MITEL

MiVB

Software Version: 13.2.0.17

Fusion SIP Trunking

NOVEMBER 2015

NOTICE

The information contained in this document is believed to be accurate in all respects but is not warranted by Mitel Networks™ Corporation (MITEL®). The information is subject to change without notice and should not be construed in any way as a commitment by Mitel or any of its affiliates or subsidiaries. Mitel and its affiliates and subsidiaries assume no responsibility for any errors or omissions in this document. Revisions of this document or new editions of it may be issued to incorporate such changes.

No part of this document can be reproduced or transmitted in any form or by any means - electronic or mechanical - for any purpose without written permission from Mitel Networks Corporation.

TRADEMARKS

Mitel is a trademark of Mitel Networks Corporation.

Windows and Microsoft are trademarks of Microsoft Corporation.

Other product names mentioned in this document may be trademarks of their respective companies and are hereby acknowledged.

Mitel Technical Configuration Notes:
Configure the Mitel Voice Business for use with Fusion SIP trunk November 2015, 15-4940-00412

CONTENTS:

OVERVIEW

This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel Voice Business to connect to Fusion 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 Mitel Voice Business and Fusion SIP trunk

INTEROP STATUS

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

COMPATIBLE	The most common certification which means Fusion 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 and the Mitel Voice Business.

Manufacturer	Variant	Software Version
Mitel	Mitel Voice Business	13.2.0.17
Mitel	5300 Series IP Sets	06.03.01.05
Mitel	6800 Series SIP Sets	4.0.0.1096
Fusion		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 Pans (08-4940-00034) for detailed test cases.

Feature	Feature Description	Issues
Basic Call	Making and receiving a call to and from the Fusion, 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.	☑
NuPoint Voicemail	Terminating calls to a NuPoint voicemail boxes and DTMF detection.	☑
Packetization	Forcing the Mitel Voice Business to stream RTP packets different intervals, of 20ms and 30ms.	⚠
Personal Ring Groups	Receiving calls from Fusion to a personal ring group. Also moving calls to/from the prime member and group members.	☑
Video	Video Calls inbound and outbound	✗
Fax	T.38 and G711 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 is connected to the Mitel Voice Business.

Feature	Problem Description
Video	Video is not supported.
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 session timer in the SIP peer profile to a value which will prevent the timer form expiring during a T.38 FAX call. Reference defect MN00607790 when contacting Mitel Product Support.
Paketization	P-times of other than 20ms are not honored by Fusion. **Recommendation**: Use the default 20ms P-time
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.

NETWORK TOPOLOGY

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

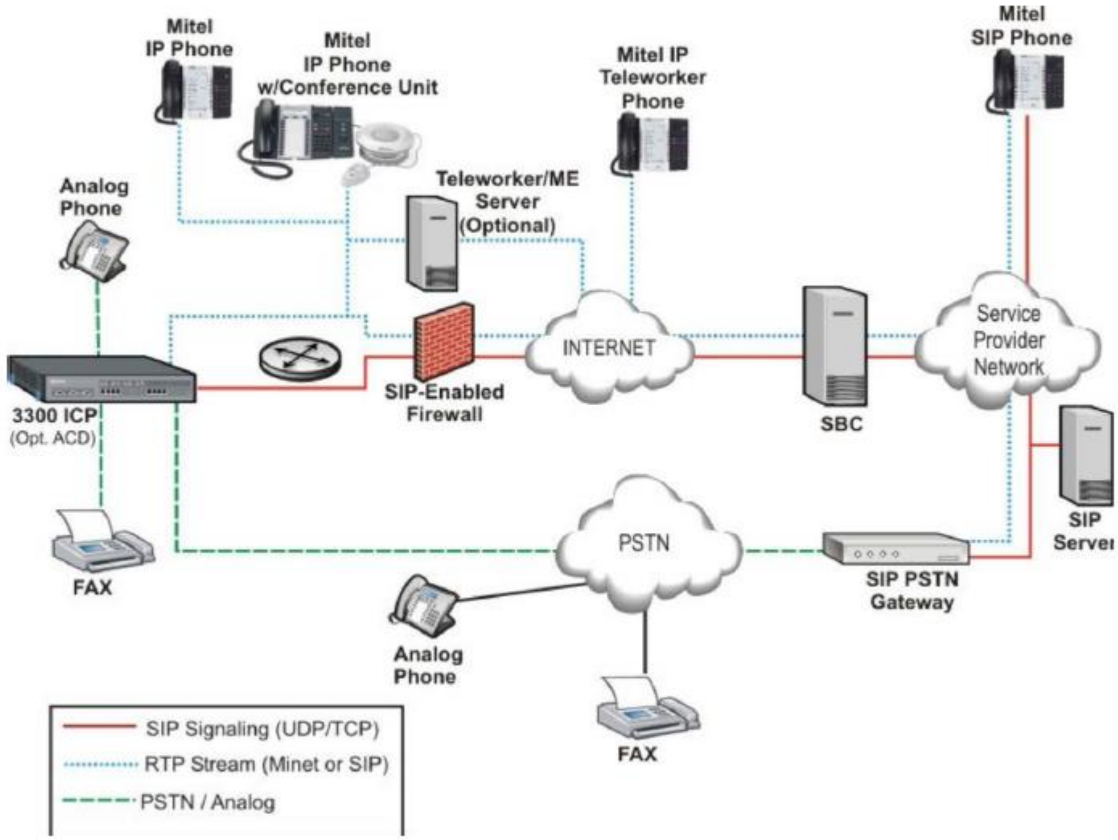


Figure 1 – Network Typology

CONFIGURATION NOTES

This section is a description of how the SIP Interop was configured. These notes should give a guideline how a device can be configured in a customer environment and how the 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 Mitel Voice Business to interconnect with Fusion SIP trunking.

Configuration Template

A configuration template can be found in the same MOL Knowledge Base 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 ICP 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 for G.729. 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 the 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 the Fusion SIP trunk. 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.

