Tieline G5 Codec Compatibility over IP and ISDN

Manual Version: 1.2
May, 2019

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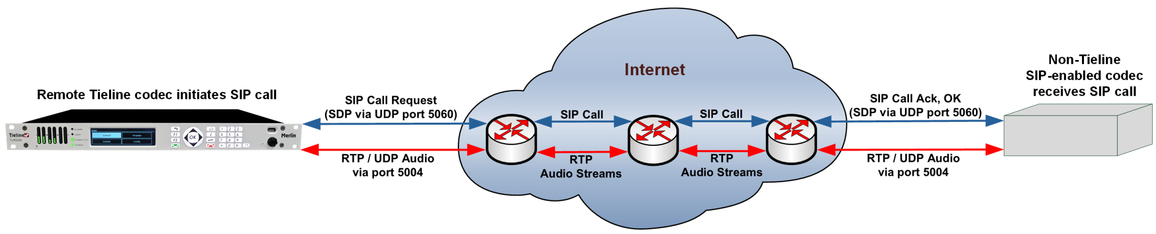
1 Connecting Tieline IP to other Codecs Using SIP

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.



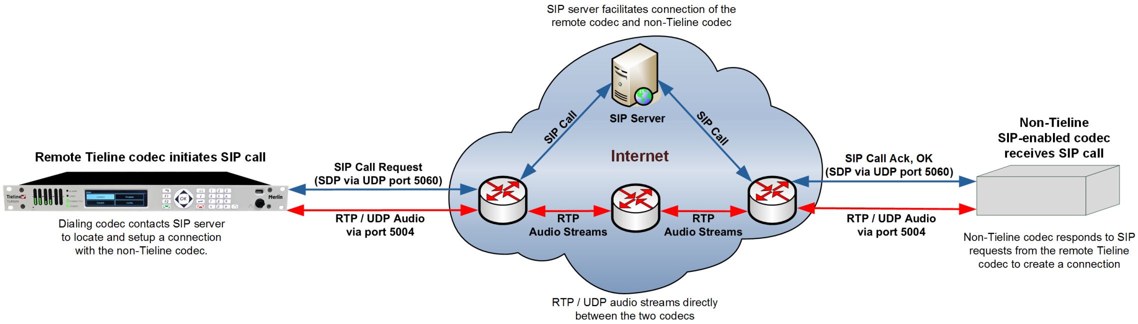
There are two very distinct parts to a call when dialing over IP. The initial stage is the call setup stage and this is what SIP and SDP is used for. The second stage is when data transfer occurs and this is left to the other protocols such as RTP/UDP to stream audio data. SDP works with a number of 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. SIP requires both codecs to be configured with the same settings before connecting or the connection will fail. 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. SIP usually uses standard ports; UDP Port 5060 for session data (SDP) and UDP Port 5004 for audio data.
3. Ports 80 and 8080 are commonly used to access the Tieline codec web server. You can add an additional layer of security by translating these ports on the WAN side of your network into non-standard port numbers. Adjust ports using the **Options panel** in the Toolbox HTML5 Web-GUI, or see "Configuring TCP/UDP Ports".
4. By default SIP interfaces are disabled to avoid unwanted traffic. The **SIP Filter Lists panel** in the Toolbox HTML5 Web-GUI allows filtering of SIP URIs and User Agents to provide greater security when using SIP. See "Configure SIP White and Blacklists" for more information.
5. An SSL security certificate can be installed on each codec in your network to ensure it is a trusted device within your network. See "Installing a Security Certificate" for more information.
6. Firewall settings to enable or disable a range of firewall-related network services, or limit ping to only work in a local subnet. Adjust settings using the **Options panel** in the Toolbox HTML5 Web-GUI, or see "Firewall Configuration."
7. Implementation of CSRF protection (Cross-Site Request Forgery). Enable and disable this setting using the **Options panel** in the Toolbox HTML5 Web-GU, or see "Enabling CSRF Security" for more info.

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

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 whitelist or blacklist 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 discover a codec user agent as some codec manufacturers allow whitelisting of calls by 'User Agent.' 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 and later releases the user agent in a Tieline G5 codec is configured as "Tieline <ProductName> <Firmware Version>". E.g. Tieline Bridge-IT v2.18.68.

- In Tieline G3 codecs the user agent is configured as "Tieline <ProductName> <Serial Number>". E.g. Tieline TLR350 8972. The model numbers for Tieline G3 codecs are as follows:

    - Commander G3 Rack TLR300 = Model Number TLR300
    - Commander G3 Rack TLR300B = Model Number TLR350
    - Commander G3 Field TLF300 = Model Number TLF300
    - 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.

Getting Started with SIP

To dial over SIP peer-to-peer without using a SIP server follow the instructions in this document. To dial over SIP using a SIP Server you will need to:

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

Important Notes:
- SIP dialing is only supported over point-to-point connections, not multi-unicast connections.
- Tieline G5 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.

