

MiVoice MX-ONE Integration with Microsoft® Exchange Server 2013 UM

QUICK SETUP GUIDE



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1 INTRODUCTION

The MiVoice MX-ONE communication system is based on an open software and hardware environment, using standard servers with a Linux SUSE operating system. This open standards approach enables Mitel to offer our customers a choice and with this in mind we have worked together with Microsoft to ensure that MiVoice MX-ONE can be integrated with the latest Microsoft UC products.

1.1 GENERAL

MX-ONE 6.x version can interwork with third party UC products using standards-based protocols, such as SIP and CSTA V3/XML. Note however, that **the configuration described in this version of the document, is valid for MX-ONE 6.0 SP2 or later releases.**

Integration of MX-ONE 6.1 (and 6.0 and 5.0 SP4 or later) can be done with the Microsoft Exchange Server 2013 Unified Messaging (UM) as a complementary solution providing end user services like voice mail, Unified Messaging and auto attendant as well as system functionalities such as load balancing and fault tolerance.

Microsoft Partner Program has certified the integration between MX-ONE 5.0 SP4 and Microsoft Exchange Server 2013 Unified Messaging (UM) via a Direct SIP connection.

1.2 SCOPE OF THIS DOCUMENT

The intent of this guide is to describe the basic integration between the MiVoice MX-ONE and Microsoft Exchange Server 2013 Unified Messaging as well as describe the configuration needed and what features are available after the integration. The following sections describe the solution integration that has been certified through the Microsoft partner program and also the tests performed in Mitel's laboratory.

For a more technical description on how this integration is set-up, as well as tested features, we refer to the relevant CPI documentation for MX-ONE or please, go to the Microsoft Exchange Server 2013 product websites.



Note! Always check the latest products documentation.

2 SOLUTION DESCRIPTION

The integration of MX-ONE 6.x and Microsoft Exchange Server 2013 Unified Messaging described in this guide is achieved via Direct SIP.

Direct SIP that is specified by Microsoft means that a SIP trunk is used to connect MX-ONE 6.x and Microsoft Exchange Server 2013 Unified Messaging. Additionally, MX-ONE can be configured with TLS and SRTP when integrated with Exchange 2013 UM to provide security in the transport between the systems as well as load balancing and failover functionalities.

2.1 MIVOICE MX-ONE 6.X INTEGRATION WITH MICROSOFT EXCHANGE SERVER 2013 UM

The solution diagram below shows how MX-ONE is connected with Exchange 2013 UM.

In the validated scenario both Client Access and Mailbox role run in the same Exchange Server



Note! Microsoft Exchange Server 2013 architecture is different than the architecture in Exchange Server 2010, read Microsoft document “Voice Architecture Changes for more information.

[http://technet.microsoft.com/en-us/library/jj150516\(v=exchg.150\).aspx](http://technet.microsoft.com/en-us/library/jj150516(v=exchg.150).aspx)

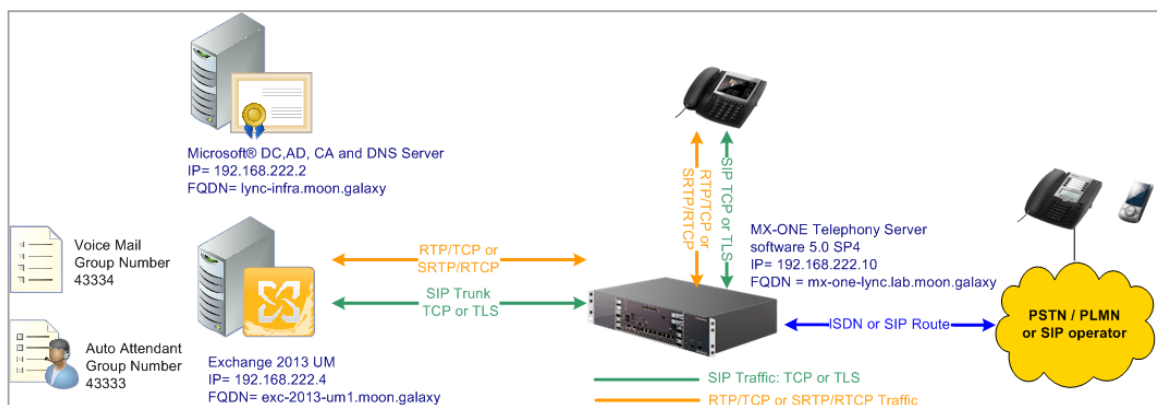


Figure 1 - MX-ONE 6.x integration with Microsoft Exchange Server 2013 UM

As described in Microsoft's documentation: "In the new model, the Client Access server running the Microsoft Exchange Unified Messaging Call Router service redirects Session Initialization Protocol (SIP) traffic that's generated from an incoming call to a Mailbox server. Then a media (Realtime Transport Protocol (RTP) or secure RTP (SRTP)) channel is established from the VoIP gateway or IP Private Branch eXchange (PBX) to the Mailbox server that hosts the user's mailbox."

In short, the Direct SIP integration works in the following way: When MX-ONE is configured to use TCP as transport, it calls to Microsoft Exchange 2013 UM by sending a SIP INVITE message to the 5060 port of Exchange Server. Then, Exchange Server sends 302 (Moved Temporarily) back to MX-ONE asking to send the INVITE on a different port (TCP: for example, 5065 or 5067). After the MX-ONE sends the INVITE to the new port, the call setup is executed and the call is established.

