Quick Configuration Guide (QCG)
TA 900/900e SIP Trunk Quick Configuration Guide

Overview

Introduction

This configuration guide explains the concepts behind transmitting voice signals over Internet packet-based networks with Session Initiation Protocol (SIP) trunks, utilizing the Total Access 900 Series of products.

Understanding SIP Trunks

SIP is designed to control call setup and tear down as well as caller ID, call transfer, and call hold features. SIP resembles the text-based client-server HTTP protocol, where the client requests services that are provided by the server. SIP is commonly used in Internet technologies such as Instant Messaging, IP voice, video web cams, and IP Centrex service. In short, a SIP trunk is an IP telephony connection between a client and server using SIP signaling to govern call control.

Although SIP is a fairly new IP signaling protocol, its simplicity and interoperability make it very popular. The Internet Engineering Task Force (IETF) has adopted SIP as the standard VoIP application layer protocol.

Requirements & Limitations

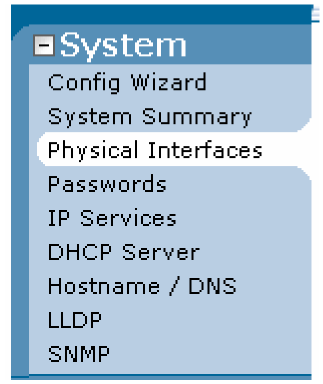
Multiple SIP trunks are not supported in firmware revisions below 12.01. Also, the ‘media-gateway’ command only affects the RTP stream and not SIP messages in firmware revisions below 13.01. In revision 13.01 and beyond, the ‘media-gateway’ command affects both SIP and RTP messages.

Web Interface Configuration

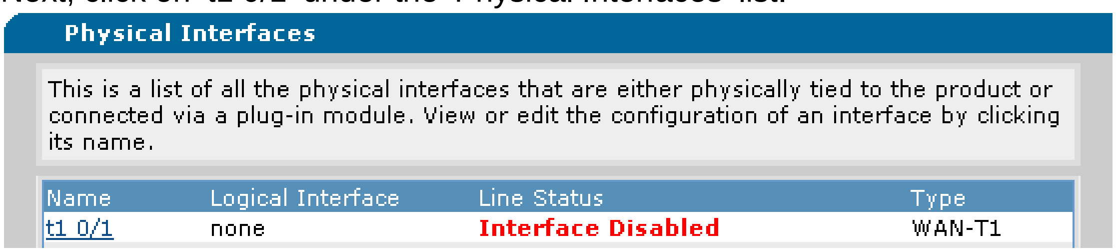
A SIP Trunk is configured over a data connection. Thus, it can be configured over an Ethernet connection or a T1 connection. If you are setting up a SIP Trunk over an Ethernet connection, you can skip to the section titled ‘Configuring the Ethernet Interface’.

Configuring the T1 Interface

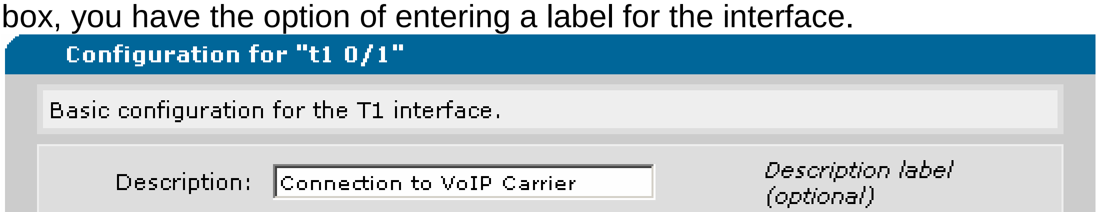
To begin configuring the T1 interface, click on ‘Physical Interfaces’ under ‘System’.



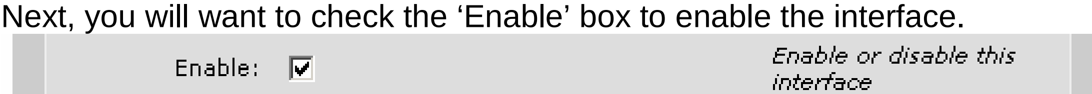
Next, click on ‘t1 0/1’ under the ‘Physical Interfaces’ list.



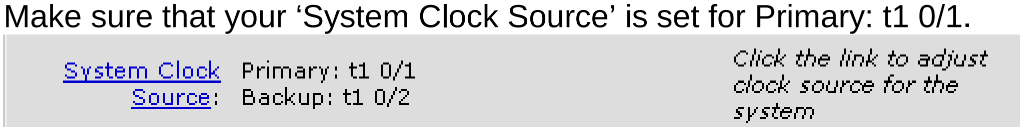
This will take you to the ‘T1 0/1’ Interface Configuration page. In the Description box, you have the option of entering a label for the interface.



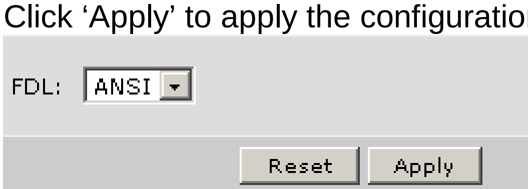
Next, you will want to check the ‘Enable’ box to enable the interface.



