



# DSP-1282 & DSP-1283 Crestron Avia™ DSP with Mitel® 8.0 PR3 Platform

Configuration Guide  
Crestron Electronics, Inc.

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# DSP-1282 & DSP-1283: SIP Endpoint with Mitel® 8.0 PR3 Platform

## Introduction

This configuration guide describes the procedures required to configure Crestron Avia™ Digital Signal Processor (DSP) devices. The devices operate on the MiVoice Business (Mitel® PBX) as basic Session Initiation Protocol (SIP) users.

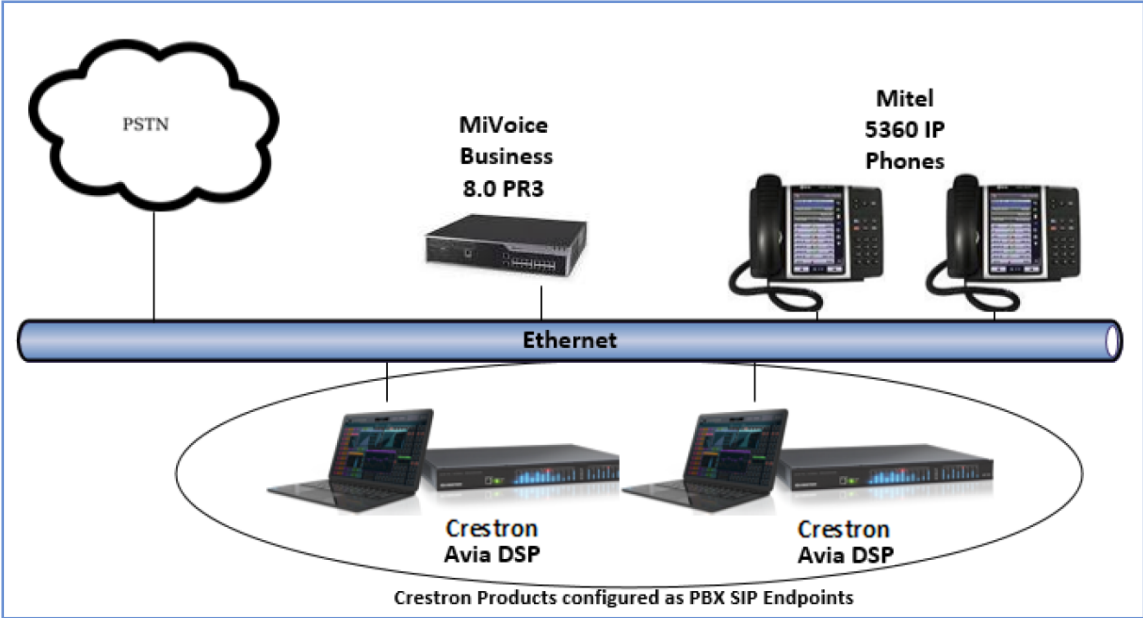
## Audience

The intended audience includes those attempting to configure and use Crestron Avia DSP devices as SIP endpoints registered to MiVoice Business (Mitel PBX).

## Topology

The diagram below shows the network topology for integration of a Crestron Avia DSP endpoint with MiVoice Business (Mitel PBX).

SIP Endpoint Integration with MiVoice Business (Mitel PBX) - Reference Network



The lab network consists of the following components:

- Mitel PBX
- Mitel phones
- Crestron Avia DSP as SIP users

## Software Requirements

- MiVoice Business (Mitel PBX): 8.0 PR3
- Crestron Avia DSP: v1.00.121

## Hardware Requirements

- MiVoice Business (Mitel PBX) either in a virtual environment or with a hardware server
- Public Switched Telephone Network (PSTN) gateway
- MiTel Phones - 5360 IP phones (2)
- Crestron Avia DSP devices (2)

## Product Description

The Crestron Avia DSP products (DSP-1282 and DSP-1283, specifically) consist of a family of programmable digital audio signal processors intended for the commercial sound market. Each version provides 12 analog mic/line inputs and eight analog line outputs. The devices include a Local Area Network (LAN) connection and a Universal Serial Bus (USB) connection for programming and control. The programmable signal flow is a fixed topology with user-configurable input and output processing chains using a library of preset signal-specific DSP blocks.

Use the Crestron Avia tool to control and configure the Crestron Avia DSP devices on the network.

Use the Crestron Toolbox™ software to discover and control all Crestron devices on the network.

## Summary

This document describes how to configure the Crestron Avia DSP devices as SIP users. It also provides information on how to register devices to the Mitel PBX with digest authentication.

Supported features include:

- Registration with digest authentication
- Basic calls with G711u and G711a codecs
- Dual-Tone Multi-Frequency (DTMF) support
- Early media support

- Retrieval of a parked call
- Transferee in a call transfer
- Conference participant
- Member of hunt group
- DND (Do Not Disturb)

Unsupported features include:

- Caller ID presentation
- Call hold and resume
- Call forwarding on the device (forwarding can be configured on the Private Branch Exchange (PBX) for the Domain Name (DN) assigned to the endpoint)
- Call waiting
- Initiating conference
- Initiating an attended call transfer
- Initiating an early attended call transfer
- Initiating a blind call transfer
- Shared line (configuration of shared line on device)
- Initiating call park
- Message Waiting Indicator (MWI)

Known issues and limitations include:

- The DSP does not support Music on Hold when integrated with the Mitel PBX.
- No support for caller ID on the Crestron Avia DSP.
- No support for MWI on the Crestron Avia DSP.
- When registered to Mitel, the DSP is not available to accept calls after a power cycle unless the previous call is disconnected.
- In a Mitel environment, a call declined by the DSP does not provide appropriate treatment to the calling party.
- The DSP fails to play a reorder tone when a call from the DSP to a PBX extension eventually times out after the called party does not answer.

# Crestron Avia DSP Configuration

This section provides the following details:

- How to set up connections to the amplifier and speaker
- How to access the DSP on the network (once powered)
- How to configure the DSP for registration and integration with the Mitel PBX

## Connections

Make the following connections:

- Connect microphone to DSP MIC/LINE INPUTS port 1
- Connect DSP LINE OUTPUTS port 1 to "Audio In" on amplifier
- Connect "Audio Out" of amplifier to speaker
- Connect LAN port to network
- Connect VOIP port to network

## Device Discovery/Access

Use the Crestron Toolbox and the Crestron Avia tool to discover and access the connected LAN and/or VOIP ports) DSP devices.

Use the Help menu to assist when performing the discovery and configuration procedure.

## Device Configuration

The basic setup for a phone call requires:

- An analog input (such as from a microphone) routed out through the phone line
- Audio coming in from the phone line routed to an analog output (such as to an amplifier or speaker)

## Configure the DSP Device

Use the Crestron Avia tool to select and configure the DSP device.

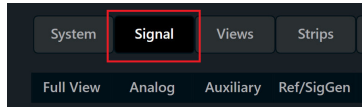


## Input Configuration

To configure the analog input:

1. Click **Signal**.

### Crestron Avia tool: Audio Input Configuration (1/4)



2. Under **Analog In 1** (first row), double click **Gain**. In the new window set the following:
  - a. Click **Mute** to **Off**.
  - b. Select **33** for the **Analog Gain**.
  - c. If a condenser microphone is being used, click **+48V** (phantom power) to **On**.

### Crestron Avia Tool: Audio Input Configuration (2/4)



3. Under **Analog In 1** (first row), click **Ref/Phone Out** (right-most column) and enter **0** as the decibel value.

### Crestron Avia Tool: Audio Input Configuration (3/4)



4. Under **Phone\Sig Gen In**, click **PHN**. In the new window set the following:
  - a. Move the **Send Level** slider to **0 db**.
  - b. Click **Mute** to **Off**.

### Crestron Avia Tool: Audio Input Configuration (4/4)

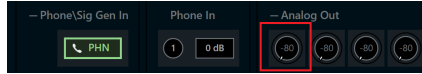


## Output Configuration

To configure the analog output:

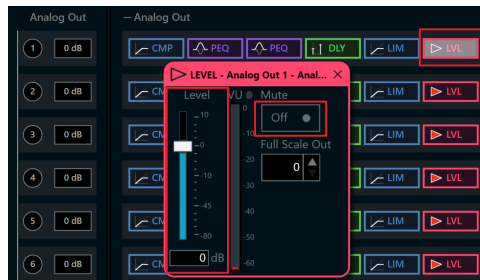
1. Under **Phone In 1** (first row), click **Analog Out** (left-most column) and enter **0** as the decibel value.

### Crestron Avia Tool: Audio Output Configuration (1/3)



2. Under **Analog Out 1**, double click **LVL**. In the new window set the following:
  - a. Move the **Level** slider to **0 db**.
  - b. Click **Mute** to **Off**.

### Crestron Avia Tool: Audio Output Configuration (2/3)



3. Under **Phone\Sig Gen In**, click **PHN**. In the new window set the following:
  - a. Move the **Receive Level** slider to **0 db**.
  - b. Click **Mute** to **Off**.

### Crestron Avia Tool: Audio Output Configuration (3/3)



## Configure the SIP Parameters

From the open **PHN - In 1 - Phone Receive** window, select and configure the SIP parameters.

1. With **SIP** selected, click the chevron at the right top corner to expand the window.  
**Crestron Avia Tool: Phone Dialer, SIP Parameters Configuration**



2. Enter the extension configured on Mitel for the **Local Extension** for this device. This example uses **5000**.
3. Enter the Mitel PBX for the **SIP Server IP Address**. This example uses **10.35.32.2**.
4. Enter the SIP server port (**5060**) for the **Port**.
5. Enter the same end user name configured for the Mitel PBX with the digest authentication credentials for the **SIP Server User Name**.
6. Enter the same password as configured for the Mitel PBX end user digest credentials for the **SIP Server Password**.

