Tieline TECHNOLOGY

COMMANDER G3

TLF300 Integrated Field Mixer-CODEC

Reference Manual

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Section 1. Safety Notices and Warnings

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.

(Related Topic: Tips for Successful Operation)

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 2. Manual Conventions

The conventions we have used in this manual are as follows:

2.1. Controls

Buttons, switches, and rotary controls are in ARIAL CAPITALS, the same font and style as the labeling on the codec. E.g. SEND/RETURN refers to the SEND/RETURN digipot.

2.2. Connector Panel

Labeling is done in Arial, reflecting the text on the codec. E.g. HEADPHONE 2 is the socket for HEADPHONE 2.

2.3. Menu Text

Menu Text is done in boxed and is the exact text in the LCD window .

2.4. Menu Navigation

When describing how to navigate through detailed codec menus, the following convention will be used. Bold Souvenir ITCTT for all characters; square brackets to surround each individual menu item; inward facing arrows (inside the square brackets, i.e. →**Voice G3**←) to indicate the menu setting; the **>** character is used to indicate movement to the next menu item.

[Pots Wizard →Setup GSM Landline←] > [Algorithm →Voice G3←] > [GSM Landline Rate →9600←] > [Pre-buffer Secs →0←] > [Auto Reconnect →Disable←].

2.5. ToolBox Software

Any reference directly attributable to the Tool**Box** software will be in Souvenir ITCTT Italic font. E.g. *Modem Max Bitrate.* If it appears in blue font color in Tool**Box** software then it will appear in the same font but in blue. E.G. *Automatic Redial* (in ***Connection Setup*** in the ***Profile Editor***)

Any Tool**Box** software section tab or Main Menu title is in Souvenir ITCTT Bold Italic font. E.g. ***Profile Editor***

2.6. Hyperlinks

If you are reading this document on a PC, within it there are many hyperlinks to websites or to other related bookmarked elements within the manual. These are characterized by being underlined as in the following example:

Connection Setup

If you are reading this document as a PDF simply click on the hyperlink to go to the destination. If you are not viewing it as a PDF, to activate the hyperlink place your mouse cursor on the hyperlink, hold down the Ctrl key on your keyboard and click the left mouse button. This will take you to the hyperlink destination.

Section 3. Welcome to our Revolution

This may seem an outrageous statement, but **Tieline** has developed and currently manufactures the world's finest POTS, ISDN, GSM, X.21, 3G and IP codecs. You will find that **Tieline** codecs have a whole range of invaluable features for creating low cost, studio quality programs from remote locations. The **Tieline** codec family includes:

- ***i*-Mix *G3***, the latest generation of 5 channel POTS, bonded POTS, ISDN, GSM, IP and satellite ISDN mixer-codec, is justifiably known as the 'sportscasters dream machine'.

- ***COMMANDER G3***, with two module slots, allows flexibility in the selection of field POTS, ISDN, GSM and IP codec with full broadcast functionality.

- ***COMMANDER G3*** 1RU and 2RU rack unit codecs with similar features to the field unit. A PC GUI (Graphical User Interface) can be connected to the 1RU codec, providing flexibility in controlling the codec and saving rack space in the studio.

The **Tieline *COMMANDER G3*** is an integrated 2 channel mixer and award winning studio quality POTS, ISDN, GSM, 3G/IP, IP and X.21 codec. The superb **Tieline** DSP^2 algorithm delivers crystal clear, studio quality bi-directional 15 kHz audio with an insignificant 100ms delay. This single unit weights only 5lb (2kg) and is easily placed in a briefcase and transported as carry-on baggage, allowing timely and convenient worldwide deployment of presenters without technical staff. With headsets you have a complete remote broadcast facility.

The ***COMMANDER G3*** concept is about combining suitable communications and programming modules together to create the right portable studio for your remote broadcast. **Tieline**'s most recent technological advances combine IP, 3G/IP cell-phone and X.21 functionality into the ***COMMANDER G3***.

The **Tieline *COMMANDER G3*** mixer accommodates up to two announcers simultaneously. Each announcer has their own input, headphone output and talkback level controls.

The superb **Tieline** *Music* and *Music Plus* algorithms deliver crystal clear, studio quality bi-directional FM quality audio with low delay. This single unit weights only 5lb (2kg) and is easily placed in a briefcase and transported as carry-on baggage, allowing timely and convenient worldwide deployment of presenters without technical staff. With headsets you have a complete remote broadcast facility.

There are a multitude of connection possibilities to suit every broadcast situation. You can use two POTS connections for stereo or dual-mono POTS. Dual-mono POTS allows programming to be sent to more than one destination from the same POTS codec.

One of the unique features of the ***COMMANDER G3*** is the ability to connect via a bonded POTS connection. For example, if you have two POTS connections that are

each achieving bitrates of 12,000 bps, you can bond these connections to create a single 24,000 bps connection.

For the first time broadcasters are able to phase lock left and right audio channels over two ordinary telephone lines to deliver stable and reliable 15kHz FM quality stereo programming – all for the cost of a couple of regular telephone calls. In some parts of the world, local telephone calls are free which could allow some FM broadcasters a studio to transmitter link with no transmission costs at all!

Combine an ISDN module with a 15 kHz POTS connection to provide IFB over POTS and program over ISDN. Using IP and 3GIP you can connect codecs over wireless 3G networks or connect over a private Local Area Network (LAN), or over different public networks such as the Internet. **Tieline** codecs can supply high bandwidth audio and communications data over national and international networks. Integration of X.21 capabilities into **Tieline** codecs adds even more flexibility in connecting across leased line networks.

The **Tieline** ***COMMANDER G3*** is designed to be operated in conjunction with ToolBox PC software. Most functions can be programmed via the codec itself, but to get the most out of your codec you can program it using ToolBox software.

One of the features of the **Tieline** ***COMMANDER G3*** platform is the Configuration File System. It enables a user to configure the codec with ToolBox software before arriving at a remote broadcast site, minimizing the amount of adjustments a broadcaster has to make when arriving at a remote site.

Tieline's Connection Manager automates the process of making a connection with the ***COMMANDER G3*** from a remote site. All a user has to do is turn the codec on, wait for the menu on the LCD screen to light up and then select START. Even dialing manually with the ***COMMANDER G3*** is simple. Connect to a preferred dialing interface and dial the number of a remote codec. The remote **Tieline** codec automatically answers the call and establishes a secure link at the best quality the line will allow.

Tieline's unique remote control feature allows the setting and continuous supervision of all switch settings and audio levels remotely from a codec. This can be done either from a studio or a laptop PC connected to a codec – leaving an announcer free to concentrate on the content of a broadcast, not the technical parameters.

In summary, **Tieline** codecs provide opportunities previously impossible to engineer, or simply too expensive to contemplate. The **Tieline** ***COMMANDER G3*** is the perfect solution for remote broadcasters. In the audio broadcast revolution - **hearing is believing**... welcome to our revolution!

Section 4. Introduction to Tieline Codecs

If you are a new user or even if you are very familiar with a previous model of **Tieline** codec, we highly recommend that you at least familiarize yourself with a few of the sections within this manual. These sections include:

- Connecting Your Codec to a PC;
- Quick Start: Connecting Quickly Using Manual Default Profiles;
- Operation of your Codec; and
- Configuration File System.

The 'Connecting Your Codec to a PC' section of this manual will describe how to configure your codec and ToolBox software, so that they will communicate effectively with each other.

The section called 'Configuration File System' will give you a good understanding of how Configuration Files are used to store profiles and how to program them into a codec. If you need further information on these areas the 'Matrix Editor' and 'Profile Editor' sections will explain this in more detail.

The 'Operation of your Codec' section will assist the experienced user to use the codec with a minimum of fuss - without using ToolBox software. It includes an explanation of the basic operations required to connect and adjust audio input levels etc. You will of course need to use ToolBox software for many codec operations. This includes creating and amending matrices for profiles and saving configuration files.

The 'Quick Start: Connecting Quickly Using Manual Default Profiles' section of the manual gives a really quick explanation of how to connect your codecs by using manual default profiles that come with the codec.

Have fun with your new codec. It is at the leading edge of codec technology and will deliver superior performance for you and your broadcast partners.

Help us to help you: We value feedback from our customers and encourage you to help us make your job easier by emailing any suggestions on how we can improve this reference manual to support@tieline.com

4.1. Features in Release Version 1.4.xx

Following is an overview of the new features incorporated into **Tieline** Firmware Release 1.4.

- It is no longer necessary to select the *ManDflt Bonded Mono* profile for bonded connections - this profile has been removed. Simply select *ManDflt MonoPgm* and if a second connection is dialed it will automatically be bonded;
- 3G has been updated to provide 3G to 3G call functionality;
- **Tieline** broadcast codecs now support ISDN 3B and 4B bonding (***COMMANDER G3*** only) in order to create connections of up to 256 kbps in bandwidth;
- It is now possible to make voice calls using V.1.2 POTS plug-in modules;
- It is now possible to make voice calls over GSM plug-in modules or cell-phones;
- The ***Phonebook Editor*** now supports speed dialing numbers and associated profiles (both manual default and custom created profiles);
- ***Phonebook Editor*** capacity has increased from 50 numbers to 80;
- It is now possible to connect using the *MP2* algorithm and simultaneously connect via *RS232* (between **Tieline** broadcast codecs only) to send data or use ToolBox;
- AGC (Automatic Gain Control) is visible in codec menus and able to be switched on and off if required;
- Aux/Phone input level can be adjusted via the codec Audio menu by pressing Softkey 1;
- Updates to session data include the ability to turn it on and off in codecs and view if it is operating properly via codec menus;
- The phone input can be routed to the codec analog outputs via the Audio menu by pressing Softkey 1 (***i*-Mix** only);
- AES/EBU functionality is fully integrated (***COMMANDER G3*** rack unit only); and
- X.21/V.35 functionality is fully integrated.

4.2. Features in Release Version 1.6.xx

Numerous changes have occurred in release 1.6.xx. Most of these relate to IP and 3G IP connections. As a result, these sections have been extracted from all codec reference manuals and have been amalgamated into a manual titled the "IP & 3GIP Streaming Reference Manual".

This manual contains all the latest connection information relating to IP and 3GIP in general. It also contains information about SIP connectivity and updated quick start guides for studio and field unit codecs, as well as for wireless 3G networks.

Following is a summary of the new and updated features that are contained in the "IP & 3GIP Streaming Reference Manual".

- 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 either; 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.

4.2.1. Adjustments in this Reference Manual

The features outlined in this reference manual supersede the information contained in the previous manual which was titled "TLR300B_Commander_G3_Rack_Unit_PLUS_Main_Man_v.4.0."

Following is a summary of the amendments made to this reference manual:

- The "IP Streaming" and "3GIP" sections have been extracted and consolidated into the "IP & 3GIP Streaming Reference Manual".
- The "LAN Tab" section discussing LAN connection of codecs has been extracted and consolidated into the "IP & 3GIP Streaming Reference Manual".
- All ToolBox 3G programming has been consolidated into the "IP & 3GIP Streaming Reference Manual".
- The ISDN section has been updated with more detailed connection and troubleshooting information.
- The TLG3 GUI application for controlling rack unit codecs has the following new features:
  - The GUI is fully resizable on a screen.
  - Multiple windows/applications of the GUI can run simultaneously.
  - Windows can be tiled or cascaded.
  - A Toolbar provides the ability to restore minimised windows.

  Caveats:
  - Supports multiple TCP connections but with only 1 UDP, 1 Serial or 1 USB connection at the same time.

4.3. Features of the Commander G3 Field Unit Codec:

- 2 balanced MIC/LINE inputs on XLR and 1 AUX input on an RCA connector.
- 2 balanced LINE outputs on XLR and 1 AUX output on an RCA connector.
- Accommodates two announcers simultaneously, each with individual industry standard, ¼ inch (6.5mm) stereo RTS headphone connectors, and each with individual volume controls and programmable sources.
- Programmable buttons can be used for cue, intercom or talkback functions - allowing private communication between announcers and/or the studio.
- 2 Programmable PROGRAM OUTPUTS. These can be for PA feeds, recorders etc.
- 2 Programmable CONTROL ports for machine control along with 2 opto-isolated CONTROL inputs.
- 4 Programmable HOTKEYS for triggering User Functions.
- PPM style meters that are programmable.
- Internal 80 Number phone book.
- The ability to use POTS, Bonded POTS, ISDN, GSM, 3G, IP and X.21 connections.
- Purpose built award-winning modems for POTS codec operation.
- High quality bi-directional audio at very low bit rates e.g. 15 kHz @ 24000 bps on POTS lines, and an amazing 7 kHz at 9600 bps.
- Choice of *Music, MusicPlus*, *G.711*, *G.722, MP2 Mono, MP2 Stereo, MP2 J-Stereo*, *Other* and *Voice G3* algorithms.
- Seamless up and down re-negotiation.
- POTS line quality of forward and reverse link displayed on both codecs.
- Programmable automatic re-connection in the event of line dropouts.
- Configurable for PSTN lines, Leased (dry) lines and ISDN lines with the in-built ISDN module.
- GSM phone capabilities using either a GSM plug-in module with an antenna connector and/or from a phone via a data cable connected to the 9 pin RS 232 connector on the back of the codec.
- A 3G plug-in module for connecting to UMTS/HSDPA and EV-DO 3G networks.
- A USB plug-in module for USB modem and 3G cell-phone connections.
- An X.21 plug-in module with a male D-15 connector for X.21 connections.
- Satellite transmission capability which is similar to ISDN use.
- Virtually overload proof *Intelligent Gain Control (IGC)*.
- 4 programmable CONTROL FUNCTION buttons
- All settings and gains of a remote **Tieline** codec can be controlled from the studio using the LOCAL/REMOTE control feature.
- Comprehensive menu and Configuration File system simplifies user operation.
- 9 pin RS232 Interface Connector for PC connection.
- USB PC interface connector.
- LAN 10/100 interface with individual MAC address.
- Fast and easy configuration using the **Tieline** Tool**Box** software running on *Windows*® 98/2000/XP.
- Inbuilt 400 Hz Oscillator.

- **Tieline** codecs are fully compatible with the Musicam[1] Liberty and Voyager codecs.

To summarize, the **Tieline** ***COMMANDER G3*** delivers a studio quality audio mixer with *Intelligent Gain Control*, comprehensive monitoring and control, and the award winning **Tieline** Codec.

4.4. Data Options Available

Every connection includes a minimum of 50 bytes per second in a remote control channel, for **Tieline** ToolBox capabilities and which can be also used to control specific **Tieline** external devices, i.e. external relay boxes.

If your codec is captioning-enabled, please see the captioning operation manual for more information on data transfer and setting your codec up.

For specific GSM data information, please see the section in this reference manual titled GSM Transparent and Non-transparent Data Modes. For all other data information, please see the section of this reference manual titled Data Transfer & Using 3rd Party Devices.

Please note:

It is possible that when you use *G.722* and MPEG algorithms to connect to a non- **Tieline** codec, the 'framing' process (where incoming bit streams are identified and distinguished for individual decoding) can be unsuccessful when a call is initiated. This can be solved by turning off the 'session data' stream sent by the Tieline codec. To do this select **[Menu] > [Configuration] > [System Settings] > [Session Data] > [Disable].** Please note that this is a global setting and will also affect all POTS, and IP connections on the codec.

If you try to connect to a **Tieline** codec with session data disabled, you will have to ensure that the connection settings on both codecs are identical because dialing codec session data normally provides this information to the remote codec. To avoid this problem, simple re-enable session data **[Menu] > [Configuration] > [System Settings] > [Session Data] > [Enable].**

[1] Musicam USA Musicam USA Holmdel, NJ USA http://www.musicamusa.com info@musicamusa.com

4.5. Compatibility across the G3 Range of Codecs

Tieline ***G3*** codecs are all compatible with each other. For example, a ***COMMANDER G3*** field unit codec will connect successfully with another field unit codec, as well as a ***COMMANDER G3*** rack unit and the ***i*-Mix *G3***. The codecs all use the same algorithms and can be set to interact with each other. Please not that there are some differences between the previous version of **Tieline's** rack unit codec, the TLR300, and the new TLR300B1 and TLR300B2 rack unit codec.

Important Note:
It is important to note that the ***i*-Mix *G3*** has several different features in comparison with the ***COMMANDER G3*** Field Unit and Rack Unit codecs. For example, it has more inputs, an on-board POTS modem and programmable HOTKEYS.

If you are using your codecs in conjunction with ToolBox PC software, management of the codecs is simple. You can create profiles for all the codecs in the **Tieline** range, program them, and then integrate and connect the codecs at the touch of a button.

(Related Topics: ToolBox Software, Connecting Your Codec to a PC, Configuring ToolBox and Your Codec to Work Together.)

Backward Compatibility Tip:
If you want to connect your ***COMMANDER G3*** to a ***COMMANDER G1*** or to an ***i*-Mix *G1*** you will need to use the *Music* algorithm.

Section 5. Codec Controls and Connections

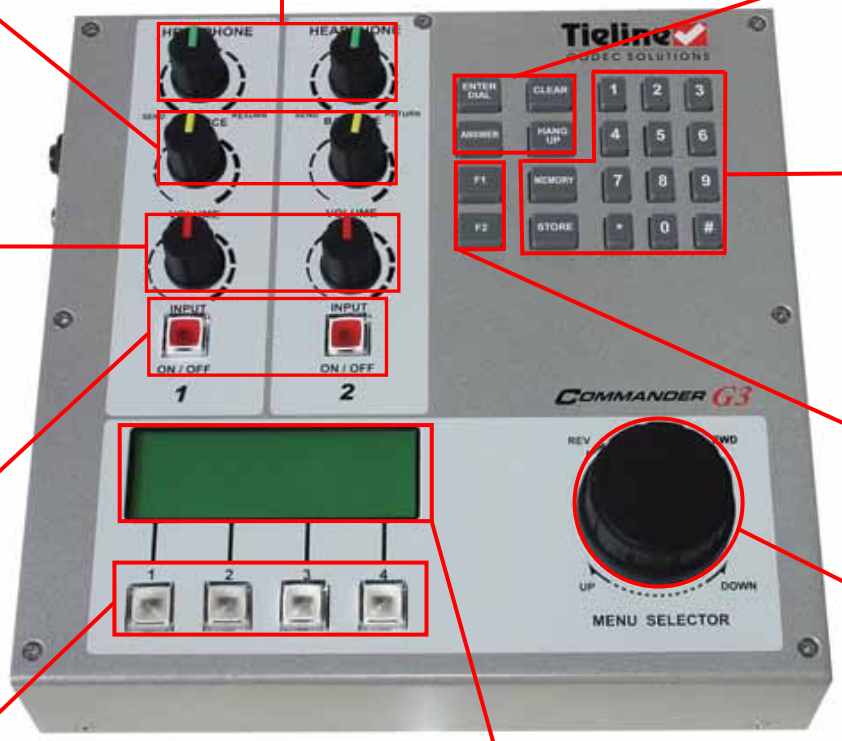
Figure 1: Front Panel of the Commander G3 Field Codec

VOLUME
Sets the volume in the headphones

BALANCE
Controls the balance between the SEND & RETURN audio to the headphones. This control does not affect the level of the transmitted or received program audio. It only controls the monitoring.

ENTER/DIAL
CLEAR
ANSWER
HANG UP
POTS/ISDN/X.21/IP/GSM phone function buttons.

INPUT
Sets the input gain for each channel. Menu items refer to the maximum gain available on each of the following settings:
+15 dBu - Line level input
+35 dBu – Unbalanced input
+55 dBu– Low gain mic input
+65 dBu - Medium gain mic input
+72 dBu - High gain mic input
(Note: The codec has 2 mic/line inputs that will handle inputs from -72 dBu to +21 dBu)



KEYPAD
MEMORY & STORE
For dialing numbers. Memory and store are for storing & accessing numbers in the 50 entry phone book.

FUNCTIONS BUTTONS
Used for initiating User Functions. These can all be programmed with ToolBox within the *Functions* menu in the *Profile Editor*.

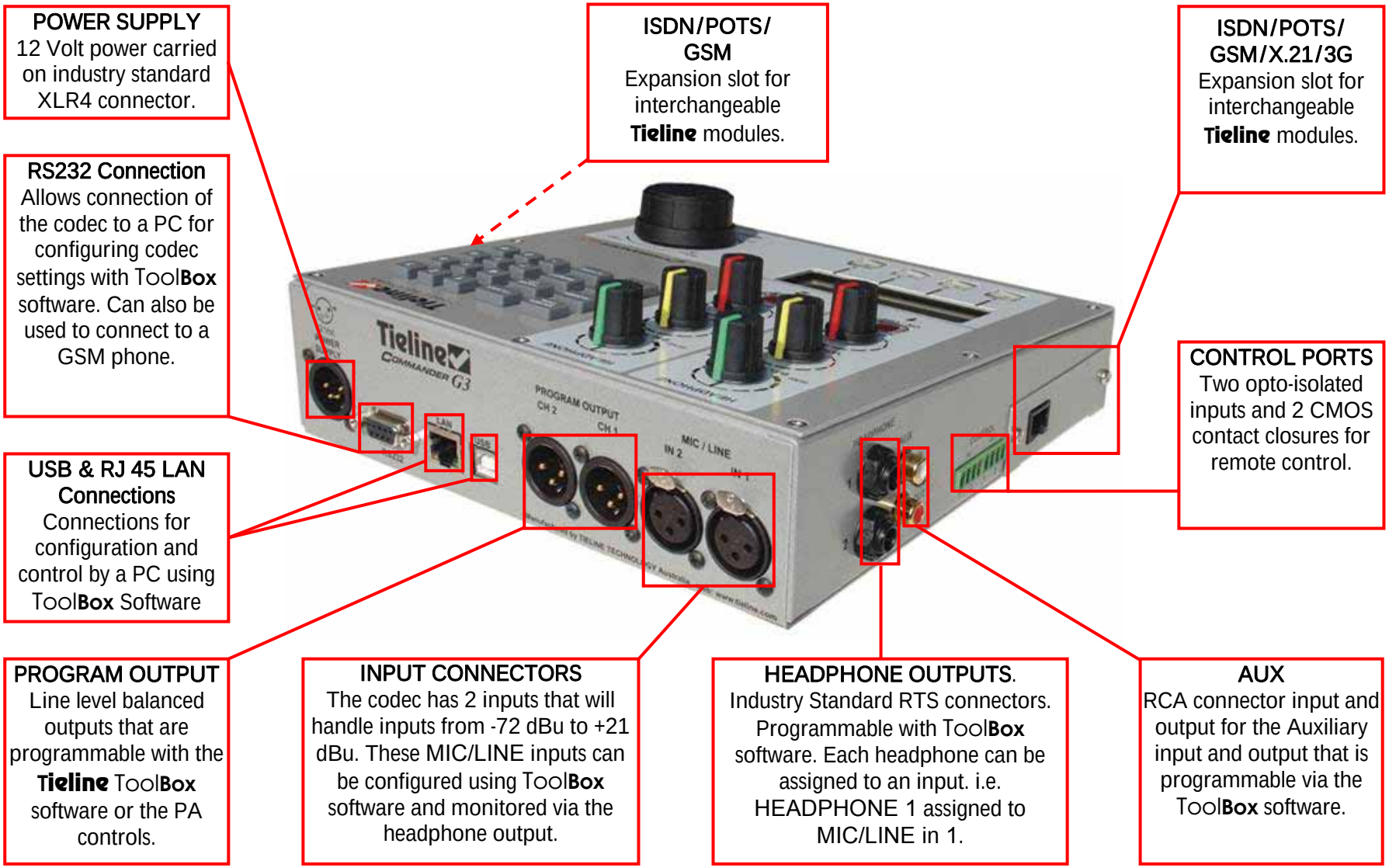
CHANNEL ON/OFF
Illuminated red when channel is active.

MENU SELECTOR (MS)
For navigation through the menu options. Pressing the MS down selects the bracketed option.

SOFTKEYS
Programmable buttons for one press initialization of User Functions and *Cue* functions.

LCD DISPLAY
4 rows of 20 characters display the menu options.

Figure 2: Rear and Side Panels of the Commander G3 Field Codec



Section 6. New Codec Menu Wizards

Tieline has developed an enviable reputation for creating quality products with unparalleled features and functionality. Now **Tieline** has developed new menu wizards to further simplify codec configuration – making life even easier for broadcasters.

All the previous functionality of **Tieline** codecs has been retained. All that has changed is that now you can select the connection on the codec that you wish to connect with, then click SOFTKEY 3, which has Wiz displayed above it, and the menu wizard for that connection will be displayed. Each type of connection, whether POTS, ISDN, GSM or IP, has its own wizard for easy connection configuration.

Figure 3: POTS Wizard Menu Screen

POTS connection surrounded by square brackets to access the POTS wizard.

The new menu wizard as displayed on **Tieline** codecs.

SOFTKEY 3 for opening the menu wizard.

In the previous example, the POTS connection is surrounded by the square brackets so the POTS wizard will open if SOFTKEY 3 is pressed. All the previous **Tieline** menu configuration options are still available by pressing SOFTKEY 4 and accessing **Tieline**'s traditional menu structure.

6.1. Easier Navigation

Tieline has also made navigation of codec menus more user-friendly. The square brackets, used to display where the MENU SELECTOR (MS) is positioned when scrolling, now flash to make it easier to see the current position. The current setting in a menu is still displayed by the use of arrows either side of the setting.

Arrow as displayed on the LCD screen of **Tieline** codecs.

If the flashing MENU SELECTOR (MS) brackets are navigated over the current selection (displaying arrows) then the brackets and arrows will flash intermittently between one and then the other.

6.2. Module Indicator Arrows

When a module is inserted into a **Tieline** codec an arrow on the LCD indicates where the module is situated in the codec.

6.2.1. Commander G3 Modules

The following image could be from a ***COMMANDER G3*** field or rack unit codec. ***COMMANDER G3*** field unit codecs can have modules inserted into the left and right sides of the codec. If the following codec screen image was from a field unit codec, it would indicate that an ISDN module is in the left side of the codec and a POTS module is in the right side of the codec.

With a ***COMMANDER G3*** rack unit codec the modules are inserted into the rear of the codec. Therefore, if the following image was from a rack unit codec, the arrows would display where the modules are positioned in the rear of the codec (when viewing from the front of the codec).

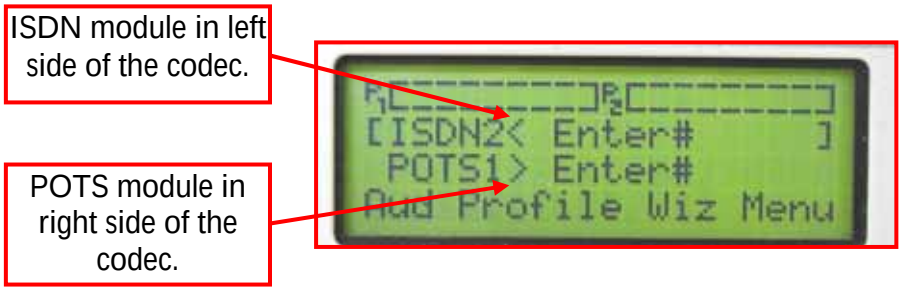


Figure 4: Commander G3 Menu Screen Module Indicator Arrows

6.2.2. I-Mix Modules

i*-Mix *G3 codecs are slightly different because they only have one module slot in the right side of the codec. In addition, there is an onboard POTS connection via an RJ11 connection on the rear of the codec. The arrows in the following image reflect how two POTS connections are displayed on an ***i*-Mix *G3*** LCD screen.

Figure 5: i-Mix G3 Menu Screen Module Indicator Arrows

6.3. Codec Module Options

This section of the reference manual will describe the module alternatives available for use in the ***COMMANDER G3*** codec. Hyperlinks to other sections of the manual will assist you to set up the various functions.

Both the ***COMMANDER G3*** field and rack units enable operators to literally build their own codec to suit the conditions of each broadcast. Each codec features two expansion slots, which accept your choice of POTS, ISDN, GSM, 3G (USB) or X.21 modules. This enables operators to simply select and plug in the desired telecommunications medium that is optimally suited to the broadcast location.

The ***COMMANDER G3*** can accept single or multiple module combinations such as:

- The on-board POTS modem used with one analog telephone line.
- The on-board POTS modem and one POTS module, each requiring one analog telephone line at the remote site.
- One ISDN module which requires one ISDN line with up to two B channels
- One ISDN line and one POTS module. ISDN can be used as the primary link and the POTS line can be used as a backup program feed.
- A GSM module can be used independently or in combination with an ISDN, POTS or other modules. GSM does not require any wiring as it uses a GSM cell-phone connection.
- Two GSM connections can be used, using one GSM module and a GSM cell-phone connected to the RS 232 port on the codec (no GSM modules can be used in rack unit codecs)
- Two IP/Internet connections using either the UDP or TCP protocols, via the Ethernet LAN connection port on **Tieline** codecs.
- A 3G/IP connection via a USB module connected to a 3G cell-phone or USB modem.
- A 3G connection via an HSDPA or EV-DO module.
- Two X.21 connections via X.21 modules with D15 connectors.

The next section of this manual explains how to get connected quickly using the manual default profiles supplied with your codec, depending on what modules you have installed into your codec.

Important note: It is only possible to use one 3G or USB module in a codec at a time. If you insert two modules into a codec then only one will be able to connect.

Section 7. Quick Start: Connecting Quickly Using Manual Default Profiles

Getting connected quickly with **Tieline** codecs is a three step process that has been simplified with the new codec menu structure. These three steps can be broken down into:

1. Set codec audio configurations settings;
2. Select the profile you wish to connect with; and
3. Use the **Tieline** codec wizard to change any connection settings.

This process is the same for all profiles and applies to both manual default profiles and custom profiles. Descriptions of how to connect quickly and successfully, using the default profiles programmed into your codec, follow in the sections ahead. If you require information about how your **Tieline** codec operates, i.e. how to navigate through menus and how to dial, please see the Operation of Your Codec section in the main reference manual for each **Tieline** codec.

WARNING for Installation of Modules

If you are installing or uninstalling any module, **please ensure that the codec is not powered up** during the installation process. If possible use anti-static precautions to help minimize the chances of static charges damaging the highly sensitive circuitry. Do not force a module into the codec. Modules should be installed slowly and gently.

WARNING: Ensure that both the local and remote codecs are connected correctly before use. If you plug a digital PBX line into the RJ11 connector in a POTS module or into the RJ 45 connector in an ISDN module, permanent damage may result from the high voltage pulses digital systems generate.

IMPORTANT ISDN CONNECTION INFORMATION

SPID1, SPID2 (Service Profile ID): Users outside North America please leave the SPID1 and SPID2 blank. North American users need to get SPID (Service profile ID) numbers from the phone company and enter them exactly as supplied.

DN1, DN2 and MSN Numbers: (Directory Numbers and Multiple Subscriber Numbers) A Directory Number is the ISDN number assigned to each B channel of an ISDN line. In North America, users must get DNs from the phone company. These must be entered exactly as provided. In most other locations MSN numbers are used when more than one unit is on the line and will be allocated by the Telco.

A Cautionary Note: You usually need to enter DNs with an area code (10 digits), while some other ISDN devices may only require 7 digit numbers. If your switch type is an AT&T 5ESS custom point-to-point, you may need to leave both *DN*s blank - even though they have been provided by the phone company. If you are using your codec in places where MSN numbers (Multiple Subscriber Numbers) are used, enter the MSN number in the DN1 and DN2 menu in the codec.

7.1. Step 1: Set Audio Configuration Settings

First, plug in your microphones and high-level audio sources into your codec and configure the relevant audio settings you require by pressing SOFTKEY 1 Aud on the codec. The settings available are displayed in the following image of the codec menu structure.

If you require any further details about any of these menu settings, please consult the relevant sections in this reference manual. Hyperlinks to the relevant sections follow. If you are reading this manual in hard copy please consult the relevant sections in the elements of this manual relating to ToolBox.

- Input Gains;
- Mic Phantom Power;
- IGC Enable;
- IGC Auto Level;
- PPM Meters;
- Audio Reference Level;
- AGC Enable;
- Route Ph IN to Ch Out;
- Aux Vol/Ctrl; and
- Phone Vol/Ctrl.

The Oscillator Enable menu item simply allows you to enable the oscillator within the codec and send 400Hz tone to all the outputs of the codec. This is particularly helpful when you are setting up the codec as it allows you to check and line up codec connections with the studio or broadcast center. It is also helpful when you are setting up because it assists you in checking that equipment such as headphones are working and that other analog output feeds (etc) are functioning correctly.

7.1.1. Codec Audio Menu

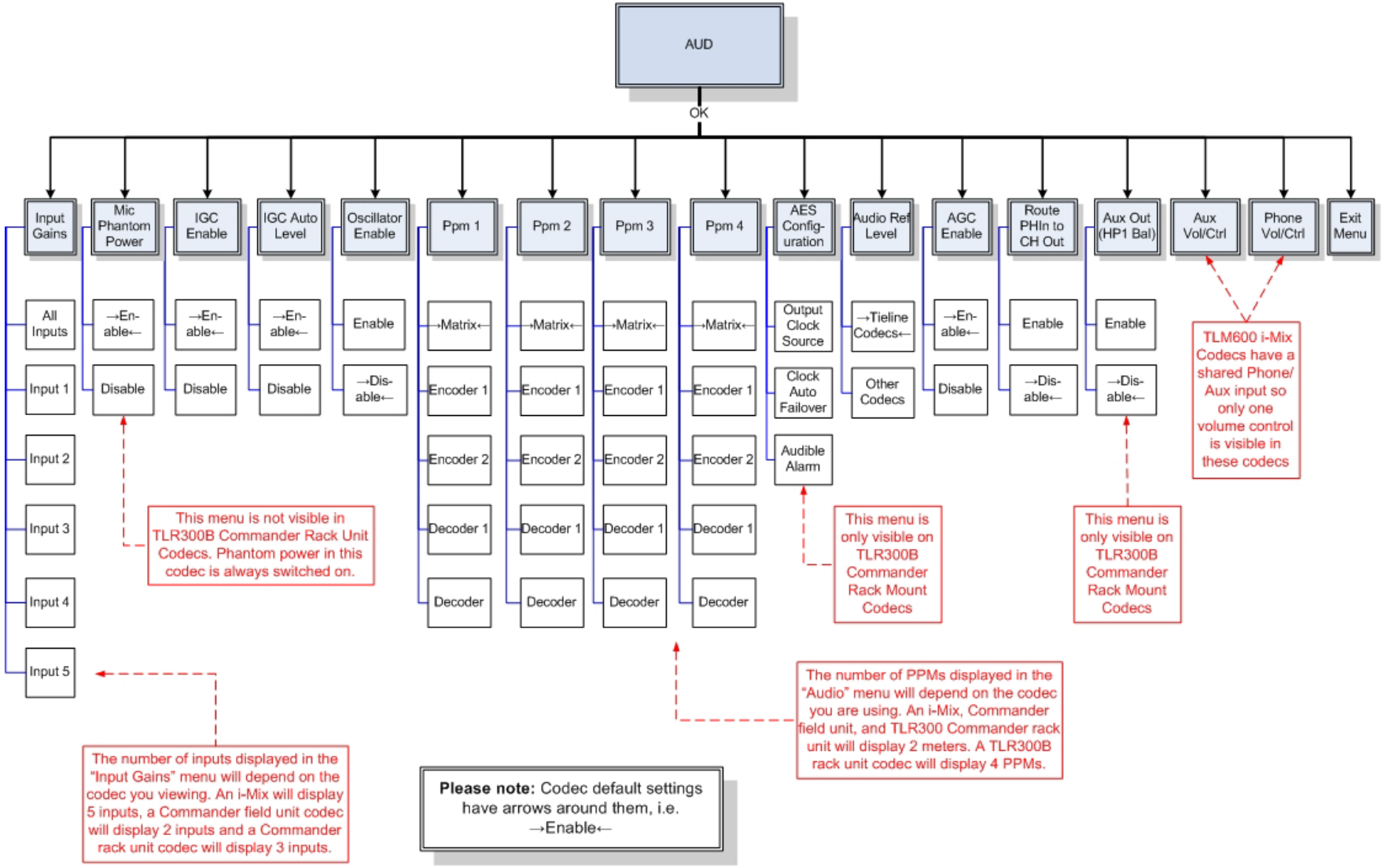


Figure 6: Codec Audio Menu Structure

7.2. Step 2: Select a Profile

Next, select the profile to suit your broadcast requirements. The manual default profiles, programmed into all **Tieline** codecs, can be selected by pressing SOFTKEY 2 Profile. The manual default profiles, and any custom profiles you have loaded onto a codec, will be displayed in this menu. Select the profile you require by scrolling with the MENU SELECTOR (MS) until the flashing square brackets surround the profile you wish to select. Press the MS to select the profile and then press SOFTKEY 2 to load the profile into the codec.

7.2.1. Which Profile Should I Select?

Some users will be familiar with the **Tieline** family of codecs. If this is the case then you will probably be familiar with the five manual default connection profiles programmed into **Tieline** codecs. You can connect your codec quickly using these profiles, or by using *Current Runtime*. These manual default profiles are explained below and they are also described in more detail within this manual in the section titled Manual Default Profiles.

7.2.2. Manual Default Mono Program

With this profile you can send bi-directional mono program audio of up to 15 kHz between two codecs over an analog phone line, a GSM cell-phone, a 3G cell-phone, an ISDN B channel, an X.21 connection, or IP (Internet, DSL, LAN, WAN).

7.2.3. Manual Default Mono/IFB

With this profile you can send bi-directional mono program audio of up to 15 kHz between two codecs and also communicate using a high-bandwidth IFB (off-air communications) circuit at the same time. Use either one or two ISDN B channels[2], two POTS connections, X.21, IP (Internet, DSL, LAN, WAN), a 3G cell-phone, or a combination of these connections.

7.2.4. Manual Default Stereo

With this profile you can send two channels of up to 15 kHz audio as a stereo program feed. Use either one or two ISDN B channels[2], two POTS lines, a 3G cell-phone, an X.21 connection, or IP (Internet, DSL, LAN, WAN).

7.2.5. Manual Default Dual Program

With this profile you can send a mix of all inputs over two channels of up to 15 kHz mono audio, or simultaneously send two separate bi-directional mono feeds over the same or two different networks. Use any combination of

[2] The bandwidth of one 64kb ISDN B channel and one satellite ISDN connection can be split to deliver two 15kHz signals. (See *Connection 2* of the *Manual Default Stereo* profile in the ***Connection Setup*** section of the *Profile Editor* in ToolBox).

POTS, GSM, 3G, X.21, ISDN or IP connections (Internet, DSL, LAN, WAN) to send these audio streams.

7.2.6. Manual Default Bonded Mono

Please note: This profile has been replaced by automatic bonding and is explained in the section of this reference manual titled Dialing Bonded Mono Connections.

If you would like more detailed information about these profiles, please click on the hyperlinks below to go to the relevant sections (if you are reading this manual on a PC). If you are reading the manual in hard-copy format go to the section titled Manual Default Profiles.

- *Manual Default Mono Program;*
- *Manual Default Mono/IFB;*
- *Manual Default Stereo;*
- *Manual Default Dual Program.*

Very Important Connection Note: Please ensure that you set both the local and remote codecs to operate using the same algorithm – otherwise your codecs will not operate correctly.

If you are using manual default profiles, when the dialing codec calls the remote codec, the dialing codec will automatically set both the profile and the default algorithm on the remote codec to match the dialing codec's setting.

In addition, it is very important to note that remote control of a codec is not possible unless you are using either the *Music, Music Plus* or *Voice G3* algorithms.

7.3. Step 3: Change Connection Settings

Next, adjust any connection settings. To do this simply scroll with the MENU SELECTOR (MS) to the connection you will be using (POTS, ISDN, GSM, 3G, X.21 or IP). Next press SOFTKEY 3 Wiz to open the wizard for that connection. Press SOFTKEY 4 OK to proceed through the wizard options for that connection.

Once you have opened the wizard, the current menu configuration settings will have the two inward pointing arrows (→ ←) next to them. Press SOFTKEY 4 OK to save any new selections that you have scrolled to with the MENU SELECTOR, or to keep any of the current settings and continue through the wizard. Press SOFTKEY 3 Exit to exit the wizard at any time (any changes prior to exiting the menu will be saved). Press SOFTKEY 1 Prev to navigate to the previous wizard menu item.

Figure 7: POTS Menu Wizard Screen Displaying Editing Functionality Available

The menu structure of the POTS, ISDN, GSM and IP connection wizards is displayed in the next few pages. Following this is a brief discussion about algorithm and connection bitrates, including suggested rule-of-thumb minimum connection bitrates for different connection types. We then complete this section of the manual with quick start connection procedures for each manual default profile and connection type.

7.3.1. New POTS Wizard

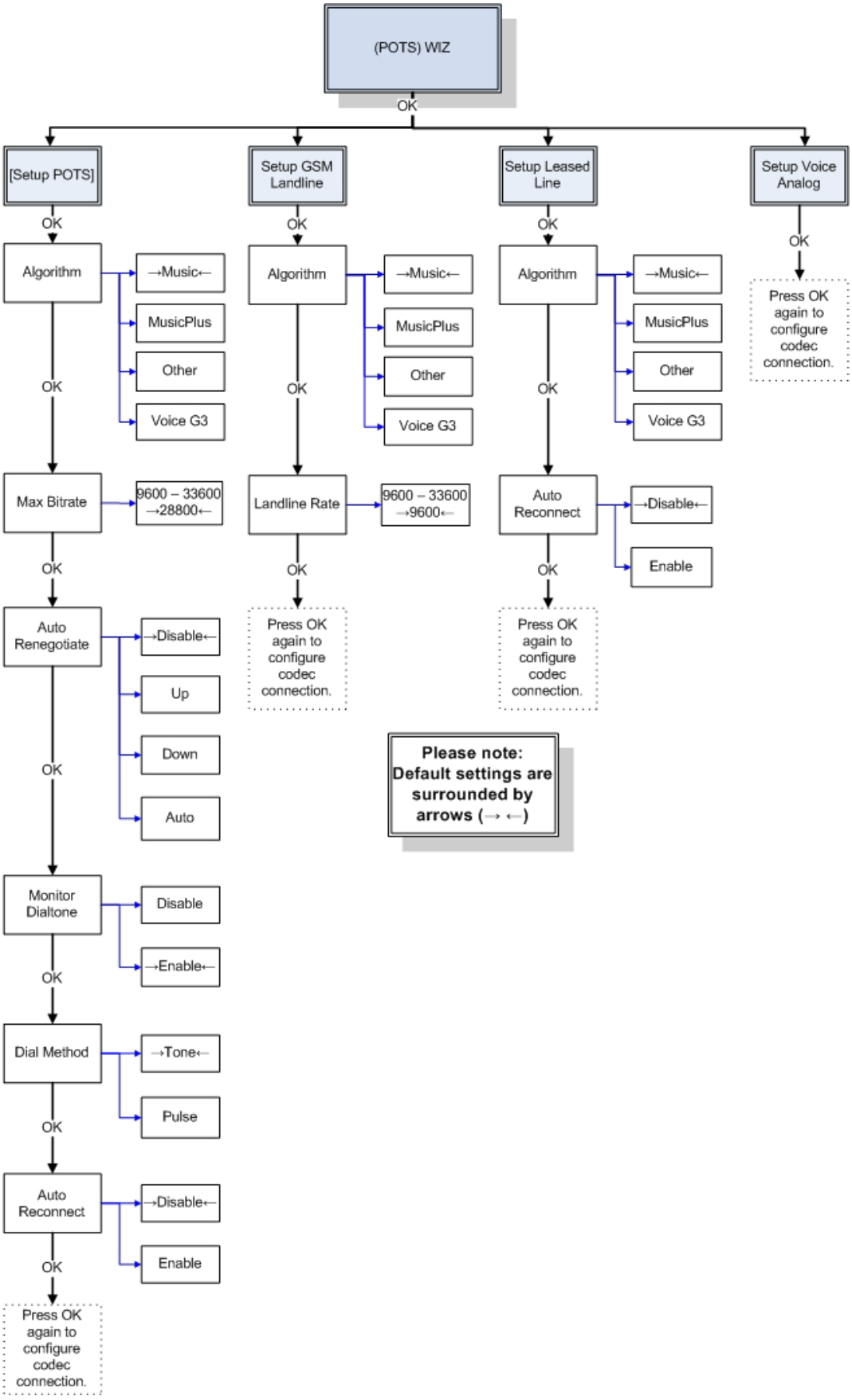


Figure 8: POTS Menu Configuration Wizards

7.3.2. New ISDN Wizard

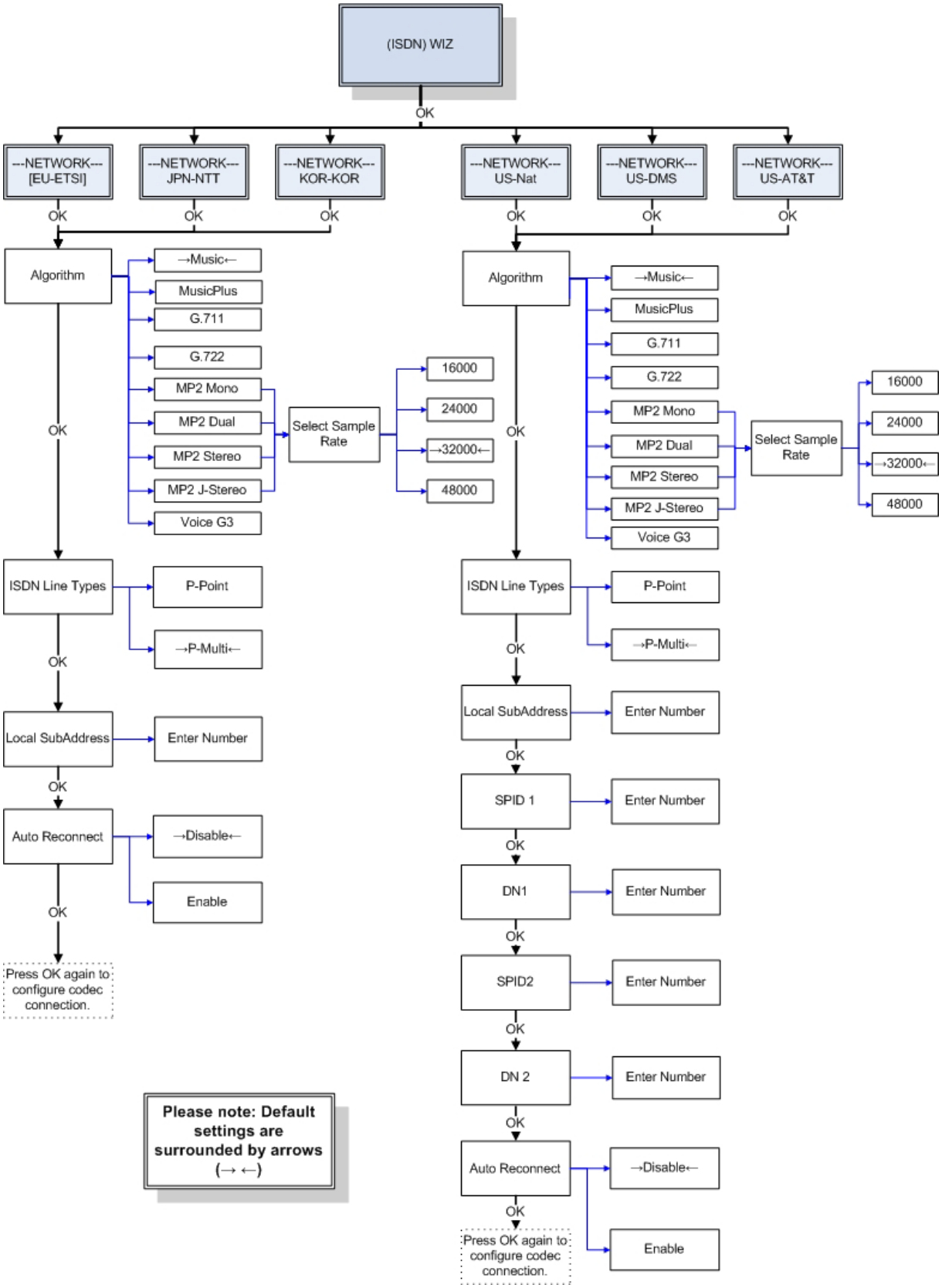


Figure 9: ISDN Menu Configuration Wizard

7.3.3. New GSM Wizard

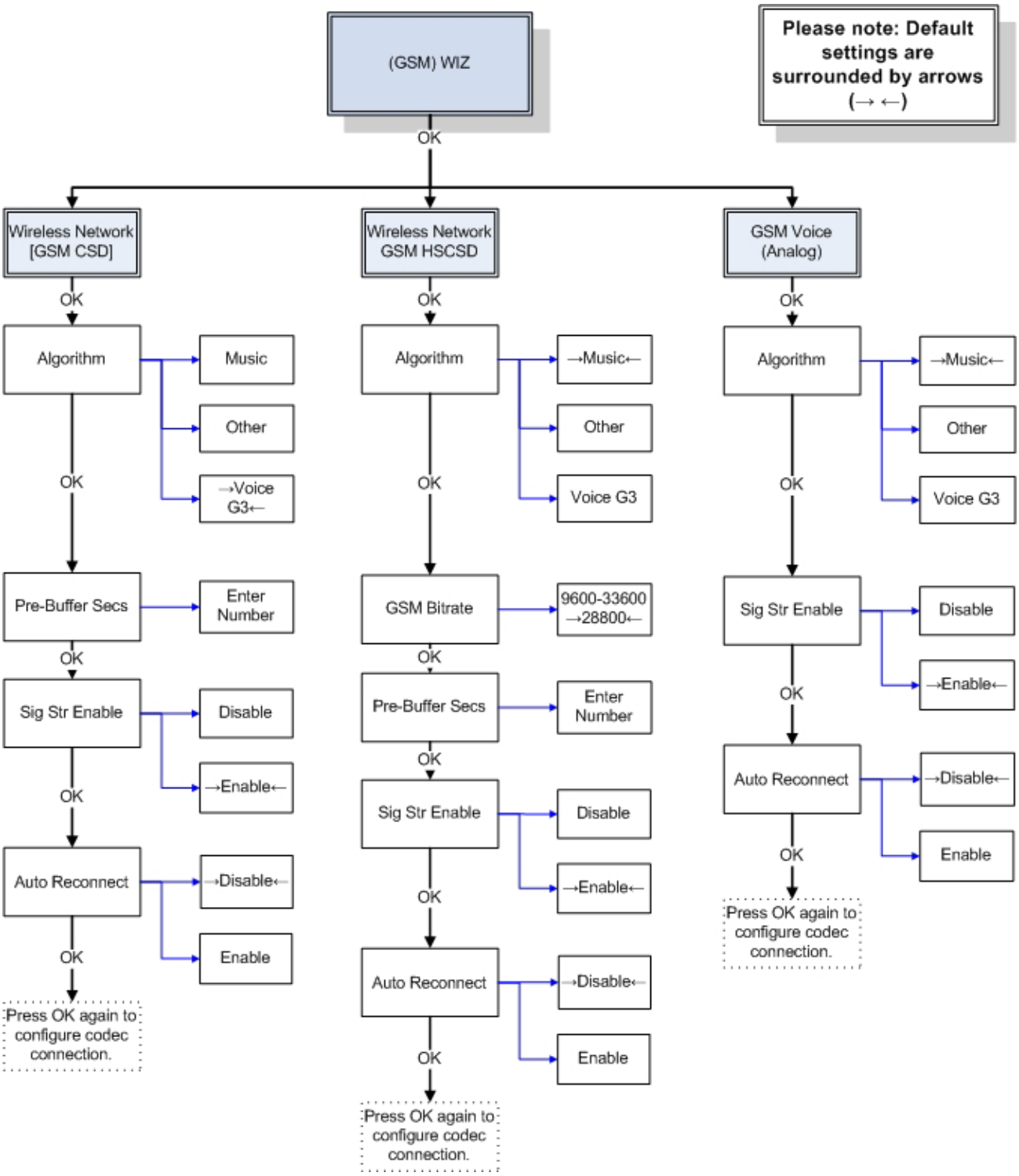


Figure 10: GSM Menu Configuration Wizard

7.3.4. New IP Wizard

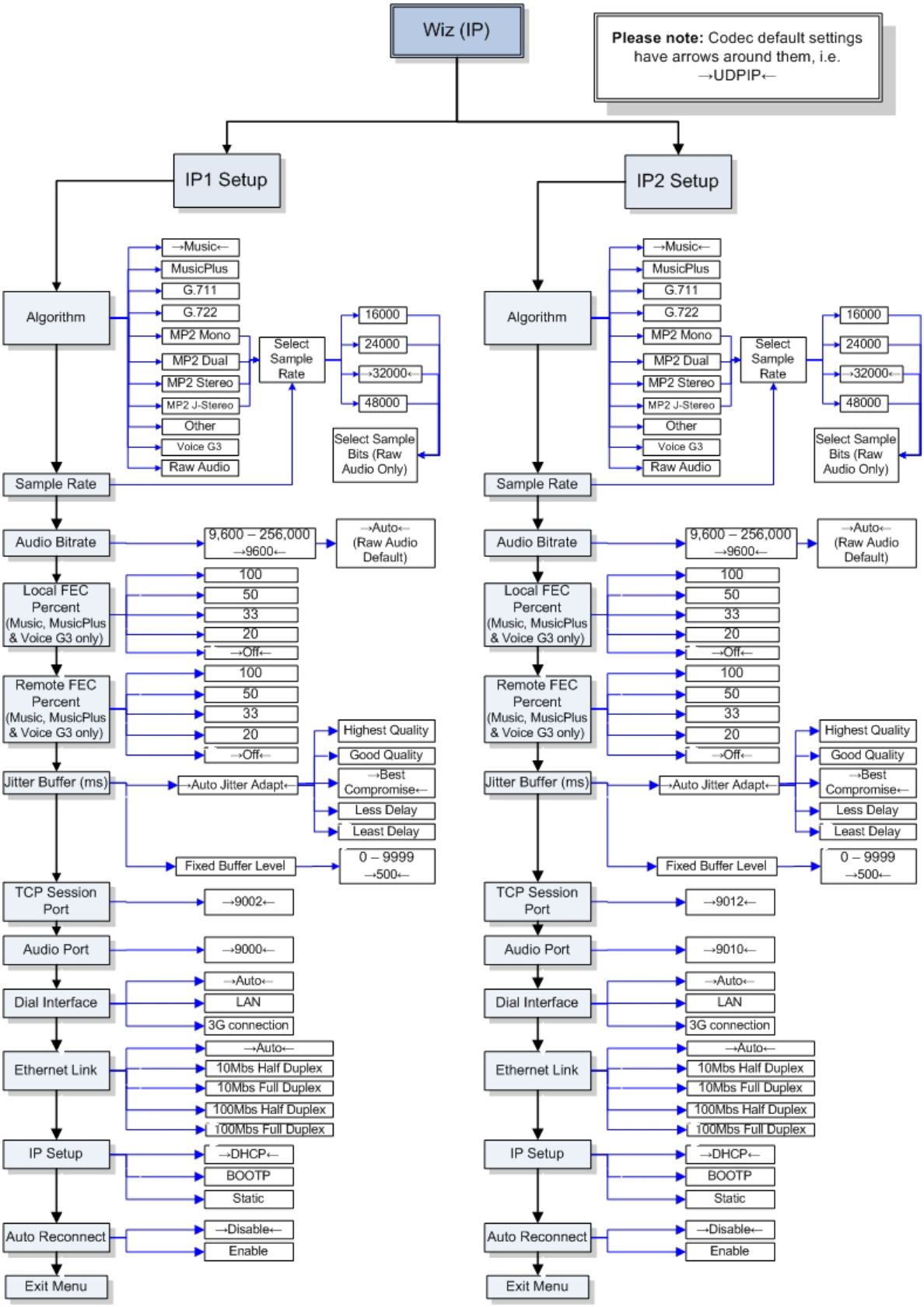


Figure 11: IP Menu Configuration Wizard

7.3.5. 3G Wizard

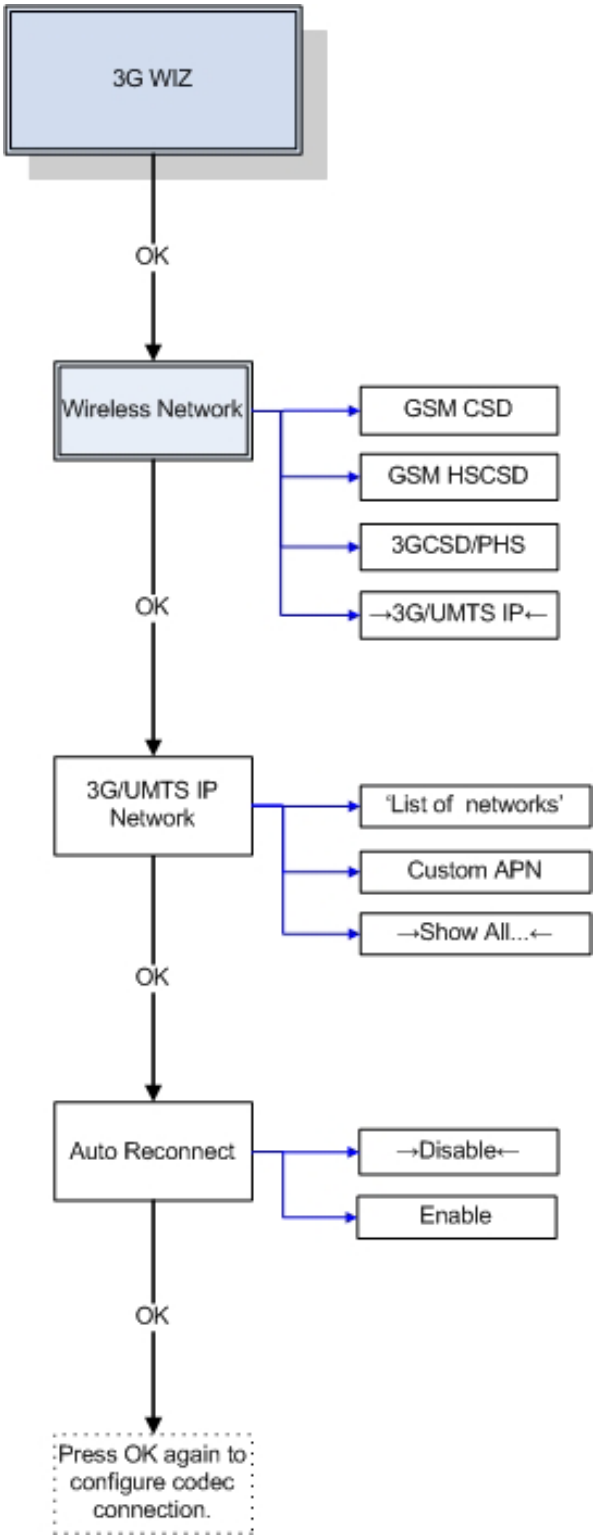


Figure 12: 3G Menu Wizard

7.3.6. New X.21 Wizard

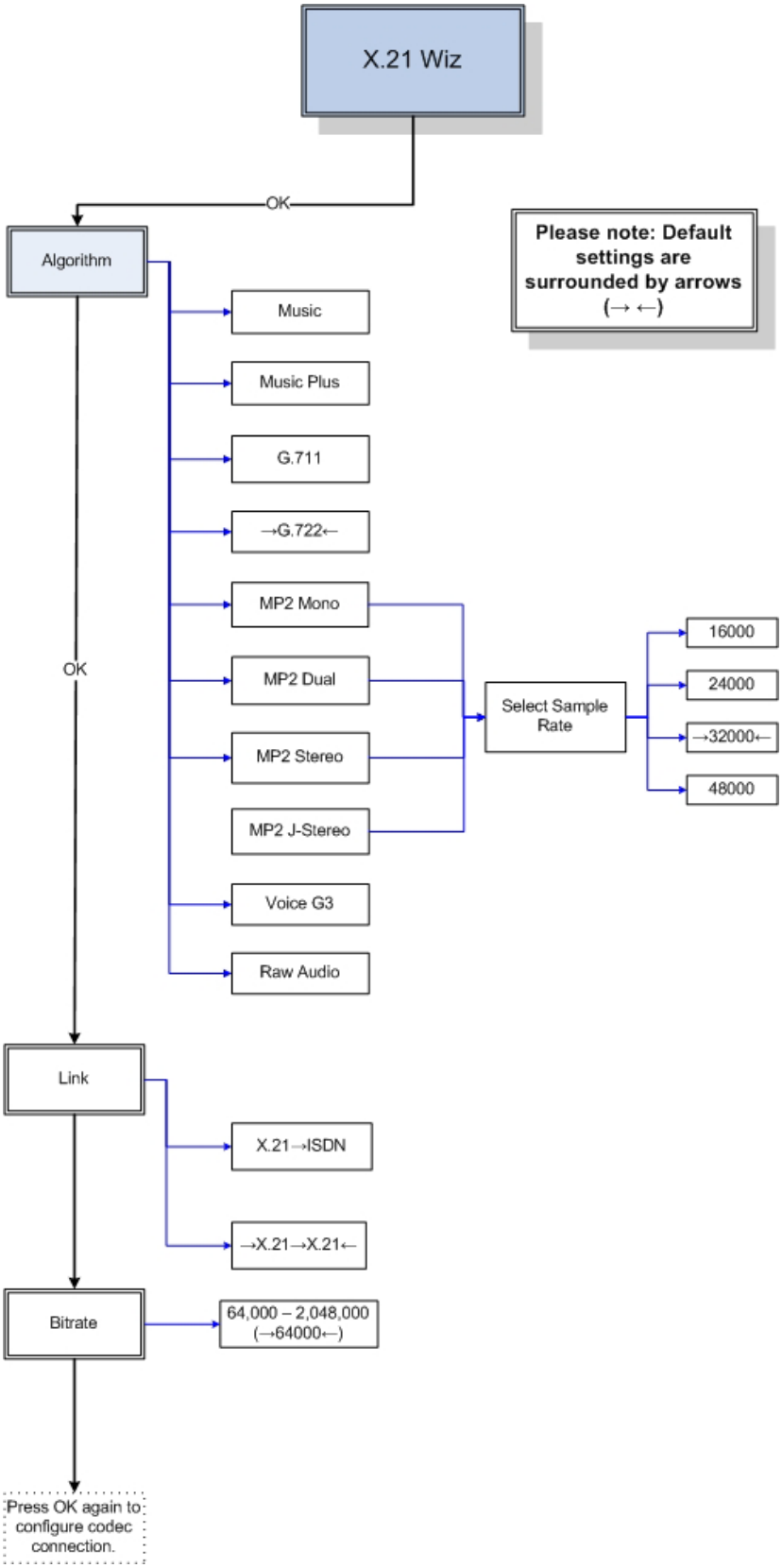


Figure 13: X.21 Menu Wizard

7.3.7. Connection Bitrates and Algorithm Options

In this section you will find the Uplink Bandwidth table for IP connections. This displays recommended rule-of-thumb settings for IP connections, based on the uplink bandwidth of your broadband connection. Following this is a table showing the recommended connection bit-rates for the factory default algorithm settings, which are used by the manual default profiles. These settings are a general guide and can be changed if required. In addition, there are tables displaying all the available codec profiles and the algorithms you can select, depending on the type of connection you are using.

The algorithm you connect with will be dependent on the program audio you are sending and the connection bit-rate you are able to achieve. *Music* is generally the preferred POTS algorithm setting if your program content contains music. *Voice G3* is generally best to use for a POTS connection if your connection bit-rate is 16,800 bps or lower. It is a good idea to listen to the quality of your program signal and to see how it sounds when it is sent at different bit-rates. This will assist you to determine what the best algorithm setting is for the signal you will be broadcasting.

7.3.8. UDP 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.)

	Dialup and DSL (ADSL) Broadband Uplink Bandwidth						
Codec Settings	**33.6kb Dialup**	**64kb DSL**	**128kb DSL**	**256kb DSL**	**512kb DSL**	**1,024kb DSL**	**Wireless Wi-Fi**
Audio Bitrate	9.6 - 14.4kb	9.6 - 16.8kb	9.6 - 28.8kb	9.6 - 64kb	9.6 - 128kb	9.6 - 128kb	9.6 - 16.8kb
Algorithm	Voice G3	Voice G3	Voice G3 or Music	Music	Music or Music Plus	Music or Music Plus	Voice G3
Forward Error Correction	Off	20% - 33%	20% - 33%	20% - 50%	20% - 100%	20% - 100%	100%
Jitter Buffer Ms	500ms	250 - 500ms	200 – 350ms	100 - 300ms	100 - 300ms	100 - 300ms	250 - 750ms
Mono Profile	✔	✔	✔	✔	✔	✔	✔
Stereo Profile	✘	✘	✘	✔	✔	✔	✘
Dual Profile	✘	✘	✘	✔	✔	✔	✘
Mono/IFB	✘	✘	✘	✔	✔	✔	✘

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.

7.3.9. MPEG 2 Layer 2 Recommended Bitrates

The following tables indicates the connection bitrates that are recommended for *MP2* (MPEG 2 Layer 2) algorithm connections, which are available in **Tieline** codecs. The possible bitrates for mono, dual mono, stereo and joint stereo (J-stereo) connections are provided for ISDN, X.21 and IP connections.

The recommended bit rates are the ones that will provide the best quality audio at different sample rates. If you configure your codec via the connection wizard then only the recommended bit rates are displayed for each connection sample rate and each MP2 algorithm.

7.3.9.1. Dual Mono MP2

If connecting using a dual mono profile with *MP2* algorithms it is only possible to use these profiles using either two ISDN B channels or two X.21 module connections. It will require either two 64kbps B channels or two 64kbps (minimum) X.21 connections.

Dual mono MP2 is not possible over IP or 3G/IP. There is only one plug-in module slot in an ***i*-Mix *G3*** so dual mono MP2 audio cannot be sent using an X.21 interface with this codec.

Important Notes:

1. If you are looking to connect at a bit rate of 256 kbps it is better to use *MP2 Stereo* than *MP2 J-Stereo* because 128 kbps per channel is adequate for high quality MP2 Stereo connections.
2. The connection wizard is the safest way to configure connections. This is because in the main connection menu via **[Menu] > [Configuration] > [ISDN/IP/X.21]**, all connection bit rates are displayed for selection. Whereas in the connection wizard only relevant algorithm and sample rate settings are available in order to optimize the quality of connections.

7.3.9.2. Recommended ISDN Bit rates

MP2-Mono	64K	128K	192k	256K
16kHz	✓			
24kHz	✓	✓	✓	
32kHz	✓	✓	✓	
48kHz	✓	✓	✓	

MP2-Dual	64K	128K	192k	256K
16kHz	✓			
24kHz	✓	✓		
32kHz	✓	✓	✓	
48kHz	✓	✓	✓	

MP2-J Stereo	64K	128K	192k	256K
16kHz	✓			
24kHz	✓	✓		
32kHz		✓	✓	
48kHz		✓	✓	

MP2-Stereo	64K	128K	192k	256K
16kHz	✓			
24kHz	✓	✓		
32kHz		✓	✓	✓
48kHz		✓	✓	✓ *TLR300B only

Table 2: ISDN MPEG 2 Layer 2 Recommended Bitrates

7.3.9.3. Recommended X.21 Bit rates

MP2-Mono	64K	128K	192k	256K	384K	768K	1M	1.5M	2M
16kHz	✓								
24kHz	✓	✓							
32kHz	✓	✓	✓						
48kHz	✓	✓	✓						

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MP2-Dual		128K	192k	256K	384K	768K	1M	1.5M	2M
16kHz	✓								
24kHz	✓	✓							
32kHz	✓	✓	✓						
48kHz	✓	✓	✓						

MP2-JStereo	64K	128K	192k	256K	384K	768K	1M	1.5M	2M
16kHz	✓								
24kHz	✓	✓							
32kHz		✓	✓						
48kHz		✓	✓						

MP2-Stereo	64K	128K	192k	256K	384K	768K	1M	1.5M	2M
16kHz	✓								
24kHz	✓	✓							
32kHz		✓	✓	✓					
48kHz		✓	✓	✓ *TLR300B only					

Figure 14: X.21 MPEG 2 Layer 2 Recommended Bitrates

7.3.9.4. Recommended IP Bit rates

MP2-Mono	64K	96K	112K	128K	192k	256K
16kHz	✓					
24kHz	✓	✓	✓	✓		
32kHz	✓	✓	✓	✓	✓	
48kHz	✓	✓	✓	✓	✓	

MP2-Dual	64K	96K	112K	128K	192k	256K
16kHz	✓					
24kHz	✓	✓	✓	✓		
32kHz	✓	✓	✓	✓	✓	
48kHz	✓	✓	✓	✓	✓	

MP2-J Stereo	64K	96K	112K	128K	192k	256K
16kHz	✓					
24kHz	✓	✓	✓	✓		
32kHz			✓	✓	✓	
48kHz			✓	✓	✓	

MP2-Stereo	64K	96K	112K	128K	192k	256K	384K
16kHz	✓						
24kHz	✓	✓	✓	✓			
32kHz	✓		✓	✓	✓	✓	
48kHz			✓	✓	✓	✓ *TLR300B only	

Figure 15: IP MPEG 2 Layer 2 Recommended Bitrates

Please note: FEC is not currently offered for use with MPEG algorithms. It is scheduled to be incorporated in future software releases. Please check at support@tieline.com for details of the implementation date of this feature.

In addition, the default connection bit rate for MP2 algorithms is 64kbps over IP. Other algorithms connect at 9,600bps by default.

Connection Type	Algorithm	Minimum Bit-rate and Bandwidth	Recommended Connection for On-air use.
POTS	Music	Up to 7.5kHz at 9,600 bps[3]	Up to 15kHz at 24,000 bps
POTS	Voice G3	7.5 kHz at 9,600 bps (Optimized for 16.8 kb, use for voice only)	7.5 kHz at 9,600 – 24,000 bps (use for voice only)
ISDN	Music Plus	20 kHz on all ISDN connections.	
ISDN	Music	15kHz on all ISDN connections.	
ISDN	MP2	24 kHz mono at 64,000 bps.	
ISDN	G.711	3 kHz at 64 kbps.	
ISDN	G.722	7 kHz at 64 kbps.	
GSM CSD	Voice G3	7.5 kHz at 9,600 bps.	7.5 kHz at 9,600 bps
GSM HSCSD	Music[4]	7.5 kHz at 14,400 bps (Using Voice G3)	15kHz at 28,800 bps (Using Music)
3G/IP	Music Plus	Up to 20kHz mono at 48,000 bps	Up to 20kHz mono at 48,000 bps
	Music[5]	Up to 15 kHz at 24,000 bps	Up to 15kHz at 24,000 bps or higher
	Voice G3	7.5 kHz at 9,600 bps (use for voice only)	7.5 kHz at 9,600 - 14,400 bps[6] (use for voice only)
IP/Internet	Music Plus	Up to 20kHz mono at 48,000 bps	Up to 20kHz mono at 48,000 bps
	Music	Up to 7.5kHz at 24,000 bps	Up to 15kHz at 24,000 bps or higher
IP/Internet	Voice G3	7.5 kHz at 9,600 bps (use for voice only)	7.5 kHz at 9,600 - 14,400 bps[7] (use for voice only)
IP	Raw Audio[8]	15 kHz at 2.3 megabit minimum.	
X.21	Music Plus	20 kHz on all X.21 connections.	
	Music	15kHz on all X.21 connections.	
X.21	MP2	24 kHz at 64,000 bps.	
X.21	G.711	3 kHz at 64 kbps.	
X.21	G.722	7 kHz at 64 kbps.	

Table 3: Algorithm Connection Bit rate Table

- All of the factory default algorithm settings can be changed for POTS, ISDN, GSM/3G and X.21 connections. For more information on changing these settings please see the main reference manual.

[3] If your connection bit-rate is 16,800 bps or lower, we recommend using the *Voice G3* algorithm. It provides up to 7.5 kHz audio at 9,600 bps.
[4] If you wish to connect using a GSM HSCSD connection and achieve up to 15 kHz in audio quality, you will need to change the default algorithm setting to *Music*.
[5] Stereo IP and 3G/IP connections using the *Music* algorithm require a minimum connection bit rate of 48kbps.
[6] Using Voice G3, there is no need to negotiate higher than 14.4 or 16.8kbps to maximize audio quality. Staying at lower bit-rates improves link stability over TCP.
[7] Using Voice G3, there is no need to negotiate higher than 14.4 or 16.8kbps to maximize audio quality. Staying at lower bit-rates improves link stability over TCP.
[8] Only available over a point-to-point crossover cable (i.e. CAT 5) and using high quality LAN switching.

- Every manual default connection includes a data channel of 50 bytes per second which can be used to send RS232 data between devices attached to the serial port of each **Tieline** codec.
- It is very important to note that remote control function of a codec is not possible unless you are using the *Music, MusicPlus* or *Voice G3* algorithms.

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Tieline G3 Codec - Profile / Algorithm Matrix

	Music	Other	Voice G3	G.711	G.722	MP2 Mono	MP2 Dual	MP2 Stereo	MP2 J-Stereo	RAW Audio	Music Plus
POTS											
ManDflt MonoPgm	✔	✔	✔								
ManDflt Mono/IFB	✔		✔								
ManDflt Stereo	✔		✔								
ManDflt DualPgm	✔		✔								
IP											
ManDflt MonoPgm	✔		✔	✔	✔	✔	✔			✔	✔
ManDflt Mono/IFB	✔		✔	✔	✔		✔			✔	✔
ManDflt Stereo	✔		✔					✔	✔	✔	✔
ManDflt DualPgm	✔		✔	✔	✔		✔			✔	✔
3G											
ManDflt MonoPgm	✔		✔			✔					✔
ManDflt Mono/IFB	✔		✔								✔
ManDflt Stereo	✔		✔								✔
ManDflt DualPgm	✔		✔								✔
SIP											
ManDflt MonoPgm				✔	✔	✔				✔	
ManDflt Mono/IFB											
ManDflt Stereo								✔	✔	✔	
ManDflt DualPgm											

GSM											
ManDflt MonoPgm	✔		✔								
ManDflt DualPgm	✔		✔								

ISDN											
ManDflt MonoPgm	✔		✔	✔	✔	✔	✔				✔
ManDflt Mono/IFB	✔		✔	✔	✔		✔				✔
ManDflt Stereo	✔		✔					✔	✔		✔
ManDflt DualPgm	✔		✔	✔	✔		✔				✔

X.21											
ManDflt MonoPgm	✔		✔	✔	✔	✔	✔			✔	✔
ManDflt Mono/IFB	✔		✔	✔	✔		✔			✔	✔
ManDflt Stereo	✔		✔					✔	✔	✔	✔
ManDflt DualPgm	✔		✔	✔	✔		✔			✔	✔

✔ **= Not recommended for use on TLF300 and TLR300 using new POTS G3 Module**

Tieline G3 Codec - Sample Rate / Algorithm Matrix

	Music	Other	Voice G3	G.711	G.722	MP2 Mono	MP2 Dual	MP2 Stereo	MP2 J-Stereo	RAW Audio	MusicPlus
Sample Rates											
8kHz				✔							
16khz					✔	✔	✔	✔	✔	✔	
24kHz						✔	✔	✔	✔	✔	
32kHz						✔	✔	✔	✔	✔	
48kHz						✔	✔	✔	✔	✔	
Proprietary	✔	✔	✔								✔

Table 4: Profile, Algorithm and Sample Rate Matrix

7.3.10. Dual Mono Program: Valid Algorithm Combinations

POTS-POTS

	Music	Other	VoiceG3
Music	✓	✓	×
Other	✓	✓	×
VoiceG3	×	×	✓

POTS-IP

	Music	VoiceG3	RawAudio-16	RawAudio-32
Music	✓	×	×	✓
Other	✓	×	×	✓
VoiceG3	×	✓	✓	×

POTS-ISDN

	Music	G711	G722	MP2-mono-16	MP2-mono-32	VoiceG3
Music	✓	✓	✓	×	✓	×
Other	✓	✓	✓	×	✓	×
VoiceG3	×	×	×	✓	×	✓

POTS-GSM

	Music	VoiceG3
Music	×	×
Other	×	×
VoiceG3	×	✓

IP-IP

	Music	VoiceG3	RawAudio-16	RawAudio-32
Music	✓	×	×	✓
VoiceG3	×	✓	✓	×
RawAudio-16	×	✓	✓	×
RawAudio-32	✓	×	×	✓

ISDN-ISDN

	Music	G711	G722	MP2-mono-16	MP2-mono-32	VoiceG3
Music	✓	✓	✓	×	✓	×
G711	✓	✓	✓	×	✓	×
G722	✓	✓	✓	×	✓	×
MP2-mono-16	×	×	×	✓	×	✓
MP2-mono-32	✓	✓	✓	×	✓	×
VoiceG3	×	×	×	✓	×	✓

ISDN-IP

	Music	VoiceG3	RawAudio-16	RawAudio-32
Music	✓	×	×	✓
G711	✓	×	×	✓
G722	✓	×	×	✓
MP2-mono-16	×	✓	✓	×
MP2-mono-32	✓	×	×	✓
VoiceG3	×	✓	✓	×

GSM-GSM

	Music	VoiceG3
Music	✓	×
VoiceG3	×	✓

GSM-IP

	Music	VoiceG3	RawAudio-16	RawAudio-24
Music	×	×	×	✓
VoiceG3	×	✓	✓	×

GSM-ISDN

	Music	G711	G722	MP2-mono-16	MP2-mono-24	VoiceG3
Music	×	×	×	×	✓	×
VoiceG3	×	×	×	✓	×	✓

Table 5: Dual Mono Program Algorithm Matrix

7.4. Getting Connected

The following pages include all the Quick Start connection information you require to connect quickly using either POTS, ISDN, GSM, X.21, 3G or IP connections.

If you are unsure about the connection bit rate you will require for the profile you have selected and the type of connection you are using, please refer to the tables in the preceding sections before you attempt to connect. They will provide rule-of-thumb guidance for different connection bit rate settings.

Please Note: The connection procedures described in this Quick Start manual are for manual connections. If you wish to use the **Tieline** Connection Manager (CXNS) to connect automatically, please see the 'Connections Manager' section in this reference manual for more information.

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.

7.5. 10 Simple Steps to Connect Tieline Codecs (Version 1.6.xx firmware or higher)

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.

IP Warnings: If you connect over IP and use auto jitter buffer then both codecs must have firmware v.1.6.xx installed. Unless the remote codec has a public IP address assigned to it and you know what the number is (or you are connecting using a SIP server), 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.

If you are unable to achieve a connection using these instructions, please refer to the relevant section of this reference manual for more detailed connection information. If you connect with IP, 3GIP or SIP, please download the "IP and 3GIP Streaming Reference Manual" from the **Tieline** website at www.tieline.com .

If you continue to have difficulty connecting, contact **Tieline** at support@tieline.com for more assistance.

Rack Mount Codec Control PC Software Information

If you are using the TLG3 GUI codec controller to connect to a rack unit codec it will emulate a 'physical' front panel for connection purposes. Please see the section in the reference manual for the TLR300B codec titled 'Operation of the TLG3 GUI', for more information on its installation and operation.

WARNING for Installation of Modules:

If you are installing or uninstalling any module, **please ensure that the codec is not powered up** during the installation process. We recommend you use anti-static precautions to help minimize the chances of static charges damaging the highly sensitive circuitry. Do not force a module into the codec. Modules should be installed slowly and gently.

GSM WARNING: Only one GSM module can be used in a ***COMMANDER G3***. If you install two GSM modules an error message will be displayed. If you wish to operate over two GSM connections you will need to use the RS232 connection on the rear of a codec and connect a GSM-compatible cell-phone such as the *Nokia*® 6310 or 6310i. If you are using an RS232 connection you will also need to configure the codec serial port for GSM use. To configure GSM cell-phone connections press SOFTKEY 4 Menu, scroll to **[Configuration] > [Advanced Menu] > [Serial Port Mode] > [GSM].** This configures the serial port and the GSM connection should appear on the main menu screen as a connection option.

VERY IMPORTANT GSM WARNING FOR RACK UNIT CODECS:

GSM plug-in modules cannot be used in ***COMMANDER G3*** rack unit codecs. If you must connect using GSM with a ***COMMANDER G3*** rack unit codec, we suggest you use a GSM cell-phone attached to the RS232 port on the rear panel of the codec. Alternatively, you could try a GSM phone or a 3G cell-phone in GSM mode with a **Tieline** USB module.

Important Notes: A POTS landline-connected codec must be in GSM Landline Mode to receive a GSM CSD data call. To configure this, select the POTS connection on the codec using the MS and press SOFTKEY 3 Wiz to use the connection wizard. Select Setup GSM Landline in the wizard. More detailed information about GSM connections is contained in the GSM Operation section of every codec's main reference manual.

Warning on Digital Phone Systems: Digital phone systems typically run off voltages greater than the 50 volts used by the PSTN. Some older ISDN systems run on approx 100 volts. Connection of a **Tieline** POTS codec to a voltage greater than the normal 50 volts will cause damage to a POTS codec and void the warranty.

7.6. 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 3GIP1 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 or USB modem for your region, check with your local **Tieline** dealer or contact **Tieline** at support@tieline.com for more information. Alternatively, check the 3G section on our website at www.tieline.com for a list of compatible devices.

7.7. Remote Control of Input Controls

The simplest way to program remote control functions is with ToolBox software. Before discussing in detail how you can remotely control a **Tieline** codec, it is useful to ensure you are clear about how two codecs operate in relation to local and remote functions.

In this example we are using a ***COMMANDER G3*** field unit and rack unit codec, but the principles are the same for all **Tieline** codecs, including the ***i*-Mix *G3***. An ***i*-Mix *G3*** codec can control the functions of a ***COMMANDER G3*** codec, or vice versa if required.

The local codec is usually the codec at the studio or broadcast center. This codec is usually the 'master' codec when using remote control functions. In other words it is able to control the input gain settings and channel on/off functions on the remote codec. The local codec may have a PC connected to it using **Tieline** ToolBox software.

The remote codec, its companion partner, is usually at the broadcast venue or remote site. This is usually the 'slave' codec, able to be controlled by the local codec at the studio or broadcast center. A typical connection setup between two codecs is displayed in the following image.

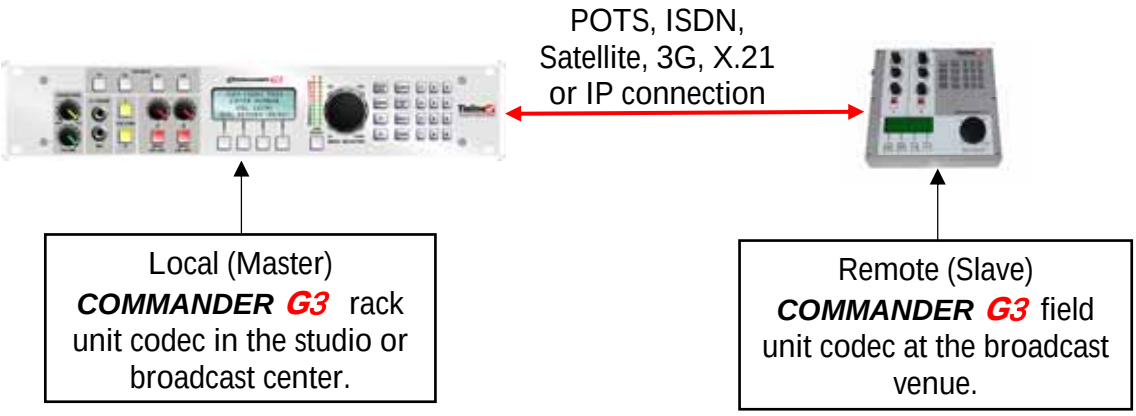


Figure 16: Typical Remote Control Setup without Using ToolBox

IMPORTANT WARNING: It is very important to note that remote control of a codec is not possible unless you are using either the *Music, Music Plus* or *Voice G3* algorithms.

Next we will explain how to configure a codec for remote control functionality. If you have a PC attached to your codec with ToolBox software installed on it, you can configure remote control settings for all codec inputs using this software. See the ToolBox section of this manual titled Remote Control for more details.

7.8. Configuring Remote Control via Codec Menus

The easiest way to explain configuring remote control is to use an example. In this example, we will show you how to configure the local (master) codec at the studio and the remote (slave) codec at a remote broadcast venue. To illustrate the power and flexibility of **Tieline**'s remote control functions, we will show how to use one channel on the master studio codec to control either a mono source, or a stereo source with two channels ganged together, on the slave venue codec.

In our example we will use channel 1 on the master codec to control channel 1 on the remote codec. We will then illustrate how to gang channel 2 to channel 1 on the slave codec – effectively allowing a single channel on the master codec to control a stereo source on the slave codec. This is a three step process.

Please note: In this example we are using channels 1 and 2 for ganging of a stereo source. These are the two channels sent left and right to encoders 1 and 2 in the default ***COMMANDER G3*** stereo profile. If you are remotely controlling an ***i*-Mix *G3***, channels 4 and 5 are configured for a stereo source (i.e., sent left and right to encoders 1 and 2), so use those channels for a stereo source with the default stereo profile.

Remote Control Recommendation:
Before commencing Remote Control configuration we recommend that you ensure that any previous settings have been deleted from the menus. To do this press SOFTKEY 4 Menu on each codec and then scroll to and select **[Remote Controls] > [Reset Remote Control] > [Yes]**.

7.8.1. Step 1: Configure the Master (Local) Codec

To configure channel 1 on the master codec to be the remote control master channel, controlling channel 1 on the remote slave codec, select SOFTKEY 4 Menu then scroll to and select **[Remote Controls] > [Setup Remote Ctrl] > [Setup Master Codec] > [Master Ctrl Ch1] > [Input 1].** It's as simple as that. Channel 1 is now configured as the remote control master of input 1 on the slave codec.

7.8.2. Step 2: Configure the Slave (Remote) Codec

Now let's configure the slave codec. Press SOFTKEY 4 Menu on the slave codec then scroll to and select **[Remote Controls] > [Setup Remote Ctrl] > [Setup Slave Codec] > [Slave Rem Lvl Ctrl1] > [Enable] > [Slave Rem OnOff Ctrl1] > [Enable].** It's as simple as that. Channel 1 is now slaved to channel 1 on the master codec. *Note: If you want to remotely control either the input gain or the channel on/off function only, simply enable only the function you require.*

7.8.3. Step 3: Gang Channels together to control a Stereo Source Remotely

To gang channel 2 on the slave codec to channel 1 on the slave codec press SOFTKEY 4 Menu on the slave codec then scroll to and select **[Remote Controls] > [Setup Remote Ctrl] > [Gang Inputs] > [Gang Input 2 To] > [Knob 1].** How easy was that! Now when input 1 on the master codec adjusts the gain setting on input 1

of the slave codec, the gain for input 2 on the slave codec will also be adjusted upwards or downwards. *Note: Channel on/off functions cannot be ganged – only input gain settings can be ganged.*

Please note: It is not possible to gang inputs together on the master codec and control several channels on a slave codec using the ganged master codec channels.

7.9. Adjusting Local (Master Codec) and Remote (Slave Codec) Input Levels When in Remote Control Channel Mode

If remote control functionality is enabled on both the master and slave codecs, and you have sources plugged into either of these codecs, you may wish to adjust the levels of these inputs locally - either before or after the codecs are connected.

To adjust the input level settings locally at either codec, simply press the CUE-COMMS button for the channel you wish to adjust and while holding it down adjust the input pot on the channel. If the CUE-COMMS button is being used for talkback then this will only work in offline mode. Please note that there is no CUE-COMMS button on a ***COMMANDER G3*** field unit codec so level adjustment of inputs by this method is not possible.

When a connection is made, the remote (slave) codec obtains gain settings from the local (master) unit and it then stores the setting in memory. (Related Topic: ***Inputs***)

7.10. Using the Menu Selector for Remote Level Control

Apart from using the channel gain knobs on the master codec for adjusting levels, it is also possible to use the MENU SELECTOR. To configure this press SOFTKEY 4 Menu on the master codec and then scroll to and select **[Remote Controls] > [Use Rem Ctrl] > [Master x].** This will allow you to control whichever channel's gain knob you select. This codec menu screen also allows you to control channel on/off for this channel.

7.11. Reset Remote Control

Remote control functionality can be reset to the factory default setting (no remote control functionality active) by pressing SOFTKEY 4 Menu on each codec and then scroll to and select **[Remote Controls] > [Reset Remote Control] > [Yes]**.

Following is a complete menu structure for the Remote Controls menu in **Tieline** codecs.

