



Setup Guide

SIP Trunking and Virtual PBX on Asterisk / FreePBX

Contents

1. Introduction	3
2. MyNetFone Support.....	3
3. Virtual PBX setup on FreePBX Distro or Asterisk	3
4.1 Configuration Steps.....	4
4.2 Configuration Using the Command Line and Viewing Trunk Configurations from CLI.....	6
4.3 Quick Reference Guide: Configuring Virtual PBX to your FreePBX Asterisk	7
4. Connecting SIP Trunk to your FreePBX Asterisk Distro.....	8
5.1 Configuration Steps.....	9
5.2 Quick Reference Guide for Connecting SIP Trunk to FreePBX Asterisk.....	12
5. Full Technical Discussion: Connecting a MyNetFone SIP Trunk to Your PBX.....	13
6.1 Authenticating a SIP Trunk.....	13
6.2 Outbound Call Breakdown	15
6. Compatible PBX Systems with MyNetFone SIP Trunking Service	16

1. Introduction

MyNetFone take the complexity out of business telephone systems and make it easy to connect to the world. MyNetFone business customers can choose between a Virtual PBX or SIP Trunking voice services, depending on what equipment they have and how many phone lines are required. Virtual PBX is a hosted phone system, so does not require any on-site PBX equipment, and is ideal for between 2-30 phone lines. Meanwhile, SIP Trunking is a voice service that connects an on-site hardware PBX to the phone network, and is ideal for 30+ phone lines.

The main differences between how the two services connect are:

1. SIP Trunking authenticates with an authorized public IP address (yours) and a unique number provided by MyNetFone.
2. Virtual PBX authenticates with a Username and Password provided by MyNetFone.

MyNetFone offers a range of support options when setting up a SIP Trunking or Virtual PBX service. For tech-savvy clients, we offer assistance over the phone if required. For the less experienced, we offer a fully managed valet service where everything is taken care of for you.

The purpose of this document is to explain several ways a MyNetFone Virtual PBX and SIP Trunking services can be connected to a third party system - we will be using Asterisk as an example.

In section 6, at the end of the document, a complete and detailed technical description is available to assist in connecting to any third-party device.

2. MyNetFone Support

If you would like to speak to MyNetFone Business support specialist, please call us on **1300 887 899** or send an email to business-support@mynetfone.com.au and we will be happy to assist you.

3. Virtual PBX setup on FreePBX Distro or Asterisk

When you sign up for a Virtual PBX service, you can either use the MyNetFone online portal to configure phone system functionality, or can use your own remote PBX system with the MyNetFone supplied Virtual PBX credentials to connect to your own FreePBX Asterisk Distro.

Below is an example of using MyNetFone supplied details to connect to a FreePBX Asterisk system. You can do this at the command line interface (CLI) or use the FreePBX GUI which will write the dial plan(s) for you.

MyNetFone will send an email with the Virtual PBX details similar to the following:

Line Number: 1

MyNetFone VoIP Number: 09xxxxxx (we will use 09112233)

MyNetFone VoIP Password: Vxxxxxxx (we will use VXX451FG)

SIP Proxy: sipXXX.mynetfone.com.au

SIP Port: 5060

Voice Codec: G.729, 40ms packet size

4.1 Configuration Steps

Step 1: From FreePBX GUI select CONNECTIVITY >> TRUNKS.

Step 2: Click on “Add SIP (chan_sip) Trunk” on the left side near the top.