- Tieline supports RFC5109 and RFC2733 compliant FEC over SIP from firmware v2.18.xx.
- Some ISPs and/or cellular networks may block SIP traffic over UDP port 5060.

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 default credentials for the Comrex Access GUI is to use any username and enter the default password **comrex**

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

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.

Configure a Tieline G5 Codec for a peer-to-peer SIP Call

1. Attach a LAN cable and any audio connections to the Tieline codec.
2. To dial peer-to-peer press the **HOME** button to return to the **Home** screen, select **Connect > IP > SIP**.
3. Use the numeric **KEYPAD** to enter the IP address of the codec you want to dial, using the or buttons to enter the periods in the IP address and use the **RETURN** button to delete numbers already entered.
4. Then press the down navigation button to select **Setup** and press to adjust the algorithm to match the Access rack codec. Also adjust the jitter buffer and encode/decode direction if required. Note: By default the Tieline codec will attempt to connect using MP2 and then G.722.
5. Press the **RETURN** button to navigate backwards to the **Connect SIP** screen.
6. Press the **CONNECT** button to make a connection.

Important Note: 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.

1.2 Connecting to a Comrex Access Portable

Important Note: Firmware installed - Comrex Access Portable codec firmware version 2.8-pre; Tieline codec firmware version 2.14.100

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.

Configure a Tieline G5 Codec to Connect peer-to-peer using SIP

1. Attach a LAN cable and any audio connections to the Tieline codec.
2. To dial peer-to-peer press the **HOME** button to return to the **Home** screen, select **Connect > IP > SIP**.
3. Use the numeric **KEYPAD** to enter the IP address of the codec you want to dial, using the * or # buttons to enter the periods in the IP address and use the **RETURN** button to delete numbers already entered.
4. Then press the down navigation button to select **Setup** and press OK to adjust the algorithm to match the Access portable codec. Also adjust the jitter buffer and encode/decode direction if required. Note: By default the Tieline codec will attempt to connect using MP2 and then G.722.
5. Press the **RETURN** button to navigate backwards to the **Connect SIP** screen.
6. Press the **CONNECT** button to make a connection.

Important Note: 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.

1.3 Connecting to a Mayah Sporty

Important Note: Mayah Sporty codec firmware version 4.9.1.0; Tieline codec firmware version 2.14.88.

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)**.

Configuring a Tieline G5 Codec for a Peer-to-Peer SIP Call

1. Attach a LAN cable and any audio connections to the Tieline codec.
2. To dial peer-to-peer press the **HOME** button to return to the **Home** screen, select **Connect > IP > SIP**.

3. Use the numeric **KEYPAD** to enter the IP address of the codec you want to dial, using the or buttons to enter the periods in the IP address and use the **RETURN** button to delete numbers already entered.
4. Then press the down navigation button to select **Setup** and press to adjust the algorithm (usually MP2 or G.722), jitter buffer and encode/decode direction if required. Note: By default the Tieline codec will attempt to connect using MP2 and then G.722
5. Press the **RETURN** button to navigate backwards to the **Connect SIP** screen.
6. Press the **CONNECT** button to make a connection.

Important Note: 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.

1.4 Connecting to a Telos Zephyr IP

Important Notes:
- Zephyr IP codec firmware version 3.1; Tieline codec firmware version 2.14.88.
- 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.

Configuring a Tieline G5 Codec for a Peer-to-Peer SIP Call

1. Attach a LAN cable and any audio connections to the Tieline codec.
2. To dial peer-to-peer press the **HOME** button to return to the **Home** screen, select **Connect > IP > SIP**.
3. Use the numeric **KEYPAD** to enter the IP address of the codec you want to dial, using the or buttons to enter the periods in the IP address and use the **RETURN** button to delete numbers already entered.
4. Then press the down navigation button to select **Setup** and press to adjust the algorithm to **MP2** and the jitter buffer and encode/decode direction if required. Note: By default the Tieline codec will attempt to connect using MP2 and then G.722.
5. Press the **RETURN** button to navigate backwards to the **Connect SIP** screen.
6. Press the **CONNECT** button to make a connection.

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; Tieline codec firmware version 2.14.88.
- 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.

Configuring a Tieline G5 Codec for a Peer-to-Peer SIP Call

1. Attach a LAN cable and any audio connections to the Tieline codec.
2. To dial peer-to-peer press the **HOME** button to return to the **Home** screen, select **Connect > IP > SIP**.
3. Use the numeric **KEYPAD** to enter the IP address of the codec you want to dial, using the * or # buttons to enter the periods in the IP address and use the **RETURN** button to delete numbers already entered.
4. Then press the down navigation button to select **Setup** and press OK to adjust the algorithm to **MP2** and the jitter buffer and encode/decode direction if required. Note: By default the Tieline codec will attempt to connect using MP2 and then G.722.
5. Press the **RETURN** button to navigate backwards to the **Connect SIP** screen.
6. Press the **CONNECT** button to make a connection.

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 Note: Prodys Prontonet codec firmware version 06.0.1; Tieline codec firmware version 2.14.88.