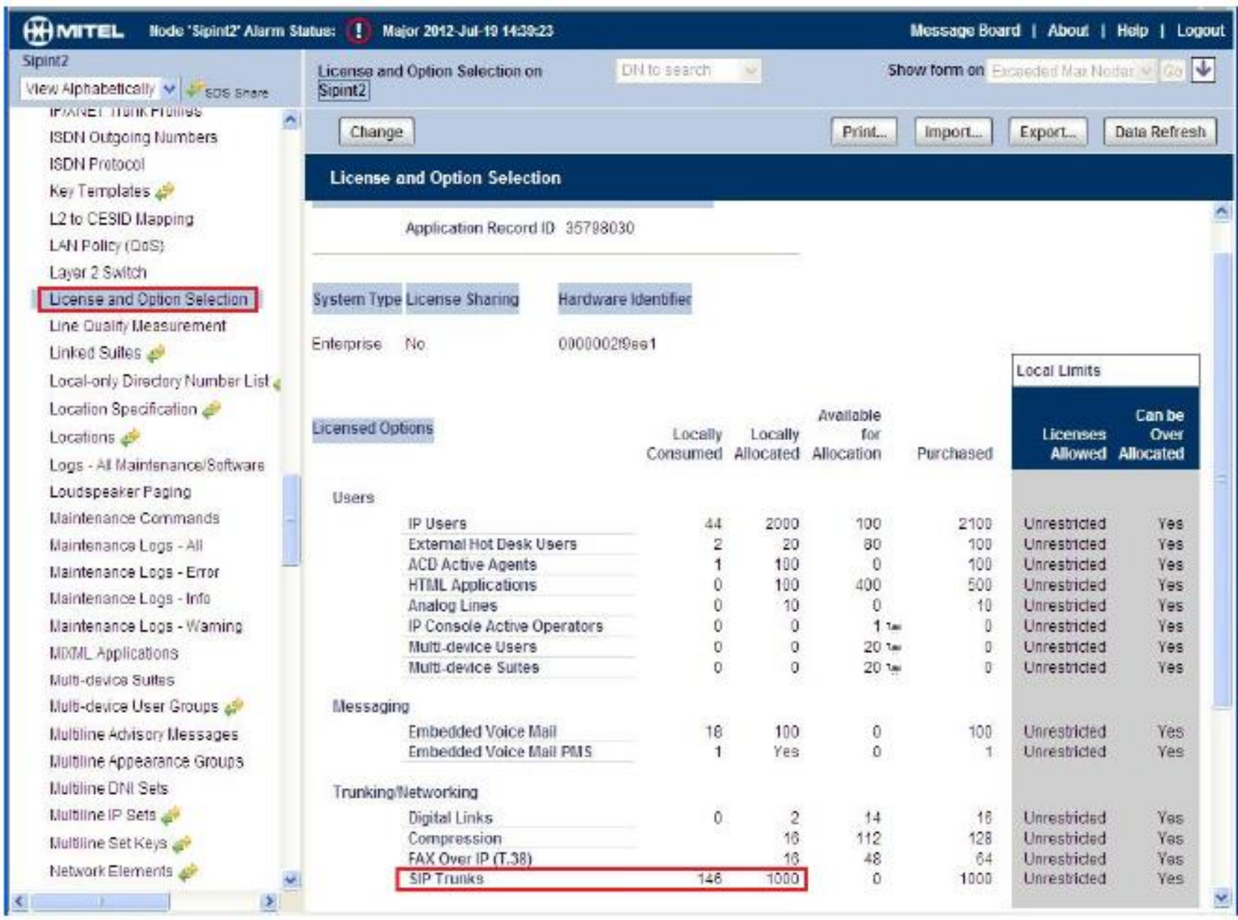


Figure 2 – License and Option Selection Form

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 Attributes 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.

Also, under General tab, ensure that the following options are enabled (see Figure 3):

- Busy Override Security (in Busy Override section) set to **Yes**
- Campon Tone Security (in Fax section) set to **Yes**
- Public Network Access via DPNSS (in Trunk section) set to **Yes**
- Fax Capable if a Fax device is connected to this port or uses this trunk **YES**

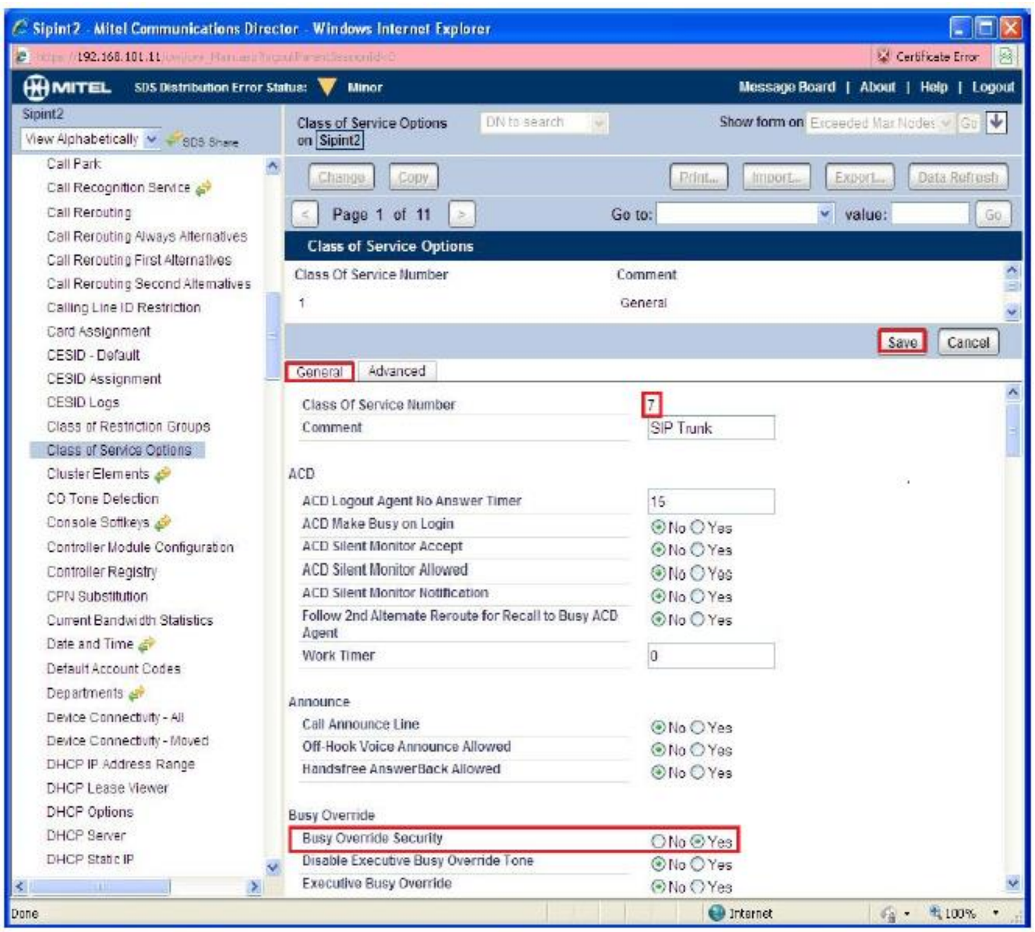


Figure 3 – Class of Service Form

Network Elements

Create a network element for a SIP Peer "Fusion" as shown in **Figure 4**.

If you want to use compression set the Zone to be a different value than that of the ICP. If no compression is required you can set the zone to that of the ICP, 1 by default.

In our setup the SIP trunks do register so the registration address needs to be filled in.

Set the transport to Default or UDP and port to 5060.

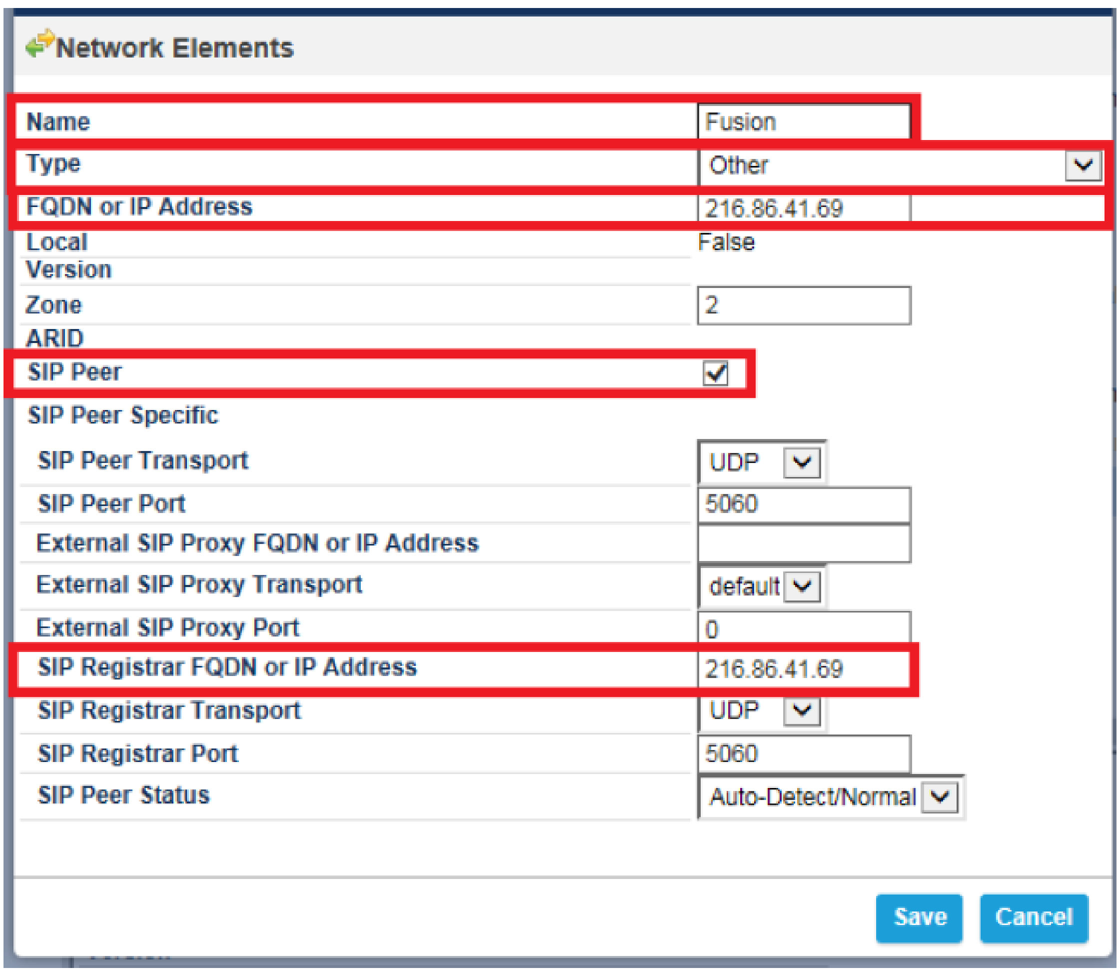


Figure 4 – Network Element Form

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).

