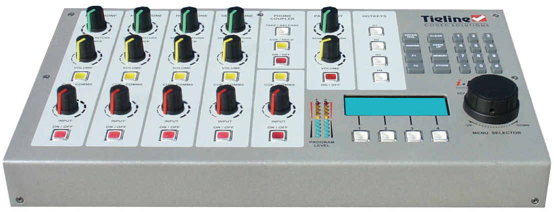

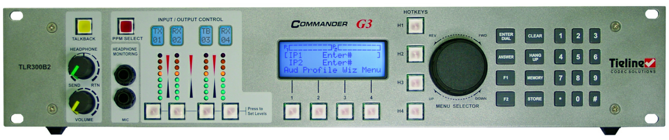

Tieline G3 Codec Compatibility

Manual Version: 1.0
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1 Introduction

When evaluating the performance of different brands of codecs the question of compatibility is often raised. It is comforting to know that Tieline is compatible with a large range of broadcast codecs. They are the only ones to support compatibility with most other brands of codecs over five different network types – IP, 3GIP, POTS/PSTN, ISDN, and X.21.

Tieline codecs offer a range of interchangeable connection modules that slot neatly into Commander G3 field, rack mount and i-Mix codecs in seconds, providing high quality connections between other Tieline codecs and a large range of different codec brands. Using interchangeable modules it is possible to build a codec that connects to your existing codec network infrastructure, as well as the codecs in other networks that you connect to.

1.1 IP Compatibility

Over IP Tieline is compatible with all major brands of codecs that have implemented the EBU N/ACIP tech 3326 specification relating to interoperability over IP using SIP (Session Initiation Protocol). SIP is an accepted and widely used technology for VoIP communications and its application has extended to include broadcast codecs. Tieline is developing SIP technology in accordance with the recommendations of the N/ACIP working group of the EBU, whose role is to standardize technology development approaches across manufacturers.

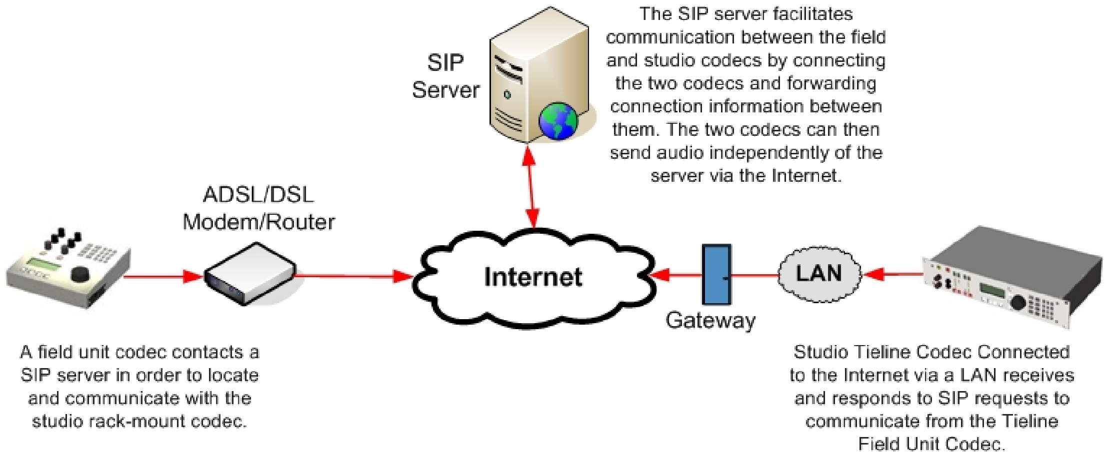
As a member of the Audio-via-IP Experts Group, Tieline is already compatible with fellow members Orban CRL, Mayah Communications and AETA. Recent tests confirmed compatibility with other manufacturers Prodys, Telos, AEQ, AVT, APT, Mandozzi and Digigram.

Tieline codecs support both peer-to-peer and SIP-server connections between all compatible codec brands.

SIP Server Connections

SIP works with a myriad of other protocols to establish connections with codecs and other devices over the Internet. It is used to find call participants and devices – even when they move regularly from place-to-place.

SIP is unique in that is allows dissimilar devices to communicate with each other once they have been registered to a SIP server. In a nutshell, a device using SIP dials another device’s SIP address to find its location. This task is performed by SIP servers, which communicate between SIP-compliant devices to set up a call. Unless you configure your own SIP server Tieline recommends that you make peer-to-peer SIP calls.



Example of a SIP Server Connection

Peer-to-Peer Connections

Tieline codecs don’t need to be registered to a SIP server to make peer-to-peer connections and are the easiest to set up and perform. A peer-to-peer connection involves two codecs connecting to each other by dialing as you would for a standard Tieline IP call. The only difference is that a standard Tieline IP call uses proprietary Tieline session data to negotiate and program the IP parameters (e.g. algorithm and bit rate), whereas a peer-to-peer SIP connection uses Session Description Protocol (SDP) that all compatible codec brands recognise for dialing and connecting.

