
# Tieline G6 Codec SIP Compatibility over IP

**Manual Version: 1.0**
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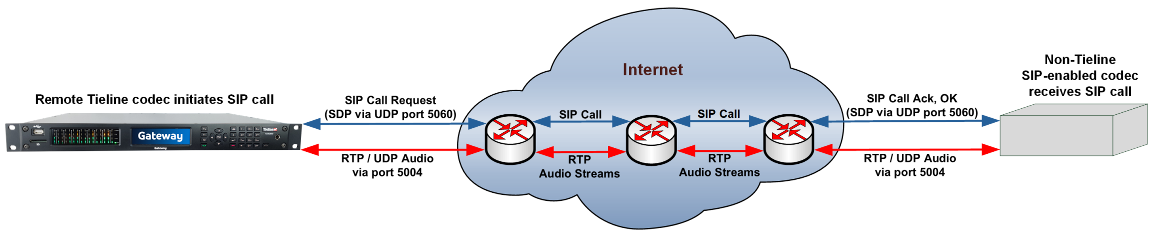
# 1 Connecting Tieline to other Codecs Using SIP

To dial between Tieline and non-Tieline codecs over IP it is necessary to configure all codecs to connect in SIP mode. SIP provides interoperability between different brands of codecs due to its standardized protocols for connecting different devices. Tieline IP codecs are EBU N/ACIP Tech 3326 compliant when connecting using SIP (Session Initiation Protocol) to other brands of IP codecs.

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

## Unregistered Peer-to-Peer SIP Connections

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



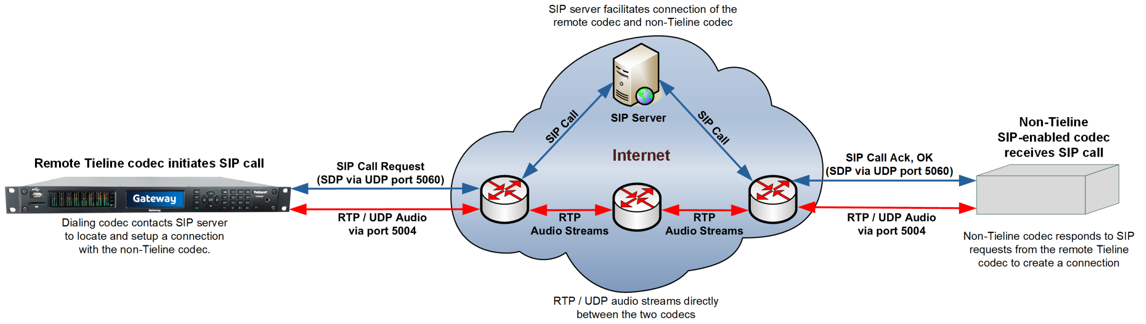

There are two very distinct parts to a call when dialing over IP. The initial stage is the call setup stage and this is what SIP and SDP is used for. The second stage is when data transfer occurs and this is left to the other protocols such as RTP/UDP to stream audio data. SDP works with a number of other protocols, to deliver the following functions when connecting devices over SIP:

- Establish a codec's location.
- Determine the availability of a codec.
- Negotiate the features to be used during a call, e.g. the algorithm and bit rate.
- Provide call management of participants.
- Adjust session management features while a call is in progress (e.g. termination and transfer of calls).

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

## Registered SIP Server Connections

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




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

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

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

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

## Advantages and Disadvantages of Using SIP

### Advantages of SIP

1. SIP provides interoperability between different brands of codecs due to its standardized protocols for connecting different devices, e.g. Session Description Protocol (SDP).
2. EBU N/ACIP Tech 3326 provides a minimum set of requirements necessary to ensure interoperability between equipment intended for the transport of audio over IP networks. It standardizes the use of ports, encoders, transport protocols and signaling to ensure codecs, and other SIP-enabled devices like smartphones and VoIP phones, can connect successfully.

### Disadvantages of SIP

1. When dialing Tieline to Tieline the dialing codec provides all connection information to the answering codec. This is not possible when using SDP in SIP mode.
2. Not all professional codec manufacturers are fully compliant with all requirements for interoperability.
3. SIP connections are more complex to configure.

4. SIP does not support advanced software enhancements which deliver redundancy and rock solid reliablity over IP, e.g. failover connections, SmartStream PLUS redundant streaming, Fuse-IP bonding, plus error concealment strategies.
5. Codecs using SIP cannot use the TieLink Traversal Server for presence and connections. In addition port forwarding is usually required.
6. Some ISPs and/or cellular networks may block SIP traffic.

## SIP Security

Tools such as Shodan make it easy for anyone to easily locate devices connected to the internet around the world. Therefore it is critical that security measures are in place for all IP and SIP connections over the public internet.

## Managing Unwanted SIP Calls

Hackers and other nefarious net-bound characters look for networks with easy access in which to ply their trade. As a starting point they look for networks with open gateways and platforms using default passwords.

## Maintaining Codec Network Security

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

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

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

# Understanding SIP Terminology

## What is a URI?

A Uniform Resource Identifier (URI) is a string of characters identifying a name or resource on the internet. A Uniform Resource Locator (URL) is a URI specifying where an identified resource is available (e.g. network location) and the mechanism for retrieving it. E.g. http://twitter.com, where http:// is the protocol used to retrieve information from "twitter.com" which is the network location.

## What is a SIP URI?

SIP URIs can be used to whitelist or blacklist devices from connecting to a codec. A SIP URI is the address or characters used to call another person via SIP. A SIP URI is essentially a user's 'contact address' and is used by VoIP phones and other devices to call using SIP. An example of a SIP URI is: "sip:tieline_test1@getonsip.com". When dialing using Tieline codecs "sip:" is automatically placed in the dial string and does not need to be entered. SIP URIs can be used to allow or block devices from connecting to a codec.

**Important Notes:**

- The default UDP audio port when using SIP for a peer-to-peer connection is 5004 in Tieline codecs.
- Enter the IP address or SIP URI, then a full colon and the session port number to change the session port from the default setting of UDP port 5060, e.g. "sip:tieline_test1@getonsip.com:5070"
- Tieline codecs automatically add "**sip:**" to the address entered in the **Address** field when dialing, so it's not necessary to add this.
- To only allow a predefined list of codecs to connect, add them to the **URI Whitelist** and add a wildcard (asterisk) * to the **URI Blacklist**: all incoming calls will be blocked except for codecs in the Whitelist.

## What is a SIP User Agent?

A user agent is a software agent acting on behalf of a user. Hackers use SIP user agents or 'botnets' to scan for open ports on the internet. When they locate open ports they can force their way into SIP servers and scan for valid accounts to use for fraudulent purposes, like making free international calls. Known user agents like "sipvicious" and "friendly-scanner" can be added to a User Agent Blacklist in Tieline codecs to stop them from accessing them.

