MITEL – SIP CoE

Technical

Configuration Notes

Configure MCD 6.X for use with VoiceHost SIP trunks

SIP CoE 13-4940-00284

NOTICE

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

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:

COMPATIBLE	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.

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 defaut 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.

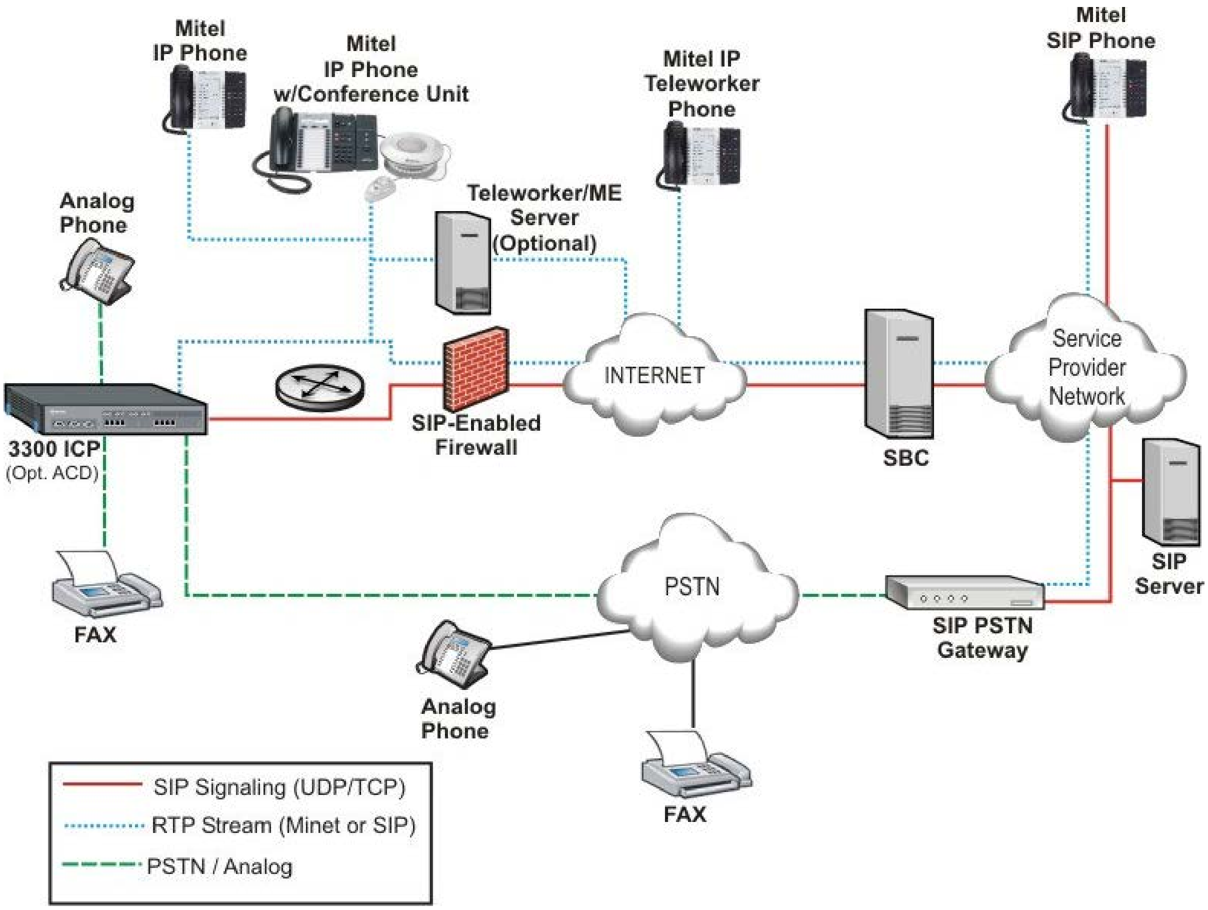

Figure 1 – Network Topology

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.

