

MITEL – SIP CoE

Technical Configuration Notes



Configure MCD 6.X for use with
VoiceHost SIP trunks

SIP CoE 13-4940-00284



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Mitel Technical Configuration Notes – Configure MCD for use with VoiceHost SIP trunks
December 2013, 13-4940-00284

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Overview


This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel 3300 ICP to connect to VoiceHost SIP trunks. 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 2013	Initial Interop with Mitel 3300 6.0 SP1 and VoiceHost

Interop Status

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

	<p>The most common certification which means VoiceHost SIP trunk 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.</p>
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Software & Hardware Setup


This was the test setup to generate a basic SIP call between VoiceHost trunk line and the 3300ICP.

Manufacturer	Variant	Software Version
Mitel	3300ICP MXe	12.0.1.24
Mitel	Minet sets: 5340, 5220, 5330	05.02.01.07
Mitel	MBG - Teleworker	8.0.12.0
Mitel	MBG - Gateway	8.0.12.0

Tested Features

This is an overview of the features tested during the Interop test cycle and not a detailed view of the test cases. Mitel Interop Test plan was executed during this testing

Feature	Feature Description	Issues
Basic Call	Making and receiving a call through the VoiceHost SIP trunk, call holding, transferring, conferencing, busy calls, long calls durations, variable codec.	
PRACK	Reliable Provisional Response	N/S
NuPoint Voicemail	Terminating calls to a NuPoint voicemail boxes and DTMF detection.	
Packetization	Forcing the 3300 ICP to stream RTP packets through its E2T card at different intervals, from 10ms to 60ms	
Personal Ring Groups	Receiving calls through VoiceHost SIP trunk to a personal ring group. Also moving calls to/from the prime member and group members.	
Teleworker	Making and receiving a call through VoiceHost SIP trunk to and from Teleworker extensions.	
Video	Making and receiving a call through VoiceHost SIP trunk with video capable devices.	
G 711 Faxing	Fax transmission with G 711 codec.	
T.38 Faxing	Fax transmission with protocol T.38	

 - 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 the VoiceHost SIP trunk is connected to Mitel 3300ICP.

Feature	Problem Description
Basic Call	VoiceHost does not support the g.729 codec. Recommendation: Do not use Intra-Zone compression.
Packetization	VoiceHost only supports a packetization rate of 20ms Recommendation: Set the packetization rate to the default 20ms value
Video	Although VoiceHost supports video calls on their SIP trunk interface video calls could not be tested due to test environment issues.
T.38 Faxing	Although VoiceHost supports T.38 FAX calls on their SIP trunk interface T.38 calls could not be tested due to test environment issues