## How do I find my Codec's User Agent

It may be necessary to identify a codec user agent as some codec manufacturers allow calls based on 'User Agent' identification. Therefore, it may be necessary to enter a Tieline codec user agent into a non-Tieline codec to allow it to connect to a Tieline SIP codec.

- From firmware v2.16.xx the user agent in the codec is configured as "Tieline <Product Name> <Firmware Version>". E.g. Tieline Bridge-IT v2.18.68
- User agent for Report-IT SIP connections: Tieline Report-IT EE (3.5.6_2894). Note: In this example "3.5.6" is the Report-IT version number and "2894" is the build number.
- In Tieline G3 codecs the user agent is configured as "Tieline <Product Name> <Serial Number>". E.g. Tieline TLR350 8972. The model numbers for Tieline G3 codecs are as follows:
    - o Commander G3 Rack TLR300 = Model Number TLR300
    - o Commander G3 Rack TLR300B = Model Number TLR350
    - o Commander G3 Field TLF300 = Model Number TLF300
    - o i-Mix G3 TLM600 = Model Number TLM600

## Troubleshooting SIP Connections

Manufacturers of professional IP codecs in most cases do not use the standard SIP ports (UDP 5060 and UDP 5004) for making IP connections. Therefore you will need to reconfigure each codec to use the standard SIP ports in most instances.

Most of the time EBU N/ACIP Tech 3326 SIP compliant codecs should connect when using the same encoders and connection settings at both ends. However, if one-way audio is encountered, it is highly likely to be a port forwarding issue. Port forwarding configuration instructions for most popular routers are available at www.portforward.com.

Some manufacturers support a subset of the EBU N/ACIP Tech 3326 recommendations so not all algorithms are supported by all manufacturers. G.722 is supported by most codec manufacturers, so if you encounter connection issues, default to using this algorithm for most troubleshooting scenarios.

## Using this Document

To dial over SIP peer-to-peer without using a SIP server follow the instructions in this document. The following sections explain:

1. How to configure a range of codecs from different vendors to connect with Tieline G6 codecs.
2. How to configuring Tieline G6 codecs for SIP using the Toolbox Web-GUI.

If you require more detailed information about configuring a Tieline codec for SIP, visit www.tieline.com/support and download the user manual for your product.

## Notes on Using Comrex Codecs

Common SIP settings Tieline recommends when connecting to non-Tieline codecs like Comrex are as follows:

- Profile: Mono
- Bit rate: 64kbps
- Algorithm: G.722
- Session port: UDP 5060
- Audio port: UDP 5004

Please note the following when connecting to Comrex codecs using SIP:

1. Some Comrex codecs do not support MPEG Layer 2, so G.722 is the preferred option for testing.
2. Opus encoding may be the default option in some Comrex codecs. Current Tieline firmware supports both CBR and VBR Opus encoding. In BRIC-Link II, and possibly other codecs, special CBR modes are offered for compatibility with Tieline products if firmware in a Tieline codec is old, i.e. prior to version 2.18.32.
3. Comrex codecs using SIP may expect a User Agent ID. Please see "How do I find my Codec's User Agent" in this document for more details on Tieline codec User Agent IDs.

# 1.1 Connecting to a Comrex Access Rack Codec

**Important Note:**
- Firmware installed; Comrex Access codec firmware version 4.0 / Flash 4.0; Tieline codec firmware version 2.14.88.
- To connect to the Comrex web-GUI you need to determine the IP address of the codec on startup. View the PPM LEDs on start up to discover the last 4 digits of the IP address; **L** (**Send**) is the 4th last digit and **R** (**Receive**) is the last digit.
- The Comrex Access does support MPEG 1 Layer 2 encoding.
- The Tieline codec should automatically select G.711 µ-law for North America and Japan, and G.711 a-law in most other regions of the world if this algorithm is used.
- The default credentials for the Comrex Access GUI is to use any username and enter the default password **comrex**

## Configure the Comrex Access Rack Codec for a Peer-to-Peer SIP Call

1. Attach a LAN cable and any audio inputs to the Comrex Access codec.
2. Apply power to the Comrex Access.
3. Determine the IP address of the codec (DHCP by default), by noting the number of "Receive" LEDs illuminated on the front panel of the Access codec.
4. Enter the IP address into a browser and open the Comrex Access Graphic User Interface (GUI).
5. Click the **System Settings** tab in the Comrex Access GUI.
6. Click to select the **Show advanced options** check-box below the **System Setting** pane.
7. Click the arrow symbol to the left of the **EBU 3326/SIP Settings** folder to expand and view all SIP settings.
8. Click **Accept incoming connections** and ensure the **Enabled** check-box is selected.
9. Click **IP Port** and ensure that port **5060** is entered in the port number text box; click **Apply** to change this setting after making changes.
10. Click **RTP IP Port** and ensure that port **5004** is entered in the port number box; click **Apply** to change this setting after making changes.
11. Click the arrow symbol for **Standard RTP Settings** to expand and view all settings.
12. Click **IP Port** and change the **Incoming network port** to **5004**, then click **Apply.**
13. Click the **Profiles tab** and click the **Add New Profile** button.
14. In the **Profile Setting** section select **General** and type a name into the **Profile Name** text box, then select **EBU 3326/SIP** in the **Channel** drop-down arrow selections.
15. In the **Profile Setting** section and select **Local**, then **Encoder** and click **VoIP** in the adjustment pane to select either **G.711** or **G.722**. Then select **Remote > Encoder > VoIP > G.722**. Note: Alternatively, select AAC, HE-AAC, AAC-LD or Opus encoding options.
16. You are now ready to connect, so click on the **Connections** tab in the Comrex Access GUI.
17. Click the **Store New Remote** button to create a new connection.
18. Enter the **Remote Name** and the **IP address** of the Tieline codec you are dialing. Then select the profile you want to use via the drop-down **Profile** list box, next click **OK**.
19. Click the large green **Connect** button.
20. Click the **Disconnect** button to hangup the connection.

# 1.2 Connecting to a Comrex Access Portable

**Important Note:**
- Firmware installed - Comrex Access Portable codec firmware version 2.8-pre.
- The Tieline codec should automatically select G.711 µ-law for North America and Japan, and G.711 a-law in most other regions of the world if this algorithm is used.

