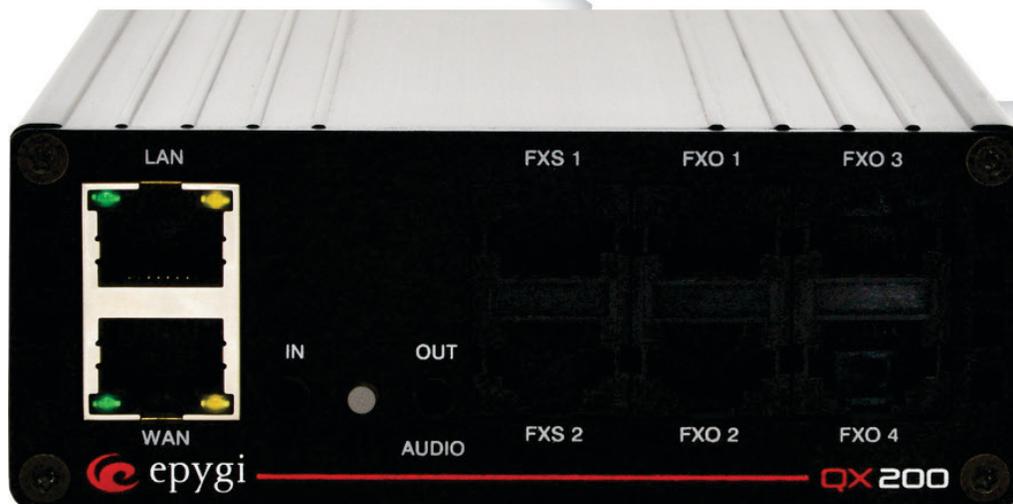




MyNetFone



Configuring Epygi QX IP PBXs with MyNetFone

Abstract: This document describes the configuration of the Epygi QX IP PBXs to use the IP-PSTN service from MyNetFone.

Document Revision History

Revision	Date	Description	Valid for SW	Valid for Models
1.0	06-Oct-2015	Initial release	6.1.x and higher	QX IP PBXs

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1 Introduction

This document describes the configuration of Epygi QX IP PBXs (hereafter QXs) to use the SIP trunking service from **MyNetFone**. The QX is capable of making IP-PSTN calls via **MyNetFone** SIP trunks. This solution allows QX users to make cost saving calls to the global PSTN.

Please Note: The described configuration is common for all QX IP PBX models.

Please Note: Security issues and calling rates are beyond the scope of this document. See the listed documents in the [References](#) section to get more information on the security related issues, as well as on the advanced configuration for QXs.

2 Scenario

Provider: MyNetFone

- offers outbound and inbound calls.
- allows parallel outbound calls to be made from the QX.
- allows parallel calls to be received to the QX.

Customer:

- will make long distance PSTN calls through the **MyNetFone** SIP trunks.

2.1 Requirements and Preparations

- The QX is connected to the network and all network settings are properly configured.
- The QX is running software 6.1.x and higher.

The MyNetFone trunks do not authenticate based on username / password combinations. The **MyNetFone** SIP Trunk authenticates based on

- IP Address that the QX is presenting to **MyNetFone** SIP server
- **MyNetFone** DID that is being sent from the QX
- Correct formatting of various fields within the SIP Packets being sent from the QX

2.2 Account Information from MyNetFone

MyNetFone will provide the following data to the customer:

- SIP Server - SIPxx.mynetfone.com.au
- Signaling port for SIP server - 5060
- Telephone Number (DIDs allocated to the customer) - 03 8376 6600 - 03 8376 6699 (samples)

3 Configuration

As the Epygi “**Voip Carrier Wizard**” typically requires User credentials and configures an extension which holds registration with the VoIP Provider, **MyNetFone** trunks will not use this **Wizards Configuration** mode.

Instead the process below will explain the configuration as manual steps.

Important to note, as no extension is configured, the trunk is only setup in the **Call Routing Table**. The **Codec** parameters are assigned to outbound calls from the System Default 00 Auto Attendant Codec Preference Listing.

Two routing rules are required to support **MyNetFone** SIP trunks.

- An Inbound rule from which **MyNetFone** will send calls to the QX
- An outbound rule from which the QX will send calls to **MyNetFone** network

Call Routing Rules are processed by metric as well as from top to bottom.

All rules that match will be evaluated, and the most distinct match will be processed.

Caution needs to be taken when configuring the rules for **MyNetFone** trunks that there is not overlapping rules that are taking precedence over the rules you have configured based on this guide.

There is also support for Failover rules as well, but this is beyond the scope of this document and should be raised with your hardware supplier.

4 Configuration for Making Outgoing Calls through MyNetFone

To make outgoing calls through the MyNetFone Trunks:

1. Go to the **Telephony**→**Call Routing**→**Call Routing Table** page.
2. Press the **Add** button to start the **Call Routing Wizard** to create a new routing rule.
3. Fill the following fields to create a new routing rule (Figure 1):
 - **Destination Pattern** - * to make all destinations to match this route.
 - **Destination Type** - IP-PSTN.
 - **Filter on Source / Modify Caller ID** - enabled (to restrict / secure access to this route and required to present the call in the correct format to **MyNetFone**).

