

## Tieline Codec Configuration for POTS Leased Line (Dry Line) Connections

### 1.1. Purpose

This document can be used to assist users of Tieline codecs to create permanent POTS connections over leased line networks using Tieline DL100 dry line interface units.

### 1.2. Requirements

- ✓ Codecs installed at the studio must use firmware version 1.6.48 or higher. Codecs at the transmitter site should retain firmware installed at the factory.
- ✓ Codecs should connect using manual default profiles that are programmed into the codec by default.
- ✓ Set up the studio codec first and commence dialing, then set up the transmitter site codec.

### 1.3. Differences between Leased Lines and Public Telco Lines

Most telephone exchanges provide 48 volts across public telephone lines. Tieline audio codecs use this voltage to power the input circuitry on their POTS modules.

Leased lines are not powered like a typical telephone exchange. As a result, to use a pair of codecs across leased line networks they must be connected to a DL100 at each end of the line. DL100s provide 24 volts to power the modem in the POTS G3 module supplied with your codec.

### 1.4. Connecting the Codec to the DL100

There are two RJ-11 phone inputs on the DL100. Connect each codec's POTS G3 module to the input labeled "Codec" and connect the Telco leased line to the input labeled "Private Line".

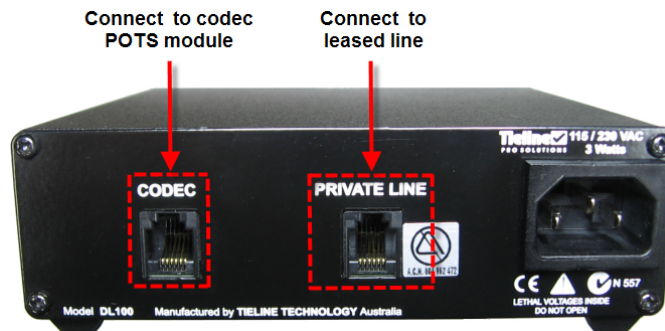


Figure 1: DL100 Dry Line Interface

**Note:** Connect the IEC power cable supplied with the unit to the power socket on the DL100 and check that the red LED on the front of the unit is illuminated. The unit is supplied to Papua New Guinea with compatibility for 230 volt AC power sources.

**WARNING:** Do not connect the DL100 to a public telephone network. To prevent electrocution, do not remove the rear or front panels with power attached to the unit. It is a good idea to have a UPS (uninterruptable power supply) attached to all STL connected codecs.

## 1.5. Programming a Codec for Leased Line Connections

Both codecs need to be programmed to connect over leased lines and there are six parts to the configuration process:

- ✓ Program both codecs for connecting in leased line mode.
- ✓ Program the codec to Auto Reconnect.
- ✓ Program both codecs with the Manual Default Mono profile.
- ✓ Program the country code in the codec to United States.
- ✓ Adjust audio inputs and check PPM meters.
- ✓ Determine the maximum allowable bit rate for the connection.

### 1.5.1. Programming Leased Line Mode and Auto Reconnect

Program leased line mode and enable the Auto Reconnect command on both the remote and local codecs so that they automatically redial and connect if the connection is lost temporarily.

1. Scroll the black rotary MENU SELECTOR on the front of the codec until the [POTS1] connection is surrounded by the square brackets, i.e. [POTS1< Enter #].



2. Press the white SOFTKEY 3 button below [Wiz] on the LCD screen.
3. Press the white SOFTKEY 4 button [OK] to continue.
4. Select [Pots Setup →Setup Leased Line←] and press [OK] to continue.
5. Select [Algorithm →Music←] and press [OK] to continue.
6. Select [Auto Reconnect →Enable←] and press [OK] to continue.
7. Press [OK] to complete programming.

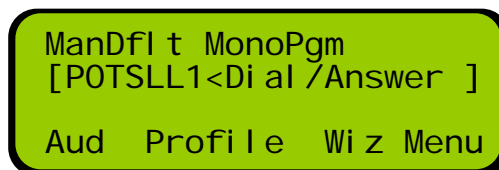


Figure 2: Codec Programmed for Leased Lines

**Note:** Do not enable both the *Automatic redial* and the *Auto reconnect* functions at the same time in a codec as they will conflict with each other.

### 1.5.2. Programming the Manual Default Mono Profile

**Note:** This is the factory default setting in the codec when shipped.

1. Select SOFTKEY 4 button **MENU**
2. Use the black rotary MENU SELECTOR to select **[Load Profile] > [ManDflt MonoPgm]**.
3. Press the CLEAR button to return to the connection screen.

### 1.5.3. Programming the Country Setting

Please ensure both codecs are programmed with USA as the country setting. This setting must be used in all countries for leased line operation. To reset this:

4. Select SOFTKEY 4 button **MENU**
5. Use the black rotary MENU SELECTOR to navigate to **[Configuration] > [System Settings] > [Country]**.
6. Scroll to **[United States]** and select this option.
7. Press the CLEAR button several times to return to the connection screen.

### 1.5.4. Adjusting Codec Input Levels

The default input level setting for the codec is **[Line Level]**. Plug your STL program audio into the codec and press the red input ON/OFF buttons to turn inputs on. Turn the INPUT knobs to adjust the audio levels until the audio peaks average at the first orange PPM LEDs.

To adjust mic/line level input gains press SOFTKEY 1 with **Aud** displayed above it and scroll to and select **[Input Gains]**. Either 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.