## Configure the Access Portable for a Peer-to-Peer SIP Call

1. Attach a LAN cable and any audio inputs to the Comrex Access codec.
2. Apply power to the Comrex Access Portable.

3. Tap the **Configure** drop-down menu in the top-left of the codec touch-screen and then tap **System Settings**.
4. Tap the **Advanced** check box at the bottom of the screen to display the full menu.
5. Scroll down the screen to **N/ACIP SIP Settings** and tap to select **Accept Incoming Connections** if it is not enabled, then tap **Edit**, next tap the **Enabled** check box and tap the **Save** button.
6. Ensure that port **5060** is entered into the port number box.
7. Tap **RTP IP Port** and enter port **5004** into the port number box.
8. Tap the **Network** drop-down menu at the top of the screen, then tap **Manage Networks** to determine the IP address (DHCP by default) of the codec as you will need this to dial into the codec from a Tieline codec.
9. Tap the **Configure** drop-down menu in the top-left of the codec touch-screen and then tap **Manage Profiles**.
10. Tap the **Add New** button to create a **New Profile** in the **Available Profiles** list.
11. Tap **New Profile** to highlight it and tap the **Edit** button.
12. Tap **Profile Name** to highlight it on the screen and tap the **Edit** button, then type a name for the profile and tap the **Save** button.
13. Tap **Channel** to highlight it on the screen and tap the **Edit** button, then tap to select **N/ACIP SIP** in the drop-down list box and tap the **Save** button.
14. Tap **Encoder** to highlight it on the screen and tap the **Edit** button, then tap **X3: VoIP G.722** in the drop-down list box and tap the **Save** button.
15. Tap the **Done** button in the **Profile Settings** screen.
16. To create a new "Remote" connection tap **Remotes**, then tap **Add New Remote**.
17. Enter the **Name** of the connection and the IP address, then tap to select the profile you have just created in the **Profile** drop-down list box, next tap the **OK** button.
18. Tap on the Remote you have just programmed with the new profile and tap the **Connect** button on the screen to connect to the Tieline codec.
19. Tap the **Disconnect** button to hangup the connection.

## 1.3 Connecting to a Mayah Sporty

**Important Notes:**
- Mayah Sporty codec firmware version 4.9.1.0
- The Tieline codec should automatically select G.711 µ-law for North America and Japan, and G.711 a-law in most other regions of the world if this algorithm is used.

### Configure a Mayah Sporty for a Peer-to-Peer SIP Call

1. Attach a LAN cable and any audio inputs to the Sporty codec.
2. Apply power to the Mayah Sporty.
3. Find the IP address of the codec via **F2 [Codec1] > F3 [Setup] > F1 [Interface] > Ethernet > OK**. Use the navigation buttons to find the IP address as you will need this to dial into the codec from the Tieline codec.
4. Use the up/down navigation arrow buttons and select **SIP**.
5. Navigate down and ensure port **5060** is entered for the UDP port, then ensure port **5004** is entered for the TCP port. Select **UDP** or **TCP** and then **OK** to edit these settings.
6. Press the **F4/ON** button several times to return to the **Home screen**.
7. Press **F2 [Codec] > F3 [Setup] > F2 [Quality]** to display the algorithm selection screen.
8. Use the navigation arrow buttons on the top of the codec to select an algorithm from G.711µ, G.711a, G.722, MPEG L2 or Linear 16 bit.
9. Press the **F4/ON** button several times to return to the **Home screen**.
10. Ensure the codec is configured to automatically accept incoming calls via **F4/ON System > F2 (Misc.) > Connections > Accept Mode > Auto**.
11. Press the **F4/ON** button several times to return to the **Home screen**.

12. Select **F2 [Codec1] > (F1) Connect > (F3) Direct > Interface (Ethernet) > Protocol (SIP) > SIP Address (enter IP address) > OK**.
13. Press the right navigation button to select **dial** and then press **OK** to dial. The screen should display **Connecting** and then connect to the destination codec.
14. To hang up from the **Home screen** select **F2 [Codec1] > F1 (End Call)**.

## 1.4 Connecting to a Telos Zephyr IP

**Important Notes:**
- Zephyr IP codec firmware version 3.1;
- The default user name "user" and password "Telos" can be used to open the Telos web-GUI interface.
- In testing Tieline was successful in connecting using MPEG Layer 2 and G.722.

### Configure the Zephyr IP for a Peer-to-Peer SIP Call

1. Apply power to the codec and when the menu appears use the arrow buttons to navigate to **Network** on the LCD screen and press **OK**.
2. Verify and note the IP address displayed via **Ethernet Config**.
3. Press the **ESC** button and return to the **Main Menu Screen**.
4. Select **Codec > Advanced Setup > Encoding Mode [Layer 2] / Minimum Bitrate [128kbps] / Maximum Bitrate [128kbps]**. Note: G.722 has also been successfully tested.
5. To add a new sip address navigate to **Auto > Add > Device Name [sip1@<enter IP address here>] >Device Type [SIP] > Save**. Note: the @ symbol is accessed via the "1" button.
6. Navigate back to the **Contacts** screen, select the contact and then select **Call** and press **OK** to dial.

**Important Notes:** The address used to dial the Zephyr from the Tieline codec over SIP was **ZEPHYR@<insert IP address here>**

## 1.5 Connecting to an APT Worldcast Equinox

**Important Notes:**
- Equinox codec firmware version 3.1;
- In testing Tieline was successful in connecting using MPEG 1 Layer 2 encoding only.

### Configure the Equinox for a Peer-to-Peer SIP Call

1. Plug your Ethernet LAN cable into the back of the codec and attach power.
2. Ensure the correct IP address is configured in the Equinox via **Main Menu > IP > Stream Port Settings**.
3. Return to the **Main Menu** and select **Audio**.
4. Next select **Audio Profile (No)** and then **MPEG - L2** as the algorithm. Select the appropriate bit rate and whether you want to dial in mono, stereo or Joint Stereo, and then the sample rate. For the profile we selected **CCS IMUX**.
5. Return to the **Main Menu** and select **User** and navigate to **Primary Conn.**, then press the **Ent Dial** button.
6. Navigate to **Codec - SIP > Master** and press **Ent Dial**.
7. Return to the **Main Menu** and select the **SIP** menu and press the **Ent Dial** button.
8. Select **Setup Address Book** and press the **Ent Dial** button.
9. Select entry entry **0** or **1** in the address book and press the **Ent Dial** button to enter the address. Note: if you use entry **0** or **1** you can use the **FD0** and **FD1** buttons on the front of the codec as speed dial buttons for these two entries.
10. Configure the address as **SIP:TIELINE@<insert codec IP address>**.
11. The codec should now be ready to dial or answer.
12. Press either the **FD0** or **FD1** button to dial one of these entries.

**Important Notes:**

- The default Local SIP address in the Equinox you need to dial is **sip:default_local_sip_user@yourdomain.com .** As an example: **default_local_sip_user@203.36.205.174**
- You can adjust the local SIP address setting in the Equinox via **Main Menu Screen > SIP > Local User Address**

# 1.6 Connecting to an Prodys Prontonet LC

**Important Notes:**

- Prodys Prontonet codec firmware version 06.0.1
- In Tieline testing the codecs were successful in connecting over MPEG Layer 2 in both directions.
- Please configure the Prontonet LC with a 20ms frame rate when attempting to connect over G.711 or G.722 with Tieline codecs. This setting can be configured using Prodys web-browser configuration software.
- The Tieline codec should automatically select G.711 µ-law for North America and Japan, and G.711 a-law in most other regions of the world if this algorithm is used.

