OVERVIEW

Session Initiation Protocol (SIP) is an application layer protocol used for establishing sessions in an IP network. SIP trunks allow the IPedge system to get PRI-like services from an Internet Telephony Service Provider using SIP.

A SIP Trunk allows an IPedge system to connect internal voice and private data traffic to the outside public network (PSTN and public data) via IP.

When a user dials a call that will be sent over the PSTN, the call routing is sent over the WAN to the Internet Telephony Service Provider (ITSP) that is providing the SIP Trunk. This ITSP will provide a connection to the PSTN through their equipment. The call will be sent from the IPedge system to the SIP provider, who will act as a proxy, and send the call to the dialed destination.

For incoming calls, the SIP Trunk acts somewhat like a DID trunk, the dialed number is sent to the SIP provider and then routed over the IP Network to the IPedge system. This routing is based on the URI and associated IP address.

Toshiba product SIP Trunk capabilities allow the IPedge system to communicate with a service provider natively over an IP circuit, which can be used to carry voice and data simultaneously. Inside the IPedge system, voice is converted to data and sent to the service provider along the same circuit as the other data packets. This allows one circuit to be used for voice and data, it also allows data to use all of the bandwidth when no voice is present. Quality of Service (QoS) is managed by the service provider, allowing voice to instantaneously take priority over data.

SIP trunks offer ISDN-like features over a data connection (e.g. a T1 circuit). However, unlike a traditional T1 circuit, a SIP Trunk enabled circuit does not have to be physically provisioned and divided to separate the voice channels from the data channels.

SIP Trunk

Session Initiation Protocol (SIP) is an application layer protocol used for establishing sessions in an IP network. SIP trunks allow the IPedge system to get PRI-like services from an Internet Telephony Service Provider using SIP.

A SIP Trunk allows an IPedge system to connect internal voice and private data traffic to the outside public network (PSTN and public data) via IP.

Media Relay Server Overview

In a NAT environment use the NAT Traversal capability of the IPedge server (Release 1.3 and later) or a SIP ALG router along with the IPedge Media Relay Server (MRS).

The MRS is configured by defining the Public IP address of the IPedge server and the port range to be used for calls. Each call requires two UDP ports for the audio streams (one port RTP, one port RTCP).

Refer to the IPedge Install manual for Media Relay Sever setup instructions.

SIP Trunk NAT Traversal

In order to support SIP Trunking on R1.2 and earlier IPedge systems in a NAT environment, the router needs to support an enterprise grade SIP ALG (Application Layer Gateway). In Release 1.3 and later, the IPedge system can support SIP Trunking with routers that do not have a SIP ALG.

When used behind a NAT firewall that does not support a SIP ALG, the IPedge server can still be given a private IP address. The SIP Trunk NAT Traversal capability (Release 1.3 and later) along with the MRS will allow the IPedge server to:

- Use it’s internal Media Relay Server to route media packets between the WAN and the LAN and
- Apply the correct IP address to SIP signaling messages so that when they are sent out though a NAT firewall, the SIP Trunk service provider will be able to send responses to the correct IP address.

Within the NAT router, port forwarding rules will need to be configured, and a range of ports opened for the Media Relay Server.

When used with a NAT firewall that does support an enterprise grade SIP ALG (such as the Cisco ASA5500 product line) the SIP ALG feature needs to be enabled. In this configuration the media packets will be routed directly from the LAN to the WAN and mid-call survivability of a PSTN call is possible.

SIP Trunk

SIP Trunk operation is transparent to the system users.

Outgoing calls are placed using features such as Pooled Line access, Least Cost Routing, Speed Dial, or others.

Incoming calls are routed via DID, UCD, transfer, ACD, and others.

PROGRAMMING

SIP TRUNK EXAMPLE

The example shown below is a general system plan. Refer to the specific provider sections of Install Manual.