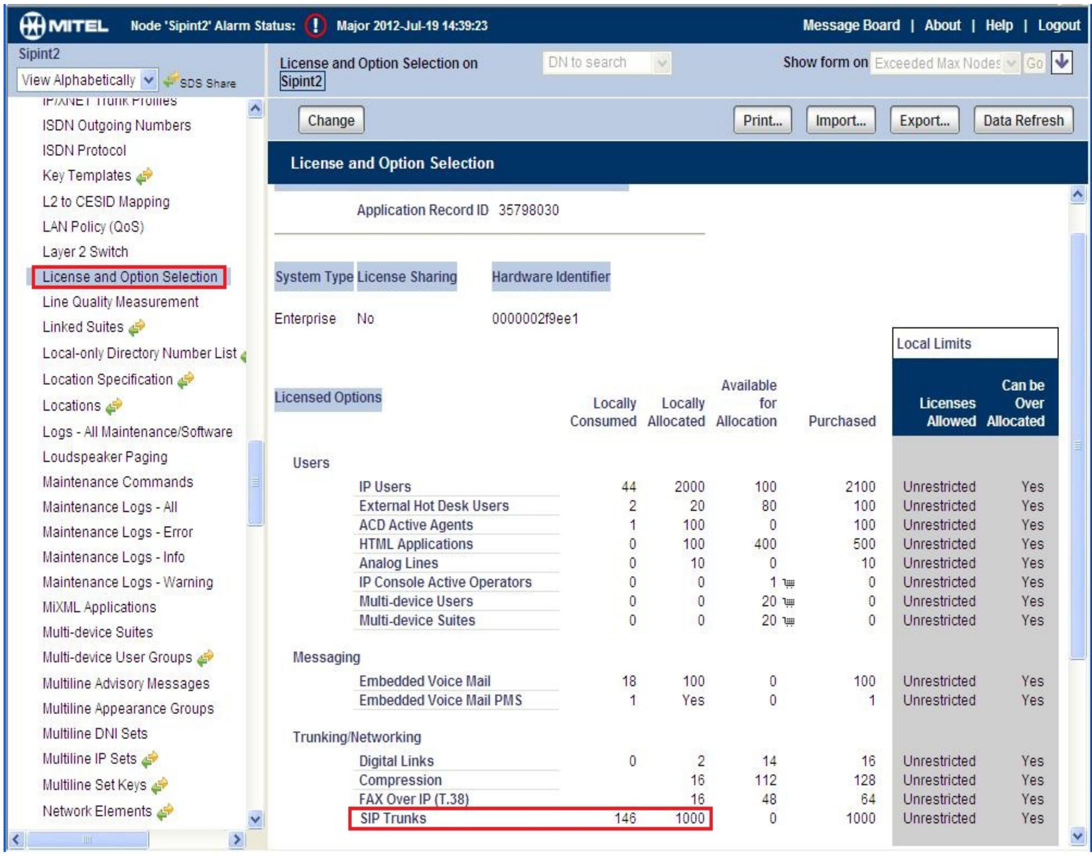

Figure 2 – License and Option Selection form

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**

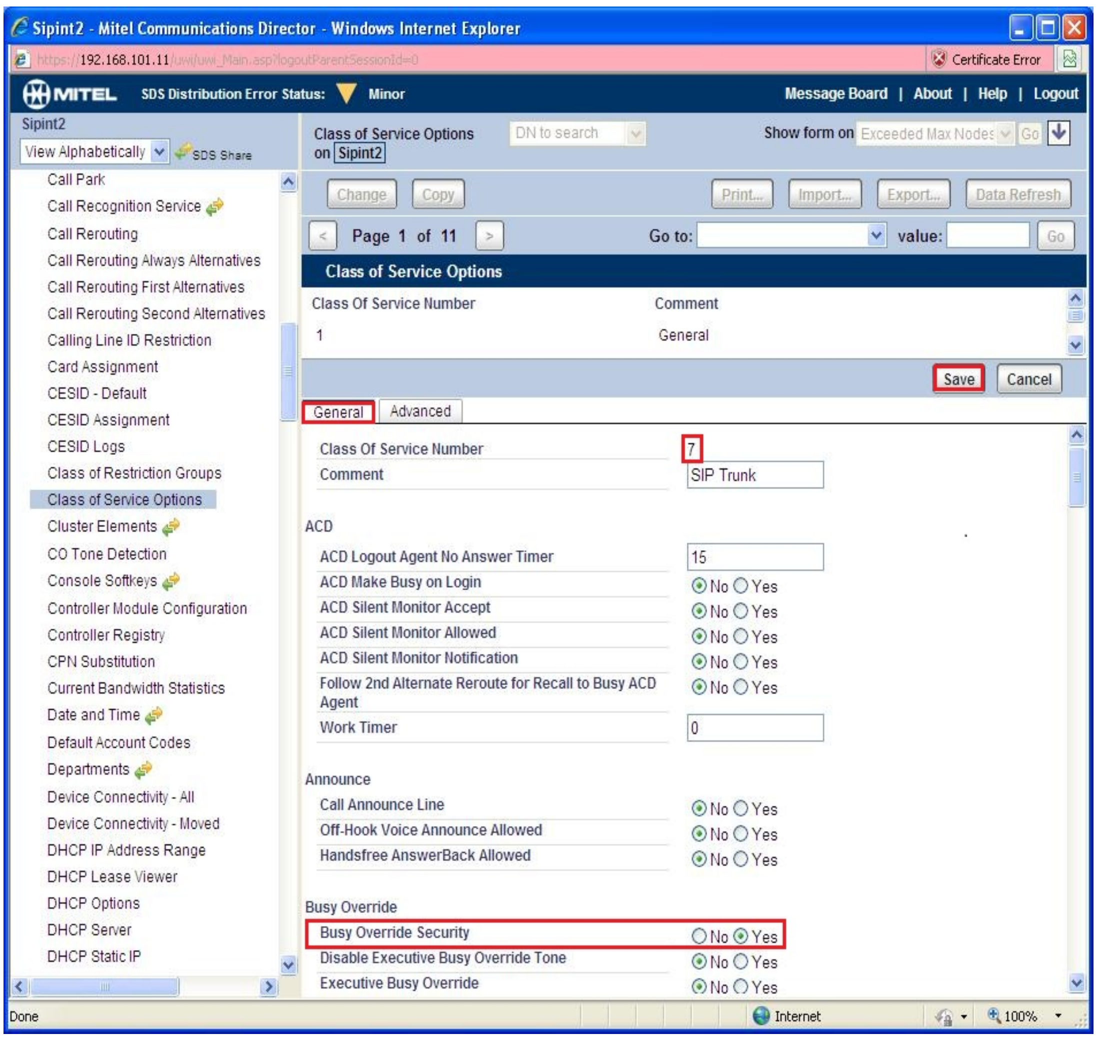

Figure 3 – Class of Service form

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.

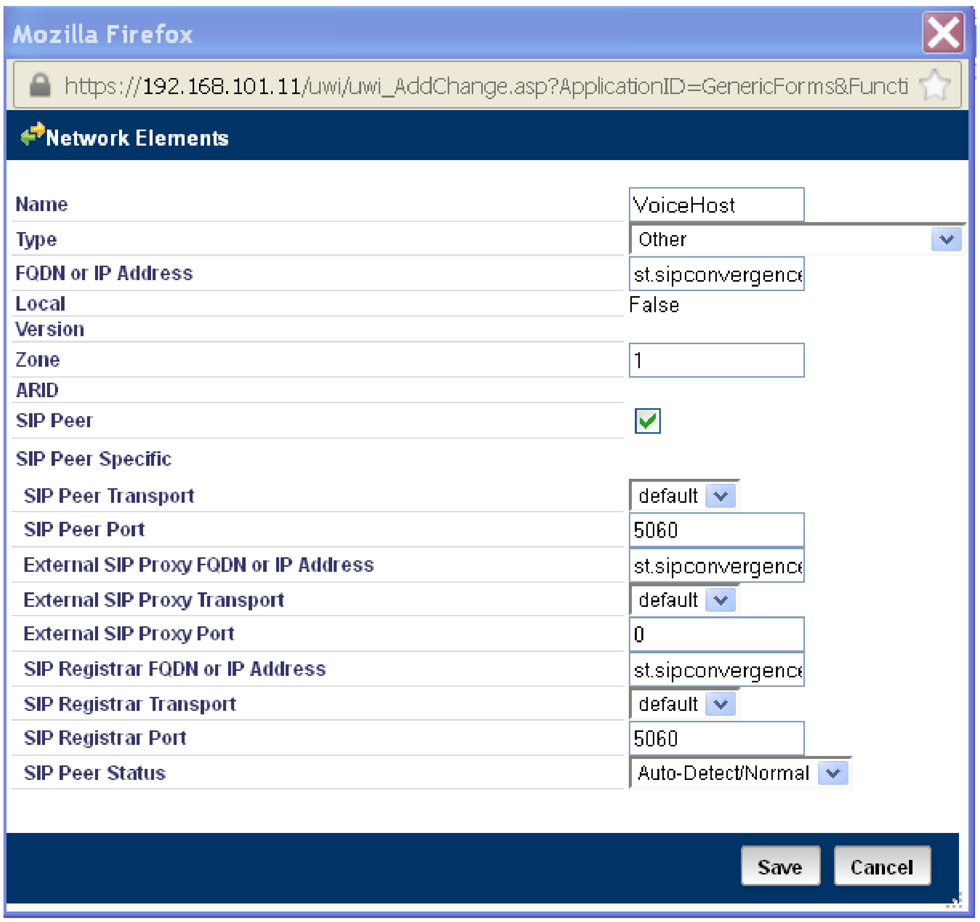

Figure 4 – Network Element form

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).