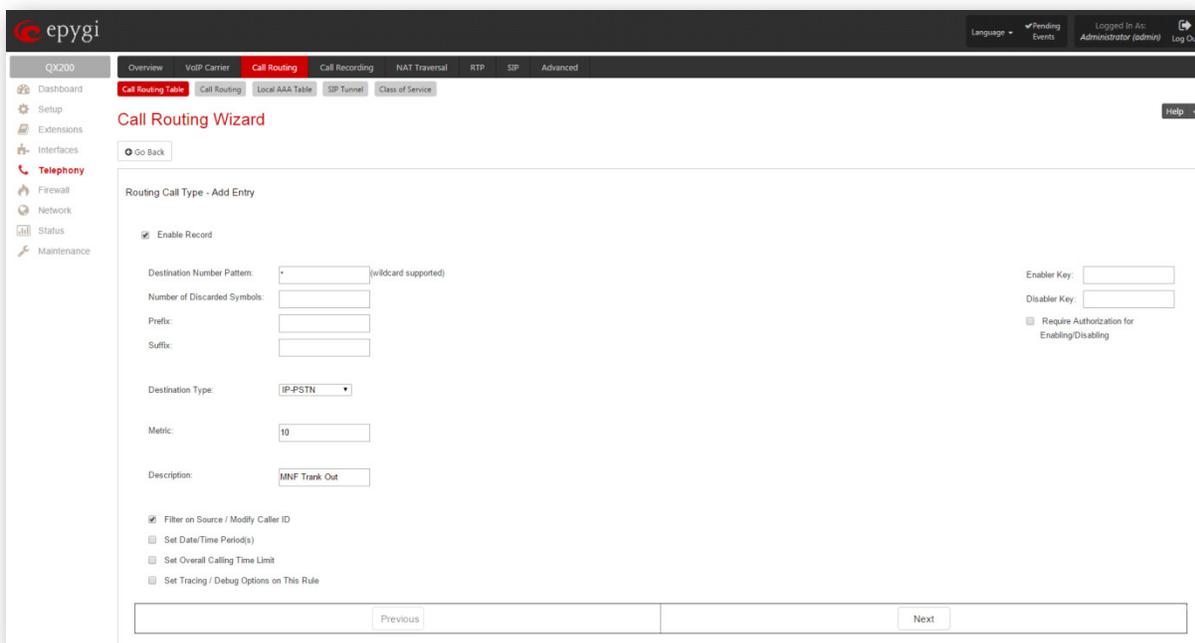


Figure 1: Call Routing Wizard - page 1

4. Press the **Next** button to continue the wizard.

Fill the following fields on the second page of the wizard:

1. **Destination Host** – SIP20.mynetfone.com.au
2. **Destination Port** – 5060
3. Ensure the **Use Extension Settings** is blank (Figure 2).

