

MITEL – SIP CoE

Technical Configuration Notes



Configure MCD 6.0 SP3 for use with
BT GS

SIP COE 11-4940-00172

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Mitel Technical Configuration Notes:

Configure the Mitel MCD 6.0 SP3 for use with BT GS
April 2014 11-4940-00172_2

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OVERVIEW	1
Interop History.....	1
Interop Status	1
Software & Hardware Setup.....	1
Tested Features.....	2
Device Limitations and Known Issues	3
Network Topology	4
CONFIGURATION NOTES	5
BT GS Configuration Notes.....	5
3300 MCD Configuration Notes	6
Network Requirements.....	6
Assumptions for the 3300 MCD Programming.....	6
Licensing and Option Selection – SIP Licensing	7
Class of Service Assignment	8
Network Element Assignment	9
Network Element Assignment (Proxy)	10
Trunk Service Assignment	11
SIP Peer Profile	12
SIP Peer Profile Assignment by Incoming DID	16
ARS Digit Modification Number.....	17
ARS Routes Assignment.....	18
ARS Digits Dialed Assignment.....	19
Fax Configuration	20
Zone Assignment	22
Mitel Border Gateway Setup	23
MBG Setup.....	23
ICP Setup	24
SIP Trunk Setup	25
Configuration Settings SIP Options.....	26

Overview

This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel 3300 MCD to connect to BT GS. 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	June 2011	Initial Interop with Mitel 3300 MCD 4.2 SP2 and BT GS
2	April 2014	Interop with Mitel 3300 MCD 6.0 SP3 and BT GS

Interop Status

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

 COMPATIBLE	<p>The most common certification which means BT GS 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 BT GS and the 3300 MCD.

Manufacturer	Variant	Software Version
Mitel	3300 MCD 6.0 SP3 – Mxe Platform	12.0.3.15
Mitel	MBG - Gateway	8.0.26.0
Mitel	MBG - Teleworker	8.0.17.0
Mitel	Nupoint Voicemail	14.1
Mitel	5330 SIP Sets	SIP (05.02.03.01)
Mitel	5330/5340 IP Sets	Minet (05.02.03.01)
PSTN Gateway	CISCO PGW	
SBC	ACME 4250	ACME Net-Net 4250 Firmware SC6.1.0 MR-11 patch 1 (Build 1036)

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 BT GS and their PSTN gateway, 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 60ms	
Personal Ring Groups	Receiving calls through BT GS and their PSTN gateway to a personal ring group. Also moving calls to/from the prime member and group members.	
Teleworker	Making and receiving a call through BT GS and their PSTN gateway to and from Teleworker extensions.	
Video	Making and receiving a call through BT GS with video capable devices.	
Fax	T.38 and G711Fax Calls	

 - No issues found

 - Issues found, cannot recommend to use

 - Issues found

Device Limitations and Known Issues

This is a list of problems or not supported features when BT GS is connected to the Mitel MCD 6.0 SP3.

Feature	Problem Description
Video	<p>BT GS does not support video calls over SIP Trunks.</p> <p>Recommendation: Consult BT GS for future deployment of SIP video calls.</p>
Packetization	<p>3300 MCD was tested with stream RTP packets through its E2T card at different intervals of 20ms to 30ms (BT recommendation) instead of the Mitel recommended range of 10ms to 60ms.</p> <p>Recommendation: Use BT GS 20ms or 30ms recommended setting.</p>
Loopback Testing and RFC 3261 Call-ID 8.1.1.4	<p>During standard Mitel SIP trunking test setup we had an issue with all DID loopback calls, whereby the inbound INVITE back to the MBG from BT re-uses the same call-id and the MBG responded back with a 491 request pending.</p> <p>Recommendation: Mitel strongly suggests that re-using a call ID violates a MUST and should be fixed. As a result no loopback calls were tested and all test calls were therefore limited to either originate/terminate from a Mitel (SIP, IP and Analog) phone set or FAX to/from a respective private BT GS PSTN, IVR or FAX extension.</p>

