

1.1. 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 1: Voice-Capable Original POTS Module 710R1.2

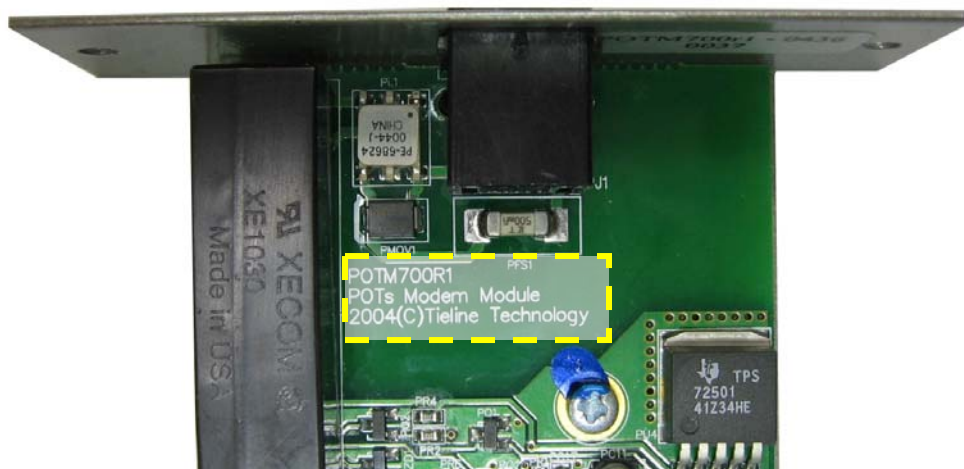


Figure 2: Original POTS Module 700R1 without Voice Capability

1.2. 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.

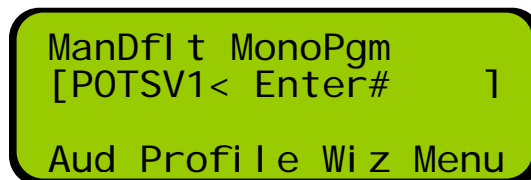
1.3. Selecting POTS Voice Mode

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.

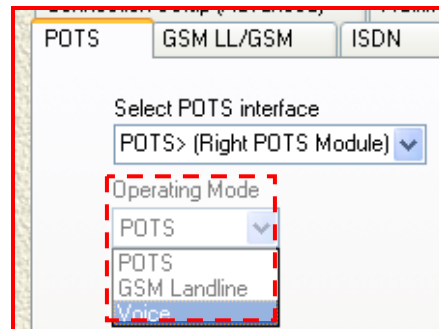
Please note: The other way to configure a plug-in module for voice mode is to select **[Menu] > [Configuration] > [POTS Left/Right Setup] > [Operating Mode] > [Voice Analog]**.

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



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.

1.3.1. 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	Auxiliary	PPM 1	PPM 2
Input 1	✓				✓	✓	✓	✓	✓	✓	✓	
Input 2	✓				✓	✓	✓	✓	✓	✓	✓	
Aux	✓				✓	✓	✓	✓	✓		✓	
Phone	✓				✓	✓	✓	✓		✓	✓	
Decoder 1			✓	✓	✓	✓	✓	✓				✓
Decoder 2												

Figure 3: 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.

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*)

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

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.

1.4. Integrating Incoming POTS voice Calls into Program Audio

COMMANDER G3 field and rack-mount codecs are not fitted with an on-board phone coupler like the **i-Mix G3**. However, it is still possible to integrate a voice call into program audio. To do this:

1. Plug a standard POTS/PSTN line into the POTS module you plan to make the voice call from.
2. Select the default mono program profile via [Menu] > [Load Profile] > [ManDflt MonoPgm] > [Really Load? →YES←].
3. Select POTS voice mode for the POTS module by pressing SOFTKEY 3 [Wiz], then press [OK] to proceed. Scroll using the rotary MENU SELECTOR to [Setup Modem Voice] and press press [OK] to configure the connection.
4. By default, your codec is programmed to automatically answer incoming calls. To manually answer incoming calls, select [Menu] > [Configuration] > [POTS> Left or Right Setup] > [Auto Answer]. Enter the number “0” into the codec and press [OK].

1.4.1. Programming the Phone/Cue Button

Using the default “ManDflt MonoPgm” profile, an incoming call will be added to program audio as soon as the incoming call is detected, including ringing tones and the caller’s voice once the call is answered.

To manage incoming voice calls off-line it is necessary to program a “Phone/Cue” button on the codec. Essentially, this takes codec input 1 off-line, along with an incoming voice call, enabling off-air call answering, dialing and communications.

