IP and 3GIP
Internet Streaming
Reference Manual

Software GUI Interface for the TLR300B

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SAFETY NOTICES and WARNINGS

THUNDERSTORMS and LIGHTNING
DO NOT USE Tieline codecs during thunderstorms and lightning.

You may suffer an injury using a phone, Tieline codec, or any device connected to a phone during a thunderstorm.

This can lead to personal injury and in extreme cases may be fatal.

Protective devices can be fitted to the line, however, due to the extremely high voltages and energy levels involved in lightning strikes, these devices may not offer protection to the users, the Tieline codec and equipment connected to the codec.

Secondary strikes can occur. These secondary strikes are induced by lightning strikes and also produce dangerously high currents and energy levels. You only need to be near an object struck by lightning to lead to personal injury or damage to equipment. e.g. if located near a lighting tower at a sports facility, water features and drains on golf courses you will be affected by these secondary strikes.

Damage to personnel and Tieline codecs may occur during thunderstorms, even if the codec is turned off but is connected to the phone system or the power.

ANY DAMAGE TO A TIELINE PRODUCT CAUSED BY LIGHTNING or an ELECTRICAL STORM WILL VOID THE WARRANTY.

WARNING: DIGITAL PHONE SYSTEMS
DO NOT CONNECT YOUR Tieline CODEC TO A DIGITAL PHONE SYSTEM. PERMANENT DAMAGE MAY OCCUR!

If you are unfamiliar with any facility, check that the line you are using is NOT a digital line. If the Tieline codec becomes faulty due to the use of a digital phone system, the WARRANTY IS VOID.

Disclaimer
Whilst every effort has been made to ensure the accuracy of this manual we are not responsible for any errors or omissions within it. The product specifications and descriptions within this manual will be subject to improvements and modifications over time without notice, as changes to software and hardware are implemented. Please visit our website at www.tieline.com

PLEASE READ OUR SOFTWARE LICENSE BEFORE USING THIS PRODUCT
Section 1. IP & 3GIP – A New Frontier in Broadcasting

Welcome to the Tieline IP and 3GIP reference manual and the new frontier in broadcasting. High-speed 3G data networks are changing the face of broadcasting and Tieline is at the forefront of these technological changes – creating solutions for the challenges they present.

As broadcasting moves inexorably towards predominantly packet-switched networks (rather than traditional circuit switched networks), Tieline is providing ground-breaking solutions for IP and 3GIP – while maintaining connectivity to existing communication infrastructure. This provides the best of both worlds and ensures your investment in Tieline products maximizes the return on your investment.

Make sure you have the latest IP software. Tieline’s new firmware release addresses IP network congestion and changing link quality when broadcasting over IP. Version 1.6.xx brings you innovative and cutting edge IP connectivity that is reliable, simpler to connect to and which reacts intuitively to network conditions. Tieline’s new automatically adaptive jitter buffer, in conjunction with forward error correction, helps you master packet-switched network connections – it really is simple!

1.1. About this Manual

This manual is divided into four sections. The first section is designed to get you connected quickly and contains four initial parts with instructions on how to quickly configure both studio and field unit codecs for IP broadcasting, as well as quick start connection guides for wired and wireless IP connections. These four parts are titled:

☑ IP streaming quick start studio setup guide;
☑ IP streaming quick start field unit setup guide;
☑ 10 simple steps to connect Tieline codecs (wired IP); and
☑ 3GIP quick start connection guide (wireless IP).

Most users will be satisfied with this information to successfully connect over IP and 3GIP networks. Following these elements are three complete reference sections that fully explain in detail all the technical elements of:

☑ IP streaming over wired studio and field unit codec IP connections;
☑ Wireless 3GIP streaming using Tieline 3G modules or USB modules with either a 3G cell-phone or USB modem attached; and
☑ IP streaming using SIP connections.

For advanced connection information and other useful IP broadcasting technical detail, please consult these reference sections within this document. Please note that Appendix 1 discusses IP line quality and firewalls with useful troubleshooting tips.
Important Configuration Note:
Please note that if you are IP streaming for the first time you will need to enable your codec for IP streaming by entering the code that you have purchased from Tieline. To do this select Menu > Configuration > Advanced Menu > IP Streaming > Enable. Then enter the code and press “OK” or “ENTER/DIAL” to input the code. Alternatively, you can trial IP streaming for 45 days via Menu > Configuration > Advanced Menu > IP Stream Demo > Enable.

The Latest Software = Best Performance
Tieline recommends that all users of IP and 3GIP networks use firmware version 1.6.xx in codecs as this will guarantee the ability to connect using auto jitter buffer settings. It will also automatically upgrade all USB modules to version 1.0.9, which supports UMTS, HSDPA & EV-DO 3G connections.

1.2. Software Updates in v.1.6.xx Codec Firmware
Following is a summary of the new and updated features contained in the “IP & 3GIP Streaming Reference Manual”, as a result of changes contained in release 1.6.xx firmware.

- Updates to how to connect a codec for IP in a studio using a static IP address;
- Updates to how to connect a codec for IP in the field using DHCP addresses;
- Updates to how to connect a codec over 3GIP using new codec menus;
- 3G Modules available for GSM, GSM Voice, UMTS, EVDO and HSDPA connections;
- Codec interoperability using SIP;
- Addition of the high quality, low delay Music Plus algorithm;
- Support for IP dial/answer without using session data;
- Permanent display of signal strength using 3G modules;
- Information about how using v.1.6.xx software guarantees the ability to use auto jitter buffer over IP/3GIP networks;
  - Also how if dialing to a lower software version than v.1.6.xx jitter buffer defaults to the default fixed setting of 500ms.
- Jitter buffer software changes;
  - New Auto Jitter Buffer use;
  - 5 new settings for auto jitter buffer;
  - 4 stages to jitter buffer when dialing and connecting;
  - Auto jitter buffer and how it works adaptively with FEC by measuring FEC on a connection and adjusting the jitter buffer appropriately to suit;
Full explanation of the "Connection Details" screen and the elements within it, including:

- How to use the new "Loss; Empty; Late; FECd" indications in the "Connection Details" LCD screen to determine the reliability and optimum IP jitter buffer and FEC settings;

How to order the right 3G data plan;

3G Antennae: how and what to select for the module purchased, i.e. EV-DO versus UMTS/HSDPA

USB module use:

- How v.1.6 version software automatically upgrades v.1.0.2 and v.1.0.4 USB software to v.1.0.9;

- Upgrades are performed; when firmware is upgraded and a USB module is in a codec; or subsequently when a module is inserted into a codec - a screen appears while the upgrade is performed and it takes about 10 seconds to perform.

- Use of USB modems and USB modules to connect over 3G.

Programming a new network into a codec using the "Custom Access Point" setting in the GSM LL/GSM/USB-3G tab in ToolBox.

Sending data using the "encode only" and "decode only" functions.

3G idle timeout feature added to minimize data costs.

IP Dialing error messages when dialing:

- To an "incompatible jitter buffer" device

- Using the Raw algorithm where jitter buffer is disabled automatically.
1.3. IP Streaming Quick Start Studio Configuration

This guide is intended to help you configure your Internet connection and Tieline codec in the studio to enable incoming calls over IP from a remote Tieline codec. It is assumed that you have an understanding of your network, its devices and configuration. In the studio you need to:

- Order a dedicated static public IP address for the codec;
- Program the IP connection and static IP address details using the IP wizard; and
- Follow the simple 10 step connection procedure to dial and connect.

1.3.1. Steps for Programming Studio Network Connectivity

- Order a high-speed broadband service from your Internet Service Provider (ISP) & do not share this connection with other devices. Order a ‘static public IP address’ from your ISP where possible.
- Get your network administrator to ‘install’ the static public IP address and perform Network Address Translation (NAT) between the public Internet and your Local Area Network (LAN).
- Connect an active RJ-45 LAN cable for the broadband service to the LAN port on the rear of the codec.
- If there is an active connection on the LAN cable, the green LED underneath the LAN port will illuminate and the orange LED will flash steadily.
- By default, the Tieline codec is set to DHCP. Reprogram static IP address settings on the codec by scrolling to the IP connection you wish to connect with on the main codec LCD screen.
- When the square brackets [ ] surround the connection press SOFTKEY 3 Wiz. Use the default settings of [Algorithm → Music←] > [Audio Bitrate →9600←] > [Local FEC Percent → Off←] > [Remote FEC Percent → Off←] > [Jitter Buffer Type → Auto Jitter Adapt←] > [Auto Jitter Priority → Best Compromise←] > [TCP Session Port → 9002 or 9012←] > [Audio Port → 9000 or 9010←] > [Ethernet Link → Auto←] > [IP Setup → Static←] > [IP Address → Enter Number←] > [Subnet Mask → 255.255.255.0←] > [Default Gateway → Enter Number←] > [Auto Reconnect → Disable←] > [RTP Configured → OK←]
- Please note: press the CLEAR button on the codec keypad to delete the existing number and then enter a new static IP Address, Subnet Mask and Default Gateway (check with your IT administrator if you are unsure).
- Check the codec’s IP Address by selecting Menu > Unit Details > IP Address. Please Note: Depending upon how your network is configured, it may also be possible to simply connect your Tieline Codec directly into your DSL/ADSL modem/router and receive a Public address from the router. A public address typically looks like 203.35.196.135 and is out of the ranges: 10.0.0.1 – 10.255.255.255,
Once you are set up in the studio you are now ready to receive an incoming call from a remote codec over the Internet. Unless the remote codec has a public IP address assigned to it and you know what the number is, you will always have to dial the public IP address of the studio from the field codec.

Once the settings have been configured on both the field and studio codecs follow the instruction titled “10 Simple Steps to Connect Tieline Codecs” to Create a new connection.

1.4. IP Streaming Quick Start Field Unit Configuration

Normally, in the field you will only need to attach your Tieline codec to a LAN with access to the Internet. You will not need to know the public IP address or have to configure network address translation. Plug in the Tieline codec and check that it has been assigned a private IP address. Then you should be able to simply enter the static public IP Address of the studio codec and dial to connect.

1.4.1. Programming Field Codec Network Connectivity

- Connect an active RJ-45 LAN cable for a broadband service to the LAN port on the rear of the codec.
- If there is an active connection on the LAN cable, the green LED underneath the LAN port will illuminate and the orange LED will flash steadily.
- By default, the Tieline codec is set to DHCP. In this mode a DHCP server (if available) usually assigns a private IP Address to connected devices automatically. Check to see if the codec has been allocated a DHCP IP Address by selecting Menu > Unit Details > IP Address. They are generally in the ranges: 10.0.0.1 – 10.255.255.255, 169.254.0.0 – 169.254.255.255, 172.16.0.0 – 172.31.255.255 and 192.168.0.0 – 192.168.255.255 and are assigned by network DHCP servers. Note: If an address has not been assigned, contact the venue’s network administrator about programming a static IP address. The procedure for programming this is similar to programming a studio codec.
- If an IP address is visible then check settings on the codec by scrolling to the IP connection you wish to connect with on the main LCD screen.
- When the square brackets [ ] surround the connection press SOFTKEY 3 Wiz. Use the default settings of [Algorithm → Music] > [Audio Bitrate → 9600] > [Local FEC Percent → Off] > [Remote FEC Percent → Off] > [Jitter Buffer Type → Auto Jitter Adapt] > [Auto Jitter Priority → Best Compromise] > [TCP Session Port → 9002 or 9012] > [Audio Port → 9000 or 9010] > [Ethernet Link → Auto] > [IP Setup → DHCP] > [Auto Reconnect → Disable] > [RTP Configured → OK].
- Double-check the codec’s IP Address by selecting Menu > Unit Details > IP Address.
1.5. 10 Simple Steps to Connect Tieline Codecs

This section outlines the Quick Start procedure for connecting codecs using IP. Unless the remote codec has a public IP address assigned to it and you know what the number is, you will always have to dial the public IP address of the studio from the field codec. I.e. always dial from the field codec to the studio codec over the Internet.

**Warning:** If you connect over IP and use auto jitter buffer then both codecs must have firmware v.1.6.xx installed.

Use the black rotary MENU SELECTOR (MS) to scroll through menus and press it to select menu items. If more detailed connection information is required, please see the ‘Quick Start’ section of each codec’s reference manual for more information.

**Step 1.** Disconnect power from the codec before installing any module into it.

**Step 2:** Plug power into the codec and attach any POTS, ISDN or Ethernet lines that are required.

**Step 3:** Turn on power to the codec and select [Menu] by pressing SOFTKEY 4. Then select [Load profile] to choose the type of connection to connect with (i.e. default profiles or any Custom Profile). Select the profile you want from the menu and press SOFTKEY 2 to load the profile.

**Step 4:** Use the black rotary MS to scroll to the connection you are using, i.e. [IP1 Enter#] etc, until it is surrounded by the square brackets [ ]. (Note: If “Unavailable” is displayed there is a connection issue that needs investigating.)

**Step 5:** Plug your microphones and/or music sources into the codec and adjust the input gain, phantom power (default is off) and other audio settings by pressing SOFTKEY 1 Aud. (If you are not using a microphone at the codec you are dialing from go to step 7).

**Step 6:** The default input level setting is Line Level. To adjust input gains press SOFTKEY 1 with Aud displayed above it and scroll to and select [Input Gains]. Select the input gain setting you require for each individual input or select [All Inputs] to change all inputs simultaneously. Press the CLEAR button on the keypad twice to return to the main LCD screen. **WARNING:** Phantom power of 15 volts is always switched on for the TLR300B rack mount codec analog microphone input.

**Step 7:** Scroll until the square brackets [ ] surround the connection you will be dialing (e.g. IP1 Enter# ) and type the number/IP address for the
connection via the keypad. (Note: the “*” key on the codec keypad inserts a period into an IP address).

Step 8: Press the ENTER DIAL button on the codec to dial and connect. To negotiate higher bit-rates press “F2” then “3”; for lower bit-rates press “F2” then “9”.

Step 9: Repeat steps 7-8 if dialing a second connection.

Step 10: On an i-Mix G3 press the yellow CUE button to send audio over the communications channel. If you are using a field unit COMMANDER G3 codec, once both channels are connected hold down the MS for 2 seconds and a secondary activation menu will appear along the bottom of the screen. You will see CUE1 and CUE2 above HOTKEYS 2 and 3. (Please note that rack unit codecs and the TLG3 GUI rack mount codec control software have dedicated CUE buttons so you will not need to do this). Pressing the CUE key on either of the 2 microphone inputs will route audio from these inputs to the off-air bi-directional communications channel only. Audio being sent will be heard in the right side of both headphone outputs. Communications audio will be displayed on PPM 2. To return to the main menu hold down the MS for 2 seconds, or it will automatically return to the main menu after two minutes. For more information on the i-Mix G3 phone coupler, please see the codec reference manual.

If you are unable to achieve a connection using these instructions, please refer to the detailed information relating to IP and 3G/IP connections that follows in this reference manual. Alternatively, contact your IT Network Administrator for assistance configuring your network or contact Tieline on support@tieline.com for support.
1.6. Quick Start Procedure for 3G IP Connections (version 1.6.xx firmware or higher)

Connecting your codec over 3G is very similar in principle to connecting over IP. The only difference is that you are wirelessly connecting to your ISP instead of connecting via a LAN. Connect to your ISP/cell-phone provider and then use the Quick Start connection procedure for your preferred connection profile (i.e. mono, stereo, mono/IFB and dual mono program) over IP.

**Very Important Warning:**

**Tieline** CDMA EV-DO 3G modules don’t use SIM cards and need to be activated and provisioned in order to connect to cell-phone networks in the U.S.A. Use the procedure outlined in this manual to program your module before use over these networks.

**Important Note:**

As a factory default, GSM/3G settings are programmed for **AUTODETECT**. If you plug a 3G or USB module (with a 3G phone connected) into your codec it will program it to operate in 3GIP mode by default.

If you use a GSM module or plug a GSM cell-phone into the serial port of your codec it will program it to operate in GSM CSD mode by default.

If you wish to use a 3G phone in GSM mode, you will need to change the **Wireless Network** setting in the **GSM/3G Wizard** to either **GSM CSD** or **GSM HSCSD** - depending on the connection you wish to use.

1. Insert a **Tieline** 3G module (with a SIM card installed for UMTS/HSDPA networks or provisioned & activated for EV-DO networks) into your codec and then power up the codec.
2. Scroll to the **3GIP1** connection with the codec **MENU SELECTOR** and press **SOFTKEY 3 Wiz** and then select **SOFTKEY 4 OK**. Next select [**Wireless Network →3G/UMTS IP**] → [**3G/UMTS IP Network →select your network**] → [**Auto Reconnect →Disable**]. Select **SOFTKEY 4 OK** to complete configuration and return to the main LCD connection screen.
3. If **3GIP1> Prs Entr** is displayed, press **ENTER/DIAL** to connect to your 3G network. If **3GIP1> Enter #** is displayed, dial the SIM card cell-phone number using the codec keypad and then press **ENTER/DIAL** to connect to your 3G network. This is because some cell-phone networks require you to dial this number to connect.
4. Once you have connected the codec connection will display **3GIP1> Cntd Goto IP**
5. Now scroll to **IP1** on the main codec LCD screen and connect using the Quick Start IP profile you have selected, i.e. mono, stereo, mono/IFB or dual mono. Type the IP address of the codec you are dialing. (Note: Use the * or # button on the codec keypad to enter the periods (.) in the IP address).
6. Press the **ENTER DIAL** button on the remote codec’s grey keypad to begin dialing. In many situations it is only possible to dial from the remote codec to...
the local codec with IP connections because only the studio codec is using a public IP address.

7. Try to maintain a link quality (LQ) reading for your connection of between 70% and 100%. To negotiate higher bit-rates press “F2” then “3”; for lower bit-rates press “F2” then “9”. For best performance, the dialing codec should be used to renegotiate connection bitrates up and down. If you hear audio drop-outs the current bit rate cannot be sustained and should be renegotiated down. To disconnect, hang up your IP connection and then hang up the 3G connection.

8. To disconnect, hang up your IP connection and then hang up the 3G connection.

9. To change other 3GIP settings select SOFTKEY 4 OK and scroll to [Configuration] > [GSM/3G Setup] > [3G Module].

Note on Cell-phone Use with Tieline USB Modules:
Turn on your 3G cell-phone and wait 30 seconds before you connect the phone to the USB module via its USB cable (this cable should be available from your cell-phone manufacturer). Within 30 seconds a 3GIP connection should appear on the main codec connection LCD screen.

Cell-Phone Warnings
If you are unsure about the compatibility of a particular cell-phone, please see the ‘How to connect over 3G with a USB Module & Cell-phone’ section of this manual, or contact Tieline at support@tieline.com for more information.

1.7. What to do if a Network is Unavailable in the Codec
An expansive list of cell-phone networks covering 52 countries is currently programmed into Tieline codecs. Due to the number of new cell-phone networks coming on stream all the time, it is possible that a network you wish to connect to may not be listed in the codec.

In this scenario it is still possible to program a ‘custom access point’ into your codec for your preferred network. To do this you will need a PC with ToolBox software installed in order to program the access point and then create a .cdc file to load onto the codec.

Before programming the access point some information is required from the network provider you are connecting to. These include:

1. An access point address;
2. A dial string;
3. An account number; and
4. A password.

Once you have been supplied with these details open ToolBox and go to the GSM LL/GSM/USB-3G tab and complete the following:
1. Check the Use Custom Access Point check-box in the GSM 3G/UMTS IP Network menu;
2. Enter the Access Point address;
3. Enter the Dial String;
4. Enter the Account details;
5. Enter the Password for the network;
6. Enter a User-Defined Label to identify the custom access point.

Save the settings into a profile, load the configuration file onto the codec and you are ready to connect to the network. The network you have programmed should appear in the 3G/UMTS IP Network codec menu at the bottom of the available networks.

**Custom Network Tip:**
If you find that there are multiple networks that you wish to connect to that are not listed in the 3G/UMTS IP Network codec menu, simply create a new profile for each network and enter the relevant ‘custom access point’ details into each new profile. Save these profiles together in a *.cdc file and load this file onto the codec you will be using over these networks.

1.7.1. Tell us about a New Network

**Tieline** will add new networks to each release of codec software. Email support@tieline.com.au with the name of any network you want added along with:

1. The access point address;
2. The dial string;
3. The account number; and
4. The password.
1.8. Uplink Bandwidth Table

Following is the UDP Uplink Bandwidth table that can be used as a rule-of-thumb for configuring all IP connections. Please note that the jitter buffer data in the table is relevant only for manually configured jitter buffer settings.

The following table sets out in detail what your codec settings should be (as a rule of thumb), based on the following variables:

- Different broadband DSL (ADSL) data uplink rates;
- The algorithm that you have selected;
- The codec audio connection bit rate setting;
- Forward Error Correction settings;
- Jitter-buffer millisecond settings; and
- The profile you wish to select (i.e. Mono, Stereo, Dual Mono or Mono IFB.)

