MITEL – SIP CoE

TECHNICAL CONFIGURATION NOTES

Configure MiVoice Office 250 for use with Fusion Connect SIP Trunking

SIP COE 15-4940-00419

NOVEMBER 2015

NOTICE

TRADEMARKS

Mitel Technical Configuration Notes –
Configure the Mitel MiVo 250 CP for use with Fusion Connect SIP Trunk Service provider

CONTENTS:

OVERVIEW

This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel MiVo 250 to connect to Fusion Connect Service Provider SIP trunk. 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	November, 2015	Initial Interop with Mitel MiVo 250 6.1 and Fusion Connect Service Provider SIP trunk

INTEROP STATUS

The Interop of Fusion Connect Service Provider has been given a Certification status. This service provider or trunking device will be included in the SIP CoE Reference Guide. The status Fusion Connect Service Provider achieved is:

COMPATIBLE	The most common certification, which means that SIP Trunk from Fusion Connect Service Provider 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 Fusion Connect service provider and the Mitel MiVo 250.

Manufacturer	Variant	Software Version
Mitel	MiVo 250 CS	6.1.27
Mitel	IP set 5324, 5330,5340, 5360	06.03.01.05
Mitel	SIP 68xx series end point	4.0.0.1096
Fusion Connect SIP Trunk		As of November, 2015

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 the SIP Service provider and their PSTN gateway, call holding, transferring, conferencing, busy calls, long calls durations, variable codec.	☑
Automatic Call Distribution	Making calls to an ACD environment with RAD treatments, Interflow and Overflow call scenarios and DTMF detection.	☑
MAS/NuPoint Voicemail	Terminating calls to a NuPoint voicemail boxes and DTMF detection.	☑
Embedded voicemail	Using the embedded voicemail system on Mitel MiVo 250.	☑
Dynamic Extension Express	Receiving a call through the SIP Service provider and their PSTN gateway to Mobile extensions and TUI interface. Also moving calls to/from Desktop and Twinned devices.	☑
Video	Making and receiving a call through SIP trunk with video capable devices.	N/A
Fax	G.711 Fax Calls	☑
Fax	T.38 Fax 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 Fusion Connect Service Provider has a SIP trunk connected to the Mitel MiVo 250 Networks Platform.

Feature	Problem Description
Outbound Private Calls	Private outbound calls are not supported as Fusion requires the main account number to be present in the "From" SIP header. Mitel removes this number when a call is marked as private.
Paketization	P-times of other than 20ms are not supported by Fusion. **Recommendation**: Use the default 20ms P-time
T.38 FAX	An issue was found during T.38 FAX calls where if a re-invite occurs due to session timer expiry during the FAX call the call will drop. **Recommendation**: Set the Keep-alive (session timer to a value, which will prevent the timer form expiring during a T.38 FAX call. Reference defect MN00607790 when contacting Mitel Product Support.
Video Calls	Video is not supported on the Mitel MiVo 250 at this time.

NETWORK TOPOLOGY

This diagram shows how the testing network is configured for reference.

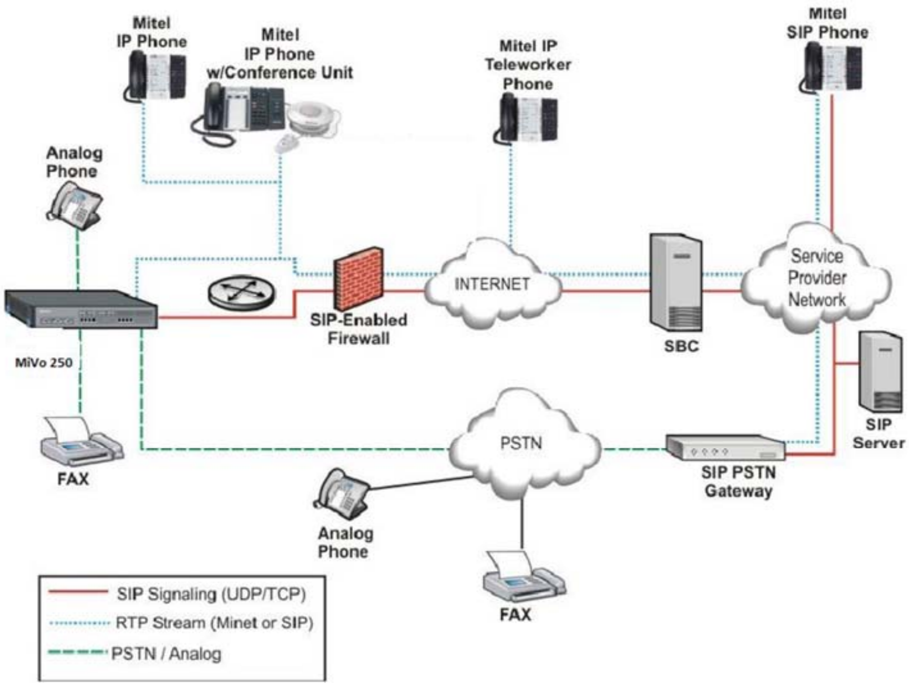

Figure 1 – Network Typology

CONFIGURATION NOTES

This section is a description of how the SIP Interop was configured.

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.

MITEL MiVO 250 COMMUNICATIONS PLATFORM CONFIGURATION NOTES

The following steps show how to program Mitel MiVo 250 Communications Platform to interconnect with Fusion Connect Service Provider using the Adtran 908e eSBC in a back to back configuration.

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 for G.729. Almost all Enterprise LAN networks can support this level of traffic without any special engineering. Please refer to the MiVo 250 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 Mitel MiVo 250 Communications Platform Programming

The SIP signaling to and from the MiVo 250 is configured to use UDP on Port 5060.

Licensing and Option Selection – SIP Licensing

Ensure that the Mitel MiVo 250 is equipped with enough SIP Trunks licences for the connection to Fusion Connect Service Provider. This can be verified within the Software License form (see **Figure 2**).

Check the total number of licenses in the SIP Trunk Licences field. This is the maximum number of SIP trunk sessions that can be configured in the Mitel MiVo 250 to be used with all service providers and applications.