1.2 POTS/PSTN Compatibility

All Tieline POTS modules create rock-solid, reliable, high quality connections because they measure the amount of fluctuation in line quality before working out the optimal bit-rate setting to connect at. POTS modules also include *SmartDAA™ technology to automatically interface with local phone line characteristics in a large number of supported countries. Just set your country of operation and the codec does the rest!

Tieline POTS G3 technology also detects and filters ADSL and DSL signals, negotiates stable bit-rates on shared lines, and provides more stable bonded connections. All Tieline codecs are also POTS-compatible with **Comrex Matrix, Access, Vector and Blue codecs.

Tieline is POTS and GSM Voice Call Compatible

Tieline POTS, GSM and UMTS/HSDPA wireless 3G modules are also capable of making standard voice calls. Tieline codecs can:

- Make a standard voice call to or from a Tieline codec.
- Integrate a standard voice call into program content.
- Integrate a standard voice call for communications.
- Dial into a hybrid telephone from a Tieline codec.

Voice calls are 3kHz in in audio quality whereas Tieline's proprietary Music coding algorithm delivers 15 kHz quality bi-directional audio at a bit rate as low as 24Kbps. Tileine Voice G3 can deliever 7 kHz audio from 9.6Kbps.

**SmartDAA is a trademark, patent pending, of Conexant Systems, Inc in the United States and/or other countries.*
***All products and trademarks mentioned are property of their respective companies.*

1.3 ISDN and X.21 Compatibility

When connecting over ISDN and X.21, Tieline codecs are compatible with all major brands of codecs using the popular G.722 and MPEG algorithms.

ISDN Connections

Tieline ISDN codecs provide simple, ultra-reliable, high-speed digital leased line connections that can connect over either ‘U' or ‘S/T' ISDN interfaces using purpose-built Tieline ISDN plug-in modules.

The module that you need depends on whether you have a NT-1 device connected to the line and most North American installations do not have an NT-1. In the USA the telephone company provides its customers with a two-wire (single pair) U interface from the phone switch, the same physical interface provided for POTS lines. It supports full-duplex data transfer over a single pair of wires, therefore only a single device can be connected to a U interface.

The situation is different in Europe, the UK, most of Asia, Australia, Africa and parts of the Middle East where the phone company is allowed to supply the NT-1 and the customer is given an S/T interface. The NT-1 is a relatively simple device that converts the 2-wire U interface into the 4-wire S/T interface supporting multiple devices. While it is still a full-duplex interface, there is now a pair of wires for receiving data, and another pair for transmitting data.

If you have an NT1 device connected to the U interface line then you will require a Tieline S/T ISDN module. If you don’t have an NT1 device installed then the Tieline U ISDN module is required. You can ring your telecommunications provider to ask if you’re not sure.

If you already own a Tieline codec, upgrading your codec for ISDN use is a simple process - just purchase the ISDN module for your region, plug it into the codec, load the profile you require and you are ready to connect.

X.21 Connections

X.21 is a protocol that runs over subscriber networks that transfers synchronous serial data. 64 - 384 kbps connections are most common and Tieline codecs support X.21 leased line mode connections from 64 - 2,048 kbps.

1.4 Satellite Compatibility

Discover the ultimate in wireless freedom using Tieline codecs over BGAN satellite terminal connections. Tieline's satellite solutions expand broadcast opportunities by delivering reliable and flexible connections from anywhere that satellite coverage is available.

Tieline i-Mix G3 or Commander G3 audio codecs connected to BGAN terminals are able to stream live bidirectional FM quality mono and stereo audio from any location. Tieline's unique solutions include:

- Live broadcast quality mono or stereo FM quality audio using 64Kbps or higher BGAN ISDN connections.
- Live broadcast quality mono FM audio using 32 or 64Kbps BGAN IP connections.

Tieline codecs can connect over ISDN through the use of interchangeable connection modules. IP connections can be created by using the LAN port available on all Tieline codecs and no extra hardware is required.

Satellite IP - the new frontier...

Tieline IP broadcasting over satellite connections is the most cost effective and dependable way to get your audio to the studio from very remote locations. Tieline codecs can deliver 15kHz quality audio over 32Kbps satellite connections, providing significant savings on data costs when compared to 64Kbps connections.

Reliability is paramount and Tieline's IP solutions feature QoS Performance Engine technology for reliably managing connections over unreliable IP networks like the Internet. All codecs include features such as Forward Error Correction (FEC) and automated jitter buffering for demanding IP broadcast situations.

Connecting is a breeze because Tieline has, in association with Inmarsat, produced a comprehensive 'how to' guide for creating satellite broadcast connections over a range of BGAN terminals. Download this document at http://www.tieline.com/files/files/244_Using-Tieline-Codecs-over-BGAN-v10.pdf.

1.5 In Summary

This means that if you have a Tieline rack unit codec in your studio with a LAN connection attached and a POTS/PSTN and ISDN module installed, you can accept calls from any ISDN codec, most IP codecs, most Comrex codecs and Tieline wireless 3G codecs – now that’s compatible!

