MITEL – SIP CoE

Technical

Configuration Notes

Configure MCD 4.1 SP1 for use with the Mobiso 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 Lyrix Speech Enabled Auto Attendant
June 2010, 10-4940-00133

OVERVIEW 1

Interop History 1

Interop Status 1

Software & Hardware Setup 1

Tested Features 2

Device Limitations and Known Issues 3

Network Topology 5

CONFIGURATION NOTES 6

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

Mobiso SEAA CONFIGURATION NOTES 16

Overview

This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel MCD 4.1 SP1 to connect to the Mobiso SEAA. The different devices can be configuredin 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 Mobiso SEAA 6.1

Interop Status

The Interop of the Mobiso SEAA has been given a Certification status. This service provider ortrunking device will be included in the SIP CoE Reference Guide. The status the Votacall SEAAachieved is:

COMPATIBLE	The most common certification which means the Mobiso SEAA has beentested and/or validated by the Mitel SIP CoE team. Product support willprovide all necessary support related to the interop, but issues uniqueor specific to the 3rd party will be referred to the 3rd party asappropriate.

Software & Hardware Setup

This was the test setup to generate a basic SIP call between the Votacall SEAA and the MitelMCD 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 Pans (08-4940-00034) for detailed test cases.

Feature	Feature Description	Issues
Basic Call	Making and receiving a call through the Mobiso SEAA, callholding, transferring, conferencing, busy calls, long callsdurations, 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 Mobiso SEAA to a personal ringgroup. Also moving calls to/from the prime member andgroup members.	☑
Mobile Extension	Receiving a call through the Mobiso 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 Mobiso SEAA to andfrom Teleworker extensions.	Not Tested
Video	Making and receiving a call through the Mobiso SEAA withvideo 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 Mobiso SEAA is connected to theMitel MCD 4.1 SP1.

Feature	Problem Description
G.729	The Mobiso SEAA doesn't support G.729 audio codec. Typically, this will not affect the interopability 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. **Recommendation:** Use G.711 audio codec for calls directed to the Lyrix SEAA
Inband DTMF	The Mobiso SEAA doesn't support inband DTMF. Typically, thiswill not affect the interopability between the 3300 MCD and theSEAA but could have issues with VOIP service providers that donot support RFC2833 on trunk calls. **Recommendation:** RFC2833 (i.e. out-of-band) DTMF must be used for calls directed to the Mobiso SEAA.
Packetization	The Mobiso SEAA only supports packetization sizes of 10, 20 or 30ms. **Recommendation:** It is recommended that the default packetization size of 20ms be used. This will not affect the interopability of the 3300 MCD and the SEAA.
Video	The Mobiso SEAA does not support Video calls. **Recommendation:** Follow the SIP Peer Profile provided below. This will not affect the interopability of the 3300 MCD and the SEAA.
PRACK	The Mobiso SEAA does not support PRACK. **Recommendation:** Follow the SIP Peer Profile provided below. This will not affect the interopability of the 3300 MCD and the SEAA.
Session Timers	The Mobiso SEAA does not support Session Timers. **Recommendation:** Follow the SIP Peer Profile provided below. This will not affect the interopability of the 3300 MCD and the SEAA.
Busy/Out-Of-Service	When the Mobiso SEAA tries to reach a device that is busy or out-of-service, the device making the call will hear a tone and thenget disconnected. No busy tones will be heard. **Recommendation:** Mobiso is aware of this issue and is working tocorrect the issue. Contact Mitel Support for further details andrefer to Third Party DPAR MN00341923.
Mobile Extension, Teleworker and Nupoint	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

	that Mobiso SEAA would have a problem functioning in a networkenvironment with these applications.

