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

Configuration Guide

Crestron Electronics, Inc.

Contents

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.

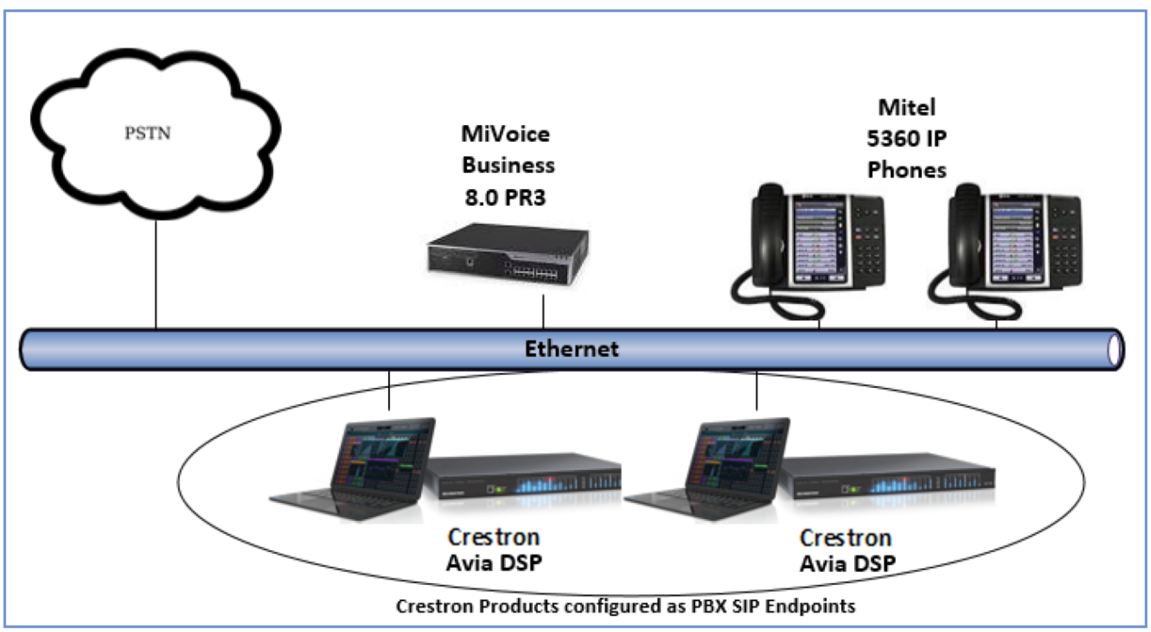
Audience

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.

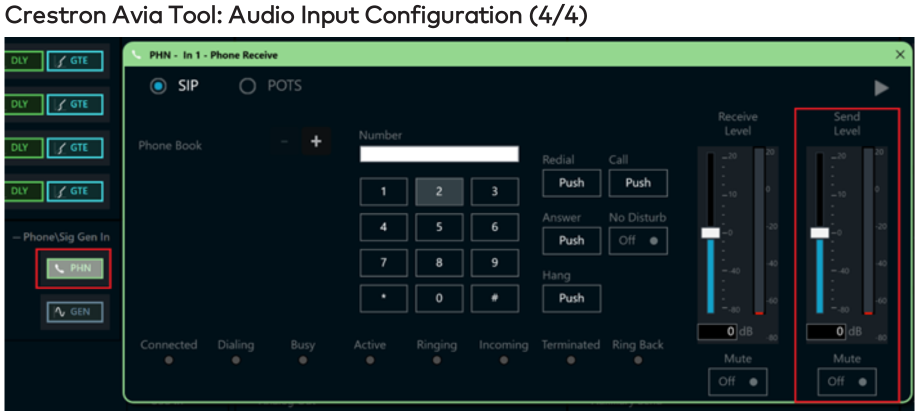
Input Configuration

To configure the analog input:

1. Click **Signal**.
   Crestron Avia tool: Audio Input Configuration (1/4)
   

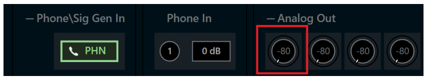
2. Under **Analog In 1** (first row), double click **Gain**. In the new window set the following:
   a. Click **Mute** to **Off**.
   b. Select **33** for the **Analog Gain**.
   c. If a condenser microphone is being used, click **+48V** (phantom power) to **On**.
   Crestron Avia Tool: Audio Input Configuration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

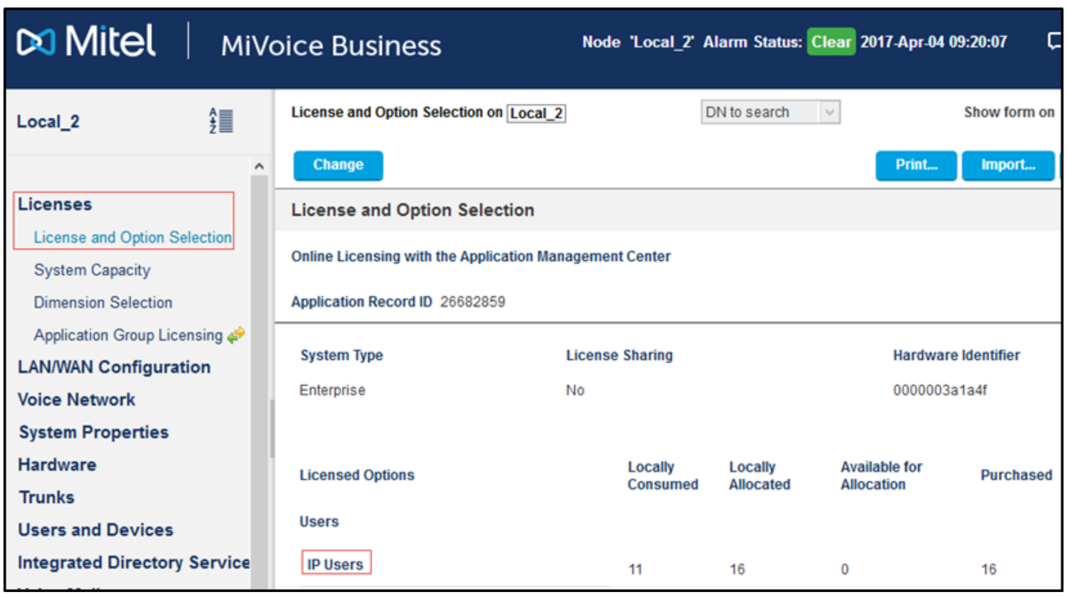
NOTE: Confirm that the general installation and basic Mitel configuration have been administered.

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

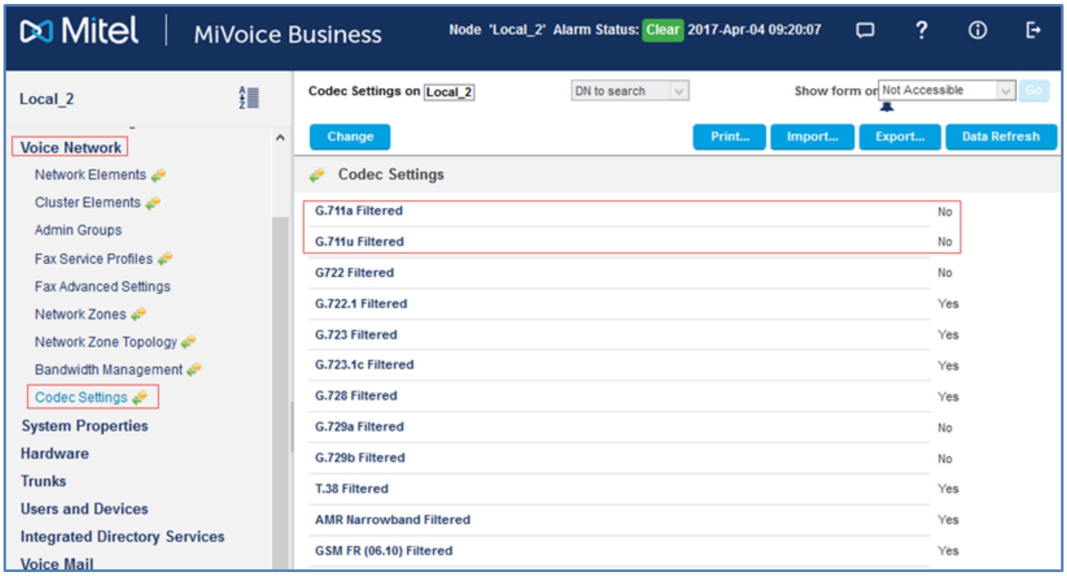


Configure Codec Settings and Network Zones

Configure codec settings to allow G711u and G711a codec negotiation.

