

MITEL – SIP CoE

# Technical Configuration Notes

## Configure MiVoice Office 6.1 SP1 PR2 for use with IntelePeer SIP Trunking

AUGUST 2016

SIP COE 16-4940-00469

TECHNICAL CONFIGURATION NOTES



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Mitel Technical Configuration Notes – Configure MiVoice Office for use with IntelPeer SIP Trunking

August 2016 16-4940-00469

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## Overview


This document provides a reference to Mitel Authorized Solutions providers for configuring the MiVoice Office 250 to connect to IntelPeer SIP Trunking. The different devices can be configured in various configurations depending on your VoIP solution. This document covers a basic setup with required option setup.

## Interop History

Version	Date	Reason
1	8/26/2016	Initial Interop with MiVoice Office 250 Release 6.1 SP1 PR2 and IntelPeer SIP Trunking

## Interop Status

The Interop of IntelPeer SIP Trunking has been given a Certification status. This service provider or Trunking device will be included in the SIP CoE Reference Guide. The status IntelPeer SIP Trunking achieved is:

	The most common certification which means IntelPeer SIP Trunking has been tested and/or validated by the Mitel SIP CoE team. Product support will provide all necessary support related to the interop, but issues unique or specific to the 3rd party will be referred to the 3rd party as appropriate.
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## Software & Hardware Setup

This was the test setup to generate a basic SIP call between IntelPeer SIP Trunking and the MiVoice Office.

Manufacturer	Variant	Software Version
Mitel	MiVoice Office	Release 6.1 SP1 PR2
Mitel	Minet Sets: 5320, 5360, 5312	6.03.00.12
Mitel	MiVoice Border Gateway	9.2.0.23
Service Provider	IntelPeer	N/A

## Tested Features

This is an overview of the features tested during the Interop test cycle and not a detailed view of the test cases. Please see the SIP Trunk Side Interoperability Test Plans (08-4940-00034) for detailed test cases.

Feature	Feature Description	Issues
Basic Call	Making and receiving a call through IntelPeer and their PSTN gateway, call holding, call forwarding, transferring, conferencing, busy calls, DTMF RFC2833, long calls durations, variable codec, G.711 and G.729 Codec, Privacy, Loop back calling, Long Ringing	
Automatic Call Distribution	Making calls to an ACD environment with RAD treatments, Interflow and Overflow call scenarios and DTMF detection	
NuPoint Voicemail	Terminating calls to a NuPoint voicemail boxes as well as Embedded voicemail and DTMF detection	
Packetization	Forcing the MiVoice Office to stream RTP packets through its E2T card at different intervals, from 10ms to 90ms	
Personal Ring Groups	Receiving calls through IntelPeer and their PSTN gateway to a personal ring group. Also moving calls to/from the prime member and group members	
Video	Making and receiving a call through IntelPeer with video capable devices	
Fax	T.38 and G711Fax Calls	

 - No issues found

 - Issues found, cannot recommend to use

 - Issues found

## Device Limitations and Known Issues

This is a list of problems or not supported features when IntelPeer SIP Trunking is connected to the MiVoice Office.

Feature	Problem Description
Video Call	IntelPeer does not support video calls <b>Recommendation:</b> Contact IntelPeer for update on this feature
Packetization	IntelPeer supports 20MS packetization only <b>Recommendation:</b> set packetization rate as listed in the configuration section later in this document.



## Network Topology

This diagram shows how the testing network is configured for reference

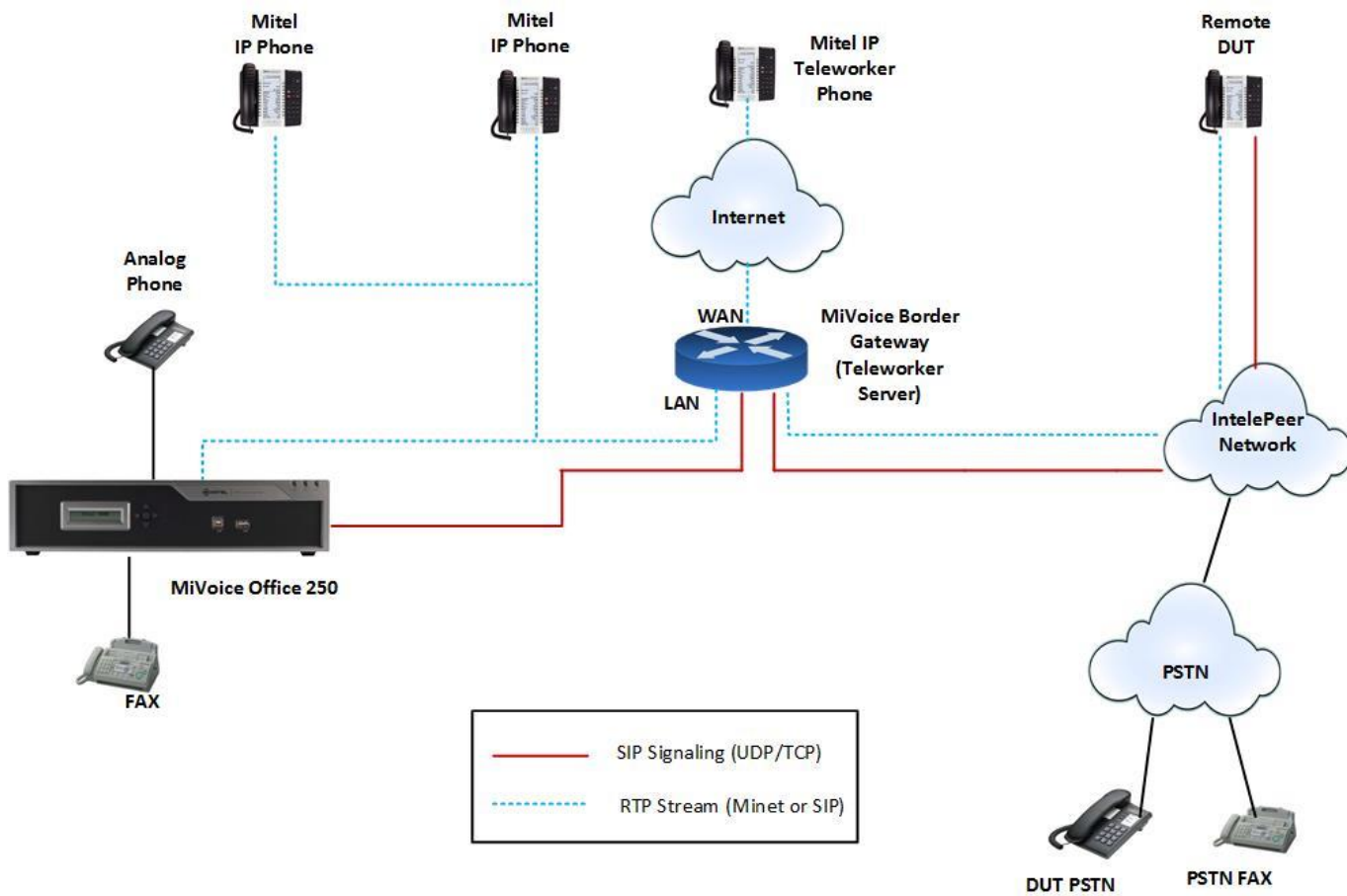


Figure 1: Network Topology

## Configuration Notes

This section is a description of how the SIP Interop was configured. These notes should give a guideline how a device can be configured in a customer environment and how MiVoice Office programming with IntelPeer SIP Trunking was configured in our test environment.

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**Disclaimer: Although Mitel has attempted to setup the interop testing facility as closely as possible to a customer premise environment, implementation setup could be different onsite. YOU MUST EXERCISE YOUR OWN DUE DILIGENCE IN reviewing, planning, implementing, and testing a customer configuration.**

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## MiVoice Office 250 Configuration Notes

The following steps show how to program a MiVoice Office 250 to interconnect with IntelPeer.

### Network Requirements

- There must be adequate bandwidth to support the voice over IP. As a guide, the Ethernet bandwidth is approx. 85 Kb/s per G.711 voice session and 29 Kb/s per G.729 voice session (assumes 20ms packetization). As an example, for 20 simultaneous SIP sessions, the Ethernet bandwidth consumption will be approx. 1.7 Mb/s for G.711 and 0.6Mb/s. Almost all Enterprise LAN networks can support this level of traffic without any special engineering. Please refer to the 3300 Engineering guidelines for further information.
- For high quality voice, the network connectivity must support a voice-quality grade of service (packet loss <1%, jitter < 30ms, one-way delay < 80ms).

### Assumptions for the MiVoice Office 250 Programming

- The SIP signaling connection uses UDP on Port 5060

## Licensing and Option Selection – SIP Licensing

Ensure that the MiVoice Office 250 is equipped with enough SIP trunk licenses for the connection to IntelPeer. This can be verified under the **Software License** form.

File View Operations Tools Favorites Help

Recent MiVoice Office 250 Software License

MiVoice Office 250

- Maintenance Accounts
- Software License
- System
- Users
- Voice Processor

Software License Feature	Value
System Type	MiVoice Office 250
ACD Hunt Group	Yes
Additional T1/E1/PRI Ports	0
Agent Help	Yes
Analog Voice Mail Hunt Group	No
Category 'A' Phones	5
Category 'B' Phones	5
Category 'C' Phones	5
Category 'D' Phones	5
Category 'E' Phones	5
Category 'F' Phones	5
Desktop Interface	No
Dynamic Extension Express	Yes
File-Based MOH Sources	5
Hot Desking	Yes
IP Networking	6
Meet-Me Conferencing	Yes
Remote ACD Hunt Groups	Yes
SIP Trunks	9
SIP Voice Mail Ports	4
System OAI Events	Yes
System OAI Third Party Call Control	Yes
Virtualized PS-1 Support	No
Voice Processor Messaging Networking	No
Unified Voice Messaging Ports	4
Unified Voice Messaging Blackberry® Integration	Yes
Unified Voice Messaging E-mail Synchronization	Yes
User Web Portal	Yes

Figure 2: License Selection

## Creating and Configuring a SIP Peer Trunk Group

Navigation: **System** -> **Device and Feature Codes** -> **SIP Peer** -> **SIP Trunk Groups**

To create a SIP Trunk Group for IntelPeer, right click in the right hand window panel under **SIP Trunk Groups** and then select “**Create SIP Trunk Group**”. A pop-up window shows and input **Start Extension, 92002** is given for this test and then click **OK**.

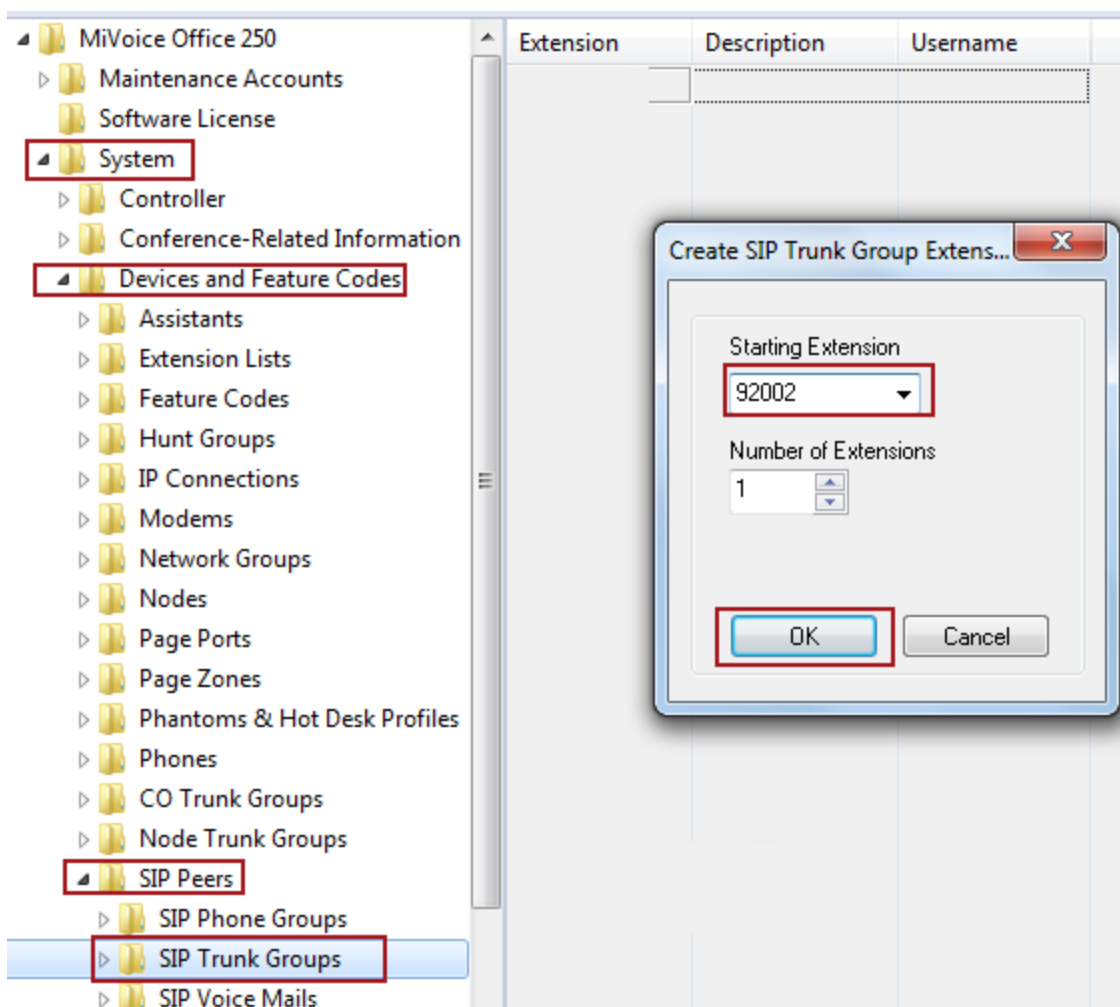


Figure 3: Create SIP Trunk Group

## Program the Configuration folder as described below

Navigation: **System -> Device and Feature Codes -> SIP Peer -> SIP Trunk Groups -> 92002 -> Configuration**