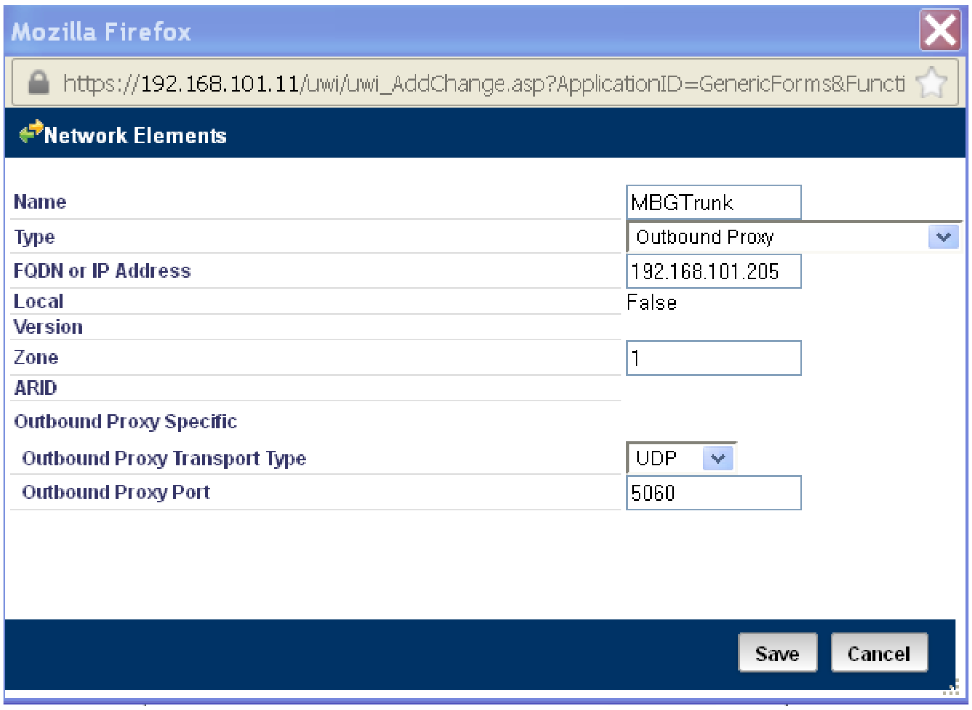

Figure 5 – Network Element (Proxy)

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.

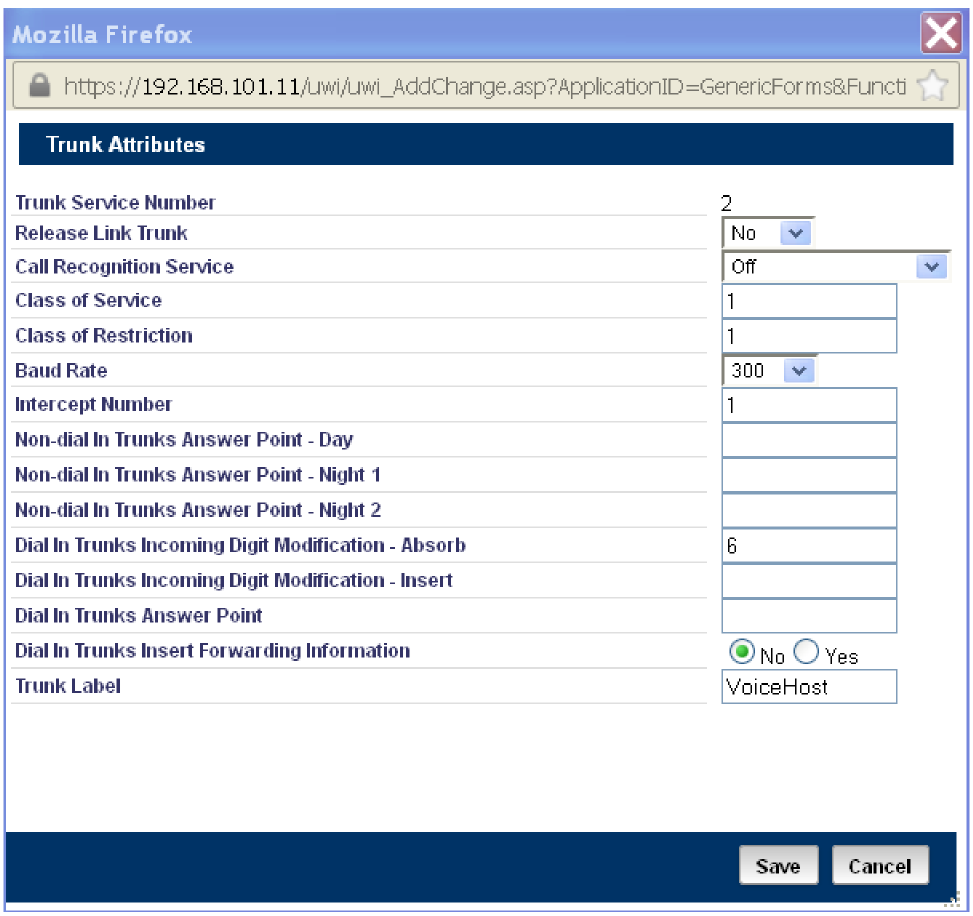

Figure 6 – Trunk Attributes (trunk service number)

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.