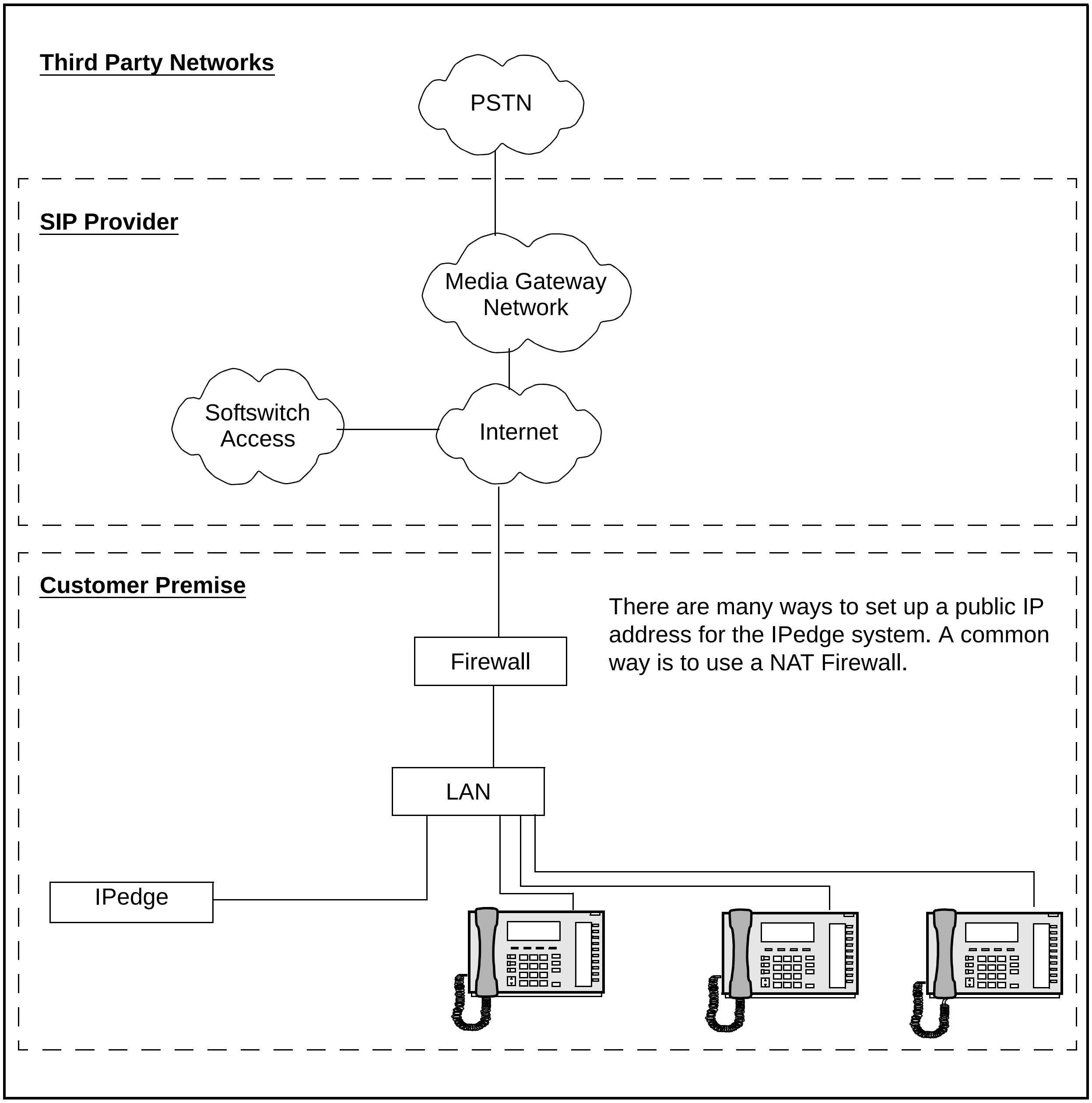


Figure 1 IPedge System with SIP Trunking

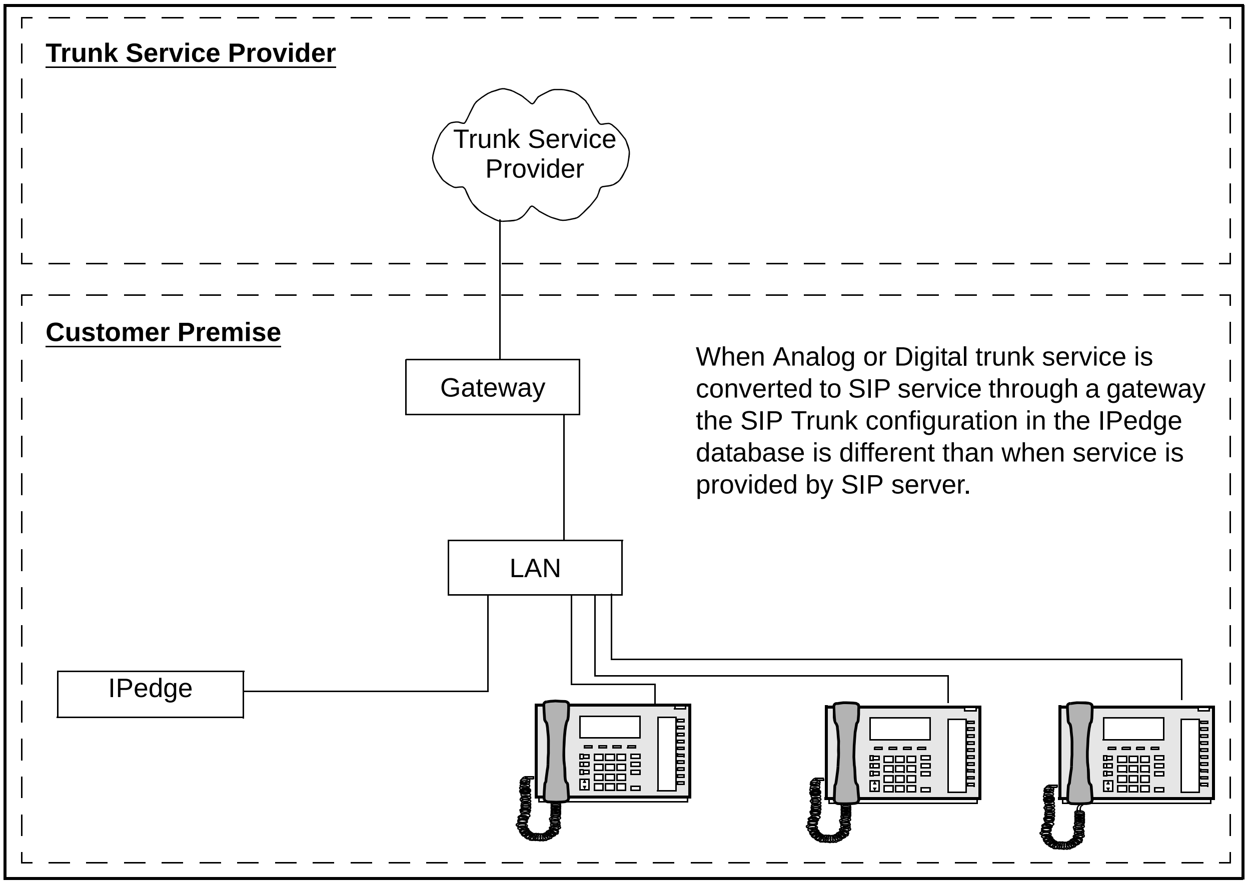


Figure 2 IPedge System with SIP Gateway

SIP TRUNK GROUP PROGRAMMING

The following pages show the general programming and configuration steps to implement a SIP Trunk. Specific procedures for each provider are in the linked tables in this document.

Note: SIP trunking requires a license for each trunk. No channel group can successfully be programmed without a license.

Programming the Incoming Line Group

1. Select **Trunk > Trunk Groups**. Click on the **New** icon.
2. Select the server to which to add the trunk group.
3. In the Group Direction pull-down select **Incoming and Outgoing**. then, click on **OK**.
4. In the **Common** tab select a trunk **Group Number** then select **Group Type** SIP. Record this number.
5. On the **Incoming** tab in the **CO Service Type** select DID then, set the number of **DID digits** (Default = 4 digits).

The DID Digits parameter sets how many of the digits received from the SIP Trunk will be used to choose the station to which the call goes. For example; if the SIP provider sends 10 digits, and the DID digits is set to four, only the last four digits are used to route the call. The additional digits will be ignored. All of the received digits must be in the URI table.

6. Click on the **Save** icon.

Note: Notice that Incoming and Outgoing trunk group with the same Trunk Group Number have been created.

Note: When all of the ILGs and OLGs have been created Toshiba recommends that you enable Intercept and program destination in **Trunk > DID Intercept**.

Programming the Outgoing Line Group

In the Outgoing tab set the parameters for outgoing calls on this trunk group. Typically the default values are used.

Note: An OLG flexible access code must be created for this group.

ASSIGN DID TRUNK DESTINATION

DID routing must be set up to route incoming SIP calls to their desired destination. This programming is the same as any other trunk group type. If the routing is not set up, incoming Invites will fail instantly.

1. Select **Trunk > DID**.
2. Select the server.
3. Click on the **New** icon.
4. Select the ILG Group Number.
5. Enter the number of DID digits in the **DID Number** field.
6. Select the MOH source.
7. Select the Tenant number (Default = 1).
8. In the DID Audio section: Set Audio Day1 DST Type, Audio Day2 DST Type and Audio Night DST Type to **Dialing Digits**.
9. Set the DST Digits to the Extension Number to which the DID calls will ring.
10. Leave the DID Data section at default.
11. Leave the remaining parameters blank.
12. Click on the **Save** icon.

Note: Least Cost Routing is, by default, programmed to use OLG 1. If SIP trunks are created using a different OLG, adjustments may be required in the LCR > Route Choice Assignments, and Route Definition Assignments.

OLG FLEXIBLE ACCESS CODE PROGRAMMING

An access code is required for the OLG that was setup for the SIP Trunks. SIP trunks can also be accessed using LCR.

1. Select **System > Flexible Access Code**.
2. Click on the **New** icon.
3. Enter the Access Code.
4. Select Feature Name; Line Group access code.
5. Select the OLG.
6. Click on the **Save** icon.

Creating the Channel Group

Important! Complete the Channel Group programming before starting the Service Definition programming.

1. Select **Trunk > SIP Trunking**.
2. In the Channel Group tab select the SIP Trunk Channel Group to be created.

Note: Choose a Channel group number that has not been assigned in another section.
When a Channel Group is selected for a SIP Trunk that Channel group number cannot be used for IPedge Net.