## Configure a Prodys Prontonet LC for a Peer-to-Peer SIP Call

1. Attach a LAN cable and any audio inputs to the Prontonet codec.
2. Apply power to the codec.
3. To find the IP address in the codec press the **INF** button on the codec until you can see the IP address listed adjacent to ADDR:
4. Press the OK button to exit the **INF** menu screen.
5. Press **OK** again to view the **MAIN MENU**.
6. Use the navigation buttons to select **NET** and press **OK**.
7. In the **NET SELECTION** screen select **IP** and press **OK**.
8. In the **SET CODEC** screen select **SIMPLE** for a single connection, then press **OK**.
9. In the **SET IP NET** screen select **OK** when **Unicast/Multicast** is displayed.
10. In the **STREAMING PROTOCOL** screen navigate to **SIP** and press **OK**.
11. Press arrow button above the **OK** button to return to the **MAIN MENU**.
12. Navigate to **ENC** and press **OK**.
13. Press **OK** in the **SET ENCODING MODE (ENCODER1)** screen.
14. In the **SET ENCODER 1** screen navigate to your preferred encoding algorithm and press **OK**.
15. Configure your preferred bit rate in the **MPEG SET BITRATE** screen, then press **OK**.
16. Select Mono, J-Stereo or Stereo in the **MPEG SET MODE** screen, then press OK.
17. Select your preferred sample rate in the **MPEG SET Fs** screen, then press **OK**.
18. In the **MPEG SET CRC** menu select **OFF** and press **OK**.
19. In the **MPEG SET AUX DATA** menu select **OFF** and press **OK**.
20. Press the **CALL 1** button and select **Unicast**, then press **OK**.
21. Select **BIDIR** for a bidirectional audio path over the connection, then press **OK**.
22. In the **LAN L1 DIAL** screen use the keypad buttons to enter the IP address of the destination codec in the **LAN L1 DIAL** screen, then press **OK** to dial.
23. On the LCD screen of the Prontonet it should display **L1->CONNECTED** and **FRAMED** is displayed in the bottom left corner of the screen. The **GREEN LED** adjacent to the **CALL1** button is also illuminated when connected.
24. Press the **CALL 1** button for approximately 2-3 seconds to hangup the call.

# 2 Configure Tieline SIP Interfaces and Accounts

Tieline G6 codecs support dialing over SIP using a registered SIP server account, or peer-to-peer using an available SIP interface, e.g. **SIP1** or **SIP 2**. To configure a peer-to-peer SIP connection see Configure Peer-to-Peer SIP Programs.

To dial over SIP using a SIP Server you will first need to:

1. Register the codec to a SIP server using SIP account credentials.
2. Configure a SIP interface in the codec. Note: The **SIP1** or **SIP2** interface will include the proxy and port settings, as well as the selected IP interface used to make the connection, e.g. **LAN1**.
3. Create a SIP program using the HTML5 Toolbox Web-GUI.

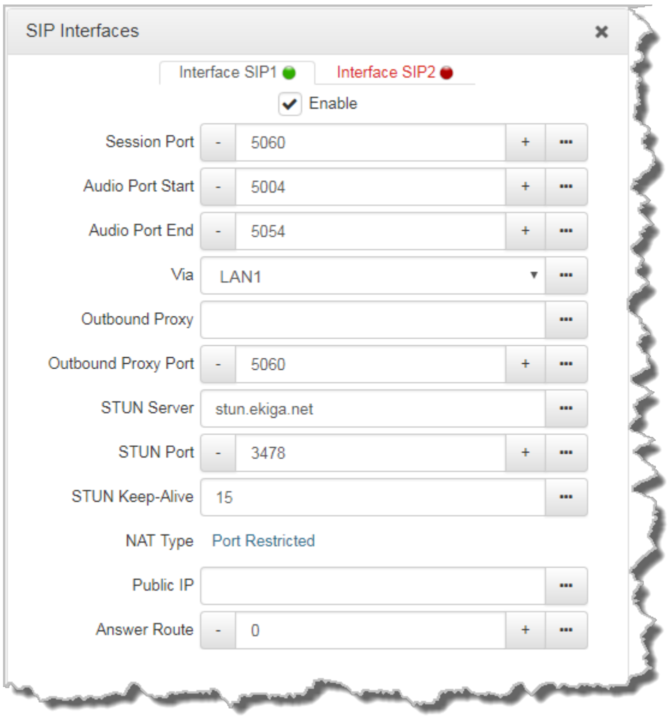
**Important Notes:**
- SIP dialing is only supported over point-to-point connections, not multi-unicast connections.
- Failover and SmartStream PLUS redundant streaming is not available with SIP.
- Tieline supports RFC5109 and RFC2733 compliant FEC over SIP from firmware v2.18.xx.
- Tieline G5 and G6 codecs support a SIP call being placed on-hold. Note: there are several different implementations of "on-hold" by various SIP providers. Some will stop streaming when a call is placed on-hold and others will replace live streaming with on-hold messages or music.
- Some ISPs and/or cellular networks may block SIP traffic over UDP port 5060.
- By default the Tieline codec will attempt to connect using MP2 and then G.722

## 2.1 Configuring SIP Interfaces

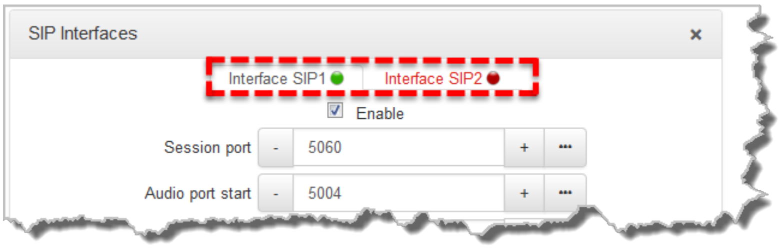
**Important Notes:**
1. SIP interfaces are disabled by default.
2. **SIP1** is configured to use **LAN1** by default, which is mapped to the **Primary** Via interface by default.
3. **SIP2** is configured to use **LAN2** by default, which is mapped to the **Secondary** Via interface by default.
4. **SIP1** and **SIP2** each need to use a separate IP interface when connecting, e.g. **LAN1** or **LAN2**.
5. **SIP1** and **SIP2** can each make multiple SIP calls, e.g. two calls can be made over **SIP1**, or two calls can be made over **SIP2**.
6. The settings for **SIP1** and **SIP2** cannot be edited if the interface has been enabled.
7. Enter a public IP address in the **Public IP** text box if you want to dial over SIP from behind a firewall. Then configure port forwarding to route traffic to the codec's local IP address behind your firewall. Note: Do not enter a **Public IP** address if STUN is configured. They cannot be used together because both will attempt to use a public IP address over SIP. STUN settings are prioritized and used if both are configured.

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

1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then click **SIP Interfaces** to view and configure SIP interface settings.
2. Default SIP settings are configured and select **Interface SIP1** or **Interface SIP2** to adjust each interface. Note: Ensure each interface uses a unique "Via" IP interface because they can't share one, e.g. **LAN1**.