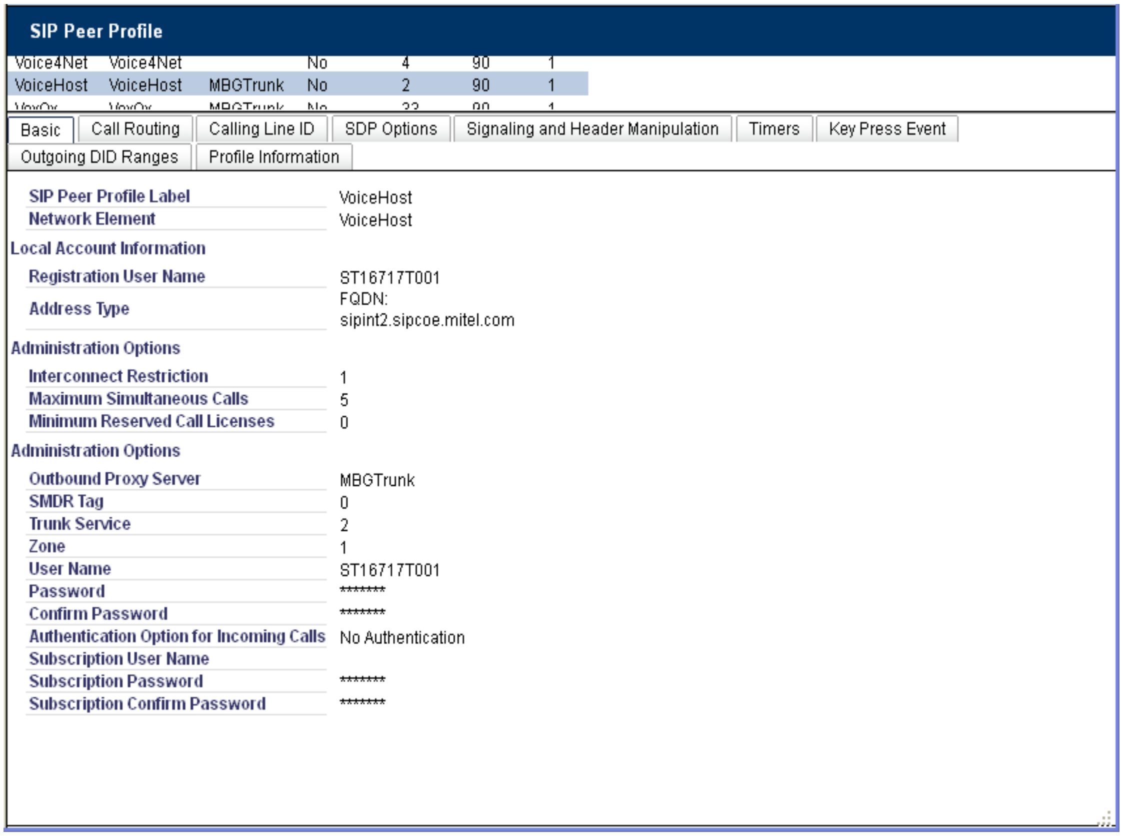

Figure 7 – SIP Peer Basic form

Call Routing (**Figure 8**):

Leave the default settings intact, as shown.

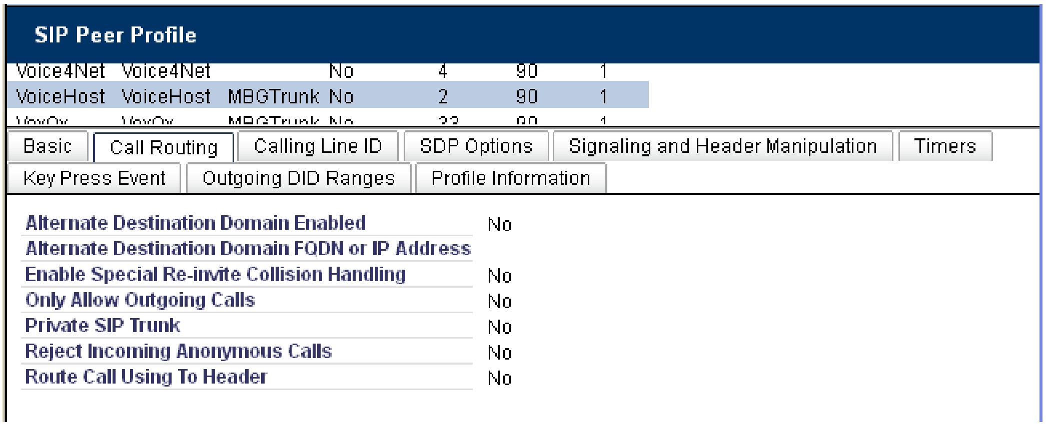

Figure 8 – SIP Peer Profile Call Routing

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.

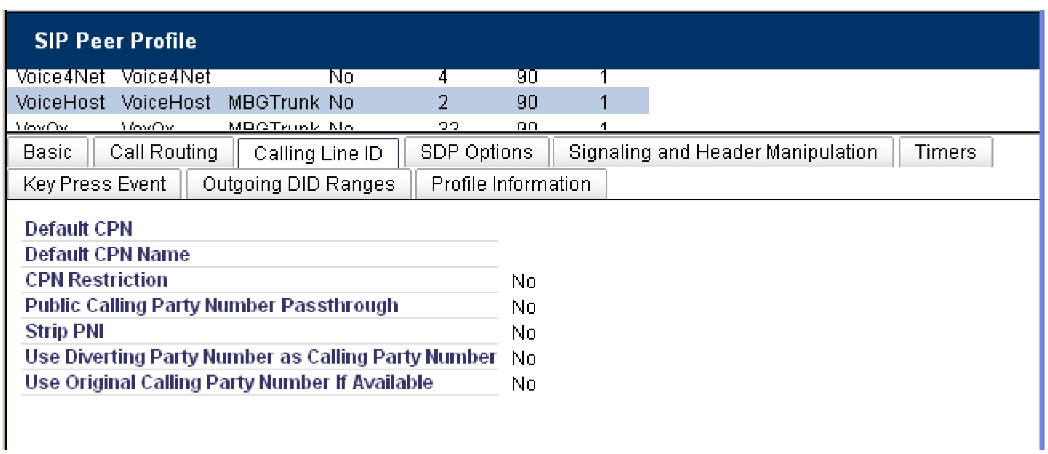

Figure 9 – SIP Peer Profile Calling Line Id

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.

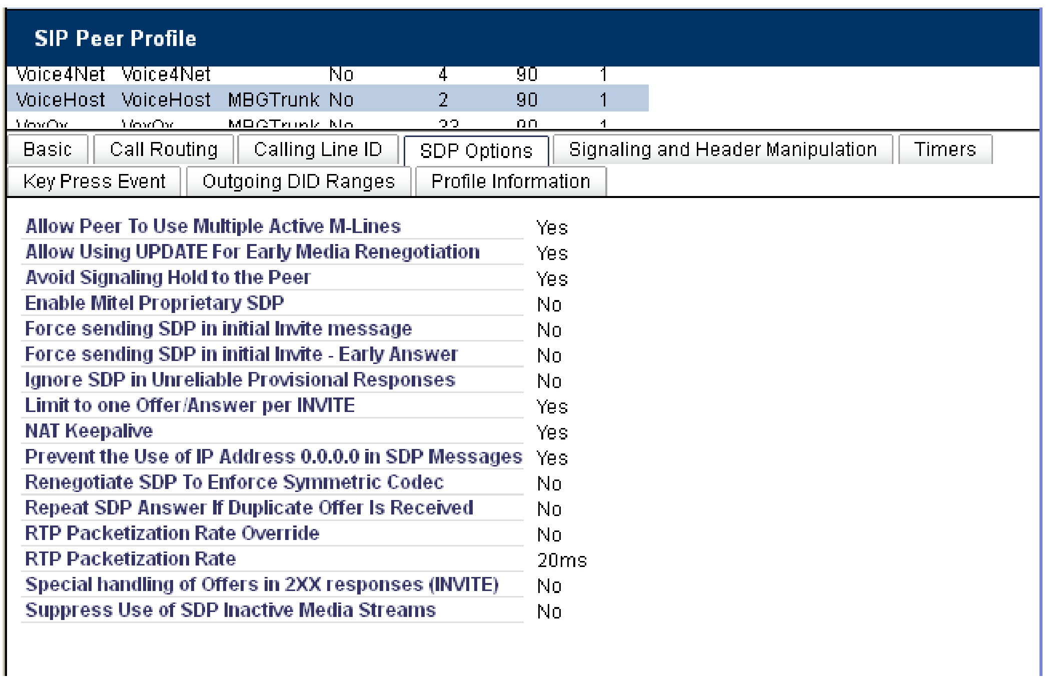

Figure 10 – SIP Peer Profile SDP Options

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.