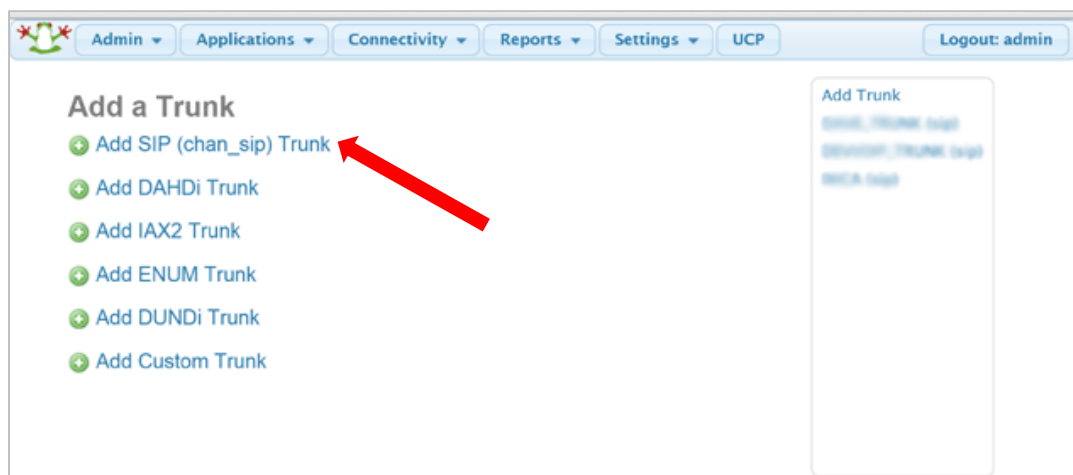


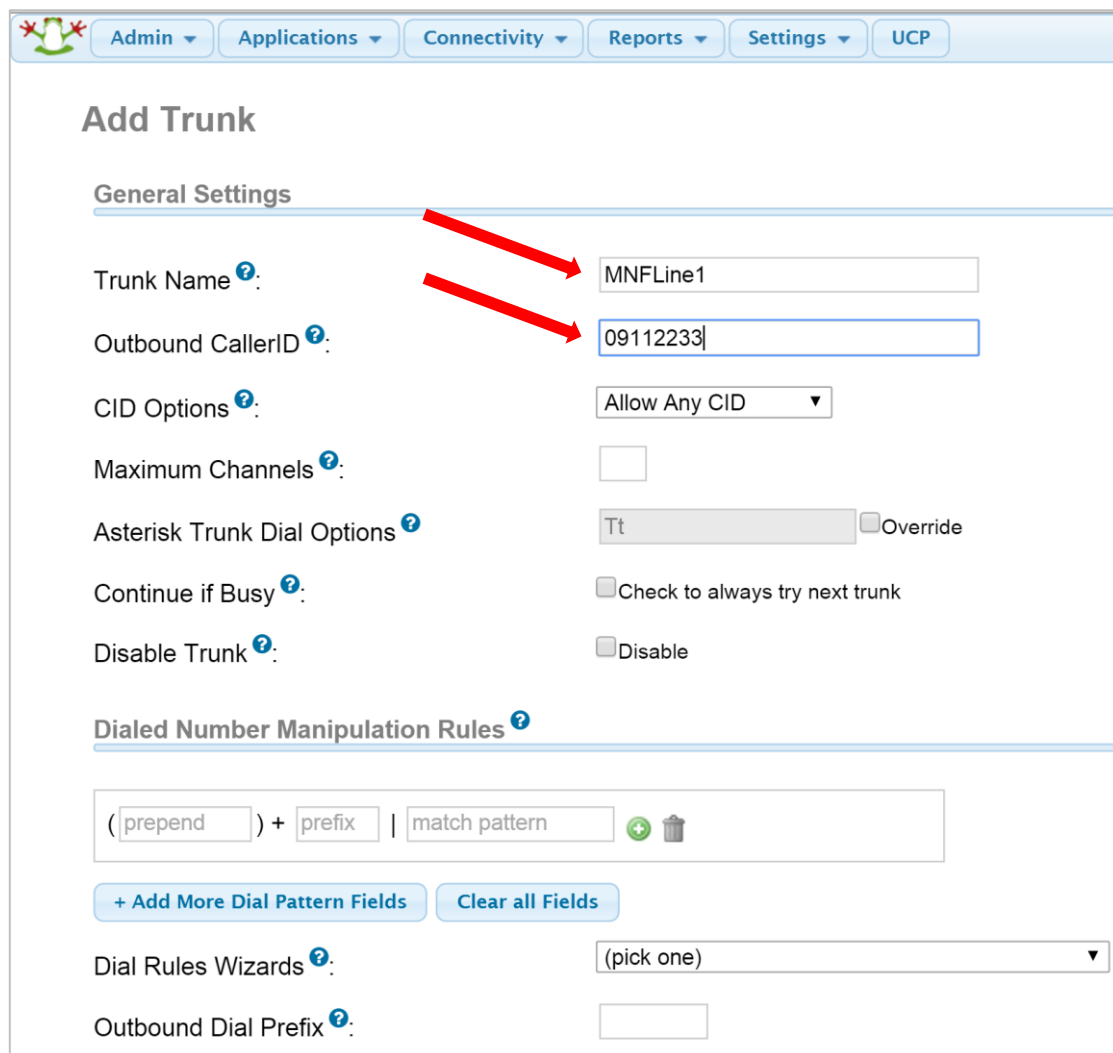
Figure 1

Step 3:

In Trunk Name, give a descriptive name for your new Virtual PBX line. In this case MNFLine1.

In Outbound CallerID, add the number assigned to your VPBX line.

Leave the rest of the settings in Figure in their default settings unless you know exactly what you are doing.



Add Trunk

General Settings

Trunk Name [?]: MNFLine1

Outbound CallerID [?]: 09112233

CID Options [?]: Allow Any CID ▼

Maximum Channels [?]:

Asterisk Trunk Dial Options [?]: Tt ☐ Override

Continue if Busy [?]: ☐ Check to always try next trunk

Disable Trunk [?]: ☐ Disable

Dial Number Manipulation Rules [?]

(prepend)	+	prefix		match pattern
<input type="text"/>		<input type="text"/>		<input type="text"/>

[+ Add More Dial Pattern Fields](#) [Clear all Fields](#)

Dial Rules Wizards [?]: (pick one) ▼

Outbound Dial Prefix [?]:

Figure 2

Step 4:

Scroll down.

In the Trunk Name enter the same descriptive name as in the top portion (in this case MNFLine1).

In the Peer Details enter in the following suggested settings:

```
type=friend
dtmfmode=auto
secret= VXX451FG ... this is the password supplied by MyNetFone
contact=from-trunk
qualify=yes
insecure=port.invite
canreinvite=no
host=sipXXX.mynetfone.com.au ← this is the "SIP proxy" supplied by MyNetFone
```

+ Add More Dial Pattern Fields
Clear all Fields

Dial Rules Wizards [?]: (pick one) ▼

Outbound Dial Prefix [?]:

Outgoing Settings

Trunk Name [?]: MNFLine1

PEER Details [?]:

```

type=friend
dtmfmode=auto