MX-ONE integrated with Microsoft Exchange Server 2013 Unified Messaging delivers the following end user features:

- Voice mail
- Auto Attendant
- Message waiting indication for MX-ONE terminals
- Outlook voice access

3 LICENSES

3.1 MIVoice MX-ONE

The MiVoice MX-ONE licenses needed for this integration are:

- SIP trunk licenses. At a minimum there is the need for one SIP trunk license (SIP route) per Microsoft Exchange 2013 UM server.
- SIP trunk channel licenses (port licenses in proportion to the maximum number of simultaneous Voice Mail messages that shall be supported).
- License for private network services is also required for the SIP route.



Note! Actual quantity of licenses will depend on the customer installation.

- An optional VoIP Encryption license is required if security (TLS/SRTP) is used.

Please, always check with your Mitel partner that your system has the correct licenses, before beginning the integration deployment.

3.2 MICROSOFT EXCHANGE 2013 UM

Microsoft licenses needed for this integration are not included as part of the scope of this guide. Please, contact Microsoft or a qualified Microsoft partner to obtain the proper license requirements for each component of the Microsoft Exchange 2013 UM solution.

4 MX-ONE INTEGRATION WITH EXCHANGE 2013 UM USING TCP

4.1 PREREQUISITES

4.1.1 MIVoice MX-ONE PREREQUISITES

Main components

MiVoice MX-ONE 6.0 SP2 (or later version) with the proper licenses.

At least the following MX-ONE components are required:

- MiVoice MX-ONE communications system
 - MX-ONE Service Node
 - MX-ONE Service Node 6.0 SP2 or later version
 - Supported media gateways with the latest compatible firmware with MX-ONE 6.0 SP2 or later version
 - MX-ONE Classic - 7U 19-inch chassis, using MGU boards or
 - MX-ONE Lite - 3U 19-inch chassis, using MGU board
 - MX-ONE 1U 19-inch chassis, using MGU board
 - MX-ONE Media Server (soft media gateway)

The following shall be configured:

- Trunk between MX-ONE and Exchange UM - SIP route.
- Message Waiting Indicator configuration in the system and in the phones that will use the service.
- Call list or Diversion for IP phones. These features are used to forward the call to the voice mail in case of no answer or busy.

The following MX-ONE type of devices can be used with Exchange 2013 UM:

- SIP – Mitel 6700i/6800i family or any device supporting baseline SIP. As the Exchange Server also supports SIP with Direct Media, MX-ONE gateway resources would not be needed for SIP devices. But, in order to guarantee interoperability with any 3rd party SIP terminal, the SIP route to Exchange UM can be setup as “forced gateway”. The effect is that SIP calls to the Exchange UM server will always transit via the MX-ONE media gateway (MGU) for a call setup and media.
- Non SIP – All non SIP devices calling into the Exchange UM server will transit via the MX-ONE Media GW (MGU based) for call setup and media. The following is the list of supported devices:
 - H.323 -MiVoice 4400 IP phones and Mitel 7400 IP phones (incl. Dialog 5446 Premium)
 - Digital phones: MiVoice 4200 series digital phones
 - Analog phones: MiVoice 4100 series analog phones
 - Mitel Cordless Phones: DT690, DT390, DT412, DT422, DT432
 - Mobiles devices (no MWI functionality) using MX-ONE’s Mobile extension service
 - External callers coming in via the MX-ONE public access, regardless of the type of terminal or network connection (SIP or TDM)

4.1.2 MICROSOFT EXCHANGE 2013 UM PREREQUISITES

This guide does not cover the Exchange 2013 UM installation, so our recommendation is that Microsoft Exchange 2013 UM shall be installed by a trained Microsoft engineer.

Before you start to install Microsoft Exchange 2013 Unified Messaging, please read the Microsoft Exchange 2013 documentation for a better understanding of the solution requirements.

The documentation can be found in the following links:

- Microsoft Exchange 2013 documentation
[http://technet.microsoft.com/en-us/library/bb124558\(v=exchg.150\).aspx](http://technet.microsoft.com/en-us/library/bb124558(v=exchg.150).aspx)
- Microsoft Exchange 2013 Unified Messaging
[http://technet.microsoft.com/en-us/library/jj150478\(v=exchg.150\).aspx](http://technet.microsoft.com/en-us/library/jj150478(v=exchg.150).aspx)

4.2 CONFIGURATION

In this configuration example, we have used the following information:

Direct SIP connection using TCP as transport

MX-ONE

- IP address: 192.168.222.10
- FQDN: mx-one-lync.lab.moon.galaxy
- Numbering Plan: 5 digits
- Access numbers (external destinations) for Voice Mail and Auto Attendant: 43334 and 43333 (could be the same number for VM and attendant, if wanted, but here different numbers are used).
- Route access codes: 43333, 43334
- Users IP extensions: 27000, 27001 and 27010.

Exchange UM

- IPv4: 192.168.222.4
- FQDN: exc-2013-um1.moon.galaxy
- Voice Mail Pilot identifier: 43334
- Auto Attendant: 43333

4.2.1 MIVoice MX-ONE CONFIGURATION

Voice Mail and Auto Attendant Numbers

The Voice Mail and the Auto Attendant numbers need to be initiated. In this example, the service number 43334 is used for Voice Mail and the service number 43333 is used for Auto Attendant.

