Mitel Technical Configuration Notes- HO3111

March 27, 2019

Configure MiVoice Business 9.0 SP1 for use with OpenIP SIP Trunking

Description: This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel MiVB to connect to Service Provider OpenIP SIP Trunking.

Environment:
(VMware Environment)

MiVB : 9.0.1.22

MiCollab :

- ⇨ MSL : 10.6.13.0
- ⇨ MBG : 10.1.0.250
- ⇨ AWV : 8.1.1.8
- ⇨ MiCollab Client Service : 8.1.1.13
- ⇨ MiCollab Client Deployment : 8.1.1.1
- ⇨ NuPoint Unified Messaging : 19.1.1.4

MBG :

- ⇨ MSL : 10.6.13.0
- ⇨ MBG : 10.1.0.250

NOTICE

TRADEMARKS

Mitel Technical Configuration Notes – Configure MiVoice Business 9.0 SP1 for use with OpenIP SIP Trunking

Mitel Configuration Guide HO3111

March 2019

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Overview

This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel MiVB to connect to Service Provider OpenIP 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	2019/02	Initial Interop with Mitel MiVB 9.0 and Service Provider OpenIP SIP trunk

Interop Status

The Interop of Service Provider OpenIP SIP Trunking has been given a Certification status. This service provider or trunking device will be included in the Mitel Interoperability Reference Guide (IRG). The status Service Provider OpenIP SIP Trunking achieved is:

COMPATIBLE	The most common certification which means Service Provider OpenIP SIP Trunking has been tested and/or validated by the Mitel Third-Party Interop Team. Mitel 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 Service Provider OpenIP SIP Trunking and the MiVB.

Note – Although this testing was performed on the below tested variants, the scope of this testing can be extended to other product variants that work with the same firmware. The list of components for which this testing can be considered applicable is given in the "Additional Applicable Variants" column of the following table –

Mark it Not Applicable (NA) in case there are no specific models to be added under Additional Applicable Variants

Manufacturer	Tested Variants	Software Version	Additional Applicable Variants
Mitel	MiVB – Mxe Platform	9.0 SP1	9.0.1.22
Mitel	MBG – SIP Trunking & Teleworker	10.1.0.250	NA
Mitel	MiCollab Client Service	8.1.1.13	NA
Mitel	NuPoint Unified Messaging	19.1.1.4	NA
Mitel	69XX MiNET Rev : A18	01.04.00.080	NA
Service Provider			

Tested Features

This is an overview of the features tested during the Interop test cycle and not a detailed view of the test cases.

Feature	Feature Description	Issues
Basic Call	Making and receiving a call through Service Provider OpenIP and their PSTN gateway, call holding, transferring, conferencing, busy calls, long calls durations, variable codec.	⚠ - Issues found
Automatic Call Distribution	Making calls to an ACD environment with RAD treatments, Interflow and Overflow call scenarios and DTMF detection.	☑ - No issues found
NuPoint Voicemail	Terminating calls to a NuPoint voicemail box and DTMF detection.	☑ - No issues found
Packetization	Forcing the Mitel MiVB to stream RTP packets through its E2T card at different intervals, from 10ms to 90ms	☑ - No issues found (tested from 10ms to 40ms)
Personal Ring Groups	Receiving calls through Service Provider OpenIP and their PSTN gateway to a personal ring group. Also moving calls to/from the prime member and group members.	☑ - No issues found
External Hot Desking	Receiving calls through Service Provider OpenIP and their PSTN gateway to PRG with EHDU . Including moving calls to/from the prime member of the PRG with the EHDU. Also placing calls from the EHDU and using mid call features with EHDU.	☑ - No issues found
Teleworker	Making and receiving a call via Service Provider OpenIP and their PSTN gateway to and from Teleworker extensions.	☑ - No issues found
Video	Making and receiving a call through Service Provider OpenIP with video capable devices.	NA
Fax	T.38 and G711 Fax Calls	☑ - No issues found (T38 not supported)

☑ - No issues found ✗ - Issues found, cannot recommend to use ⚠ - Issues found

Device Limitations and Known Issues

This is a list of problems or unsupported features when Service Provider OpenIP SIP Trunking is connected to the MiVB via MBG.