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

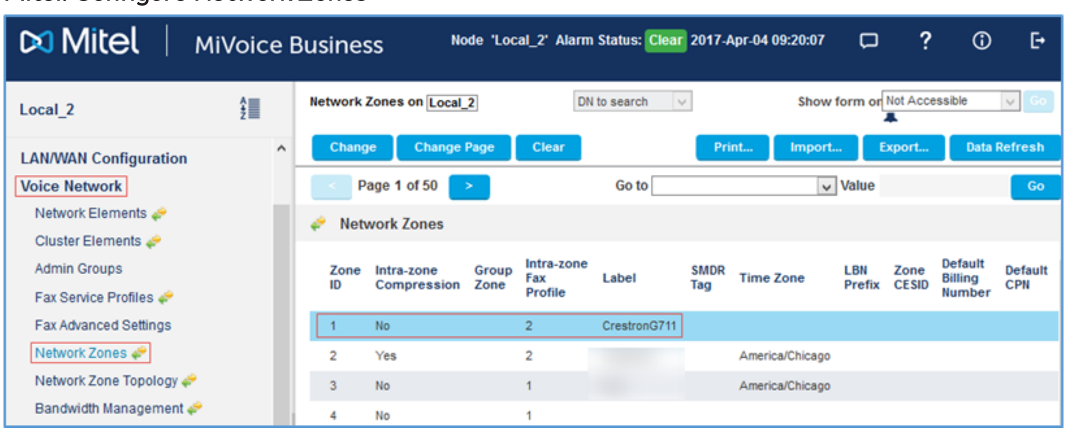
Mitel: Codec Settings



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



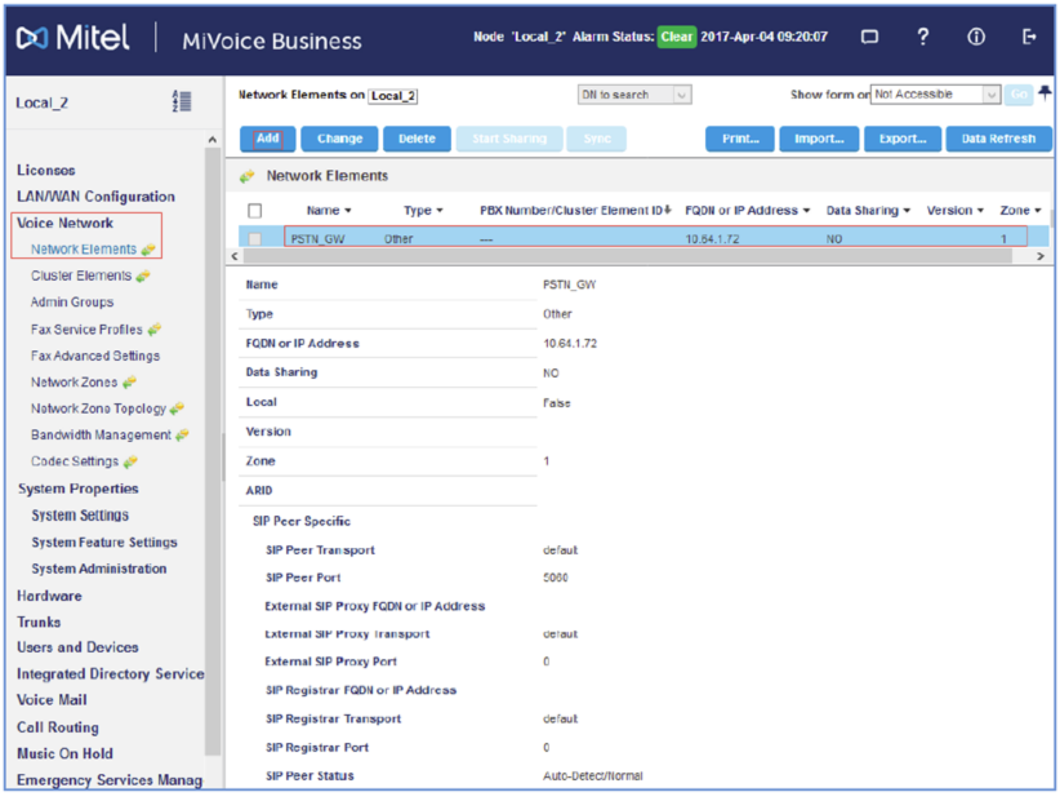
2. Select a **Zone ID** to modify. This example uses **1**.
3. Select **No** for the **Intra-zone Compression**.

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



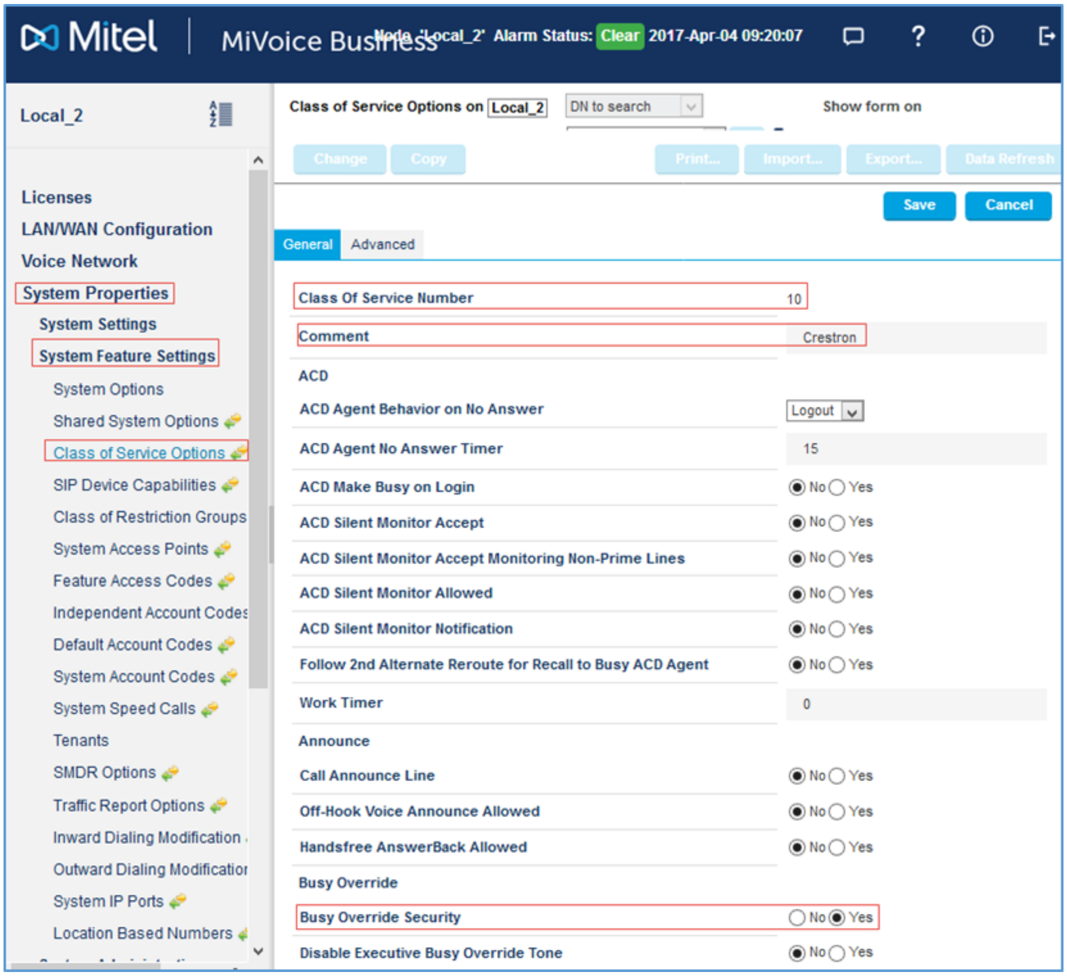
3. Enter **PSTN_GW** for the **Name** (for this example).
4. Enter **Other** for the **Type**.
5. Enter **10.64.1.72** for the **FQDN or IP Address** (for this example, which is the IP address of the PSTN gateway).
6. Enter **1** for the **Zone**. This setting ensures a G711u and G711a codec.
7. In the **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:

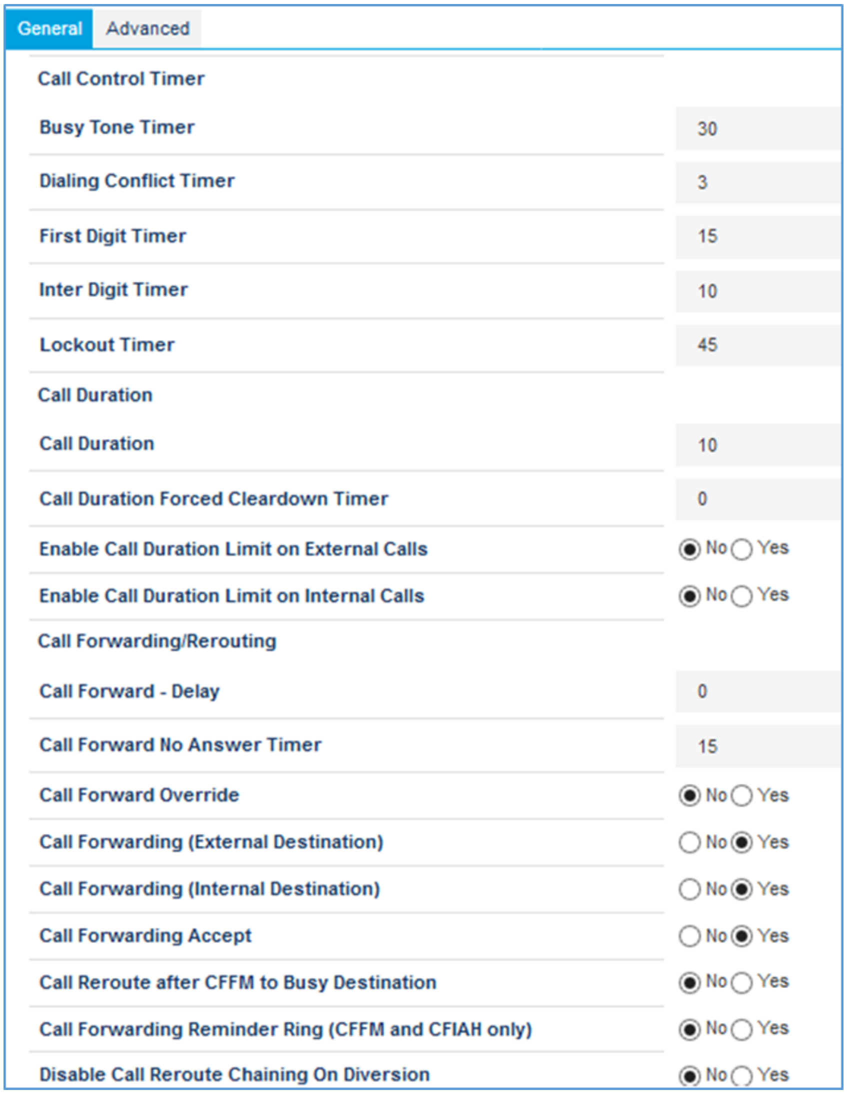
1. Click **System Properties** > **System Feature Settings** > **Class of Service Options**.
2. Select **10** for the **Class Of Service Number** (for this example).
3. Click **Change**.