Network Topology

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

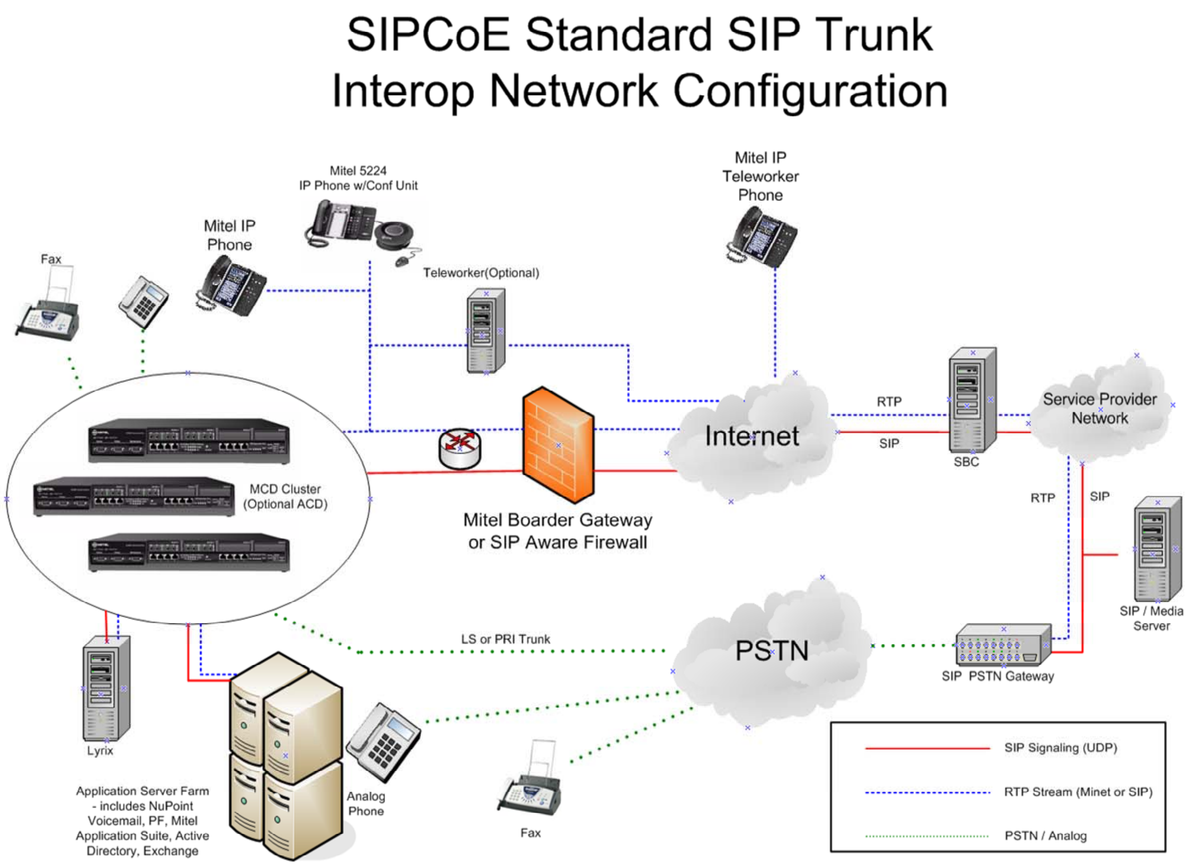


Figure 1 – Network Topology

Configuration Notes

This section is a description of how the SIP Interop was configured. These notes should give a guideline how a device can be configured in a customer environment and how the Mobiso 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 Mobiso 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 Mobiso SEAA. This can be verified within the License and Option Selectionform.

