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

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

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Overview

This document provides a reference to Mitel Authorized Solutions providers for configuring the MiVoice Office 250 to connect to IntelePeer 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 IntelePeer SIP Trunking

Interop Status

The Interop of IntelePeer 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 IntelePeer SIP Trunking achieved is:

COMPATIBLE	The most common certification which means IntelePeer 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.

Software & Hardware Setup

This was the test setup to generate a basic SIP call between IntelePeer 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	IntelePeer	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 Pans (08-4940-00034) for detailed test cases.

Feature	Feature Description	Issues
Basic Call	Making and receiving a call through IntelePeer 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 IntelePeer 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 IntelePeer 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 IntelePeer SIP Trunking is connected to the MiVoice Office.

Feature	Problem Description
Video Call	IntelePeer does not support video calls **Recommendation**: Contact IntelePeer for update on this feature
Packetization	IntelePeer supports 20MS packetization only **Recommendation**: 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

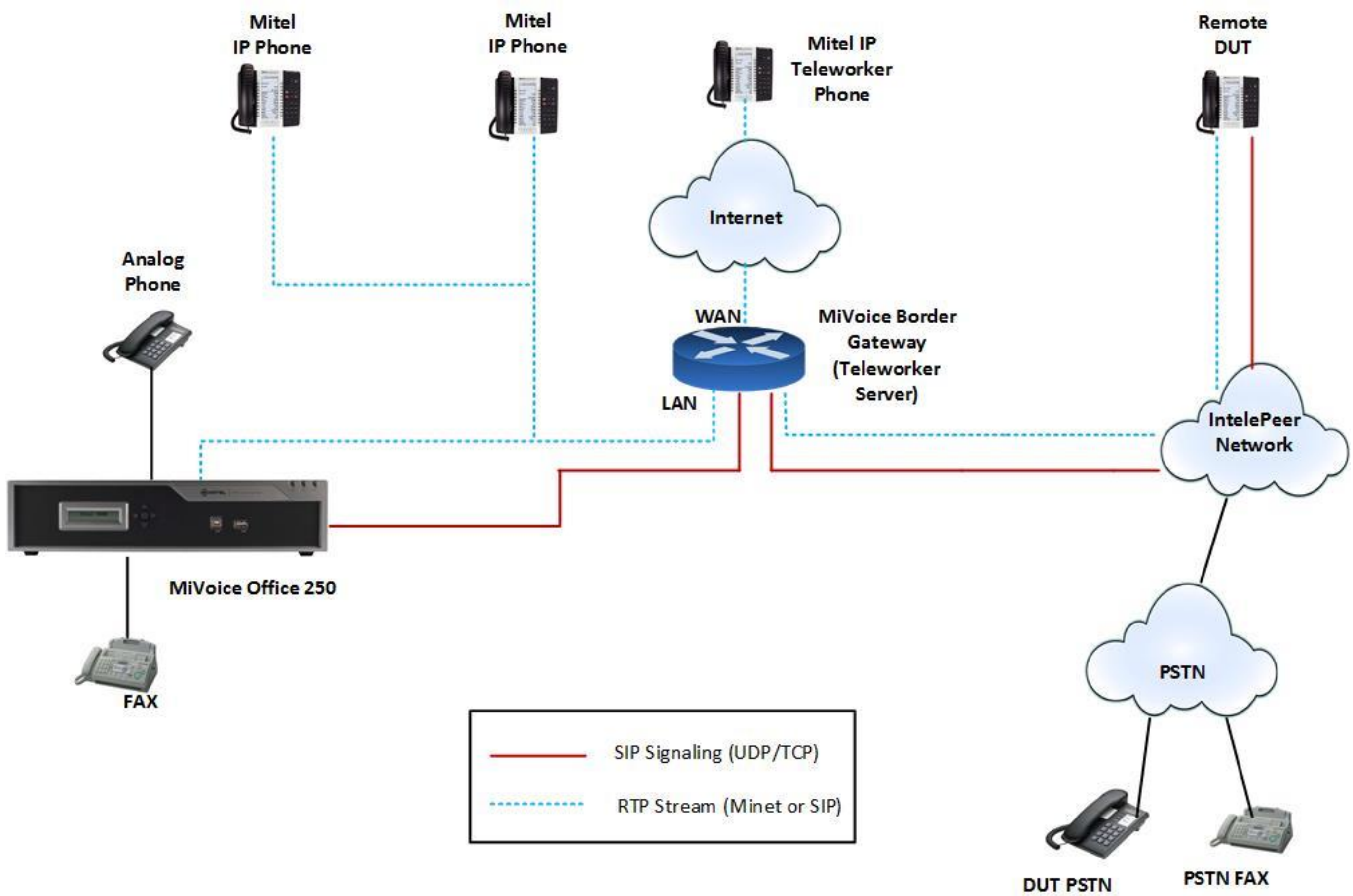

Figure 1: Network Topology

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 IntelePeer SIP Trunking was configured in our test environment.

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.

MiVoice Office 250 Configuration Notes

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

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 IntelePeer. This can be verified under the **Software License** form.

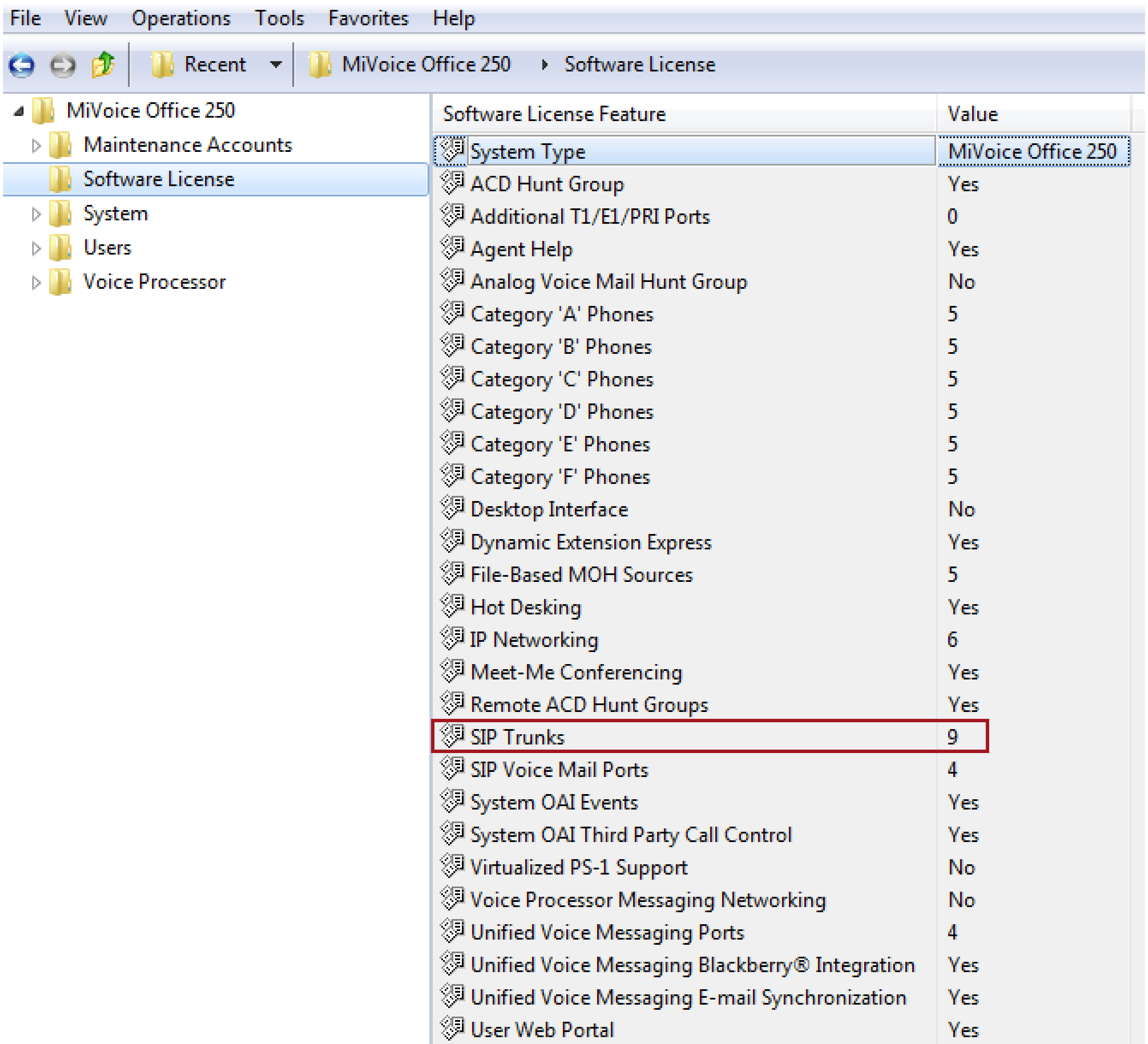

Figure 2: License Selection

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 IntelePeer, 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**.