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

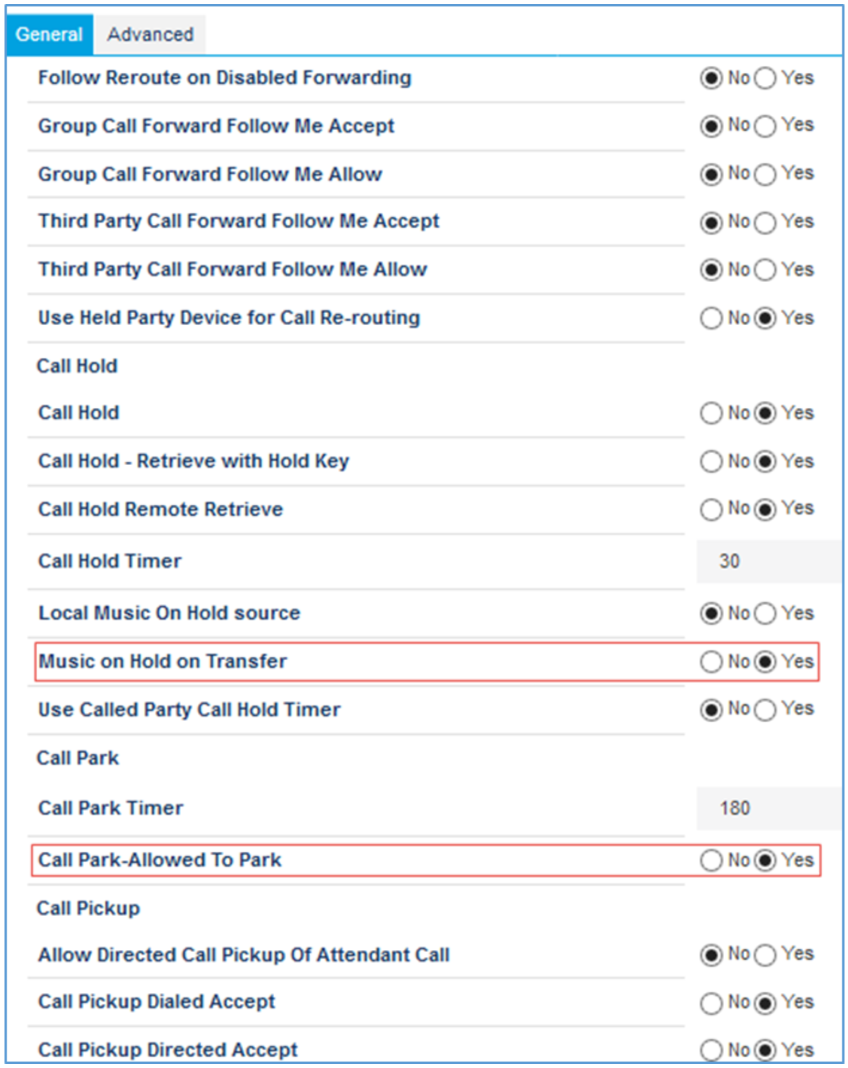
   

4. In the **General** tab, do the following:
   a. Click **Yes** for **Busy Override Security**.

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



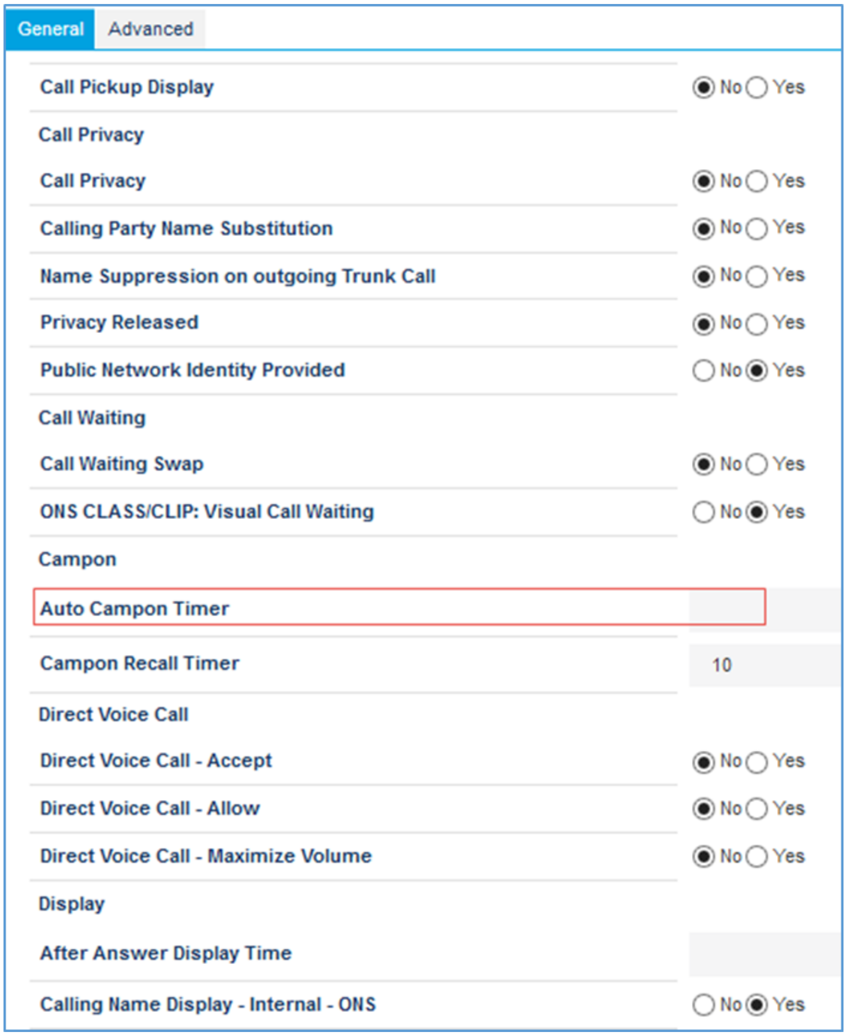
Mitel: Class of Service Options 10 (3/9)



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)



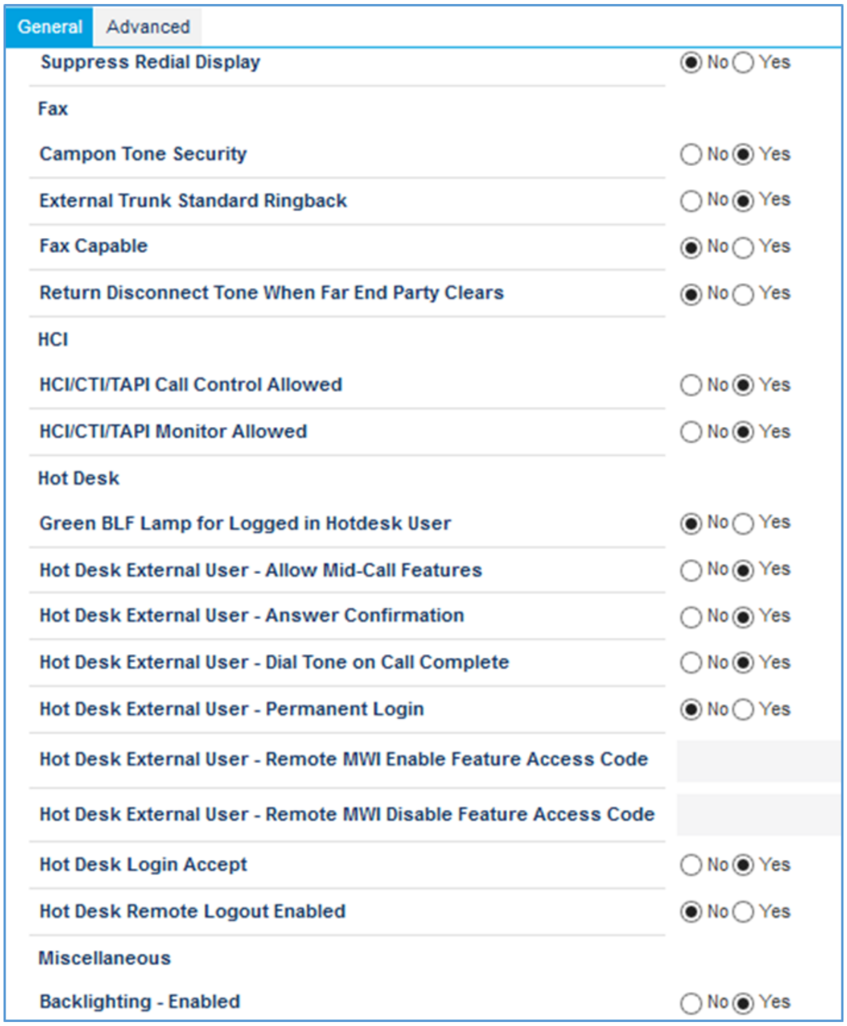
d. Clear the value for **Auto Campon Timer**.

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

General | Advanced

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

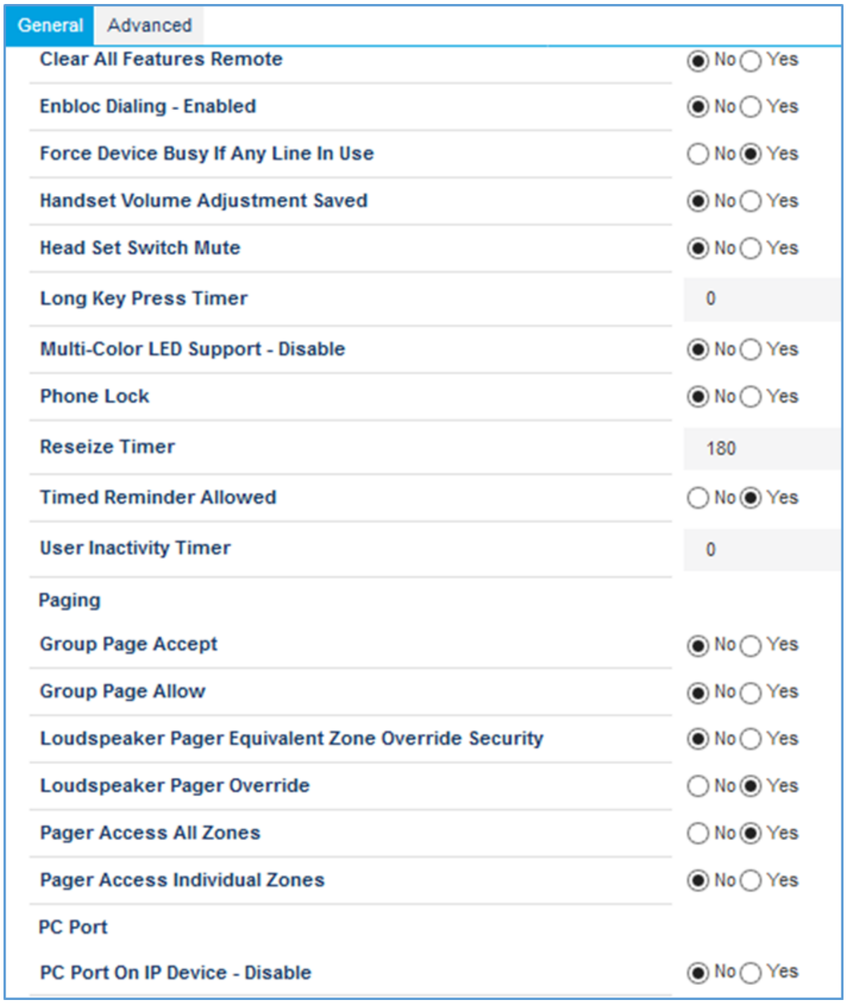
Mitel: Class of Service Options 10 (6/9)