Feature	Problem Description
Directory user name is sent to ISP in Display SIP Info (From Header, PAI, Contact)	User name is displayed in ISP side. **Recommendation**: From MBG SIP Trunking, use the SIP Adaptation
PSTN_to_Busy_Device	If you try incoming call from an IP Phone (MiNet) you have a busy Tone and the message Busy displayed on the screen. But through the SIP Trunk the call present to the IP phone.
Do_Not_Disturb_PSTN_MiCollab	MiCollab Client alert and PSTN doesn't hear busy tone. The phone ringing. If you try incoming call from an IP Phone you have a busy Tone and the message DND displayed on the screen.
Authentication for incoming calls	Service Provider OpenIP does not support Authentication for Incoming Calls **Recommendation**: Contact Service Provider OpenIP for updates for supporting authentication for incoming calls
Video	Service Provider OpenIP does not support video calling **Recommendation**: Contact Service Provider X for updates for supporting Video calling
G722	Service Provider OpenIP does not support G722 Codec **Recommendation:** Contact Service Provider OpenIP for updates for supporting G722 Codec
Fax	Service Provider OpenIP does not support T38 fax calling. **Recommendation:** Contact Service Provider OpenIP for updates for supporting T38 fax calling.
TLS / SRTP	Service Provider OpenIP does not support RTP/AVP + Crypto **Recommendation:** Contact Service Provider OpenIP for updates for supporting RTP/AVP + Crypto
inband-DTMF	MITEL does not support inband DTMF in Virtual MiVB **Recommendation:** Contact MITEL for updates for supporting inband DTMF