Network Topology

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

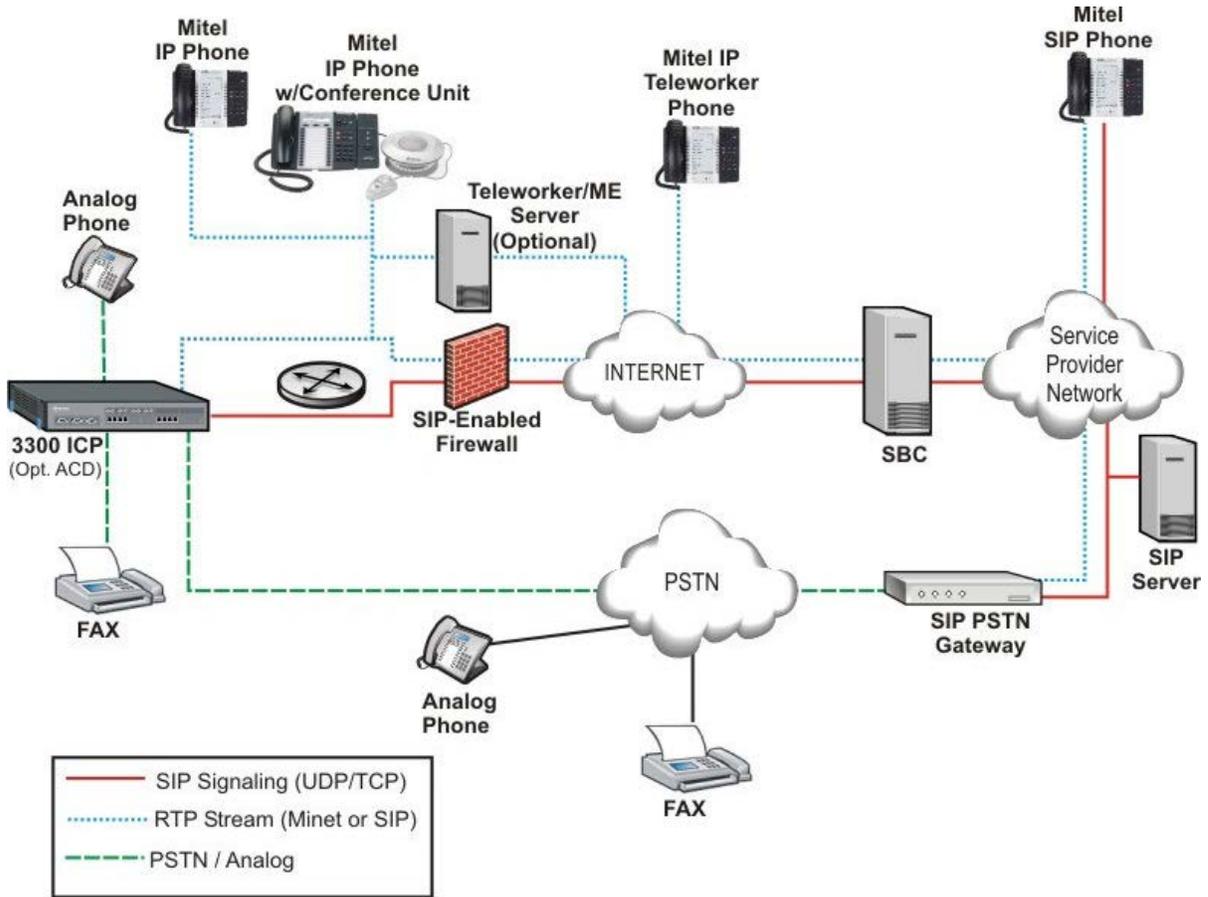


Figure 1 – Network Topology

Configuration Notes

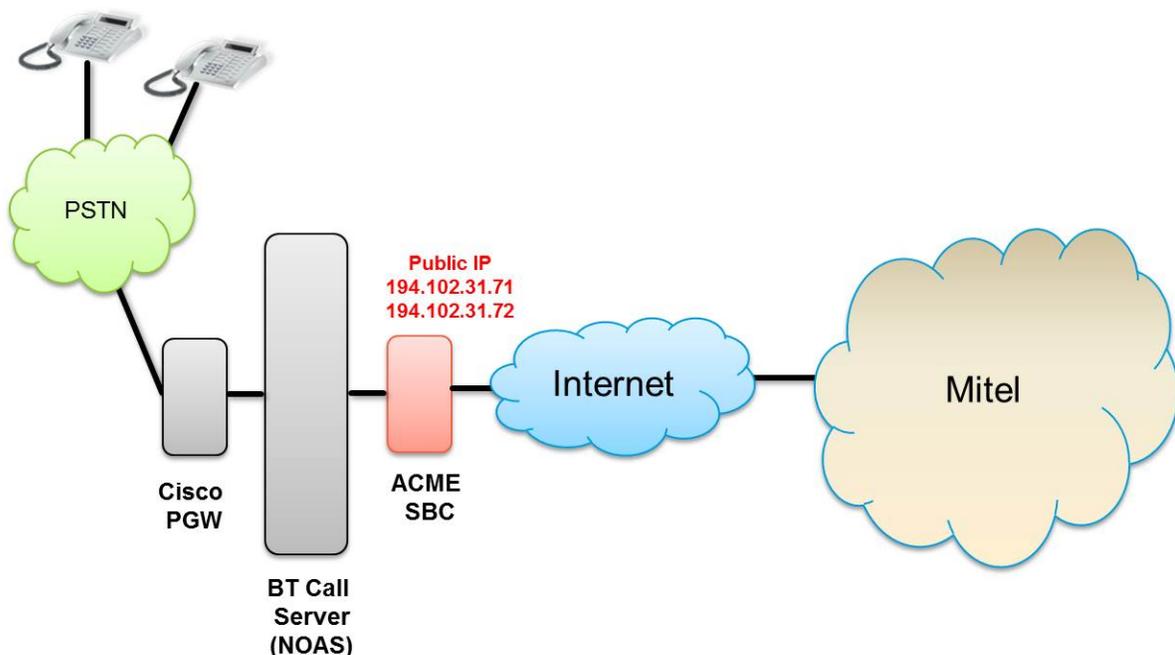
This section is a description of how the SIP Interop was configured. These notes should give a guideline how a device can be configured in a customer environment and how BT GS 3300 programming was configured in our test environment.

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.

BT GS Configuration Notes

The BT GS testing environment was an internal/private Sandbox setup with no connection to outside public network other than the Mitel MBG via redundant BT SBC.

BT NOAS – Mitel Testing Setup



SIP Service Provider Server IP address	SBC IPs: 194.102.31.71 and 194.102.31.72
Media Server	N/A
Registration and Authentication	N/A
Pilot Number	N/A
Username/Password	05511500200
DIDs	05511500200-05511500202
PSTN	01912500753-01912500754
FAX	01912226651
IVR	08207232007
Preferred Codec	G711A,G711u,G729
SIP port	5060
Transport Type	UDP
Session Timer	Requests will be ignored

3300 MCD Configuration Notes

The following steps show how to program a 3300 MCD to interconnect with BT GS.

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 3300 MCD Programming

- The SIP signaling connection uses UDP on Port 5060

Licensing and Option Selection – SIP Licensing

Ensure that the 3300 MCD is equipped with enough SIP trunking licenses for the connection to BT GS. 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 'License and Option Selection' page for Sipint5. The left sidebar contains a navigation menu with 'Licenses' expanded to 'License and Option Selection'. The main content area shows the 'License and Option Selection' page for application record ID 17943750. A table lists various licensed options with their respective consumption and allocation details. The 'SIP Trunks' row is highlighted with a red box, indicating 791 locally consumed licenses and 1000 locally allocated licenses.