General | Advanced

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

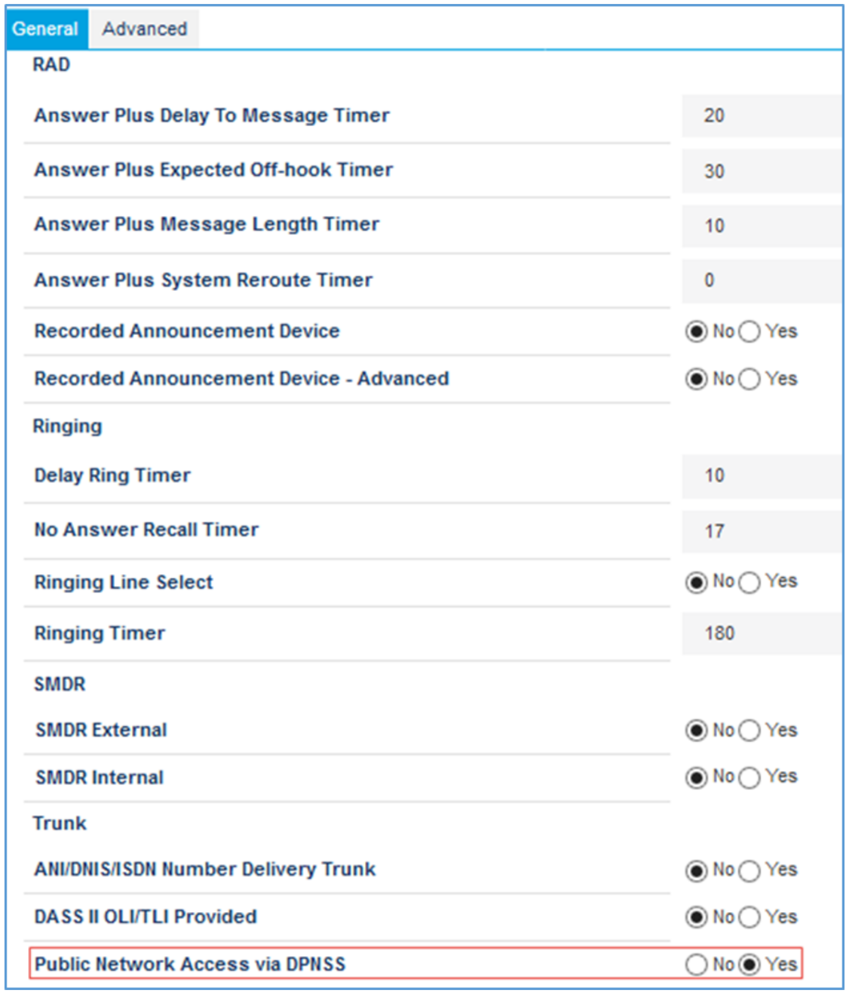
Mitel: Class of Service Options 10 (7/9)



General | Advanced

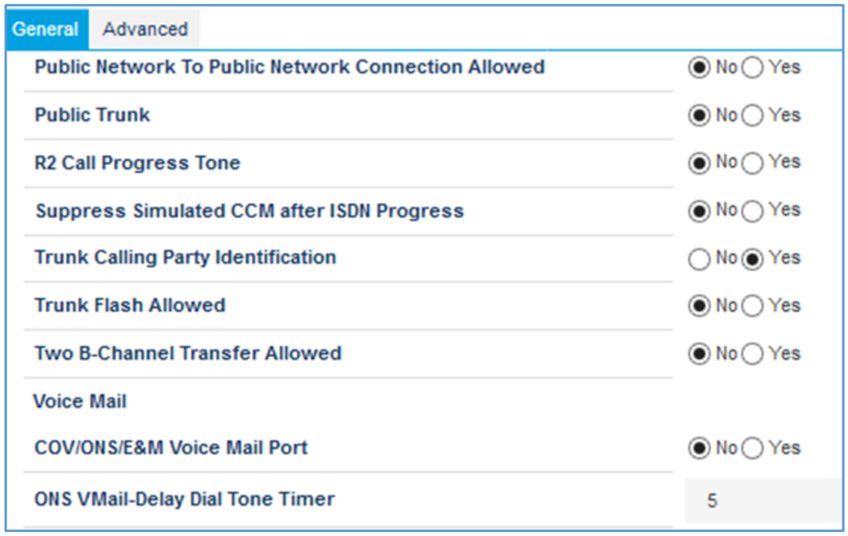
Option	Value
Clear All Features Remote	(•) No ( ) Yes
Enbloc Dialing - Enabled	(•) No ( ) Yes
Force Device Busy If Any Line In Use	( ) No (•) Yes
Handset Volume Adjustment Saved	(•) No ( ) Yes
Head Set Switch Mute	(•) No ( ) Yes
Long Key Press Timer	0
Multi-Color LED Support - Disable	(•) No ( ) Yes
Phone Lock	(•) No ( ) Yes
Reseize Timer	180
Timed Reminder Allowed	( ) No (•) Yes
User Inactivity Timer	0
Paging	
Group Page Accept	(•) No ( ) Yes
Group Page Allow	(•) No ( ) Yes
Loudspeaker Pager Equivalent Zone Override Security	(•) No ( ) Yes
Loudspeaker Pager Override	( ) No (•) Yes
Pager Access All Zones	( ) No (•) Yes
Pager Access Individual Zones	(•) No ( ) Yes
PC Port	
PC Port On IP Device - Disable	(•) No ( ) Yes

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



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

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



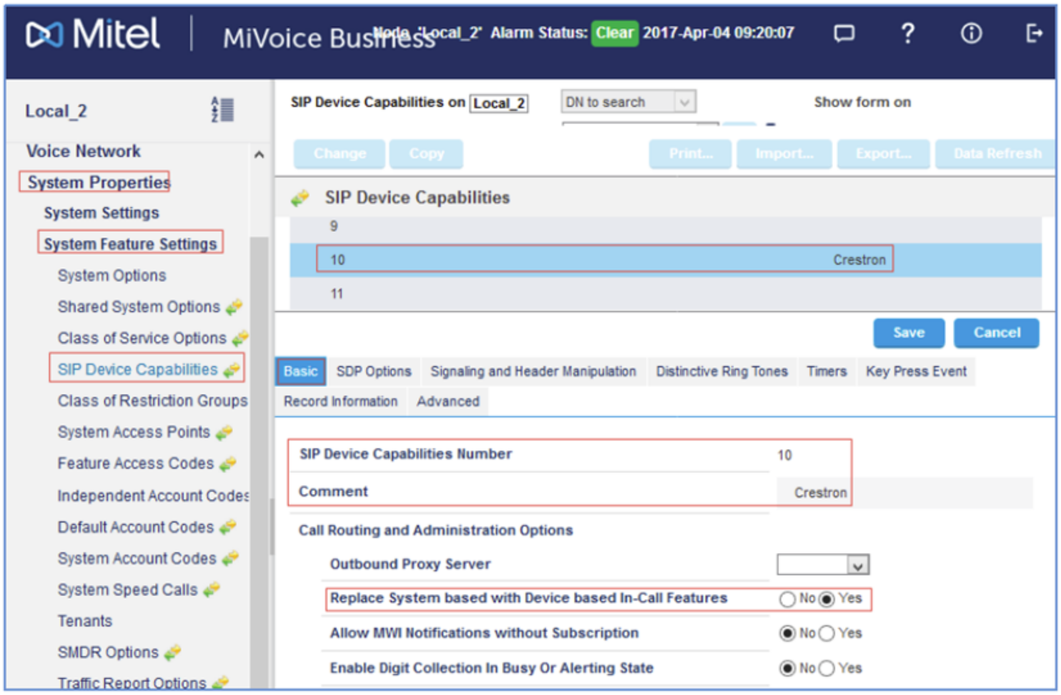
f. Leave all other fields at the default values.

5. Click **Save**.

Configure SIP Device Capabilities

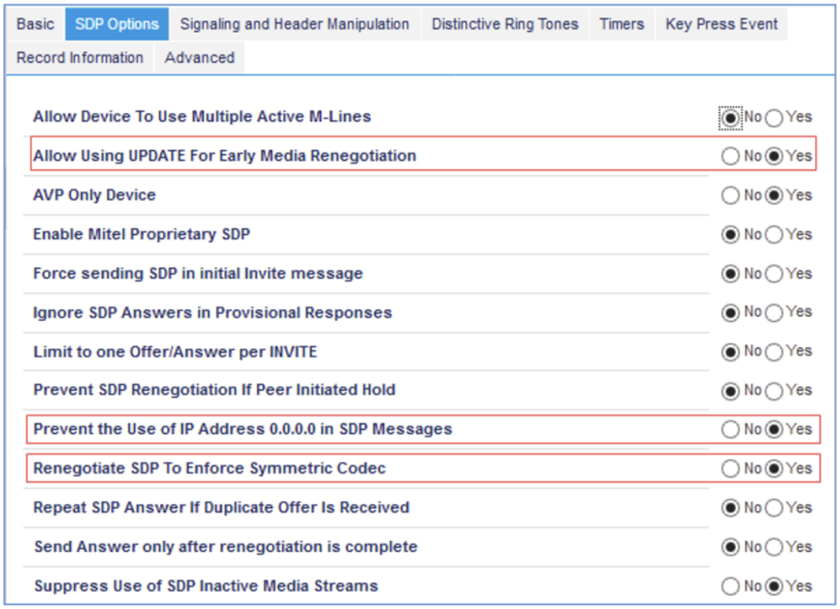
The **SIP Device Capabilities** window allows customization of features and options that the Mitel MiVoice System uses and accepts when communicating with Crestron devices. This example uses **SIP Device Capabilities Number 10**.

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

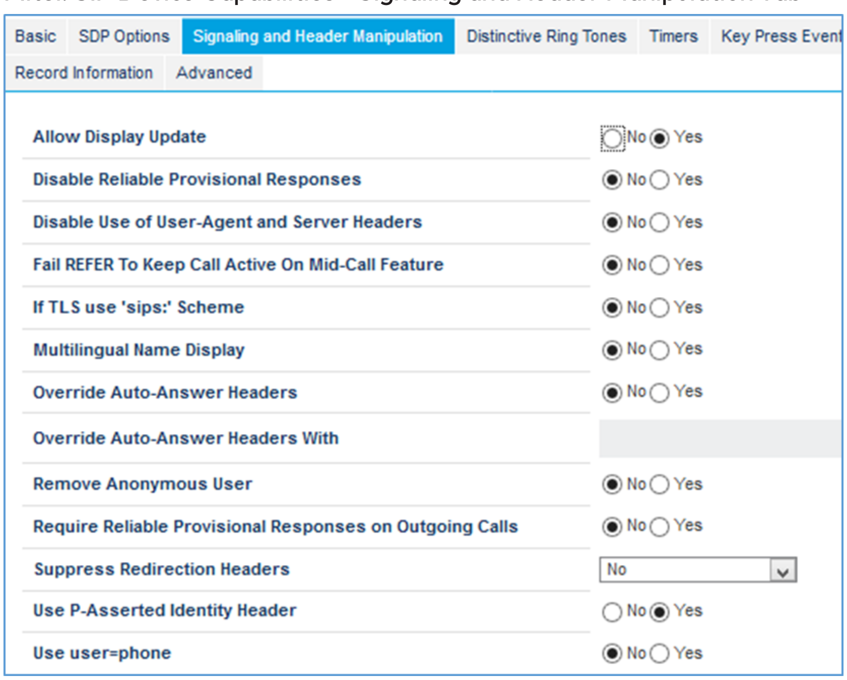
2. In the **Basic** tab, do the following:
   a. Enter **Crestron** for the **Comment** (for this example).
   b. Click **Yes** for **Replace System based with Device based In-Call Features**.
   c. Leave all other fields at the default values.