secret=V4XXMT45
context=from-trunk
qualify=yes
insecure=port,invite
canreinvite=no
host=sipXXX.mynetfone.com.au

```

Incoming Settings

Figure 3

MyNetFone strongly recommends you use the above settings for the non- MyNetFone supplied information unless you know exactly what you are doing.

Step 5: Hit “SUBMIT” at the bottom and then “APPLY CONFIG” at the top.

4.2 Configuration Using the Command Line and Viewing Trunk Configurations from CLI

If you do not wish to use the FreePBX GUI to edit or enter your SIP Trunking credentials, you can always modify the SIP.CONF and SIP_ADDITIONAL.CONF file directly – this assumes you will be using FreePBX as a “read-only” application.

To view or edit any file in the FreePBX Distro of Asterisk, you will have to SSH into your Asterisk PBX server, navigate to the SIP_ADDITIONAL.CONF file location and then use your favourite editor. You will see output similar to:

```

type=peer
qualify=yes
nat=auto
host=192.168.1.100
dtmfmode=auto
context=from-psstn-4555-45
canreinvite=no
allow=ulaw,gsm,alaw,g729

[MNFLine1]
type=friend
dtmfmode=auto
secret=VXX451FG
context=from-trunk
qualify=yes
insecure=port,invite
canreinvite=no
host=sipXX.mynetfone.com.au

[secret]
type=friend
dtmfmode=auto
secret=VXX451FG
context=from-trunk
qualify=yes
insecure=port,invite
canreinvite=no
host=sipXX.mynetfone.com.au

```

Figure 4

4.3 Quick Reference Guide: Configuring Virtual PBX to your FreePBX Asterisk

You will receive your Virtual PBX details from MyNetFone via email. You will need the following information:

- Phone Number Associated with your Virtual PBX Account
- SIP Proxy
- Password

Step 1: From FreePBX GUI select CONNECTIVITY >> TRUNKS

Step 2: Click on “Add SIP (chan_sip) Trunk” in the left side near the top

Step 3: In Trunk Name, give a descriptive name for your new VPBX line

Step 4: In Outbound CallerID add the number assigned to your VPBX line

Step 5: In the Trunk Name enter the same descriptive name as in the top

Step 6: In Peer Details enter the following:

```

type=friend
dtmfmode=auto
secret= VXX451FG ← this is the password supplied by MyNetFone

```

```
context=from-trunk
qualify=yes
insecure=port.invite
canreinvite=no
```

```
host=sipXXX.mynetfone.com.au ← this is "SIP proxy" supplied by MyNetFone
```

Step 7: Hit Submit

Step 8: Apply the Configuration

To Configure at the Command Line (or view your configuration) simply:

1. SSH into your Virtual PBX
2. Navigate to the file location which is most likely in:
 /etc/asterisk/sip_additional.conf
3. Open the SIP_ADDITIONAL.CONF file in your favourite editor

Above procedure should work on any Asterisk system, but may require some slight modifications.

4. Connecting SIP Trunk to your FreePBX Asterisk Distro

When you sign up for a MyNetFone SIP Trunk service, you can connect your PBX system directly at your CLI, or you can use a FreePBX Distro of Asterisk. Similarly you could use Trixbox, Elastix or any other Asterisk distro.

Below is an example of using MyNetFone SIP Trunk supplied details to connect to a FreePBX Asterisk system. Using the FreePBX GUI will allow it to write the dial plan(s) for you, and give you full PBX functionality.

MyNetFone will send an email with the SIP Trunk with details similar to the following:

DID's

0288112233

0288112234

Sip Proxy

sipXXX.mynetfone.com.au

The following CODEC's are supported by your SIP trunk.

a=rtpmap:18 G729/8000

a=rtpmap:2 G726-32/8000

a=rtpmap:8 PCMA/8000

a=rtpmap:0 PCMU/8000

5.1 Configuration Steps

Step 1: From FreePBX GUI select CONNECTIVITY >> TRUNKS.

Step 2: Click on “Add SIP (chan_sip) Trunk” in the left side near the top.



Figure 5


Step 3:

In Trunk Name, give a descriptive name for your new SIP Trunk line. In this case MNFLine3.


In Outbound CallerID add the number assigned to your SIP Trunk.

Note: This Outbound CallerID will override all CallerID settings in your Extensions or other applications. If you wish to pass through caller ID, you will need to contact your sales representative to assist you in requesting a pass-through caller ID – although you will need to provide a valid reason for the request.

Leave the rest of the settings in Figure in their default settings unless you know exactly what you are doing.


Admin
Applications
Connectivity
Reports
Settings
UCP

Edit Trunk

 Delete Trunk MNFLine3

In use by 1 route ?

General Settings

Trunk Name ?:	MNFLine3
Outbound CallerID ?:	0288112233<0288112233>
CID Options ?:	Allow Any CID ▾
Maximum Channels ?:	<input type="text"/>
Asterisk Trunk Dial Options ?	Tt <input type="checkbox"/> Override
Continue if Busy ?:	<input type="checkbox"/> Check to always try next trunk
Disable Trunk ?:	<input type="checkbox"/> Disable

Dialed Number Manipulation Rules ?