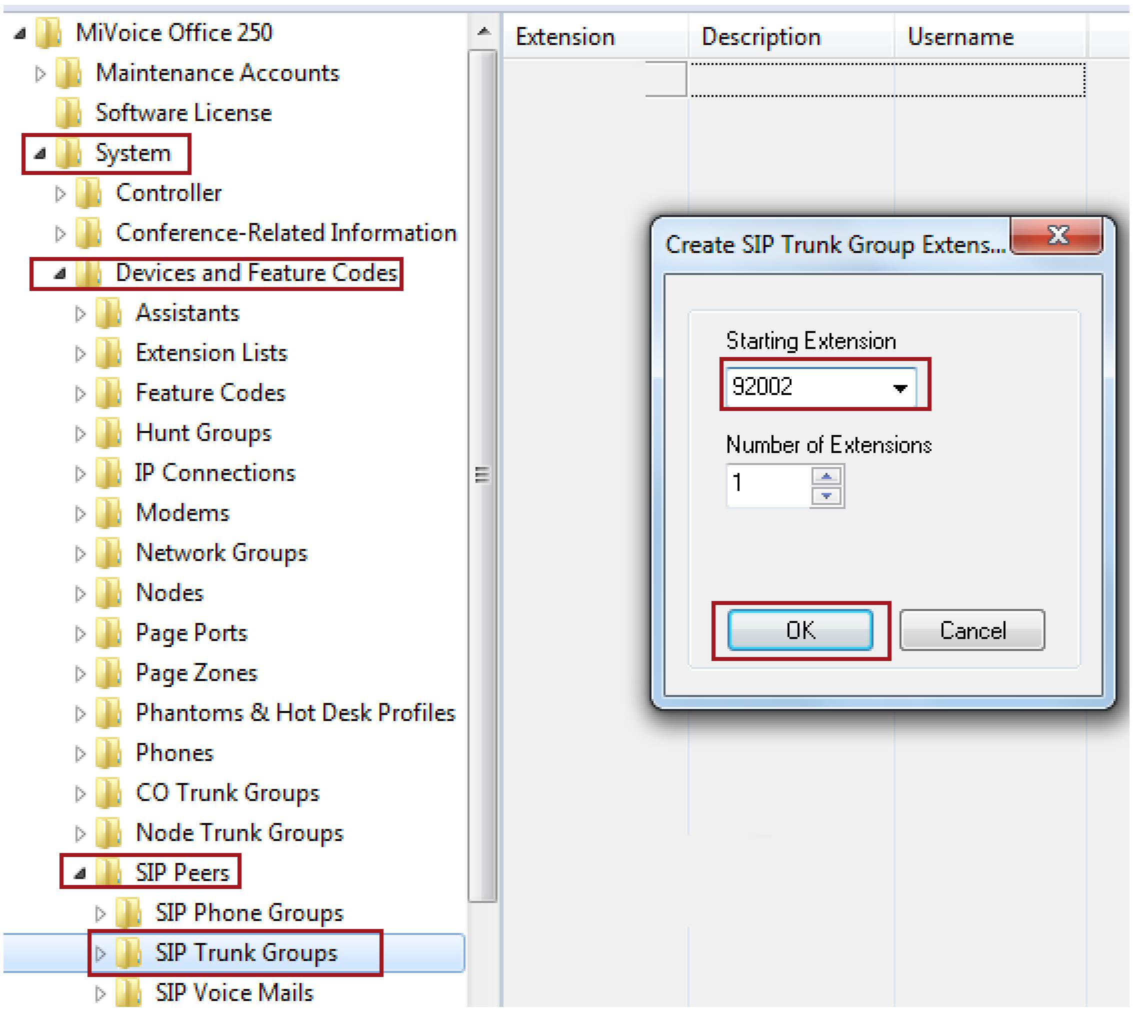

Figure 3: Create SIP Trunk Group

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

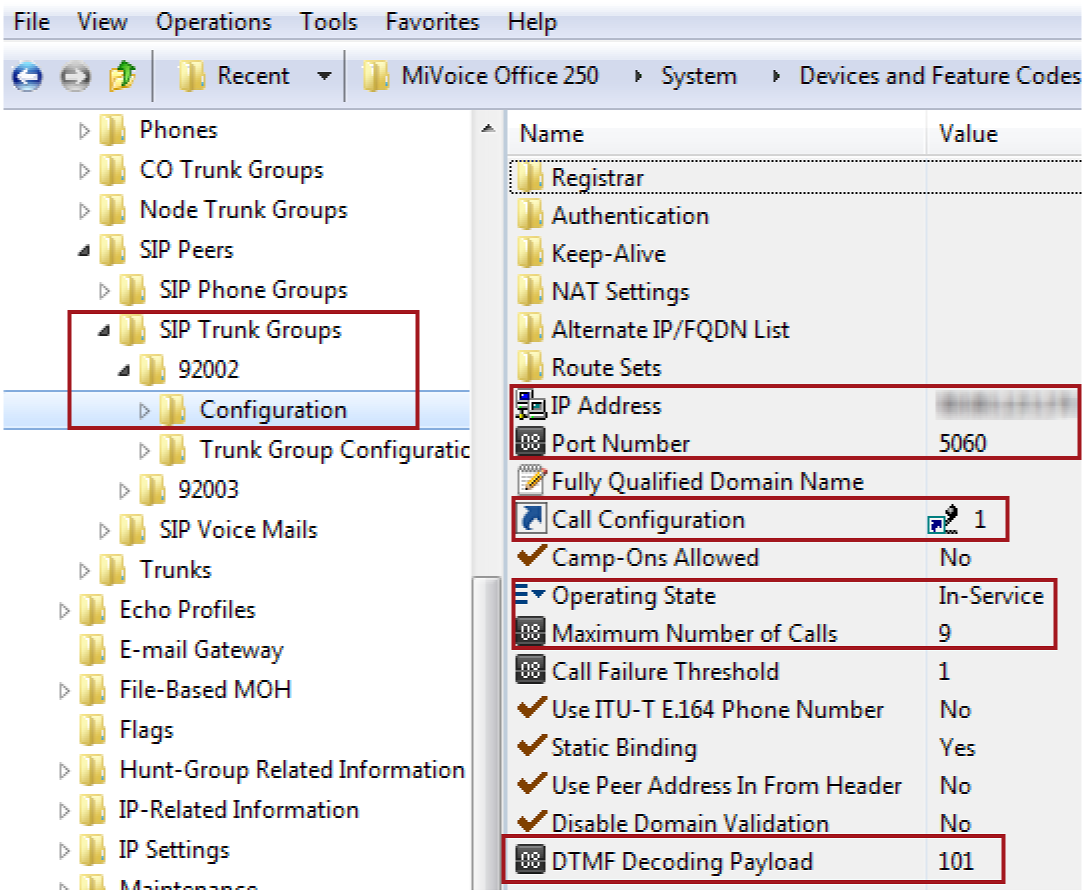

Figure 4: SIP Trunk Group for IntelePeer

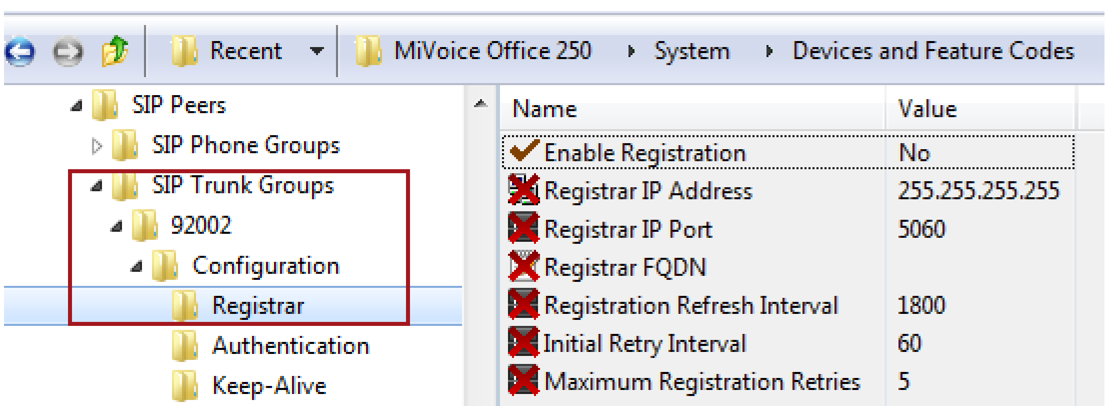

Figure 5: Registration not required for IntelePeer

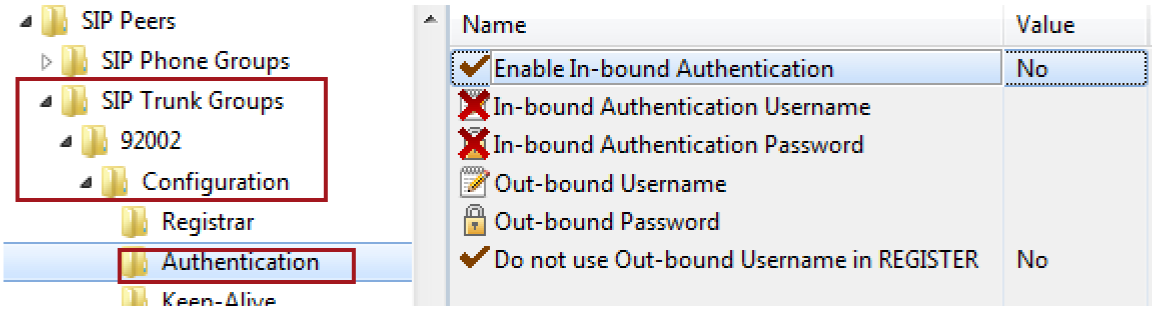

Figure 6: Authentication Not Required for IntelePeer

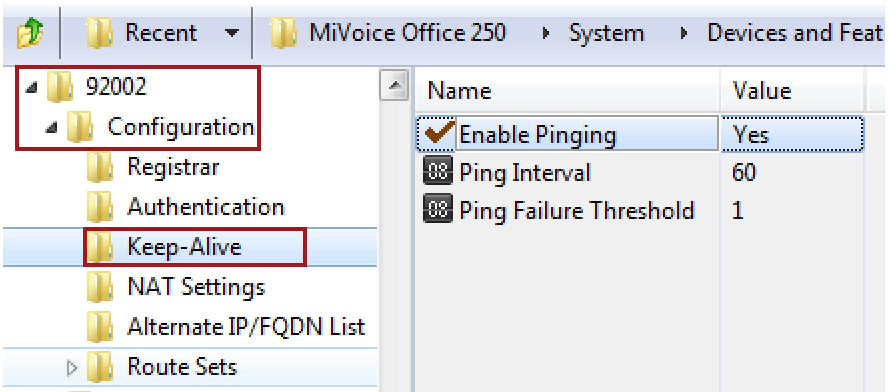

Figure 7: IntelePeer SIP Trunk Group - Keep-Alive

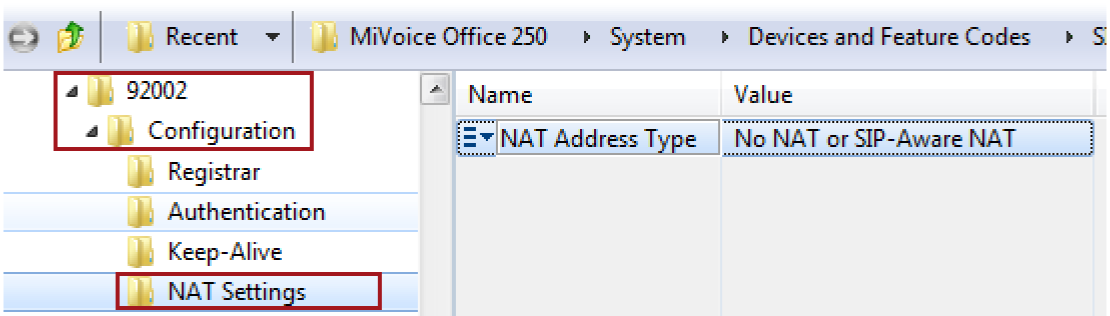

Figure 8: IntelePeer SIP Trunk Group: NAT Setting

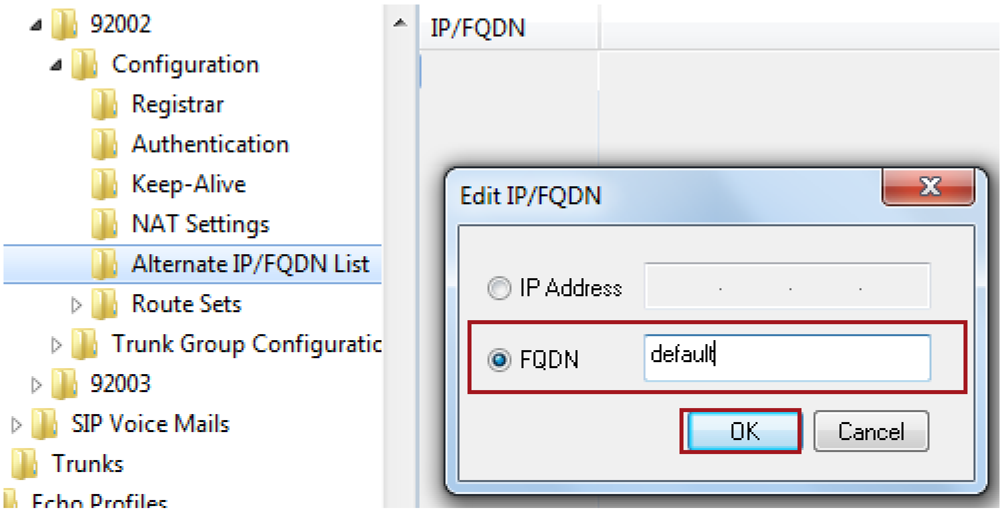

Figure 9: IntelePeer SIP Trunk Group: Alternate IP/FQDN

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)

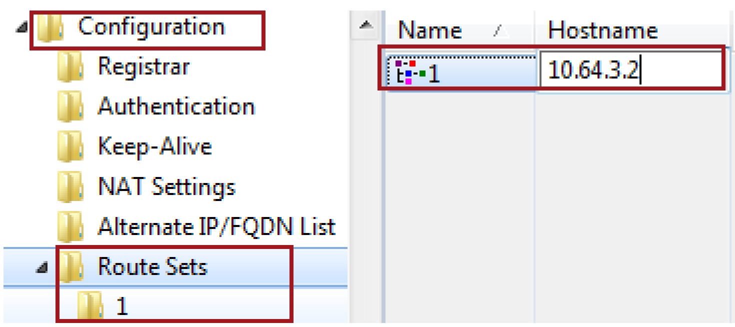

Figure 10: IntelePeer SIP Trunk Group - Route Set

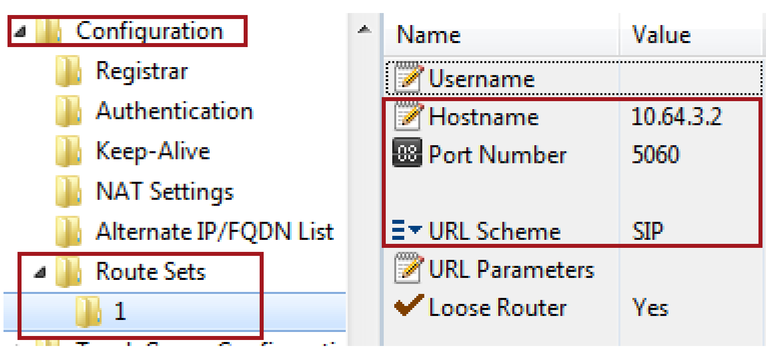

Figure 11: IntelePeer SIP Trunk Group - Route Set – Cont.

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

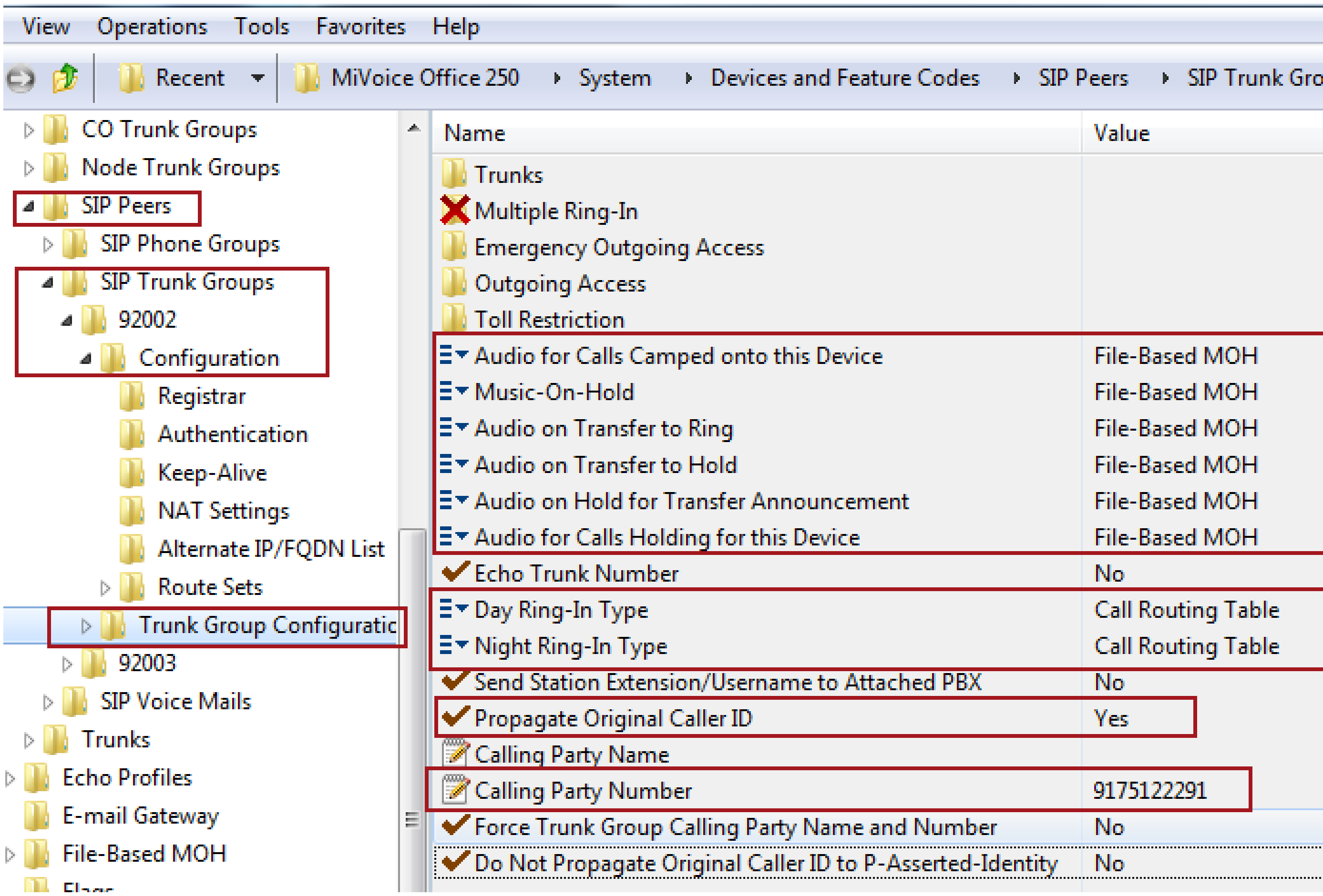

Figure 12: IntelePeer Trunk Group Configuration

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 the 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**