3. In the SIP Trunk Channels box select the TOTAL number of ports to be dedicated to the SIP Trunk channel group.
4. Click on the **Save** icon.

Service Definition

1. Click on the Service Definition tab.
2. Click on the **New** icon.
3. Select a Service Definition Index number then, enter the following based on the SIP Trunk Provider:
   Registration Mode - Client or none
   Domain Name - The domain name of the SIP Trunk provider (FQDN) or the IP address.
   SIP Server - The SIP Trunk provider outbound proxy or blank.
4. Enter the ILG and OLG created above.
5. Select the number of trunks/channels provided by this SIP Trunk provider as the Effective Channel Number.
6. Click on the **Save icon.**

Note: If you experience one-way speech on local IPT to SIP Trunk calls; enable the Media Relay Server in the System Settings then, set Connection to Media Relay Server to **Manual** in the SIP Trunk service definition.

Note: When using a NAT router, the private IP address in the SIP header is not changed. The result is an unsuccessful call.

A SIP ALG router will be required to change the private IP address to public IP address in fields in the SIP header (such as the contact field), MRS is not a SIP ALG. Refer to the NAT Traversal chapter in the IPedge Install manual.

Service Assignment

1. Click on the Service Assignment tab.
2. Click on the **New** icon.
3. SIP Trunk Channel Group = Channel Group tab number (Use the Channel group created above.)
4. Service number = Row number (Enter the digit 1 for the first assignment. Increment for each new assignment.)
5. Service Definition Index = Value create in service definitions tab.

Service URI

The SIP URI is the Telephone Number (TN) from the SIP Trunk provider.

1. Click on the Service URI tab.
2. Click on the **New** icon.
3. Service Definition Index: The service index that defines the SIP provider. This is the number assigned in .
4. SIP URI Number: This is the TN of the URI, typically this is the same as the CLID.
5. SIP URI User Name: Refer to your SIP Trunk provider.
6. SIP URI password: Refer to your SIP Trunk provider.
7. SIP URI Attribution: Typically the value is MAIN. If your SIP Trunk provider registers only the Primary number set the remaining numbers to SUB. When SUB is used the URI number cannot be used as the Calling Number.

Important! If a SIP URI (TN) is entered into more than one Service Definition Index certain system features may not function as expected. When processing a SIP call the system searches for the URI until the first match is found. If a URI is recorded in two Service Definition Indexes, assigned to two ILGs the SMDR records will only show the calls in one ILG.

CALL FORWARD ACTIONS (R1.2 and Later)

When a call, on a SIP Trunk, is forwarded out on another trunk, some SIP Trunk providers will allow the originating caller’s ID to display on the call forward destination phone as the Caller ID, rather than the IPedge URI. However, some providers may not support this.

Caller ID of Originating Caller Sent

By default **Number Verification** (Programmed in Enterprise Manager: Trunk > Calling Number > **Calling Number Identification**) is set to **Disable**. If the SIP Trunk provider supports this function the call will forward and the originating caller ID will be sent (The forwarded INVITE will contain the calling phone's PSTN ID in the FROM header). If the SIP provider does not support this function the call will not forward.

Caller ID That IPedge Sends

Some SIP Trunk providers require that the IPedge system send a valid, provisioned, calling number. In these cases set the program the IPedge SIP OLG as follows.

In Enterprise Manager or select **Trunk > Calling Number > Calling Number Identification** and set **Number Verification** to **Enable** for the SIP OLG.

The call will forward. The forwarding IPedge system URI will be displayed in the destination phone Caller ID display (The forwarded INVITE will contain the IPedge SIP Trunk URI in the FROM header.).

Note: The above discussion is call forward operation not Diversion Headers. IPedge systems do not support diversion header operation or Assert Identity.

Sending Caller ID from each station

Some SIP Trunk providers do not require that the IPedge system send a valid, provisioned, calling number. In these cases set the program the IPedge SIP OLG as follows.

1. In Enterprise Manager or select **Trunk > Calling Number > Calling Number Identification** and set **Number Verification** to **Disable** for the SIP OLG.
2. **System > System Data** set Default Calling number to enable.
3. **SIP Trunking > SIP URI Table** enter the number to be sent as a Main or Sub as determined by the pattern for your SIP provider. Refer to Table 2-1.

Note: If the SIP Trunk provider does not support this function the forwarded call will fail.

SIP TRUNK CONFIGURATION PATTERNS

The SIP trunks from service providers typically require IPedge configuration that conforms to one of the patterns shown in Table 2-2 through Table 2-8.

Patterns A and B are the most common. Some SIP Trunk providers and the typically used pattern are shown in Table 2-1.

Table 2-1 SIP Trunks Pattern Reference

Provider	Pattern	T.38 Support	Notes
123.net	B	Note 1	Enable Network Transfer (Service Def.)
8x8 (Note 2)	Other	Note 1	Contact 8x8 L2 setup for “No Plus”
AccessLine	A	Note 1	
AT&T	Other	Yes	Refer to AT&T IPedge configuration guide
Bright House Networks	B	No	SIP Trunk Option interval must be 0
Broadsoft	Note 3	Note 1	Refer to your SIP Trunk service provider
Broadvox	C	Note 1	Set the SIP URI attribute for additional numbers to SUB. Set the SIP Trunk Option Interval to 180.
Cbeyond (Note 2)	A	No	E911 Emergency destination can not be used on IPedge R1.2 and earlier systems.
Charter	B	Note 1	Ccontact Charter for a configuration guide.
Firstcomm (Note 2)	B	Note 1	Leave Domain Blank. Enter IP address provided by Firstcomm in SIP Server parameter.
Metaswitch	Note 3	Note 1	Refer to your SIP Trunk service provider
MM Internet	B	Note 1	Enable Network Transfer (Service Def.)
N2Net	B	No	Set the following to the SIP Server IP Address: SIP Trunk Message Option SIP Trunk Register Message From Header Option SIP Trunk Message To Header Option SIP Trunk Register Message to Header Option
Optimum	Other	Note 1	Contact Optimum for a configuration guide
TDS	B	Note 1	
Tierzero	A	Note 1	
Toshiba’s SIP Trunking I-VoIP Service	VIPedge SIP Trunk Pattern	No	Refer to Table 2-2. (IPedge systems require software TGZ 1.06.0026 or later)
Twist	A	Yes	

Notes:

1. Check with your SIP service provider about T.38 fax support.
2. Field tested
3. Refer to your SIP Trunk service provider for the appropriate configuration for this installation. (Sheet 1 of 2)

(Sheet 1 of 2)

Table 2-1 SIP Trunks Pattern Reference (continued)

Provider	Pattern	T.38 Support	Notes
Verizon (Note 2)	B	Note 1	Contact Verizon for the configuration guide for settings between IPedge and Acme packets.
Voice Carrier	B	Note 1	Disable Network Transfer (Service Definition)
XO Communications	B	Yes	SIP Trunk Option interval must be 0

Notes:

1. Check with your SIP service provider about T.38 fax support.
2. Field tested
3. Refer to your SIP Trunk service provider for the appropriate configuration for this installation. (Sheet 2 of 2)

(Sheet 2 of 2)

SIP Trunk Configuration Tables

The following tables show the typical SIP Trunk configuration patterns. The tables show the data entered in to the IPedge database using Enterprise Manager.

