Tieline®
The Codec Company

Codec Quick Start for Genie STL, Genie Distribution, Merlin and Merlin PLUS

Software Version: 2.20.xx
Manual Version: 3.0
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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 THE ANALOG POTS MODULE 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.
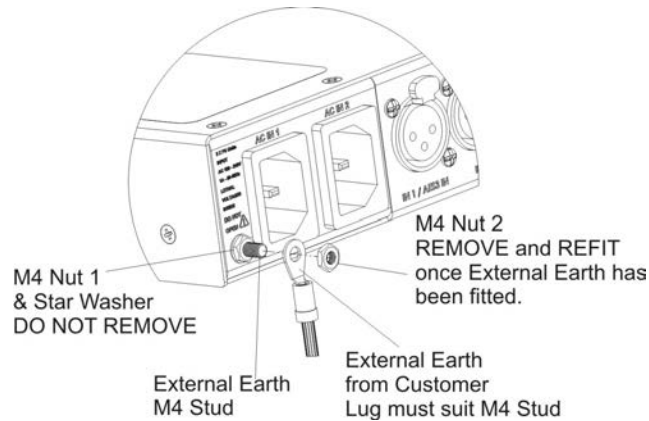
WARNING:
HIGH LEAKAGE CURRENT. EARTH CONNECTION ESSENTIAL BEFORE CONNECTING SUPPLY.
If the total leakage current exceeds 3.5 mA, or if the leakage current of the connected loads is unknown, connect the supplementary ground terminal to a reliable ground connection in your facility.

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.

M4 Nut 1
& Star Washer
DO NOT REMOVE

M4 Nut 2
REMOVE and REFIT
once External Earth has
been fitted.

External Earth
M4 Stud

External Earth
from Customer
Lug must suit M4 Stud

SAFE LISTENING GUIDANCE

WARNING: LISTENING TO AUDIO AT EXCESSIVE VOLUMES CAN CAUSE PERMANENT HEARING DAMAGE. USE AS LOW A VOLUME AS POSSIBLE.

Over exposure to excessive sound levels can damage your ears resulting in permanent noise-induced hearing loss (NIHL). Please use applicable health and safety authority guidelines on maximum exposure limits. As a rule of thumb, avoid extended periods listening to sound pressure levels (SPLs) of 85dBA or higher.

Warranty and Disclaimer

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

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

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

Whilst every effort has been made to ensure the accuracy of this manual we are not responsible for any errors or omissions within it. The product specifications and descriptions within this manual will be subject to improvements and modifications over time without notice, as changes to software and hardware are implemented. Tieline takes no responsibility for any damage to equipment attached to the codec.

2   Install the Codec and Configure Network Settings

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.

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 https://portforward.com/router.htm
   c. Select your router manufacturer from the list.
   d. Next, select your router model from the list.
   e. Follow the instructions to complete port forwarding. Note: It is necessary to select the device you are configuring in the drop-down "Application" menu (usually step 4). Select "Tieline Bridge-IT", "Tieline-G3" or "Tieline -G5" to suit the codec you are configuring.

6. Visit www.portforward.com and download the port checking application to verify your router's ports are open.
7. Configure the static IP address in your codec using the instructions in the next section. To allow multiple codecs to share a single public static IP address behind a firewall and route