Mitel: SIP Device Capabilities - SDP Options Tab



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



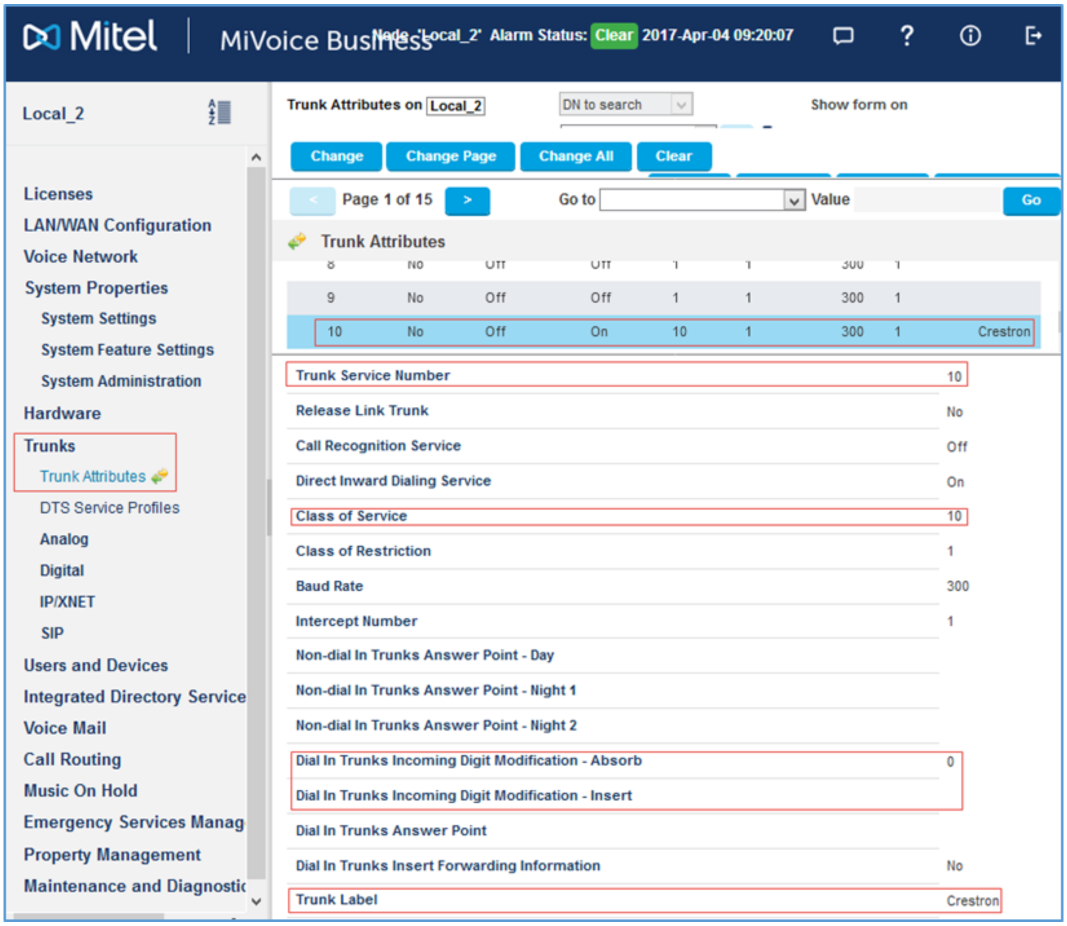
Configure Trunk Attributes

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



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.

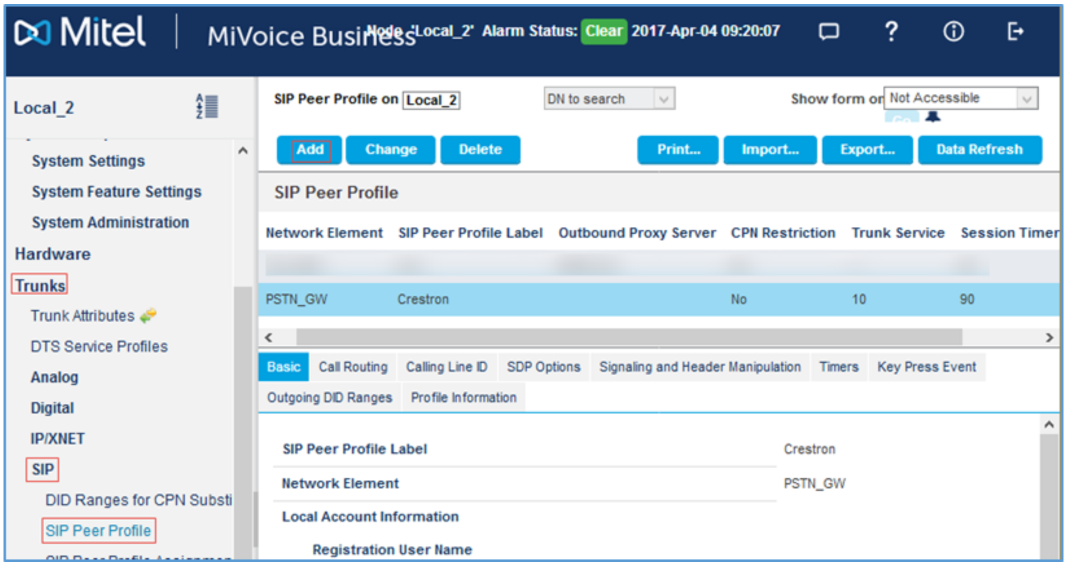
7. Leave **Dial In Trunks Incoming Digit Modification - Insert** blank.

Configure SIP Peer Profile

To configure SIP peer profile:

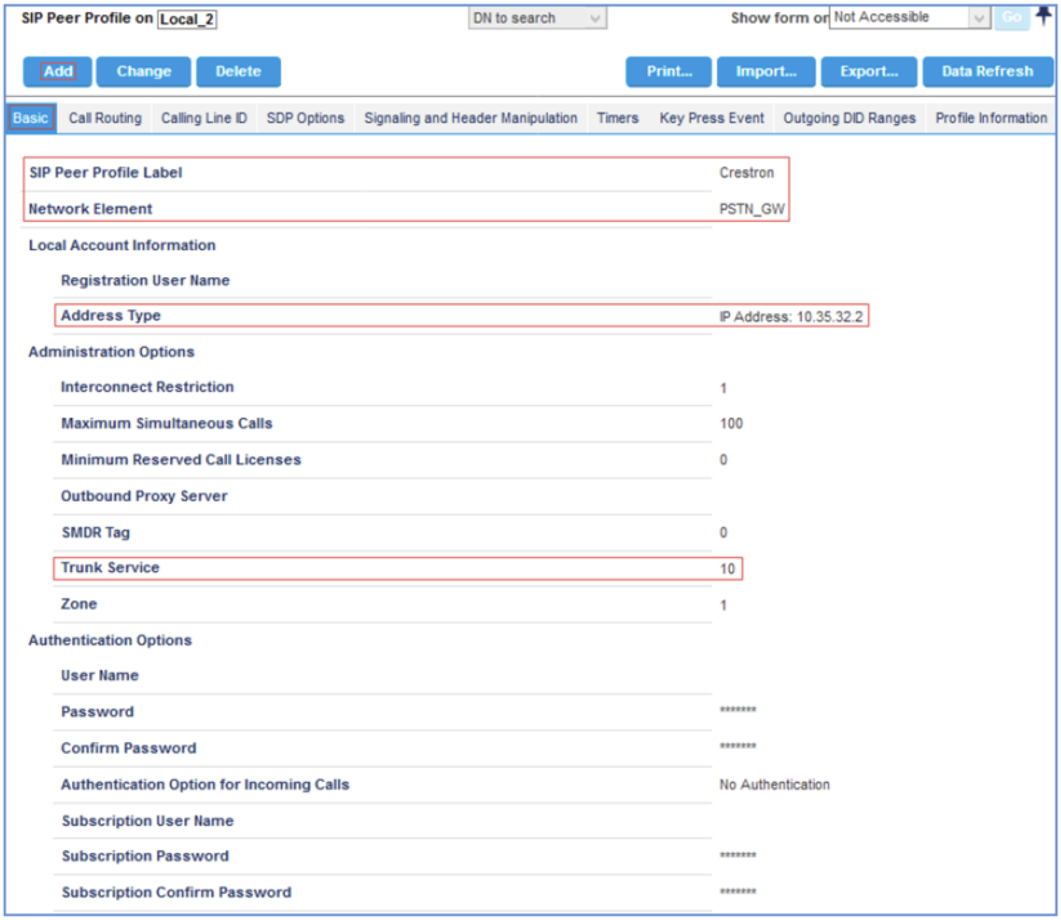
1. Click **Trunks** > **SIP** > **SIP Peer Profile**.

   Mitel: Add SIP Peer Profile

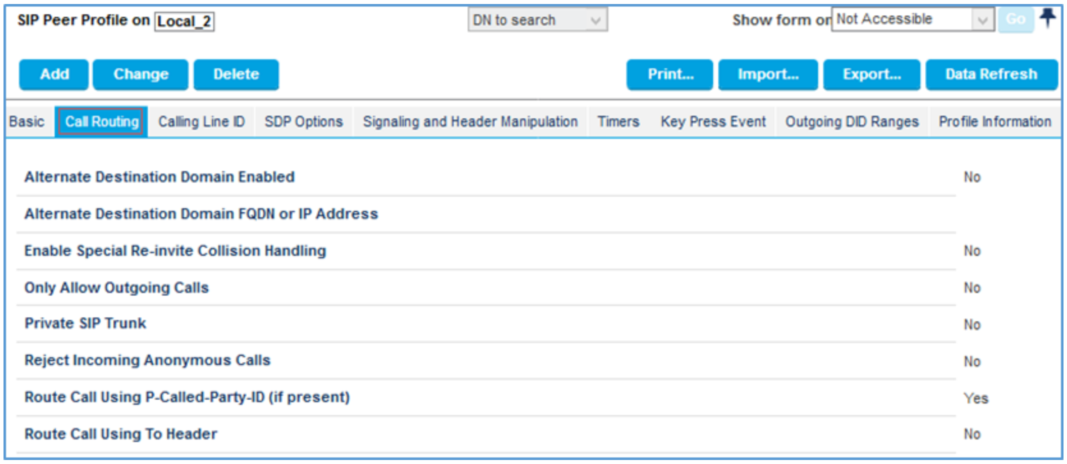
2. Click **Add**.