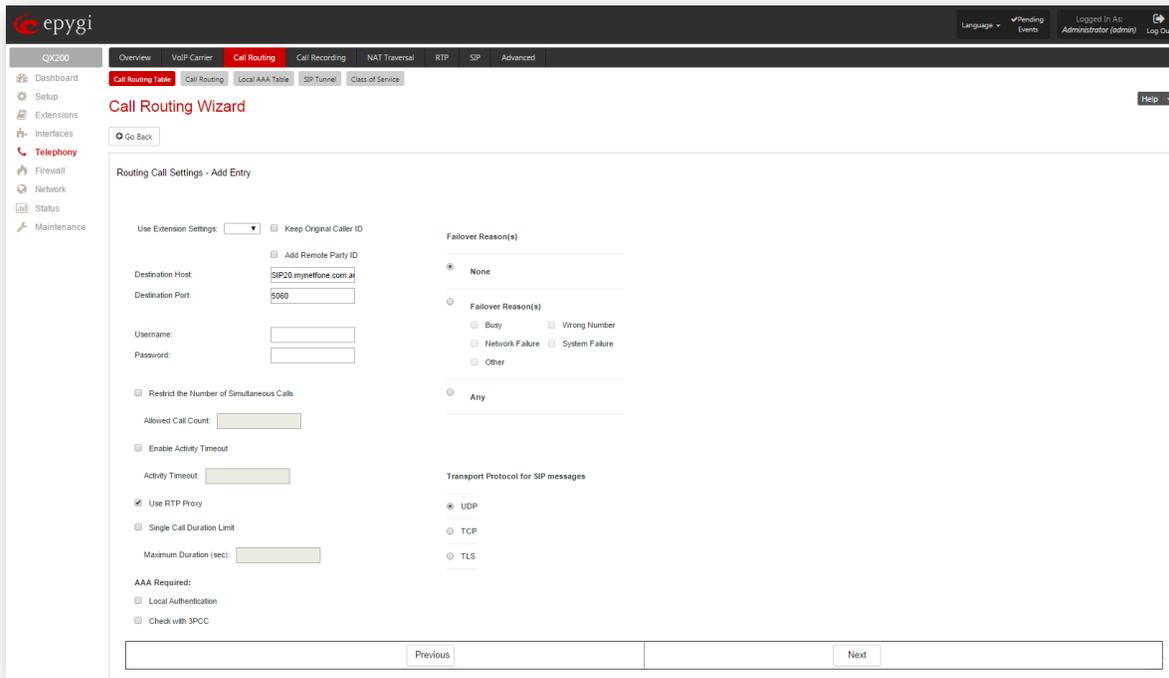
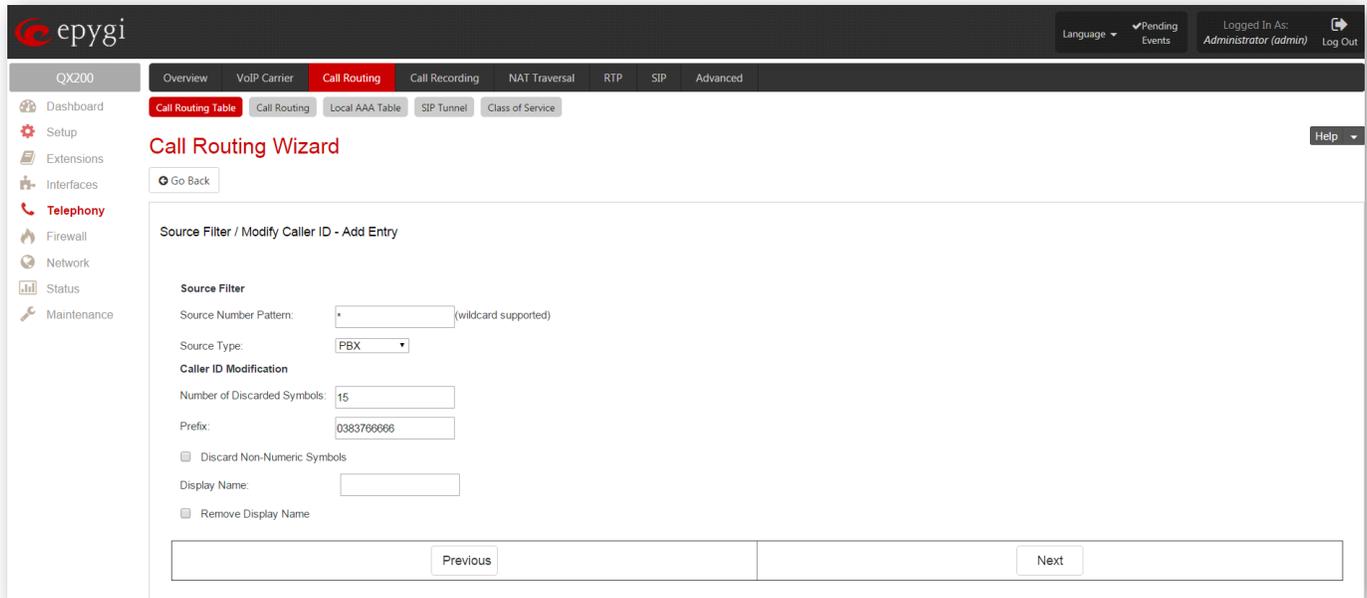


Figure 2: Call Routing Wizard - page 2

Fill the following fields on the third page of the wizard:

1. **Source Number Patter** - * for any source call pattern (could be more restrictive if needed)
2. **Source Type** – PBX, this is important to help secure the QX
3. **Number of Discarded Symbols** - 15
4. **Prefix** – 0383766666 (Figure 3)

Last two entries ensure that the address presented in the invite to **MyNetFone** can be authenticated with a DID that is leased to that service. If we did not Discard Digits from the original caller ID then the extension ID calling would be included. i.e. if extension 182 was calling an external number then the from CLI would include 182.



Source Filter / Modify Caller ID - Add Entry

Source Filter

Source Number Pattern: * (wildcard supported)

