Configure MiVO 6.1 with MBG 9.1 for use with the XO Communications SIP Trunking

NOVEMBER 2015
SIP COE 10-4940-00106
TECHNICAL CONFIGURATION NOTES
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Mitel Technical Configuration Notes:

Configure the MiVoice Office 250 6.1 for use with XO Communications
November 2015, 10-4940-00106_6

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Overview

This document provides a reference to Mitel Authorized Solutions providers for configuring the MiVoice Office 250 to connect to XO Communications. 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

<table>
<thead>
<tr>
<th>Version</th>
<th>Date</th>
<th>Reason</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>January 11, 2010</td>
<td>Initial Interop with Mitel 5000 CS-5200 and XO Communications</td>
</tr>
<tr>
<td>2</td>
<td>March 09, 2011</td>
<td>Retest with Mitel 5000 Ver. 4.0 Release 30</td>
</tr>
<tr>
<td>3</td>
<td>July 2013</td>
<td>Interop with MiVoice Office and XO Communications</td>
</tr>
<tr>
<td>4</td>
<td>July 2014</td>
<td>Documentation update</td>
</tr>
<tr>
<td>5</td>
<td>July 2014</td>
<td>Documentation update</td>
</tr>
<tr>
<td>6</td>
<td>November 2015</td>
<td>Refresh interop MiVO 250 Ver 6.1</td>
</tr>
</tbody>
</table>

Interop Status

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

![COMPATIBLE](image)

The most common certification which means XO Communications 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 XO Communications and the MiVoice Office.

<table>
<thead>
<tr>
<th>Manufacturer</th>
<th>Variant</th>
<th>Software Version</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mitel</td>
<td>MiVO 250</td>
<td>6.1</td>
</tr>
<tr>
<td>Mitel</td>
<td>MBG - Gateway</td>
<td>9.1.1.37</td>
</tr>
<tr>
<td>Mitel</td>
<td>Nupoint Voicemail</td>
<td>V17.0.0.24</td>
</tr>
<tr>
<td>Mitel</td>
<td>5320, 5330e, 5340 IP Sets</td>
<td>6.01.00.06</td>
</tr>
<tr>
<td>Mitel</td>
<td>6863i</td>
<td>3.3.1.7040</td>
</tr>
<tr>
<td>Sonus</td>
<td>SBC 9000</td>
<td>V07.03.01,F009</td>
</tr>
<tr>
<td>Broadsoft</td>
<td>Broadsoft AS</td>
<td>As of November 2015</td>
</tr>
</tbody>
</table>
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.

<table>
<thead>
<tr>
<th>Feature</th>
<th>Feature Description</th>
<th>Issues</th>
</tr>
</thead>
<tbody>
<tr>
<td>Basic Call</td>
<td>Making and receiving a call through XO Communications and their PSTN gateway, call holding, transferring, conferencing, busy calls, long calls durations, variable codec.</td>
<td>✔️</td>
</tr>
<tr>
<td>Automatic Call Distribution</td>
<td>Making calls to an ACD environment with RAD treatments, Interflow and Overflow call scenarios and DTMF detection.</td>
<td>✔️</td>
</tr>
<tr>
<td>NuPoint Voicemail</td>
<td>Terminating calls to a NuPoint voicemail boxes and DTMF detection.</td>
<td>✔️</td>
</tr>
<tr>
<td>Packetization</td>
<td>Forcing the MiVoice Office to stream RTP packets through its E2T card at different intervals, from 10ms to 60ms</td>
<td>✔️</td>
</tr>
<tr>
<td>Personal Ring Groups</td>
<td>Receiving calls through XO Communications and their PSTN gateway to a personal ring group. Also moving calls to/from the prime member and group members.</td>
<td>✔️</td>
</tr>
<tr>
<td>Fax</td>
<td>T.38 and G711Fax Calls</td>
<td>☑️</td>
</tr>
</tbody>
</table>

✔️ - 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 XO Communications is connected to the MiVoice Office.

<table>
<thead>
<tr>
<th>Packetization</th>
<th>Exclusively tested with stream RTP packets through its E2T card at 20ms (XO COMMUNICATIONS recommendation) instead of the Mitel testing range of 20ms to 30ms.</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Recommendation:</strong></td>
<td>Use XO COMMUNICATIONS 20ms recommended setting.</td>
</tr>
<tr>
<td>T.38 Fax</td>
<td>XO COMMUNICATIONS does not support T.38 Fax with multiple m-lines over SIP Trunks; however both G.711 and T.38 (single m-lines) were tested.</td>
</tr>
<tr>
<td><strong>Recommendation:</strong></td>
<td>Consult XO COMMUNICATIONS for future deployment.</td>
</tr>
</tbody>
</table>
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 XO Communications and MiVoice Office programming 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 Configuration Notes

The following steps show how to program a MiVoice Office to interconnect with XO Communications.