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




## 2.2 Configuring SIP Accounts

Up to 16 SIP accounts can be configured in the codec and registering codecs for SIP connectivity is simple. First, select the SIP server to which you will register your codec. On a LAN this may be your own server, or it could be one of the many internet servers available. When you register an account with a SIP server you will be provided with:

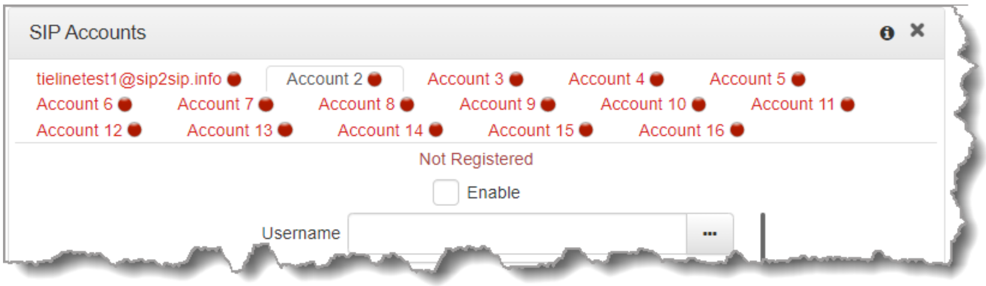
- Username
- Authorized User
- SIP address
- Domain
- Realm
- Registrar
- Registar port
- Outbound Proxy
- Proxy port
- Password
- Registration Timeout (this shouldn't need to be adjusted from the default setting).

**Important Notes:**
- In most situations it is best to configure a SIP account when the codec is configured with a public IP address.
- Each SIP account can only be mapped to a single SIP interface, i.e. **SIP1** or **SIP 2**.
- Up to 16 SIP accounts can be added to the codec.

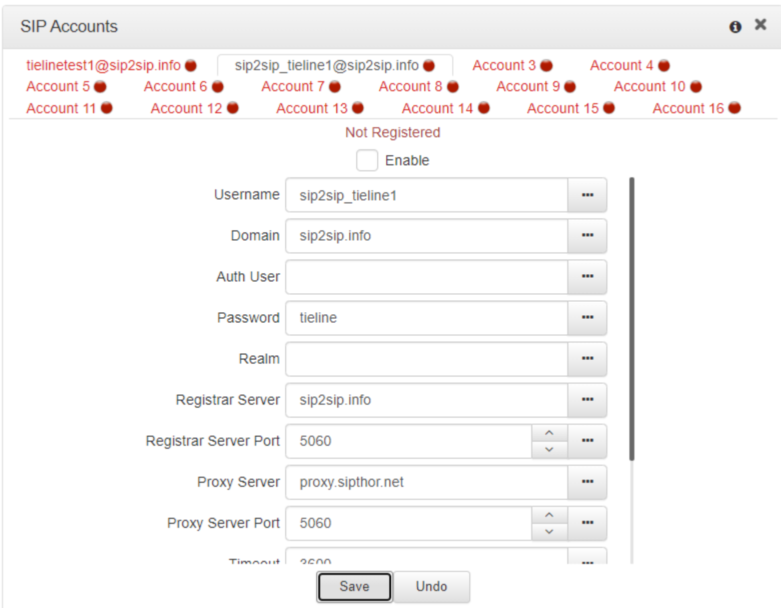
## Adding a SIP Account

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

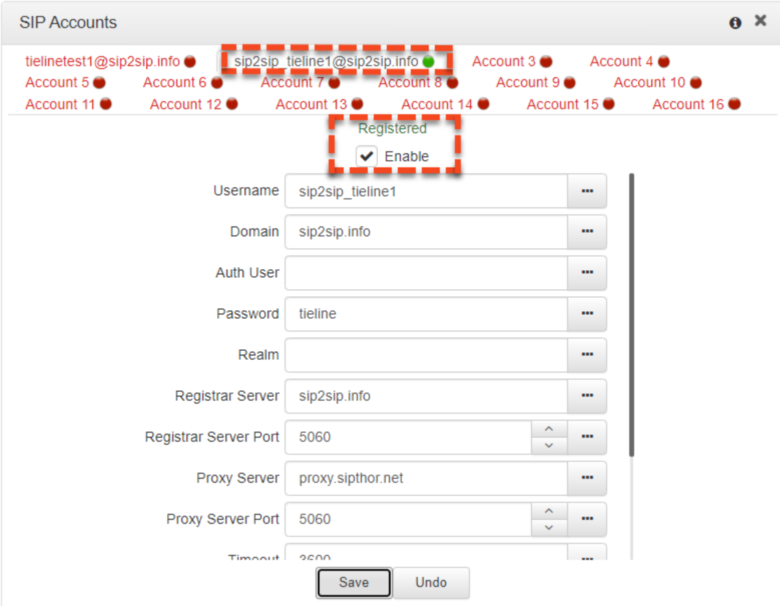
1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then click **SIP Accounts** to view and configure SIP account settings.
2. Click to select one of the unused **Accounts** at the top of the **SIP Accounts panel**.




3. Enter the SIP account details into the relevant text boxes, including the registration **Timeout** (which shouldn't need to be adjusted from the default setting). Also ensure a SIP **Interface** is selected (e.g. **SIP1** or **SIP2**.) The SIP interface contains settings related to ports and the selected **Via** interface, e.g. **LAN1** or **LAN2**.




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




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


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

**Important Notes:**
- Some ISPs may block SIP traffic over UDP port 5060.
- By default, the session port used for each SIP account is the associated SIP interface session port. The default session port is the registered UDP port number 5060. It is also possible to configure a custom local session port for each SIP account for compatibility with Cisco Unified Communications Manager (CUCM). Ensure your firewall has the required TCP and UDP ports open when receiving multiple SIP calls.


## Troubleshooting SIP Registration

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

1. Confirm all account registration information has been entered correctly.
2. Confirm the SIP interface (**SIP1** or **SIP2**) configured as the **Via** in the account is enabled.
3. Verify that the **Via** selection in the **SIP1** or **SIP2** interface settings corresponds with the network interface being used by the codec to register the account. E.g. **LAN1**, **LAN2**.