# Mitel Configuration

This section describes the MiVoice Business system (Mitel PBX) configuration necessary to support registration of Crestron devices and connectivity to Public Switched Telephone Network (PSTN).

**NOTE:** Confirm that the general installation and basic Mitel configuration have been administered.

## Verify Licenses

Ensure that adequate licenses are available in the MiVoice Business System to support the Mitel phones and Crestron devices.

Click **Licenses > License and Option Selection** in the MiVoice Business controller. Each Crestron device uses one IP user license.

### Mitel: License Verification

The screenshot shows the Mitel MiVoice Business web interface. The top navigation bar includes the Mitel logo, 'MiVoice Business', and the alarm status 'Local\_2' with a 'Clear' button and the timestamp '2017-Apr-04 09:20:07'. The left sidebar contains a menu with categories like 'Licenses', 'LAN/WAN Configuration', 'Voice Network', 'System Properties', 'Hardware', 'Trunks', 'Users and Devices', and 'Integrated Directory Service'. The 'Licenses' section is expanded, showing 'License and Option Selection' as the active page. The main content area is titled 'License and Option Selection on Local\_2' and includes a 'Change' button, a 'DN to search' dropdown, and 'Print...' and 'Import...' buttons. Below this, there is a section for 'Online Licensing with the Application Management Center' showing an 'Application Record ID' of 26682859. A table displays system information:

System Type	License Sharing	Hardware Identifier
Enterprise	No	0000003a1a4f

Below the system information, there is a table for 'Licensed Options' with columns for 'Locally Consumed', 'Locally Allocated', 'Available for Allocation', and 'Purchased'. A sub-section for 'Users' is also visible, with a table showing 'IP Users' with 11 locally consumed, 16 locally allocated, 0 available for allocation, and 16 purchased licenses.

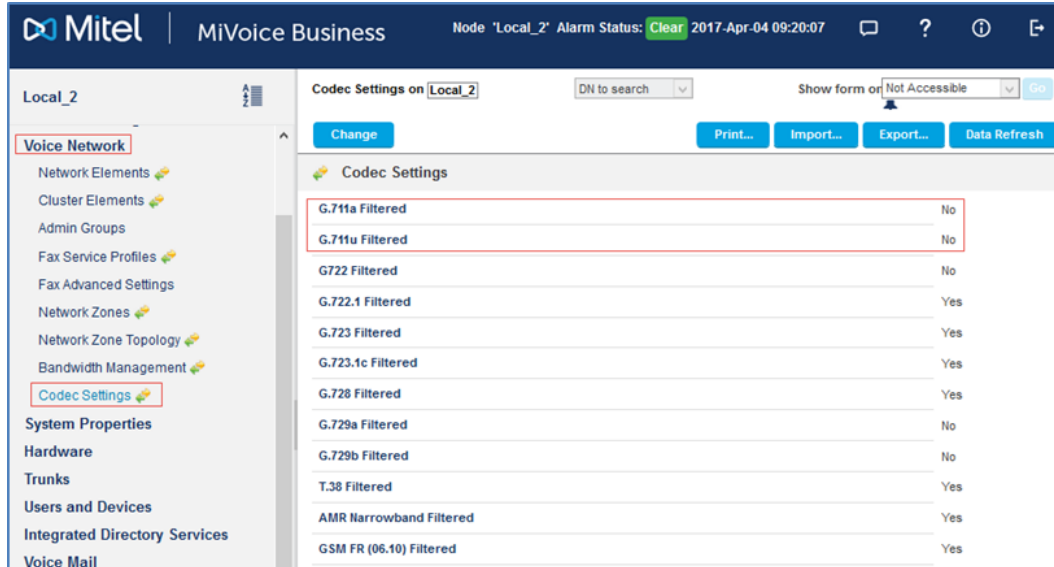
Licensed Options	Locally Consumed	Locally Allocated	Available for Allocation	Purchased
Users				
IP Users	11	16	0	16

## Configure Codec Settings and Network Zones

Configure codec settings to allow G711u and G711a codec negotiation.

Click **Voice Network > Codec Settings** and configure the codec filtering as shown.

Mitel: Codec Settings



Local\_2

Codec Settings on Local\_2

Change Print... Import... Export... Data Refresh

Codec Settings

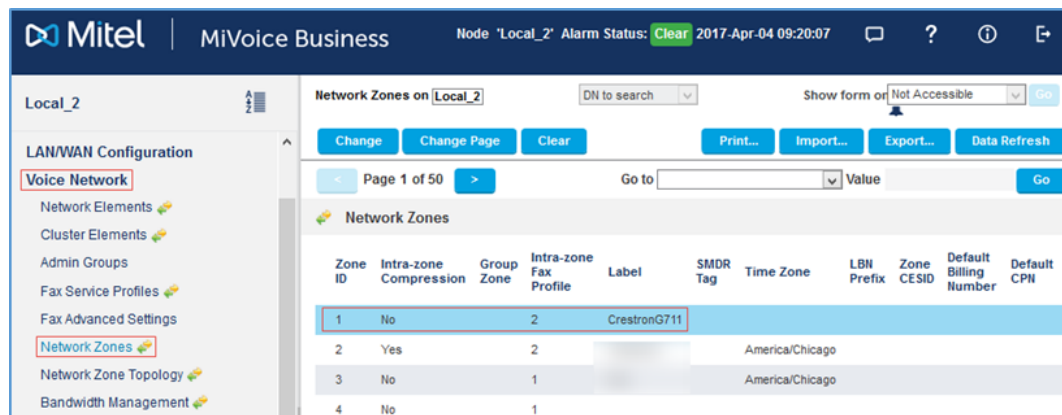
Codec	Filtered
G.711a Filtered	No
G.711u Filtered	No
G.722 Filtered	No
G.722.1 Filtered	Yes
G.723 Filtered	Yes
G.723.1c Filtered	Yes
G.728 Filtered	Yes
G.729a Filtered	No
G.729b Filtered	No
T.38 Filtered	Yes
AMR Narrowband Filtered	Yes
GSM FR (06.10) Filtered	Yes

**NOTE:** By default, use zone 1 to negotiate G711u as the preferred codec.

To configure Network Zone 1 to offer G711u as the preferred codec:

1. Click **Voice Network > Network Zones**.

Mitel: Configure Network Zones



Local\_2

Network Zones on Local\_2

Change Change Page Clear Print... Import... Export... Data Refresh

Page 1 of 50 Go to Value Go

Network Zones

Zone ID	Intra-zone Compression	Group Zone	Intra-zone Fax Profile	Label	SMDR Tag	Time Zone	LBN Prefix	Zone CESID	Default Billing Number	Default CPH
1	No	2		CrestronG711						
2	Yes	2				America/Chicago				
3	No	1				America/Chicago				
4	No	1								

2. Select a **Zone ID** to modify. This example uses **1**.
3. Select **No** for the **Intra-zone Compression**.

## Configure Network Element

To create a network element for the PSTN gateway:

1. Click **Voice Network > Network Elements**.
2. Click **Add**.

Mitel: Configure Network Element

The screenshot displays the Mitel MIVoice Business web interface. The top header shows 'Local\_2' and 'MIVoice Business' with a status indicator 'Node 'Local\_2' Alarm Status: Clear' and a timestamp '2017-Apr-04 09:20:07'. The left sidebar contains a navigation menu with categories like 'Licenses', 'LAN/WAN Configuration', 'Voice Network', 'System Properties', 'Hardware', 'Trunks', 'Users and Devices', 'Integrated Directory Service', 'Voice Mail', 'Call Routing', 'Music On Hold', and 'Emergency Services Manag'. The 'Voice Network' section is expanded, and 'Network Elements' is selected. The main content area shows a table of Network Elements with columns for Name, Type, PBX Number/Cluster Element ID, FQDN or IP Address, Data Sharing, Version, and Zone. A table with one row is visible:

Name	Type	PBX Number/Cluster Element ID	FQDN or IP Address	Data Sharing	Version	Zone
PSTN_GW	Other	---	10.64.1.72	NO	1	1

Below the table, the configuration form for the selected 'PSTN\_GW' element is shown. The fields are:

- Name: PSTN\_GW
- Type: Other
- FQDN or IP Address: 10.64.1.72
- Data Sharing: NO
- Local: False
- Version: 1
- Zone: 1
- ARID: (empty)
- SIP Peer Specific:
  - SIP Peer Transport: default
  - SIP Peer Port: 5060
  - External SIP Proxy FQDN or IP Address: (empty)
  - External SIP Proxy Transport: default
  - External SIP Proxy Port: 0
  - SIP Registrar FQDN or IP Address: (empty)
  - SIP Registrar Transport: default
  - SIP Registrar Port: 0
  - SIP Peer Status: Auto-Detect/Normal

3. Enter **PSTN\_GW** for the **Name** (for this example).
4. Enter **Other** for the **Type**.
5. Enter **10.64.1.72** for the **FQDN or IP Address** (for this example, which is the IP address of the PSTN gateway).
6. Enter **1** for the **Zone**. This setting ensures a G711u and G711a codec.
7. In the **SIP Peer Specific** section, do the following:
  - a. Enter **5060** for the **SIP Peer Port** (for this example).
  - b. Leave all other fields at the default values.
8. Click **Save**.