the calls correctly, your codecs and the firewall need to be configured similarly to the example diagram which follows. Ensure the port, IP address, Subnet Mask and Gateway settings in your codecs match those configured in your router.
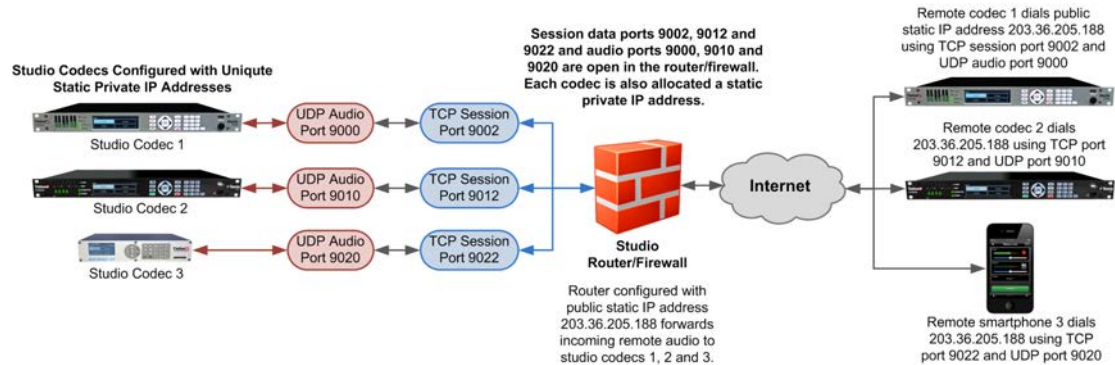
Port Forwarding to 3 Studio Codecs Sharing a Public Static IP Address

Important Note:
- The most common studio configuration issue is a firewall which blocks the incoming and/or outgoing TCP and UDP ports, or not configuring NAT and port forwarding correctly. The following table lists the firewall ports you need to open for each model of Tieline codec if they are dialing your router at the studio. If the remote codec is also connected to a LAN with a firewall you may also need to open the ports at the remote end of the link to connect successfully.
- Some firewalls require symmetric port configuration. The codec supports configuration of the "send" audio port (codec port at the remote end of the link to which you are sending audio) and "return" audio port (port used by the local codec to receive audio from the remote codec).

Firewall Ports								
Commander G3 / i-Mix G3		Bridge-IT / Bridge-IT XTRA		Merlin and Genie Codec Families		ViA Codec		Cloud Codec Controller
TCP	UDP	TCP	UDP	TCP	UDP	TCP	UDP	TCP
IP1 Session Port: 9002	IP1 Audio Port: 9000	Session Port (Sess): 9002	Audio (Proto): 9000	Session Port: 9002	Audio Port Stream 1: 9000	Session Port: 9002	Audio Port Stream 1: 9000	HTTP 80
IP2 Session Port: 9012	IP2 Audio Port: 9010	Web-GUI: 80	SIP Session: 5060	Alternative Session: 9012	Audio Port Stream 2: 9010	Alternative Session: 9012	Audio Port Stream 2: 9010	HTTPS 443
Toolbox Software: 5550	Toolbox Software: 5550	Alternative Session: 9012	SIP Audio: 5004	Web-GUI: 80	Audio Port Stream 3: 9020	Web-GUI: 80	SIP Session: 5060	
	SIP Session: 5060	Alternative Web-GUI: 8080	Fuse-IP 8999	Alternative Web-GUI: 8080	Audio Port Stream 4: 9030	Alternative Web-GUI: 8080	SIP Audio: 5004-5054	
	SIP Audio: 5004	TLS/SSL 443		TLS/SSL 443	Audio Port Stream 5: 9040	TLS/SSL 443	Fuse-IP 8999	
					Audio Port Stream 6: 9050			
					SIP Session: 5060			
					SIP Audio: 5004-5054			
					Fuse-IP 8999			

Configuring a Static Public or Private IP Address in Genie, Merlin and Bridge-IT (v.2.x firmware) Codecs

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. To enter a static IP Address into the codec for NAT:

1. Press the **SETTINGS** ⌱ button.
2. Select **Network** and press the [OK] button.
3. Use the down ▼ navigation button to select **ETH1**, **ETH2**, **Wi-Fi** or a **VLAN** interface.
4. Select **Config** and then **Usage** and then the appropriate control and/or streaming mode for the connection. Next, press the [OK] button.
5. Select **IPv4 Mode** and press the [OK] button.
6. Select **Static** and press the [OK] button.
7. Navigate to **IPv4 Static** and enter the IP address, then press the [OK] button.
8. Navigate to **IPv4 Subnet** and enter the Subnet Mask, then press the [OK] button.
9. Navigate to **IPv4 Gateway** and enter the Gateway details, then press the [OK] button.

