

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

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## 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:

	<p>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.</p>
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## 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 Plans (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	<p>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.</p> <p><b>Recommendation:</b> This was a defect found with the Mitel MBG during testing. Please contact Mitel Product support and provide defect ID MN426329 for further updates.</p>
Varibale Packetization	<p>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.</p> <p><b>Recommendation:</b> If customer requires packetization rate other than 20ms, they can contact Thinktel for implementation.</p>
Transfer to external call when using G.729	<p>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.</p> <p><b>Recommendation:</b> In the network zone that is being used for compression set the Intra-zone compression to <b>Yes</b>. This fixes the issue with no audio</p>
Using G.729 with Nupoint Messenger	<p>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.</p> <p><b>Recommendation:</b> It is planned to add G.729 Nupoint Messenger in a future release. Please contact Mitel Support or MOL for futher updates.</p>
Loopback calling	<p>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.</p> <p><b>Recommendation:</b> Please contact Thinktel for futher information regading this issue.</p>

## Network Topology

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

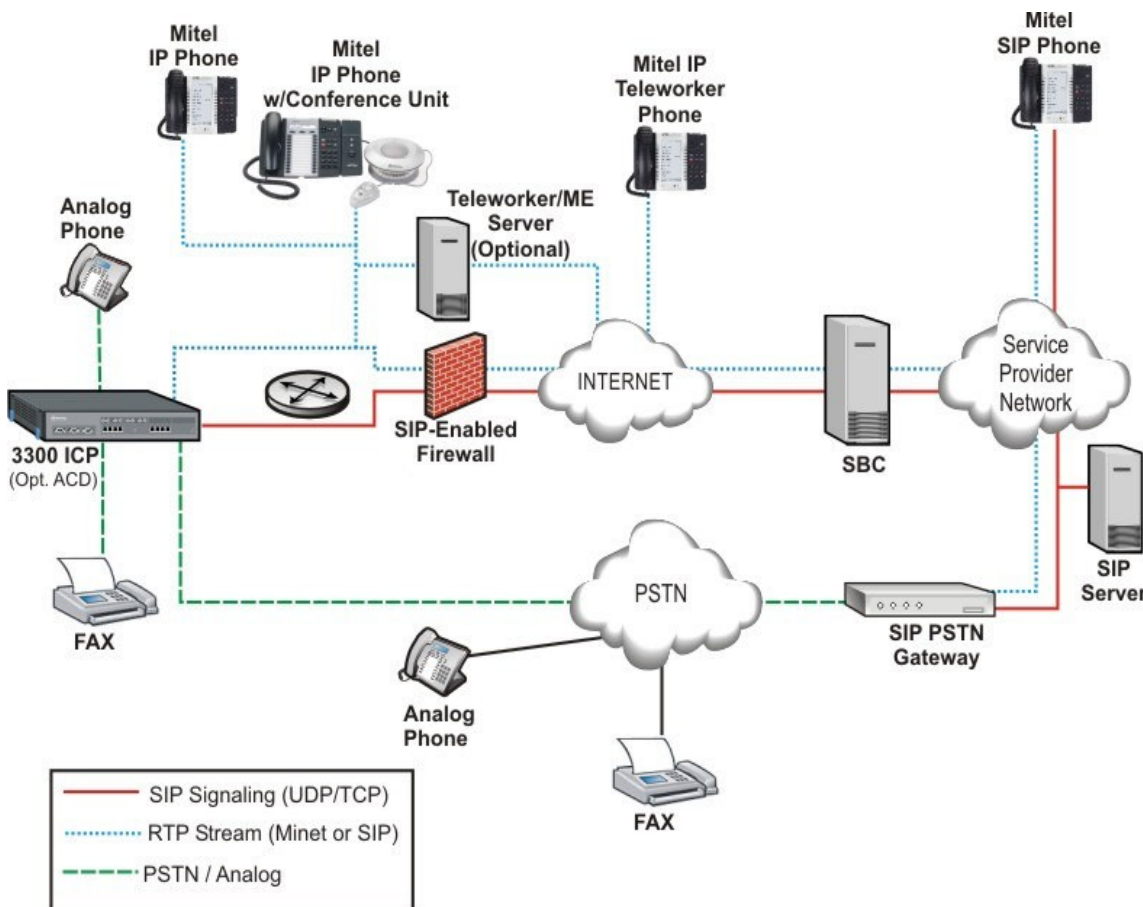


Figure 1 – Network Topology

## 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.

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**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.**

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## 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**

The screenshot shows the Mitel Node 'Sipint4' Alarm interface. The top bar displays the status as 'Major' on 2009-Jun-16 at 13:32:48. The left navigation menu lists various configuration options, with 'Class of Service Options Assignment' selected. The main content area shows the 'Class of Service Options Assignment' form, which includes a search bar and a table of Class of Service Options.

