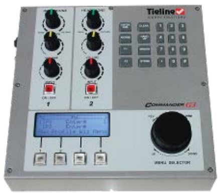
Quick Start Guide for TLF300, TLR300B and TLM600 Integrated Mixer-Codecs

Full product manuals are available for download at www.tieline.com/support/documentation

TLR300B

TLM600



TLF300

Software GUI Interface for the TLR300B

ToolBox Software Note: We recommend using a LAN or serial connection when installing ToolBox G3 software on operating systems other than XP. Please read http://www.tieline.com/Support/toolbox-G3 before attempting to connect ToolBox via USB.

Broadcast Codec Quick Start Manual Version 14.0
Software Version: Tieline Toolbox V.4.14.62 RPTP version 106
Firmware Version: 1.6.122; RPTP version 106
January, 2015

Table of Contents

(This manual includes connection procedures for POTS, ISDN, GSM, X.21, IP and 3G/4G cellular IP Connections. For Safety, Compliance, Warranty and Software License information, please see each codec's main reference manual.)

Section 1. Quick Start

1.1. Getting Connected Fast

The following pages include all the Quick Start connection information you require to connect quickly using either POTS, ISDN, GSM, X.21, 3G/4G cellular or wired IP connections.

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

If you are unsure about which buttons do what on your codec, please see the section titled 'Operation of your Codec: Codec LCD Displays and Dialing Connections' in your codec's reference manual. This can be downloaded from www.tieline.com/support/documentation.

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

1.2. Quick Start Procedure for New Connections

Before attempting to connect ask yourself the following questions:

- How many microphones are being used and is the broadcast mono or stereo?
- What is the likely connection bandwidth available for the chosen connection transport (i.e. POTS, ISDN, and 3G/4G etc.)?
- Which is the profile best suited to this broadcast?

As an example, if you have a single remote commentator microphone you wish to send to the studio via a POTS connection, a profile such as *Manual Default Mono Program* would be ideal. This can provide one channel of high quality 15 kHz audio over a single POTS line. To choose this profile and connect:

1. Select **[Menu] > [Load Profile] > [ManDflt MonoPgm]**.
2. Load the profile by selecting **SOFTKEY 2 [YES]**.
3. Scroll to the POTS connection on the main codec LCD screen that you wish to connect with.
4. Dial the phone number at the **POTS1 Enter#** prompt on the codec LCD using the codec keypad and press **ENTER/DIAL** to initiate the call and connect.

1.3. Which Profile Should I Select?

Now that you have a feel for how to select a profile and connect, think about the broadcast you are setting up. How many microphones are you using and what connection transport is available to you? Will the broadcast be mono or stereo and will IFB communications be required? By answering these questions you can then look at the following Manual Default Profiles in order to select your preferred connection profile.

1.3.1. 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 3G or 4G connection, an ISDN B channel, an X.21 connection, or IP (Internet, DSL, LAN, WAN).

1.3.2. 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[1], two POTS connections, X.21, IP (Internet, DSL, LAN, WAN), a 3G or 4G connection, or a combination of these connections.

1.3.3. 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[1], two POTS lines, a 3G or 4G connection, an X.21 connection, or IP (Internet, DSL, LAN, WAN).

1.3.4. Manual Default Dual Program

With this profile you can send a ‘mix of all inputs’ over two separate mono channels to two different locations, or send two separate inputs or bi-directional mono feeds over the same or two different networks. Use any combination of POTS, GSM, 3G, X.21, ISDN or IP connections (Internet, DSL, LAN, WAN) to send these audio streams. Now you need only three codecs to do two separate remotes!

[1] 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).

1.4. 10 Simple Steps to Connect Tieline Codecs

Use the black rotary **MENU SELECTOR** 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 **MENU SELECTOR** 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:** TLR300B rack unit codecs supply 15 volts of phantom power to the *Aux* microphone input only. This is always switched on.

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 **MENU SELECTOR** 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** button 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 **MENU SELECTOR** 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 TIPS: 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 at the studio from the field codec. I.e. always dial from the field codec to the studio codec over the Internet.

In most situations when connecting over IP we recommend using the default automatic jitter buffer setting of “best compromise”. This automatically minimizes the IP packet jitter buffer to allow for your network’s capacity to carry IP packets.

If you are unable to achieve a connection using these instructions, please refer to the reference manual for your codec for more detailed connection information. If you are connecting using IP, cellular wireless 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.

1.5. Quick Start Procedure for Cellular Wireless Connections

Connecting your codec over wireless 3G and 4G networks 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.

