Tieline® The Codec Company

Merlin
IP / ISDN / POTS Codec
User Manual

Software Version: 2.14.88
Manual Version: 1.5_20150923
September, 2015

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:

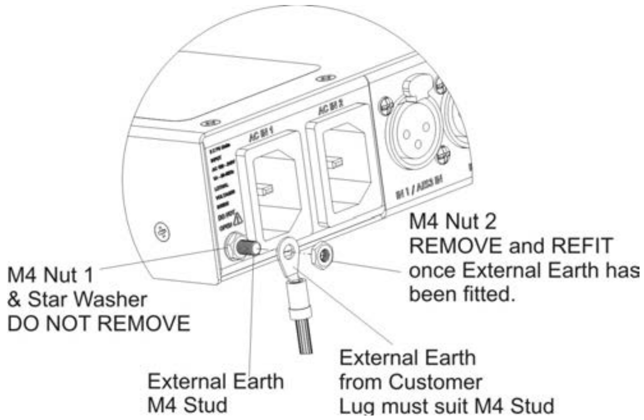
DO NOT CONNECT YOUR TIELINE CODEC TO A DIGITAL PHONE SYSTEM. PERMANENT DAMAGE MAY OCCUR! If you are unfamiliar with any facility, check that the line you are using is NOT a digital line. If the Tieline codec becomes faulty due to the use of a digital phone system, the WARRANTY WILL BE VOID.

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.

POTS — Information specific to POTS 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
Failover	Method of switching to an alternative backup audio stream if the primary connection is lost.
GUI	Graphical User Interface
IFB	Interrupted Foldback/Interruptible Foldback: an intercom circuit consisting of a mix-minus program feed sent to talent, which can be interrupted and replaced by a producer's or director's intercom microphone
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.
Packet	A formatted unit of data carried over packet-switched networks.
Port Address Translation (PAT)	Related to NAT; a feature of a network device that allows IP packets to be routed to specific ports of devices communicating between public and private IP networks
POTS	Plain old telephone system: copper phone network infrastructure
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: Simple Network Management Protocol: a protocol used mostly in network management systems to monitor devices for conditions that warrant administrative attention.
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
TTL	Time-to-Live is the setting used in muliticast servers to ensure data packets have a finite life and don't cause congestion over networks.
UDP	User Datagram Protocol: the most commonly used protocol for sending internet audio and video streams. UDP packets include information which allows them to travel independently of previous or future packets in a data stream
Unicast	Broadcasting of a single stream of data between two points
VLAN	Virtual Local Area Network: partitioning of a single layer-2 network to create multiple distinct broadcast domains
WAN	Wide Area Network; a computer network spanning regions and/or countries to connect separate LANs

4 Getting to know Merlin

The 1RU Merlin rack mount IP codec is designed for the latest digital IP broadcast networks and delivers high quality bidirectional stereo and full duplex communications for remote broadcast connections. Connect to IP codecs or smartphones using Report-IT, as well as ISDN and POTS codecs via optional plug-in transport modules. Merlin has multiple levels of power, audio and network redundancy.

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

Built upon the success of the renowned Commander G3 rack mount codec, Merlin delivers high quality bidirectional stereo audio and bidirectional mono IFB communications for remote broadcast connections over LANs, WANs, the internet and satellite IP. It is ideal for studio and remote truck installations and features:

- High reliability over IP networks without Quality of Service.
- Powerful audio and data routing.
- Simple local or remote command and control.
- Recallable connection 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.
- Supports ISDN and POTS connections via optional interface modules.
- Asymmetric algorithmic encode/decode*.
- SNMP and integrated alarm management.
- Java or HTML5 Toolbox GUI enables remote codec control over WANs.
- 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.

5 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.

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

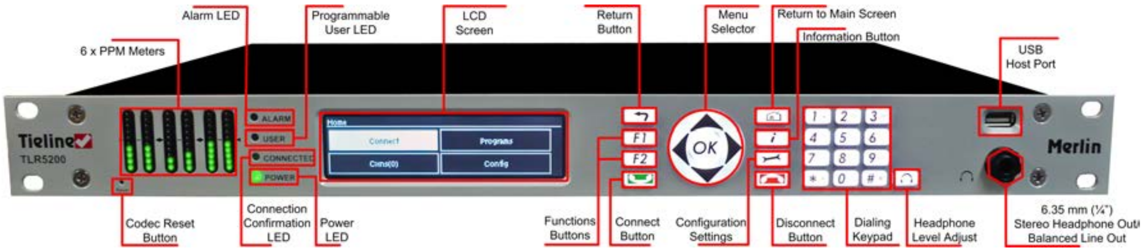
Two additional module slots for inserting optional POTS or ISDN modules.

Supplementary Ground Terminal

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

6 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

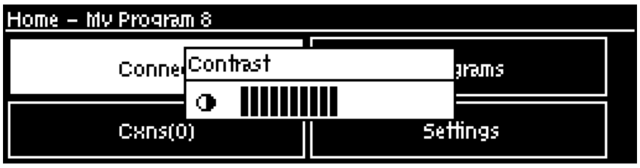
The keypad has alpha-numeric buttons, plus star and hatch (pound) buttons, which can be used to enter contact and program information into the codec.

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	Reset Button	Press to reboot the codec

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

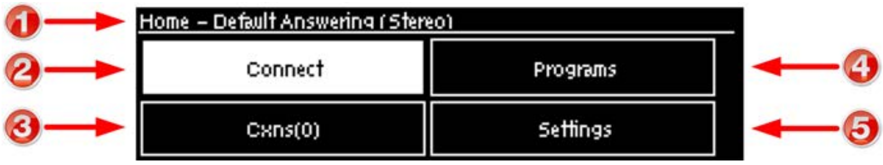
The codec has a 6.35mm (1/4") RTS stereo **HEADPHONE** output for audio monitoring and this can also be switched to a balanced mono auxiliary line output. The front panel **HEADPHONE** output and rear panel **HP/AUX OUT** share the same hardware output. This means both are switched and configured together.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

7 Navigating Menus

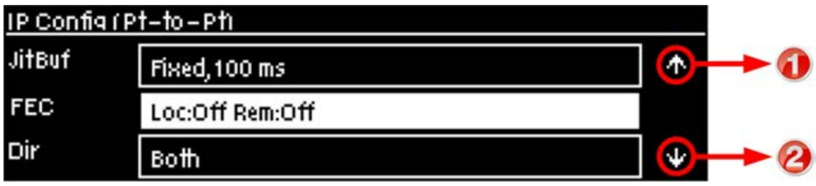
All main codec menus can be launched from the **Home** screen which includes:



	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.

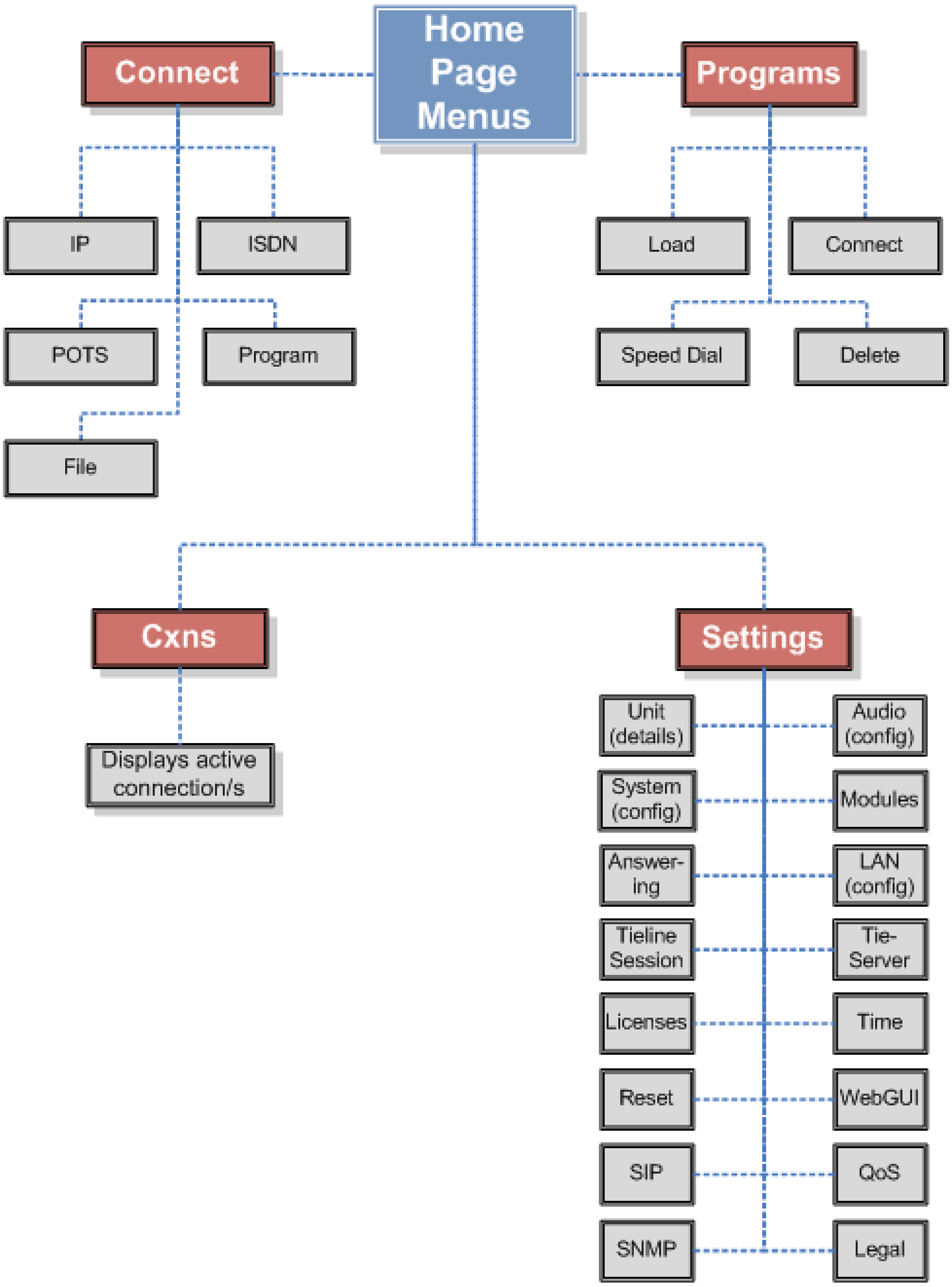
If a complete menu cannot be viewed on a single codec screen, arrows on the right hand side of the screen indicate that the current menu has options below and/or above the visible items. Use the navigation arrows to scroll up and down.



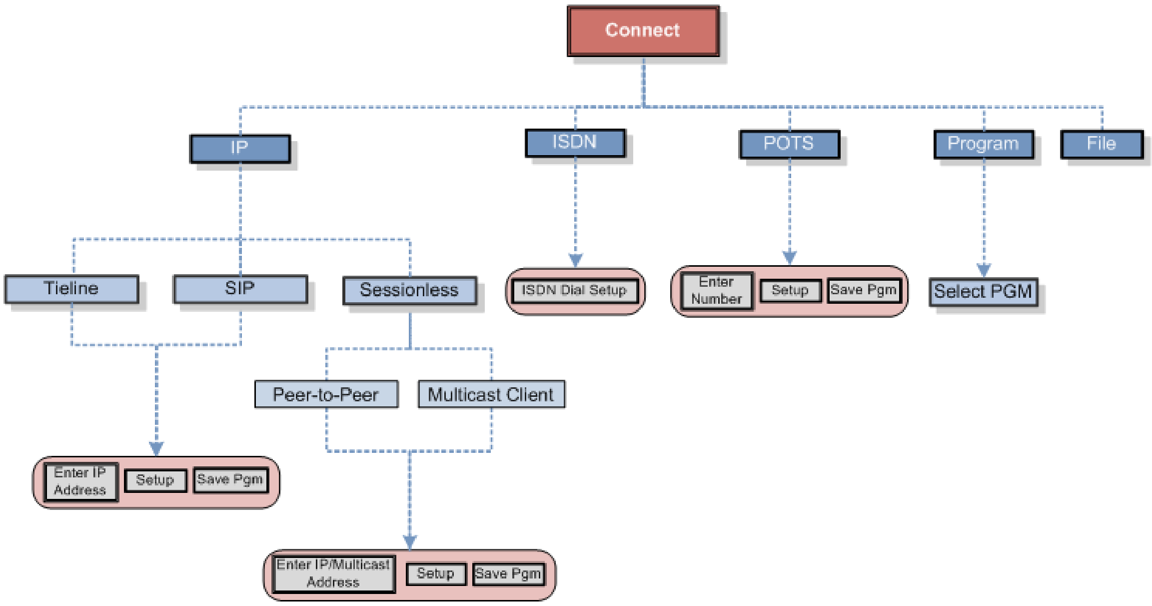
	Features	Codec Home Screen Elements
1	Up Arrow	Arrow indicating menus can scroll upwards
2	Down Arrow	Arrow indicating menus can scroll downwards

Codec Menu Overview

Following is an overview of the codec menus from the **Home** screen. Note: file playback may not be supported in all codecs.

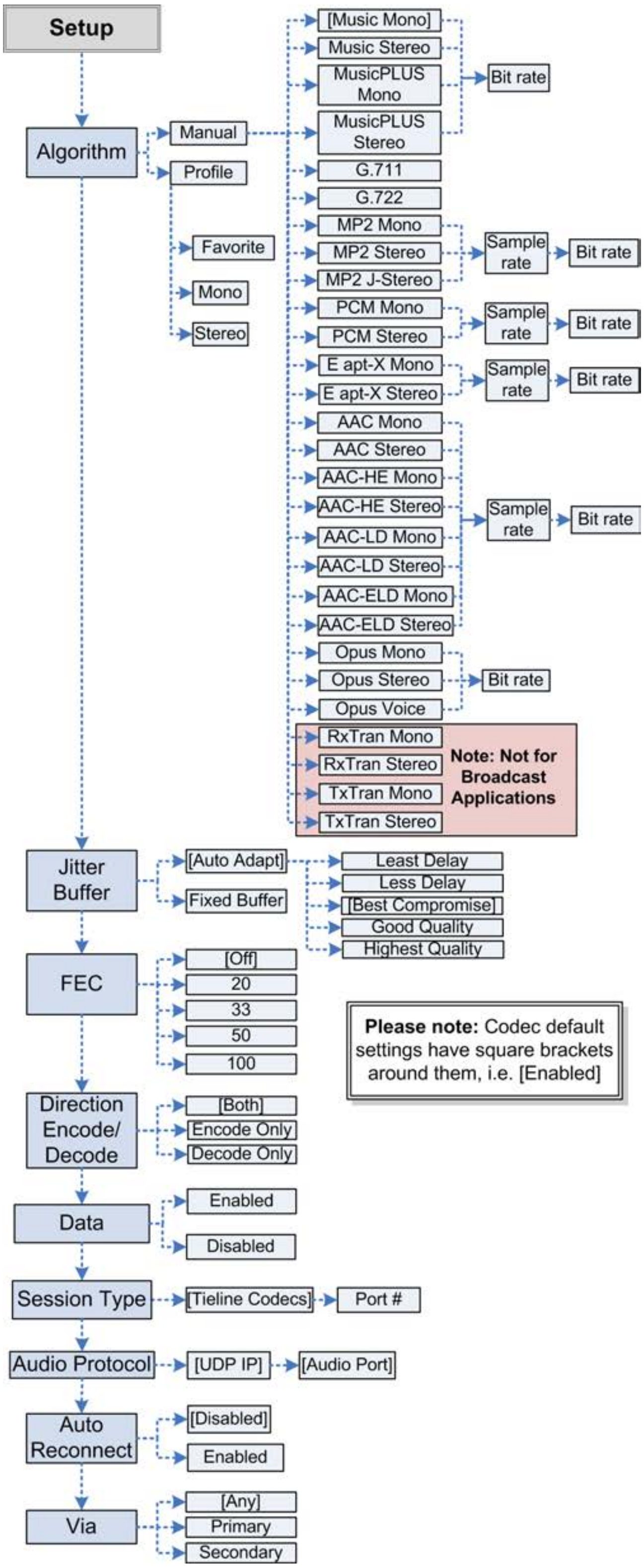


Connect Menu



IP Setup Menu Navigation

After selecting **IP** and a connection mode use **Setup** to adjust connection settings.



Important Notes:

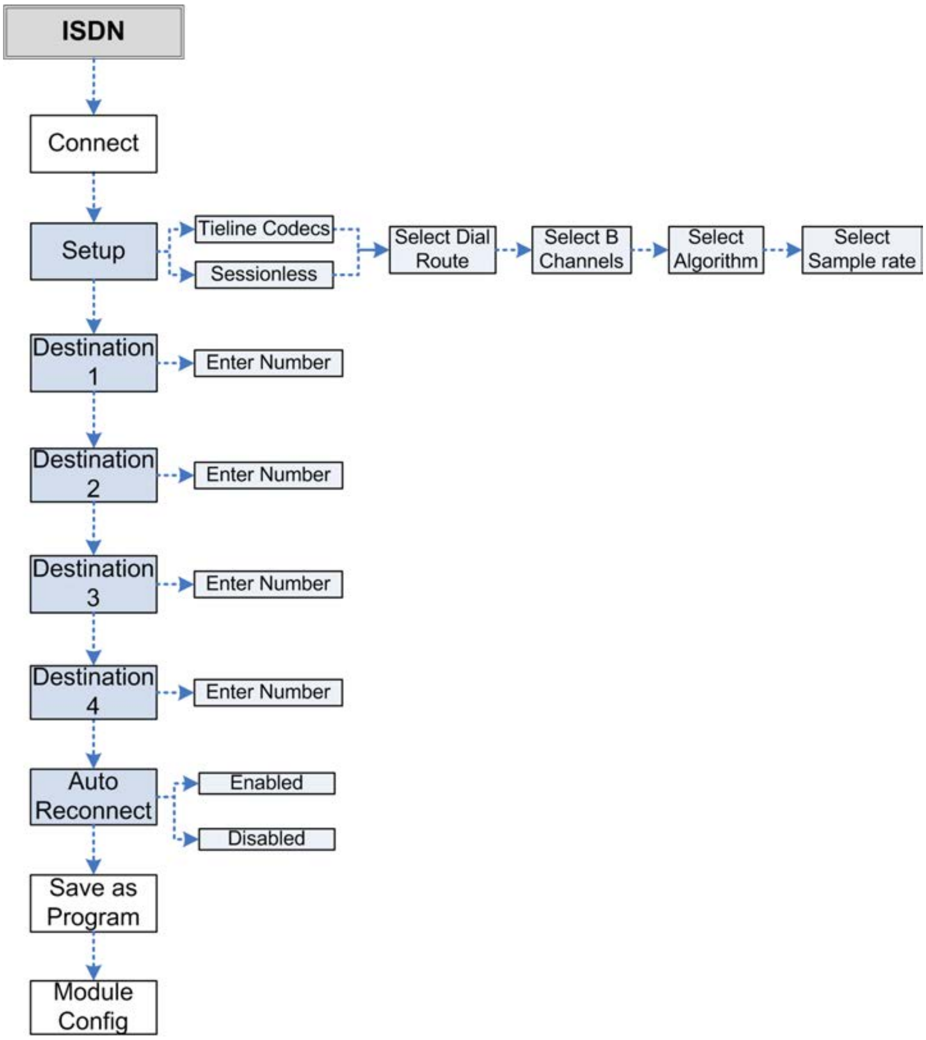
- Depending on the session type selected in the codec, not all options are displayed.

E.g. **Session Type** and **Data** are not displayed when configuring **Sessionless** IP connections.

- Default settings may also change depending on the session type selected, e.g. **Tieline Session** versus **SIP** or **Sessionless**.

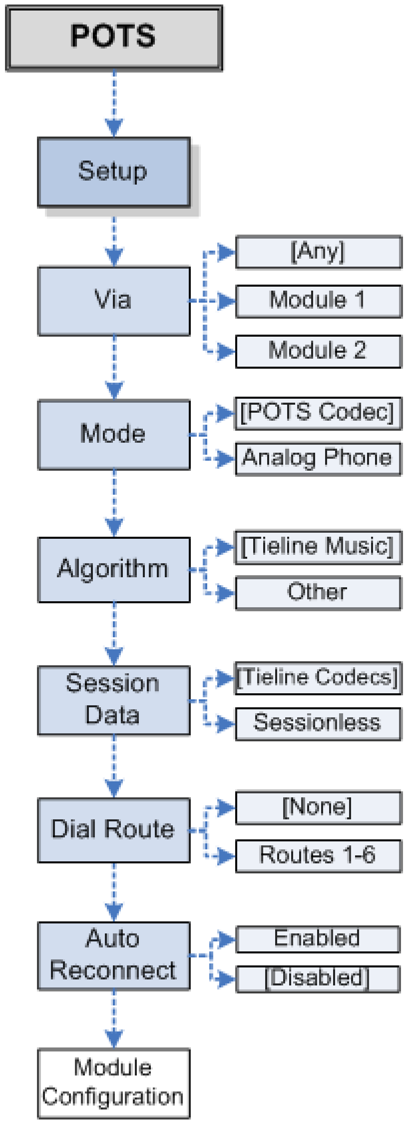
ISDN Menu Navigation

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



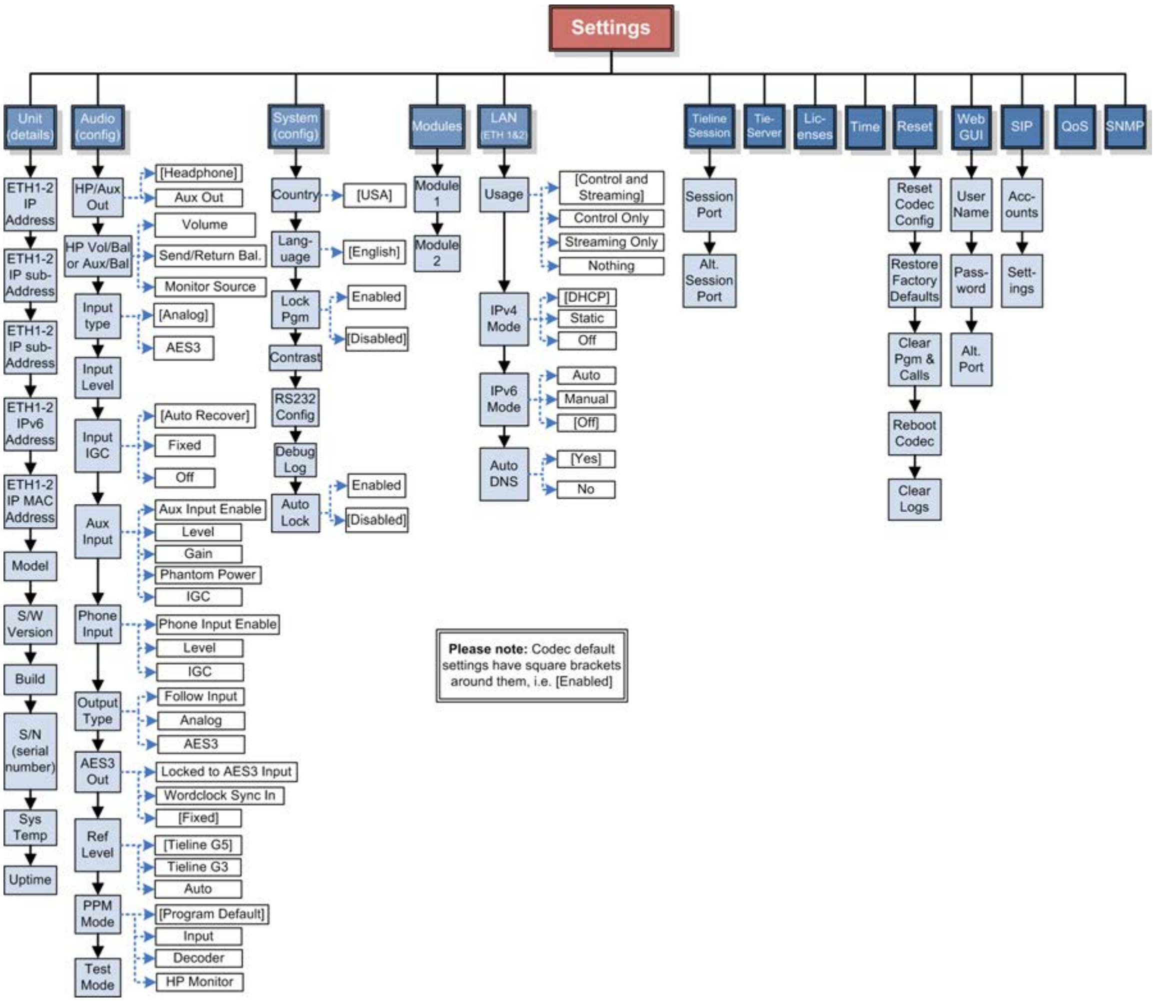
POTS Menu Navigation

Select **Connect > POTS** to configure POTS dialing settings using the codec front panel. Note: default settings are surrounded by square brackets.



Settings Menu

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

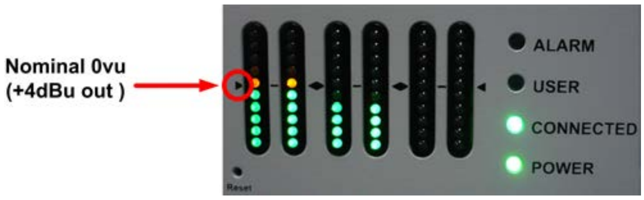


8 Merlin Input Levels and PPMs

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.

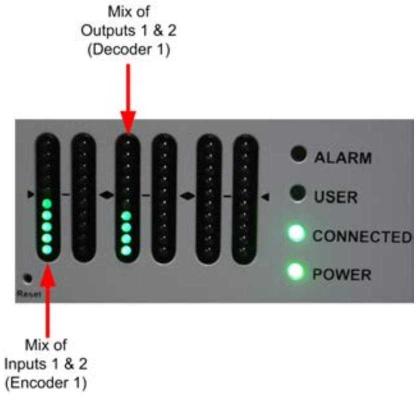
Audio Levels and Meters

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.

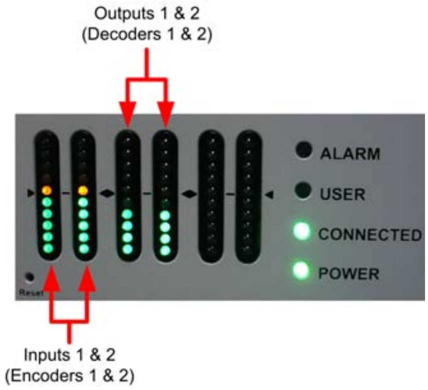


Mono and Stereo Metering

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.

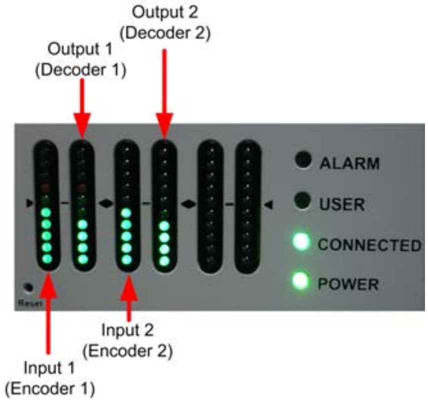


When connecting with a stereo program, the codec displays audio on **PPM1** and **2** for inputs 1 and 2 and **PPM 3** and **4** for the return audio.



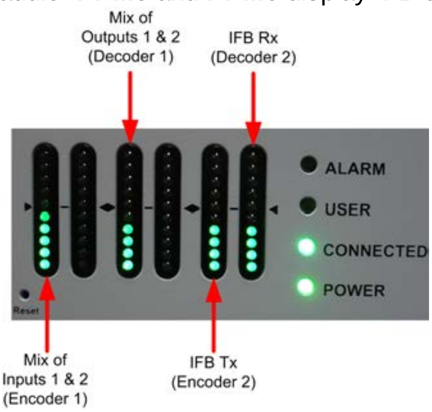
2 x Mono Peer-to-Peer Connection Metering

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.



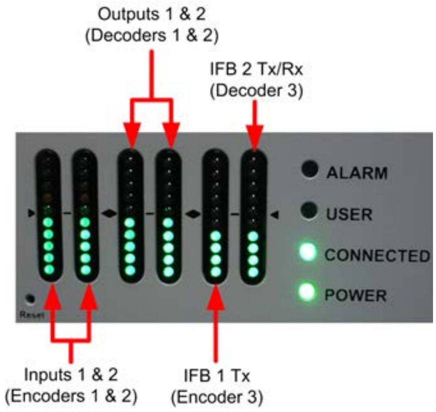
1 x Mono Peer-to-Peer + IFB Metering

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

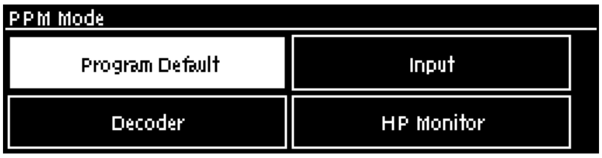
This program transmits a bidirectional stereo audio stream and a separate bidirectional mono IFB communications audio stream.



Adjusting Default PPM Metering

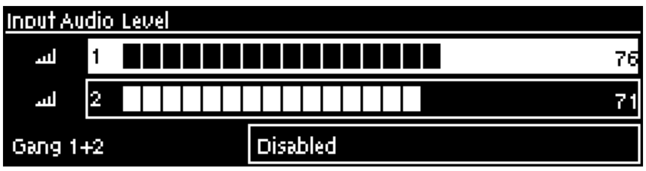
The default PPM metering settings can be adjusted via **Settings > Audio > PPM Mode**. The options include:

	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]**.

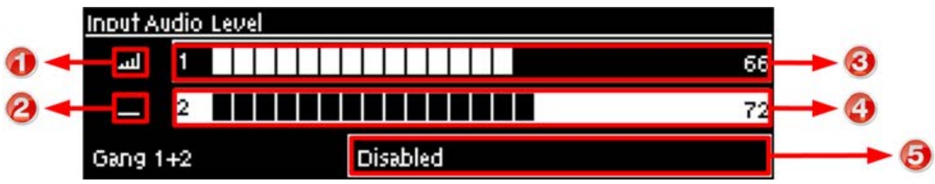


Selecting Analog Inputs and Adjusting Input Levels

1. Press the **SETTINGS** button.
2. Navigate to **Audio** and press .
3. Inputs are grouped in pairs 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.

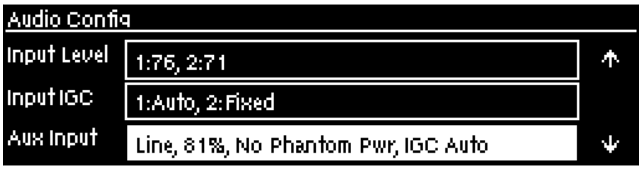


	Input Audio Features	Description
1	Channel On Symbol	Symbol indicates a channel is turned on
2	Channel Off Symbol	Symbol indicates a channel is turned off
3	Input 1 Level Control	Ch 1 level indication with percentage of gain indicated, i.e. **66**
4	Input 2 Level Control	Ch 1 level indication with percentage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

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)

When the broadcast action really starts to heat up, the codec's inbuilt DSP limiter automatically takes care of any instantaneous audio peaks that occur in demanding broadcast situations. **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 .

Configure Audio Reference 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 the audio metering reference scale 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.

9 Configuring AES3 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.

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.

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.

10 Merlin Headphone/Aux Output

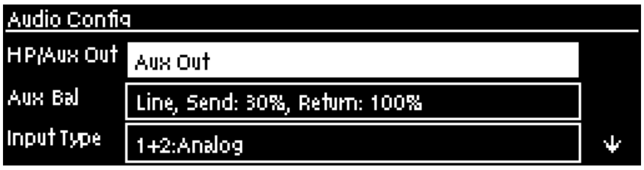
The codec has a 6.35mm (1/4") RTS stereo **HEADPHONE** output for monitoring inputs and return audio. If you are using analog inputs or digital inputs you will see audio metering on the PPMs and can monitor it with the headphones.

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

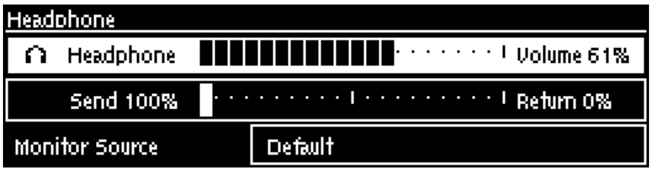
Both the front panel **HEADPHONE** and rear panel **HP/AUX** outputs are configured as stereo headphone outputs by default. To adjust this setting:

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.

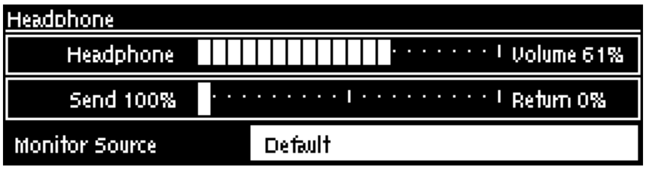


3. Press the down navigation button to select the **Send/Return** audio balance and use the left or right navigation buttons to adjust the balance. The **Send/Return** audio balance dictates whether the front panel **HEADPHONE** output and the rear panel **HP/AUX** output monitors send (input/encoder) 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 to exit the menu.

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 OK. 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
1 x Mono Peer-to-Peer + IFB	Inputs 1&2/ Outputs 1&2	Aux In/Aux Out
1 x Stereo Peer-to-Peer + IFB	Inputs 1&2/ Outputs 1&2	Aux In/Aux Out

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 **RETURN** when you have finished to exit the menu.

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

11 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.

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 fitted with an ISDN G5 module 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. 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.

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.

How to Configure ISDN G5 Modules

To configure the codec to dial using ISDN for the first time:

1. Ensure that the correct country setting is configured in your codec via **Settings > System > Country**.
2. If you are dialing between two Tieline codecs you normally only need to configure an ISDN dialing program via **Connect > ISDN**. See Configuring ISDN to adjust settings using the Java Toolbox Web-GUI, or click here to adjust settings using the HTML5 Toolbox Web-GUI.

Other more advanced settings can also be configured:

1. Select **Settings > Modules > [Select ISDN Module]** to adjust ISDN module settings specific to your codec site. See Configure ISDN Module Settings for more information. See Configuring ISDN Modules to adjust settings using the Java Toolbox Web-GUI, or click here to adjust settings using the HTML5 Toolbox Web-GUI.
2. 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 (sessionless calls).

   Adjust answering configuration via **Settings > Answering > ISDN Answer Configs > [Select Config]** and see ISDN Answering Configuration for more information (see Configuring ISDN Answering to adjust settings using the Java Toolbox Web-GUI, or click here to adjust settings using the HTML5 Toolbox Web-GUI).

12.1 ISDN Module Settings

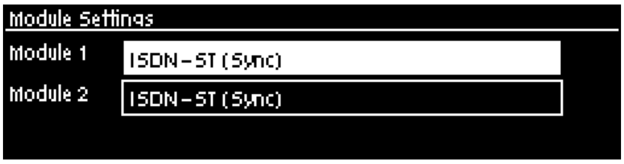
ISDN settings in the **Module** menu will determine how each installed module operates at a particular site. This allows you to copy programs between codecs installed at different locations and separately configure site-specific settings for how each ISDN module should connect. Other answering-related settings are available in the **Answering** menu via **Settings > Answering > [Select ISDN Config]**.

Configuring ISDN G5 Modules

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

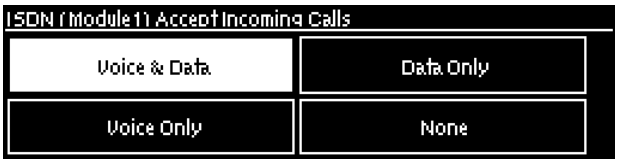
Important Note: You can also configure your ISDN module by pressing the **HOME** button to return to the **Home** screen and select **Connect > ISDN**. Then use the down navigation button to select **Module Configuration** and press the button.

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



Important Note: ISDN Sync should be displayed when an ISDN line is connected to the codec. This appears regardless of whether you have configured the 'ISDN Line Type' correctly.

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.

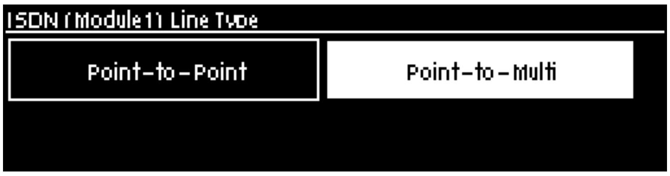


Important Note: G.711 is the default algorithm for incoming connections when **Voice Only** is selected. There are two G.711 algorithms and the one used by the codec depends on the country setting in the codec. The µ-law algorithm is used in the USA, Japan and Canada, whereas the A-law algorithm is used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

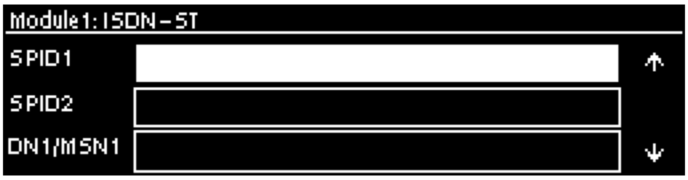
4. Navigate to **Network** and press the button. Select the **Network Type** corresponding to the region in which you are using the codec, then press the button.

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 (UK, Europe, Australia and most other countries)
JPN-NTT	If you are in the Japan and your network is NTT

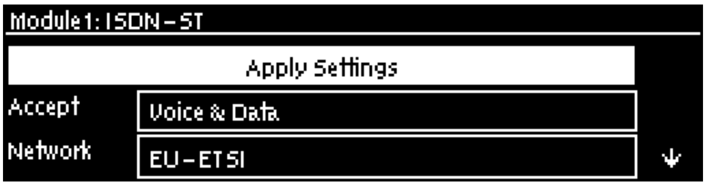
5. Navigate to **Line Type** and press the button. Ask your Telco whether your ISDN line is Point-to-Point or Point-to-Multipoint. By default select **Point-to-Multipoint**, unless your switch type is point-to-point, your Telco says the line is point-to-point, or you are connected to a PABX system. Most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



8. Navigate up to **Apply Settings** and press the 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.

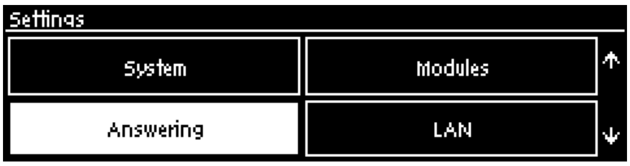
Note: SPID numbers normally include the phone number and additional prefix or suffix digits up to 20 digits long.

12.2 ISDN Answering Configuration

Important Note: For more detailed information about **ISDN Answer Config** parameters, including bonding and 'route' configuration etc., please see Configuring ISDN Answering in the Java Toolbox Web-GUI manual, or Configuring ISDN Answering in the HTML5 Toolbox Web-GUI manual.

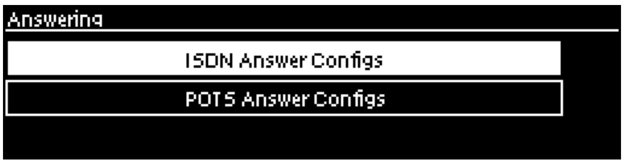
ISDN Answering Configs are used to determine how codec ISDN modules will behave when answering ISDN calls.

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

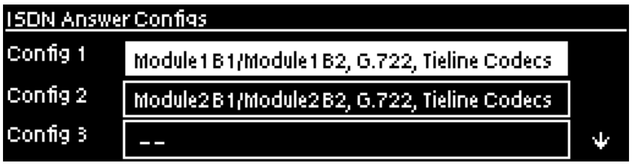


Important Note: You can also configure your ISDN module by pressing the **HOME** button to return to the **Home** screen and select **Connect > ISDN**. Then use the down navigation button to select **Module Configuration** and press the button.

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

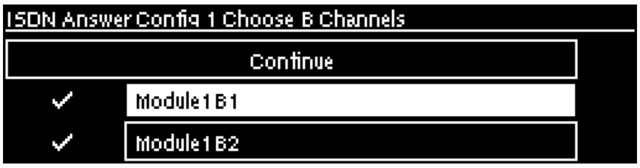


3. Navigate to one of the four available **Configs** and press the button.



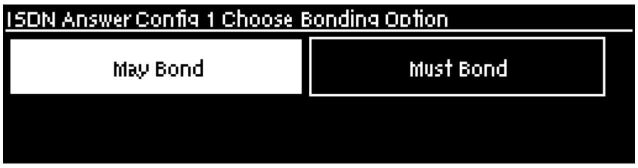
4. Navigate to **Edit** and press the button.

5. 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.

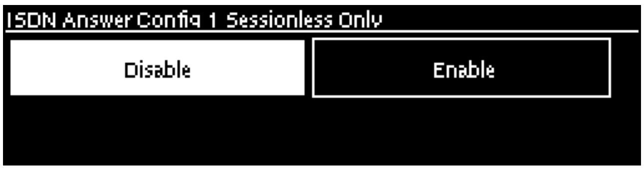


Important Note: If a B channel has been selected within another **Config** it will not be visible. Only available B channels are displayed.

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



7. Select **Disable** when connecting to Tieline codecs using session data, or select **Enable** if connecting to non-Tieline codecs only, then press the button.

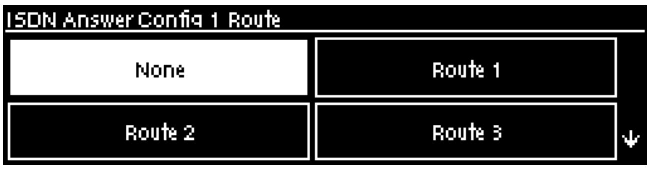


Important Note: Select **Disable** if the codec is expected to receive ISDN calls from Tieline codecs, or both Tieline and non-Tieline codecs (i.e. you are not sure which type of codec may call). In this mode, once the codec answers a call, it expects to receive Tieline session data from the caller and configure its own algorithm settings according to that. If it fails to receive Tieline session data within 5 seconds (i.e. a non-Tieline codec is calling, or a Tieline codec with session data disabled), it will use the settings in the **ISDN Answering Config** instead.

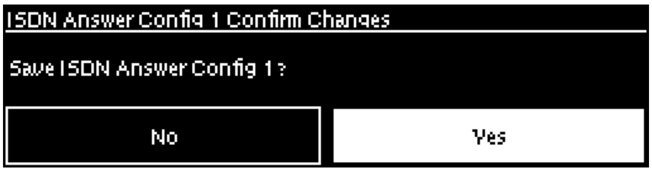
Enable **Sessionless Only** when answering ISDN calls from non-Tieline codecs only. When **Sessionless** is configured, the codec will not wait for the session data. This reduces the time taken to answer an inbound sessionless call.

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

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



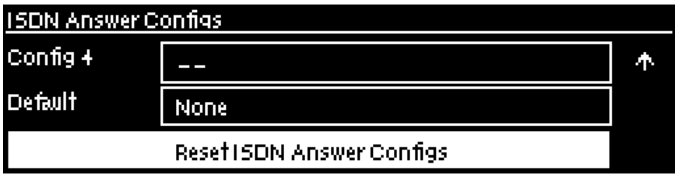
10. Select **Yes** and then press the button to confirm all changes.



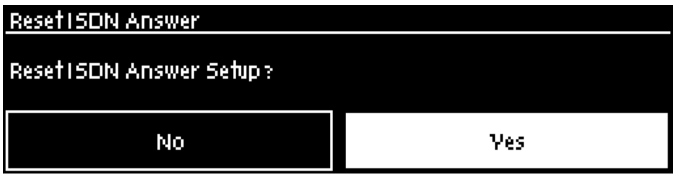
Reset ISDN Answer Configs

To reset ISDN answering settings to factory defaults:

1. Navigate to **Settings > Answering > ISDN Answer Configs > Reset ISDN Answer Configs** and then press the button.



2. Select **Yes** and press the button.



13 About POTS Modules

The Tieline POTS G5 module can be used in the codec to stream high quality audio over a POTS (PSTN) phone line. The codec has two module slots available and it is possible to insert one or two POTS G5 modules, or a mix of POTS G5 and ISDN G5 modules as required. The Tieline Music algorithm can deliver 15 kHz quality bi-directional audio at bit rates as low as 24kbps over a POTS connection.

Modem Negotiation and Line Quality

The codec can send and receive high-speed digital information over a standard POTS telephone line via the modem in the POTS G5 module.

G5 POTS modems initially attempt to establish a link at the lowest **Max Bit rate** setting configured in the two modules being connected. If the POTS line doesn't support this bit rate, the modems will attempt to connect at the highest possible bit rate to suit the prevailing line quality at each end of the link. The modem then perform a process called ‘training’, during which the codecs at each end of the link analyze the line. The codecs will then 'renegotiate' the link downwards to the highest possible bit rate where line quality is greater than 70%.

The POTS G5 module contains a SmartDAA™ (Smart Data Access Arrangement) line interface, which isolates the modem from voltages on phone lines. It is important to select the correct country in the codec from which you are dialing. This allows the SmartDAA to automatically adjust for the line voltage present in that country.

Important Note: It is possible to connect two concurrent POTS connections, however the codec will not bond two POTS connections.

Connecting to G3 Codecs using POTS

The codec will successfully connect to Tieline Commander G3 and i-Mix G3 codecs over POTS. These Tieline G3 codecs may use:

- **POTS** modules (older superseded version)
- **POTS G3** modules (current version)

Connecting to POTS G3 Modules

POTS G3 modules operate in the same way as POTS G5 modules when connecting, e.g. they establish a link at the default bit rate of 28.8kbps and then 'renegotiate' the link downwards to the highest possible bit rate where line quality is greater than 70%.

Connecting to Legacy POTS Modules in G3 Codecs

These modules have slightly different characteristics when connecting. When dialing from a POTS G5 module to these older POTS modules the codecs will attempt to connect initially at 19.2kbps. If line quality is above 80% at this bit rate then the codec will ‘retrain’ the connection up to a maximum of 28,800bps (depending on modem handshaking). The codec will then renegotiate the link downwards to the highest possible bit rate where line quality is greater than 70%.

How to Configure POTS G5 Codec Connections

To configure the codec to dial using POTS for the first time:

1. Ensure that the correct country setting is configured in your codec via **Settings > System > Country**. This ensures the correct settings are used by the codec when making POTS connections.
2. When dialing between two Tieline codecs you normally only need to configure a POTS dialing program via **Connect > POTS**.

Other more advanced settings can also be configured:

1. Select **Settings > Modules > [Select POTS Module]** to adjust POTS module settings specific to your codec site. See Configure POTS Module Settings for more information. See Configuring POTS Modules to adjust settings using the Java Toolbox Web-GUI, or click here to adjust settings using the HTML5 Toolbox Web-GUI.
2. If you are connecting to non-Tieline codecs you may need to create an answering "Config" via **Settings > Answering > POTS Answer Configs > [Select POTS Config]**, which will determine the module used and relevant settings for answering a non-Tieline POTS call. See Configure POTS Module Answering for more information. See Configuring POTS Answering to adjust settings using the Java Toolbox Web-GUI, or click here to adjust settings using the HTML5 Toolbox Web-GUI.

Important Note: The codec has a single analog phone input shared by both modules (default setting is **Off**). This phone input is used to monitor modem tones in **POTS Codec** mode and for receiving audio in **Analog Phone** mode. **Phone Input** settings can be adjusted via **Settings > Audio > Phone Input**.

Making Analog Phone (Voice) Calls

All POTS G5 modules are capable of making analog voice calls. It may be necessary to make an analog call to dial a telephone hybrid, or to use for communications, or because there is no Tieline codec at the other end of the link. Remember analog voice calls are only 3 kHz audio quality. To select analog phone answering mode in a POTS G5 module navigate to **Settings > Modules > POTS > Answer Mode [Analog phone]**.

Important Analog Phone Note: The codec has a single analog phone input which is shared by two POTS G5 modules when installed. As a result, two concurrent analog phone connections are not recommended because both connections share the same input and audio will be accepted from the oldest active connection only.

13.1 POTS Module Settings

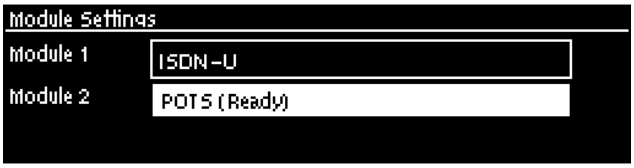
POTS settings in the **Module** menu determine how your codec behaves at a particular site. This allows you to copy programs between codecs installed at different locations and configure site-specific settings for how each module should connect. Other answering-related settings are available in the **Answering** menu via **Settings > Answering > [Select POTS Config**] if you are connecting to non-Tieline codecs over POTS.

Important Notes: When **POTS (Ready)** is displayed throughout POTS menus it means the POTS module has initialized and is ready to accept or make a call.

How to Configure POTS G5 Modules

1. Press the the **SETTINGS** button, then navigate to **Modules** 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. Complete configuration changes as per the following options and then navigate up to **Apply Settings** and press the button to apply all module settings.

Module (Site) Settings

Answer Mode (Affects Answering Only)

Answer Mode selects how the selected module in the codec will be able to answer incoming POTS line calls. Options include:

- **POTS Codec:** allows the POTS G5 module to receive encoded audio data over a POTS line.
- **Analog Phone:** configures the POTS G5 module to receive a standard analog phone call.
- **Disabled:** disables the POTS G5 module from receiving a **POTS Codec** or **Analog Phone** call.

Calls are answered based on the settings in **Config 1 & 2** via **Settings > Answering > POTS Answer Configs**. Adjustments to these **Config** settings are not normally necessary when connecting between Tieline codecs. Default settings may need to be adjusted when connecting to non-Tieline codecs over POTS (see POTS Answering Configuration for more info).

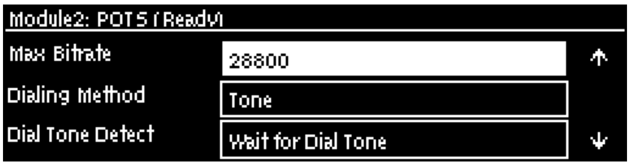
Maximum Bit rate (Affects Dialing and Answering)

The default setting for the **Max Bitrate** is **28800** (28.8kbps) and this only affects **POTS Codec** calls. The range of the setting is 9.6kbps to 33.6kbps. Even if the line is capable of establishing a connection at a higher bit rate, the **Max Bitrate** setting is the highest bit rate that will be attempted.

G5 POTS modems initially attempt to establish a link at the lowest **Max Bit rate** setting configured in the two modules being connected. If the POTS line doesn't support this bit rate, the modems will attempt to connect at the highest possible bit rate to suit the prevailing line quality at each end of the link.

In the initial connection phase, the modems perform a process called 'training', to analyze the line and compensate for frequency and phase response. This also cancels out any echo that may be present. The codec will then 'renegotiate' the link downwards to the highest possible bit rate where line quality is greater than 70%. Negotiation is the process of bit rate adjustment.

Reducing this value can improve connection reliability on poor quality lines. If two codecs are not configured the same, they will always attempt to connect at the lowest of the two **Max Bit rate** settings.



Dialing Method (Affects Dialing only)

Use this menu to select **Tone** (DTMF) or **Pulse** dialing over **POTS Codec** connections. Tone dialing is used always when the **Answer Mode** is **Analog Phone**.

Dial Tone Detect (Affects Dialing only)

There are two settings in this menu:

- **Dial Tone Detect**: The module will only be allowed to dial when a dial tone is present on the line.
- **Blind Dialing**: Allows the module to dial when no dial tone is present.

Monitor Modem Tone (Affects Dialing and Answering)

This setting can be **Enabled** or **Disabled**. If enabled the module will allow audio monitoring of modem tones during connection in **POTS Codec** mode via the phone input. By default, the following phone input monitoring rules apply when multiple POTS G5 modules are installed in a codec and multiple POTS connections are dialed.

Module 1	Module 2	Audio Rule
POTS Codec (Monitor Modem Tone)	POTS Codec (Monitor Modem Tone)	The phone input receives a mix of modem tone audio from both modules
POTS Codec (Monitor Modem Tone)	Analog Phone	The phone input receives analog phone input audio only and mutes modem tone monitoring
Analog Phone	Analog Phone	The phone input accepts audio from the oldest active connection only

Important Notes:
- POTS modem tones are audible in the left side of the headphone output, irrespective of the type of program loaded in the codec.
- Modem tone monitoring will work even if **Phone Input Enable** is **Off** via **Settings > Audio > Phone Input > Phone Input Enable [Off]**.
- Modem tone monitoring is only enabled during the initial connection training and negotiation period in **POTS Codec** mode.
- The monitoring volume can be adjusted via **Settings > Audio > Phone Input > Level**.

Country

This displays the current country setting in the codec. To adjust this setting select **Settings > System > Country**.

Caution: Don't forget to navigate up to **Apply Settings** and press the OK button to apply all module settings before leaving this menu!

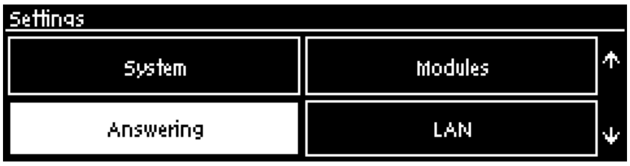
13.2 POTS Answering Configuration

Connection setting preferences are normally exchanged via session data sent between two Tieline codecs when a connection is established. If you answer a call from a non-Tieline codec you will need to create an answering "Config" to determine the settings used when connecting, and designate which module will answer the call (if more than one POTS module is installed).

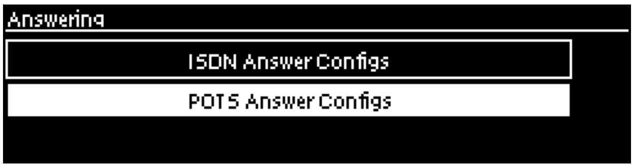
Important Notes:

- **POTS Answer Config** settings are applied to **POTS Codec** connections and not **Analog Phone** connections.
- When receiving a call from a Tieline codec sending session data (i.e. not a **Sessionless** connection), the algorithm setting from the dialing codec overrides the setting in the **POTS Answer Config** menu.
- The default **POTS Answer Configs** accept a call from an incoming Tieline codec with session data enabled. They will also answer a call from a Comrex POTS codec by using the **Other** algorithm.
- For more information about POTS answering parameters, including 'route' configuration, please see Configuring POTS Answering in the Java Toolbox Web-GUI manual, or Configuring POTS Answering in the HTML5 Toolbox Web-GUI manual.

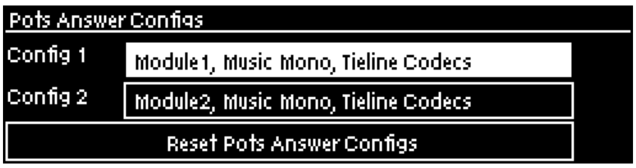
1. Press the the **SETTINGS** button, then navigate to **Answering** and press the OK button.



2. Navigate to **POTS Answer Configs** and press the OK button.

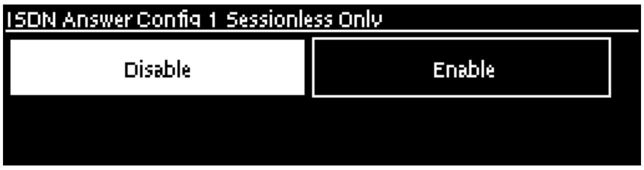


3. Navigate to **Config 1** and press OK to configure a POTS module in module slot 1, or navigate to **Config 2** and press OK to configure a POTS module in module slot 2.



4. Select **Info** to view current settings or **Edit** to adjust **Config** settings, then press the OK button.

5. Select **Disable** when connecting to Tieline codecs using session data, or select **Enable** if connecting to non-Tieline codecs only, then press the button.



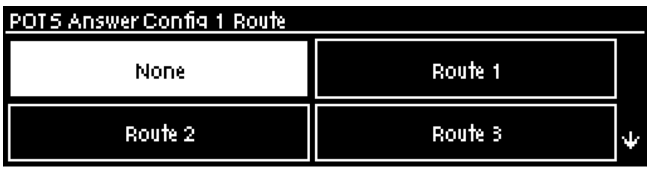
Important Note: Select **Disable** if the codec is expected to receive POTS calls from Tieline codecs, or both Tieline and non-Tieline codecs (i.e. you are not sure which type of codec may call). In this mode, once the codec answers a call, it expects to receive Tieline session data from the caller and configure its own algorithm settings according to that. If it fails to receive Tieline session data within 5 seconds (e.g. a Comrex POTS codec is calling), it will use the settings in the **ISDN Answering Config** instead.

Enable **Sessionless Only** when answering POTS calls from non-Tieline codecs only. When **Sessionless** is configured, the codec will not wait for the session data. This reduces the time taken to answer an inbound sessionless call.

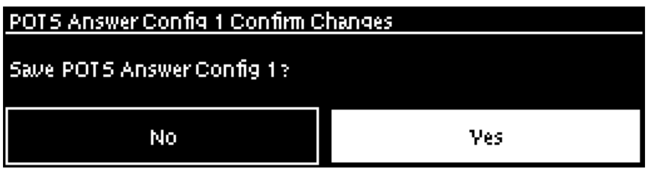
6. Select **Other** when connecting to Comrex® Vector, Matrix® and BlueBox® codecs, then press the button.

Important Note: On the Comrex codec select its "Music" algorithm. Please note that 9.6kbps connections are not supported by the Comrex codecs.

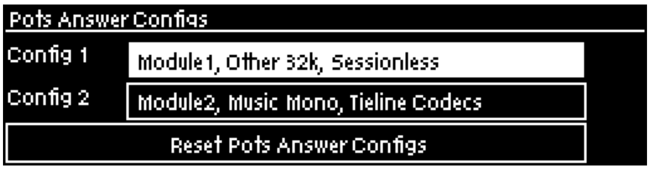
7. If required you can specify the audio stream **Route** when answering a call from a non-Tieline codec, then press the button.



8. Select **Yes** and press the button to confirm the new **Config** settings.



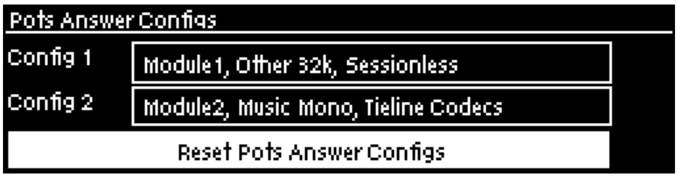
9. The new **Config** will be displayed showing the updated settings.



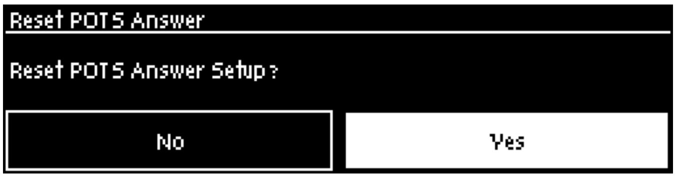
Reset POTS Answer Configs

To reset POTS answering settings to factory defaults:

1. Navigate to **Settings > Answering > POTS Answer Configs > Reset POTS Answer Configs** and then press the OK button.



2. Select **Yes** and press the OK button.



14 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 OK.
3. Use the navigation buttons to select **Language** and press OK.
4. Select a language and press OK.

15 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 acknowledgement 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 cannot be unloaded by a codec dialing in with different program settings.

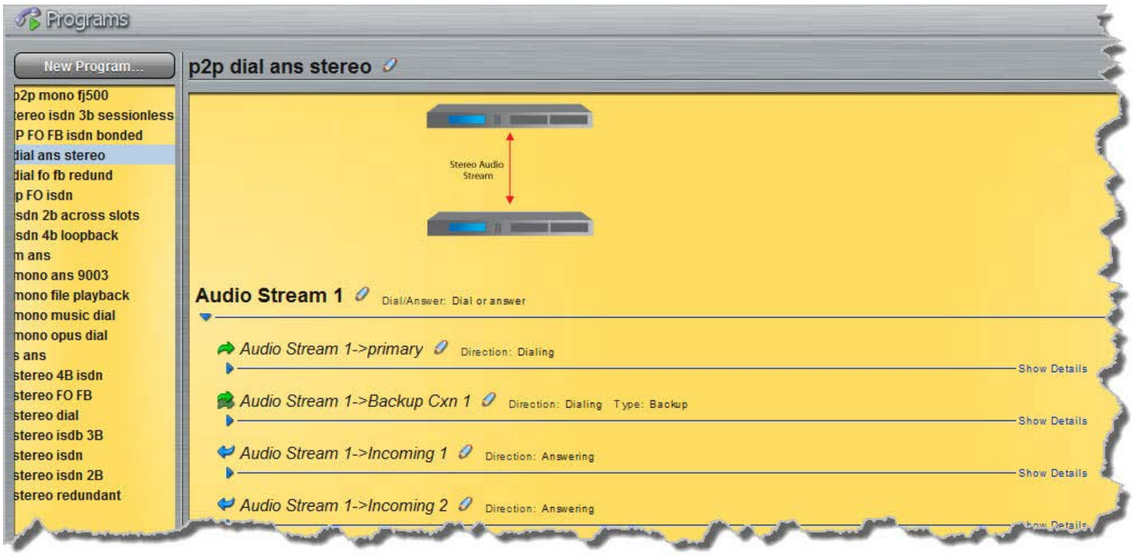
For example, if your routing requirements require the codec at the studio to always connect in mono, simply load and lock a mono program in the codec. Generally programs will be up or down-mixed by the answering codec to match the loaded program type. In some situations incompatible program types will 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 Java 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 Java Toolbox web-GUI contains a **Programs panel** with a wizard for configuring program settings and backup connections. Edit settings easily at the touch of a button and use existing programs as templates for creating other programs.