## Configure Class of Service

To configure class of service:

1. Click **System Properties > System Feature Settings > Class of Service Options**.
2. Select **10** for the **Class Of Service Number** (for this example).
3. Click **Change**.

Mitel: Class of Service Options 10 (1/9)

The screenshot shows the Mitel MiVoice Business configuration interface. The top navigation bar includes the Mitel logo, 'MiVoice Business', and a status bar with 'Local\_2', 'Alarm Status: Clear', and a timestamp '2017-Apr-04 09:20:07'. The main content area is titled 'Class of Service Options on Local\_2'. On the left, a sidebar menu lists various configuration categories, with 'Class of Service Options' highlighted. The main panel is divided into 'General' and 'Advanced' tabs, with 'General' selected. The 'General' tab contains several configuration fields: 'Class Of Service Number' (set to 10), 'Comment' (set to Crestron), 'ACD Agent Behavior on No Answer' (set to Logout), 'ACD Agent No Answer Timer' (set to 15), 'ACD Make Busy on Login' (radio buttons for No/Yes), 'ACD Silent Monitor Accept' (radio buttons for No/Yes), 'ACD Silent Monitor Accept Monitoring Non-Prime Lines' (radio buttons for No/Yes), 'ACD Silent Monitor Allowed' (radio buttons for No/Yes), 'ACD Silent Monitor Notification' (radio buttons for No/Yes), 'Follow 2nd Alternate Reroute for Recall to Busy ACD Agent' (radio buttons for No/Yes), 'Work Timer' (set to 0), 'Announce' section with 'Call Announce Line' (radio buttons for No/Yes), 'Off-Hook Voice Announce Allowed' (radio buttons for No/Yes), and 'Handsfree AnswerBack Allowed' (radio buttons for No/Yes). The 'Busy Override' section includes 'Busy Override Security' (radio buttons for No/Yes, with 'Yes' selected) and 'Disable Executive Busy Override Tone' (radio buttons for No/Yes). Buttons for 'Change', 'Copy', 'Print...', 'Import...', 'Export...', 'Data Refresh', 'Save', and 'Cancel' are visible at the top of the configuration area.

4. In the **General** tab, do the following:
  - a. Click **Yes** for **Busy Override Security**.

Mitel: Class of Service Options 10 (2/9)

General	Advanced
<b>Call Control Timer</b>	
Busy Tone Timer	30
Dialing Conflict Timer	3
First Digit Timer	15
Inter Digit Timer	10
Lockout Timer	45
<b>Call Duration</b>	
Call Duration	10
Call Duration Forced Cleardown Timer	0
Enable Call Duration Limit on External Calls	<input checked="" type="radio"/> No <input type="radio"/> Yes
Enable Call Duration Limit on Internal Calls	<input checked="" type="radio"/> No <input type="radio"/> Yes
<b>Call Forwarding/Rerouting</b>	
Call Forward - Delay	0
Call Forward No Answer Timer	15
Call Forward Override	<input checked="" type="radio"/> No <input type="radio"/> Yes
Call Forwarding (External Destination)	<input type="radio"/> No <input checked="" type="radio"/> Yes
Call Forwarding (Internal Destination)	<input type="radio"/> No <input checked="" type="radio"/> Yes
Call Forwarding Accept	<input type="radio"/> No <input checked="" type="radio"/> Yes
Call Reroute after CFFM to Busy Destination	<input checked="" type="radio"/> No <input type="radio"/> Yes
Call Forwarding Reminder Ring (CFFM and CFIAH only)	<input checked="" type="radio"/> No <input type="radio"/> Yes
Disable Call Reroute Chaining On Diversion	<input checked="" type="radio"/> No <input type="radio"/> Yes



### Mitel: Class of Service Options 10 (3/9)

General	Advanced
Follow Reroute on Disabled Forwarding	<input checked="" type="radio"/> No <input type="radio"/> Yes
Group Call Forward Follow Me Accept	<input checked="" type="radio"/> No <input type="radio"/> Yes
Group Call Forward Follow Me Allow	<input checked="" type="radio"/> No <input type="radio"/> Yes
Third Party Call Forward Follow Me Accept	<input checked="" type="radio"/> No <input type="radio"/> Yes
Third Party Call Forward Follow Me Allow	<input checked="" type="radio"/> No <input type="radio"/> Yes
Use Held Party Device for Call Re-routing	<input type="radio"/> No <input checked="" type="radio"/> Yes
<b>Call Hold</b>	
Call Hold	<input type="radio"/> No <input checked="" type="radio"/> Yes
Call Hold - Retrieve with Hold Key	<input type="radio"/> No <input checked="" type="radio"/> Yes
Call Hold Remote Retrieve	<input type="radio"/> No <input checked="" type="radio"/> Yes
Call Hold Timer	30
Local Music On Hold source	<input checked="" type="radio"/> No <input type="radio"/> Yes
Music on Hold on Transfer	<input type="radio"/> No <input checked="" type="radio"/> Yes
Use Called Party Call Hold Timer	<input checked="" type="radio"/> No <input type="radio"/> Yes
<b>Call Park</b>	
Call Park Timer	180
Call Park-Allowed To Park	<input type="radio"/> No <input checked="" type="radio"/> Yes
<b>Call Pickup</b>	
Allow Directed Call Pickup Of Attendant Call	<input checked="" type="radio"/> No <input type="radio"/> Yes
Call Pickup Dialed Accept	<input type="radio"/> No <input checked="" type="radio"/> Yes
Call Pickup Directed Accept	<input type="radio"/> No <input checked="" type="radio"/> Yes

- b. Click **Yes** for **Music on Hold on Transfer**.
- c. Click **Yes** for **Call Park-Allowed To Park**.

Mitel: Class of Service Options 10 (4/9)

General	Advanced
Call Pickup Display	<input checked="" type="radio"/> No <input type="radio"/> Yes
Call Privacy	
Call Privacy	<input checked="" type="radio"/> No <input type="radio"/> Yes
Calling Party Name Substitution	<input checked="" type="radio"/> No <input type="radio"/> Yes
Name Suppression on outgoing Trunk Call	<input checked="" type="radio"/> No <input type="radio"/> Yes
Privacy Released	<input checked="" type="radio"/> No <input type="radio"/> Yes
Public Network Identity Provided	<input type="radio"/> No <input checked="" type="radio"/> Yes
Call Waiting	
Call Waiting Swap	<input checked="" type="radio"/> No <input type="radio"/> Yes
ONS CLASS/CLIP: Visual Call Waiting	<input type="radio"/> No <input checked="" type="radio"/> Yes
Campon	
Auto Campon Timer	
Campon Recall Timer	10
Direct Voice Call	
Direct Voice Call - Accept	<input checked="" type="radio"/> No <input type="radio"/> Yes
Direct Voice Call - Allow	<input checked="" type="radio"/> No <input type="radio"/> Yes
Direct Voice Call - Maximize Volume	<input checked="" type="radio"/> No <input type="radio"/> Yes
Display	
After Answer Display Time	
Calling Name Display - Internal - ONS	<input type="radio"/> No <input checked="" type="radio"/> Yes

- d. Clear the value for **Auto Campon Timer**.

Mitel: Class of Service Options 10 (5/9)

General	Advanced
Calling Number Display - Internal - ONS	<input type="radio"/> No <input checked="" type="radio"/> Yes
Display ANI/DNIS/ISDN Calling/Called Number	<input checked="" type="radio"/> No <input type="radio"/> Yes
Display ANI/ISDN Calling Number Only	<input checked="" type="radio"/> No <input type="radio"/> Yes
Display Caller ID on multicall/keylines	<input checked="" type="radio"/> No <input type="radio"/> Yes
Display Caller ID On Multicall/Keylines Timer	5
Display Caller ID On Single Line Displays For Forwarded Calls	<input checked="" type="radio"/> No <input type="radio"/> Yes
Display Dialed Digits during Outgoing Calls	<input checked="" type="radio"/> No <input type="radio"/> Yes
Display DNIS/Called Number Before Digit Modification	<input checked="" type="radio"/> No <input type="radio"/> Yes
Display DNIS on Key Label	<input checked="" type="radio"/> No <input type="radio"/> Yes
Display Held Call ID on Transfer	<input checked="" type="radio"/> No <input type="radio"/> Yes
Display Transfer Destination on Recall	<input checked="" type="radio"/> No <input type="radio"/> Yes
Hot Desk External User - Display Internal Calling ID	<input checked="" type="radio"/> No <input type="radio"/> Yes
Maintain Ringing Party During Recall	<input checked="" type="radio"/> No <input type="radio"/> Yes
Non-Prime Public Network Identity	<input checked="" type="radio"/> No <input type="radio"/> Yes
Originator's Display Update In Call Forwarding/Rerouting	<input checked="" type="radio"/> No <input type="radio"/> Yes
Prefer Call Forwarding/Rerouting Information	<input checked="" type="radio"/> No <input type="radio"/> Yes
Prefer Name for Call Information	<input checked="" type="radio"/> No <input type="radio"/> Yes
Suppress Delivery of Caller ID Display between Sets	<input checked="" type="radio"/> No <input type="radio"/> Yes
Suppress Delivery of Caller ID Display between Sets - Override	<input checked="" type="radio"/> No <input type="radio"/> Yes
Suppress Display Of Account Code Numbers	<input checked="" type="radio"/> No <input type="radio"/> Yes

Mitel: Class of Service Options 10 (6/9)

