

MITEL – SIP CoE

# Technical Configuration Notes

Configure MCD 4.1 SP1 for use with  
the Lyrix Speech Enabled Auto  
Attendant



## NOTICE

The information contained in this document is believed to be accurate in all respects but is not warranted by Mitel Networks™ Corporation (MITEL®). The information is subject to change without notice and should not be construed in any way as a commitment by Mitel or any of its affiliates or subsidiaries. Mitel and its affiliates and subsidiaries assume no responsibility for any errors or omissions in this document. Revisions of this document or new editions of it may be issued to incorporate such changes.

No part of this document can be reproduced or transmitted in any form or by any means - electronic or mechanical - for any purpose without written permission from Mitel Networks Corporation.

## TRADEMARKS

Mitel is a trademark of Mitel Networks Corporation.

Windows and Microsoft are trademarks of Microsoft Corporation.

Other product names mentioned in this document may be trademarks of their respective companies and are hereby acknowledged.

Mitel Technical Configuration Notes – Configure MCD 4.1 SP1 for use with the Lyrinx Speech  
Enabled Auto Attendant  
June 2010, 10-4940-00133

®,™ Trademark of Mitel Networks Corporation  
© Copyright 2010, Mitel Networks Corporation  
All rights reserved

---

<b>OVERVIEW .....</b>	<b>1</b>
Interop History .....	1
Interop Status .....	1
Software & Hardware Setup .....	1
Tested Features.....	2
Device Limitations and Known Issues .....	3
Network Topology.....	5
<b>CONFIGURATION NOTES .....</b>	<b>6</b>
MCD 4.1 SP1 Configuration Notes .....	6
Network Requirements.....	6
Assumptions for the Mitel MCD 4.1 SP1 Programming .....	6
Licensing and Option Selection – SIP Licensing .....	7
Class of Service Options .....	8
Network Element Assignment .....	9
Trunk Attributes .....	10
SIP Peer Profile .....	11
ARS Digit Modification Plans .....	13
ARS Routes.....	14
ARS Digits Dialed.....	15
<b>LYRIX SEAA CONFIGURATION NOTES.....</b>	<b>16</b>



## Overview


This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel MCD 4.1 SP1 to connect to the Lyrix SEAA. 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 15, 2010	Initial Interop with Mitel 3300 MCD 4.1 SP1 and the Lyrix SEAA 6.1

## Interop Status

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

	<p>The most common certification which means the Lyrix SEAA 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>
--	---





## Software & Hardware Setup

This was the test setup to generate a basic SIP call between the Lyrix SEAA and the Mitel MCD 4.1 SP1.

Manufacturer	Variant	Software Version
Mitel	3300 MCD – Mxe Platform	10.1.1.11_2
Lyrix	SEAA	6.1
Cisco	CallManager	6.x

## 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 Lyrix SEAA, 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.	Not Tested
Packetization	Forcing the Mitel MCD 4.1 SP1 to stream RTP packets through its E2T card at different intervals, from 10ms to 90ms	
Personal Ring Groups	Receiving calls through the Lyrix SEAA to a personal ring group. Also moving calls to/from the prime member and group members.	
Mobile Extension	Receiving a call through the Lyrix SEAA to Mobile extensions and TUI interface. Also moving calls to/from Desktop and Twinned devices.	Not Tested
Teleworker	Making and receiving a call through the Lyrix SEAA to and from Teleworker extensions.	Not Tested
Video	Making and receiving a call through the Lyrix SEAA with video capable devices.	Not Supported
Fax	T.38 and G711Fax Calls	Not Supported

 - No issues found     - Issues found, cannot recommend to use     - Issues found

## Device Limitations and Known Issues

This is a list of problems or not supported features when the Lyrix SEAA is connected to the Mitel MCD 4.1 SP1.