Licensed Options	Locally Consumed	Locally Allocated	Available for Allocation	Purchased	Local Limits	
					Licenses Allowed	Can be Over Allocated
Users						
IP Users	107	1000	100	1100	Unrestricted	Yes
External Hot Desk Users	3	100	0	100	Unrestricted	Yes
ACD Active Agents	0	100	0	100	Unrestricted	No
HTML Applications	2	10	0	10	Unrestricted	Yes
Analog Lines	0	20	0	20	Unrestricted	Yes
IP Console Active Operators	0	0	0	0	0	No
Multi-device Users	0	0	20	0	Unrestricted	Yes
Multi-device Suites	0	0	0	0	0	No
Messaging						
Embedded Voice Mail	83	750	0	750	Unrestricted	Yes
Embedded Voice Mail PMS	1	Yes	0	1	Unrestricted	Yes
Trunking/Networking						
Digital Links	0	8	0	8	Unrestricted	Yes
Compression	16	16	0	16	Unrestricted	Yes
FAX Over IP (T.38)	16	16	0	16	Unrestricted	Yes
SIP Trunks	791	1000	0	1000	Unrestricted	Yes
Others						
MCD IDS Connection	0	No	1	0	Unrestricted	Yes
MLPP	0	No	0	0	Unrestricted	No

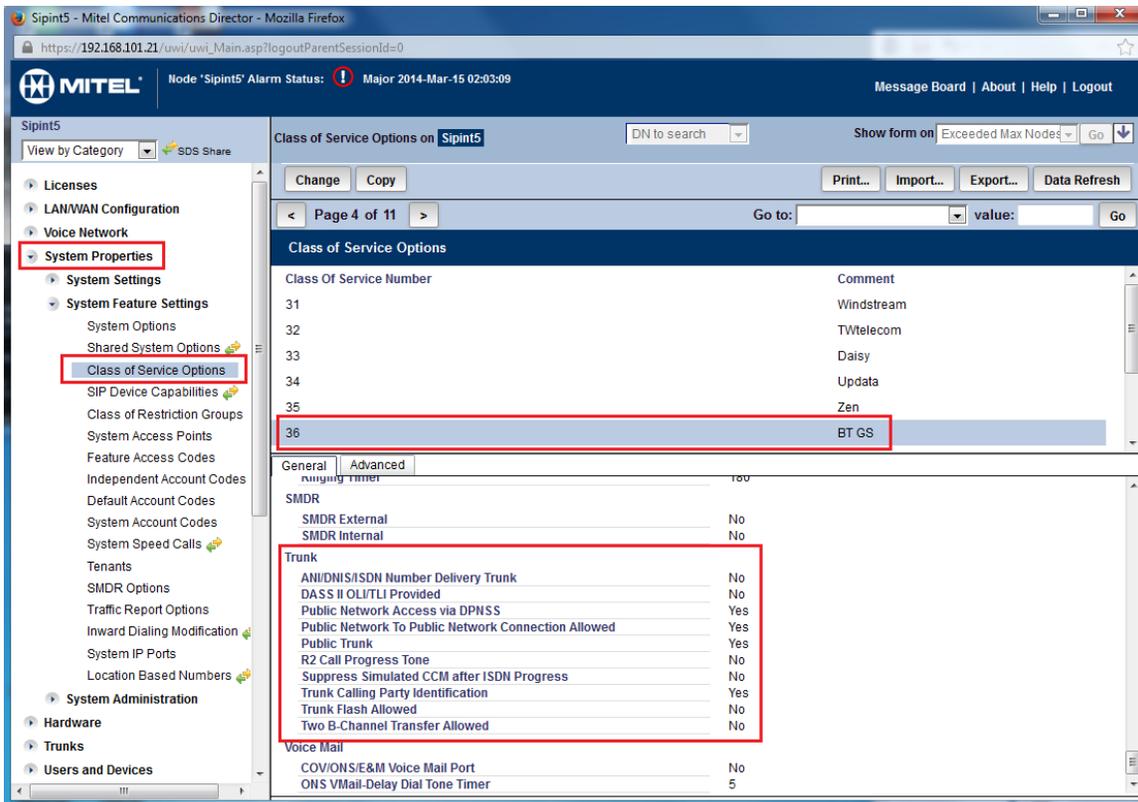
Figure 2 – License and Option Selection

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**
- Clear Auto Campon Timer to enable 486 BUSY testing



Campon
Auto Campon Timer
Campon Recall Timer 10

Figure 3 – Class of Service

Network Element Assignment

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

If your service provider trusts your network connection by asking for your gateway external IP address, then programming the IP address for the SIP Peer, Outbound Proxy and Registrar is not required for SIP trunk integration. This will need to be verified with your service provider. Set the transport to UDP and port to 5060.

The screenshot displays the Mitel SIPint5 configuration interface. On the left, a navigation tree shows 'Voice Network' > 'Network Elements' selected. The main area shows a table of network elements:

Name	Type	FQDN or IP Address	Data Sharing	Local	Version	Zone	AKID
BT	Other	194.102.31.71	NO	False			
BT_GS	Other	194.102.31.71	NO	False		2	
btestq2	Other	65.57.109.232	NO	False			
BTi	Other	btsiptk.bt.com	NO	False			
BTItalia	Other	213.213.83.151	NO	False			
CandW	Other	212.165.24.6	NO	False			
CBeyond	Other	sipconnect-fca.at0.cbeyond.net	NO	False			

The configuration form for the selected 'BT_GS' element is shown below:

Name: BT_GS
 Type: Other
 FQDN or IP Address: 194.102.31.71
 Data Sharing: NO
 Local: False
 Version:
 Zone: 2 (Zone 1 was used for G.711 Fax testing)
 AKID:
 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:
 SIP Registrar Transport: default
 SIP Registrar Port: 0
 SIP Peer Status: Auto-Detect/Normal

Figure 4 – Network Element Assignment

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 3300 ICP 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).