- **Registration:** If the SIP peer does not require registration, the fields in this folder do not need to be configured. The **Enable Registration** option is set to **No** by default and the remaining fields appear with a red “X”.
- **Authentication:**
  - **Username:** This field applies only if the SIP peer requires registration or call authentication
  - **Password:** This field applies only if the SIP peer requires registration or call authentication
- **Keep-Alive:** The Keep-Alive option keeps refreshing the NAT bindings for any Firewall/NAT in the path. It also helps in determining whether the SIP peer is reachable or not.
- **NAT Settings:** Specifies the NAT address type. The default is “No NAT or SIP-Aware NAT” (for systems that are using a SIP-aware firewall). If you are not using a SIP-aware firewall, you must change the setting to “Non SIP-Aware NAT”.
- **Alternate IP/FQDN List:** Some providers use multiple IP addresses to send SIP messages to the MiVoice Office 250. You must add All IP addresses or FQDNs other than the primary IP/FQDN to the list for all calls to be successful. To make the anonymous inbound calls to work, “default” is given as FQDN as shown in figure below.
- **Route Sets:** Add the IP address of the MBG LAN to the route set, **10.64.3.2** is given for this test
- **IP Address:** Indicates the **IP address** of the **IntelePeer** side. Please contact IntelePeer for your deployment.
- **Port Number:** Indicates the port that the system listens on the system for SIP peer messages. The range is 0–65535, **5060** is used for this setup.
- **Fully Qualified Domain Name:** Indicates the domain name of the SIP peer trunk group. Leave it blank.
- **Call Configuration:** **Call Configuration 1** is used for this setup
- **Operating State:** Indicates the operating state of the SIP peer. Set it to **In-Service**.
- **Maximum Number of Calls:** Indicates the maximum number of concurrent calls that are permitted towards the SIP peer. DB Programming restricts this field based on the number of the SIP Trunks and SIP trunk licenses.
- **Use ITU-T E.164 Phone Number:** If set to Yes, the MiVoice Office 250 handles ITU-T E.164 formatted phone numbers as part of the incoming SIP INVITE messages from the SIP peer. **No** is set for this setup.
- **DTMF Decoding Payload:** **101** is used for the setup as IntelePeer uses the same payload for DTMF

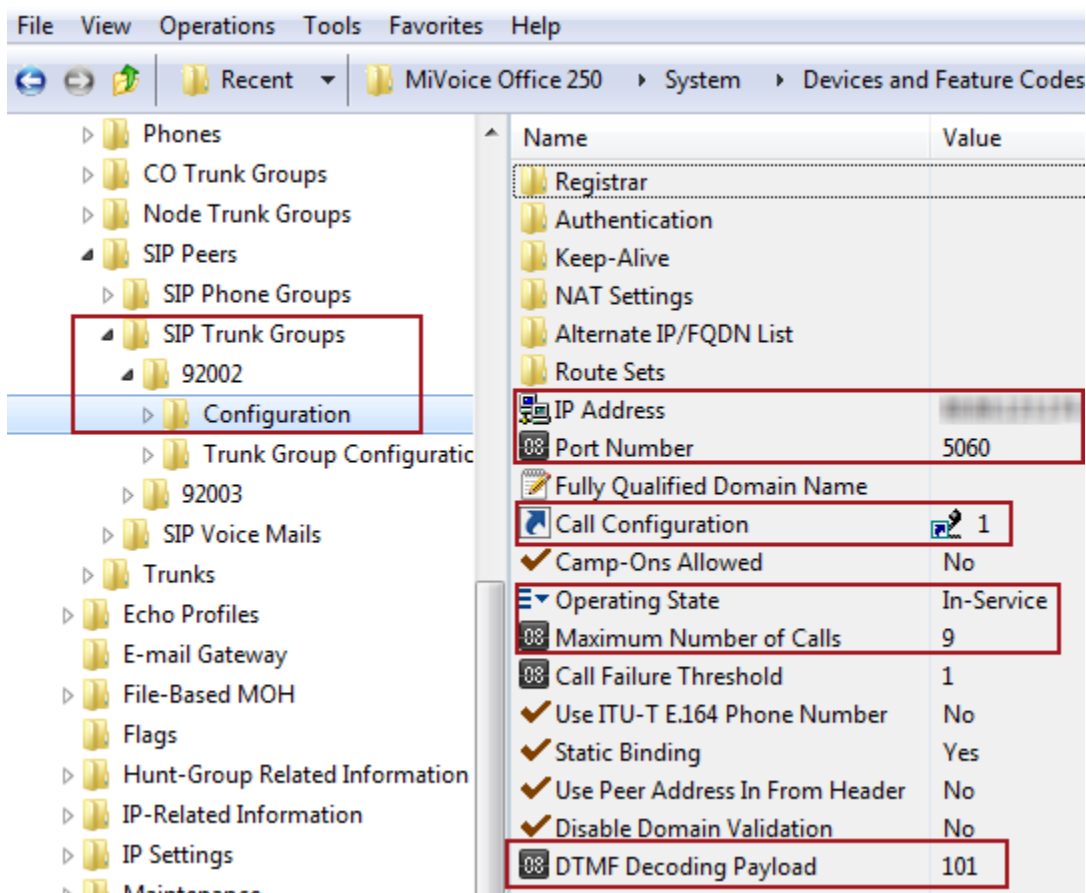


Figure 4: SIP Trunk Group for IntelPeer

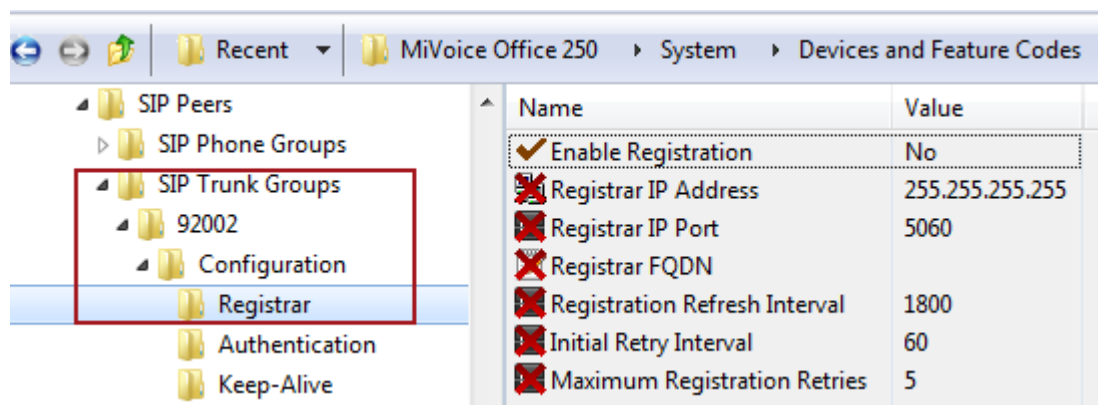


Figure 5: Registration not required for IntelPeer

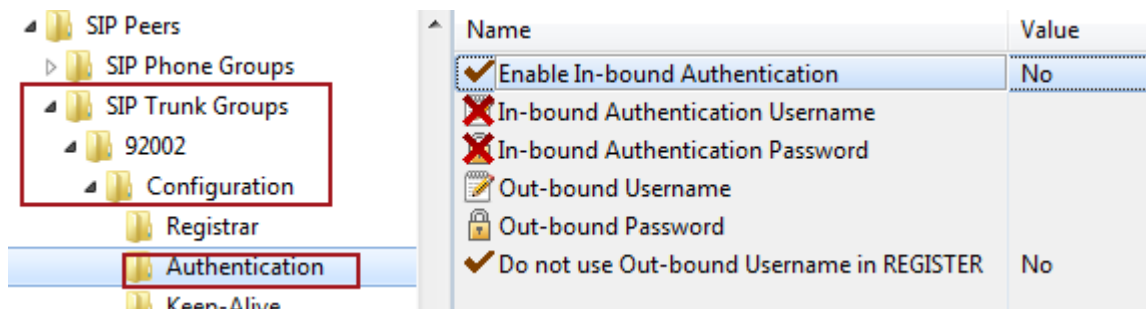


Figure 6: Authentication Not Required for IntelPeer

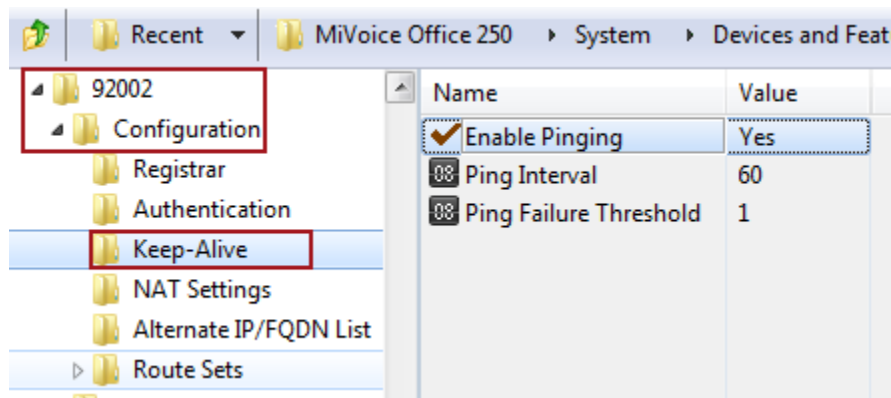


Figure 7: IntelPeer SIP Trunk Group - Keep-Alive

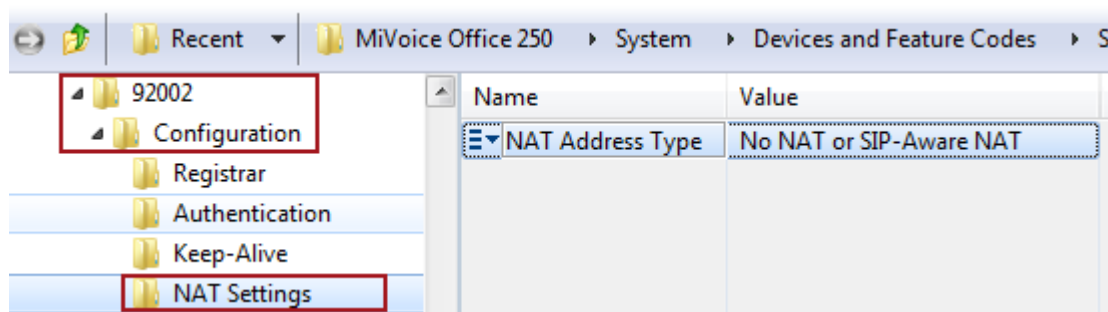


Figure 8: IntelPeer SIP Trunk Group: NAT Setting

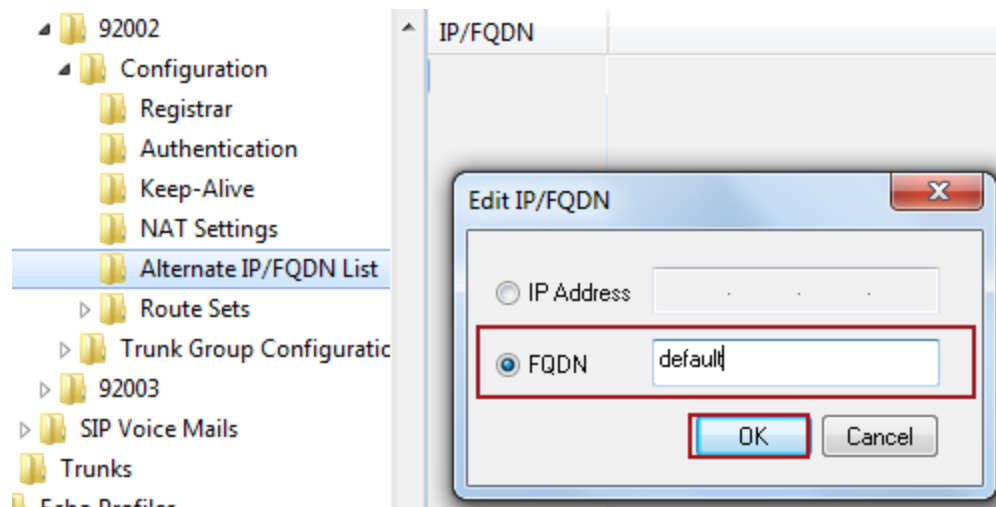


Figure 9: IntelPeer SIP Trunk Group: Alternate IP/FQDN

### Create Route Set for MBG

**Add to Route Sets List:** Under **SIP Peer – SIP Trunk Group – Configuration**, add **Route Set** using IP address of the MBG (Mitel Border Gateway)