Some SIP Trunk providers may use a trunk number to activate a trunk. That trunk number will be the Main number. All of the rest of the directory numbers will be set to Sub.

Toshiba's SIP Trunking I-VoIP Service - The VIPedge SIP trunk portal will provide the Username and Password. Refer to Table 2-2.

Pattern A - Registration Mode With or Without Authentication - The SIP provider will typically provide the Username and Password. Refer to Table 2-3.

Pattern B - No Registration Mode and No Authentication - The IPedge server requires a static IP address. This address will be used instead of registration. Refer to Table 2-4.

Pattern C - Registration Mode with or without Authentication - The SIP provider will typically provide the Username and Password. The Port may be different than 5060 or no SRV records. Refer to Table 2-5.

Pattern D - No Registration Mode and No Authentication- The IPedge server requires a static IP address. This address will be used instead of registration. The Port may be different than 5060 or no SRV records. Refer to Table 2-6.

Pattern E - No Registration Mode With Authentication On - The SIP provider will typically provide the Username and Password although the provider generally does not require registration. Refer to Table 2-7.

Other: Different Than Patterns A ~ E - Consult with your SIP Trunk provider and Toshiba's Technical Support group. Refer to Table 2-8.

Table 2-2 Toshiba's SIP Trunking I-VoIP Service Pattern

Parameter	Entry
Trunk > SIP Trunking > Service Definition	
Registration Mode	Client
Domain Name	sip.outbound.vipedge.com
SIP Server	Leave blank
Primary Voice Packet Configuration	1
Primary Audio Codec	G711
Secondary Voice Packet Configuration	1
Secondary Audio Codec	G729
Connection to Media Relay Server	Manual (IPedge systems)
SIP Trunk Option Interval	60
SIP Trunk Message Option	FQDN (Default)
SIP Trunk Message to Header Option	FQDN (Default)
SIP Trunk Register Message From Header Option	FQDN (Default)
SIP Trunk Register Message To Header Option	FQDN (Default)
Trunk > SIP Trunking > Service URI	
The following values are obtained from the VIPedge SIP Trunk Admin portal.	
Trunk Number	
SIP URI	37412345 (example trunk number)
SIP URI User Name	37412345 (example trunk number)
SIP URI Password	1234 (example trunk password)
SIP URI Attribute	Main
DID Telephone Numbers	
SIP URI	19495833001 (1+10 digits) (TN example)
SIP URI User Name	37412345 (example trunk number)
SIP URI Password	1234 (example trunk password)
SIP URI Attribute	SUB

Table 2-3 Pattern A - Registration Mode With or Without Authentication

Parameter	Entry
Trunk > SIP Trunking > Service Definition	
Registration Mode	Client
Domain Name	SIP Provider IP address or domain name
SIP Server	Use an OutBound proxy if the SIP Provider requires
Primary Voice Packet Configuration	1
Primary Audio Codec	G729 or G711 (Consult your SIP provider.)
Secondary Voice Packet Configuration	1
Secondary Audio Codec	G711 or G729 (Assign the codec not used for as the primary.)
Network transfer	Typically Disabled (Test transfer with on and off to see which works.)
SIP Trunk Option Interval	0
SIP Trunk Message Option	Typically FQDN
SIP Trunk Message to Header Option	Typically FQDN
SIP Trunk Register Message From Header Option	Typically the same as SIP Trunk Message Option
SIP Trunk Register Message To Header Option	Typically the same as SIP Trunk Message to Header Option
Trunk -> SIP Trunking -> Service URI	
Primary Number	
SIP URI	9495833000 (example only)
SIP URI User Name	9495833000 (example only)
SIP URI Password	1234 (example only)
SIP URI Attribute	Main
Additional Numbers	
SIP URI	9495833001 (example only)
SIP URI User Name	9495833000 (example only)
SIP URI Password	1234 (example only)
SIP URI Attribute (When Reg mode is Client - Use sub if you do not want the number to register)	Main

Table 2-4 **Pattern B - No Registration Mode and No Authentication**

Parameter	Entry
Trunk > SIP Trunking > Service Definition	
Registration Mode	None
Domain Name	SIP Provider IP address or domain name
SIP Server	Use an OutBound proxy if the SIP Provider requires
Primary Voice Packet Configuration	1
Primary Audio Codec	G729 or G711 (Consult your SIP provider.)
Secondary Voice Packet Configuration	1
Secondary Audio Codec	G711 or G729 (Assign the codec not used for as the primary.)
Network transfer	Typically Disabled (Test transfer with on and off to see which works.)
SIP Trunk Option Interval	60
SIP Trunk Message Option	Typically FQDN
SIP Trunk Message to Header Option	Typically FQDN
SIP Trunk Register Message From Header Option	Typically the same as SIP Trunk Message Option
SIP Trunk Register Message To Header Option	Typically the same as SIP Trunk Message to Header Option
Trunk -> SIP Trunking -> Service URI	
Primary Number	
SIP URI	9495833000 (example only)
SIP URI User Name	
SIP URI Password	
SIP URI Attribute	Main
Additional Numbers	
SIP URI	9495833001 (example only)
SIP URI User Name	
SIP URI Password	
SIP URI Attribute (When Reg mode is Client - Use sub if you do not want the number to register)	Main

Table 2-5 Pattern C - Registration Mode with or without Authentication
The Port may be different than 5060 or no SRV records

Parameter	Entry
Trunk > SIP Trunking > Service Definition	
Registration Mode	Client
Domain Name	IP or domain name
SIP Server	IP or domain name: 5060 (Your SIP provider may use a different port)
Primary Voice Packet Configuration	1
Primary Audio Codec	G729 or G711 (Consult your SIP provider.)
Secondary Voice Packet Configuration	1
Secondary Audio Codec	G711 or G729 (Assign the codec not used for as the primary.)
Network transfer	Typically Disabled (Test transfer with on and off to see which works.)
SIP Trunk Option Interval	0
SIP Trunk Message Option	Typically FQDN
SIP Trunk Message to Header Option	Typically FQDN
SIP Trunk Register Message From Header Option	Typically the same as SIP Trunk Message Option
SIP Trunk Register Message To Header Option	Typically the same as SIP Trunk Message to Header Option
Trunk -> SIP Trunking -> Service URI	
Primary Number	
SIP URI	9495833000 (example only)
SIP URI User Name	9495833000 (example only)
SIP URI Password	1234 (example only)
SIP URI Attribute	Main
Additional Numbers	
SIP URI	9495833001 (example only)
SIP URI User Name	9495833000 (example only)
SIP URI Password	1234 (example only)
SIP URI Attribute (When Reg mode is Client - Use sub if you do not want the number to register)	Main

Table 2-6 Pattern D - No Registration Mode and No Authentication
The Port may be different than 5060 or no SRV records

