

How to Configure the Fonality trixbox CE for use with Integra Telecom SIP Solutions

Overview: This document provides a reference for configuration of the Fonality trixbox CE IP PBX to connect to Integra Telecom SIP Trunks. The document covers basic setup and required steps for interoperability with Integra Telecom only. Values used are for example purposes only. Your values will differ!

Hardware and Software: The following hardware and software were employed to test interoperability between the trixbox CE IP PBX and Integra.

Manufacturer	Model	Software Version
Fonality	trixbox CE	2.8.0.1
MetaSwitch	MetaSphere	7.3
Adtran	NV3305	17.09.02

Tested Features: The following is a list of features that were tested.

Feature	Description	Issue (if any)
Basic Call	Making and receiving a call between the IP-PBX and Integra Telecom service provider with both G.711 and G.729 codec.	None
Call Hold	Placing a call in On Hold state and retrieval of a call from same station.	None
Call Transfer	Relocation of an active call from one station to another. Both internal and external transfers were tested.	None
Call Forward	Forwarding of calls from one station to another.	None
Fax	Fax Transmission	T.38 fax not supported

Network Topology: Figure 1 shows how the network was configured for interoperability testing.

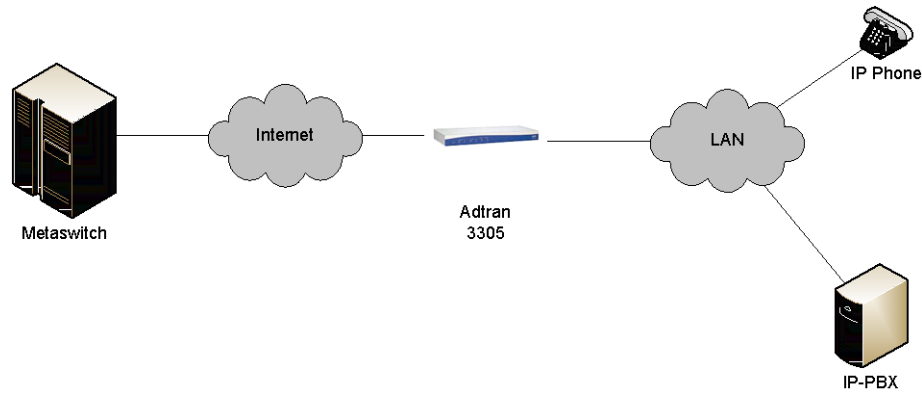


Figure 1: interoperability network diagram

Configuration Notes: This section contains a detailed description of how the Fonality trixbox CE was configured for interoperability testing with Integra Telecom SIP trunks.

Network Requirements: As in any VoIP deployment there must be adequate bandwidth to support VoIP traffic. A proper network assessment should be performed prior to any VoIP deployment.

Assumptions:

All SIP Signalling uses UDP on port 5060.

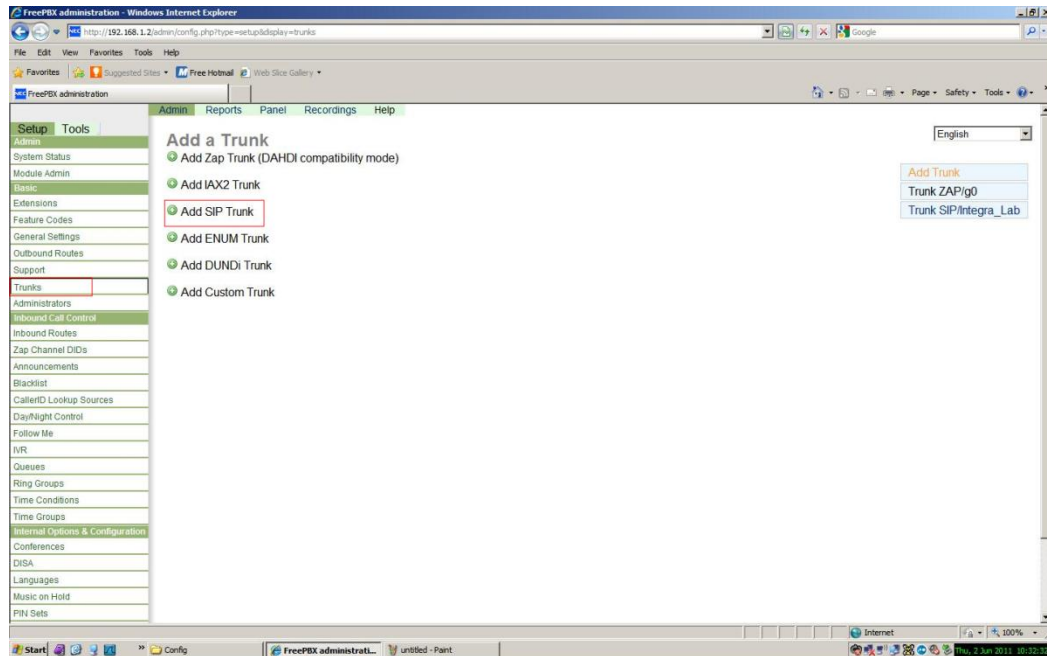
SIP Signalling packets use Differentiated Services Code Point (DSCP) 24.