Number Initiation:

```
number_initiate -numbertype ED -number 43333
```

```
number_initiate -numbertype ED -number 43334
```


Creating SIP trunk

The following commands shall be executed in MX-ONE to configure a SIP Trunk.

Basic Setup:

```
rocai:rou=55,sel=7110000000000010,sig=0111110000A0,traf=03151515,rm=4,serv=3110000001,bcap=000100;
```

Wanted options for the SIP route are: Forced gateway, and 9 sec timeout for 100 Trying.

Since SIP route profiles will be used, the var-parameter settings do not matter, but we still must enter RODAI, so we will set all var parameters 0, but the profile shall set Forced gateway and 9 s timeout.

```
rodai:rou=55,type=TL66,vari=00000000,varc=00000000,varo=00000000;
```

Please note that Message Waiting Indication number needs to be defined in the SIP route via mwinumber parameter as shown in the example.

SIP Route Setting:

```
sip_route -set -route 55 -profile ExchangeUM_TCP -uristring0 sip:?@192.168.222.4 -match 192.168.222.4 -accept REMOTE_IP -codecs PCMA,PCMU -mwinumber 43334
```

mwinumber is the Message Waiting Indication number

Route equipment and destination data:

#Node 1 (as in node x, as in TRU=x-1) for MX-ONE SIP access, in this case the IP address 192.168.222.10 is configured in the MX-ONE Service Node 1.

```
roeqi:rou=55,tru=1-1&&1-xx;
```

Define external destinations to the SIP route 55

```
roddi:rou=55,dest=43333,adc=0005000000000250000001010000,SRT=1;
```

```
roddi:rou=55,dest=43334,adc=0005000000000250000001010000,SRT=1;
```

User configuration to forward to Voice Mail

Any third party terminal registered in MX-ONE may subscribe on Message Waiting Indicator (MWI) according to RFC 3842.

The commands below enable a user to forward calls to Exchange Server voice mail.

The example shows how calls will be forwarded to Exchange 2013 UM Voice Mail number 43334 if a call is made to extension 27000 on no answer.

Call List Setup:

```
call_list -i -d 27000 --dest-number 27000 --position 1 --busy-position 2
```

```
call_list -i -d 27000 --dest-number 43334 --position 2 --ird-bypass true
```

The extensions can also use Call Diversion instead of the Call List.

Other Extension User configuration related to Voice Mail

For H.323 Dialog terminals: If there is no fixed key for Voice Mail on the terminal, a function key, 'Message Waiting' must be enabled in order to enable speed dial. The key is enabled in a common phone configuration file (for example d42x02-config.txt).

For SIP 6700i terminal family: In the common phone configuration file, aastra.cfg, set "sip line1 vmail: 43334" to enable speed dial to voice mail. Similarly for 6800i terminals.

4.2.2 MICROSOFT EXCHANGE 2013 UM

In order to setup the Exchange 2013 UM, please check Microsoft's documentation:

Deploy Exchange 2013 UM

[http://technet.microsoft.com/en-us/library/jj673564\(v=exchg.150\).aspx](http://technet.microsoft.com/en-us/library/jj673564(v=exchg.150).aspx)

After the installation of the Exchange 2013 UM roles, the following steps need to be executed to create the integration between MX-ONE and Exchange 2013 UM.

UM Dial Plan

A "UM Dial Plan" needs to be created in the Exchange UM.

Before you create a UM dial plan, please read the Microsoft's document, UM Dial Plans.

[http://technet.microsoft.com/en-us/library/bb125151\(v=exchg.150\).aspx](http://technet.microsoft.com/en-us/library/bb125151(v=exchg.150).aspx)

To create a New UM dial plan, please follow the step 1 in Microsoft's document, Create a UM dial plan.

[http://technet.microsoft.com/en-us/library/jj673564\(v=exchg.150\).aspx](http://technet.microsoft.com/en-us/library/jj673564(v=exchg.150).aspx)

Example:

- UM Dial Plan: Integration_MX-ONE
- Number of digits in extensions numbers: 5 - It needs to match the number of digits of the MX-ONE extensions.
- VoIP Security: Unsecured. In this example TCP is used.

The screen below shows the required configuration for the example.

new UM dial plan

Use UM dial plans to manage the UM features for a group of users who are enabled for voice mail.
[Learn more](#)

*Name:
Integration_MX-ONE

*Extension length (digits):
5

*Dial plan type:
Telephone extension

*VoIP security mode:
Unsecured

*Audio language:
English (United States)

*Country/Region code:
46

After you click Save, select this dial plan and click Edit to configure dial codes, Outlook Voice Access, voice mail settings, and dialing rules.