Parameter	Entry
Trunk > SIP Trunking > Service Definition	
Registration Mode	None
Domain Name	IP or domain name
SIP Server	IP or domain name: 5060 (Your SIP provider may use a different port)
Primary Voice Packet Configuration	1
Primary Audio Codec	G729 or G711 (Consult your SIP provider.)
Secondary Voice Packet Configuration	1
Secondary Audio Codec	G711 or G729 (Assign the codec not used for as the primary.)
Network transfer	Typically Disabled (Test transfer with on and off to see which works.)
SIP Trunk Option Interval	60
SIP Trunk Message Option	Typically FQDN
SIP Trunk Message to Header Option	Typically FQDN
SIP Trunk Register Message From Header Option	Typically the same as SIP Trunk Message Option
SIP Trunk Register Message To Header Option	Typically the same as SIP Trunk Message to Header Option
Trunk -> SIP Trunking -> Service URI	
Primary Number	
SIP URI	9495833000 (example only)
SIP URI User Name	
SIP URI Password	
SIP URI Attribute	Main
Additional Numbers	
SIP URI	9495833001 (example only)
SIP URI User Name	
SIP URI Password	
SIP URI Attribute (When Reg mode is Client - Use sub if you do not want the number to register)	Main

Table 2-7 **Pattern E - No Registration Mode With Authentication On**

Parameter	Entry
Trunk > SIP Trunking > Service Definition	
Registration Mode	None
Domain Name	SIP Provider IP address or domain name
SIP Server	Use an OutBound proxy if the SIP Provider requires
Primary Voice Packet Configuration	1
Primary Audio Codec	G729 or G711 (Consult your SIP provider.)
Secondary Voice Packet Configuration	1
Secondary Audio Codec	G711 or G729 (Assign the codec not used for as the primary.)
Network transfer	Typically Disabled (Test transfer with on and off to see which works.)
SIP Trunk Option Interval	60
SIP Trunk Message Option	Typically FQDN
SIP Trunk Message to Header Option	Typically FQDN
SIP Trunk Register Message From Header Option	Typically the same as SIP Trunk Message Option
SIP Trunk Register Message To Header Option	Typically the same as SIP Trunk Message to Header Option
Trunk -> SIP Trunking -> Service URI	
Primary Number	
SIP URI	9495833000 (example only)
SIP URI User Name	9495833000 (example only)
SIP URI Password	1234 (example only)
SIP URI Attribute	Main
Additional Numbers	
SIP URI	9495833001 (example only)
SIP URI User Name	9495833000 (example only)
SIP URI Password	1234 (example only)
SIP URI Attribute (When Reg mode is Client - Use sub if you do not want the number to register)	Main

Table 2-8 Other: Different Than Patterns A ~ E

Parameter	Entry
Trunk > SIP Trunking > Service Definition	
Registration Mode	Consult with your SIP Trunk provider.
Domain Name	
SIP Server	
Primary Voice Packet Configuration	
Primary Audio Codec	
Secondary Voice Packet Configuration	
Secondary Audio Codec	
Network transfer	
SIP Trunk Option Interval (in a few cases use 0 when reg mode is none)	
SIP Trunk Message Option	
SIP Trunk Message to Header Option	
SIP Trunk Register Message From Header Option	
SIP Trunk Register Message To Header Option	
Trunk -> SIP Trunking -> Service URI	
Primary Number	
SIP URI	Consult with your SIP Trunk provider.
SIP URI User Name	
SIP URI Password	
SIP URI Attribute	
Additional Numbers	
SIP URI	Consult with your SIP Trunk provider.
SIP URI User Name	
SIP URI Password	
SIP URI Attribute (When Reg mode is Client - Use sub if you do not want the number to register)	

SIP RESPONSE MESSAGES

SIP response messages usually come from one of two sources:

- The SIP provider
- The IPedge server

From the SIP Provider

The conditions causing these messages may require consultation with the SIP Trunk service provider to resolve.

- **401** - Typically a challenge from the SIP service provider. Check the the user name and password set in the IPedge Service URI table.
- **403** - Typically a message that the URI may have an incorrect number of digits set in the IPedge Service URI table or:

  SIP Trunk Message options and SIP Trunk Register Message From Header option set in the Service Definition table is incorrect. Sometimes occurs when set to FQDN but should be set to IPU IP address (IPedge server IP address).

- **501** - Typically occurs when the Registration Mode is incorrect (change Client to None).

From the IPedge Server

The conditions causing these messages generally indicate incomplete or missing database programming.

- **403** - The DN digits sent by the SIP Trunk provider do not match the URI table entries. For example 9495833000 is sent from the SIP Trunk provider but this number is not in the URI table or was entered as 5833000.
- **403** (when an Adtran Gateway attempts a call to the IPedge system) - The From Header Host Type must be set to Local. Refer to the Adtran gateway configuration guide.
- **404 -** The DID number is missing (Trunk > DID programming).
- **480** - The DID number in the IPedge database is incorrect. Also caused if the destination IPT is: unplugged, set to DND, no System Call Forward is assigned, or is otherwise unreachable.
- **503** - Not enough channels assigned or all channels are in use.

Other Indicators

If a call drops at 32 seconds enable the NAT Transversal and MRS (R1.3 and later) or use a public IP address for the IPedge server (R1.2 and later). Refer to the NAT Traversal chapter.

- If there is no audio on a call check the IPT firmware version.
- If there is no MOH or no 3-way conference check the Media Server configuration.
- Jitter, Echo, Voice Quality issues; check bandwidth, router settings, perform a network assessment.
- SIP Trunks and voicemail were working have stopped working. Check for network security problems.

FIREWALL SETUP

Firewall setup is critical to IPedge and SIP Trunk operation. Refer to the IPedge Install manual for additional port and specific application information.

Note: Setup your firewall to ensure that the public WAN IP address is pointed to the IPedge server address.

SIP Trunk RTP Routing

For traversal of NAT firewalls without using a SIP ALG, the MRS is enabled and is set to manual, the RTP stream will flow through the IPedge rather than peer to peer. The MRS also changes the IP address and port in the Session Description Protocol (SDP). SDP connection information controls where the RTP stream is sent.

When using a NAT router with the IPedge server's NAT Traversal function disabled (the IPedge Public IP Address and Port for NAT field left blank), the private IP address in the SIP header is not changed. In this configuration a SIP ALG router will be required to change the private IP address to public IP address in fields in the SIP header (such as the contact field). In IPedge systems running R1.3 and later the SIP Trunk NAT Traversal feature can be used instead of the SIP ALG function in a router/firewall.

Note: Turning off SIP ALG in the router/firewall is recommended when using the NAT traversal feature.