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.

Important Note:
- 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.

Configuring a Tieline G5 Codec for a Peer-to-Peer SIP Call

1. Attach a LAN cable and any audio connections to the Tieline codec.
2. To dial peer-to-peer press the **HOME** button to return to the **Home** screen, select **Connect > IP > SIP**.
3. Use the numeric **KEYPAD** to enter the IP address of the codec you want to dial, using the or buttons to enter the periods in the IP address and use the **RETURN** button to delete numbers already entered.
4. Then press the down navigation button to select **Setup** and press to adjust the algorithm (usually MP2 or G.722), jitter buffer and encode/decode direction if required. Note: By default the Tieline codec will attempt to connect using MP2 and then G.722.
5. Press the **RETURN** button to navigate backwards to the **Connect SIP** screen.
6. Press the **CONNECT** button to make a connection.

Important Note: 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.

2 Connecting Tieline ISDN to other Codecs

To dial from a Tieline codec to a non-Tieline codec it is necessary to disable 'Session Data' and use an algorithm like G.722 or MPEG Layer 2 for compatibility. The same settings must be configured at both ends for:

- Mono or stereo
- Encoding (Algorithm)
- Sample Rate
- Other relevant settings on the non-Tieline codec

Following are configuration instructions for dialing to several non-Tieline codec brands over ISDN.

2.1 Connecting to APT Wordcast Equinox ISDN

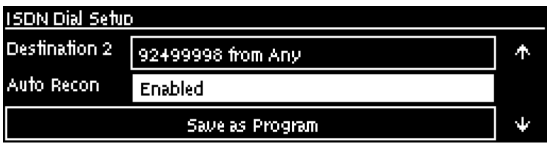
Configuring the WorldCast Equinox to Make an ISDN Call

1. Plug your ISDN line into the back of the codec and attach power.
2. Press the "Menu" button on the codec to access the codec menus.
3. Press the "Menu" button to select the "USER" menu.
4. Select "Primary Connection" and press the "Ent/Dial" button.
5. Next select "Codec - ISDN" > "Use Audio Profile" [No] > "Eapt-X16", or "Eapt-X24", "MPEG1-L2" or G.722.
6. Select the appropriate bit rate and whether you are dialing in mono, stereo or Joint Stereo, and then the sample rate.
7. For bonded "MPEG1-L2" connections select "CCS IMUX".
8. Complete the profile setup. The codec is now ready to dial or answer.

Configuring the Tieline Codec to Dial the Equinox over ISDN

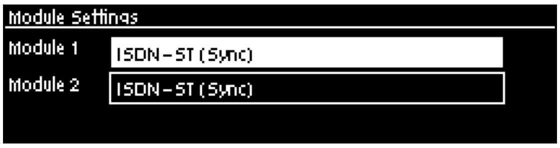
1. Press the **HOME** button to return to the **Home** screen and select **Connect > ISDN**.
2. Navigate to **Setup** and press the button.

3. Select **Session Type [Sessionless] > Select Dial Route [None] > Number of B Channels [**Choose the number of B-channels (between 1 and 4) required for your connection**] > Algorithm [**Choose **G.722** , **E apt-X Mono** or **Stereo**, **MP2 Mono or Stereo** or **MP2 J-Stereo (**Note: select **32kHz** or **48kHz** sample rate for **MP2** and **E apt-X** depending on available B-channels**)]**.
4. Navigate to a Destination (e.g. **Dest 1** or **Dest 2**) and press the button to select each one in turn. Enter the number for each B channel you want to dial and press the button, then select which B channel will dial using that number and press the button.
5. Navigate down to **Auto Reconnect** and press the button to toggle between **Enabled** and **Disabled**. Note: This is normally enabled on the dialing codec only.



6. Navigate down to **Save as Program** and press the button to save these settings as a program.
7. Navigate down to **Module Configuration** and press the button.

8. Select the ISDN module you want to configure and press the button.



9. Configure the following settings:
   - **Accept > Voice and Data**
   - **Network >** Check with your Telco (**EU-ETSI** in Australia; Europe & most countries outside North America; **[US Nat]** is the most common in the US, but check with your Telco).
   - **Line Type >** Check with your Telco and select either **Point-to-Multi** (point-to-multipoint) or **Point-to--Point** (point-to-point).
   - **DN/MSN >** Enter the "SPID" and "DN" numbers if required in your region, e.g. a SPID is normally required in the US.
10. Navigate up to **Apply Settings** and press the button.

Dialing from the Tieline Codec

Program Dialing

1. If you have saved the ISDN program as previously instructed, press the **HOME** button to return to the **Home** screen and select **Connect > Programs**.
2. Select the saved program you want to load and press the button.
3. Select **Load** and press the button to load the program.
4. Press the **CONNECT** button to dial the ISDN program connections.

Ad Hoc Dialing

1. If you haven't save the program but have entered the dialing numbers and other settings, press the **HOME** button to return to the **Home** screen and select **Connect > ISDN > Connect**.
2. Press the button to dial using the settings previously entered.

