
ThinkTel



THINKTEL COMMUNICATIONS CUDATEL PHONE SYSTEM 270

High Availability and SIP-TRUNK Configuration

A decorative graphic consisting of several thin, white, curved lines that sweep across the bottom of the page, starting from the left and curving towards the right.



TABLE OF CONTENTS

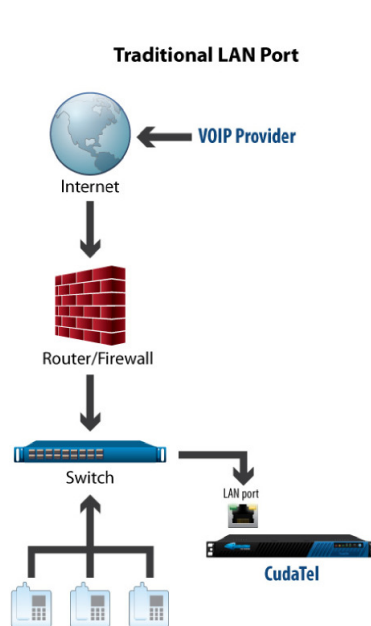
1.1	CONFIGURING TELEPHONE SERVICE PROVIDER (THINKTEL)	3
1.2	OUTBOUND CALL ROUTING.....	5
1.3	INBOUND CALL FROM THINKTEL SIP-TRUNK	5
2.1	HIGH AVAILABILITY FEATURE	9
2.2	HIGH AVAILABILITY TERMINOLOGY	9
2.3	HIGH AVAILABILITY PREPARATION	9
2.4	HIGH AVAILABILITY ACTIVATION	10
2.5	HIGH AVAILABILITY FAILOVER	11
3.1	FREWALL CONFIGURATION.....	14

1.1 CONFIGURING TELEPHONE SERVICE PROVIDER (THINKTEL)

This Chapter explains how to configure the SIP-TRUNK connection between the BARRACUDA Phone System and the telephone service provider THINKTEL COMMUNICATIONS.

The Primary way to connect the Cudatel Communication Server to the outside world is via IP connection and a SIP account called SIP-TRUNK.

THINKTEL COMMUNICATIONS as a VOIP Provider gives you an account with accompanying credentials (username - password - Proxy IP Address)



-Log in to the CCS Web Access, Navigate to “**PROVIDERS**” > “**SIP ACCOUNTS**” and select “**NEW SIP ACCOUNT**” In the displayed window, you will see a number of configuration options, many of which are required. These options are:

- Provider Type : <Generic SIP>
- Name : <THINKTEL>
- Host : Proxy IP Address Provided by THINKTEL or <tor.trk.tprm.ca> or <edm.trk.tprm.ca>
- Port : <5060>
- Realm : Realm IP Address Provided by THINKTEL or <tor.trk.tprm.ca> or <edm.trk.tprm.ca>
- Username : Pilot Number provided by THINKTEL
- Password : Provided by THINKTEL

- Registration : Unchecked
- Services : Select the three services
- Caller ID Number : You can customise the sent outbound caller ID sent on this connection by Specifying a phone number that should be sent under these conditions “ALWAYS SENT” or “ALWAYS SENT UNLESS OVERRIDEN” or “NEVER SENT”
- Outgoing Music on Hold : <default>
- Restrict Codec to : Don’t specify any of the codecs ULAW will be the default Codec
- Inbound Registration : Unchecked

Then click on “CREATE GATEWAY”

Inbound Registration Allow Inbound Registration

Create Gateway

The screenshot shows the CUDATEL web interface for configuring a provider. The provider name is 'THINKTEL'. The configuration includes the following fields and options:

- Provider/Type:** Generic SIP
- Host:** tor.trk.tpm.ca
- Port:** 5060
- Realm:** tor.trk.tpm.ca
- Username:** 4388997551
- Auth. Username:** 4388997551
- Password:** [Redacted]
- Registration:** Requires Registration, 300 second interval, Refresh Registration Available
- Services:** Inbound Calls, Outbound Calls, Faxes
- CallerID Number:** [Redacted], Never use a custom CallerID number, Use a custom CallerID number unless overridden, Always use a custom CallerID number
- Outgoing Music on Hold:** default
- Restrict Codecs To:** 0/21 selected
- Inbound Registration:** Allow Inbound Registration, Apply Gateway Settings
- External Numbers:** (438) 899-7551, Add External Numbers
- Outbound Routing:** 10 Digit Dialing, Manage Routes

1.2 OUTBOUND CALL ROUTING

Call routing allows you to customize how outbound calls are routed over THINKTEL telephony network. You can specify which connection to use based on the digits that the user dialed.

