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 retreival 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.

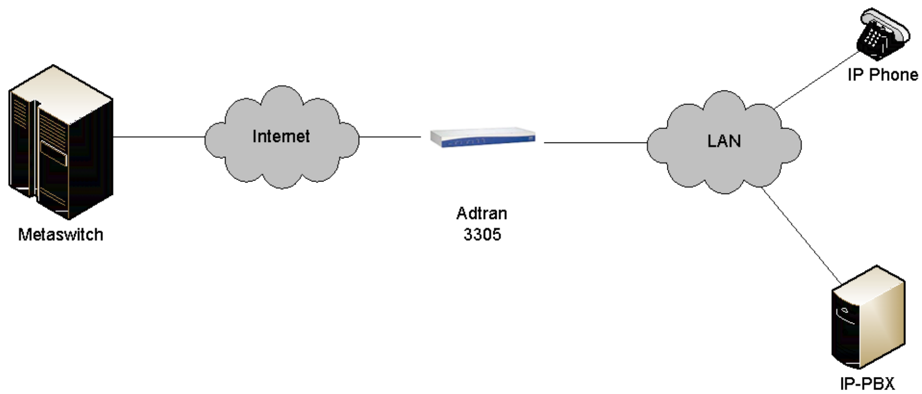


Figure 1: interoperability network diagram

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 assessement should be performed prior to any VoIP deployment.

Assumptions:

All SIP Signalling uses UDP on port 5060.

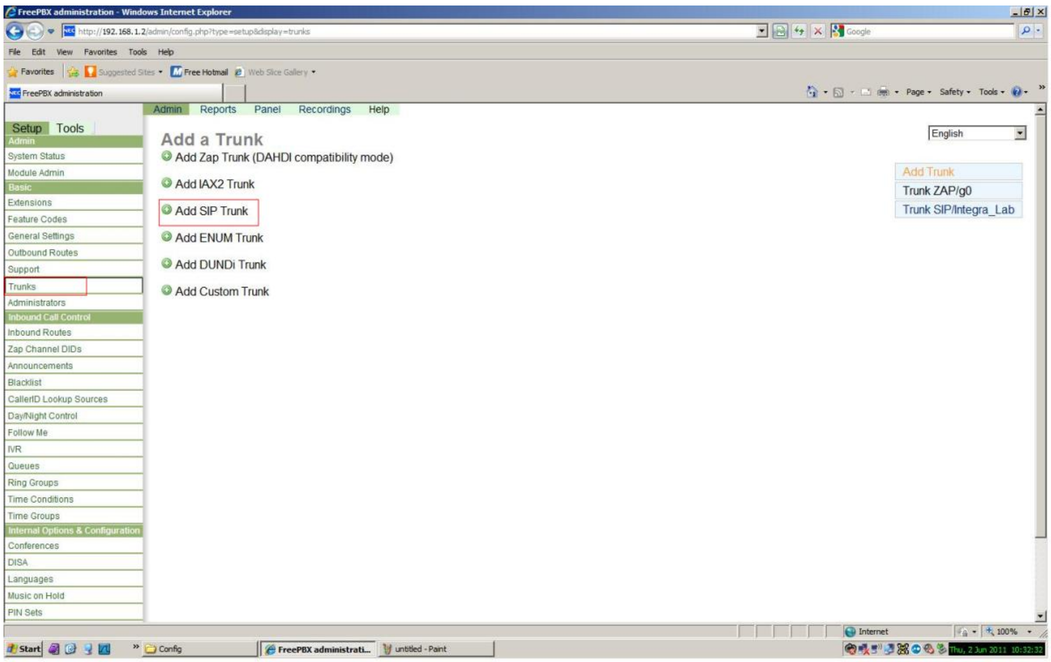
SIP Signalling packets use Differentiated Services Code Point (DSCP) 24.