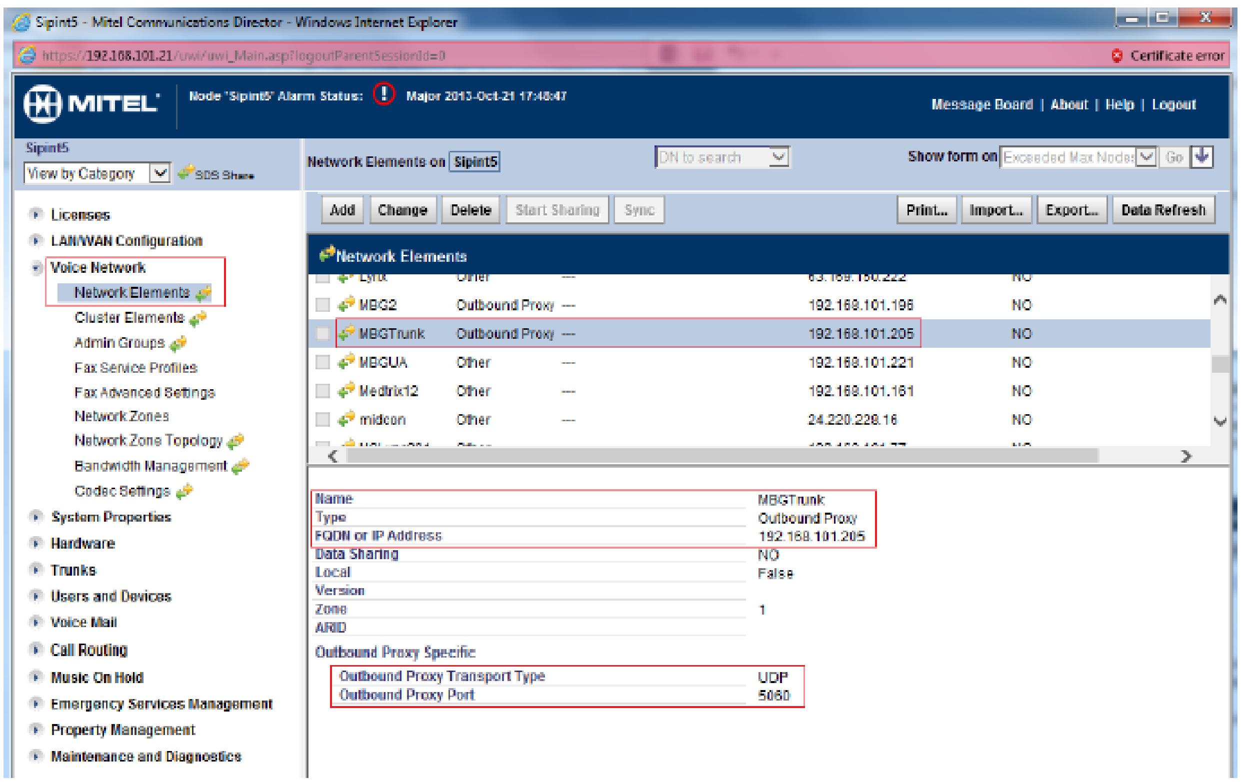


Figure 5 – Network Element Assignment (Proxy)

Trunk Attributes (trunk service number)

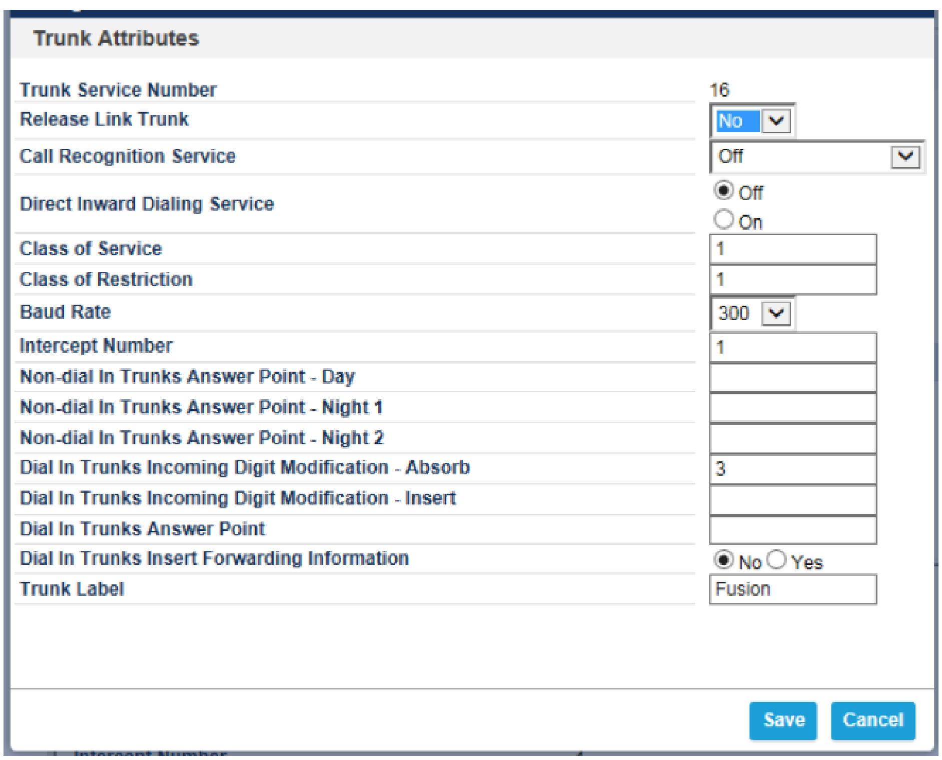
The Trunk Attributes is defined for Trunk Service Number (**16**), which will be used to direct incoming calls to an answer point in the MiVb.

Set the number of Class of Service that was configured in the section above.

Program the Non-dial In Trunks Answer Point according to the site requirements and what type of service was ordered from your service provider.

The figure below shows configuration for incoming DID calls. The MiVb will absorb the first 3 digits of the DID number received from the Fusion SIP trunk leaving 6 digits for the MiVb to translate and route the call.

For example, the Fusion SIP trunk delivers number 613-592-5660 to the MiVb. The MiVb will absorb the first 3 digits (613) leaving the Mitel MiVb 5925660 to routre the call. The digits 5925660 must be programmed as a valid dialable number in the MiVb. As an alternative way, you can create a System Speed Call number to associate number 5925660 with the real telephone extension on MiVb. Please refer to the MiVb System Administration documentation for further programming information.


