Tieline®

The Codec Company


# Merlin PLUS IP Codec
# User Manual

**Software Version: Build 2.8.30**
**Manual Version: v1.2_20140327**
**March, 2014**

# Table of Contents

# 1 Warnings & Safety Information

1. Both appliance power cables must be removed from the device for Power Disconnection.
2. Remove the phone cable from the POTS interface before servicing.

**THUNDERSTORM AND LIGHTNING WARNING:**

DO NOT USE Tieline codecs during thunderstorms and lightning. You may suffer an injury using a phone, Tieline codec, or any device connected to a phone during a thunderstorm. This can lead to personal injury and in extreme cases may be fatal. Protective devices can be fitted to the line, however, due to the extremely high voltages and energy levels involved in lightning strikes, these devices may not offer protection to the users, or 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 you are 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 remains connected to the phone or ISDN system, LAN 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.

**DIGITAL PHONE SYSTEM WARNING:**

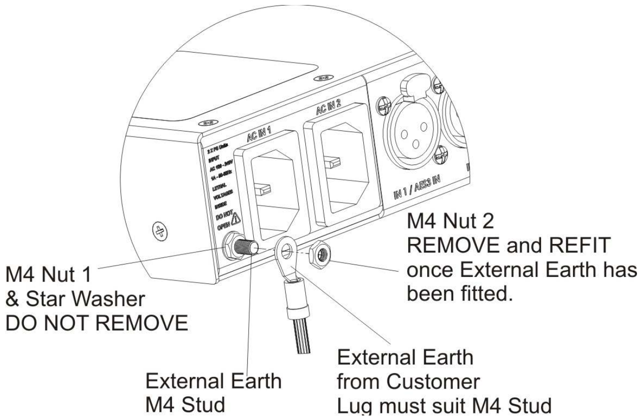
DO NOT CONNECT YOUR Tieline CODEC TO A DIGITAL PHONE SYSTEM. PERMANENT DAMAGE MAY OCCUR! If you are unfamiliar with any facility, check that the line you are using is NOT a digital line. If the Tieline codec becomes faulty due to the use of a digital phone system, the WARRANTY WILL BE VOID.

**WARNING:**

HIGH LEAKAGE CURRENT. EARTH CONNECTION ESSENTIAL BEFORE CONNECTING SUPPLY.
If the total leakage current exceeds 3.5 mA, or if the leakage current of the connected loads is unknown, connect the supplementary ground terminal to a reliable ground connection in your facility.

**Supplementary ground connection**

A supplementary ground terminal is provided on the codec to connect the unit to a ground connection. The ground terminal has an M4 stud with M4 retaining nuts and is compatible with all grounding wires. Remove only **NUT** 2 to connect your ground wire. The ground wire must have a suitable lug. When refitting **NUT** 2 ensure that both **NUT** 1 & **NUT** 2 are correctly tightened to establish and maintain a proper earth connection.




### Disclaimer

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

# 2 How to Use the Documentation

### Manual Conventions

**Warnings:** Instructions that, if ignored, could result in death or serious personal injury caused by dangerous voltages or 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.

IP Information specific to IP connections.

ISDN Information specific to ISDN connections.

### 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 *i* (information/help) button when navigating codec menus to display a dialog suggesting the actions which can be performed from within the current menu.

# 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 |
| **BRI** | Basic Rate Interface for ISDN services |
| **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 |
| **Fail over** | Method of switching to an alternative backup audio stream if the primary connection is lost. |
| **GUI** | Graphical User Interface |
| **ISDN** | Integrated Services Digital Network |
| **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 |
| **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 |
| **Network Address Translation (NAT)** | A system for forwarding data packets to different private IP network addresses that reside behind a single public IP address. |
| **Port Address Translation (PAT)** | Related to NAT; a feature of a network device that allows IP packets to be routed to specific ports of devices communicating between public and private IP networks |
| **PSU** | Power Supply Unit |
| **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 |

| | |
|---|---|
| **SNMP** | Simple Network Management Protocol |
| **SPID** | Service Profile ID for identifying devices over ISDN networks |
| **STL** | Studio-to-transmitter link for program audio feeds |
| **STS** | Studio-to-studio audio link |
| **TCP** | TCP protocol ensures reliable in-order delivery of data packets between a sender and a receiver |
| **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 |
| **WAN** | Wide Area Network; a computer network spanning regions and/or countries to connect separate LANs |

# 4 Getting to know Merlin PLUS

The 1RU Merlin PLUS rack mount IP codec is designed to rapidly expand your remote capability.

Merlin PLUS has the ability to manage up to 6 bidirectional mono connections with IP codecs or smartphones using Report-IT, saving money on codec hardware costs and reducing studio rack space requirements. It can also deliver high quality bidirectional stereo and full duplex communications for remote broadcast connections.


## Overview of this User Manual

Use this manual to learn how to:

- Configure codec 'programs' (please read About Program Dialing for more info).
- Adjust audio and connection settings within the codec.

Please read Getting Connected Quickly for an overview of how to adjust and store audio and connection settings in your codec using 'programs'.

## Applications

Merlin PLUS is ideal for studio and remote truck installations and is capable of:

- 6 simultaneous bidirectional mono remote connections.
- 1 or 2 Bidirectional mono or stereo remotes, each with a separate bidirectional IFB channel for communications.
- Simple local or remote command and control.
- Recallable codec configurations via ‘programs’.

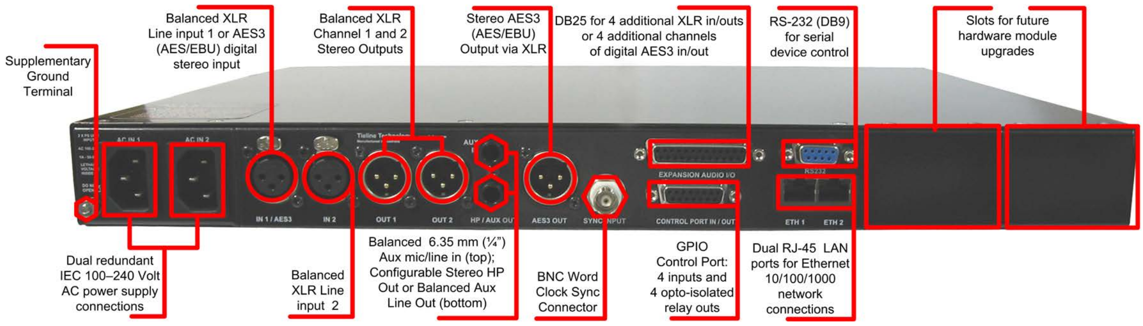
## Codec Features

- DSP-based architecture designed for continuous operation.
- Dual Gigabit (10/100/1000) Ethernet ports with automatic switching for redundancy.
- Auto switching, dual redundant AC power supplies.
- Uncompressed PCM audio plus the low-delay, cascade resilient aptX® Enhanced algorithm (capable of up to 24bit, 48kHz audio sampling)
- Other popular algorithms including LC-AAC, HE-AAC v1 and v2, AAC-LD, AAC-ELD, AAC-ELDv2, Opus, MPEG-1 Layer II and III, Tieline Music and MusicPLUS, G.722 and G.711.
- SmartStream PLUS redundant streaming for high reliability over IP networks without Quality of Service.
- IPv4 & IPv6 compatible and ready.
- Asymmetric algorithmic encode/decode*.
- Integrated alarm management.
- G5 Toolbox GUI enables remote codec control over WANs.
- Compatible with Tieline Codec Management System**.
- Low latency in-band RS-232 auxiliary data channel.
- Programmable software rules engine via a GUI for Control Port functions.
- Streamlined codec wizards and GUI for configuration and control.
- *Support for multiple languages: English, Spanish, Portuguese, French and Chinese.

- Connect to all Tieline IP codecs and Report-IT iOS/Android suite of Apps.

*  Supported in later releases.
** Separate product

# 5 Merlin PLUS Rear Panel Connections




## *XLR Analog and AES3 Inputs*

XLR **IN1/AES3** and **IN 2** are balanced line inputs.

Input 1 can also be used as an AES3 (AES/EBU) digital input. This input accepts both mono and stereo digital AES3 signals.

## *XLR Analog and AES3 Outputs*

XLR **OUT 1** and **2** are balanced analog audio line outputs.

**AES3 OUT** is an AES/EBU digital 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.

## *DB-25 Input/Output Expansion Connector*

A female DB25 connector can be used to attach a male DB25 Tascam (pin-out) breakout cable, providing 4 additional analog XLR in/outs, or 4 additional channels of digital AES3 audio in/out. Please note that expansion inputs/outputs 3, 4, 5 and 6 must be either all analog or all digital AES.

The additional audio inputs/outputs are grouped in pairs: inputs 3 and 4 and inputs 5 and 6. By default, if you program analog or digital audio in on expansion inputs then you will get the same output; i.e. analog in = analog out, or digital in = digital out. This can be adjusted if required. E.g. you can configure analog in on inputs 3, 4, 5 and 6 and AES3 digital out on outputs 3, 4, 5 and 6, or vice versa. Adjust the output setting independently if required via **SETTINGS** **> Audio > Output Type > [Select Outputs 3 to 6] > [Select output type].**

For analog sources use inputs 3, 4, 5 and 6. For digital AES3 sources use input 3 for channels 3 and 4 and use input 4 for channels 5 and 6. Similarly, for analog outs use outputs 3, 4, 5 and 6. For digital AES3 audio out use output 3 for channels 3 and 4 and use output 4 for channels 5 and 6.

DB25 pin outs are available in Appendix B.

## *Dual Gigabit Ethernet Ports*

The codec features two Gigabit (10/100/1000) RJ-45 Ethernet ports for IP connections. By default, the codec assumes **ETH1** is the primary LAN connection and **ETH2** is the backup LAN connection when in use. If you are only using one Ethernet port, always use **ETH1**.

## Aux Mic/Line Input

**AUX IN** 6.35mm (1/4") balanced auxiliary mic or line input.

## Headphone Out/Aux Line Out

**HP/AUX OUT** 6.35mm (1/4") software configurable stereo headphone output, or balanced auxiliary line output. The front panel **HEADPHONE** output and rear panel **HP/AUX OUT** share the same hardware output. This means both are switched and configured together. I.e. both outputs are either a stereo headphone output (default setting), or a balanced mono auxiliary output.

## Sync Input

BNC type **SYNC INPUT** for attaching Word Clock sync to the codec.

## Command & Control Interfaces

1. Four relay inputs and four opto-isolated outputs for machine control via the DB15 **CONTROL PORT IN/OUT** connector.
2. A nine pin female **RS-232** serial connection for local and remote control of equipment at either end of the link.

## Dual Redundant AC Power Inputs

The codec is powered by dual 100-240 volt redundant AC power supplies, which use standard IEC connectors.

## Dual Module Slots

Two additional module slots for inserting optional POTS or ISDN modules.

## Supplementary Ground Terminal

Supplementary ground terminal for connecting the unit to a ground connection. See Warnings and Safety Information for more details.

# 6 Inserting Hardware Modules

Two slots are available for inserting optional ISDN or POTS connection modules into the codec. The module slots are numbered as follows.


## *Inserting or Removing a Module*

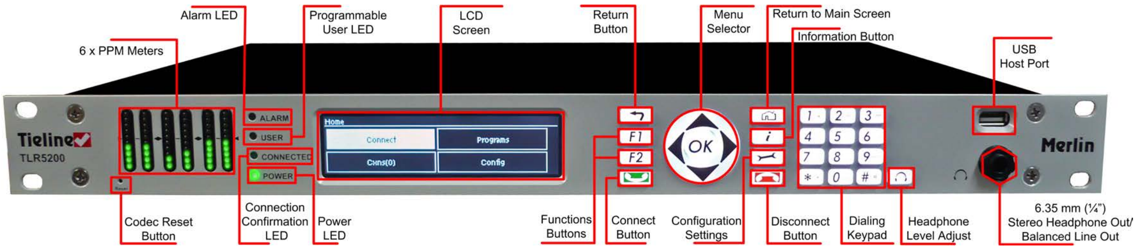
Ensure the codec is not powered up when inserting or removing modules. Where possible use anti-static precautions to help minimize the chance of static charges damaging the highly sensitive circuitry. Do not force a module into the codec. Modules should be installed slowly and gently.

1. Remove power from the codec and then remove the 4 screws from the blanking panel or module installed in the codec.
2. Carefully slide the new module into the module slot and ensure the base of the module remains flat during insertion, to ensure it lines up correctly with the module connector within the codec.
3. Reinsert the 4 screws to hold the module firmly in place.
4. Power up the codec.
5. Press the **SETTINGS** button to verify it is installed correctly.
6. Navigate to **Modules** and press the button.
7. The newly installed module should be visible as **Module 1** or **Module 2**.

**Important Note:** If the module does not appear in the **Modules** menu in the codec, it is possible that the connector on the module has not lined up correctly with the connector inside the codec. Remove the module and reinsert it carefully to resolve this issue.

# 7 Merlin Front Panel Controls

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




## Navigation Buttons

The codec has four arrow shaped navigation buttons for navigating codec menus and adjusting levels, and an **OK** button for selecting menu items.

## Dialing Keypad

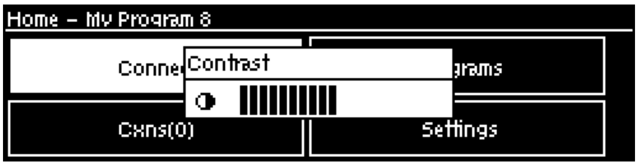
The keypad has alpha-numeric buttons, plus star and hatch (pound) buttons, which can be used to enter contact and program information into the codec.

## Operation Button Descriptions

| | Features | Operation Button Descriptions |
|---|---|---|
| | Return Button | Press to move back through menus & delete characters |
| F1 | Function Button 1 | Press to activate codec user functions |
| F2 | Function Button 2 | Press to activate codec user functions |
| | Connect Button | Press to create an IP connection |
| | Home Button | Press to return to home screen |
| i | Information Button | Press to view a help menu onscreen |
| | Settings Button | Press to adjust codec settings |
| | Disconnect Button | Press to end a connection |
| | Headphone Button | Press to adjust headphone audio levels |
| | Reset Button | Press to reboot the codec (feature not yet enabled) |

## Adjusting LCD Screen Contrast Levels

1. Press and hold the F1 button and then press and release 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**.

## *Stereo RTS Headphone Output*

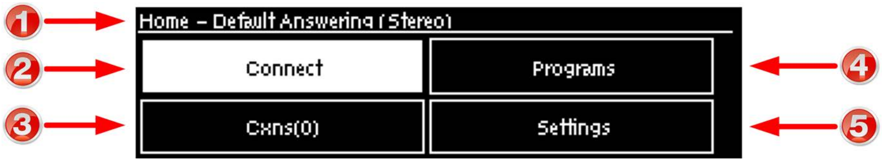
The codec has a 6.35mm (1/4") RTS stereo **HEADPHONE** output for audio monitoring and this can also be switched to a balanced mono auxiliary line output. The front panel **HEADPHONE** output and rear panel **HP/AUX OUT** share the same hardware output. This means both are switched and configured together. I.e. both outputs are either a stereo headphone output (default setting), or a balanced mono auxiliary output.

## *USB 2.0 Host Port*

USB 2.0 host port, which can be used for playback of backup audio files and firmware upgrades.

# 8 Navigating Menus

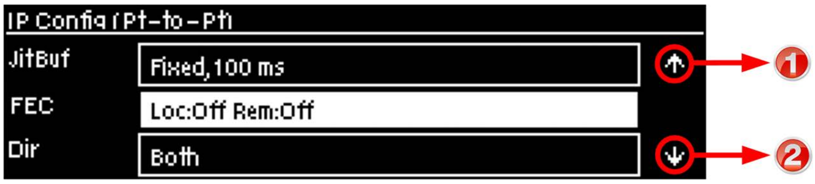
All main codec menus can be launched from the **Home** screen which includes:




| | Features | Codec Home Screen Elements |
|---|---|---|
| 1 | Screen Name | The name of the current screen |
| 2 | Connect | Select to connect and adjust connection settings |
| 3 | Cxns | Displays the number of current connections and connection details |
| 4 | Programs | View and edit Program configurations |
| 5 | 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.

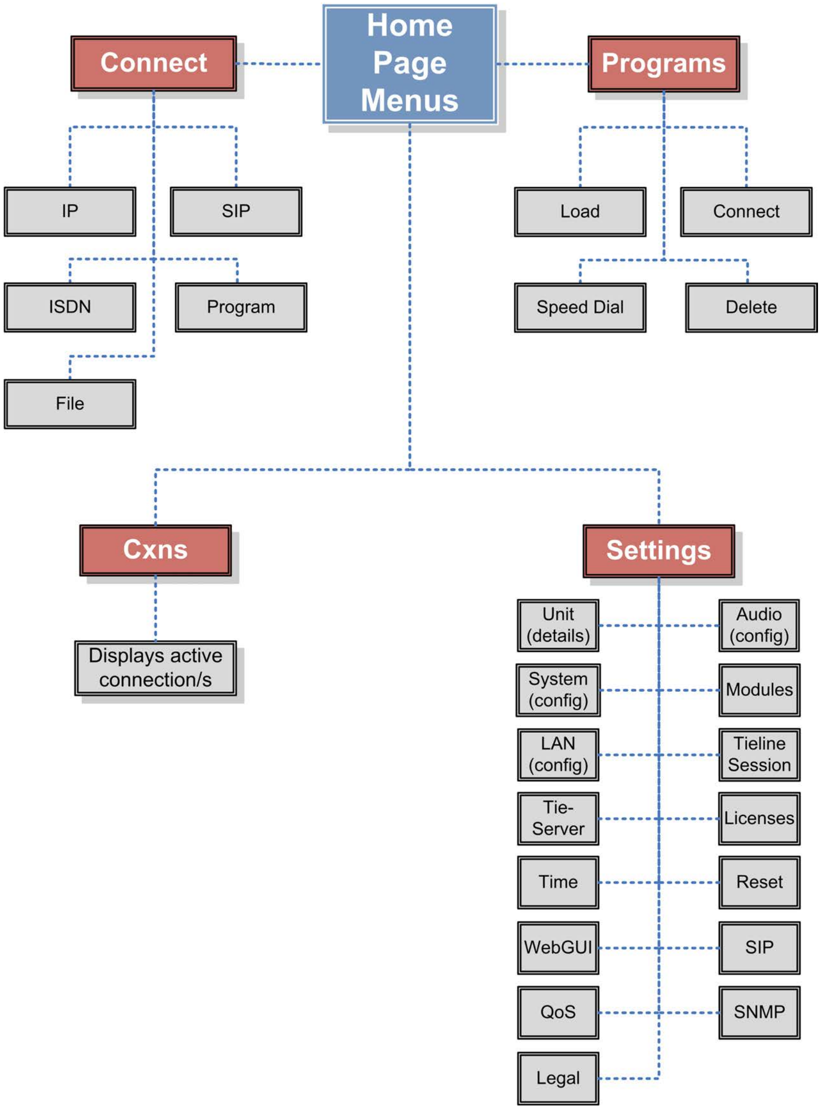
If a complete menu cannot be viewed on a single codec screen, arrows on the right hand side of the screen indicate that the current menu has options below and/or above the visible items. Use the navigation arrows to scroll up and down.



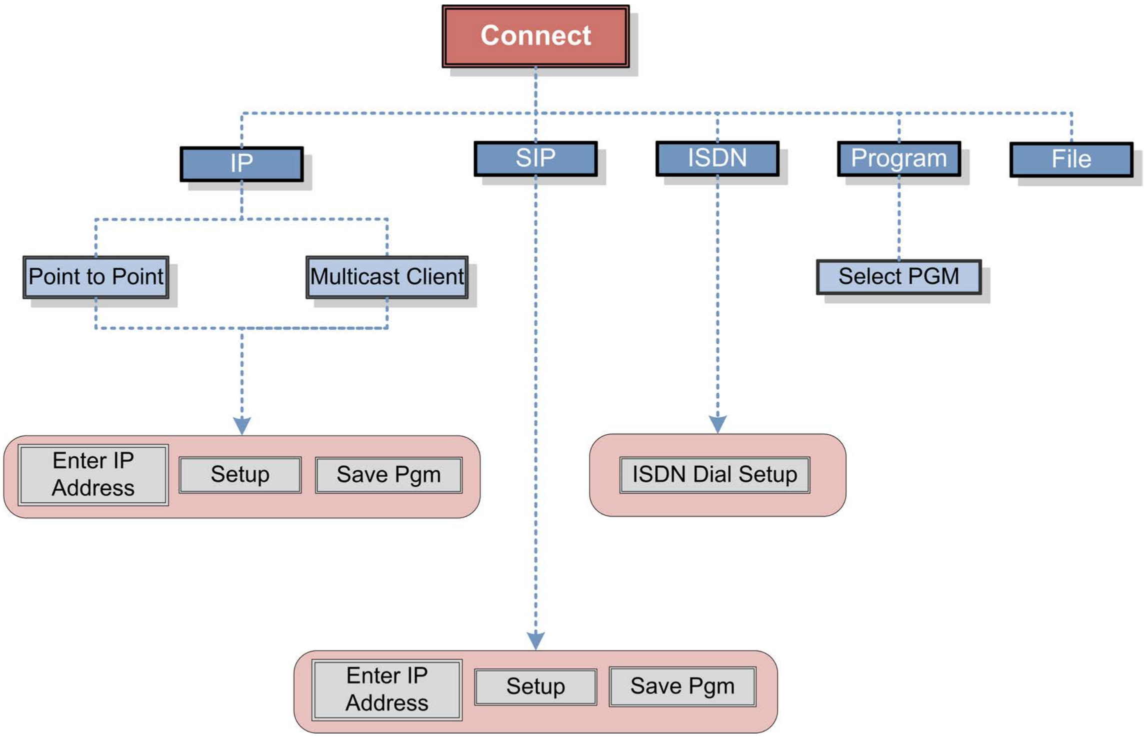

| | Features | Codec Home Screen Elements |
|---|---|---|
| 1 | Up Arrow | Arrow indicating menus can scroll upwards |
| 2 | Down Arrow | Arrow indicating menus can scroll downwards |

## *Codec Menu Overview*

Following is an overview of the codec menus. The **Connect** and **Settings** selections on the main screen provide a range of configuration settings. Note: file playback may not be supported in all codecs.

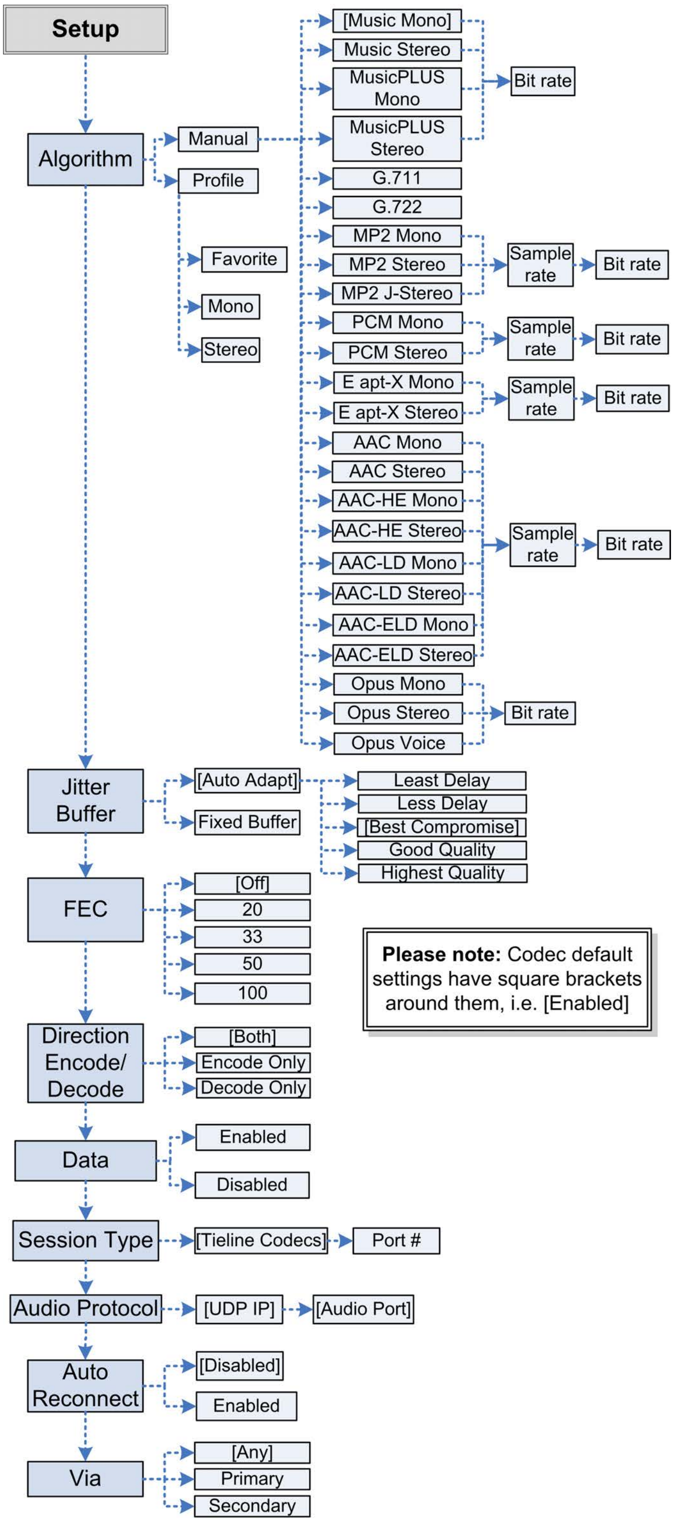


## Connect Menu



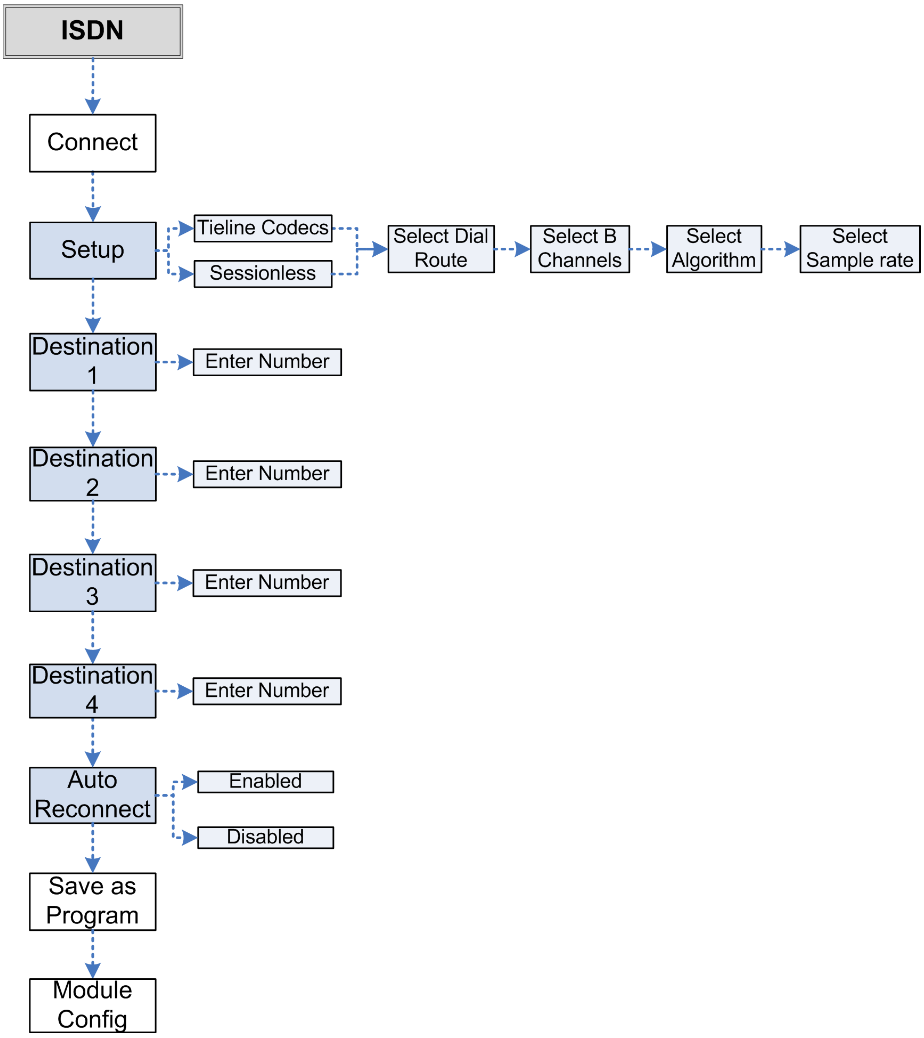
### *IP Setup Menu Navigation*

After selecting **IP** and a connection mode, or **SIP** in the **Connect** menu, select **Setup** to adjust connection settings.



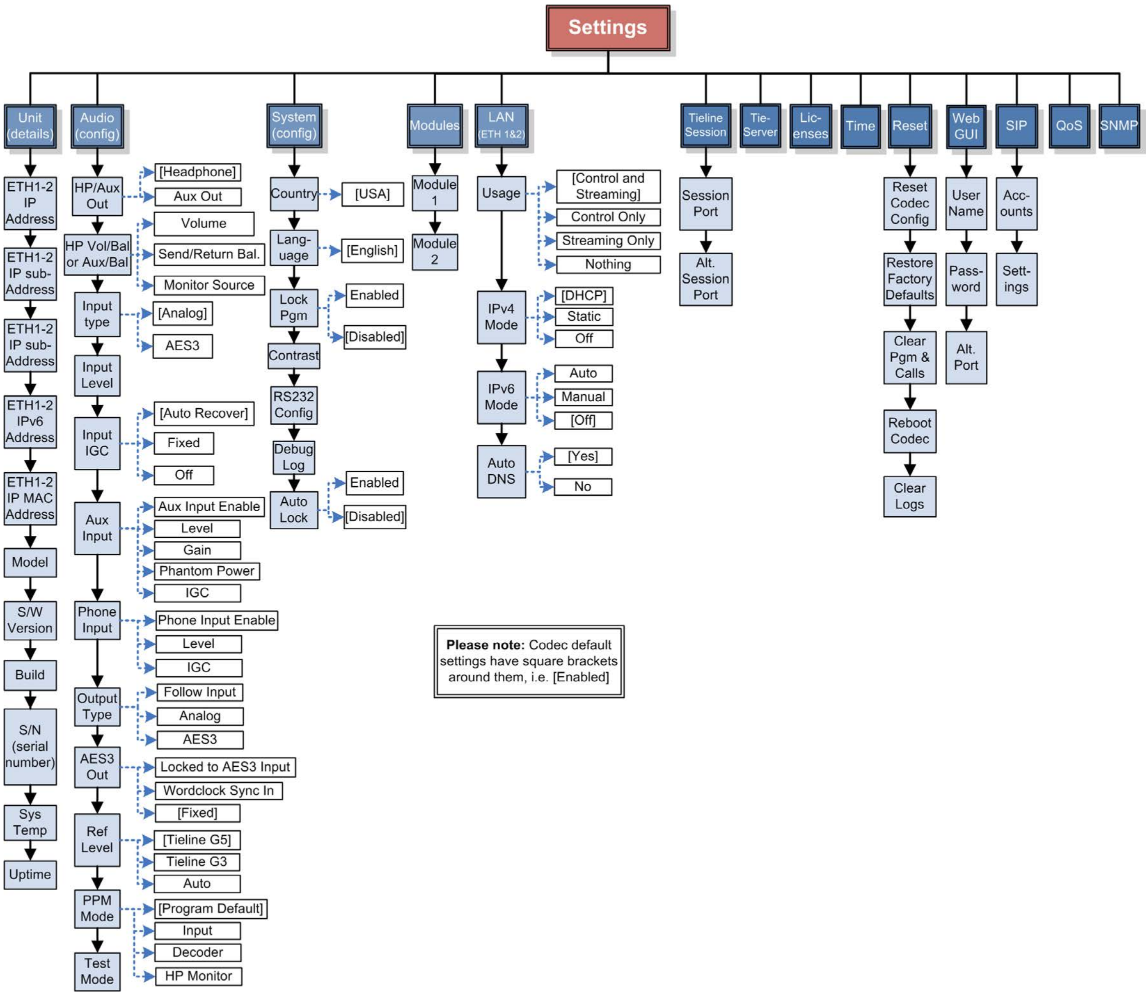
### ISDN Menu Navigation

Select **Connect** and then **ISDN** to configure ISDN dialing settings using the codec front panel.



## Settings Menu

Press the **SETTINGS** button on the codec front panel to access a wide range of configuration settings.



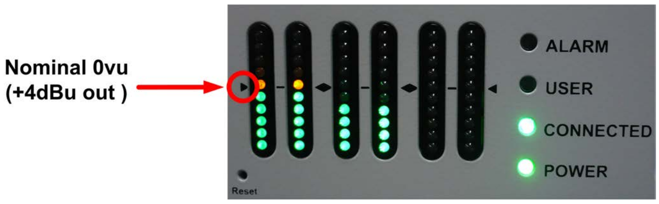
# 9 Merlin PLUS Input Levels and PPMs

**Important Note:**

- Input levels can only be adjusted on analog inputs. Digital AES3 source audio is not adjustable. See Configuring AES3 input audio for more information about digital in/outs. Input audio functions can also be configured using the Toolbox web-GUI. See Configuring Input/Output Settings.
- Merlin PLUS supports input expansion with inputs/outputs 3, 4, 5 and 6. Expansion inputs 3 to 6 must be either all analog or all digital. Expansion outputs 3 to 6 must also be either all analog or all digital. By default the output selection follows the input selection, however, inputs and outputs can be configured differently. For example, if you configure expansion inputs 3 to 6 as analog, you can configure all expansion outputs 3 to 6 as digital AES3. Adjust the output setting independently if required via **SETTINGS**  **> Audio > Output Type > [Select Outputs] > [Select Analog or Digital]**

## Audio Levels and Default PPM Metering

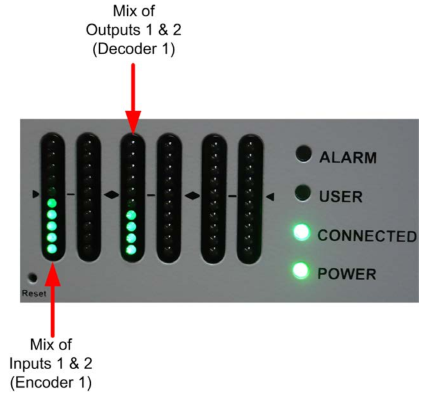
The PPM meters use dBu to express nominal operating, headroom and noise floor levels. Set audio levels so that audio peaks average at the nominal 0vu point indicated on the front panel PPM meters. This represents a program level of +4 dBu leaving the codec. Audio peaks can safely reach +22 dBu without clipping, providing 18dBu of headroom from the nominal 0vu point.



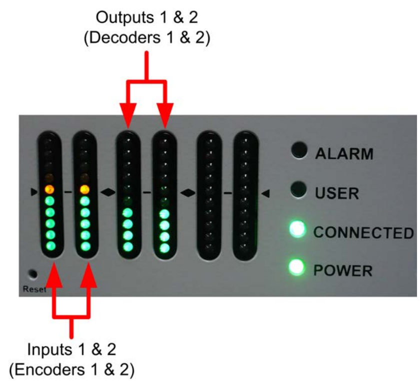

The following PPM settings are displayed by default in the codec. The default settings can be adjusted via **Settings > Audio > PPM Mode**.

### Mono and Stereo 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.



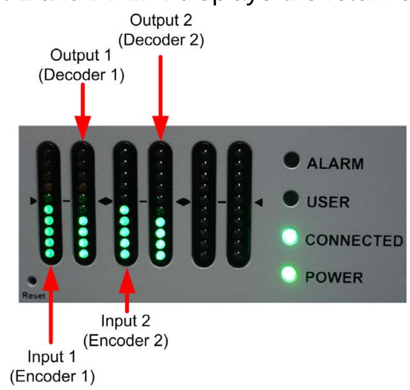

When connecting with a stereo program, the codec displays audio on **PPM1 & 2** for inputs 1 and 2 and **PPM 3 & 4** for the return program audio.




**Important Note:** When sending an audio stream to multiple endpoints using a multi-unicast program, the codec PPMs will display the same audio monitoring as for a default mono or stereo connection. The return audio PPM metering will display return audio from the first connection dialed within the multi-unicast program.

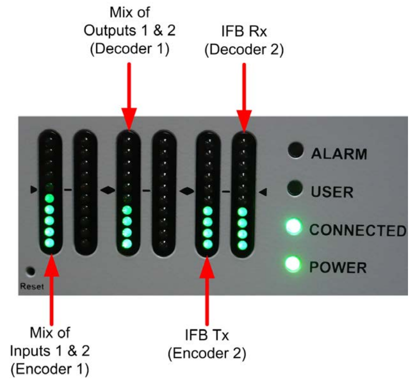
### *2 x Mono Peer-to-Peer Connection Metering*

The codec is capable of creating two independent mono audio stream connections simultaneously. In this situation the codec will display outgoing connection 1 audio on **PPM1** for input 1 and use **PPM 2** to display return audio. Outgoing connection 2 audio will be displayed on **PPM3** for input 2 and **PPM 4** displays the return audio.



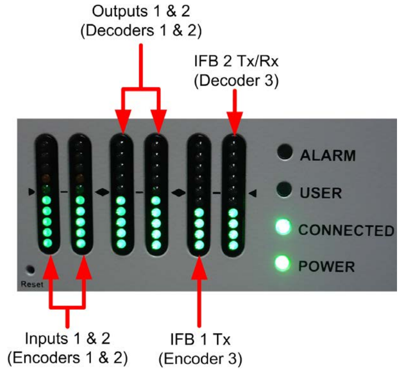

### *1 x Mono Peer-to-Peer + IFB Metering*

This program transmits a bidirectional mono audio stream and a separate bidirectional mono IFB communications audio stream. A mix of inputs 1 and 2 is displayed on **PPM1**. **PPM 3** displays the level of return audio. **PPM5** and **PPM6** display IFB audio in/out.



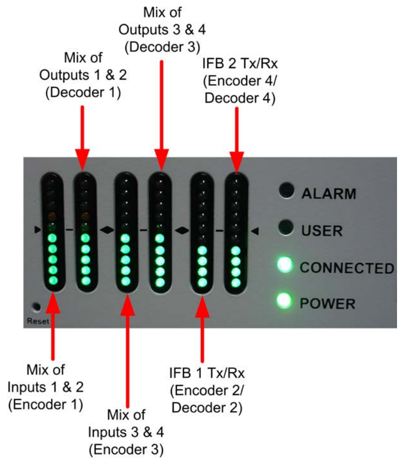
### *1 x Stereo Peer-to-Peer + IFB Metering*

This program transmits a bidirectional stereo audio stream and a separate bidirectional mono IFB communications audio stream. Incoming and outgoing audio for both audio streams is displayed on the PPMs.



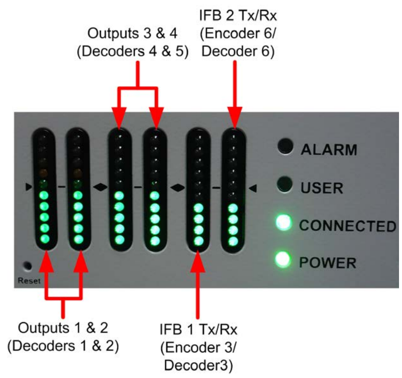

### *2 x Mono Peer-to-Peer + IFB Metering*

This program transmits two bidirectional mono audio streams and two bidirectional mono IFB audio streams. Incoming and outgoing audio for both audio streams is displayed on the PPMs. A mix of inputs 1 and 2 is displayed on **PPM1** and a mix of inputs 3 and 4 is displayed on **PPM3**. **PPMs 2 & 4** display each return audio stream. **PPM5** and **PPM6** display a mix of IFB audio in/out for each IFB audio stream.