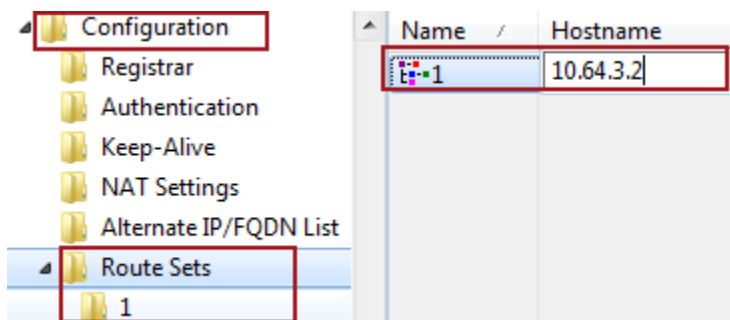


Figure 10: IntelPeer SIP Trunk Group - Route Set

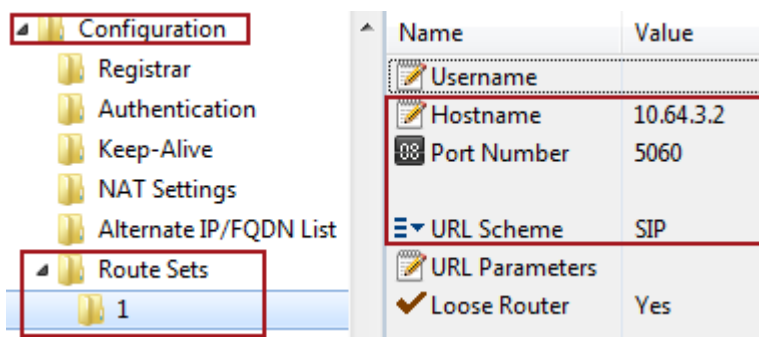


Figure 11: IntelPeer SIP Trunk Group - Route Set – Cont.



## Programming the Trunk Group Configuration Folder

Navigation: **System -> Device and Feature Codes -> SIP Peer -> SIP Trunk Groups -> 92002 -> Trunk Group Configuration.**

- **Ring-In Type Day/Night:** Set **Call Routing Table 1** for both **Day and Night Ring-In Type** for this setup, please refer to section [Call Routing Table](#)
- **Music-On-Hold:** **File-based MOH** is selected for this test
- **Audio on Transfer/Hold:** **File-Based MOH** is selected

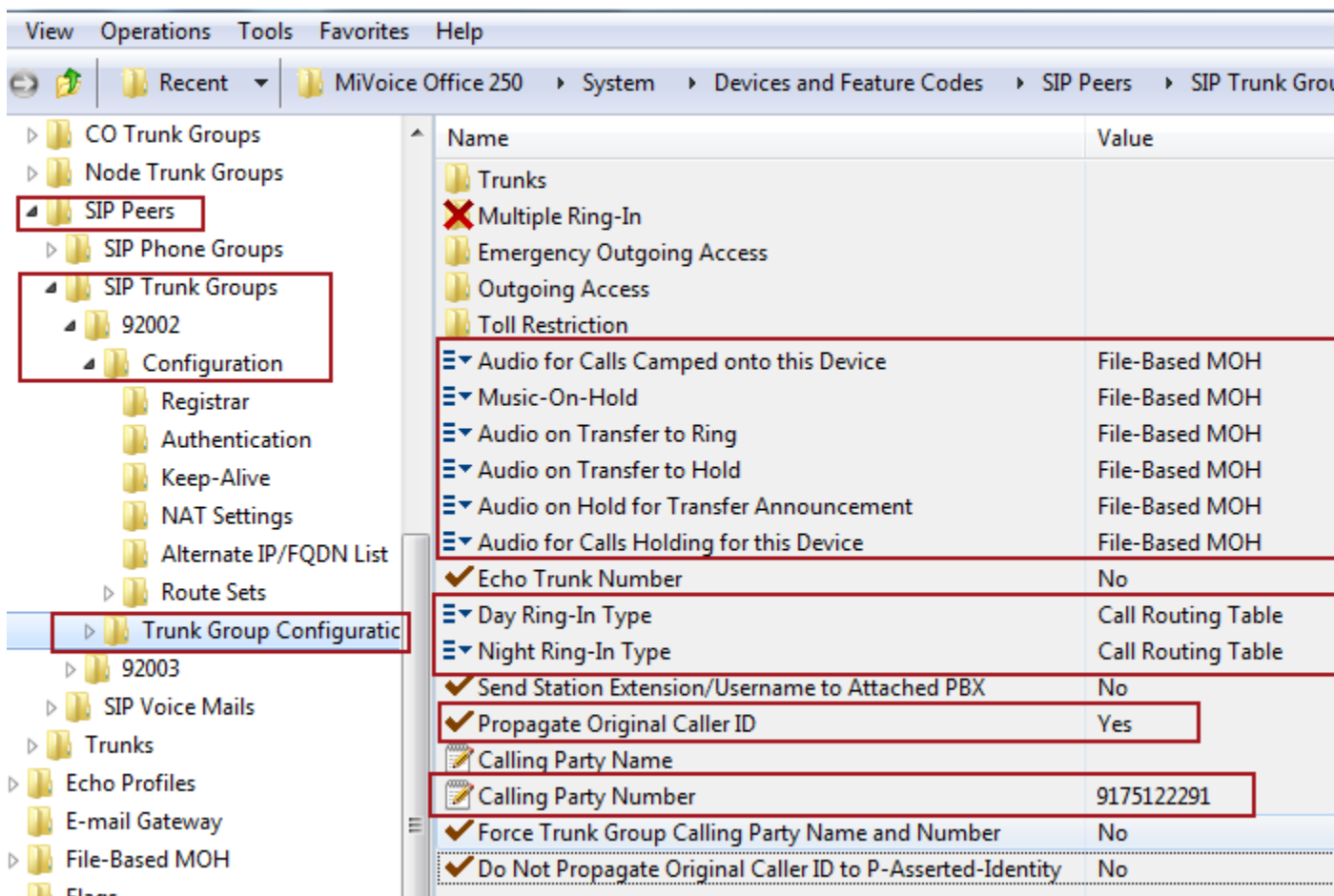


Figure 12: IntelePeer Trunk Group Configuration

Create the SIP peer trunks as follows:

Navigation: **System -> Device and Feature Codes -> SIP Peer -> SIP Trunk Groups -> 92002 -> Trunk Group Configuration -> Trunks**

- Right-click the right pane, and then select **Create SIP Peer Trunk**. The Create SIP Peer Trunk Extension dialog box appears.
- Select the extension number you want to use for the item in the **Starting Extension** field. The recommended range is 94000–94999; **94000** is used in this lab setup.
- Indicate the number of extensions you want to create in the **Number of Extensions** field. If the system is set to have more than one extension, the new trunks are assigned sequentially to the next available numbers. **9** is set for this example. The number SIP Peer trunk is restricted by the number of available SIP Trunks license.
- Click **OK**