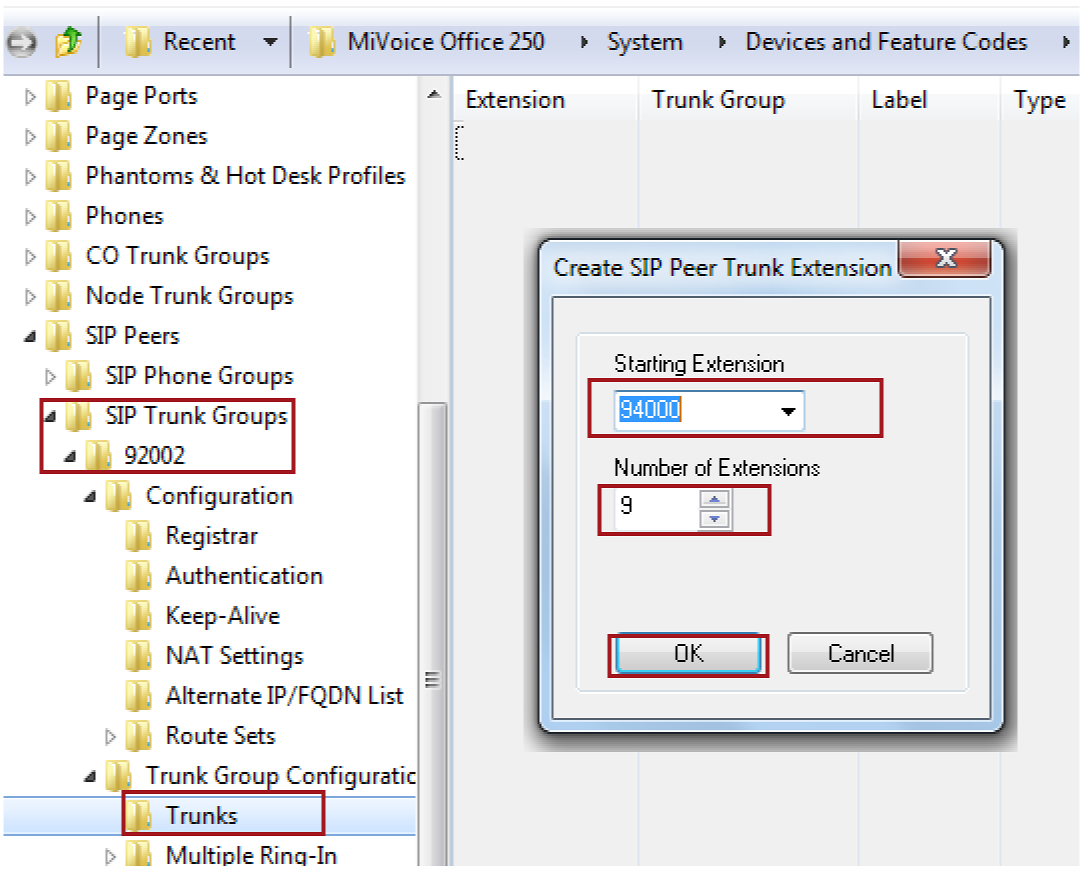

Figure 13: Create SIP Trunks

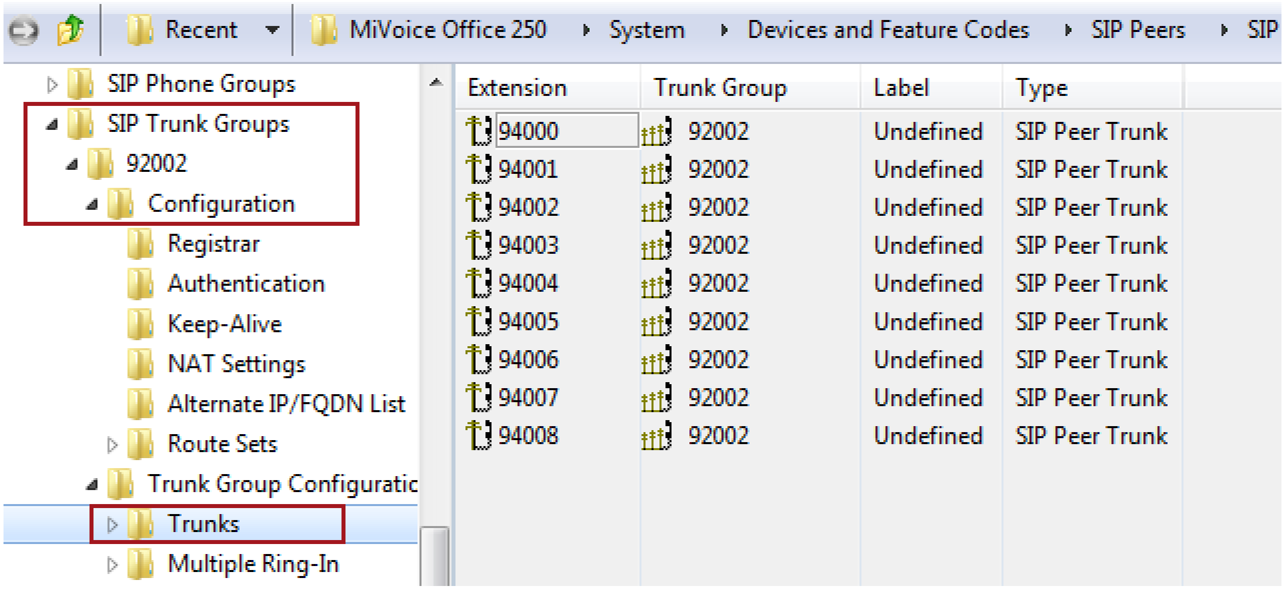

Figure 14: SIP Trunks – Cont.

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.

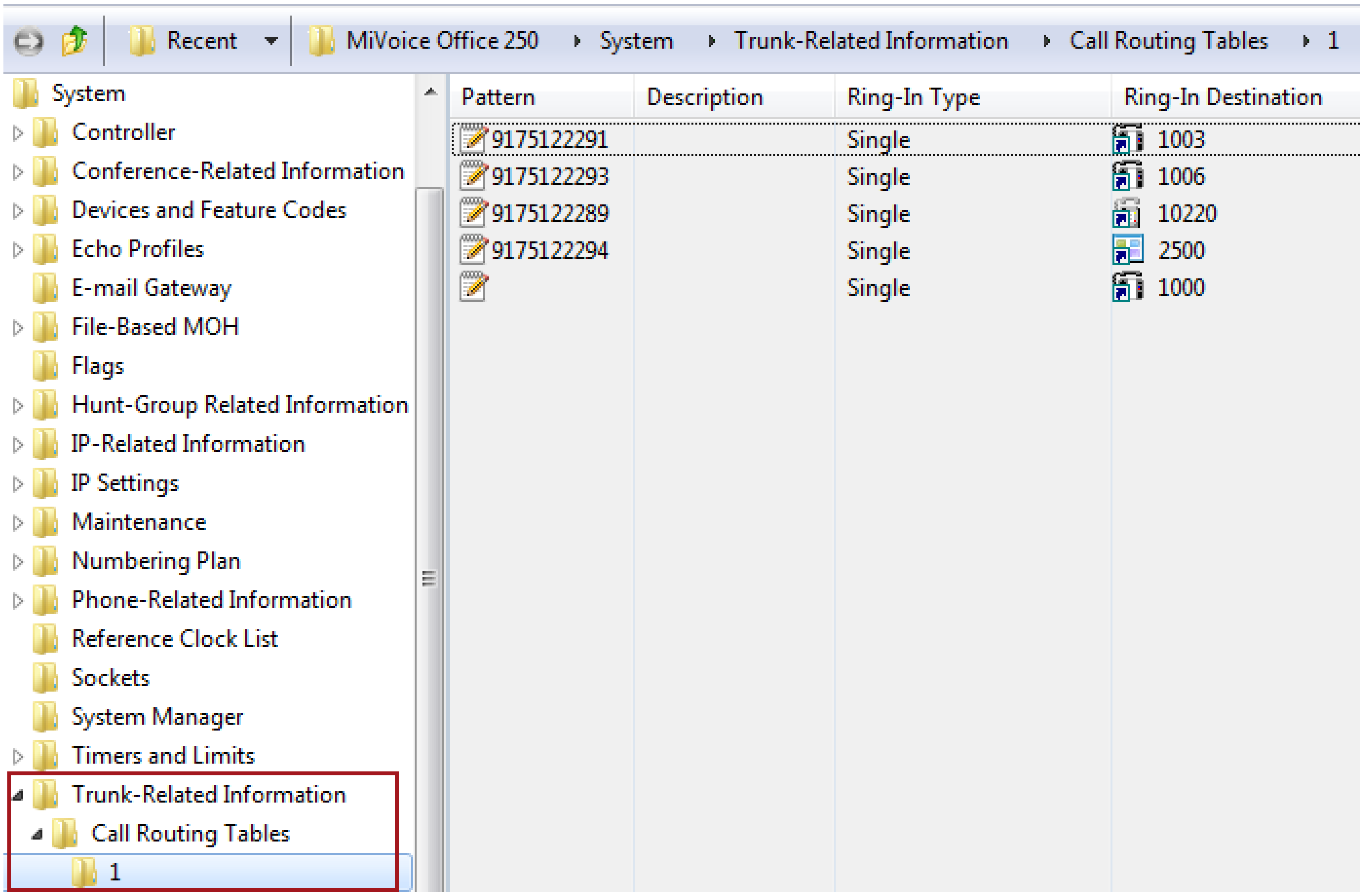

Figure 15: Call Routing Table

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 IntelePeer supports G711 Codecs only
- **Fax Encoding Setting**: IntelePeer 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

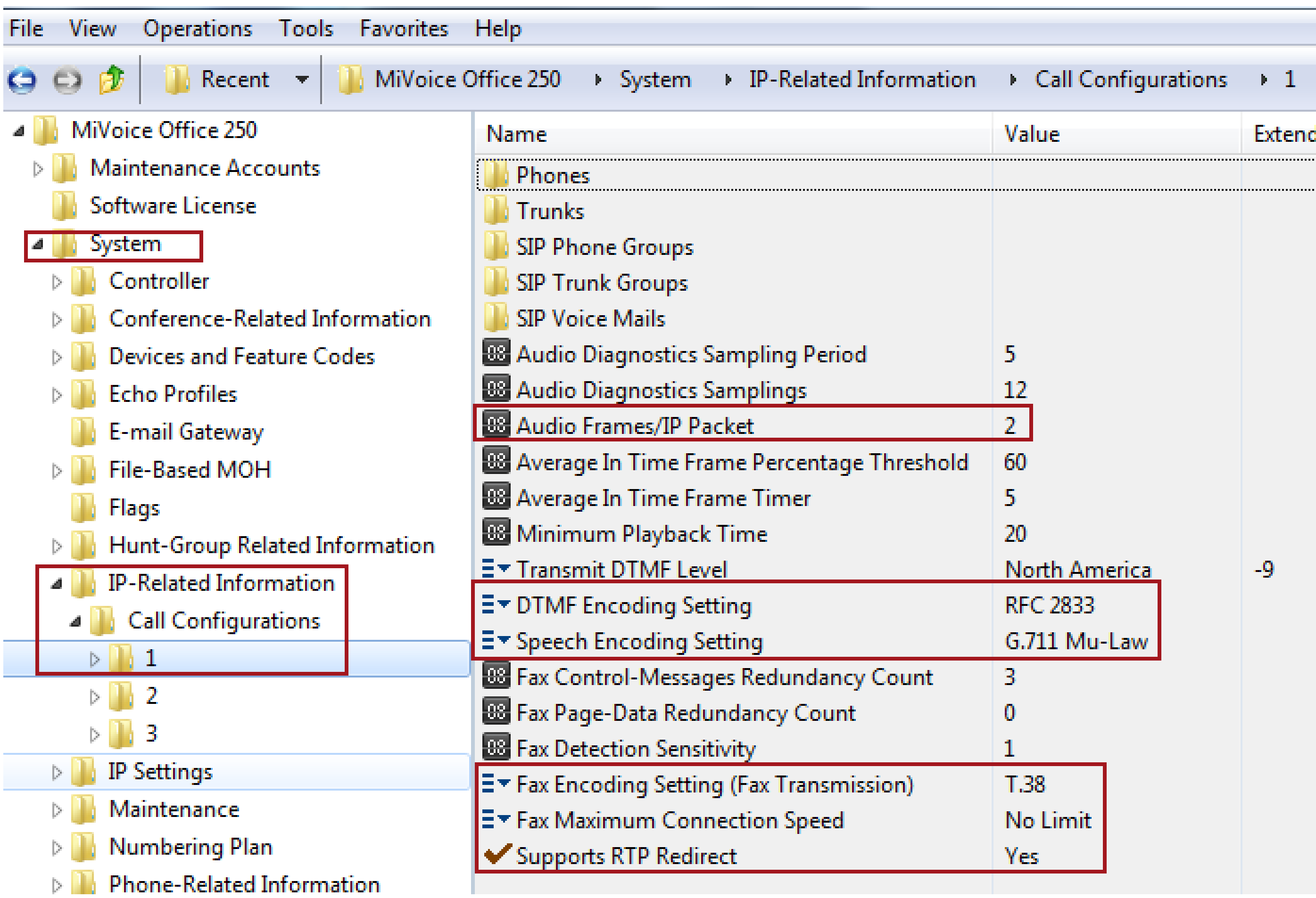

Figure 16: Call Configuration

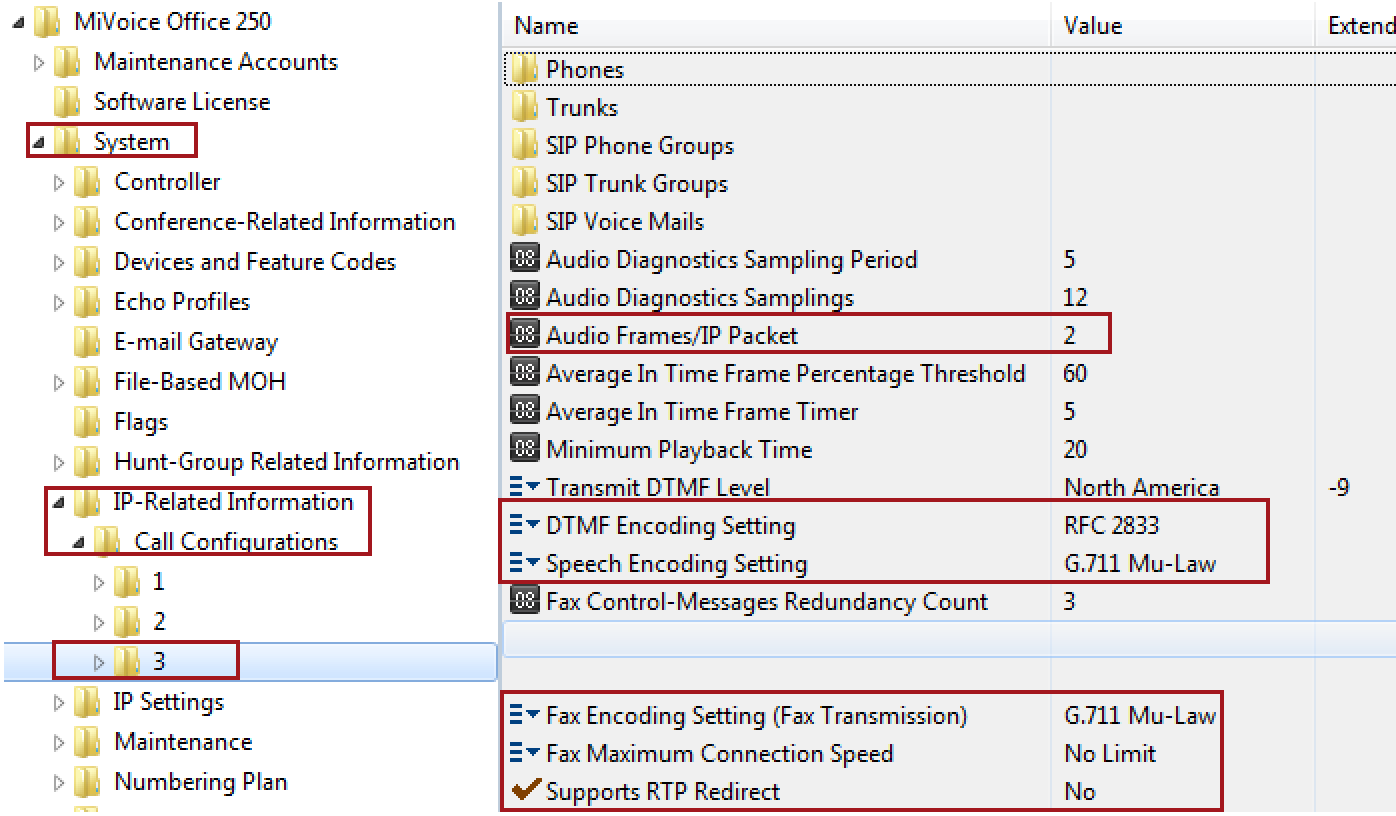

Figure 17: Call Configuration for NuPoint Voice mail

Call Routing

By default, the Feature Code for Outgoing Calls in MiVoice Office 250 is set to “8”. User can dial ‘8” then follow by the 10/11 digits (i.e.:8 1 214242XXXX) for outgoing calls. The User may also dial by the SIP trunk Group’s extension (i.e.:92002 1 214242XXXX) or dial by each SIP trunk extension (i.e.: 94030 1 214242XXXX).