Source Type: PBX

Caller ID Modification

Number of Discarded Symbols: 15

Prefix: 0383766666

Discard Non-Numeric Symbols

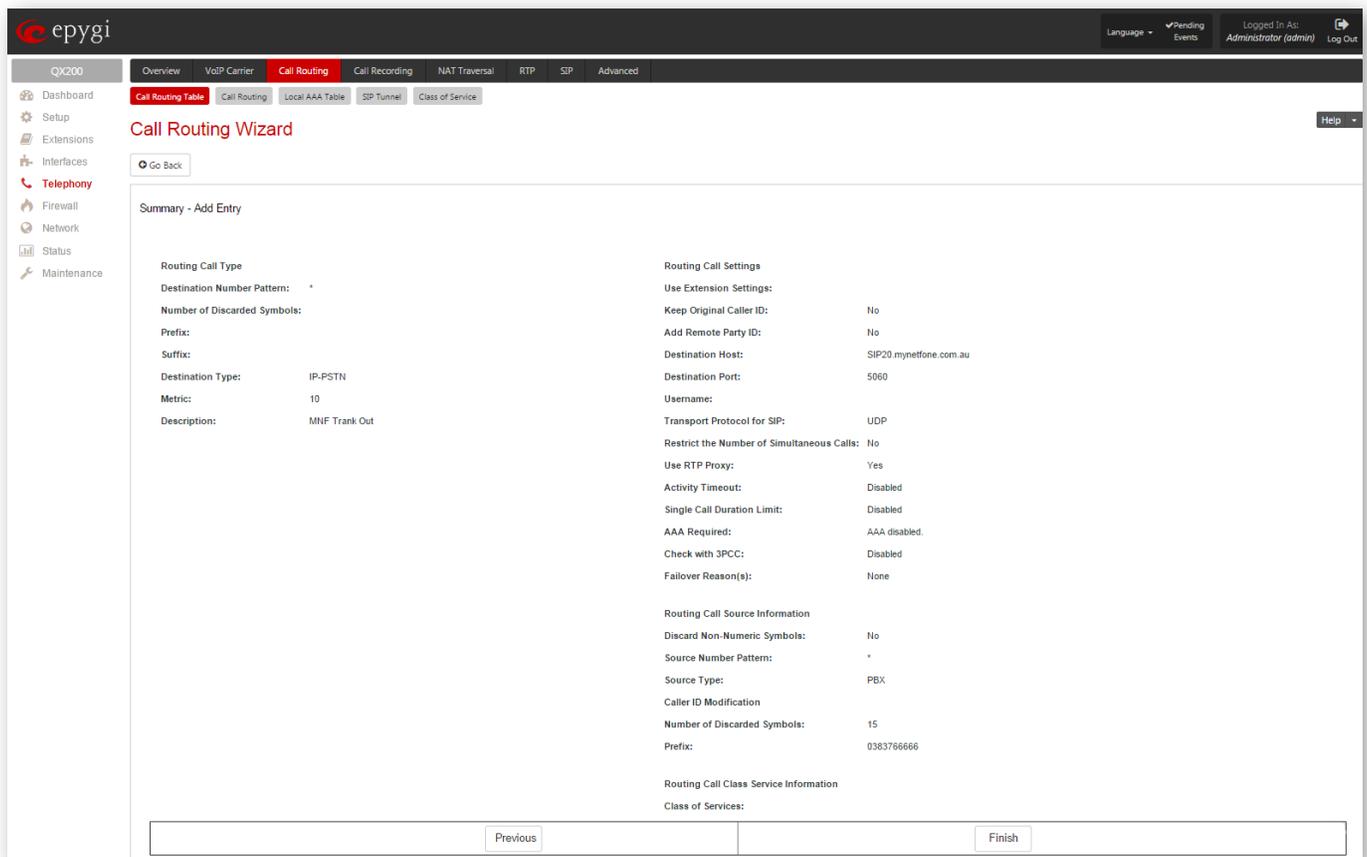
Display Name:

Remove Display Name

Previous Next

Figure 3: Call Routing Wizard - page 3

Summary of the **Call Routing** entry will appear on the forth page of the wizard.



Summary - Add Entry

<p>Routing Call Type</p> <p>Destination Number Pattern: *</p> <p>Number of Discarded Symbols:</p> <p>Prefix:</p> <p>Suffix:</p> <p>Destination Type: IP-PSTN</p> <p>Metric: 10</p> <p>Description: MNF Trunk Out</p>	<p>Routing Call Settings</p> <p>Use Extension Settings:</p> <p>Keep Original Caller ID: No</p> <p>Add Remote Party ID: No</p> <p>Destination Host: SIP20.mynetfone.com.au</p> <p>Destination Port: 5060</p> <p>Username:</p> <p>Transport Protocol for SIP: UDP</p> <p>Restrict the Number of Simultaneous Calls: No</p> <p>Use RTP Proxy: Yes</p> <p>Activity Timeout: Disabled</p> <p>Single Call Duration Limit: Disabled</p> <p>AAA Required: AAA disabled.</p> <p>Check with 3PCC: Disabled</p> <p>Failover Reason(s): None</p>
<p>Routing Call Source Information</p> <p>Discard Non-Numeric Symbols: No</p> <p>Source Number Pattern: *</p> <p>Source Type: PBX</p> <p>Caller ID Modification</p> <p>Number of Discarded Symbols: 15</p> <p>Prefix: 0383766666</p>	<p>Routing Call Class Service Information</p> <p>Class of Services:</p>

Previous Finish

Figure 4: Call Routing Wizard - page 4

5 Configuration for Receiving Inbound Calls from MyNetFone

1. Go to the **Telephony**→**Call Routing**→**Call Routing Table** page.
2. Press the **Add** button to start the **Call Routing Wizard** to create a new routing rule.
3. Fill the below presented entries to map the **MyNetFone** DID to an internal Extension (Figure 5). This could be the system Auto Attendant or as shown in the case below “*83766666” will result at “00” Auto Attendant. More complex inbound routes could be performed including multiple inbound routes to map more than one DID’s to more than one internal QX extension.

Example to support 100 number block.