Navigate to “PROVIDERS” > “CALL ROUTING” to see the current routes on the system. The routes are listed in the order that they will be attempted. In the order that they will be attempted, you can move a route up or down to adjust its priority. You can also add a new route or edit existing routes from this page.

To add a route, click on “ADD ROUTING ENTRY”

- Call Type : choose “ 10 Digit Dialing”
- Destination : “THINKTEL”
- All 10 digits outgoing calls will be established through THINKTEL SIP-TRUNK

1.3 INBOUND CALL FROM THINKTEL SIP-TRUNK

- Navigate to “PROVIDERS” > “THINKTEL”

Name	Account	Provider	Services	Status
THINKTEL	4388997551@tor.trk.tprn.ca	Generic SIP	In Out	Available
asterisk	403@elsa-canada.getmyip.com	Generic SIP	In Out	Registered

- Click on “ADD EXTERNAL NUMBERS”

- ADD the DID number (E.g.: 4388997551 and then click on “ADD NUMBERS”

To ADD the DID to an Extension

- Navigate to “EXTENSIONS”

The screenshot shows the CUDATEL web interface. At the top, there is a navigation bar with icons for Dashboard, Switchboard, Extensions, Providers, Reports, and Configuration. On the left, a sidebar menu is open to 'Extensions', with sub-items like People, Groups, Inbound Call Queues, etc. The main content area is titled 'People' and contains a table of extension records:

Ext.	Name
4000	spa942 linksys
4001	bt200 hold
4002	bt200 2
4003	xlite Xlite
4004	bt200 hold
4005	Elie Iphone
4006	xlite4 pchome
4012	Elie BOUNAJM

- Click on one of the extensions E.g. “4000”

The screenshot shows the configuration page for extension 'x4000 - spa942 linksys'. The page is divided into several sections:

- Contact Information:** A message states 'There is no contact information associated with this person yet.' There is a checkbox for 'Show this person in Contact Directory searches' which is checked.
- Voice Mail:** Includes a 'Disable Voice Mail' section with a checkbox and an 'Apply Settings' button. Below it is a 'Change PIN/Password' section with two input fields and a 'Change PIN' button.
- Send Voice Mail to E-mail:** Includes a dropdown menu, a 'Voicemail Format' dropdown, and a checkbox for 'Do not save e-mailed voicemail'.
- Call Recording Policy:** Includes a checkbox for 'Record calls and save for' followed by a days input field, and a 'Send to Email' checkbox with an 'Address' input field.
- Phones:** Shows a list of phones, including 'spa942's Generic SIP Device'.
- Secondary Numbers and Extensions:** Includes a section for adding secondary numbers with a dropdown for 'Next Free Extension' and an 'Add Extension' button.

- Click on an existing phone E.g. “spa942’s Generic SIP Device”

- “EDIT PHONE”

spa942's Generic SIP Device

Rename Remove

1: x4000

Assign Line 2

Edit Phone: spa942's Generic SIP Device ✕

Click a line on the left to modify it, or “Assign Line 2” to add a line to the phone. The “Rename” and “Remove” buttons will rename the phone or remove it from this user.

Phone Information » spa942's Generic SIP Device

Display Name	spa942's Generic SIP Device
Manufacturer	Generic SIP device
Model	Generic SIP device (Linksys/SPA942-6.1.5(a))
MAC Address	(Generic SIP device: See individual lines for registration details.)
IP Address	192.168.1.101
Last Registration	2012-07-04 13:42:57.662085

- Click on “x4000” which is the extension phone number

spa942's Generic SIP Device

Rename Remove

1: x4000

Assign Line 2

spa942's Generic SIP Device » Line 1 ✕

Extensions and Numbers

This line can be reached at x4000

Add a Number or Extension
Valid extension ranges: 4000-40050
Next Free Extension
✓ Valid extension selected

Test Line

This will test this line on this phone by calling the phone and entering an echo test. It will not call other phones sharing this line.