IMPORTANT NOTES:

1. Tieline 4G USB modules support the use of the following modems only:
   - Pantech® 4G LTE USB Modem UML290 (available from Verizon Wireless™ in the US)
   - Telstra™ USB 4G Modem 320U (available from Telstra™ retailers in Australia)

2. Tieline 4G-LTE modules (using SIM cards) are available to support different wireless network bands globally. Visit http://www.tieline.com/products/accessories/4G-LTE-Module for more details.

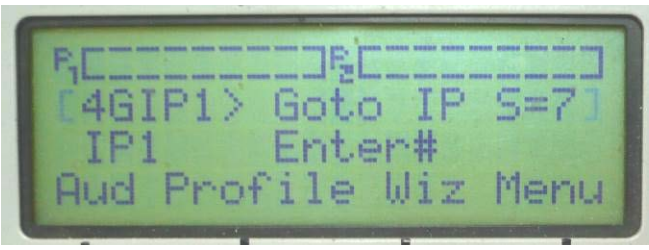
3. Tieline CDMA EV-DO 3G modules don't use SIM cards and need to be activated and provisioned in order to connect to cell-phone networks in the U.S.A. Use the procedure outlined in this manual to program your module before use over these networks.

4. By default, wireless network settings are configured in **Autodetect** mode. If you insert a 3G/4G LTE or 4G USB module into your codec it will be configured to connect over 4G if available, and 3G if 4G is unavailable.

5. We recommend you turn Forward Error Correction (FEC) off for wireless IP connections.

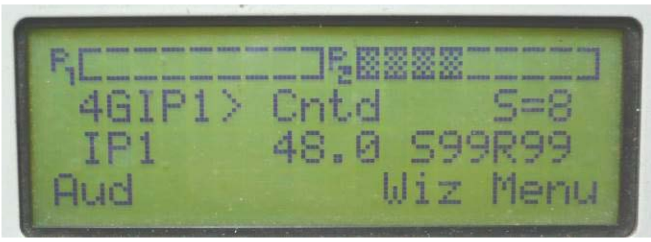
Procedure to Connect over 4G or 3G:

1. Insert a Tieline USB or cellular module into your codec and then power up the codec. Please note that by default the codec LCD screen will display the cellular network currently available, e.g. **3GIP1** or **4GIP1**.
2. Scroll to **3GIP1** or **4GIP1** with the black rotary **MENU SELECTOR** and press **SOFTKEY 3 [Wiz]** and then select **SOFTKEY 4 [OK]**. Next select **[Wireless Network →3G/4G IP←] > [3G/4G 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** or **4GIP1 > Prs Entr** is displayed, press **ENTER/DIAL** to connect to your cellular network. Note: If **3GIP1> Enter #** or **4GIP1> Enter #** is displayed, enter the cell-phone number (for your EV-DO or HSDPA network) using the codec keypad. Some cell-phone networks require you to dial this number to connect. Next, press **ENTER/DIAL** to connect to your cellular network.
4. Once connected, the codec will display **3GIP1> or 4GIP Cntd Goto IP**.



5. Next scroll to **IP1** on the main codec LCD screen and 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 codec's grey keypad to dial and connect using the connection profile you have selected, e.g. mono, stereo, mono/IFB or dual mono. Note: In most situations it is only possible to dial from the remote codec to the studio codec over IP because only the studio codec is using a public IP address.



7. Try to maintain a link quality reading for your connection of between 70% and 100%. This is indicated by the **S99** (Send) and **R99** (Return) indication on the previous codec screen image. **S=8** refers to the cellular signal strength being received. Note: In firmware prior to v1.6.122 this indication was represented by **L** (Local) and **R** (Remote).

8. To negotiate higher bit-rates press **F2** then **3**; for lower bit-rates press **F2** then **9**. If you hear audio drop-outs the current bit rate cannot be supported and should be renegotiated down.
9. To disconnect, hang up your IP connection and then hang up the 3G/4G connection.
10. To change other cellular settings select **SOFTKEY 4 [OK]** and scroll to **[Configuration] > [GSM/3G/4G Setup] > [3G Module].**

1.6. Connection Bitrates and Algorithm Options

The algorithm you select will depend 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 and your connection bit-rate is quite good.

In addition, some algorithms, such as *MP2*, have a longer inherent delay than others such as *Voice G3*, *Music* and *MusicPlus.* This can be a major consideration for live applications that integrate talkback callers or live remote-crosses into a broadcast.

The following algorithm table displays the algorithm options available for POTS, ISDN and IP connections.