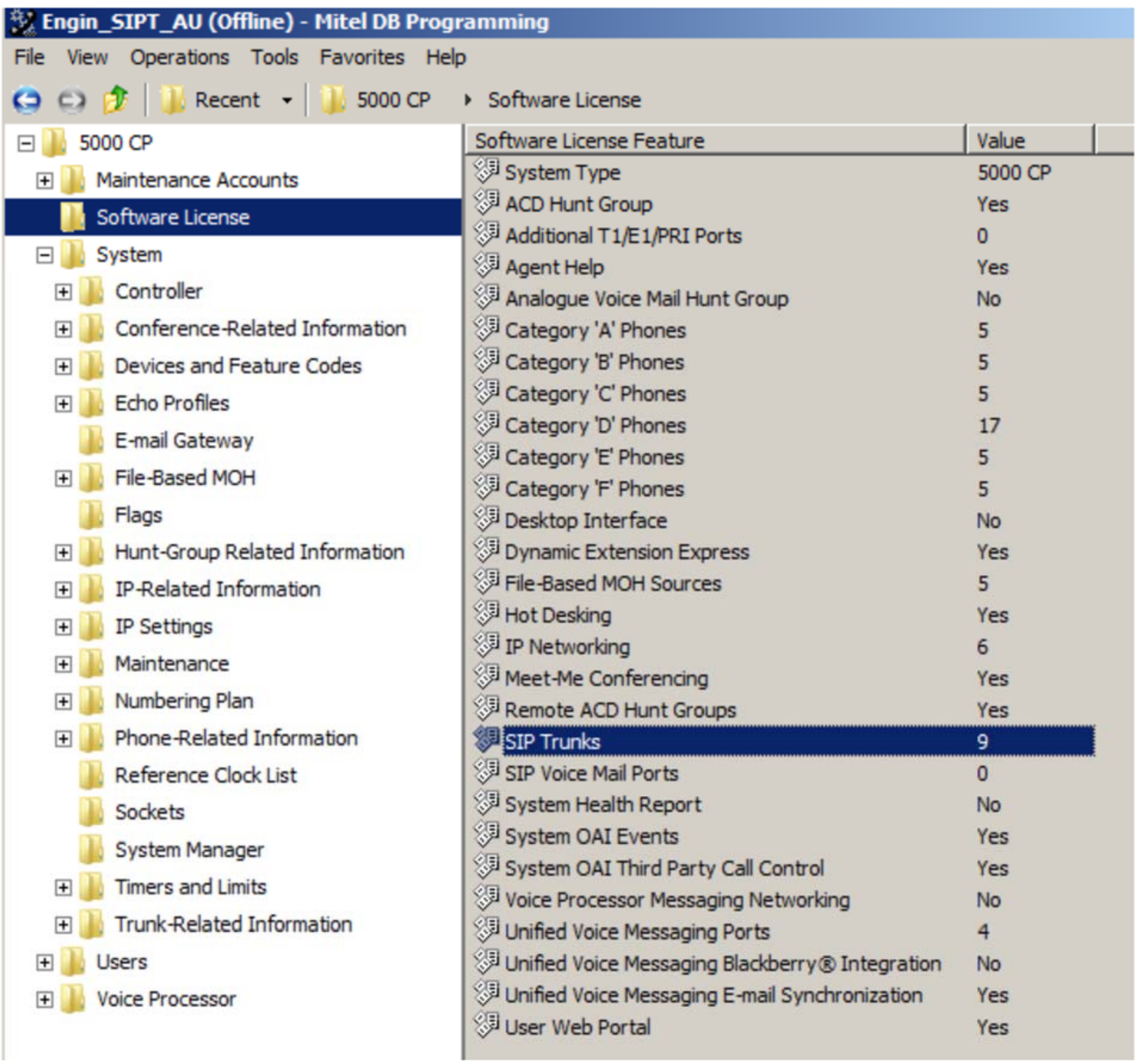

Figure 2 – Example of SIP Licensing

Creating and Configuring a SIP Peer Trunk Group

To support SIP trunks through a SIP trunk service provider, the SIP Trunk Groups folder was added to the SIP Peers folder in DB Programming.

To create a SIP Trunk Group for Fusion Connect Service Provider, navigate to System->Device and Feature Codes->SIP Peers->SIP Trunk Groups and right click in the right hand pane. Then select “Create SIP Trunk Group”. (See **Figure 3**)

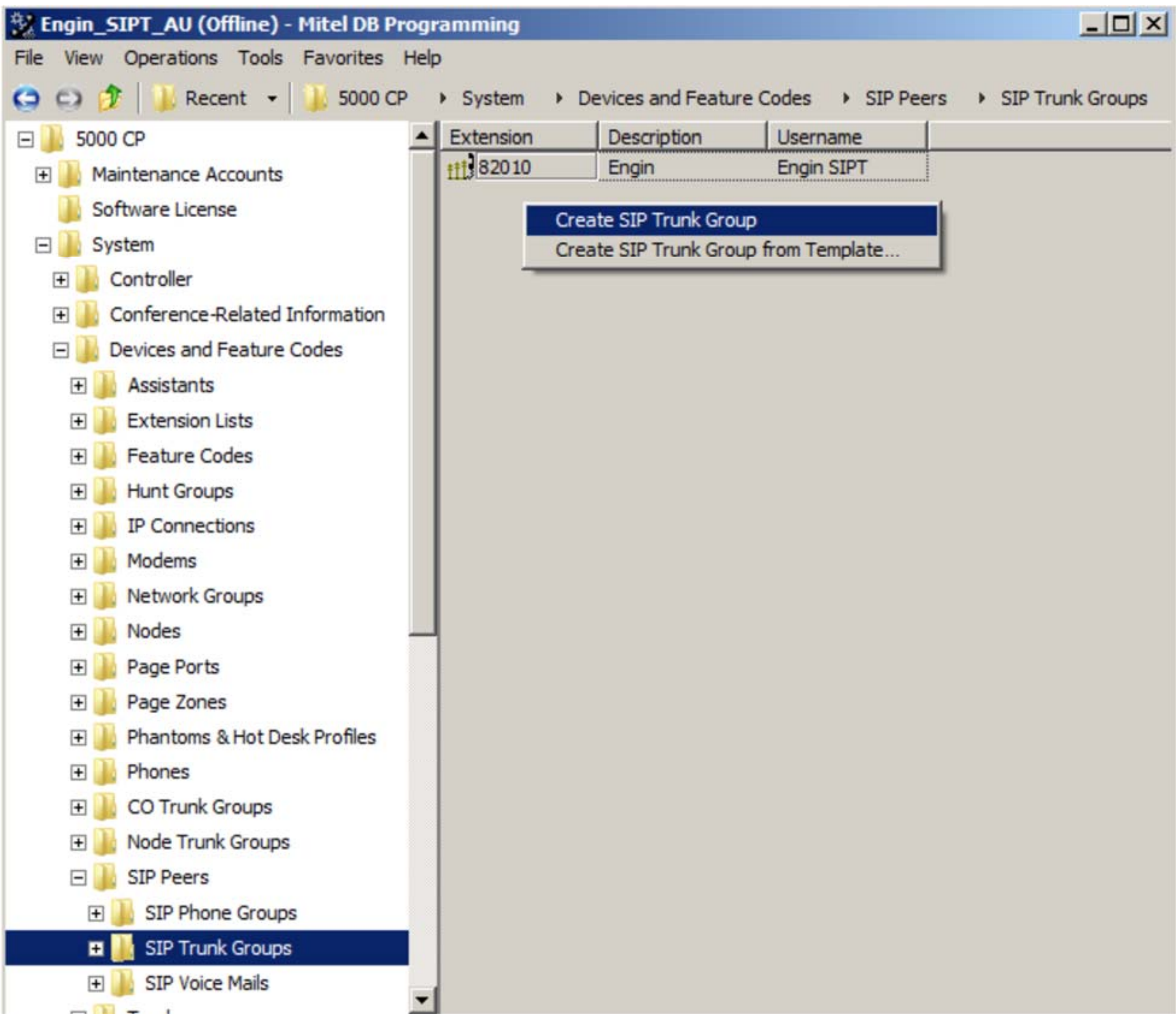

Figure 3 – Example of Create SIP Trunk Group

When you create a SIP Peer trunk group without using a template, you must obtain the necessary information from the SIP trunk service provider, and then configure this information in DB Programming.

When the Trunk group for Fusion Connect Service Provider is being created, we have to configure parameters in Configuration and Trunk Group Configuration nodes.

The example configuration for Fusion Connect is provided in this document to manually configure the SIP trunk group.

Programming the Configuration settings

Under Configuration configure the following: (see **Figure 4**)

- **IP Address**: Configure the address of Fusion Connect.
- **Port Number**: Indicates the port that the system listens on the system for SIP peer messages. The range is 0–65535. Leave the default value of 5060.
- **Fully Qualified Domain Name**: Indicates the domain name of the SIP peer trunk group. Not required if using IP addresses
- **Call Configuration**: Enter the call configuration number in Value field that you want to use with this trunk group. (See Figure 12)

  Double clicking **Call Configuration** takes you to the Call Configuration folder where you can add a new call configuration profile or configure the existing profile(s) (e.g. codecs for voice and faxing, DTMF settings, etc. See section Call Configurations). (System->IP-Related Information->Call Configurations-><call configuration number>).