save cancel

UM IP gateway

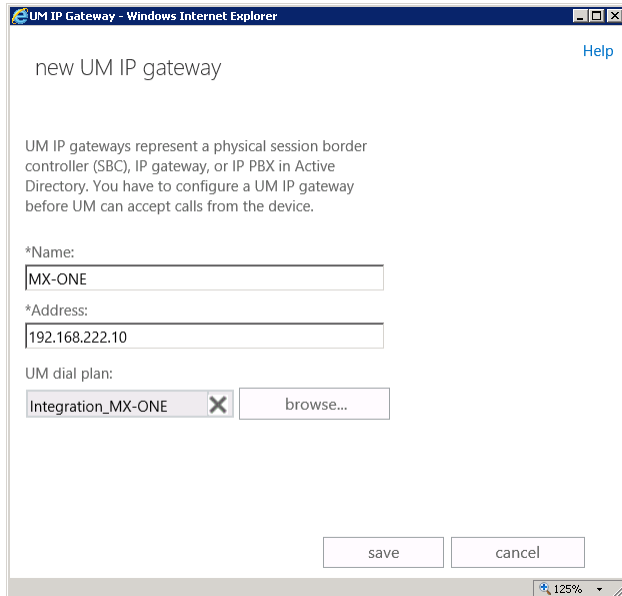
A UM IP gateway needs to be created in the Exchange UM.

To create a UM IP gateway, follow the step 2 in Microsoft's document:

[http://technet.microsoft.com/en-us/library/jj673564\(v=exchg.150\).aspx](http://technet.microsoft.com/en-us/library/jj673564(v=exchg.150).aspx)

Example:

- Name of the gateway: MX-ONE
- IP address: 192.168.222.10
- Dial Plan: It is the same one created previously.



The screenshot shows a web browser window titled "UM IP Gateway - Windows Internet Explorer". The page content is titled "new UM IP gateway" and includes a "Help" link. Below the title, there is explanatory text: "UM IP gateways represent a physical session border controller (SBC), IP gateway, or IP PBX in Active Directory. You have to configure a UM IP gateway before UM can accept calls from the device." The form contains three input fields: "*Name:" with the value "MX-ONE", "*Address:" with the value "192.168.222.10", and "UM dial plan:" with a dropdown menu showing "Integration_MX-ONE" and a "browse..." button. At the bottom of the form are "save" and "cancel" buttons. The browser's status bar at the bottom right shows "125%" zoom.

UM Hunt Group

A Hunt group shall be created to the voice mail.

To create a UM Hunt Group, follow the steps in Microsoft's document:

<https://technet.microsoft.com/en-us/library/aa997679.aspx>

Example:

- Associated UM IP Gateway: MX-ONE
- Name: Voice_Mail_43334
- Dial Plan: Integration_MX-ONE
- Pilot identifier: 43334. It must be the same number that was previously created in MX-ONE.

The screenshot shows a web browser window titled 'UM Hunt Group - Windows Internet Explorer'. The page has a blue header bar with the title and a 'Help' link. The main content area is titled 'new UM hunt group'. Below the title, there is a paragraph explaining that UM hunt groups determine which UM IP gateways to accept calls from for the users of this UM dial plan. The 'UM dial plan' is set to 'Integration_MX-ONE'. There are three input fields: '*Name:' with the value 'Voice_Mail_43334', '*UM IP gateway:' with a dropdown menu showing 'MX-ONE' and a 'browse...' button, and 'Pilot identifier:' with the value '43334'. At the bottom right, there are 'save' and 'cancel' buttons. The browser's status bar at the bottom shows a zoom level of 125%.

UM Mailbox Policies

A new UM mailbox policy can be created or the default policy can be used. Please, follow step 4 in Microsoft's document:

[http://technet.microsoft.com/en-us/library/jj673564\(v=exchg.150\).aspx](http://technet.microsoft.com/en-us/library/jj673564(v=exchg.150).aspx)

UM Auto Attendant

To setup the Exchange 2013 UM Auto Attendant, please follow the steps below:

Create an UM Auto Attendant

To create an UM Auto Attendant, please follow the step 5 in Microsoft's document:

[http://technet.microsoft.com/en-us/library/jj673564\(v=exchg.150\).aspx](http://technet.microsoft.com/en-us/library/jj673564(v=exchg.150).aspx)

Example:

- Name: MX-ONE Auto Attendant
- Dial Plan: Integration_MX-ONE
- Pilot identifier: 43333. It needs to be the same number that was previously created in MX-ONE.

The screenshot shows a web browser window titled 'UM Auto Attendant - Windows Internet Explorer'. The page content is as follows:

- Page title: new UM auto attendant
- Help link: [Help](#)
- Text: UM auto attendants allow you to automatically answer and route calls for your organization.
- Text: UM dial plan: Integration_MX-ONE
- Text: *Name:
- Form elements:
 - ☒ Create this auto attendant as enabled
 - ☐ Set the auto attendant to respond to voice commands
- Text: Access numbers:
- Form elements:
 - Icon:
 - Text:
 - Input field:
 - Highlighted number: 43333
- Information icon and text: After you click Save, select this UM auto attendant and click Details to set greetings,
- Buttons:
- Zoom: 125%

Enable the Unified Messaging

To enable Voice Mail for a user follow Microsoft's document:

Enable a User for Voice Mail

[http://technet.microsoft.com/en-us/library/bb124147\(v=exchg.150\).aspx](http://technet.microsoft.com/en-us/library/bb124147(v=exchg.150).aspx)

5 MIVOICE MX-ONE INTEGRATION WITH EXCHANGE 2013 UM USING TLS

5.1 PREREQUISITES

5.1.1 MIVOICE MX-ONE PREREQUISITES

Main components

MiVoice MX-ONE 6.0 SP2 or later version with the proper licenses.