Network Topology

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

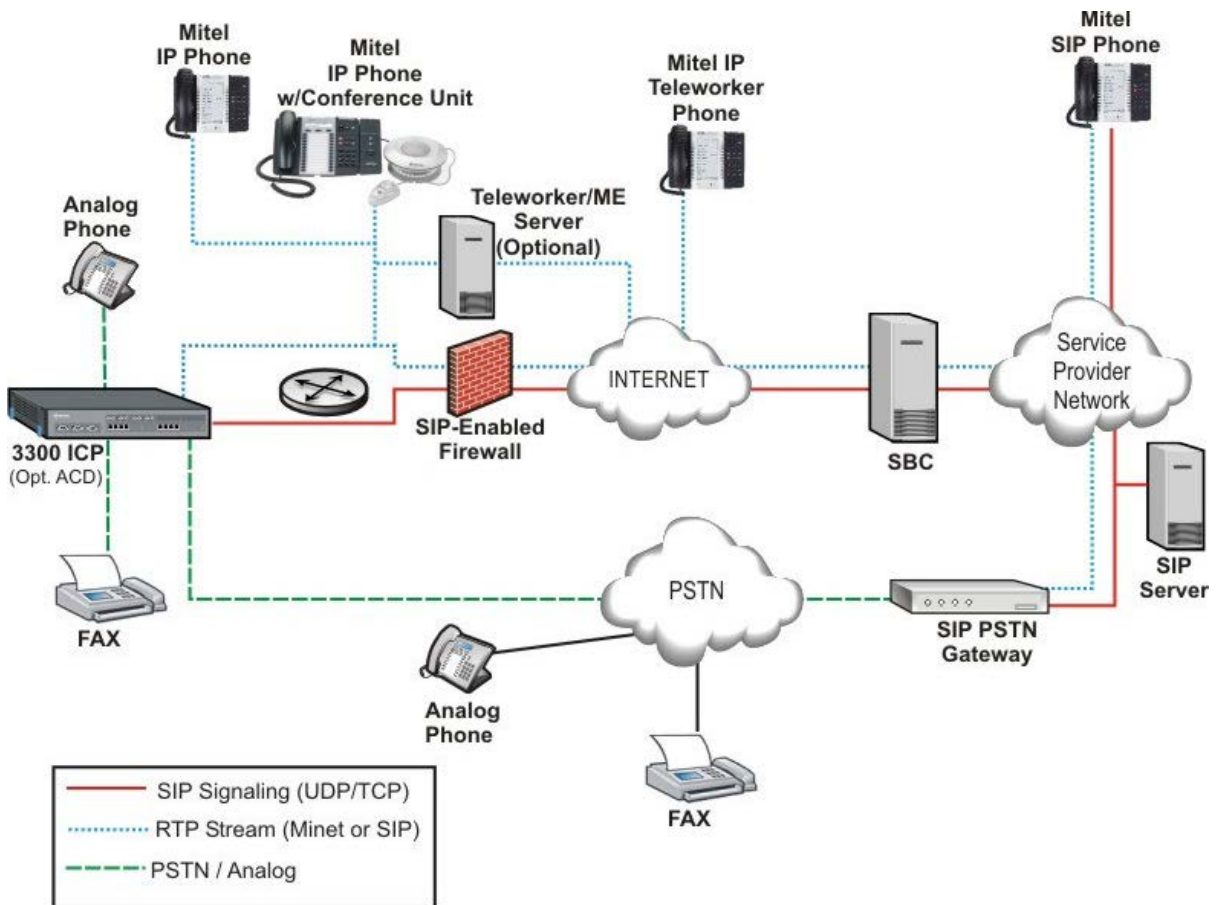


Figure 1 – Network Topology

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 the 3300ICP 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.

3300ICP Configuration Notes

The following steps show how to program a 3300 MCD to interconnect with VoiceHost 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 MCD 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 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 3300ICP Programming

- The SIP signaling connection uses UDP on Port 5060.

Licensing and Option Selection – SIP Licensing

Ensure that the 3300 ICP is equipped with enough SIP trunking licenses for the connection to the VoiceHost SIP trunks. 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 3300 to be used with all service providers, applications and SIP trunking devices.

The screenshot shows the MITEL Sipint2 web interface. The left sidebar contains a navigation menu with 'License and Option Selection' highlighted in red. The main content area displays the 'License and Option Selection' form for Application Record ID 35798030. The form includes a table of 'Licensed Options' with columns for System Type, License Sharing, Hardware Identifier, Locally Consumed, Locally Allocated, Available for Allocation, Purchased, Local Limits (Licenses Allowed), and Can be Over Allocated. The 'SIP Trunks' row is highlighted in red, showing 146 locally consumed licenses and 1000 locally allocated licenses.

System Type	License Sharing	Hardware Identifier	Locally Consumed	Locally Allocated	Available for Allocation	Purchased	Local Limits Licenses Allowed	Can be Over Allocated
Enterprise	No	0000002f9ee1						
Licensed Options								
Users								
IP Users			44	2000	100	2100	Unrestricted	Yes
External Hot Desk Users			2	20	80	100	Unrestricted	Yes
ACD Active Agents			1	100	0	100	Unrestricted	Yes
HTML Applications			0	100	400	500	Unrestricted	Yes
Analog Lines			0	10	0	10	Unrestricted	Yes
IP Console Active Operators			0	0	1	0	Unrestricted	Yes
Multi-device Users			0	0	20	0	Unrestricted	Yes
Multi-device Suites			0	0	20	0	Unrestricted	Yes
Messaging								
Embedded Voice Mail			18	100	0	100	Unrestricted	Yes
Embedded Voice Mail PMS			1	Yes	0	1	Unrestricted	Yes
Trunking/Networking								
Digital Links			0	2	14	16	Unrestricted	Yes
Compression				16	112	128	Unrestricted	Yes
FAX Over IP (T.38)				16	48	64	Unrestricted	Yes
SIP Trunks			146	1000	0	1000	Unrestricted	Yes

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 3300ICP.