- **Operating State**: Indicates the operating state of the SIP peer. If required, the status could be changed to "Out-of-Service – Maintenance".
- **Maximum Number of Calls**: Indicates the maximum number of concurrent calls that are been permitted towards the SIP peer. This number is not configurable and is controlled by the number of SIP trunks that have been added to the SIP Trunk Group >Device and Feature Codes->SIP Peers->SIP Trunk Groups-><SIP Trunk group #>->Trunk Group Configuration->Trunks (see section Programming the Trunks in Trunk Group Configuration Folder for details).
- **Use ITU-T E.164 Phone Number**: If set to Yes, the Mitel MiVo 250 handles ITU-T E.164 formatted phone numbers as part of the incoming SIP INVITE messages from the SIP peer. For testing Fusion Connect this was set to No however Fusion Connect does support E.164.
- **Static Binding**: It specifies whether a static binding exist for the corresponding SIP Peer. If set to Yes, then IP address and listening port for the SIP Peer must be configured. Leave this setting to Yes.
- **Use Peer Address In From Header**: Set to No
- **Route Sets**: Enter the address of an outbound proxy if one is being used. In our test setup the address of the MBG was used here. **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. Leave the default values here.

- **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”. Leave the default values here.

  Alternate IP/FQDN List: Not required.

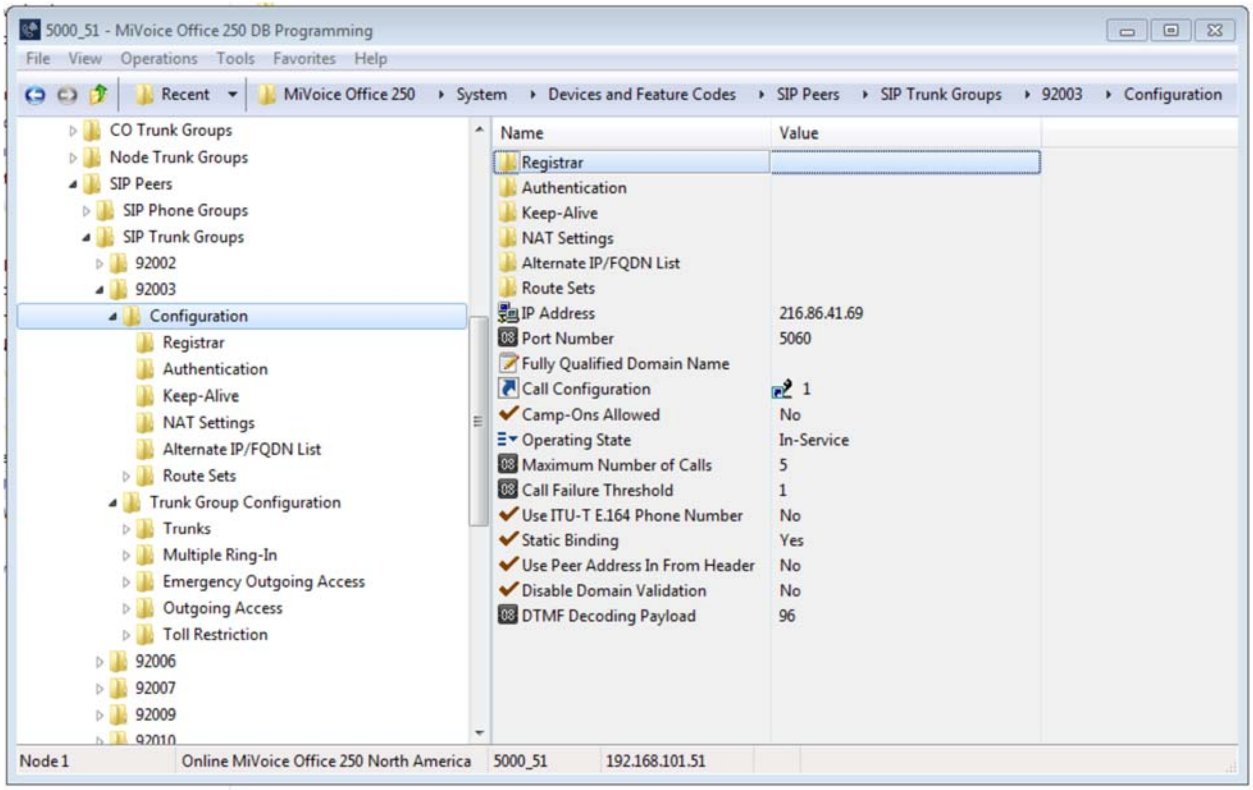

Figure 4 – Example of Configuration

- **Registrar**: Fusion Connect Service Provider' SIP trunk does require registration. Set Enable Registration option to yes as shown in **Figure 5**.

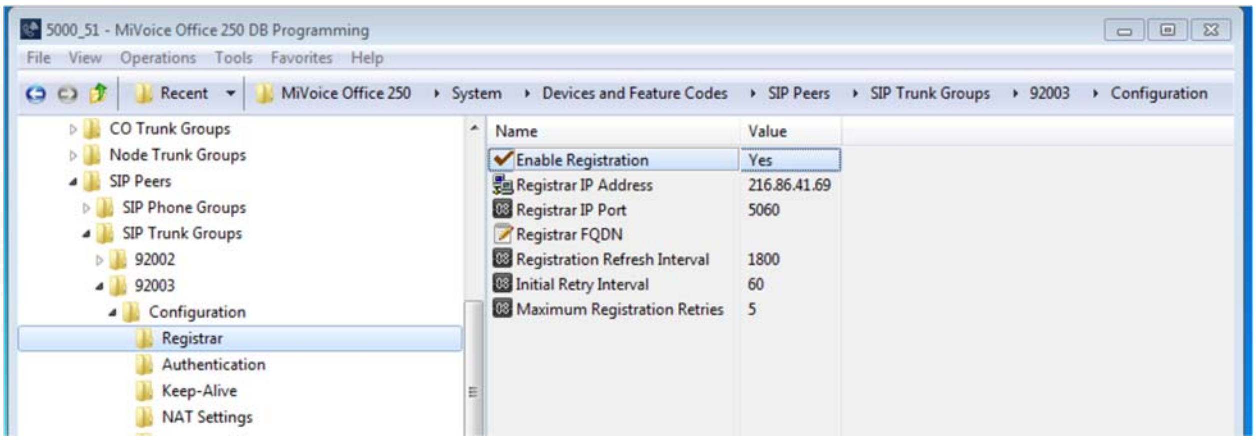

Figure 5 – Example of Registration

- **Authentication**:

  In this form, make sure that Enable In-Bound Authentication is set to No.

  The Out-Bound Username and Password must be configured with the Fusion Connect SIP account information or outgoing calls will fail.