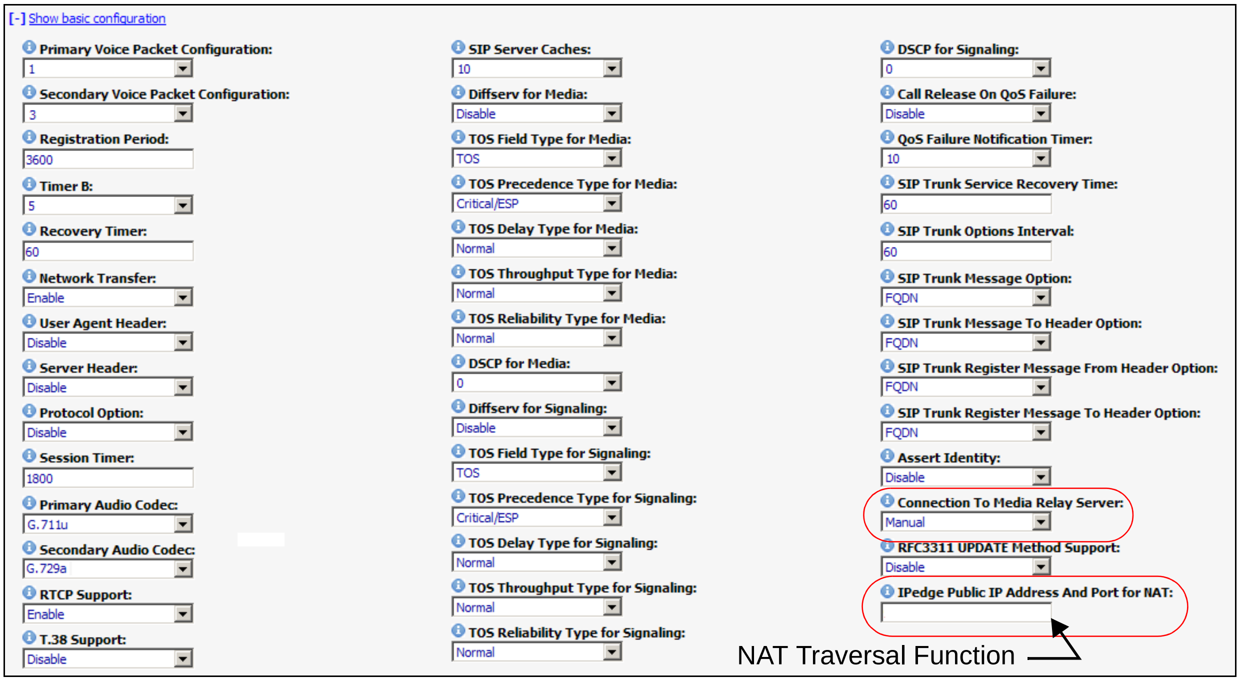
To set the SIP Trunk Connection to use the Media Relay Server with NAT Traversal capability use these steps.

1. In Enterprise Manager select **Trunk > SIP Trunking**.
2. Click to select the Service Definition tab.
3. Select the **Service Definition** number of the SIP Trunk.
4. Click on **Show advanced configuration.**
5. In the **Connection to Media Relay Server** field select **Manual**.
6. For the **IPedge Public IP Address and Port for NAT** field, enter the public IP address of the WAN interface for the router.

Firewall Ports

Refer to the IPedge Install manual for the IPedge Ports lists.

1. Click on the **Save** icon.



2. Configure a port forwarding rule in the NAT firewall to forward packets sent to the “IPedge Public IPaddress” and “Port for NAT”, to the IPedge servers local IP address, and to port 5060.

SIP TRUNK WIZARD

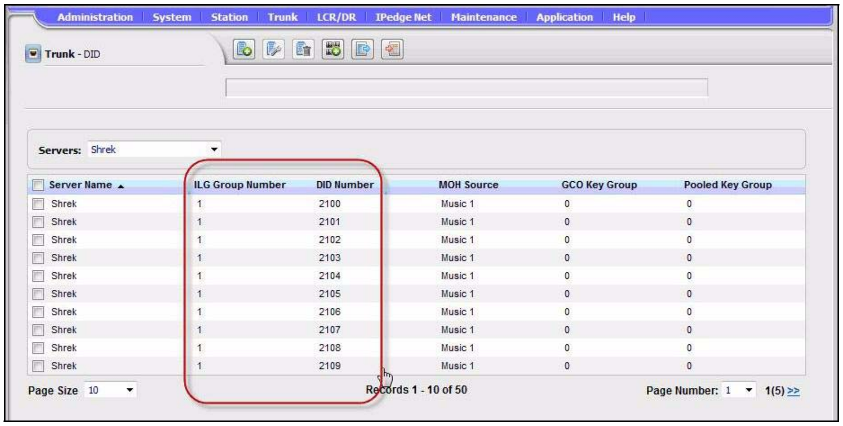
The SIP Trunk URI Wizard automates the creation of the URIs. The URIs can be created as a range of numbers and individually. (e.g. 2500 - 2574, 2602, 2605, 2700). In most IPedge systems using DID the SIP Trunk URIs include all of the DID numbers in the system. The SIP Trunk Wizard has the option to import the DID numbers already configured in the system. Using the wizard, depending on the size of the system, can save hours of programming time.

The procedure outline is:

- Program DID Numbers
- Setup ILGs and OLGs
- Setup SIP Trunks
- Use the wizard to create URIs for the DID numbers

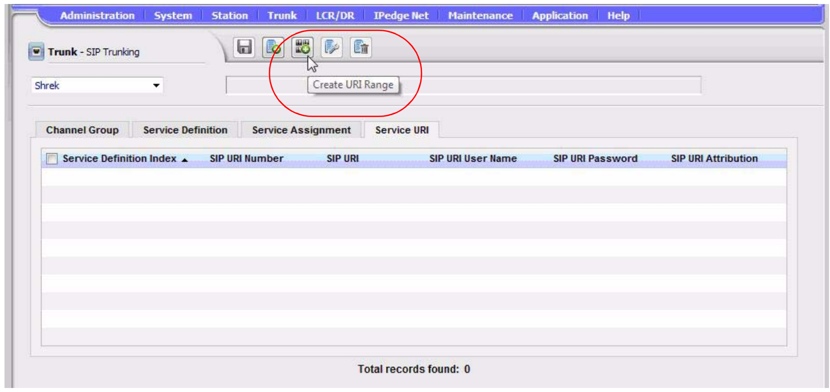
Create DID Numbers

1. Create the DID numbers.
   In the example below, 50 DID numbers, 2100 ~ 2149, were created for ILG=1. The ILG=1 group was created as a SIP/CO type incoming line group.



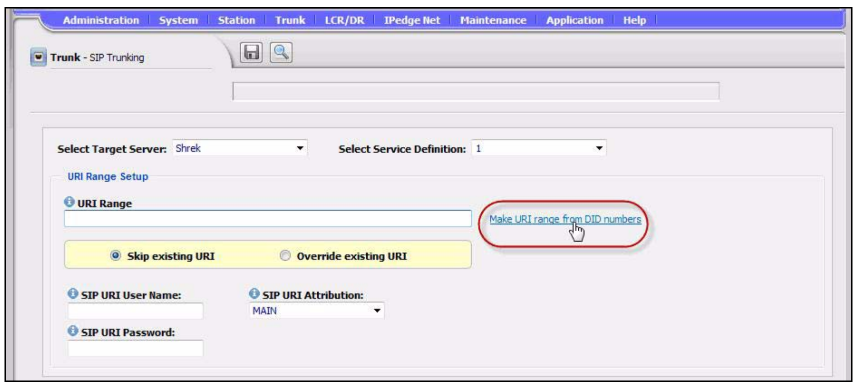
2. Because URIs belong to SIP trunks, the SIP trunks are created first. After the SIP Trunk channel group is assigned and the SIP Trunk service is defined, go to the Service URI tab and select the Create URI Range function.
3. On the Create URI Range form, the URIs are entered in the URI Range box. Multiple ranges or individual URIs can be entered manually, or the URI can be composed automatically by importing DID numbers already defined in the system. The wizard can be set to

skip or override any URI that already exists.

