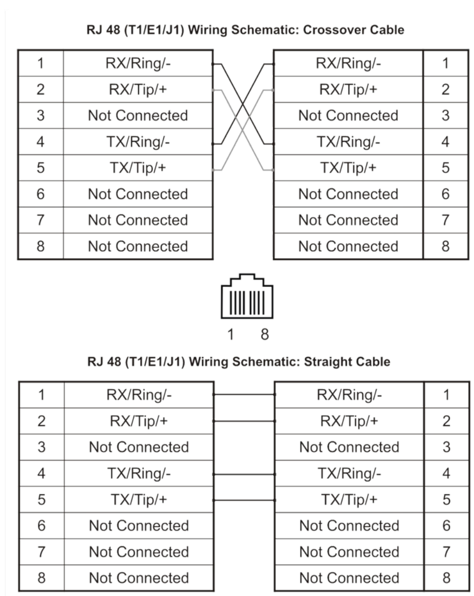
Integration Guide for Azblink SBC and Alcatel-Lucent OXO/OXE

Abstract: This document is provided as reference for the integration between Azblink SBC and Alcatel-Lucent OXO/OXE via T1/E1 link or SIP trunk.

Introduction

It is noted that deploying Azblink SBC will need a **public IP address** facing the Internet and a **private IP address** for the office network. To connect Azblink SBC with Alcatel-Lucent OXO/OXE, T1/E1 link or SIP trunk can be used. If Alcatel-Lucent OXO/OXE is with SIP trunk module, it will need an extra private IP address for itself as well. In this document, the example in the setting for the IP address part should be replaced by your own IP addresses.

For T1/E1 link, Alcatel-Lucent's OXO provides the sockets for the following schemes:



Thus, only straight cable is needed between Azblink SBC and Alcatel-Lucent's OXO for T1/E1 link. For OXE, you have to make cross-wiring cable by yourself.

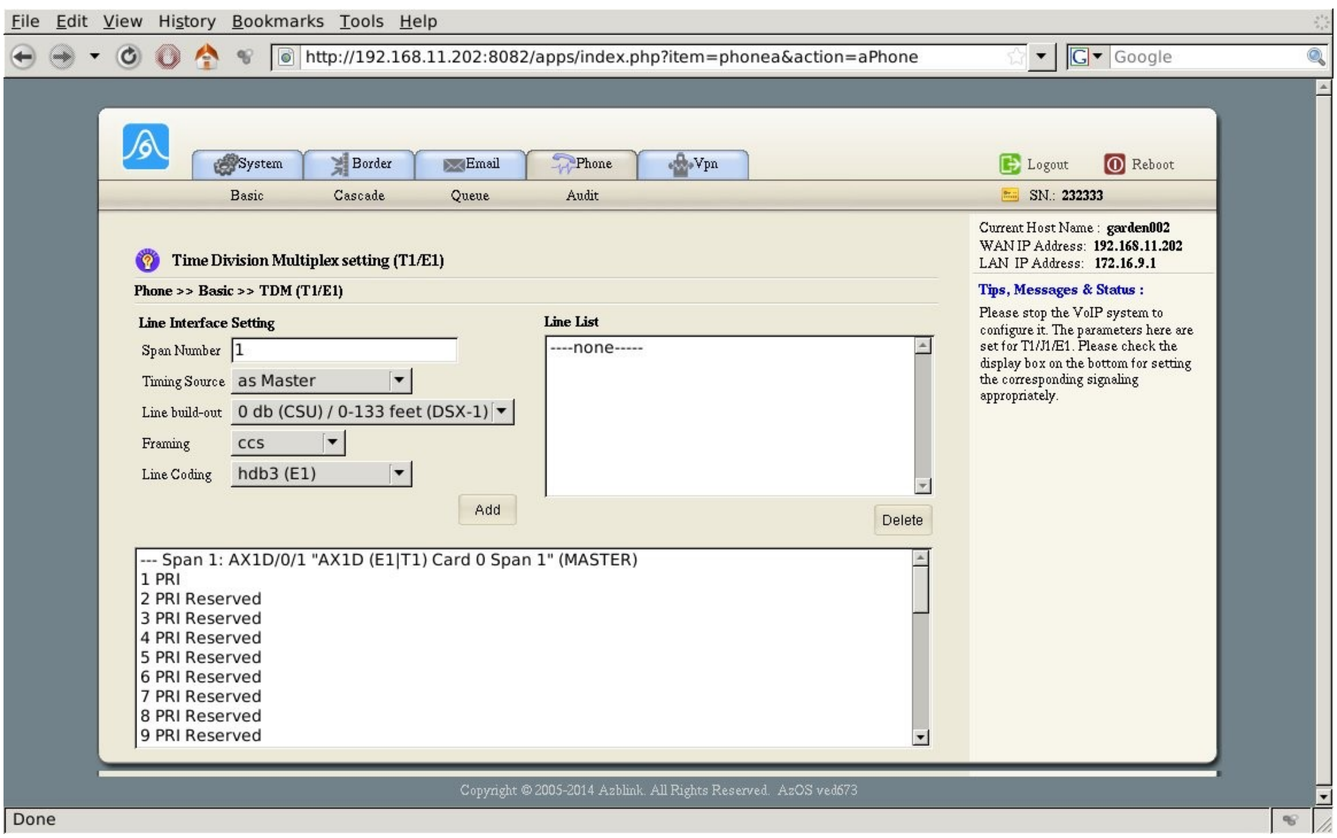
And each of the Alcatel-Lucent's T1/E1 module can have its own **System Clock Timing Source** (BITS clock). Thus, it is flexible to select the system timing source even when multiple T1/E1 modules are present on OXO/OXE system. With the description above in mind, you should adapt the changes to the deployment scenario you encounter.