7.12. Codec Remote Control Menu Structure

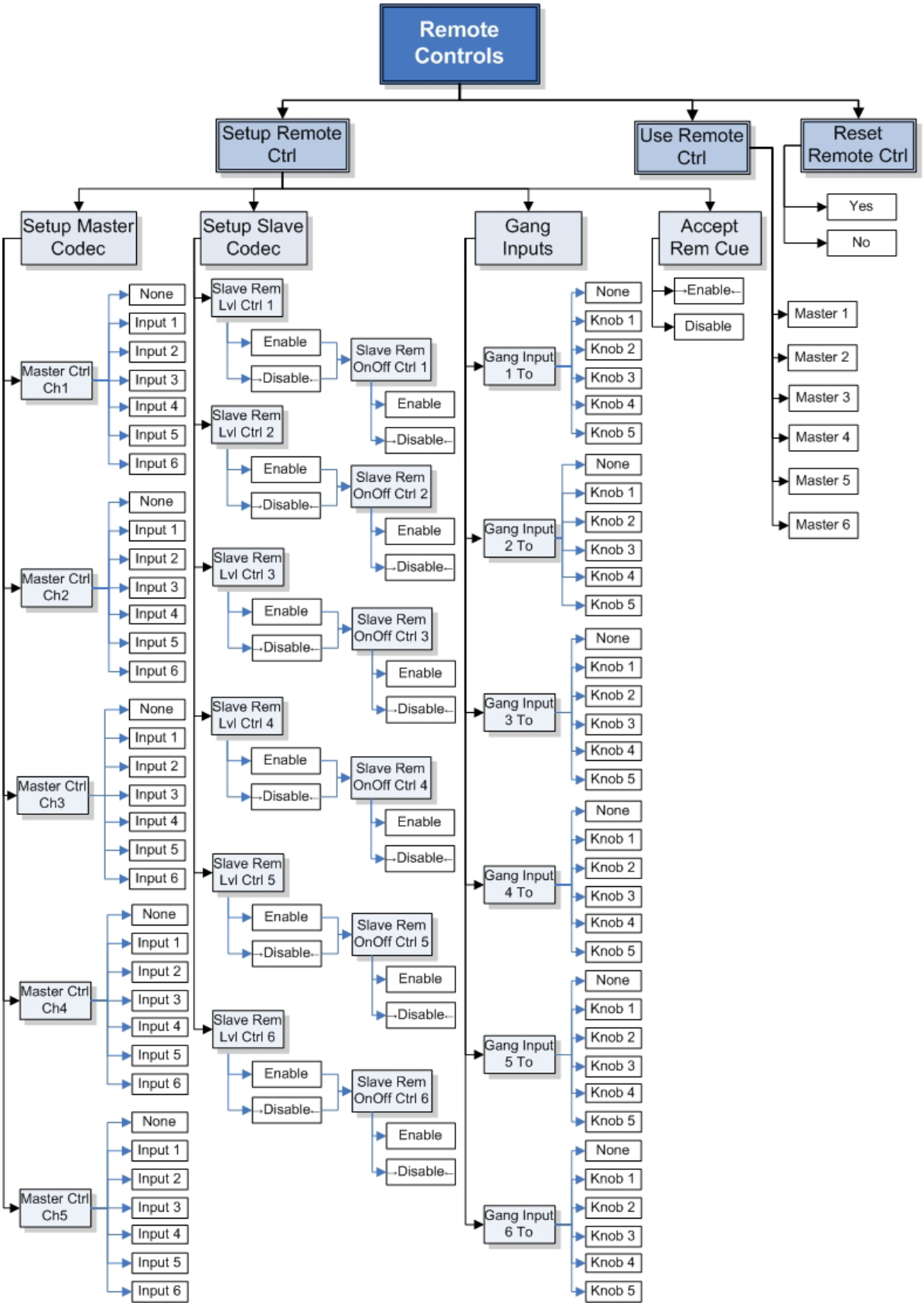


Figure 17: Remote Control Codec Menus

Section 8. POTS Operation and Usage Tips: How it works

All analogue audio inputs are digitized by a high quality stereo 24-bit analogue-to-digital (A to D) converter. This serial data stream is then fed to an extremely fast 32-bit Digital Signal Processor (DSP) that encodes the audio data using the **Tieline** proprietary digital audio compression algorithm. After compression in the DSP the resulting low bit rate serial data stream is fed to a data modem when in POTS mode, or to an ISDN data modem when in ISDN mode.

At the receiving **Tieline** codec, the reverse process takes place. The audio data is decompressed, fed to a digital-to-analogue (D to A) converter, then amplified to line level and fed to a balanced output connector.

ISDN operation is very similar to POTS operation except that a fixed bit rate of connection is used. The quality of sound at ISDN bit rates is near CD quality and the delay is an insignificant 100 milliseconds.

8.1. Historical Reflections

8.1.1. Coding Algorithms

Over the past two decades, there have been great improvements in the way that digital audio data can be condensed, while retaining the quality and subtle nuances of the original signal. Many different algorithms have been developed, including well-known MPEG layer 2 and 3.

These algorithms require a reasonably high bit rate, typically 64 kbps or higher, for high quality wide band mono operation. Many ISDN codecs use MPEG layer 2 and 3 for mono communications at 64 kbps and 128 kbps. However, for stereo, the inherent MPEG encoding and decoding delays can become a problem for real time operation.

While most MPEG algorithms may be suitable for ISDN operation, they are generally inadequate for the low bit rates available with POTS operation. Some manufacturers have tried to use MPEG for low bit rate POTS operation but have found the results unsatisfactory. A typical POTS line will achieve bit rates of less than 28,800 bps and few algorithms can deliver full bandwidth, high quality audio at these very low bit rates.

Fortunately, significant advances in the development of coding algorithms have made the design of digital audio codecs like the **Tieline** ***G3*** range possible. The proprietary music and voice coding algorithms used by **Tieline** achieve compression factors of the order of twenty times or more. This enables 15 kHz bandwidth high quality bi-directional audio to be transported at a bit rate as low as 24,000bps. 7 kHz bandwidth voice quality audio can be transported at bit rates down to 9,600bps. All bit rates have an end to end latency of only 100 milliseconds. This insignificant delay provides the kind of codec performance required for real time operation.

8.2. Modem Negotiation and Line Quality for POTS Mode

The ***i*-Mix *G3*** sends and receives high-speed digital information over a standard telephone line via an internal modem or a POTS module. A ***COMMANDER G3*** codec uses POTS modules only. The POTS modems communicate over the phone line to establish the best possible configurations for data transmission and to achieve the highest reliable data rate. This also corresponds to best possible audio quality.

Upon connection the modems initially establish a link at the default bit rate of 19,200 bps. The modems then perform a process called 'training' in which they analyze the line and compensate for frequency and phase response. This process also cancels out any echo that may be present.

The training process continues until the modems establish the highest reliable speed for data transfer using a process of 'negotiation'. Negotiation is the process of bit-rate adjustment. To negotiate bit-rates up press "**F2**" then "**3**", or for lower bit-rates press "**F2**" then "**9**".

The relationship between Line Quality, modem negotiation and retraining is described in further detail in the ***POTS*** menu section of the full manual under *Auto Renegotiate*.

8.3. POTS Operation Precautions

Tieline codecs are very reliable but a codec's performance will only be as good as the line that it is operating over. When using a POTS line, proper precautions MUST be taken to ensure that only the **Tieline** codec uses the line. Please ensure that you remove these possible interference sources:

- DSL or ADSL Modems;
- Other telephone handsets;
- Portable phone base stations;
- Unused parallel phone sockets;
- Fax machines;
- Computer modems;
- Burglar Alarm systems; and
- Extension bells.

8.4. Call Waiting

Call waiting tones may cause the codec to malfunction. Most phone companies supply call waiting as a feature and you will need to turn it off. In the internal phone book, program the number for disabling call waiting, and dial it before calling anyone

8.5. Private Branch Exchanges (PBX), Private Automatic Branch Exchanges (PABX), Business Systems

Avoid connecting a **Tieline** codec to a PBX, PABX, Key station, business system or any other local switchboard. Easily said, the tricky bit is working out if you are on one of these systems.

As a general guide, these devices have two characteristics:

a) Requires the dialing of an additional digit to access the PSTN[9].
b) Have more buttons and features than your average space shuttle.

WARNING: Many of these systems are digital and have non standard telephone line operating voltages. If you plug a digital PABX/PBX line into a **Tieline** codec, permanent damage may result from the high voltage pulses these systems generate. Even if the PBX is not digital, the performance of the codec is unlikely to be as good as a normal POTS/PSTN line.

If you have no option than using a PBX/PABX, search for a fax machine. Thankfully, the overwhelming majority of fax machines are designed for analogue POTS line operation and are normally on an extension optimized for fax machines and data transmission. Substitute a normal phone for the fax machine and verify correct operation. Use a normal phone, not a house supplied phone as the house phone may have characteristics to match the existing PBX/PABX, not a POTS line. After confirming correct phone operation unplug the phone and attach the phone line to the codec.

8.6. Line Checks

8.6.1. Length of the line from your site to the Central Office or Local Exchange

It is desirable to have a local loop (the line from your location to the local CO or Exchange) as short as possible. Optimum performance can be expected for lines up to about 2 miles (3 kilometers) in length. Lengths in excess of this can be expected to perform at lower bit rates and line quality depending on the age, condition and type of cabling. E.g. plastic insulation, paper insulation, water or moisture entering the cable, age and state of repair of joins.

8.6.2. Presence of stub or party lines on your local loop

In some countries, it was the practice to have more than one phone service attached to one line - sometimes called a Party Line. In this situation, as more lines are installed, services are separated but the redundant cabling may remain connected across the line. This scenario will cause problems with the operation of your **Tieline** codec.

[9] PSTN Public Switched Telephone Network

As an example, a POTS service has been provided to a customer a long way from the CO, this customer has since moved on. The phone company needs to provide a phone service in the same area, but closer to the CO. Let us assume half way. The service is provided, however the phone company simply jumpers the new service onto the old (further) service. The cable from the jumpering point to the old service point is redundant and will act as a stub, loading the line down. This will not matter for a phone service. However it will affect the performance of a **Tieline** codec connected to the line.

A common variation of this problem is in high rise buildings. Most high rise buildings have a MDF[10] in the basement and a series of smaller IDFs[11] on the floors. If a line was originally used on an upper floor, it would have a connection from the CO to the MDF and then from the MDF to IDF for the appropriate floor. Over time the service could be redeployed to a lower floor. However, for a quick installation the phone company may not disconnect the upper floor IDF, they could just bridge into the lower floor IDF leaving the upper floor IDF still connected. The unused stub from the lower to upper floors could then cause problems.

8.7. Earth Leakage Problems on the Line

A good line should have an earth isolation of better than ten meg-ohms. If your line is located in an area where water is a problem, ask your Telco to check out the earth leakage.

8.8. Equipment Problems at the CO or Local Exchange

Although there are many factors at the Telco end that can cause problems, a problem that does occasionally occur is that the clock on the interface codec to your line is not synchronized to the network. A drifting clock will cause instability and unreliable codec performance. If you suspect that this could be the problem, contact your local Telco.

The **Tieline** family of codecs is designed to achieve solid and reliable operation with a line quality of 30% or better. On most good POTS lines, **Tieline** ***G3*** codecs will normally achieve 28,800 bps and a line quality of approximately 50% or greater. If you are not able to achieve this level of operation, you may have transmission problems with your line or the line at the other end of the connection. Using the **Tieline** test facility on 317 913 6911 (USA) will assist you to find where the problem lies.

[10] MDF Main Distribution Frame.
[11] IDF Intermediate Distribution Frame

Warning on Digital Phone Systems: Digital phone systems typically run off voltages greater than the 50 volts used by the PSTN. Some older ISDN systems run on approx 100 volts. Connection of a **Tieline** POTS codec to a voltage greater than the normal 50 volts will cause damage to a POTS codec and void the warranty.

Handy Hint: To find out if you are about to connect a **Tieline** codec to a digital line, use your normal analog phone to check the line first.

WARNING for Installation of Modules:
If you are installing or uninstalling any module, **please ensure that the codec is not powered up** during the installation process. We recommend you use anti-static precautions to help minimize the chances of static charges damaging the highly sensitive circuitry. Do not force a module into the codec. Modules should be installed slowly and gently.

8.9. Tips for Successful Operation

Following are a few simple and handy hints that could save you a lot of blood, sweat and tears when you are setting up your codec at a remote site:

1. It can be a good idea to take your own phone when you are doing a remote broadcast. If possible, dial the number you are going to use and check for any unusual noises. These may be due to other devices connected to the line.

2. **Tieline** USA has a test unit on 317 913 6911 to facilitate testing. A **Tieline** ***G3*** codec will automatically negotiate the optimal connect rate for each individual circuit on connection

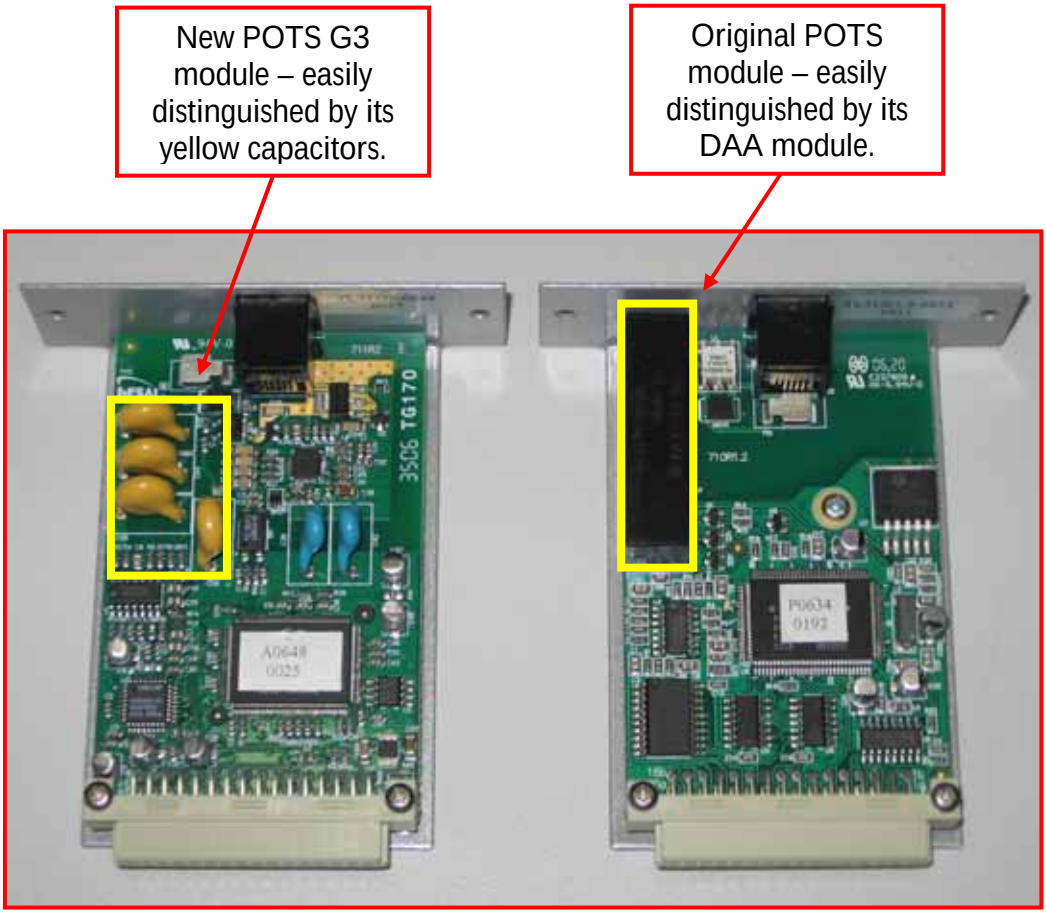
3. Some facilities, especially large hotels, are able to provide phone connection points to allow a dial-up connection for lap top Computers. Lap top Computers use technology similar to **Tieline** POTS codecs, so a **Tieline** codec will invariably work if the computer connection speed is 24,000 bps or higher.

4. Take an ADSL/DSL filter to all remote locations. ADSL/DSL modems can generate noise on a line which will degrade the performance of your codec. Due to the increasing popularity of ADSL/DSL services on phone lines, this is likely to be an increasing problem, but one that is simple to solve. Simply place the ADSL/DSL filter between the POTS line and your codec to remove the interference.

All this obviously takes time, often requiring assistance from hotel staff, resident IT professional etc. If you have no other option than connecting your codec to a PBX/PABX, **Tieline** recommends that you confirm that the line works BEFORE you try to do a live program.

8.10. POTS versus new POTS G3 Modules

Tieline has a new POTS module called a POTS G3 module. The new module is distinguishable from the old one because the original DAA module has been removed – as displayed in the following image.



The modules are also named 'POTS' and 'POTS G3' on the front of the module, to distinguish one module from the other. This is displayed in the following images.

Figure 18: Original POTS Module

Figure 19: New POTS G3 Module

Recommendation for Bonded Mono & Bonded Stereo Connections

Where possible we recommend that you use the same type of POTS modules in the dialing codec when creating bonded connections between two codecs (i.e. bonded mono and bonded stereo connections). It is possible that bonded connections may not bond reliably when dialing bonded connections and using different POTS modules. To circumvent this possibility use the same type of POTS modules in the dialing codec. If you must use different types of POTS modules, always dial the first connection of a bonded connection from a POTS G3 module. (Please note: The on-board POTS hardware on an ***i*-Mix *G3*** codec is the same as a POTS module, not a POTS G3 module. Dial this connection second for bonded connections).

8.11. Stability of Tieline POTS Modules

Both **Tieline** POTS modules create stable connections because they measure the amount of fluctuation in line quality before working out the optimal bit-rate setting to connect at. This provides very stable POTS connections.

8.11.1. Features of the POTS G3 Module

DAAs are required for codecs, and any other devices, that connect to POTS/PSTN networks. The new **Tieline** POTS G3 module contains a SmartDAA™ [12] (Smart Data Access Arrangement), which is different to the original **Tieline** POTS module DAA.

DAAs are line interfaces that isolate electronic devices from voltages on phone lines. The new SmartDAA is more tolerant of variations in line voltage. In addition, when you select the country that you are dialing from in your codec, the SmartDAA automatically adjusts for the line voltage present in each particular country.

Important note:

Please ensure that the *Country Selection* in ***Unit Options*** is set correctly for your POTS and POTS G3 modules to ensure the correct settings are programmed into the codec before dialling connections.

[12] SmartDAA is a trademark, patent pending, of Conexant Systems, Inc in the United States and/or other countries.

Recommendation

When possible we recommend that you use the same pair of modules when connecting between two codecs. This is not always possible, so please note the following differences in dialing with different module combinations.

8.11.2. POTS G3 Module Dialing Screen

The new POTS G3 module dials a little differently to the original POTS module. The dialing process explained in this section will occur when dialing from a POTS G3 module to a POTS module, or a new POTS G3 module.

When dialing is initiated from the codec the screen displays as shown in the following image.

```
ManDflt MonoPgm
 POTS1^ Enter#
⌈POTS2> Connctng 59⌉
Aud            Menu
```

Figure 20: New POTS G3 Module Connecting

The codec dials out and displays a countdown timer and the word 'Connecting' until a call has been established and retraining and renegotiation has been completed. Once this process is completed the standard connection screen is displayed.

```
ManDflt MonoPgm
 POTS1^ Enter#
⌈POTS2> 28.8 L91R89⌉
Aud            Menu
```

Figure 21: Standard POTS Connection Screen

This is a little different to dialing with the original **Tieline** POTS module. When dialing with this module the codec displays intermediate connection states such as answering, negotiating and stabilizing in real-time for each connection, as well as Rx and Tx numbers - as shown in the right-hand image.

```
ManDflt MonoPgm
 POTS1^ Enter#
[POTS2> Tx300:Rx307]
Aud            Menu
```

Very Important Notes:

Quick Negotiation is not necessary with the new POTS G3 module and it has been removed from the POTS G3 module menus.

In addition, the Auto and Up options have been removed from the Reneg (Fast) POTS G3 module menus.

8.11.3. Dialing from a POTS G3 Module to a POTS Module

When dialing from the new POTS G3 module to an old POTS module, the codec will attempt to connect initially at 19,200 bps. If line quality is above 80% at this bit rate then the codec will 'retrain' the connection up to a

maximum of 28,800bps (depending on modem handshaking). The codec will then renegotiate the link downwards to the highest possible bit rate where line quality is greater than 70%.

8.11.4. Dialing from a POTS G3 Module to a POTS G3 Module

When initially connecting to another POTS G3 module, **Tieline** POTS G3 modules will initially attempt to establish a connection at 28,800 bps. The codec will then renegotiate the link downwards to the highest possible bit rate where line quality is greater than 70%.

Very Important Renegotiation Note:
After initially connecting, if you attempt to shift the connection bit-rate **upwards locally** with a POTS G3 module, the codec will **'retrain' upwards, not renegotiate** like the original POTS modules. Retraining takes significantly longer than renegotiation – up to 30 seconds.

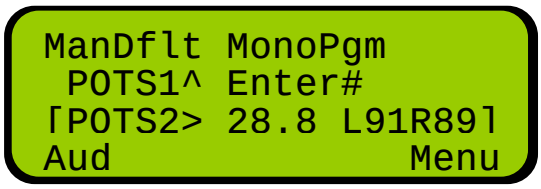
Connection Tip:
If you are connecting a POTS module to a new POTS G3 module, use the POTS module in the local codec. This will allow you to renegotiate the connection upwards seamlessly if required, without the delay of retraining.

8.11.5. Dialing from a POTS Module to a POTS or POTS G3 Module

Tieline POTS modules initially attempt to establish a link at 19,200 bps. They will then renegotiate the link to the highest possible bit rate, either up or down.

The codec modem can also be set for Quick Negotiation in the main codec POTS menu, which will more quickly negotiate a connection at the codec bit rate setting within this menu.

Once the connection process has completed, the main connection screen will be displayed. The bit-rate of the connection and the Line Quality of both codecs will be displayed.



8.12. Making POTS Voice Calls

The ***i*-Mix *G3*** is capable of making and accepting voice calls via the on-board phone coupler. Now it is possible to make voice calls using a **Tieline** POTS module with a ***COMMANDER G3*** field or rack unit codec.

Important Note:
All POTS G3 modules are capable of making voice calls. Original POTS modules are also capable of making voice calls if they are version 1.2 modules. Most original modules are version "710R1.2" modules but some are version 1.0. This can be checked by looking at the version number on the module itself, as indicated in the following images.

Figure 22: Voice-Capable Original POTS Module 710R1.2

Figure 23: Original POTS Module 700R1 without Voice Capability

8.13. Why Make POTS Voice Calls?

There are several reasons for using a POTS plug-in module in voice mode and they include:

- To make a standard voice call from a **Tieline** codec;
- To integrate a voice call into program content; or
- To dial into a hybrid telephone from a **Tieline** codec.

Sometimes it may be necessary to make a voice call because there is no **Tieline** codec at one end of a connection. At other times it may be necessary to integrate a voice call into a broadcast, or you may simply want to make a voice phone call for communications. Remember, voice calls are only 3 kHz in bandwidth.

8.14. Selecting POTS Voice Mode

Programming POTS voice mode can be done in one of two ways – either using the codec menus or using ToolBox software.

To program via the codec menus there are two alternatives. The easiest way is to use the POTS wizard. Select the POTS connection that you wish to connect with using the MENU SELECTOR. Next, select SOFTKEY 3 Wiz then press OK and scroll to and select Setup Voice Analog. Press OK to complete the configuration.

Please note: The other way to configure a plug-in module for voice mode is to select **[Menu] > [Configuration] > [POTS Left/Right Setup] > [Operating Mode] > [Voice Analog]**.

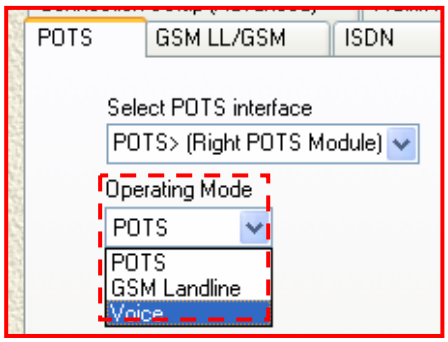
The connection should display on the screen as shown in the following image.

```
ManDflt MonoPgm
[POTSV1< Enter#    ]

Aud Profile Wiz Menu
```

Note that the connection is called *POTSV* and in this example we are looking at the LCD menu screen of a ***COMMANDER G3*** rack unit codec.

It is also possible to program GSM voice mode using the ***POTS*** tab in ToolBox software. To do this use the *Operating Mode* drop-down menu and select *Voice Analog*.



Important Configuration Note: This setting is shared by the *Operating Mode* menu in the *GSM Landline* section of the ***GSM LL/GSM/USB-3G*** tab. In other words, both sections named *Operating Mode* in these tabs change simultaneously when either one is changed.

8.14.1. Default Settings for POTS Voice Calls

By default, a POTS voice call is sent to a codec's encoders as per the connection profile loaded into the codec. This is a very useful way of integrating voice calls into a broadcast and sending the composite program signal to the studio encoded. The program matrix in this situation is as displayed in the following image.

	Encoder 1	Encoder 2	CH 1 Out	CH 2 Out	HP1 L	HP1 R	HP2 L	HP2 R	Phone Out	Auxilary	PPM 1	PPM 2
Input 1	✓				✓	✓	✓	✓	✓	✓	✓	
Input 2	✓				✓	✓	✓	✓	✓	✓	✓	
Aux	✓				✓	✓	✓	✓	✓		✓	
Phone	✓				✓	✓	✓	✓		✓	✓	
Decoder 1			✓	✓	✓	✓	✓	✓				✓
Decoder 2												

Figure 24: Phone Audio Routed to Encoder by Default

In this situation audio is not sent to the analog codec outputs. Incoming audio is sent to the *Encoder/s* and can be monitored via the *Mix (Send/Return) Pan Matrix.*

As displayed in the following image, phone audio by default is routed to the *Send* section of the matrix – along with the other inputs.

Mix (Send/Return) Pan Matrix

	Send	Return
Input 1	✓	
Input 2	✓	
Aux	✓	
Phone	✓	
Decoder 1		✓
Decoder 2		✓

The BALANCE knobs on **Tieline** codecs control the balance between the SEND & RETURN audio sent to the codec headphone outputs. The *Mix (Send/Return) Pan Matrix* provides the flexibility of selecting which sources are allocated to SEND and RETURN.

The audio balance between the *Decoder* audio monitored (via *Return*) and the inputs (via *Send*) can be controlled via the BALANCE knob on a **Tieline** codec.

Important Note: This control does not affect the level of the transmitted or received program audio. It only controls the monitoring of these sources in the headphones.

8.15. Routing POTS Voice Calls to a Codec's Analog Outputs

It is possible to route POTS and GSM voice calls to a codec's analog outputs by pressing SOFTKEY 1 Audio and selecting **[Route PhIn to CHOut] > [Enable]**. The default setting is Disable.

This can also be configured using ToolBox software via the *Inputs* tab, as displayed in the following image.

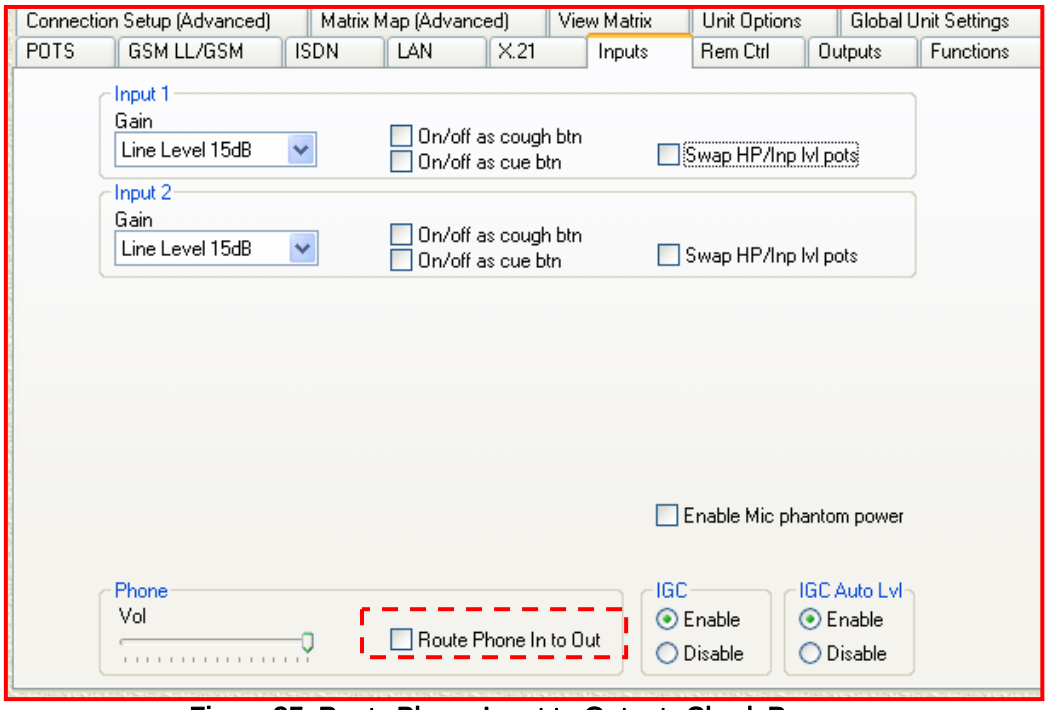


Figure 25: Route Phone Input to Outputs Check-Box

Once this has been configured, analog audio is routed to the analog outputs instead of the *Encoder/s* – allowing this audio to be used as an analog program, foldback or IFB feed.

In this situation audio is deselected from the *Encoders* and is instead sent to analog *CH1 & 2 Out.* The program matrix effectively looks like the one displayed in the following image when the *Route Phone In to Out* check-box has a tick in it.

	Encoder 1	Encoder 2	CH 1 Out	CH 2 Out	HP1 L	HP1 R	HP2 L	HP2 R	Phone Out	Auxilary	PPM 1	PPM 2
Input 1	✓				✓	✓	✓	✓	✓	✓	✓	
Input 2	✓				✓	✓	✓	✓	✓	✓	✓	
Aux	✓				✓	✓	✓	✓	✓		✓	
Phone			✓	✓	✓	✓	✓	✓		✓	✓	
Decoder 1			✓	✓	✓	✓	✓	✓				✓
Decoder 2												

Figure 26: Audio Routed to Analog Outputs

8.15.1. Monitoring when POTS Voice is Routed to Analog Outputs

When *Route Phone In to Out* is enabled, audio is sent to the *Return* section of the *Mix (Send/Return) Pan Matrix.* As a result, the *Decoders* and the POTS voice (*Phone*) audio is monitored via *Return* and the other inputs can be controlled with the BALANCE knob via *Send*.

Mix (Send/Return) Pan Matrix

	Send	Return
Input 1	✓	
Input 2	✓	
Aux	✓	
Phone		✓
Decoder 1		✓
Decoder 2		✓

8.16. POTS Summary:

It is always preferable to obtain a dedicated POTS phone line from the Local Exchange or Central Office, nothing is better than keeping it simple. The simpler the installation the less there is to go wrong, and the quicker faults can be rectified. It is common sense, not rocket science.

If you have any doubt about a line being a digital or analogue line, plug your own normal phone into the line first. If the phone works correctly, the line is a POTS line. If the phone does not work correctly, something is not quite right and further investigation is required.

Support Tip: The **Tieline** USA POTS test number 317 913 6911, and the codec's Line Quality Display is there to help you test your codec. If you need any assistance in setting up your codec, you can contact us at support@tieline.com

Section 9. ISDN

The acronym ISDN stands for Integrated Services Digital Network. The Basic Rate Interface (BRI) of ISDN consists of 2 bearer (B) channels at 64 Kbps each and 1 data (D) channel at 16 Kbps, i.e. (2B +D). This can be provided over a 2 wire facility and the two B channels can be bonded together to form a single 128Kbps channel. The B channel can carry user information such as voice, video or data. The D channel carries signaling information between a user and the network.

Tieline *G3* codecs can provide high bandwidth stereo audio over a single B channel using the **Tieline** *Music* algorithm. If you have 2 B channels you can use one as a standby, or operate in higher bandwidth stereo or mono/IFB mode using a variety of algorithms. This can be set with ToolBox under the *Connection Setup* page within the *Profile Editor*.

9.1. Important Considerations

There are a number of things you will need to consider if you are using your codec in ISDN mode. Some of these things include:

- Will you be using a single B channel or 2 B channels?
- Which network will you be using?
- Is your ISDN line Point-to-Point or Point-to-Multipoint?
- What is your Service Profile ID (SPID) and what are your Directory Numbers (DN1, DN2)?
- What is your Multiple Subscriber Number (MSN)?

Many of the answers to these questions will be influenced by the country you are operating in. For example, a SPID does not need to be entered into a **Tieline** codec for operation within Europe - but it does in the USA. A detailed explanation of all the points listed above, and how the codec should be configured for ISDN operation, is discussed in a logical sequence in the ToolBox *ISDN* section of this manual.

9.2. What ISDN module do I need?

Tieline Technology has two different ISDN Modules available. The module that you need depends on whether you have a NT-1 device connected to the line. Most North American installations do not have an NT-1.

In the U.S., the telephone company provides its BRI customers with a U interface. The U interface is a two-wire (single pair) interface from the phone switch, the same physical interface provided for POTS lines. It supports full-duplex data transfer over a single pair of wires, therefore only a single device can be connected to a U interface.

In North America, the Customer Equipment (CE) is usually connected directly to the 'U' Interface.

The situation is different in Europe, the UK, most of Asia, Australia, Africa and parts of the Middle East where the phone company is allowed to supply the NT-1, and thereby the customer is given an S/T interface.

The NT-1 is a relatively simple device that converts the 2-wire U interface into the 4-wire S/T interface. The S/T interface supports multiple devices (up to 7 devices can be placed on the S/T bus) because, while it is still a full-duplex interface, there is now a pair of wires for receive data, and another for transmit data.

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

In Japan, you will need the **Tieline** S/T ISDN module.

North America 'U' interface		**European 'S' interface**	
Model	**Tieline Part Number**	**Model**	**Tieline Part Number**
Commander Field	TLISDNUS	Commander Field	TLISDNEU
i-Mix	TLISDNUS	i-Mix	TLISDNEU

Table 6: ISDN Modules

9.3. How do I install the ISDN module?

You can purchase a **Tieline** codec with the ISDN module already installed. If you already own a **Tieline** codec, upgrading your codec for ISDN use is an easy process – simply purchase the ISDN module for your region, put it into the codec, load the profile you require and you are ready to connect.

WARNING for Installation of Modules:

If you are installing or uninstalling any module, **please ensure that the codec is not powered up** during the installation process. If possible use anti-static precautions to help minimize the chances of static charges damaging the highly sensitive circuitry. Do not force a module into the codec. Modules should be installed slowly and gently.

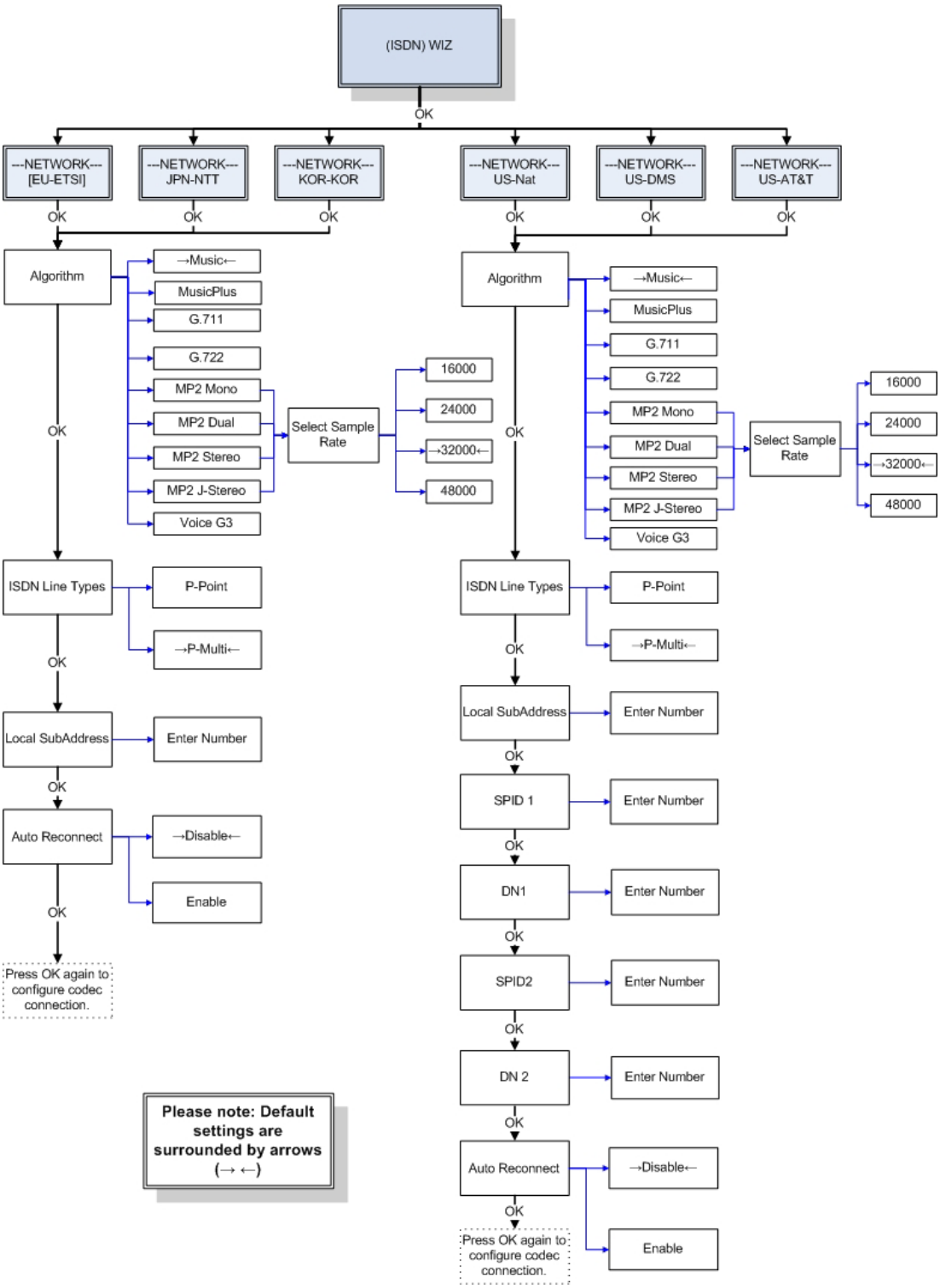
If you are having trouble connecting, please see the ISDN Troubleshooting section in this reference manual. If you do this and find you are still having difficulties, contact **Tieline** Technology to find out more, or to locate your nearest support and upgrade office. We have support and upgrade offices in the USA, England, Netherlands, Singapore and Australia.

9.4. Connecting over ISDN

The following information is intended to provide users with the necessary tools to make successful ISDN calls on **Tieline** codecs under a variety of ISDN line conditions

9.4.1. Important Settings

First, it is important to know what Wizard settings are available to configure ISDN. The ISDN Wizard menu structure is displayed in the following image.



The full and expanded codec ISDN menu settings are located at **[Menu] > [Configuration] > [ISDN< Left Setup]** (or ISDN> Right Setup, depending on the slot a module has been inserted into).

9.4.2. Factory Default Settings

These are indicated on the codec by arrows either side of the setting, e.g. →P-Point←. To restore Factory Default Settings select **[Menu] > [Reset Functions] > [Set Sys Defaults] > [All Profiles] > [Yes]** and wait 10 seconds. Reboot the codec via **[Menu] > [Reset Functions] > [Reboot System] > [Yes]**

9.4.3. ISDN Line Types

The ISDN Line Types, 'P-Point' (Point-to-Point) & 'P-Multi' (Point-to-Multipoint), are designated by the network service provider and may vary between venues and locations.

Another ISDN setting is 'On Demand'. This may be designated 'On' or 'Off' by the network service provider and is also discussed further below

9.4.4. Session Data

By default, when **Tieline** codecs establish communications the dialing codec sends its settings to the remote codec over **Tieline** Session Data. The receiving codec inherits the following settings from the dialing codec:

- Profile;
- Algorithm;
- Sample Rate; and
- Bit rate.

This does not occur when dialing to non-**Tieline** devices so 'Session Data' must be turned off to provide compatibility. To disable 'Session Data' navigate to: [**Menu] > [Configuration] > [System Settings] > [Session Data] > [Disable].** Note: Remember to re-enable 'Session Data' when dialing to another **Tieline** codec.

9.4.5. Configuration Profile

'ManDflt MonoPgm' (default) should be displayed at the top of the main menu screen on the LCD display. If not press SOFTKEY 2 Profile and using the MENU SELECTOR scroll through the available options until ManDflt MonoPgm is surrounded by square braces. Press the Enter/Dial keypad button to choose your selection and then press SOFTKEY 2 Yes to confirm.

9.4.6. Audio Input Levels

Line Level 15db is the default input gain for inputs. If you require an alternate input gain press SOFTKEY 1 Audio then select **[Input Gains] > [All Inputs] > [*select desired input level*]**

9.4.7. Configuring ISDN

ISDN can be programmed using the wizard by pressing SOFTKEY 3 Wiz and then pressing SOFTKEY 4 OK to proceed. Selection is made by scrolling with the MENU SELECTOR and pressing SOFTKEY 4 OK to accept the setting and to progress to the next menu option. Typical ISDN settings are:

- Network → EU-ETSI
- Algorithm → G.722
- ISDN Line Type → P-Point or P-Multi as required

Once all settings are chosen ISDN Configured! is displayed. Pressing SOFTKEY 4 OK again will return to the main menu

9.4.8. Testing the ISDN Line Type

With an ISDN line connected to the codec ISDN Sync should be displayed. This appears regardless of whether you have the 'ISDN Line Type' set correctly. To test whether you have the correct 'ISDN Line Type' selected simply dial your own number.

- Rotate the MENU SELECTOR until 'ISDN1.....' is surrounded by square braces I.e. **[ISDN1< Enter# Sync]**
- Enter the number to be dialed using the numeric keypad
- Press the Enter/Dial button to dial the connection.

This will perform a loopback test by dialing out on the main ISDN number and receiving on the auxiliary ISDN number

If this results in a 'Disconnect: 1' message being displayed, change the 'ISDN Line Type' to the other option. For example, if 'P-Point' was selected change to 'P-Multi'. Redialing your own number again will result in the following screen with a mix of the inputs received on the 'De 02' (Decoder 2) PPM.

9.4.9. On Demand

If Sync appears for approximately 60 seconds upon initial connection of the ISDN line and then disappears, or if Sync does not appear and you know you are connected to an active ISDN interface, then it may have 'On Demand' enabled by the network service provider. To test this, dial a codec on a known working ISDN line.

Rotate the MENU SELECTOR until ISDN1 is surrounded by square braces, i.e. **[ISDN1< Enter#].**

- Enter the number using the numeric keypad.
- Press the Enter/Dial button to dial the connection.
- If Sync now appears beside the second ISDN connection this indicates the service has now been activated.
- Immediately disconnect the call by pressing the Hang Up button then SOFTKEY 4 Yes to confirm the hang up.
- Dial again.
- If this dial is successful 'On Demand' is enabled. We recommend you contact your network service provider to have 'On Demand' disabled where possible to alleviate future issues.

9.4.10. Connecting Tieline to Tieline

1. Rotate the MENU SELECTOR until 'ISDN1' is surrounded by square braces

2. Enter the number to be dialed using the numeric keypad

3. Press the Enter/Dial button to dial the connection

4. During the framing process the algorithm will flash, i.e. G.722 and remain solid once the framing process is complete.

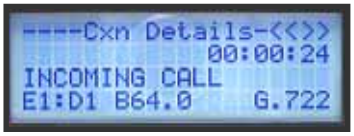
5. To bond two 'B' channels (connect at 128kb) scroll until ISDN 2 is surrounded by square braces and proceed by following previous steps 2-4.

6. When fully connected at 128kb both ISDN 'B' channels will be connected with the algorithms stable (not flashing).

7. Pressing the Enter/Dial button during a connection will display the Cnx Details screen. This provides:

   - Call status.
   - Call duration.
   - Direction of call.
   - Assigned Encoders and Decoders.
   - Bit rate for a particular 'B' Channel.
   - Algorithm.

8. To disconnect a call:
   - Press the Hang Up button;
   - Select SOFTKEY 2 Yes to disconnect the call; or
   - Select SOFTKEY 3 ALL to hang up both 'B' Channels.



9.4.11. Connecting Tieline to AEQ/Telos

To dial from a **Tieline** codec to a non-**Tieline** codec it is necessary to disable 'Session Data' and use either G.722 or MP Layer 2 algorithms for compatibility. The following settings must be manually set on both ends:

- Profile: Mono/Stereo.
- Algorithm.
- Sample Rate.
- Other settings as required by the non-**Tieline** codec.

Note: For a full description of operation please refer to the specific Codec Reference Manual available for download from: http://www.tieline.com/support/manuals.html.

Section 10. GSM

GSM networks deliver wireless high quality and secure mobile voice and data. Today's highly successful GSM wireless technology has become the world's leading and fastest growing mobile standard, spanning over 190 countries.

Figure 27: Tieline GSM Module and Antenna

Cell phone networks provide a voice channel and a data channel. The **Tieline** GSM data facility connects over the data channel and is therefore relying on the Telco's commitment to wireless data integrity, bit rate allocation and link reliability.

VERY IMPORTANT WARNING:

GSM plug-in modules cannot be used in TLR300B ***COMMANDER G3*** rack unit codecs. If you must connect using GSM with aTLR300B ***COMMANDER G3*** rack unit codec, we suggest you use a GSM cell-phone attached to the RS232 port on the rear panel of the codec. Alternatively, you could try a GSM phone or a 3G cell-phone in GSM mode with a **Tieline** USB module.

10.1. How does it work?

GSM uses a process called Circuit Switching. Circuit-switching is a method of communicating where a communications path is established between two devices through one or more switching nodes. The digital data is sent as a continuous stream of bits, allowing the data to be heard from the receiving end whilst the data is still being sent. This makes circuit switching an ideal method of sending real time audio. This is different from GPRS which sends packets of information which need to be reassembled before they can be heard and is therefore not suitable for real time audio broadcast applications. Circuit Switched Data (CSD) enables single channel data at 9,600bps.

10.2. HSCSD Information courtesy of GSMworld.com

HSCSD (High speed circuit switched data) is a real time delivery protocol compared with GPRS which is a non-real-time packet based technology. HSCSD allows users to send and receive data at speeds of 28.8kbit/sec and 43.2kbit/sec by enabling up to 4 traffic channels of a GSM network.

HSCSD is only available in a few countries. Some of these are the UK, Germany, Switzerland, Singapore and Canada. Go to www.gsmworld.com to see if HSCSD is available in your region.

10.3. What do I need for GSM?

To use a **Tieline** codec in GSM mode it requires a GSM plug-in module with an antenna connector, which is available from **Tieline** as an additional module option.

Only one GSM module can be used in a codec. You will need to remove one of the modules if you have two installed in a ***COMMANDER G3*** codec (the ***i*-Mix *G3*** only has one module slot available). If you wish to use two GSM connections you will also need to use the RS 232 connection attached to the rear of the codec and connect a GSM-compatible cell-phone (with an in-built GSM data modem) such as the *Nokia*® 6310 and 6310i. You will also need a serial data cable to connect between the RS232 connection and your cell-phone.

WARNING for Installation of Modules

If you are installing or uninstalling any module, **please ensure that the codec is not powered up** during the installation process. If possible use anti-static precautions to help minimize the chances of static charges damaging the highly sensitive circuitry. Do not force a module into the codec. Modules should be installed slowly and gently.

A Cautionary Note: Some service providers package voice and data as a standard subscription and others require you to enable "data calls" from your mobile separately. Please check with your service provider to ensure they have enabled data calls for your service and your phone has data calling enabled.

10.4. Default GSM Connection Settings

10.4.1. CSD Default Settings

CSD connections provide up to 7kHz audio connections and the default settings use the Voice G3 algorithm to connect at a bit rate of 9,600.

10.4.2. HSCSD Connections

HSCSD connections provide up to 15 kHz audio connections and the default settings use the Music algorithm to connect at a bit rate of 28800.

If you are having trouble connecting at this bit rate, change the bit rate and algorithm at both codecs to 14400 and Voice G3 respectively. Try to reconnect with these new settings. If you are still having trouble connecting it could be that you don't have an HSCSD connection available. If this is the case you could try to connect in CSD mode, i.e. select GSM CSD as the wireless network and use the Voice G3 algorithm to connect at a bit rate of 9,600. (Related topic: GSM Troubleshooting)

10.5. Configuring GSM Connections Using a GSM Module

If you are dialing out from a codec via a GSM module, then you will need to configure the GSM codec connection. The easiest way to do this is to use the codec GSM menu wizard.

10.5.1. Recommended Settings

To configure the GSM module, find the connection on the codec LCD screen using the MENU SELECTOR. Then press SOFTKEY 3 Wiz to activate the menu wizard. Select OK in the wizard to proceed. This will lead you through the configuration steps required to get connected.

10.5.1.1. CSD Connections

For CSD connections select GSM CSD as the Wireless Network in the wizard. Connect using the default settings which use the Voice G3 algorithm to connect at a bit rate of 9,600. Ensure the following settings in the wizard are selected:

[GSM Wizard] > [Wireless Network →GSM CSD←] > [Algorithm →Voice G3←] > [Pre-buffer Secs →0←] > [Sig Str Enable →Enable←] > [Auto Reconnect →Disable←]

10.5.1.2. HSCSD Connections

If you are connecting via an HSCSD connection (up to 15kHz), the GSM bit rate and codec algorithm need to be set to 28800bps and *Music* respectively. When you select an HSCSD connection during the configuration process these settings become the default. To configure HSCSD settings, go to the main codec menu by pressing SOFTKEY 3 Wiz. Ensure the following settings in the wizard are selected:

[Configuration] > [GSM Setup →GSM Module 1←] > [Wireless Network →GSM HSCSD←] > [Algorithm →Music→] > [GSM Bitrate →28800←] > [Pre-Buffer Secs →0←] > [Sig Str Enable →Enable←] > [Auto Reconnect →Disable←].

10.6. Configuring GSM Connections via the RS 232 Serial Port

If you are connecting via the RS232 port on the rear of the codec then you will need to use the RS232 connection on the rear of a codec and connect a GSM-compatible cell-phone such as the *Nokia*® 6310 or 6310i. You will also require a *Nokia*® (Type: DAN-1) null-modem DB9 male to DB9 male connector, between the cell-phone and the codec RS232 port. You will also need to configure the codec serial port for GSM use.

10.6.1. Recommended Settings

To display a GSM serial port connection on the codec LCD and to configure the RS232 port for this connection, connect your phone to the RS232 port

on the rear of the codec. Then press SOFTKEY 4 Menu, scroll to **[Configuration] > [Advanced Menu] > [Serial Port Mode] > [GSM].** This configures the serial port and the GSM connection should appear on the main menu screen as a connection option. If it doesn't appear on the main menu screen, power the codec down and then power it up and it should appear.

The serial port mode can also be set in ToolBox software via the *Serial Port Mode* drop-down menu in the ***Global Unit Settings*** tab.

10.6.1.1. CSD Connections

Now configure the CSD GSM connection by using the codec GSM menu wizard. This is done in a similar way to configuring GSM module connections. For CSD connections select GSM CSD as the Wireless Network in the wizard. Connect using the default settings which use the Voice G3 algorithm to connect at a bit rate of 9,600.

First select the RS 232 GSM connection on the codec LCD screen using the MENU SELECTOR. Then press SOFTKEY 3 Wiz to activate the menu wizard. Select OK in the wizard to proceed and then ensure the following wizard settings are selected:

[GSM Wizard] > [Wireless Network →GSM CSD←] > [Algorithm →Voice G3←] > [Pre-buffer Secs →0←] > [Sig Str Enable →Enable←] > [Auto Reconnect →Disable←]

10.6.1.2. HSCSD Connections

If you are connecting via an HSCSD connection, you may wish to change other settings such as the GSM Bitrate and Algorithm. To configure HSCSD connection settings, access the main codec menu by pressing SOFTKEY 3 Wiz. Select OK in the wizard to proceed and then ensure the following wizard settings are selected:

[Configuration] > [GSM Setup →Ext CellPhone←] > [Wireless Network →GSM HSCSD←] > [Algorithm →Music→] > [GSM Bitrate →28800←] > [Pre-Buffer Secs →0←] > [Sig Str Enable →Enable←] > [Auto Reconnect →Disable←].

For more information see the ***GSM LL/GSM/USB-3G*** section in this manual).

Please Note:
Initially set the Pre-Buffer Secs for GSM connections to 0 and adjust the setting if required. (Related topics: GSM Pre Buf Seconds and GSM Landline Pre Buf Seconds).

If your codec is captioning enabled, you will also see Transcriptor TX and Transcriptor RX as options in the algorithm menu.

For more information on these settings see the *GSM LL/GSM/USB-3G* section in this manual or the Quick Start connection procedure for GSM connections.

10.7. Configuring a POTS Landline Codec for GSM Connections

If you are dialing a GSM-connected codec via a POTS landline connection, then you need configure the POTS-connected codec specifically for this purpose. A POTS landline-connected codec must be in GSM Landline Mode to receive a GSM call.

10.7.1. Recommended Settings

The easiest way to do this is to use the POTS connection wizard. To enable GSM landline-mode via the wizard, use the MENU SELECTOR to select the POTS connection on the codec LCD screen that you are connecting with. Next press SOFTKEY 3 Wiz to use the connection wizard. Select OK in the wizard to proceed and then ensure the following wizard settings are selected:

[Pots Wizard →Setup GSM Landline←] > [Algorithm →Voice G3←] > [GSM Landline Rate →9600←] > [Pre-buffer Secs →0←] > [Auto Reconnect →Disable←].

Please Note:

Initially set the Pre-Buffer Secs for GSM connections to 0 and adjust the setting if required. (Related topics: GSM Pre Buf Seconds and GSM Landline Pre Buf Seconds).

Use the Voice G3 algorithm if you are connecting in CSD mode and select either Voice G3 or Music if you are connecting via HSCSD (depending on what bit rate you are able to achieve over your connection). Use Music for 28,800 bps connections).

Set Auto Reconnect to Enable if you want a codec to redial a connection if it drops out. With this enabled a codec can automatically redial until a link is re-established. See the 'Auto Reconnect' section in this reference manual for more info.

If you are having trouble connecting, please see the GSM Troubleshooting section in this reference manual. If you do this and find you are still having difficulties, contact **Tieline** Technology to find out more, or to locate your nearest support and upgrade office. We have support and upgrade offices in the USA, England, Netherlands, Singapore and Australia.

10.7.2. GSM Transparent and Non-transparent Data Modes

If you are using a GSM connection, we recommend you ensure that your SIM card has been set to send data in non-transparent mode. Check this with your telco when you order your data service. (Some cell-phone networks only provide non-transparent data capabilities). When transparent mode is used, the bit error rate could be as high as 1/1000. This will lead to dropouts in audio.

When data is sent in transparent mode you require sustained data flow to transfer data reliably. This is network dependent and if the network is unreliable there may be some loss of data packets, more bit errors (1/1000) and therefore dropouts in audio.

In non-transparent data mode, Radio Link Protocol (RTP) is used, providing better error correction and less bit errors (1/100,000,000). Data is buffered when it is being sent and received from a GSM module or cell-phone in this mode. This mode is more reliable for a data call and **Tieline** GSM modules only support the sending of non-transparent data. We also recommend that you use non-transparent data mode when you use a *Nokia*® 6310i cell-phone.

If your SIM card is set to transparent mode you will be unable to receive any calls to your cell-phone or GSM module. Contact your telco to check if your SIM card has been set to non-transparent data mode and get them to switch it if it hasn't.

10.8. Making GSM Voice Calls

The ***i*-Mix *G3*** and the ***COMMANDER G3*** field unit codecs are capable of making both data and voice calls via **Tieline** GSM plug-in modules. This section specifically deals with analog GSM call capabilities using this module.

Very Important Note: A GSM voice call from a **Tieline** codec can only be made via a GSM plug-in module.

10.9. Why make GSM Voice Calls?

There are several reasons why it could be useful to use a GSM plug-in module in voice mode and they include:

- To make a voice call from a **Tieline** codec;
- To integrate a voice call into program content;
- To dial into a hybrid telephone from a **Tieline** codec;
- In some areas data calls may not be available and a voice call may be the only option (this is common if you are 'roaming' with the phone in different areas); or
- If you are using the GSM phone connection for communications, a voice call may be preferable because it has much lower delay.
- In some cases network congestion over a GSM data call may cause the codec latency to become several seconds. For bi-directional conversations

this becomes difficult to work with. Switching to GSM voice will eliminate the codec latency problem.

A GSM voice call is similar in quality to standard POTS/PSTN voice calls - in other words a 3 kHz connection. It can dial into a studio hybrid or another **Tieline** codec which is set to POTS Voice Mode.

10.10. Selecting GSM Voice Mode

Programming GSM voice mode can be done in one of two ways – either using the codec menus or using Tool**Box** software.

To program via the codec menus there are two alternatives. The easiest way is to use the GSM wizard. Select the GSM connection that you wish to connect with using the MENU SELECTOR. Next, select SOFTKEY 3 Wiz then press OK and scroll to and select **[GSM VOICE (Analog)] > [Signal Strength Enable →Enable←] > [Auto Reconnect →Disable←]** and press OK to complete the configuration.

Please note: The other way to configure a plug-in module for voice mode is to select **[Menu] > [Configuration] > [GSM Setup] > [GSM Module1] > [Wireless Network →GSM VOICE (Analog)←]**.

After a few seconds the connection should display on the screen as shown in the following image.

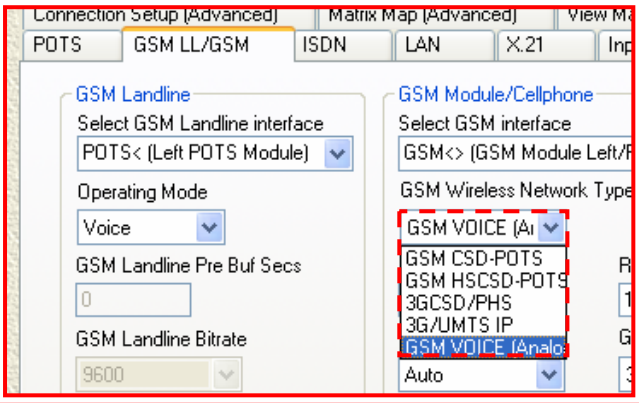
```
ManDflt MonoPgm
⌈GSMV1< Enter# Ss=6⌉

Aud Profile Wiz Menu
```

Note that the connection displayed is called *GSMV* and in this example we are looking at the LCD menu screen of a ***COMMANDER G3*** field unit codec (before PPM metering is displayed).

You can now connect your GSM voice enabled codec from the field to the studio. If you choose to dial another **Tieline** codec in the studio it must be set to POTS Voice mode.

It is also possible to program GSM voice mode using the ***GSM LL/GSM/USB-3G*** menu in Tool**Box** software. To do this use the *GSM Wireless Network Type* drop-down menu and select *GSM VOICE (Analog).*



10.10.1. Default Settings for GSM Voice Calls

By default, a voice call is sent to a codec's encoders as per the connection profile loaded into the codec. This is a very useful way of integrating voice calls into a broadcast and sending the composite program signal to the studio encoded. The program matrix in this situation is as displayed in the following image.

	Encoder 1	Encoder 2	CH 1 Out	CH 2 Out	HP1 L	HP1 R	HP2 L	HP2 R	Phone Out	Auxilary	PPM 1	PPM 2
Input 1	✓				✓	✓	✓	✓	✓	✓	✓	
Input 2	✓				✓	✓	✓	✓	✓	✓	✓	
Aux	✓				✓	✓	✓	✓	✓		✓	
Phone	✓				✓	✓	✓	✓		✓	✓	
Decoder 1			✓	✓	✓	✓	✓	✓				✓
Decoder 2												

Figure 28: Phone Audio Routed to Encoder

As displayed in the right-hand image, phone audio by default is routed to the *Send* section of the matrix.

Mix (Send/Return) Pan Matrix

	Send	Return
Input 1	✓	
Input 2	✓	
Aux	✓	
Phone	✓	
Decoder 1		✓
Decoder 2		✓

You may recall that the BALANCE knobs on **Tieline** codecs control the balance between the SEND & RETURN audio sent to the codec headphone outputs. The *Mix (Send/Return) Pan Matrix* provides the flexibility of selecting which sources are allocated to SEND and RETURN.

The audio balance between the *Decoder* audio monitored (via *Return*) and the inputs (via *Send*) can be controlled via the BALANCE knob on a **Tieline** codec.

Important Note: This control does not affect the level of the transmitted or received program audio. It only controls the monitoring of these sources in the headphones.

10.11. Routing GSM Voice Calls to a Codec's Analog Outputs

It is possible to route GSM voice calls to a codec's analog outputs by pressing SOFTKEY 1 Audio and selecting **[Route PhIn to CHOut] > [Enable]**. The default setting is Disable.

This can also be configured using ToolBox software via the *Inputs* tab, as displayed in the following image.

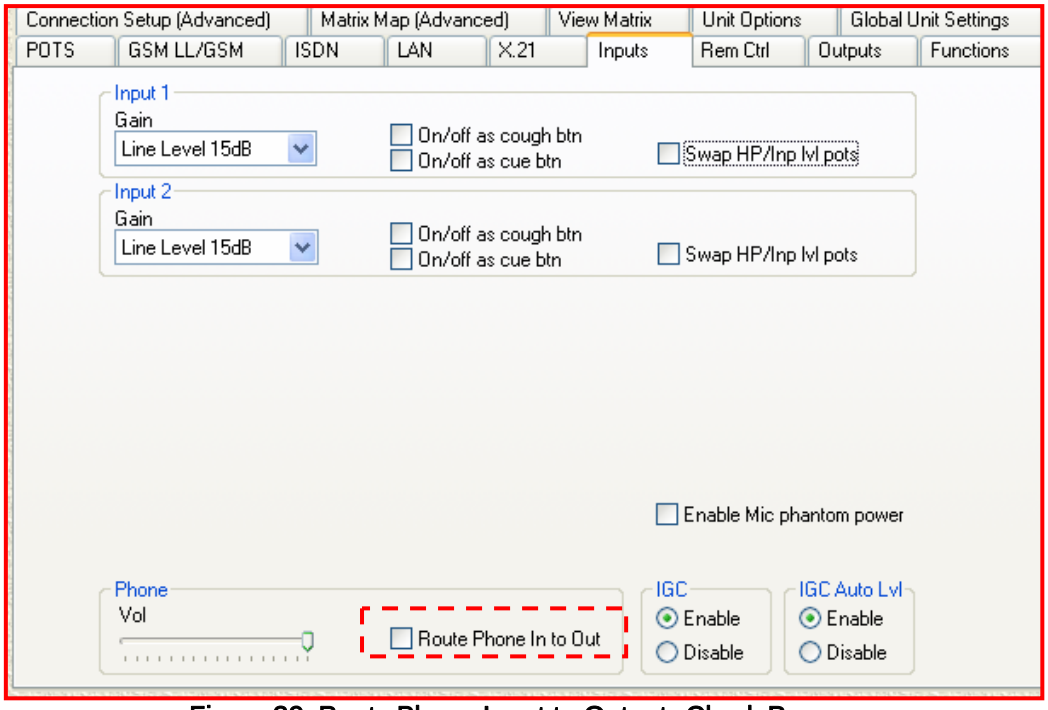


Figure 29: Route Phone Input to Outputs Check-Box

Once this has been configured, analog audio is routed to the analog outputs instead of the *Encoder/s* – allowing this audio to be used as an analog program, foldback or IFB feed.

In this situation audio is deselected from the *Encoders* and is instead sent to analog *CH1 & 2 Out.* The program matrix effectively looks like the one displayed in the following image when the *Route Phone In to Out* check-box has a tick in it.

	Encoder 1	Encoder 2	CH 1 Out	CH 2 Out	HP1 L	HP1 R	HP2 L	HP2 R	Phone Out	Auxilary	PPM 1	PPM 2
Input 1	✓				✓	✓	✓	✓	✓	✓	✓	
Input 2	✓				✓	✓	✓	✓	✓	✓	✓	
Aux	✓				✓	✓	✓	✓	✓		✓	
Phone			✓	✓	✓	✓	✓	✓		✓	✓	
Decoder 1			✓	✓	✓	✓	✓	✓				✓
Decoder 2												

Figure 30: Audio Routed to Analog Outputs

10.11.1. Monitoring when GSM Voice is Routed to Analog Outputs

When *Route Phone In to Out* is enabled, audio is sent to the *Return* section of the *Mix (Send/Return) Pan Matrix.* As a result, the *Decoders* and the GSM voice (*Phone*) audio is monitored via *Return* and the other inputs can be controlled with the BALANCE knob via *Send*.

Mix (Send/Return) Pan Matrix

	Send	Return
Input 1	✓	
Input 2	✓	
Aux	✓	
Phone		✓
Decoder 1		✓
Decoder 2		✓

10.12. A Final Note on GSM Configuration

For more detailed information on configuring your codec and using ToolBox, please see the ***GSM LL/ GSM*** section in this reference manual.

Operational Tip: If you want total wireless freedom consider the **Tieline** codec battery kit which can deliver up to 2.5 hours of broadcasting. For more information please see the Portable Powering Solutions section within this manual.

Figure 31: The Tieline Portable Solutions Range

Section 11. 3GIP

The sections on IP, 3GIP and SIP IP connections are contained in the "IP & 3GIP Streaming Reference Manual", which is a separate document. This manual contains all the relevant information for streaming audio over IP – including SIP connectivity.

Contact **Tieline** support at support@tieline.com or visit our website at http://www.tieline.com/support/manuals.html to download the latest manual.

Section 12. Satellite Operation

(Information courtesy of Xantic at www.xantic.net)

Your **Tieline** codec can also be used on the Inmarsat GAN (Global Area Network). Using a satellite connection you can take advantage of the ultimate in mobility for your codec requirements. GAN is an Inmarsat-based service which provides global coverage, targeted at major land and sea locations internationally.

Satellite operation of your codec is similar in setup to ISDN configurations. The GAN ISDN service allows data connection rates of up to 64 Kbits/s in full duplex or up to 56 Kbits/s when converted to American and Japanese standards. Universal Serial Bus (USB) interface and ETSI-standard ISDN connectivity, allow you to easily set up your codec for satellite use.

12.1. Broadband Global Area Network (BGAN)

BGAN is a global high-speed mobile satellite data network. It is capable of sending broadband data at bit rates up to 492 kbps. Guaranteed bandwidth rates are offered and this network can be used over the Internet as well as for voice, SMS and ISDN communications.

The additional bandwidth of this service allows the use of multiple devices – providing a flexible mobile solution for many remote broadcasting situations. **Tieline** codecs can interface easily with BGAN to provide high quality audio broadcast solutions.

For more details on BGAN and other satellite operations, please visit www.xantic.net

WARNING for Installation of Modules

If you are installing or uninstalling any module, **please ensure that the codec is not powered up** during the installation process. If possible use anti-static precautions to help minimize the chances of static charges damaging the highly sensitive circuitry. Do not force a module into the codec. Modules should be installed slowly and gently.

Section 13. X.21

X.21 is a protocol that runs over subscriber networks (i.e. leased line connections). X.21 can operate as a full duplex connection with data rates from between 9,600 bps to 2 megabits.

X.21 is an ITU-T data interface standard for connecting a DTE to a DCE and transferring synchronous serial data. The properties of X.21 define the physical, electrical and call control procedures for interfacing using this standard. The basic interface signals performed in most X.21 signalling include transmit, receive, control, indicate and timing. X.21 connections use balanced data, clock and control signal lines. X.21 connections have been fully integrated into **Tieline** codecs.

64 – 384 kbps connections are most common and **Tieline** codecs support X.21 leased line mode connections from 64 – 2,048 kbps. Ensure that the bit rate setting in your codec is matched to that of your ISDN TA/X.21 DCE device. If you are unsure of the bit rate of your ISDN TA/X.21 DCE device, please check its user manual to discover the correct setting.

The physical interface to a **Tieline** codec is via a DB15 connector (ISO 4903). The pin-outs for D-15 X.21 module connections are displayed in the Connector Wiring section of this reference manual. A straight-through cable is required from the male D15 connector on the X.21 module to a female DB15 connector, which will attach to an X.21 DCE Device (such as an ISDN Terminal Adaptor/TA). The connection profiles supported are mono, stereo, dual mono and mono/IFB.

Once a call has been made successfully between two codecs that are programmed to use the X.21 protocol, any data can be transferred over the connection. In **Tieline** codecs, choose from *G.711, G.722,* MPEG 2 Layer 2 (*MP2)* and **Tieline** *Music* Algorithms (**Tieline** *Music* is capable of 15 kHz mono or stereo over 64kbps or 128kbps with 70ms delay) as standard.

13.1. How Do X.21 Modules Connect to the Network?

X.21 modules use a D-15 connector to connect to leased line networks. The X.21 module has a male connector and it connects to a female version of the same connector.

13.2. How does it Work?

First you will require a **Tieline** X.21 module in each of your codecs. Next, you will need to ensure that you have correctly connected the codecs to your leased line network. A diagram follows with an example of how this might look.

Figure 32: X.21 ISDN Interface Connection

13.3. Connecting Via X.21

First install the X.21 module into your codec. The connection should then appear in the root menu of the main codec LCD screen. For the procedure to connect over X.21, please see the Quick Start section of this reference manual. This will display the menu wizard for X.21 module connections and explain the Quick Start connection procedure to get you connected.

13.3.1. X.21 over Leased Lines

With C/DTR call establishment, a 'C' (Control Signal) is used to enable and disable call control protocols. An 'I Signal' (incoming call signal) is used to indicate the presence of an incoming call.

When dialing from one codec to another, the dialing codec (DTE1) changes the C Signal from off to on to initiate an outbound call. At the codec (DTE2) receiving the call an 'I Signal' ('Indication' or incoming call signal) changes from off to on, indicating the presence of an incoming call. The codec receiving the call then initiates a C Signal state change, from off to on, and replies to the dialing codec, which receives the incoming signal and changes its I Signal state from off to on. This completes the call connection process over X.21.

To clear a call, press HANG UP on either codec to change the C Signal from on to off. This will terminate the connection between the codecs.

For more information on how to connect via X.21 using **Tieline** codecs, please see the X.21 or Quick Start section of this reference manual, or contact customer support at support@tieline.com for more info.

Section 14. IP Streaming Configurations

The sections on IP, 3GIP and SIP IP connections are contained in the "IP & 3GIP Streaming Reference Manual", which is a separate document. This manual contains all the relevant information for streaming audio over IP – including SIP connectivity.

Contact **Tieline** support at support@tieline.com or visit our website at http://www.tieline.com/support/manuals.html to download the latest manual.

Section 15. Operation of Your Codec: Codec LCD Displays and Dialing Connections

The objective of this section is to get you started without using Tool**Box** software. You can complete most of the programming of your codec with the codec MENU SELECTOR (MS), SOFTKEYS and the codec KEYPAD. If you are the average skilled professional operator and the **Tieline** ***COMMANDER G3*** is your new toy, this section and the Quick Start section together provide enough information to get you up and running quickly.

This section provides information about how to navigate menus using the codec and it explains what the various connection state screens (POTS, ISDN, GSM, X.21, 3G and IP) look like at different stages of connection. In addition, the difference between manual and Connection Manager (Automatic) dialing is explained.

Important note: Any changes you make to a profile using the codec menus will be stored as the *Current Runtime* settings. These settings are lost when a new profile is loaded. To permanently edit and save any changes to profiles, or to create new profiles, you must use Tool**Box** software. For a more comprehensive and detailed description of *Current Runtime* and editing and saving profiles, please go to the Tool**Box** section of this reference manual.

15.1. Getting Started: Operating the Menu Selector (MS)

The ***COMMANDER G3*** is designed to be used and configured in conjunction with Tool**Box** software. The codec can however be used as a stand-alone unit and the menu settings can be altered using the MENU SELECTOR (MS).

You can rotate the MS to scroll up, down or sideways and navigate through the lists viewable on the LCD screen. Press the MS downwards to select an item in the screen which is highlighted in [BRACKETS]. This is similar in concept to using the enter button on a PC. Any menu item that has been selected will have two arrows on either side of it, i.e. → ←. This shows you what the current setting is. On many LCD menus you can use SOFTKEYS 2 & 3 to navigate and scroll through the menu selections. They are displayed as either << or >> on the LCD screen. For simplicity, in this manual the MS will only be described for navigation as it is able to navigate through every codec menu item and select it.

15.2. Clear

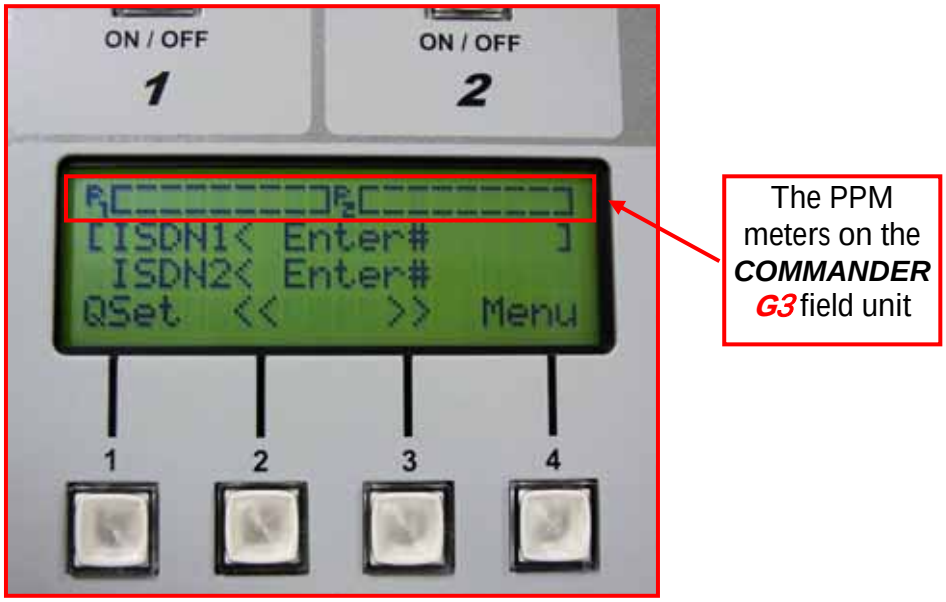
The CLEAR button is a general Escape (Esc) button like on a PC. Pressing the CLEAR button takes you back to the previous screen. Pressing the CLEAR button a second time will take you back a second level – similar to the ESC button on a computer.

15.3. Audio Monitoring On the Codecs

The ***COMMANDER G3*** field unit codec displays the levels of incoming and outgoing audio streams on the top row of the LCD.

When the main power-up LCD is displayed, the loaded profile on the field unit codec is displayed on the top row of the LCD. This is displayed for a few seconds each time you return to this screen.

Subsequently the PPM 1 and PPM2 meters at the top of the LCD are displayed. These meters are programmable via ToolBox software or via the Audio menu on the codec. They display as shown below before audio is metering on them. (Related Topic: PPM Displays)



15.4. Adjusting Input Levels

The audio levels of codec inputs can be adjusted locally via the Input knobs on the front panel of the codec. If further adjustment is required they can be adjusted by selecting SOFTKEY 1 Audio and then selecting **[Input Gains].** Select the input that needs adjusting individually, or select All Inputs to adjust both inputs at the same time.

15.5. Adjusting Auxiliary and Phone Input Levels

The incoming level of the auxiliary and phone inputs can be adjusted using the Audio menu via SOFTKEY 1.

15.5.1. Auxiliary Input Levels

To adjust the incoming level of the auxiliary input select SOFTKEY 1 Audio and then select **[Aux Vol/Ctrl] > [Disable Remote Control →Yes←].**

Use the MENU SELECTOR to adjust the audio gain up and down and use SOFTKEYS 1 & 2 to turn the channel on or off. The default setting is for the auxiliary input to be switched on and the default gain setting is 0%. The level of the input will need to be adjusted upwards to hear any source plugged into the auxiliary input.

Once the settings have been adjusted at the codec, press SOFTKEY 4 Exit and remote control functionality can be enabled again if required. This provides the opportunity to adjust the auxiliary input level remotely with the *Control* menu in ToolBox software. (Related topic: Control Tab).

Important Note on Aux Input Gain:
If you are using the *Aux* input on the codec and you want to adjust the gain of the input there are another two ways of doing it. You can use the Aux input fader displayed in the *Control* menu or use the *Gang to Knob* function in the *Rem Ctrl* menu with one of the INPUT digipots to control input levels.

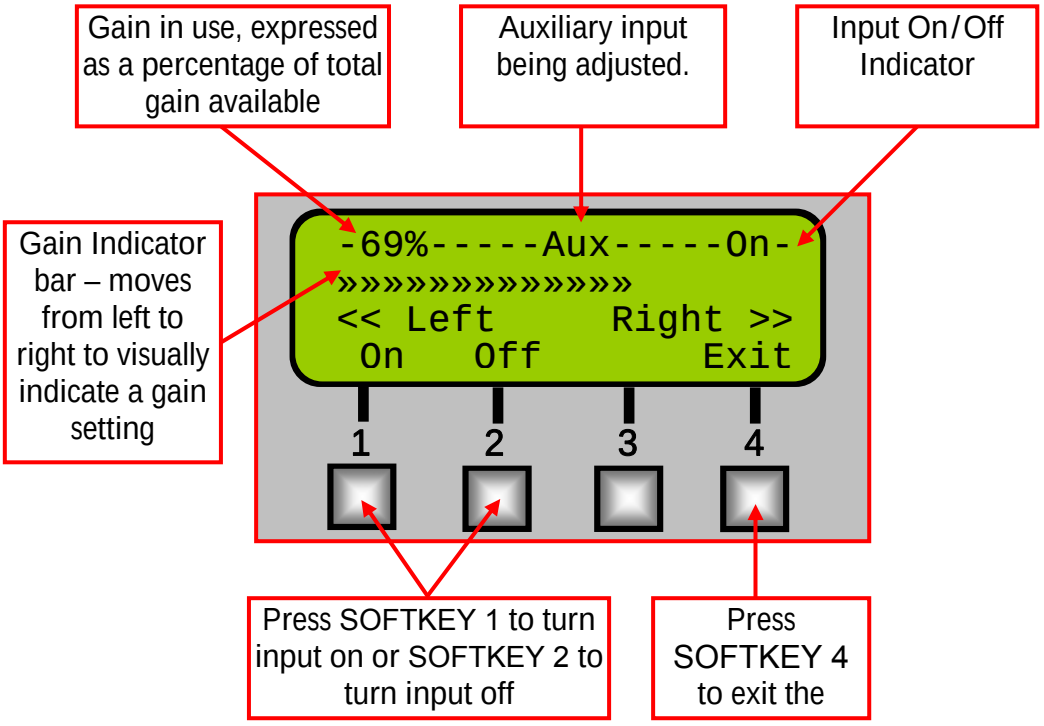


Figure 33: Auxiliary Input Level Control LCD Screen

15.5.2. Phone Input Levels

To adjust the incoming level of phone inputs (if two POTS voice calls are in use both phone inputs are adjusted up or down) select SOFTKEY 1 Audio and then select **[Phone Vol/Ctrl] > [Disable Remote Control →Yes←].**

Use the MENU SELECTOR to adjust the audio gain up and down and use SOFTKEYS 1 & 2 to turn the channel on or off. The default setting is for the phone input to be switched off and the default gain setting is 100%. The level of the input can be adjusted as required.

Once the settings have been adjusted at the codec, press SOFTKEY 4 Exit and remote control functionality can be enabled again if required. This provides the opportunity to adjust the phone input level remotely with the *Control* menu in ToolBox software.

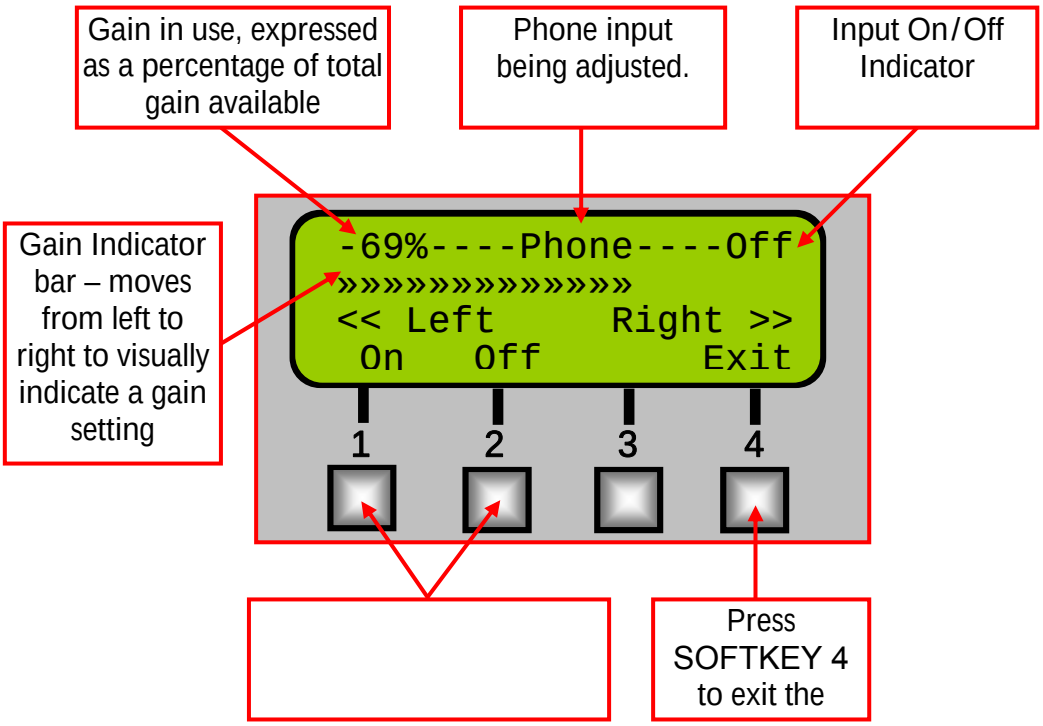


Figure 34: Phone Input Level Control LCD Screen

15.6. Pre-Flight Connection Checks

Before operation, try to verify that the following connections are made:

1. A POTS phone line, an ISDN line, an Ethernet connection, a GSM module, a 3G USB module, an X.21 module or satellite ISDN connection is connected to both the local and remote codecs.
2. An audio source (e.g. microphone, CD player oscillator etc) is connected and that audio levels have been set using either the Aud menu via SOFTKEY 1 on the codec or the ***Inputs*** menu within ToolBox.
3. Headphones are plugged in to check operation of the codec. While they are not essential, they are an excellent way to monitor performance of the codec.

15.7. Menu Navigation

Every **Tieline** codec uses 14 menus within the ***Profile Editor*** in ToolBox related to editing profiles. They are fully documented in the section in this manual titled ToolBox. Most of the elements within these profile menus can be amended into *Current Runtime* (*Current Runtime* stores non-permanent 'runtime' profile amendments) with the use of the MENU SELECTOR, KEYPAD and SOFTKEYS.

Preset user profiles and matrices for several codec operations are supplied with your **Tieline** codec. For more information in relation to these, please see the sections titled ***Manual Default Profiles*** and ***Matrix Editor*** within this manual. Matrices within profiles must be edited using ToolBox. (Related Topic: ***Profile Editor***)

Connection Tip: If you feel you need to check the audio, profile or connection settings for either Manual or Connections Manager connections (over POTS, ISDN, IP, X.21, 3G or GSM), please read through the Quick Start elements within this manual. In this section codec menu structures are displayed - including the Audio menu and all the wizard and remote control connection menus. For more detailed information see the relevant menu sections within the ToolBox section of this manual.

15.8. Getting Started

When you power up the codec the backlight on the LCD SCREEN (LCD) will be illuminated instantly, confirming that power is reaching the codec. Wait for about 15 seconds for the unit to start up and load the operating software and complete its pre-flight checks. During this time various screens will be displayed. These screens may include:

- The Codec Initialization screen;
- A screen displaying if the codec is in IP demo mode;
- Whether the software is a Beta release; and
- The main codec connection screen.

When the codec powers up it will always display the currently loaded profile at the top of the screen.

15.9. The LCD Screen on Initial Powerup

When your codec is delivered from the factory, the LCD after the initial start-up procedure will display the screen as shown[13]. You will also note that the manual default profile factory setting is Man Dflt Mono Pgm. The menu screen shown has only a left-hand POTS module connection displayed.

```
ManDflt MonoPgm
⌈POTS1< Enter#     ⌉

Aud Profile Wiz Menu
```

Depending on the communications module options that you have purchased from **Tieline**, and installed in your codec, the connections available to you will be displayed on the LCD.

15.10. Loading Profiles

Before you attempt to connect you must ensure you have the correct profile loaded into the codec. The currently loaded profile is displayed at the top of the main LCD screen. To change the codec profile press SOFTKEY 2 Profile. This will open the profile menu displaying all the profiles loaded into the codec. The right-hand image shows this menu displaying manual default profiles programmed into every **Tieline** codec.

```
----Load Profile----
⌈ManDflt MonoPgm   ⌉
 ManDflt Mono/IFB
 ManDflt Stereo
```

The currently selected profile will have two arrows on either side of it, i.e. →Manual Default Mono←. If you wish to change the profile scroll the rotate the MENU SELECTOR until the square brackets surround the profile you want to select and press it or press SOFTKEY 4 to select the profile. Then press SOFTKEY 2 to confirm the selection.

[13] Depending on what module you may have installed.

Very Important Connection Note:
Please ensure that you set both the local and remote codecs to operate using the same algorithm – otherwise your codecs will not operate correctly.

If you are using manual default profiles, when the dialing codec calls the remote codec, the dialing codec will automatically set both the profile and the default algorithm on the remote codec to match the dialing codec's setting.

In addition, it is very important to note that remote control of a codec is not possible unless you are using either the *Music, Music Plus* or *Voice G3* algorithms.

Important Profile Note: The profile on powering up can be set to your preference via **[Menu]** > **[Configuration]** > **[Advanced Menu]** > **[Powerup Profile].** This can also be configured via *Powerup Profile* within the ***Global Settings*** menu in ToolBox software.

Connection Tip: If a loaded profile has a connection or connections that are not supported by the current hardware (i.e. it is a profile set for ISDN connections but no ISDN modules are in the codec), the LCD won't display any connections but instead will display an error message indicating it is not supported – either adjust the profile settings or install the correct module if this occurs.

15.11. Initial Connection State of Codec LCD Displays

Following are the displays for each connection type as they should display initially on your codec.

15.11.1. POTS Connection States

When you turn your codec on, the POTS module will display as POTS1< Enter#. If a second POTS module is installed, this connection will display as POTS2> Enter#.

```
ManDflt MonoPgm
[POTS1< Enter#     ]
 POTS2> Enter#
Aud Profile Wiz Menu
```

15.11.2. ISDN Connection States

If you have an ISDN module installed, ISDN1> Unavailable and ISDN2> Unavailable will be displayed for a few seconds after the codec is powered up.

```
ManDflt MonoPgm
[ISDN1> Unavailable]
 ISDN2> Unavailable
Aud Profile Wiz Menu
```

It will then display the two B channels as ISDN1> Enter# and ISDN2> Enter#.

```
ManDflt MonoPgm
[ISDN1> Enter#     ]
 ISDN2> Enter#
Aud Profile Wiz Menu
```

Finally, if an ISDN line is attached to the codec and the ISDN card has been read, the Network has been detected and it is ready to connect, the two B channels connections will display as ISDN1> Enter# Sync and ISDN1> Enter# Sync.

```
ManDflt MonoPgm
[ISDN1> Enter# Sync]
 ISDN2> Enter# Sync
Aud Profile Wiz Menu
```

Important Note: ISDN connections will always appear as available on the codec because some networks are 'on-demand' and need to detect a valid network. These networks will not send you ISDN sync information until an ISDN connection is detected. If you try to dial an ISDN connection and the line is not connected, the codec will display No signaling data. Check wiring

15.11.3. GSM Connection States

The right-hand image displays how a codec screen will display both a GSM module connection and a GSM cell-phone connection – without SIM cards inserted, or before the connections are detected by the cell-phone network.

```
ManDflt MonoPgm
[GSMCS1>Unavailable]
 GSMCS2^Unavailable
Aud Profile Wiz Menu
```

Once a SIM card is inserted into both the module and the cell-phone, the connections will display as shown. The GSM signal strength, the Ss= value, will appear within 30 secs of inserting a SIM card, or after turning on a codec with a SIM card in a module or a cell-phone.

```
ManDflt MonoPgm
[GSMCS1>Enter# Ss=6]
 GSMCS2^Enter# Ss=7
Aud Profile Wiz Menu
```

15.11.4. IP Connection States

Please note: If IP functionality is not available on your codec menu screen, contact **Tieline** *to purchase an IP enable code for your codec. All customers can trial IP streaming for 45 days via* Menu > Configuration > Advanced Menu > IP Stream Demo > Enable.

If you don't have an Ethernet cable attached to the LAN connection of the codec the connections will be displayed as shown in the right hand image.

```
ManDflt MonoPgm
[IP1>   Unavailable]
 IP2>   Unavailable
Aud Profile Wiz Menu
```

Once you have an Ethernet cable attached to the LAN connection of the codec the connections will be displayed as shown in the right hand image. They are then ready to connect.

```
ManDflt MonoPgm
[IP1>   Enter#     ]
 IP2>   Enter#
Aud Profile Wiz Menu
```

Please note:
If you connect an Ethernet cable to the LAN port and the screen still displays IP1 Unavailable # and IP2 Unavailable #, you may not have a DHCP server which assigns IP addresses automatically. In this case, you may need to enter a static address using the IP configuration wizard. Please see your IT system administrator for more information.

15.11.5. 3G Connection States

Once a SIM card is inserted into a 3G cell-phone and it is switched on and has network coverage, plug it into a USB module. The right-hand screen is displayed initially.

Within approximately 30 seconds the right-hand codec screen should be displayed.

```
ManDflt MonoPgm
[3GIP1> Press Enter]

Aud Profile Wiz Menu
```

Press the ENTER/DIAL button to connect to the Internet Service Provider. Once connected, the screen will display as shown in the right-hand image. (Please note: If 3GIP1> Enter # is displayed, dial the 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.

```
ManDflt MonoPgm
[3GIP1>Cntd Goto IP]

Aud                Menu
```

To make the IP connection you must scroll now to the IP1 connection on the codec screen and it should display like a normal Ethernet-connected codec, as shown in the right-hand image. Simply use the keypad to add the IP address you wish to dial and connect to the other codec. This is just like connecting over any IP connection in terms of its functionality.

```
ManDflt MonoPgm
[IP1>   Enter#     ]
 IP2>   Enter#
Aud Profile Wiz Menu
```

15.11.6. X.21 Connection States

The right-hand image displays how a codec screen will display an X.21 connection when a codec has an X.21 module installed in a right-hand module slot.

```
ManDflt MonoPgm
[X21 1> Dial/Answer]

Aud Profile Wiz Menu
```

Once you press the ENTER/DIAL button to connect, a screen similar to that displayed will appear.

```
ManDflt MonoPgm
[X21 1> 128k Music ]

Aud Profile Wiz Menu
```

15.12. Initiating Manual Connections: Dialing a Number

This section describes how to dial over each of the various connections that are available with **Tieline** codecs – POTS, ISDN, GSM, IP, 3G/IP and X.21. For more detailed information in relation to connections over each type of connection, and using the various default profiles that are programmed into your

codec, please see the Quick Start section of this reference manual. It describes in detail how to connect successfully.

15.12.1. New POTS G3 Modules

Tieline has a new generation of POTS module called a POTS G3 module. This supersedes the previous model which was simply called a POTS module. Modules are clearly marked to delineate the two modules from each other. Dialling displays are slightly different when using each different module and the differences are outlined in the following sections.

For more information about the new POTS G3 module and its features, please see the section titled POTS versus new POTS G3 Modules in the POTS Operation section of this reference manual.

15.12.2. Manual POTS Connections

The manual default mono profile is loaded in the right-hand image. In this example we will attempt to connect with the connection displayed as POTS2> Enter#.

```
ManDflt MonoPgm
[POTS1< Enter#     ]
 POTS2> Enter#
Aud Profile Wiz Menu
```

If you are confident that the correct POTS profile with all the connection settings you want is loaded into your codec, follow these steps to connect.

