MITEL – SIP CoE

Technical

Configuration Note

Configure MCD for use with Thinktel SIP Trunking Service

SIP CoE 12-4940-00197

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Mitel Technical Configuration Notes – Configure MCD for use with Thinktel SIP Trunking Service

March 2012, 12-4940-00197

Overview

This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel 3300 MCD to connect to Thinktel 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	March 2012	Initial Interop with MCD 5.0 SP1 and Thinktel SIP Trunking

Interop Status

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

COMPATIBLE	The most common certification which means Thinktel SIP trunking 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 Thinktel SIP Trunking and the 3300 MCD.

Manufacturer	Variant	Software Version
Mitel	3300 MCD – Mxe Platform	11.0.1.20
Mitel	MBG – Teleworker	7.1.13.0
Mitel	Nupoint Voicemail	14.1.0.45
ThinkTel	Mediaswitch	As of March 2012

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 through the SIP Service provider and their PSTN gateway or SIP trunking device, 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 3300 MCD to stream RTP packets through its E2T card at different intervals, from 10ms to 90ms	⚠
Personal Ring Groups	Receiving calls through the SIP Service provider and their PSTN gateway or SIP trunking device to a personal ring group. Also moving calls to/from the prime member and group members.	☑
Teleworker	Making and receiving a call through the SIP Service provider and their PSTN gateway or SIP trunking device to and from Teleworker extensions.	☑
Video	Making and receiving a call through the SIP Service provider or SIP trunking device with video capable devices.	Not Supported
Fax	T.38 and G711Fax Calls	☑

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

Device Limitations and Known Issues

Feature	Problem Description
Unsupervised Transfer	When doing an unsupervised transfer to an external PSTN number of call that originated on a Thinktel SIP trunk there is no audible ringback to the callee during the unsupervised transfer but two way audio is established once the far end answers. **Recommendation:** This was a defect found with the Mitel MBG during testing. Please contact Mitel Product support and provide defect ID MN426329 for further updates.
Varibale Packetization	Thinktel will not support a packetization rate outside of the default 20ms unless every element in the path is set to dynamic packetization then their switch will honor the packetization rate specified. **Recommendation:** If customer requires packetization rate other than 20ms, they can contact Thinktel for implementation.
Transfer to external call when using G.729	If a call the originated on a Thinktel sip trunk is then is unsupervised transferred to an external number outbound on a the Thinktel trunk the result is no audio. **Recommendation:** In the network zone that is being used for compression set the Intra-zone compression to **Yes**. This fixes the issue with no audio
Using G.729 with Nupoint Messenger	If a SIP trunk (not only Thinktel) is using G.729 and a call is placed to Nupoint Messenger the call will fail because Nupoint Messenger only supports G.711. Recommendation: It is planned to add G.729 Nupoint Messenger in a future release. Please contact Mitel Support or MOL for futher updates.
Loopback calling	When making an outgoing SIP trunk call from the MCD to another Thinktel DID terminated at a busy extension on the same MCD, there is a delay of 30 seconds in receiving the busy signal from Thinktel. **Recommendation:** Please contact Thinktel for futher information regading this issue.

Network Topology

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

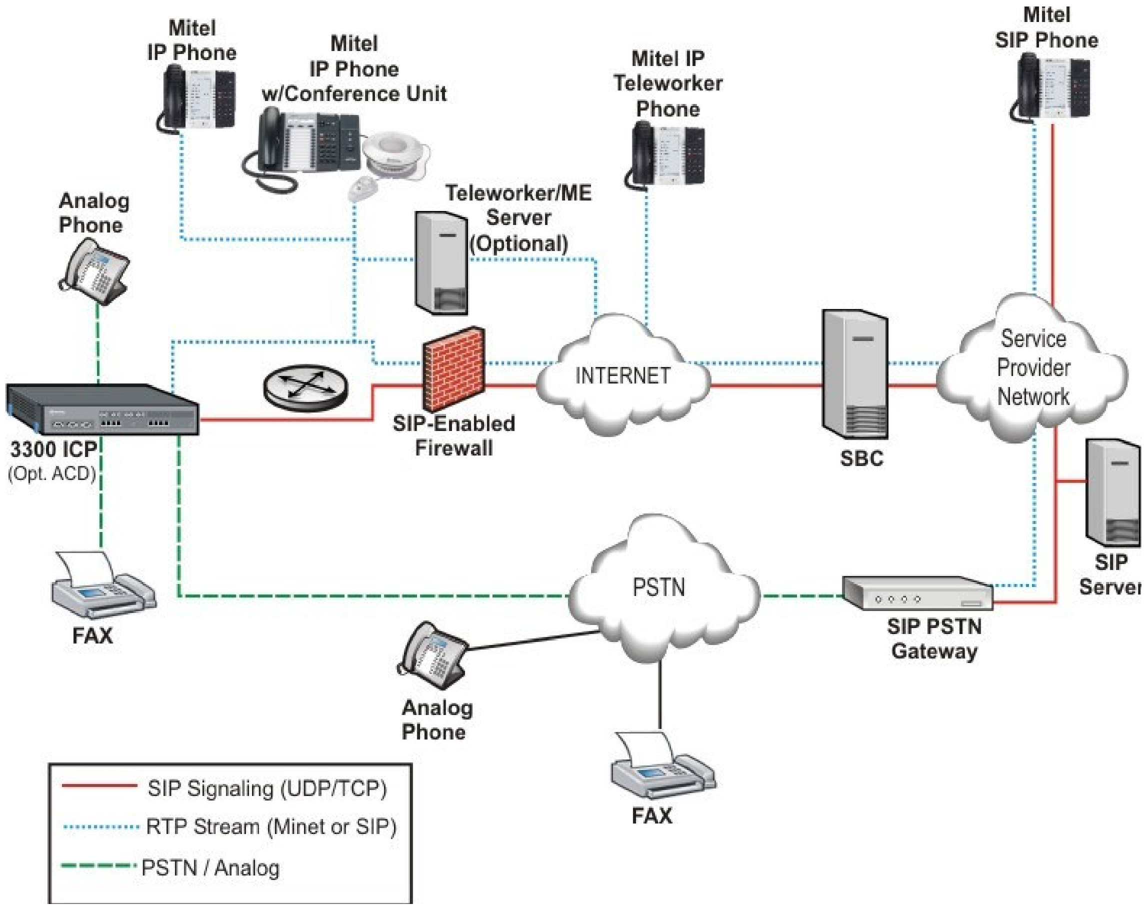


Figure 1 – Network Topology

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 Thinktel SIP Trunking 3300 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.

MCD Configuration Notes

The following steps show how to program the MCD to interconnect with Thinktel 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 MCD Programming