T1/E1 setting

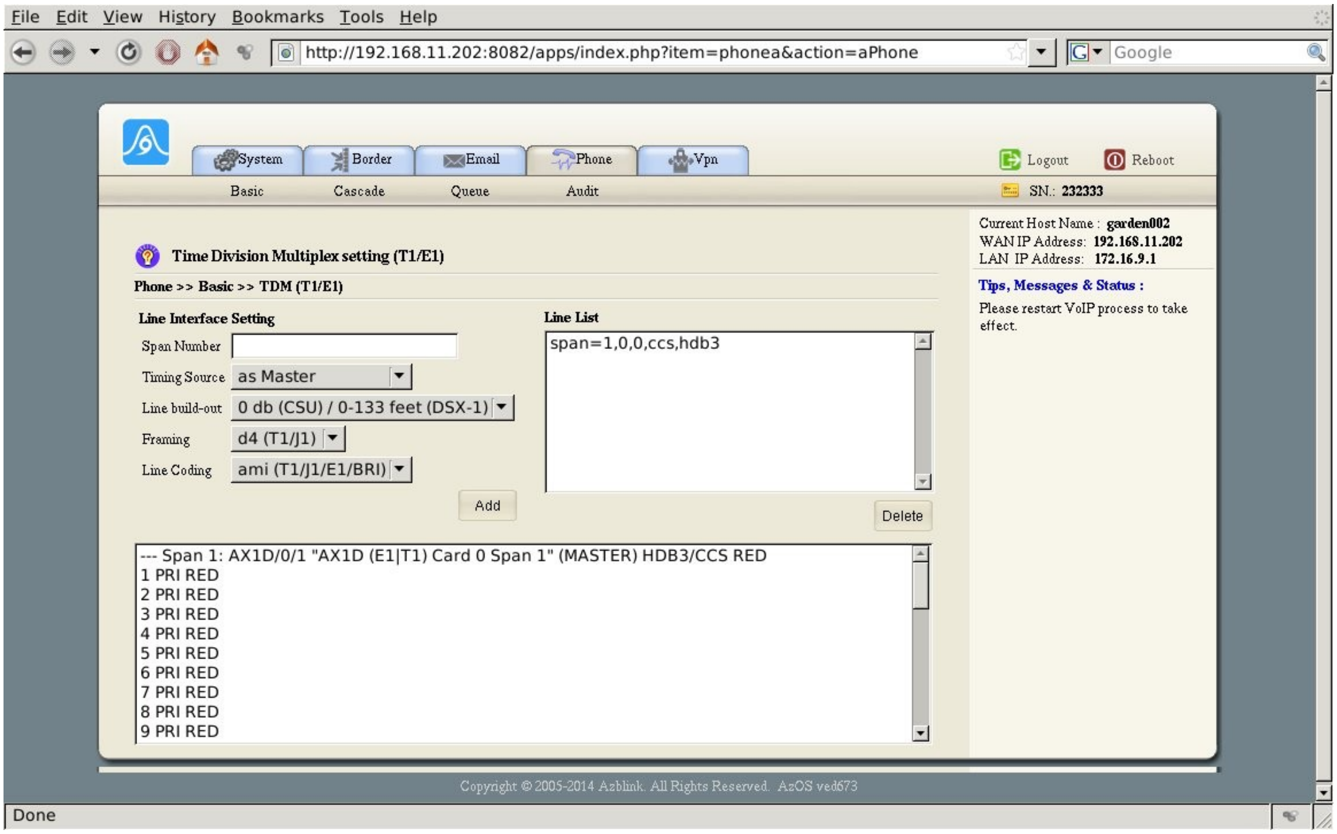
In the following, we assume that Azblink SBC's T1/E1 will be used as the System Clock Timing Source so that OXO/OXE will lock it from the other side. And we intend to run ISDN PRI over E1 on this link; OXO/OXE will act as “PRI NET” and Azblink SBC will act as “PRI CPE”.

Before setting T1/E1 link on Azblink SBC, please stop VoIP process in order for the configuration to go on smoothly.