ETH1: Config (Primary)
IPv4 StaticIP	10.1.1.10	↑
IPv4 Subnet	255.255.255.0	
IPv4 Gateway	10.1.1.254	↓

10. Use the up ▲ navigation button to scroll to the top of the menu and select **Apply Setting**, then press the ⌷OK button to confirm the new settings.
11. Check the **Unit Details** menu to ensure the new static IP address has been entered correctly.

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.

> ℹ️ **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.
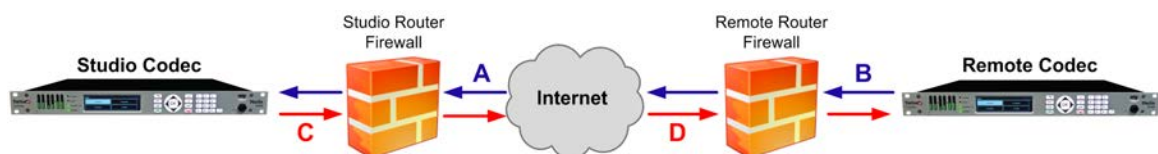
Diagnosing Port Blocking via the Remote Codec LQ

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, Using & Maintaining Your Codec

- Visit www.tieline.com 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.

- Visit http://www.speedtest.net/ to test the upload and download speed of your IP connections. This site will also identify your public IP address.
- Visit www.portforward.com and download the port checking application to verify your router's ports are open.
- To download the latest firmware visit www.tieline.com.
- To download the latest main user manual for your codec visit www.tieline.com.

3   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 OK button.
iii. Navigate to **Country** and press the OK button.
iv. Use the navigation buttons to select your country of operation and press the OK 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 ring tones and line impedances 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.
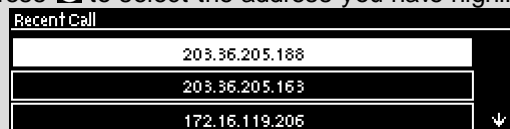
3.1   Steps to Connect over IP

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.

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 > IP > Tieline** and press the [OK] 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 (or use the **Virtual Keyboard**). Next, press the down ▼ navigation button to select **Setup** and press [OK].

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

4. Navigate to **Algorithm** and press [OK].

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

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 [OK] after each option is selected.

Bit Rate	
48kbps	64kbps
96kbps	112kbps

7. Press the down ▼ navigation button to select **Jitter Buffer** and press [OK] 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.

IP Connect Setup (Module 1:Ready)	
Algorithm	Music Mono,32k,28.8kbps
Jitter Buffer	Auto,Best Compromise
FEC	Loc:Off Rem:Off

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

IP Connect Setup (Module 1:Ready)	
Algorithm	Music Mono,32k,28.8kbps
Jitter Buffer	Auto,Best Compromise
FEC	Loc:Off Rem:Off

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

IP Setup:Music Mono 32k 28.8kbps	
Routing Type	Default
Auto Recon	Disabled
Via	Any

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

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

> **Important Note:** The G3 profile setting supports maintaining specific G3 codec settings when answering a call from a G5 codec.
>
> 1. **Auto**: The codec will dial the G3 codec and connect in mono or stereo. Note: This is overridden in a ViA codec when a G3 Main + IFB use-case is configured.
> 2. **Dual Program**: This allows the codec to dial a G3 codec with a Dual Program profile loaded and support two simultaneous mono connections.
> 3. **Runtime**: The G3 codec will retain runtime settings when answering a call from a G5 codec.
> 4. **Custom**: The G3 codec will load a specified profile, e.g. profile 6, which is the first custom profile number.

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

12. 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.

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

14. 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.

15. 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.

16. 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**.

3.2   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 [OK] button.
3. If 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 [OK] 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.