- On Add a Number or Extension choose “EXTERNAL NUMBER” “4388997551”

spa942's Generic SIP Device

Rename Remove

1: x4000

Assign Line 2

spa942's Generic SIP Device » Line 1 ✕

Extensions and Numbers

This line can be reached at x4000

Add a Number or Extension
Valid extension ranges: 4000-40050
External Number
✓ Valid extension selected

Test Line

This will test this line on this phone by calling the phone and entering an echo test. It will not call other phones sharing this line.

- Click on “ADD EXTENSION”
- All calls to the DID “4388997551” will ring on the extension 4000

2 HIGH AVAILABILITY

2.1 HIGH AVAILABILITY FEATURE

The Cudatel Communication Server (versions 270 and above) supports High Availability to ensure that if a CCS unit fails, agents will continue to be able to make calls.

2.2 HIGH AVAILABILITY TERMINOLOGY

- HA PAIR - Two connected CCS units configured for the High Availability Feature
- HA PORT - A designed Ethernet port for connecting two CCS units together to implement HA
- HA CABLE - A Gigabit Ethernet cable connecting the two CCS
- MASTER - The primary CCS unit in the HA PAIR
- SLAVE - The secondary unit in the HA PAIR
- FAILOVER - The process where the slave units takes over all CCS functions from the MASTER unit
- PAIRING - The process of putting two CCS units into a HA PAIR
- UNPAIRING - The process of removing two CCS units from pair state
- SHARED ADDRESS - The HA pair's collective LAN IP Address , this will be the shared LAN IP Address for the HA PAIR

2.3 HIGH AVAILABILITY PREPARATION

- The two CCS should be the same model
- The HA ports of both units must be connected by an HA CABLE
- Both CCS must be running the same firmware version
- The LAN IP Addresses of each unit and the SHARED IP Address must all be on the same subnet
- The two units must be powered ON

For our example:

IP Address of the MASTER: 192.168.1.120 (this IP will become the Share IP Address)

IP Address of the SLAVE : 192.168.1.126

New internal IP Address of the MASTER: 192.168.1.125

2.4 HIGH AVAILABILITY ACTIVATION

- Using a web browser , connect to the IP Address of the primary unit
- Navigate to “CONFIGURATION” > “HIGH AVAILABILITY”



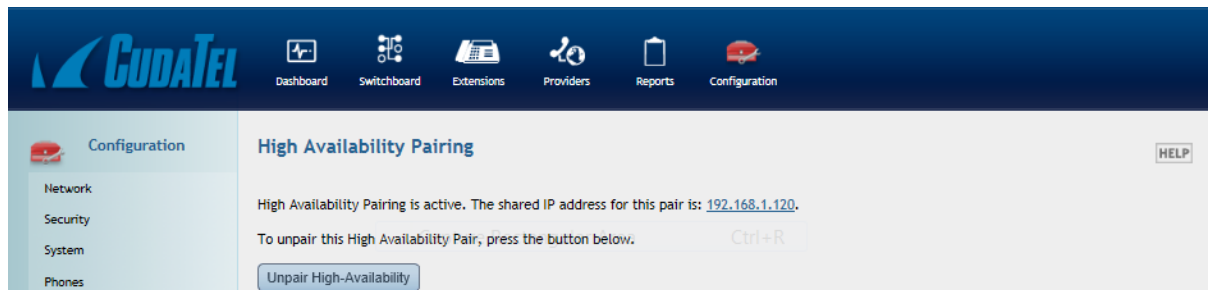
- Enter the Communication Server’s new internal IP Address “192.168.1.125” and press “CREATE HIGH-AVAILABILITY PAIR”



Create High Availability Pair?

