Mitel Technical Configuration Notes – HO3868

August 13, 2020

Configure MiVoice Business 9.1 for use with Centurion CARES IVR

Description: This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel MiVB to connect to Centurion CARES IVR

Environment: MiVoice Business 9.1 (9.1.0.92), Mitel 69xx MiNET 01.05.02.024, Mitel 53xx MiNET 06.05.00.24

NOTICE

TRADEMARKS

Mitel Technical Configuration Notes – Configure MiVoice Business for use with Centurion CARES IVR

Aug 2020 - HO3868

Table of Contents

Overview

This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel MiVB to connect to Centurion CARES IVR. 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	August 2020	MiVoice Business 9.1 with Centurion CARES IVR

Interop Status

The Interop of Centurion CARES IVR has been given a Certification status. Enghouse will be included in the Mitel Interoperability Reference Guide (IRG). The status Centurion CARES IVR achieved is:

COMPATIBLE	The most common certification which means Centurion CARES IVR has been tested and/or validated by the Mitel Third-Party Interop Team. Mitel Product Support will provide all necessary support related to the interop, but issues unique or specific to the 3rd party will be referred to the 3rd party as appropriate.

Software & Hardware Setup

This was the test setup to generate a basic SIP call between Centurion CARES IVR and the MiVB.

Note – Although this testing was performed on the below tested variants, the scope of this testing can be extended to other product variants that work with the same firmware. The list of components for which this testing can be considered applicable is given in the "Additional Applicable Variants" column of the following table –

Manufacturer	Tested Variants	Software Version	Additional Applicable Variants
Mitel	MiVoice Business	Release 9.1 (9.1.0.92)	Legacy flex GCP Flex – MPLS Variant
Mitel	69xx MiNET	01.05.02.024	N/A
Mitel	5330/5340 IP Sets	06.05.00.24	N/A
Centurion CARES	Centurion Cares IVR	14.4	

Tested Features

This is an overview of the features tested during the Interop test cycle and not a detailed view of the test cases. The features tested are applicable to two different topologies covered during the testing.

Note:
Both Normal and Call center IVR deployment has been tested

Feature	Feature Description	Issues
Basic Call	Making and receiving calls through Agent. Call in to the queue and Agent picks the call.	☑
Hold/Retrieve	Holding/Retrieving the current call. Making/receiving the second call	☑
Call Transfer	Unattended and Attended transfer. Completing and Canceling the transfer before the call is answered	☑
Conference	Agent 3-way Calling	☑
DTMF	Supports for Both Inband and RFC 2833	☑
Long Call	30 minutes Calls between PSTN user and Agent	☑
2B channel	2B Channel Transfer from IVR Server	☑
Bridge	Call Bridge from IVR Server	☑
End Call	Call End from CARES Soft Client	☑
Hold	Hold from CARES Soft Client	☑
Mute	Mute from CARES Soft Client	☑
Hold	Hold from PSTN	☑
Reconnect	Reconnect Line from CARES Soft Client	☑
Make a Call	Make a Call from CARES Soft Client	☑
Queue	Call in Queue and Answered by Agent once Ready	☑
Resiliency	Resilient IVR for making and receiving calls	⚠
TLS	Making and Receiving Calls through Secure Mode	⚠

☑ - No issues found ✕ - Issues found, cannot recommend using ⚠ - Issues found

Device Limitations and Known Issues

This is a list of problems or unsupported features when Centurion CARES IVR is connected to the MiVB.

Feature	Problem Description
Call Disconnect	Agent need to disconnect call manually in device **Recommendation: Please contact Centurion for more Details**
CTI Integration	There is no CTI Integration between Mitel and Centurion **Recommendation: Please contact Centurion for more Details**
Codec	Centurion only Support G711uLaw **Recommendation: Please contact Centurion for more Details**
PRACK	Centurion does not Support PRACK **Recommendation: Please contact Centurion for more Details**
TLS	Not Supported by Centurion **Recommendation: Please contact Centurion for more Details**