Navigation: **System** ->**Device and Feature Codes** ->**Feature Codes**

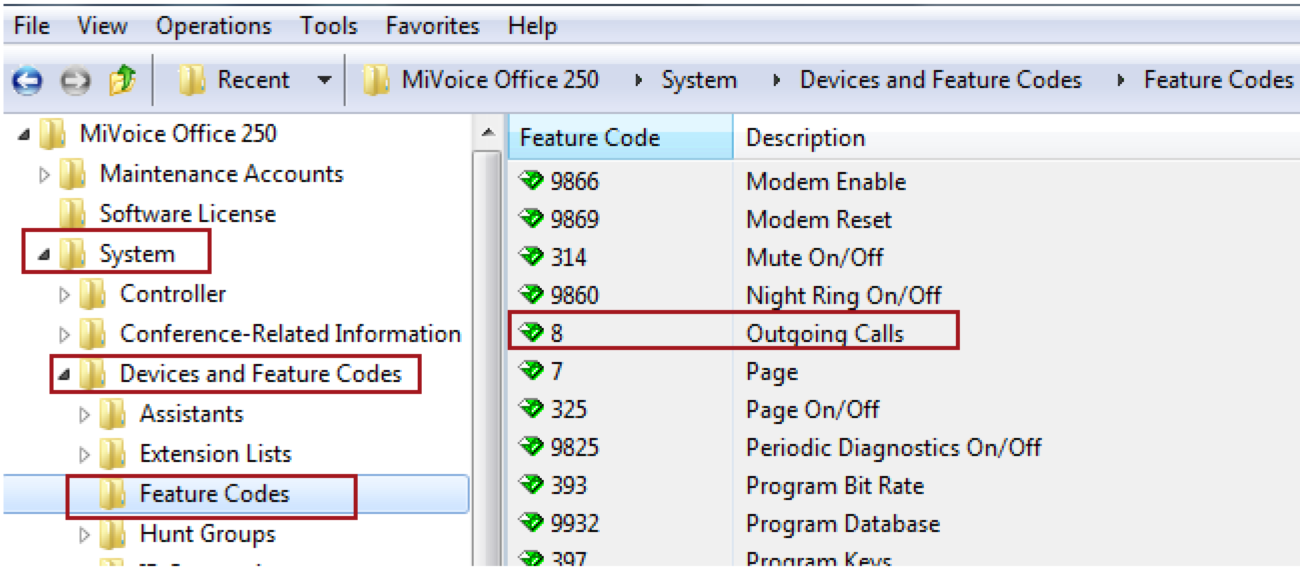

Figure 18: Feature Codes

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**

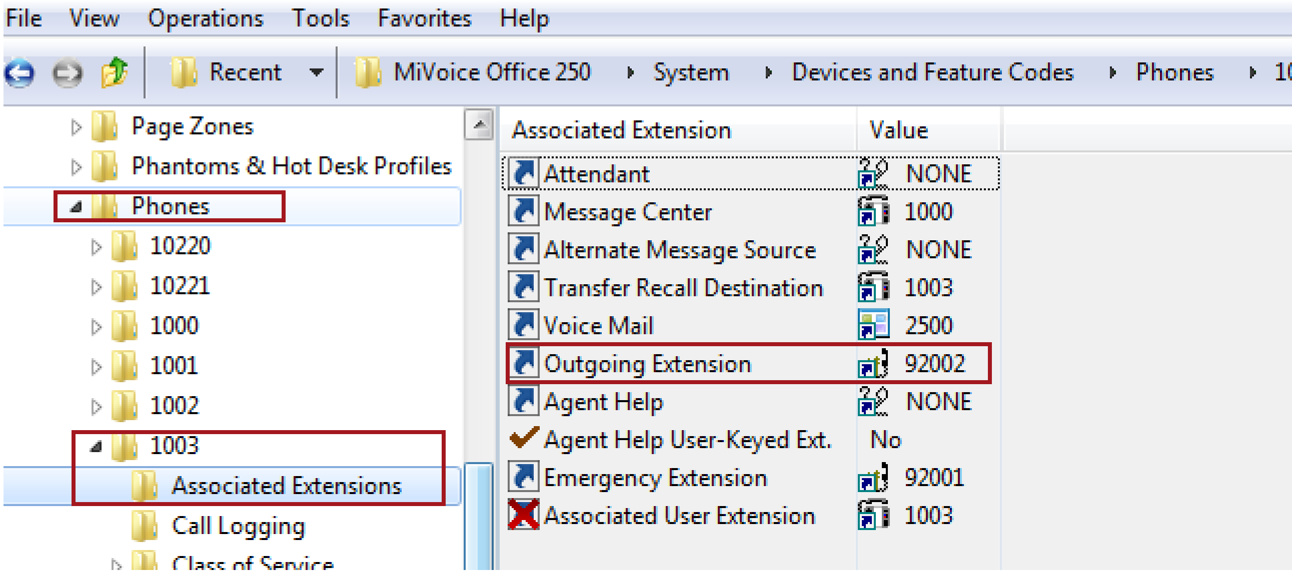

Figure 19: Associated Phone Extensions

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**

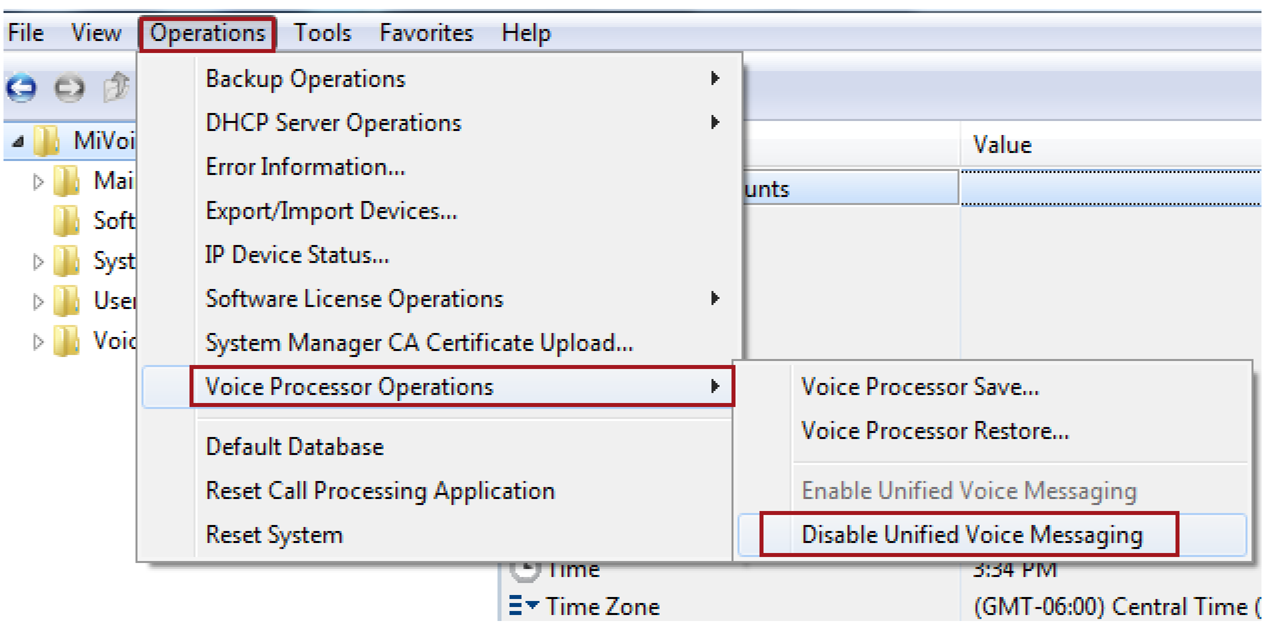

Figure 20: Disable Basic Voice Mail

Create SIP Voice Mail

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

- First, right-click the right pane, and the 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**

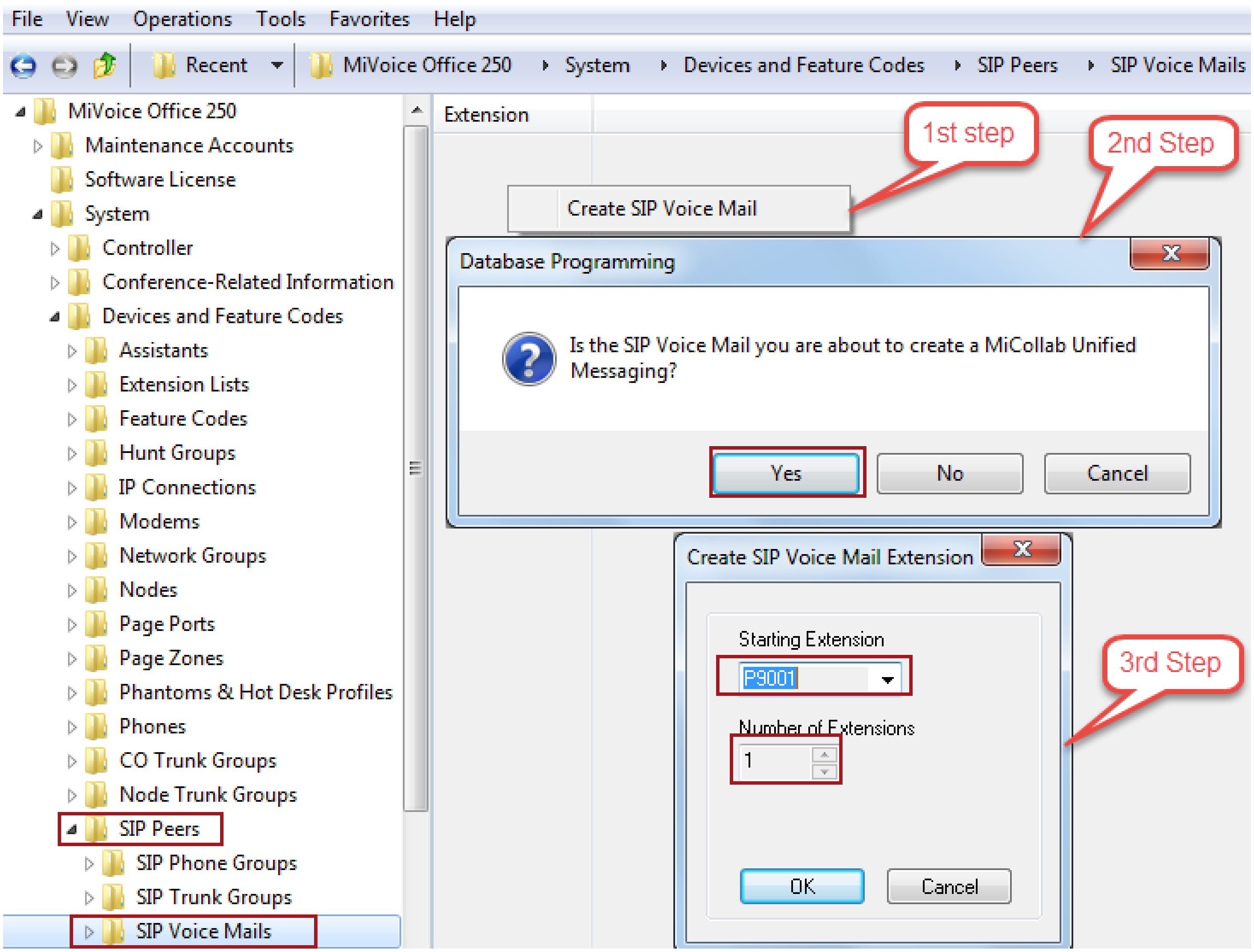

Figure 21: Create SIP Voice Mail

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 IntelePeer DTMF payload
- Leave all other fields as default

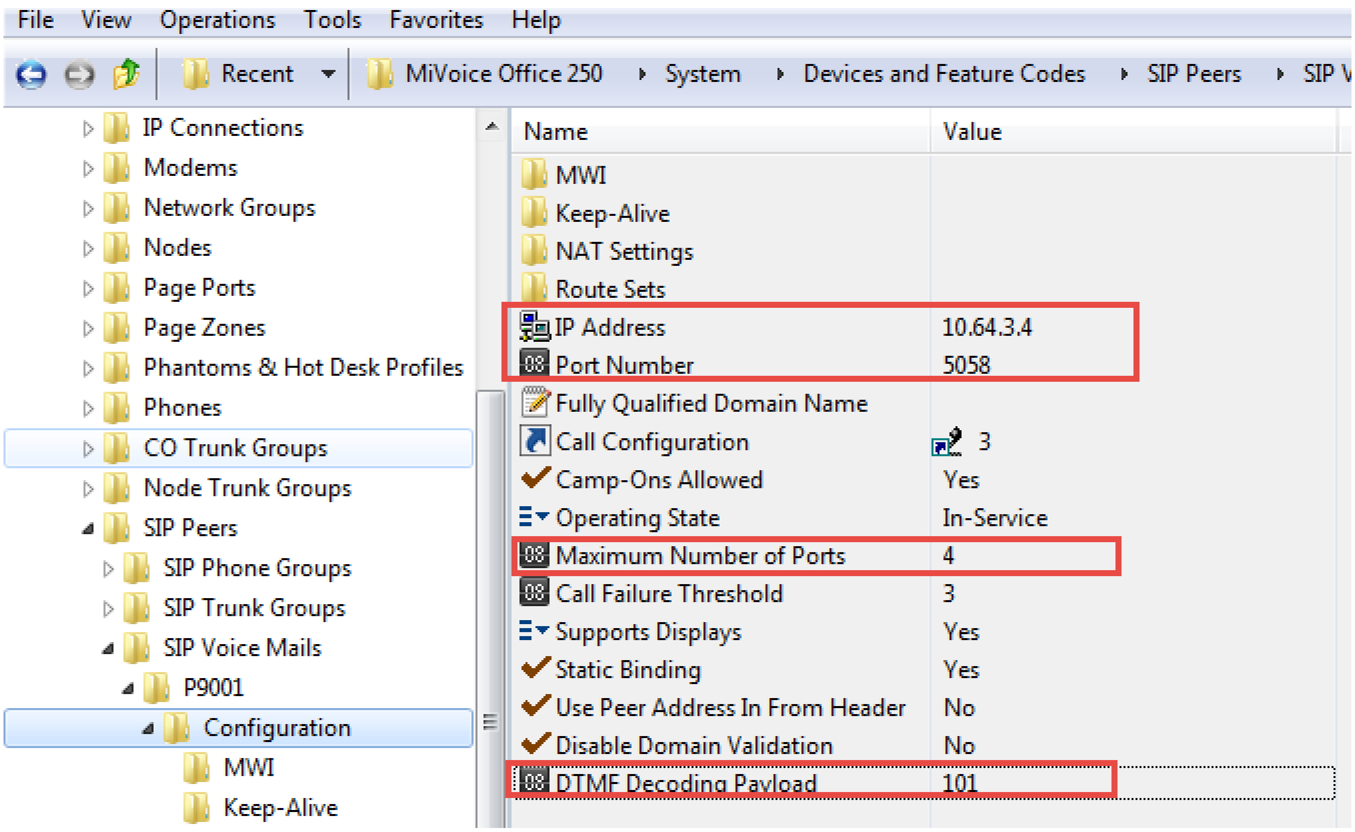

Figure 22: SIP Voice Mail Configuration

SIP Voice Mail Pilot (NuPoint)