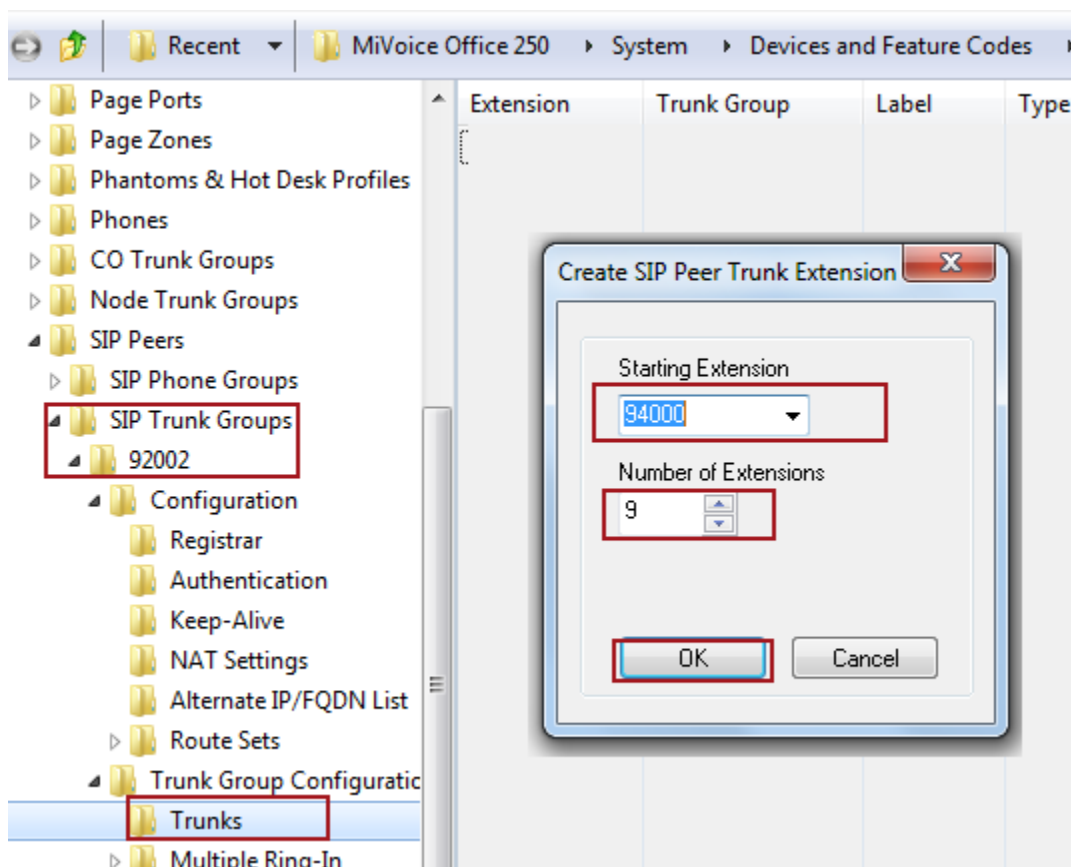


Figure 13: Create SIP Trunks

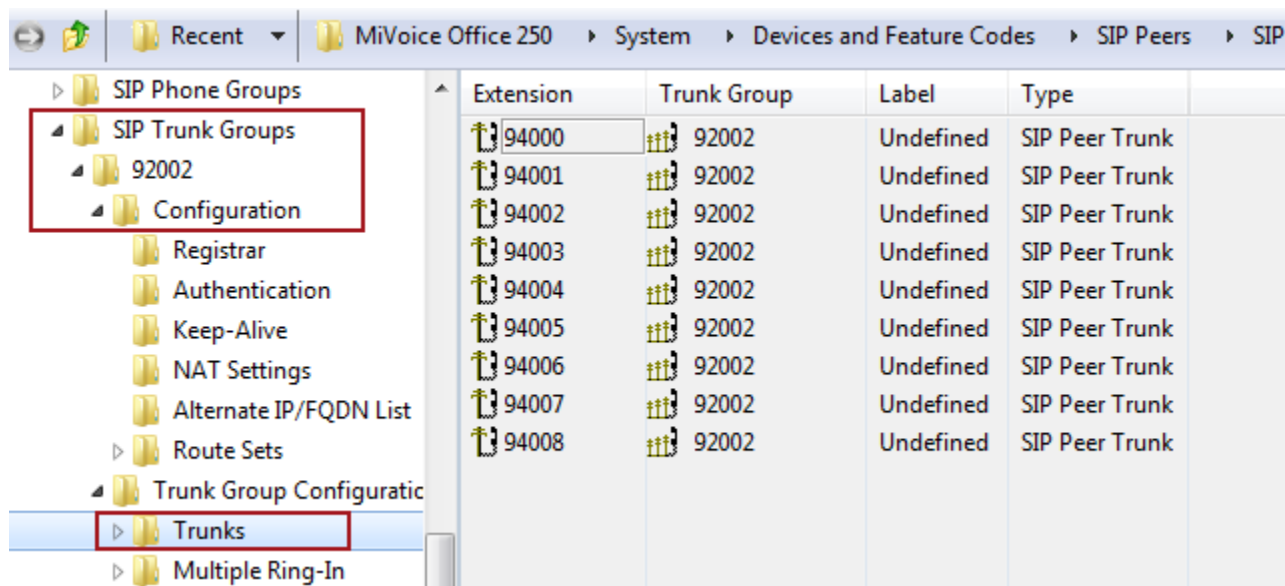
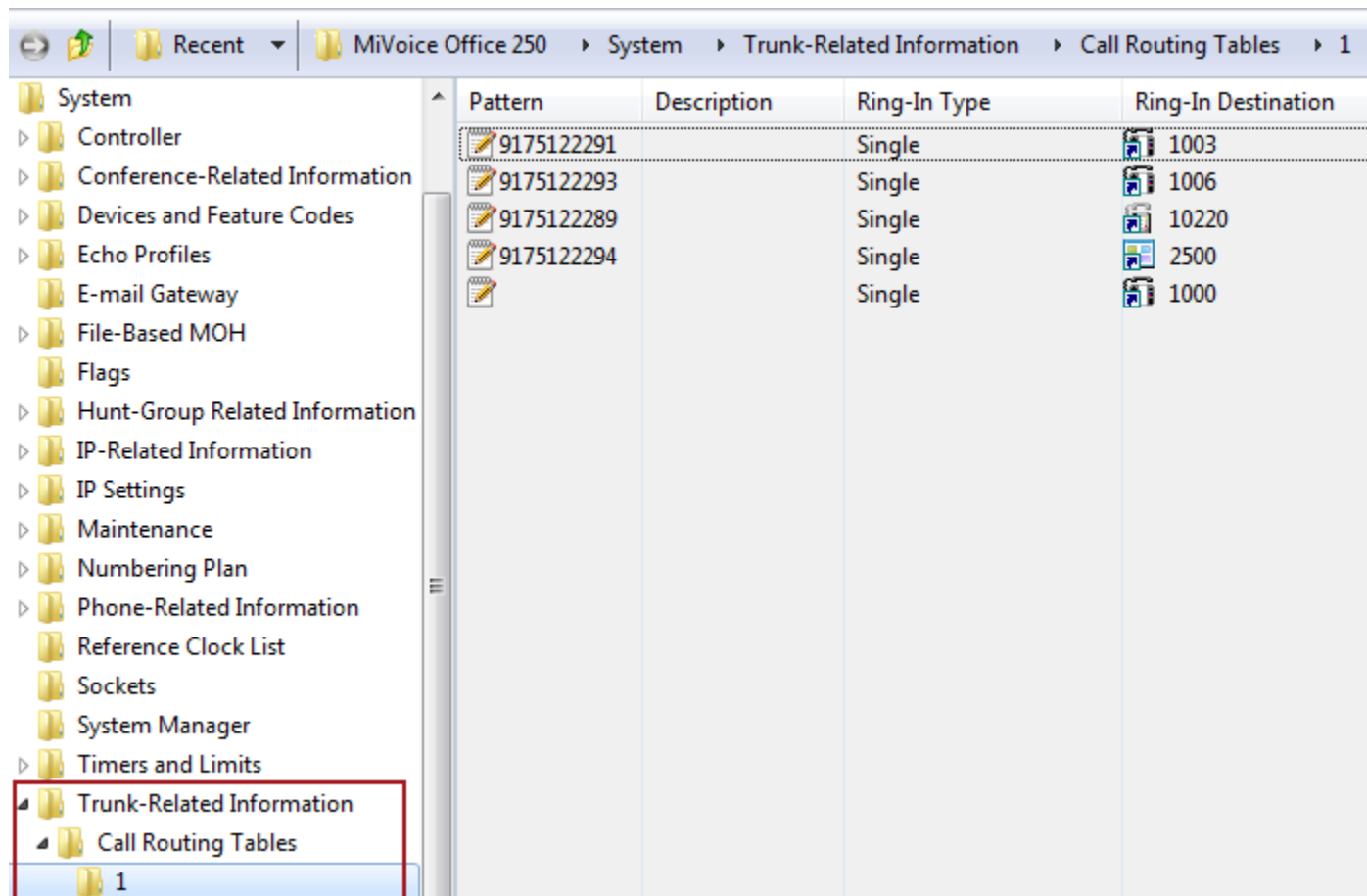


Figure 14: SIP Trunks – Cont.

## Call Routing Table

Navigation: **System** -> **Trunk-Related Information** -> **Call Routing Tables** -> **Table 1**

- **Pattern:** Set with the DID numbers assigned by IntelePeer.
- **Ring-In Type:** Default value **Single** is used for all DIDs.
- **Ring-In Destination:** set the proper target for the call to be routed to.



Recent MiVoice Office 250 System Trunk-Related Information Call Routing Tables 1

System	Pattern	Description	Ring-In Type	Ring-In Destination
Controller	9175122291		Single	1003
Conference-Related Information	9175122293		Single	1006
Devices and Feature Codes	9175122289		Single	10220
Echo Profiles	9175122294		Single	2500
E-mail Gateway			Single	1000
File-Based MOH				
Flags				
Hunt-Group Related Information				
IP-Related Information				
IP Settings				
Maintenance				
Numbering Plan				
Phone-Related Information				
Reference Clock List				
Sockets				
System Manager				
Timers and Limits				
Trunk-Related Information				
Call Routing Tables				
1				

Figure 15: Call Routing Table

## IP Call Configurations

Call configurations define the settings that IP endpoints and gateways use when connected to calls. You can assign multiple devices to a specific call configuration.

Navigation: **System -> IP-Related Information -> Call Configurations**

By default, all IP devices are placed in Call Configuration 1, which is programmable. You do not need to add SIP endpoints to Call Configurations, because these devices negotiate call configurations before establishing a connection. You can program up to 25 different Call Configurations. Call Configuration 1 was used for phone and SIP trunk, while Call Configuration 3 was used for NuPoint voice mail.

- Set **Audio Frames/IP Packet: 2** (20ms packetization rate) is set for this test
- **DTMF Encoding Setting: RFC2833** is selected for this test
- Set **Speech Encoding Setting: G711 Mu-Law** is select as IntelPeer supports G711 Codecs only
- **Fax Encoding Setting:** IntelPeer supports both G711 Mu-Law Pass-through and T.38 for fax.
- **Support RTP redirect:** for Call Configuration 1, YES is set, and No is set for Configuration 3
- Leave all other fields as default

Name	Value	Extension
Phones		
Trunks		
SIP Phone Groups		
SIP Trunk Groups		
SIP Voice Mails		
Audio Diagnostics Sampling Period	5	
Audio Diagnostics Samplings	12	
Audio Frames/IP Packet	2	
Average In Time Frame Percentage Threshold	60	
Average In Time Frame Timer	5	
Minimum Playback Time	20	
Transmit DTMF Level	North America	-9
DTMF Encoding Setting	RFC 2833	
Speech Encoding Setting	G.711 Mu-Law	
Fax Control-Messages Redundancy Count	3	
Fax Page-Data Redundancy Count	0	
Fax Detection Sensitivity	1	
Fax Encoding Setting (Fax Transmission)	T.38	
Fax Maximum Connection Speed	No Limit	
Supports RTP Redirect	Yes	

Figure 16: Call Configuration



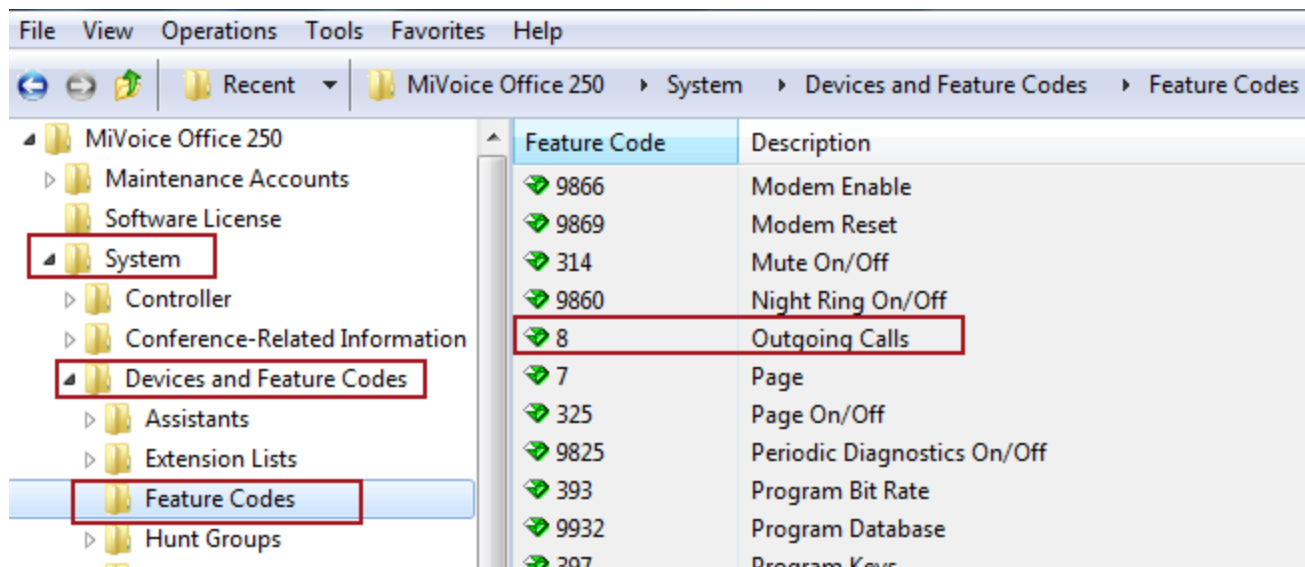


Figure 18: Feature Codes

In order to let user pickup correct trunk group for outgoing call, need to assign the proper SIP trunk Group extension to the phone:

Navigation: **System ->Device and Feature Code ->Phones ->Local -> XXXX (i.e. 1003)**  
**-> Associated Extension**

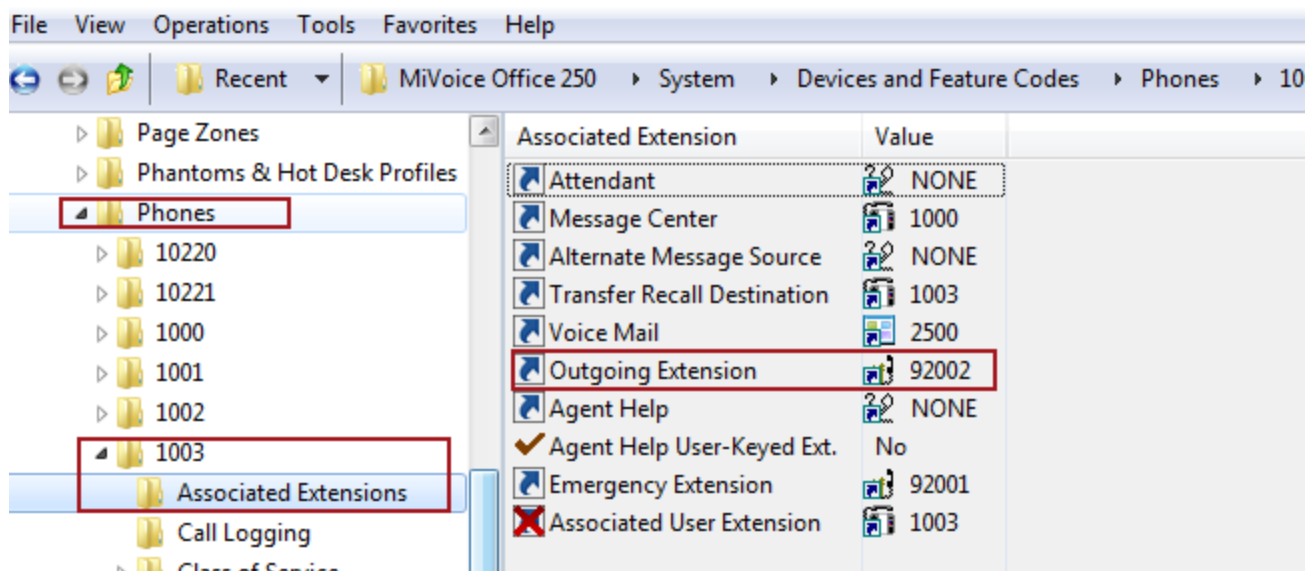


Figure 19: Associated Phone Extensions

## SIP Voice Mail Configuration (NuPoint)

MiVoice Office 250 can use embedded Basic Voice Mail or integrated with NuPoint Voice Mail. Before configure NuPoint SIP Peer Voice mail, please make sure BVM (Basic Voice Mail) is disabled.