Network Topology

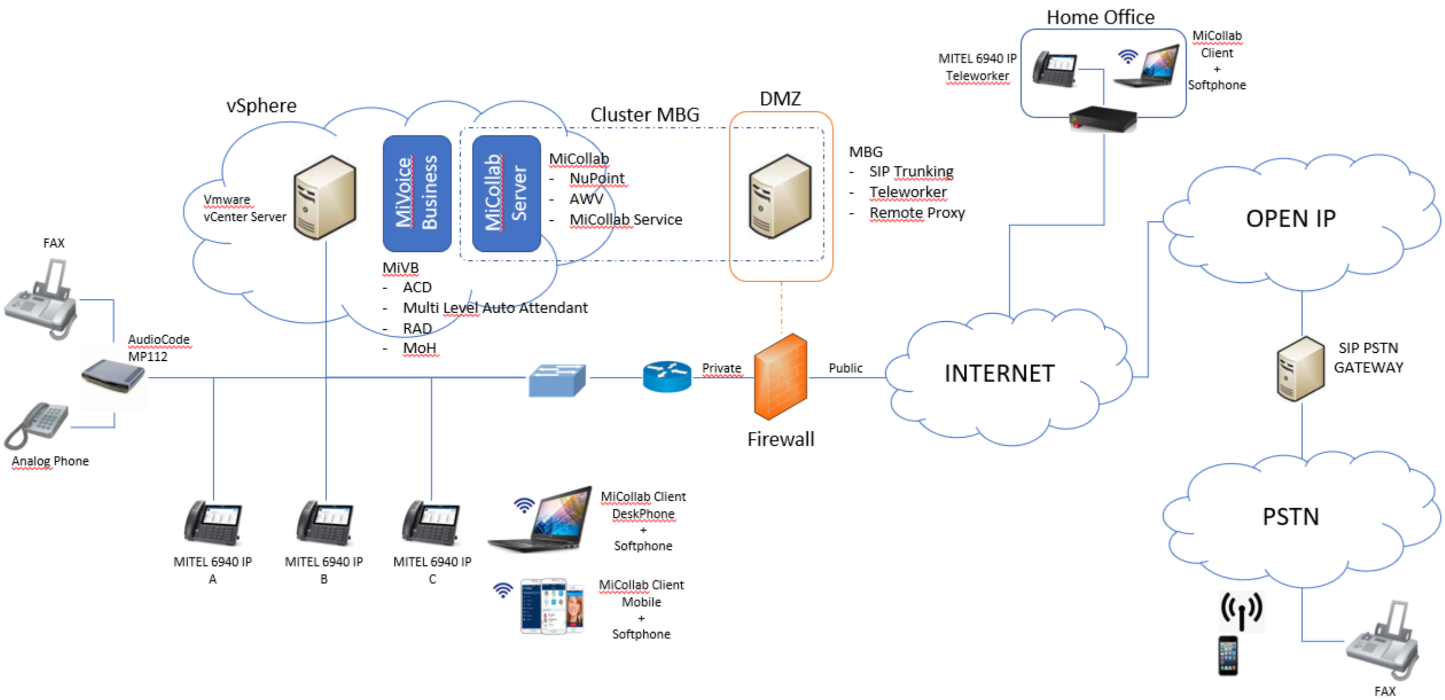

Figure 1 – Network Topology

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 Service Provider OpenIP SIP Trunking MiVB 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.

MiVB Configuration Notes

The following steps show how to program a MiVB to interconnect with Service Provider OpenIP SIP Trunking.

Configuration Template

A configuration template can be found in the same Mitel Knowledge Management System (KMS) article as this document. The template is a Microsoft Excel spreadsheet (.csv format) **solely** consisting of the SIP Peer profile option settings used during Interop testing. All other forms should be programmed as indicated below. Importing the template can save you considerable configuration time and reduce the likelihood of data-entry errors. Refer to the MiVB documentation on how the Import functionality is used.

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 MiVB 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 MiVB Programming

The SIP signaling connection uses UDP on Port 5060.

Licensing and Option Selection – SIP Licensing

Ensure that the MiVB is equipped with enough SIP trunking licenses for the connection to Service Provider OpenIP SIP Trunking. This can be verified within the License and Option Selection form.

Enter 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 MiVB to be used with all service providers, applications and SIP trunking devices.

License and Option Selection

MiVoice Business Console Active Operators	0	0	20	0	Unrestricted	No
Multi-device Users	1	4	0	4	Unrestricted	Yes
Multi-device Suites	0	0	20	0	Unrestricted	Yes
Messaging						
Embedded Voice Mail	9	24	0	24	Unrestricted	Yes
Embedded Voice Mail PMS	0	No	1	0	Unrestricted	Yes
Trunking / Networking						
Digital Links	0	0	1	1	0	No
Compression		8	0	8	Unrestricted	Yes
FAX Over IP (T.38)		0	4	4	0	No
SIP Trunks	0	14	0	14	Unrestricted	Yes

Figure 2 – License and Option Selection

Class of Service Assignment

The Class of Service Options Assignment form is used to create or edit a Class of Service and specify its options. Classes of Service, identified by Class of Service numbers, are referenced in the Trunk Service Assignment form for SIP trunks.

Many different options may be required for your site deployment but ensure that "Public Network Access via DPNSS" Class of Service Option is configured for all devices that make outgoing calls through the SIP trunks in the MiVB.

- Busy Override Security set to **Yes**
- Campon Tone Security set to **Yes**
- Public Network Access via DPNSS set to **Yes**

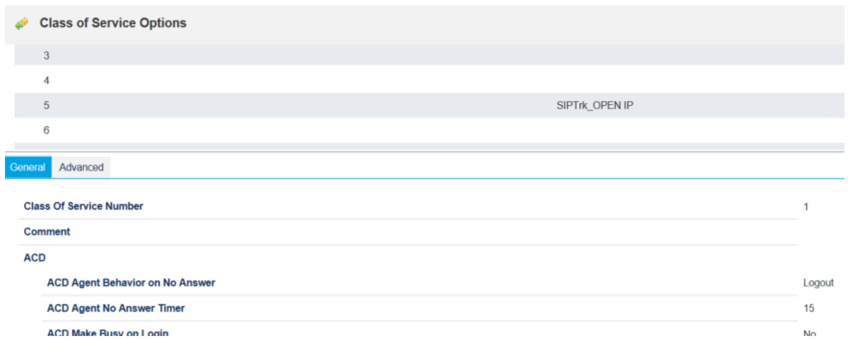

Figure 3 – Class of Service

Network Element Assignment

Create a network element for Service Provider OpenIP SIP Trunking. In this example, the softswitch is reachable by an FQDN and is defined as “OpenIP” in the network element assignment form. **The FQDN or IP addresses of the SIP Peer (Network Element), the External SIP Proxy and Registrar are provided by your service provider.**