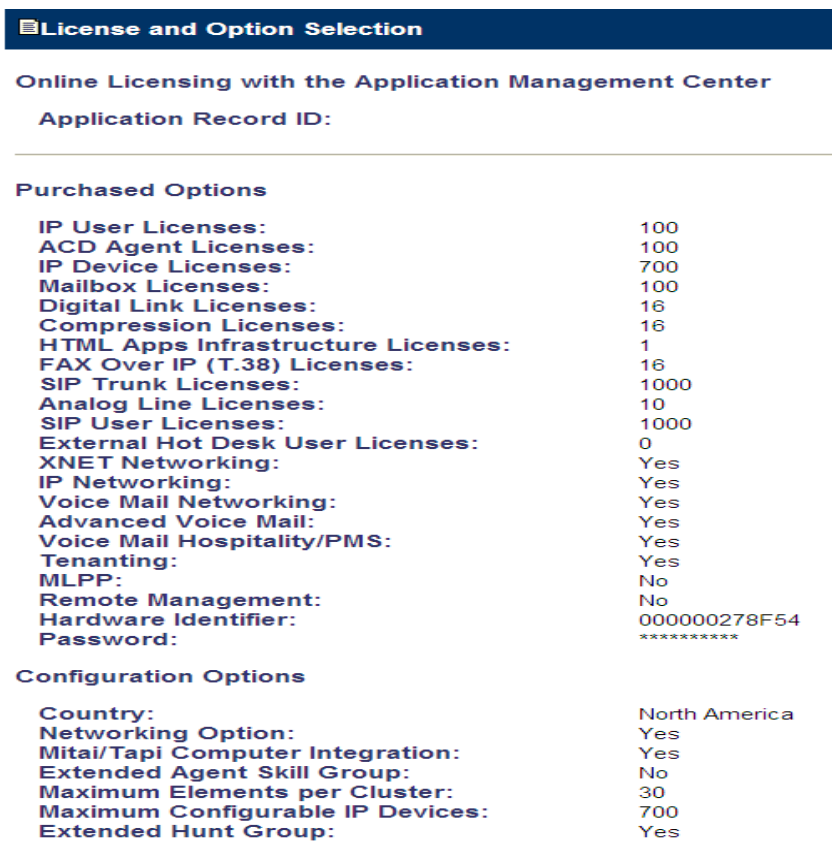
- The SIP signaling connection uses UDP on Port 5060.

Licensing and Option Selection – SIP Licensing

Ensure that the MCD is equipped with enough SIP trunking licenses for the connection to Thinktel 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 MCD to be used with all service providers, applications and SIP trunking devices.

Thinktel SIP Trunking can use T.38 over SIP trunk to communicate with the MCD. To do so, hardware and software should be ready. DSP II Card needs to be installed for dealing with T.38. Assign the required number of “Fax over IP (T.38) Licenses” and the required number of “Compression” licenses. Based on DSP engineering, 16 is the number of “Fax over IP (T.38) Licenses”. If the number of T.38 licenses programmed exceeds the available DSP resources, a DSP alarm is raised and a maintenance log is generated. Reboot the system to enable the “Fax over IP (T.38) Licenses” and “Compression” licenses.



License and Option Selection

Online Licensing with the Application Management Center

Application Record ID:

Purchased Options

IP User Licenses:	100
ACD Agent Licenses:	100
IP Device Licenses:	700
Mailbox Licenses:	100
Digital Link Licenses:	16
Compression Licenses:	16
HTML Apps Infrastructure Licenses:	1
FAX Over IP (T.38) Licenses:	16
SIP Trunk Licenses:	1000
Analog Line Licenses:	10
SIP User Licenses:	1000
External Hot Desk User Licenses:	0
XNET Networking:	Yes
IP Networking:	Yes
Voice Mail Networking:	Yes
Advanced Voice Mail:	Yes
Voice Mail Hospitality/PMS:	Yes
Tenanting:	Yes
MLPP:	No
Remote Management:	No
Hardware Identifier:	000000278F54
Password:	**********

Configuration Options

Country:	North America
Networking Option:	Yes
Mitai/Tapi Computer Integration:	Yes
Extended Agent Skill Group:	No
Maximum Elements per Cluster:	30
Maximum Configurable IP Devices:	700
Extended Hunt Group:	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 3300.