Feature	Problem Description
G.729	<p>The Lyrix SEAA doesn't support G.729 audio codec. Typically, this will not affect the interoperability between the 3300 MCD and the SEAA but could have issues with VOIP service providers using only G729 codec for trunk calls streaming audio to the Lyrix SEAA.</p> <p><b>Recommendation:</b> Use G.711 audio codec for calls directed to the Lyrix SEAA</p>
Inband DTMF	<p>The Lyrix SEAA doesn't support inband DTMF. Typically, this will not affect the interoperability between the 3300 MCD and the SEAA but could have issues with VOIP service providers that do not support RFC2833 on trunk calls.</p> <p><b>Recommendation:</b> RFC2833 (i.e. out-of-band) DTMF must be used for calls directed to the Lyrix SEAA.</p>
Packetization	<p>The Lyrix SEAA only supports packetization sizes of 10, 20 or 30 ms.</p> <p><b>Recommendation:</b> It is recommended that the default packetization size of 20ms be used. This will not affect the interoperability of the 3300 MCD and the SEAA.</p>
Video	<p>The Lyrix SEAA does not support Video calls.</p> <p><b>Recommendation:</b> Follow the SIP Peer Profile provided below. This will not affect the interoperability of the 3300 MCD and the SEAA.</p>
PRACK	<p>The Lyrix SEAA does not support PRACK.</p> <p><b>Recommendation:</b> Follow the SIP Peer Profile provided below. This will not affect the interoperability of the 3300 MCD and the SEAA.</p>
Session Timers	<p>The Lyrix SEAA does not support Session Timers.</p> <p><b>Recommendation:</b> Follow the SIP Peer Profile provided below. This will not affect the interoperability of the 3300 MCD and the SEAA.</p>
Busy/Out-Of-Service	<p>When the Lyrix SEAA tries to reach a device that is busy or out-of-service, the device making the call will hear a tone and then get disconnected. No busy tones will be heard.</p> <p><b>Recommendation:</b> Lyrix is aware of this issue and is working to correct the issue. Contact Mitel Support for further details and refer to Third Party DPAR MN00341923.</p>
Mobile Extension, Teleworker and Nupoint	<p>These Mitel applications were not tested during the course of this interop and no assertions can be made about their functionality with the Lyrix SEAA. That being said, there are no indications</p>

	that Lyrix SEAA would have a problem functioning in a network environment with these applications.
--	--