General	Advanced
Suppress Redial Display	<input checked="" type="radio"/> No <input type="radio"/> Yes
<b>Fax</b>	
Campon Tone Security	<input type="radio"/> No <input checked="" type="radio"/> Yes
External Trunk Standard Ringback	<input type="radio"/> No <input checked="" type="radio"/> Yes
Fax Capable	<input checked="" type="radio"/> No <input type="radio"/> Yes
Return Disconnect Tone When Far End Party Clears	<input checked="" type="radio"/> No <input type="radio"/> Yes
<b>HCI</b>	
HCI/CTI/TAPI Call Control Allowed	<input type="radio"/> No <input checked="" type="radio"/> Yes
HCI/CTI/TAPI Monitor Allowed	<input type="radio"/> No <input checked="" type="radio"/> Yes
<b>Hot Desk</b>	
Green BLF Lamp for Logged in Hotdesk User	<input checked="" type="radio"/> No <input type="radio"/> Yes
Hot Desk External User - Allow Mid-Call Features	<input type="radio"/> No <input checked="" type="radio"/> Yes
Hot Desk External User - Answer Confirmation	<input type="radio"/> No <input checked="" type="radio"/> Yes
Hot Desk External User - Dial Tone on Call Complete	<input type="radio"/> No <input checked="" type="radio"/> Yes
Hot Desk External User - Permanent Login	<input checked="" type="radio"/> No <input type="radio"/> Yes
Hot Desk External User - Remote MWI Enable Feature Access Code	
Hot Desk External User - Remote MWI Disable Feature Access Code	
Hot Desk Login Accept	<input type="radio"/> No <input checked="" type="radio"/> Yes
Hot Desk Remote Logout Enabled	<input checked="" type="radio"/> No <input type="radio"/> Yes
<b>Miscellaneous</b>	
Backlighting - Enabled	<input type="radio"/> No <input checked="" type="radio"/> Yes

Mitel: Class of Service Options 10 (7/9)

General	Advanced
Clear All Features Remote	<input checked="" type="radio"/> No <input type="radio"/> Yes
Enbloc Dialing - Enabled	<input checked="" type="radio"/> No <input type="radio"/> Yes
Force Device Busy If Any Line In Use	<input type="radio"/> No <input checked="" type="radio"/> Yes
Handset Volume Adjustment Saved	<input checked="" type="radio"/> No <input type="radio"/> Yes
Head Set Switch Mute	<input checked="" type="radio"/> No <input type="radio"/> Yes
Long Key Press Timer	0
Multi-Color LED Support - Disable	<input checked="" type="radio"/> No <input type="radio"/> Yes
Phone Lock	<input checked="" type="radio"/> No <input type="radio"/> Yes
Reseize Timer	180
Timed Reminder Allowed	<input type="radio"/> No <input checked="" type="radio"/> Yes
User Inactivity Timer	0
<b>Paging</b>	
Group Page Accept	<input checked="" type="radio"/> No <input type="radio"/> Yes
Group Page Allow	<input checked="" type="radio"/> No <input type="radio"/> Yes
Loudspeaker Pager Equivalent Zone Override Security	<input checked="" type="radio"/> No <input type="radio"/> Yes
Loudspeaker Pager Override	<input type="radio"/> No <input checked="" type="radio"/> Yes
Pager Access All Zones	<input type="radio"/> No <input checked="" type="radio"/> Yes
Pager Access Individual Zones	<input checked="" type="radio"/> No <input type="radio"/> Yes
<b>PC Port</b>	
PC Port On IP Device - Disable	<input checked="" type="radio"/> No <input type="radio"/> Yes

Mitel: Class of Service Options 10 (8/9)

General	Advanced
<b>RAD</b>	
Answer Plus Delay To Message Timer	20
Answer Plus Expected Off-hook Timer	30
Answer Plus Message Length Timer	10
Answer Plus System Reroute Timer	0
Recorded Announcement Device	<input checked="" type="radio"/> No <input type="radio"/> Yes
Recorded Announcement Device - Advanced	<input checked="" type="radio"/> No <input type="radio"/> Yes
<b>Ringing</b>	
Delay Ring Timer	10
No Answer Recall Timer	17
Ringing Line Select	<input checked="" type="radio"/> No <input type="radio"/> Yes
Ringing Timer	180
<b>SMDR</b>	
SMDR External	<input checked="" type="radio"/> No <input type="radio"/> Yes
SMDR Internal	<input checked="" type="radio"/> No <input type="radio"/> Yes
<b>Trunk</b>	
ANI/DNIS/ISDN Number Delivery Trunk	<input checked="" type="radio"/> No <input type="radio"/> Yes
DASS II OLI/TLI Provided	<input checked="" type="radio"/> No <input type="radio"/> Yes
Public Network Access via DPNSS	<input type="radio"/> No <input checked="" type="radio"/> Yes

- e. Click Yes for Public Network Access via DPNSS.

### Mitel: Class of Service Options 10 (9/9)

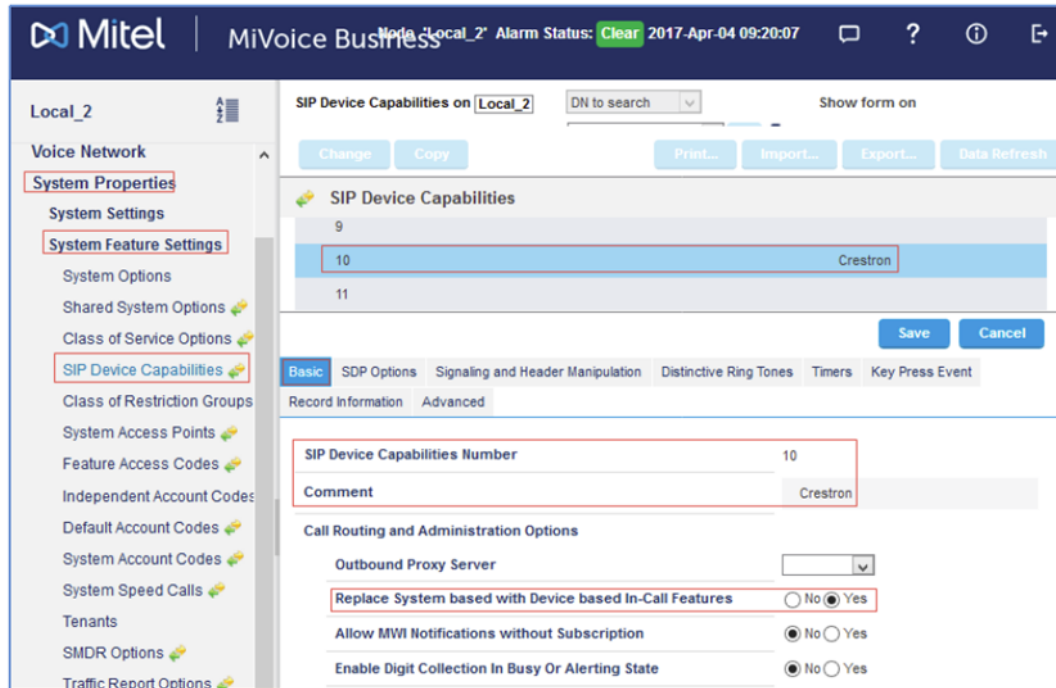
General	Advanced
Public Network To Public Network Connection Allowed	<input checked="" type="radio"/> No <input type="radio"/> Yes
Public Trunk	<input checked="" type="radio"/> No <input type="radio"/> Yes
R2 Call Progress Tone	<input checked="" type="radio"/> No <input type="radio"/> Yes
Suppress Simulated CCM after ISDN Progress	<input checked="" type="radio"/> No <input type="radio"/> Yes
Trunk Calling Party Identification	<input type="radio"/> No <input checked="" type="radio"/> Yes
Trunk Flash Allowed	<input checked="" type="radio"/> No <input type="radio"/> Yes
Two B-Channel Transfer Allowed	<input checked="" type="radio"/> No <input type="radio"/> Yes
Voice Mail	
COV/ONS/E&M Voice Mail Port	<input checked="" type="radio"/> No <input type="radio"/> Yes
ONS VMail-Delay Dial Tone Timer	5

- f. Leave all other fields at the default values.
5. Click **Save**.

## Configure SIP Device Capabilities

The **SIP Device Capabilities** window allows customization of features and options that the Mitel MiVoice System uses and accepts when communicating with Crestron devices. This example uses **SIP Device Capabilities Number 10**.

1. Click **System Properties > System Feature Settings > SIP Device Capabilities**.  
**Mitel: SIP Device Capabilities - Basic Tab**



The screenshot displays the Mitel MiVoice Business configuration interface. The left sidebar shows the navigation menu with 'System Properties' > 'System Feature Settings' > 'SIP Device Capabilities' highlighted. The main content area shows the 'SIP Device Capabilities' configuration page for 'Local\_2'. The page title is 'SIP Device Capabilities on Local\_2'. Below the title, there are buttons for 'Change', 'Copy', 'Print...', 'Import...', 'Export...', and 'Data Refresh'. A table lists SIP Device Capabilities with columns for 'SIP Device Capabilities Number' and 'Comment'. The number 10 is selected, and the comment is 'Crestron'. Below the table, there are 'Save' and 'Cancel' buttons. The 'Basic' tab is active, showing the following fields:

- SIP Device Capabilities Number:** 10
- Comment:** Crestron
- Call Routing and Administration Options:**
  - Outbound Proxy Server:** [Empty dropdown]
  - Replace System based with Device based In-Call Features:**  Yes
  - Allow MWI Notifications without Subscription:**  No
  - Enable Digit Collection In Busy Or Alerting State:**  No

2. In the **Basic** tab, do the following:
  - a. Enter **Crestron** for the **Comment** (for this example).
  - b. Click **Yes** for **Replace System based with Device based In-Call Features**.
  - c. Leave all other fields at the default values.