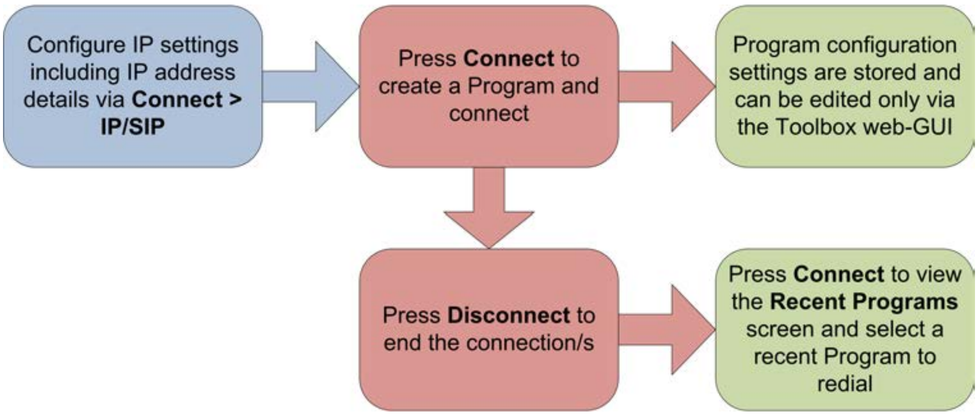
Mono and Stereo Peer-to-Peer Programs

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

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

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

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



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

16 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

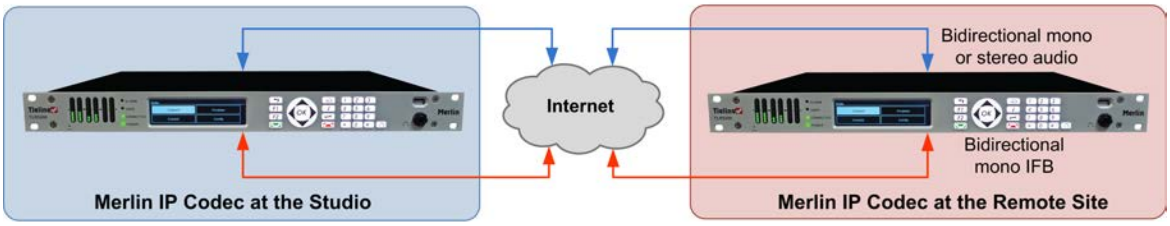
A **2 x Mono Peer-to-Peer** program includes two mono audio streams. Each audio stream includes a separate peer-to-peer connection to a different destination, which can also be configured with different transport, audio and backup settings.

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 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 or POTS 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 line numbers for dialing over ISDN or POTS to the destination codec.

Important Note: It is important to set the correct country setting for connections over POTS to adjust the POTS G5 module for varying line voltages in different countries. The country setting also affects whether G.711 µ-Law (North America/Japan) or A-Law (Europe/Australasia) coding is used over IP, SIP and ISDN connections.

18.1 Steps to Connect over IP

The following procedure will create a custom peer-to-peer connection program using the codec front panel keypad and navigation buttons. It instructs how to connect your codec over IP for the very first time without using the Toolbox Web-GUI and your computer for configuration.

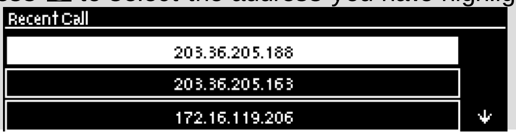
Important Notes:
- See Using the Java Toolbox Web-GUI for details on configuring connections remotely via a computer. Creation of programs is not currently supported in the HTML5 Toolbox Web-GUI.
- See Installing the Codec at the Studio for valuable information about installing your codec, negotiating firewalls and port forwarding.
- See Tips for Creating Reliable IP Connections for a range of IP information to assist with setting up IP services for your codecs.
- See Testing IP Network Connections to learn how you can test and verify the reliability of your IP connection.

1. Press 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 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.

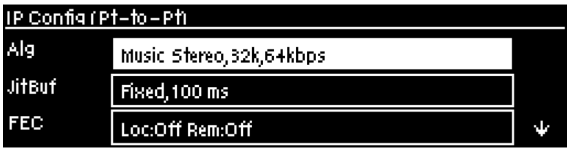
2. Press the **HOME** button to return to the **Home** screen, select **Connect > IP > Tieline** and press the button. Note: Select **SIP** or **Sessionless** instead of **Tieline** if these connections are required.

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.



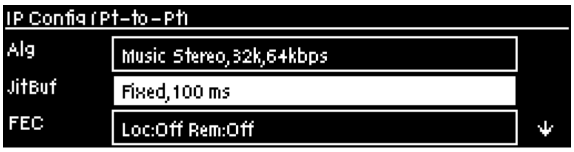
4. Navigate to **Algorithm** and press .



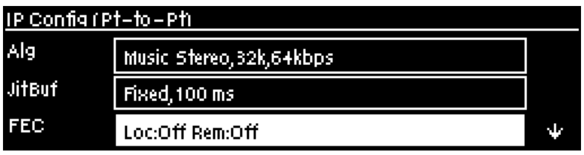
5. Use the navigation buttons to select an algorithm profile or manually enter algorithm settings, then press .

6. 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.

7. 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.



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



9. When configuration is complete press the **RETURN** button to navigate backwards to the

Connect IP screen that the IP address was entered into.

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

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

11. 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.

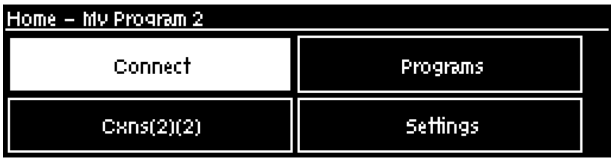
12. When dialing, the **CONNECTED LED** on the front of the unit will flash green. When connected, the **CONNECTED LED** on the front of the unit will illuminate solid green.

13. From the **Home** screen use the down navigation button to select **Cxns** and view connection **Status** and press to view connection statistics for IP packets being sent over the connection. To negotiate higher bit rates press F2 then **3** on the numeric **KEYPAD**; for lower bit rates press F2 then **9**.

18.2 Monitoring IP Connections

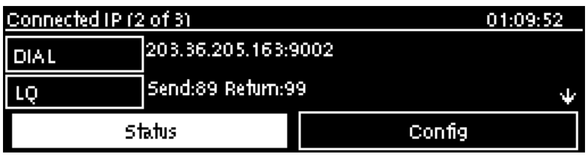
Connection Details

The number of active audio streams and connections is 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.



1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Cxns** and press the button.
3. If only a single IP audio stream is connected, the **Connected IP** screen displays details of the active connection. When multiple audio streams are connected, navigate to the one you want to view in the **Connections** screen and press the button to view more details.

For IP connections you can view the IP address dialed and the **LQ** (link quality) on the screen. 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 the IP network.



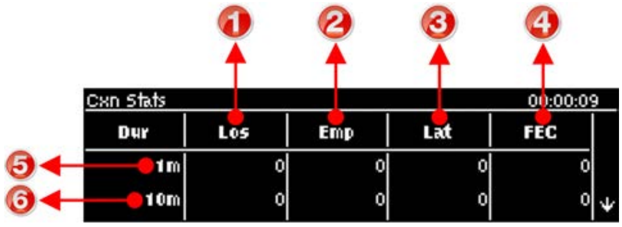
Link Quality (LQ) Readings

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

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

Viewing Connection Statistics

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



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

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

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

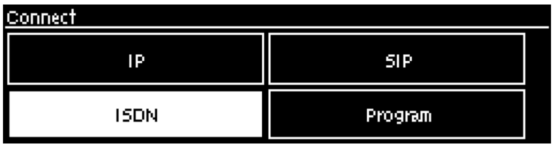
18.3 Steps to Connect over ISDN

The following procedure explains how to create a custom peer-to-peer program and dial another Tieline codec over ISDN using the front panel keypad and navigation buttons.

Important Notes:
- See Testing ISDN Connections for valuable information about setting up and maintaining reliable ISDN connections.
- 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 the Java Toolbox Web-GUI, or click here for details on using the HTML5 Toolbox Web-GUI to configure.

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.



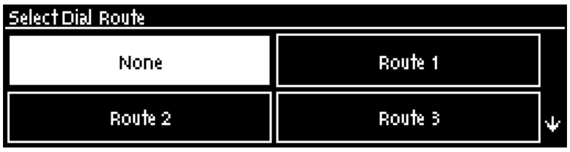
3. Navigate to **Setup** and press the OK button.

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

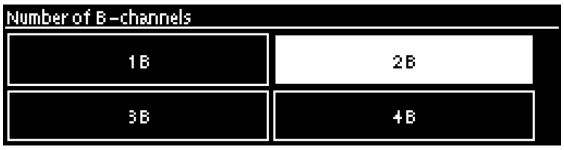
Important Note: By default, when Tieline codecs dial they send call configuration settings to the remote codec using Tieline Session Data. This configures the codec receiving the call with matching algorithm, sample rate and bit rate settings. This does not occur when dialing to non-Tieline devices,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

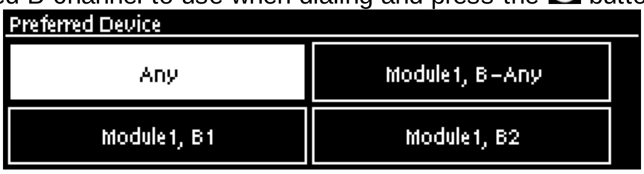


7. Select an algorithm, then press the button.

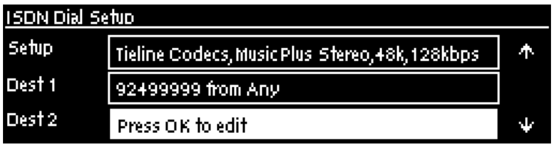
8. Select the sample rate if required, then press the button.

9. Select **Destination 1** and press the 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 button.

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

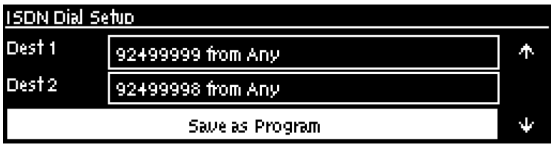


11. If you are dialing over multiple B channels to create a bonded connection select the next destination, e.g. **Destination 2**, and use the numeric **KEYPAD** to enter the next ISDN number you want to dial. Do this for all B channel destinations.



12. Navigate down to **Auto Reconnect** and press the OK button to toggle between **Enabled** and **Disabled**. Note: This is normally enabled on the dialing codec only.

13. 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 OK button.



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

15. 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 OK to connect.

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.

Important Note: 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.

18.4 Monitoring ISDN Connections

Each new audio stream connection becomes visible in the **Cxns** menu via the **Home** screen.

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Cxns** and press the button.
3. If a single ISDN audio stream is connected, the **Connected ISDN** screen will display details of the active ISDN connection. When multiple audio streams are connected, navigate to the one you want to view in the **Connections** screen, e.g. ISDN, and press the button to view more details.

18.5 Steps to Connect over POTS

The following procedure explains how to create a custom peer-to-peer program and dial another Tieline codec over POTS using the front panel keypad and navigation buttons.

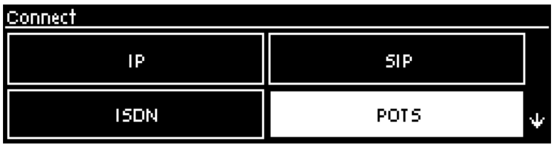
Important Notes:

- See POTS Connection Tips and Precautions for valuable information about setting up and maintaining reliable POTS connections.
- See POTS Module Settings for details on module settings.
- See POTS Answering Configuration for details on POTS answering settings (required for answering calls from non-Tieline POTS codecs)
- See Configuring POTS for details on configuring codec connections via a computer.
- The **Local** and **Remote** line quality displayed for **POTS Codec** connections is related to the actual POTS line quality at either end of the link. This reading affects the maximum allowable bit rate when the codec is training and negotiating a connection.

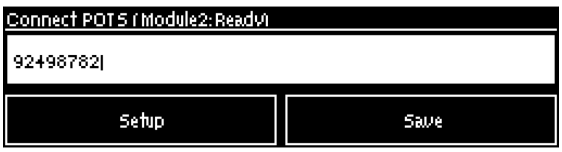
1. Press F1 and press and release the right arrow button to open the **Input Audio Level** adjustment screen.
   - Press the number on the keypad corresponding to the channel you want to toggle on or off. E.g. press 1 on the numeric keypad to toggle channel 1 on and off.

- 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.

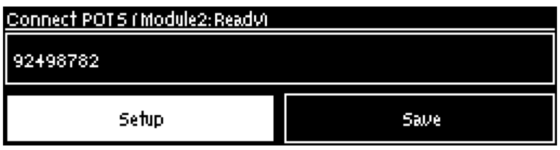
2. Press the **HOME** button to return to the **Home** screen, select **Connect > POTS** and press the OK button.



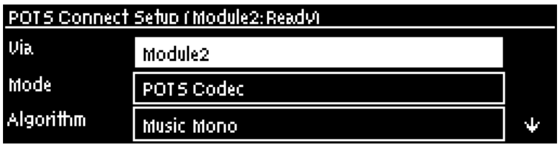
3. Use the **RETURN** button to delete any numbers if already entered, then use the numeric **KEYPAD** to enter the number you want to dial. Note: When **POTS (Ready)** is displayed throughout POTS menus it means the POTS module has initialized and is ready to accept or make a call.



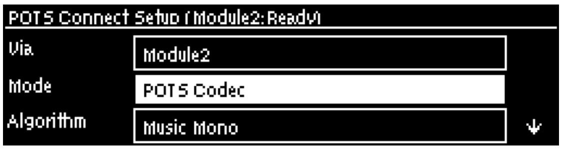
4. Navigate to **Setup** and press the OK button.



5. Select **Via** to nominate the module used when dialing a connection, or select Any to use any available module in the codec, then press the OK button.

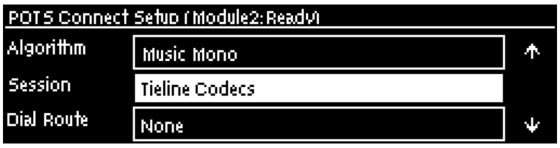


6. Navigate to the connection **Mode** and press the button to toggle between selecting either **POTS Codec** or **Analog Phone**. Note: Redundant settings in the menu will disappear If you select **Analog Phone**.



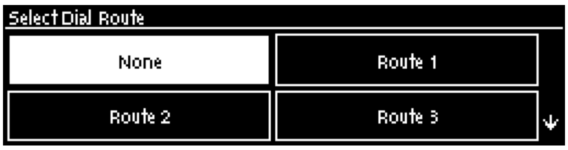
7. Select an algorithm, then press the button.

8. Select **Tieline Codecs** session data when connecting to another Tieline codec or **Sessionless** when dialing to non-Tieline POTS codecs.

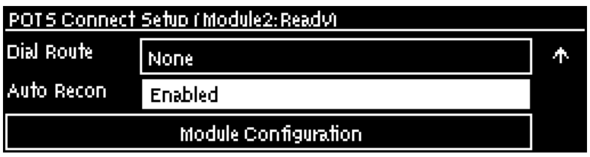


Important Notes: To dial a Comrex® Vector, Matrix® or BlueBox® codec over POTS select the **Other** algorithm and **Sessionless**. Please note that 9.6kbps connections are not supported by Comrex codecs.

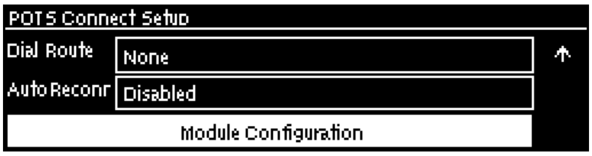
9. Select the **Dial Route** to use for this audio stream if one is required, then press the button. Note: See Configuring POTS Answering for more information on **Dial Route** and **Answer Route** tags. These can be useful when routing multiple audio streams.



10. Navigate to **Auto Reconnect** and press the button to **Enable** or **Disable** this setting as required.



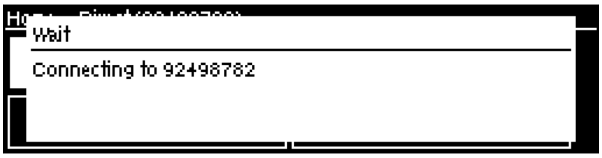
11. Select **Module Configuration** to adjust other settings specific to how you want POTS modules to dial and answer, e.g. Maximum connection bit rate and dial tone settings etc.



12.When configuration is complete press the **RETURN** button to navigate back to the **Connect POTS** screen.

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

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



Note: 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.

14.To negotiate higher bit rates press F2 then **3** on the numeric **KEYPAD**; for lower bit rates press F2 then **9**.

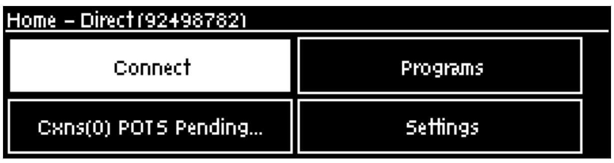
> **Important Note:** To load a saved program and dial press the **HOME** button, navigate to **Programs**, select the program you want to dial and press the **CONNECT** button to load the program and dial.

See Monitoring POTS Connections for more details on monitoring the different POTS connection states.

18.6 Monitoring POTS Connections

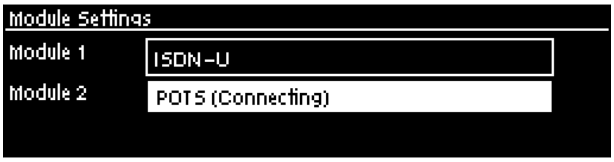
Monitoring POTS Calls when Dialing and Connecting

1. When dialing and connecting:
   - The **CONNECTED LED** on the front of the unit will flash green.
   - The **Cxns** section on the **Home** screen displays **Pending** while the call is connecting (prior to streaming audio data).



Cxns Displays Pending

   - **Connecting** is displayed in the **Modules** menu via **Settings > Modules**.



Modules Displays Connecting

2. While connecting you can also monitor dial tones and modem handshaking etc., via the left channel of the headphone output. For more details see the **Monitor Modem Tones** section in POTS Module Settings.

Monitoring POTS Calls when Connected

1. The **CONNECTED LED** on the front of the codec illuminates solid green when connected.

2. The newly connected audio stream connection becomes visible in the **Cxns** menu via the **Home** screen. To view connection details:
   a. Use the down navigation button to select **Cxns** and press the button.
   b. When multiple audio streams are connected, navigate to the one you want to view and press the button to view connection details.

The **Local** and **Remote** line quality displayed for **POTS Codec** connections is related to the actual POTS line quality at either end of the link. This reading affects the maximum allowable bit rate when the codec is training and negotiating a connection. It also indicates the stability of the connection when a call has been connected for a long period of time. If the line quality starts drop quite low after being connected for a long period, we recommend you retrain the connection to improve the line quality and avoid loss of audio.

18.7 Load and Dial Custom Programs

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

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

18.8 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.

18.9 Redialing a Connection

Press the **CONNECT** button from any codec menu to redial previous connections (except menus accessed via the **Connect** > **IP**, **ISDN** or **POTS** screens).

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 Programs** redial screen using either a previously entered name, or by a dialing address or number (manually dialed connections).

18.10 Configuring Auto Reconnect

Auto Reconnect is disabled by default. When enabled, the dialing codec attempts to reconnect if data is temporarily lost over a connection.

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.

Auto Reconnect using IP

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

Auto Reconnect using ISDN

1. Press the **HOME** button to return to the **Home** screen, select **Connect,** then select **ISDN** and press the button.
2. Navigate to **Auto Recon** and press to toggle between **Enabled** and **Disabled**.

Auto Reconnect using POTS

1. Press the **HOME** button to return to the **Home** screen, select **Connect,** then select **POTS** and press the button.
2. Select **Setup** and press .
3. Navigate to **Auto Recon** and press to toggle between **Enabled** and **Disabled**.

18.11 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 program you want to assign a speed dial number, then 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.

18.12 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.

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.

18.13 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).

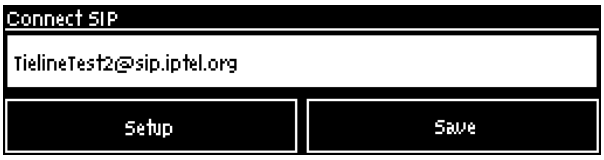
1. To dial peer-to-peer press the **HOME** button to return to the **Home** screen, select **Connect > IP > SIP**.
2. Use the numeric **KEYPAD** to enter the IP address of the codec you want to dial, using the or buttons to enter the periods in the IP address and use the **RETURN** button to delete numbers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

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

18.14 Dialing SIP Addresses

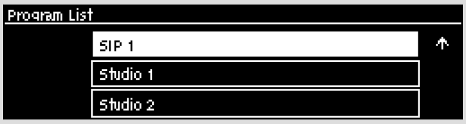
Dialing a SIP Address via the Codec Front Panel

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

18.15 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:

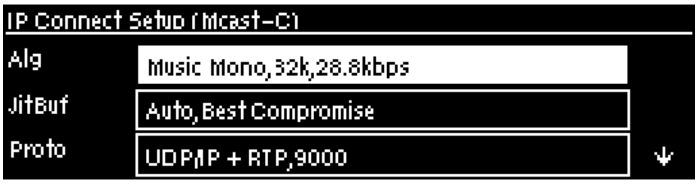
- You cannot edit a program when it is currently loaded in the codec.
- Ensure all connection related settings like the port, algorithm, bit rate (etc) match on both multicast server and client programs or they will not be able to join multicast streaming sessions.
- The default UDP audio port is 9000 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 > Sessionless** and press the button.

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

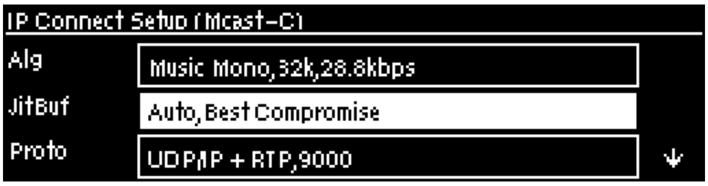
3. Use the **RETURN** button to delete any numbers already entered, then use the numeric **KEYPAD** to enter the multicast IP address you want to dial, using the or buttons to enter the periods in the IP address. The same multicast address and audio port must be used for both the server and client programs.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

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



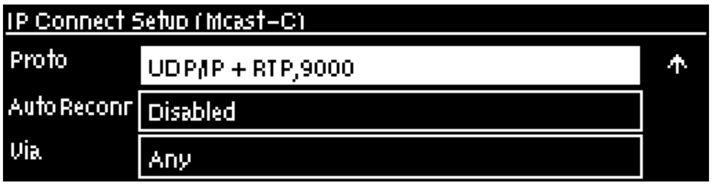
5. Use the navigation buttons to select an algorithm profile or manually choose algorithm settings, then press .

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 .

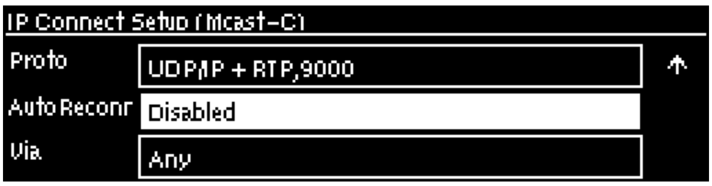


Important Notes: Automatic or fixed jitter buffer settings can be adjusted on individual client codecs as required. There is no jitter buffer setting on the server codec because it never receives audio packets.

7. Select **Protocol** to select the audio protocol and adjust the **Return 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.

Connecting a Multicast Client Program

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

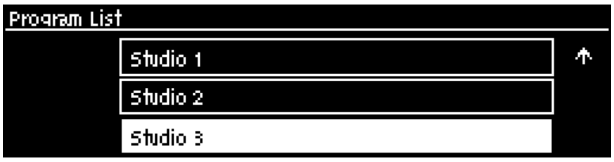
2. Select and load the multicast client program on each of the multicast client codecs and dial the multicast IP address to begin receiving multicast audio packets.

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

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.

18.16 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.



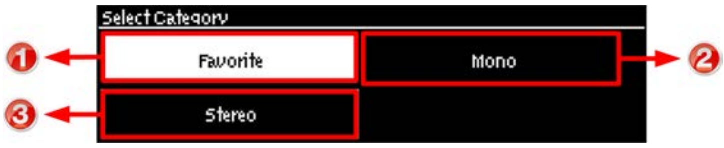
4. Navigate to **Delete** and press the button.

5. Confirm the deletion and press the button.

18.17 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 **Tieline** session mode and press the button. Note: algorithm profiles are only available for Tieline session connections.
5. Use the down navigation button to select **Setup** and press the button.
6. Select **Algorithm** and press the button.
7. Use the right navigation button to select **Profile**.
8. Choose the profile you want from the **Favorite**, **Mono or Stereo** menus.

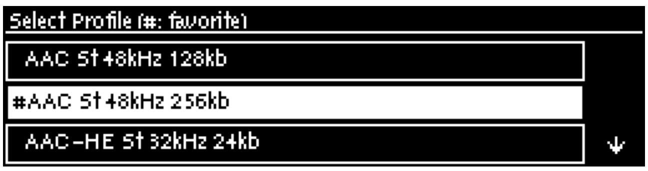


	Features	Codec Home Screen Elements
1	Favorite	Displays a list of favorite profiles that have been selected manually within the codec by users
2	Mono	Displays preprogrammed mono profiles within the codec
3	Stereo	Displays preprogrammed stereo profiles within the codec

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 Session** mode and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Press the button to select **Algorithm**.
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. Select your preferred **IP Session** mode and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Press the button to select **Algorithm**.
7. Use the right navigation button to select **Profile**.
8. Select the profile you want to delete from the **Favorite** menus.
9. Press the hatch (pound) button to delete the selected profile from the favorite menu.

18.18 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

18.19 Merlin Backup Options

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

Tieline Audio Codec Backup Features			
Backup Option	**Transport: IP, ISDN or POTS**	**Time Required to Respond**	**How to Enable**
SmartStream PLUS	IP Only (Note: concurrent packet stream sent; codec detects IP packet loss or delayed packets)	No time delay - simultaneous dual redundant streaming	Enabled in dialing codec program; configures local decoding, or remote decoding via session data
On-demand (cold) Failover	All transports (Note: codec detects loss of data or connection and redials the backup connection)	User configurable detection parameters during program configuration*. Delay is equal to detection time plus the time required to dial the alternative connection	Dialing codec program monitors streaming and manages failover
FEC (Forward Error Correction)	IP Only (Note: decoding codec detects IP packet loss or delayed packets)	No time delay - packet replacement occurs in real-time	Dialing codec configures local and remote FEC settings via session data transfer when connecting
Auto Reconnect	All transports; codec will redial continuously to try and reconnect	Immediately redials after loss of IP stream detected	Enabled in dialing codec program

* Note: POTS can take up to 60 seconds to connect successfully.

Important Note: Failover and SmartStream PLUS redundant streaming are not available with SIP or sessionless IP connections.

SmartStream PLUS Redundant 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.

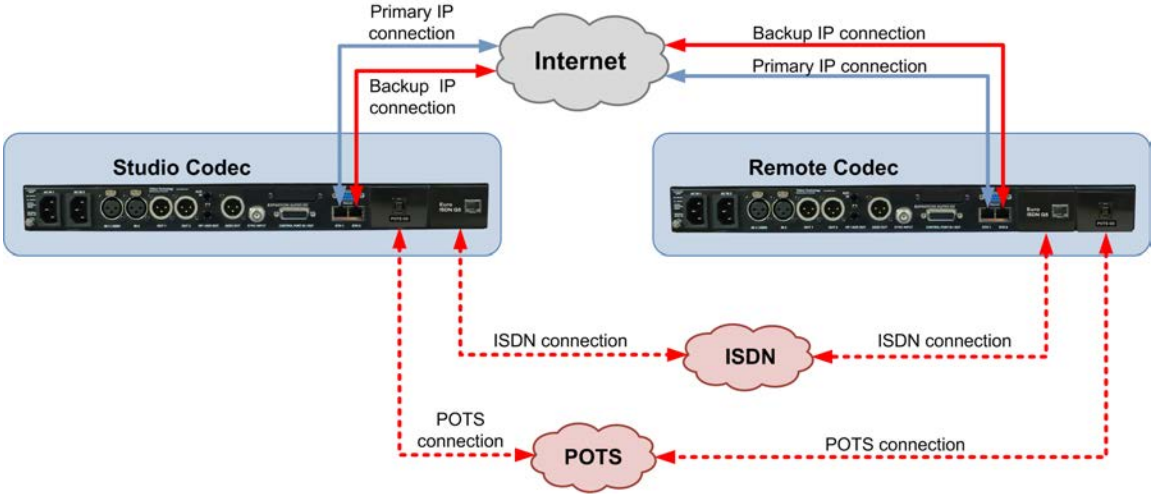
1. The codec can stream simultaneous redundant data streams from both Ethernet ports and deliver seamless redundancy by switching back and forth, without loss of audio, from the nominated primary data link to the backup link if one fails and then subsequently recovers. Use IP links from two different 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. See the procedures for configuring different programs using the web-GUI for more configuration details.

On-Demand Failover (IP, ISDN or POTS)

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

The codec can be configured to switch to a backup connection over IP, ISDN or POTS as required. For example, you can create a program with IP as the primary connection and also create a backup ISDN or POTS connection in the same program. For details on configuring backup connections using failover see Configuring Merlin Point-to-Point Programs.



Forward Error Correction (FEC)

FEC transmits a secondary stream of audio data packets over a single connection. If packets are lost or corrupted over the connection then replacement FEC data packets can be substituted to replace them.

Note: FEC should not be confused with SmartStream PLUS. FEC packets are sent over a single data stream connection, whereas SmartStream PLUS dual redundant streaming transmits two completely redundant audio data streams. FEC is also a subset of features within SmartStream PLUS, which means you can configure SmartStream PLUS dual redundant data streams and also configure FEC on each of these data streams. For more info on FEC see Configuring Forward Error Correction.

Auto Reconnect

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

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

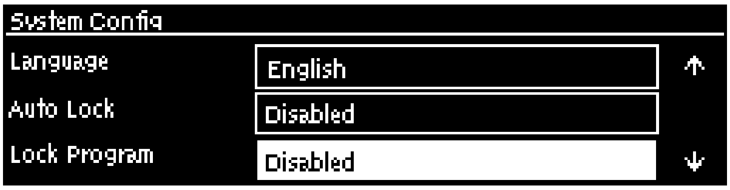
Auto reconnect can be enabled when configuring a codec program designed to dial another codec or codecs. See the procedures for configuring different programs using the web-GUI for more configuration details.

18.20 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. On the answering codec, you may wish to configure the codec to always use a particular jitter buffer or FEC setting.

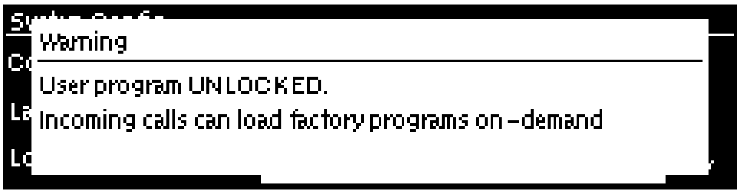
Generally programs will be up or down-mixed by the answering codec to match the loaded program type. In some situations incompatible program types will be rejected. A compatible program type can still connect and specify different connection parameters such as algorithm preferences and bit rates via 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.

18.21 Locking the Front Panel

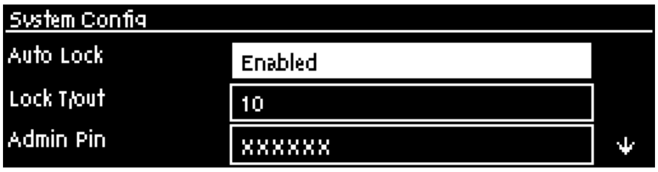
The codec features a front panel lock feature for tamper-proof operation. This feature is disabled by default.

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

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

Enabling the Front Panel Lock Feature

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



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

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

19 Connecting to the ToolBox Web-GUI

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

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

About the Java Toolbox Web-GUI

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

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

Java Toolbox Web-GUI Prerequisites

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

About the HTML5 Toolbox Web-GUI

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

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

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

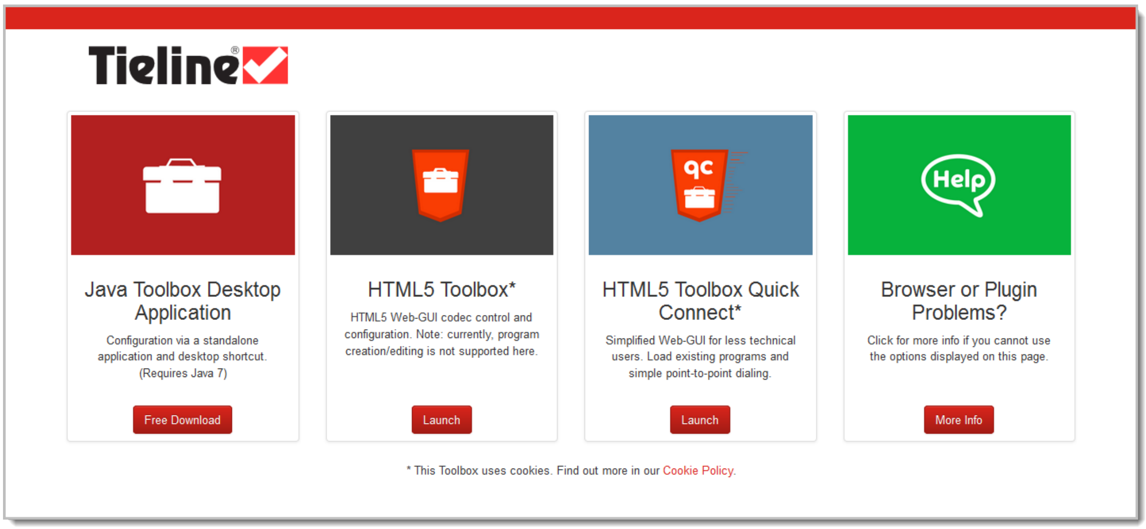
- Dial and hangup existing programs only (currently program creation wizard is not available).
- Quick connect option available for simple peer-to-peer connections: configure IP, SIP, ISDN and POTS connections including the algorithm, sample rate, bit-rate and dial number/address.
- Extensive command and control of codec settings.

About the HTML5 Toolbox Quick Connect

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

19.1 Opening the Java or HTML5 Web-GUI & Login

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



Launching the Java Toolbox Web-GUI

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

Desktop Icon

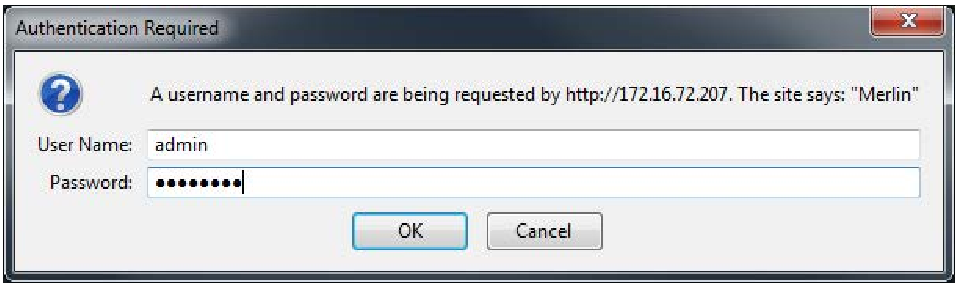
2. When you launch Toolbox an authentication dialog prompts you to enter a password to login. The first time you log in you can enter the default setting "**password**" and click the **OK** button. Tieline highly recommends you click the hyperlink in the login dialog or visit Changing the Default Password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

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

Launching the HTML5 Toolbox Web-GUI

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

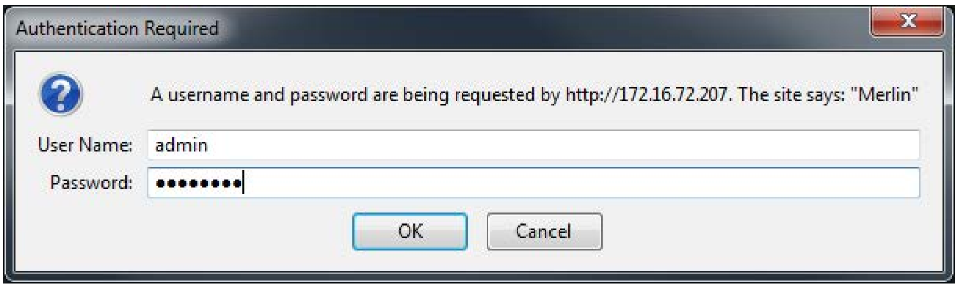
2. When you launch Toolbox for the first time an authentication dialog prompts you to enter the user name "**admin**" and password "**password**" to login, then click the **OK** button. Tieline highly recommends you change the password (see Changing the Default Password). This will provide better network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



Launching the HTML5 Toolbox Quick Connect

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

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



Using the Web-GUI over the Internet

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

LAN Troubleshooting

PC LAN Settings

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

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

Port Selection

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

E.g. **192.168.0.176:8080**

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

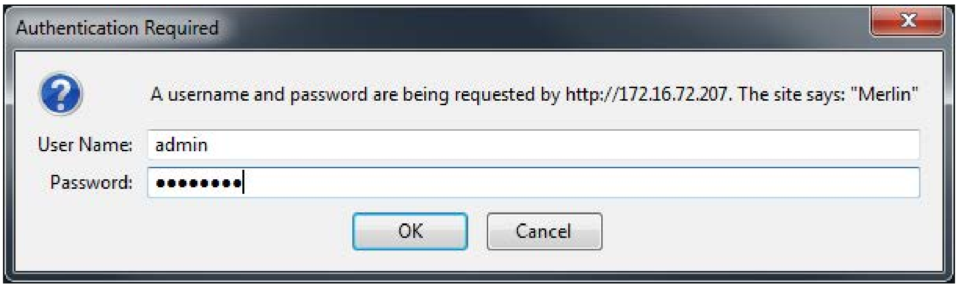
1. Press the **HOME** button on the codec to return to the **Home** screen.
2. Use the navigation buttons to select **Settings** and press the button.
3. Use the navigation button to navigate down to **WebGUI** and press the button.
4. Select **Alt. Port** and press .
5. Use the **KEYPAD** to enter a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

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

19.2 Changing the Default Password

The default password for the Toolbox Web-GUI is **password**. Enter this in the authentication dialog to use the Web-GUI initially and then Tieline highly recommends changing the default password to protect your codec from being tampered with during live broadcasts.

Toolbox Java Web-GUI Login Dialog



Toolbox HTML5 Web-GUI Login Dialog on a Merlin Codec

Caution: Codecs connected to the internet can be accessed by anyone with knowledge of the codec's public IP address. Setting a strong password protects your equipment from being tampered with 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 button.
3. Select **Password** and press .
4. Use the **KEYPAD** to enter a new password and press the button to save the new setting (Note: there is no character limit for passwords).

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

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

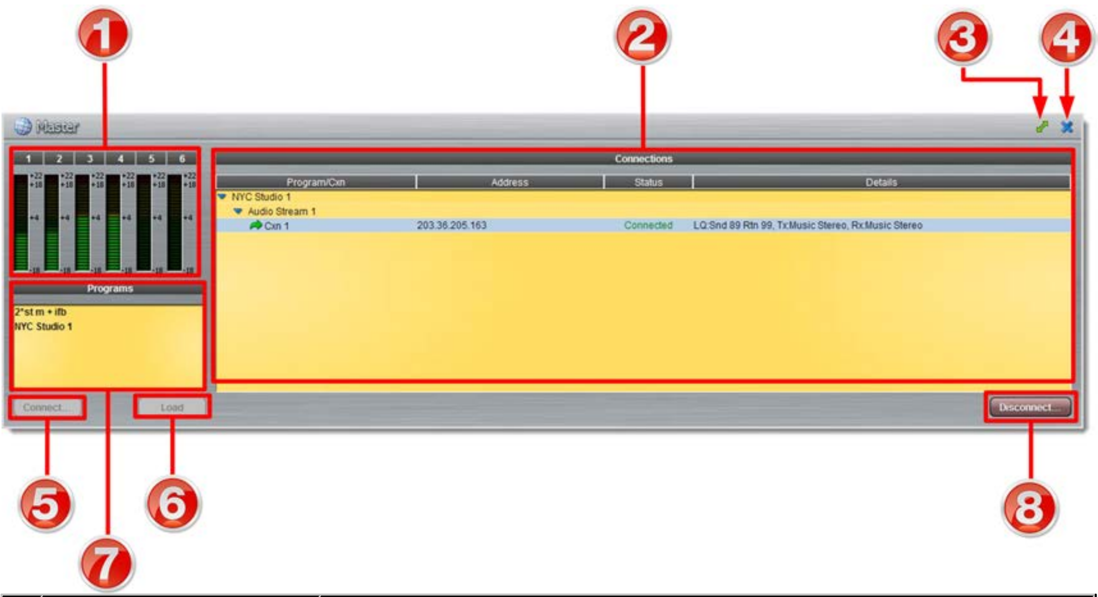
20 Using the Java Toolbox Web-GUI

The following sections provide an overview of the different programming panels available within the JavaToolbox 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

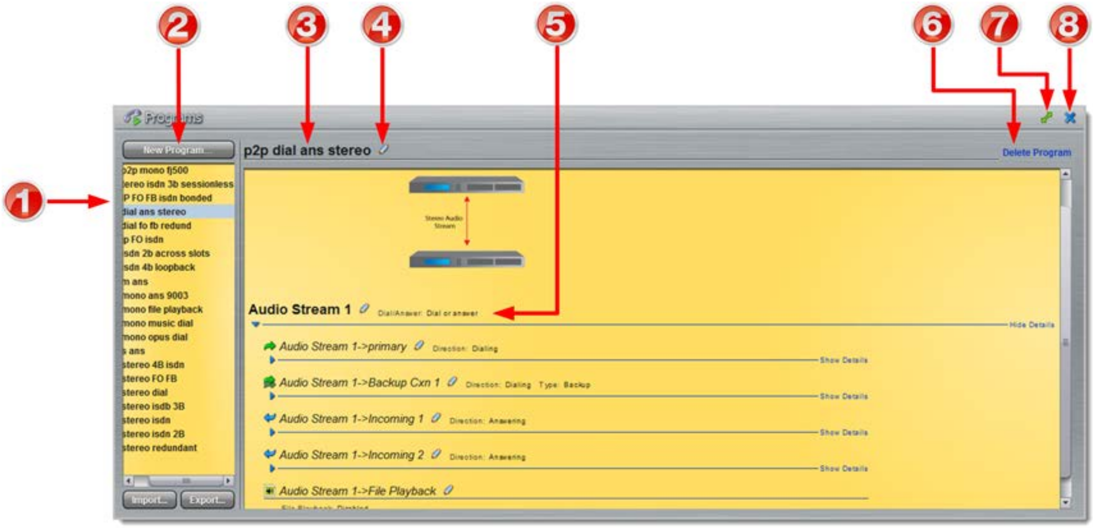
The most recently opened panel is displayed underneath the **Master panel** by default. Click the **Maximize/Minimize** symbol to view a panel in full-screen mode, or click to minimize back to the default panel size.

Master Panel to Load Programs and Connect Audio Streams



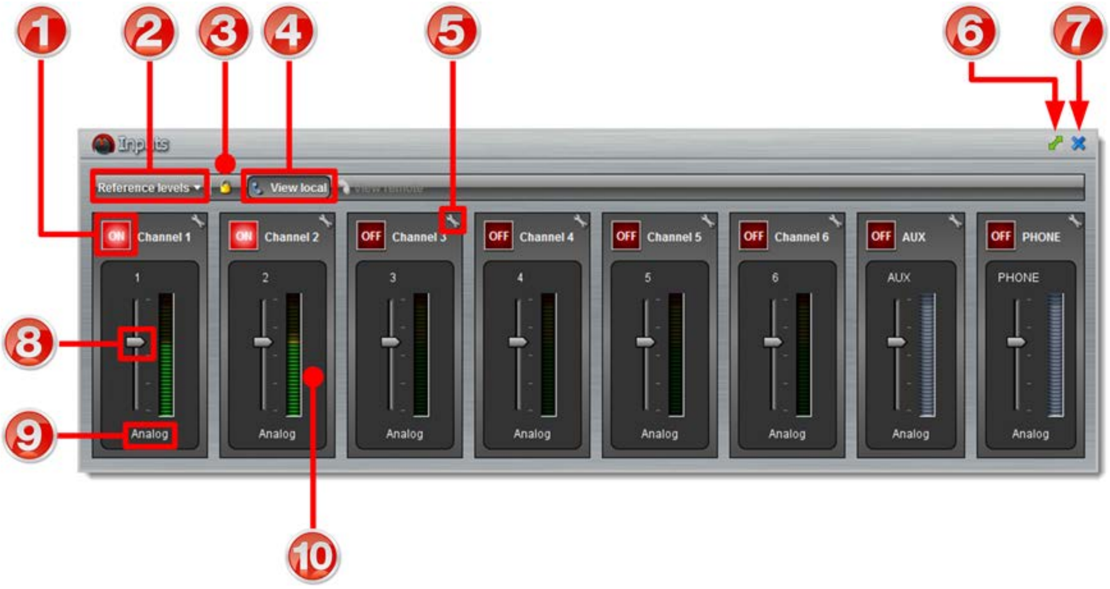
	Feature	Description
1	Input/Output PPMs	6 PPM meters to display audio levels for inputs and outputs
2	**Connections**	Provides a summary of connection details and audio streams
3	Locked program	Symbol indicates the currently loaded program is locked
4	Maximize/Minimize	Click to maximize a panel to view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

Programs Panel for Connection Configuration



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

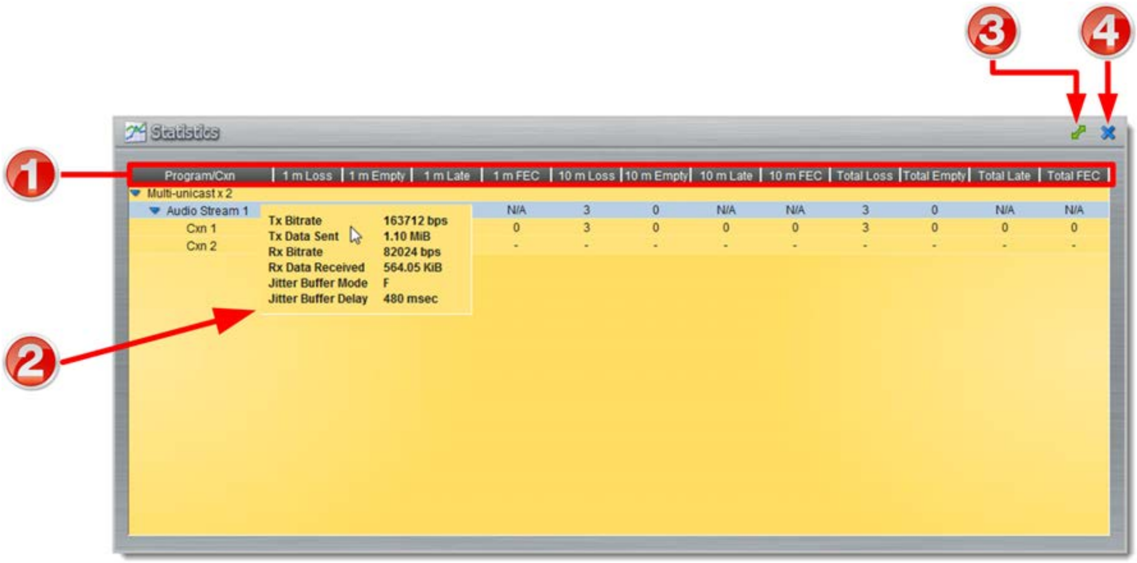
Inputs Panel for Input Adjustments



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

Statistics Panel for Monitoring Connection Stability

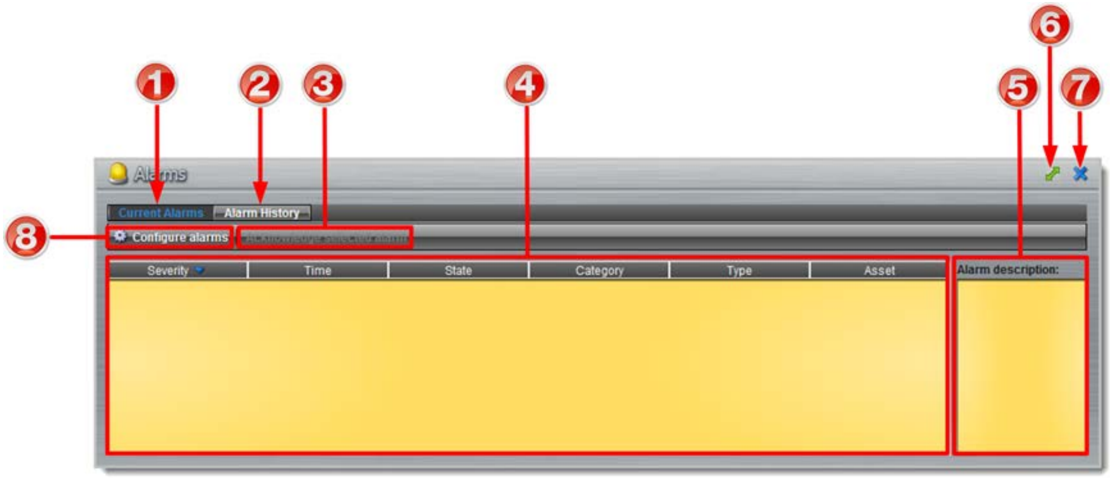


	Feature	Description
1	Headings	Headings for the various packet arrival statistics available
2	Connection Statistics	Right-click to view audio stream bit-rate and jitter buffer statistics
3	Maximize/Minimize	Click to maximize a panel to view it in full-screen mode, or click to minimize back to the default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

Rules Panel for Creating Relay Activation Rules

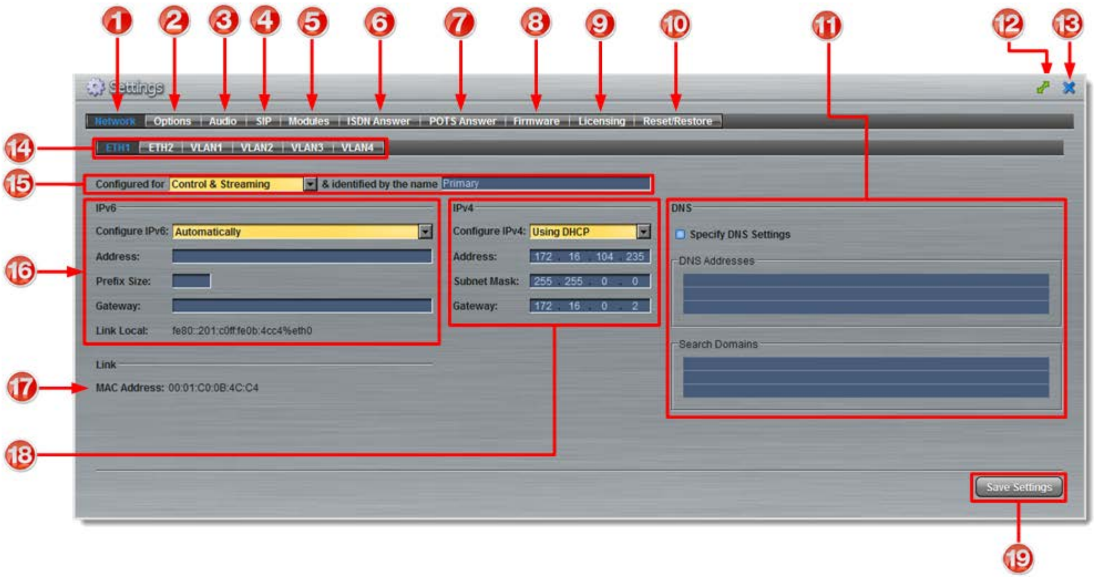
	Rule	Description
1	Connect and disconnect a program when an input is toggled	Click to configure Connection and Disconnection by toggling an input
2	Connect when an input is switched **ON**; Disconnect when another input is switched **ON**	Click to configure Connection and Disconnection after different relay inputs are switched **ON**
3	Connect and Disconnect on Audio Detection	Click to configure the codec to connect when audio is detected and disconnect when silence is detected
4	Synchronise a local relay input with a remote relay output	Click to configure a local relay input to synchronise with the state of a remote relay output
5	Toggle a relay based on a connection's status	Click to configure a relay to toggle based on connection status
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 **Rules** panel

Alarms Panel



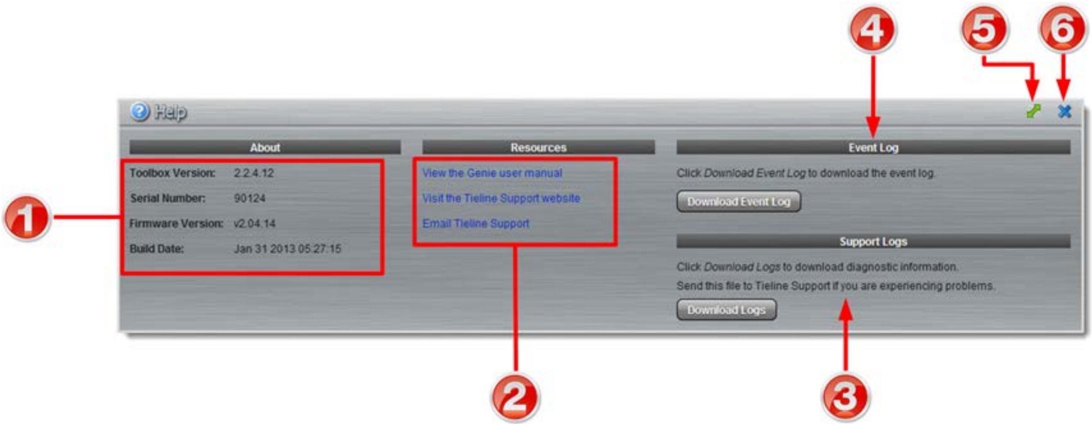
	Feature	Description
1	**Current Alarms**	Click to view current device alarms
2	**Alarm History**	Click to view the history of device alarms
3	**Acknowledge Selected Alarm**	Click to acknowledge an alarm after activation
4	**Alarm details** pane	Displays alarm details
5	**Alarm description** pane	Troubleshooting 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.

Panel



	Feature	Description
1	**Network tab**	Click to edit or view codec network configuration settings
2	**Options tab**	Click to configure RS232 and QoS data settings, lock a loaded user Program and adjust Session Port settings 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	**POTS Answer tab**	Click to configure POTS Answering settings
8	**Firmware tab**	Click to view software versions and perform an upgrade
9	**Licensing tab**	Click to select a license file and install it into the codec
10	**Reset/Restore tab**	Click to reset codec default settings and perform backup/restore of codec programs and settings
11	**DNS** Pane	Activate to specify DNS addresses and domains to search.
12	Maximize/Minimize	Click to maximize a panel to view it in full-screen mode, or click to minimize back to the default panel size
13	**Close** button	Click to close the panel
14	Network Interface	Select a network interface for configuration options
15	Network Interface Identifier	Control and streaming configuration options for each network interface, e.g. Ethernet Port 1 or 2.
16	**IPv6** details	IPv6 addressing details and configuration
17	**MAC Address**	Device MAC address
18	**IPv4** details	IPv4 addressing details and configuration
19	**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.

20.1 Configuring IP Settings

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

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

IPv4 versus IPv6

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

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

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

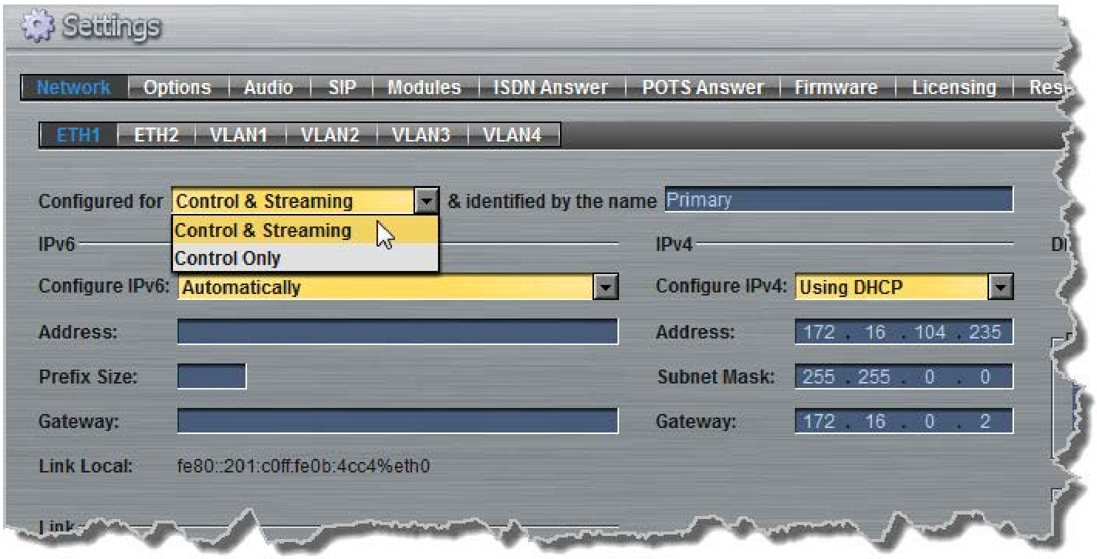
Configuring Ethernet Ports and VLANs

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

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

As an example, if only one physical Ethernet interface is available, VLANs can be used to operate SmartStream PLUS or to separate codec Control and Streaming functions if required. Ethernet and VLAN interfaces can be configured for:

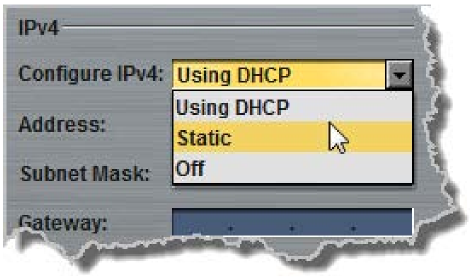
- Controlling audio: codec control and command only from the Ethernet port.
- Controlling and Streaming: stream audio and control and command the codec via the Ethernet port.
- Streaming audio: stream audio only from an Ethernet port (**ETH2** and **VLANs** only).
- Nothing: Disable the Ethernet port from streaming audio and codec command and control (**VLANs** only).



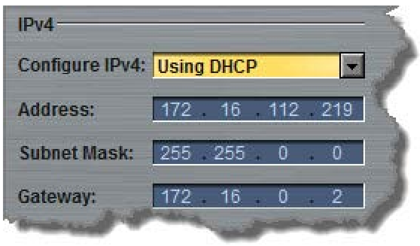
The name entered into the right-hand text box, e.g. **Primary** or **Secondary**, is an interface identifier used when configuring new programs via the **Programs panel**.

IPv4 Address Configuration

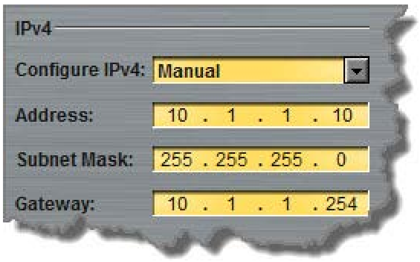
The codec is capable of automatic DHCP address assignment, or manually configured static IPv4 address configuration via the drop-down **Configure IPv4** menu. If you want to ignore IPv4 settings select **Off**.



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

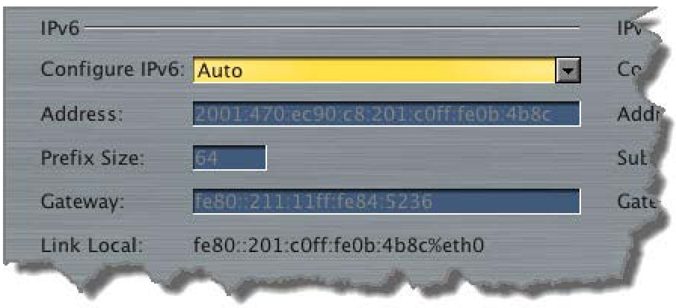
Types of IPv6 Addresses

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.