## 2.3 Configure SIP Allow and Block Lists

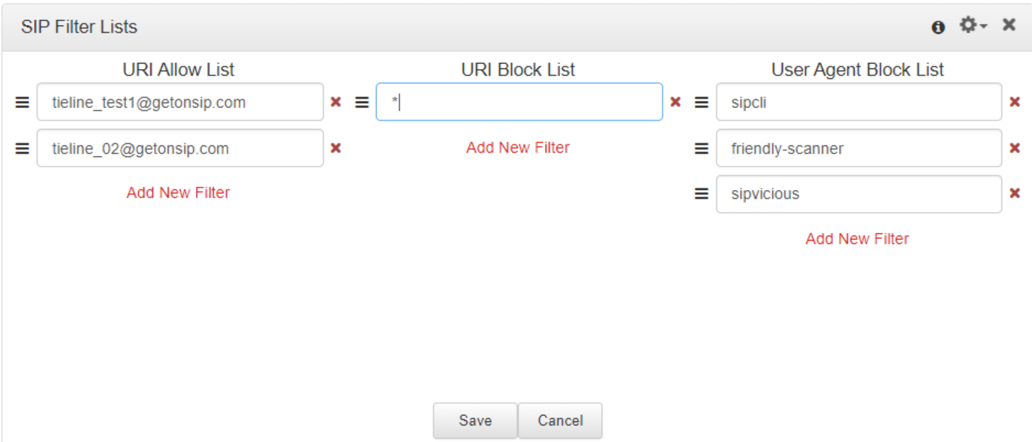
The **SIP Filter Lists panel** allows filtering of SIP URIs and User Agents to provide greater security for connections. For example, add trusted network codecs to the **URI Allow List** in this panel and only codecs using these SIP URIs will be able to connect. This is like saying, "if you have the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

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

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

### Filter URIs and User Agents

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




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

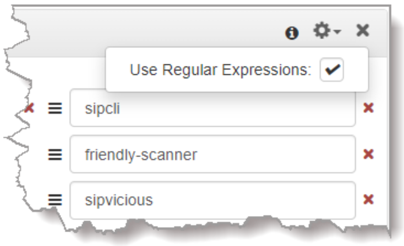
**Important Note:** Some codec manufacturers allow calls based on 'User Agent' identification. It may be necessary to enter a Tieline codec user agent into a non-Tieline codec to connect to a Tieline SIP-enabled codec.

- From firmware v2.16.xx the user agent in the codec is configured as "Tieline <Product Name> <Firmware Version>". E.g. Tieline Bridge-IT v2.18.68
- User agent for Report-IT SIP connections: Tieline Report-IT EE (3.5.6_2894). Note: In this example "3.5.6" is the Report-IT version number and "2894" is the build number.
- In Tieline G3 codecs the user agent is configured as "Tieline <Product Name> <Serial Number>". E.g. Tieline TLR350 8972. The model numbers for Tieline G3 codecs are as follows:

- o Commander G3 Rack TLR300 = Model Number TLR300
- o Commander G3 Rack TLR300B = Model Number TLR350
- o Commander G3 Field TLF300 = Model Number TLF300
- o i-Mix G3 TLM600 = Model Number TLM600

## Using Regular Expressions

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




**Important Note:** Regular expressions should not use ^ and $ anchors because searches implicitly try to match anywhere in the line.

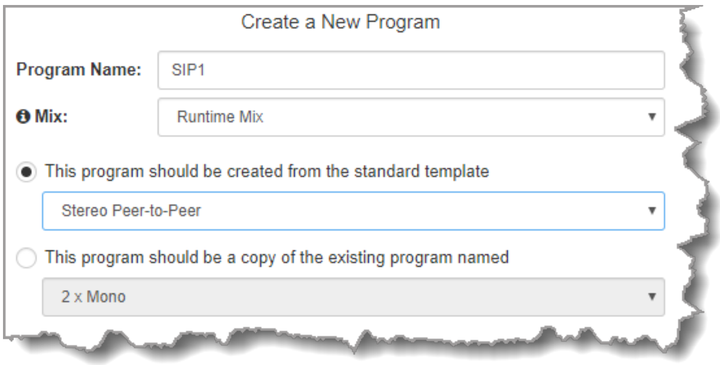
# 3 Configure Peer-to-Peer SIP Programs

The codec supports dialing over SIP using a registered SIP server account, or peer-to-peer using one of the two SIP interfaces **SIP1** and **SIP 2**. It is also necessary to select SIP as the **Session Protocol**.

To configure a SIP multiple stream program simply create a new program and configure each SIP audio stream like a single SIP Peer-to-Peer program. The codec is capable of registering up to 16 SIP accounts, each of which has an associated **Answer Route** field, which can be matched to a loaded answering program's audio stream Answer Route. Without using SIP accounts, each SIP interface also has an **Answer Route** field. However, only 2 SIP interfaces are supported, limiting this method of routing configuration to a maximum of 2 audio streams. Note: An account's Answer Route setting is applied first.

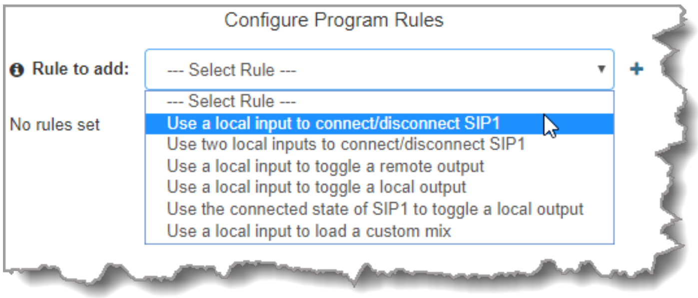
**Important Notes:** Before commencing program configuration please note:
- You cannot edit a program when it is currently loaded in the codec.
- Some drop-down menus and settings may be greyed out intentionally depending on features available.
- Failover and SmartStream PLUS redundant streaming is not available when connecting using SIP.
- Lock a loaded custom program or multistream program in a codec to ensure it cannot be unloaded by a codec dialing in with a different type of program. For example, if a multistream program is not locked it will be unloaded by a mono or stereo call.
- Ensure the appropriate TCP and UDP audio ports are open in your firewall to allow SIP audio streams to connect.

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




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

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

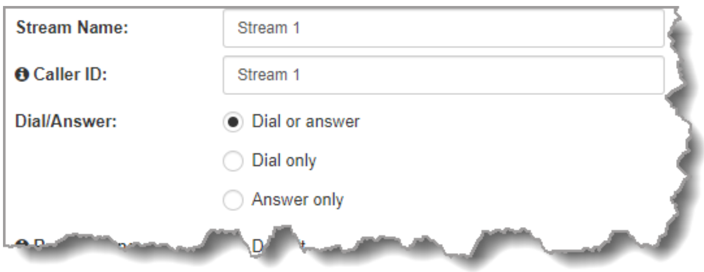



**Important Notes for Rules:**

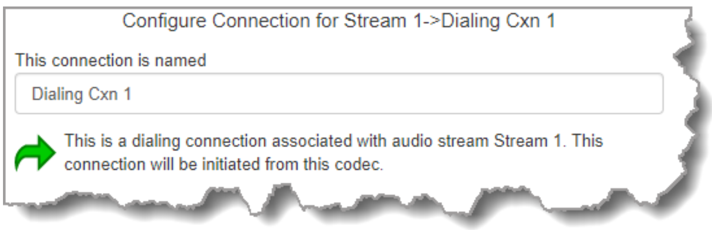
- The Gateway 4 codec has 8 hardware GPIOs and 56 logical outputs, and the Gateway 8/16 has 16 hardware GPIOs and 48 logical outputs; both codecs also have 3 virtual inputs, and 64 WheatNet Logic Inputs/Outputs. (WheatNet logic I/Os allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network. WheatNet logical inputs are only available if a codec has a WheatNet-IP card installed).
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- Tieline WheatNet-IP codecs require Wheatstone Razor firmware version 1.4.22 or later to support WheatNet LIOs. In addition, the WheatNet-IP codec must have the **WNet Enable LIO** checkbox selected in the **Options panel** of the HTML5 Toolbox Web-GUI.
- Relay reflection is not available for SIP and Multicast Client programs.
- For more details about rules see download the product user manual at www.tieline.com/support.