Creating the High Availability Pair will remove all data on the other (secondary) Communication Server

- Press on “CREATE PAIR” this process may take up to 10 minutes

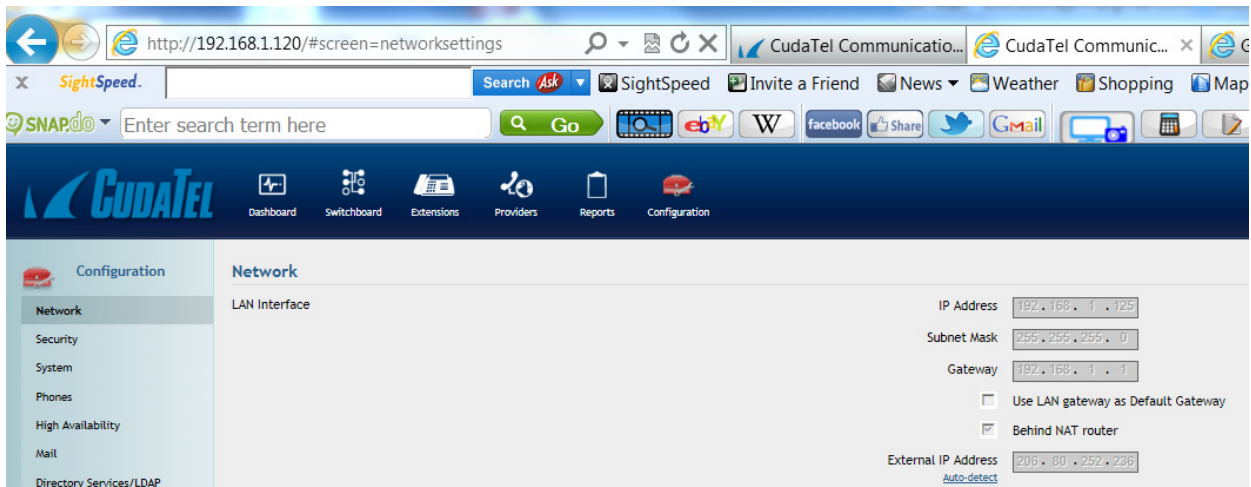


- The Shared IP Address for the pair is now : 192.168.1.120
- This Shared IP will be used for all SIP Connection with the HA PAIR

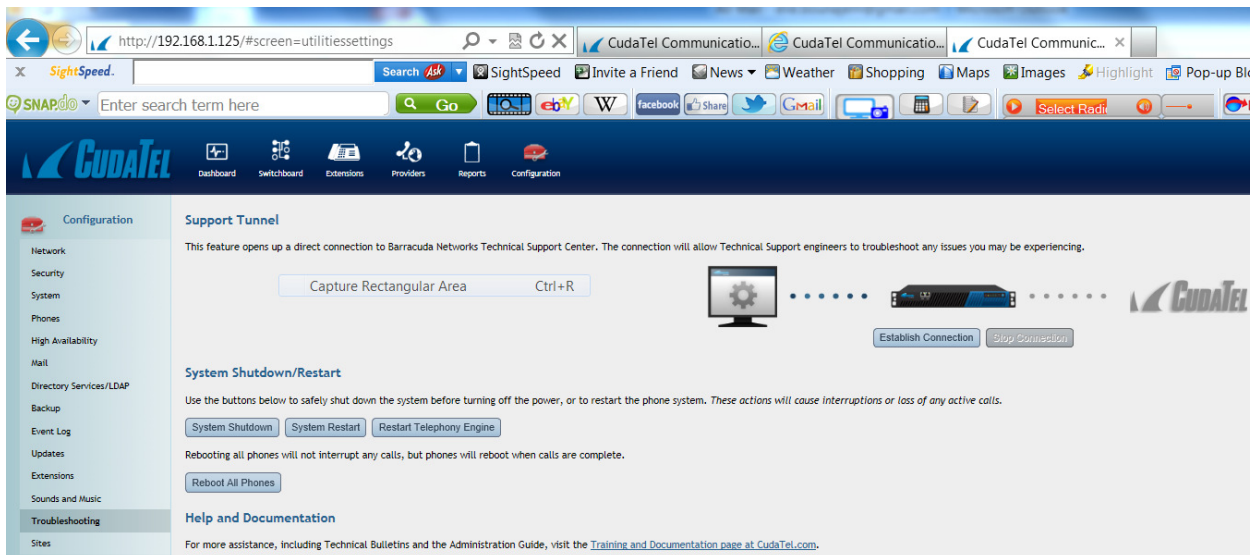
2.5 HIGH AVAILABILITY FAILOVER

MASTER IP Address : 192.168.1.125
SLAVE IP Address : 192.168.1.126
SHARED PAIR IP Address : 192.168.1.120