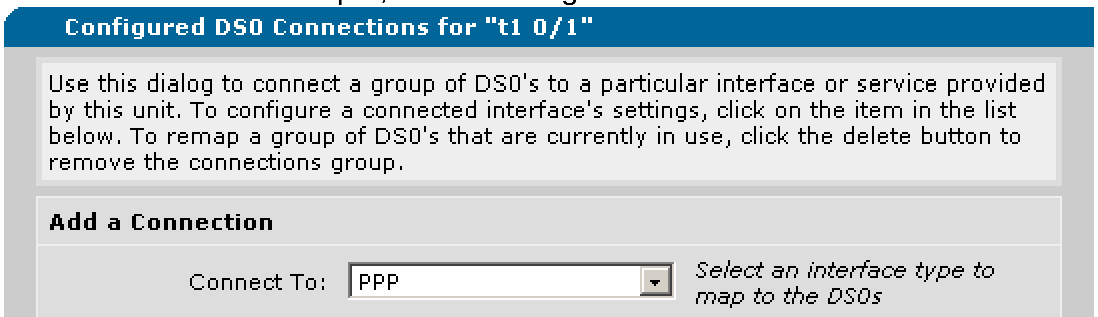
Make sure that your ‘System Clock Source’ is set for Primary: t1 0/1.



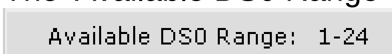
Click ‘Apply’ to apply the configuration above.



The T1 interface must be connected to a Layer 2 logical interface. To specify the Layer 2 protocol, choose the appropriate interface type in the ‘Connect To:’ drop-down box. In this example, we are using PPP.

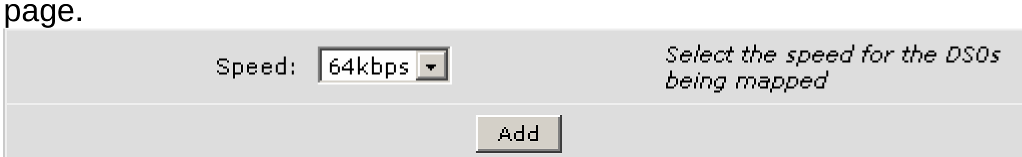


The ‘Available DS0 Range’ shows the DS0s on the T1 that are available for use.

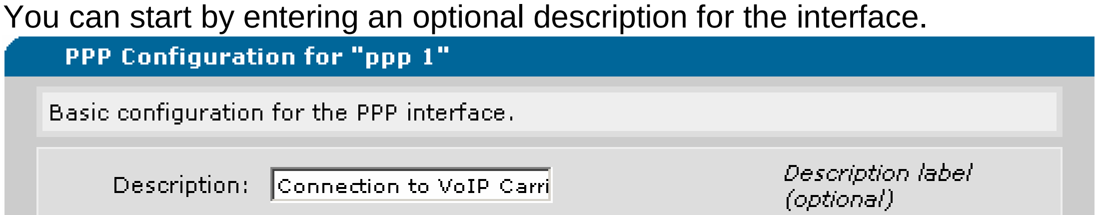


Specify the appropriate DS0 Range using the drop-down boxes.

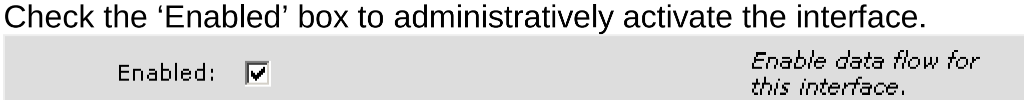
If you need to change the Speed of each DS0, you can do that with the ‘Speed’ drop-down box. Then, click on ‘Add’ to go to the PPP interface configuration page.



At the PPP Configuration page, you can specify appropriate PPP parameters. You can start by entering an optional description for the interface.

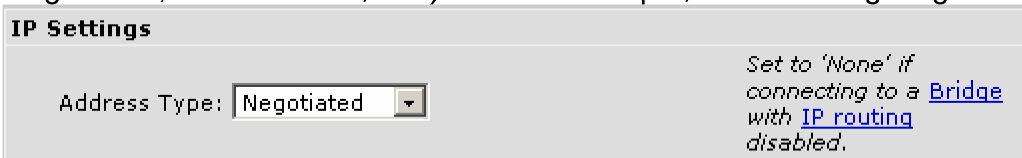


Check the ‘Enabled’ box to administratively activate the interface.

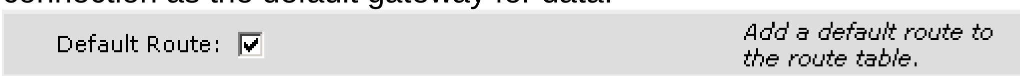


The ‘Physical Interface’ shows the physical interface that the logical interface is connected to.

Under ‘IP Settings’, you can specify the ‘Address Type’ being used (i.e. Static, Negotiated, Unnumbered, etc.). In this example, we are using Negotiated.



You can check the ‘Default Route’ box if you are going to be using this connection as the default gateway for data.