Mitel: SIP Peer Profile - Basic Tab



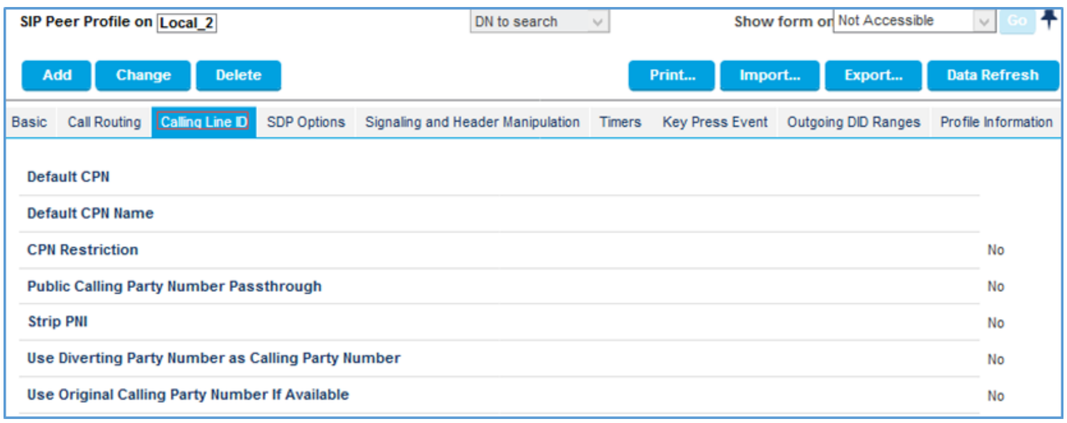
3. In the **Basic** tab, do the following:
    a. Enter a descriptive name for the **SIP Peer Profile Label.** This example uses **Crestron**.
    b. Select the **Network Element** from the drop-down menu. This example uses **PSTN_GW**.
    c. In the **Local Account Information** section, select **IP 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



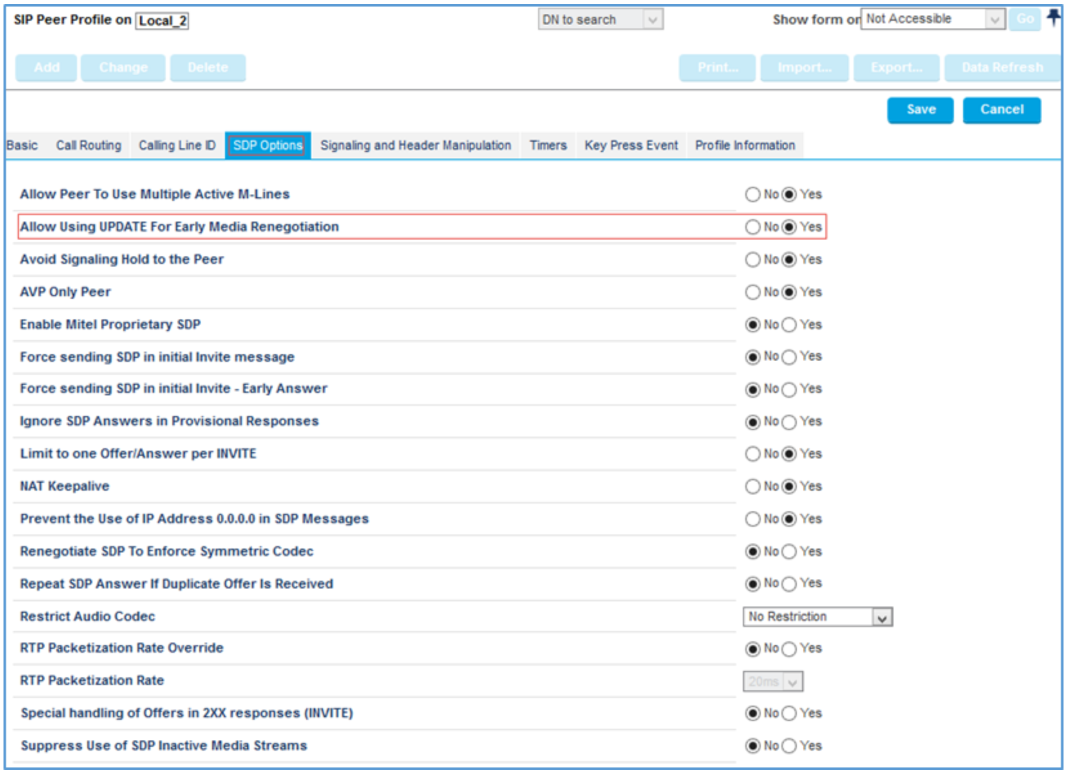
4. In the **Call Routing** tab, leave all fields at the default values.

Mitel: SIP Peer Profile - Calling Line ID Tab



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

Mitel: SIP Peer Profile - SDP Options Tab



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

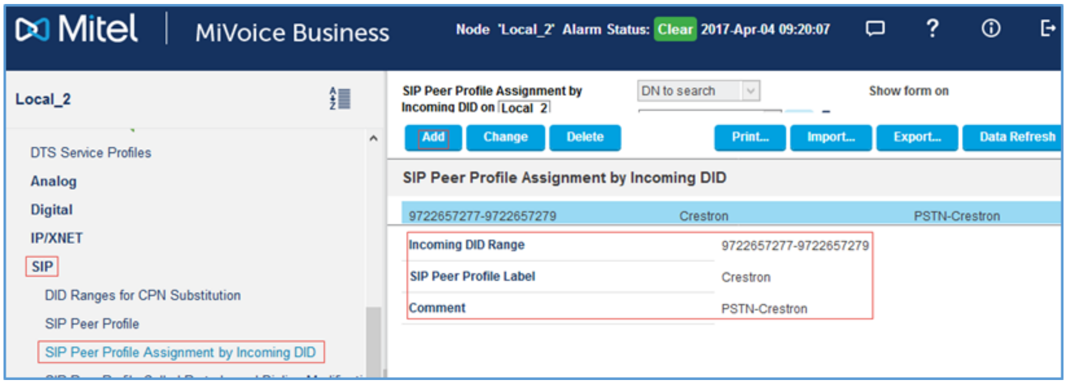
7. In the **Signaling and Header Manipulation** tab, do the following:
    a. Select **Yes** for **Allow Display Update**.
    b. Select **No** for **Require Reliable Provisional Responses on Outgoing Calls.**
8. Leave all other tabs at the default values.

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

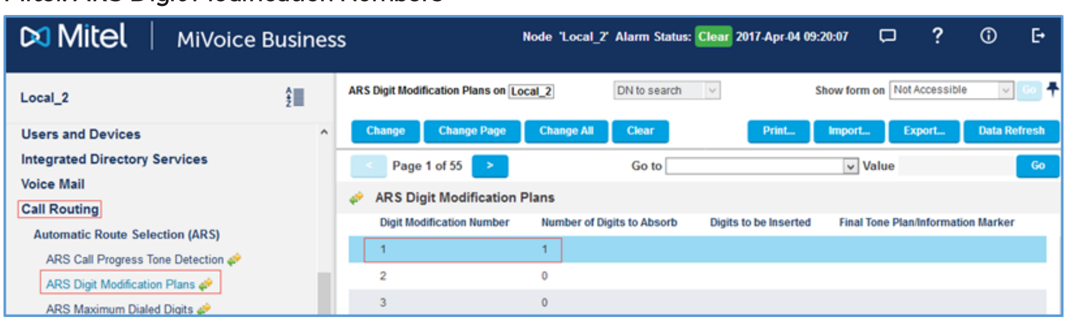


2. Click **Add**.
3. Enter **9722657277-9722657279** for the **Incoming DID Range**.
4. Enter **PSTN-Crestron** as an optional **Comment**.

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

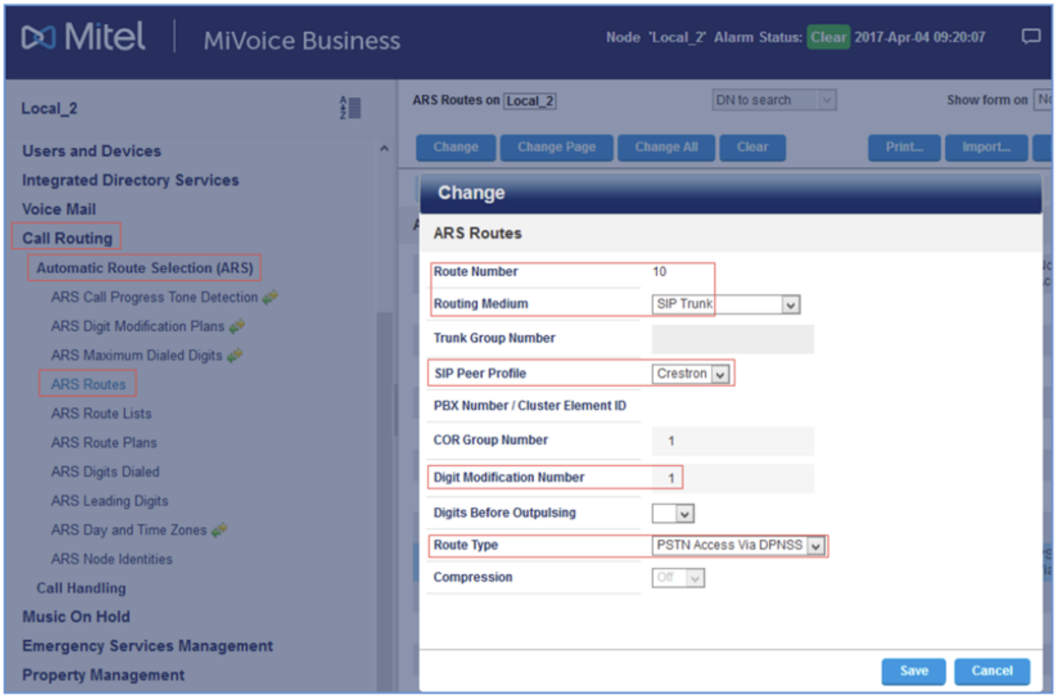


2. Change **Digit Modification Number 1**, by selecting **1** for the **Number of Digits to Absorb** while dialing out to PSTN.
3. Click **Save**.