### Mitel: SIP Device Capabilities - SDP Options Tab

Basic	SDP Options	Signaling and Header Manipulation	Distinctive Ring Tones	Timers	Key Press Event
Record Information    Advanced					
Allow Device To Use Multiple Active M-Lines					<input checked="" type="radio"/> No <input type="radio"/> Yes
Allow Using UPDATE For Early Media Renegotiation					<input type="radio"/> No <input checked="" type="radio"/> Yes
AVP Only Device					<input type="radio"/> No <input checked="" type="radio"/> Yes
Enable Mitel Proprietary SDP					<input checked="" type="radio"/> No <input type="radio"/> Yes
Force sending SDP in initial Invite message					<input checked="" type="radio"/> No <input type="radio"/> Yes
Ignore SDP Answers in Provisional Responses					<input checked="" type="radio"/> No <input type="radio"/> Yes
Limit to one Offer/Answer per INVITE					<input checked="" type="radio"/> No <input type="radio"/> Yes
Prevent SDP Renegotiation If Peer Initiated Hold					<input checked="" type="radio"/> No <input type="radio"/> Yes
Prevent the Use of IP Address 0.0.0.0 in SDP Messages					<input type="radio"/> No <input checked="" type="radio"/> Yes
Renegotiate SDP To Enforce Symmetric Codec					<input type="radio"/> No <input checked="" type="radio"/> Yes
Repeat SDP Answer If Duplicate Offer Is Received					<input checked="" type="radio"/> No <input type="radio"/> Yes
Send Answer only after renegotiation is complete					<input checked="" type="radio"/> No <input type="radio"/> Yes
Suppress Use of SDP Inactive Media Streams					<input type="radio"/> No <input checked="" type="radio"/> Yes

3. In the **SDP Options** tab, do the following:
  - a. Click **Yes** for **Allow Using UPDATE For Early Media Renegotiation**.
  - b. Click **Yes** for **Prevent the Use of IP Address 0.0.0.0 in SDP Messages**.
  - c. Click **Yes** for **Renegotiate SDP To Enforce Symmetric Codec**.
  - d. Leave all other fields at the default values.
4. Click **Save**.

### Mitel: SIP Device Capabilities - Signaling and Header Manipulation Tab

Basic	SDP Options	Signaling and Header Manipulation	Distinctive Ring Tones	Timers	Key Press Event
Record Information    Advanced					
Allow Display Update					<input type="radio"/> No <input checked="" type="radio"/> Yes
Disable Reliable Provisional Responses					<input checked="" type="radio"/> No <input type="radio"/> Yes
Disable Use of User-Agent and Server Headers					<input checked="" type="radio"/> No <input type="radio"/> Yes
Fail REFER To Keep Call Active On Mid-Call Feature					<input checked="" type="radio"/> No <input type="radio"/> Yes
If TLS use 'sips:' Scheme					<input checked="" type="radio"/> No <input type="radio"/> Yes
Multilingual Name Display					<input checked="" type="radio"/> No <input type="radio"/> Yes
Override Auto-Answer Headers					<input checked="" type="radio"/> No <input type="radio"/> Yes
Override Auto-Answer Headers With					
Remove Anonymous User					<input checked="" type="radio"/> No <input type="radio"/> Yes
Require Reliable Provisional Responses on Outgoing Calls					<input checked="" type="radio"/> No <input type="radio"/> Yes
Suppress Redirection Headers					No <input type="button" value="v"/>
Use P-Asserted Identity Header					<input type="radio"/> No <input checked="" type="radio"/> Yes
Use user=phone					<input checked="" type="radio"/> No <input type="radio"/> Yes

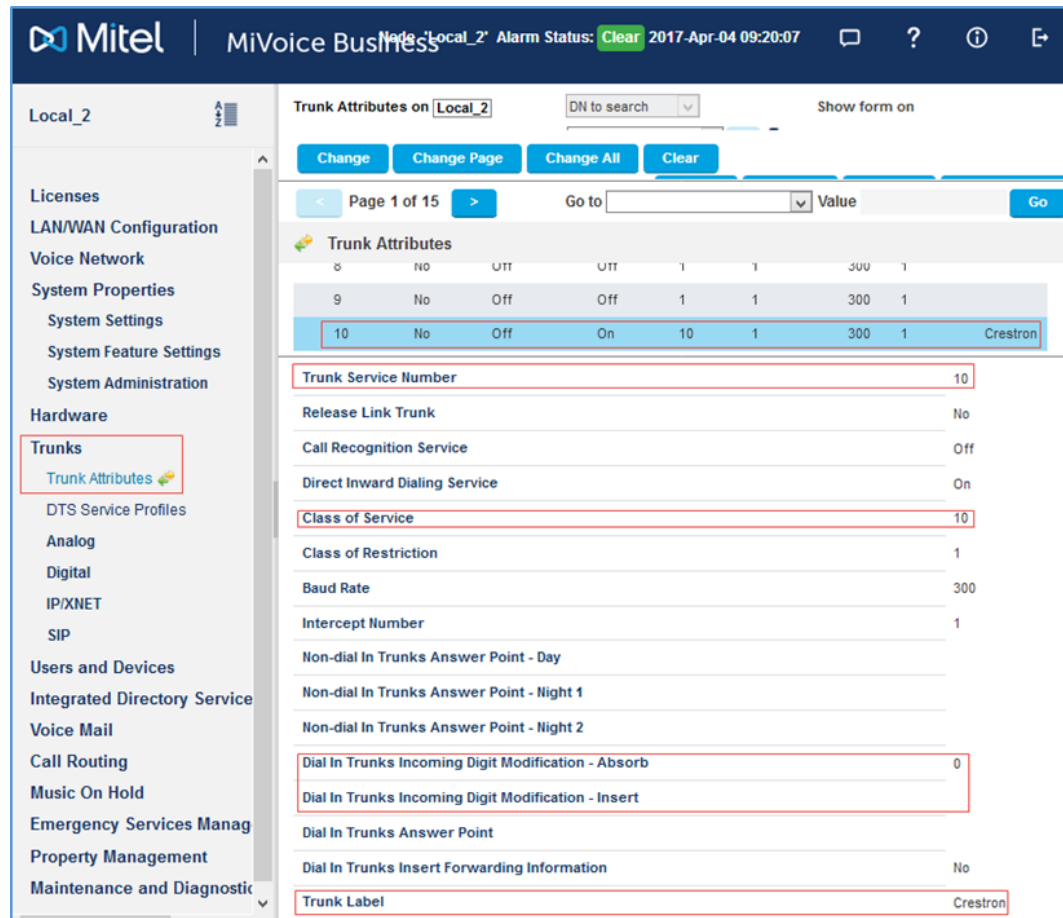
## Configure Trunk Attributes

Define attributes for the trunk used for PSTN calls. This example modified the attributes of Trunk Number 5, as shown below.

To configure trunk attributes (for this example):

1. Click **Trunks > Trunk Attributes**.
2. Select an unused **Trunk Service Number**. This example uses **10**.

**Mitel: Configure Trunk Attributes**



The screenshot displays the Mitel MiVoice Business web interface. The left sidebar shows the navigation menu with 'Trunks' selected. The main content area shows 'Trunk Attributes on Local\_2' with a table of attributes. The table has columns for Trunk Number, NO, UTT, UTT, T, T, 300, and T. The row for Trunk Number 10 is highlighted. Below the table, the configuration form for Trunk Number 10 is shown, with fields for Trunk Service Number (10), Release Link Trunk (No), Call Recognition Service (Off), Direct Inward Dialing Service (On), Class of Service (10), Class of Restriction (1), Baud Rate (300), Intercept Number (1), Non-dial In Trunks Answer Point - Day, Non-dial In Trunks Answer Point - Night 1, Non-dial In Trunks Answer Point - Night 2, Dial In Trunks Incoming Digit Modification - Absorb (0), Dial In Trunks Incoming Digit Modification - Insert, Dial In Trunks Answer Point, Dial In Trunks Insert Forwarding Information (No), and Trunk Label (Crestron).

3. Click **Change**.
4. Enter a descriptive name such as **Crestron** for the **Trunk Label**.
5. Enter **10** (configured earlier) for the **Class of Service**.
6. Enter **0** for **Dial In Trunks Incoming Digit Modification - Absorb**.

**NOTE:** Mitel absorbs none of the incoming digits on an incoming PSTN call to the desired PBX extension based on the configured translation.