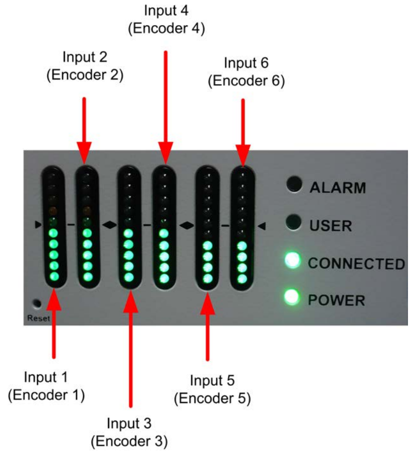

### *2 x Stereo Peer-to-Peer + IFB Metering*

This program transmits two bidirectional stereo audio streams and two bidirectional mono IFB audio streams. Decoder audio for both stereo audio streams is displayed on **PPMs 1-4**. **PPM5** and **PPM6** display a mix of IFB audio in/out for each IFB audio stream.



### Metering 6 Bidirectional Peer-to-Peer Mono Audio Streams

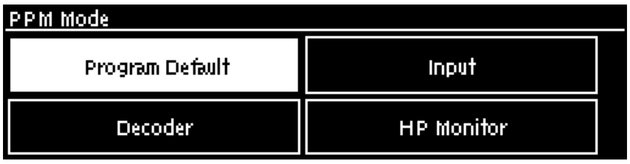
By default the PPMs display input (encoder) audio levels when up to 6 mono audio streams are configured within a program. Inputs 1 to 6 are mapped to **PPMs 1-6**.




## Adjusting Default PPM Metering

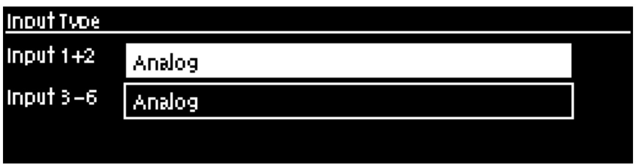
The default PPM metering settings can be adjusted via **Settings > Audio > PPM Mode**. The options include:

| | PPM Mode | Description |
|---|---|---|
| **1** | Program Default (default) | Displays default program PPM meter settings (i.e. the settings described previously for mono, stereo programs etc.). |
| **2** | Input | Maps input encoders 1 to 6 with PPM meters 1 to 6. |
| **3** | Decoder | Maps decoders 1 to 6 with PPM meters 1 to 6. |
| **4** | HP Monitor | Maps PPM meters to inputs/outputs currently selected via the headphone monitoring function. The default headphone monitoring setting is accessed via **HEADPHONE** **> Monitor Source > [Select audio Source]**. |

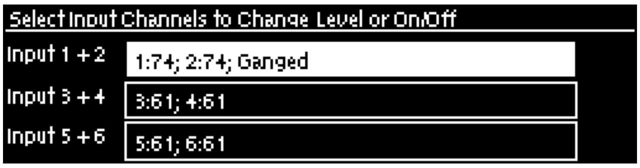



## Selecting Analog Inputs and Adjusting Input Levels

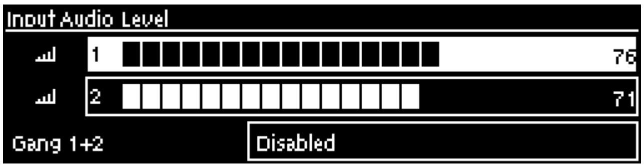
1. Press the **SETTINGS** button.
2. Navigate to **Audio** and press .
3. Inputs 1 and 2 and inputs 3 to 6 are displayed under **Input Type** and should be set to **Analog**; press to toggle between **Analog** and **AES3** and press the **RETURN** button to exit the menu.



4. Use the down navigation button to highlight **Input Level** and press the button.




5. Navigate to the channels you want to adjust and press .
6. Press the number on the keypad corresponding to the channel you want to toggle on or off. E.g. press on the numeric keypad to toggle channel 1 on and off.
7. Use the left or right navigation buttons to select the appropriate gain setting, then press the button to save the settings.




**Important Note:**

- To adjust levels quickly press and press and release the right arrow button to open the **Input Audio Level** adjustment screen.
- 15 volt phantom power can only be supplied on the auxiliary input; this is disabled by default.


| | Input Audio Features | Description |
|---|---|---|
| **1** | Channel On Symbol | Symbol indicates a channel is turned on |
| **2** | Channel Off Symbol | Symbol indicates a channel is turned off |
| **3** | Input 1 Level Control | Ch 1 level indication with percentage of gain indicated, i.e. **66** |
| **4** | Input 2 Level Control | Ch 1 level indication with percentage of gain indicated, i.e. **72** |
| **5** | Ch1/2 Gang Indication | Indicates whether ganging is enabled or disabled |

## *Quick Level Adjustment of Input 1 and 2*

1. Press and press and release 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.

## Auxiliary Input Adjustment

The codec has 1 x 6.35mm (1/4") Mic/Line level Jack on the rear panel. By default the input is **Off** and can be configured by:

1. Selecting the **SETTINGS** button.
2. Navigate to **Audio** and press the button.
3. Use the arrow-down button to select **Aux Input** and press the button to view menu options.


Input settings which can be adjusted include:

- Input on/off.
- Input level.
- Input Type: High Gain Mic, Medium Gain Mic, Low Gain Mic, Unbalanced and Line Level.
- Phantom power (15V available when enabled).
- IGC.

**Important Note:** When the auxiliary input (**AUX IN**) is **On** the default mixer configuration sends audio to all inputs. If you are not using the auxiliary input ensure it is **Off** to avoid additional noise in program audio.

## Ganging Audio Channels

It is possible to gang channels together and adjust the audio level of the ganged channels simultaneously. When channels are ganged together:

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

1. Press the **SETTINGS** button.
2. Navigate to **Audio** and press .
3. Use the down navigation button to highlight **Input Level** and press the button.
4. Navigate to the channels you want to gang and press the button.
5. Navigate to the **Gang** function and press the button to toggle between **Enabled** or **Disabled.**
6. Use the up and down arrow buttons to highlight and select the audio channels.
7. Use the left and right arrow buttons to adjust the levels for both inputs up or down simultaneously.
8. Press the **RETURN** or **HOME** buttons to exit the screen.

**Important Note:**

- To gang channels quickly press F1 and press and release the right arrow button to open the **Input Audio Level** adjustment screen and follow the preceding instructions.

## Intelligent Gain Control (IGC)

The codec's inbuilt DSP limiter automatically takes care of any instantaneous audio peaks that occur in demanding broadcast situations. **Input IGC** (Intelligent Gain Control) is enabled by default and is automatically activated at +20 dBu (G5 audio scale) and +14dBu (G3 audio scale) to prevent audio clipping.

There are three settings; **Auto**, **Fixed** and **Off**. If **Auto** is configured the codec will detect when incoming audio levels have reduced sufficiently and automatically 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 return the levels to the previous setting within half a second. This response is linear.

To adjust this setting in the codec:

1. Press the **SETTINGS** button.
2. Navigate to **Audio** and press .
3. Navigate to **Input IGC** and press .
4. Select the channel you want to adjust and press .


5. Navigate to the preferred setting and press .

## Programming Audio Metering when Connecting to Tieline G3 Codecs

New generation Genie, Merlin and Bridge-IT IP codecs have more audio headroom than Tieline G3 audio codecs, therefore metering needs to be adjusted when connecting to a Commander or i-Mix G3 codec with one of these codecs. The G3 metering scale is between -11dBu and +18dBu. Tieline codecs perform this metering adjustment automatically when they connect to each other or this can be programmed to occur by default.

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

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

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

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

# 10 Configuring AES3 Input Audio

The codec has an **IN1/AES3** input on the rear panel of the codec for AES3 (AES/EBU) format audio. This balanced 110 ohm female XLR input can operate effectively over distances of up to 100 meters and accepts both mono and stereo AES3 signals. If your codec supports multiple channels of AES audio use the DB25 **EXPANSION AUDIO I/O** connection to send/receive up to 4 additional channels of AES3 audio. Please note that expansion inputs/outputs 3, 4, 5 and 6 must be either all analog or all digital AES. See Appendix B for the DB25 wiring pin-outs.

1. Press the **SETTINGS** button.
2. Navigate to **Audio** and press .
3. Select **Input Type** and press the button.
4. Navigate to the inputs you want to configure and press the button to toggle between **Analog** and **AES3**.

The 3 pin male XLR **AES3 OUT** connector is capable of sending both mono and stereo AES3 signals.

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

## AES3 Sample Rate Conversion

The codec contains two sample rate converters.

### Input Sample Rate Converter

The codec implements an Asynchronous Sample Rate Converter (ASRC) to convert the sample rate of an 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.

By default the codec will up-sample all channel 1 and 2 AES3 input sources to 96kHz sampling unless your audio source uses a 44.1kHz sample rate.

**Important Notes:** If your codec supports up to six AES3 inputs, the three AES3 stereo inputs are multiplexed onto analog inputs 1, 3 and 4. There are also three AES3 stereo outputs via the **AES3 Out** XLR and two are multiplexed onto outputs 3 and 4 (see Appendix B for pin-outs of inputs 3 and 4 and outputs 3 and 4).

The inputs have asynchronous sample rate converters to facilitate sources with different sample rates being attached to the codec. For example, **IN1-AES** (Inputs 1 and 2) can be 48kHz AES-IN2 (IN3&4) can be 32kHz and AES-IN3 (IN5&6) can be 96kHz

### Output Sample Rate Converter

The sample rate of the AES3 output is currently configured using the clock source setting via the **SETTINGS** button and then **Audio > Input Type > AES3 Out**. This configures the sample rate frequency of all AES3 output signals and there are three possible settings.


**Locked to AES3 Input**
If this setting is used, the codec will use the sync information received by the AES3 XLR input (this is the same as the **AES Rx Clock** setting in Tieline G3 codecs) to set the sample rate within the codec. This codec input also carries AES3 audio data.

**Wordclock Sync In**
This setting configures the codec for a word clock source via the **SYNC INPUT** on the codec rear panel (this is the same as the **External Word Clock** setting in Tieline G3 codecs). Often this will be a studio reference signal (D.A.R.S., or Digital Audio Reference Signal). In television broadcasting facilities, the audio reference signal should be locked to the video reference if there is one available. The sample rate being received is recognized by the codec and automatically adjusted within it. Sample rates from 32 kHz to 96 kHz are accepted, including the most popular rates of 32 kHz, 44.1 kHz and 48 kHz.

**Fixed Sample Clock**
Select from a range of fixed output sample rates.

# 11 Merlin PLUS Headphone/Aux Output

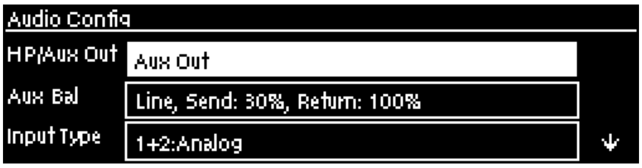
The codec has a 6.35mm (1/4") RTS stereo **HEADPHONE** output for monitoring inputs and return audio. If you are using analog inputs or digital inputs you will see audio metering on the PPMs and can monitor it with the headphones.

**Important Note:** The front panel **HEADPHONE** output and rear panel **HP/AUX** output share the same hardware output. This means both are switched and configured together. I.e. both outputs are either a stereo headphone output (default setting), or a balanced mono auxiliary output.

## *Configure for Headphone and Aux Output*

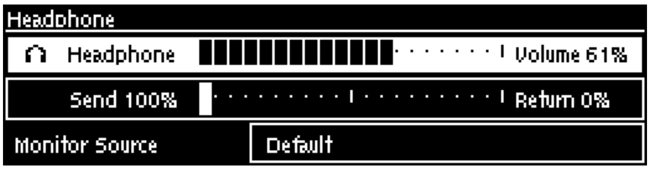
Both the front panel **HEADPHONE** and rear panel **HP/AUX** outputs are configured as stereo headphone outputs by default. To adjust this setting:

1. Press the **SETTINGS** button.
2. Navigate to **Audio** and press .
3. Select **HP/Aux Out** and press to toggle between **Headphone** and **Aux Out**.




## *Adjust Headphone Output Settings*

1. Press the **HEADPHONE** button to display the headphone monitoring adjustment screen.
2. Use the left or right navigation buttons to adjust the volume level up or down. The screen displays level adjustments in real-time.

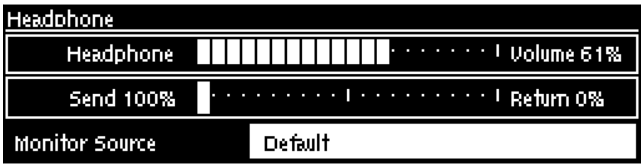



3. Press the down navigation button to select the **Send/Return** audio balance and use the left or right navigation buttons to adjust the balance. The **Send/Return** audio balance dictates whether the front panel **HEADPHONE** output and the rear panel **HP/AUX** output monitors send (input/encoder) audio only, return audio only (decoder audio from a connected device), or a mix of both send and return audio.
4. Press **RETURN** when you have finished.

Note: Headphone levels can also be adjusted by pressing the **SETTINGS** button, navigate to **Audio** and then **HP Vol/Bal** and press .

### *Adjusting the Monitor Source*

In headphone listen mode it is possible to select monitoring sources via **HEADPHONE** **> Monitor Source > [Select audio Source]**.




Navigate to the source you want to monitor and press . Options include:

1. **Default**: the default factory program headphone mix
2. **Audio Stream**: monitors the selected codec audio stream.
3. **Inputs**: monitors the codec inputs (i.e. encoders).

The default headphone mixes for factory programs are displayed in the following table.

| Codec Programs | Left | Right |
|---|---|---|
| 1 x Peer-to-Peer Mono | Inputs 1&2/ Outputs 1&2 | Inputs 1&2/ Outputs 1&2 |
| 1 x Peer-to-Peer Stereo | Input1 /Output 1 | Input 2/Output 2 |
| 2 x Mono Peer-to-Peer | Input 1/Output 1 | Input 2/Output 2 |
| 6 x Mono Peer-to-Peer | Input1 /Output 1 | Input 2/Output 2 |
| 1 x Mono Peer-to-Peer + IFB | Inputs 1&2/ Outputs 1&2 | Aux In/Aux Out/ Input3/ Output3 |
| 1 x Stereo Peer-to-Peer + IFB | Inputs 1&2/ Outputs 1&2 | Aux In/Aux Out/ Input3/ Output3 |
| 2 x Mono Peer-to-Peer + IFB | Inputs 1&2/ Outputs 1&2 | Input 5/Output 5 |
| 2 x Stereo Peer-to-Peer + IFB | Inputs 1&2/ Outputs 1&2 | Input 5/Output 5 |

## Adjust Auxiliary Output Settings

Settings for the auxiliary output audio are adjusted similarly to the **HEADPHONE** output, except that the output level is fixed at line level. Configure the front panel **HEADPHONE** output and rear panel **HP/AUX** output as an **Aux Out** and then:

1. Press the **HEADPHONE** button to display the aux output adjustment screen.
2. Use the left or right navigation buttons to adjust the **Send/Return** audio balance.


3. Press when you have finished.

Note: Send/Return balance can also be adjusted by pressing the **SETTINGS** button, navigate to **Audio** and then **Aux Bal** and press .

# 12 About ISDN Modules

ISDN stands for Integrated Services Digital Network. The Basic Rate Interface (BRI) of ISDN consists of 2 bearer (B) channels at 64 kbps each and 1 data (D) channel at 16 kbps, i.e. (2B +D). This can be provided over a 2 wire facility and the two B channels can be bonded together to form a single 128kbps channel. The B channel can carry user information such as voice, video or data. The D channel carries signaling information between a user and the network.

Tieline codecs can provide high quality mono or stereo audio over a single B channel using the Tieline Music algorithm. If you have 2 B channels you can use one as a standby, or configure higher bandwidth mono or stereo connections using algorithms such as MusicPLUS and MPEG.

## *Important Considerations*

There are a number of things to consider if you are using your codec in ISDN mode. Some of these things include:

- Will you be operating within North America or other countries?
- Will you be using a single B channel, 2 B channels, or 4 B channels?
- Which network will you be using?
- Is your ISDN line Point-to-Point or Point-to-Multipoint?
- What are your directory numbers (DN)?
- If you are in the US, what are your Service Profile ID (SPID) numbers?
- What is your Multiple Subscriber Number (MSN) if you need to enter this outside North America?

The answers to these questions will be influenced by the country in which you operate. For example, a SPID does not need to be entered into a Tieline codec for operation within Europe, but it does in North America.

## *U and S/T ISDN Interfaces*

In North America the telephone company provides its BRI customers with a U interface. The U interface is a two-wire (single pair) interface from the phone switch. It supports full-duplex data transfer over a single pair of wires, therefore only a single device can be connected to a U interface.

The situation is different in Europe, the UK, most of Asia, Australia, Africa and parts of the Middle East, where the phone company is allowed to supply the NT-1 and the customer is given an S/T interface. The NT-1 is a relatively simple device that converts a 2-wire U interface into the 4-wire S/T interface.

If you have an NT-1 device connected to the U interface line then you will require a Tieline Euro ISDN G5 module (S/T interface - model: TLISDNEUROG5). If you don't have an NT-1 device installed then the Tieline US ISDN G5 module (U interface - model: TLISDNUSG5) will be required. You can ring your telecommunications provider to ask if you're not sure. Note: In Japan use the Tieline Euro ISDN module.

**Important Note:** Tieline S/T Euro ISDN G5 modules do not have internal terminating resistors. When you connect terminating equipment such as a Tieline codec to an NT-1, 100 ohm termination resistors must be connected between pins 3 and 6 and between pins 4 and 5 at the last socket on the ISDN line. Check your NT-1 device user manual as this may be supported. Suppliers of electronic components sell suitable plugs with termination resistors when required. Please note: U interface ISDN terminations do not require terminating resistors.

## 12.1 ISDN Module Settings

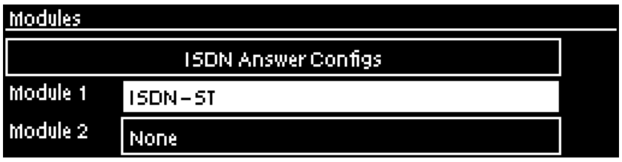
The codec has two module slots available. Each module supports 2 B channels and it is possible to insert two ISDN modules and bond 4 B channels together. This will increase connection bandwidth to 256 kbps for connections using high quality algorithms like aptX Enhanced.

### *Configuring the ISDN Module*

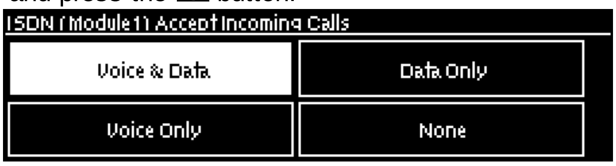
1. Press the the **SETTINGS** button, then navigate to **Modules** and press the button.


**Important Note:** You can also configure your ISDN module by pressing the **HOME** button to return to the **Home** screen and select **Connect > ISDN**. Then use the down navigation button to select **Module Configuration** and press the button.

2. Navigate to the module you want to configure and press the button. Note: Module 1 is on the left when viewing the codec rear panel.

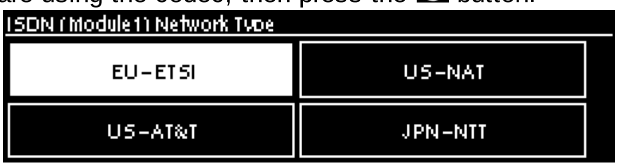



3. Navigate to **Accept** and press the button. This menu is a call filter to allow or deny voice or data calls according to your preferences. The default setting allows both **Voice & Data**. Select your preferred option and press the button.



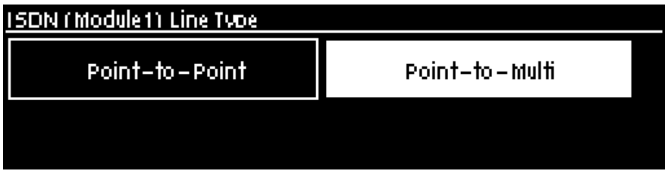

**Important Note:** G.711 is the default algorithm for incoming connections when **Voice Only** is selected. There are two G.711 algorithms and the one used by the codec depends on the country setting in the codec. The µ-law algorithm is used in the USA, Japan and Canada, whereas the A-law algorithm is used in other countries.

4. Navigate to **Network** and press the button. Select the **Network Type** corresponding to the region in which you are using the codec, then press the button.

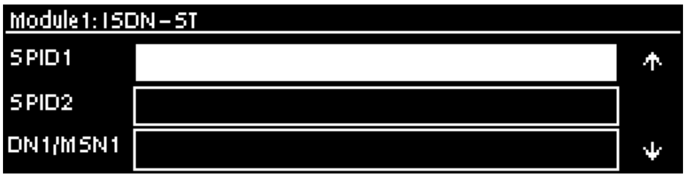


| Networks | Select |
|---|---|
| US-Nat | If switch type is National ISDN-1 and 2 |
| US-AT&T | If switch Type is AT&T 5ESS |
| EU-ETSI | If Switch Type is ETSI<br>(UK, Europe, Australia and most other countries) |
| JPN-NTT | If you are in the Japan and your network is NTT |

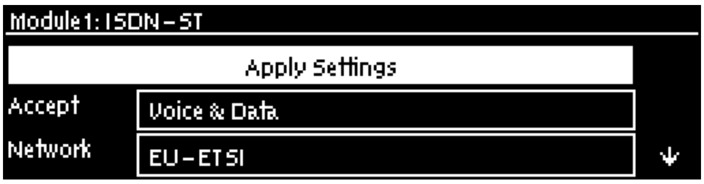
5. Navigate to **Line Type** and press the OK button. Ask your Telco whether your ISDN line is Point-to-Point or Point-to-Multipoint. By default select **Point-to-Multipoint**, unless your switch type is point-to-point, your Telco says the line is point-to-point, or you are connected to a PABX system. Most PABX systems are point-to-point. Next, press the OK button.




6. If you are in the US enter DN and SPID numbers as required, or in other regions enter DN or MSN numbers as required. Navigate to each **DN**, **SPID** or **MSN** and press the OK button before entering each number, then press the OK button to store each number.




8. Navigate up to **Apply Settings** and press the OK button to apply all module settings.




**Important Notes:**

***Directory Numbers and Multiple Subscriber Numbers***
Directory Numbers (DN) in North America and Multiple Subscriber Numbers (MSN) in the rest of the world are simply phone numbers associated with an ISDN B channel, like lines listed in a typical phone directory. Your Telco will normally supply 2 DN/MSN numbers for each pair of B channels. However, these numbers may or may not be associated with a specific B channel.

Often broadcasters prefer to predict which B channel will answer an incoming call to ensure audio routing is consistent. However, if a DN or MSN number is not entered in the codec and multiple B channels are available, the codec may use any channel to answer an incoming call. To ensure calls are routed consistently, enter a DN/MSN number (without the country or area code) as the DN/MSN for a B channel, then only that corresponding B channel will answer an incoming call to that number. Programming DN/MSN numbers for each B channel allows the codec to ignore calls without matching DN/

MSN numbers. This is the best way to answer calls from codecs in a predictable manner.

***SPID Numbers in North America***
ISDN relies on an initialization procedure for associating Service Profiles with specific terminating equipment (e.g. your audio codec) rather than lines. In the US Telcos assign a Service Profile ID (SPID) number which assists in identifying different ISDN services across the network. Your Telco must provide a SPID for each B channel you order when connecting over US-Nat or US-AT&T networks in the US. A SPID is not required when using the AT&T PTP protocol.

Typically, each ISDN BRI service in the US will have two SPIDs and these must be entered correctly. When you enter a SPID into your codec and connect it to an ISDN line, an initialization and identification process takes place, whereby the terminating equipment (your codec) sends the SPID to the switch. The switch then associates the SPID with a specific Service Profile and directory number.

Note: SPID numbers normally include the phone number and additional prefix or suffix digits up to 20 digits long.

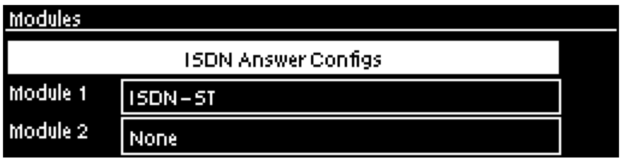
## 12.2 ISDN Answering Configuration

**Important Note:** For more information about ISDN Answering parameters, including bonding and 'route' configuration etc., please see the web-GUI section of this manual titled Configuring ISDN Answering.

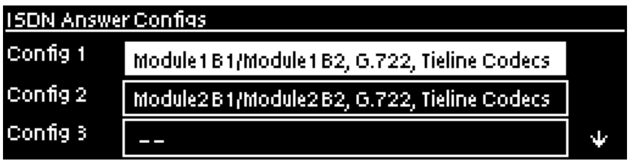
1. Press the the **SETTINGS** button, then navigate to **Modules** and press the button.


**Important Note:** You can also configure your ISDN module by pressing the **HOME** button to return to the **Home** screen and select **Connect > ISDN**. Then use the down navigation button to select **Module Configuration** and press the button.

2. Navigate to **ISDN Answer Configs** and press the button.

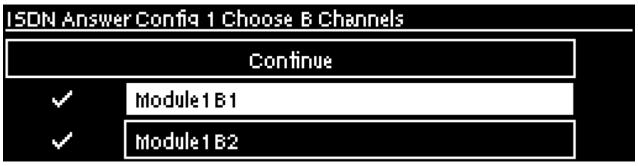



4. Navigate to one of the four available **Configs** and press the button.



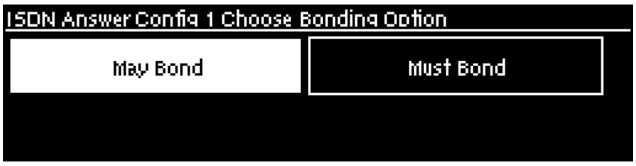

5. Navigate to **Edit** and press the button.


6. Navigate to each B channel and press the button if you want to select/deselect a B channel within the selected **Config**. Navigate to **Continue** and press the button. Notes: The tick symbol confirms a B channel has been selected.

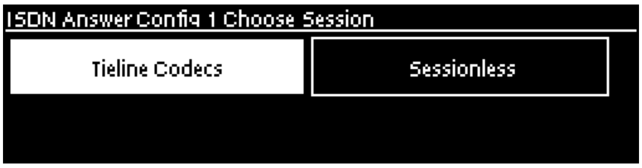



**Important Note:** If a B channel has been selected within another **Config** it will not be visible. Only available B channels are displayed.

7. Choose the bonding method if multiple B channels have been selected, then press the button.

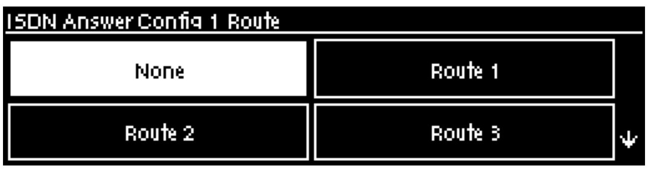



8. Choose to enable Tieline session data or select no session data enabled, then press the button.

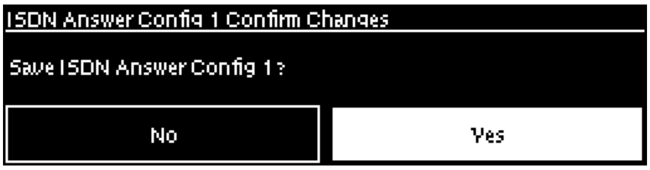



9. Select the default algorithm when receiving a call from a non-Tieline codec, then press the button.


10. Specify the audio stream **Route** when receiving a call on the answering codec from a non-Tieline codec, then press the button.




11. Select **Yes** and then press the button to confirm all changes.




# 13 Language Selection

English is the default language in the codec. To select a new language:

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 .

# 14 About Program Dialing

## What Defines a Program?

Tieline Genie and Merlin codecs use programs to connect to another codec. A **Program** configures a Tieline codec to send or receive one or more **Audio Streams** based upon the particular application the codec is being used for at any given time. The attributes of the audio stream and associated connections are embodied within a program when it is created, including the configuration, dialing and answering parameters.

Tieline Genie and Merlin 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 acknowledgment to the dialing codec and streaming can commence.

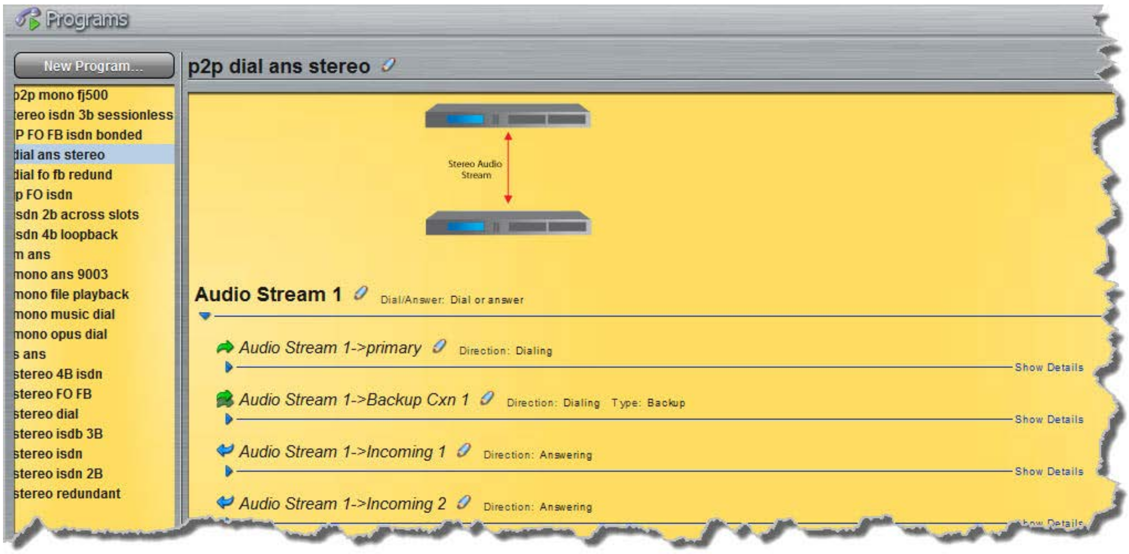
For example, if you configure a standard stereo program on the dialing codec using a particular algorithm and bit rate settings etc., these settings will be configured on the dialing codec when the codec connects. It is also possible to lock a loaded program in a codec to ensure the currently loaded program type cannot be unloaded by a codec dialing in with a different program type. For example, if your routing requirements 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.

## Defining Audio Streams within Programs

Each audio stream within a program can be defined separately and contain a variety of settings relating to the number of connections (e.g. primary and backup) and the number of destinations to which each audio stream is distributed. Each audio stream is capable of being configured to include dial and answer connections, dial connections only, or answer connections only. Each audio stream has its own:

- Name.
- Connection, Transport, and Destination settings.
- Backup configuration options.

The following image displays a simple peer-to-peer program in the **Programs panel** within the Toolbox web-GUI, which can be used to configure and edit all program parameters. The program displayed is configured to send a single stereo audio stream and will allow the codec to both answer and dial (via dialing and answering connections) if required. A backup dialing connection is configured in case the primary connection fails.




## *Creating Programs*

Only the simplest peer-to-peer (point-to-point) programs can be created using the codec front panel. The Toolbox web-GUI contains a feature-rich program creation wizard, which can tailor settings and create backup connections. Use the Toolbox web-GUI to retrieve or edit settings easily at the touch of a button. Once programs have been created they can also be used as templates for creating other programs using the web-GUI.

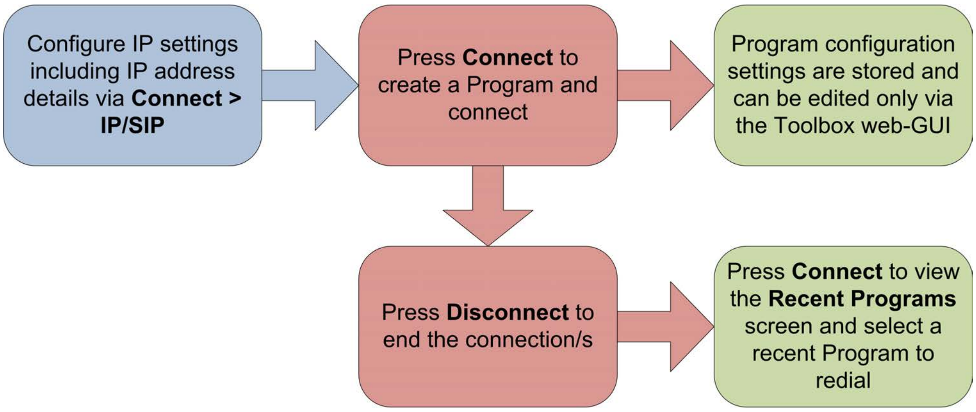
### *Mono and Stereo Peer-to-Peer Programs*

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

Front panel configured programs are automatically saved as **Recent Programs** - retaining all the audio stream dialing and configuration information programmed into the codec. 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.

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

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




*Peer-to-peer connection configured via the codec front panel*

# 15 Multiple Stream Programs

Programs can be configured to connect a single audio stream to multiple destinations, or connect multiple audio streams to different destinations.

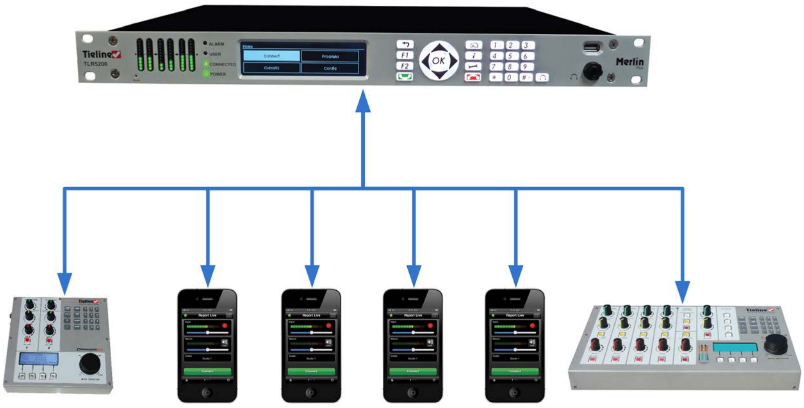
### *2 x Mono Peer-to-Peer Programs*

A **2 x Mono Peer-to-Peer** program includes two mono audio streams. Each audio stream includes a separate peer-to-peer connection to a different destination, which can also be configured with different transport, audio and backup settings.

# 16 6 x Mono Peer-to-Peer Programs

A **6 x Mono Peer-to-Peer** program includes six bidirectional mono audio streams. Each audio stream is a separate peer-to-peer connection with a different remote codec or smartphone using the Report-IT codec application. Each connection can also be configured with unique settings.

Merlin PLUS codec at the studio



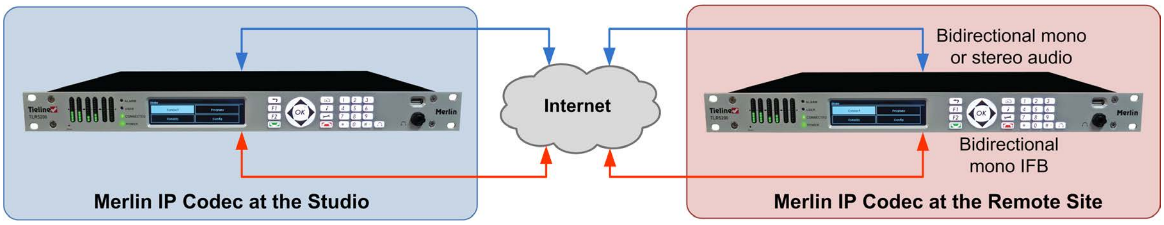
Up to 6 bidirectional mono remotes simultaneously connected to Merlin PLUS

# 17 Mono or Stereo + IFB Programs

This program is designed to allow Merlin and Merlin PLUS codecs to answer a call from an incoming codec and receive:

1. A bidirectional mono or stereo audio stream connection.
2. A separate bidirectional mono IFB audio stream for communications.

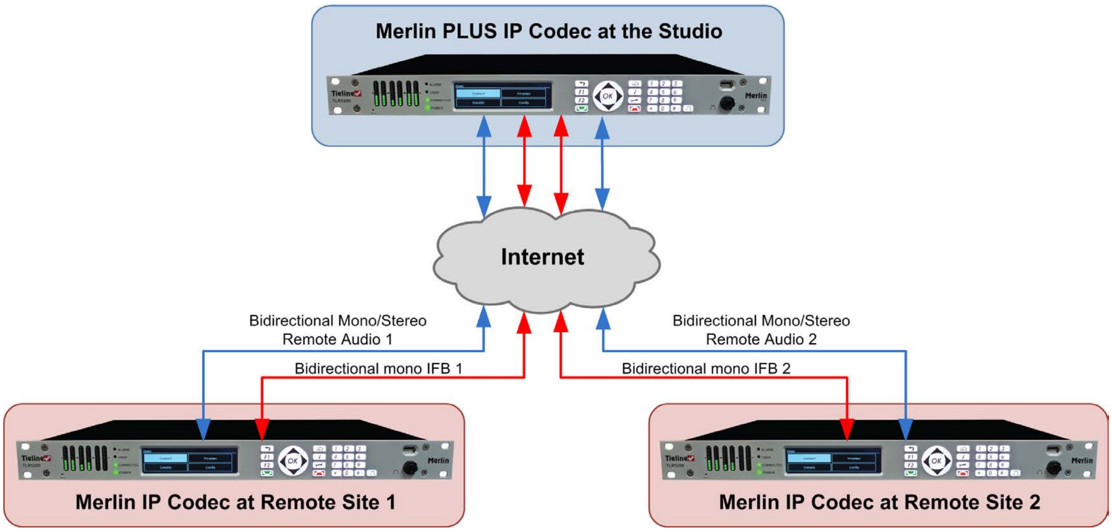
**1 x Mono or Stereo Peer-to-Peer + IFB via Merlin (or Merlin PLUS)**




# 18 2 x Mono or Stereo + IFB Programs

This program is designed to allow a Merlin PLUS codec at the studio to answer calls from two incoming Merlin or Merlin PLUS codecs and receive:

1. 2 bidirectional mono or stereo audio streams connections.
2. 2 separate bidirectional mono IFB audio streams for communications.



# 19 Getting Connected Quickly

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

1. Attach power to the codec.
2. For IP connections, attach RJ45 Ethernet cables to at least one of the **ETH** ports on the codec's rear panel. Attach cables to ISDN modules inserted in your codec as required.
3. Attach headphones to the 6.35mm (1/4") headphone jack on the codec's front panel.
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 and press the button.

5. Make sure you know the IP address, or ISDN dialing numbers for the destination codec.

## 19.1 Steps to Connect over IP

**Important Note:** The following procedure will create a custom point-to-point 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 Web-GUI Introduction for details on configuring connections remotely via a computer.

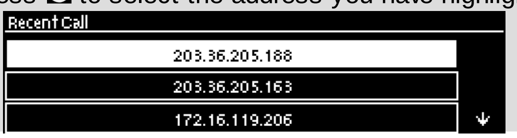
1. Press F1 and press and release the right arrow button to open the **Input Audio Level** adjustment screen.
   - Press the number on the keypad corresponding to the channel you want to toggle on or off. E.g. press 1 on the numeric keypad to toggle channel 1 on and off.

- Use the up and down navigation buttons to select the gang function 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.

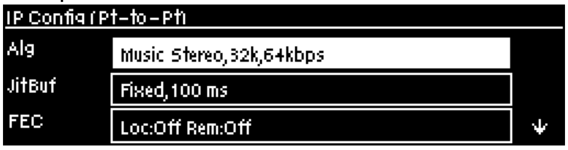
2. Press the **HOME** button to return to the **Home** screen, select **Connect > IP > Point to Point** and press the button.

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


**Important Note:** The codec remembers recent IP addresses just like a cell-phone. To view these addresses just press 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 to select the address you have highlighted.

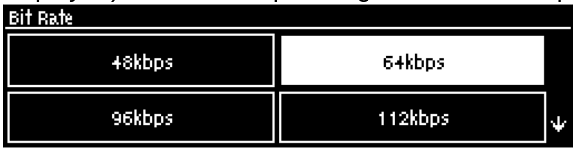


4. Navigate to **Algorithm** and press .

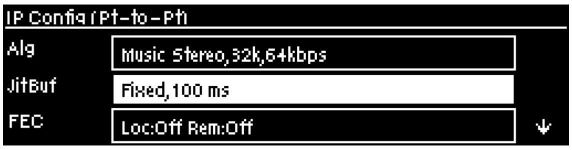



5. Use the navigation buttons to select an algorithm profile or manually enter algorithm settings, then press .

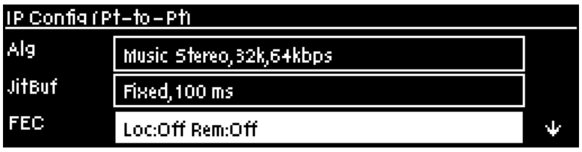

If you decide to manually program the algorithm, use the navigation buttons to select your preferred algorithm sample rate (if displayed) and bit rate, pressing  after each option is selected.