2 Tieline Codec Interoperability

When Tieline codecs establish communications during the dialing and connection process, the dialing codec sends settings to the remote codec using Tieline Session Data. The receiving codec inherits the following settings from the dialing codec:

- Connection Profile (when using default profiles).
- Algorithm.
- Sample Rate.
- Bit rate.

Other brands of codecs do not recognise Tieline Session Data so this must be turned off in the Tieline codec to allow them to connect.

2.1 POTS/PSTN Interoperability

Tieline codecs are compatible over POTS/PSTN with *Comrex Matrix, Access, Vector and Blue codecs.

**All products and trademarks mentioned are property of their respective companies.*

2.1.1 Connect to Comrex Matrix POTS

Comrex Matrix rack mount codecs can connect over POTS and ISDN. Over POTS you need to:

a. Connect a single POTS line to each codec.
b. Select the "Music" algorithm in the Comrex codec.
c. Select the "Other" algorithm in the Tieline codec.

Important Note: The "Other" algorithm in Tieline codecs is specifically used with Comrex® Vector, Matrix® and BlueBox® codecs. Bitrates of 26,400bps and 9,600bps are not supported with this algorithm. If you try to connect at these bit rates an error message will be displayed on the codec LCD next to the connection you are using. In addition, the Line Quality at the remote Comrex® codec will not be displayed on the Tieline codec LCD screen.

Programming the Matrix to Make a POTS Call

1. Connect a POTS line to the Matrix codec and power up the unit.
2. Press "Enter" view the main menu.
3. Press "4" to enter the "Config" menu.
4. Press "2" to select an algorithm.
5. Press "2" to select "Music".
6. Press "3" to select "Max Rate" and program the maximum bit rate.

Important Note: As a rule of thumb we recommend a bit rate of 21.6Kbps, although 24Kbps is possible depending on line quality. If you connect at 24Kbps and find the quality of the connection is poor, you will need to disconnect and reset the maximum bit rate in both codecs to 21.6Kbps or lower.

7. Press "3" to select "21.6".
8. Press "4" to select "More" if you wish to program auto answering, dialing (Tone/ Pulse), modem monitoring, input and output levels.
9. Press the "Cancel" button several times until you return to the main connection screen.
10. Check your settings are displayed correctly. The codec is now programmed to connect.

Comrex Matrix Screen after Programming

Programming the Tieline Codec to Connect to the Matrix over POTS

1. Scroll using the black rotary **MENU SELECTOR** to the POTS connection on the LCD screen that you will be using to dial the connection.
2. Press **SOFTKEY 2 [Profile] > [ManDflt MonoPgm] > [Really Load – Select YES]**
3. Press **SOFTKEY 3 [Wiz] > [OK]** and use the wizard to select **[Setup POTS] > [Algorithm "Other"] > [Max Bitrate "21600"] > [Auto Renegotiate "Disable"] > [Monitor Dialtone "Enable"] > [Dialling Method "Tone"] > [Auto Reconnect "Disable"]**
4. Select **[Menu] > [Configuration] > [System Settings] > [Session Data] > [Disable]**

Important Note: Please ensure you program both codecs with the same settings to provide the best chance of connecting first time.

Dialing from the Tieline Codec

1. Rotate the black rotary **MENU SELECTOR** until the square brackets surround **[POTS1>Enter #].**
2. Use the numeric keypad to enter the phone number and press the "Enter/Dial" button.
3. The Tieline codec connection should display **[POTS 1> 21.6]** when connected. The local line quality "L" is also displayed.

Making a Call Using the Comrex Matrix

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

Comrex Matrix Screen after it has Dialed and Connected

2.1.2 Connect to Comrex Access Rack over POTS

To connect to a Comrex Access 1RU rack mount codec you need to:

a. Connect a single POTS line to each codec.
b. Select the "Other" algorithm in the Tieline codec.
c. Dial the Comrex Access codec from the Tieline codec.

Important Note: The "Other" algorithm in Tieline codecs is specifically used with Comrex® Vector, Matrix® and BlueBox® codecs. Bitrates of 26,400bps and 9,600bps are not supported with this algorithm. If you try to connect at these bit rates an error message will be displayed on the codec LCD next to the connection you are using. In addition, the Line Quality at the remote Comrex® codec will not be displayed on the Tieline codec LCD screen.

Programming the Tieline Codec to Connect to the Access Rack Mount Codec in Mono

1. Scroll using the black rotary **MENU SELECTOR** to the POTS connection on the LCD screen that you will be using to dial the connection.
2. Press **SOFTKEY 2 [Profile] > [ManDflt MonoPgm] > [Really Load – Select YES]**
3. Press **SOFTKEY 3 [Wiz] > [OK]** and use the wizard to select **[Setup POTS] > [Algorithm "Other"] > [Max Bitrate "21600"] > [Auto Renegotiate "Disable"] > [Monitor Dialtone "Enable"] > [Dialling Method "Tone"] > [Auto Reconnect "Disable"]**
4. Select **[Menu] > [Configuration] > [System Settings] > [Session Data] > [Disable]**

Important Note: As a rule of thumb we recommend a bit rate of 21.6Kbps, although 24Kbps is possible depending on line quality. If you connect at 24Kbps and find the quality of the connection is poor, you will need to disconnect and reset the maximum bit rate in both codecs to 21.6Kbps or lower.