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

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

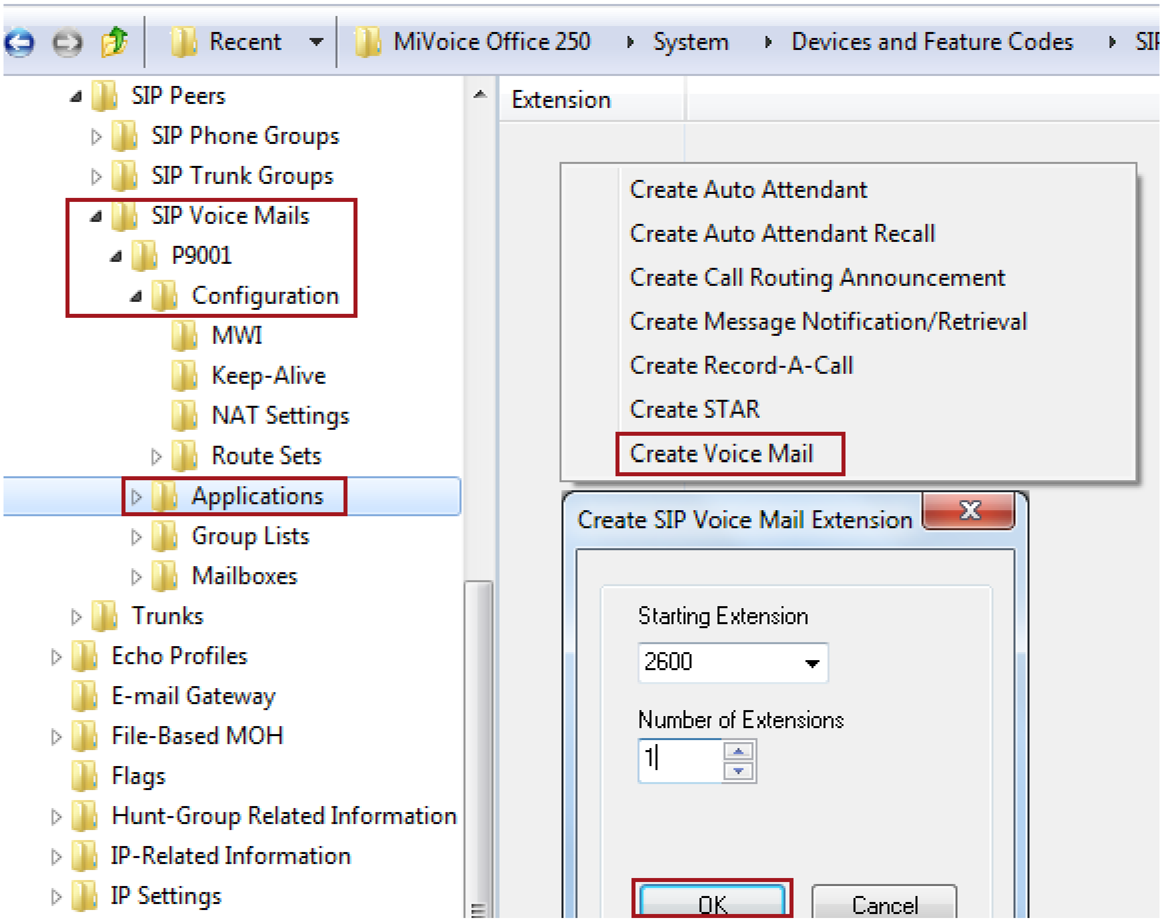

Figure 23: SIP Voice Mail Application

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

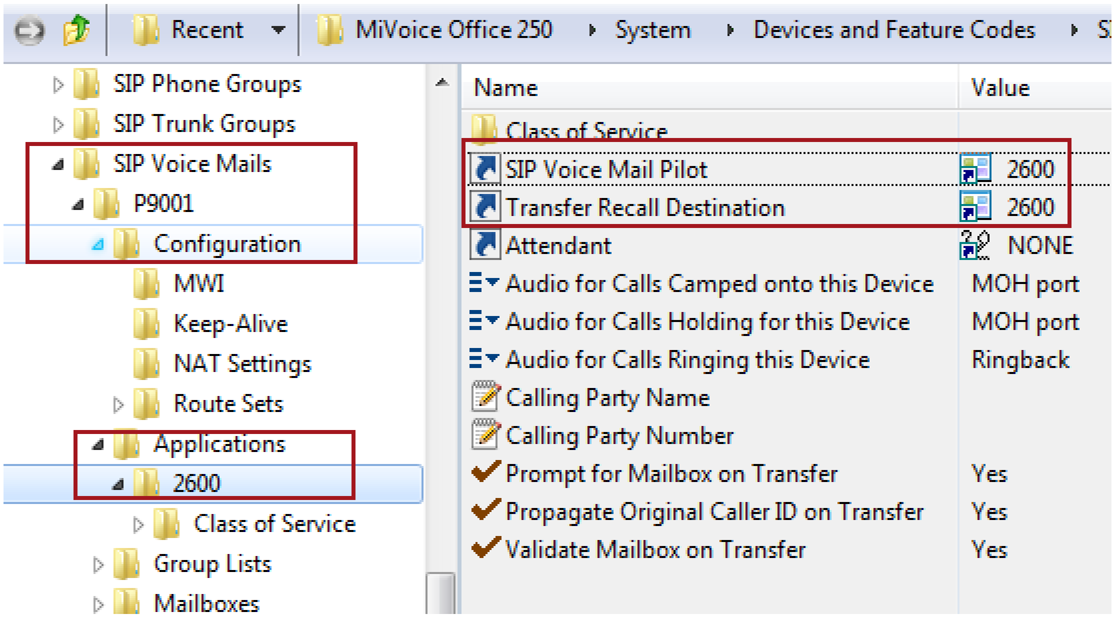

Figure 24: SIP Voice Mail Pilot

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**

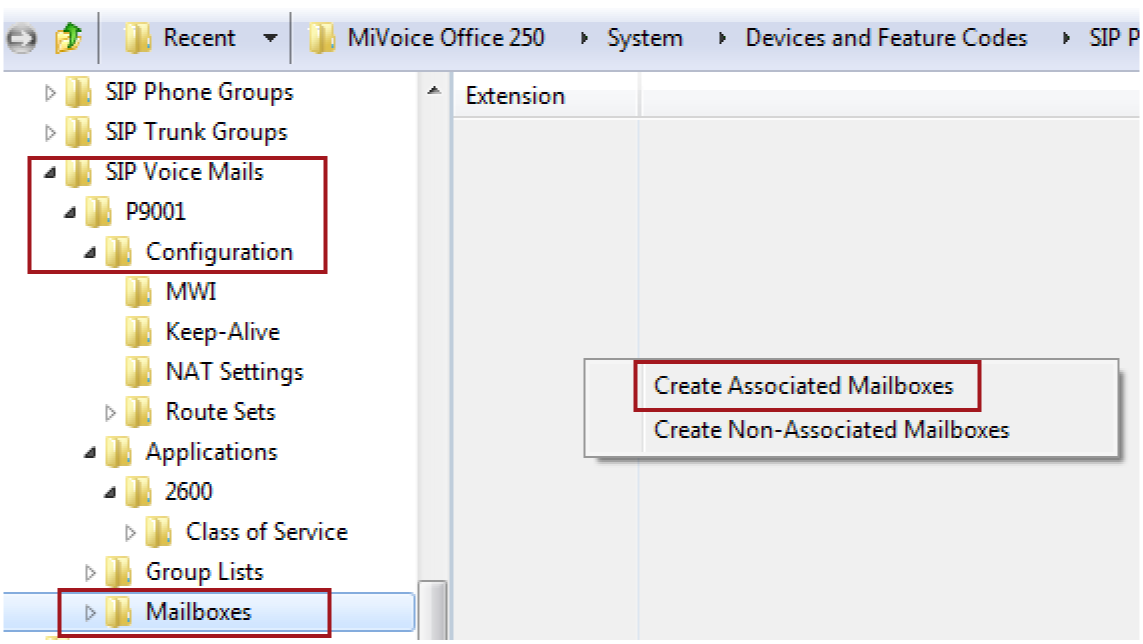

Figure 25: Create Associated Mailbox

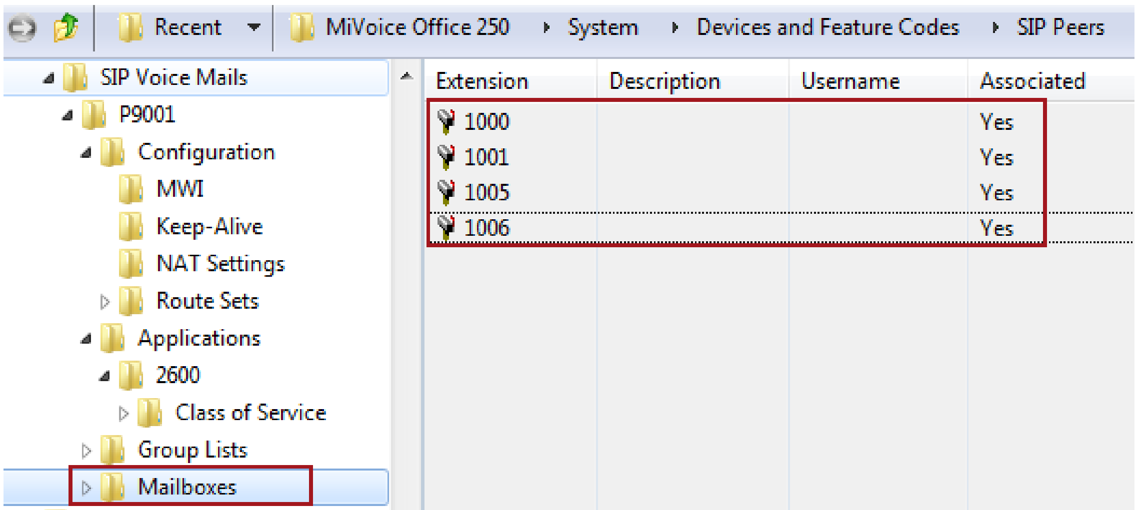

Figure 26: Associated Mailboxes

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**

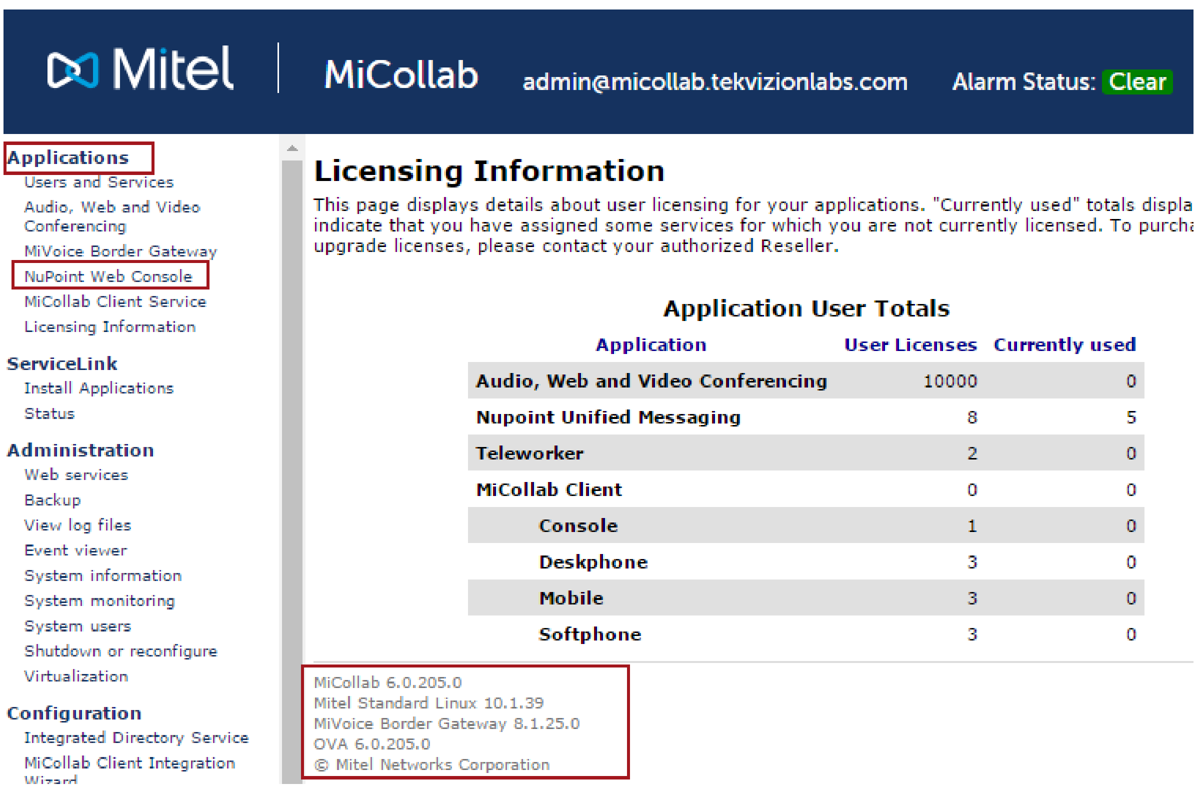

Figure 27: MiCollab Server-Manager

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

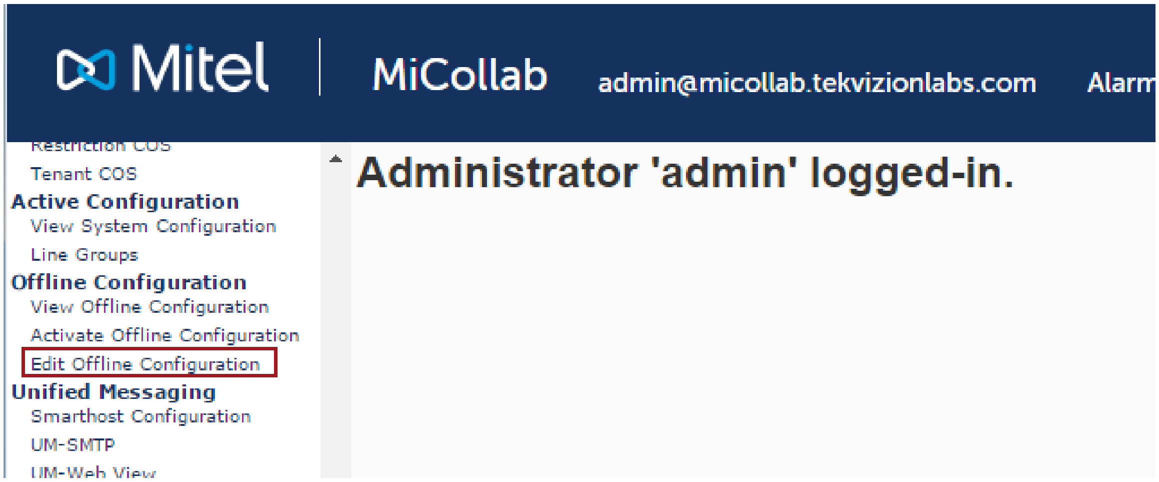

Figure 28: Offline Configuration

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

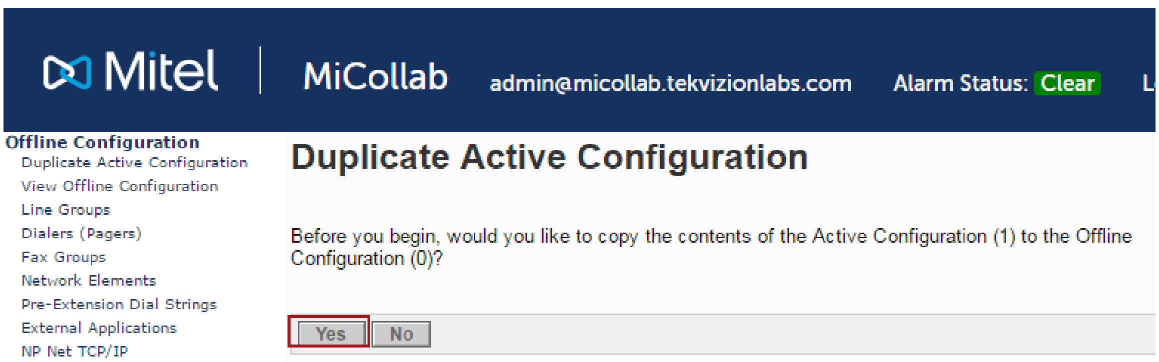

Figure 29: Duplicate Active Configuration

Add SIP Gateway Network Element