Algorithm	Very Low Delay	Moderate to High Delay	Excellent Performance at Low Bit rates	Preferred for Live Remotes	Preferred for STLs and Audio Distribution	Highly Compatible with other Codecs	Auto Jitter Buffer Capability: non-SIP connections	Auto Jitter Buffer Capability: SIP connections
Linear PCM	✓	✓			✓	✓		
Voice G3 (when available)	✓		✓	✓			✓	
Tieline Music	✓		✓	✓			✓	
Tieline Music PLUS	✓		✓	✓	✓		✓	
MPEG Layer 2	✓					✓	✓	✓
G.722	✓					✓		✓
G.711	✓					✓		✓

Please note: Psychoacoustic measurement methodologies have been used to determine the audio quality of connections.

Connection Type	Algorithm	Minimum Bit-rate and Bandwidth	Recommended Connection for On-air use.
POTS	Music	Up to 7.5kHz at 9,600 bps[2]	Up to 15kHz at 24,000 bps
	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.	
	Music	15kHz on all ISDN connections.	
	MP2	24 kHz mono at 64,000 bps.	
	G.711	3 kHz at 64 kbps.	
	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[3]	7.5 kHz at 14,400 bps (Using Voice G3)	15kHz at 24,000 bps (Using Music)
4G/3G/IP	Music PLUS	Up to 20kHz mono at 48,000 bps	Up to 20kHz mono at 48,000 bps
	Music[4]	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[5] (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
	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	Linear PCM[7]	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.	
	MP2	24 kHz at 64,000 bps.	
	G.711	3 kHz at 64 kbps.	
	G.722	7 kHz at 64 kbps.	

Table 1: Algorithm Connection Bit rate Table

NOTE: From the 31st March 2011 Tieline has discontinued support for the Voice G3 algorithm. As a result it has been omitted from firmware release v.1.6.72 and all codecs purchased after this date. Codecs with firmware v.1.6.72 or higher can connect to G3 codecs running older firmware versions, but will have to use an algorithm other than Voice G3. Voice G3 is usually used for low-bitrate connections over GSM and POTS. We recommend using Tieline Music as a substitute for these low bit-rate connections.

[2] 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.
[3] 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*.
[4] Stereo IP and 3G/IP connections using the *Music* algorithm require a minimum connection bit rate of 48kbps.
[5] 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.
[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] Only available over a point-to-point crossover cable (i.e. CAT 5) and using high quality LAN switching.

- 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 for your codec.
- 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. For more information on data please see the section in each codec's reference manual titled 'Data Transfer and Using 3rd Party Devices'.
- Remote control function of a codec is not possible unless you are using the Music, MusicPlus or Voice G3 algorithms.

1.6.1. 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 only relevant 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 2: UDP IP Broadband Uplink Bandwidth Table

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

1.6.2. MPEG 2 Layer 2 Recommended Bitrates

The following tables indicate the connection bit rates that are recommended for MP2 (MPEG 1 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.

1.6.2.1. 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.

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. In the connection wizard only acceptable algorithm and sample rate settings are available in order to optimize the quality of connections. In the main connection menu via **[Menu] > [Configuration] > [ISDN/IP/X.21]**, all connection bit rates are displayed as options.

1.6.2.2. 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 3: ISDN MPEG 2 Layer 2 Recommended Bitrates

1.6.2.3. 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	

Table 4: IP MPEG 2 Layer 2 Recommended Bitrates

Please Note: The default connection bit rate for MP2 is 64kbps over IP. Other algorithms connect at 9.6 kbps by default.

1.7. POTS versus new POTS G3 Modules

The POTS G3 module dials, negotiates and retrains differently to the original Tieline POTS module. The connection screen displays differently as well. We strongly recommend you read the section in the main reference manual for your codec titled 'POTS versus new POTS G3 Modules'. This will explain the differences in dialing each module.

1.8. Troubleshooting

If audio levels are low once two codecs have connected, adjust audio input levels as per step 6 in the '10 Step Connection Guide'. If no audio is metering check that input gain settings are correct, inputs are switched on, input gain is turned up and that you are supplying power to microphones if required. The default phantom power setting is off. TLR300B rack unit codecs supply 15 volts of phantom power to the *Aux* microphone input only. This is always switched on.

To adjust input gain or phantom power settings on the remote codec during a broadcast it is necessary to use ToolBox software. For more information on ToolBox please see the sections in each codec's reference manual titled 'ToolBox Software Operation' and 'Configuration File System'.

WARNING on Module Installation: Please ensure that the codec is not powered up when inserting and removing modules. 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.

Warning on Digital Phone Systems: Digital phone systems typically run off voltages greater than the 50 volts used by the POTS/PSTN system. 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.

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.