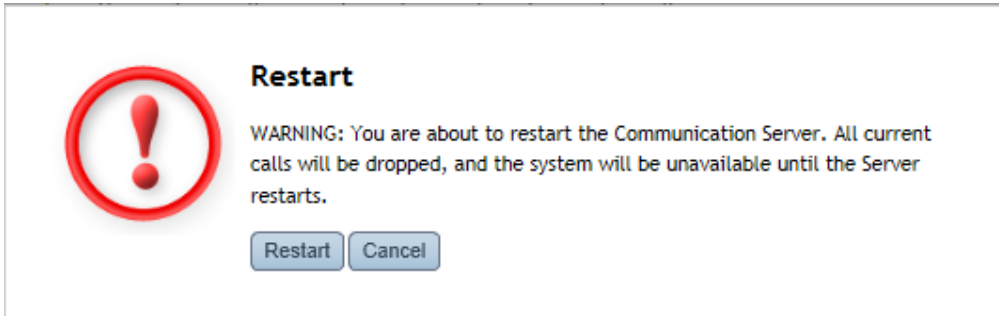
- Open a Web Page with the Shared IP Address
- Navigate to “Configuration” > “Network”
- Notice that we are not in Failover status because the IP Address shown is 192.168.1.125
Which is the IP Address of the MASTER



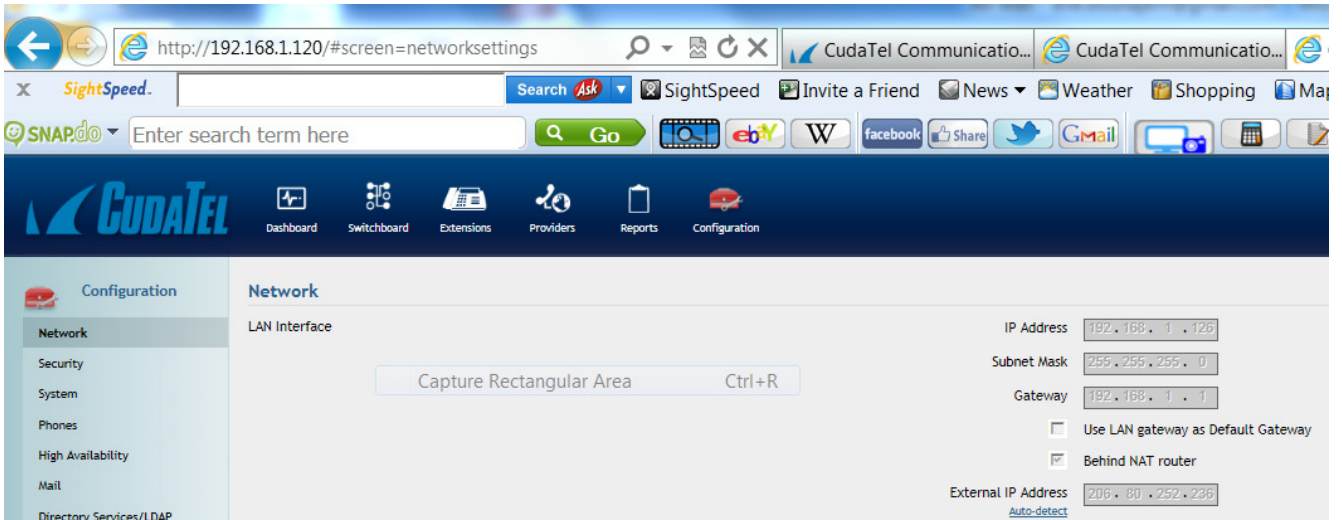
- Open a Web page with the IP Address of the MASTER and Click on “System Restart”