Trunk Attributes

Trunk Service Number	16
Release Link Trunk	No
Call Recognition Service	Off
Direct Inward Dialing Service	(•) Off ( ) On
Class of Service	1
Class of Restriction	1
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	3
Dial In Trunks Incoming Digit Modification - Insert	
Dial In Trunks Answer Point	
Dial In Trunks Insert Forwarding Information	(•) No ( ) Yes
Trunk Label	Fusion

Save Cancel

Figure 6 – Trunk Attributes (trunk service number)

SIP Peer Profile

The recommended connectivity via SIP Trunking does not require additional physical interfaces. IP/Ethernet connectivity is the part of the MiVb platform. The SIP Peer Profile should be configured as shown in **Figures 7 through 12**.

Basic (Figure 7):

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

Registration User Name: Leave this field blank.

Address Type: Select the IP Address of your Mitel MiVb.

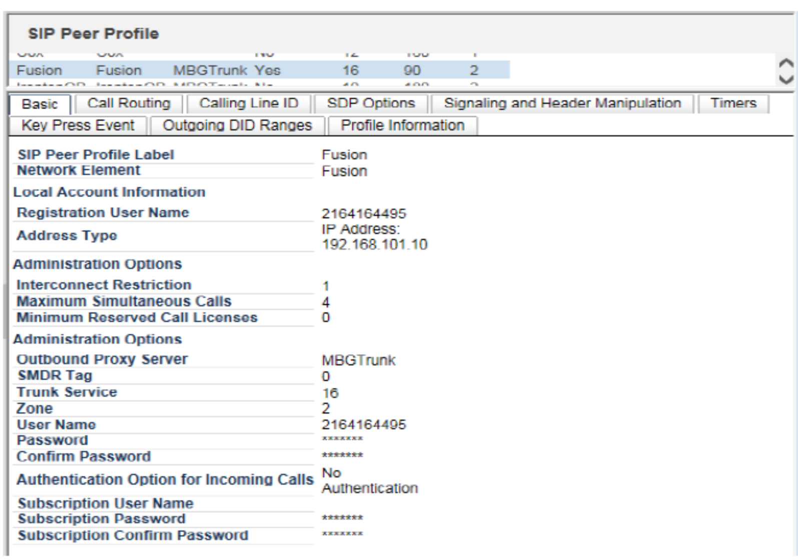
Maximum Simultaneous Calls: This entry should be configured to maximum number of SIP trunks provided by Fusion.

Outbound Proxy Server: Not required in our test setup.

SMDR Tag: 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).

Trunk Service: Enter the trunk attributes number that was previously configured (16) in this configuration.

Authentication Options: User name and password as supplied by Fusion.


SIP Peer Profile

Fusion Fusion MBGTrunk Yes 16 90 2

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	Fusion
Network Element	Fusion
Local Account Information	
Registration User Name	2164164495
Address Type	IP Address: 192.168.101.10
Administration Options	
Interconnect Restriction	1
Maximum Simultaneous Calls	4
Minimum Reserved Call Licenses	0
Administration Options	
Outbound Proxy Server	MBGTrunk
SMDR Tag	0
Trunk Service	16
Zone	2
User Name	2164164495
Password	*******
Confirm Password	*******
Authentication Option for Incoming Calls	No Authentication
Subscription User Name	
Subscription Password	*******
Subscription Confirm Password	*******

Figure 7 – SIP Peer Profile Form

Call Routing (Figure 8):

Leave the default settings intact, as shown.

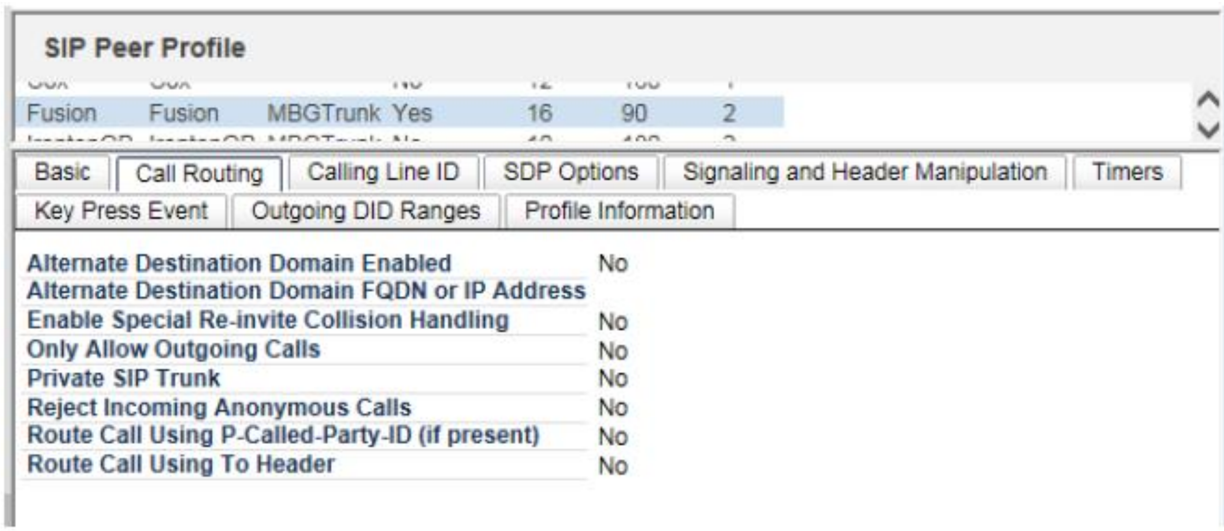


Figure 8 – SIP Peer Profile Form (continues)

Calling Line ID (Figure 9):

The **Default CPN** (Calling Party Number) is applied to all outgoing calls. You can use the one of DID numbers assigned on the trunk by the provider.

CPN Restriction: By default, this parameter is set to "**No**" to not hide the caller's number. You can enable it if required.

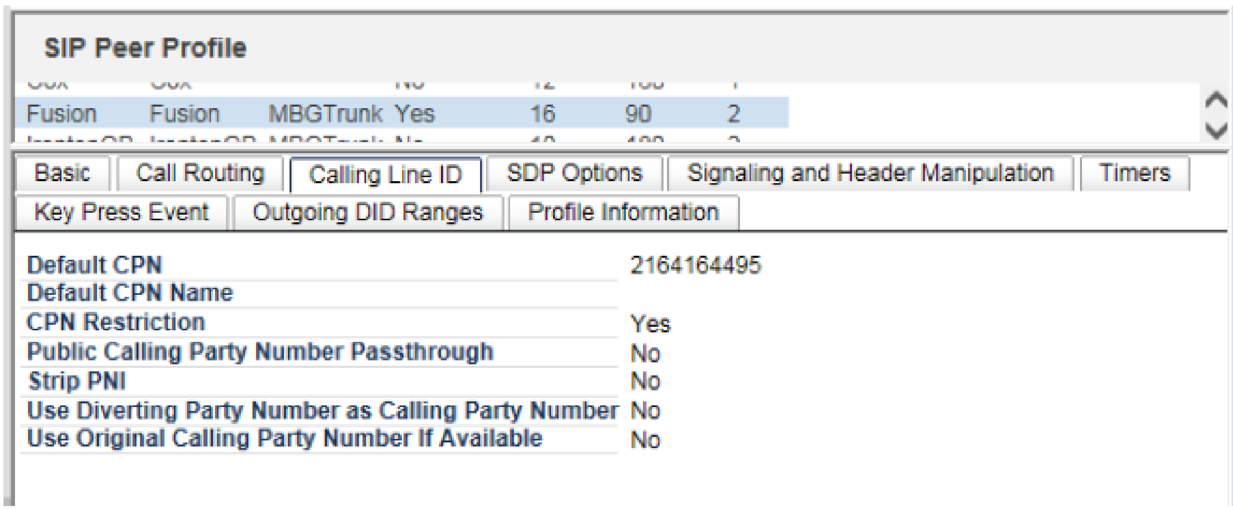


Figure 9 – SIP Peer Profile Form (continues)

SDP Options (Figure 10):

Set the options as depicted below unless there is a specific reason to change them.