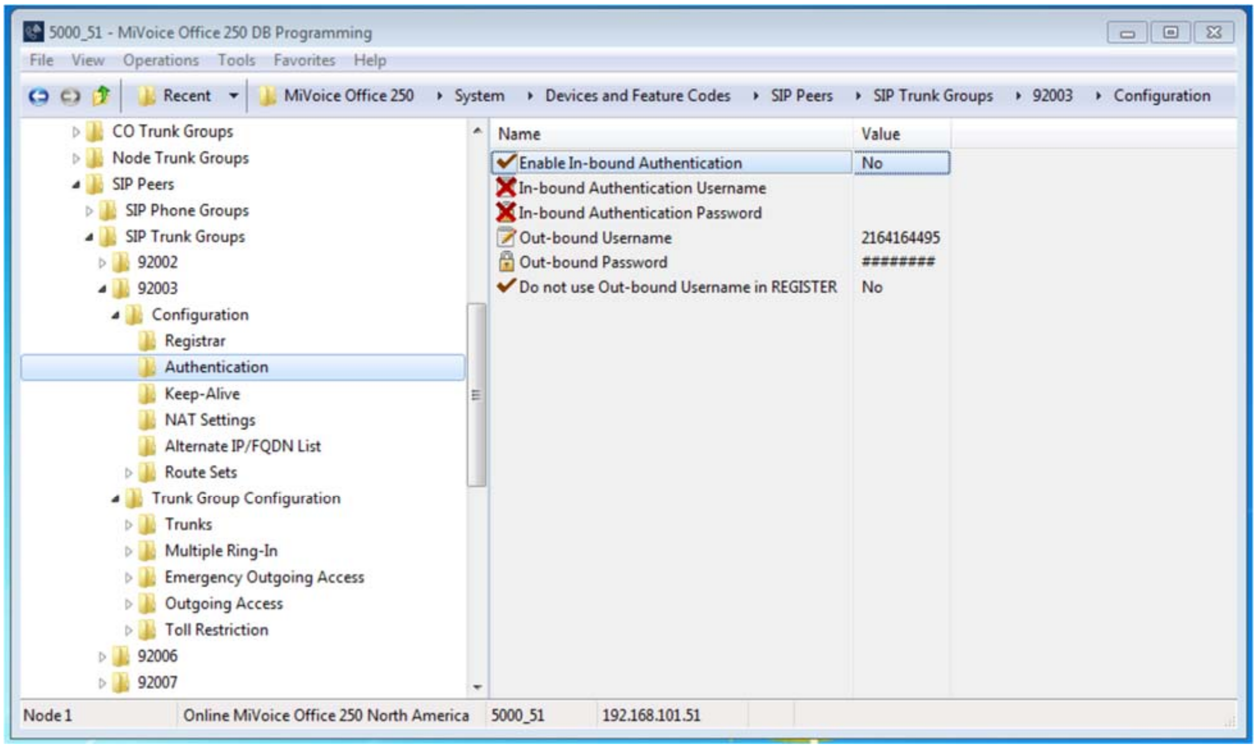

Figure 6 – Example of Authentication Form

Programming the Trunk Group Configuration properties

To program the Trunk Group Configuration properties, navigate to System->Device and Feature Codes->SIP Peers->SIP Trunk Groups-><SIP Trunk group #>->Trunk Group Configuration:

As per **Figure 7**, we need to configure three most important parameters:

- **Day** and **Night Ring-In Type** – The various Ring-In Types can be used. In the test environment the Single Ring-in was configured
- **Calling Party Number** (CPN) – is the default calling party number, which MiVo 250 presents to the provider's SIP trunk. Fusion Connect Service Provider should provide this number. This field must be populated. The MiVo 250 system will use this CPN to send to Fusion Connect as the Calling Party Number. If this value is missing Fusion Connect Service Provider will reject the outgoing call.

For the rest of the settings, refer to the DB Programming Help for trunk programming.

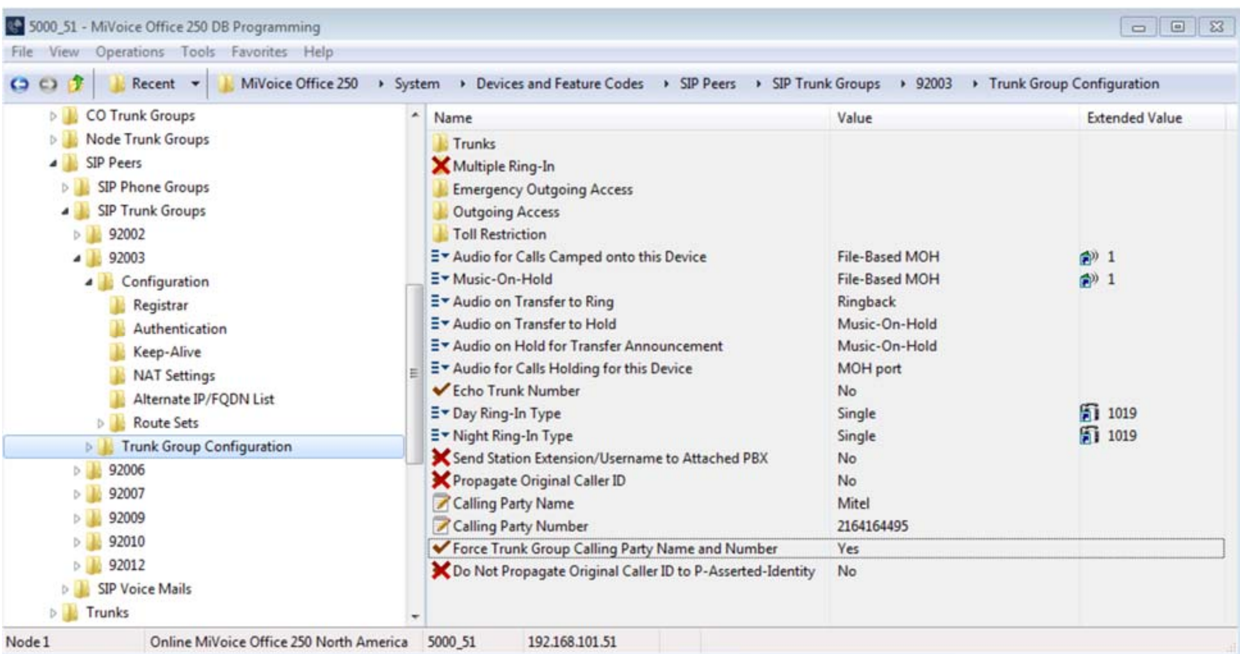

Figure 7 – Example of Trunk Group Configuration

Programming the Trunks in Trunk Group Configuration Folder

The number of SIP trunks that we create in here will appear as **Maximum Number of Calls** in the Configuration screen (see **Figure 6**).

Create the SIP peer trunks as follows:

- Navigate to System->Device and Feature Codes->SIP Peers->SIP Trunk Groups-><SIP Trunk group #>->Trunk Group Configuration->Trunks
- Right-click the right pane and the select **Create SIP Peer Trunk**. The Create SIP Peer Trunk dialog box appears (see **Figure 8**).
- Select the extension number you want to use for the item in the Starting Extension field. Choose a range that is recommended for your system. If using CSM it is recommended to use unique Trunk Group numbers in a multi-node environment;
- 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 will assign sequentially to the next available numbers.
- Click **OK**. See **Figure 8**. **Note**: The number of available SIP Trunks licenses restricts the number of SIP peer trunks.