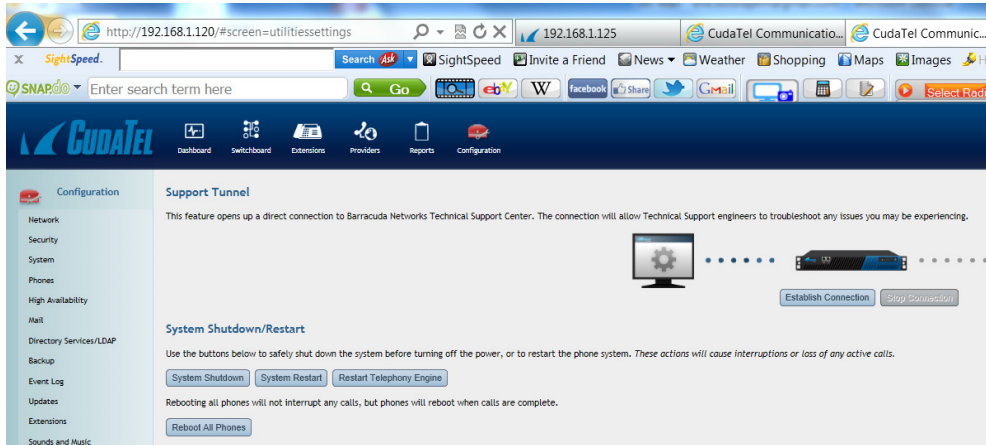
- Then Click on Restart



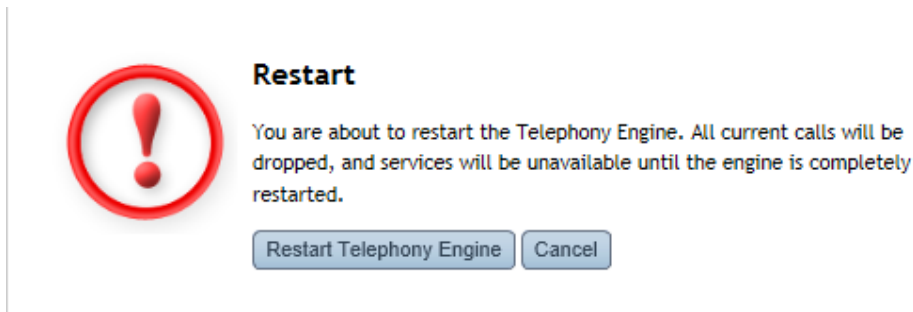
- Wait for 2 minutes and then Open a Web page with the Shared IP Address
- Notice that we are in Failover Status because the IP Address shown is 192.168.1.126
Which is the IP Address of the SLAVE



- In order to accelerate the failover process
- Navigate to “Configuration” > “Troubleshooting” and Click on “System Restart”



- Click on “Restart Telephony Engine”



- Now the Slave CCS takes over all CCS functions from the Master Unit and processing all Calls
- In Order to return back to the normal status , Log in to a web page with the IP Address of the Slave
- Navigate to “Configuration” > “Troubleshooting”
- Click on “System Restart”
- Click on “Restart”
- Log in to a Web page with the Shared IP Address
- Navigate to “Configuration” > “Troubleshooting” and Click on “System Restart”
- Click on “Restart Telephony Engine”
- Now the MASTER CCS takes over all CCS functions from the SLAVE Unit and processing all Calls

3 FIREWALL

3.1 FIREWALL CONFIGURATION

- Open the following ports in your Firewall

Port	Direction	TCP	UDP	Usage
21	Out	Yes	No	FTP (System Backups)
53	Out	Yes	Yes	DNS
69	In/Out	No	Yes	TFTP (Phone Provisioning)
80	In/Out	Yes	No	Firmware provisioning & Doc Updates
123	In/Out	No	Yes	NTP
843	In/Out	Yes	No	Flash Policy Server
5060-5070	In/Out	Yes	Yes	SIP Ports
7838	In/Out	Yes	No	Web Sockets
16384-32768	In/Out	No	Yes	RTP