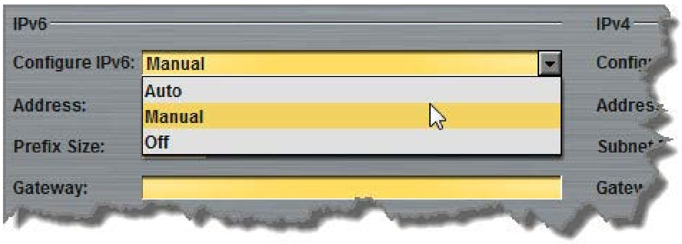
Auto Address Assignment

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



Manual IPv6 Address Assignment

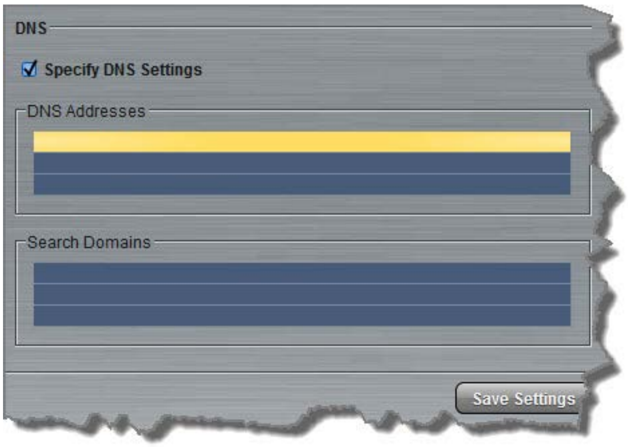
To configure IPv6 address details into the codec manually, select **Manual** and enter details into the **Address, Prefix** and **Gateway** text boxes.



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

Specifying DNS Settings

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



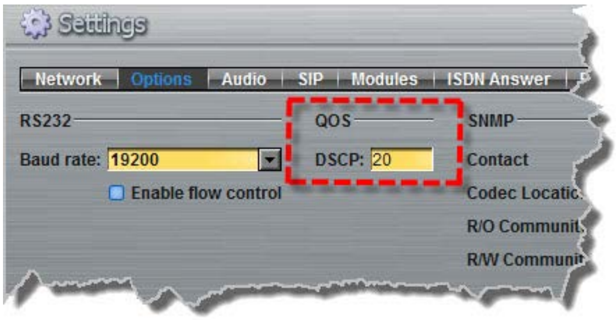
The codec can be configured to tag IP data packets sent across a network by entering a value into the Differentiated Services Code Point (DSCP) field within the header of data packets transmitted over the network.

Configuring QoS

1. Open the Java Toolbox 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.

20.2 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). See About ISDN Modules for additional information on ISDN.

You can use the Java Toolbox Web-GUI to configure a dial and/or answer program with ISDN settings. You may also need to:

1. Configure ISDN module settings.
2. Configure ISDN Answering settings.

20.2.1 Configuring ISDN Modules

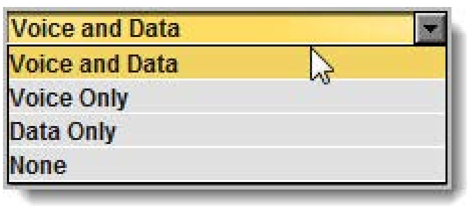
ISDN settings in the **Module** menu determine how each codec module operates at a particular site. You can copy similar programs between codecs installed at different locations and also configure site-specific settings for how each ISDN module should connect. ISDN module settings may need to be adjusted depending on your country and network requirements.

1. Open the Java Toolbox 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. Select Module 1 or Module 2.

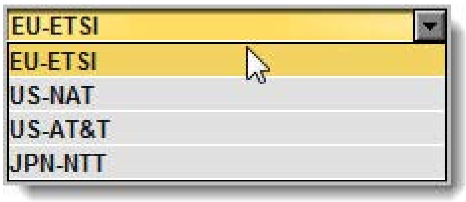
4. Click the drop down arrow for **Accept** and select whether to allow or disallow circuit switched voice or data calls. The default setting allows **Data Only**.



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

5. 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).



6. 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.

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

8. 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.

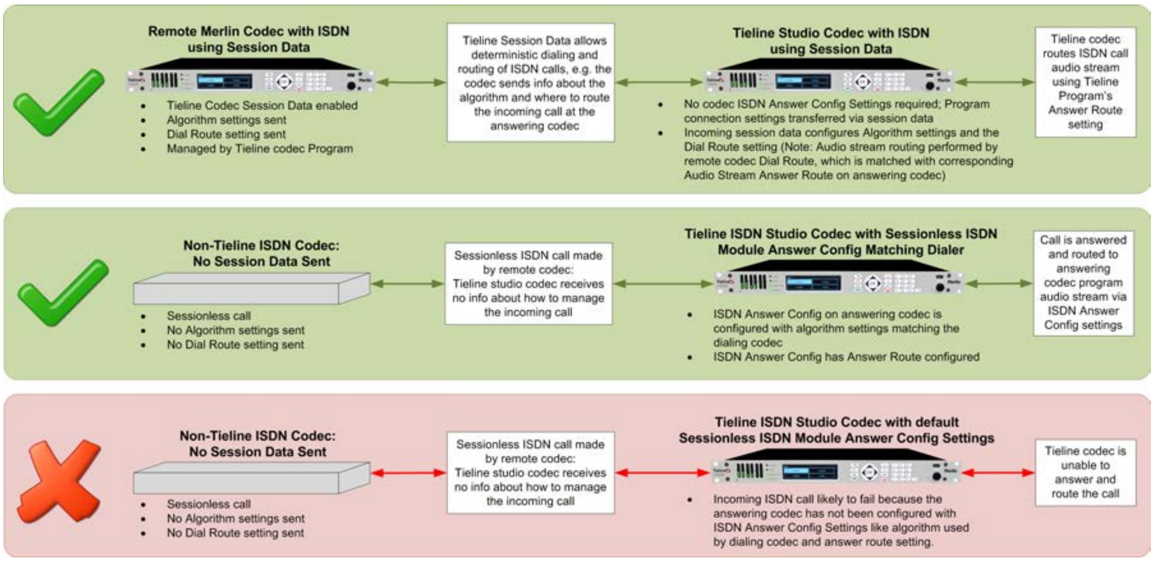
Typically, each ISDN BRI service in the US will have two SPIDs and these must be entered correctly. When you enter a SPID into your codec and connect it to an ISDN line, an initialization and identification process takes place, whereby the terminating equipment (your codec) sends the SPID to the switch.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

20.2.2 Configuring ISDN Answering

ISDN Answer Configs are used to determine how codec ISDN modules will behave when answering ISDN calls.

The following image explains the difference between answering calls from Tieline codecs sending session data, and non-Tieline codecs making sessionless ISDN calls. Codecs sending Tieline Session Data contain all the information required to connect, e.g. algorithm and audio stream routing settings. When answering sessionless calls it is necessary to configure the answering codec with an **ISDN Answer Config**, which tells the answering codec how a sessionless call will try and connect.



It is possible to save up to four different **ISDN Answer Configs**, which allow up to 4 ISDN B channels to be individually configured for unique answering behaviors. ISDN answering can be configured to suit:

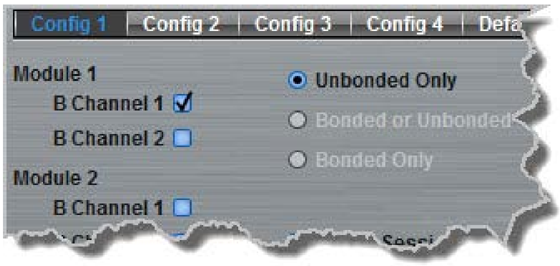
- Hardware available in the codec, i.e. the number of B channels available.
- Expected dialing behaviors, e.g. if B channels should bond or not, and whether audio streams need to use **Dial** and **Answer Route** tags.
- The type of call being received by the codec, e.g. Tieline (with Tieline Session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

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**.

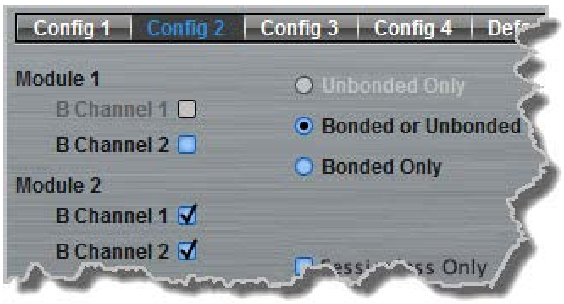
Single B Channel Config

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.



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**.

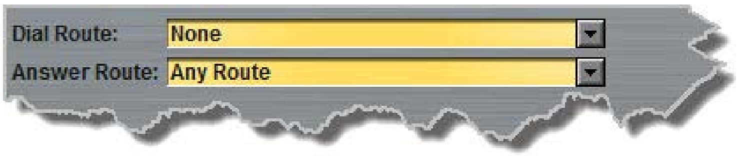


Configure the bonding setting that best suits the audio stream with which this **Config** will be associated. **Bonded or Unbonded** is the best setting in most situations. Note: Click the **Save settings** button to apply changes to the **Config**.

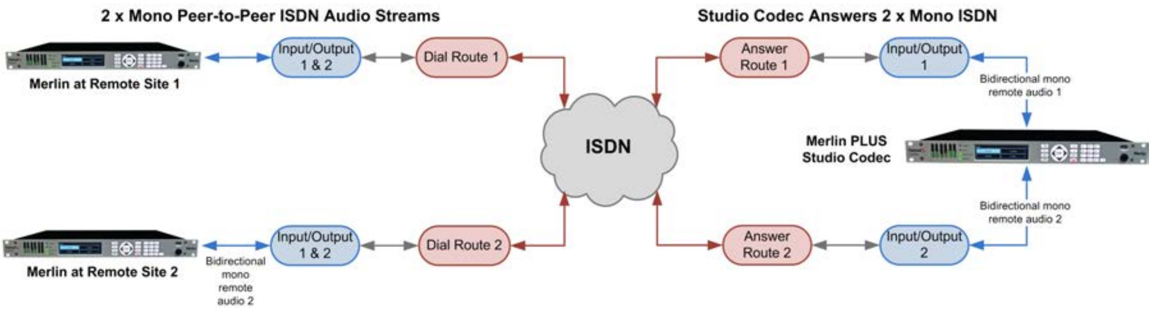
Bonding Setting	Behavior
Unbonded Only	Unbonded single B Channel
Bonded or Unbonded (May Bond)	Calls using the same algorithm from the same Tieline codec, or sessionless calls, will attempt to bond when received. Calls using incompatible algorithms will not be bonded
Bonded Only	Will only bond compatible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

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. 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.



In principle, the concept of 'routes' 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 codec audio streams, and therefore the same inputs and outputs. Following is an example of how to consistently route incoming ISDN audio streams using dial and answer routes.

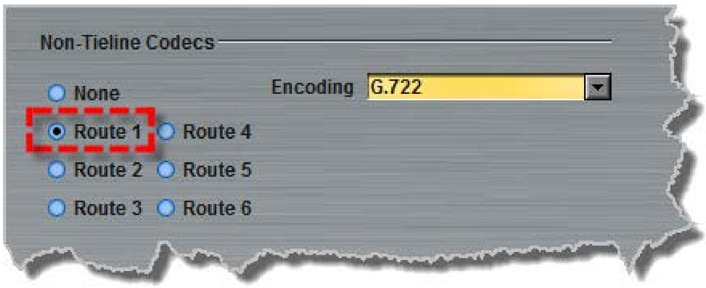


Answer Routes for Non-Tieline (Sessionless) ISDN Calls

In some situations you may receive a call from a non-Tieline codec which doesn't support session data and **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.

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. Click **Save Settings** to store the new **Config** settings.

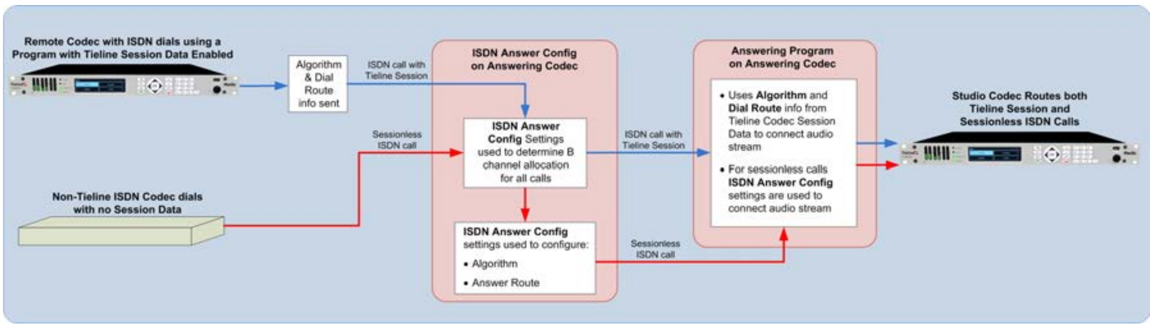
3. This will associate the incoming call with a corresponding **Answer Route** configured in the answering codec program, e.g. **Answer Route 1**.

More detailed information about how to configure the codec to answer and route multiple sessionless ISDN calls is available in Using ISDN Answer Routes for Sessionless ISDN Calls. This uses examples to explain how to set up consistent deterministic routing of multiple incoming sessionless calls.

Answering both Tieline Session and Sessionless ISDN Calls

Leave the **Sessionless Only** checkbox in the **ISDN Answering Config** unchecked if the codec is expected to receive ISDN calls from Tieline codecs, or both Tieline and non-Tieline codecs (i.e. you are not sure which type of codec may call). In this mode, when the codec answers a call it initially expects to receive Tieline session data from the dialing codec and configure its own algorithm settings according to that. If it fails to receive Tieline session data within 5 seconds (i.e. a non-Tieline codec is calling, or a Tieline codec with session data disabled), it will use the settings in the **ISDN Answering Config** instead.

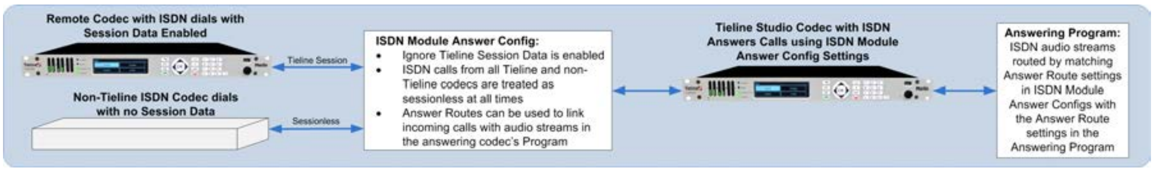
The following image displays how the answering codec will behave in this mode when receiving calls from both Tieline and non-Tieline codecs.



Allow Answering of Sessionless ISDN Calls Only

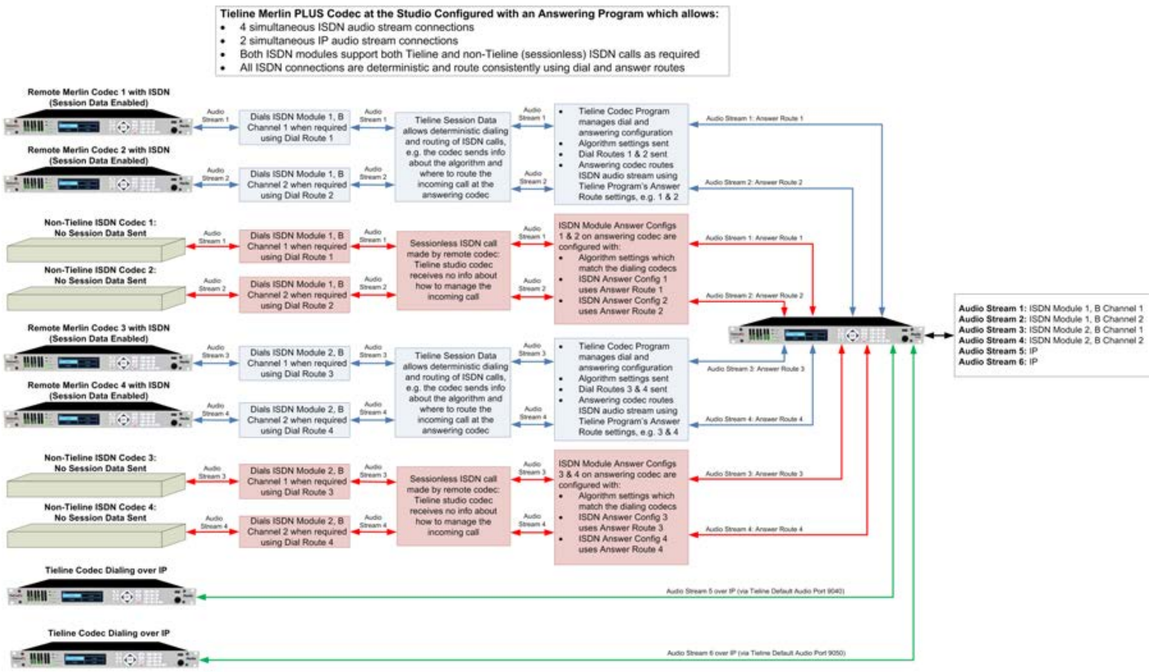
Select **Sessionless Only** when answering ISDN calls from non-Tieline codecs only. When **Sessionless Only** is selected, the codec will not wait to receive the Tieline session data. This reduces the time taken to answer an inbound sessionless call.

The following image displays how the answering codec will respond with **Sessionless Only** selected, i.e. calls from both Tieline and non-Tieline codecs are always regarded as sessionless.



Answering Multiple ISDN Calls from Tieline and non-Tieline Codecs

Tieline codecs capable of answering multiple incoming audio streams can be configured to answer both Tieline session data and sessionless ISDN calls at different times. They can also support connections using other transports such as IP or POTS. The following example shows how a Tieline codec can be configured to answer up to 4 separate mono ISDN calls at different times from both Tieline and non-Tieline codecs, as well as two mono IP audio streams.



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

20.3 Configuring POTS

Two slots are available for inserting optional POTS modules into the codec. These can be configured using the codec front panel or the Toolbox graphical user interface (GUI). See About POTS Modules for additional information on POTS.

You can use the Web-GUI to configure a dial and/or answer program with POTS settings. You may also need to:

1. Configure POTS module settings.
2. Configure POTS Answering settings.

20.3.1 Configuring POTS Modules

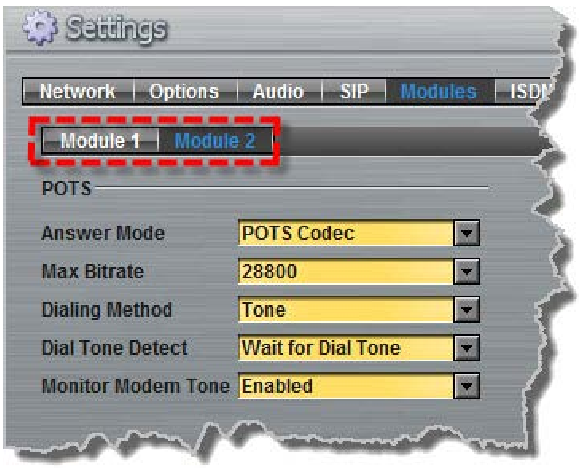
POTS settings in the **Module** menu determine how your codec will connect at a particular site. You can copy similar programs between codecs installed at different locations and also configure site-specific settings for how each module should connect. The default **Config** settings for POTS modules are designed to suit Tieline codecs. These settings will need to be adjusted to connect to non-Tieline POTS codecs or connect in **Analog Phone** mode.

Configuring POTS G5 Modules

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

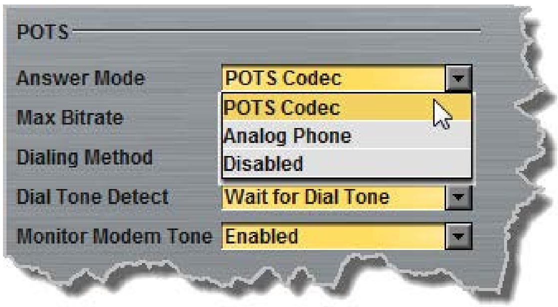
2. Click the **Modules** button at the top of the **Settings panel**.

3. Select **Module 1** or **Module 2**.



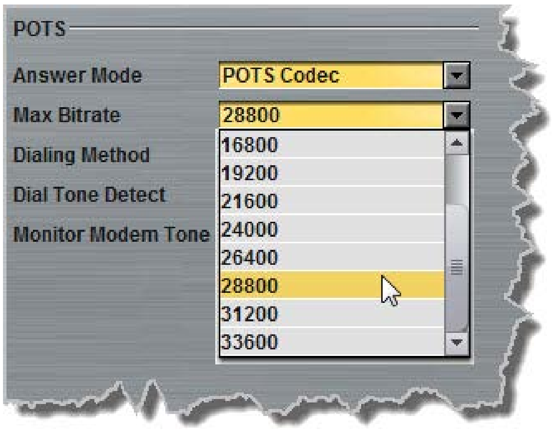
4. Click the drop down arrow to adjust the **Answer Mode** and select how the module in the codec will be able to answer incoming POTS calls. Options include:

- **POTS Codec:** allows the POTS G5 module to receive incoming audio data over a POTS line.
- **Analog Phone:** configures the POTS G5 module to receive a standard analog phone call.
- **Disabled:** disables the POTS G5 module from receiving a **POTS Codec** or **Analog Phone** call.



Calls are answered based on the **POTS Answer settings** in **Config 1** & **2**. Adjustments to these **Config** settings are not normally necessary when connecting between Tieline codecs. They are normally adjusted when connecting to non-Tieline codecs over POTS (see Configuring POTS Answering for more info).

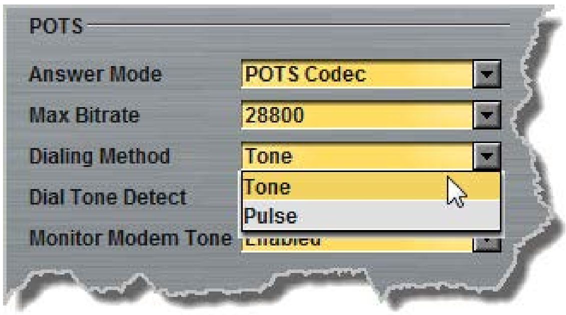
5. Click the **Max Bitrate** drop-down arrow to adjust the maximum bit rate (dialing and answering). The default setting is **28800** (28.8kbps) and this only affects **POTS Codec** calls. The range of the setting is 9.6kbps to 33.6kbps. Even if the line is capable of establishing a connection at a higher bit rate, the **Max Bitrate** setting is the highest bit rate that will be attempted. Reducing this value can improve connection reliability on poor quality lines. If two codecs are not configured the same, they will attempt to connect at the lowest of the two **Max Bit rate** settings.



Important Note: G5 POTS modems initially attempt to establish a link at the lowest **Max Bitrate** setting configured in the two modules being connected. If the POTS line doesn't support this bit rate, the modems will attempt to connect at the highest possible bit rate to suit the prevailing line quality at each end of the link.

Dialing Method (Dialing only)

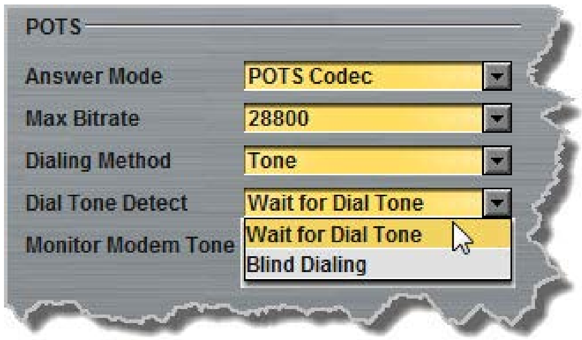
Click the drop-down arrow for **Dialing Method** to select **Tone** (DTMF) or **Pulse** dialing over POTS Codec connections. Tone dialing is used always when the **Answer Mode** is **Analog Phone**.



Dial Tone Detect (Dialing only)

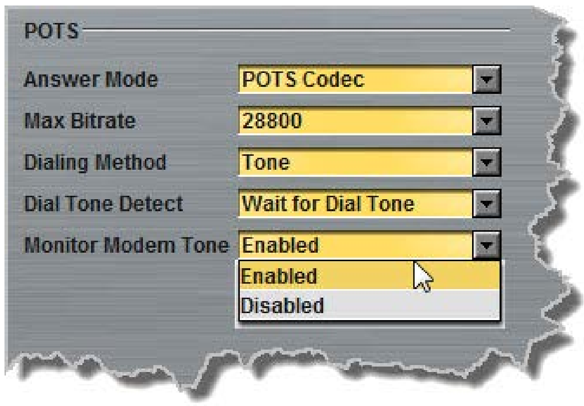
Click the drop-down arrow for Dial Tone detect to select either:

- **Dial Tone Detect**: The module will only be allowed to dial when a dial tone is present on the line.
- **Blind Dialing**: Allows the module to dial when no dial tone is present.



Monitor Modem Tone (Dialing and Answering)

Click the drop-down arrow for **Monitor Modem Tone** to select either **Enabled** or **Disabled**.



When enabled the module will allow audio monitoring of modem tones during connection in **POTS Codec** mode via the phone input. By default, the following phone input monitoring rules apply when multiple POTS G5 modules are installed in a codec and multiple POTS connections are dialed.

Module 1	Module 2	Audio Rule
POTS Codec (Monitor Modem Tone)	POTS Codec (Monitor Modem Tone)	The phone input receives a mix of modem tone audio from both modules
POTS Codec (Monitor Modem Tone)	Analog Phone	The phone input receives analog phone input audio only and mutes modem tone monitoring
Analog Phone	Analog Phone	The phone input receives audio from the oldest active connection only

Important Notes:
- Modem tone monitoring will work even if **Phone Input Enable** is **Off** via **Settings > Audio > Phone Input > Phone Input Enable [Off]**.
- Modem tone monitoring is only enabled during the initial connection training and negotiation period in **POTS Codec** mode.
- The monitoring volume can be adjusted using the codec front panel via **Settings > Audio > Phone Input > Level**, or by opening the **Inputs panel** in the Web-GUI and adjusting the **Phone** input volume slider.

Country

This displays the current country setting in the codec. To adjust this setting select **Settings > System > Country**.

20.3.2 Configuring POTS Answering

It is possible to store a different **POTS Answer Config** for each POTS module installed in the codec. POTS answering can be configured to suit:

- The type of call being made, e.g. Tieline (with Tieline Session Data) versus non-Tieline (Sessionless).
- Expected dialing behaviors and encoding, e.g. whether audio streams use **Route** tags and which algorithm is used.

If you answer a call from a non-Tieline codec you will need to create an answering "Config" to determine which module in the codec will answer the call and the settings used when connecting.

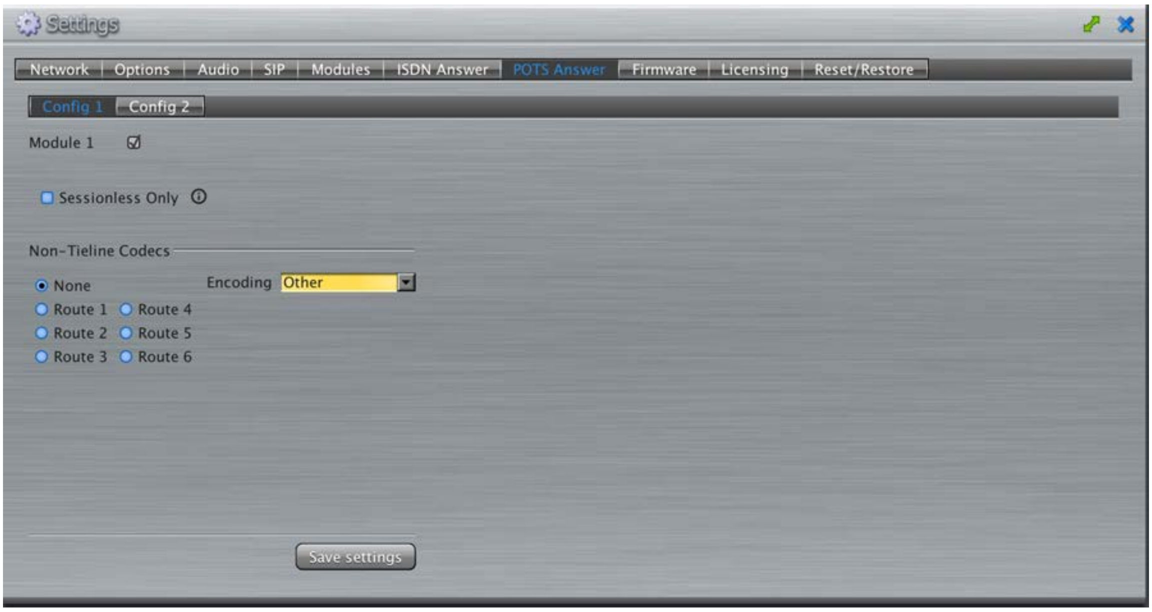
Important Notes:
- **POTS Answer Config** settings are applied to **POTS Codec** connections and not **Analog Phone** connections.
- When receiving a call from a Tieline codec with session data enabled (i.e. not **Sessionless**), the algorithm setting from the dialing codec overrides the setting in the **POTS Answer Config** menu.

POTS Config Settings

The default **POTS Answer** module **Config** settings are:

- **Tieline Codecs** Session Data,
- The **Other** algorithm.

This configuration will accept the settings from an incoming Tieline codec when it dials with session data enabled. It will also allow the codec to answer a call from a Comrex POTS codec supporting the **Other** algorithm setting.



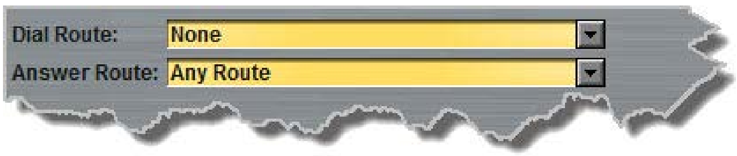
Answering Calls from Non-Tieline POTS Codecs

Select the **Sessionless Only** check-box when only non-Tieline codecs are dialing a Tieline codec over POTS. 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 if required.

Important Note: Select **Other** in the **Encoding** drop-down menu when connecting to Comrex® Vector, Matrix® and BlueBox® codecs. On the Comrex codec select its "Music" algorithm. Please note that 9.6kbps connections are not supported by the Comrex codecs.

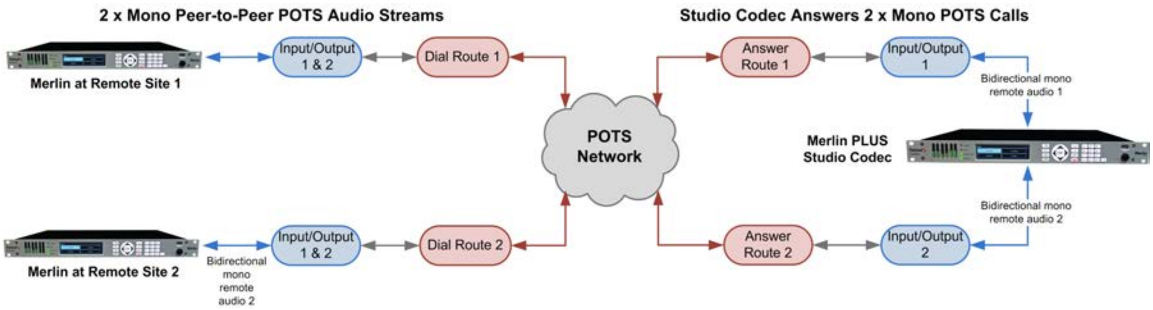
Dial and Answer Route Settings in Programs

Dial Route and **Answer Route** tags allow you to associate a POTS **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 POTS audio stream configurations, however it is very useful in multiple audio stream codecs which support POTS connections. When dialing Tieline to Tieline over POTS 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.

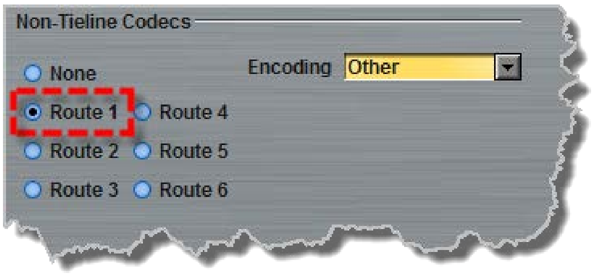


Answer Routes for Non-Tieline POTS Codecs

In some situations you may receive a call from a non-Tieline POTS 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 or 2** in **POTS Answer**. You can also select the default algorithm.

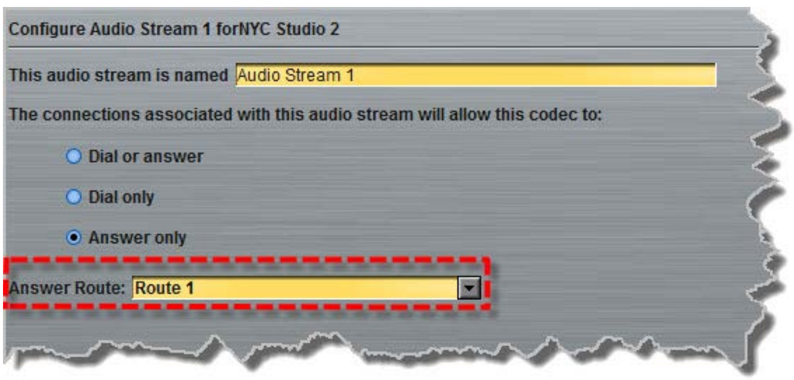
For example, if a call from a non-Tieline codec is received via POTS **Module 1** (i.e. no **Dial Route** has been specified in the dialing codec):

1. Select an answering **Route** for this POTS module in one of the two **Configs** within **POTS Answer**, e.g. **Route1**, then select the default **Encoding** algorithm to use when connecting (Note: **Other** is used for connecting to Comrex POTS codecs).



2. Click **Save Settings** to store the new **Config** settings.

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



20.4 Configuring Input/Output Settings

Click the **Inputs** button to view input controls available within the Java 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

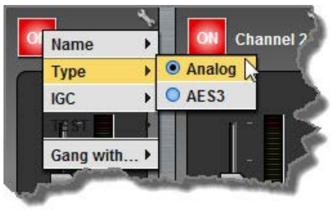
Renaming Input Channels:

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.

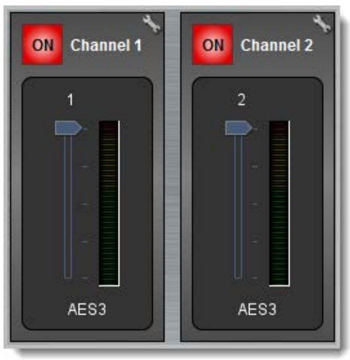
Selecting Analog and Digital Audio Sources:

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 Audio for more information about the digital inputs and outputs.

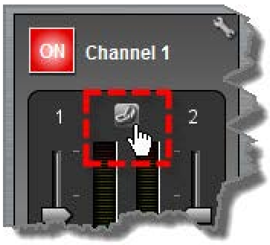
Ganging Channels:

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.

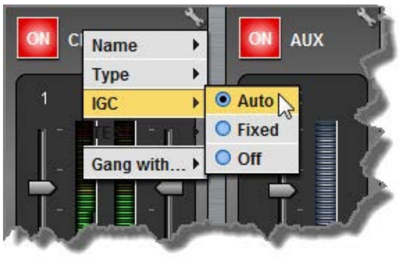


Setting Analog Audio Levels

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. Note: this lock function does not affect the codec front panel controls.

AES3 Output Sample Rate Configuration

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

1. Open the Java Toolbox 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**.

20.5 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.

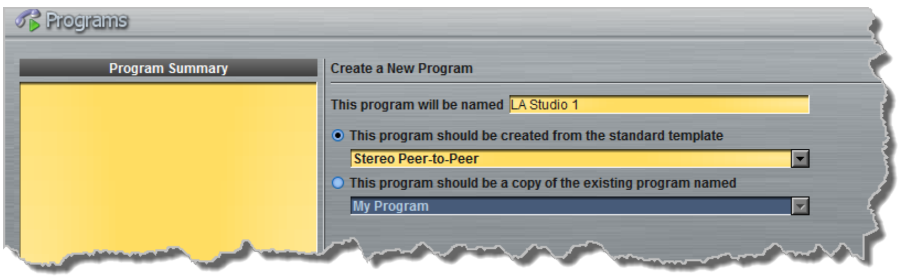
Configuring Merlin Peer-to-Peer Programs: Dialing

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 cannot be unloaded by a codec dialing in with a different type of program.
- Some drop-down menus and settings may be greyed out intentionally depending on features available 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.
- Failover and SmartStream PLUS redundant streaming are not available with SIP or sessionless IP connections.
- POTS is not supported for stereo audio stream connections.
- To learn more about programs see the section titled About Program Dialing.

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

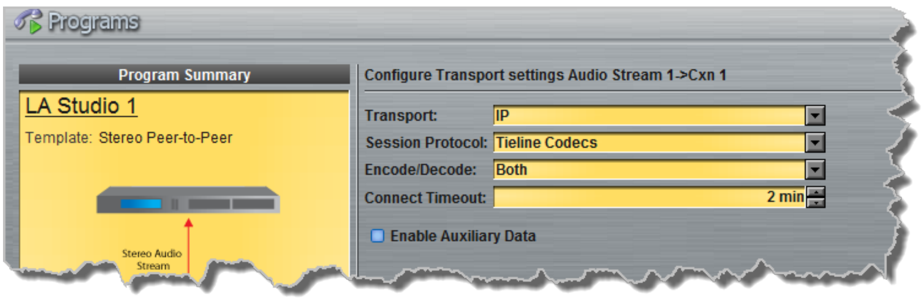
3. Enter a name for the **Audio Stream** and configure the codec to dial, answer or dial and answer.

Then click **Next**.

It is also possible to select a **Dial Route** or **Answer Route** if required. When routing multiple audio streams over transports like ISDN or POTS, you can use **Dial** and **Answer Routes** to configure deterministic routing of audio streams. Use of **Dial** and **Answer Routes** is not recommended over IP. See Configuring ISDN Answering or Configuring POTS 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**.

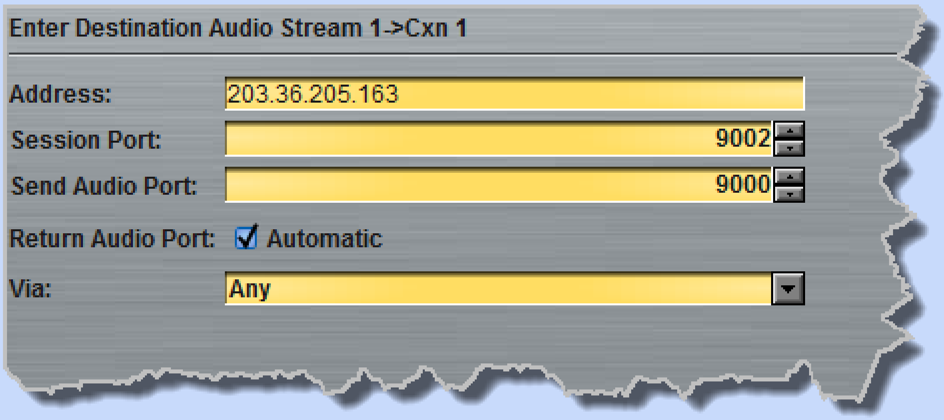
5. Follow the instructions on the right-hand side of the panel to configure the transport settings for the connection, then click **Next**.



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

6. Configure destination codec dialing and encoding settings:

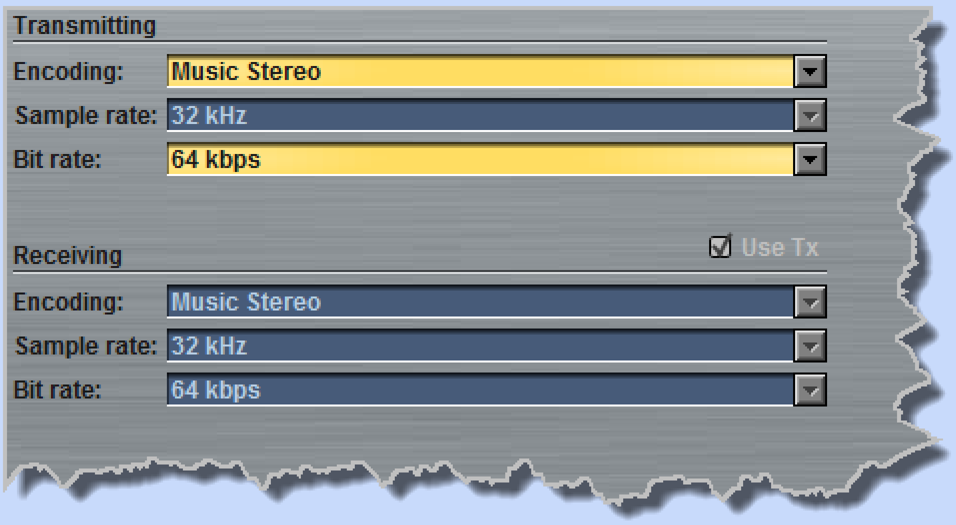
IP For IP connections configure the IP address, ports, and then specify which streaming interface is used to dial this connection, e.g. **Primary** (port **ETH1**) or **Secondary** (port **ETH2**). Note: By default **Any** will select **ETH1** if it is available and **ETH2** if it is unavailable.



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

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

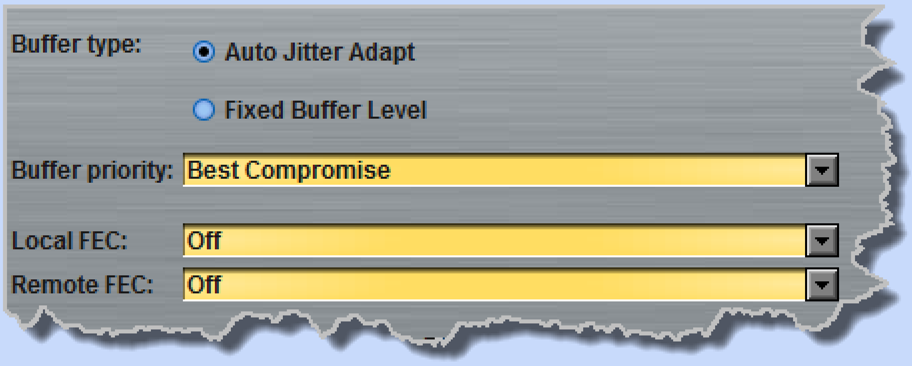
Click the drop-down arrows on the right-hand side of each text box to adjust the **Encoding**, **Sample rate** and **Bit rate** options.



For IP connections click to configure:

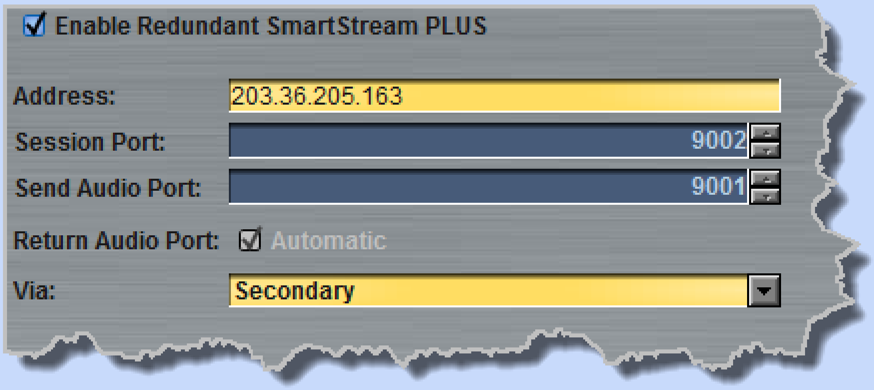
- **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**, or

- **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 in case one IP connection fails. Alternatively, click **Next** to configure **Auto Reconnect** or a backup connection, whereby the alternative connection is dialed if the primary connection fails.

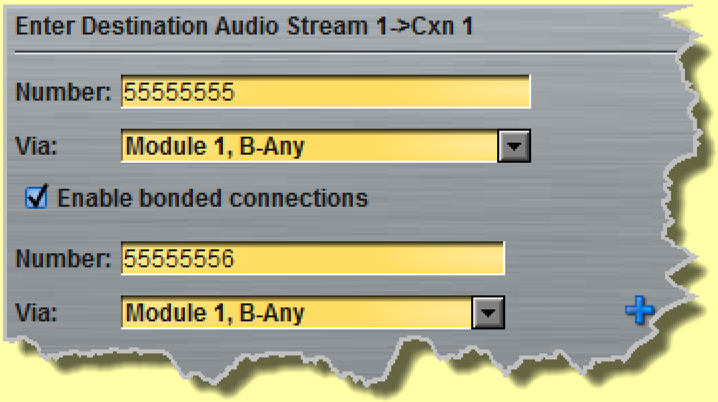
By default, primary IP streaming is via **ETH1**. To achieve the maximum level of redundancy select **Secondary** to configure redundant streaming from the secondary IP port **ETH2**. The redundant stream uses **Send Audio Port 9001** by default and the **Return Audio Port** allocated is **Automatic**. Note: **Automatic** indicates that the codec will arbitrarily allocate the return port value and send this information to the codec to which you are dialing.



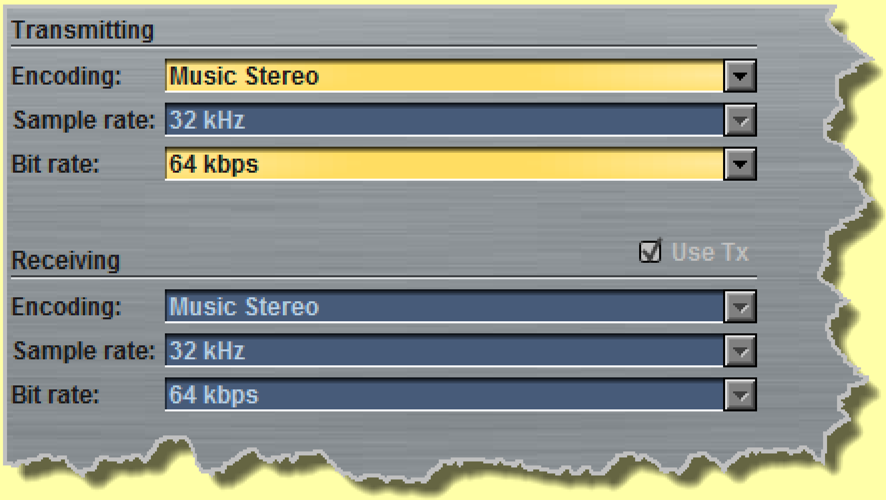
Important Note: Dual SmartStream PLUS redundant streaming over both Ethernet ports mitigates lost packets on either link and will provide IP network backup if an IP link is lost. To learn more about SmartStream PLUS redundant IP streaming see http://www.tieline.com/Transports/SmartStream-IP

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.



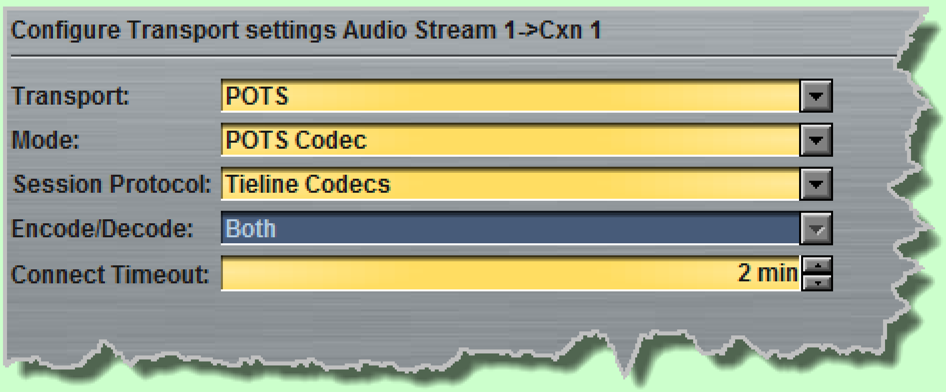
Next, click **Save Program** to save the program with default algorithm settings, or click **Next** to specify a different algorithm and configure a backup connection if required. (recommended).



Dialing settings for this ISDN audio stream are now complete.

POTS

Select **POTS Codec** in the **Mode** drop-down menu to encode/decode using POTS, or select **Analog Phone** to configure a standard analog phone call, then click **Next**.



Next, enter the phone number of the codec or device you want to dial. When multiple POTS modules are installed, click the **Via** drop-down menu and select **Module 1** or **Module 2 to** specify which POTS module will dial. Next, click **Save Program** to save the program with default settings, or click **Next** to specify algorithm settings and configure a backup connection if required (recommended).

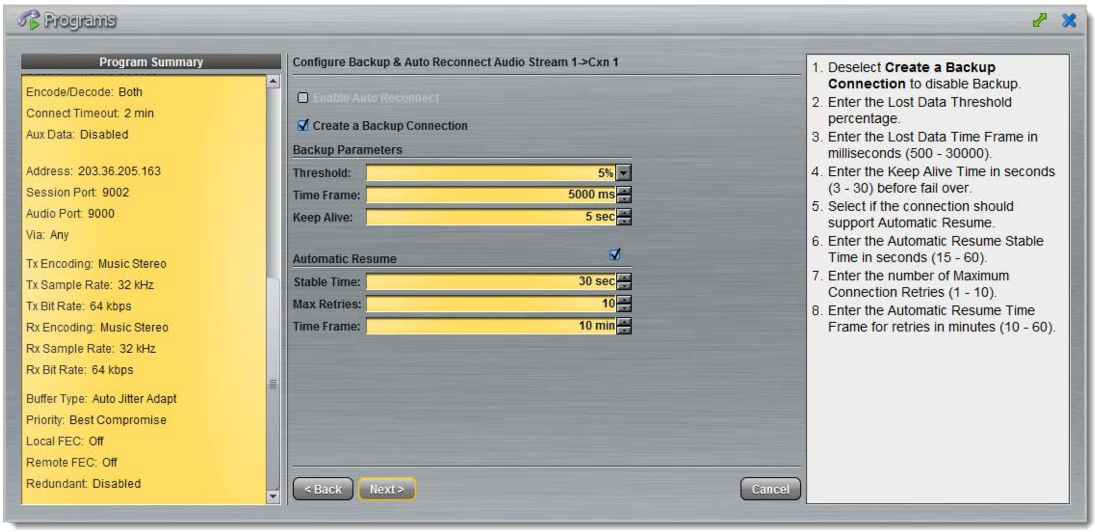
Dialing settings for this POTS audio stream are now complete.

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.

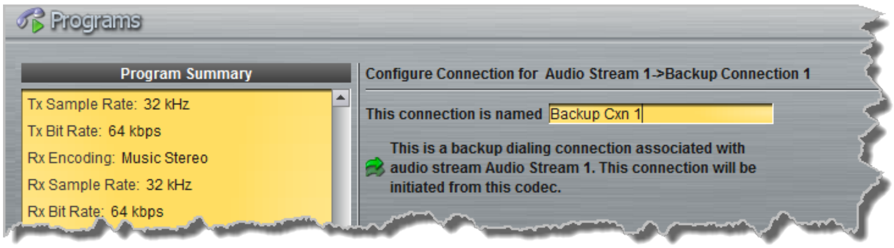
1. Click to select the check-box for **Create a Backup Connection**. Adjust the parameters and click **Next**.



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**.

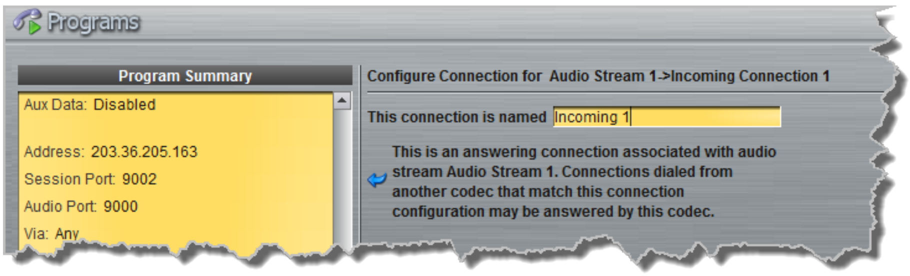


3. Click **Next** to continue through the wizard and configure the backup connection in a similar manner to how you configured the primary connection.

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:

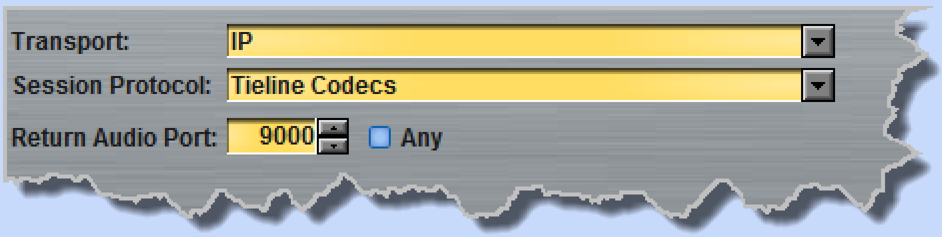
1. Enter a name for the answering connection and click **Next**.



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.



Important Note: The **Return Audio Port** is the port used by the local codec to receive audio from the remote codec. When **Tieline Codecs** is the **Session Protocol** selected (using Tieline session data), the **Return Audio Port** is automatically configured as UDP audio port 9000 by default for the first audio stream connection. Click to deselect the **Any** check-box to adjust this setting.

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.

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

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

3. After configuring all settings there are 2 options:

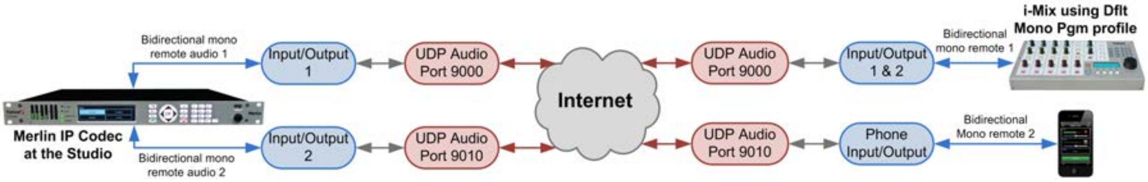
   i. If you want to create another answering connection, select the check-box for **Create another answering connection** and continue through the wizard.
   ii. Click **Save Program** to save the program at this point.

4. Click **Finish** to exit the wizard.

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.

20.6 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.

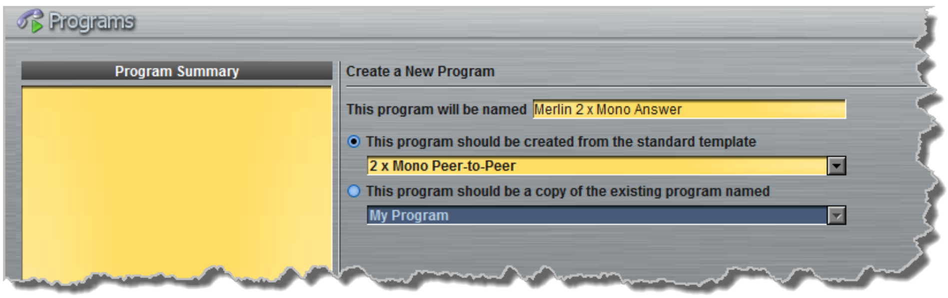
Routing 2 Incoming Mono Audio Streams to Specific Codec Outputs

Important Notes: Before you start program configuration please note:

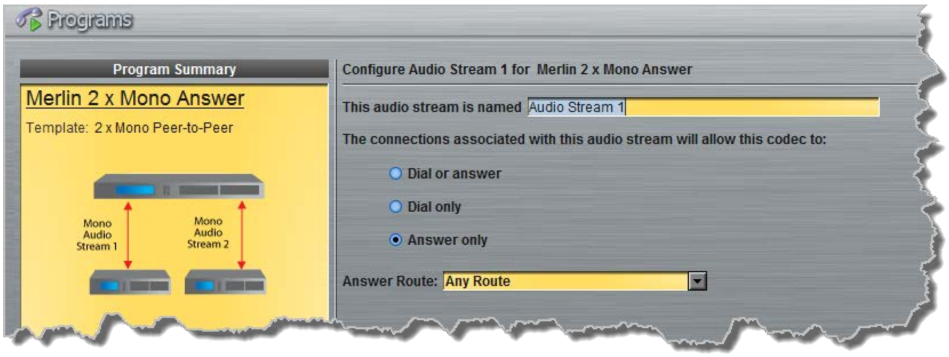
- You cannot edit a program when it is currently loaded in the codec.
- You can lock a loaded custom program in a codec to ensure the currently loaded program cannot be unloaded by a codec dialing in with a different type of program.
- Some drop-down menus and settings may be greyed out intentionally depending on features available 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.
- Failover and SmartStream PLUS redundant streaming are not available with SIP or sessionless IP connections.
- 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 theJava Toolbox 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**.

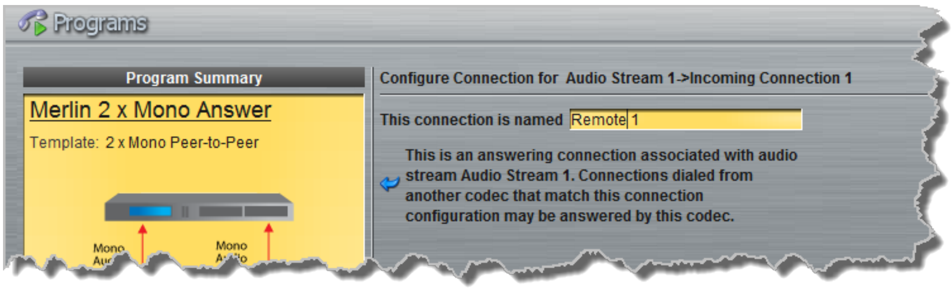


3. Enter a name for the **Audio Stream** and select **Answer only**. Then click **Next**. Note: if the codec you are configuring needs to dial connections as well as answer, you can select **Dial or answer** and configure both dialing and answering connections. For the purposes of this example dialing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



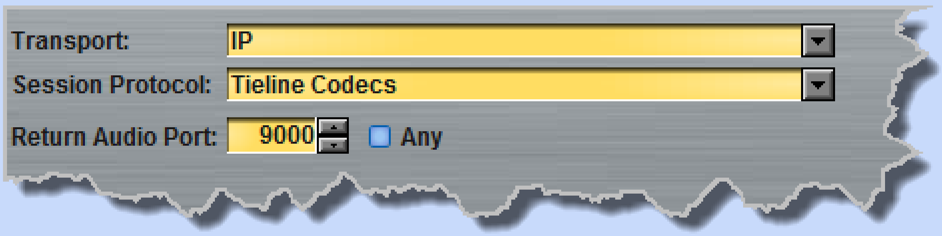
It is also possible to select an **Answer Route** if required. When routing multiple audio streams over transports like ISDN or POTS, you can use **Dial** and **Answer Routes** to configure deterministic routing of audio streams. Use of **Dial** and **Answer Routes** is not recommended over IP. See Configuring ISDN Answering or Configuring POTS 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:

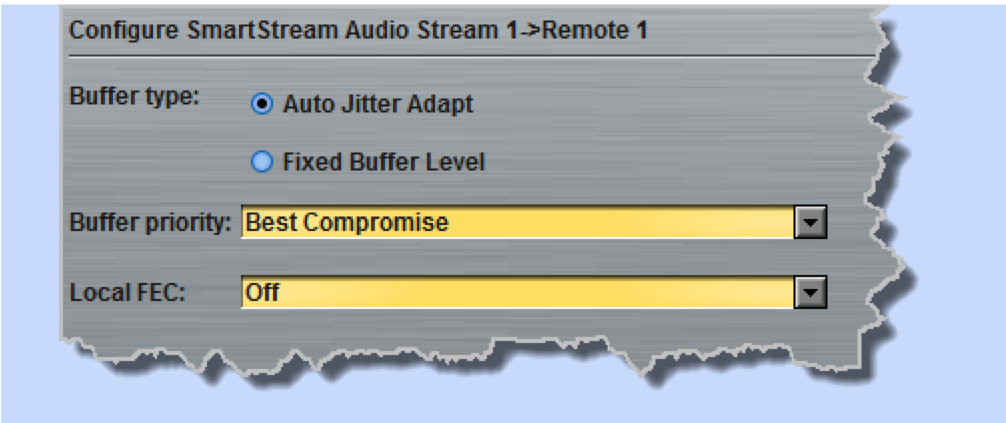
For IP click the drop-down **Session Protocol** menu and select **Tieline Codecs** and ensure the **Any** check-box is not selected, then click **Next**.