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/FAX Machine (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**

The screenshot shows the 'Class of Service Options' form in the Sipint2 web interface. The form is for Class of Service Number 7, with Comment 'SIP Trunk'. The 'Busy Override Security' option is highlighted with a red box and set to 'Yes'. Other options include 'ACD Logout Agent No Answer Timer' (15), 'ACD Make Busy on Login' (No), 'ACD Silent Monitor Accept' (No), 'ACD Silent Monitor Allowed' (No), 'ACD Silent Monitor Notification' (No), 'Follow 2nd Alternate Reroute for Recall to Busy ACD Agent' (No), 'Work Timer' (0), 'Call Announce Line' (No), 'Off-Hook Voice Announce Allowed' (No), 'Handsfree AnswerBack Allowed' (No), 'Disable Executive Busy Override Tone' (No), and 'Executive Busy Override' (No).

Class Of Service Number	Comment
1	General
7	SIP Trunk

General | Advanced

Class Of Service Number: 7
 Comment: SIP Trunk

ACD

ACD Logout Agent No Answer Timer: 15
 ACD Make Busy on Login: No Yes
 ACD Silent Monitor Accept: No Yes
 ACD Silent Monitor Allowed: No Yes
 ACD Silent Monitor Notification: No Yes
 Follow 2nd Alternate Reroute for Recall to Busy ACD Agent: No Yes
 Work Timer: 0

Announce

Call Announce Line: No Yes
 Off-Hook Voice Announce Allowed: No Yes
 Handsfree AnswerBack Allowed: No Yes

Busy Override

Busy Override Security: No Yes
 Disable Executive Busy Override Tone: No Yes
 Executive Busy Override: No Yes

Figure 3 – Class of Service form

Network Elements

Create a network element for a SIP Peer (VoiceHost) as shown in **Figure 4**.

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

Our setup uses an external proxy. Set the address for you installation appropriately.

In our setup the SIP trunks used authentication.

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

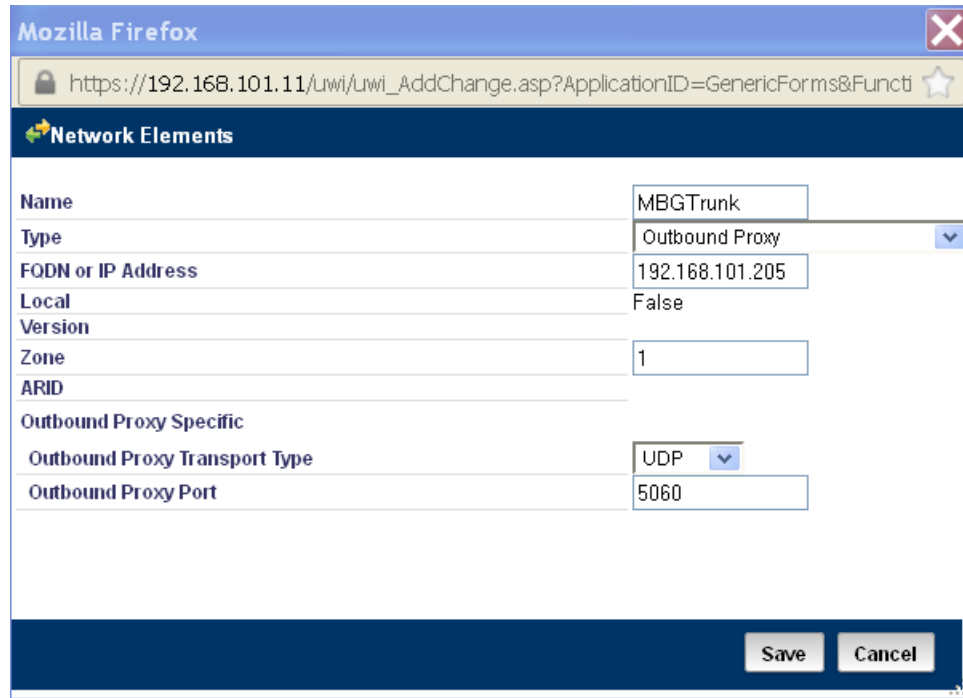
Network Elements	
Name	VoiceHost
Type	Other
FQDN or IP Address	st.sipconvergence
Local Version	False
Zone	1
ARID	
SIP Peer	<input checked="" type="checkbox"/>
SIP Peer Specific	
SIP Peer Transport	default
SIP Peer Port	5060
External SIP Proxy FQDN or IP Address	st.sipconvergence
External SIP Proxy Transport	default
External SIP Proxy Port	0
SIP Registrar FQDN or IP Address	st.sipconvergence
SIP Registrar Transport	default
SIP Registrar Port	5060
SIP Peer Status	Auto-Detect/Normal

Save Cancel

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 3300ICP will require knowledge of this by programming the Proxy as a network element then referencing this proxy in the SIP Peer Profile form (later in this document).



