



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

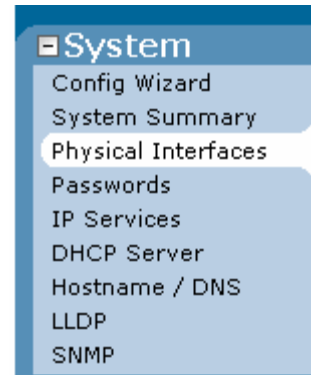
Multiple SIP trunks are not supported in firmware revisions below 12.01. Also, the 'media-gateway' command only affects the RTP stream and not SIP messages in firmware revisions below 13.01. In revision 13.01 and beyond, the 'media-gateway' command affects both SIP and RTP messages.

Web Interface Configuration

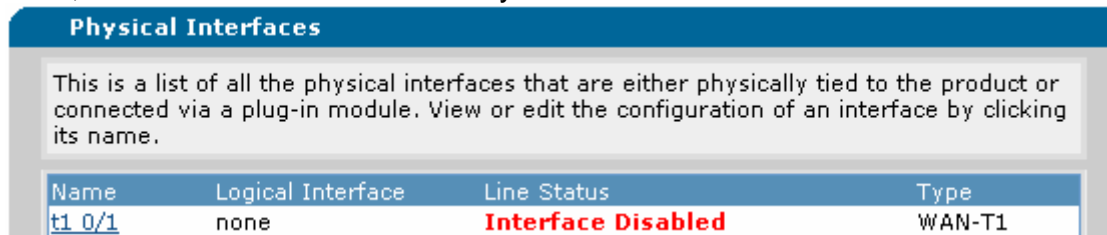
A SIP Trunk is configured over a data connection. Thus, it can be configured over an Ethernet connection or a T1 connection. If you are setting up a SIP Trunk over an Ethernet connection, you can skip to the section titled 'Configuring the Ethernet Interface'.

Configuring the T1 Interface

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



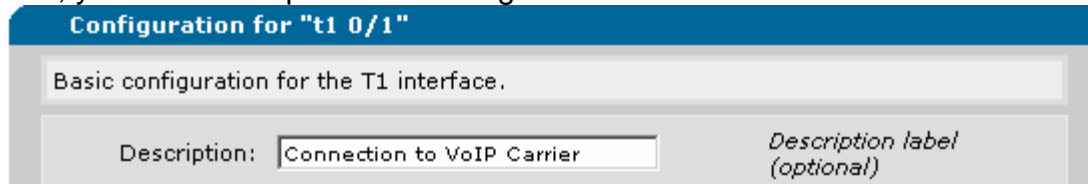
Next, click on 't1 0/1' under the 'Physical Interfaces' list.



A screenshot of the 'Physical Interfaces' configuration page. It shows a table with the following data:

Name	Logical Interface	Line Status	Type
t1 0/1	none	Interface Disabled	WAN-T1

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.



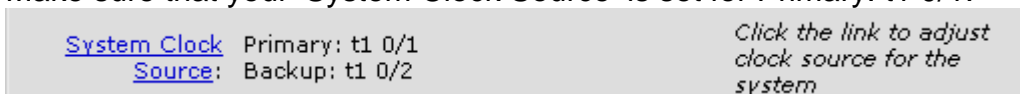
A screenshot of the 'Configuration for "t1 0/1"' page. It shows a 'Description' field with the text 'Connection to VoIP Carrier' and a note: 'Description label (optional)'.

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



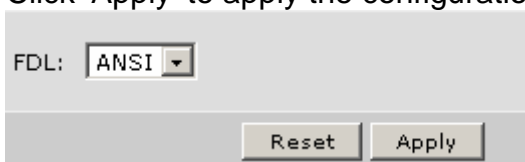
A screenshot of the 'Enable' section of the configuration page. The 'Enable' checkbox is checked, and there is a note: 'Enable or disable this interface'.

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



A screenshot of the 'System Clock Source' configuration page. It shows the following text: 'System Clock Source: Primary: t1 0/1 Backup: t1 0/2' and a note: 'Click the link to adjust clock source for the system'.

Click 'Apply' to apply the configuration above.



A screenshot of the configuration page showing the 'FDL' dropdown menu set to 'ANSI' and the 'Apply' button.

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.