Important Note: If you select different algorithm settings on each codec and dial from the Tieline codec, the connection will be unsuccessful and the **CONNECTED LED** on the front panel of the Tieline codec will continuously flash. Adjust the algorithm settings and attempt to reconnect.

Dialing from the WorldCast Equinox

Important Note: Configure **ISDN Answer Config** settings in the codec before attempting to dial from the Equinox to the Tieline codec. Select the following settings in the Tieline codec in one of the **Configs** (see ISDN Answering Configuration for more detail):

- **May bond**.
- **Sessionless**.
- Algorithm: **G.722**, **MP2 Mono**, **MP2 Stereo**, **MP2 J-Stereo** or **E apt-X Mono** or **Stereo**.
- Sample Rate: **32kHz** or **48kHz**

1. Navigate to the B-channel you want to dial over and press the "Ent/Dial" button.
2. Use the keypad to enter the number of the line you are dialing.
3. Press the "Ent/Dial" button again to make the outgoing call from the codec.
4. If dialing two B-channels, navigate to the second B-channel and use the keypad to enter the number, then press the "Ent/Dial" button. Note: the codec screen will display IMUX UNLOCKED until you dial additional connections when bonding multiple channels.

Important Note: When dialing a mono or stereo connection over two B-channels audio is not available until the second connection is successful.

2.2 Connecting to CDQ Prima ISDN

Use the following information to connect a Tieline codec to a Musicam CDQ Prima codec.

Configuring the CDQ Prima for a Mono Connection

Select a mono profile in the Prima codec for the connection:

1. Press the “SDIAL” button on the front panel of the codec.
2. When “ID NUM” is displayed press “8” and then press "Enter" using the down arrow.
3. “MPEG2/64K:QS” will be displayed briefly followed by “WORKING”.
4. “OK” will be displayed momentarily and then the LCD screen will return to the default screen and be programmed for:
   - A Mono connection.
   - 64Kbps Bit-Rate.
   - 48K Sample Rate.
   - MPEG Layer 2 algorithm.
   - 1 ISDN B channel.
   - Decoder Independent – No.
5. Press the right arrow on the "Enter" button and navigate to "Interface". Push the down arrow on the "Enter" button to select this menu.
6. Use the "Enter" button and navigate to the type of interface you are using. Note: During Tieline tests we used an "Internal TA".
7. Select the actual terminal adapter connected to your codec. Note: During Tieline tests we used the internal "TA301".
8. Use the "Enter" button and select the switch type for the country you are in. Check with your Telco for the correct setting if you are unsure. Note: During Tieline tests we used the internal "NI1" setting for the USA.
9. Use the "Enter" button and keypad to enter the "SPID 1" and "SPID 2" numbers if required.
10. Use the "Enter" button and keypad to enter the "ID 1" and "ID 2" (Directory/MSN) numbers if required.
11. The codec should now be configured.

Configuring the CDQ Prima for a Stereo Connection

1. Press the “SDIAL” button on the front panel of the codec.
2. When “ID NUM” is displayed press “27” or "35" (in testing this varies depending on the firmware version in the CDQ Prima) and then press "Enter" using the down arrow. Note: if "27" or "35" don't work, look for "Zephyr/128K:QS" in the dial list.

3. “Zephyr/128K:QS” will be displayed briefly followed by “WORKING”.
4. “OK” will be displayed momentarily and then the LCD screen will return to the screen displayed prior to programming. The codec is now programmed for:
   - A Joint Stereo connection.
   - 128Kbps Bit-Rate.
   - 48K Sample Rate.
   - MPEG Layer 2 algorithm.
   - 2 ISDN B Channels
   - Decoder Independent – Yes
5. Press the right arrow on the "Enter" button and navigate to "Interface". Push the down arrow on the "Enter" button to select this menu.
6. Use the "Enter" button and navigate to the type of interface you are using. Note: During Tieline tests we used an "Internal TA".
7. Select the actual terminal adapter connected to your codec. Note: During Tieline tests we used the internal "TA301".
8. Use the "Enter" button and select the switch type for the country you are in. Check with your Telco for the correct setting if you are unsure. Note: During Tieline tests we used the internal "NI1" setting for the USA.
9. Use the "Enter" button and keypad to enter the "SPID 1" and "SPID 2" numbers if required.
10. Use the "Enter" button and keypad to enter the "ID 1" and "ID 2" (Directory/MSN) numbers if required.
11. The codec should now be configured.

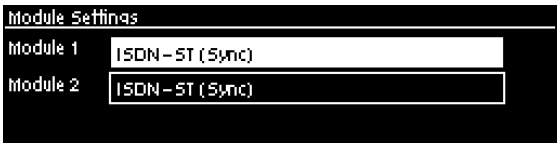
Configuring the Tieline Codec to Connect to the CDQ Prima

1. Press the **HOME** button to return to the **Home** screen and select **Connect > ISDN**.
2. Navigate to **Setup** and press the button.