Real-Time Transport Protocol (RTP) uses DSCP 46.

Licenses: Note that trixbox CE does not include G.729 licenses. They can be purchased through Digium.

SIP Trunk Setup

1. Using the FreePBX web client, select Trunks. On the Add a Trunk page, select Add SIP Trunk.



2. On the Edit SIP Trunk page, enter Outbound Caller ID if required. Enter the Maximum Channels as provided by your Integra Telecom implementation team. Enter a Trunk Name of your choosing. This field is informational only. Enter the User Name as supplied by Integra Telecom in the USER Context field.
3. Complete the Outgoing PEER Details as follows:
 - disallow=all
 - allow=ulaw
 - context=from-trunk
 - insecure=very
 - dtmfmode=auto
 - fromdomain= **<private IP address of Adtran CPE LAN Interface>**
 - fromuser=**<User Name as provided by Integra Telecom>**
 - host=**<URL as provided by Integra Telecom>**
 - canreinvite=update
 - nat=no
 - qualify=yes

type=peer
username=<User Name as provided by Integra Telecom>
secret=<password as supplied by Integra Telecom>
port=5060

Complete the Incoming USER Details as follows:

disallow=all
allow=ulaw
context=from-trunk
insecure=very
dtmfmode=auto
fromdomain=<private IP address of Adtran CPE LAN Interface>
fromuser=<User Name as provided by Integra Telecom>
host=<private IP address of Adtran CPE LAN Interface>
nat=no
qualify=yes
type=peer
username=<User Name as provided by Integra Telecom>
secret=<password as supplied by Integra Telecom>
port=5060

4. Enter the Registration String in the following format:

<User Name>:<Password>@<URL>/<User Name>

For example: 3605551212:secret@sip.integra.net/3605551212

FreePBX administration

Admin Reports Panel Recordings Help

Setup Tools

Admin

- System Status
- Module Admin
- Basic
- Extensions
- Feature Codes
- General Settings
- Outbound Routes
- Support
- Trunks
- Administrators
- Inbound Call Control
- Inbound Routes
- Zap Channel DIDs
- Announcements
- Blacklist
- CallerID Lookup Sources
- Day/Night Control
- Follow Me
- IVR
- Queues
- Ring Groups
- Time Conditions
- Time Groups
- Internal Options & Configuration
- Conferences

Edit SIP Trunk

⊘ Delete Trunk Integra_Lab

In use by 1 route

General Settings

Outbound Caller ID:

Never Override CallerID:

Maximum Channels:

Disable Trunk: Disable Enable

Monitor Trunk Failures: Enable

Outgoing Dial Rules

Dial Rules:

Clean & Remove duplicates

Dial Rules Wizards: (pick one)

Outbound Dial Prefix:

Outgoing Settings

Trunk Name:

PEER Details:

```

disallow=all
allow=ulaw
context=from-trunk
insecure=very
dtmfmode=auto
fromdomain=192.168.1.1
fromuser=3608529765
host=proxy1.integravoip.net
canreinvite=update
nat=no

```

Incoming Settings

USER Context:

USER Details:

```

disallow=all
allow=ulaw
context=from-trunk
insecure=very
dtmfmode=auto
fromdomain=192.168.1.1
fromuser=3608529765
host=192.168.1.1
nat=no
qualify=yes

```

Registration

Register String:

Submit Changes

- DISA
- Languages
- Music on Hold
- PIN Sets
- Paging and Intercom
- Parking Lot
- System Recordings
- VoiceMail Blasting

5. You should now be able to verify successful SIP Trunk Registration on the System Status page.

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Day/Night Control

Follow Me

IVR

Queues

System Status

Notices
No new notifications
[show all](#)

Statistics

Total active calls	0
Internal calls	0
External calls	0
Total active channels	0

Connections

IP Phones Online	3
IP Trunks Online	1
IP Trunk Registrations	1

Uptime

System Uptime: 6 hours, 26 minutes
Asterisk Uptime: 6 hours, 25 minutes
Last Reload: 6 hours, 25 minutes

System Statistics

Processor

Load Average	0.00
CPU	0%

Memory

App Memory	7%
Swap	0%

Disks

/	4%
/boot	20%
/dev/shm	0%

Networks

eth1 receive	0.00 KB/s
eth1 transmit	0.00 KB/s
eth2 receive	0.00 KB/s
eth2 transmit	0.00 KB/s
eth0 receive	0.00 KB/s
eth0 transmit	0.00 KB/s

Extension Setup

1. Browse to Extensions. From Device pulldown select the appropriate type of phone to be added. For SIP Phones select Generic SIP Device. For FXS/ analog ports select Generic ZAP Device. Click Submit.