If your service provider trusts your network connection by asking for your gateway external IP address, then programming the IP address for the SIP Peer, Outbound Proxy and Registrar is not required for SIP trunk integration. This will need to be verified with your service provider. Set the transport to UDP and port to 5060.

Name	OpenIP
Type	Other
FQDN or IP Address	voip.myopenip.fr
Data Sharing	NO
Local	False
Version	
Zone	1
ARID	
SIP Peer Specific	
SIP Peer Transport	UDP
SIP Peer Port	5060
External SIP Proxy FQDN or IP Address	
External SIP Proxy Transport	default
External SIP Proxy Port	0
SIP Registrar FQDN or IP Address	voip.myopenip.fr
SIP Registrar Transport	UDP
SIP Registrar Port	5060
SIP Peer Status	Auto-Detect/Normal

Figure 4 – Network Element Assignment

Network Element Assignment (Proxy)

In addition, depending on your configuration, a Proxy may need to be configured to route SIP data to the service provider. If you have a Proxy server installed in your network, the MiVB will require knowledge of this by programming the Proxy as a network element then referencing this proxy in the SIP Peer profile assignment (later in this document).

Name	MBG_SIP
Type	Outbound Proxy
FQDN or IP Address	MBG IP Address
Data Sharing	NO
Local	False
Version	
Zone	1
ARID	
Outbound Proxy Specific	
Outbound Proxy Transport Type	UDP
Outbound Proxy Port	5060

Figure 5 – Network Element Assignment (Proxy)

Trunk Attributes

This is configured in the Trunk Attributes form. In this example the Trunk Attributes is defined for Trunk Service Number 5 which will be used to direct incoming calls to an answer point in the Mitel MiVB.

The example below shows configuration for incoming DID calls. The Mitel MiVB will absorb 0 digits of the DID number from Service Provider OpenIP because we use a DID Service Form. Please refer to the Mitel MiVB System Administration documentation for further programming information.

Trunk Service Number	5
Release Link Trunk	No
Call Recognition Service	Off
Direct Inward Dialing Service	On
Caller Based Routing Service	Off
Class of Service	5
Class of Restriction	5
Baud Rate	300
Intercept Number	1
Non-dial In Trunks Answer Point - Day	
Non-dial In Trunks Answer Point - Night 1	
Non-dial In Trunks Answer Point - Night 2	
Dial In Trunks Incoming Digit Modification - Absorb	0
Dial In Trunks Incoming Digit Modification - Insert	
Dial In Trunks Answer Point	
Dial In Trunks Insert Forwarding Information	No
Trunk Label	Ext.

Figure 6 – Trunk Attributes

SIP Peer Profile

The recommended connectivity via SIP Trunking does not require additional physical interfaces. IP/Ethernet connectivity is part of the base MiVB Platform. The SIP Peer Profile should be configured with the following options:

Network Element: The selected SIP Peer Profile needs to be associated with previously created "OpenIP" Network Element.

Registration User Name: The Mitel MiVB does not support Bulk Registration; therefore trunks will have to be registered individually. This information will be supplied by the provider.

Address Type: Select IP address.

Outbound Proxy Server: Select the Network Element previously configured for the Outbound Proxy Server.

Calling Line ID: The default CPN is applied to all calls unless there is a match in the "Outgoing DID Ranges" of the SIP Peer Profile. **This number will be provided by Service Provider OpenIP**. Do not use a Default CPN if you want public numbers to be preserved through the SIP interface. Add private numbers into the DID ranges for CPN Substitution form (see DID Ranges for CPN Substitution). Then select the appropriate numbers in the Outgoing DID Ranges in this form (SIP Peer Profile).