Dialing from the Tieline Codec

1. Rotate the black rotary **MENU SELECTOR** until the square brackets surround **[POTS1>Enter #].**
2. Use the numeric keypad to enter the phone number and press the "Enter/Dial" button.
3. The Tieline codec connection should display **[POTS 1> 21.6]** when connected. The local line quality "L" is also displayed.
4. The Access codec displays incoming audio on the two "Receive" PPM meters on the codec front panel. Consult Comrex user documentation to adjust Access codec audio levels if required.

2.1.3 Connect to Comrex Access Portable

To connect to a Comrex Access Portable codec you need to:

a. Connect a single POTS line to each codec.
b. Select the "Other" algorithm in the Tieline codec.

Important Note: The "Other" algorithm in Tieline codecs is specifically used with Comrex® Vector, Matrix® and BlueBox® codecs. Bitrates of 26,400bps and 9,600bps are not supported with this algorithm. If you try to connect at these bit rates an error message will be displayed on the codec LCD next to the connection you are using. In addition, the Line Quality at the remote Comrex® codec will not be displayed on the Tieline codec LCD screen.

Programming the Tieline Codec to Connect to the Access Portable Codec in Mono

1. Scroll using the black rotary **MENU SELECTOR** to the POTS connection on the LCD screen that you will be using to dial the connection.
2. Press **SOFTKEY 2 [Profile] > [ManDflt MonoPgm] > [Really Load – Select YES]**
3. Press **SOFTKEY 3 [Wiz] > [OK]** and use the wizard to select **[Setup POTS] > [Algorithm "Other"] > [Max Bitrate "21600"] > [Auto Renegotiate "Disable"] > [Monitor Dialtone "Enable"] > [Dialling Method "Tone"] > [Auto Reconnect "Disable"]**
4. Select **[Menu] > [Configuration] > [System Settings] > [Session Data] > [Disable]**

Important Note: As a rule of thumb we recommend a bit rate of 21.6Kbps, although 24Kbps is possible depending on line quality. If you connect at 24Kbps and find the quality of the connection is poor, you will need to disconnect and reset the maximum bit rate in both codecs to 21.6Kbps or lower.

Dialing from the Tieline Codec

1. Rotate the black rotary **MENU SELECTOR** until the square brackets surround **[POTS1>Enter #]**.
2. Use the numeric keypad to enter the phone number and press the **ENTER/DIAL** button.
3. The Tieline codec connection should display **[POTS 1> 21.6]** when connected. The local line quality "L" is also displayed.
4. The Access portable codec displays incoming and outgoing audio on the LCD screen PPM meters.

Dialing from the Access Portable Codec

1. To create a new "Remote" connection use the stylus to tap **Remotes**, then tap **Add New Remote**.
2. Enter the **Name** of the connection and the phone number, then tap to select the **POTS** profile from the drop-down list box, next tap the **OK** button.
3. Tap on the Remote connection you have just added and tap the **Connect** button on the screen to dial the Tieline codec.
4. Tap the **Disconnect** button to hangup the connection.

Important Note: Once connected, audio levels may need to be adjusted at either end because there is approximately 10dB difference between the output levels of each codec.

2.2 ISDN Interoperability

Ensure that you have the correct ISDN module in your Tieline codec. A 'U' module is required in North America and an 'S/T' module is required in most other countries. In general the following settings need to be configured in your Tieline codec when connecting to other codec brands over ISDN.

2.2.1 Connecting to CDQ Prima ISDN

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

Programming the CDQ Prima for a Mono Connection

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

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

Programming the Tieline Codec for Connecting in Mono to the CDQ Prima

1. Press **SOFTKEY 2 [Profile] > [ManDflt MonoPgm] > [Really Load – Select YES]**
2. Scroll using the black rotary **MENU SELECTOR** to the ISDN connection on the LCD screen that you will be using to dial the connection.
3. Press **SOFTKEY 3 [Wiz] > [OK]** and use the wizard to select:
   - 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).
- Algorithm: Select **[MP2 Mono]** and choose a sample rate of **[48000]**.
- ISDN Line Type: Check with your Telco and select either "P-Multi" (point-to-multipoint) or "P-Point" (point-to-point).
- Local SubAddress: (Not required in Australia)
- Enter the "SPID" and "DN" numbers if required in your region, e.g. in the US.
- Auto Reconnect: Select: Enable or Disable (as required)

4. Select **[Menu] > [Configuration] > [System Settings] > [Session Data] > [Disable]**

Important Note: If Sample Rate is not available in the configuration Wizard (pre v.1.4.xx codec firmware releases), select **[Menu] > [Configuration] > [ISDN <Left (or Right) Setup] > [Algorithm] > [Sample Rate] > [48K].**

Making a Mono Call from the Tieline Codec

1. Rotate the black rotary **MENU SELECTOR** until the square brackets surround **[ISDN 1> Enter #Sync].**
2. Enter the number to dial using the numeric keypad.
3. Press the "Enter/Dial" button.