Navigation: **Operations** -> **Voice Processor Operations** -> **Disable Unified Voice Messaging**

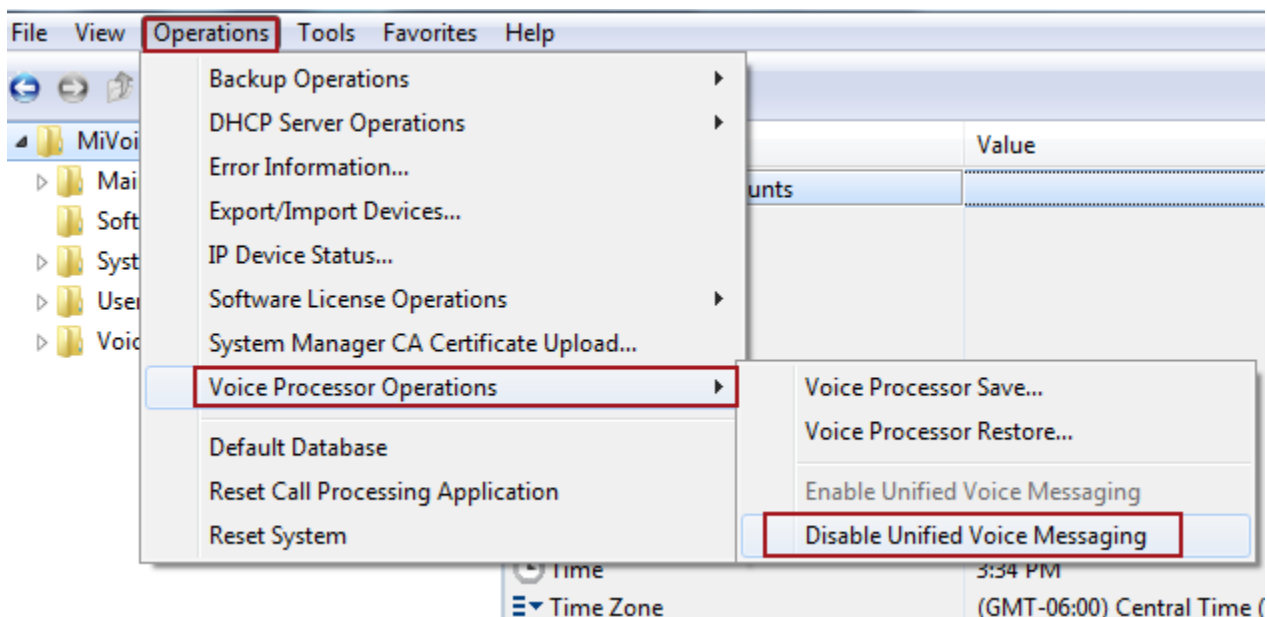


Figure 20: Disable Basic Voice Mail



## Create SIP Voice Mail

Navigation: **System** -> **Devices and Feature Codes** -> **SIP Peers** -> **SIP Voice Mails**

- First, right-click the right pane, and then select **Create SIP Voice Mail**
- A pop-up window appears and click **“YES”** to confirm this SIP Voice Mail is NuPoint UM
- The next pop-up window **“Create SIP Voice Mail Extension”** appears and set **P9001** as **Starting Extension** and **1** as **Number of Extensions**
- Click **OK**

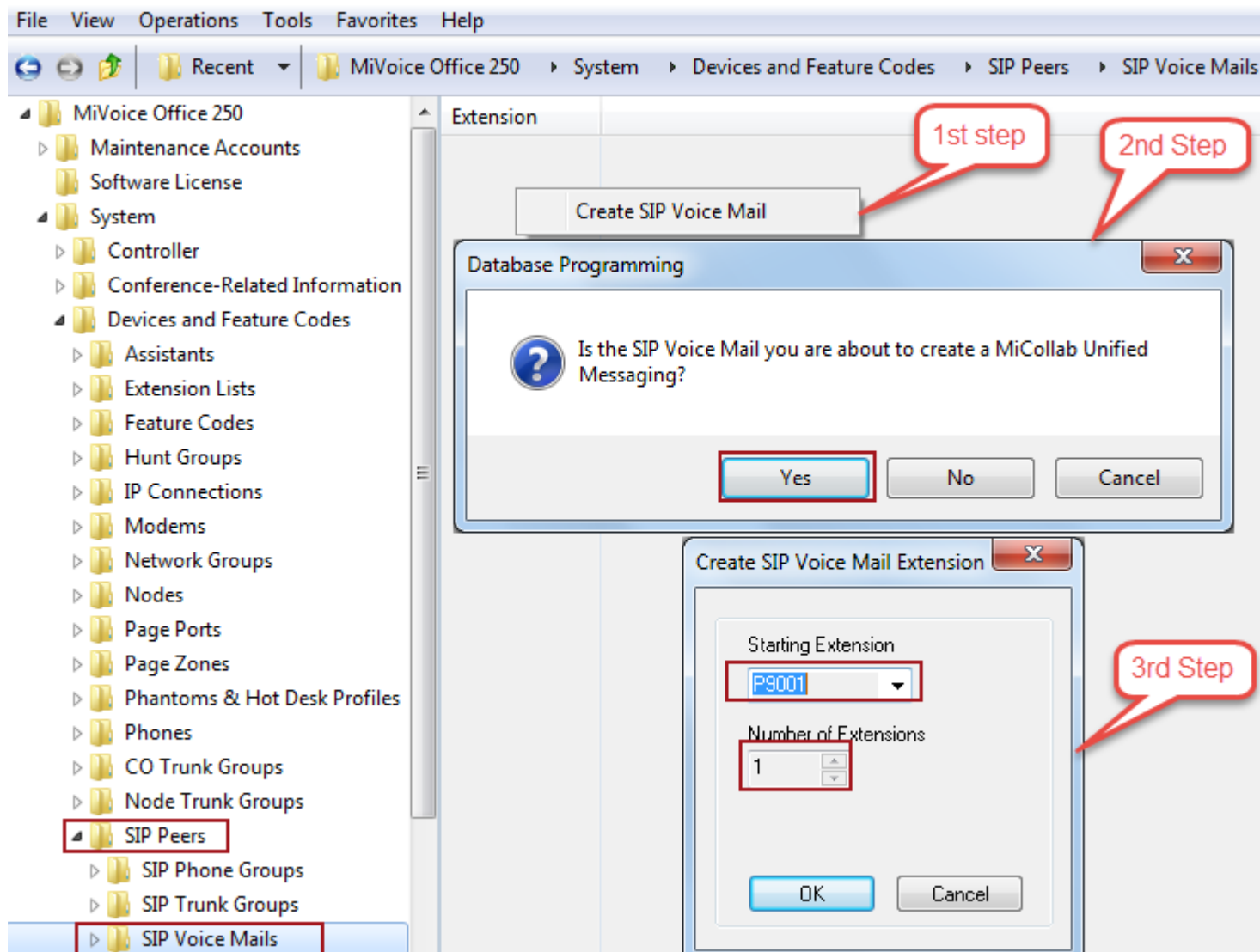


Figure 21: Create SIP Voice Mail

## SIP Voice Mail Configuration (NuPoint)

Navigation: **System -> Devices and Feature Codes -> SIP Peers -> SIP Voice Mails -> P9001 -> Configuration**

- Set **IP Address**: NuPoint UM IP Address **10.64.3.4** is given here
- Set **Port Number**: Port **5058** is given for this test as we are using NuPoint UM on MiCollab, if it is NuPoint UM Standalone, then Port 5060 will be used
- Set **Call Configuration**: Call Configuration 3 (see Section [IP Call Configurations](#)) is used for this test
- **Maximum Number of Ports**: **4** is given for this test, this number should be same as the ports under the Line Group 1 in [NuPoint UM Configuration](#)
- **DTMF Decoding Payload**: **101** is given to match SIP trunk and IntelPeer DTMF payload
- Leave all other fields as default

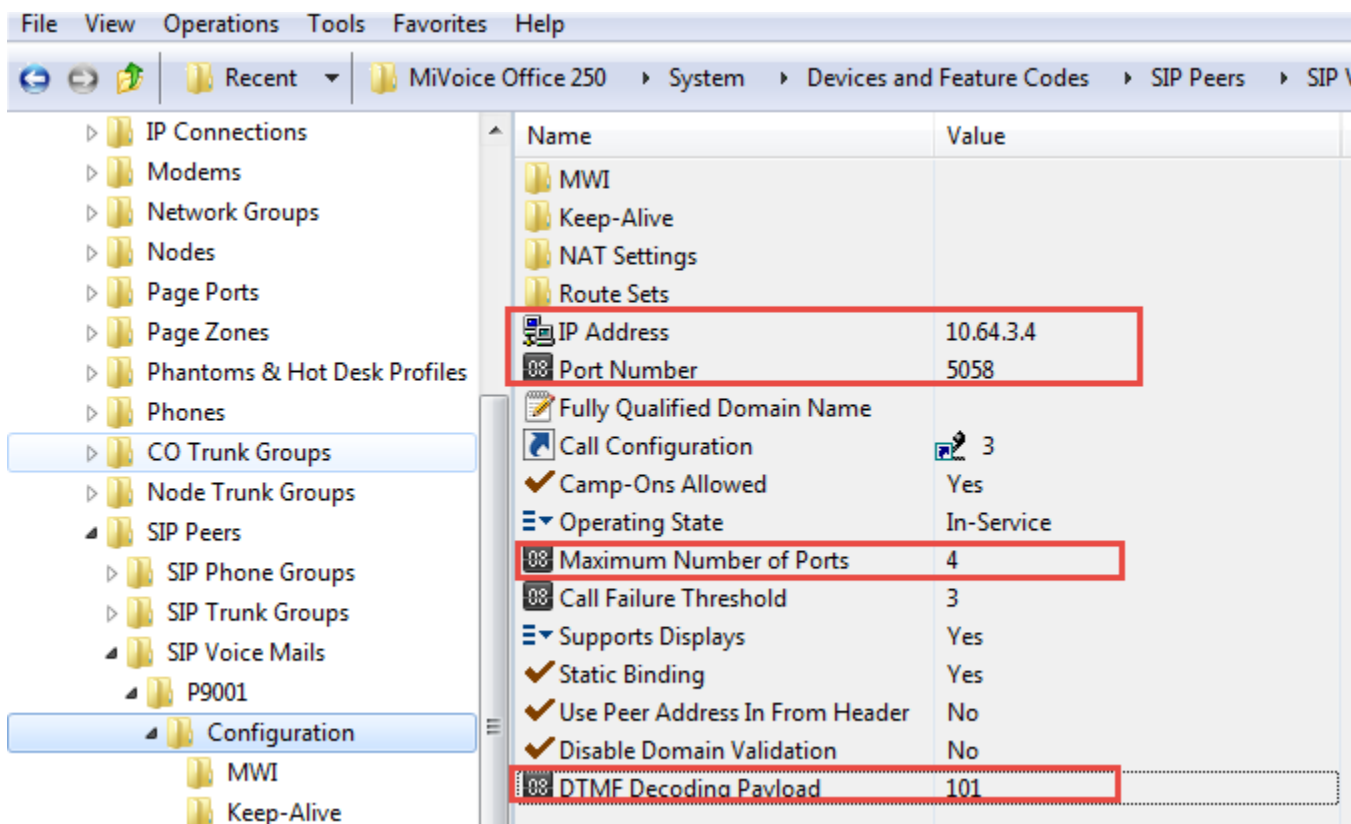


Figure 22: SIP Voice Mail Configuration

## SIP Voice Mail Pilot (NuPoint)

Navigation: **System -> Devices and Feature Codes -> SIP Peers -> SIP Voice Mails -> P9001 -> Applications**

- Right-click the right pane, and then select **Create Voice Mail**
- At new pop-up window, set **2600** as **Starting Extension** and **1** as **Number of Extensions**
- Click **OK**

