

## Configuring the PBX Call Routing Table for outbound calls (with security against unsecured calls)

The Quadro's Call Routing Table (CRT) defines how incoming and outgoing calls will be handled by the Quadro. In the CRT below there are a number of call routes that have been configured for different types of outbound calls. Handling inbound calls through the CRT will be covered in other documents.

To access the CRT go to the System menu, select Call Routing, and then click the [Call Routing Table](#) link.

The first 4 entries in the table are default entries. The first 2 entries cannot be edited or deleted.

The **Pattern** column defines the pattern that the Quadro will attempt to match to the number that has been dialed from an extension. The Quadro will look in the CRT and find the call route with the pattern that best matches (longest digit match) the number that has been dialed and use that route to place the call.

Patterns can include number digits (1-9) as well as the \* and ? wildcard characters.

The \* wildcard is a multi-digit wildcard character and the ? is a single-digit wildcard character. Where there are 2 or more routes with identical patterns in the table, the Quadro will match the one with the lowest ID number (highest in the table). Having 2 or more matching patterns is typically used when configuring Failover call routes. The Quadro will automatically attempt to use a failover call route when the call route with matching pattern and lower ID has failed.

The **NDS** column shows how many digits will be stripped from the front of the number dialed. The remainder of numbers will be sent through to the carrier. This will be explained further into the document.

The **Call Type** identifies how the call will be made. For example, FXO for the PSTN lines, ISDN for ISDN Trunks (xi models), IP-PSTN for an ITSP (VoIP Carrier), SIP for direct VoIP calls between two SIP based devices e.g. Quadro's at remote sites.

The **Fail Reason** will allow you to select the conditions under which the Quadro will consider a call route to have failed. When this failure reason has been detected by the Quadro, it will try to make another call via a failover route, if one has been configured. You can select individual fail reasons or set the fail reason to All (recommended), which will mean the Quadro will try a failover route to place a call when the call did not work for any possible reason using the first route e.g. if the internet is down and a VoIP call is attempted.

The **Description** field allows you to give a description of the call route as a reference.

For outbound calls, these are the most important columns to understand. The system administrator will specify the settings when configuring the call routes using the Call Routing Wizard and the VoIP Carrier Wizard.

## Call Routing Table

Show Detailed View >>>

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	ID	State	Pattern	Pattern Modification	Call Settings	Fail Reason	Local Authentication	Inbound Pattern/Modification	Inbound Settings	DT	UES / URP	Metric	Description
	1	Enabled	000		FXO port: Any Port	None	No	*	PBX			10	Emergency Call
	2	Enabled	0?*	NDS: 1	FXO port: Any Port	Any	No	*	PBX			10	Make PSTN Call
<input type="checkbox"/>	3	Enabled	8*	NDS: 1	SIP sip.epygi.com	None	No	*	PBX		URP: No	10	Make SIP call
<input type="checkbox"/>	4	Enabled	??		PBX	None	No					10	Call to Extensions
<input type="checkbox"/>	5	Enabled	1??	NDS: 1	SIP 203.52.18.211:5060	None	No	*	PBX		URP: No	10	SIP call
<input type="checkbox"/>	6	Enabled	3*	NDS: 1	IP-PSTN sip.phonet.net.au:5060, ML : Yes	Any	No	*	PBX		UES: 99 URP: No	10	PhoNet
<input type="checkbox"/>	7	Enabled	3*	NDS: 1	FXO port: Any Port	None	No	*	PBX			10	PhoNet
<input type="checkbox"/>	8	Enabled	?????????		IP-PSTN sip.phonet.net.au:5060, ML : Yes	Any	No	*	PBX		UES: 99 URP: No	10	Local landlines
<input type="checkbox"/>	9	Enabled	?????????		FXO port: Any Port	None	No	*	PBX			10	Local landlines Failover
<input type="checkbox"/>	10	Enabled	???????????		IP-PSTN sip.phonet.net.au:5060, ML : Yes	Any	No	*	PBX		UES: 99 URP: No	10	Mobile / Other Area Code
<input type="checkbox"/>	11	Enabled	???????????		FXO port: Any Port	None	No	*	PBX			10	Mobile / Other Area Code Failover
<input type="checkbox"/>	12	Enabled	1800???????		FXO port: Any Port	None	No	*	PBX			10	1800 (free) numbers
<input type="checkbox"/>	13	Enabled	0011*		IP-PSTN sip.phonet.net.au:5060, ML : Yes	Any	No	{15,21,22}	PBX		UES: 99 URP: No	10	O/S numbers (restricted)
<input type="checkbox"/>	14	Enabled	0011*		FXO port: Any Port	None	No	{15,21,22}	PBX			10	O/S numbers (restricted) Failover
<input type="checkbox"/>	15	Enabled	12??	NDS: 2	PBX-Voicemail	None	No					10	Direct to Voicemail
<input type="checkbox"/>	16	Enabled	13??	NDS: 2	PBX-Intercom	None	No					10	Direct Intercom Call