The call should now connect and “FRAMED” should illuminate on the CDQ Prima screen. Tieline codecs also support 16K, 24K and 32K sample rates.

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.

Programming the CDQ Prima for a Stereo Connection

1. Press the “SDIAL” button on the front panel of the codec.
2. When “ID NUM” is displayed press “27” and then press "Enter" using the down arrow.
3. “Zephyr/128K:QS” will be displayed briefly followed by “WORKING”.
4. “OK” will be displayed momentarily and then the LCD screen will return to the screen displayed prior to programming. The codec is now programmed for:
   - A Joint Stereo connection.
   - 128Kbps Bit-Rate.
   - 48K Sample Rate.
   - MPEG Layer 2 algorithm.
   - 2 ISDN B Channels
   - Decoder Independent – Yes
5. Press the right arrow on the "Enter" button and navigate to "Interface". Push the down arrow on the "Enter" button to select this menu.
6. Use the "Enter" button and navigate to the type of interface you are using. Note: During Tieline tests we used an "Internal TA".

7. Select the actual terminal adapter connected to your codec. Note: During Tieline tests we used the internal "TA301".
8. Use the "Enter" button and select the switch type for the country you are in. Check with your Telco for the correct setting if you are unsure. Note: During Tieline tests we used the internal "NI1" setting for the USA.
9. Use the "Enter" button and keypad to enter the "SPID 1" and "SPID 2" numbers if required.
10. Use the "Enter" button and keypad to enter the "ID 1" and "ID 2" (Directory/MSN) numbers if required.
11. The codec should now be configured.

Programming the Tieline Codec for a Stereo Connection

1. Press **SOFTKEY 2 [Profile] > [ManDflt Stereo] > [Really Load – Select YES]**
2. Scroll using the black rotary **MENU SELECTOR** to the ISDN connection on the LCD screen that you will be using to dial the connection.
3. Press **SOFTKEY 3 [Wiz] > [OK]** and use the wizard to select:
- Network: Check with your Telco (EU-ETSI in Australia; Europe & most countries outside North America).
- Algorithm: Select **[MP2 J-Stereo]** and choose a sample rate of **[48000]**.
- ISDN Line Type: Check with your Telco and select either "P-Multi" (point-to-multipoint) or "P-Point" (point-to-point).
- Local SubAddress: (Not required in Australia)
- Auto Reconnect: Select: Enable or Disable (as required)

5. Select **[Menu] > [Configuration] > [System Settings] > [Session Data] > [Disable]**

Important Note: If Sample Rate is not available in the configuration Wizard (pre v.1.4.xx codec firmware releases), select **[Menu] > [Configuration] > [ISDN <Left (or Right) Setup] > [Algorithm] > [Sample Rate] > [48K].**

Making a Stereo Call from the Tieline Codec

1. Rotate the black rotary **MENU SELECTOR** until the square brackets surround **[ISDN 1> Enter #Sync]**.
2. Enter the number to dial using the numeric keypad.
3. Press the "Enter/Dial" button.
4. The Tieline codec connection displays **[ISDN 1> 64.0 WAITING]** until the second connection is dialed.
5. Rotate the black rotary **MENU SELECTOR** until the square brackets surround **[ISDN 2> Enter #Sync].**
6. Press the "Enter/Dial" button.
7. When the second call connects both connections on the Tieline codec display **[ISDN> +MP2 J-]**

Both calls should now connect and “FRAMED” should illuminate on the CDQ Prima screen. Tieline codecs also support 16K, 24K and 32K sample rates.

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; simply ignore the warning messages on the Tieline LCD screen.

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.

2.2.2 Connecting to Zephyr Xstream ISDN

Programming 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.

Programming the Tieline Codec to Connect to the Xstream over ISDN

1. Scroll using the black rotary **MENU SELECTOR** to the ISDN connection on the LCD screen that you will be using to dial the connection.
2. Press **SOFTKEY 2 [Profile] > [ManDflt MonoPgm or Stereo] > [Really Load – Select YES]**
3. Press **SOFTKEY 3 [Wiz] > [OK]** and use the wizard to select:
    - 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).
    - Algorithm: Select from "MP2" or "G.722" algorithm options.
        - Note: If selecting "MP2", choose the "48000" sample rate.
    - ISDN Line Type: Check with your Telco and select either "P-Multi" (point-to-multipoint) or "P-Point" (point-to-point).
    - Local SubAddress: (Not required in Australia).
    - Enter the "SPID" and "DN" numbers if required in your region, e.g. in the US.
    - Auto Reconnect: Select "Enable" or "Disable"

3. Select **[Menu] > [Configuration] > [System Settings] > [Session Data] > [Disable]**

Dialing from the Zephyr Xstream

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 codec 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, please not that it can take up to a minute for audio to commence streaming after the second connection is successful.