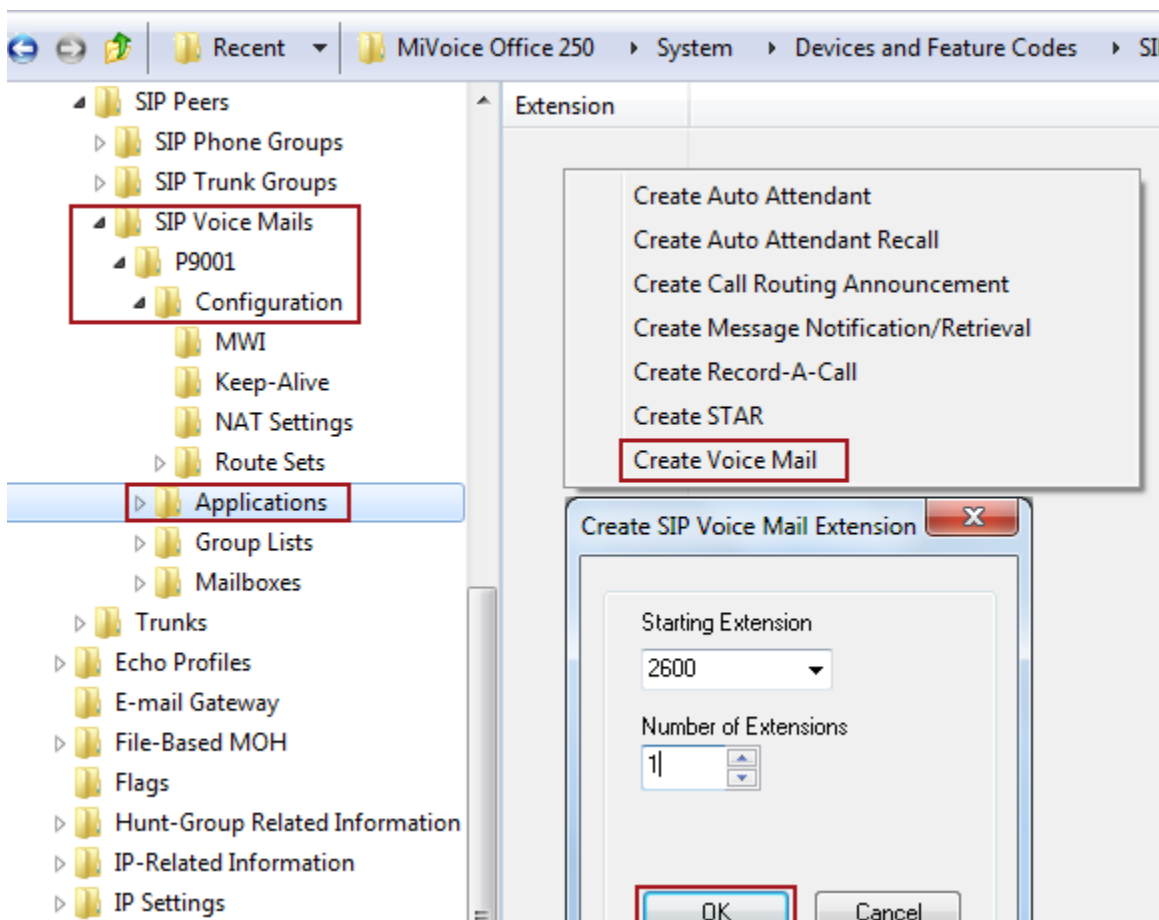


Figure 23: SIP Voice Mail Application

Navigation: **System -> Devices and Feature Codes -> SIP Peers -> SIP Voice Mails -> P9001 -> Applications -> 2600**

Set **SIP Voice Mail Pilot** to **2600** and leave all other fields as default

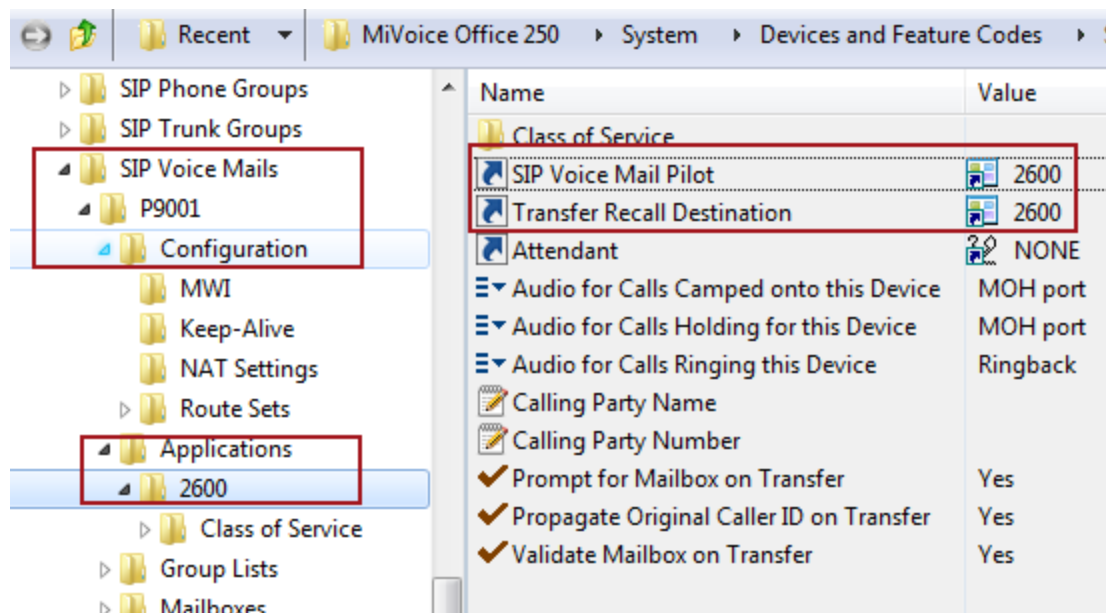


Figure 24: SIP Voice Mail Pilot

### SIP Voice Mail Mailbox (NuPoint)

Navigation: **System** -> **Devices and Feature Codes** -> **SIP Peers** -> **SIP Voice Mails** -> **P9001** -> **Mailboxes**

- Right-click the right pane, and select **Create Associated Mailboxes**
- Select **52xx/53xx** as **Type** in next pop-up window, then click **Next**
- Select desire extensions and click **Add Items**, then **Finish**

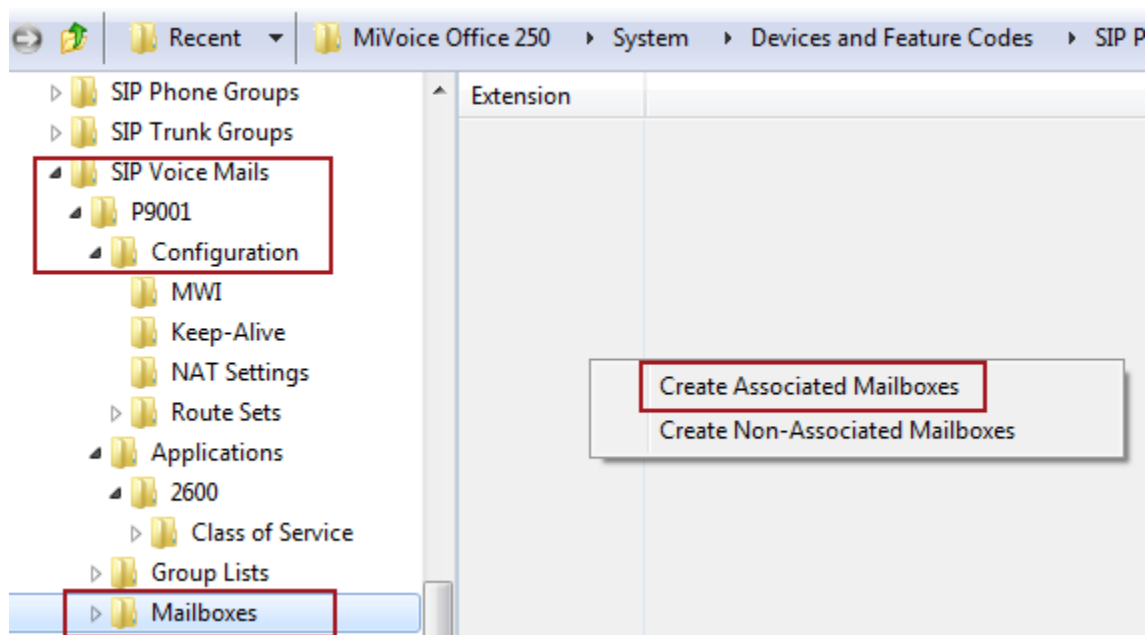


Figure 25: Create Associated Mailbox

The screenshot shows the 'SIP Peers' configuration page in the MiVoice Office 250 interface. The left sidebar shows a tree view with 'Mailboxes' selected. The main table lists SIP peers with their extensions, descriptions, usernames, and associated status.

Extension	Description	Username	Associated
1000			Yes
1001			Yes
1005			Yes
1006			Yes

Figure 26: Associated Mailboxes

## NuPoint UM on MiCollab Configuration Notes

This section provide detail steps to configure NuPoint UM on MiCollab.

- Click **NuPoint Web Console** under **Applications** in the navigation pane after log into **MiCollab server-manager**

The screenshot shows the MiCollab Server-Manager interface. The top header includes the Mitel logo, 'MiCollab', the user 'admin@micollab.tekvizionlabs.com', and an 'Alarm Status: Clear' indicator. The left sidebar shows the 'Applications' section with 'NuPoint Web Console' selected. The main content area displays 'Licensing Information' and a table of 'Application User Totals'.

**Licensing Information**

This page displays details about user licensing for your applications. "Currently used" totals display indicate that you have assigned some services for which you are not currently licensed. To purchase upgrade licenses, please contact your authorized Reseller.

Application	User Licenses	Currently used
Audio, Web and Video Conferencing	10000	0
Nupoint Unified Messaging	8	5
Teleworker	2	0
MiCollab Client	0	0
Console	1	0
Deskphone	3	0
Mobile	3	0
Softphone	3	0

**System Information:**

- MiCollab 6.0.205.0
- Mitel Standard Linux 10.1.39
- MiVoice Border Gateway 8.1.25.0
- OVA 6.0.205.0
- © Mitel Networks Corporation

Figure 27: MiCollab Server-Manager

- Navigate to **Offline Configuration > Edit Offline Configuration**

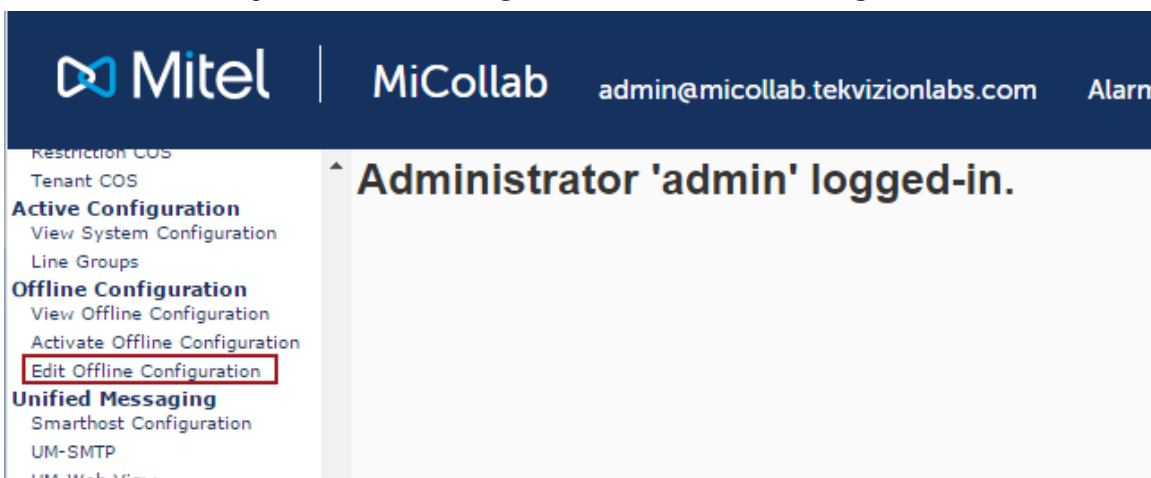


Figure 28: Offline Configuration

- Click **YES** to duplicate the active configuration to the offline configuration for editing purpose

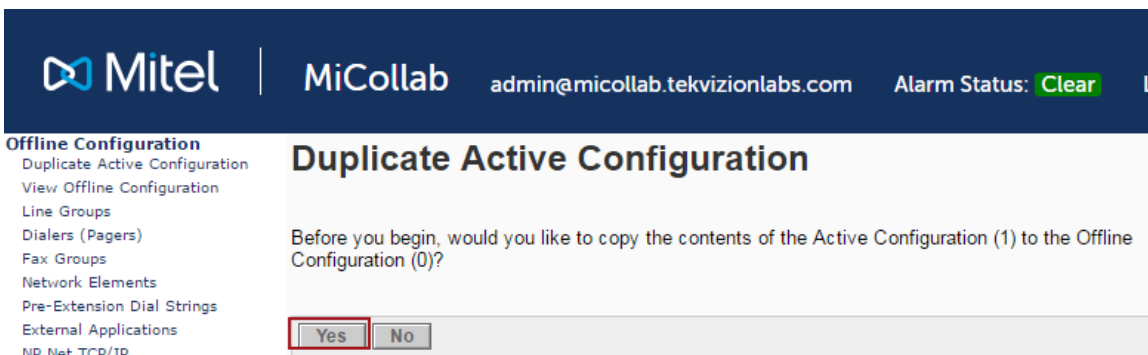


Figure 29: Duplicate Active Configuration

## Add SIP Gateway Network Element

- Navigate to **Offline Configuration > Network Elements**
- Click **Add**

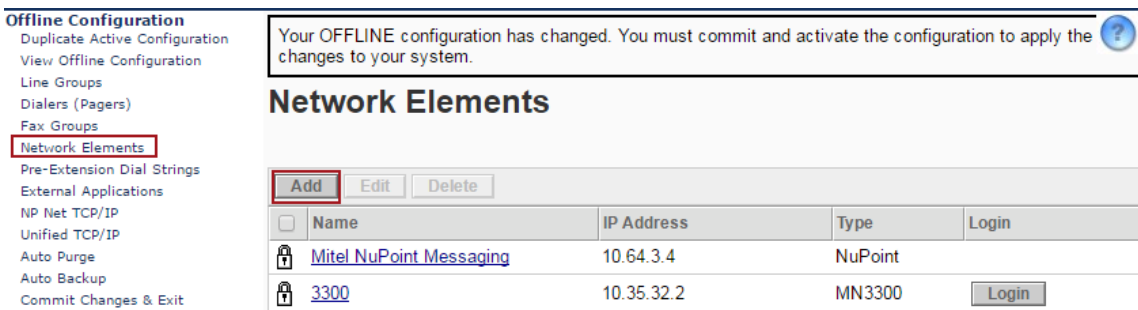


Figure 30: Network Elements

At Add Network Element Page

- Set **Type**: Select **SIP Gateway** from drop-down
- Set **Name**: **MiVoice Office** is given for this setup
- Set **IP Address**: This is the MiVoice Office 250 Base Server IP address (if your deployment with MiVoice Office 250 equipped with a Processing Server, then enter the IP address of Processing Server). **10.70.62.2** is given in this setup.
- Set **Number of Ports**: **4** is given here
- Click **Save**

