

Multi-Service Business Gateways

Enterprise Session Border Controllers

VoIP Media Gateways

Configuration Guide

IP-to-IP Application



Version 6.8

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1 Introduction

The IP-to-IP application refers to call routing of calls received from the IP and forwarded to an IP destination.

The device's IP-to-IP application supports IP-to-IP VoIP call routing (or SIP Trunking). The IP-to-IP call routing application enables enterprises to seamlessly connect their IP-based PBX (IP-PBX) to SIP trunks, typically provided by Internet Telephony Service Providers (ITSP). The device enables the enterprise to communicate with the PSTN network (local and overseas) through the ITSP, which interfaces directly with the PSTN. Therefore, the IP-to-IP application enables enterprises to replace the bundles of physical PSTN wires with SIP trunks provided by ITSPs and use VoIP to communicate within and outside the enterprise network using its standard Internet connection. At the same time, the device can also provide an interface with the traditional PSTN network, enabling PSTN fallback in case of IP connection failure with the ITSPs.

The device also supports multiple SIP Trunking. This can be useful in scenarios where if a connection to one ITSP fails, the call can immediately be transferred to another ITSP. In addition, by allowing multiple SIP trunks where each trunk is designated a specific ITSP, the device can route calls to an ITSP based on call destination (e.g., country code).

In addition to providing VoIP communication within the enterprise's LAN, the device enables the enterprise to communicate outside of the corporate LAN using SIP Trunking. This includes remote (roaming) IP-PBX users, for example, employees using their laptops to communicate with one another from anywhere in the world such as at airports.

The IP-to-IP application can be implemented by enterprises in the following example scenarios:

- VoIP between an enterprise's headquarters and remote branch offices
- VoIP between an enterprise and the PSTN through an ITSP

The IP-to-IP call routing capability is feature-rich, allowing interoperability with different ITSPs:

- Easy and smooth integration with multiple ITSP SIP trunks.
- Supports SIP registration and authentication with ITSP servers (on behalf of the enterprise's IP telephony system) even if the enterprise's IP telephony system does not support registration and authentication.
- Supports SIP-over-UDP, SIP-over-TCP, and SIP-over-TLS transport protocols, one of which is generally required by the ITSP.
- Provides alternative routing to different destinations (to another ITSP or the PSTN) when the connection with an ITSP network is down.
- Provides fallback to the legacy PSTN telephone network upon Internet connection failure.
- Provides Transcoding from G.711 to G.729 coder with the ITSP for bandwidth reduction.
- Supports SRTP, providing voice traffic security toward the ITSP.
- IP-to-IP routing can be used in combination with the regular Gateway application. For example, an incoming IP call can be sent to an E1/T1 span or it can be forwarded to an IP destination.

Therefore, the device provides the ideal interface between the enterprise IP-PBX and the ITSP SIP trunk.

The device's IP-to-IP application handles and terminates SIP methods and responses at each leg independently:

- Initiating-dialog INVITE: terminated at one leg and initiated on the other leg, 180\182\183\200\4xx uses the same logic and same limitations, in some cases the result may be a different response code.
- OPTIONS: terminated at each leg independently.

- INFO: only specific INFO's (such as DTMF) are handled; other types are omitted.
- UPDATE: terminated at each leg independently and may cause only changes in the RTP flow - Hold\Retrieve are the only exceptions that traverse the two legs.
- Re-INVITE: terminated at each leg independently and may cause only changes in the RTP flow - Hold\Retrieve are the only exceptions that traverse the two legs.
- PRACK: terminated at each leg independently.
- REFER (within a dialog): terminated at each leg independently.
- 3xx Responses: terminated at each leg independently.
- 401\407 responses to initial INVITE: in case the back-to-back session is associated with an Account, the responses are terminated at the receiving leg; in other cases, the responses are passed transparently.
- REGISTER: handled only in cases associated with a User-type IP Group - Contact, To, From specific parameters are omitted.

2 Theory of Operation

The device's IP-to-IP SIP session is performed by implementing Back-to-Back User Agent (B2BUA). The device acts as a user agent for both ends (*legs*) of the SIP call (from call establishment to termination). The session negotiation is performed independently for each call leg, using global parameters such as coders or using IP Profiles associated with each call leg to assign different configuration behaviors for these two IP-to-IP call legs.

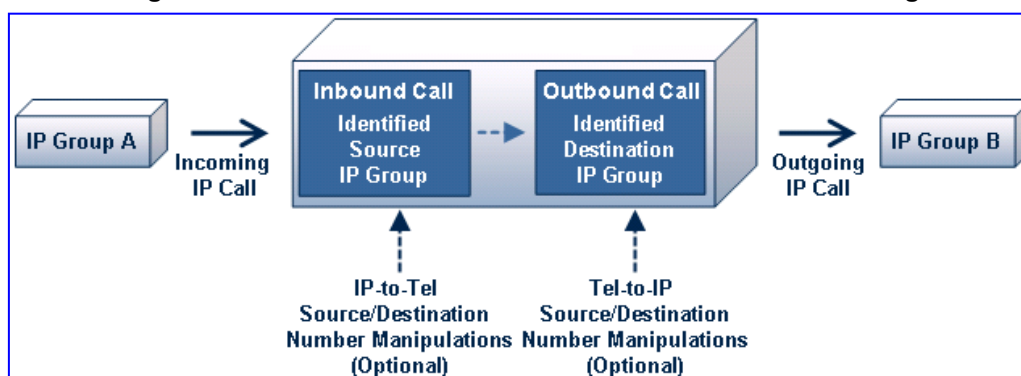
If transcoding is required, the RTP streams for IP-to-IP calls traverse the device and two DSP channels are allocated per IP-to-IP session. Therefore, the maximum number of IP-to-IP sessions is 60 (corresponding to a maximum of 120 media channels that can be designated for IP-to-IP call routing is).