## Network Topology

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

### SIPCoE Standard SIP Trunk Interop Network Configuration

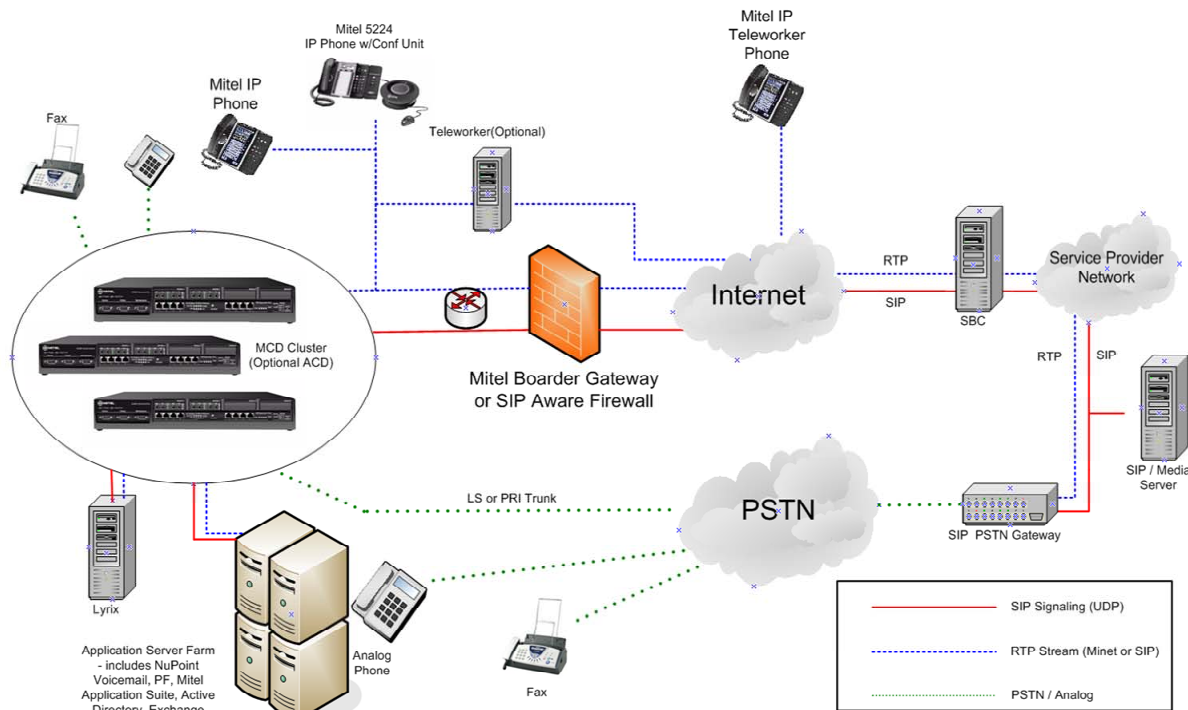


Figure 1 – Network Topology

## Configuration Notes

This section is a description of how the SIP Interop was configured. These notes should give a guideline how a device can be configured in a customer environment and how the Lyrix SEAA MCD 4.1 SP1 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 4.1 SP1 Configuration Notes

The following steps show how to program the Mitel MCD 4.1 SP1 to interconnect with the Lyrix SEAA.

#### 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 for G729. Almost all Enterprise LAN networks can support this level of traffic without any special engineering. Please refer to the MCD 4.1 SP1 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 Mitel MCD 4.1 SP1 Programming

- The SIP signaling connection uses UDP on Port 5060.

## Licensing and Option Selection – SIP Licensing

Ensure that the MCD 4.1 SP1 is equipped with enough SIP trunking licenses for the connection to the Lyrix SEAA. 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 4.1 SP1 to be used with all service providers, applications and SIP trunking devices.

### Online Licensing with the Application Management Center

Application Record ID

#### Purchased Options

##### Users

IP User Licenses	2300
External Hot Desk User Licenses	100
ACD Active Agent Licenses	100
HTML Apps Infrastructure Licenses	100
Analog Line Licenses	10

##### Voicemail

Mailbox Licenses	100
Voice Mail Networking	Yes
Advanced Voice Mail	Yes
Voice Mail Hospitality/PMS	Yes

##### Trunking/Networking

Digital Link Licenses	16
Compression Licenses	16
FAX Over IP (T.38) Licenses	16
SIP Trunk Licenses	1000
XNET Networking	Yes
IP Networking	Yes

##### Others

Tenanting	Yes
MLPP	No
Remote Management	Yes
Hardware Identifier	00000003BB45
Password	*****

#### Configuration Options

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

**Figure 2 – License and Option Selection**

## Class of Service Options

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 Mitel MCD 4.1 SP1.

- 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 Lyrix SEAA web interface. At the top, there is a navigation bar with the Mitel logo, group name 'System Defaulted', alarm status 'Major', and links for 'Message Board', 'About', 'Help', and 'Logout'. Below this, the page title is 'Class of Service Options on Sipint1'. There is a search bar with a dropdown menu set to 'Class Of Service Number' and a 'Search' button. Below the search bar, there are buttons for 'Change', 'Copy', 'Print...', 'Import...', 'Export...', and 'Data Refresh'. A table titled 'Class of Service Options' is displayed with the following data:

Class Of Service Number	Comment
1	
2	IP Sets
3	NPM VM Ports
4	NPM MWI
5	IP Sets DND

Figure 3 – Class of Service

## Network Element Assignment

Create a network element for the Lyrix SEAA. In this example, the softswitch is reachable by an IP Address and is defined as “Lyrix” in the network element assignment form. **The FQDN or IP addresses of the Lyrix SEAA should be provided by your network administrator.**

Set the transport to UDP and port to 5060.

Network Elements	
Name	Lyrix
Type	Other
FQDN or IP Address	8.20.177.246
Local	False
Version	
Zone	1
SIP Peer	<input checked="" type="checkbox"/>
<b>SIP Peer Specific</b>	
SIP Peer Transport	UDP
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
<input type="button" value="Save"/> <input type="button" value="Cancel"/>	

Figure 4 – Network Element Assignment

## Trunk Attributes

This is configured in the Trunk Attributes form. In this example the Trunk Attributes is defined for Trunk Service Number 20 which will be used to direct incoming calls to an answer point in the Mitel MCD 4.1 SP1.