1. Use the MS to scroll to the POTS2> Enter# connection you want on the codec startup LCD. To select the number you wish to dial, either:

    a. Press the MEMORY key to get a number from the phone book; or

    b. Press the ENTER/DIAL key to display the last dialed number for that connection; or

    c. Simply input the number for the destination codec using the keypad. It should display the telephone number as shown in the right-hand image.

```
ManDflt MonoPgm
 POTS1< Enter#
[POTS2> 55555555]
Aud Profile Wiz Menu
```

2. Press the ENTER/DIAL button on the keypad to make the call.

3. Once the connection process has completed, the connection screen will be displayed as shown - the bit rate of connection and the Line

```
ManDflt MonoPgm
[POTS1< Enter#     ]
[POTS2> 28.8 L91R89
Aud            Menu
```

Quality of both codecs will be displayed.

In the previous example we have connected using ManDflt MonoPgm profile settings. The codec has connected and the LCD is displaying the POTS2 bit rate (28,800 bps) as well as the line quality line quality of the local and remote POTS codec connections.

The first L91 figure refers to the line quality at the local codec. The R89 figure refers to the line quality at the remote codec we have dialed. These numbers would be reversed if viewed from the remote codec. The line quality is a dynamic display that will adjust constantly to reflect the prevailing line conditions at the local and the remote codecs.

When you hang up the connection, the codec LCD will display Disconnect.

4. If you were connecting two POTS lines for a different profile, i.e. *ManDflt Stereo*, you would simply repeat the procedure for the second POTS connection. (Related Topics: Manual Default Profiles, *Powerup Profile* and *POTS*)

Connection Tip: Scrolling to an active connection and pressing ENTER/DIAL will display the Connection Details for that connection.

15.12.3. POTS G3 Module Dialing Screen

The new POTS G3 module dials a little differently to the original POTS module. The dialing process explained in this section will occur when dialing from a POTS G3 module to a POTS module, or a new POTS G3 module.

When dialing is initiated from the codec the screen displays as shown in the following image.

```
ManDflt MonoPgm
 POTS1^ Enter#
⌈POTS2> Connctng 59⌉
Aud            Menu
```

Figure 35: New POTS G3 Module Connecting

The codec dials out and displays a countdown timer and the word 'Connecting' until a call has been established and retraining and renegotiation has been completed. Once this process is completed the standard connection screen is displayed.

```
ManDflt MonoPgm
 POTS1^ Enter#
[POTS2> 28.8 L91R89]
Aud              Menu
```

Figure 36: Standard POTS Connection Screen

This is a little different to dialing with the original **Tieline** POTS module. When dialing with this module the codec displays intermediate connection states such as answering, negotiating and stabilizing in real-time for each connection, as well as Rx and Tx numbers - as shown in the right-hand image.

```
ManDflt MonoPgm
 POTS1^ Enter#
[POTS2> Tx300:Rx307]
Aud              Menu
```

Very Important Notes:

Quick Negotiation is not necessary with the new POTS G3 module and it has been removed from the POTS G3 module menus.

In addition, the Auto and Up options have been removed from the Reneg (Fast) POTS G3 module menus.

15.12.4. Dialing from a POTS G3 Module to a POTS Module

When dialing from the new POTS G3 module to an old POTS module, the codec will attempt to connect initially at 19,200 bps. If line quality is above 80% at this bit rate then the codec will 'retrain' the connection up to a maximum of 28,800bps (depending on modem handshaking). The codec will then renegotiate the link downwards to the highest possible bit rate where line quality is greater than 70%.

15.12.5. Dialing from a POTS G3 Module to a POTS G3 Module

When initially connecting to another POTS G3 module, **Tieline** POTS G3 modules will initially attempt to establish a connection at 28,800 bps. The codec will then renegotiate the link downwards to the highest possible bit rate where line quality is greater than 70%.

Very Important Renegotiation Note:

After initially connecting, if you attempt to shift the connection bit-rate **upwards locally** with a POTS G3 module, the codec will **'retrain' upwards, not renegotiate** like the original POTS modules. Retraining takes significantly longer than renegotiation – up to 30 seconds.

Connection Tip:

If you are connecting a POTS module to a new POTS G3 module, use the POTS module in the local codec. This will allow you to renegotiate the connection upwards seamlessly if required, without the delay of retraining.

15.12.6. Dialing from a POTS Module to a POTS or POTS G3 Module

Tieline POTS modules initially attempt to establish a link at 19,200 bps. They will then renegotiate the link to the highest possible bit rate, either up or down.

The codec modem can also be set for Quick Negotiation in the main codec POTS menu, which will more quickly negotiate a connection at the codec bit rate setting within this menu.

Once the connection process has completed, the main connection screen will be displayed. The bit-rate of the connection and the Line Quality of both codecs will be displayed.

```
ManDflt MonoPgm
 POTS1^ Enter#
⌈POTS2> 28.8 L91R89⌉
Aud            Menu
```

15.12.7. Unsuccessful POTS Connections

If the RJ11 POTS line is not connected to the codec properly the following error message will be displayed on the LCD after you attempt to connect – No Dial Tone. If the two codecs can't connect then an error message will be displayed and you will have to attempt to reconnect again.

A Cautionary Note: The maximum number of digits and pauses in the dial string is 27.

15.13. Manual ISDN Connections

If you are confident that the ISDN profile with all the connection settings you want is loaded in your codec, then once you power-up the codec simply follow the next steps to connect.

```
ManDflt MonoPgm
⌈ISDN1> Enter#     ⌉
 ISDN2> Enter#
Aud Profile Wiz Menu
```

1. Use the MS to scroll to the ISDN connection you want on the codec startup LCD (ISDN1 or ISDN2 in the example above). To select the number you wish to dial either:

   a. Press the MEMORY key to get a number from the phone book, or

   b. Press the ENTER/DIAL key to display the last dialed number for that connection, or

   c. Simply dial the number for the destination codec using the keypad.

```
ManDflt MonoPgm
⌈ISDN1> 55555555   ⌉
 ISDN2> Enter# Sync
Aud Profile Wiz Menu
```

2. Press the ENTER/DIAL button on the keypad to make the call.

3. The connection you are using should then display Dialling… on the LCD. While the codec is dialing it will display as shown in the right-hand image.

```
ManDflt MonoPgm
⌈ISDN1> Dialling...⌉
 ISDN2> Enter# Sync
Aud              Menu
```

By default, a **Tieline** codec receiving a call will auto answer, or if this is not programmed, the operator will need to press ANSWER to connect the two codecs.

Once a call is answered the connection screen will display as shown in the right-hand image. In this example the connection is at 64k and the algorithm selected is Music. Only one channel of ISDN is connected.

```
ManDflt MonoPgm
⌈ISDN1> 64.0  Music⌉
 ISDN2> Enter# Sync
Aud              Menu
```

(Related topic: Manual Default Profiles, *Powerup Profile* and ***ISDN***)

15.13.1. Unsuccessful ISDN Connections

If you attempt to connect and your RJ45 ISDN line connection is not attached correctly to the codec, the following error message will be displayed on the codec LCD – No signaling data. Check wiring

Connection Tip: Scrolling to an active connection and pressing ENTER/DIAL will display the Connection Details for that connection.

15.14. Manual GSM Connections

If you are confident that the GSM profile with all the connection settings you want is loaded in your codec, then once you power-up the codec simply follow the next steps to connect. The right-hand image displays how a codec screen will display by default, for both a GSM module connection and a GSM cell-phone connection in CSD mode.

```
ManDflt MonoPgm
[GSMCS1>Enter# Ss=6]
 GSMCS2^Enter# Ss=7
Aud Profile Wiz Menu
```

1. Use the MS to scroll to the GSM connection you want on the codec startup LCD (GSMCS1 or GSMCS2 in the example above). To select the number you wish to dial either:

    a. Press the MEMORY key to get a number from the phone book, or

    b. Press the ENTER/DIAL key to display the last dialed number for that connection, or

    c. Simply dial the number for the destination codec using the keypad.

2. Press the ENTER/DIAL button on the keypad to make the call.

3. Initially the connection you are using will display Dialling… briefly before displaying Connecting, as displayed in the right-hand image.

```
ManDflt MonoPgm
[GSMCS1>Connecting.]
 GSMCS2^Connecting.
Aud Profile Wiz Menu
```

4. Once the connections have been established they will be displayed as shown in the right-hand image.

```
ManDflt MonoPgm
[GSMCS1> 9.6 VoiceG3
 GSMCS2^ 9.6 VoiceG3
Aud Profile Wiz Menu
```

15.15. Manual 3G Connections

If you are confident that the 3G profile with all the connection settings you want is loaded in your codec (for more information see the sections titled 3G Connection States or 3G Operation), then once you power-up the codec simply follow the next steps to connect. The right-hand image displays how a codec screen will display by for most 3G cell-phone connections in 3GIP mode. There is no need to enter a number to dial in this example. Simply press ENTER/DIAL to dial into your ISP/3G cell-phone network.

```
ManDflt MonoPgm
[3GIP1> Press Enter]

Aud Profile Wiz Menu
```

However, there are situations when it is necessary to enter the number of your cell-phone before dialing into your ISP/3G cell-phone network. If this number is required for dialing, when you select the network you are dialing over in the 3G configuration menu, an LCD screen similar to the right-hand one will be displayed. To dial in this situation:

```
ManDflt MonoPgm
[3GIP1> Enter#     ]

Aud Profile Wiz Menu
```

1. Use the MS to scroll to the 3GIP connection you want on the codec LCD. To select the number you wish to dial either:

    a. Press the MEMORY key to get a number from the phone book, or

    b. Press the ENTER/DIAL key to display the last dialed number for that connection, or

    c. Simply dial the number for the destination codec using the keypad.

2. Press the ENTER/DIAL button on the keypad to make the call.

3. Initially the connection you are using will display Connecting, as displayed in the right-hand image.

```
ManDflt MonoPgm
[3GIP1 >Connecting.]

Aud Profile Wiz Menu
```

4. Once a connection has been established it will be displayed as shown in the right-hand image.

```
ManDflt MonoPgm
[3GIP >Cntd Goto IP]

Aud                Menu
```

To connect to the other codec, scroll to IP1 on the LCD screen and connect as you would for any Ethernet-connected IP connection, as explained in the following section.

15.16. Manual IP Connections

If you are confident that the IP profile with all the connection settings you want is loaded in your codec, then once you power-up the codec simply follow the next steps to connect.

```
ManDflt MonoPgm
[IP1>   Enter#     ]
 IP2>   Enter#
Aud Profile Wiz Menu
```

1. Use the MS to scroll to the IP connection you want to use on the codec startup LCD (IP1 or IP2 in the example above). To select the number you wish to dial either:

   a. Press the MEMORY key to get an IP address from the phone book (in this case the IP address will display star characters * representing the periods in the address).

```
ManDflt MonoPgm
[    203*36*205*163]
 IP2>   Enter#
Aud Profile Wiz Menu
```

   b. Press the ENTER/DIAL key to display the last dialed IP address for that connection; or

   c. Simply dial the IP address for the destination codec using the keypad. (Note: Use the * or # button on the codec keypad to enter the periods (.) in the IP address).

```
ManDflt MonoPgm
[    203.36.205.163]
 IP2>   Enter#
Aud Profile Wiz Menu
```

2. Press the ENTER/DIAL button on the keypad to make the call.

3. The connection you are using should then display Connecting… on the LCD. While the codec is dialing it will display as shown in the right-hand image.

```
ManDflt MonoPgm
[IP1    Connecting.]
 IP2>   Enter#
Aud             Menu
```

4. Once you have connected to the codec you have dialed, the connection should display in a similar manner to the right-hand screen image.

```
ManDflt MonoPgm
[IP1    27.2 L95R93]
 IP2>   Enter#
Aud             Menu
```

15.16.1. Unsuccessful IP Connections

If you attempt to connect and the codec you are dialing is busy, i.e. already connected, the following error message will be displayed.

```
IP1    Disconnect: 0
Connection refused
```

15.17. Manual X.21 Connections

If you are confident that the X.21 profile with all the connection settings you want is loaded in your codec, then once you power-up the codec simply follow the next steps to connect.

```
ManDflt MonoPgm
[X21 1< Dial/Answer]
 X21 2> Dial/Answer
Aud Profile Wiz Menu
```

1. Use the MS to scroll to the X21 connection you want on the codec startup LCD (X.21 1< or X21 2> in the example above). There is no requirement to dial a number with X.21 connections. The connection will display Press Enter once an X.21 module has a cable attached to a connection from your ISDN TA/X.21 DCE Device.

2. Press the ENTER/DIAL button on the keypad to make the call. The connection you are using should then display Connecting on the LCD. While the codec is connecting it will display as shown in the right-hand image.

```
ManDflt MonoPgm
⌈X21 1< Connecting.⌉
 X21 2> Dial/Answer
Aud              Menu
```

   a. By default, a **Tieline** codec receiving a call will auto answer.

   b. Once a call is answered the connection screen will display as shown in the right-hand image. In this example the connection is at 64k and the algorithm selected is Music. Only one X.21 module is connected in this screen.

```
ManDflt MonoPgm
⌈X21 1< 64.0  Music⌉
 X21 2> Dial/Answer
Aud              Menu
```

   c. (Related topic: Manual Default Profiles, *Powerup Profile* and X.21 Connections)

15.18. The Connections Manager (CXNS)

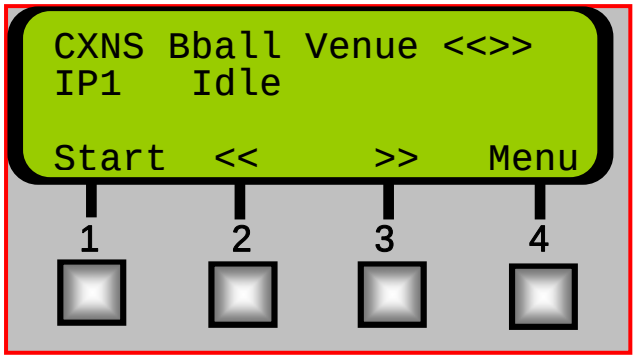
Tieline has developed the Connection Manager (CXNS) to automate the process of making a connection with **Tieline** codecs. The principle of what is explained here applies equally to POTS, ISDN and IP connections.

Tieline codecs have two modes of operation: Connection Manager (Automated and pre-programmed dialing) mode or Manual (manual dialing) mode. Previous discussions have described connecting in manual mode, i.e. dialing a number manually by using the codec keypad. In Connection Manager mode, all a user has to do is turn the codec on, wait for the menu on the LCD screen to light up and then select SOFTKEY 4, START from the CXNS (Connections Manager) screen – it's as simple as that.

Most of the time the codec Connection Manager will be used on a codec deployed at a broadcast venue, simplifying the connection process to the studio. In the following example we have created a custom profile called "Bball Venue" to facilitate using the Connection Manager. This is because a custom profile must be used when dialing via the Connection Manager. Why? Unlike the factory default profiles, custom profiles you create can be configured for a particular type of connection and you can program the number or the IP address to be dialed into the profile.

15.18.1. Operating the Connection Manager

On powering up, a **Tieline** codec will be either in Connection Manager CXNS mode (automated dialing) or manual mode depending on the profile loaded (the factory default mode of operation is manual connection mode). An example of an IP connection programmed to use the Connection Manager is displayed in the following image.

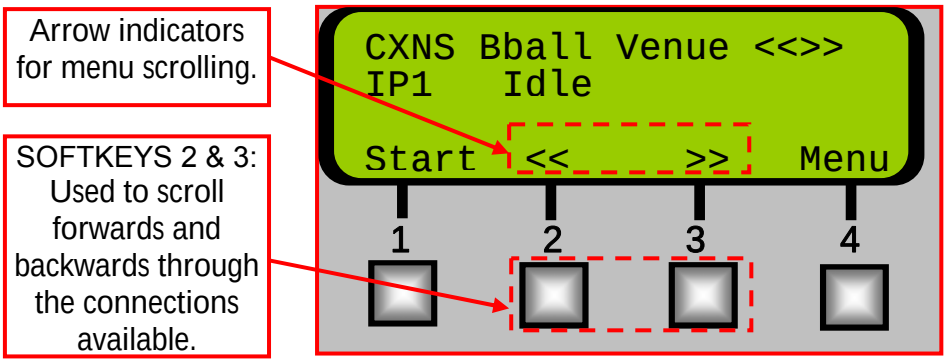


If you are not able to access the CXNS screen on powering up the codec, press SOFTKEY 4 to access the Menu submenu and select the Connection Mode submenu. Scroll to Auto and select this with the MS. The codec will then display a message telling you that you need to use a custom profile in Connection Manager mode. Simply click SOFTKEY 4 to exit this message and return to the main codec CXNS connection screen.

If you do not have a custom profile loaded in your codec, use Tool**Box** software to load a custom profile that you have saved in a configuration (.cdc) file onto the codec. Both the codecs that you are connecting with will require matching custom profiles. If you are not sure how to create profiles or how to use Tool**Box**, please consult the sections in this manual titled Tool**Box**, Configuration File System and Profile Editor.

15.18.2. Selecting Connections and Dialing Using the Connections Manager

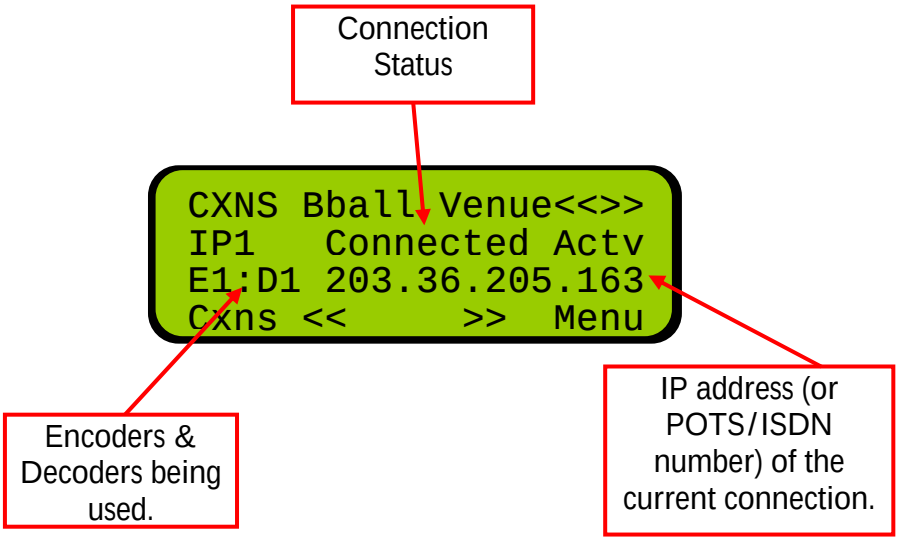
To scroll through the state of codec connections when you are viewing the main Cxns screen you can use SOFTKEYS 2 & 3. The LCD above these keys is displayed as << and >>. Use these SOFTKEYS to scroll forward and backwards through the connections available to select the one you want.



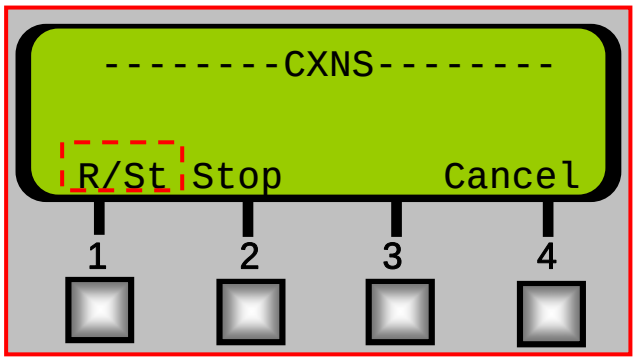
If you have more than one connection programmed into the profile you are using, (i.e. you have programmed a redundant failover connection to connect along-side your main connection - also programmed to be inactive unless failover is initiated) all programmed connections will be dialed by the Connection Manager when it is activated by pressing START.

15.18.3. The Display When Connected

If you connect successfully the connection should display as shown in the following image. The elements of the screen are described in detail.

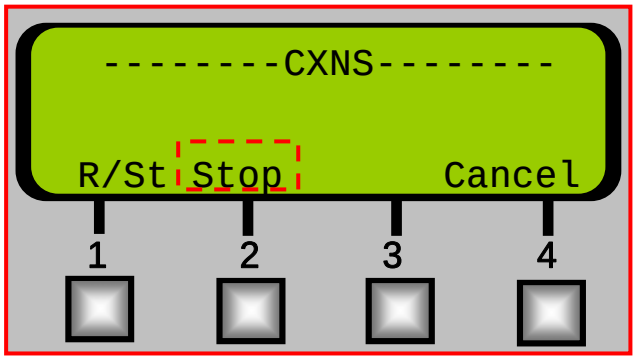


15.18.4. Reconnecting Using the Connections Manager



If you have connected using the Connections Manager and you lose the connection between the codecs for some reason, to reconnect simply press SOFTKEY 1 on the main Connections Manager LCD, which will have Cxns displayed above it. Then press R/St (Restart) using SOFTKEY 1 and the codec will attempt to reconnect.

15.18.5. Disconnecting Using the Connections Manager



If you wish to end a connection simply press SOFTKEY 1 on the main Connections Manager LCD which will have Cxns displayed above it. Then press SOFTKEY 2 which will have Stop displayed above it.

15.19. Summary:

The Quick Start and Operation of Your Codec sections of this manual are designed to allow users to connect successfully without using ToolBox software. Although complete codec menu structures and descriptions of their elements follow, more in-depth details of many codec functions are contained in the ToolBox software section of this manual.

15.20. Menu Settings

The Menu option on the startup LCD is just above SOFTKEY 4 on the codec. If you press SOFTKEY 4, the submenu for editing profiles is displayed. This submenu is more comprehensive than the wizard used to configure different connections. All of the wizard menu items are available and there are many other codec menus displayed that allow you to adjust different settings within the codec to suit your broadcast requirements.

```
ManDflt MonoPgm
⌈IP1>   Enter#     ⌉
 IP2>   Enter#  ----
Aud Profile Wiz Menu
```

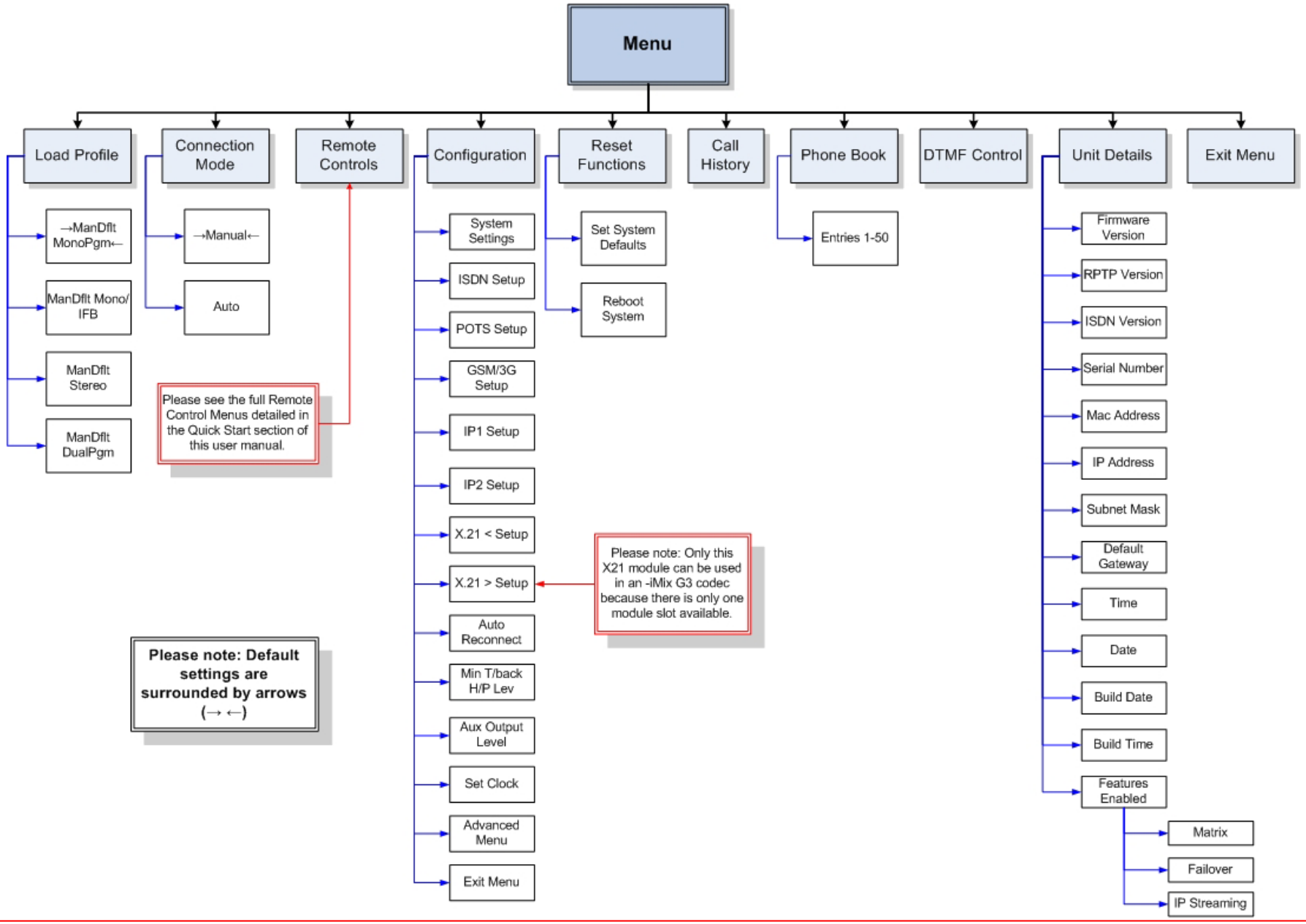
Following is a table defining the contents of each element of this profile setup submenu. These submenu categories are essentially the codec's in-built profile editor.

Submenu	Description
Load Profile	Allows you to load new profiles.
Connection Mode	Enables you to access either the Connection Manager CXNS or Manual connection.
Remote Controls	Allows you to select the inputs on the remote codec and control several functions with the local codec MENU SELECTOR i.e. turning channels on and off and input gains.
Configuration	The submenus of this are numerous and cover a broad array of system and connection settings.
Reset Functions	Menu items allow you to reboot the codec or reset codec settings to the system defaults.
Call History	Shows recent calls made by the codec. Use the MS to scroll through the list of calls.
Phone Book	Allows you to scroll through up to 50 numbers that can be listed in the Phone Book of a profile.
DTMF Control	Provides DTMF control functionality for controlling 3rd party devices.
Unit Details	Contains the details of the serial number, firmware, and other relevant codec information.
Exit Menu	The exit menu takes you out of the current submenu and returns you to the current connection's main LCD.

On the following page there is a diagram with all the codec menu items displayed. Following this is a more in-depth description of the elements within these menus. These descriptions are in a table format, itemizing the selections available and what their function is within the codec.

Please Note: If you would like to see menu diagrams for the Audio and Wizard menus in the codec, please see the Quick Start section of this reference manual.

Figure 37: Menu Submenus



15.21. Menu: Submenu Details

Submenu	Submenu Options	Description
Load Profile	Lists available codec profiles.	Allows you to choose a particular profile to load.
Connection Mode	Manual	Takes you to the Manual connection display.
	CXNS	Displays the CXNS LCD.
Remote Controls	Remote 1 - 6	Using the local codec's MS you can adjust the levels of a remote codec's inputs and switch inputs on and off.
	Exit Menu	Exits to the Menu submenu.
Configuration	System Settings	Sets a range of codec mic input functions as well as PPM monitoring, country selection, IGC and Locking functions.
	ISDN< Left Setup & ISDN> Right Setup	Sets a range of ISDN functions relevant to ISDN use.
	POTS < Left Setup & POTS > Right Setup	Sets a range of modem functions relevant to POTS use.
	GSM/3G Setup	Sets functions relevant to GSM/3G module and cell-phone/modem use.
	IP1 and IP2 Setup	Sets a range of connection functions relevant to IP use.
	X21 1 and X21 2	Sets a range of connection functions relevant to X21 use.
	Auto Reconnect	If a connection fails a codec can automatically initiate a call.
	Min T/Back H/P Lev	Sets the minimum headphone talkback level. This is independent of other program audio feeds.
	Aux Output Level	Sets the output level for the RCA AUX OUT on the left side of the codec.
	Set Clock	Sets the clock in the codec.
	Advanced Menu	Sets ToolBox, CAN, and LAN connection settings as well as system defaults and system reset functions.
	Exit Menu	Exits to the Menu submenu.

Reset Functions	Set System Defaults	Reconfigures the codec to the factory default settings.
	Reboot System	Reboots the codec.
Call History	Displays the previous numbers the codec has used to connect.	
Phone Book	Entries 1 - 50	Displays the phone book entries loaded in the codec.
DTMF Control	Enter DTMF Digits	Provides DTMF control.
Unit Details	Firmware Version	Displays the codec's Firmware Version.
	RPTP Version	The RPTP version of the codec firmware.
	ISDN Version	The ISDN software version.
	Serial Number	Displays the codec's Serial Number.
	Mac Address	The codec Mac address.
	IP Address	Displays the relevant IP address for LAN connections.
	Subnet Mask	Displays the Subnet Mask for LAN connections.
	Default Gateway	Displays the Default Gateway for LAN connections.
	Time	The internal codec clock.
	Date	The date within the codec.
	Build Date	Displays the build date of the codec's operating system.
	Build Time	Displays the build time of the codec's operating system.
	Features Enabled	Displays the features enabled within the codec. Includes: Matrix Editor; Failover; IP Streaming.
Exit Menu	Exits the Menu submenu to either the Manual or CXNS LCD screen.	

The next image displays the Configuration submenus of the codec's Menu submenu. For more detail of these secondary submenu items, check the tables that follow which outline each individual selection available within these menus.

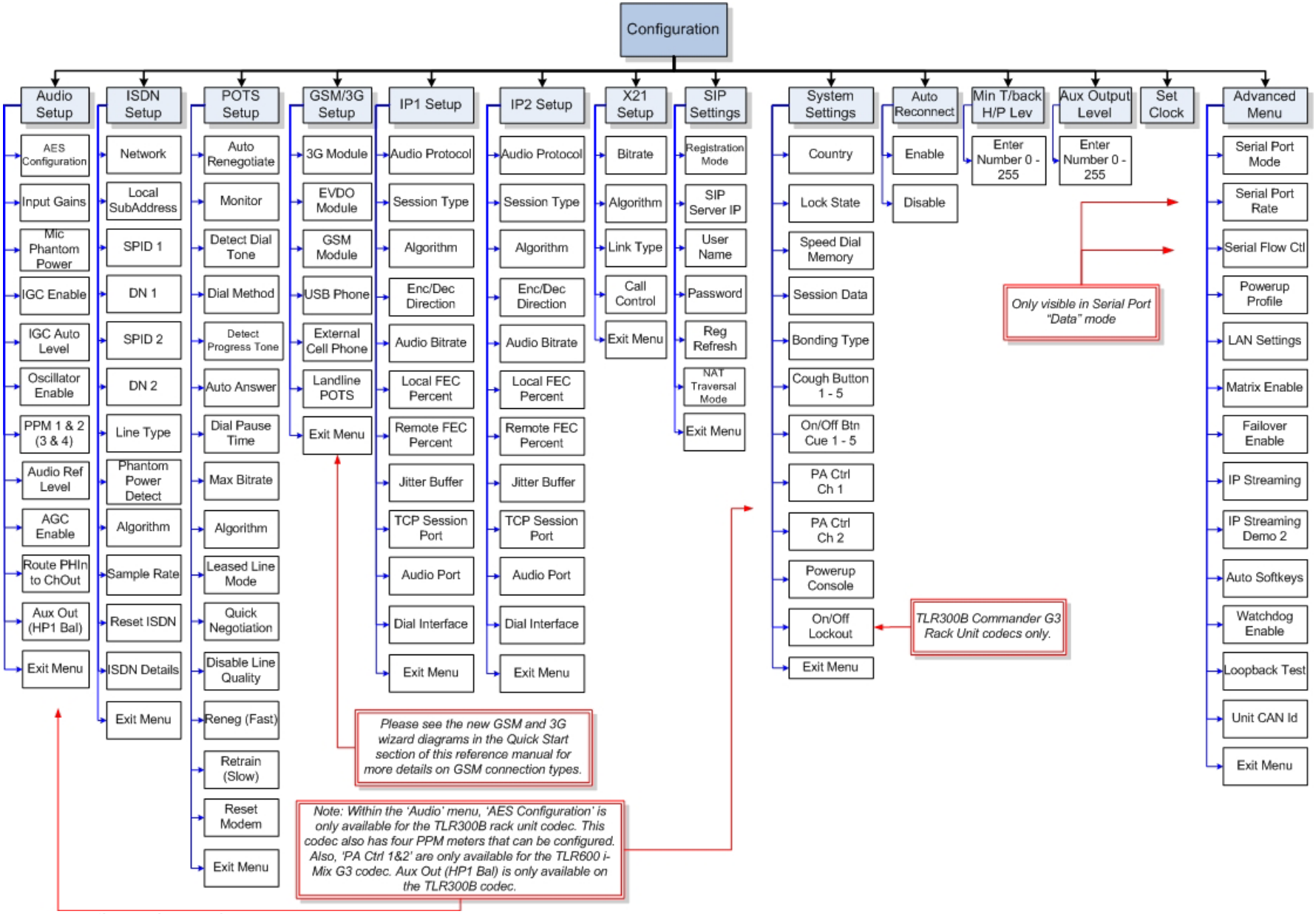


Figure 38: Configuration Submenu Items

Menu Submenu	Secondary Submenus	Secondary Sub-menu Options	Description of Secondary Submenu Options	Additional Options	Description of Additional Options
Load Profile	ManDflt MonoPgm	Really Load? No/Yes	This menu allows you to select the profile you wish to load in the codec. Any custom profiles that are loaded into the codec will also be listed within this menu. By pressing either SOFTKEY 1 or 2, you can select whether you will load the profile.		
	ManDflt Mono/IFB				
	ManDflt Stereo				
	ManDflt DualPgm				
	Other user profiles				
Connection Mode	Auto	Takes the user to the Manual connection LCD			
	Remote 1-6	Takes the user to the connection manager LCD			
Remote Controls		On	Switches the remote codec channel control function on at the local codec.		
		Off	Switches the remote codec channel control function off at the local codec.		

Menu Submenu	Secondary Submenus	Secondary Sub-menu Options	Description of Secondary Submenu Options	Additional Options	Description of Additional Options
	Remote 1-6 cont...	Input levels and mic On/Off	Adjusts the input gains on remote codec inputs. The second row of the local codec's LCD screen will show a bar display of the remote codec's input level. This will move up (to the right) and down (to the left) when the MS is turned. The remote input On/Off function is also controlled by SOFTKEYS 1 & 2.		
	Exit Menu	Exits the Remote Control menu to the Menu submenu			
Configuration	Audio Setup	AES Configuration **(TLR300B only)**	Output Clock Source	AES Rx Clock	Codec sample rate set to the rate of the signal being received by the AES/EBU input.
				External Word Clock	Configures the codec for a word clock source.
				Codec Sample Clock (Default)	Sets the sample rate based on the algorithm you select within the codec.
			Clock Auto Failover	Enable	Programs the codec to failover to the internal sample clock if a sync source is removed from the codec.
				Disable (Disable)	Disables this function.
			Audible Alarm	Enable	Enables a piezo alarm within the codec that sounds if there is a loss of AES/EBU sync in the codec.
				Disable (Default)	Disables this function.

Menu Submenu	Secondary Submenus	Secondary Sub-menu Options	Description of Secondary Submenu Options	Additional Options	Description of Additional Options
	Audio Setup	Input Gains	Allows you to set the input gains for mic inputs on the codec by scrolling with the MS and pressing it to select the option.	All Inputs	Sets all input gains to your selection.
				Input 1 - x	Allows individual input gain selection.
		Mic Phantom Power	The codec by default supplies 15 volts of phantom power to the all balanced mic inputs. (Note: **For TLR300B codecs phantom power cannot be disabled in the codec & doesn't appear in the menu)**	Enable	Turns on phantom power to all inputs.
				Disable	Turns off phantom power to all inputs. (Factory Default)
		IGC Enable	The inbuilt DSP limiter automatically limits high input gains.	Enable	Enables IGC (Default)
				Disable	Disables IGC
		IGC Auto Level	The inbuilt DSP limiter automatically lowers high input gains and then returns incoming levels to the previous setting when they are acceptable.	Enable	Enables IGC Auto Level (Default)
				Disable	Disables IGC Auto Level
		Oscillator Enable	Enable the oscillator within the codec to send 400 Hz tone to all outputs.	Enable	Turns the oscillator on.
				Disable	Turns the oscillator off.
		PPM 1 - 4	Sets the PPM monitoring for all meters on the codec. (	Matrix (Default)	The PPM meters display as per the matrix settings (Factory Default)
				Encoder 1	Displays the level entering the encoder.
				Encoder 2	Displays the level entering the encoder.
				Decoder 1	Displays the level leaving the encoder.
				Decoder 2	Displays the level leaving the encoder.
		Audio Ref Level	Solves audio level incompatibility between a **Tieline** codec and another brand of codec. (Related topic: *Inputs*)	Tieline Codecs	The default setting for **Tieline** codecs.
				Other Codecs	Use this setting to fix incompatibilities.

Menu Submenu	Secondary Submenus	Secondary Sub-menu Options	Description of Secondary Submenu Options	Additional Options	Description of Additional Options
		Route PhIn to ChOut	Routes POTS and GSM voice calls to a codec's analog outputs.	Enable	Enables routing.
				Disable	Disables routing (Default).
		Aux Out (HP1 Bal)	The codec's 1/4" (6.5mm) headphone output can be programmed to be a balanced mono output.	Enable	Enables balanced mono output.
				Disable (Default)	Disables balanced mono output, enables auxiliary output.
		Exit Menu	Exits this submenu and goes to the Configuration submenu.		
Configuration	ISDN< Left Setup (and/or ISDN> Right Setup).	Network	This menu selects the ISDN software you wish to use The list of possible selection items here depends on the ISDN software type loaded into the ISDN module in the codec you are using.	EU-ETSI	If Switch Type is ETSI (UK, Europe, Australia and most other countries)
				JPN-NTT	If you are in Japan and your network is NTT
				KOR-KOR	If you are in the Korea
				US-Nat	If switch type is National ISDN-1 and 2
				US-DMS	If switch type is Northern Telecom DMS100
				US-AT&T	If switch Type is AT&T Custom

Table continued on the next page...

Menu Submenu	Secondary Submenus	Secondary Sub-menu Options	Description of Secondary Submenu Options	Additional Options	Description of Additional Options
Configuration cont...	ISDN< Left Setup (and/or ISDN> Right Setup).	Local SubAddress	This setting must be blank by default. Use it only if your codec shares the line with another ISDN device.	Enter Number	
		SPID 1 and 2	Users outside North America please leave the SPID1 and SPID2 blank. North American users need to get SPID (Service profile ID) numbers from the phone company and enter them exactly as supplied.	Enter Number	
		DN 1 and 2 or MSN numbers.	A Directory Number is the ISDN number assigned to each B channel of an ISDN connection. It may also be known as the Multiple Subscriber Number (MSN).	Enter Number	
		Line Type	Selects point-to-multipoint configurations (multiple but dissimilar devices) or point-to-point (one codec connected to another)	Point-to-Point	Selects point-to-point configuration
				Point-to-Multipoint	Selects point-to-multipoint configuration (Default)
		Phantom Power Detect	The codec will detect phantom power loss when using ISDN lines.	Enable	Enables loss detection
				Disable	Disables loss detection
		Algorithm (Default Algorithm setting in the codec for ISDN)	This selects the default algorithm that you will be using within the codec. MPEG algorithms allow you to select the sample rate from the four available options.	Music/Default	
				Music Plus	
				G.711	
				G.722	

Menu Submenu	Secondary Submenus	Secondary Sub-menu Options	Description of Secondary Submenu Options	Additional Options	Description of Additional Options
Configuration cont...	ISDN< Left Setup (and/or ISDN> Right Setup).	Algorithm cont...	If you select one of the MPEG algorithms this menu appears and allows you to adjust the sample rate of the selected algorithm.	MP2 Mono	16,000; 24,000; 32,000; 48,000.
				MP2 Dual	16,000; 24,000; 32,000; 48,000.
				MP2 Stereo	16,000; 24,000; 32,000; 48,000.
				MP2 J-Stereo	16,000; 24,000; 32,000; 48,000.
		Sample Rate	Adjusts the sample rate setting in the codec.	16,000	Sets this sample rate in the codec.
				24000	Sets this sample rate in the codec.
				32000 (Default)	Sets this sample rate in the codec.
				48000	Sets this sample rate in the codec.
		Reset ISDN	Resets the ISDN connection		
		ISDN Details	This menu provides ISDN hardware interface and firmware details.	Hardware Interface	Provides hardware interface details. This is an auto-detect feature in the codec and shows whether your ISDN module is interfacing with S/T or U type line interfaces.
				Firmware	Provides firmware details.
		Exit Menu	Exits this submenu and goes to the Configuration submenu.		

Table continued on the next page...

Menu Submenu	Secondary Submenus	Secondary Sub-menu Options	Description of Secondary Submenu Options	Additional Options	Description of Additional Options
Configuration cont...	POTS Left Setup (and/or POTS Right Setup)	Auto Renegotiate	This menu allows you to select from various modem connection renegotiation settings.	Disable	The setting for live program links because it prevents data renegotiation if line conditions change (default).
				Up	A higher bit rate is used if line quality at both ends is above 80%
				Down	A lower bit rate is used if line quality falls below 20% at either end
				Auto	Selected auto-renegotiation occurs as line conditions vary and that can be either up or down
		Monitor	The modem dialing and connection process can be heard through the headphones.	Enable	Enables monitoring (default).
				Disable	Disables monitoring.
		Detect Dial Tone	By enabling the *Detect dial tone* function the codec will wait until a dial tone is present before dialling.	Enable	(Default) Enables this function.
				Disable	Disables this function. Use if there is any difficulty in detecting a dial tone.
		Dial Method	This menu selects either Tone (DTMF) dialling or Pulse dialling.	Tone	(Default) Selects *Tone* dialing.
				Pulse	Selects *Pulse* dialing.
		Det. Progress Tone	If this function is enabled, the codec tests for phone system connection tones at the conclusion of dialing (e.g."busy" tone).	Disable	Disables monitoring.
				Disable	Disables Progress tone detection.
		Auto Answer	Allows the user to enable or disable automatic answering of incoming calls to start operation.	Enter Number 0-9	Inserts the number of rings before answering. Insert 0 to auto-answer.
		Dial Pause Time	Determines the length of any dialing pause inserted into the telephone number.	Enter Number	Sets the actual pause in milliseconds.
		Max Bitrate	Sets the maximum bit rate at which the codec will attempt a connection.	9600bps – 33600bps	Use the MS to scroll and set the maximum bit rate.

Menu Submenu	Secondary Submenus	Secondary Sub-menu Options	Description of Secondary Submenu Options	Additional Options	Description of Additional Options
Configuration cont...	POTS Left Setup (and/or POTS Right Setup)	Algorithm (Default Algorithm)	This selects the default algorithm that you will be using within the codec.	Music	Selects this algorithm. (Default)
				Voice G3	Selects this algorithm
				Other	Enables G3 codecs to connect to other brands of codec. (Please note: Connection bitrates of 9,600 and 26,400 are not supported.)
		Leased Line Mode	This needs to be checked for operation over leased 2 wire bi-directional lines. As the codec modem requires a loop voltage source for correct operation, an optional line interface unit is available for use with dedicated 2 wire circuits where the loop voltage is not present.	Enable	Enables this for leased 2-wire lines
				Disable (Default)	Disables this function (for non-leased 2-wire lines).
		Quick Negotiation	If enabled, the modem will attempt to connect at the highest bit-rate automatically. If disabled, the modem will attempt to connect initially at 19,200 bps and then negotiate the highest bit-rate.	Enable	Enables this function.
				Disable	Disables this function (Default setting)
		Disable Line Quality	Disables the line quality display on the codec LCD. See POTS section for details.	Enable (Default)	Enables the line quality display.
				Disable	Disables the line quality display.
		Reneg (Fast)	With this setting, the codec will quickly renegotiate the connection as per the settings in this menu. The modem is not recalibrated in this process.	Auto	Auto-renegotiation occurs as line conditions vary and that can be either up or down.
				Up	A higher bit rate is renegotiated if line quality at both ends is above 80%.

Table continued on the next page...

Menu Submenu	Secondary Submenus	Secondary Sub-menu Options	Description of Secondary Submenu Options	Additional Options	Description of Additional Options
Configuration cont...	POTS Left Setup (and/or POTS Right Setup) cont...	Reneg (Fast) cont...		Down	A lower bit rate is renegotiated if line quality falls below 20% at either end.
				Current	Renegotiates to the current bit rate setting.
				9600bps - 33600bps	Renegotiates to the bit rate selected from the menu.
		Retrain (Slow)	In this setting the modem will be recalibrated and then it will retrain the connection. This process is slower than Reneg (Fast) 1. Over time two connected modems can drift (causing phase discrepancies). Therefore retraining at the same bit rate can potentially improve LQ (Line Quality).	Auto	The modem retrains as line conditions vary and that can be either up or down.
				Up	A higher bit rate is retrained if line quality at both ends is above 80%.
				Down	A lower bit rate is renegotiated if line quality falls below 20% at either end.
				Current	The modem will retrain to the current bit rate setting.
				9600bps - 33600bps	Retrains to the bit rate selected from the menu.
		Reset Modem	Resets the modem.		
		Exit Menu	Exits this menu and goes to the Configuration submenu.		

Table continued on the next page...

Menu Submenu	Secondary Submenus	Secondary Sub-menu Options	Description of Secondary Submenu Options	Additional Options	Description of Additional Options
Configuration cont...	GSM/3G Setup	GSM Module/3G Module/ EVDO Module/USB Phone	Selecting this menu allows you to set the connection bit rate, wireless network, pre-buffer seconds, signal strength enable, and algorithm for your GSM, 3G (HSDPA), EV-DO or USB module. Only one GSM module can be installed in a codec or an error message will be displayed.		
		Ext CellPhone	If you are connecting via a cell-phone using the RS 232 port on the rear of the codec then you will need to enable this function with this menu. You can then set the connection bit rate, wireless network setting, pre-buffer seconds, signal strength enable, and algorithm settings for your GSM phone. For more information, please see the *GSM LL/GSM/USB-3G* section, or the GSM Operation and Quick Start sections within this manual.	Enable	Enables this function.
				Disable	Disables this function

Table continued on the next page...

Menu Submenu	Secondary Submenus	Secondary Sub-menu Options	Description of Secondary Submenu Options	Additional Options	Description of Additional Options
Configuration cont...	GSM/3G Setup cont...	Landline POTS < or > or ^	Enable if the codec you are using is to dial via a landline and will be communicating with another codec using a GSM cell-phone. Only the landline-connected codec needs to check this function. This function needs to be left unchecked unless you are using a GSM cell-phone. Selecting Enable in this menu allows you to then set the connection bit rate, Pre-Buffer seconds and algorithm for your POTS module. For more information please see the ***GSM LL/GSM/USB-3G*** section within this manual.	GSM Landline Mode Enable	Enables this function.
				GSM Landline Mode Disable (Default)	Disables this function.
		Exit Menu	Exits this submenu and goes to the Configuration submenu.		
	IP1 & IP2 Setup	Audio Protocol	Sets the audio protocol used to connect the codecs over IP.	UDPIP	Selects the UDP IP protocol (Default)
				TCPIP	Selects the TCP IP protocol
		Session Type	Programs Tieline session data, SIP or no session data for IP connections.	Tieline Codecs	Tieline Session Data used to connect devices.
				None	No session data sent when connecting.
				SIP	Session Description Protocol used to connect devices.
		Algorithm	Selects the algorithm used to connect over the IP connection.	Music	Selects this algorithm. (Default)
				Voice G3	Selects this algorithm.

Menu Submenu	Secondary Submenus	Secondary Sub-menu Options	Description of Secondary Submenu Options	Additional Options	Description of Additional Options
Configuration cont...	IP1 & IP2 Setup cont...	Algorithm	Selects the algorithm used to connect over the IP connection.	Music	Selects this algorithm (Default).
				Music Plus	Selects this algorithm.
			If you select one of the MPEG algorithms this menu appears and allows you to adjust the sample rate of the selected algorithm.	MP2 Mono	16,000; 24,000; 32,000; 48,000.
				MP2 Dual	16,000; 24,000; 32,000; 48,000.
				MP2 Stereo	16,000; 24,000; 32,000; 48,000.
				MP2 J-Stereo	16,000; 24,000; 32,000; 48,000.
				Other	Selects this algorithm.
				Voice G3	Selects this algorithm.
				Raw Audio	Selects this algorithm.

Menu Submenu	Secondary Submenus	Secondary Sub-menu Options	Description of Secondary Submenu Options	Additional Options	Description of Additional Options
Configuration cont...	IP1 & IP2 Setup	Enc/Dec Direction	Programs a codec to encode or decode only, or encode and decode.	Both	Programs a codec to encode and decode bidirectional audio.
				Encode Only	Programs a codec to encode only.
				Decode Only	Programs a codec to decode only.
		Audio Bitrate	Sets the maximum bit rate at which the codec will attempt a connection.	9600bps – 33600bps (default 9600bps)	Use the MS to scroll and set the maximum bit rate.
		Local FEC Percent (UDP only)	Configures the forward error correction setting on the local codec.	Off (Default), 20%, 33%, 50%, 100%.	There are five settings in this menu for UDP connections. For more info see the IP sections of this manual.
		Remote FEC Percent (UDP only)	Configures the forward error correction setting on the remote codec.	Off (Default), 20%, 33%, 50%, 100%.	There are five settings in this menu for UDP connections. For more info see the IP sections of this manual.
		Jitter-Buffer	Programs the jitter-buffer setting on the codec - either “Auto” or “Fixed”.	Auto Jitter Adapt	Choose from: Least Delay; Less Delay; Best Compromise (Default); Good Quality; Highest Quality.
				Fixed Buffer	Enter the jitter-buffer milliseconds.
		TCP Session Port	Reprograms the TCP Session Port number.		
		Audio Port	Reprograms the codec Audio Port number.		
		Exit Menu	Exits this submenu and goes to the Configuration submenu.		

Table continued on the next page...

Menu Submenu	Secondary Submenus	Secondary Sub-menu Options	Description of Secondary Submenu Options	Additional Options	Description of Additional Options
	X21 > (or X21< in Commander Codecs only).	Bitrate	Sets the X.21 bit rate to match the bit rate of the ISDN TA/X.21 DCE Device that the codec is connecting to. (Or X.21 to X.21 connections when this is implemented).	384000	X.21 to X.21 only.
				320000	X.21 to X.21 only.
				256000	Sets this connection bit rate
				192000	Sets this connection bit rate
				128000	Sets this connection bit rate
				64000	Sets this connection bit rate
		Algorithm	Sets the algorithm in the codec.	Music	Selects this algorithm.
				G.711	Selects this algorithm.
				G.722	(Default) Selects this algorithm.
				MP2 Mono	Selects this algorithm.
				MP2 Dual	Selects this algorithm.
				MP2 Stereo	Selects this algorithm.
				MP2 J-Stereo	Selects this algorithm.
		Sample Rate	Sets the sample rate for an MP2 algorithm if it is selected.	16000	Selects this sample rate.
				24000	Selects this sample rate.
				32000	Selects this sample rate.
				48000	Selects this sample rate.
		Link Type	**Tieline** codec X.21/V.35 modules currently only support connections via an ISDN Terminal Adapter (TA). Ensure that you select *X21 < - > ISDN* in this menu.	X.21 - ISDN	Select this options
				X.21 – X.21	Check at support@tieline.com for the implementation date of this feature.
		Call Control	Selects the call mode used. See the X.21 section for more info.	C Signal	Selects this option
				Permanent Link	Selects this option
		Exit Menu	Exits this submenu and goes to the Configuration submenu.		

Table continued on the next page...

Menu Submenu	Secondary Submenus	Secondary Sub-menu Options	Description of Secondary Submenu Options	Additional Options	Description of Additional Options
Configuration	SIP Settings	Registration Mode	Enables registration of a codec to a SIP Server.	Enable	Enables a codec to be registered.
				Disable (Default)	A codec remains unregistered in disable mode.
		SIP Server IP	Enter the IP address of the SIP server that a codec is being registered to.		
		User Name	Enter the SIP user name for the codec.		
		Password	Enter the SIP password into the codec.		
		Reg Refresh (Sec)	Enter the registration refresh setting in seconds. The default setting is one hour.		
		Nat Traversal Mode	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. This is necessary if a device has a private IP address behind a public IP address and a firewall.	Auto (Default)	NAT (Network Address Translation) will be performed if required.
				On	NAT (Network Address Translation) will be performed if required.
				Off	NAT Traversal mode is switched off.
		Exit Mode			

Menu Submenu	Secondary Submenus	Secondary Sub-menu Options	Description of Secondary Submenu Options	Additional Options	Description of Additional Options
Configuration	System Settings	Country	Set country-specific phone settings like ring cadence from the list of countries.		
		Lock State	Allows you to lock or unlock the operational settings of codec.	Unlocked	Unlocks the codec. (Default)
				Locked	Locks the codec.
		Speed Dial Memory	Enables Speed-dialing of phonebook numbers only, or dials numbers and changes profiles as well.	Profile	Changes profiles and dials a number.
				Phonebook	(Default) Dials a number only.
		Session Data	This menu can disable and enable session data sent from a codec. Disabling session data may assist to connect via G.722 and MPEG algorithms.	Enable	Enables session data to be sent from a codec.
				Disable	Disables the sending of session data from a codec.
		Bonding Type	CCS bonding, used by MPEG algorithms, is only capable of bonding two connections together. **Tieline** codecs can bond up to four channels using **Tieline's** bonding.	Auto	Automatic selection. See 'ISDN 3B and 4B Bonding' for details.
				Tieline	Selects Tieline Bonding
				CCS	Selects CCS Bonding
		Cough Buttons 1 – x	Switches the ON/OFF button on the codec into a cough button. **(Not available on the TLR300B).**	Enable	Enables the cough button
				Disable	Disables the cough button
		On/Off Btn Cue 1 - x	Switches the ON/OFF button on the codec into a CUE-COMMS button. **(Not available on the TLR300B).**	Enable	Enables the Cue functionality.
				Disables	Disables the Cue functionality.

Table continued on the next page...

Menu Submenu	Secondary Submenus	Secondary Sub-menu Options	Description of Secondary Submenu Options	Additional Options	Description of Additional Options
		PA Control Ch 1 and Ch 2 Out	If enabled, the sources sent to these two outputs can be controlled by the PA OUTPUT controls on the front panel of an *i*-Mix *G3* codec. (This function is only available on the TLM600 codec).	Enable	Enables PA Control functionality.
				Disables	Disables PA Control functionality.
		Powerup Console	Powerup console relates only to the LCD menu screen on the codec. The setting you choose within this menu will decide which menu screen the codec will display on power up. It does not control which profiles are loaded into the codec prior to powering the unit down.	Manual	Manual refers to the manual connection LCD. (Default setting)
				Cxn Mgr	Cxn Mgr refers to the Connection Manager CXNS LCD
				Last Selected	This refers to the last the last screen selected with console navigation, either Cxn Mgr or Manual.
		On/Off Lockout	Enables and disables this feature in TLR300B codecs only.	Disable	Disables this function.
				Enable	(Default) Enables this function.
		Exit Menu	Exits this submenu and goes to the Configuration submenu.		
	Auto Reconnect	Enable	If the link fails the codec will automatically initiate redialing.		
		Disable (Default)	If the link fails the codec will **not** automatically initiate redialing.		
	Min T/back H/P Level	Enter Number (Enter a number between 0 & 255)	Sets the minimum headphone talkback level. Even if a commentator turns down the headphone digipot completely on the codec, talkback will still be heard in the headphones.		

Table continued on the next page...

Menu Submenu	Secondary Submenus	Secondary Sub-menu Options	Description of Secondary Submenu Options	Additional Options	Description of Additional Options
Configuration cont...	Aux Output Level	Enter Number	Sets the output level of the RCA AUX OUT on the left-hand side panel of the codec. (Enter a number between 0 & 255) **(Not available on the TLR300B codec)**		
	Set Clock	Set/Exit	Sets the clock in the codec. Use SOFTKEYS 2 & 3 to navigate around the LCD menu items and use the MS to scroll to the desired setting. Press set to save your settings and start the clock.		
	Advanced Menu	Serial Port Mode	Before you can communicate between your codec other devices successfully, you must ensure that both your codec and your ToolBox software are configured correctly. (Related Topic: Connecting Your Codec to a PC).	ToolBox	Configures for ToolBox connections.
				GSM	Enables the codec for connection to a GSM enabled cell-phone.
				Data	Tells the codec you will be sending data via the serial port.
		Serial Port Rate	Sets the data rate for data connections using the RS232 serial port on the rear codec panel.	1,200 – 115,200	Choose your required data rate from a range of settings.
		Serial Flow Ctl	If enabled, this allows a device connected to the codec to regulate the flow of data.	Enable	Enables RS232 flow control.
				Disable	Disables RS232 flow control (Default)
		Powerup Profile	This menu allows you to set the profile that you would like loaded into the codec when it is initially powered up.	Last User Selected	Selects the last profile loaded into the codec.
				Current Runtime	(Default) Selects Current Runtime as the active profile.
				ManDflt MonoPgm	Selects this profile on powerup.
				ManDflt Mono/IFB	Selects this profile on powerup.

Table continued on next page...

Menu Submenu	Secondary Submenus	Secondary Sub-menu Options	Description of Secondary Submenu Options	Additional Options	Description of Additional Options
Configuration cont...	Advanced Menu cont...	Powerup Profile cont...		ManDflt Stereo	Selects this profile on powerup.
				ManDflt Dual Pgm	Selects this profile on powerup.
		LAN Settings	Ethernet setup allows the user to select the Ethernet data transfer speed. IP setup allows the user to input the IP addressing information that is necessary when using a LAN connection.	Ethernet Link	Auto (Default)
					10 Mbs Half Duplex
					10 Mbs Full Duplex
					100 Mbs Half Duplex
					100 Mbs Full Duplex
				IP Setup	DHCP
					BOOTP
					Static: IP Address
					Static: Subnet Mask
					Static: Default Gateway
				Exit Menu	Exits to the Advanced submenu.
		Matrix Enable	If you have purchased the ***Matrix Editor*** option from **Tieline**, this menu will allow you to enter your 12 digit unlocking-key.	Enable	Enables this function, press OK via SOFTKEY 4 after entering the number.
				Disable	Disables this function.
		Failover Enable	If you have purchased the Failover Profile option from **Tieline**, this menu will allow you to enter your 12 digit unlocking-key. After entering the key, it will enable or disable the failover function in the codec.	Enable	Enables this function, press OK via SOFTKEY 4 after entering the number.
				Disable	Disables this function.

Table continued on the next page...

Menu Submenu	Secondary Submenus	Secondary Sub-menu Options	Description of Secondary Submenu Options	Additional Options	Description of Additional Options
Configuration cont...	Advanced Menu cont...	IP Streaming	If you have purchased the IP Streaming function for your codec, input the key into your codec by selecting 'Enable' within this menu.	Enable	Enables this function, press OK via SOFTKEY 4 after entering the number.
				Disable	Disables this function.
		IP Stream Demo 2	Allows users of Tieline codecs to enable a 45 day trial of IP streaming. After 45 days the IP streaming key must be purchased to continue to use this function.	Enable	Enables the IP Stream demo.
				Disable	Disables the IP Stream demo. .
		Auto Softkeys	In TLF300 field codecs the "Cue" button LCD display is enabled automatically by this feature.	Enable	Enable this feature.
				Disable	Disables this feature (Default).
		Watchdog Enable	Enables resetting of codecs automatically if they require this intermittently.	Enable	Enable this feature.
				Disable	Disables this feature (Default).
		Unit CAN Id	Not required for standard setups. In some situations two codecs and an external relay box may attached to each other. In essence the codecs are 'networked' and the relay box may need to be attributed to one of the codecs. The CAN Id does this.	Enter Number	Enters the CAN Id #
		Exit Menu	Exits this submenu and goes to the Configuration submenu.		
Reset Functions		Set System Defaults	This menu will allow you to set the codec to the factory default of Current Runtime (*ManDflt Mono* profile). The option to set all the manual profiles to the factory defaults is also available.	Runtime Profile	Current Runtime only will be reset to the factory default.
				All profiles	Current Runtime and the factory default manual profiles will be reset. Custom profiles will also be deleted.
				Exit menu	Exits this menu selection.
		Reboot System	This is akin to powering the codec down and then up again.	No	Exits the reboot option.
				Yes	Reboots the codec.

Menu Submenu	Secondary Submenus	Secondary Sub-menu Options	Description of Secondary Submenu Options	Additional Options	Description of Additional Options
Call History					
Phone Book	Entries 1 - 80	Dial	Allows you to edit the phone number of an entry.	Enter Number	Enter the number you wish to save.
		Edit	Allows you to edit the name of the phone entry using the MS.	Enter Name	Scroll the MS to select alphanumeric letters and name the entry.
		Delete	Delete function for the phone entry.	NO	Exits delete function.
				YES	Deletes the number.
DTMF Control	Enter DTMF Digits				
Unit Details	Firmware Version	The software version within the codec.			
	RPTP Version	The RPTP version of the codec software.			
	ISDN version	Displays the version of the software in ISDN modules.			
	Serial Number	The serial number of the codec.			
	Audio Card Type	Displays the input module inserted in the codec.			
	Mac Address	Displays the Media Access Control address (MAC). A MAC address identifies each node of a network.			
	IP Address	Displays the IP address for LAN connections.			
	Subnet Mask	Displays the Subnet Mask for LAN connections.			

Table continued on the next page...

Menu Submenu	Secondary Submenus	Secondary Sub-menu Options	Description of Secondary Submenu Options	Additional Options	Description of Additional Options
Unit Details cont...	Default Gateway	Displays the Default Gateway for LAN connections.			
	Time	Displays the time set in the codec.			
	Date	Displays the date set in the codec.			
	Build Date	Displays the build date of the codec's operating system.			
	Build Time	Displays the build time of the codec's operating system.			
	Features Enabled	Lists the features enabled in the codec.			
Exit Menu	Returns you to the connection screen of the profile you have loaded into the codec.				

Section 16. Data Transfer & Using 3rd Party Devices

This section discusses the data sending options available when using **Tieline** codecs. First we look at the session data that is sent between codecs when they first connect. We also explore the data sending capabilities of codecs for remote controlling audio channels and for sending data used to control third party devices.

16.1. Data Options Available

Every connection includes a minimum of 50 bytes per second in a remote control channel to provide **Tieline** ToolBox capabilities, and which can be also used to control specific **Tieline** external devices, i.e. external relay boxes.

If your codec is captioning-enabled, please see the captioning user manual for more information on data transfer and setting your codec up.

For specific GSM data information, please see the section in this reference manual titled 'GSM Transparent and Non-transparent Data Modes'.

16.2. An Introduction to Session Data

The information that follows provides relevant information about how session data is used when sending data between **Tieline** codecs. We will also discuss the implications of connecting a **Tieline** codec to a non-**Tieline** codec. Before discussing session data in detail it is useful to outline some information on data packets and the OSI (Open System Interconnection) model of computer communications. This provides an understanding of where session data sits in relation to computer network protocol design.

16.3. The Serial Port Data Wizard

A codec serial port can be configured to send data by pressing SOFTKEY 4 and then using the MENU SELECTOR (MS) to scroll to and select **[Configuration] > [Advanced Menu] > [Serial Port Mode] > [Data].** A wizard will then ask an operator to confirm the following:

1. The serial port data rate setting (default is 9600); and
2. Whether to enable serial port flow control, allowing a device connected to the codec to regulate the flow of data (default is enabled);

These elements are discussed in more detail later in this section of this reference manual.

16.4. 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).

16.5. The OSI Model Explained

To assist in understanding the operation of session data within **Tieline** codecs following is a diagram of the OSI model of computer network protocol design. In the table below, the functions of the protocol are divided into layers to describe each element individually.

The OSI Model Explained			
	Data Unit	Layer	Function
Application	Data	Layer 7: Application	End user processes to application
		Layer 6: Presentation	Data representation/encryption
		Layer 5: Session	Establish, manage and terminate communication.
Data Transport	Segments	Layer 4: Transport	End-to-end data transfer, error recovery and flow control (The TCP & UDP protocols are used on the Internet).
	Packets	Layer 3: Network	Switching and routing (IP)
	Frames	Layer 2: Data link	Physical addressing (i.e. MAC & LLC)
	Bits	Layer 1: Physical	Hardware providing the ability to send and receive data.

Table 7: The OSI Model Explained

This model is roughly adhered to throughout the IT industry and it is important to recognize that each layer only uses the functions of the layer below it and only performs services for the layer above it. The implementation of several layers is often referred to as a 'stack'.

It is appropriate to now explain the Session Layer in some more detail, as well as the attributes of session data and how it is sent between two codecs.

16.6. The Session Layer

The OSI model is a good example of how the session layer fits into the scheme of network protocol design.

Essentially, the session layer is used to establish, manage and terminate communication sessions between local and remote applications – in our case two codecs.

16.7. How Tieline Codec Session Data Works

Session data operates over the full bandwidth of a codec connection. This is a minimum of 9,600 bps and is dependent on the connection type being used. As with audio data, session data is sent bidirectionally from both the remote and local codecs.

Please Note:

It is possible that when you use *G.722* and *MP2* algorithms to connect to a non-**Tieline** codec, the 'framing' process (where incoming bit streams are identified and distinguished for individual decoding) can be unsuccessful when a call is initiated. This can be solved by turning off the 'session data' stream sent by the **Tieline** codec. To do this select **[Menu] > [Configuration] > [System Settings] > [Session Data] > [Disable].** Please note that this is a global setting and will also affect all POTS, and IP connections on the codec.

If you try to connect to a **Tieline** codec with session data disabled, you will have to ensure that the connection settings on both codecs are identical because dialing codec session data normally provides this information to the remote codec. To avoid this problem, simple re-enable session data **[Menu] > [Configuration] > [System Settings] > [Session Data] > [Enable].**

16.7.1. Information Contained within Session Data

When session data is enabled in **Tieline** codecs the following occurs when a connection is established:

1. The two codecs negotiate and initiate the sending of session data;
2. The master codec, which is usually the dialing codec, then sends information pertaining to:
    a. The profile loaded;
    b. Which encoders and decoders are being used;
    c. The algorithm loaded into the codec;
    d. The model of the codec and its serial number; and
    e. The codec sample rate.
3. Once the codec receiving session data from the master codec has received session data information successfully, it sends an acknowledgement to the master codec that it has been received successfully.
4. Session data is then simultaneously shut down between the two codecs and audio data replaces session data at the codecs.

16.7.2. Session Data and Non-Tieline Codecs

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

Most non-**Tieline** codecs do not use session data when connecting. As a result, they will not respond to session data sent by a **Tieline** codec. 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.

We recommend that it is best to turn off session data if you are connecting your **Tieline** codec to a non-**Tieline** codec over any connection. 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).

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.

16.7.3. Enabling and Disabling Session Data

Turning session data on and off is a simple task. To configure this setting press SOFTKEY 4 Menu and select **[Configuration] > [System Settings] > [Session Data]**. The default setting is *Enable*. Select *Disable* if you wish to disable this feature.

It is also possible to program session data via Tool**Box** software. To do this access the ***Unit Options*** tab as displayed in the following image. If the check-box has a tick in it then session data is enabled.

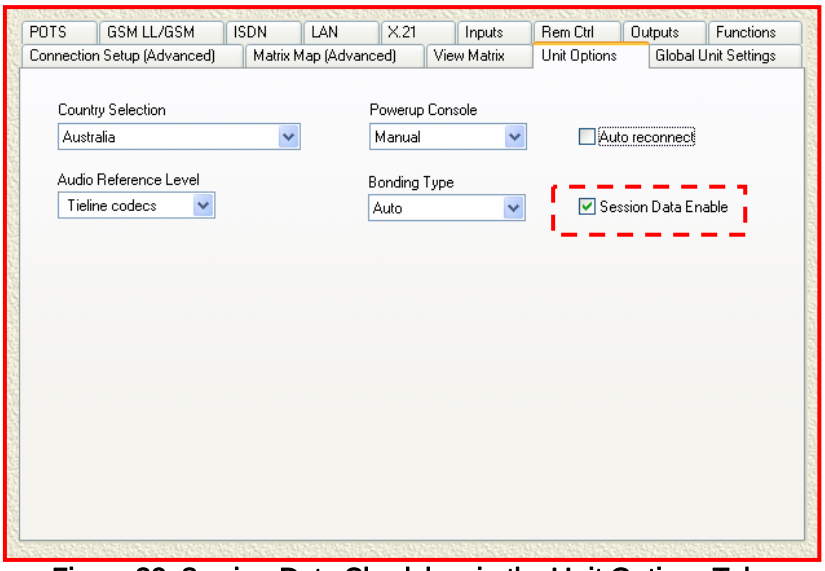


Figure 39: Session Data Check-box in the Unit Options Tab

Remember, if you try to connect to a **Tieline** codec with session data disabled, you will have to ensure that the connection settings on both codecs are identical because the dialing codec's session data normally provides this information to the remote codec. To avoid this problem, simple re-enable session data.

Please note:
This is a global setting and will affect all connections on the codec.

16.7.4. Checking Session Data is not Corrupted

It is possible to check the current state of session data being sent between two codecs. To check this press the ENTER/DIAL button or the MENU SELECTOR on the codec once connected. Scroll until you see the screen as displayed in the following image.

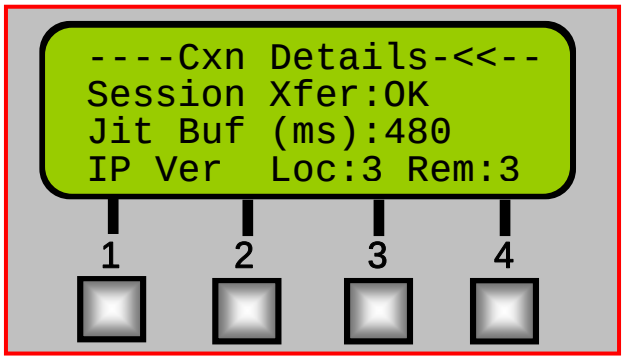


Figure 40: Session Data Status Display

Some of the other messages you may see in the Session Xfer session include:

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

16.7.5. Limitations of Session Data Configuration

In most situations session data can be sent successfully over different connections. Please take note of the following limitations in relation to session data use.

16.7.5.1. GSM Limitations

There can be some limitations on its use with GSM connections because of the delay introduced over these connections.

GSM data calls can have long delays as a result of network loss and GSM delays. This can create problems in successfully transferring session data between two codecs. We **highly recommend** that both codecs are programmed with the same settings if you are connecting using GSM - in case session data fails. This will ensure that you are still able to connect successfully.

16.7.5.2. X.21 Use

Session data does not time-out in X.21 connections. It is therefore **imperative** that you **turn session data off** if you are connecting a **Tieline** codec to a non-**Tieline** codec. In this situation ensure that both codecs are pre-programmed to operate using X.21.

Important Connection Note:
Disabling session data may assist to connect via G.722 and MPEG 2 Layer 2 algorithms in some situations. Session data should only be disabled if you are having difficulty connecting. The default setting (enabled) should be used in most situations.

16.8. Configuring the Serial Port to Send Data

By default, the RS232 connection is programmed to communicate with ToolBox software. To configure your RS232 connection to send data press SOFTKEY 4 Menu and select **[Configuration] > [Advanced] > [Serial Port Mode] > [Data] > [Serial Port Rate]**. Ensure you match the serial port rate to match the rate of the external device you are connecting to. Ideally the settings on both codecs should match or you could have data overflow issues. Once this is done the serial port is configured to send data. Remember, this needs to be configured on both codecs.

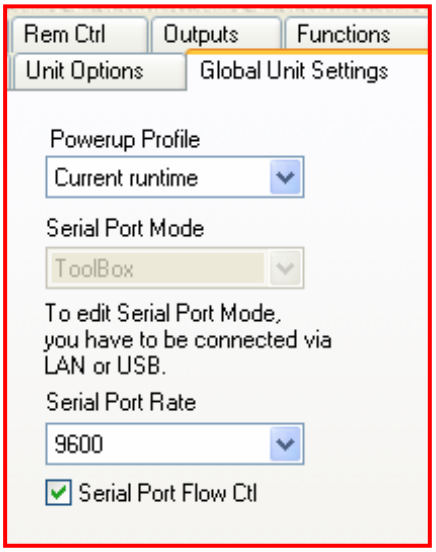
16.9. Enabling Serial Port Data Flow Control

It is also important that you enable serial port flow control within the codec. Flow control, when enabled, regulates the flow of data through the serial port. If disabled, data will flow unregulated and some may be lost.

To enable flow control access the Menu submenu by pressing SOFTKEY 4. Then scroll to **[Configuration] > [Advanced] > [Serial Flow Control]**. Select Enable to enable flow control. Remember, this needs to be configured on both codecs.

16.9.1. Configuring the Serial Port Using ToolBox

All of the settings in the codec menus can be configured using ToolBox software as well as with the codec itself. The Serial Port Mode, Serial Port Rate and Serial Port Flow Control can all be programmed - as displayed in the *Global Unit Settings* tab image shown.



Use the drop-down menus to select the Serial Port Mode and Serial Port Rate. Check the Serial Port Flow Control box to enable flow control. Default settings are shown.

16.9.2. The Serial Port (RS232) Auxiliary Data Channel for MP2 and Tieline Music Connections

The RS232 connection on **Tieline** codecs is capable of sending in-band data between two codecs using either **Tieline** *Music* or *MP2* algorithms. In-band data is encoded in the same bit stream as the audio that is sent between two codecs. Simply configure your RS232 connection to send data, turn serial port flow control on and you are ready to go.

If you are connecting over POTS, 50 bits of in-band data is available per second using **Tieline** *Music*. Over other connections, i.e. ISDN, IP etc, up to 200 bits per second may be available using **Tieline** *Music*. This algorithm will provide greater data sending capabilities if more channels are bonded together, i.e. by increasing the amount of B channels that are bonded using **Tieline** *Music* (this algorithm uses **Tieline** bonding).

Using *MP2,* a 64 kbps connection will provide a maximum data rate of 30 characters (240 bits) per second over ISDN, X.21 and IP. Unlike with *Tieline* bonding, with CCS bonding this rate will not increase - even if you bond more than two channels together.

Please Note: If you are using the serial port for data and you are using ToolBox software for controlling your codecs, you will need to connect to a PC using either the LAN or USB port on the codec.

16.10. GSM Transparent and Non-transparent Data Modes

If you are using a GSM connection, we recommend you ensure that your SIM card has been set to send data in non-transparent mode. Check this with your telco when you order your data service. (Some cell-phone networks only provide non-transparent data capabilities). When transparent mode is used, the bit error rate could be as high as 1/1000. This will lead to dropouts in audio.

When data is sent in transparent mode you require sustained data flow to transfer data reliably. This is network dependent and if the network is unreliable there may be some loss of data packets, more bit errors (1/1000) and therefore dropouts in audio.

In non-transparent data mode, Radio Link Protocol (RTP) is used, providing better error correction and less bit errors (1/100,000,000). Data is buffered when it is being sent and received from a GSM module or cell-phone in this mode. This mode is more reliable for a data call and **Tieline** GSM modules only support the sending of non-transparent data. We also recommend that you use non-transparent data mode when you use a *Nokia*® 6310i cell-phone.

If your SIM card is set to transparent mode you will be unable to receive any calls to your cell-phone or GSM module. Contact your telco to check if your SIM card has been set to non-transparent data mode and get them to switch it if it hasn't.

16.11. DTMF Control of 3rd Party Devices

It is possible to control 3rd party devices using the DTMF control menu on your codec. To access this menu select **[Menu] > [DTMF Control]**.

To utilize DTMF control from a codec you require a POTS module connected to a POTS line. The POTS module needs to be in voice mode. To program this select the POTS connection that you wish to use for DTMF with the MENU SELECTOR. Next, select SOFTKEY 3 Wiz then press OK and scroll to and select Setup Voice Analog. Press OK to complete the configuration.

Please note: The other way to configure a plug-in module for voice mode is to select **[Menu] > [Configuration] > [POTS Left/Right Setup] > [Operating Mode] > [Voice Analog]**.

Plug the other end of the POTS connection into the device you wish to control.

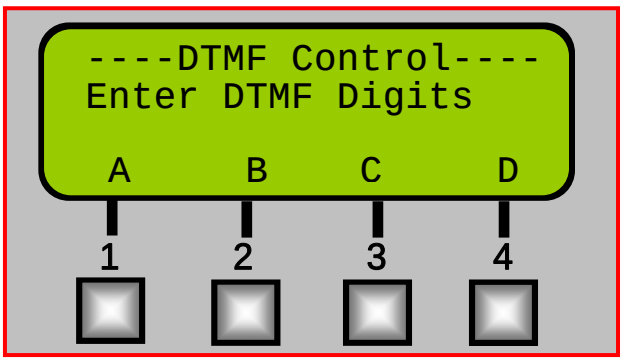


Figure 41: DTMF Control LCD Screen

16.11.1. DTMF Control via ISDN

An ISDN module can be used for DTMF control of a device if the module is used in ISDN voice call mode. To program an ISDN module for voice mode simply select the G.711 algorithm and the codec will make an outgoing analog call. This is automatically selected if an analog call is received by the codec.

Section 17. Relay and Control Port Operation

All **Tieline** codecs have two CMOS solid state relays for the control of equipment. They consist of two relay closures and two opto-isolators and can toggle or pulse. Relays behave in the same manner for all connection modes.

These two relays have been designated Control Port 1 and Port 2. They can be activated by User Function buttons which can be set via the *Functions* menu within **Tieline** Tool**Box** software. User Functions are activated by a sequence of the F1 or F2 button being pressed, followed by a number on the keypad. To read more about setting up the relay functions see the Tool**Box** *Control* and *Functions* menu sections within this manual.

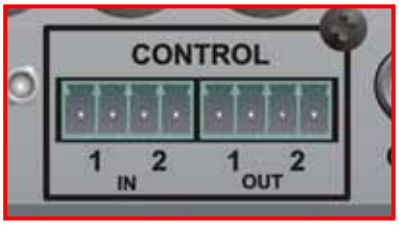


Figure 42: Control Ports on a Codec

As mentioned, Control Port 1 and Control Port 2 relays can be used in two different modes:

- Pulse: When activated the relay will close one second.
- Toggle: When activated the relay will toggle, i.e. if the relay is open, it will close. If closed it will open.

The Control Port 1 and 2 relay controls are assignable to any combination of FUNCTIONS buttons or to one of the four SOFTKEY buttons (or HOTKEY buttons on the ***COMMANDER G3*** rack unit codec front panel and TLG3 GUI, or the ***i*-Mix *G3***) - for faster one button operation. Please see the Tool**Box** *Functions* menu section within this manual about assigning the SOFTKEY or HOTKEY buttons.

17.1. External Relay Box

Each codec user profile includes a 50 byte per second Remote Control Channel[14] for remote control of external **Tieline** devices such as an external relay box. Up to four[15] external relay boxes, each with eight CMOS relays and eight input controls can be added to a system via the CAN port - this allows up to 32 external relays to be utilized.

CAN is a Multi-Node serial buss to which several **Tieline** products can be connected simultaneously each with their own unique ID address. Each opto-isolated input can be used to trigger a range of user programmable functions and can be connected to a relay closure or an open collector driver.

A master device can provide power or an external plug pack can be utilised where there are long cable runs. In addition, depending on the codec in use, current consumption will limit some products being powered from the buss. In

[14] *50 bps is the minimum data capability. ISDN, X.21 and IP connections may provide greater data bandwidth.*
[15] *Please check with us at support@tieline.com for updates on the implementation of this feature. Currently only one relay box can be connected.*

these cases a local power supply should be connected to the CAN 8+8. This should be 9 volts DC, 500mA.

An external relay box is connected via the CAN port on the rear of the codec.

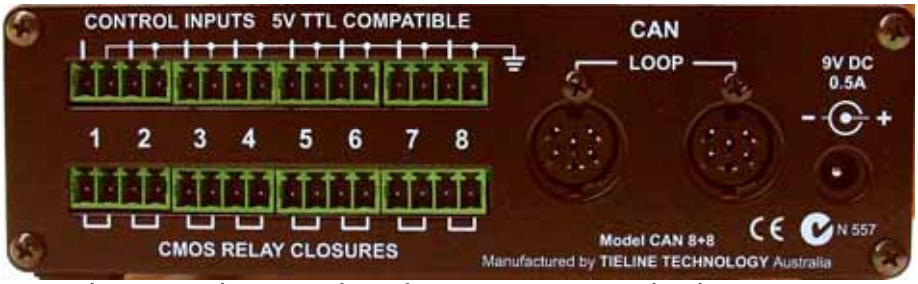


Figure 43: The Rear View of a CAN 8+8 External Relay Box

17.2. Cabling and Operation Distances

A number of factors determine the maximum length. Firstly, there is the CAN spec itself. This specifies a maximum length of 130m @ 500Kbps (the rate **Tieline** codecs use). The second factor, which is often the limiting one, is the voltage drop in the cable. Cable types, thickness, quality and length will all determine any limitations in a unit's performance.

Shielded dual twisted pair cables are recommended. Where cable runs are long and master power source is being used, heavier conductors will be required. Screw terminal plugs are provided to allow for easy on site wiring. These can be hot plugged without any impact upon a unit's performance.

In general, **Tieline** recommends that the relay box should operate effectively on a cable of 0.5 meters up to a maximum length of 75 meters from the codec. Distances greater than this could be achievable. Following is a formula for calculating maximum cable length:

- V in(min) at relay box = 8V (minimum input voltage);
- I (peak) at relay box = 250mA (peak current);
- V out(min) at TLM600 (*i*-**Mix** *G3*) = 11V;
- Max DC resistance (DCR) of cable = (11-8)/0.25 = 12 ohms.

Typically, 24AWG CAT5 cable has a DCR of about 0.095 ohms/meter. Therefore, maximum length = (12/0.095)/2; dividing by 2 accounts for the return path as well. In this example, the maximum theoretical length using this cable's specifications is about 63m.

A CAN cable wiring diagram is available in Appendix 1 of this reference manual. This section is titled CAN Cable Wiring Configuration.

17.3. CAN Bus Cable Terminations

It is a technical requirement for the CAN Bus to be terminated at each end. Where there are a number of units on the Bus it may be necessary to remove terminations from inside the CAN 8+8 box. For a small number of units on a local network this is often unnecessary.

Some **Tieline** products have the termination resistor included on the Bus by soldering a number of pins together in the connector. Please refer to the product manual in these cases.

For the CAN 8+8 resistors R12 and R13 are removed if needed. Keep in mind that a complete network requires at least two terminations and these should be at the end of the lines.

17.4. Inputs

The input signal is referenced to chassis ground. All eight ground terminals are connected together and to the chassis. The input device is an opto-coupler with a 680 ohm resistor in series with 5 volts. DO NOT feed voltages into the inputs.

Operation is as simple as jointing the input pin to the ground terminal. This can be via a remote relay contact or the open circuit collector of a transistor or FET.

17.5. Outputs

CMOS field effect transistors switch a low impendence path between the two pins when activated. These are opto-isolated and floating above ground. It is important to current limit the source as damage will result where the current exceeds 100mA. No more than 48 volts should be used as a safety precaution. The resistance of the CMOS element is approximately 18 ohms in the ON state.

17.6. Setup

A clear understanding of the units being connected to the CAN system is essential. If there are conflicts on the buss (units with the same node ID and/or operational capabilities) it is possible that the system will fail. All products must be set within the strict **Tieline** set-up protocols.

Internal DIP switches set the units node ID and operational features. These must be selected to the correct positions for operation to be possible. The unit is shipped with a default selection. If changes are required, disconnect power, remove the rear screws and grounding screw on the bottom of the box. Slide the PCB card out and make changes as needed. Ensure the grounding screw is secure when reassembling.

17.7. DIP Settings

Function (Sw 2,1)	Setting
OFF, OFF	First CAN 8+8 unit
OFF, ON	Second CAN 8+8 unit (future software feature)
ON, OFF	Third CAN 8+8 unit (future software feature)
ON, ON	Fourth CAN 8+8 unit (future software feature)
Function (Sw 4,3)	**Setting**
Switch 3 Off	Pulse Relay Mode
Switch 3 On	Continuous On Relay Mode
Switch 4 Off	Piezo Alarm Off
Switch 4 On	Piezo Alarm On

It may be necessary to set a particular mode of operation for the desired control system when used with different **Tieline** codec products. This should be done once the control system is determined in consultation with the main codec operation manual.

Normally, one CAN 8+8 will communicate with another one at the end of a link between two codecs. Please see the ToolBox ***Functions*** menu section within this manual about setting up codec User Functions and assigning the SOFTKEY or HOTKEY buttons.

17.8. CMOS Relay Operational Mode

As mentioned, the CMOS relays can be set to work in two ways. They can pulse ON for one second or they can stay ON until another command is sent to turn them OFF. The default mode is for a one second pulse. To change the relay operational mode, make changes to DIP Switch 3 as per the previous table.

17.8.1. CMOS Solid State Relay Specifications

These semiconductor devices provide a circuit closure when they are activated. They are rated to 350 volts peak across the contact closure and they have a maximum current rating of 120 milliamps. The contact closure resistance is typically 18Ω. (Related Topic: CMOS Solid State Relay Connectors)

17.9. Front Panel LED Indicators

The LED's on the front panel of the unit indicate the state of the relays and inputs. They also indicate a connection problem on the CAN buss by scanning back and forth.

17.10. Piezo Alarm

An internal piezo alarm is fitted to act as an audible indicator when either an input or an output function occurs. The default for this is off. It can be activated by changing DIP switch 4 settings per the table provided.

Section 18. ToolBox Operation

This program will enable you to configure **Tieline** codecs before and during field deployment. ToolBox software is used to create and load new custom user profiles onto codecs. Profiles you create are saved in configuration files with a .cdc file extension.

With ToolBox software you can configure connections, change codec settings, upgrade codec software, store telephone numbers, operate local and remote CMOS relays and much more. It is also capable of controlling both local and remote codec input levels while on air - from either end of the link.

18.1. ToolBox Software Downloads

ToolBox is a PC compatible software application that works with all **Tieline** codecs, providing a convenient way to configure, remotely operate and monitor them. Free updates of ToolBox software can be downloaded from the **Tieline** web site at http://www.tieline.com/support/software.html. To determine the version of ToolBox software you require for your codec, go to our website at http://www.tieline.com/support/software.html and check for the most suitable version to upgrade your codec.

If you are not sure what version of ToolBox software you should be running, regardless of whether you have a new G3 codec or an older G1 legacy codec, simply navigate to **[Menu] >[Unit Details]** on your codec. The firmware version is listed in this menu. Compare this to what version of ToolBox is currently available for downloading. Download the appropriate version of ToolBox to your PC and if you require an updated version of codec firmware, fill out the Upgrade Request Form on our website at http://www.tieline.com/support/upgrades.html.

18.1.1. Operating System Requirements:

We recommend that ToolBox software be run on one of the following operating systems: Windows®[16] XP or 2000.

Please Note: This software is not available for Apple®[17] computers and Windows® 95 is not supported.

ToolBox software requires Internet Explorer® 5.5 or higher.

Please Note: Netscape®[18] is not supported.

[16] *Windows® is a registered trademark of Microsoft Corporation in the United States and/or other countries.*

[17] *Apple® is a trademark of Apple Computer, Inc., registered in the U.S. and other countries.*

[18] *Netscape® is a registered trademark and the exclusive property of AOL Time Warner, Inc., or their respective subsidiaries, divisions and/or related companies.*

18.1.2. Hardware Requirements

As a minimum, ToolBox software requires a system with:

- Intel® Pentium® 500 MHz (equivalent or higher);
- 64Mb of free RAM;
- 5 MB of hard disk space; and
- A free USB or serial port.

18.2. Preparing to Use ToolBox Software with Your Codec

ToolBox software is distributed as a compressed file ".zip" file. Extract the program file to the PC hard drive using a program such as WinZip. Click on this hyperlink to download Winzip or visit www.winzip.com.

Once you have unzipped and saved the file, go the where you have saved it and double-click on the TieTool.exe file to open the program. The program will appear as displayed in the following image. For the procedure to upgrade you codec's software, see the Updating Firmware section in this reference manual.

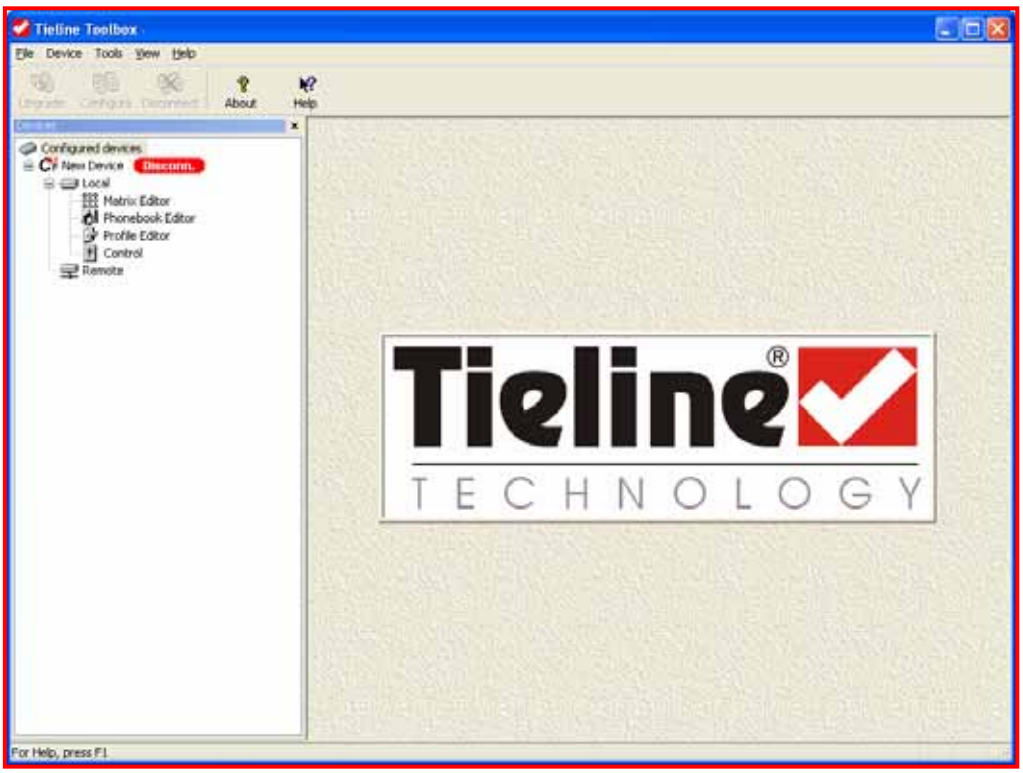


Figure 44: Main Menu Screen of the ToolBox Program

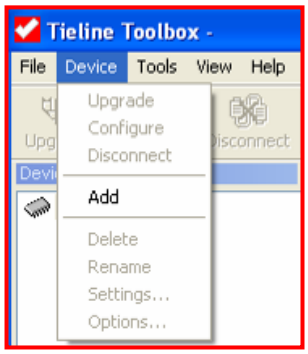
18.3. Connecting Your Codec to a PC

A **Tieline** codec serial port connects to any COM port on the PC using a standard modem cable. If you don't have a serial port on your PC, you can use a USB to serial port adapter cable to connect.