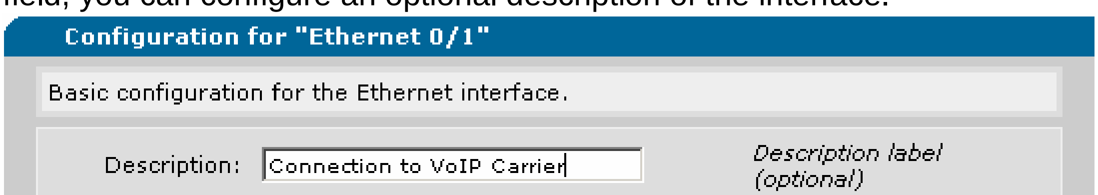


Under ‘Media-Gateway’, you can specify which address is to be used for RTP traffic. In this example, the primary address will be used. Once you are finished, click on ‘Apply’ to apply all of the changes. The T1 should now be setup correctly.

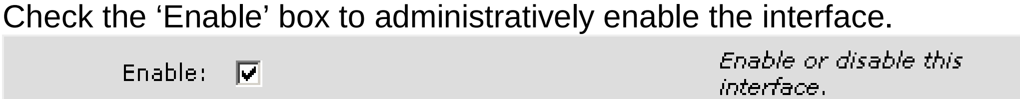
Configuring the Ethernet Interface

If you are using the Ethernet port as the connection to the VoIP Network, you will need to click on the appropriate Ethernet interface under ‘Physical Interfaces’.

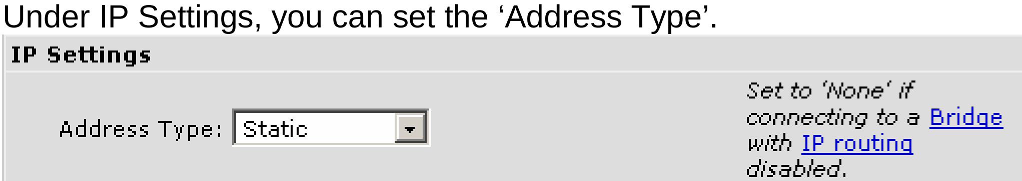
This takes you to the Ethernet interface configuration page. In the ‘Description’ field, you can configure an optional description of the interface.



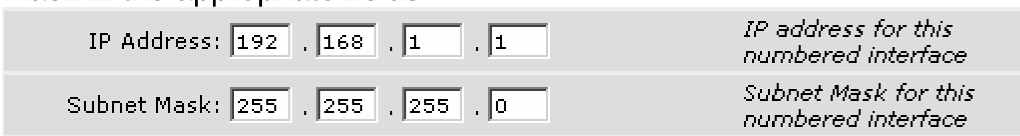
Check the ‘Enable’ box to administratively enable the interface.



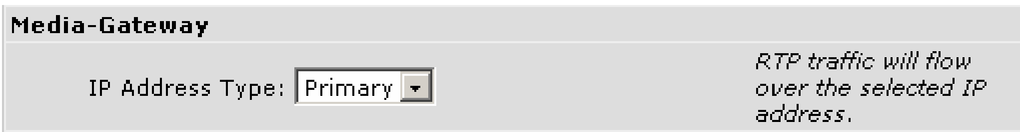
Under IP Settings, you can set the ‘Address Type’.



If you are using a Static Address Type, configure the IP Address and Subnet Mask in the appropriate fields.



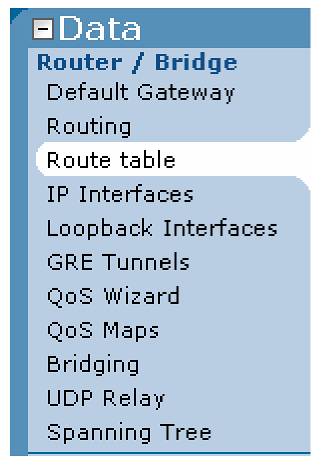
Configure the ‘Media-Gateway’ setting to reflect which address will be used on this interface for RTP.



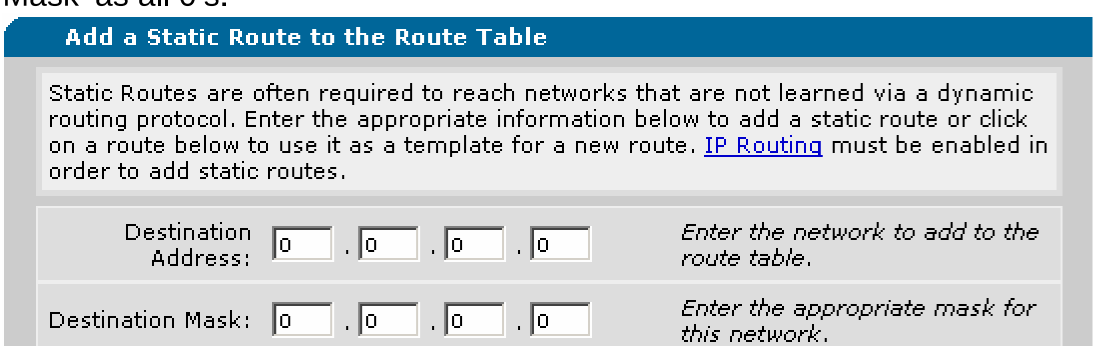
Configuring a Default Route

Note: This step can be skipped if the ‘Default Route’ box was checked when setting up the PPP interface.