<table>
<thead>
<tr>
<th>Codec Settings</th>
<th>33.6kb Dialup</th>
<th>64kb DSL</th>
<th>128kb DSL</th>
<th>256kb DSL</th>
<th>512kb DSL</th>
<th>1,024kb DSL</th>
<th>Wireless Wi-Fi</th>
</tr>
</thead>
<tbody>
<tr>
<td>Audio Bitrate</td>
<td>9.6 - 14.4kb</td>
<td>9.6 - 16.8kb</td>
<td>9.6 - 28.8kb</td>
<td>9.6 - 64kb</td>
<td>9.6 - 128kb</td>
<td>9.6 - 128kb</td>
<td>9.6 - 16.8kb</td>
</tr>
<tr>
<td>Algorithm</td>
<td>Voice G3</td>
<td>Voice G3</td>
<td>Voice G3 or Music</td>
<td>Music</td>
<td>Music or Music Plus</td>
<td>Music or Music Plus</td>
<td>Voice G3</td>
</tr>
<tr>
<td>Forward Error Correction</td>
<td>Off</td>
<td>20% - 33%</td>
<td>20% - 33%</td>
<td>20% - 50%</td>
<td>20% - 100%</td>
<td>20% - 100%</td>
<td>100%</td>
</tr>
<tr>
<td>Jitter Buffer Ms</td>
<td>500ms</td>
<td>250 - 500ms</td>
<td>200 - 350ms</td>
<td>100 - 300ms</td>
<td>100 - 300ms</td>
<td>100 - 300ms</td>
<td>250 - 750ms</td>
</tr>
<tr>
<td>Mono Profile</td>
<td>✔️</td>
<td>✔️</td>
<td>✔️</td>
<td>✔️</td>
<td>✔️</td>
<td>✔️</td>
<td>✔️</td>
</tr>
<tr>
<td>Stereo Profile</td>
<td>❌</td>
<td>❌</td>
<td>❌</td>
<td>✔️</td>
<td>✔️</td>
<td>✔️</td>
<td>❌</td>
</tr>
<tr>
<td>Dual Profile</td>
<td>❌</td>
<td>❌</td>
<td>❌</td>
<td>✔️</td>
<td>✔️</td>
<td>✔️</td>
<td>❌</td>
</tr>
<tr>
<td>Mono/IFB</td>
<td>❌</td>
<td>❌</td>
<td>❌</td>
<td>✔️</td>
<td>✔️</td>
<td>✔️</td>
<td>❌</td>
</tr>
</tbody>
</table>

Table 1: UDP IP Broadband Uplink Bandwidth Table

Please note: Tieline recommends that your broadband service in your studio is not shared with other users as this will decrease the available bandwidth for your broadcast signals and may cause instability.
1.9. Checklist for Obtaining High Quality IP Connections

The following 10 tips are provided to help obtain the best possible IP connection between two codecs, without paying for QOS (Quality of Service).

1) Always use the best quality Internet Service Provider (ISP). Tier 1 service providers are best as their infrastructure actually makes up the Internet ‘backbone’. Wikipedia lists the major service providers that make up the Internet backbone at: [http://en.wikipedia.org/wiki/Internet_backbone](http://en.wikipedia.org/wiki/Internet_backbone). In Australia Telstra is equivalent to one of these service providers.

2) You will get the best quality connection if both the local (studio) and remote codecs use the same Internet service provider. This can substantially increase reliability, audio bandwidth and reduce audio delay. Using the same service provider nationally can give better results than using different local service providers. This is especially true if one of the service providers is a cheap, low-end domestic service provider, which buys its bandwidth from other ISPs. Second and third tier providers sublease bandwidth from first tier providers and can result in connection reliability issues due to multiple switch hops. We also highly recommend using First Tier ISPs if connecting two codecs in different countries.

3) Sign up for a business plan that provides better performance than domestic or residential plans. Business plans typically have a fixed data limit per month with an additional cost for data beyond that limit. In addition, Service Level Agreements (SLA) will often provide better support and response times in the event of a connection failure. Domestic plans are often speed-limited or “shaped” when usage exceeds a predefined limit. These plans are cheap but they are dangerous for streaming broadcast audio.

4) Ensure that the speed of the connection for both codecs is adequate for the job. The minimum upload speed recommended is 256 kbps for a studio codec and 64 kbps for a field unit connection.

5) Use good quality equipment to connect your codecs to the Internet. [Tieline](http://www.tieline.com) successfully uses Cisco® switching and routing equipment.:
   a. If you are using a DSL or ADSL connection make sure you purchase a high quality modem that can easily meet your speed requirements. This is especially important if you are over 4 kms from an exchange.
   b. If you have multiple codecs connected to a local area network (LAN) please ensure that your network infrastructure is designed for media streaming and not domestic usage. [Tieline](http://www.tieline.com) has tested several cheap 8-port switches that lose more packets between local computers than an international IP connection between Australia and the USA!

Please Note: You should be able to stream audio between two codecs on your LAN and get ‘link quality’ readings of L99R99. If you see anything less than this then you should get a network engineer to investigate the issue.

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1 Cisco is a registered trademark of Cisco Systems, Inc. and/or its affiliates in the U.S. and certain other countries
c. If using a wireless connection ensure that the antenna signal strength received is strong. The type of antenna used and the amount of output gain also affects connection quality.

d. If you are using a 3G phone please make sure the battery is fully charged and that you are close to a cell-phone base station.

6) Once your Internet connection is installed at the studio check that the connection performance is approximately what you ordered and are paying for. A connection can perform below advertised bit rates if:

   a. There is an error in ISP configuration;
   b. There is an error in modem configuration;
   c. There is a poor quality line between the studio and the exchange;
   d. There are too many phones or faxes connected to the phone line; or
   e. Line filters have been connected incorrectly.

You can test the Internet connection speed by connecting a PC to the Internet and using http://www.speedtest.net/index.php. If the bandwidth detected is low then something is wrong. Get it fixed before going live!

7) Use a dedicated DSL/ADSL line for your codecs. Do not share a link with PCs or company networks. The only exception to this rule is if an organisation has network equipment and engineers that can implement and manage quality of service (QOS) on its network.

8) Use UDP as the preferred audio transport protocol. TCP generally results in lower bitrates and random drop-outs of audio over the Internet. Only use TCP if UDP is blocked by firewalls and you are unable to connect.

9) When using UDP ensure the total bit rate (audio bit rate plus header bit rate) is no more than 80% of the ISP connection rate. IP headers require around 20 kbps in addition to the audio bit rate. For example, with a 64 kbps connection the audio bit rate should be (64-20) x 0.8 = 31.2 kbps or lower. For TCP we suggest a limit of 50% or less.

10) Wireless IP connections can easily become congested and result in packet loss and audio drop-outs. It is very difficult to guarantee connection quality when there is no way of knowing how many people are sharing the same wireless connection. Examples of wireless connections include:

   a. WiFi connections within your LAN (These are unmanaged connections and should not be used to distribute audio when setting up IP connections);
   b. GSM CSD and HSCSD connections; and
   c. EV-DO, 3G UMTS or HSDPA phone connections.

Note: Be careful when using cell-phone connections at special events where thousands of people have mobile phones. This can result in poor quality connections and audio drop-outs if cell-phone base stations are overloaded.
Complete the following check list and aim for a score of at least 8 out of 10 before going live.

<table>
<thead>
<tr>
<th>Number</th>
<th>Check</th>
<th>Result</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Using a reputable Tier1 ISP that’s part of Internet backbone.</td>
<td></td>
</tr>
<tr>
<td>2</td>
<td>The same ISP is being used for both codec connections.</td>
<td></td>
</tr>
<tr>
<td>3</td>
<td>The ISP Plan is a Business Plan or equivalent.</td>
<td></td>
</tr>
<tr>
<td>4</td>
<td>The ISP connection speed is adequate (256 kbps or higher).</td>
<td></td>
</tr>
<tr>
<td>5</td>
<td>Equipment is high quality and suitable for media streaming.</td>
<td></td>
</tr>
<tr>
<td>6</td>
<td>The ISP connection speed has been tested and is suitable.</td>
<td></td>
</tr>
<tr>
<td>7</td>
<td>The ISP connection is not shared with PCs or other devices.</td>
<td></td>
</tr>
<tr>
<td>8</td>
<td>UDP is being used as the audio transport protocol.</td>
<td></td>
</tr>
<tr>
<td>9</td>
<td>Only up to 80% of ISP connection bandwidth is being used.</td>
<td></td>
</tr>
<tr>
<td>10</td>
<td>There are no wireless connections being used.</td>
<td></td>
</tr>
</tbody>
</table>

1.10. How to Order the Right Plan for your 3G Service:

There are many data services offered by Telcos. At the time of writing this, HSDPA and EV-DO Rev A offer the highest bit rates and therefore the best opportunity for delivering stable high quality audio. Try to use these services from a reliable provider in your region.

One of the most expensive mistakes you can make is borrowing a 3G SIM card for a broadcast that will last a couple of hours. It is likely that this type of 3G plan is optimized for voice and not IP data. Don’t find out the hard way – it could be an expensive mistake! We recommend you purchase a plan that includes unlimited data for a fixed price per month. Then you can broadcast for as long as you need for a fixed price per month.

If this type of plan is not available, estimate the number of remote broadcast minutes/hours you need per month and buy a plan that bundles large blocks of data for one price. Some telcos also offer ‘timed’ or ‘minutes’ plans, which offer unlimited data for fixed amounts of time.

**Warning:** Some 3G network providers prohibit streaming multimedia of any kind on certain accounts. Also, some plans charge very high rates for data, or may ‘throttle’ or ‘shape’ your available bandwidth after a certain amount of data has been transferred. Check with your telco before subscribing to a plan.
1.11. Troubleshooting Network Device Settings

1.11.1. Port Settings

For your Tieline Codec to communicate over the public Internet an IP Address alone is not sufficient. The Internet Protocol pre-defines TCP and UDP protocol port numbers for specific functions.

Tieline codecs use TCP ports for setting up the communication session and UDP ports for streaming audio. While TCP ports are generally open, UDP ports are generally blocked by network devices which contain firewalls and will stop you delivering your audio. You need to configure your firewall to allow TCP and UDP protocols to pass through on the specific ports listed in the table below.

<table>
<thead>
<tr>
<th>Application</th>
<th>Session Port</th>
<th>Audio Port</th>
</tr>
</thead>
<tbody>
<tr>
<td>IP1</td>
<td>TCP 9002</td>
<td>UDP 9000</td>
</tr>
<tr>
<td>IP2</td>
<td>TCP 9012</td>
<td>UDP 9010</td>
</tr>
<tr>
<td><strong>Tieline</strong></td>
<td><strong>UDP 5550</strong></td>
<td><strong>UDP 5060</strong></td>
</tr>
<tr>
<td><strong>ToolBox &amp; TLG3GUI</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>SIP</td>
<td>UDP 5060</td>
<td>UDP 5004</td>
</tr>
</tbody>
</table>

Table 2: Default Tieline Port Allocations

Figure 1 illustrates the firewall port settings on a sample ADSL Internet router. Consult your Network Administrator if you are unsure how to do this.

**Important Note:** Appendix 1 of this reference manual contains a Tech Note which explains how to interpret and troubleshoot firewall problems using the LQ (link quality) display on a codec.
1.11.2. DMZ (De-Militarized Zone)
An alternative method of communicating over the Internet is to enter a Tieline codec’s IP Address into the DMZ of network devices. The DMZ opens a hole in the firewall only for the IP address of 1 device. All other devices behind the firewall remain completely protected. This will freely allow bi-directional communication over the network. Figure 2 below illustrates the DMZ settings on a sample ADSL Internet router.

![Figure 2: DMZ Configuration Example](image)

1.12. Troubleshoot IP Connections Using “Connection Details”
The Connection Details screen displays information that assists in appraising the performance of IP connections. Access this screen by pressing the ENTER/DIAL button after initiating dialing, or after connecting. The screen display is described in detail in the following sections.

1.12.1. Connection Status and Number Dialed
The first element at the top of the screen displays the connection, e.g. IP1, and its connection state – as well as time it has been connected for in days/hours/minutes and seconds. The states displayed include:

- ✔︎ “Connecting”: the connection is in the process of connecting;
- ✔︎ “Connected Active”: the connection is live and encoders and decoders are active;
- ✔︎ “Connected Stby”: a connection is available and in standby mode (e.g. failover connections);
- ✔︎ “Unavailable”: The connection is unavailable due to network connectivity or there is a problem, e.g. a cable is disconnected.
- ✔︎ “Idle”: There is no connection, usually displayed after disconnection.
- ✔︎ “Disconnected”: The connection has been disconnected.
The next element is the number dialed. In the following example it is the number for one of Tieline’s IP test codecs.

![Codec Settings Example]

1.12.2. Codec Settings
Scroll down further to display connection settings programmed into the codec. An example of the next three lines on the screen follows.

![Codec Settings Example]

The elements displayed include:
1. The encoders and decoders being used “E1:D1” in the previous image;
2. The actual bit rate of the connection (with “B” preceding the bit rate). This will differ from the setting displayed on the codec main connection screen. In this example “B274.2” kbps.
3. The algorithm being used over the connection is displayed – in this example “Music”.
4. The PPM meter settings – in this example “Matrix”.
5. The current active state of IGC is displayed. The options displayed include:
   a. “Off”: IGC is switched off in the codec, or
   b. “- - -”: IGC is switched on but not currently in use, or
   c. “On”: IGC is switched on and currently in use.
6. The number of dialing retries remaining, based on settings programmed into a codec, are listed as “Rtry Remaining”.
1.12.3. Checking Session Data, Jitter Buffer & IP Versions

Scroll down the Connection Details screen until you see the screen details displayed in the following image.

![Figure 3: Session Data Status Display](image)

1.12.3.1. Session Data Transfer

If “OK” is displayed then session data has been transferred successfully. Some of the other messages you may see in the Session Xfer section include:

- **Timeout:** If session data times-out this message will appear. This may occur if session data is switched off, when attempting to connect to a non-Tieline codec or if you are attempting to connect using GSM.

- **Wait:** This appears when session data transfer is being negotiated.

- **Error:** This appears when a software error has occurred. If you see this message attempt to connect again. If repeated attempts to connect fail, please contact support@tieline.com in order to troubleshoot the problem.

1.12.3.2. Jitter Buffer Displays

Current jitter buffer settings are displayed in the menu below the session data transfer section. The jitter buffer in the previous image is 480 ms and it is a fixed setting, i.e. auto jitter buffer is not enabled. This is indicated by the “F” character next to the millisecond value.

The jitter buffer state displayed may include any of the following:

1. **Stabilization period “a1”:** This is a few seconds at the start of a connection where no action is taken at all while the establishment of a stable connection means analysis of jitter data is not valid.

2. **Stage 2 “a2”:** A compatibility check to ensure the RTP connection is compliant and RTP clocks are synchronized enough to perform jitter analysis.

3. **Stage 3 “a3”:** If the compatibility check is successful, this is the analysis hold-off period. During a minute, the jitter buffer is held at a
safe, fixed value of 500ms while enough history is recorded to start jitter buffer adaptation.

4. **Stage 4 Live “A”:** This is where the codec determines it is safe enough to start broadcasting using the auto-jitter buffer level. We recommend running the codec for a few more minutes to obtain a more comprehensive history of the connection’s characteristics.

5. **Fixed “F”:** This state is displayed if the jitter buffer is fixed and in auto mode.

### 1.12.3.3. IP Version Display

The IP version in both codecs over a connection is displayed after the jitter buffer settings. Following is a table displaying the possible versions in a codec and the firmware versions they represent.

<table>
<thead>
<tr>
<th>Codec Version Number</th>
<th>Firmware Version Number</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>1.2.xx</td>
</tr>
<tr>
<td>2</td>
<td>1.3.xx</td>
</tr>
<tr>
<td>3</td>
<td>1.4.xx</td>
</tr>
<tr>
<td>4</td>
<td>1.6.xx</td>
</tr>
</tbody>
</table>

IP versions are all backward compatible in connecting but only version 1.6.xx software allows you to connect and use auto jitter buffer.

### 1.12.4. Packet Arrival Analysis

Details about the arrival times of packets over a connection are also displayed. This information can be used to determine how effective current codec connection settings are in relation to:

- Jitter buffer settings;
- Forward Error Correction; and
- Renegotiation of connection bit rates.

See the following table for information about the details displayed.
<table>
<thead>
<tr>
<th>Packet Analysis</th>
<th>Displays</th>
<th>Possible Causes</th>
<th>Possible Solutions</th>
</tr>
</thead>
</table>
| **Loss**        | Expected packets that failed to arrive. | • LAN/WAN congestion  
• Unreliable ISPs  
• Unreliable networks  
• Inferior IP hardware | • Renegotiate connection bit rate downwards  
• If link quality good add or increase FEC as required  
• Assess ISP’s QOS if very bad performance |
| **Empty**       | Indicates how often the jitter buffer ‘reservoir’ empties causing loss of audio. | • High number of packets being lost or arriving late  
• Signal dropouts using 3G cell networks  
• Renegotiation causes the jitter buffer reservoir to empty | • Once could be an anomaly – assess lost & late packets  
• If many lost packets network is unreliable – renegotiate bit rate and /or FEC down  
• If many late packets increase jitter buffer |
| **Late**        | The number of packets that arrive late and after audio play out. | • Network congestion  
• Jitter Buffer depth is too low | • Auto-jitter buffer will adjust automatically  
• For manual jitter buffer settings increase jitter buffer depth 50-100 ms & reassess (if only a few packets arrive late over time, audio repairs will be automatic and may not require buffer changes). |
| **FECd**        | Indicates the number of FEC repaired packets if FEC active. | • Packets have been lost or corrupted over the network | • Assess audio quality & the number of FEC repairs – if many packets are being ‘lost’ perhaps reduce FEC &/or renegotiate bit rate down. |

If settings such as the jitter buffer, FEC or the connection bit rate are changed, it is best to assess a minute of connection performance in preference to the 10 minute historical performance. This is because the 10 minute performance will include historical data, which includes connection settings that my no longer be relevant. It is only when a connection is hung up that ‘packet arrival history’ is cleared.

### 1.13. IP Dialing Error Messages

Following are some error messages that may be displayed on a codec when dialing over IP.

#### 1.13.1. Incompatible Device

If the ‘remote incompatible’ message is displayed then the device being dialed is either:
A Tieline codec running incompatible software lower than v.1.6.xx; or
An incompatible device or competitor codec that is unable to use auto jitter buffer.

![Figure 4: Incompatible Device Message](image1)

1.13.2. Raw Audio Jitter Buffer Disabled
If a codec is programmed to connect using the Raw algorithm then it will automatically disable auto jitter buffer if it is enabled.

![Figure 5: Jitter Disabled Message](image2)

1.14. Troubleshooting Tip
Every computer (and codec) has an internal IP address of 127.0.0.1, which when dialed provides a loopback test. This will allow audio to be looped from the encoder to the decoder and you can listen to this audio via the headphone output on the codec. If this test works, it is likely that you have a network problem.
Section 2.  IP Streaming Configurations

Using IP you can connect codecs attached to different computers over a private Local Area Network (LAN), or over different public networks such as the Internet. In addition, you can use a 3G cell-phone to connect wirelessly over IP.

Put simply, a computer network is a system for communication between computers and a LAN is a network covering a small local area. Networks can connect via hard-wired Ethernet connections or a combination of hard-wired connections and wireless technology.

In this reference manual we discuss the basics of IP connections and provide information pertaining to the factors affecting the quality of IP connections. We will also discuss IP addressing configurations for studio based IP codecs as well as remote broadcast codecs.

If you feel you have a good understanding of IP technology, you can go straight to the section titled IP Settings on the Codec, which discusses configuring your Tieline codec for IP. Within this section the codec menu structure is displayed and there are explanations of how to configure your codec correctly. Used in conjunction with the IP (and 3G IP) Quick Start setups for Mono, Mono/IFB, Stereo and Dual Mono profiles, it will equip you to connect using IP successfully.

It is useful to have a basic understanding of how IP networks operate before attempting to connect your IP codecs.

2.1. Quality versus Reliability

The bandwidth of your connection can provide broadcasting limitations. It is also important to be aware that there can be a trade-off between the quality and the reliability of an IP connection, depending on various factors.

Your connection will only be as good as the minimum free bandwidth that your uplink and downlink can provide. We recommend that your broadband service is not shared with other users as this will decrease the available bandwidth for your broadcast signals and may cause instability. Be wary of wireless networks in the studio and the field, which are prone to interference from many different environmental elements over which the codec will have no control.

2.2. Public versus Private IP Addresses

An IP address is a unique number that allows devices to communicate over networks and the Internet using the Internet Protocol standard. There are two types of IP addresses – public and private.

IP address numbers currently range from 0.0.0.0 to 255.255.255.255 and there is currently a shortage of IP addresses, so every device in the world is unable to have a unique IP address.
Due to the shortage of IP addresses, certain IP address ranges have been allocated for private use. These numbers can be used by anyone on a LAN but computers or devices using these numbers are unable to connect directly over the Internet without using Network Address Translation and a public IP address.

Conceptually, IP addresses operate similarly to phone numbers because an IP number can be public or private. For example, a standard PBX telephone system allows people to call you on a single public telephone number and performs the translation and routing of the public number into a particular private PBX extension. Private and public IP addresses operate in a similar way to private and public phone numbers - so similar dialing principles apply.

If you want to dial a codec with a private IP address you will require network address translation (NAT). NAT allows a single device, such as a broadband router, to act as an agent between the public Internet and a local private LAN. Usually this will be set up at the studio end so you will dial into the studio from the remote codec. Following is a table describing the different types of IP addresses you may encounter and how they originate.

<table>
<thead>
<tr>
<th>Type of IP Address</th>
<th>How the IP Address is Allocated</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Static Public IP Address</strong></td>
<td>Internet Service Providers (ISPs)</td>
<td>ISP’s allocate a static public IP address to allow network devices to communicate with each other over the Internet. It works like a public telephone number and will allow your remote codec to call your studio codec over the Internet.</td>
</tr>
<tr>
<td><strong>Dynamically Assigned Public IP Address</strong></td>
<td>Internet Service Providers (ISPs)</td>
<td>ISP’s usually allocate dynamically (automatically) assigned public IP addresses to allow network devices to communicate with each other over the Internet. (Not recommended for studio installations because each time you connect to your ISP the IP address can change).</td>
</tr>
<tr>
<td><strong>Dynamically Assigned Private IP Address</strong></td>
<td>DHCP Server from your own private LAN network.</td>
<td>A DHCP server-allocated IP address that is automatically assigned to a device on a LAN to allow it to communicate with other devices and the Internet. This address can change each time a device connects.</td>
</tr>
<tr>
<td><strong>Static Private IP Address</strong></td>
<td>LAN Administrator</td>
<td>A network administrator-allocated static address which is programmed into a device to allow it to connect to a LAN. Often a security measure to only allow access to devices approved by a network administrator.</td>
</tr>
</tbody>
</table>

Table 3: IP Address Types
If you want to dial a codec with a public IP address you simply dial the IP address to connect. For more detailed information about programming IP addresses into your codecs, please see the sections in this manual that describe studio and remote codec configuration for IP.