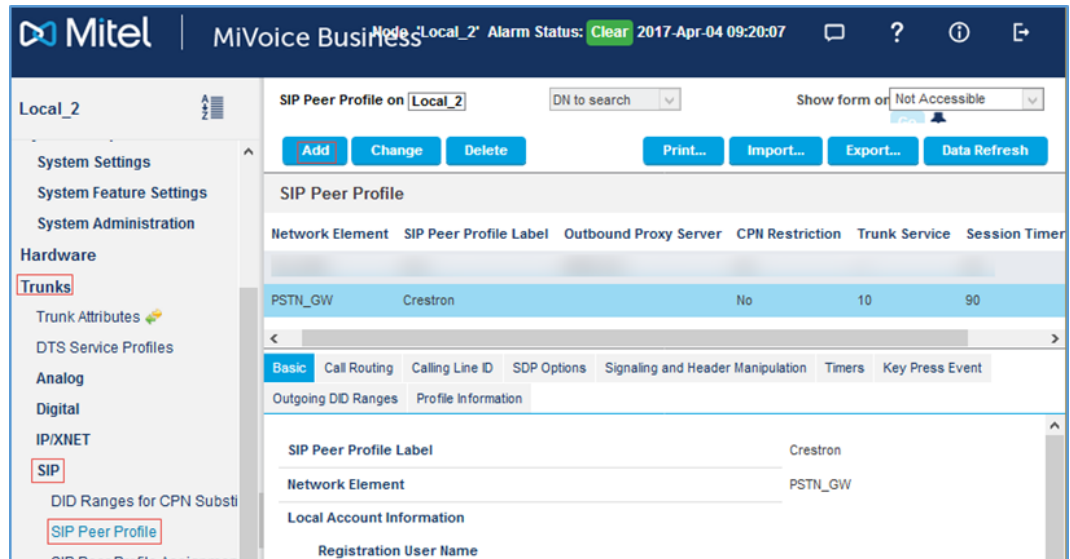
7. Leave **Dial In Trunks Incoming Digit Modification - Insert** blank.

## Configure SIP Peer Profile

To configure SIP peer profile:

1. Click **Trunks > SIP > SIP Peer Profile**.

**Mitel: Add SIP Peer Profile**



The screenshot displays the Mitel MiVoice Business configuration interface. The top navigation bar shows the Mitel logo, the product name 'MiVoice Business', and the current node 'Local\_2' with an alarm status of 'Clear' and a timestamp of '2017-Apr-04 09:20:07'. The left sidebar contains a menu with categories: System Settings, System Feature Settings, System Administration, Hardware, Trunks (highlighted), Trunk Attributes, DTS Service Profiles, Analog, Digital, IP/XNET, SIP (highlighted), and DID Ranges for CPN Substitution. The 'SIP Peer Profile' option is selected under the SIP category. The main content area is titled 'SIP Peer Profile on Local\_2' and includes a search dropdown and a 'Show form of' dropdown set to 'Not Accessible'. Below this are buttons for 'Add', 'Change', 'Delete', 'Print...', 'Import...', 'Export...', and 'Data Refresh'. The 'Add' button is highlighted with a red box. A table lists SIP Peer Profiles with columns for Network Element, SIP Peer Profile Label, Outbound Proxy Server, CPN Restriction, Trunk Service, and Session Timer. One entry is visible: PSTN\_GW, Crestron, No, 10, 90. Below the table are tabs for 'Basic', 'Call Routing', 'Calling Line ID', 'SDP Options', 'Signaling and Header Manipulation', 'Timers', and 'Key Press Event'. The 'Basic' tab is active, showing 'Outgoing DID Ranges' and 'Profile Information'. The 'Profile Information' section includes fields for 'SIP Peer Profile Label' (Crestron) and 'Network Element' (PSTN\_GW). The 'Local Account Information' section includes a field for 'Registration User Name'.

2. Click **Add**.

### Mitel: SIP Peer Profile - Basic Tab

SIP Peer Profile on Local\_2    DN to search    Show form on Not Accessible    Go

**Add**    **Change**    **Delete**    **Print...**    **Import...**    **Export...**    **Data Refresh**

**Basic**    Call Routing    Calling Line ID    SDP Options    Signaling and Header Manipulation    Timers    Key Press Event    Outgoing DID Ranges    Profile Information

**SIP Peer Profile Label**    Crestron

**Network Element**    PSTN\_GW

**Local Account Information**

Registration User Name

**Address Type**    IP Address: 10.35.32.2

**Administration Options**

Interconnect Restriction	1
Maximum Simultaneous Calls	100
Minimum Reserved Call Licenses	0
Outbound Proxy Server	
SMDR Tag	0
<b>Trunk Service</b>	<b>10</b>
Zone	1

**Authentication Options**

User Name	
Password	*****
Confirm Password	*****
Authentication Option for Incoming Calls	No Authentication
Subscription User Name	
Subscription Password	*****
Subscription Confirm Password	*****

3. In the **Basic** tab, do the following:
  - a. Enter a descriptive name for the **SIP Peer Profile Label**. This example uses **Crestron**.
  - b. Select the **Network Element** from the drop-down menu. This example uses **PSTN\_GW**.
  - c. In the **Local Account Information** section, select **IP Address** for the **Address Type** and enter **10.35.32.2** (for this example).
  - d. Select **10** (configured earlier as the **Trunk Group**) for the **Trunk Service**.
  - e. Leave all other fields at the default values.

#### Mitel: SIP Peer Profile - Call Routing Tab

SIP Peer Profile on Local\_2      DN to search      Show form on Not Accessible      Go

**Add**   **Change**   **Delete**      **Print...**   **Import...**   **Export...**   **Data Refresh**

Basic   **Call Routing**   Calling Line ID   SDP Options   Signaling and Header Manipulation   Timers   Key Press Event   Outgoing DID Ranges   Profile Information

Alternate Destination Domain Enabled	No
Alternate Destination Domain FQDN or IP Address	
Enable Special Re-invite Collision Handling	No
Only Allow Outgoing Calls	No
Private SIP Trunk	No
Reject Incoming Anonymous Calls	No
Route Call Using P-Called-Party-ID (if present)	Yes
Route Call Using To Header	No

4. In the **Call Routing** tab, leave all fields at the default values.

#### Mitel: SIP Peer Profile - Calling Line ID Tab

SIP Peer Profile on Local\_2      DN to search      Show form on Not Accessible      Go

**Add**   **Change**   **Delete**      **Print...**   **Import...**   **Export...**   **Data Refresh**

Basic   Call Routing   **Calling Line ID**   SDP Options   Signaling and Header Manipulation   Timers   Key Press Event   Outgoing DID Ranges   Profile Information

Default CPN	
Default CPN Name	
CPN Restriction	No
Public Calling Party Number Passthrough	No
Strip PNI	No
Use Diverting Party Number as Calling Party Number	No
Use Original Calling Party Number If Available	No

5. In the **Calling Line ID** tab, leave all fields at the default values.

## Mitel: SIP Peer Profile - SDP Options Tab

SIP Peer Profile on Local\_2 DN to search Show form on Not Accessible Go

Add Change Delete Print... Import... Export... Data Refresh

Save Cancel

Basic Call Routing Calling Line ID **SDP Options** Signaling and Header Manipulation Timers Key Press Event Profile Information

Allow Peer To Use Multiple Active M-Lines	<input type="radio"/> No <input checked="" type="radio"/> Yes
Allow Using UPDATE For Early Media Renegotiation	<input type="radio"/> No <input checked="" type="radio"/> Yes
Avoid Signaling Hold to the Peer	<input type="radio"/> No <input checked="" type="radio"/> Yes
AVP Only Peer	<input type="radio"/> No <input checked="" type="radio"/> Yes
Enable Mitel Proprietary SDP	<input checked="" type="radio"/> No <input type="radio"/> Yes
Force sending SDP in initial Invite message	<input checked="" type="radio"/> No <input type="radio"/> Yes
Force sending SDP in initial Invite - Early Answer	<input checked="" type="radio"/> No <input type="radio"/> Yes
Ignore SDP Answers in Provisional Responses	<input checked="" type="radio"/> No <input type="radio"/> Yes
Limit to one Offer/Answer per INVITE	<input type="radio"/> No <input checked="" type="radio"/> Yes
NAT Keepalive	<input type="radio"/> No <input checked="" type="radio"/> Yes
Prevent the Use of IP Address 0.0.0.0 in SDP Messages	<input type="radio"/> No <input checked="" type="radio"/> Yes
Renegotiate SDP To Enforce Symmetric Codec	<input checked="" type="radio"/> No <input type="radio"/> Yes
Repeat SDP Answer If Duplicate Offer Is Received	<input checked="" type="radio"/> No <input type="radio"/> Yes
Restrict Audio Codec	No Restriction <input type="text"/>
RTP Packetization Rate Override	<input checked="" type="radio"/> No <input type="radio"/> Yes
RTP Packetization Rate	20ms <input type="text"/>
Special handling of Offers in 2XX responses (INVITE)	<input checked="" type="radio"/> No <input type="radio"/> Yes
Suppress Use of SDP Inactive Media Streams	<input checked="" type="radio"/> No <input type="radio"/> Yes

6. In the **SDP Options** tab, click **Yes** for **Allow Using UPDATE For Early Media Renegotiation**.

## Mitel: SIP Peer Profile - Signaling and Header Manipulation Tab

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Outgoing DID Ranges	Profile Information
Trunk Group Label								
Allow Display Update								Yes
Build Contact Using Request URI Address								No
De-register Using Contact Address not *								Yes
Disable Reliable Provisional Responses								No
Disable Use of User-Agent and Server Headers								No
Domain for Trunk Context								
E.164: Enable sending '+'								No
E.164: Add '+' if digit length > N digits								0
E.164: Do not add '+' to Emergency Called Party								No
E.164: Do not add '+' to Called Party								No
Force Max-Forward: 70 on Outgoing Calls								No
If TLS use 'sips:' Scheme								No
Ignore Incoming Loose Routing Indication								No
Include Diversion Header for EHDU								No
Multilingual Name Display								No
Only use SDP to decide 180 or 183								Yes
Prefer From Header for Caller ID								No
Require Reliable Provisional Responses on Outgoing Calls								No
Signal Privacy (if enabled) on Emergency Calls								No
Suppress Redirection Headers								No
Use Fixed Retry Time for 491								No
Use Privacy: none								No
Use P-Asserted Identity Header								Yes
Use P-Asserted Identity for Billing								No
Use P-Call-Leg-ID Header								No
Use P-Early-Media Header								No
Use P-Preferred Identity Header								No
Use Restricted Character Set For Authentication								No
Use To Address in From Header on Outgoing Calls								No
Use user=phone								No
Use user=phone for Diversion Header								No

7. In the **Signaling and Header Manipulation** tab, do the following:
  - a. Select **Yes** for **Allow Display Update**.
  - b. Select **No** for **Require Reliable Provisional Responses on Outgoing Calls**.
8. Leave all other tabs at the default values.