Network Elements	
Name	MBGTrunk
Type	Outbound Proxy
FQDN or IP Address	192.168.101.205
Local	False
Version	
Zone	1
ARID	
Outbound Proxy Specific	
Outbound Proxy Transport Type	UDP
Outbound Proxy Port	5060

Save Cancel

Figure 5 – Network Element (Proxy)

Trunk Attributes (trunk service number)

The Trunk Attributes is defined for Trunk Service Number (2), which will be used to direct incoming calls to an answer point in the 3300ICP.

Set the number of Class of Service that was configured in the section above (1).

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 3300ICP will absorb the first 6 digits of the DID number received from the VoiceHost SIP trunk leaving 4 digits for the 3300 to translate and ring the 4-digit extension.

For example, if the VoiceHost SIP trunk delivers number 613-592-5660 to the 3300. The 3300 will absorb the first 6 digits (613-592) leaving the Mitel 3300 to ring extension 5660. Extension 5660 must be programmed as a valid dialable number in the 3300ICP. As an alternative way, you can create a System Speed Call number to associate number 5660 with the real telephone extension on 3300ICP. Please refer to the 3300 System Administration documentation for further programming information.

Trunk Attributes	
Trunk Service Number	2
Release Link Trunk	No
Call Recognition Service	Off
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	6
Dial In Trunks Incoming Digit Modification - Insert	
Dial In Trunks Answer Point	
Dial In Trunks Insert Forwarding Information	<input checked="" type="radio"/> No <input type="radio"/> Yes
Trunk Label	VoiceHost

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 3300ICP 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 "VoiceHost" Network Element.

Registration User Name: VoiceHost uses registration so fill this field in with the information provided to you.

Address Type: Select the IP Address of your Mitel 3300ICP.

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

Outbound Proxy Server: Select the Network Element previously configured for the Outbound Proxy Server ("MBG Trunk" in our test environment).

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, **2** in this configuration.

Authentication Options: In this example proxy server authentication was used therefore the user name and password must be filed in. This should not be confused with incoming call authentication.

SIP Peer Profile						
Voice4Net	Voice4Net	No	4	90	1	
VoiceHost	VoiceHost	MBGTrunk	No	2	90	1
View On	View On	MBGTrunk	No	22	90	1

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	VoiceHost
Network Element	VoiceHost
Local Account Information	
Registration User Name	ST16717T001
Address Type	FQDN: sipint2.sipcoo.mitel.com
Administration Options	
Interconnect Restriction	1
Maximum Simultaneous Calls	5
Minimum Reserved Call Licenses	0
Administration Options	
Outbound Proxy Server	MBGTrunk
SMDR Tag	0
Trunk Service	2
Zone	1
User Name	ST16717T001
Password	*****
Confirm Password	*****
Authentication Option for Incoming Calls	No Authentication
Subscription User Name	
Subscription Password	*****
Subscription Confirm Password	*****

Figure 7 – SIP Peer Basic form

Call Routing (Figure 8):

Leave the default settings intact, as shown.

SIP Peer Profile					
Voice4Net	Voice4Net	No	4	90	1
VoiceHost	VoiceHost	MBGTrunk No	2	90	1
View Or	View Or	MBGTrunk No	22	90	1
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 No Only Allow Outgoing Calls No Private SIP Trunk No Reject Incoming Anonymous Calls No Route Call Using To Header No					

Figure 8 – SIP Peer Profile Call Routing

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.

SIP Peer Profile					
Voice4Net	Voice4Net	No	4	90	1
VoiceHost	VoiceHost	MBGTrunk No	2	90	1
View Or	View Or	MBGTrunk No	22	90	1
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 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 Calling Line Id

SDP Options (Figure 10):

Allow Peer to use Multiple Active M-Lines “YES”

Allow Using Update for Early Media Renegotiation “YES”

Avoid Signaling Hold to the PEER to “YES”

Enable Mitel Proprietary SDP to “NO”

Limit to one Offer/Answer per INVITE to “YES”

NAT Keepalive to “YES”

Prevent the Use of IP Address 0.0.0.0 in SDP Messages to “YES”

Leave the other options at the default settings unless there is a specific reason to change them.