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 Mitel MCD 4.1 SP1 will not absorb any DID number from the Lyrix SEAA leaving all digits for the MCD 4.1 SP1 to translate. For example, the Lyrix SEAA delivers 4009 through the SIP trunk to the MCD 4.1 SP1. The MCD 4.1 SP1 will ring extension 4009. Extension 4009 must be programmed as a valid dialable number in the MCD 4.1 SP1. Please refer to the Mitel MCD 4.1 SP1 System Administration documentation for further programming information.

Trunk Attributes	
Trunk Service Number	20
Release Link Trunk	No <input type="button" value="v"/>
Call Recognition Service	Off <input type="button" value="v"/>
Class of Service	5
Class of Restriction	1
Baud Rate	300 <input type="button" value="v"/>
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	0
Dial In Trunks Incoming Digit Modification - Insert	
Trunk Label	Lyrix

Figure 5 – 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 MCD 4.1 SP1 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 "Lyrix" Network Element.

**Address Type:** Select IP address.

**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 the Lyrix SEAA.

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

SIP Peer Profile	
SIP Peer Profile Label	Lyrix
Network Element	Lyrix
<b>Local Account Information</b>	
Registration User Name	
Address Type	<input type="radio"/> FQDN: siplab3.sipint.com <input checked="" type="radio"/> IP Address: 66.46.196.195
<b>Call Routing and Administration Options</b>	
Interconnect Restriction	1
Maximum Simultaneous Calls	10
Outbound Proxy Server	
SMDR Tag	0
Trunk Service	20
Zone	1
Alternate Destination Domain Enabled	<input checked="" type="radio"/> No <input type="radio"/> Yes
Alternate Destination Domain FQDN or IP Address	
Enable Special Re-invite Collision Handling	<input checked="" type="radio"/> No <input type="radio"/> Yes
Private SIP Trunk	<input checked="" type="radio"/> No <input type="radio"/> Yes
Route Call Using To Header	<input checked="" type="radio"/> No <input type="radio"/> Yes
<b>Calling Line ID Options</b>	
Default CPN	
CPN Restriction	<input checked="" type="radio"/> No <input type="radio"/> Yes
Public Calling Party Number Passthrough	<input checked="" type="radio"/> No <input type="radio"/> Yes
Use Diverting Party Number as Calling Party Number	<input checked="" type="radio"/> No <input type="radio"/> Yes
<b>Authentication Options</b>	
User Name	
Password	
Confirm Password	
Authentication Option for Incoming Calls	No Authentication
<b>SDP Options</b>	
Allow Peer To Use Multiple Active M-Lines	<input checked="" type="radio"/> No <input type="radio"/> Yes
Allow Using UPDATE For Early Media Renegotiation	<input type="radio"/> No <input checked="" type="radio"/> Yes
Avoid Signaling Hold to the Peer	<input checked="" type="radio"/> No <input type="radio"/> Yes
Enable Mitel Proprietary SDP	<input checked="" type="radio"/> No <input type="radio"/> Yes
Force sending SDP in initial Invite message	<input checked="" type="radio"/> No <input type="radio"/> Yes
Force sending SDP in initial Invite - Early Answer	<input checked="" type="radio"/> No <input type="radio"/> Yes
Limit to one Offer/Answer per INVITE	<input checked="" type="radio"/> No <input type="radio"/> Yes
NAT Keepalive	<input type="radio"/> No <input checked="" type="radio"/> Yes
Prevent the Use of IP Address 0.0.0.0 in SDP Messages	<input type="radio"/> No <input checked="" type="radio"/> Yes
Renegotiate SDP To Enforce Symmetric Codec	<input checked="" type="radio"/> No <input type="radio"/> Yes
Repeat SDP Answer If Duplicate Offer Is Received	<input checked="" type="radio"/> No <input type="radio"/> Yes
RTP Packetization Rate Override	<input checked="" type="radio"/> No <input type="radio"/> Yes
RTP Packetization Rate	20ms
Special handling of Offers in 2XX responses (INVITE)	<input checked="" type="radio"/> No <input type="radio"/> Yes
Suppress Use of SDP Inactive Media Streams	<input checked="" type="radio"/> No <input type="radio"/> Yes
<b>Signaling and Header Manipulation Options</b>	
Session Timer	0
Allow Display Update	<input type="radio"/> No <input checked="" type="radio"/> Yes
Build Contact Using Request URI Address	<input checked="" type="radio"/> No <input type="radio"/> Yes
Disable Reliable Provisional Responses	<input type="radio"/> No <input checked="" type="radio"/> Yes
Enable sending '+' for E.164 numbers	<input checked="" type="radio"/> No <input type="radio"/> Yes
Force Max-Forward: 70 on Outgoing Calls	<input checked="" type="radio"/> No <input type="radio"/> Yes
Ignore Incoming Loose Routing Indication	<input checked="" type="radio"/> No <input type="radio"/> Yes
Use P-Asserted Identity Header	<input type="radio"/> No <input checked="" type="radio"/> Yes
Use P-Preferred Identity Header	<input checked="" type="radio"/> No <input type="radio"/> Yes
Use Restricted Character Set For Authentication	<input checked="" type="radio"/> No <input type="radio"/> Yes
Use To Address in From Header on Outgoing Calls	<input checked="" type="radio"/> No <input type="radio"/> Yes