At least the following MX-ONE components are required:

- MiVoice MX-ONE communications system
 - MX-ONE Service Node
 - MX-ONE Service Node 6.0 SP2 or later version
 - Supported media gateways with the latest compatible firmware with MX-ONE 6.0 SP2 or later version
 - MX-ONE Classic - 7U 19-inch chassis, using MGU boards or
 - MX-ONE Lite - 3U 19-inch chassis, using MGU board
 - MX-ONE 1U 19-inch chassis, using MGU board
 - MX-ONE Media Server (soft media gateway)
 - Licenses
 - All licenses described in the item 3.1 MX-ONE Licenses
 - VoIP Encryption license is required (TLS/SRTP) as TLS and SRTP will be used.

The following shall be configured:

- Trunk between MX-ONE and Exchange UM – SIP route configured with TLS.
- Two access numbers (external destinations) to be used as Pilot numbers (groups) in Exchange UM.
- Message Waiting Indicator configuration in the system and in the phones that will use the service.
- Call list or Diversion for IP phones. These features are used to forward the call to the voice mail in case of no answer or busy.

5.1.2 MICROSOFT EXCHANGE 2013 UM PREREQUISITES

This guide does not cover the Exchange 2013 UM installation. Our recommendation is that Microsoft Exchange 2013 UM shall be installed by a trained Microsoft engineer.

Before you start to install Microsoft Exchange 2013 Unified Messaging server role, please read the Microsoft Exchange 2013 documentation for a better understanding of the solution requirements, the documentation can be found in the following links:

- Microsoft Exchange 2013 documentation
[http://technet.microsoft.com/en-us/library/bb124558\(v=exchg.150\).aspx](http://technet.microsoft.com/en-us/library/bb124558(v=exchg.150).aspx)
- Microsoft Exchange 2013 Unified Messaging
[http://technet.microsoft.com/en-us/library/jj150478\(v=exchg.150\).aspx](http://technet.microsoft.com/en-us/library/jj150478(v=exchg.150).aspx)

5.2 CONFIGURATION

In this configuration example, we used the following:

Direct SIP connection using TLS as transport

MX-ONE

- IP address: 192.168.222.10
- FQDN: mx-one-lync.lab.moon.galaxy
- Numbering Plan: 5 digits
- Access numbers for Voice Mail and Auto Attendant: 43334 for VM and 43333 for attendant.
- Route access code: 043
- Users IP extensions: 27000, 27001 and 27010.

Exchange UM

- IPv4: 192.168.222.4
- FQDN: exc-2013-um1.moon.galaxy
- Voice Mail Pilot identifier, Hunt Group: 43334
- Auto Attendant: 43333

Certificate:

To use TLS between MX-ONE and Exchange 2013 UM a certificate must be created.

The common Microsoft Enterprise CA used for signing server certificates for Mediation Server and Exchange 2013 UM is assumed to be used to create a server certificate for MX-ONE as well.

A server certificate is signed to the FQDN (Fully Qualified Domain Name) of MX-ONE Service Node.

5.2.1 CREATE A CERTIFICATE

When using security, an appropriate certificate needs to be installed in MX-ONE as well as the encryption licenses. Please, check Certificate Management on MX-ONE CPI documentation in case you need more details regarding certificates.

5.2.2 IMPORT THE CERTIFICATE TO MIVOICE MX-ONE SERVICE NODE

Import the server certificate, in the example, mx-one-lync.lab.moon.galaxy.pfx to MX-ONE Service Node accessing the Exchange 2013 UM.

On the access Server, for example, MX-ONE Service Node 1 run the command:

Install certificate in MX-ONE Service Node

mxone_certificate, with the certificate mx-one-lync.lab.moon.galaxy.pfx

5.2.3 MIVOICE MX-ONE CONFIGURATION:

Creating SIP trunk with TLS

The following commands shall be executed in MX-ONE to configure a SIP Trunk with TLS, the others commands are the same as in a TCP configuration.

SIP Route settings for TLS

```
sip_route -set -route 55 -profile ExchangeUM_TLS_SRTP -uristring0 sip:?@192.168.222.4 -match  
192.168.222.4 -accept REMOTE_IP -codecs PCMA,PCMU -mwinumber 43334
```

mwinumber is the Message Waiting Indication number



Note! -accept REMOTE_IP will match the IP address sent in the IPv4 source IP header.

Enable Media Encryption in the route:

```
media_encryption_enable -type route
```

5.2.4 MICROSOFT EXCHANGE 2013 UM

In order to setup Exchange 2013 UM to use TLS, please follow Microsoft's documentation.

[http://technet.microsoft.com/en-us/library/jj150478\(v=exchg.150\).aspx](http://technet.microsoft.com/en-us/library/jj150478(v=exchg.150).aspx)

6 LOAD BALANCING AND FAILOVER BETWEEN MIVOICE MX-ONE AND TWO EXCHANGE SERVERS

6.1 LOAD BALANCING

MiVoice MX-ONE 6.x supports load balancing when connected with more than one Exchange Server UM. To be able to use such a scenario, the Microsoft DNS Load Balancing functionality is used.

MX-ONE 6.x supports DNS SRV and multiple A record query where a list with multiple entries can be used. When properly configured, MX-ONE will attempt to send INVITE to the entries in the list until the call is successful. No answer or 503 Service Unavailable will trigger MX-ONE to try the next entry.