Class Of Service Number	Comment
1	
2	voicemail
3	Mobile Ext
4	Me Outgoing
5	Bandwidth

Figure 3 – Class of Service

## Network Element Assignment

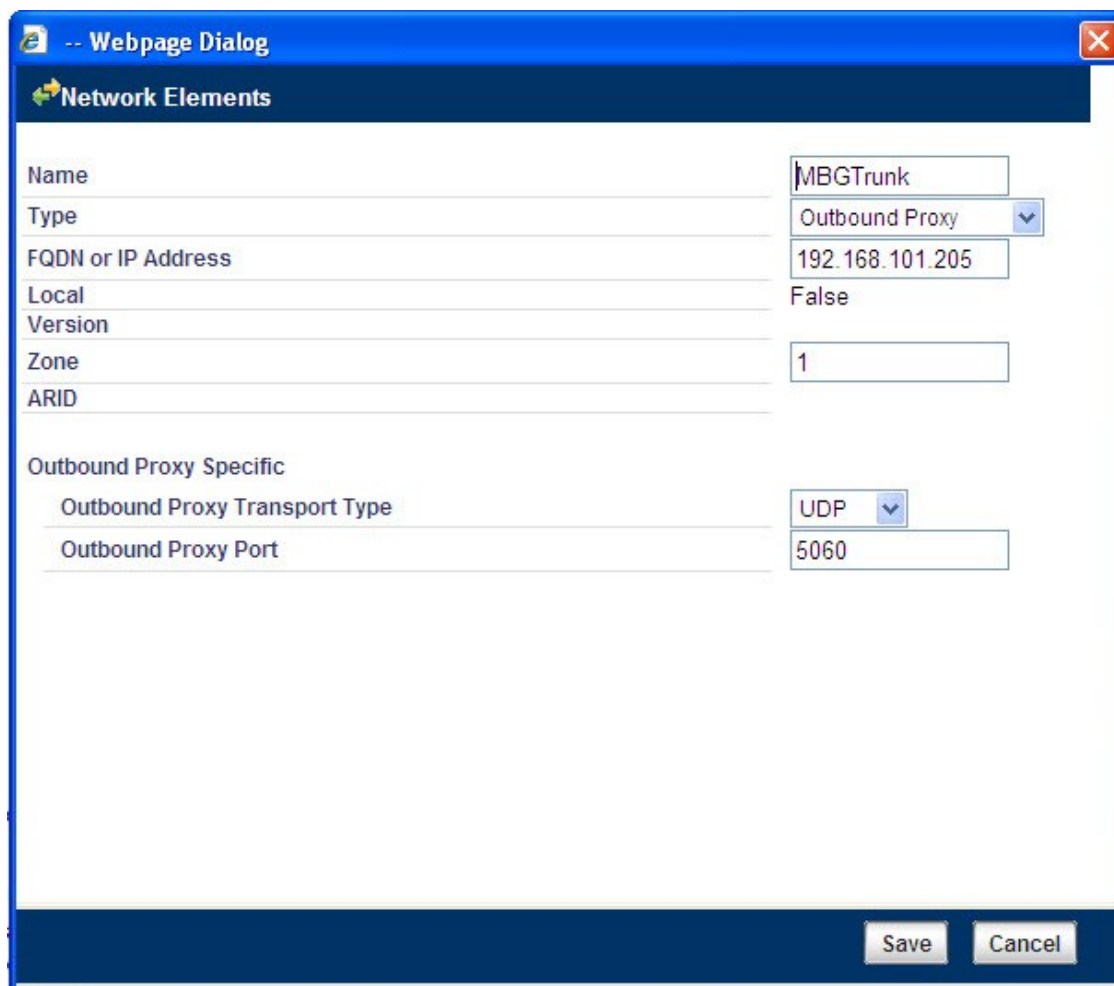
Create a network element for Thinktel SIP Trunking. In this example, the softswitch is reachable by an IP Address and is defined as “ThinkTel” in the network element assignment form. **The IP addresses of the SIP Peer (Network Element), the External SIP Proxy and Registrar are provided by your service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

Name	ThinkTel
Type	Other
FQDN or IP Address	208.68.17.52
Local	False
Version	
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	
External SIP Proxy Transport	default
External SIP Proxy Port	0
SIP Registrar FQDN or IP Address	208.68.17.52
SIP Registrar Transport	default
SIP Registrar Port	5060
SIP Peer Status	Auto-Detect/Normal

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).