3. Select **Session Type [Sessionless] > Select Dial Route [None] > Number of B Channels [**Choose **1B** (mono) **or 2B** (stereo)**] > Algorithm [**Choose **MP2 Mono or MP2 J-Stereo** (Note: select **48kHz** sample rate for MP2 algorithms)**]**.
4. Navigate to a Destination (e.g. **Dest 1** or **Dest 2**) and press the button to select each one in turn. Enter the number for each B channel you want to dial and press the button, then select which B channel will dial using that number and press the button. Note: In testing it was necessary to enter the ISDN number, followed by the SPID, to dial successfully.
5. Navigate down to **Auto Reconnect** and press the button to toggle between **Enabled** and **Disabled**. Note: This is normally enabled on the dialing codec only.

6. Navigate down to **Save as Program** and press the button to save these settings as a program.
7. Navigate down to **Module Configuration** and press the button.

8. Select the ISDN module you want to configure and press the OK button.



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

9. Configure the following settings:
   - **Accept > Voice and Data**
   - **Network >** Check with your Telco (**EU-ETSI** in Australia; Europe & most countries outside North America; **[US Nat]** is the most common in the US, but check with your Telco).
   - **Line Type >** Check with your Telco and select either **Point-to-Multi** (point-to-multipoint) or **Point-to--Point** (point-to-point).
   - **DN/MSN >** Enter the "SPID" and "DN" numbers if required in your region, e.g. a SPID is normally required in the US.
10. Navigate up to **Apply Settings** and press the OK button.

Dialing from the Tieline Codec

Program Dialing

1. If you have saved the ISDN program as previously instructed, press the **HOME** button to return to the **Home** screen and select **Connect > Programs**.
2. Select the saved program you want to load and press the OK button.
3. Select **Load** and press the OK button to load the program.
4. Press the **CONNECT** button to dial the ISDN program connections.

Ad Hoc Dialing

1. If you haven't save the program but have entered the dialing numbers and other settings, press the **HOME** button to return to the **Home** screen and select **Connect > ISDN > Connect**.
2. Press the OK button to dial using the settings previously entered.

After dialing successfully "FRAMED" should illuminate on the CDQ Prima screen. Tieline codecs also support 32kHz sampling.

Important Note: Configure **ISDN Answer Config** settings in the codec before attempting to dial from the Equinox to the Tieline codec. Select the following settings in the Tieline codec in one of the **Configs**:

- **May bond**.
- **Sessionless**.
- Algorithm: **MP2 Mono**, **MP2 J-Stereo**.
- Sample Rate: **48kHz**

Making a Mono Call from the CDQ Prima Codec

1. Press the "Dial" button on the front panel of the codec.
2. Navigate right using the "Enter" button and select "1".
3. Enter the number to dial using the numeric keypad.
4. Press the "Enter" button (bottom arrow) and the screen will briefly display "Working", then "Connect" and then the green "Framed" light should illuminate on the front panel.

Making a Stereo Call from the CDQ Prima Codec

1. Press the "Dial" button on the front panel of the codec.
2. Use the "Enter" button and select "Both".
3. Enter the first number to dial using the numeric keypad.
4. Press the "Enter" button (bottom arrow) and the screen will briefly display "Dialling line 1" and then "Connect".
5. Enter the second number to dial using the numeric keypad and press the "Enter" button (bottom arrow).
6. The screen will briefly display "Dialling line 2", then "Connect" and then the green "Framed" light should illuminate on the front panel.

Important Note: When connecting in stereo, the Prima expects both B channel dials to occur within 5 seconds. This can be performed by the Tieline codec.

It has also been noted that the CDQ Prima codec will not connect if no audio is present when dialing. It may connect Prima > Tieline, but not Tieline > Prima. If audio is present, the codec should connect and stay connected even if audio is removed subsequently. The J-Stereo light on the Prima may also flash when in this mode.

Ideally, have audio connected when dialing and the codec will frame immediately after the first dial and then dial the second B channel quickly afterwards.

2.3 Connecting to Mayah ISDN

Configuring the Mayah Sporty to Make an ISDN Call

1. Plug your ISDN line into the codec and attach power.
2. Press "F2 Codec".
3. Press "F3 Setup"
4. Press "F2 Quality"
5. Use the navigation buttons to select an algorithm setting from "G.722", "L2 Mono, Stereo or J-Stereo" or "E apt-X Mono or Stereo", then press the "OK" button to save the setting.
6. Press "F4 ESC" to return to the home screen.
7. Press "F2 Codec", then "F1 Connect", then "F3 Direct".
8. Navigate to "interface" and press "OK" to select ISDN.
9. Navigate to "number1" and press "OK" to enter the ISDN number using the keypad, then press "OK".
10. If you are bonding multiple channels navigate to "number2" and press "OK" to enter the ISDN number using the keypad, then press "OK".