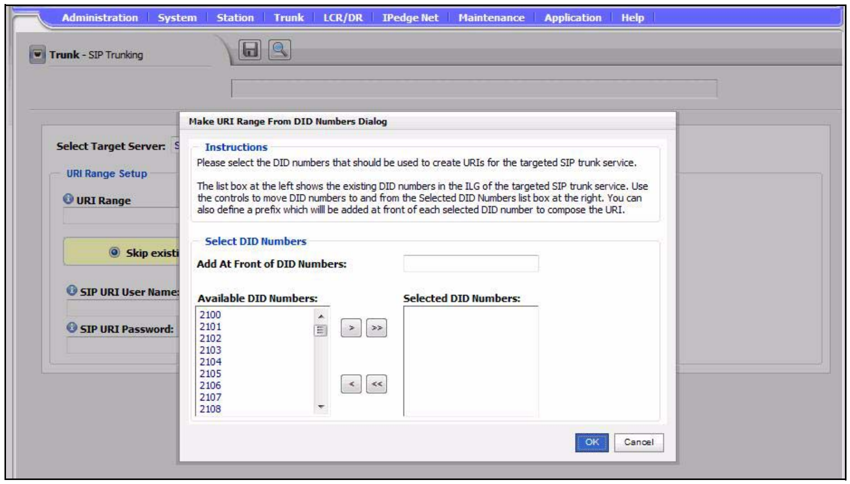


4. To enter the URIs for the DID number manually you must know all of the DID numbers. Then the numbers must be typed into the URI range field in the SIP Trunking screen in Enterprise Manager.

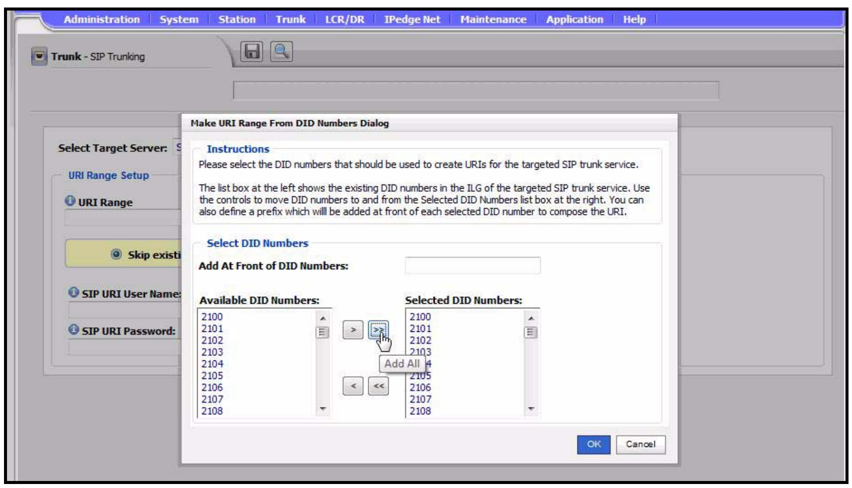
   To use the wizard click on the Create URI Range icon in the SIP Trunking, Service URI tab. In the URI Range Setup section click on Make URI range from DID numbers.



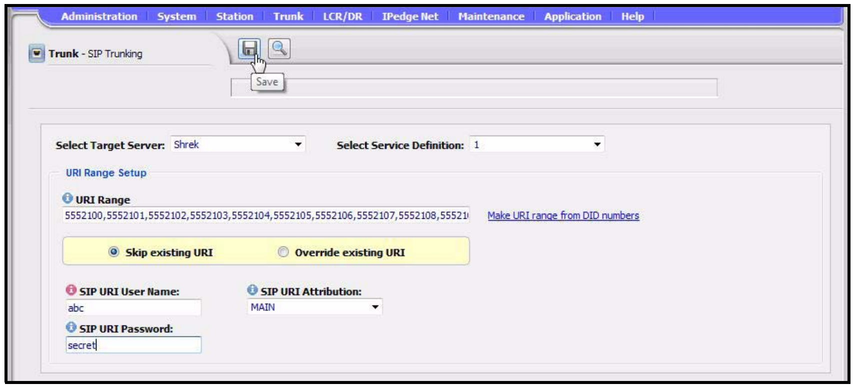
5. The Make URI Range From DID Numbers Dialog box will open. All of the DID numbers will be shown in the Available DID Numbers list.



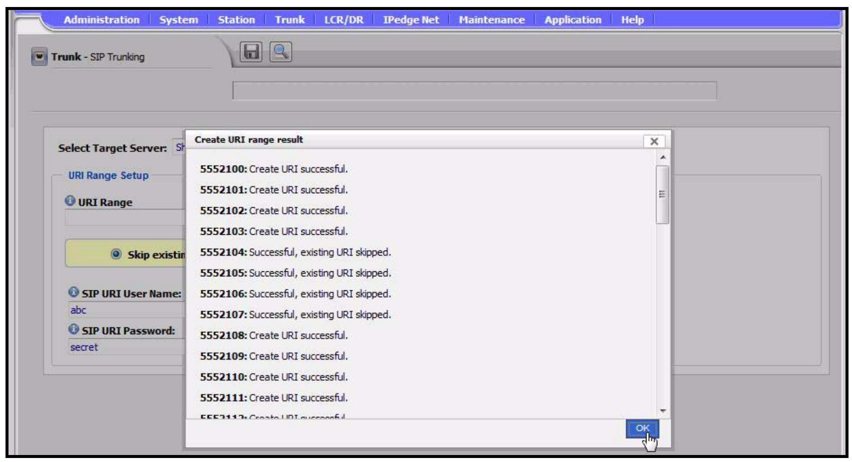
6. Move the DID numbers to Selected list. Note that there is an “Add All” button. Click on the OK button.



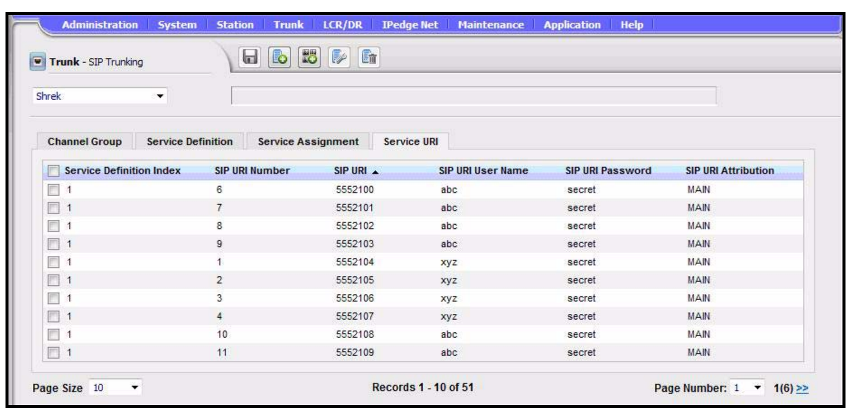
7. The URI range field on the SIP Trunking screen will be populated with the URIs (DID numbers) you selected. Click on the Save icon.