2.3. Factors Affecting IP Connection Reliability

There are many factors that will affect the stability of an IP connection. Some of these include:

- Whether the connection is a wireless connection;
- Whether a connection is local, national or international;
- Whether a connection is a TCP or UDP connection;
- Codec jitter buffer and Forward Error Correction (FEC) settings; and
- Whether a connection is shared with other devices like computers.

Tieline recommends using wired connections and where possible you should attempt to use a broadband connection that is not being shared.

2.3.1. Some Background on Data Packets

Circuit switching, as used in GSM CSD and HSCSD connections, creates a dedicated connection between two nodes (in this case cell-phones) to send data exclusively between these two devices.

Packet switching, as used in computer networks and increasingly in telecommunications devices (i.e. 3G cell-phones), is where data packets can be individually routed between two nodes (in our case two codecs) over shared LAN and WAN connections.

Packet switching optimizes the use of bandwidth over computer and wireless networks by dividing data streams into packets with destination addresses embedded within them. In this way packets are routed through ISP routing tables to find the best route to their destinations.

The exact form of a packet is determined by the protocol a network is using. Packets are generally split into three parts which include:

- A Header: This section contains instructions about the data contained within the packet;
- The Payload: This contains the actual data that is being sent to the destination; and
- A Trailer (Footer): This tells the receiving device that it has received the entire packet and it may also contain error checking information (used to send a packet resend request if a packet is corrupted).

We will discuss more about lost and corrupted data packets later in the sections relating to Forward Error Correction (FEC).
2.3.2. Latency – Delay over Packet Switched Networks

Latency, or delay, is the amount of time it takes for a packet of data to get from one point to another. Over packet-switched networks delay is variable, depending on the path packets take from their source to their destination. Latency is an important issue when using packet-switched networks – particularly when broadcasting audio or video in live situations. Latency over packet-switched networks is created by:

- Network transmission delay;
- Physical processing delay over the network via switchers and routers etc.;
- Packet delay including algorithm compression delays.

All of these factors contribute to the total latency over a network. If the total latency over a network is too high then it may be difficult to sync up with other audio and/or video sources when conducting interviews etc.

A jitter buffer is often used over networks to solve problems caused when packets arrive in varying intervals – particularly when broadcasting streaming audio and video. Latency is explained in more detail in the sections of this manual discussing **Jitter** and **Forward Error Correction**.

Now let’s discuss a little about the TCP and UDP Internet transport protocols used by *Tieline* codecs.

2.3.3. TCP versus UDP Connections

TCP (Transmission Control Protocol) is an Internet transport protocol most commonly used for many of the Internet’s applications such as email and the World Wide Web.

The TCP protocol ensures reliable in-order delivery of data packets between a sender and a receiver. Its two functions include controlling the transmission rate of data and ensuring reliable transmission occurs. Remote broadcasting over TCP connections will typically require a higher delay than UDP connections. This is because buffering is employed to ensure data packets are received in order.

UDP is a protocol most commonly used for sending Internet audio and video streams. The UDP protocol is different to the TCP protocol in that it sends datagram packets. These packets include information which allows them to travel independently of previous or future packets in a data stream. UDP is a much faster and more efficient method of sending audio over IP. *Tieline* has written special Forward Error Correction software (FEC) for UDP data streams, which significantly increases the stability of a connection and lowers audio delay. UDP is the protocol generally recommended by *Tieline* for codec IP connections.
2.3.4. Local, National and International IP Connections

There are several different attributes of local, national and international IP connections. Following are some assumptions you can make about these connection types. First, a local IP connection will:

- Often route data using the same service provider;
- Achieve higher bitrates and better quality audio connections;
- Require low rates of FEC or none at all;
- Require lower pre-buffer delays;
- Allow the use of all possible profiles; and
- Be generally more reliable.

A national IP connection may:

- Require data to be routed through more Internet router points;
- Achieve good bitrates and good quality audio connections;
- Require medium rates of FEC;
- Require medium pre-buffer delay settings;
- Allow the use of mono and stereo profiles; and
- Be reliable.

An international IP connection may:

- Require data to be routed through many Internet router points and many service providers;
- Achieve lower bitrates and hence lower quality audio connections;
- Require medium to high rates of FEC;
- Require the highest pre-buffer delay settings;
- Allow the use of mono and stereo profiles; and
- Be less reliable.

An awareness of these factors when you are setting up your IP connection will assist you to configure each IP connection successfully, and obtain the best performance.

2.4. Factors Affecting IP Connectivity

Issues that can affect the connectivity of IP connections include:

- Firewalls; and
- Attempting to dial a private IP address from a public IP address.

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Additional Information:

*Tieline* codecs also utilize RTP (Real-time Transport Protocol), which is a standardized packet format for sending audio and video data streams over the Internet. In essence, RTP ensures consistency in the delivery order of voice data packets.
To ensure you are familiar with these factors, please read the following sections about studio and remote codec installations. They discuss private versus public IP addresses, Network Address Translation (NAT) and firewalls.

Although we recommend that you connect using the UDP protocol where possible, if you are having difficulties obtaining a successful connection we recommend you attempt to connect using TCP. In general, you are more likely to get through firewalls with TCP connections.

**Important Note:** Appendix 1 of this reference manual contains a Tech Note which explains how to interpret and troubleshoot firewall problems using the LQ (link quality) display on a codec.

### 2.5. Detailed Instructions on How to Install a Codec in the Studio: IP Connections and NAT

Preparing your studio to receive IP calls from your remote codec is similar to ordering a POTS or ISDN service. If you're not a network IT professional, you should consider using your organization’s network administrator, or hire an IT professional, to install and configure the service properly before you connect your codec. Following is the procedure for successfully installing your Tieline codec at the studio.

#### 2.5.1. Step 1: Order Your Broadband Connection

Let’s start with the basics. First you will need to order a broadband service from your ISP (Internet Service Provider). For best results, we recommend a high-speed Internet connection.

It comes in many flavors such as DSL, ADSL, SDSL, SHDSL, cable and satellite. For the studio, we recommend a wired high-speed service with a downlink speed of at least 256 kbps and an uplink speed of at least 256 kbps. Faster installations provide greater reliability and more broadcasting options.

**WARNING:** We highly recommend that you don’t share your broadband connection with other devices. For a start, other devices will use up valuable uplink bandwidth that you may require for your broadcast codec connection. In addition, telephones sharing broadband connections can interfere with the bit streams sent over IP connections. In particular, some customers have reported lower bit stream reliability when sharing DSL/ADSL connections with phones. This can depend upon the brand of modem and DSL/ADSL filter being used.

#### 2.5.1.1. DSL (ADSL)

A DSL (ADSL) connection is common and transmits bi-directional digital data over a POTS line. DSL (ADSL) connections typically use most of the data channel bandwidth to download data to a subscriber and the rest of the bandwidth is used to send data from a subscriber.
IP connections will only transmit data as fast as the DSL (ADSL) data uplink will provide. This outbound data rate can vary greatly and Tieline recommends that operators check with their DSL (ADSL) broadband service provider to discover the speed of their connection. Common broadband connection bitrates are 512 kbps downlink and 256 kbps uplink, or 1524 kbps downlink and 512 kbps uplink.

2.5.1.2. SDSL and SHDSL

Some parts of North America have Symmetric DSL capabilities and SHDSL (Symmetric High-Bit rate Digital Subscriber Line) technology is now being used increasingly as the IT industry moves to more internationally standardized technology.

SHDSL connections send symmetrical data (i.e. 512 kbps downlink and 512 kbps uplink) as opposed to ADSL connections which send asymmetrical data (i.e. 512 kbps downlink and 256 kbps uplink). Symmetrical data is preferred in most IP broadcast situations because you will most likely achieve higher uplink speeds than with ADSL connections. This increases the stability and quality of your connections.

Unlike DSL (ADSL), SHDSL cannot be transported on top of a POTS line so line sharing is not possible.

2.5.1.3. Studio-to-Transmitter Links

Tieline also offers uncompressed (linear) audio suitable for Studio-to-Transmitter links (STLs) and studio-to-studio links. Choose from 15-20 kHz mono or stereo audio.

We recommend you consider purchasing a priority data service and a dedicated link from your ISP - to guarantee link stability and data integrity. The ISP can tag your data packets to ensure they receive priority over other network traffic. Uncompressed 20 kHz stereo STL links will require at least a 3 megabit bi-directional connection for successful transmission of just the audio. If you have additional network traffic, you will need more bandwidth.

2.5.2. Step 2: Order a Static Public IP Address

The next thing to ensure is that you order a static public IP address from your ISP. Remember the previous discussion on static and public IP addresses and the PBX example we used? (See the section in this manual titled Public Versus Private IP Addresses).

A public IP address is a permanent address for your Internet connection. Instead of street names and house numbers, it's a series of numbers and dots (called a dotted quad) that allows computers to locate and talk to each other on the Internet, similar to the way in which we use phone numbers to locate and talk to people.
A static public IP address allows you to host your own services on your broadband connection without needing to notify everyone each time you reconnect to the Internet.

It's worth noting that with a standard “always connected” broadband service, your ISP dynamically assigns you a public IP address. This IP address can be changed every time you connect to the Internet and even while you are connected, which is why you cannot reliably use a dynamically assigned IP address in the studio for remote broadcasts and should consider obtaining a permanent static public IP address.

In short, you will need a static public IP address in the studio to ensure that you will be able to receive a call from your remote codecs (unless you are connecting using the dual mono profile – see the section titled Dual Mono Profiles Using a Static Public IP Address in this operation manual).

2.5.3. Step 3: Configure Network Address Translation (NAT)
We recommend you hire the services of your ISP or IT professional to "install" the public IP address and perform the Network Address Translation (NAT) between the public Internet and your private Local Area Network (LAN).

The best way to explain NAT is to use the PBX example again. A standard PBX telephone system allows people to call you on a single public telephone number and performs the translation and routing of the public number into a particular private PBX extension. Similarly, in order to receive an IP call from a remote codec over the public Internet, the same network address translation principle applies. NAT allows a single device, such as a broadband router, to act as an agent between the public Internet and a local private LAN.

The relationship between public and private IP addresses and NAT is displayed in the following diagram.
NAT devices such as broadband routers also act as a simple firewall preventing unwanted traffic. Depending on how your firewall is configured, your system may require Port Forwarding to enable a call from a remote codec to be received by a studio codec (for more information see the section titled Codec TCP and UDP Connection Ports). This is often the case if you have several codecs in the studio sharing the same static public IP address. You will require different port numbers in each studio codec to connect them to your remote codecs successfully.

The Tieline default ports are displayed in the section titled Tieline Codec Default Port Settings in this manual. These port settings can also be changed in a codec if you have other applications or devices on your network that are already routed to the default port numbers.

2.5.4. Step 4: Install the Studio Codec and Configure NAT
Once you have a broadband service with a static public IP address installed and your IT professional is on hand to perform the network address translation, you can go ahead and install the codec (i.e. connect the LAN port on the rear of the codec to your LAN network port).

As displayed in the following image, the solid green LED on a codec port indicates an active LAN port connected at 100Mbps. If a codec is connected at 10Mbps then the green LED will not be visible. The flashing orange LED indicates port data activity.
2.5.4.1. DHCP Networks

A DHCP network (Dynamic Host Configuration Protocol) is a fancy name for a network host that automatically assigns a private IP address (or private extension number if you think back to the PBX example) to any piece of equipment that is connected to your LAN. It will also configure the subnet mask and default gateway in the codec. If you have a DHCP enabled network then the minute you plug in your Tieline codec, it will be assigned a number that looks much like 192.168.X. This is the private extension number of the codec on your LAN. Your static public IP address should be routed to this private DHCP IP address through your NAT configuration.

Think of the PBX example again. You can dial from a private extension number to a public number, but you cannot dial from a private number to another private number because the public network will not recognize the call without PBX translation. Similarly, your remote field codec will be assigned a private LAN IP address from which you can initiate a call to your public IP address at the studio and NAT will successfully route your call to the private IP address of the studio codec - but it will not work the other way.

If your installation is only temporary, (i.e. for 1 broadcast or a short period of time) then you should be able to rely on the DHCP auto assigned IP address. However, be aware that as soon as you cycle the power on the codec, a new DHCP IP address is likely to be assigned by your network to the codec. The NAT translation will still route the incoming call to the original private DHCP assigned IP address and you will not get a connection.

2.5.4.2. Recommended Permanent Studio Installation Procedure

For permanent or long-term installations we recommend that your IT professional configures your DHCP server to assign a permanent static private IP address into the codec every time the power is cycled, or the codec is connected or disconnected from the network. The DHCP server
can be programmed to assign this based on the unique MAC address of the codec you are using.

The benefit of this is that you can connect to any location on the LAN and it will always assign the same static private IP address into the codec. Alternatively, you can assign a static private IP address directly into the codec and configure your local area network to obtain the IP address from the codec.

One last important point: Normally, you will not be able to initiate an outbound call from the studio codec over the Internet to the remote codec. The only time you will need to dial from the studio codec to the remote codec is if you are using the Dual Mono program profile, which connects a single studio codec to two remote codecs via individual IP connections.

In this situation you will require static public IP addresses in your remote codecs (in the same way as you should have set up and configured a public IP address with NAT in your studio). This type of connection is discussed in the following section of this manual.

2.6. Detailed Instructions on how to Configure a Codec at a Remote Broadcast Location

Following is the procedure for setting up a codec at a remote broadcast location. The configurations discussed include:

- Connecting using DHCP servers
- Connecting using a static private IP address; and
- Connecting using a static public IP address.

**WARNING:**
We highly recommend that you don’t share your broadband connection with other devices. For a start, other devices will use up valuable uplink bandwidth that you may require for your broadcast codec connection. In addition, telephones sharing broadband connections can interfere with the bit streams sent over IP connections. In particular, some customers have reported lower bit stream reliability when sharing DSL/ADSL connections with phones. This can depend upon the brand of modem and DSL/ADSL filter being used.

2.6.1. Step 1: Plug into a DHCP Server at the Remote Site and Dial the Studio

If the remote venue or remote broadcast site has a DHCP LAN which can access a broadband Internet service, then all you need to do is connect the codec to the LAN port, dial the studio and you’re on the air. It’s that simple. The DHCP LAN will auto assign your codec an IP address, subnet mask and default gateway.
Even if the venue does not have a corporate LAN, many of the latest DSL routers have the ability to assign an IP address to equipment and you can attach a codec to a LAN port on a DSL router.

### 2.6.1.1. Checking IP Address, Subnet Mask and Default Gateway Settings

Once you have connected to the LAN at the remote broadcast location, wait five seconds and then reboot the codec.

To confirm your codec has the new IP address, Subnet Mask and Default Gateway settings, ensure you are connected to your LAN and select [Menu] > [Unit Details] > [IP Address] > [Subnet Mask] > [Default Gateway]. If you are not connected to a LAN the codec menu will display **LAN not connected**.

### 2.6.2. Step 2: DHCP Server Unavailable: Program a Static Private IP Address

If the venue does not have a DHCP network then you will need to configure a private static IP address and enter the Subnet Mask and Default Gateway information into the codec. This is a simple task and you can obtain this information from the venue network administrator.

#### 2.6.2.1. What is the Subnet Mask and Default Gateway

Sub-netting allows network administrators to be flexible in the way that users communicate on a network. Host devices using different sub-networks can only communicate if they use a network gateway device. This is used to filter traffic and limit access to certain users.

The default gateway is usually a computer and router for a network that is an access point to other networks. This is programmed into your codec to allow it to send data.

Codecs using static private IP addresses will need to have the subnet mask and default gateway programmed into it, in addition to the IP address. Please read the following sections in this manual titled Entering a Static IP Address, Entering the Subnet Mask and Entering the Default Gateway, which explains how to program this information into your codec.

**Important Configuration Tip:** The static IP address for the codec will be given to you by a network administrator. To obtain the Subnet Mask and Default Gateway settings for your codec, go to a PC connected to the LAN you will be connecting to and do the following. For Windows® XP  

1. Click on the “Start” menu and select “All Programs”. Now select “Accessories” and then “Command Prompt”. Once the “Command Prompt” window appears, type “ipconfig” and press “Enter” on your keyboard. The Subnet Mask and Default Gateway setting for the LAN will be displayed and you should input these settings into your codec.

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2. Windows is a registered trademark of Microsoft Corporation in the United States and/or other countries.
2.6.2.2. Entering a Private Static IP Address

To enter a private static IP address access the codec [Menu] by pressing SOFTKEY 4 (the button closest to the bottom right-hand corner of the LCD), then rotate the MENU SELECTOR (MS) and select [Configuration] > [Advanced Menu] > [LAN Settings] > [IP Setup] > [Static] > [IP address]. Press the CLEAR button to delete any existing address details and enter the new address.

When this is done select [OK] by pressing SOFTKEY 4 to confirm the IP address. Without leaving the IP Cfg menu you should now configure the Subnet Mask into the codec.

**WARNING**

Only use a private static IP address assigned by your network administrator if you are connecting to a DHCP server. This will ensure that you don’t use an address that is already in use. Using an IP address already in use can cause a conflict on the network.

2.6.2.3. Entering the Subnet Mask

Without leaving the IP Cfg menu, scroll to Subnet Mask in the same menu. Press SOFTKEY 4 once to enter the subnet mask into the codec. Press the CLEAR button to delete any existing address details and enter the new address. We suggest you input the following number in this section 255.255.255.0.

When this is done select [OK] by pressing SOFTKEY 4 to confirm the new Subnet Mask. Without leaving the IP Cfg menu, once you have set the Subnet Mask details into the codec, you should configure the Default Gateway in a similar manner.

2.6.2.4. Entering the Default Gateway

Without leaving the IP Cfg menu, scroll to Default Gateway. Press SOFTKEY 4 once to enter the default gateway into the codec. Press the CLEAR button to delete any existing address details and enter the new details.

When this is done select [OK] by pressing SOFTKEY 4 to confirm the new Default Gateway.

2.6.3. Step 3: Dual Mono Profiles Using a Static Public IP Address

With a dual mono profile you can send two channels of up to 15 kHz mono audio including all inputs, or send two individually configured mono feeds from two different remote broadcast codecs.

If you are using a dual mono profile, you will need to use static public IP address in each remote codec. This is because the codec that makes calls to two different IP addresses needs to be the dialing codec – in this case the...
studio codec. Remember that if you are making a call you need to dial a public IP address. That is why the remote codecs require static public IP addresses when you use this profile. This is different to other profiles where the remote codec usually dials into the studio codec using a static public IP address.

2.7. IP Codec Menus in Detail
The full codec menu structure for IP connections is displayed on the following page. Following this diagram is a description of each element of the menu and how to configure your codec for IP connections. Remember to set both the local and remote codecs with identical settings to ensure optimal performance over your IP connection.
Figure 7: Full Codec IP Menu Structure
2.8. Advanced Programming of IP Connections using ToolBox

Configuration of codec IP and LAN settings can be completed by using the IP/LAN tab in ToolBox software. It is actually much simpler than using the codec menu structure because it provides an overview of all the IP and LAN settings in the codec and gives users the opportunity to program all settings at once. Once settings have been programmed save them in a configuration file and load the file onto the codec.

Figure 8: IP/LAN Tab in ToolBox

Following is an overview of the sections in the tab.

2.9. IP Setup

IP setup allows the user to input the IP addressing information that is necessary when using a LAN connection.

The settings here are for configuring a static IP address in a codec. If DHCP or BOOTP addressing is selected, the settings in this menu are greyed out and the IP address allocated to a codec can be viewed in the Global Unit Settings tab.

For more information, please see the section in this manual titled Detailed Instructions on How to Install a Codec in the Studio.

2.9.1. Ethernet Setup

Ethernet setup allows the user to select the Ethernet data transfer speed. Tieline recommends that the Link speed be set to Auto detect.
2.10. IP Stream Setup

Section 1 of this manual explains how to connect quickly using IP with manual default profiles. This includes detailed information about the best rule-of-thumb settings for IP streaming. We highly recommend that you consult both these sections before adjusting and configuring any of the following ToolBox menu settings.

2.10.1. Select IP Interface

When IP streaming using Tieline codecs you can have up to two connections. Select IP interface allows you select either IP1 or IP2 as the connection that you will be configuring.

2.10.2. RTP Session Type

This menu provides the ability to program the session data protocol used by the codec when dialing. There are three options:

- Tieline: Uses Tieline codec session data;
- None (Sessionless); or
- SIP.

If you are connecting two Tieline codecs use the default Tieline setting unless you are connecting over SIP. Select SIP if you are connecting using SIP and select None (Sessionless) if you are connecting to a non-Tieline codec. For more information see the Session Type section in this manual.

2.10.3. Bitrate

This sets the Bitrate that you wish to connect with. The Bitrate possible will be dependent on a variety of factors - including the uplink bandwidth of your broadband connection and the profile you wish to use. Initially set the audio bit rate to 9,600 bps on both the local and remote codecs. Once the codecs are connected, and the call has stabilized you can manually negotiate higher bit-rates. For best performance, the dialing codec should be used to renegotiate connection bit rates up and down.

2.10.4. Encode/Decode Direction

By default Tieline codecs send data bidirectionally. It is possible to reduce the amount of data being sent over a connection by only sending data in one direction. This can assist in reducing the demands on uplinks or downlinks when the bandwidth available is limited. It also
assists in compatibility with some non-Tieline codecs that only send audio data in one direction.