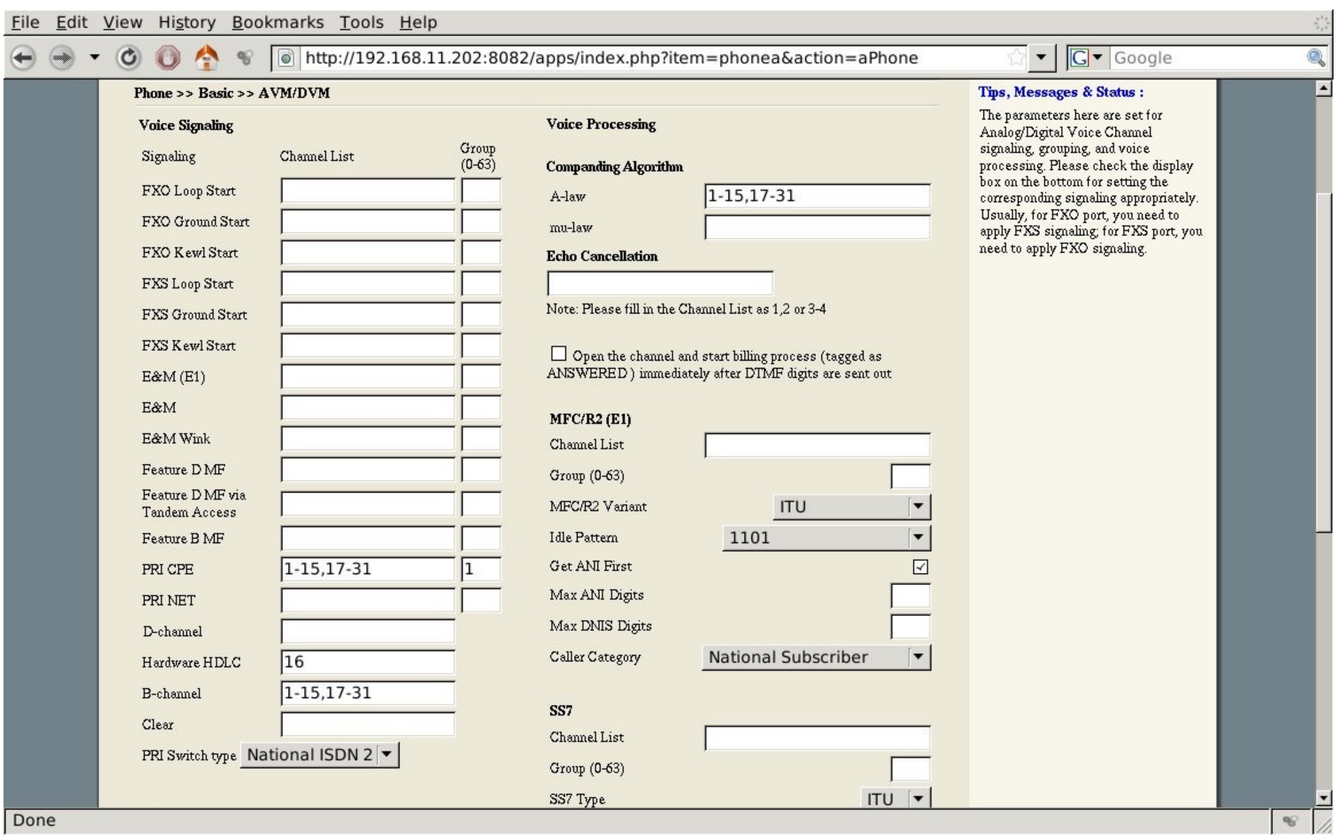
ISDN PRI belongs to CCS (Common Channel Signaling); thus, we start as follows:



After “Add” button is pressed, it takes a long while to load the parameters. After it is completed, you will find “**RED alarm**” is raised if the cable is unplugged:

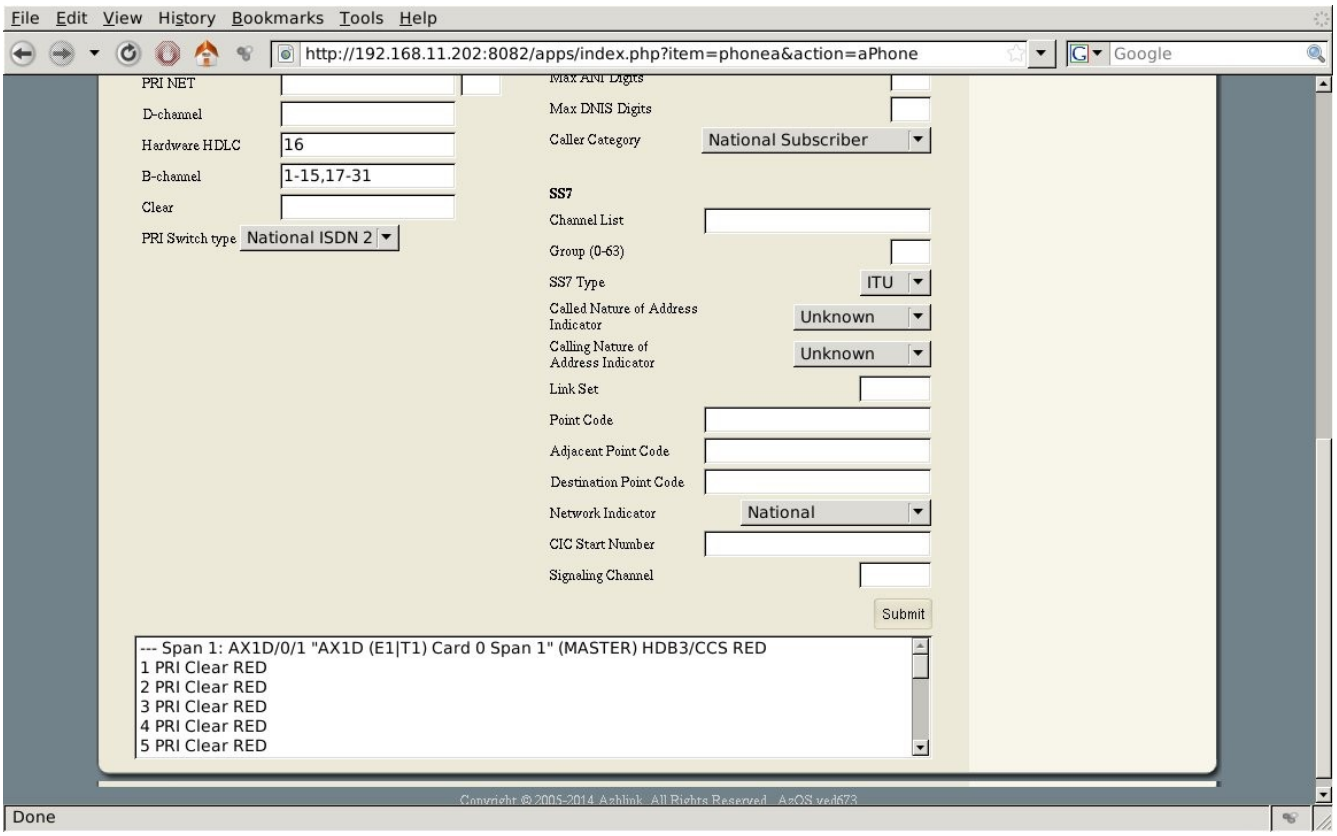


We can move on by ignoring it at this moment.