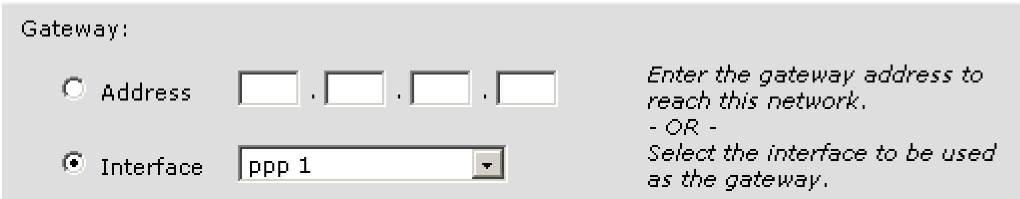
To configure a Default Route, click on ‘Route table’ under ‘Data’.



To add a static default route, configure the ‘Destination Address’ and ‘Destination Mask’ as all 0’s.

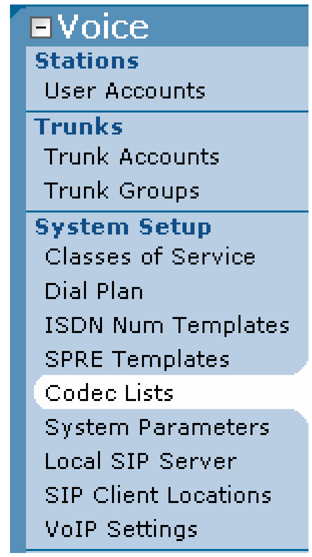


Then, specify the Address or Interface to use as the default gateway.

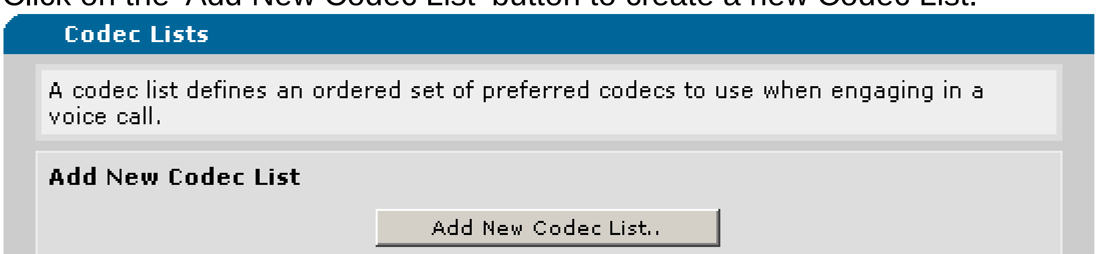


Configuring the SIP Trunk

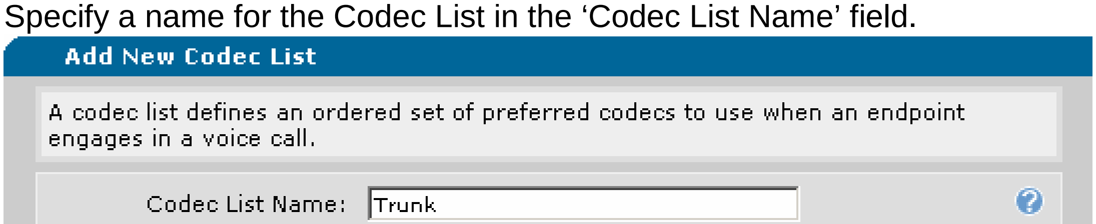
First, we will configure a Codec List to be used by the SIP trunk. To do this, click on ‘Codec Lists’ under ‘Voice’.



Click on the ‘Add New Codec List’ button to create a new Codec List.

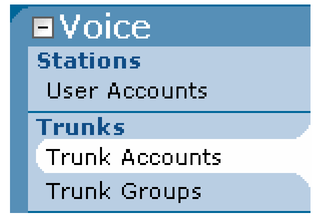


Specify a name for the Codec List in the ‘Codec List Name’ field.

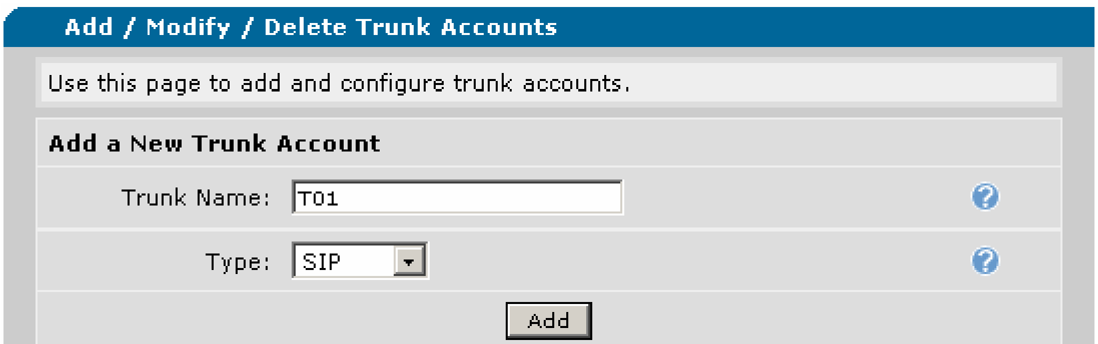


Specify the primary Codec to be used as ‘Codec #1’ and the secondary Codec to be used as ‘Codec #2’. Once those are configured, click on ‘Apply’.