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**

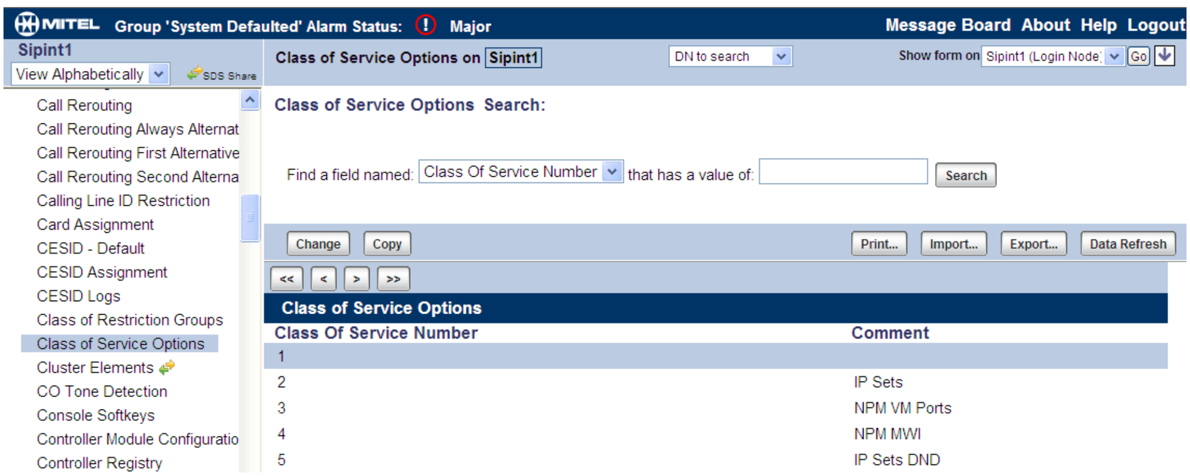


Figure 3 – Class of Service

Network Element Assignment

Create a network element for the Mobiso SEAA. In this example, the softswitch is reachable byan IP Address and is defined as Mobiso in the network element assignment form. **The FQDNor IP addresses of the Mobiso SEAA should be provided by your network administrator.**

Set the transport to UDP and port to 5060.