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




7. Press the down navigation button to select **FEC** (forward error correction) and press  to view selection options. Use the navigation buttons to choose the FEC percentage you want to use and press .




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

> **Important Note:** At this point you can navigate to **Save** on the **Connect IP** screen and press  to use the numeric **KEYPAD** to name the program and press  to save the program.

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

10. Alternatively, to load a saved program and dial press the **HOME** button, navigate to **Programs**, select the program you want to dial and press the **CONNECT** button to load the program and dial.

11.When dialing, the **CONNECTED LED** on the front of the unit will flash green. When connected, the **CONNECTED LED** on the front of the unit will illuminate solid green. Use the down navigation button to select **Cxns** and view connection **Status** and press OK to view connection statistics for IP packets being sent over the connection. To negotiate higher bit rates press F2 then **3** on the numeric **KEYPAD**; for lower bit rates press F2 then **9**.

## 19.2 Steps to Connect over ISDN

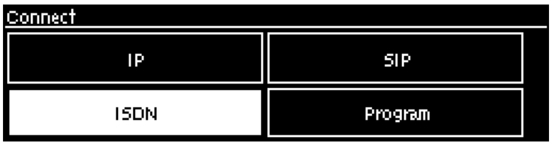
**Important Note:** The following procedure shows how to create a custom peer-to-peer connection program using the front panel keypad and navigation buttons. It instructs how to connect to another Tieline codec using ISDN for the very first time without using the Toolbox web-GUI and your computer for configuration.

- See ISDN Module Configuration for details on module settings.
- See ISDN Answering Configuration for details on ISDN answering settings.
- See Configuring ISDN for details on configuring connections via a computer.

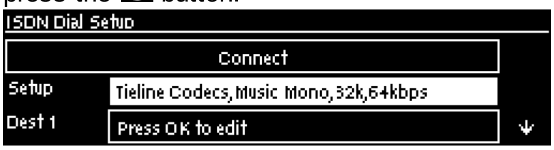
1. Press F1 and press and release the right arrow button to open the **Input Audio Level** adjustment screen.
   - Press the number on the keypad corresponding to the channel you want to toggle on or off. E.g. press 1 on the numeric keypad to toggle channel 1 on and off.

- Use the up and down navigation buttons to select the gang function and press the OK 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.

2. Press the **HOME** button to return to the **Home** screen, select **Connect > ISDN** and press the OK button.



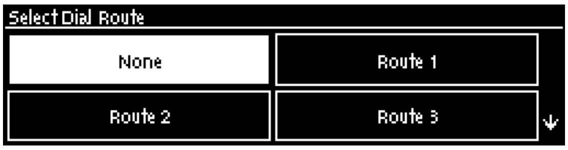

3. Navigate to **Setup** and press the OK button.



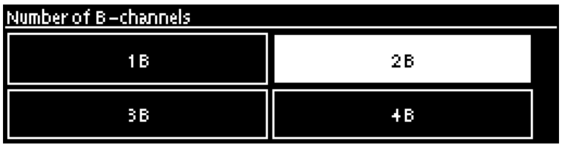

4. Select whether to dial with Tieline Session Data or select **Sessionless** if dialing a non-Tieline codec, then press the OK button.


**Important Note:** By default, when Tieline codecs dial they send call configuration settings to the remote codec using Tieline Session Data. This configures the codec receiving the call with matching algorithm, sample rate and bit rate settings. This does not occur when dialing to non-Tieline devices, therefore **Sessionless** must be selected to provide compatibility.

5. Select the **Dial Route** to use for this audio stream if one is required, then press the OK button. Note: See Configuring ISDN Answering for more information on **Dial Route** and **Answer Route** tags. These are useful when routing multiple audio streams over transports like ISDN.

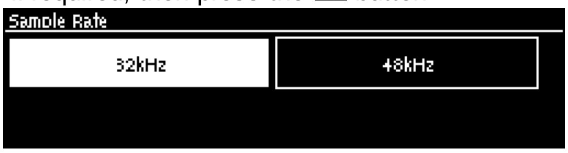



6. Select the number of B channels being used for the audio stream connection, then press the OK button.




7. Select an algorithm, then press the OK button.

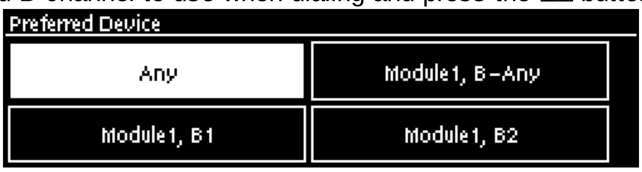

8. Select the sample rate if required, then press the OK button.



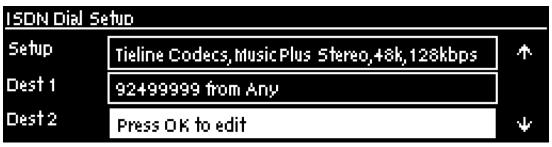

9. Select **Destination 1** and press the OK button, then use the numeric **KEYPAD** to enter the ISDN number you want to dial and use the **RETURN** button to delete any numbers already entered. Then press the OK button.



10. Select the preferred B channel to use when dialing and press the OK button.

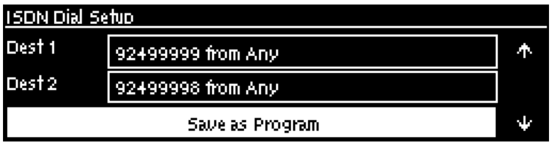



11. If you are dialing over multiple B channels to create a bonded connection select the next destination, e.g. **Destination 2**, and use the numeric **KEYPAD** to enter the next ISDN number you want to dial. Do this for all B channel destinations.

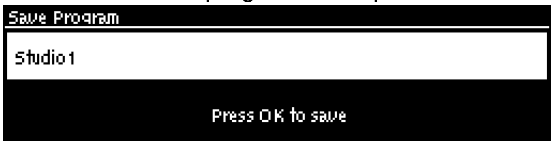



12. At this point we recommend you save a program to simplify dialing and to store this

configuration for future use. Use the up navigation button to select **Save as Program** and press the button.




13. Use the numeric **KEYPAD** to name the program, then press to save the program.




14. It is possible to dial the B channels associated with this audio stream from this menu. Use the up navigation button to select **Connect** and press to connect.

15. Alternatively, to load a saved program and dial press the **HOME** button, navigate to **Programs**, select the program you want to dial and press the **CONNECT** button to load the program and dial.

16. When dialing, the **CONNECTED LED** on the front of the unit will flash green. When connected, the **CONNECTED LED** on the front of the unit will illuminate solid green. Use the down navigation button to select **Cxns** and view connection **Status** which displays the program, numbers dialed, algorithm and connection bit rate.


## 19.3 Creating a Multicast Client 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.

**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.
- The default UDP audio port is 9000 for a multicast client program configured via the codec front panel.
- You can lock a loaded custom program in a codec to ensure the currently loaded program cannot be unloaded by a codec dialing in with a different program type.
- Always dial the multicast server codec connection first before connecting multicast client codecs.
- Multicast client codecs will display return link quality (LQ) only. The **Return** reading represents the audio being downloaded from the network locally.
- Forward Error Correction (FEC) is not available for multicast connections.
- It is not possible to send auxiliary data using multicast connections.

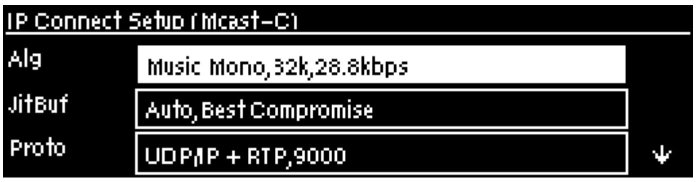
- It is not possible to connect to a G3 codec and receive multicast IP audio streams.
- To copy multicast client programs onto multiple codecs see Save and Restore Configuration Files.
- To learn more about programs see the section titled About Program Dialing.
- See Toolbox web-GUI documentation for more detailed information about Configuring Multicast Client Programs

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


2. Select **Multicast Client** to configure a client codec program.

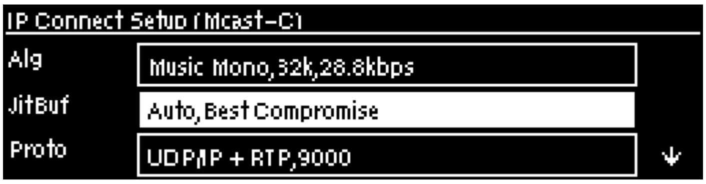

3. Use the **RETURN** button to delete any numbers already entered, then use the numeric **KEYPAD** to enter the multicast IP address you want to dial, using the or buttons to enter the periods in the IP address. The same multicast address and audio port must be used for both the server and client programs.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


4. Press the down navigation button to select **Algorithm** and press .



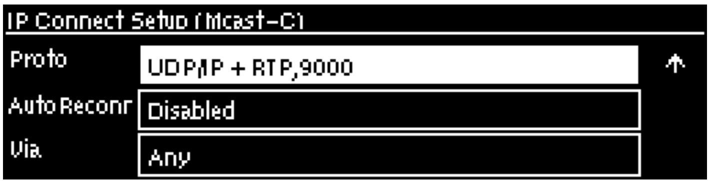

5. Use the navigation buttons to select an algorithm profile or manually choose algorithm settings, then press .

6. Click to configure the Jitter Buffer from either **Auto Jitter Adapt** or **Fixed Buffer Level** , then and enter the **Jitter Depth**, which must be between 12ms and 5000ms depending on the algorithm you select, then press .

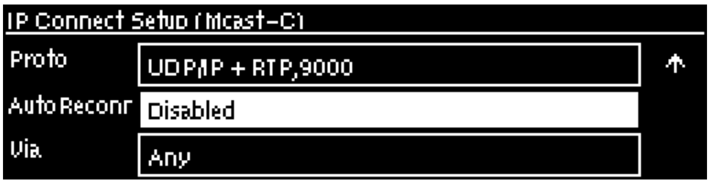



**Important Notes:** Automatic or fixed jitter buffer settings can be adjusted on individual client codecs as required. There is no jitter buffer setting on the server codec because it never receives audio packets.

7. Select **Protocol** to adjust the audio protocol and audio port. Select **UDP/IP +RTP** for RFC compliant IP streaming. Press  to save settings.




8. If required, enable **Auto Reconnect** and use **Via** to specify which IP streaming interface is used to dial this connection, e.g. **Primary** (port **ETH1**) or **Secondary** (port **ETH2**). Note: By default **Any** will select **ETH1** if it is available and **ETH2** if it is unavailable.




9. 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.


10. 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.

11. 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.

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

## 19.4 Load and Dial Custom Programs

Programs 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 use, then press the **CONNECT** button to load the program and make a connection.
4. The **Wait Connecting** screen appears during the connection process and then connection details are displayed.

## 19.5 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 disconnection.


## 19.6 Dialing SIP Peer-to-Peer

**Important Note:** 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 IP Connections*

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

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

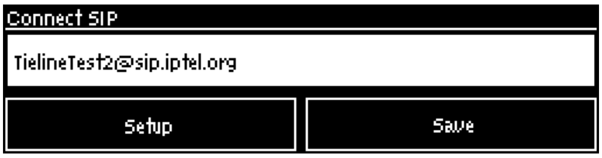
- Find out the IP address of the remote codec being dialed.
- Program each codec with a compatible algorithm and sample rate etc.
- Dial using **SIP** within the **Connect** menu.
- If the remote codec has a private IP address then it should be configured for port forwarding and should dial the public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

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

## 19.7 Dialing SIP Addresses

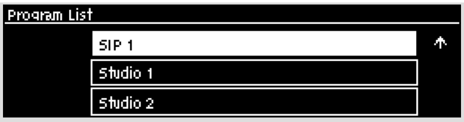
### *Dialing a SIP Address via the Codec Front Panel*

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




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

**Important Notes:**
- See Configuring SIP Settings for instructions on entering SIP account details into the codec. If your codec is registered with same SIP registrar as the destination codec then you only need to enter the SIP user name to dial successfully.
- 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. See the section titled Configuring SIP Programs for more information.

## 19.8 Dial/Disconnect Multiple Audio Stream Programs

### *Multiple Audio Streams within Programs*

Some programs are created to allow simultaneous audio stream connections with different destination codecs, e.g. 2 x Mono peer-to-peer programs. These programs can only be created using the Toolbox web-GUI.

There are two ways to simultaneously dial multiple audio stream connections within these types of programs:

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

### *Dialing Multiple Audio Stream Programs with 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 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, see Redialing a Connection for more information.

#### *Connection Details*

The number of active audio streams and connections are displayed on the **Home** screen via **Cxns**. In the following image two connections (left bracketed number) and two audio streams (right bracketed number) are currently in use.


#### *Disconnect All Audio Stream 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.

***Disconnect a Single Audio Stream (not available for multi-unicast connections)***

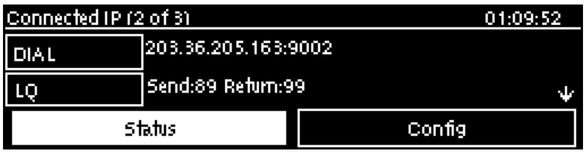
1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Cxns** and press the button.
3. Use the up and down navigation buttons to select the connection you want to disconnect.


3. Press the red **DISCONNECT** button on the numeric **KEYPAD**.
4. Use the right navigation button to select **Yes** and press the **DISCONNECT** button or the button to confirm the disconnection.

## 19.9 Monitoring IP Connections

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 audio streams which are connected. Press the button again to view connection details. The IP address which has been dialed and the **LQ** (link quality) is displayed on the screen and you can use the down navigation button to view the algorithm being used, the connection bit rate, total bytes used and the amount of jitter buffer delay over IP network connections.




### *Link Quality (LQ) Readings*

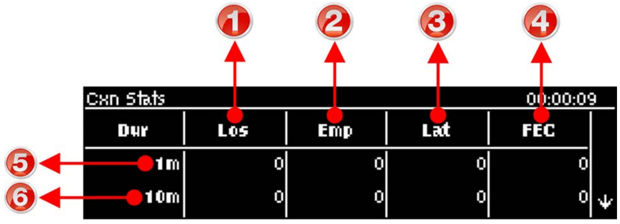
Send and return LQ numbers can also help you to determine if a problem is occurring at either end of a connection. For example, on an IP connection the **Return** reading represents the audio being downloaded from the network locally (i.e. audio data is being sent by the remote codec). Conversely, the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

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

Following is a packet arrival analysis table with solutions for any noticeable packet loss statistics displayed on the screen.

| Packet Analysis | Displays | Possible Causes | Possible Solutions |
|---|---|---|---|
| Loss | Packets sent and that failed to arrive. | • LAN/WAN congestion<br>• Unreliable ISPs<br>• Unreliable networks<br>• Inferior IP hardware | • Renegotiate connection bit rate downwards<br>• If link quality good add or increase FEC as required<br>• Assess ISPs 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 dropouts using 3G cell networks<br>• Renegotiation causes the jitter buffer reservoir to empty | • Once could be an anomaly – assess lost & late packets<br>• If many lost packets and 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. |

## 19.10 Monitoring ISDN Connections

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 ISDN** screen displays all audio streams which are connected. Press the button again to view connection details.


## 19.11 Redialing a Connection

Press the **CONNECT** button from any codec menu to redial previous connections (except menus accessed via the **Connect** > **IP** or **Connect > ISDN** 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 a dialing address or number (manually dialed connections).

### *Redialing Manually from the Connect IP Screen*

From the **Home** screen select **Connect > IP > Select an IP mode** and the codec assumes you want to dial a new manual connection. Press the **CONNECT** button when the **Connect IP** screen is displayed to retrieve previously dialed IP addresses. Codec settings used for a connection dialed from this screen include the current settings in the **Setup** menu, which can be accessed via this screen.

## 19.12 Programming Auto Reconnect

**Auto Reconnect** is disabled by default. When enabled the dialing codec attempts to reconnect if audio 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 mode** you are using to connect.
3. Select **Setup** and press .
4. Navigate to **Auto Reconnect** and press to toggle between **Enabled** and **Disabled**.

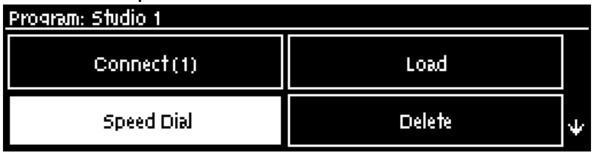
**Important Note:** When **Auto Reconnect** is enabled, the dialing codec will continue to attempt a connection with the remote codec until **Disconnect** is pressed either on the dialing codec's keypad, or in the web-GUI.

## 19.13 Speed Dialing Connections

### *Assigning Speed Dial Numbers*

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

2. Use the navigation buttons to select **Programs** and press the button.
3. Navigate to the program you want to assign a speed number to and press the button.
4. Navigate to **Speed Dial** and press the button.



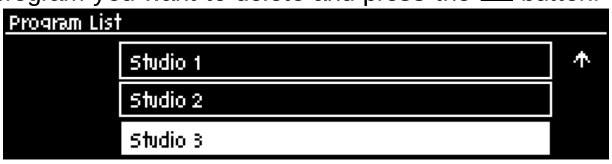

5. Navigate to the speed dial number you want to assign to the selected program and press the button.
6. A confirmation message will display the number assigned.


### *Speed Dialing*

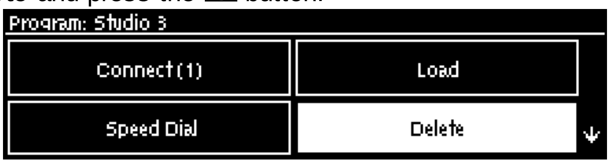
1. Press the **HOME** button to return to the **Home** screen.
2. Use the numeric **KEYPAD** to enter the speed dial number.
3. When the **Speed Dial** screen appears, press the button or the **CONNECT** button to connect.

## 19.14 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.



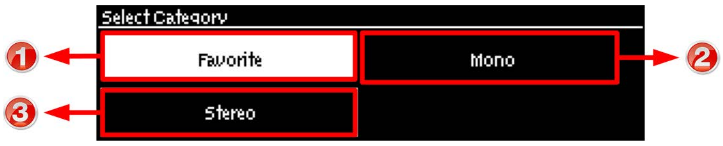

5. Confirm the deletion and press the button.

## 19.15 Selecting Algorithm Profiles

A number of pre-programmed mono and stereo dialing profiles are available for programming the codec quickly without individually selecting algorithms and bit rates etc. These profiles have been programmed with the most popular settings that provide high quality connections using each available algorithm.

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 Mode** and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Select **Alg** and press the button.
7. Use the right navigation button to select **Profile**.
8. Choose the profile you want from the **Favorite**, **All**, **Mono or Stereo** menus.

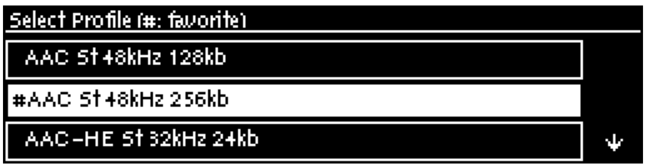



| | Features | Codec Home Screen Elements |
|---|---|---|
| 1 | Favorite | Displays a list of favorite profiles that have been selected manually within the codec by users |
| 2 | Mono | Displays preprogrammed mono profiles within the codec |
| 3 | Stereo | Displays preprogrammed stereo profiles within 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 Mode** and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Press the button to select **Alg**.
7. Use the right navigation button to select **Profile**.
8. Select the profile you want from the **Mono or Stereo** menus.
9. Press the hatch (pound) 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. Use the down navigation button to select **Setup** and press the button.
5. Press the button to select **Alg**.
6. Use the right navigation button to select **Profile**.
7. Select the profile you want to delete from the **Favorite** menus.
8. Press the hatch (pound) button to delete the selected profile from the favorite menu.

## 19.16 Merlin Algorithm Profiles

The following algorithm profiles are programmed into Merlin codecs.

| | 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 | 24 |
| 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-ELD | Stereo | 32 | 48 |
| 11 | aptX Enhanced | Mono | 32 (16 bit) | 128 |
| 12 | aptX Enhanced | Mono | 48 (24 bit) | 288 |
| 13 | aptX Enhanced | Stereo | 32 (16 bit) | 256 |
| 14 | aptX Enhanced | 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 |
| 34 | Opus Mono | Mono | 48 | 64 |
| 35 | Opus Stereo | Stereo | 48 | 128 |

## 19.17 Merlin Backup Connections

Tieline codecs feature comprehensive backup audio options which include automatic fail over to an alternative connection when streaming over IP. Two IP backup options are available:

1. Concurrent: Redundant IP streaming using SmartStream PLUS.
2. 'Cold' backup whereby the second connection is dialed after the codec determines that the primary connection has failed.

### *Redundant SmartStream PLUS IP Streaming*

Tieline's proprietary SmartStream PLUS IP technology ensures you're always on the air. The codec features dual Ethernet IP ports allowing two completely independent IP connections.

There are three levels to SmartStream PLUS IP streaming.

1. The codec can stream simultaneous redundant data streams from both Ethernet ports 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.

### *Backup to ISDN*

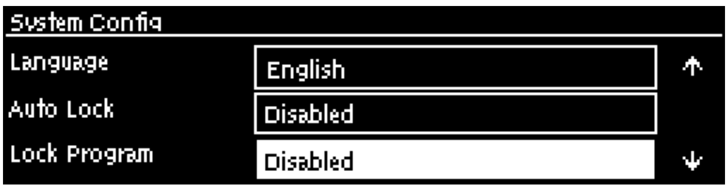
The codec can be configured to fail over to a backup connection over ISDN. Create a program with IP as the primary connection and also create a backup ISDN connection in the same program. For details on configuring backup connections using fail over see Configuring Merlin Point-to-Point Programs.

## 19.18 Lock or Unlock a Program in the Codec

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 your routing requirements 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.

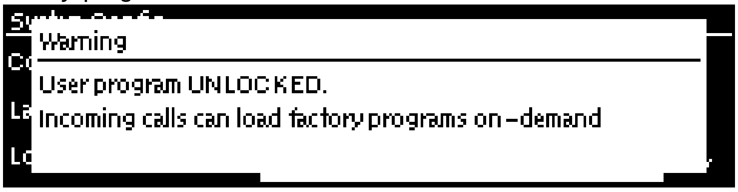
A compatible program type can still connect and specify different connection parameters such as algorithm preferences and bit rates via session data.

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 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.

## 19.19 Locking the Front Panel

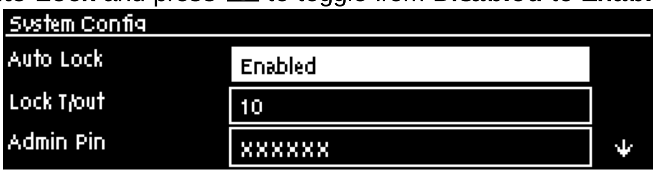
The codec features a front panel lock feature for tamper-proof operation. This feature is disabled by default.

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 to the panel **Lock Timeout** field and press to enter the desired time-out period in seconds.
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.

# 20 Connecting to the ToolBox Web-GUI

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

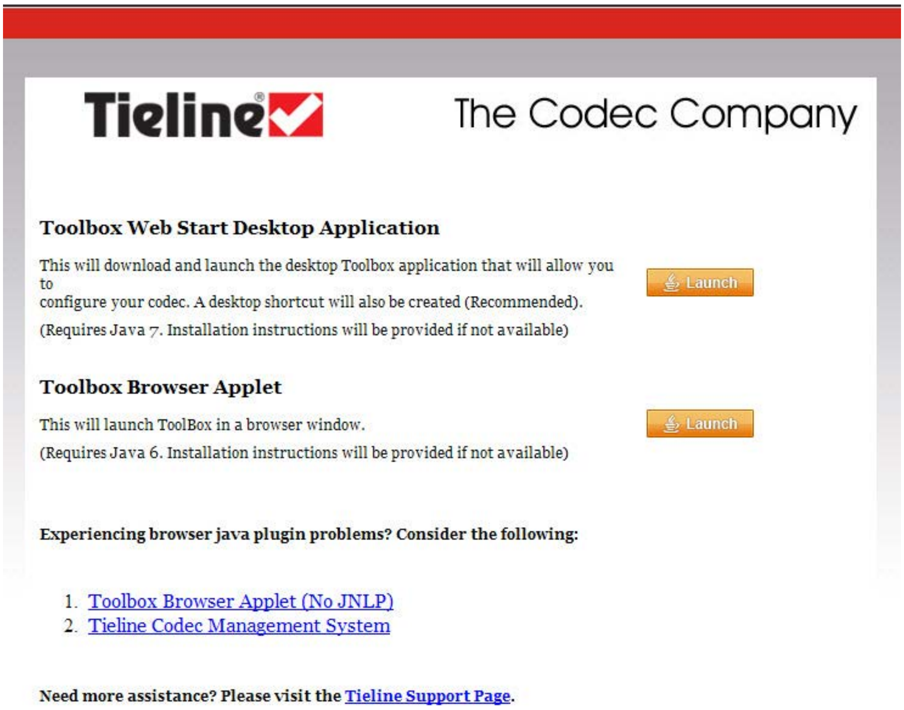
- Internet Explorer 6 or greater on Windows® XP, Windows Vista ® and Windows 7 ®.
- Firefox® 3 or greater on Windows® XP, Windows Vista ® and Windows 7 ®, Solaris™ and Linux®.

### *Web-GUI Prerequisites*

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

## 20.1 Opening the Web-GUI & Login

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




6. Click to launch the ToolBox Web Start Desktop Application (recommended). Note: When you launch for the first time the application will download and launch the desktop Toolbox application

that will allow you to configure your codec. A desktop shortcut will also be created.


*Desktop Icon*

7. When you launch Toolbox an authentication dialog prompts you to enter a password to login. The first time you log in you can enter the default setting "password" and click the **OK** button. Tieline highly recommends you click the hyperlink in the login dialog or visit Changing the Default Password to change the password and have greater security during live broadcasts.


**Important Note:** If you update Java software or clear the Java cache on your computer you will need to repeat the preceding steps.

## 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.

## IPv6 Configuration

It is only possible to configure IPv6 connections using the ToolBox web-GUI.

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

### Port Selection

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

E.g. **192.168.0.176:8080**

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

1. Press the **HOME** button on the codec to return to the **Home** screen.
2. Use the navigation buttons to select **Settings** and press the OK button.
3. Use the navigation button to navigate down to **WebGUI** and press the OK button.
4. Select **Alt. Port** and press OK.
5. Use the **KEYPAD** to enter a new port number and press the OK button to save the new setting (Note: there is no character limit for passwords).
6. Type the IP address of your codec into your browser with a full colon and then the new port number.

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

### *Launching the Toolbox web-GUI*

If you have trouble launching the web-GUI in a browser, type http://<insert codec IP address>.htm directly in your browser.

## 20.2 Changing the Default Password

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


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

### *Creating a New Password*

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

1. Press the **SETTINGS** button.
2. Use the navigation button to select **WebGUI** and press the OK button.
3. Select **Password** and press OK.
4. Use the **KEYPAD** to enter a new password and press the OK 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 menu is permanently set to Tieline and cannot be changed; only the **Password** can be changed.

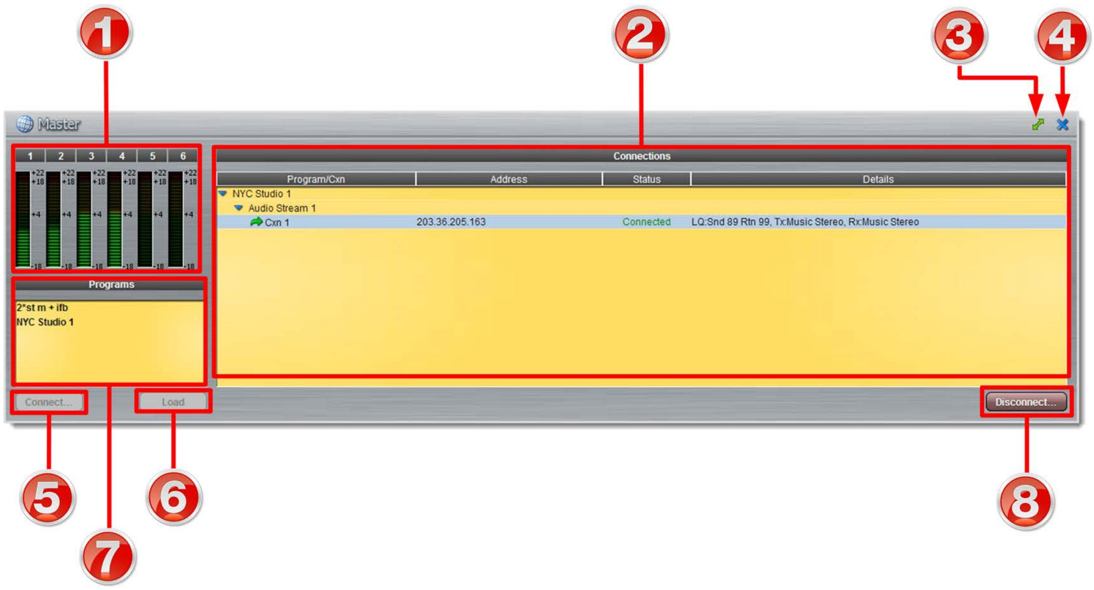
# 21 Using the Web-GUI

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


*Web-GUI Symbols for Opening Panels*

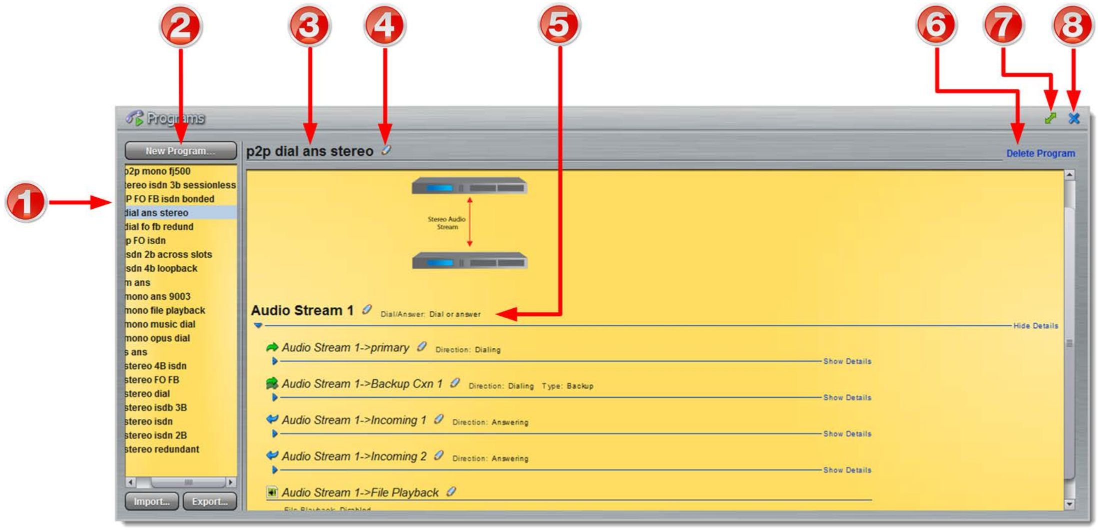
The most recently opened panel is displayed underneath the **Master panel** by default. Click the **Maximize/Minimize** symbol to view a panel in full-screen mode, or click to minimize back to the default panel size.

**Master Panel to Load and Connect Programs, Audio Streams and Connections**



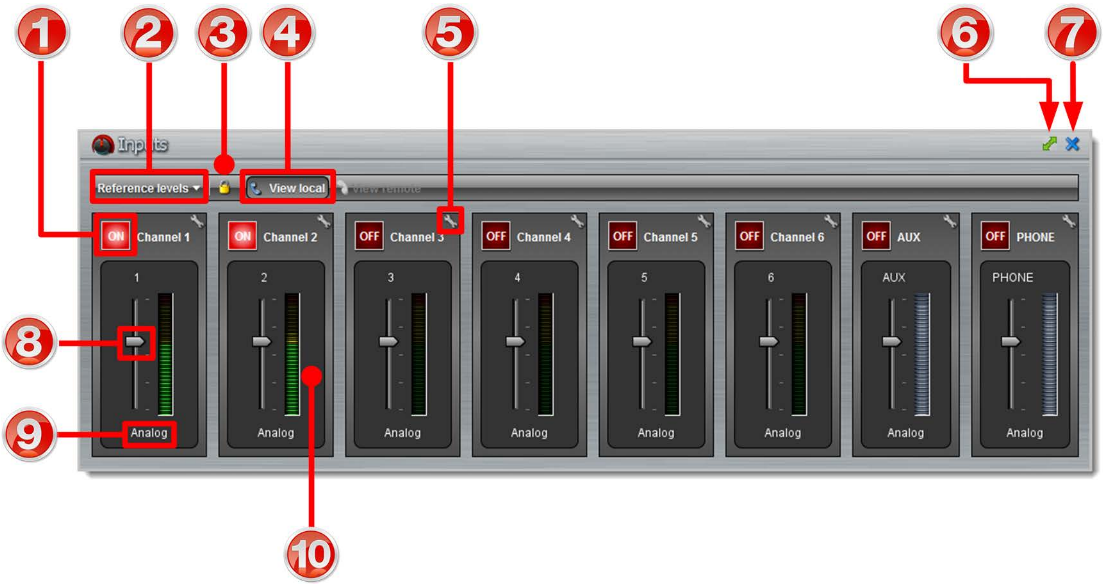

| | Feature | Description |
|---|---|---|
| 1 | Input/Output PPMs | 6 PPM meters to display audio levels for inputs and outputs |
| 2 | **Connections** | Provides a summary of connection details and audio streams |
| 3 | Maximize/Minimize | Click to maximize a panel to view it in full-screen mode, or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

## Programs Panel for Connection Configuration




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

## Inputs Panel for Input Adjustments




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

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

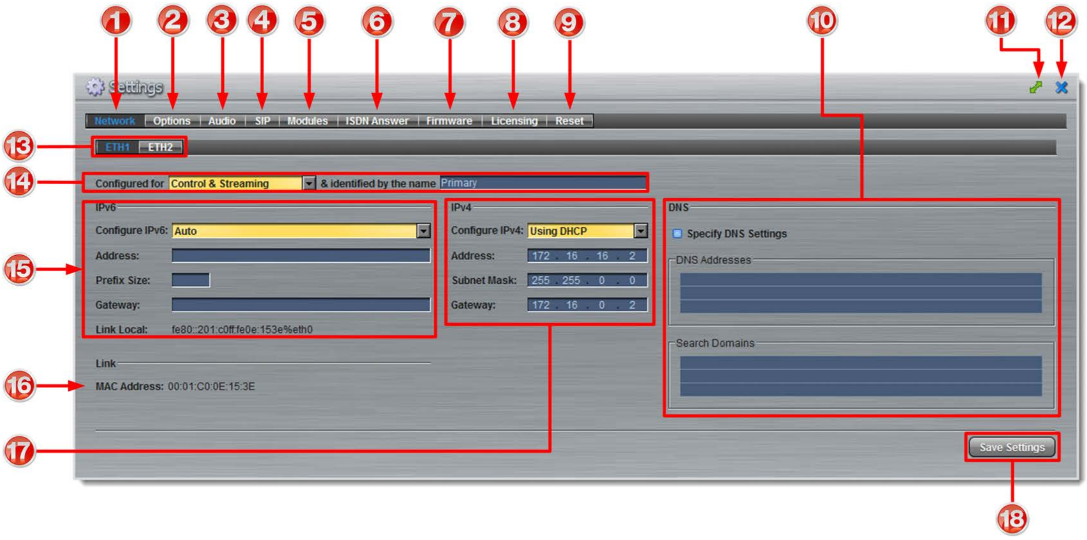
### Rules Panel for Creating Relay Activation Rules


| | Rule | Description |
|---|---|---|
| **1** | Connect/Disconnect a program by toggling a relay input | Click to program Connection and Disconnection by toggling an input |
| **2** | Connect when an input is switched **ON**; Disconnect when another input is switched **ON** | Click to program Connection and Disconnection after different relay inputs are switched **ON** |
| **3** | Synchronise a local relay input with a remote relay output | Click to program a local relay input to Synchronise with the state of a remote relay output |
| **4** | Toggle a relay based on connection status | Click to program a relay to toggle based on connection status |

## Alarms Panel

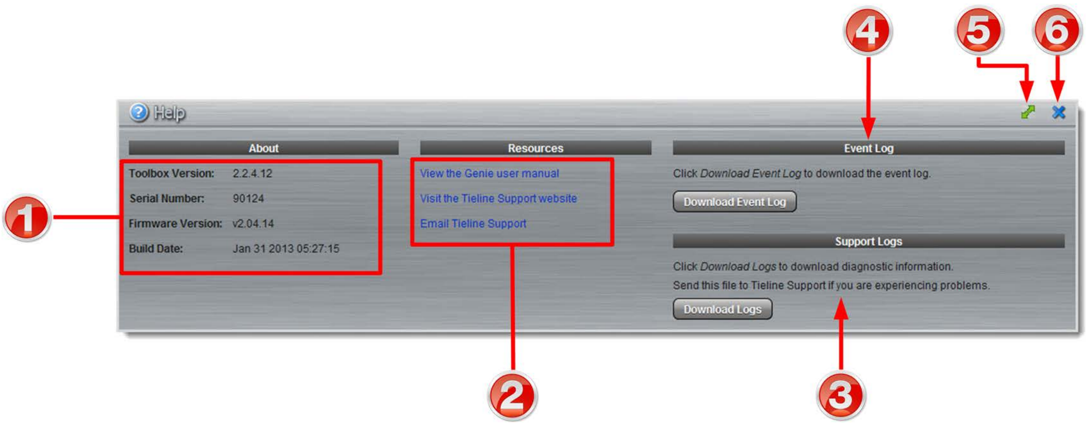

| | Feature | Description |
|---|---|---|
| 1 | **Current Alarms** | Click to view current device alarms |
| 2 | **Alarm History** | Click to view the history of device alarms |
| 3 | **Acknowledge Selected Alarm** | Click to acknowledge an alarm after activation |
| 4 | **Alarm details** pane | Displays alarm details |
| 5 | **Alarm description** pane | Troubleshooting information to assist users when alarms occur |
| 6 | Maximize/Minimize | Click to maximize a panel to view it in full-screen mode, or click to minimize back to the default panel size |
| 7 | **Close** button | Click to close the **Alarms** panel |
| 8 | **Configure alarms** | Click to create or edit alarms. |

## Settings Panel




| | Feature | Description |
|---|---|---|
| 1 | **Network tab** | Click to edit or view codec network configuration settings |
| 2 | **Options tab** | Click to configure RS232 and QoS data settings, Session Port settings and SNMP. |
| 3 | **Audio tab** | Click to configure the AES Output Clock sample rate |
| 4 | **SIP tab** | Click to edit or view SIP configuration settings |
| 5 | **Modules tab** | Click to edit hardware module configuration |
| 6 | **ISDN Answer tab** | Click to configure ISDN Answering settings |
| 7 | **Firmware tab** | Click to view software versions and perform an upgrade |
| 8 | **Licensing tab** | Click to select a license file and install it into the codec |
| 9 | **Reset tab** | Click to reset codec default settings and perform backup/restore of codec programs and settings |
| 10 | **DNS** Pane | Activate to specify DNS addresses and domains to search. |
| 11 | Maximize/Minimize | Click to maximize a panel to view it in full-screen mode, or click to minimize back to the default panel size |
| 12 | **Close** button | Click to close the panel |
| 13 | Network Interface | Select a network interface for configuration options |
| 14 | Network Interface Identifier | Control and streaming configuration options for each network interface, e.g. Ethernet Port 1 or 2. |
| 15 | **IPv6** details | IPv6 addressing details and configuration |
| 16 | **MAC Address** | Device MAC address |
| 17 | **IPv4** details | IPv4 addressing details and configuration |
| 18 | **Save Settings** button | Saves all configuration settings |

## Help Panel




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

## Language Selection

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

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


3. Click to refresh your web-browser and display the new language selected.

# 21.1 Configuring IP Settings

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

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

## *IPv4 versus IPv6*

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

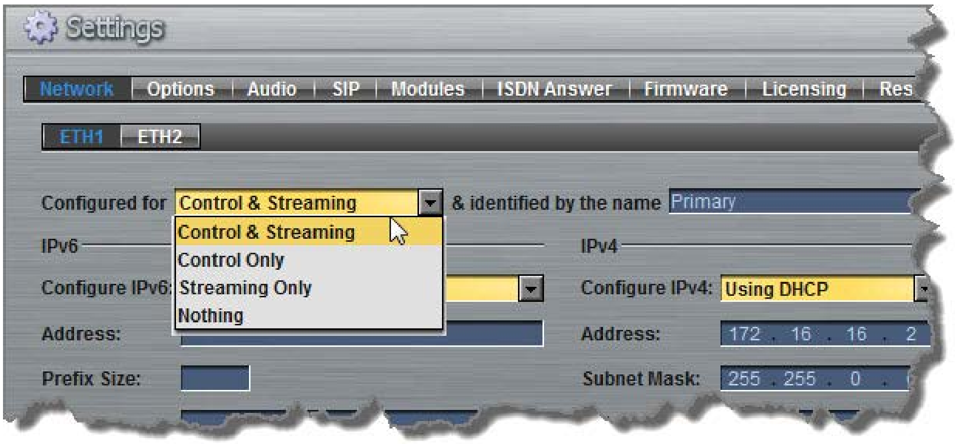
If you want to dial a codec with a public IP address you simply dial the IP address to connect. If you want to dial a codec with a private IP address you need to perform network address translation (NAT). NAT allows a single device, such as a broadband router, to act as an agent between the public internet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

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

## *Configuring Ethernet Ports*

Dual Ethernet ports allow two different IP network connections. These connections can be configured for:

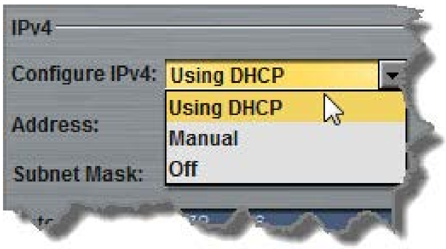
- Streaming audio: stream audio only from an Ethernet port.
- Controlling audio: codec control and command only from the Ethernet port.
- Controlling and Streaming: stream audio and control and command the codec via the Ethernet port.
- Nothing: Disable the Ethernet port from streaming audio and codec command and control.