SIP Peer Profile																																						
Voice4Net	Voice4Net	No	4	90	1																																	
VoiceHost	VoiceHost	MBGTrunk No	2	90	1																																	
VowOr	VowOr	MBGTrunk No	22	90	1																																	
<div style="display: flex; justify-content: space-between; border-bottom: 1px solid black;"> Basic Call Routing Calling Line ID SDP Options Signaling and Header Manipulation Timers </div> <div style="display: flex; justify-content: space-between; border-bottom: 1px solid black;"> Key Press Event Outgoing DID Ranges Profile Information </div>																																						
<table border="0" style="width: 100%;"> <tr> <td style="width: 80%;">Allow Peer To Use Multiple Active M-Lines</td> <td>Yes</td> </tr> <tr> <td>Allow Using UPDATE For Early Media Renegotiation</td> <td>Yes</td> </tr> <tr> <td>Avoid Signaling Hold to the Peer</td> <td>Yes</td> </tr> <tr> <td>Enable Mitel Proprietary SDP</td> <td>No</td> </tr> <tr> <td>Force sending SDP in initial Invite message</td> <td>No</td> </tr> <tr> <td>Force sending SDP in initial Invite - Early Answer</td> <td>No</td> </tr> <tr> <td>Ignore SDP in Unreliable Provisional Responses</td> <td>No</td> </tr> <tr> <td>Limit to one Offer/Answer per INVITE</td> <td>Yes</td> </tr> <tr> <td>NAT Keepalive</td> <td>Yes</td> </tr> <tr> <td>Prevent the Use of IP Address 0.0.0.0 in SDP Messages</td> <td>Yes</td> </tr> <tr> <td>Renegotiate SDP To Enforce Symmetric Codec</td> <td>No</td> </tr> <tr> <td>Repeat SDP Answer If Duplicate Offer Is Received</td> <td>No</td> </tr> <tr> <td>RTP Packetization Rate Override</td> <td>No</td> </tr> <tr> <td>RTP Packetization Rate</td> <td>20ms</td> </tr> <tr> <td>Special handling of Offers in 2XX responses (INVITE)</td> <td>No</td> </tr> <tr> <td>Suppress Use of SDP Inactive Media Streams</td> <td>No</td> </tr> </table>							Allow Peer To Use Multiple Active M-Lines	Yes	Allow Using UPDATE For Early Media Renegotiation	Yes	Avoid Signaling Hold to the Peer	Yes	Enable Mitel Proprietary SDP	No	Force sending SDP in initial Invite message	No	Force sending SDP in initial Invite - Early Answer	No	Ignore SDP in Unreliable Provisional Responses	No	Limit to one Offer/Answer per INVITE	Yes	NAT Keepalive	Yes	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	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
Allow Peer To Use Multiple Active M-Lines	Yes																																					
Allow Using UPDATE For Early Media Renegotiation	Yes																																					
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Enable Mitel Proprietary SDP	No																																					
Force sending SDP in initial Invite message	No																																					
Force sending SDP in initial Invite - Early Answer	No																																					
Ignore SDP in Unreliable Provisional Responses	No																																					
Limit to one Offer/Answer per INVITE	Yes																																					
NAT Keepalive	Yes																																					
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																																					
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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 SDP Options

Signaling and Header Manipulation (Figure 11):

Figure 11 shows the settings used for VoiceHost SIP trunk interop testing. Ensure that the option in your configuration match these.

SIP Peer Profile					
Voice4Net	Voice4Net	No	4	90	1
VoiceHost	VoiceHost	MBGTrunk No	2	90	1
Voice4Net	Voice4Net	MBGTrunk No	22	90	1

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
E.164: Enable sending '+'	No
E.164: Add '+' if digit length > N digits	12
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
Only use SDP to decide 180 or 183	Yes
Prefer From Header for Caller ID	No
Require Reliable Provisional Responses on Outgoing Calls	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-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

asp?parseOnClient="true"#tab

Figure 11 – SIP Peer Profile Signaling and Header Manipulation

Timers (Figure 12):

Session Timers: Figure 12 shows how the timers were set for our test environment. These may vary for other installations.

SIP Peer Profile					
VoiceHost	VoiceHost	MBGTrunk No	2	90	1
VovOv	VovOv	MBGTrunk No	33	90	1
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					90
Subscription Period					180
Subscription Period Minimum					300
Subscription Period Refresh (%)					80
Invite Ringing Response Timer					0

Figure 12 – SIP Peer Profile Timers

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 3300 ICP with Not Found message.

To deliver such calls to Mitel's extensions, make sure to associate VoiceHost's DID number(s) with the SIP Peer Profile we configured earlier. See **Figure 13** as a guide.