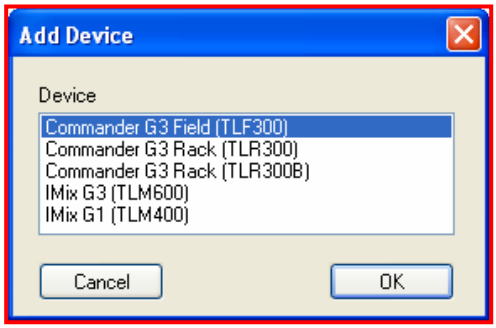
In addition, it is also possible to connect between a USB port on a PC and the USB port on the rear of your codec. Please see the Connecting via USB section of this manual for more details.

18.4. Configuring ToolBox and Your Codec to Work Together

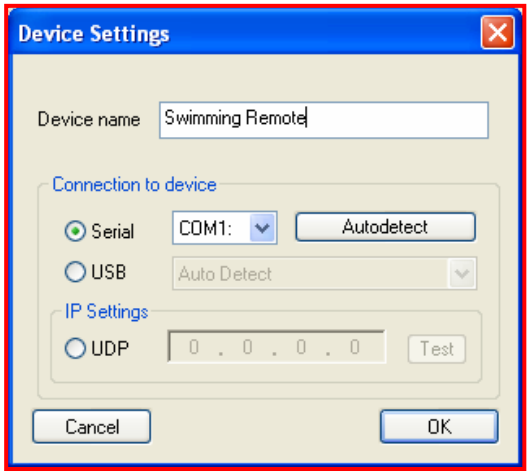
Before you can communicate between your codec and your PC successfully, you must ensure that both your codec and your ToolBox software are configured correctly. The COM port is set up by ToolBox. When you first connect to the codec, in order to operate ToolBox effectively you will first need to add the codec to your *Devices* menu tree. The steps to configure ToolBox successfully are shown below. Please Note: ToolBox will automatically connect to a PC at the highest bit rate possible.



1. Open ToolBox and click on the ***Device*** drop-down menu.
2. Click on *Add* which will open the *Add Device* menu.



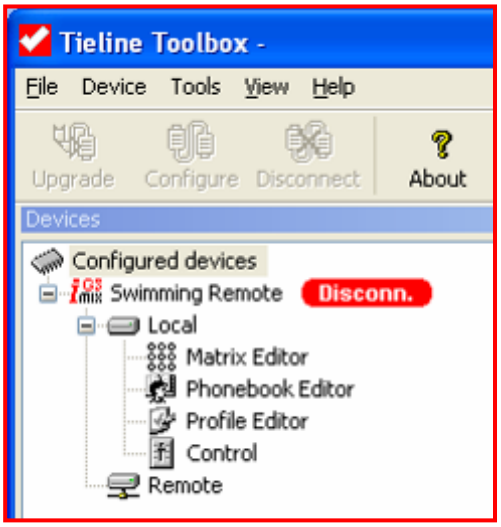
3. Click on the device you wish to add and then click OK which will open the *Device Settings* menu.



4. Name the device in the *Device name* section of the menu, in this example we will call it Swimming Remote.
5. Select the relevant connection from *Serial, IP* or *USB*.
6. If you are connecting via the serial port, select the COM port that you are connected to (if you are unsure click on *Autodetect* and your ToolBox software will detect this automatically for you).

7. Please note that if you are connecting via a USB connection you will also need to ensure the correct drivers for USB connection are installed on your PC. If you do not have these drivers please contact us at support@tieline.com

8. Press *OK* and your device will be added to the ***Configured devices*** root menu as shown. Once you have added your codec to this list, it becomes a local codec. Various codec function menus can be accessed by clicking on the options for the ***Matrix Editor***, ***Profile Editor***, ***Phonebook Editor*** and ***Control*** menu. The remote codec's function menus can also be accessed by clicking on ***Remote*** in this menu screen once you are connected to it.



18.5. Connecting ToolBox via USB

If you are connecting via a USB port to the codec you will need to ensure you have installed the appropriate USB software drivers to connect successfully. (If you don't have the appropriate USB drivers please contact us at support@tieline.com). Please follow these steps to successfully install the drivers:

1. Before you begin, ensure that you have the appropriate USB drivers saved in a file on your PC.
2. If you are using software such as Windows® XP on your PC, connect the USB cable to your PC before you power it up.
3. When the computer powers up the "New Hardware Wizard" should recognize the codec is connected and ask you to either install driver software automatically or from a specific location. Choose to "install from a list or specific location".
4. Next choose "Search for the best driver in these locations" and check the box for "Include this location in the search". Click "Browse" and then go to the file in which you have saved the driver software on your PC.

5. Once you have selected the appropriate file, click “Next” and the wizard should find and install the software for the codec. Open ToolBox and click on the Device drop-down menu.
6. Click on *Add* which will open the *Add Device* menu.
7. Click on the device you wish to add and then click *OK* which will open the *Device Settings* menu.
8. Please note that if you are connecting via a USB connection and you have only one device connected to your codec, the *Auto Detect* setting will automatically select the device you are connected to. If you have more than one device connected, you will need to select the codec you wish to connect to from the drop-down menu as displayed. You will also need to ensure the correct drivers for USB connection are installed on your PC. If you do not have these drivers please contact us at support@tieline.com .

IMPORTANT: Each time you connect a new codec to your PC via USB you will need to reinstall the USB software drivers. This is because the driver uses the hard coded serial number of the codec as an identifier.

If you don't see the device driver in the drop down list next to USB selection in *Device Settings*, you will not be able to connect to the codec via USB. This could indicate a driver problem.

To check whether the USB driver is installed correctly, with the codec powered and connected to USB go to "My Computer... Properties... Hardware... Device Manager" and look in the USB section down the bottom and see if there is a Codec entry. This indicates that the driver is installed correctly and it should appear in the list.

If don't find a "Tieline Codec" and you see a yellow exclamation mark in "USB" or "Other Devices" then you should re-install the driver. To do this you have to right click on yellow exclamation device, select Update Driver and select driver, etc.

18.6. LAN Connection: Static, DHCP and BOOTP IP Addresses

If you are connecting ToolBox software to your codec via a LAN IP connection you will need to input an IP address into your ToolBox software. Static addresses can be entered manually. Alternatively, many networks have automated DHCP or BOOTP IP address systems.

18.6.1. Automated DHCP and BOOTP LAN Connections

First connect your network cable to the LAN port on your codec and another to your PC. If you are connecting via a DHCP network, your codec and your PC should be automatically detected and receive an IP address from the network. If the network is using a BOOTP IP addressing system, then both the PC and the codec should still be automatically detected and receive allocated IP addresses.

Once the two devices have received addresses (both the codec and PC), input the IP address of the codec into the *Device Settings* menu - as shown in the previous image. To find out the codec IP address simply navigate to **[Menu] > [Unit Details]** on your codec and scroll to the IP address. If the codec is still obtaining a DHCP address it will display the message Requesting IP….

18.6.2. Static Addresses: Connecting via a Crossover Cable Connection

If you are connecting your PC directly to your codec via a crossover cable, you will need to setup the codec and your PC using a static IP address. Follow the instructions in the next section to do this.

18.7. Inserting Static IP Addresses into a Codec and PC

To manually insert a static IP address into your codec and your PC please do the following. (Please note that the IP address of the codec **must be different** to the IP address of the PC)

18.7.1. Adding a Static IP Address Manually into the Codec

- First connect a cable from your PC to the LAN port on your codec;
- Then access the Menu submenus by pressing SOFTKEY 4;
- Next use the MS to select the Configuration Menu;
- Then use the MS to select the Advanced Menu;
- Next scroll with the MS and select LAN Settings;
- Then scroll with the MS and select IP Setup;
- Next scroll with the MS and select Static;
- Then select the IP Address using the MS;
- Next, press CLEAR to delete any old IP address details and input the new static IP address using the codec keypad (This number needs to be unique if you are on a network and there is no DHCP server. We suggest using the following number as shown above for your codec 192.168.0.1);

- Once it is set correctly, press SOFTKEY 4 or the MS to set the number into the codec;
- Then select Subnet Mask in the same menu by scrolling with the MS. We suggest you input the following number in this section 255.255.255.0;
- Once you have programmed the Subnet Mask number into the codec in the same manner as the IP Address, adjust the Default Gateway if required. Press SOFTKEY 4 OK to confirm the new details.
- Your codec is now configured. You must now set up your PC to enable it to communicate with the codec.

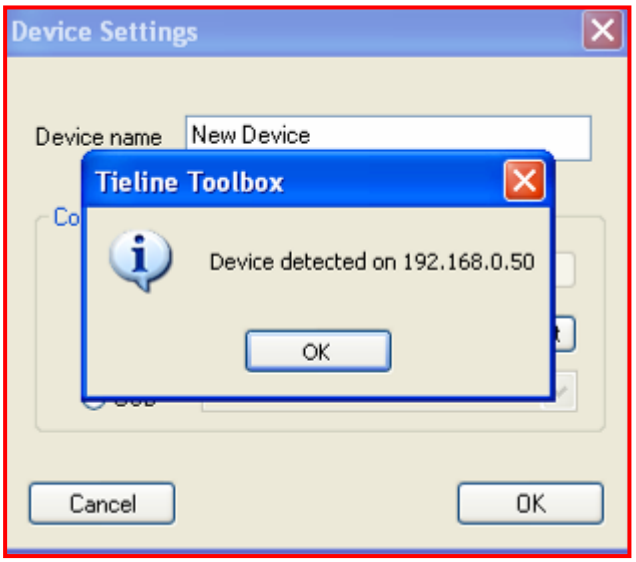
18.7.2. Adding an IP Address Manually into your PC

The following information relates to connecting with a PC using Windows® XP.

- First connect a cable from your PC to the LAN port on your codec;
- Then click on the 'Start' menu on your computer and select the 'Control Panel';
- Then select 'Network Connections';
- Next select your codec 'LAN connection' by right-clicking on it and selecting 'Properties';
- Then select 'Internet Protocol (TCP/IP)' by clicking on it once with your mouse and then click on 'Properties';
- Next select 'Use the following IP address' from the menu and in the relevant section input the IP address for your computer (This number needs to be unique if you are on a network and there is no DHCP server. We suggest using the following number, which is one number on from the IP address inserted into the codec previously. This number is 192.168.0.2);
- Then in the 'Subnet Mask' section input the same address that you have put into your codec, 255.255.255.0;
- Then click on *OK* to save the changes and now both your PC and your codec should be configured to communicate with each other.

18.7.3. Testing ToolBox IP Connections

Once the IP and Subnet Mask addresses have been either automatically detected, or manually inserted, press the *Test* button in ToolBox to ensure that the connection has been detected. If there is a problem detecting it, recheck that you have the correct numbers in the IP address and Subnet Mask sections in both the PC and the codec.

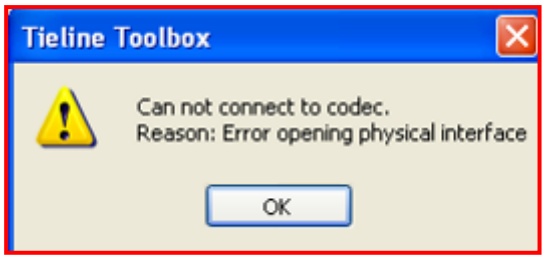


If you are still having problems connecting, please see your resident IT professional for more information.

18.8. Serial Port Connections: Configuration at the Codec Itself

If you are using the serial port to connect with, it is also necessary to configure the codec serial port itself. Once you have initially powered up the codec, select the Menu option from the startup LCD screen as shown. This is done by pressing SOFTKEY 4.

Then use the MENU SELECTOR (MS) to scroll to and select **[Configuration] > [Advanced Menu] > [Serial Port Mode] > [Toolbox]**.



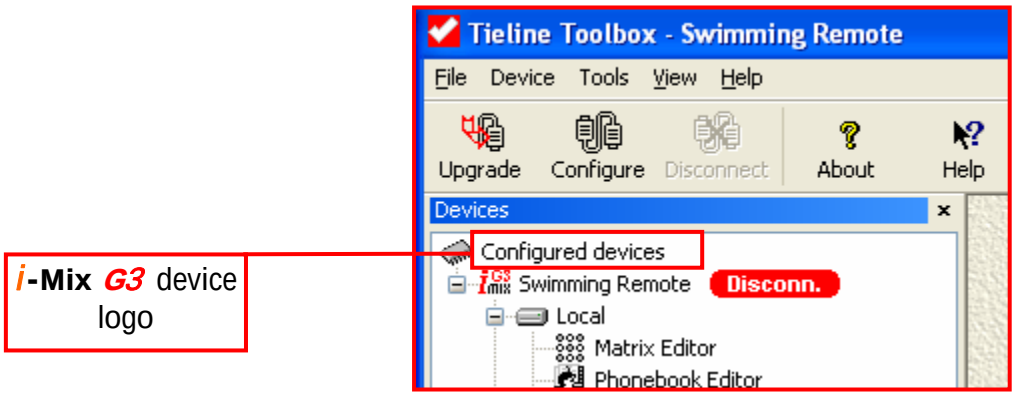
Your ToolBox software and your codec will now be configured to 'talk' to each other. If this message appears on your PC when you are trying to connect your codec to ToolBox software, you have not configured your codec correctly and you will need to check that you have selected the appropriate connection type i.e. Serial, LAN or USB. It could also be a good idea to check that the cables are connected and seated properly. If you are still having problems connecting, it could be a good idea to reboot the codec and try again.

18.9. Connecting your Codec to Toolbox Software

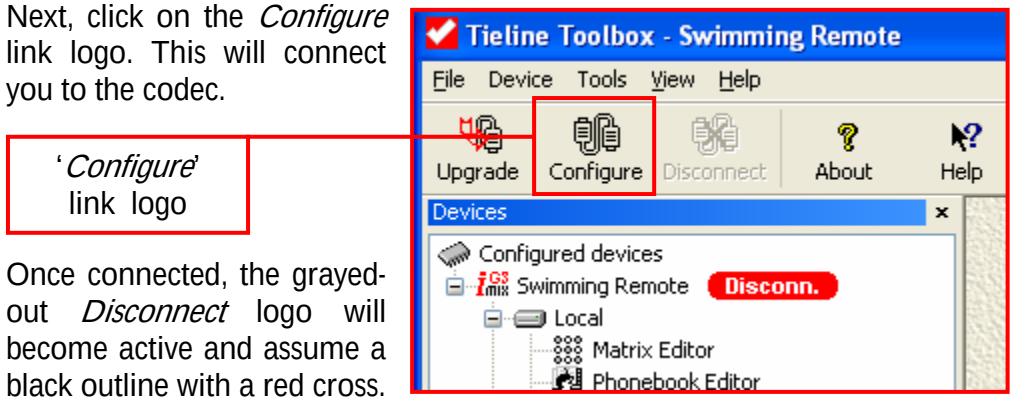
Connection of your codec to a PC with ToolBox is possible via a USB, LAN or serial cable. After ensuring the cable between your computer and your codec are connected properly, you will also need to determine if the firmware version on your codec is the same RPTP version as the version of ToolBox that you are attempting to connect with. This should be displayed in the file name of the software. Older versions of software may not display the RPTP version number. For more information on software compatibility, please see the Updating Firmware section within this reference manual or visit the software support page on our website at http://www.tieline.com/support/software.html .

Once you have the correct software installed on your PC and in your codec, you can initiate connection between your codec and your computer by following these steps. (In the following example we are using an *i*-Mix *G3* codec but the connection procedure is the same for all **Tieline** codecs).

1. Click on the device logo for the codec in the *Configured devices list* that you wish to connect to. This selects the device as the one that you will connect to.



2. Next, click on the *Configure* link logo. This will connect you to the codec.

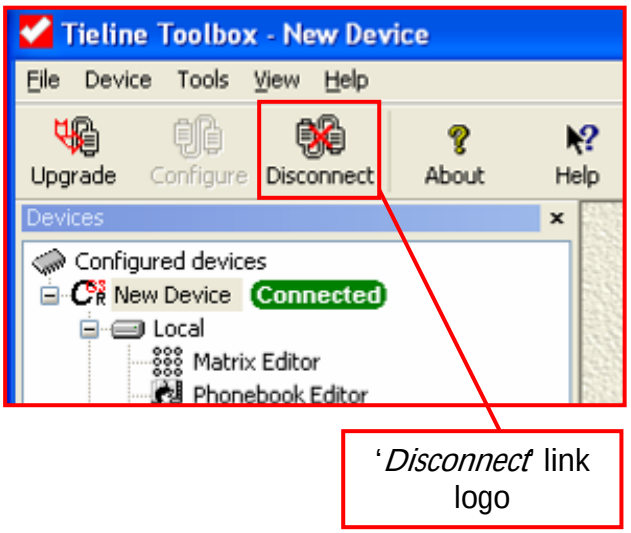


Once connected, the grayed-out *Disconnect* logo will become active and assume a black outline with a red cross.

The codec is now able to communicate with ToolBox on the PC. The red disconnected *(Disconn.)* device goes to a green *connected* mode. You can change the settings on your codec and any remote units to which your local codec is connected via ToolBox software.

(If an error message appears during the connection process, please see the Toolbox Connection Error Messages section that follows.)

3. Simply click on the *Disconnect* logo to disconnect ToolBox from the codec.



If you have connected to a PC and are ready to dial using a manual default profile, please see the Quick Start section in this manual for the startup procedure for these profiles. If you are using custom designed profiles or are having trouble dialing the remote codec, please see the Operation of your Codec section within this manual.

18.9.1. ToolBox Connection Error Messages

If the following error messages appear you will need to download the appropriate RPTP firmware version or the appropriate RPTP ToolBox version so that they match.

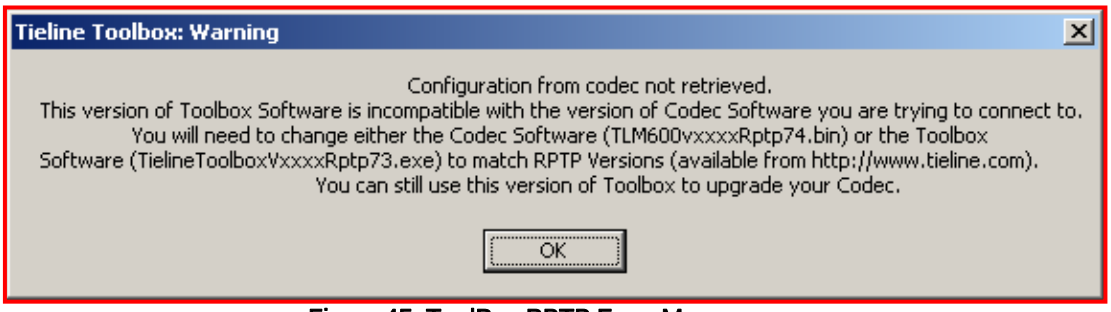


Figure 45: ToolBox RPTP Error Message

In the previous example, the RPTP versions of both the codec firmware and ToolBox are displayed to enable you to easily determine which RPTP version of software you will need to download.

In the next example, an older version of ToolBox without an RPTP version is installed on the PC. You will need to check the software support page on our website at http://www.tieline.com/support/software.html to determine what version of codec firmware will work with your version of ToolBox. Alternatively, you can download a newer version of ToolBox to match the codec software you have installed.

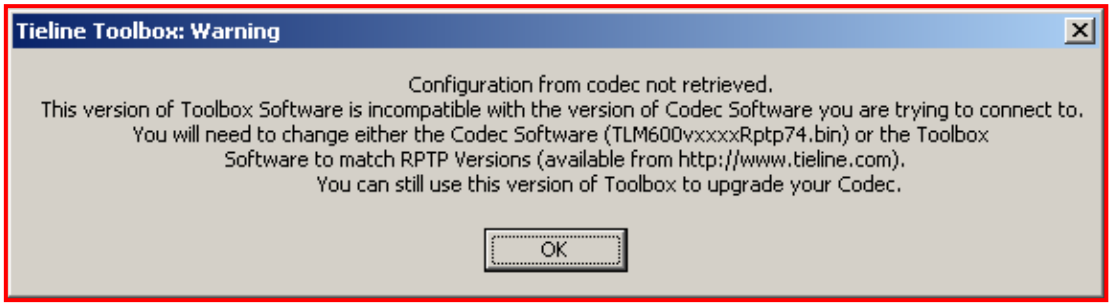


Figure 46: ToolBox Incompatibility Error Message

18.10. Updating Firmware

From time to time it may be necessary for you to update the firmware on your codec. This is a fairly simple procedure. To check for the latest software available, and to download ToolBox, please visit the software support page on our website at http://www.tieline.com/support/software.html

If you require new codec firmware, click on the "I need an upgrade" link on the software support page and complete the Software Upgrade Request Form.

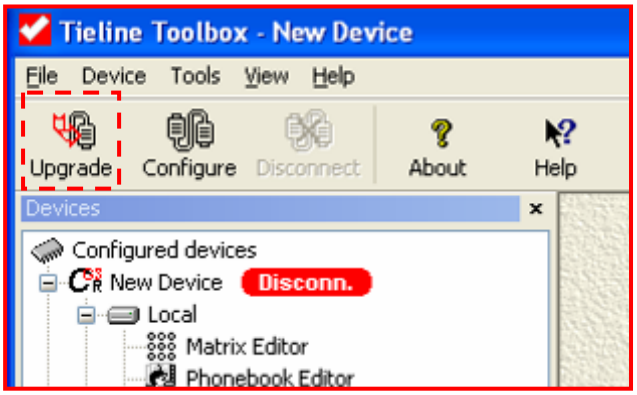
If you are upgrading or downgrading the firmware version in your codec, you will require the matching version of ToolBox software for your firmware - if you wish to use ToolBox to configure your codec. You can match the version of firmware and ToolBox software by checking the RPTP version in the file name for each software package.

For example, if you are using ToolBox software v.XXXX RPTP81, then the firmware version you install in your codec must be an RPTP81 version or ToolBox will not be able to communicate with your codec.

If you are simply wishing to upgrade the firmware in your codec, you can use any version of ToolBox to do this.

To find out what version of software you are running on your codec, go to the Unit Details menu item on the codec. Unit Details will list all the programs installed along with their versions. To scroll down the list simply rotate the MENU SELECTOR. The firmware updating process is as follows (please note that in this example we are upgrading a ***COMMANDER G3*** rack unit codec but the principles apply to all **Tieline** codecs):

1. Once you have attached your codec to a PC via a USB, LAN or serial cable, open ToolBox and you will see the *Upgrade* button on the software toolbar (there is no need to 'connect' your codec to ToolBox software). Press this button and the *Firmware upgrade* window will open.



If you try to upgrade firmware while ToolBox software is connected to the codec, the following error message will appear. Click yes to disconnect the codec and proceed with the software upgrade.

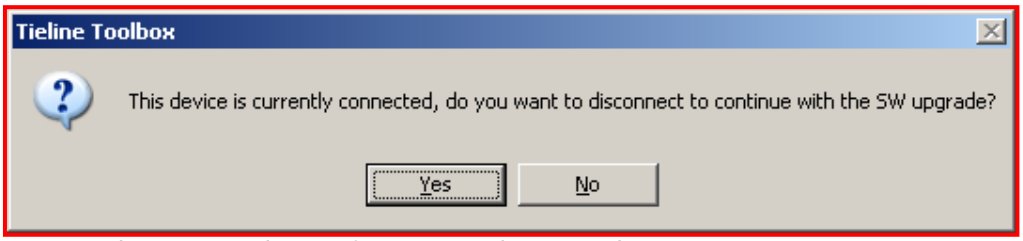
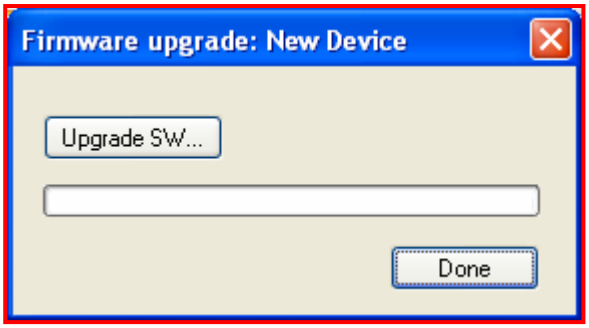
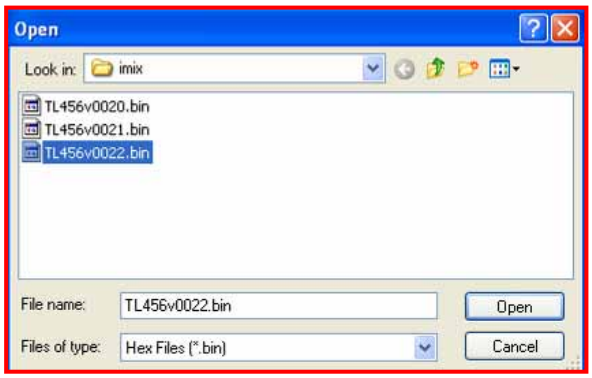


Figure 47: ToolBox Software Upgrade Connection Error Message

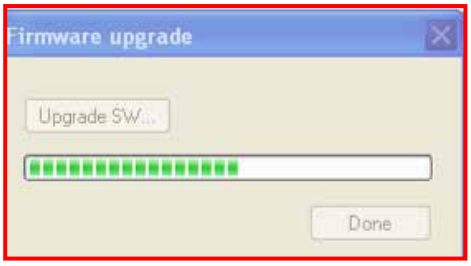
2. Click on *upgrade SW...* to upgrade the software.



3. Click on the *.bin file that you wish to load into the codec and then click on *Open*. (If an error message appears during this step please see the Software Upgrade Error Messages section that follows.)



4. The file loading process will continue and progress can be monitored by viewing the green update bar. The update process can take several minutes.

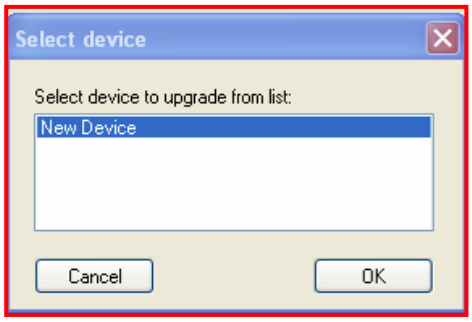


5. Once the update has been successfully completed a new message will advise you of this. Click *OK* and proceed with using the codec. The codec will automatically reboot itself after upgrading.

Once you have completed the upgrade process, you are advised to set the system defaults on the codec. To do this press SOFTKEY 4 to access the Menu submenus, then scroll down to **[Reset Functions] > [Set Sys Defaults] > [All Profiles] > [Yes]**. This will return the codec to default profile settings.

18.10.1. Upgrading Via the Tools Menu

You can also initiate the firmware upgrade process by clicking on the *Tools* drop-down menu in ToolBox and selecting *Firmware upgrade.* The process of upgrading is very similar to using the *Upgrade* button in the toolbar. The only difference is an additional step when you first select *Firmware upgrade*. The *Select device* window will open and ask you to select the device you wish to upgrade. Simply select the device you want to connect to and click *OK*.



Remember, once you have completed the upgrade process, you are advised to set the system defaults on the codec. To do this press SOFTKEY 4 to access the Menu submenus, then scroll down to **[Reset Functions] > [Set Sys Defaults] > [All Profiles] > [Yes]**.

18.10.2. Software Upgrade Error Messages

Some error messages may appear during step 3 of the upgrade process. If the *.bin file that you select to load into the codec is a different RPTP version to the ToolBox RPTP version you have on your PC, the following message will appear:

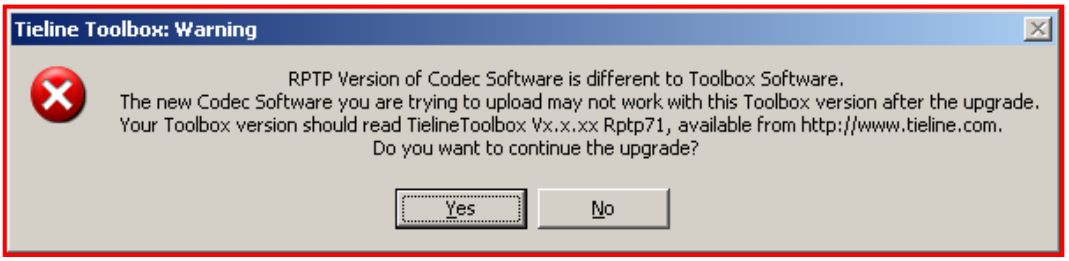


Figure 48: RPTP Error Messages

If the *.bin file that you select to load into the codec is an old firmware version without an RPTP version number, the following error message will appear:

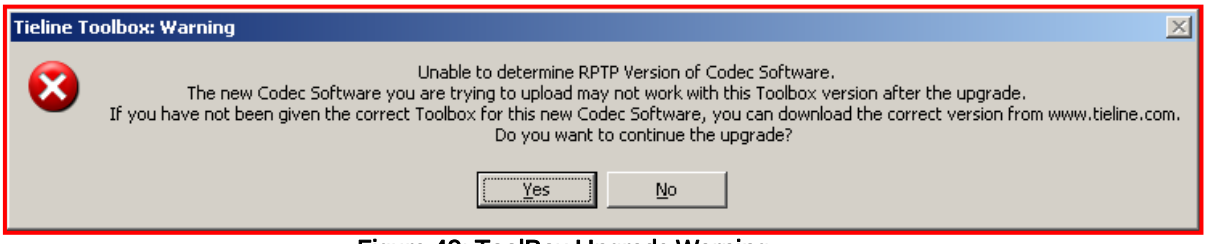


Figure 49: ToolBox Upgrade Warning

With all of these error messages, you can proceed with the upgrade process and download the matching version of ToolBox by visiting our website's software support page at http://www.tieline.com/support/software.html.

Firmware Upgrade Troubleshooting Tip: If you are having trouble performing the upgrade, try powering the codec down and then up again. If you are upgrading via a USB connection, check to see that the USB cable is seated well in the codec.

Section 19. Configuration File System

One of the great new features of the **Tieline** ***G3*** platform is the Configuration File system. It enables a user to configure a codec before they arrive at a remote broadcast site, minimizing the amount of adjustments a broadcaster has to make when arriving at the site. One of the last things that an announcer wants to do is to have to make last minute adjustments to their equipment before they are due to go 'on-air'. ***G3*** codecs are designed to eliminate that inconvenience completely.

Configuration Files are created by the ToolBox PC application, supplied with every ***G3*** codec. This software application has comprehensive editing facilities to enable the user to configure their codec for every remote broadcast that they are likely to do. Each Configuration File can contain:

1. Default and User Created Profiles,
2. Default and User Created Routing Matrices, and
3. A Phone Book.

19.1. The Relationship between Configuration Files, Profiles and Matrices

Tieline codecs operate in pairs. Typically, one will be a field unit codec at a venue and the other will be in the studio. The studio codec could be a field unit or rack unit codec. It could also be a rack unit codec with a GUI (Graphical User Interface).

Codecs are programmed with Configuration Files which contain 'Profiles'. These profiles contain code that programs a codec to perform as either the 'local' or 'remote' codec. In other words, it defines how a ***G3*** codec is configured. Each Configuration File contains up to 36 unique profiles – each one can configure the codec for a different broadcast situation. There is a set of default profiles in each **Tieline** codec that will suit many broadcast applications.

Each profile can be individually tailored to suit a user's requirements. It can then be assigned a name that relates to the remote broadcast site or event that it is to be used for. Using profiles to manage a codec's operation enables a user to go to a remote site, plug in the codec and simply press START to connect to the studio. It can't get much simpler than that.

Within each profile is a set of matrices that define the routing of audio throughout a codec. There is a set of default matrices in each codec and users can adjust these and create new custom matrices to suit individual broadcast requirements. The following image displays how this system works.

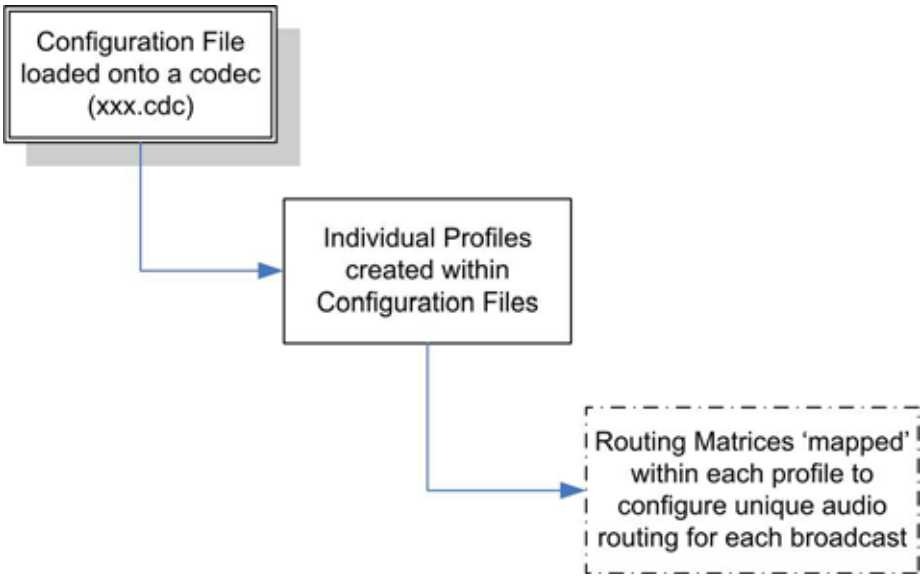


Figure 50: Elements within a Configuration File

19.2. Different Codec Configuration Files

Configuration files differ between different codecs, i.e. an ***i*-Mix *G3*** versus a ***COMMANDER G3*** (or a ***COMMANDER G3*** rack unit codec and a field unit codec) etc. This is because profiles, and the matrices within them, differ between different codecs depending on the routing functionality available in a particular codec.

If you use different **Tieline** codecs it is important that you try to load appropriate configuration files into each codec you are using. When you create new configuration files it is a good idea to clearly identify the file as being for a particular codec. This will assist you to load the correct file onto your codecs.
For more detailed information on codec matrices and profiles, please read the following sections of this reference manual titled *Matrix Editor* and *Profile Editor*.

Very Important Note:
If you connect using manual default profiles, during the connection process the dialing codec will ensure the correct profile is loaded in the remote codec.

However, when you are not using manual default profiles (i.e. you are using specific profiles you have created), different profiles need to be loaded onto both the local and remote codecs. These profiles will reflect the fact that each codec's matrices are set to send and receive different audio streams.

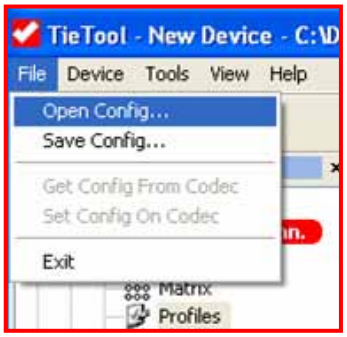
For example, it is important to remember that local and remote matrices need to be matched if features such as talkback are to be used. Also, if the local codec is in the Broadcast Center, the matrix settings on the codec will reflect the fact that it is receiving program audio from the remote or field codec, whereas the remote codec will be set to send program audio. Check that the talkback matrices of both the local and remote codecs are configured correctly. (Related Topics: Manual Default Profiles, *Matrix Editor*, *Profile Editor*, The LCD Screen on Initial Powerup)

WARNING: Please note that if you want to change profiles you need to hang up the codec before making the change. Then load the new profile that you require and redial to make a connection.

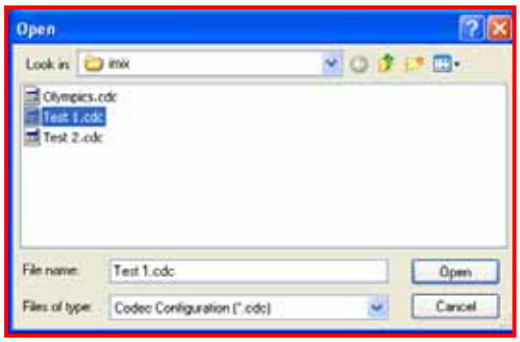
19.3. Set, Get, Save and Open Functions of Configuration Files

All of these functions have their commands displayed in the drop-down box from the file menu in ToolBox software.

To open a Configuration File you that you have created and saved on your PC, first open ToolBox software. Then click on *File* in the top left corner of the ToolBox main menu. A drop-down box will appear as shown below.



Click on *Open Config,* which will open a new menu showing the Codec Configuration files you have saved on your PC. These files have a *.cdc file extension. Select the file you wish to open and click on it once, in the example below we are selecting the file *Test 1.cdc*. Now click on *Open* to open the file. Your settings for this file are then loaded into ToolBox and you can view, adjust and save any changes you make. In addition, you can download the profiles onto a codec.



To save a new or amended Configuration File simply click on *File* in the top left corner of the ToolBox main menu. A drop-down box will appear and then click on *Save Config.* This will open a new menu screen allowing you to name the file and then click save to save it.
The complete list of the configuration file menu functions is explained below.

Open Config	Opens a Codec Configuration *.cdc file from on the PC
Save Config	Saves the Configuration settings into the PC as a codec configuration *.cdc file
Get Config From Codec	Loads the Configure settings into the PC from the codec.
Set Config On Codec	Writes the displayed Configure menu to the codec from the PC.

Table 8: Configuration File Menu Options

A discussion of how to adjust all of a profile's individual settings with the *Matrix Editor* and the *Profile Editor* follows.

Section 20. Matrix Editor

20.1. An Overview

In each ***COMMANDER G3*** field unit Configuration File there is a set of routing matrices used by the various profiles defined within the file. Basically, the matrices tell each codec how to route audio throughout the unit.

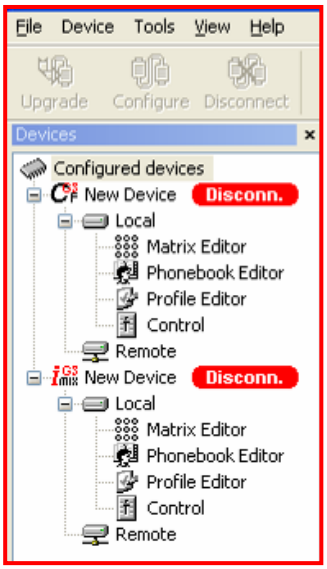
There can be up to 160 user matrices in a Configuration File. Each profile has 10 routing matrices that have to be defined by the user. Matrix mapping is the process used to assign a matrix to each routing function within a profile. This is discussed within the *Matrix Map* section of this manual.

20.2. How do I use Matrices in the Codec?

The process of creating, saving and loading profiles begins with the *Matrix Editor*. Once you have used the *Matrix Editor* to create the audio routing matrices for your profile, then you can 'map' the matrices in the *Matrix Map* section of Tool**Box**, finalize your connection settings for the profile and save it in a configuration file. This is then loaded onto your codec and you are ready to connect – it's as simple as that.

Very Important Note on Matrices:

It is very important to note that ***i*-Mix *G3*** *and* ***COMMANDER G3*** *TLR300 rack, TLR300B rack plus and field unit matrices are different to each other. This is because each codec has slightly different audio routing capabilities. Therefore each codec has different audio matrix routing requirements.*



If you are connecting an ***i*-Mix *G3*** to a ***COMMANDER G3***, (or a ***COMMANDER G3*** rack unit codec to a field unit codec), you will need to create separate configuration files with profiles and matrices for each codec using the correct *Matrix Editor* for each codec. To do this you will need to add a new configured device, i.e. an ***i*-Mix *G3*** and a ***COMMANDER G3*** codec if you are connecting these two codecs, to the *Configured Devices* menu in Tool**Box** (as displayed in the right-hand image that follows).

Once you have done this, you can then select the *Matrix Editor* for each codec and amend, create and save matrices to a profile as required. For more information on ***COMMANDER G3*** matrices, please consult the ***COMMANDER G3*** reference manuals. (Related Topics: *Matrix Map*, *View Matrix*, Preparing to Use Tool**Box** Software with your Codec.)

20.3. Routing Matrices Explained

At the heart of **Tieline** codecs is a digital routing switcher. It enables the codec to be configured for a whole range of possible broadcast applications, particularly where different configurations for inputs, outputs and monitoring are required. It also enables the user to configure for POTS, ISDN, GSM, 3G, X.21, satellite ISDN, IP/Internet, single channel or dual channel operation. There may also be a requirement for communications as well. The router can even configure full talkback operation between a studio producer and the remote broadcast codec.

Each codec profile contains 10 programmable routing matrices (as displayed in the Matrix Map menu), that enable the user to program audio routing for normal program, cue functions, remote cue, phone cue, local talkback and remote talkback. Two separate 'Global' matrices for muting program and monitoring are also provided but are not included in the profile.

All ***COMMANDER G3*** field unit routing matrices have six inputs - four are hard-wired and two are virtual. Of the four wired inputs, two are balanced mic/line inputs, one is a phone input (used for monitoring dial tones when dialing and for use as a phone input via a POTS module) and there is an auxiliary input. The two virtual inputs are for the two decoder channels.

There are 12 outputs in the router of which 8 are wired and 4 are virtual outputs. Of the 8 wired outputs, 2 are balanced outputs, 4 are for headphone monitoring, one is for a phone connection (via a POTS module) and one is for the *Aux* (auxiliary) output. Of the 4 virtual outputs, 2 are for the *Encoders* and 2 are for the *PPM* meters.

	Encoder 1	Encoder 2	CH 1 Out	CH 2 Out	HP1 L	HP1 R	HP2 L	HP2 R	Phone Out	Auxilary	PPM 1	PPM 2
Input 1												
Input 2												
Aux												
Phone												
Decoder 1												
Decoder 2												

Figure 51: The 6 inputs and 12 outputs of the field unit routing matrix.

A set of reference matrices is provided with the ***COMMANDER G3*** to enable the user to have a starting point for setting up the codec. Other matrices can be created and amended by the ***Matrix Editor*** if you have purchased this option from **Tieline**. The main menu is shown below.

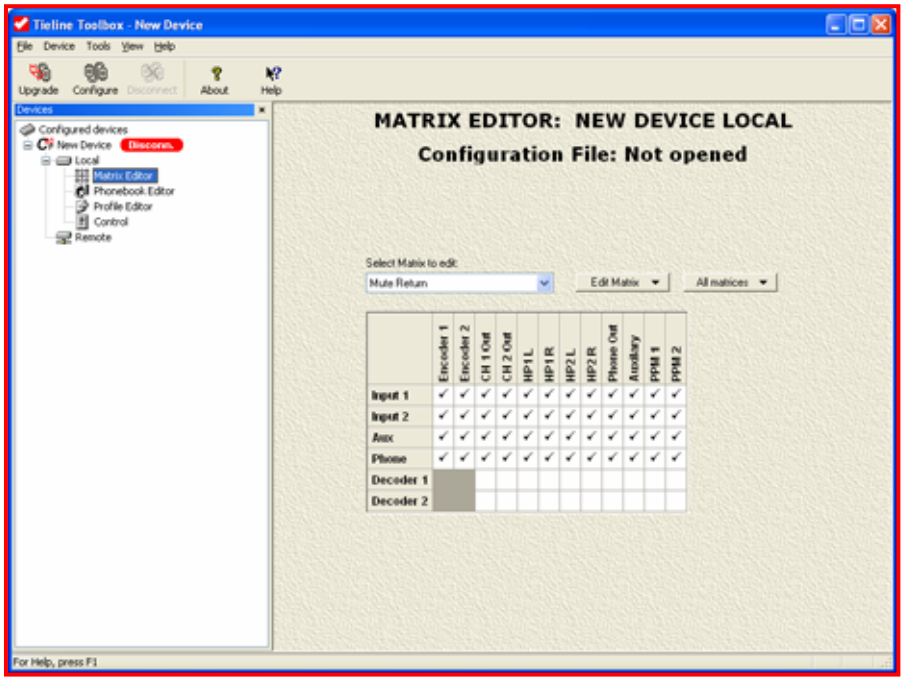


Figure 52: The Matrix Editor Main Menu Page

20.4. Activating the Matrix Editor

If you have purchased the *Matrix Editor* option, to activate it within the codec you will need to input your 12 digit key into it to enable matrix functionality. To do this please do the following:

- First, you will need to access Matrix Enable menu within the codec.
- Press SOFTKEY 4 on the codec to access the Menu submenus;
- Scroll using the MS to Configuration and press the MS to select it;
- Then scroll to Advanced with the MS and select this option;
- Scroll to Matrix Enable and select this option;
- Select Enable and then enter the code that you have been given by **Tieline**.

Your codec will now be enabled for matrix functionality.

20.5. Matrices –Where Do I Start?

The *Matrix Editor* is the first stage of a three stage process that allows you to set the routing of mixer functions within profiles for the ***COMMANDER G3***. The three stage process is as follows:

1. Set the cross point connections for your matrices within the *Matrix Editor*. This is where you set the individual signal path routing for each mixer function of a profile within the codec; i.e. cue and talkback functions and the inputs.
2. Select which matrix will be attached to each mixer function listed within the *Matrix Map* menu and save the settings to a profile.
3. View and check your matrix settings within the *View Matrix* menu. This allows you to view the overall mixer settings you have made within the codecs.

Connection Tip: When you connect two codecs using manual default profiles, the dialing codec will instruct the codec that receives the call as to which profile should be active in the codec. If you are using other user created profiles that use matrices that you have created yourself, you need to ensure that you have the correct profiles (with matching 'mirror-image' matrices) loaded into both codecs, along with the correct talkback and connection settings, i.e. *Auto answer* and *Automatic Redial* set correctly.

Operational Tip: It is important to remember that local and remote matrices need to be matched if features such as talkback are to be used. For example if the local codec is in the Broadcast Center, the matrix settings on the codec will reflect the fact that it is receiving the program audio from the remote or field codec, whereas the remote codec will be set to send program audio. Check that the talkback matrices of both the local and remote codecs are configured correctly. (Related topics: Manual Default Profiles and The Default Matrices)

Following is a diagram showing the process used to create matrices in the ***COMMANDER G3***.

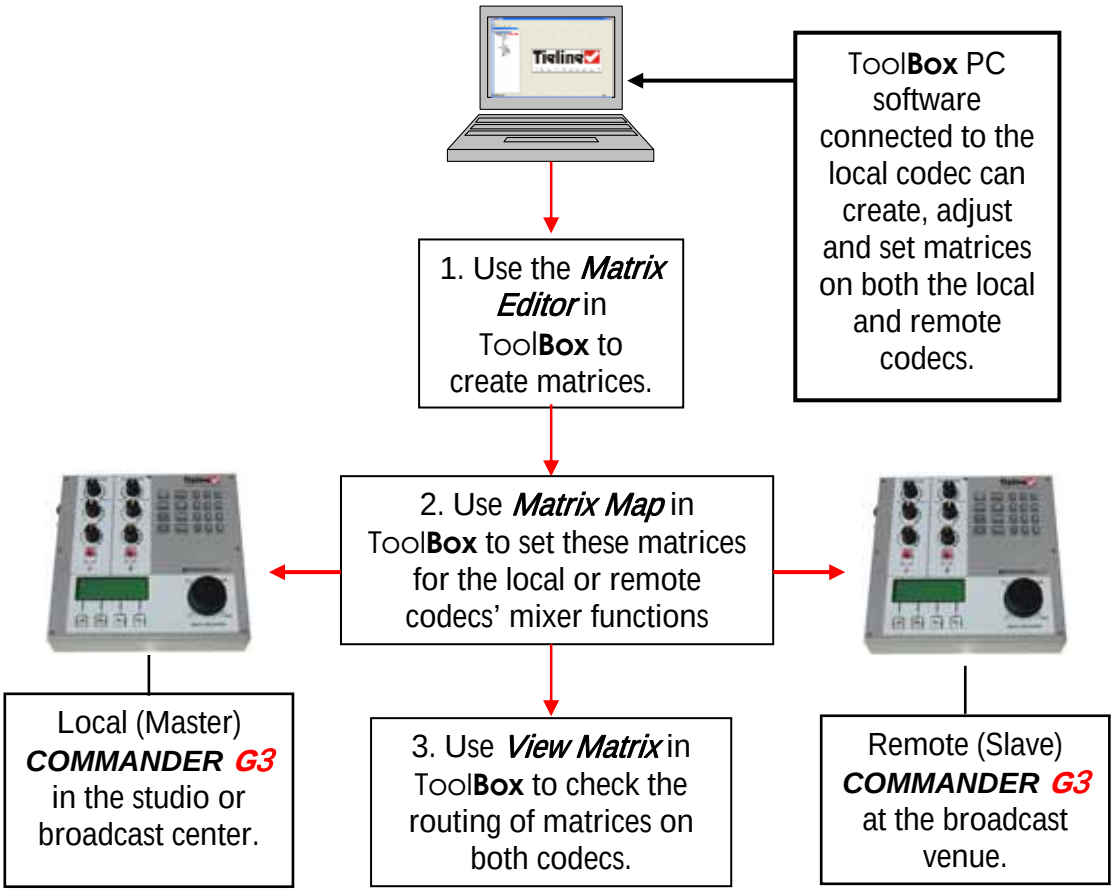


Figure 53: Diagram of Typical Matrix Setting Configuration Using ToolBox

The details of how to use the *Matrix Editor* are discussed in the following sections.

20.6. Defining the Matrix Editor

For some unknown reason the word “matrix” scares some broadcast personnel. It is simply a device for switching selected sources to selected destinations, thereby controlling their route. Hence the title router, DMR (Digital Matrix Router), routing matrix, routing switcher or just plain matrix switcher or router. All mean the same thing and perform the same task. Experienced matrix users will recognize the usual ‘check the path you require’ format. For those who don’t, please read on.

	Encoder 1	Encoder 2	CH 1 Out	CH 2 Out	HP1 L	HP1 R	HP2 L	HP2 R	Phone Out	Auxilary	PPM 1	PPM 2
Input 1												
Input 2												
Aux												
Phone												
Decoder 1												
Decoder 2												

The list on the left hand side: *Input 1, Input 2* etc, are inputs and sources from where signal flows from.

The list across the top *Encoder 1, CH 1 Out* etc, are destinations or outputs where signals flow to.

Any source can be sent to all or no destinations. The intersections of these sources and destinations are called cross points.

20.7. Checking of Cross points

If a cross point is checked a connection is made. If unchecked, a connection is not made, i.e. no audio path.

	Encoder 1	Encoder 2	CH 1 Out	CH 2 Out	HP1 L	HP1 R	HP2 L	HP2 R	Phone Out	Auxilary	PPM 1	PPM 2
Input 1					✓	✓						
Input 2												
Aux												
Phone												
Decoder 1												
Decoder 2												

In this example, *Input 1* is checked to *HP1L* & *HP1R*, being the left and right side of the headphones plugged into the HEADPHONE 1 socket on the left side of the ***COMMANDER G3***. If these boxes were not checked nothing would be heard in the headphones. *HP1 L* & *HP1 R* are the only destinations enabled to receive audio in this matrix.

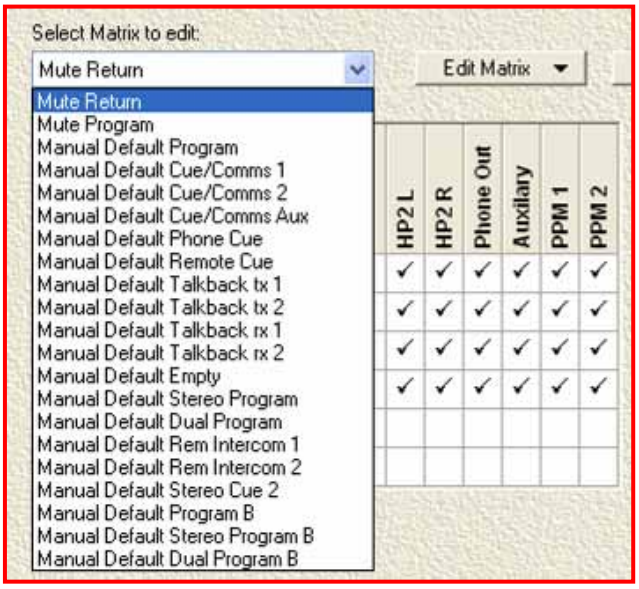
	Encoder 1	Encoder 2	CH 1 Out	CH 2 Out	HP1 L	HP1 R	HP2 L	HP2 R	Phone Out	Auxilary	PPM 1	PPM 2
Input 1					✓	✓						✓
Input 2												
Aux												
Phone												
Decoder 1												
Decoder 2												

In this example, *Input 1 PPM2* has now been checked.

The signal appearing on *Input 1* can now be heard in the headphones (as in the previous example) and also will display on the PPM2 meter on the ***COMMANDER G3*** field unit LCD, or on the PPM2 LEDs on the rack unit.

20.8. The Default Matrices

The ***COMMANDER G3*** comes with a set of default matrices. You will have 21 preset default matrices in your ***Matrix Editor***. These are displayed by clicking on the pull-down menu labeled *Select Matrix to edit*. Simply click on the desired preset matrix to display the signal paths that have been created. These can be left the same, changed to suit your unique needs and even saved as a new matrix with a unique name.



The intentions of the default matrices are two-fold. First of all to assist in conceptualizing how the matrices are set up for different routing functions. Secondly, they allow you to easily choose a preset matrix that will suit your requirements.

The great thing about the default matrices is that you can save a lot of time and effort in creating new matrices by copying, adapting and saving the default ones as new matrices. This is discussed in more detail in the following sections.

There are 21 default matrices within the ***COMMANDER G3*** that are set for the following:

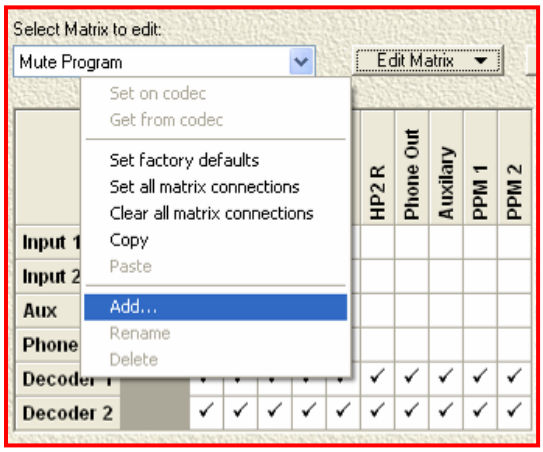
- *Mute Return;*
- *Mute Program;*
- *Manual Default Program;*
- *Manual Default Cue/Comms 1 & 2;*
- *Manual Default Cue/Comms Aux;*
- *Manual Default Phone Cue;*
- *Manual Default Remote Cue;*
- *Manual Default Talkback tx 1 &2;*

- *Manual Default Talkback rx 1 &2;*
- *Manual Default Empty;*
- *Manual Default Stereo Program;*
- *Manual Default Dual Program;*
- *Manual Default Rem Intercom 1 & 2;*
- *Manual Default Stereo Cue 2*;
- *Manual Default Program B*;
- *Manual Default Stereo Program B*; and
- *Manual Default Dual Program B*.

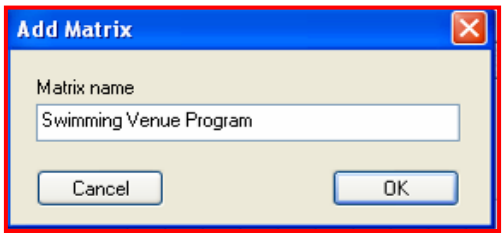
The *Mute Return* matrix mutes the return feed from the studio, as per the routing rules set by the user. *Mute Program* mutes the outgoing program feed, as per the routing rules set by the user.

These default matrices have been selected and mapped for the five default operational profiles that come with the ***COMMANDER G3***. The manual default profiles represent the most common uses of the codec operationally. Default matrices can be viewed and selected using ***Matrix Map*** within the ***Profile Editor*** in Tool**Box**. If you have purchased the ***Matrix Editor*** option you can also amend and change the individual default matrix settings and save them as new matrices. (Related Topics: ***Matrix Map***, ***View Matrix***, Manual Default Profiles and Selecting Manual Default Profiles)

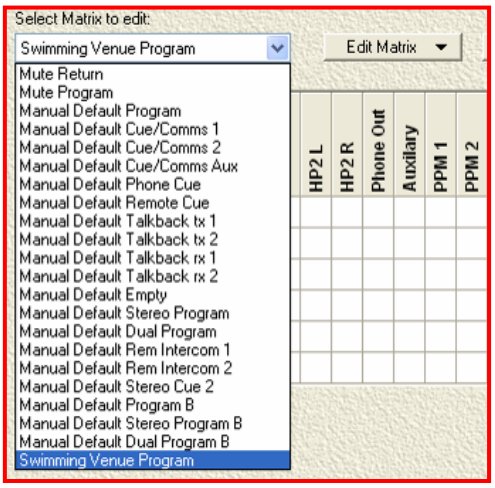
20.9. Creating, Saving and Amending Matrices



To create a new matrix simply click on the *Edit Matrix* button and click on *Add*. This will open a new menu screen which will allow you to name the matrix.



In the example to the left we have called it *Swimming Venue Program*. Names can be amended at a later date by clicking on *Rename* in the *Edit Matrix* drop-down menu.



Swimming Venue Program will now appear as an additional matrix in the drop-down *Matrix* menu, as shown to the left.

No connection will be checked in the matrix initially. You can create your matrix from scratch by checking the connection signal paths you desire. Alternatively, to save time you can copy the cross point settings from an existing matrix to your newly created matrix.

To do this, simply go to the matrix you wish to copy from, i.e. *Manual Default Program B*, by selecting it with the *Select Matrix to edit* drop-down menu. Once selected, click on *Edit Matrix* and then *Copy*. Then go to the matrix you wish to copy to, i.e. *Swimming Venue Program* as shown above, and once selected click on *Edit Matrix* and then *Paste*. The cross point connections from the *Manual Default Program B* matrix will be copied to the *Swimming Venue Program* matrix. They can be amended as required.

To delete a matrix that is no longer required, simply select the matrix no longer required with the *Select Matrix to edit* drop-down menu, click the *Edit Matrix* button and then click *Delete*. The matrix will be deleted from the list of matrices.

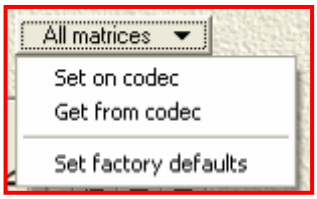
20.10. Additional Edit Matrix Functions

Set factory defaults sets the current matrix displayed by the *Matrix* drop-down menu, to the factory default settings.

Set all matrix connections makes all of the cross points in the currently shown matrix active connections.

Clear all matrix connections makes all of the cross points in the currently shown matrix inactive connections.

20.11. All Matrices Menu Functions



Set on codec transfers all of the matrices in the active profile in ToolBox onto the codec.

Get from codec retrieves all of the matrices from the codec and imports them into the ToolBox software.

Set factory defaults returns all of the factory default matrices in the *Select Matrix to edit* drop-down menu to their original factory default settings.

Section 21. Profile Editor

One of the new features of the ***G3*** platform is the profile system for configuring the codec. A profile enables a user to configure their codec for every event where the unit is to be used. A profile consists of all the configuration details required for a remote broadcast to be done successfully, including phone numbers, input settings, audio routing, ISDN setup, function buttons, talkback operation and many more.

Once you have set up a profile, the **Tieline** connection manager can automatically connect your codecs. All you have to do is turn the codec on and select START from the codec's Connection Manager (Cxns) LCD screen - the Connection Manager takes over from there.

21.1. User Profiles

Up to 98 user profiles can be stored on a ***COMMANDER G3***. These include *Current Runtime*, *Manual Default Presets* (for setting manual default profile settings), the five manual default profiles and any custom profiles you create.

These can all be adjusted on a PC using Tool**Box** software and downloaded into the codec. With Tool**Box** software it is possible to copy and paste between profiles, reducing much of the effort required to create variations for each remote site.

To assist a user to identify each profile, it is possible to assign a unique 16 character name for each profile. For example, a typical profile name would be "football park" or "basketball" or "town hall". When you go to the site where the remote broadcast is to take place, simply select the profile for that site. The codec will then be configured and ready to use for broadcasting from that venue.

By using Tool**Box** software, a user can set up a profile by entering all the details required in a sequence of tabulated menus. Alternatively, it is possible to copy from a reference profile and paste it into a new profile. Then only a few details such as phone numbers need to be changed, saving a lot of time and effort. Reference profiles are provided with factory default settings. Users can work from these to develop their own profiles and save them in Configuration Files with new names.

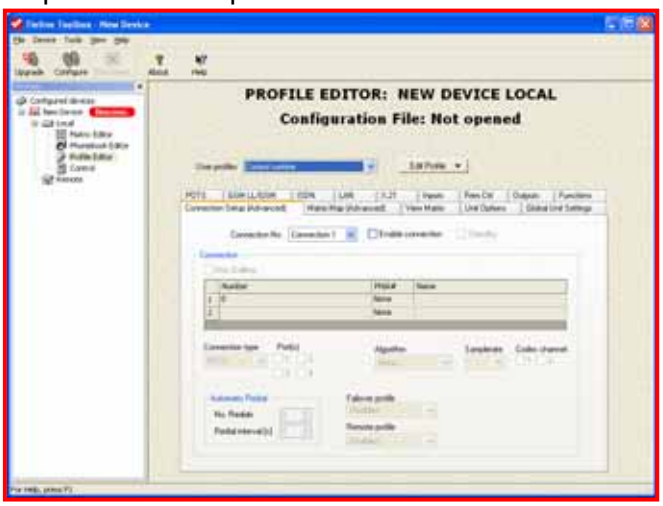


Figure 54: The Profile Editor Menu Screen for the i-Mix G3 in ToolBox Software

WARNING: Please note that if you want to change profiles you need to hang up the codec before making the change. Then load the new profile that you require and redial to make a connection.

21.2. Set Factory Defaults

When you purchase your codec it will be loaded with several factory default profiles. *Set factory defaults* sets the profile settings on the codec to the factory defaults.

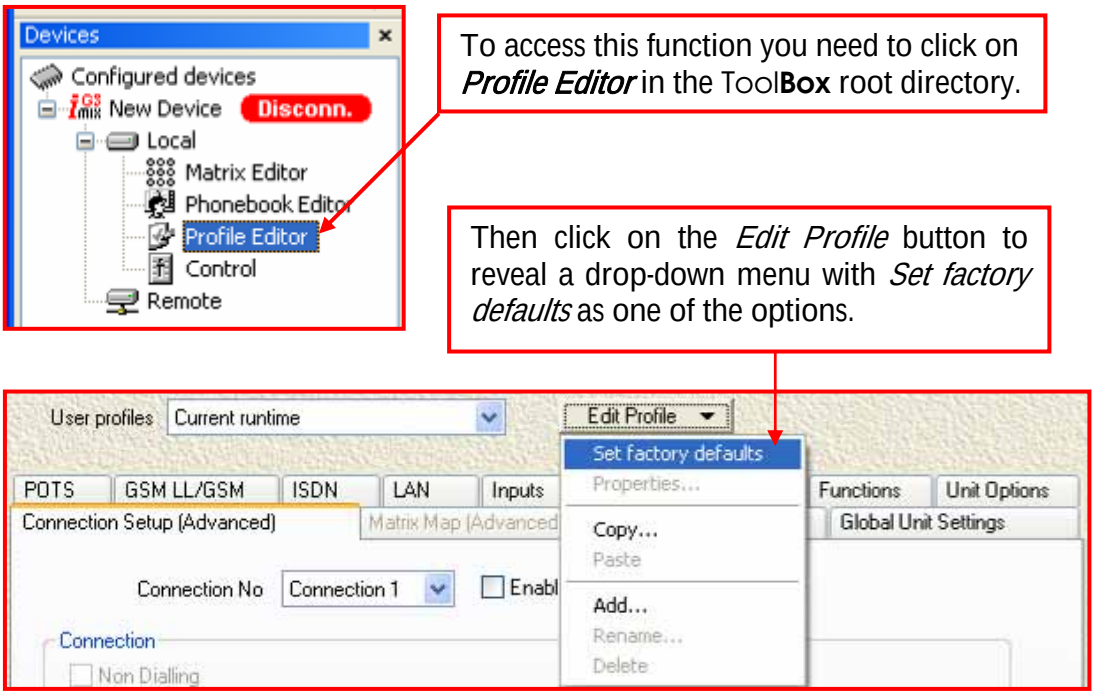


Figure 55: The Set factory defaults drop-down menu.

Tieline codecs when powered up will initialize with the same settings that were in operation just before being turned off. These settings are stored in Electronically Erasable RAM within the codec. In normal operational use the day-to-day settings will be different to the factory default settings.

Factory defaults are stored in flash memory in the codec at the factory during manufacture, or by uploading software updates containing revised defaults. *Set factory defaults* copies these settings from Flash memory to EE memory. An internal battery ensures Flash memory is retained.

21.3. Manual Default Profiles: Overview

Several manual default profiles have been programmed into each codec. These profiles allow you to easily select a preset profile that will suit typical requirements. These default profiles have certain matrices attributed to them. Your codec is shipped with several preset default matrices that route audio to program, talkback and cue.

The great thing about the default matrices and profiles is that by copying and pasting you can save a lot of the time and effort required to create new matrices and profiles from the beginning. By purchasing the 'Profile and Router' (*Matrix Editor*) option from **Tieline**, you can also adapt and save the default matrices as new profiles in your codec.

The manual default profiles within **Tieline** codecs include:

- *Current Runtime;*
- *Manual Default Presets;*
- *Manual Default Mono Program;*
- *Manual Default Mono/IFB;*
- *Manual Default Stereo; and*
- *Manual Default Dual Program.*

The default ***Matrix Map (Advanced)*** selections for the manual default profiles cannot be changed in a codec unless you have purchased the 'Profile and Router' option. If you have purchased this option, you can amend and create new matrices. See the ***Matrix Editor*** section within this reference manual for more details on editing codec matrices.

21.3.1. Adjusting Tab Settings within the Manual Default Profiles

Manual Default Presets allows you to adjust the following tab settings in relation to **all** the manual default profiles except *Current Runtime*:

- *POTS;*
- *GSM LL/GSM/USB-3G;*
- *ISDN;*
- *IP/LAN;*
- *SIP*
- *X.21;*
- *Inputs;*
- *Rem Ctrl;*
- *Outputs;*
- *Unit Options;*
- *Functions;* and
- *Global Unit Settings.*

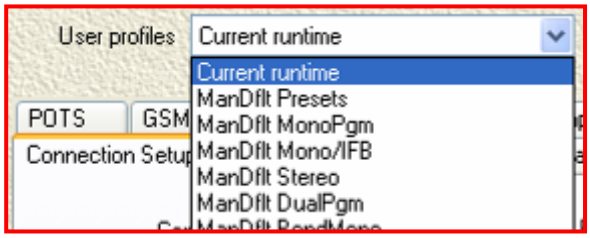
To adjust settings within these tabs, simply select *Manual Default Presets* with the *User profiles* drop-down menu. This will reveal the profile 'mask' (Related Topic: *Properties*) which will allow you to adjust settings within the various tab settings available. These settings will be standard for all the manual default profiles.

The *Manual Default Presets* settings are retained for later use if you switch from a manual default profile to a user created profile with different settings. Therefore you can load a user created profile with different settings onto your codec and when a manual default profile is subsequently reloaded, the previous *Manual Default Presets* settings are loaded back into *Current Runtime* (i.e. current codec settings).

(Related topics: The Default Matrices, *Powerup Profile*, *Connection Setup* and Selecting Manual Default Profiles)

Connection Tip: When you connect two codecs using manual default profiles, the dialing codec will instruct the codec that receives the call as to which profile should be active in the codec. If you are using custom profiles that you have created, you need to ensure that you have the correct profiles loaded into both codecs with the correct talkback and connection settings i.e. *Auto answer* and *Automatic Redial*. Following is a discussion of each of the manual default profiles and their features.

21.4. Selecting Manual Default Profiles



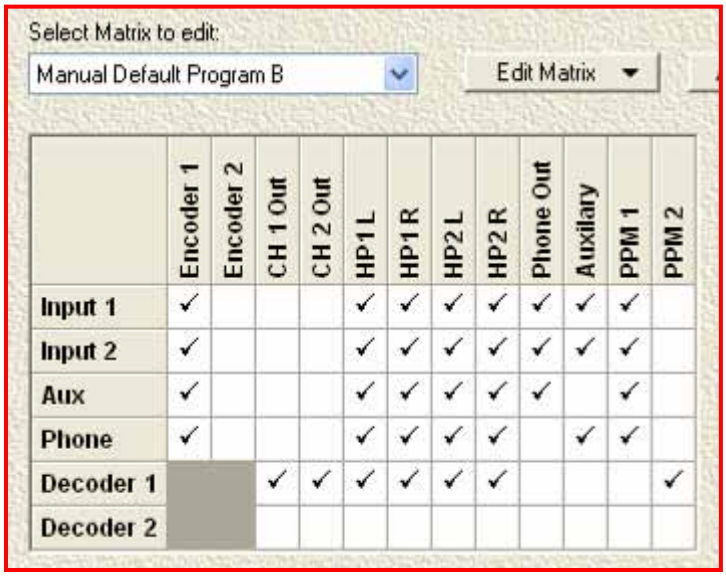
To select a manual default profile you simply go to the ***Profile Editor*** and click on the drop-down menu for *User profiles*. This will display the manual default profiles for selection.

WARNING: Please note that if you want to change profiles you need to hang up the codec before making the change. Then load the new profile that you require and redial to make a connection.

21.5. General Attributes of Manual Default Profiles

As mentioned, the ***Matrix Map*** tab sets the matrices for manual default profiles and all other profiles that you create.

In the manual default profiles, the *Phone* input is set for a mono program source such as a talkback radio caller and the *Aux* input is set for an auxiliary mono audio source.



	Encoder 1	Encoder 2	CH 1 Out	CH 2 Out	HP1 L	HP1 R	HP2 L	HP2 R	Phone Out	Auxilary	PPM 1	PPM 2
Input 1	✓				✓	✓	✓	✓	✓	✓	✓	
Input 2	✓				✓	✓	✓	✓	✓	✓	✓	
Aux	✓				✓	✓	✓	✓	✓		✓	
Phone	✓				✓	✓	✓	✓		✓	✓	
Decoder 1			✓	✓	✓	✓	✓	✓				✓
Decoder 2												

Figure 56: The Manual Default Program B (Mono) Matrix

In the preceding manual default matrix example, both the *Auxiliary* and *Phone* outputs are mix-minus feeds of their respective inputs. The functionality of these outputs is similar for all the manual default profiles. A brief description of the various manual default profiles follows.

21.6. Current Runtime: Overview

Current runtime includes the profile settings that are current within the codec, including subsequent changes to a profile that have not been saved to a Configuration File. (Related topic: Powerup Profile)

Important note: Any changes you make to a profile using the codec menus will be stored as the *Current Runtime* settings. These settings are lost when a new profile is loaded. To permanently edit and save any changes to profiles, or to create new profiles, you must use ToolBox software.

21.7. Manual Default Presets

Selecting *Manual Default Presets* reveals the profile 'mask' (Related topic: *Properties*) showing several active tabs for programming settings for all the manual default profiles. The settings made within the *Manual Default Presets* profile become standard for all the manual default profiles.

Manual Default Presets provides the ability to program commonly required codec settings for copying into any new custom profile, e.g. Mic Input Level as opposed to the default Line Level.

It is simply a repository for those settings, which when a new profile is created, must be copied into the new profiles. The current *ManDflt Preset* settings can be copied and pasted automatically by using the *Profile Properties* dialog when a new profile is created.

Once programmed into a codec, these settings become the default settings when changing between manual default profiles, e.g. *ManDflt Mono*, *ManDflt Stereo* etc.

If a custom profile is loaded into a codec, this profile's settings are programmed to be active in the codec and manual default settings are not used. If a manual default profile is subsequently selected, *Manual Default Presets* settings are restored into the codec.

Important Note: If a factory reset is performed on the codec these settings will be discarded and will need to be sent to the codec again and reloaded.

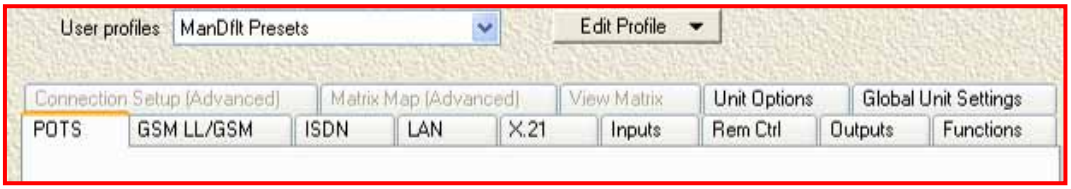


Figure 57: The Profile 'Mask' for Manual Default Presets

The inactive or 'grayed-out' tab settings within *Manual Default Presets* can be accessed and amended once a profile has been selected with the *User profiles* drop-down menu.

(Related Topic: Adjusting Tab Settings within the Manual Default Profiles)

21.8. Manual Default Mono Program

Following are the program matrix settings as shown within the ***View Matrix*** tab in the ***Profile Editor***. The default matrix used is *Manual Default Program B*.

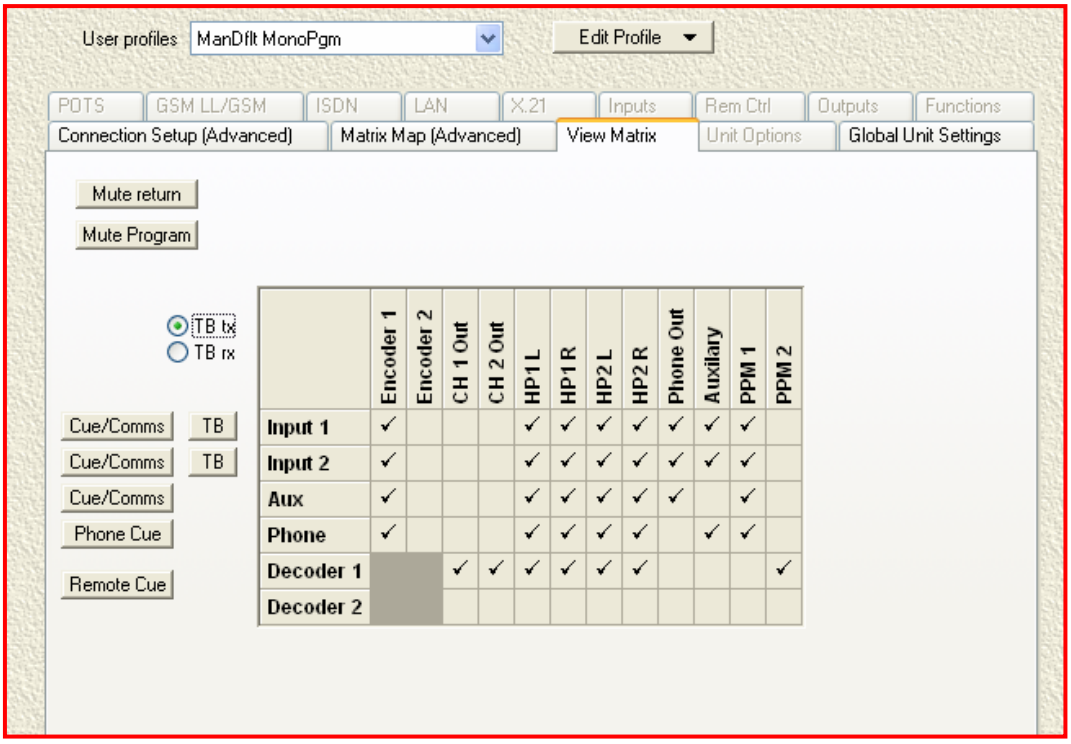


Figure 58: Manual Default Mono Program Matrix Settings

The matrix settings show that *Inputs 1-2,* the *Phone* input and the *Aux* input are all being sent to *Encoder 1, HP 1 & 2 Left* & *Right* and *PPM 1*. *Inputs 1 & 2* are also being sent to *Phone Out* and the *Auxiliary* output (RCA out). Incoming audio using *Decoder 1* is being sent to *Ch 1 & 2 Out* (analog program outputs via 3 pin XLRs), *HP 1 & 2 Left* & *Right* and *PPM 2*. This program matrix is also used for the *ManDflt Mono/IFB* profile.

21.9. Manual Default Mono/IFB

This profile uses the same *Manual Default Program B* matrix as the previous profile for routing program signals. In addition it allows IFB communications. The following image displays the IFB settings for this profile when viewing the default *Cue/Comms* and *Phone Cue* matrices.

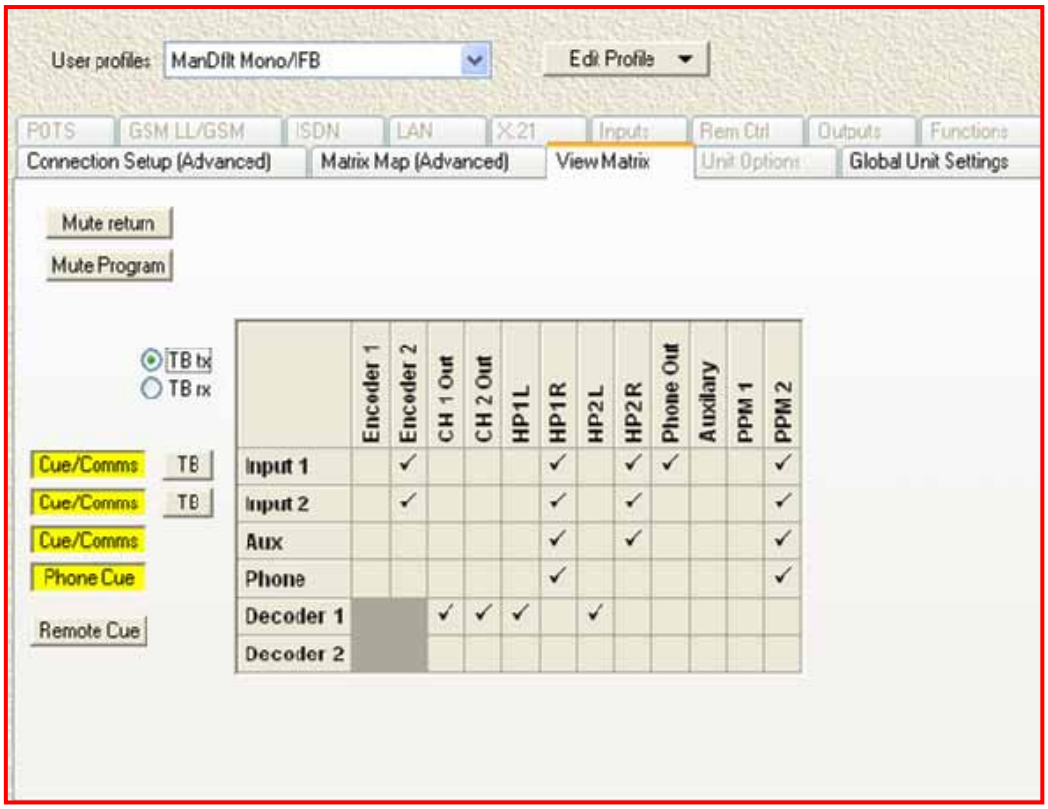


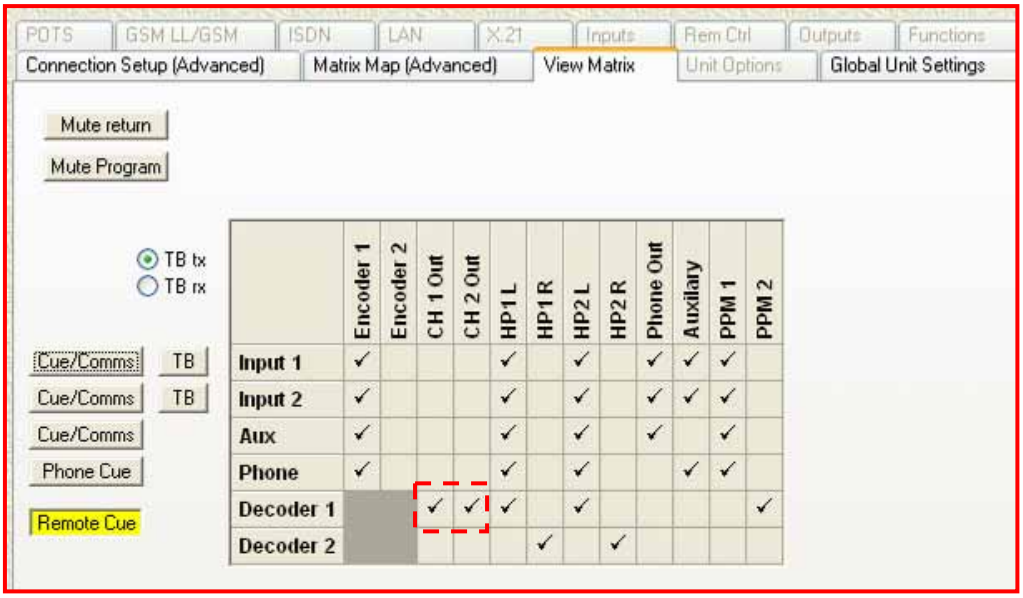
Figure 59: Manual Default Mono/IFB Matrix Settings

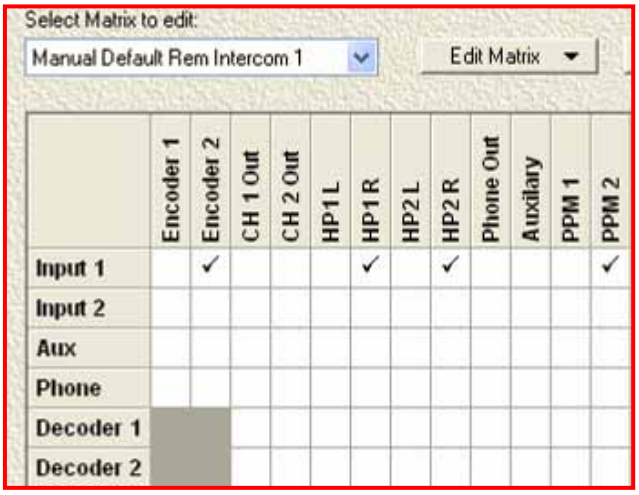
The *Cue/Comms* matrix settings are set for *Inputs 1 & 2* to be sent to *Encoder 2* and to the right side of the headphone outputs. This would be an ideal setup for commentators' microphones which can send outbound talkback signals via *Encoder 2* and at the same time monitor each commentator's talkback in the headphones.

The *Aux* input is being sent to the right side of *HP 1 & 2* but not to the *Encoder,* so *Cue/Comms* on this input can be used as a cue or pre-fade listen for a program audio source such as music or sound effects. The *Phone* input's audio is being monitored in the right side of HP 1. *Phone Out* audio is being sent from the *Input 1* audio source.

Via *Decoder 1*, the incoming mix minus audio feed is being monitored in the left side of each *HP* output and is also being sent to *CH 1 & 2 Out*. All wired audio inputs are being monitored via *PPM 2*.

Remote Cue is the matrix that determines where incoming talkback will be monitored. Pressing the *Remote Cue* button in the ***View Matrix*** tab will display the matrix talkback routing. In the following image, you will note that incoming talkback is being monitored in *HP1R* and *HP2R.*





To the left is an example of the *Manual Default Rem Intercom 1* matrix. This matrix is the default for *Input 1* within this profile. Changes to this matrix can be made if you have purchased the 'Profile and Router' option from **Tieline**. Matrices can be set for various codec functions within ***Matrix Map*** in the ***Profile Editor***.

As seen in the ***View Matrix*** section of the ***Profile Editor***, you will note that audio for *Input 1* is being sent on *Encoder 2* and monitored via the two right-hand headphone outputs and *PPM 2*.

21.10. Manual Default Stereo

The *Manual Default Stereo Program B* matrix used by this profile is set for stereo program signals. The program matrix settings are displayed in the following Tool**Box** image:

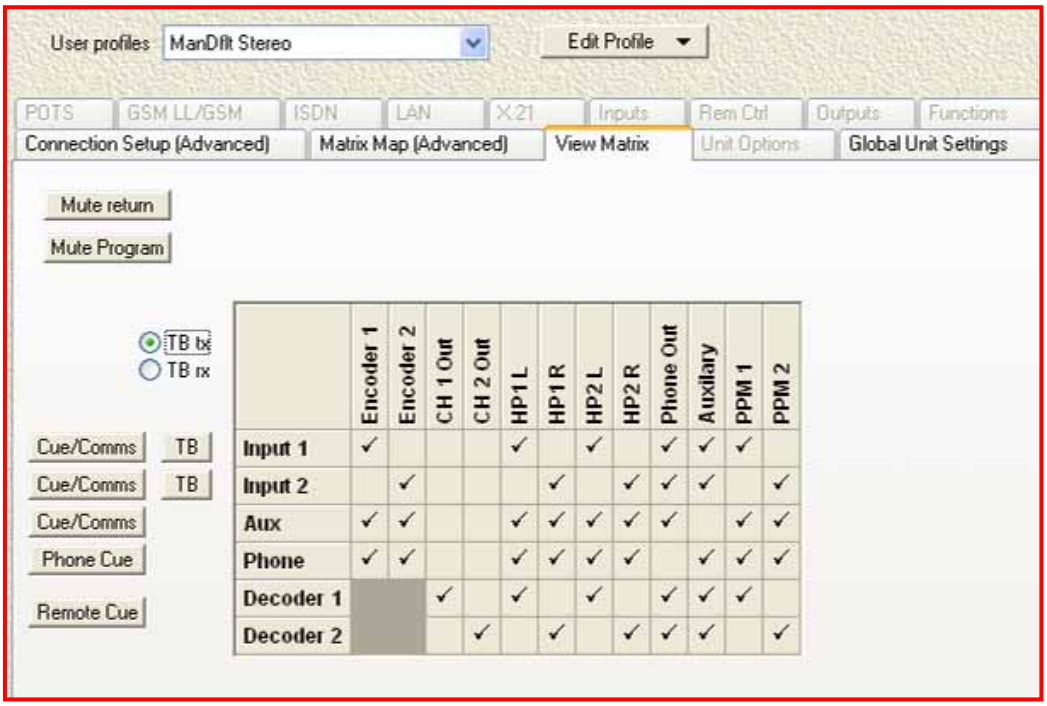


Figure 60: Manual Default Stereo Matrix Settings

You will notice that the *Phone* and *Aux* inputs are set for mono audio sources and are being sent to both *Encoder 1* and *2* (the left and right channel encoders). Each of the analog outputs, *CH 1 Out* and *Ch 2 Out*, are receiving incoming audio via *Decoders 1 & 2* respectively. *Inputs 1* and *2* are set for a stereo program audio source. *Input 1* is being sent to *Encoder 1* and *Input 2* is being sent to *Encoder 2*.

PPM 1 is monitoring left channel program sound and *PPM 2* is monitoring the right channel.

Important Stereo Connection Note:

If you are using the *Manual Default Stereo* profile (or a custom stereo profile you have created), and you are connecting using two POTS lines, if for some reason you lose one connection, the bandwidth of the single remaining connection will be split to provide two channels of audio (audio quality will be reduced if this occurs). If the bandwidth of this POTS connection drops below 21,600 bps, audio will default to provide mono audio on the left channel only. This will continue until the bit rate increases to 21,600 bps. At this bit rate two channels will be returned. This will occur automatically.