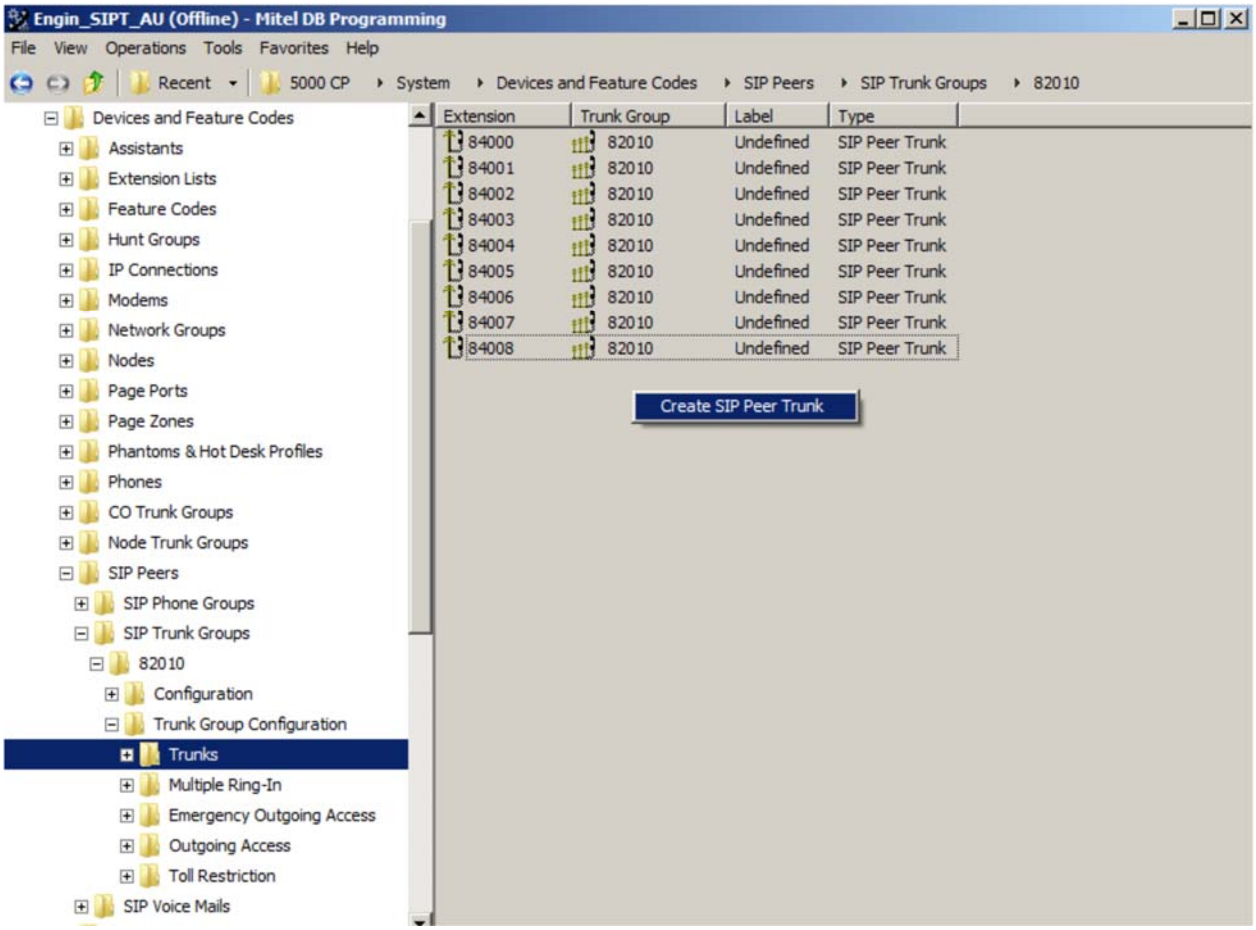

Figure 8 – Example of SIP Trunks creation

Phone Configurations

Some configuration settings need to be updating for the extensions. In our test environment, when Outgoing key is pressed on the phone, we wanted to direct outbound calls to the Fusion Connect Service Provider' SIP trunk. To do this:

- Navigate to System->Device and Feature Codes->Phones-><Phone's extension number>->Associated Extension
- In right hand pane, select **Outgoing Extension** and enter the number of SIP Trunk Group corresponding to Fusion Connect Service Provider' trunk. See **Figure 9** for an example.

You can configure a voicemail extension in the same screen.

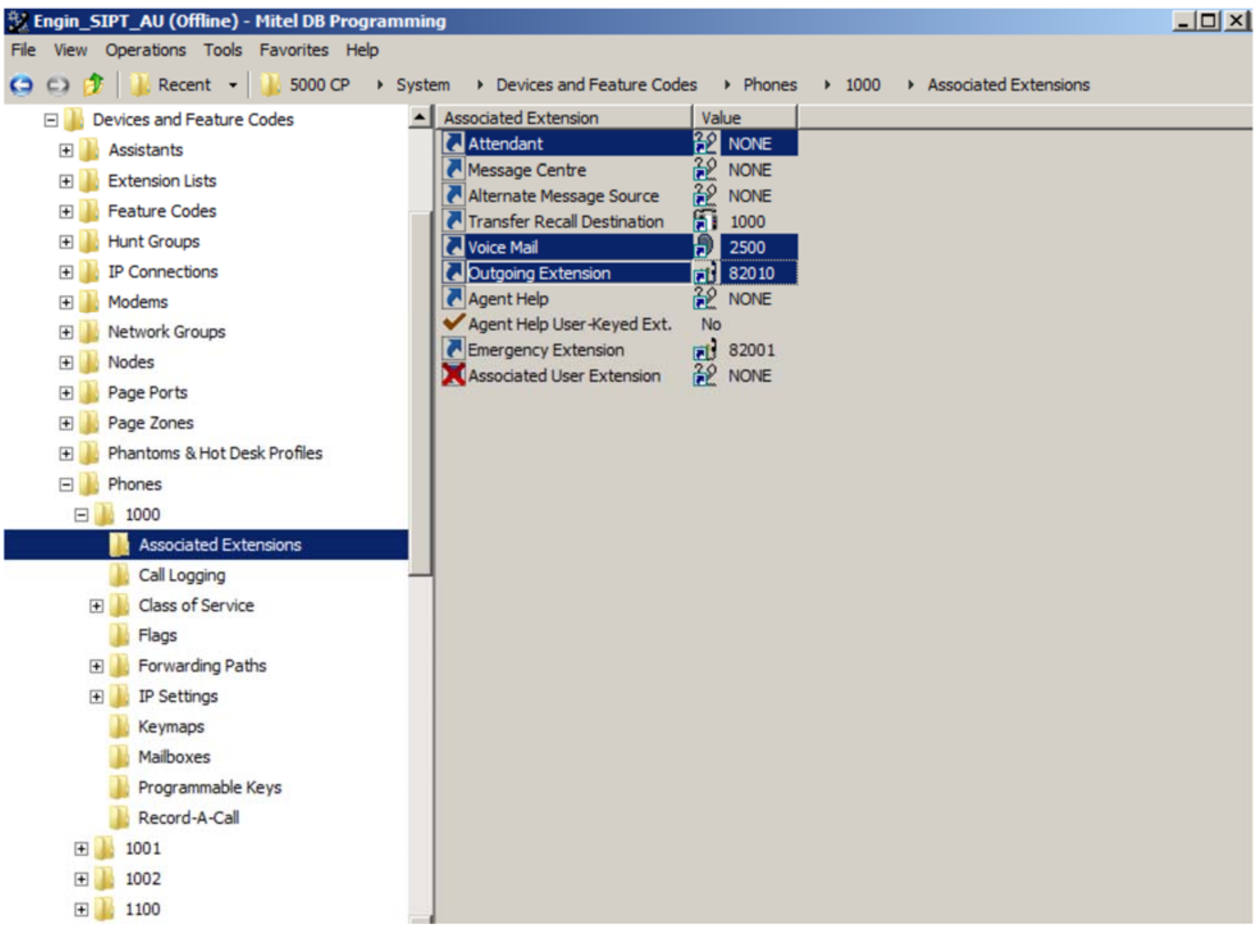

Figure 9 – Configure outgoing extension for the phone

When configuring call forwarding to a voicemail box, it is not enough to create the mailbox and assign it to the phone. You also have to define the Forwarding Path and assign it to the phone. To do this:

- Navigate to System->Phone Related Information->System Forwarding Paths

Define at least Forwarding Point 1 for the selected path. In the example in **Figure 10**, we defined extension 2500 as the forwarding point for path #1. Extension 2500 is the UVM (internal Voice Processor) pilot number. **NOTE**: For easier identification later on, we would recommend to enter a text in the Description field as shown in **Figure 10**.