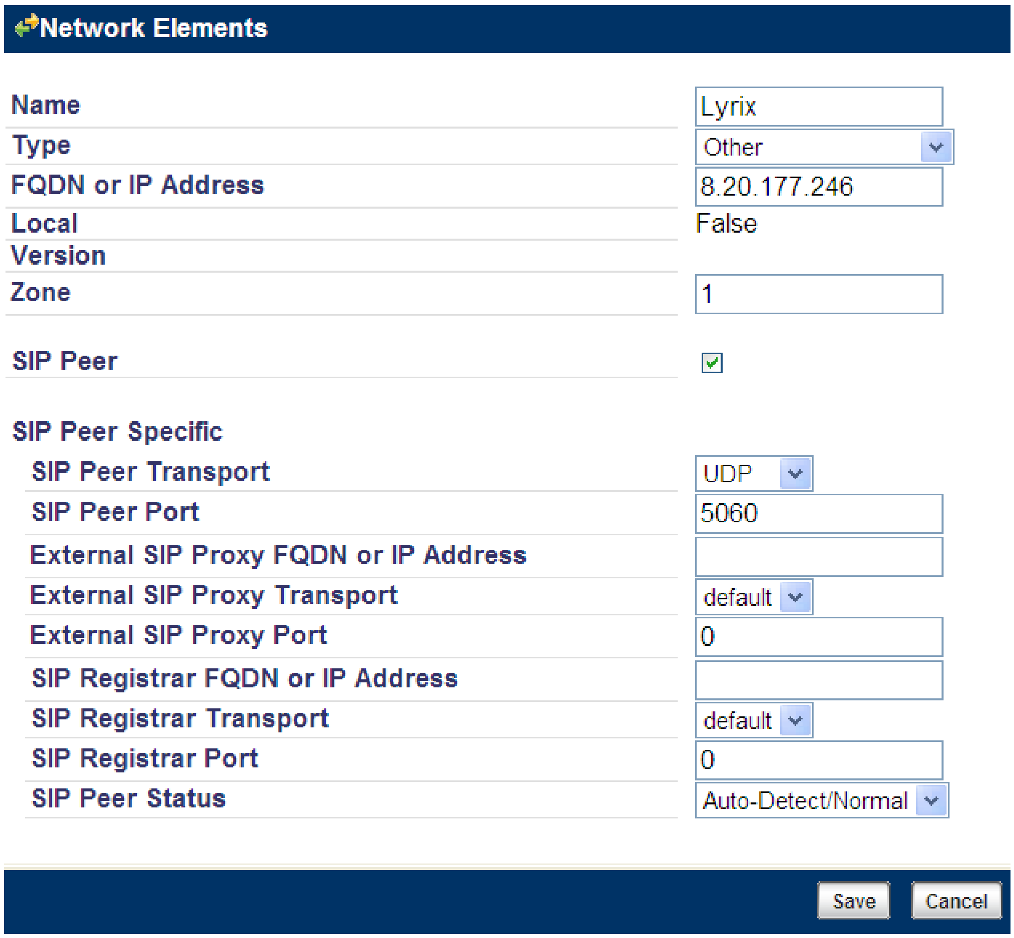


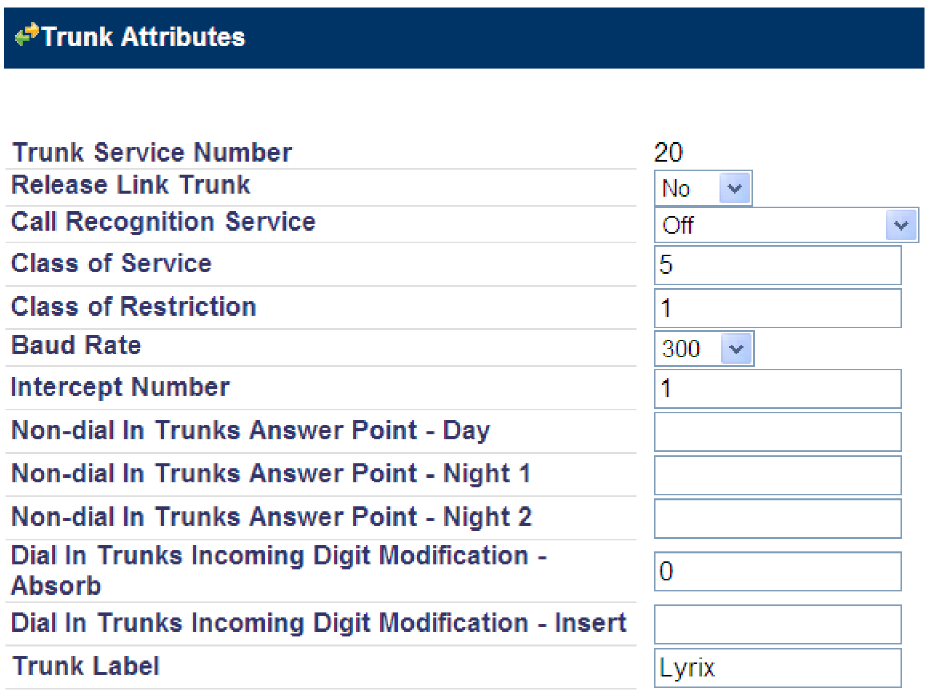
Figure 4 – Network Element Assignment

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
Call Recognition Service	Off
Class of Service	5
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	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 Mobiso 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

Local Account Information

Registration User Name	
Address Type	○ FQDN: siplab3.sipint.com ◉ IP Address: 66.46.196.195

Call Routing and Administration Options

Interconnect Restriction	1
Maximum Simultaneous Calls	10
Outbound Proxy Server	
SMDR Tag	0
Trunk Service	20
Zone	1
Alternate Destination Domain Enabled	◉ No ○ Yes
Alternate Destination Domain FQDN or IP Address	
Enable Special Re-invite Collision Handling	◉ No ○ Yes
Private SIP Trunk	◉ No ○ Yes
Route Call Using To Header	◉ No ○ Yes