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

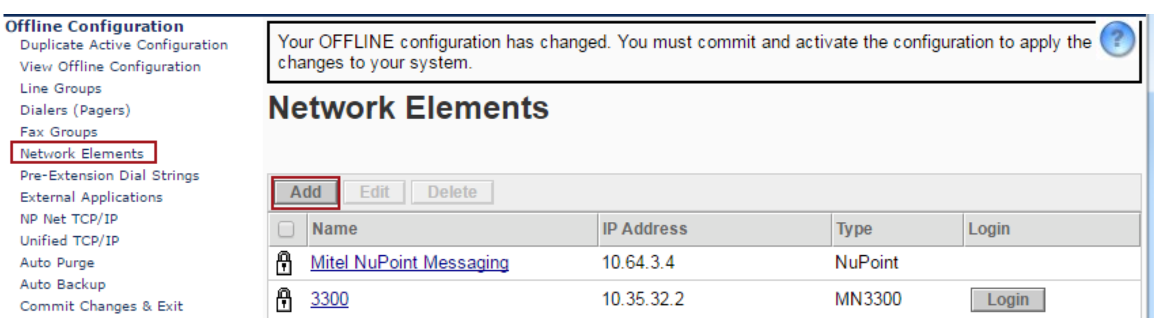

Figure 30: Network Elements

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**

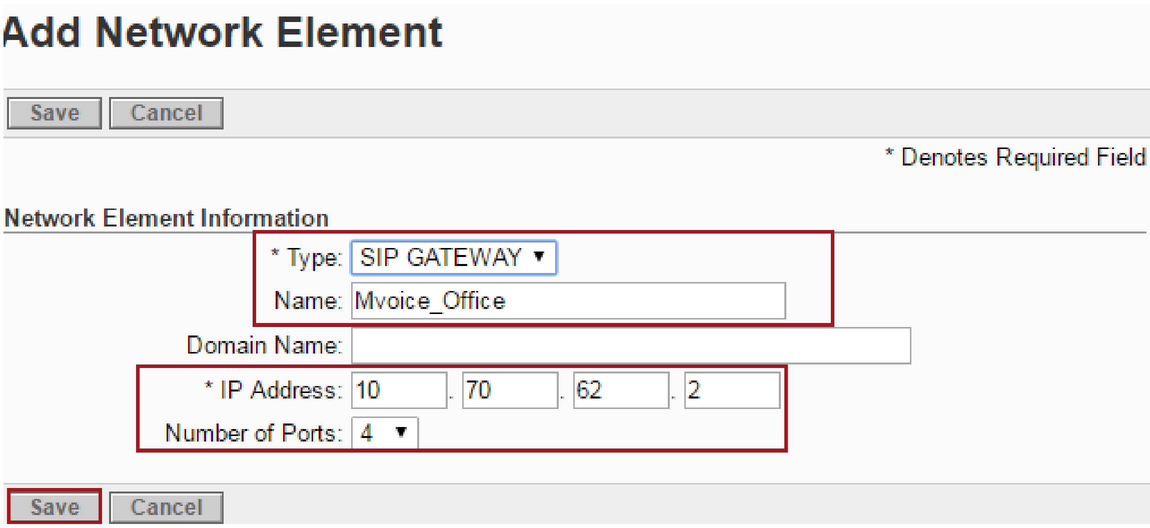

Figure 31: Add Network Element

Add Voice Mail Line Group

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

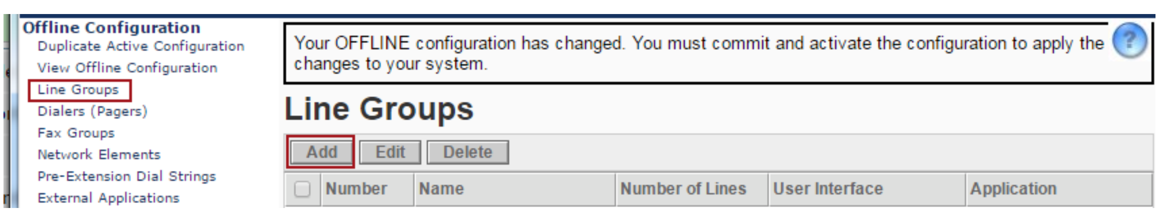

Figure 32: Line Groups

- 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

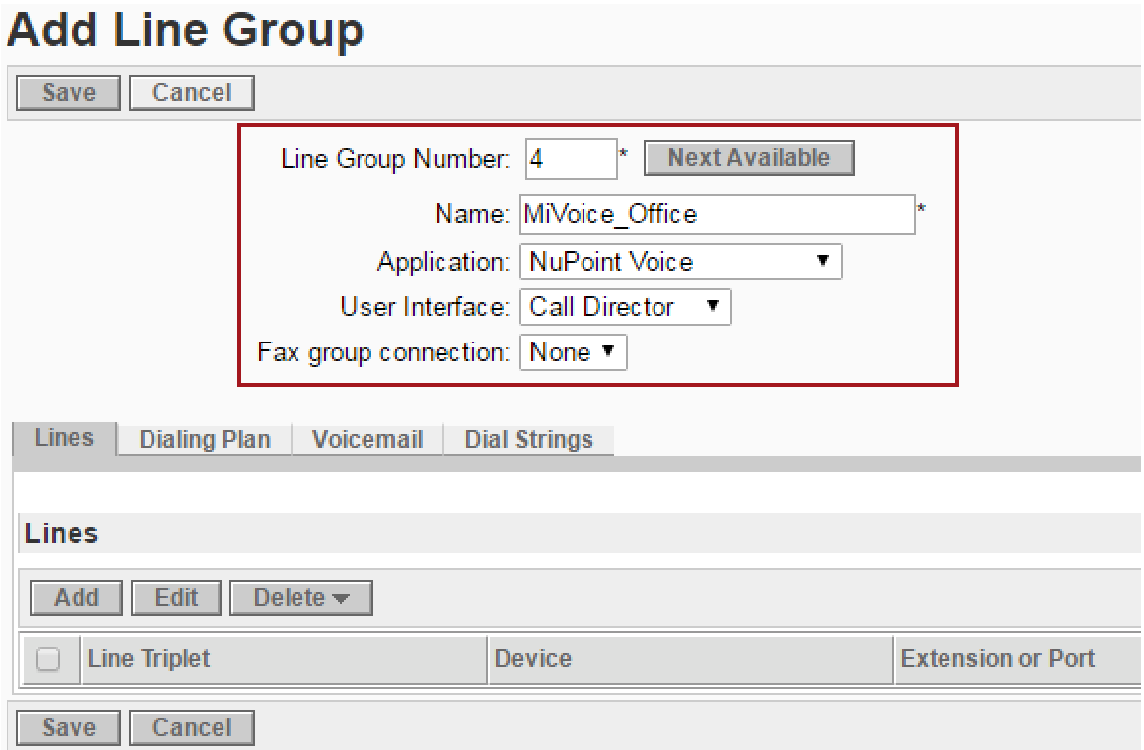

Figure 33: Add Line Group

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

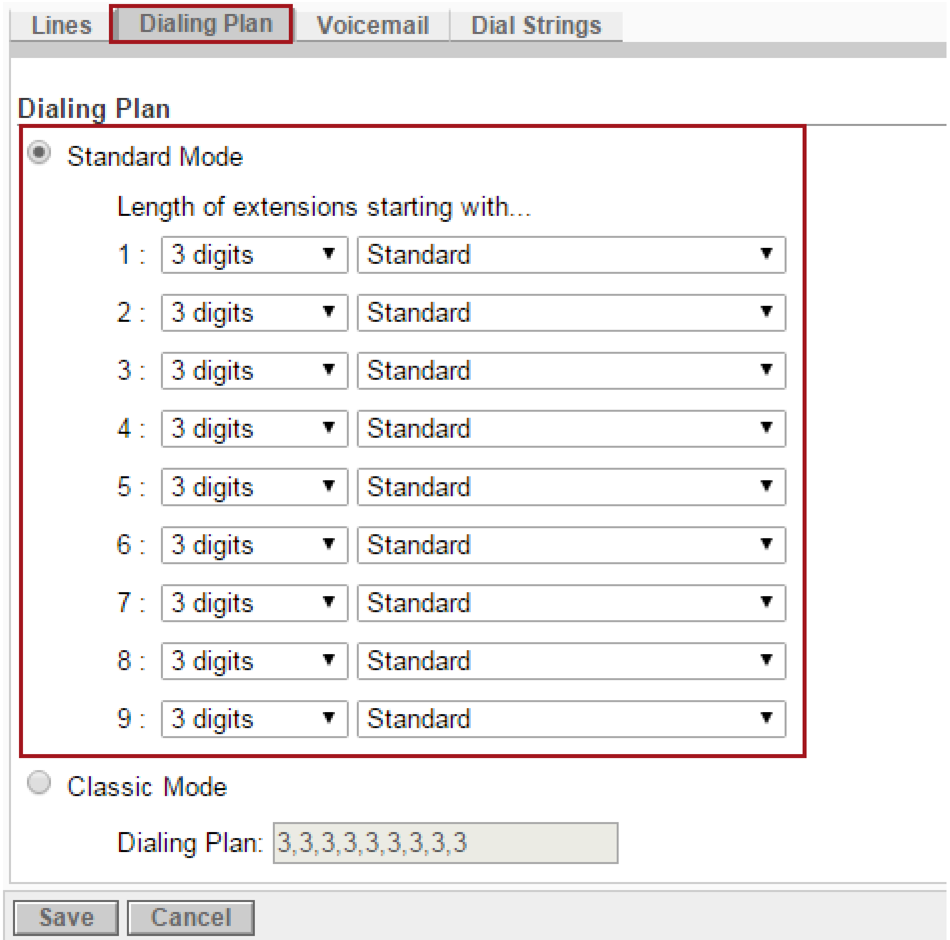

Figure 34: Line Group - Dialing Plan

- 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 1st 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**

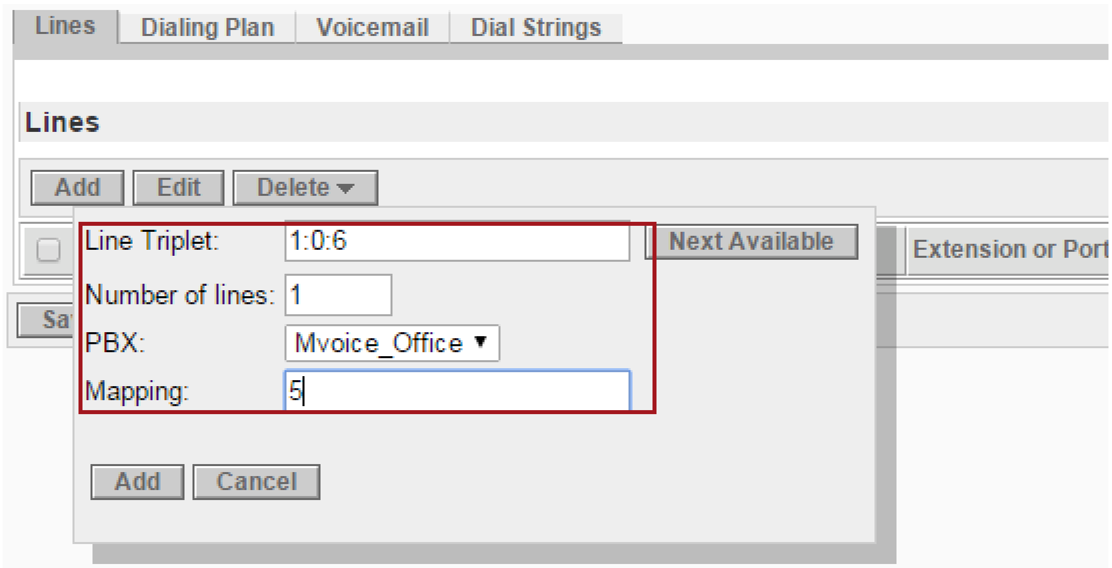

Figure: 35 Add Line Triplet

- 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

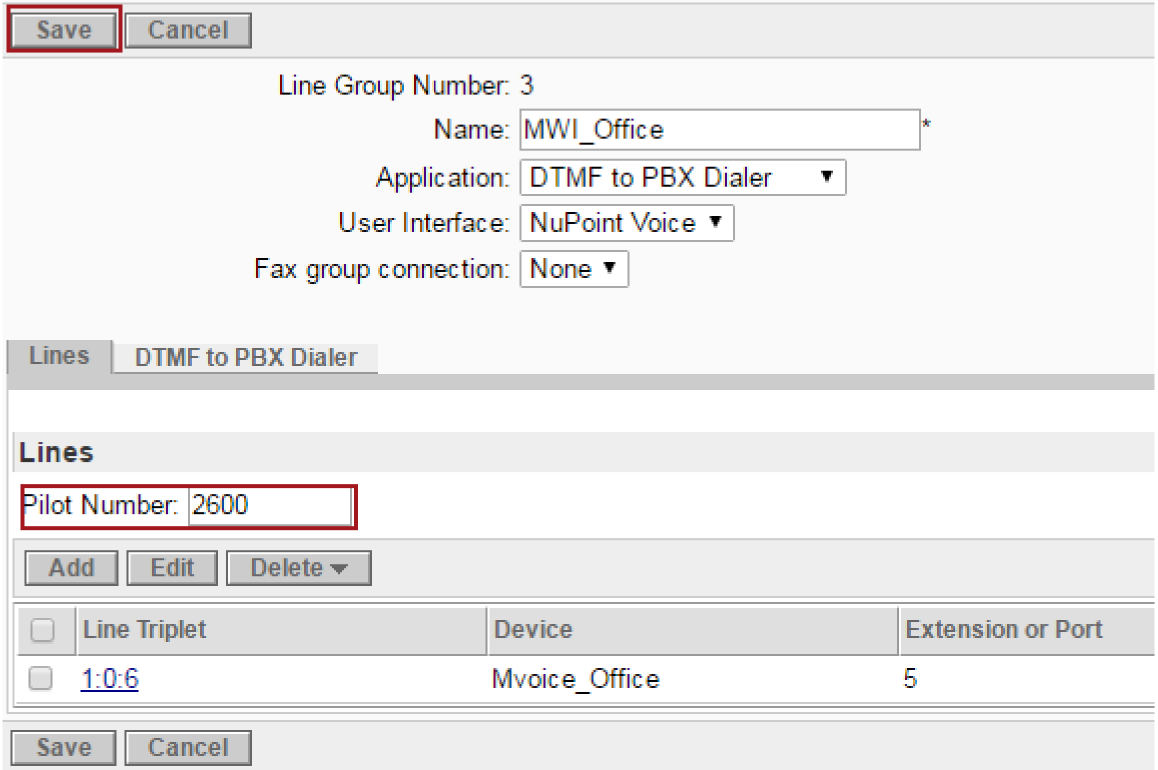

Figure 36: Add Line Group – Cont.