The image shows a Windows-style dialog box titled "-- Webpage Dialog" with a sub-header "Network Elements". The dialog contains several fields for configuring a network element:

Name	MBGTrunk
Type	Outbound Proxy
FQDN or IP Address	192.168.101.205
Local	False
Version	
Zone	1
ARID	
<b>Outbound Proxy Specific</b>	
Outbound Proxy Transport Type	UDP
Outbound Proxy Port	5060

At the bottom right of the dialog are "Save" and "Cancel" buttons.

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.

Trunk Attributes	
Trunk Service Number	7
Release Link Trunk	No
Call Recognition Service	Off
Class of Service	7
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	5
Dial In Trunks Incoming Digit Modification - Insert	
Trunk Label	ThinkTel

Save Cancel

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.

SIP Peer Profile				
Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Man
SIP Peer Profile Label	ThinkTel			
Network Element	ThinkTel <input type="button" value="v"/>			
<b>Local Account Information</b>				
Registration User Name	7808092201			
Address Type	<input type="radio"/> FQDN: sipint2.mitel.com <input checked="" type="radio"/> IP Address: 192.168.101.11			
<b>Administration Options</b>				
Interconnect Restriction	1			
Maximum Simultaneous Calls	5			
Outbound Proxy Server	MBGTrunk <input type="button" value="v"/>			
SMDR Tag	0			
Trunk Service	7			
Zone	1			
<b>Authentication Options</b>				
User Name	7808092201			
Password	●●●●●●			
Confirm Password	●●●●●●			
Authentication Option for Incoming Calls	No Authentication <input type="button" value="v"/>			
Subscription User Name				
Subscription Password				
Subscription Confirm Password				

Figure 7 – SIP Peer Profile Assignment- Basic

Basic	Call Routing	Calling Line ID	SDP Options
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 Assignment- Call Routing**

Basic	Call Routing	Calling Line ID	SDP Options	Signaling
Profile Information				
Default CPN		7808092201		
Default CPN Name				
CPN Restriction		No		
Public Calling Party Number Passthrough		No		
Strip PNI		No		
Use Diverting Party Number as Calling Party Number		No		

**Figure 9 – SIP Peer Profile Assignment- Calling Line ID**

Basic	Call Routing	Calling Line ID	SDP Options	Signal
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				Yes
Force sending SDP in initial Invite - Early Answer				No
Limit to one Offer/Answer per INVITE				No
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

**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				
Build Contact Using Request URI Address				
De-register Using Contact Address not *				
Disable Reliable Provisional Responses				
Disable Use of User-Agent and Server Headers				
E.164: Enable sending '+'				
E.164: Add '+' if digit length > N digits				
E.164: Do not add '+' to Emergency Called Party				
E.164: Do not add '+' to Called Party				
Force Max-Forward: 70 on Outgoing Calls				
Ignore Incoming Loose Routing Indication				
Only use SDP to decide 180 or 183				
Require Reliable Provisional Responses on Outgoing Calls				
Use Privacy: none				
Use P-Asserted Identity Header				
Use P-Preferred Identity Header				
Use Restricted Character Set For Authentication				
Use To Address in From Header on Outgoing Calls				
Use user=phone				

**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					
Registration Period					
Registration Period Refresh (%)					
Registration Maximum Timeout					
Session Timer					
Subscription Period					
Subscription Period Minimum					
Subscription Period Refresh (%)					
Invite Ringing Response Timer					

**Figure 12 – SIP Peer Profile Assignment- Timers**

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

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

**Figure 13 – SIP Peer Profile Assignment- Key Press Event**

Outgoing DID Ranges	Profile Information
---------------------	---------------------

Index	DID Range	CPN Substitution

**Figure 14 – SIP Peer Profile Assignment- Outgoing DID Ranges**

Outgoing DID Ranges	Profile Information
---------------------	---------------------

Creator	_____
Date Created	_____
Created on MCD Version	_____
Service Provider	_____
Vendor Notes	_____