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

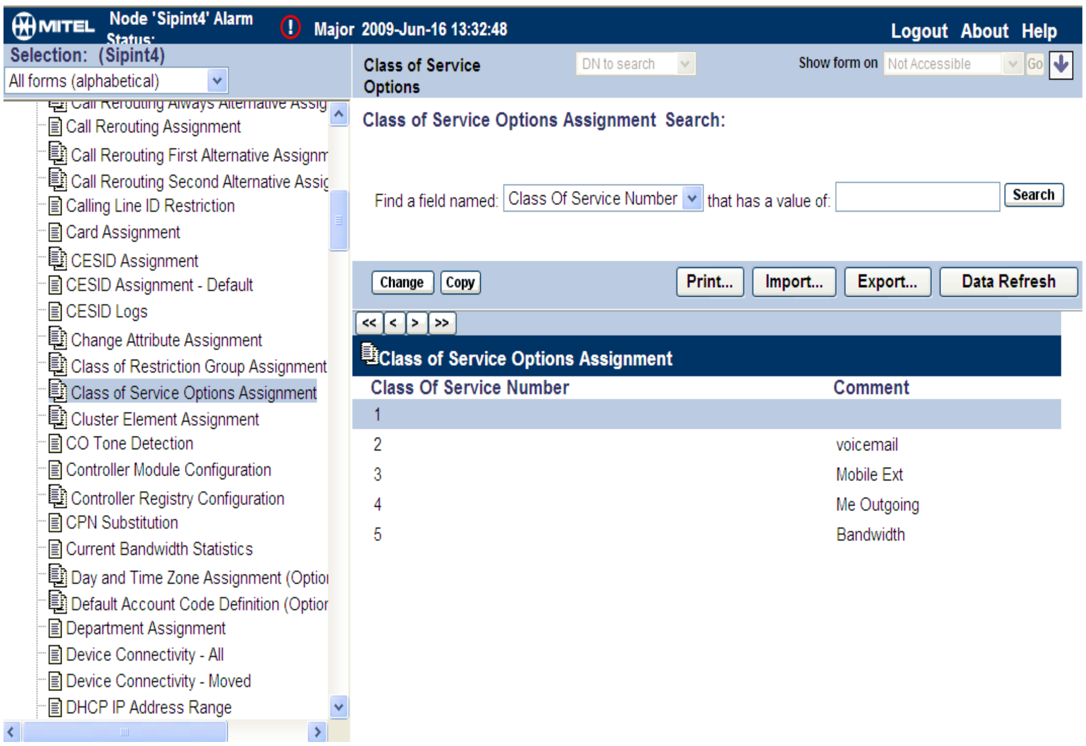


Figure 3 – Class of Service

Network Element Assignment

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

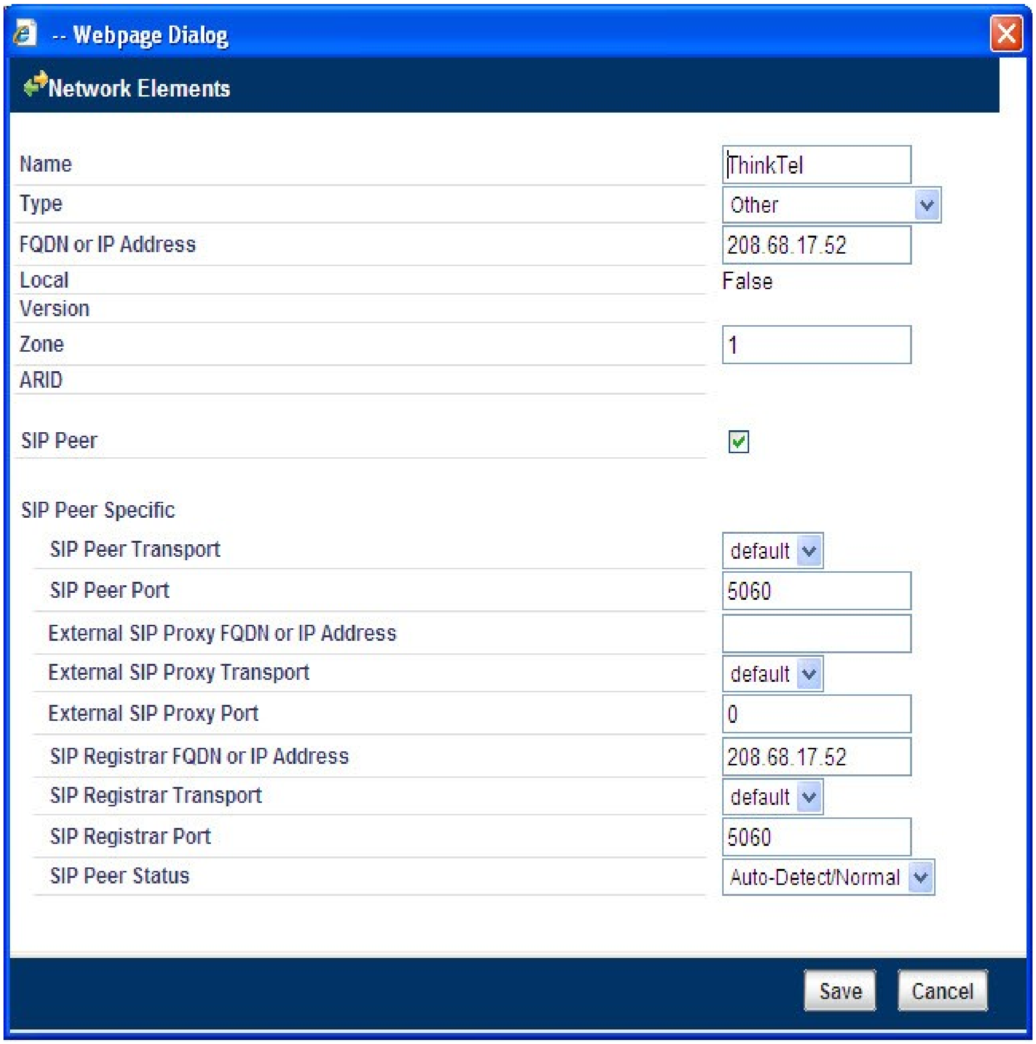


Figure 4 – Network Element Assignment

Network Element Assignment (Proxy)

In addition, depending in 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 3300 MCD 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).

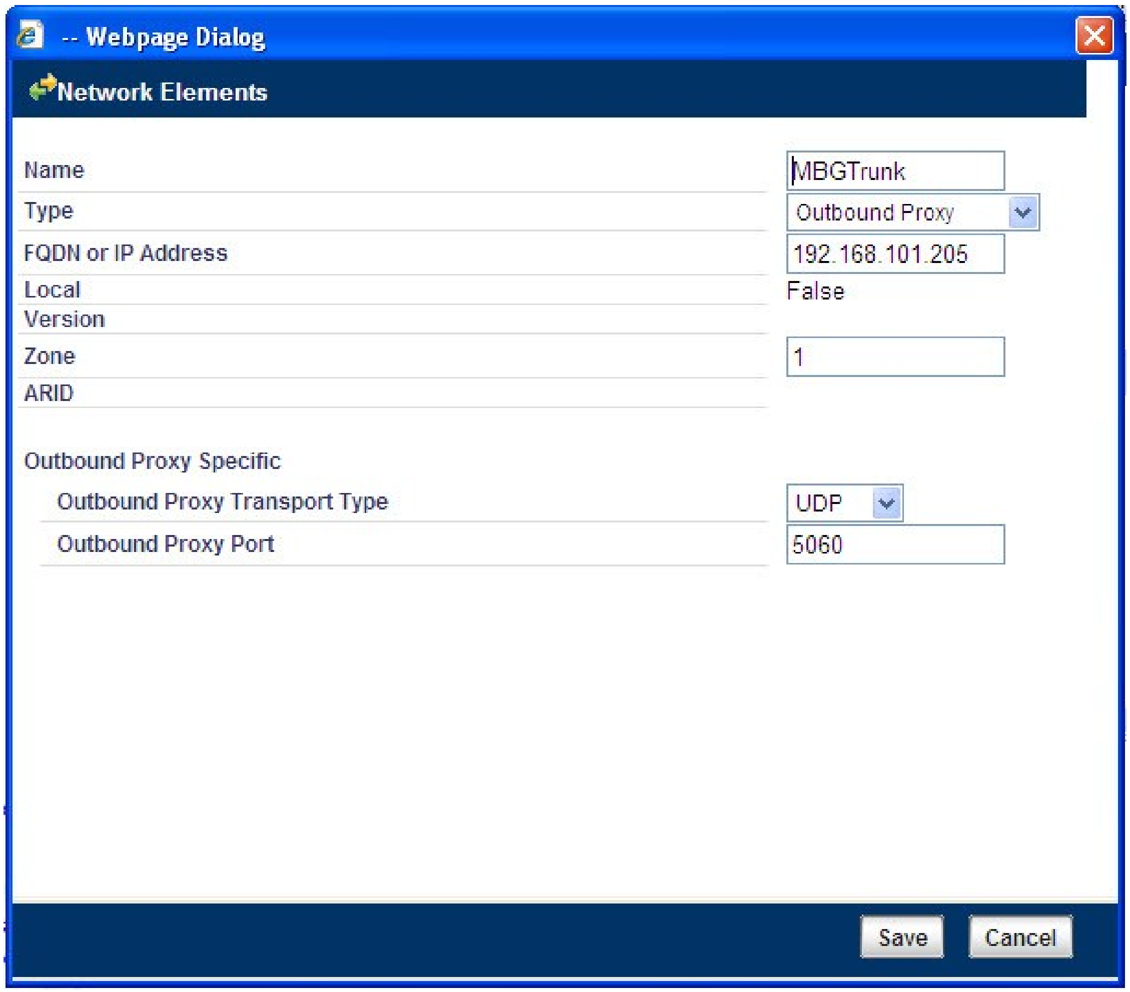


Figure 5 – Network Element Assignment (Proxy)

Trunk Service Assignment

This is configured in the Trunk Service Assignment form. In this example the Trunk Service Assignment is defined for Trunk Service Number 7 which will be used to direct incoming calls to an answer point in the 3300.

Program the Non-dial In or Dial In Trunks (DID) according to the site requirements and what type of service was ordered from your service provider.

The example below shows configuration for incoming DID calls. The 3300 will absorb the first 5 digits of the DID number from Thinktel SIP Trunking leaving 5 digits for the 3300 to translate and ring the remaining 5 digit extension. For example, Thinktel SIP Trunking delivers 732-321-4009 through the SIP trunk to the 3300. The 3300 will absorb the first 5 digits (732321) leaving the 3300 to ring extension 14009. Extension 14009 must be programmed as a valid dialable number in the 3300. Please refer to the 3300 System Administration documentation for further programming information.

