a LAN. Often a security measure to only allow access to devices approved by a network administrator. |

## Obtaining Public IP Addresses

To send audio streams over the public internet you need to use a public IP address assigned to you by your ISP (Internet Service Provider).

A public IP address is like your public telephone number and allows you to be contacted over the internet in much the same way people dial your public telephone number. They come in two forms; dynamic (DHCP) and static. Most ISPs assign a dynamic public IP address by default, which can often change without you knowing. This is suitable for a quick demo of your Tieline codec, but for a permanent installation you will need to request a permanent static public IP address.

Once the Static Public IP address is assigned to your internet connection (router) at the studio you need to create a link between the public IP address and your codec's private IP address on the LAN. This is called Network Address Translation.

Depending upon how your network is configured, it may also be possible to simply connect your Tieline codec directly into your ADSL modem/router and receive a public address from the router.

## Private LAN IP Addresses

By default your Tieline codec will normally be automatically assigned a private IP address when you connect it to a typical router over a LAN.

Private IP Addresses are associated with LANs and normally reside behind a firewall and are not visible to the internet. They are generally in the ranges: 10.0.0.1 – 10.255.255.255, 169.254.0.0 – 169.254.255.255, 172.16.0.0 – 172.31.255.255 and 192.168.0.0 – 192.168.255.255 and are assigned by network DHCP servers and routers.

These IP Addresses are generally assigned for a predefined period (known as a lease) by your network's DHCP server or router. This IP address will generally expire after the lease period. DHCP assigned IP Addresses may also change if the device is disconnected for lengthy periods or if power to the device is turned off and back on. As a result, it is advised that you make this IP address permanent by assigning it as a Static DHCP IP Address. This will ensure you are able to always forward incoming audio packets to your codec using the same private IP address at the studio using port forwarding (see the section on port forwarding for more details). Consult your Network Administrator if you are unsure how to do this.

## Network Address Translation (NAT)

Network Address Translation (NAT) is a method of connecting multiple devices to the internet using one public IP address.

The best way to explain NAT is to use the example of a phone system at an office that has one public telephone number and multiple extensions. This type of telephone system allows people to call you on a single public telephone number and performs the translation and routing of the public number to a particular private extension. Similarly, in order to receive an IP call from a remote codec over the public internet, the same network address translation principle applies. NAT and port forwarding allows a single device, such as a broadband router, to act as an agent between the public internet and a local private LAN.

The relationship between public and private IP addresses and NAT is displayed in the following diagram and the following section explains port forwarding configuration in more detail.

### Port Forwarding: Tieline TCP and UDP Port Settings

For your Tieline Codec to communicate over the public internet an IP Address alone is not sufficient. In TCP/IP and UDP networks the codec port is the endpoint of your connection. Ports are doorways for IP devices to communicate with each other. Picture a house and imagine the front door is the entry point represented by a public or private IP address. Then you want to get to several codecs in different rooms of the same house and ports represent the doors to each of those rooms. In principle this is how port addressing works.

For example, several codecs may dial into your studio using the same public static IP address. In this situation it is necessary to configure codec 'programs' with audio streams using different audio ports for discretely routing each incoming and outgoing audio stream. By doing this your studio's network routers know where IP packets for each audio stream should be routed, i.e. to which codec and respective audio outputs.

When data packets are received from remote codecs at a particular public IP address, port information is translated from data packets to ensure the correct packets are sent to the correct studio codecs. This process is performed by PAT (Port Address Translation), which is a feature of NAT (Network Address Translation) devices.

Tieline codecs use TCP ports for setting up the communication session and UDP ports for streaming audio. While TCP ports are generally open, UDP ports are generally blocked by network devices which contain firewalls and will stop you delivering your audio. Depending on the codecs you are using, you need to configure your firewall to allow TCP and UDP protocols to pass through the ports listed in the table below.