(prepend) + prefix | match pattern

Figure 6

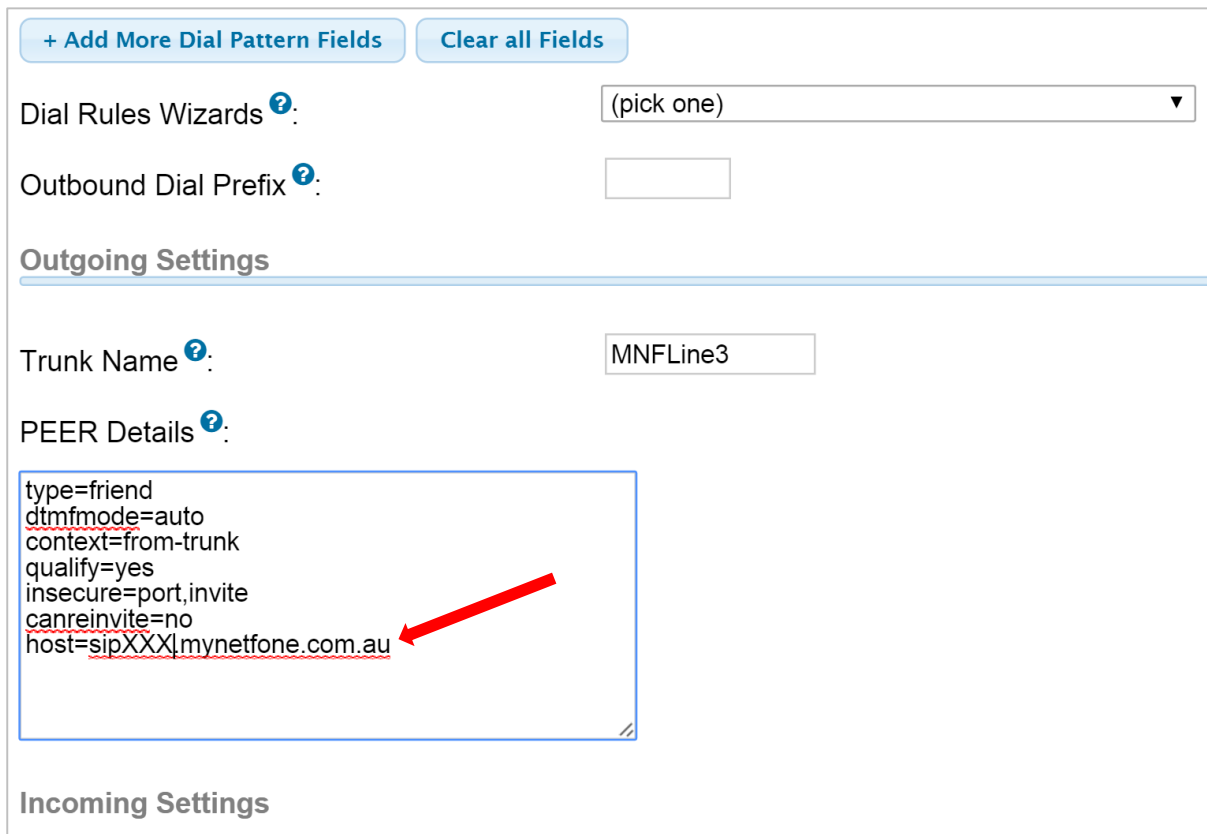
Step 4:

Scroll down.

In the Trunk Name enter the same descriptive name as in the top portion (in this case MNFLine1).

In the Peer Details enter in the following suggested settings:

```
type=friend
dtmfmode=auto
context=from-trunk
qualify=yes
insecure=port,invite
canreinvite=no
host=sipXXX.mynetfone.com.au ← this is the "SIP proxy" supplied by MyNetFone
```



+ Add More Dial Pattern Fields Clear all Fields

Dial Rules Wizards ? (pick one) ▼

Outbound Dial Prefix ?

Outgoing Settings

Trunk Name ? MNFLine3

PEER Details ?

```
type=friend
dtmfmode=auto
context=from-trunk
qualify=yes
insecure=port,invite
canreinvite=no
host=sipXXX|mynetfone.com.au
```

Incoming Settings

Figure 7

MyNetFone strongly recommends you use the above settings for the non- MyNetFone supplied information unless you know exactly what you are doing.

Step 5: Hit "SUBMIT" at the bottom and then "APPLY CONFIG" at the top.