If transcoding is not needed, the device supports up to 150 IP-to-IP SIP sessions (without using DSP channels).

The device also supports NAT traversal for SIP clients behind NAT, where the device is defined with a global IP address.

The figure below provides a simplified illustration of the device's handling of IP-to-IP call routing:

Figure 2-1: Basic Schema of the Device's IP-to-IP Call Handling



The basic IP-to-IP call handling process can be summarized as follows:

1. Incoming IP calls are identified as belonging to a specific logical entity in the network referred to as a *Source IP Group*, according to Inbound IP Routing rules.
2. The Source IP Group is sent to a specific IP Group referred to as a *Destination IP Group*; the IP destination address being as configured by the *Proxy Set* associated with the Destination IP Group.
3. Number manipulation can be done on inbound and outbound legs.

The following subsections discuss the main terms associated with the IP-to-IP call routing application.

2.1 Proxy Sets

A Proxy Set is a group of Proxy servers (for Proxy load balancing and redundancy) defined by IP address or fully qualified domain name (FQDN). The Proxy Set is assigned to Server-type IP Groups only, representing the address of the IP Group to where the device sends the INVITE message (i.e., the **destination** of the call). Typically, for IP-to-IP call routing, two Proxy Sets are defined for call destination – one for each leg (i.e., one for each IP Group) of the call (i.e., both directions).

2.2 IP Groups

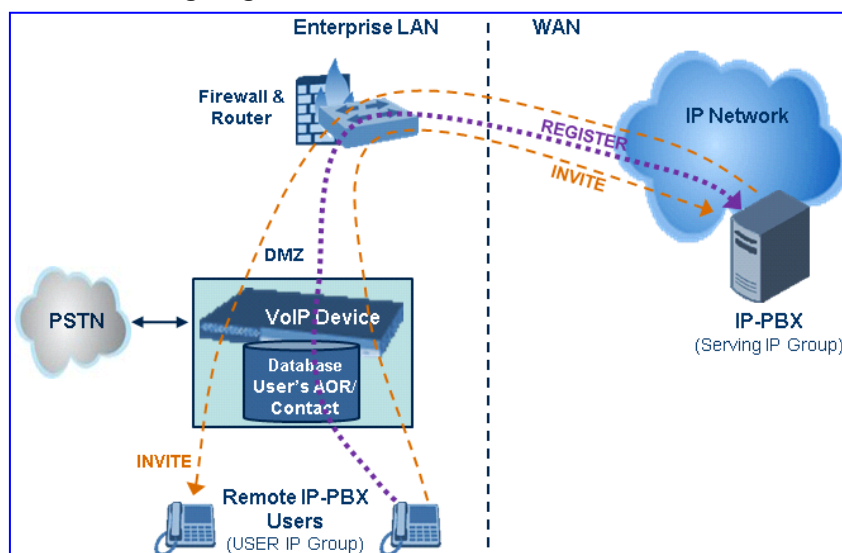
An IP Group represents a logical SIP entity in the device's network environment such as an ITSP SIP trunk, Proxy/Registrar server, IP-PBX, or remote IP-PBX users. The address of the IP Group is typically defined by its associated Proxy Set.

The opposite legs of the call are each presented by an IP Group; one being a *Serving* IP Group the other a *Served* IP Group. The Serving IP Group denotes the IP Group that provides service (e.g., ITSP) to the Served IP Group (e.g., IP-PBX). This is the IP Group to where the device sends INVITE messages received from the Served IP Group as well as REGISTER messages for registering on behalf of the Served IP Group.

IP Group can be a *Server* or *User* type. For Server-type IP Groups (e.g., ITSP or IP-PBX), the destination address (defined by the Proxy Set) is known. In contrast, User-type IP Groups represents groups of users whose location is dynamically obtained by the device when REGISTER requests and responses traverse (or are terminated) by the device. Generally, these are remote IP-PBX users (e.g., IP phones and soft phones).

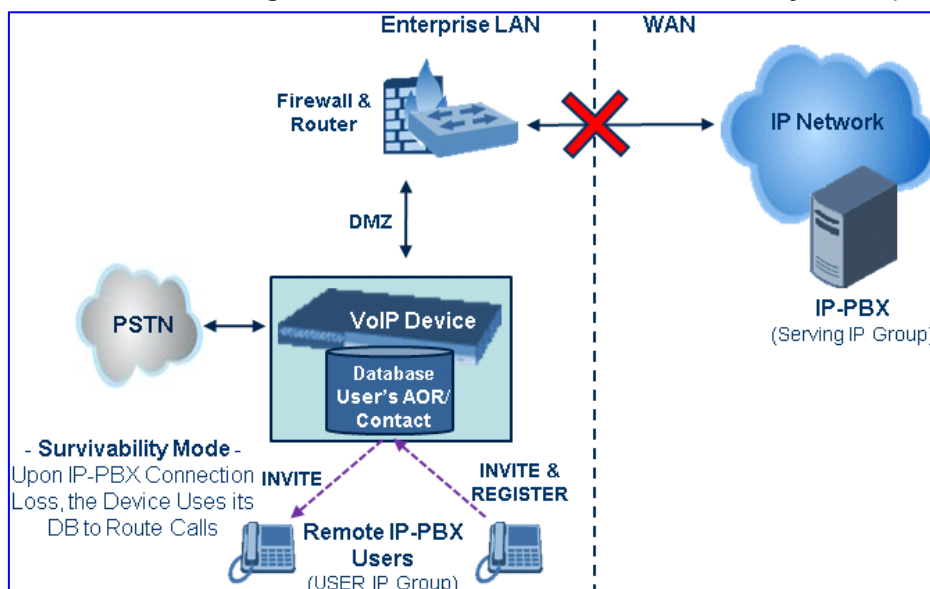
For registrations of User-type IP Groups, the device updates its internal database with the AOR and Contacts of the users (see the figure below) Digest authentication using SIP 401/407 responses, if needed, is done by the Serving IP Group (e.g., IP-PBX). The device forwards these responses directly to the remote SIP users. For a call to a registered remote user, the device searches its dynamic database using the Request URI for an entry that matches a registered AOR or Contact. Once an entry is found, the IP destination is obtained and a SIP request is then sent to the user.