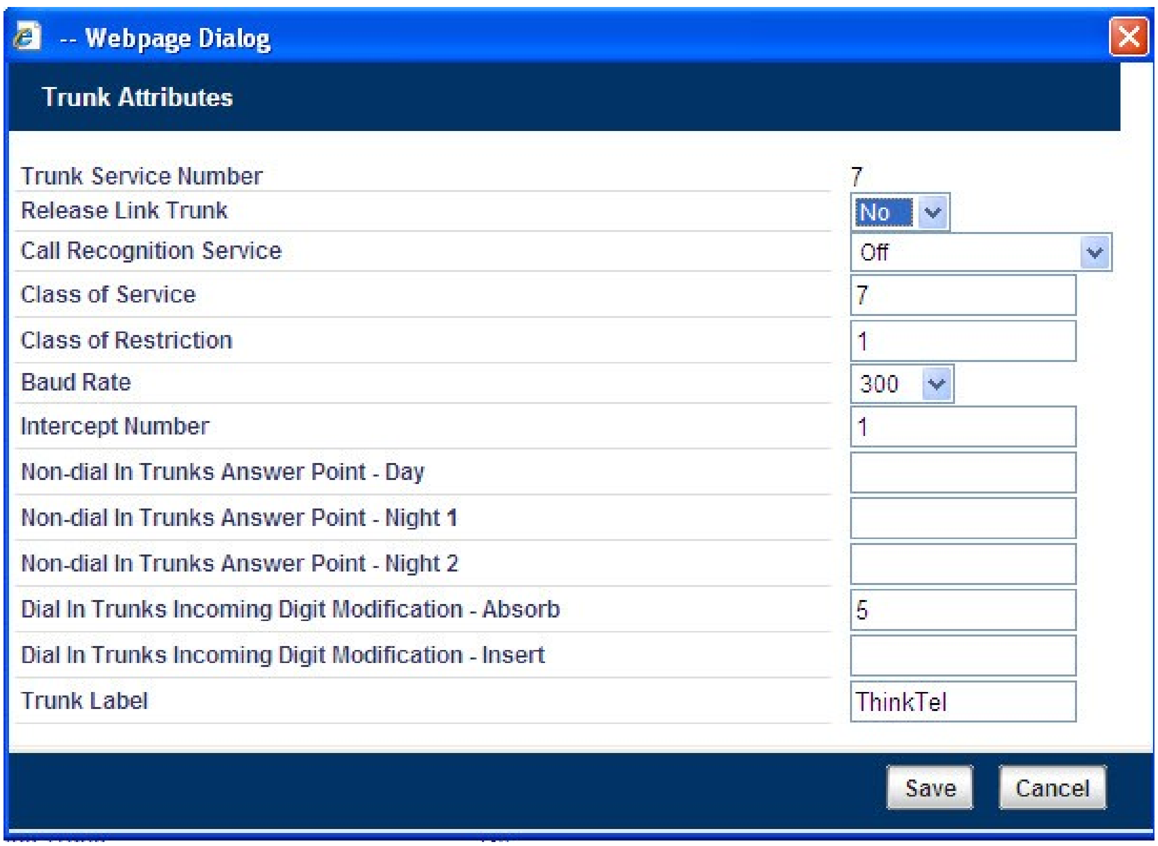


Figure 6 – Trunk Attributes

SIP Peer Profile

The recommended connectivity via SIP Trunking does not require additional physical interfaces. IP/Ethernet connectivity is part of the base 3300 MCD 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 “ThinkTel” Network Element.

Registration User Name: ThinkTel uses registration. The 3300 does not support Bulk Registration, therefore trunks will have to be registered individually. Enter the DIDs assigned by Thinktel SIP Trunking. Enter one or more numbers. The field has a maximum of 60 characters. The maximum number of digits per number is 26. You can enter a mix of ranges and single numbers (for example, "6135554000-6135554400, 6135554500"). 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.

Address Type Use 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** ThinkTel SIP Trunking. 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 ThinkTel SIP Trunking.

NOTE: Ensure the remaining SIP Peer profile policy options are similar the screens capture below.