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

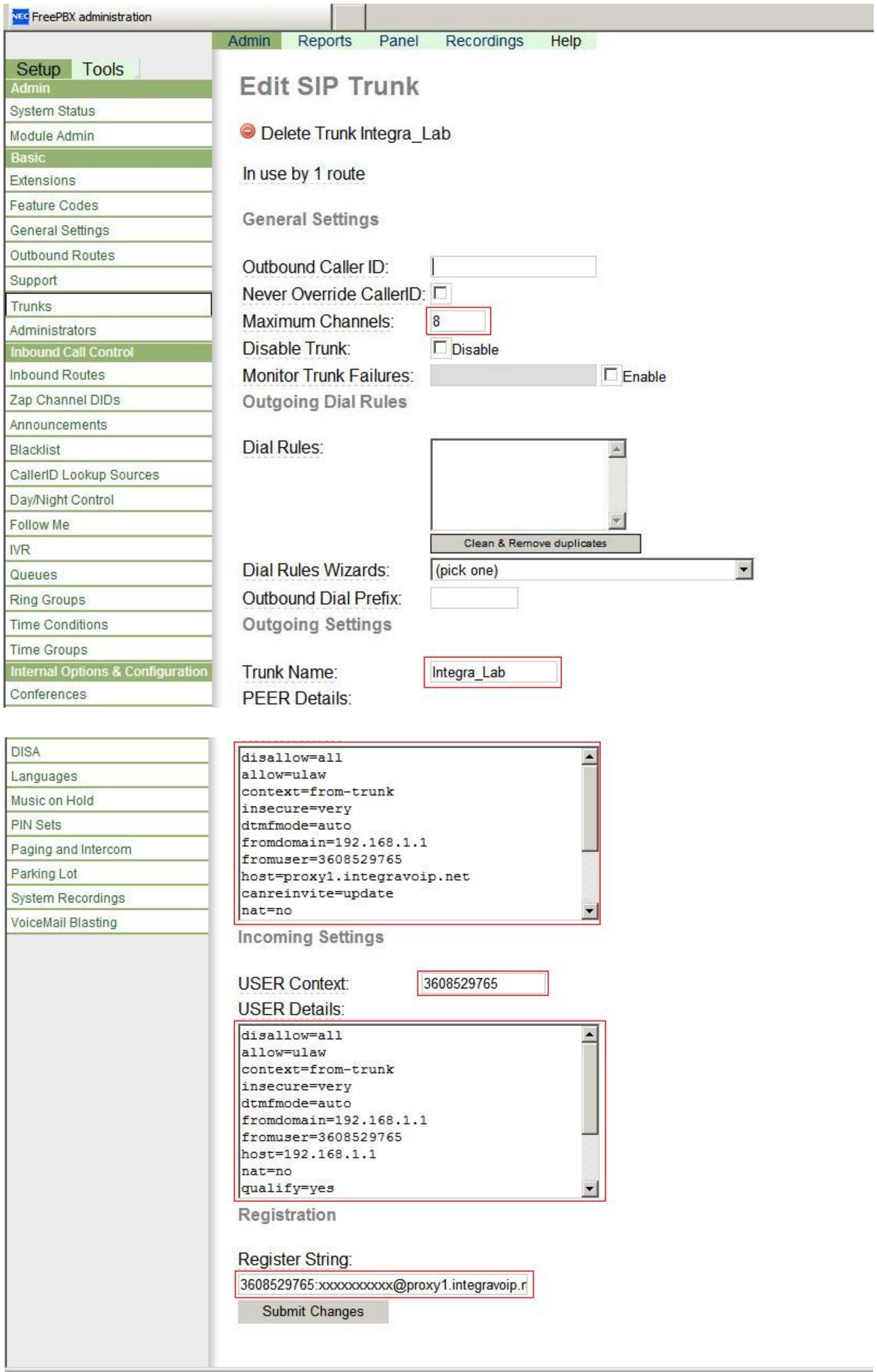
type=peer
username=**<User Name as provided by Integra Telecom>**
secret=**<password as supplied by Integra Telecom>**
port=5060

Complete the Incoming USER Details as follows:

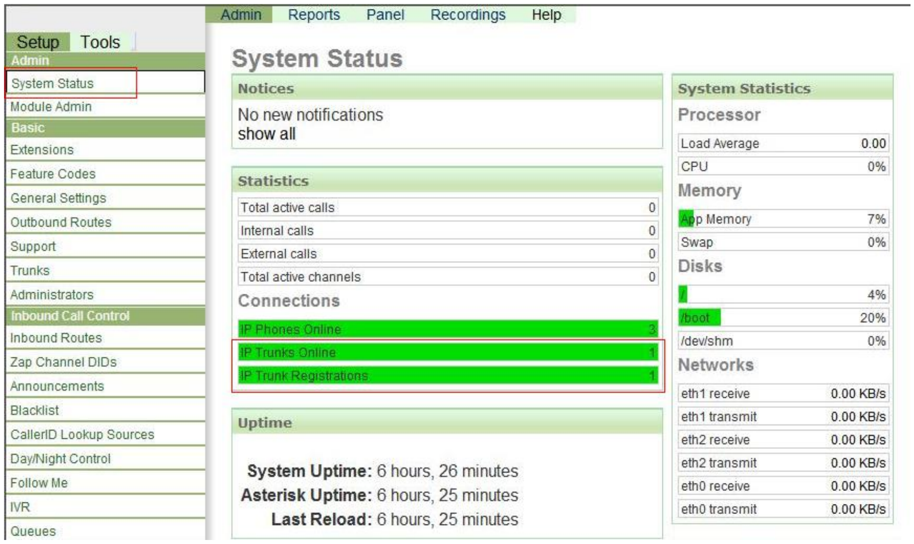
disallow=all
allow=ulaw
context=from-trunk
insecure=very
dtmfmode=auto
fromdomain=**<private IP address of Adtran CPE LAN Interface>**
fromuser=**<User Name as provided by Integra Telecom>**
host=**<private IP address of Adtran CPE LAN Interface>**
nat=no
qualify=yes
type=peer
username=<**User Name as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