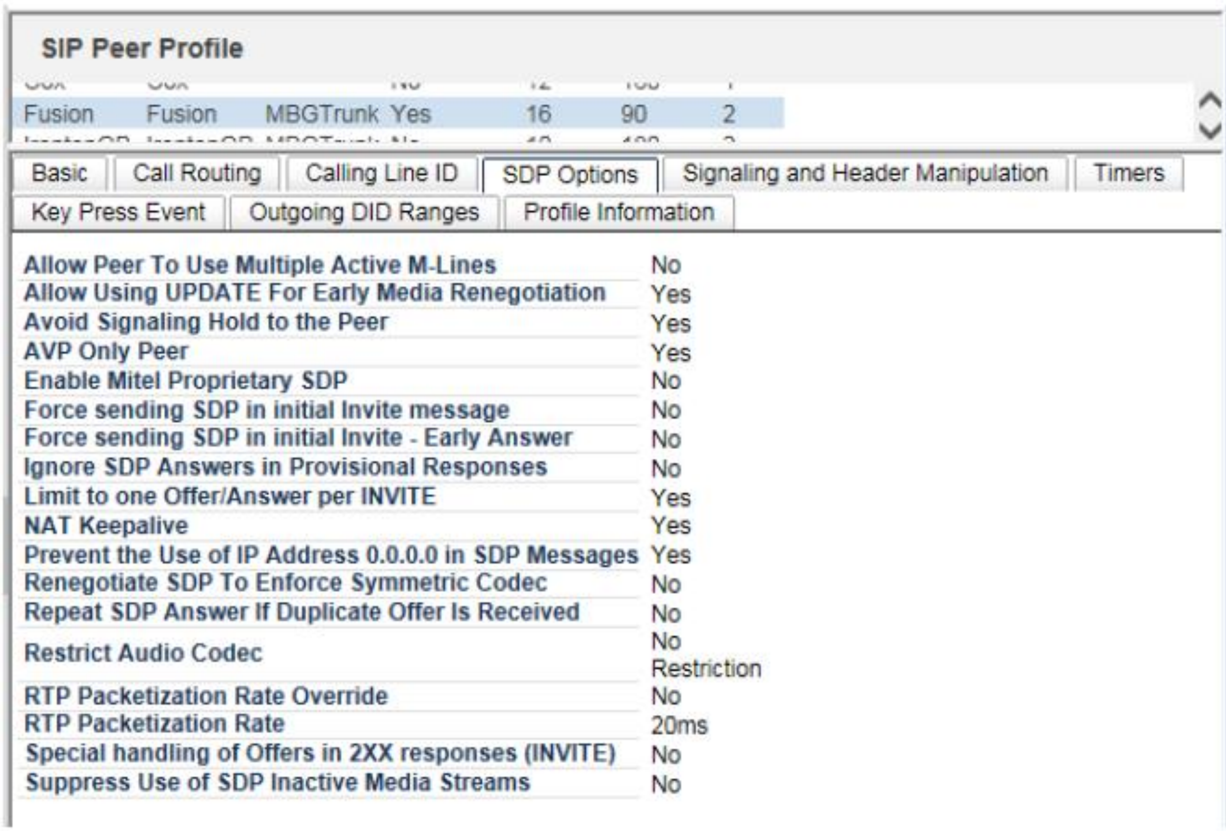
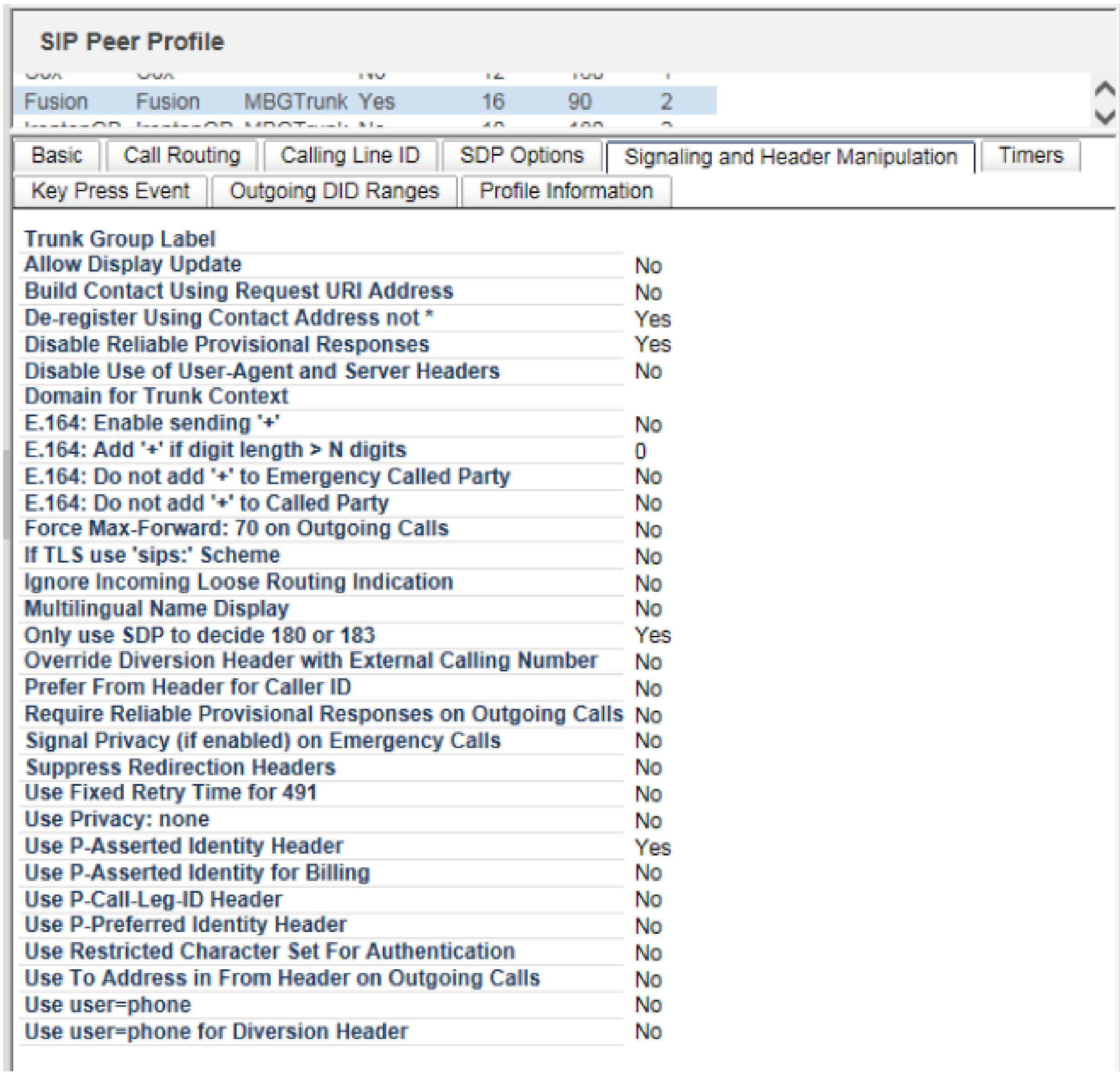


Figure 10 – SIP Peer Profile Form (continues)

Signaling and Header Manipulation (Figure 11):

Figure 11 depicts how the test environment was configured.


SIP Peer Profile

Fusion	Fusion	MBGTrunk	Yes	16	90	2

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 *	Yes
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
Multilingual Name Display	No
Only use SDP to decide 180 or 183	Yes
Override Diversion Header with External Calling Number	No
Prefer From Header for Caller ID	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	No
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-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 form

Timers (Figure 12):

Session Timers: Figure 12 is how the timers were set for our test environment. The session timer was increased from the default to accommodate T.38 FAX calls as stated in Device Limitations and Know Issues section.