**Figure 15 – SIP Peer Profile Assignment- Profile Information**



### 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).

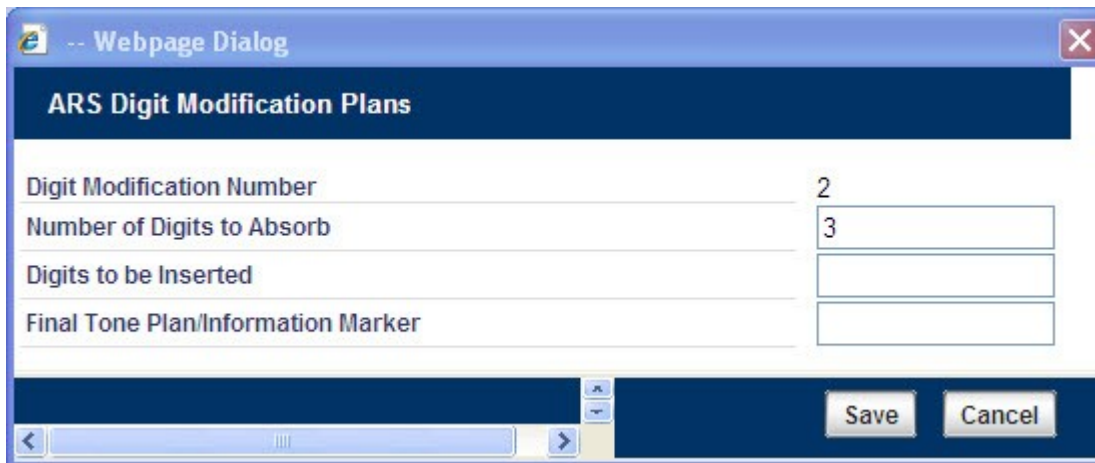
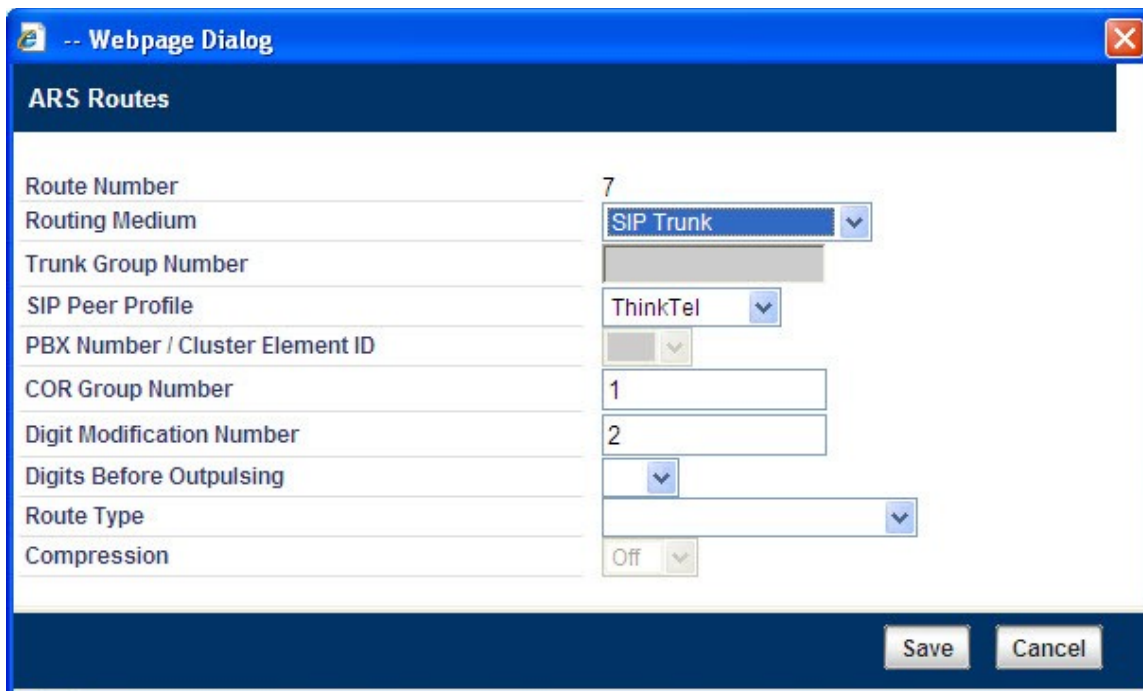


Figure 17 – Digit Modification Assignment

## 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.



The image shows a web browser dialog box titled "ARS Routes" with a blue header and a close button in the top right corner. The dialog contains a form with the following fields and values:

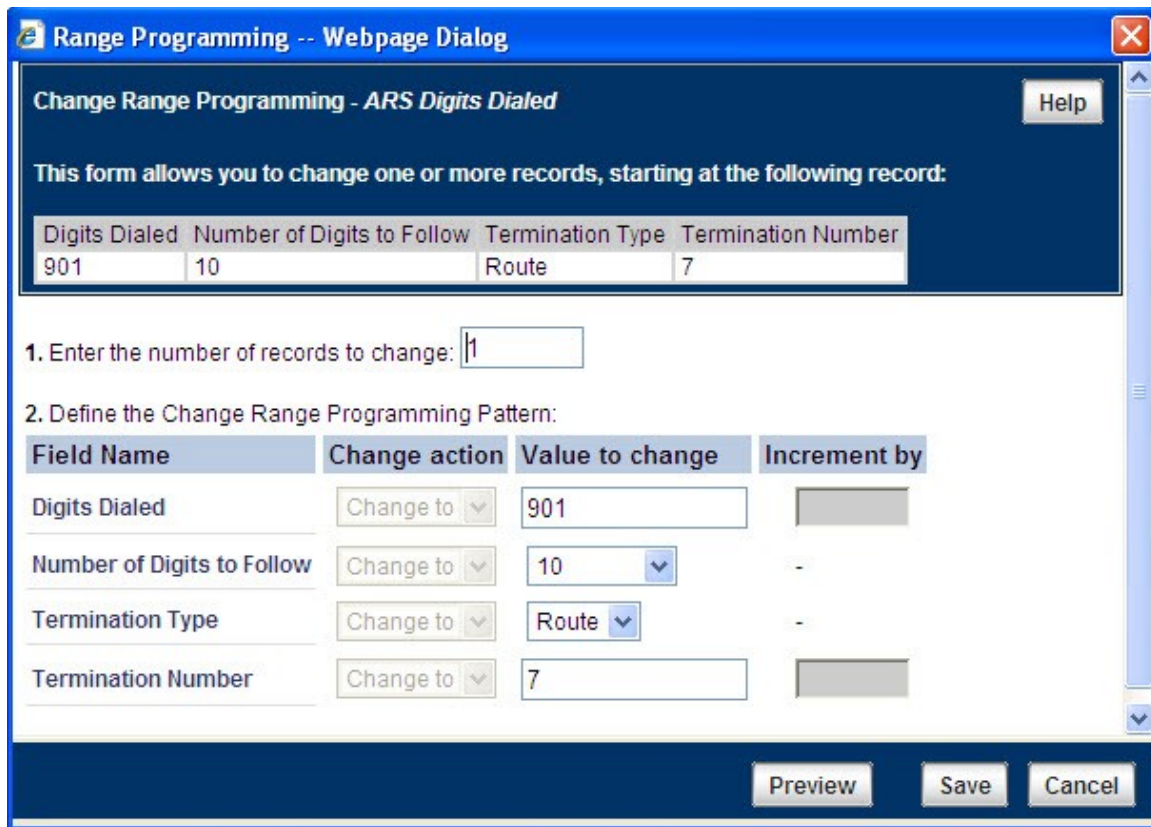
Field	Value
Route Number	7
Routing Medium	SIP Trunk
Trunk Group Number	
SIP Peer Profile	ThinkTel
PBX Number / Cluster Element ID	
COR Group Number	1
Digit Modification Number	2
Digits Before Outpulsing	
Route Type	
Compression	Off

At the bottom right of the dialog, there are two buttons: "Save" and "Cancel".

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).



The screenshot shows a web-based configuration dialog titled "Range Programming -- Webpage Dialog". The main heading is "Change Range Programming - ARS Digits Dialed". A "Help" button is in the top right. Below the heading, a message states: "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
901	10	Route	7

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 <input type="button" value="v"/>	<input type="text" value="901"/>	<input type="text"/>
Number of Digits to Follow	Change to <input type="button" value="v"/>	<input type="text" value="10"/> <input type="button" value="v"/>	-
Termination Type	Change to <input type="button" value="v"/>	<input type="text" value="Route"/> <input type="button" value="v"/>	-
Termination Number	Change to <input type="button" value="v"/>	<input type="text" value="7"/>	<input type="text"/>

At the bottom of the dialog are three buttons: "Preview", "Save", and "Cancel".

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).

The screenshot displays the 'Fax Service Profiles' configuration page in the Mitel Communications Director. The left-hand navigation pane shows the 'Voice Network' section expanded to 'Fax Service Profiles'. The main content area is titled 'Fax Service Profiles on Sipint2' and includes a 'Change' button and several utility buttons (Print, Import, Export, Data Refresh). Below this, the 'Inter-Zone Fax Profile' section is visible, showing 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 the inter-zone profile is a table for 'Intra-Zone Fax Service Profiles'. The table has the following columns: Profile, Maximum Fax Rate, High Speed Redundancy, Low Speed Redundancy, Error Correction Mode, NSF Override, NSF Vendor Code Value, NSF Country Code Value, and Label. The data rows are as follows:

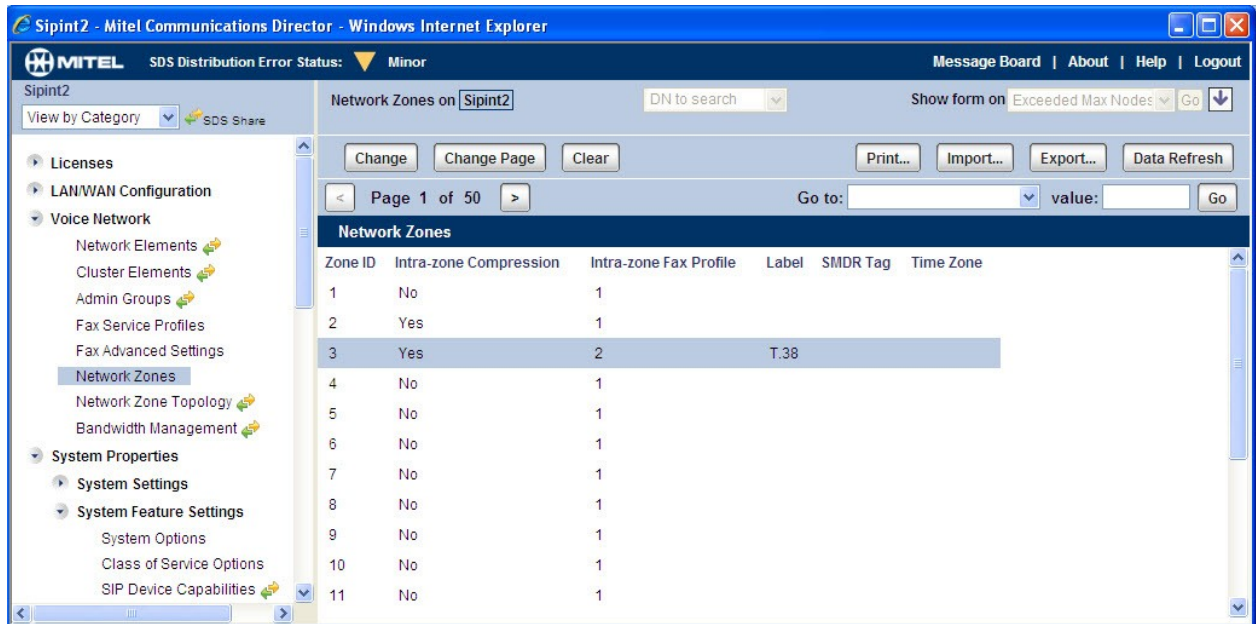
Profile	Maximum Fax Rate	High Speed Redundancy	Low Speed Redundancy	Error Correction Mode	NSF Override	NSF Vendor Code Value	NSF Country Code Value	Label
1	-	-	-	-	-	-	-	G.711
2	14400 (V.17, 14400bps)	0	3	Disabled	Disabled	.	.	T.38
3	7200 (V.29, 7200bps)	0	3	Disabled	Disabled	.	.	T.38 Mike
4	-	-	-	-	-	-	-	
5	-	-	-	-	-	-	-	
6	-	-	-	-	-	-	-	
7	-	-	-	-	-	-	-	
8	-	-	-	-	-	-	-	
9	-	-	-	-	-	-	-	
--	-	-	-	-	-	-	-	

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



The screenshot displays the Sipint2 - Mitel Communications Director web interface. The main content area shows the 'Network Zones' configuration page for 'Sipint2'. The interface includes a navigation menu on the left with categories like Licenses, LAN/WAN Configuration, Voice Network, and System Properties. The 'Network Zones' section is currently selected. The main area displays a table of network zones with the following data:

Zone ID	Intra-zone Compression	Intra-zone Fax Profile	Label	SMDR Tag	Time Zone
1	No	1			
2	Yes	1			
3	Yes	2	T.38		
4	No	1			
5	No	1			
6	No	1			
7	No	1			
8	No	1			
9	No	1			
10	No	1			
11	No	1			

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)

The screenshot shows the 'Manage Mitel Border Gateway' interface. The 'ICPs' tab is active, displaying a table of ICP information. The table has columns for 'Default for MiNet', 'Default for SIP', 'Name', 'Address', 'Type', 'Installer Password', and 'SIP DNS Hostname(s)'. The row for 'sipint4' at address '192.168.101.20' is highlighted in red. Below the table is an 'Update Default ICPs' button.