Figure 2-2: IP-to-IP Routing/Registration/Authentication of Remote IP-PBX Users (Example)



The device also supports the IP-to-IP call routing Survivability mode feature (see the figure below) for User-type IP Groups. The device stores in its database REGISTER messages sent by the clients of the User-type IP Group. If communication with the Serving IP Group (e.g., IP-PBX) fails, the User-type IP Group enters into Survivability mode in which the device uses its database for routing calls between the clients of the User-type IP Group. The RTP packets between the clients traverse through the device. When the Serving IP Group is available again, the device returns to normal mode, sending INVITE and REGISTER messages to the Serving IP Group.

Figure 2-3: IP-to-IP Routing for IP-PBX Remote Users in Survivability Mode (Example)



2.3 Inbound and Outbound IP Routing Rules

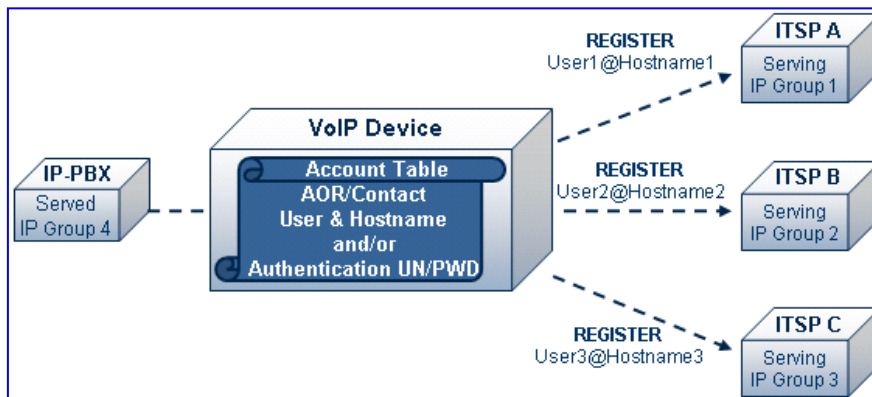
The device's IP-to-IP call routing is performed using the following two routing rule stages:

- 1. Inbound IP Routing Rule:** Identifies the received call as an IP-to-IP call based on various characteristics such as the call's source IP address, and assigns it to an IP Group.
- 2. Outbound IP Routing Rule:** Determines the destination (i.e., IP address) to where the incoming call associated with a specific source IP Group is finally routed. The destination address is typically denoted by another IP Group (destination IP Group) and therefore, the call is sent to the IP address that is defined by the Proxy Set associated with this IP Group. If the destination is a User-type IP Group, the device searches for a match between the request-URI of the received INVITE to an AOR registration record in the device's database. If a match is found, the INVITE is sent to the IP address of the registered contact.

2.4 Accounts

Accounts are used by the device to register to a Serving IP Group (e.g., an ITSP) on behalf of a Served IP Group (e.g., IP-PBX). This is necessary for ITSPs that require registration to provide services. Accounts are also used for defining user name and password for digest authentication (with or without registration) if required by the ITSP. Multiple Accounts per Served IP Group can be configured for registration to more than one Serving IP Group (e.g., an IP-PBX that requires registering to multiple ITSP's).

Figure 2-4: Registration with Multiple ITSP's on Behalf of IP-PBX



3 IP-to-IP Routing Configuration Example

This section provides a typical example on how to configure IP-to-IP call routing. In this example, the device serves as the communication interface between the enterprise's IP-PBX (located on the LAN) and the following network entities:

- ITSP SIP trunks (located on the WAN)
- Remote IP-PBX users (located on the WAN)
- Local PSTN network

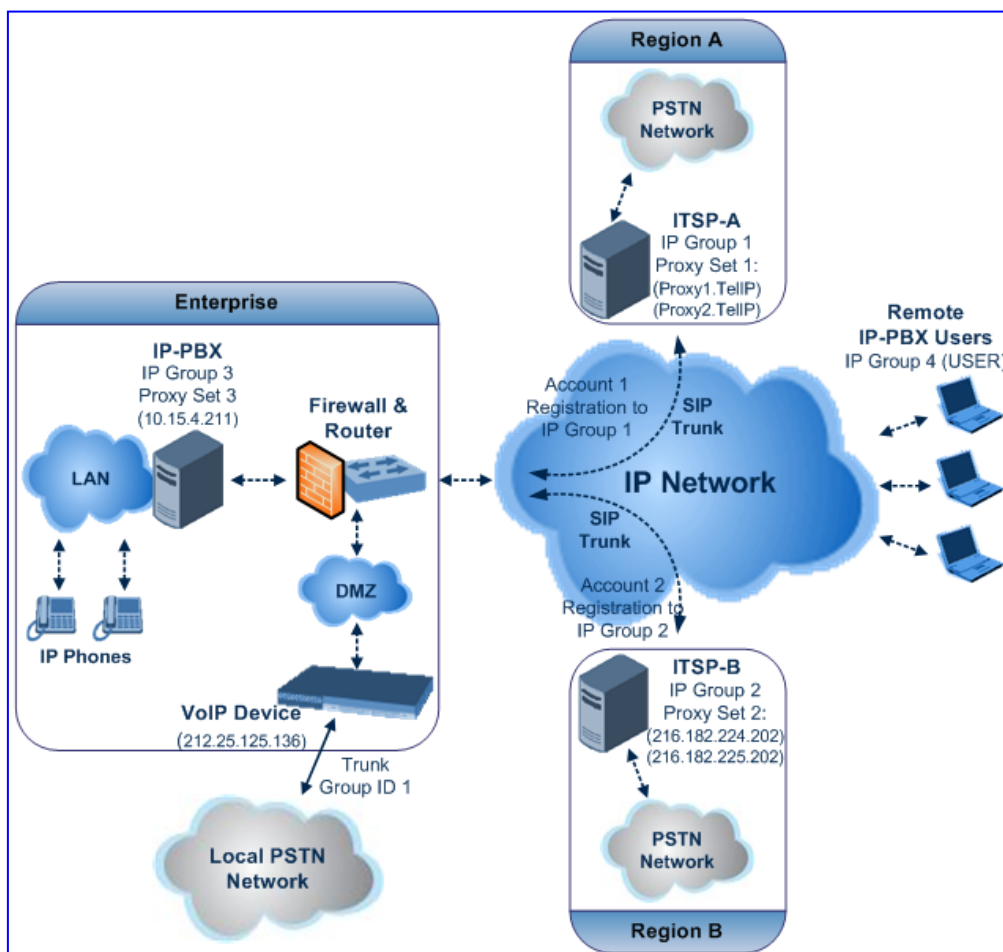
Calls from the Enterprise are routed according to destination.