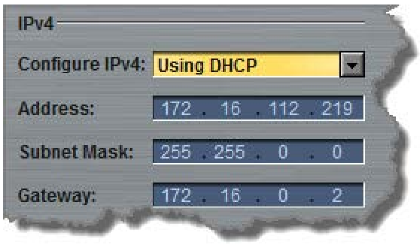

The name entered into the right-hand text box, e.g. **Primary** or **Secondary**, is an identifier used when configuring new programs via the **Programs panel**.

## IPv4 Address Configuration

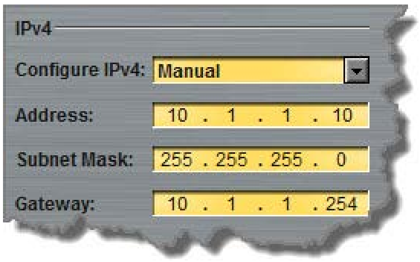
The codec is capable of automatic DHCP address assignment, or manually programmed static IPv4 address configuration via the drop-down **Configure IPv4** menu. If you want to ignore IPv4 settings select **Off**.




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 programmed for DHCP-assigned IP addresses.




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




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

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

## IPv6 Address Configuration

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

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

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

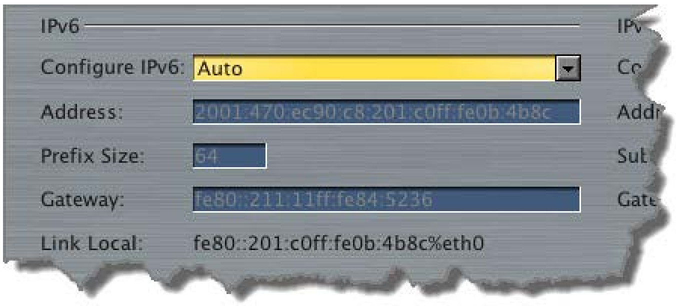
### *Types of IPv6 Addresses*

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

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

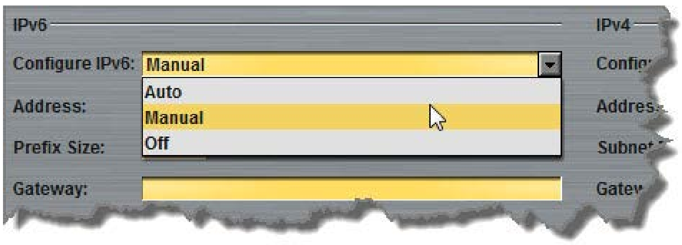
### *Auto Address Assignment*

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




### *Manual IPv6 Address Assignment*

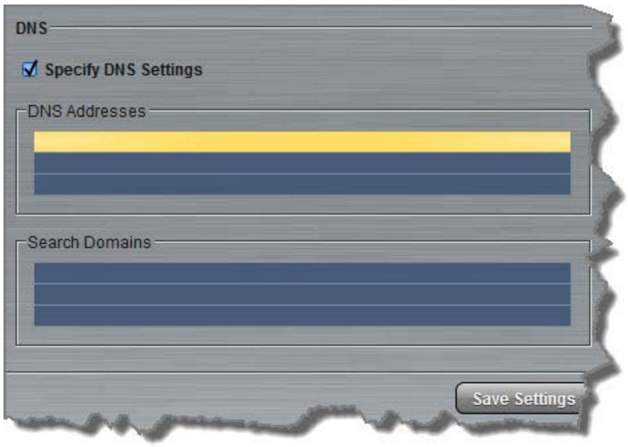
To program IPv6 address details into the codec manually, select **Manual** and enter details into the **Address, Prefix** and **Gateway** text boxes.




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

## *Specifying DNS Settings*

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




### *Configure QoS*

1. Open the web-GUI and click the **Settings** symbol at the top of the screen to display the **Settings panel**.
2. Click the **Options** button.
3. Click in the **DSCP** field and enter the priority setting recommended by your IT administrator.
4. Click **Save settings**.

## 21.2 Configuring ISDN

Two slots are available for inserting optional ISDN modules into the codec. These can be configured using the codec front panel or the Toolbox graphical user interface (GUI).

To configure ISDN using the web-GUI it is necessary to:

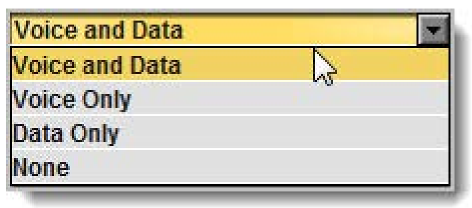
1. Configure all ISDN module settings.
2. Configure ISDN Answering settings.
3. Configure dial and/or answer connection settings in the codec programs.

### 21.2.1 Configuring ISDN Modules

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

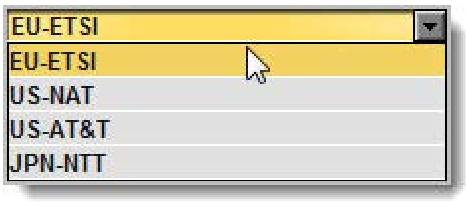
2. Click the **Modules** button at the top of the **Settings panel**.

3. Click the drop down arrow for **Accept** and select your preference of whether to allow or deny circuit switched voice or data calls according to your preferences. The default setting allows both **Voice & Data**.




**Important Note:** G.711 is the algorithm used when **Voice Only** is selected.

4. Click the drop down arrow for **Network** and select the **Network Type** corresponding to the region in which you are using the codec (see ISDN Module Configuration for more details).




5. Click the drop-down arrow for **Line Type** and select your preferred option. Ask your Telco whether your ISDN line is Point-to-Point or Point-to-Multipoint. By default select **Point-to-Multipoint**, unless your switch type is an AT&T 5ESS custom point-to-point.

6. If you are in the US enter DN and SPID numbers as required, or in other regions enter DN or MSN numbers as required.

7. Click the **Save Settings** button when configuration is complete.

**Important Notes:**

***Directory Numbers and Multiple Subscriber Numbers***
Directory Numbers (DN) in North America and Multiple Subscriber Numbers (MSN) in the rest of the world are simply phone numbers associated with an ISDN B channel, like lines listed in a typical phone directory. Your Telco will normally supply 2 DN/MSN numbers for each pair of B channels. However, these numbers may or may not be associated with a specific B channel.

Often broadcasters prefer to predict which B channel will answer an incoming call to ensure audio routing is consistent. However, if a DN or MSN number is not entered in the codec and multiple B channels are available, the codec may use any channel to answer an incoming call. To ensure calls are routed consistently, enter a DN/MSN number (without the country or area code) as the DN/MSN for a B channel, then only that corresponding B channel will answer an incoming call to that number. Programming DN/MSN numbers for each B channel allows the codec to ignore calls without matching DN/MSN numbers. This is the best way to answer calls from codecs in a predictable manner.

***SPID Numbers in North America***
ISDN relies on an initialization procedure for associating Service Profiles with specific terminating equipment (e.g. your audio codec) rather than lines. In the US Telcos assign a Service Profile ID (SPID) number which assists in identifying different ISDN services across the network. Your Telco must provide a SPID for each B channel you order when connecting over US-Nat or US-AT&T networks in the US. A SPID is not required when using the AT&T PTP protocol.

Typically, each ISDN BRI service in the US will have two SPIDs and these must be entered correctly. When you enter a SPID into your codec and connect it to an ISDN line, an initialization and identification process takes place, whereby the terminating equipment (your codec) sends the SPID to the switch. The switch then associates the SPID with a specific Service Profile and directory number.

Note: SPID numbers normally include the phone number and additional prefix or suffix digits up to 20 digits long.

## 21.2.2 Configuring ISDN Answering

It is possible to store up to four different ISDN Answering configurations, which allows up to 4 ISDN B channels to be individually configured for unique answering behaviors.

ISDN answering can be configured to suit:

- Hardware available in the codec, i.e. the number of B channels available.
- Expected dialing behaviors, e.g. if B channels should bond or not, and whether audio streams need to use **Route** tags.
- The type of call being made, e.g. Tieline (with Tieline Session Data) versus non-Tieline.

Each of the four available **Configs** allows you to select which B channel or channels are used to answer a call or calls from incoming ISDN codecs. A maximum of up to 4 B channels can be selected if 2 ISDN modules are installed in the codec.


**Important Note:** B channels can only be selected once and are greyed out once they have been selected in one of the four ISDN **Configs**.

### *Multiple B Channel Bonding Config*

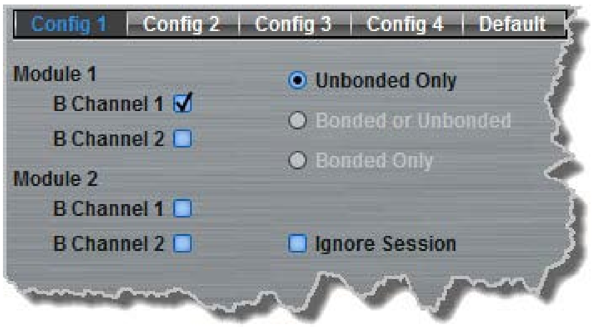
A point-to-point audio stream can also bond multiple B channels to create higher bandwidth connections. In the following example, two B channels from **Module 2** have been selected within **Config 2**. Note that **B Channel 1** in **Module 1** has already been selected in **Config 1** and is therefore unavailable in **Config 2**.


Configure the bonding setting which best suits the audio stream with which this **Config** will be associated. **Bonded or Unbonded** is the best setting in most situations.

| Bonding Setting | Behavior |
|---|---|
| Unbonded Only | Unbonded single B Channel |
| Bonded or Unbonded (May Bond) | Calls using the same algorithm from the same Tieline codec, or sessionless calls, will attempt to bond when received. Calls using incompatible algorithms will not be bonded. Calls may be from Tieline or non-Tieline codecs. |
| Bonded Only | Will only bond compatible algorithms. This mode will reject incompatible calls which cannot be bonded, e.g. G.711 and G.722. |

## *Single B Channel Config*

To use a single 64kbps B channel for a connection (e.g. a 1 x Mono Peer-to-Peer audio stream) simply select a B channel from those available and click the **Save settings** button. If only one B channel is selected then **Unbonded Only** is the default setting.

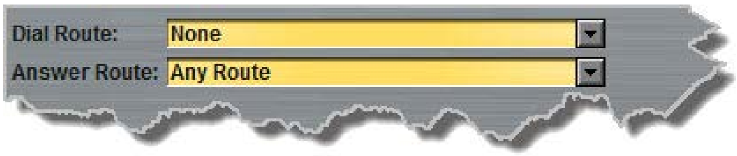



## *Non-Tieline Codecs*

Select the **Ignore Session** check-box when a non-Tieline codec is dialing a Tieline codec over ISDN. This allows you to choose the default encoding setting and **Route** the incoming call to a nominated audio stream via a corresponding **Answer Route** in the answering codec program.

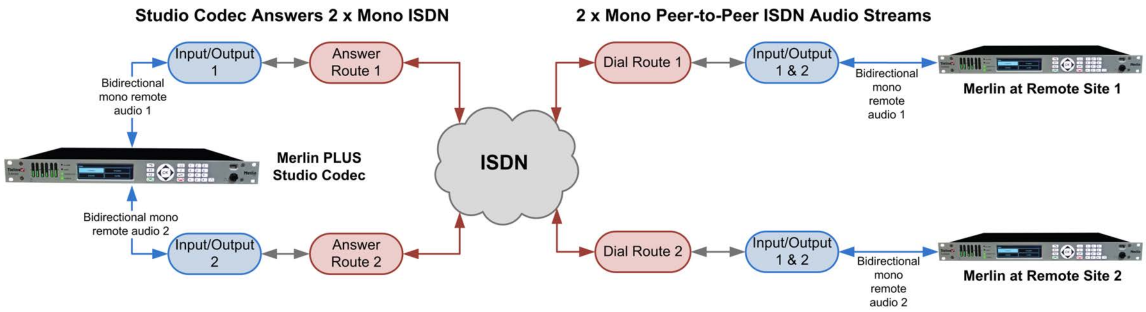
## *Dial and Answer Route Settings in Programs*

**Dial Route** and **Answer Route** tags allow you to associate a B channel (or channels) in a **Config** with a particular incoming audio stream from either Tieline or non-Tieline codecs.




In principle, this operates similarly to how audio ports are used to route multiple audio streams over IP. Selecting different IP audio port numbers allows users to define which incoming IP audio stream is routed to a specific answering audio stream configuration on the codec. This ensures inbound calls from multiple codecs can be consistently routed to the same answering audio streams, and therefore the same inputs and outputs.

This is not necessary in simple point-to-point ISDN audio stream configurations, however it is very useful in multiple audio stream codecs using multiple B channels. When dialing Tieline to Tieline over ISDN using the Merlin or Genie family of codecs, you can configure a **Dial Route** in the dialing codec's program and a corresponding **Answer Route** in the answering codec's program. This will ensure a particular audio stream is routed between two codecs consistently.

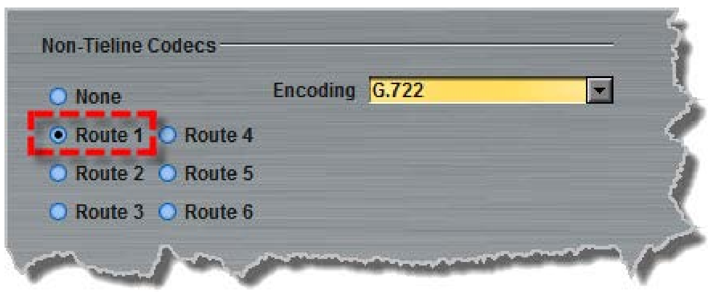



## *Non-Tieline Codecs*

In some situations you may receive a call from a non-Tieline codec which doesn't support **Dial Route** tags. In this situation you can still specify the audio stream **Route** on the answering codec using **Config 1-4** in **ISDN Answer**. You can also select the default algorithm to use.

For example, if a call from a non-Tieline codec is received via **B Channel 1** on **Module 1** (i.e. no **Dial Route** has been specified in the dialing codec):

1. Select a **Route** for this B channel in one of the four **Configs** within **ISDN Answer**, e.g. **Route1**, then select the default **Encoding** algorithm to use when connecting (default setting is **G.722**).




2. This will associate the incoming call with a corresponding **Answer Route** configured in the answering codec program, e.g. **Dial Route 1**.


## *Default Answering Settings*

When a B channel is not associated with a **Config** it inherits the following default settings:

- Tieline Session
- Unbonded
- G.722 algorithm
- Audio route: None

# 21.3 Configuring Input/Output Settings

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

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

## *Configuring Input Channel Settings*

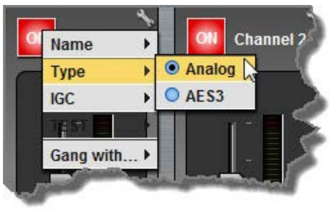
### *Renaming Input Channels:*

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

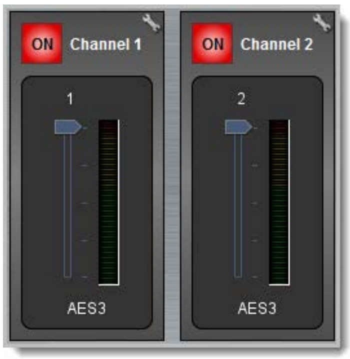

### *Selecting Analog and Digital Audio Sources:*

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

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




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




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

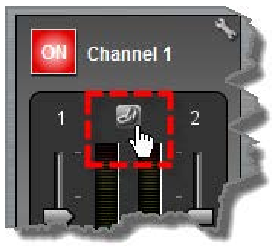
### *Ganging Channels:*

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

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


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


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

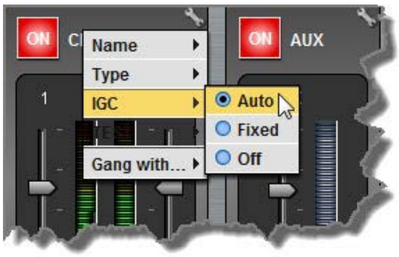



## *Setting Analog Audio Levels*

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

## Other Input Controls

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




**Important Note:** When the auxiliary input (**AUX IN**) is **On** the default mixer configuration sends audio to all inputs. If you are not using the auxiliary input ensure it is **Off** to avoid additional noise in program audio.

## Locking Input Settings

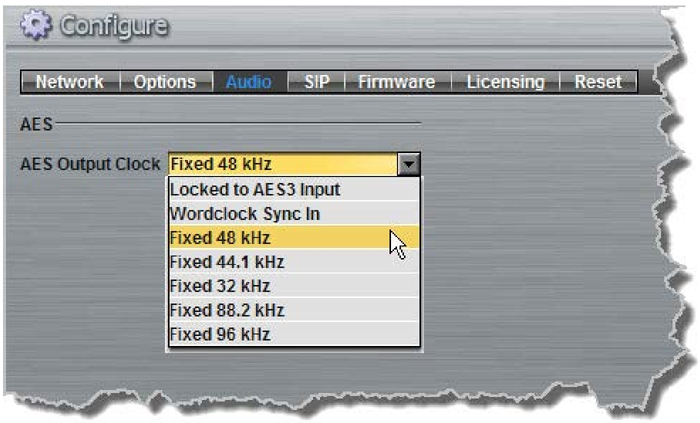
1. Click the **Lock** symbol to lock all **Input panel** settings.
2. When locked, the **Input panel** is greyed out and the lock symbol appears in the bottom-left corner.

## AES3 Output Sample Rate Configuration

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

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

2. Click the **Audio tab** and use the drop-down menu to select your preferred **AES Output Clock** setting, then click **Save Settings**.



# 21.4 Configure Mono or Stereo Peer-to-Peer Programs in Merlin

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

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

This section contains instructions for:

1. Configuring Merlin Peer-to-Peer Programs: Dialing
2. Configuring a Merlin Backup Connection or Auto Reconnect
3. Configuring Merlin to Answer Connections

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

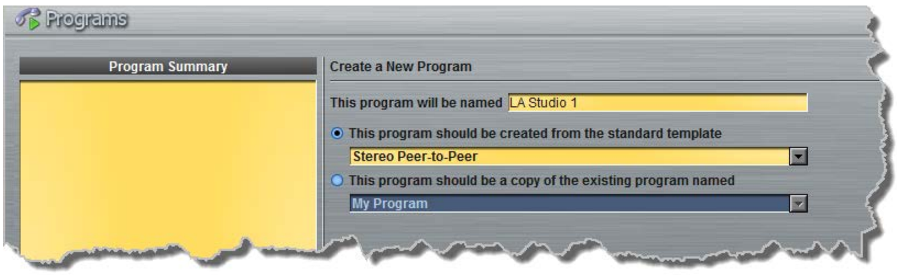
## *Configuring Merlin Peer-to-Peer Programs: Dialing*

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

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

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

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



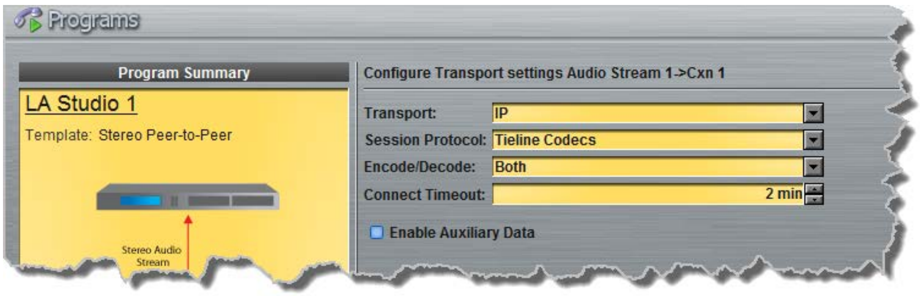

**Important Notes:** When you decide to use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

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


ISDN It is also possible to select a **Dial Route** or **Answer Route** if required. This is useful when routing multiple audio streams over transports like ISDN and is not recommended for use over IP. See Configuring ISDN Answering for more information. Use the default settings for IP connections.

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


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



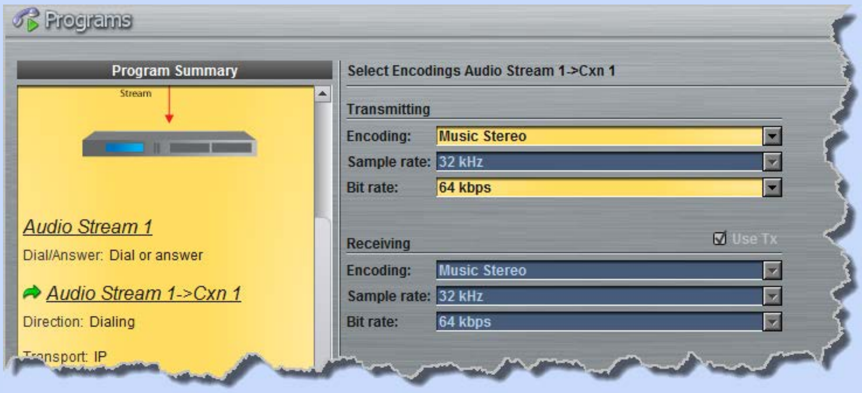

**Important Note:** See RS232 Data Configuration for detailed information on RS232 data and see Enabling Relays and RS232 Data for more information on relay operations.

6. Configure destination codec dialing and encoding settings:

IP For IP connections configure the IP address, ports, and then specify which streaming interface is used to dial this connection, e.g. **Primary** (port **ETH1**) or **Secondary** (port **ETH2**). Note: By default **Any** will select **ETH1** if it is available and **ETH2** if it is unavailable.

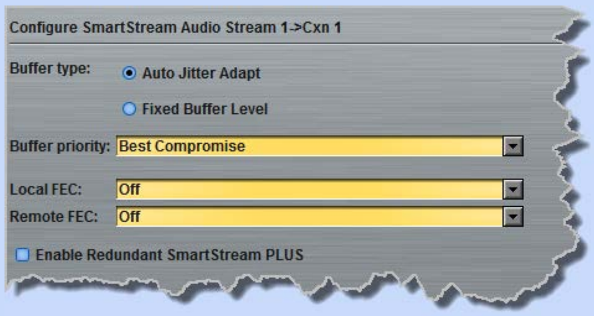
Click **Save Program** to save the program with the default algorithm, jitter and FEC settings which are physically entered in the codec. Alternatively, click **Next** to specify individual algorithm, jitter buffer and FEC settings and configure a backup connection or SmartStream PLUS for this audio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



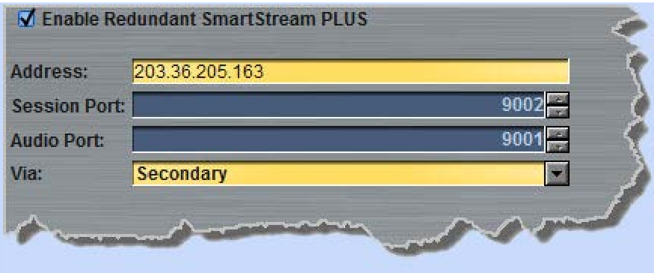

For IP connections click to configure:

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



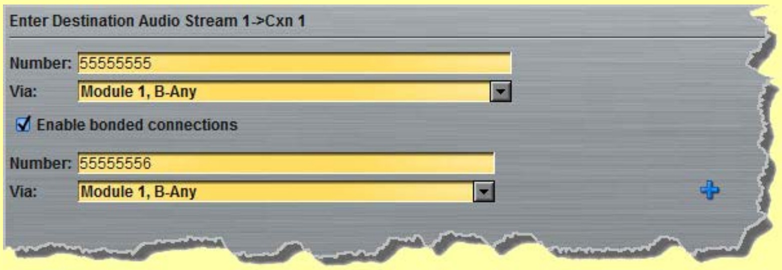

Click the check-box to select **Enable Redundant SmartStream PLUS** and configure dual Ethernet SmartStream IP streaming. Alternatively, click **Next** to configure **Auto Reconnect** or a backup connection, whereby the alternative connection is dialed if the primary connection fails.

By default, primary IP streaming is via **ETH1**. To achieve the maximum level of redundancy select **Secondary** to configure redundant streaming from the secondary IP port **ETH2**. This will stream using **Audio Port 9001** by default and provide automatic IP streaming backup in case one IP connection fails.




**Important Note:** Dual SmartStream PLUS redundant streaming over both Ethernet ports mitigates lost packets on either link and will provide IP network backup if an IP link is lost. To learn more about SmartStream PLUS redundant IP streaming see [http://www.tieline.com/Transports/SmartStream-IP](http://www.tieline.com/Transports/SmartStream-IP)

For ISDN connections enter a number and select which B channel to use. Select the **Enable bonded connections** check-box to configure and bond multiple B channels.




Next, click **Save Program** to save the program with default algorithm settings, or click **Next** to specify a different algorithm and configure a backup connection if required. (recommended).

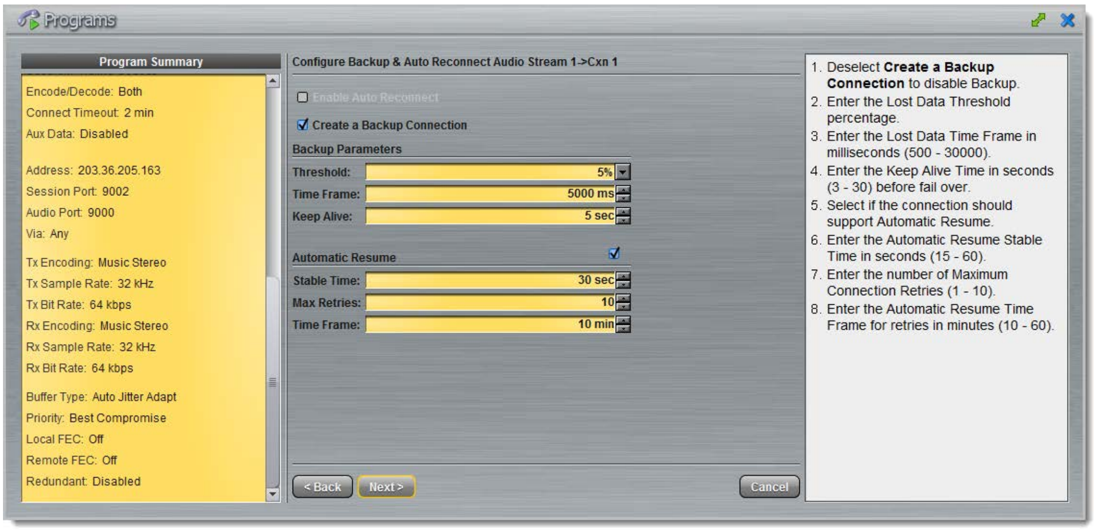

Dialing settings for this ISDN audio stream are now complete.

## Configuring a Merlin Backup Connection or Auto Reconnect

At this point in the wizard you can choose to configure **Auto Reconnect** or create a backup connection for the audio stream you are configuring.

**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.

1. Click to select the check-box for **Create a Backup Connection**. Adjust the parameters and click **Next**.




*Note: The explanations within the following table can be used to assist with backup connection configuration.*

| | Screen Display | Description |
|---|---|---|
| **1** | Threshold | The percentage of lost data measured during a given time frame |
| **2** | Time Frame | The time frame against which lost data is measured |
| **3** | Keep Alive | The keep connection alive time before failing over to a backup connection; Tieline RTP pings every second to confirm connectivity |
| **4** | Automatic Resume | Select the check-box to configure fail back to a higher priority connection |
| **5** | Stable Time | The amount of time a primary connection must remain stable before attempting to fail back from the backup connection |
| **6** | Maximum Retries | The maximum number of fail back retries a codec can try before ending fail back attempts |
| **7** | Time Frame | The time frame used to measure the number of fail back retries attempted |

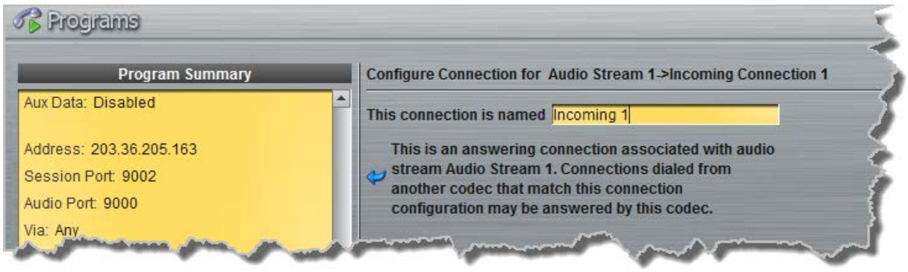
2. Enter a name for the backup connection and click **Next**.


3. Click **Next** to continue through the wizard and configure the backup connection in a similar manner to how you configured the primary connection.

## *Configuring Merlin to Answer Connections*

The codec is capable of being configured to accept calls via different transports (e.g. IP and ISDN), or to accept calls using different audio ports. If you are configuring the codec to allow it to answer one or more incoming audio stream connections:

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




2. Configure the transport settings:

IP

For IP select the **Session Protocol** and **Audio Port**, then click **Next** to configure jitter buffer and FEC settings.


Click to configure:

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

ISDN

For ISDN, settings are determined by ISDN module answering settings. For more details see Configuring ISDN Answering.

3. After configuring all settings there are 2 options:

i. If you want to create another answering connection, select the check-box for **Create**

**another answering connection** and continue through the wizard.

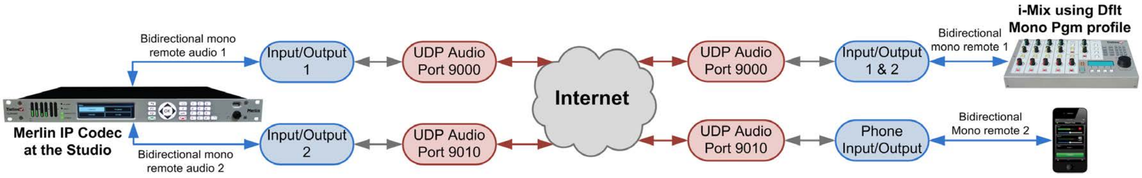
ii. Click **Save Program** to save the program at this point.

4. After you save the program you can select the check-box if you want to connect the program immediately, then click **Finish**.


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

## 21.5 Configure 2 Mono Peer-to-Peer Answering Programs in Merlin

It is possible to create two simultaneous mono peer-to-peer audio stream connections with different codecs. This is similar to a 'Dual Mono' profile in G3 Tieline codecs.




The following program wizard procedure displays the configuration screens to create an answering connection for each incoming call. See Configuring Merlin Point-to-Point Programs for more details about individual settings within the program wizard.

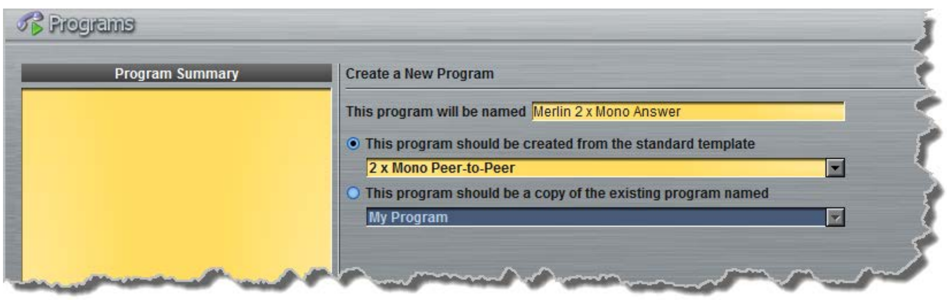
### *Routing 2 Incoming Mono Audio Streams to Specific Codec Outputs*

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

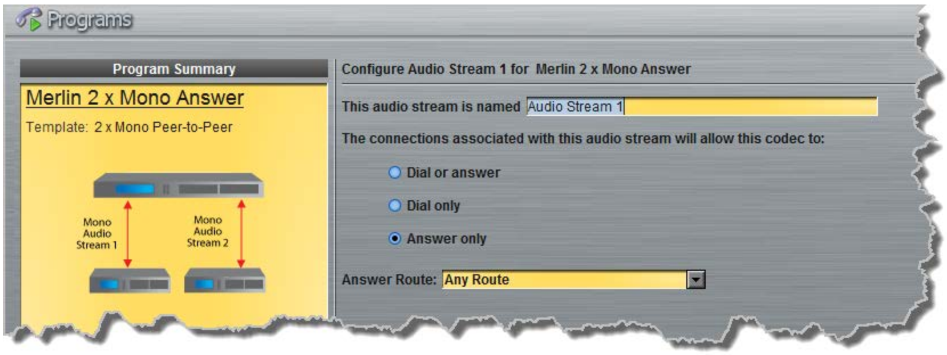
- You cannot edit a program when it is currently loaded in the codec.
- You can lock a loaded custom program in a codec to ensure the currently loaded program type cannot be unloaded by a codec dialing in with a different program type.
- Some drop-down menus and settings may be greyed out intentionally depending on features available and the transport selected (e.g. IP or ISDN).
- 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.

If your intention is to ensure 2 incoming mono peer-to-peer audio stream connections are always routed to the same outputs, configure a new answering program as follows:

1. Open the web-GUI and click the **Programs**  symbol at the top of the screen to display the **Programs panel**.
2. Click the **New Program** button to open the wizard and:
   - Click in the text box to name the new program.
   - Select **2 x Mono Peer-to-Peer,** or if you want to use an existing program as a template, select this option. Then click **Next**.



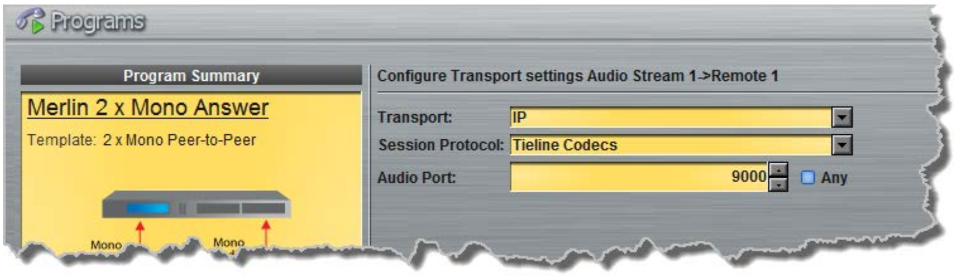

3. Enter a name for the **Audio Stream** and select **Answer only**. Then click **Next**. Note: if the codec you are configuring needs to dial connections as well as answer, you can select **Dial or answer** and configure both dialing and answering connections. For the purposes of this example dialing is excluded.




ISDN It is also possible to select an **Answer Route** if required. This is useful when routing multiple audio streams over transports like ISDN and is not recommended for use over IP. See Configuring ISDN Answering for more information. Use the default settings for IP connections.

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

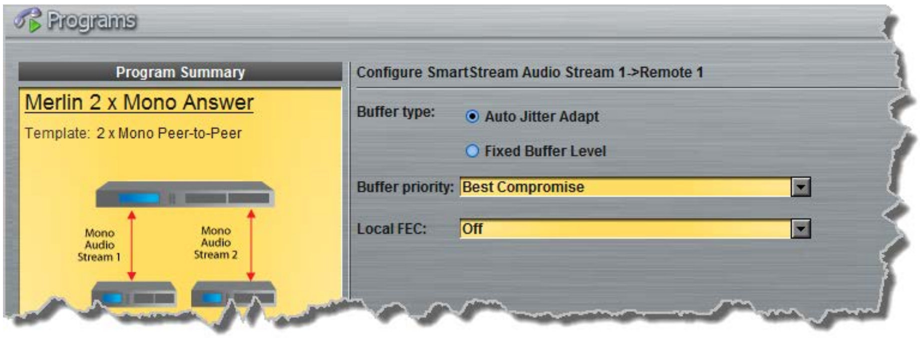

5. Click your preferred connection **Transport** and use the drop-down **Session Protocol** menu to select **Tieline Codecs**. Ensure the **Any** check-box is not selected, then click **Next**.




**Important Notes:** If **Any** is not selected then a codec dialing this connection and using the port specified will always be routed to output 1 on the codec receiving the call. By default port 9000 is the port specified for the first IP connection made by any Tieline codec; port 9010 is used by default for the second connection in a 2 x mono peer-to-peer program. If you change the default port settings on the dialing codec, copy this setting into the codec you are configuring to answer the call.

ISDN ISDN module answering settings are used if you select ISDN as the connection transport. For more details see Configuring ISDN Answering.

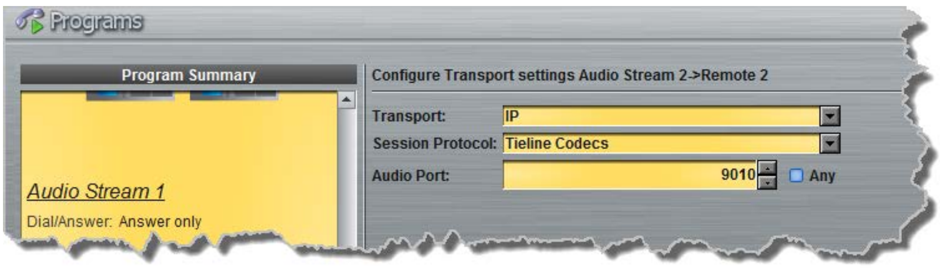
6. Click to configure the jitter buffer and FEC settings if required.




7. Enter a name for the second **Audio Stream** and select **Answer only**. Then click **Next**.


8. Enter the name of the second audio stream connection in the text box and click **Next**.


9. Click your preferred connection **Transport** and use the drop-down **Session Protocol** menu to select **Tieline Codecs**. Ensure the **Any** check-box is not selected, then click **Next**.




ISDN

ISDN module answering settings are used if you select ISDN as the connection transport. For more details see Configuring ISDN Answering.