5.2 Quick Reference Guide for Connecting SIP Trunk to FreePBX Asterisk

You will receive your Virtual PBX details from MyNetFone via email. You will need the following information:

- Phone Number(s) Associated with your Virtual PBX Account
- SIP Proxy

Step 1: From FreePBX GUI select CONNECTIVITY >> TRUNKS

Step 2: Click on “Add SIP (chan_sip) Trunk” in the left side near the top

Step 3: In Trunk Name, give a descriptive name for your new SIP Trunk

Step 4: In Outbound CallerID add the number assigned to your SIP Trunk

Step 5: In the Trunk Name enter the same descriptive name as in the top

Step 6: In Peer Details enter the following:

type=friend

dtmfmode=auto

context=from-trunk

qualify=yes

insecure=port.invite

canreinvite=no

host=sipXXX.mynetfone.com.au ← this is “SIP proxy” supplied by MyNetFone

Step 7: Hit Submit

Step 8: Apply the Configuration

To Configure at the Command Line (or view your configuration) simply:

1. SSH into your VPBX
2. Navigate to the file location which is most likely in:
 /etc/asterisk/sip_additional.conf
3. Open the SIP_ADDITIONAL.CONF file in your favourite editor.

Above procedure should work on any Asterisk system, but may require some slight modifications.

5. Full Technical Discussion: Connecting a MyNetFone SIP Trunk to Your PBX

6.1 Authenticating a SIP Trunk

You can connect a MyNetFone SIP Trunking service to virtually any PBX that supports Session Initiated Protocol (SIP). The MyNetFone SIP Trunking service authenticates based on the following information:

1. The Public IP Address of the Internet Connection on your PBX system.
2. The MyNetFone DID that is being sent from your PBX system.
3. Correctly formatting the configuration files of your PBX system to send SIP Packets from PBX system as follows below.

The figure below illustrates a typical “INVITE” SIP Packet that a PBX system sends out. The example below was from a successful outgoing call attempt from a properly authenticated MyNetFone SIP Trunk.

```

Session Initiation Protocol
  Request-Line: INVITE sip:0280088000@sip20.mynetfone.com.au:5060 SIP/2.0
  Message Header
    Via: SIP/2.0/UDP 115.187.240.24:5060;branch=z9hG4bK-d8754z-0b32fb20a71de607-1---d8754z-;rport
    Max-Forwards: 70
    Contact: <sip:0280049174@115.187.240.24:5060>
    To: <sip:0280088000@sip20.mynetfone.com.au:5060>
    From: "0280049174"<sip:0280049174@sip20.mynetfone.com.au:5060>;tag=ef0bd55f
    Call-ID: YTFjMzVIZTR1MDM5OWRlYTlkMjA0NWNhM2RmYzQwNzk.
    CSeq: 1 INVITE
    Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REGISTER, SUBSCRIBE, NOTIFY, REFER, INFO, MESSAGE
    Content-Type: application/sdp
    Supported: replaces
    User-Agent: 3CXPhoneSystem 9.0.13530.0
    Content-Length: 279
    Remote-Party-ID: "0280049174"<sip:0280049174@sip20.mynetfone.com.au:5060>;party=calling
  Message Body
    Session Description Protocol
      Session Description Protocol version (v): 0
      Owner/Creator, Session Id (o): 3cxps 520093696000 200722612225 IN IP4 115.187.240.24
      Session Name (s): 3cxps Audio call
      Connection Information (c): IN IP4 115.187.240.24
      Time Description, active time (t): 0 0
      Media Description, name and address (m): audio 7002 RTP/AVP 8 0 3 101
      Connection Information (c): IN IP4 115.187.240.24
      Media Attribute (a): rtpmap:8 PCMA/8000
      Media Attribute (a): rtpmap:0 PCMU/8000
      Media Attribute (a): rtpmap:3 GSM/8000
      Media Attribute (a): rtpmap:101 telephone-event/8000
      Media Attribute (a): sendrecv
  
```

Figure 8

The items outlined in **RED** are the important parts. The capture above was obtained using a Protocol Analysis tool called “Wireshark.”

Further details or each of the outlined fields are provided below.

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]@[ProxyServer) must be used.

FROM: "0280049174"<sip:0280049174~ip20.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 are 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.