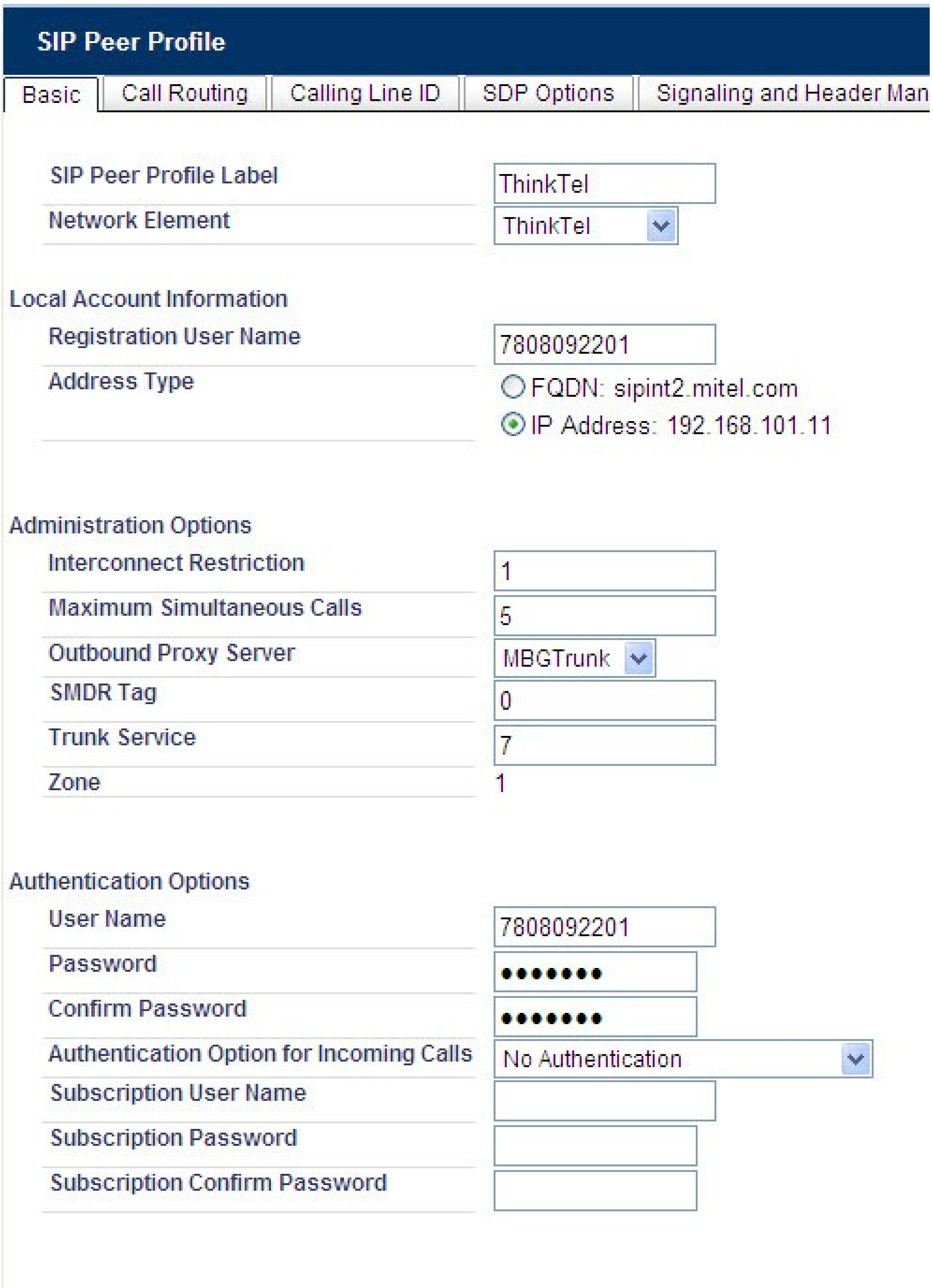


Figure 7 – SIP Peer Profile Assignment- Basic

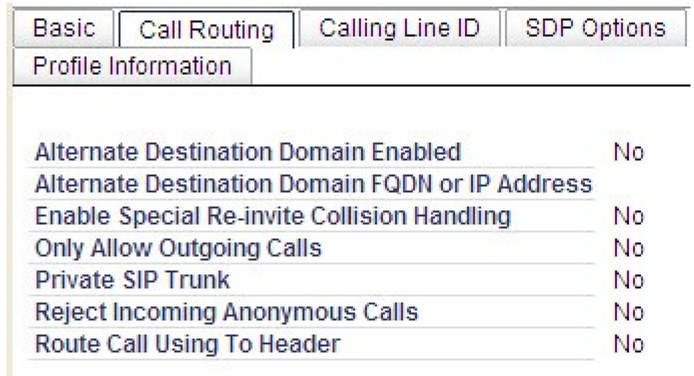


Figure 8 – SIP Peer Profile Assignment- Call Routing

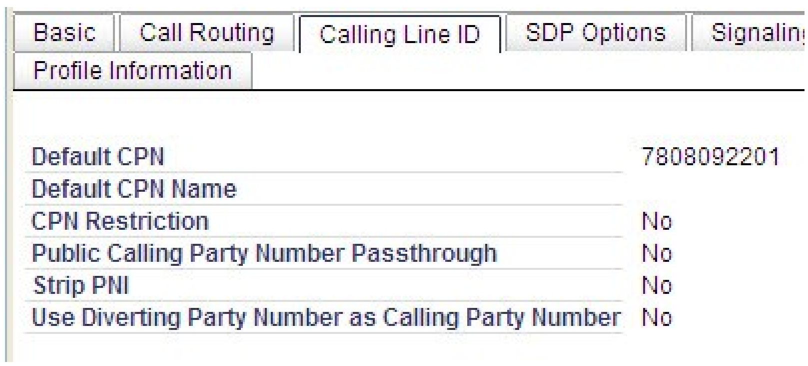


Figure 9 – SIP Peer Profile Assignment- Calling Line ID

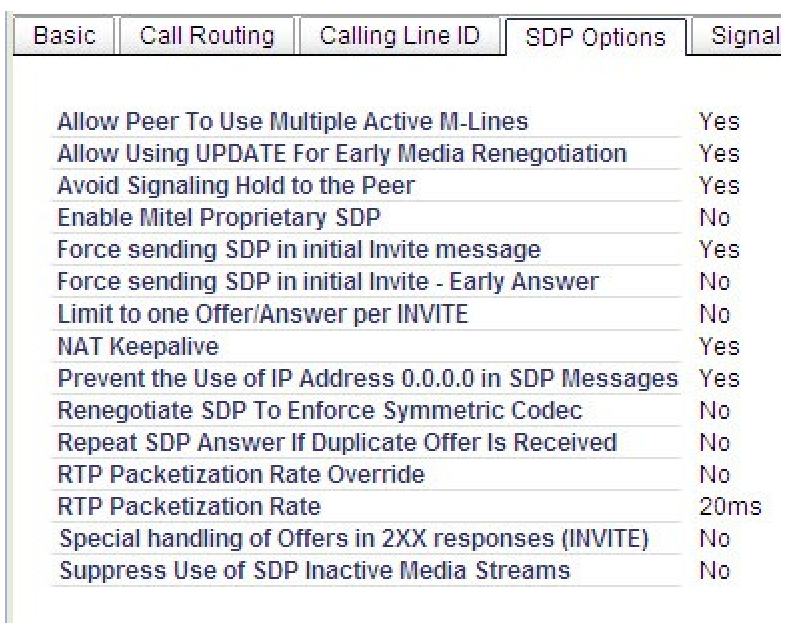


Figure 10 – SIP Peer Profile Assignment- SDP Options

Basic | Call Routing | Calling Line ID | SDP Options | Signaling and Header Manipulation | 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	No
Disable Use of User-Agent and Server Headers	No
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
Ignore Incoming Loose Routing Indication	No
Only use SDP to decide 180 or 183	No
Require Reliable Provisional Responses on Outgoing Calls	No
Use Privacy: none	No
Use P-Asserted Identity Header	Yes
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

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

Basic | Call Routing | Calling Line ID | SDP Options | Signaling and Header Manipulation | Timers

Keep-Alive (OPTIONS) Period	120
Registration Period	3600
Registration Period Refresh (%)	50
Registration Maximum Timeout	90
Session Timer	90
Subscription Period	3600
Subscription Period Minimum	300
Subscription Period Refresh (%)	80
Invite Ringing Response Timer	0

Figure 12 – SIP Peer Profile Assignment- Timers

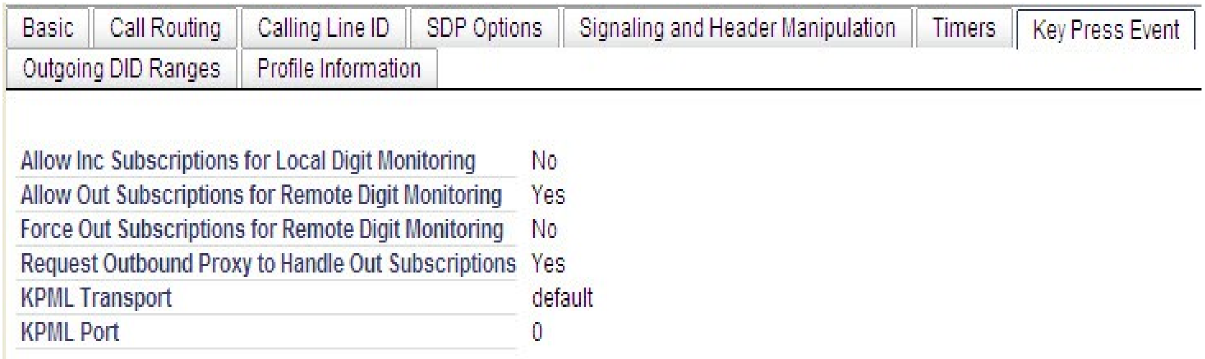


Figure 13 – SIP Peer Profile Assignment- Key Press Event

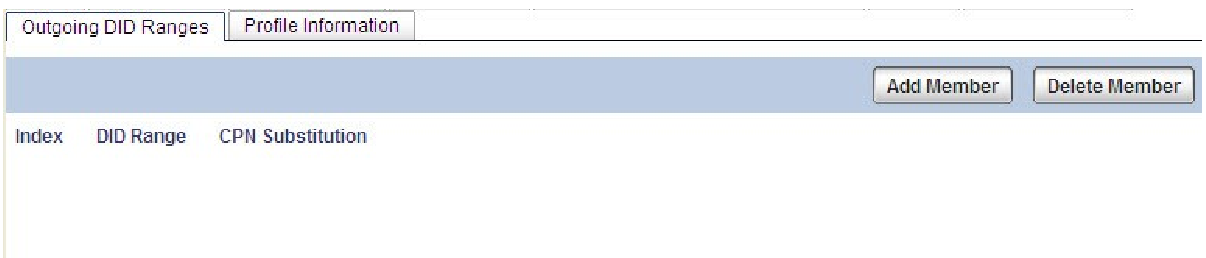


Figure 14 – SIP Peer Profile Assignment- Outgoing DID Ranges

Figure 15 – SIP Peer Profile Assignment- Profile Information

SIP Peer Profile Assignment by Incoming DID

This form is used to assign incoming digits from ThinkTel DID range numbers assigned by ThinkTel and are associated to a particular SIP Peer.

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, "6135554000-6135554400, 6135554500"). 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 character 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.

Figure 16 – SIP Peer Profile Assignment by Incoming DID

Digit Modification Number

Ensure that Digit Modification for outgoing calls on the SIP trunk to ThinkTel SIP Trunking absorbs or inject additional digits according to your dialling plan. In this example, we will be absorbing 3 digits (in this case will be 901 to dial out).

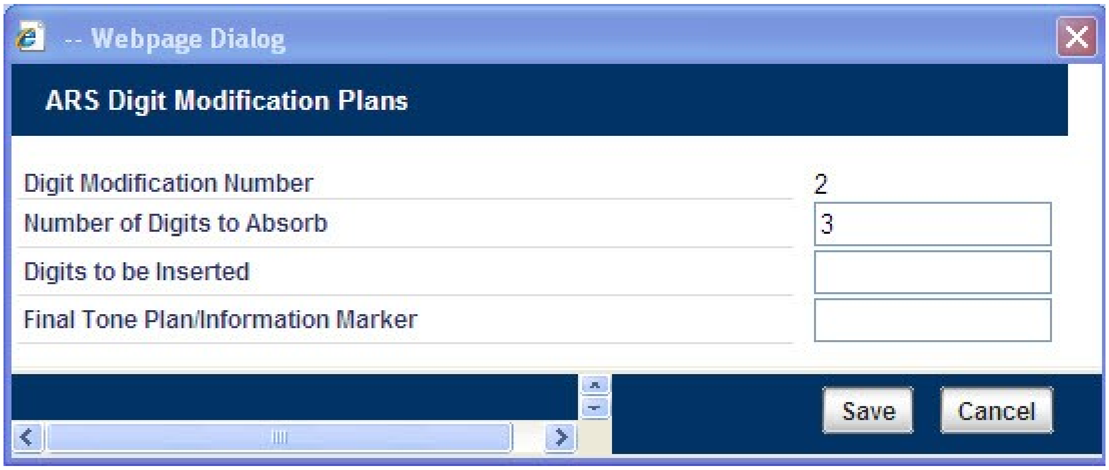


Figure 17 – Digit Modification Assignment

Route Assignment

Create a route for SIP Trunks connecting a trunk to Thinktel SIP Trunking. In this example, the SIP trunk is assigned to Route Number 7. Choose SIP Trunk as a routing medium and choose the SIP Peer Profile and Digit Modification entry created earlier.