10. Continue through the steps in the wizard to complete configuration in the same way as the first connection was configured. Click **Save Program** at the end of this process. The newly created program will be displayed in the left pane within the **Programs panel** and in the **Master panel**.

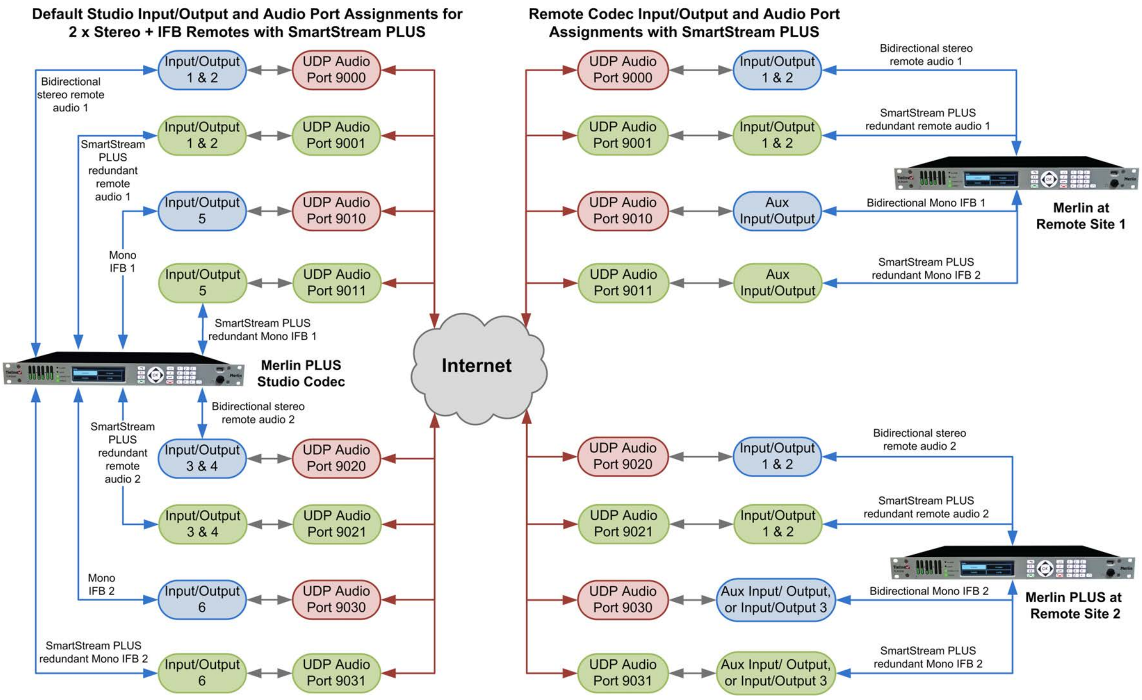
When this program is loaded any codec dialing in using IP1 (using default Tieline IP port settings) will be routed to output 1 on the codec and a codec dialing in using IP2 will be routed to output 2.

# 21.6 Configure Mono or Stereo + IFB Dialing Programs

This program is designed to allow remote Merlin and Merlin PLUS codecs to dial a Merlin or Merlin PLUS codec at the studio and transmit:

1. A bidirectional mono or stereo audio stream connection.
2. A separate bidirectional mono IFB audio stream for communications.

This program can also include SmartStream PLUS dual IP streaming. The following diagram indicates the default input, output and port assignments for **Mono or Stereo Peer-to-Peer + IFB** Programs using SmartStream PLUS and dialing a Merlin PLUS codec at the studio.




*2 x Mono/Stereo Peer-to-Peer + IFB Remotes dialing Merlin PLUS at the studio*

The following setup instructions describe how to configure a stereo audio stream and IFB audio stream, with a backup connection, in order to connect with a Merlin or Merlin PLUS codec at the studio.

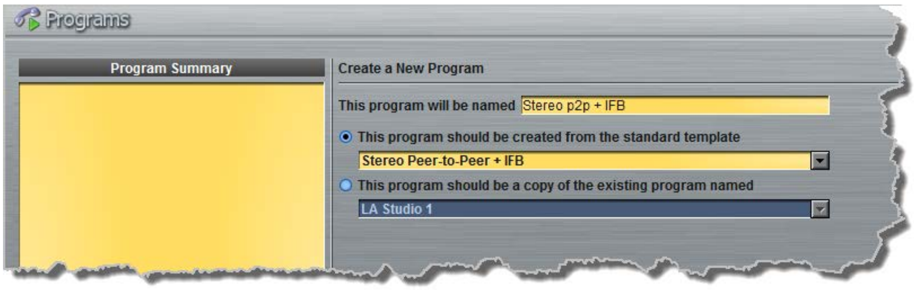
## *Configuring a Mono or Stereo Audio Stream: Dialing*

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

- The auxiliary input is used by default for the IFB communications channel in Merlin codecs. In Merlin PLUS codecs the auxiliary input and XLR input 3 are mixed together by default for 1 x Mono/Stereo Peer-to-Peer + IFB programs.
- You cannot edit a program when it is currently loaded in the codec.
- You can lock a loaded custom program in a codec to ensure the currently loaded program type cannot be unloaded by a codec dialing in with a different program type.
- Some drop-down menus and settings may be greyed out intentionally depending on features available and the transport selected (e.g. IP or ISDN).
- It is possible to save a program at several points throughout the program wizard and use default settings to save configuration time.
- To learn more about programs see the section titled About Program Dialing.

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

2. Click the **New Program** button to open the wizard and:
   - Click in the text box to name the new program.
   - Select **Mono/Stereo Peer-to-Peer + IFB,** or if you want to use an existing program as a template, select this option. Then click **Next**. Note: The following example is configured to connect a stereo audio stream and mono IFB stream.



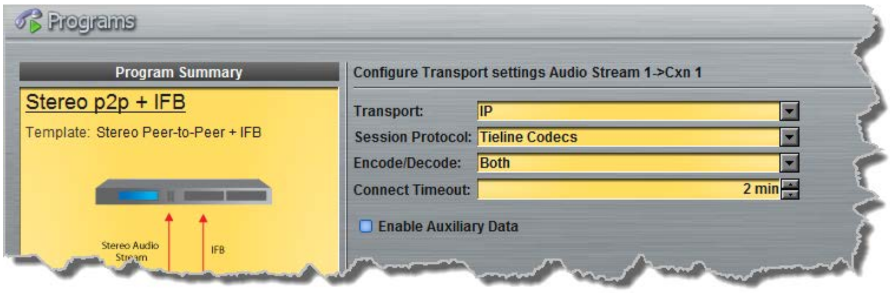

**Important Notes:** When you decide to use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

3. Enter the **Audio Stream** name and configure the codec to **Dial only**. Then click **Next**.


ISDN It is also possible to select a **Dial Route** if required. This is useful when routing multiple audio streams over transports like ISDN and is not recommended for use over IP. See Configuring ISDN Answering for more information. Use the default settings for IP connections.

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


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




Important Note: Bidirectional auxiliary IP data is available on one audio stream. When auxiliary data is enabled on one stream the option is greyed out for the other audio stream in the program wizard. See RS232 Data Configuration for detailed information on RS232 data and see Enabling Relays and RS232 Data for more information on relay operations.

6. Configure destination codec dialing and encoding settings:

IP

For IP connections configure the IP address, ports, and then specify which streaming interface is used to dial this connection, e.g. **Primary** (port **ETH1**) or **Secondary** (port **ETH2**). Note: By default **Any** will select **ETH1** if it is available and **ETH2** if it is unavailable.

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 a backup connection or SmartStream PLUS for this audio stream (recommended).

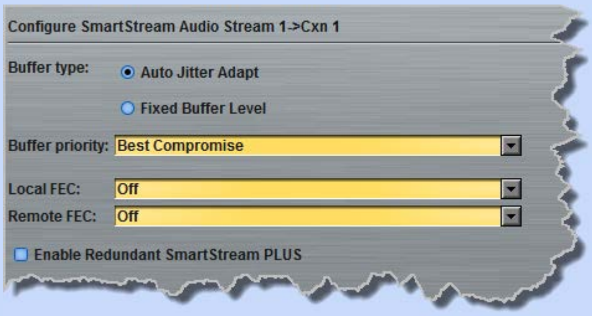

**Note:** If you connect multiple remote codecs simultaneously to a Merlin PLUS codec at the studio to create 2 x Mono or Stereo Peer-to-Peer + IFB connections, use audio port 9020 to dial the second mono/stereo connection at the studio. Mono or stereo audio over this audio stream connection will be routed via audio inputs/outputs 3 and 4 on the studio Merlin PLUS codec.

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:

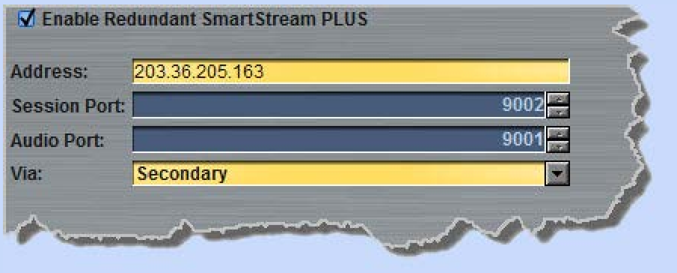
- **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**, or

- **Fixed Buffer Level** and enter the **Jitter Depth**, which must be between 12ms and 5000ms depending on the algorithm you select.
- **Local** and **Remote FEC** settings if required.




Click the check-box to select **Enable Redundant SmartStream PLUS** and configure dual Ethernet SmartStream IP streaming. Alternatively, click **Next** to configure **Auto Reconnect** or a backup connection, whereby the alternative connection is dialed if the primary connection fails.

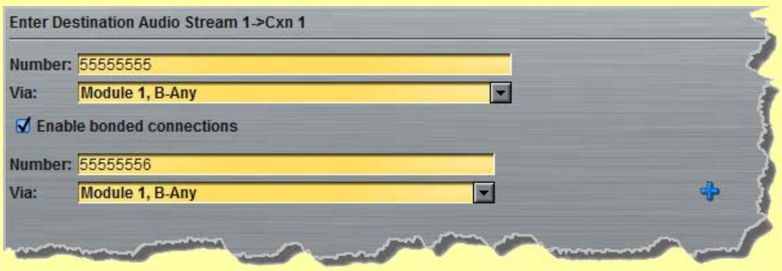
By default, primary IP streaming is via **ETH1**. To achieve the maximum level of redundancy select **Secondary** to configure redundant streaming from the secondary IP port **ETH2**. This will stream using **Audio Port 9001** by default and provide automatic IP streaming backup in case one IP connection fails.




**Important Note:** Dual SmartStream PLUS redundant streaming over both Ethernet ports mitigates lost packets on either link and will provide IP network backup if an IP link is lost. To learn more about SmartStream PLUS redundant IP streaming see [http://www.tieline.com/Transports/SmartStream-IP](http://www.tieline.com/Transports/SmartStream-IP)

ISDN

For ISDN connections enter a number and select which B channel to use. Select the **Enable bonded connections** check-box to configure and bond multiple B channels.




Next, click **Save Program** to save the program with default algorithm settings, or click **Next** to specify a different algorithm and configure a backup connection if required. (recommended).


Dialing settings for this ISDN audio stream are now complete.

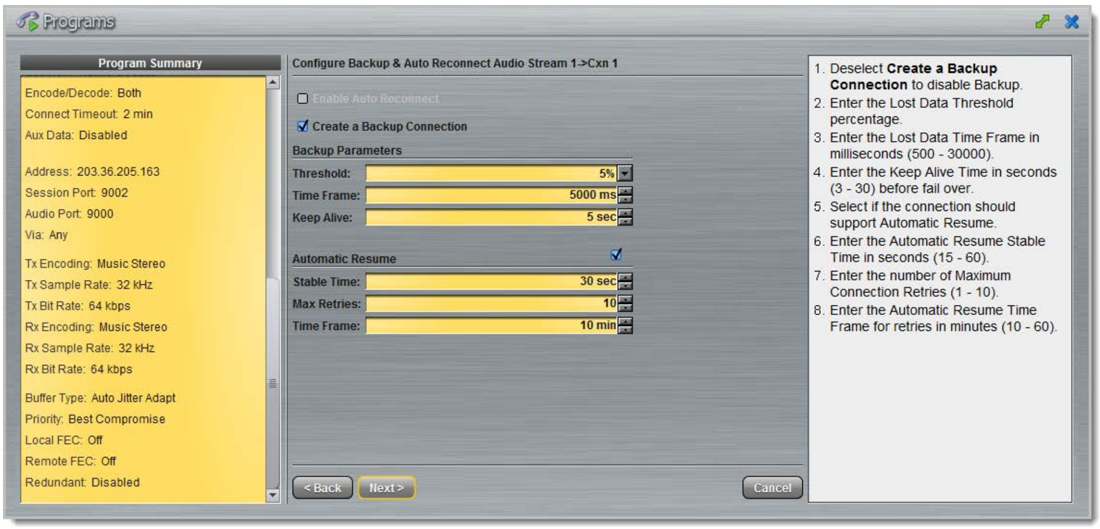
## *Configuring a Backup Connection or Auto Reconnect*

At this point in the wizard you can choose to configure **Auto Reconnect** or create a backup connection for the audio stream you are configuring.

**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.

To configure a backup connection:

1. Click to select the check-box for **Create a Backup Connection**. Adjust the parameters and click **Next**.




*Note: The explanations within the following table can be used to assist with backup connection configuration.*

| | Screen Display | Description |
|---|---|---|
| 1 | Threshold | The percentage of lost data measured during a given time frame |
| 2 | Time Frame | The time frame against which lost data is measured |
| 3 | Keep Alive | The keep connection alive time before failing over to a backup connection; Tieline RTP pings every second to confirm connectivity |
| 4 | Automatic Resume | Select the check-box to configure fail back to a higher priority connection |
| 5 | Stable Time | The amount of time a primary connection must remain stable before attempting to fail back from the backup connection |
| 6 | Maximum Retries | The maximum number of fail back retries a codec can try before ending fail back attempts |
| 7 | Time Frame | The time frame used to measure the number of fail back retries attempted |

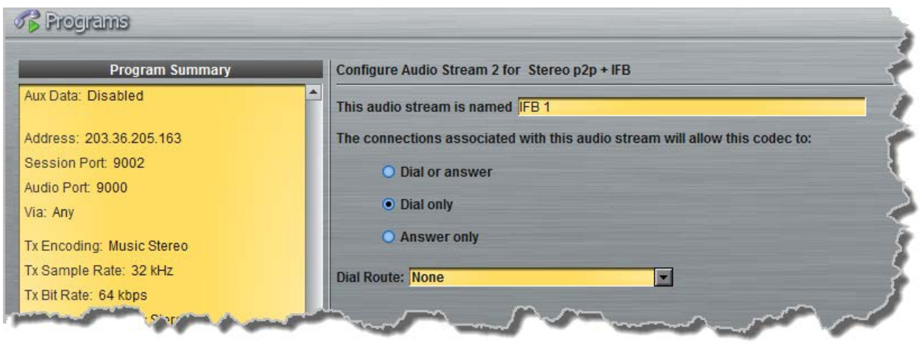
2. Enter a name for the backup connection and click **Next**.


3. Click **Next** to continue through the wizard and configure the backup connection in a similar manner to how you have configured the primary connection.

## Configure the Bidirectional IFB Audio Stream

When you have finished configuring SmartStream PLUS, Auto Reconnect or a backup connection, proceed with configuration of the IFB audio stream in the wizard.

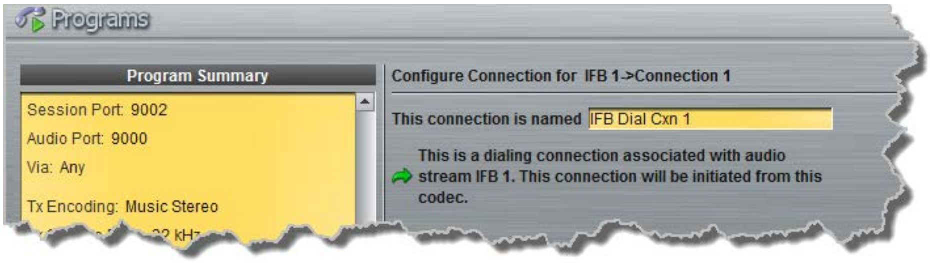
1. Enter the IFB Audio Stream name and configure the codec to **Dial only**. Then click **Next**.



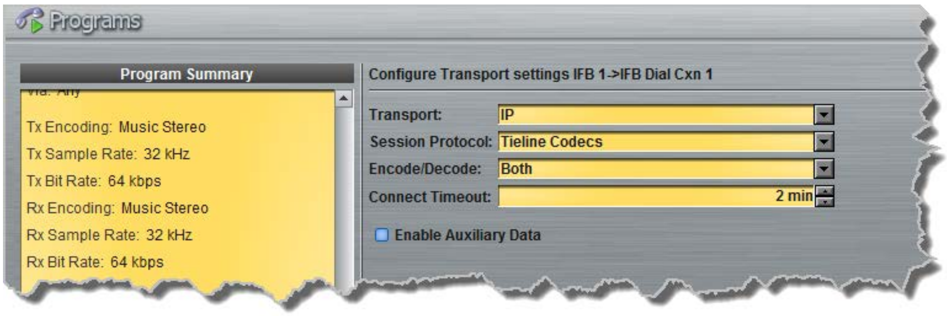

ISDN It is also possible to select a **Dial Route** if required. This is useful when routing multiple audio streams over transports like ISDN and is not recommended for use over IP. See Configuring ISDN Answering for more information.

2. This audio stream connection in the wizard will allow the codec to dial. Enter the connection

name in the text box, then click **Next**




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




**Important Note:** Bidirectional auxiliary IP data is available on one audio stream. When auxiliary data is enabled on one stream the option is greyed out for the other audio stream in the program wizard. See RS232 Data Configuration for detailed information on RS232 data and see Enabling Relays and RS232 Data for more information on relay operations.

4. Configure destination codec dialing and encoding settings:

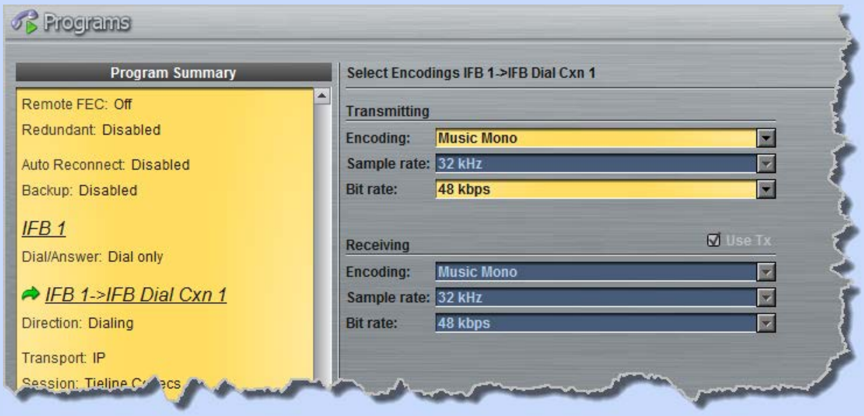
IP

For IP connections configure the IP address, ports, and then specify which streaming interface is used to dial this connection, e.g. **Primary** (port **ETH1**) or **Secondary** (port **ETH2**). Note: By default **Any** will select **ETH1** if it is available and **ETH2** if it is unavailable.

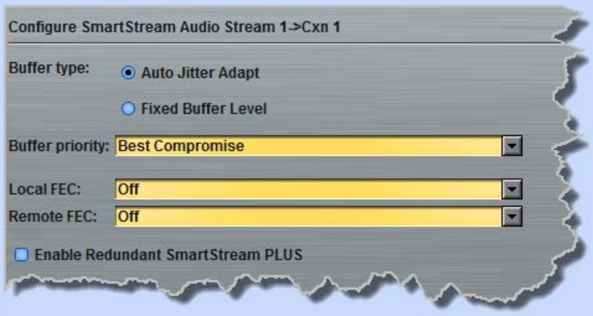
Click **Save Program** to save the program with the default algorithm, jitter and FEC settings which are physically entered in the codec. Alternatively, click **Next** to specify individual algorithm, jitter buffer and FEC settings and configure a backup connection or SmartStream PLUS for this audio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


**Note:** The default **Audio Port** is 9010 for this IP audio stream. If you connect multiple remote codecs simultaneously to a Merlin PLUS codec at the studio to create 2 x Mono or Stereo Peer-to-Peer + IFB connections, use audio port 9030 to dial the second IFB connection at the studio. IFB audio over this audio stream connection will be routed via audio input/output 6 on the studio Merlin PLUS codec.

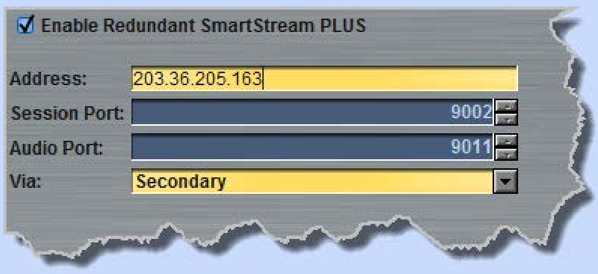
Click the drop-down arrows on the right-hand side of each text box to adjust the **Encoding**, **Sample rate** and **Bit rate** options.




Click the check-box to select **Enable Redundant SmartStream PLUS** and configure dual Ethernet SmartStream IP streaming. Alternatively, click **Next** to configure **Auto Reconnect** or a backup connection, whereby the alternative connection is dialed if the primary connection fails.



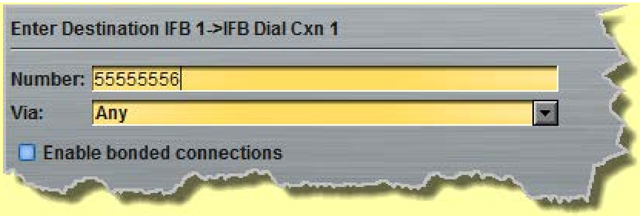

By default, primary IP streaming is via **ETH1**. To achieve the maximum level of redundancy select **Secondary** to configure redundant streaming from the secondary IP port **ETH2**. This will stream using **Audio Port 9011** by default and provide automatic IP streaming backup in case one IP connection fails.




**Important Note:** Dual SmartStream PLUS redundant streaming over both Ethernet ports mitigates lost packets on either link and will provide IP network backup if an IP link is lost. To learn more about SmartStream PLUS redundant IP streaming see [http://www.tieline.com/Transports/SmartStream-IP](http://www.tieline.com/Transports/SmartStream-IP)

ISDN

For ISDN connections enter a number and select which B channel to use. Select the **Enable bonded connections** check-box to configure and bond multiple B channels.




Next, click **Save Program** to save the program with default algorithm settings, or click **Next** to specify a different algorithm and configure a backup connection if required. (recommended). Dialing settings for this ISDN audio stream are now complete.

5. Click **Save Program** to complete configuration. After you save the program you can select the check-box if you want to connect the program immediately, then click **Finish**.

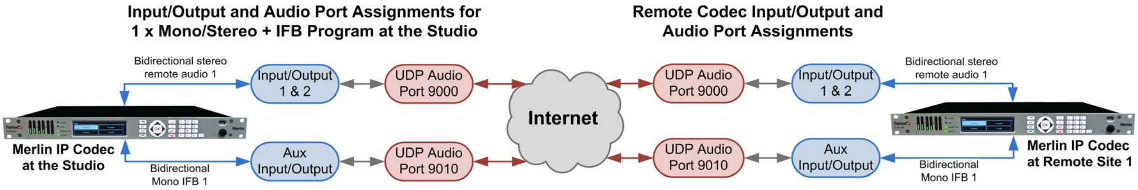

6. The newly created program will be displayed in the left pane within the **Programs panel** and in the **Master panel**. Select and connect audio streams in a program using the **Master panel**, or dial the program manually using the codec front panel.

# 21.7 Configure Mono or Stereo + IFB Answering Programs

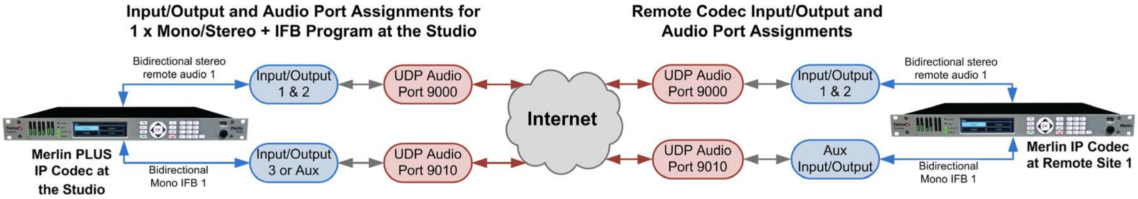
This program is designed to allow Merlin and Merlin PLUS codecs to answer a call from an incoming codec and receive:

1. A bidirectional mono or stereo audio stream connection.
2. A separate bidirectional mono IFB audio stream for communications.

A remote Merlin or Merlin PLUS codec can dial into a studio Merlin or Merlin PLUS codec to create these audio stream connections.




*Merlin codec at the studio connects to a remote Merlin codec*

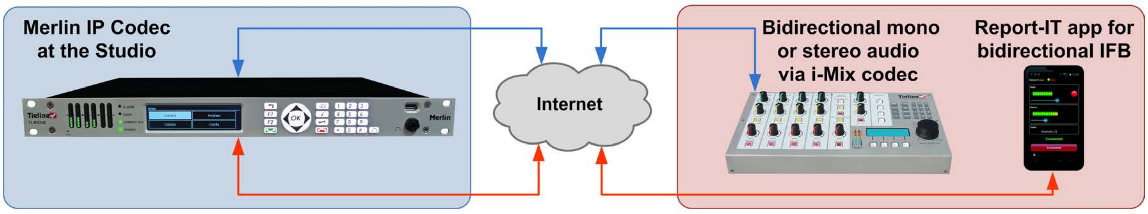



*Merlin PLUS at the studio connects to a remote Merlin (Note Aux/input 3 IFB routing option)*

## *Connecting other Tieline Codecs*

When a codec which only supports a mono or stereo audio stream attempts to connect (e.g. Commander G3, i-Mix-G3 or Bridge-IT codecs not supporting a separate IFB audio stream), the Merlin or Merlin PLUS codec at the studio will accept the call and stream mono or stereo audio only. A second IP codec or smartphone running the Report-IT application can also be configured to connect and deliver the bidirectional mono IFB audio stream.

**1 x Mono or Stereo Peer-to-Peer + IFB via Two Remote Devices**




**Important Note:** Remember to lock the program when connecting a **Mono or Stereo Peer-to-Peer + IFB** program using two devices at the remote site. This will avoid the first mono or stereo call unloading the **Peer-to-Peer + IFB** program at the studio and loading a mono or stereo peer-to-peer program, which would cause the second connection to fail.

## *Configuring Mono or Stereo Peer-to-Peer + IFB Programs*

In most situations the studio codec will answer incoming audio stream connections from the remote site. The following procedure outlines configuration of an answering program for the studio codec.

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

- The auxiliary input is used by default for the IFB communications channel in Merlin codecs. In Merlin PLUS codecs the auxiliary input and XLR input 3 are mixed together by default for 1 x Mono/Stereo + IFB programs.
- You cannot edit a program when it is currently loaded in the codec.
- You can lock a loaded custom program in a codec to ensure the currently loaded program type cannot be unloaded by a codec dialing in with a different program type.
- If the codec at the studio will receive both mono and stereo peer-to-peer + IFB calls from different remote sites at different times, we recommend you configure and load a **1 x Stereo Peer-to-Peer + IFB** answering program and lock this in the codec at the studio. This will accept both mono and stereo audio stream connections. If a codec with a **Mono Peer-to-Peer + IFB** program calls the studio, the incoming mono stream will be mixed to both the left and right outputs at the studio.
- Some drop-down menus and settings may be greyed out intentionally depending on features available and the transport selected (e.g. IP or ISDN).
- It is possible to save a program at several points throughout the program wizard and use default settings to save configuration time.
- To learn more about programs see the section titled About Program Dialing.

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

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


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

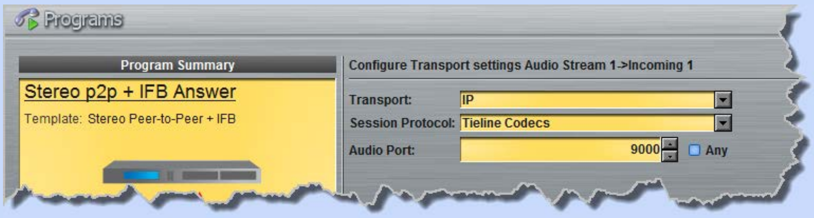
3. Enter the **Audio Stream** name and select **Answer only**. Then click **Next**. Note: If the codec you are configuring needs to dial connections as well as answer, you can select Dial or answer and configure both dialing and answering connections. For the purposes of this example dialing is excluded in configuring a stereo and mono IFB stream.

ISDN It is also possible to select an **Answer Route** if required. This is useful when routing multiple audio streams over transports like ISDN and is not recommended for use over IP. See Configuring ISDN Answering for more information. Use the default settings for IP connections.

4. Enter the connection name in the text box, then click **Next.**

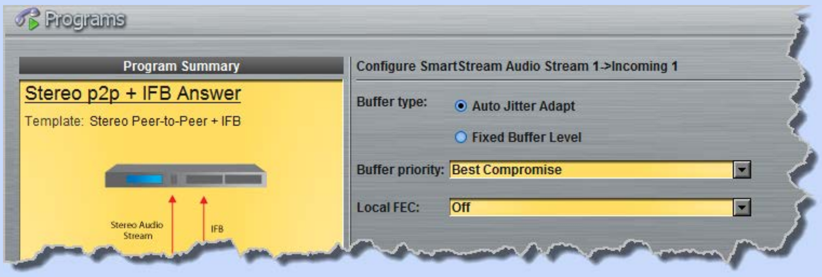

5. Configure the transport settings:

IP For IP click the drop-down **Session Protocol** menu and select **Tieline Codecs** and ensure the **Any** check-box is not selected, then click **Next**.




Click to configure:

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



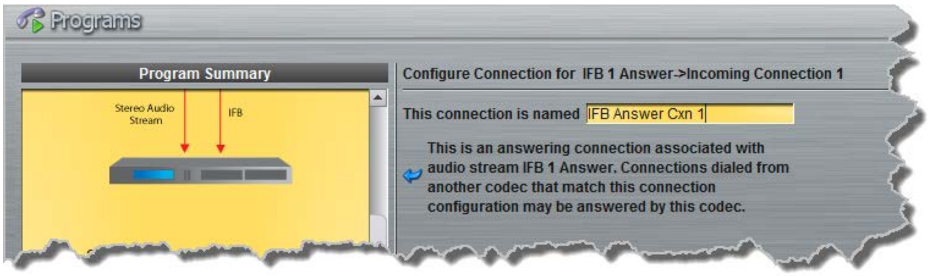

ISDN For ISDN, settings are determined by ISDN module answering settings. For more details see Configuring ISDN Answering.

6. After configuring all settings there are 2 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 **Next Stream** to configure the IFB audio stream.

7. Enter the IFB **Audio Stream** name and select **Answer only**, then click **Next**.

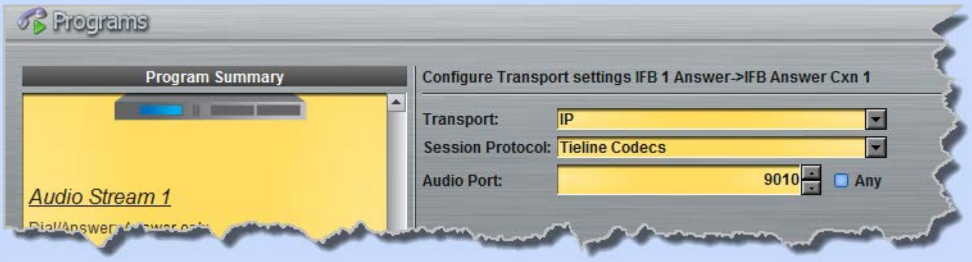

8. Enter the IFB audio stream connection name in the text box and click **Next.**




9. Configure the transport settings:

IP

For IP click the drop-down **Session Protocol** menu and select **Tieline Codecs** and ensure the **Any** check-box is not selected, then click **Next**.




Click to configure:

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

ISDN

For ISDN settings are determined by ISDN module answering settings. For more details see Configuring ISDN Answering.

10. After configuring all settings there are 2 options:

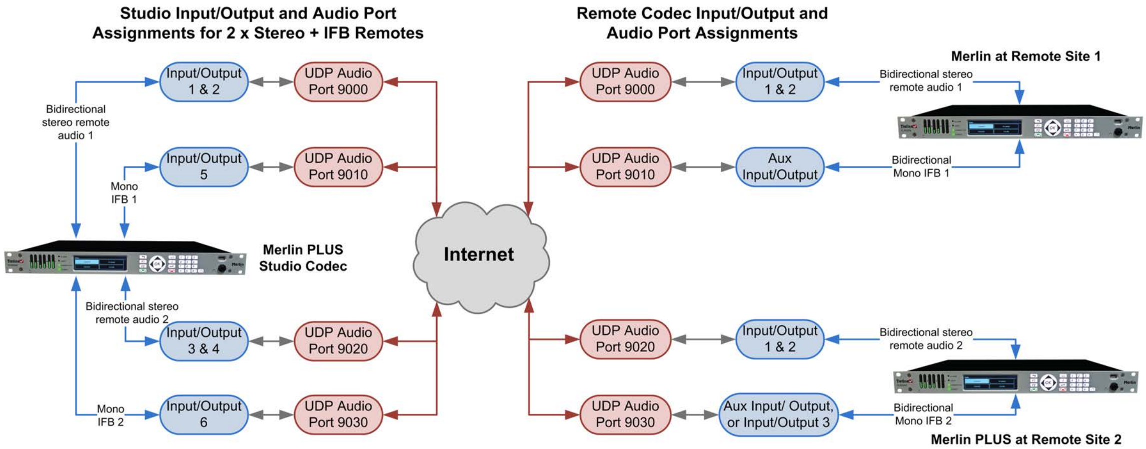
i. If you want to create another answering connection, select the check-box for **Create another answering connection** and continue through the wizard.
ii. Click **Save Program** to save all program settings.

The newly created program will be displayed in the left pane within the **Programs panel** and in the **Master panel**. This program will now allow incoming codec calls to establish **1 x Stereo Peer-to-Peer + IFB** connections and stream audio according to the input/output and port assignments indicated at the beginning of this section.

## 21.8 Configure 2 Mono or Stereo + IFB Answering Programs

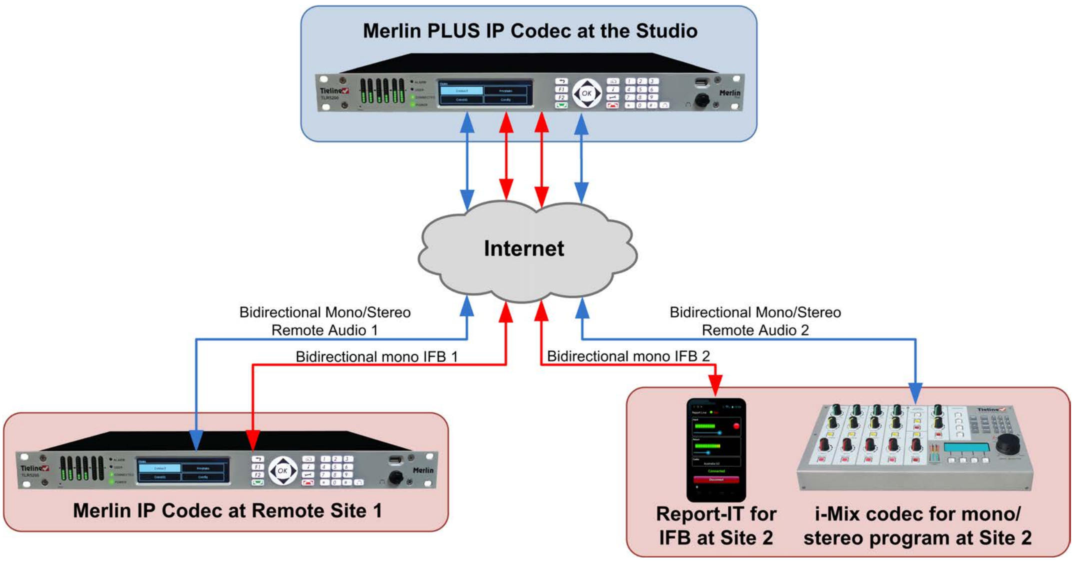
This program is designed to allow a Merlin PLUS codec at the studio answer calls from two incoming Merlin or Merlin PLUS codecs and receive:

1. Two bidirectional mono or stereo audio stream connections.
2. Two separate bidirectional mono IFB audio streams for communications.




### *Connecting other Tieline Codecs*

When a codec which only supports a mono or stereo audio stream attempts to connect (e.g. Commander G3, i-Mix-G3 or Bridge-IT codecs not supporting a separate IFB audio stream), a Merlin PLUS codec at the studio will accept the call and stream mono or stereo audio only. A second IP codec or smartphone running the Report-IT application can also be configured to connect and deliver the bidirectional mono IFB audio stream.




**Important Note: Important Note:** Remember to lock the program when connecting a **Mono or Stereo Peer-to-Peer + IFB** program using two devices at the remote site. This will avoid the first mono or stereo call unloading the **Peer-to-Peer + IFB** program at the studio and loading a mono or stereo peer-to-peer program, which would cause the second connection to fail.

## Configuring 2 x Mono or Stereo Peer-to-Peer + IFB Programs

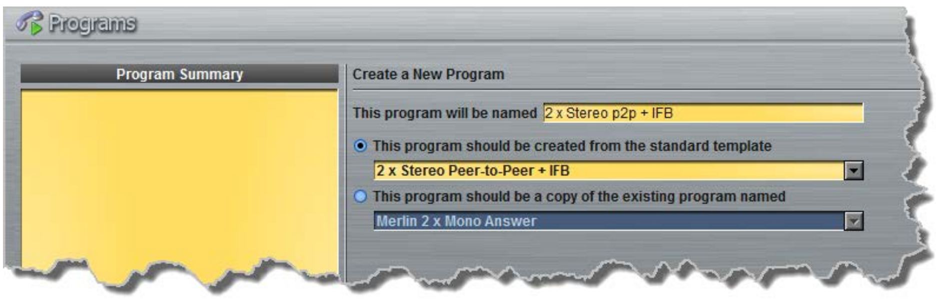
In most situations the studio codec will answer incoming audio stream connections from 2 remote sites. As a result, the following procedure explains configuration of the answering connections required at studio codec. The following configuration will always route audio from each audio stream connection to separate audio outputs.

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

- The auxiliary input is used by default for the IFB communications channel in Merlin codecs.
- 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 your Merlin PLUS codec at the studio will receive both mono and stereo peer-to-peer + IFB calls from different remote sites at different times, we recommend you configure and load a **2 x Stereo Peer-to-Peer + IFB** answering program and lock this in the codec at the studio. This will accept both mono and stereo audio stream connections. If a codec with a **Mono Peer-to-Peer + IFB** program calls the studio, the incoming mono stream will be mixed to both the left and right outputs at the studio.
- Some drop-down menus and settings may be greyed out intentionally depending on features available and the transport selected (e.g. IP or ISDN).
- It is possible to save a program at several points throughout the program wizard and use default settings to save configuration time.
- To learn more about programs see the section titled About Program Dialing.

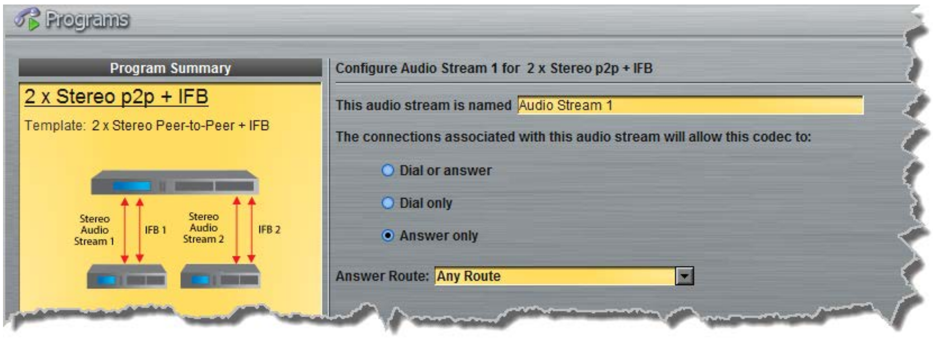
1. Open the web-GUI and click the **Programs** symbol at the top of the screen to display the **Programs panel**.