Select Encode Only at the codec sending audio and select Decode Only at the codec receiving audio only.

2.10.5. **Man Dflt Algorithm**

The Manual Default Algorithm setting relates to the Manual Default Profiles within ToolBox. The setting you choose here will relate to any Manual Default Profiles you use that utilize IP connections. For example, if you select Music from the drop-down menu, any manual default profile using an IP connection will be programmed to use the Tieline Music algorithm.

Choose Voice G3 at low bitrates for greater connection stability. Tieline Music or MusicPlus will provide higher quality audio but requires higher connection bitrates. You may have lower connection reliability with these algorithms, depending on your available uplink bandwidth. For more information see the Set Algorithm section in this manual.

**Very Important Note:** Voice G3, Music and Music Plus algorithms cannot be used over SIP connections. Use MP2 algorithms at 64kbps mono or 128kbps stereo for high quality connections when using SIP.

2.10.6. **Samplerate**

Samplerate is grayed-out for the Music and Voice G3 algorithms. It is possible to set the Samplerate for Raw Audio and MPEG algorithms.

2.10.6.1. **Raw Audio Samplerate**

Once you select the Raw Audio algorithm, it will become active and provide selectable settings as displayed.

**Important Note:**
FEC is not available when using the Raw Audio algorithm and never use this algorithm over the Internet. Raw Audio should only be used over a well-managed (low traffic) LAN or when using a crossover cable between codecs.

Tieline recommends setting the Samplerate at no more than 32000. In addition, we recommend using dedicated switchers and associated infrastructure that is designed to handle high volumes of data. This is because sustained high bitrates are often not handled well by cheaper
equipment and busy networks – even if Ethernet link speeds are set to 10 or 100Mbps.

2.10.7. Jitter Buf Msecs
This sets the jitter-buffer milliseconds for your IP connection. This will be dependent on a variety of factors including FEC settings in your codecs. Jitter buffer settings can be fixed or automatically set by the codec. In most situations involving Tieline codecs we recommend using the auto jitter buffer set at Best Compromise.

For detailed information on which jitter-buffer setting is best to use, check the Jitter Buffer section in this manual.

2.10.8. Audio RTP Protocol
You can choose from either UDPIP or TCPIP. The TCP protocol sends reliable in-order data between a sender and a receiver. Remote broadcasting over TCP connections typically will require a higher delay than UDP connections.

UDP is a protocol most commonly used for sending Internet audio and video streams. The UDP protocol is different to the TCP protocol in that it sends datagram packets. These packets include information which allows them to travel independently of previous or future packets in a data stream. UDP is a much faster and more efficient method of sending audio over IP. Tieline has written special Forward Error Correction software (FEC) for UDP data streams, which significantly increases the stability of a connection and lowers audio delay. Where possible, UDP is the protocol generally recommended by Tieline for codec IP connections.

2.10.9. Audio Port
This sets the audio port number for an IP connection. As a default, Tieline codecs use a TCP session port to send session data and either a TCP or UDP port to send audio. The audio port is configured when you select the audio protocol you wish to connect with. The reason the session port always uses the TCP protocol is that TCP is the most likely protocol to get through firewalls – ensuring critical session data (including dial, connect and hang-up data) will be received reliably.
Generally we recommend using the default port settings. There are a few reasons why you may wish to adjust the port setting on your codec. These include:

- Creating individual port settings for gateways and firewalls;
- Another device is already using a codec’s port number; or
- More than one studio codec is in use and each codec requires a different port number. For a studio and remote codec to connect, they need to be programmed with matching port numbers.

If there is a need to change codec port settings, in most situations you should consult your organization’s resident IT professional.

2.10.9.1. Session Port (TCP)

This sets the session port number for an IP connection. As a default, Tieline codecs use a TCP session port to send session data and either a TCP or UDP port to send audio. See the earlier section titled Port Number for more information on configuring this setting.

For more information see the Set the TCP Session Port and (UDP) Audio Port section in this manual.

2.10.10. Local FEC%

This menu configures Forward Error Correction on the local codec. There are four settings available. Forward Error Correction is designed to increase the stability of UDP/IP connections. It is configurable for UDP data streams only. If you select TCP as the Audio RTP Protocol then the FEC menus will be grayed-out and inactive.

The way FEC works is by sending a secondary stream of audio packets so that if your primary audio packets are lost or corrupted, then packets from the secondary stream can be substituted to correct the primary stream. The amount of FEC that you require is will depend on how many data packets are being lost over the network connection.

To fully understand the relationship between what your FEC setting should be, based upon your codec connection bit rate and your broadband uplink speed, see the section titled Set Forward Error Correction in this manual.

2.10.11. Remote FEC%

This menu sets Forward Error Correction on the remote codec. As with the Local FEC% setting (discussed previously), there are four settings available.

To fully understand the relationship between what your FEC setting should be, based upon your codec connection bit
rate and your broadband uplink speed, see the section titled Set Forward Error Correction in this manual.

2.11. Programming IP/LAN Connections via a Codec

The following sections explain in detail how to program IP and LAN connections using only a codec itself.

2.12. Setting the Audio Protocol

The first setting in the codec IP menus is for selecting the IP protocol that you require. You can choose from either UDPIP or TCPIP. As previously discussed, UDP is the protocol generally recommended by Tieline for codec IP connections.

2.12.1. Wireless Network Connections

**WARNING:**

Tieline recommends using UDP connections for wireless setups.

You can purchase an off-the-shelf mini Wi-Fi bridge and connect it directly to the LAN port on the rear of the Tieline codec to interface with wireless LANs. Combine this with Tieline’s portable in-car power or battery options and you have a fully mobile live reporter kit.

You can select Tieline’s ultra low bit-rate Voice G3 algorithm and deliver up to 7.5 kHz with connections as low as 9.6 kbps. Use the Music algorithm and deliver up to 15 kHz with connections as low as 19.2 kbps. The smaller the stream you are sending over a network, the less interference it will encounter on the way. However, (like a cell phone network) be aware that wireless networks sometimes simply drop out due to external influences beyond the codec’s control.

In most cases, connecting codecs over wireless routers is simple because it will usually automatically assign a private IP address to a codec. If the router is unsecured, you should be able to successfully access the Internet immediately.

2.12.1.1. Using Secure Wireless Routers

In some cases, a wireless router may be secured using specific IP or MAC addresses. You can overcome these obstacles by programming a static private IP address into the codec and enabling the wireless router to use that IP address, or by injecting the unique MAC address of the Tieline codec into the router. The most difficult case we have seen in our field tests required us to program a static private IP address, subnet mask and default gateway into the codec, then program the MAC address into the router, and finally, enter the codec’s IP address into the router’s DMZ (Demilitarized Zone).
2.12.1.2. Wireless Reliability

There are several caveats to be aware of before deciding to conduct a wireless broadcast compared with a wired IP or traditional POTS and ISDN broadcast.

Wireless networks are far less reliable than wired networks. Since numerous factors influence wireless transmission you'll rarely approach the maximum networking speeds set out by the various wireless protocols. Data transfer can vary between 30-60% of the stated maximum depending on adapter interface, distance from the wireless router, number of users connected to the router, number of obstacles such as walls and types of building materials, packet length, number of packet collisions and packet retransmissions... the list goes on!

Most of these obstacles can be overcome by using Tieline’s unique FEC and buffering strategies.

2.13. Session Type

Session data is a great tool for programming codecs to operate according to a series of preprogrammed instructions. It makes the operation of Tieline codecs a simple task and can eliminate the need to adjust codecs each time they are connected.

The **Session Type** IP menu provides the ability to program the session data protocol used by the codec when dialing over IP. There are three options:

- ✔ Tieline Codecs: Using Tieline codec session data;
- ✔ None (Sessionless); or
- ✔ SIP.

It is useful to read about TCP and UDP ports to fully understand the implications of selecting these different session data settings. See the **Set the TCP Session Port and (UDP) Audio Port** section in this manual for more information.

If you are connecting two Tieline codecs use the default **Tieline Codecs** setting unless you are connecting over SIP.

The ports used for sending session data and audio using Tieline session data are displayed in the following image.

![Figure 9: Tieline Session Data Default Ports](image_url)
2.13.1. SIP

This is an established format for describing streaming media initialization parameters. When connecting two devices, SDP performs similar tasks to Tieline’s proprietary session data, which is used to configure all non-IP codec connections and non-SIP IP connections.

SIP provides superior interoperability due to its standardized protocols for connecting devices and is intended to be used when connecting Tieline codecs to non-Tieline devices. SIP is only used to negotiate a connection between two devices and media data is sent separately. This process is displayed in the diagrams that follow.

If you wish to connect using SIP (Session Initiation Protocol) then select this setting in the codec menu. Session Description Protocol (SDP) is used to connect devices and negotiate connection characteristics such as the algorithm and connection bit rate when using SIP.

Different ports are used for SIP connections and the following images explain peer-to-peer and SIP Server routing.

![Figure 10: SIP Peer-to-Peer Port Configuration](image-url)
SIP uses UDP port 5060 for signaling data. When SIP Session is selected, two other ports are used and these can be programmed in the IP configuration menu in a codec or via ToolBox software. The port number should be an even number and the codec automatically allocates the port one number higher for RTCP control (e.g. UDP 9000 and 9001, or UDP 9010 and 9011 for the default codec UDP audio port settings).

To contact a codec that is behind a firewall or NAT-enabled router it is essential that all relevant ports are open and forwarded to the other device.

2.13.2. None (Sessionless)
This setting turns off session data completely for IP connections only. This is unlike the System Settings session data Disable function – which still sends some session data when initially configuring a connection.

Most non-Tieline codecs and other devices do not respond to session data sent by a Tieline codec. We recommend that it is best to turn off session data if you are connecting your Tieline codec to a non-Tieline device over IP. We also recommend that you ensure the same algorithm settings are programmed into both codecs. This will not only reduce the chance of a failed connection, it will also reduce the time it takes for the two codecs to connect.

If a non-Tieline codec is set to auto-scan for the algorithm used by the codec it is connecting to, it can take quite some time to connect and in some situations may fail (this process is called ‘framing’ and is when incoming bit streams are identified and distinguished for decoding).

Session data times-out for all connections except X.21, enabling non-Tieline codecs to connect to a Tieline codec after session data is switched off. You must turn session data off if connecting to a non-Tieline codec over X.21.
Whatever you decide to do, we highly recommend you do a test of the connection well before you go on-air, to ensure your settings are compatible.

2.14. Set Algorithm

Next, choose the algorithm that you wish to connect with. As previously mentioned, this will be dependent on a variety of factors, including the uplink bandwidth of your broadband connection and the profile you wish to use. It will also depend on whether you are connecting using SIP or not.

**Very Important Note:** Voice G3, Music and Music Plus algorithms cannot be used over SIP connections. Use MP2 algorithms at 64kbps mono or 128kbps stereo for high quality connections when using SIP.

Choose Voice G3 at low bitrates for greater connection stability. Tieline Music or Music Plus will provide higher quality audio but both algorithms require higher connection bitrates. You may have lower connection reliability with these algorithms, depending on your available uplink bandwidth.

It can be a good idea to listen to the quality of your program signal using each algorithm and to see how it sounds when it is sent at different connection bitrates (as well as different FEC and jitter-buffer millisecond settings). This will assist you to determine what the best algorithm setting is for the connection you are setting up.

The algorithm selected will not only affect the quality of the broadcast but it will also contribute to the amount of latency or delay introduced. For example, if MP2 algorithms are used, program delays will be much longer than when using Tieline Music, Music Plus or Voice G3 algorithms. This is due to the additional inherent encoding delays involved when using MP2 algorithms.

This can be a major consideration for live applications that integrate talkback callers or live remote-crosses into a broadcast. Particularly when using 3GIP networks where transport delays are generally greater than over fixed line networks.

Tieline now offers two optimized low-delay Music algorithms for IP and wireless IP connections.

2.14.1. Tieline Music

The Tieline Music algorithm is optimized for audio bit rates as low as 19.2kbps with only a 20 millisecond encode delay. It offers 15 kHz mono from 24kbps to 48kbps.

2.14.2. Tieline Music Plus

Tieline Music Plus offers 20 kHz mono from 48kbps upwards. It can also deliver 15-20 kHz stereo from 64-96kbps upwards.
2.14.3. Voice G3

In rare cases where a wireless broadband IP connection will only support 16 kbps or below we recommend using the Tieline Voice G3 algorithm which delivers 7 kHz G.722 audio quality with low delay.

For detailed rule-of-thumb information on which algorithm is best to use with the available uplink bandwidth of your connection, check the uplink bandwidth table.

2.15. Set Audio Bitrate

Next, set the audio bit rate that you wish to connect with. As previously mentioned, this will be dependent on a variety of factors including the uplink bandwidth of your broadband connection and the profile you wish to use. Initially set the audio bit rate to 9,600 bps on both the local and remote codecs (unless you are connecting using SIP as renegotiation is unavailable over SIP connections). Once the codecs are connected, and the call has stabilized you can manually negotiate higher bit-rates by holding down the “F2” button on the keypad and then pressing “3”. To negotiate lower bit-rates hold down “F2” and then press “9”. If you hear audio dropouts or artifacts, the current bit rate cannot be sustained and should be renegotiated down. For best performance, the dialing codec should be used to renegotiate connection bitrates up and down.

Important note: Use MP2 algorithms at 64kbps mono or 128kbps stereo for high quality connections when using SIP.

2.15.1. Audio Bitrate and Link Quality Relationship

The link quality reading on the codec LCD screen is similar to the Line Quality (LQ) reading when you are using Tieline codecs in POTS mode. It is important to recognize the importance of the link quality of your connections and how setting the audio bit rate should be based on the prevailing link quality. As a rule of thumb, Tieline recommends that you try to maintain a link quality reading for your connection of between 70% and 100%. The link quality reading should remain between these points to ensure you maintain a stable IP connection over long periods. In the following image, the local codec link quality is 96% and the remote codec link quality is 92%.

![Figure 12: Link Quality Display on the Codec LCD](image)

By using this percentage range as a guide, negotiate your connection bit rate upwards to maximize the quality of your connection. If the link quality starts to become unstable and bounce around outside the recommended range, there is too much traffic at either the local or remote codec. If this is the case, reduce your connection bit rate.
Remember that other factors such as the FEC setting and jitter-buffer millisecond settings will also impact on the stability of your connection. During the setup phase of your IP connection, adjust all these settings to maximize the audio quality and stability of your connection.

Please note: If you are using a dual mono profile with two connections, when you renegotiate the bit rate up or down using the codec F2, 3 and 9 keys, both connections will be renegotiated simultaneously. If you wish to renegotiate each connection individually, please adjust the connection bit rate using the codec menus.

2.15.2. Conserving Bandwidth: Encode/Decode Direction
By default Tieline codecs send data bidirectionally. It is possible to reduce the amount of data being sent over a connection by only sending data in one direction. This can assist in reducing the demands on uplinks or downlinks when the bandwidth available is limited. It also assists in compatibility with some non-Tieline codecs that only send audio data in one direction.

To program this select [Menu] > [Configuration] > [IP Setup] > [Enc/Dec Direction] ➔ Encode/Decode Only. Select Encode Only at the codec sending audio and select Decode Only at the codec receiving audio only. This can also be programmed via the IP/LAN tab in ToolBox.

2.16. Set Forward Error Correction (FEC)
Next, set FEC on the codecs that you wish to connect with. As previously mentioned, this will be dependent on a variety of factors including the uplink bandwidth of your broadband connection and the profile you wish to use. Detailed information about FEC operation follows and you should also consult the uplink bandwidth table for the profile you wish to connect with.

2.16.1. Understanding DSL (ADSL) and Forward Error Correction
In general, the higher the bit rate you require for a codec’s IP connection, the more likely it is that you will encounter connection instability. The relationship between the codec connection bit rate and your broadband uplink speed is useful to help you choose the IP settings you should use for different IP profile connections. In the next few sections we will explain Forward Error Correction as well as the settings you should use for different algorithm and profile selections.

2.16.2. Forward Error Correction (FEC)
Forward Error Correction is designed to increase the stability of UDP/IP connections. It is configurable for UDP data streams only. Both the local and remote codec FEC settings can be configured before dialing and once the codecs are connected ‘on the run’.

Important Notes: FEC can only be programmed for use with the Music, Music Plus and Voice G3 algorithms. FEC is only a benefit if link quality displayed is below “L99 R99.”
The way FEC works is by sending a secondary stream of audio packets so that if your primary audio packets are lost or corrupted, then packets from the secondary stream can be substituted to correct the primary stream. The amount of FEC that you require is will depend on how many data packets are being lost over the network connection.

There are four FEC settings possible within Tieline codecs. To program FEC select [Menu] > [Configuration] > [IP1 Setup] > [Local/Remote FEC Percent]. It is possible to program FEC on both codecs that are being connected.3

The codec settings are outlined in the following table along with the bit rate ratios.

<table>
<thead>
<tr>
<th>FEC Setting</th>
<th>Bitrate Ratios</th>
<th>Connection Use</th>
</tr>
</thead>
<tbody>
<tr>
<td>100% (Lowest delay)</td>
<td>A simultaneous dual-redundant stream (1:1 ratio) is sent from the codec. Twice the connection bit rate is required to operate the codec using the 100% setting. For example, if your connection is at a bit rate of 14,400, you will require an additional 14,400 kilobytes of bandwidth to allow for the FEC data stream — a total of 28,800.</td>
<td>Recommended to be used over wireless and international connections.</td>
</tr>
<tr>
<td>50%</td>
<td>Additional data is sent by FEC in a ratio of 2:1.</td>
<td>Recommended to be used over international &amp; national connections.</td>
</tr>
<tr>
<td>33%</td>
<td>Additional data is sent by FEC in a ratio of 3:1.</td>
<td>Recommended to be used over national and local connections.</td>
</tr>
<tr>
<td>20% (Highest delay)</td>
<td>Additional data is sent by FEC in a ratio of 5:1.</td>
<td>Recommended to be used over local and LAN connections.</td>
</tr>
<tr>
<td>Off</td>
<td>FEC is off in the codec and the connection bandwidth is equal to the connection bit rate setting in the codec.</td>
<td>Recommended to be used over wired LAN connections as well as managed T1 &amp; E1 connections for STLs that have connections that aren’t shared &amp; have quality of service (QOS).</td>
</tr>
</tbody>
</table>

Table 4: Explanation of FEC Rates in Tieline Codecs

As an example of how FEC works, if you have a FEC setting of 20% and you are losing one packet in every five that is sent, this packet will be replaced by

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3 FEC is visible via the connection details LCD screen, which is visible when the MENU SELECTOR or the ENTER/DIAL button is pressed when a codec is connected.
FEC to maintain the quality of the connection. If you are losing more packets than this, say one in three, it will be necessary to increase the FEC setting to 33% to compensate.

**Please note:** There is an inverse relationship between FEC settings and the jitter-buffer millisecond setting that you use for IP connections.

So why not use 100% FEC every time? The answer is because you need twice the bit rate to achieve full redundancy and depending on the link conditions, this could potentially cause more dropouts because of network congestion than it fixes. Here is a simple rule to remember: Your maximum uplink speed is all the bandwidth you have to play with. As a rule of thumb, try not to exceed more than 80% of your maximum bandwidth. If your link is shared, be even more conservative.

You should also consider the remote end too. What is their maximum upload speed? Is the connection shared at either end? Your bit rates, FEC settings and buffer rates must be pre-configured at both ends before you connect therefore so it's always better to set your connection speed and balance your FEC according to the available uplink bandwidth at each end for best performance.

As an example, if you want 15 kHz mono (using the Tieline Music Algorithm) you will need at least a 24kbps connection for audio. Adding 100% FEC will add another 24kbps making your bit rate 48kbps plus some overhead of around 10kbps. If you're on a 64kbps uplink, you should consider reducing your FEC or switching to a 7kHz audio link at 14.4kbps with the Voice G3 algorithm and 100% FEC.

Here is another example, if you want 15 kHz stereo, you need at least 56kbps for the audio. 100% FEC requires at least 112kbps and 50% FEC requires at least 84kbps. If your uplink speed is 256kbps and you're on a shared connection, then choosing a lower FEC% of 20%-33% may give you better results.

### 2.16.3. Conserving Bandwidth with FEC

As mentioned previously, there is a trade-off between the quality and the reliability of an IP connection – particularly when FEC is activated on your codecs. However, it is possible in certain situations to set different FEC on each codec to conserve bandwidth and create more stable IP connections.

For example, if your broadcast is a one-way broadcast from a remote site, i.e. you are not using the return path from the studio, or only using it for communications purposes, it is possible to reduce or turn off FEC at the studio codec. This effectively reduces the bandwidth required over the return link (communications channel) and increases the overall bandwidth available for the incoming broadcast signal from the remote site. This could be particularly useful if you have limited uplink bandwidth at the remote location.
Keep in mind that as you move from local to national to international connections, you should be more conservative with your FEC and connection rates.

2.17. What is Jitter?
Packet jitter occurs over packet-switched networks and is created when data packets sent over a network do not arrive in regular intervals. This occurs because the packets sent can travel over any route to their destination – despite being sent in regular time intervals. The random delays that occur, and the severity and frequency of these delays, will be different for every connection.

2.18. What is a Jitter Buffer?
A jitter buffer is a temporary storage buffer used to capture incoming data packets. It is used in packet-based networks to ensure the continuity of audio streams is maintained. A jitter-buffer smoothes out packet arrival times during periods of network congestion. Data packets travel independently and arrival times can vary greatly depending on network congestion and the type of network used, i.e. LAN versus wireless networks.

In a way, a jitter-buffer can be looked upon as insurance for packets not turning up in time. The trade-off, or cost of increasing jitter-buffer is increased latency. The greater the jitter-buffer delay programmed, the greater the program delay.