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



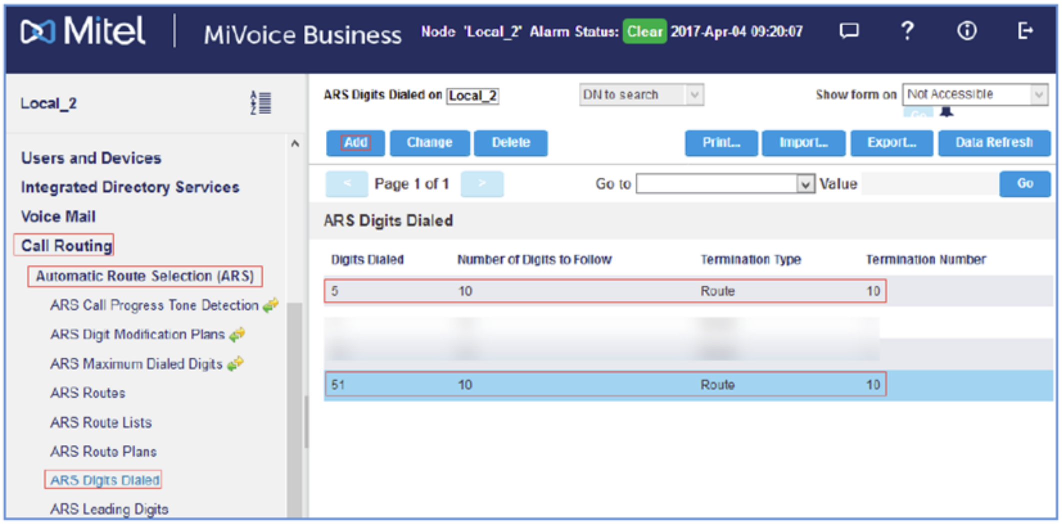
2. Select an unused **Route Number**. This example uses **10**.
3. Click **Change**.
4. Select **SIP Trunk** for the **Routing Medium**.
5. Select **Crestron** for the **SIP Peer Profile.**
6. Enter **1** for the **Digit Modification Number**.
7. Select **PSTN Access Via DPNSS** for the **Route Type**.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



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).

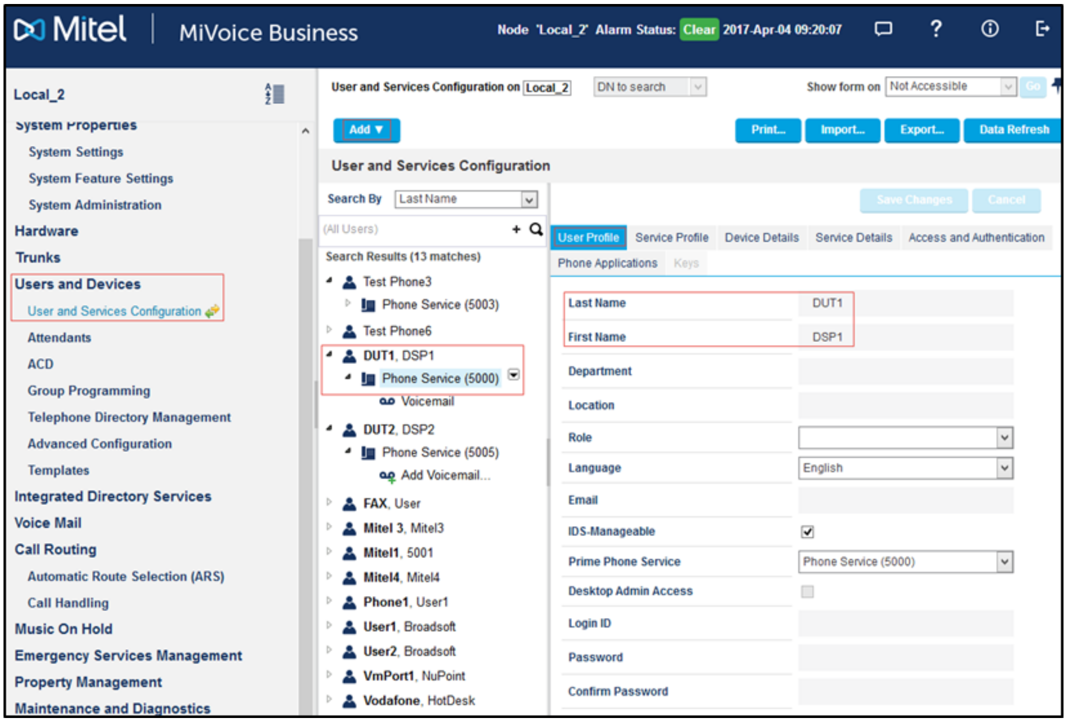
Similarly, add another entry for starting digits **51**.

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.

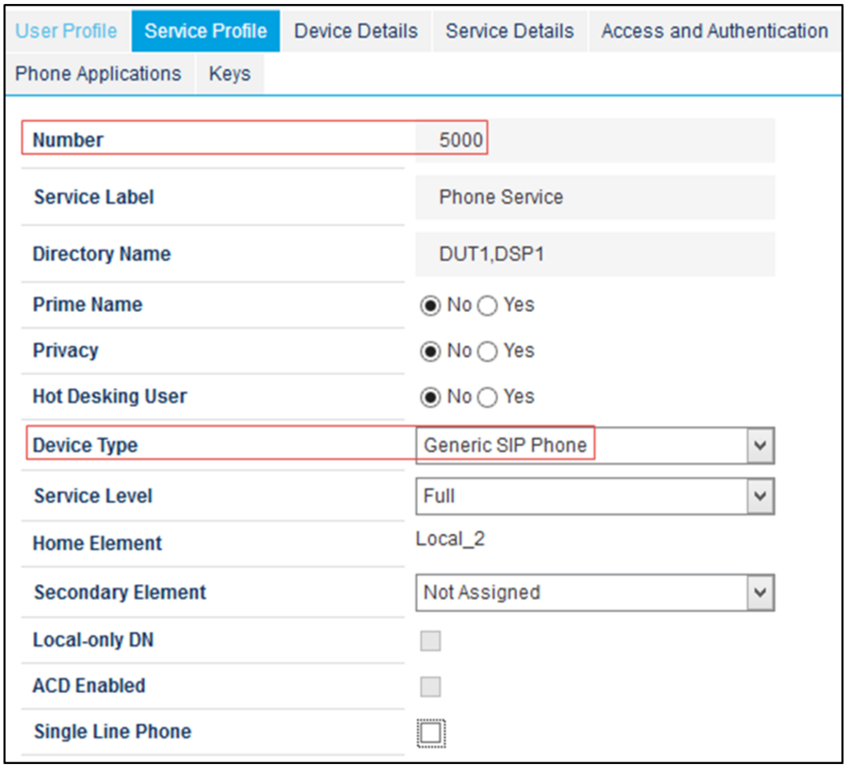
To configure a user for each device/phone (for this example):

1. Click **Users and Devices** > **User and Services Configuration**.
2. Click **Add** > **Default Users and Device**.
   Mitel: Add User - User Profile



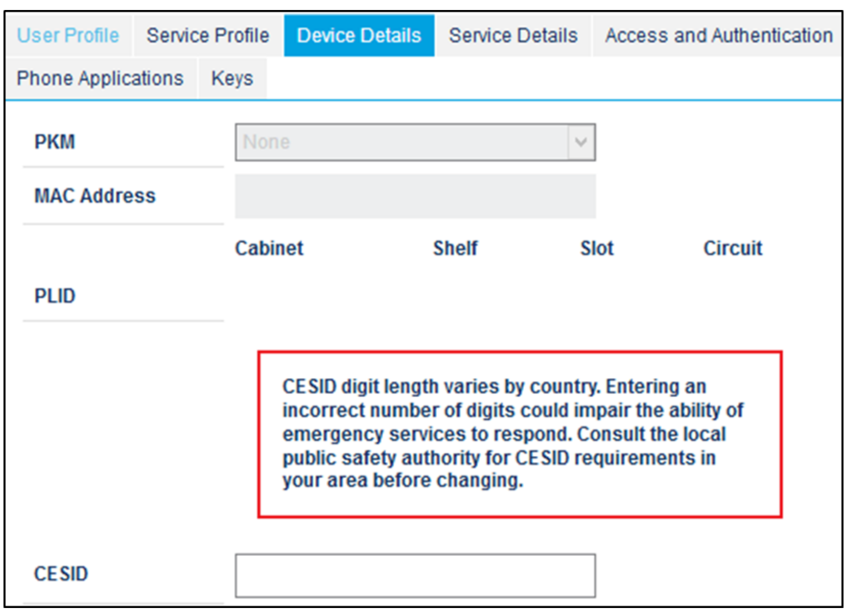
3. In the **User Profile** tab, do the following:
   a. Enter **DUT1** for the **Last Name**.
   b. Enter **DSP1** for the **First Name**.

Mitel: Add User - Service Profile

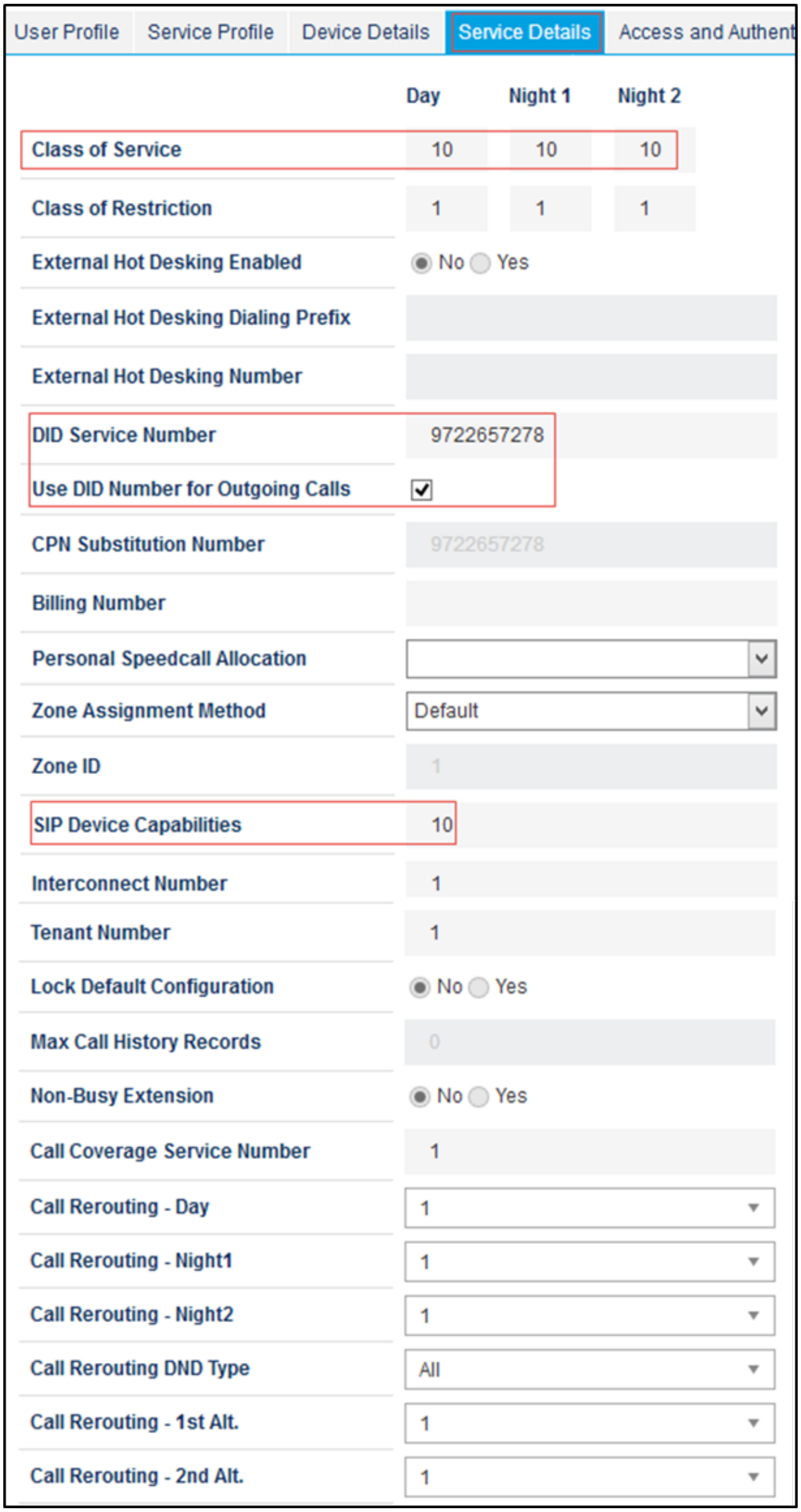


4. In the **Service Profile** tab, do the following:
    a. Enter **5000** (available DN) for the **Number**.
    b. Select **Generic SIP Phone** for the **Device Type**.