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




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

3. Enter a name for the first stereo remote **Audio Stream** and select **Answer only**. Then click **Next**. Note: If the codec you are configuring needs to dial connections as well as answer, you can select **Dial or answer** and configure both dialing and answering connections. For the purposes of this example dialing is excluded in configuring 2 stereo and 2 IFB audio streams.




ISDN

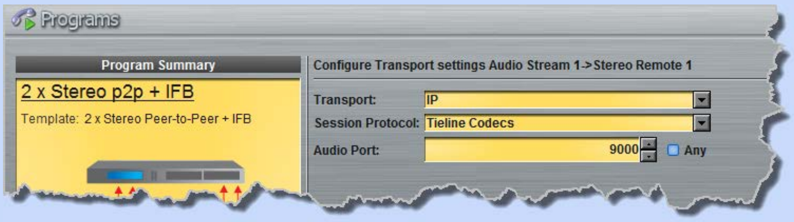
It is also possible to select an **Answer Route** if required. This is useful when routing multiple audio streams over transports like ISDN and is not recommended for use over IP. See Configuring ISDN Answering for more information. Use the default settings for IP connections.

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


5. Configure the transport settings:

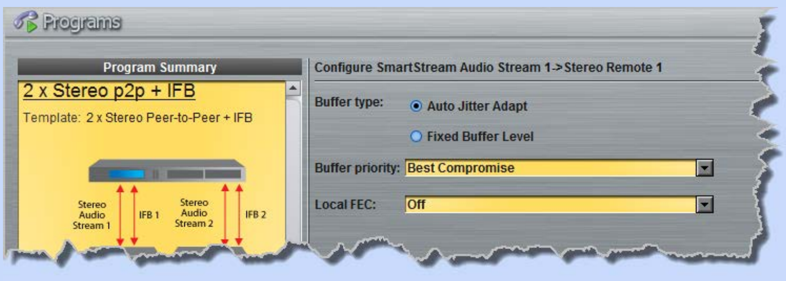
IP

For IP click the drop-down **Session Protocol** menu and select **Tieline Codecs** and ensure the **Any** check-box is not selected, then click **Next**.




Click to configure:

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

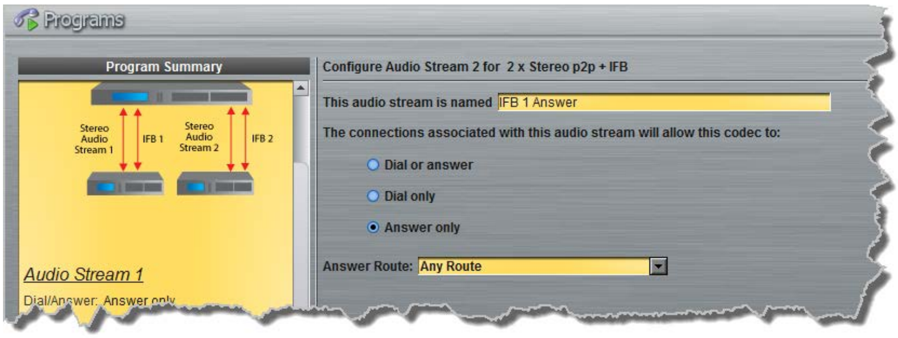


ISDN For ISDN settings are determined by ISDN module answering settings. For more details see Configuring ISDN Answering.

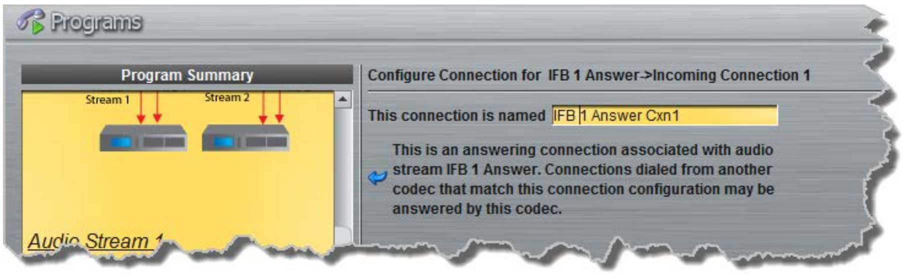
6. After configuring all settings there are 2 options:

   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 **Next Stream** to configure the first IFB audio stream.

7. Enter a name for the first IFB **Audio Stream** and select **Answer only**, then click **Next**.

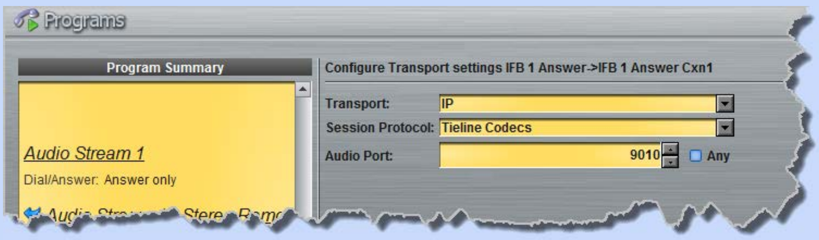



8. Enter the name of the IFB audio stream connection in the text box and click **Next.**



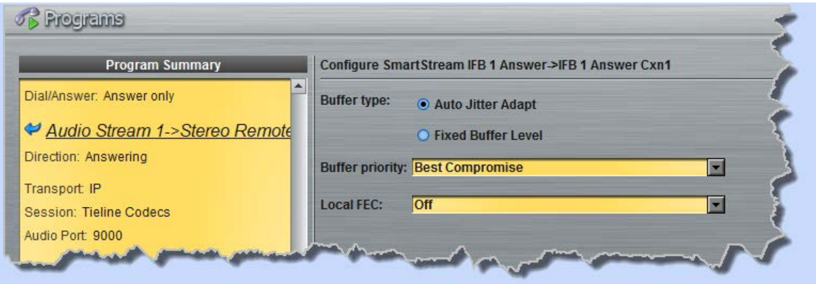

9. Configure the IFB transport settings:

IP For IP click the drop-down **Session Protocol** menu and select **Tieline Codecs** and ensure the **Any** check-box is not selected, then click **Next**.




Click to configure:

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

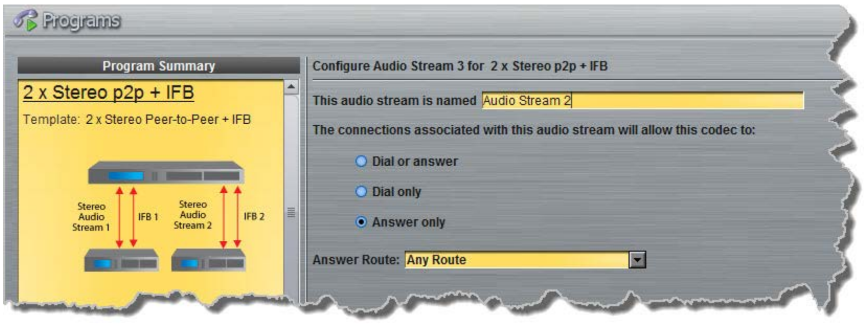



ISDN For ISDN settings are determined by ISDN module answering settings. For more details see Configuring ISDN Answering.

10.After configuring all settings there are 2 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 **Next Stream** to configure the second stereo audio stream and IFB stream.

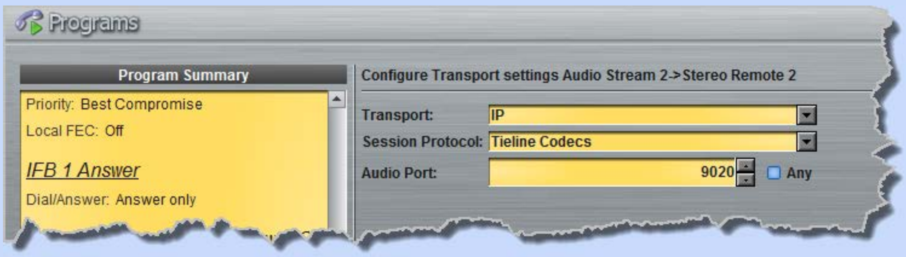
11.Enter a name for the second remote **Audio Stream** and select **Answer only**. Then click **Next**.




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

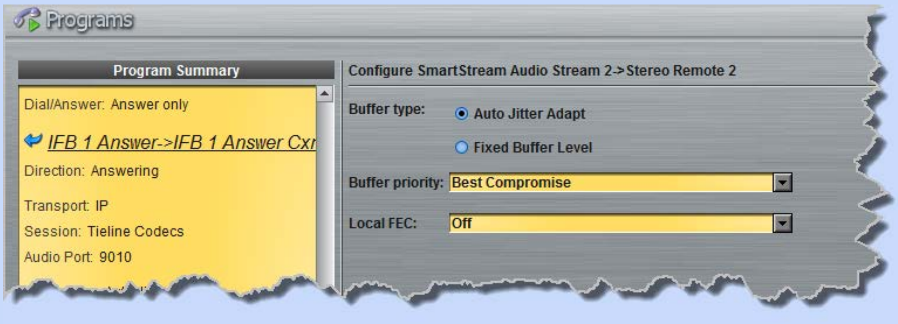

13.Configure the transport settings:

IP For IP click the drop-down **Session Protocol** menu and select **Tieline Codecs** and ensure the **Any** check-box is not selected, then click **Next**.




Click to configure:

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

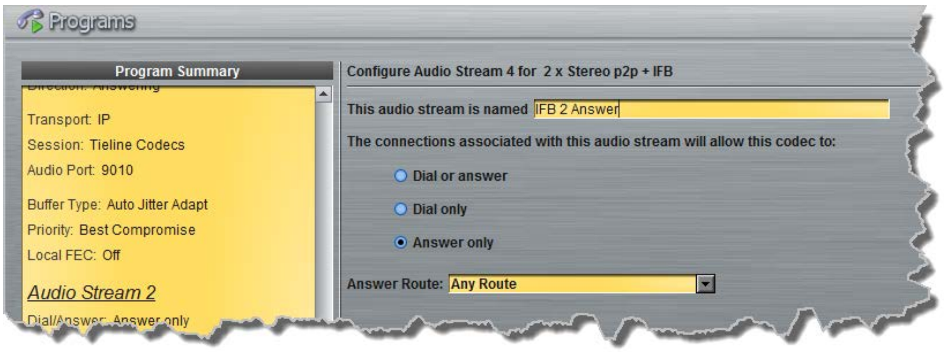



ISDN For ISDN, settings are determined by ISDN module answering settings. For more details see Configuring ISDN Answering.

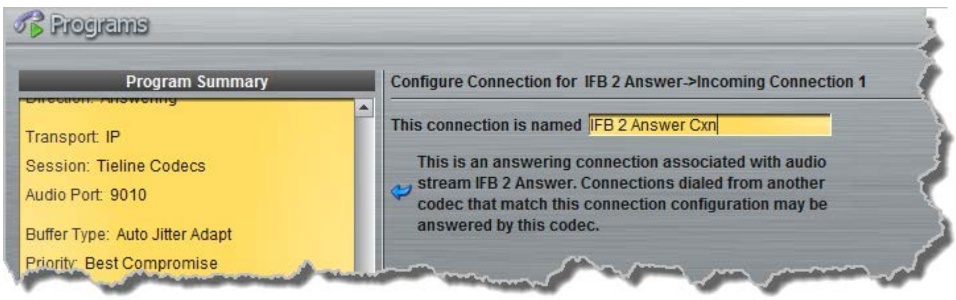
14.After configuring all settings there are 2 options:

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 **Next Stream** to configure the second IFB audio stream.

15.Enter a name for the second IFB **Audio Stream** and select **Answer only**, then click **Next**.

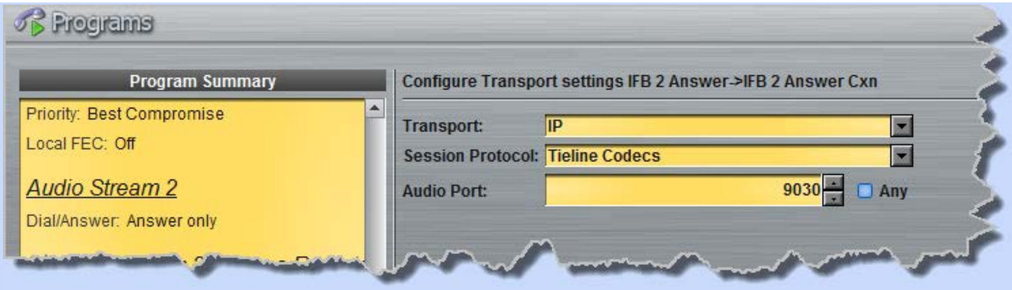



16.Enter the name of the second IFB audio stream connection in the text box and click **Next**.



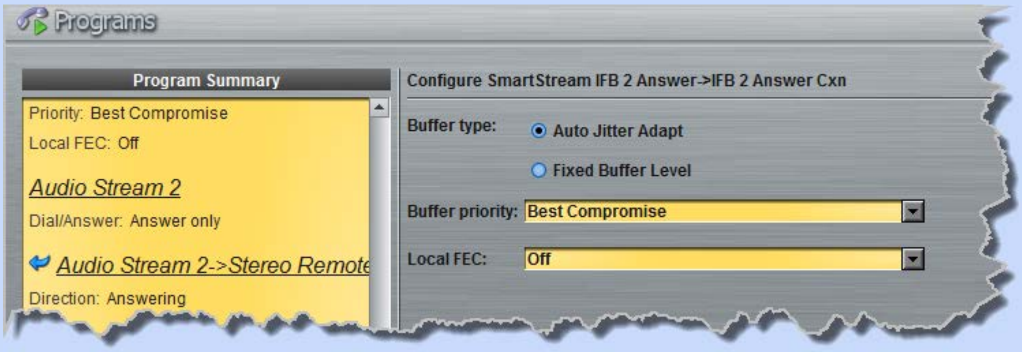

17.Configure the IFB transport settings:

IP For IP click the drop-down **Session Protocol** menu and select **Tieline Codecs** and ensure the **Any** check-box is not selected, then click **Next**.




Click to configure:

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




ISDN For ISDN, settings are determined by ISDN module answering settings. For more details see Configuring ISDN Answering.

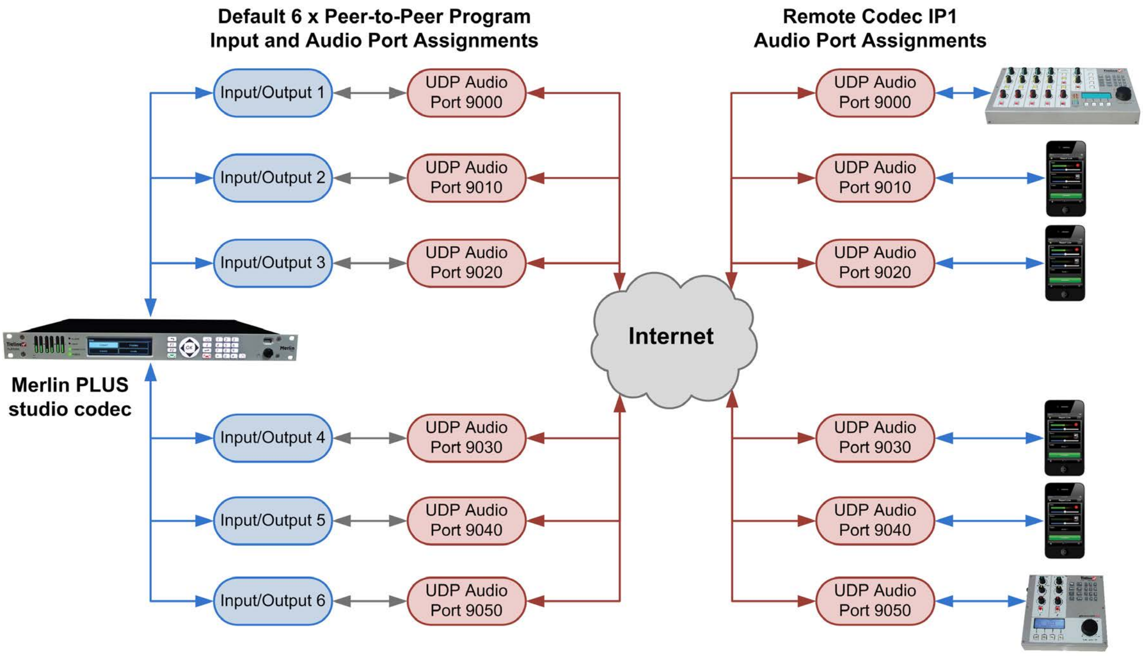
18. After configuring all settings there are 2 options:

   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 all program settings.

The newly created program will be displayed in the left pane within the **Programs panel** and in the **Master panel**. This program will now allow incoming codec calls to establish **2 x Stereo Peer-to-Peer + IFB** connections and stream audio as per the input/output and port assignments indicated at the beginning of this section.

## 21.9 Configuring 6 x Mono Peer-to-Peer Answering Programs

It is possible for a Merlin PLUS codec at the studio to create six simultaneous bidirectional mono peer-to-peer audio stream connections with different remote codecs or smartphones using the Report-IT app. The following image displays the default audio port settings used when creating a 6 x Mono Peer-to-Peer Program. Remote codecs configured with corresponding audio port settings will route audio to specific inputs/outputs of the Merlin PLUS codec as shown.




Select the **6 x Mono Peer-to-Peer** option from the drop-down menu in the first screen of the program wizard to configure a program allowing up to six mono audio stream connections with different IP codecs or smartphones using Report-IT.

In nearly all situations the studio codec will receive all calls. As a result, the following program wizard procedure displays the various configuration screens to configure an answering connection for each incoming call from up to 6 different devices.

Both mono and stereo audio stream connections can be answered by a Merlin PLUS codec when using a **6 x Mono Peer-to-Peer** program. To do this load and lock a **6 x Mono Peer-to-Peer** answering program in a Merlin PLUS codec at the studio. If a codec using a stereo program attempts to connect then the studio codec will accept the call and mix the two incoming audio channels into a single audio channel to allow a mono peer-to-peer connection.

See Configuring Merlin Point-to-Point Programs for more details about individual settings within the program wizard.

### *Routing 6 Incoming Mono Audio Streams to Specific Codec Outputs*

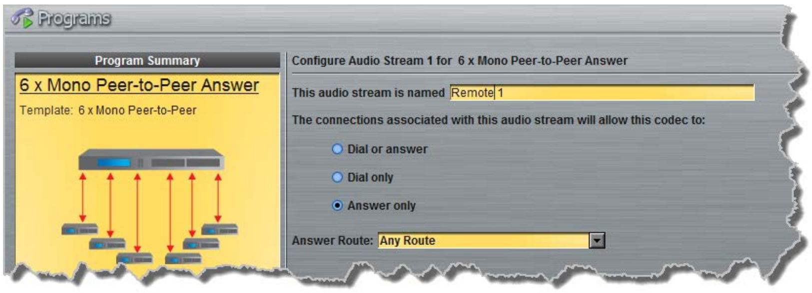
**Important Notes:** Before you start program configuration please note:
- You cannot edit a program when it is currently loaded in the codec.
- Some drop-down menus and settings may be greyed out intentionally depending on features available and the transport selected (e.g. IP or ISDN).
- It is possible to save a program at several points throughout the program wizard and use default settings to save configuration time.
- 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.
- To learn more about programs see the section titled About Program Dialing.
- 6 audio streams must be configured.

If your intention is to ensure 6 incoming mono peer-to-peer audio stream connections are always routed to the same outputs, configure a new answering program as follows:

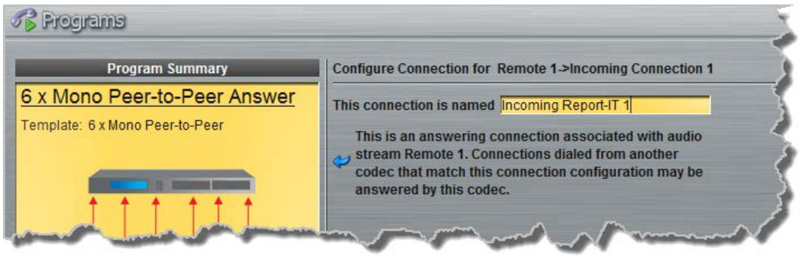
1. Open the web-GUI and click the **Programs** symbol at the top of the screen to display the **Programs panel**.
2. Click the **New Program** button to open the wizard and:
   - Click in the text box to name the new program.
   - Select **6 x Mono Peer-to-Peer,** or if you want to use an existing program as a template, select this option. Then click **Next**.
3. Enter a name for the first **Audio Stream** and select **Answer only**. Then click **Next**.

Note: if the codec you are configuring needs to dial connections as well as answer, you can select Dial or answer and configure both dialing and answering connections. For the purposes of this example dialing is excluded.



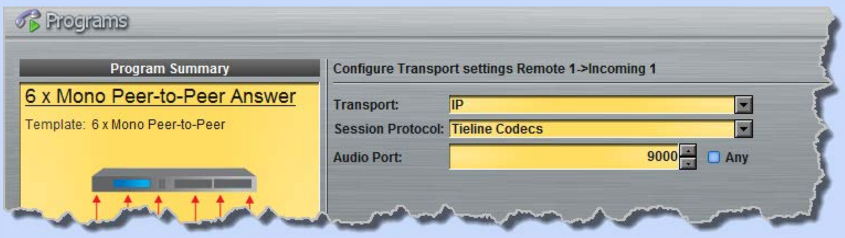

ISDN It is also possible to select an **Answer Route** if required. This is useful when routing multiple audio streams over transports like ISDN and is not recommended for use over IP. See Configuring ISDN Answering for more information. Use the default settings for IP connections.

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



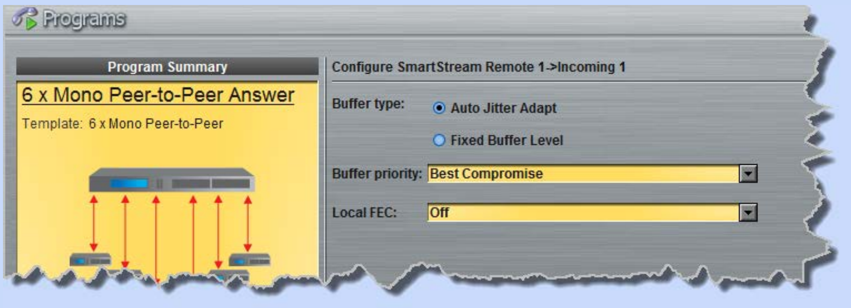

5. Configure the transport settings:

IP For IP click the drop-down **Session Protocol** menu and select **Tieline Codecs** and ensure the **Any** check-box is not selected, then click **Next**.



**Note:** Deselecting the **Any** check-box ensures that any codec dialing and using the audio port specified will always be routed to codec output 1. By default port 9000 is the port specified for the first IP audio stream connection made by any Tieline codec; port 9010 is used by default for a second IP connection. Merlin PLUS default ports for mono connections 3 to 6 are 9020, 9030, 9040 and 9050. If you need to change this setting on your dialing codec, or in your Report-IT smart phone application, this will need to be adjusted accordingly in the answering codec.

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

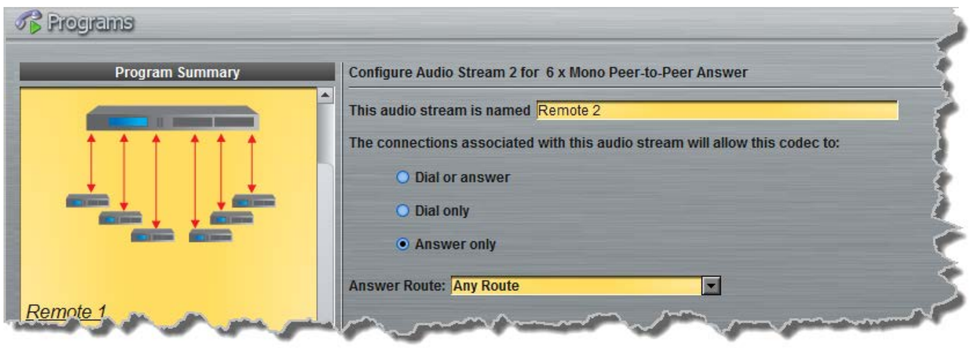



ISDN For ISDN settings are determined by ISDN module answering settings. For more details see Configuring ISDN Answering.

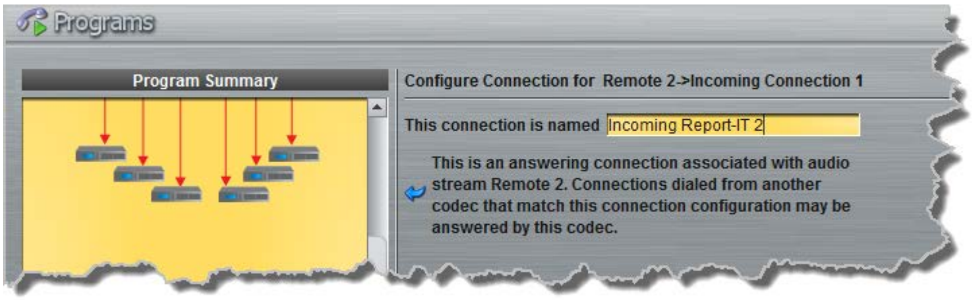
6. After configuring all settings there are 2 options:

   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 all program settings.

7. Enter a name for the second **Audio Stream** and select **Answer only**. Then click **Next**.

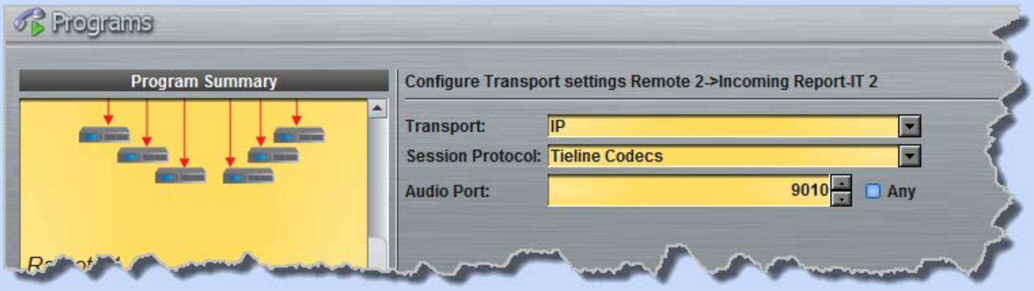



8. Enter the name of the audio stream connection in the text box and click **Next**.




9. Configure the IFB transport settings:

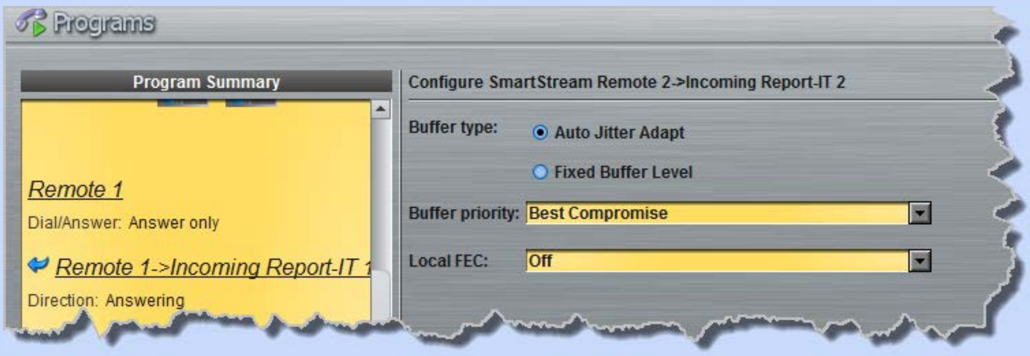
For IP click the drop-down **Session Protocol** menu and select **Tieline Codecs** and ensure the **Any** check-box is not selected, then click **Next**.




Port 9010 is used by default for a second IP connection. Merlin PLUS default ports for mono connections 3 to 6 are 9020, 9030, 9040 and 9050. If you need to change this setting on your dialing codec, or in your Report-IT smart phone application, this will need to be adjusted accordingly in the answering codec.

Click to configure:

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



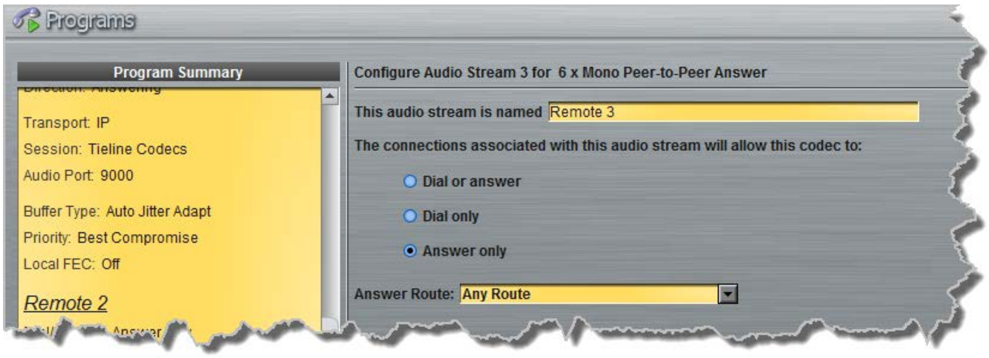

ISDN For ISDN settings are determined by ISDN module answering settings. For more details see Configuring ISDN Answering.

10. After configuring all settings there are 2 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 all program settings.

11. Enter a name for the next **Audio Stream** and select **Answer only**. Then click **Next**.




12. Enter the name of the audio stream connection in the text box and click **Next**.


13. Continue through the program wizard and configure all 6 connections, then click **Save Program** to save program settings.

**Important Note:** The default Merlin PLUS answering program uses audio port 9020 for a third audio stream IP connection. Merlin PLUS default ports for mono connections 3 to 6 are 9020, 9030, 9040 and 9050. If you need to change this setting on your dialing codec, or in your Report-IT smart phone application, this will need to be adjusted accordingly in the answering codec.

14. The newly created program will be displayed in the left pane within the **Programs panel** and in the **Master panel**.

## 21.10 Configuring Multicast Client 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

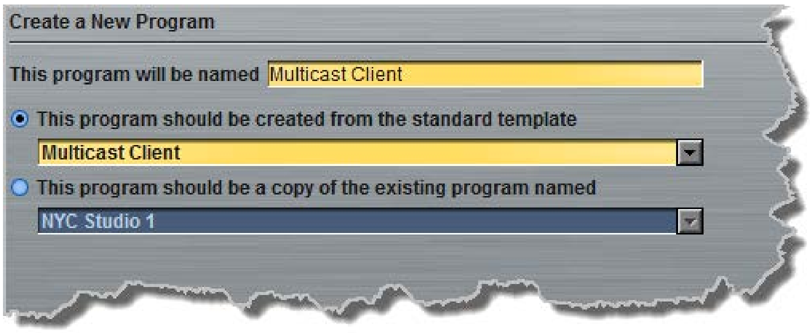
represents the audio being downloaded from the network locally. Multicast server codecs do not display LQ readings.

- The default UDP audio port setting is 9000 for the first multicast, 9010 for the second multicast and 9020 for the third multicast. The client and server port settings must match to receive an audio stream. E.g. if a client codec wishes to receive multicast audio stream 2 then it must use audio port 9010.
- Forward Error Correction (FEC) is not available for multicast connections.
- It is not possible to send auxiliary data using multicast connections.
- It is not possible to connect to a G3 codec and receive multicast IP audio streams.
- To copy multicast client programs onto multiple codecs see Save and Restore Configuration Files.
- If the codec is answering more than one mono or stereo multicast connection it is necessary to create an answering program to suit the answering configuration and lock this program in the codec.

## Configuring Multicast Client Programs

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

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

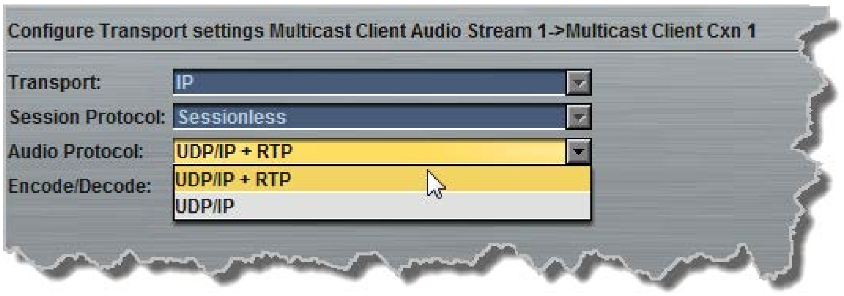



**Important Notes:** When you decide to use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

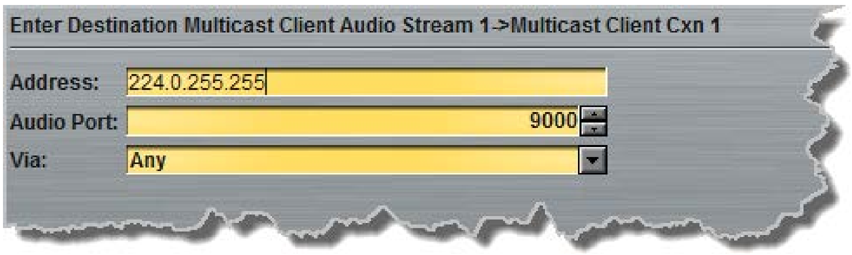
3. Enter a name for the **Audio Stream**, then click **Next**.


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

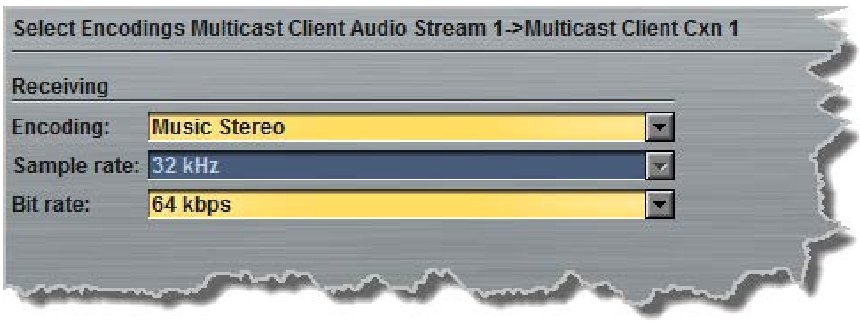

5. Follow the instructions on the right-hand side of the panel to configure the transport settings for the connection, then click **Next**. Note: select **UDP/IP +RTP** for RFC compliant streaming.



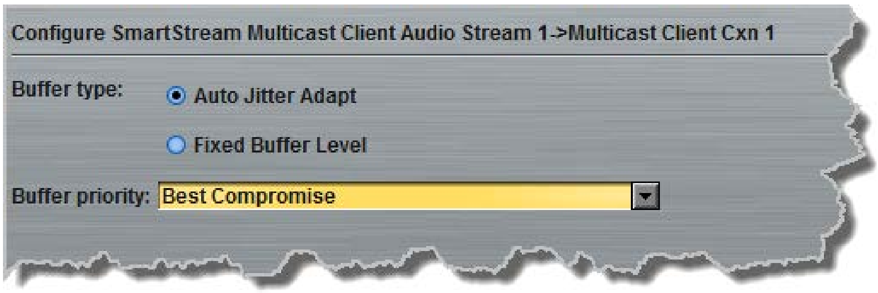

6. Configure the multicast IP address and audio port (the same multicast address and port must be used for both the server and client programs), then specify which IP streaming interface is used to dial this connection, e.g. **Primary** (port **ETH1**) or **Secondary** (port **ETH2**), then click **Next**. Note: By default **Any** will select **ETH1** if it is available and **ETH2** if it is unavailable.




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

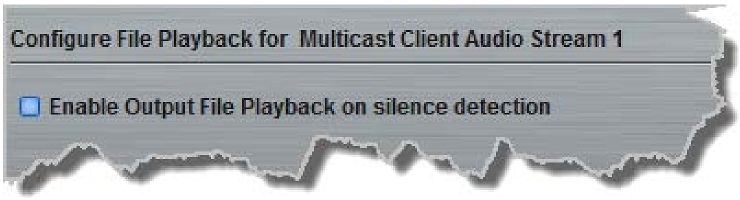



8. Click to configure:
   - **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer Priority**, or
   - **Fixed Buffer Level** and enter the **Jitter Depth**, which must be between 12ms and 5000ms depending on the algorithm you select.




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

9. Select the **Enable File Playback on silence detection** check-box to configure the codec to play back audio from a file via a drive attached to the USB port.



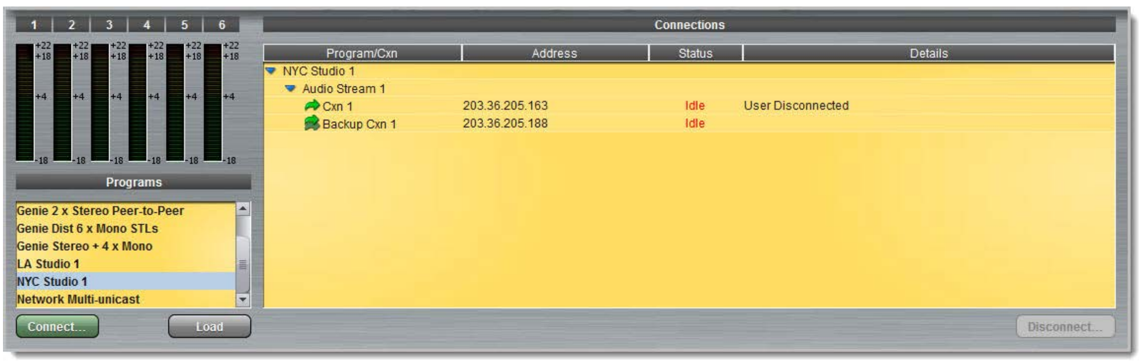

10. Click **Save Program** to complete configuration of the program.

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

## 21.11 Dial and Disconnect a Program

### *Connecting a Program*

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




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

### *Disconnecting a Program*

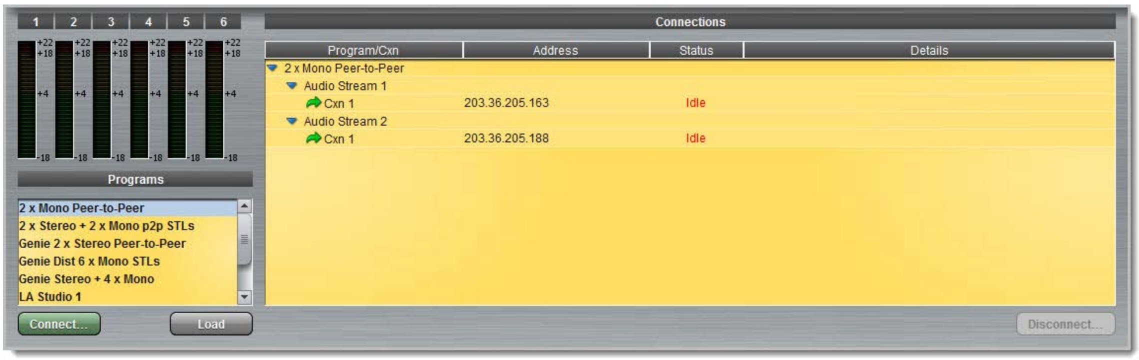
1. Click to highlight the audio stream in the **Connections** pane.