DID range supported on Trunk = 03 8376 6600 - 03 8376 6699

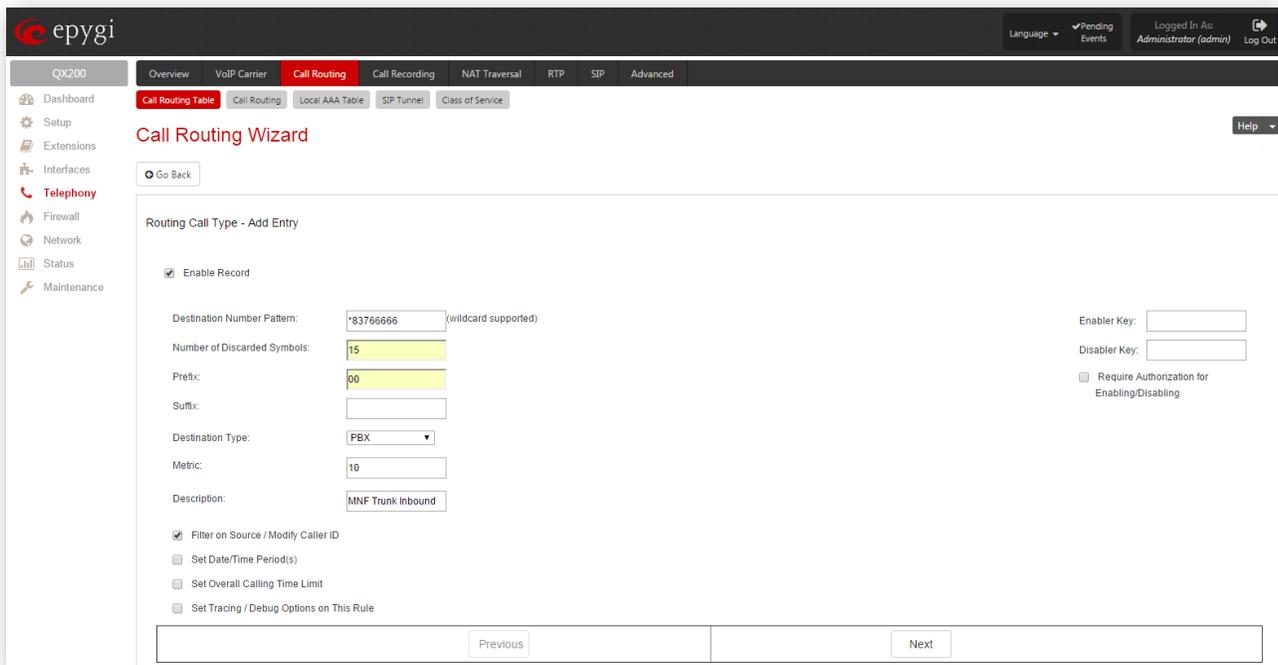
QX Extensions = 100-199

Destination Number Pattern = “03837666??”

Number of Discarded Symbols = “8”

Prefix = “1”

Would map “03 8376 6687” to PBX Extension “187”



The screenshot shows the Epygi web interface for configuring a call routing rule. The main content area is titled "Call Routing Wizard" and "Routing Call Type - Add Entry". The form includes the following fields and options:

- Enable Record
- Destination Number Pattern: (wildcard supported)
- Number of Discarded Symbols:
- Prefix:
- Suffix:
- Destination Type:
- Metric:
- Description:
- Enabler Key:
- Disabler Key:
- Require Authorization for Enabling/Disabling
- Filter on Source / Modify Caller ID
 - Set Date/Time Period(s)
 - Set Overall Calling Time Limit
 - Set Tracing / Debug Options on This Rule

Navigation buttons "Previous" and "Next" are located at the bottom of the form.

Figure 5: Call Routing Wizard - page 1

The second page of the wizard has no settings required.

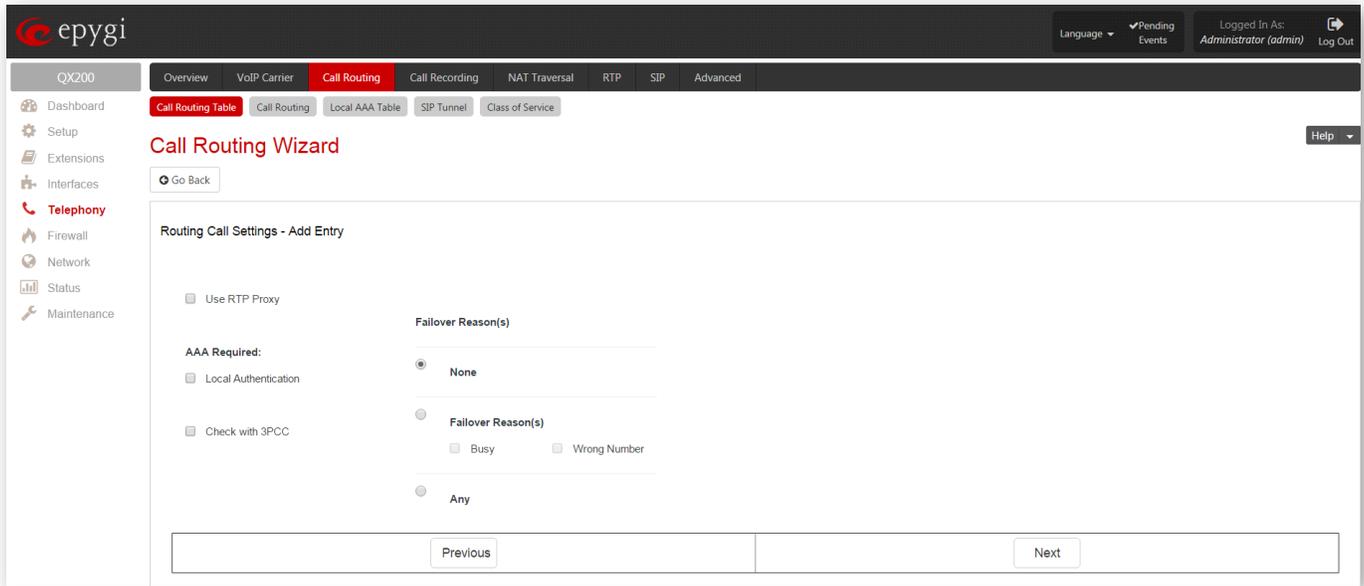


Figure 6: Call Routing Wizard - page 2

The **Source Type** field could be used to filter only for external SIP calls (but **Any** can be defaulted to). This allows any calls to the destination *83766666 from any source.