Default for MiNet	Default for SIP	Name	Address	Type	Installer Password	SIP DNS Hostname(s)		
<input type="radio"/>	<input type="radio"/>	5000_1	192.168.101.50	3300 ICP			<a href="#">Modify</a>	<a href="#">Delete</a>
<input type="radio"/>	<input type="radio"/>	5000_2	192.168.101.52	3300 ICP			<a href="#">Modify</a>	<a href="#">Delete</a>
<input type="radio"/>	<input type="radio"/>	5000_56	192.168.101.56	3300 ICP			<a href="#">Modify</a>	<a href="#">Delete</a>
<input type="radio"/>	<input type="radio"/>	MICD	192.168.101.85	3300 ICP			<a href="#">Modify</a>	<a href="#">Delete</a>
<input type="radio"/>	<input checked="" type="radio"/>	sipint3	192.168.101.21	3300 ICP		cust3-tor.vaac.bell.ca	<a href="#">Modify</a>	<a href="#">Delete</a>
<input type="radio"/>	<input type="radio"/>	sipint1	192.168.101.10	3300 ICP			<a href="#">Modify</a>	<a href="#">Delete</a>
<input checked="" type="radio"/>	<input type="radio"/>	sipint2	192.168.101.11	3300 ICP			<a href="#">Modify</a>	<a href="#">Delete</a>
<input type="radio"/>	<input type="radio"/>	sipint4	192.168.101.20	3300 ICP			<a href="#">Modify</a>	<a href="#">Delete</a>
<input type="radio"/>	<input type="radio"/>	vMCD	192.168.101.30	3300 ICP			<a href="#">Modify</a>	<a href="#">Delete</a>

**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)

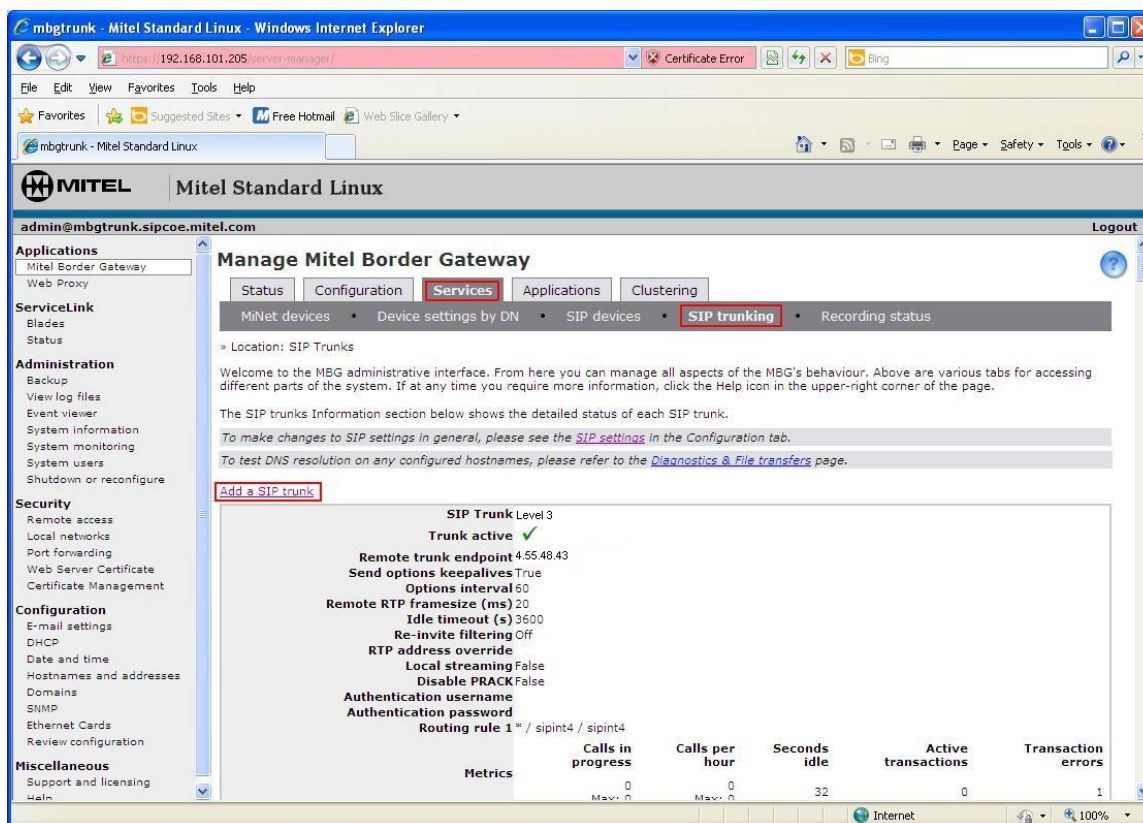


Figure 23 – SIP trunking configuration page

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

**Manage Mitel Border Gateway**

Status Configuration **Services** Applications Clustering

MINet devices • Device settings by DN • SIP devices • **SIP trunking** • Recording status

**There is an outstanding alarm on this system. Please see the [MSL event viewer](#) for details.**

> Location: [SIP Trunks](#) / Edit SIP Trunk - ThinkTel

Welcome to the MBG administrative interface. From here you can manage all aspects of the MBG's behaviour. Above are various tabs for accessing different parts of the system. upper-right corner of the page.