This example assumes the following:

- The device has the public IP address 212.25.125.136 and is connected to the enterprise's firewall/NAT demilitarized zone (DMZ) network, providing the interface between the IP-PBX, and two ITSP's and the local PSTN.
- The enterprise has an IP-PBX located behind a Firewall/NAT:
 - IP-PBX IP address: 10.15.4.211
 - Transport protocol: UDP
 - Voice coder: G.711
 - IP-PBX users: 4-digit length extension number and served by two ITSPs.
 - The enterprise also includes remote IP-PBX users that communicate with the IP-PBX via the device. All dialed calls from the IP-PBX consisting of four digits starting with digit "4" are routed to the remote IP-PBX users.
- Using SIP trunks, the IP-PBX connects (via the device) to two different ITSP's:
 - **ITSP-A:**
 - ◆ Implements Proxy servers with fully qualified domain names (FQDN): "Proxy1.ITSP-A" and "Proxy2.ITSP-B", using TLS.
 - ◆ Allocates a range of PSTN numbers beginning with +1919, which is assigned to a range of IP-PBX users.
 - ◆ Voice coder: G.723.
 - **ITSP-B:**
 - ◆ Implements Proxy servers with IP addresses 216.182.224.202 and 216.182.225.202, using TCP.
 - ◆ Allocates a range of PSTN numbers beginning with 0200, which is assigned to a range of IP-PBX users.
 - ◆ Voice coder: G.723.
- Registration and authentication is required by both ITSP's, which is performed by the device on behalf of the IP-PBX. The SIP REGISTER messages use different URI's (host name and contact user) in the From, To, and Contact headers per ITSP as well as username and password authentication.
- Outgoing calls from IP-PBX users are routed according to destination:
 - If the calls are dialed with the prefix "+81", they are routed to ITSP-A (Region A).
 - If the calls are dialed with the prefix "9", they are routed to the local PSTN network.
 - For all other destinations, the calls are routed to ITSP-B.
- The device is also connected to the PSTN through a traditional T1 ISDN trunk for local incoming and outgoing calls. Calls dialed from the enterprise's IP-PBX with prefix '9' are sent to the local PSTN. In addition, in case of Internet interruption and loss of connection with the ITSP trunks, all calls are rerouted to the PSTN.

The figure below provides an illustration of this example scenario:

Figure 3-1: SIP Trunking Setup Scenario Example



The steps for configuring the device according to the scenario above can be summarized as follows:

- Enable the IP-to-IP feature - see 'Step 1: Enable the IP-to-IP Capabilities' on page 15.
- Configure the number of media channels - see 'Step 2: Configure the Number of Media Channels' on page 15.
- Configure a Trunk Group for interfacing with the local PSTN - see 'Step 3: Define a Trunk Group for the Local PSTN' on page 16.
- Configure Proxy Sets - see 'Step 4: Configure the Proxy Sets' on page 17.
- Configure IP Groups - see 'Step 5: Configure the IP Groups' on page 19.
- Configure Registration Accounts - see 'Step 6: Configure the Account Table' on page 20.
- Configure IP Profiles - see 'Step 7: Configure IP Profiles for Voice Coders' on page 21.
- Configure inbound IP routing rules - see 'Step 8: Configure Inbound IP Routing' on page 22.
- Configure outbound IP routing rules - see 'Step 9: Configure Outbound IP Routing' on page 25.
- Configure destination phone number manipulation - see 'Step 10: Configure Destination Phone Number Manipulation' on page 27.

3.1 Step 1: Enable the IP-to-IP Capabilities

This step describes how to enable the device's IP-to-IP application.

➤ **To enable IP-to-IP capabilities:**

1. Open the Applications Enabling page (**Configuration** tab > **VoIP** menu > **Applications Enabling** submenu > **Applications Enabling**).
2. From the 'IP to IP Application' drop-down list, select **Enable**.
3. Click **Submit**, and then save the setting to flash memory ("burn") with a device reset.



Note: For the IP-to-IP Application feature, the device must also be installed with the appropriate Software License Key.

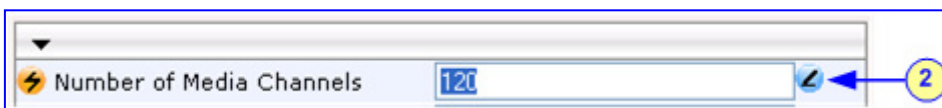
3.2 Step 2: Configure the Number of Media Channels

The number of media channels represents the number of digital signaling processors (DSP) channels that the device allocates to IP-to-IP calls. The remaining DSP channels can be used for PSTN calls. Two IP media channels are used per IP-to-IP call.