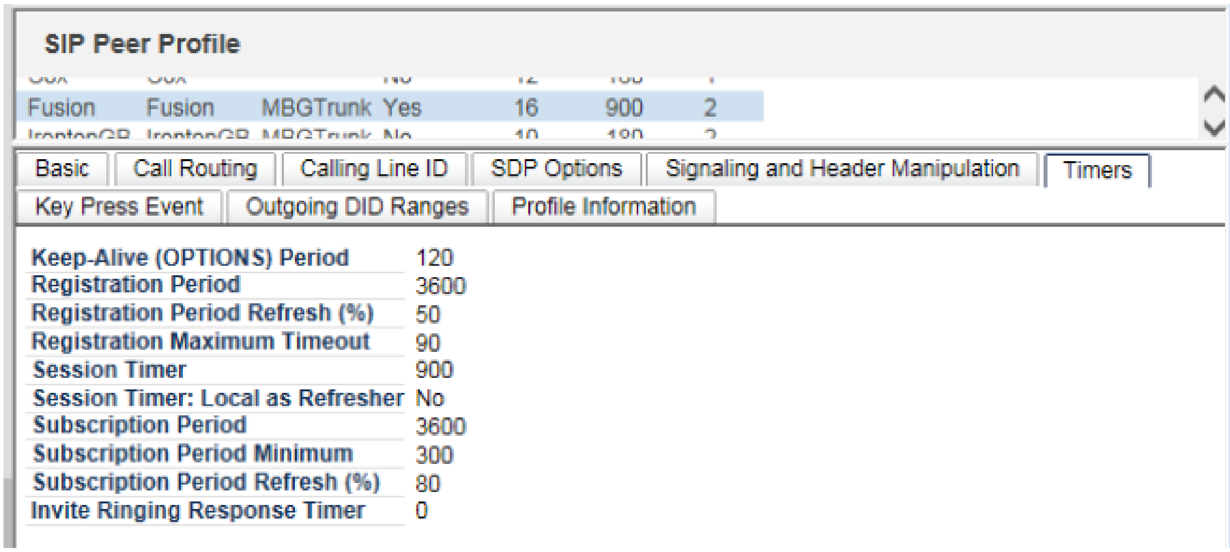


Figure 12 – SIP Peer Profile form (continues)

For Key Press Event and Profile Information tabs, leave the default settings intact.

SIP Peer Profile Assignment by Incoming DID

In some situations calls from anonymous PSTN callers may be rejected at MiVb with Not Found message. To deliver such calls to Mitel's extensions, make sure to associate Fusion's DID number(s) with the SIP Peer Profile we configured earlier.

ARS Digit Modification Plan

Ensure that Digit Modification for outgoing calls to Fusion SIP trunk absorbs or inject additional digits according to your dialling plan. In our test environment, we will be absorbing 1 digit and will not inject any digits, as shown in Figure 13. As per our test environment, we need to dial 952 to access Fusion SIP trunk; thus, digit 952 will be absorbed and no digits will be preceding the dialled number. For instance, if caller dials 9526135555660, MiVb will send to the SIP trunk the following: 6135555660.

ARS Digit Modification Plans

Digit Modification Number	3
Number of Digits to Absorb	3
Digits to be Inserted	
Final Tone Plan/Information Marker	

Figure 13 – ARS Digit Modification Form

ARS Routes

Create a route to Fusion SIP trunk. In this test environment, the SIP trunk is assigned to Route Number 9. Choose **SIP Trunk** as a routing medium and choose the SIP Peer Profile and ARS Digit Modification entry created earlier.

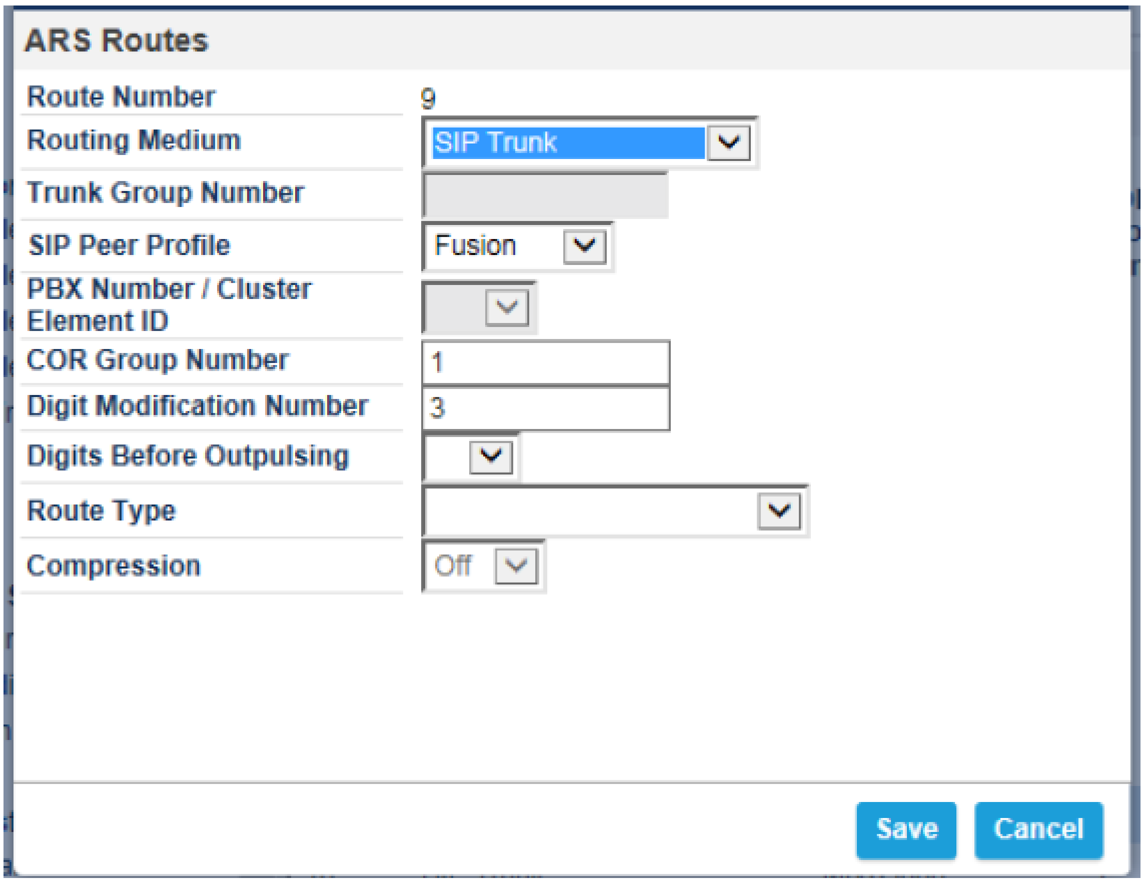
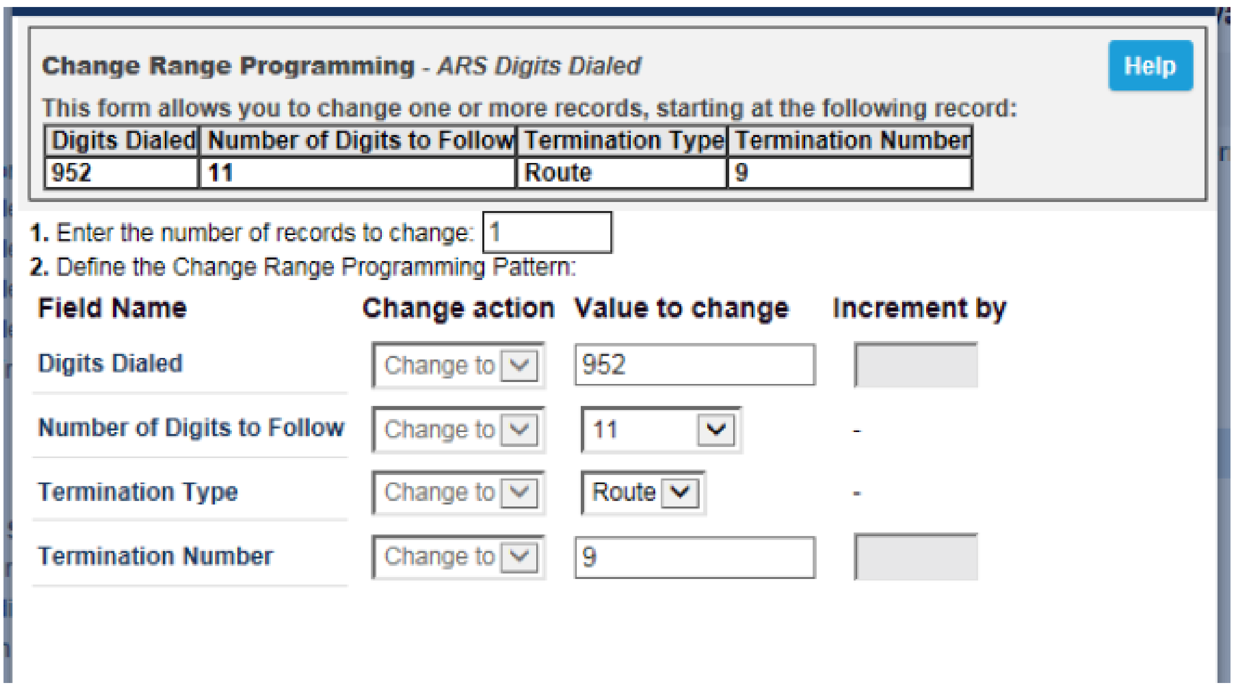