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 Programming

- The SIP signaling connection uses UDP on Port 5060
Licensing and Option Selection – SIP Licensing

Ensure that the MiVoice Office is equipped with enough SIP trunking licenses for the connection to XO Communications. This can be verified under the Software License form.

Figure 2 – License Selection
Creating and Configuring a SIP Peer Trunk Group

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

To create a SIP Trunk Group for XO Communications, right click in the right hand window panel of a SIP Trunk and then select “Create SIP Trunk Group”.

![Figure 3 – Examples of Created SIP Trunk Group](image)

Program the Configuration folder as described below:

- **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. You must add all IP addresses or FQDNs other than the primary IP/FQDN to the list for all calls to be successful.

- **IP Address**: Indicates the IP address of the SIP peer trunk group.

- **Port Number**: Indicates the port that the system listens on the system for SIP peer messages. The range is 0–65535.

- **Fully Qualified Domain Name**: Indicates the domain name of the SIP peer trunk group.

- **Call Configuration**: Clicking Call Configuration takes you to the Call Configuration folder (System\IP-Related Information\Call Configurations\<call configuration number>).

- **Operating State**: Indicates the operating state of the SIP peer.

- **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 handles ITU-T E.164 formatted phone numbers as part of the incoming SIP INVITE messages from the SIP peer.

Figure 4: Configured XO Communications SIP Trunk
Figure 5: Registration not required for XO Communications

Figure 6: Authentication not required for XO Communications
Figure 7: Basic Configuration folder provisioning for XO Communications
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 8: MBG ip address](image1)

![Figure 9: MBG Route Sets for XO Communications](image2)

**Programming the Trunk Group Configuration Folder**

To program the Trunk Group Configuration folder as per Figure 10

- XO Communications requires CPN = 2146355855
- Call Routing Table 2 used to associate and direct incoming calls to ip-phone sets (see Figure 11)

Figure 10: Trunk Group Configuration folder

Figure 11: Call Routing Table 3

Create the SIP peer trunks as follows:
- Double-click 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 94001–94999;
- 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.
- Click OK. For the XO Communications, 4 extensions were created. The number of SIP peer trunks is restricted by the number of available SIP Trunks licenses.

Figure 12: Trunk Extensions

**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.

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 11 was used for testing.
Figure 13: Call Configuration 12 Settings

Figure 14: Call Configuration Extensions
Figure 15: Call Configuration SIP Trunk Association

Figure 16: Call Configuration SIP Voice Mail Association
Programming Basic Users and IP-Phone Sets

2 basic users were created: 1049 and 1054

Figure 17: Basic Users

Figure 18: User Example
Figure 19: Associated Extensions Programming
DTMF Decoding Payload Setting

Figure 19: DTMF Decoding Payload
SIP Voice Mail Configuration (NuPoint)

MiVoice Office connected to NuPoint Voice Mail server 192.168.101.215:5061 and Call Configuration 11.

Figure 20: SIP Voice Mail
SIP Voice Mail Pilot (NuPoint)

MiVoice Office has 2502 as the SIP voice mail pilot number.

Figure 21: SIP Voice Mail Pilot
Mitel Border Gateway Setup

ICP Setup

To program an MCD into the MBG, click on ICP’s→Add an ICP.

Enter a name for the MCD.

Enter the IP address of the MCD and select the Type as MCD.

Figure 22: ICP setup
SIP Trunk Setup

Under the Services tab, click on SIP trunking and then "Add a SIP Trunk". Enter the SIP trunk’s details as shown in Figure 23:

Name – is the name of 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. Bell).

Local/Remote RTP framesize (ms) – is the packetization rate you want to set on this trunk

PRACK – Use master setting.

Routing rule one – it allows routing of any digits to the selected MiVoice Office

The rest of the settings are optional and could be configured if required. Click Save button.

Figure 23: Services - SIP Trunking setup
Mitel NuPoint Basic Setup

Line Groups Setup
Dot50_Line was created for use with MiVoice Office.

Network Elements Setup
5000_CP_Dot50 was created for use with MiVoice Office.
Voice Mailbox Setup

As an example, 1001 was created for use with MiVoice Office. The mailbox number would be the same as the extension.