## 18.3 Tips for Creating Reliable IP Connections

The following 10 tips are provided to help obtain the best possible IP connection between two codecs, without paying for Quality of Service (QoS).

1. Always use the best quality Internet Service Provider (ISP). Tier 1 service providers are best as their infrastructure actually makes up the internet ‘backbone’. Wikipedia lists the major service providers that make up the internet backbone at: http://en.wikipedia.org/wiki/internet_backbone. In Australia Telstra is equivalent to one of these service providers.

2. You will get the best quality connection if both the local (studio) and remote codecs use the same ISP. This can substantially increase reliability, audio bandwidth and reduce audio delay. Using the same service provider nationally can give better results than using different local service providers. This is especially true if one of the service providers is a cheap, low-end domestic service provider, which buys its bandwidth from other ISPs. Second and third tier providers sublease bandwidth from first tier providers and can result in connection reliability issues due to multiple switch hops. We also highly recommend using First Tier ISPs if connecting two codecs in different countries.

3. Sign up for a business plan that provides better performance than domestic or residential plans. Business plans typically have a fixed data limit per month with an additional cost for data beyond that limit. In addition, Service Level Agreements (SLA) will often provide better support and response times in the event of a connection failure. Domestic plans are often speed-limited or “shaped” when usage exceeds a predefined limit. These plans are cheap but they are dangerous for streaming broadcast audio.

4. Ensure that the speed of the connection for both codecs is adequate for the job. The minimum upload speed recommended is 256 kbps for a studio codec and 64 kbps for a field unit connection.

5. Use good quality equipment to connect your codecs to the internet. (Tieline successfully uses Cisco® switching and routing equipment.):

- If you are using a DSL or ADSL connection make sure you purchase a high quality modem that can easily meet your speed requirements. This is especially important if you are over 4 kms from an exchange.
- If you have multiple codecs connected to a local area network (LAN) please ensure that your network infrastructure is designed for media streaming and not domestic usage. Tieline has tested several cheap 8-port switches that lose more packets between local computers than an international IP connection between Australia and the USA!
- If using a wireless connection ensure that the antenna signal strength received is strong. The type of antenna used and the amount of output gain also affects connection quality.

**Important Note:** You should be able to stream audio between two codecs on your LAN and get 'link quality' readings of S99R99. If you see anything less than this then you should get a network engineer to investigate the issue.

6. Once your internet connection is installed at the studio check that the connection performance is approximately what you ordered and are paying for. A connection can perform below advertised bit rates if:

- There is an error in ISP configuration;
- There is an error in modem configuration;
- There is a poor quality line between the studio and the exchange;
- There are too may phones or faxes connected to the phone line; or
- Line filters have been connected incorrectly.

You can test your internet connection speed by connecting a PC to the internet and using http://www.speedtest.net/index.php. If the bandwidth detected is low then something is wrong. Get it fixed before going live!

7. Use a dedicated DSL/ADSL line for your codecs. Do not share a link with PCs or company networks. The only exception to this rule is if an organisation has network equipment and engineers that can implement and manage quality of service (QoS) on its network.

8. Use UDP as the preferred audio transport protocol. TCP generally results in lower bit rates and random drop-outs of audio over the internet. Only use TCP if UDP is blocked by firewalls and you are unable to connect.

9. When using UDP ensure the total bit rate (audio bit rate plus header bit rate) is no more than 80% of the ISP connection rate. IP headers require around 20 kbps in addition to the audio bit rate. For example, with a 64 kbps connection the audio bit rate should be (64-20) x 0.8 = 31.2 kbps or lower. For TCP we suggest a limit of 50% or less.

10. Wireless IP connections can easily become congested and result in packet loss and audio drop-outs. It is very difficult to guarantee connection quality when there is no way of knowing how many people are sharing the same wireless connection.

**Important Note:** Be careful when using cell-phone connections at special events where thousands of people have mobile phones. This can result in poor quality connections and audio drop-outs if cell-phone base stations are overloaded.