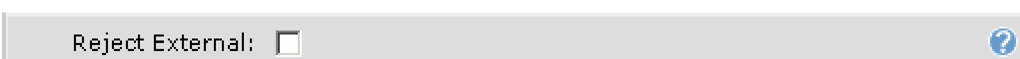
Next, we need to create the SIP trunk. To do this, start by clicking on 'Trunk Accounts' under the 'Voice' tab.



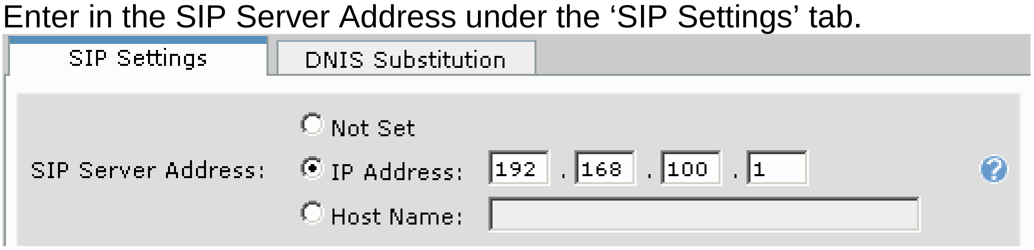
Give the Trunk a name and make sure that 'Type' is set to SIP. When finished, click on the 'Add' button.



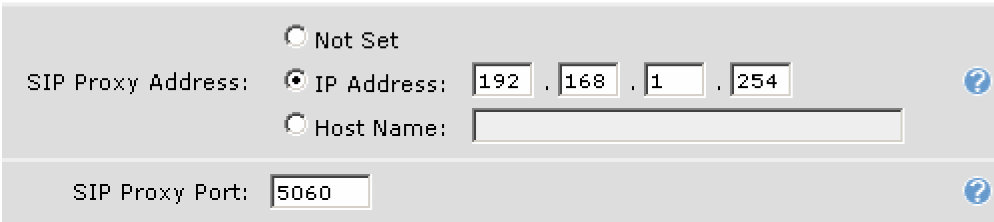
You will probably want to un-check the 'Reject External' box to allow trunk-to-trunk calls to work properly. Un-checking this box allows the SIP trunk to accept calls from other trunks that may be configured (i.e. E&M Wink or PRI). This command is necessary for deployments where more than one trunk is being used (i.e. a SIP trunk and a PRI/RBS trunk to the CPE).



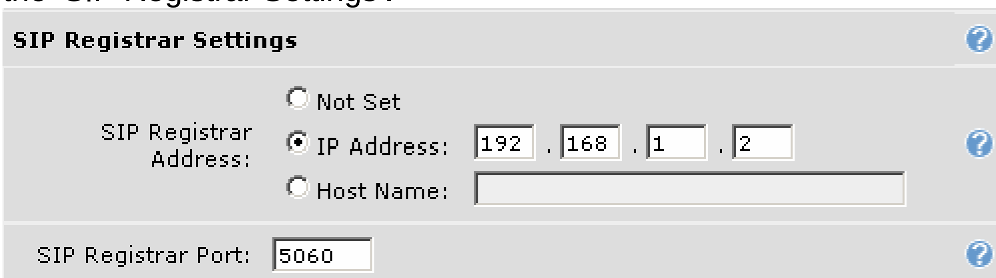
Enter in the SIP Server Address under the 'SIP Settings' tab.



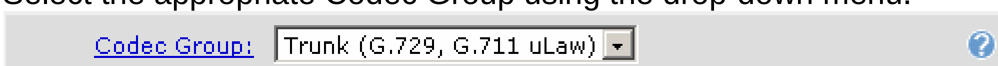
Enter in the correct IP Address and Port for the SIP Proxy if one is being used. If a SIP Proxy is not being used, make sure that 'Not Set' is selected.



If you are using a SIP Registrar, set the appropriate IP Address and Port under the 'SIP Registrar Settings'.

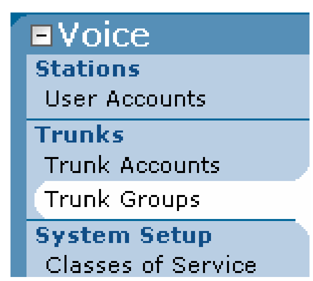


Select the appropriate Codec Group using the drop-down menu.