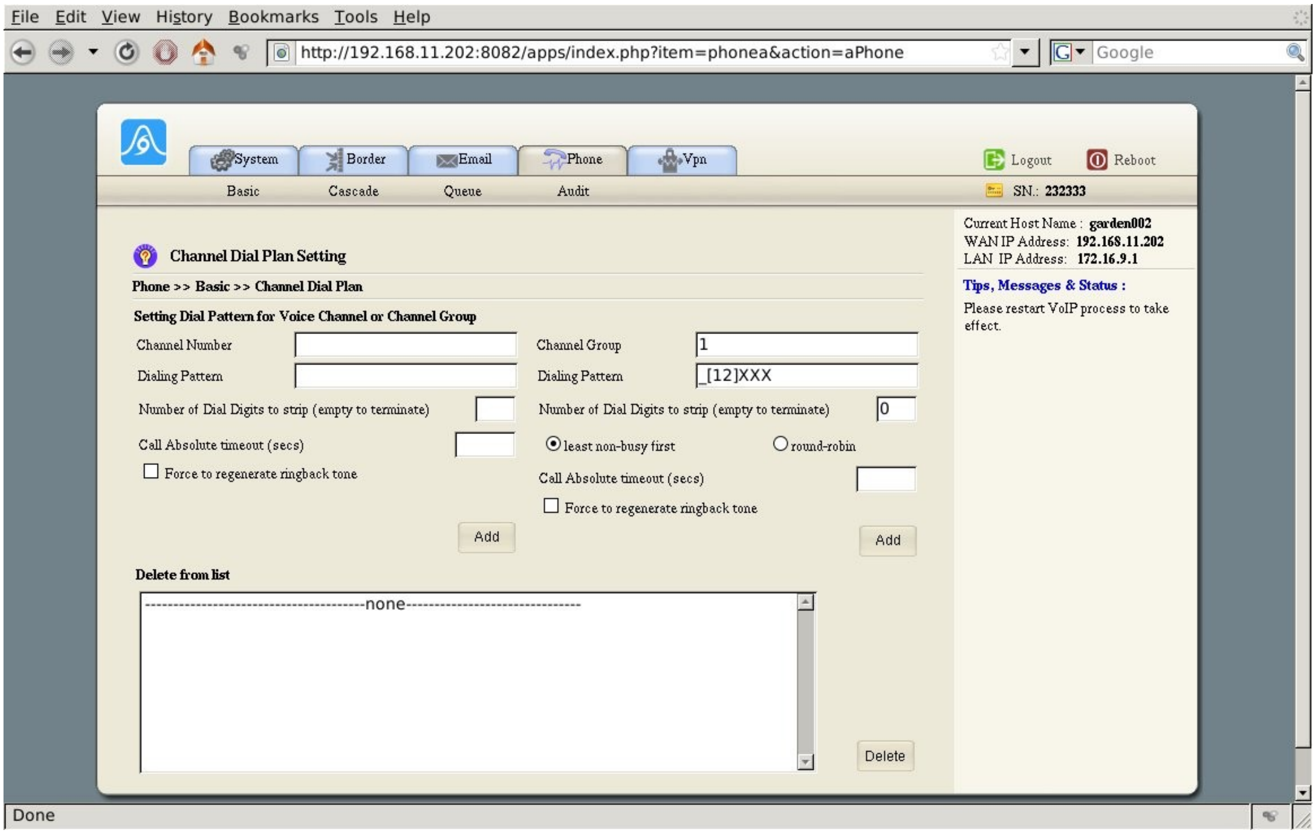


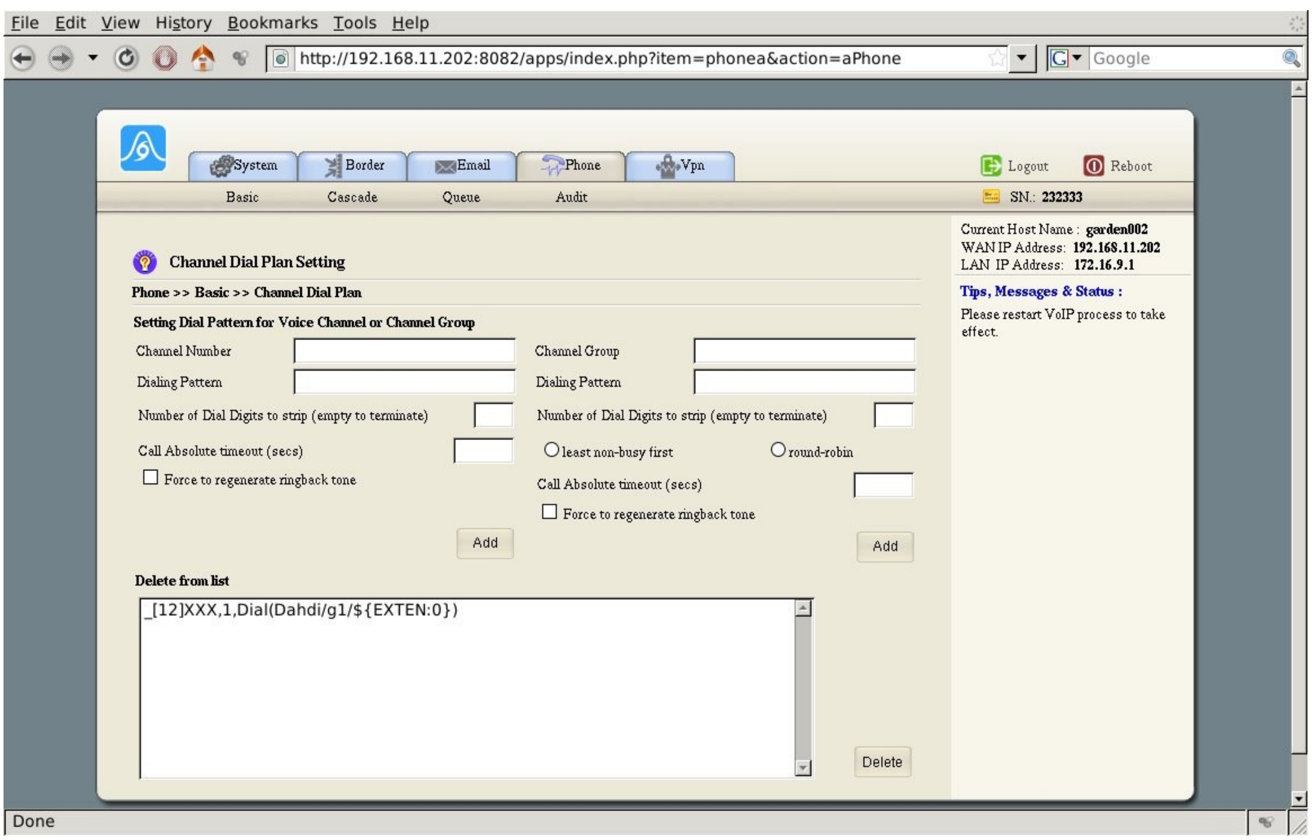
For the signaling part, ISDN PRI is with 30 B-channels plus 1 D-channel. And we let the channels “1-15,17-31” use “**PRI CPE**” on the side of Azblink SBC.

“**Hardware HDLC**” is same as “D-channel” except that it is implemented in hardware driver.   Thus, we set “Channel 16” in “**Hardware HDLC**” and just use “**National ISDN 2**” standard.