## IP Connection Checklist

Complete the following check list and aim for a score of at least 8 out of 10 before going live.

| Number | Check | Result |
|---|---|---|
| 1 | Using a reputable Tier1 ISP that's part of internet backbone. | |
| 2 | The same ISP is being used for both codec connections. | |
| 3 | The ISP Plan is a Business Plan or equivalent. | |
| 4 | The ISP connection speed is adequate. | |

| | | |
|---|---|---|
| 5 | Equipment is high quality and suitable for media streaming. | |
| 6 | The ISP connection speed has been tested and is suitable. | |
| 7 | The ISP connection is not shared with other PCs or devices. | |
| 8 | UDP is being used as the audio transport protocol. | |
| 9 | No more than 80% of ISP connection bandwidth is being used. | |
| 10 | There are no wireless connections being used. | |

# 18.4 Testing IP Network Connections

There are a few very simple tools that you can use to test whether a codec can be reached over an IP network.

- Visit http://www.speedtest.net/ to test the upload and download speed of your IP connections and identify your public IP address.
- Visit http://www.ipfingerprints.com/portscan.php to verify your router's ports are open. Note: Using a port scanner to test a codec will be unsuccessful if you try to scan and the port is already in use, i.e. the codec is connected.
- Visit www.subnetonline.com and use an online port scanner to check for open and closed TCP ports. This site also has numerous other software tools, including an online ping web-tool for IPv4, plus TraceRoute and TracePath software tools.

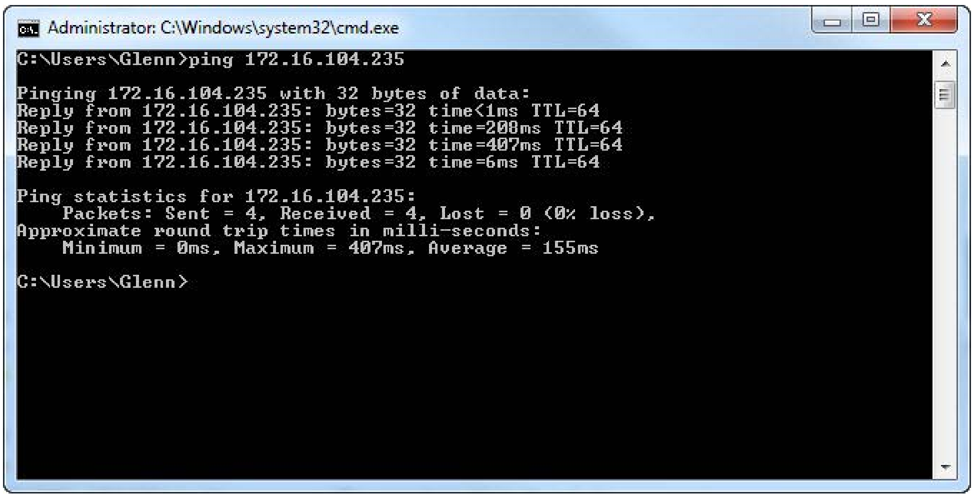
## Ping the Codec

A ping test can be used to test whether it is possible to reach a codec or any device over an IP network. A ping test measures:

- The round-trip time of packets.
- Any packet loss.

There are two types of ping tests:

1. **Short test:** sends 4 packets and delivers statistics.
   i. Point to the **start** menu on your PC and click once.
   ii. In the search text box type **Run** and press **Enter.**
   iii. Type **CMD** in the **Run dialog** text box and click **OK**.
   iv. Type **ping** and the IP address of the codec you are pinging (i.e. **ping 192.168.0.159**) and press the **Enter** key on your keyboard.
   v. The round trip time of the packets is displayed, as well as any packet loss.