Configured DS0 Connections for "t1 0/1"

Use this dialog to connect a group of DS0's to a particular interface or service provided by this unit. To configure a connected interface's settings, click on the item in the list below. To remap a group of DS0's that are currently in use, click the delete button to remove the connections group.

Add a Connection

Connect To: *Select an interface type to map to the DS0s*

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

Available DS0 Range: 1-24

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

DS0 Range: to *Set the range of DS0s to be mapped*

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.

Speed: *Select the speed for the DS0s being mapped*

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

PPP Configuration for "ppp 1"

Basic configuration for the PPP interface.

Description: *Description label (optional)*

Check the 'Enabled' box to administratively activate the interface.

Enabled: *Enable data flow for this interface.*

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

Physical Interface: **t1 0/1** *Physical interface connection for this interface.*

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

IP Settings	
Address Type: <input type="text" value="Negotiated"/>	<i>Set to 'None' if connecting to a Bridge with IP routing disabled.</i>

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

Default Route: <input checked="" type="checkbox"/>	<i>Add a default route to the route table.</i>
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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.

Media-Gateway	
IP Address Type: <input type="text" value="Primary"/>	<i>RTP traffic will flow over the selected IP address.</i>
<input type="button" value="Reset"/> <input type="button" value="Apply"/>	

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'.

eth 0/1	none	Up	Ethernet
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This takes you to the Ethernet interface configuration page. In the 'Description' field, you can configure an optional description of the interface.

Configuration for "Ethernet 0/1"	
Basic configuration for the Ethernet interface.	
Description: <input type="text" value="Connection to VoIP Carrier"/>	<i>Description label (optional)</i>

Check the 'Enable' box to administratively enable the interface.

Enable: <input checked="" type="checkbox"/>	<i>Enable or disable this interface.</i>
---	--

Under IP Settings, you can set the 'Address Type'.

IP Settings	
Address Type: <input type="text" value="Static"/>	<i>Set to 'None' if connecting to a Bridge with IP routing disabled.</i>

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

IP Address:	<input type="text" value="192"/> . <input type="text" value="168"/> . <input type="text" value="1"/> . <input type="text" value="1"/>	<i>IP address for this numbered interface</i>
Subnet Mask:	<input type="text" value="255"/> . <input type="text" value="255"/> . <input type="text" value="255"/> . <input type="text" value="0"/>	<i>Subnet Mask for this numbered interface</i>

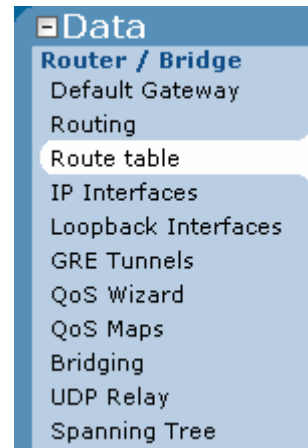
Configure the 'Media-Gateway' setting to reflect which address will be used on this interface for RTP.

Media-Gateway		
IP Address Type:	<input type="text" value="Primary"/> <input type="button" value="v"/>	<i>RTP traffic will flow over the selected IP address.</i>

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.

Add a Static Route to the Route Table		
Static Routes are often required to reach networks that are not learned via a dynamic routing protocol. Enter the appropriate information below to add a static route or click on a route below to use it as a template for a new route. IP Routing must be enabled in order to add static routes.		
Destination Address:	<input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/>	<i>Enter the network to add to the route table.</i>
Destination Mask:	<input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/>	<i>Enter the appropriate mask for this network.</i>

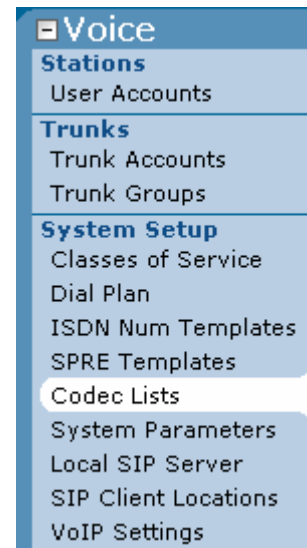
Then, specify the Address or Interface to use as the default gateway.

Gateway:

Address . . . *Enter the gateway address to reach this network.*
 - OR -
 Interface *Select the interface to be used as the 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.