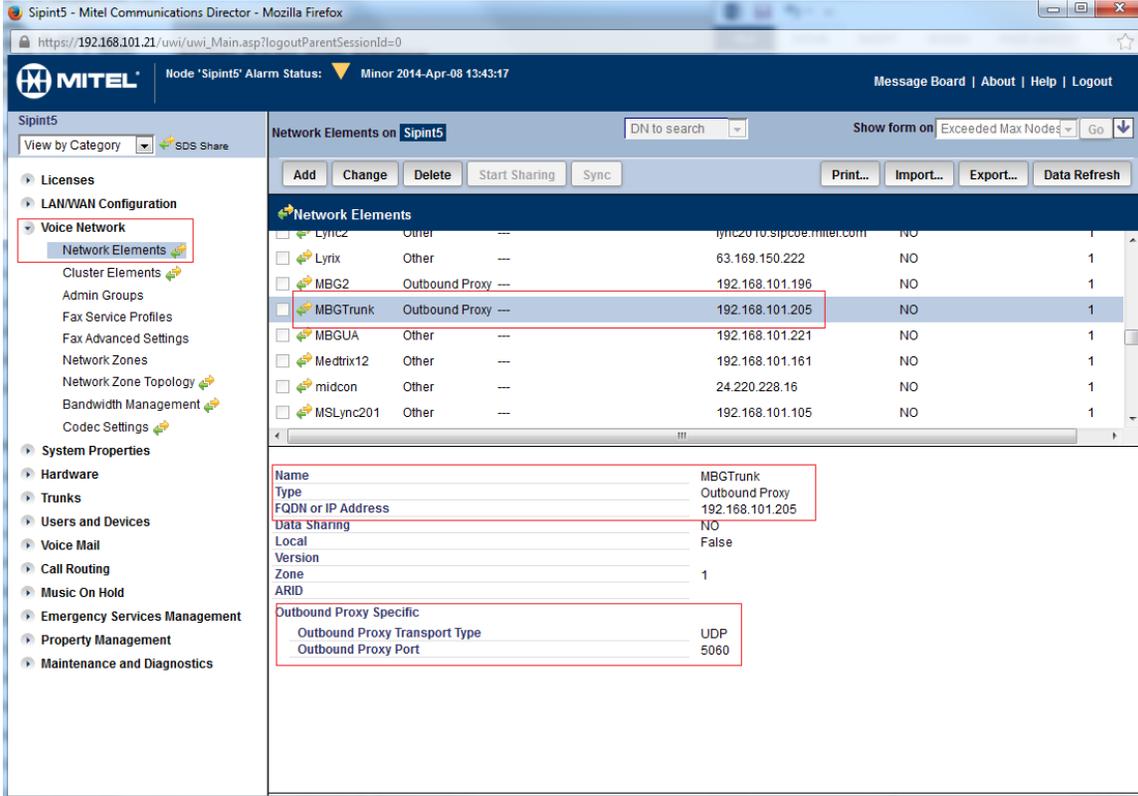


Figure 5 – Network Element (Proxy)

Trunk Service Assignment

This is configured in the Trunk Service Assignment form. In this example the Trunk Service Assignment is defined for Trunk Service Number 25 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 7 digits of the DID number from BT GS leaving 5 digits for the 3300 to translate and ring a 4 digit extension. For example, BT GS delivers +44-551-150-0200 through the SIP trunk to the 3300. The 3300 will absorb the first 7 digits (4455115) leaving the 3300 to ring extension 00200. Extension 00200 must be programmed as a valid 4 digit dialable number in the 3300 via system speed call. Please refer to the 3300 System Administration documentation for further programming information.

The screenshot displays the Mitel SIPint5 web interface for configuring trunk services. The left sidebar shows a navigation tree with 'Trunks' selected. The main content area shows a table of trunk attributes and a detailed configuration form for Trunk Service Number 25.

Trunk Service Number	Release Link Trunk	Call Recognition Service	Class of Service	Class of Restriction	Baud Rate	Intercept Number	Trunk Label
21	No	Off	9	1	300	1	Poly_DMA
22	No	Off	9	1	300	1	Poly_RMX
23	No	Off	1	1	300	1	
24	No	Off	1	1	300	1	
25	No	Off	36	1	300	1	BT_GS
26	No	Off	35	1	300	1	Zen
27	No	Off	34	1	300	1	Updata
28	No	Off	33	1	300	1	Daisy

Trunk Service Number	25
Release Link Trunk	No
Call Recognition Service	Off
Class of Service	36
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	7
Dial In Trunks Incoming Digit Modification - Insert	
Dial In Trunks Answer Point	
Dial In Trunks Insert Forwarding Information	No
Trunk Label	BT_GS

Figure 6 – Trunk Service Assignment

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 "BT GS" Network Element.

Registration User Name: BT GS does require the use of a Registration User Name.

Address Type: Select IP address.

Default CPN: The default CPN "05511500200" 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 BT GS. 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 BT GS.

NOTE-1: Ensure the remaining SIP Peer profile policy options are similar the screen capture below.

NOTE-2: SDP options had Updates disabled (not supported by BT) and Force SDP sent in the initial invite enabled.

NOTE-3: BT Session Timer not supported or used and set to 0.