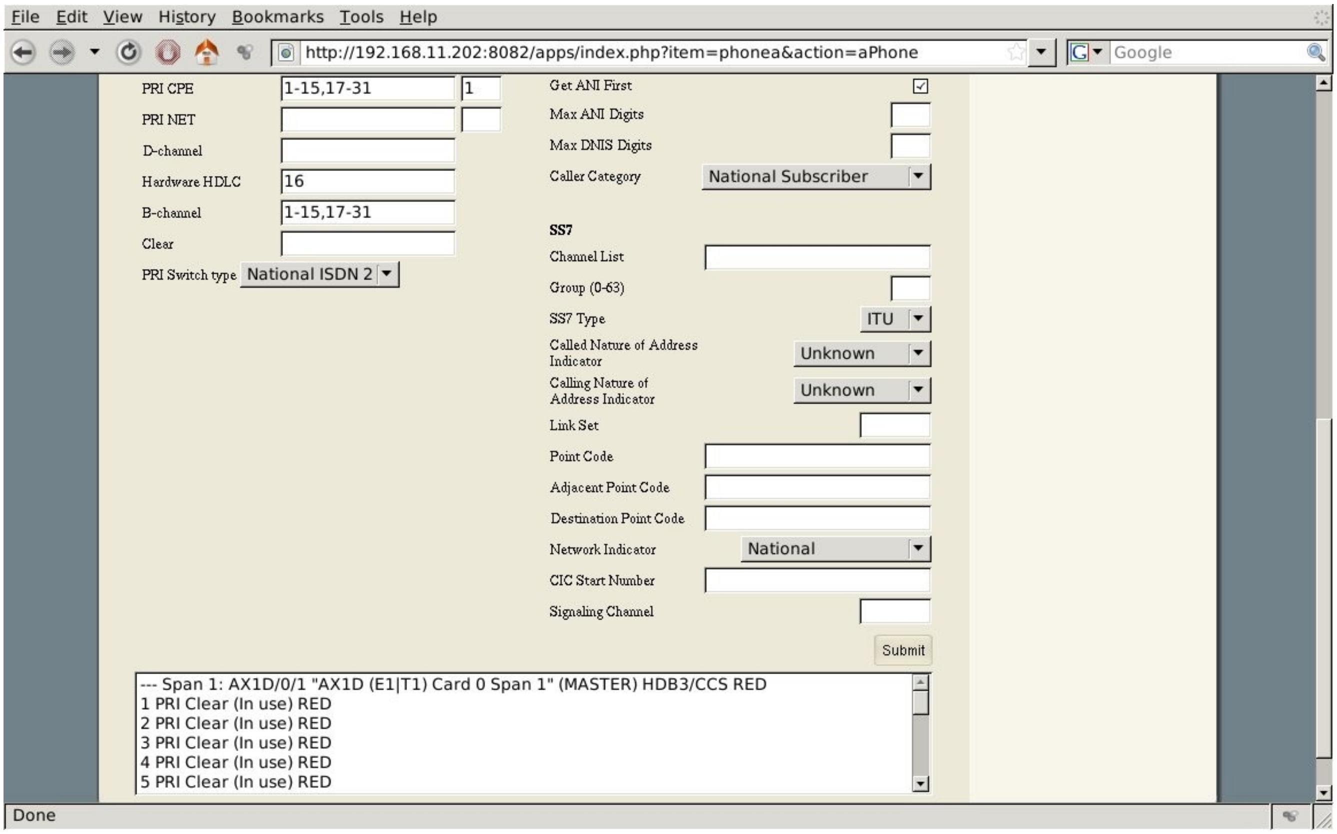


Then, we turn to “Channel Dial Plan”.   Assume the numbers 1XXX and 2XXX will go to the the E1 trunk to the other end; we set as follows:


File Edit View History Bookmarks Tools Help
http://192.168.11.202:8082/apps/index.php?item=phonea&action=aPhone
Google
System
Border
Email
Phone
Vpn
Logout
Reboot
Basic
Cascade
Queue
Audit
SN.: 232333
Channel Dial Plan Setting
Phone >> Basic >> Channel Dial Plan
Setting Dial Pattern for Voice Channel or Channel Group
Channel Number
Dialing Pattern
Number of Dial Digits to strip (empty to terminate)
Call Absolute timeout (secs)
Force to regenerate ringback tone
Add
Channel Group
1
Dialing Pattern
_[12]XXX
Number of Dial Digits to strip (empty to terminate)
0
least non-busy first
round-robin
Call Absolute timeout (secs)
Force to regenerate ringback tone
Add
Delete from list
------------------------------------none------------------------------
Delete
Current Host Name : garden002
WAN IP Address: 192.168.11.202
LAN IP Address: 172.16.9.1
Tips, Messages & Status :
Please restart VoIP process to take effect.
Done