Important Note: The **Return Audio Port** is used by the local codec to receive audio from the remote codec. When **Tieline Codecs** is the **Session Protocol** selected (using Tieline session data), the **Return Audio Port** is automatically configured as UDP audio port 9000 by default for the first audio stream connection. Click to deselect the **Any** check-box to adjust this setting. A codec dialing this connection and using the port specified will always be routed to output 1 on the codec receiving the call.

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

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

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

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

7. Configure the 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**.

Important Note: When **Tieline Codecs** is the **Session Protocol** selected (using Tieline session data), the **Return Audio Port** is automatically configured as UDP audio port 9010 by default for the second audio stream connection. Click to deselect the **Automatic** check-box to adjust this setting. A codec dialing this connection and using the port specified will always be routed to output 2 on the codec receiving the call.

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

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

8. 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**.

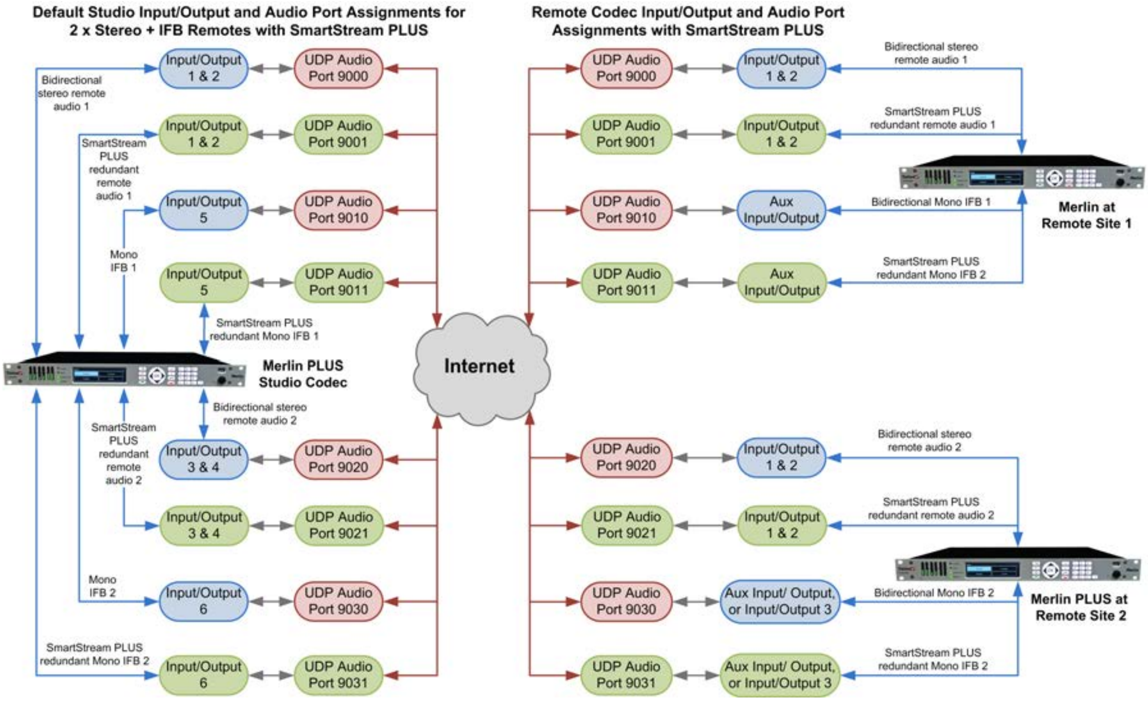
When this program is loaded any codec dialing in using IP1 (using default Tieline IP port settings) will be routed to output 1 on the codec and a codec dialing in using IP2 will be routed to output 2.

20.7 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

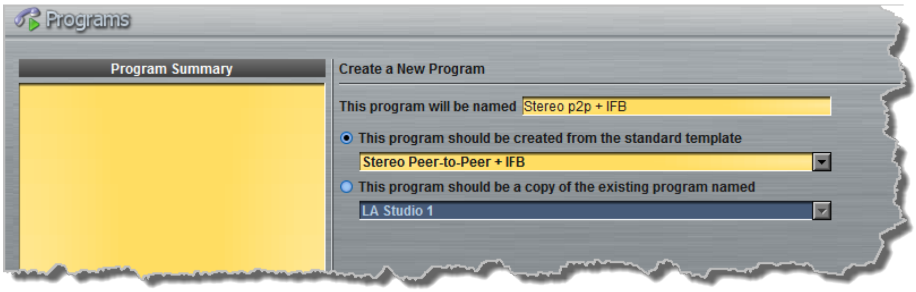
Important Notes: Before you commence program configuration please note:

- 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 cannot be unloaded by a codec dialing in with a different type of program.
- Some drop-down menus and settings may be greyed out intentionally depending on features available 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.
- Failover and SmartStream PLUS redundant streaming are not available with SIP or sessionless IP connections.

- POTS is not supported for stereo audio stream connections.
- To learn more about programs see the section titled About Program Dialing.

1. Open the Java Toolbox 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.

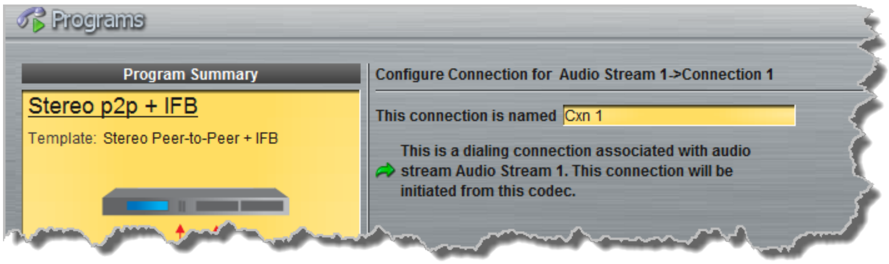


Important Note: When you decide to use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

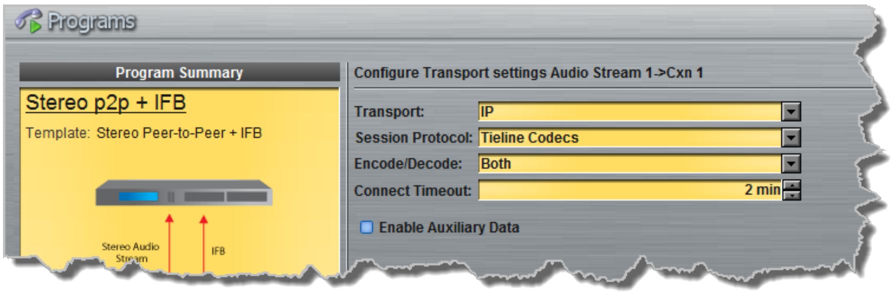
3. Enter the **Audio Stream** name and configure the codec to **Dial only**. Then click **Next**.

It is also possible to select an **Dial Route** if required. When routing multiple audio streams over transports like ISDN or POTS, you can use **Dial** and **Answer Routes** to configure deterministic of routing audio streams. Use of **Dial** and **Answer Routes** is not recommended over IP. See Configuring ISDN Answering or Configuring POTS 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**.

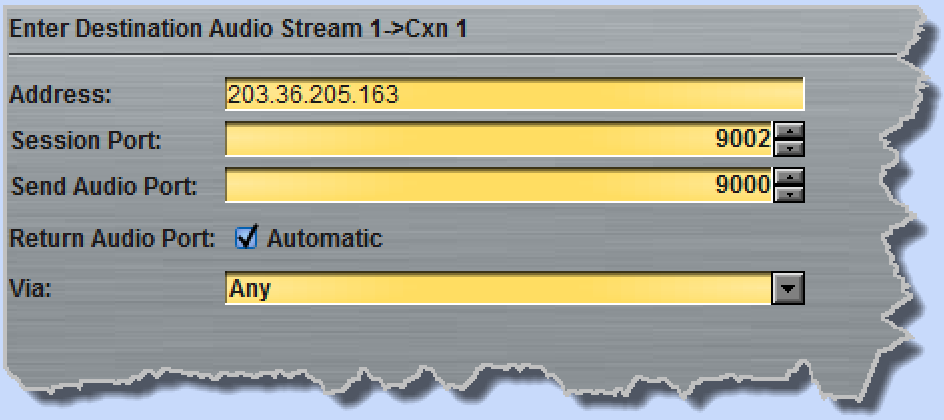


Important Note: Bidirectional auxiliary IP data is available on one audio stream. When auxiliary data is enabled on one stream the option is greyed out for the other audio stream in the program wizard. See RS232 Data Configuration for detailed information on RS232 data and see Enabling Relays and RS232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



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

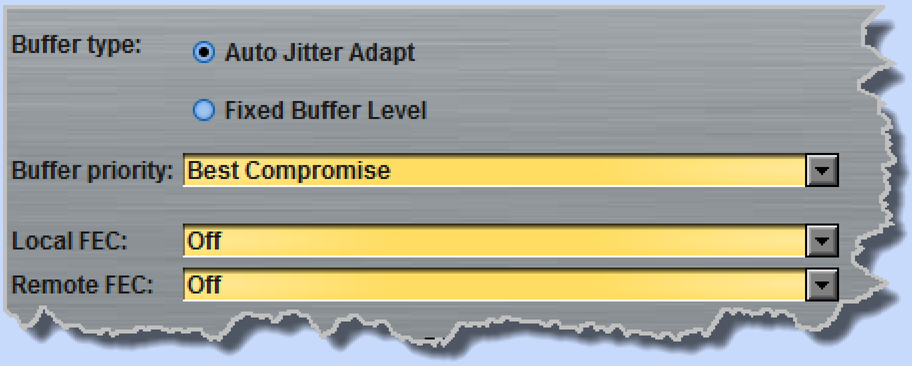
Click **Save Program** to save the program with the default algorithm, jitter and FEC settings which are physically entered in the codec. Alternatively, click **Next** to specify individual algorithm, jitter buffer and FEC settings and configure a backup connection or SmartStream PLUS for this audio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

Note: If you connect multiple remote codecs simultaneously to a Merlin PLUS codec at the studio (when creating 2 x Mono or Stereo Peer-to-Peer + IFB connections), use **Send Audio Port 9020** to configure the second mono/stereo dialing 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.

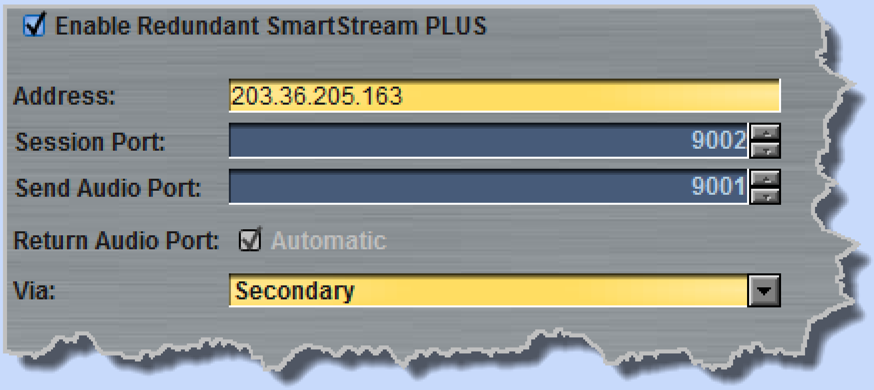
For IP connections click to configure:

- **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.

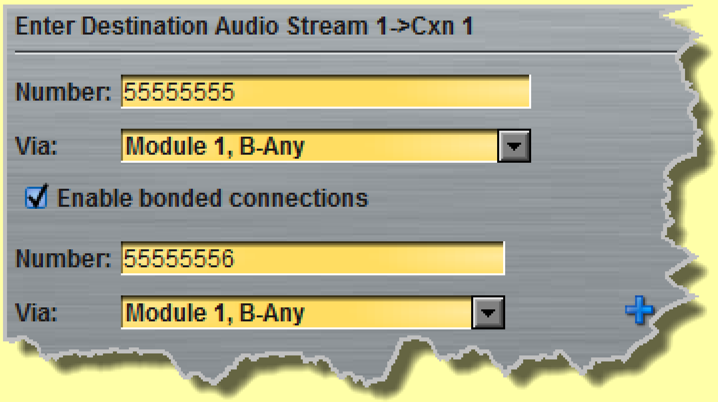
By default, primary IP streaming is via **ETH1**. To achieve the maximum level of redundancy select **Secondary** to configure redundant streaming from the secondary IP port **ETH2**. The redundant stream uses **Send Audio Port 9001** by default and the **Return Audio Port** allocated is **Automatic**. Note: **Automatic** indicates that the codec will arbitrarily allocate the return port value and send this information to the codec to which you are dialing.



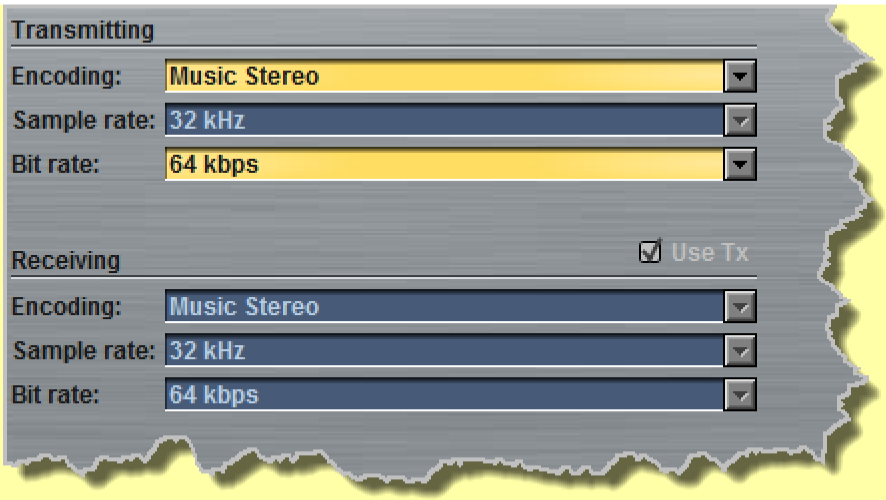
Important Note: Dual SmartStream PLUS redundant streaming over both Ethernet ports mitigates lost packets on either link and will provide IP network backup if an IP link is lost. To learn more about SmartStream PLUS redundant IP streaming see http://www.tieline.com/Transports/SmartStream-IP

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.



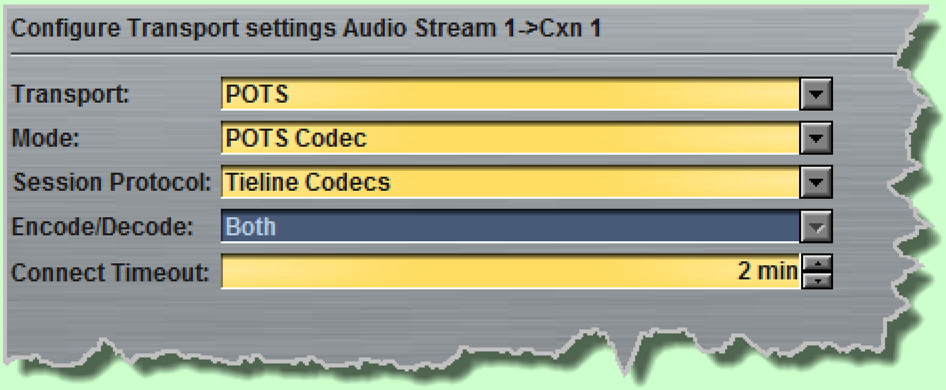
Next, click **Save Program** to save the program with default algorithm settings, or click **Next** to specify a different algorithm and configure a backup connection if required. (recommended).



Dialing settings for this ISDN audio stream are now complete.

POTS

Select **POTS Codec** in the **Mode** drop-down menu to encode/decode using POTS, or select **Analog Phone** to configure a standard analog phone call, then click **Next**.



Next, enter the phone number of the codec or device you want to dial. When multiple POTS modules are installed, click the **Via** drop-down menu and select **Module 1** or **Module 2 to** specify which POTS module will dial. Next, click **Save Program** to save the program with default settings, or click **Next** to specify algorithm settings and configure a backup connection if required (recommended).

Dialing configuration settings for this POTS audio stream are now complete.

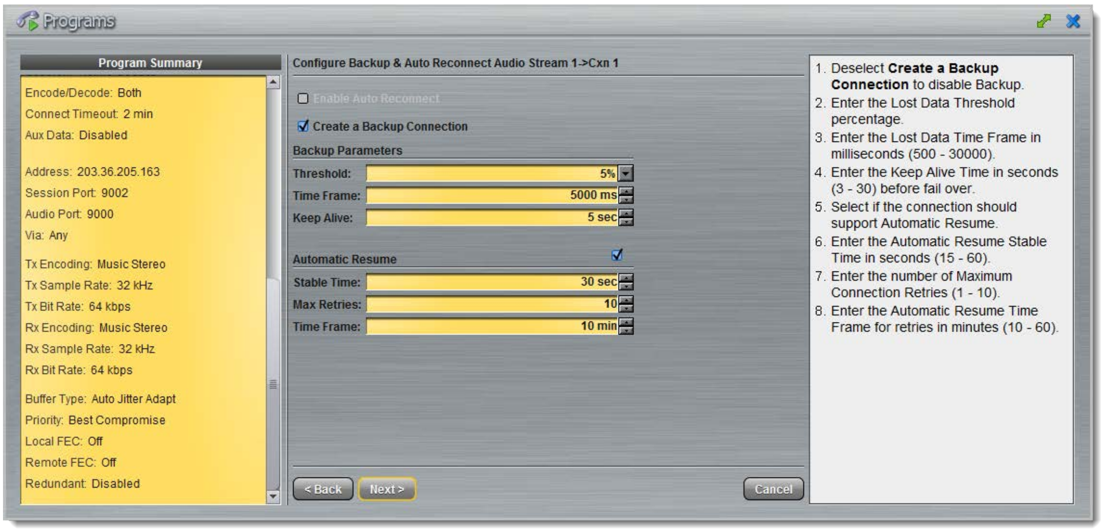
Configuring a Backup Connection or Auto Reconnect

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:

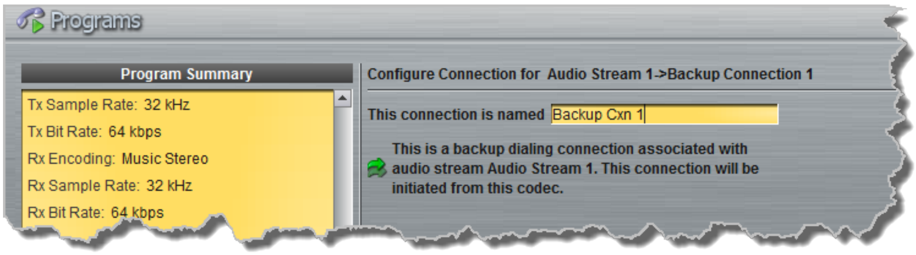
1. Click to select the check-box for **Create a Backup Connection**. Adjust the parameters and click **Next**.



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**.

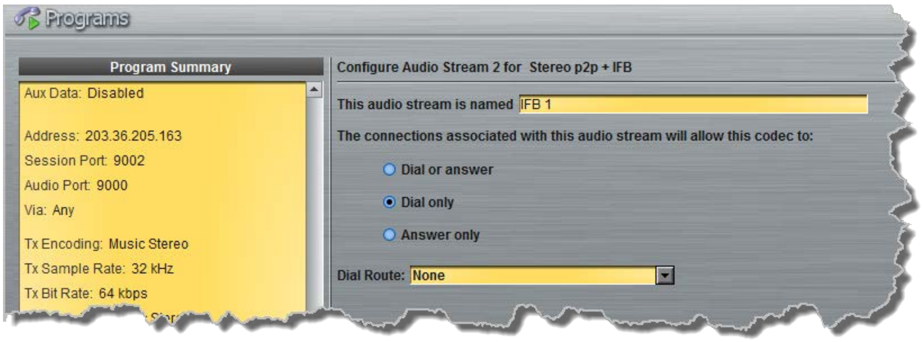


3. Click **Next** to continue through the wizard and configure the backup connection in a similar manner to how you have configured the primary connection.

Configure the Bidirectional IFB Audio Stream

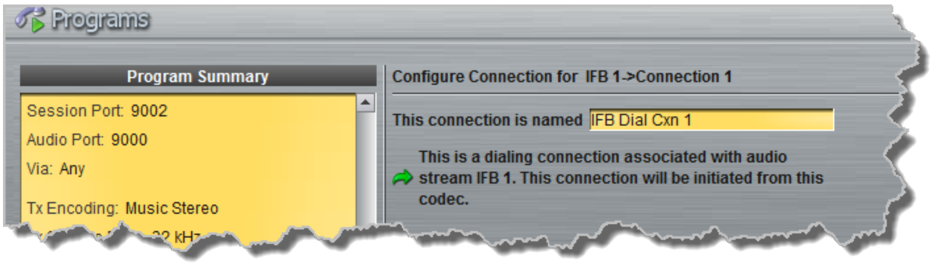
When you have finished configuring SmartStream PLUS, Auto Reconnect or a backup connection, proceed with configuration of the IFB audio stream in the wizard.

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

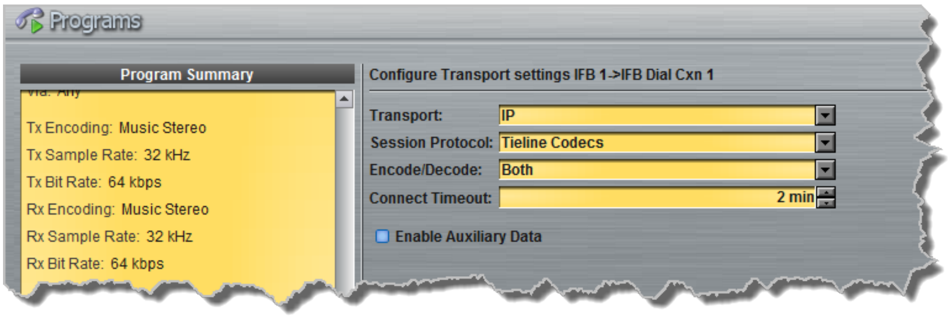


It is also possible to select an **Dial Route** if required. When routing multiple audio streams over transports like ISDN or POTS, you can use **Dial** and **Answer Routes** to configure deterministic of routing audio streams. Use of **Dial** and **Answer Routes** is not recommended over IP. See Configuring ISDN Answering or Configuring POTS Answering for more information. Use the default settings for IP connections.

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**.

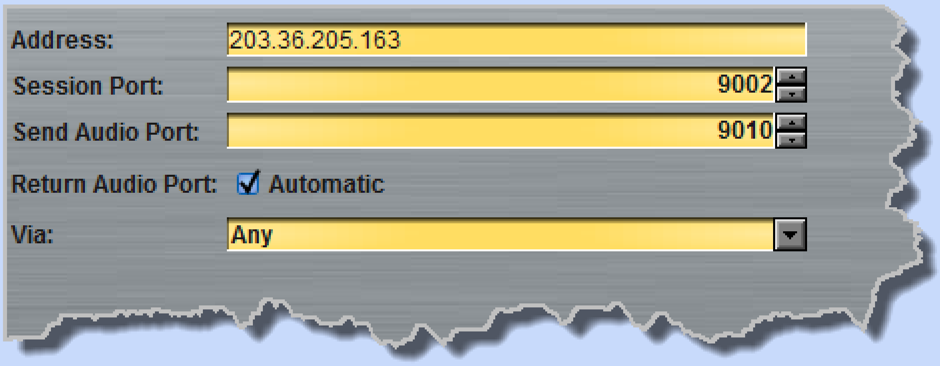


Important Note: Bidirectional auxiliary IP data is available on one audio stream. When auxiliary data is enabled on one stream the option is greyed out for the other audio stream in the program wizard. See RS232 Data Configuration for detailed information on RS232 data and see Enabling Relays and RS232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

4. Configure destination codec dialing and encoding settings:

IP

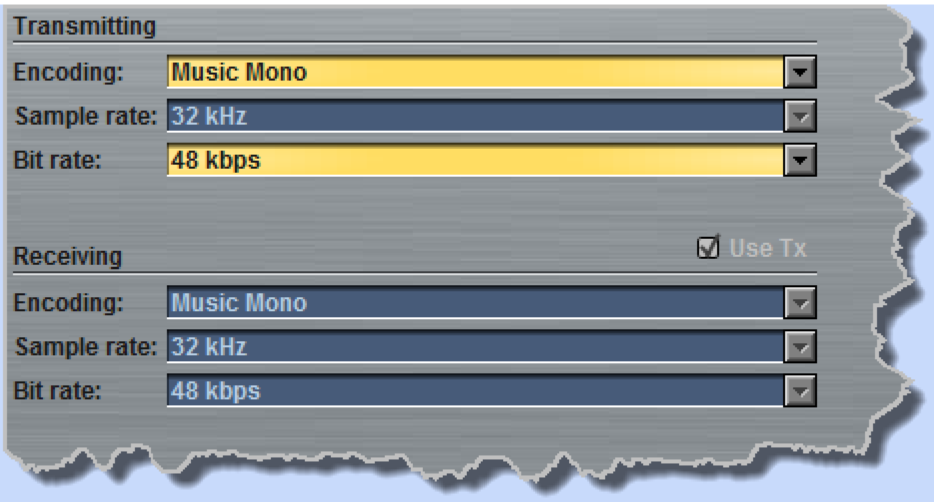
For IP connections configure the IP address, ports, and then specify which streaming interface is used to dial this connection, e.g. **Primary** (port **ETH1**) or **Secondary** (port **ETH2**). Note: By default **Any** will select **ETH1** if it is available and **ETH2** if it is unavailable.



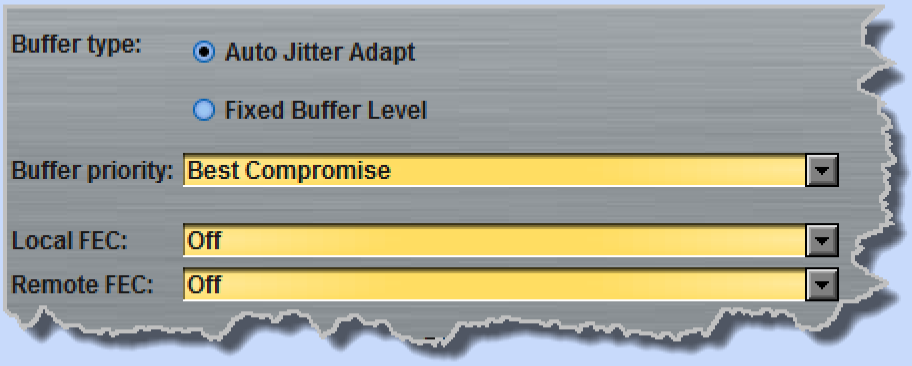
Click **Save Program** to save the program with the default algorithm, jitter and FEC settings which are physically entered in the codec. Alternatively, click **Next** to specify individual algorithm, jitter buffer and FEC settings and configure a backup connection or SmartStream PLUS for this audio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

Note: The default **Send Audio Port** is **9010** for this IP audio stream. If you connect multiple remote codecs simultaneously to a Merlin PLUS codec at the studio (when creating 2 x Mono or Stereo Peer-to-Peer + IFB connections), use **Send Audio Port 9030** to configure 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.

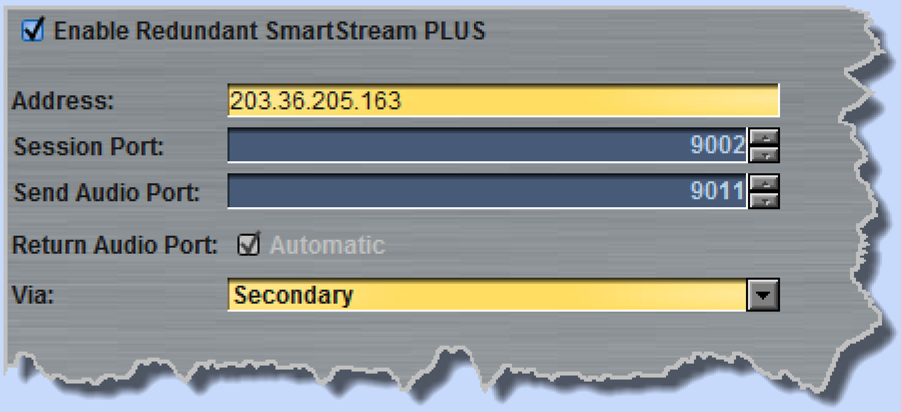
Click the drop-down arrows on the right-hand side of each text box to adjust the **Encoding**, **Sample rate** and **Bit rate** options.



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.

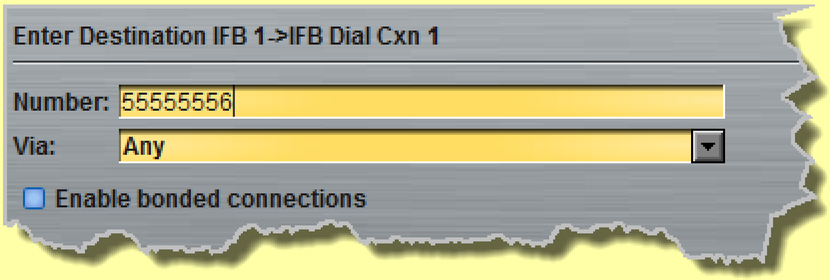


By default, primary IP streaming is via **ETH1**. To achieve the maximum level of redundancy select **Secondary** to configure redundant streaming from the secondary IP port **ETH2**. The redundant stream uses **Send Audio Port 9011** by default, and provides automatic IP streaming backup in case one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



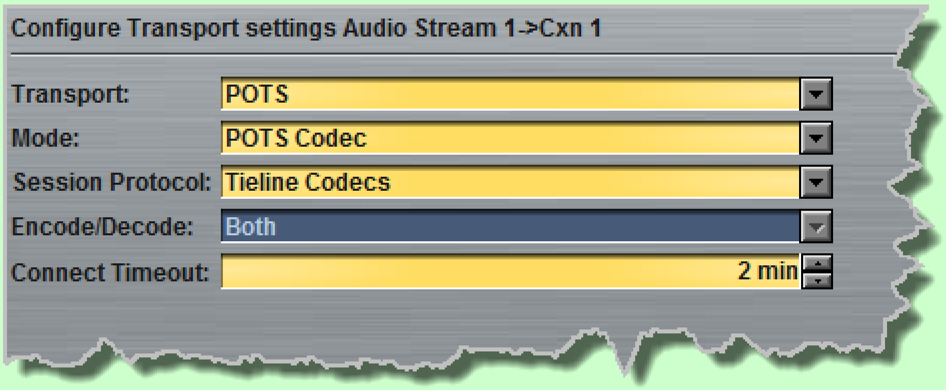
Important Note: Dual SmartStream PLUS redundant streaming over both Ethernet ports mitigates lost packets on either link and will provide IP network backup if an IP link is lost. To learn more about SmartStream PLUS redundant IP streaming see http://www.tieline.com/Transports/SmartStream-IP

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.

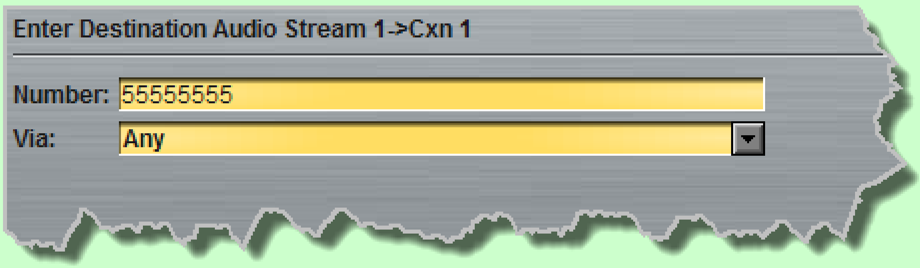


Next, click **Save Program** to save the program with default algorithm settings, or click **Next** to specify a different algorithm and configure a backup connection if required. (recommended). Dialing settings for this ISDN audio stream are now complete.

Select **POTS Codec** in the **Mode** drop-down menu to encode/decode using POTS, or select **Analog Phone** to configure a standard analog phone call, then click **Next**.



Next, enter the phone number of the codec or device you want to dial. When multiple POTS modules are installed, click the **Via** drop-down menu and select **Module 1** or **Module 2 to** specify which POTS module will dial. Next, click **Save Program** to save the program with default settings, or click **Next** to specify algorithm settings and configure a backup connection if required (recommended).



Dialing configuration settings for this POTS audio stream are now complete.

5. Click **Save Program** to complete configuration, then click **Finish** to exit the wizard.

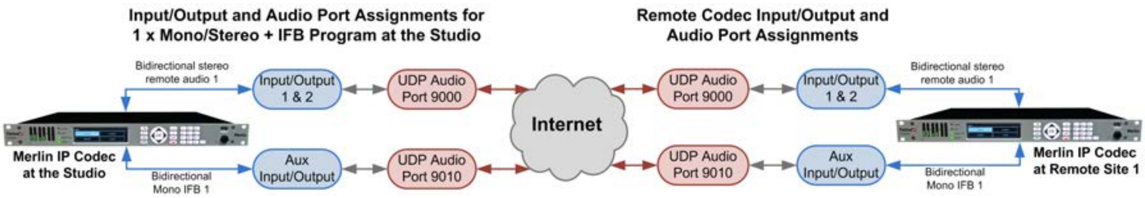
6. The newly created program will be displayed in the left pane within the **Programs panel** and in the **Master panel**. Select and connect audio streams in a program using the **Master panel**, or dial the program manually using the codec front panel.

20.8 Configure Mono or Stereo + IFB Answering Programs

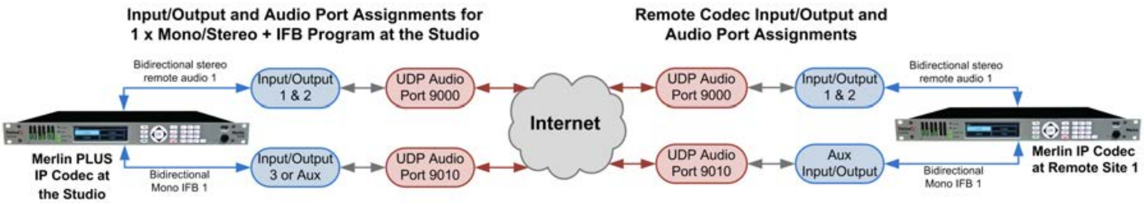
This program is designed to allow Merlin and Merlin PLUS codecs to answer a call from an incoming codec and receive:

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

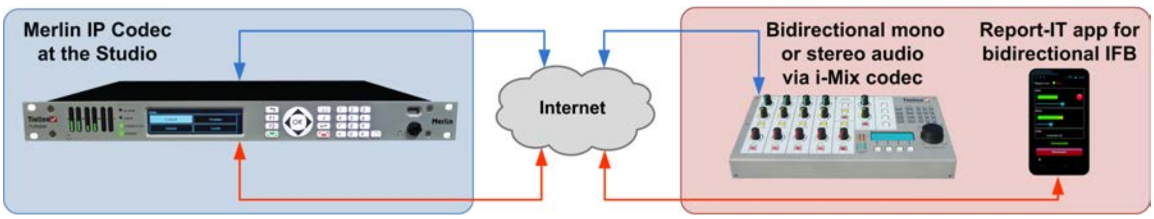


Merlin PLUS at the studio connects to a remote Merlin (Note Aux/input 3 IFB routing option)

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:

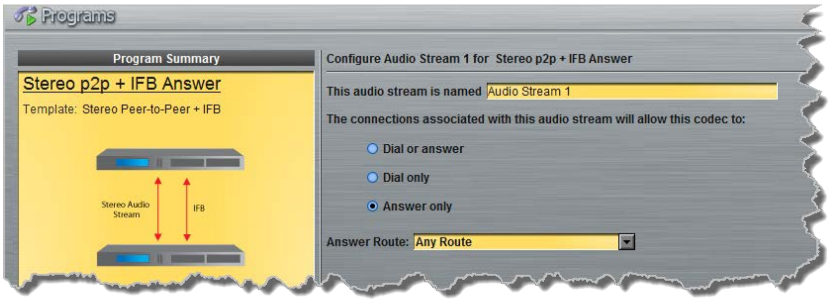
  - The auxiliary input is used by default for the IFB communications channel in Merlin codecs. In Merlin PLUS codecs the auxiliary input and XLR input 3 are mixed together by default for 1 x Mono/Stereo + IFB programs.
  - You cannot edit a program when it is currently loaded in the codec.
  - You can lock a loaded custom program in a codec to ensure the currently loaded program cannot be unloaded by a codec dialing in with a different type of program.
- 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.
  - Failover and SmartStream PLUS redundant streaming are not available with SIP or sessionless IP connections.
  - POTS is not supported for stereo audio stream connections.
  - To learn more about programs see the section titled About Program Dialing.

1. Open the Java Toolbox 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.

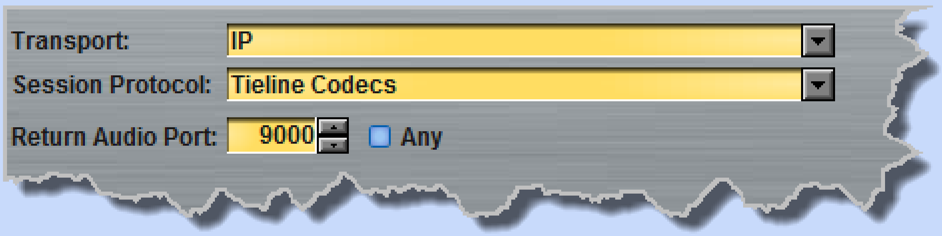


It is also possible to select an **Answer Route** if required. When routing multiple audio streams over transports like ISDN or POTS, you can use **Dial** and **Answer Routes** to configure deterministic routing of audio streams. Use of **Dial** and **Answer Routes** is not recommended over IP. See Configuring ISDN Answering or Configuring POTS Answering for more information. Use the default settings for IP connections.

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

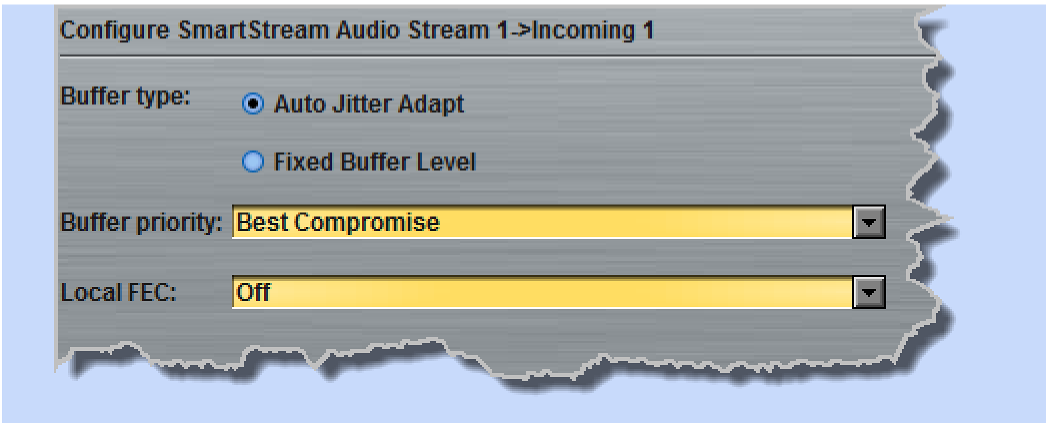
5. Configure the transport settings:

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**.



Important Note: The **Return Audio Port** is the port used by the local codec to receive audio from the remote codec. When **Tieline Codecs** is the **Session Protocol** selected (using Tieline session data), the **Return Audio Port** is automatically configured as UDP audio port 9000 by default for the first audio stream. Click to deselect the **Any** check-box to adjust this setting.

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.



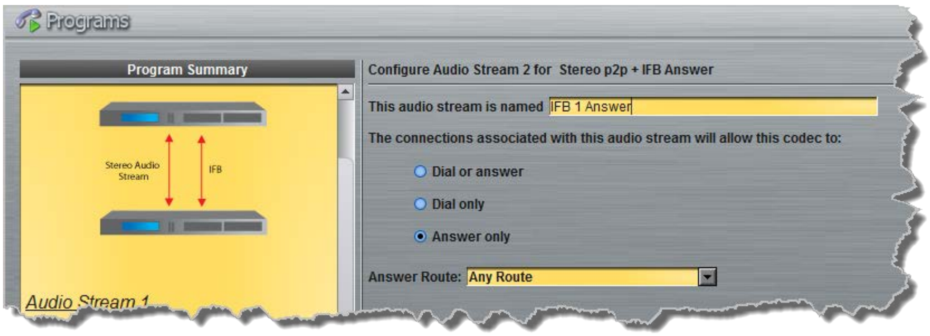
For ISDN, settings are determined by ISDN module answering settings. For more details see Configuring ISDN Answering.

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

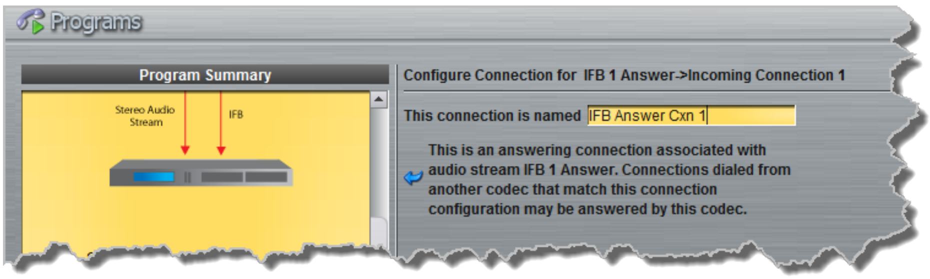
6. After configuring all settings there are 2 options:

   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:

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**.

Important Note: When **Tieline Codecs** is the **Session Protocol** selected (using Tieline session data), the **Return Audio Port** is automatically configured as UDP audio port 9010 by default for the second audio stream.

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.

POTS For POTS, settings are determined by POTS module answering settings. For more details see Configuring POTS 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, then click **Finish** to exit the wizard.

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.

20.9 Configure Multicast Client Programs

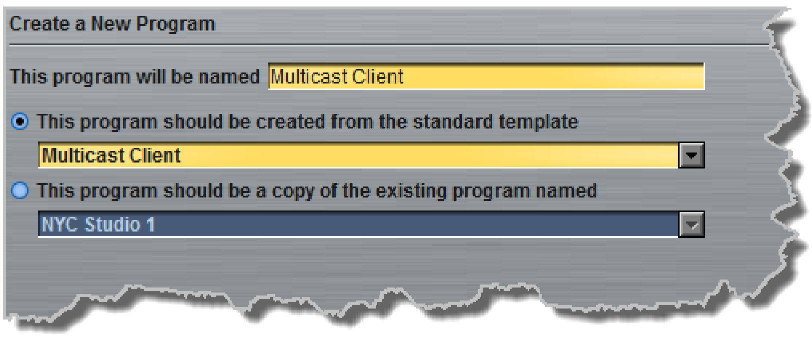
Important Notes: Before you commence program configuration please note:

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

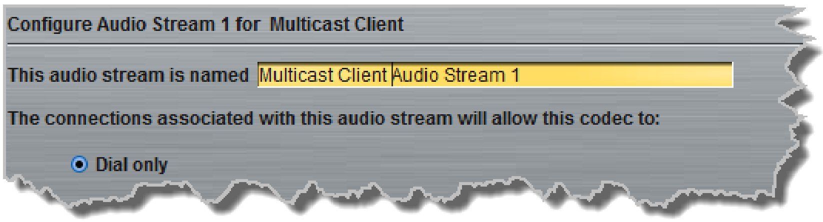
1. Open the Java Toolbox Web-GUI and click the **Programs** symbol at the top of the screen to display the **Programs panel**.

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**.

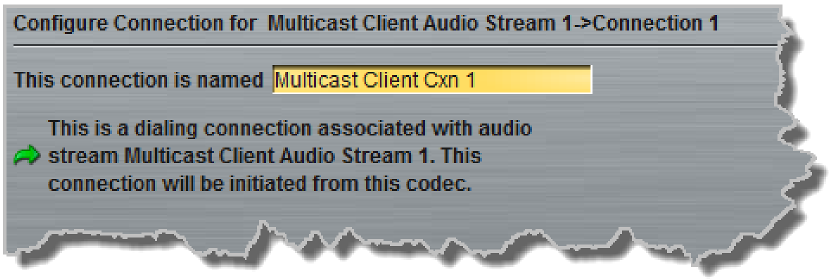


Important Notes: When you decide to use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

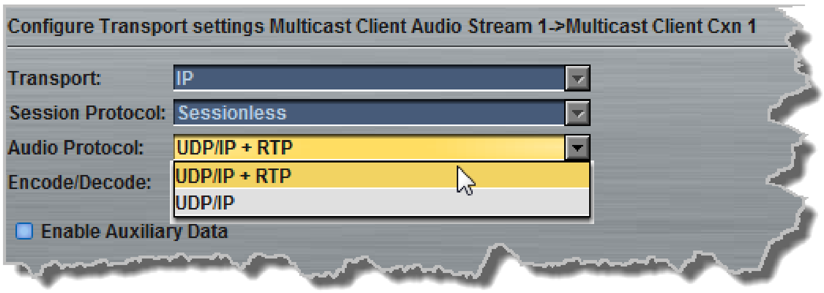
3. Enter a name for the **Audio Stream**, then click **Next**.



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**.

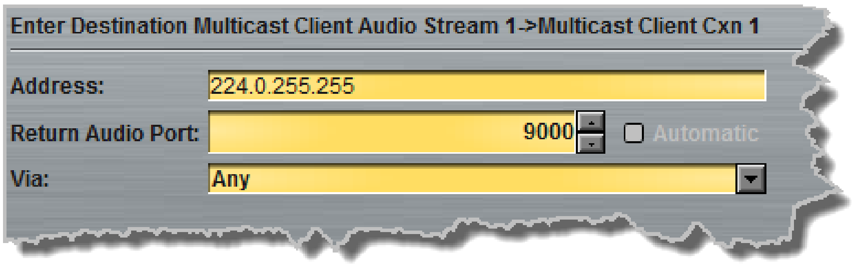


5. Follow the instructions on the right-hand side of the panel to configure the transport settings for the connection, then click **Next**. Note: select **UDP/IP +RTP** for RFC compliant streaming.

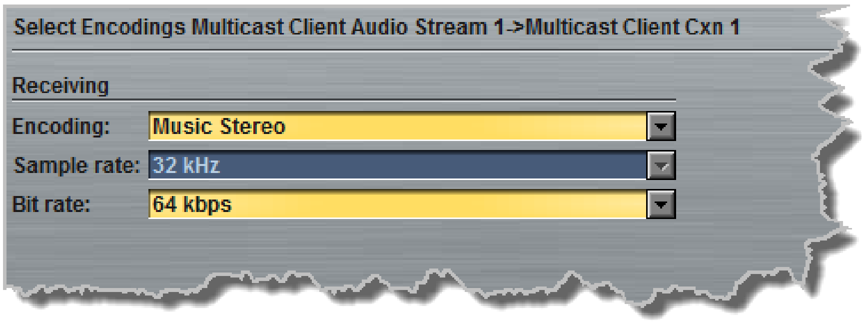


Important Note: Bidirectional auxiliary IP data is available on one audio stream when multicast dialing programs. When auxiliary data is enabled on one stream the option is greyed out for other audio streams in the program wizard. See RS232 Data Configuration for detailed information on RS232 data and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

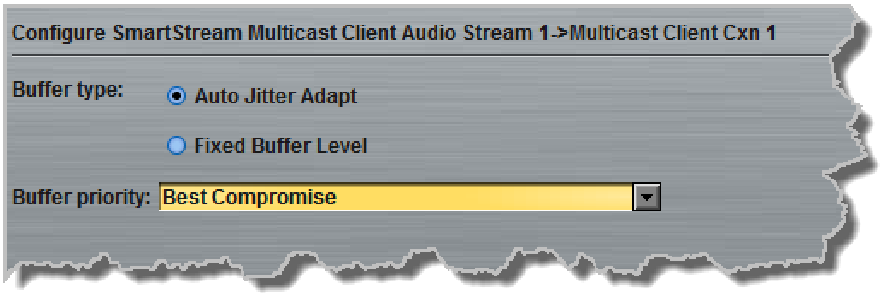
6. Configure the multicast IP address and audio port (the same multicast address and port must be used for both the server and client programs), then specify which IP streaming interface is used to dial this connection, e.g. **Primary** (port **ETH1**) or **Secondary** (port **ETH2**), then click **Next**. 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.

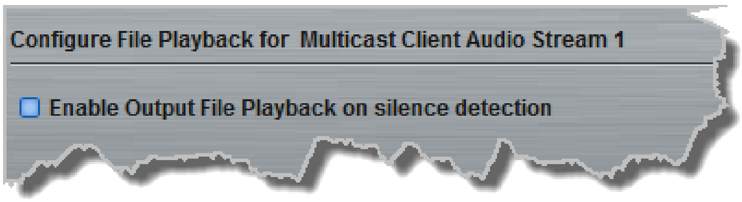


8. Click to configure:
   - **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer Priority**, or
   - **Fixed Buffer Level** and enter the **Jitter Depth**, which must be between 12ms and 5000ms depending on the algorithm you select.



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.

20.10 Configure SIP Settings

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

About SIP

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

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

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

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

SIP Server Connections: Getting Started

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

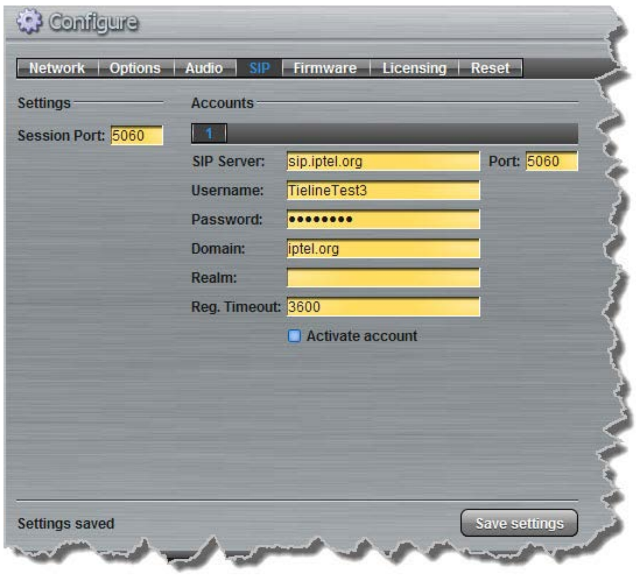
When you register an account with a SIP server you will be provided with:

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

Program the Codec for SIP using the Java Web-GUI

Use the Java Toolbox Web-GUI to configure 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 Java Toolbox Web-GUI and click the **Settings** symbol at the top of the screen to display the **Settings panel**.
2. Click the **SIP** button in the top-left corner of the **System panel**.
3. Enter the account details into the relevant text boxes.
4. Enter the **Registration Timeout** (this shouldn't need to be adjusted from the default setting).
5. Click to select **Activate Account** and click the **Save Settings** button to create the account in the codec. **Settings Saved** is displayed in the bottom-left corner of the **Settings panel** if the account details are saved successfully.



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

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

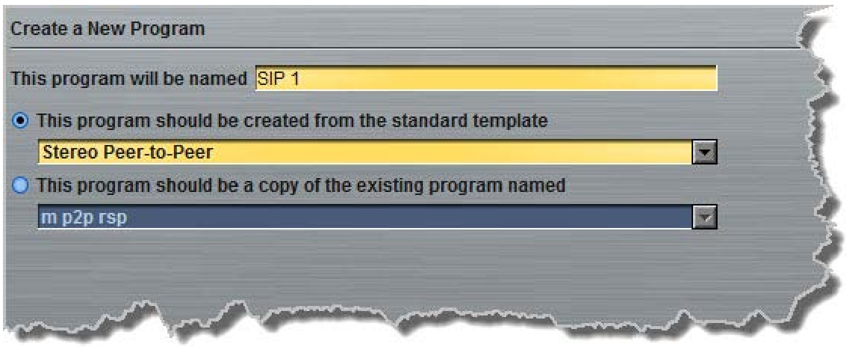
20.11 Configure Peer-to-Peer SIP Programs

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

Important Notes: Before you start program configuration please note:
- You cannot edit a program when it is currently loaded in the codec.
- Some drop-down menus and settings may be greyed out intentionally depending on features available.
- Failover and SmartStream PLUS redundant streaming are not available with SIP connections.
- To learn more about programs see the section titled About Program Dialing.

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

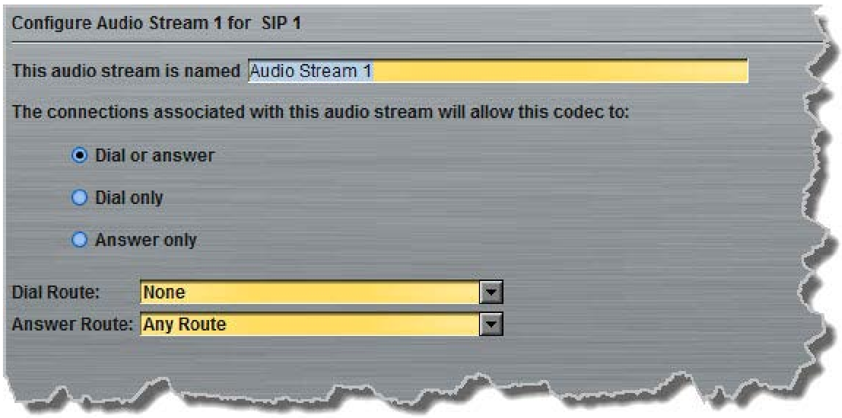
2. Click the **New Program** button to open the wizard and:
   - Click in the **Program Name** text box to name the new program.
   - Select **Mono/Stereo Peer-to-Peer,** or if you want to use an existing program as a template, select this option. Then click **Next**.



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

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

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

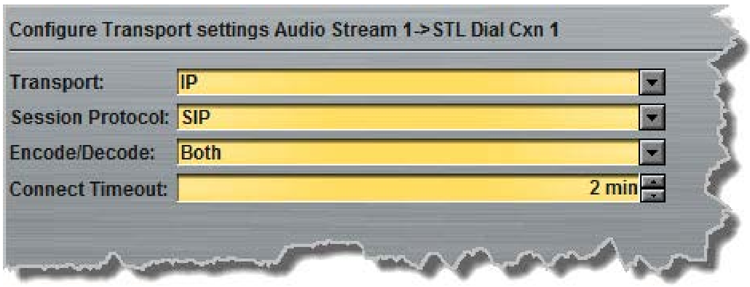


4. This audio stream connection in the wizard will allow the codec to dial. Enter the name of the connection in the text box, then click **Next**.

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

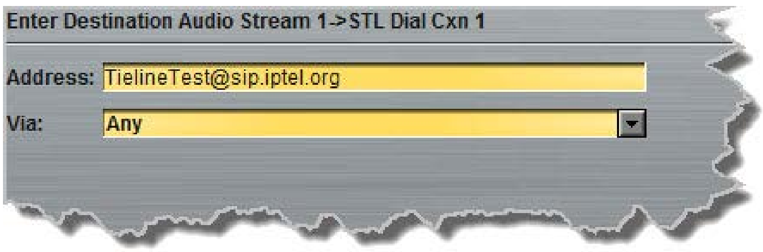
- **IP** as the **Transport**.
- **SIP** from the **Session Protocol** menu option.

Then click **Next**.



6. Configure the destination codec **Address**, then specify 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 default algorithm and jitter settings. Alternatively, click **Next** to confirm and specify algorithm and jitter settings for this connection and configure backup audio settings (recommended).



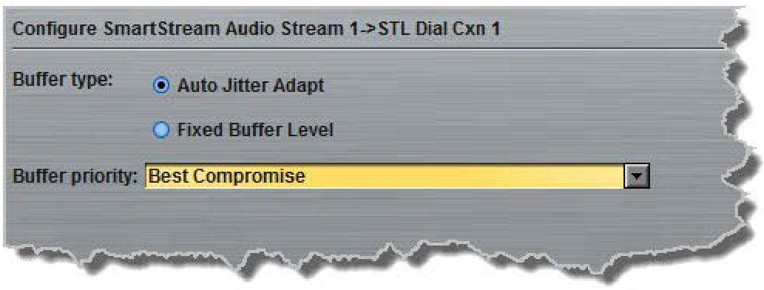
Important Notes:

- 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 for a peer-to-peer connection is 5004 in Tieline codecs. To contact a codec that is behind a firewall or NAT-enabled router, it is essential that this and all other relevant ports are open and forwarded to the other device.

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.

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.



Click **Save Program** to save all settings, or click **Next** to configure **Auto Reconnect**. 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.

20.12 Multiple Peer-to-Peer SIP Programs

The codec is also capable of creating multiple SIP Peer-to-Peer connections. Configure a new program and configure each SIP audio stream as you would for a single SIP Peer-to-Peer program.

The SIP UDP audio ports are automatically allocated by the Java Toolbox Web-GUI when you create SIP programs incorporating multiple audio streams. The first stream uses UDP audio port 5004 and then each subsequent stream created will in the following order use UDP audio ports 5006, 5008, 5010, 5012 and 5014. These audio ports need to be open in your firewall at each end of the connection to allow the successful transfer of audio packets.

Answering Multiple SIP Calls

To answer multiple SIP calls you need to create and lock a suitable SIP answering program in the codec, or it will be unloaded by the first SIP call and a default peer-to-peer program will be loaded. There is no Tieline session data transferred during SIP calls to assist with configuring the codec.

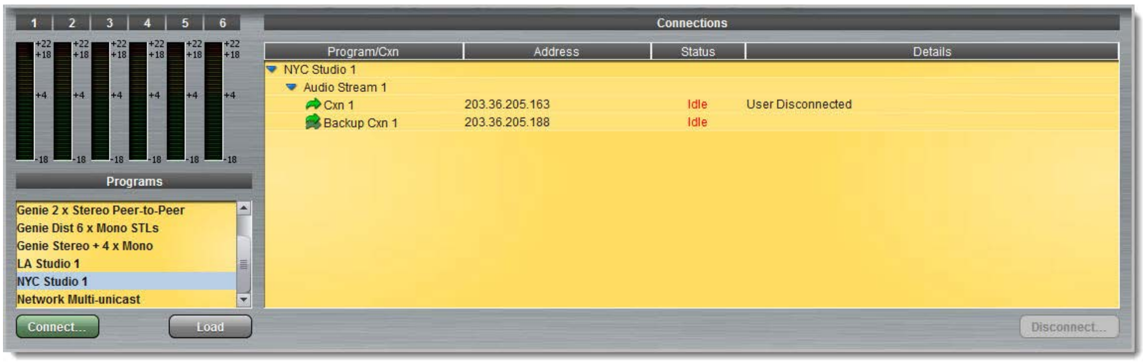
Important Notes:

- Remember to lock an answering program in a codec when answering multiple SIP calls.
- When multiple calls are answered by the codec they are routed to audio inputs and outputs on a first come, first served basis.
- Ensure the appropriate UDP audio ports are open in your firewall to allow multiple SIP audio streams to connect. See Installing the Codec at the Studio for more information.
- Failover and SmartStream PLUS redundant streaming are not available with SIP connections.

20.13 Dial and Disconnect a Program

Connecting a Program

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



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

Disconnecting a Program

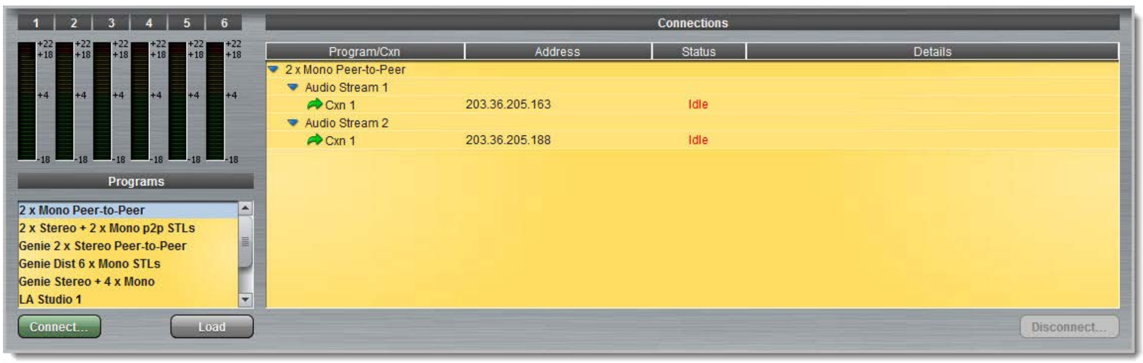
1. Click to highlight the audio stream in the **Connections** pane of the **Master panel**.