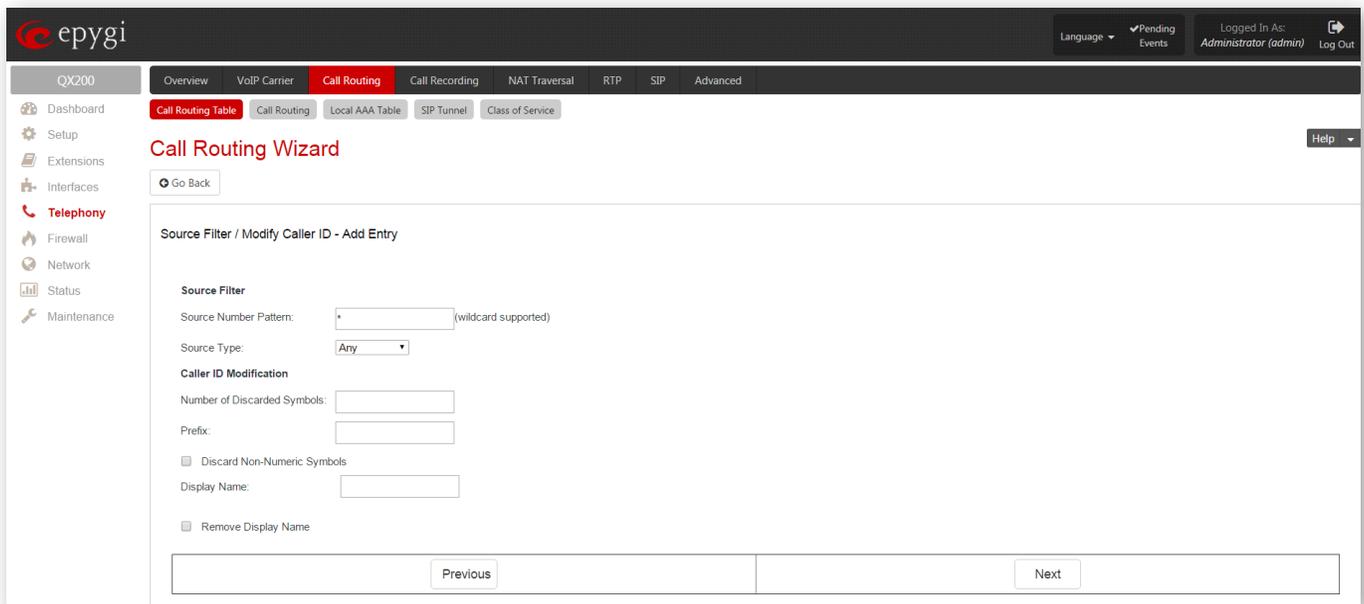


Figure 7: Call Routing Wizard - page 3

Summary of the **Call Routing** entry will appear on the forth page.