21.11. Manual Default Dual Program

The *Manual Default Dual Program B* matrix used by this profile is programmed to send dual mono feeds from the codec. The matrix settings are displayed in the following ToolBox image:

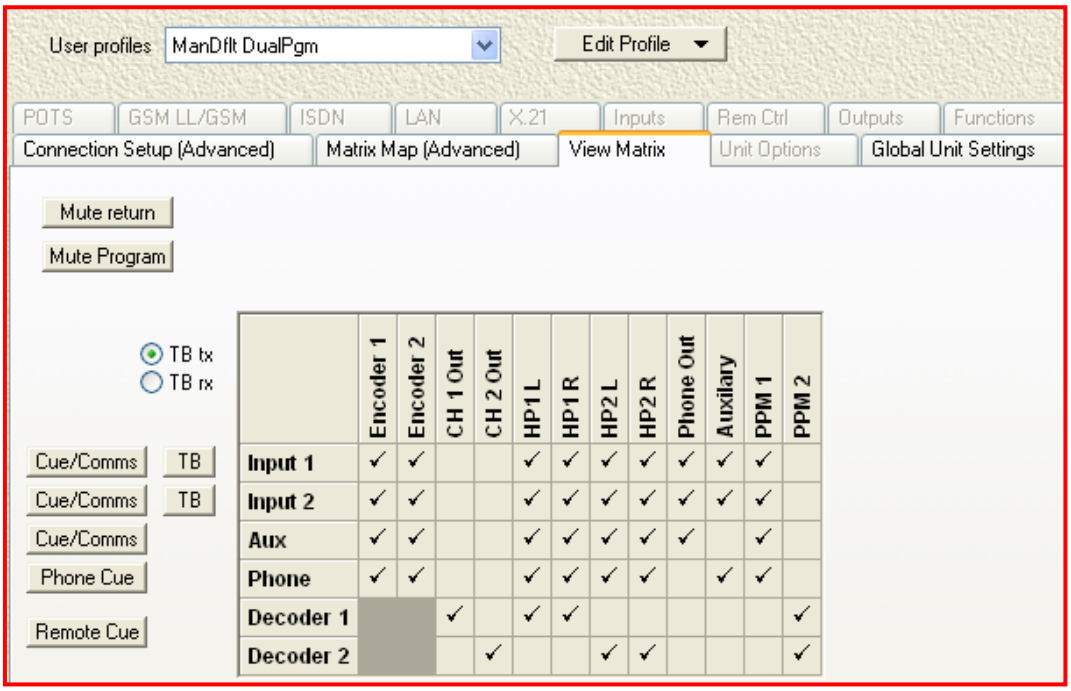


Figure 61: Manual Default Dual Program Matrix Settings

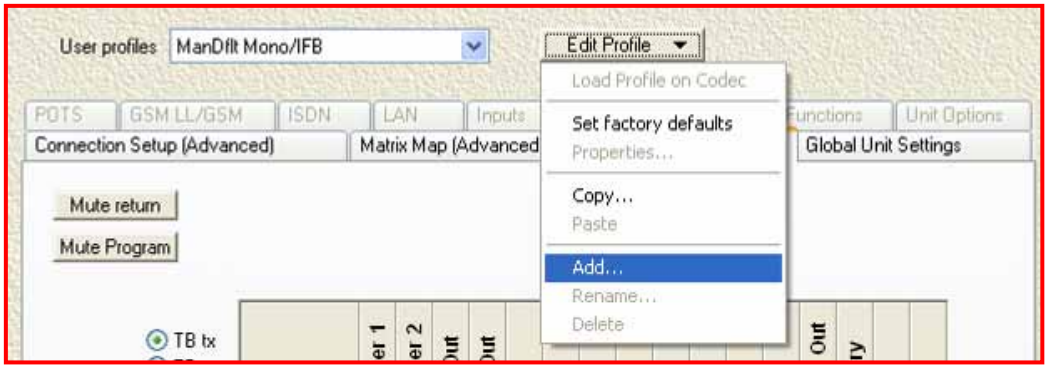
All four inputs are being sent to both *Encoders 1* and *2* as dual mono signals out of the codec. *Decoders 1 & 2* are being sent to *Ch 1 & 2 Out* respectively. Each *Input* is sent to both *HP L* and *R* for each headphone output and all *Inputs* are being monitored on *PPM 1*.

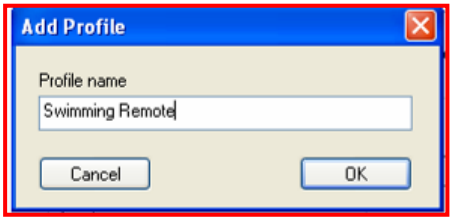
21.12. Manual Default Bonded Mono

Please note: This profile has been replaced by automatic bonding and is explained in the section of this reference manual titled Dialing Bonded Mono Connections.

21.13. Creating a New Profile

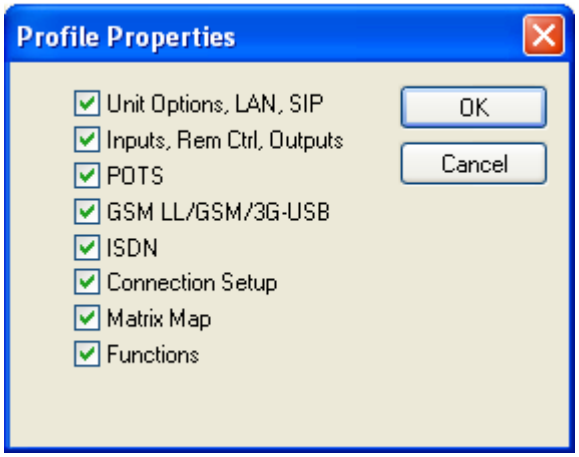
Creating a new profile within a Configuration File is made simple with ToolBox software. Simply click on the *Edit Profile* button within the ***Profile Editor***. Select *Add* from the drop-down menu as the following diagram shows.





When you click on *Add* the *Add Profile* menu will be displayed. Simply name your profile, in this case *Swimming Remote*, and click on *OK*. Your new profile is then saved to the Configuration File and can be viewed under *User Profiles*.

21.14. Properties



Properties is a very important profile setting as it determines which profile settings are set on the codec within each **custom profile**. To access the *Profile Properties* menu screen, you must first *Add* a new profile. Once you have named the new profile and clicked on *OK*, the *Profile Properties* menu screen will open. *Properties* relates to the tabs within the ***Profile Editor***. The tabs affected are shown below.

When checked, the ***Profile Editor*** tabs displayed becomes a profile's 'mask'. What this means is that all of the checked tab settings will be saved in the Configuration File for the new profile, and will be loaded onto the codec when you transfer the Configuration File onto it. The grayed-out and inactive tabs are set to their *Current runtime* settings.

Important Note: *Properties* can only be accessed and the 'mask' changed within user created profiles. These settings cannot be changed for manual default profiles. The common settings for manual default profiles can be viewed and changed within *Manual Default Presets*. Individual profile settings can be changed by viewing a profile's 'active' tabs when a profile is selected. (Related Topics: Manual Default Profiles.)

Current runtime includes the profile settings that were current within the codec, including subsequent changes to a profile that have not been saved to a profile. These settings will be retrieved on power up. The *Current runtime* settings can be viewed by selecting *Current runtime* from the *User profiles* drop-down menu

In the example that follows, you will see that the *Man Dflt Stereo* profile 'mask' is displayed by the active tabs in the ***Profile Editor*** main menu. The inactive tabs are grayed-out. What this means is that the four tabs for *Connection Setup, Matrix Map, View Matrix* and *Global Unit Settings* can be viewed and amended when the profile is opened in ToolBox. The other tabs can only be viewed and changed by making *Current runtime* the active profile within ToolBox.

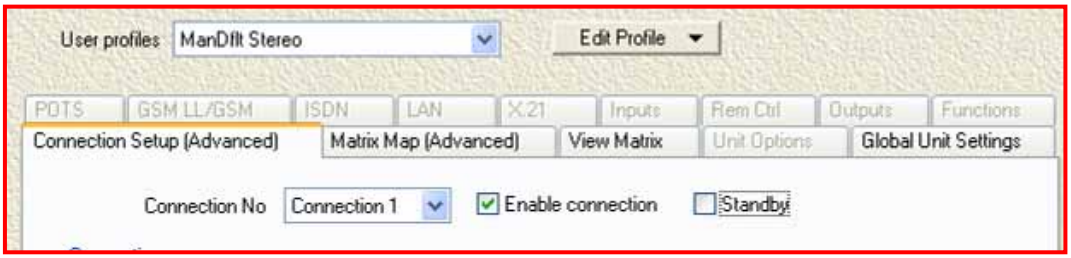


Figure 62: The 'Mask' for Man Dflt Stereo

21.15. How Do I Utilize Profile Masks?

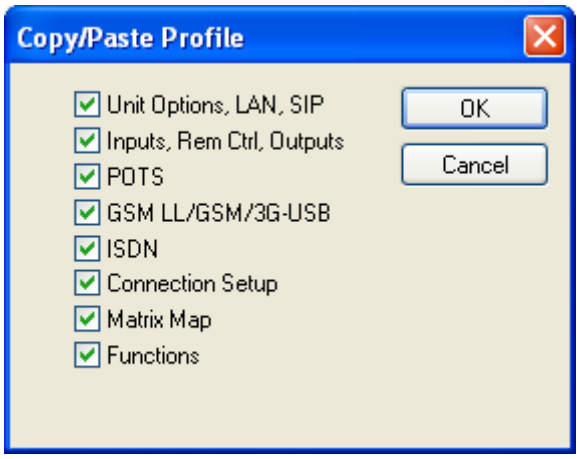
The reason you may wish to adjust your profile 'mask' for custom profiles is so that you can easily create common settings among different profiles by utilizing the *Current runtime* profile.

First you would set the tab settings you want to use for several profiles in *Current Runtime*. You can then create and name new profiles, changing only the tabs that require different settings by creating individual 'masks'.

If you decide subsequently that you wish to change a user created profile's 'mask', you can do this by opening the profile, clicking on the *Edit Profile* drop-down menu, and then *Properties*. This will open the *Profile Properties* menu box and allow you to select or deselect menu items.

21.16. Copy and Paste Functions

ToolBox software makes it incredibly easy to copy and paste settings from one profile to another. To copy settings from one profile to another, simply select the profile you wish to copy from with the *User Profile* drop down menu. Then click on the *Edit Profile* button to reveal the drop-down menu. Select *Copy* from this menu and the *Copy/Paste Profile* menu will be displayed.



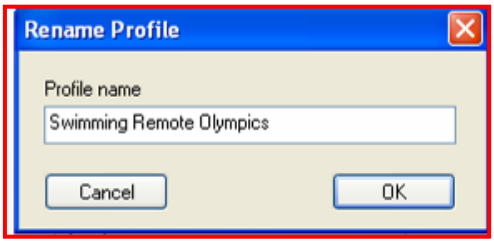
As you can see it is very similar to the *Profile Properties* menu. Similar principles apply in that you simply select the ***Profile Editor*** tabs which you wish to copy settings from by checking the boxes for the required tabs. In this example all ***Profile Editor*** tabs have been selected.

Once you have made your selections click *OK* and select the profile you wish to copy to with the *User profiles* drop-down menu. If you need to create a new profile - do this now. Once you have selected the profile you are copying to, click on *Edit Profile* and select *Paste* from the drop-down menu. The settings will then be copied to the selected profile.

21.17. Deleting a Profile

To delete a profile that is no longer required, select the profile you wish to delete with the *User profile* drop-down menu. Then simply click on the *Edit Profile* button and click *Delete* in the drop-down menu. The profile will be deleted instantly.

21.18. Renaming a Profile



It is simple to rename a profile. Simply select the profile you need to rename with the *User profile* drop-down menu. Then click on the *Edit Profile* button and select *Rename* from the drop-down menu. The *Rename Profile* menu will be displayed. Simply type in the new name for the profile and click *OK* to save it.

21.19. Making Adjustments within Profiles

If you need to amend the matrices within profiles you must use the ***Matrix Editor*** (To do this you need to purchase the 'Profile and Router' option from **Tieline**). To adjust any other profile settings first click on ***Profile Editor*** on the Tool**Box** root menu for your codec and the ***Profile Editor*** menu will open. Each tab in the ***Profile Editor*** contains the many parameters required to configure your codec. These parameters fall under the following general tab headings:

- *Connection Setup (Advanced).*
- *POTS.*
- *GSM LL/GSM/USB-3G*
- *ISDN.*
- *IP/LAN.*
- *SIP.*
- *X.21*
- *Inputs.*
- *Remote Control.*
- *Outputs.*
- *Unit Options.*
- *Functions.*
- *Matrix Map (Advanced).*
- *View Matrix.*
- *Global Unit Settings.*

A detailed discussion of each of the ***Profile Editor*** tabs, and the elements within them, will follow in this manual.

Section 22. Connection Setup

Connection Setup is the first of several tabs within the *Profile Editor* menus. It allows you to program dialing of connections and failover operation within your codec. If you don't have ToolBox software and a PC attached to your codec, most of the dialing functions can be programmed via the codec connection wizards. For information about these wizards see the Quick Start section of this manual. Dialing procedures are also explained in the section of this manual titled Operation of your Codec.

Within this section of the manual we also explore algorithms and coding in more detail, to assist you in selecting the most reliable connection profile for your broadcast situation.

22.1. The Connection Manager

Tieline has developed a new Connection Manager that automates the process of making a connection between **Tieline** codecs. All a user has to do is turn the codec on, wait for the menu on the LCD screen to light up and then select START.

The Connection Manager takes the number set up in a profile's *Connection Setup* and dials that number, be it POTS, ISDN, satellite ISDN, IP or a cell-phone number. From that point on, everything is set up to work, as per the selected profile. It is up to the user to ensure that there is a receiving codec at the other end for the dialing codec to call.

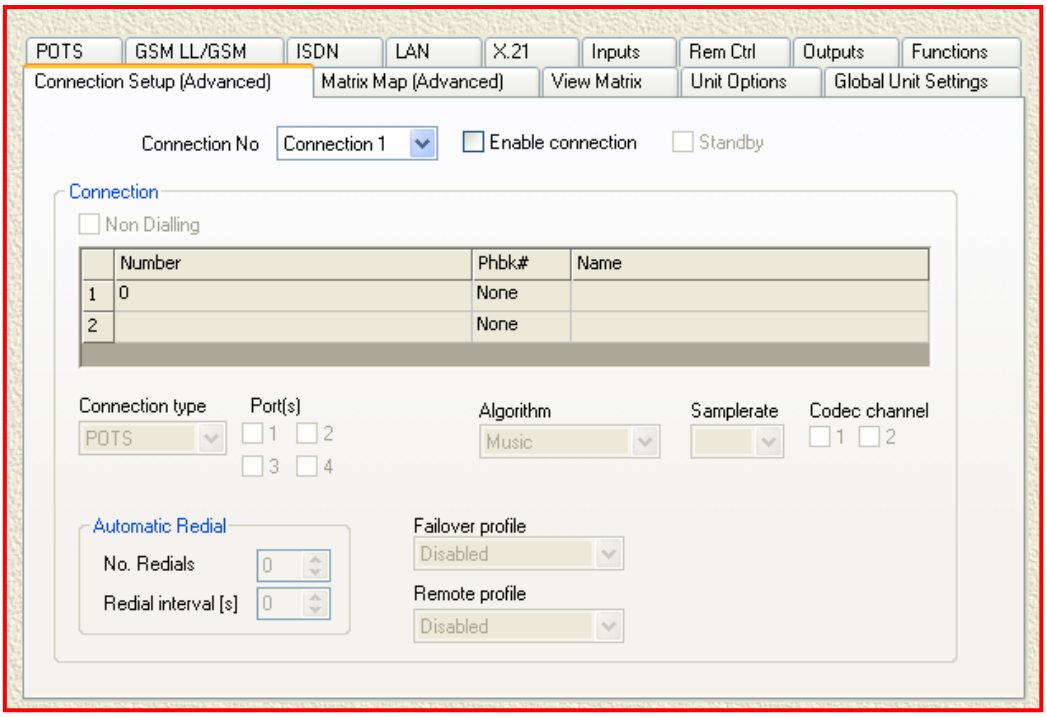
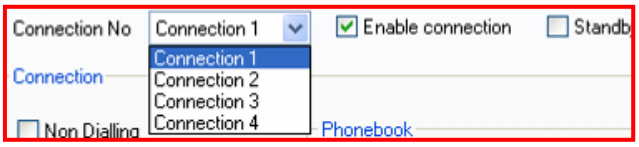


Figure 63: The Connection Setup (Advanced) Menu in the Profile Editor

22.2. Connection No.

The Connection Manager will handle multiple connections, such as dual channel ISDN, POTS or IP. It is also possible to purchase the option of using main and standby *Failover profile* connections. An example of this could be an ISDN connection with one or two 'B' channels in use and a POTS connection that can be set as the *Standby* connection. The Connection Manager can detect a failure on an ISDN 'B' channel and then automatically change over to the POTS circuit to prevent a loss of program audio.



Up to two main connections can be programmed into the Connection Manager and two standby connections. To select a connection number click on the pull-down menu for *Connection No.* and select the connection number you want to set. Then check the *Enable connection* box which allows a user to set the options for the functions within the ***Connection Setup*** tab. Prior to this the functions within this menu are 'grayed' out. Once *Enable connection* is checked the functions become active. If the connection you are setting is a standby connection then also check the *Standby* box next to the *Enable connection* box.

Operational tip:

If the codec you are using will not be dialing (i.e. you will be receiving a call from another codec) you must check the *Non Dialling* box which will then grey-out the number display.

Important 3G Connection Tip:

When connecting over 3G, it is important to not confuse dialing into your 3G network provider with dialing your IP connections. When connecting over 3G you first connect to your ISP, and then you can subsequently dial the IP address of the codec you wish to connect to. It is not possible currently to connect using the Connection Manager over IP. This type of connection needs to be done manually as a two step process. For more information please see the '3G Quick Start' section of this reference manual.

22.3. Bonding: An Overview

One of the new features of the ***G3*** platform is the ability to have a *Bonded POTS* or *Bonded ISDN* connection. With this latest technology, **Tieline** gives broadcasters the ability to synchronize channels and create seamless stereo broadcast capabilities over two analogue telephone lines. In addition, you can bond up to four ISDN channels together to achieve greater connection bandwidth for a mono or stereo audio broadcast. The bonded POTS and ISDN connections that are available using **Tieline** codecs are described in detail in the following section.

Important Connection Information:

Use only one phone number in the dialing section of the *Connection* menu if you are using non-bonded connections. For example, if you are using two connections that are not bonded (as you may for the *Man Dflt Mono/IFB* profile), create two connections using the *Connection No* drop-down menu and input one phone number for each connection.

If you are connecting using a bonded connection, i.e. *Bonded POTS* or *Bonded ISDN*, you will need to input multiple numbers into one connection. To do this you will need to select the bonded *Connection type* you desire from the drop-down menu. It is possible to dial up to two POTS and four ISDN numbers once you select a bonded connection.

If you connect using the Connections Manager the codec will dial up to two POTS and four ISDN numbers for you. If you connect manually after you have loaded the information onto the codec, you will need to dial the numbers from the codec via the displayed connections on the LCD screen.

22.4. Dialing Bonded Mono Connections

Dialing bonded mono connections has never been simpler. Previously, if dialing a bonded connection, a bonded profile needed to be selected. Now all you need to do is select the *ManDflt MonoPgm* profile and if **Tieline** codecs detect two connections they will automatically be bonded to increase connection bandwidth.

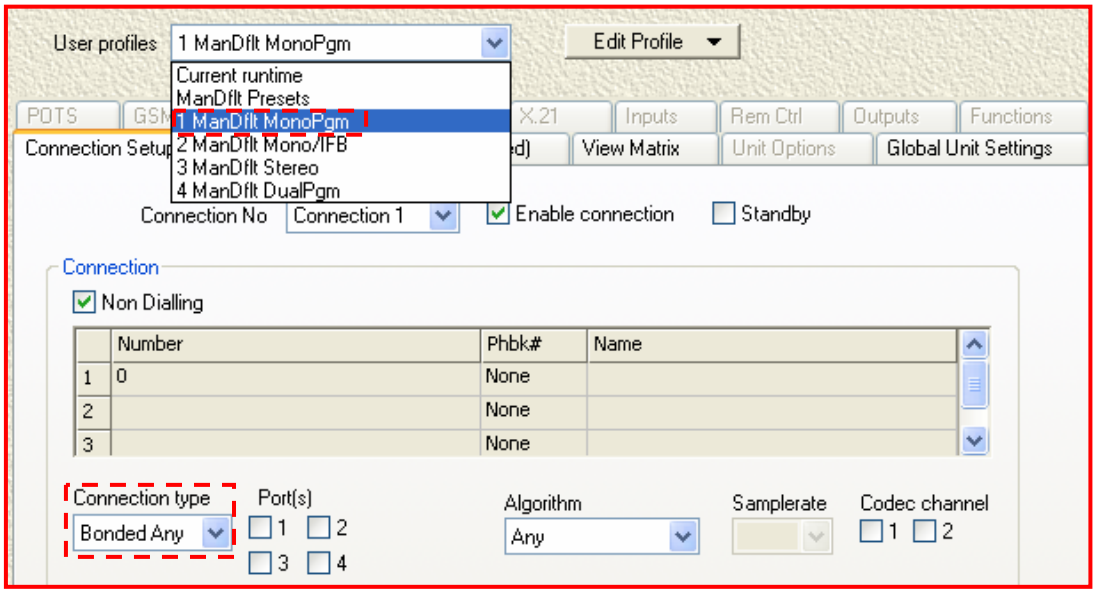
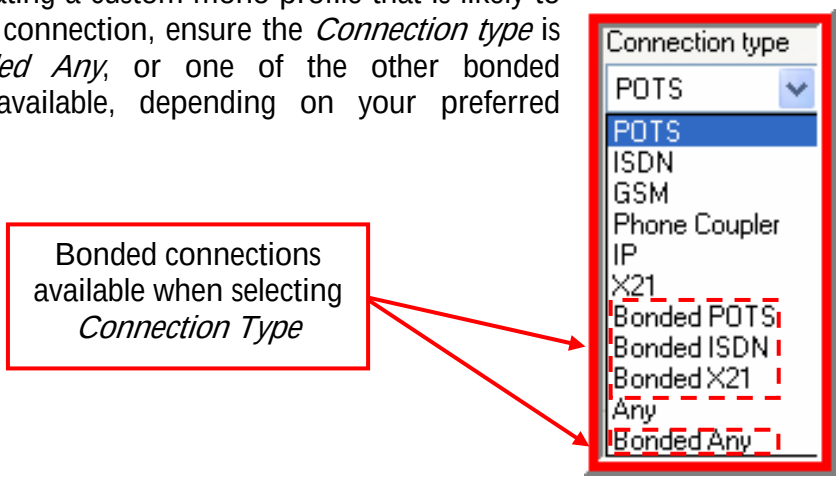


Figure 64: List of Manual Default Profiles Displaying Connection Type Setting

If you are creating a custom mono profile that is likely to use a bonded connection, ensure the *Connection type* is set to *Bonded Any*, or one of the other bonded connections available, depending on your preferred connection.



Please note:
This type of mono profile bonding will work for all algorithms if you are connecting between **Tieline** codecs. If you are connecting to a non-**Tieline** codec then bonding will only work effectively using *MP2 Mono*. This algorithm use CCS bonding.

22.5. ISDN 3B and 4B Bonding

4B channel ISDN bonding is available in **Tieline** codecs, which can provide 256 kbps connections. CCS bonding, used by MPEG algorithms, is only capable of bonding two connections together. However, **Tieline** codecs can bond up to four channels together using **Tieline's** own proprietary bonding. By selecting **Tieline** bonding you can take advantage of this capability and increase the quality of *MP2* algorithm connections.

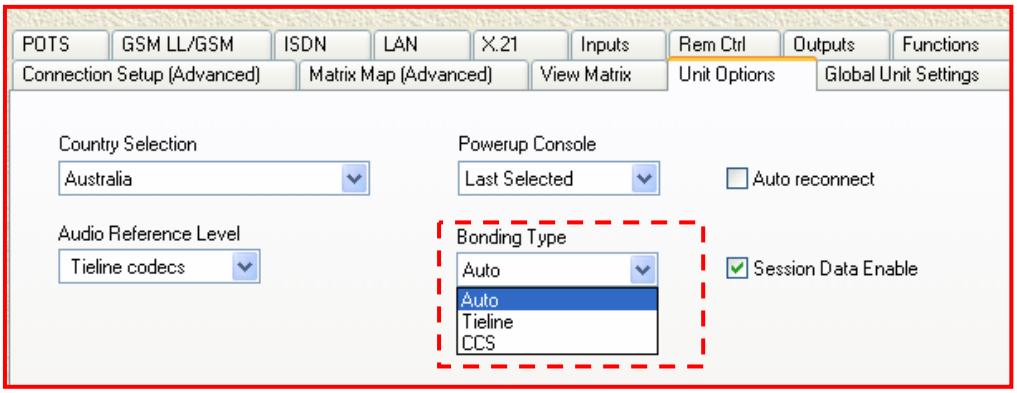


Figure 65: Bonding Type Drop-down Menu

If *Auto* is selected then the codec will automatically use CCS bonding if you select an *MP2* algorithm in the codec with a 2 x 64kbps dialing profile. Following is a table showing the codec settings based on what is programmed in this menu.

Bonding Type	Algorithm	No. of Connections	Bonding Used
Auto	*MP2 Mono, Stereo, J-Stereo*	2	CCS
Auto	*MP2 Mono, Stereo, J-Stereo*	More than 2	Tieline
Auto	**Tieline** *Music*	2 or more	Tieline
Auto	*Voice G3*	2 or more	Tieline
CCS	*MP2 Mono, Stereo, J-Stereo*	2	CCS
CCS	*MP2 Mono, Stereo, J-Stereo*	More than 2	Tieline
CCS	**Tieline** *Music*	2 or more	Tieline
CCS	*Voice G3*	2 or more	Tieline
Tieline	*MP2 Mono, Stereo, J-Stereo*	2	Tieline
Tieline	*MP2 Mono, Stereo, J-Stereo*	More than 2	Tieline
Tieline	**Tieline** *Music*	2 or more	Tieline
Tieline	*Voice G3*	2 or more	Tieline

Table 9: Bonding Type Codec Settings

If you wish to select either *CCS* or *Tieline* bonding as the default setting within a codec then select either of these options as desired from the menu.

To program this setting **with the codec itself** press SOFTKEY 4 Menu and select **[Configuration] > [System Settings] > [Bonding Type]**. Then select the bonding setting you require.

22.6. Configuring 4B Channel Connections

To configure the dialing of 4B channel connections via ToolBox software simply enter the four B channel numbers into the ***Connection Setup*** tab.

To enable this make sure you have checked the *Enable Connection* section on the page, choose *Bonded ISDN* as the *Connection Type* and then click on each connection number in turn and enter each B channel's number. An example of numbers that have been entered in this way are displayed in the following image.

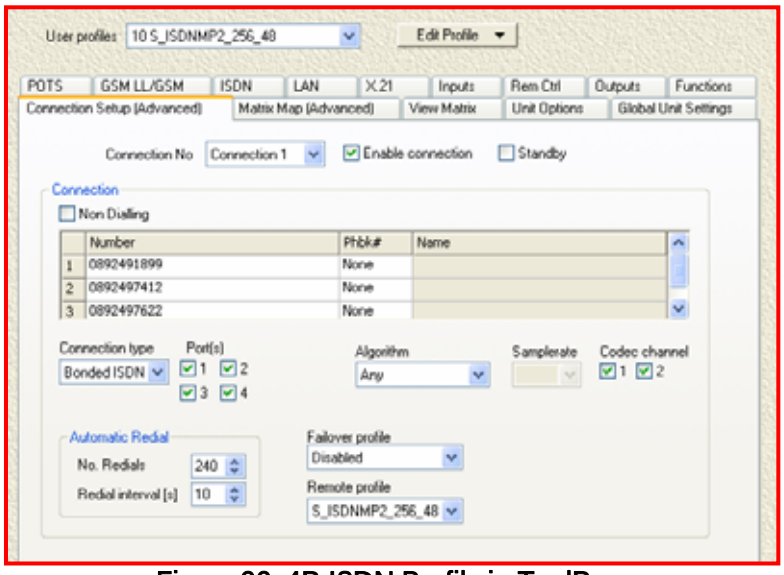
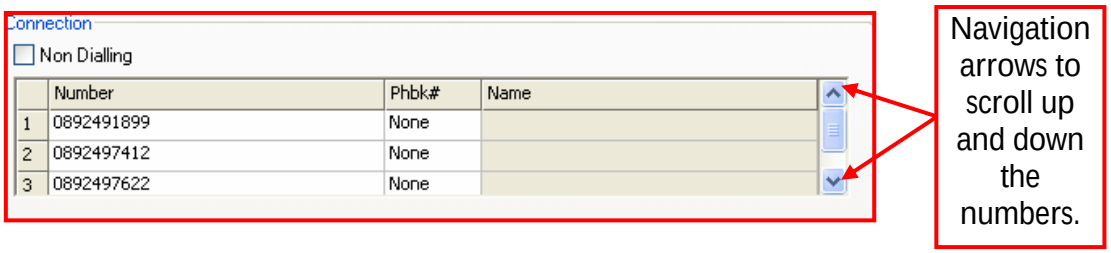


Figure 66: 4B ISDN Profile in ToolBox

To scroll up and down the ISDN numbers click on the right hand arrows as displayed in the following image.



It is also necessary to ensure four *Ports* are checked. If you are connecting using a mono profile check one *Codec channel* and if you are connecting using a stereo profile, check two *Codec channels.*

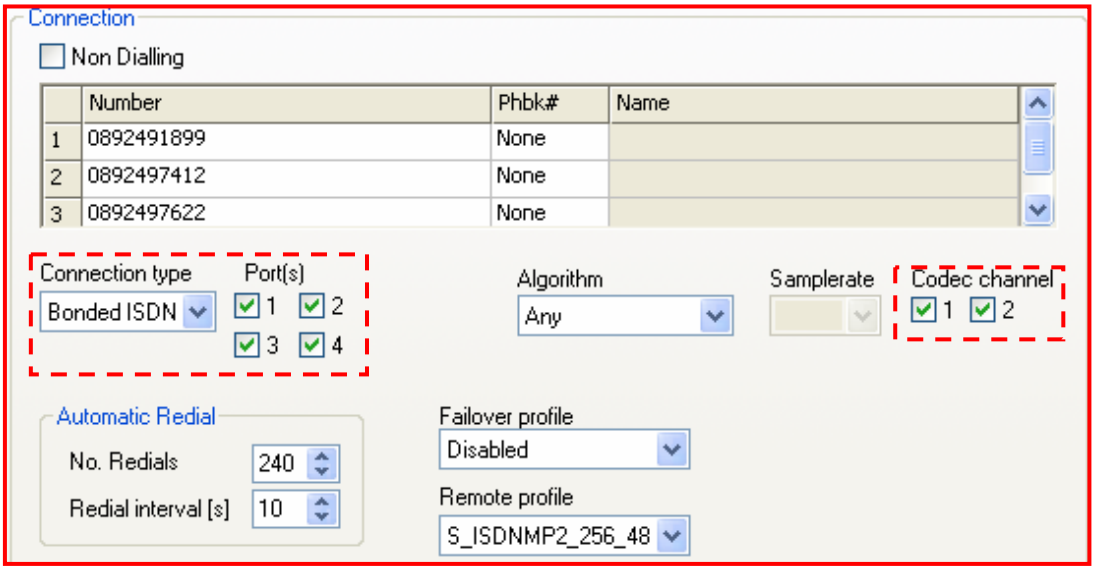
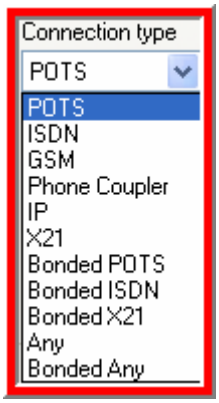


Figure 67: Port and Codec Assignments for Stereo using 4B Channels

22.7. Connection Type



There are several ways of connecting **Tieline** codecs. A drop-down menu allows users to select from several different connection methods. This is displayed in the right-hand image.

Manual default profiles have preset *Connection type* (and *Algorithm*) settings. Generally these settings reflect how you will want to connect using the profile you wish to use.

In certain situations you will need to adjust the settings for *Connection type*. For example, if you are connecting using a stereo connection, i.e. *ManDflt Stereo*, the default *Connection type* is *Bonded Any*. This is so that you can connect using either POTS lines or ISDN B channels with a bonded connection. However, if you have a POTS connection attached to your codec, as well as an ISDN module (with 2 B channels), if you dial using *Bonded Any* the connection may select one B channel and the POTS line as the two available connections to connect with – even if you have selected your bonded connection numbers. In this situation it would be best to change the *Connection type* to *Bonded ISDN*, ensuring this sort of confusion does not occur. The POTS connection could be used as a failover connection in this situation. (Related topics: Algorithms, Failover profile, and Phonebook Numbers.)

Important Note: If you change a *Connection type* setting for a manual default profile, you need to ensure that this setting is also changed on the remote codec you are dialing.

22.7.1. POTS

Use this connection mode to connect via a standard Plain Old Telephone Service (POTS or PSTN) telephone line. *Music* is the factory default algorithm setting.

22.7.2. ISDN

Select this connection mode to connect using an ISDN line. Attach an ISDN line with an RJ-45 connector to an ISDN module installed in your codec to provide this type of connection. Two B channels can provide two 64 kbps connections.

22.7.3. GSM

To use the GSM connection mode, ***G3*** codecs require a GSM plug-in module with an antenna connector which is available from **Tieline** as an additional option. A connection can also be made with a cell-phone via a data cable connected to the RS 232 serial port connection on the back of the codec. (For cable pin configurations please see the Connector Wiring section of this reference manual.)

Figure 68: Tieline GSM Plug-in Module

22.7.4. Phone Coupler (on i-Mix codecs)

Plug your RJ-11 POTS line into Coupler Line 2 on the rear panel of an ***i*-Mix *G3*** codec to utilize this function. The operation of this is explained more fully in the section within this manual titled Phone Input: Operation of the i-Mix G3 Phone Coupler.

22.7.5. IP

Select this mode for IP/Internet connections. Attach an Ethernet cable to the LAN port on your codec to use this type of connection. If the cable is connected successfully to the network, the lights will be illuminated below where the Ethernet cable is attached to the codec.

For 3G/IP connections, insert a 3GIP or USB module into your codec and connect to your 3G provider/ISP. These connections use IP as you would for an Ethernet connection over a LAN.

22.7.6. X.21

Select this mode for X.21 connections. Attach a D15 cable from your ISDN TA/X.21 DCE Device to an installed X.21 plug-in module in your codec.

22.7.7. Bonded POTS

If you are using POTS lines, all you need to do is select the option you require on the drop-down menu, allocate the two ports you will be using, the codec channels that will correspond to these ports and the algorithm you require. You will also need to input a phone number for each channel to be connected. (Related Topic: Phonebook Numbers)

22.7.8. Bonded ISDN

Use this mode to connect with, and bond up to four ISDN B channels. Two channels can operate in stereo and in sync with each other, or up to four channels can be bonded to create connections from 64kbps up to 256 kbps. These higher bandwidth connections can be used for broadcasting with MPEG algorithms.

Recommendation for Bonded Mono & Bonded Stereo Connections

Where possible we recommend that you use the same type of POTS modules in the dialing codec when creating bonded connections between two codecs (i.e. bonded mono and bonded stereo connections). It is possible that bonded connections may not bond reliably when dialing bonded connections and using different POTS modules. To circumvent this possibility use the same type of POTS modules in the dialing codec. If you must use different types of POTS modules, always dial the first connection of a bonded connection from a POTS G3 module. (Please note: The on-board POTS hardware on an ***i*-Mix *G3*** codec is the same as a POTS module, not a POTS G3 module. Dial this connection second for bonded connections).

22.7.9. Bonded X.21

Use this connection mode to bond two X.21 module connections if you are connecting using a ***COMMANDER G3*** codec, which has two module slots available. This can be useful if you are dialing to an ISDN endpoint connection using MPEG algorithms, which may require ISDN bonding to be successful.

22.7.10. Any

This setting is only used for manual default profiles. Primarily this setting is used on the remote codec when you are unsure what the profile connection settings are on the local codec. It allows you to let the remote codec itself determine what connection is possible. When using this mode, the remote codec will pick the first possible manual default dialing profile (POTS, ISDN or GSM) for a single channel mono connection and connect to the local codec. The codec will automatically detect what connections are possible. The only time you would use this setting on the local codec is if you are unsure what the remote codec's connection settings are, i.e. POTS, ISDN or GSM. Otherwise if you have both POTS and ISDN connections available for example, the connection you wish to use may not be selected.

22.7.11. Bonded Any

Bonded Any is also only used with manual default profiles. It is used in the same situations as the *Any* setting except it will be a bonded connection. It will connect using either two POTS lines or in dual channel ISDN mode. (see the preceding description for the *Any* setting for further details). This connection can be used for a stereo signal or it is possible to bond two channels to gain greater bandwidth. For example if you have two POTS connections with a bit rate of 14,400, you can bond them to get one 28,800 bps connection. This would achieve better quality for mono audio. (Related Topic: Phonebook Numbers)

22.8. Coding and Algorithms

It is appropriate at this point to discuss a little about how coding works and which algorithms are available within your **Tieline** codec. Our discussion will primarily focus on audio coding but some of the information is relevant for video coding as well.

22.8.1. What is coding and how does it work?

Coding is performed by a codec (Coder/Decoder), which is simply an integrated circuit device that encodes an analog signal into a digital bit stream and can decode this bit stream back to an analog signal.

Codecs can use digital signal processor (DSP) software algorithms to compress and decompress speech and other audio signals. Algorithms are clever little beasts that can convert analog signals into digital signals at low bit rates.

In broadcast, they assist us to conserve valuable bandwidth and send high quality audio, using minimal bandwidth, which in the real-world would be impossible to do without their clever audio compression techniques.

In summary, coding algorithms facilitate a reduction in the size of data files and provide more efficient transmission of audio, video and data.

22.8.2. Sampling Frequencies, Word Size and Quantization

There is a trade-off between the sampling frequency and the 'word size' chosen to convert an analog signal into a digital signal, and the resulting quality of the converted signal. To understand these concepts we also need to understand the relationship between 'sampling' and 'quantization'.

Technical Spiel

Quantization is part of the process of converting an analog signal into a digital one. Sampling and quantization are inextricably linked. Sampling represents the times that an analog signal is measured during A to D conversion (in the case of audio the measurement of amplitude), and quantization is the numerical value of that measurement in the digital domain.

How precisely the analog signal is represented after A to D conversion is a function of the sampling frequency (how many samples per second are recorded), how many bits are available for each sampled value (word size, i.e. 16, 24, 32 or 48 bits etc). 'Quantization noise' is the sum of conversion and rounding errors during sampling. Each additional bit of sampling can reduce 'quantization noise' by 6dB – a factor of 2.

Analog signals have 'signal-to-noise' ratios that measure the ratio of the power of a signal and the noise floor of that signal. In the digital realm, 'signal-to-error' measures the errors caused during the quantization of an audio signal. As an example, 16 bit (word size) sampling, used for CD production, has an S/E ratio of about 98 dB.

It is important to understand that errors that occur in the digitization process are non-linear and depend on the signal that is being converted. This is the reason that some algorithms are better than others for speech than for music (etc) and at varying bit rates. Perceptual coding is also important to understanding this and is explained later in this section.

The challenge in the digitization process when using broadcast codecs is to accurately convert an analog signal in the best way possible, and at the same time use minimum signal processing and data bandwidth to broadcast that signal. This will depend largely on the bandwidth reliability of the connection being used, i.e. POTS, ISDN, GSM, IP, 3G or X.21.

22.8.3. Data Compression

There are basically two sorts of codec compression. In its simplest form, codec compression can be lossless, where all the original data transmitted can be retrieved after compression and decompression. This is particularly useful for archiving original recordings in a form as close as possible to their original.

Alternatively, codec compression can be what is termed 'lossy'. This means that the information compressed does not include all the source audio information. Instead, only enough data is encoded in order to provide an accurate and good quality audio signal after decoding.

Please Note: Data compression is in no way related to audio compression for limiting the dynamic range of audio signals.

22.8.4. Perceptual Coding (Modeling)

Lossless data compression is limited in its applications because of its voracious appetite for bits. As a result, there has been a considerable investment of resources into the application of perceptual modeling and hence perceptual coding techniques, which employ psychoacoustic theory.

How is this relevant you may ask? Well in simple terms, the human ear only perceives relatively small portions an audio signal's information. For example, the human ear can only hear audio between its minimum and

maximum hearing thresholds and certain loud tones in an audio signal will mask softer tones.

By replicating the information gathered from psychoacoustic theory, large elements of an audio signal that the human ear perceives as irrelevant, or that are masked, can be removed.

'Lossy' codecs that use these sophisticated predictive perceptual models to analyze the human voice, music and other audio sources, can then encode these signals using fewer bits.

Any information that is essentially imperceptible or is 'masked' within the source audio is removed by frequency filters. This is the case with most codecs used for audio broadcasting, streaming and communications.

Perceptual encoding will only encode the data required to provide an accurate representation of the audio that is being captured for a broadcast – reducing the total connection data bandwidth required for codec audio signals.

How is this useful to users of **Tieline** codecs? By using perceptual models that are tailored for speech and music etc, it is possible to utilize algorithms that are best suited to a broadcast's content and connection bandwidth availability. For example, if your broadcast content is voice only, by using **Tieline's** *Voice G3* algorithm you can achieve good voice-quality audio at very low POTS line bit rates.

22.9. MPEG Explained

The collaborative MPEG (Moving Picture Experts Group) working group standards provide open international multimedia standards for both audio and video encoding at a range of data rates. MPEG coding utilizes The first standards, MPEG-1 (ISO/IEC 11172), were defined in 1991 and approved in 1992.

Why use MPEG coding? Well MPEG coding is by far the most popular method of audio coding and the proliferation of MPEG-compatible players is all-pervasive in both broadcast and non-broadcast environments.

For example, there is little doubt that MP3 is the dominant compression algorithm for creating music files for domestic consumption and many decoders/players exist for this particular algorithm.

22.9.1. What's the difference between MPEG-1, MPEG-2, MPEG-3 & MPEG-4?

The differences between MPEG 1, 2, 3, and 4 have resulted from evolutionary advances in audio coding over the past decade or so. Each subsequent set of MPEG standards has provided greater compression efficiency and more options for coding not only audio, but video as well.

In the following sections we will give a brief overview of MPEG audio. If you require more detailed information on MPEG audio it can be sourced from http://www.chiariglione.org/mpeg/

22.9.1.1. MPEG-1

This was the first set of MPEG standards produced and it has three operation modes or 'layers'. These are:

- **Layer-1:** The compression used in Digital Compact Cassettes;
- **Layer-2 (MP2)**: The MUSICAM®[19] compression format used in CDs and Video CDs ; and
- **Layer-3 (MP3)**: the MP3 compression format used widely to rip tracks from CDs for portable digital music players.

MPEG-1 layers can provide single channel, dual channel, stereo and joint stereo coding and all of these operating modes are supported in **Tieline** codecs. Each 'layer' has improved upon the previous one by performing better at lower bitrates.

The sampling frequencies available for MPEG-1 include 32kHz, 44.1kHz and 48kHz. **Tieline** codecs support 32kHz and 48kHz sampling frequencies for MPEG-1 algorithms.

MPEG 1 Layer 2 (*MP2)* is included in **Tieline** codecs.

22.9.1.2. MPEG-2

MPEG-2 standards followed the MPEG-1 standards and were designed to be the successor to the popular MP3 format. The standards included the attributes of MPEG-1 as well as:

- Backwards compatibility with MPEG-1;
- Surround sound 5.1 capability (please note that MP3 surround has since become available); and
- Coding at the lower sampling frequencies (LSF) of 16 kHz, 22.05 kHz and 24kHz using Layer-1, Layer-2 and Layer-3.

Of particular relevance to **Tieline** codec users is the ability to encode at lower sample frequencies. This enables mono and stereo audio to be encoded using fewer bits and requiring less connection bandwidth.

22.9.2. Artifacts and Cascading, what should I be aware of ?

Artifacts are the enemy of the codec broadcaster. Artifacts are to a certain extent introduced by quantization in data compression and may be inaudible at low levels. When audible they can sound ' hissy', 'grainy', 'metallic' or like

[19] *Musicam is a registered trademark of Musicam USA.*

'warbling'. Artifacts may become audible when caused by a variety of factors such as:

- Overly aggressive compression rates;
- Packet errors or loss of packets from a bit stream;
- Loss of sync (i.e. AES/EBU connections); or
- Cascading the audio signal.

Cascading is when an audio signal is processed through more than one codec. The effects of cascading through multiple codecs may evidenced by increases in artifacts and their audibility. This is particularly noticeable with 'lossy' codecs, which reduce source audio data by nature.

The best way to avoid cascading artifacts is to choose the highest bit rate possible for your codec connection. In addition, plan for your end-to-end broadcast chain to include the least amount of signal compression and cascading as possible.

To listen to an example of quantization noise, visit http://en.wikipedia.org/wiki/Quantization_noise and download the audio file available from this website.

22.9.3. Which Algorithm is best for my Broadcast Situation?

In general, it is a question of what connection quality can be achieved depending on the connection bit rate available. It may also be a question of trying to be compatible with other equipment. In this section we will provide you with some rule-of-thumb settings for how to configure your codec – depending on the broadcast situation you are faced with. Here is your connection checklist:

- What is the connection bandwidth available on my connection?
- How stable is the bit rate likely to be on this connection over time?
- Will I be sending music, voice, data or a combination of these elements?
- Will I be broadcasting live and if so will I be requiring communications with the studio?

The algorithm you connect with will be dependent on the program audio you are sending and the connection bit-rate you are able to achieve. For example, *Music* is generally the preferred POTS algorithm setting if your program content contains music. *Voice G3* is generally best to use for a POTS connection if your connection bit-rate is 16,800 bps or lower.

One of the unique features of the **Tieline** *Music* algorithm is its ability to send a 15 kHz stereo audio signal over a single B-channel, with a delay of only 70 ms... pretty impressive huh!

This brings us another point – delay. This is an important issue to consider if you are broadcasting live. An MPEG-2 Layer 2 (*MP2*) algorithm would not only require 128 kbps of connection bandwidth, but it would also have a

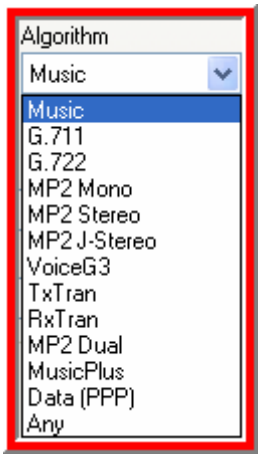
delay of over 150 milliseconds. This would most likely be above the comfortable threshold of delay for your broadcast.

When dealing with variable bit rate connections, such as over POTS, IP and 3G connections, it is a good idea to listen to the quality of your program signal and to see how it sounds when it is sent at different bit-rates. For IP and 3G/IP connections this may also include assessing your connection stability at different rates of FEC (Forward Error Correction).

If you have limited or varying bandwidth you may be better off using an efficient high-bandwidth algorithm like **Tieline** *Music*, rather than *MusicPlus* or a more bit-hungry and higher-delay MPEG algorithm.

Remember that audio quality to a certain extent is subjective. Ultimately, by testing connection stability at different bit rates (and FEC settings), you will learn how to optimize your settings depending on your connection situation.

22.10. Tieline Algorithms Available



The algorithm settings in **Tieline** codecs vary according to the *Connection type* selected. **Tieline** now offers **two** optimized low-delay *Music* algorithms which are ideal for IP and wireless IP connections.

The algorithms available include the following: *Music*, *MusicPlus, G.711*, *G.722, MP2 Mono, MP2 Stereo, MP2 J-Stereo*, *Voice G3* and *MP2 Dual* (*Transcriptor Tx* and *Transcriptor Rx* are used in captioning enabled codecs only).

Following is a description of each of the algorithms in your codec and following this is a handy reference guide in a tabular format, for deciding which algorithm is best to use in which broadcast situation – depending on your connection type and the connection bit rate that is possible.

Important Connection Note: Please ensure that you set both the local and remote codecs to operate using the same algorithm – otherwise your codecs will not operate correctly. If you are using manual default profiles, when the dialing codec calls the remote codec, the dialing codec will automatically set both the profile and the default algorithm on the remote codec to match the dialing codec's setting.

This will only occur if both the local and remote codecs have either the *Voice G3*, *Music* or *MusicPlus* algorithms set in the active profile. If either the local or remote codec have been set to an algorithm other than these then you will need to change the algorithm in each codec to either *Voice G3*, *Music* or *MusicPlus* before trying to connect.

In addition, it is very important to note that remote control of a codec is not possible unless you are using either the *Music*, *MusicPlus* or *Voice G3* algorithms.

Algorithm Tip: It is possible that when you use MPEG algorithms to connect to a non-**Tieline** codec, the 'framing' process (where incoming bit streams are identified and distinguished for individual decoding) can be unsuccessful when a call is initiated. This can be solved by turning off the 'session data' stream sent by the **Tieline** codec. To do this select **[Menu] > [Configuration] > [System Settings] > [Session Data] > [Disable].** Please note that this is a global setting and will also affect all POTS, and IP connections on the codec.

If you try to connect to a **Tieline** codec with session data disabled, you will have to ensure that the connection settings on both codecs are identical because dialing codec session data normally provides this information to the remote codec. To avoid this problem, simple re-enable session data **[Menu] > [Configuration] > [System Settings] > [Session Data] > [Enable].**

22.10.1. Tieline Music

If you will be transmitting music over a connection with limited bandwidth we recommend you use the *Music* algorithm. The audio bandwidth of the ***COMMANDER G3*** in *Music* mode is 15 kHz. This can be achieved with bit rates of 24,000 bps or higher and a delay of only 70 milliseconds. The music algorithm is effective at connection bitrates over 16,800. If your connection bit rate is lower than 16,800 bps we recommend you use the *Voice G3* algorithm. For more information on connection bitrates, please see the section in this manual titled Overview of the Manual Default Profiles.

22.10.2. Tieline Music Plus

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

22.10.3. G711 (For ISDN use)

The *G.711* algorithm is an international standard telephony encoding format. It is a 64 kbps PCM voice-coding technique for digital voice delivery via a public phone network or PBXs. Bandwidth for this algorithm is 3 kHz. If you use a POTS line to dial an ISDN line that is connected to your codec, the codec will select the G.711 algorithm to connect. This will occur automatically when you dial from the POTS end, but if you dial a POTS line using an ISDN line connected to the codec, you will need to set the G.711 algorithm in the codec before you dial.

22.10.4. G.722 (For ISDN use)

The *G.722* algorithm is for ISDN use. Audio data is sampled at 16 kHz and the audio quality provided is 7 kHz in bandwidth. It doesn't use perceptual coding and can be used for voice–based broadcasts.

It is possible that when you use this algorithm to connect to a non- **Tieline** codec, the 'framing' process (where incoming bit streams are identified and distinguished for individual decoding) can be unsuccessful when a call is initiated. This can be solved by turning off the 'session data' stream sent by the Tieline codec. To do this select **[Menu] > [Configuration] > [System Settings] > [Session Data] > [Disable].** Please note that this is a global setting and will also affect all POTS, and IP connections on the codec.

If you try to connect to a **Tieline** codec with session data disabled, you will have to ensure that the connection settings on both codecs are identical because dialing codec session data normally provides this information to the remote codec. To avoid this problem, simple re-enable session data **[Menu] > [Configuration] > [System Settings] > [Session Data] > [Enable].**

22.10.5. MP2 Mono (For ISDN/X.21/IP use)

This algorithm is used for the transmission of a mono signal and will provide 20 kHz quality audio. It is possible to use the *Bonded Mono* profile to join two ISDN B channels together and therefore achieve a better quality mono connection in MP2 mode. Bonded ISDN connections using this algorithm are only possible with **Tieline** codecs.

22.10.6. MP2 Stereo (For ISDN/X.21/IP use)

This algorithm is used for the transmission of a stereo broadcast signal and will provide 20 kHz quality audio. We recommend using 128 kbps connections for this algorithm. (Related topic: Sample Rate)

22.10.7. MP2 J-Stereo (For ISDN/X.21/IP use)

This algorithm employs 'joint stereo' techniques. Intensity stereo coding is used to take advantage of commonality between the left and right signals and this provides more effective use of connection bandwidth – meaning slightly better quality signals can be provided. *MP2 J-Stereo* can provide you with better quality signals than the *MP2 Stereo* algorithm.

We recommend using 128 kbps connections for this algorithm. (Related topic: Sample Rate)

22.10.8. MP2 Dual (For ISDN/X.21/IP use)

Normally, when Mono/IFB profiles are used over POTS or ISDN, the two streams are bonded and bi-directional audio is sent simultaneously over each channel. MP2 streams cannot be bonded so the *MP2 Dual* algorithm allows Mono/IFB profiles to send audio via the MPEG layer 2 algorithm. It can also be used for dual mono profiles. We recommend using 128 kbps connections for this algorithm.

22.10.9. Voice G3

Voice G3 bandwidth is 7 kHz and is achieved with all bit rates of 9,600 bps or higher. The *Voice G3* algorithm is incompatible with the *Voice* algorithm supported in **Tieline** ***G1*** codecs and therefore they cannot be used together.

Connection Tip:
If you want to connect a ***COMMANDER G3*** to a ***COMMANDER G1*** codec or an ***i*-Mix *G1*** you will need to use the *Music* algorithm. This will allow you to use your codecs at a bit rate as low as 9,600bps.

22.10.10. Other

This algorithm is specifically for when you are integrating **Tieline** ***G3*** codecs with Comrex®[20] Vector, Matrix® and BlueBox® codecs. If you are connecting to a Comrex® codec select its Music algorithm and select the *Other* algorithm on the **Tieline** ***G3*** codec. Bitrates of 26,400bps and 9,600bps are not supported with this algorithm. If you try to connect at these bitrates an error message will be displayed on the codec LCD next to the connection you are using. In addition, the Line Quality at the remote Comrex® codec will not be displayed on the **Tieline** codec LCD.

22.10.11. Transcriptor Tx

This feature is only used in captioning-enabled codecs. Please see the **Tieline** ***G3*** codec captioning manual if you are using this feature.

22.10.12. Transcriptor Rx

This feature is only used in captioning-enabled codecs. Please see the **Tieline** ***G3*** codec captioning manual if you are using this feature.

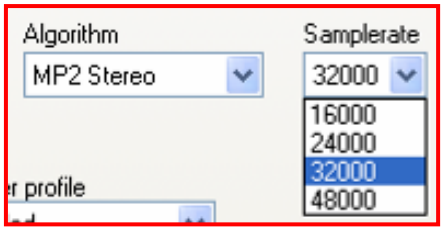
22.10.13. Any

The *Any* setting will set the algorithm you connect with according to the default settings that have been set for POTS, ISDN and GSM connections

[20] *Comrex® Vector, Matrix® and BlueBox® are all products of Comrex Devens, MA 01432 USA info@comrex.com*

within the ***POTS***, ***ISDN*** and ***GSM LL/GSM/USB-3G*** tabs in the ***Profile Editor***.

22.11. Samplerate



If you select one of the MPEG algorithms you can also select the sample rate you wish to use from the four available options. These settings are for user-created profiles. The *Samplerate* setting for manual default profiles is made within the ***ISDN*** tab within the ***Profile Editor***.

The sample rate you select will be dependent on the connection or connections you are using. For example, if you have only one 64k ISDN B channel available you will not be able to use a stereo algorithm with a sample rate of 48,000 bps. If you do connect at this *Samplerate* the quality will not be very good. The maximum you should set it at is 32,000 bps for this type of connection.

22.12. Algorithm Connection Matrix

Following is a comprehensive table that outlines the algorithms available in **Tieline** codecs and the minimum connection bit rates required for successful broadcasting. By extrapolating this information, you will be able to decide which is the best algorithm to use for your broadcast requirements.

Connection Type	Algorithm	Minimum Bit-rate and Bandwidth	Recommended Connection for On-air use.
POTS	Music	Up to 7.5kHz at 9,600 bps[21]	Up to 15kHz at 24,000 bps
POTS	Voice G3	7.5 kHz at 9,600 bps (Optimized for 16.8 kb, use for voice only)	7.5 kHz at 9,600 – 24,000 bps (use for voice only)
ISDN	Music Plus	20 kHz on all ISDN connections.	
ISDN	Music	15kHz on all ISDN connections.	
ISDN	MP2	24 kHz mono at 64,000 bps.	
ISDN	G.711	3 kHz at 64 kbps.	
ISDN	G.722	7 kHz at 64 kbps.	
GSM CSD	Voice G3	7.5 kHz at 9,600 bps.	7.5 kHz at 9,600 bps
GSM HSCSD	Music[22]	7.5 kHz at 14,400 bps (Using Voice G3)	15kHz at 28,800 bps (Using Music)
3G/IP	Music Plus	Up to 20kHz mono at 48,000 bps	Up to 20kHz mono at 48,000 bps
	Music[23]	Up to 15 kHz at 24,000 bps	Up to 15kHz at 24,000 bps or higher
	Voice G3	7.5 kHz at 9,600 bps (use for voice only)	7.5 kHz at 9,600 - 14,400 bps[24] (use for voice only)
IP/Internet	Music Plus	Up to 20kHz mono at 48,000 bps	Up to 20kHz mono at 48,000 bps
	Music	Up to 7.5kHz at 24,000 bps	Up to 15kHz at 24,000 bps or higher
IP/Internet	Voice G3	7.5 kHz at 9,600 bps (use for voice only)	7.5 kHz at 9,600 - 14,400 bps[25] (use for voice only)
IP	Raw Audio[26]	15 kHz at 2.3 megabit minimum.	
X.21	Music Plus	20 kHz on all X.21 connections.	
	Music	15kHz on all X.21 connections.	
X.21	MP2	24 kHz at 64,000 bps.	
X.21	G.711	3 kHz at 64 kbps.	
X.21	G.722	7 kHz at 64 kbps.	

Table 10: Algorithm Connection Bit rate Table

- All of the factory default algorithm settings can be changed for POTS, ISDN, GSM/3G and X.21 connections. For more information on changing these settings please see the main reference manual.

[21] If your connection bit-rate is 16,800 bps or lower, we recommend using the *Voice G3* algorithm. It provides up to 7.5 kHz audio at 9,600 bps.
[22] If you wish to connect using a GSM HSCSD connection and achieve up to 15 kHz in audio quality, you will need to change the default algorithm setting to *Music*.
[23] Stereo IP and 3G/IP connections using the *Music* algorithm require a minimum connection bit rate of 48kbps..
[24] Using Voice G3, there is no need to negotiate higher than 14.4 or 16.8kbps to maximize audio quality. Staying at lower bit-rates improves link stability over TCP.
[25] Using Voice G3, there is no need to negotiate higher than 14.4 or 16.8kbps to maximize audio quality. Staying at lower bit-rates improves link stability over TCP.
[26] Only available over a point-to-point crossover cable (i.e. CAT 5) and using high quality LAN switching.

- Every manual default connection includes a data channel of 50 bytes per second which can be used to send RS232 data between devices attached to the serial port of each **Tieline** codec.
- It is very important to note that remote control function of a codec is not possible unless you are using the *Music, MusicPlus* or *Voice G3* algorithms.

Please note: See the section titled 'Connection Bit-rate and Algorithm Options' in this reference manual for more detailed information on UDP/IP and MPEG connection bitrates.

22.13. Ports and Codec channels: An Overview

Depending on your setup you could be using one or two *Ports* and one or two *Codec channels*. The number of ports and codec channels that you will require is dependent on the profile that you are using to connect with.

22.13.1. Manual Default Profile Connections

If you are connecting using manual default profiles, you do not physically have to 'check' the *Port* and one *Codec channel* in the ***Connection Setup*** section of ToolBox software. This will automatically occur when you commence the dialing process from the codec. This is also the case for any connection where the *Any* setting is the algorithm setting in ***Connection Setup***.

For example, if you are connecting using a mono profile, one *Port* and one *Codec channel* will be used to connect over either a POTS line, ISDN B channel, IP or GSM connection. If you are using stereo or dual mono profiles, two ports and two *Codec channels* for each connection will be used.

22.13.2. Connection Manager Connections and Custom Profiles

It is always necessary to specify *Ports* and *Codec channels* in the ***Connection Setup*** menu when you are using the Connection Manager to connect your codecs. This is because these connections are made using custom profiles and ports will need to be selected for custom profiles.

Important note: If you copy manual default profiles into another profile, you will need to physically select the *Ports* and *Codec channels* required in the ***Connection Setup*** menu.

22.13.2.1. Using the Connection Manager

If you are not using manual default profiles to connect, ensure the *Port* and *Codec channel* settings on the non-dialing codec resemble the settings on the dialing codec. Ensure you don't select conflicting *Port* and *Codec channel* settings if you are configuring more that one connection.

22.13.3. I-Mix Port Configurations

22.13.3.1. POTS

If you are using two POTS connections with an ***i*-Mix *G3***, *Port 1* refers to the on-board POTS modem and *Port 2* refers to the right hand module.

22.13.3.2. ISDN

If you are using two ISDN B channels, *Port 1* refers to Channel 1 and *Port 2* refers to Channel 2.

22.13.3.3. GSM

If you are only using one GSM connection, either via a GSM module or a cell-phone connected to the RS232 port, it will always connect using *Port 1.* If you are using two GSM connections together, *Port 1* will be allocated to the GSM module and *Port 2* will be allocated to the cell-phone connection.

22.13.3.4. IP

When configuring IP connections only one *Port* is used for each connection. The IP1 connection on your codec will use *Port 1* and the IP2 connection on your codec will use *Port 2.*

Remember to select a different *Codec channel* for each connection if you are configuring two IP connections.

22.13.3.5. X.21

When configuring X.21 connections only one *Port* is used for each X.21 module connection. For an ***i*-Mix *G3***, the X21 1> connection on your codec will use *Port 1.*

22.13.4. Commander Port Configurations

22.13.4.1. POTS

If you are using two POTS connections with a ***COMMANDER G3***, *Port 1* refers to the left-hand module POTS modem and *Port 2* refers to the right hand module modem. If only one module is inserted it will use *Port 1.*

22.13.4.2. ISDN

As you can insert two ISDN modules into a ***COMMANDER G3***, you could have four *Ports* available which correspond to the four available B channels. As you can connect with a maximum of up to four B channels, you will need to select the *Ports* corresponding to the B channels you wish to connect with.

If two ISDN modules are installed, *Ports 1* and *2* correspond to the two left hand module B channels and *Ports 3* and *4* correspond with the two right hand module B channels.

22.13.4.3. GSM

If you are only using one GSM connection, either via a GSM module or a cell-phone connected to the RS232 port, it will always connect using *Port 1.* If you are using two GSM connections together, *Port 1* will be allocated to the GSM module and *Port 2* will be allocated to the cell-phone connection.

22.13.4.4. IP

When configuring IP connections only one *Port* is used for each connection. The IP1 connection on your codec will use *Port 1* and the IP2 connection on your codec will use *Port 2.*

Remember to select a different *Codec channel* for each connection if you are configuring two IP connections.

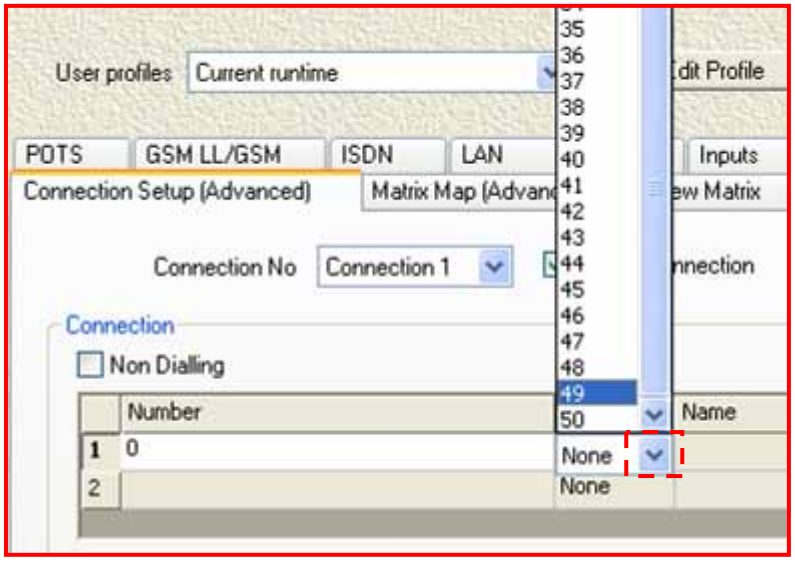
22.13.4.5. X.21

When configuring X.21 connections only one *Port* is used for each X.21 module connection. The X21 1< connection on your codec will use *Port 1* and the X21 2> connection on your codec will use *Port 2.*

Remember to select a different *Codec channel* for each connection if you are configuring two X.21 connections.

22.14. Phonebook Numbers

To select a phone number from the *Phonebook* menu, you first have to enter the number and details with the ***Phonebook Editor*** within ToolBox. (See the section titled ***Phonebook Editor*** within this manual). Up to 80 phone numbers and names can be stored in the configuration phonebook. To recall a stored phone number from the phonebook, click on the right-hand side of the *Phbk#* selection box as shown.



Once chosen, the name of the selected entry will be displayed in the *Name* box and the phone number will be displayed in the *Number* box.

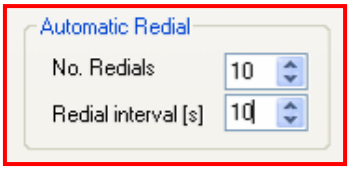
Phone numbers can also be manually entered by clicking on the *Number* box and typing the numbers in. If you are using *Bonded POTS, ISDN* or other bonded connections, you will need to input a phone number for each individual line to be connected. Please note that you can only recall numbers and type in numbers for custom profiles, not manual default profiles.

Important Connection Information:
Use only one phone number in the dialing section of the *Connection* menu if you are using non-bonded connections. For example, if you are using two connections that are not bonded (as you may for the *Man Dflt Mono/IFB* profile), create two connections using the *Connection No* drop-down menu and input one phone number for each connection.

If you are connecting using a bonded connection, i.e. *Bonded POTS* or *Bonded ISDN*, you will need to input multiple numbers into one connection. To do this you will need to select the bonded *Connection type* you desire from the drop-down menu. It is possible to dial up to two POTS and four ISDN numbers once you select a bonded connection.

If you connect using the Connections Manager the codec will dial up to two POTS and four ISDN numbers for you. If you connect manually after you have loaded the information onto the codec, you will need to dial the numbers from the codec via the displayed connections on the LCD screen.

22.15. Automatic Redial



If a **Tieline** *G3* codec loses its connection it can automatically redial until the link is re-established. If *Automatic redial* is not programmed, the codec will reconnect or redial initially as per the *Failover profile* settings (if a *Failover profile* has been programmed), or as per the *Auto reconnect* settings in the codec.

To redial utilizing the *Automatic Redial* function, simply press dial when a connection is lost and the codec will redial as per the *No. Redials* setting.

A Cautionary Note: when using a pair of **Tieline** codecs in this mode, one needs to be set for *Automatic redial* and one for *Auto answer* if you want the codecs to answer automatically. Do not set the *Auto reconnect* function if you are using *Automatic redial* as they will conflict with each other.

22.15.1. No. Redials

As per this setting, a codec will automatically redial until the number of redials selected is reached, or a link is established.

22.15.2. Redial Interval

This represents the time in seconds between redial attempts. An interval is inserted into the number by pressing the # key on the codec down for about a second. Using the parameters shown in the previous image, the codec will automatically dial a maximum of ten times or until a link is established. There is a delay of ten seconds set between dialing attempts.

22.16. Failover Profile in Detail

As previously mentioned, the Connection Manager will handle multiple connections, such as dual channel ISDN or POTS, or a POTS and an ISDN connection together. The codec can reconnect a lost connection either by using the *Auto reconnect* or *Automatic Redial* functions within the codec.

It is also possible to purchase the optional failover (*Failover Profile*) solution to create a backup program feed. An example of this could be an ISDN connection with one or two 'B' channels in use and a POTS connection that can be in *Standby* mode.

Failover Profile is a very advanced function and many users will not need to use it. Essentially it allows the user to create backup profile connections to ensure the continuity of critical signals such as main program links. For this function to work satisfactorily, you must ensure that matching configuration files (i.e. a POTS backup for an ISDN connection) are loaded on both the master and slave codecs. These files contain matrix, profile and phonebook information. The following information will explain reconnecting lost connections in detail and help you to understand what the best reconnection solution for your situation is.

22.16.1. How Can We Manage Connections?

To understand how reconnecting and how failover operates, it is useful to know how the following functions operate within **Tieline** *G3* codecs:

- *Automatic Redial*;
- *Failover profile*; and
- *Auto reconnect*.

If you are programming a failover connection, it is important understand how the codec will operate if a main connection is lost. The codec will:

- First look at the *Automatic Redial* settings and if the codec is programmed to redial a certain number of times, it will attempt to redial the previous connection.
- If *Automatic Redial* is not set, or the redials have completed, then the codec will look to see if a *Failover profile* has been programmed within the codec's main connection profile. If programmed, it will attempt to activate the *Failover profile* settings.
- If *Failover profile* is disabled, the codec will redial the lost connection if *Auto reconnect* is enabled within the codec.

Now that you have an understanding of the connection prioritization procedure within the codec, we can look at how to set a failover connection in more detail.

Important note:
If you are using the *Failover profile* function with your codec, we recommend that you don't activate the *Auto reconnect* function as they will conflict with each other.

22.16.2. Which Solution is Best for My Situation

Determining what you want the codec to do and how to program it is easy - once you understand how a codec manages connections.

In simple terms, if you simply want the codec to redial a connection if it is interrupted for some reason, it is best to either set *Automatic Redial* or *Auto reconnect* within the codec. The basic difference between these two connections is that you can set the amount of redials using *Automatic Redial*, whereas *Auto reconnect* will continue to redial until a connection is made. In addition, with *Automatic Redial* you will need to initiate the call by dialing manually when the connection is lost. It will then automatically try to redial. With *Auto reconnect*, the codec will automatically try to redial when a connection is lost.

If you purchase the failover option from **Tieline** you can also program more complex failover reconnections into your codec. Failover can be divided into two parts. First, there is simple failover which utilises codec user functions to reconnect. Second, there is a more complex failover setup which allows you to use *Standby* connections and which gives you the ability to program profiles into the codec if a connection is lost.

22.16.3. Enabling a Codec for Failover Operation

Once you have purchased the option from **Tieline**, to use failover functions you will need to input a 12 digit code into your codec. To access the Failover Enable menu on the codec, which allows you to enable failover, please do the following:

- Press SOFTKEY 4 on the codec to access the Menu submenus;
- Scroll using the MS to Configuration and press the MS to select it;
- Then scroll to Advanced with the MS and select this option;
- Scroll to Failover Enable and select this option;
- Select Enable and then enter the code that you have been given by **Tieline**.

Your codec will now be programmable for failover configurations.

22.16.4. Programming Simple Failover via a User Function

Simple failover can be programmed by simply enabling failover in the codec via the Failover Enable menu in the codec and then programming a User Function to initiate the failover. The benefit of using simple failover is that you won't need to create a new profile to program as the *Failover profile*. You can simply use the same manual default profile or other profile you have created, and that you are using for your main connection. (This can't be done if you are using a *Standby* connection running concurrently. You will need to program a more complex failover connection).

With the new **Tieline** *Phonebook Editor* it is also possible to change the profile by dialing a new phone book entry that has a different profile associated with it.

Once you have enabled Failover Enable within the codec you will need to create a User Function within the codec by using the *Functions* menu within ToolBox software. This User Function will tell the codec to failover to a new connection as it is programmed within the *Add/Edit Function* menu. In the following example, the main connection is via an ISDN B channel and the failover connection is POTS.

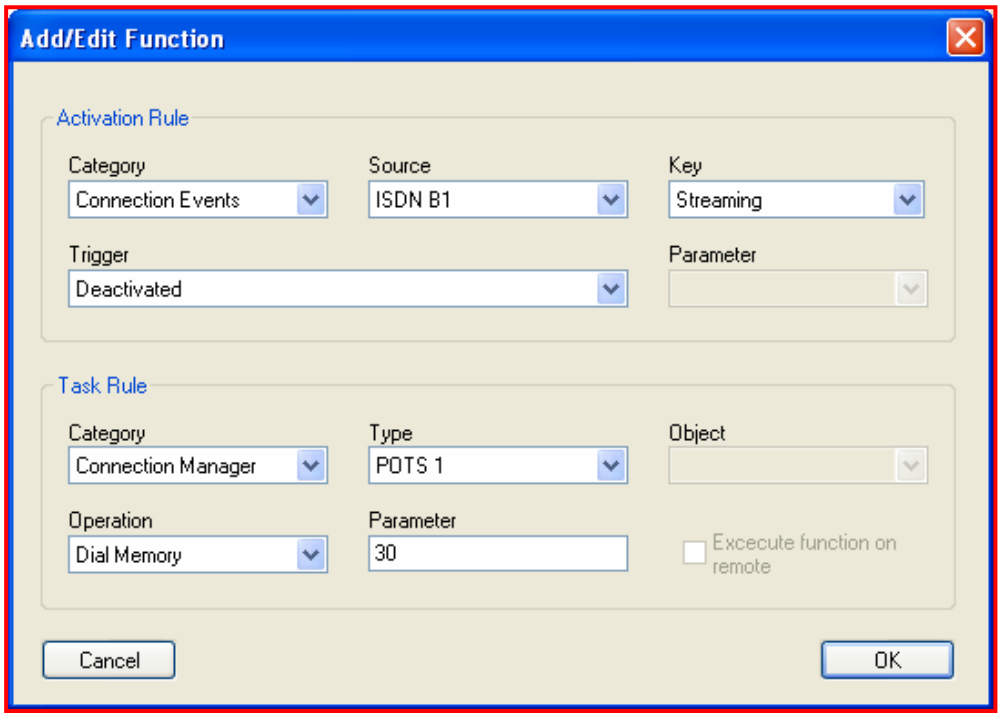


Figure 69: Simple Failover User Function

Note that if audio streaming is deactivated for some reason on the *ISDN B1* connection, the codec *Connection Manager* is set to dial a POTS connection via *POTS 1*. The number to dial is set to be phone number 30, which has been programmed into the *Phonebook Editor.* This phone number can have a POTS dialing profile associated with it so the codec will automatically load the new profile when the user function is activated.

22.16.5. Limitations of Simple Failover

The main limitation of simple failover via user functions is that the connection is down for the period that it takes to redial a new connection. This can be eliminated by using more complex failover methods, which are explained in the following sections.

For more information on programming User Functions, please see the *Functions* menu section within this manual.

WARNING: Please ensure that you don't set *Automatic redial* or *Auto reconnect* if you are creating a User Function as previously shown – otherwise it will conflict with the Connection Manager settings.

22.16.6. Complex Failover Setups: The Standby Connection and Failover Profile Relationship

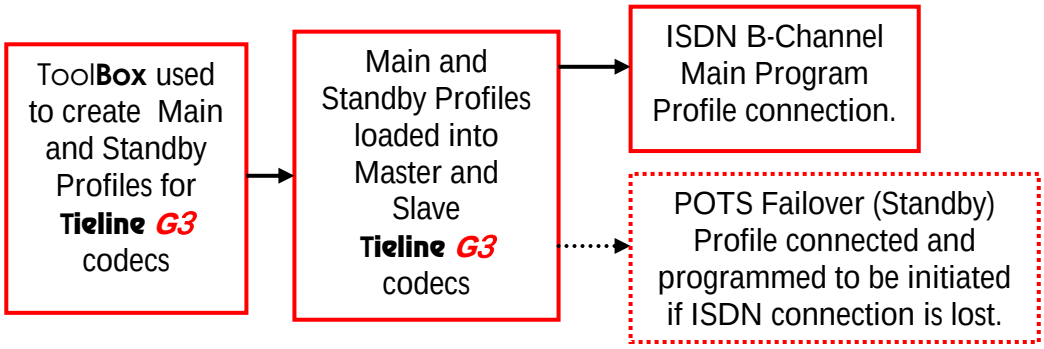
A *Standby* connection can run concurrently with a main connection as a backup program connection. Previous elements of this section of the manual have discussed how to set up a connection. A *Standby* connection is set the same way as other connections - the only difference is that you check the *Standby* box in ToolBox to create the connection.

If you wish to use set up a *Standby* connection and utilize the *Failover profile* facility, you will need to ensure the following has been set up:

1. A *Standby* connection, i.e., *Connection no. 2*, is programmed with a phone number and the *Port* and *Codec channel* has been programmed into the main program connection's profile.
2. *Failover profile* on the main connection has been set to the profile which will be used for the *Standby* connection.

For example, if your main program connection is via either one or two ISDN B channels, a *Standby* POTS connection can be connected alongside the main connection, but no audio will be sent unless the main connection fails. Only then will the connection go active and an encoder be assigned to this connection.

To assist in conceptualizing this, in the flow chart example below, the codecs have been configured to have a main connection for program sound via an ISDN B-channel and a *Failover* or *Standby* connection via a POTS connection.



To set up this configuration, you need to set up the main ISDN *Connection no 1* as the primary connection. In *Connection no 2* set up the same connection information as the settings in the *Failover profile*, in this case the POTS connection. You need to ensure that the same phone number and *Port* is set up in *Connection No 2* of the ISDN main profile as the POTS failover profile that you have set, and check it as the *Standby connection*.

Following is an image showing how you would set up *Connection 2* in your main profile, as per our example. Note that the *Standby* box is checked, the *Connection type* is POTS and that *Port 1* is checked. All these settings are the same as those made in the *Failover profile*. In addition the phone Number matches the POTS Failover profile that is programmed.

The *Codec channel* doesn't need to be allocated in this connection because it will not be sending audio until failover is initiated, i.e. it is in *Standby* running concurrently with the main connection. When failover occurs, the *Failover profile* settings will be activated and these should have the *Codec channel* allocated.

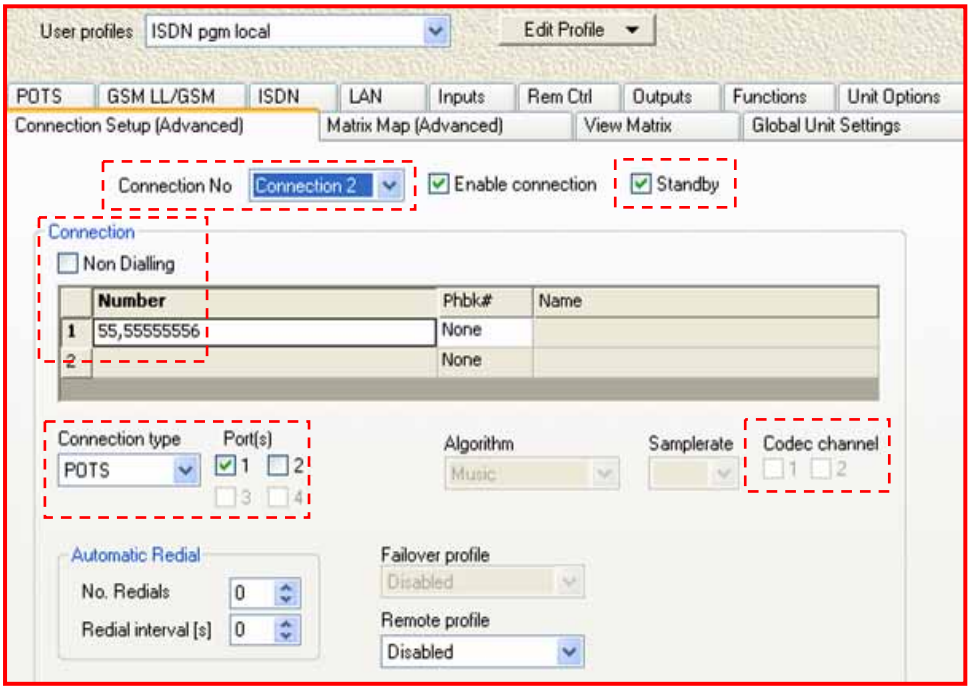


Figure 70: Failover Connection Dialing and Port Allocation Programmed into a Main Profile

The next image shows the *Failover profile's* settings. Please note that the phone *Number* and *Port* settings are the same as in *Connection 2* (shown previously) of the main ISDN connection. In addition, a codec channel has been allocated which will facilitate sending a program signal if the failover profile is activated. Also, please note that the *Remote profile* is set to be the same profile as the *POTS failover* profile. This is so that the *POTS failover* profile will be loaded onto both codecs as required in the event of failover being activated.

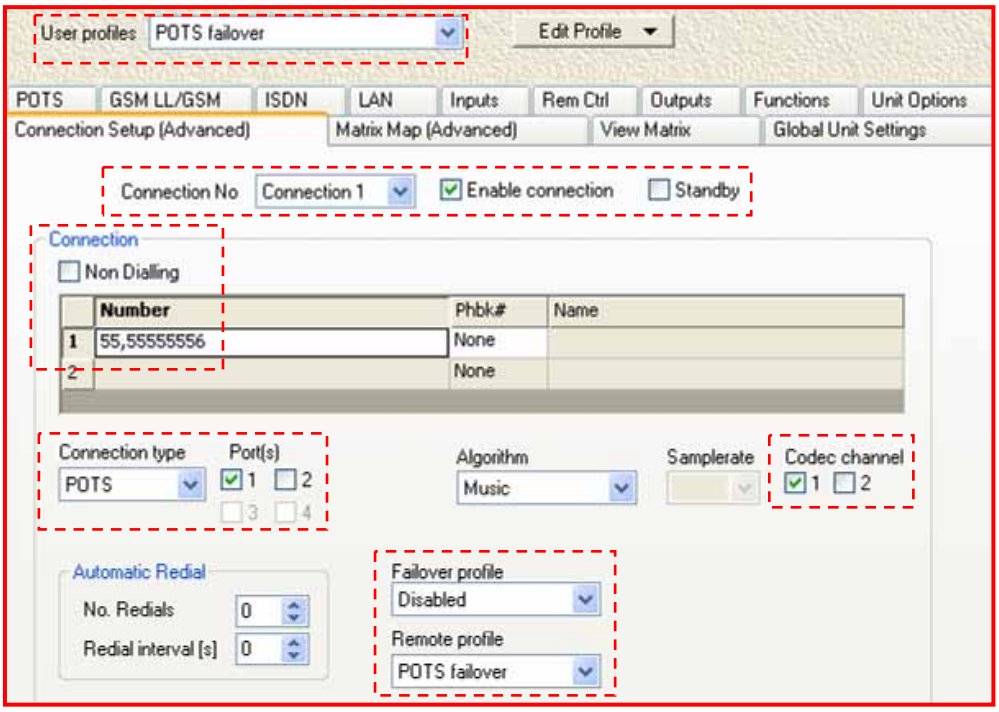


Figure 71: Failover Profile Connection Settings

The next image displays the main connection settings. You will note that the main connection *User profile* is called *ISDN pgm local* and is set as a single B channel connection. You will also note that the *Failover profile* is set to be a preprogrammed *POTS failover* profile. *Remote profile* on the remote codec is set to match the local ISDN main connection and is called *ISDN pgm remote*. *Remote profile* settings match the local codec's settings and will initiate the backup POTS connection if it is required (in other words, a matching *POTS failover* profile is also loaded into the remote codec and will become active in the event of failover being activated).

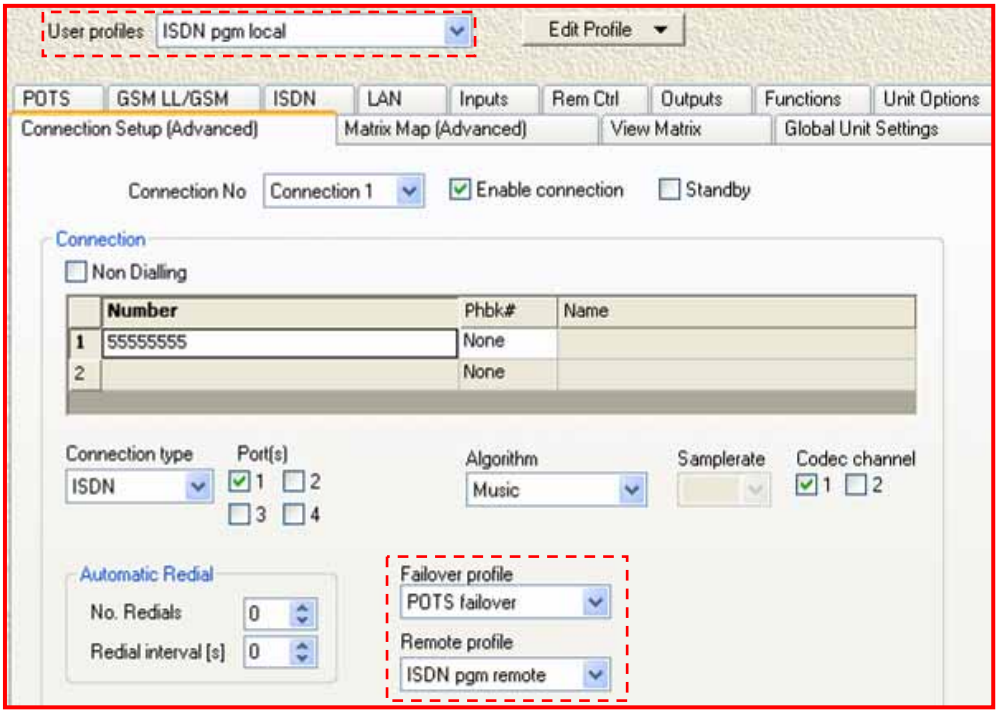


Figure 72: Main Profile with Remote and Failover Profile Settings

If the all the preceding settings are done correctly and the codec detects a lost main connection, it will automatically switch to the selected *Standby* connection and the POTS *Failover profile* will become the active profile on the codec LCD and within the connected codecs.

Important Connection Note:
If you don't put the same phone number (etc) in both *Connection no 2* of the ISDN profile and the POTS *Failover profile*, then if the failover connection is initiated, it will not automatically change over to the second connection running concurrently. Instead, it will dial the POTS *Failover profile* connection (this may cause up to 30 seconds delay to your program signal).

22.16.7. Non-Concurrent Connections

If you do not want your failover connection to be concurrently connected (i.e. connected while the main program connection is in use), then you wouldn't set *Connection no 2* as a *Standby* connection in your main program ISDN profile. Just set the POTS *Failover profile* as per your dialing configuration etc, and if the main ISDN program feed is lost, the POTS *Failover profile* will dial the connection to connect.

22.16.7.1. Another Reconnection Option

If you are using a single POTS connection, you could protect against a loss of signal by setting the *Failover profile* to the same profile you are using as your main POTS connection. Remember, if you decide not to use *Failover profile* and you want the codec to reconnect after the loss of a connection, you will need to either program *Automatic redial* or 'check' the *Auto reconnect* function which is within the ***Unit Options*** menu.

Please note: The *Failover profile* facility is 'grayed' out and inactive until the *Standby* box is checked to enable the setting up of a standby connection.

Important Failover Footnote:
If you are using *Failover profile* we recommend that you should failover to a different connection type than your main connection. For example if your main connection is via an ISDN line, we recommend you failover to IP, POTS or GSM etc. This will safeguard you in case your incoming lines have been damaged or unplugged.

In addition, we recommend that you only use the *Music* or *Voice G3* algorithms for failover connections. This is because if you use other algorithms to connect, you will need to manually change the profile setting at the remote codec (*Remote profile*) in the event of a failover connection being initiated. The remote profile can change automatically if you use the *Music* or *Voice G3* algorithms.

22.17. Remote Profile

Remote profile sets the profile on the remote codec. The remote profile should match the profile on the local codec. If you are using manual default profiles this will automatically be set to reflect the profile selected on the dialling codec.

22.18. A Typical Connection Setup Procedure

A typical setup procedure for the ***Connection Setup*** menu may include the following:

- Select your desired *Connection No.*;
- Check the *Enable Connection* Box to allow parameters to be adjusted;
- Decide if the connection will be a main or Standby signal path;
- Decide if you are dialing out on the codec and if you are, whether to retrieve a number from the *Phonebook*, or type in the desired phone number;
- For custom profiles, select your *Connection Type*;
- For custom profiles, select the *Port/s* (modem) you will be using;
- For custom profiles, select the *Codec Channel/s* you will be using;
- For custom profiles, select the desired *Algorithm* you require for each channel;
- Select the desired *Automatic redial* parameters if this function is being used;
- For User Created profiles, select the *Failover profile* (for advanced users); and
- For User Created profiles, select the *Remote profile*.

A Cautionary Note on Auto Dial/Redial/Reconnect:
It is recommended that one codec be considered the local (master) codec and the other codec will therefore be the remote (slave). An easy to remember arrangement is for the studio codec to be considered the local codec and the field codec the remote unit. At least everyone knows which way things are supposed to be. (Related Topic: Local and Remote Codec Operation Explained)

If in point-to-point configuration (ISDN operation), the dialing codec should be considered the local (master) unit and the answering unit the remote (slave) codec. The codec that initially made the successful link-establishing call will redial a number to reconnect should a link failure occur.

Section 23. POTS Tab

When you click on the *POTS* tab in the *Profile Editor* menu there are several settings available for programming the function of your codec in POTS mode. (Please note: Even if you don't have ToolBox software connected to your codec, these functions are available by accessing your codec menu via SOFTKEY 4 Menu. Next select Configuration and select the POTS connection you wish to configure. Now scroll through the menu and adjust the settings you wish to change. See the Configuration menu diagram for details of this menu structure.)

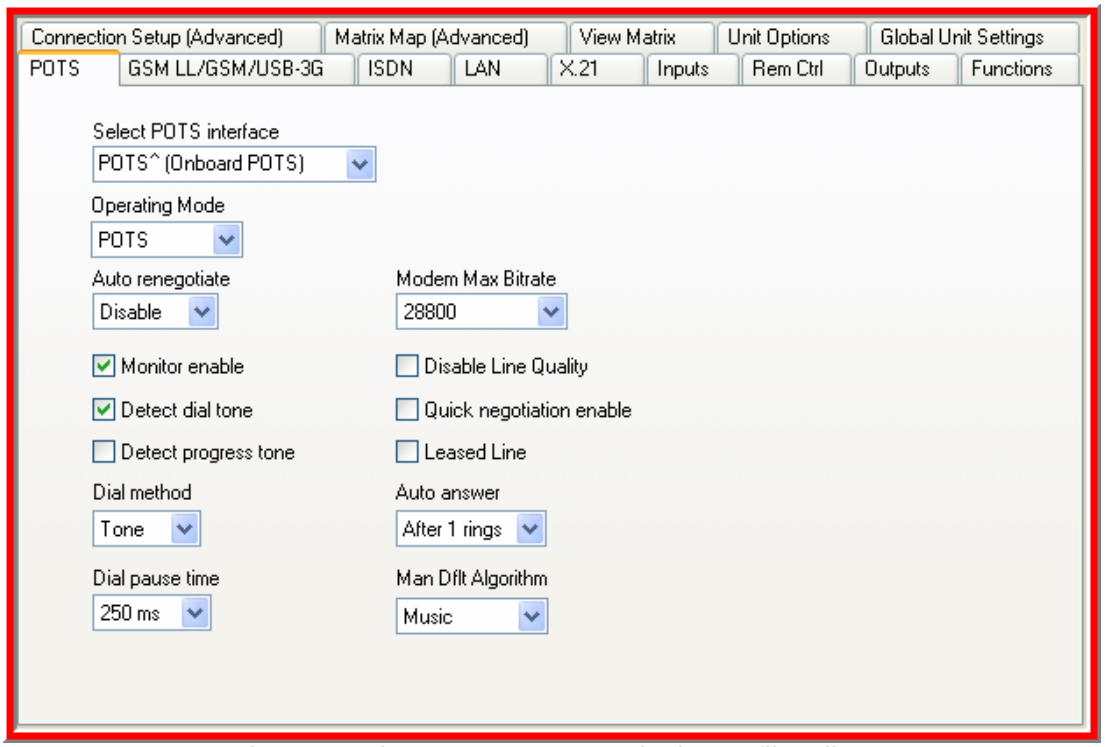


Figure 73: The POTS Menu Page in the Profile Editor

23.1. POTS and POTS G3 Module Differences

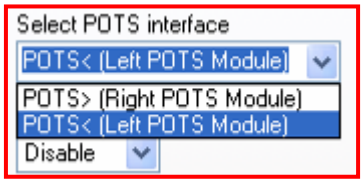
The new POTS G3 module dials a little differently to the original POTS module. Please see the section in this manual titled POTS versus new POTS G3 Modules for detailed information about using these modules, differences in connecting and default connection bit rates.