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

20.14 Dial/Disconnect Multiple Audio Streams

Load and Connect Multiple Audio Streams within a Program

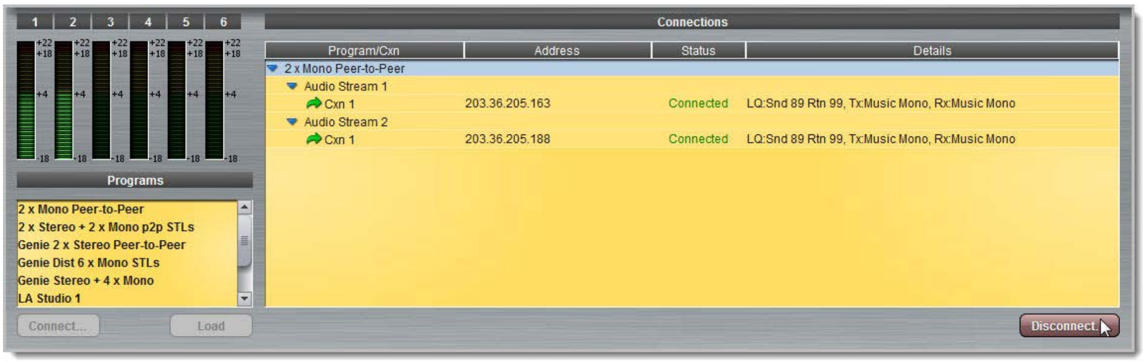
1. Click to select the program you want to load from the **Programs** list.



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

Disconnect All Audio Stream Connections

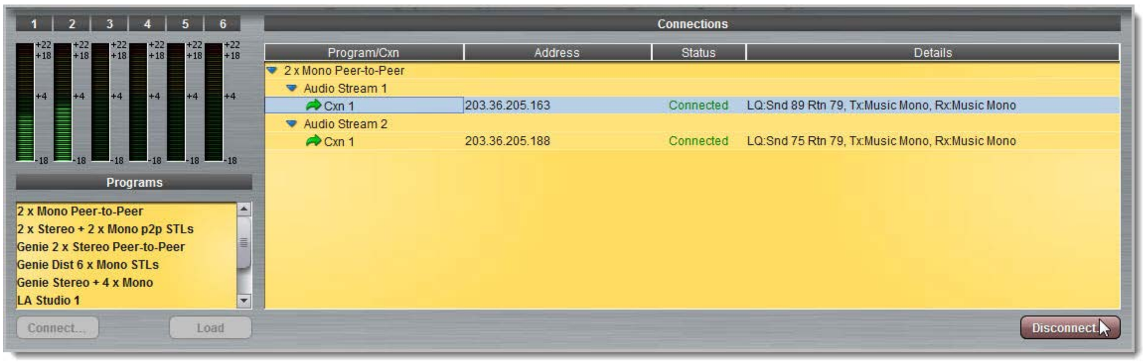
1. Click to select the program in the **Connections** pane, e.g. **2 x Mono Peer-to-Peer** in the following example.



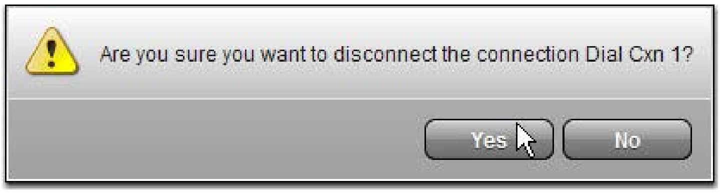
2. Click the **Disconnect** button.
3. Click **Yes** in the confirmation dialog to disconnect all audio stream connections.

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.



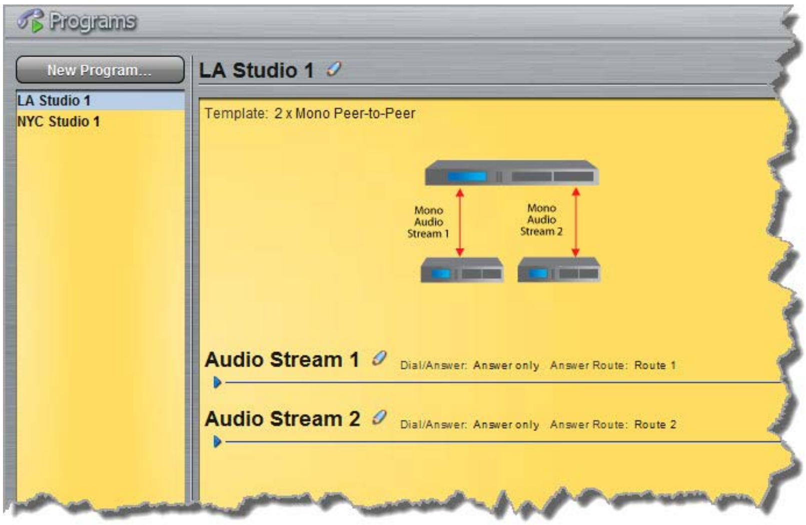
20.15 View/Edit/Delete Programs

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 Java Toolbox 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.

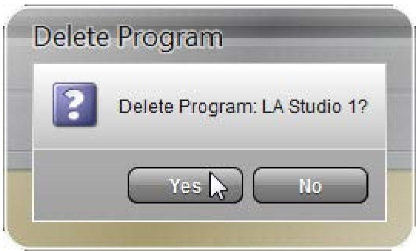
4. The program wizard will open at the relevant point to facilitate editing of connection parameters. Click **Save Program** to store settings.

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 in the top-right corner of the **Programs panel**.

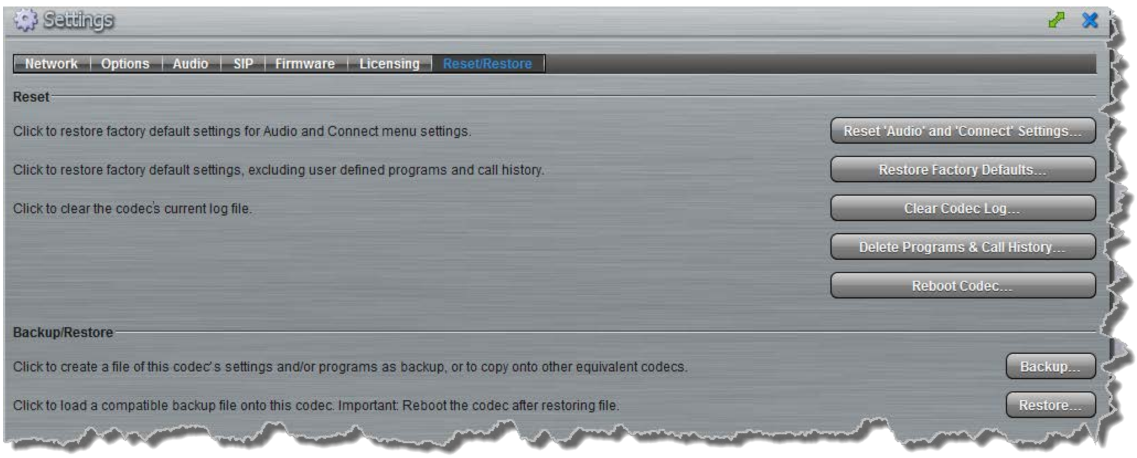
20.16 Reset Factory Default Settings

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

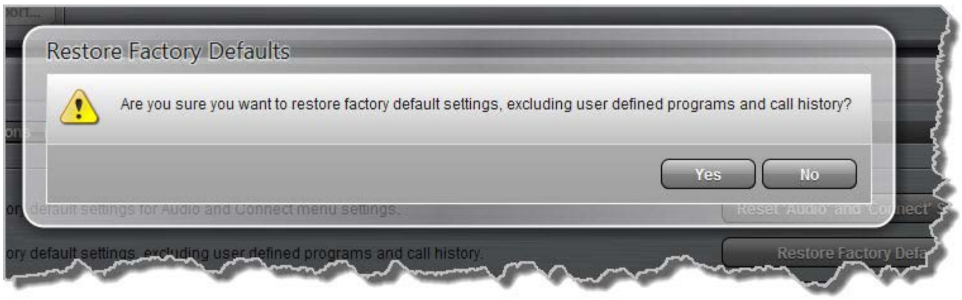
1. Open the Web-GUI and click the **Settings**  symbol at the top of the screen to display the **Settings panel**.

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

3. Click one of the four reset options available.



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



20.17 Backup and Restore Functions

The Java 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:

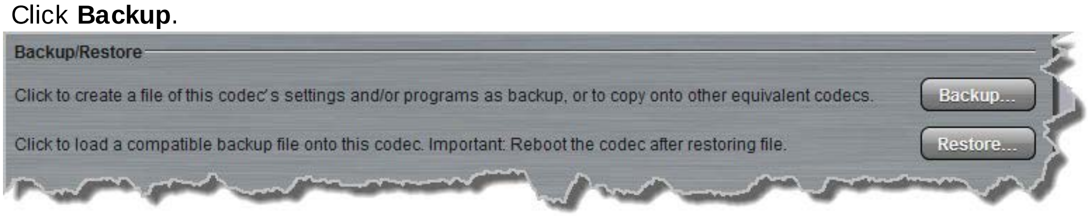
- Program, audio stream and connection names.
- IP address, port, algorithm, jitter buffer, FEC and bit rate settings (etc.) for audio stream connections.

Creating Backup Files

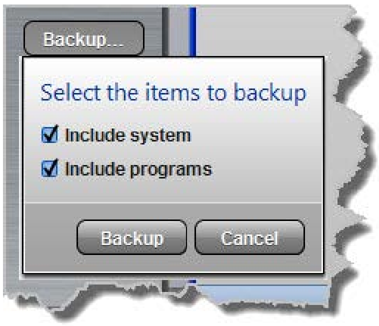
1. Open the Java Toolbox Web-GUI and click the **Settings** symbol at the top of the screen to display the **Settings panel**.

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

3. Click **Backup**.



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



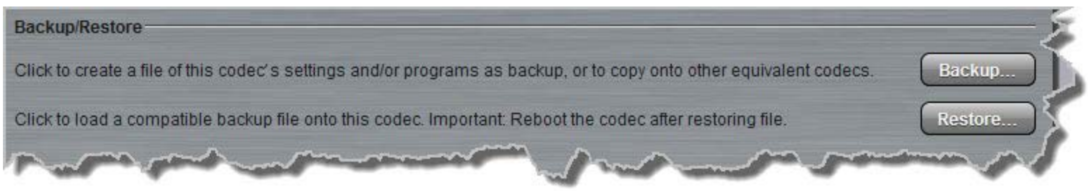
5. Click **Save** and select a location on your PC to save the configuration file.

Restoring Configuration File Settings

1. Open the Java Toolbox 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**.

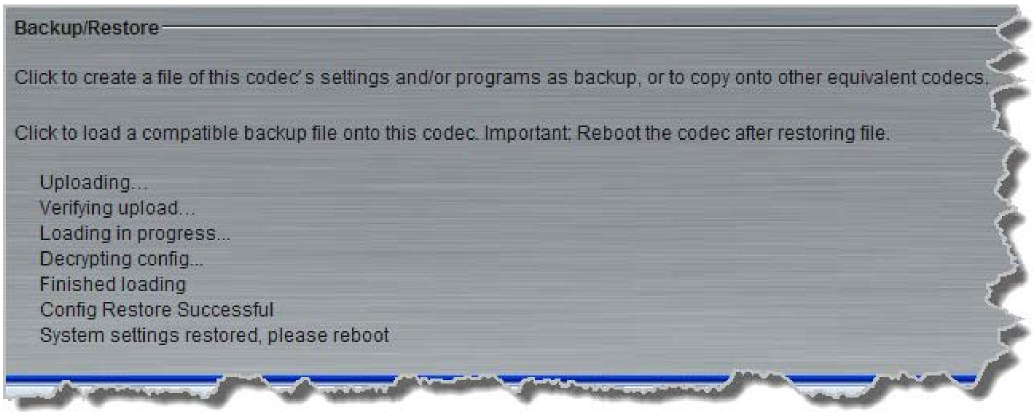
3. Click **Restore**.



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.



Note: The codec will automatically reboot to ensure the restored configuration takes effect in the codec.

20.18 Lock or Unlock Programs

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

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

4. Click **Save Settings** to save the new configuration.

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

20.19 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 Java Toolbox 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

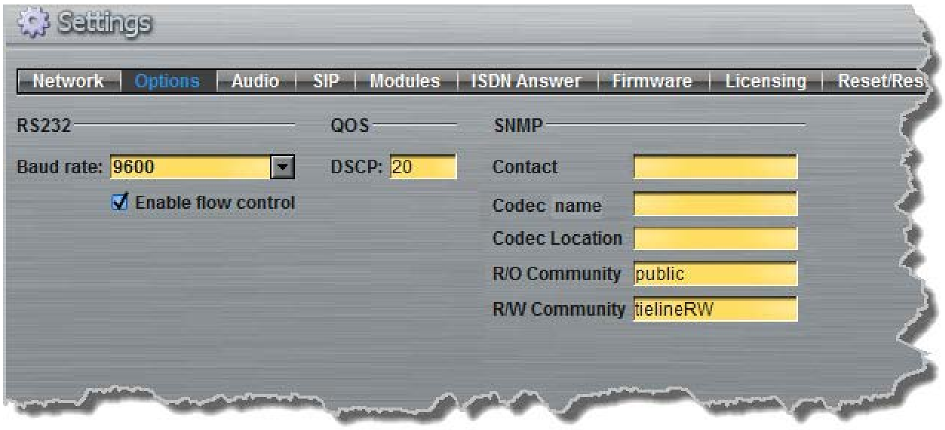
Features	Operation Button Descriptions
Codec Name	A user-specified alphanumeric identifier which may be used by third-party SNMP software to identify a device. The device name corresponds to the ".iso.org.dod.internet.mgmt.mib-2.system.sysName" SNMP attribute and is completely independent of DNS, NIS, 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 Java Toolbox 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 settings.

MIB Files for SNMP Configuration

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

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

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

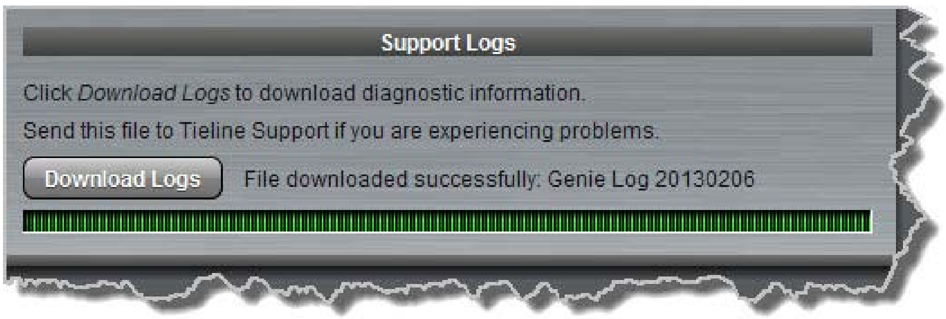
Important Note: The codec supports the attributes specified in the MIB-II standard. Please verify that your SNMP software contains the required files as specified in RFC 1213.

20.20 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 Java Toolbox Web-GUI and click the **Help** symbol at the top of the screen to display the **Help panel**.
2. Click **Download Logs**.



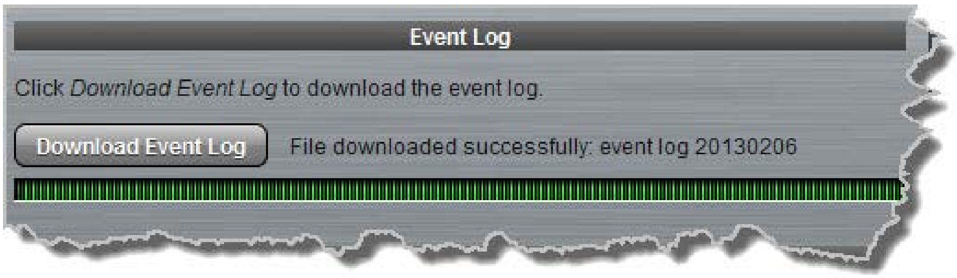
3. Save the file to your computer and then send it as a .zip file to Tieline support via

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



Clearing Logs

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

20.21 Configuring Alarms

Click the **Alarm symbol**  at the top of the Java Toolbox 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**.

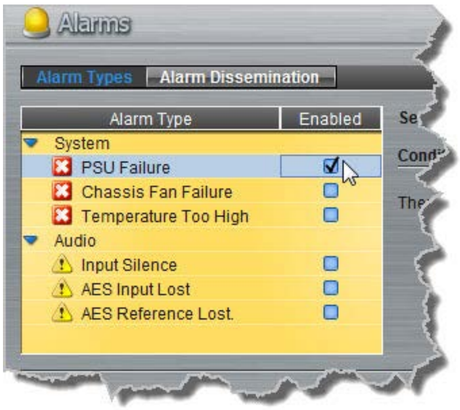
The following **System** and **Audio** alarms are available:

Alarm	Alarm Type	Explanation
PSU Failure	System	Raises an alarm if one or both PSUs fail
Chassis Fan Failure	System	Raises an alarm if the internal fan fails
Temperature Too High	System	Raises an alarm if the temperature is too high
Input Silence	Audio	Raises an alarm if input audio is lost (according to preconfigured silence detection threshold parameters)
AES Input Lost	Audio	Raises an alarm if the AES input signal is lost
AES Reference Lost	Audio	Raises an alarm if the AES reference clock signal is lost

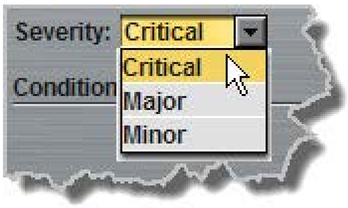
Configuring an Alarm's Severity Level

Codec alarms can be configured for 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.

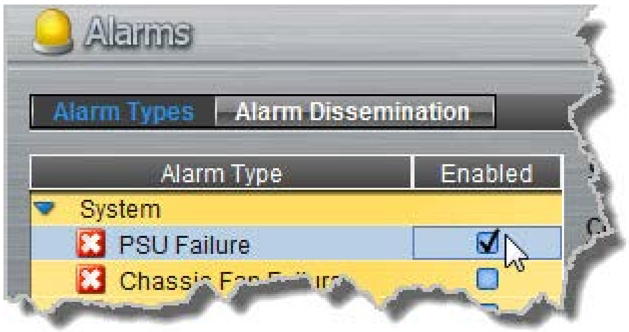


3. Perform this for each alarm you want to configure and then click **Apply** or **OK** to save settings.

Enabling Alarms

To enable and disable alarms:

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

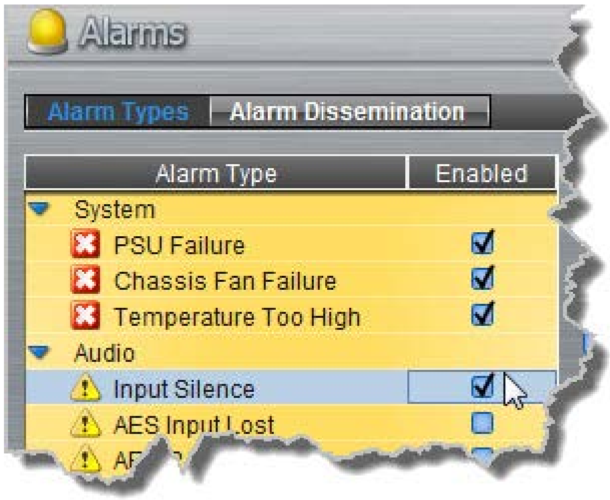


2. Click **Apply** or **OK** to save settings.

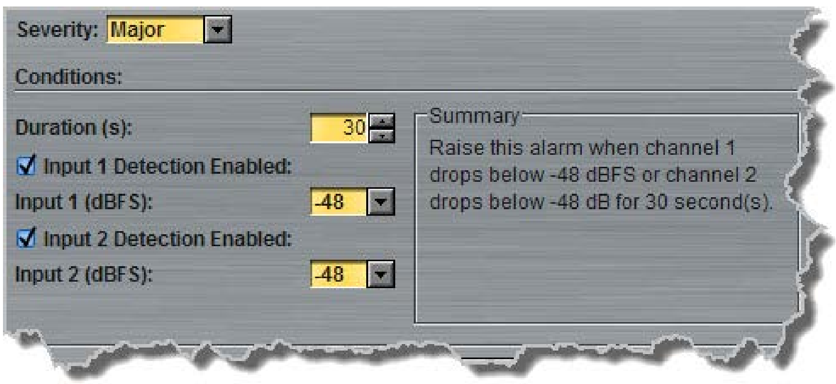
Configuring Input Silence Detection Parameters

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 and ensure the input check-boxes are selected. An alarm will be raised when these thresholds are breached.



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

Configuring Alarm Dissemination 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.

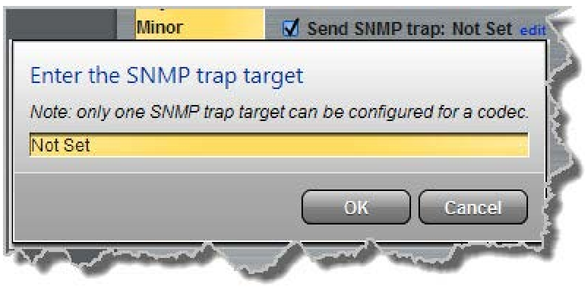
3. Click **Apply** or **OK** to save settings.

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.

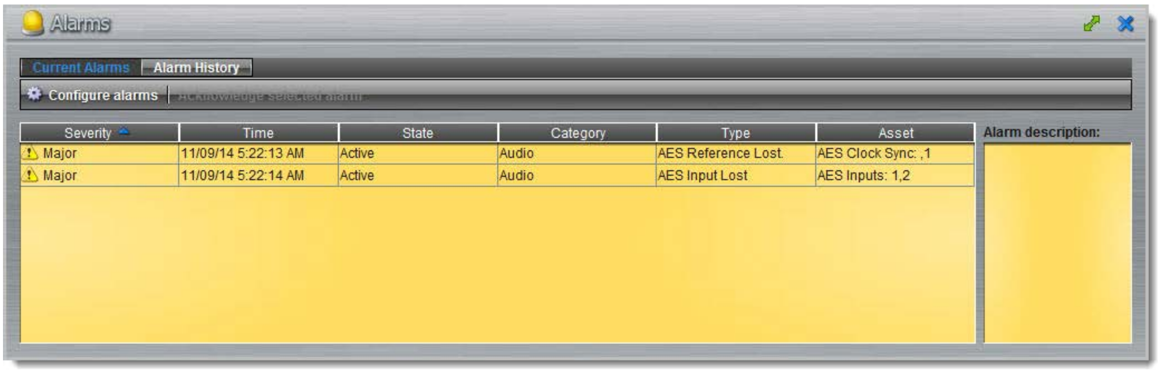
1. Click to select the **Send SNMP trap** check-box.

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



20.21.1 Managing Alarms

Active codec alarms are indicated in the **Current Alarms screen** in the Java Toolbox Web-GUI .



The user is alerted to active alarms by:

1. The **Alarm Symbol** flashing in the top right-hand corner of the Java 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.

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 Java Toolbox 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.

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

Deactivating Alarms

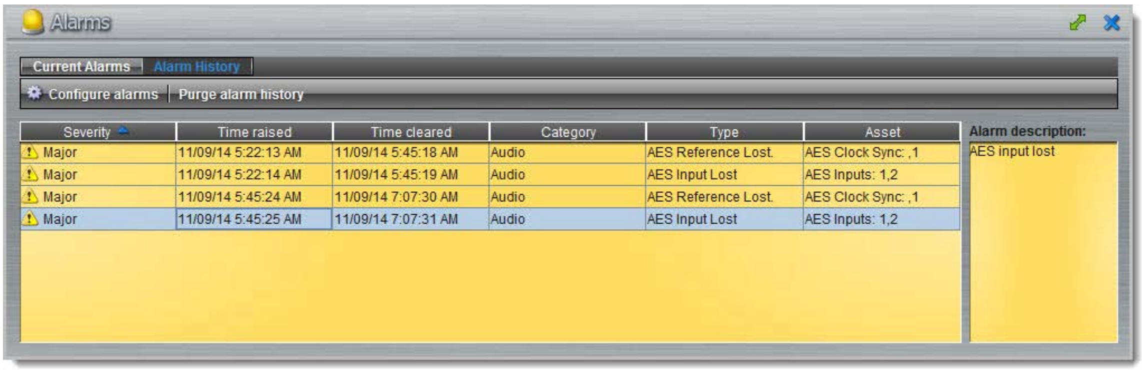
An alarm is deactivated automatically when the alarm state is reversed. E.g. if power is restored after a **PSU Failure** alarm, or if audio is restored after an **Input Silence** alarm.

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 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.

20.22 RS232 Data Configuration

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

Algorithm Selected	IP	ISDN and POTS
Tieline Music and MusicPLUS	• In-band RPTP data enabled automatically • Synchronized out-of-band data can be enabled and disabled	• In-band RPTP data enabled automatically
All other algorithms	• Synchronized out-of-band data can be enabled and disabled	• No in-band or out-of-band data available

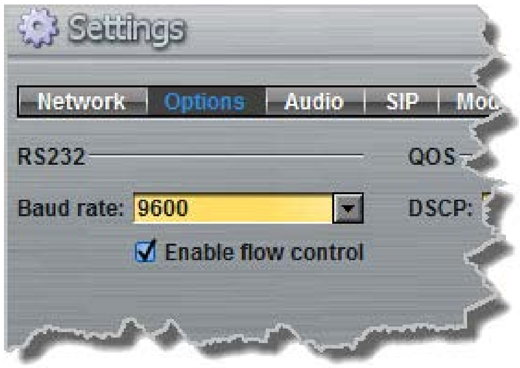
The codec can be connected to external devices and send RS232-compatible data via the serial port on the rear panel of the codec. To enable RS232 data within a connection, select **Enable Auxiliary Data** when creating a program in the **Programs panel** wizard. Alternatively, select using the codec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

Setting RS232 Data Rates and Flow Control

1. Open the Java Toolbox Web-GUI and click the **Settings** symbol at the top of the screen to display the **Settings panel**.
2. Click the **Options** button.
3. Click the **Baud rate** drop-down menu arrow to select the serial port baud rate which matches the

baud rate of the external device connected to the RS232 port on the codec.

4. Click to select the **Enable flow control** check box and enable flow control, then click **Save settings**.



Important Notes:

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

20.23 Creating Rules

The **Rules panel** in the Java 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.

Configuring Rules

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

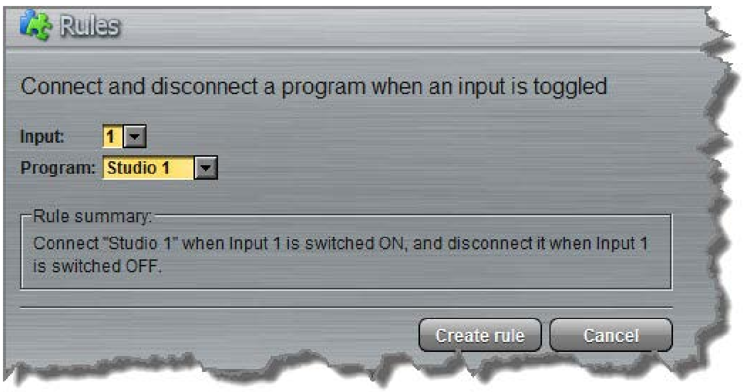
1. Click the **Rules** symbol at the top of the Java Toolbox Web-GUI screen to open the **Rules panel**.
2. Click **Add New Rule**.
3. Click to select the appropriate programming rule for your requirements. See the [Web-GUI Introduction] section for explanations of the actions each rule can perform.

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

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

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

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

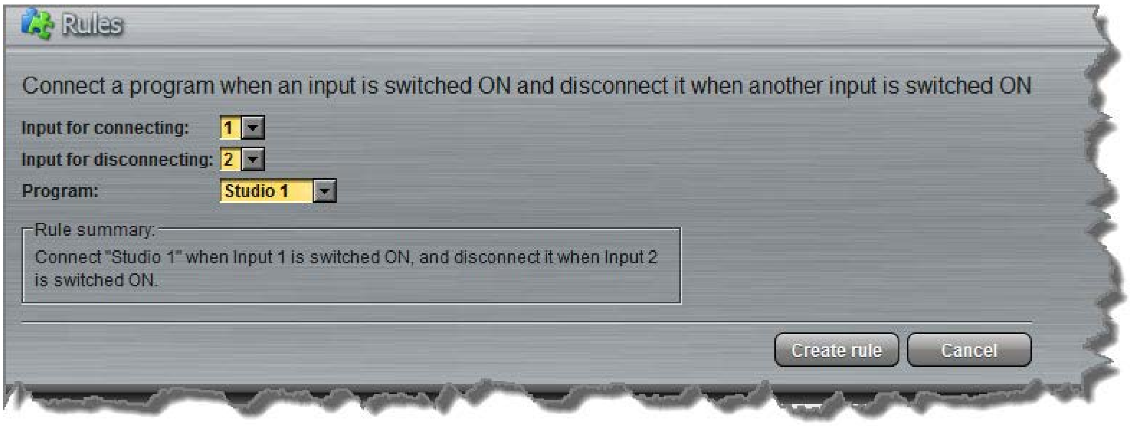


4. Check the **Rule Summary** and click **Create Rule** to save the settings.

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

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

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

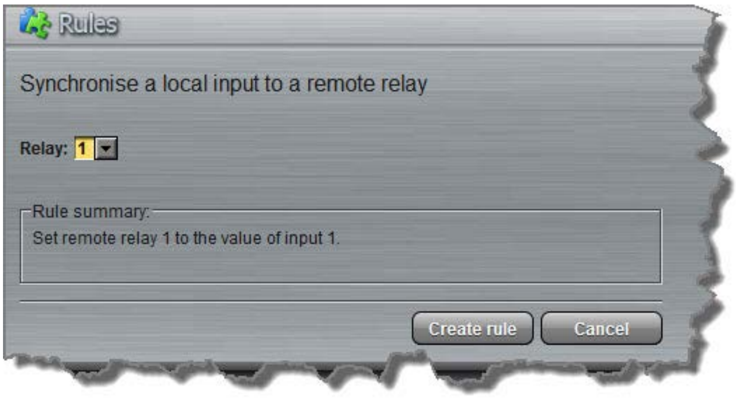


4. Check the **Rule summary** and click **Create Rule** to save the settings.

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

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

1. Click the fourth 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 fifth 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 Java Toolbox 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.

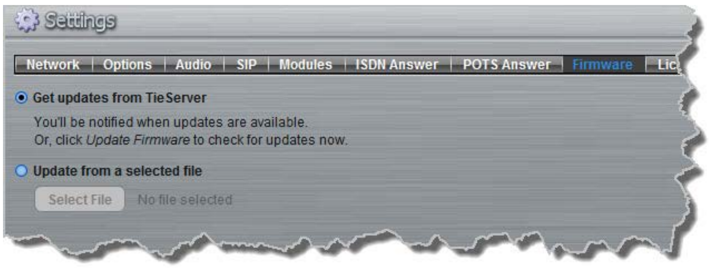
20.24 Upgrading Codec Firmware

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

Manual Firmware Upgrades

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

1. Click the **Settings** symbol at the top of the Java Toolbox 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.

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 Java Toolbox Web-GUI (See Connecting to the Web GUI)
2. If new software is available the **Update** symbol appears in the top-left of the screen.

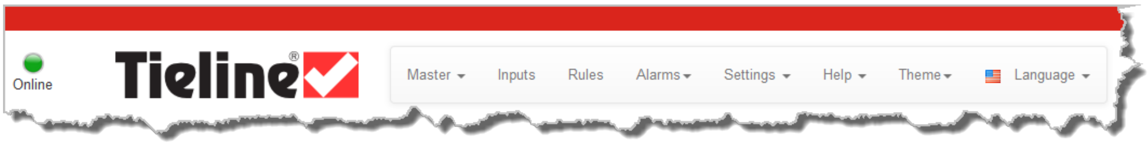
3. Position your mouse-pointer over the **Update** symbol and click the update dialog when it appears to download the new software.

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

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

21 Using the HTML5 Toolbox Web-GUI

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

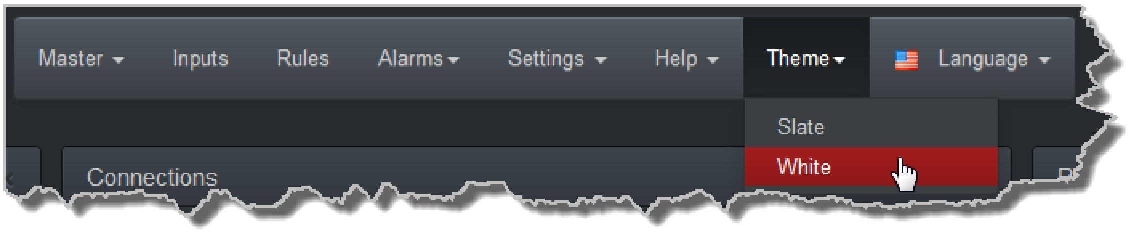


HTML GUI Menu Bar for Opening Panels

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

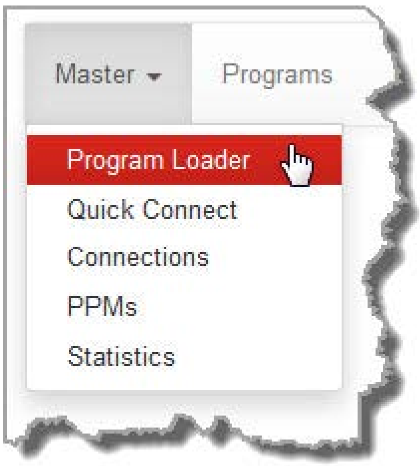
Adjusting the Theme

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



Opening a Panel & Adjusting Screen Position

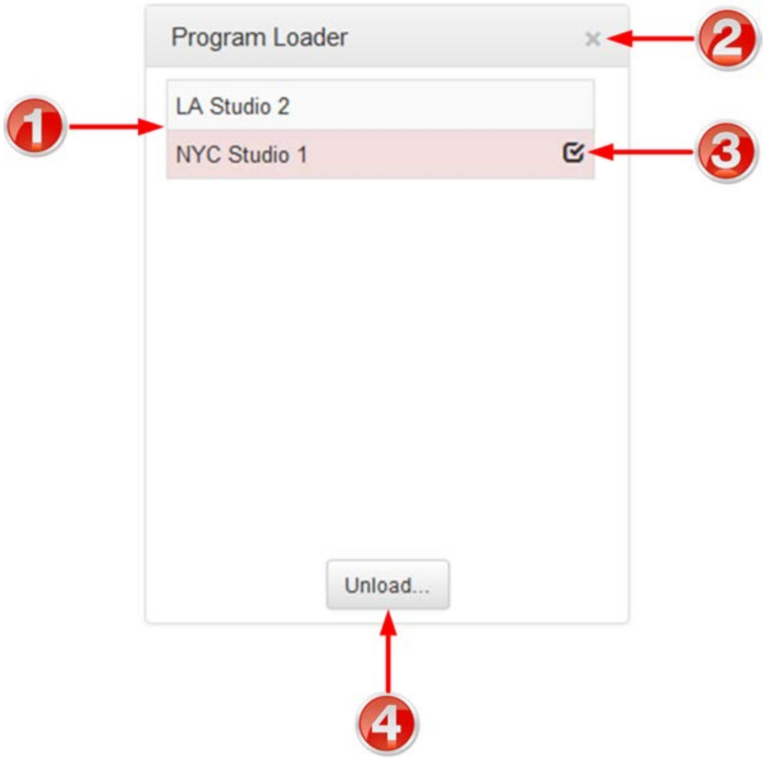
Click an item in the **Menu bar** to display available panel options, then click to select and open a panel. New panels automatically open in the top left of the screen.



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

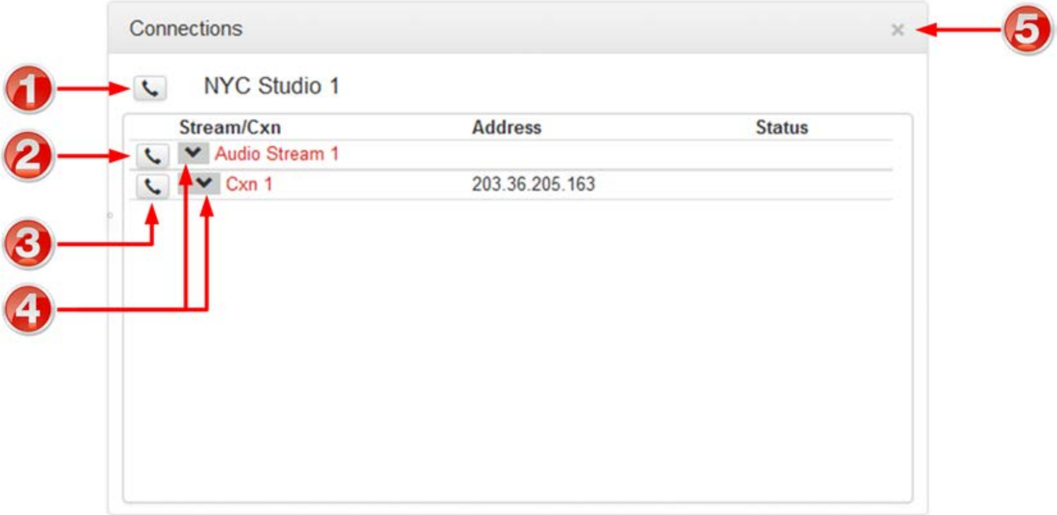
Master Panels: Load Programs & Manage Audio Streams

Program Loader Panel



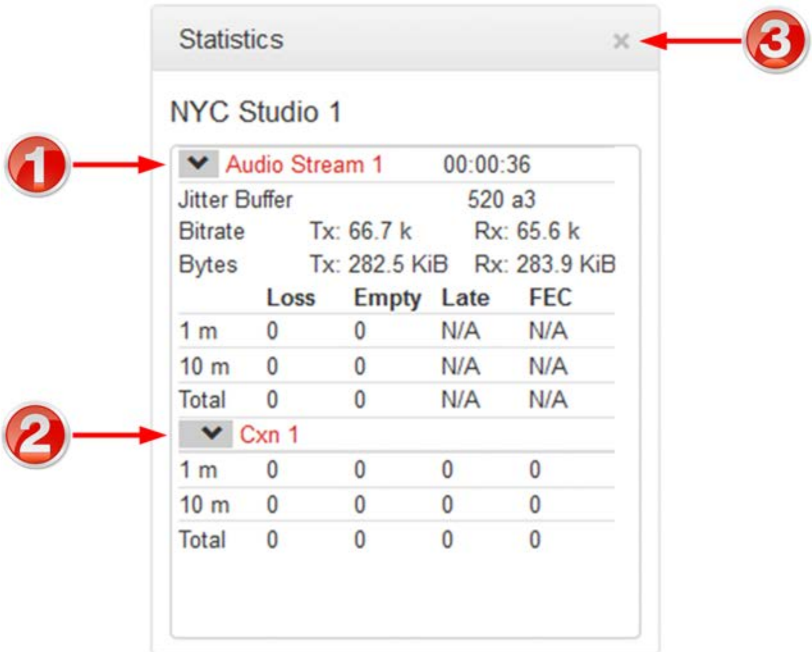
	Feature	Description
1	Programs list	Lists all configured programs which have been added into the codec. Click to select a program before loading.
2	**Close** button	Click to close the panel.
3	**Check-box symbol**	The **Check-box symbol** identifies the currently loaded program in the codec.
4	**Load**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

Connections Panel



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

Statistics Panel

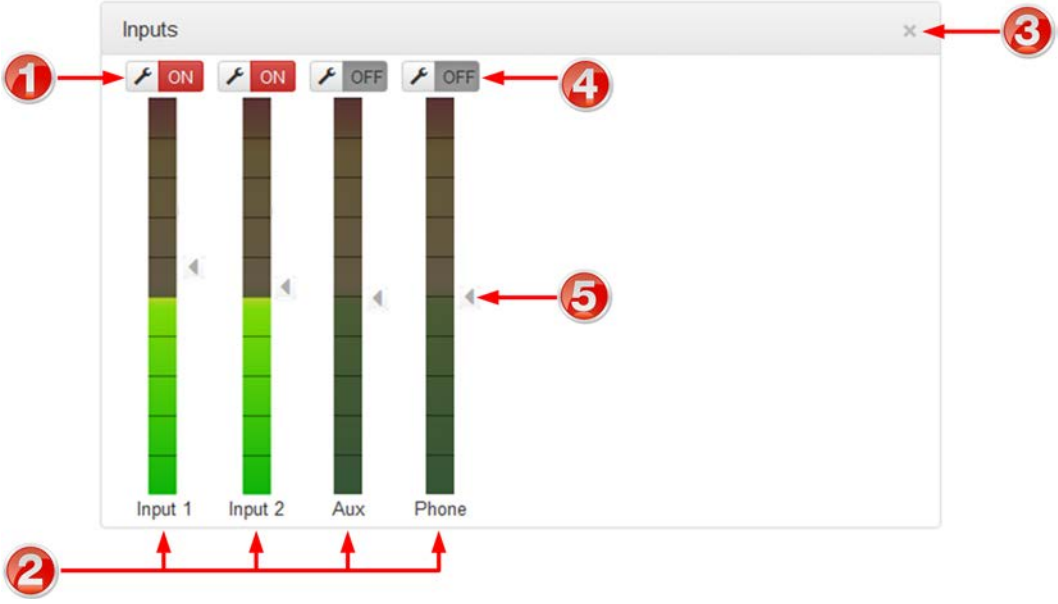


	Feature	Description
1	Show/Hide Arrow	Click to show/hide audio stream statistics.
2	Show/Hide Arrow	Click to show/hide individual connection statistics.
3	**Close** button	Click to close the panel.

PPMs Panel

	Feature	Description
1	PPM Meters	6 PPM meters.
2	**Close** button	Click to close the panel.

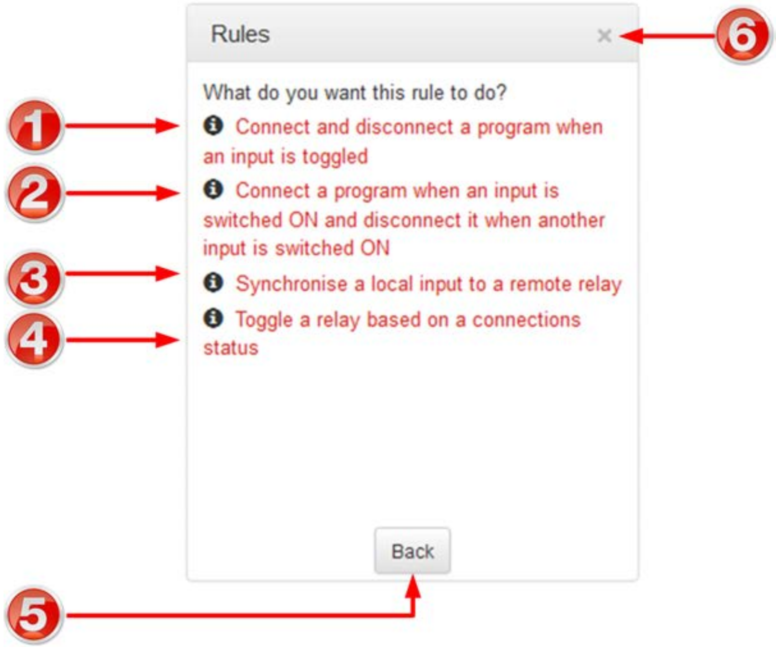
Inputs Panel for Input Adjustments



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

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

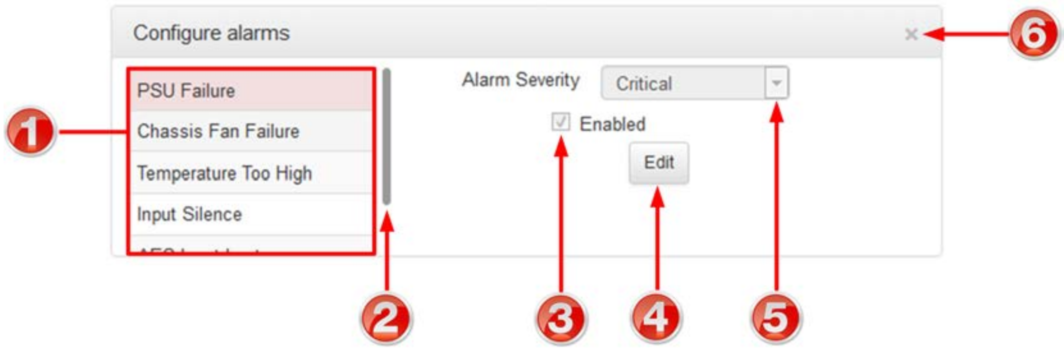
Rules Panel for Creating Relay Activation Rules



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

Alarm Panels: Configure & Monitor Alarms

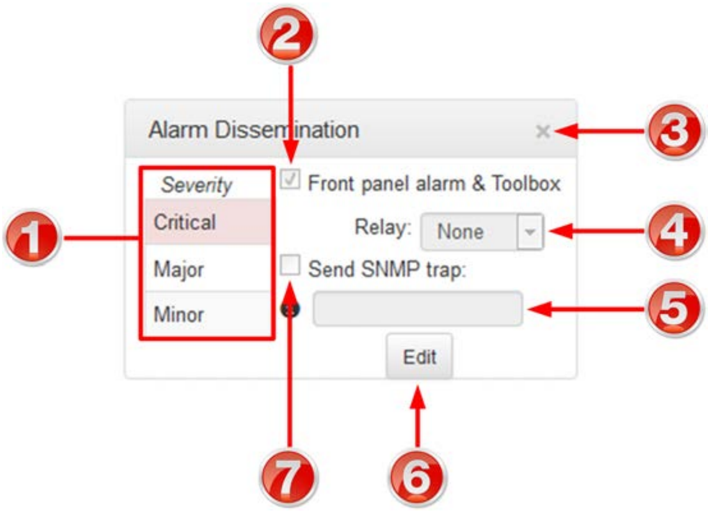
Configure Alarms Panel



	Feature	Description
1	List of alarm types	Click to select an alarm type to configure.
2	Scroll bar	Click and drag the scroll bar to view all listed alarms.
3	Enable Alarm check-box	Click the **Enabled** check-box to enable the currently selected alarm.
4	**Edit** / **Save** button	Click to edit an alarm, or save configured alarm settings when in edit mode.
5	Alarm Severity Setting	Click the drop-down arrow to select an alarm severity setting.
6	**Close** button	Click to close the panel

Alarm Dissemination Panel

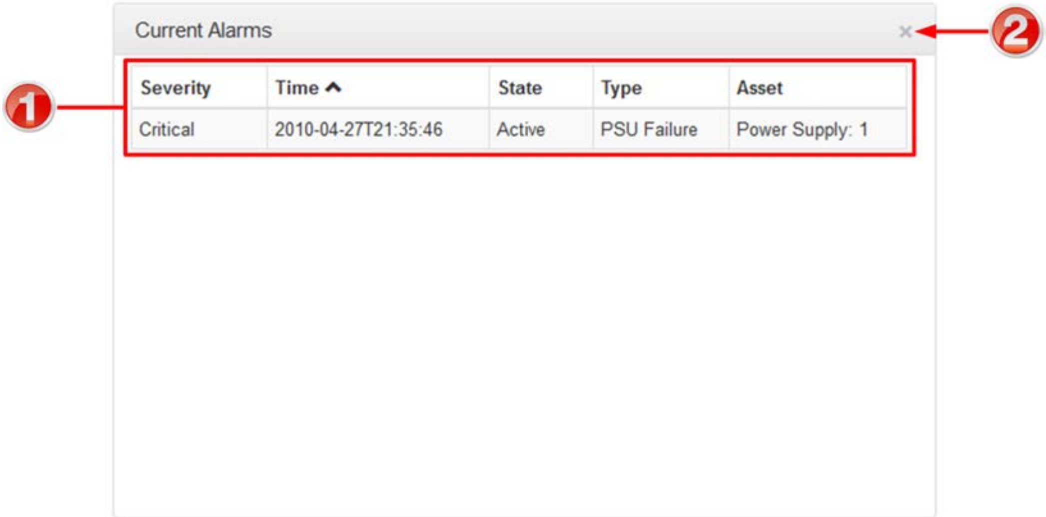
Alerts for each alarm severity level are configured using the **Alarm Dissemination panel**.



	Feature	Description
1	List of alarm severity levels	Click to select an alarm severity level to configure it.
2	**Front panel alarm & Toolbox** check-box	Select the check-box (default enabled) to deliver front panel **ALARM LED** notifications and HTML5 Toolbox Web-GUI alarm notifications.
3	**Close** button	Click to close the panel.
4	**Relay** drop-down selection	Click the drop-down arrow to select a relay to open when an alarm using the current severity level is activated.

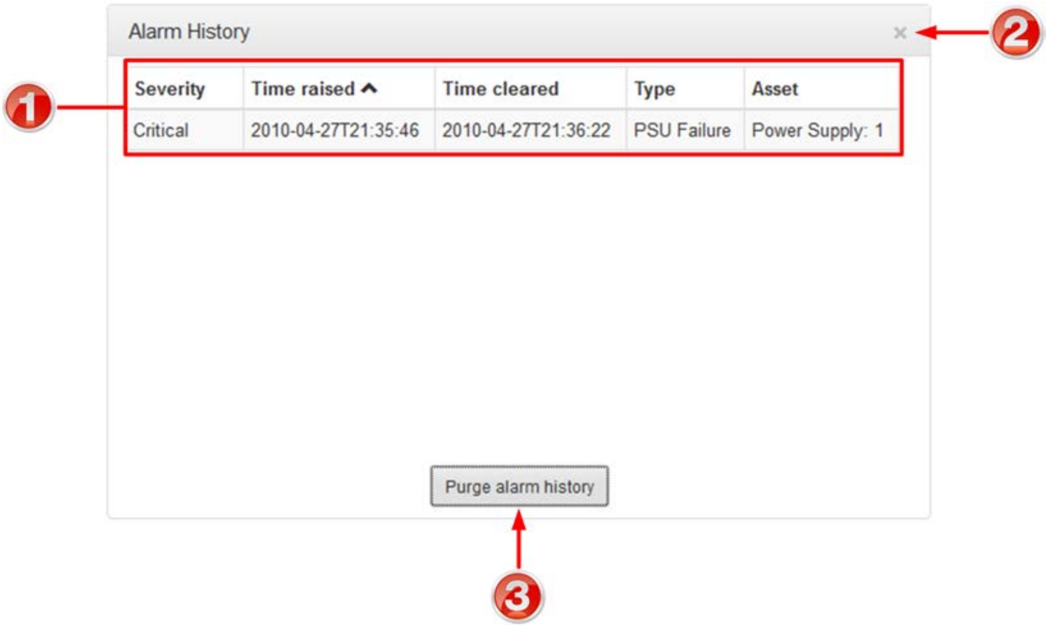
5	SNMP Trap Target text-box	Click in the text box in edit mode to enter the SNMP trap target for alarms using the currently selected severity level.
6	**Edit** / **Save** button	Click to edit alarm dissemination settings, or save configured settings when in edit mode.
7	**Send SNMP trap** check-box	Select the check-box to enable SNMP traps to be sent (for alarms using the selected severity level).

Current Alarms



	Feature	Description
1	Current alarm description	View a list of active alarms in the codec
2	**Close** button	Click to close the panel

Alarm History

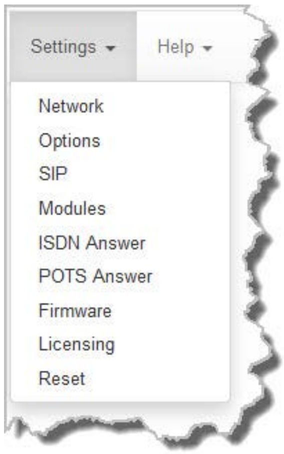


	Feature	Description
1	Alarm history description	View the history of previous alarms in the codec.
2	**Close** button	Click to close the **Alarms** panel.
3	**Purge History** button	Click to clear the alarm history.

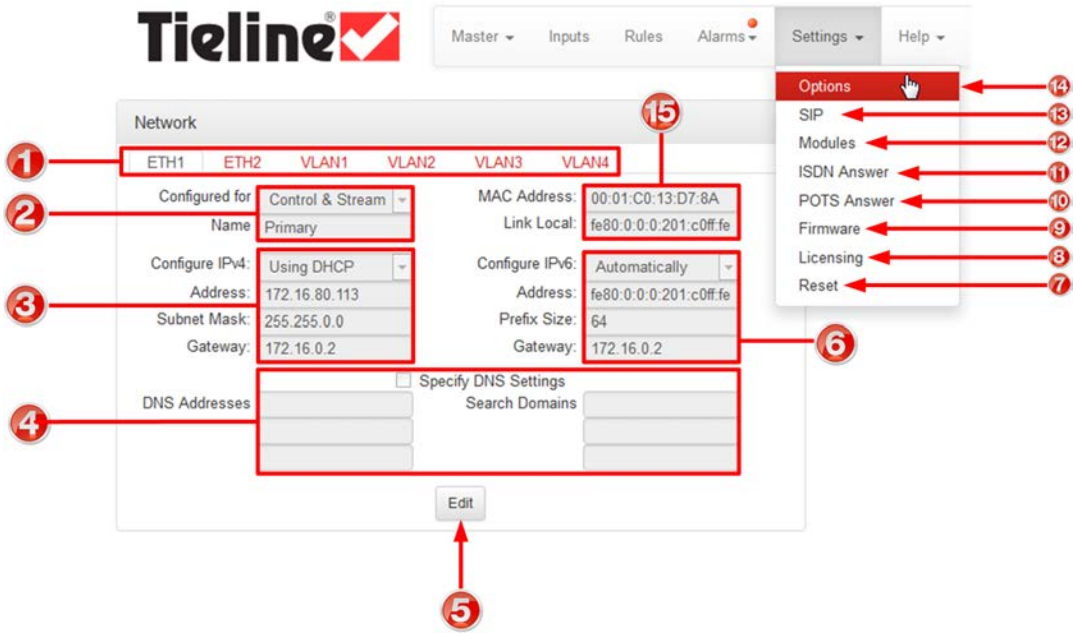
Settings Panels

There are 9 **Settings** panels which can be opened in the Web-GUI. Each panel provides specific codec configuration settings and options. Click to select and open each panel.

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



Settings panels

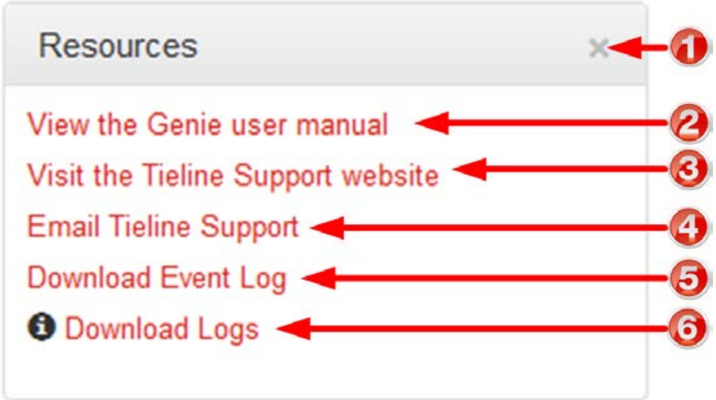


	Feature	Description
1	**Network tab**	Click to select and edit, or view network configuration settings for each Ethernet and VLAN interface.
2	Network Interface	Control and streaming configuration options for each network interface.
3	IPv4 details	IPv4 address details and configuration.
4	DNS details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

9	**Firmware tab**	Click to open the panel; view software versions and perform an upgrade.
10	**POTS Answer tab**	Click to open the panel and configure POTS Answering settings.
11	**ISDN Answer tab**	Click to open the panel and configure ISDN Answering settings.
12	**Modules tab**	Click to edit hardware module configuration.
13	**SIP tab**	Click to open the panel and edit or view SIP configuration settings.
14	**Options tab**	Click to open the panel; configure RS232 and QoS data settings, lock a loaded user Program and adjust Session Port settings and SNMP. Also configure the AES Output Clock sample rate.
15	**MAC Address / Link Local**	Click to open the panel and view the device MAC address and IPv6 local network address created by the codec.

Help Panels

Resources



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

About Panel

Details of the codec Toolbox and firmware version, as well as the codec serial number. Note: the codec name displayed will vary by product type.

Language Selection

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

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

21.1 Using the HTML5 Toolbox Quick Connect Web-GUI

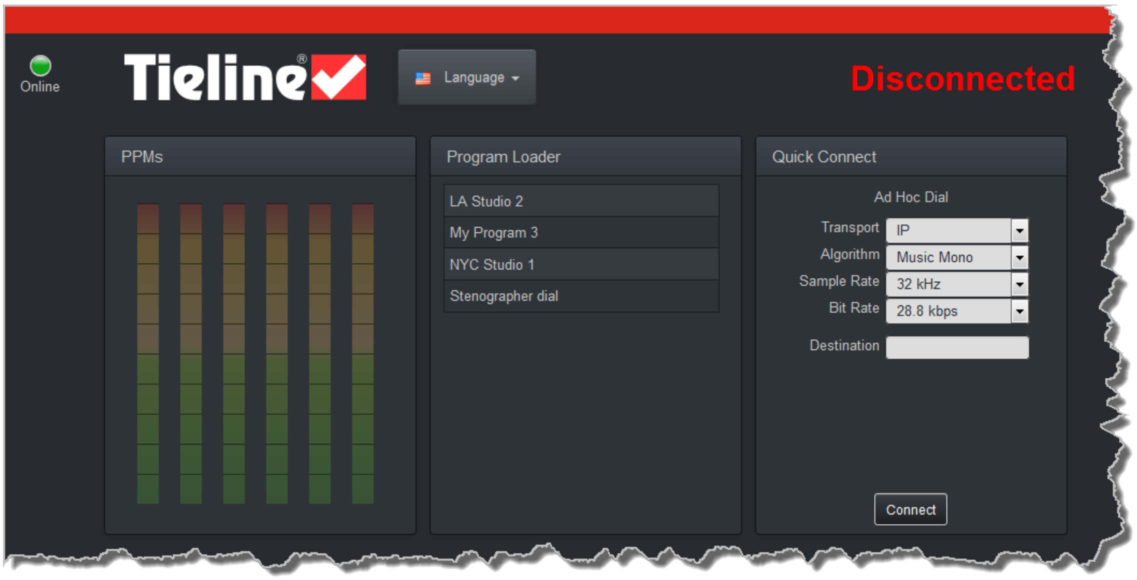
The HTML5 Quick Connect Web-GUI is designed for simple peer-to-peer connections and non-technical users. It has a reduced feature-set and allows users to:

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

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

Launching the HTML5 Quick Connect Web-GUI

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

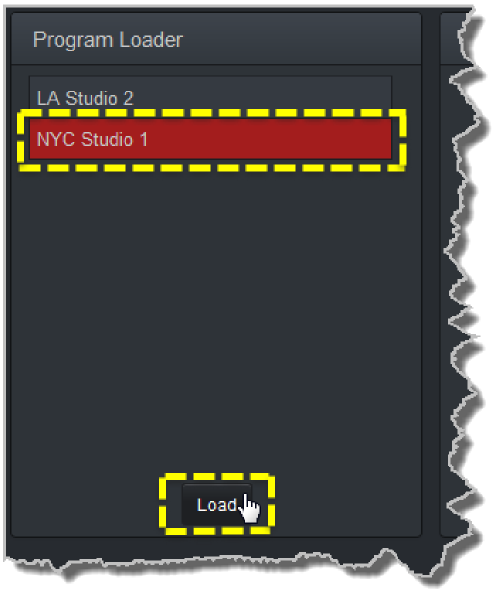


Important Note: To change the password using the codec front panel navigate to **Settings > WebGUI > Password** and press the OK button. Use the keypad to enter a new password and press the OK button to save the new setting.