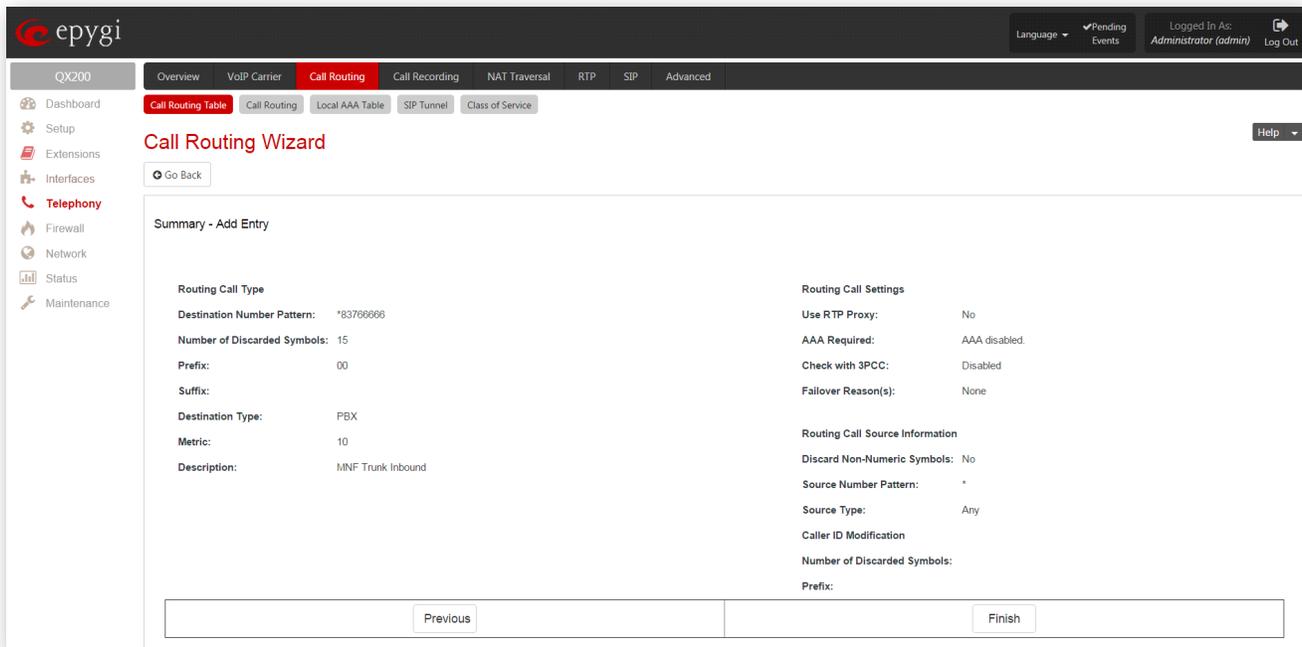


Figure 8: Call Routing Wizard - page 4

That should complete the requirements to allow Inbound and Outbound SIP trunk Calls with MyNetFone. Trouble shooting the configuration can be done from the Call History page and the “\systemlogs.cgi” page (needs to be appended to end of web management URL) Select the “SIP User Agent” report – and refer section below on correct SIP invite format.

6 Important Security Considerations

MyNetFone would recommend customers consult their hardware supplier for advice on securing their QX from fraudulent external abuse. Our advice on Call Routing Configuration does not include advise on ensuring the QX is secure, as this is outside the scope of supply of our services.

7 Expected SIP behavior

In order for the customers QX/gateway to function properly, **MyNetFone** is expecting standards based SIP behavior as described in the sections below.

7.1 Packet Examples

The illustration below shows a standard “INVITE” SIP Packet that has been sent from QX connected to a **MyNetFone** trunk. This particular outgoing call attempt will successfully authenticate with **MyNetFone**.

Please Note: Transpose the correct SIPXX address into the examples shown below based on the advice provided by MyNetFone.

7.2 Header Specifics

VIA: SIP/2.0/UDP 115.187.240.24:5060

In the VIA field, “115.187.240.24” is the Public IP address of your Internet Connection. This section of the field must display the Public IP Address of the Internet Connection that your PBX system is using, and not an Internal LAN IP Address. “5060” is the SIP signalling port that will be used to control the call.

CONTACT: <sip:0280049174@115.187.240.24:5060>

In the CONTACT field, “0280049174” is one of the MyNetFone DID’s that has been assigned to your account as part of the MyNetFone SIP Trunking service. “115.187.240.24” is the Public IP address of your Internet Connection. This section of the field must display the Public IP Address of the Internet Connection that your PBX system is using, and not an Internal LAN IP Address.

TO: <sip:0280088000@sip20.mynetfone.com.au

In the TO field, “0280088000” is the number that has been dialled from your PBX. “sip20.mynetfone.com.au” is the Proxy Server for the MyNetFone SIP Trunk Service.

All outgoing calls must be sent to this proxy. The IP address can be used instead of hostname if desired.

The format of [DID]@[Proxy Server] must be used.

FROM: “0280049174”<sip:0280049174@sip20.mynetfone.com.au:5060;

In the FROM field, “0280049174” is one of the MyNetFone DID’s that have been assigned to your account as part of the SIP Trunking service. “sip20.mynetfone.com.au” is the Proxy Server for the MyNetFone SIP Trunk Service. All outgoing calls must be sent to this proxy.

Session Description Protocol

In this portion of the SIP Packet, you will notice that the following 3 fields highlighted; The “Owner/Creator” field and two “Connection Information” fields.

All of these fields must display the Public IP Address of the Internet Connection that your PBX system is using, and not an Internal LAN IP Address.

7.3 NAT and SIP ALG behavior

In some cases the router connecting your PBX back to the MyNetFone Core may modify the presentation of the SIP headers such that MyNetFone cannot identify the originating host. This usually occurs in the NAT or SIP ALG process of the router. In some cases this can be controlled within the router to ensure valid modifiers are applied (or not applied as required).

An example of this would be the IP Address in the “CONTACT” or the “Session Description Protocol” field/s. With the LAN IP address being used in the “CONTACT” or “Session Description Protocol” field/s, MyNetFone may send some SIP Packets and/or RTP Traffic to the LAN IP Address (instead of the IP Address from which our core network receives the invite).

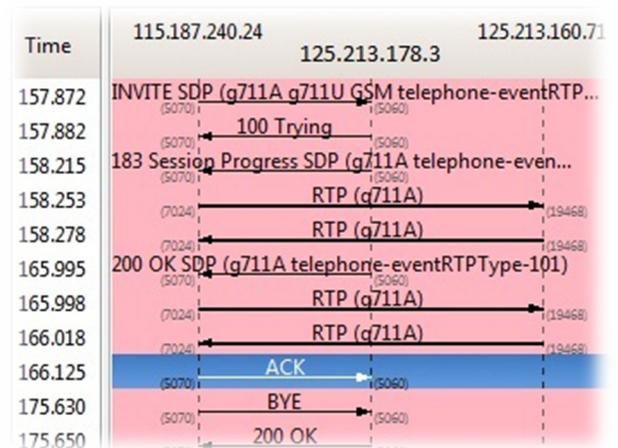
Some examples of possible issues caused by such a scenario include (but are not limited to):

- Outgoing Calls being rejected during the call setup phase
- Incoming Call failure, or abnormal behaviour
- SIP and/or RTP Traffic being sent to the wrong destination.