4. Enter the **Stream Name** and configure the codec to dial, answer or dial and answer. Then click **Next**.

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



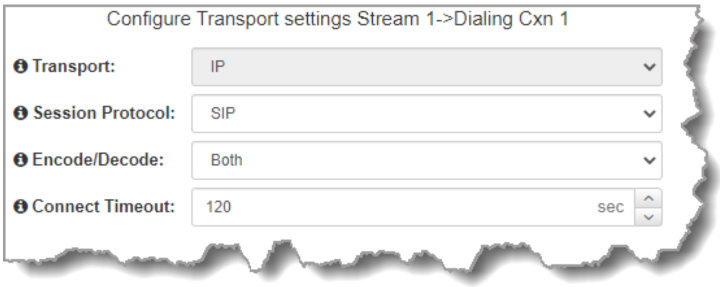

5. This audio stream connection in the wizard will allow the codec to dial. Enter the name of the connection in the text box, then click **Next**.




6. Configure the transport settings for the connection. Ensure that you select:

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

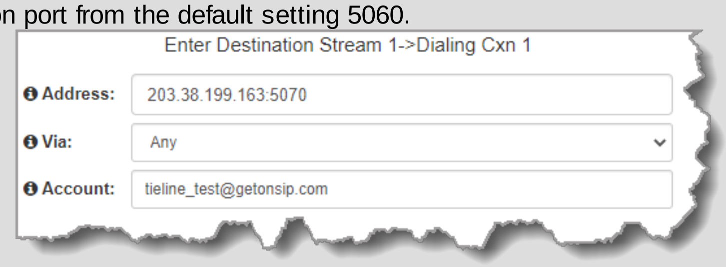
Then click **Next**.



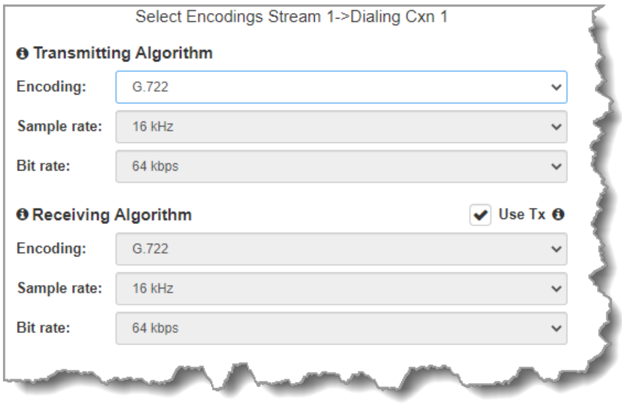

7. Configure the destination codec **Address** if you are dialing peer-to-peer, then specify the network interface used to dial the connection, e.g. **Primary** (Ethernet port 1). Enter the name of a registered SIP account if you are using a SIP server to establish a connection. If you wish to dial from one of the codec's registered accounts, then enter the account name in the Account field using the format accountname@sipserverdomain, e.g. tieline_test@getonsip.com. In this configuration the account interface will be used rather than the specified **Via**, e.g. if the account is using **SIP2** and this is configured to use LAN2, then the call will proceed using LAN2. If you do not wish to use an account for the dial then leave the **Account** field blank and select the required interface. Note: the interface must be associated with either **SIP1** or **SIP2** for the call to be able to proceed.


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

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

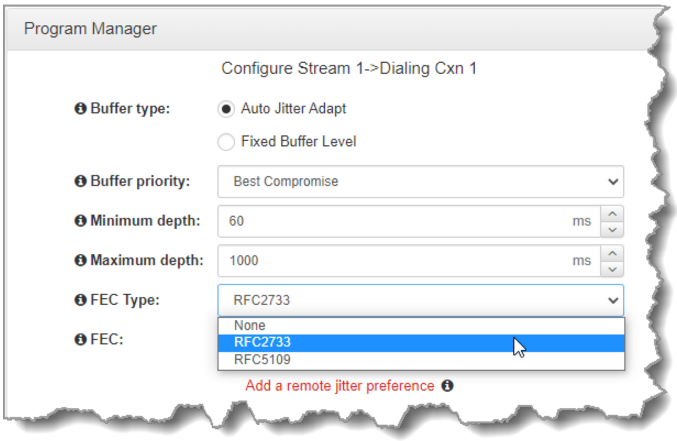



8. Click the drop-down arrows on the right-hand side of each active drop-down menu to adjust the **Encoding, Sample rate** or **Bit rate** parameters. Click **Next** to continue.



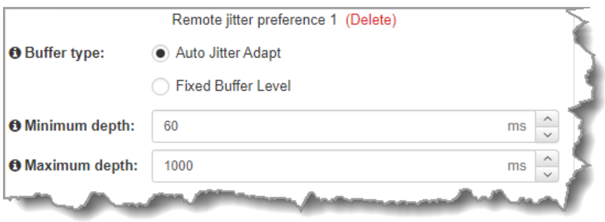

9. Click to configure:

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

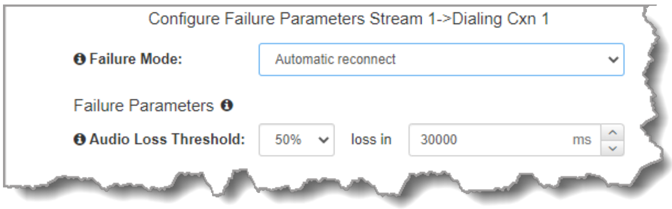



10. Click **Add a remote jitter preference** to send preferred jitter settings to a remote codec. Note: this is just a preference as per EBU Tech 3368 and there is no guarantee that the remote codec will accept or support these jitter configuration settings. Verify configuration settings on the remote codec to ensure settings are correct. Recommended jitter buffer limits are as follows:

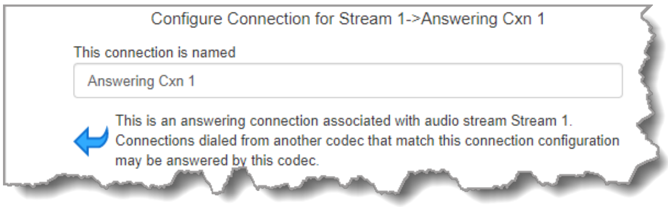
- 1,000ms for PCM and G.711, G.722 and aptX Enhanced encoding.
- 2,500ms for AAC ELD, AAC LD.
- 5,000 for all other algorithms including Opus, MP2, AAC, AAC-HE, Tieline Music and Music PLUS.