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

## 21.12 Dial/Disconnect Multiple Audio Streams

### *Load and Connect Multiple Audio Streams within a Program*

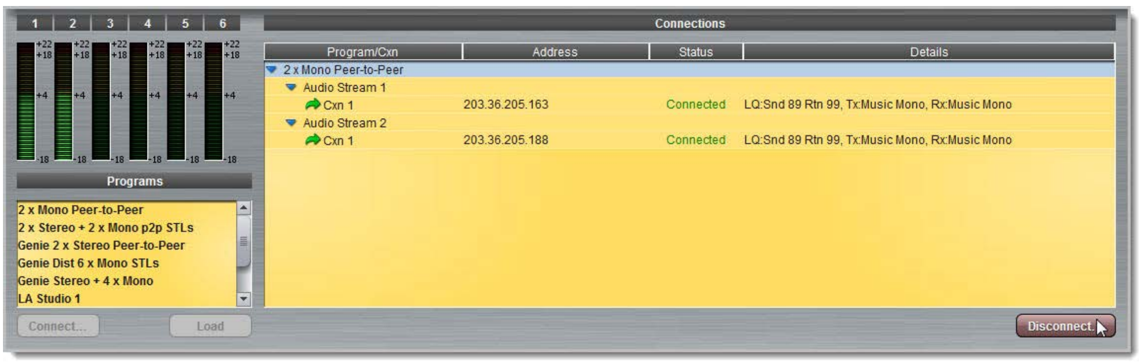
1. Click to select the program you want to load from the **Programs** list.




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

### *Disconnect All Audio Stream Connections*

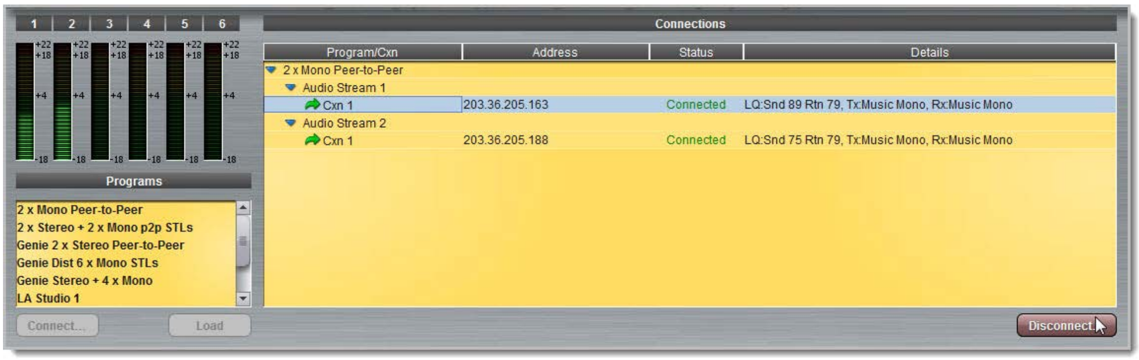
1. Click to select the program in the **Connections** pane, e.g. **2 x Mono Peer-to-Peer** in the following example.



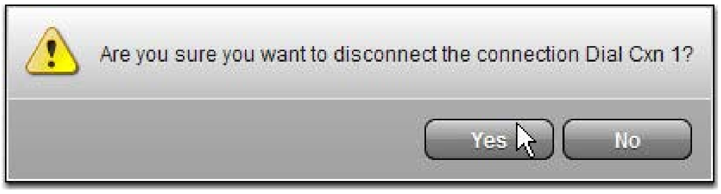
2. Click the **Disconnect** button.
3. Click **Yes** in the confirmation dialog to disconnect all audio stream connections.


### *Disconnect a Single Audio Stream Connection*

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




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



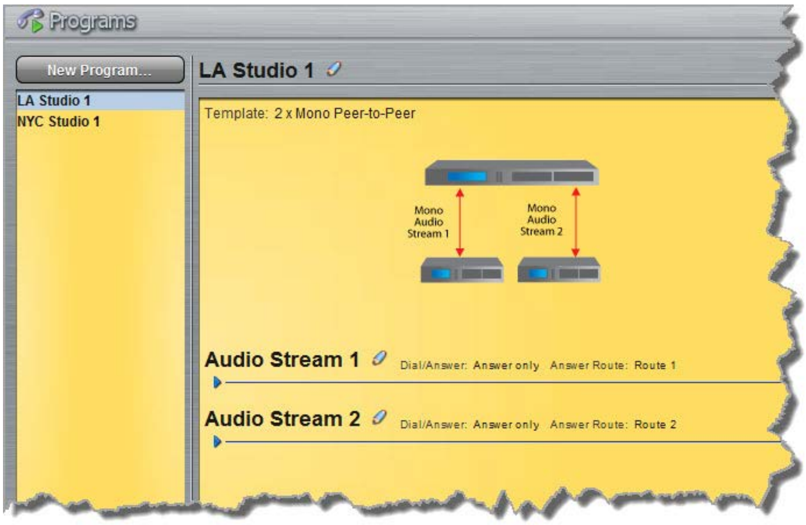

## 21.13 View/Edit/Delete Programs

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

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

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

2. Click to select a program in the left-hand pane.




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

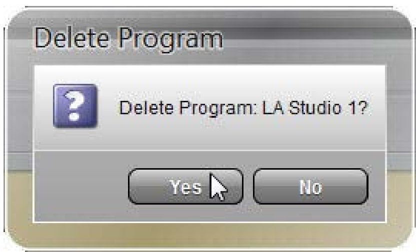
4. The program wizard will open at the relevant point to facilitate editing of connection parameters. Click **Save Program** to store settings.

## *Deleting Programs*

There are two ways to delete a program.

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


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




5. Alternatively, click **Delete Program** next to the program name.


## 21.14 Configuring SIP Settings

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

### About SIP

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

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

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

**Important Notes:**
- Each codec should be registered to a different SIP server account to avoid connection conflicts.
- SIP dialing is only supported over point-to-point connections, not multi-unicast connections.
- Tieline G3 codecs do not support connections using AAC and will default to MPEG Layer 2 if an incoming call is programmed to use this algorithm.

### SIP Server Connections: Getting Started

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

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

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

- Realm details (sometimes).

## *Program the Codec for SIP using the Web-GUI*

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

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


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


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

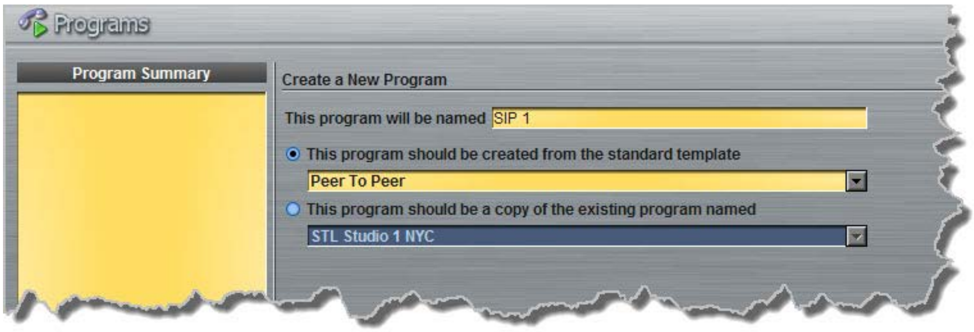
# 21.15 Configuring SIP Programs

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

**Important Notes:** Before you start program configuration please note:
- SIP can only operate using port **ETH1** on the rear panel of the codec.
- You cannot edit a program when it is currently loaded in the codec.
- Some drop-down menus and settings may be greyed out intentionally depending on features available.
- To learn more about programs see the section titled About Program Dialing.

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

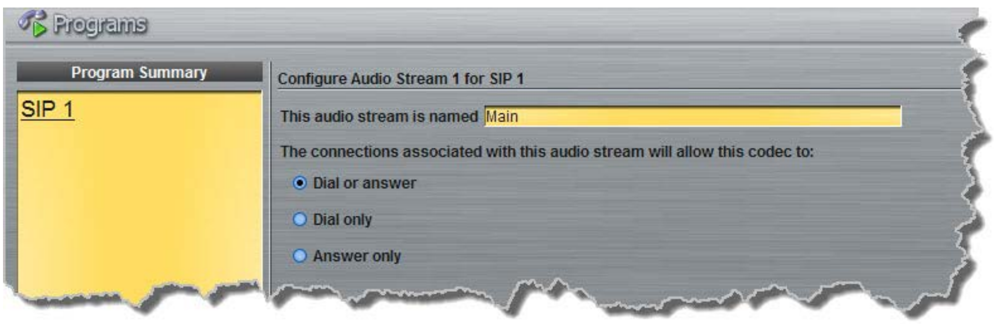
2. Click the **New Program** button to open the wizard and:
   - Click in the **Program Name** text box to name the new program.
   - Select **Peer to Peer,** or if you want to use an existing program as a template, select this option. Then click **Next**.



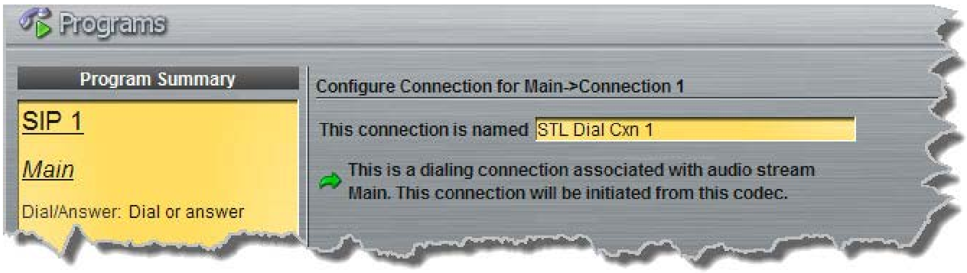

**Important Notes:** When you choose to use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

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

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

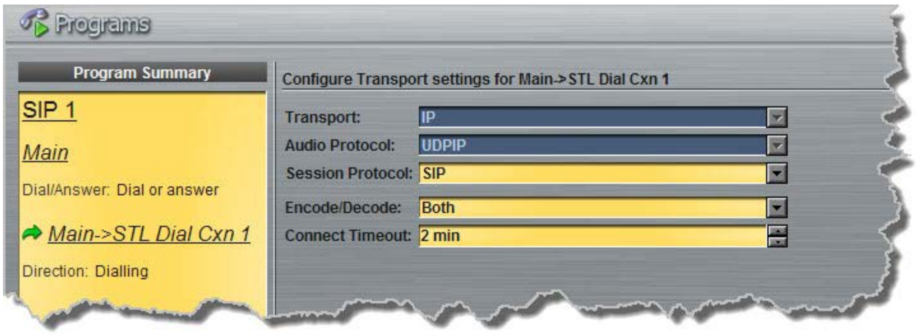



4. This audio stream connection in the wizard will allow the codec to dial. Enter the name of the connection in the text box, then click **Next**.



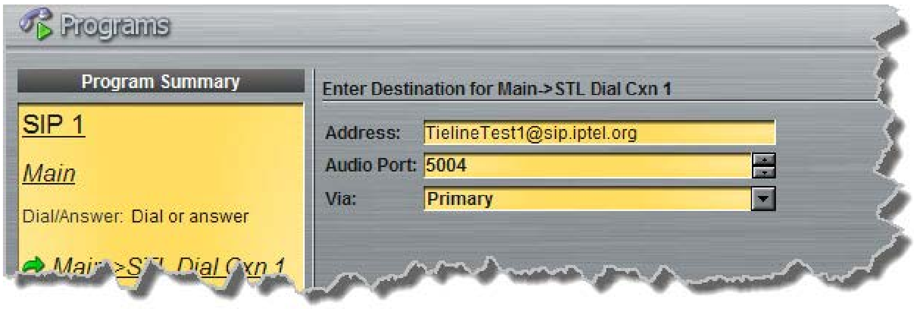

5. Follow the instructions on the right-hand side of the panel to configure the transport settings for the connection: Ensure that you select:
- **IP** as the **Transport**.
- **UDPIP** as the **Audio Protocol**.
- **SIP** from the **Session Protocol** menu option.

Then click **Next**.



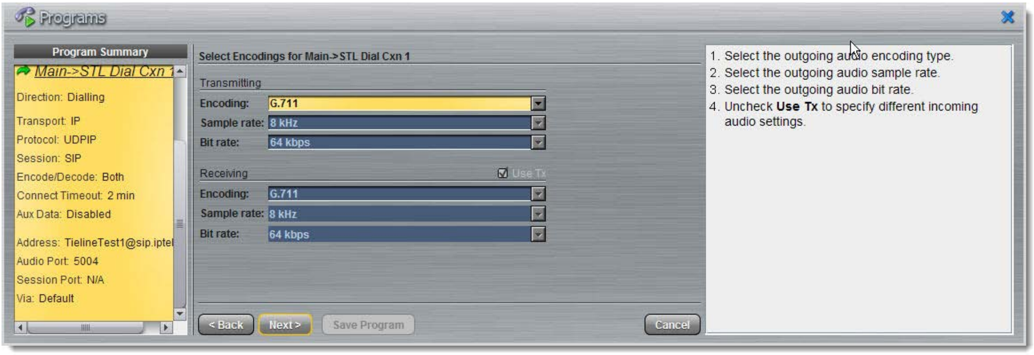

6. Configure the destination codec **Address** and **Audio Port**, then specify **ETH1** as the network interface used to dial the connection, e.g. **Primary** (Ethernet port 1).

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

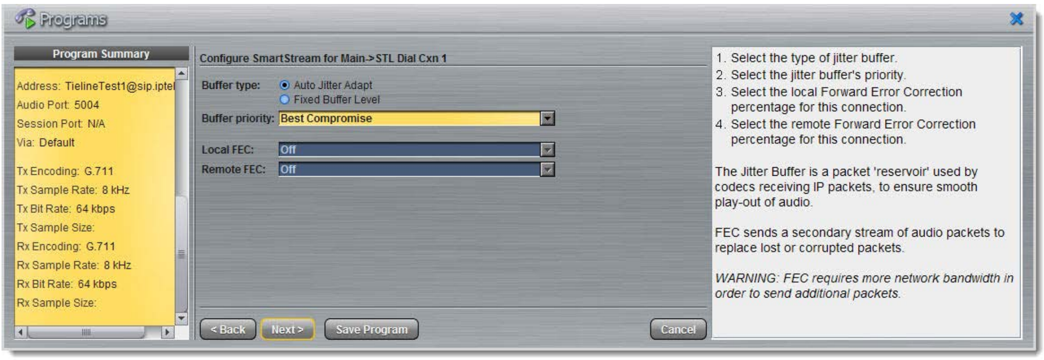



**Important Notes:**
- If your codec is registered with same SIP registrar as the destination codec then you only need to enter the SIP user name to dial successfully.
- The default UDP audio port when using SIP is 5004 in Tieline codecs. To contact a codec that is behind a firewall or NAT-enabled router,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

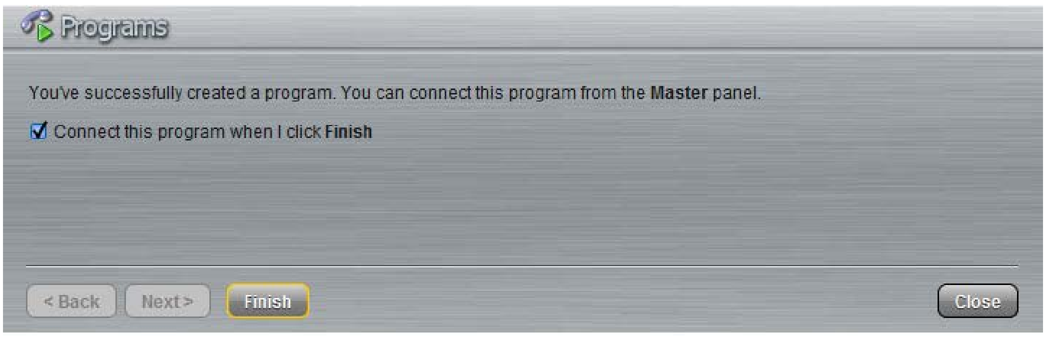
7. Click the drop-down arrows on the right-hand side of each active drop-down menu to adjust the **Encoding, Sample rate** or **Bit rate** parameters. Click Next to continue.




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




Click **Save Program** to save all settings, or click **Next** to configure **Auto Reconnect** or a backup connection using fail over (Note: any backup connection must also use the **ETH1** network port on the codec). If you click **Save Program,** select the check-box if you want to connect the program immediately, then click **Finish**.




9. The newly created program will be displayed in the **Programs panel** and in the **Master panel**. Dial the program by loading and connecting using the **Master panel**, or dial the program manually using the codec front panel.

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

## 21.16 Reset Factory Default Settings

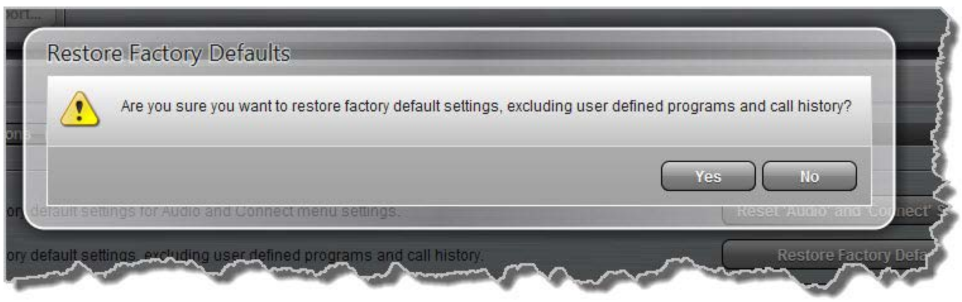
There are several options which allow you to restore factory default settings within the codec.

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

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


3. Click one of the four reset options available.


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




## 21.17 Backup and Restore Functions

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

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

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

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

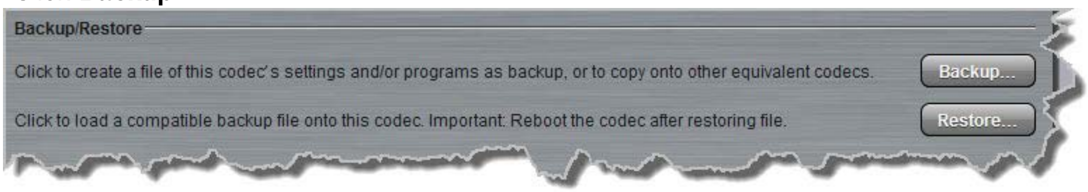
connections.

## Creating Backup Files

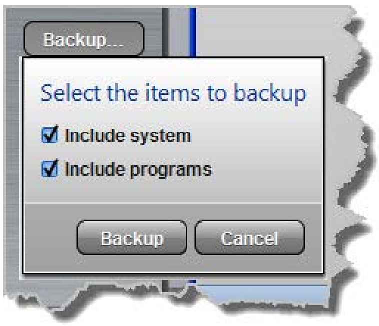
1. Open the web-GUI and click the **Settings** symbol at the top of the screen to display the **Settings panel**.

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


3. Click **Backup**.




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

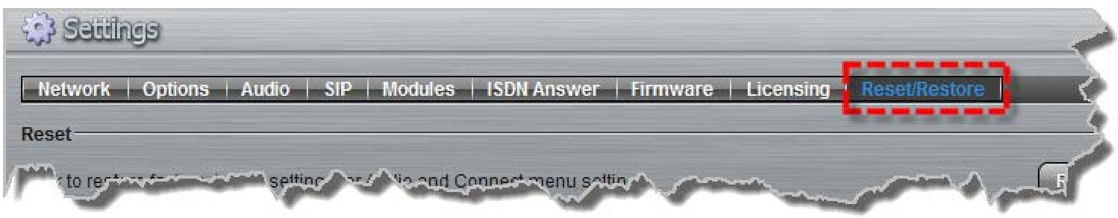



5. Click **Save** and select a location on your PC to save the configuration file.

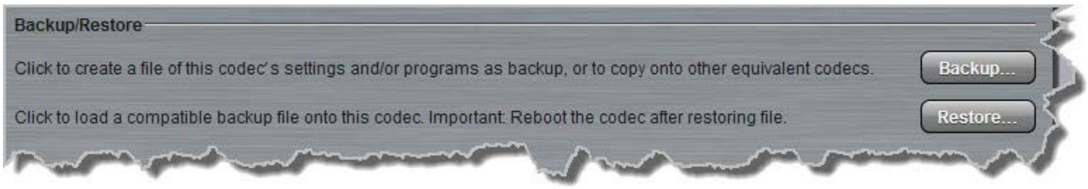
## Restoring Configuration File Settings

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

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



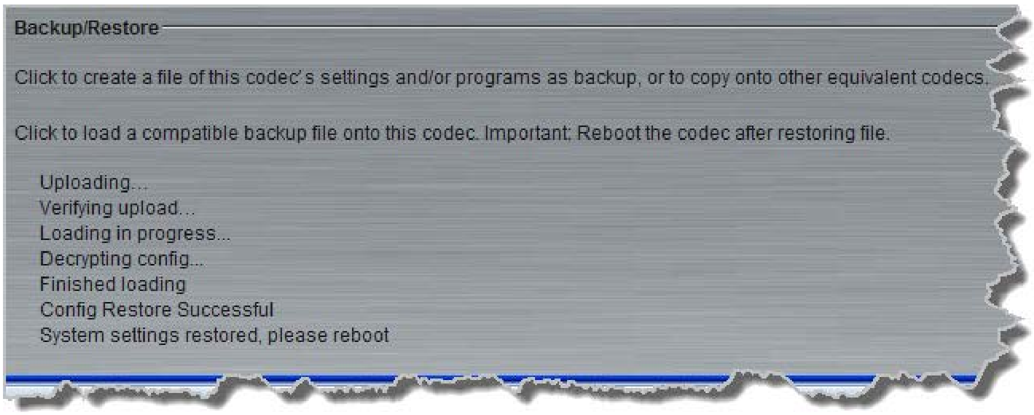

3. Click **Restore**.




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

5. Use your mouse-pointer to click and select the check boxes for restoring items. For example, you could select the **Include programs** check-box and deselect the **Include system** check-box if you are only copying programs onto codecs.


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




7. Reboot the codec to ensure the restored configuration will take effect in the codec.

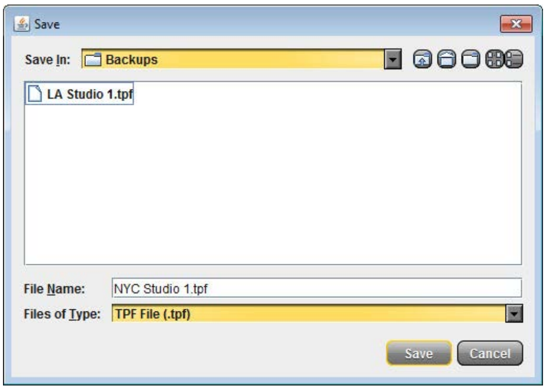
## 21.18 Import and Export Programs

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

### *Export a Program*

1. Open the web-GUI and click the **Programs**  symbol at the top of the screen to display the **Programs panel**.
2. Click the **Export** button in the bottom-left corner of the **Programs panel**.


3. Navigate to the file folder in which you are saving the **.tpf** program file.

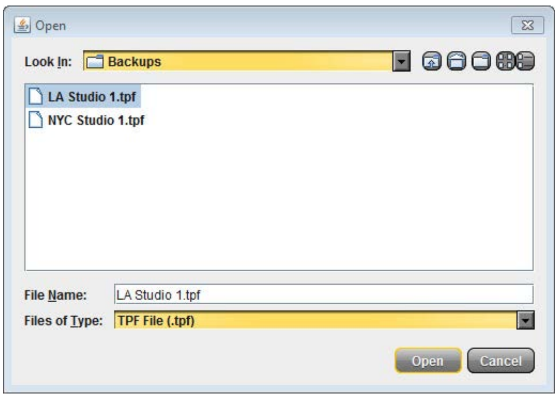



4. Click **Save** to save the program file.

## *Import a Program*

1. Open the web-GUI and click the **Programs**  symbol at the top of the screen to display the **Programs panel**.
2. Click the **Import** button in the bottom-left corner of the **Programs panel**.


3. Navigate to the file folder containing the **.tpf** program file you want to import.
4. Click to select the file and click **Open** to import it.



## 21.19 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 your routing requirements 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 web-GUI and click the **Settings** symbol at the top of the screen to display the **Settings panel**.

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

3. Click the **Lock Loaded User Program** check-box to lock or unlock a user program in the codec.


**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.

## 21.20 Configuring IP Packet QoS

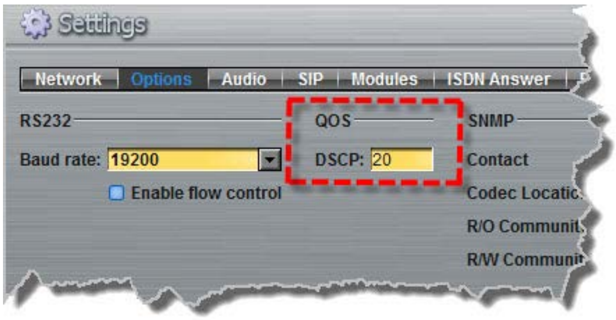
The codec can be programmed to tag IP data packets sent across a network by entering a value into the Differentiated Services Code Point (DSCP) field within the header of data packets transmitted over the network.

### *Configuring QoS*

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

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

3. Click in the **QoS** text box and enter the new value.




4. Click the **Save Settings** button 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.

## 21.21 Configuring SNMP in the Codec

The codec supports Simple Network Management Protocol (SNMP ) for managing devices on IP networks. There are two elements to configuring SNMP in your codec:

1. Configure SNMP Device settings in your codec.
2. Configure SNMP Traps via the **Alarms Panel** in the web-GUI (see SNMP Trap Configuration in Configuring Alarms, or to configure using the codec front panel see Configuring SNMP Settings).

### *Description of SNMP Settings in the Codec*

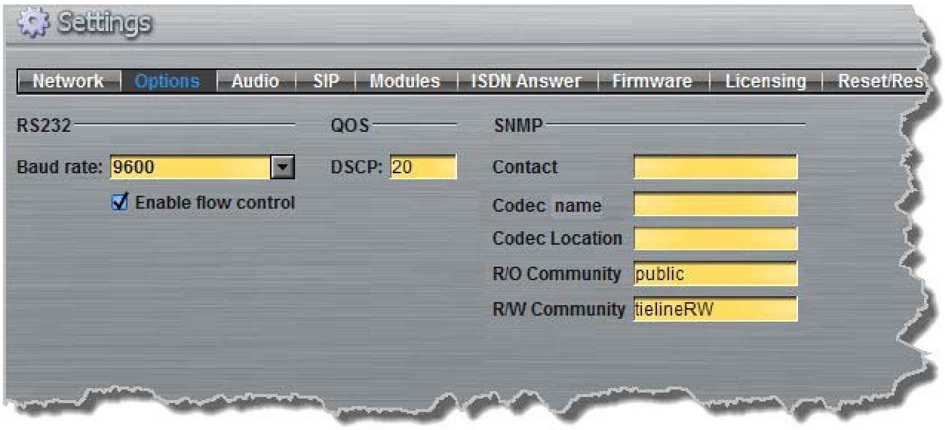
| Features | Operation Button Descriptions |
|---|---|
| **Codec Name** | A user-specified alphanumeric identifier which may be used by third-party SNMP software to identify a device. The device name corresponds to the ".iso.org.dod.internet.mgmt.mib-2.system.sysName" SNMP attribute and is completely 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 web-GUI and click the **Settings**  symbol at the top of the screen to display the **Settings panel**.

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

3. Click in the text boxes to enter SNMP configuration settings.




4. Click the **Save Settings** button to save the new setting.

### *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 links in a PC web browser connected to the same network as your codec:

http://<CODEC_ADDRESS>/mibs/TIELINE-SMI-MIB.mib
http://<CODEC_ADDRESS>/mibs/TIELINE-ALARM-MGT-MIB.mib
http://<CODEC_ADDRESS>/mibs/TIELINE-AUDIO-INPUTS-MIB.txt
http://<CODEC_ADDRESS>/mibs/TIELINE-CONNECTION-TABLE-MIB.txt
http://<CODEC_ADDRESS>/mibs/TIELINE-DECODER-TABLE-MIB.txt
http://<CODEC_ADDRESS>/mibs/TIELINE-ENCODER-TABLE-MIB.txt
http://<CODEC_ADDRESS>/mibs/TIELINE-PRODUCTS-MIB.mib
http://<CODEC_ADDRESS>/mibs/TIELINE-SMI-MIB.mib
http://<CODEC_ADDRESS>/mibs/TIELINE-SYSINFO-MIB.txt
http://<CODEC_ADDRESS>/mibs/TIELINE-TC-MIB.mib

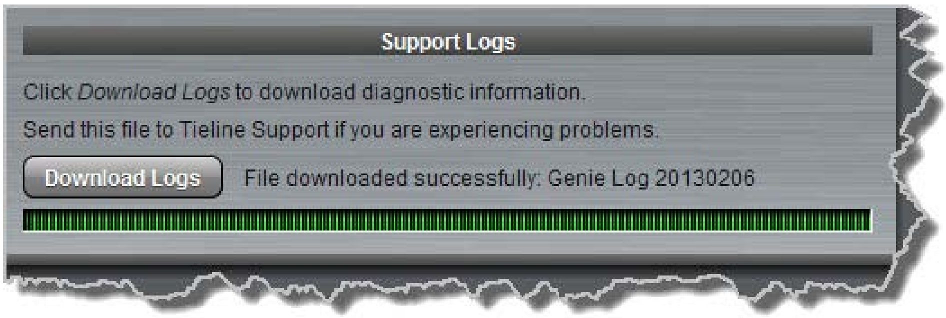
**Important Note:** The codec supports the attributes specified in the MIB-II standard. Please verify that your SNMP software contains the required files as specified in RFC 1213.

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



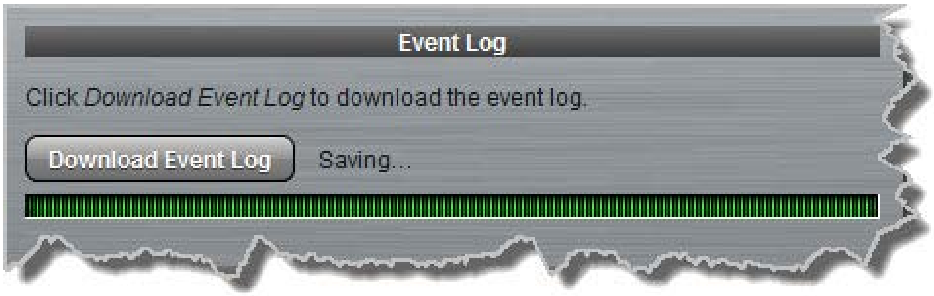

3. Save the file to your computer and then send it as a .zip file to Tieline support via support@tieline.com

### *Download Event Logs*

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

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

2. Click **Download Event Log**.



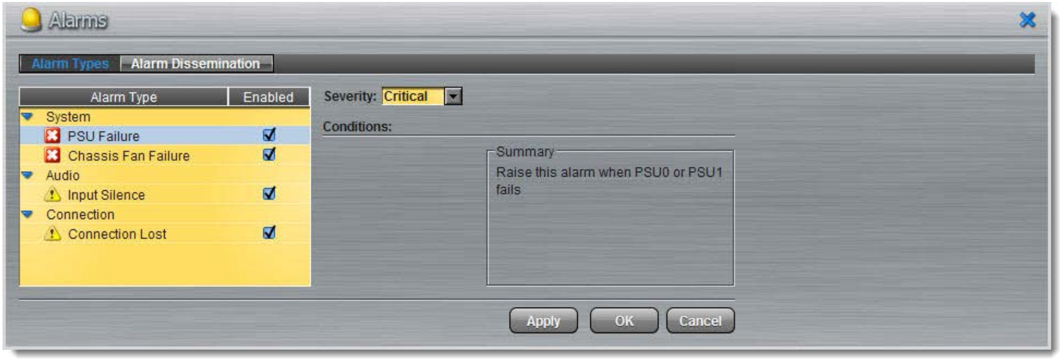

3. Save the file to your computer and then


## 21.23 Configuring Alarms

Click the **Alarm symbol** at the top of the web-GUI to view and configure a range of alarms, which can provide alerts as required.

### *Alarm Types*

Click **Alarm Types** to display the alarm overview pane within the **Alarms Panel**.



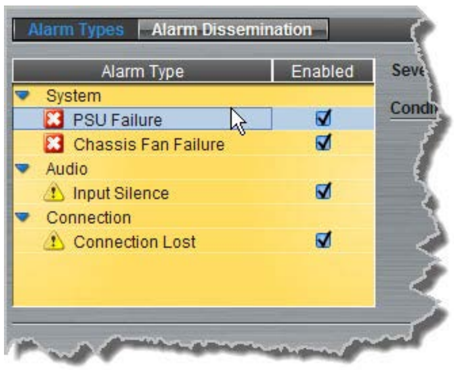

**System**, **Audio** and **Connection** alarms are available, including:

- Power Supply Failure: Enabling the **PSU Failure** alarm will configure the codec to send alerts if one or both PSUs fail.
- Chassis Fan Failure: A **Chassis Fan Failure** alarm configures the codec to send alerts if this fails.
- Input Silence Detection: An **Input Silence** detection alarm can be configured to deliver alerts if input audio is lost.
- Connection failure: A **Connection Lost** failure alarm can be configured to deliver alerts if the codec loses a connection.Note: This feature is not currently enabled.

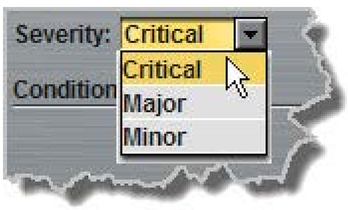
## *Configuring an Alarm's Severity Level*

Codec alarms can be rated at three different severity levels:

1. Click an alarm in the **Alarm Type** pane to highlight it.




2. Click the **Severity** drop-down menu and select the preferred severity level.




3. Perform this for each alarm you want to configure and then click **Apply** or **OK** to save settings.

## Enabling Alarms

To enable and disable alarms:

1. Click the **Enabled** check-box to toggle enabling and disabling of an alarm.

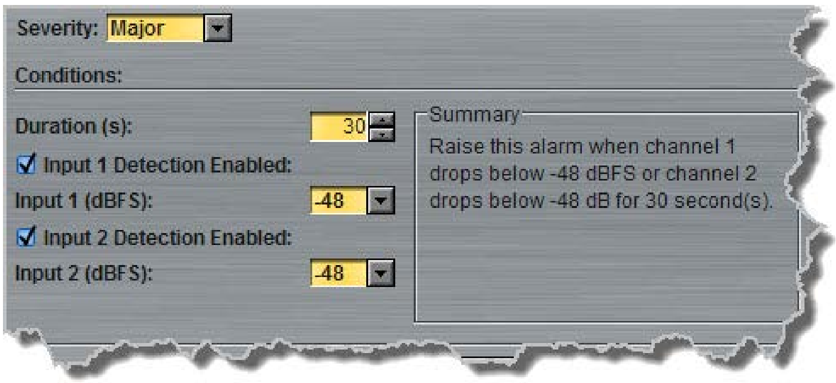

2. Click **Apply** or **OK** to save settings.

## Configuring Input Silence Detection Parameters

When configuring an **Input Silence** alarm it is also necessary to configure the audio silence thresholds and timeout duration.

1. Click **Input Silence** to highlight the alarm and ensure it is **Enabled**.


2. Configure the dBFS threshold and timeout duration in seconds within the **Conditions** pane. An alarm will be raised when these parameters are breached.




3. Click **Apply** or **OK** to save settings.

## Configuring Alarm Severity Alerts

Alerts for each alarm severity level are configured using the **Alarm Dissemination** tab.

1. Click **Alarm Dissemination**.


2. Click to highlight the **Alarm Severity** level you want to configure, then select and configure the alerts as required.

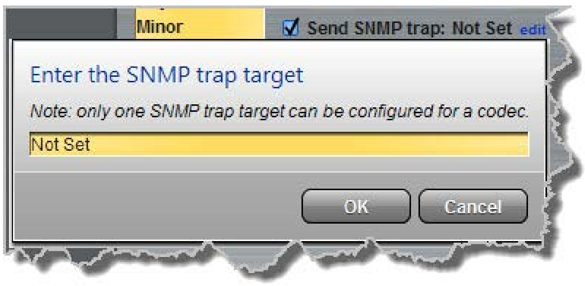

3. Click **Apply** or **OK** to save settings.

### *SNMP Trap Configuration*

Simple Network Management Protocol (SNMP) is a protocol used to manage devices on IP networks. SNMP provides the ability to send traps (notifications or alerts), which are packets containing data relating to a system component. These packets are generated by agents on a managed device and may be either statistic or status related. Please see your system administrator if you require more information.

1. Click to select the **Send SNMP trap** check-box.


2. Click **edit** to open the **Enter the SNMP trap target** dialog and enter the SNMP trap target, then click **OK**.



### 21.23.1 Managing Alarms

Active codec alarms are indicated on the web-GUI in the **Current Alarms** screen.


The user is alerted to active alarms by:

1. The **Alarm Symbol** flashing in the top right-hand corner of the Toolbox web-GUI screen.
2. All new alarms being listed in the **Current Alarms** tab within the **Alarms Panel**.
3. Other alerts as per Alarm Dissemination settings.
4. The codec front panel **ALARM LED** flashing red.

| Alarm State | Front Panel Alarm LED | Web-GUI Alarm Symbol |
|---|---|---|
| **Active** | Flashing red | Flashing |
| **Acknowledged** | Solid red | Stops flashing, remains visible |

**Important Note:** When a connection is active the front panel **CONNECTED LED** is illuminated solid green. Illumination will cease if a connection is lost.

## *Acknowledging Alarms*

To acknowledge an alarm:

1. Click to select the alarm in the **Current Alarms** tab.
2. Click **Acknowledge selected alarm**.

After acknowledging the alarm:

1. The **State** will change from **Active** to **Acknowledged**.
2. The **Alarm Symbol** will stop flashing but remain visible in the top right-hand corner of the web-GUI screen.
3. The codec front panel **ALARM LED** will stop flashing and illuminate solid red.
4. The state of other alerts may change, as per Alarm Dissemination settings.

## *Deactivating Alarms*

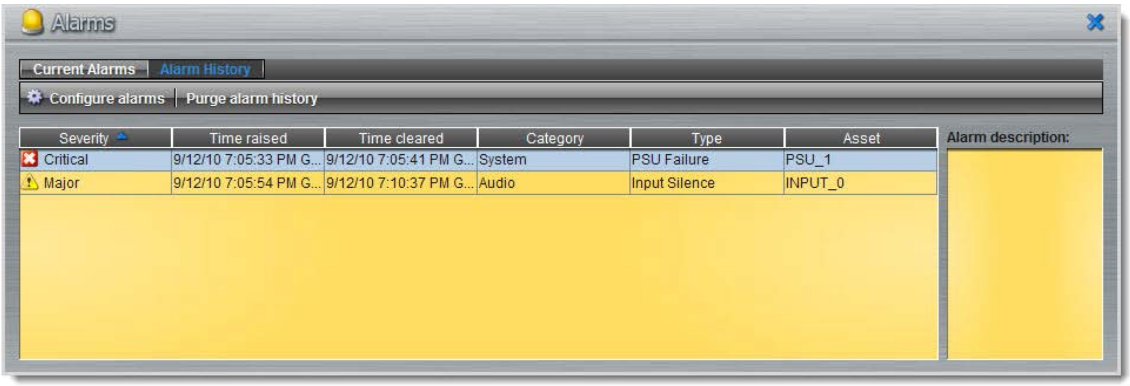
An alarm is deactivated automatically when the alarm state is reversed. E.g. if power is restored after a **PSU Failure** alarm, or if audio is restored after an **Input Silence** alarm.

### *Deactivating Input Silence Alarms*

An **Input Silence** alarm is activated when the configured audio and duration thresholds have been breached. To recover from this alarm state the codec must detect input audio 10% higher than the failure threshold. When audio at this level is detected, the codec monitors input audio to ensure it doesn't drop below the recovery threshold setting more than 5 times within the nominated **Input Silence** duration time. The alarm is then deactivated automatically.

## *Alarm History*

Click the **Alarm History** tab within the **Alarms Panel** to display a record of all system alarms which have been raised.



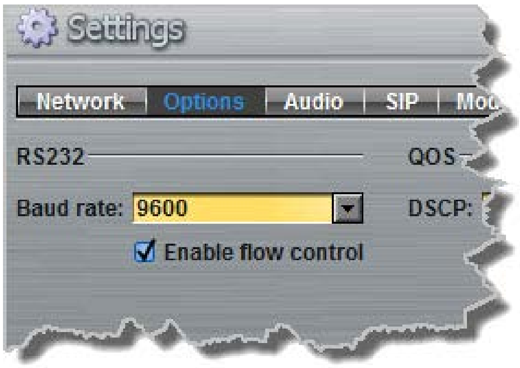

Click the **Purge Alarm History** button to clear all alarms from the **Alarm History** tab.