➤ **To configure the number of media channels:**

1. Open the IP Media Settings page (**Configuration** tab > **VoIP** menu > **IP Media** > **IP Media Settings**).

Figure 3-2: Defining Required Media Channels



2. In the 'Number of Media Channels' field, enter the required number of media channels (in the example above, "120" to enable up to 60 IP-to-IP calls).
3. Click **Submit**, and then save the settings to flash memory ("burn") with a device reset.

3.3 Step 3: Define a Trunk Group for the Local PSTN

For incoming and outgoing local PSTN calls with the IP-PBX, you need to define the Trunk Group (ID #1) for the T1 ISDN trunk connecting the device to the local PSTN. This Trunk Group is also used for alternative routing to the PSTN if connectivity with the ITSP fails.

➤ **To configure a Trunk Group for local PSTN:**

1. Open the Trunk Group Table page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** > **Trunk Group** > **Trunk Group**).
2. Configure Trunk Group ID #1 (as shown in the figure below):
 - 'From Trunk' and 'To Trunk':1 (to indicate Trunk 1 for this Trunk Group)
 - 'Channels': Enter the Trunk channels or ports assigned to the Trunk Group (e.g. 1-31 for E1 and 1-24 for T1)
 - 'Phone Number': Enter any phone number (logical) for this Trunk (e.g. 1000)
 - 'Trunk Group ID': "1"

| Add Phone Context As Prefix | | Disable | | | | | |
|-----------------------------|--------------|------------|----------|----------|--------------|----------------|----------------|
| Trunk Group Index | | 1-12 | | | | | |
| Group Index | Module | From Trunk | To Trunk | Channels | Phone Number | Trunk Group ID | Tel Profile ID |
| 1 | Module 1 PRI | 1 | 1 | 1-31 | 1000 | 1 | |
| 2 | | | | | | | |

3. Click **Submit** to apply your settings.
4. Configure the Trunk in the Trunk Settings page (**Configuration** tab > **VoIP** menu > **PSTN** > **Trunk Settings**).

3.4 Step 4: Configure the Proxy Sets

The Proxy Sets represent the actual destination (IP address or FQDN) to which the call is routed. These Proxy Sets are later assigned to IP Groups (see 'Step 5: Configure the IP Groups' on page 19). This step describes how to configure the following Proxy Sets:

- Proxy Set ID #1 with two FQDN's for ITSP-A
- Proxy Set ID #2 with two IP addresses for ITSP-B
- Proxy Set ID #3 with an IP address for the IP-PBX

➤ **To configure the Proxy Sets:**

1. Open the Proxy Sets Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **Proxy Sets Table**).
2. Configure Proxy Set ID #1 for ITSP-A:
 - a. 'Proxy Set ID': 1
 - b. 'Proxy Address': Enter the FQDN of ITSP-A SIP trunk Proxy servers (e.g., "Proxy1.ITSP-A" and "Proxy2. ITSP-A")
 - c. 'Transport Type': **TLS**
 - d. 'Enable Proxy Keep Alive': **Using Options**
 - e. 'Proxy Load Balancing Method': **Round Robin**

Figure 3-3: Proxy Set ID #1 for ITSP-A

The screenshot displays the configuration interface for Proxy Set ID #1. At the top, a dropdown menu shows 'Proxy Set ID' set to '1'. Below this is a table with two columns: 'Proxy Address' and 'Transport Type'. The table contains five rows, with the first two rows populated with 'Proxy1.ITSP-A' and 'Proxy2.ITSP-A', both with 'TLS' selected in the Transport Type column. Below the table is a form with several fields:

| | |
|-----------------------------|----------------|
| Proxy Name | dOE |
| Enable Proxy Keep Alive | Using Options |
| Proxy Keep Alive Time | 60 |
| Proxy Load Balancing Method | Round Robin |
| Is Proxy Hot Swap | No |
| Proxy Redundancy Mode | Not Configured |
| SRD Index | 0 |
| Classification Input | IP only |

3. Configure Proxy Set ID #2 for ITSP-B:
 - a. 'Proxy Set ID': 2
 - b. 'Proxy Address': Enter the IP addresses of the ITSP-B SIP trunk (e.g., "216.182.224.202" and "216.182.225.202")
 - c. 'Transport Type': **UDP**
 - d. 'Enable Proxy Keep Alive': **Using Options**

- e. 'Proxy Load Balancing Method': **Round Robin**

Figure 3-4: Proxy Set ID #2 for ITSP-B

| | | | |
|-----------------------------|-----------------|----------------|---|
| Proxy Set ID | | | 2 |
| | Proxy Address | Transport Type | |
| 1 | 216.182.224.202 | UDP | |
| 2 | 216.182.225.202 | UDP | |
| 3 | | | |
| 4 | | | |
| 5 | | | |
| Proxy Name | | | |
| Enable Proxy Keep Alive | | Using Options | |
| Proxy Keep Alive Time | | 60 | |
| Proxy Load Balancing Method | | Round Robin | |
| Is Proxy Hot Swap | | No | |
| Proxy Redundancy Mode | | Not Configured | |
| SRD Index | | 0 | |
| Classification Input | | IP only | |

4. Configure Proxy Set ID #3 for the IP-PBX:
 - a. 'Proxy Set ID': **3**
 - b. 'Proxy Address': Enter the IP address of the IP-PBX (e.g., "10.15.4.211")
 - c. 'Transport Type': **UDP**
 - d. 'Enable Proxy Keep Alive': **Using Options** (this is used in Survivability mode for remote IP-PBX users)

Figure 3-5: Proxy Set ID #3 for the IP-PBX

| | | | |
|-----------------------------|---------------|----------------|---|
| Proxy Set ID | | | 3 |
| | Proxy Address | Transport Type | |
| 1 | 10.15.4.211 | UDP | |
| 2 | | | |
| 3 | | | |
| 4 | | | |
| 5 | | | |
| Proxy Name | | | |
| Enable Proxy Keep Alive | | Using Options | |
| Proxy Keep Alive Time | | 60 | |
| Proxy Load Balancing Method | | Disable | |
| Is Proxy Hot Swap | | No | |
| Proxy Redundancy Mode | | Not Configured | |
| SRD Index | | 0 | |
| Classification Input | | IP only | |

3.5 Step 5: Configure the IP Groups

This step describes how to create the IP Groups for the following entities in the network:

- ITSP-A SIP trunk
- ITSP-B SIP trunk
- IP-PBX server
- IP-PBX remote users

These IP Groups are later used by the device for routing calls.