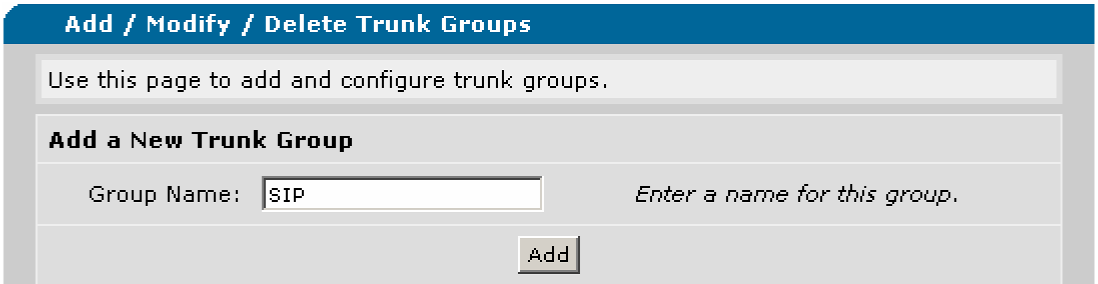


Click on 'Apply' to apply the configuration changes.

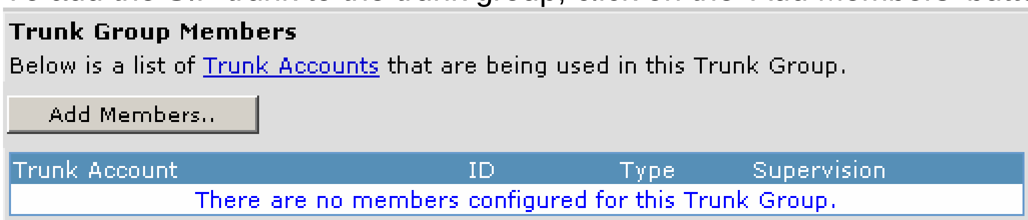
The last step is to create a trunk group for the SIP trunk. To start this process, click on 'Trunk Groups' under the 'Voice' tab.



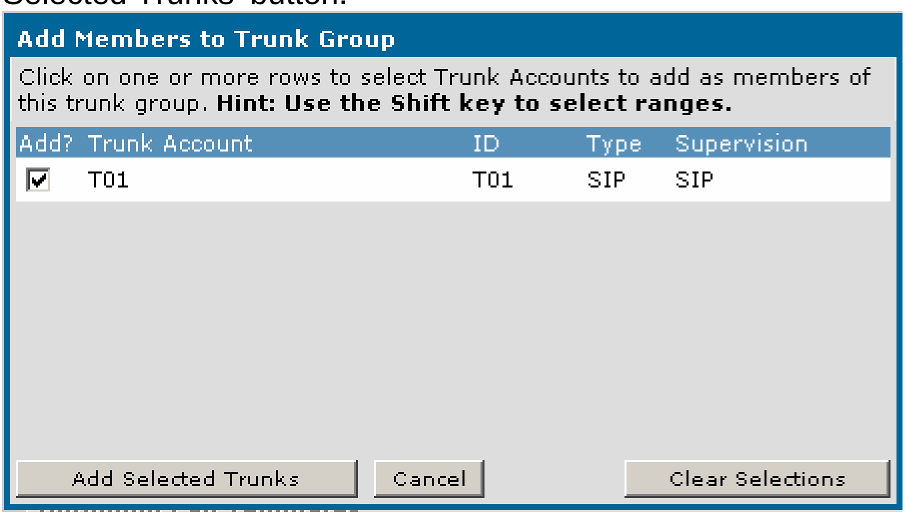
Give the Trunk Group an appropriate name using the 'Group Name' field and then click 'Add'.



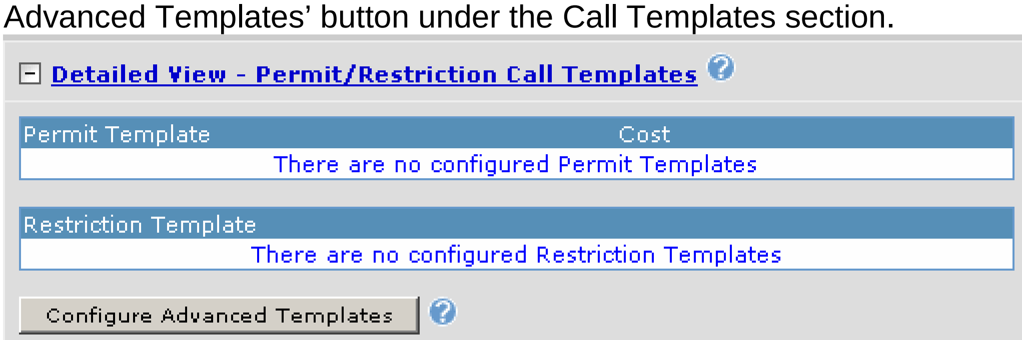
To add the SIP trunk to the trunk group, click on the ‘Add Members’ button.