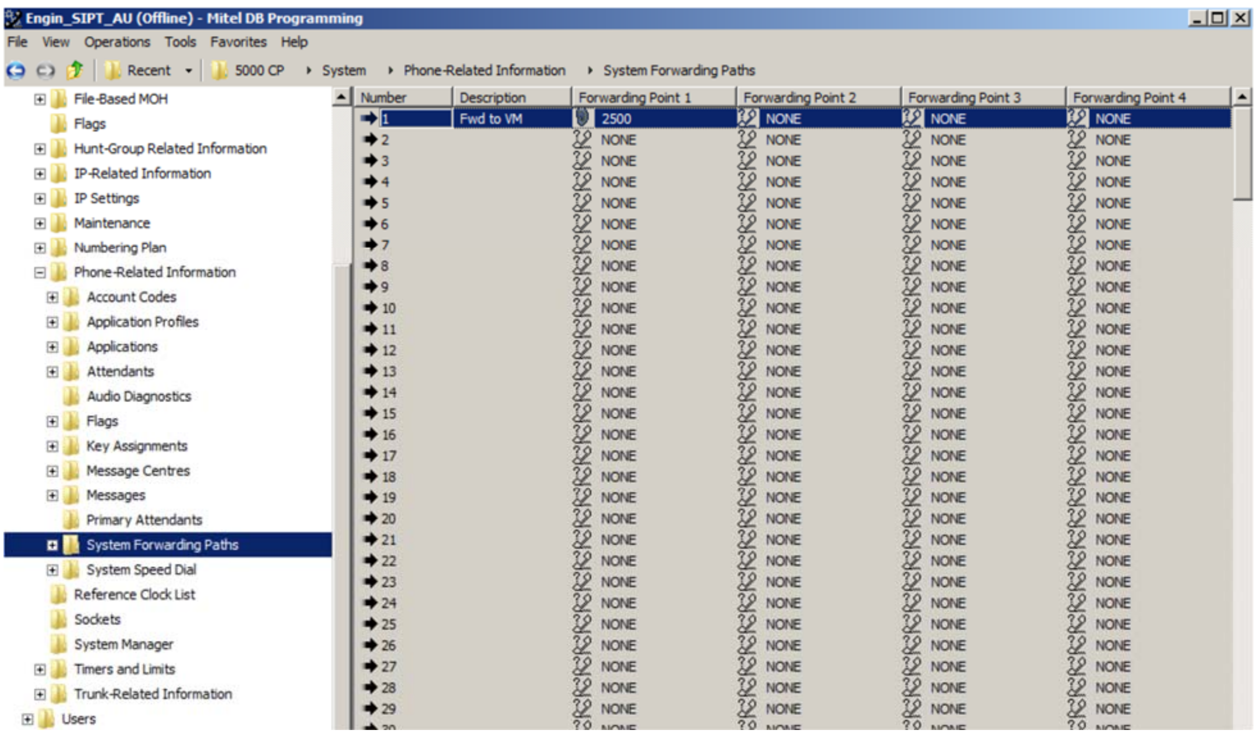

Figure 10 – Example of Forwarding Path definition

Now, when Forwarding Path #1 is defined, we can assign it to the phone:

- Navigate to System->Device and Feature Codes->Phones-><Phone’s extension number>->Forwarding Paths
- Right click in right hand pane and select **Add to Forwarding Paths List**
- Select the Forwarding Paths and click Next
- Select the required Forwarding Path’s number and click **Add Items** button
- Click Finish

NOTE: If you wish to forward unanswered internal calls to the defined Forwarding Point, set parameter **Fwd Call Type – IC Calls** to “**Yes**” as shown on **Figure 11**.

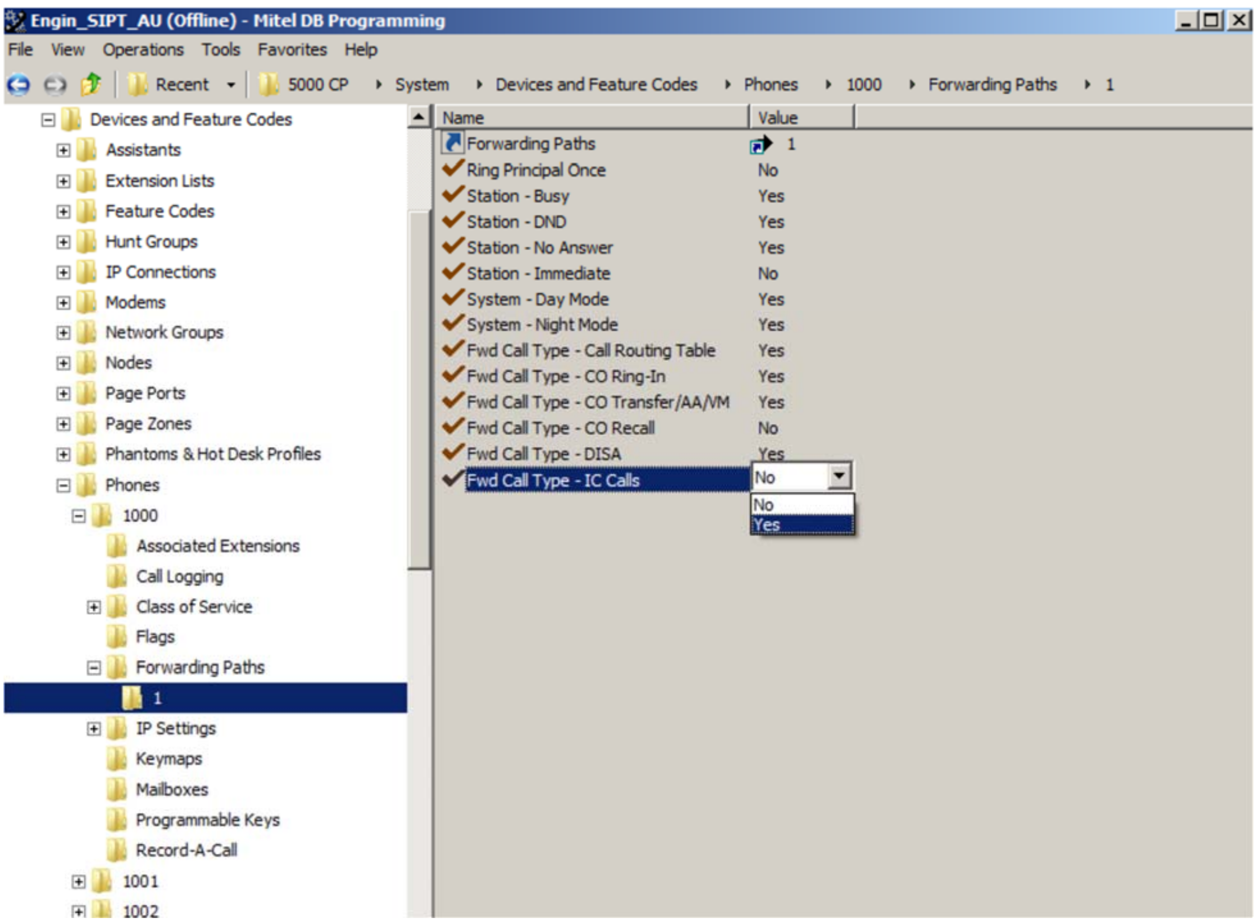

Figure 11 – Example of the properties for defined Forwarding Path

Call Configurations

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

By default, all IP devices have been placed in Call Configuration 1, which is programmable. You do not need to add SIP phones to Call Configurations, because these devices negotiate call configurations before establishing a connection. You can program up to 25 different Call Configurations.