For more details, check MX-ONE SIP Route command description in CPI or sip_route -help, parameter remotepoint.

6.2 FAILOVER

The failover functionality also requires Microsoft DNS Load Balancing functionality. When integrating MX-ONE and Exchange UM, the same configuration is valid for both failover and load balancing.

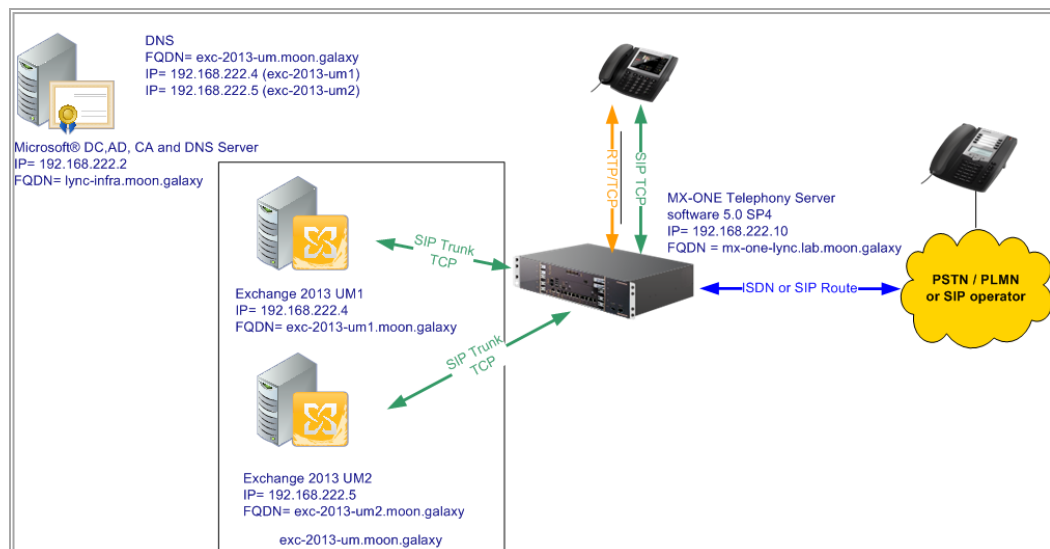
In a scenario where 2 Exchange UM servers are used and one of the servers is unavailable, the first call will be attempted to set up to the first server, but it will be redirected after a few seconds and answered. Then subsequent calls will be redirected and answered in the second Exchange UM.

The reason why it takes some seconds before getting answer is that the INVITE is sent to the first server, then the system waits 4 seconds for an answer. If no answer is received, the host is grey-listed for 32 seconds and an INVITE is sent to the second server.

For more details, check MX-ONE SIP Route command description in the CPI or sip_route -help, parameter remotepoint.

6.3 LOAD BALANCING AND FAILOVER SCENARIO

The figure below shows the validated setup:



6.3.1 CONFIGURATION

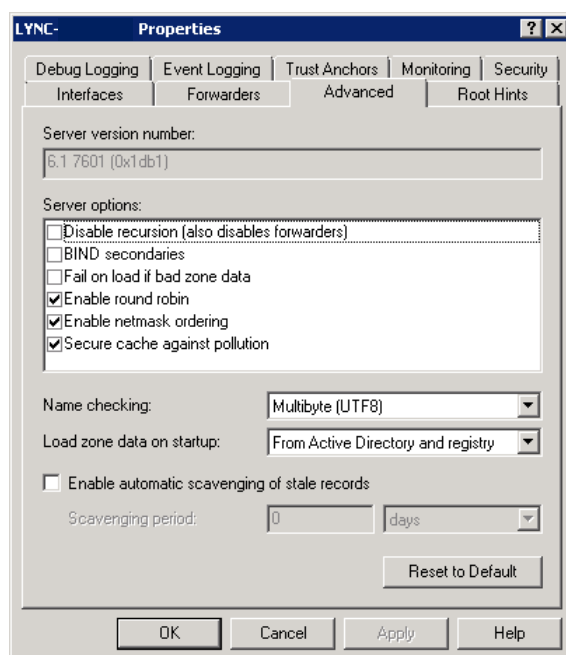
DNS Setup

Microsoft environment needs to be configured to support Round Robin as described in the TechNet article “Configure DNS for Load Balancing”.

Please, see the link below, item “To enable round robin for Windows Server”.

<http://technet.microsoft.com/en-us/library/gg398251.aspx>

The figure below shows the Round Robin option enabled.



DNS SRV setup

Go to DNS Manager Tool and create a pool entry. After that, add a DNS SRV record to each Exchange UM Server that participates in the DNS Load Balancing. In the following example the FQDN pool name is exc-2013-um.