Codec Lists

A codec list defines an ordered set of preferred codecs to use when engaging in a voice call.

Add New Codec List

Add New Codec List.

Specify a name for the Codec List in the 'Codec List Name' field.

Add New Codec List

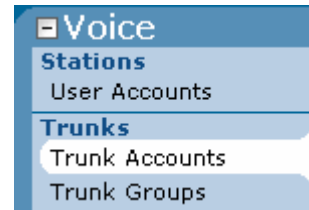
A codec list defines an ordered set of preferred codecs to use when an endpoint engages in a voice call.

Codec List Name: ?

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'.

Codec #1:	<input type="text" value="G.729"/>	
Codec #2:	<input type="text" value="G.711 uLaw"/>	
<input type="button" value="Cancel"/> <input type="button" value="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.

Add / Modify / Delete Trunk Accounts

Use this page to add and configure trunk accounts.

Add a New Trunk Account

Trunk Name:

Type:

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).

Reject External: <input type="checkbox"/>	
---	--

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

Not Set

SIP Server Address:
 IP Address:

Host Name:

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.

Not Set
 SIP Proxy Address: IP Address: . . . ?
 Host Name:
 SIP Proxy Port: ?

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

SIP Registrar Settings ?
 Not Set
 SIP Registrar Address: IP Address: . . . ?
 Host Name:
 SIP Registrar Port: ?

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

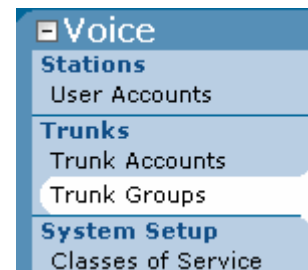
[Codec Group:](#) ?

Click on 'Apply' to apply the configuration changes.

Registration Settings ?

Register value	End (if range)	Authname
There are no Register entries for this Trunk.		

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'.

Add / Modify / Delete Trunk Groups

Use this page to add and configure trunk groups.

Add a New Trunk Group

Group Name: *Enter a name for this group.*

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

Trunk Group Members

Below is a list of [Trunk Accounts](#) that are being used in this Trunk Group.

Trunk Account	ID	Type	Supervision
There are no members configured for this Trunk Group.			

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.

Add Members to Trunk Group

Click on one or more rows to select Trunk Accounts to add as members of this trunk group. **Hint: Use the Shift key to select ranges.**

Add?	Trunk Account	ID	Type	Supervision
<input checked="" type="checkbox"/>	T01	T01	SIP	SIP

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

Detailed View - Permit/Restriction Call Templates [?](#)

Permit Template	Cost
There are no configured Permit Templates	

Restriction Template
There are no configured Restriction Templates

[?](#)

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.

Add/Delete Override Templates

Use this form to add and delete specific outbound permit call templates.

Add Outbound Permit Template

Template: Valid characters: 0-9 , () - M N X [] \$ All calls matching the specified pattern will be permitted ?

Cost: Enter cost value between 0-499 for this template (optional) ?

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

View/Delete Permit Templates

These are all of the Permit templates currently defined for trunk group ' SIP '. You can delete an existing template by clicking on the 'Delete' button. You can use an existing template as the basis for a new template by clicking on a entry row. The form above will be initialized to that template's values.

Permit Template	Cost	
\$	Low (0)	<input type="button" value="Delete"/>

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

Restriction Template

There are no configured Restriction Templates

Click on the 'Apply' button to apply the configuration changes.

Detailed View - Permit/Restriction Call Templates ?

Permit Template	Cost
\$	Low (0)

Restriction Template

There are no configured Restriction Templates

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. **example**) 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 Connection to VoIP Carrier
(config-t1 0/1)#tdm-group 1 timeslots 1-24
(config-t1 0/1)#no shutdown
(config-t1 0/1)#exit
(config)#clock source t1 0/1
```

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 negotiated with the IP address and subnet mask that you are using.

```
(config)#interface ppp 1
(config-ppp 1)#ip address negotiated
(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 T01 type sip
(config-T01)#sip-server primary 192.168.100.1 udp 5060
(config-T01)#registrar primary 192.168.100.2 udp 5060
(config-T01)#outbound-proxy primary 192.168.1.254 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
```

If you experience any problems using your ADTRAN product, please contact [ADTRAN Technical Support](#).

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