Dialing from the Tieline Codec

1. Rotate the black rotary **MENU SELECTOR** until the square brackets surround **[ISDN 1>Enter #Sync].**
2. Enter the number to dial using the numeric keypad.
3. Press the "Enter/Dial" button.
4. If only one connection is required, i.e. mono, the connection will be displayed as **[ISDN 1> 64.0 MP2]**; If a second connection is required, i.e. for J-Stereo, the first connection dialed will display **[ISDN 1> 64.0 WAITING]** until the second connection is dialed.
5. If you are making a stereo connection you will need to dial the number second B channel connection (Auxiliary number). Rotate the black rotary **MENU SELECTOR** until the square brackets surround **[ISDN 2> Enter #Sync].**
6. Press the "Enter/Dial" button.
7. When the second call connects both connections on the Tieline codec display **[ISDN> +MP2 J-]**

2.2.3 Connecting to Comrex Matrix ISDN

Comrex Matrix rack mount codecs can connect over POTS and ISDN. Over ISDN you need to:

a. Use the G.722 algorithm to connect your Tieline codec to the Matrix.
b. Connect using only one 64Kbps ISDN B Channel (bonding of G.722 over two ISDN B channels is not possible).

Programming 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 "2" 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.

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

1. Scroll using the black rotary **MENU SELECTOR** to the ISDN connection on the LCD screen that you will be using to dial the connection.
2. Press **SOFTKEY 2 [Profile] > [ManDflt MonoPgm] > [Really Load – Select YES]**
3. Press **SOFTKEY 3 [Wiz] > [OK]** and use the wizard to select:
    - 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).
    - Algorithm: Select from "G.722" algorithm options.
    - ISDN Line Type: Check with your Telco and select either "P-Multi" (point-to-multipoint) or "P-Point" (point-to-point).
    - Local SubAddress: (Not required in Australia).
    - Enter the "SPID" and "DN" numbers if required in your region, e.g. in the US.
    - Auto Reconnect: Select "Enable" or "Disable"

3. Select **[Menu] > [Configuration] > [System Settings] > [Session Data] > [Disable]**

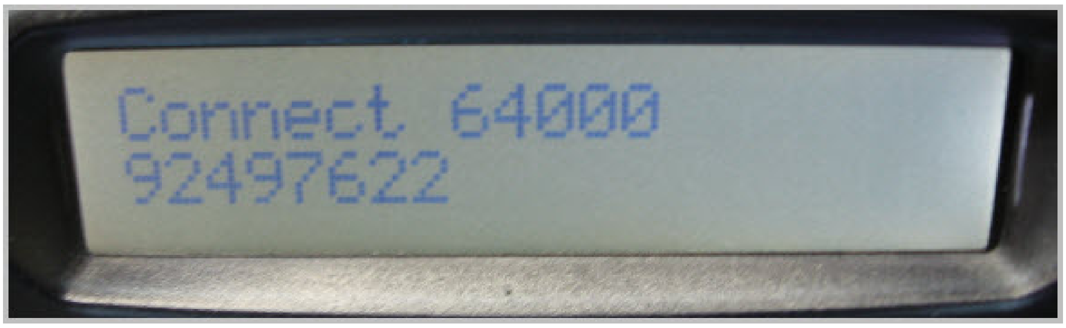
Dialing from the Tieline Codec

1. Rotate the black rotary **MENU SELECTOR** until the square brackets surround **[ISDN 1>Enter #Sync].**
2. Press the "Enter/Dial" button.
3. The Tieline codec connection displays **[ISDN 1> 64.0 G.722]** when connected.

Important Note: The "G.722" algorithm display on the Tieline LCD screen flashes until audio begins streaming successfully. This should occur within 20 seconds of connection.

Making a Call Using the Comrex Matrix

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

Important Note: If you have difficulty connecting, reprogram the following setting in the Tieline codec. Select **[Menu] > [Configuration] > [Audio Setup] > [Audio Ref Level] > [Other Codecs].**

2.3 IP SIP Interoperability

The following tests were devised as mandatory tests that should be adhered to as a minimum for interoperability of different codec brands over IP. As a general rule Tieline codecs are compatible with all major brands of codecs that have implemented the EBU N/ACIP tech 3326 specification relating to interoperability over IP using SIP (Session Initiation Protocol). The benefit of using SIP to make a peer-to-peer call is that Session Description Protocol can theoretically connect to any SIP enabled codec or device.

All the following Tieline G3 connection tests were performed with the listed codec manufacturers in May 2009.