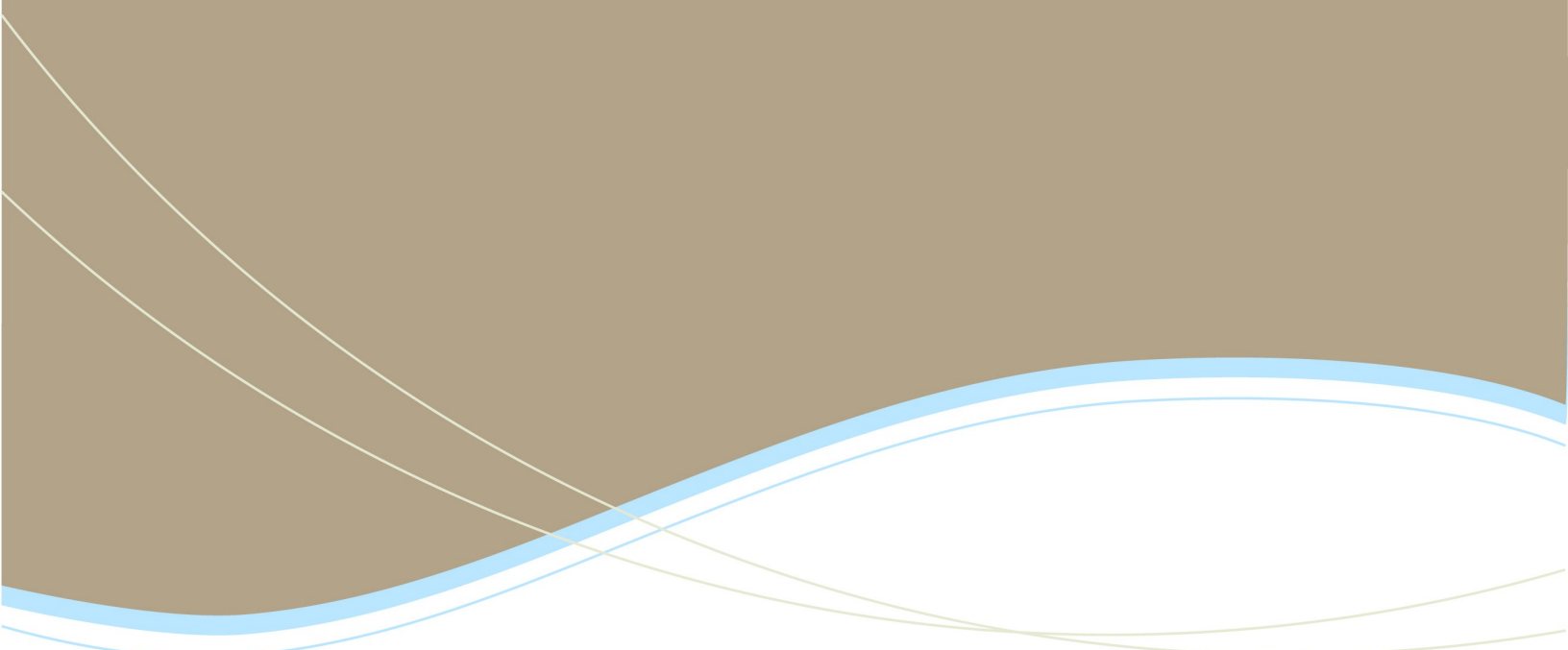
This interface provides the ability to edit a SIP trunk's details. Edit below, and click the "Save" button to commit the changes. If you do not wish to save, simply navigate elsewhere

<b>Name:</b>	ThinkTel
<b>Remote trunk endpoint address:</b>	208.68.17.52
<b>Remote trunk endpoint port:</b>	5060
<b>Options keepalives:</b>	Use master setting
<b>Options interval:</b>	60
<b>Remote RTP framesize (ms):</b>	20ms
<b>Idle timeout (s):</b>	3600
<b>Re-invite filtering:</b>	Off
<b>RTP address override:</b>	---
<b>Local streaming:</b>	<input type="checkbox"/>
<b>PRACK support:</b>	Use master setting
<b>Log verbosity:</b>	Use master setting
<b>Authentication username:</b>	7808092201
<b>Authentication password:</b>	
<b>Confirm authentication password:</b>	

[Add additional routing rule](#)

*	sipint2	-----	<a href="#">Up</a> <a href="#">Down</a> <a href="#">Delete</a>
---	---------	-------	--

Figure 24 – SIP Trunk configuration settings



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

For more information on our worldwide office locations, visit our website at [www.mitel.com/offices](http://www.mitel.com/offices)

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