Very Important Renegotiation Note:

After initially connecting, if you attempt to shift the connection bit-rate **upwards locally** with a POTS G3 module, the codec will **'retrain' upwards, not renegotiate** like the original POTS modules. Retraining takes significantly longer than renegotiation – up to 30 seconds.

Connection Tip:
If you are connecting a POTS module to a new POTS G3 module, use the POTS module in the local codec. This will allow you to renegotiate the connection upwards seamlessly if required, without the delay of retraining.

23.2. Select POTS Interface

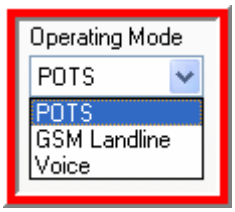


Your codec is capable of using two POTS connections. This drop-down menu allows the user to select between two possible connections and in turn configure both connections. To select the POTS connection you want to configure, simply click on the right-hand arrow, highlight the desired connection and click on it to select it.

Please note: Depending on the codec you are using, this menu will display slightly differently. The preceding image shows a ***COMMANDER G3*** codec with two POTS modules installed. An ***i*-Mix *G3*** codec will display the on-board POTS connection and a right-hand POTS module if this is installed.

23.3. Operating Mode

This setting is the same setting as the setting in the ***GSM LL/GSM/USB-3G*** tabs. If a selection is made in either menu it will change in both menus. There are three operation modes that include the following selections.



23.3.1. POTS

Select this operation mode when connecting via a standard POTS connection.

23.3.2. GSM Landline

This function needs to be selected if the codec you are using is to dial via a landline and will be communicating with another codec using a GSM cell-phone. Only the landline-connected codec needs to select this function.

23.3.3. Voice

Select this operation mode when connecting using a standard voice call. If *Voice* is selected then redundant settings in the ***POTS*** tab will be greyed out.

23.4. Auto Renegotiate: Overview

Very Important Notes:
Quick Negotiation Enable is not necessary with the new POTS G3 module and it has been removed from the POTS G3 module menus. It is still visible in ToolBox software but has no impact if you are using a POTS G3 module.

In addition, the Auto and Up options have been removed from the Reneg (Fast) POTS G3 module menus.

In the modem stabilization period this setting is not active as *Auto* is the default setting.

During the modem connection procedure the modem will renegotiate up if the line quality is above 60% in stabilization mode. Renegotiation will occur up to a possible maximum of 33,600 bps (as per the maximum bit rate connection settings in the codec).

If line quality falls below five per cent then the modem will always renegotiate down. Audio will be unusable if line quality is this low.

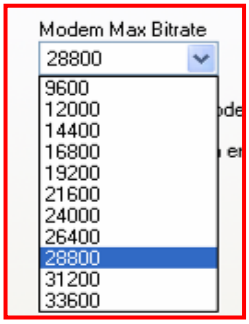
There are four different settings in relation to Line Quality (LQ) in the pull-down menu within ToolBox.

Disable **is the setting for live program links** because it prevents data renegotiation if line conditions change. The use of **auto-renegotiation can cause unexpected program breaks** with changes in line conditions.

If *Up* is selected, a higher bit rate is used if line quality at both ends is above 80%. If *Down* is selected, a lower bit rate is used if line quality falls below 20% at either end. If *Auto* is selected auto-renegotiation occurs as line conditions vary and this can be either up or down.

While *Auto renegotiate* is disabled, monitor the Line Quality to ensure it remains higher than 20%, otherwise the codec may become unstable.

23.5. Modem Max Bit-rate



This pull-down menu sets the maximum bit rate at which the codec will attempt a connection. The line may be capable of establishing a connection at a higher bit rate, but this is the highest figure that connection will be attempted at. Reducing this value can improve connection reliability on poor quality lines.

The search for a higher bit rate will stop when either of the following happens:

- The bit rate reaches the maximum bit rate specified by the menu;
- The bit rate reaches the maximum codec bit rate of 33,600 bps; or
- The codec is using a less than perfect line and the codec can only negotiate a bit rate lower than the specified maximum bit rate. If the codec is operating at a line quality of less than 70%, it won't renegotiate up any further.

Actual line quality is displayed on the codec LCD screen.

23.6. Monitor Enable

The modem dialing and connection process can be heard through the headphones by checking the *Monitor enable* function. When checked, dial up (DTMF) tones are heard in the headphones. If you are having difficulty connecting, it I useful to hear the connection tones to ensure your line is operating effectively.

23.7. Detect Dial Tone

By checking the *Detect dial tone* function the codec will wait until a dial tone is present before dialling.

If left unchecked, the codec will **not** wait until a dial tone is present before dialling, it just commences dialling. Use this setting if there is any difficulty in detecting a dial tone.

23.8. Detect Progress Tone

If this function is unchecked, the codec does **not** test for a telephone system connection tone at the conclusion of dialing. (e.g. "busy" tone). It assumes successful link establishment, and starts line negotiation.

If this function is checked, the codec tests for telephone system connection tones at the conclusion of dialing. (e.g. "busy" tone).

23.9. Disable Line Quality

In rare situations it is possible that a POTS connection may drop out for no apparent reason, even if the line quality is good. If this occurs, it is possible that stability may be increased by disabling the line quality (LQ) display on a codec's LCD screen. This will not affect codec audio settings at all – it will only remove the line quality display from the LCD screen. Do not check the *Disable Line Quality* box unless you are having disconnection problems.

23.10. Quick Negotiation Enable: Old POTS Module Only

When this box is checked the modem will attempt to connect at the highest bit-rate automatically. If this box is not checked the modem will attempt to connect initially at 19,200 bps and then negotiate the highest bit-rate.

Very Important Notes:

Quick Negotiation Enable is not necessary with the new POTS G3 module and it has been removed from the POTS G3 module menus. It is still visible in ToolBox software but has no impact if you are using a POTS G3 module.

In addition, the Auto and Up options have been removed from the Reneg (Fast) POTS G3 module menus.

23.11. Dial Method

This pull-down menu selects either *Tone* (DTMF) dialling or *Pulse* dialling.

23.12. Leased Line

This option needs to be checked for operation over leased 2-wire bi-directional lines. As the codec modems require a loop voltage source for correct operation, an optional line interface unit is available for use with dedicated 2-wire circuits where the loop voltage is not present.

Important note on operating in Leased Line mode:

When leased line mode is enabled the dial tone and progress tones are disabled. In this mode it is necessary to manually answer the call at the remote codec initially. If you want the remote codec to automatically re-answer after losing a connection, you should enable Auto Reconnect on both the remote and local codecs. Then if you lose a connection for some reason, after you have already made a successful call, the codec should auto-reconnect.

To enable Auto Reconnect using the codec menus, press SOFTKEY 4 Menu, then select **[Configuration] > [Auto Reconnect] > [Enable]**. The Auto Reconnect function will now be enabled.

If Auto reconnect is enabled in the codec menu or via ToolBox software, it will attempt to redial indefinitely in the manual default profiles. If you wish to adjust the redial setting in the codec to only dial a certain amount of times, you will need to use the *Automatic Redial* function in ToolBox software. Do not enable both the *Automatic redial* and the *Auto reconnect* functions at the same time as they will conflict with each other. For more information please see the sections on *Automatic Redial* and *Auto reconnect* in this manual.

23.13. Auto Answer

This pull-down menu allows the user to enable or disable automatic answering of incoming calls to start operation. The number of rings before answering can also be set from between zero and ten rings.

When *Auto answer* is selected, the codec will automatically answer incoming calls and begin the link-establishing process.

If *Auto answer* is not selected, the incoming call must be answered by pressing the ANSWER button on the codec front panel before the negotiating process will commence.

Operational tip: You can use *Auto answer* with *Automatic Redial* to maintain a program link over extended periods. If these functions are set correctly, the codec will automatically restore the link if the connection fails.

23.14. Dial Pause Time

A pause is inserted by holding down the # key until a comma (,) appears on the display. This function determines the length of any dialing pause if any are inserted into the telephone number. The length of the pause is set with the pull-down menu. This feature is useful when dialing through older type PBX systems that have latency in connecting to the PSTN.

When dialing out through a PBX, the codec must find a line on two occasions:

- At the beginning of the call, and
- When the call goes into the Public Switched Telephone Network (PSTN) from the PBX. The PSTN takes time to prepare the line for an outgoing call, i.e. disabling the incoming call circuitry.

It is also useful when making a long distance or international calls e.g. dialing **Tieline** 1,3178458000. Pauses can be set from zero up to two seconds in 250 millisecond increments. (Related Topics: *Phonebook Editor*)

23.15. Manual Default Algorithm

The *Manual Default Algorithm* setting in the *POTS* menu relates to the *Manual Default Profiles* within ToolBox. The setting you choose here will be the algorithm setting for any POTS *Manual Default Profiles* that you connect with. The setting will also be used by custom profiles with connections that use the *Any* setting within the *Algorithm* drop-down menu in the *Connection Setup* tab. (Related Topics: Manual Default Profiles and *Algorithms*)

Section 24. GSM LL/GSM/USB-3G Tab

This menu allows users to configure GSM, GSM landline and USB-3G connection settings within **Tieline** codecs. (Please note: Even if you don't have Tool**Box** software connected to your codec, these functions are available by accessing your codec menu via SOFTKEY 4 Menu. Next select Configuration and select the GSM/3G or POTS (GSM landline) connection you wish to configure. Now scroll through the menu and adjust the settings you wish to change. See the Configuration menu diagram for details of this menu structure.)

The *GSM Landline* section includes setup functions for a codec which is receiving a call from a codec with a GSM module or cell-phone in use. The *GSM Module/USB-3G Module/Cellphone* section includes setup functions for a codec that is connecting using a USB, 3G or GSM module or a cell-phone. The *GSM CSD/HSCSD Settings* section programs GSM settings. The other sections are specific to 3G connections and are explained in detail within the "IP & 3GIP Streaming Reference Manual".

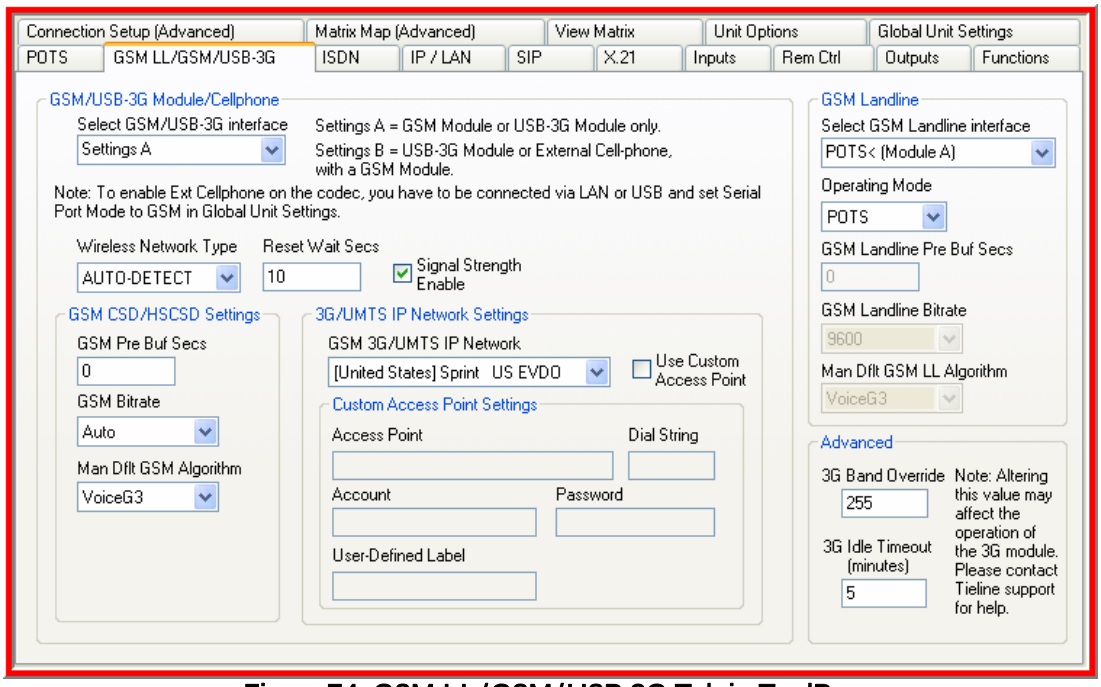


Figure 74: GSM LL/GSM/USB-3G Tab in ToolBox

VERY IMPORTANT WARNING:

GSM plug-in modules cannot be used in TLR300B ***COMMANDER G3*** rack unit codecs. If you must connect using GSM with aTLR300B ***COMMANDER G3*** rack unit codec, we suggest you use a GSM cell-phone attached to the RS232 port on the rear panel of the codec. Alternatively, you could try a GSM phone or a 3G cell-phone in GSM mode with a **Tieline** USB module.

24.1. GSM Connections

When you insert a GSM module into your ***G3*** codec you will notice that the GSM connection will appear in the list of available connections on the connection LCD screen. Relevant GSM menu elements are displayed in the codec configuration wizard. If you are not using ToolBox software to configure your connection, use the wizard via SOFTKEY 3.

Only one GSM module can be used in a ***COMMANDER G3***. If you wish to use two GSM connections you will also need to use the RS232 connection on the rear of the codec and connect a GSM-compatible cell-phone such as the *Nokia*® 6310 and 6310i. ***i*-Mix *G3*** codecs only have one module slot available.

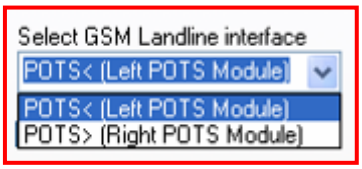
If you are connecting via the RS 232 port on the rear of the codec, or if you need to configure for either CSD, HSCSD or GSM landline modes, please see the GSM Operation section of this reference manual for relevant configuration information.

24.2. 3G Connections

When you insert a 3G module (or a USB module with a 3G cell-phone attached) into your ***G3*** codec you will notice that a 3GIP connection will appear in the list of available connections on the connection LCD screen. Relevant 3G menu elements are also displayed in the codec configuration wizard. If you are not using ToolBox software to configure your connection, use the wizard via SOFTKEY 3.

Please see the "IP & 3GIP Streaming Reference Manual" for more information relating to programming and dialing 3GIP and IP connections and for the latest ToolBox software changes for 3GIP connections.

24.3. Select GSM Landline Interface



This drop-down menu allows the user to select between two possible connections and in turn configure both connections. To select the POTS connection you want to configure, simply click on the right-hand arrow, highlight the desired connection and click on it to select it. Once you have selected the module you can then view and amend the following settings as required.

Please note: Depending on the codec you are using, this menu will display slightly differently. The preceding image shows a ***COMMANDER G3*** codec with two POTS modules installed. An ***i*-Mix *G3*** codec will display the on-board POTS connection and a right-hand POTS module if this is installed.

24.4. Operating Mode

This setting is the same setting as the one in the *POTS* tab. If a selection is made in either menu it will change in both menus. There are three operation modes that include the following selections.

24.4.1. POTS

Select this operation mode when connecting via a standard POTS connection.

24.4.2. GSM Landline

This function needs to be selected if the codec you are using is to dial via a landline and will be communicating with another codec using a GSM cell-phone. Only the landline-connected codec needs to select this function.

24.4.3. Voice

Select this operation mode when connecting using a standard voice call.

24.5. GSM Landline Pre Buffer Secs

The nature of GSM transmissions means information is sent in 'bursts'. Sometimes this can lead to program drop-outs. If you find you are getting program drop-outs, you can attempt to circumvent them by delaying the time-frame in which packets of information are received. In effect you are creating a buffer.

1-2 seconds is usually sufficient to avoid any problems but the delay can be set up to 9 seconds if required. Program signals will be delayed by the amount of time set in this menu. You only need to adjust this setting if the Landline connected codec is having problems receiving audio. If you are still getting program drop-outs after adjusting this setting, it could be due to network congestion. Hang up the connection and try to reconnect in a short while. (Related topic: GSM Operation)

24.6. GSM Landline Bitrate

The GSM landline bitrates needs to be set **at the codec using a POTS connection**. This setting is only active if you have checked the *GSM Landline Mode* box for your POTS connections. If you have not checked these boxes, the *GSM Bitrate* settings will be greyed-out.

You only need to set the *GSM Bitrate* setting if you are connecting your codecs using a GSM and landline connection. If you are connecting two codecs that are both using GSM connections it is not necessary to set the bit rate - this will automatically be determined.

To determine the correct setting it is best to check the connection bit rate possible at the codec which will be using a GSM connection, i.e. try to connect at

the GSM end and see what the connection bit rate is. Then enter the bit rate possible at the codec using the POTS landline connection and attempt to connect.

24.7. Manual Default GSM Landline Algorithm

The *Man Dflt GSM LL Algorithm* setting in this menu relates to all manual default profiles within Tool**Box.** The algorithm setting you choose here will relate to any GSM *Manual Default Profile* connections.

Important Algorithm Selection Information

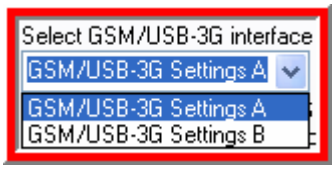
The manual default profile algorithm settings in the ***POTS, GSM LL/GSM/USB-3G***, and ***ISDN*** menus of the Tool**Box** ***Profile Editor***, select which algorithm will be the default for manual default profiles using these connections. For example, if *Music* is selected in the *Man Dflt Algorithm* setting in the ***POTS*** menu, all manual default profiles connecting via a POTS connection will use the *Music* algorithm.

This can be overridden in two ways. If you are using the ***Connection Setup (Advanced)*** menu to set your connections, you will note that a *Connection Type*'s *Algorithm* setting can be changed. If *Any* is the selected algorithm, the algorithm for each *Connection Type* will be the one selected for manual default profiles, i.e. the default ones selected in the ***POTS, GSM LL/GSM***, and ***ISDN*** menus. Any other algorithm selections here will override default algorithm settings in the ***POTS, GSM LL/GSM/USB-3G***, and ***ISDN*** menus.

Secondly, you can reset the algorithm using the codec itself by accessing the algorithm selections in the ***G3*** codec menus.

24.8. GSM/USB-3G Module/Cellphone Setup: 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:

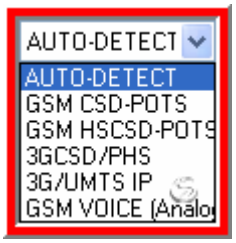


Setting	Module Types Supported
GSM/USB-3G Settings A	Supports the use of either a GSM or USB-3G module only.
GSM/USB-3G Settings B	Supports the use of either a USB-3G Module or external cell-phone, with a GSM module.

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 G3*** 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.

24.9. 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 GSMCSD call.

If a 3G 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.

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

24.10. GSM Pre Buf Secs (for a codec connecting with a GSM connection)

The nature of GSM transmissions means information is sent in 'bursts'. Sometimes this can lead to program drop-outs. If you find you are getting program drop-outs, you can attempt to circumvent them by delaying the time-frame in which packets of information are sent. In effect you are creating a buffer. 1-2 seconds is usually sufficient to avoid any problems but the delay can

be set up to 9 seconds if required. Program signals will be delayed by the amount of time set in this menu. You only need to adjust this setting if the GSM connected codec is having problems receiving audio. If you are still getting program drop-outs after adjusting this setting, it could be due to network congestion. Hang up the connection and try to reconnect in a short while. (Related topic: GSM Operation)

24.11. GSM Bitrate (for a codec connecting with a GSM connection)

This setting allows you to set the connection bit rate for your GSM connections. The default setting is *Auto* and this is the setting you should use. This is because the GSM connection will automatically detect what the possible network connection bit rate is. If you use this setting the connection bit rate will be detected automatically when you attempt to connect and will be displayed on the codec LCD. (Related topic: GSM Operation)

24.12. Manual Default GSM Algorithm (for a codec connecting with a GSM connection)

The *Man Dflt GSM Algorithm* setting in this menu relates to the manual default profiles within ToolBox. The setting you choose here will relate to any GSM connections in the *Manual Default Profiles* that you select to use with the codec.

24.13. 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*.

24.14. A Final Note on GSM Configuration

For more detailed information on configuring your codec by using the menus on the codec itself, please see the GSM Operation section within this reference manual.

Section 25. ISDN Tab

This menu allows users to configure ISDN connection settings within **Tieline** codecs. (Please note: Even if you don't have ToolBox software connected to your codec, these functions are available by accessing your codec menu via SOFTKEY 4 Menu. Next select Configuration and select the ISDN connection you wish to configure. Now scroll through the menu and adjust the settings you wish to change. See the Configuration menu diagram for details of this menu structure.)

Each **Tieline** codec can provide mono or stereo capabilities using a single B channel. In addition, if you have 2 B channels available, you can use one as a standby or operate in stereo, dual mono or mono IFB modes.

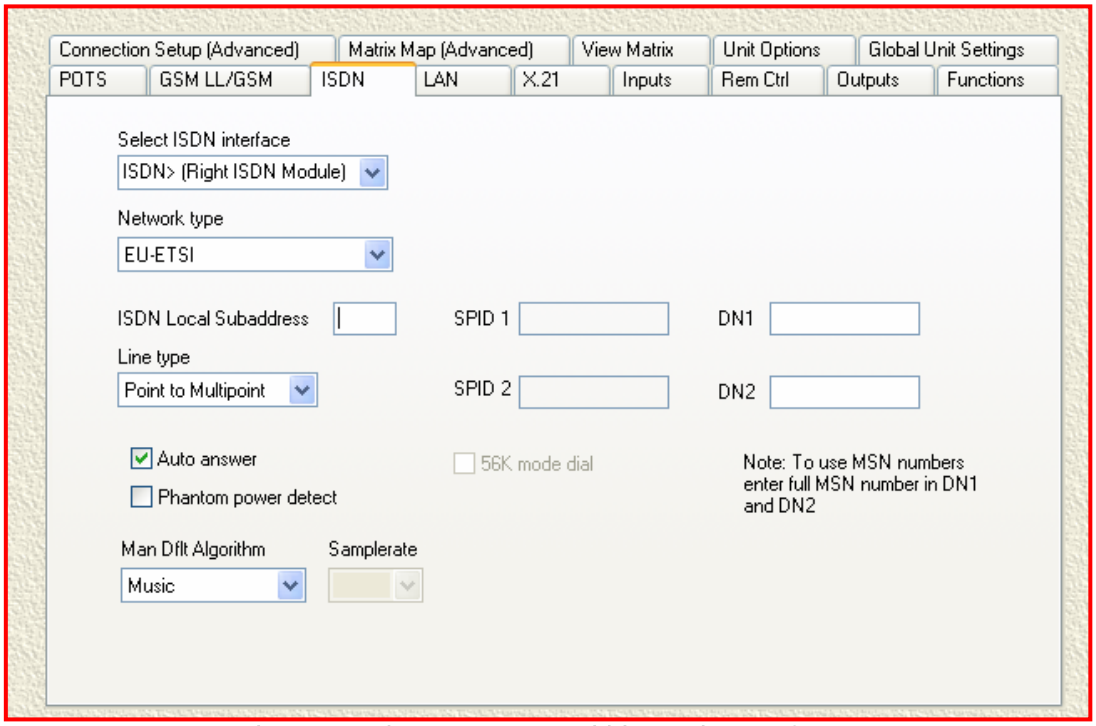


Figure 75: The ISDN menu within ToolBox software

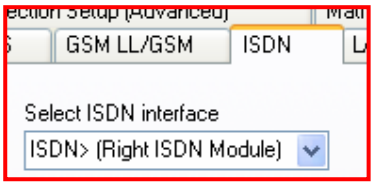
COMMANDER G3 codecs have two module slots available and it is possible to bond up to four ISDN B channels together, using two ISDN modules, to increase the connection bandwidth available for a broadcast. 256 kbps connections can be achieved using this method. This is particularly useful when using MPEG algorithms for broadcasts.

In order to use a **Tieline** ***G3*** codec in ISDN mode you must insert either one or two ISDN plug-in modules and configure these modules by programming the following information in the ***ISDN*** setup menu.

Important Note: ISDN connections will always appear as available on the codec because some networks are 'on-demand' and need to detect a valid network. These networks will not send you ISDN sync information until an ISDN connection is detected. If you try to dial an ISDN connection and the line is not connected, the codec will display No signaling data. Check wiring

25.1. Select ISDN Interface

This drop-down menu allows the user to select the ISDN module you are configuring. To select the ISDN connection you want to configure, simply click on the right-hand arrow, highlight the desired connection and click on it to select it. Once you have selected the module you can then view and amend the following settings as required.



Please note: Depending on the codec you are using, this menu will display slightly differently. The preceding image shows a right-hand ISDN module. An ***i*-Mix *G3*** will only display this module option, whereas a ***COMMANDER G3*** codec will display a left and right-hand module because there are two module slots in these codecs.

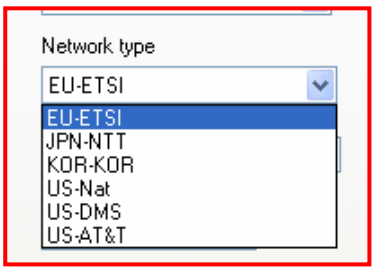
25.1.1. Bonding Four ISDN Channels Using Two ISDN Modules

As previously mentioned, ***COMMANDER G3*** codecs have two module slots available and it is possible to bond up to four ISDN B channels together, using two ISDN modules, to increase the connection bandwidth available for a broadcast.

To program the SPIDs and DNs etc for each module, simply select one module at a time in the *Select ISDN Interface* drop-down menu and enter the relevant information for each modules B channels. Once you have programmed both modules you will need to enter the ISDN number for each B channel in the ***Connection Setup*** tab within the ***Profile Editor***.

25.2. ISDN Network Type

You will need to select the *Network type* in your codec that corresponds with the region in which you are using your codec. To select the network, click on the right-hand blue arrow in this menu to reveal a drop down menu and highlight the network you require – click it to select it.



The list of possible networks that can be used with an ISDN module in ***G3*** codecs is shown in the following table. It will assist you to decide which network type you need to select and program into your codec. If you are in North America, you need to ask your phone Company for the switch type. In other places, EU-ETSI most probably will work.

25.2.1. ISDN Interface for North America and Other Regions

If you are using your codec in North America, it will most likely be connected to an ISDN line with a two-wire 'U' interface. You will need to have a 'U' interface ISDN module installed in your codec to use this type of line.

If your codec is connected to an NT-1 device with a four-wire cable (common configuration outside North America), you need to have an S/T interface ISDN module installed in your codec.

ISDN Software Type Loaded	Networks	Select
US/CAN ISDN (ISDN Type=SGW)	US-Nat	If switch type is National ISDN-1 and 2
	US-DMS	If switch type is Northern Telecom DMS100
	US-AT&T	If switch Type is AT&T Custom
European ISDN (ISDN Type=SGW)	EU-ETSI	If Switch Type is ETSI (UK, Europe, Australia and most other countries)
Asian ISDN (ISDN Type=SGW)	JPN-NTT	If you are in the Japan and your network is NTT
	KOR-KOR	If you are in the Korea

Table 11: ISDN Network Settings

25.3. ISDN Line Type

Ask your phone company whether your ISDN line is Point-to-Point or Point-to-Multipoint. You will need this information to enter into your **Tieline** codec. By default, select *Point to Multipoint*, unless your switch type is an AT&T 5ESS custom point-to-point.

25.4. ISDN Local Subaddress

This setting must be blank by default. Use it only if your codec shares the same phone number with another ISDN device. In this case, enter an arbitrary number (unique for each device) as the Subaddress. To ring a codec with a non-blank Subaddress you must dial the normal ISDN number, followed by "*", followed by the Subaddress.

A Cautionary Note: A Subaddress value of 1 is not permitted in some locations.

25.5. SPID Explained

Each telephone line is assigned a single directory number. When a customer orders services or features they are programmed into a Service Profile stored at the switch. Any call associated with that line can be assumed to apply only to that customer's Directory Number and Service Profile. When a single ISDN device is connected and serviced this way, the configuration is said to be point-to-point.

However, ISDN also supports point-to-multipoint configurations, with multiple and dissimilar devices connected to a single line. In many cases, it is not desirable to share the same directory number and Service Profile. Consequently ISDN relies on an initialization procedure for associating Service Profiles with individual terminating equipment, rather than lines. Terminating equipment that supports this procedure is said to be Auto-Initializing.

The ISDN supplier gives the customer a Service Profile ID (SPID) of up to 20 digits long. The customer programs the number into the terminating equipment which sends it to the switch. The switch then associates the SPID with a specific Service Profile and Directory number. When the switch sends setup messages to a point-to-multipoint line, only equipment with the matching SPID will respond.

25.6. SPID1, SPID2 (Service Profile ID)

Users outside North America please leave the SPID1 and SPID2 blank. North American users need to get SPID (Service profile ID) numbers from the phone company and enter them exactly as supplied.

Some ISDN lines use two SPIDs, some one and some use none at all. If your line was configured for two SPIDs, you must enter the appropriate numbers exactly as they have been given to you by the phone company into the *SPID1* and *SPID2* boxes in the ***ISDN*** menu within Tool**Box**. If your line has only one SPID, program that number into both the *SPID 1* and *SPID 2* boxes. If you have ordered service on only one B channel (1B+D), enter the SPID number into *SPID 1* and leave *SPID 2* blank. Finally, if your line has no SPIDs, verify that there are no numbers programmed into *SPID 1* and *SPID 2*.

The phone company should provide this number exactly as it is to be entered. In many cases however, a Terminal IDentifier (TID) is required at the end of the SPID (usually 00 ,01, or 11) and the phone company will neglect to mention this. If your switch type is an AT&T 5ESS custom point-to-point, you may need to leave both *SPIDs* blank - even though they have been provided by the phone company.

25.7. DN1, DN2 and MSN Numbers

(Directory Numbers and Multiple Subscriber Numbers)
A Directory Number is the ISDN number assigned to each B channel of an ISDN line. In North America users must get DNs from the phone company. These must be entered exactly as provided. For most other locations, the Telco will allocate MSN numbers to be used when more than one unit is on the line.

A Cautionary Note: In ToolBox you usually need to enter *DNs* with area code (10 digits), while some other ISDN devices may only require 7 digits numbers. If your switch type is an AT&T 5ESS custom point-to-point, you may need to leave both *DNs* blank - even though they have been provided by the phone company.

If you are using your codec in other places where MSN numbers (Multiple Subscriber Numbers) are used, enter the MSN number in the *DN1* and *DN2* fields within your codec or ToolBox.

25.8. Auto answer

When checked, the codec will automatically answer incoming calls. If unchecked, upon receiving an ISDN ring, the codec makes periodic beeps and the user must press the ANSWER or HANGUP buttons on the keypad of the codec.

A Cautionary Note: Tieline codecs react to ISDN incoming calls only if it is in ISDN operation mode.

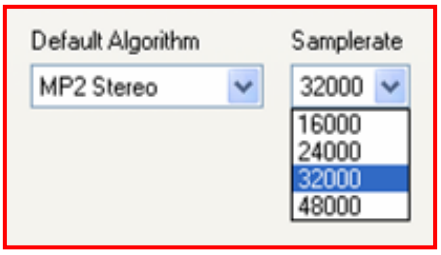
25.9. Phantom Power Detect

Some ISDN lines have phantom power on them. When checked, the codec will detect phantom power when using ISDN lines. This is useful in the detection of lost connections. If there is phantom power on the ISDN line and it suddenly disappears at the codec, the codec will display a Disconnect message. This will indicate a lost connection and assist in tracing the fault.

25.10. Manual Default Algorithm

The *Manual Default Algorithm* setting in the ***ISDN*** menu relates to the *Manual Default Profiles* within ToolBox. The setting you choose here will relate to any *Manual Default Profiles* that utilize ISDN connections. For example, if you select *G.722* from the drop-down menu, any manual default profile using an ISDN connection will be programmed to use the G.722 algorithm.

25.11. Samplerate



If you select one of the MPEG algorithms you can also select the sample rate you wish to use from the four available options. The *Samplerate* setting is for all manual default profiles and will also be used by custom profiles with connections that use the *Any* setting within the *Algorithm* drop-down menu in the ***Connection Setup*** tab.

The sample rate you select will be dependent on the connection or connections you are using. For example, if you have only one 64k ISDN B channel available

you will not be able to use a stereo algorithm with a sample rate of 48,000bps. If you do connect at this *Samplerate* the quality will not be very good. The recommended maximum you should set it at is 32,000bps for this type of connection. The default *Samplerate* for MP2 algorithms is 32,000 bps.

Section 26. IP/LAN and SIP Tabs

When you insert a 3G module (or a USB module with a 3G cell-phone attached) into your ***G3*** codec you will notice that a 3GIP connection will appear in the list of available connections on the connection LCD screen. Relevant 3G menu elements are also displayed in the codec configuration wizard. If you are not using ToolBox software to configure your connection, use the wizard via SOFTKEY 3.

Please see the "IP & 3GIP Streaming Reference Manual" for more information relating to programming and dialing 3GIP, IP and SIP IP connections and for the latest ToolBox software changes for LAN connections.

Section 27. X.21 Tab

When you click on the *X21* tab in the *Profile Editor* menu there are several settings available for programming the function of your codec in X.21 mode. (Please note: Even if you don't have ToolBox software connected to your codec, these functions are available by accessing your codec menu via SOFTKEY 4 Menu. Next select Configuration and select the X.21 connection you wish to configure. Now scroll through the menu and adjust the settings you wish to change. See the Configuration menu diagram for details of this menu structure.)

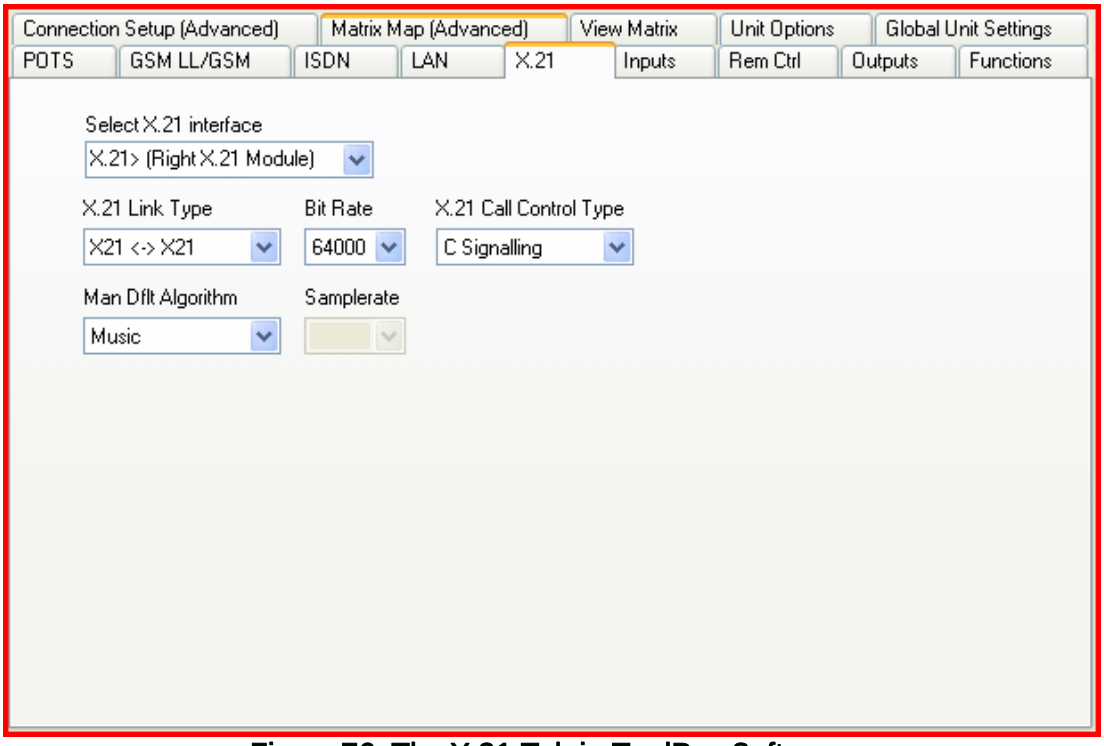


Figure 76: The X.21 Tab in ToolBox Software

27.1. Select X.21 Interface

An ***i*-Mix *G3*** has a single module slot and ***COMMANDER G3*** codecs have two module slots. It is therefore possible to run two X.21/V.35 digital leased line hardware modules from a ***COMMANDER G3*** codec. Using this menu you are able to select the X.21 module in a codec that you wish to configure. Simply click on the blue arrow to reveal the drop-down menu and make your selection.

27.2. X.21 Link Type

Tieline codec X.21/V.35 digital leased modules currently only support connections via an ISDN Terminal Adapter (TA). Ensure that you select *X21 < - > ISDN* in this menu.

27.3. Bit Rate

Currently, **Tieline** codecs support X.21 leased line mode with C/DTR call establishment over 64 to 384 kbps connections. Please select the bit rate you require from this menu and ensure that your ISDN TA/X.21 DCE Device supports this bit rate. The bit rate setting of your codec will need to match that of your ISDN TA/X.21 DCE Device.

27.4. Manual Default Algorithm

The *Manual Default Algorithm* setting in the ***X21*** menu relates to the *Manual Default Profiles* within ToolBox. The setting you choose here will relate to any *Manual Default Profiles* that utilize X21 connections. For example, if you select *G.722* from the drop-down menu, any manual default profile using an ISDN connection will be programmed to use the G.722 algorithm.

Important Algorithm Selection Information

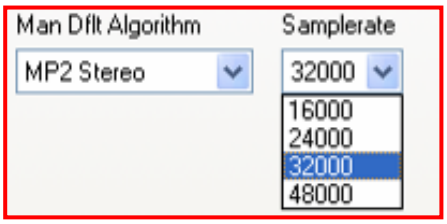
The manual default profile algorithm settings in the ***POTS, GSM LL/GSM/USB-3G***, ***X21*** and ***ISDN*** menus of the ToolBox ***Profile Editor***, select which algorithm will be the default for manual default profiles using these connections. For example, if *Music* is selected in the *Man Dflt Algorithm* setting in the ***POTS*** menu, all manual default profiles connecting via a POTS connection will use the *Music* algorithm.

This can be overridden in two ways. If you are using the ***Connection Setup (Advanced)*** menu to set your connections, you will note that a *Connection Type*'s *Algorithm* setting can be changed. If *Any* is the selected algorithm, the algorithm for each *Connection Type* will be the one selected for manual default profiles, i.e. the default ones selected in the ***POTS, GSM LL/GSM***, and ***ISDN*** menus. Any other algorithm selections here will override default algorithm settings in the ***POTS, GSM LL/GSM/USB-3G***, ***X21*** and ***ISDN*** menus.

Secondly, you can reset the algorithm using the codec itself by accessing the algorithm selections in the ***G3*** codec menus.

27.5. Sample Rate

If you select one of the MPEG algorithms you can also select the sample rate you wish to use from the four available options. The *Samplerate* setting is for all manual default profiles and will also be used by custom profiles with connections that use the *Any* setting within the *Algorithm* drop-down menu in the ***Connection Setup*** tab.

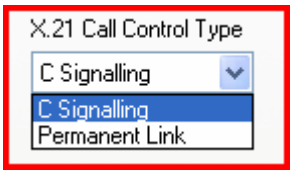


The sample rate you select will be dependent on the connection profile you are using. For example, if you are connecting using a stereo profile with a 64 kbps X.21 connection, you will not be able to use a *Samplerate* of 48,000bps. If you do connect at this *Samplerate* the quality will not be very good. The

recommended maximum you should set it at is 32,000bps for this type of connection. The default *Samplerate* for MP2 algorithms is 32,000 bps.

27.6. X.21 Call Control Type

This menu configures whether you are connecting using X.21 in either *C Signaling* or *Permanent Link* modes.



27.6.1. C Signaling Mode

In *C Signaling* mode, a 'C' (Control Signal) is used to control the connection. An 'I Signal' (incoming call signal) is used to indicate the presence of an incoming call.

When dialing from one codec to another, the dialing codec (DTE1) changes the C Signal from off to on to initiate an outbound call. At the codec (DTE2) receiving the call an 'I Signal' ('Indication' or incoming call signal) changes from off to on, indicating the presence of an incoming call. The codec receiving the call then initiates a C Signal state change, from off to on, which causes the I signal at the calling end to change its state from off to on. This completes the call connection process over X.21.

To clear a call, press HANG UP on either codec to change the C Signal from on to off. This will terminate the connection between the codecs.

27.6.2. Permanent Link Mode

In essence, X.21 connections in *Permanent Link* mode behave similarly to making a call in POTS leased-line mode.

Very Important Note:
Session data is automatically turned off in *Permanent Link* mode. In this mode ensure that both codecs are pre-programmed to operate using X.21 and with the same algorithm and bit rate settings etc.

When dialing from one codec to another, the dialing codec (DTE1) changes the C Signal from off to on to initiate an outbound call. A call is initiated when a user presses the ENTER/DIAL button on the dialing codec. Unlike in *C Signalling* mode, the I (CTS) signal is ignored.

An operator must manually press the ANSWER button on the answering codec (DTE2) to accept the call. Please note that there is no auto-answering in this mode.

To clear a call, press HANG UP on both codecs. This will terminate the connection between the codecs.

If you require more information on session data and how it works, please see the section titled Data Transfer & Using 3rd Party Devices.

Section 28. Inputs Tab

The *Inputs* menu is used to configure input settings as well as *IGC* (intelligent gain control) settings. The input settings that can be adjusted on the *Inputs* page of the *Profile Editor* are displayed in the following image.

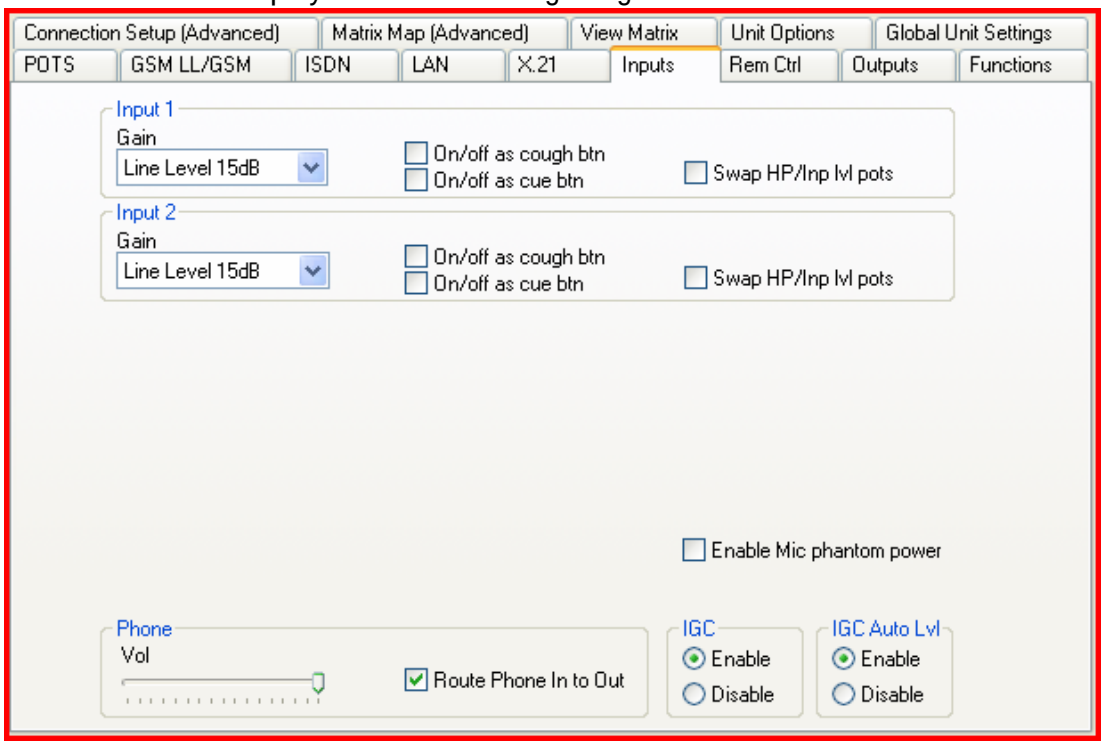
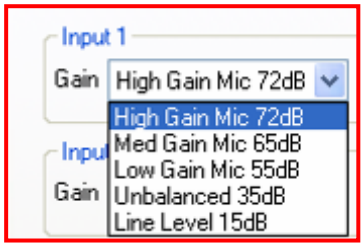


Figure 77: The Inputs Menu in ToolBox

28.1. Input Gains

Tieline codecs will handle inputs levels from -72 dB to +21 dBu. Depending on the codec you are using, the number of inputs available will vary. There are 5 gain settings on each input as shown in the pull down menu to the right.



For information in relation to setting levels on all **Tieline** broadcast codecs, please see the section in this manual titled PPM Displays.

28.2. Input Types

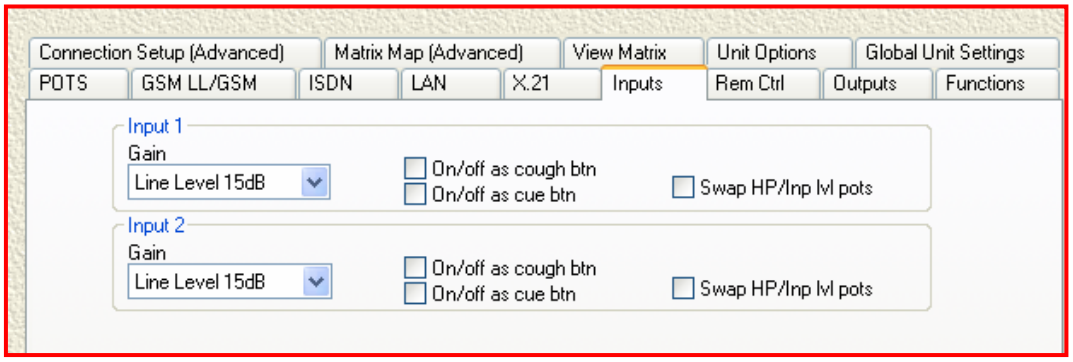
Following are some suggested input settings for various sources that can be plugged into **Tieline** codecs.

Gain setting	Max Gain	Example of equipment
High Gain:	Gain + 72dB.	ECM77, Lip Ribbon
Med Gain:	Gain + 65dB.	Sennheiser HMD25/1
Low Gain:	Gain + 55dB	SM57/58 etc.
Unbalanced:	Gain + 35dB.	Consumer items fitted with RCA connectors.
Line Level:	Gain + 15dB.	Professional equipment.

Table 12: Input Gain Setting Options

28.3. Additional Input Settings

Each balanced codec input has several other signal routing and processing functions that can be set individually on each input. To select a function simply click on the box for that function and a green tick will appear in the box confirming your selection. Simply click again if you wish to deselect the function.



A description of the functions available on the inputs is detailed in the following table:

1.	*On/Off as cough button*	Switches the ON/OFF button on the codec into a cough button.
2.	*On/Off as cue button*	Switches the CUE-COMMS button on the codec into a cough button.
3.	*Swap HP/Inp Lvl pots*	This swaps the functionality of these two features so that the INPUT level digipot will control the headphone levels and vice versa.

28.4. Phantom Power

Tieline codecs also have the ability to supply 15 volts of phantom power to the microphone inputs. Simply check the *Enable Mic phantom power* box to switch this function on.

WARNING: Please check the powering specifications of the microphones you are using with your codec to ensure they will operate effectively and to avoid damaging them.

28.5. Intelligent Gain Control

All codecs also have the **Tieline** Intelligent Gain Control (*IGC*) system, which makes controlling levels a simple non-technical exercise. *IGC* was developed to enable announcers to forget about panel controls and to focus on program content.

28.5.1. How it works:

The microphone input level is controlled by the input level digipot on the codec. All analogue audio inputs are digitized by a high quality stereo 24-bit analogue-to-digital (A to D) converter. This serial data stream is then fed to an extremely fast 32-bit Digital Signal Processor (DSP) that encodes the audio data.

The DSP measures the amplitude of incoming signals and if the levels exceed acceptable limits, the DSP will automatically control the digipot to decrease the level of the incoming signal back to an acceptable level. This *IGC* is enabled almost instantaneously.

28.5.2. IGC Enable Mode

IGC has two operational modes. In *IGC Enable* mode, audio levels will be automatically adjusted downwards until they are acceptable. The input level will remain at this point until the input digipot is adjusted again by the user. In addition, the channel on light will flash until the input pot is adjusted. The flashing button notifies the codec operator that *IGC* has been initiated.

28.5.3. IGC Auto Level

In *IGC Auto Lvl* mode (the default setting for manual default profiles), audio levels are automatically adjusted downwards until they are acceptable. However, once the codec detects that the incoming audio levels have reduced sufficiently, the input level will be returned to the previous setting before *IGC Auto Lvl* control was enabled. The codec takes just 250 milliseconds to detect audio levels have returned to normal (after *IGC Auto Lvl* has been initiated) and will then return the digipot levels to the previous setting in half a second. This response is linear. The channel on light will flash while *IGC Auto Lvl* is activated.

28.5.4. IGC Summary

In summary, when the broadcast action really starts to heat up, the inbuilt DSP limiter automatically takes care of the instantaneous peaks that occur in demanding broadcast situations. **Tieline** recommends enabling the use of *IGC* when using your codec. *IGC* on the inputs, if enabled, is activated at +14 dBu. As mentioned, *IGC Auto Lvl* is enabled as a default in the manual default profiles. (Related topic: PPM Displays)

28.6. Routing Voice Calls to a Codec's Analog Outputs

It is possible to route POTS and GSM voice calls to a codec's analog outputs by pressing SOFTKEY 1 Audio and selecting **[Route PhIn to CHOut] > [Enable]**. The default setting is Disable.

This can also be configured using ToolBox software via the ***Inputs*** tab, as displayed in the following image.

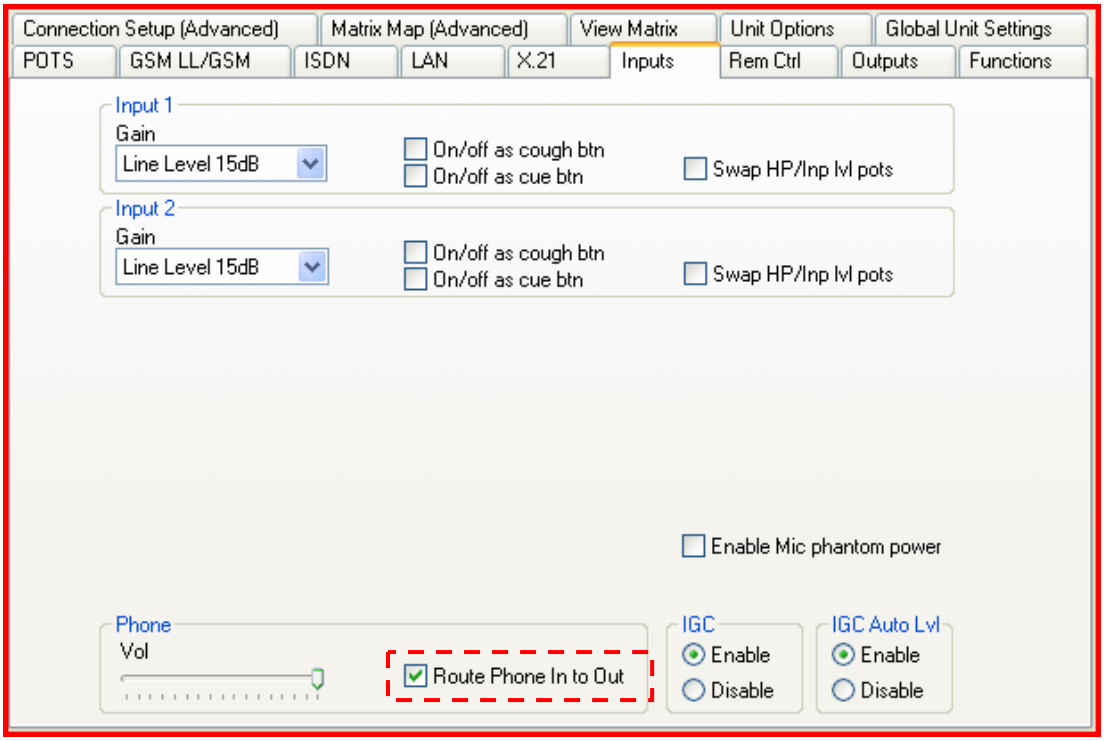


Figure 78: Route Phone Input to Outputs Check-Box

Once this has been configured, analog audio is routed to the analog outputs instead of the *Encoder/s* – allowing this audio to be used as an analog program, foldback or IFB feed.

28.7. Phone Input: Operation of the I-Mix G3 Phone Coupler

28.7.1. Simultaneous Off-Air Communications

The ***i*-Mix *G3*** has a telephone coupler system which can be used either for bi-directional communications, or for integrating callers into program signals. In other words, a caller can dial into or out of the codec, and by using the CUE/HOLD function, set up bi-directional 'off-air' communications. This system operates by 'ducking' the incoming phone call when any commentator is speaking. The procedure for establishing a phone call for bi-directional communications, and for mixing calls into program feeds, is discussed in the following section.

28.7.2. Making Out-bound Calls Using the Phone Coupler

The procedure to make a call using the PHONE COUPLER is as follows:

1. Ensure that you have a phone line plugged into the Line 2 COUPLER input on the rear panel of the codec.
2. Use the MS (Menu Selector) to go to the PHONE Enter# connection on the codec LCD.
3. Next press the TAKE/RELEASE button on the codec and you should hear the phone line dial tone in the headphones plugged into HEADPHONE 1 on the rear panel of the codec.
4. Next dial the phone number using the keypad on the codec and press the ENTER/DIAL button to make the call.
5. The TAKE/RELEASE button will flash red when there is an incoming call. Simply press the TAKE/RELEASE button and the CUE/HOLD button will illuminate yellow and allow you to talk 'off-air'.

If the announcer in INPUT 1 wants to go back on-air after chatting off-air via the PHONE COUPLER, simply press the illuminated PHONE COUPLER CUE/HOLD button to turn it off and make sure the channel ON/OFF button is switched on for INPUT 1. The announcer is now back on-air.

28.7.3. Taking Live Callers in the Field

By using the PHONE COUPLER's ON/OFF function you can also mix a call into the main program feed. The TAKE/RELEASE button will flash red if there is an incoming call. Simply press the TAKE/RELEASE button and the CUE/HOLD button will illuminate yellow and allow you to talk 'off-air'. If you wish to put the PHONE COUPLER on-air once you have established a call, simply press the ON/OFF button in the PHONE COUPLER section and it will go live to *Encoder 1 Out* (and is also sent to CH 1 Out which can be useful for PA feeds etc.)

Please Note:

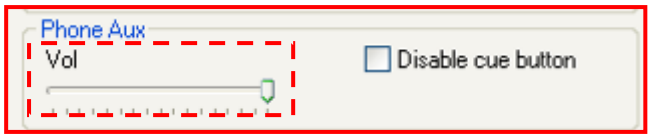
1. The PHONE COUPLER for all the manual default profiles is set up to work as a communications channel for the announcer or commentator using INPUT 1 and the HEADPHONE 1 output on the codec.
2. The announcer plugged into INPUT 1 will be able to speak using the PHONE COUPLER even if the channel ON/OFF button is switched off.
3. Incoming and outgoing PHONE COUPLER audio will be monitored in the right side of the headphones in HEADPHONE 1.
4. The auxiliary input connections on the rear panel, AUX IN and AUX OUT, are also controlled by the PHONE COUPLER section on the front panel of the codec. The buttons on the front panel that can control the auxiliary input source are the CUE/HOLD button and the ON/OFF button. The TAKE/RELEASE button only controls the PHONE connection on the codec LCD.

The Phone Coupler Controls on the *i*-Mix *G3*.

To the left the PHONE COUPLER is in 'off-air cue' mode and to the right it is in 'on-air' mode.

28.7.4. Phone Coupler Functionality within ToolBox

The functions available on the codec front panel are also available within ToolBox software.



The incoming volume *(Vol)* setting for the *Phone Aux* input is presented in ToolBox as displayed. It controls the level of the incoming signal. To adjust the level simply click on the volume indicator (as highlighted) and drag it until the level is set as desired. This setting is the 'powerup' level setting when you initially power the codec up.

The operational functions for the PHONE COUPLER are also adjustable within the ***Control*** menu within ToolBox. The *Take*, *Cue/Hold* and *Off/On*

buttons operate as they do on a codec front panel. The incoming audio level can also be adjusted using the virtual fader. Simply click on the volume indicator and drag it to adjust the level setting. You will note that the incoming audio level will 'duck' when *Phone Aux* audio is sent from the codec.

You can also control audio levels and the other PHONE COUPLER functions remotely by using **Tieline** ToolBox software. This functionality is programmed within the *Rem Ctrl* tab under *Phone.*

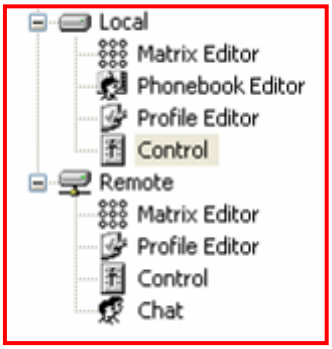
28.8. Auxiliary Input: Output Level Adjustment

The unbalanced AUX input on the codec is a standard RCA line level input for interfacing with consumer electronic devices. If you are connecting the codec AUX output to a device that is expecting a microphone level source, i.e. a cell-phone, this high-level output can be adjusted using the codec menus. To adjust the output level do the following:

- Press SOFTKEY 4 to access the Menu submenu;
- Scroll using the MENU SELECTOR (MS) to Configuration and press to select this option;
- Next scroll to Aux Output Level and press the MS to select it;
- The default setting is 255 in the menu. Use the CLEAR button to delete the numbers and input a number lower than 255 to lower the output level; and
- Press SOFTKEY 4 which will have OK displayed above it to enable the change.

Related topic: Phone/Aux Input Level Control

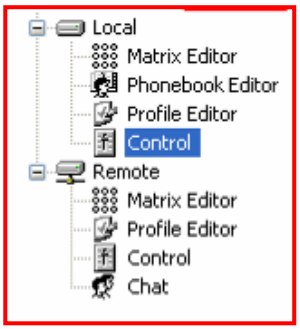
28.9. Local and Remote Codec Level Adjustment with ToolBox Software



To adjust the input levels of the local codec using ToolBox, first connect your PC to the codec and click on *Control* under *Local* on the Root Menu of ToolBox. This will display the *Local Control* screen within ToolBox and allow adjustment of input levels. To adjust the input levels of the remote codec, click on *Control* under *Remote* on the Root Menu of ToolBox and make the necessary input adjustments.

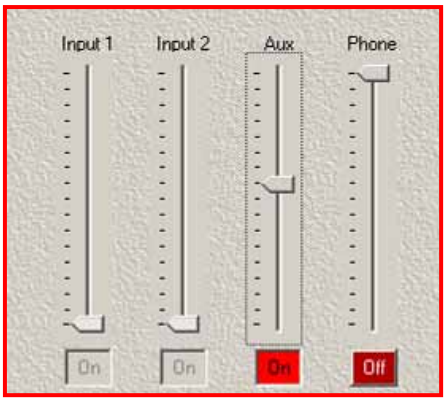
28.10. Control Menu: Adjusting Input Levels Using ToolBox

All input levels on **Tieline** codecs are processed with digipots. Normal potentiometers are controlled by rotating a shaft. Digipots are potentiometers with digital control and behave similarly to VCAs (Voltage Controlled Amplifiers).



The *Local Control* menu for adjusting input gain settings can be found by clicking on *Control* on the Root Menu within the ToolBox software.

Once you have a connection confirmed between ToolBox on your PC and a local **Tieline** codec, you can control the input levels on the local codec if the appropriate boxes within ToolBox have been checked. Refer to the section on Local and Remote Codec Control for more details, or if you wish to control codec functions by remote control over the codec connection.



To switch a channel on or off using ToolBox, just click on the *On/Off* button with a mouse. To adjust the audio levels of the individual channels just click and drag the virtual fader slider on each input.

Figure 79: Virtual Input Faders in a Commander G3 Codec

Section 29. Remote Control Tab

Remote control functions for codec inputs can be programmed from within this menu in the *Profile Editor*. You can also set remote control functionality with your codec quite simply and this is explained within this manual in the section titled Remote Control of Input Controls. The ToolBox menu is displayed below.

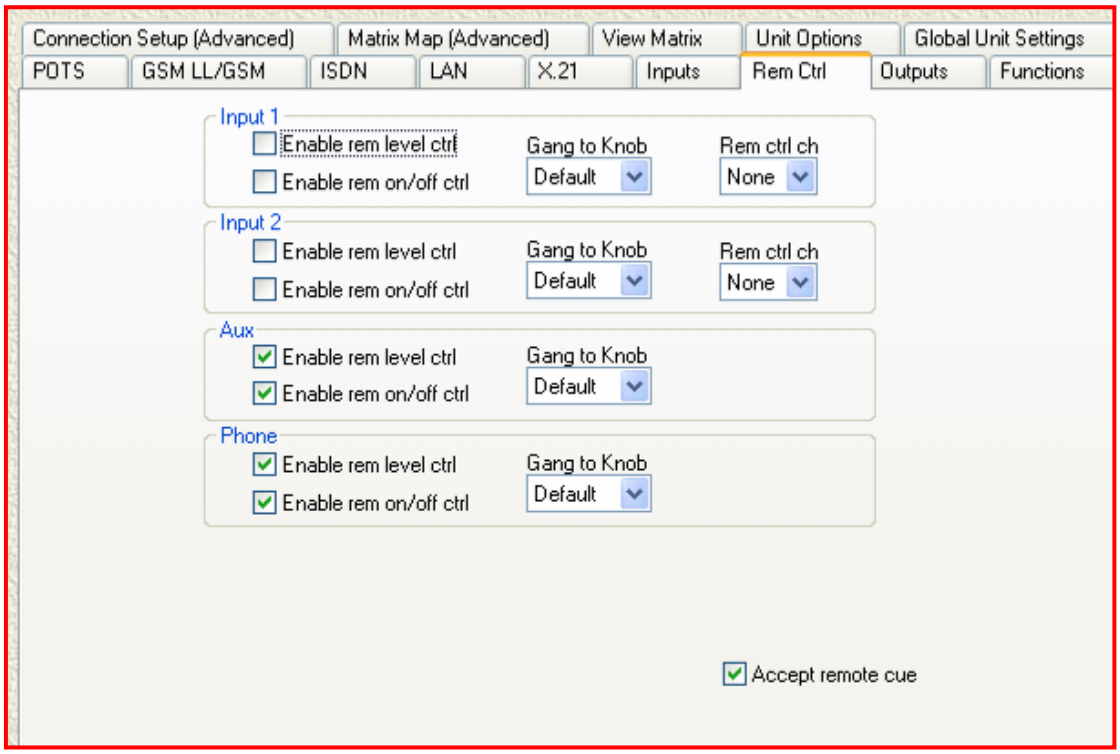


Figure 80: ToolBox Remote Control Menu

Following is an outline of the master and slave principles and how they operate within **Tieline** codecs.

29.1. Local and Remote Codec Operation Explained

Another simple way to program remote control functions is with ToolBox software. Before discussing in detail how you can remotely control a **Tieline** codec, it is useful to ensure you are clear about how two codecs operate in relation to local and remote functions.

In this example we are using a ***COMMANDER G3*** field unit and rack unit codec, but the principles are the same for all **Tieline** codecs, including the ***i*-Mix *G3***. An ***i*-Mix *G3*** codec can control the functions of a ***COMMANDER G3*** codec, or vice versa if required.

The local codec is usually the codec at the studio or broadcast center. This codec is usually the 'master' codec when using remote control functions. In other words it is able to control the input gain settings and channel on/off functions on the remote codec. The local codec may have a PC connected to it using **Tieline** ToolBox software.

The remote codec, its companion partner, is usually at the broadcast venue or remote site. This is usually the 'slave' codec, able to be controlled by the local codec at the studio or broadcast center. A typical connection setup between two codecs is displayed in the following image.

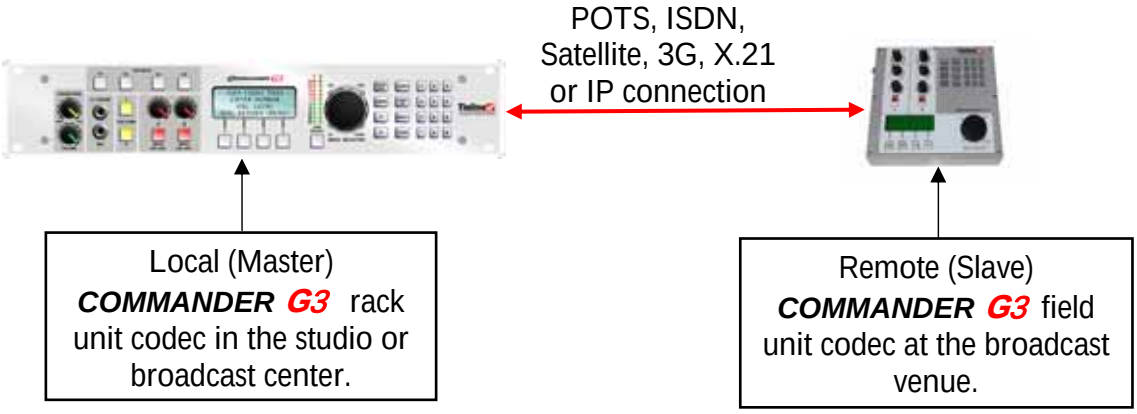


Figure 81: Typical Remote Control Setup without Using ToolBox

IMPORTANT WARNING: It is very important to note that remote control of a codec is not possible unless you are using either the *Music, Music Plus* or *Voice G3* algorithms.

Next we will explain how to configure a codec for remote control functionality. If you have a PC attached to your codec with ToolBox software installed on it, you can configure remote control settings for all codec inputs using this software. An operator at the local codec will then be able to control the levels on the remote codec using **Tieline** ToolBox software, or with the input level controls on the local codec.

Remote Control Recommendation:
Before commencing Remote Control configuration we recommend that you ensure that any previous settings have been deleted from the menus. To do this press SOFTKEY 4 Menu on each codec and then scroll to and select **[Remote Controls] > [Reset Remote Control] > [Yes]**.

29.2. Remote Control of Input Gains

All **Tieline** codecs can be configured so that the input levels, and the phone coupler input level (if you are using an ***i*-Mix *G3***), as well as channel on/off functions, can be controlled by a codec that has been designated the status of 'master' codec. This enables a codec to be used by non-technical people or to be unmanned by technical staff in most applications.

Tieline codecs are designed to operate in conjunction with ToolBox software. This proprietary **Tieline** software program easily facilitates the programming of

remote control functions. With the use of virtual faders it also provides a convenient way to control both the local (master) and remote (slave) input gains.

For example, the codec located in the studio or broadcast center could be set up as the local unit and then a **Tieline** codec at the remote location would be set up as the remote unit. A PC attached to the local codec with ToolBox software could control input gains and channel on/off functions for all channels of both the local and remote codecs.

29.3. Profile Requirements for Remote Control

Two separate profiles are required to enable remote control operation between two codecs. The local codec's profile will need to be configured with master codec remote control input settings and the remote codec will need to be configured with slave remote control settings. These profiles will need to be configured and then loaded onto each codec before remote control is possible.

Remote control can also be configured in *Current Runtime* (settings configured on a codec subsequent to a profile being loaded onto a codec). This is useful if you don't have a PC with ToolBox software and you wish to configure a codec for remote control use. Remember, *Current Runtime* changes are not permanent profile changes. If you wish to save configuration settings it is best to use ToolBox software to create and save your profile settings.

Please see the section titled Configuring Remote Control in the Quick Start section of this manual for more information on using the codec to configure remote control settings without using ToolBox software.

29.4. Configuring the Remote (Slave) Codec to be Controlled

Either a local codec itself, or a PC with ToolBox software connected to it, can be configured to adjust the input levels and channel on/off for a remote codec. The following settings on the remote (slave) codec profile need to be enabled to successfully allow it to be controlled by another codec.

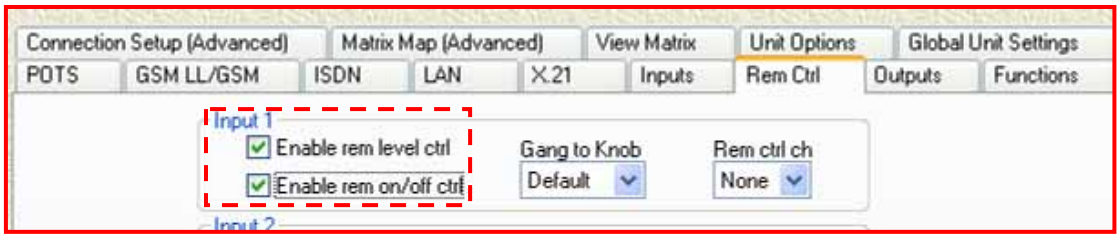
First, enable the remote (slave) codec to receive messages and be controlled by the local (master) codec or ToolBox. This is configured in the *Rem Ctrl* menu in the *Profile Editor* within **Tieline** ToolBox software. There are three options for remote level control within this menu and they are available on all inputs of the codec:

1. *Enable remote level control*;
2. *Enable remote on/off control*; and
3. *Remote control channel*.

The first two items relate to the codec being controlled – the 'slave' codec. The third item relates to the local or 'master' codec and is used to select which channel on the local (master) codec will control which channel on the remote (slave) codec.

29.4.1. Enabling a Codec to be Controlled Remotely (Slaved)

To enable a codec to be 'slaved', check *Enable remote level control* and *Enable remote on/off control,* as displayed in the following image. If you only want to have control over one of these two functions, just check the function you require.



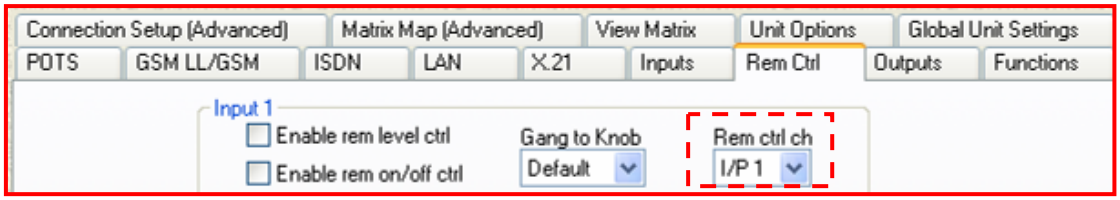
A PC with ToolBox software installed, which is connected to the local codec, can now assume control of the audio levels of the remote codec using the *Control* menu within ToolBox.

If you want to use the input level pots on the local codec to control the input levels on the remote codec, *Remote control channel* in ToolBox configures which input level pots on the local codec will control the input levels on the remote codec. It is discussed in detail in the next section.

29.5. Configuring the Local (Master) Codec to Control the Remote Codec

A master codec with ToolBox connected can adjust remote codec input settings via the *Control* menu. Alternatively, you can configure which input gain knobs on the local (master) codec will control which input gain settings on the remote codec. To do this go into the *Rem Ctrl* menu within the *Profile Editor* and adjust the *Remote control channel* settings for each channel.

In the following example, the digipot on INPUT 1 of the local codec has been set to control the levels of INPUT 1 on the remote codec.



Configure each input setting as per your remote control requirements. If you wish to control two channels on the remote codec, select the two channels on the local codec that you wish to use as master gain knobs.

Remember, for this to work you need to make sure that the remote codec has been 'enabled' by the checking of the *Enable rem level ctrl* and *Enable rem on/off ctrl* functions on the remote codec.

Please note: It is not possible to gang inputs of a master codec together and use this function to control inputs remotely on a slave codec. However, if you have a stereo source or two or more inputs which you would like to gang together and control on the slave codec, this can be done at the remote (slave) codec. Simply gang the inputs you would like controlled on the slave codec to a single input that is being remotely controlled by an input on the master codec. See the section titled Configuring Remote Control for more details.

29.6. Adjusting Local (Master Codec) and Remote (Slave Codec) Input Levels When in Remote Control Channel Mode

When a connection is made, the remote (slave) codec obtains gain settings from the local (master) unit and it then stores the settings in memory.

If remote control functionality is enabled on both the master and slave codecs, and you have sources plugged into either of these codecs, you may wish to adjust the levels of these inputs locally - either before or after the codecs are connected.

To adjust the input level settings locally at either codec, simply press the CUE-COMMS button for the channel you wish to adjust and while holding it down adjust the input pot on the channel. If the CUE-COMMS button is being used for talkback then this will only work in offline mode. Please note that there is no CUE-COMMS button on ***COMMANDER G3*** field and TLR300B rack unit codecs - so level adjustment of inputs by this method is not possible.

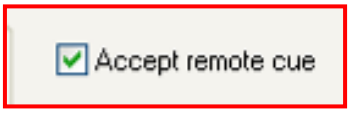
29.6.1. TLR300B Local and Remote Input Level Control

With the TLR300B codec, adjust levels locally and remotely by programming the local or remote inputs onto the codec's PPM meters. This is done by holding down the LEVEL SELECT button for a PPM meter and then pressing the PPM SELECT button to sequentially scroll through the menu options.

To adjust the local or remote input levels, simply continue to press the LEVEL SELECT button for the input and use the MENU SELECTOR to adjust the audio gain up and down, or use SOFTKEYS 1 & 2 to turn the channel on or off. For more information please see the section in this reference manual titled 'Menu Selector Input Controls'.

Please note: A remote codec's channels, that are to be controlled by a rack unit codec, will need to be programmed for remote control operation.

29.7. Accept Remote Cue



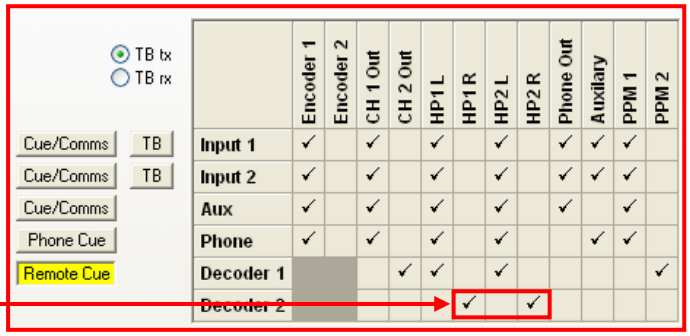
Remote cue simply enables the *Cue/Comms* send within the remote codec and the local codec. As a default it is checked. When *Accept remote cue* is checked on a codec, it enables the *Remote Cue*

Matrix Map functions set on **Tieline** codecs. Once this is enabled, to utilize the cue facility simply press the CUE/COMMS button on the codec. Remember that for ***COMMANDER G3*** field unit codecs the CUE/COMMS button when activated uses SOFTKEYS 2 & 3 on the codec (or the virtual CUE-COMMS button on the rack unit GUI codec).

To access the *Cue* button facility on the ***COMMANDER G3*** field unit simply press the MENU SELECTOR down for a few seconds and the two Cue button functions will become visible on the codec LCD. To deactivate the buttons simply hold the MS down until they disappear. (Related Topics: Cue and Talkback)

Figure 82: Active Cue Softkey Functions on the Commander G3 Field Unit

To view the ***Matrix Editor*** settings for *Remote Cue* go to the ***View Matrix*** menu and click on the *Remote Cue* button which is the lowest on the menu page.

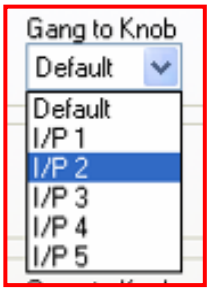


Remote Cue active and as displayed in the ***View Matrix*** menu window.

Please Note: The preceding example displays a ***COMMANDER G3*** field unit ToolBox menu but the functionality is similar across the entire range of **Tieline** codecs.

Please note: The only manual default profile that has a matrix attributed to the Remote Cue function in ***Matrix Map*** is *Manual Default Mono/IFB*. These settings are displayed in the preceding image. Use this profile for sending program and IFB.

29.8. Gang to Knob



This feature allows the user to 'gang' inputs together. Any of the input gain digipots on a codec can be ganged to each other or to the AUX or PHONE inputs. This function allows the input with the gang function selected to replicate any gain adjustments via the selected digipot. This is particularly useful for stereo audio sources or for sources with similar gain-setting characteristics. If *Default* is selected no ganging is activated.

29.8.1. Ganging in Remote Control

It is not possible to gang inputs of a master codec together and use this function to control inputs remotely on a slave codec. However, if you have a stereo source or two or more inputs which you would like to gang together and control on the slave codec, this can be done at the remote (slave) codec. Simply gang the inputs you would like controlled on the slave codec to a single input that is being remotely controlled by an input knob on the master codec.

Section 30. Outputs Tab

This tab within the *Profile Editor* allows adjustment of the output functions of **Tieline** codecs.

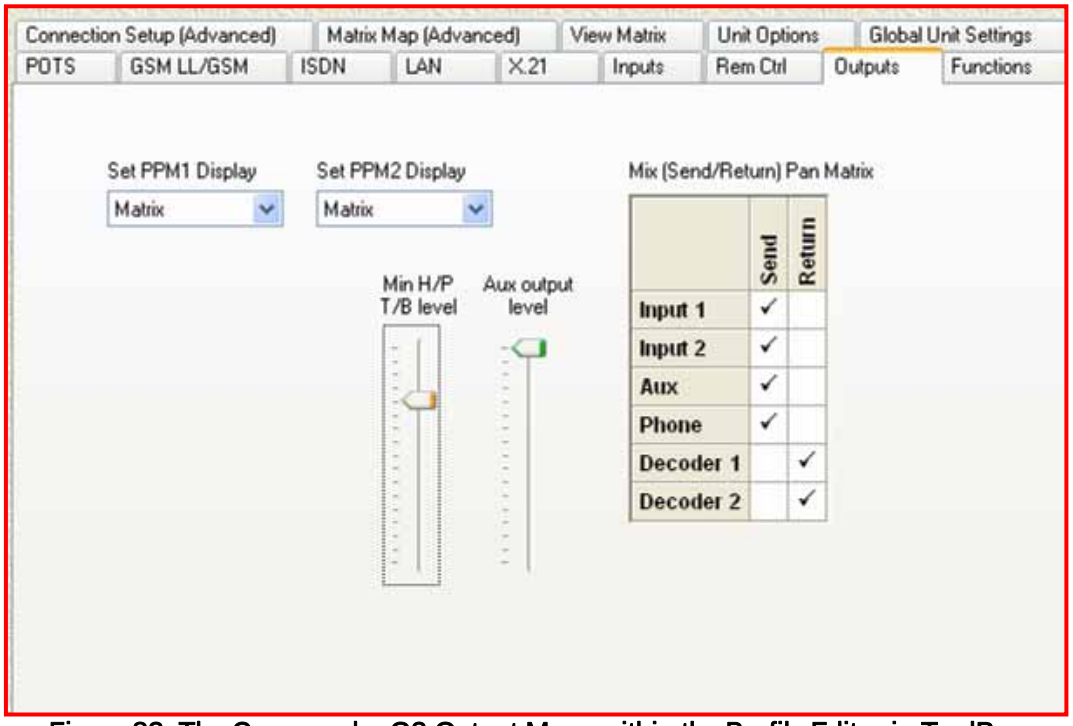
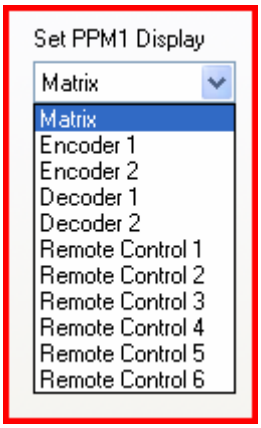


Figure 83: The Commander G3 Output Menu within the Profile Editor in ToolBox

30.1. PPM Displays



The PPM displays on the codec can be set to show the input level of audio signals entering the codec's encoders or the output level of audio signals leaving the decoders. Remote controlled channels can also be displayed. The PPMs can also be set to display mic input levels and/or other sources which have been selected via matrices and are displayed under *PPM 1* or *PPM 2* in the ***View Matrix*** section of the ***Profile Editor*** menu.

Each PPM display is set independently by clicking on the pull-down menu (see left) for *Set PPM1 Display* and *Set PPM2 Display*.

30.1.1. Setting Audio Levels: i-Mix and Commander Rack Unit Codecs

If you are using an ***i*-Mix *G3*** or a ***COMMANDER G3*** rack unit, levels should be set so that audio peaks average at the first set of orange PPM LEDs. In addition, the PPM settings can be adjusted by using the Audio menu on the codec itself. (This can be accessed by pressing SOFTKEY 1 and using the MENU SELECTOR to select either PPM1 or PPM2.)

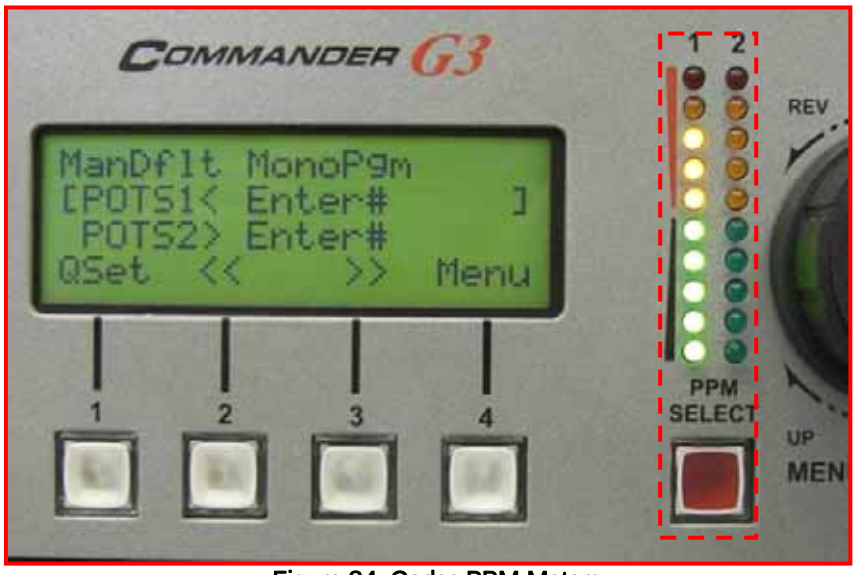


Figure 84: Codec PPM Meters

30.1.2. Setting Commander G3 Field Unit Audio Levels

On the field unit ***COMMANDER G3***, codec audio levels are displayed on the LCD screen. Set your levels so that audio peaks average at the first solid bar. This represents a program level of +4 dBu leaving the codec. Audio peaks can safely reach +21 dBu without clipping. We recommend that you adjust levels so that peaks don't exceed +16 dBu on all **Tieline** codecs (indicated by the red LEDs on the ***COMMANDER G3*** rack and the ***i*-Mix *G3*** codecs, and the fourth solid LED on the field unit codec LCD).

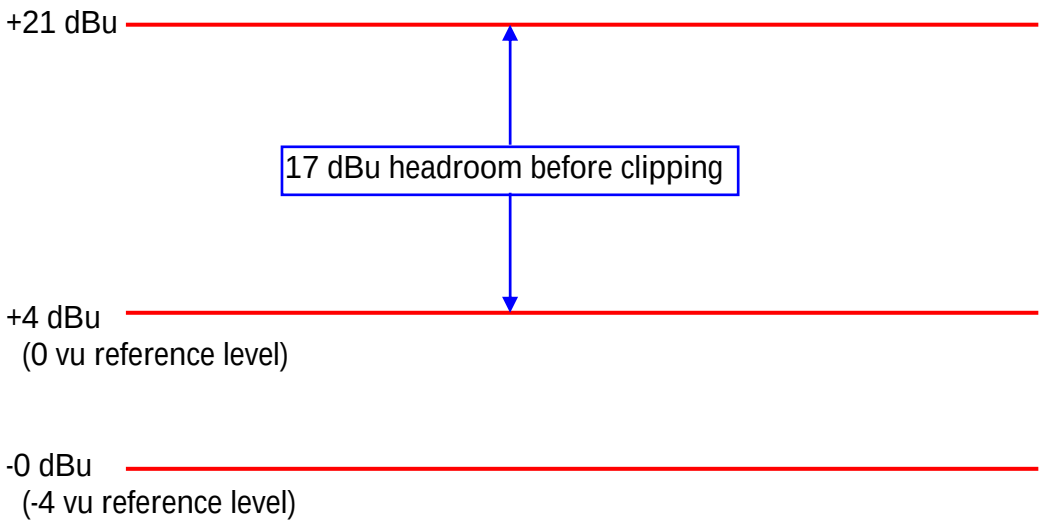


Figure 85: Diagram Showing Codec Headroom

The preceding diagram shows the amount of operating headroom the codec has. If you put 0 vu into the codec, when the codec input digipot is set at the one o'clock position, the nominal operating reference level is 0 vu. An

additional 15 dB of gain is available via the digipot from this reference position.

IGC (Intelligent Gain Control) on the inputs, if enabled, is activated at +14 dBu. *IGC* is enabled as a default for the manual default profiles. (Related topic: Intelligent Gain Control, Input Gains)

Figure 86: The LCD Screen of the COMMANDER G3 Field Unit

In the example above, *PPM1* is the only channel of the codec that is metering an audio signal at this time.

30.2. Automatic Gain Control

As a default setting, automatic gain control (AGC) is enabled on the outputs of **Tieline** codecs. AGC is independent of Intelligent Gain Control (*IGC*).

AGC is triggered if audio signals approach +20dBu (+16 vu). This is to avoid distortion of audio, which occurs if signals go beyond the analog stage limit of +21dBu.

ACG has a very fast reaction time and will decrease high audio levels by approximately 1dB every 2 milliseconds, up to a maximum of 44 dB of attenuation. AGC tries to return levels to normal once the output dips below +15dBu (+11 vu). This will increase the levels by 1dB every 2 milliseconds.

Another way of expressing AGC in as follows:

Output = Mix x AGC.

- If output > +16 vu (+20dBu), AGC decreases levels by 1dB to a maximum of – 44dB.
- If output < +11 vu (+15dBu), ACG increases levels by 1dB to a maximum 0dB.

30.3. Relationship of IGC to AGC

Tieline recommends the use of both *IGC* and AGC in most situations. *IGC* is a transparent level adjustment mechanism across each codec input that keeps levels in check, without coloring audio content with unwanted additional processing. It also allows each input to be adjusted individually, retaining the levels of other inputs when one is adjusted. This is different to AGC which is across the outputs, and for which any gain reduction affects the mix of all audio inputs simultaneously.

These thresholds are complementary to the settings for *IGC,* which is across codec inputs. If *IGC* is enabled in **Tieline** codecs it is enabled at +14dBu. Enabling *IGC* avoids hard clipping of signals, which occurs in the digital input stages if signals are higher than +14dBu. With *IGC,* an input will remain at the lower level until it is readjusted by an operator.

IGC Auto Level reduces high input levels to within acceptable limits and returns levels automatically to the original settings when levels have returned to normal.

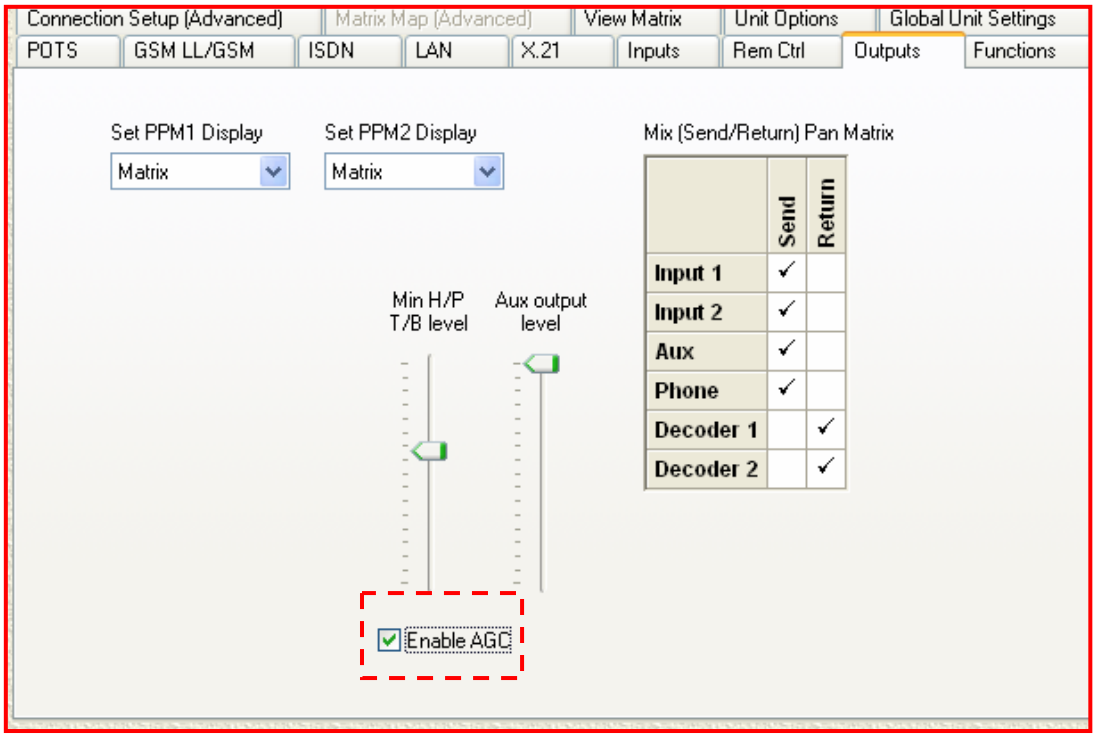


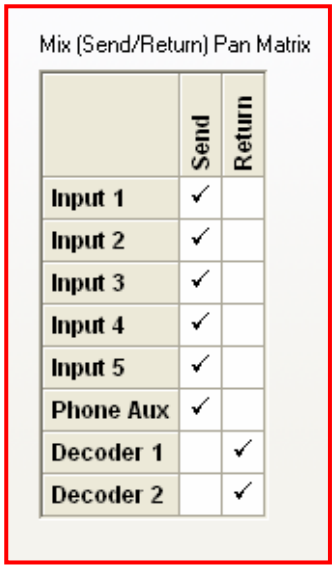
Figure 87: AGC Enable in the Outputs Tab

30.4. PA Controls

This feature is only available on ***i*-Mix *G3*** codecs. Before explaining the *PA Controls* menu and its implications on codec functionality, it is necessary to understand how the *Mix (Send/Return) Pan Matrix* operates.