# 21.24 RS232 Data Configuration

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

## *Setting RS232 Data Rates and Flow Control*

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




**Important Notes:**
- The codec cannot send RS232 data or activate relays on IP-enabled Tieline G3 codecs.
- Auxiliary data is not available for 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 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 programmed to send RS232 data. Session data sent from the dialing codec will program all other compatible codecs (non-G3) when you connect.

## 21.25 Creating Rules

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

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

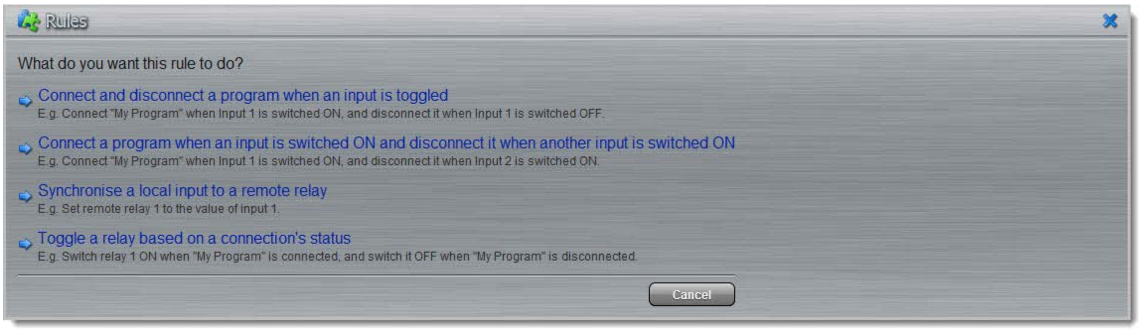
1. Press the **HOME** button to return to the **Home** screen
2. Use the navigation buttons on the front panel to select **Connect** and press the button
3. Select **IP** and press the button
4. Select your preferred **IP Mode** and press the button.
5. Use the down navigation button to select **Setup**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

For more information please see Enabling Relays & RS232 Data.

### *Programming Rules*

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

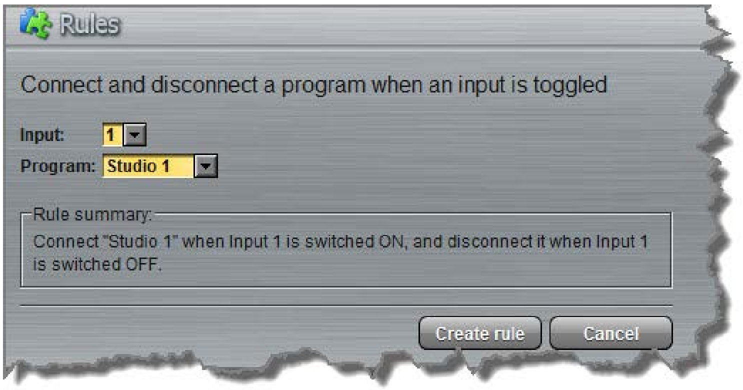
1. Click the **Rules** symbol at the top of the web-GUI screen to open the **Rules panel**.
2. Click **Add New Rule**.
3. Click to select the appropriate programming rule for your requirements. See the Web-GUI Introduction section for explanations of the actions each rule can perform.




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

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

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

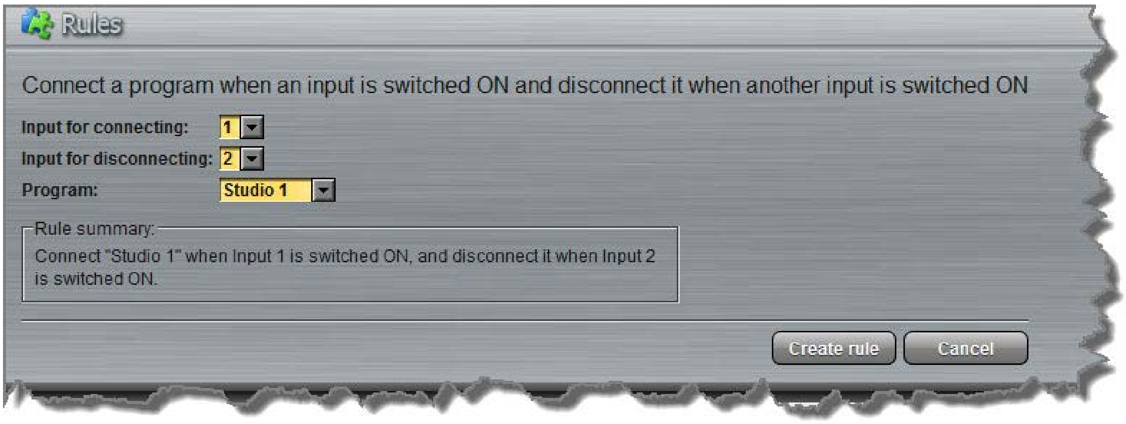



4. Check the **Rule Summary** and click **Create Rule** to save the settings.

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

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

1. Click the second rule in the **Rules panel**.
2. Click the drop-down arrows to select the control port input for connecting and the alternative one for disconnecting.
3. Click the drop-down **Program** arrow to select an individual program which will be connected and disconnected by the change in the control port input states.

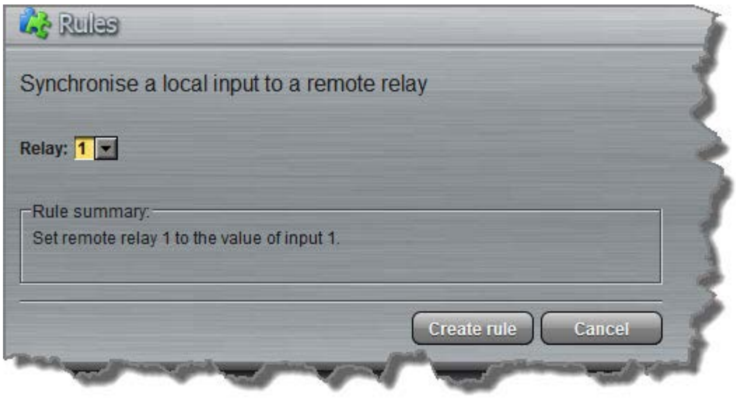



4. Check the **Rule summary** and click **Create Rule** to save the settings.

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

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

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




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

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

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

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


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

### *Deleting Rules*

1. Click the **Rules** symbol at the top of the web-GUI screen to open the **Rules panel**.
2. Click the **Delete** button next to the rule you want to delete.
3. Click **Yes** in the confirmation dialog.

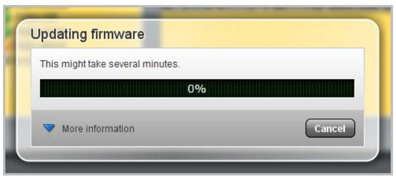
## 21.26 Upgrading Codec Firmware

### *Automatic Firmware Upgrades*

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

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


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




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

### *Manual Firmware Upgrades*

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

1. Click the **Settings**  symbol at the top of the web-GUI screen if the **Settings panel** is not displayed.
2. Click **Firmware**.


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

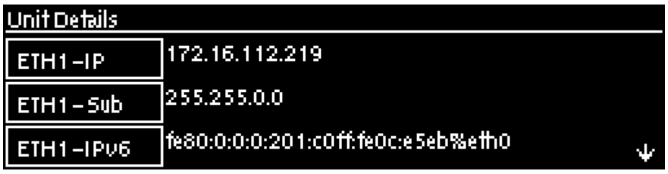
# 22 Front Panel Configuration Tasks

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

## 22.1 Configuring IP via the Front Panel

### *Checking IP Address Details in the Codec*

1. Press the **SETTINGS** button.
2. Select **Unit** and press the button.
3. IP address details and other unit details are listed. Use the arrow up and down buttons to scroll and view all details listed.




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

### *Configure an IPv4 DHCP Address*

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

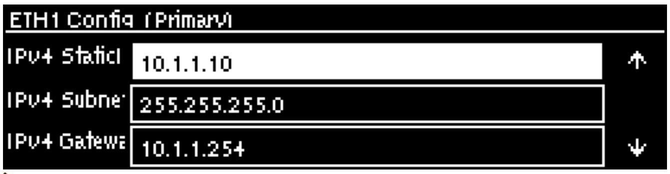
1. Press the **SETTINGS** button.
2. Select **LAN** and press the button.
3. Use the down navigation button to select **ETH1** or **ETH2**.
4. Select **Usage** and choose the appropriate control and/or streaming mode for the connection, then press the button.
5. Select **IPv4 Mode** and press the button.
6. Select **DHCP** and press the button.
7. Use the up navigation button to scroll to the top of the menu and select **Apply Setting**, then press the button to confirm the new settings.

### *Configure a Static IPv4 Address*

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

1. Press the **SETTINGS** button.
2. Select **LAN** and press the button.
3. Use the down navigation button to select **ETH1** or **ETH2**.
4. Select **Usage** and choose the appropriate control and/or streaming mode for the connection, then press the button.

5. Select **IPv4 Mode** and press the button.
6. Select **Static** and press the button.
7. Navigate to **IPv4 Static** and enter the IP address, then press the button.
8. Navigate to **IPv4 Subnet** and enter the Subnet Mask, then press the button.
9. Navigate to **IPv4 Gateway** 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 each Ethernet port on the codec.

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

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

To adjust this setting:

1. Press the **SETTINGS** button.
2. Select **LAN** and press the button.
3. Use the down navigation button to select **ETH1** or **ETH2**.
4. Select **IPv6 Mode** and press the button.
5. Select **Auto**, **Manual** or **Off** and press the button.

By default the codec is programmed 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 **IPv6 Static** (Address)**, IPv6 Prefix** and IPv6 **Gateway** fields in the codec to manually program address details.

## *DNS Server*

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

1. Press the **SETTINGS** button.
2. Use the navigation buttons on the front panel to select **LAN** and press the button.

3. Use the down navigation button to select **ETH1** or **ETH2**.
4. Use the arrow down button to scroll to **Auto DNS**.
5. Press the button to toggle between **Yes** and **No**.

## 22.2 Selecting an Algorithm

The codec offers uncompressed linear audio as well as aptX® Enhanced, LC-AAC, HE-AAC v.1 and HE-AAC v.2, AAC-LD, AAC-ELD, AAC-ELDv2, MPEG Layer 2, G.711 and G.722, Tieline Music and MusicPLUS algorithms. There is a range of pre-programmed connection profiles to simplify codec configuration. 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*

#### *AAC-LC*

LC-AAC is optimized 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 bandwidth 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. HE-AAC v.1, when used for mono connections, 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.

#### *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

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

32kHz or 48kHz sample 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.

aptX® Enhanced is supported over ISDN at the following sample and bit rates:

| Encoding | Bit rate Required | B Channels Required |
|---|---|---|
| aptX® Enhanced Mono 16 bit, 32 kHz | 128 kbps | 2 |
| aptX® Enhanced Mono 16 bit, 48 kHz | 192 kbps | 3 |
| aptX® Enhanced Mono 24 bit, 32 kHz | 192 kbps | 3 |
| aptX® Enhanced Stereo 16 bit, 32 kHz | 256 kbps | 4 |

## 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) |

## Programming an Algorithm into the Codec

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the button.
3. Select **IP** and press the button.
4. 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 **Alg** and press .
7. Navigate to **Manual** to configure all settings manually, or **Profile** to choose a pre-configured algorithm profile, then press .

## How do I Choose an 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 remotes 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 and will default to MPEG Layer 2 if an incoming connection is programmed to use this algorithm.

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 which is the best algorithm setting for the connection you are setting up. Please see the following table for details on the connection requirements of the different algorithms available.

| Algor-ithm | Audio Band-width | Algor-ithmic Delay | IP bit rate per channel | IP over-head per connection | Recommended connection for on-air use |
|---|---|---|---|---|---|
| Linear (Uncom-pressed) | 16/24 bit up to 24kHz | 0ms | sample rate x bits per sample x no. channels | 80kbps | Extremely high quality uncompressed audio distribution and STLs |
| Tieline Music | Up to 15kHz | 20ms | 24 kbps minimum | 16kbps | High quality low bit rate remotes, STLs and audio distribution |
| Tieline Music-PLUS | Up to 22kHz | 20ms | 48 kbps minimum | 16kbps | Very high quality low bit rate remotes, STLs and audio distribution |
| G.711 | 3kHz | 1ms | 64kbps minimum | 80kbps | Voice quality connections to other brands of audio codec |
| G.722 | 7kHz | 1ms | 64kbps minimum | 80kbps | Voice quality connections to other brands of audio codec |
| MPEG Layer 2 | Up to 22kHz | 24 to 36ms | 64kbps minimum | 8.5 - 13.3kbps | Very high quality audio connections between Tieline or other brands of codec. |
| MPEG Layer 3 | Up to 15kHz | 100ms | 64kbps | 8.5 - 13.3kbps | High quality low bit rate remotes, STLs and audio distribution |
| LC-AAC | Up to 15kHz | 64ms | 64kbps | 15kbps | High quality low bit rate remotes, STLs and audio distribution |
| HE-AAC v.1 | Up to 15kHz | 128ms | 48kbps | 7.4kbps | High quality low bit rate remotes, STLs and audio distribution |
| HE-AAC v.2 | Up to 15kHz | 128ms | Minimum 16kbps (Mono); 24kbps (stereo) | 7.4kbps | DAB+ radio streaming and high quality low bit rate remotes, STLs and audio distribution |
| AAC-LD | Up to 20kHz | 20ms at 48kHz | 48kbps minimum | 30kbps | High quality low bit rate remotes, STLs and audio distribution |
| AAC-ELD | Up to 20kHz | 15-30ms | 24 kbps minimum | 15-30kbps | High quality low bit rate remotes, STLs and audio distribution |
| AAC-ELDv.2 | Up to 20kHz | 35ms | Pending release | Pending release | High quality low bit rate remotes |
| aptX Enhanced | 10Hz-24kHz | 2.5ms at 48kHz | 128kbps minimum (16bit; 32kHz) to 288kbps (24bit;48kHz) | 80kbps | Very high quality STLs and audio distribution |
| Opus | 4Hz-20kHz | 20ms | 9.6-256kbps | 16kbps | Very high quality remotes, STLs and audio distribution |

## 22.3 Configuring the Jitter Buffer

A jitter buffer is a temporary storage buffer used to capture incoming data packets. It is used in packet-based networks to ensure the continuity of audio streams by smoothing out packet arrival times during periods of network congestion. Data packets travel independently and arrival times can vary greatly depending on network congestion and the type of network used, i.e. LAN versus wireless networks. The jitter-buffer is encompassed with 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 programmed into the codec will be enabled (default is 500ms). Non-compliant devices include some other brands of codec, web streams and other devices.

### Tieline 'Auto Jitter Buffer' Settings

**Least Delay:** This setting attempts to reduce the jitter buffer to the lowest possible point, while still trying to capture the majority of data packets and keep audio quality at a reasonable level. This setting is the most aggressive in its adaptation to prevailing conditions, so jitter buffer may vary more quickly than with the other settings. It is not recommended in situations where jitter variation is significant and/or peaky. (E.g. 3G/multi-user wireless networks). It is best for stable and reliable links such as dedicated or lightly-loaded WAN/LANs.

**Highest Quality:** This setting is the most conservative in terms of adapting down to reduce delay. The jitter-buffer setting will actually stay high for a longer period after a jitter spike is detected – just in case there are more spikes to follow.  This setting is best used where audio quality is most highly desired and delay is not so critical. Unless delay is irrelevant, this setting is also not recommended over peaky jitter networks (such as 3G) and is best used on more stable networks where large jitter peaks are not as common.

**Best Compromise:** This (default) setting is literally the midpoint between the jitter buffer levels that would have been chosen for the Highest Quality and Least Delay settings. It is designed to provide the safest level of good audio quality without introducing too much extra delay.

**Good Quality** and **Less Delay**: These two settings lie between the mid-point setting of Best Compromise and two settings Highest Quality and Least Delay. They indicate a slight preference and may assist in achieving better performance from a connection without incurring extreme delays in transmission or packet loss.

### Which Algorithms can use Automatic Jitter Buffering?

The following table provides an overview of which algorithms are capable of using the automatic jitter buffer feature over SIP and non-SIP connections.

| Algorithm | Non-SIP Connections | SIP Connections |
|---|---|---|
| Linear (Uncompressed) | ✗ | ✗ |
| Tieline Music | ✓ | ✗ |
| Tieline MusicPLUS | ✓ | ✗ |
| G.711 | ✗ | ✓ |
| G.722 | ✗ | ✓ |
| MPEG Layer 2 | ✓ | ✓ |
| MPEG Layer 3 | ✓ | ✗ |
| LC-AAC | ✓ | ✓ |
| HE-AAC v.1 | ✓ | ✓ |
| HE-AAC v.2 | ✓ | ✓ |
| AAC-LD | ✓ | ✓ |
| AAC-ELD | ✓ | ✓ |
| Opus | ✓ | ✓ |
| aptX Enhanced | ✗ | ✗ |

## Programming 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 Mode** and press the OK button.
5. Use the down navigation button to select **Setup** and press the OK button.
6. Navigate to **JitBuf** and press OK.
7. Select **Auto Jitter Adapt** and press OK.
8. Select your preferred jitter buffer setting and press OK.

**Important Notes:**
- Automatic jitter buffering is disabled for a PCM (linear uncompressed) audio connection.
- There is no jitter buffer setting on a multicast server codec because it only sends and never receives audio packets.

## How to get the Best Jitter Buffer Results

When programming 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 jitter buffer states. Jitter buffer and connection status statistics can be viewed via **HOME** > **Cxns** 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.

## *Auto Jitter Buffer and Forward Error Correction (FEC)*

If forward error correction is programmed then additional data packets are sent over a connection to replace any data packets lost. There is no need to modify jitter buffer settings if you are sending FEC data, only if you are receiving FEC data.

The jitter buffer depth on the receive codec needs to be increased if forward error correction is employed. We recommend you add 100ms to the jitter buffer on a codec receiving FEC at a setting of 20% and 20ms at a setting of 100%.

Tieline’s auto jitter buffer detects the amount of FEC that is being used and automatically compensates to increase the codec jitter buffer if FEC is being used.

## *Fixing Jitter Buffer Settings*

The default jitter-buffer setting in Tieline codecs is 500 milliseconds. This is a very reliable setting that will work for just about all connections. However, this is quite a long delay and we recommend that when you set up an IP connection you test how low you can set the jitter-buffer in your codec.

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select your preferred **IP Mode** and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Navigate to **JitBuf** and press .
7. Select **Fixed Buffer Level** and press .
8. Use the numeric **KEYPAD** to enter the fixed buffer value in milliseconds and press .

If you change the jitter buffer setting in a codec it will only adjust to the new level when link quality is high (e.g. above 70%). This is done to ensure audio quality is not compromised. When manually programming 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 programming jitter-buffer delays into your codec.

| Connection | Jitter-Buffer Recommendation |
|---|---|
| **Private LAN** | 60 milliseconds |
| **Local** | 100 - 200 milliseconds |
| **National** | 100 - 300 milliseconds |
| **International** | 100 – 400 milliseconds |
| **Wireless Network** | 250 - 750 milliseconds |
| **Satellite IP** | 500 - 999 milliseconds |

**Important Note:** The preceding table assumes the use of the Tieline Music algorithms. Do not use PCM (uncompressed) audio over highly contended DSL/ADSL connections without enough bandwidth to support the high connection bit rates required.

## 22.4 Configuring Forward Error Correction

Forward Error Correction (FEC) is designed to increase the stability of UDP/IP connections in the event that data packets are lost. FEC works by sending a secondary stream of audio packets over a connection so that if your primary audio stream packets are lost or corrupted, then packets from the secondary stream can be substituted to replace them. The amount of FEC required depends on the number of data packets lost over the IP connection.

Both the local and remote codec FEC settings can be configured in your codec before dialing. These settings can also be changed 'on the run' while the codecs are connected. FEC should only be used if the Send/Return link quality percentage displayed on the codec is below 99, as it is of no benefit otherwise.

### *Programming FEC into the Codec*

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select **Peer-to-Peer** 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 .

The four FEC settings in Tieline codecs are outlined in the following table with their bit rate ratios.

| FEC Setting | Bit rate Ratios | Connection Use |
|---|---|---|
| **100% (Lowest delay)** | A simultaneous dual-redundant stream (1:1 ratio) is sent from the codec. Twice the connection bit rate is required to operate the codec using the 100% setting. E.g. if your connection is 14,400kbps, you will require an additional 14,400 kbps of bandwidth to allow for the FEC data stream. | Recommended to be used over wireless and international connections. |
| **50%** | Additional data is sent by FEC in a ratio of 2:1. | Recommended for international & national connections |
| **33%** | Additional data is sent by FEC in a ratio of 3:1. | Recommended for national and local connections. |
| **20% (Highest delay)** | Additional data is sent by FEC in a ratio of 5:1. | Recommended for local and LAN connections. |
| **Off** | FEC is off in the codec and the connection bandwidth is equal to the connection bit rate setting in the codec. | Recommended for wired LAN connections & managed T1 & E1 connections for STLs that have connections that aren't shared & have quality of service (QoS). |

**Important Note:** FEC can only be programmed for use with the Music and MusicPLUS algorithms.

## How does FEC work?

If you program a FEC setting of 20% and you are losing one packet in every five sent, the lost packet will be replaced by FEC to maintain the quality of the connection. If you are losing more packets than this, say one in three, it will be necessary to increase the FEC setting to 33% to compensate.

**Note**: There is an inverse relationship between FEC settings and the jitter-buffer millisecond setting that you use for IP connections.

So why not use 100% FEC every time? The answer is because you need twice the bit rate to achieve full redundancy and depending on the link conditions, this could potentially cause more dropouts because of network congestion than it fixes. Here is a simple rule to remember: Your maximum uplink speed is all the bandwidth you have to play with. As a rule of thumb, try not to exceed more than 80% of your maximum bandwidth. If your link is shared, be even more conservative.

You should also consider the remote end too. What is their maximum upload speed? Is the connection shared at either end? Your bit rates, FEC settings and buffer rates must be pre-configured at both ends before you connect, so it's always better to set your connection speed and balance your FEC according to the available uplink bandwidth at each end for best performance.

As an example, if you want 15 kHz mono (using the Tieline Music Algorithm) you will need at least a 24kbps connection for audio. Adding 100% FEC will add another 24kbps making your bit rate 48kbps plus some overhead of around 10kbps is required. If you're on a 64kbps uplink, you should consider reducing your FEC to 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.

## 22.5 Configuring Encode/Decode Direction

By default the codec by is configured to both encode and decode data. However, it is possible to configure the codec to either 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.

Program the transmitting codec to encode only and program the receive codec to decode only when using this feature. To adjust this setting:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the button.
3. Select **IP** and press .
4. Select **Peer-to-Peer** and press the button.
5. Use the down navigation button to select **Setup** and press .
6. Navigate to **Dir** and press .
7. Select the encode or decode direction setting you want and press .

## 22.6 Enabling Relays & RS232 Data

Data must be enabled to activate contact **CONTROL PORT** closure operation and RS232 data.

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select **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**)

**Important Note:** Data transmission is disabled by default.

### *Configuring Control Port Contact Closure Operation*

The **Rules panel** on the web-GUI can be used to configure switch inputs and relay outputs. See the section titled Creating Rules for more information.

### *Configuring RS232 Data*

Once **Data** is enabled, the codec can be connected to external devices and transport RS232-compatible data via the serial port on the rear panel of the codec.

1. Press the **SETTINGS** button.
2. Navigate to **System** and press .
3. Select **RS232 Cfg** and press .
4. Use the navigation buttons to select the correct baud rate.
5. Select **Enable** for flow control and press to save all settings.

**Important Notes:**
- The codec cannot send RS232 data or activate relays on Tieline G3 codecs.
- It is important that you enable serial port flow control within the codec. Flow control regulates the flow of data through the serial port. If disabled, data will flow unregulated and some may be lost.
- Ensure you match the serial port baud rate to match the rate of the external device you are connecting to. Ideally the settings on both codecs should match, or you could have data overflow issues.
- Only the dialing codec needs to be programmed to send RS232 data. Session data sent from the dialing codec will program all other compatible codecs (non-G3) when you connect.
- RS232 data can be sent from the dialing codec to all endpoints of a multi-unicast connection if your codec is capable of these connections. Note: Bidirectional RS232 data is only available on the first connection dialed when multi-unicasting.

## 22.7 Configuring TCP/UDP Protocols

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

### *Tieline Codec Default Port Settings*

By default, the codec uses a TCP session port to send session data and a UDP port to send audio. The session port  is programmed to use the TCP protocol because it is the most likely protocol 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 following table.

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

Note: Using a port scanner to test a codec will be unsuccessful if you try to scan and the port is already in use, i.e. the codec is connected.

### *Changing Codec Port Numbers*

Reasons for adjusting the port setting on your codec include:

- Creating a path through gateways and firewalls.
- Another IP device is already using a codec’s port number.

- More than one studio codec is in use and each codec requires a different port number.

### Changing the Tieline Session Port Number

To adjust the local Tieline session data port used by your codec to send connection data:

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 **Session Port** or **Alternative Session Port** and press .
5. Adjust the setting and press the button to store the new configuration.

### Configure Port Numbers for Connections within Programs

For two codecs to connect, they need to be programmed with matching port numbers. If there is a need to change codec port settings, in most situations you should consult your organization's resident IT professional. To adjust either the session or audio port numbers for a particular connection within a program:

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 either **Sess** (session protocol) or **Proto** (audio protocol) and press .
7. Select the session or audio protocol you want and and press .
8. Use the numeric **KEYPAD** to add a new port number and press .

## 22.8 Configuring QoS for IP Packets

It is possible for IP networks to prioritize and differentiate between data packets 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

IP audio data packets can be programmed for expedited or assured forwarding (Quality of Service or QoS) when traversing different networks. Routers can also be programmed to ignore these forwarding priorities so they are not assured across all networks.

The codec can be programmed 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. 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.

### Configuring QoS

1. Press the **SETTINGS** button.

2. Use the navigation buttons to select **QoS** and press the OK button.


3. Press the OK button and use the **RETURN** button to delete numbers already entered, then use the numeric **KEYPAD** to enter the new setting recommended by your IT administrator.
4. Press the OK 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 respect the QoS setting of the packets sent. Any bandwidth partitioning schemes should partition over a small interval to ensure the codec jitter buffer does not empty and audio remains continuous.

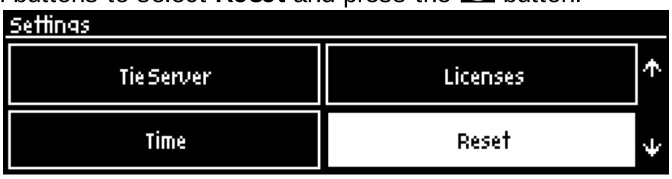
## 22.9 Reset and Restore Factory Default Settings

There are several options in the **Reset** menu which allow you to restore factory default settings within the codec.

| | Function | Description |
|---|---|---|
| **1** | Reset Audio and 'Connect' Settings | Click to restore factory default settings for Audio and Connect menu settings |
| **2** | Restore Factory Defaults | Click to restore factory default settings, excluding user defined programs and call history |
| **3** | Delete Programs & Call History | Deletes custom programs and recent calls in the codec; speed dial contacts are retained |
| **4** | Reboot Codec | Click to restart the codec |

> **Important Note:** After restoring factory defaults, always reboot the codec using the **Reboot Codec** function, not by removing power from the codec.

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Reset** and press the OK button.




3. Navigate to the preferred option from those available and press the OK button.


4. Select **Yes** and press the OK button to confirm the reset function.

### *Reset and Restore Factory Defaults using the Web-GUI*

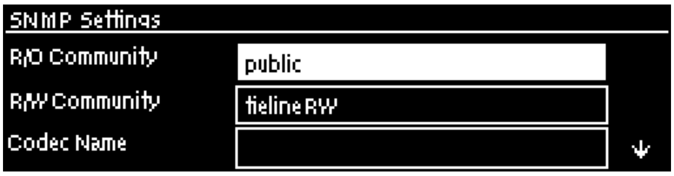
The web-GUI can also be used to reset and restore factory defaults. See Reset Factory Default Settings for more details.

## 22.10 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 see Configuring SNMP in the Codec.

## 22.11 Test Mode

Test mode is used by the codec to perform an input/output loopback test of audio. E.g. channel 1 is routed to channel 1 out.

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

# 23 Reference

The following sections contain reference and troubleshooting information.

## 23.1 Regular Maintenance

Tieline recommends the codec undergoes regular maintenance to ensure operational efficiency and prolong its life.

**WARNINGS:** All work should be carried out by suitably qualified personnel. Remove both power leads from the codec before removing the cover. All parts are mounted on plugs and only a Philips screwdriver is required. Ensure that fan mounting lugs are not hooked out by the cover.

### *Maintenance Schedule*

Tieline recommends a three year maintenance schedule which includes the following procedures to be completed:

1. Evacuate all dust from the unit and clean vents.
2. Replace both PSUs.
3. Replace the fan.

Controlled rack environments may allow a longer maintenance cycle. Uncontrolled environments, where temperatures are elevated, may require a shorter maintenance cycle.

Tieline recommends that the racks in which codecs are installed are thoroughly evacuated to ensure proper airflow from the bottom to the top. Where space is available, a 1RU gap between codecs will assist in minimizing internal temperature build up. Tieline has incorporated dual redundant PSUs and backup alarm features to assist in maintaining reliable operations. The fan has been carefully chosen for long life operation and should not be replaced by a cheaper equivalent. Fan speed control circuitry reduces the fan speed as internal rack temperatures fall below 25 degrees Celsius. This greatly extends the working life of the fan and the codec. If rack temperatures are elevated above 25 degrees Celsius, the fan speed will increase to reduce CPU temperature.

## 23.2 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 high percentage send/return 'link quality' readings of around 99. 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.

**Tech Note**: 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 organization 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.

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.

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. | |

## 23.3 Merlin Compliances and Certifications

### *Declaration of Conformity*

This Merlin 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 Merlin 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 Merlin) does not exceed the Class B limits for radio-noise emissions from digital apparatus as set out in the Radio Interference Regulations of the Canadian Department of Communications.

### Règlement sur le brouillage radioélectrique du ministère des Communications

Cet appareil numérique (Tieline Merlin) respecte les limites de bruits radioélectriques visant les appareils numériques de classe B prescrites dans le Règlement sur le brouillage radioelectrique du ministère des Communications du Canada.

## 23.4 FCC Compliance Statements

### FCC Part 15

Compliance: TIELINE PTY LTD, 25 Irvine Drive, Malaga. Western Australia 6090.

This equipment has been tested and found to comply with the limits for a class B 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. Changes or modifications not expressly approved by Tieline Pty Ltd could void the user’s authority to operate the equipment.

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 and correct the problem by one or more of the following measures:

1. Increase the separation between the equipment and the receiver;
2. Connect the equipment into an outlet on a circuit different to that used by the receiver;
3. Consult the dealer or an experienced radio/TV technician.

### FCC Part 68

FCC Registration Number: 6NAAUS-34641-MD-E
Ringer Equivalence Number (REN):0.5B

A label containing, among other information, the FCC registration and Ringer Equivalence Number (REN) for this equipment is prominently posted on the bottom, near the rear of the equipment. If requested, this information must be provided to your telephone company. USOC Jacks: This device uses RJ11C terminal jacks. The REN is used to determine the quantity of devices, which may be connected to the telephone line. Excessive RENs on the telephone line may result in the devices not ringing in response to an incoming call. In most, but not all areas, the sum of RENs should not exceed five (5). To be certain of the number of devices that may be connected to the line, as determined by the total RENs, contact the telephone company to obtain the maximum RENs for the calling area.

If this equipment causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of the service may be required. If advance notice is not practical, the company will notify the customer as soon as possible. Also you will be advised of your right to file a complaint with the FCC if you believe it is necessary.

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

If you experience problems with this equipment, contact TIELINE Pty Ltd, 25 Irvine Drive, Malaga. Western Australia, 6090. Ph +61 8 9249 6688 Fax +61 8 9249 6858 email info@tieline.com (web page www.tieline.com) for repair and warranty information.

If the problem is causing harm to the telephone network, the Telephone Company may request you remove the equipment from the network until the problem is resolved.

No user serviceable parts are contained in this product. If damage or malfunction occurs, contact TIELINE Pty Ltd for instructions on repair or return. This equipment cannot be used on a telephone company provided coin service. Connection to Party Line service is subject to state tariffs.

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# 24 Merlin PLUS Specifications

| Input/Output Specifications | |
|---|---|
| Analog Audio Inputs | 2 x Female XLR line inputs |
| 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 |
| Auxiliary Input | 1 x 6.35mm (1/4") Mic/Line level Jack on rear panel |
| Female DB-25 | 4 additional analog or digital inputs/outputs |
| Headphones Out/ Aux Out | 1 x 6.35mm (1/4") Jack on rear panel and 1 x 6.35mm (1/4") Jack on the front panel |
| Control Port In/Out | Four relay inputs and four opto-isolated outputs for machine control via a DB15 connector. |
| Audio Input Impedance | High Impedance > 5K ohm |
| Output Impedance | <50 ohm Balanced |
| Clipping Level | +22dBu (input and outputs) |
| A/D & D/A Converters | 24 bit |
| Frequency Response | 20Hz to 22kHz |
| Total Harmonic Distortion | <0.0035% at +16dBu or -89dBu unweighted |
| Signal To Noise Ratio | >98.5dB at +22dBu, unweighted |
| **Sample Frequencies** | |
| IP Sample Frequencies | 16kHz, 32kHz, 44.1kHz, 48kHz |
| **Algorithms** | |
| IP | Tieline Music, Tieline MusicPLUS, G.711, G.722, MPEG-1 Layer 2, MP3, LC-AAC, HE-AAC and HE-AACv2, AAC-LD, AAC-ELD, Opus, 16/24 bit aptX Enhanced |
| IP (uncompressed) | Linear PCM16/24 bit 48kHz sampling |
| **Data and Control Interfaces** | |
| USB | USB 2.0 Host port on the front panel |
| LAN | 2 x 10/100/1000 RJ45 connectors |
| 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 | Tieline; SIP (EBU N/ACIP 3326 compliant); IPv4/IPv6 compatible |
| ISDN via module* | Optional via module slot |
| POTS via module* | Optional via module slot |
| **Front Panel Interfaces** | |
| Display | 256 x 64 monochrome LCD |
| Keypad | 21 button keypad |
| Navigation | 5 button keypad |
| **General** | |
| Size | 1U x 19" Rackmount |
| Dimensions | 19" x 1.75" x 13.5" [482mm (W) x 44mm (H) x 343mm (D) including rear connectors] |
| Weight | 6lb 7.7oz/2.94Kg |
| Power Consumption | Dual AC 100-240V IEC power inlets; 1A - 50-60Hz |
| Operating | 0°C to 45°C (32°F to 113°F) |

| | |
|---|---|
| Temperature | |
| Humidity Operating Range | 20% ≤RH ≤70% (0 to 35°C/32°F to 95°F), non-condensing |

*Available in later releases

# 25 Appendix A: RS232 and Control Port Wiring

## Relays

The codec uses a DB15 connector to facilitate use of four CMOS solid state relays for the control of equipment, consisting of four relay closures and four opto-isolated outputs.

### Inputs

The input signal is referenced to chassis ground, i.e. the ground reference terminal on the connector is connected the chassis. The input device is a high impedance CMOS device with a 330 ohm pull-up resistor to +5 volts.

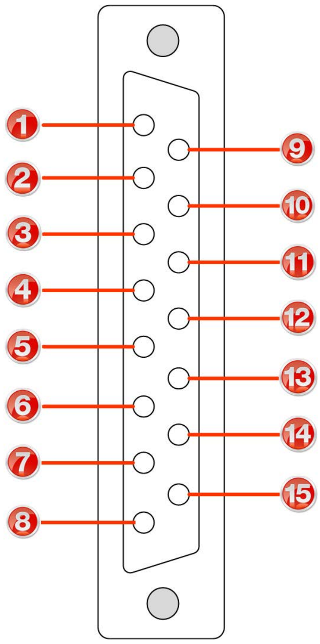
Operation is as simple as joining the input pin to the ground terminal. This can be via a remote relay contact or the open circuit collector of a transistor or FET. DO NOT feed voltages into the inputs.

### 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.

## Control Port Pin-outs

A closing contact across Inputs 1-4 to Ground will provide a closing contact on the remote codec Outputs 1 to 4. If your codec supports 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 connected codecs closed.



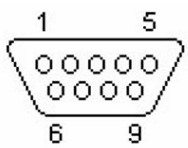

*Female DB-15 Codec Connector*

| Pins | Pin Function |
|---|---|
| 1 | Ground |
| 2 | Output 4 |
| 3 | Output 3 |
| 4 | Output 2 |
| 5 | Output 1 |
| 6 | Ground |
| 7 | Input 3 |
| 8 | Input 1 |
| 9 | Output 4 |
| 10 | Output 3 |
| 11 | Output 2 |
| 12 | Output 1 |
| 13 | Ground |
| 14 | Input 4 |
| 15 | Input 2 |

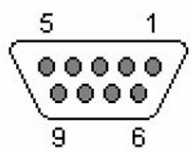
**Important Note:** For more information about how to program relay operations with a PC using the Toolbox web-GUI, please see Creating Rules.

### 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:**
- The codec cannot send RS232 data to, or activate relays on Tieline G3 codecs.
- It is important that you enable serial port flow control within the codec. Flow control regulates the flow of data through the serial port. If disabled, data will flow unregulated and some may be lost.
- Ensure you match the serial port baud rate to match the rate of the external device you are connecting to. Ideally the settings on both codecs should match, or you could have data overflow issues.
- Only the dialing codec needs to be programmed to send RS232 data. Session data sent from the dialing codec will program all other compatible codecs (non-G3) when you connect.

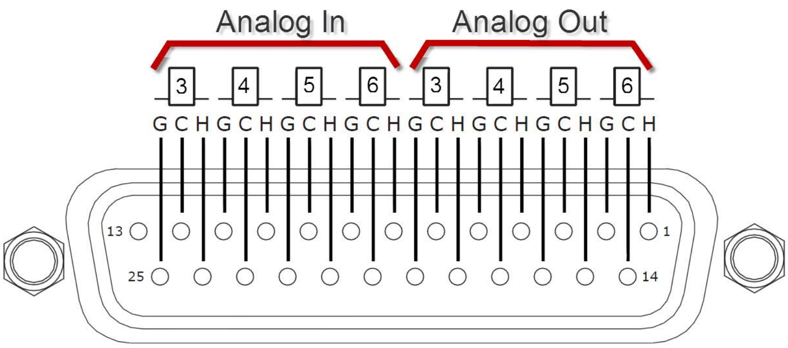
# 26 Appendix B: DB25 Wiring

### DB25 Audio Expansion Pin Outs

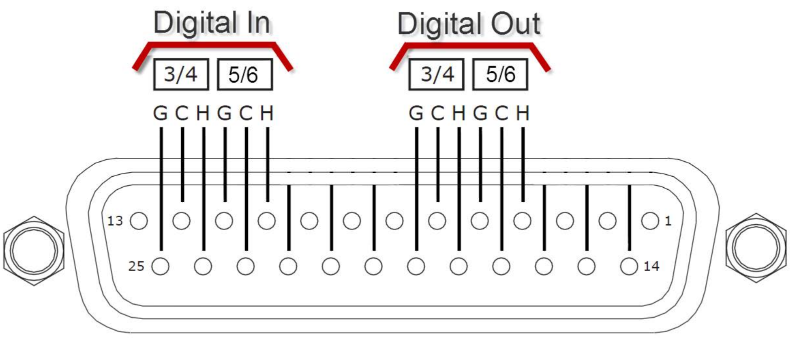
In the following images:

- G = Ground
- C = Cold
- H = Hot

### Analog DB-25 Wiring Pinouts




### Digital AES3 (AES/EBU) DB-25 Wiring Pinouts



# Index

## - 2 -



## - 6 -



## - A -



## - B -



## - C -

# - D -



# - E -



# - F -

# - G -



# - H -



# - I -



# - J -

# - K -



# - L -



# - M -



# - N -



# - O -



# - P -

# - U -



# - W -



# - X -