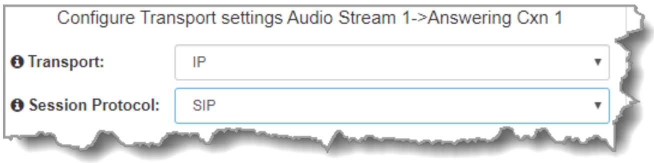

11. Click **Next** to select the check-box if you want to **Enable Auto Reconnect**.



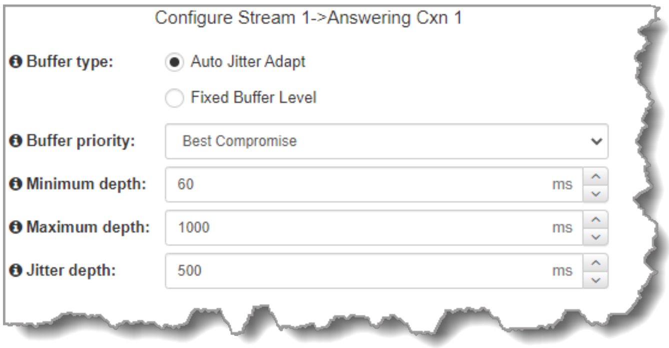

12. Click **Next** to name the answering connection for when calls are received by the codec.



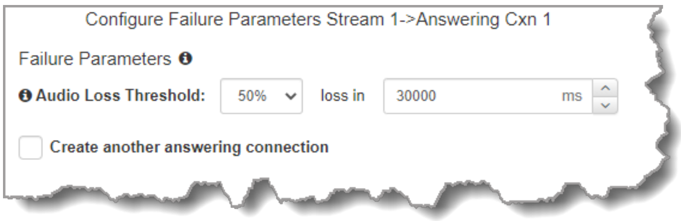

13. Click **Next** to configure the **Session Protocol** as **SIP** for the answering connection to receive a SIP call.




14. Click **Next** to configure the jitter buffer settings for the answering connection. Note: it is also possible to configure remote jitter preferences if the remote codec supports RFC5109.




15. Click **Next** to configure **Failure Parameters** for the answering connection if required. **Please note**: In most situations the default answering **Failure Parameters** do not need adjustment. These settings may be useful to troubleshoot certain connections, e.g. satellite IP links.




16. After configuring all settings there are 3 options:

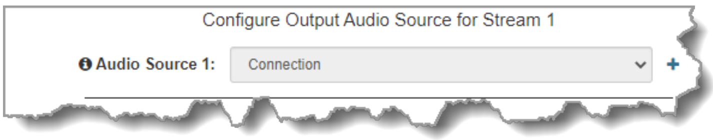
    i. If you want to create another answering connection, select the check-box for **Create another answering connection** and continue through the wizard.

ii. Click **Save** to save the program at this point.
iii. Click **Next** to configure **Output Audio Source** options.

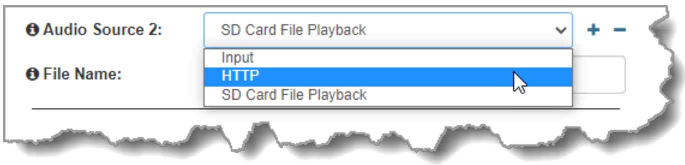
## Configuring Output Audio Source Options

1. Click **Next** to configure **Output Audio Source** options and automatically switch between up to 4 backup audio sources to maintain program
 audio at transmitter sites. **Output Audio Source** options include:

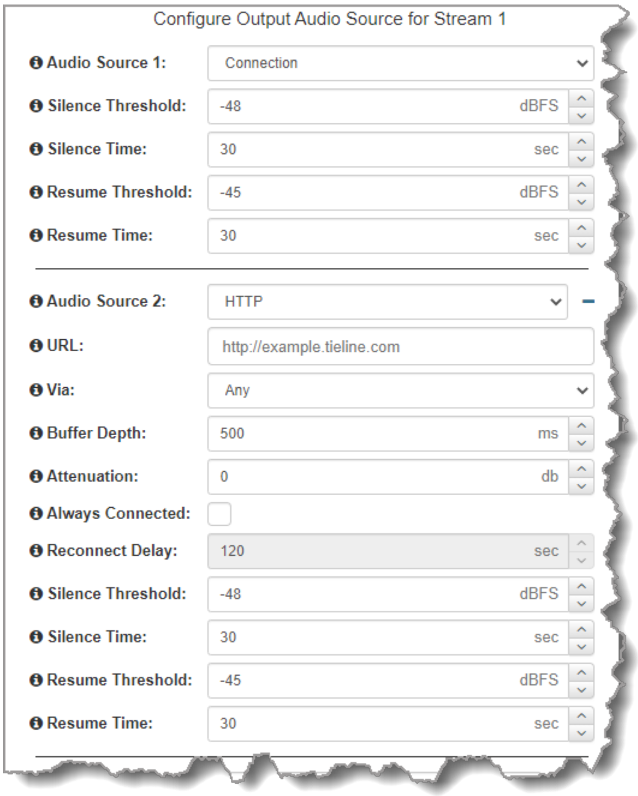
   - Connection: Decoded connection audio sent from a remote codec (Note: this must be selected as one of the configured sources).
   - **Input**: Input audio looped to the physical codec outputs.
   - **HTTP**: Icecast client mode to allow media server streaming from a specified URL.
   - **USB File Playback**: Audio file playback from a USB stick.




2. Click the blue **Plus symbol** to add a backup **Output Audio Source**, or click the **Minus symbol** to remove an **Output Audio Source**. Click the drop-down arrow to select an **Output Audio Source** option.




3. Configure silence threshold parameters for enabling a preferred backup option, as well as resume thresholds for reactivating a previous source. Then click **Save Program** to save program settings.

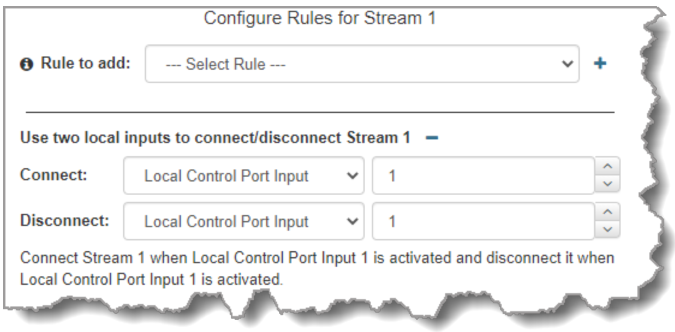



4. After configuring **Output Audio Source** options you can:

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

## Configuring Rules

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




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

2. Click **Save** to save the program.
3. Click **Finish** to exit the wizard.
4. The newly created program can be loaded from within the **Program Manager panel**, **Connections panel** and the **Program Loader panel** (in the Quick Connect web-GUI). Select and connect audio streams in a program using the **Connections panel**, or connect the program manually using the front panel.