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Inbound Routes

Zap Channel DIDs

Add an Extension

Please select your Device below then click Submit

Device

Submit

Device

- Generic SIP Device
- Generic SIP Device
- Generic IAX2 Device
- Generic ZAP Device
- Other (Custom) Device

- To add a SIP phone, enter a Display Name, CID Num Alias and SIP Alias to suit your installation. Enter a valid Outbound CID and Emergency CID. Enter the DID that will be directed to this phone (if applicable) in the Add Inbound DID field. This will automatically create the Inbound Route for the DID to extension mapping. Enter the secret to be used to register your SIP Phone with the trixbox CE. Click Submit.

The screenshot displays the configuration interface for Extension 200. The sidebar on the left lists various system settings, with 'Internal Options & Configuration' currently selected. The main configuration area is divided into several sections:

- Display Name:** TestPhoneA
- CID Num Alias:** 3608529765
- SIP Alias:** Test_Phone_A
- Extension Options:**
 - Outbound CID:** 3608529765
 - Ring Time:** Default
 - Call Waiting:** Enable
 - Call Screening:** Disable
 - Emergency CID:** 3608529765
- Assigned DID/CID:**
 - DID Description:** (empty field)
 - Add Inbound DID:** (empty field)
 - Add Inbound CID:** 3608529765

PIN Sets	
Paging and Intercom	
Parking Lot	
System Recordings	
VoiceMail Blasting	

This device uses sip technology.

secret	xxxxxxx
dtmfmode	rfc2833
canreinvite	no
context	from-internal
host	dynamic
type	friend
nat	yes
port	5060
qualify	yes
callgroup	
pickupgroup	
disallow	
allow	
dial	SIP/200
accountcode	
mailbox	200@device
deny	0.0.0.0/0.0.0.0
permit	0.0.0.0/0.0.0.0
Language	

Language Code

- For an analog extension follow the above SIP Phone guidelines with the following exceptions. Enter channel numbers to match the ports on your FXS PCI card. Set context to “from-zaptel”. Ensure signaling is set to “fxs_ks”.

Time Conditions	
Time Groups	
Internal Options & Configuration	
Conferences	
DISA	
Languages	
Music on Hold	
PIN Sets	
Paging and Intercom	
Parking Lot	
System Recordings	
VoiceMail Blasting	

DID Description

Add Inbound DID	
Add Inbound CID	
3608529768 (Fax)	

Device Options

This device uses zap technology. (Via DAHDI compatibility mode)

channel	1&2
context	from-zaptel
immediate	yes
signalling	fxs_ks
echocancel	yes
echocancelwhenbridged	no
echotraining	800
busydetect	no
busycount	7
callprogress	no
dial	ZAP/2
accountcode	
callgroup	
pickupgroup	
mailbox	202@device

ZAP Channel DID Setup

1. If a ZapTEL/ DAHDI analog port is required, select Zap Channel DID's. In the Add Zap Channel options, enter the Channel number and DID associated as entered in the ZapTEL device configuration above and click Submit Change.

Add Zap Channel

Zap Channel DIDs allow you to assign a DID to specific Zap Channels if you have multiple POTS lines that are on a hunt group from your provider. It will be a line that looks like:

```
context = from-zapTEL
```

in your zapata.conf configuration effecting the specified channel(s) for calls.

Add Channel

Channel:

Description:

DID:

Outbound Route Setup

1. Setup outbound dialing rules to meet your system requirements. In the example below, all calls are directed out the SIP Trunks created in the SIP Trunk Setup section above. Enter a Route Name such as Integra SIP Trunks. From the Dial patterns wizards select all of the necessary outbound dialing patterns that should be directed out the Integra SIP Trunks. In the first Trunk Sequence pull-down, select the Sip Trunks created above. Click Submit Changes.

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Queues

Ring Groups

Time Conditions

Time Groups

Add Route

Route Name:

Route Password:

PIN Set:

Emergency Dialing:

Intra Company Route:

Music On Hold?

Dial Patterns

Dial patterns wizards:

Trunk Sequence

(pick one)

Local 7 digit

Local 7/10 digit

Toll-free

Long-distance

International

Information

Emergency

Lookup local prefixes

When configuration is complete on the trixbox CE, be sure to apply the changes and reset services as instructed by the administrative client!