8. The URIs created will be displayed in a dialog box. Click on OK.



9. The SIP Trunking - Service URI screen will now show the updated list of URIs.



CAPACITY

VIPedge Capacities Within each VIPedge system SIP trunks are used to communicate with the PSTN. The SIP trunk will support the number of channels provisioned for the system.

IPedge Capacities The IPedge system can support up to 1000 URI entries. Refer to the table below.

Trunk Capacities

Trunks	EP Server	EC Server	EM Server
IPedge Net IP channels	20	96	440
SIP Trunk channels	20	96	440
Total Analog, T1, and ISDN trunk channels connected by gateways.	20	96	440
Channel Groups (One group for SIP trunks and one group for IPedge Net.)	2	2	2

911/E911 CALLS

It is imperative to ensure that E911 calls are routed correctly in all cases according to local and state laws.

Important! IP Phone users in locations where 911 calls can not be routed to the correct Public Safety Answering Point (PSAP) must maintain a wired land line phone or cell phone in order to make 911 calls to the correct PSAP.

SIP SIGNALING

IPedge system SIP Trunks will send SIP message 100 trying and 180 ringing in response to Invites, message 183 is not available as a session progress response.

REQUIREMENTS

- Contact the Toshiba Sales Applications Desk for the latest SIP Trunk Service provider list.
- License: I-CP-TRUNK
- IPedge system software TGZ 1.06.0026 or later required to support Toshiba’s SIP Trunking I-VoIP Service.

HARDWARE

No additional IPedge hardware is necessary to support SIP trunks.

Some system configurations may require a gateway. Refer to the IPedge Install document for gateway information.

FEATURE INTERACTION

Account Code	Voluntary Account code can be used while hearing DT if make a SIP trunk call. If incoming party is SIP terminal, SIP trunk, or terminals or trunks which are connected to gateway by SIP protocol, and outgoing party is SIP trunk, as forced account code cannot be entered, IPedge server disconnects incoming call. If incoming call comes from Loop trunk without release supervision, IPedge server cannot disconnect the call so that the administrator shall not set external number specifying the trunk with forced account code as the destination.
Automatic Busy Redial	If 486 Busy here is responded to INVITE, it is provided.
Automatic Call Back	Provided if all trunks of OLG are busy.
Automatic Campon	Provided if incoming call arrives at busy station.
Automatic Release of CO	When analog trunk is connected as SIP trunk via gateway, the call hangs up automatically by receiving BYE message which disconnecting signal is translated from analog network while talking while SIP trunk.
Call Forward	Provided.
Call History	Provided.
Call Transfer With Campon	If SIP trunk is set as transferred-to party or transferred party, the call does not camp on and terminate on transferring party as a recall. Because transferred party cannot enter external number while calling state.
Call Transfer	When 2 trunks connecting to IPedge server directly and one terminal belonging to IPedge server are talking and then transferring, signals can be transferred without IPedge server. (i.e. rerouting) If the call across nodes is transferred, the signals may be keep join connection. If public trunk supports, transferring above is possible. **Note:** Regarding Verizon, currently this is not mandatory. This is treated as Future.
Conferencing	SIP trunk can be entered as conference member.

Consultation Hold	SIP trunk cannot create the call state of similar Consultation Hold state.
Credit Card Calling	Provided if public trunk supports. Currently there is no service.
Dialed Number Identification Service	Provided.
Station Message Detail Record (SMDR)	Provided.
Tandem CO Line Connection	Provided.
Intercept	Provided.
Direct Inward Dialing	URI user name of To header which is received in SIP trunk termination is treated as DID number.
Direct Inward Termination	Not provided.
Do not Disturb	Provided.
Enhanced 911	Provided.
Least Cost Routing	Provided.
Line Group	Only One service index can be set for one ILG.
Manual Voice Recording	Provided.
Message Waiting	Provided if public trunk supports. **Note:** Regarding Verizon, currently this is not mandatory. This is treated as Future.
Music On Hold	Holding music source can be specified per each DID number.
Ring Transfer	Provided if public trunk supports. Regarding Verizon, currently this is not mandatory. This is treated as Future.

Station CO Line Access Provided.

Tenant Service Destination tenant cannot be set per DID number. Tenant service in the IPedge system is based on Stations not upon Trunks. DID routing to the station will determine the associated tenant

System Call Forward Provided as DID call.

Call Pick Up Provided.

- Phase 2 or later; Picking up the call by replace INVITE from the trunk can be done if the call state, CoS, and feature allow.

Station Hunting Provided.

Caller Identification Display Text and URI user name of From header which is received in SIP trunk termination is treated as caller number.

Audible Tone When originating to SIP public network, if “183 Session Progress” is received instead of “180 Ringing”, the caller hears tone which is provided by public network.

Other cases the caller hears tone which telephone or gateway provides.

If the far end party is in remote country, hearing tone may be different from tone defined in local country. (If inband tone)

Make Busy The trouble make busy is set per service index.

While conferencing, if SIP trunk goes made busy, the conference continues. At this time, the conference again includes SIP trunk party if SIP trunk becomes made idle.

Specified Caller Identification If Calling Number Verification is set to “Enable”:

- Specified caller number which can be sent via SIP trunk complies with
- SIP Trunking specification. To send a specified caller number via SIP trunk, URI whose user name is the same as the specified caller number is registered on SIP trunk provider. For making a call to SIP trunk with specified caller number, following all conditions shall be satisfied.
    - Subscribe to SIP trunk provider by URI with specified caller number in advance.
    - Specified caller number that a user wants to send to SIP trunk is set to IPedge.

	If a user uses specified caller number that is not registered as URI, CIX replace it by default caller number and send. If Calling Number Verification is set to “Disable”, Caller Number is notified to SIP Trunk even if it is the Caller Number without doing a) and b) above. **Note:** If SIP trunk call is transferred (CT) to SIP trunk by using Specified Caller Number with CONF key operation, REFER transfer is failed. (Call is connected by Join, so that SIP trunk is not released and SIP related resources for each trunk are consumed.) This is because URI when terminating and URI when originating a consultation call are different URI.
T.38 FAX Over IP	Depends on the service provider.
Secure communication	Provided secure communication by using TLS and SRTP.
End-to-End signalling	RFC2833/4733 or inband tone is used for both sending and receiving.
Line Hold	Holding operation from SIP network is specified by “Send Only” in SDP media direction. This is notified to held party as media information. Holding operation from IPedge extensions is not specified by media direction. By setting Music On Hold, ordinary holding music is played to the network side, and IPedge server does not notify holding operation.
SIP Extension	SIP trunk can be set as terminate-rejecting destination.
Through Dialing	SIP trunk can be set as the transferred-to destination of Through Dialing.
ISDN Basic Call Control	IPedge server connects ISDN trunk via gateway.
Call by Call service selection	This feature does not work even if IPedge server connects ISDN trunk via gateway.
2 B-channel transfer	This feature does not work even if IPedge server connects ISDN trunk via gateway.
Malicious Call Identification (MCID)	This feature does not work even if IPedge server connects ISDN trunk via gateway.
CTI Link Protocol	B channel number is stored on CSTA message for originating and terminating SIP trunk, however this B channel has no relationship with

	trunk URI.
Tracer for SIP trunk	Only calls via SIP trunk can be recorded by Tracer.