Sipint5 - Mitel Communications Director - Mozilla Firefox
 https://192.168.101.21/uwi/uwi_Main.asp?logoutParentSessionId=0

Node 'Sipint5' Alarm Status: Minor 2014-Apr-08 13:43:17
 Message Board | About | Help | Logout

Sipint5
 View by Category SDS Share

SIP Peer Profile on Sipint5
 DN to search Show form on Exceeded Max Node

AltitudeV	Altitude	Ingate	No	35	120	1
BTI	BT	Ingate	No	55	0	1
BT_GS	BT_GS	MBGTrunk	No	25	0	2
IngatePeer	BandTel	No	No	22	0	1
inactive	Bell	MBGTrunk	No	14	90	1

Outgoing DID Ranges Profile Information

SIP Peer Profile Label BT_GS
 Network Element BT_GS

Local Account Information
 Registration User Name 05511500200
 Address Type IP Address: 192.168.101.21

Administration Options
 Interconnect Restriction 1
 Maximum Simultaneous Calls 6
 Minimum Reserved Call Licenses 0

Administration Options
 Outbound Proxy Server MBGTrunk
 SMDR Tag 0
 Trunk Service 25
 Zone 2
 User Name
 Password *****
 Confirm Password *****
 Authentication Option for Incoming Calls No Authentication
 Subscription User Name
 Subscription Password *****
 Subscription Confirm Password *****

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

Outgoing DID Ranges Profile Information

Default CPN 05511500200
 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

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event
Outgoing DID Ranges		Profile Information				
Allow Peer To Use Multiple Active M-Lines						Yes
Allow Using UPDATE For Early Media Renegotiation						No
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
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

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						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
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						Yes
Suppress Redirection Headers						No
Use Fixed Retry Time for 491						No
Use Privacy: none						No
Use P-Asserted Identity Header						Yes
Use P-Asserted Identity for Billing						No
Use P-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

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						0
Subscription Period						3600
Subscription Period Minimum						300
Subscription Period Refresh (%)						80
Invite Ringing Response Timer						0

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

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	
Outgoing DID Ranges	Profile Information						
						Add Member	Delete Member
Index	DID Range	CPN Substitution					
-							
Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	
Outgoing DID Ranges	Profile Information						
Creator							
Date Created							
Created on MCD Version							
Service Provider							
Vendor Notes							

Figure 7 – SIP Peer Profile Assignment

ARS Digit Modification Number

Ensure that Digit Modification for outgoing calls on the SIP trunk to BT GS absorbs or inject additional digits according to your dialling plan. In this example, we will be absorbing 3 digits for both prefixes used: 924 prefix to dial out followed by 11 digits and 925 prefix followed by 13 digits.

The screenshot displays the Mitel SIPint5 web interface. The left-hand navigation menu is expanded to 'Call Routing', with 'Automatic Route Selection (ARS)' and 'ARS Digit Modification Plans' highlighted. The main content area shows a table of ARS Digit Modification Plans for the 'Sipint5' system. The table has four columns: Digit Modification Number, Number of Digits to Absorb, Digits to be Inserted, and Final Tone Plan/Information Marker. The third row of the table is highlighted with a red box, showing a digit modification number of 3 and 3 digits to absorb.

Digit Modification Number	Number of Digits to Absorb	Digits to be Inserted	Final Tone Plan/Information Marker
1	0		
2	2		
3	3		
4	3		
5	1		
6	2		
7	0		
8	2		
9	0		
10	3		
11	0		
12	0		
13	0		
14	0		
15	0		

Figure 9 – Digit Modification Assignment

ARS Routes Assignment

Create a route for SIP Trunks connecting a trunk to BT GS. In this example, the SIP trunk is assigned to Route Number 31. Choose SIP Trunk as a routing medium and choose the SIP Peer Profile and Digit Modification entry created earlier.

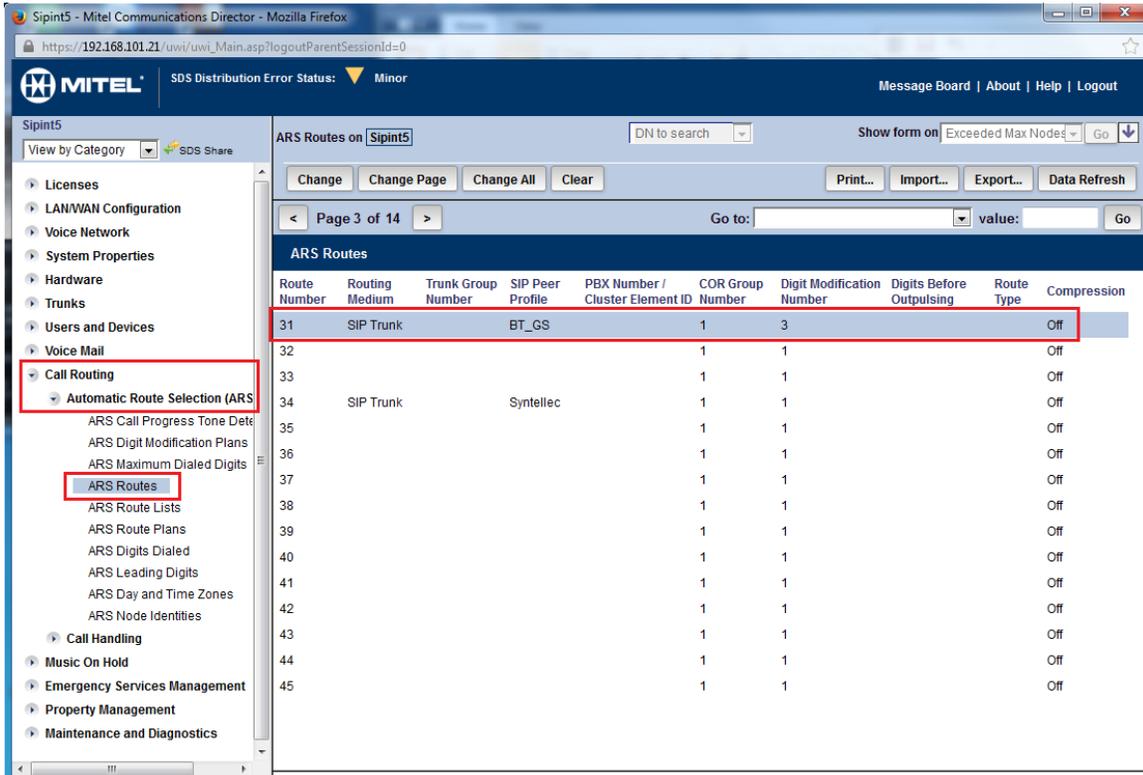


Figure 10 – SIP Trunk Route Assignment

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 924 or 925, the call will be routed to BT GS (ie. Route 31).