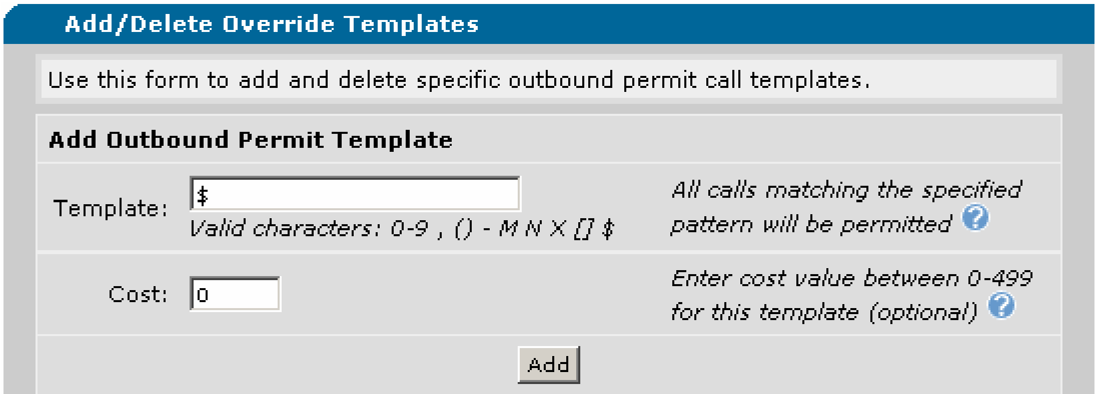
A new window will open with the available trunk accounts. Check the box under the ‘Add?’ column for the appropriate trunk account and then click on the ‘Add Selected Trunks’ button.



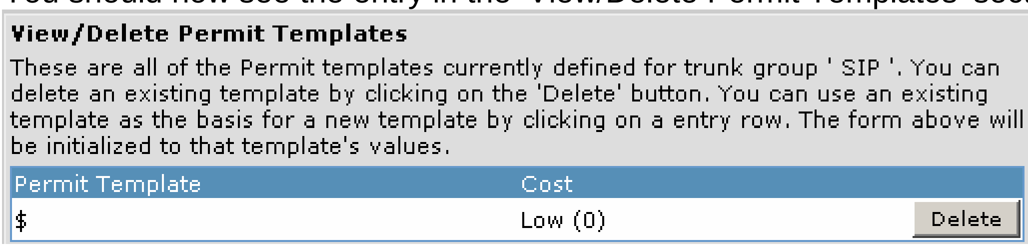
To configure all calls to route out of the SIP trunk, click on the ‘Configure Advanced Templates’ button under the Call Templates section.



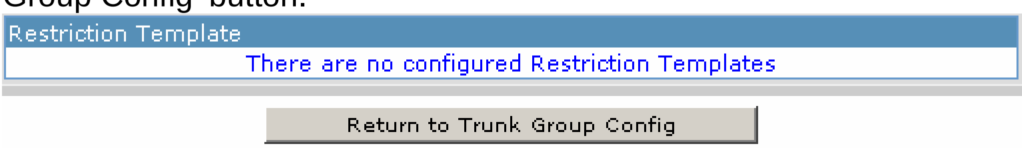
Add an appropriate entry by configuring the ‘Template’ and ‘Cost’ according to your call routing strategy. In this case, all calls will be routed out of the SIP trunk so the Template should be ‘$’ and the Cost should be ‘0’. When finished, click on the ‘Add’ button.



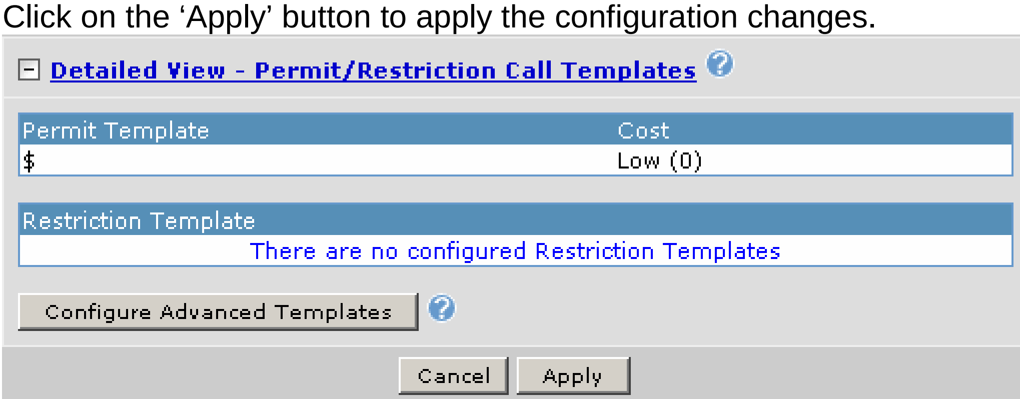
You should now see the entry in the ‘View/Delete Permit Templates’ section.



Once the Permit Template(s) have been verified, click on the ‘Return to Trunk Group Config’ button.



Click on the ‘Apply’ button to apply the configuration changes.



Always remember to save your changes by clicking on ‘Save’ at the top of the Web GUI.

Command Line Interface Configuration

A SIP Trunk is configured over a data connection. Thus, it can be configured over an Ethernet connection or a T1 connection. If you are setting up a SIP Trunk over an Ethernet connection, you can skip to the section titled ‘Configuring the Ethernet Interface’.

Note: The configuration parameters used in the examples are for instructional purposes only. Please replace all underlined entries (i.e. **<u>example</u>**) with your specific parameters to configure your application.

Configuring the T1 Interface

Step 1:
First, you must create a TDM group. The following example creates a TDM group of 24 DS0s (Channels 1-24 at 64kbps each) on the T1 connection. The ‘clock source’ command configures the IAD to use T1 0/1 as the clock source for the entire device.