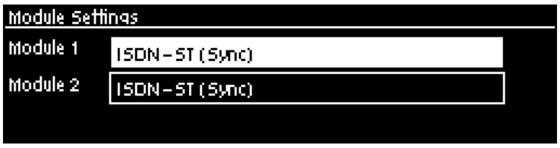
Configuring the Tieline Codec to Dial the Mayah Sporty over ISDN

1. Press the **HOME** button to return to the **Home** screen and select **Connect > ISDN**.
2. Navigate to **Setup** and press the button.

3. Select **Session Type [Sessionless] > Select Dial Route [None] > Number of B Channels [**Choose the number of B-channels (between 1 and 4) required for your connection**] > Algorithm [**Choose **G.722** , **E apt-X Mono** or **Stereo**, **MP2 Mono or Stereo** or **MP2 J-Stereo (**Note: select **32kHz** or **48kHz** sample rate for **MP2** depending on available B-channels**)]**.
4. Navigate to a Destination (e.g. **Dest 1** or **Dest 2**) and press the OK button to select each one in turn. Enter the number for each B channel you want to dial and press the OK button, then select which B channel will dial using that number and press the OK button.
5. 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.

6. Navigate down to **Save as Program** and press the OK button to save these settings as a program.
7. Navigate down to **Module Configuration** and press the OK button.

8. Select the ISDN module you want to configure and press the OK button.



9. Configure the following settings:
   - **Accept > Voice and Data**
   - **Network >** Check with your Telco (**EU-ETSI** in Australia; Europe & most countries outside North America; **[US Nat]** is the most common in the US, but check with your Telco).
   - **Line Type >** Check with your Telco and select either **Point-to-Multi** (point-to-multipoint) or **Point-to--Point** (point-to-point).
   - **DN/MSN >** Enter the "SPID" and "DN" numbers if required in your region, e.g. a SPID is normally required in the US.
10. Navigate up to **Apply Settings** and press the OK button.

Dialing from the Tieline Codec

Program Dialing

1. If you have saved the ISDN program as previously instructed, press the **HOME** button to return to the **Home** screen and select **Connect > Programs**.

2. Select the saved program you want to load and press the button.
3. Select **Load** and press the button to load the program.
4. Press the **CONNECT** button to dial the ISDN program connections.

Ad Hoc Dialing

1. If you haven't save the program but have entered the dialing numbers and other settings, press the **HOME** button to return to the **Home** screen and select **Connect > ISDN > Connect**.
2. Press the button to dial using the settings previously entered.

Important Note: If you select different algorithm settings on each codec and dial from the Tieline codec, the connection will be unsuccessful and the **CONNECTED LED** on the front panel of the Tieline codec will continuously flash. Adjust the algorithm settings and attempt to reconnect.

Dialing from the Mayah Sporty

Important Note: Configure **ISDN Answer Config** settings in the Tieline codec before attempting to dial from the Equinox to the Tieline codec. Select the following settings in the Tieline codec in one of the **Configs**:
- **May bond**.
- **Sessionless**.
- Algorithm: **G.722**, **MP2 Mono**, **MP2 Stereo**, **MP2 J-Stereo** or **E apt-X Mono** or **Stereo**.
- Sample Rate: **32kHz** or **48kHz**

1. Press "F4 ESC" to return to the home screen.
2. Press "F2 Codec", then "F1 Connect", then "F3 Direct".
3. Use the navigation buttons to select "dial" and press the "OK" button to dial all B-channels.

2.4 Connecting to Telos Zephyr Xstream ISDN

Configuring the Xstream to Make an ISDN Call

1. Plug your ISDN line into the back of the codec and press the "Codec" button below the LCD screen on the Xstream.
2. "Transmit" should be highlighted and this lets you select your transmit algorithm of choice. If it is not selected use the arrow buttons on the right-hand side of the LCD screen to navigate to this menu item and press the "SEL" button to the right of the LCD screen to select the menu.
3. Use the arrow buttons to navigate to:

- "G.722".
- "L2 J-Stereo" (for an MPEG Layer 2 stereo connection), or
- "L2 Mono 64" or "L2 Mono 128" (for a mono connection, depending on whether you have one or two B channels available).

4. Press the "SEL" button to store your setting and use the arrow down button to navigate to "Receive".
5. Press the "SEL" button and select the same algorithm that you selected for "Transmit" previously and then press the "SEL" button to store your setting.

Important Note: It you don't select the same algorithm for "Transmit" and "Receive" algorithms then it can take a long time to connect as the algorithms are scanned by the codec, or the wrong algorithm could be selected.