2. **Long test:** sends packets continuously until stopped.
   i. Point to the **start** menu on your PC and click once.
   ii. In the search text box type **Run** and press **Enter.**
   iii. Type **CMD** in the **Run dialog** text box and click **OK**.
   iv. Type **ping**, the IP address of the codec you are pinging, and then **-t** (i.e. **ping 203.36.205.163 -t**) and press the **Enter** key on your keyboard.
   v. Let the test run for several minutes and then press **CTRL C**.
   vi. The round trip time of the packets is displayed, as well as any packet loss for the period of time that the test occurred.

## Trace the Route of IP Packets

Another utility available on your PC is traceroute. This tool can be used to determine the route and number of hops that data packets are taking to their destination (codec). This is useful because the more routers that packets traverse, the more latency your connection will have, and the less reliable it will be.

i. Point to the **start** menu on your PC and click once.
ii. In the search text box type **Run** and press **Enter.**
iii. Type **CMD** in the **Run dialog** text box and click **OK**.
iv. Type **tracert**, the IP address of the codec you are contacting (i.e. **tracert 203.36.205.163**) and press the **Enter** key on your keyboard.

## 18.5 Bridge-IT Declaration of Conformity

**COMMUNICATION CERTIFICATION LABORATORY**
1940 West Alexander Street
Salt Lake City, UT 84119
801-972-6146

**Test Report**

Declaration of Conformity

TEST OF:

BRIDGE-IT

To EN 55024:1998

Test Report Serial No: 2321

Applicant:

Tieline Technology
25 Irvine Drive
Malaga, WA Australia 6090

Date of Test: September 14 – 15, 2009

Issue Date: September 15, 2009

Accredited Testing Laboratory By:

NVLAP

NVLAP Lab Code 100272-0

**COMMUNICATION CERTIFICATION LABORATORY**

**1940 West Alexander Street**
**Salt Lake City, UT 84119**
**801-972-6146**

## Test Report

Declaration of Conformity

Test Of:

BRIDGE-IT

Test Specifications:

EN 55022: 2006
FCC Part 15, Subpart B
ICES-003

Test Report Serial No: 2320

Applicant:

Tieline Technology
25 Irvine Drive
Malaga, WA Australia 6090

Date of Test: September 9, 2009

Issue Date: September 10, 2009

Accredited Testing Laboratory By:

NVLAP Lab Code 100272-0

## 18.6 Compliances and Certifications

### FCC Compliance Notice

This equipment has been tested and found to comply with the limits for a class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area may cause harmful interference, in which case the user will be required to correct the interference at his/her own expense. There is no guarantee, however, that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment to an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio or TV technician for help.

**CAUTION:**
Changes or modifications to this unit not expressly approved by the party responsible for compliance could void the user's authority to operate this equipment.

### Declaration of Conformity

The Tieline Bridge-IT IP codec meets the requirements of directives for CE and C-Tick certifications. Technical documentation required by the conformity assessment procedure is kept at the head office of Tieline Technology; 1/25 Irvine Drive, Malaga, Western Australia 6090.

### EN 55 022 Statement

This is to certify that Tieline Bridge-IT is shielded against the generation of radio interference in accordance with the application of EN 55 022: 2006 Class A. Technical documentation required by the conformity assessment procedure is kept at the head office of Tieline Technology; 1/25 Irvine Drive, Malaga, Western Australia 6090.

### Canadian Department of Communications Radio Interference Regulations

This digital apparatus (Tieline Bridge-IT) does not exceed the Class A limits for radio-noise emissions from digital apparatus as set out in the Radio Interference Regulations of the Canadian Department of Communications.

### Règlement sur le brouillage radioélectrique du ministère des Communications

Cet appareil numérique (Tieline Bridge-IT) respecte les limites de bruits radioélectriques visant les appareils numériques de classe A prescrites dans le Règlement sur le brouillage radioelectrique du ministère des Communications du Canada.