Network Topology

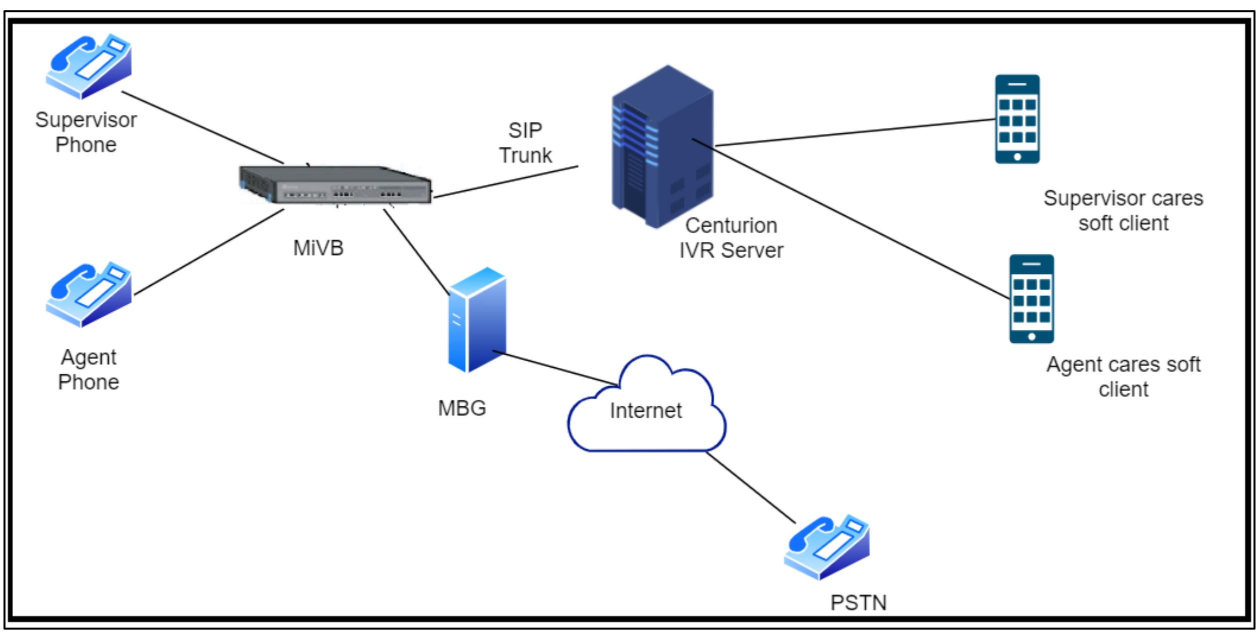


Figure 1 – Network Topology

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 Centurion CARES IVR connected with MiVB 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.*

MiVB Configuration Notes

The following steps show how to program a MiVB to interconnect with Centurion CARES IVR.

Configuration Template

A configuration template can be found in the same Mitel Knowledge Management System (KMS) 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 MiVB 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 MiVB 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 MiVB Programming

The SIP signaling connection uses UDP on Port 5060.

Licensing and Option Selection – SIP Licensing

Ensure that the MiVB is equipped with enough SIP Trunking licenses for the connection to Centurion CARES IVR. 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 MiVB to be used with all service providers, applications and SIP trunking devices.