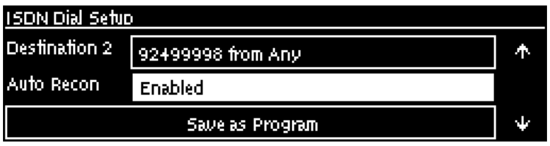
6. Use the arrow buttons to navigate to "Bitrate" and check that it displays "64kbps" - this is a per channel rate so both ISDN channels are programmed.
7. Use the arrow buttons to navigate to "Sample" and check that the sample rate is set at "48kHz". Press the "SEL" button and use the arrow buttons to make any adjustments to the current setting.
8. Press the "Tel" button below the codec LCD screen and press it again to display the "SPID" and "DN/MSN" screen. If these numbers need to be entered (check with your Telco), use the arrow buttons to navigate to each SPID and DN/MSN field in turn and when it is highlighted press the "SEL" button and enter the number using the keypad. Press "SEL" again to store each number once it has been entered.
9. Press the "Tel" button if you are not entering these SPID/DN/MSN numbers, or if you have already entered them, and check the local ISDN switch type setting is configured for your region.
10. Press the "SEL" button and use the arrow buttons to adjust the setting.
- Select "ETS300" if you are connecting to a Euro ISDN service.
- "Natl.I-1" is the most common in the US but check with your Telco.
11. Press the "SEL" button to store the ISDN switch type setting that you have selected.

Your codec should now be configured. Press the "Tel" button on the front panel until it displays the "ISDN Status" screen. "Ready" should be displayed next to any active lines. If this is not displayed, check your connections and settings to make sure they are correct.

Configuring the Tieline Codec to Dial the Xstream over ISDN

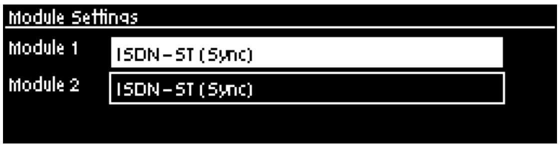
1. Press the **HOME** button to return to the **Home** screen and select **Connect > ISDN**.
2. Navigate to **Setup** and press the button.

3. Select **Session Type [Sessionless] > Select Dial Route [None] > Number of B Channels** [Choose **1B or 2B] > Algorithm** [Choose **G.722** , **MP2 Mono** or **MP2 Stereo** (Note: select **48kHz** sample rate for **MP2)]**.
4. Navigate to a Destination (e.g. **Dest 1** or **Dest 2**) and press the button to select each one in turn. Enter the number for each B channel you want to dial and press the button, then select which B channel will dial using that number and press the button.
5. Navigate down to **Auto Reconnect** and press the button to toggle between **Enabled** and **Disabled**. Note: This is normally enabled on the dialing codec only.



6. Navigate down to **Save as Program** and press the button to save these settings as a program.
7. Navigate down to **Module Configuration** and press the button.

8. Select the ISDN module you want to configure and press the button.



9. Configure the following settings:
   - **Accept > Voice and Data**
   - **Network >** Check with your Telco (**EU-ETSI** in Australia; Europe & most countries outside North America; **[US Nat]** is the most common in the US, but check with your Telco).
   - **Line Type >** Check with your Telco and select either **Point-to-Multi** (point-to-multipoint) or **Point-to--Point** (point-to-point).
   - **DN/MSN >** Enter the "SPID" and "DN" numbers if required in your region, e.g. a SPID is normally required in the US.
10. Navigate up to **Apply Settings** and press the button.

Dialing from the Tieline Codec

Program Dialing

1. If you have saved the ISDN program as previously instructed, press the **HOME** button to return to the **Home** screen and select **Connect > Programs**.
2. Select the saved program you want to load and press the button.
3. Select **Load** and press the button to load the program.
4. Press the **CONNECT** button to dial the ISDN program connections.

Ad Hoc Dialing

1. If you haven't save the program but have entered the dialing numbers and other settings, press the **HOME** button to return to the **Home** screen and select **Connect > ISDN > Connect**.
2. Press the button to dial using the settings previously entered.

Important Note: If you select different algorithm settings on each codec and dial from the Tieline codec, the connection will be unsuccessful and the **CONNECTED LED** on the front panel of the Tieline codec will continuously flash. Adjust the algorithm settings and attempt to reconnect.

Dialing from the Zephyr Xstream

Important Note: Configure **ISDN Answer Config** settings in the codec before attempting to dial from the Xstream to the Tieline codec. Select the following settings in the Tieline codec in one of the **Configs**:
- **May bond**.
- **Sessionless**.
- Algorithm: **G.722** or **MP2 Mono**, or **MP2 Stereo**.
- Sample Rate: **48kHz**

1. Press the "Dial" button once.

2. Use the keypad to enter the number of the line you are dialing.
3. Press the "Dial" button again to make the outgoing call from the Xstream.
4. The codec screen will briefly display "Outgoing Ring" and then "Conn" is displayed after a successful connection.
5. If you are making a stereo connection and need to dial the second line press the "Dial" button again and a screen for "Line 2" is displayed.
6. Use the keypad to enter the second number and press the "Dial" button again.
7. The "TEL" screen will briefly display "Outgoing Ring" and then "Conn" is displayed after a successful connection.

Two ISDN B Channels Connected

Important Note: When dialing a stereo connection over two ISDN B lines audio is not heard until the second connection is successful.

2.5 Connecting to Comrex Matrix ISDN

To connect your Tieline codec to a Comrex Matrix rack mount codec:

1. Use the G.722 algorithm.
2. Connect using only one 64Kbps ISDN B Channel (bonding of G.722 over two ISDN B channels is not possible).