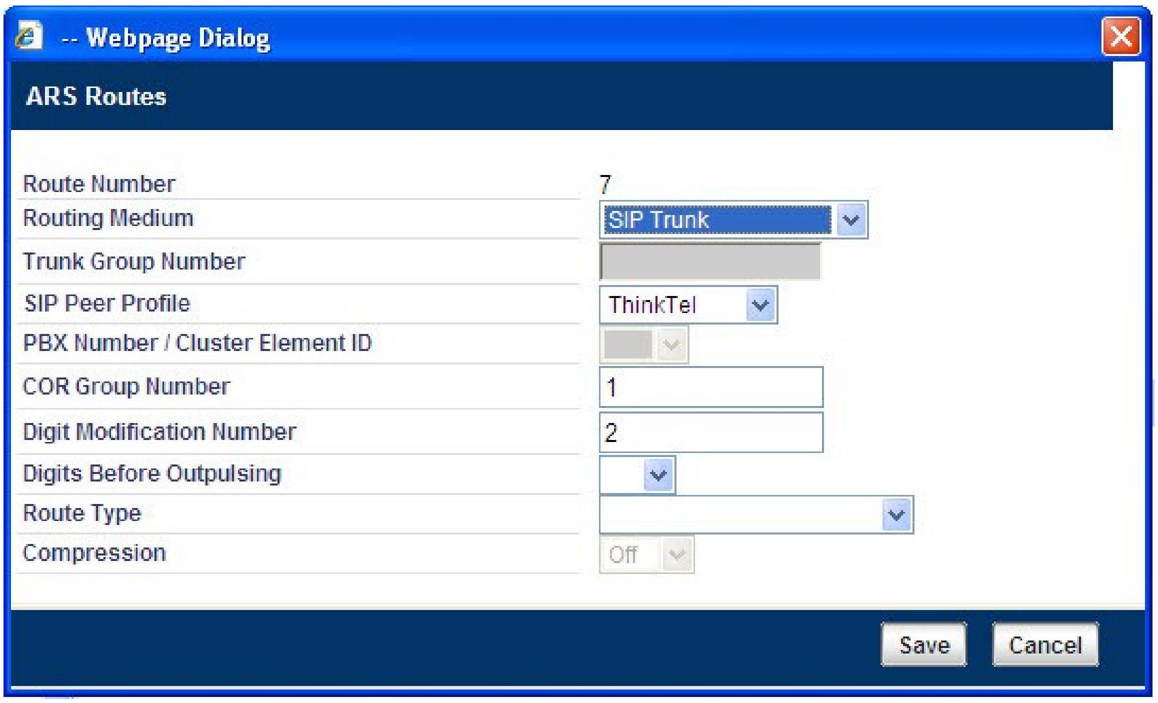


Figure 18 - Trunk ARS Route Assignment

ARS Digits Dialed Assignment

ARS initiates the routing of trunk calls when certain digits are dialed from a station. In this example, when a user dials 901, the call will be routed to Thinktel SIP Trunking (ie. Route 7).

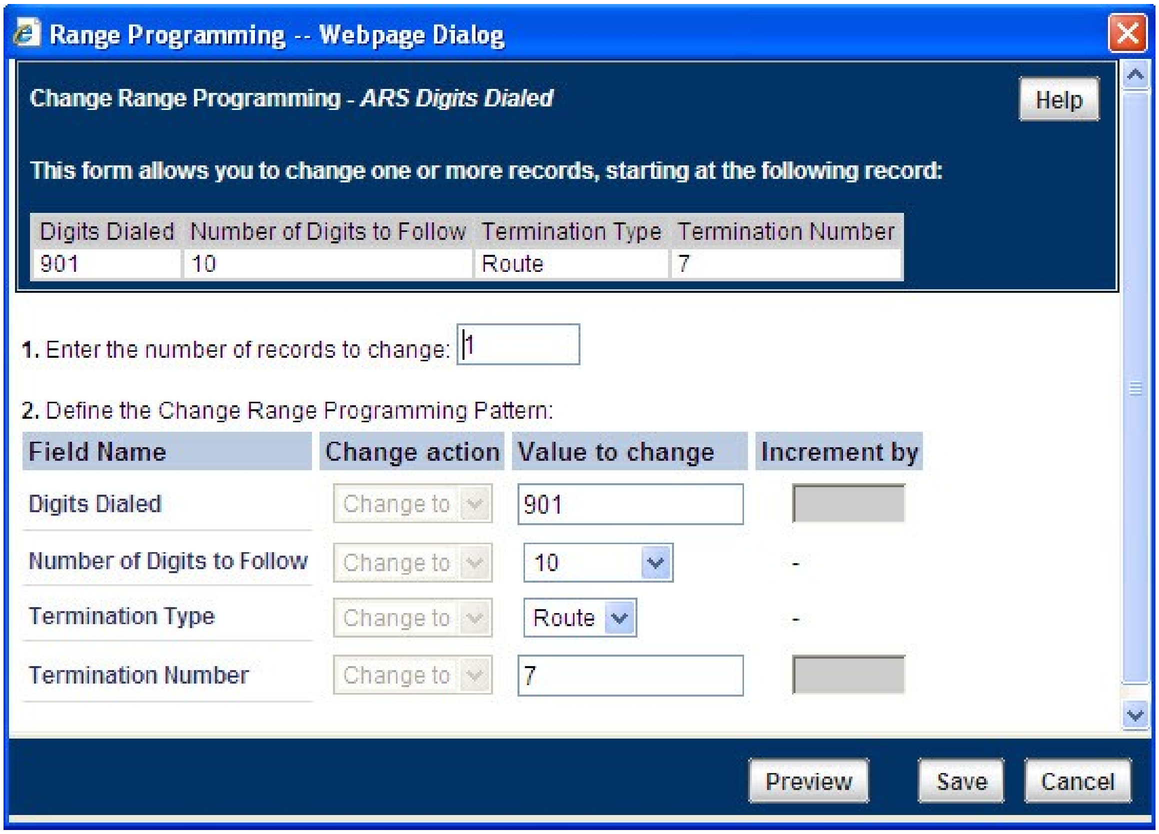


Figure 19 - ARS Digit Dialed Assignment

T.38 Fax Configuration

Thinktel SIP Trunking 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).