Set **Audio Frames/IP Packet** to "**2**" which corresponds to the packet rate of 20ms.
DTMF Encoding setting - RFC 2833(preferred) & G711 were tested and either one can be used.
Speech Encoding Setting – G711(preferred) or G729 were tested and either one can be used.
Fax Detection Sensitivity – Default value of 1.
Fax Encoding Setting (Fax Transmission) – Set to G711 or T38

To view the list of IP phones which are currently assigned to the call configuration:

- Navigate to System->IP-Related Information->Call Configurations-><call configuration number>
- Click **Phones**

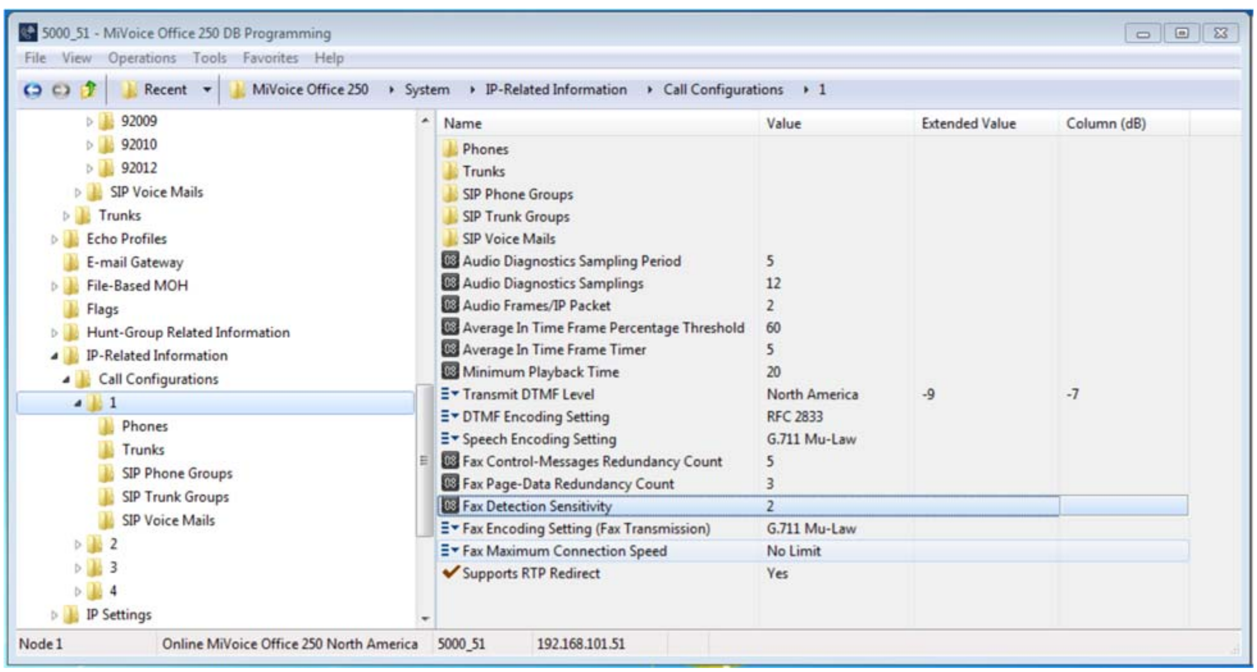

Figure 12 – Call Configuration Options

You can move extensions to this call configuration profile. To do this:

- Right click in right hand pane and select **Move to Phones List**
- Select the device type, e.g. 53xx and click Next
- Select extensions that you want to move and click **Move Items** button
- Click Finish

To view and move SIP Trunk Groups to the Call Configuration:

- Navigate to System->IP-Related Information->Call Configurations-><call configuration number>
- Click **SIP Trunk Groups**
- Right click in right hand pane and select **Move to SIP Trunk Groups List**
- Select the type, e.g. SIP Trunk Group and click Next
- Select the required trunk group that you want to move and click **Move Items** button
- Click Finish

Mitel Border Gateway Configuration Notes

This section explains how to configure Mitel Border Gateway (MBG).

Firstly, you need to identify or add "the" MiVoice Office 250 where the MBG will forward SIP messages to and then to configure the SIP trunk.

To do this:

- Login to the MBG and click Mitel Border Gateway.
- In the right pane, click the **Service Configuration** tab and then **ICP's** (see **Figure 13** for details).
- On the **ICP's** page ensure that the "MiVoice office 250 is configured. If needed, click the **Add ICP** link and add a new Mitel switch.
- Click the **Save** button when complete.

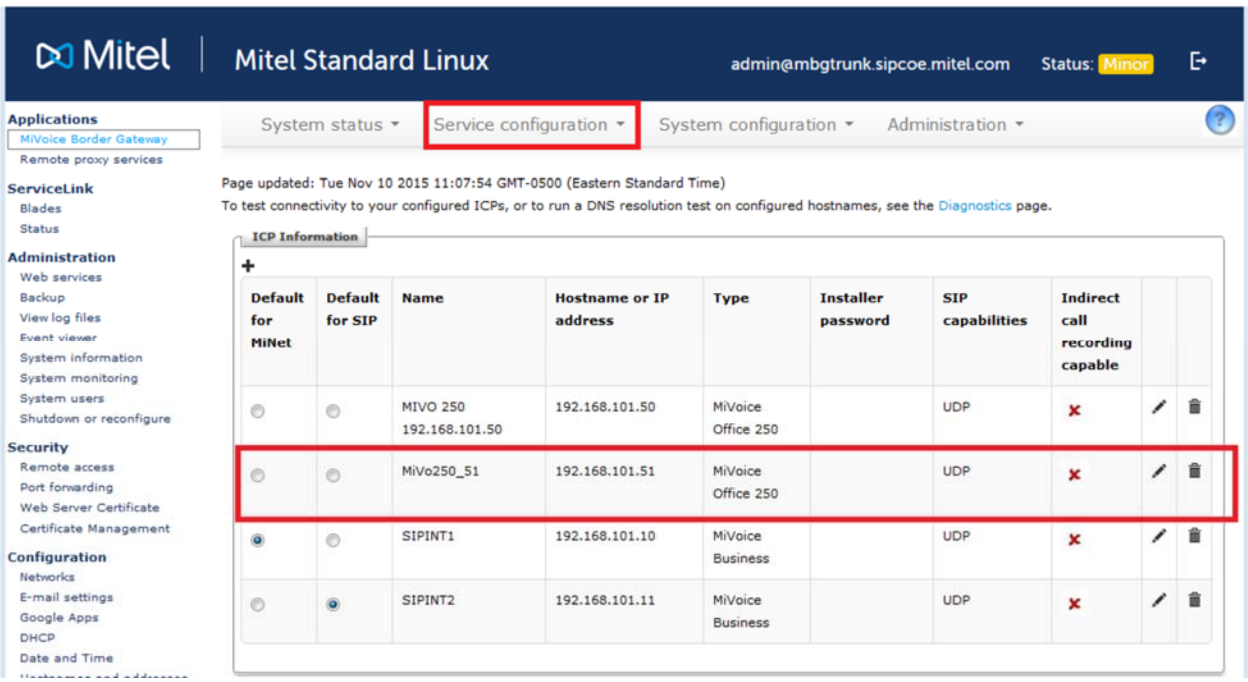

Figure 13 – ICP's Configuration page

Under the **System Configurations** tab, click on **Settings** and ensure that SIP options are sets as in **Figure 14**.