## Description of call routes shown in table

The first route in the table (**ID 1**) is for emergency calls. The Quadro will always try to place a call out through the PSTN and will use an FXO/ISDN interface to do this. The 000 pattern seen in the table is specified in the System Configuration Wizard by the administrator when first setting up the Quadro. The NDS (Number of Discarded Symbols) is 0 so the Quadro will send all digits to the FXO port to make the call. It will use any available FXO/ISDN interface to make the call (see Port ID column).

**ID 2** is the call route used when the user wants to place a call through an FXO port. User dials an additional 0 at the start of the destination number. This route can be removed from the table by running the System Configuration Wizard and leaving the PSTN Access Code setting blank. If there is a need to select a specific FXO port(s) to dial out, additional routes can be created with the call type FXO and selecting the specific FXO port number required.

**ID 3** is the call route used to make SIP calls to an extension of another Quadro via the internet. The user will dial 8 at the start of the number portion of the Epygi SIP address ([e.g.12345631@sip.epygi.com](mailto:e.g.12345631@sip.epygi.com)) and the Quadro will route the call through Epygi's Public SIP Server. This is a free service for Quadro users. This is also useful for testing the SIP connectivity of your Quadro. By dialing 899 from any extension of the Quadro, you will dial extension 99 of Epygi's public SIP server and can listen to a message.

**ID 4** is used for calls between internal extensions of the Quadro (PBX calls). Dialing any 2 digit number will match the ?? pattern and the Quadro will call the dialed extension. Use a ??? pattern if you are using 3 digit extension numbers.

**ID 5** uses a 3-digit pattern to make direct VoIP calls to another SIP device e.g. another Quadro. Typically this would be used in a multi-site business. This will allow a SIP call to be placed directly between the Quadro's at each site via the public IP address of each unit. As there is no carrier involvement this type of call is completely free. When a number matching the 1?? pattern shown is dialed, the Quadro will strip the 1 and send the remaining 2 digits to the remote Quadro. For example, dialing 150 at site A will call extension 50 at site B. This type of route can also be used to dial SIP devices on the LAN e.g. a SIP speaker or 3G/GSM gateway.

**ID's 6 & 7** were added by the VoIP Carrier Wizard, which is used to configure ITSP (VoIP Carrier) accounts. When configuring the Accounts through this wizard, the administrator will be asked to choose an Access Code. This is the number that will be set in the pattern field (3 in this case) and will be dialed at the start of the destination number to route calls through this ITSP account\*. You are also given the option of setting a Failover to PSTN. If you select this option, the wizard will also create a second route underneath (**ID 7** in table shown) that the Quadro will use if the ITSP account is unreachable, for example if your internet connection fails. Once you have configured the account with the VoIP Carrier Wizard you can then configure some direct dialing entries in the table to dial the destination number without having to prefix it with a pattern. The remaining routes in the CRT are used for this as well as a PSTN failover for each of these routes to be used in case of ISTP call failure.

**\*Caution-** By leaving the Access Code blank the wizard will configure a \* pattern for the ITSP account route that will allow you to dial through the ITSP without any prefix, however this is not recommended as it can cause incorrect routing issues e.g. a default route pattern may be a better match than the \*. The purpose of this document is to explain how to avoid this type of issue by using specific call routing logic.

**ID 8** is used for calls to local landline numbers (8 digits) through the ITSP. If the call fails, call route with **ID 9** will send the call through the PSTN.

**ID 10** is used for calls to 10 digit numbers which includes landline numbers in different area codes and mobiles through the ITSP. If the call fails, call route with **ID 11** will send the call through the PSTN.

**ID 12** is used for calling 1800 numbers. As calls to these numbers are free for the caller, the example shows placing the calls through the PSTN.

**ID 13** is used for calling international numbers through the ITSP. The call route has a restriction placed on it so only specified extensions can make overseas calls. This is achieved by enabling the Filter on Caller/Call Type/modify Caller ID setting on page 1 of the call routing wizard and specifying the extensions which may access the call route. If the call fails, call route with **ID 14** will send the call through the PSTN.

**ID 15** is a call route used for dialing directly to another internal extensions voicemail. This can also be used to transfer calls directly to someone's voicemail. With a pattern of 12?? and 2 discarded digits, dialing 1250 would call the voicemail of extension 50 directly.