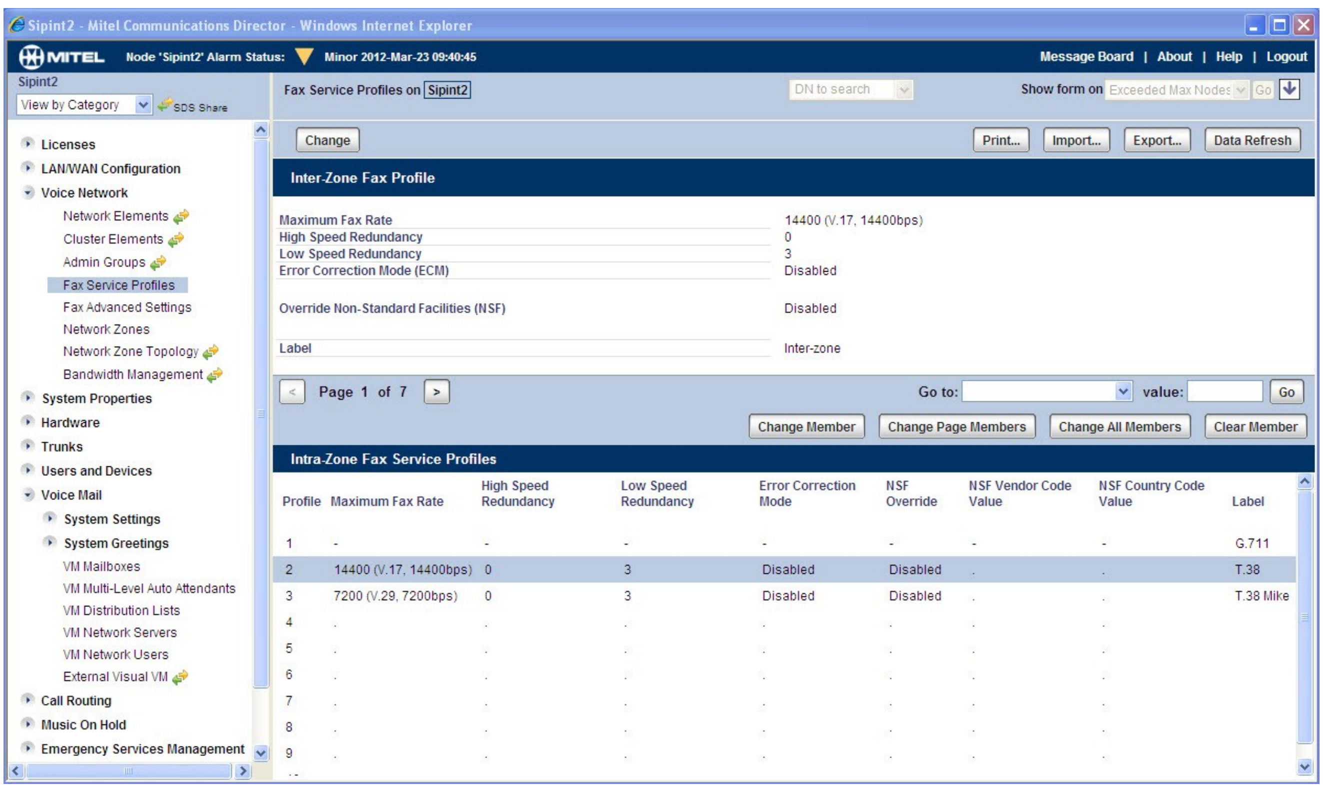


Figure 20 - Fax Configuration

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”. ThinkTel SIP Trunking uses the Inter-zone FAX Profile 3

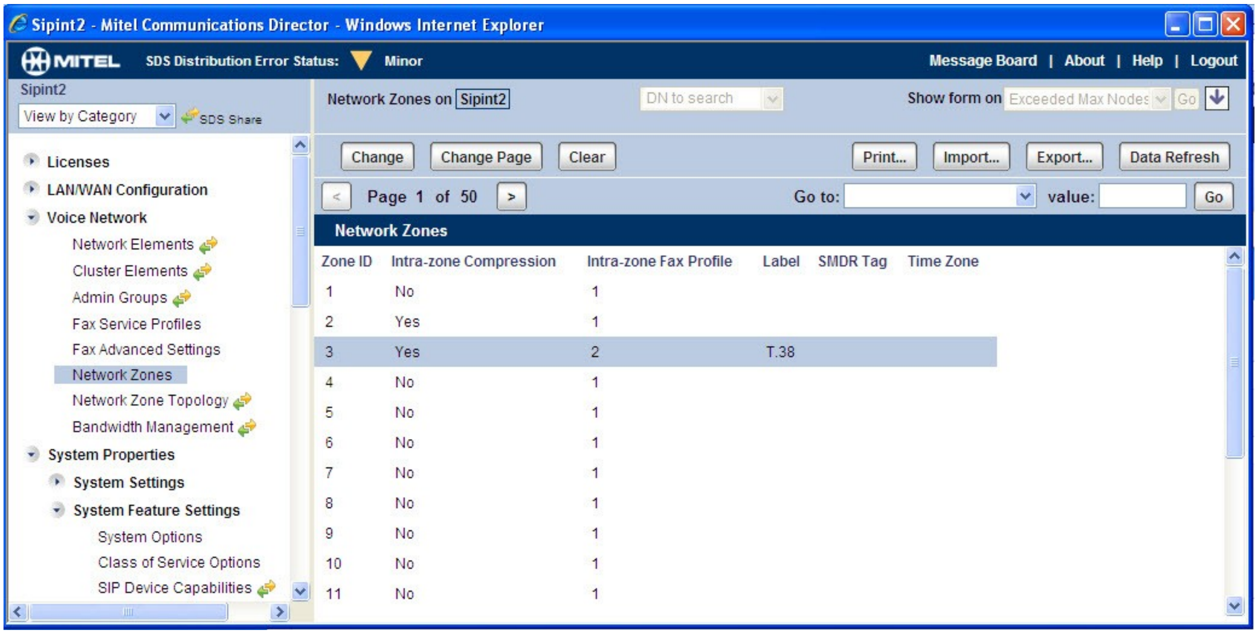


Figure 21 – Zone Assignment

Mitel Border Gateway Configuration Notes (Optional)

When configuring Mitel Border Gateway (MBG), you need to identify the working 3300 ICP where to forward SIP messages to and then to configure the SIP trunk.

When using ThinkTel SIP Trunking the MBG will do the call authentication, you will need to enter the authentication information as described further below.

To do this:

- Login to MBG and click **Mitel Border Gateway**
- In right pane, click **Configuration** tab and then **ICPs** (see Figure 22 for details)