(config)#interface t1 0/1
(config-t1 0/1)#description <u>Connection to VoIP Carrier</u>
(config-t1 0/1)#tdm-group <u>1</u> timeslots <u>1-24</u>
(config-t1 0/1)#no shutdown
(config-t1 0/1)#exit
(config)#clock source <u>t1 0/1</u>

Step 2:
A Layer 2 interface will need to be configured according to the Layer 2 protocol that is being used. The TA 900 series currently supports PPP, Frame Relay, and HDLC. For this example, PPP is going to be used. The PPP interface is going to be configured to have an assigned IP address. If you are using a static address, replace the word <u>negotiated</u> with the IP address and subnet mask that you are using.

(config)#interface ppp <u>1</u>
(config-ppp 1)#ip address <u>negotiated</u>
(config-ppp 1)#media-gateway ip primary
(config-ppp 1)#no shutdown
(config-ppp 1)#exit

Note: The command ‘media-gateway ip primary’ is used to specify the address to be used for Real-Time Transport Protocol operation when this particular interface is being used. It can be set to the primary address, secondary address, or a loopback address.

Step 3:
Use the following command at the Global Configuration prompt to create a cross-connect map from a TDM group on a physical T1 interface to the logical Layer 2 interface created in Step 2.

(config)#cross-connect 1 t1 0/1 1 ppp 1

Configuring the Ethernet Interface

If you are using a SIP trunk over an Ethernet interface, use the following commands to configure the Ethernet interface accordingly.

(config)#int eth 0/1
(config-eth 0/1)#ip address 192.168.1.1 255.255.255.0
(config-eth 0/1)#media-gateway ip primary
(config-eth 0/1)#no shutdown

Note: The command 'media-gateway ip primary' is used to specify the address to be used for Real-Time Transport Protocol operation when this particular interface is being used. It can be set to the primary address, secondary address, or a loopback address.

Configuring a Default Route

It is necessary to have at least one route configured for traffic to be transmitted/received correctly. In this example, a default route is configured to use the 'PPP 1' interface. The 'ppp 1' in the command below should be configured appropriately as the IP address or interface that should be used as the default gateway in your application.

(config)#ip route 0.0.0.0 0.0.0.0 ppp 1

Configuring the SIP Trunk

Step 1:
First, it is a good idea to create an appropriate Codec List to be used by the SIP trunk. In a Codec List, you can specify which codec should be prioritized by the trunk when negotiating the call setup. In this example, a Codec List named 'Trunk' is going to be configured to use G.729 as the primary codec and G.711 as the secondary codec.

(config)#voice codec-list Trunk
(config-codec)#codec g729
(config-codec)#codec g711ulaw
(config-codec)#exit

Step 2:
Next, a voice trunk will need to be configured. This voice trunk is where the specific SIP server attributes should be configured. In this example, the SIP server's IP address is 192.168.100.1, the registrar server's IP address is 192.168.100.2, and the Session Border Controller's IP address is 192.168.1.254. The command 'no reject-external' allows the SIP trunk to accept calls from other trunks that may be configured (i.e. E&M Wink or PRI). This command is necessary for deployments where more than one trunk is being used (i.e. a SIP trunk and a PRI/RBS trunk to the CPE).

Note: IP addresses or host names can be used to configure the sip-server, registrar, and outbound-proxy.

(config)#voice trunk <u>T01</u> type sip
(config-T01)#sip-server primary <u>192.168.100.1</u> udp 5060
(config-T01)#registrar primary <u>192.168.100.2</u> udp 5060
(config-T01)#outbound-proxy primary <u>192.168.1.254</u> udp 5060
(config-T01)#no reject-external
(config-T01)#codec-group Trunk
(config-T01)#exit

Note: The 'sip-server primary' command is the only one that ***HAS*** to be configured on the SIP trunk. The 'registrar primary' (if a Registrar server is being used) and 'outbound-proxy primary' (if a Session Border Controller is used) commands are completely optional and should only be used if they are needed.

Step 3:
Since this is a voice trunk and not a voice user, a voice grouped trunk will need to be configured to specify which numbers should be routed out the SIP trunk. In this example (and in most applications), the SIP trunk is going to be the connection to the outside world. Thus, it will be configured with an 'accept $'. This configures the unit to send all calls out the SIP trunk if there is not a more specific voice user/trunk match.

(config)#voice grouped-trunk SIP
(config-SIP)#trunk T01
(config-SIP)#accept $ cost 0
(config-SIP)#exit

Step 4:
Once everything is configured correctly, you will want to save your changes.

(config)#exit
#copy running-config startup-config

Example Configuration

```
ip sip
!
interface t1 0/1
  tdm-group 1 timeslots 1-24 speed 64
  no shutdown
!
interface ppp 1
  ip address  negotiated
  media-gateway ip primary
  no shutdown
  cross-connect 1 t1 0/1 1 ppp 1
!
ip route 0.0.0.0 0.0.0.0 ppp 1
!
voice codec-list Trunk
  codec g729
  codec g711ulaw
!
voice trunk T01 type sip
  sip-server primary 192.168.100.1
  registrar primary 192.168.100.2
  outbound-proxy primary 192.168.1.254
  no reject-external
!
voice grouped-trunk SIP
  no description
  trunk T01
  accept $ cost 0
```

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