7.4 Protocol Inspection

Below is an overview of a successful outbound call from the Protocol Analysis tool “Wireshark”. The network traffic of the outbound call was captured and Wireshark was used for analysis. A breakdown of the outbound call is below:

1. The “INVITE SDP” Packet is sent from the PBX system to MyNetFone.
This is to request that an outgoing call be made.
4. The “100 Trying” Packet is sent from MyNetFone to the PBX system.
This is to inform the PBX that the call attempt is being processed.
5. The “183 Session Progress SDP” Packet is sent from MyNetFone to the PBX system.
This is to inform the PBX that the other party is currently ringing the relevant destination.
6. The “RTP (g711A)” packet is then sent from the PBX system to MyNetFone. This begins the audio stream between the PBX to MyNetFone.
7. The “RTP (g711A)” packet is then sent from the MyNetFone to the PBX system. This begins the return audio stream between MyNetFone & the PBX system.
8. The “200 OK SDP” Packet is sent from MyNetFone to the PBX system.
This is to inform the PBX that the call has been answered by the relevant destination.
9. The “RTP (g711A)” packet is then sent from the PBX system to MyNetFone. Please note this is the exact same stream continuing from above, this is the way Wireshark displays the call progression in this scenario.
10. The “RTP (g711A)” packet is then sent from the MyNetFone to the PBX system. Please note this is the exact same stream continuing from above, this is the way Wireshark displays the call progression in this scenario.
11. The “ACK” Packet is sent from the PBX system to MyNetFone. This is to inform MyNetFone that the PBX has successfully received the previous “200 OK SDP” Packet.
12. The “BYE” Packet is sent from the PBX system to MyNetFone.
This is to inform MyNetFone that the PBX has chosen to disconnect the call.
13. The “200 OK” Packet is sent from MyNetFone to the PBX system.
This is to inform the PBX that MyNetFone has successfully received the previous “BYE” Packet, and that the call will now be disconnected.



Depending on the abilities and/or features of your Router or Gateway device, it may not be possible to view SIP Packets as they are being sent to MyNetFone. Modification of SIP Packets may occur as they pass through your Router or Gateway device, which would result in different data being displayed from the LAN-Side compared to the WAN-Side.

For further technical assistance, please contact MyNetFone Business Support on 1300 984 238. Alternatively, you can send a request for technical assistance to tasgovtincidents@mynetfone.com.au.

8 Example of a Successful Invite out from Epygi QX

Environment Variables:

Example Destination Number = Mobile 0435 911 111

Example DID On MNF Trunk = 03 8376 6666

Process to Extract Log:

SIP Messages from web management Page : “\systemlogs.cgi”

(needs to be appended to end of web management URL)

Select the “SIP User Agent” report.

```
***** SIP message buffer start INVITE sip:0435911111@SIP20.mynetfone.com.au
SIP/2.0
Via: SIP/2.0/UDP 131.131.199.142:5060;rport;branch=z9hG4bKEPSVBUS650d2cc7-7472-4bf7-a449-58b84fcedbd6
To: <sip:0435911111@SIP20.mynetfone.com.au>
From: <sip:0383766666@131.131.199.142>;tag=13870384014d69053a-7797-4cba-a9f4-1b1a98c92ee2
CSeq: 474 INVITE
Call-ID: 48879631226708331_992caadc-1ef8-494d-9ff7-5f29050e9292@localhost
Allow: INVITE, ACK, CANCEL, BYE, OPTIONS, INFO, SUBSCRIBE, NOTIFY, REFER, MESSAGE, UPDATE
Contact: <sip:0383766666@131.131.199.142:5060>
Content-Type: application/sdp
Supported: replaces, norefersub
User-Agent: Epygi Quadro SIP User Agent/v5.2.3 (MACRO-IPPBX)
Max-Forwards: 70
Content-Length: 364

v=0
o=- 247 789 IN IP4 131.131.199.142
s=-
c=IN IP4 131.131.199.142
t=0 0
m=audio 7020 RTP/AVP 0 8 18 96 97 2 98 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:96 G726-16/8000
a=rtpmap:97 G726-24/8000
a=rtpmap:2 G726-32/8000
a=rtpmap:98 G726-40/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
***** SIP message buffer start *****
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 131.131.199.142:5060;rport;branch=z9hG4bKEPSVBUS650d2cc7-7472-4bf7-a449-58b84fcedbd6
To: <sip:0435911111@SIP20.mynetfone.com.au>
From: <sip:0383766666@131.131.199.142>;tag=13870384014d69053a-7797-4cba-a9f4-
```

1b1a98c92ee2

Call-ID: 48879631226708331_992caadc-1ef8-494d-9ff7-5f29050e9292@localhost

CSeq: 474 INVITE

Content-Length: 0

9 References

The following documents can be helpful for further configuration of the QX. They can be downloaded from Epygi's WEB portal at www.epygi.com:

- QX IP PBX Manual I – Installation Guide
- QX IP PBX Manual II – Administrator's Guide
- Preventing Unauthorized Calls on the Epygi QX IP PBX
- Web Access Control and Privileges on the Epygi QX IP PBX

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