Codec	G.711 A-law	G.722	RAW 20bit 48KHz Stereo	MP2 256kbps 48kHz Stereo
Prodys	pass	pass	pass	pass
Telos	pass (Note 3&4)	pass (Note 3&4)	fail (Note 8)	pass (Note 3&4)
Orban	pass	pass	pass	pass
Digigram	pass	pass	pass	pass
Mayah	pass	pass	pass	pass
AETA	pass	pass	not tested	pass
AEQ	pass	pass	not supported	not supported
AVT	pass @ 8ms	pass	pass	pass
APT (Note 9)	pass	fail (Note6)	not tested	pass
Mandozzi	fail (Note 7)	pass	fail (Note 7)	pass (Note 9)

	Notes:
1	Tieline firmware version 1.6.48 used.
2	All tests in SIP-Registered mode. Point-to-point only tested if SIP-Registration mode failed.
3	Telos could not make registered calls to anyone.
4	Telos could not disconnect a registered connection when Tieline initiated the call.
5	Successfully tested send and receive only originated from Mayah codec
6	APT sent non-compliant audio packets.
7	Mandozzi sent non-compliant audio packets.
8	Telos only run at a 5ms frame rate.
9	The Mandozzi and APT codec does not follow the correct rules for the RTP timestamp increment which disables Tieline auto-jitter buffer.

Additional Tests Performed

A number of optional tests were performed by various manufacturers over uncompressed (RAW) audio connections as well as other MPEG sample rates and bit rates. The results are listed in the following tables.

Codec	RAW 12bit 32kHz Stereo	RAW 16bit 48kHz Mono	RAW 20bit 48KHz mono	RAW 24bit 48KHz Stereo
Prodys	pass (8ms)	pass (8ms)		pass
Orban	pass			pass
Digigram				pass
AEQ		pass	pass	
AVT		pass	pass	pass

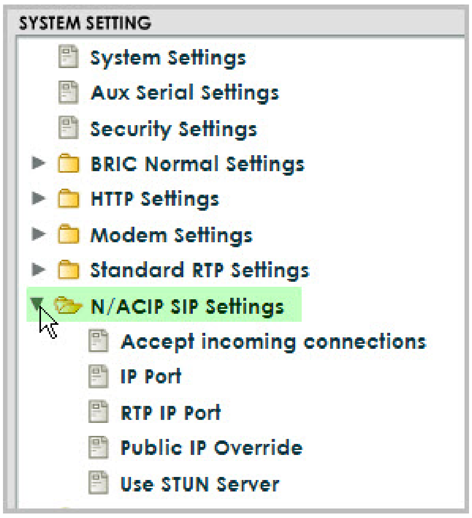
Codec	MP2 Mono 24Khz 64Kbps	MP2 Stereo 48Khz 128Kbps	MP2 Mono 48Khz 128Kbps	Uni directional test
Digigram	pass	pass		
Mayah				pass (Note 5)
AETA	pass	pass		
AEQ			pass	

2.3.1 Connect to Comrex Access Rack Unit

Important Note: Firmware installed; Comrex Access codec firmware version 2.7-p5; Tieline codec firmware version 1.6.48.

Programming the Comrex Access Rack Attached to a LAN to Connect using SIP

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). Note: Ensure the firmware version in the Access codec is 2.7p5 or higher and the GUI version is 2.7c or higher; download the Comrex Device Manager and the latest Access firmware for installation into the codec.
5. Click the "System Settings" tab in the Comrex Access GUI.
6. Click the arrow symbol to the left of the "N/ACIP SIP Settings" folder to view all SIP settings.



7. Click on "Accept incoming connections" and click on the "Enabled" option box.
8. Click on "IP Port" and ensure that port "5060" is entered into the port number box.
9. Click on "RTP IP Port" and ensure that port "5004" is entered into the port number box.

Programming a Tieline G3 Codec to Connect using SIP

1. Attach a LAN cable and any audio connections to the Tieline codec.
2. Press **SOFTKEY 2 [Profile] > [ManDflt MonoPgm] > [Really Load – Select YES]** to select the mono profile.
3. Program the Tieline codec to be in SIP mode by selecting **[Menu] > [Configuration] > [IP1 Setup] > [Session Type] > [SIP].**
4. Select the **SIP** connection on the Tieline codec LCD screen and press **SOFTKEY 3 [Wiz]** to configure the Tieline codec using the IP Configuration Wizard.
5. Select either the G.711 or G.722 algorithms over a public internet connection or WAN. Raw audio (uncompressed) may be selected over a private LAN.

Dialing the Connection using the Tieline Codec

1. Tieline recommends dialing from the Tieline codec to the Comrex Access codec. To connect enter the IP address for the Access codec (private IP address over a LAN or public IP address if connecting over a WAN) and press **ENTER/DIAL** to connect to the Comrex Access codec.
2. Press the **HANGUP** button to disconnect the connection.

Dialing the Connection using the Comrex Access

It is also possible to dial from the Comrex Access to the Tieline codec.

1. Click on the "Connections" tab in the Comrex Access GUI.
2. Click the **Store New Remote** button to create a new connection.
3. Complete the connection details and select the appropriate profile ("LPCM16" for raw uncompressed audio; or click the "Profiles" tab and in the "Profile Setting" section select "Local", then "Encoder" and click "VoIP" in the adjustment pane to select either "G.711" or "G.722".

4. Click on the connection you have just created in the **Connections** pane and click the "Connect" button.
5. Click the "Disconnect" button to hangup the connection.