Please Note: It may not be possible to edit all the above fields within the PBX system itself. Some of the IP Addresses in the above fields may need to be modified on your Router. Some Routers may be flexible enough to allow modification of Nat Parameters or "**SIP ALG**" behaviour, but some routers may not.

An example of this would be the IP Address in the "**CONTACT**" or the "**Session Description Protocol**" fields. With the LAN IP address being used in the "**CONTACT**" or "**Session Description Protocol**" fields, MyNetFone may send some SIP Packets and/or RTP Traffic to the LAN IP Address (instead of the Public IP Address of your Internet Connection).

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.

6.2 Outbound Call Breakdown

The network traffic of the outbound call was captured and Wireshark was used for analysis.

A breakdown of the outbound call is below:

- a. The **"INVITE SDP"** Packet is sent from the PBX system to MyNetFone. This is to request that an outgoing call be made.
- b. 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.
- c. 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.
- d. The **"RTP (g711A)"** packet is then sent from the PBX system to MyNetFone.

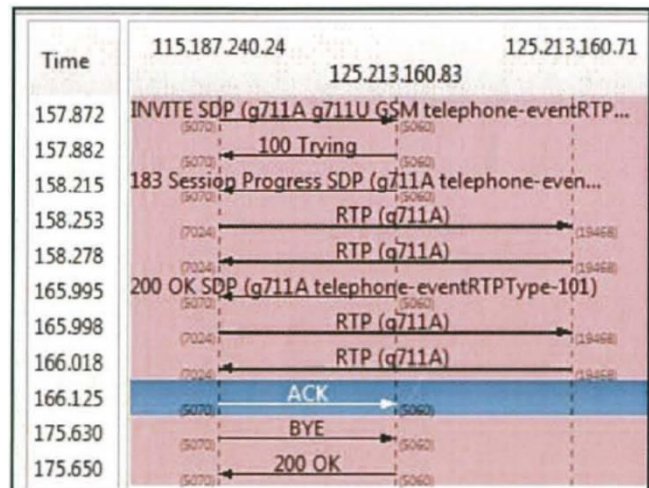


Figure 9

- This begins the audio stream between the PBX to MyNetFone.
- e. S. 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.
- f. 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.
- g. 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.*
- h. The **"RTP (g711A)"** packet is then sent from the MyNetFone to the PBX system.
- i. *Please note this is the exact some stream continuing from above, this is the way Wireshark displays the coli progression in this scenario.*
- j. 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.
- k. 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.
- l. 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.

In this case, MyNetFone is able to capture the SIP and RTP Traffic that is sent/received to and from your Public IP Address for troubleshooting purposes.

For further technical assistance, please contact MyNetFone Business Support on **1300 887 899**.

Alternatively, you can email a request for technical assistance to → business-support@mynetfone.com.au

6. Compatible PBX Systems with MyNetFone SIP Trunking Service

Below is an interoperable and certified hardware list of vendor products with the MyNetFone SIP Trunking Service.

Vendor	Model(s)	Interoperable	Certified
Alcatel-Lucent	Omni PCX Office	✓	✗
Asterisk/TrixBox/FreePBX		✓	✗
Avaya	IP Office	✓	✗
Cisco	Call Manager, Call Manager Express	✓	✗
Cisco	Unified Communications 500 Series	✓	✓
Epygi	2x, 4x, 16x, 4Li, 6L	✓	✓
Epygi	BRI ISDN, E1 Gateway	✓	✓
Grandstream		✗	✗
LG	Aria 24IPE, Aria 130, Aria 300, Aria 600	✗	✗
Mitel		✗	✗
NEC	Univerge IPX	✗	✗
Panasonic	KX-TDE	✓	✓
Quintum	Tenor	✓	✗
Samsung	OfficeServ 7100, 7200	✓	✓
Splicecom	Maximizer 5100	✓	✗
Toshiba	Strata CIX40, CIX100, CIX200, CIX670/CIX1200	✓	✓
Uniden		✗	✗
Zultys	MX 250	✓	✗

If your PBX system is not listed in the table above that does not mean it will not work with the MyNetFone SIP Trunking service. If your PBX system follows standard SIP protocol then your system should work – however, an interop would need to be done to verify compatibility.