To program the “Phone/Cue” button, connect your codec to a PC with ToolBox. Use the ToolBox drop-down menu to select “File” → “Get Config From Codec”.

Next, go to the “Profile Editor” and select the **Functions** tab. Then click on the *Add Function* button to open the *Add/Edit Function* dialog.

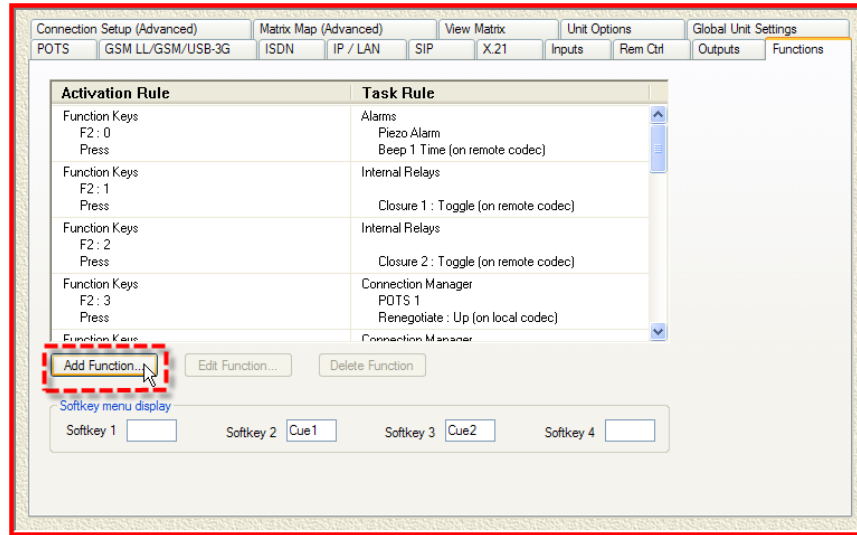


Figure 4: Add Function Button in Toolbox

In the *Add/Edit Function* dialog program one of the SOFTKEYS to toggle the Phone/Cue function on and off. This is programmed onto SOFTKEY 1 of a **COMMANDER G3** field unit codec in the following example.

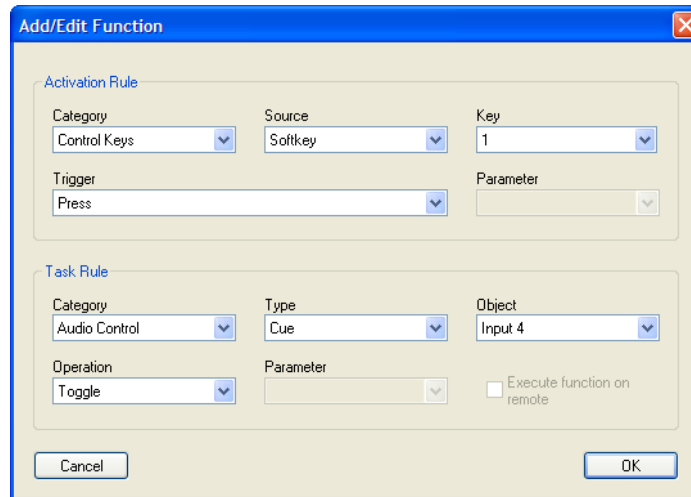
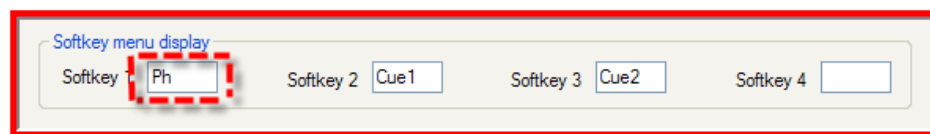
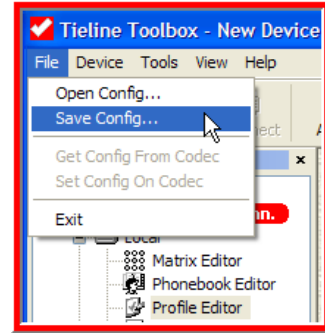


Figure 5: Softkey 1 Configured for Phone/Cue

Press “OK” to save the user function and note that it is also necessary to program a name in the *Softkey Menu Display* for SOFTKEY 1. In the following example we have named it “Ph”.



Once programmed, save the settings as a configuration file by selecting “File” → “Save Config” in the ToolBox drop-down menu. Next, connect to the codec using the “Configure” button and transfer the Config file settings to the codec using “File” → “Set Config On Codec”.