Figure 14 – ARS Route Form

ARS Digits Dialed

ARS initiates the routing of trunk calls when certain digits are dialed from an extension. In this test environment, when user dials 952, the call will be routed to Fusion SIP trunk (i.e. to Route 9). For outbound calling, MiVb expects 11 digits to be dialed after dialing of 957. See Figure 15 for details.



Change Range Programming - *ARS Digits Dialed*

Help

This form allows you to change one or more records, starting at the following record:

Digits Dialed	Number of Digits to Follow	Termination Type	Termination Number
952	11	Route	9

1. Enter the number of records to change: 1

2. Define the Change Range Programming Pattern:

Field Name	Change action	Value to change	Increment by
Digits Dialed	Change to	952	
Number of Digits to Follow	Change to	11	-
Termination Type	Change to	Route	-
Termination Number	Change to	9	

Figure 15 – ARS Digit Dialed form

Fax Service Profiles

This form allows you to define the settings for FAX communication over the IP network. You can modify the default settings for the:

Inter-zone FAX profile: defines the FAX settings between different zones in the network. There is only one Inter-zone FAX profile; it applies to all inter-zone FAX communication. It defaults to V.29, 7200bps. It defines the settings for FAX Relay (T.38) FAX communication.

Intra-zone FAX profile: defines the FAX settings within each zone in the network.

- Profile 1 defines the settings for G.711 pass through communication.
- Profile 2 to 64 define the settings for FAX Relay (T.38) FAX communication.
- All zones default to G.711 pass through communication (Profile 1).

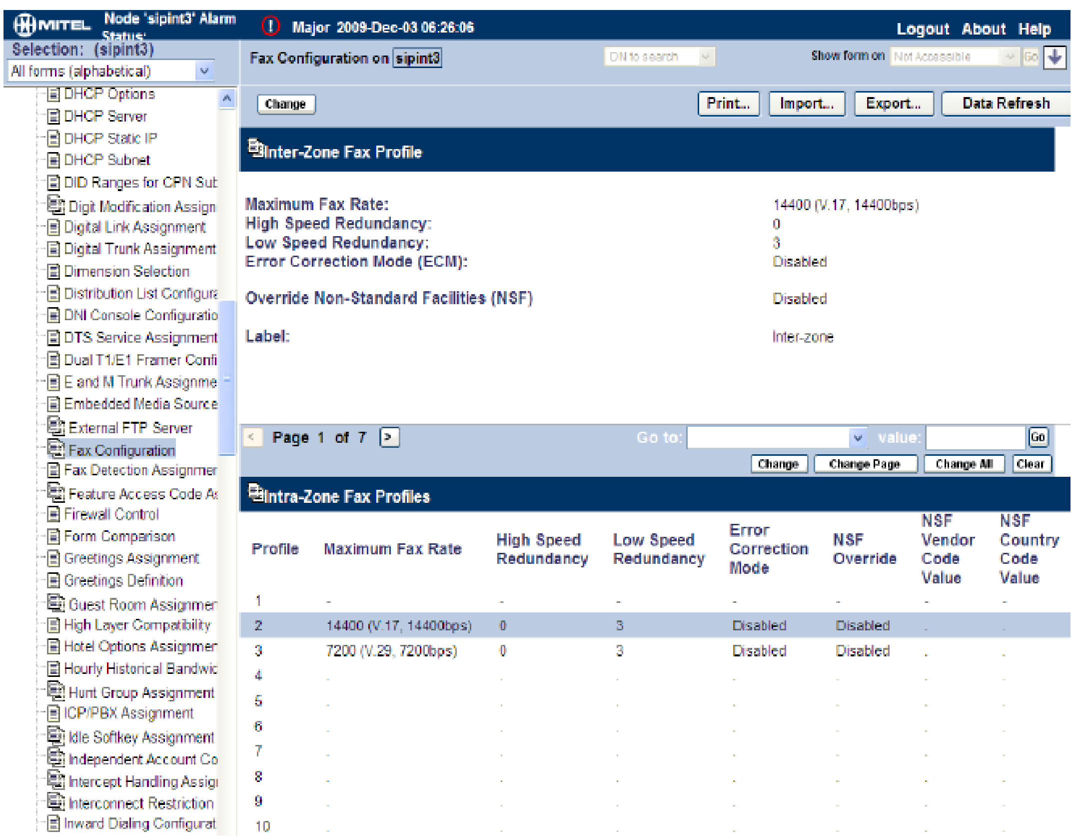


Figure 16 – Fax Configuration

Zone Assignment

By default, all zones are set to Intra-zone FAX Profile 1.
Based on your network diagram, assign the Intra-zone FAX Profiles to the Zone IDs of the zones. If audio compression is required within the same zone, set Intra-Zone Compression to "Yes". Fusion uses the Inter-zone FAX Profile.