The screenshot shows the Mitel SIPint5 web interface. The left sidebar contains a navigation tree with the following items: Licenses, LAN/WAN Configuration, Voice Network, System Properties, Hardware, Trunks, Users and Devices, Voice Mail, Call Routing, Automatic Route Selection (ARS), ARS Call Progress Tone Det, ARS Digit Modification Plans, ARS Maximum Dialed Digits, ARS Routes, ARS Route Lists, ARS Route Plans, ARS Digits Dialed (highlighted), ARS Leading Digits, ARS Day and Time Zones, ARS Node Identities, Call Handling, Music On Hold, Emergency Services Management, Property Management, and Maintenance and Diagnostics. The main content area is titled 'ARS Digits Dialed on Sipint5' and shows a table of configurations. The table has four columns: Digits Dialed, Number of Digits to Follow, Termination Type, and Termination Number. The rows are as follows:

Digits Dialed	Number of Digits to Follow	Termination Type	Termination Number
920	11	Route	24
921	13	Route	24
922	11	Route	30
923	13	Route	30
924	11	Route	31
925	13	Route	31
928	4	Route	18
960	10	Route	17
9601	10	Route	17
961	11	Route	13
962	11	Route	15
963	10	List	1
9631	10	List	1
964	10	Route	14
9641	10	Route	14
965	10	Route	12
9651	10	Route	12
98	Unknown	Route	51

Figure 11 – ARS Digit Dialed Assignment

Fax Configuration

BT GS 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 SIPint5 web interface. The left navigation pane is expanded to 'Voice Network' > 'Fax Service Profiles'. The main content area displays the 'Inter-Zone Fax Profile' configuration form, which is currently empty. Below this, there is a table for 'Intra-Zone Fax Service Profiles'.

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	4800 (V.27ter, 4800bps)	0	3	Disabled	Disabled	.	.	4800
4	-	-	-	-	-	-	-	-

Figure 12 – Fax Service Profiles



The screenshot displays the Mitel Sipint5 configuration interface in a Mozilla Firefox browser window. The address bar shows the URL https://192.168.101.21/uwi/uwi_Main.asp?logoutParentSessionId=0. The page title is "Sipint5 - Mitel Communications Director - Mozilla Firefox". The navigation bar includes the Mitel logo, "Node 'Sipint5' Alarm Status: Minor 2014-Feb-03 13:06:17", and links for "Message Board", "About", "Help", and "Logout".

The main content area is titled "Fax Advanced Settings on Sipint5". It features a "View by Category" dropdown menu set to "LAN/WAN Configuration" and a search field labeled "DN to search". A "Show form on" dropdown is set to "Exceeded Max Nodes" with a "Go" button. A "Change" button is located above the settings table. Below the table are buttons for "Print...", "Import...", "Export...", and "Data Refresh".

The left sidebar contains a tree view of configuration categories: Licenses, LAN/WAN Configuration, Voice Network (highlighted), Network Elements, Cluster Elements, Admin Groups, Fax Service Profiles, Fax Advanced Settings (highlighted), Network Zones, Network Zone Topology, Bandwidth Management, Codec Settings, and System Properties.

Fax Advanced Settings	
Enable V.34 Fax Interop at V.17 speeds with SIP GWs	Enabled
Enable T.38 FAX Logging	Enabled
T.38 RX Level	-43 dBm (default)
Fax Detect Level	-35 dBm (default)
Fax Detect Timeout	60 s (default)

Figure 13 – Fax Advanced Settings

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". BT GS uses the Inter-zone FAX Profile.

The screenshot displays the Mitel Sipint5 Network Zones configuration page. The left sidebar shows the navigation menu with 'Network Zones' highlighted. The main content area shows a table of Network Zones. The table has the following data:

Zone ID	Intra-zone Compression	Intra-zone Fax Profile	Label	SMDR Tag	Time Zone	LBN Prefix	Zone CESID	Default Billing Number	Default CPN
1	No	1							
2	No	2							
3	No	1							
4	No	1							
5	No	1							
6	No	1							