You are now ready to connect so press the rotary MENU SELECTOR for two seconds until the SOFTKEY names are displayed, e.g. “Ph”, “Cue 1” and “Cue 2”. These names remain displayed until the MENU SELECTOR is pressed in and held again. To connect:

1. Dial and connect over the transport you are sending program audio over. E.g. IP, 3G/3.5G, ISDN, POTS or X.21.
2. Engage Phone-Cue by pressing SOFTKEY 1.
3. When an incoming call is detected by either listening to Headphone 1 or viewing the codec LCD screen, you can press the ANSWER button on the keypad to answer the call. Use Mic 1 and Headphone 1 to manage call dialing and connection off-air.
4. Press SOFTKEY 1 again to disengage Phone-Cue and put the call to air.

Please Note: it is possible to create a different profile using the ToolBox *Matrix Editor*, that allows Input 1 to remain on-line while Phone-Cue is engaged. In this situation the codec could be programmed to trigger a relay and activate a light or similar to indicate when an incoming call is detected.

1.4.2. Adjusting Voice Call Audio Levels

To adjust the level of the voice call:

1. Select SOFTKEY 1 **Aud** and scroll using the rotary MENU SELECTOR to **[Phone Vol/Ctrl] > [Disable Rem Ctrl? →YES←]**.
2. Use the black rotary MENU SELECTOR to adjust the audio level as required.
3. Press SOFTKEY 4 **Exit** to return to the main codec connection screen.

1.5. Integrating Outgoing POTS voice Calls into Program Audio

It is also possible to make a POTS voice call from a Commander codec with a POTS module installed and integrate the call into program audio. Typically, this would be done when dialing out to interview someone by phone. Basically the codec should be programmed as described previously for handling incoming calls. The only difference is that the codec should be programmed with a dual program profile, such as the default ManDflt Dual Pgm.

To do this select **[Menu] > [Load Profile] > [ManDflt DualPgm] > [Really Load? →YES←]**. Once selected:

1. Dial and connect over the transport you are sending mono program audio over. E.g. IP, 3G/3.5G, ISDN, POTS or X.21.
2. Engage Phone-Cue by pressing SOFTKEY 1.
3. Use the rotary MENU SELECTOR to scroll to the **POTSV1** connection on your codec LCD screen.
4. Use the keypad to enter the POTS/PSTN number that you want to dial and then press the ENTER/DIAL button.
5. Use Mic 1 and Headphone 1 to manage the call dialing and connection off-air.
6. Press SOFTKEY 1 again to disengage Phone-Cue and put the call to air.

1.6. 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.

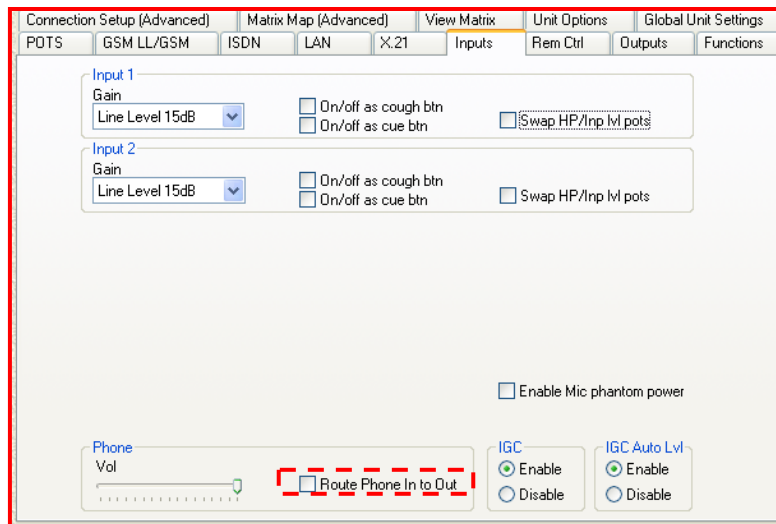


Figure 6: Route Phone Input to Outputs Check-Box

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	Auxiliary	PPM 1	PPM 2
Input 1	✓				✓	✓	✓	✓	✓	✓	✓	
Input 2	✓				✓	✓	✓	✓	✓	✓	✓	
Aux	✓				✓	✓	✓	✓	✓		✓	
Phone			✓	✓	✓	✓	✓	✓		✓	✓	
Decoder 1			✓	✓	✓	✓	✓	✓				✓
Decoder 2												

Figure 7: Audio Routed to Analog Outputs

Important Note: This feature cannot be used with default profiles if you mixing POTS voice call into program because *Phone* audio is deselected from program (*Encoder 1* above) when *Route Phone In to Out* is engaged.

1.6.1. 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*.

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