SIP Peer Profile Assignment by Incoming DID	
441235380003,448435573962	VoiceHost
Incoming DID Range	441235380003,448435573962
SIP Peer Profile Label	VoiceHost
Comment	

Figure 13 – SIP Peer Profile Assignment by Incoming DID form

ARS Digit Modification Plan

Ensure that Digit Modification for outgoing calls to VoiceHost SIP trunk absorbs or inject additional digits according to your dialing plan. In our test environment, we will be absorbing 1 digits and will not inject any digits, as shown in **Figure 14**.

As per our test environment, we need to dial **9** to access VoiceHost SIP trunk; thus, digits 9 will be absorbed and no digits will be preceding the dialled number. For instance, if caller dials 9613555660, 3300 ICP will send to the SIP trunk the following: 613555660.

ARS Digit Modification Plans	
Digit Modification Number	3
Number of Digits to Absorb	<input type="text" value="1"/>
Digits to be Inserted	<input type="text"/>
Final Tone Plan/Information Marker	<input type="text"/>

Save Cancel

Figure 14 – ARS Digit Modification form

ARS Routes

Create a route to VoiceHost SIP trunk. In this test environment, the SIP trunk is assigned to Route Number **27**. Choose **SIP Trunk** as a routing medium and choose the SIP Peer Profile and ARS Digit Modification entry created earlier.

ARS Routes	
Route Number	27
Routing Medium	SIP Trunk
Trunk Group Number	
SIP Peer Profile	VoiceHost
PBX Number / Cluster Element ID	
COR Group Number	1
Digit Modification Number	3
Digits Before Outpulsing	
Route Type	
Compression	Off

Save Cancel

Figure 15 – 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 905, the call will be routed to VoiceHost SIP trunk (i.e. to Route 18). For outbound calling, 3300 ICP expects 10 digits to be dialed after dialing of 905. See **Figure 16** for details.

Help

Change Range Programming - ARS Digits Dialed

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
905	10	Route	18

1. Enter the number of records to change:

2. Define the Change Range Programming Pattern:

Field Name	Change action	Value to change	Increment by
Digits Dialed	Change to	9	<input style="width: 50px;" type="text"/>
Number of Digits to Follow	Change to	10	-
Termination Type	Change to	Route	-
Termination Number	Change to	27	<input style="width: 50px;" type="text"/>

Preview
Save
Cancel

Figure 16 – ARS Digit Dialed form

Fax Configuration

VoiceHost uses the inter-zone FAX profile. 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).

The screenshot shows the MITEL configuration interface for Node 'sipint3'. The top status bar indicates a Major alarm on 2009-Dec-03 at 06:26:06. The main configuration area is titled 'Fax Configuration on sipint3'. It features a 'Change' button and a 'Data Refresh' button. The 'Inter-Zone Fax Profile' section displays the following settings:

- Maximum Fax Rate: 14400 (V.17, 14400bps)
- High Speed Redundancy: 0
- Low Speed Redundancy: 3
- Error Correction Mode (ECM): Disabled
- Override Non-Standard Facilities (NSF): Disabled
- Label: Inter-zone

Below this is a navigation bar showing 'Page 1 of 7' and a search field. The 'Intra-Zone Fax Profiles' section contains a table with the following data:

Profile	Maximum Fax Rate	High Speed Redundancy	Low Speed Redundancy	Error Correction Mode	NSF Override	NSF Vendor Code Value	NSF Country Code Value
1	-	-	-	-	-	-	-
2	14400 (V.17, 14400bps)	0	3	Disabled	Disabled	.	.
3	7200 (V.29, 7200bps)	0	3	Disabled	Disabled	.	.
4
5
6
7
8
9
10

Figure 22 – 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”. VoiceHost Communications uses the Inter-zone FAX Profile.