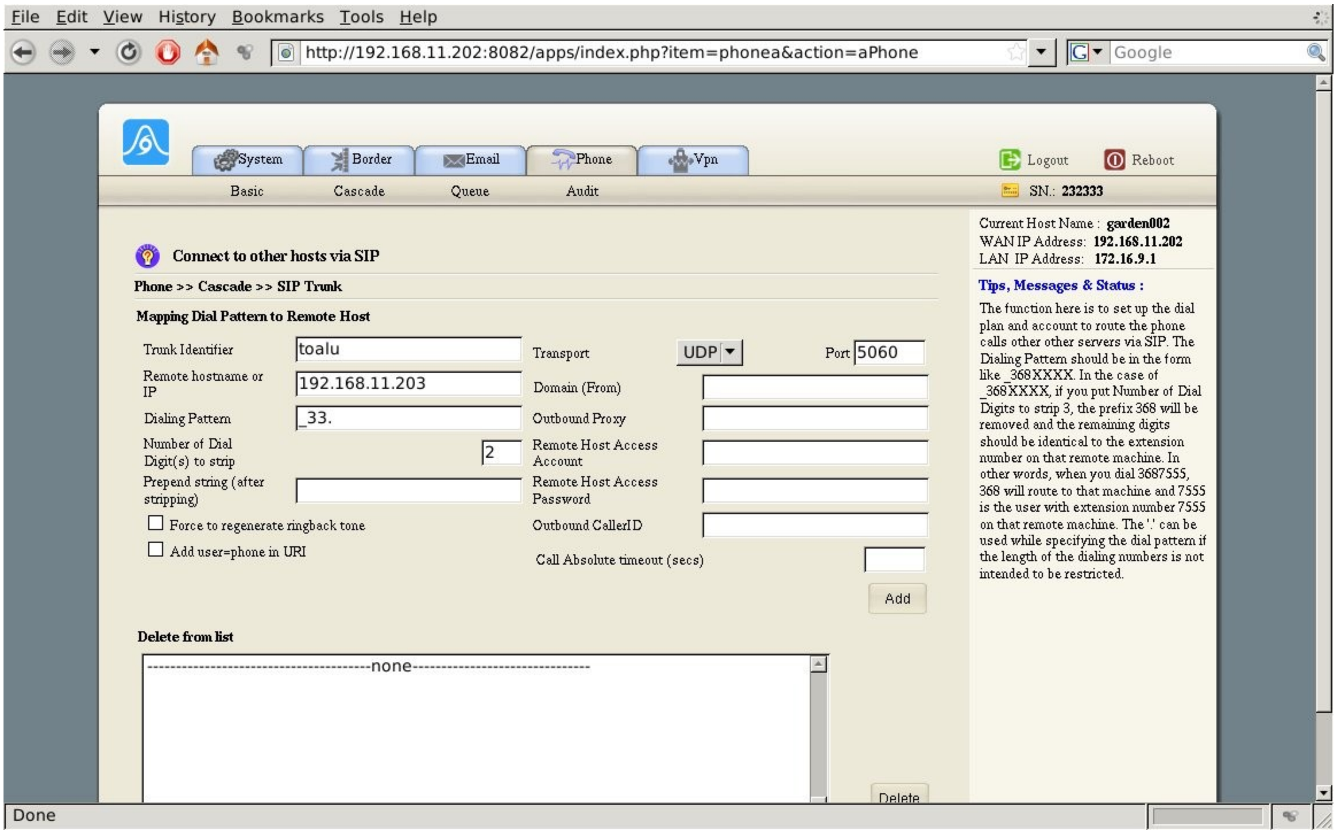

File Edit View History Bookmarks Tools Help
http://192.168.11.202:8082/apps/index.php?item=phonea&action=aPhone
Google
System
Border
Email
Phone
Vpn
Logout
Reboot
Basic
Cascade
Queue
Audit
SN.: 232333
Channel Dial Plan Setting
Phone >> Basic >> Channel Dial Plan
Setting Dial Pattern for Voice Channel or Channel Group
Channel Number
Dialing Pattern
Number of Dial Digits to strip (empty to terminate)
Call Absolute timeout (secs)
Force to regenerate ringback tone
Add
Channel Group
Dialing Pattern
Number of Dial Digits to strip (empty to terminate)
least non-busy first
round-robin
Call Absolute timeout (secs)
Force to regenerate ringback tone
Add
Delete from list
_[12]XXX,1,Dial(Dahdi/g1/${EXTEN:0})
Delete
Current Host Name : garden002
WAN IP Address: 192.168.11.202
LAN IP Address: 172.16.9.1
Tips, Messages & Status :
Please restart VoIP process to take effect.
Done