Packets are retrieved from the jitter buffer at regular intervals by a device’s decoder in order to provide a smooth and regular play out of audio streams. The concept of jitter buffering is displayed visually in the following image.

If a jitter buffer is not adequate then it is likely that interruptions to streams will occur as a result of late packets. If the time value or ‘depth’ of the jitter buffer is set at a point larger than the longest experienced jitter delay, then all packets received by a device will be delivered to the decoder and the best possible audio quality is recreated.

Unfortunately there are two problems with this scenario:

1) There is no way to predict for sure what the longest jitter delay will be; and
2) The larger a jitter buffer is (to increase the chance of catching all late packets) the longer the end-to-end and round trip delay of data becomes. (This becomes unacceptable for applications that need low delay or feedback.)

2.18.1. Tieline Jitter Buffer Settings

**Tieline’s** Jitter-buffer is smart because of its ability to:

- Remove duplicate packets;
- Re-order packets if they arrive out-of-order;
- Repair the stream in the event of packet loss (error concealment);
- Manage delay intuitively based on current network congestion; and
- Manage forward error correction (FEC).

In the past, **Tieline** codecs have used a fixed but configurable jitter buffer ‘depth’ and a user could increase this value if late packets caused audio artifacts or dropouts. Jitter buffer could be reduced if less delay was desired, at the risk of an increased loss of packets and hence reduced audio quality. Largely this was set by a user through trial and error after assessing network conditions and the most appropriate setting. Fixing the codec jitter buffer is still possible and is called **Fixed Buffer Level** in the **Jitter Buffer Type** codec menu.

2.18.2. Tieline’s Smart Auto Jitter Buffer Solution

The new **Auto-Jitter Buffer** feature is designed to adaptively analyze the history of observed jitter over a connection and then set the jitter buffer depth automatically based on this result.

**Warning:**

If you are connecting over IP and using auto jitter buffer then both codecs must have firmware v.1.6.xx installed. If a codec is using a version of software lower than this then auto jitter buffering is automatically disabled and the last fixed setting programmed into the codec is enabled.

Unfortunately, due to the random nature of jitter, there is no magic method that provides the perfect solution. It is always a trade-off between reducing loss and reducing further delay. As a result, **Tieline** has decided to get an indication from a user as to where their priorities lie (or what they value more).

Codec menus and the IP wizard now give the choice of **Fixed Jitter Buffer** or **Auto-Jitter Buffer**. If **Fixed Jitter Buffer** is selected, the same screen options are displayed - as in firmware versions prior to version 1.6.xx. These menus prompt a user to input the value of the jitter buffer in milliseconds.
Important Auto Jitter Buffer Compatibility Note:
If a **Tieline** codec connects to a device that is using non-compliant RTP streams then the last fixed setting programmed into the codec will be enabled. Non-compliant devices include legacy **Tieline** codecs, some other brands of codec, web streams and other devices.

2.18.3. Programming ‘Auto Jitter Buffer’ Settings

If a user selects *Auto-Jitter Buffer*, they are prompted to indicate which of the five available settings best describes their preferences. These preferences provide a simple method of dealing with what has previously been a complex concept to program.

**Tieline**’s simple settings include the following options:

- Highest Quality;
- Good Quality;
- Best Compromise;
- Less Delay; and
- Least Delay.

**Least Delay:**
This setting attempts to reduce the jitter buffer to the lowest possible point, while still trying to capture the majority of data packets and keep audio quality at a reasonable level. This setting is the most aggressive in its adaptation to prevailing conditions, so jitter buffer may vary more quickly than with the other settings. It is not recommended in situations where jitter variation is significant and/or peaky. (E.g. 3G/multi-user wireless networks). It is best for stable and reliable links such as dedicated or lightly-loaded WAN/LANs.

**Highest Quality:** This setting is the most conservative in terms of adapting down to reduce delay. The jitter-buffer setting will actually stay high for a longer period after a jitter spike is detected – just in case there are more spikes to follow. This setting is best used where audio quality is most highly desired and delay is not so critical.

Unless delay is irrelevant, this setting is also not recommended over peaky jitter networks (such as 3G) and is best used on more stable networks where large jitter peaks are not as common.

**Best Compromise:** This (default) setting is literally the midpoint between the jitter buffer levels that would have been chosen for the *Highest Quality* and *Least Delay* settings. It is designed to provide the safest level of good audio quality without introducing too much extra delay.

**Good Quality** and **Less Delay:** These two settings lie between the mid-point setting of *Best Compromise* and two settings *Highest Quality* and *Least Delay*. They indicate a slight preference and may assist in achieving better performance from a connection without incurring extreme delays in transmission or packet loss.
2.18.4. How to get the Best Jitter Buffer Results

The decision of where a jitter buffer setting is programmed is based on observed network conditions. The best results are seen if the connection is established for a while before ‘going live’, so there is enough data to start making well-informed decisions about network conditions.

The initial jitter buffer setting when a codec first connects is 500ms and it is kept at this level for the first minute of connection (if observed delay values are lower than this point).

After this initial period, adaptation of the jitter buffer delay commences and it usually reduces.

Important Note:
It is recommended that a connection is established for at least 5 minutes prior to broadcasting, so that the codec has been provided with enough jitter history to ensure a reliable connection.

There are five states or stages that jitter buffer may display and these can be observed in the connection status screen by pressing the Enter/Dial button while connected. The first four stages are observed in “auto” jitter buffer mode.

1. **Stabilization period (a1):** This is a few seconds at the start of a connection where no action is taken at all while the establishment of a stable connection means analysis of jitter data is not valid.

2. **Stage 2 (a2):** A compatibility check to ensure the RTP connection is compliant and RTP clocks are synchronized enough to perform jitter analysis.

3. **Stage 3 (a3):** If the compatibility check is successful, this is the analysis hold-off period. During a minute, the jitter buffer is held at a safe, fixed value of 500ms while enough history is recorded to start jitter buffer adaptation.

4. **Stage 4 “live” (A):** This is where the codec determines it is safe enough to start broadcasting using the auto-jitter buffer level. We recommend running the codec for a few more minutes to obtain a more comprehensive history of the connection’s characteristics.

5. **Fixed (F):** This state is displayed if the jitter buffer is fixed.

In auto mode if the codec jitter buffer needs to be adjusted then it adjusts gradually and imperceptibly on most occasions.

If a significant jitter ‘spike’ is detected and causes audio loss, the jitter buffer is instantly raised above the observed spike. Audio may drop out momentarily while the codec jumps to a new jitter buffer level.

It is held at this point for a short period in case more spikes follow, or in case there is a general rise in the average level of jitter over the network. After a hold-off period, the system re-assesses jitter values and reduces the buffer level again if it was just a temporary spike.
The ‘instant’ reaction to a jitter spike will not mean audio quality is maintained at 100% but it does help to prevent further loss in the near future. The period in which the jitter buffer is temporarily raised, while jitter values are reassessed, is indicated by the state “a3” on the connection status screen.

**Very Important Notes:**
The Auto-jitter Priority/preference can be changed ‘on the fly’ if conditions and audio output quality dictate such an action. Auto jitter buffer is disabled for the Raw algorithm.

### 2.18.5. Different Auto Jitter Buffer Values

It is likely that auto jitter buffer values on two codecs may vary over a connection. This is because the data uplink bandwidth over a 3G cell-phone network can vary greatly depending on the following factors:

- Cell-phone data is asymmetrical and data uplinks are lower in bandwidth than data downlinks;
- Cell-phone data uplink capabilities vary from network to network;
- Cell-phone data uplinks are dynamic and may vary depending on how many users are connected to the network.

As a result, the auto jitter buffer value at the studio codec may be significantly higher than at the field codec. For example, it is not unusual to see a value of say 150 milliseconds at the field codec and 300 milliseconds at the studio codec. This is so that there is a sufficient jitter buffer in place for the studio codec to reliably receive all the data being sent in varying network conditions.

### 2.18.6. Auto Jitter Buffer and FEC

FEC uses data in the jitter buffer to recover lost packets. There is no need to modify jitter buffer settings if you are sending FEC data, only if you are receiving FEC data. FEC requires added jitter buffer time to work successfully so the use of FEC will increase jitter buffer depth.

Tieline’s auto jitter buffer detects the amount of FEC that is being used and automatically compensates to increase the codec jitter buffer if FEC is being used.

**Please note:** There is an inverse relationship between FEC settings and the jitter-buffer millisecond setting that you use for IP connections.

### 2.18.7. Manually Programmed Jitter Buffer Settings

Manual programming of fixed jitter buffer settings is only recommended if the jitter is predictable over a network. It may also be useful in situations where:
Audio from a codec needs to be synced to a video feed;  
Audio from a codec needs to be synced to other audio feeds; or  
Latency is not an issue and maximum audio quality is desired, at the expense of higher delay.

When manually programming the jitter-buffer delay in a codec it is necessary to think carefully about the type of connection you will be using. Following is a table displaying rule of thumb settings for programming jitter-buffer delays into your codec.

<table>
<thead>
<tr>
<th>Connection</th>
<th>Jitter-Buffer Recommendation</th>
</tr>
</thead>
<tbody>
<tr>
<td>Private LAN</td>
<td>60 milliseconds</td>
</tr>
<tr>
<td>Local</td>
<td>100 - 200 milliseconds</td>
</tr>
<tr>
<td>National</td>
<td>100 - 300 milliseconds</td>
</tr>
<tr>
<td>International</td>
<td>100 - 400 milliseconds</td>
</tr>
<tr>
<td>Wireless Network</td>
<td>250 - 750 milliseconds</td>
</tr>
</tbody>
</table>

Table 5: Jitter Buffer Recommendations

Please note: The preceding table assumes the use of either Tieline Music or Voice G3 algorithms. Do not use Raw Audio over DSL/ADSL connections.

If you connect initially with a 500 millisecond jitter-buffer, adjust it downwards until the connection becomes unstable. This can be adjusted while you are connected and will help to minimize program delay.

Once you have determined where a connection becomes unstable, increase the jitter-buffer so that there is an acceptable amount of ‘jitter-buffer headroom’, to allow for any network congestion that may occur. We recommend a minimum of 60 milliseconds from the point that a connection becomes unstable.

Jitter-buffer readings are visible by pressing the ENTER/DIAL button or the MENU SELECTOR on the codec once connected. This reading will reflect the setting programmed into the codec but it will rarely be exactly the same figure as that programmed. This is because it reflects current codec jitter buffering and will be in a range just above or below the setting programmed.
The default jitter-buffer setting in Tieline codecs is 500 milliseconds. This is a very reliable setting that will work for just about all connections. However, this is quite a long delay and we recommend that when you set up an IP connection you test how low you can set the jitter-buffer in your codec.

If you change the jitter buffer setting in a codec it will only adjust to the new level when link quality is high (e.g. above 70%). This is done to ensure audio quality is not compromised.

**Important Note on FEC:**
If FEC is being used we recommend that approximately 100ms be added to the jitter buffer on a codec receiving FEC at a setting of 20%, and 20ms at a setting of 100%.

### 2.19. Set the TCP Session Port and (UDP) Audio Port

Depending on whether you choose to connect using TCP or UDP, you may need to adjust the Session Port and Audio Port settings within the codec. The Session Port and the Audio Port are used in both TCP and UDP connections. Detailed information relating to these ports follows.

#### 2.19.1. Codec TCP and UDP Connection Ports

In TCP/IP and UDP networks the codec port is the endpoint of your connection. Software network ports are doorways for systems to communicate with each other. For example, several codecs in your studio may use the same public static IP address. Therefore it is necessary to allocate port numbers to these codecs so that when an incoming call comes in, the network knows which codec to send the call to.

Picture a house and imagine the front door is the entry point represented by an IP address. You want to get to several codecs in different rooms of the same house and the doors to each of those rooms are represented by different port numbers. In principle this is how port addressing works.

When data is received from several field unit codecs at a particular public IP address, port information is translated from data packets to ensure the correct packets are sent to the correct studio codecs. This process is performed by PAT (Port Address Translation), which is a feature of NAT (Network Address Translation) devices. For more info on PAT, please go to the Wikipedia page at [http://en.wikipedia.org/wiki/Port_address_translation](http://en.wikipedia.org/wiki/Port_address_translation)

#### 2.19.2. Tieline Codec Default Port Settings

As a default, Tieline codecs use a TCP session port to send session data and either a TCP or UDP port to send audio. The audio port is configured when you select the audio protocol you wish to connect with. The reason the session port always uses the TCP protocol is that TCP is the most likely protocol to get through firewalls – ensuring critical session data (including dial, connect and hang-up data) will be received reliably.
When the dialing codec makes a call, the session data from that codec configures both the local and remote codec session ports, as well as the audio ports, with the settings on the dialing codec. IP1 and IP2 connection ports are configured separately.

The default session and audio port settings for both TCP and UDP connections are the same and are displayed as follows.

<table>
<thead>
<tr>
<th>IP Connection</th>
<th>Audio Protocol</th>
<th>TCP Session Port (Used to send session data)</th>
<th>Audio Port (Used for audio)</th>
</tr>
</thead>
<tbody>
<tr>
<td>IP1</td>
<td>TCP</td>
<td>9002</td>
<td>9000</td>
</tr>
<tr>
<td>IP1</td>
<td>UDP</td>
<td>9002</td>
<td>9000</td>
</tr>
<tr>
<td>IP2</td>
<td>TCP</td>
<td>9012</td>
<td>9010</td>
</tr>
<tr>
<td>IP2</td>
<td>UDP</td>
<td>9012</td>
<td>9010</td>
</tr>
</tbody>
</table>

Table 6: Default Tieline TCP/IP and UDP/IP Port Numbers

Following is a list of the ports used for all codec connections.

<table>
<thead>
<tr>
<th>Default Tieline Ports</th>
<th>Application</th>
<th>Session Port</th>
<th>Audio Port</th>
</tr>
</thead>
<tbody>
<tr>
<td>IP1</td>
<td>TCP 9002</td>
<td>UDP 9000</td>
<td></td>
</tr>
<tr>
<td>IP2</td>
<td>TCP 9012</td>
<td>UDP 9010</td>
<td></td>
</tr>
<tr>
<td>Tieline Toolbox &amp; TLG3GUI</td>
<td>UDP 5550</td>
<td>UDP 5500</td>
<td></td>
</tr>
<tr>
<td>SIP</td>
<td>UDP 5060</td>
<td>UDP 5004</td>
<td></td>
</tr>
</tbody>
</table>

Using a port scanner to test a codec will be unsuccessful if you try to scan and the port is already in use, i.e. the codec is connected.

2.19.3. Changing Codec Port Numbers

The previous table displays the default ports used by Tieline codecs for TCP and UDP connections. There are a few reasons why you may wish to adjust the port setting on your codec. These include:

- Creating individual port settings for gateways and firewalls;
- Another device is already using a codec’s port number; or
- More than one studio codec is in use and each codec requires a different port number. For a studio and remote codec to connect, they need to be programmed with matching port numbers.

If there is a need to change codec port settings, in most situations you should consult your organization’s resident IT professional. Following is a diagram displaying codec port configurations for a scenario whereby multiple studio codecs use a single static public IP address to connect to different remote broadcast codecs.

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4 If you have connected successfully via UDP/IP connections but you are not hearing any audio, please check your firewall settings with your system administrator to see if the UDP audio port is open.
Figure 14: Multiple Codec Configuration using a Single Static Public IP Address

Please note: In the configuration displayed, each remote site broadcast codec is connected to a different studio codec using one IP connection. Each remote codec has been programmed with the port numbers of the destination codec it is dialing into at the studio.
2.19.3.1. TCP Session Port Settings
TCP Session Ports need to be configured for both TCP and UDP audio connections, as displayed in the table previously. If you are operating multiple codecs in the studio using the same IP address, you can change the TCP session port number by accessing the codec menu by selecting [Menu] > [Configuration] > [IP1 Setup] > [TCP Session Port] (you can also select [IP2 Setup] if you would prefer to configure this connection). Press CLEAR several times to delete the old port number and use the codec keypad to insert a new port number. Then press OK to enter it.

2.19.3.2. UDP Port Settings
For TCP and UDP connections, it is also necessary to configure the audio port. To do this select [Menu] > [Configuration] > [IP1 Setup] > [Audio Port]. Then Press CLEAR several times to delete the old port number and use the codec keypad to insert a new port number. Then press OK to enter it.

2.20. Dial Interface
This menu is only visible if you have two active IP connections – one over a LAN and one over a 3GIP connection. For example, this may be the case if you are using two IP connections, one as the primary broadcast connection and one as a standby connection. This scenario may also occur if you are sending dual mono signals to two separate locations.

A codec is programmed by default to operate in Auto mode, whereby a codec will by default dial over a 3G interface if it is present, in preference to a LAN IP connection. However, it is possible to force a codec to select which connection will dial using which IP interface.

To force the codec to use a particular interface select [Menu] > [Configuration] > [IP1 or IP2 Setup] > [Dial Interface] > [LAN or 3G connection]. Select the interface you wish to use for both IP1 and IP2 connections as desired.

2.20.1. Dialing IP Connections
Dialing is done in the normal way and you can use the * or # button on the codec keypad to enter the periods (.) in the IP address. It is important to remember that in all configurations (except the dual mono profile) you cannot initiate an outbound call from a studio codec over the Internet to a remote codec. Think of the PBX example again. You can dial from a private extension number to a public number, but you cannot dial from a private number to another private number because the public network will not recognize the call without PBX translation.

Your remote field codec will normally be assigned a private LAN IP address from which you can initiate a call to your public IP address at the studio and
NAT will successfully route your call to the private IP address of the studio codec, but this will not work the other way.

The only time you will need to dial from the studio codec to the remote codec is if you are using the dual mono program profile. If you are using a dual mono profile, you will need to use static public IP address in each remote codec. This is because the codec that makes calls to two different IP addresses needs to be the dialing codec – in this case the studio codec. Remember that if you are making a call you need to dial a public IP address. That is why the remote codecs require static public IP addresses when you use this profile. This type of connection is discussed in the section of this manual titled Detailed Instruction on how to Configure a Codec at a Remote Broadcast Location.

2.21. Testing IP Connections

If you are unable to connect after repeated attempts, please contact Tieline customer support at support@tieline.com. If you are unsure of any of the settings described in this manual, contact your LAN Administrator.

For an overview of broadcasting over LANs, WANs and the Internet, visit www.tieline.com/ip

2.22. Suggested Reading

If you are just starting out and require further information about IP connectivity, we suggest the following book as a good additional reference source.

**Title:** TCP/IP Jumpstart™: Internet Protocol Basics, Second Edition  
**Author:** Andrew G. Blank  
**Publisher:** Sybex® Inc., 1151 Marina Village Parkway, Alameda, CA 4501; www.sybex.com
Section 3. An Introduction to 3G Broadcasting

The services available over Third-generation (3G) cell-phone networks provide the ability to make both voice and non-voice data calls. By using a data service with a Tieline 3G module, or a 3G cell-phone connected to a USB module, you can deliver high quality audio from remote location. This also provides maximum flexibility in locating a codec at a difficult or mobile remote location.

The main advantages of 3G networks are the increased voice capacity and the higher data rates provided. Multicast services can be utilized and you can send video and audio, as well as data, information, emails and instant messages.

One of the major advantages of 3G networks is the increased connection bandwidth available in comparison to GSM cell-phone connections. This provides greater flexibility and higher quality broadcast audio.

Connections of up to 1920 kbps can be achieved over 3G networks. Most connections in the real world are up to 384 kbps for data downlinks and uplink speeds of up to 144 kbps can be achieved.

Unfortunately, in the real world network providers often limit these speeds and you should check with your service provider to see what uplink and downlink speeds you can achieve. This can also vary from country to country.

3.1. 3G – How to Get the Best Results

The purpose of this document is to assist Tieline codec users to connect over 3G cell-phone networks. It contains a brief overview of how to connect using different types of 3G cell-phone networks and some useful information in relation to:

- Ordering the right plan for your 3G service;
- Choosing the right ISP;
- Overview of 3G standards and EV-DO and UMTS/HSDPA modules;
- 3G bands and the selection of appropriate antennae;
- Choice of the most appropriate algorithm;
- 3G Data Usage; and
- Batteries available for remote broadcasting.

3.2. Overview of 3G Standards

The International Telecommunication Union (ITU) has produced a standard called IMT-2000 which consists of five operating modes. The two dominant modes are W-CDMA (Wideband Code Division Multiple Access) and CDMA2000® and these are the standards used in Tieline 3G modules. Following is some information on these standards.
3.2.1. W-CDMA

W-CDMA is the technology behind the UMTS (Universal Mobile Telecommunications System) standards for 3G. It is compatible with 2G CSD and HSCSD GSM standards and 2G packet switched applications such as GPRS and EDGE.

UMTS standards are being implemented in Europe, the United States, Australia and Japan. W-CDMA is not compatible with the CDMA family of standards, which includes CDMA2000®.

3.2.1.1. HSDPA (High-Speed Downlink Packet Access)

HSDPA is commonly referred to as 3.5G and extends UTMS technology to provide higher data uplink and downlink bitrates than traditional W-CDMA.

Maximum network download speeds of up to 3.6Mbps and upload speeds of up to 384kbps can be achieved over HSDPA networks (future download data rates of up to 14.4 Mbps are expected).

Speeds vary from network to network are also be affected by the hardware used (i.e. type of antenna) and environmental factors. Real-world connections should be capable of download bit rates of around 500-1100 kbps. Both UMTS and HSDPA networks provide low latency of around 100 to 200 milliseconds.

3.2.1.2. HSUPA (High-Speed Uplink Packet Access)

HSUPA is commonly referred to as 3.75G and extends HSDPA technology by increasing uplink speeds up to 5.76 Mbps.