## Assign SIP Peer Profile by Incoming DID

Use the **SIP Peer Profile Assignment by Incoming DID** window to assign incoming digits from the PSTN to the Mitel.

To assign a profile (for this example):

1. Click **Trunks > SIP > SIP Peer Profile by Incoming DID**.  
**Mitel: SIP Peer Profile Assignment by Incoming DID**

Incoming DID Range	SIP Peer Profile Label	Comment
9722657277-9722657279	Crestron	PSTN-Crestron

2. Click **Add**.
3. Enter **9722657277-9722657279** for the **Incoming DID Range**.
4. Enter **PSTN-Crestron** as an optional **Comment**.

## Automatic Route Selection (ARS) Digit Modification Number

Configure digit modification for outgoing calls on the SIP trunk to PSTN to absorb or inject additional digits according to the chosen dialing plan. This example absorbs one digit.

1. Click **Call Routing > Automatic Route Selection (ARS) > ARS Digit Modification Plans**.  
**Mitel: ARS Digit Modification Numbers**

Digit Modification Number	Number of Digits to Absorb	Digits to be Inserted	Final Tone Plan Information Marker
1	1		
2	0		
3	0		

2. Change **Digit Modification Number 1**, by selecting **1** for the **Number of Digits to Absorb** while dialing out to PSTN.
3. Click **Save**.



## ARS Routes

Configure a route for SIP trunk Connectivity to PSTN (for this example):

1. Click **Call Routing > Automatic Route Selection (ARS) > ARS Routes**.  
Mitel: ARS Routes

The screenshot displays the Mitel MiVoice Business configuration interface. On the left, a navigation sidebar is visible with the following categories: Users and Devices, Integrated Directory Services, Voice Mail, Call Routing (highlighted), Automatic Route Selection (ARS) (highlighted), ARS Call Progress Tone Detection, ARS Digit Modification Plans, ARS Maximum Dialed Digits, ARS Routes (highlighted), ARS Route Lists, ARS Route Plans, ARS Digits Dialed, ARS Leading Digits, ARS Day and Time Zones, ARS Node Identities, Call Handling, Music On Hold, Emergency Services Management, and Property Management. The main content area is titled 'ARS Routes on Local\_2' and features a 'Change' form. The form includes the following fields: Route Number (10), Routing Medium (SIP Trunk), Trunk Group Number (empty), SIP Peer Profile (Crestron), PBX Number / Cluster Element ID (empty), COR Group Number (1), Digit Modification Number (1), Digits Before Outpulsing (empty), Route Type (PSTN Access Via DPNSS), and Compression (Off). The 'Save' and 'Cancel' buttons are located at the bottom right of the form.

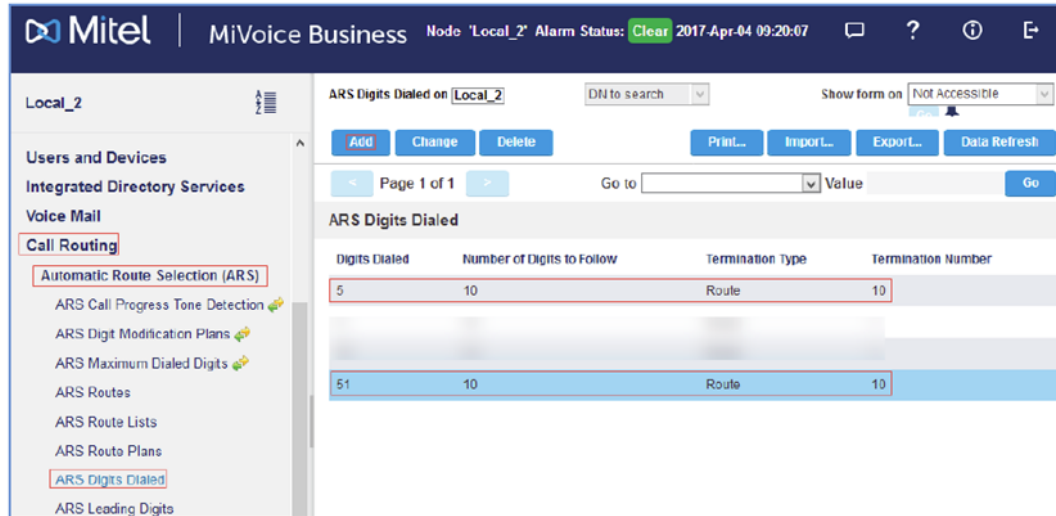
2. Select an unused **Route Number**. This example uses **10**.
3. Click **Change**.
4. Select **SIP Trunk** for the **Routing Medium**.
5. Select **Crestron** for the **SIP Peer Profile**.
6. Enter **1** for the **Digit Modification Number**.
7. Select **PSTN Access Via DPNSS** for the **Route Type**.
8. Click **Save**.

## ARS Digits Dialed

ARS initiates the routing of trunk calls when a station dials certain digits. This example uses the prefix 5 to route calls to PSTN using Route 10.

To configure ARS digits dialed:

1. Click **Call Routing > Automatic Route Select (ARS) > ARS Digits Dialed**.  
Mitel: ARS Digits Dialed



The screenshot shows the Mitel MiVoice Business configuration interface. The left sidebar contains a navigation menu with the following items: Users and Devices, Integrated Directory Services, Voice Mail, Call Routing (highlighted), Automatic Route Selection (ARS) (highlighted), ARS Call Progress Tone Detection, ARS Digit Modification Plans, ARS Maximum Dialed Digits, ARS Routes, ARS Route Lists, ARS Route Plans, ARS Digits Dialed (highlighted), and ARS Leading Digits. The main content area is titled 'ARS Digits Dialed on Local\_2'. It includes a search bar for 'DN to search', a 'Show form on' dropdown set to 'Not Accessible', and buttons for 'Add', 'Change', 'Delete', 'Print...', 'Import...', 'Export...', and 'Data Refresh'. Below these are pagination controls showing 'Page 1 of 1' and a 'Go to' field. The main table displays the following data:

Digits Dialed	Number of Digits to Follow	Termination Type	Termination Number
5	10	Route	10
51	10	Route	10

2. Click **Add**.
3. Enter **1** for the number of records to add.
4. Enter **5** for the **Digits Dialed** (for this example).
5. Enter **10** for the **Number of Digits to Follow**.
6. Enter **Route** for the **Termination Type**.
7. Enter **10** for the **Termination Number** (for this example).

Similarly, add another entry for starting digits **51**.

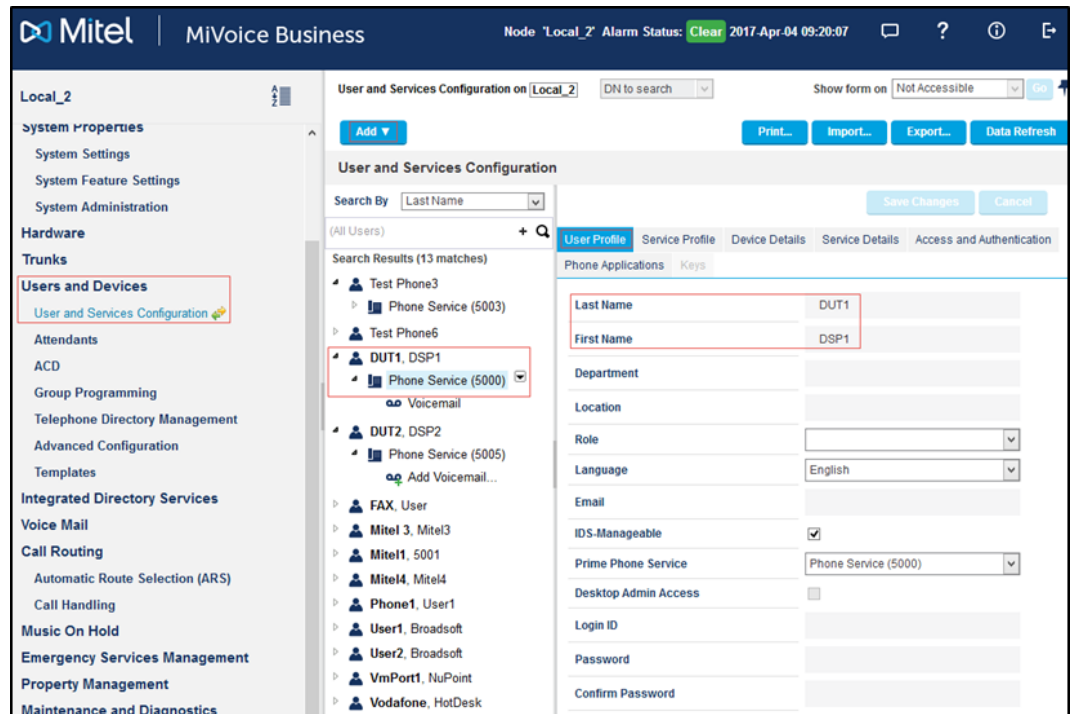
## Configure a User for Each Device/Phone

Configure the Crestron Avia DSP device as a generic SIP phone that registers to the Mitel PBX. Configure a user for each phone and Crestron device used in this example.