2.3.2 Connect to Comrex Access Portable

Important Note: Firmware installed; Comrex Access Portable codec firmware version 2.7-p6; Tieline codec firmware version 1.6.48.

Programming the Access Portable Attached to a LAN to Connect in Mono using SIP

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.

Programming a Tieline G3 Codec to Connect using SIP

1. Attach a LAN cable and any audio connections to the Tieline codec.
2. Press **SOFTKEY 2 [Profile] > [ManDflt MonoPgm] > [Really Load – Select YES]** to select the mono profile.
3. Program the Tieline codec to be in SIP mode by selecting **[Menu] > [Configuration] > [IP1 Setup] > [Session Type] > [SIP].**
4. Select the **SIP** connection on the Tieline codec LCD screen and press **SOFTKEY 3 [Wiz]** to configure the Tieline codec using the IP Configuration Wizard.
5. Select either the G.711 or G.722 algorithms over a public internet connection or WAN. Raw audio (uncompressed) may be selected over a private LAN.

Important Note: If you are in the USA or Japan set your country code to the correct country in the Tieline codec by selecting **[Menu] > [Configuration] > [System Settings] > [Country].** This selects G.711 µ-law for North America and Japan, whereas a-law is prevalent in other regions of the world.

Dialing the Connection using the Tieline Codec

1. Tieline recommends dialing from the Tieline codec to the Comrex Access codec if possible because it is simpler to program. To connect enter the IP address for the Access codec (private IP address over a LAN or public IP address if connecting over a WAN) and press **ENTER/DIAL** to connect to the Comrex Access codec.
2. Press the **HANGUP** button to disconnect the connection.

Dialing the Connection using the Comrex Access

It is also possible to dial from the Comrex Access to the Tieline codec. To do this you need to create a new "remote" and a new "profile" in the Access portable.

1. Tap the **Configure** drop-down menu in the top-left of the codec touch-screen and then tap **Manage Profiles**.
2. Tap the **Add New** button to create a **New Profile** in the **Available Profiles** list.
3. Tap **New Profile** to highlight it and tap the **Edit** button.
4. 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.
5. 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.
6. 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.
7. Tap the **Done** button in the **Profile Settings** screen.
8. To create a new "Remote" connection use the stylus to tap **Remotes**, then tap **Add New Remote**.
9. 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.
10. 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.
11. Tap the **Disconnect** button to hangup the connection.

Important Note: Once connected, audio levels may need to be adjusted at either end because there is approximately 10dB difference between the output levels of each codec.

2.3.3 Connect to Mayah Sporty IP

Important Note: Firmware installed; Mayah Sporty codec firmware version 4.0.0.0; Tieline codec firmware version 1.6.48.

Programming the Mayah Sporty Attached to a LAN to Connect in Mono using SIP

1. Attach a LAN cable and any audio inputs to the Sporty codec.
2. Apply power to the Mayah Sporty.
3. Press the **F4/ON** button to return to the Home screen.
4. Press **F2 [Codec1] > F3 [Setup] > F1 [Interface].**
5. Use the up/down navigation arrow buttons to select **interface**, then press the **OK** button to view a drop-down menu and set this to **Ethernet** if it is not already selected.
6. Use the up/down navigation arrow buttons to select **protocol**, then press the **OK** button to view a drop-down menu and set this to **SIP** if it is not already selected.
7. Press **F2 [Codec] > F3 [Setup] > F2 [Quality]** to present 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. Tap the **Advanced** check box at the bottom of the screen to display the full menu.

10. 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.
11. Ensure that port **5060** is entered into the port number box.
12. Tap **RTP IP Port** and enter port **5004** into the port number box.
13. 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.

Programming a Tieline G3 Codec to Connect using SIP

1. Attach a LAN cable and any audio connections to the Tieline codec.
2. Press **SOFTKEY 2 [Profile] > [ManDflt MonoPgm] > [Really Load – Select YES]** to select the mono profile.
3. Program the Tieline codec to be in SIP mode by selecting **[Menu] > [Configuration] > [IP1 Setup] > [Session Type] > [SIP].**
4. Select the **SIP** connection on the Tieline codec LCD screen and press **SOFTKEY 3 [Wiz]** to configure the Tieline codec using the IP Configuration Wizard.
5. Select either the G.711 or G.722 algorithms over a public internet connection or WAN. Raw audio (uncompressed) may be selected over a private LAN.

Important Note: If you are in the USA or Japan set your country code to the correct country in the Tieline codec by selecting **[Menu] > [Configuration] > [System Settings] > [Country].** This selects G.711 µ-law for North America and Japan, whereas a-law is prevalent in other regions of the world.

Dialing the Connection using the Tieline Codec

1. Tieline recommends dialing from the Tieline codec to the Comrex Access codec if possible because it is simpler to program. To connect enter the IP address for the Access codec (private IP address over a LAN or public IP address if connecting over a WAN) and press **ENTER/DIAL** to connect to the Comrex Access codec.
2. Press the **HANGUP** button to disconnect the connection.