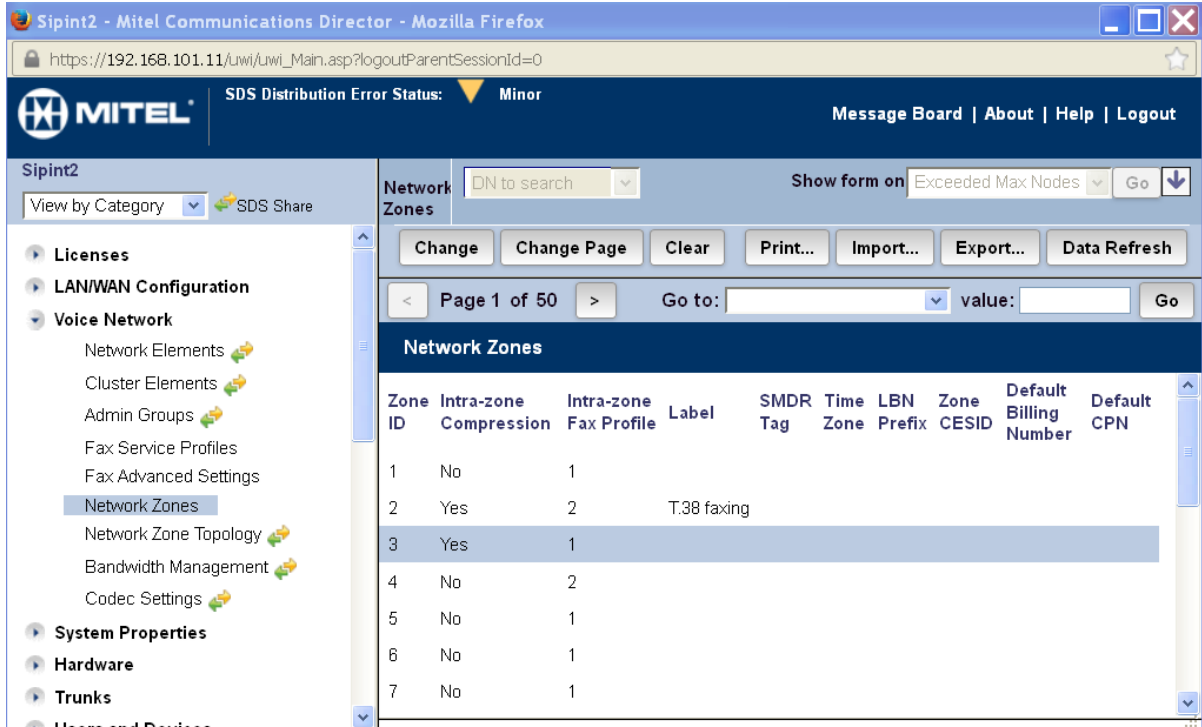


Figure 23 – Zone Assignment

Mitel Border Gateway Configuration Notes (Optional)

This section explains how to configure Mitel Border Gateway (MBG) if you use it as a SIP-aware gateway.

Firstly, you need to identify or add “the working” 3300 ICP where MBG will forward SIP messages to and then to configure the SIP trunk.

To do this:

- Login to the MBG and click Mitel Border Gateway.
- In the right pane, click the **Configure** tab and then **ICP's** (see **Figure 17** for details).
- On the **ICP's** page ensure that the “working” 3300ICP is configured. If needed, click the **Add ICP** link and add a new Mitel switch.
- Click the **Update** button when complete.

The screenshot shows the Mitel Standard Linux web interface. The left sidebar contains navigation menus for Applications, ServiceLink, Administration, Security, and Configuration. The main content area is titled 'Manage Mitel Border Gateway' and has several tabs: Status, Configuration (selected), Services, Applications, and Clustering. Under the Configuration tab, there are sub-tabs: Settings, Network profiles, ICPs (selected), IP Translations, Bandwidth management, and Alarms. The page displays a table of ICP information with one entry highlighted in red:

Default for MiNet	Default for SIP	Name	Hostname or IP address	Type	Installer password	Indirect call recording capable	Indirect call recording password		
<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	spint2	192.168.101.11	MCD		X		Modify	Delete

Below the table is an 'Add ICP' link and an 'Update Default ICPs' button. The footer of the page reads: 'Mitel Standard Linux 9.4.28.0 Copyright 1999-2012 Mitel Corporation All rights reserved.'

Figure 17 – ICP's Configuration page

To add a new SIP trunk:

- Click **Services** tab and then click **SIP trunking**
- Click **Add a SIP trunk** link (see **Figure 18**)

SIP Trunk VoiceHost_reg

Trunk status ●

Remote trunk endpoint st.sipconvergence.co.uk : 5060

Send options keepalives Use master setting

Options interval 60

Rewrite host in PAI True

Remote RTP framesize (ms) 20

Idle timeout (s) 3600

Re-invite filtering Off

RTP address override

Local streaming False

PRACK support Use master setting

Log verbosity Use master setting

Authentication username

Authentication password

Routing rules

Rule number	Header match rule	Pattern	Primary destination	Secondary destination
1	req	*	sipint2	None

Filter rules list (Pattern or destination)

Metrics

Calls in progress	Calls per hour	Seconds idle	Active transactions	Transaction errors
0 Max: 2	0 Max: 359	62	0	2

[Reset metrics](#)

Figure 18 – SIP Trunking Configuration Page

Enter the SIP trunk's details as shown in **Figure 18**:

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. VoiceHost).