➤ **To configure the IP Groups:**

1. Open the IP Group Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **IP Group Table**).
2. Define IP Group #1 for ITSP-A:
 - a. 'Type': **Server**
 - b. 'Description': Enter an arbitrary name for the IP Group (e.g., "ITSP A")
 - c. 'Proxy Set ID': **1** (represents the IP addresses for communicating with this IP Group)
 - d. 'SIP Group Name': Enter the host name sent in the SIP Request From\To headers for this IP Group, as required by ITSP-A (e.g., "RegionA")
 - e. 'Contact User': Enter a name that is sent in the SIP Request's Contact header for this IP Group (e.g., "ITSP-A")
3. Define IP Group #2 for ITSP-B:
 - a. 'Type': **Server**
 - b. 'Description': Enter an arbitrary name for the IP Group (e.g., "ITSP B")
 - c. 'Proxy Set ID': **2**
 - d. 'SIP Group Name': Enter the host name sent in SIP Request From\To headers for this IP Group, as required by ITSP-B (e.g., "RegionB")
 - e. 'Contact User': Enter a name that is sent in the SIP Request Contact header for this IP Group (e.g., "ITSP-B")
4. Define IP Group #3 for the IP-PBX:
 - a. 'Type': **Server**
 - b. 'Description': Enter an arbitrary name for the IP Group (e.g., "IP-PBX")
 - c. 'Proxy Set ID': **3**
 - d. 'SIP Group Name': Enter the host name that is sent in SIP Request From\To headers for this IP Group (e.g., "IPPBX")
 - e. 'Contact User': Enter a name that is sent in the SIP Request Contact header for this IP Group (e.g., "PBXUSER")
5. Define IP Group #4 for the remote IP-PBX users:
 - a. 'Type': **User**
 - b. 'Description': Enter an arbitrary name for the IP Group (e.g., "IP-PBX")
 - c. 'SIP Group Name': Enter the host name that is used internal in the device's database for this IP Group (e.g., "RemoteIPPBXusers")
 - d. 'Serving IP Group ID': **3** (i.e. the IP Group for the IP-PBX)



Note: No Serving IP Groups are defined for ITSP-A and ITSP-B. Instead, the Outbound IP Routing table (see 'Step 9: Configure Outbound IP Routing' on page 25) is used to configure outbound IP call routing for calls originating from these ITSP IP Groups.

3.6 Step 6: Configure the Account Table

The Account table is used by the device to register to an ITSP on behalf of the IP-PBX. As described previously, the ITSP requires registration and authentication to provide service. In this example, the Served IP Group is the IP-PBX (IP Group ID #3) and the Serving IP Groups are the two ITSPs (IP Groups #1 and #2).

➤ **To configure the Account table:**

1. Open the Account Table page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Account Table**).

Figure 3-6: Defining Accounts for Registration

| Index | Served Trunk Group | Served IP Group | Serving IP Group | Username | Password |
|-------|--------------------|-----------------|------------------|----------|----------|
| 1 | -1 | 3 | 1 | itsp_a | * |
| 2 | -1 | 3 | 2 | itsp_b | * |

| Host Name | Register | Contact User | Application Type |
|-----------|----------|--------------|------------------|
| regiona | Yes | ITSP-A | GWIP2IP |
| regionb | Yes | ITSP-B | GWIP2IP |

2. Configure Account ID **#1** for IP-PBX authentication and registration with ITSP-A:
 - a. 'Served IP Group': "3" (authentication is performed on behalf of IP Group #3, i.e., the IP-PBX)
 - b. 'Serving IP Group': "1" (registration/authentication is with IP Group #1, i.e., ITSP-A)
 - c. 'User Name': Enter the SIP username for authentication supplied by ITSP-A (e.g., "itsp_a")
 - d. 'Password': Enter the SIP password for authentication supplied by ITSP-A (e.g., "12345")
 - e. 'Register': **Yes** (enables registration with ITSP-A)
3. Configure Account ID **#2** for IP-PBX registration) with ITSP-B Registrar server:
 - 'Served IP Group': "3" (registration is performed on behalf of IP Group #3, i.e., the IP-PBX)
 - 'Serving IP Group': "2" (registration is with IP Group #3, e.g., ITSP-B)
 - 'User Name': Enter the SIP username for the registration/authentication supplied by ITSP-B (e.g., "itsp_b")
 - 'Password': Enter the SIP password for registration/authentication supplied by ITSP-B (e.g., "11111")
 - 'Register': **Yes**

3.7 Step 7: Configure IP Profiles for Voice Coders

Since different voice coders are used by the IP-PBX (G.711) and the ITSPs (G.723), you need to define two IP Profiles:

- Profile ID #1 - configured with G.711 for the IP-PBX
- Profile ID #2 - configured with G.723 for the ITSPs

These profiles are later used in the Inbound IP Routing table and Outbound IP Routing table.

➤ **To configure IP Profiles for voice coders:**

1. Open the Coder Group Settings page (**Configuration** tab > **VoIP** menu > **Coders and Profiles** > **Coders Group Settings**)
2. Configure Coder Group ID #1 for the IP-PBX (as shown in the figure below):
 - a. 'Coder Group ID': 1.
 - b. 'Coder Name': **G.711A-law**.
 - c. Click **Submit**.