Figure 14 – Zone Assignment

Mitel Border Gateway Setup

MBG Setup

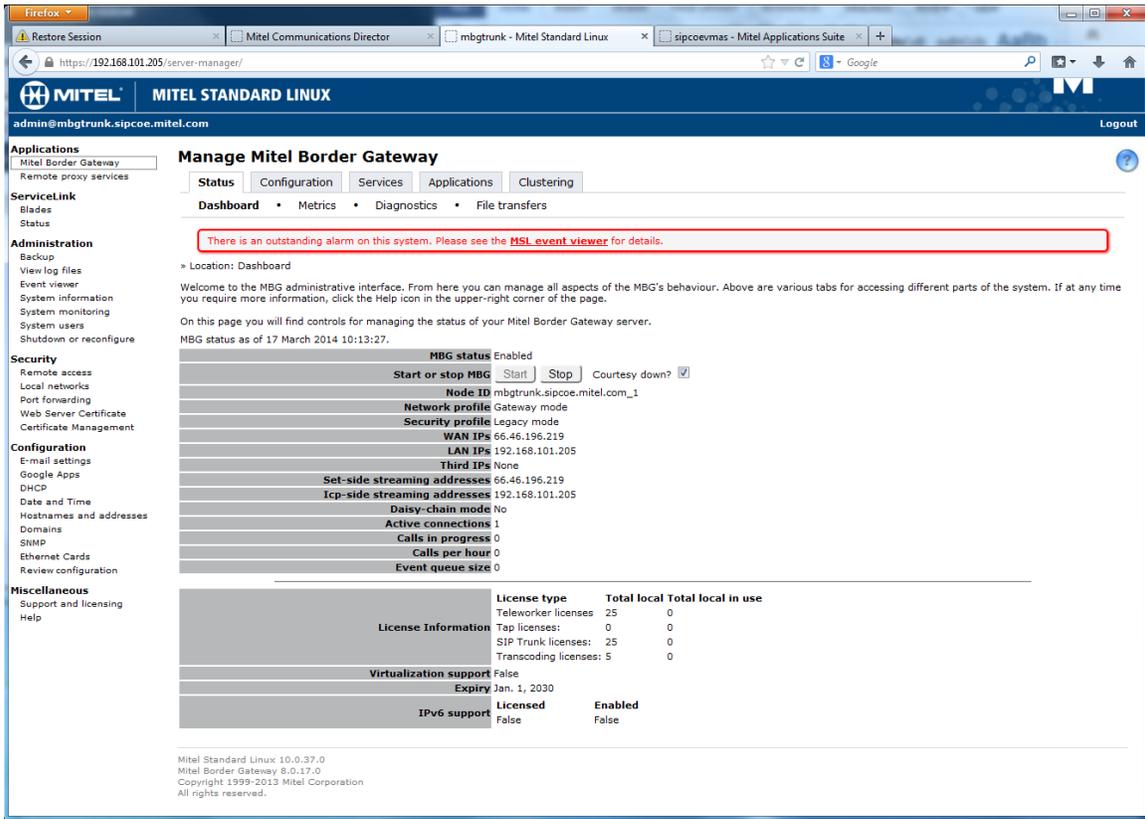


Figure 15 – MBG setup

ICP Setup

To program an MCD into the MBG, click on ICP's → Add an ICP.

Enter a name for the MCD.

Enter the IP address of the MCD and select the Type as MCD.

The screenshot shows the Mitel Standard Linux web interface in a Mozilla Firefox browser. The browser tabs include 'mbgtrunk - Mitel Standard Linux', 'Mitel Communications Director', 'Mitel 5330 SIP Phone', and 'Mitel Collaboration Advanced'. The address bar shows 'https://192.168.101.205/server-manager/'. The page title is 'MITEL STANDARD LINUX' and the user is logged in as 'admin@mbgtrunk.sipcoe.mitel.com'. The main content area is titled 'Manage Mitel Border Gateway' and has tabs for 'Status', 'Configuration', 'Services', 'Applications', and 'Clustering'. Under the 'Configuration' tab, there are sub-tabs for 'Settings', 'Network profiles', 'ICPs', 'Bandwidth management', 'Alarms', and 'Overrides'. A red-bordered warning box states: 'There is an outstanding alarm on this system. Please see the [MSL event viewer](#) for details.' Below this, the location is set to 'ICPs / Modify'. The main text reads: '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. If at any time you require more information, click the Help icon in the upper-right corner of the page. The following is a form for modifying an icp entry. You may edit this information as you wish, and click on the "Save" button below when you are done.' The form fields are: 'Name: Sipint5', 'Hostname or IP address: 192.168.101.21', 'Type: MCD', 'Installer password:', 'Indirect call recording capable: ', and 'Indirect call recording password(*):'.

Applications
 Mitel Border Gateway
 Remote proxy services

ServiceLink
 Blades
 Status

Administration
 Backup
 View log files
 Event viewer
 System information
 System monitoring
 System users
 Shutdown or reconfigure

Security
 Remote access
 Local networks
 Port forwarding
 Web Server Certificate
 Certificate Management

Configuration
 E-mail settings
 Google Apps
 DHCP
 Date and Time
 Hostnames and addresses
 Domains
 SNMP
 Ethernet Cards
 Review configuration

Miscellaneous
 Support and licensing
 Help

Manage Mitel Border Gateway

Status Configuration Services Applications Clustering

Settings • Network profiles • ICPs • Bandwidth management • Alarms • Overrides

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

» Location: [ICPs](#) / Modify

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. If at any time you require more information, click the Help icon in the upper-right corner of the page. The following is a form for modifying an icp entry. You may edit this information as you wish, and click on the "Save" button below when you are done.

Name: Sipint5
 Hostname or IP address: 192.168.101.21
 Type: MCD
 Installer password:
 Indirect call recording capable:
 Indirect call recording password(*):

Save

(*) Password must match corresponding password on MCD, once the capability to implement it is introduced.

Mitel Standard Linux 10.0.37.0
 Mitel Border Gateway 8.0.17.0
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Figure 16 – ICP setup

SIP Trunk Setup

Under the Services tab, click on SIP trunking and then “Add a SIP Trunk”. Enter the SIP trunk’s details as shown in Figure 17:

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).

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

The rest of the settings are optional and could be configured if required. Click **Save** button

The screenshot shows the Mitel Standard Linux administrative interface. The main content area is titled "Manage Mitel Border Gateway" and is under the "Services" tab. A red box highlights an alarm message: "There is an outstanding alarm on this system. Please see the MSL event viewer for details." Below this, the "SIP Trunk BT_GS" configuration is displayed. A red box highlights the "Trunk status" section, which includes the following settings:

- Remote trunk endpoint: 194.102.31.71 : 5060
- Send options keepalives: Use master setting
- Options interval: 60
- Rewrite host in PAI: True
- Remote RTP framesize (ms): 0
- Idle timeout (s): 3600
- Re-invite filtering: Off
- RTP address override: False
- Local streaming: False
- PRACK support: Use master setting
- Log verbosity: Use master setting

Below the trunk status, the "Routing rules" section is shown. A red box highlights the "Rule number", "Header match rule", "Pattern", and "Primary destination" columns. The table contains one rule:

Rule number	Header match rule	Pattern	Primary destination	Secondary destination
1	req	*	Sipint5	None

At the bottom of the configuration area, there is a "Filter rules list (Pattern or destination)" section with "Apply" and "Clear" buttons. Below that is a "Metrics" table:

	Calls in progress	Calls per hour	Seconds idle	Active transactions	Transaction errors
	0	0	6	0	13
Max:	2	285			