30.4.1. Mix (Send/Return) Pan Matrix

The BALANCE knobs on **Tieline** codecs control the balance between the SEND & RETURN audio sent to the codec headphone outputs. The *Mix (Send/Return) Pan Matrix* provides the flexibility of selecting which sources are allocated to SEND and RETURN.

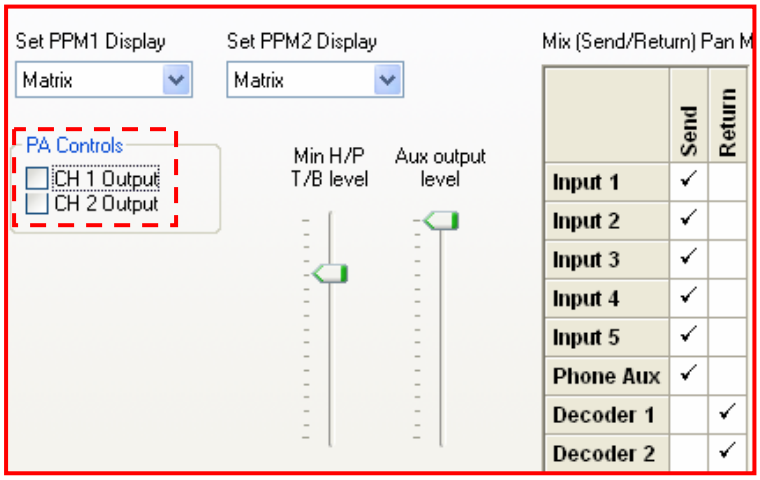


As shown in the right-hand image, sources are selected by 'checking' the matrix boxes. The balance between how much level you receive from either the SEND or RETURN matrix sources is then controlled by the BALANCE knob. This control does not affect the level of the transmitted or received program audio. It only controls the monitoring of these sources in the headphones.

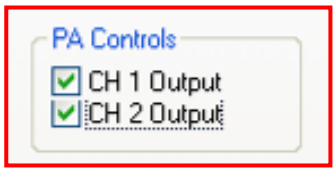
The example shown is the default *Mix (Send/Return) Pan Matrix* setting in an ***i*-Mix *G3*** codec. It many situations it is likely that the codec decoders will be providing return audio from a studio codec, so this example would allow the commentators to adjust the balance between the program audio at the venue codec (provided by *Inputs 1 - 5* and the *Phone Aux* input) and an incoming return audio foldback or IFB feed provided by *Decoders 1 & 2.*

30.4.2. Using the Mix (Send/Return) Pan Matrix for PA Output Settings

As a default, if the *Ch1* and *Ch2 PA Controls* boxes are unchecked, audio is sent to analog Program Output Ch1 and Ch 2 on the rear of the codec, as per the matrix settings of the profile you have loaded into the codec. Audio is sent at +4 from the codec.



If the *Ch1* and *Ch2 PA Controls* boxes are checked, the sources sent to these two outputs can be controlled by the PA OUTPUT controls on the front panel of the ***i*-Mix *G3*.**



The PA ON, VOLUME and BALANCE elements will now control what is sent via analog Output Channels 1 and 2. Either one or both of the analog Output Channels 1 and 2 can be selected within this ToolBox menu and controlled in this way.

Assuming that both the *Ch1* and *Ch2 PA Control* boxes are checked, the PA ON button will turn analog Output Channels 1 and 2 on and off. The VOLUME knob will adjust the output level of both analog outputs and the BALANCE digipot (in the PA OUTPUT section of the codec) will control the balance of *Mix (Send/Return) Pan Matrix* settings sent to these outputs. Previously this was only affecting the headphone balance of *Mix (Send/Return) Pan Matrix* settings. Now it is also actively able to impact upon the balance of what is sent to analog Output Channels 1 and 2.

Remember, the PA ON button will need to be switched on for audio to be sent to the analog outputs if operating in this mode. For an example of a matrix that would take advantage of this sort of functionality, please see section within the ***i*-Mix *G3*** main manual titled Manual Default Program TLM400.

Important Note:
If you use the *Mix (Send/Return) Pan Matrix* when the *Ch1* and *Ch2 PA Control* boxes are checked, it is important to note that any settings you make will not only affect the PA OUTPUT BALANCE digipot but also the BALANCE digipot on each of the first four inputs of the codec. In other words it will also affect the sources monitored via the headphone outputs.

30.5. Talk Back Level Adjustment

This setting allows the user to set the minimum headphone talkback level. This means that even if a commentator turns down the headphone level pot completely on the codec, talkback will still be heard in the headphones. This is a great safety feature to ensure continuity of communications. (Related Topic: Talkback)

Section 31. Functions Tab

The *Functions* menu within the *Profile Editor* enables you to set user-programmable functions with your codec, which can trigger events automatically via FUNCTION KEYS, SOFTKEYS, HOTKEYS and Internal or External Inputs. These user-programmable functions will be referred to as User Functions.

The SOFTKEYS are the four buttons below the menu screen on **Tieline** codecs. The HOTKEYS are on the front panel of the ***i*-Mix *G3***, or the ***COMMANDER G3*** rack unit codec front panel or GUI. There are no HOTKEYS on the ***COMMANDER G3*** field unit codec.

Please Note: Some hardware elements such as the HOTKEYS are not displayed in the *Functions* menu in ToolBox software for the ***COMMANDER G3*** field unit codec (or other codecs that do not have these hardware elements). In addition, the menu may display more inputs etc, than are available on the codec you are using. Ensure you only program user functions that match the hardware profile of the codec you are using. For example, only program inputs 1 or 2 on ***COMMANDER G3*** codecs, not inputs 3-5 which are available on ***i*-Mix *G3*** codecs (it is possible to program input 3 on a rack unit codec).

Tieline ***G3*** codecs have several preset User Functions programmed into them. These differ slightly depending on the model that you have purchased. The preset user functions are outlined in the main reference manual for each codec in the section titled Preset User Functions. User Functions are advanced software setups that can be programmed into **Tieline** codecs. We recommend that you read this section in full to understand the concepts behind User Functions and how they are programmed.

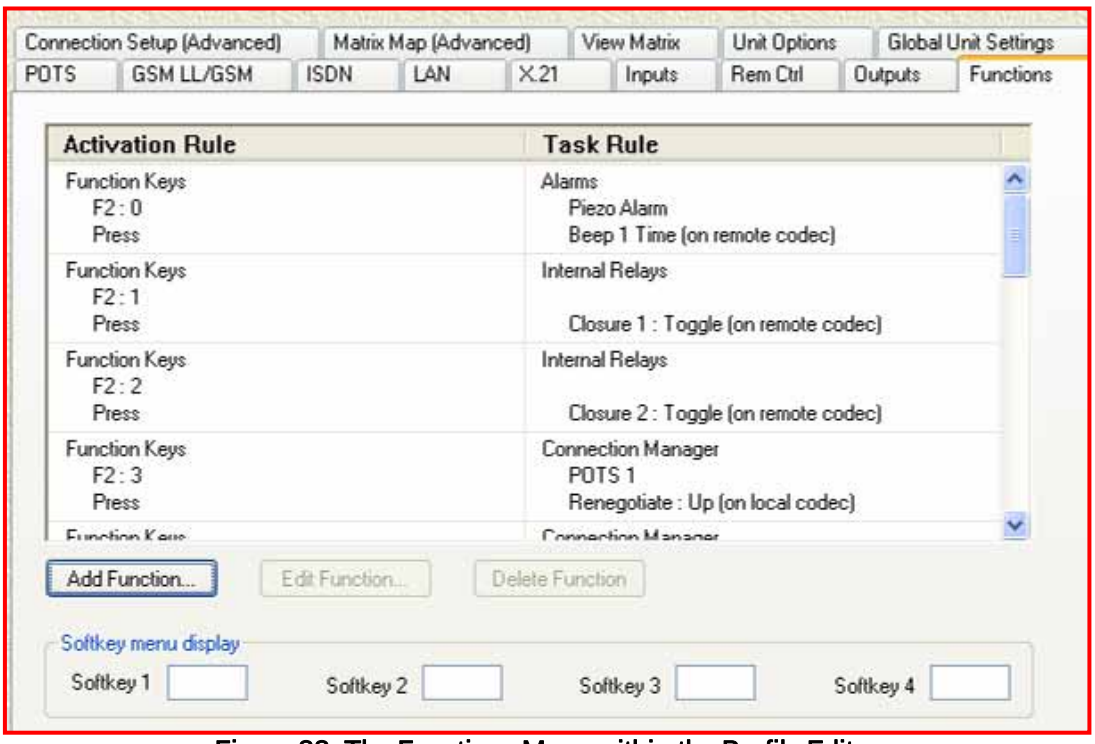


Figure 88: The Functions Menu within the Profile Editor

User Functions trigger certain events or allow you to access menu items. You have total flexibility in setting up these functions. Examples of some events you could program include:

- Programming an alarm to sound when the codec has connected.
- Programming the Connection Manager to renegotiate up or down.
- Programming the activation of internal or external relays.
- Programming simple failover connections (see Failover Profile).

Preset User Functions that have been set by **Tieline** are visible within the *Functions* menu and are explained fully later in this section of the manual.

31.1. Creating a New User Function

To create a new User Function click on the *Add Function* button on the *Functions* main menu screen within the *Profile Editor*. This will open the following menu as shown below.

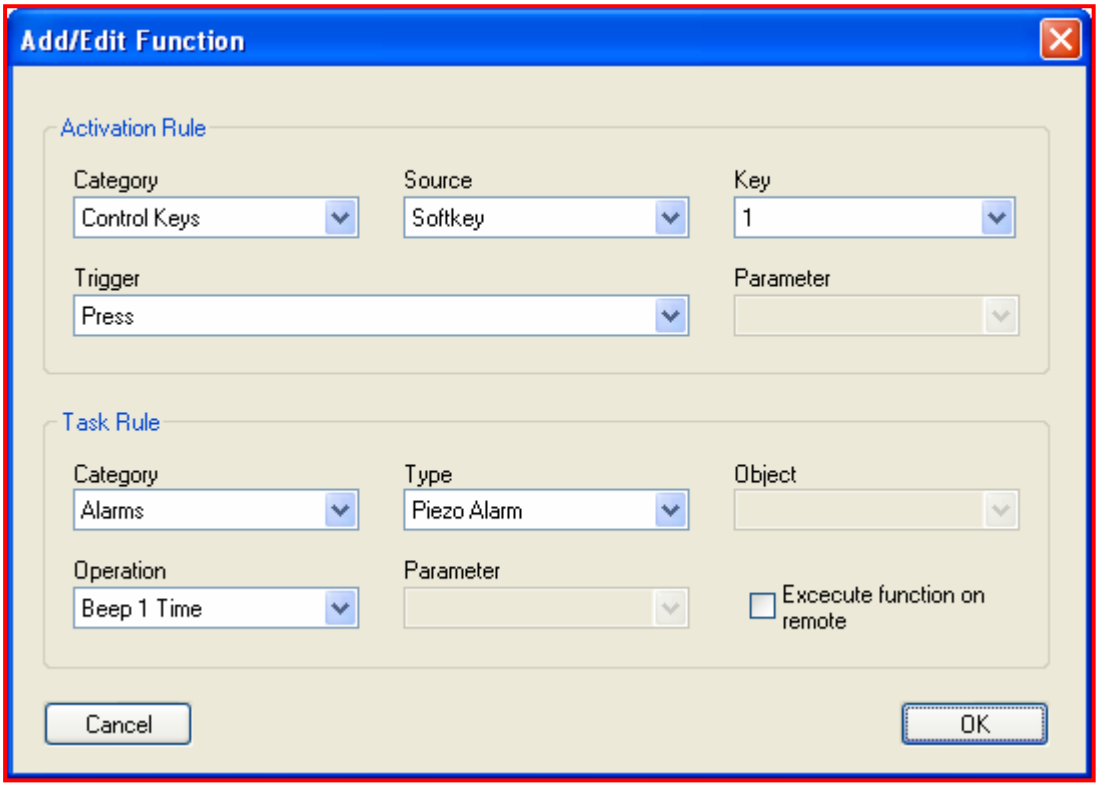


Figure 89: Add/Edit Menu in the Functions Menu

This screen will allow you to set all the required elements for each individual User Function. There are two sections to the *Add/Edit Function* menu –*Activation Rule* and *Task Rule*. *Activation Rule* defines what actions will initiate a User Function and *Task Rule* sets what the effect of that action will be.

31.2. Activation Rule

One way of looking at a User Function is to think in terms of a cause and effect relationship. There are two aspects to a User Function – an action which causes it to occur and the effect of that action or what it will actually change as a result.

Before you can program what the effect of a function will be, you must program what will cause a User Function to execute – its *Activation Rule*. In other words which functional element of the codec will initiate a User Function. For example, which external input, internal input, FUNCTION KEY or SOFTKEY will initiate a User Function.

Perhaps the easiest way to conceptualize the *Activation Rule* element of a User Function is to know that three things need to be programmed to set how it will be executed:

1. Decide whether a FUNCTION KEY, SOFTKEY, HOTKEY, internal input, external input or a connection event will be used to execute a User Function via *Category*;
2. Decide which specific FUNCTION KEY, SOFTKEY, HOTKEY, internal input, external input or connection event will actually be used to execute a User Function via *Source* and *Key*, and
3. Decide which *Trigger* will facilitate the specific FUNCTION KEY, SOFTKEY, HOTKEY, internal input, external input or connection event executing a User Function (Set via *Trigger* drop-down menu).

Following is a list of the elements within the *Category* section of the menu and a description of them.

31.2.1. Category Menu

As previously mentioned, *Category* sets whether a FUNCTION KEY, SOFTKEY, HOTKEY, internal input, external input or connection event, will be used to execute a User Function. There are five categories in the drop-down menu as shown below.

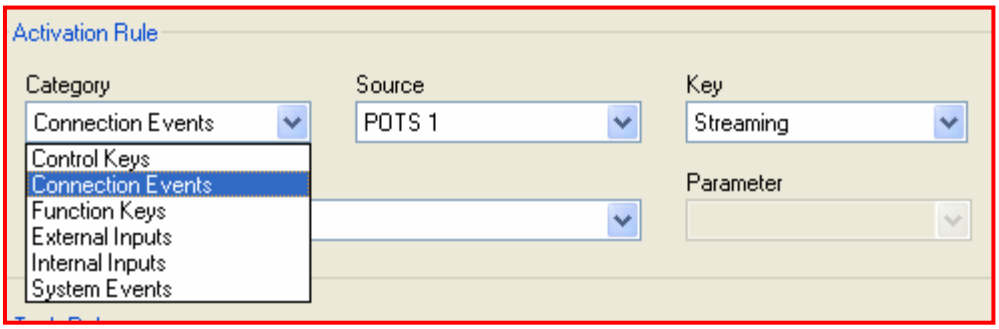


Figure 90: Functions Menu - Category Drop-down Menu

Following is a brief overview of the elements within the *Category* drop-down menu:

Category	Description
Control Keys	Allows you to select whether User Functions will be activated by a SOFTKEY or HOTKEY.
Connection Events	Allows you to select if User Functions will be initiated by connection functions of the codec.
Function Keys	Allows you to select whether User Functions will be activated by the FUNCTION KEYS (F1 and F2). Pressing the F1 or F2 keys and a number (0-9) on the keypad of a codec activates these User Functions.
External Inputs	Allows you to select whether User Functions will be activated by CAN 8+8 expansion box sources plugged into the CAN port on the rear panel of the codec.
Internal Inputs	Allows you to select whether User Functions will be activated by sources which are plugged into Control Ports 1 and 2 on the rear panel of the codec.
System Events	Allows you to select whether a system event, such as AES sync loss will activate a User Function. **(TLR300B only)**

Table 13: Functions Menu - Category Menu

Once you have decided which *Category* will be used to activate a User Function you will need to specify the *Source* of the *Category* you select. The relationship between a *Category* and its *Source* is shown in the table in the next section of this manual.

31.2.2. Source Menu

As mentioned previously, *Source* allows you to program how you wish to activate a User Function for each *Category*. It is simply a matter of specifying which FUNCTION KEY, SOFTKEY, HOTKEY, internal input, external input or connection event will actually be used to execute a User Function.

Category	Source
Control Keys	Selects whether a SOFTKEY or HOTKEY (or the PPM SELECT button on the TLR300 ***COMMANDER G3*** 2RU front panel or GUI) will activate a User Function.
Connection Events	Selects a method of connection you want to use to activate a User Function - POTS, ISDN, IP, X.21 or GSM.
Function Keys	Selects whether the F1 or the F2 FUNCTION KEYS will be used to activate a User Function. An Item key also needs to be pressed. For example F1 and a number between 0 and 9 on the keypad.
External Inputs	Up to four relay boxes with eight relays can be connected to the codec. The relay box you wish to use in activating a User Function is selectable in this menu. The selections are named *Modules 1 - 4*.
Internal Inputs	This section is 'grayed-out' as it is only necessary to select the relevant input with the *Key* menu.
System Events	This section is 'grayed-out' as it is only necessary to select the relevant input with the *Key* menu. **(TLR300B only)**

Table 14: Functions Menu - Source Menu

In addition to your selection in the source menu, you still need to specify the number of the SOFTKEY, HOTKEY, FUNCTION KEY, relay etc. – this will be discussed in the next section titled *Key*.

31.2.3. Key Menu

Along with the *Source* drop-down menu, *Key* selects which specific button or relay you want to use to activate a User Function. In the case of the *Category* called *Connection Events*, *Key* refers to selecting various modem and ISDN connection states. These are described in detail in the following table.

Category	Source	Key	
Control Keys	SOFTKEYS	Sets the number of the SOFTKEY that needs to be pressed to initiate a User Function.	
	HOTKEYS	Sets the number of the HOTKEY that needs to be pressed to initiate a User Function.	
Connection Events	POTS 1 & 2	Streaming (status)	Allows the user to set the *Trigger* for a User Function based on audio streaming status.
		Bitrate	Allows the user to set the *Trigger* for a User Function based on the bit-rate of the codec connection. The bit-rate needs to be set in the menu item *Parameter*.
		Status (connection)	Allows the use of an element of the POTS modem connection process to trigger a User Function.
	IP1 and IP2	Streaming (status)	Allows the user to set the *Trigger* for a User Function based on audio streaming status.
		Bitrate	Allows the user to set the *Trigger* for a User Function based on the bit-rate of the codec connection. The bit-rate needs to be set in the menu item *Parameter*.
		Status (connection)	Allows the use of an element of the IP connection process to trigger a User Function.
	GSM 1 & 2	Streaming (status)	Allows the user to set the *Trigger* for a User Function based on audio streaming status.
		Bitrate	Allows the user to set the *Trigger* for a User Function based on the bit-rate of the codec connection. The bit-rate needs to be set in the menu item *Parameter*.
		Status (connection)	Allows the use of an element of the connection process to trigger a User Function.

Table 15: Key Menu

Continued...

Category	Source	Key	
	ISDN B1, B2, B3, B4.	**Status** (connection)	Allows the use of an element of the ISDN connection process to trigger a User Function.
		Streaming (status)	Allows the user to set the *Trigger* for a User Function based on audio streaming status.
	X.21 - 1 & 2	**Status** (connection)	Allows the use of an element of the X.21 connection process to trigger a User Function.
		Streaming (status)	Allows the user to set the *Trigger* for a User Function based on audio streaming status.
	Any Connection	**Status** (connection)	Sets which activity in both the modem and the ISDN connection process will trigger a User Function. It will execute the User Function whether it is an ISDN or modem connection occurring.
		Streaming (status)	Allows the user to set the Trigger for a User Function based on audio streaming status. It will execute the User Function whether it is an ISDN or modem connection.
Function Keys	F1 and F2 FUNCTION KEYS	The numeric keys 0-9 on the keypad, that once pressed after the FUNCTION KEY (F1 or F2), will activate a User Function.	
External Inputs	Modules 1-4	Sets which relay from numbers 1 – 8 will activate a User Function. There are provisions for four external relay boxes with eight opto-isolated inputs in each.	
Internal Inputs	'Grayed-out' and inactive.	Sets which of the two Control port inputs, or input power, will activate a User Function.	
System Events	'Grayed-out' and inactive.	Sets AES sync loss as the system event to activate a User Function. **(TLR300B only)**	

31.2.4. Trigger Menu

For four of the five *Activation Rule Categories*, this menu selects what switching *Trigger* will actually execute the *Source* and *Key* selections for a User Function.

The complete *Activation Rule* options for the *Trigger* pull-down menus are included in the table in following pages:

Table 16: Functions Menu - Trigger Menu

Category	Source	Key	Trigger	Description
Control Keys	SOFTKEYS and HOTKEYS	1 - 4	Press	A User Function is triggered for as long as a Control Key is pressed (i.e. off to on).
			Release	A User Function is triggered for as long as a Control Key is released (i.e. on to off).
			Momentary	A User Function is triggered by any momentary depression of a Control Key (i.e. either off to on, or on to off).
Connection Events	POTS 1 & 2	Strea-ming	Activated	When the codec begins streaming audio a User Function is triggered.
			Deactivated	A User Function is triggered when streaming is deactivated.
			Any change	A User Function is triggered when streaming is either activated or deactivated.
		Bitrate	Greater than	The connection bit-rate can be used to *Trigger* a User Function. This menu setting allows the User Function to be triggered after surpassing a specified bit-rate – this must be specified in the *Parameter* menu.
			Less than	The connection bit-rate can be used to *Trigger* a User Function. This menu setting allows the Function to be triggered after falling below a specified bit-rate – this must be specified in *Parameter*.
		Status	Connecting	When modem connection is initiated a User Function will be executed.
			Connected	When the modem connects a User Function will be executed.

Table continued on the next page...

Category	Source	Key	Trigger	Description
	POTS 1 & 2 cont...		Dis-connected	When the modem disconnects a User Function will be executed.
			Ringing	When the codec starts to call and the ring tone is heard a User Function will be executed.
	IP 1 & 2	Strea-ming	Activated	When the codec begins streaming audio a User Function is triggered.
			Deactivated	A User Function is triggered when audio streaming is deactivated.
			Any change	A User Function is triggered when audio streaming is either activated or deactivated.
		Bitrate	Greater than	The connection bit-rate can be used to *Trigger* a User Function. This menu setting allows the User Function to be triggered after surpassing a specified bit-rate – this must be specified in the *Parameter* menu.
			Less than	The connection bit-rate can be used to *Trigger* a User Function. This menu setting allows the Function to be triggered after falling below a specified bit-rate – this must be specified in *Parameter*.
		Status	Connecting	When dialing is initiated a User Function will be executed.
			Connected	When the codec connects a User Function will be executed.
			Dis-connected	When the codec disconnects a User Function will be executed.
			Ringing	When the codec starts to call and the ring tone is heard a User Function will be executed.
	GSM 1 & 2	Strea-ming	Activated	When the codec begins streaming audio a User Function is triggered.
			Deactivated	When the codec is streaming audio a User Function is triggered when streaming is deactivated.

Continued on next page...

Category	Source	Key	Trigger	Description
Connection Events continued...			Any change	A User Function is triggered when audio streaming is either activated or deactivated.
		Bitrate	Greater than	The connection bit-rate can be used to *Trigger* a User Function. This menu setting allows the User Function to be triggered after surpassing a specified bit-rate – this must be specified in the *Parameter* menu.
			Less than	The connection bit-rate can be used to *Trigger* a User Function. This menu setting allows the Function to be triggered after falling below a specified bit-rate – this must be specified in *Parameter*.
		Status	Connecting	When a GSM connection is initiated a User Function will be executed.
			Connected	When the GSM call connects, a User Function will be executed.
			Dis-connected	When the GSM call disconnects, a User Function will be executed.
			Ringing	When the codec starts to call and the ring tone is heard a User Function will be executed.
	ISDN B1 – B4	Status	Connecting	When ISDN connection is initiated a User Function will be executed.
			Connected	When ISDN connection occurs a User Function will be initiated.
			Dis-connected	When ISDN connection ceases a User Function will be initiated.
			Ringing	When the codec begins to call a User Function is initiated.

Table continued on the next page...

Category	Source	Key	Trigger	Description
Connection Events continued...	ISDN B1 –B4 continued.	Streaming	Activated	When the codec begins streaming audio in ISDN mode, a User Function is triggered.
			De-activated	When the codec is streaming audio in ISDN mode, a User Function is triggered when streaming is deactivated.
			Any change	When the codec is streaming audio a User Function is triggered when streaming is either activated or deactivated.
	X.21-1& 2	Status	Connecting	When X.21 connection is initiated a User Function will be executed.
			Connected	When X.21 connection occurs a User Function will be initiated.
			Disconnected	When X.21 connection ceases a User Function will be initiated.
			Ringing	When the codec begins to call a User Function is initiated.
		Streaming	Activated	When the codec begins streaming audio in X.21 mode, a User Function is triggered.
			De-activated	When the codec is streaming audio in X.21 mode, a User Function is triggered when streaming is deactivated.
			Any change	When the codec is streaming audio a User Function is triggered when streaming is either activated or deactivated.
	Any Connection	Status	Connecting	When the codec connection process is initiated, in any connection mode, it will trigger a User Function.
			Connected	When codec connection occurs, in any connection mode, a User Function will be initiated.
			Disconnected	When the codec disconnects, in any connection mode, a User Function will be initiated.
			Ringing	When the codec begins to call, in any connection mode, a User Function is executed.

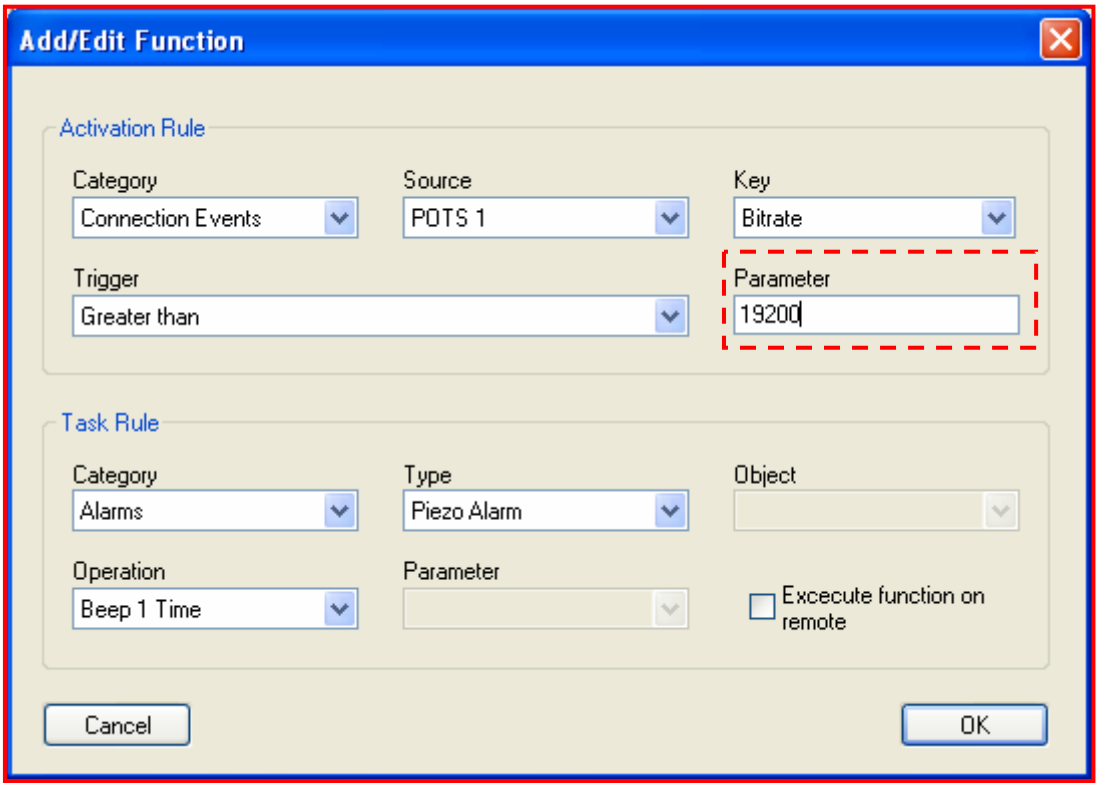
Table continued on the next page...

Category	Source	Key	Trigger	Description
	Any Connection cont...	Strea-ming	Activated	When the codec begins streaming audio in any connection mode, a User Function is triggered.
			De-activated	When the codec is streaming audio in any connection mode, a User Function is triggered when streaming is deactivated.
			Any change	When the codec is streaming audio in any connection mode, a User Function is triggered when streaming is either activated or deactivated.
FUNCTION KEYS	F1 & F2 Keys	0 - 9	Press	A User Function is triggered for as long as a Function Key is pressed (i.e. off to on).
External Inputs	Modules 1- 4	Inputs 1 - 8	Activated	Switches from off to on to trigger the User Function.
			De-activated	Switches from on to off to trigger the User Function.
			Any change	Either switches from on to off, or from off to on to trigger the User Function.
Internal Inputs		Inputs 1 & 2 & Input Power	Activated	Switches from off to on to trigger the User Function.
			De-activated	Switches from on to off to trigger the User Function.
			Any change	Either switches from on to off, or from off to on to trigger the User Function.
System Events (TLR300B only)		AES Sync Loss	Activated	Switches from off to on to trigger the User Function.
			De-activated	Switches from on to off to trigger the User Function.
			Any change	Either switches from on to off, or from off to on to trigger the User Function.

Please note: *Source* is 'grayed-out' and inactive for the category *Internal Inputs*.

31.2.5. Parameter Menu

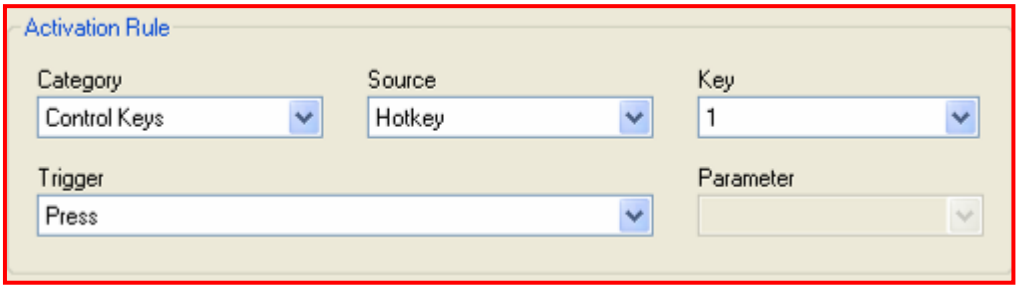
Parameter is used to set the actual bit-rate at which a User Function is activated for the *Category* called *Connection Events*. A User Function can be set to trigger either above or below the figure that is set within *Parameter*. In the following example the bit-rate has been set at 19,200 bps.



31.3. Some Activation Rule Examples

Following are some examples of *Activation Rule* settings used to execute User Functions.

1. In the following example, HOTKEY 1 has been set to initiate a User Function when the HOTKEY is pressed.



2. In the following example, a User Function has been programmed to execute once the *POTS 1* modem has connected to the remote codec.

Further examples will follow in the next section which focuses on setting the *Task Rule*. Remember, this is the action that will be initiated by the *Activation Rule* settings.

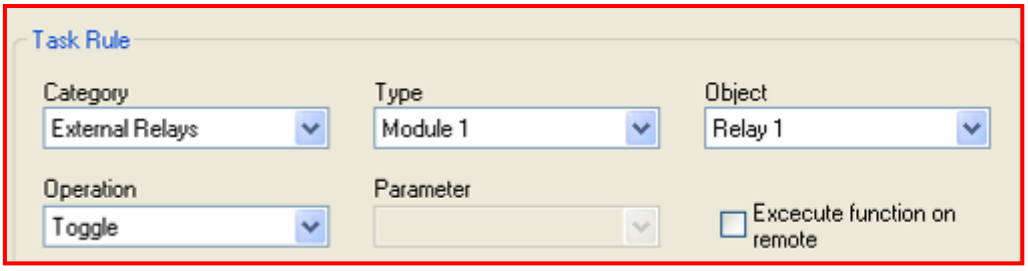
31.4. Task Rule

In the previous section which described how the *Activation Rule* was set, we showed that three things need to be programmed to set how a User Function will be executed. *Task Rule* sets what the effect of that *Activation Rule* setting will be. Therefore, the *Task Rule* section of the *Add/Edit* menu allows you to specify the effect of a User Function being executed.

Three things need to be programmed in order to create a valid function that can be executed:

1. Decide which *Category* of codec operation will change once a User Function is activated;
2. Decide which element of that category will be affected i.e. relay module or audio input (set with *Type* and *Object*); and
3. Decide how it will be affected i.e. the relay will toggle or the input gain setting will change (set with *Operation* and *Parameter*).

Following is a description of how these settings are created within the *Task Rule* menu. There are six elements to the *Task Rule* section as shown below.

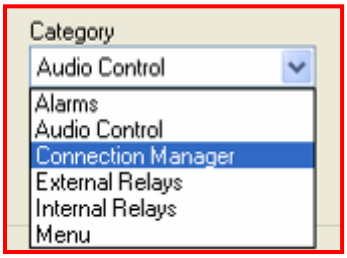


The *Category* pull down menu within the *Task Rule* menu lists the areas affected by a User Function. The pull-down menu allows you to select this area and you then need to set what element will be affected in the *Type* menu.

The individual element that will be affected is set with the *Object* menu i.e. which input, channel or relay. How you want it to be affected is set with the *Operation* and *Parameter* menus. In the example above, *Relay 1* in external relay *Module 1*

is set to toggle each time the *Activation Rule* settings for the User Function are activated.

31.4.1. Category Menu



The *Category* pull-down menu has six options that can be selected. In the following table the six options are described in detail.

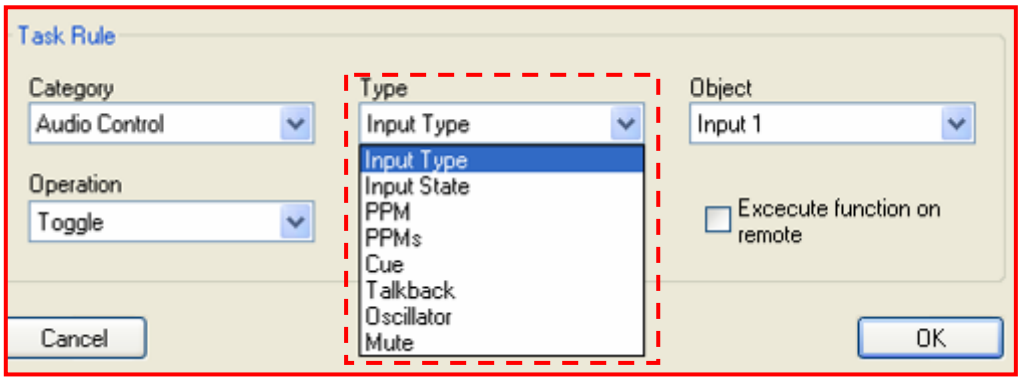
Category	Description
Alarms	This category allows you to set a Piezo or Battery alarm on the codec.
Audio Control	This category allows you to reset a variety of Audio Control features on the codec such as the inputs, talkback, PPM meters, the oscillator and muting functions.
Connection Manager	This category allows you to reset a variety of connection function features for POTS, GSM, IP, X.21 and ISDN operation.
External Relays	This category allows you to reset the switching functions for the codec's external relays.
Internal Relays	This category allows you to reset the switching functions for sources plugged into the codec's Control port inputs.
Menu	This category allows you to lock and unlock the console and also program the unit to display certain menus on the LCD screen on the codec.

Table 17: Functions Menu - Task Rule Category Menu

Following is a description of the various menu settings that can be set within the six *Categories*.

31.4.2. Type Menu

The *Type* menus allow you to set which element of the codec will be affected. For example whether you want to change an input setting, an external relay module or a POTS or ISDN connection.



For most selections in this menu you will also have to specify in more detail what will be affected i.e. which particular input, relay or ISDN connection. This will be done subsequently in the *Object* menu. Full descriptions of the options available for each *Category,* via the *Type* menu, are displayed in the table that follows.

Table 18: Functions Menu - Type Menu

Category	Type	Description
Alarms	Piezo Alarm	A piezo-electric beeper in the codec can be set as an alert.
	Battery	This allows you to display a variety of messages on the codec warning of low battery etc.
Audio Control	Input Type	Allows you to activate input options such as gain settings, unbalanced input and toggle.
	Input State	Select input state options such as input *Off, On, Toggle, Key and Inv Key.*
	PPM	Select from various options and change the setting of individual meters on the codec.
	PPMs	Select from various options and simultaneously change the setting for all meters on the codec.
	Cue	Select from various switching options to change Cue operation.
	Talkback	Select from various switching options to change talkback operation.
	Oscillator	Program the oscillator to send 400Hz tone to all codec outputs.
	Mute	Program either the Mute Program or Mute Return functions to activate.
Connection Manager	POTS 1 & 2	Program the codec to renegotiate or retrain one or two modems, to dial a phonebook entry or hang up a call.
	ISDN B1, B2, B3 & B4	Program the codec to dial a phonebook entry or hang up/answer a call.
	GSM 1 & GSM 2	Program the codec to dial a phonebook entry or hang up/answer a call.
	IP1 & 2	Program the codec to dial a phonebook entry, renegotiate or hang up/answer a call.
	3G + IP1	Program the codec to dial/hangup a 3G network & dial or hang up an IP call.
	X.21 – 1 & 2	Program the codec to dial a phonebook entry, renegotiate or hang up/answer a call.

Category	Type	Description
External Relays	Module 1	This allows you to select which out of four possible external relay boxes you wish to program. You will also need to select the individual relay itself and how you want it to be affected via the *Object* and *Operation* menus.
	Module 2	
	Module 3	
	Module 4	
Internal Relays	Not active	'Grayed-out' and inactive.
Menu	Remote Input	Allows you to program the codec to display the remote control menu for a specified input.
	Switch Console	Allows you to go to the next console listed in the codec's LCD menu via Console Navigation.
	Lock Console	Allows you to program locking and unlocking of the codec console.
	One RU LEDs (TLR300B use only)	Allows you to program different states of the two one RU LEDs named Assign 1 & 2.

Now that you have a broad understanding of the elements within the *Type* menu for each *Category* of *Task Rule*, we will provide a detailed table showing how each setting in the associated *Object* and *Operation* menus can impact on your *Task Rule* settings.

31.4.3. Object and Operation Menus

The next two menu selections for *Task Rule*, *Object* and *Operation*, will be looked at together but in relation to relevant *Category* and *Type* settings. The *Object* menu sets the specific input, relay etc. and the *Operation* menu sets the actual effect on that *Object*.

Table 19: Functions Menu - Object and Operation Menus

Category	Type	Object	Operation	Description
Alarms	Piezo Alarm	'Grayed-out' and inactive.	Beep 1-10 times or forever.	The piezo alarm can be set to beep between one and ten times or until cancelled by the operator.
Audio Control	Input Type	Inputs 1-5	Toggle	If set to *Toggle*, the User Function will toggle the setting through the list of mic input level settings.
			High Gain Mic	Sets the mic input level to *High Gain Mic*.
			Med Gain Mic	Sets the mic input level to *Med Gain Mic*.
			Low Gain Mic	Sets the mic input level to *Low Gain Mic*.
			Un-balanced	Sets the mic input level to *Unbalanced Gain Mic*.
			Line Level	Sets the mic input level to *Line Level* mic gain.
	Input State	Inputs 1-6	Off	The input will be switched off when the User Function is initiated.
			On	The input will be switched on when the User Function is initiated.
			Toggle	The input will toggle between on and off each time the User Function is executed.
			Key	The input will be on for the duration a key is pressed i.e. a SOFTKEY (The *Activation Rule Trigger* setting should be *Momentary*.)

Table continued on the next page...

Category	Type	Object	Operation	Description
Audio Control Continued			Inv Key	The input will be switched off for the duration a key is pressed i.e. a SOFTKEY. (The *Activation Rule Trigger* setting should be *Momentary*.)
	PPM	PPM 1, 2, 3 & 4	Toggle	If set to *Toggle*, the User Function will toggle individual PPM meters through the list of Encoders, Decoders and Matrix as listed under *Operation*.
			Matrix	Sets individual PPM meters to monitor the PPM settings for a profile, as set in the ***Matrix Editor***.
	PPMs	(Grayed out and inactive. All codec meters are adjusted with this setting.)	Toggle	If set to *Toggle*, the User Function will toggle all PPM meters through the list of Encoders, Decoders and Matrix as listed under *Operation*.
			Matrix	Sets all PPM meters to monitor the PPM settings for a profile, as set in the ***Matrix Editor***.
	Cue	Inputs 1-6	Off	The Cue function for a specified input will be switched off when the User Function is initiated.
			On	The Cue function for a specified input will be switched on when the User Function is initiated.
			Toggle	The Cue function for a specified input will toggle between on and off each time the User Function is executed.
			Key	The Cue function will be active on an input for the duration a key is pressed i.e. a CONTROL KEY (The *Activation Rule Trigger* setting should be *Momentary*.)
			Inv Key	The Cue function will be inactive on an input for the duration a key is pressed i.e. a CONTROL KEY. (The *Activation Rule Trigger* setting should be *Momentary*.)

Category	Type	Object	Operation	Description
Audio Control continued	Talkback	Channels 1-4	Off	The talkback function for a specified channel will be switched off.
			On	The talkback function for a specified channel will be switched on.
			Toggle	The talkback function for a specified channel will toggle between on and off each time the User Function is executed.
			Key	The talkback function will be active for the duration a key is pressed i.e. a CONTROL KEY (The *Activation Rule Trigger* setting should be *Momentary*.)
			Inv Key	The talkback function will be inactive for the duration a key is pressed i.e. a CONTROL KEY. (The *Activation Rule Trigger* setting should be *Momentary*.)
	Oscillator	'Grayed-out' and inactive.	Toggle	The User Function will activate the oscillator and send 400Hz tone at +4 dBu to each output of the codec.
	Mute	Mute Return	Off	The User Function will turn the *Mute Return* function off.
			On	The User Function will turn the *Mute Return* function on.
			Toggle	The User Function will toggle between turning the *Mute Return* function off and on.
			Key	*Mute Return* will become active while a key is pressed i.e. a CONTROL KEY. (The *Activation Rule Trigger* setting should be *Momentary*.)
			Inv Key	*Mute Return* will be inactive while a key is pressed i.e. a CONTROL KEY. (The *Activation Rule Trigger* setting should be *Momentary*.)

Category	Type	Object	Operation	Description
Audio Control continued...		Mute Program	Off	The User Function will turn the *Mute Program* function off.
			On	The User Function will turn the *Mute Program* function on.
			Toggle	The User Function will toggle between turning the *Mute Program* function off and on.
			Key	*Mute Program* will become active while a key is pressed i.e. a CONTROL KEY. (The *Activation Rule Trigger* setting should be *Momentary*.)
			Inv Key	*Mute Program* will be inactive while a key is pressed i.e. a CONTROL KEY. (The *Activation Rule Trigger* setting should be *Momentary*.)
Connection Manager	POTS 1 & 2	'Grayed-out' and inactive.	Reneg-otiate	The User Function will make the modem renegotiate according to the associated *Parameter* setting.
			Retrain	The User Function will make the modem retrain according to the associated *Parameter* setting.
			Dial Memory	If this is used in conjunction with the *Parameter* (phone #) setting, a number from the *Phonebook Editor* will be dialed.
			Answer	This setting will answer an incoming call.
			Hangup	This setting will cause a connection to disconnect.
	ISDN , B2, B3 & B4	'Grayed-out' and inactive.	Dial Memory	If this setting is used in conjunction with the *Parameter* (phone entry #) setting, a number from the *Phonebook Editor* will be dialed.
			Answer	This setting will answer an incoming call.
			Hangup	This setting will cause a connection to disconnect.
	GSM 1 & GSM 2	'Grayed-out' and inactive.	Dial Memory	If this setting is used in conjunction with the *Parameter* (phone entry #) setting, a number from the *Phonebook Editor* will be dialed.
			Answer	This setting will answer an incoming call.

Table continued on the next page...

Category	Type	Object	Operation	Description
	GSM 1 & GSM 2 cont...		Hangup	This setting will cause a connection to disconnect.
	IP1 & IP2	'Grayed-out' and inactive.	Reneg-otiate	The User Function will make the codec renegotiate according to the associated *Parameter* setting.
			Dial Memory	If this is used in conjunction with the *Parameter* (phone #) setting, a number from the *Phonebook Editor* will be dialed.
			Answer	This setting will answer an incoming call.
			Hangup	This setting will cause a connection to disconnect.
	X.21 – 1 & 2	'Grayed-out' and inactive.	Dial Memory	If this is used in conjunction with the *Parameter* (phone #) setting, a number from the *Phonebook Editor* will be dialed.
			Answer	This setting will answer an incoming call.
			Hangup	This setting will cause a connection to disconnect.
External Relays	Modules 1-4	Relays 1-8	Off	Relay switch from the on position to off.
			On	The relay will switch from the off position to on.
			Toggle	The relay will toggle between closed and open each time the User Function is executed.
			Key	The User Function will close the relay while a key is pressed i.e. a CONTROL KEY. (The *Activation Rule Trigger* setting should be *Momentary*.)
			Inv Key	The User Function will open the relay while a key is pressed i.e. a CONTROL KEY. (The *Activation Rule Trigger* setting should be *Momentary*.)
			Pulse	The relay will close for the duration in milliseconds set in the *Parameter* menu.

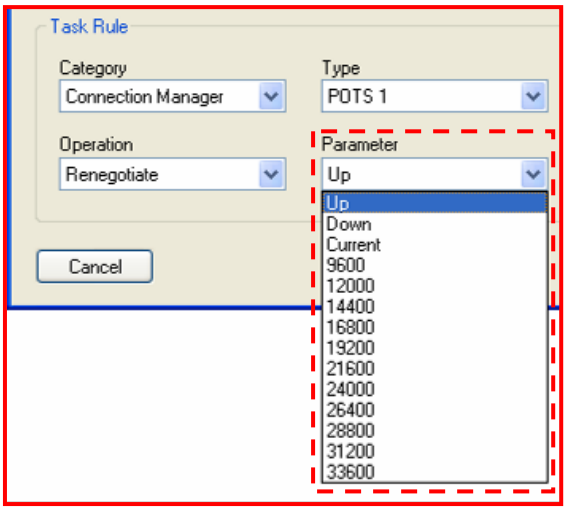
Table continued on next page...

Category	Type	Object	Operation	Description
Internal Relays	'Grayed-out and inactive.	Closures 1-4	Off	The relay will switch from the on position to off.
			On	The relay will switch from the off position to on.
			Toggle	The relay will toggle between open and closed each time the User Function is executed.
			Key	The User Function will close the relay while a key is pressed i.e. a CONTROL KEY.
			Inv Key	The User Function will open the relay while a key is pressed i.e. a CONTROL KEY.
			Pulse	The relay will close for the duration in milliseconds set in the *Parameter* menu.

Table continued on next page...

Category	Type	Object	Operation	Description
Menu	Remote Input	Inputs 1 - 4	'Grayed-out' and inactive.	The User Function displays the remote control menu for a specified input.
	Switch Console	'Grayed-out and inactive.	'Grayed-out' and inactive.	The User Function displays the next console listed via the Console Navigation menu, on the codec's LCD screen.
	Lock Console	'Grayed-out and inactive.	Lock	Locks the codec console.
			Unlock	Unlocks the codec console.
			Toggle	Toggles the codec to its opposite state, either from locked to unlocked or vice versa, each time the User Function is activated.
	One RU LEDs (TLR300B use only)	Assign 1 & 2	Off	Turns the LED off.
			On	Turns the LED on.
			Key	The User Function will turn the LED on while a key is pressed i.e. a CONTROL KEY.
			Inv Key	The User Function will turn the LED on while a key is pressed i.e. a CONTROL KEY.
			Blink	The LED is programmed to blink on and off. Use the *Parameter* setting in the menu to select the blink on/off time in milliseconds.
			State Blink	The LED is programmed to blink (illuminate) once for a pre-programmed time-frame. Use the *Parameter* setting in the menu to select the blink time in milliseconds.
			State Inv Blink	The LED is programmed to turn off once (from an illuminated state) for a pre-programmed time-frame. Use the *Parameter* setting in the menu to select the blink time in milliseconds.

31.4.4. Parameter: Renegotiating and Retraining



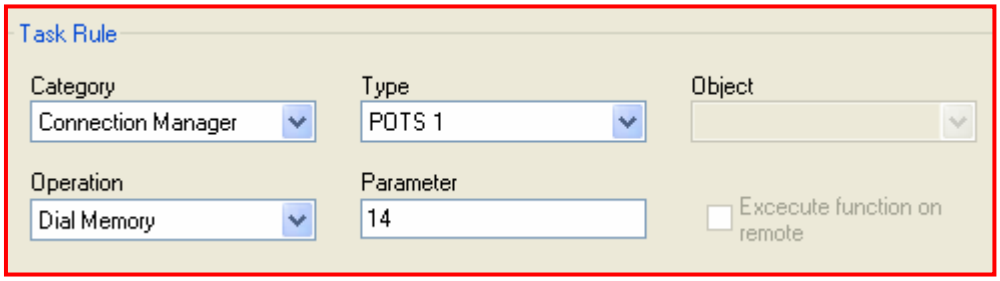
The *Parameter* menu within the *Task Rule* menu is linked to the *Connection Manager Category* as shown in this example. It allows you to set the bit-rate at which you want the codec to renegotiate to in POTS or IP modes.

You can also set the POTS modem to retrain up or down, depending on current line quality, or to connect at the current setting as set in the POTS menu within the ***Profile Editor***.

In the example above the modem is being programmed to renegotiate up when the User Function is executed. *Parameter* is also used to set the *Pulse* time in milliseconds for relay use.

31.4.5. Parameter: Setting Dialing User Functions

Parameter is also used if you are setting a User Function which uses *Dial Memory* in the *Operation* menu. *Parameter* will then allow you to select a phone number entry from the ***Phonebook Editor***. In the following example entry 14 is programmed to be dialed.



Please see the Failover Profile section in this manual for more information in relation to the previous *Task Rule* example.

31.4.6. Parameter: Setting One RU LED Blink Times

Parameter is also used if you are setting a User Function which uses the indicator LEDs on a ***COMMANDER G3*** one RU codec. The *Parameter* setting in this circumstance is used to enter the blink time in milliseconds of the LED that is being programmed via a user function.

31.4.7. Execute Function on Remote

If you are setting User Functions on the local codec, you can also set User Functions to be executed on the remote codec. A list of the functions that can be set by the local codec for the remote codec, are contained in the table that follows.

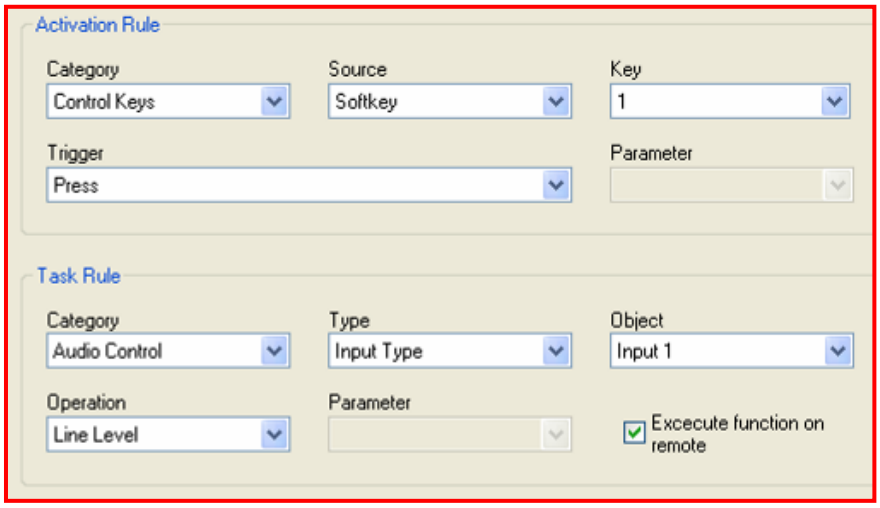
Category	Type	Object	Executable on Remote
Alarms	Piezo Alarm		Yes
Audio Control	Input Type	Inputs 1-5	Yes
	Input State	Inputs 1-6	Yes
	PPM	PPM 1 & PPM 2	Yes
	Cue	Inputs 1-6	No
	Talkback	Channels 1-4	No
	Oscillator		Yes
	Mute	Mute Return & Program	Yes
Connection Manager	POTS 1 & 2		No
	ISDN B1, B2, B3 & B4		No
	GSM 1 & 2		No
	IP1 & IP2		No
	3G + IP1		No
	X.21 – 1 & 2		No
External Relays	Modules 1-4	Relays 1-8	Yes
Internal Relays		Closures 1, 2, 3 & 4	Yes
Menu	Remote Input	Inputs 1-4	No
	Switch Console		No
	Lock Console		Yes
	One RU LEDs		Yes

Table 20: Functions Menu – Executable Function on a Remote Codec

When you are using HOTKEYS and SOFTKEYS on a local codec, if a User Function is initiated the relevant button will light up and then switch off when the opposite state is executed. No indication is given if a User Function is initiated by the use of FUNCTION KEYS.

When you execute a User Function on a remote codec, no indication is given on the local codec that the User Function has been initiated.

In the following example, the local codec has a programmed User Function whereby *Input 1* on the remote codec will be changed to a line-level input once the User Function is executed.



Note the *Execute function on remote* box has been checked and that it will be executed when SOFTKEY 1 is pressed on the local codec.

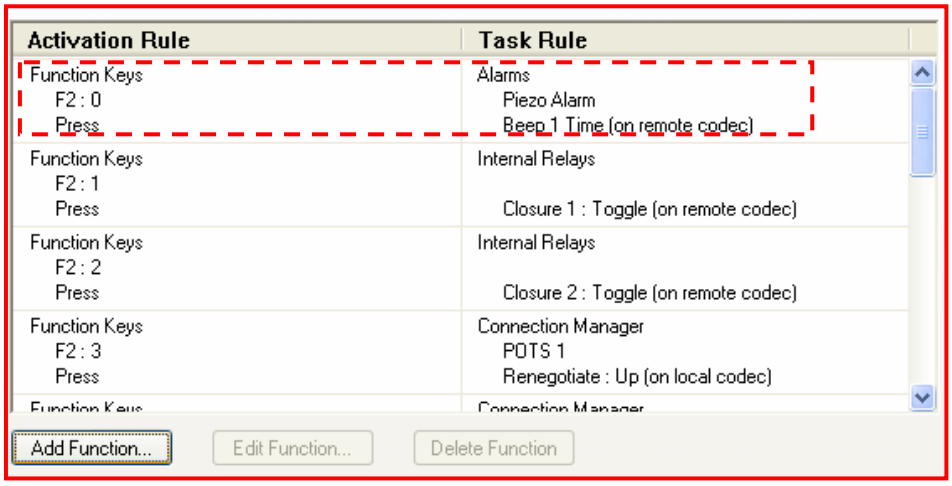
31.5. Preset User Functions

Tieline codecs have several preset User Functions programmed into them. Following is an image of each User Function as it is viewed within Tool**Box** and an explanation of its function within the codec.

Setup Tip: To view the setup for any of the following User Functions, simply select the User Function by clicking on it once to highlight it blue. Then press the *Edit Function* button in this *Functions* tab.

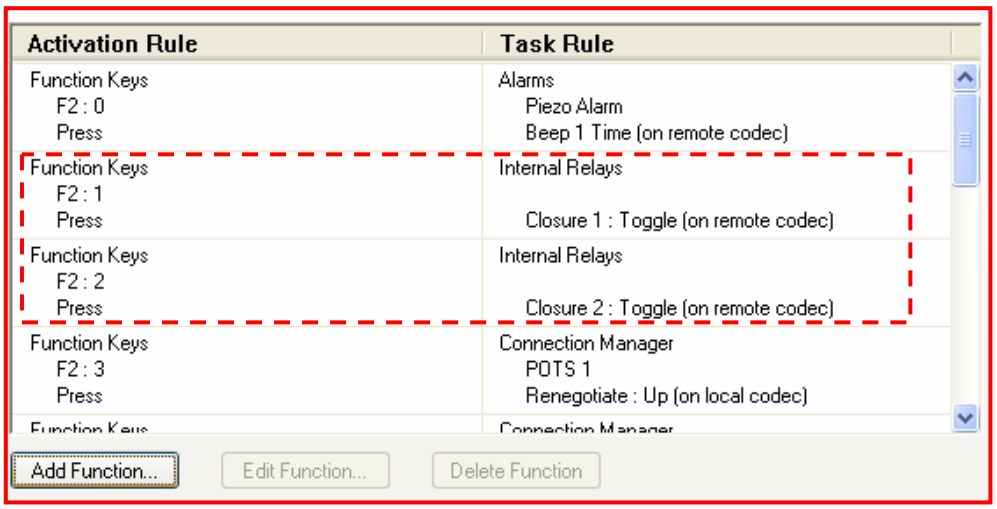
31.5.1. Piezo Alarm

This User Function will allow an announcer at the remote codec to use a Piezo Alarm to alert the studio if TB/comms is not available. It is activated by pressing the F2 button and then 0 on the remote codec's keypad.



31.5.2. Toggle Internal Relays 1 and 2

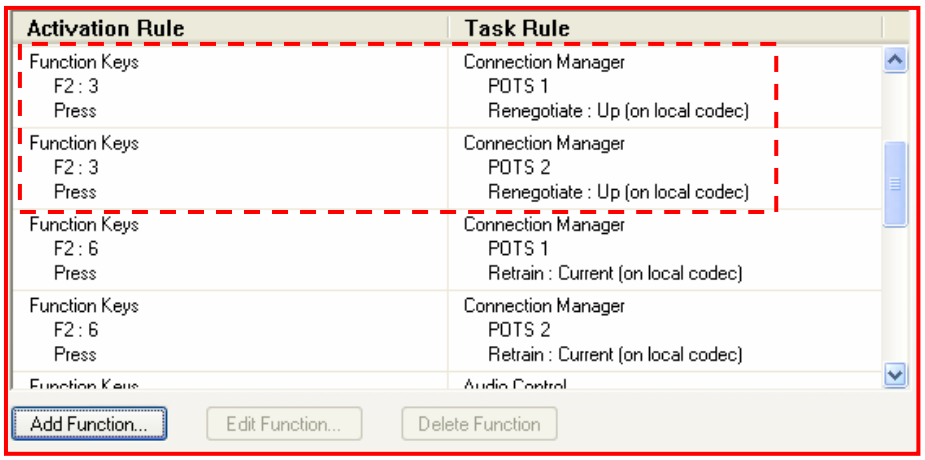
The next two codec User Functions are programmed to toggle internal relays on the remote codec. They affect these relays when they are activated via the F2 key and numeric keys 1 and 2 on the remote codec.



An example of using this User Function would be if a remote announcer wanted to put their program feed on-air in an unattended studio from the remote location, or if they wanted to activate a remote device which had radio stings or music loaded for integration into program.

31.5.3. Renegotiate POTS Modem Up

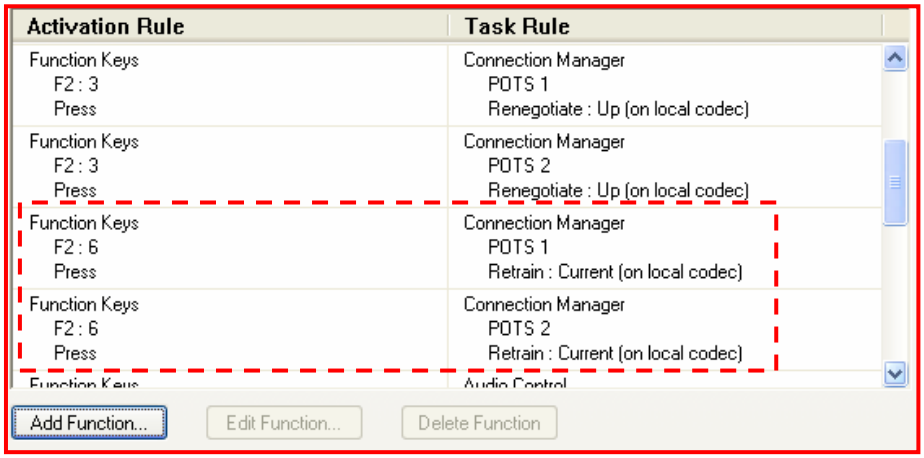
This User Function is set to make the modem (or modems if two POTS modules are installed) in the local codec renegotiate the connection upwards (as set by the *Parameter* setting in the *Add/Edit Function* menu). Each time you activate this User Function the codec will negotiate upwards to the next setting. For example, if you are currently connected at a bit rate of 19,200 bps the codec will renegotiate up to a bit rate of 21,600 bps. The setting the codec will renegotiate up to is also governed by the *Modem Max Bitrate* setting in the *POTS* menu.



This User Function would be used if line conditions have improved and you would like increase the connection bit rate to improve the quality of your connection. To activate the User Function simply press the F2 key and then 3 on the codec keypad. If two modems are being used then the renegotiation process will be initiated for both codecs.

31.5.4. Retrain POTS Modem: Current

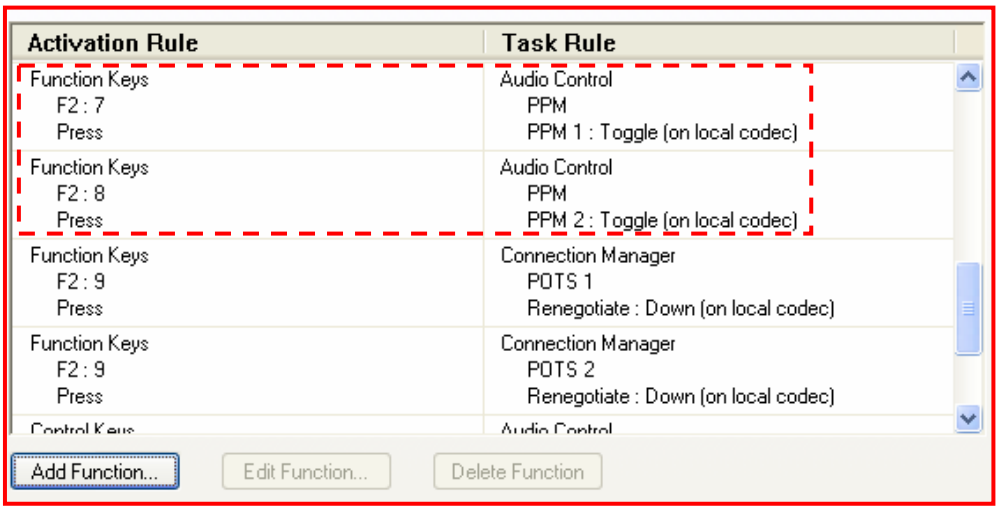
This User Function is set to make the local codec modem (or modems if two POTS modules are installed) retrain to the *Current* setting, as set in the *POTS* menu within the *Profile Editor*. To activate this User Function press the F2 key and 6 on the codec keypad. If two modems are being used then the retraining process will be initiated for both codecs.



This User Function would be useful if line quality has dropped over time and you would like to retrain the connection to see if you can improve it.

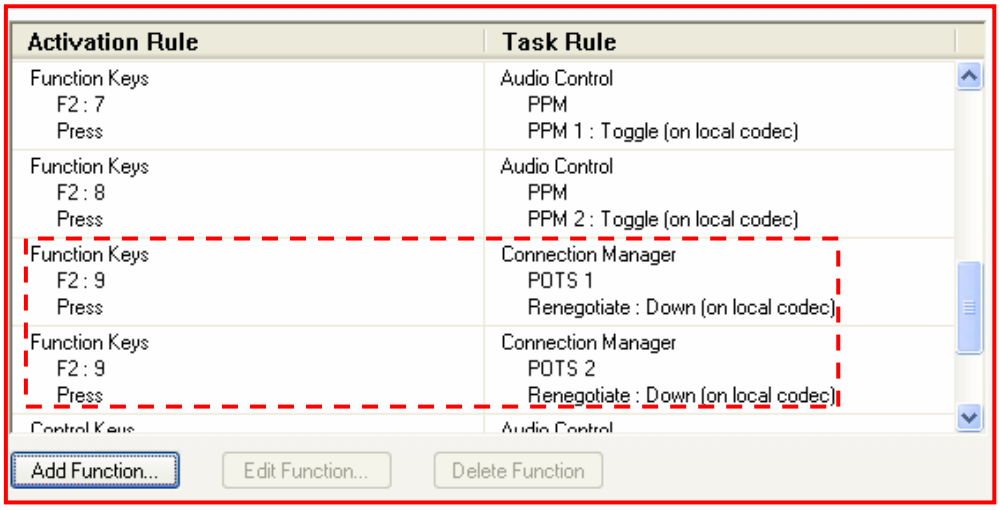
31.5.5. Toggle PPM 1 and 2

These User Functions are set to cause the PPM metering to toggle through the list of available options in the *Operation* menu within the *Add/Edit Function* menu. To toggle through the settings available for PPM1 simply press F2 and then 7 on the codec keypad. To toggle PPM2 simply press F2 and then 8 on the local codec's keypad. Each time this is done the next selection will be activated sequentially.



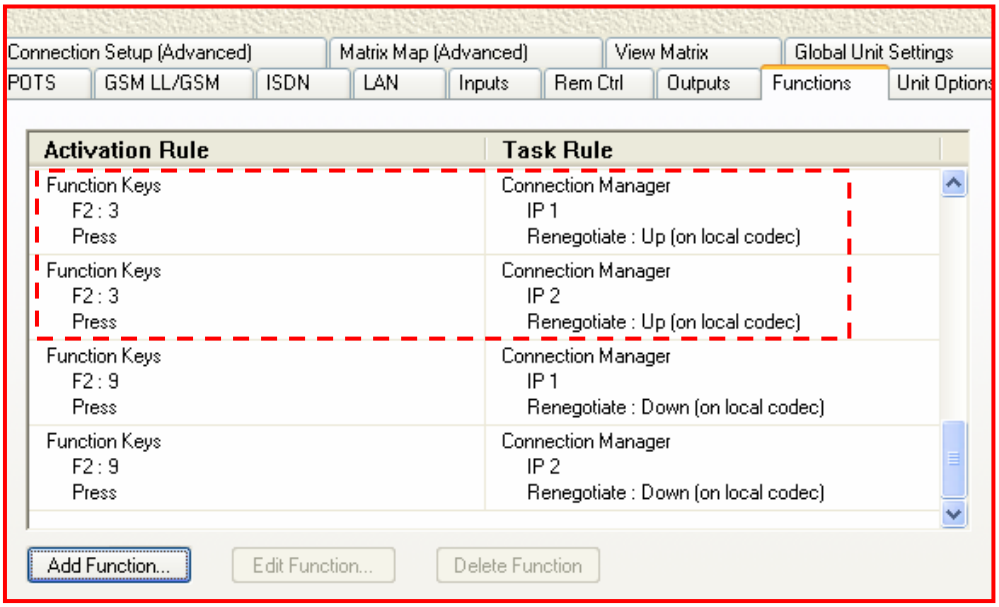
31.5.6. Renegotiate POTS Modem Down

This User Function will cause the local codec modem (or modems if two POTS modules are installed in the codec) to renegotiate down. This User Function could be useful if line conditions have worsened and the prevailing connection bit rate is likely to be unusable. To activate the function simply press the F2 key and then 9 on the local codec's keypad. If two modems are being used then the retraining process will be initiated for both codecs.



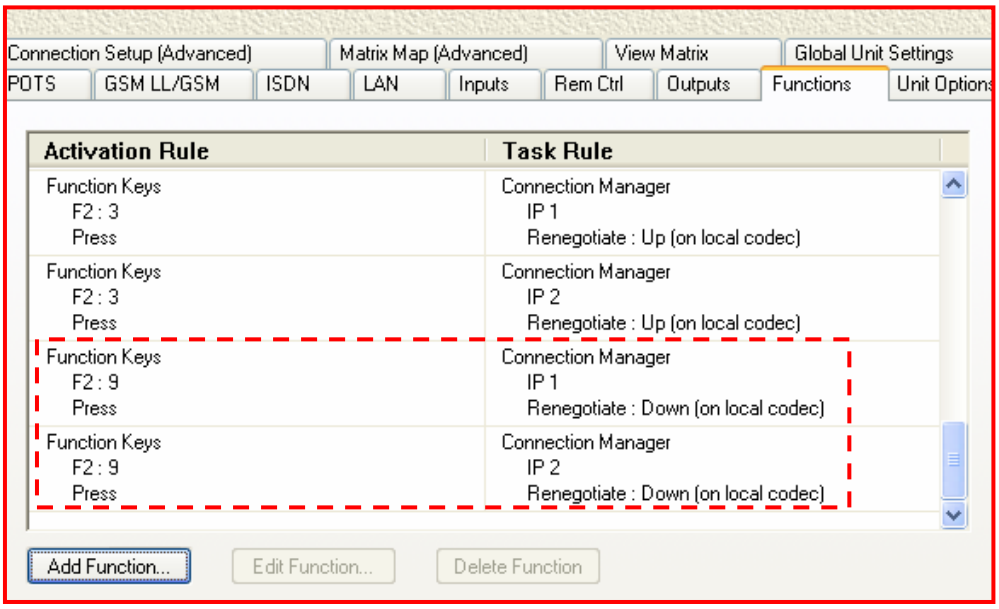
31.5.7. Renegotiate IP Connection Up

This User Function is programmed to make the IP connection on the local codec renegotiate the connection upwards (as set by the *Parameter* setting in the *Add/Edit Function* menu). Each time you activate this User Function the codec will negotiate upwards to the next bit rate setting. If you have two IP connections active then they will both attempt to renegotiate upwards.



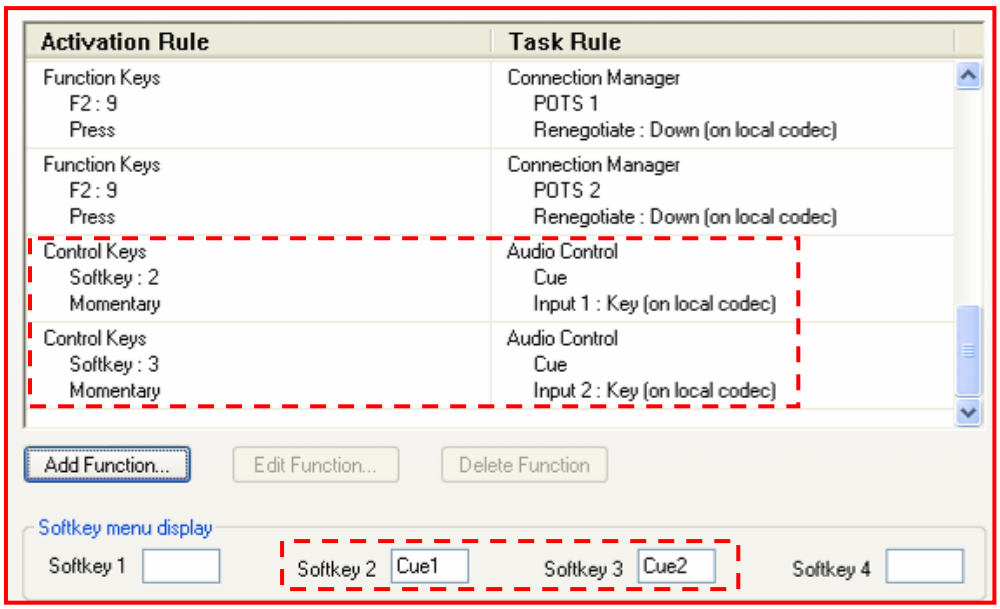
31.5.8. Renegotiate IP Connection Down

This User Function is programmed to make the IP connection on the local codec renegotiate the connection downwards (as set by the *Parameter* setting in the *Add/Edit Function* menu). Each time you activate this User Function the codec will negotiate upwards to the next bit rate setting. If you have two IP connections active then they will both attempt to renegotiate downwards.



31.5.9. Cue Function Key

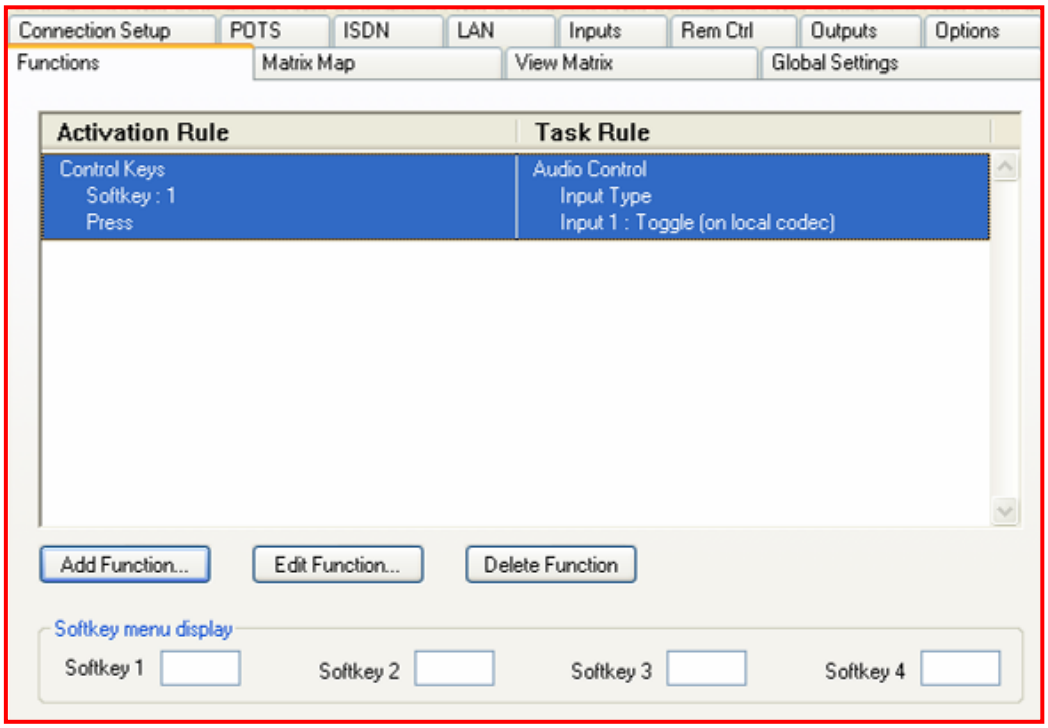
These User Functions enable SOFTKEYS 2 & 3 on ***COMMANDER G3*** field unit codecs to be used as Cue buttons for comms setups. The User Functions actually *Key* Inputs 1 & 2 on the local codec and enable routing as per the *Cue/Comms 1 & 2* matrices selected for these functions in the ***Matrix Map*** menu within the ***Profile Editor***.



You will also note that the *Softkey menu display* in the preceding image shows the alpha-numeric names attributed to SOFTKEYS 1 & 2 on the codec. To display these keys on the ***COMMANDER G3*** LCD screen simply press the MENU SELECTOR until they appear. To activate the User Function for each Cue function simply press either SOFTKEY 2 or 3 on the ***COMMANDER G3***. (Related topic: *Remote Cue*, *Cue*, Talkback, ***Matrix Map***)

31.6. Edit Function

To edit a previously set User Function, simply go to the ***Functions*** main menu screen. Click on the User Function in the list which you want to adjust which will highlight it in blue. Then click on the *Edit Function* button and the menu for that User Function will be displayed. Adjust the menu parameters as required and click *OK* to save the changes.



31.7. Delete Function

To delete a function no longer required, simply click on the function to highlight it in the ***Functions*** main menu and click the *Delete Function* button. The selected function will be deleted.

31.8. Softkey Menu Display

Another feature is the ability of the four SOFTKEYS on **Tieline** codecs to trigger several pre-programmed User Functions simultaneously. Up to five User Functions can be allocated to an individual SOFTKEY. An alphanumeric name can be given to each SOFTKEY with a total of four letters or numbers. This name will be displayed on the LCD screen of the codec when you press the

MENU SELECTOR continuously for two seconds. Programmed functions will be displayed for two minutes and then disappear. To remove the displayed functions from the screen press the MENU SELECTOR continuously for two seconds.

31.9. A Final Note...

Ultimately User Functions are an extremely flexible way for you to program your codec to perform in whatever way you want.

Section 32. Matrix Map

32.1. Matrix Management – A Three Step Process

It is important to realize that matrix management with the **Tieline** codecs is a three step process. These three steps are as follows.

1. Set the cross point connections for your matrices within the *Matrix Editor*. This is where you set the individual signal path routing for each mixer function of a profile within the codec, i.e. cue and talkback functions and the inputs.
2. Assign which matrix will be set to each mixer function listed within the *Matrix Map* menu and save the settings to a profile.
3. View and check your matrix settings within the *View Matrix* menu. This allows you to view the overall mixer settings you have made within the codec.

Step one above is described in detail in the section of this manual titled *Matrix Editor*. A thorough knowledge of this is required before proceeding with matrix mapping within the *Matrix Map* section of ToolBox.

A Cautionary Note:
It is strongly recommended that if you are using your codec for the first time you should thoroughly read the *Matrix Editor* section within this manual before attempting to set matrices with the *Matrix Map* menu.

Important Note on Matrices:
It is very important to note that *COMMANDER G3* rack unit matrices and profiles are different to field unit matrices and profiles.

If you are connecting a ***COMMANDER G3*** field unit to a rack unit codec and you are not using manual default profiles, you will need to create separate profiles and matrices for each codec using the correct *Matrix Editor* for each codec. To do this you will need to add a new configured device, i.e. a rack unit codec as well as a field unit codec, to the *Configured Devices* menu in ToolBox.

Once you have added a rack unit codec to the menu you can then select the *Matrix Editor* for it and amend, create and save matrices to a profile as required. (Please note that matrices for the TLR300 are different to the TLR300B rack unit codec matrices).

Very Important Note:

If you connect using manual default profiles, during the connection process the dialing codec will ensure the correct profile is loaded in the remote codec.

However, when you are not using manual default profiles (i.e. you are using custom profiles that you have created), different profiles need to be loaded onto both the local and remote codecs. These profiles will reflect the fact that each codec's matrices are set to receive send and receive different audio streams.

For example, it is important to remember that local and remote matrices need to be matched if features such as talkback are to be used. Also, if the local codec is in the Broadcast Center, the matrix settings on the codec will reflect the fact that it is receiving program audio from the remote or field codec, whereas the remote codec will be set to send program audio. Check that the talkback matrices of both the local and remote codecs are configured correctly.

Connection Tip: When you connect two codecs using manual default profiles, the dialing codec will instruct the codec that receives the call as to which profile should be active in the codec. If you are using other user created profiles that use matrices that you have created yourself, you need to ensure that you have the correct profiles (with matching 'mirror-image' matrices) loaded into both codecs, along with the correct talkback and connection settings i.e. *Auto answer* and *Automatic Redial* set correctly.

32.2. Matrix Mapping – An Overview

In each ***COMMANDER G3*** Configuration File, there is a set of routing matrices used by the various profiles defined within the file. There can be up to 160 user matrices in the Configuration File. Each profile has 10 routing matrices that have to be defined by the user. Matrix mapping is used to assign a matrix to each routing function in a profile. Each profile has the following routing functions:

- One normal 6 input by 12 output program routing matrix,
- 5 Cue routing matrices,
- 2 Talkback send routing matrices, and
- 2 Talkback receive routing matrices.

A set of reference matrices is provided with the ***COMMANDER G3*** codec to enable users to set the codec up for the most common functions that are required.

32.3. 'Mapping' Matrices for Codec Functions

Operational Tip: It is important to remember that local and remote matrices need to be matched if features such as talkback are to be used. For example, if the local codec is in the Broadcast Center, the matrix settings on the codec will reflect the fact that it is receiving the program audio from the remote or field codec, whereas the remote codec will be set to send program audio. Check that the talkback matrices of both the local and remote codecs are configured correctly. (Related topics: Manual Default Profiles and The Default Matrices)

To assign a particular matrix to a particular routing function, click with a mouse on the routing function listed under the heading *Mode* on the ***Matrix Map*** main menu page. This selects the function to be set. Then move your mouse cursor to the right hand end of the *Matrix* column and click once to display a drop-down menu listing the matrices to choose from. Move your cursor to the particular matrix you want and then click on it to select it. The drop-down menu listing the default matrices in the codec is shown in the following image.

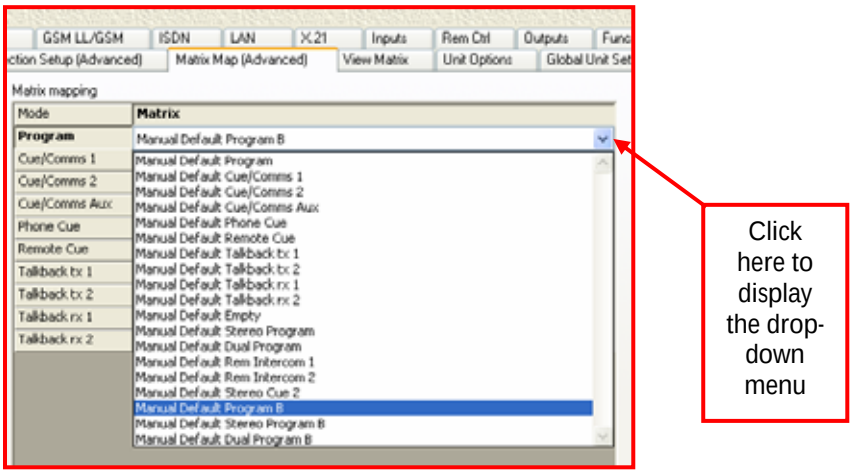


Figure 91: The drop-down menu for selecting matrices.

Alternatively, you can perform the same function without a mouse by using the arrow keys to scroll down and select the routing function under *Mode*, then press Enter on your keyboard. Then use the arrow keys to scroll to the matrix you want and press Enter to select it.

The *Program* routing matrix defines which inputs can be routed to which outputs. ToolBox software can be used to define the cross points selected in the matrix if you have purchased this option from **Tieline**. The logistics of this is discussed further in the ***Matrix Editor*** section of this manual.

The C*ue/Comms* matrices are used to define how the *Cue/Comms* functions will work in your codec.

32.4. Cue: An Overview

Please Note: The images in this section relate to ***COMMANDER G3*** field unit codecs but the general cue function principles explained here are similar for all **Tieline** codecs. The only differences are in the number of inputs and headphone outputs displayed in each codec's matrix tab.

The cue function can be used in different ways because its functionality is programmable with ToolBox. Every ***i*-Mix *G3*** and ***COMMANDER G3*** rack unit codec has a CUE-COMMS button associated with each of the balanced inputs.

32.4.1. Commander G3 Field Unit Cue Button

Both Inputs 1 & 2 on the ***COMMANDER G3*** have an associated Cue button facility which has been programmed into SOFTKEYS 2 & 3 via preset User Functions. For more information please see the Preset User Functions or Cue sections in this manual. When a cue-programmed SOFTKEY is pressed, the associated *Cue/Comms* matrix is utilized to route the audio.

As a default setting when you purchase your ***COMMANDER G3***, SOFTKEYS 2 & 3 are programmed to be *Cue/Comms* buttons for *Inputs 1 & 2*. To display them on the LCD, press the MENU SELECTOR and hold it down for a few seconds and the Cue1 and Cue2 buttons will appear and become active as displayed below.

Figure 92: The Cue Function as Displayed on a Commander G3 LCD

To understand how the SOFTKEYS can be programmed on your codec, please see the following sections within this manual: Preset User Functions; ***Functions*** and Remote Cue.

The various ways that cue functions can be programmed to perform different communications tasks are discussed in the following sections.

Operational Tip: A SOFTKEY which is programmed to operate for talkback, cue or intercom purposes will operate even if a mic input's CHANNEL ON button is not switched on.

32.5. Cue/Comms in Prefade or Off-line Monitoring Mode

The Cue function can be used in the typical way as a '**prefade**' function. In other words, the user is able to cue things up '**offline**', i.e. CD players and other external audio sources. In the following setup there are two inputs set up for announcers and the *Aux* input is set up for an external mono program source. If a SOFTKEY has been programmed to activate the *Cue/Comms* matrix for the *Aux* input, then every time it is pressed it will activate off-line or prefade monitoring of the *Aux* input in *HP1R* and *HP2R*. *Aux* input audio levels will also be monitored on PPM2.

TB tx
TB rx

		Encoder 1	Encoder 2	CH 1 Out	CH 2 Out	HP1 L	HP1 R	HP2 L	HP2 R	Phone Out	Auxilary	PPM 1	PPM 2
Cue/Comms	TB	Input 1	✓				✓		✓		✓	✓	✓	
Cue/Comms	TB	Input 2	✓				✓		✓		✓	✓	✓	
Cue/Comms		Aux						✓		✓				✓
Phone Cue		Phone	✓				✓		✓			✓	✓	
Remote Cue		Decoder 1			✓	✓	✓		✓					
		Decoder 2												

Figure 93: The Manual Default Cue/Comms Aux Matrix as Viewed in View Matrix

In the preceding example, the *Manual Default Cue/Comms Aux* matrix has been selected using the ***Matrix Map*** menu. This matrix has been used for the *Aux* input in all the manual default profiles.

32.6. Cue/Comms as 'Local' Intercom

Typically local intercom is set up so that commentators can talk amongst themselves at their own codec, without going 'to line' or talking to another codec. In the following example, the *Cue/Comms* matrices for Inputs 1 & 2 have been set so that when the cue function is activated for each of these inputs, audio from the commentators will be monitored in the right ear of both *HP1 & HP 2* only. Audio levels will also be monitored on PPM 2.

TB tx
TB rx

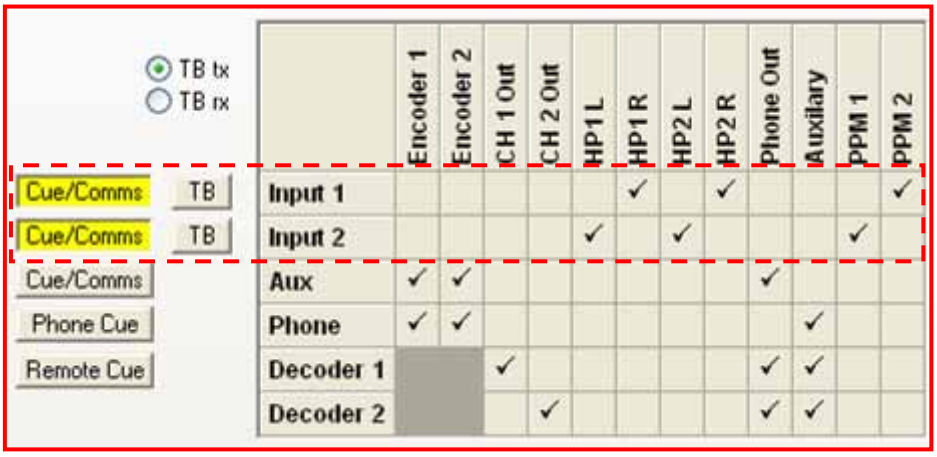
			Encoder 1	Encoder 2	CH 1 Out	CH 2 Out	HP1 L	HP1 R	HP2 L	HP2 R	Phone Out	Auxilary	PPM 1	PPM 2
Cue/Comms	TB	Input 1						✓		✓				✓
Cue/Comms	TB	Input 2						✓		✓				✓

Figure 94: Local Intercom Using Manual Default Cue/Comms Matrices

The matrices used in the preceding example were *Manual Default Cue/Comms 1 & 2*. These matrices have been applied, as displayed above, in all the manual default profiles except *Man Dflt Mono/IFB* and *Man Dflt Stereo*.

32.7. Using Cue/Comms as 'Local' Intercom in Stereo

If you are using the codec in stereo modes and utilizing *Cue/Comms* features (for example when using the *Man Dflt Stereo* profile), then the codec can monitor program audio in stereo and also monitor *Cue/Comms* in separate sides of the headphones. To do this the *Manual Default Stereo Cue 2* matrix has been set as the default for *Cue/Comms 2* in *Man Dflt Stereo*. Its settings are displayed in the following image.



Note that *Cue Comms 1* is set to be sent to the right side of both *HP1* and *HP2* via the *Manual Default Cue/Comms 1* matrix. *Cue Comms 2* is set to be set to the left side of both *HP 1* and *HP2* via the *Manual Default Stereo Cue 2* matrix .

32.8. Cue as Codec-to-codec Intercom

In the following example, the *Cue/Comms* function for *Input 1 & 2* is programmed to send audio both to *Encoder 2* for studio communications and to the right side of the headphones for both commentators to monitor. Audio levels are also being monitored via PPM 2. This matrix would be enabled when the pre-programmed Cue SOFTKEY is pressed for each of these inputs.

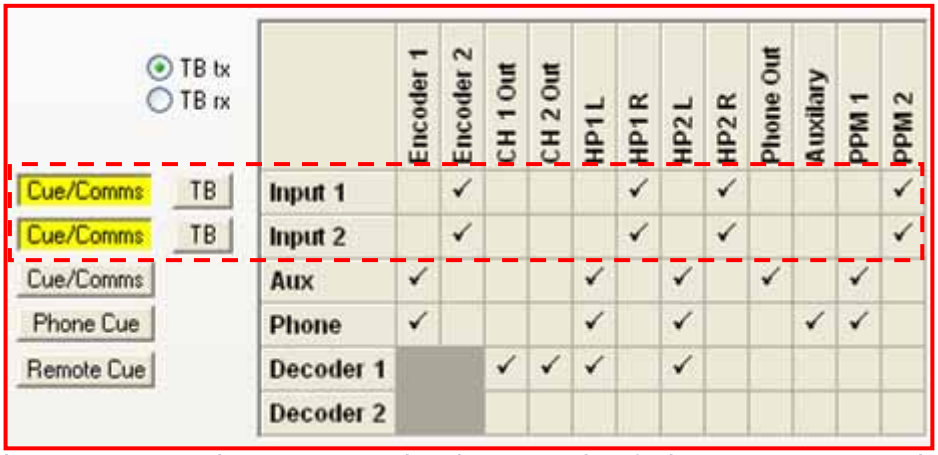


Figure 95: Inter-codec Intercom Using the *Manual Default Rem Intercom* Matrices

Effectively, commentators can listen to their own intercom, send talkback simultaneously to another codec and monitor audio levels being sent. The matrices shown in the preceding example are *Manual Default Rem Intercom 1 &*

2. These matrices are only preprogrammed to be used in the manual default profile *Man Dflt Mono/IFB*. Any codecs using the *Manual Default Mono/IFB* profile, or a similar profile, need to ensure that incoming audio is set to be received via *Accept remote cue*. This is enabled as a default for the *Manual Default Mono/IFB* profile.