## Add Network Element

\* Denotes Required Field

Network Element Information

\* Type: SIP GATEWAY ▼

Name: Mvoice\_Office

Domain Name:

\* IP Address: 10 . 70 . 62 . 2

Number of Ports: 4 ▼

Save Cancel

Figure 31: Add Network Element

## Add Voice Mail Line Group

- Navigate to **Offline Configuration > Line Groups**
- Click **Add**

Offline Configuration

Duplicate Active Configuration

View Offline Configuration

Line Groups

Dialers (Paggers)

Fax Groups

Network Elements

Pre-Extension Dial Strings

External Applications

Your OFFLINE configuration has changed. You must commit and activate the configuration to apply the changes to your system.

### Line Groups

Add Edit Delete

Number	Name	Number of Lines	User Interface	Application
--------	------	-----------------	----------------	-------------

Figure 32: Line Groups

- Set **Line Group Number**: Specify a number or click **Next Available**. 1 is given for this setup.
- Set **Name**: **MiVoice\_Office** is used here
- Set **Application**: **NuPoint Voice** is selected from drop-down
- Set **User Interface**: **Call Director** is selected from drop-down

## Add Line Group

Line Group Number:  \*

Name:  \*

Application:  ▼

User Interface:  ▼

Fax group connection:  ▼

Lines | Dialing Plan | Voicemail | Dial Strings

**Lines**

▼

<input type="checkbox"/>	Line Triplet	Device	Extension or Port
--------------------------	--------------	--------	-------------------

Figure 33: Add Line Group



- Under **Dialing Plan tab**, create a dialing plan based on site requirements

The screenshot shows a configuration window with four tabs: 'Lines', 'Dialing Plan', 'Voicemail', and 'Dial Strings'. The 'Dialing Plan' tab is active. Under the 'Dialing Plan' section, there are two radio buttons: 'Standard Mode' (selected) and 'Classic Mode'. Below 'Standard Mode' is a table titled 'Length of extensions starting with...'. The table has 9 rows, each with a number (1-9), a dropdown menu set to '3 digits', and another dropdown menu set to 'Standard'. Below 'Classic Mode' is a text field labeled 'Dialing Plan:' containing the value '3,3,3,3,3,3,3,3'. At the bottom of the window are 'Save' and 'Cancel' buttons.

Length of extensions starting with...		
1 :	3 digits	Standard
2 :	3 digits	Standard
3 :	3 digits	Standard
4 :	3 digits	Standard
5 :	3 digits	Standard
6 :	3 digits	Standard
7 :	3 digits	Standard
8 :	3 digits	Standard
9 :	3 digits	Standard

Dialing Plan: 3,3,3,3,3,3,3,3

Figure 34: Line Group - Dialing Plan

- Select the **Lines** tab, then click **Add**
- Set **Line Triplet**: Click **Net Available**, it will populate automatically. **1:0:6** is showed as this is the 1<sup>st</sup> Line Triplet configured in NuPoint Voice Mail.
- Set **Number of Lines**: This number should match the number configured in previous section [SIP Voice Mail Configuration](#). 1 is given in this setup.
- Set **PBX**: Select **MiVoice Office** programed in section [Network Element](#) from drop-down
- Set **Mapping**: **5** is set for this test as the starting mapping number
- Click **Add**

The screenshot shows a web-based configuration interface with tabs for 'Lines', 'Dialing Plan', 'Voicemail', and 'Dial Strings'. The 'Lines' tab is active. Below the tabs, there's a 'Lines' section with 'Add', 'Edit', and 'Delete' buttons. A modal dialog box is open for adding a new line triplet. It contains the following fields: 'Line Triplet' with the value '1:0:6' and a 'Next Available' button; 'Number of lines' with the value '1'; 'PBX' with a dropdown menu showing 'Mvoice\_Office'; and 'Mapping' with the value '5'. At the bottom of the dialog are 'Add' and 'Cancel' buttons. A red rectangle highlights the 'Line Triplet', 'Number of lines', 'PBX', and 'Mapping' fields.

Figure: 35 Add Line Triplet

- Enter pilot number in the field that matches the **Pilot Number** defined in MiVoice Office 250 [SIP Voice Mail Pilot](#) section, **2600** is given in this example
- Click **Save** to complete the Line Group configuration

Line Group Number: 3

Name:

Application:

User Interface:

Fax group connection:

Lines

Lines

Pilot Number:

<input type="checkbox"/>	Line Triplet	Device	Extension or Port
<input type="checkbox"/>	<a href="#">1:0:6</a>	Mvoice_Office	5

Figure 36: Add Line Group – Cont.

### Add Message Waiting Indicator (MWI) Line Group

- At **Line Groups** page, Click **Add**

## Line Groups

<input type="checkbox"/>	Number	Name	Number of Lines	User Interface	Application
--------------------------	--------	------	-----------------	----------------	-------------

Figure 37: Add MWI Line Group

- Set **Line Group Number**: It will automatically populate or you can set a number. **2** is given for this test
- Set **Name**: **MWI\_Office** is given for this test
- Set **Application**: Select **DTMF to PBX Dialer** from drop-down
- Set **User Interface**: Select **NuPoint Voice** from drop-down
- Set **Fax Group Connection**: Leave the default value **None**

## Add Line Group

Line Group Number: 4 \* Next Available

Name: MWI\_Office \*

Application: DTMF to PBX Dialer ▼

User Interface: NuPoint Voice ▼

Fax group connection: None ▼

Figure 38: Add MWI Line Group – Cont.

- Select the **DTMF to PBX Dialer** tab
- Set **Pre-DN On Dial String**: **1** is given here
- Set **Pre-DN Off Dial String**: **0** is given for the test
- Set **Initial Dialtone Detect**: **Checked**
- Set **Suppress Updates to MWI**: **Checked**
- Leave all other fields either empty or unchecked

Lines DTMF to PBX Dialer

DTMF to PBX Dialer

PBX Special Access Code:

Pre-DN On Dial String: 1

Pre-DN Off Dial String: 0

Post-DN On Dial String:

Post-DN Off Dial String:

Maximum PBX Message Count:

Options

☒ Initial Dialtone Detect

☐ Dial Tone Confirmation

☒ Suppress Updates to MWI

☐ Wait for Dial Tone

☐ Enable Alternate Code

☐ Use Same Port to Turn On/Off MWI

Save Cancel

Figure 39: DTMF to PBX Dialer

- Select the **Lines** tab
- Click **Add**
- Click **Next Available** to select **Line Triplet**
- Set **Number of Lines**: **1** is given for the test
- Set **PBX**: Select **MiVoice Office** from drop-down, this was configured in section [Network Element](#)
- Set **Mapping**: Set this to the next number according to the sequential mapping set for the line groups under same SIP Gateway. **5** is given in this example
- Click **Add**

The screenshot shows the 'Lines' configuration interface. At the top, there's a 'Pilot Number' field set to '2600'. Below it are 'Add', 'Edit', and 'Delete' buttons. A table lists line triplets with columns 'Line Triplet' and 'Device'. One triplet is listed: '1:0:6' with device 'Mvoice\_Office'. A 'Save' button is at the bottom left. A modal dialog is open over the table, showing details for the selected triplet: 'Line Triplet: 1:0:6', 'PBX: Mvoice\_Office' (from a dropdown), and 'Mapping: 5'. The dialog has 'Save' and 'Cancel' buttons at the bottom.

**Figure 40: Add MWI Line Triplet**

- Set **Pilot Number**: **2600** which was configured as Pilot Number in MiVoice Office 250 section [SIP Voice Mail Pilot](#) is given here
- Click **Save** to complete the configuration

Save Cancel

Line Group Number: 3

Name: MWI\_Office \*

Application: DTMF to PBX Dialer ▼

User Interface: NuPoint Voice ▼

Fax group connection: None ▼

Lines DTMF to PBX Dialer

Lines

Pilot Number: 2600

Add Edit Delete ▼

<input type="checkbox"/>	Line Triplet	Device	Ext
<input checked="" type="checkbox"/>	1:0:6	Mvoice_Office	5

Save Cancel

Figure 41: Add MWI Line Group – Cont.

### Activate Offline Configuration

- Navigate to **Offline Configuration > Commit Change & Exit**
- Click **Commit** at **Commit Offline Changes** page

**Offline Configuration**

- Duplicate Active Configuration
- View Offline Configuration
- Line Groups
- Dialers (Paggers)
- Fax Groups
- Network Elements
- Pre-Extension Dial Strings
- External Applications
- NP Net TCP/IP
- Unified TCP/IP
- Auto Purge
- Auto Backup
- Commit Changes & Exit**
- Discard Changes & Exit

**Server Manager**

- Return to Server Manager

Your OFFLINE configuration has changed. You must commit and activate the config changes to your system.

## Commit Offline Changes

Do you wish to commit the changes you have made to the Offline Configuration (0):

Commit Continue Editing

Figure 42: Commit Changes

**Figure 43: Commit Changes – Cont.**

- Click **Activate** link
- Uncheck **Wait for MWI/pager queue to be empty**
- Click **Activate**

Your OFFLINE configuration has changed. You must [activate](#) the configuration to apply the changes to your system.

## Activate Offline Configuration

Changes have been made to the Offline Configuration (0).  
Press Activate to apply this configuration to your system.

☐ Wait for MWI queue to be empty.  
☐ Wait for pager queue to be empty.

The following users are currently logged in and will possibly lose changes if you activate now.

ID	Name	Login Time
admin		May 23, 4:49 PM

[Activate](#)

**Figure 44: Activate the Configuration**

- Click **OK** at pop-up window to confirm

10.64.3.4 says:

Do you want to activate the system configuration?

This will cause the system to be restarted.  
The system will be inaccessible for a few moments.

☐ Prevent this page from creating additional dialogs.

[OK](#) [Cancel](#)

**Figure 45: Activate the Configuration – Cont.**

- Click **OK** at Activation complete page

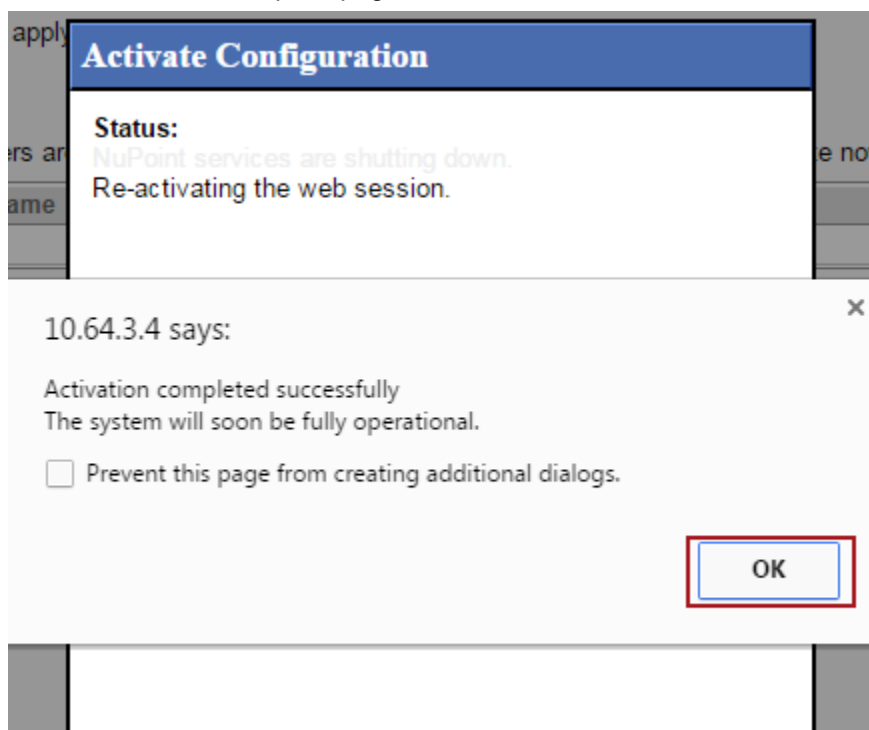


Figure 46: Activate the Configuration – Cont.