Figure 6 – SIP Peer Profile Assignment



### ARS Digit Modification Plans

Ensure that Digit Modification for outgoing calls on the SIP trunk to the Lyrix SEAA absorbs or inject additional digits according to your dialling plan. In this example, we will be absorbing 0 digits.

ARS Digit Modification Plans	
Digit Modification Number	5
Number of Digits to Absorb	<input type="text" value="0"/>
Digits to be Inserted	<input type="text"/>
Final Tone Plan/Information Marker	<input type="text"/>

Figure 7 – Digit Modification Assignment

### ARS Routes

Create a route for SIP Trunks connecting a trunk to the Lyrix SEAA. In this example, the SIP trunk is assigned to Route Number 20. Choose SIP Trunk as a routing medium and choose the SIP Peer Profile and Digit Modification entry created earlier.

ARS Routes	
Route Number	20
Routing Medium	SIP Trunk
Trunk Group Number	
SIP Peer Profile	Lyrix
COR Group Number	1
Digit Modification Number	5
Digits Before Outpulsing	
Route Type	
Compression	Off

Figure 8 – SIP Trunk Route Assignment

### ARS Digits Dialed

ARS initiates the routing of trunk calls when certain digits are dialed from a station. In this example, when a user dials 9700, the call will be routed to the Lyrix SEAA (ie. Route 20).

**Change Range Programming - ARS Digits Dialed** Help

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
9700	0	Route	20

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 style="width: 100px;" type="text" value="9700"/>	<input style="width: 40px;" type="text" value=""/>
Number of Digits to Follow	Change to <input type="button" value="v"/>	<input style="width: 50px;" type="text" value="0"/> <input type="button" value="v"/>	-
Termination Type	Change to <input type="button" value="v"/>	<input style="width: 50px;" type="text" value="Route"/> <input type="button" value="v"/>	-
Termination Number	Change to <input type="button" value="v"/>	<input style="width: 100px;" type="text" value="20"/>	<input style="width: 40px;" type="text" value=""/>

**Figure 9 – ARS Digit Dialed Assignment**

## Lyrix SEAA Configuration Notes

Consult the Lyrix SEAA documentation for installation and configuration.



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

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

THIS DOCUMENT IS PROVIDED TO YOU FOR INFORMATIONAL PURPOSES ONLY. The information furnished in this document, believed by Mitel to be accurate as of the date of its publication, is subject to change without notice. Mitel assumes no responsibility for any errors or omissions in this document and shall have no obligation to you as a result of having made this document available to you or based upon the information it contains.

M MITEL (design) is a registered trademark of Mitel Networks Corporation. All other products and services are the registered trademarks of their respective holders.

© Copyright 2008, Mitel Networks Corporation. All Rights Reserved.

[www.mitel.com](http://www.mitel.com)