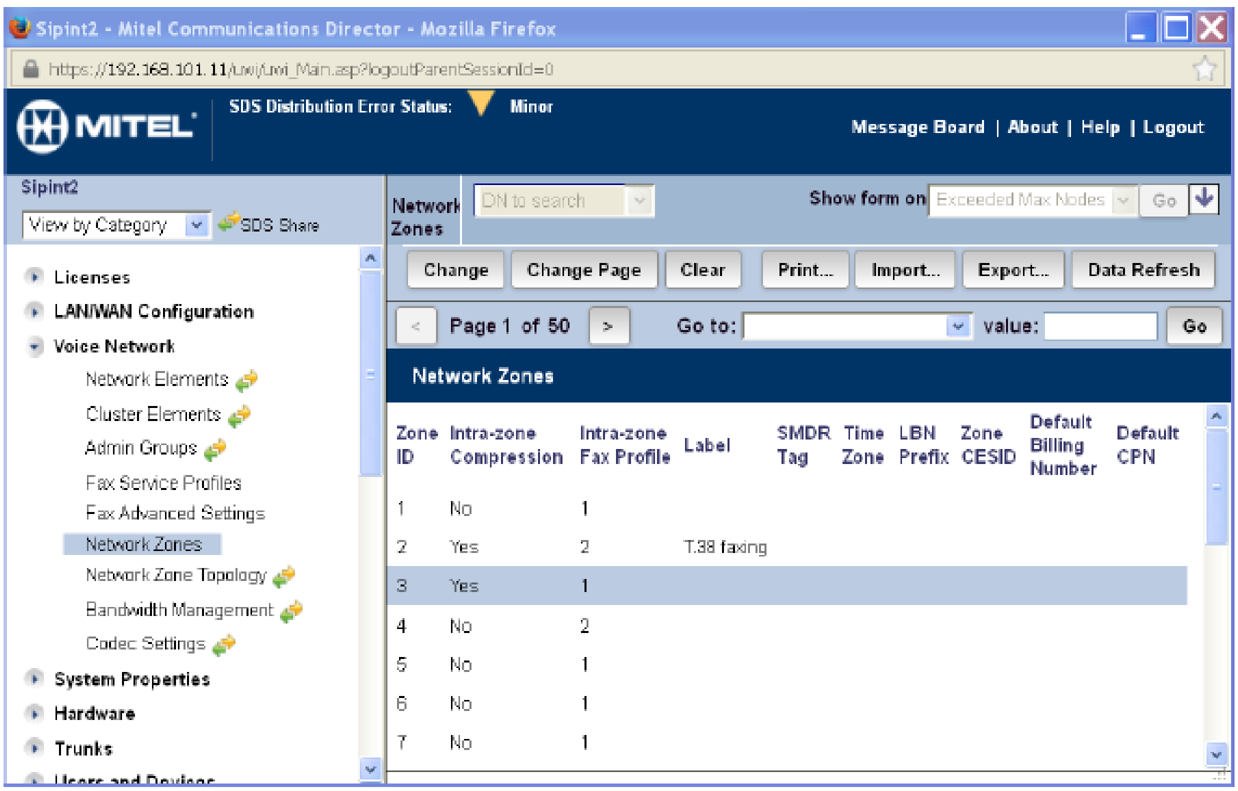


Figure 17 – Zone Assignment

MITEL BORDER GATEWAY SETUP

MiVB SETUP

1. To program an MIVB into the MBG, click on Service Configuration -> ICP's -> +.
2. Enter a name for the MIVB.
3. Enter the IP address of the MIVB and select the Type as Mitel Voice Business.

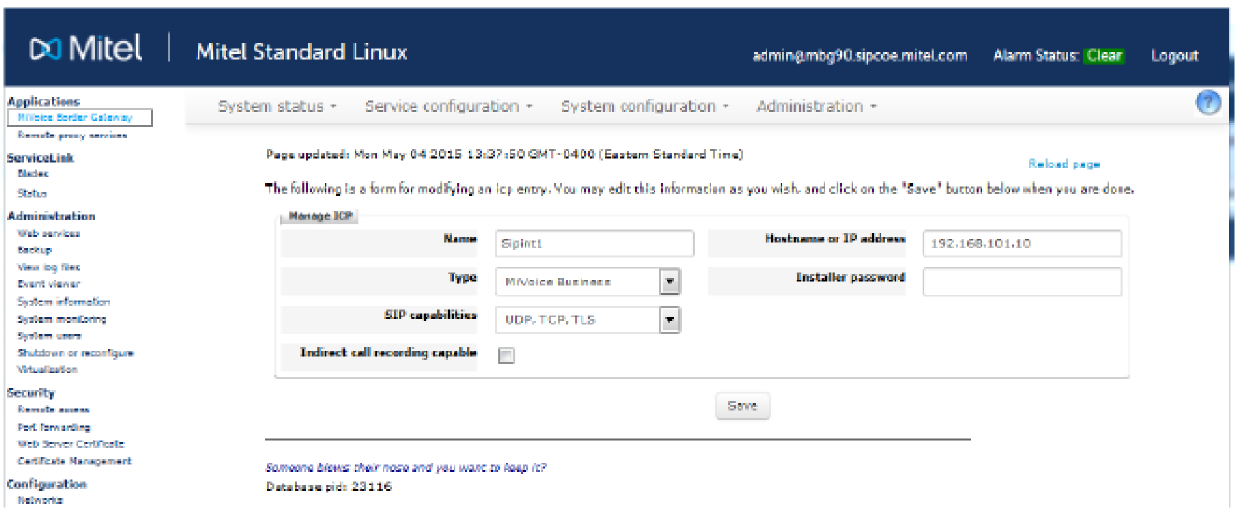


Figure 18 – ICP setup

SIP TRUNKS SETUP

1. Under the Services Configuration tab, click on SIP trunking and then click on the + sign to add a SIP trunk". Enter the SIP trunk's details for the as provided by Fusion.
    a. **Name** – is the name of the trunk.
    b. **Remote trunk endpoint address** – the public IP address/FQDN address of the provider's switch or gateway (this address should be given to you by the provider, e.g. Fusion).
    c. **Local/Remote RTP framesize (ms)** – is the packetization rate you want to set on this trunk, Auto or 20ms in this case.
    d. **PRACK** – Use master setting.
    e. **Routing rule one** – it allows routing of any digits to the selected MiVb's
2. The rest of the settings are optional and could be configured if required. Click **Save** button.

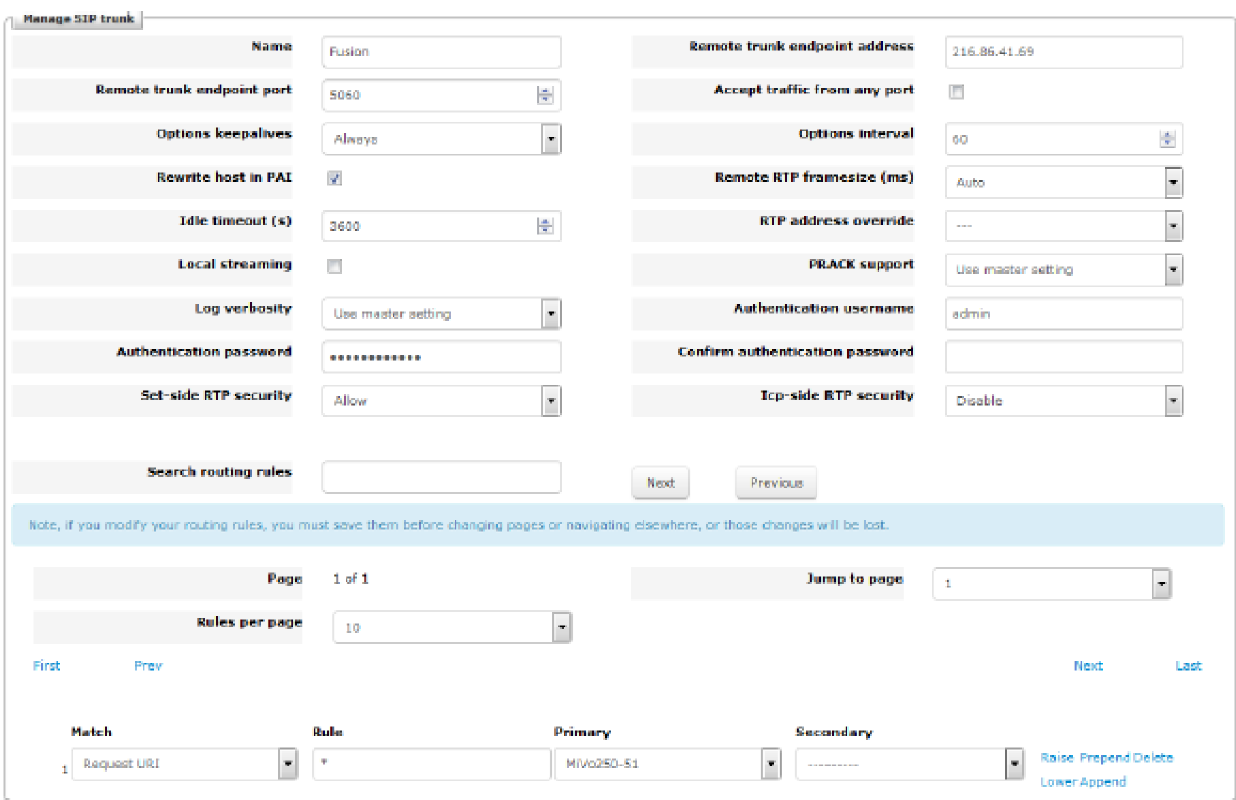


Figure 19 – Services - SIP Trunking Setup