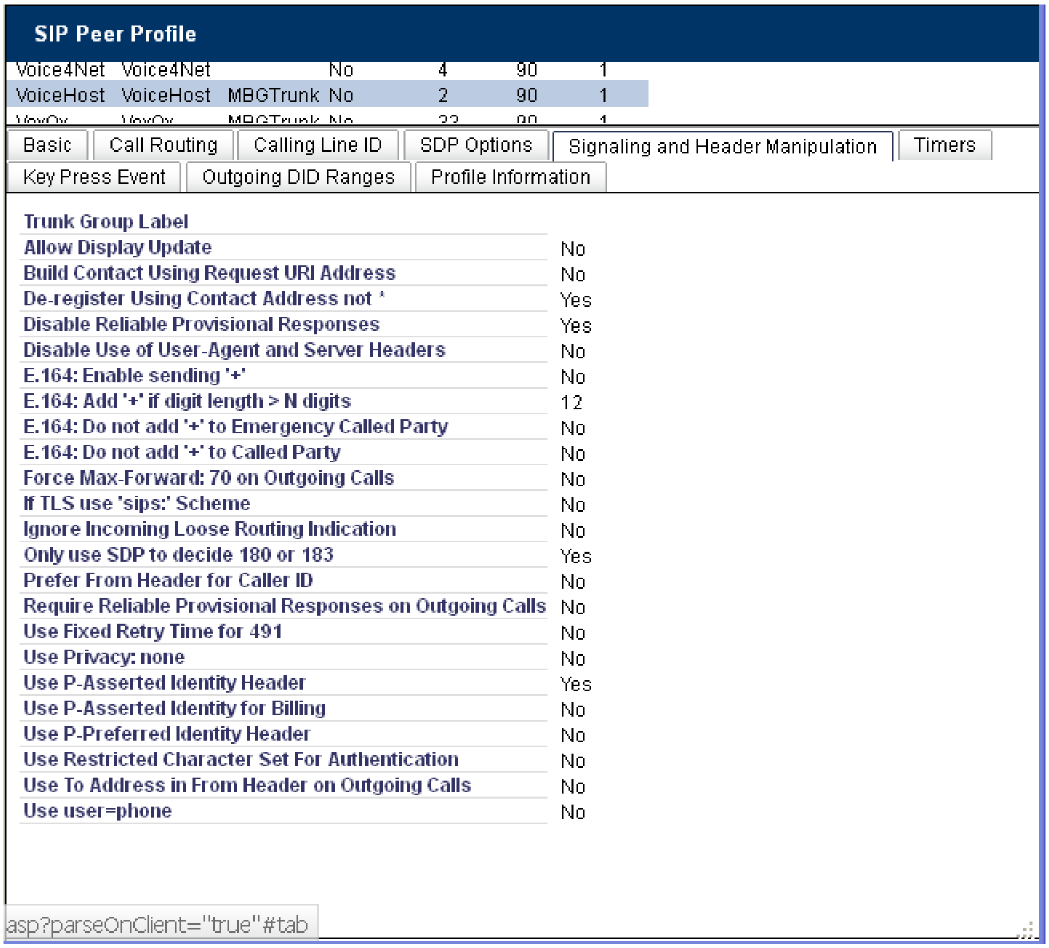

Figure 11 – SIP Peer Profile Signaling and Header Manipulation

Timers (**Figure 12**):

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

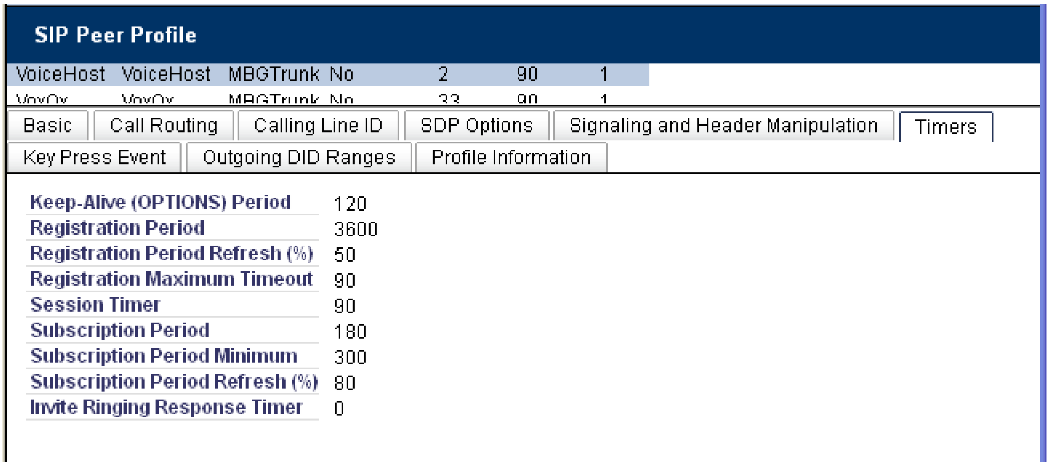

Figure 12 – SIP Peer Profile Timers

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.

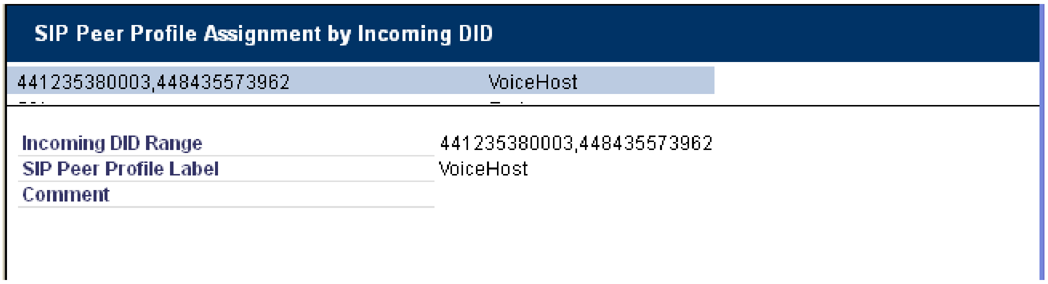

Figure 13 – SIP Peer Profile Assignment by Incoming DID form

ARS Digit Modification Plan

Ensure that Digit Modification for outgoing calls to VoiceHost SIP trunk absorbs or inject additional digits according to your dialling 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 96135555660, 3300 ICP will send to the SIP trunk the following: 6135555660.