3.2.1.3. HSPA (Evolved HSPA, HSPA Evolved, HSPA+, I-HSPA)

Evolved HSPA is a 3G mobile data protocol that provides data rates up to 42 Mbps on the downlink and 11Mbps on the uplink.
3.2.1.4. HSOPA
HSOPA succeeds HSDPA and HSUPA technologies and is unrelated to and incompatible with W-CDMA technology. It promises to provide peak transfer rates of 100Mbps for the downlink and 50Mbps for the uplink.

3.2.2. CDMA2000®
There are several CDMA2000® standards and they are based upon CDMA principles and are backward-compatible with older CDMA technologies.

The CDMA2000 1X (also known as 1xRTT) standard is currently implemented throughout the Americas, much of Asia, Australia and New Zealand. CDMA2000 1X is designed to operate on all the existing allocated spectrum for wireless CDMA communications and can provide both voice and data services over a standard connection.

3.2.2.1. CDMA2000 1xEV-DO (1x Evolution Data Optimized)
Like HSDPA, CDMA2000 1xEV-DO provides higher capacity data transfer speeds than its technological predecessor - CDMA2000 1x. CDMA2000 1xEV-DO supports data downloads of up to 3.1 Mbps and data uplinks of up to 1.8 Mbps.

For more information on CDMA2000® standards please see visit the following website http://www.evdoinfo.com/

3.2.3. Tieline 3G Modules
It is simple to connect over 3G networks and send high quality audio using Tieline 3G modules. Tieline has developed two plug-in 3G modules, one which connects over UMTS and HSDPA-capable 3G networks and the other connects over CDMA EV-DO 3G networks.

Tieline 3G modules make it possible to roam globally and seamlessly connect to the network that is most suitable to your environment. Depending on your geographic location, you can select the best network option that suits your circumstances.

3.2.4. Detailed 3G UMTS/HSDPA Module Features
The Tieline 3G UMTS/HSDPA module has the following features:

- Connects to all three UMTS/HSDPA frequency bands: 850MHz; 1900MHz; 2100MHz;
- Connects to all four EDGE/GPRS frequency bands: 850MHz; 900MHz; 1800MHz; 1900MHz;

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5 CDMA2000® is a registered certification mark of the Telecommunications Industry Association.
Peak download data rates of up to 3.6 Mbps and upload data rates of up to 384kbps (dependent on network capabilities);

Data connection rates are highest over HSDPA networks. If coverage is not available then the Tieline 3G UMTS/HSDPA module can connect to widely available UMTS networks. The module can seamlessly switch between these networks without a call being interrupted.

The Tieline TL3GHSDPA, 3G HSDPA/UMTS/GSM module supports:

- HSDPA;
- UMTS;
- 3G Voice;
- GSM CSD (Commander G3 Field and i-Mix codecs only); and
- GSM Voice (Commander G3 Field and i-Mix codecs only).

This module does not support GSM HSCSD data calls in any codec.

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**Important Warning:**
No GSM CSD, HSCSD or voice call capability is supported on a rack unit TLR300B Commander codec using a GSM SIM card in a GSM module, or using a 3G UMTS/HSDPA module. In addition, if “analog voice” is selected in a codec when using a 3G SIM card then a 3G voice call will be dialed.

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### 3.2.4.1 Troubleshooting UMTS/HSDPA Connections

If there is no SIM card in a UMTS/HSDPA module when it is inserted into a codec, the user is prompted with the message “Insert SIM” on the main connection screen. If a SIM card is locked the screen will display “SIM Locked”. A SIM cannot be unlocked by a codec so ensure any SIM cards used are unlocked.

If you need to contact Tieline customer support for assistance with a UMTS/HSDPA 3G module it is useful to take note of relevant information listed in the Unit Details menu of the codec. This information may be required to debug the connection issue.

Select [Menu] > [Unit Details] and scroll to find information relating to:

- The “3G Carrier”;
- The “3G IMEI” number (a unique number used to identify valid devices over a GSM/UMTS/HSDPA cell-phone network);
- The “3G SIM Lock” status; and
- ‘3G Module Firmware’.

When contacting Tieline customer support at support@tieline.com please supply this information to assist in debugging the issue.
### Important Warning:
If a module switches between different networks and the current connection bit rate is not supported by the new network, the connection will be interrupted. The connection bit rate will need to be renegotiated downwards until the currently supported network connection bit rate is found.

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### 3.2.5. Detailed 3G EV-DO Module Features

The **Tieline** 3G EV-DO module has the following features:

- Provides data services over:
  - CDMA 1xEV-DO Revision A networks;
  - CDMA 1xEV-DO Rel 0 networks; and
  - CDMA 1xRTT networks.

- Dual band compatible with:
  - 800MHz North American Cellular Band; and
  - 1900MHz North American PCS Band.

- Supports RX diversity in both the 800MHz and 1900MHz bands;

- Peak download data rates of up to 3.1 Mbps and upload rates of up to 1.8 Mbps (dependent on network capabilities);

Data connection rates are highest over CDMA 1xEV-DO Revision A networks. If coverage is not available then the **Tieline** 3G EV-DO module can connect to the widely available CDMA 1xEV-DO Rel 0 and CDMA 1xRTT networks. The module can seamlessly switch between these networks without a call being interrupted.

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### Important Warning:
If a module switches between different networks and the current connection bit rate is not supported by the new network, the connection will be interrupted. The connection bit rate will need to be renegotiated downwards until the currently supported network connection bit rate is found.

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### Please Note:
In order to connect to Sprint and Verizon EV-DO networks in the U.S.A., **Tieline** EV-DO modules must be programmed via a codec. The procedure for completing this process follows.

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### 3.2.6. Procedure for Programming 3G EV-DO Modules

**Tieline** CDMA EV-DO 3G modules do not use a SIM card but they need to be programmed in order to connect to either the Sprint™ or Verizon™ cell-phone networks in the U.S.A. There are two different EV-DO modules available – one for Sprint and one for Verizon. Each of the modules requires

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6 Sprint is a trademark of Sprint Nextel in the U.S. and/or other countries.

7 The Verizon and Verizon Business names and logos and all other names, logos, and slogans identifying Verizon’s products and services are trademarks and service marks or registered trademarks and service marks of Verizon Trademark Services LLC or its affiliates in the United States and/or other countries.
‘factory provisioning’ for the particular network it will be connecting to. For this reason it is necessary to choose your preferred network provider before purchasing an EV-DO module.

Two additional steps are required to get ‘on the air’ once you have selected, purchased and received a module:

1. Activate the module using a Tieline codec; and
2. Perform IOTA (Internet Over The Air) provisioning after activation. This is also performed using a Tieline codec.

Once you have decided upon a carrier and purchased a module contact your cell-phone network provider and open an account. The procedure for opening an account for a new EV-DO module and then activating it is as follows.

**Important Note:** EV-DO menus are only visible in Tieline codecs if an EV-DO module is in the unit. If you insert a 3G module and no EV-DO menus appear, check that the version of firmware in the codec is version 1.5.xx or higher. If the firmware version is correct and no menus appear then the module is probably an HSDPA module.

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### 3.2.6.1. Programming a New EV-DO Module

1. Find out the ESN (Electronic Serial Number) of your module. To discover this power down your codec, insert the EV-DO 3G module and power the codec up. Go to [Menu] > [Unit Details] and then scroll with the **MENU SELECTOR** until you see the listed **EVDO ESN** number.

![Figure 15: ESN Number in the Unit Details Screen](image)

2. Contact your service provider and:
   a. Tell them the ESN number; and
   b. Request that an account be opened.

3. You will need a credit card to open the account and your service provider will provide you with three numbers to program into your codec module to activate it and complete IOTA provisioning. The three numbers include:
   a. The MSL Code (sometimes called the ‘Activation Code’ or the ‘Unlock Code’);
   b. The MDN (Mobile Directory Number) of the module SIM card. This is the actual phone number of the card; and
c. The MIN (Mobile Identification Number), also sometimes referred to as the MSID or the IMSI.

**Important Note:** The telco may also provide additional numbers called the “System ID” and “Network ID”. If these numbers are provided take note of them and program them into the codec as well.

4. Once you have these numbers for your account go to the codec and select [Menu] > [Configuration] > [GSM/3G Setup] > [EVDO Activation]. The following screen will then briefly appear:

![Unlock Code Screen](image1)

Figure 16: Unlock Code Screen in the Activation Wizard

5. Next select [Activate] to open the ‘activation wizard’ and program the six digit MSL Unlock Code (Activation Code) into the codec using the keypad. This is one of the codes supplied by your cell-phone provider. Once entered press SOFTKEY 4 OK to continue.

![MDN Screen](image2)

Figure 17: Cell-Phone Number Screen in the Activation Wizard

6. Next enter the MDN (the cell-phone number) for the module using the codec keypad. This should also have been supplied by your cell-phone provider. Once entered press SOFTKEY 4 OK to continue.

![Mobile ID Screen](image3)

Figure 18: Mobile Identification Number Screen in the Activation Wizard

7. Next enter the Mobile Identification Number (MIN, MSID or IMSI) supplied by your cell-phone provider. Once entered press SOFTKEY 4 OK to continue.
8. If a ‘System ID’ number has been supplied by your cell-phone network provider enter this into the next screen of the activation wizard. If no number is provided leave it as the default setting of “0”. Press **SOFTKEY 4 OK** to continue.

![Figure 19: Default System ID Screen in the Activation Wizard](image)

9. If a ‘Network ID’ number has been supplied by your cell-phone network provider enter this into the next screen of the activation wizard. If no number is provided leave it as the default setting of “65535”. Press **SOFTKEY 4 OK** to continue.

![Figure 20: Default Network ID Screen in the Activation Wizard](image)

10. The next screen in the wizard actually activates the module. Press **SOFTKEY 2 YES** to continue and activate the module.

![Figure 21: Initiate Activation Screen](image)

11. The codec should then proceed to activate the module and provision IOTA (Internet Over The Air). This is a two-stage process and the following screens should appear.

![Figure 22: Module Activation Screen](image)

The second screen that displays while the codec completes IOTA provisioning can remain for 1-2 minutes.
12. The process of activation and provisioning is complete when the following screen appears.

![Figure 24: Provisioning Completed Screen](image)

Press SOFTKEY 4 OK to return to the EVDO Activation menu in the codec. To view the settings programmed into the codec scroll down to the EVDO Details section in the menu and select this to view the programmed EV-DO settings.

### 3.2.6.2. Troubleshooting Activation and Provisioning

Activation and provisioning can fail for several reasons. The following screen appears if activation of a module fails.

![Figure 25: Activation Failure Screen](image)

The activation process may fail if the wrong activation codes have been programmed into a codec. Check these by selecting the EVDO Details section in the EVDO Activation menu. This can be viewed from the main connection screen via [Menu] > [Configuration] > [GSM/3G Setup] > [EVDO Activation] > [EVDO Details].

If the wrong codes have been entered, use the activation wizard to re-enter the codes and reactivate the module. If provisioning fails (the stage after activation where the module is configured for data calls) the following screen appears:
The reasons that provisioning may fail include:

1. There is no network coverage available; or
2. The account has not been set up by your cell-phone network provider.

Network coverage can be checked by looking at the signal strength of the module’s 3GIP connection on the main connection screen of the codec.

If signal strength is good (5-6 on the codec LCD display) it is most likely that the account has not been set up correctly. Call your cell-phone network provider and check that data capability has been configured on your account. In some cases an ‘account reset’ by your provider may solve the problem.

If after trying all this you are still having issues with either activation or provisioning, contact Tieline customer support for assistance at support@tieline.com.

3.2.6.3. Previously Programmed EV-DO Modules

If you are not sure if a module has been programmed previously you can check the status of the module in the Unit Details menu via [Menu] > [Unit Details] and then scroll with the MENU SELECTOR until you see the EVDO Status of the module. The three possible status states include:

1. Not Provisioned: The module has been activated for voice calls but not for data calls.
2. Not Activated: The module has not been activated for voice or data calls.
3. Active: The module is programmed correctly for data calls and no additional programming is required.

If a module has been programmed previously then there are three possible scenarios that may occur.

1. If a module has been used previously in another codec then the card within the module may already have been activated for data calls. In this situation there should be no need to activate the card.
2. If a module has been used previously for voice calls only then it may need to be ‘ provisioned ’ for data calls.
3. If the module in a codec has been used by someone else using another data account then the module may need to be reactivated. This is because the module is going to be used with a different cell-phone data account.
3.2.6.4. Troubleshooting EV-DO Connections

If you need to contact Tieline customer support for assistance in activating and provisioning an EV-DO 3G module it is useful to take note of the EV-DO information listed in the Unit Details menu of the codec. This information will be required to debug activation and provisioning.

Select [Menu] > [Unit Details] and scroll to find information relating to:

- The ‘EV-DO Provider’;
- The ‘EV-DO ESN’;
- The ‘EV-DO status’; and
- ‘EV-DO Module Firmware’.

When contacting Tieline customer support at support@tieline.com please supply this information to assist in debugging the issue.

3.3. Advanced Programming: 3G Band Selection (HSDPA Modules Only)

**Important Information:** These menus are not available in 3G EV-DO modules or when using USB modules with HSDPA cell-phones.

Tieline codecs are programmed to automatically choose the most appropriate band for a connection. Europe and Asia both use only the 2100 MHz band for 3G and there shouldn’t be any need to configure 3G band settings within a codec. As a guide, the default “Auto” setting in Tieline codecs will:

- Use a GSM network for CSD connections;
- Use a GSM or WCDMA network for GSM voice calls; and
- Use a WCDMA network for 3GIP connections.

In most situations this should not need to be adjusted. Tieline codecs are programmed to automatically choose the most appropriate band for a connection. In the US the situation is slightly different and cell-phone networks, such as Cingular\(^8\) and T-MOBILE\(^9\), use different bands. This shouldn’t be an issue because a Cingular SIM card will only ever find the Cingular network, and similarly for T-MOBILE etc.

If more than one band is available, the fastest available network will be selected for the codec connection.

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\(^8\) AT&T, AT&T logo, Cingular and Cingular logos are trademarks of AT&T Knowledge Ventures and/or AT&T affiliated companies.

\(^9\) T-MOBILE is a registered and/or unregistered trademark of Deutsche Telekom AG in the US and/or other countries.
Important Warnings:
One dangerous and potentially expensive scenario can eventuate if a codec has been changed from the factory default band setting. If a user has chosen a band override setting that forces the card onto a GSM band – then the codec would use GPRS or EDGE for what should be a 3G connection. If the setting is programmed as ‘Auto’ this will not happen.

Do not change this setting in a codec using ToolBox software before contacting support@tieline.com. This is because a regional-specific number will need to be entered when programming specific bands using the ToolBox software interface.

To check a codec’s settings are correct when a 3G connection is established, the band and network type are displayed on the connection details LCD screen (pressing ENTER on the 3GIP connection). If the user has played with the band override setting, they should verify on every connection that they are using UMTS or HDSPA, and not GPRS or EDGE.

3.3.1. Other 3G Band Settings
It is possible to force an HSDPA module to select certain bands only. This is not normally necessary and is most likely to be required for troubleshooting specific connection issues. Following is description of the selection options available.

3.3.1.1. All bands
This leaves the band selection up to the 3G module. The band is unpredictable. Generally the module will try a WCDMA band first and fall back to a GSM band, but this cannot be relied upon. The band will (typically) remain the same if the user switches between GSM modes (e.g. changing from “GSM Voice” to “3GIP”).

3.3.1.2. WCDMA ALL
Only WCDMA (3G) bands will be used. GSM bands will not be selected. This will prevent the use of GPRS or EDGE (which are often billed at very high rates).

3.3.1.3. GSM ALL
Only GSM bands will be used.

3.3.1.4. Others
The remaining options depend on the region that the module has been configured for. This regional configuration must be done at the factory, and adapts both the network scanning algorithm, as well as the set of available bands for the networks in that region. The region can be found in “Unit Details” under “Module Firmware”. The firmware version will typically start with “H1_1_8_3MCAP,XX”. These last two letters indicate the region the module has been configured for. It is possible that a module may not have been configured for a specific
region, in which case these last two letters will not be present. The list of possible regions follows.

**AU (Australia):**
- GSM ALL
- WCDMA ALL
- WCDMA 850 GSM 900/1800
- WCDMA 850

**EU (Europe):**
- WCDMA 2100
- GSM 900/1800
- GSM ALL
- WCDMA ALL

**JP (Japan):**
- GSM ALL
- WCDMA ALL
- WCDMA 800/2100

**No region:**
- WCDMA 2100
- WCDMA 850/1900
- GSM 900/1800
- GSM 850/1900
- GSM ALL
- WCDMA 2100 GSM 900/1800
- WCDMA 850/1900 GSM 850/1900
- WCDMA ALL
- WCDMA 850/2100
- WCDMA 800/2100
- WCDMA 850/2100 GSM 900/1800
- WCDMA 850 GSM 900/1800
- WCDMA 850

### 3.3.2. Antennae

The antenna is your 3G lifeline. The quality and gain of your antenna will make the difference between a good and average connection. **Tieline** supplies a multi-band antenna, which is designed to work over many different frequencies in most countries around the world. We have discovered that each telco around the world usually recommends a specific single band antenna, which is optimized for their specific network properties.

We recommend purchasing a high-gain single or dual-band antenna recommended by your telco for optimal results. For telcos that support 2100 MHz and above, we recommend a high gain antenna. You'll really see the difference when you get into low signal strength areas. Signals can be received on both antennae but are transmitted on the first antenna only. The
second connection is designed to support a coaxial cable connection to the second antenna.

The antenna you use should have an SMA type connector. When using two antennae it is recommended that they are at least a quarter of a wavelength apart. If this is not possible, try to vertically polarize one antenna and horizontally polarize the other.

### 3.3.2.1. Multi-Band Antenna

**Tieline** 3G EV-DO and 3G UMTS/HSDPA modules are factory-fitted with a standard 5-band antenna. It is also possible to fit an antenna which is more specific to the frequency bands of the module you are using and achieve better performance.

### 3.3.2.2. EV-DO Compatible Antenna

An antenna specifically suited to EV-DO bands should be connected to the **Tieline** 3G EV-DO module. Select an antenna to suit the band in your region.

- 800MHz North American Cellular Band; or
- 1900MHz North American PCS Band.

Antenna gain, including cable loss, should not exceed:

- 4.15dBi for the 1900MHz North American PCS Band, and
- 5.1dBi on other EV-DO bands.

### 3.3.2.3. UMTS/HSDPA Compatible Antenna

An antenna specifically suited to UMTS/HSDPA bands should be connected to the **Tieline** 3G UMTS/HSDPA module. Select an antenna to suit the band in your region, which may include:

- 850MHz;
- 1900MHz; or
- 2100MHz.

Antenna gain, including cable loss, should not exceed:

- 4dBi for the 1900MHz PCS Band, and
- 8dBi on other UMTS/HSDPA bands.
3.3.3. Choosing the Best Antenna for the Network

Significant efficiencies in transmission and reception of RF can be achieved by simply matching your 3G module with an antenna (50ohm antenna load impedance) that is best suited to the 3G network you are connecting to.

The HSDPA module uses the “Main” antenna port only and the “Aux” port is programmed for GPS use (which is not supported).

EV-DO 3G modules support the use of one or two antennae. If used, the second connection creates a diversity system whereby signal reception is improved to minimize dropouts and maximize network connectivity and signal strength. This is particularly useful in mobile remote broadcast situations.

3.4. USB Module Software Versions

**Tieline** USB modules support USB 2.0 high-speed connections. If you are using a USB module with either a USB modem or a cell-phone to connect over 3G, it is also important to know that there are currently three different software versions available in **Tieline** USB modules. The simplest way to avoid incompatibilities is to use a **Tieline** 3G module which connects over UMTS and HSDPA 3G networks seamlessly.
With USB software versions 1.0.2 and 1.0.4 some cell-phones will connect in either one version or the other, depending on whether they are UMTS 3G phones only or whether they are HSDPA-compatible or not. Unfortunately this is largely a case of trial and error.

The software version of a module inserted in a codec can be viewed in the Unit Details by selecting SOFTKEY 4 Menu and choosing Unit Details > [USB Host 1> Version 1.0.2/4/9]. The USB host will display as Host 1 if a module is in the left side of a codec and as Host 2 if a module is in the right side of a codec.

The difficulties with compatibility are compounded now that EV-DO networks are supported. As a result we recommend that all users of IP and 3GIP networks use firmware version 1.6.xx in codecs as this will automatically upgrade all USB modules to version 1.0.9 when installed in a codec. This version supports UMTS, HSDPA & EV-DO connections.

3.4.1. Connecting over 3G using a USB Module and 3G Modem

USB modems can be used to connect over 3G networks. Usually they attach directly into a codec’s USB module, but sometimes they require a cable to connect. UMTS/HSDPA modems require a SIM card. Remember to disable the PIN on the SIM card as it cannot be unlocked by a codec.

It is best to check the manufacturer’s specifications for each USB modem, particularly in relation to whether they can connect over GSM CSD and HSCSD bands.

Important Compatibility Note:
Make sure a 3G modem does not draw any more than the USB standard of 500 mA otherwise it may not work effectively. Some have been tried that draw up to 700 mA.

3.4.1.1. Compatible USB Modems

In the USA the following USB modems have been used successfully to connect over 3G networks:

1. Sierra Wireless Aircard 875U (HSDPA); and
2. Sierra Wireless Aircard 595U (Sprint & Verizon EV-DO).

Please do not purchase a modem without checking with your local dealer for the latest list of compatible modems for the region you are operating in. If a dial prompt does not appear when a modem is inserted into a USB module it means that either the modem is not supported by Tieline, or the codec firmware has not been updated to v.1.6.xx.