Local/Remote RTP framesize (ms) – is the packetization rate you want to set on this trunk. Ensure that this option is set to 20ms or Auto.

Disable PRACK – VoiceHost Does not support PRACK

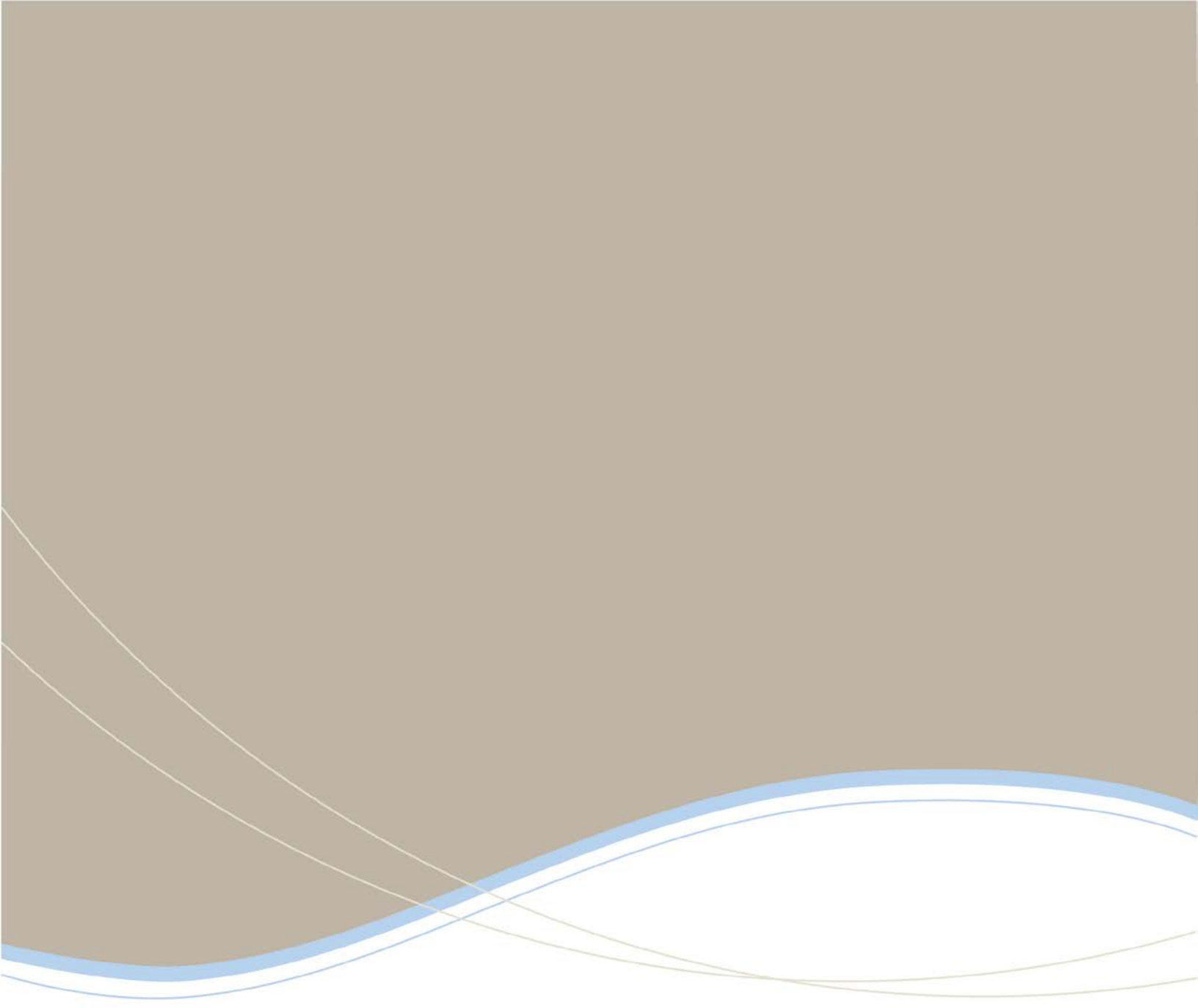
Routing rule one – it allows routing of any digits to the selected Mitel 3300 ICP

VoiceHost uses Authentication - Fill in the user name and password as provided if required.

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

In some installations you may require 2 SIP trunk configuration entries to handle incoming and outgoing calls differently.

Click **Save** button



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