Trunk Service Assignment: Enter the trunk service assignment previously configured.
SMDR: If Call Detail Records are required for SIP Trunking, the SMDR Tag should be configured (by default there is no SMDR and this field is left blank).

Maximum Simultaneous Calls: This entry should be configured to maximum number of SIP trunks provided by Service Provider OpenIP.

> *NOTE: Ensure the remaining SIP Peer profile policy options are similar the screen capture below.*

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Outgoing DID Ranges	Profile Information

SIP Peer Profile Label	OpenIP
Network Element	OpenIP
Local Account Information	
Registration User Name	
Address Type	IP Address: 192.168.32.10
Administration Options	
Interconnect Restriction	1
Maximum Simultaneous Calls	12
Minimum Reserved Call Licenses	2
Outbound Proxy Server	MBG_SIP
SMDR Tag	5
Trunk Service	5
Zone	1
Authentication Options	
User Name	
Password	*******
Confirm Password	*******
Authentication Option for Incoming Calls	No Authentication
Subscription User Name	
Subscription Password	*******
Subscription Confirm Password	*******
Gateway Options	
Digital Trunk Licenses	0
Maximum Digital/Analog Channels	0

Information provided by Service provider

Figure 7 – SIP Peer Profile Assignment- Basic

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Outgoing DID Ranges	Profile Information

Alternate Destination Domain Enabled	No
Alternate Destination Domain FQDN or IP Address	
Enable Special Re-invite Collision Handling	Yes
Only Allow Outgoing Calls	No
Private SIP Trunk	No
Reject Incoming Anonymous Calls	No
Route Call Using P-Called-Party-ID (if present)	No
Route Call Using To Header	No

Figure 8 – SIP Peer Profile Assignment- Call Routing

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Outgoing DID Ranges	Profile Information

Default CPN	
Default CPN Name	
CPN Restriction	No
Override From Header with Default CPN	No
Public Calling Party Number Passthrough	No
Strip PNI	No
Use Diverting Party Number as Calling Party Number	No
Use Original Calling Party Number If Available	No

Figure 9 – SIP Peer Profile Assignment- Calling Line ID

Basic | Call Routing | Calling Line ID | **SDP Options** | Signaling and Header Manipulation | Timers | Key Press Event | Outgoing DID Ranges | Profile Information

Allow Peer To Use Multiple Active M-Lines	No
Allow Using UPDATE For Early Media Renegotiation	Yes
Avoid Signaling Hold to the Peer	Yes
AVP Only Peer	Yes
Enable Mitel Proprietary SDP	No
Force sending SDP in initial Invite message	Yes
Force sending SDP in initial Invite - Early Answer	No
Ignore SDP Answers in Provisional Responses	No
IP Media Default	ipv4
Limit to one Offer/Answer per INVITE	Yes
NAT Keepalive	No
Prevent the Use of IP Address 0.0.0.0 in SDP Messages	Yes
Renegotiate SDP To Enforce Symmetric Codec	No
Repeat SDP Answer If Duplicate Offer Is Received	No
Restrict Audio Codec	No Restriction
RTP Packetization Rate Override	No
RTP Packetization Rate	20ms
Special handling of Offers in 2XX responses (INVITE)	No
Suppress Use of SDP Inactive Media Streams	No

Figure 10 – SIP Peer Profile Assignment- SDP Options

Basic | Call Routing | Calling Line ID | SDP Options | **Signaling and Header Manipulation** | Timers | Key Press Event | Outgoing DID Ranges | Profile Information

Trunk Group Label	
Allow Display Update	No
Build Contact Using Request URI Address	No
De-register Using Contact Address not *	No
Disable Reliable Provisional Responses	Yes
Disable Use of User-Agent and Server Headers	No
Domain for Trunk Context	
E.164: Enable sending '+'	No
E.164: Add '+' if digit length > N digits	0
E.164: Do not add '+' to Emergency Called Party	No
E.164: Do not add '+' to Called Party	No
Force Max-Forward: 70 on Outgoing Calls	No
If TLS use 'sips:' Scheme	No
Ignore Incoming Loose Routing Indication	No
Include Diversion Header for EHDU	No
Mode for Out-of-Band DTMF	RFC 4733 DTMF
Multilingual Name Display	No
Only use SDP to decide 180 or 183	Yes