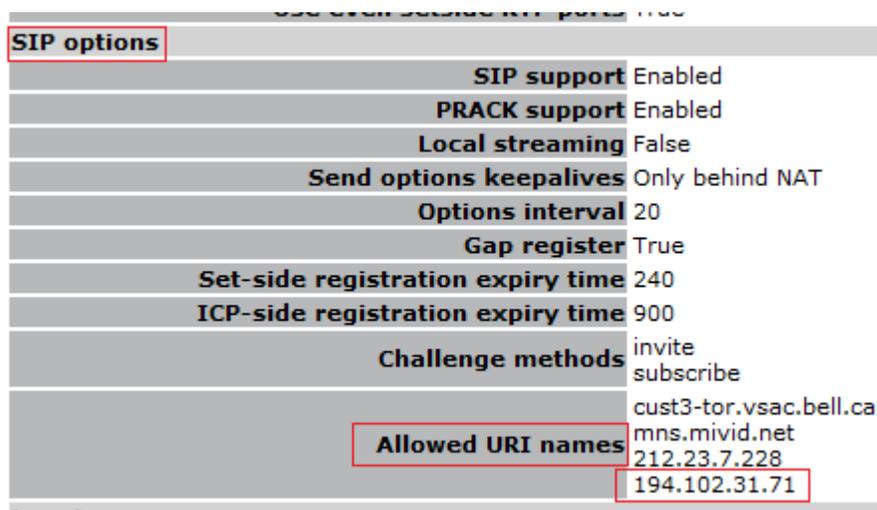
At the bottom right of the configuration area, there are "Modify" and "Delete" buttons. The footer of the page contains the following text:

Mitel Standard Linux 10.0.46.0
 Mitel Border Gateway 8.0.26.0
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Figure 17 – Services - SIP Trunking setup

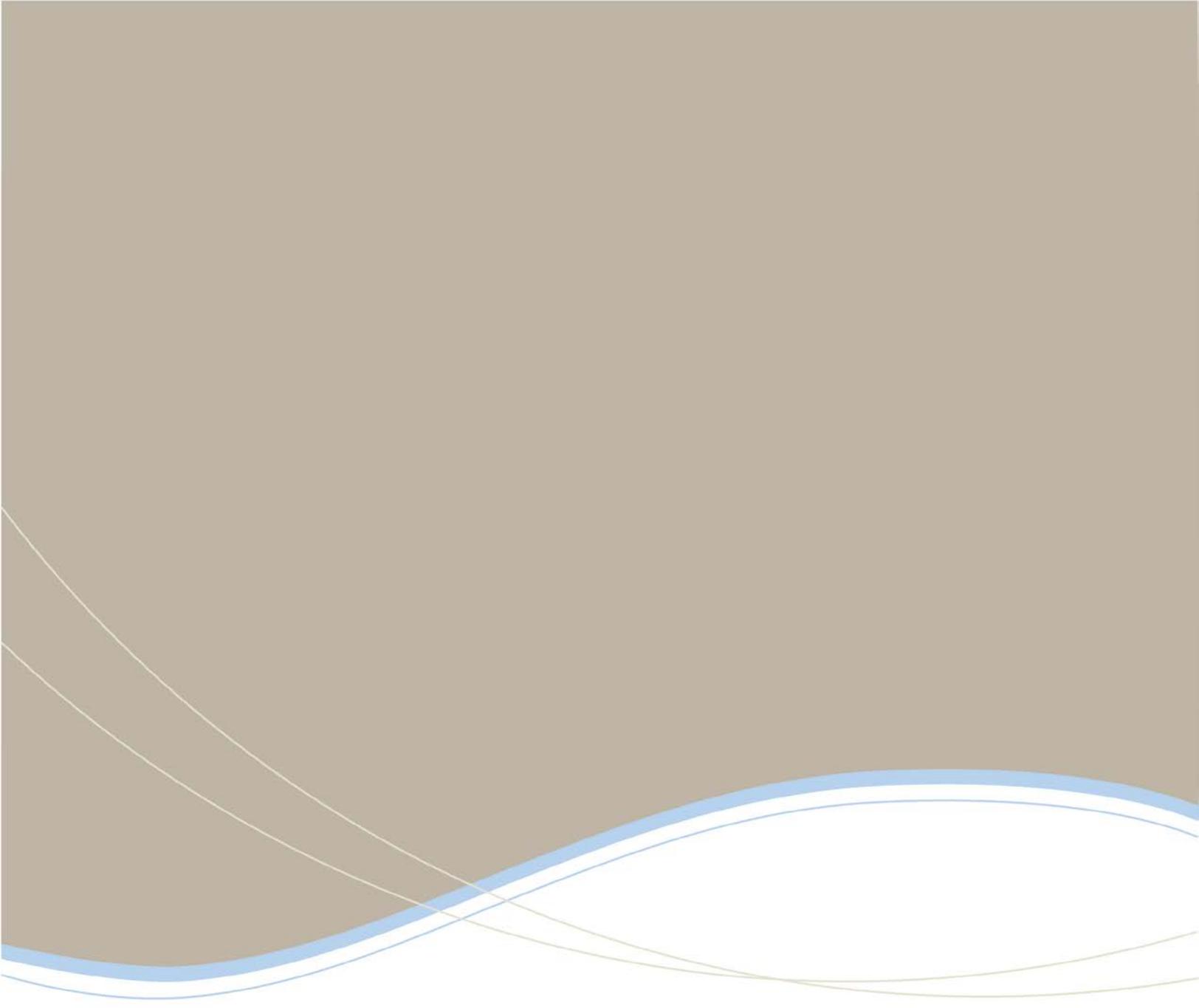
Configuration Settings SIP Options

Under the Configurations tab, click on Settings.



SIP options	
SIP support	Enabled
PRACK support	Enabled
Local streaming	False
Send options keepalives	Only behind NAT
Options interval	20
Gap register	True
Set-side registration expiry time	240
ICP-side registration expiry time	900
Challenge methods	invite subscribe
Allowed URI names	cust3-tor.vsaac.bell.ca mns.mivid.net 212.23.7.228 194.102.31.71

Figure 18 – Configuration – Settings - SIP Options



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