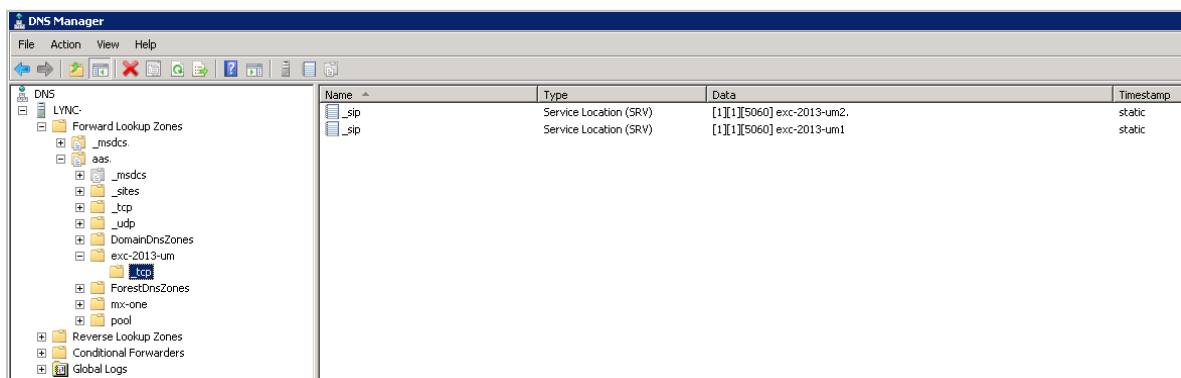
The values below needs to be configured in the DNS SRV record for each Exchange UM that is part of the pool.

DNS SRV	Values
Service:	_sip
Protocol:	_tcp
Priority:	1
Weight:	1
Port Number:	5060
Host offering this service:	exc-2013-um1 exc-2013-um2

Check the following Microsoft article for more details.

<http://technet.microsoft.com/en-us/library/gg398680>

The figure below shows the exc-2013-um pool after the SRV Records configuration:



The screenshot shows the DNS Manager console with the 'exc-2013-um' zone selected. The right pane displays two SRV records:

Name	Type	Data	Timestamp
_sip	Service Location (SRV)	[1][1][5060] exc-2013-um2.	static
_sip	Service Location (SRV)	[1][1][5060] exc-2013-um1	static

Test DNS SRV record setup

Using the Windows command nslookup, test the configuration:

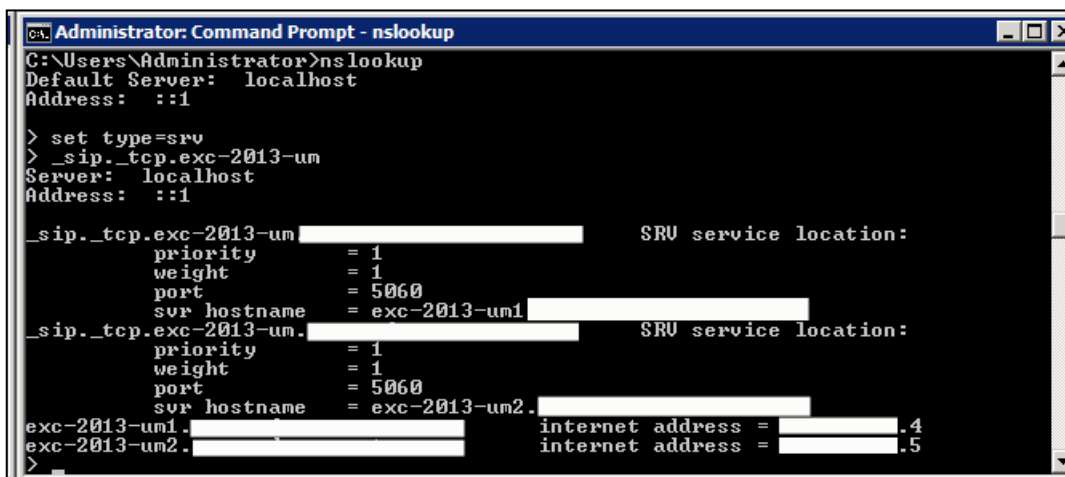
```
nslookup
```

```
set type=srv
```

```
_sip._tcp.exc-2013-um
```

The expected result is presented in the 2 next screens. Please note the domain name and the IP addresses are just partially presented.

The first query the DNS replies with exc-2013-um1.

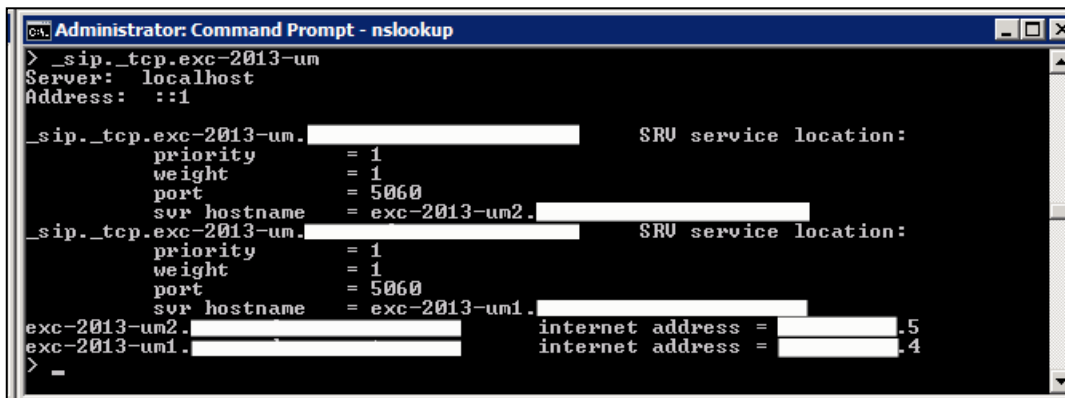


```
Administrator: Command Prompt - nslookup
C:\Users\Administrator>nslookup
Default Server: localhost
Address: ::1

> set type=srv
> _sip._tcp.exc-2013-um
Server: localhost
Address: ::1

_sip._tcp.exc-2013-um.           SRV service location:
    priority = 1
    weight   = 1
    port     = 5060
    svr hostname = exc-2013-um1.
_sip._tcp.exc-2013-um.           SRV service location:
    priority = 1
    weight   = 1
    port     = 5060
    svr hostname = exc-2013-um2.
exc-2013-um1.                    internet address = .4
exc-2013-um2.                    internet address = .5
>
```