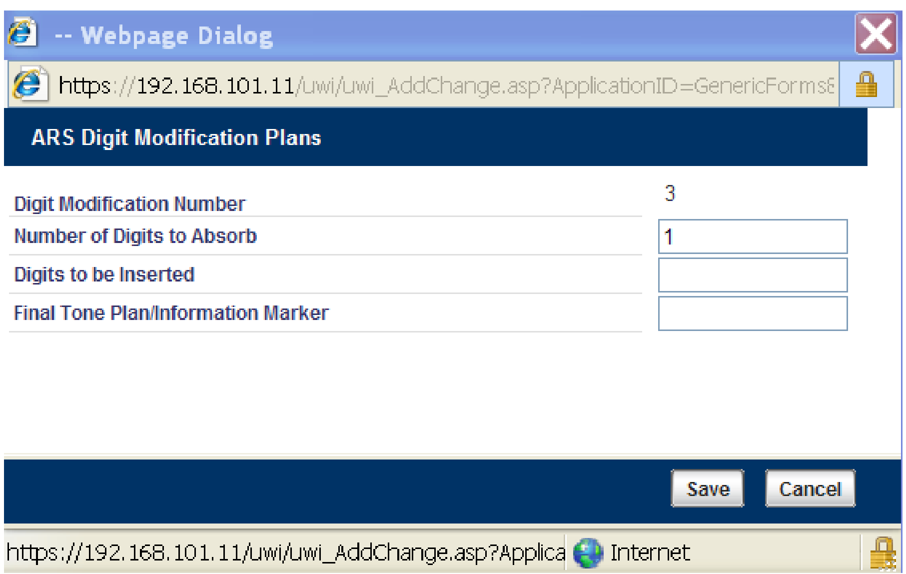

Figure 14 – ARS Digit Modification form

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.

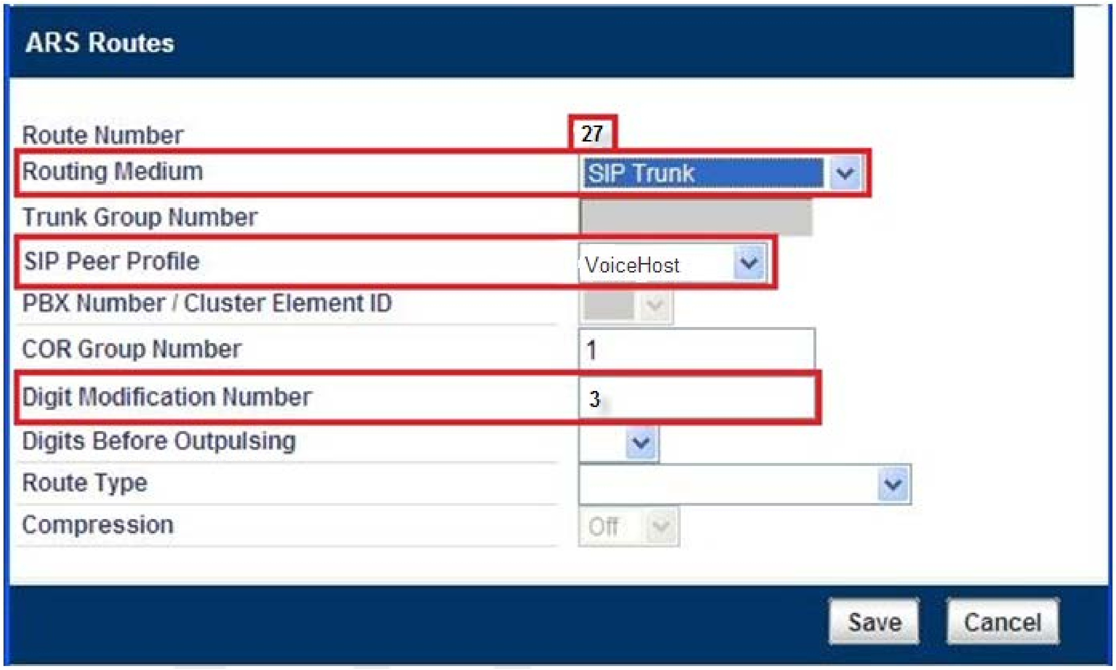

Figure 15 – ARS Route form

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.