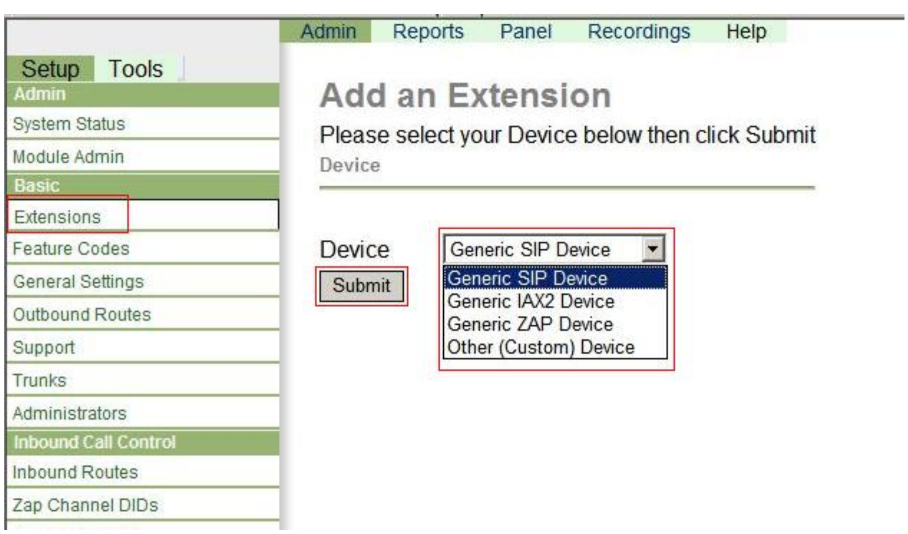


5. You should now be able to verify succesful SIP Trunk Registration on the System Status page.

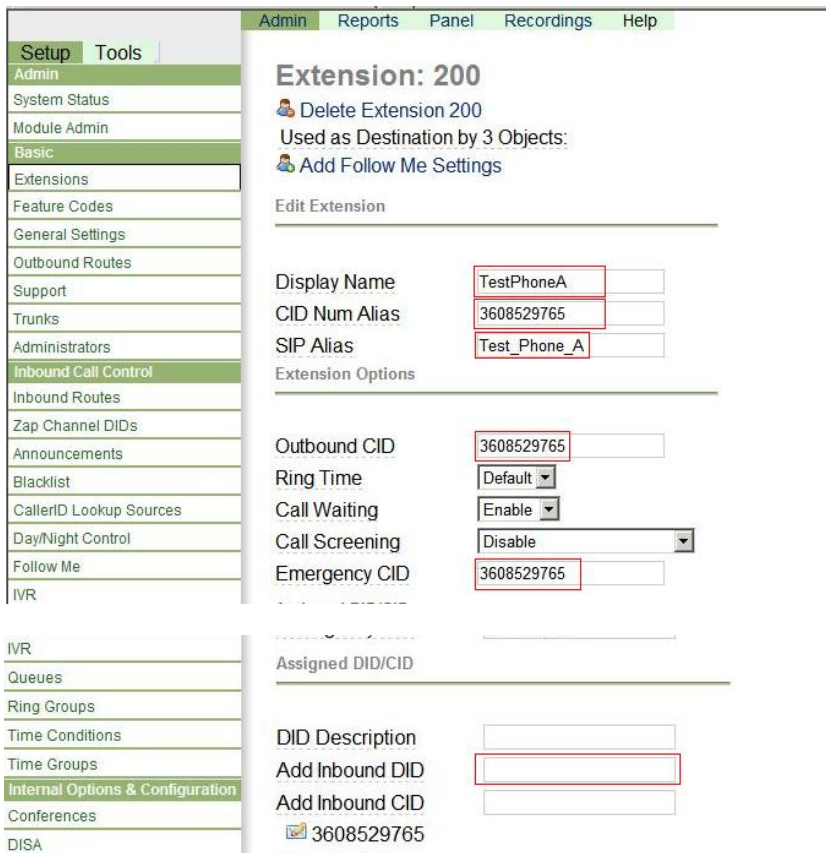


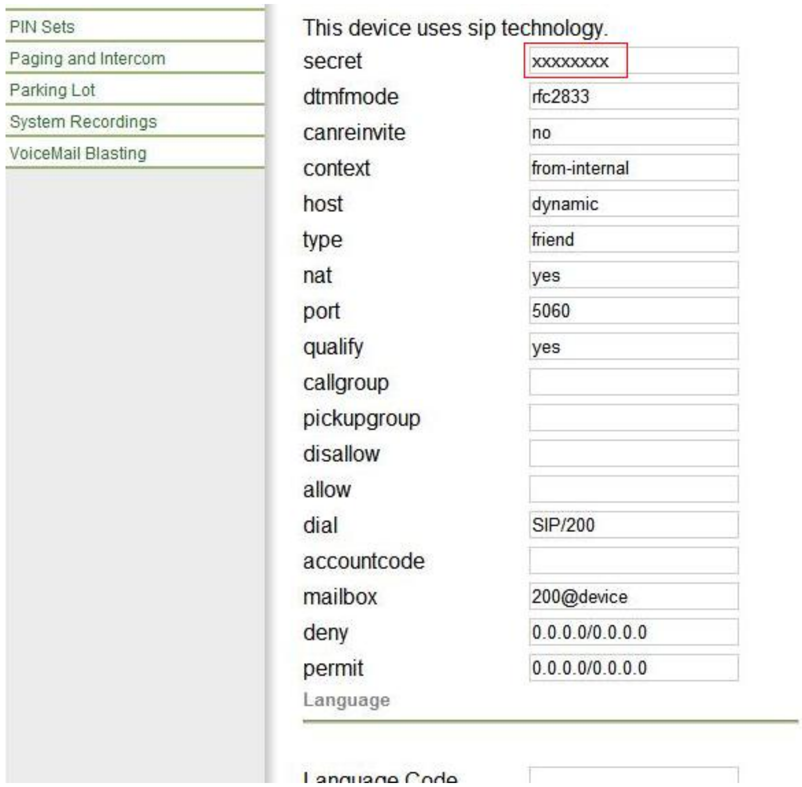
Extension Setup

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.