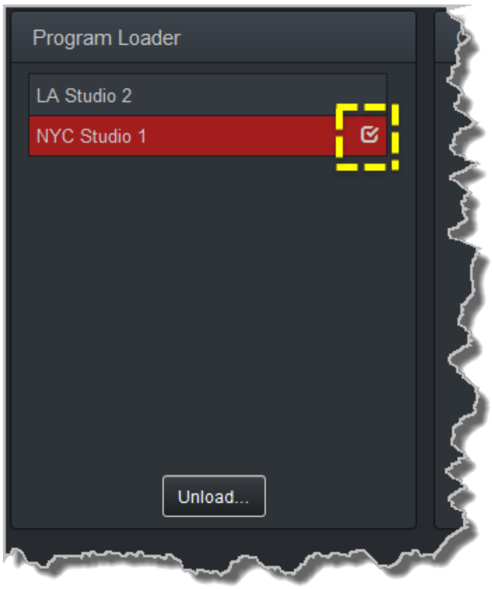
Loading and Unloading an Existing Program

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

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



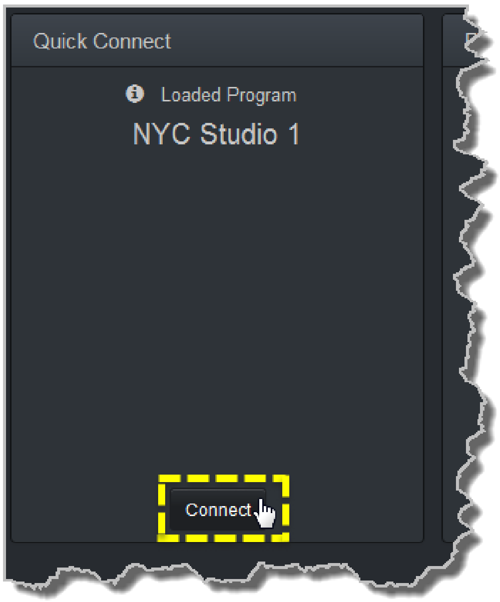
2. The **Check-box symbol** appears next to the program name to confirm it has been loaded and the **Load button** changes to an **Unload button**.



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

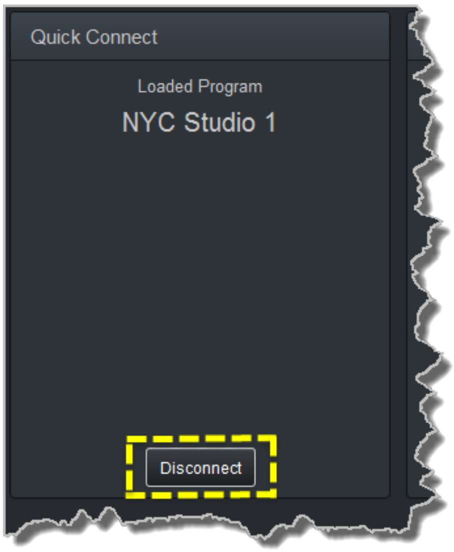
Dial a Loaded Program

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

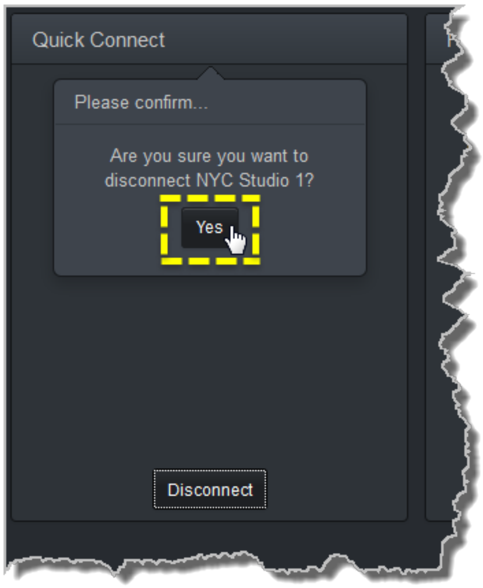


Disconnect a Loaded Program

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



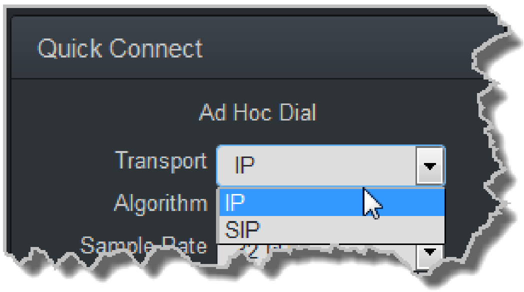
2. Click **Yes** in the confirmation dialog to disconnect the connection.



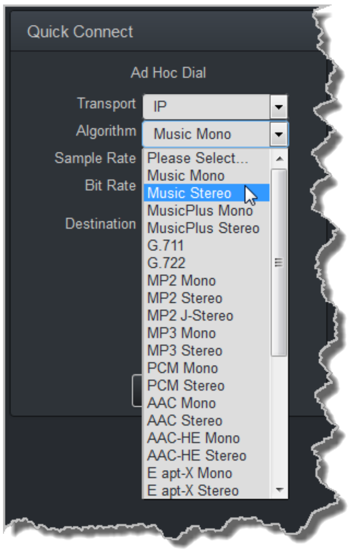
Dial Peer-to-Peer over IP with Quick Connect

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

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

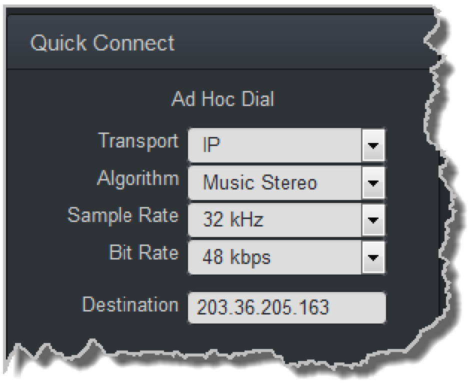


2. Click the drop-down **Algorithm** menu and select an algorithm.



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

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



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

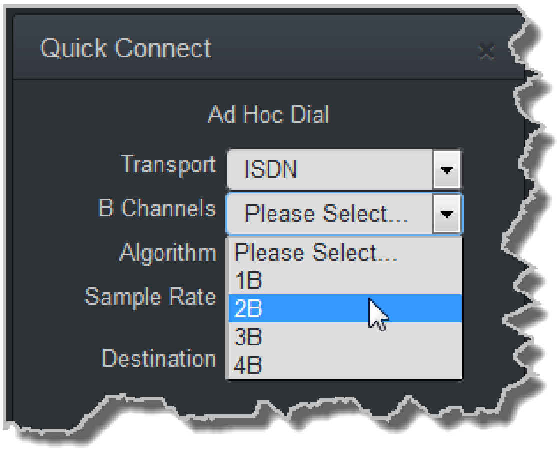
Dial Peer-to-Peer over ISDN with Quick Connect

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

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

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

3. Click the drop-down **B Channels** menu and select the number of channels required for this connection.



4. Click to select the appropriate **Sample Rate** for the connection.

5. Click in each **Destination** text box in turn to enter the ISDN number for each B Channel.

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

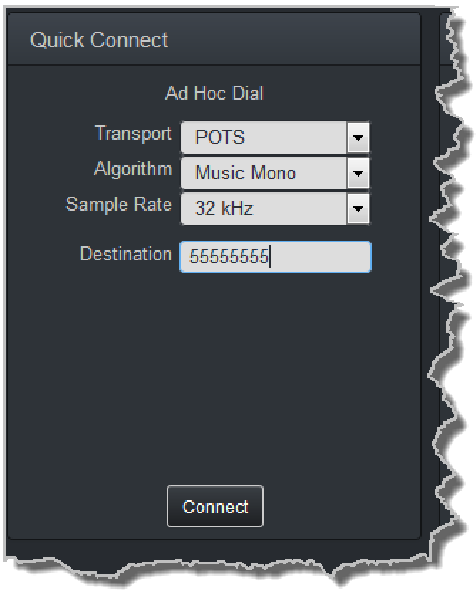
Dial Peer-to-Peer Over POTS with Quick Connect

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

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

2. Click the drop-down **Algorithm** menu and select an algorithm. The connection bit rate is configured automatically. Note: G5 POTS modems initially attempt to establish a link at the lowest **Max Bit rate** setting configured in the two modules being connected. If the POTS line doesn't support this bit rate, the modems will attempt to connect at the highest possible bit rate to suit the prevailing line quality at each end of the link.

3. Enter the phone number in the **Destination** text box.



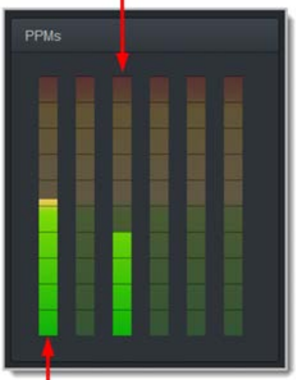
4. Click the **Connect button** to dial.

Monitoring PPMs

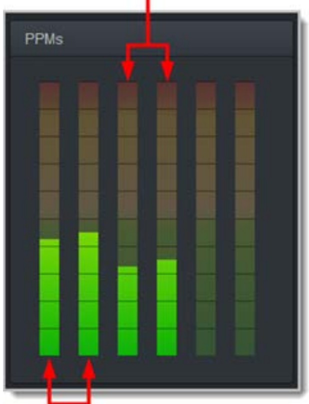
Set audio levels so that audio peaks average at the nominal 0vu point indicated below on the PPM meters. This represents a program level of +4 dBu leaving the codec. Audio peaks can safely reach +22 dBu without clipping, providing 18dBu of headroom from the nominal 0vu point. Note: the audio metering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

Mono and Stereo PPM Metering

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



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



21.2 Configuring IP Settings

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

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

IPv4 versus IPv6

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

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

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

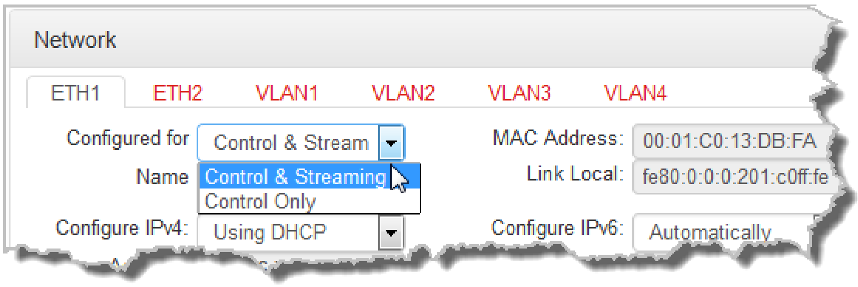
Configuring Ethernet Ports and VLANs

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

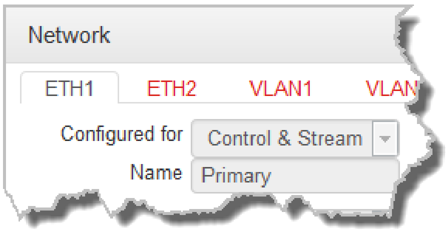
VLAN interfaces have features similar to physical Ethernet interfaces. However, your network administrator will need to configure VLAN support throughout your network for them to be supported in your codec.

As an example, if only one physical Ethernet interface is available, VLANs can be used to operate SmartStream PLUS or to separate codec **Control and Streaming** functions if required. Ethernet and VLAN interfaces can be configured for:

- 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.
- Streaming audio: stream audio only from an Ethernet port (**ETH2** and **VLANs** only).
- Nothing: Disable the Ethernet port from streaming audio and codec command and control (**VLANs** only).

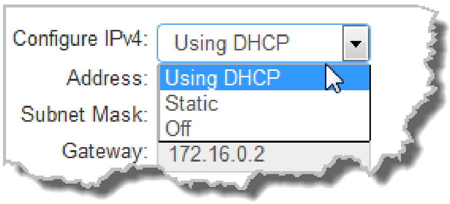


The **Name** text box, e.g. **Primary** or **Secondary**, is an interface identifier used when configuring new programs via the **Programs panel**.

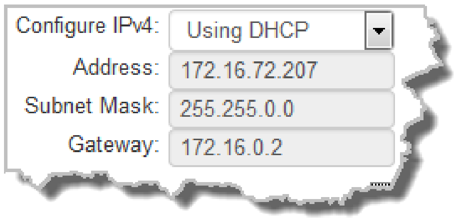


IPv4 Address Configuration

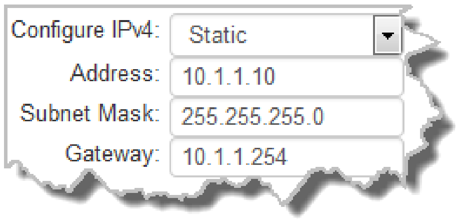
Click the **Edit** button in the **Network** panel to configure settings. The codec is capable of automatic DHCP address assignment, or manually configured static IPv4 address configuration via the drop-down **Configure IPv4** menu. If you want to ignore IPv4 settings select **Off**.



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

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

IPv6 Address Configuration

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

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

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

Types of IPv6 Addresses

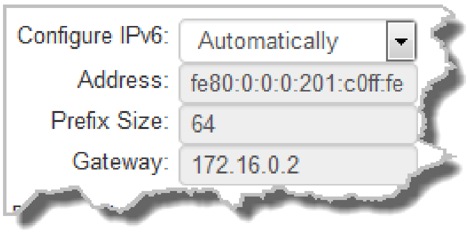
There are two types of addresses displayed in the IPv6 section:

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

over a LAN. This address is allocated by the codec internally based on MAC address details.

Auto Address Assignment

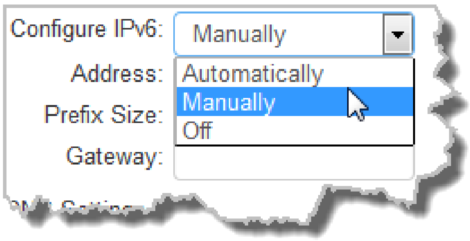
1. Click the **Edit** button in the **Network** panel to configure settings.
2. By default the codec is programmed for connecting to an IPv6 router which automatically allocates IPv6 address details, as displayed in the following example.



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

Manual IPv6 Address Assignment

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

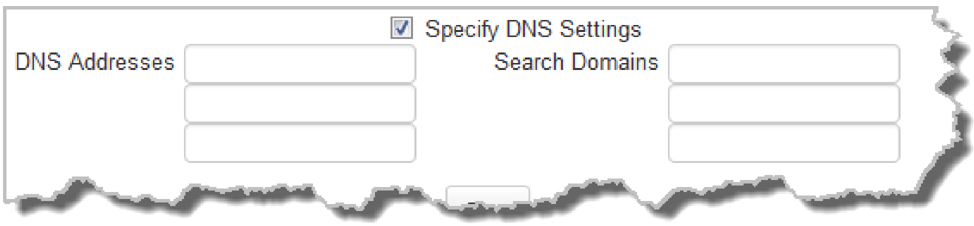


3. Click **Save** to store all configuration settings.

Specifying DNS Settings

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

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

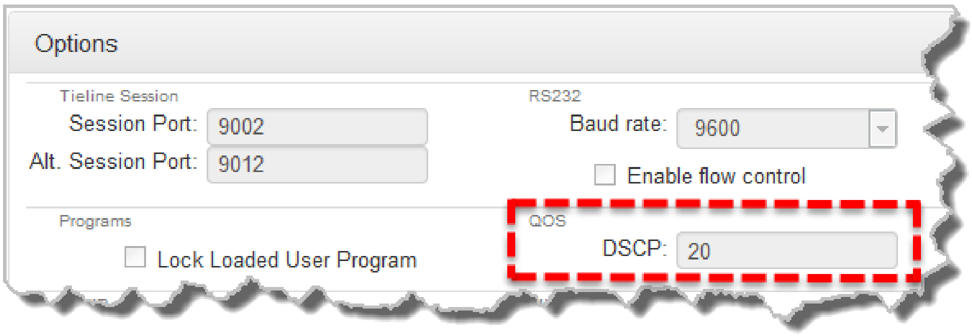


2. Click **Save** to store all configuration settings.

Configuring QoS

1. Open the HTML5 Toolbox Web-GUI and click **Settings** and then click **Options** to open the **Options panel**.

2. Click the **Edit** button.
3. Click in the **DSCP** field and enter the priority setting recommended by your IT administrator.
4. Click **Save** to store configuration settings.



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

Important Note: Check with your IT administrator before changing this setting. By default the codec is configured for Assured Forwarding and more details about DSCP are available on Wikipedia at http://en.wikipedia.org/wiki/Dscp.

21.3 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 HTML5 Toolbox Web-GUI. See About ISDN Modules for additional information on ISDN.

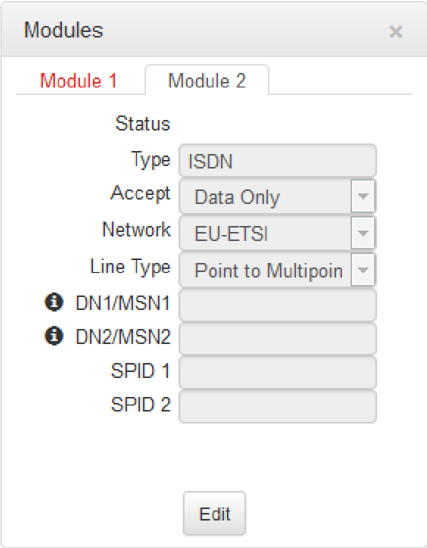
You can use the Java Toolbox Web-GUI to configure a dial and/or answer program with ISDN settings. You may also need to:

1. Configure ISDN module settings.
2. Configure ISDN Answering settings.

21.3.1 Configuring ISDN Modules

ISDN settings in the **Modules panel** determine how each codec module operates at a particular site. You can copy similar programs between codecs installed at different locations and also configure site-specific settings for how each ISDN module should connect. ISDN module settings may need to be adjusted depending on your country and network requirements.

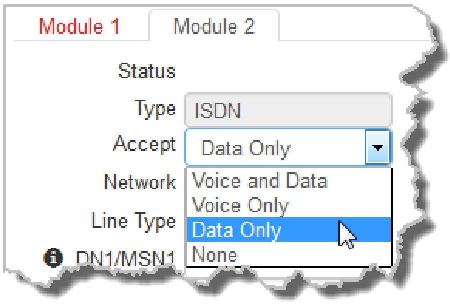
1. Open the HTML5 Toolbox Web-GUI and click **Modules** to view and configure ISDN site settings.



2. Click to select **Module 1** or **Module 2**.

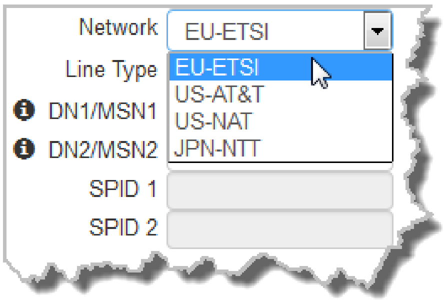
3. Click the **Edit** button to configure settings.

4. Click the drop-down arrow for **Accept** to select whether to allow or disallow circuit switched voice and data calls. The default setting allows **Data only**.

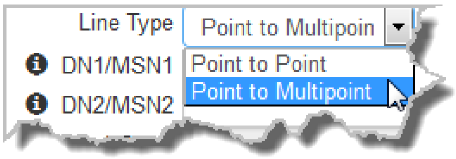


Important Note: G.711 is the algorithm used when **Voice Only** is selected.

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



6. Click the drop-down **Line Type** arrow 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.



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

8. Click **Save** 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.

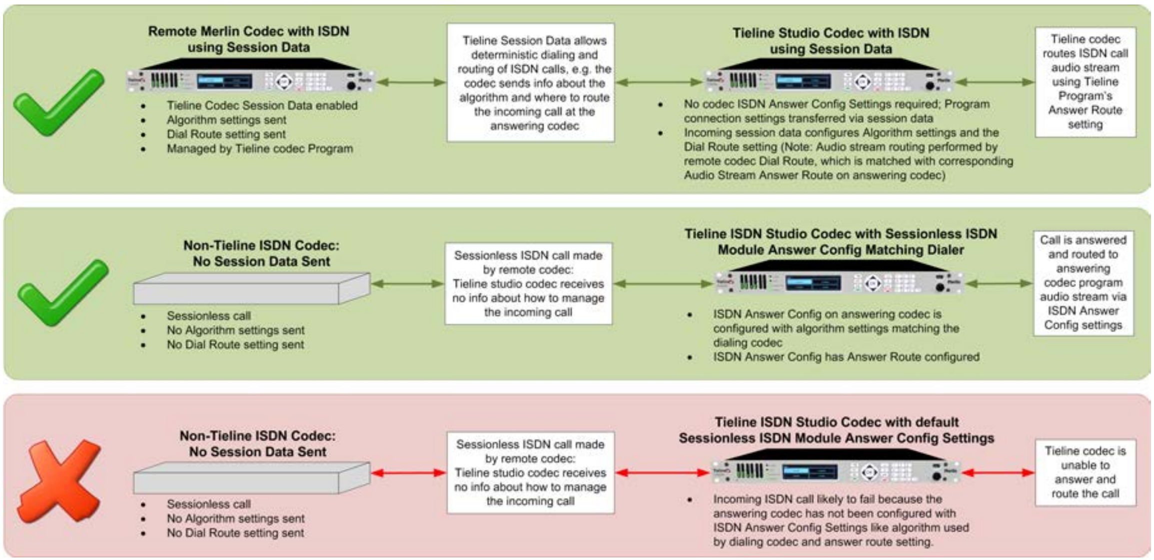
Typically, each ISDN BRI service in the US will have two SPIDs and these must be entered correctly. When you enter a SPID into your codec and connect it to an ISDN line, an initialization and identification process takes place, whereby the terminating equipment (your codec) sends the SPID to the switch.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

21.3.2 Configuring ISDN Answering

ISDN Answer Configs are used to determine how codec ISDN modules will behave when answering ISDN calls.

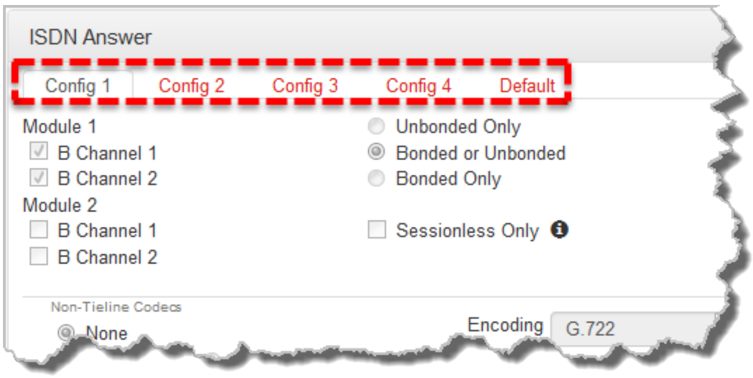
The following image explains the difference between answering calls from Tieline codecs sending session data, and non-Tieline codecs making sessionless ISDN calls. Codecs sending Tieline Session Data contain all the information required to connect, e.g. algorithm and audio stream routing settings. When answering sessionless calls it is necessary to configure the answering codec with an **ISDN Answer Config**, which tells the answering codec how a sessionless call will try and connect.



It is possible to save up to four different **ISDN Answer Configs**, which allow 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 **Dial** and **Answer Route** tags.
- The type of call being received by the codec, e.g. Tieline (with Tieline Session Data) versus non-Tieline sessionless calls.
- The algorithm expected when receiving sessionless calls.

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.



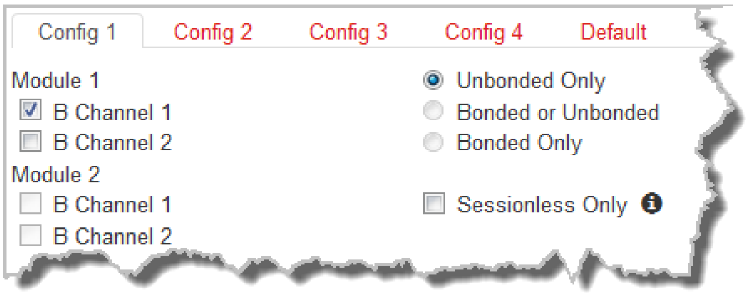
Important Note: B channels can only be selected once and are greyed out once they have been selected in one of the four ISDN **Configs**.

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):

1. Click the **Edit** button to configure settings.
2. Select a B channel from those available and then click **Save**. **Unbonded Only** is the default

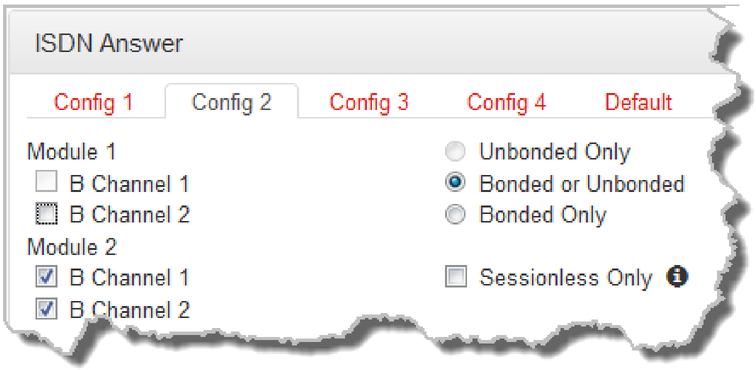
setting if only one B channel is selected.



Multiple B Channel Bonding Config

A point-to-point audio stream can also bond multiple B channels to create higher bandwidth connections.

1. Click the **Edit** button to configure settings.
2. 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 greyed out and unavailable in **Config 2**.



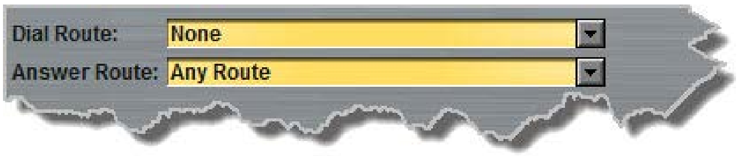
3. Configure the bonding setting that best suits the audio stream with which this **Config** is associated. **Bonded or Unbonded** is the best setting in most situations.

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
Bonded Only	Will only bond compatible algorithms. This mode will reject incompatible calls which cannot be bonded, e.g. G.711 and G.722

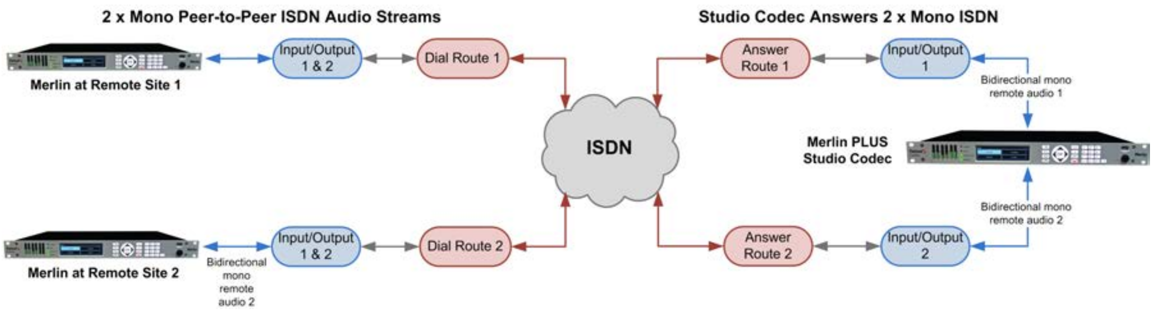
3. Click **Save** to apply changes to the **Config**.

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. 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.



In principle, the concept of 'routes' 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 codec audio streams, and therefore the same inputs and outputs. Following is an example of how to consistently route incoming ISDN audio streams using dial and answer routes.

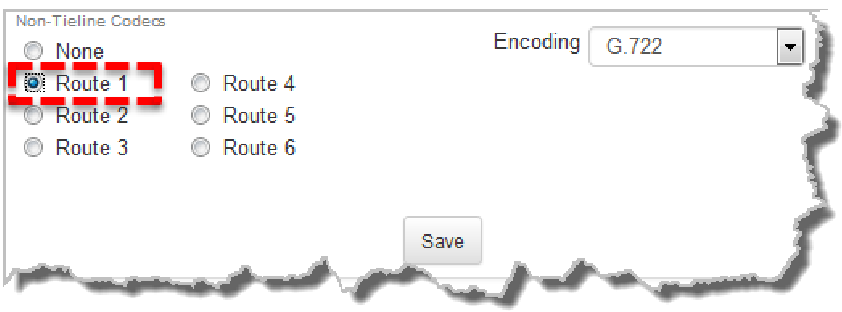


Answer Routes for Non-Tieline (Sessionless) ISDN Calls

In some situations you may receive a call from a non-Tieline codec which doesn't support session data and **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.

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

1. Click the **Edit** button to configure settings.
2. 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**).



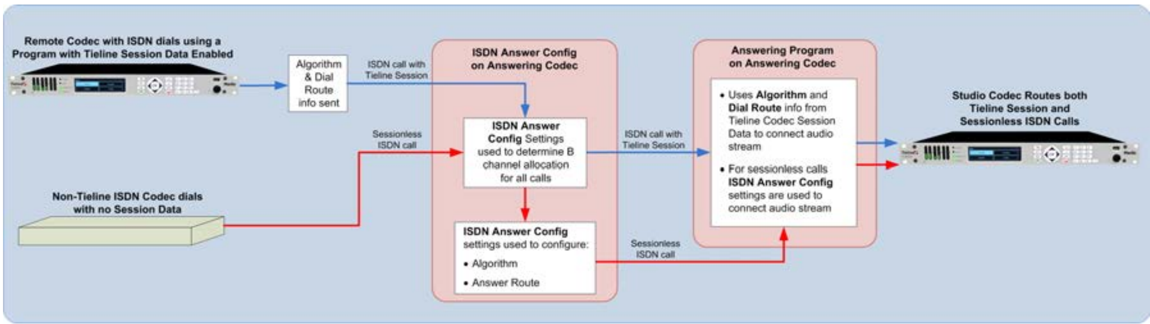
3. Click **Save** when configuration is complete to store the new **Config** settings.
4. This will associate the incoming call with a corresponding **Answer Route** configured in the answering codec program, e.g. **Answer Route 1**.

More detailed information about how to configure the codec to answer and route multiple sessionless ISDN calls is available in Using ISDN Answer Routes for Sessionless ISDN Calls. This uses examples to explain how to set up consistent deterministic routing of multiple incoming sessionless calls.

Answering both Tieline Session and Sessionless ISDN Calls

Leave the **Sessionless Only** check-box in the **ISDN Answering Config** unchecked if the codec is expected to receive ISDN calls from Tieline codecs, or both Tieline and non-Tieline codecs (i.e. you are not sure which type of codec may call). In this mode, when the codec answers a call it initially expects to receive Tieline session data from the dialing codec and configure its own algorithm settings according to that. If it fails to receive Tieline session data within 5 seconds (i.e. a non-Tieline codec is calling, or a Tieline codec with session data disabled), it will use the settings in the **ISDN Answering Config** instead.

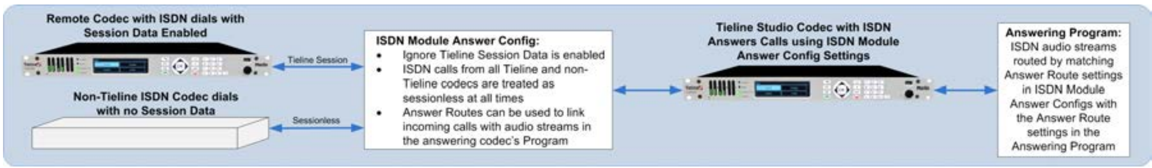
The following image displays how the answering codec will behave in this mode when receiving calls from both Tieline and non-Tieline codecs.



Allow Answering of Sessionless ISDN Calls Only

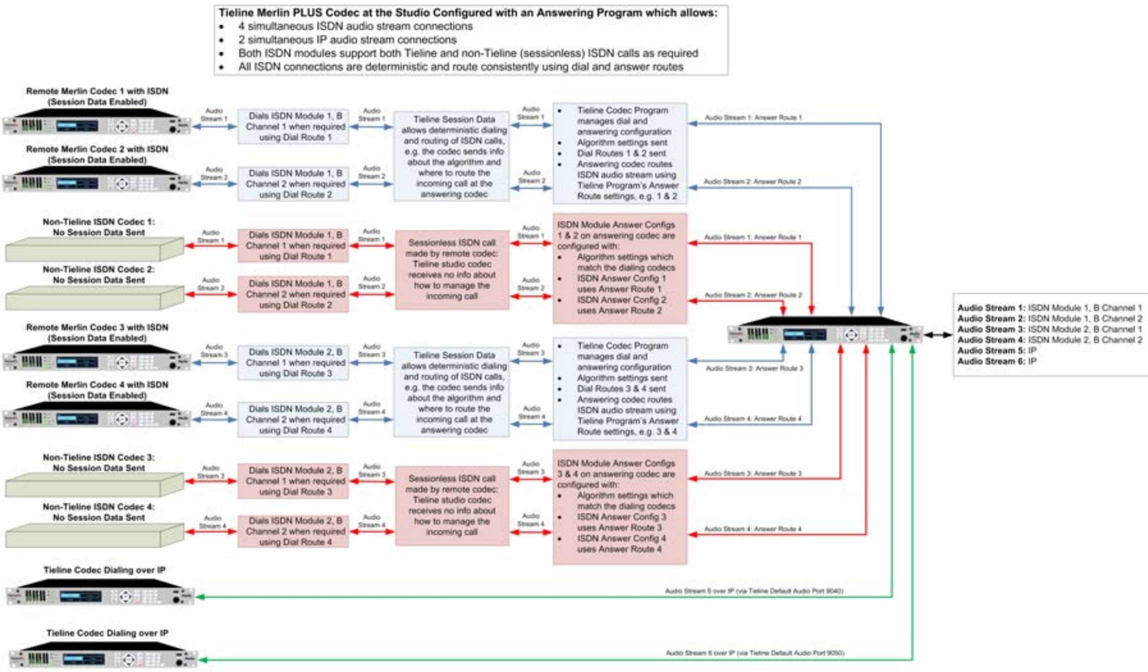
Select **Sessionless Only** when answering ISDN calls from non-Tieline codecs only. When **Sessionless Only** is selected, the codec will not wait to receive the Tieline session data. This reduces the time taken to answer an inbound sessionless call.

The following image displays how the answering codec will respond with **Sessionless Only** selected, i.e. calls from both Tieline and non-Tieline codecs are always regarded as sessionless.



Answering Multiple ISDN Calls from Tieline and non-Tieline Codecs

Tieline codecs capable of answering multiple incoming audio streams can be configured to answer both Tieline session data and sessionless ISDN calls at different times. They can also support connections using other transports such as IP or POTS. The following example shows how a Tieline codec can be configured to answer up to 4 separate mono ISDN calls at different times from both Tieline and non-Tieline codecs, as well as two mono IP audio streams.



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

21.4 Configuring POTS

Two slots are available for inserting optional POTS modules into the codec. These can be configured using the codec front panel or the HTML5 Toolbox Web-GUI. See About POTS Modules for additional information on POTS.

You can use the Java Toolbox Web-GUI to configure a dial and/or answer program with POTS settings. You may also need to:

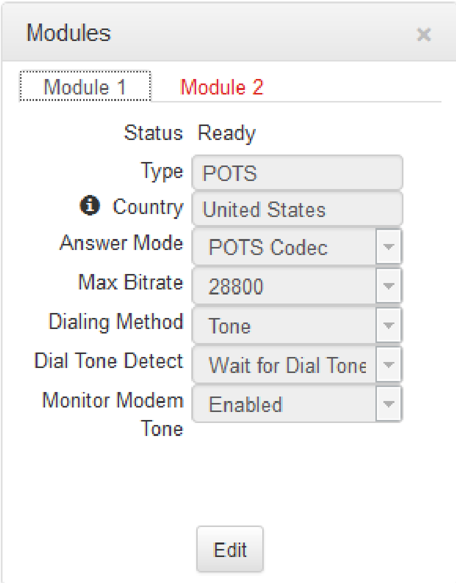
1. Configure POTS module settings.
2. Configure POTS Answering settings.

21.4.1 Configuring POTS Modules

POTS settings in the **Modules panel** menu determine how your codec will connect at a particular site. You can copy similar programs between codecs installed at different locations and also configure site-specific settings for how each module should connect. The default **Config** settings for POTS modules are designed to suit Tieline codecs. These settings will need to be adjusted to connect to non-Tieline POTS codecs or connect in **Analog Phone** mode.

Configuring POTS G5 Modules

1. Open the HTML5 Toolbox Web-GUI and click **Modules** to view and configure POTS site settings.



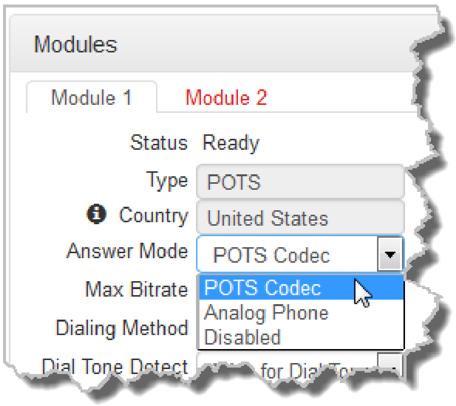
Important Notes: The POTS module **Status** is displayed in the **Modules panel**. **No Phone Line** is displayed when a cable is detached; **Ready** is displayed when a cable is attached and the line voltage is good.

2. Click to select **Module 1** or **Module 2**.

3. Click the **Edit** button to configure settings.

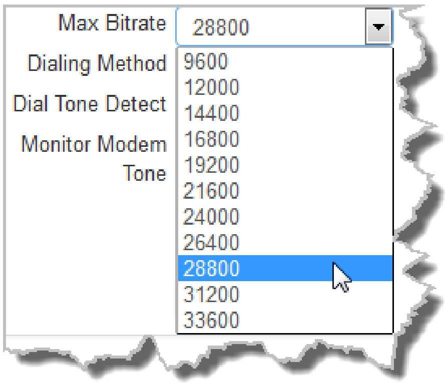
4. Click the drop down arrow to adjust the **Answer Mode** and select how the module in the codec will be able to answer incoming POTS calls. Options include:

- **POTS Codec:** allows the POTS G5 module to receive incoming audio data over a POTS line.
- **Analog Phone:** configures the POTS G5 module to receive a standard analog phone call.
- **Disabled:** disables the POTS G5 module from receiving a **POTS Codec** or **Analog Phone** call.



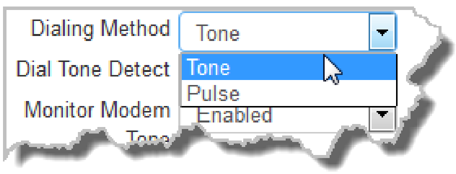
Calls are answered based on the **POTS Answer settings** in **Config 1** & **2**. Adjustments to these **Config** settings are not normally necessary when connecting between Tieline codecs. They are usually adjusted when connecting to non-Tieline codecs over POTS (see Configuring POTS Answering for more info).

5. Click the **Max Bitrate** drop-down arrow to adjust the maximum bit rate (dialing and answering). The default setting is **28800** (28.8kbps) and this only affects **POTS Codec** calls. The range of the setting is 9.6kbps to 33.6kbps. Even if the line is capable of establishing a connection at a higher bit rate, the **Max Bitrate** setting is the highest bit rate that will be attempted. Reducing this value can improve connection reliability on poor quality lines. If two codecs are not configured with the same setting, they will attempt to connect at the lowest of the two **Max Bit rate** settings.



Important Note: G5 POTS modems initially attempt to establish a link at the lowest **Max Bitrate** setting configured in the two modules being connected. If the POTS line doesn't support this bit rate, the modems will attempt to connect at the highest possible bit rate to suit the prevailing line quality at each end of the link.

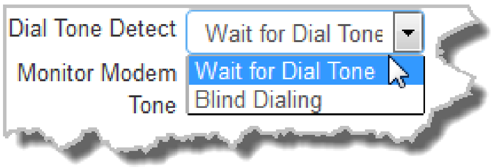
6. Click the drop-down arrow for **Dialing Method** to select **Tone** (DTMF) or **Pulse** dialing over POTS Codec connections. Tone dialing is used always when the **Answer Mode** is **Analog Phone**.



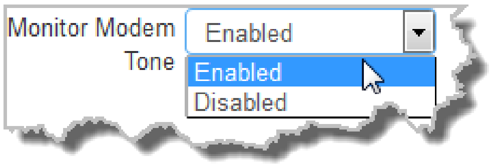
7. Click the drop-down arrow for **Dial Tone Detect** to select either:

- **Wait for Dial Tone**: The module will only be allowed to dial when a dial tone is present on the line.

- **Blind Dialing**: Allows the module to dial when no dial tone is present.



8. Click the drop-down arrow for **Monitor Modem Tone** to select either **Enabled** or **Disabled**.



When enabled the module will allow audio monitoring of modem tones during connection in **POTS Codec** mode via the phone input. By default, the following phone input monitoring rules apply when multiple POTS G5 modules are installed in a codec and multiple POTS connections are dialed.

Module 1	Module 2	Audio Rule
POTS Codec (Monitor Modem Tone)	POTS Codec (Monitor Modem Tone)	The phone input receives a mix of modem tone audio from both modules
POTS Codec (Monitor Modem Tone)	Analog Phone	The phone input receives analog phone input audio only and mutes modem tone monitoring
Analog Phone	Analog Phone	The phone input receives audio from the oldest active connection only

Important Notes:
- Modem tone monitoring will work even if **Phone Input Enable** is **Off** via **Settings > Audio > Phone Input > Phone Input Enable [Off]**.
- Modem tone monitoring is only enabled during the initial connection training and negotiation period in **POTS Codec** mode.
- The monitoring volume can be adjusted using the codec front panel via **Settings > Audio > Phone Input > Level**, or by opening the **Inputs panel** in the Web-GUI and adjusting the **Phone** input volume slider.

9. **Country** displays the current country setting in the codec. To adjust this setting select **Settings > System > Country**.

10. Click **Save** when configuration is complete.

21.4.2 Configuring POTS Answering

It is possible to store a different **POTS Answer Config** for each POTS module installed in the codec. POTS answering can be configured to suit:

- The type of call being made, e.g. Tieline (with Tieline Session Data) versus non-Tieline (Sessionless).
- Expected dialing behaviors and encoding, e.g. whether audio streams use **Route** tags and which algorithm is used.

If you answer a call from a non-Tieline codec you will need to create an answering "Config" to determine which module in the codec will answer the call and the settings used when connecting.

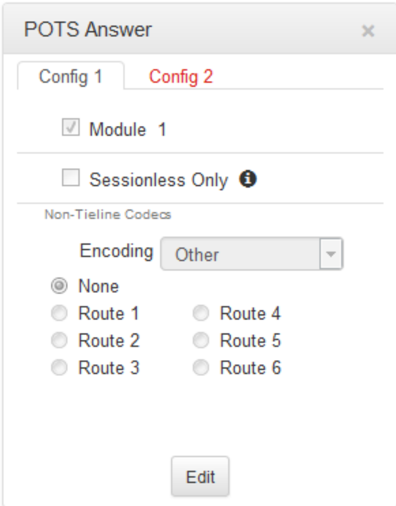
Important Notes:
- **POTS Answer Config** settings are applied to **POTS Codec** connections and not **Analog Phone** connections.
- When receiving a call from a Tieline codec with session data enabled (i.e. not **Sessionless**), the algorithm setting from the dialing codec overrides the setting in the **POTS Answer Config** menu.

POTS Config Settings

The default **POTS Answer** module **Config** settings, which can be viewed in the **POTS Answer panel** are:

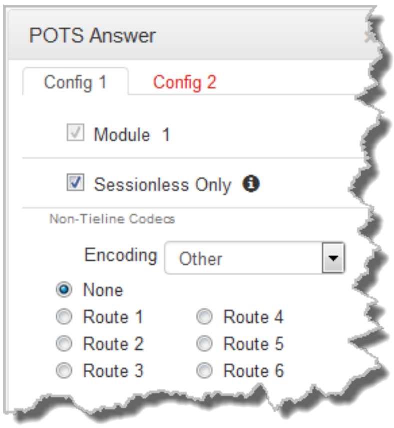
- **Tieline Codecs** Session Data,
- The **Other** algorithm.

This configuration will accept the settings from an incoming Tieline codec when it dials with session data enabled. It will also allow the codec to answer a call from a Comrex POTS codec supporting the **Other** algorithm setting.



Answering Calls from Non-Tieline POTS Codecs

1. Click the **Edit** button to configure settings.
2. Select the **Sessionless Only** check-box when only non-Tieline codecs are dialing a Tieline codec over POTS. 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 if required.

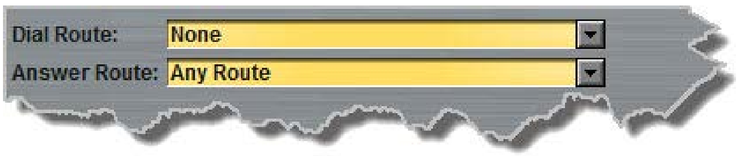


3. Click **Save** to apply changes to the **Config**.

Important Note: Select **Other** in the **Encoding** drop-down menu when connecting to Comrex® Vector, Matrix® and BlueBox® codecs. On the Comrex codec select its "Music" algorithm. Please note that 9.6kbps connections are not supported by the Comrex codecs.

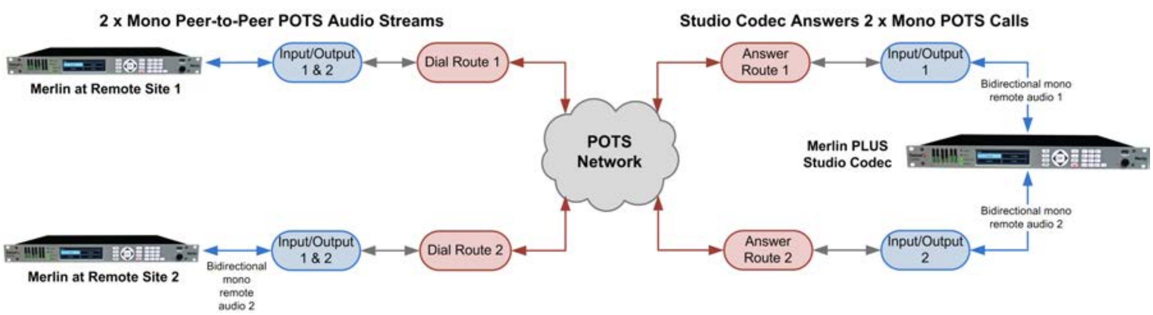
Dial and Answer Route Settings in Programs

Dial Route and **Answer Route** tags allow you to associate a POTS **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 POTS audio stream configurations, however it is very useful in multiple audio stream codecs which support POTS connections. When dialing Tieline to Tieline over POTS 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.



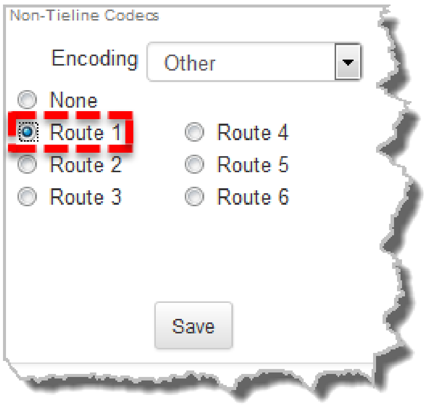
Answer Routes for Non-Tieline POTS Codecs

In some situations you may receive a call from a non-Tieline POTS 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 or 2** in **POTS Answer**. You can also select the default algorithm.

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

1. Click the **Edit** button to configure settings.
2. Select an answering **Route** for this POTS module in one of the two **Configs** available in the **POTS Answer panel**, e.g. **Route1**, then select the default **Encoding** algorithm **Other** (Note: **Other** is used for connecting to Comrex POTS codecs).



3. Click **Save Settings** to store the new **Config** settings.

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

21.5 Configuring Input/Output Settings

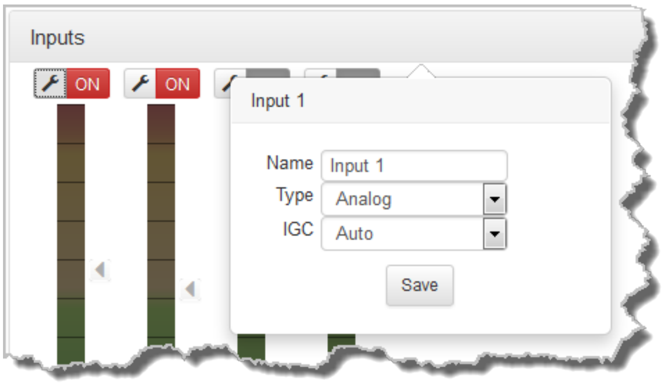
Open the HTML5 Toolbox Web-GUI and click **Inputs** to display the **Inputs panel**.

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

Configuring Input Channel Settings

Renaming Input Channels:

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

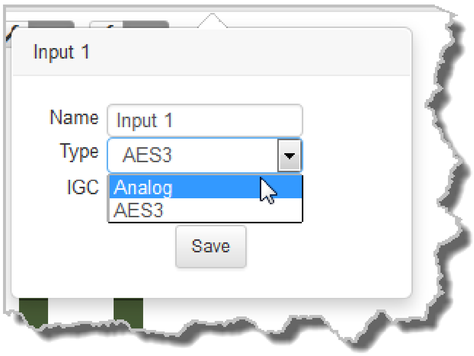


3. Click **Save** to confirm the name change.

Selecting Analog and Digital Audio Sources:

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

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



3. When you select AES3, an input is automatically configured for 100% input levels; input level and input on/off controls are unable to be adjusted.

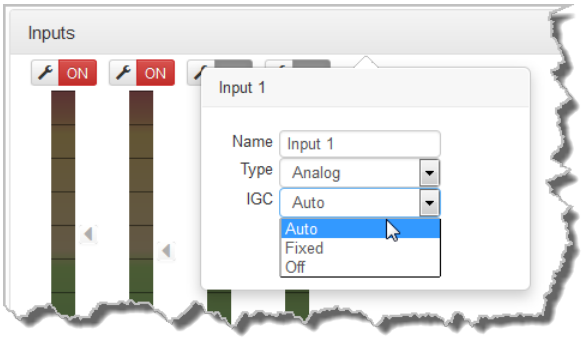
Important Note: Input levels can only be adjusted on analog inputs. See Configuring AES3 Audio for more information about the digital inputs and outputs.

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.

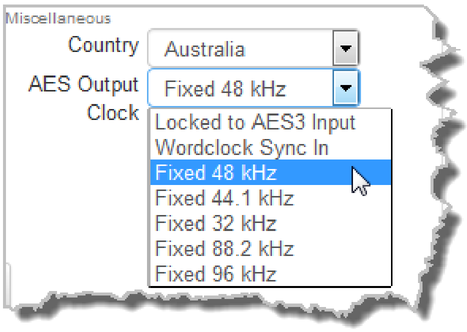


Important Note: When the auxiliary input (**AUX IN**) is **On** the default mixer configuration sends audio to all inputs. If you are not using the auxiliary input ensure it is **Off** to avoid additional noise in program audio.

AES3 Output Sample Rate Configuration

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

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



21.6 Configure SIP Settings

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

About SIP

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

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

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

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

SIP Server Connections: Getting Started

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

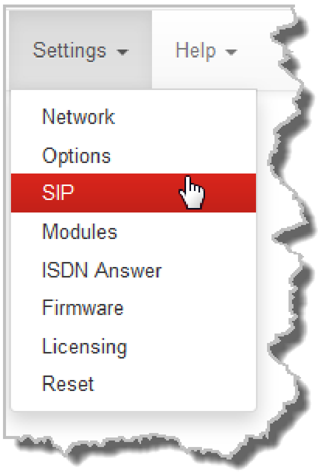
When you register an account with a SIP server you will be provided with:

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

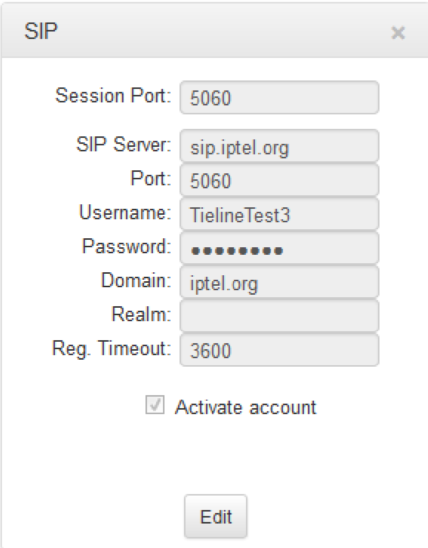
Configure the Codec for SIP using the Web-GUI

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

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



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



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

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

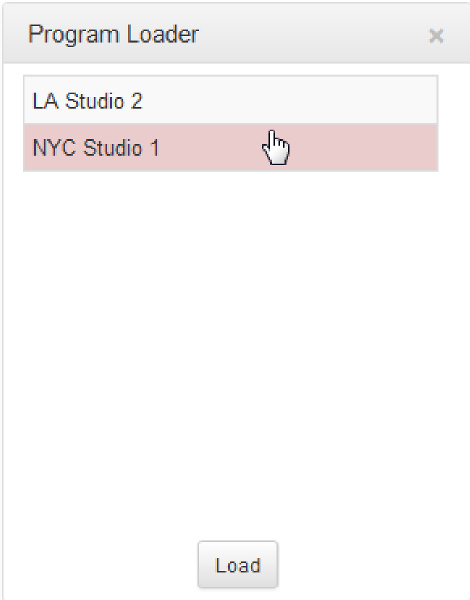
21.7 Load, Unload and Dial a Program

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

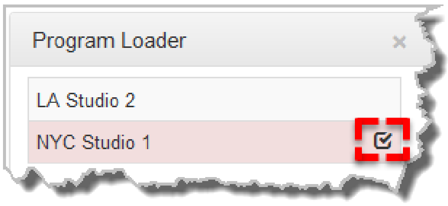
Important Notes: The HTML5 Toolbox Web-GUI currently does not support the creation of new programs. Use the Java Toolbox Web-GUI to create a new program.

Loading a New Program

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



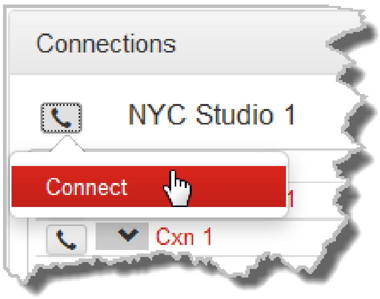
Note: the currently loaded program has the check-box symbol displayed next to its name.



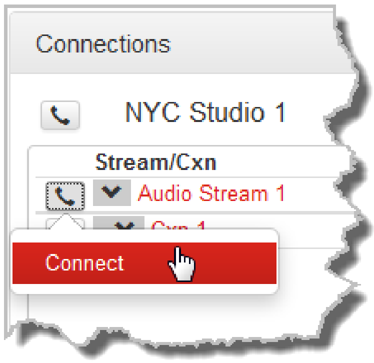
Connecting a Program

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

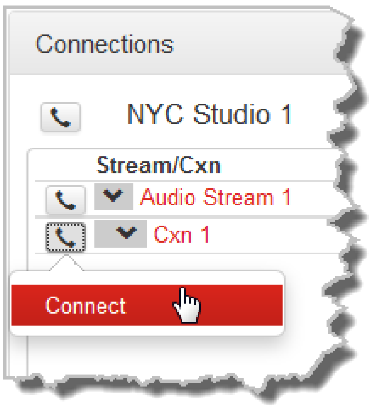
1. Click the program **Connect/Disconnect** symbol and then click **Connect**; this connects all active audio streams and connections associated with the program.



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



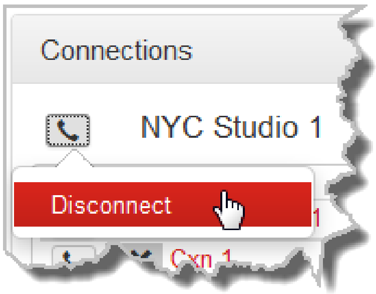
3. Click the connection **Connect/Disconnect** symbol and then click **Connect**; this connects an individual audio stream connection.



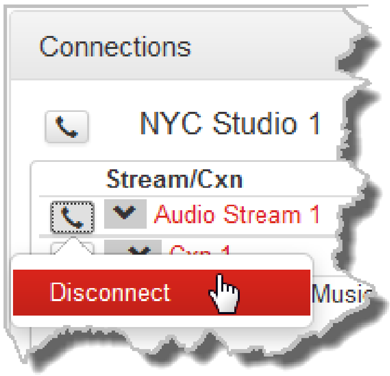
Disconnect a Program

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

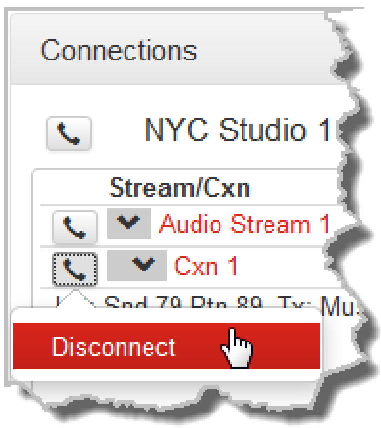
1. Click the program **Connect/Disconnect** symbol and then click **Disconnect** to disconnect a program; this includes all audio streams and connections associated with the program.



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

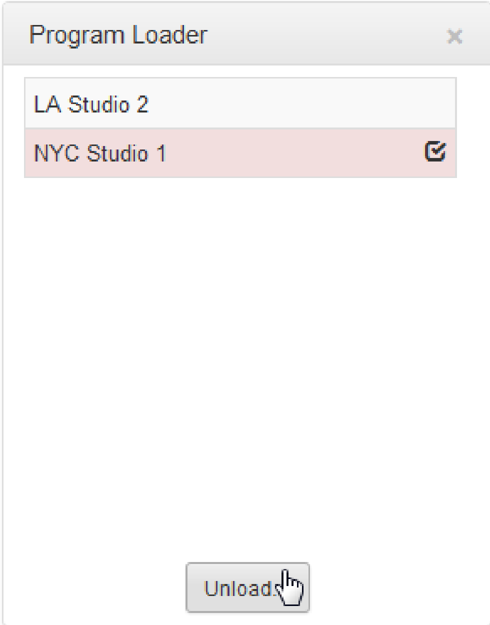


3. Click the connection **Connect/Disconnect** symbol to disconnect an individual audio stream connection.



Unloading a Program

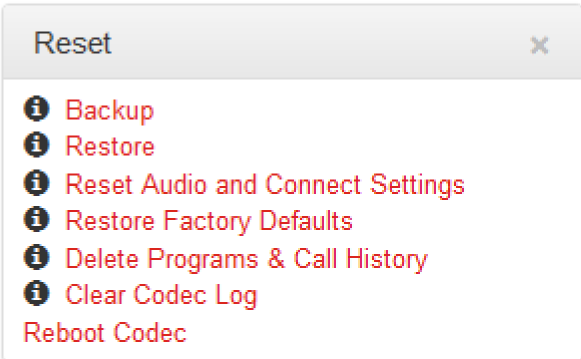
1. Click to select the loaded program in the **Program Loader panel** and then click **Unload** to unload the program in the codec.



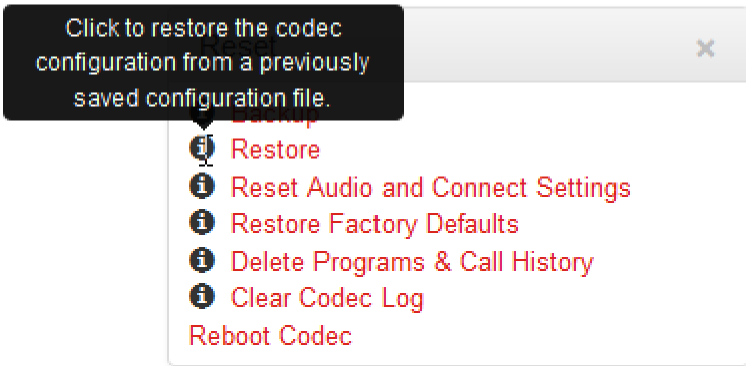
21.8 Reset Factory Default Settings

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

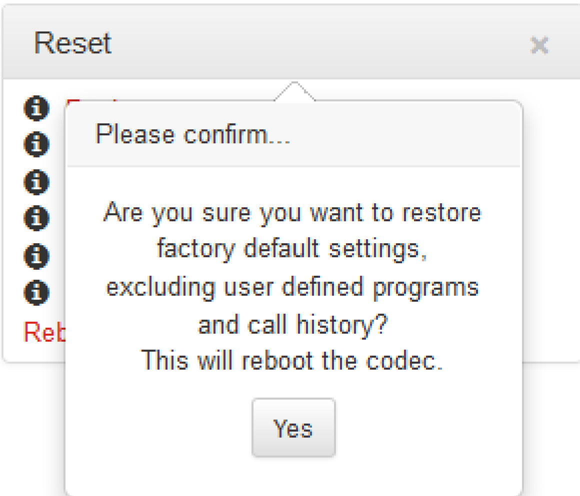
1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Reset** to display the **Reset panel**.



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



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



21.9 Backup and Restore Functions

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

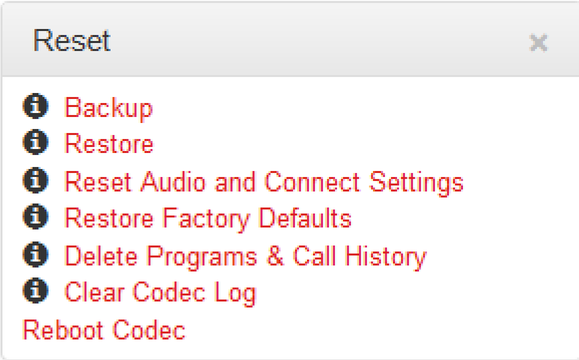
- Programs containing a variety of connection settings.
- All system settings that have been adjusted to change the factory default codec settings (current runtime settings).