Calling Line ID Options

Default CPN	
CPN Restriction	◉ No ○ Yes
Public Calling Party Number Passthrough	◉ No ○ Yes
Use Diverting Party Number as Calling Party Number	◉ No ○ Yes

Authentication Options

User Name	
Password	
Confirm Password	
Authentication Option for Incoming Calls	No Authentication

SDP Options

Allow Peer To Use Multiple Active M-Lines	◉ No ○ Yes
Allow Using UPDATE For Early Media Renegotiation	○ No ◉ Yes
Avoid Signaling Hold to the Peer	◉ No ○ Yes
Enable Mitel Proprietary SDP	◉ No ○ Yes
Force sending SDP in initial Invite message	◉ No ○ Yes
Force sending SDP in initial Invite - Early Answer	◉ No ○ Yes
Limit to one Offer/Answer per INVITE	◉ No ○ Yes
NAT Keepalive	○ No ◉ Yes
Prevent the Use of IP Address 0.0.0.0 in SDP Messages	○ No ◉ Yes
Renegotiate SDP To Enforce Symmetric Codec	◉ No ○ Yes
Repeat SDP Answer If Duplicate Offer Is Received	◉ No ○ Yes
RTP Packetization Rate Override	◉ No ○ Yes
RTP Packetization Rate	20ms
Special handling of Offers in 2XX responses (INVITE)	◉ No ○ Yes
Suppress Use of SDP Inactive Media Streams	◉ No ○ Yes

Signaling and Header Manipulation Options

Session Timer	0
Allow Display Update	○ No ◉ Yes
Build Contact Using Request URI Address	◉ No ○ Yes
Disable Reliable Provisional Responses	○ No ◉ Yes
Enable sending '+' for E.164 numbers	◉ No ○ Yes
Force Max-Forward: 70 on Outgoing Calls	◉ No ○ Yes
Ignore Incoming Loose Routing Indication	◉ No ○ Yes
Use P-Asserted Identity Header	○ No ◉ Yes
Use P-Preferred Identity Header	◉ No ○ Yes
Use Restricted Character Set For Authentication	◉ No ○ Yes
Use To Address in From Header on Outgoing Calls	◉ No ○ Yes

Figure 6 – SIP Peer Profile Assignment

ARS Digit Modification Plans

Ensure that Digit Modification for outgoing calls on the SIP trunk to the Mobiso SEAA absorbsor inject additional digits according to your dialling plan. In this example, we will be absorbing0 digits.