Prefer From Header for Caller ID	Yes
Q.850 Reason Headers	No
Require Reliable Provisional Responses on Outgoing Calls	No
Signal Privacy (if enabled) on Emergency Calls	No
Suppress Redirection Headers	No
Use Fixed Retry Time for 491	Yes
Use Privacy: none	No
Use P-Asserted Identity Header	Yes
Use P-Asserted Identity for Billing	No
Use P-Call-Leg-ID Header	No
Use P-Early-Media Header	No
Use P-Preferred Identity Header	No
Use Restricted Character Set For Authentication	No
Use To Address in From Header on Outgoing Calls	No
Use user=phone	No
Use user=phone for Diversion Header	No

Figure 11 – SIP Peer Profile Assignment- Signaling and Header Manipulation

Basic | Call Routing | Calling Line ID | SDP Options | Signaling and Header Manipulation | Timers | Key Press Event | Outgoing DID Ranges | Profile Information

Keep-Alive (OPTIONS) Period	120
Registration Period	3600
Registration Period Refresh (%)	50
Registration Maximum Timeout	90
Session Timer	1800
Session Timer: Local as Refresher	No
Subscription Period	3600
Subscription Period Minimum	300
Subscription Period Refresh (%)	80
Invite Ringing Response Timer	0

Figure 12 – SIP Peer Profile Assignment- Timers

Basic | Call Routing | Calling Line ID | SDP Options | Signaling and Header Manipulation | Timers | Key Press Event | Outgoing DID Ranges | Profile Information

Allow Inc Subscriptions for Local Digit Monitoring	No
Allow Out Subscriptions for Remote Digit Monitoring	No
Force Out Subscriptions for Remote Digit Monitoring	No
Request Outbound Proxy to Handle Out Subscriptions	No
KPML Transport	default
KPML Port	0

Figure 13 – SIP Peer Profile Assignment- Key Press Event

Basic | Call Routing | Calling Line ID | SDP Options | Signaling and Header Manipulation | Timers | Key Press Event | Outgoing DID Ranges | Profile Information

Add Member | Delete Member

Index	DID Range	CPN Substitution

Figure 14 – SIP Peer Profile Assignment- Outgoing DID Ranges

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Outgoing DID Ranges	Profile Information

Creator	
Date Created	
Created with Version	
Service Provider	
Vendor Notes	

Figure 15 – SIP Peer Profile Assignment- Profile Information

SIP Peer Profile Assignment by Incoming DID

This form is used to associate DID range numbers from Service Provider OpenIP SIP trunk to a particular SIP Peer profile. The configured settings here help matching the incoming DID numbers with the SIP Peer Profile when call is arriving from anonymous caller.

Enter one or more telephone numbers. The maximum number of digits per telephone number is 26. You can enter a mix of ranges and single numbers (for example, "33970008470-33970008472, 33970008475"). The entire field width is limited to 60 characters.

Use a comma to separate telephone numbers and ranges. Use a dash (-) to indicate a range of telephone numbers. The first and last characters cannot be a comma or a dash. If the numbers do not fit within the 60 characters maximum, you can create a new entry for the same profile.

Use a '*' to reduce the number of entries that need to be programmed. This is a type of "prefix identifier", and cannot be used as a range with '-'. For example, the string "11*" would be used to associate a peer with any number in the range from 110 up to the maximum digits per telephone number (In this case, 11999999999999999999999999.) Note that the string "11" by itself would not count as a match, as the '*' represents 1 or more digits.

Note :
If a SIP "404 Not Found" message is returned when the number does exist, you may need to enter the DID range in this form and associate a SIP Peer profile to the number(s).

SIP Peer Profile Assignment by Incoming DID

Incoming DID Range	SIP Peer Profile Label	Comment
0970196760-0970196764	OpenIP	Open IP
1600-1699,0211-0213,9100-9199	CUCM	CUCM

Incoming DID Range	0970196760-0970196764
SIP Peer Profile Label	OpenIP
Comment	Open IP

Figure 16 – SIP Peer Profile Assignment by Incoming DID

ARS Digit Modification Plans

Ensure that Digit Modification for outgoing calls on the SIP trunk to Service Provider OpenIP absorbs or injects additional digits according to your dialling plan. In this example, we will be absorbing 1 digit (in this case will be 6 to dial out).