Add Message Waiting Indicator (MWI) Line Group

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

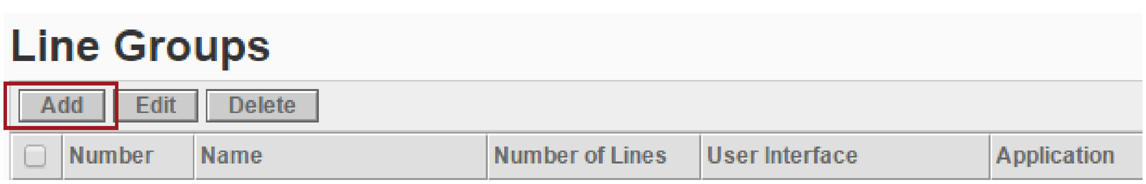

Figure 37: Add MWI Line Group

- Set **Line Group Number**: It will automatically populates 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**

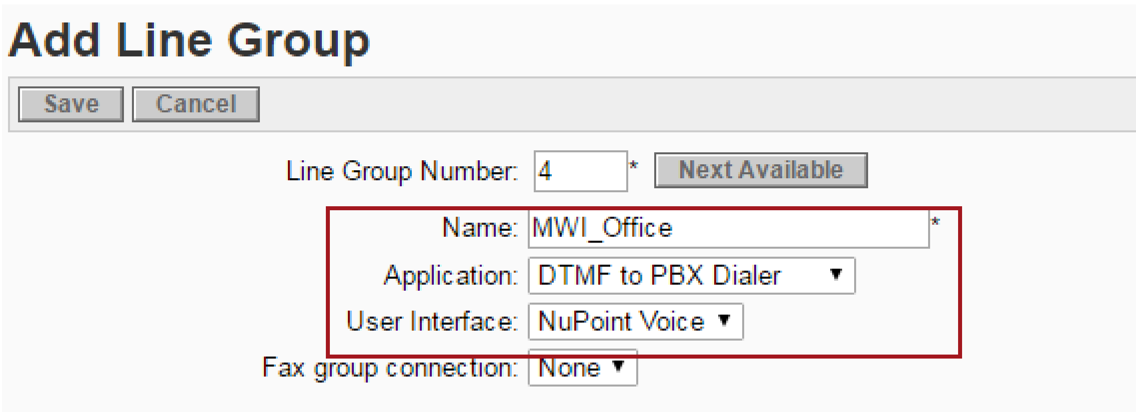

Figure 38: Add MWI Line Group – Cont.

- 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

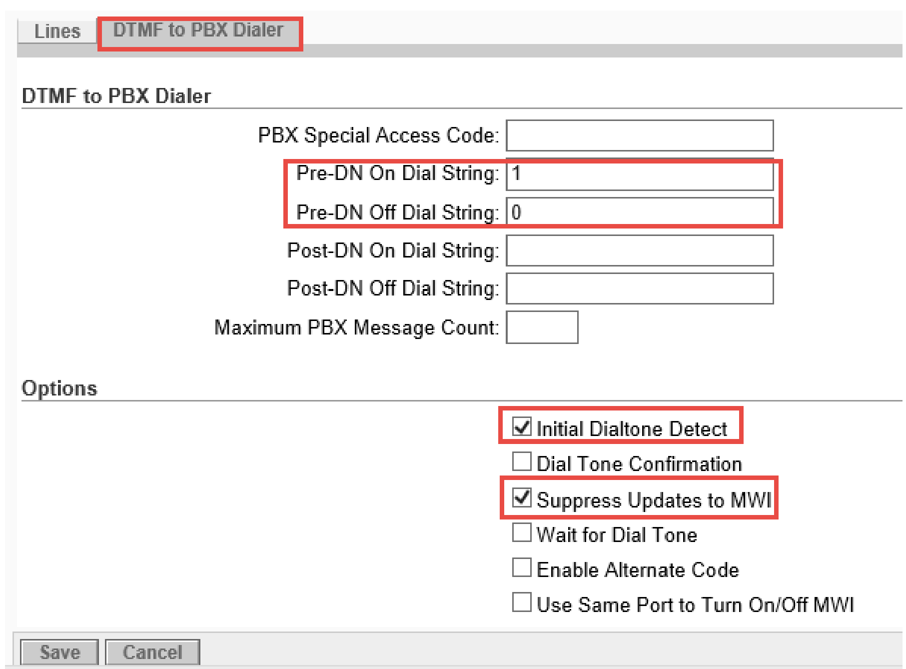

Figure 39: DTMF to PBX Dialer

- 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**

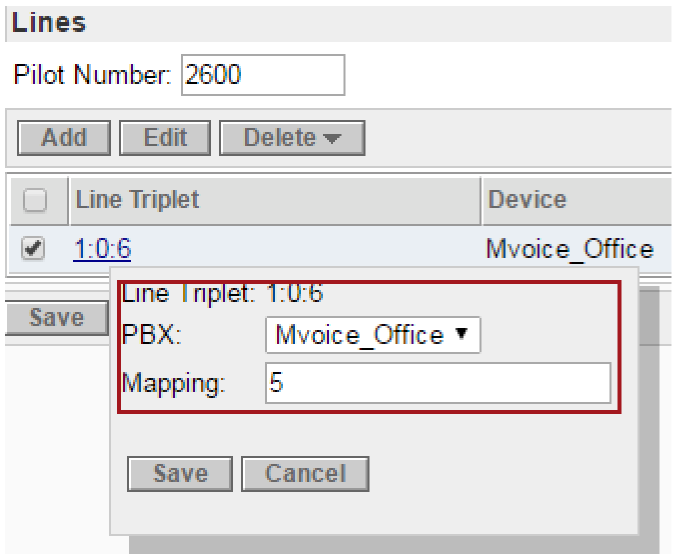

Figure 40: Add MWI Line Triplet

- 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

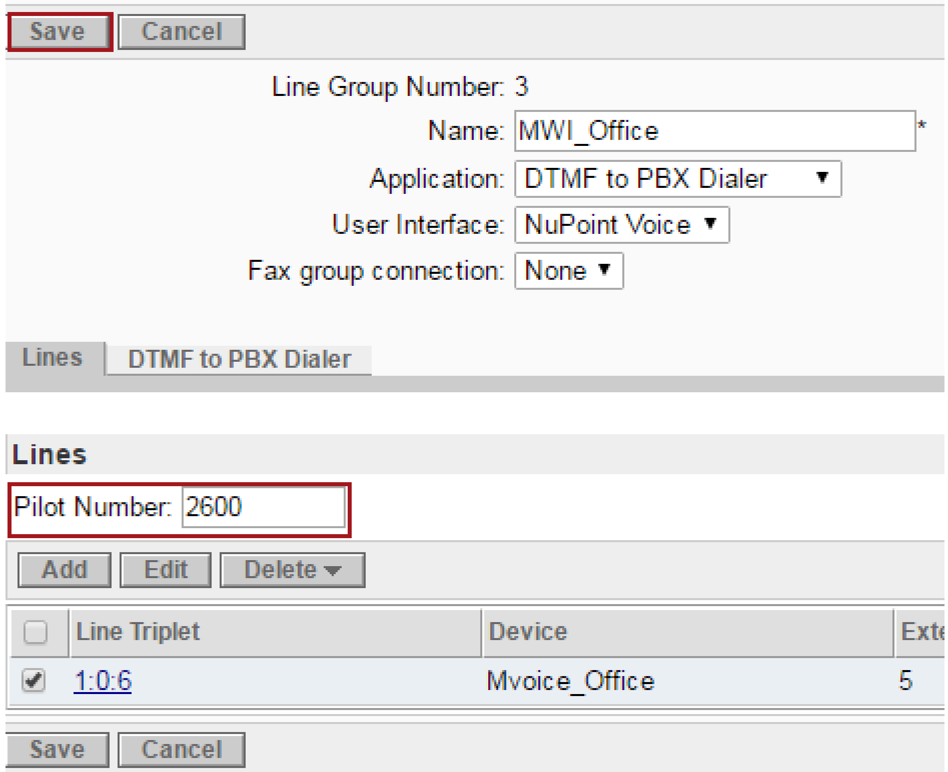

Figure 41: Add MWI Line Group – Cont.

Activate Offline Configuration

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

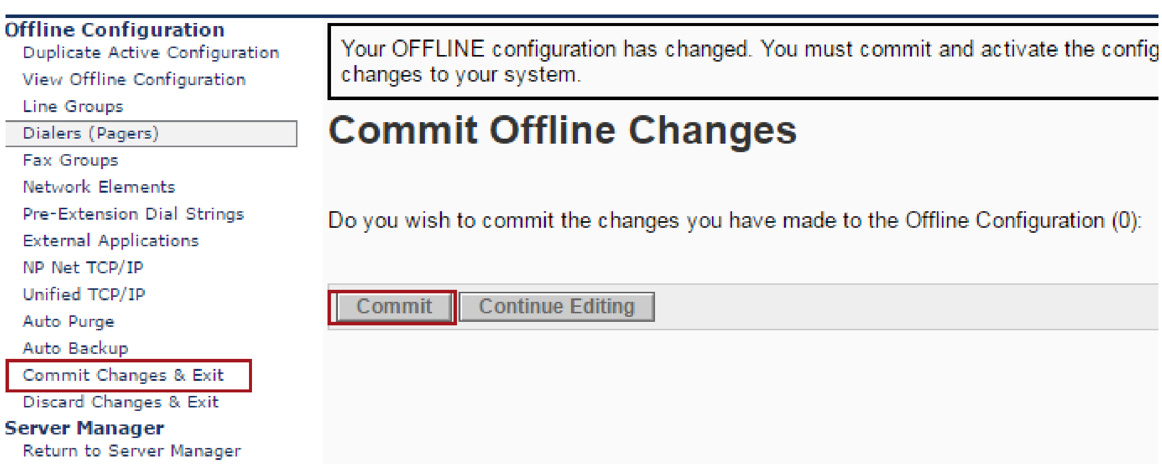

Figure 42: Commit Changes

Figure 43: Commit Changes – Cont.

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

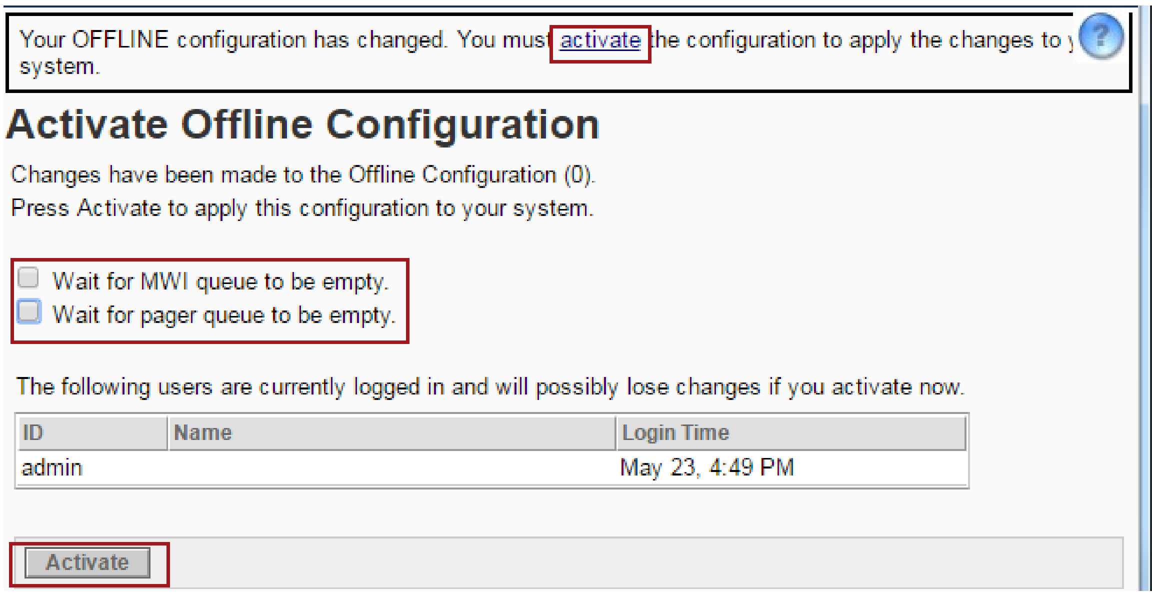

Figure 44: Activate the Configuration

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

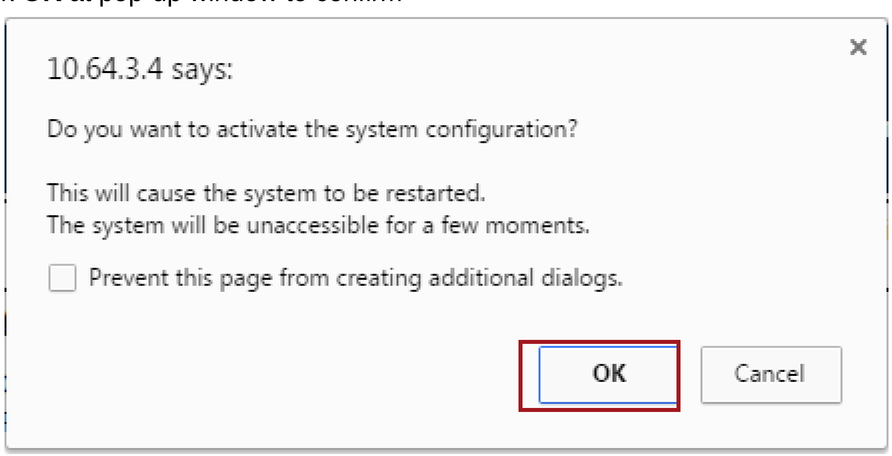

Figure 45: Activate the Configuration – Cont.

- Click **OK** at Activation complete page

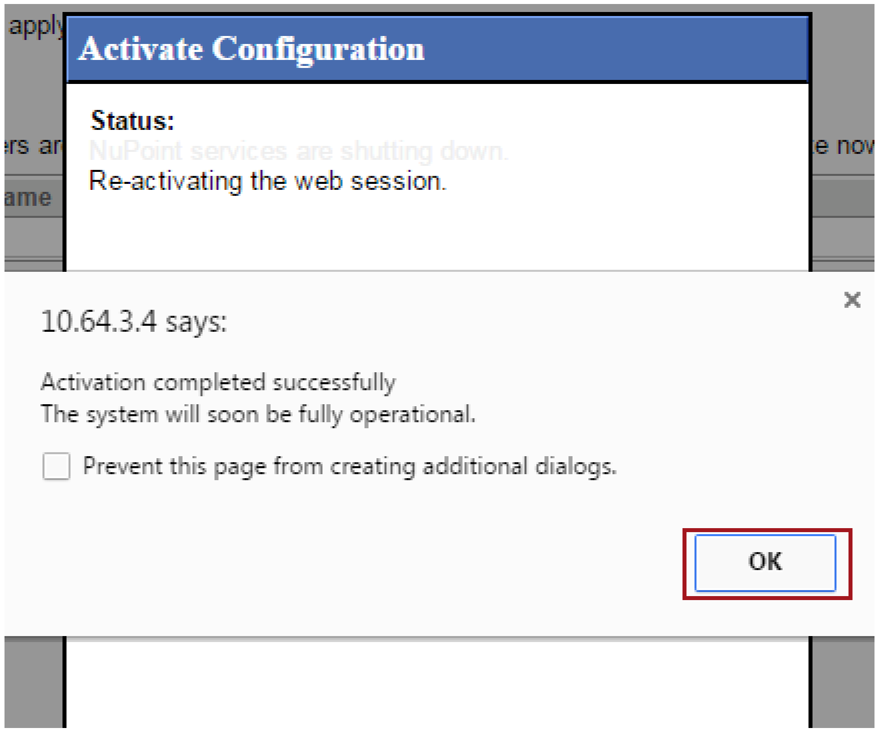

Figure 46: Activate the Configuration – Cont.