It is important when purchasing a new USB modem that it be fully activated prior to putting it into a codec to make a 3G connection. To complete this process at a PC do the following:
1. Load the connection software located on the CD that comes with each USB modem.
2. Insert the USB modem into a PC (it may look for drivers on the CD or be pre-loaded by the connection manager software).
3. Simply follow the instructions that pop up in the software to finish activating the modem. This may require updating the user profile.

Once you have successfully connected to the 3G network via a PC and can connect to the Internet, you can then attach the USB modem to the USB module, set it for the correct network, and as long as it shows a good signal strength (preferably higher than “4”) connect and proceed to make an IP connection.

### 3.4.2. Connecting over 3G using a USB module and 3G Cell-phone

It is also possible to connect using a Tieline USB module and a compatible 3G cell-phone. To achieve uninterrupted connection over long periods, we recommend you purchase a mobile phone capable of being externally powered and connected via USB simultaneously.

![Figure 27: 3G Cell-phone Connected to a Tieline Codec](image)

### 3.5. How to connect over 3G with a USB Module & Cell-phone

Following is a series of issues that you need to be aware of when configuring a Tieline codec for 3G monitoring capability.

1. It is essential that you purchase a 3G phone that provides you with the opportunity to interface with a Tieline USB module. Check with your cell-phone manufacturer about whether this is possible.

Specifically, you need to ensure that you are purchasing a phone that is able to connect to a PC via USB and act as a modem for connection of a PC to the Internet. (This means your phone will need to support connecting to ACM USB devices in order to connect to the codec. This may be called ‘tethering’ or ‘phone as modem’ capability).
If your phone is able to connect via either Linux®\textsuperscript{10}, Mac®\textsuperscript{11} or Windows®\textsuperscript{12} standard dial-up networking, then there is a good chance it should connect successfully to Tieline codecs.

The **ONLY** way to be 100% certain your choice of cell-phone is going to work with a Tieline codec is to choose a model from the list of Tieline-tested cell-phones provided in the following table.

<table>
<thead>
<tr>
<th>Mobile Make</th>
<th>Mobile Model</th>
<th>Country</th>
<th>Service Provider</th>
<th>Comments</th>
</tr>
</thead>
<tbody>
<tr>
<td>Nokia</td>
<td>N70</td>
<td>Australia/Asia</td>
<td>3, Telstra,*Telstra/</td>
<td>*EVDO</td>
</tr>
<tr>
<td>Nokia</td>
<td>N70</td>
<td>UK/Europe</td>
<td>3/Vodafone</td>
<td></td>
</tr>
<tr>
<td>Nokia</td>
<td>U6680</td>
<td>South Africa</td>
<td>MTN</td>
<td></td>
</tr>
<tr>
<td>Sony Ericsson</td>
<td>K8001</td>
<td>Australia</td>
<td>3</td>
<td></td>
</tr>
<tr>
<td>Vodafone</td>
<td>702NK</td>
<td>Japan</td>
<td>Softbank</td>
<td>SIM lock Version of NOKIA 6630</td>
</tr>
<tr>
<td>LG</td>
<td>U8330</td>
<td>Australia</td>
<td>3</td>
<td></td>
</tr>
<tr>
<td>LG</td>
<td>U8360</td>
<td>Australia</td>
<td>3</td>
<td></td>
</tr>
<tr>
<td>LG</td>
<td>TU500</td>
<td>Australia</td>
<td>Telstra</td>
<td>HSDPA</td>
</tr>
</tbody>
</table>

The following cell-phones are only compatible in the U.S.A.

| LG          | CU320        | America           | Cingular         |                           |
| LG          | VX8100       | America           | *Verizon         | Simultaneous Power / USB (with optional data cable LG Part No: SGDY001060) *EVDO |
| LG          | VX8300       | America           | *Verizon         |                           |
| Motorola    | E815         | America           | *Alltel Axcess Broadband | Simultaneous Power / USB *EVDO |
| Samsung     | SPH-A900     | America           | *Sprint          | *EVDO Low Speed Cnx      |

Table 7: 3G-Compatible Phone Table

\textsuperscript{10} Linux® is the registered trademark of Linus Torvalds in the U.S. and other countries.

\textsuperscript{11} Mac is a trademark of Apple Computer, Inc., registered in the U.S. and other countries.

\textsuperscript{12} Windows is a registered trademark of Microsoft Corporation in the United States and/or other countries.
If you wish to use an existing model phone or purchase an alternate brand, guidelines are provided in the following table.

<table>
<thead>
<tr>
<th>Assurance</th>
<th>Method</th>
</tr>
</thead>
<tbody>
<tr>
<td>100%</td>
<td>Choose a model in the previous table.</td>
</tr>
<tr>
<td>80%</td>
<td>If the cell-phone can connect a Linux PC to the Internet using AT Commands over USB.</td>
</tr>
<tr>
<td>Add 5%</td>
<td>If the cell-phone product description includes words to the effect: “Modem to access the Internet via your PC”</td>
</tr>
</tbody>
</table>

Table 8: 3G Cell-Phone Compatibility Assurance Levels

If there is no SIM card in a UMTS/HSDPA module when it is inserted into a codec, the user is prompted with the message “Insert SIM” on the main connection screen. If a SIM card is locked the message will display “SIM Locked”. A SIM cannot be unlocked by a codec so ensure any SIM cards used are unlocked.

Please note: Some 3G cell-phones require extra software on a PC to use it as a modem. As your codec will not have this software, these phones are not able to be used for connecting your codec to a 3G network.

2. As a factory default, your new 3G capable cell-phone is probably programmed for GSM use. You will need to ensure that you change the mode of operation to 3G IP mode before attempting to connect your codec using the 3G network. Please consult the user manual for your cell-phone for instructions on how to do this.

3. Before you purchase a 3G phone be sure to check that it will operate in the regions you will be broadcasting in. In the USA the 1900 MHz and 850 MHz bands are used (depending on the region you are in) and the rest of the world uses the 2100 MHz band.

4. 3G cell-phones can be either ‘locked’ or ‘unlocked’ to networks. If your cell-phone is locked, you will only be able to use the SIM card of the network you are locked to. If you are only using your 3G phone in one particular region this may not be a major issue for you.

If you are planning to use the cell-phone across multiple regions and/or countries, you will probably require an ‘unlocked’ phone, allowing you to insert the SIM cards for different 3G networks.

Unlocked 3G cell-phones are available from on-line stores, airports and duty free shops – as well as some 3G network providers and phone shops.

5. You will need to purchase a data plan, either in addition to, or instead of, a voice call plan with your cell-phone network provider. You should opt for blocks of data for a monthly price, i.e. $20 per month for up to 300 megabytes of data.
6. In summary, the specific requirements for choosing a Tieline Codec 3G-compatible mobile phone are:

- ✔ USB connection to a PC and capable of USB 1.0 connection;
- ✔ Modem capability;
- ✔ Capable of connecting a PC to the Internet;
- ✔ 3G Data Service; and
- ✔ Internet Plan.

If you are unsure about a particular cell-phone or whether it can connect to Tieline codecs, contact Tieline at covert@tieline.com for more information. Email the model of your phone, the name of your service provider and the details of your data plan.
3.6. 3GIP Codec Menus in Detail

![Diagram of 3GIP Codec Menu]

*Figure 28: 3GIP Codec Menu*
3.7. Advanced Programming of 3GIP Connections using ToolBox

The **GSM LL/GSM/USB-3G** tab in ToolBox software allows users to configure GSM, GSM landline and USB-3G connection settings within Tieline codecs. Please note: Even if you don’t have ToolBox software connected to your codec, all these functions are available by accessing your codec menu via [Menu] > [Configuration] > [GSM/3G Setup].

This manual only covers the settings applicable for 3G connections. Please see the reference manual for your codec for information relating to GSM and GSM landline connections. The three sections relevant to 3G in ToolBox software include:

- **GSM/USB-3G Module Cellphone**
- **3G/UMTS IP Network Settings**
- **Advanced**

3.8. GSM/USB-3G Module/Cellphone Setup

3.8.1. Select GSM Interface (for a codec connecting with a GSM connection)

This drop-down menu allows a user to select between different module and connection combinations in a codec. The options are as follows:

<table>
<thead>
<tr>
<th>Setting</th>
<th>Module Types Supported</th>
</tr>
</thead>
<tbody>
<tr>
<td>GSM/USB-3G Settings A</td>
<td>Supports the use of either a GSM or USB-3G module only.</td>
</tr>
<tr>
<td>GSM/USB-3G Settings B</td>
<td>Supports the use of either a USB-3G Module or external cell-phone, with a GSM module.</td>
</tr>
</tbody>
</table>
To select the most appropriate connection scenario for your situation, simply click on the right-hand blue arrow, highlight the desired setting and click on it to select it.

**Important Note:** Only one GSM module can be used in a **COMMANDER** codec. If you wish to use two GSM connections you will also need to use the RS 232 connection on the rear of the codec and connect a GSM-compatible cell-phone such as the *Nokia®* 6310 and 6310i. If you install two GSM modules an error message will be displayed.

### 3.8.2. Wireless Network Type

This drop-down menu allows the appropriate network connection to be selected.

By default **Tieline** codecs are programmed to auto-detect the network that a module supports. If a GSM module is inserted into a codec it will automatically detect that the module is for GSM use and display a connection preprogrammed to make a 9,600bps GSM CSD call.

If a 3G or 3G-enabled USB module is inserted into a codec, it will automatically detect that the module is for 3GIP use and display a connection preprogrammed to make a 3G call. The easiest way to change any default settings is to use the codec configuration wizard.

To change the default setting, click on the drop-down menu arrow and select the network you require by highlighting and clicking on it. Following is an explanation of the settings in this menu.

<table>
<thead>
<tr>
<th>Codec Settings</th>
<th>Explanation of Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>AUTO-DETECT</td>
<td>Auto-detects a GSM, 3G or USB module inserted into a codec and configures the module for preprogrammed settings.</td>
</tr>
<tr>
<td>GSM CSD – POTS</td>
<td>Programs a codec GSM module (or 3G module in GSM mode) for a GSM CSD call to a POTS line.</td>
</tr>
<tr>
<td>GSM HSCSD – POTS</td>
<td>Programs a codec GSM module for a GSM HSCSD call to a POTS line.</td>
</tr>
<tr>
<td>3GCSD/PHS</td>
<td>This setting is used for 3GCSD connections in Japan.</td>
</tr>
<tr>
<td>3G/UMTS IP</td>
<td>Programs a codec USB or 3G module for a 3GIP call.</td>
</tr>
<tr>
<td>GSM VOICE –(Analog)</td>
<td>Programs a GSM module to make a voice call over a POTS line.</td>
</tr>
</tbody>
</table>

### 3.8.3. Signal Strength Enable and Reset Wait Seconds

*Signal Strength Enable* should be checked as a default. It ensures that the signal strength of your GSM connection will be displayed on the codec LCD screen. *Reset Wait Seconds* should be set to the default setting of 10 seconds because this relates to the time it takes to for the GSM cell-phone signal to
connect to the network. If your network doesn’t support signal strength being
displayed, disable Signal Strength Enable.

3.9. 3G UMTS IP Network Settings
This section is used to configure the 3G network that you connect over. Select your
network from the drop-down list of networks.

If your network is not in the list, see the What to do if a Network is Unavailable in
the Codec section in this manual. This describes in detail the procedure for adding
a network to the list in your codec. This feature can only be implemented using
ToolBox software.

3.10. Advanced Settings

3.10.1. 3G Band Override
This can be programmed within the codec menus and this is described in the
3G Band Override section of this manual. Do not change this setting using
ToolBox software unless you contact support@tieline.com.au beforehand.
This is because there are different numerical codecs required and these vary
depending on the region you are connecting from within.

3.10.2. 3G Idle Timeout (minutes)
The timeout setting represents the duration (in minutes) after a 3G/IP call
terminates, that the 3G connection automatically disconnects. The default
setting is 5 minutes and a ‘0’ turns this function off completely.

3.11. Battery Kit
The Tieline TLBMOD lithium ion battery is a 2300mAH hour module which will
power a Commander G3 Field codec for around 2 hours and 15 minutes over a
An Introduction to 3G Broadcasting

wireless connection with good signal strength. Lower signal strength may affect battery life.

3.12. 3G Data Usage Table
Following is a table which is helpful for calculating approximate data usage while connecting over 3G mobile networks using Teline codecs.

<table>
<thead>
<tr>
<th>CODEC BITRATE</th>
<th>FORWARD ERROR CORRECTION % (FEC)</th>
<th>TOTAL BITRATE (±/ +2%)</th>
<th>TOTAL USAGE: PER SECOND (Kilobytes)</th>
<th>TOTAL USAGE: PER MINUTE (Kilobytes)</th>
</tr>
</thead>
<tbody>
<tr>
<td>9600</td>
<td>0%</td>
<td>27712</td>
<td>6.77</td>
<td>405.94</td>
</tr>
<tr>
<td>9600</td>
<td>20%</td>
<td>30160</td>
<td>7.36</td>
<td>441.80</td>
</tr>
<tr>
<td>9600</td>
<td>33%</td>
<td>31544</td>
<td>7.70</td>
<td>462.07</td>
</tr>
<tr>
<td>9600</td>
<td>50%</td>
<td>33280</td>
<td>8.13</td>
<td>487.50</td>
</tr>
<tr>
<td>14400</td>
<td>0%</td>
<td>32512</td>
<td>7.94</td>
<td>476.25</td>
</tr>
<tr>
<td>14400</td>
<td>20%</td>
<td>35792</td>
<td>8.74</td>
<td>524.30</td>
</tr>
<tr>
<td>14400</td>
<td>33%</td>
<td>37528</td>
<td>9.16</td>
<td>549.73</td>
</tr>
<tr>
<td>14400</td>
<td>50%</td>
<td>40136</td>
<td>9.80</td>
<td>587.93</td>
</tr>
<tr>
<td>16800</td>
<td>0%</td>
<td>34912</td>
<td>8.52</td>
<td>511.41</td>
</tr>
<tr>
<td>16800</td>
<td>20%</td>
<td>38912</td>
<td>9.50</td>
<td>570.00</td>
</tr>
<tr>
<td>16800</td>
<td>33%</td>
<td>41296</td>
<td>10.08</td>
<td>604.92</td>
</tr>
<tr>
<td>16800</td>
<td>50%</td>
<td>43432</td>
<td>10.60</td>
<td>636.21</td>
</tr>
<tr>
<td>24000</td>
<td>0%</td>
<td>42112</td>
<td>10.28</td>
<td>616.88</td>
</tr>
<tr>
<td>24000</td>
<td>20%</td>
<td>47216</td>
<td>11.53</td>
<td>691.64</td>
</tr>
<tr>
<td>24000</td>
<td>33%</td>
<td>50320</td>
<td>12.29</td>
<td>737.11</td>
</tr>
<tr>
<td>24000</td>
<td>50%</td>
<td>54656</td>
<td>13.34</td>
<td>800.63</td>
</tr>
</tbody>
</table>

Please Note: Kilobyte usage has been summarized to two decimal points.
3.13. Data Usage for HSDPA and EV-DO High Bit-rate Connections

The following tables can be used to estimate the approximate data usage when using higher bandwidth connections over HSDPA and EV-DO 3G networks.

**Important Note:**
The 'MB Per Min (Both Ways)' calculation is assuming the same FEC amount has been set in both directions. Should FEC be less in one direction, this would lower the 'Both Ways' amount.

<table>
<thead>
<tr>
<th>Audio/Data Bitrate</th>
<th>Actual Bitrate</th>
<th>MB Per Min (One Way)</th>
<th>MB Per Min (Both Ways)</th>
</tr>
</thead>
<tbody>
<tr>
<td>128000</td>
<td>146000</td>
<td>1.1</td>
<td>2.2</td>
</tr>
<tr>
<td>112000</td>
<td>130100</td>
<td>0.98</td>
<td>1.96</td>
</tr>
<tr>
<td>96000</td>
<td>114100</td>
<td>0.86</td>
<td>1.72</td>
</tr>
<tr>
<td>64000</td>
<td>82100</td>
<td>0.62</td>
<td>1.24</td>
</tr>
<tr>
<td>48000</td>
<td>66100</td>
<td>0.5</td>
<td>1</td>
</tr>
</tbody>
</table>

3.13.1. UDP Bitrate with Forward Error Correction “Off”

<table>
<thead>
<tr>
<th>Audio/Data Bitrate</th>
<th>Actual Bitrate</th>
<th>MB Per Min (One Way)</th>
<th>MB Per Min (Both Ways)</th>
</tr>
</thead>
<tbody>
<tr>
<td>128000</td>
<td>146000</td>
<td>1.31</td>
<td>2.62</td>
</tr>
<tr>
<td>112000</td>
<td>130100</td>
<td>1.16</td>
<td>2.32</td>
</tr>
<tr>
<td>96000</td>
<td>114100</td>
<td>1.04</td>
<td>2.08</td>
</tr>
<tr>
<td>64000</td>
<td>82100</td>
<td>0.72</td>
<td>1.44</td>
</tr>
<tr>
<td>48000</td>
<td>66100</td>
<td>0.57</td>
<td>1.14</td>
</tr>
</tbody>
</table>

3.13.2. UDP Bitrate with Forward Error Correction at 20%

<table>
<thead>
<tr>
<th>Audio/Data Bitrate</th>
<th>Actual Bitrate</th>
<th>MB Per Min (One Way)</th>
<th>MB Per Min (Both Ways)</th>
</tr>
</thead>
<tbody>
<tr>
<td>128000</td>
<td>146000</td>
<td>1.27</td>
<td>2.54</td>
</tr>
<tr>
<td>112000</td>
<td>130100</td>
<td>1.11</td>
<td>2.22</td>
</tr>
<tr>
<td>96000</td>
<td>114100</td>
<td>0.78</td>
<td>1.56</td>
</tr>
<tr>
<td>48000</td>
<td>66100</td>
<td>0.62</td>
<td>1.24</td>
</tr>
</tbody>
</table>

3.13.3. UDP Bitrate with Forward Error Correction at 33%
3.13.4. UDP Bitrate with Forward Error Correction at 50%

<table>
<thead>
<tr>
<th>Audio/Data Bitrate</th>
<th>Actual Bitrate</th>
<th>Megabyte/Min</th>
<th>MB Per Min (Both Ways)</th>
</tr>
</thead>
<tbody>
<tr>
<td>128000</td>
<td>146000</td>
<td>1.59</td>
<td>3.18</td>
</tr>
<tr>
<td>112000</td>
<td>130100</td>
<td>1.41</td>
<td>2.82</td>
</tr>
<tr>
<td>96000</td>
<td>114100</td>
<td>1.23</td>
<td>2.46</td>
</tr>
<tr>
<td>64000</td>
<td>82100</td>
<td>0.87</td>
<td>1.74</td>
</tr>
<tr>
<td>48000</td>
<td>66100</td>
<td>0.69</td>
<td>1.38</td>
</tr>
</tbody>
</table>


Another way to ensure data costs are kept to a minimum is to program the 3GIP “Idle Timeout” minutes in a codec. The timeout setting represents the duration (in minutes) after a 3G/IP call terminates, that the 3G connection automatically disconnects. The default setting is 5 minutes and a ‘0’ (zero) turns this function off completely. To reprogram the settings select [Menu > Configuration > GSM/3G Setup > 3G Module > 3G/UMTS IP > Wireless Network > Sig Str Enable > Reset Wait Secs > 3G/UMTS IP Network > Idle Timeout (mins)].

One minute before disconnection the “IDLE Timeout” will begin to display a countdown timer, alternating with the “Connected” display and the “Goto IP” display. I.e. [3GIP1> IDLE 59 S=n] & [3GIP1> Cntd 52 S=n] & [3GIP1>Goto IP S=n]

At the conclusion of the timeout “Disconnect 13 IDLE Timeout Expired” is displayed and the 3G module will reset momentarily displaying “S=?”.

3.15. A Final Note on 3G/IP

If you require more information on IP connections using 3G cell-phone networks, please check the latest information on the Tieline website at www.tieline.com.
Section 4. IP Connectivity Using SIP

The information contained in this section discusses connection of Tieline codecs and other devices over IP using SIP. It complements existing literature about IP connections and discusses how to connect codecs:

- Directly over peer-to-peer connections using SIP settings; and
- Using a SIP server to connect.

It is important to remember that it is still possible to connect over IP without using SIP connectivity.

4.1. SIP and EBU N/ACIP Compatibility

SIP is an accepted and widely used technology for VoIP communications and its application has extended into other communication devices, including broadcast codecs. Tieline is developing SIP technology in accordance with the recommendations of the N/ACIP working group of the EBU. This role of this group is to standardize technology development approaches.

Technology development is being driven by the different approaches required for broadcast and telephony applications. For example, VoIP and telephony applications typically use algorithms such as G.711 and G.722, whereas broadcast applications would prefer to use higher quality MPEG algorithms.

4.2. Advantages of SIP

There are several benefits in using SIP to connect devices. It is possible to dial from both the remote or local codecs when using SIP and it also makes it easier to:

- Locate and connect to devices other than Tieline codecs;
- Locate codecs if they are changing locations and IP addresses regularly, or if they are not using static IP addresses; and
- Connect and select the best compatible algorithm.

SIP is easy to configure. Simply register your codec, enter the IP address of your SIP server along with your user name and password, and you are free to connect anywhere at any time. It really is simple.

4.3. Disadvantages of SIP

Most SIP servers are for VoIP and telephony in general and support a limited range of algorithms. If you are using a standard Internet SIP server to connect two codecs the only supported algorithms will usually be G.711 or GSM algorithms. The quality of these algorithms is lower than those preferred for most codec connections. This may be ok if you wish to connect to a VoIP phone, but not if connecting to another codec.
Unless you configure your own SIP server to connect between codecs, **Tieline** recommends that you make peer-to-peer SIP calls between dissimilar devices, e.g. a **Tieline** codec and a non-**Tieline** codec. By choosing to do this you take advantage of the reliable interoperability of Session Description Protocol (used for negotiating SIP connections) and can connect using the most appropriate and highest quality algorithm that both devices support.