The second query the DNS replies with exc-2013-um2.



```
Administrator: Command Prompt - nslookup
> _sip._tcp.exc-2013-um
Server: localhost
Address: ::1

_sip._tcp.exc-2013-um.           SRV service location:
    priority = 1
    weight   = 1
    port     = 5060
    svr hostname = exc-2013-um2.
_sip._tcp.exc-2013-um.           SRV service location:
    priority = 1
    weight   = 1
    port     = 5060
    svr hostname = exc-2013-um1.
exc-2013-um2.                    internet address = .5
exc-2013-um1.                    internet address = .4
>
```

Creating MiVoice MX-ONE SIP trunk

The MX-ONE SIP Route Outbound setting needs to be configured with the FQDN pool name that is used to solve the Exchange UM Servers that are part of the “load balancing cluster”. Please, note that `remoteport` should be configured equal 0. This is needed by MX-ONE in order to use the DNS SRV option.

Outbound Setting:

```
sip_route -set -route 55 -uristring0 sip:?@exc-2013-um.moon.galaxy -remoteport 0 -protocol tcp -codecs PCMA,PCMU -mwinumber 43334
```

mwinumber is the Message Waiting Indication number

Please note that Exchange 2013 UM IP addresses needs to be defined in the parameter **match**, as shown in the example.

Inbound Setting:

```
ip_route -set -route 55 -accept REMOTE_IP -match 192.168.222.4,192.168.222.5
```

match = Exchange 2013 UM IP addresses

7 HOW TO TEST THE INTEGRATION

To execute the integration test, the configuration in both sides shall be ready.

7.1 BASIC TESTS

1. Dial the pilot number from a phone extension that is NOT enabled for Unified Messaging and logon to a user's mailbox.
Confirm hearing the prompt: "<Microsoft Exchange Earcon>. To access your mailbox, enter your extension..."
2. Navigate mailbox using the Voice User Interface (VUI).
3. Navigate mailbox using the Telephony User Interface (TUI).
4. Dial user extension and leave a voicemail.
 - Dial user extension and leave a voicemail from an internal extension.
Confirm that the Active Directory name of the calling party is displayed in the sender field of the voicemail message.
 - Dial user extension and leave a voicemail from an external phone.
Confirm that the correct phone number of the calling party is displayed in the sender field of the voicemail message.
5. Dial Auto Attendant (AA).
Dial the extension for the AA and confirm that the AA answers the call.
6. Call Transfer by Directory Search.
 - Call Transfer by Directory Search and have the called party answer.
Confirm that the correct called party answers the phone.
 - Call Transfer by Directory Search when the called party's phone is busy.
Confirm that the call is routed to the called party's voicemail.
 - Call Transfer by Directory Search when the called party does not answer.
Confirm that the call is routed to the called party's voicemail.
 - Setup an invalid extension number for a particular user.
Call Transfer by Directory Search to this user. Confirm that the number is reported as invalid.
7. Outlook Web Access (OWA) Play-On-Phone Feature.
 - Listen to voicemail using OWA's Play-On-Phone feature to a user's extension.
Listen to voicemail using OWA's Play-On-Phone feature to an external number.
8. Configure a button on the phone of a UM-enabled user to forward the user to the pilot number.
Press the voicemail button.
Confirm that you are sent to the prompt: "<Microsoft Exchange UM Earcon>. <User>. Please enter your pin and press the pound key."
9. MWI.
Ensure that a UM-enabled user's mailbox does not have any new voice mails.
 - Dial the user's extension and leave a voicemail. Confirm that the MWI lamp on the phone lights up.
 - Mark the voice mail email as read in OWA. Confirm that the MWI lamp on the phone turns off.

10. Load balancing

- Open the Wireshark tool and configure it to collect SIP packets.

Dial several times to the voice mail number from an SIP extension. Use the Wireshark tool to analyze the SIP packet in order to verify that the load balancing is working properly.

11. Failover

- Disconnect the Ethernet cable of Exchange 2013 UM 1 to simulate a failure.

Dial several times to the voice mail number from an SIP extension. Check that the calls are answered in the Exchange 2013 UM 2.

8 REVISION HISTORY

Document Version	Comments	Date
Rev A	First version	January 2014
Rev B	Rebranding	May 2015
Rev C	Change of installation and certificate handling command; mxone_certificate, and updates on MX-ONE version info.	2015-10-27
Rev C1	Correction due to TR 113453. Configuration changed since 6.0 SP2. SIP route profile used. Trademark text update also done.	2016-03-03
Rev C2	Further correction due to TR 113453, comments from Stefan Andersson.	2016-03-04
Rev C3	Further correction indirectly due to TR 113453, comments from Niklas Windahl and others.	2016-03-18
Rev C4	Included TTL value 0 in the DNS SRV entries.	2019-06-14