Add Mailbox

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

- Click **Add**

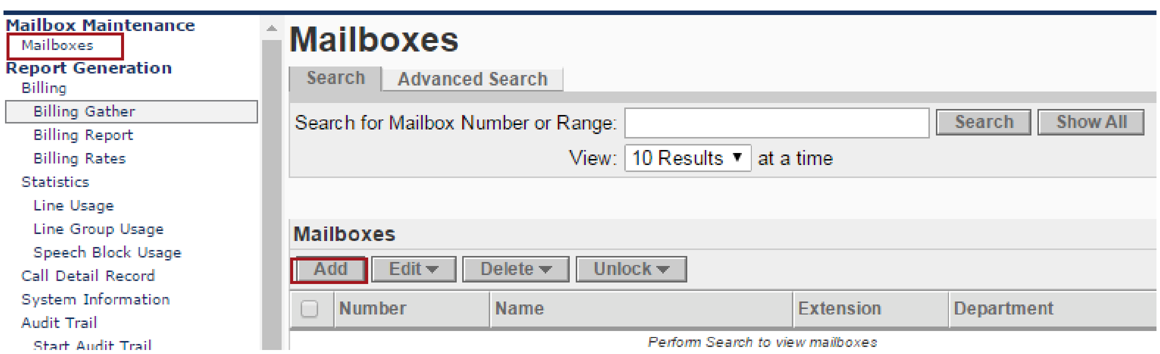

Figure 47: Add Mailbox

- 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

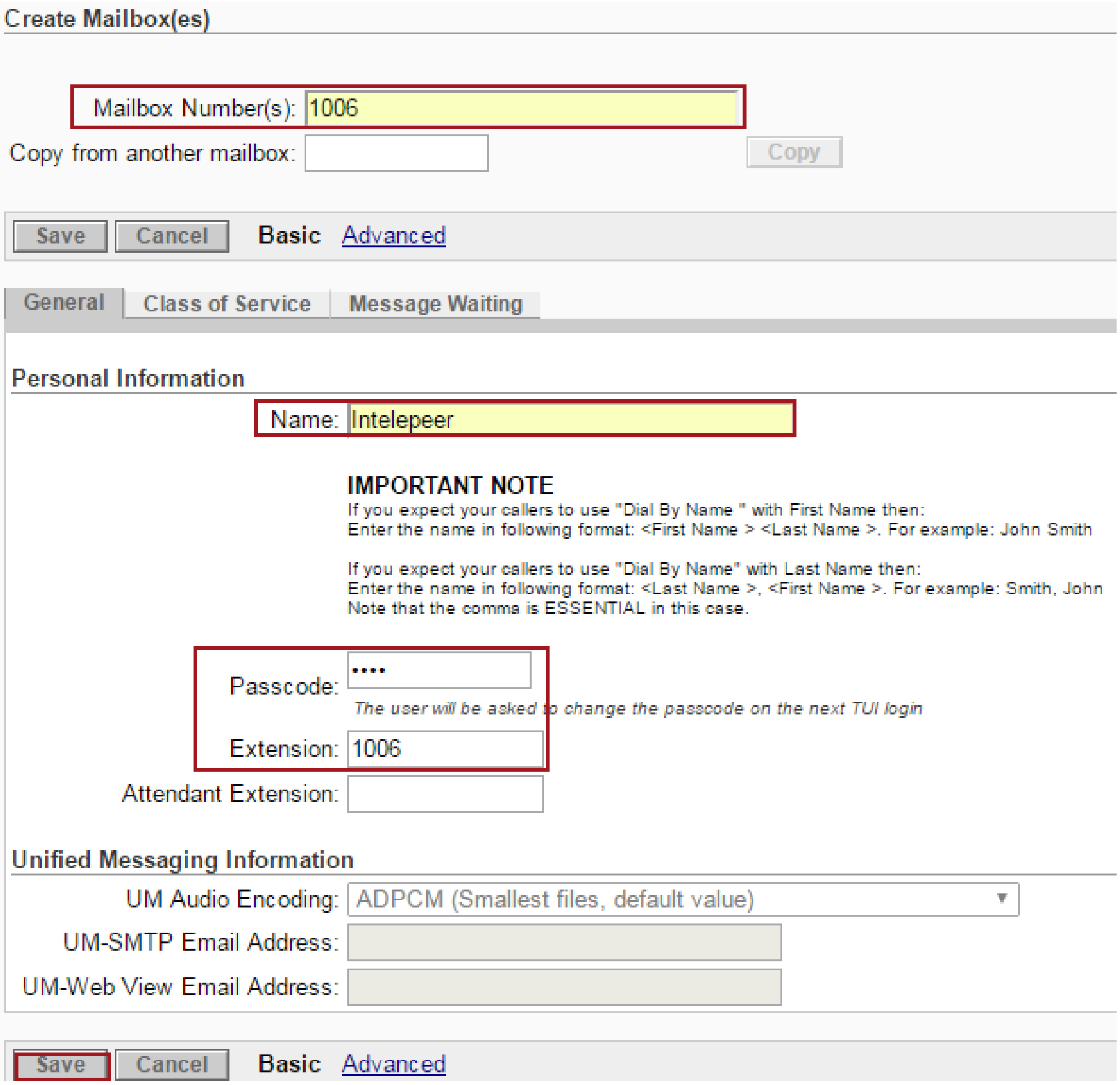

Figure 48: Add Mailbox – Cont.

- 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**

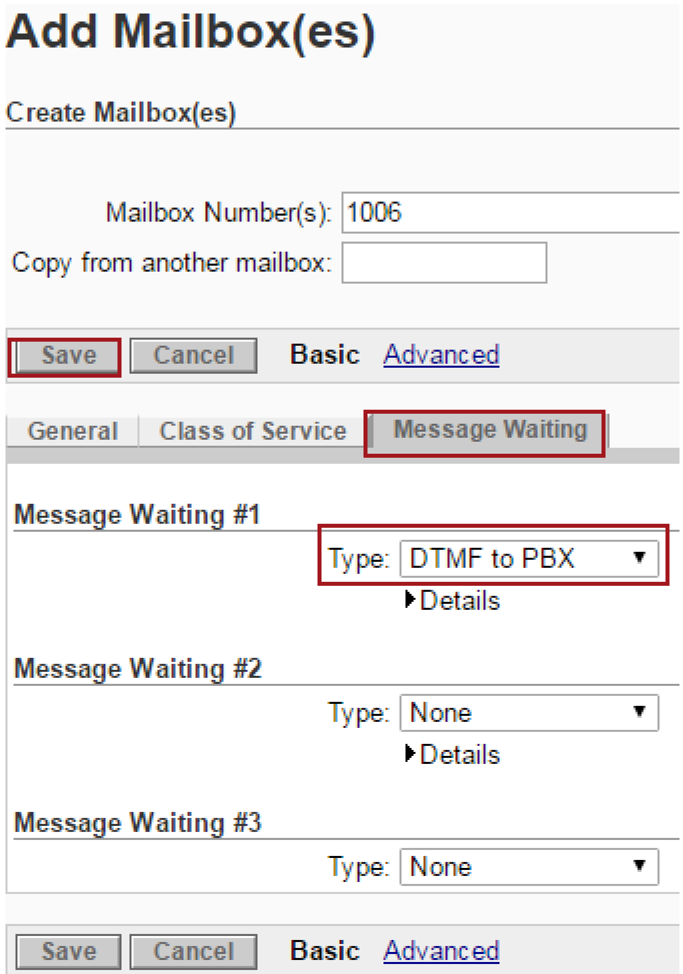

Figure 49: Message Waiting

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**

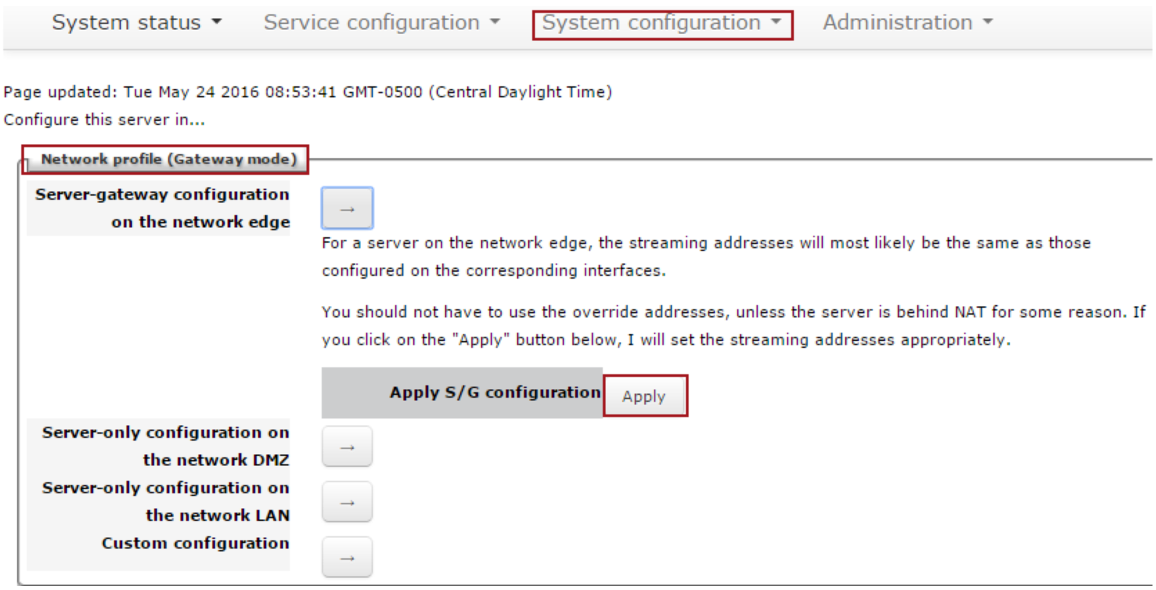

Figure 50: Network Profiles

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**

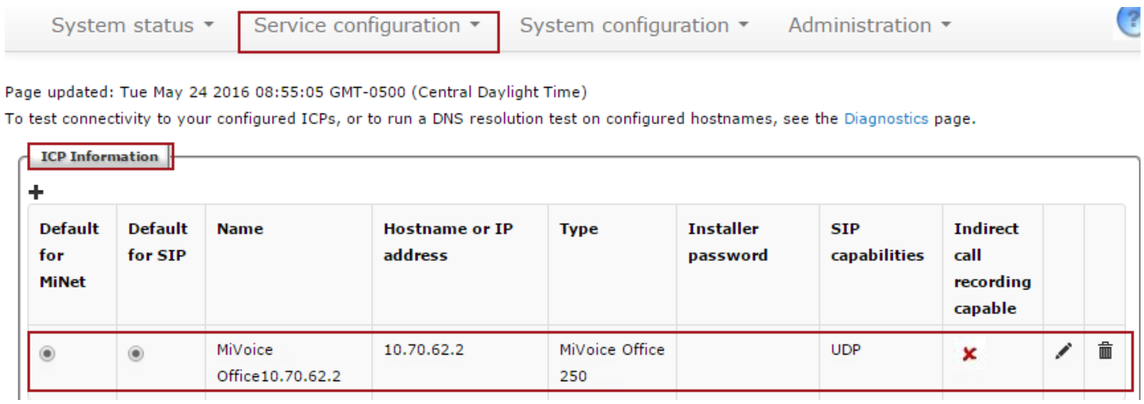

Figure 51: MIVOICE BORDER GATEWAY Configuration

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

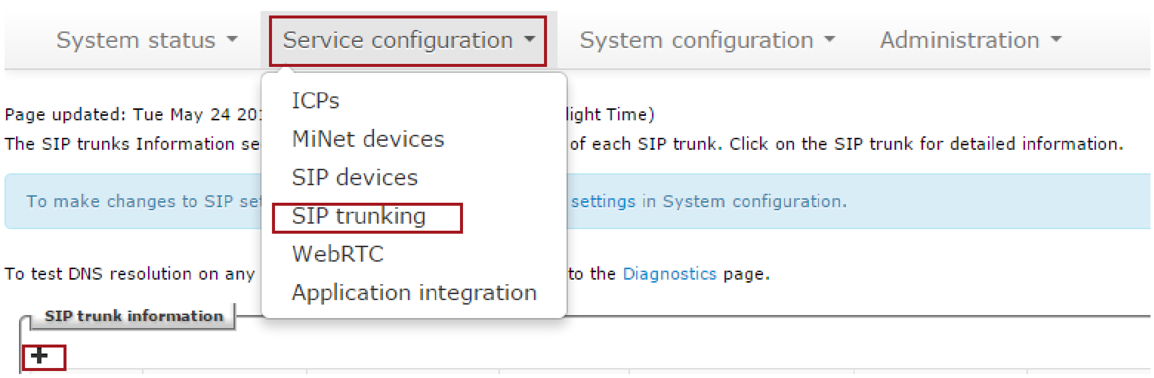

Figure 52: SIP Trunking Configuration

Enter the SIP trunk details as follows:

Set **Name**: IntelePeer 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**

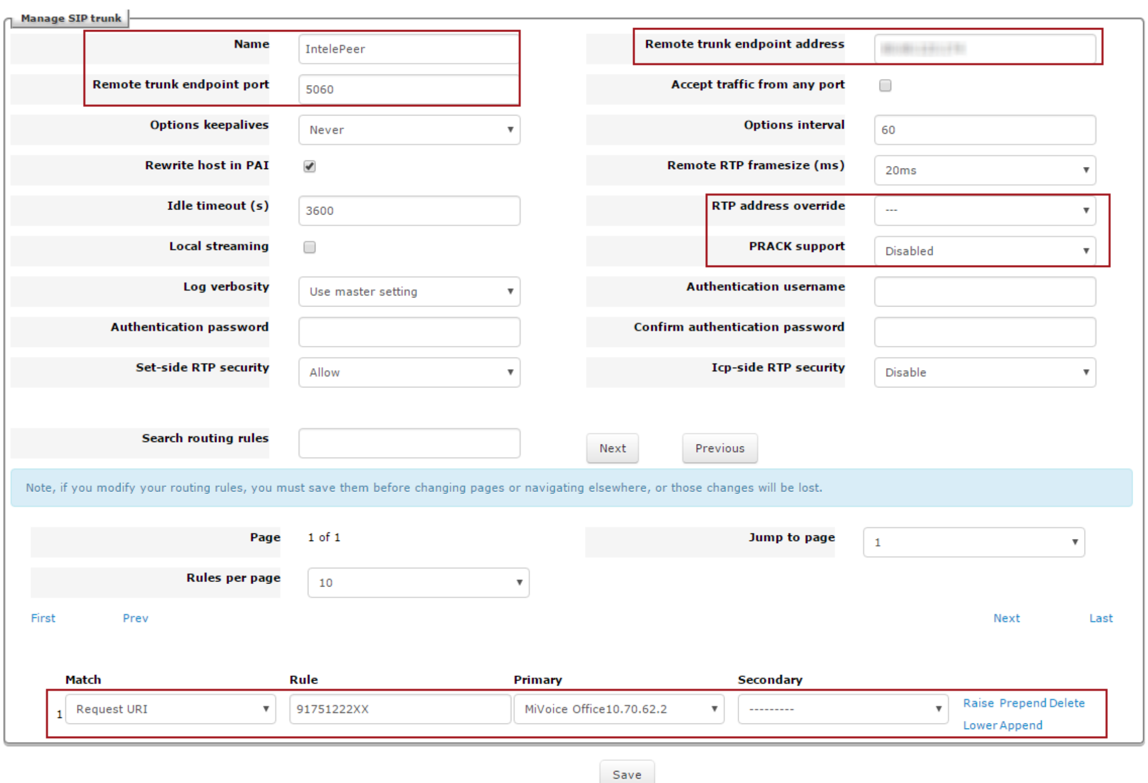

Figure 53: SIP Trunk Configuration Settings

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