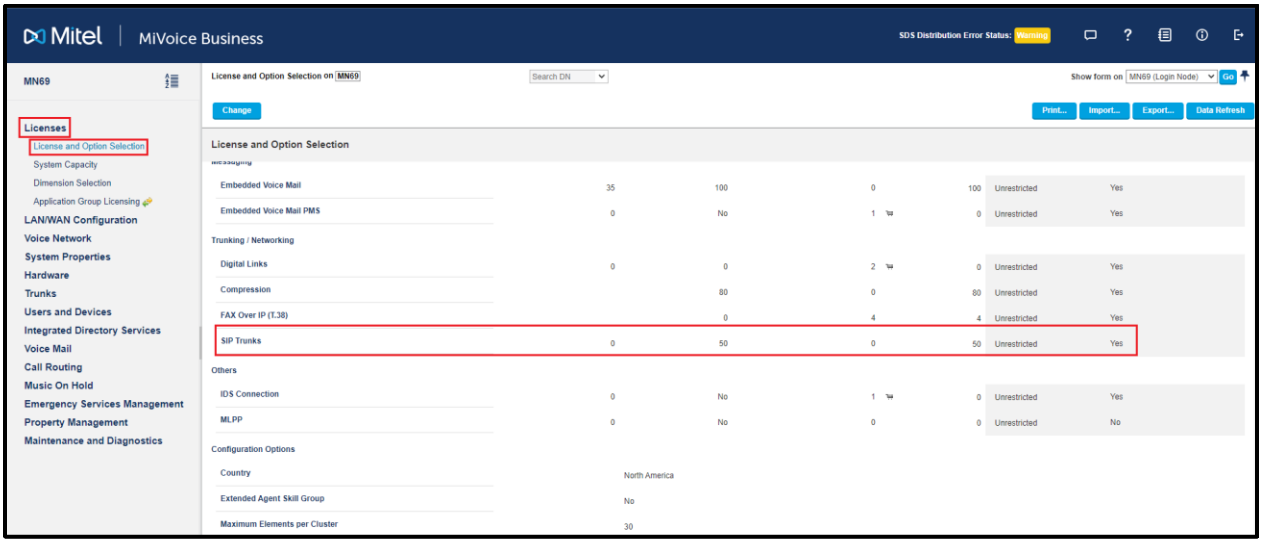


Figure 2 – License and Option Selection

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 MiVB.

- Public Network Access via DPNSS set to Yes
- Campon Tone Security/FAX Machine set to Yes
- Busy Override Security set to Yes
- Trunk Flash Allowed set to Yes
- Two B-Channel Transfer Allowed set to yes

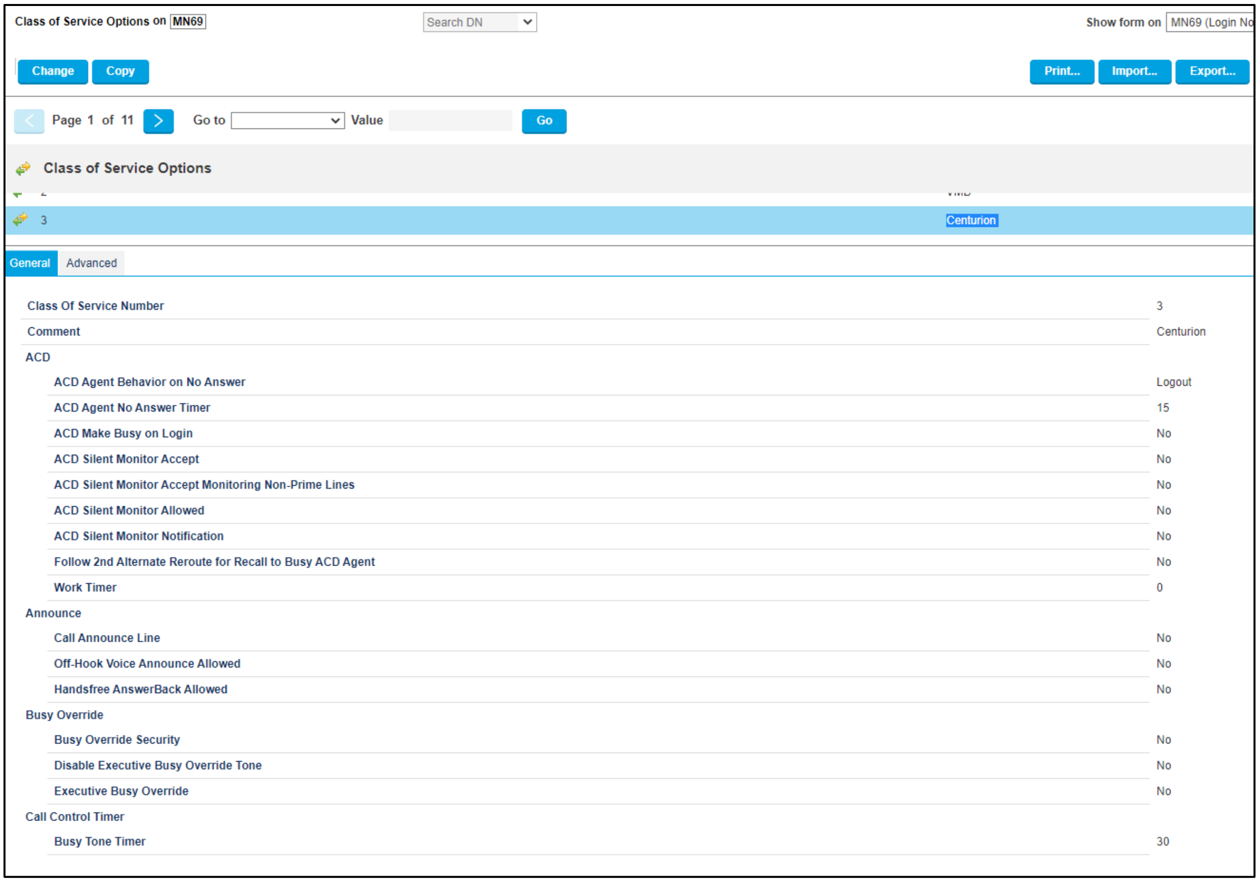


Figure 3 – Class of Service

Network Element Assignment

Create a network element for Centurion CARES IVR. In this example, the soft switch is reachable by an IP Address and is defined as "Enghouse" in the network element assignment form.