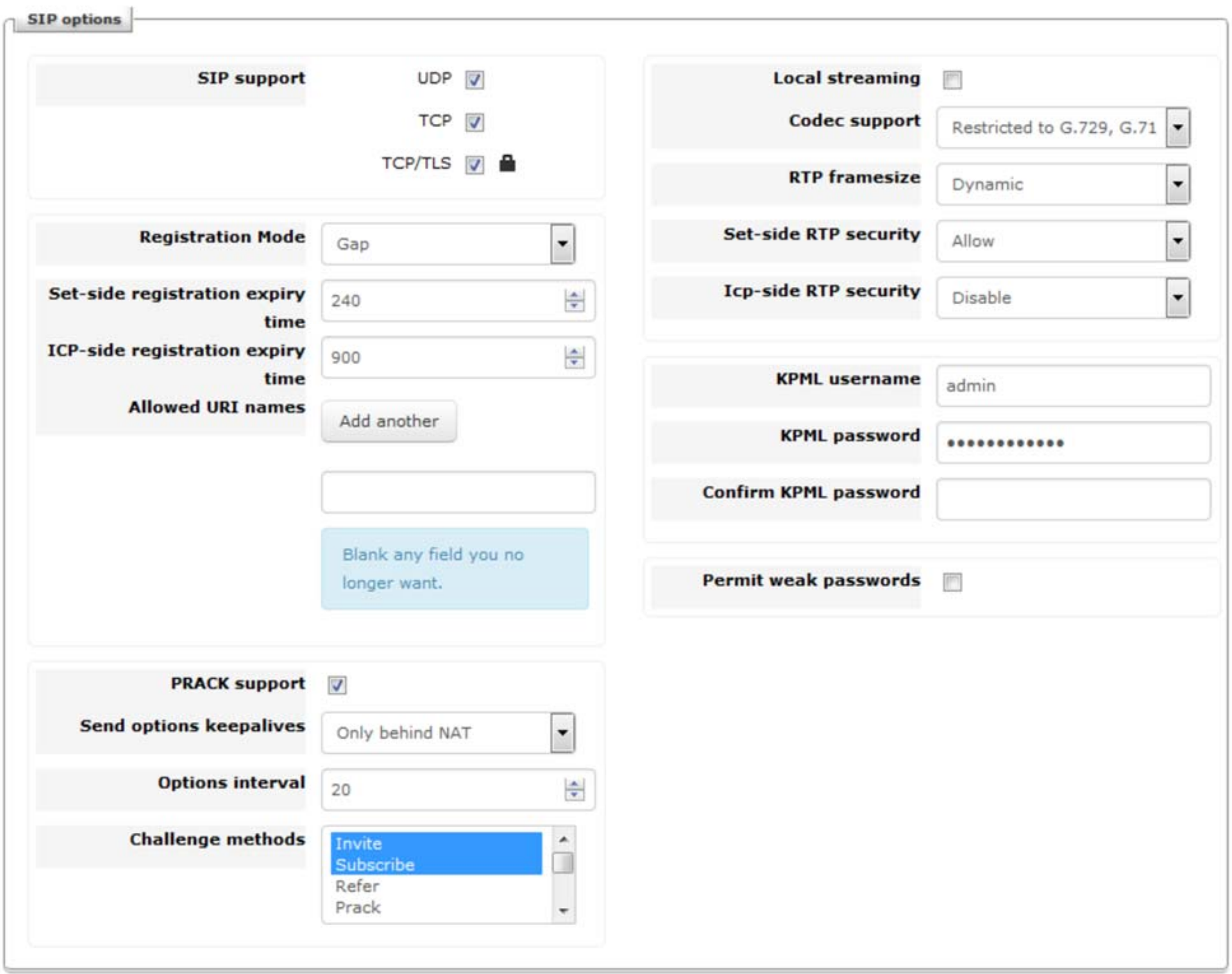

Figure 14 – Configuration – Settings – SIP Options

To add a new SIP trunk click on **SIP trunking** under the **Service configuration** tab and then click on the **+** sign to add a new SIP trunk.

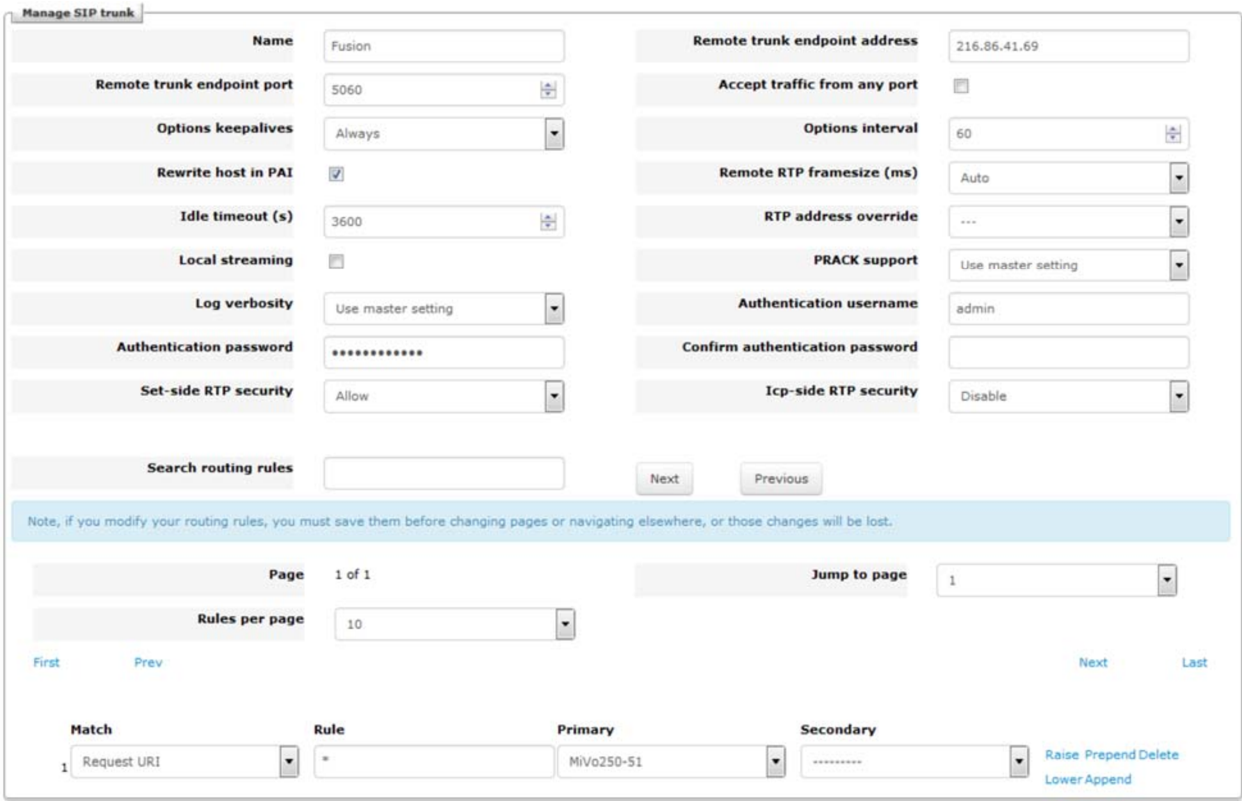

Figure 15 – SIP trunking configuration page

Enter the SIP trunk's details as shown in **Figure 15**:

Name – is the name you want to call the trunk
Remote trunk endpoint address – the public IP address of the provider's switch or gateway (this address should be given to you by the provider, e.g. Fusion Connect).
Remote trunk endpoint port – typically 5060
Options Keepalive – Set to Always
Options interaval – Set to 60 seconds in this case.
Remote RTP framesize (ms) – is the packetization rate you want to set on this trunk. This option is typically set to Auto.
RTP address override – Leave blank.
PRACK support – Leave this option at the default setting.
Routing rule one – it allows routing of any digits to the selected MiVoice Office 250 switch
The rest of the settings are optional and could be configured as required.

Click **Save** button.