**Important note:** It is possible to easily control up to four remote codec inputs with the TLR300B2 rack unit codec because this functionality is already preprogrammed into the codec. PPMs 1-4 are already configured to control a remote codec's audio inputs 1-4, as long as you assign **RC** (Remote Control) to each PPM meter you wish to use for remote controlling inputs. You need to also ensure that your transmitter codec is enabled for remote control use. To do this:

Press SOFTKEY 4 **Menu** on the slave (transmitter) 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 studio 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. See the main codec reference manual for more information on remote control of codec inputs.*

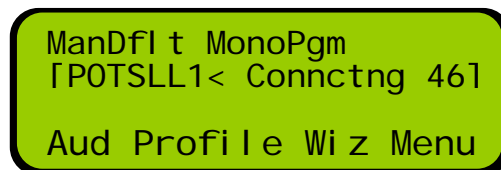
## 1.6. Dialing in Leased Line Mode

When leased line mode is enabled the dial tone and progress tones are disabled. You must physically press the DIAL button on the studio codec and press the ANSWER button on the transmitter codec to successfully connect.

### 1.6.1. Dialing from the Studio Codec

It is logical to set up the studio codec as the dialing codec and Tieline recommends setting up the studio codec before the transmitter codec. This is so you can program the studio codec to dial out before travelling to the transmitter site and installing this codec. With Automatic Reconnect enabled the studio codec will continually attempt a connection until it is successful.

Once you have programmed your codec and set up the DL100 at the studio, press the ENTER/DIAL button on the codec. This codec is now in 'dial' mode and the codec 'connecting' screen will appear and attempt to make a connection with the transmitter codec.



**Figure 3: Codec Dialing Screen Example**

The screen counts down to zero and dialing occurs continuously until a connection is made. When you connect the transmitter site codec you can answer the calls being made continually from the studio codec.

### 1.6.2. Answering at the Transmitter Codec

Program your codec and set up the DL100 at the transmitter and if the studio codec is dialing press the ANSWER button on the codec. The codec should answer the call from the studio and negotiate a connection. Please note that this may take a minute or two, depending on where the two codecs are at in the dialing sequence.

If you decide to set up the codec at the transmitter first, press the ANSWER button on the codec after the codec and DL100 have been set up. The codec will remain in 'answer' mode until it detects an incoming call from the studio. The screen looks the same as the screen on the codec that is dialing.

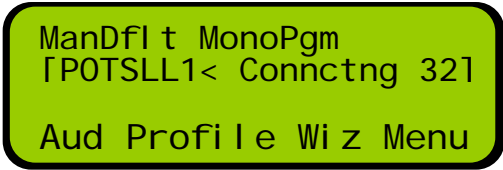


Figure 4: Codec Answer Screen Example

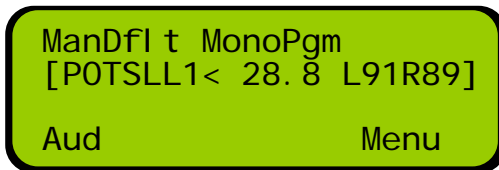
When the codec at the transmitter detects the studio codec's incoming call, the two codecs will negotiate the connection to the maximum supported bit rate over the leased line connection. Once the connection is made adjust audio levels as required.

### 1.6.3. Programming the Maximum Bit Rate

When connecting over POTS for STL connections it is important to determine the maximum bit rate that the line can support and allow some leeway for the connection to deteriorate slightly over time.

When connecting between two codecs with POTS G3 modules they initially attempt to establish a connection at 28,800 bps. If this is not possible, the codec will then renegotiate the link downwards to the highest possible bit rate where line quality is greater than 70%.

The line quality is displayed on the codec connection screen next to the connection bit rate. The "L" reading is the percentage at the codec you are at and the "R" reading is the percentage at the remote codec you are connected to.



Tieline recommends that the line quality reading on the codec connection screen should be 80% or higher for STL connections to ensure stability of the signal over time. Negotiate the connection until you achieve this percentage reliably.

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 bit rates up and down. If you hear audio drop-outs the current bit rate cannot be sustained and should be renegotiated down.

**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 Tieline's original POTS modules. Retraining takes significantly longer than renegotiation – up to 30 seconds.

Once you have determined the maximum reliable bit rate that the leased line can support, program this bit rate into the dialing codec at the studio. To do this:

1. Press SOFTKEY 4 **Menu** and select **[Configuration] > [POTS 1 or 2] > [Max Bitrate] > [select bitrate]**.
2. Press the CLEAR button several times to return to the connection screen.
3. Dial the connection.

**Important Note:** Default profiles are automatically programmed to connect using the Tieline Music algorithm, which provides up to 15kHz audio quality at 24Kbps. To maximize stability over STL connections, even if the connection is stable at 28.8Kbps, set the maximum bit rate to 24Kbps at the dialing codec.

## 1.7. Troubleshooting Connections

### 1.7.1. Poor Line Quality

If the line quality of the connection is particularly poor you may need to contact your Telco and investigate having the lines equalized to improve performance.

### 1.7.2. Retraining of Connections

The nature of POTS modems is that over long periods of time they will drift and line quality will be reduced. This requires the connection to be periodically retrained. If you notice over time that the line quality reading is lower than it should be, retrain the connection to improve the quality. You should generally choose to perform this at a time when listeners are unlikely to be listening to the connection because audio will drop out for the a few seconds during the retraining period.

To retrain the connection select **[Menu] > [Configuration] > [POTS Right/Left Setup] > [Retrain (Slow)] > [Current]**. The connection screen should display **Retran Dtctd** during this process and the line quality reading should improve after it has been completed. If it doesn't improve then the line may no longer support your previous connection bit rate and the maximum bit rate of the connection may need to be lowered.

### 1.7.3. Lost Connections

If a connection temporarily drops out the transmitter site codec will enter 'Answer' mode automatically and the studio codec will enter 'Dial' mode automatically and continuously attempt to reconnect.