> ℹ️ **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.
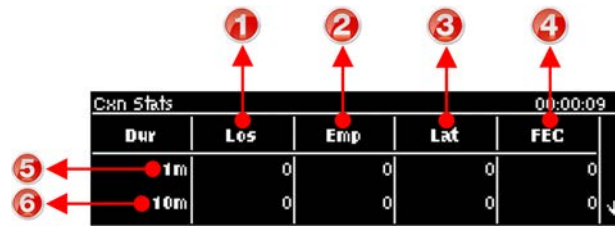
	Feature	Description
1	Lost Packets	Packets sent that failed to arrive
2	Empty (Jitter Buffer)	Indicates how often the jitter buffer 'reservoir' empties causing loss of audio
3	Late Packets	The number of packets that arrive late, i.e. after audio play out
4	FEC Packets	Indicates the number of forward error correction (FEC) packets that have been used (replaced) when FEC is enabled
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.

3.3   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 to configure ISDN settings using the HTML5 Toolbox Web-GUI.
5. For detailed information about connecting with other brands of codec using ISDN visit www.tieline.com.

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.

IP	SIP
ISDN	Program

Connect

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

ISDN Dial Setup

Connect	
Setup	Tieline Codecs, Music Mono, 32k, 64kbps
Destination 1	1 from Any ↓

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

Session Type

Tieline Codecs	Sessionless

> **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.

Select Dial Route

None	Route 1
Route 2	Route 3 ↓

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

Number of B-channels

1B	2B
3B	4B

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

Algorithm

MP2 Mono	MP2 Stereo
Music Mono	Music Stereo ↓

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

Sample Rate	
32kHz	48kHz

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

ISDN Dial Setup	
Connect	
Setup	Tieline Codecs,MP2 Stereo,32k,128kbps
Destination 1	Press OK to edit

Destination 1 Number
92499999	

10. Select the preferred B channel or channels to use when dialing and press the [OK] button.

Select B-Channel	
Any	Module2, B-Any
Module2, B1	Module2, B2

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.

ISDN Dial Setup		
Setup	Tieline Codecs,MusicPlus Stereo,48k,128kbps	↑
Dest 1	92499999 from Any	
Dest 2	Press OK to edit	↓

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.

ISDN Dial Setup		
Destination 2	92499998 from Any	↑
Auto Recon	Enabled	
Save as Program		↓

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.

ISDN Dial Setup		
Destination 2	92499998 from Any	↑
Auto Recon	Enabled	
Save as Program		↓

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

Save Program
Studio1
Press OK to save

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.

> (i) **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.

3.4 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 [OK] 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 [OK] button to view more details.

| Connected ISDN		00:00:40
DIAL	55555555	↑
DIAL	55555556	
Alg	E apt-X Stereo 256kbps	

3.5 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.

> (i) **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 🆗 button.

Connect	
IP	SIP
ISDN	POTS ↓

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.

Connect POTS (Module2:Ready)

92498782|

Setup	Save

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

Connect POTS (Module2:Ready)

92498782

Setup	Save

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 🆗 button.

POTS Connect Setup (Module2:Ready)

Via	Module2
Mode	POTS Codec
Algorithm	Music Mono ↓

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**.

POTS Connect Setup (Module2:Ready)

Via	Module2
Mode	POTS Codec
Algorithm	Music Mono ↓

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

Algorithm

Music Mono	Other

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 [OK] 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 [OK] 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 [OK] to use the numeric **KEYPAD** to name the program. Then press [OK] 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**.

<table>
<tr>
<td>ⓘ</td>
<td>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.</td>
</tr>
</table>

See <u>Monitoring POTS Connections</u> for more details on monitoring the different POTS connection states.

3.6   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).
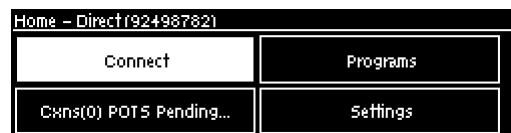
Cxns Displays Pending

   - **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 [OK] button.
   b. When multiple audio streams are connected, navigate to the one you want to view and press the [OK] 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.

3.7 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 OK 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.

3.8 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 OK button to confirm the disconnection.

Warning	
Hangup all connections?	
No	Yes