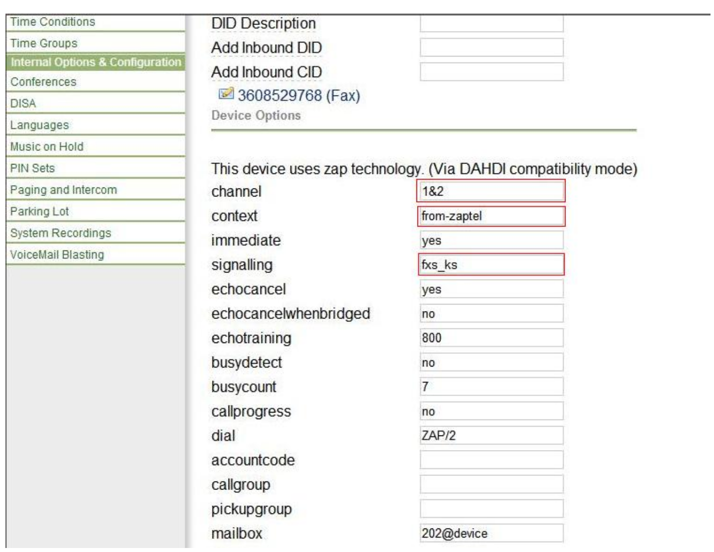


2. To add a SIP phone, enter a Display Name, CID Num Alias and SIP Alias to suit your installation. Enter a valid Outbound CID and Emergency CID. Enter the DID that will be directed to this phone (if applicable) in the Add Inbound DID field. This will automatically create the Inbound Route for the DID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



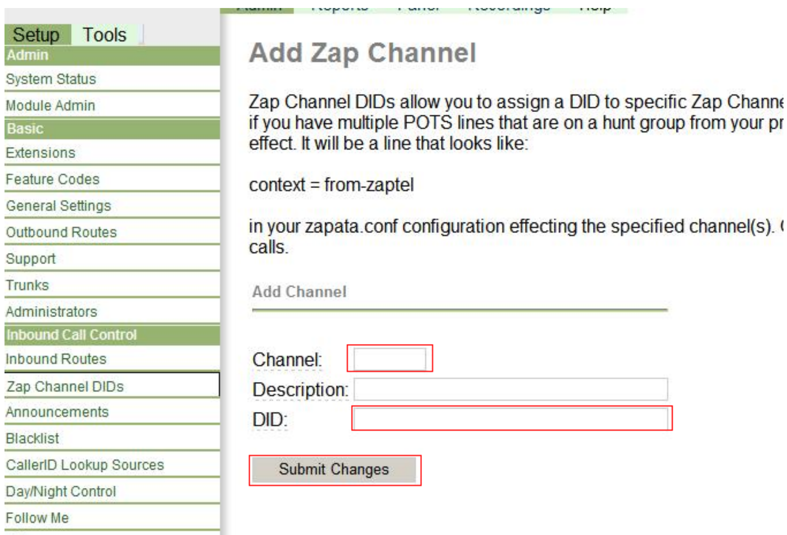


3. For an analog extension follow the above SIP Phone guidelines with the following exceptions. Enter channel numbers to match the ports on your FXS PCI card. Set context to “from-zaptel”. Ensure signaling is set to “fxs_ks”.



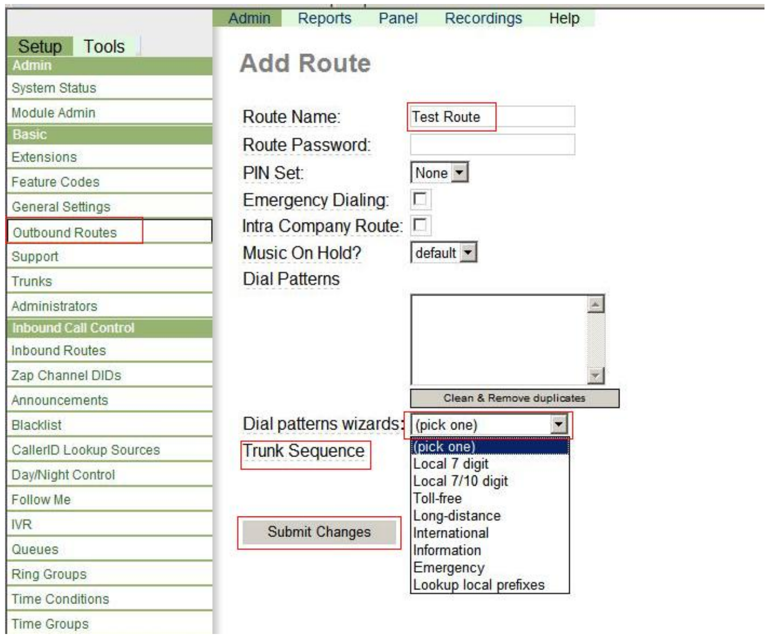
ZAP Channel DID Setup

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.



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.



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