Mitel: Add User - Device Details

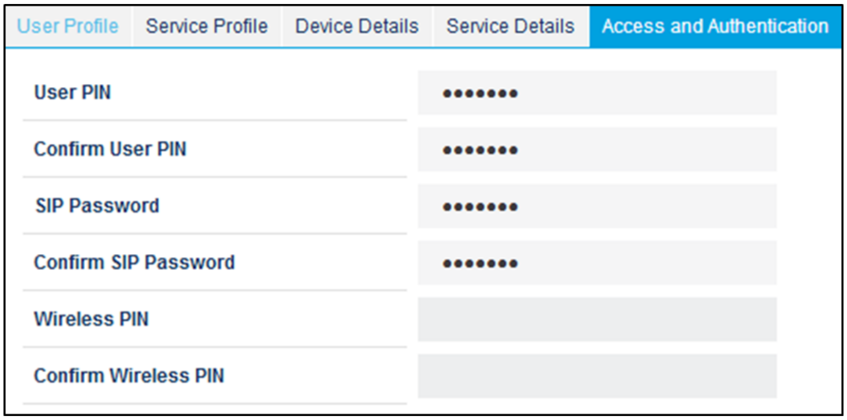


Mitel: Add User - Service Details



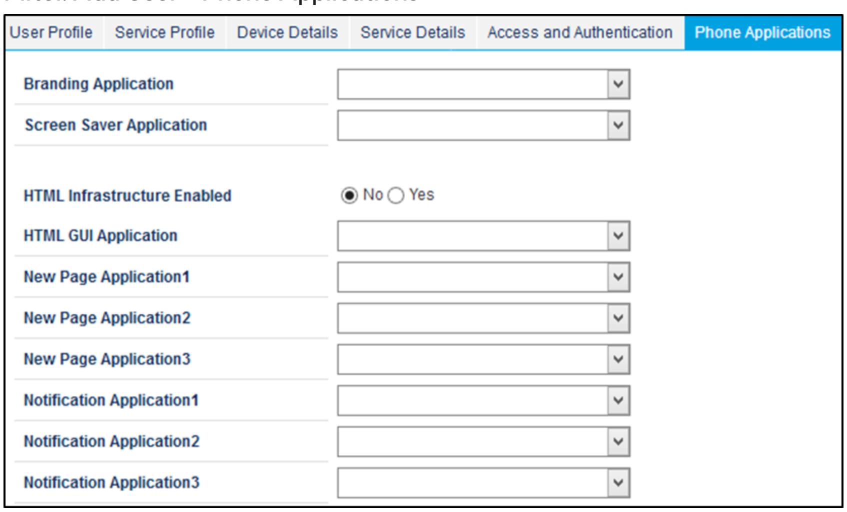
5. In the **Service Details** tab, do the following:
   a. Enter **10** for the **Class of Service**.
   b. Enter **9722657278** for the **DID Service Number**.
   c. Check **Use DID Number for Outgoing Calls**.
   d. Enter **10** for the **SIP Device Capabilities**.

Mitel: Add User - Access and Authentication



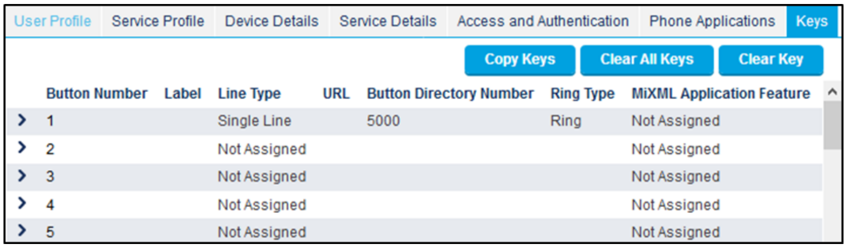
6. In the **Access and Authentication** tab, do the following:
   a. Assign the **SIP Password**. This example uses **123156**.
   b. Confirm the password by entering the same password used in the previous step.

Mitel: Add User - Phone Applications



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

Mitel: Add User - Keys



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

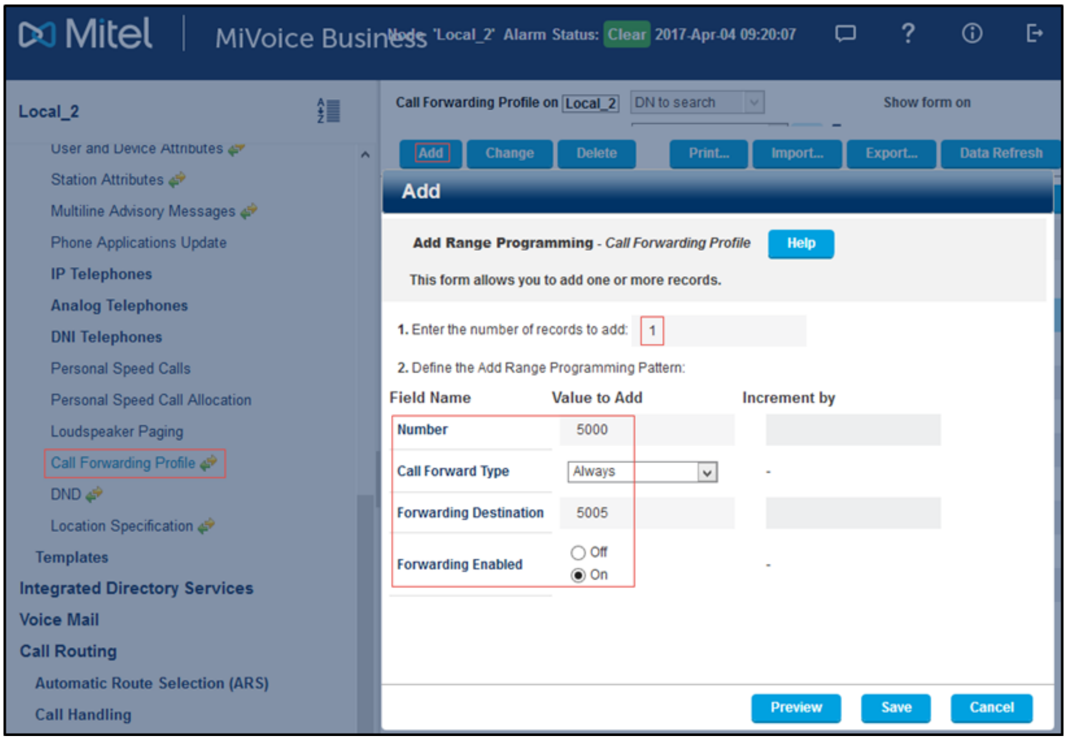
This example configures another user with DN 5005.

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

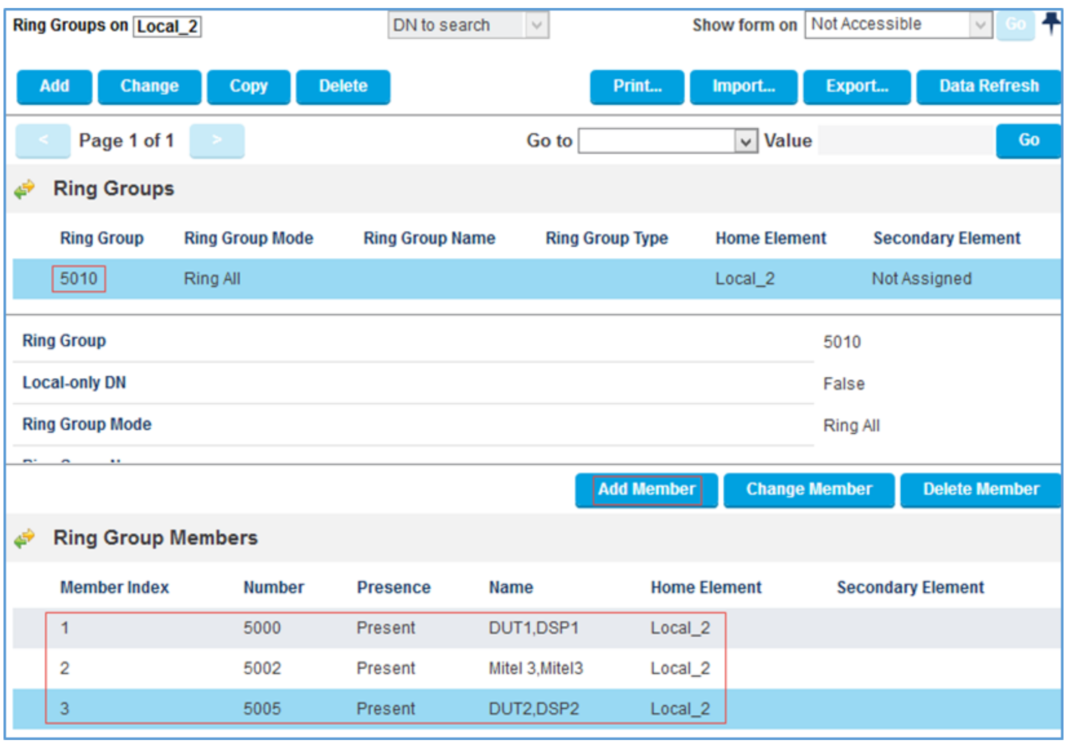
3. Enter **1** for **Enter the number of records to add**.
4. In the **Define the Add Range Programming Pattern** section, do the following:
   a. Enter **5000** for the **Number**.
   b. Select **Always** for the **Call Forwarding Type** (for this example). Other options include Busy Internal/External 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



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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