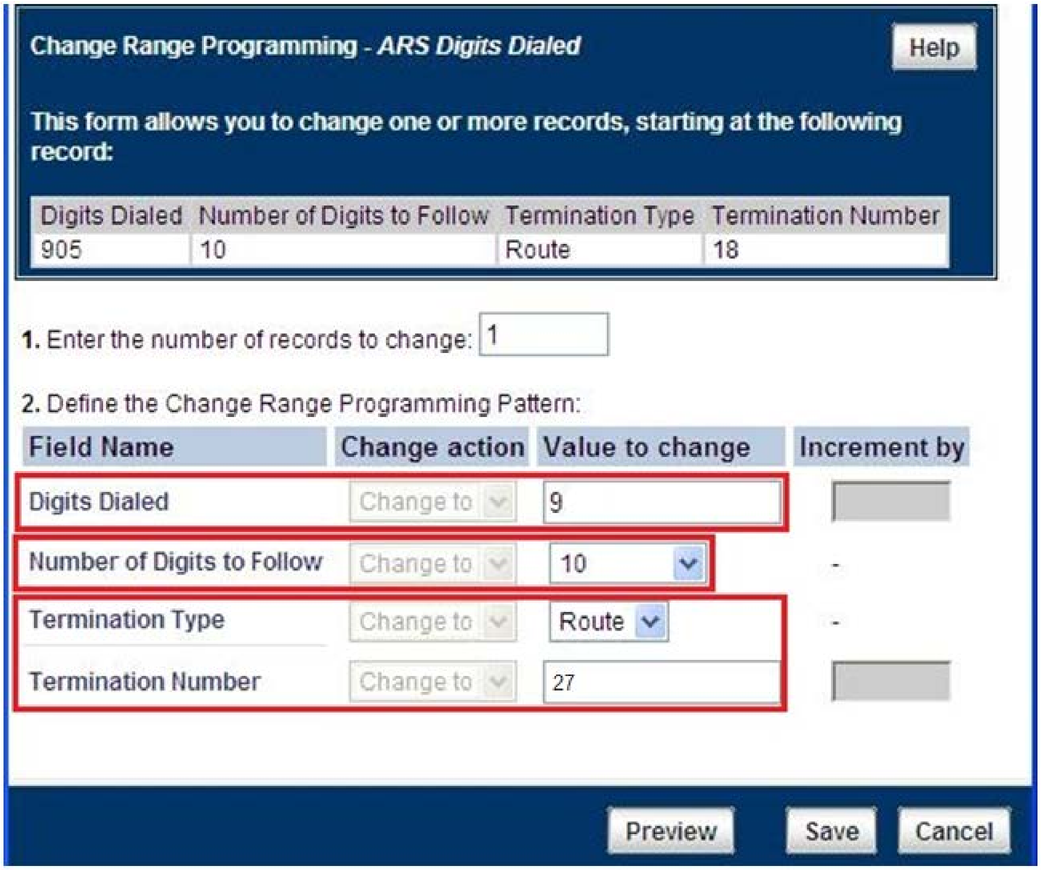

Figure 16 – ARS Digit Dialed form

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).

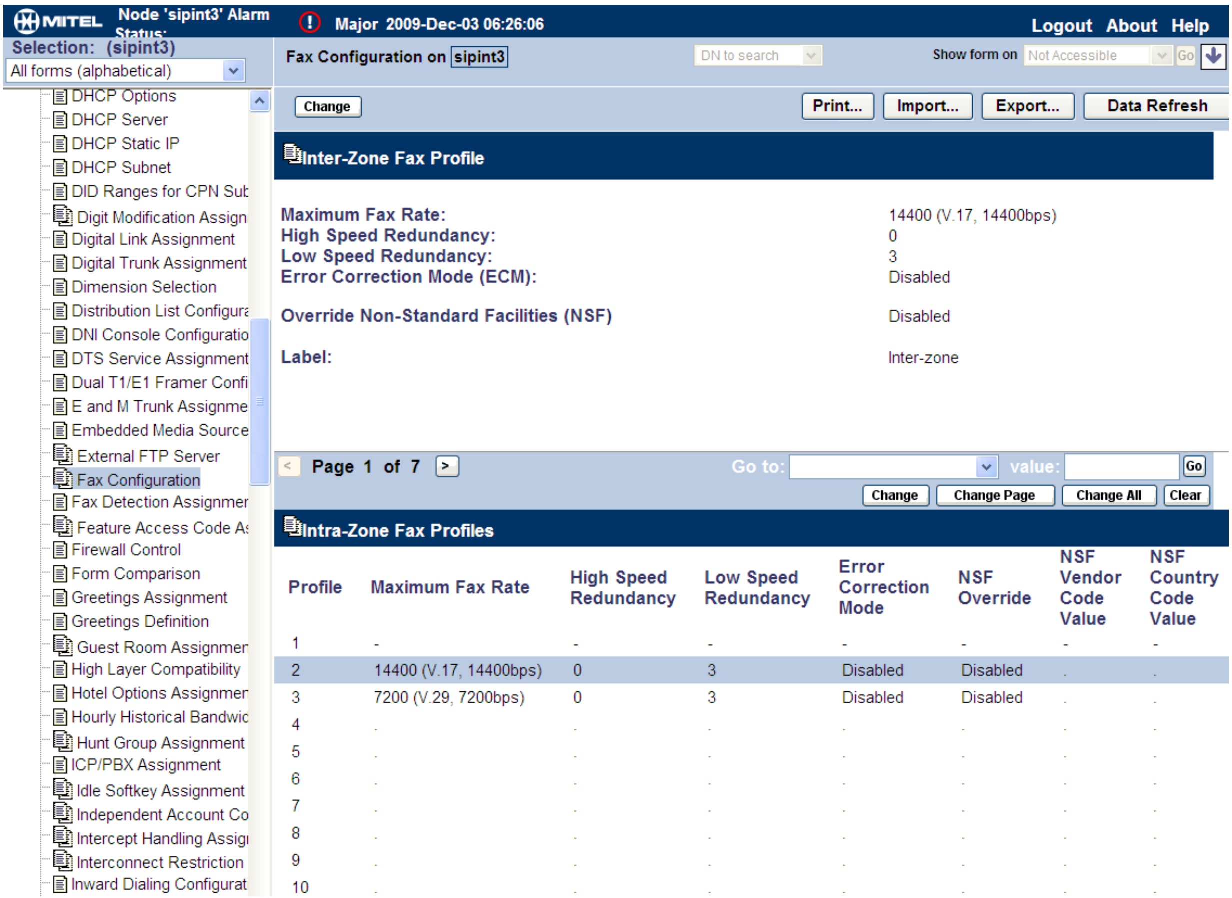

Figure 22 – Fax Configuration

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.

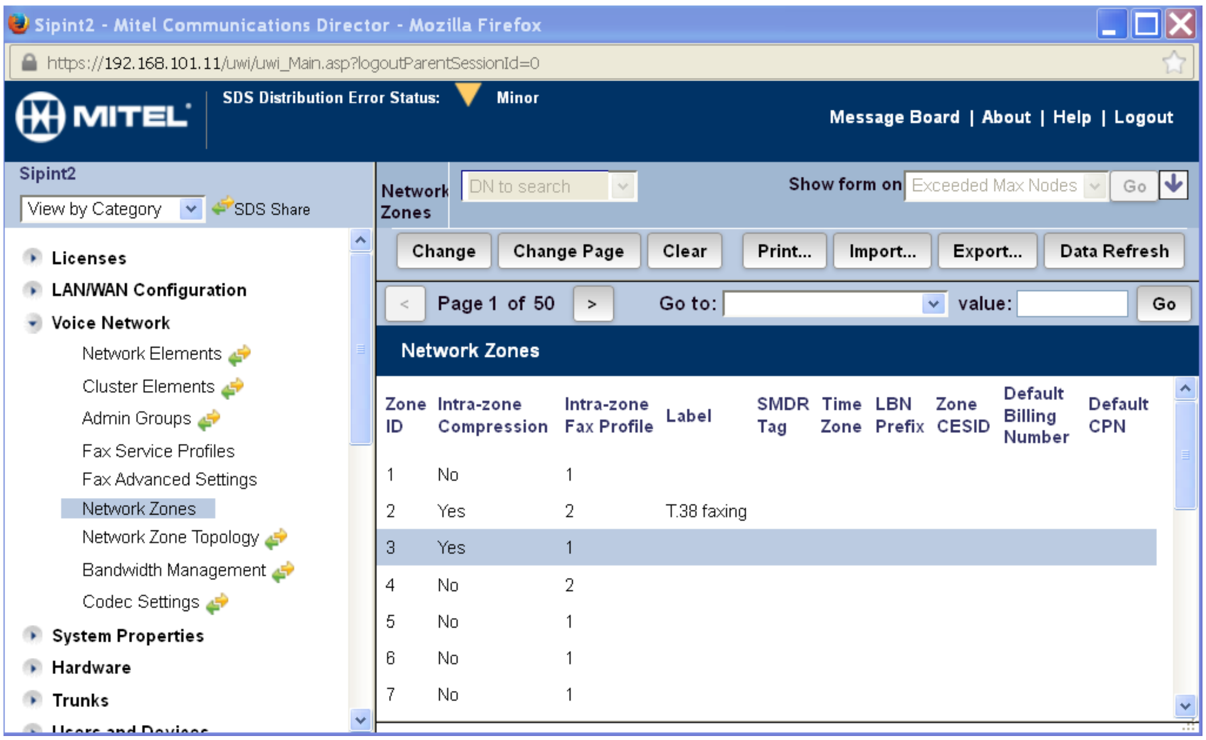

Figure 23 – Zone Assignment

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.