Configuring the Matrix to Make an ISDN Call

1. Connect an ISDN line to the Matrix codec and power up the unit.
2. Press "2" to select "ISDN Status".
3. Press "Enter" to configure the connection.
4. Press "4" to select the "Configure" menu.
5. Press "2 to select the "Network" menu.
7. Press "4" to select "Profiles" and then press "1" to select "Load Profile".
8. Press "2" to select "Store" and program a new profile using the codec wizard.
9. Press ""Enter" to enter a profile number between 1 and 10. Note: This will overwrite any previously stored profile.
10. Next select the number for the ISDN "Switch Type" setting that is appropriate for your region.
- Press "4" to select "Euro" if you are connecting to a Euro ISDN service.
- Press "1" to select "NI1", which is the most common in the US, but check this with your Telco.
11. If prompted by the menu, use the keypad to enter the "SPID" number the line being used if this is required in your region. Press "Enter" to store the new number. Note: Use the "Cancel" button to delete numbers.
12. Next use the keypad to enter the "LDN" (DN/MSN) number for the line being used. Press "Enter" to store the new number. Note: Use the "Cancel" button to delete numbers.
13. Enter a "Qdial" (Quick Dial) number.
14. Press "1" to select "G.722" as the algorithm.

15. Press "2" to select "64" as the bit rate.
16. The codec is now programmed to dial.

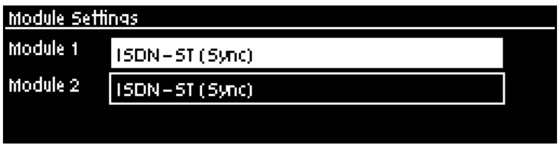
Configuring the Tieline Codec to Connect to the Matrix over G.722

1. Press the **HOME** button to return to the **Home** screen and select **Connect > ISDN**.
2. Navigate to **Setup** and press the button.

3. Select **Session Type [Sessionless] > Select Dial Route [None] > Number of B Channels [Select 1B] > Algorithm [Select G.722]**.
4. Navigate to a Destination (e.g. **Dest 1** or **Dest 2**) and press the button to select it. Then enter the number you want to dial and press the button, then select which B channel will dial using that number and press the button.
5. Navigate down to **Auto Reconnect** and press the button to toggle between **Enabled** and **Disabled**. Note: This is normally enabled on the dialing codec only.

6. Navigate down to **Save as Program** and press the button to save these settings as a program.
7. Navigate down to **Module Configuration** and press the button.

8. Select the ISDN module you want to configure and press the button.



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

9. Configure the following settings:
   - **Accept > Voice and Data**
   - **Network >** Check with your Telco (**EU-ETSI** in Australia; Europe & most countries outside North America; **[US Nat]** is the most common in the US, but check with your Telco).
   - **Line Type >** Check with your Telco and select either **Point-to-Multi** (point-to-multipoint) or **Point-to--Point** (point-to-point).

- **DN/MSN >** Enter the "SPID" and "DN" numbers if required in your region, e.g. a SPID is normally required in the US.

10. Navigate up to **Apply Settings** and press the button.

Dialing from the Tieline Codec

Program Dialing

1. If you have saved the ISDN program as previously instructed, press the **HOME** button to return to the **Home** screen and select **Connect > Programs**.
2. Select the saved program you want to load and press the button.
3. Select **Load** and press the button to load the program.
4. Press the **CONNECT** button to dial the ISDN program connections.

Ad Hoc Dialing

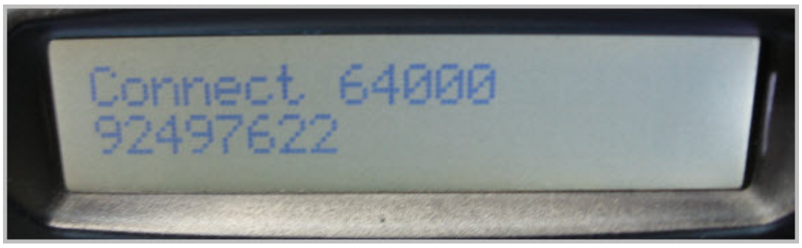
1. If you haven't save the program but have entered the dialing numbers and other settings, press the **HOME** button to return to the **Home** screen and select **Connect > ISDN > Connect**.
2. Press the button to dial using the settings previously entered.

Dialing from the Comrex Matrix

Important Note: Configure **ISDN Answer Config** settings in the codec before attempting to dial from the Comrex Matrix to the Tieline codec. Select the following settings in the Tieline codec in one of the **Configs**:

- **May bond**.
- **Sessionless**.
- **G.722** Algorithm

1. Use the "Cancel" button to return to the main LCD connection screen.
2. Press "Enter", then press "1" ("Dial") and use the numeric keypad to enter the number you wish to dial.
3. Press "Enter" to make the call.



Matrix Codec Screen when Connected