32.9. Talkback

The talkback (*TB*) function is usually provided for 'discreet' producer/director talkback facilities between the local and remote codecs. There are two sets of matrices for talkback communication. Two are for talkback send (*TB tx*) and two are for talkback receive (*TB rx*). The send matrices are activated by pressing a pre-programmed SOFTKEY on the codec. The receive matrices are used when a talkback command is received over the POTS, ISDN or IP link.

The *View Matrix* settings for *TB tx* (talkback send) in the following image show that *Input 1 TB* is being sent to *Encoder 2* and *HP1R*. *Input 2 TB* is being sent to *Encoder 2* and *HP2R*. The talkback settings are 'discreet' in that *TB* listen is only activated on the input from which audio is being sent. (Related Topic: Talkback Level Adjustment)

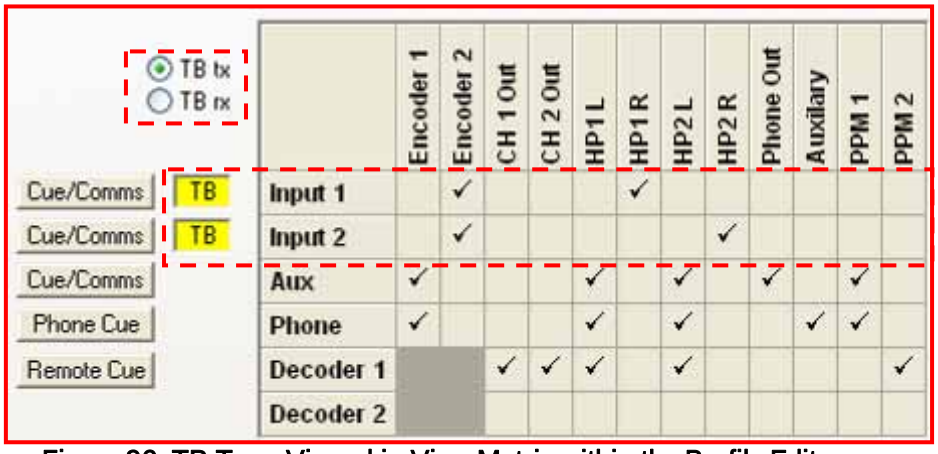


Figure 96: TB Tx as Viewed in View Matrix within the Profile Editor

The matrices used in the above image are *Manual Default Talkback tx 1 & 2*. These matrices are pre-selected in all manual default profiles except *Man Dflt Stereo* and *Man Dflt Dual Pgm*.

TB *Rx* (talkback receive) is also 'discreet' in that incoming audio is routed to either commentator 1 or 2 (*HP1R* and *HP2R*) as required via *Decoder 2* as shown in the following image.

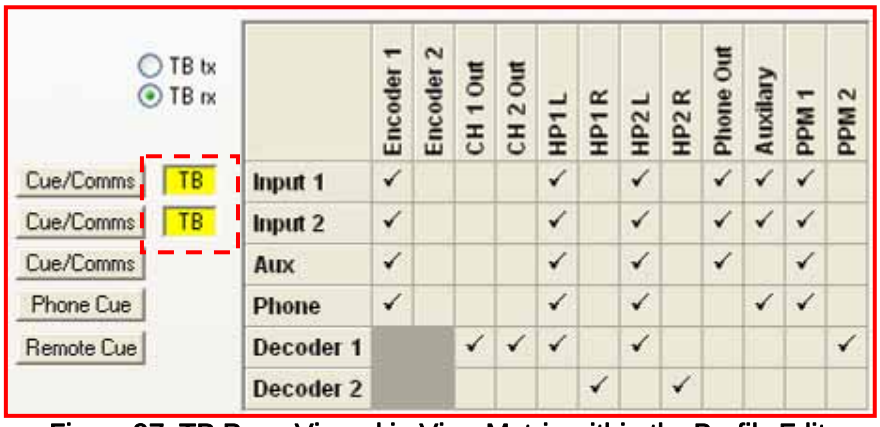
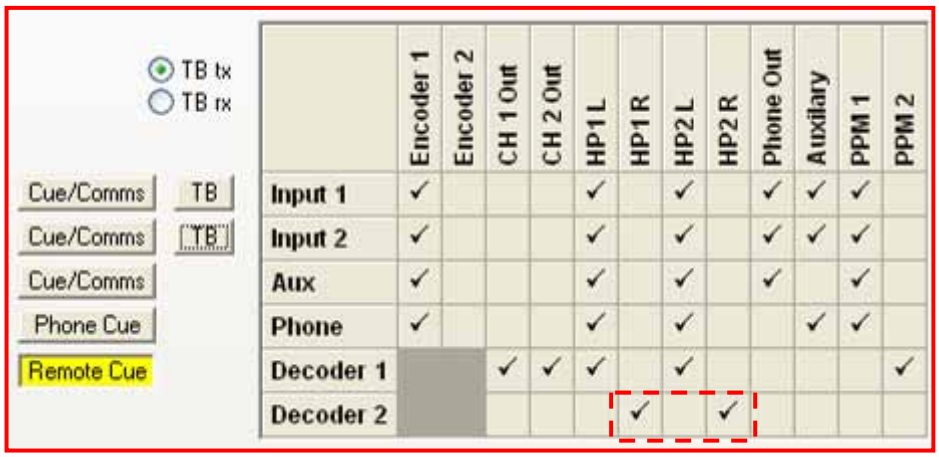


Figure 97: TB Rx as Viewed in View Matrix within the Profile Editor

The matrices used in the previous image are *Manual Default Talkback rx 1 & 2*. These matrices are pre-selected in all manual default profiles except *Man Dflt Stereo* and *Man Dflt Dual Pgm*.

32.10. Remote Cue

Remote Cue activates the listening function at a codec receiving audio via *Cue/Comms.* The default setting for this function in all codecs is on. For example, when *Accept remote cue* is checked on the remote codec, it allows the local codec to activate the *Remote Cue* matrix functions set on the remote codec. To do this simply press the *Cue* button as set via SOFTKEYS 2 & 3 on the ***COMMANDER G3*** field unit (or the CUE/COMMS buttons on the rack unit ***COMMANDER G3*** and ***i*-Mix *G3***). This will activate the ***Matrix Map*** settings for *Remote Cue* on the remote codec.



The settings for *Remote Cue* displayed previously are the settings within the *Manual Default Remote Cue* matrix. This matrix is only preprogrammed into the *Man Dflt Mono/IFB* profile because this is the only manual default profile set to utilize communications.

32.11. Phone as Talkback

If you are using an ***i*-Mix *G3*** codec there is a PHONE COUPLER which can be used for communications. This is discussed in the section of this manual titled Phone Input: Operation of the Phone Coupler.

The POTS modules within the ***COMMANDER G3*** are also capable of being used as a phone input. A discussion on how *Cue/Comms* can be integrated into overall codec communications is discussed in the next section.

32.12. Creating Profiles

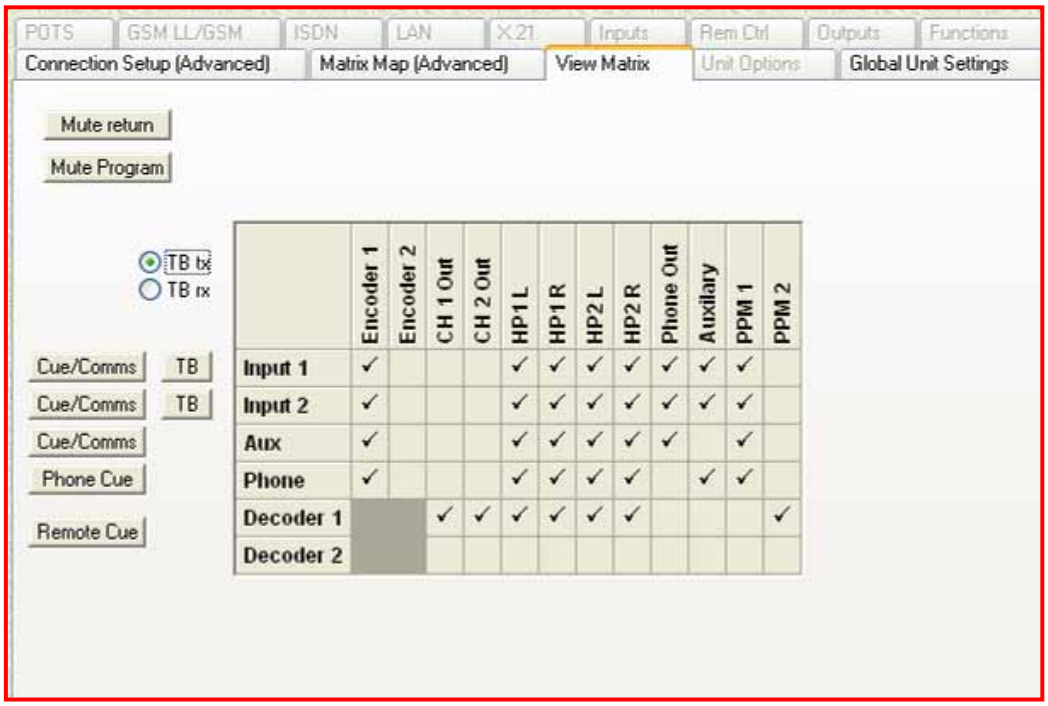
Once you have completed matrix mapping, you can create a new profile and name your setup for the broadcast you wish to use it for. Simply click on the *Edit Profile* drop-down menu and then click on *Add*. Name the file in the menu screen that appears and then save it. Your settings can then be saved as a Configuration File for later retrieval or to program onto other codecs. To do this click on *File* in the top left corner of the Tool**Box** menu and click on *Save Config*. Name the file and then click *Save*.

Section 33. View Matrix

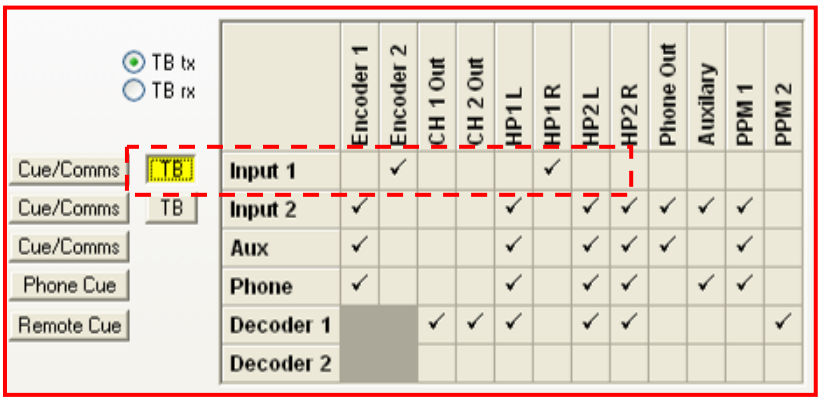
33.1. View Matrix Menu

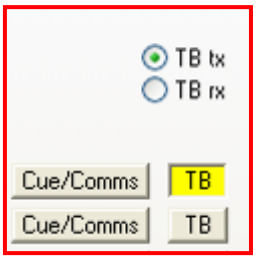
This menu allows the user to view the overall routing for the matrices that have been assigned within a profile via the ***Matrix Map*** menu, and to check that the mute, cue, intercom and talkback routing matrices work as intended. This is a view only menu screen. Changes to cross point routing within matrices can only be done with the ***Matrix Editor***. Matrix mapping (attributing matrices for mixer functions within the codec) can only be done in ***Matrix Map*** within ToolBox.

An example of the default program matrix for the *ManDflt Mono Pgm* profile, as it is displayed in the ***COMMANDER G3*** field unit's ***View Matrix*** tab, is shown below.



It is easy to check all of the routing for the various inputs and outputs via the checked cross points as shown previously. In addition, if you want to see what the routing for *TB* on *Input 1* is, simply click on the *TB* button adjacent to *Input 1* and this will turn yellow and show the routing as displayed in the following image.





The previous example showed *TB tx* routing for *Input 1*. You will note that when *TB* is activated it is being sent to *Encoder 2* and to *HP1R*. To switch between *TB Tx* and *Rx* settings in the profile simply click on the *Tx* and *RX* buttons as required, and as shown in the left-hand image.

The same principles apply for viewing the *Cue/Comms* sends on each codec input. As a default, Cue is always an outgoing send (except Remote Cue which displays Rx settings).

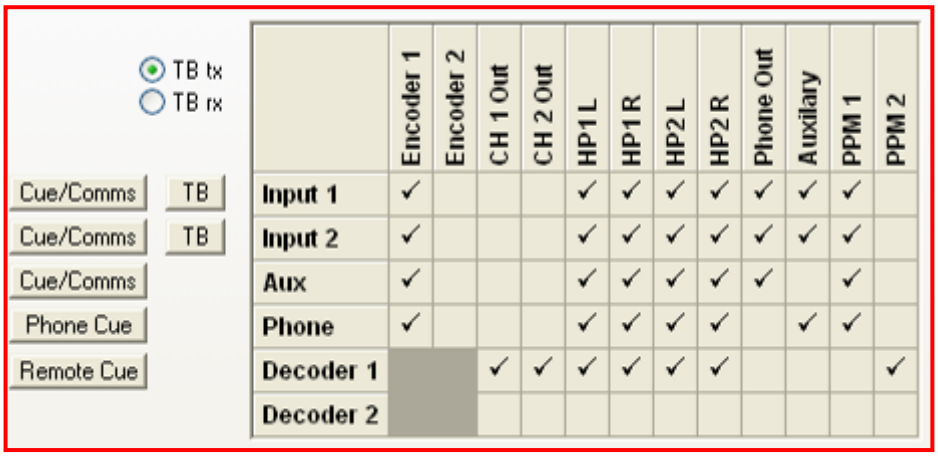
33.2. An Example: Integrated Communications with a Tieline Broadcast Codec

The following is an example of a typical ***COMMANDER G3*** setup at a sports event. The profile settings are very similar on other **Tieline** broadcast codecs. This configuration could be replicated on an ***i*-Mix *G3***. More inputs and expanded communications capabilities are available using the ***i*-Mix *G3***.

A typical announce booth team mixer configuration could consist of:

Input 1	Main play x play announcer
Input 2	Color announcer, statistics etc.
Aux	Mono FX source.

In the example below, using the *Man Dflt Mono/IFB* profile, announcer 1 and 2 in *Inputs 1 & 2* are identical. Both commentators hear *Inputs 1 & 2* and the *Aux* (FX) input in the left and right sides of their headphones. The commentators in *Inputs 1* and *2* hear the studio via *Decoder 1* in both sides of their headphones.



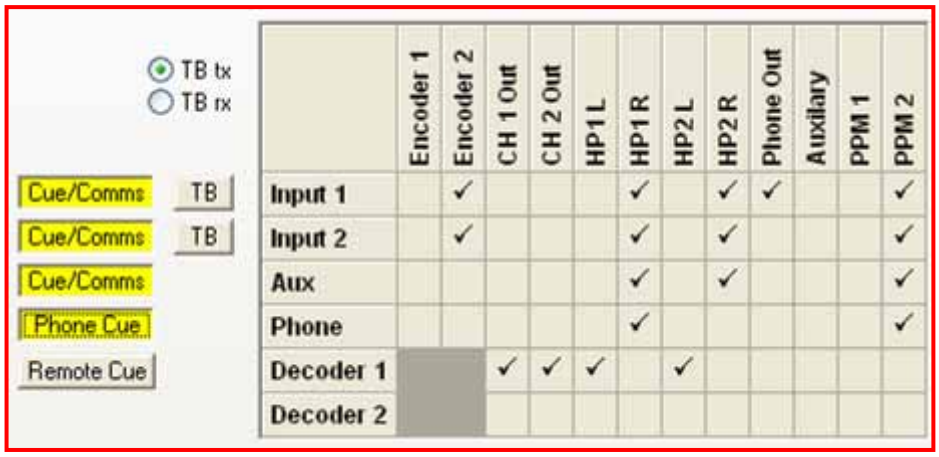
The commentators' audio and the *Aux* input FX are being sent via *Encoder 1.* The audio levels of the commentators and FX are set to be monitored on *PPM1* and incoming audio is being monitored via *PPM 2*. The incoming *Decoder 1*

audio is also able to be monitored via *HP 1 & 2* and is also being sent to *Ch1 Out* and *Ch 2 Out*.

The communications setup includes:

- Intercom (Comms) via *Cue/Comms* between commentators at the codec;
- A separate cue function for the commentators to listen to FX 'off-line' if required; and
- Comms via *Cue/Comms* to a producer/director at the other codec.

Following is an example of the ***View Matrix*** menu which shows routing for the functions described previously. The *Cue/Comms* and *Phone Cue* buttons for the four inputs are set to be monitored in the right side of the headphones only. The *Cue/Comms* button for the *Aux* input and the *Phone Cue* button for the *Phone* input are set for pre-fade listen off-line. The *Phone* input is set to be monitored only by the headphones in *Input 1* on the left side. Audio levels are also set to be monitored via *PPM 2*.



	Encoder 1	Encoder 2	CH 1 Out	CH 2 Out	HP1 L	HP1 R	HP2 L	HP2 R	Phone Out	Auxilary	PPM 1	PPM 2
Input 1		✓				✓		✓	✓			✓
Input 2		✓				✓		✓				✓
Aux						✓		✓				✓
Phone						✓						✓
Decoder 1			✓	✓	✓		✓					
Decoder 2												

Operational Tip: A SOFTKEY which is programmed to operate for talkback, cue or intercom purposes will operate even if a mic input's CHANNEL ON button is not switched on.

Section 34. Unit Options

Following are the options available in this menu within the ***Profile Editor***.

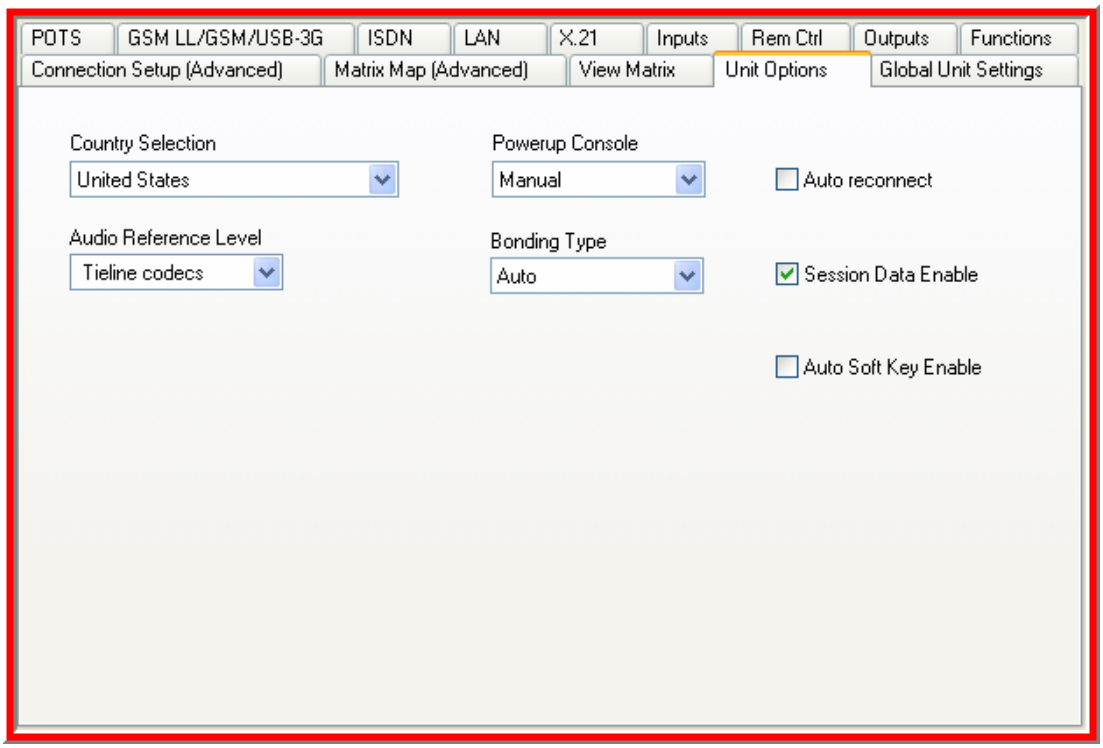


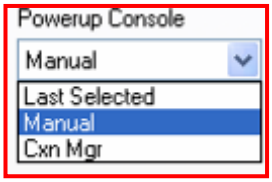
Figure 98: Unit Options Menu in ToolBox

34.1. Country Selection

Country Selection sets country specific telephone settings like ring cadence from the drop-down list of countries. The default country is the United States.

Important note:
Please ensure that the *Country Selection* in ***Unit Options*** is set correctly for your POTS and POTS G3 modules to ensure the correct settings are programmed into the codec before dialling connections.

34.2. Powerup Console



Powerup console relates only to the LCD menu screen on the codec. The setting you choose within this drop-down menu will decide which menu screen the codec will display on power up. It does not control which profiles are loaded into the codec prior to or subsequent to powering the unit down. *Manual* refers to the manual connection LCD, *Cxn Mgr* refers to the Connection Manager (*CXNS*) LCD and *Last Selected* refers to the last the last screen selected with console navigation, either *Cxn Mgr* or *CXNS*.

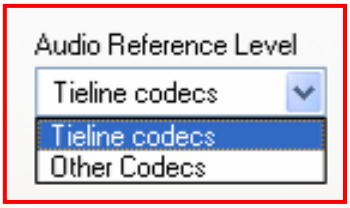
To set the *Powerup Profile* for the codec please see *Powerup Profile* within the ***Global Unit Settings*** menu in the ***Profile Editor***.

34.3. Auto reconnect

If the codec loses its connection it can automatically redial until the link is re-established (the codec will continue to redial until a connection is established) if you are using *Auto reconnect*. This can occur automatically even if power to the codec is lost. If the codec loses power and *Auto reconnect* is selected within the ***Profile Editor Unit Options*** menu, the codec will redial the connection/s as programmed under *Powerup Profile* in ***Global Unit Settings***.
On a pair of **Tieline** codecs, the codec that successfully initiated the original call will attempt to make the reconnection. Therefore *Auto reconnect* should be enabled on the dialing codec.

Please note: *Automatic redial* and *Failover profile* should not be programmed in codecs at the same time. This is because they will conflict with each other.

34.4. Audio Reference Level



This drop-down menu is used to fix audio level incompatibility between a **Tieline** codec and another brand of codec.

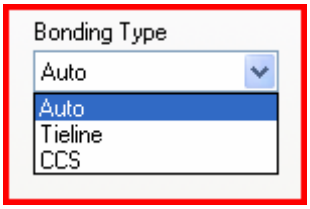
34.4.1. Tieline Codecs (-18dB Full Scale)

This is the default setting for **Tieline** codecs.

34.4.2. Other Codecs (-14dB Full Scale)

This setting solves audio level incompatibility between a **Tieline** codec and another brand of codec.

34.5. Bonding Type



This menu provides the option to select either *Tieline* or *CCS* bonding, or to have the codec automatically select the bonding type used. For more information about this menu, please see the section of this manual titled ISDN 3B and 4B Bonding.

34.6. Session Data Enable

When checked, session data is enabled within **Tieline** codecs. For more information on the attributes of session data please see the section of this manual titled Data Transfer & Using 3rd Party Devices.

34.7. Auto Softkey Enable

This setting is used to automatically enable the "Cue" button LCD display in TLF300 field codecs. Ordinarily this feature is enabled by pressing and holding down the codec MENU SELECTOR.

Section 35. Global Unit Settings

This tab on the *Profile Editor* menu contains relevant technical information about ISDN, the CPU, and the Digital Signal Processor (DSP). You will need this information when visiting our website for details and/or for software upgrades. This information can also be obtained from your codec by pressing **[Menu] > [Unit Details]** and then scrolling through the listed items. The main page for *Global Unit Settings* is shown below.

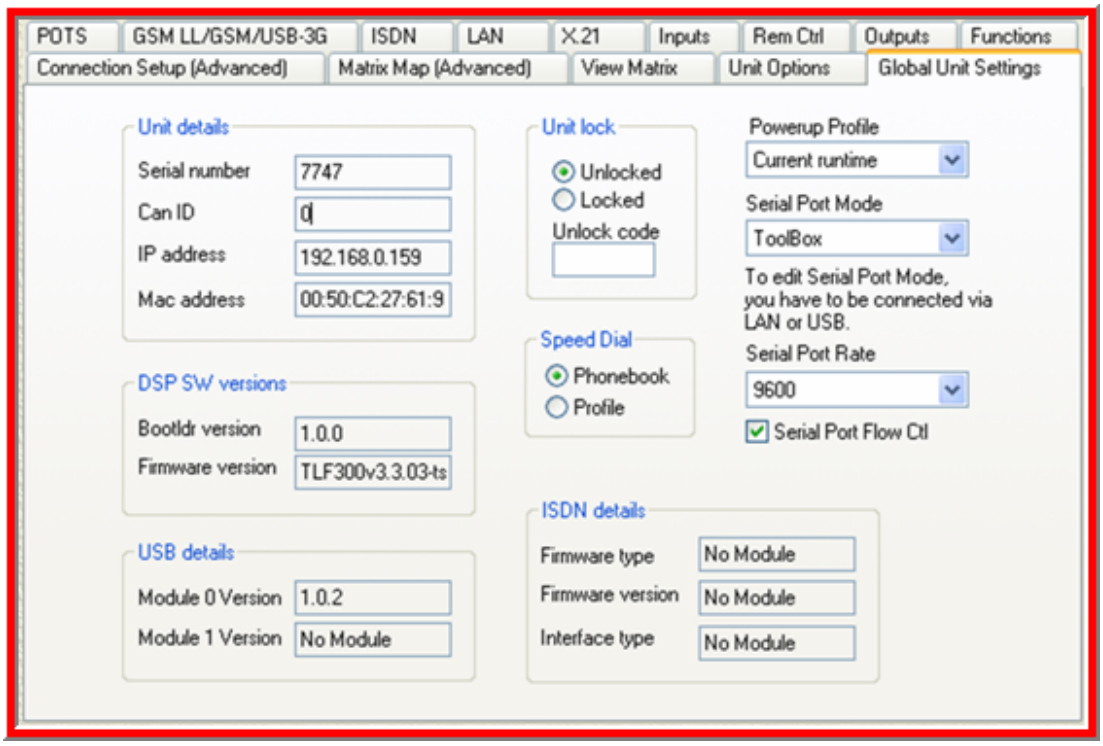
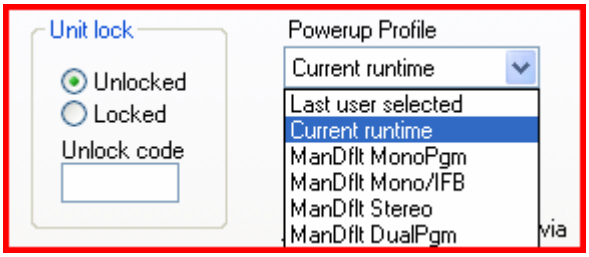


Figure 99: Unit Details Menu in ToolBox

The other menu items that are available include the following.

35.1. Unit Lock

Enables password security to lock the codec front panel. This is a useful safety feature if you would like to lock out users from changing settings on the codec without authorisation. Enter your personalised four digit number to lock and unlock the codec.



35.2. Speed Dial

Select between *Phonebook* or *Profile* dialing with the *Speed Dial* menu. See the How does Speed Dialing Work? section for more information about speed dialing.

35.3. Powerup Profile

Basically this is where you can set the profile that you would like loaded into the codec when it is initially powered up.

Last user selected will load the last profile that you loaded into the codec before it was powered down.

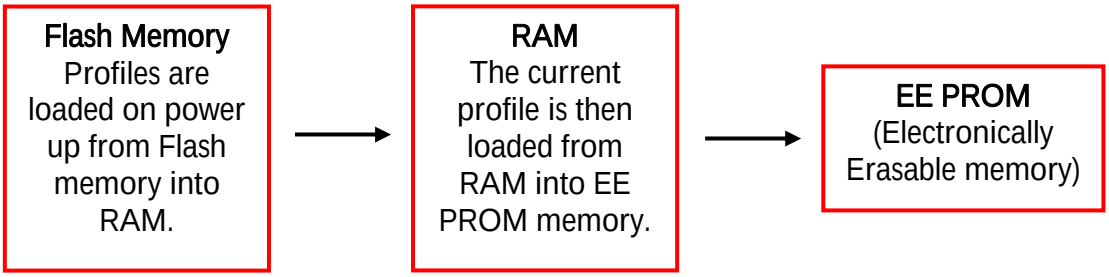
Current runtime is the profile settings that were current within the codec, including subsequent changes to a profile that have not been saved. These settings will be retrieved on power up. This setting should be used if you know you are going to make changes to an active profile 'on the run' and they are not likely to be saved. In case the unit suddenly loses power you will want these settings to be recovered.

The manual default profiles are also listed in this drop-down menu.

If you have *Auto reconnect* selected within the ***Unit Options*** menu in the ***Profile Editor***, the codec will redial automatically if it loses power or a connection is interrupted.

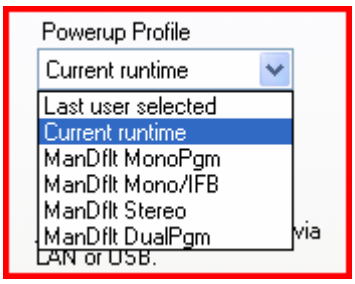
A Cautionary Note: This is not a backup profile option for the codec. *Failover profile* within ***Connection Setup*** will allow you to set a standby connection in case of a loss of a main codec connection. When using the *Failover profile* function the codec will detect a lost main connection and automatically switch to the selected *Standby* profile and attempt to connect with this. *Failover profile* is an additional option available from **Tieline**.

To assist in understanding codec memory principles, following is a diagram showing the relationships between the various types of memory within **Tieline** codecs.



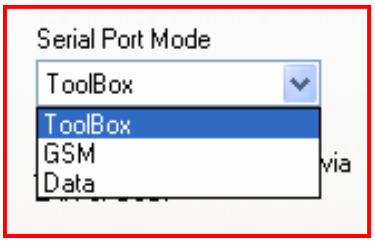
35.4. Powerup Profile: Manual Default Profile Settings

Several manual default profiles have been programmed into all codecs. These profiles are listed in the ***Global Settings*** menu under *Powerup Profile*. To view the matrices for these profiles look in the ***Matrix Editor*** or once you have loaded a profile view the settings in the ***View Matrix*** section of the ***Profile Editor***.



If you have one of the manual default profiles listed as the *Powerup Profile* then it will be the default profile when you power up your codec. If you have loaded a manual default profile into your codec and made changes to it, you may find it useful to select *Current runtime* as your *Powerup Profile* as this will power the unit up with the profile and any changes you have subsequently made to it. (Related Topics: Manual Default Profiles and Selecting Manual Default Profiles)

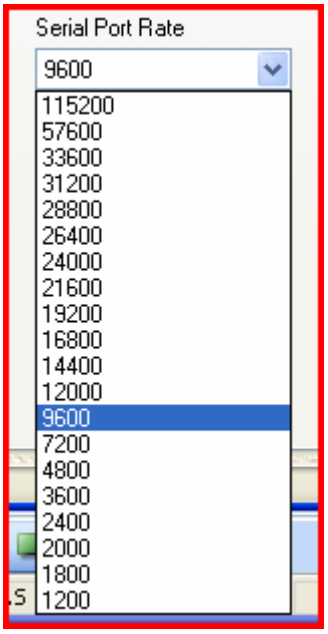
35.5. Serial Port Mode



This setting allows you to set the mode that you will be using the RS232 serial port in. Select the option you want from the drop-down menu. This menu is only available when you are connected to the codec via a LAN or USB connection. The modes available include: *ToolBox* (for connecting to a PC with Tool**Box** software installed); *GSM* (for connecting to a GSM cell-phone); *Data* (for sending data via the serial port).

This can also be configured via the codec menu. To do this select **[Menu] > [Configuration] > [Advanced] > [Serial Port Mode]** and then select the option you require. (Related topic: Configuring GSM Connections via the RS232 Serial Port).

35.6. Serial Port Rate



This sets the baud rate of data streaming from the serial port. Use the drop-down menu to select the data rate required. Ensure you select the rate that is compatible with the device you are connecting to the serial port. (Related topic: Configuring GSM Connections via the RS232 Serial Port).

35.7. Serial Port Flow Control

If enabled, *Serial Port Flow Control* allows a device connected to the codec to regulate the flow of data. Check the box to enable this function. Configuring GSM Connections via the RS232 Serial Port).

Section 36. Phonebook Editor

With the profile system, a phonebook is not entirely necessary. However, it can be convenient for situations where users do not want to use the Connection Manager (CXNS), and prefer to select the dialing numbers from the phonebook manually. Users can set up a default profile and dial the numbers from the *Phonebook Editor* if they wish to. This is done using the *Connection Setup* page within the *Profile Editor*. This process is explained in the Phonebook Numbers section of this manual.

Phone numbers can be stored within ToolBox using the *Phonebook Editor* which is displayed below and accessed via the ToolBox root menu. Operators can then set up default profiles and dial a remote codec by retrieving stored numbers from the *Phonebook Editor*.

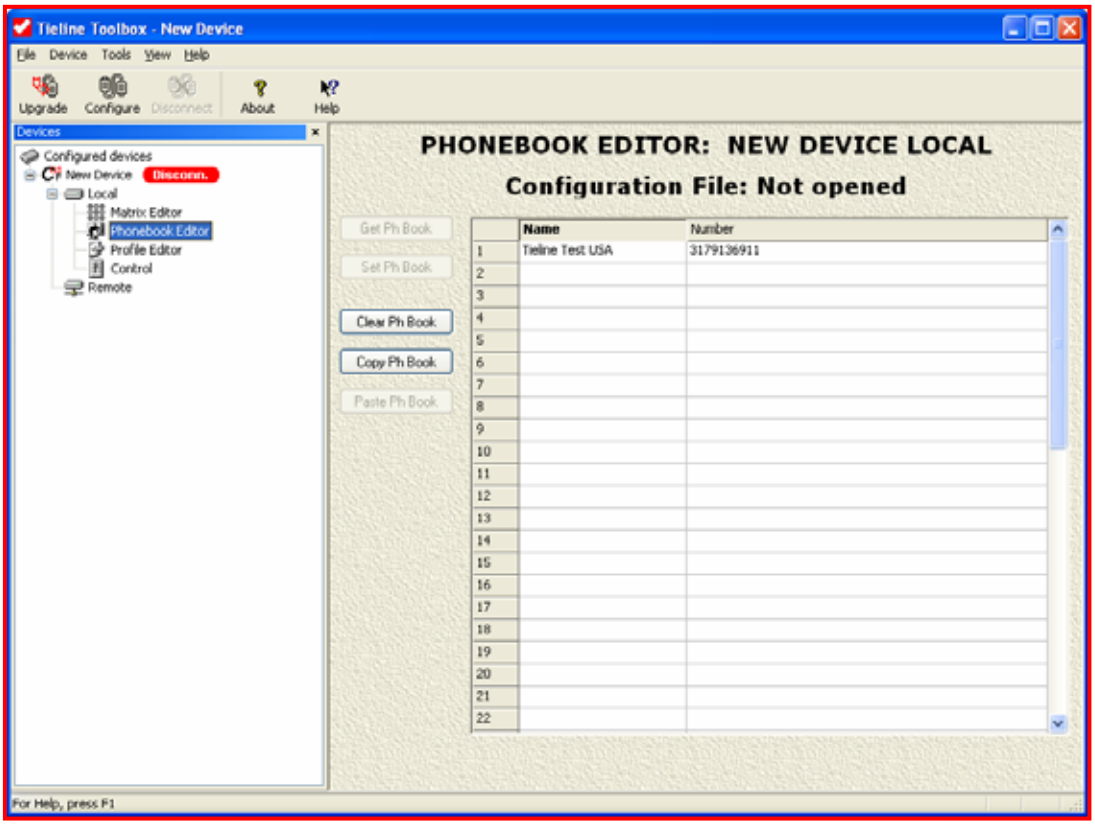


Figure 100: The Phonebook Editor in ToolBox

Using **Tieline** codecs is now easier than ever before with the ability to speed dial locations using phonebook entries linked to specific profiles. Simply program the number to be dialed into the *Phonebook Editor* for your codec, associate the profile you require (either a manual default profile or a custom profile) with this number and you are ready to go.

Speed dialing is particularly useful as it means operators can program speed dial numbers for many different locations and associate a profile for each particular broadcast. An announcer with no knowledge of how to operate a codec can load a profile and dial the studio by simply recalling a stored memory number and pressing ENTER/DIAL on the codec.

In addition, it is now possible to enter 80 phonebook entries into your codec - this has been increased from 50 previously.

36.1. Name and Number

Up to 80 phone numbers and names can be stored in the configuration file phonebook. Each number can be up to 27 digits, including commas, and the name can be up to 16 characters long. A phonebook can be copied from one Configuration File to another, enabling users to maintain a consistency between books.

To enter a name and a number simply click on the *Name* button for a particular entry number, in the above example entry *1*, and type in the name. Then do the same for the *Number* button. The number must be entered without brackets, dashes or spaces. Numbers 0-9 only. The number for the **Tieline** test line in the US is (317) 913 6911 and this would be entered as 3179136911 as shown above.

36.2. International Prefixes and Dialing Out of a PBX

If you are calling long distance an area code will need to be inserted. In addition, if you are dialing internationally, all countries have different international dialing codes as well as area codes.

Example:
The phone number for **Tieline** in the US is (317) 913 6911
If you were dialing from within the 317 area you would dial 9136911
If you are dialing from a PBX within the 317 area you would dial 0,9136911
If dialing from outside the 317 area you would dial 1,3179136911
Dialing from overseas 011,1,3179136911
And if you were using a PBX overseas you would dial 0,011,1,3179136911

Please consult your phone company to confirm what prefix(s) needs to be used. Some phone companies use different prefixes for data and voice traffic. It is preferable to use the data lines where possible.

36.3. How Does Speed Dialing Work?

Speed dialing works in two ways. If you simply wish to dial phonebook entries via speed dial you can program the codec to dial the phonebook number only. Alternatively, if you want to speed dial a phonebook entry and have the profile change to the one associated with that entry, select this option in the Speed Dial Memory menu.

36.3.1. Programming a Codec to Speed Dial Phonebook Numbers Only

To program phonebook speed dialing press SOFTKEY 4 Menu and select **[Configuration] > [System Settings] > [Speed Dial Memory] > [Phonebook]**. Press CLEAR several times to return to the main menu screen.

Ensure that the **[ ]** on the main codec screen are surrounding the connection (e.g. POTS, ISDN, GSM, IP, 3G/IP or X.21) that you wish to connect with. Now press the MEMORY button on the keypad and enter the number of the phonebook entry to recall the stored phonebook number. Press ENTER/DIAL on the codec to dial this number.

36.3.2. Programming a Codec to Speed Dial and Change a Profile Simultaneously

To program phonebook and profile change speed dialing press SOFTKEY 4 Menu and select **[Configuration] > [System Settings] > [Speed Dial Memory] > [Profile].** Press CLEAR several times to return to the main menu screen.

Ensure that the **[ ]** on the main codec screen are surrounding the connection (e.g. POTS, ISDN, GSM, IP, 3G/IP or X.21) that you wish to connect with. Now press the MEMORY button on the keypad and the following screen will appear.

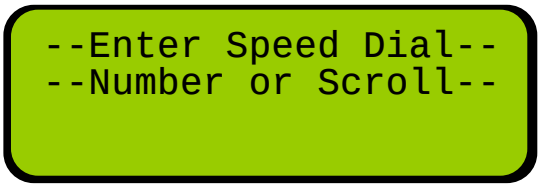


Figure 101: Speed Dial Profile and Number Screen

If you know the number of the entry you require simply use the keypad to recall the profile and dialing settings of that entry. If you would like to scroll through the available entries, simply use the MENU SELECTOR to scroll through the entries and select the one you require.

If your codec is already connected the following screen will appear on the codec.

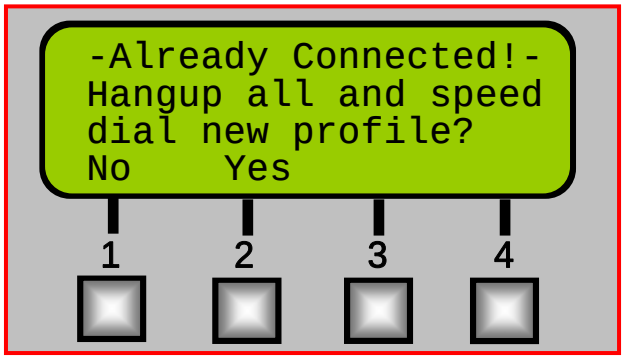


Figure 102: Hangup & Profile Change Confirmation Message

Select Yes if you wish to continue and change the profile or select No if you wish to exit the menu. If you select Yes by pressing SOFTKEY 2, the codec will disconnect any current calls, load the new profile settings and all you

need to do is press ENTER/DIAL on the codec to dial and connect with the new profile's settings.

Important note: We highly recommend that you create your own custom profiles for speed dial purposes, particularly if you require many profiles for different situations.

It is also possible to use speed dial with manual default profiles but if you do this we recommend you do not enter the number into the *Connection Setup* page of the *Profile Editor.* If you do, this will limit the scope of use for these profiles. This is **not recommended**. Instead, we recommend that you program the dialing number into the *Phonebook Editor* and select the manual default profile next to the number. An example of how to program manual default profiles in this way is displayed in the next section.

Alternatively, if you wish to use the settings of a particular manual default profile and save it as a new profile, simply save the profile's settings with its own unique name. This is simple to do and is fully explained in the section of each codec's main manual titled "Creating a New Profile".

36.4. Programming Profiles Using ToolBox

Following is an image of the *Phonebook Editor* in Tool**Box** software. This software can be used to program numbers and profiles into your codec easily.

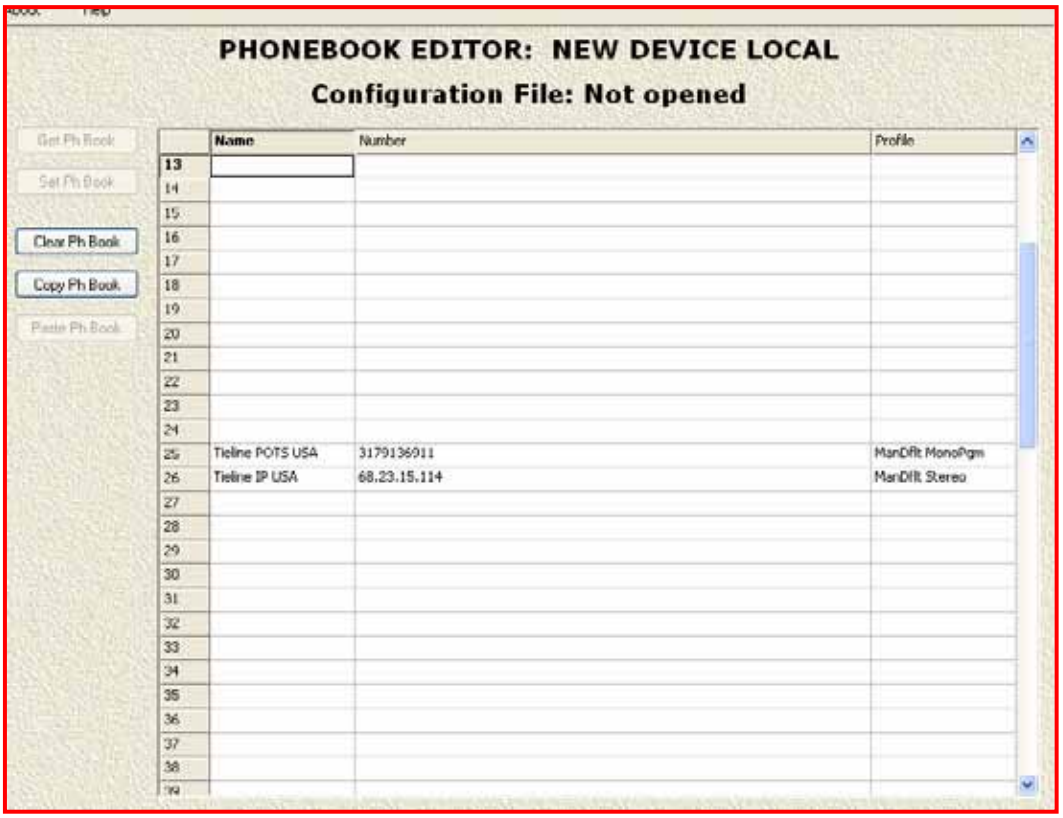
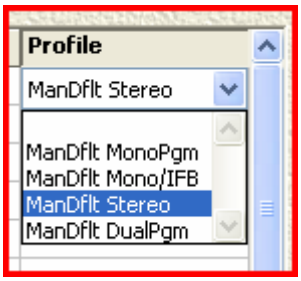


Figure 103: Phonebook Editor in ToolBox

The principles for programming the profile and associating it to a particular number for dialing are similar to selecting matrices in the ***Matrix Map*** section of the ***Profile Editor.*** First, click on the right side of the *Profile* column in the ***Phonebook Editor*** (please ensure you click on the right side of the section next to the number you have just entered and that you wish to program the profile for).

An arrow for a drop-down menu will then appear. Click on this arrow to reveal a drop-down menu for selecting the profile you wish to associate with the speed dial number. All manual default and custom profiles will be listed in this menu.

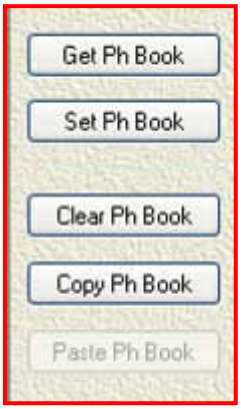


36.4.1. Loading the Phonebook Editor Settings into a Codec

Once you have completed this process, please see the section in each codec's main reference manual titled 'Set, Get, Save and Open Functions of Configuration Files' if you need more information on loading profiles onto a codec.

If you wish to copy the entries from one codec's phonebook to another codec's phonebook, please read the following section.

36.5. Storing, Saving and Copying Phonebook Data



The phonebook is stored within a codec in EE memory. EE memory is non volatile. A copy of the phone book can also be backed up onto disc with a PC.

Get Ph Book: Loads the phone book from the codec into a PC.

Set Ph Book: Loads the phone book from a PC into the EE ram within the codec.

Clear Ph Book: Clears the contents of the phone book, but does not affect the contents of the phone book in memory or on file.

Copy Ph Book: Saves the phone book into a file stored on a PC.

Paste Ph Book: Loads the phone book from a PC file into the ToolBox *Phonebook Editor* screen, for inspection and editing.

Phonebook file format: Phonebook files are stored as a Codec Phonebook (*.cpb) File.

To save the entries that you put into the *Phonebook Editor* you will need to save the entries to a profile within a Configuration File. To understand how to save a profile in a Configuration File, please go to the section titled 'Set, Get, Save and Open Configuration Files' in this manual.

Very Important Note:

You can only copy entries from one ***i*-Mix *G3*** listed in your *Devices* root menu in ToolBox, into the *Phonebook Editor* of a separate ***i*-Mix *G3*** codec listed in the same *Devices* root menu

In addition, you can only copy entries from one ***COMMANDER G3*** field unit codec listed in your *Devices* root menu in ToolBox, into the *Phonebook Editor* of a separate ***COMMANDER G3*** field unit codec listed in the same *Devices* root menu, or if you are copying from a ***COMMANDER G3*** rack unit codec then you must copy to another ***COMMANDER G3*** rack unit (please ensure it is the same model of rack unit codec because the TLR300 and TLR300B rack units have different matrices).

This is because the Configuration Files for different models of codecs are different to each other and cannot be loaded onto a different model of codec.

Following is the procedure to copy entries from those listed in the *Phonebook Editor* and to paste them into another profile.

- First, go to the profile you wish to copy entries from and open it within ToolBox ;
- Next go to the *Phonebook Editor* where the entries are listed and click on *Copy Ph Book* from the options available;
- Then add a new device to the *Devices* root menu (Please ensure that the device that you are copying from is the same as the one you are copying to, i.e. field unit to field unit, or rack unit to rack unit).
- Go to the *Phonebook Editor* within the new device you have just created and select the *Paste Ph Book* option to paste the entries into the new phone book. If you wish to save these entries you must save your profile settings to a Configuration File.

Very Important Note:

You can only copy entries from one ***i*-Mix *G3*** listed in your ***Devices*** root menu in ToolBox, into the ***Phonebook Editor*** of a separate ***i*-Mix *G3*** codec listed in the same ***Devices*** root menu

In addition, you can only copy entries from one ***COMMANDER G3*** field unit codec listed in your ***Devices*** root menu in ToolBox, into the ***Phonebook Editor*** of a separate ***COMMANDER G3*** field unit codec listed in the same ***Devices*** root menu, or if you are copying from a ***COMMANDER G3*** rack unit codec then you must copy to another ***COMMANDER G3*** rack unit (please ensure it is the same model of rack unit codec because the TLR300 and TLR300B have different matrices).

This is because the Configuration Files for different models of codecs are different to each other and cannot be loaded onto a different model of codec.

Following is the procedure to copy entries from those listed in the ***Phonebook Editor*** and to paste them into another profile.

- First, go to the profile you wish to copy entries from and open it within ToolBox ;
- Next go to the ***Phonebook Editor*** where the entries are listed and click on *Copy Ph Book* from the options available;
- Then add a new device to the ***Devices*** root menu (Please ensure that the device that you are copying from is the same as the one you are copying to, i.e. field unit to field unit, or rack unit to rack unit).
- Go to the ***Phonebook Editor*** within the new device you have just created and select the *Paste Ph Book* option to paste the entries into the new phone book. If you wish to save these entries you must save your profile settings to a Configuration File.

Section 37. Troubleshooting Tips

Following are some troubleshooting tips for getting connected using **Tieline** codecs.

37.1. POTS Troubleshooting

1. To find out if you are about to connect a **Tieline** codec to a digital line, use a normal analog phone to check the line first.

2. It can be a good idea to take your own phone when you are doing a remote broadcast. If possible, dial the number you are going to use and check for any unusual noises. These may be due to other devices connected to the line.

3. Some facilities, especially large hotels, are able to provide phone connection points to allow a dial-up connection for laptop Computers. Laptop Computers use technology similar to POTS codecs, so a **Tieline** codec will invariably work if the computer connection speed is 24,000 bps or higher.

4. Take an ADSL/DSL filter to all remote locations. ADSL/DSL modems can generate noise on a line which will degrade the performance of your codec. Due to the increasing popularity of ADSL/DSL services on phone lines, this is likely to be an increasing problem, but one that is simple to solve. Simply place the ADSL/DSL filter between the POTS line and your codec to remove the interference.

5. **Tieline** USA has a test unit on 317 913 6911 to facilitate testing. A **Tieline** ***G3*** codec will automatically negotiate the optimal connect rate for each individual circuit on connection

Important note on operating in Leased Line mode:

When leased line mode is enabled the dial tone and progress tones are disabled. In this mode is necessary to manually answer the call at the remote codec initially. If you want the remote codec to automatically re-answer after losing a connection, you should enable Auto Reconnect on both the remote and local codecs. Then if you lose a connection for some reason, after you have already made a successful call, the codec should auto-reconnect.

To enable Auto Reconnect using the codec menus, press **Softkey 4** to access the **[Menu]** screen, then scroll using the MS to **[Auto Reconnect]** and select this option. Then scroll using the MS to the **[Enable]** and select this. The Auto Reconnect function will now be enabled.

If Auto reconnect is enabled in the codec menu or via ToolBox software, it will attempt to redial indefinitely in the manual default profiles. If you wish to adjust the redial setting in the codec to only dial a certain amount of times, you will need to use the *Automatic Redial* function in ToolBox software. Do not enable both the *Automatic redial* and the *Auto reconnect* functions at the same time as they will conflict with each other. For more information please see the sections on *Automatic Redial* and *Auto reconnect* in this manual.

37.2. ISDN Troubleshooting (See the ISDN section for more details about connecting over ISDN)

1. First, if you have an ISDN line connected to your ISDN module, ensure that you have "SYNC" displayed on the LCD next to the line you wish to connect with – as displayed in the following image.

```
ManDflt MonoPgm
⌈ISDN1> Enter# Sync⌉
 ISDN2> Enter# Sync
Aud              Menu
```

   If not ensure you have an ISDN service available.

2. Following is a checklist for you to go through in order to check the ***ISDN*** setup menu items one by one. Double check that all the information has been entered exactly as provided by the phone company and matches to the ISDN setup guide above. Here is the list:

Setup Item	Your Entered value	Correct?
ISDN AutoAnswer		
ISDN Network		
ISDN Line Type		
ISDN Interface		
ISDN 56K Dial		
Local SubAddress		
SPID1		
DN1/MSN1		
SPID2		
DN2/MSN2		

Table 21: ISDN Connection Checklist

3. Unplug the ISDN line and leave it unplugged. Turn the codec off, pause and turn it back on. Wait about a minute, or until you see ISDN No Line on the LCD codec screen. Plug in the ISDN line again and wait for ISDN Line Ready.

4. You can sometimes get a line working without the proper SPIDs by dialing into it. It is a temporary fix and the line will fail next time the codec is booted.

5. If your ISDN line has two B channels, try to ring each codec from itself. It must get connected instantly and remain connected for a few seconds. This way you can see which side the problem is from.

37.3. GSM Troubleshooting

VERY IMPORTANT WARNING:

GSM plug-in modules cannot be used in TLR300B ***COMMANDER G3*** rack unit codecs. If you must connect using GSM with aTLR300B ***COMMANDER G3*** rack unit codec, we suggest you use a GSM cell-phone attached to the RS232 port on the rear panel of the codec. Alternatively, you could try a GSM phone or a 3G cell-phone in GSM mode with a **Tieline** USB module.

Following are some troubleshooting tips if you are having trouble with your GSM connection:

1. Use the balanced inputs and outputs where possible to minimize the likelihood of cell-phone interference.

2. Attach the male to female antenna adaptor to your GSM module. This adapter is provided with the antenna kit when you purchase a GSM module from **Tieline**. If you don't use the adapter, connections could be unreliable and audio problems may be introduced.

3. If you are dialing from the landline connected codec, check that you are dialing the special GSM data number and not the standard GSM voice number. You may have to purchase this data number from your telco. Most telco's have a data number that is separate to the standard voice number. You can get the data number by calling your telco and requesting it.

4. If you are having problems with dialing from the landline side, but you are successful dialing from your cell-phone to the landline connected codec, it could be that your GSM data number has been set up in 'Transparent' mode. Ring up your telco and change over to 'Non-transparent' mode (Related topic: GSM Transparent and Non-transparent Data Modes).

5. If you encounter audio drop-outs try increasing the jitter-buffer second setting. If you adjust this while you are connected, the settings will only take effect when you next try to connect. In other words, you must hang up and redial the connection.

37.4. 3G Troubleshooting

Consult the "IP and 3G Streaming Reference Manual" for detailed troubleshooting and connection information.

37.5. IP Troubleshooting

Please check our website regularly for updates to the IP knowledge base at www.tieline.com/IP . Consult the "IP and 3G Streaming Reference Manual" for detailed troubleshooting and connection information.

37.6. X.21 Troubleshooting

If you are not able to connect successfully please check the following:

1. Check that you have configured the codec for *X.21 < > ISDN* use and not *X21< > X21*[27] use.

2. 64 – 2,048 kbps bit rates are currently supported by **Tieline** codecs for X.21 connections. Ensure that the bit rate setting in your codec is matched to that of your ISDN TA/X.21 DCE device. If you are unsure of the bit rate of your ISDN TA/X.21 DCE device, please check its user manual to discover the correct setting.

For more information on how to connect via X.21 using **Tieline** codecs, please see the Quick Start or X.21 Connections sections of this reference manual or contact customer support at support@tieline.com for more info.

[27] Please contact **Tieline** via support@tieline.com for details of the implementation date for *X21 < > X21* connections.

Section 38. Glossary

Basic Rate Interface (BRI) The Basic Rate Interface (BRI) of ISDN consists of 2 bearer (B) channels at 64 Kbps each and 1 delta (D) channel at 16 Kbps, i.e. (2B +D). The total raw data rate is 144Kbps on a single twisted pair. This can be provided over a 2 wire facility and the two B channels can be bonded together to form a single 128Kbps channel.

Central Office (CO) One of a local telephone company's offices. Circuits are run from here to support the surrounding area and ISDN circuits can extend up to 3.42 miles (5.5 km) from the CO.

Codec A codec (coder/decoder) is an integrated circuit device that encodes an analog signal into a digital bit stream and then decodes this bit stream into an analog signal.

CSD Circuit Switched Data (CSD) is provided as part of the GSM system. The basic data rate is 9,600bps. (see also WAP and HSCSD)

Customer Equipment (CE) Telecommunications equipment used by an organization and connected to an ISDN line.

B (Bearer) Channel A 64Kbps ISDN channel that can be used for carrying voice, video or data. The channels are known as B1 and B2.

BOOTP Bootstrap protocol is used by a network node when determining the IP address of its Ethernet interfaces.

D (Delta) Channel The channel carrying all signaling between the network and the customer's equipment when using ISDN channels.

DHCP This is a method of dynamically allocating IP addresses so that the addresses are able to be reused when hosts no longer require them.

Directory Number A Directory Number is the ISDN number assigned to each B channel of an ISDN connection.

EDGE Further enhancements in data capability over the core GSM network will be provided with the introduction of Enhanced Data rates for GSM Evolution - known as EDGE*. This will achieve the delivery of advanced mobile services such as the downloading of video and music clips, full multimedia messaging, high-speed color Internet access and e-mail on the move. (Information courtesy of GSMworld.com)

FDD	Frequency Division Duplex operates when a cell-phone connection uses and up link and a down link that use different frequencies.
GSM	(Global System for Mobile communications) Today's second-generation GSM networks deliver high quality and secure mobile voice and data services (such as SMS/Text Messaging) with full roaming capabilities across the world. (Information courtesy of GSMworld.com)
3GSM (3G)	3GSM* is the latest addition to the GSM family. 3GSM* is about having third generation mobile multimedia services available globally. (Information courtesy of GSMworld.com)
GPRS	General Packet Radio Service (GPRS) enabled networks offer 'always-on', higher capacity, Internet-based content and packet-based data services. (Information courtesy of GSMworld.com)
GUI	The term General User Interface is described in this manual as a device for externally connecting to the ***COMMANDER G3*** rack unit to control the 1RU codec.
HSCSD	High Speed Circuit Switched Data utilises circuit switched channels to achieve data rates up to 43,200bps. (See also WAP and CSD)
Internet Protocol (IP)	This is an Internet protocol containing addressing information and some control information which enables packets of data to be routed. IPv4 is in common use today and IPv6 is the emerging standard which will enable greater volumes of IP addresses.
ISDN	The acronym ISDN stands for Integrated Services Digital Network.
Local Area Network (LAN)	This is a high-speed data network with low errors which covers a fairly small geographic area. These networks connect workstations, terminals, peripherals and other devices in a building or similarly limited geographical area.
Local Exchange	Provides ISDN services - the Network side of the ISDN loop.
Local Exchange Carrier	A local phone company such as regional Bell operating companies.

MPEG	MPEG is the international standard for multimedia which incorporates both audio and video encoding at a range of data rates.
MP3	MPEG-1 Layer 3 is commonly used to compress sound files to around one twelfth of the original size of the file.
O.B.	An Outside Broadcast or Remote Broadcast site external to the studio.
Point-to-Multipoint	ISDN supports point-to-multipoint configurations, with multiple but dissimilar CE devices connected to a single line.
Point-to-Point	When a single ISDN device is connected to an ISDN line i.e. one codec connected to another codec.
Remote	A Remote Broadcast or Outside Broadcast site external to the studio.
Service Profile ID	Each telephone line is assigned a single directory number. When a customer orders services or features they are programmed into a Service Profile stored at the switch. Any call associated with that line can be assumed to apply only to that customer's Directory Number and Service Profile. When a single ISDN device is connected and serviced this way, the configuration is said to be point-to-point.
Stub or Party Lines	In some countries, it was the practice to have more than one phone service attached to one line. This is called a Stub or Party Line. As more lines have been installed, services have been separated but the redundant cabling may remain connected across your line and it will cause problems with operation of your codec.
S/T Interface	A Network Terminator 1 (NT-1) converts a two-wire U interface into four-wire S/T interface which will allow the use of multiple devices when using ISDN. (Common outside North America)
TCP	A transport layer protocol which sends reliable in-order data between hosts.
TDD	A technique used by cell-phone networks where the up link and the down link use the same frequencies for transmission.
UDP	A transport layer protocol which sends data between two hosts which can be received out-of-order.

U – Interface	When using ISDN in the U.S.A., a phone company will provide its BRI (Basic Rate Interface) customers with a U interface. It is a two-wire interface from the phone switch that supports full-duplex data transfer. Only a single device can be connected to this interface. (Common in the U.S.A.)
WAP	Wireless Application Protocol (WAP) is the leading standard for information services on wireless terminals such as digital mobile phones (see also CSD and HSCSD).

Section 39. Portable Powering Solutions

39.1. Tieline Battery Module

The **Tieline** battery module is the ideal wireless powering solution if you on the road and need to power your codec in remote locations. Simply fit the battery module into the codec and attach the 4-pin female XLR connector into the power input on a **Tieline** codec. This battery will power your codec for up to 2.5 hours if it is fully charged.

Figure 104: The Tieline Battery Module

The battery is rechargeable and is charged by **Tieline**'s 12 volt 'mouse charger'. This charger is designed to be charged via a 12 volt in-vehicle cigarette lighter. Do not connect to a voltage source greater than 12 volts. The cable from the cigarette charger requires 12 volts and the output of this cable is 24 volts into the mouse charger itself.

The mouse charger can also be used with a 110 volt or 240 volt mains adapter, depending on the voltage requirements in your country. Please specify your requirements when you order your mains adapter.

Figure 105: The Tieline Mouse Charger and Cigarette Lighter Cable

When you connect the cable from the cigarette lighter to the mouse charger, the red LED on the charger should illuminate to notify you that a voltage source is connected to it. When you connect the **Tieline** battery module to the charger the other LED should illuminate red indicating that the battery is charging. The battery should take between four and five hours to fully charge and the red charging LED will turn green when it is charged. When this LED turns green the battery is approximately 85% charged. To fully charge the battery it is a good idea to leave the charger connected for about another hour after the green LED appears.

WARNINGS ON BATTERY USE AND CHARGING

1. Do not wire the charger directly to the battery.
2. Do not connect your mouse charger to a voltage source greater than 12 volts. **24 volt sources are not acceptable**. Correct connection is the responsibility of the user.
3. Do not charge the battery module for more that 12 hours.

39.2. The 12 volt Vehicle Power Supply Cable

Tieline can also supply a tailor-made 12 volt cable for connecting your codec directly to your 12 volt in-vehicle cigarette lighter. This is the perfect solution when you are broadcasting with your codec for long periods in remote locations.

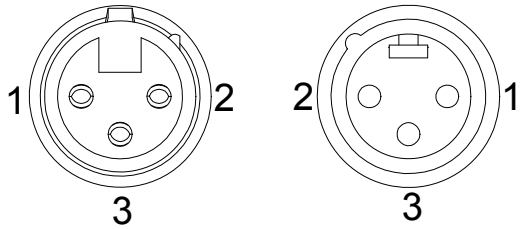
When you connect to a vehicle battery please ensure that the voltage source is 12 volts. Connecting in this manner will provide up to 10 hours of powering to the codec from a fully charged 12 volt battery in good condition.

WARNING ON USE OF THE 12 VOLT VEHICLE POWER SUPPLY CABLE

1. Do not wire the codec directly to the battery.
2. Only connect your codec to a 12 volt cigarette lighter socket in your vehicle.
3. We **do not recommend** that you connect and broadcast while the engine of the vehicle is running. Connecting with the engine of the vehicle running may interfere with the performance of your codec or your mobile connection.
4. We take no responsibility for unforeseen damage to a vehicle's battery.

Appendix 1. Connector Wiring

XLR 3 Pin Audio Connectors
Connectors viewed from wiring side



Female Input

Male Output

XLR Male Output Connectors.
Never connect pins 2 or 3 to each other, or to, pin 1, or earth, chassis.
This will result in damage to the audio output IC of the codec and the warranty being voided.

Pin	Balanced Mic & Line	Input to the codec from equipment RCA Connectors	Output from the codec to equipment with RCA Connectors
1	Ground	See note below	Ground
2	+ve audio	audio	Signal
3	-ve audio	ground	Not used.

Appendix 1.1. Interconnection of RCA and XLR Connectors

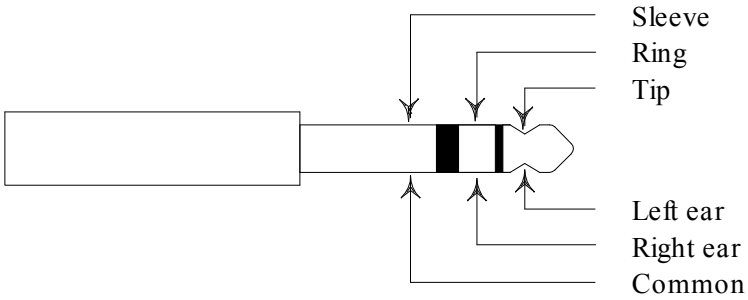
XLR Female Input Connectors:

In some unbalanced installations pins 1 & 3 may need to be joined. There is no rule set in stone that is guaranteed to work in all circumstances. It is very much a case by case scenario depending on, amongst other things:

- Is the circuit being driven by a transformer? If so is one side of the transformer is earthed, or a center tap earthed?
- Is the driving equipment “semi-professional”, i.e. it is not truly balanced having one side of the audio tied to O volts (ac ground) and the other side active?
- Is the equipment wired to “rock and roll” standard with pins 2 & 3 reversed, whereupon the preceding goes out the window... and the list goes on.

The entire implementation of the balanced to unbalanced is the epitome of the time worn don’t fix it if is not broken rule. In other words, if it works and doesn’t hum it must be wired satisfactorily.

Appendix 1.2. RTS Headphone Connector



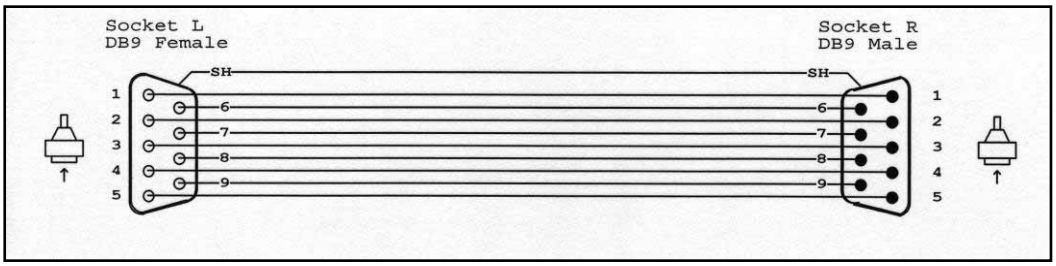
Appendix 1.3. Codec Connection Cable Configurations

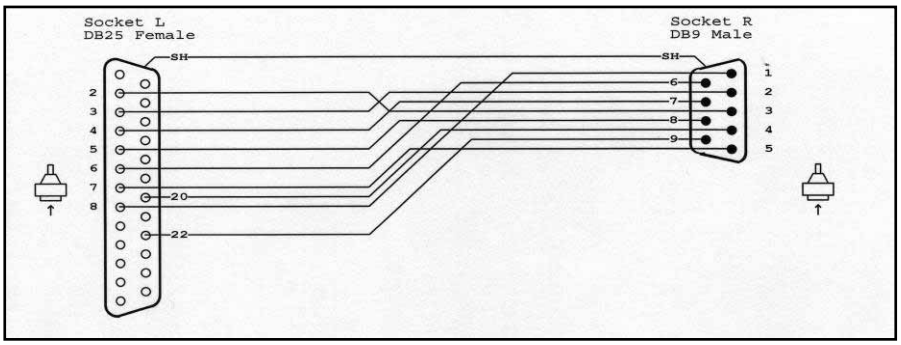
A cable can be constructed using the following components and connections. A normal D9 to RS 232 modem cable will work well. A null modem cable will not work at all. Following are the pin and signal configurations for these connectors.

	Pin Configurations		
Signal Description	Codec DB 9 male	PC DB 9 female DCE	PC DB 25 female DCE
TxD	Pin 3	Pin 2	Pin 2
RxD	Pin 2	Pin 3	Pin 3
GnD	Pin 5	Pin 5	Pin 7

For example, if you are connecting your codec using the codec's DB 9 male to the PC DB 9 female, you would connect Pin 2 RxD on the male connector to Pin 2 TxD on the female connector.

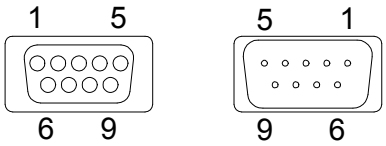
Following are diagrams of the pin configurations of the DB9 and DB25 connectors that connect to a codec.





Appendix 1.4. D9 (RS 232) Data and Interface Connectors

Viewed from wiring side



Female Male

Pin	INTERFACE Female DCE	DATA Male DTE
1	No Connection	No connection
2	TX Data	RX Data
3	RX Data	TX Data
4	No connection	No connection
5	Signal Ground	Signal Ground
6	No Connection	No connection
7	CTS	RTS
8	RTS	CTS
9	No connection	No connection

Table 22: D9 Data and Interface Connector

Appendix 1.5. D15 X.21 Interface Connector

The pin-outs for D-15 X.21 module connections are displayed in the following image. A straight-through cable is required from the male D15 connector on the X.21 module to a female DB15 connector, which will attach to an ISDN TA/X.21 DCE Device.

Pin	Signal	DTE	DCE
1	Shield	-	-
2	Transmit (A)	Out	In
3	Control (A)	Out	In
4	Receive (A)	In	Out
5	Indication (A)	In	Out
6	Signal Timing (A)	In	Out
7	Unassigned		
8	Ground	-	-
9	Transmit (B)	Out	In
10	Control (B)	Out	In
11	Receive (B)	In	Out
12	Indication (B)	In	Out
13	Signal Timing (B)	In	Out
14	Unassigned		

Table 23: X.21 Pin-outs for D-15 Connectors

With this pin configuration all signals are balanced. The DCE (Typically ISDN TA) provides Signal Timing, ensuring X.21 data is synchronous.

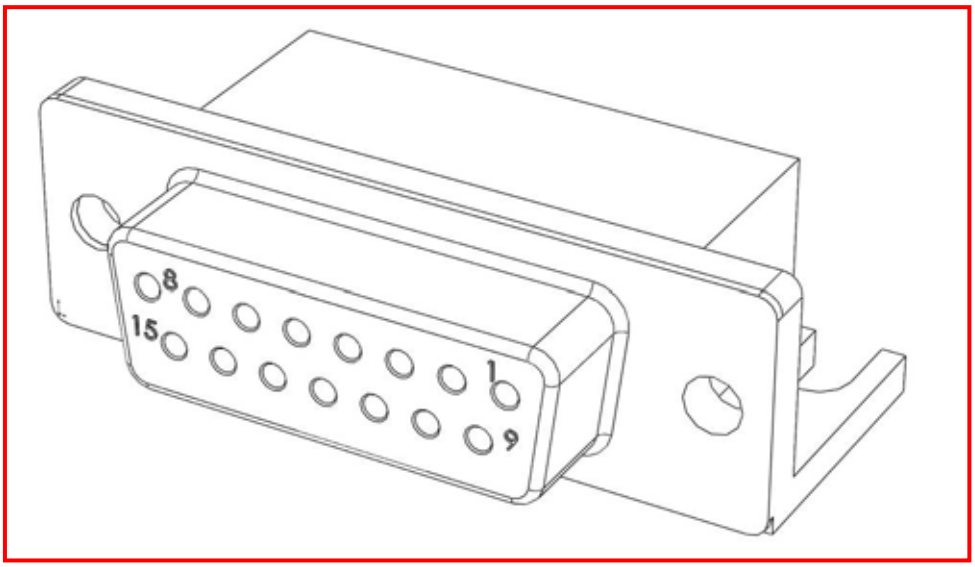


Figure 106: Female D15 Interface Connector

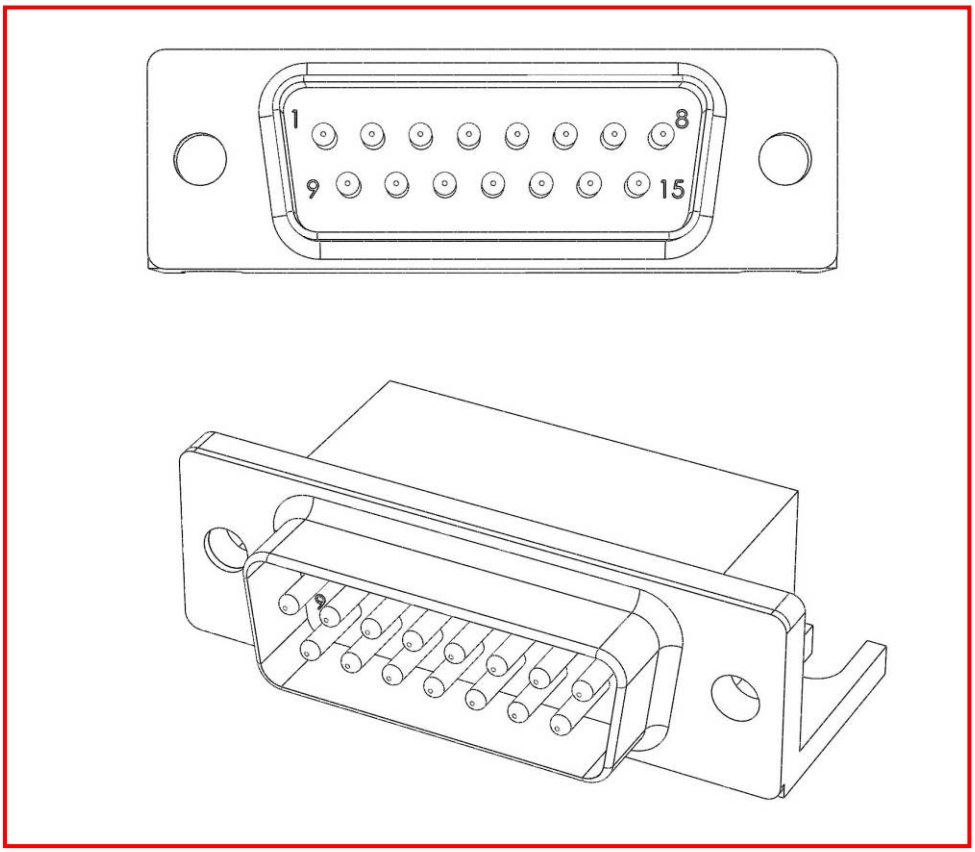


Figure 107: Male D15 Interface Connector

Appendix 1.6. X.21 Connections

The physical interface to a **Tieline** codec is via a DB15 connector (ISO 4903). The pin-outs for D-15 X.21 module connections are displayed in the main reference manual for all **Tieline** codecs. A straight-through cable is required from the male D15 connector on the X.21 module to a female DB15 connector, which will attach to an X.21 DCE Device (such as an ISDN Terminal Adaptor/TA).

Some DCE modems may require an adapter cable from the male DB15 connector on **Tieline** X.21 plug-in modules. Following is a common pin configuration for a DB15 female to a DB25 male (ISO2110), for connecting between a **Tieline** X.21 plug-in module and a DCE modem with a DB25 female connector. This is a fairly common pin configuration but please check the user manual of your modem to ensure it is compatible with your device.

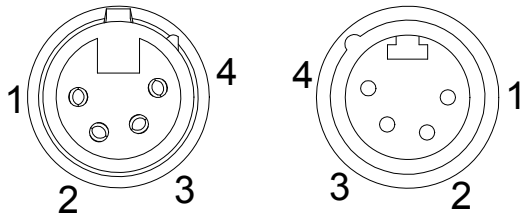
Tieline (DTE)	Tieline Pin-outs for DB15 Female	Signal	Pin–outs for DB25 Male	Modem (DCE)
-	1	Shield	1	-
Out	2	DTE Transmit Data (A)	2	In
Out	3	Request to send (A)	4	In
In	4	DTE Receive Data (A)	3	Out
In	5	Receiver ready (A)	8	Out
In	6	Signal Timing (A)	17	Out
-	7	Unassigned	-	-
-	8	Signal Ground	7	-
Out	9	DTE Transmit Data (B)	14	In
Out	10	Request to send (B)	19	In
In	11	DTE Receive Data (B)	16	Out
In	12	Receiver ready (B)	10	Out
In	13	Signal Timing (B)	9	Out
-	14	Unassigned	-	-
-	15	Unassigned	-	-

Please note: The following pins on the DB15 connector should be twisted pairs;

- Pins 2 and 9
- Pins 3 and 10
- Pins 4 and 11
- Pins 5 and 12
- Pins 6 and 13

Appendix 1.7. XLR 4 Pin Power Connectors

Viewed From Wiring Side



Female Male

Pin No.	Function
1	Ground (O V DC)
2	No Connection
3	No Connection
4	+12 V DC (10.5 to 17 V DC)

Table 24: XLR 4 Pin Power Connector

Appendix 1.8. CMOS Solid State Relay Connectors

This is the interface connector for the CMOS relays in the external relay box which connects to the codecs.

Appendix 1.9. CMOS Solid State Relay Specifications

These are semiconductor devices that provide a circuit closure when they are activated. They are rated to 350 volts peak across the contact closure and they have a maximum current rating of 120 milliamps. The contact closure resistance is approximately 35Ω.

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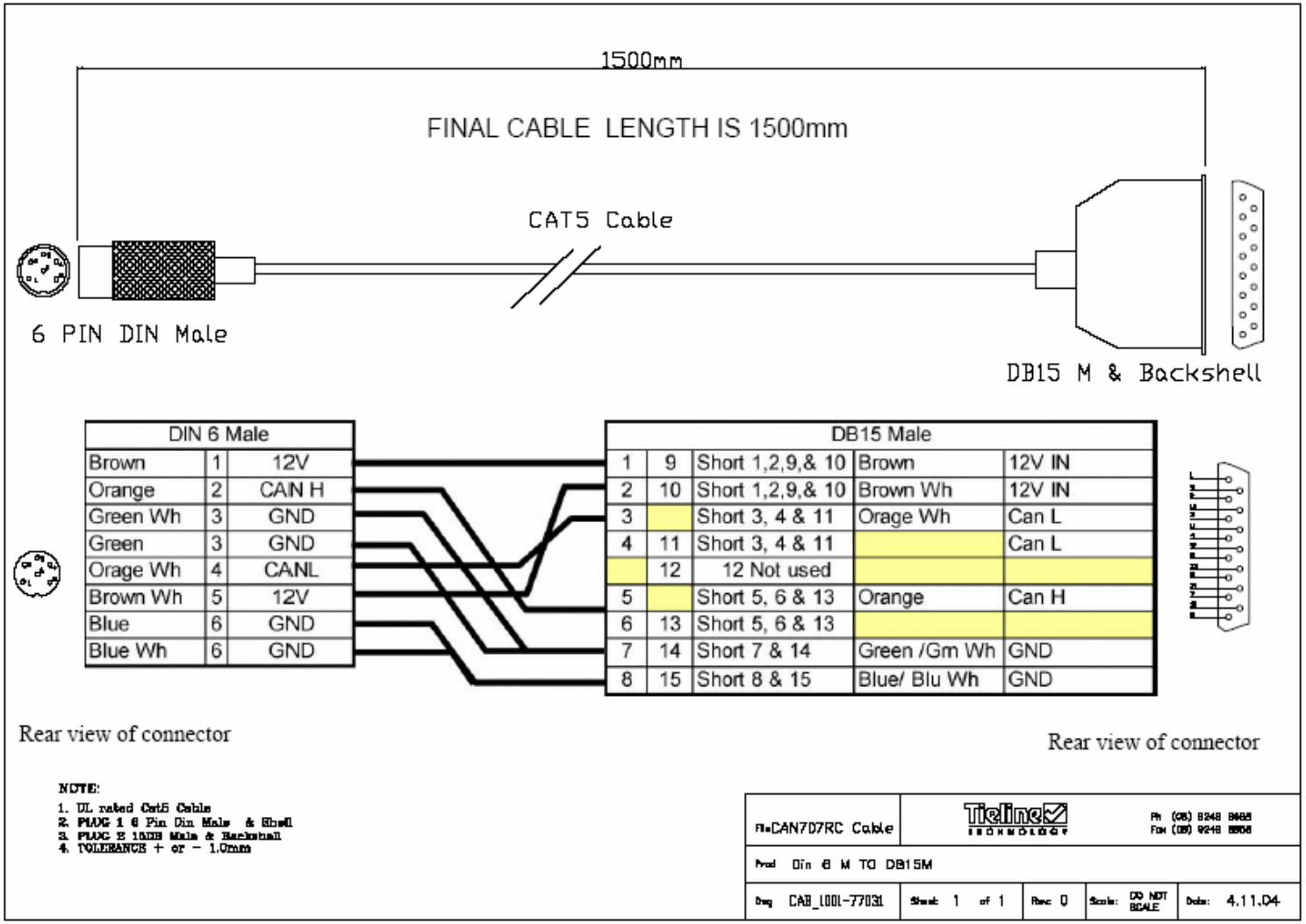
Appendix 1.10. CAN Cable Wiring Configuration



DIN 6 Male		
Brown	1	12V
Orange	2	CAN H
Green Wh	3	GND
Green	3	GND
Orage Wh	4	CANL
Brown Wh	5	12V
Blue	6	GND
Blue Wh	6	GND

DB15 Male				
1	9	Short 1,2,9,& 10	Brown	12V IN
2	10	Short 1,2,9,& 10	Brown Wh	12V IN
3		Short 3, 4 & 11	Orage Wh	Can L
4	11	Short 3, 4 & 11		Can L
	12	12 Not used		
5		Short 5, 6 & 13	Orange	Can H
6	13	Short 5, 6 & 13		
7	14	Short 7 & 14	Green /Grn Wh	GND
8	15	Short 8 & 15	Blue/ Blu Wh	GND

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TIELINE TECHNOLOGY (Manufacturer) warrants that this product is free of defects in both materials and workmanship. Should any part of this product be defective, the Manufacturer agrees, at its sole option, to:

a) Repair or replace any defective part free of charge for a period of two years from the date of the original purchase, provided the owner returns the equipment to the manufacturer. No charge will be made for parts or labor during this period. Transportation charges are the responsibility of the purchaser
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.

This Warranty excludes assembled products not manufactured by the Manufacturer whether or not they are incorporated in a Manufactured product or sold under a Manufacturer part or model number.

THE WARRANTY IS TERMINATED WHEN ANY OF THE FOLLOWING HAS OCCURRED TO THE EQUIPMENT:

a) Damage by negligence, accident, act of God, mishandling, or it has been operated incorrectly to the various instructions described in this Operation Manual.
b) Altered or repaired by other than the Manufacturer or an authorized service representative of the Manufacturer.
c) Any adaptations or accessories other than those manufactured or provided by the Manufacturer have been made or attached to the equipment, which, in the sole determination of the Manufacturer, shall have affected the performance, safety or reliability of the equipment.
d) The original serial number has been modified or removed.

NO OTHER WARRANTY, EXPRESS OR IMPLIED, INCLUDING WARRANTY OF MERCHANTABILITY OR FITNESS FOR ANY PARTICULAR USE, APPLIES TO THE PRODUCT, nor is any person or company authorized to assume any warranty for the Manufacturer or any other liability in connection with the sale of the Manufacturer's products. The Manufacturer does not assume any responsibility for consequential damages, to personnel, equipment, expenses, or loss of revenue or property, inconvenience, or interruption in operation experienced by the customer due to a malfunction in the purchased equipment.

No warranty service performed on any product shall extend the applicable warranty period.
In case of unsatisfactory operation, the purchaser shall promptly notify the Manufacturer in writing, giving full particulars as to the defects or unsatisfactory operation. Upon receipt of such notice, the Manufacturer will give instructions regarding the shipment of the equipment, or such other matters as it elects to honor this warranty as above provided.

This warranty does not cover damage to the equipment during shipping and the Manufacturer assumes no responsibility for such damage. The customer shall pay all (transportation) costs.

This warranty extends only to the original purchaser and is not assignable or transferable. Tieline Technology is continually seeking ways of improving our products and service to customers. Please visit http://www.tieline.com/Tieline_04/warranty/warnty.htm for the current warranty.

Subject to change without notice.

Appendix 4. Compliances

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 RENs 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.

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. Commander G3 Codec Specifications

Input/Output Specifications

TLR300B Rackmount Codec

Analog Audio Input Card - optional card	2 x Female XLR and 1 x 6.35mm (1/4 inch) jack Aux balanced
Analog Audio Outputs	2 x Male XLR and 1 x 6.35mm (1/4 inch) jack Aux
AES/EBU Audio Card - optional 1	1 x stereo Female XLR input and 1 Stereo Male output with Word Clock on BNC
AES/EBU Audio Card - optional 2	1 x stereo Female XLR input and 1 Stereo Male output with STR Ref Clock
Headphones- on all card options	1 x 6.5mm (1/4 inch) at rear shared with Aux out

TLF300 Field Codec

Audio Inputs	2 x Female XLR and 1 x RCA Aux
Audio Outputs	2 x Male XLR and 1 x RCA Aux
Headphones	2 x 6.5mm (1/4") Jacks
Audio Input Impedance	High Impedence - > 5K ohm
Output Impedance	50 ohm Balanced
Clipping Level	+18dBu (inputs and outputs)
Frequency Response	20Hz to 15kHz over POTS/GSM and 20Hz to 20kHz over ISDN.
Total Harmonic Distortion	<0.01% at +4dBu
Signal To Noise Ratio	Typically 84dB at +4dBu

Sample Frequencies

A/D &D/A Converters 24 bit Sigma-Delta	
IP Sample Frequencies	16kHz, 24kHz, 32kHz, 48kHz
3G Sample Frequencies	16kHz, 24kHz, 32kHz, 48kHz
POTS Sample Frequencies	16kHz, 32kHz
ISDN Sample Frequencies	16kHz, 24kHz, 32kHz, 48kHz
X.21/V.35 Sample Frequencies	16kHz, 24kHz, 32kHz, 48kHz
GSM Sample Frequencies	16kHz, 32kHz

Compression Algorithms

POTS	Tieline Music and Tieline Voice G3
ISDN	G.711, G.722, Mpeg Layer 2, Mpeg, Tieline Music, Tieline Music Plus
GSM	Tieline Music and Tieline Voice G3
X.21	G.711, G.722, Mpeg Layer 2, Tieline Music, Tieline Music Plus
IP	Tieline Music, Tieline Music Plus and Tieline Voice G3
3G	Tieline Music, Tieline Music Plus and Tieline Voice G3

Module Interfaces

3G - (Software module)	3G GSM/UMTS/HSDPA/HSUPA or EVDO Rev 0/Rev A interal cards available or broadband cell phone via Tieline USB master module.
USB	3 x USB master inputs
IP - (Software module)	TCP/UDP, SIP, over LAN, WAN, open internet, WiFi, WiMax
ISDN	Basic Rate S/T or U modules with up to 2 B channels supported per module with RJ45 interface
POTS / PSTN / Analogue Telephone	POTS Modem module with RJ11 connector
GSM	HSCSD 14,400 to 28,800bps and CSD 9600bps (requires sim card)
X.21	DB15 male connector. Up to 384KBits/S Compressed, up to 2 Mb Uncompressed
Satellite	External BGAN or Inmarsat Satellite phone can be connected to the ISDN module or IP LAN port

Data and Control Interfaces

USB Slave	USB -1.1 slave
USB - Master optional module	USB 1.1 Master
LAN	10/100 base T RJ45 connector
Serial	RS232 up to 115kbps with or without CTS/RTS flow control Female DB9 Connector
RS232	Data interface port can be used as a proprietary data channel
CAN	Differential CAN, 6 pin female DIN connector

Front Panel Interface

1RU Rackmount codec	12 x Status LED 1 x Red = Power 1 x Green = Audio is active and present 4 x Green = Input and audio outputs are active 2 x Green = Aux input and output audio is present 2 x Orange = CXN 1 and 2 are connected 2 x Orange = Assignable
Reset Button	Recessed pin hole reset button (Not fitted on 2RU)

Front Panel Interface

2RU Rackmount codec	Keypad 20 Key
Display	4 x 20 line LCD
Display	4 x configurable Tx and Rx indicator LCDs
Level Indicators	4 x 8 level LED's, Configurable source with selector button.
Channel On/Off	On/Off, Gain controlled and matrix router configurable
Cue/intercom	Matrix Router configurable (Software option)
Microphone input	1 x 6.35mm (1/4inch) jack Aux balanced
Headphone Monitoring	1 x 6.35mm (1/4inch) shared with Aux out with volume and mix adjustment
Send Return Mix	Mix between audio send/return or a mix of both
4 x Programmable hotkeys	Function manager configurable
Send Return Mix	Switch between audio send/return or a mix of both

GPIO Relay Controls

Inputs	2 x inputs: Contact Closure to ground with internal 5V feed at 5mA 0.15" pitch male pheonix connectors. (12V power connector has two relay inputs for power sensing one independent and 1 duplicated with phoenix connector)
Outputs	2 x outputs: floating solid state switching 48V 100mA
Relay Expansion	Expandable to 8 in and 8 out using optional CAN8+8 product

Weights and Dimensions

1RU Rackmount codec

Dimensions TLR300B 1RU	48.3cm x 23.7cm x 4.5cm - 19 x 9.33 x 1.75 inches
Weight TLR300	1.87kg or 4.12lbs

2RU Rackmount codec

Dimensions TLR300B 2RU	48.3cm x 27.3cm x 8.8cm - 19 x 10.7 x 3.46 inches
Weight TLR300+303 attached	2.83kg or 6.5lbs

TLF300 Field Codec

Dimensions	21.6 x 21.8 x 7.5cm or 8.56 x 8.58 x 2.95 inches
Weight	1.2kg or 4lbs

General

TLR300B Rackmount Codec	Operating Temperature 5°C to 45°C (40°F to 115°F) Power 90-250VAC, Auto Sensing 3 PIN IECconnector

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