Figure 3-7: Defining Coder Group ID 1

| Coder Group ID: 1 | | | | |
|-------------------|--------------------|------|--------------|---------------------|
| Coder Name | Packetization Time | Rate | Payload Type | Silence Suppression |
| G.711A-law | 20 | 64 | 8 | Disabled |
| | | | | |

3. Configure Coder Group ID #2 for the ITSP's (as shown in the figure below):
 - a. 'Coder Group ID': 2.
 - b. 'Coder Name': **G.723.1**.
 - c. Click **Submit**.

Figure 3-8: Defining Coder Group ID 2

| Coder Group ID: 2 | | | | |
|-------------------|--------------------|------|--------------|---------------------|
| Coder Name | Packetization Time | Rate | Payload Type | Silence Suppression |
| G.723.1 | 30 | 5.3 | 4 | Disabled |
| | | | | |

4. Open the IP Profile Settings page (**Configuration** tab > **VoIP** menu > **Coders and Profiles** > **IP Profile Settings**).
5. Configure Profile ID #1 for the IP-PBX (as shown below):
 - a. 'Profile ID': 1.
 - b. 'Coder Group ID': **Coder group 1**.

- c. Click **Submit**.

Figure 3-9: Defining IP Profile ID 1

| Common | | GW | SBC |
|--|----------------|----|-----|
| Index | 1 | | |
| Coders Group ID | Coders group 1 | | |
| Fax Signaling Method | No Fax | | |
| Remote RTP Base UDP Port | 0 | | |
| CNG Detector Mode | Disable | | |
| Vxx Modem Transport Type | Enable Bypass | | |
| NSE Mode | Disable | | |
| Is DTMF Used | Disable | | |
| Play RB Tone to IP | Disable | | |
| Early Media | Disable | | |
| Progress Indicator to IP | Not Configured | | |
| Copy Destination Number to Redirect Number | Disable | | |
| Media Security Behavior | Preferable | | |
| Number of Calls Limit | -1 | | |
| First Tx DTMF Option | RFC 2833 | | |
| Second Tx DTMF Option | | | |
| Rx DTMF Option | Supported | | |

6. Configure Profile ID #2 for the ITSP's:
 - a. 'Profile ID': 2.
 - b. 'Coder Group': **Coder Group 2**.
 - c. Click **Submit**.

3.8 Step 8: Configure Inbound IP Routing

This step defines how to configure the device for routing inbound (i.e., received) IP-to-IP calls. The table in which this is configured uses the IP Groups that you defined in 'Step 5: Configure the IP Groups' on page 19.

➤ **To configure inbound IP routing:**

1. Open the Inbound IP Routing Table page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** submenu > **Routing** submenu > **IP to Trunk Group Routing**).

Figure 3-10: Defining Inbound IP Routing Rules

| | Dest. Host Prefix | Source Host Prefix | Dest. Phone Prefix | Source Phone Prefix | Source IP Address | Trunk Group ID | IP Profile ID | Source IP Group ID |
|---|-------------------|--------------------|--------------------|---------------------|-------------------|----------------|---------------|--------------------|
| 1 | | | 9 | * | * | 1 | 0 | |
| 2 | | | * | * | 10.15.4.211 | -1 | 1 | 3 |
| 3 | | | +1919 | * | * | -1 | 2 | 1 |
| 4 | | | 0200 | * | * | -1 | 2 | 2 |
| 5 | | pbxremote | * | * | * | -1 | 0 | 4 |
| 6 | | | * | * | 10.15.4.211 | 1 | 0 | -1 |

2. **Index #1:** Routes calls with prefix 9 (i.e., local calls) dialed from IP-PBX users to the local PSTN:
 - 'Dest Phone Prefix': "9" (dialing prefix for local calls).
 - 'Trunk Group ID': "1" (these calls are routed to Trunk Group #1 connected between the device and the local PSTN network).
3. **Index #2:** Identifies IP calls received from the IP-PBX as IP-to-IP calls and assigns them to the IP Group ID configured for the IP-PBX:
 - 'Dest Phone Prefix': "*" (all destinations).
 - 'Source IP Address': Enter the IP address of the IP-PBX (i.e., "10.15.4.211").
 - 'Trunk Group ID': "-1" (indicates that these calls are IP-to-IP calls).
 - 'IP Profile ID': "1" (assigns call to Profile ID #1 to use G.711).
 - 'Source IP Group ID': "3" (assigns call to the IP Group pertaining to the IP-PBX).
4. **Index #3:** Identifies IP calls received from ITSP-A as IP-to-IP calls and assigns them to the IP Group ID configured for ITSP-A:
 - 'Dest Phone Prefix': ITSP-A assigns the Enterprise a range of numbers that start with +1919. Enter this prefix to indicate calls received from this ITSP.
 - 'Trunk Group ID': "-1" (indicates IP-to-IP calls).
 - 'IP Profile ID': "2" (assigns call to Profile ID #2 to use G.723).
 - 'Source IP Group ID': "1" (assigns call to IP Group pertaining to ITSP-A).
5. **Index #4:** Identifies IP calls received from ITSP-B as IP-to-IP calls and assigns them to the IP Group ID configured for ITSP-B:
 - 'Dest Phone Prefix': ITSP-B assigns the Enterprise a range of numbers that start with 0200. Enter this prefix to indicate calls coming from this ITSP.
 - 'Trunk Group ID': "-1" (indicates IP-to-IP calls).
 - 'IP Profile ID': "2" (assigns call to Profile ID #2 to use G.723).
 - 'Source IP Group ID': "2" (assigns call to IP Group pertaining to ITSP-B).