To configure a user for each device/phone (for this example):

1. Click **Users and Devices > User and Services Configuration**.
2. Click **Add > Default Users and Device**.

Mitel: Add User - User Profile



The screenshot displays the Mitel MiVoice Business configuration interface. The main window is titled 'User and Services Configuration on Local\_2'. The left sidebar shows a navigation menu with 'Users and Devices' highlighted. The main content area shows a search for 'DUT1, DSP1' with 13 matches. The 'User Profile' tab is selected, showing the following fields:

Last Name	DUT1
First Name	DSP1
Department	
Location	
Role	
Language	English
Email	
IDS-Manageable	<input checked="" type="checkbox"/>
Prime Phone Service	Phone Service (5000)
Desktop Admin Access	<input type="checkbox"/>
Login ID	
Password	
Confirm Password	

3. In the **User Profile** tab, do the following:
  - a. Enter **DUT1** for the **Last Name**.
  - b. Enter **DSP1** for the **First Name**.

#### Mitel: Add User - Service Profile

User Profile	<b>Service Profile</b>	Device Details	Service Details	Access and Authentication
Phone Applications	Keys			
Number	5000			
Service Label	Phone Service			
Directory Name	DUT1,DSP1			
Prime Name	<input checked="" type="radio"/> No <input type="radio"/> Yes			
Privacy	<input checked="" type="radio"/> No <input type="radio"/> Yes			
Hot Desking User	<input checked="" type="radio"/> No <input type="radio"/> Yes			
Device Type	Generic SIP Phone			
Service Level	Full			
Home Element	Local_2			
Secondary Element	Not Assigned			
Local-only DN	<input type="checkbox"/>			
ACD Enabled	<input type="checkbox"/>			
Single Line Phone	<input type="checkbox"/>			

4. In the **Service Profile** tab, do the following:
  - a. Enter **5000** (available DN) for the **Number**.
  - b. Select **Generic SIP Phone** for the **Device Type**.

#### Mitel: Add User - Device Details

User Profile	Service Profile	<b>Device Details</b>	Service Details	Access and Authentication
Phone Applications	Keys			
PKM	None			
MAC Address				
	Cabinet	Shelf	Slot	Circuit
PLID				
	<div style="border: 1px solid red; padding: 5px;">CESID digit length varies by country. Entering an incorrect number of digits could impair the ability of emergency services to respond. Consult the local public safety authority for CESID requirements in your area before changing.</div>			
CESID				

Mitel: Add User - Service Details

	Day	Night 1	Night 2
<b>Class of Service</b>	10	10	10
<b>Class of Restriction</b>	1	1	1
<b>External Hot Desking Enabled</b>	<input checked="" type="radio"/> No <input type="radio"/> Yes		
<b>External Hot Desking Dialing Prefix</b>			
<b>External Hot Desking Number</b>			
<b>DID Service Number</b>	9722657278		
<b>Use DID Number for Outgoing Calls</b>	<input checked="" type="checkbox"/>		
<b>CPN Substitution Number</b>	9722657278		
<b>Billing Number</b>			
<b>Personal Speedcall Allocation</b>			
<b>Zone Assignment Method</b>	Default		
<b>Zone ID</b>	1		
<b>SIP Device Capabilities</b>	10		
<b>Interconnect Number</b>	1		
<b>Tenant Number</b>	1		
<b>Lock Default Configuration</b>	<input checked="" type="radio"/> No <input type="radio"/> Yes		
<b>Max Call History Records</b>	0		
<b>Non-Busy Extension</b>	<input checked="" type="radio"/> No <input type="radio"/> Yes		
<b>Call Coverage Service Number</b>	1		
<b>Call Rerouting - Day</b>	1		
<b>Call Rerouting - Night1</b>	1		
<b>Call Rerouting - Night2</b>	1		
<b>Call Rerouting DND Type</b>	All		
<b>Call Rerouting - 1st Alt.</b>	1		
<b>Call Rerouting - 2nd Alt.</b>	1		

5. In the **Service Details** tab, do the following:
  - a. Enter **10** for the **Class of Service**.
  - b. Enter **9722657278** for the **DID Service Number**.
  - c. Check **Use DID Number for Outgoing Calls**.
  - d. Enter **10** for the **SIP Device Capabilities**.

**Mitel: Add User - Access and Authentication**

User Profile	Service Profile	Device Details	Service Details	Access and Authentication
User PIN		<input type="password"/>		
Confirm User PIN		<input type="password"/>		
SIP Password		<input type="password"/>		
Confirm SIP Password		<input type="password"/>		
Wireless PIN		<input type="password"/>		
Confirm Wireless PIN		<input type="password"/>		

6. In the **Access and Authentication** tab, do the following:
  - a. Assign the **SIP Password**. This example uses **123156**.
  - b. Confirm the password by entering the same password used in the previous step.

**Mitel: Add User - Phone Applications**

User Profile	Service Profile	Device Details	Service Details	Access and Authentication	Phone Applications
Branding Application		<input type="text"/>			
Screen Saver Application		<input type="text"/>			
HTML Infrastructure Enabled		<input checked="" type="radio"/> No <input type="radio"/> Yes			
HTML GUI Application		<input type="text"/>			
New Page Application1		<input type="text"/>			
New Page Application2		<input type="text"/>			
New Page Application3		<input type="text"/>			
Notification Application1		<input type="text"/>			
Notification Application2		<input type="text"/>			
Notification Application3		<input type="text"/>			

7. Leave all other fields on all tabs at the default values.

## Mitel: Add User - Keys

User Profile							Service Profile	Device Details	Service Details	Access and Authentication	Phone Applications	Keys
							Copy Keys	Clear All Keys	Clear Key			
Button Number	Label	Line Type	URL	Button Directory Number	Ring Type	MiXML Application Feature						
> 1		Single Line		5000	Ring	Not Assigned						
> 2		Not Assigned				Not Assigned						
> 3		Not Assigned				Not Assigned						
> 4		Not Assigned				Not Assigned						
> 5		Not Assigned				Not Assigned						

This example configures another user with DN 5005.

## Configure a Call Forwarding Profile

Configure call forwarding on the device via the Call Forwarding Profile. The example below describes the procedure to configure a **Call Forward Type** of **Always** from DN 5000 to DN 5005.

1. Click **Users and Devices > Advanced Configuration > Call Forwarding Profile**.
2. Click **Add**.

**Mitel: Call Forwarding Profile**

Field Name	Value to Add	Increment by
Number	5000	
Call Forward Type	Always	-
Forwarding Destination	5005	
Forwarding Enabled	<input type="radio"/> Off <input checked="" type="radio"/> On	-

3. Enter **1** for **Enter the number of records to add**.
4. In the **Define the Add Range Programming Pattern** section, do the following:
  - a. Enter **5000** for the **Number**.
  - b. Select **Always** for the **Call Forwarding Type** (for this example). Other options include Busy Internal/External and No Answer Internal/External.
  - c. Enter **5005** for the **Forwarding Destination**.
  - d. Click **On** for **Forwarding Enabled**.
5. Click **Save**.



## Configure the Ring Group

To configure the ring group:

1. Click **Users and Devices > Group Programming > Ring Groups**.
2. Click **Add** (not shown).
3. Enter **1** for **Enter the number of records to change**.
4. In the **Define the Add Range Programming Pattern** section, do the following (for this example):
  - a. Enter **5010** for the **Ring Group**.
  - b. Select **Ring All** for the **Ring Group Mode**.
  - c. Enter **10** for the **Class of Service - Day**, **Class of Service - Night1**, and **Class of Service - Night2**.
  - d. Enter **1** for the **Zone ID**.
  - e. Leave all other fields at the default values.
5. Click **Save**.

### Mitel: Add Members to Ring Group

The screenshot displays the configuration interface for adding members to a ring group. The top section, 'Ring Groups', shows a table with one entry: Ring Group 5010, Ring Group Mode Ring All, Ring Group Name (empty), Ring Group Type (empty), Home Element Local\_2, and Secondary Element Not Assigned. Below this is a form with fields for Ring Group (5010), Local-only DN (False), and Ring Group Mode (Ring All). The bottom section, 'Ring Group Members', shows a table with three members: Member Index 1 (Number 5000, Presence Present, Name DUT1,DSP1, Home Element Local\_2), Member Index 2 (Number 5002, Presence Present, Name Mitel 3,Mitel3, Home Element Local\_2), and Member Index 3 (Number 5005, Presence Present, Name DUT2,DSP2, Home Element Local\_2). Buttons for 'Add Member', 'Change Member', and 'Delete Member' are visible.

Ring Group	Ring Group Mode	Ring Group Name	Ring Group Type	Home Element	Secondary Element
5010	Ring All			Local_2	Not Assigned

Member Index	Number	Presence	Name	Home Element	Secondary Element
1	5000	Present	DUT1,DSP1	Local_2	
2	5002	Present	Mitel 3,Mitel3	Local_2	
3	5005	Present	DUT2,DSP2	Local_2	

6. Click **Add Member**.
7. Enter **1** for number of records to add.
8. In the **Ring Group Members** section, do the following:
  - a. Enter **5000** for the **Number**.
  - b. Enter **Present** for **Presence**.

9. Click **Save**.
10. Similarly, add other members. This example added 5000, 5002, and 5005.

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