Digit Modification Number	Number of Digits to Absorb	Digits to be Inserted	Final Tone Plan/Information Marker
1	0		
2	0		
3	0		
4	0		
5	1		

Figure 17 – Digit Modification Assignment

ARS Routes

Create a route for SIP Trunks connecting a trunk to Service Provider OpenIP. In this example, the SIP trunk is assigned to Route Number 21. Choose SIP Trunk as a routing medium and choose the SIP Peer Profile and Digit Modification entry created earlier.

Route Number	Routing Medium	Trunk Group Number	SIP Peer Profile	PBX Number / Cluster Element ID	COR Group Number	Digit Modification Number	Digits Before Outpulsing	Route Type	Compression
16					1	1	4		Off
17					1	1	4		Off
18					1	1	4		Off
19					1	1	4		Off
20					1	1	4		Off
21	SIP Trunk		OpenIP		5	5	4		Off

Figure 18 – SIP Trunk Route Assignment

ARS Digits Dialed

ARS initiates the routing of trunk calls when certain digits are dialed from a station. In this example, when a user dials 6, the call will be routed to Service Provider OpenIP (i.e., Route 21).

Digits Dialed	Number of Digits to Follow	Termination Type	Termination Number
16	2	Route	2
202	1	Route	2
23	2	Route	12
40	2	Route	2
6	Unknown	Route	21

Figure 19 – ARS Digit Dialed Assignment

MiVoice Border Gateway Configuration Notes

When configuring MiVoice Border Gateway (MBG), you need to identify the working MiVB ICP where to forward SIP messages to and then to configure the SIP trunk.

To do this:

- Login to MBG and click **Mitel Border Gateway**
- In right pane, click **Service Configuration** tab and then **ICPs**

Figure 20 – MBG's Configuration page

- On **ICPs** page, ensure that the "working" MiVB is configured. If needed, click **"+"** link and add a new Mitel switch.
- Click **Save** button

To add a new SIP trunk:

- Click **Service Configuration** tab and then click **SIP trunking**
- Click **"+"** link

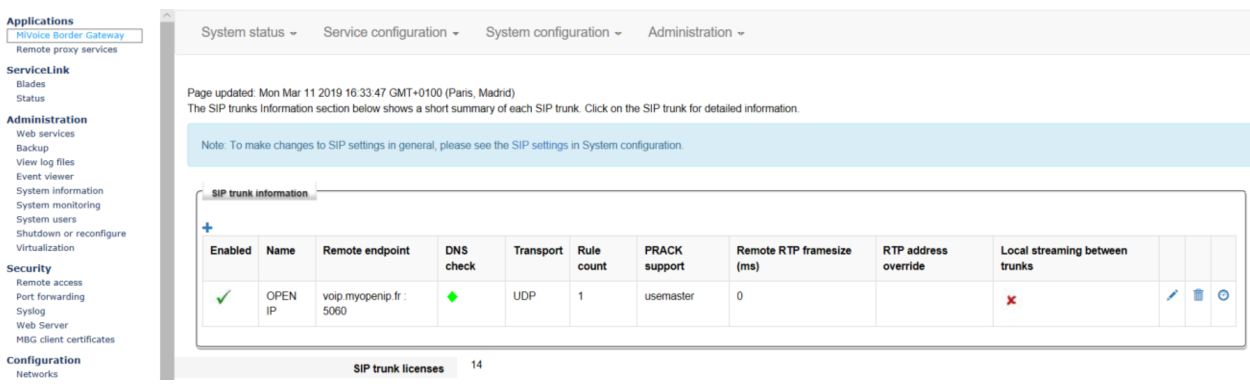

Figure 21 – SIP trunking configuration page

Enter the SIP trunk's details as shown in Figure 22:

Enabled – Tick it

Name – is the name of the trunk

Remote trunk endpoint address – FQDN of the provider's switch or gateway (this address should be given to you by the provider, e.g. Service Provider OpenIP).