6. **Index #5:** Identifies all IP calls received from IP-PBX remote users:
 - 'Source Host Prefix': "PBXuser" (host name in the From header of the Request-URI received from remote IP-PBX users).
 - 'Trunk Group ID': enter "-1" (indicates IP-to-IP calls).
 - 'Source IP Group ID': "4" (assigns call to the IP Group pertaining to the remote IP-PBX users).
7. **Index #6:** Identifies IP calls received from the IP-PBX that can't reach the ITSP's servers (e.g. loss of connection with ITSP's) and routes them to the local PSTN network (i.e., alternative routing):
 - 'Dest Phone Prefix': "*" (all destinations).
 - 'Source IP Address': Enter the IP address of the IP-PBX (i.e., "10.15.4.211").
 - 'Trunk Group ID': "1" (routes call to the Trunk Group connected to the local PSTN).
 - 'Source IP Group ID': "-1" (indicates that these calls are not assigned to any source IP Group).

3.9 Step 9: Configure Outbound IP Routing

This step defines how to configure the device for routing outbound (i.e., sent) IP-to-IP calls. In our example scenario, calls from both ITSPs must be routed to the IP-PBX, while outgoing calls from IP-PBX users must be routed according to destination. If the calls are destined to the Japanese market, then they are routed to ITSP-B; for all other destinations, the calls are routed to ITSP-A. This configuration uses the IP Groups defined in 'Step 5: Configure the IP Groups' on page 19 and IP Profiles defined in 'Step 7: Configure IP Profiles for Voice Coders' on page 21.

➤ **To configure outbound IP routing rules:**

1. Open the Outbound IP Routing Table page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** > **Routing** > **Tel to IP Routing**).
2. **Index #1:** Routes IP calls received from ITSP-A to the IP-PBX:
 - 'Source IP Group ID': **1** (received / inbound calls identified as belonging to the IP Group configured for ITSP-A).
 - 'Dest Phone Prefix' and 'Source Phone Prefix': "*" (all destinations and callers).
 - 'Dest IP Group ID': **3** (destination IP Group to where the calls are sent, i.e., to the IP-PBX).
 - 'IP Profile ID': "2" (IP Profile configured for G.723).
3. **Index #2:** Routes IP calls received from ITSP-B to the IP-PBX:
 - 'Source IP Group ID': **2** (received / inbound calls identified as belonging to the IP Group configured for ITSP-B).
 - 'Dest Phone Prefix' and 'Source Phone Prefix': "*" (all destinations and callers).
 - 'Dest IP Group ID': **3** (destination IP Group to where these calls are sent, i.e., to the IP-PBX).
 - 'IP Profile ID': "2" (IP Profile configured for G.723).
4. **Index #3:** Routes calls received from the local PSTN network to the IP-PBX:
 - 'Source Trunk Group ID': "1" (calls received on the trunk connecting the device to the local PSTN network).
 - 'Dest IP Group ID': **3** (destination IP Group to where the calls must be sent, i.e., to the IP-PBX).
5. **Index #4:** Routes IP calls received from the IP-PBX to ITSP-A:
 - 'Source IP Group ID': **3** (received calls identified as belonging to the IP Group configured for the IP-PBX).
 - 'Dest Phone Prefix': "+81" (calls to Japan, i.e., with prefix +81).
 - 'Source Phone Prefix': "*" (all sources).
 - 'Dest IP Group ID': **1** (destination IP Group to where the calls must be sent, i.e., to ITSP-A).
 - 'IP Profile ID': "1" (IP Profile configured for G.711).
6. **Index #5:** Routes IP calls received from the IP-PBX to ITSP-B:
 - 'Source IP Group ID': **3** (received calls identified as belonging to the IP Group configured for the IP-PBX).
 - 'Dest Phone Prefix' and 'Source Phone Prefix': "*" (all destinations besides Japan, and all sources).
 - 'Dest IP Group ID': **2** (destination IP Group to where the calls must be sent, i.e., to ITSP-A).
 - 'IP Profile ID': "1" (IP Profile configured for G.711).

7. **Index #6:** Routes dialed calls, four digits starting with digit 4, from IP-PBX to remote IP-PBX users. The device searches its database for the remote users registered number and then sends an INVITE to the remote user's IP address (listed in the database):
- 'Source IP Group ID': **3** (received calls identified as belonging to the IP Group configured for the IP-PBX).
 - 'Dest Phone Prefix': "4xxx#" (all calls dialed from IP-PBX that include four digits and start with the digit 4).
 - 'Dest IP Group ID': **4** (destination IP Group to where the calls must be sent, i.e., to remote IP-PBX users).
 - 'IP Profile ID': "1" (IP Profile configured for G.711).

3.10 Step 10: Configure Destination Phone Number Manipulation

This step defines how to manipulate the destination phone number. The IP-PBX users in our example scenario use a 4-digit extension number. The incoming calls from the ITSPs have different prefixes and different lengths. This manipulation leaves only the four digits of the user's destination number coming from the ITSPs.

➤ **To configure destination phone number manipulation rules:**

1. Open the Destination Phone Number Manipulation Table for IP -> Tel calls page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** > **Manipulations** > **Destination Number Tel->IP**).
2. Configure Index #1 to manipulate destination number of IP calls received from ITSP-A. The phone number of calls received with prefix +1919 (i.e., from ITSP-A) are removed except for the last four digits:
 - 'Destination Prefix': "+1919".
 - 'Source Prefix': "*" (all sources).
 - 'Number of Digits to Leave': "4" (to leave only the last four digits).
3. Configure Index #2 to manipulate destination number of IP calls received from ITSP-B. The phone number of calls received with prefix 0200 (i.e., from ITSP-B) are removed except for the last four digits:
 - 'Destination Prefix': "0200".
 - 'Source Prefix': "*" (all sources).
 - 'Number of Digits to Leave': "4" (to leave only the last four digits).



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