**ID 16** is a call route used for placing an Intercom call to another internal extension. This will open a full-duplex channel between both extensions via the called extensions hands-free speaker (only if the called phone is not currently on another call). With a pattern of 13?? and 2 discarded digits, dialing 1325 would make an intercom call to extension 25 directly.

## Call Routing Wizard

### Routing Call Type - Edit Entry

Pattern:	<input type="text" value="????????"/>	(wildcard supported)	Enabler Key:	<input type="text"/>
Number of Discarded Symbols:	<input type="text"/>		Disabler Key:	<input type="text"/>
Prefix:	<input type="text"/>		<input type="checkbox"/> Require Authorization for Enabling/Disabling	
Suffix:	<input type="text"/>			
Call Type:	<input type="text" value="IP-PSTN"/>			
Metric:	<input type="text" value="10"/>			
Description:	<input type="text" value="Local landlines"/>			
<input checked="" type="checkbox"/> Filter on Caller / Call Type / Modify Caller ID				
<input type="checkbox"/> Set Date/Time Period(s)				

To configure an ITSP account, refer to the 'Configuring ITSP Accounts' document which is available for download on Alloys VoIP website in the Support section.

To configure call routes to dial a destination number without needing to prefix it with an extra pattern digit(s), first configure your ITSP account using the VoIP Carrier Wizard. Then go to the Quadro's CRT and Add a call route. This will open the Call Routing Wizard. To configure a call route for dialing local landlines phones, follow these steps.

On the Routing Call Type – Edit Entry page configure the following settings.

**Pattern:** ?????????

**Call Type:** IP-PSTN

**Description:** e.g. Local landlines

**Filter on Caller/Call Type/Modify Caller ID:** Enable this setting. It will be used for securing the route against unauthorised access.

**Call Routing Wizard**

Routing Call Settings - Edit Entry

Use Extension Settings: 99  Keep Original Caller ID  
 Add Remote Party ID

Destination Host: sip.phonet.net.au

Destination Port: 5060

Username:

Password:

Multiple Logons  
 Enable Activity Timeout  
 Activity Timeout:

Use RTP Proxy

AAA Required:  
 Local Authentication

Fail Reason: Any

**Use Extension Settings:** Set this to the extension number that is used by the ITSP account. To find out this number go to the CRT and look in the UES column of the call route created by the VoIP Carrier Wizard.

**Destination Host:** SIP Server hostname of ITSP

**Destination Port:** 5060

**Fail Reason:** Any

Leave all other settings as default

**Secure the route against unauthorised access!**

By default, when you set up call routes using either the System configuration wizard (for PSTN Access Code) or the VOIP Carrier Wizard (Access Code), the resulting call routes will be restricted with an inbound setting of 'PBX-\*' in the Inbound Settings columns of the CRT. This means that only registered extensions can place calls through these routes. **BUT** if a call routes for outbound calls is manually added, this is left completely unrestricted as the 'Filter on Caller/Call Type/Modify Caller ID' box is disabled. This means that any person who can dial into the Quadro and access the Call Routing Table can dial out any unsecured route.

One way to make the configuration easier is to always Duplicate the routes set up by the VoIP Carrier Wizard and change the pattern, NDS etc. this will ensure the inbound settings are restricted by default.

**Inbound Call Type - Edit Entry**

Inbound Caller Pattern: \*  (wildcard supported)

Inbound Number of Discarded Symbols:

Inbound Prefix:

Inbound Call Type: PBX

## Call Routing Wizard

### Summary - Edit Entry

Routing Call Type		Routing Call Settings	
<b>Pattern:</b>	?????????	<b>Use Extension Settings:</b>	99
<b>Number of Discarded Symbols:</b>		<b>Keep Original Caller ID:</b>	No
<b>Prefix:</b>		<b>Add Remote Party ID:</b>	No
<b>Suffix:</b>		<b>Destination Host:</b>	sip.phonet.net.au
<b>Call Type:</b>	IP-PSTN	<b>Destination Port:</b>	5060
<b>Metric:</b>	10	<b>Username:</b>	
<b>Description:</b>	Local landlines	<b>Transport Protocol for SIP:</b>	UDP
		<b>Multiple Logons:</b>	Yes
		<b>Use RTP Proxy:</b>	No
		<b>Activity Timeout:</b>	Disabled
		<b>AAA Required:</b>	AAA disabled.
		<b>Fail Reason:</b>	Any

Review the settings and click Finish.

The new call route will appear in the CRT. To configure a failover route, tick the box for this call route in the CRT and select Duplicate from the options at the top of the table. In the Call Routing Wizard change the Call Type to FXO on the first page of the wizard. Leave all other settings as default and Finish. The failover route will now be in the table underneath the original route that was duplicated.