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 Mitel MiVB ICP

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

Click **Save** button

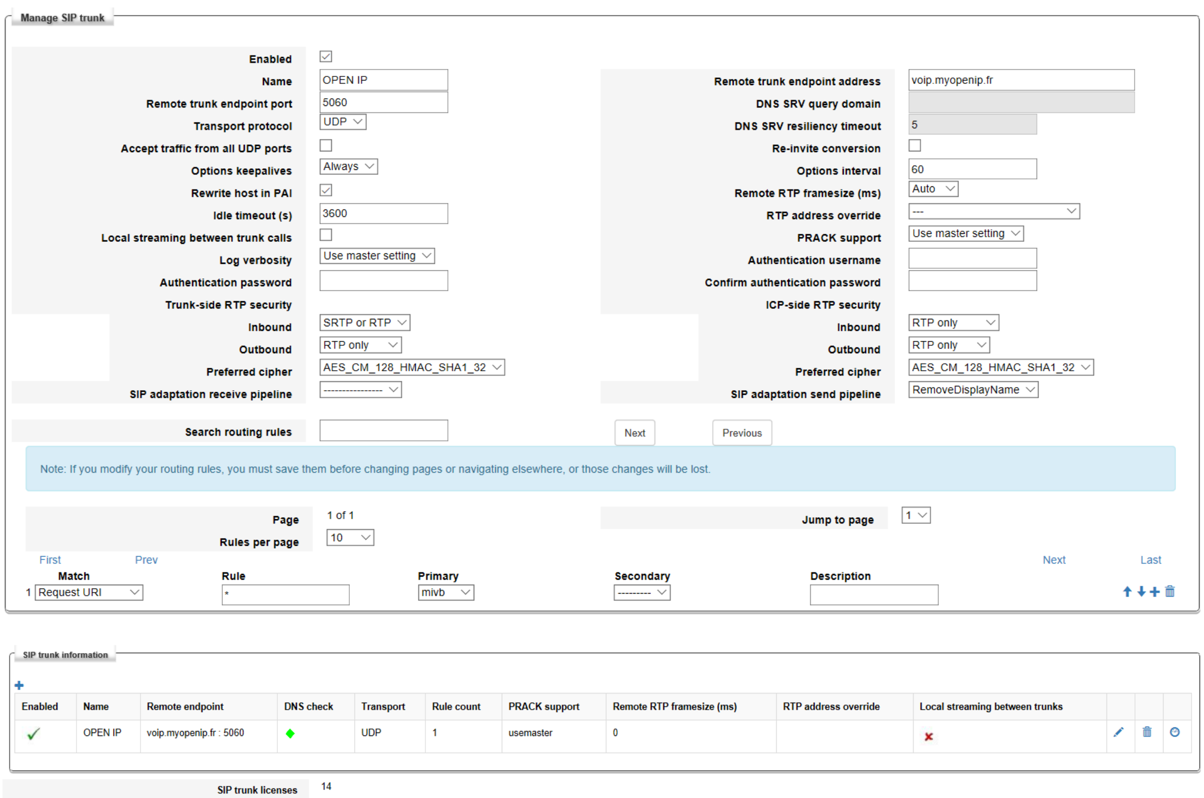

Figure 22 – SIP Trunk configuration settings

Glossary

MiVoice Business	MiVB
MiContact Center Business	MiCC-B
MiVoice Call Recorder	MiVCR
MiVoice Border Gateway	MBG
MiCollab	MiCollab
Mitel Open Integration Gateway	OIG
MiTAI API	MiTAI
MiNET Interface	MiNET
Secure Recording Connector Interface	SRC
Mitel Solutions Alliance	MSA
Personal Ring Group	PRG
External Hot Desk User	EHDU
Knowledge Management System	KMS
MiVoice Office 250	MiVO-250
MiVoice Office 400	MiVO-400
MiVoice MX-ONE	MX-ONE
MiVoice 5000	MiV-5000
MiContact Center Enterprise	MiCC-E
Class of Service	COS