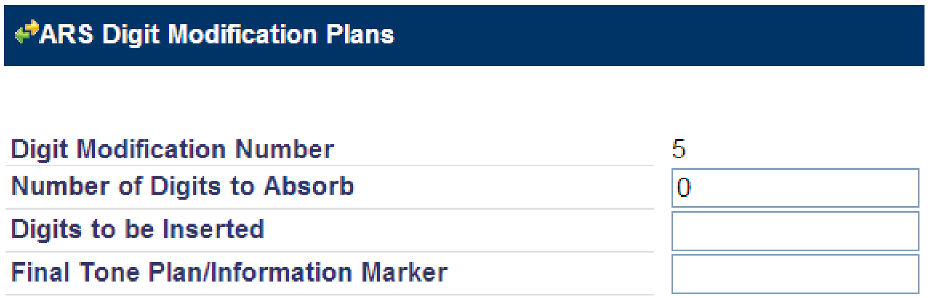


Figure 7 – Digit Modification Assignment

ARS Routes

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

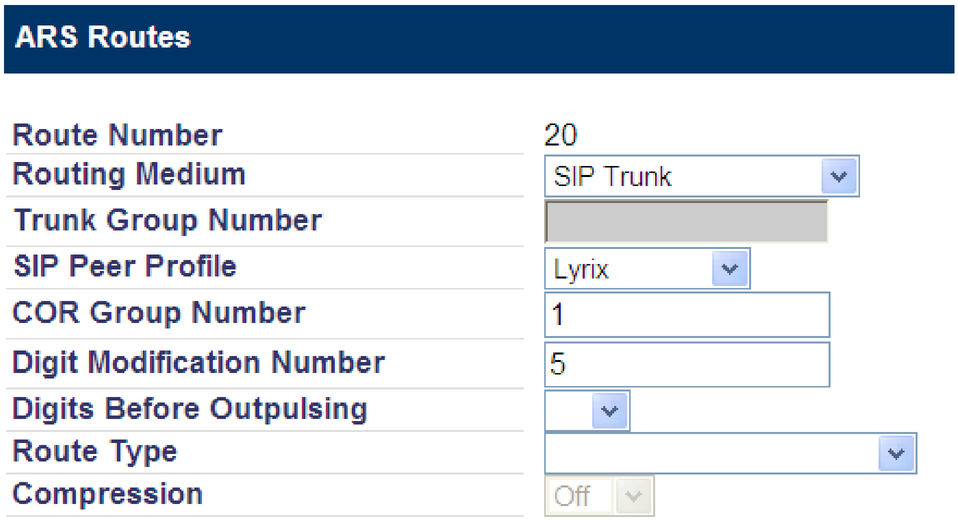
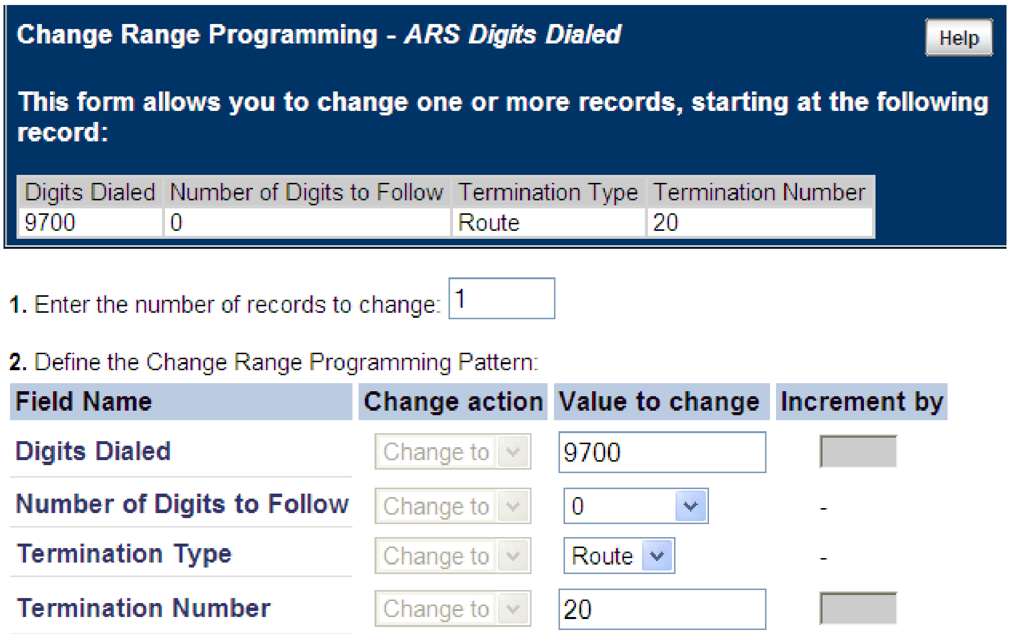


Figure 8 – SIP Trunk Route Assignment

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 Mobiso 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: 1

2. Define the Change Range Programming Pattern:

Field Name	Change action	Value to change	Increment by
Digits Dialed	Change to	9700	
Number of Digits to Follow	Change to	0	-
Termination Type	Change to	Route	-
Termination Number	Change to	20	

Figure 9 – ARS Digit Dialed Assignment

Lyrix SEAA Configuration Notes

Consult the Mobiso SEAA documentation for installation and configuration.

www.mitel.com

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