## Add Mailbox

Navigation: **Mailbox Maintenance** -> **Mailboxes**

- Click **Add**

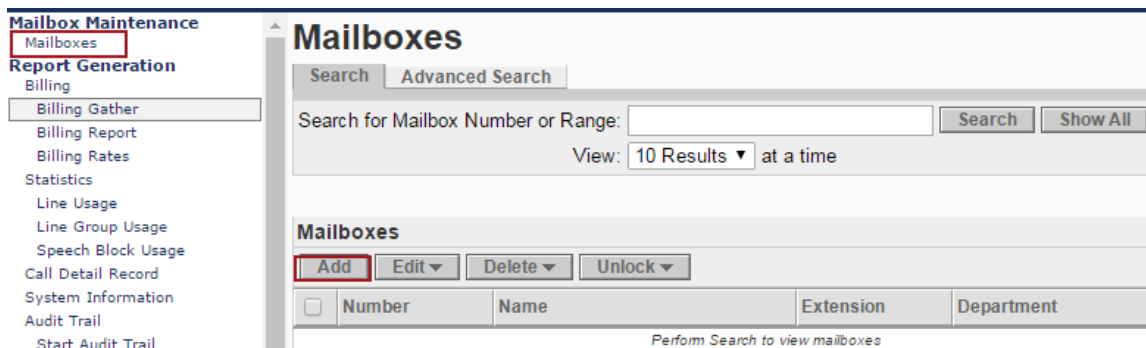


Figure 47: Add Mailbox

- Set **Mailbox Number**: **1006** is given in this example
- Set **Name**: **IntelePeer** is given in this setup
- Set **Passcode**: input proper passcode for the mailbox
- Set **Extension**: input associated MiVoive Office 250 Extension, **1006** is used here



**Create Mailbox(es)**

Mailbox Number(s):

Copy from another mailbox:

**Basic** [Advanced](#)

**General** **Class of Service** **Message Waiting**

**Personal Information**

Name:

**IMPORTANT NOTE**  
 If you expect your callers to use "Dial By Name" with First Name then:  
 Enter the name in following format: <First Name > <Last Name >. For example: John Smith  
 If you expect your callers to use "Dial By Name" with Last Name then:  
 Enter the name in following format: <Last Name >, <First Name >. For example: Smith, John  
 Note that the comma is ESSENTIAL in this case.

Passcode:   
*The user will be asked to change the passcode on the next TUI login*

Extension:

Attendant Extension:

**Unified Messaging Information**

UM Audio Encoding:

UM-SMTP Email Address:

UM-Web View Email Address:

**Basic** [Advanced](#)

Figure 48: Add Mailbox – Cont.

- Click **Message Waiting** tab
- Set **Message Waiting #1 Type: DTMF to PBX** is selected from drop-down
- Leave all other fields as default
- Click **Save**

## Add Mailbox(es)

### Create Mailbox(es)

Mailbox Number(s): 1006

Copy from another mailbox:

**Basic**[Advanced](#)

#### Message Waiting #1

Type:

►Details

#### Message Waiting #2

Type:

►Details

#### Message Waiting #3

Type:

**Basic**[Advanced](#)

Figure 49: Message Waiting

## MiVoice Border Gateway Configuration Notes

When configuring MiVoice Border Gateway (MIVOICE BORDER GATEWAY), you need to specify the Network profile, gateway mode used in this setup

Navigate to: **Applications > MiVoice Border Gateway > System Configuration > Network Profiles**

Click the “→” beside **Server-gateway configuration on the network edge**

Click **Apply**

The screenshot displays the 'System configuration' tab in the MiVoice Border Gateway configuration interface. The 'Network profile (Gateway mode)' section is active, showing a list of configuration options on the left and a detailed description on the right. The 'Server-gateway configuration on the network edge' option is selected, indicated by a blue arrow button. Below this, a text block explains that for a server on the network edge, streaming addresses will likely be the same as those configured on the corresponding interfaces. It also notes that override addresses should not be used unless the server is behind NAT. An 'Apply S/G configuration' button is present, with an 'Apply' sub-button highlighted by a red box. Below the selected option, there are three more options: 'Server-only configuration on the network DMZ', 'Server-only configuration on the network LAN', and 'Custom configuration', each with a grey arrow button.

System status ▾ Service configuration ▾ **System configuration ▾** Administration ▾

Page updated: Tue May 24 2016 08:53:41 GMT-0500 (Central Daylight Time)  
Configure this server in...

**Network profile (Gateway mode)**

**Server-gateway configuration on the network edge** →

For a server on the network edge, the streaming addresses will most likely be the same as those configured on the corresponding interfaces.

You should not have to use the override addresses, unless the server is behind NAT for some reason. If you click on the "Apply" button below, I will set the streaming addresses appropriately.

**Apply S/G configuration** **Apply**

Server-only configuration on the network DMZ →


Server-only configuration on the network LAN →

Custom configuration →

Figure 50: Network Profiles




Then identify the working MiVoice Office ICP where to forward SIP messages to and then to configure the SIP trunk.

Navigate to **MiVoice Border Gateway > Service Configuration > ICPs**

System status ▾ **Service configuration ▾** System configuration ▾ Administration ▾ 

Page updated: Tue May 24 2016 08:55:05 GMT-0500 (Central Daylight Time)  
To test connectivity to your configured ICPs, or to run a DNS resolution test on configured hostnames, see the [Diagnostics](#) page.

**ICP Information**

Default for MiNet	Default for SIP	Name	Hostname or IP address	Type	Installer password	SIP capabilities	Indirect call recording capable		
<input checked="" type="radio"/>	<input checked="" type="radio"/>	MiVoice Office10.70.62.2	10.70.62.2	MiVoice Office 250		UDP			

**Figure 51: MIVOICE BORDER GATEWAY Configuration**

On **ICPs** page, ensure that the “working” MiVoice Office is configured. If needed, click **Add ICP** link and add a new Mitel switch.

Click **Update** Default ICPs

To add a new SIP trunk:

- Click **Service Configuration** tab and then click **SIP trunking**
- Click **Add a SIP trunk** link

System status ▾ **Service configuration ▾** System configuration ▾ Administration ▾


Page updated: Tue May 24 2016 08:55:05 GMT-0500 (Central Daylight Time)  
The SIP trunks Information section displays the configuration of each SIP trunk. Click on the SIP trunk for detailed information.

To make changes to SIP settings, click on the [SIP settings](#) link in System configuration.

To test DNS resolution on any configured hostnames, click on the [Diagnostics](#) page.

**SIP trunking**

**SIP trunk information**



**Figure 52: SIP Trunking Configuration**

Enter the SIP trunk details as follows:

Set **Name**: IntelPeer is given in this setup

Set **Remote Trunk Endpoint Address**: Enter the IP address / FQDN for your deployment

Set **Remote Trunk Endpoint Port**: 5060 is used

Set **Remote RTP Framesize (ms)**: This is the Packetization rate you want to set on this trunk. Set to Auto.

Set **PRACK Support**: Disabled for this configuration.

Set **Routing rules**: This allows routing of calls with certain range of dialed digits to the selected MiVoice Office ICP

The remaining settings are optional and could be configured as required

Click **Save**

Manage SIP trunk

**Name**: IntelPeer

**Remote trunk endpoint port**: 5060

**Options keepalives**: Never

**Rewrite host in PAI**: ☒

**Idle timeout (s)**: 3600

**Local streaming**: ☐

**Log verbosity**: Use master setting

**Authentication password**:

**Set-side RTP security**: Allow

**Search routing rules**:

**Remote trunk endpoint address**:

**Accept traffic from any port**: ☐

**Options interval**: 60

**Remote RTP framesize (ms)**: 20ms

**RTP address override**: ---

**PRACK support**: Disabled

**Authentication username**:

**Confirm authentication password**:

**Icp-side RTP security**: Disable

**Next** **Previous**

Note, if you modify your routing rules, you must save them before changing pages or navigating elsewhere, or those changes will be lost.

**Page**: 1 of 1 **Jump to page**: 1

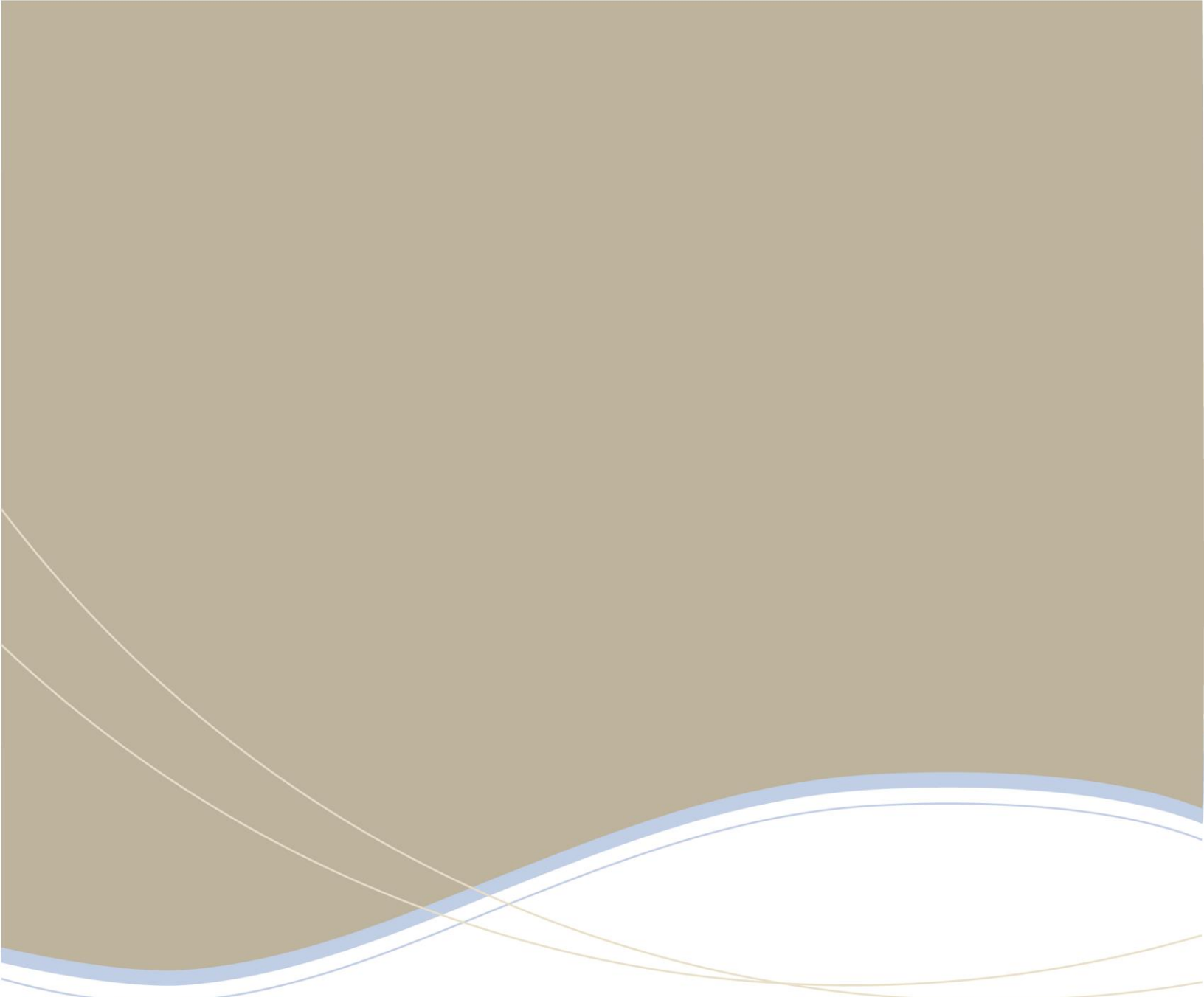
**Rules per page**: 10

**First** **Prev** **Next** **Last**

Match	Rule	Primary	Secondary	
1 Request URI	91751222XX	MiVoice Office10.70.62.2	.....	<a href="#">Raise</a> <a href="#">Prepend</a> <a href="#">Delete</a> <a href="#">Lower</a> <a href="#">Append</a>

**Save**

Figure 53: SIP Trunk Configuration Settings



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