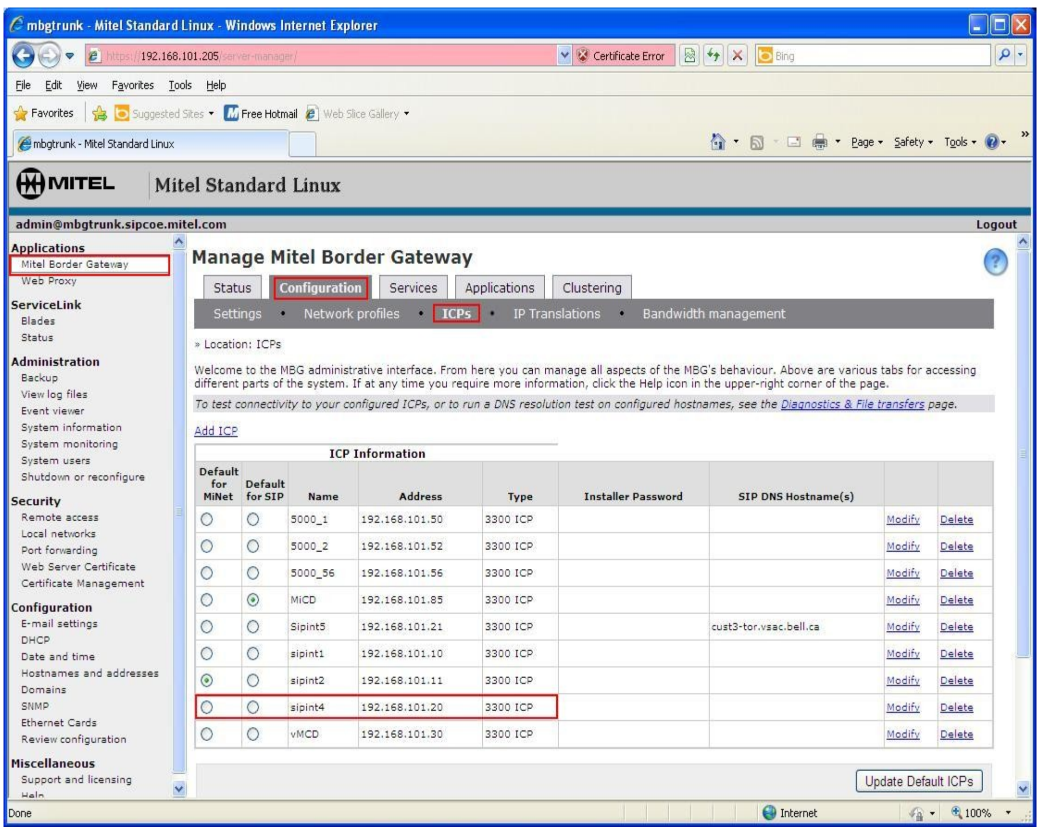


Figure 22 – MBG's Configuration page

- On **ICPs** page, ensure that the "working" 3300ICP is configured. If needed, click **Add ICP** link and add a new Mitel switch.
- Click **Update** button

To add a new SIP trunk:

- Click **Services** tab and then click **SIP trunking**

- Click **Add a SIP trunk** link (see Figure 23)

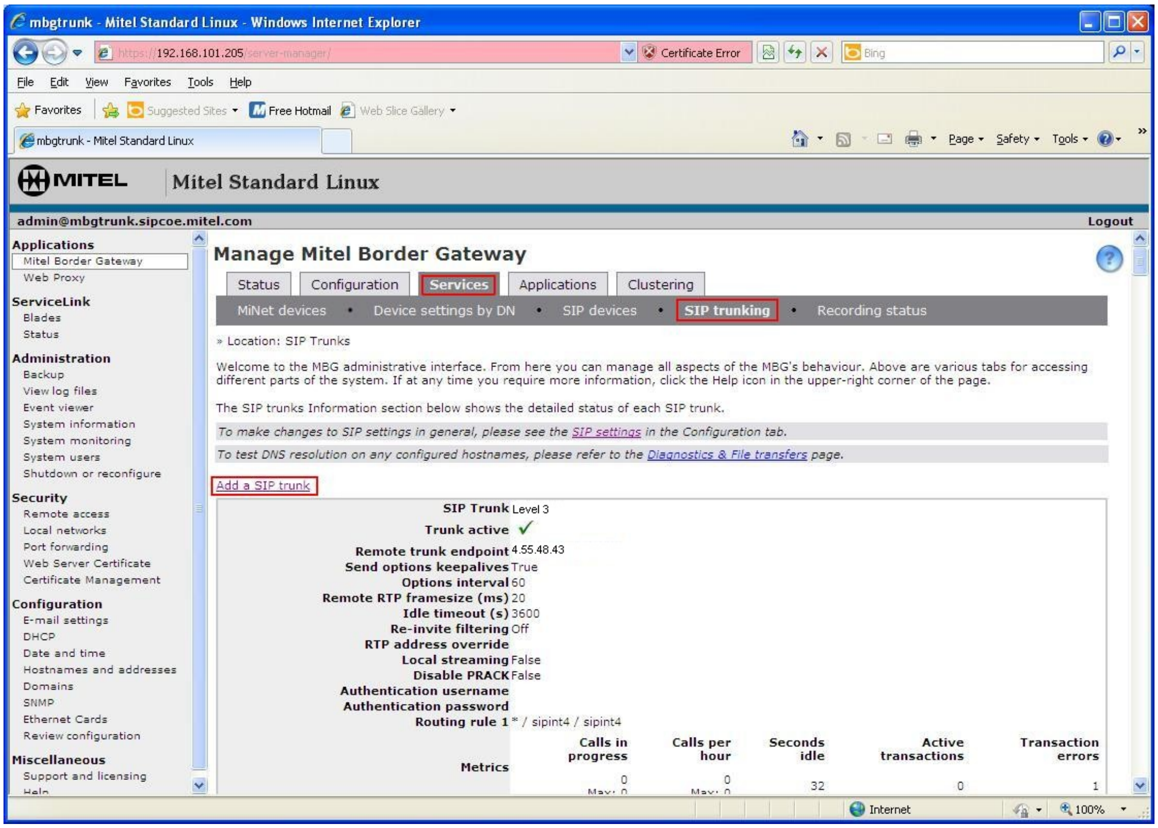


Figure 23 – SIP trunking configuration page

Enter the SIP trunk's details as shown in Figure 24:

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. Thinktel).

Local/Remote RTP framesize (ms) – is the packetization rate you want to set on this trunk

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

Authentication Username – Enter the username provided by ThinkTel

Password – Enter the password provided by ThinkTel

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

Click **Save** button

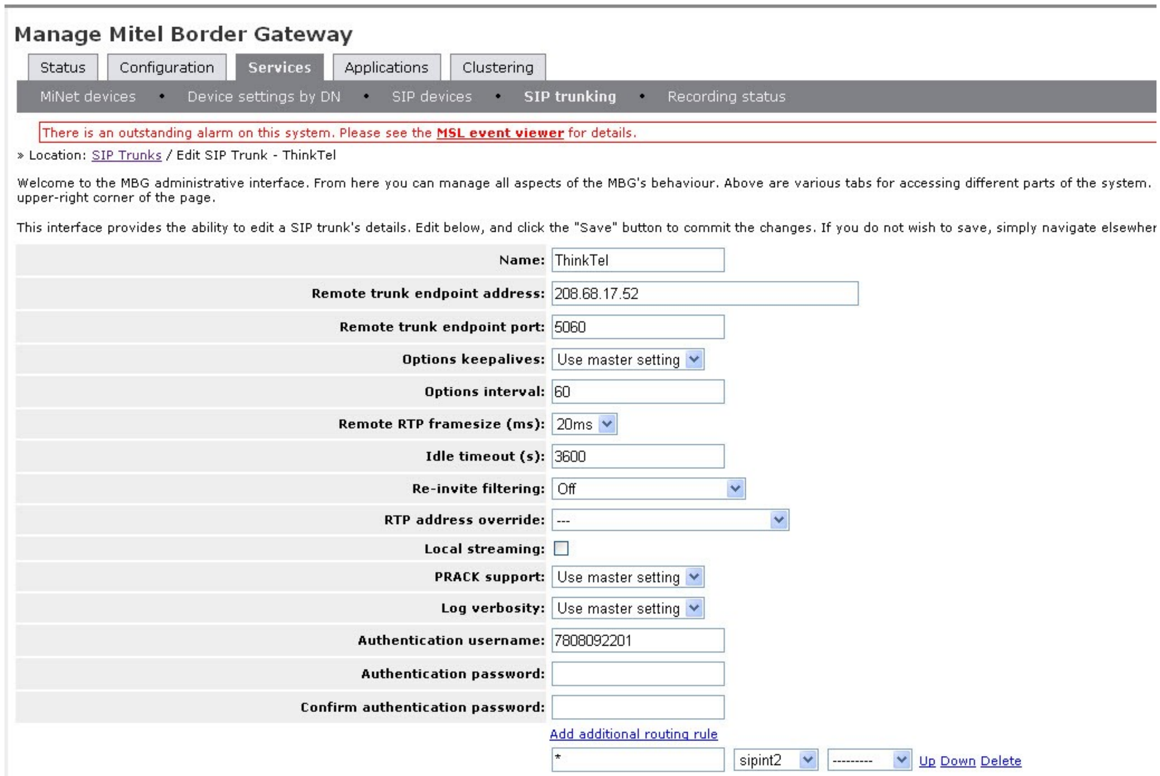


Figure 24 – SIP Trunk configuration settings

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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