### Safety of Electrical and Electronic Products and Components

The IECEE CB Scheme is an international system for mutual acceptance of CB test reports and certificates covering the safety of electrical and electronic products and components. The IEC CB is a multilateral scheme among participating countries and certification organizations, based on the use of international (IEC) standards.

This product has been tested by an independent certifying company and has been certified to comply with IEC 60950-1(ed.2);am1

## 18.7 Trademarks and Credit Notices

1. Windows is a registered trademark of Microsoft Corporation in the United States and/or other countries.
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8. Java is a trade mark Sun Microsystems Inc. in the United States and/or other countries.
9. Other product names mentioned within this document may be trademarks or registered trademarks, or a trade name of their respective owner.

# 19 Specifications

| Input/Output Specifications | |
|---|---|
| Analog Audio Inputs | 2 x Female XLR (Channel 1 mic/line; channel 2 line only) |
| Analog Audio Outputs | 2 x Male XLR |
| AES3 In | 1 x female XLR (Channel 1 in; shared with Ch1 analog input) |
| AES3 Out | 1 x male XLR |
| Headphones | 1 x 6.35mm (1/4") Jack on rear panel |
| Control Ports In/Out | Two relay inputs and two opto-isolated outputs for machine control via Phoenix connector |
| Audio Input Impedance | High Impedance > 5K ohm |
| Output Impedance | <50 ohm Balanced |
| Clipping Level | +22dBu (input and outputs) |
| 24 bit A/D & D/A Converters | |
| Frequency Response | 20Hz to 20kHz |
| Total Harmonic Distortion | <0.0039% at +16dBu, or -88dBu unweighted |
| Signal To Noise Ratio | >90dB at +22dBu, unweighted |
| **Sample Frequencies** | |
| IP Sample Frequencies | 16kHz, 32kHz, 44.1kHz, 48kHz |
| **Algorithms** | |
| IP | Opus, Tieline Music, Tieline MusicPLUS, G.711, G.722, MPEG Layer 2 |
| IP (Pro version only) | AAC-LD, AAC-ELD, AAC-LC, AAC-HE, AAC-HEv2 |
| IP (Pro version only) | 16/24 bit Enhanced apt-X |
| IP (uncompressed) | Linear PCM16 |
| **Data and Control Interfaces** | |
| USB | USB 2.0 (Type B) host port on the rear panel |
| LAN | 10/100 base T RJ45 connector |
| Serial | RS232 up to 115kbps with or without CTS/RTS flow control via female DB9 connector, can be used as a proprietary data channel |
| Protocols / RFC support | Tieline, DHCP, SNMP, DNS, HTTP, IGMP, ICMP, VLAN, IPv4/v6, FEC, SIP/SDP (EBU N/ACIP Tech 3326 compliant), RTP, RTCP, STUN, SSL, CSRF, AES67, RFC5109, RFC5956, RFC5588, RFC4756, RFC3388, RFC5956, RFC 5588, RFC2733, RFC3190 |
| **Front Panel Interfaces** | |
| Display | 128 x 64 monochrome LCD |
| SD/SDHC Card Slot | Supports SDHC Flash Cards up to 32GB capacity |
| Keypad | 20 button keypad |
| Navigation | 5 button keypad |
| **General** | |
| Dimensions | 8 ½" x 5 9/10" x 1 ¾" (216mm x 150mm x 44mm) |
| Weight | 2.07lb/940g |
| Power Consumption | 12V DC, 400mA |
| Operating Temperature | 0°C to 50°C (32°F to 122°F) |
| Humidity Operating Range | 20% ≤RH ≤70% (0 to 35°C), non-condensing |
| Internal Battery | Panasonic CR2032, 3V coin type 20mm |

# Index

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# - E -



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# - G -

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# - L -



# - M -

# - N -



# - O -



# - P -



# - Q -



# - R -



# - S -

# - T -



# - U -



# - V -



# - W -