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

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

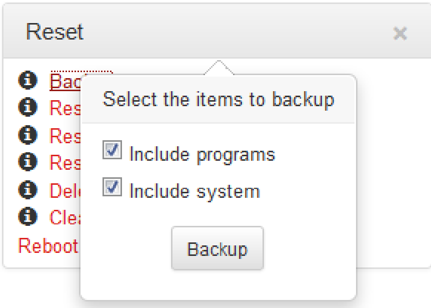
Creating Backup Files

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



2. Click **Backup**.

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



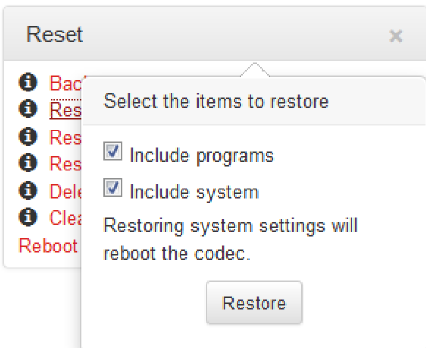
4. Select a location on your PC to save the configuration file. Note: You may need to "allow" your browser to display the pop-up dialog.

Restoring Configuration File Settings

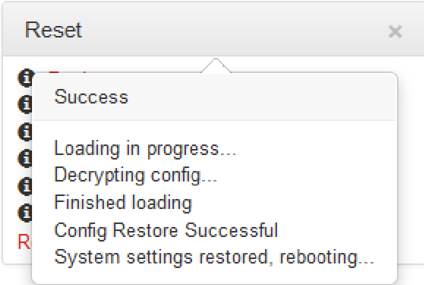
1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Reset** to display the **Reset panel**.

2. Click **Restore**.

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



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

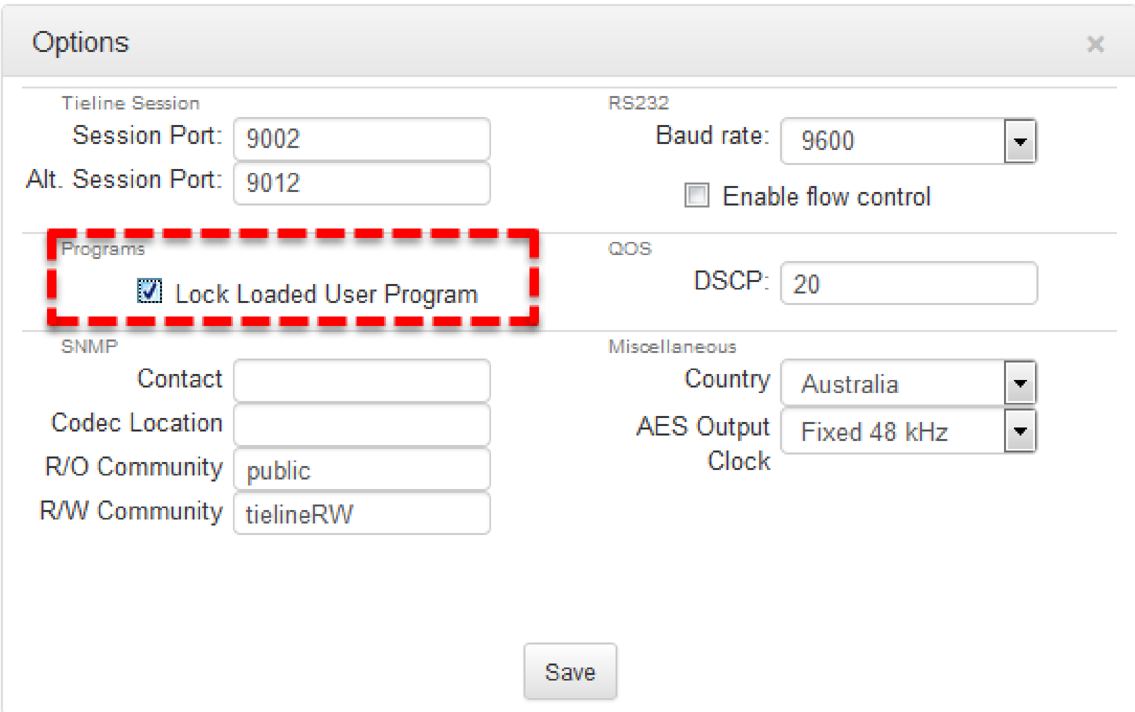


Note: The codec will automatically reboot if you restore system settings.

21.10 Lock or Unlock Programs

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

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



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

Important Note:
- A black **Padlock** symbol appears next to the program name in the **Connections panel** and in the **Program Loader panel**, to indicate a program is locked in the

codec.

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

21.11 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:

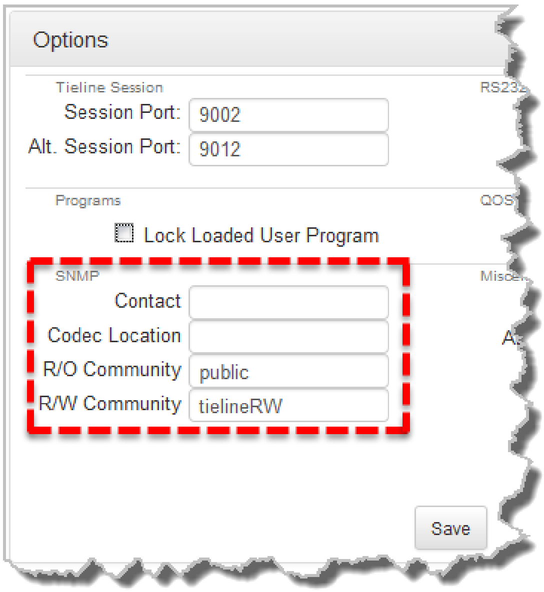
1. Configure SNMP Device settings in your codec.
2. Configure SNMP Traps via the **Alarms Panel** in the Web-GUI (see SNMP Trap Configuration in Configuring Alarms, or to configure using the codec front panel see Configuring SNMP Settings).

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



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

MIB Files for SNMP Configuration

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

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

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

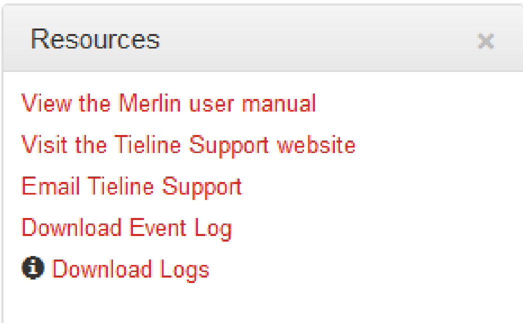
Important Note: The codec supports the attributes specified in the MIB-II standard. Please verify that your SNMP software contains the required files as specified in RFC 1213.

21.12 Download Logs

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

Procedure for Sending Logs to Tieline

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

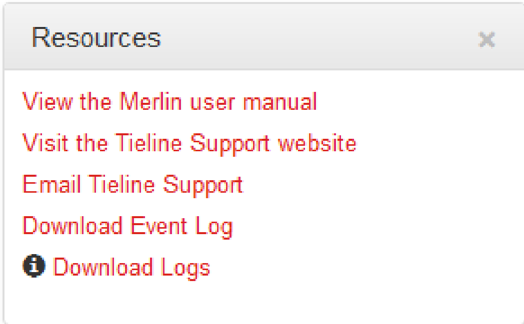


3. Save the file to your computer and then send it as a .zip file to Tieline support via support@tieline.com

Download Event Logs

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

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

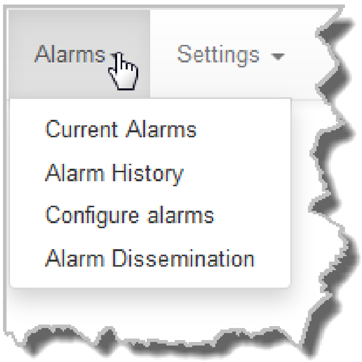


Clearing Logs

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

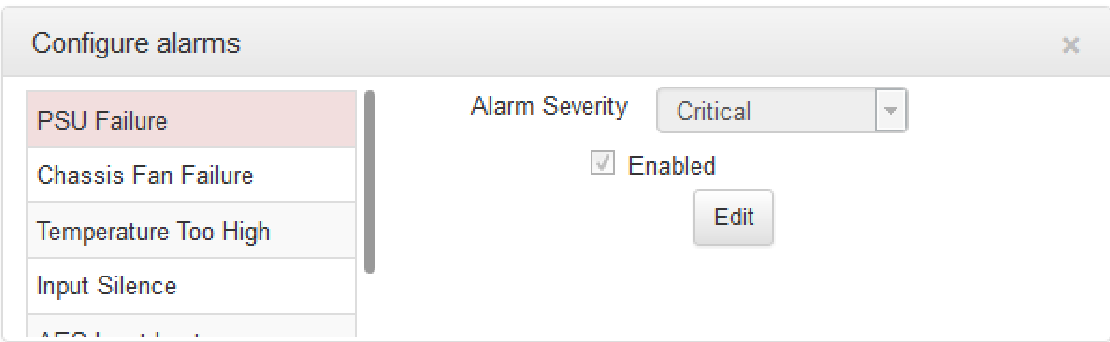
21.13 Configuring Alarms

Open the HTML5 Toolbox Web-GUI and click **Alarms** to open and view panels used to configure and monitor a range of alarms.

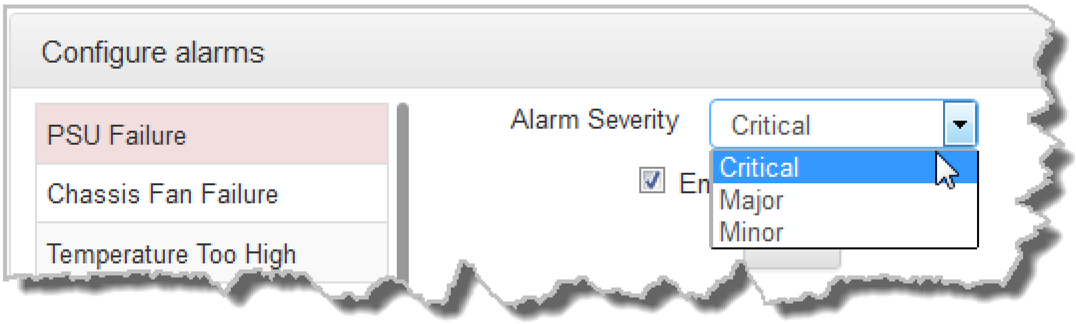


Configure and Enable Alarms

1. Open the HTML5 Toolbox Web-GUI and click **Alarms**, then click **Configure alarms** to open the panel.



2. Click to select an alarm from the list on the left side of the panel.
3. Click **Edit** to configure alarm settings.
4. Click the **Enabled** check-box to activate the alarm and then select an Alarm Severity level from the drop-down menu.



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

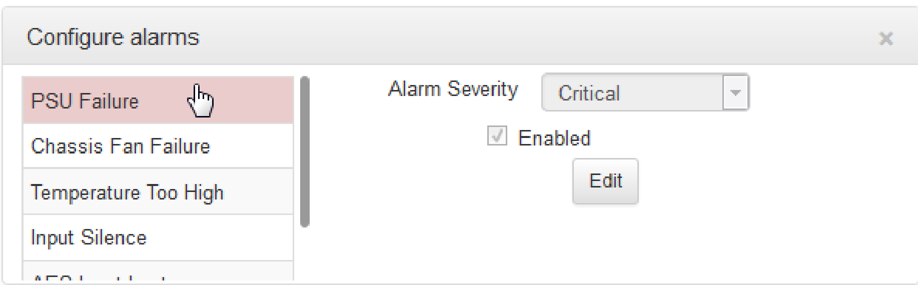
Note: The following **System** and **Audio** alarms are available:

Alarm	Alarm Type	Explanation
PSU Failure	System	Raises an alarm if one or both PSUs fail
Chassis Fan Failure	System	Raises an alarm if the internal fan fails
Temperature Too High	System	Raises an alarm if the temperature is too high
Input Silence	Audio	Raises an alarm if input audio is lost (according to preconfigured silence detection threshold parameters)
AES Input Lost	Audio	Raises an alarm if the AES input signal is lost (not available in WheatNet-IP capable codecs)
AES Reference Lost	Audio	Raises an alarm if the AES reference clock signal is lost (not available in WheatNet-IP capable codecs)

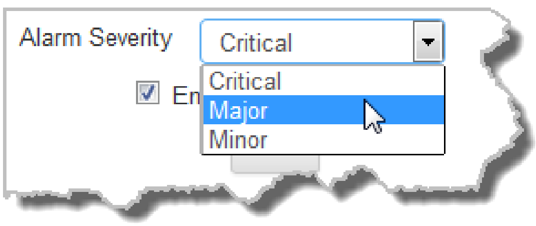
Configuring an Alarm's Severity Level

Codec alarms can be configured for three different severity levels:

1. Click to select an alarm from those displayed in the **Configure alarms** panel.



2. Click **Edit** to configure alarm settings.
3. Click the **Alarm Severity** drop-down menu and select the preferred severity level.

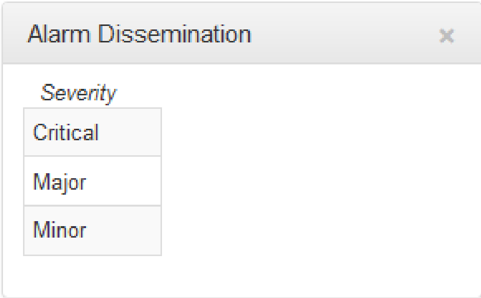


4. Click **Save** to store the new settings for the selected alarm.

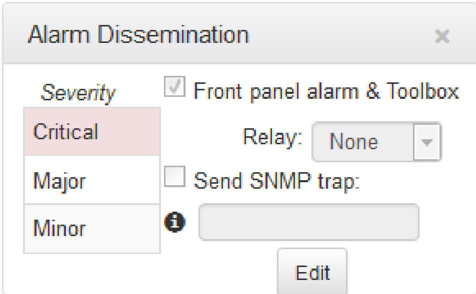
Configuring Alarm Dissemination Severity Alerts

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

1. Open the HTML5 Toolbox Web-GUI and click **Alarms**, then click **Alarm Dissemination** to open the panel..



2. Click to highlight the **Alarm Severity** level you want to configure.

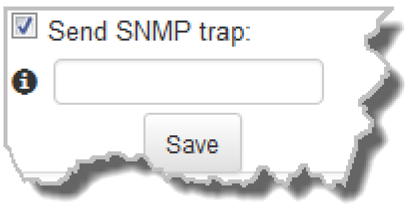


3. Click **Edit** to configure notification settings.
4. Click **Save** to store the new settings.

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.

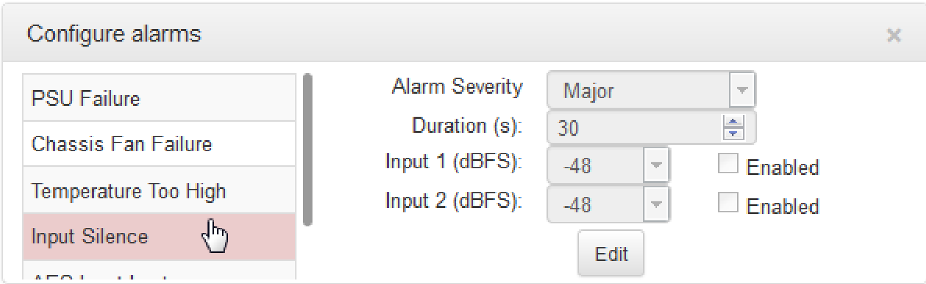


2. Enter the SNMP trap target in the text box, then click **Save** to store the new settings.

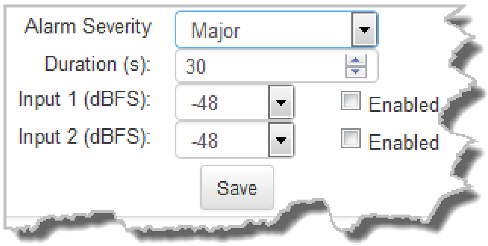
Configuring Input Silence Detection Parameters

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

1. Click **Input Silence** to select the alarm.



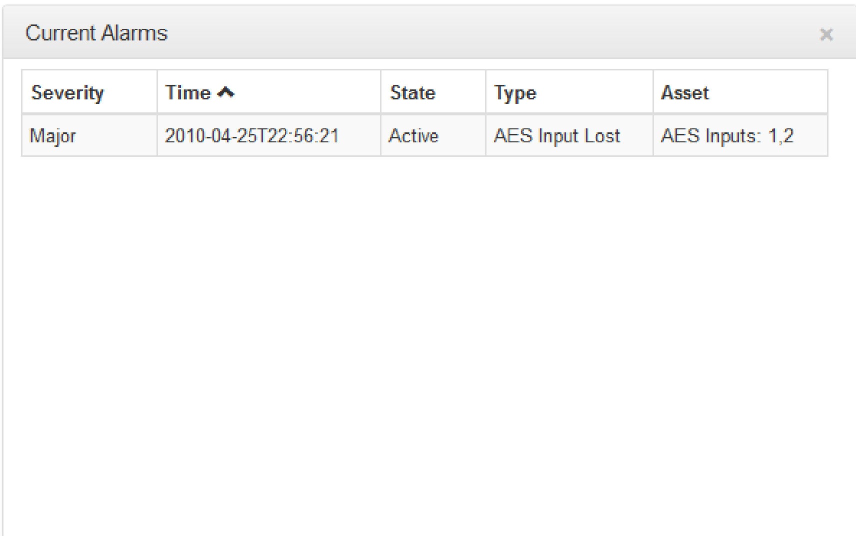
2. Click **Edit** to configure alarm settings.
3. Configure the dBFS threshold and timeout duration in seconds and ensure the input **Enabled** check-boxes are selected. An alarm will be raised when these thresholds are breached.



4. Click **Save** to store the new input silence alarm settings.

21.13.1 Managing Alarms

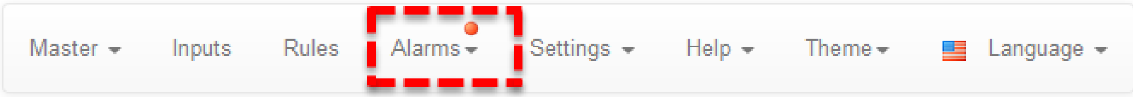
Open the HTML5 Toolbox Web-GUI and click **Alarms**, then click **Current Alarms** to view active alarms.



Viewing Current Alarms

Active alarms are indicated by:

1. The red **Alarm Symbol** flashing in the toolbar of the HTML5 Toolbox Web-GUI screen.



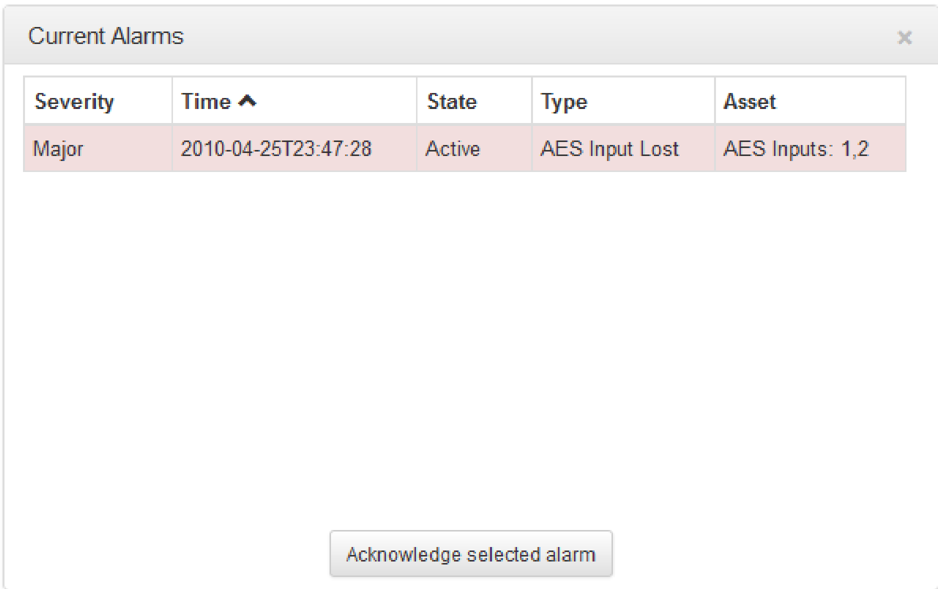
2. All new alarms being listed in the **Current Alarms panel**.
3. Other alerts as per **Alarm Dissemination panel** settings.
4. The codec front panel **ALARM LED** flashing red.

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

Acknowledging Alarms

To acknowledge an alarm in the **Current Alarms panel**:

1. Click to select the alarm in the **Current Alarms** panel.



2. Click **Acknowledge selected alarm**.

After acknowledging the alarm:

1. The **State** will change from **Active** to **Acknowledged**.
2. The red **Alarm Symbol** will stop flashing but remain visible in the toolbar of the HTML5 Toolbox 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 panel** settings.

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

Deactivating Alarms

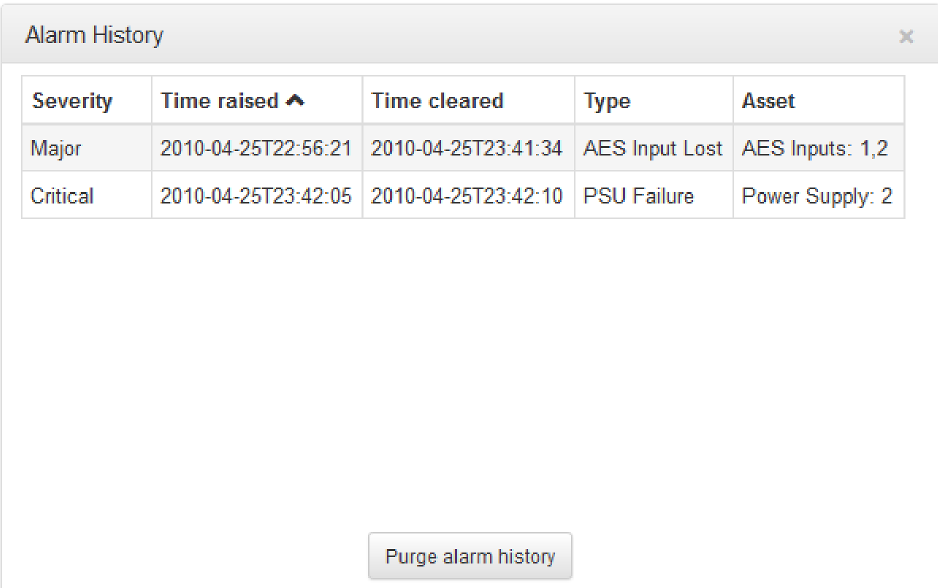
An alarm is deactivated automatically when the alarm state is reversed. E.g. if power is restored after a **PSU Failure** alarm, or if audio is restored after an **Input Silence** alarm.

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 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

1. Open the HTML5 Toolbox Web-GUI and click **Alarms**, then click **Alarm History** 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** panel.

21.14 RS232 Data Configuration

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

Algorithm Selected	IP	ISDN and POTS
Tieline Music and MusicPLUS	• In-band RPTP data enabled automatically • Synchronized out-of-band data can be enabled and disabled	• In-band RPTP data enabled automatically
All other algorithms	• Synchronized out-of-band data can be enabled and disabled	• No in-band or out-of-band data available

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

Important Notes:

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

21.15 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.

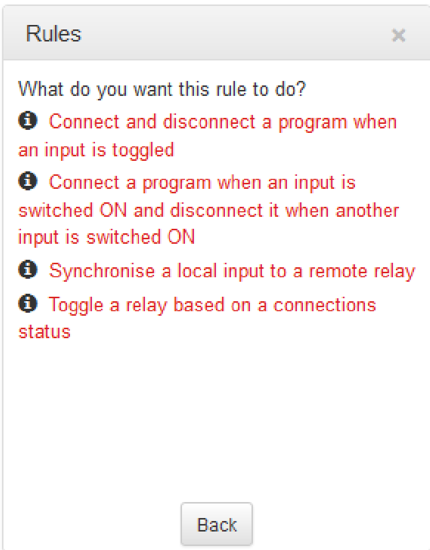
1. Press the **HOME** button to return to the **Home** screen
2. Use the navigation buttons on the front panel to select **Connect** and press the button
3. Select **IP** and press the button
4. Select your preferred **IP Mode** and press the button.
5. Use the down navigation button to select **Setup** 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. Open the Toolbox HTML5 Web-GUI and click **Rules** in the **Menu Bar** to display the **Rules panel**.
2. Click **Add New Rule**.
3. Click to select the appropriate rule for your requirements. See the Rules panel section in Using the Toolbox HTML5 Web-GUI for an explanation of the action each rule can perform.

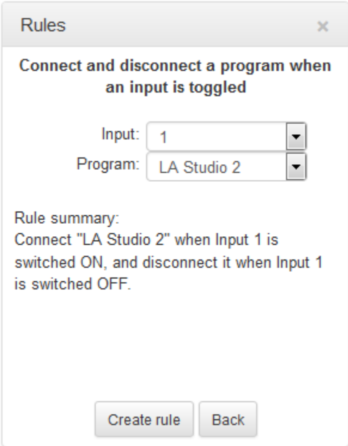


Note: When rules have been configured previously they are listed when the **Rules panel** is opened.

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

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

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

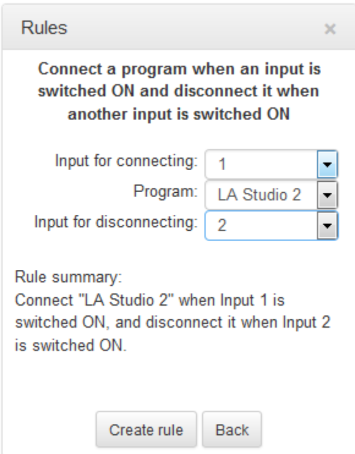


4. Check the **Rule Summary** and click **Create Rule** to save the settings.

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

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

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

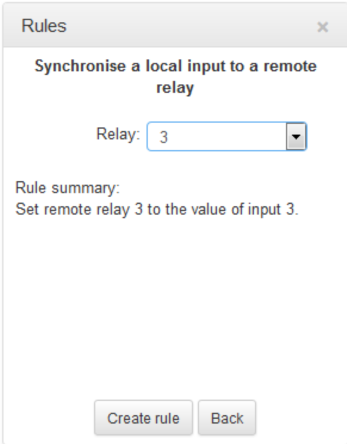


4. Check the **Rule summary** and click **Create Rule** to save the settings.

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

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

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

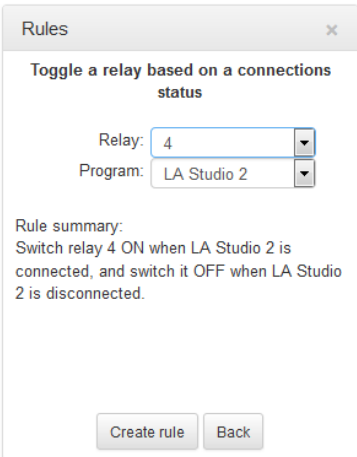


3. Check the **Rule summary** and click **Create Rule** to save the settings.

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

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

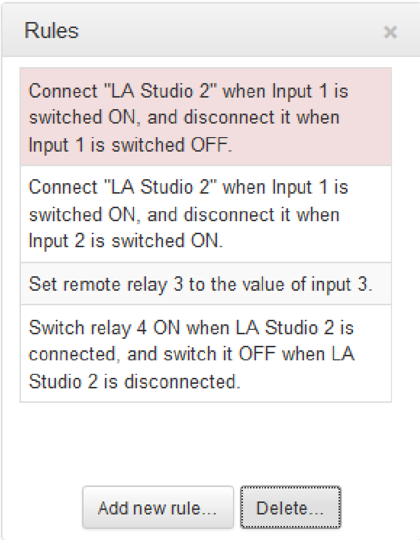
1. Click the fourth rule in the **Rules panel** titled **Toggle a relay based on a connection's status**.
2. Click the drop-down **Relay** arrow and select the relay output you want to toggle.
3. Click the drop-down **Program** arrow to select a specific program which will affect the relay toggle function,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



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

Deleting Rules

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



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

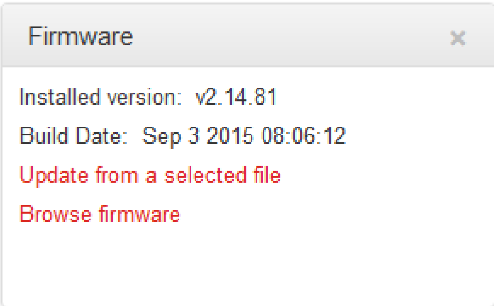
21.16 Upgrading Codec Firmware

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

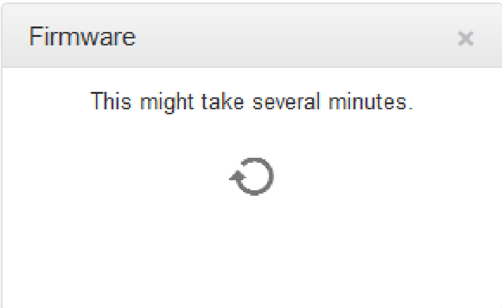
Firmware Upgrades

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

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



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



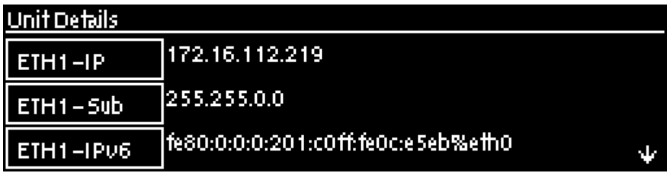
22 Front Panel Configuration Tasks

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.

Ethernet and VLAN Configuration Options

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

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

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

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

Configure an IPv4 DHCP Address

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

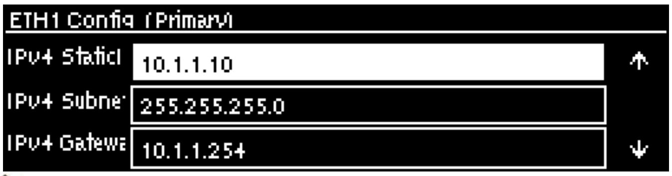
1. Press the **SETTINGS** button.
2. Select **LAN** and press the button.
3. Use the down navigation button to select **ETH1**, **ETH2** or a **VLAN** interface.

4. Select **Usage** and choose the appropriate control and/or streaming mode for the connection, then press the button.
5. Select **IPv4 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**, **ETH2** or a **VLAN** interface.
4. Select **Usage** and choose the appropriate control and/or streaming mode for the connection, then press the button.
5. Select **IPv4 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 and any VLANs which are configured.

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

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

To adjust this setting:

1. Press the **SETTINGS** button.
2. Select **LAN** and press the button.
3. Use the down navigation button to select **ETH1**, **ETH2** or a **VLAN** interface.
4. Select **IPv6 Mode** and press the button.

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

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

Manual IPv6 Address Assignment

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

DNS Server

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

1. Press the **SETTINGS** button.
2. Use the navigation buttons on the front panel to select **LAN** and press the button.
3. Use the down navigation button to select **ETH1**, **ETH2** or a **VLAN** interface.
4. Use the down navigation button to scroll to **Auto DNS**.
5. Press the button to toggle between **Yes** and **No**.

VLAN ID (VLAN configuration only)

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

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

VLAN Priority (VLAN configuration only)

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

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

VLAN Interface (VLAN configuration only)

This setting applies the VLAN settings to a physical Ethernet port in the codec.

1. Press the **SETTINGS** button.
2. Use the navigation buttons on the front panel to select **LAN** and press the button.
3. Use the down navigation button to select a **VLAN** interface.
4. Select **Usage** and press the button.
5. Select the mode of operation for this VLAN (e.g. Control & Streaming, Streaming only, Control Only) and press the button.
6. Use the down navigation button to scroll to **Interface**.
7. Press the button to select **ETH1** or **ETH2**, then press the button.

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. When used for mono connections, HE-AAC v.1 performs best at bit rates of 24kbps per channel or higher. HE-AAC v.1 is also used for stereo connections when audio connection bandwidth is 48kbps or higher.

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

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

AAC-LD

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

AAC-ELD

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

AAC-ELD v 2

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

Overview of aptX Enhanced Audio Coding

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)

Configuring an Algorithm in the Codec

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the button.

3. Select **IP** and press the button.
4. Select your preferred **IP Session** mode and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Navigate to **Algorithm** 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 the right algorithm?

The algorithm you select will not only affect the quality of the broadcast but it will also contribute to the amount of latency or delay introduced. For example, if MP2 algorithms are used, program delays will be much longer than when using Tieline Music or MusicPLUS algorithms. This is due to the additional inherent encoding delays involved when using MP2 algorithms. This can be a major consideration for live applications that integrate 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	Audio Quality and Features	Recommended applications for on-air use
Linear/PCM (Uncom-pressed)	16/24 bit up to 45kHz	0ms	sample rate x bits per sample x no. channels; 512kbps minimum (16bit;32kHz) to 4.6 Mbps (24bit; 96 kHz)	80kbps	• Full bandwidth, perfect audio quality for voice and music • No error concealment/ correction or artefacts	• Extremely high quality uncompressed audio for STLs and audio distribution. • Ideal for fiber or high bandwidth links.
Tieline Music	Up to 15kHz	20ms	24 kbps minimum	16kbps	• High quality voice and music • Very low delay at low bit rates	• Great for live voice or music remotes as well as STLs and audio distribution with limited connection bandwidth (e.g. POTS or 3G wireless) • Suitable when bidirectional communication between announcers is required • Deliver 15kHz stereo over 1 x 64kbps ISDN B Channel.
Tieline Music-PLUS	Up to 22kHz	20ms	48 kbps minimum (Optimised for 64kbps per audio channel)	16kbps	• Very high quality voice and music • Very low delay at low to moderate bit-rates	• Very high quality, very low delay STLs and audio distribution • Remote connections able to achieve 48kbps for each audio channel • Suitable when bidirectional communication between announcers is required
G.711	3kHz	1ms	64kbps minimum	80kbps	• Low quality 3kHz POTS phone quality audio • Very low delay at moderate bit rates	• Highly compatible with other brands of audio codec • Low quality and used generally for compatibility
G.722	7kHz	1ms	64kbps minimum	80kbps	• Good quality 7kHz voice • Better quality than a standard POTS phone call • Very low delay at moderate bit rates	• Highly compatible with other brands of audio codec • Good voice quality audio for remotes and other voice quality applications
MPEG Layer 2	Up to 22kHz	24 to 36ms	64kbps minimum	8.5 - 13.3kbps	• Very high quality voice and music • Low to moderate delay at moderate to high bit rates	• Highly compatible with other brands of audio codec • Very high quality audio for remotes, STLs and audio distribution

MPEG Layer 3	Up to 15kHz	100ms	64kbps	8.5 - 13.3kbps	• High quality voice and music • Moderate bit rates • High delay	• High quality remotes, STLs and audio distribution • Use when bidirectional communication between announcers is not required
LC-AAC	Up to 15kHz	64ms	64kbps	15kbps	• High quality voice and music at lowest bit rate; better quality at higher bit rates • Moderate delay at moderate to high bit rates	• Voice or music remotes as well as STLs and audio distribution where some delay is tolerable • Tieline Music or MusicPLUS deliver lower delay
HE-AAC v.1	Up to 15kHz	128ms	48kbps	7.4kbps	• High quality voice and music at the lowest bit rate; better quality at higher bit rates • Low to Moderate bit rates • High delay	• Live voice or music remotes as well as STLs and audio distribution with limited connection bandwidth • Use when bidirectional communication between announcers is not required
HE-AAC v.2	Up to 15kHz	128ms	Minimum 16kbps (Mono); 24kbps (stereo)	7.4kbps	• High quality voice and music • Low bit rates • High delay	• Used for DAB+ radio streaming • Ideal for low bit rate remotes • Use when bidirectional communication between announcers is not required
AAC-LD	Up to 20kHz	20ms at 48kHz	48kbps minimum	30kbps	• Very high quality voice and music • Very low delay at low to moderate bit rates	• Very high quality, very low delay STLs and audio distribution • Remote connections able to achieve 48kbps for each audio channel requiring • Suitable when bidirectional communication between announcers is required
AAC-ELD	Up to 20kHz	15-30ms	24 kbps minimum	15-30kbps	• Very high quality voice and music • Very low delay at low bit rates	• Great for live voice or music remotes • Suitable when bidirectional communication between announcers is required
AAC-ELDv.2	Up to 20kHz	35ms	Pending release	Pending release	• High quality voice and music • Low delay at low bit rates	• Great for live voice or music remotes where limited connection bandwidth is available • Suitable when bidirectional communication between announcers is required
aptX Enhanced	10Hz-24kHz	2.5ms at 48kHz	128kbps minimum (16bit; 32kHz) to 288kbps	80kbps	• Very high quality voice and music • Extremely low delay at high bit	• Ideal for STLs and audio distribution where high connection bandwidth

			(24bit;48kHz)		rates • Highly cascade resilient	is available and very low delay is highly desirable. • Resilient with multiple encodes/decodes when required
Opus	4Hz-20kHz	20ms	9.6-256kbps	16kbps	• Very high quality voice and music • Very low delay at low bit rates	• "Opus Voice" is ideal for high quality, and low delay voice quality remotes at extremely low bit rates. • "Opus Mono" and "Opus Stereo" are perfect for high fidelity remotes, STLs and audio distribution at higher bit rates
TxTran / RxTran					NOT FOR BROADCAST USE	NOT FOR BROADCAST USE

Algorithm Selection Guide

Algorithm	Very Low Delay	Moderate to High Delay	Excellent Performance at Low Bit rates	Preferred for Live Remotes	Preferred for STLs and Audio Distribution	Highly Compatible with other Codecs
Linear/PCM	✓				✓	✓
Opus	✓		✓	✓		
Tieline Music	✓		✓	✓		
Tieline MusicPLUS	✓		✓	✓	✓	
apt-X Enhanced	✓				✓	
LC-AAC		✓			✓	
HE-AACv1		✓			✓	
HE-AACv2		✓	✓	✓*		
AAC-LD	✓			✓	✓	
AAC-ELD	✓		✓	✓		
AAC-ELDv2	✓		✓	✓		
MPEG Layer 2	✓				✓	✓
MPEG Layer 3		✓				✓
G.722	✓					✓
G.711	✓					✓

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

IP Connection Bit rates Supported

Algorithm	Sample Rate	Genie and Merlin Connection Bit Rates Supported																										
AAC HE Mono	32kHz				16	20		24		32	40	48	56															
AAC HE Mono	44.1kHz							24		32	40	48	56	64														
AAC HE Mono	48kHz							24		32	40	48	56	64														
AAC HE Stereo	32kHz							24		32	40	48																
AAC HE Stereo	44.1kHz										40	48	56	64														
AAC HE Stereo	48kHz										40	48	56	64														
AAC LD Mono	32kHz										40	48	56	64														
AAC LD Mono	44.1kHz												56	64	96	112	128	160		192								
AAC LD Mono	48kHz												56	64	96	112	128	160		192								
AAC LD Stereo	32kHz													64	96	112	128											
AAC LD Stereo	44.1kHz														96	112	128	160		192	256			320		384		
AAC LD Stereo	48kHz														96	112	128	160		192	256			320		384		
AAC ELD Mono	32kHz							24		32	40	48	56	64														
AAC ELD Mono	44.1kHz							24		32	40	48	56	64														
AAC ELD Mono	48kHz							24		32	40	48	56	64														
AAC ELD Stereo	32kHz											48		64	96	112	128											
AAC ELD Stereo	44.1kHz											48	56	64	96	112	128											
AAC ELD Stereo	48kHz											48	56	64	96	112	128											
AAC LC Mono	32kHz									32	40	48																
AAC LC Mono	44.1kHz											48	56	64	96		128	160										
AAC LC Mono	48kHz											48	56	64	96		128	160										
AAC LC Stereo	32kHz													64	96													
AAC LC Stereo	44.1kHz													64	96	112	128	160		192	256			320				
AAC LC Stereo	48kHz													64	96	112	128	160		192	256			320				
MP2 Mono	32kHz													64	96	112	128	160		192								
MP2 Mono	44.1kHz													64	96	112	128	160		192								
MP2 Mono	48kHz													64	96	112	128	160		192								
MP2 Stereo	32kHz															112	128	160		192	256							
MP2 Stereo	44.1kHz															112	128	160		192	256			320		384		
MP2 Stereo	48kHz															112	128	160		192	256			320		384		
MP2 J-Stereo	32kHz															112	128			192								
MP2 J-Stereo	44.1kHz															112	128			192								
MP2 J-Stereo	48kHz															112	128			192								
MP3 Mono	32kHz													64	96	112	128	160										
MP3 Mono	44.1kHz													64	96	112	128	160										
MP3 Mono	48kHz													64	96	112	128	160										
MP3 Stereo	32kHz															112	128	160		192	256							
MP3 Stereo	44.1kHz																128	160		192	256			320				
MP3 Stereo	48kHz																128	160		192	256			320				
aptX Enhanced Mono	32kHz																128			192								
aptX Enhanced Mono	44.1kHz																		176.4			264.6						
aptX Enhanced Mono	48kHz																			192			288					
aptX Enhanced Stereo	32kHz																				256					384		
aptX Enhanced Stereo	44.1kHz																								352.8		529.2	
aptX Enhanced Stereo	48kHz																									384		576
Opus Voice	48kHz	9.6	12	14.4	16.8	19.2	21.6	24	28.8	33.6	38.4	48		64	96	112	128											
Opus Mono	48kHz											48		64	96	112	128											
Opus Stereo	48kHz													64	96	112	128	160		192	256							
Music Mono	32kHz	9.6	12	14.4	16.8	19.2	21.6	24	28.8	33.6	38.4	48		64	96	112	128	160		192	256							
Music Stereo	32kHz	9.6	12	14.4	16.8	19.2	21.6	24	28.8	33.6	38.4	48		64	96	112	128	160		192	256							
Music PLUS Mono	48kHz	9.6	12	14.4	16.8	19.2	21.6	24	28.8	33.6	38.4	48		64	96	112	128	160		192	256							
Music PLUS Stereo	48kHz	9.6	12	14.4	16.8	19.2	21.6	24	28.8	33.6	38.4	48		64	96	112	128	160		192	256							

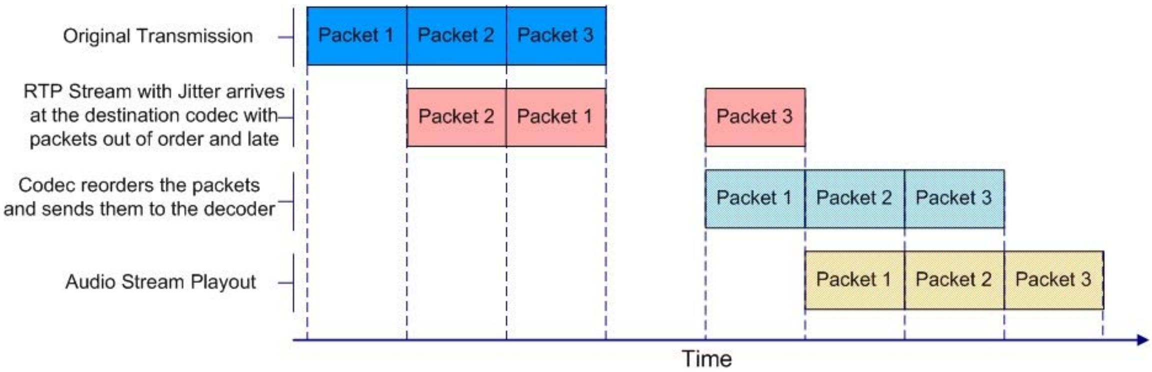
ISDN Encoding Options

The codec supports ISDN connections using the following algorithms and B Channel assignments.

ISDN Encoding	1B	2B	3B	4B
E-AptX Mono 16bit 32KHz		✓		
E-AptX Mono 16bit 48KHz			✓	
E-AptX Mono 24bit 32KHz			✓	
E-AptX Stereo 16bit 32KHz				✓
Music Mono	✓	✓	✓	✓
Music Stereo	✓	✓	✓	✓
Music Plus Mono	✓	✓	✓	✓
Music Plus Stereo	✓	✓	✓	✓
MP2 Mono 32KHz	✓	✓		
MP2 Stereo 32KHz		✓		
MP2 Mono 48KHz	✓	✓		
MP2 Stereo 48KHz		✓		
MP2 J-Stereo 32KHz		✓		
MP2 J-Stereo 48KHz		✓		
G.711	✓			
G.722	✓			

22.3 Configuring the Jitter Buffer

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



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

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

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

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

CAUTION: If a Tieline codec connects to a device that is using non-compliant RTP streams then the last fixed setting 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	✗	✗

Configuring Automatic Jitter Buffering (Default Setting)

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select your preferred **IP Session** mode and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Navigate to **Jitter Buffer** and press .
7. Select **Auto Jitter Adapt** and press .
8. Select your preferred jitter buffer setting and press .

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 Session** mode and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Navigate to **Jitter Buffer** 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 Tieline Music is the algorithm in use. Do not use PCM (uncompressed) audio over highly contended DSL/ADSL connections without enough bandwidth to support the high connection bit rates required.

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 **Tieline** and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Navigate to **FEC** and press .
7. Select the local codec FEC setting in the **Local FEC** screen and press .
8. Select the remote codec FEC setting in the **Remote FEC** screen and press .

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 using FEC in a ratio of 2:1.	Recommended for international & national connections
33%	Additional data is sent using FEC in a ratio of 3:1.	Recommended for national and local connections.
20% (Highest delay)	Additional data is sent using 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.

Conserving Bandwidth with FEC

There is a trade-off between the quality and the reliability of an IP connection – particularly when FEC is activated on your codecs. However, it is possible in certain situations to set different FEC on each codec to match connection bandwidth requirements at either end of the link, conserve bandwidth and create more stable IP connections.

For example, if your broadcast is a one-way broadcast from a remote site, i.e. you are not using the return path from the studio, or only using it for communications purposes, it is possible to reduce or turn off FEC at the studio codec. This effectively reduces the bandwidth required over the return link (communications channel) and increases the overall bandwidth available for the incoming broadcast signal from the remote site.

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 your preferred **IP Session** mode and press the button.
5. Use the down navigation button to select **Setup** and press .
6. Navigate to **Dir** and press .
7. Select **Encode Only** or **Decode Only** and press .

22.6 Enabling Relays & RS232 Data

Data must be enabled to activate contact **CONTROL PORT** closure operation and RS232 data. Please see Appendix A for RS232 and Control Port Wiring information.

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select **Tieline** (or **Sessionless**) and press the button.
5. Select **Peer-to-Peer** and press the button.
6. Use the down navigation button to select **Setup** and press the button.
7. 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** in 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 Config** 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:
- When connecting to G3 codecs over IP, ISDN or POTS only in-band data is available via the Music and MusicPLUS algorithms. See RS232 Data Configuration for more details.
- 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.

22.7 Configuring TCP/UDP Ports

In TCP and UDP networks the codec port is the endpoint of your connection. Software network ports are doorways for systems to communicate with each other. For example, several codecs in your studio may use the same public static IP address. Unique port numbers can be used to 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 Installing the Codec at the Studio section of the manual. This section also contains useful information for configuring port forwarding and troubleshooting IP connections.

Changing Codec Port Numbers

Reasons for adjusting the port setting on your codec include:

- 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.

Configuring the Session and Audio Port Numbers used when Dialing a Program

For two codecs to connect, they need to be configured with matching port numbers. If there is a need to change codec port settings, in most situations you should consult your organization's resident IT professional. To adjust 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 Session** mode and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Navigate to either **Session** (session protocol) or **Protocol** (audio protocols) and press .
7. Select the session or audio port you want and press .
8. Use the numeric **KEYPAD** to add a new port number and press .

Changing the Tieline Session Ports when Answering

To adjust the local Tieline session data port used by your codec:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Settings** and press the button.
3. Select **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.

Audio Port Settings for Tieline Session Data and Sessionless IP Calls

The codec supports sessionless IP streaming, whereby the codec does not send Tieline session data when attempting to connect. When using this mode you need to configure the "send" audio port (codec port at the remote end of the link to which you are sending audio) and "return" audio port (port used by the local codec to receive audio from the remote codec).

It is also possible to configure the send and return audio ports for a codec using Tieline session data to establish IP connections. This may be required because some firewalls require symmetric port configuration.

Sessionless Audio Port Configuration

When you select **Sessionless** as the **Session Protocol**:

- The default value for both the **Send** and **Return** (audio) **Ports** is 9000

- The range of values for the audio ports is 2000 to 65535
- The audio port values can be set independently
- Both audio ports can always be configured, i.e. there is no dependency on encode/ decode direction

"Tieline Codec" Port Configuration

If using the **Tieline Codec** setting for call establishment (i.e. Tieline session data is enabled), you can also change the default audio ports if required.

- The default value for the **Send** (audio) **Port** is 9000
- The range of values for the **Send Port** is 2000 to 65535
- The default port value for the **Return** (audio) **Port** is **Automatic**. Note: **Automatic** indicates that the codec will allocate the return port value and send this information to the codec to which you are dialing
- The range of values for the **Return Port** is 2000 to 65535

Sessionless Multicast Connections

For a sessionless multicast server connection:

- Only the **Send Port** is available
- The default value for the port is 9000
- The range of values for the port is 2000 to 65535

For a sessionless multicast client connection:

- Only the **Return Port** is available
- The default value for the port is 9000
- The range of values for the port is 2000 to 65535

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 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.

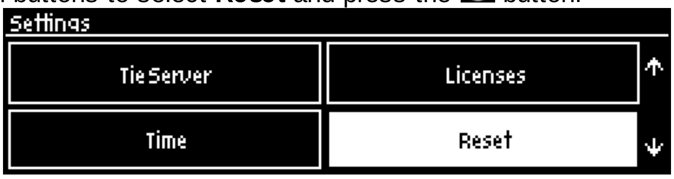
22.9 Reset and Restore Factory Default Settings

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
5	Clear Logs	Deletes codec event and log history. Note: This should only be performed if instructed to by Tieline support staff.

Important Note: After restoring factory defaults, always reboot the codec using the **Reboot Codec** function, not by removing power from the codec.

1. Press the **SETTINGS** button.
2. 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.

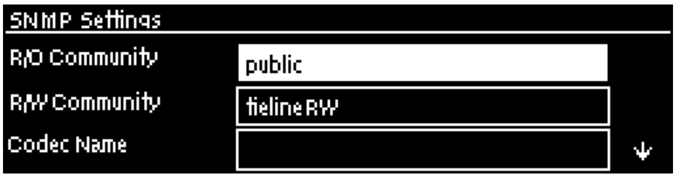
22.10 Configuring SNMP Settings

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.



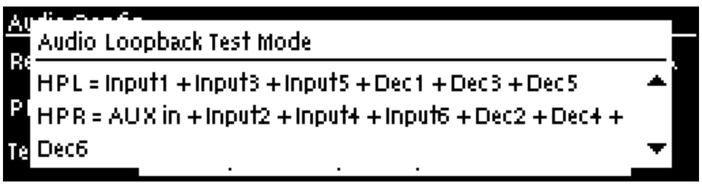
Important Note: For more information on SNMP codec settings see Configuring SNMP in the Codec.

22.11 Test Mode

Test mode is used by the codec to perform an input/output loopback test of audio. E.g. Input 1 is routed to Output 1, Input 2 is routed to Output 2 etc.

1. Press the **SETTINGS** button.
2. Navigate to **Audio** and press .
3. Navigate to **Test Mode** and press
4. Navigate to **Info** and press to view the **Audio Loopback Test Mode** summary.

Loopback input/output test mode is enabled while the **Audio Loopback Test Mode** dialog appears on the screen. When you navigate out of this screen this test mode ceases.



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 Installing the Codec at the Studio

Studio IP Streaming Setup for Tieline Audio Codecs

The following instructions are intended to help you configure your internet connection and Tieline codecs at the studio to enable incoming calls over the internet from a remote Tieline codec. It is assumed that you have a basic understanding of your IP network and how to configure IP devices. If you have limited IT network knowledge, we recommend you engage the services of an IT professional to install the public IP address and perform the Network Address Translation (NAT) and port forwarding between the public internet and your private Local Area Network (LAN) at the studio.

Prerequisites

The following procedures are valid for:

- All firmware versions in the Genie and Merlin codec families.
- All Bridge-IT Basic and Pro and Bridge-IT XTRA codecs with firmware release v.2.x or higher.
- All Commander G3 and i-Mix G3 codecs.

Getting Started at the Studio

To perform a typical codec installation at the studio you will need to:

1. Contact your Internet Service Provider and organize a dedicated high speed broadband connection at the studio for your codec with a public static IP address. Do not share this connection with other devices.
2. Install your codec at the studio and attach an active RJ-45 LAN cable to the “LAN” or “Ethernet” port on the rear of the codec. Please note:
   - The green LED underneath the “LAN” or "Ethernet" port will illuminate and the orange LED will flash steadily if you are connected to an active LAN connection.
   - The Genie and Merlin families of IP codecs support two simultaneous Ethernet connections.
3. If you are connecting a single codec to a router without a firewall you can enter the public IP address, Subnet Mask and Gateway directly into the codec and your work is done. Note: your Telco should be able to provide this information.
4. Alternatively, if you are connected to a router with a firewall, configure Network Address Translation (NAT) in your router. NAT is performed between the public internet and your private Local Area Network (LAN) by your router. Your remote codec sends IP data packets to the studio router's public static IP address and the router performs NAT, which forwards these data packets to the private IP address allocated by the router to your codec. As part of this process we recommended you:
   - Connect to your router using a web-browser.
   - Configure it to allocate a static private IP address for each codec.

Important Note: The IP address may change if the codec is allocated a DHCP IP address by the router and it loses power or is temporarily disconnected from the LAN. This will cause problems for remote codecs attempting to dial and connect.

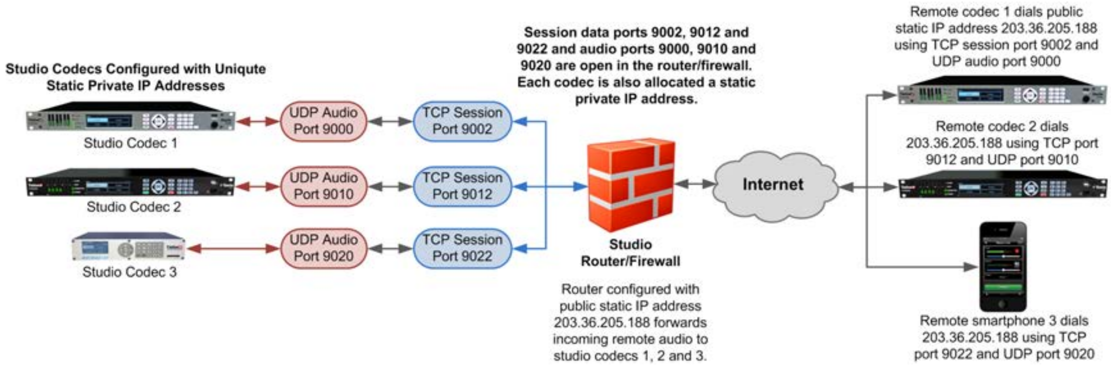
5. Ensure your router's firewall is configured with the relevant TCP and UDP IP ports open to allow data traffic between your codec and the remote codec. The process is fairly simple if you use the following procedure:

   a. Connect to your router using a web-browser.
   b. Navigate to http://portforward.com/english/applications/port_forwarding/Tieline-G5/default.htm (Note: when configuring a Commander or i-Mix G3 codec at the studio use

http://portforward.com/english/applications/port_forwarding/Tieline-G3/default.htm)

c. Click to select your router manufacturer from the list.
d. Next, click to select your router model from the list.
e. Follow the instructions to complete port forwarding

6. Visit www.portforward.com and download the port checking application to verify your router's ports are open.
7. Configure the static IP address in your codec using the instructions in the next section. To allow multiple codecs to share a single public static IP address behind a firewall and route the calls correctly, your codecs and the firewall need to be configured similarly to the example diagram which follows. Ensure the port, IP address, Subnet Mask and Gateway settings in your codecs match those configured in your router.



Port Forwarding to 3 Studio Codecs Sharing a Public Static IP Address

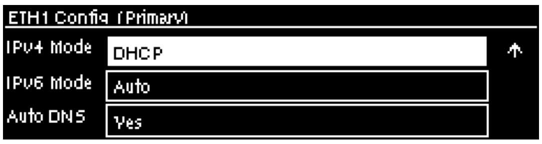
Important Note:
- The most common studio configuration issue is a firewall which blocks the incoming and/or outgoing TCP and UDP ports, or not configuring NAT and port forwarding correctly. The following table lists the firewall ports you need to open for each model of Tieline codec if they are dialing your router at the studio. If the remote codec is also connected to a LAN with a firewall you may also need to open the ports at the remote end of the link to connect successfully.
- Some firewalls require symmetric port configuration. The codec supports configuration of the "send" audio port (codec port at the remote end of the link to which you are sending audio) and "return" audio port (port used by the local codec to receive audio from the remote codec).

Firewall Ports					
Commander G3/i-Mix G3		Bridge-IT/Bridge-IT XTRA		Merlin and Genie Codec Families	
TCP	UDP	TCP	UDP	TCP	UDP
IP1 Session Port: 9002	IP1 Audio Port: 9000	Session Port (Sess): 9002	Audio (Proto): 9000	Session Port: 9002	Audio Port Stream 1: 9000
IP2 Session Port: 9012	IP2 Audio Port: 9010	Web-GUI: 80	SIP Session: 5060	Alternative Session: 9012	Audio Port Stream 2: 9010
Toolbox Software: 5550	Toolbox Software: 5550	Alternative Session: 9012	SIP Audio: 5004	Web-GUI: 80	Audio Port Stream 3: 9020
	SIP Session: 5060	Alternative Web-GUI: 8080		Alternative Web-GUI: 8080	Audio Port Stream 4: 9030
	SIP Audio: 5004				Audio Port Stream 5: 9040
					Audio Port Stream 6: 9050
					SIP Session: 5060
					SIP Audio: 5004, 5006, 5008, 5010, 5012, 5014

Configuring a Static Public or Private IP Address in Genie, Merlin and Bridge-IT (v.2.x firmware) Codecs

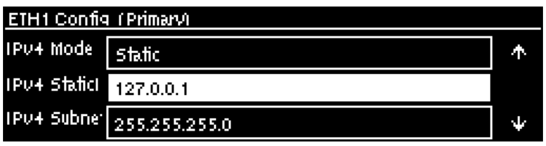
To enter a static IP Address into the codec for NAT:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Settings** and press .
3. Use the down navigation button to select **LAN** and press .
4. Select **Eth1** and navigate to IPv4 mode and press .



5. DHCP is enabled by default. Select **Static** and press .

6. The Static IP address menu is revealed after DHCP is disabled. Use the navigation buttons to select **IPv4 Static IP** and press .



7. Use the numeric **KEYPAD** to enter the IP address and press to store the setting. Note: use the * or # buttons to enter the periods in the IP address and use the **RETURN** button to delete any numbers already entered.

8. Enter changes to the **IPv4 Subnet** (Subnet Mask) or **IPv4 Gateway** (Default Gateway) in the same way if they are required (check with your network administrator for these settings).
9. After all changes have been made use the navigation buttons to scroll to the top of the menu and select **Apply Setting**, then press the OK button to save all changes.
10. From the **Home** screen select **Settings > Unit > Eth** in the codec menus to ensure the new static IP address has been entered correctly.

Configuring a Static IP Address in Commander G3 and i-Mix G3 Codecs

To set up a static IP address in Commander G3 and i-Mix G3 codecs select **Menu > Configuration > Advanced > LAN settings > IP Setup > Setup > Static > IP Address > [enter IP address] > press OK > Subnet Mask [enter Subnet Mask] > press OK > Gateway [enter Gateway] > press OK > reboot the codec**.

Record IP Address Details

IPv4 Static IP Address	
IP Address	. . .
Subnet Mask	. . .
Default Gateway	. . .
IPv6 Mode: Manual	**(Bridge-IT, Genie and Merlin codecs only)**
IP Address	: : : : : : :
IPv6 Prefix Size	
IPv6 Gateway	: : : : : : :

Getting Connected

Once the studio codec is configured you are now ready to receive an incoming call from the remote codec over the internet. Always dial from the field codec to the studio codec over the internet unless the remote codec is assigned a public static IP address and you know this address.

If you dial the studio using a cell-phone data network at the remote site you will not normally experience any firewall or port blocking issues at the remote end of the link using default Tieline ports.