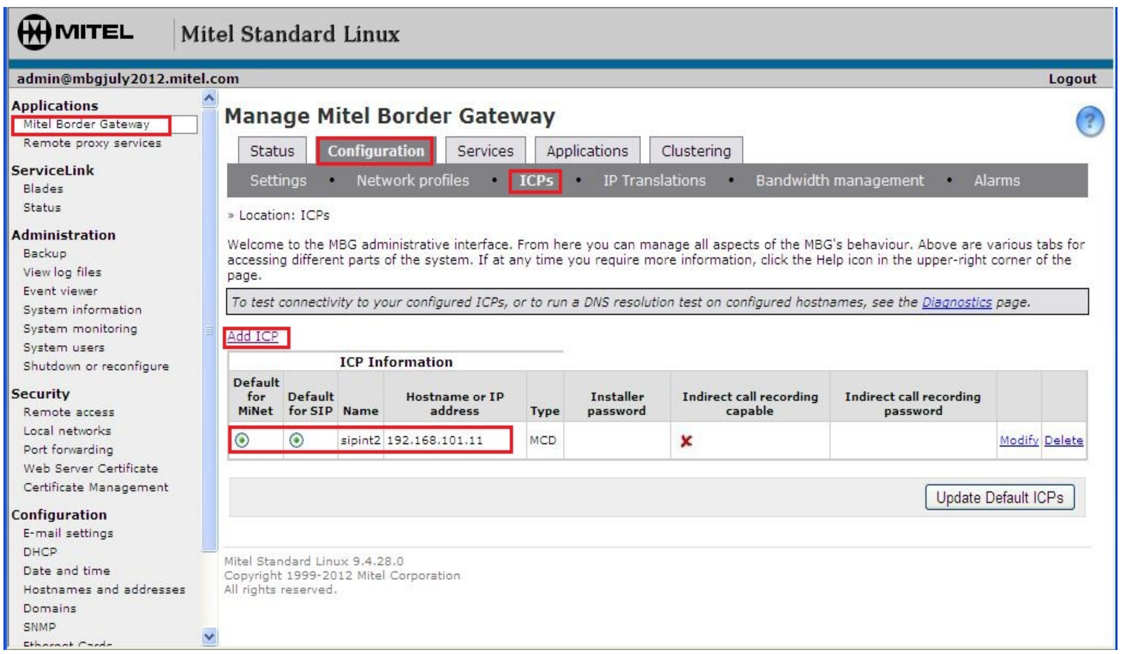

Figure 17 – ICP's Configuration page

To add a new SIP trunk:

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

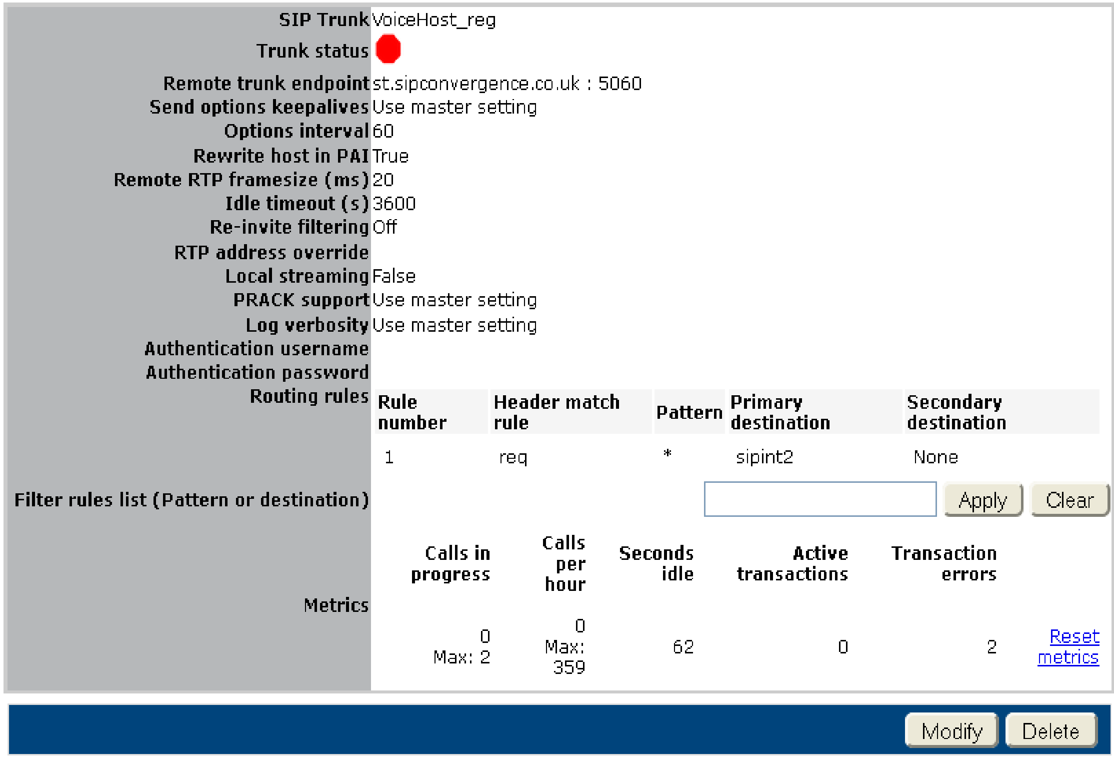

Figure 18 – SIP Trunking Configuration Page

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

Global Headquarters	U.S.	EMEA	CALA	Asia Pacific
Tel: +1(613) 592-2122	Tel: +1(480) 961-9000	Tel: +44(0)1291-430000	Tel: +1(613) 592-2122	Tel: +852 2508 9780
Fax: +1(613) 592-4784	Fax: +1(480) 961-1370	Fax: +44(0)1291-430400	Fax: +1(613) 592-7825	Fax: +852 2508 9232

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