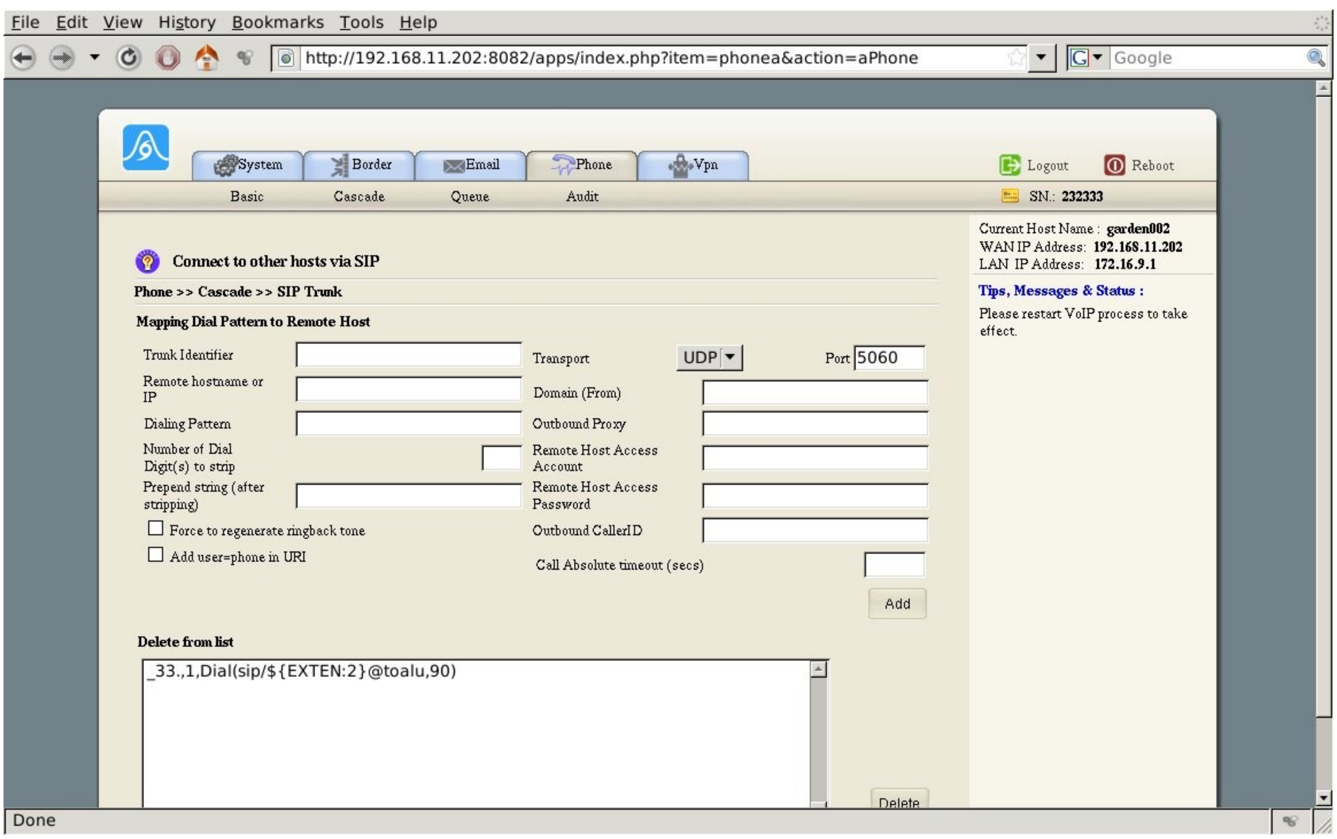
After finishing the setting of dialing rules and restart VoIP process, we come back to check the channel status. It indicates the channels are "In Use". Once the wire is connected and the other side is configured well, the RED alarm should disappear.



SIP Trunk Setting

If there is no need to set up account authentication for SIP trunk, the setting is simple as follows:


File Edit View History Bookmarks Tools Help
http://192.168.11.202:8082/apps/index.php?item=phonea&action=aPhone
Google
System Border Email Phone Vpn
Logout Reboot
Basic Cascade Queue Audit
SN.: 232333
Current Host Name : garden002
WAN IP Address: 192.168.11.202
LAN IP Address: 172.16.9.1
Connect to other hosts via SIP
Phone >> Cascade >> SIP Trunk
Mapping Dial Pattern to Remote Host
Trunk Identifier
toalu
Remote hostname or IP
192.168.11.203
Dialing Pattern
_33.
Number of Dial Digit(s) to strip
2
Prepend string (after stripping)
Force to regenerate ringback tone
Add user=phone in URI
Transport
UDP
Port
5060
Domain (From)
Outbound Proxy
Remote Host Access Account
Remote Host Access Password
Outbound CallerID
Call Absolute timeout (secs)
Add
Delete from list
--------------------------------------none------------------------------
Delete
Tips, Messages & Status :
The function here is to set up the dial plan and account to route the phone calls other other servers via SIP. The Dialing Pattern should be in the form like _368XXXX. In the case of _368XXXX, if you put Number of Dial Digits to strip 3, the prefix 368 will be removed and the remaining digits should be identical to the extension number on that remote machine. In other words, when you dial 3687555, 368 will route to that machine and 7555 is the user with extension number 7555 on that remote machine. The '.' can be used while specifying the dial pattern if the length of the dialing numbers is not intended to be restricted.
Done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Mapping Dial Pattern to Remote Host
Trunk Identifier
Remote hostname or IP
Dialing Pattern
Number of Dial Digit(s) to strip
Prepend string (after stripping)
Force to regenerate ringback tone
Add user=phone in URI
Transport
UDP
Port
5060
Domain (From)
Outbound Proxy
Remote Host Access Account
Remote Host Access Password
Outbound CallerID
Call Absolute timeout (secs)
Add
Delete from list
_33.,1,Dial(sip/${EXTEN:2}@toalu,90)
Delete
Tips, Messages & Status :
Please restart VoIP process to take effect.
Done