### 4.4. Peer-to-Peer SIP Connections

Codecs don’t need to be registered for peer-to-peer connections. 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 (i.e. algorithm and bit rate) whereas a peer-to-peer SIP connection uses Session Description Protocol.

#### 4.4.1. Programming Peer-to-peer SIP Connections (Not recommended for VoIP Phones and Softphones)

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. Peer-to-peer connections require the following:

- The IP address of the remote unit being dialed;
- Programming of the public IP address of the router that a codec is connecting through via [Menu] > [Configuration] > [SIP Settings] > [Public IP] (if the codec has a private IP address when connected to a LAN);
- Both codecs to be in SIP mode by selecting [Menu] > [Configuration] > [IP1 Setup] > [Session Type] > [SIP];
- Normal configuration of both codecs using the IP Configuration Wizard;
- A call should originate from a field unit codec to the studio if it has a private IP address.

When codecs are enabled for SIP the codec connection screen will appear as displayed in the following image.

![Figure 30: SIP Codec Connection Screen](image)

To make a call enter an IP address for the destination codec and press ENTER/DIAL to connect to the remote codec.
4.5. What is SIP?
SIP is an acronym for Session Initiation Protocol. SIP is an OSI model layer 5 signaling protocol used to connect, monitor and disconnect connections over the Internet such as telephone calls, conferencing and multimedia distribution. It is capable of supporting audio, video and instant messaging.

SIP is a unique protocol specified by the IETF that provides multi-user/device sessions and connections without regard for the particular device or the media content that is delivered. It is independent and uses its own servers to provide connectivity between devices.

SIP works with a myriad of other protocols to establish connections with other devices over the Internet. It is used to find call participants and devices – even when they move from place-to-place. SIP is the session control mechanism used by 3G cellular networks and companies such as Microsoft® have chosen SIP for deploying their real-time software communications strategies.

4.6. How does it work?
SIP is unique in that it allows dissimilar devices to communicate with each other. In a way it is an extension of the ‘open standards’ that exist for Internet connectivity. It leverages on the use of Web architectures such as DNS. SIP addresses are similar in appearance to email addresses.

There are many ways to connect using SIP. It depends on the devices being connected and how they access the Internet. 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. The following diagram shows some examples of how Tieline codecs and other devices can connect using SIP over a LAN. In this example there are three codecs and a VoIP phone connected to a LAN-configured SIP server. Each device has been registered with this server.

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13 Microsoft is a registered trademark of Microsoft Corporation in the United States and/or other countries.
In the next example, a multitude of SIP-compatible devices are displayed. All of these devices can connect to Tieline codecs via the Internet and SIP servers. Several examples of remote codec configurations for Tieline codecs are displayed.
Figure 32: SIP-Compliant Devices Connected to the Internet
There are two very distinct parts to a call when dialing over IP. The initial stage is the call setup stage and this is what SIP is used for. The second stage is when data transference occurs and this is left to the other protocols used by a device (i.e. using UDP to send audio data). SIP can also be used for other elements of a call but it is important to remember that SIP only defines the way in which a communication session between devices should be managed. It does not define the type of communication session that is established.

4.6.1. Session Description Protocol and SIP

Session Description Protocol (SDP) is used to connect devices and describe end device characteristics when using SIP. This is an established format for describing streaming media initialization parameters. When connecting two codecs, SDP performs similar tasks to Tieline's proprietary session data, which is used to configure all non-IP codec connections and some non-SIP IP connections. When configuring a Tieline codec to operate over SIP via the Session Type setting in a codec, a codec is actually being programmed to use SDP rather than Tieline session data.

SDP works with a number of other protocols, such as UDP in Tieline codecs, and it provides the following functions when connecting two codecs over SIP:

- Establishes a codec’s location;
- Determines the availability of a codec;
- Negotiates the features to be used during a call, i.e. the algorithm and bit rate;
- Provides call management of participants; and
- Adjusts session management features while a call is in progress (i.e. termination and transfer of calls etc).

4.7. Getting Started: Registering a Device for SIP

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

4.7.1. What happens when I dial over SIP?

Connecting devices over SIP is very simple. When two codecs are SIP registered they find each other by connecting to SIP servers and exchanging connection information. In practice, the following diagram displays a simplified version of what happens when connecting SIP-registered codecs.
4.7.2. **SIP Server Connections**

To dial a codec via a SIP server a codec user requires:

- Both devices to be registered with the same SIP Server;
- Both codecs to be set to operate in SIP mode;
- The IP address of the SIP server;
- The SIP URI of the codec (the SIP Address);
- Normal configuration of both codecs using the IP Configuration Wizard;
- An IT administrator to open UDP port 5060 to enable SIP traffic, as well as UDP audio port 5004, both of which are programmed into the codec by default automatically.

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

- The SIP server IP address;
- A SIP number;
- A username (often the same as a SIP number);
As long as your codec and the device you are dialing to are registered to a SIP server you can dial that device by simply dialing its SIP number.

The benefit of using a SIP server to connect is that a device does not require a static or public IP address to be dialed and both ends can dial each other. Any device can be ‘discovered’ via its SIP server registration. This is particularly useful if a codec is being used in multiple locations with a variety of IP addresses that are DHCP assigned.

Important Note:
Some 3G service providers may block SIP traffic over UDP port 5060. It is possible to reprogram this but we recommend that you contact our support desk at support@tieline.com before attempting this.

4.8. Quick Start Connection of Codecs using SIP Servers

_Tieline_ recommends using your own SIP server if connecting using this method. That way you can be sure that you support high quality algorithms, unlike most SIP servers configured for Internet telephony.

Following is an example of how the process of SIP configuration operates to connect two new _Tieline_ codecs. In the example shown a codec with the SIP address of sip:1234@tieline.com in the studio is dialing a codec in the field over the Internet with the address sip:5678@tieline.com. (Please note: Codec SIP addresses and the IP address of your SIP registrar server are provided by whoever is managing the SIP server you are planning to connect with.)

1. Connect both codecs to the Internet and on each codec select [Menu] > [Configuration] > [IP1 Setup] > [Session Type] > [SIP]. Press CLEAR several times to return to the main connection screen.
2. Next select [Menu] > [Configuration] > [SIP Settings] > [Registration Mode] > [Enable].
3. In each codec enter the IP address of the SIP registrar server being used and select [OK].
4. Enter a unique SIP User Name (case sensitive) into each codec (this is often a number) and select [OK].
5. Enter a SIP Password (case sensitive) into each codec and select [OK].

If the codec has been registered correctly the codec connection screen displays “Rgd.” as shown in the following image.
6. Enter the SIP Address of the studio codec into the field codec as shown in the following image.

7. Press ENTER/DIAL to connect the studio and field codecs.
8. Once connected the connection screen will appear with the connection bit rate and link quality percentage of the local codec.

Very Important Note:
If your codecs don’t have public IP addresses and are using NAT, or are behind a firewall, we recommend that the public IP address of the router you are connecting through is programmed into each codec via [Menu] > [Configuration] > [SIP Settings] > [Public IP]. This ensures that SIP traffic will pass through NAT and firewall configurations.

4.9. Limitations of SIP
SIP doesn’t recognize the Tieline Music or Voice G3 algorithms, so if you try to connect using these algorithms, the server you connect through will most probably default to either the G.711 or G.722 algorithm if it is configured for VoIP telephony.

SIP is compatible with MP2 algorithms - so for high quality audio these algorithms are the preferred choice in Tieline codecs.

4.9.1. Renegotiation Limitations
Tieline codecs are currently unable to renegotiate connection bit rates over SIP connections. The connection bit rate should be programmed on both codecs before dialing.

4.9.2. SIP Server Registration
Both devices that are connecting need to be registered to the same SIP Server for a SIP call to be successful. This is because a codec will always dial the server that it has been registered to.

If the other device you are calling is not registered to the same SIP server then it will not find the address of it and the call will fail. In this situation it may be better to make a peer-to-peer IP call using the public IP address of one of the devices.
4.9.3. Other Settings for SIP Connections

4.9.3.1. NAT Transversal Mode

There are three settings available in the SIP menus – Auto (default), On or Off. Generally this should be set to Auto. If set to Auto or On then NAT (Network Address Translation) will be performed if required. This is necessary if the remote device has a private IP address or is behind a firewall.

In NAT traversal mode a codec ignores the contact information from a remote codec (inside a LAN) and uses the address from which the data packets are being received.

4.9.3.2. Public IP (Address)

If you are making a peer-to-peer SIP call, or if a codec has a private IP address when connected to a LAN using NAT or a firewall, Tieline recommends that you program the public IP address of the router that a codec is connecting through via [Menu] -> [Configuration] -> [SIP Settings] > [Public IP].

4.9.3.3. FEC

When connecting two Tieline codecs using SIP do not use FEC. FEC can only be used with Music, MusicPlus and Voice G3 algorithms and SIP does not recognize these algorithms.
4.10. Connecting to Non-Tieline Devices using SIP

Following is some useful information in relation to connecting to other devices using SIP.

4.10.1. Connecting a Tieline Codec to a Different Codec using SIP

This is similar to connecting to a Tieline codec. Suggested settings when configuring a codec using the IP Configuration Wizard include:

- Audio Protocol: Select UDP/IP in the codec IP Wizard;
- Algorithm setting: Non-Tieline codecs are unable to connect using proprietary algorithms such as Music and Voice G3. Select Raw, G.711, G.722 or MP2.
- Audio Bitrate: Select 64kbps (default) for mono and 128kbps for stereo connections.
- FEC: Leave this setting disabled.
- Leave auto jitter buffer programmed as the default setting.
- IP Setup: DHCP (default).
- Auto Reconnect: Disabled (default).
- Both ends should use the default SIP and audio port numbers unless there is port translation across the network.

4.10.2. Connecting a Tieline Codec to a Softphone or IP Phone using SIP

Tieline codecs can connect to SIP-compliant Softphones and VoIP (Voice Over IP) phones and send audio if required. Many of these phones only provide G.711 algorithm capabilities so the quality of these connections is likely to be inferior to connections between two Tieline codecs.

**Warning:**

There are literally thousands of different VoIP phones or softphones and peer-to-peer connections may be unsuccessful. For maximum reliability in connecting to these devices Tieline recommends only connecting to these devices using SIP.

4.10.3. Connecting a Tieline Codec to a POTS/PSTN Line using SIP

Once a codec is registered to a VoIP provider’s SIP Server a codec should be able to make calls to POTS/PSTN lines using their POTS/PSTN copper wire interface.
4.11. Other SIP Programming

Other elements of SIP can be programmed via codec menus if required to troubleshoot SIP connectivity issues. Following is a diagram of the SIP Setting menus as displayed in Tieline codecs and an explanation of other elements that may need to be programmed for SIP.

![SIP Configuration Menus Diagram]

Figure 35: SIP Configuration Menus
4.12. Advanced Programming of SIP Connections using ToolBox

The SIP tab in ToolBox software provides the ability to program SIP registration into a codec.

Registration of a codec can be performed by using this tab instead of a codec. Check the Register check-box and enter the SIP Server IP address, a User Name and Password and the Registration Refresh interval in seconds. It is also possible to program the public IP address of a router you are connecting through if the codec has a private IP address on a LAN.

Once these details have been entered into ToolBox and saved as a configuration file, the file can be loaded onto a codec and registration can be performed by pressing [Menu] > [Configuration] > [IP1 Setup] > [Session Type] > [SIP]. Enter the SIP address of the device you wish to call and dial to connect.
4.13. **Advanced Understanding of SIP: Terminology and Components**

To understand SIP and the way that it works it may be useful to understand in more detail the hardware components that could be involved in SIP connections. There can be up to four components involved in creating codec SIP connections. These include:

- User Agents (UA): Codecs and/or other end-user devices;
- SIP Registrar Servers;
- SIP Proxy Servers; and
- SIP Redirect Servers.

There are thousands of devices that use SIP and they all may have slightly different implementations of SIP technology. In reality, a registrar server and a proxy server may reside alongside each other and not all components may be used to set up a SIP call. **Tieline** codecs operate with most devices without requiring complicated configuration. Simply register your codec, enter the IP address of your SIP server along with your user name and password and you are free to connect anywhere at any time. It really is simple.

**4.13.1. User Agents (UA)**

A User Agent is another name for any device that acts as a gateway to convert audio into data packets for routing through networks. A **Tieline** codec is a UA, as is an IP telephone and a soft phone. **Tieline** codecs can use SIP to connect to these devices.

**4.13.2. SIP Registrar Servers**

Registrar servers accept register information such as the SIP address and IP address of UAs for the domain that they are currently servicing. This is regularly updated automatically and all UAs must be registered before they can be accessed via a SIP server. Operationally, registrar servers retain and provide information about the location of local and remote **Tieline** codecs for making calls. These servers are often located with a proxy or redirect server.

**4.13.3. SIP Proxy Servers**

A proxy server is normally the conduit to the ‘outside world’ for a LAN. A SIP proxy server is a server that allows a **Tieline** codec or a UA to make an indirect connection to another server. For example, a **Tieline** codec would request a connection to a server by first connecting to a proxy server and then the proxy server would forward the requests on behalf of the codec.

SIP proxy servers route SIP requests to the servers used by other devices. A proxy server will attempt to route a message from itself to a UA server or another proxy server closer to the destination. Sometimes several proxy servers over several ‘hops’ may be used to deliver SIP requests and responses.
4.13.4. **SIP Redirect Servers**

A redirect server responds to a request for the current location of a user by replying with the current information in a registrar’s location database.

4.13.5. **Registration Refresh (Secs)**

Every SIP registrar server requires that a device is reregistered at regular intervals to ensure its current location is accurate. By default this setting in each codec is one hour but it can be adjusted if required by selecting [Menu] > [Configuration] > [SIP Settings] > [Reg Refresh (Secs)].
Appendix 1.  Tech Note on Line Quality and Firewalls

Release Date
8th September 2006

Purpose
To Interpret Link Quality ($LQ$) and LQ Relationship to Firewall Blocking

Background
Tieline codecs connect over IP using TCP session connections which in turn establish the audio data path over UDP Ports. TCP Ports are generally accessible over an IP connection however UDP Ports are often blocked by router firewalls. Tieline’s inbuilt Link Quality diagnostics assists in determining problems affecting link and audio quality and possible blocks in firewalls, impeding throughput of audio. Tieline recommends using UDP for remote broadcasts as it offers significantly higher link stability and lower audio delay than TCP.

Firmware Requirements
1.2.00 onwards

IP Link Quality
The IP link quality value consists of two digits. The value actually encodes two pieces of information. Refer to Table 1 below.
- The Whole Number provides a general indication of the link quality over the last 8 seconds
- The Last Digit can be interpreted to give an indication of the worst reception event that has occurred in the last second.
Figure 1 shows a connection over IP at a rate of 128 kbps. ‘L89’ and ‘R99’ on the local codec, refer to Local and Remote incoming link qualities respectively. I.e. ‘L89R99’. And vice versa at the remote codec. I.e. ‘L99R89’.

For a full description of the numbering schema refer to Table 1 above.

**Firewall Blocking**

Figure 2 demonstrates a connection over IP. ‘L01’ and ‘R99’ on the local codec, refer to Local and Remote incoming (receive) link qualities respectively. I.e. ‘L01R99’. And the reverse ‘L99R01’ on the remote codec. ‘01’ specifically refers to blocked communication by a firewall.
To Determine the Direction

If the Local Codec, local link quality displays 01, then the Local Codec receive path is blocked by a firewall at either point (A) or (B) in Figure 3. (The data being transmitted from the Remote Codec to the Local Codec). This indicates that one or more of the ports in Table 2 is being blocked.

Default Tieline Codec Port Settings

<table>
<thead>
<tr>
<th>Application</th>
<th>TCP Session Port</th>
<th>UDP Audio Port</th>
</tr>
</thead>
<tbody>
<tr>
<td>IP 1</td>
<td>9002</td>
<td>9000</td>
</tr>
<tr>
<td>IP 2</td>
<td>9012</td>
<td>9010</td>
</tr>
<tr>
<td>Tieline Toolbox and TLG3GUI</td>
<td>-</td>
<td>5550</td>
</tr>
</tbody>
</table>

Table 2

If the Local Codec, remote link quality displays 01 then the Remote Codec receive path is blocked by a firewall at either point (C) or (D) in Figure 2. (The data being transmitted from the Local Codec to the Remote Codec). Indicating that one or more of the ports in Table 2 is being blocked.
How To Determine Which End is Blocking Data Flow

The firewall at either end of the link can still be the cause of the impedance of data as firewall settings can apply to both incoming and outgoing data paths as illustrated in Figure 3.

To determine which end is blocked, connect to a remote codec that is known to have ports open on the remote firewall. Any issues with blocked data can then be limited to the local firewall. If this is not possible, contact Tieline Technology for assistance by emailing support@tieline.com

Alternatively, have your IT Administrator, enter the codecs IP address into the routers ‘Demilitarized Zone’ (DMZ), which should allow freedom for the codec to communicate over the internet.

Try this Simple Test Procedure

1) Call another Codec that has no firewall issues so that you can determine which end has a firewall blockage. If the firewall blockage problem is at your end, go to step 2.

2) Open ports in the firewall by asking the Network Administrator or read your Modem/Router manual and open it yourself.

3) If opening ports in the router is proving too complicated or difficult for you to perform then check to see if it has a DMZ. This is a simple, single mechanism to open ALL ports for a particular IP address. Add the Codec IP Address to the DMZ. If this step fails go to step 4.

4) Change the Audio protocol in the menu from UDP to TCP.

   Menu ⇒ Configuration ⇒ IP1 Setup ⇒ Audio Protocol ⇒ UDP/TCP

If this works then get some assistance to implement step 2 so that you can change back to UDP as soon a possible.
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or

b) Supply replacement for any defective parts in the equipment for a period of two years from the date of original purchase. Replacement parts shall be supplied without charge, except labor and transportation.

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Appendix 4.1. FCC Part 15

Compliance – TIELINE TECHNOLOGY, 25 Irvine Drive, Malaga. Western Australia 6090.

This equipment has been tested and found to comply with the limits for a class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area may cause harmful interference, in which case the user will be required to correct the interference at his/her own expense. Changes or modifications not expressly approved by Tieline Pty Ltd could void the user’s authority to operate the equipment.

If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try and correct the problem by one or more of the following measures:

1. Increase the separation between the equipment and the receiver;
2. Connect the equipment into an outlet on a circuit different to that used by the receiver;
3. Consult the dealer or an experienced radio/TV technician.

Appendix 4.2. FCC Part 68

FCC Registration Number: 6NAAUS-34641-MD-E
Ringer Equivalence Number (REN): 0.5B
A label containing, among other information, the FCC registration and Ringer Equivalence Number (REN) for this equipment is prominently posted on the bottom, near the rear of the equipment. If requested, this information must be provided to your telephone company. USOC Jacks: This device uses RJ11C terminal jacks. The REN is used to determine the quantity of devices, which may be connected to the telephone line. Excessive RENs on the telephone line may result in the devices not ringing in response to an incoming call. In most, but not all areas, the sum of REN’s should not exceed five (5). To be certain of the number of devices that may be connected to the line, as determined by the total RENs, contact the telephone company to obtain the maximum RENs for the calling area.

If this equipment causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of the service may be required. If advance notice is not practical, the company will notify the customer as soon as possible. Also you will be advised of your right to file a complaint with the FCC if you believe it is necessary.
Appendix 4: Compliances

The Telephone Company may make changes in its facilities, equipment, operations or procedures that could affect the operation of the equipment. If this happens the Telephone Company will provide advance notice for you to make the necessary modifications in order to maintain uninterrupted service.

If you experience problems with this equipment, contact TIELINE Pty Ltd, 25 Irvine Drive, Malaga, Western Australia, 6090. Ph +61 8 9249 6688 Fax +61 8 9249 6858 email info@tieline.com (web page www.tieline.com) for repair and warranty information.

If the problem is causing harm to the telephone network, the Telephone Company may request you remove the equipment from the network until the problem is resolved.

No user serviceable parts are contained in this product. If damage or malfunction occurs, contact TIELINE Pty Ltd for instructions on repair or return. This equipment cannot be used on a telephone company provided coin service. Connection to Party Line service is subject to state tariffs.

Appendix 4.3. IC

NOTICE: The Industry of Canada label identifies certified equipment. This certification means that the equipment meets certain telephone network protective operational and safety requirements. The Department does not guarantee the equipment will operate to the user’s satisfaction.

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local Telecommunications Company. The equipment must also be installed using an acceptable method of connection. In some cases the companies inside wiring associated with a single line individual service may be extended by means of a certified connector assembly (telephone extension cord). The customer should be aware that compliance with the above conditions might not prevent degradation of the service in some situations.

Repairs to certified equipment should be made by an authorized Canadian maintenance facility designated by TIELINE Pty Ltd. Any repairs or alterations made by the user to this equipment, or equipment malfunctions may give the telecommunications company cause to request the user disconnect the equipment. Users should ensure for their own protection that the electrical ground connections of the power utility, telephone lines and internal metallic water pipe system, if present, are connected together. This precaution may be particularly important in rural areas.

Appendix 4.4. CE & CE Tick

This product has been extensively tested to ensure compliance with Australian “C Tick” and European CE requirements.

Because high frequency circuits are used, it is possible that induced radiation may enter the signal path. Care should be taken to avoid high levels of radio frequency exposure to the unit as this may result in some distortion or failure of the audio signal.
Appendix 5. Credit Notices

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2. Windows XP is a trademark of Microsoft Corporation in the United States and/or other countries.

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7. Comrex, Matrix and BlueBox are registered trademarks of Comrex Corporation.

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