Troubleshooting: How to Determine Where Firewall Port Blocking is Occurring

If you find you are unable to either send or receive audio between the studio and remote codecs you can use Tieline's Link Quality reading to diagnose where ports are being blocked. LQ can be displayed on the front LCD screen of Tieline's Bridge-IT, Merlin and Genie codecs by selecting

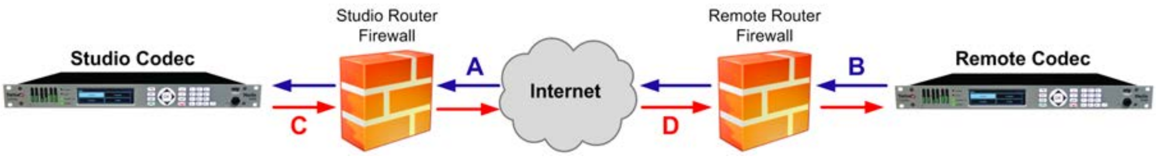
Cxns, then select the connection you want to view and press the **OK** button. LQ readings are also displayed on the home screen of all Commander and i-Mix G3 codecs.

Link Quality (LQ) Readings

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

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

Diagnosing Port Blocking via the Studio Codec LQ
If the studio codec **Return** LQ reading is **01** then incoming audio from the remote codec is being blocked by a firewall at either point A or B in the following diagram. If the studio codec **Send** LQ reading is **01** then outgoing audio from the studio is being blocked by a firewall at either point C or D in the following diagram.



Diagnosing Port Blocking via the Remote Codec LQ
If you attach your Tieline codec at the remote site to a LAN with access to the internet you can often dial and connect to the studio without any problem. It is less likely that a firewall will block outgoing TCP and UDP ports. However, if there is a firewall at the remote site it may block incoming data packets from the studio.

The principle is the same at the remote codec for diagnosing blocked ports. If the remote codec **Return** LQ reading is 01 then incoming audio from the studio codec is being blocked by a firewall at either point C or D in the preceding diagram. If the remote codec **Send** LQ reading is 01 then the outgoing audio from the remote codec is being blocked by a firewall at either point A or B in the preceding diagram.

Troubleshooting TCP Port Blocking

Error messages on the codec screen can help to diagnose TCP port blocking.

1. **"Connection Refused"** usually means that the firewall is configured correctly but the codec is not using the expected port. For example, the firewall is set up to forward via port 9002 but codec is 'listening' to port 10,000. “Connection Refused” is not normally shown if the firewall is not configured correctly because a firewall will by design silently drop any forwarding requests to ports that it doesn’t have open (see next point). Note: "Connection Refused" will also be displayed if the Commander G3 or i-Mix G3 codec you are calling is already connected.
2. **“Connection Timeout”** can mean one of two things:
   - The firewall is not configured correctly and the attempted codec connection is being

silently dropped, e.g. a remote codec is dialing to port 9002 but the studio firewall port forwarding is not configured.

- The UDP port is not port forwarded correctly. Tieline codecs send test data during connection establishment to make sure that the audio path is configured correctly; if this process fails then it will also result in a “Connection Timeout”.

How do I determine which end is blocking data flow?

Tieline test codec firewalls have the default Tieline TCP and UDP ports open. You can dial into these test codecs (or other codecs you know are configured correctly) from your recently configured studio and remote codecs and use the LQ readings to diagnose whether your studio or remote codec firewall is blocking your data packets. If one codec connects ok and the other one doesn't, then you will know which end is likely to be causing the problem. As an example:

1. Dial from site 1 to a Tieline test codec.
2. Dial from site 2 to Tieline test codec.

If both of these connect successfully then the “outbound” TCP path for session data is OK, and the inbound UDP audio path is OK.

3. Dial to site 1 from a codec you know is configured correctly.
4. Dial to site 2 from a codec you know is configured correctly.

If either of these calls fail then TCP and/or UDP inbound data is being blocked on the failed connection (see "Troubleshooting TCP Port Blocking" above).

Testing your Codec

- Visit http://www.tieline.com/Support/Test-lines for a list of test IP codec addresses you can use to verify your codec is configured correctly.
- See Testing IP Network Connections for more IP test information.

Learning More About IP Networks

For more IP network information please see the section titled Understanding IP Networks which discusses:

- Private versus public IP addresses.
- Static versus DHCP assigned IP addresses.
- Network Address Translation (NAT), port forwarding and firewalls.

23.3 Understanding IP Networks

Types of IP Addresses Available

	Type of IP Address	How the IP Address is Allocated	Description
Public	Static Public IP Address	Internet Service Providers (ISPs)	ISP's allocate a static public IP address to allow network devices to communicate with each other over the internet. It works like a public telephone number and will allow your remote codec to call your studio codec over the Internet.
	Dynamically Assigned Public IP Address	Internet Service Providers (ISPs)	ISP's usually allocate dynamically (automatically) assigned public IP addresses to allow network devices to communicate with each other over the Internet. (Not recommended for studio installations because each time you connect to your ISP the IP address can change).
Private	Dynamically Assigned Private IP Address	DHCP Server/Router on your own private LAN network.	A DHCP server-allocated IP address that is automatically assigned to a device on a LAN to allow it to communicate with other devices and the internet. This address can change each time a device connects.
	Static Private IP Address	LAN Administrator	A network administrator-allocated static address which is programmed into a device to allow it to connect to a LAN. Often a security measure to only allow access to devices approved by a network administrator.

Obtaining Public IP Addresses

To send audio streams over the public internet you need to use a public IP address assigned to you by your ISP (Internet Service Provider).

A public IP address is like your public telephone number and allows you to be contacted over the internet in much the same way people dial your public telephone number. They come in two forms; dynamic (DHCP) and static. Most ISPs assign a dynamic public IP address by default, which can often change without you knowing. This is suitable for a quick demo of your Tieline codec, but for a permanent installation you will need to request a permanent static public IP address.

Once the Static Public IP address is assigned to your internet connection (router) at the studio you need to create a link between the public IP address and your codec's private IP address on the LAN. This is called Network Address Translation.

Depending upon how your network is configured, it may also be possible to simply connect your Tieline codec directly into your ADSL modem/router and receive a public address from the router.

Private LAN IP Addresses

By default your Tieline codec will normally be automatically assigned a private IP address when you connect it to a typical router over a LAN.

Private IP Addresses are associated with LANs and normally reside behind a firewall and are not visible to the internet. They are generally in the ranges: 10.0.0.1 – 10.255.255.255, 169.254.0.0 – 169.254.255.255, 172.16.0.0 – 172.31.255.255 and 192.168.0.0 – 192.168.255.255 and are assigned by network DHCP servers and routers.

These IP Addresses are generally assigned for a predefined period (known as a lease) by your network's DHCP server or router. This IP address will generally expire after the lease period. DHCP assigned IP Addresses may also change if the device is disconnected for lengthy periods or if power to the device is turned off and back on. As a result, it is advised that you make this IP address permanent by assigning it as a Static DHCP IP Address. This will ensure you are able to always forward incoming audio packets to your codec using the same private IP address at the studio using port forwarding (see the section on port forwarding for more details). Consult your Network Administrator if you are unsure how to do this.

Network Address Translation (NAT)

Network Address Translation (NAT) is a method of connecting multiple devices to the internet using one public IP address.

The best way to explain NAT is to use the example of a phone system at an office that has one public telephone number and multiple extensions. This type of telephone system allows people to call you on a single public telephone number and performs the translation and routing of the public number to a particular private extension. Similarly, in order to receive an IP call from a remote codec over the public internet, the same network address translation principle applies. NAT and port forwarding allows a single device, such as a broadband router, to act as an agent between the public internet and a local private LAN.

The relationship between public and private IP addresses and NAT is displayed in the following diagram and the following section explains port forwarding configuration in more detail.

Port Forwarding: Tieline TCP and UDP Port Settings

For your Tieline Codec to communicate over the public internet an IP Address alone is not sufficient. In TCP/IP and UDP networks the codec port is the endpoint of your connection. Ports are doorways for IP devices to communicate with each other. Picture a house and imagine the front door is the entry point represented by a public or private IP address. Then you want to get to several codecs in different rooms of the same house and ports represent the doors to each of those rooms. In principle this is how port addressing works.

For example, several codecs may dial into your studio using the same public static IP address. In this situation it is necessary to configure codec 'programs' with audio streams using different audio ports for discretely routing each incoming and outgoing audio stream. By doing this your studio's network routers know where IP packets for each audio stream should be routed, i.e. to which codec and respective audio outputs.

When data packets are received from remote codecs at a particular public IP address, port information is translated from data packets to ensure the correct packets are sent to the correct studio codecs. This process is performed by PAT (Port Address Translation), which is a feature of NAT (Network Address Translation) devices.

Tieline codecs use TCP ports for setting up the communication session and UDP ports for streaming audio. While TCP ports are generally open, UDP ports are generally blocked by network devices which contain firewalls and will stop you delivering your audio. Depending on the codecs you are using, you need to configure your firewall to allow TCP and UDP protocols to pass through the ports listed in the table below.

23.4 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.

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.	

23.5 Testing IP Network Connections

There are a few very simple tools that you can use to test whether a codec can be reached over an IP network.

- Visit http://www.speedtest.net/ to test the upload and download speed of your IP connections and identify your public IP address.
- Visit www.portforward.com and download the port checking application to verify your router's ports are open. Note: Using a port scanner to test a codec will be unsuccessful if you try to scan and the port is already in use, i.e. the codec is connected.
- Visit www.subnetonline.com and use an online port scanner to check for open and closed TCP ports. This site also has numerous other software tools, including an online ping webtool for IPv4, plus TraceRoute and TracePath software tools.

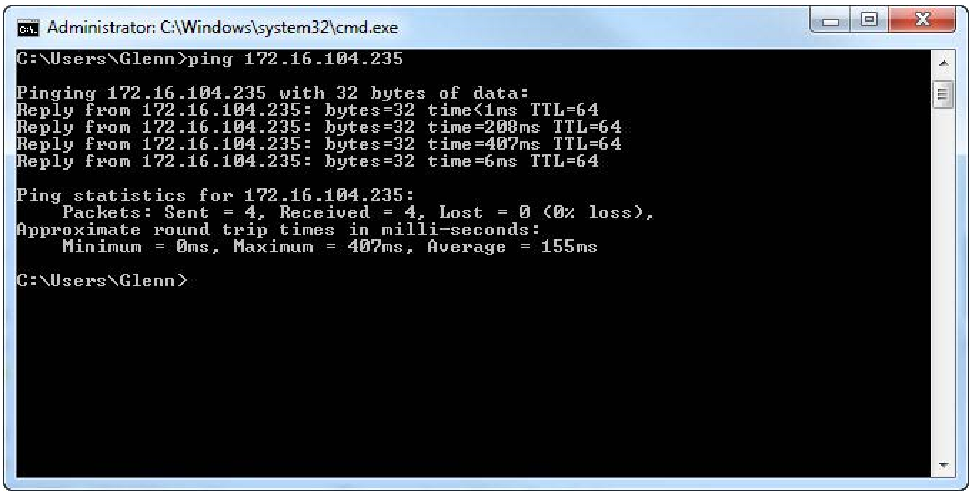
Ping the Codec

A ping test can be used to test whether it is possible to reach a codec or any device over an IP network. A ping test measures:

- The round-trip time of packets.
- Any packet loss.

There are two types of ping tests:

1. **Short test:** sends 4 packets and delivers statistics.
   i. Point to the **start** menu on your PC and click once.
   ii. In the search text box type **Run** and press **Enter.**
   iii. Type **CMD** in the **Run dialog** text box and click **OK**.
   iv. Type **ping** and the IP address of the codec you are pinging (i.e. **ping 192.168.0.159**) and press the **Enter** key on your keyboard.
   v. The round trip time of the packets is displayed, as well as any packet loss.



2. **Long test:** sends packets continuously until stopped.
   i. Point to the **start** menu on your PC and click once.
   ii. In the search text box type **Run** and press **Enter.**
   iii. Type **CMD** in the **Run dialog** text box and click **OK**.
   iv. Type **ping**, the IP address of the codec you are pinging, and then **-t** (i.e. **ping 203.36.205.163 -t**) and press the **Enter** key on your keyboard.
   v. Let the test run for several minutes and then press **CTRL C**.
   vi. The round trip time of the packets is displayed, as well as any packet loss for the period of time that the test occurred.

Trace the Route of IP Packets

Another utility available on your PC is traceroute. This tool can be used to determine the route and number of hops that data packets are taking to their destination (codec). This is useful because the more routers that packets traverse, the more latency your connection will have, and the less reliable it will be.

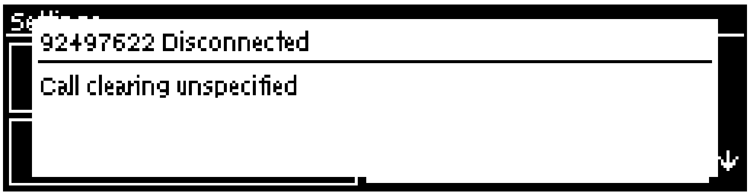
i. Point to the **start** menu on your PC and click once.
ii. In the search text box type **Run** and press **Enter.**
iii. Type **CMD** in the **Run dialog** text box and click **OK**.
iv. Type **tracert**, the IP address of the codec you are contacting (i.e. **tracert 203.36.205.163**) and press the **Enter** key on your keyboard.

23.6 Testing ISDN Connections

To test your ISDN line is working you can dial a standard phone line or your cell-phone number. If the call is successful this verifies the line is active. To verify ISDN data is being sent you can:

- Dial a codec you know is connected to an active ISDN line, e.g. another codec in your network or a Tieline test codec.
- Dial the test ISDN data number provided by your Telco (when available).
- Create a program and perform a loopback test by dialing out on the main ISDN number and receive the call on the auxiliary ISDN number. (**Note:** To create a loopback program create a 2 x Mono or Stereo Peer-to-Peer program and configure a dial only audio stream using your main ISDN number. For the second audio stream create an answer only audio stream connection configured for ISDN. If you dial the connection and can hear the audio you are sending on the return B channel, you have confirmed ISDN data is being sent successfully.

If you dial using a loopback program and a "disconnect" error message similar to the following image appears, you may have the incorrect **Line Type** configured.



Change the **Line Type** setting and this should hopefully resolve the issue.

On-Demand ISDN Services

If **Sync** appears for approximately 60 seconds when you connect an ISDN line to the codec and then disappears, or if **Sync** does not appear and you know you are connected to an active ISDN interface, then the line may have 'On-demand' enabled by your Telco. To test this you can dial a codec on an ISDN line known to be operational. Dial over ISDN and if **Sync** appears after connecting it indicates the service has now been activated. Disconnect and then dial again. If this dial is successful 'On Demand' is enabled. We recommend you contact your network service provider and get them to disable 'On Demand' to circumvent any possible connection issues.

23.7 Connecting Tieline ISDN to other Codecs

To dial from a Tieline codec to a non-Tieline codec it is necessary to disable 'Session Data' and use an algorithm like G.722 or MPEG Layer 2 for compatibility. The same settings must be configured at both ends for:

- Mono or stereo
- Encoding (Algorithm)
- Sample Rate
- Other relevant settings on the non-Tieline codecFollowing are configuration instructions for dialing to several non-Tieline codec brands over ISDN.

23.7.1 Connecting to APT Wordcast Equinox ISDN

Configuring the WorldCast Equinox to Make an ISDN Call

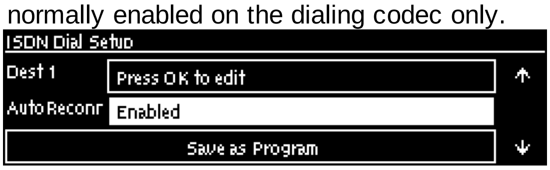
1. Plug your ISDN line into the back of the codec and attach power.
2. Press the "Menu" button on the codec to access the codec menus.
3. Press the "Menu" button to select the "USER" menu.

4. Select "Primary Connection" and press the "Ent/Dial" button.
5. Next select "Codec - ISDN" > "Use Audio Profile" [No] > "Eapt-X16", or "Eapt-X24", "MPEG1-L2" or G.722.
6. Select the appropriate bit rate and whether you are dialing in mono, stereo or Joint Stereo, and then the sample rate.
7. For bonded "MPEG1-L2" connections select "CCS IMUX".
8. Complete the profile setup. The codec is now ready to dial or answer.

Configuring the Tieline Codec to Dial the Equinox over ISDN

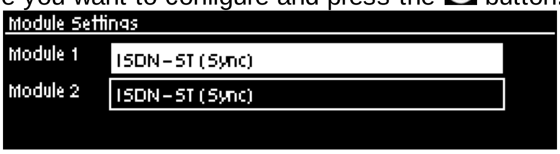
1. Press the **HOME** button to return to the **Home** screen and select **Connect > ISDN**.
2. Navigate to **Setup** and press the button.

3. Select **Session Type [Sessionless] > Select Dial Route [None] > Number of B Channels [**Choose the number of B-channels (between 1 and 4) required for your connection**] > Algorithm [**Choose **G.722** , **E apt-X Mono** or **Stereo**, **MP2 Mono or Stereo** or **MP2 J-Stereo (**Note: select **32kHz** or **48kHz** sample rate for **MP2** and **E apt-X** depending on available B-channels**)]**.
4. Navigate to a Destination (e.g. **Dest 1** or **Dest 2**) and press the button to select each one in turn. Enter the number for each B channel you want to dial and press the button, then select which B channel will dial using that number and press the button.
5. Navigate down to **Auto Reconnect** and press the button to toggle between **Enabled** and **Disabled**. Note: This is normally enabled on the dialing codec only.



6. Navigate down to **Save as Program** and press the button to save these settings as a program.
7. Navigate down to **Module Configuration** and press the button.

8. Select the ISDN module you want to configure and press the button.



9. Configure the following settings:
   - **Accept > Voice and Data**
   - **Network >** Check with your Telco (**EU-ETSI** in Australia; Europe & most countries outside North America; **[US Nat]** is the most common in the US, but check with your Telco).
   - **Line Type >** Check with your Telco and select either **Point-to-Multi** (point-to-multipoint) or **Point-to--Point** (point-to-point).
   - **DN/MSN >** Enter the "SPID" and "DN" numbers if required in your region, e.g. a SPID is

normally required in the US.

10. Navigate up to **Apply Settings** and press the OK button.

Dialing from the Tieline Codec

Program Dialing

1. If you have saved the ISDN program as previously instructed, press the **HOME** button to return to the **Home** screen and select **Connect > Programs**.
2. Select the saved program you want to load and press the OK button.
3. Select **Load** and press the OK button to load the program.
4. Press the **CONNECT** button to dial the ISDN program connections.

Ad Hoc Dialing

1. If you haven't save the program but have entered the dialing numbers and other settings, press the **HOME** button to return to the **Home** screen and select **Connect > ISDN > Connect**.
2. Press the OK button to dial using the settings previously entered.

Important Note: If you select different algorithm settings on each codec and dial from the Tieline codec, the connection will be unsuccessful and the **CONNECTED LED** on the front panel of the Tieline codec will continuously flash. Adjust the algorithm settings and attempt to reconnect.

Dialing from the WorldCast Equinox

Important Note: Configure **ISDN Answer Config** settings in the codec before attempting to dial from the Equinox to the Tieline codec. Select the following settings in the Tieline codec in one of the **Configs** (see ISDN Answering Configuration for more detail):

- **May bond**.
- **Sessionless**.
- Algorithm: **G.722**, **MP2 Mono**, **MP2 Stereo**, **MP2 J-Stereo** or **E apt-X Mono** or **Stereo**.
- Sample Rate: **32kHz** or **48kHz**

1. Navigate to the B-channel you want to dial over and press the "Ent/Dial" button.
2. Use the keypad to enter the number of the line you are dialing.
3. Press the "Ent/Dial" button again to make the outgoing call from the codec.
4. If dialing two B-channels, navigate to the second B-channel and use the keypad to enter the number, then press the "Ent/Dial" button. Note: the codec screen will display IMUX UNLOCKED until you dial additional connections when bonding multiple channels.

Important Note: When dialing a mono or stereo connection over two B-channels audio is not available until the second connection is successful.

23.7.2 Connecting to CDQ Prima ISDN

Use the following information to connect a Tieline codec to a Musicam CDQ Prima codec.

Programming the CDQ Prima for a Mono Connection

Select a mono profile in the Prima codec for the connection:

1. Press the “SDIAL” button on the front panel of the codec.
2. When “ID NUM” is displayed press “8” and then press "Enter" using the down arrow.
3. “MPEG2/64K:QS” will be displayed briefly followed by “WORKING”.
4. “OK” will be displayed momentarily and then the LCD screen will return to the default screen and be programmed for:
    - A Mono connection.
    - 64Kbps Bit-Rate.
    - 48K Sample Rate.
    - MPEG Layer 2 algorithm.
    - 1 ISDN B channel.
    - Decoder Independent – No.
5. Press the right arrow on the "Enter" button and navigate to "Interface". Push the down arrow on the "Enter" button to select this menu.
6. Use the "Enter" button and navigate to the type of interface you are using. Note: During Tieline tests we used an "Internal TA".
7. Select the actual terminal adapter connected to your codec. Note: During Tieline tests we used the internal "TA301".
8. Use the "Enter" button and select the switch type for the country you are in. Check with your Telco for the correct setting if you are unsure. Note: During Tieline tests we used the internal "NI1" setting for the USA.
9. Use the "Enter" button and keypad to enter the "SPID 1" and "SPID 2" numbers if required.
10. Use the "Enter" button and keypad to enter the "ID 1" and "ID 2" (Directory/MSN) numbers if required.
11. The codec should now be configured.

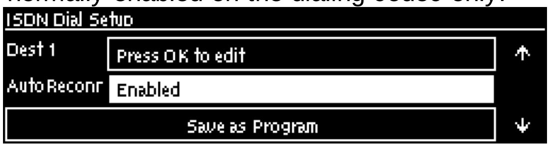
Programming the CDQ Prima for a Stereo Connection

1. Press the “SDIAL” button on the front panel of the codec.
2. When “ID NUM” is displayed press “27” and then press "Enter" using the down arrow.
3. “Zephyr/128K:QS” will be displayed briefly followed by “WORKING”.
4. “OK” will be displayed momentarily and then the LCD screen will return to the screen displayed prior to programming. The codec is now programmed for:
    - A Joint Stereo connection.
    - 128Kbps Bit-Rate.
    - 48K Sample Rate.
    - MPEG Layer 2 algorithm.
    - 2 ISDN B Channels
    - Decoder Independent – Yes
5. Press the right arrow on the "Enter" button and navigate to "Interface". Push the down arrow on the "Enter" button to select this menu.
6. Use the "Enter" button and navigate to the type of interface you are using. Note: During Tieline tests we used an "Internal TA".
7. Select the actual terminal adapter connected to your codec. Note: During Tieline tests we used the internal "TA301".
8. Use the "Enter" button and select the switch type for the country you are in. Check with your Telco for the correct setting if you are unsure. Note: During Tieline tests we used the internal "NI1" setting for the USA.
9. Use the "Enter" button and keypad to enter the "SPID 1" and "SPID 2" numbers if required.
10. Use the "Enter" button and keypad to enter the "ID 1" and "ID 2" (Directory/MSN) numbers if required.
11. The codec should now be configured.

Configuring the Tieline Codec to Connect to the CDQ Prima

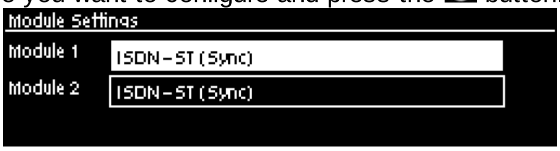
1. Press the **HOME** button to return to the **Home** screen and select **Connect > ISDN**.
2. Navigate to **Setup** and press the button.

3. Select **Session Type [Sessionless] > Select Dial Route [None] > Number of B Channels [**Choose **1B** (mono) **or 2B** (stereo)**] > Algorithm [**Choose **MP2 Mono or MP2 J-Stereo** (Note: select **48kHz** sample rate for MP2 algorithms)**]**.
4. Navigate to a Destination (e.g. **Dest 1** or **Dest 2**) and press the button to select each one in turn. Enter the number for each B channel you want to dial and press the button, then select which B channel will dial using that number and press the button.
5. Navigate down to **Auto Reconnect** and press the button to toggle between **Enabled** and **Disabled**. Note: This is normally enabled on the dialing codec only.



6. Navigate down to **Save as Program** and press the button to save these settings as a program.
7. Navigate down to **Module Configuration** and press the button.

8. Select the ISDN module you want to configure and press the button.



Important Note: ISDN Sync should be displayed when an ISDN line is connected to the codec. This appears regardless of whether you have configured the 'ISDN Line Type' correctly.

9. Configure the following settings:
   - **Accept > Voice and Data**
   - **Network >** Check with your Telco (**EU-ETSI** in Australia; Europe & most countries outside North America; **[US Nat]** is the most common in the US, but check with your Telco).
   - **Line Type >** Check with your Telco and select either **Point-to-Multi** (point-to-multipoint) or **Point-to--Point** (point-to-point).
   - **DN/MSN >** Enter the "SPID" and "DN" numbers if required in your region, e.g. a SPID is normally required in the US.
10. Navigate up to **Apply Settings** and press the button.

Dialing from the Tieline Codec

Program Dialing

1. If you have saved the ISDN program as previously instructed, press the **HOME** button to return to the **Home** screen and select **Connect > Programs**.

2. Select the saved program you want to load and press the OK button.
3. Select **Load** and press the OK button to load the program.
4. Press the **CONNECT** button to dial the ISDN program connections.

Ad Hoc Dialing

1. If you haven't save the program but have entered the dialing numbers and other settings, press the **HOME** button to return to the **Home** screen and select **Connect > ISDN > Connect**.
2. Press the OK button to dial using the settings previously entered.

After dialing successfully “FRAMED” should illuminate on the CDQ Prima screen. Tieline codecs also support 32kHz sampling.

Important Note: Configure **ISDN Answer Config** settings in the codec before attempting to dial from the Equinox to the Tieline codec. Select the following settings in the Tieline codec in one of the **Configs** (see ISDN Answering Configuration for more detail):

- **May bond**.
- **Sessionless**.
- Algorithm: **MP2 Mono**, **MP2 J-Stereo**.
- Sample Rate: **48kHz**

Making a Mono Call from the CDQ Prima Codec

1. Press the "Dial" button on the front panel of the codec.
2. Navigate right using the "Enter" button and select "1".
3. Enter the number to dial using the numeric keypad.
4. Press the "Enter" button (bottom arrow) and the screen will briefly display "Working", then "Connect" and then the green "Framed" light should illuminate on the front panel.

Making a Stereo Call from the CDQ Prima Codec

1. Press the "Dial" button on the front panel of the codec.
2. Use the "Enter" button and select "Both".
3. Enter the first number to dial using the numeric keypad.
4. Press the "Enter" button (bottom arrow) and the screen will briefly display "Dialling line 1" and then "Connect".
5. Enter the second number to dial using the numeric keypad and press the "Enter" button (bottom arrow).
6. The screen will briefly display "Dialling line 2", then "Connect" and then the green "Framed" light should illuminate on the front panel.

Important Note:
When connecting in stereo, the Prima expects both B channel dials to occur within 5 seconds. This can be performed by the Tieline codec.

It has also been noted that the CDQ Prima codec will not connect if no audio is present when dialing. It may connect Prima > Tieline, but not Tieline > Prima. If audio is present, the codec should connect and stay connected even if audio is removed subsequently. The J-Stereo light on the Prima may also flash when in this mode.

Ideally, have audio connected when dialing and the codec will frame immediately after the first dial and then dial the second B channel quickly afterwards.

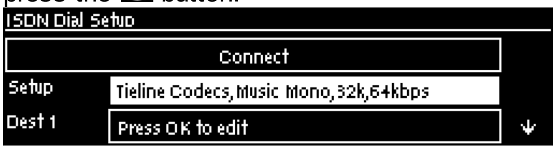
23.7.3 Connecting to Mayah ISDN

Configuring the Mayah Sporty to Make an ISDN Call

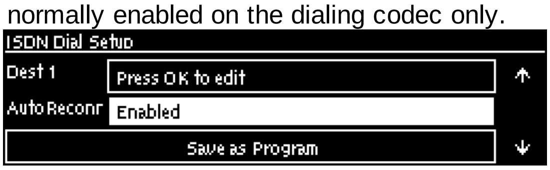
1. Plug your ISDN line into the codec and attach power.
2. Press "F2 Codec".
3. Press "F3 Setup"
4. Press "F2 Quality"
5. Use the navigation buttons to select an algorithm setting from "G.722", "L2 Mono, Stereo or J-Stereo" or "E apt-X Mono or Stereo", then press the "OK" button to save the setting.
6. Press "F4 ESC" to return to the home screen.
7. Press "F2 Codec", then "F1 Connect", then "F3 Direct".
8. Navigate to "interface" and press "OK" to select ISDN.
9. Navigate to "number1" and press "OK" to enter the ISDN number using the keypad, then press "OK".
10. If you are bonding multiple channels navigate to "number2" and press "OK" to enter the ISDN number using the keypad, then press "OK".

Configuring the Tieline Codec to Dial the Mayah Sporty over ISDN

1. Press the **HOME** button to return to the **Home** screen and select **Connect > ISDN**.
2. Navigate to **Setup** and press the button.

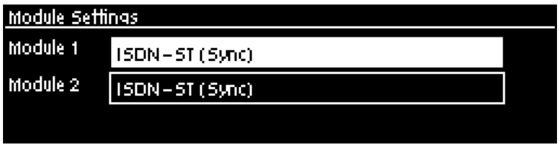


3. Select **Session Type [Sessionless] > Select Dial Route [None] > Number of B Channels [**Choose the number of B-channels (between 1 and 4) required for your connection**] > Algorithm [**Choose **G.722** , **E apt-X Mono** or **Stereo**, **MP2 Mono or Stereo** or **MP2 J-Stereo (**Note: select **32kHz** or **48kHz** sample rate for **MP2** depending on available B-channels**)]**.
4. Navigate to a Destination (e.g. **Dest 1** or **Dest 2**) and press the button to select each one in turn. Enter the number for each B channel you want to dial and press the button, then select which B channel will dial using that number and press the button.
5. Navigate down to **Auto Reconnect** and press the button to toggle between **Enabled** and **Disabled**. Note: This is normally enabled on the dialing codec only.



6. Navigate down to **Save as Program** and press the button to save these settings as a program.
7. Navigate down to **Module Configuration** and press the button.

8. Select the ISDN module you want to configure and press the button.



9. Configure the following settings:
   - **Accept > Voice and Data**
   - **Network >** Check with your Telco (**EU-ETSI** in Australia; Europe & most countries outside North America; **[US Nat]** is the most common in the US, but check with your Telco).
   - **Line Type >** Check with your Telco and select either **Point-to-Multi** (point-to-multipoint) or **Point-to--Point** (point-to-point).
   - **DN/MSN >** Enter the "SPID" and "DN" numbers if required in your region, e.g. a SPID is normally required in the US.
10. Navigate up to **Apply Settings** and press the button.

Dialing from the Tieline Codec

Program Dialing

1. If you have saved the ISDN program as previously instructed, press the **HOME** button to return to the **Home** screen and select **Connect > Programs**.
2. Select the saved program you want to load and press the button.
3. Select **Load** and press the button to load the program.
4. Press the **CONNECT** button to dial the ISDN program connections.

Ad Hoc Dialing

1. If you haven't save the program but have entered the dialing numbers and other settings, press the **HOME** button to return to the **Home** screen and select **Connect > ISDN > Connect**.
2. Press the button to dial using the settings previously entered.

Important Note: If you select different algorithm settings on each codec and dial from the Tieline codec, the connection will be unsuccessful and the **CONNECTED LED** on the front panel of the Tieline codec will continuously flash. Adjust the algorithm settings and attempt to reconnect.

Dialing from the Mayah Sporty

Important Note: Configure **ISDN Answer Config** settings in the Tieline codec before attempting to dial from the Equinox to the Tieline codec. Select the following settings in the Tieline codec in one of the **Configs** (see ISDN Answering Configuration for more detail):

- **May bond**.
- **Sessionless**.
- Algorithm: **G.722**, **MP2 Mono**, **MP2 Stereo**, **MP2 J-Stereo** or **E apt-X Mono** or **Stereo**.
- Sample Rate: **32kHz** or **48kHz**

1. Press "F4 ESC" to return to the home screen.
2. Press "F2 Codec", then "F1 Connect", then "F3 Direct".
3. Use the navigation buttons to select "dial" and press the "OK" button to dial all B-channels.

23.7.4 Connecting to Telos Zephyr Xstream ISDN

Configuring the Xstream to Make an ISDN Call

1. Plug your ISDN line into the back of the codec and press the "Codec" button below the LCD screen on the Xstream.
2. "Transmit" should be highlighted and this lets you select your transmit algorithm of choice. If it is not selected use the arrow buttons on the right-hand side of the LCD screen to navigate to this menu item and press the "SEL" button to the right of the LCD screen to select the menu.
3. Use the arrow buttons to navigate to:

- "G.722".
- "L2 J-Stereo" (for an MPEG Layer 2 stereo connection), or
- "L2 Mono 64" or "L2 Mono 128" (for a mono connection, depending on whether you have one or two B channels available).

4. Press the "SEL" button to store your setting and use the arrow down button to navigate to "Receive".
5. Press the "SEL" button and select the same algorithm that you selected for "Transmit" previously and then press the "SEL" button to store your setting.

Important Note: It you don't select the same algorithm for "Transmit" and "Receive" algorithms then it can take a long time to connect as the algorithms are scanned by the codec, or the wrong algorithm could be selected.

6. Use the arrow buttons to navigate to "Bitrate" and check that it displays "64kbps" - this is a per channel rate so both ISDN channels are programmed.
7. Use the arrow buttons to navigate to "Sample" and check that the sample rate is set at "48kHz". Press the "SEL" button and use the arrow buttons to make any adjustments to the current setting.
8. Press the "Tel" button below the codec LCD screen and press it again to display the "SPID" and "DN/MSN" screen. If these numbers need to be entered (check with your Telco), use the arrow buttons to navigate to each SPID and DN/MSN field in turn and when it is highlighted press the "SEL" button and enter the number using the keypad. Press "SEL" again to store each number once it has been entered.
9. Press the "Tel" button if you are not entering these SPID/DN/MSN numbers, or if you have already entered them, and check the local ISDN switch type setting is configured for your region.
10. Press the "SEL" button and use the arrow buttons to adjust the setting.

- Select "ETS300" if you are connecting to a Euro ISDN service.
- "Natl.I-1" is the most common in the US but check with your Telco.

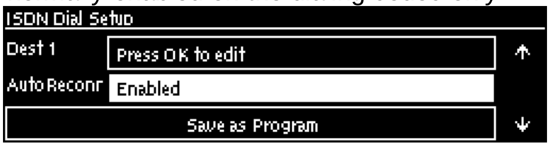
11. Press the "SEL" button to store the ISDN switch type setting that you have selected.

Your codec should now be configured. Press the "Tel" button on the front panel until it displays the "ISDN Status" screen. "Ready" should be displayed next to any active lines. If this is not displayed, check your connections and settings to make sure they are correct.

Configuring the Tieline Codec to Dial the Xstream over ISDN

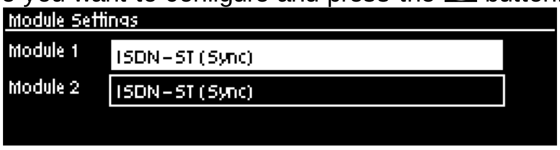
1. Press the **HOME** button to return to the **Home** screen and select **Connect > ISDN**.
2. Navigate to **Setup** and press the button.

3. Select **Session Type [Sessionless] > Select Dial Route [None] > Number of B Channels** [Choose **1B or 2B] > Algorithm** [Choose **G.722** , **MP2 Mono** or **MP2 Stereo (**Note: select **48kHz** sample rate for **MP2)]**.
4. Navigate to a Destination (e.g. **Dest 1** or **Dest 2**) and press the button to select each one in turn. Enter the number for each B channel you want to dial and press the button, then select which B channel will dial using that number and press the button.
5. Navigate down to **Auto Reconnect** and press the button to toggle between **Enabled** and **Disabled**. Note: This is normally enabled on the dialing codec only.



6. Navigate down to **Save as Program** and press the button to save these settings as a program.
7. Navigate down to **Module Configuration** and press the button.

8. Select the ISDN module you want to configure and press the button.



9. Configure the following settings:
   - **Accept > Voice and Data**
   - **Network >** Check with your Telco (**EU-ETSI** in Australia; Europe & most countries outside North America; **[US Nat]** is the most common in the US, but check with your Telco).
   - **Line Type >** Check with your Telco and select either **Point-to-Multi** (point-to-multipoint) or **Point-to--Point** (point-to-point).
   - **DN/MSN >** Enter the "SPID" and "DN" numbers if required in your region, e.g. a SPID is normally required in the US.
10. Navigate up to **Apply Settings** and press the button.

Dialing from the Tieline Codec

Program Dialing

1. If you have saved the ISDN program as previously instructed, press the **HOME** button to return to the **Home** screen and select **Connect > Programs**.
2. Select the saved program you want to load and press the button.
3. Select **Load** and press the button to load the program.
4. Press the **CONNECT** button to dial the ISDN program connections.

Ad Hoc Dialing

1. If you haven't save the program but have entered the dialing numbers and other settings, press the **HOME** button to return to the **Home** screen and select **Connect > ISDN > Connect**.

2. Press the  button to dial using the settings previously entered.

Important Note: If you select different algorithm settings on each codec and dial from the Tieline codec, the connection will be unsuccessful and the **CONNECTED LED** on the front panel of the Tieline codec will continuously flash. Adjust the algorithm settings and attempt to reconnect.

Dialing from the Zephyr Xstream

Important Note: Configure **ISDN Answer Config** settings in the codec before attempting to dial from the Xstream to the Tieline codec. Select the following settings in the Tieline codec in one of the **Configs** (see ISDN Answering Configuration for more detail):

- **May bond**.
- **Sessionless**.
- Algorithm: **G.722** or **MP2 Mono**, or **MP2 Stereo**.
- Sample Rate: **48kHz**

1. Press the "Dial" button once.
2. Use the keypad to enter the number of the line you are dialing.
3. Press the "Dial" button again to make the outgoing call from the Xstream.
4. The codec screen will briefly display "Outgoing Ring" and then "Conn" is displayed after a successful connection.
5. If you are making a stereo connection and need to dial the second line press the "Dial" button again and a screen for "Line 2" is displayed.
6. Use the keypad to enter the second number and press the "Dial" button again.
7. The "TEL" screen will briefly display "Outgoing Ring" and then "Conn" is displayed after a successful connection.

Two ISDN B Channels Connected

Important Note: When dialing a stereo connection over two ISDN B lines audio is not heard until the second connection is successful.

23.7.5 Connecting to Comrex Matrix ISDN

To connect your Tieline codec to a Comrex Matrix rack mount codec:

1. Use the G.722 algorithm.
2. Connect using only one 64Kbps ISDN B Channel (bonding of G.722 over two ISDN B channels is not possible).

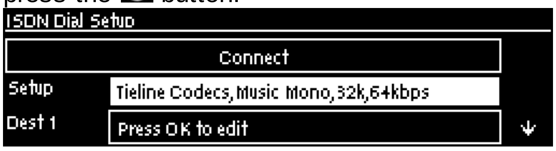
Configuring the Matrix to Make an ISDN Call

1. Connect an ISDN line to the Matrix codec and power up the unit.
2. Press "2" to select "ISDN Status".
3. Press "Enter" to configure the connection.
4. Press "4" to select the "Configure" menu.

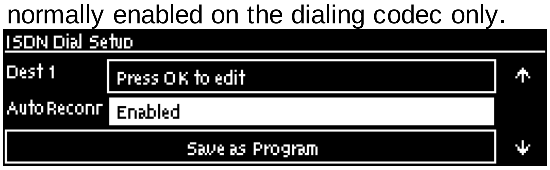
5. Press "2 to select the "Network" menu.
7. Press "4" to select "Profiles" and then press "1" to select "Load Profile".
8. Press "2" to select "Store" and program a new profile using the codec wizard.
9. Press ""Enter" to enter a profile number between 1 and 10. Note: This will overwrite any previously stored profile.
10. Next select the number for the ISDN "Switch Type" setting that is appropriate for your region.
- Press "4" to select "Euro" if you are connecting to a Euro ISDN service.
- Press "1" to select "NI1", which is the most common in the US, but check this with your Telco.
11. If prompted by the menu, use the keypad to enter the "SPID" number the line being used if this is required in your region. Press "Enter" to store the new number. Note: Use the "Cancel" button to delete numbers.
12. Next use the keypad to enter the "LDN" (DN/MSN) number for the line being used. Press "Enter" to store the new number. Note: Use the "Cancel" button to delete numbers.
13. Enter a "Qdial" (Quick Dial) number.
14. Press "1" to select "G.722" as the algorithm.
15. Press "2" to select "64" as the bit rate.
16. The codec is now programmed to dial.

Configuring the Tieline Codec to Connect to the Matrix over G.722

1. Press the **HOME** button to return to the **Home** screen and select **Connect > ISDN**.
2. Navigate to **Setup** and press the button.

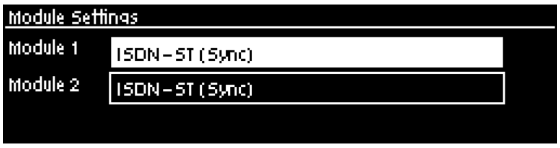


3. Select **Session Type [Sessionless] > Select Dial Route [None] > Number of B Channels [Select 1B] > Algorithm [Select G.722]**.
4. Navigate to a Destination (e.g. **Dest 1** or **Dest 2**) and press the button to select it. Then enter the number you want to dial and press the button, then select which B channel will dial using that number and press the button.
5. Navigate down to **Auto Reconnect** and press the button to toggle between **Enabled** and **Disabled**. Note: This is normally enabled on the dialing codec only.



6. Navigate down to **Save as Program** and press the button to save these settings as a program.
7. Navigate down to **Module Configuration** and press the button.

8. Select the ISDN module you want to configure and press the button.



Important Note: ISDN Sync should be displayed when an ISDN line is connected to the codec. This appears regardless of whether you have configured the 'ISDN Line Type' correctly.

9. Configure the following settings:
   - **Accept > Voice and Data**
   - **Network >** Check with your Telco (**EU-ETSI** in Australia; Europe & most countries outside North America; **[US Nat]** is the most common in the US, but check with your Telco).
   - **Line Type >** Check with your Telco and select either **Point-to-Multi** (point-to-multipoint) or **Point-to--Point** (point-to-point).
   - **DN/MSN >** Enter the "SPID" and "DN" numbers if required in your region, e.g. a SPID is normally required in the US.
10. Navigate up to **Apply Settings** and press the button.

Dialing from the Tieline Codec

Program Dialing

1. If you have saved the ISDN program as previously instructed, press the **HOME** button to return to the **Home** screen and select **Connect > Programs**.
2. Select the saved program you want to load and press the button.
3. Select **Load** and press the button to load the program.
4. Press the **CONNECT** button to dial the ISDN program connections.

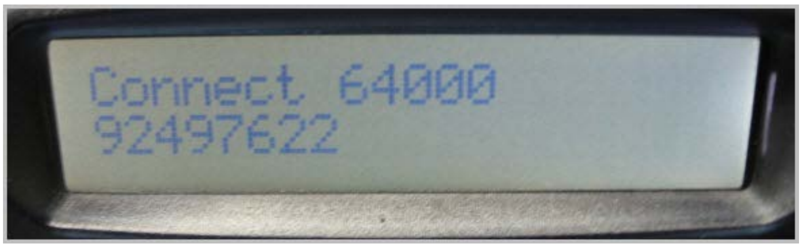
Ad Hoc Dialing

1. If you haven't save the program but have entered the dialing numbers and other settings, press the **HOME** button to return to the **Home** screen and select **Connect > ISDN > Connect**.
2. Press the button to dial using the settings previously entered.

Dialing from the Comrex Matrix

Important Note: Configure **ISDN Answer Config** settings in the codec before attempting to dial from the Comrex Matrix to the Tieline codec. Select the following settings in the Tieline codec in one of the **Configs** (see ISDN Answering Configuration for more detail) :
- **May bond**.
- **Sessionless**.
- **G.722** Algorithm

1. Use the "Cancel" button to return to the main LCD connection screen.
2. Press "Enter", then press "1" ("Dial") and use the numeric keypad to enter the number you wish to dial.
3. Press "Enter" to make the call.



Matrix Codec Screen when Connected

23.8 Using Answer Routes for Sessionless ISDN Calls

Tieline Genie Distribution and Merlin PLUS audio codecs support multiple connections using a variety of connection transports such as IP, ISDN and POTS. Tieline codecs support using Tieline session data, which assists with configuration and routing of multiple incoming calls to these codecs. In addition, audio ports can be used to successfully route IP calls to your preferred codec inputs/outputs.

If you are accepting calls from multiple non-Tieline ISDN codecs then you will be making "sessionless" connections which require the codecs at both ends to be configured with the same connection settings. In addition you can use “Answer Routes” and 'site-specific' module settings in Genie Distribution and Merlin PLUS to route incoming calls to specific codec outputs. (Note: Merlin codecs can also be configured to accept 2 ISDN calls from non-Tieline codecs and would use similar settings).

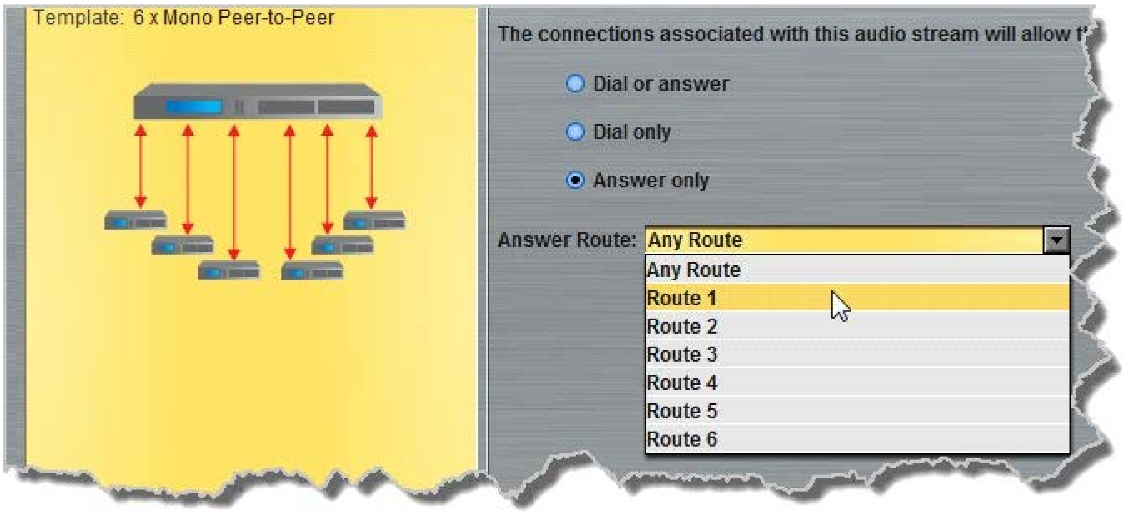
In the following example we will configure two incoming sessionless ISDN audio stream connections (Note: Genie Distribution and Merlin PLUS support up to 4 sessionless ISDN audio streams/connections using 2 ISDN modules and 4 B channels).

If you want 2 incoming mono ISDN calls to use input/outputs 1 and 2, then use answering audio stream connections 1 and 2 in your codec program. If you want to use other inputs/outputs then simply select the corresponding audio stream, e.g. answering audio streams 5 and 6 will route audio via inputs/outputs 5 and 6.

So let's get started. There are 2 or 3 steps to ensure this is configured correctly, depending on whether you want specific incoming calls to always use the same B channels and codec outputs or not.

Step 1: Configure the Answer Route for the two ISDN Audio Stream answering connections in the codec program.

Setup two ISDN audio stream answering connections in your program and use the **Answer Route** setting in the program wizard (as displayed in the following image):

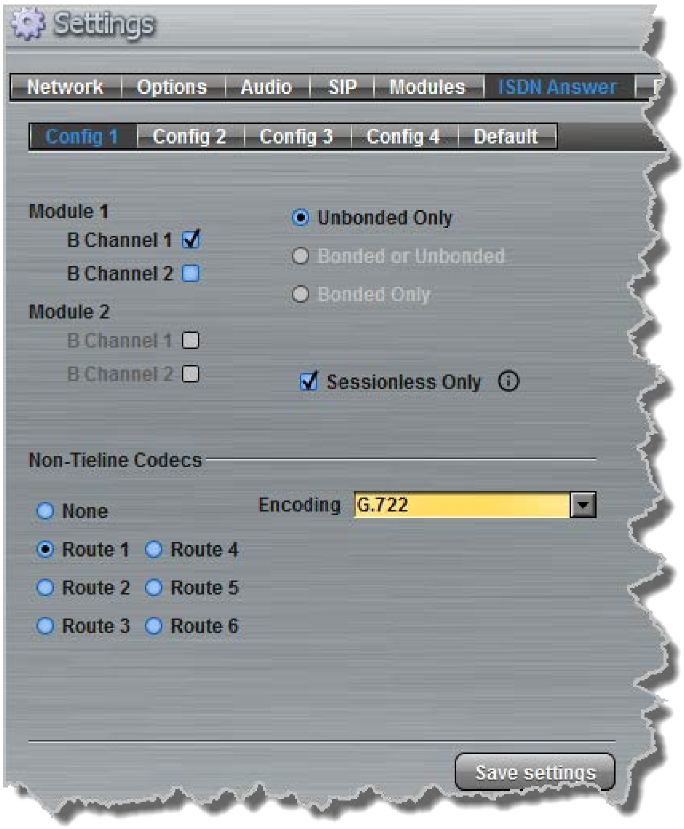


You can use any **Answer Route**, for example **Route 1** for ISDN Audio Stream 1 and **Route 2** for ISDN Audio Stream 2. The **Answer Route** number doesn’t have to match the audio stream number because the route you select will be used by the incoming ISDN call. This is similar to how an “extension number” is used to route a phone call.

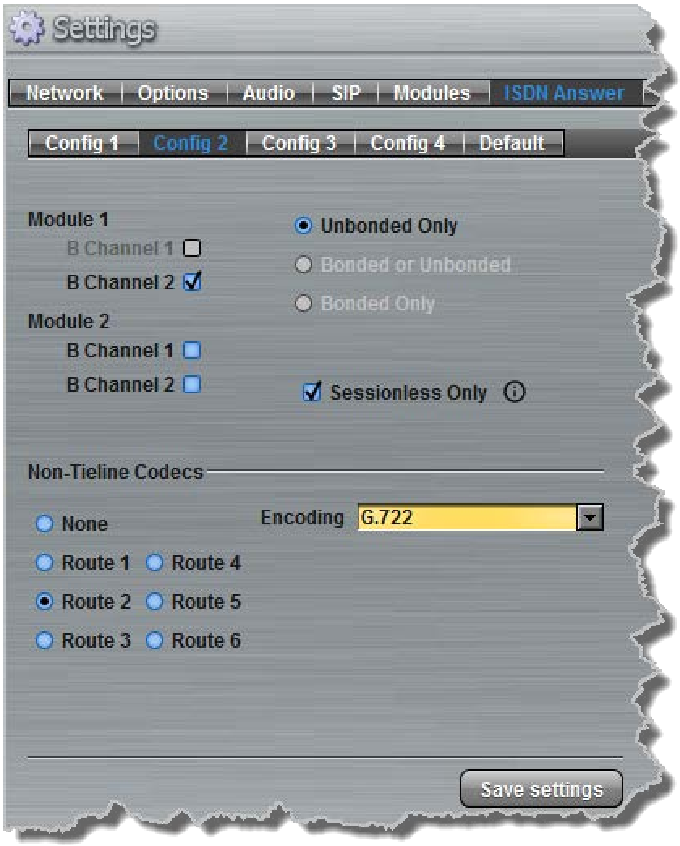
Step 2: Configure the ISDN Module to accept two sessionless ISDN calls.

This can be configured via **Settings > Modules** or use the Toolbox Web-GUI via **Settings panel > Modules**.

1. Select **Config 1** and **Sessionless Only** and **Route 1**. Select your preferred algorithm, then click **Save Settings**. This means that Module 1 B channel 1 will answer a sessionless ISDN call using these settings.



2. Next select **Config 2** and **Sessionless Only** and **Route 2**. Select your preferred algorithm, then click **Save Settings**. This means that Module 1 B channel 2 will answer a sessionless ISDN call using these settings.

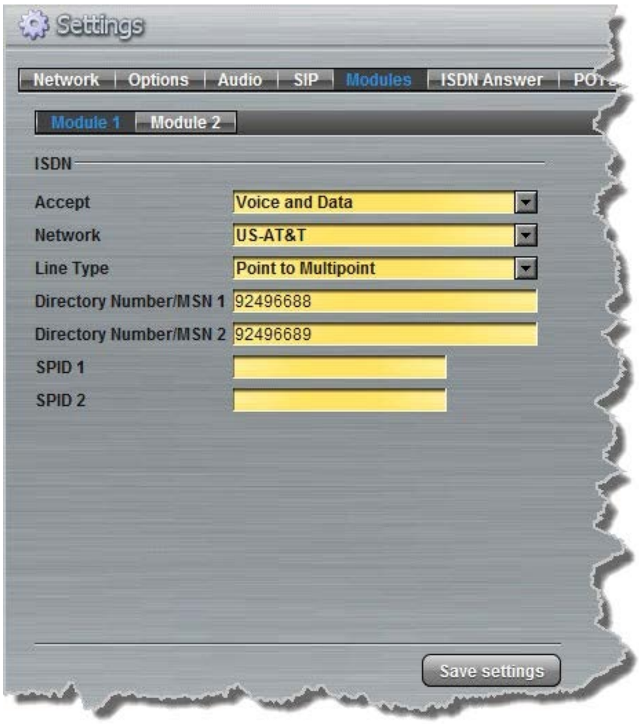


Both ISDN B channels can now answer incoming sessionless ISDN calls. If it doesn't matter which incoming codec call is answered by which B channel then that's all you need to do. If, however, you want each non-Tieline codec to use the same B channel and be routed to the same codec output consistently, you must configure this in the site config for the ISDN module via **Settings > Answering > ISDN Answer Configs**, or via the Web-GUI using the **Settings panel > Modules**.

Step 3: Configuring the module to answer calls from a specific non-Tieline codec consistently.

If a Directory Number (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.

Enter the number for the first B channel into the field for **Directory Number/MSN1**. (This has been allocated **Route 1** previously.) Enter the number for the second B channel into the field for **Directory Number/MSN2**. (This has been allocated **Route 2** previously.) Next, click **Save Settings**.



If codec 1 always uses the first directory number to call then it will always be routed via **Route 1** to the Answering Audio Stream Connection using **Answer Route 1** (configured in step 1). Codec 2 should always use the second directory number and then it will always be routed via **Route 2** to the Answering Audio Stream Connection configured with **Answer Route 2**.

23.9 POTS Connection Tips & Precautions

POTS Operation Precautions

POTS performance is greatly affected by the quality of the line being used. Precautions must be taken to ensure the Tieline codec is not sharing the line with other devices. Please remove these possible sources of interference:

- DSL or ADSL Modems
- Other telephone handsets
- Portable phone base stations
- Unused parallel phone sockets
- Fax machines
- Computer modems
- Burglar alarm systems
- Extension bells
- Call waiting

Call Waiting

Call waiting tones may cause the codec to malfunction. Most phone companies supply call waiting as a feature and you will need to turn it off. Your Telco should be able to provide a number you can dial to disabling the call waiting feature on the line.

Private Branch Exchanges

Avoid connecting the codec to a digital PBX or PABX system, key station, business system or any other local switchboard. It may sometimes be tricky to detect if you are connected to one of these systems, however, as a general guide, these devices require you to dial an additional digit to access the POTS network.

WARNING: Many of these systems are digital and have non standard telephone line operating voltages. If you attach your POTS module to a digital PABX or PBX system permanent damage may result from the high voltage pulses these systems generate. Even if the PBX is not digital, the performance of the codec is unlikely to be as good as a normal POTS line.

If you have no option other than to use a PBX or PABX System, search for a fax machine. The overwhelming majority of fax machines are designed for analog POTS line operation and are normally on an extension optimized for fax machines and data transmission. Substitute a normal phone for the fax machine to verify correct operation. Use a normal phone, not a venue-supplied phone, because this may have characteristics to match the existing PBX/PABX and not a POTS line. After confirming correct phone operation, you can unplug the phone and attach the phone line to the codec.

Check the Length of the Line

It is desirable to have a local loop which is as short as possible, i.e. the line from your location to the local Central Office or Local Exchange. Optimum performance can be expected for lines up to about 2 miles (3 kilometers) in length. Line quality will be reduced over longer distances and the codec can be expected to perform at lower bit rates. Line quality will also be affected by the age, condition and type of cabling used, e.g. plastic insulation or paper insulation, water or moisture entering the cable, age and state of repair of joins.

POTS Party Lines or Stubs

In some countries, it was the practice to have more than one phone service attached to one line - sometimes called a 'Party Line'. As more lines are installed, services are separated but the redundant cabling may remain connected across the line, causing problems with the operation of your codec.

Leakage Problems on the Line

A good line should have an earth isolation of better than ten mega-ohms. If your line is located in an area where water is a problem, ask your Telco to check out the earth leakage.

Equipment Problems at the CO or Local Exchange

Although there are many factors at the Telco end that can cause problems, a problem that does occasionally occur is that the clock on the interface codec to your line is not synchronized to the network. A drifting clock will cause instability and unreliable codec performance. If you suspect that this could be the problem, contact your local Telco.

POTS Exchange Problems

On most good POTS lines, Tieline codecs can achieve a 28.8kbps connection at a line quality of approximately 50% or greater. If you are not able to achieve this level of operation, you may have a problem with your line, or the line at the other end of the connection.

Tips for Successful POTS Operation

1. Take a phone when you are doing a remote broadcast. Connect it to the line you want to use and

dial the number to check for any unusual noises. If present, these may be caused by other devices connected to the line.

2. Take an ADSL/DSL filter to all remote locations. ADSL/DSL modems can generate noise on a line which will degrade the performance of your codec. Simply place the ADSL/DSL filter between the POTS line and your codec to remove the interference.
3. Tieline USA has a POTS test codec you can dial on +1-317-913 6911 to facilitate line tests at each end of your connection to diagnose line problems.
4. Tieline recommends that you confirm your broadcast POTS line works well before you try to go live.

23.10 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.

23.11 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 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
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 at 48kHz	20Hz to 22kHz
THD and Noise (Analog)	<0.0035% at +16dBu or -89dBu unweighted
THD and Noise (Digital)	<0.000056%
Analog 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
Advanced Networking	VLAN tagging (IEEE 802.1Q,802.1p)
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 Supported	Tieline, DHCP, SNMP, DNS, HTTP, IGMP, ICMP, VLAN, IPv4/v6, FEC, SIP/SDP (EBU N/ACIP Tech 3326 compliant), RTP, I3P EBU3347 compliant
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 Temp.	0°C to 45°C (32°F to 113°F)
Humidity Operating Range	20% ≤RH ≤70% (0 to 35°C/32°F to 95°F), non-condensing

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.

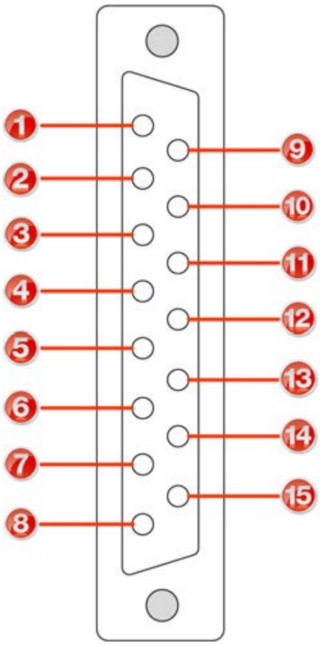
Operation is as simple as joining the input pin to the ground terminal. This can be via a remote relay contact or the open circuit collector of a transistor or FET. DO NOT feed voltages into the inputs.

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.



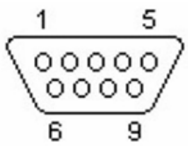
Female DB-15 Codec Connector

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

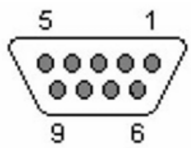
Important Note: For more information about how to program relay operations with a PC using the Toolbox Web-GUI, please see Creating Rules.

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.

Index

- 2 -

- A -

- B -

- C -

- D -

- J -

- K -

- L -

- M -

- X -