The network element is required to allow the MiVoice Business to connect to the CCE server using SIP trunks.

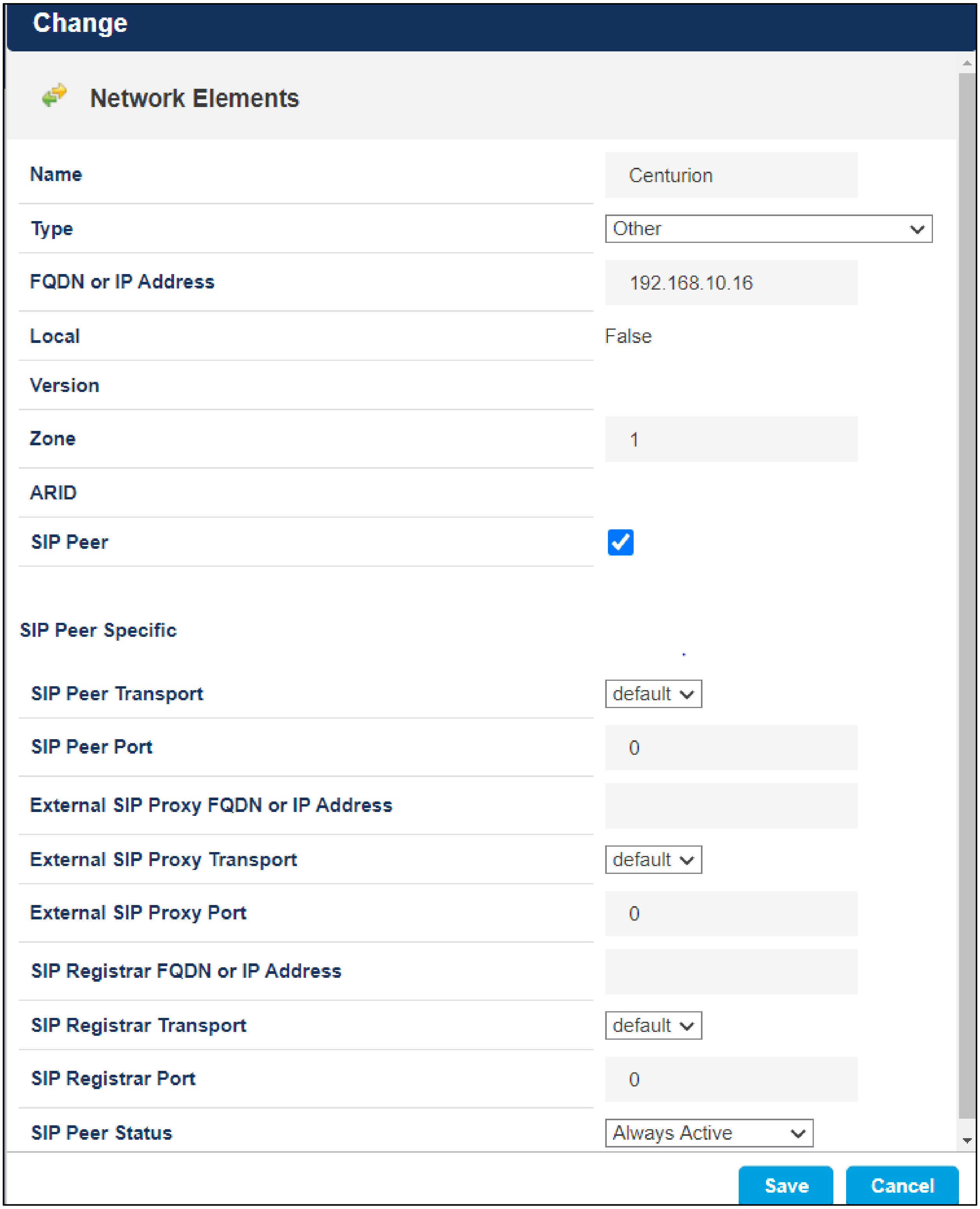
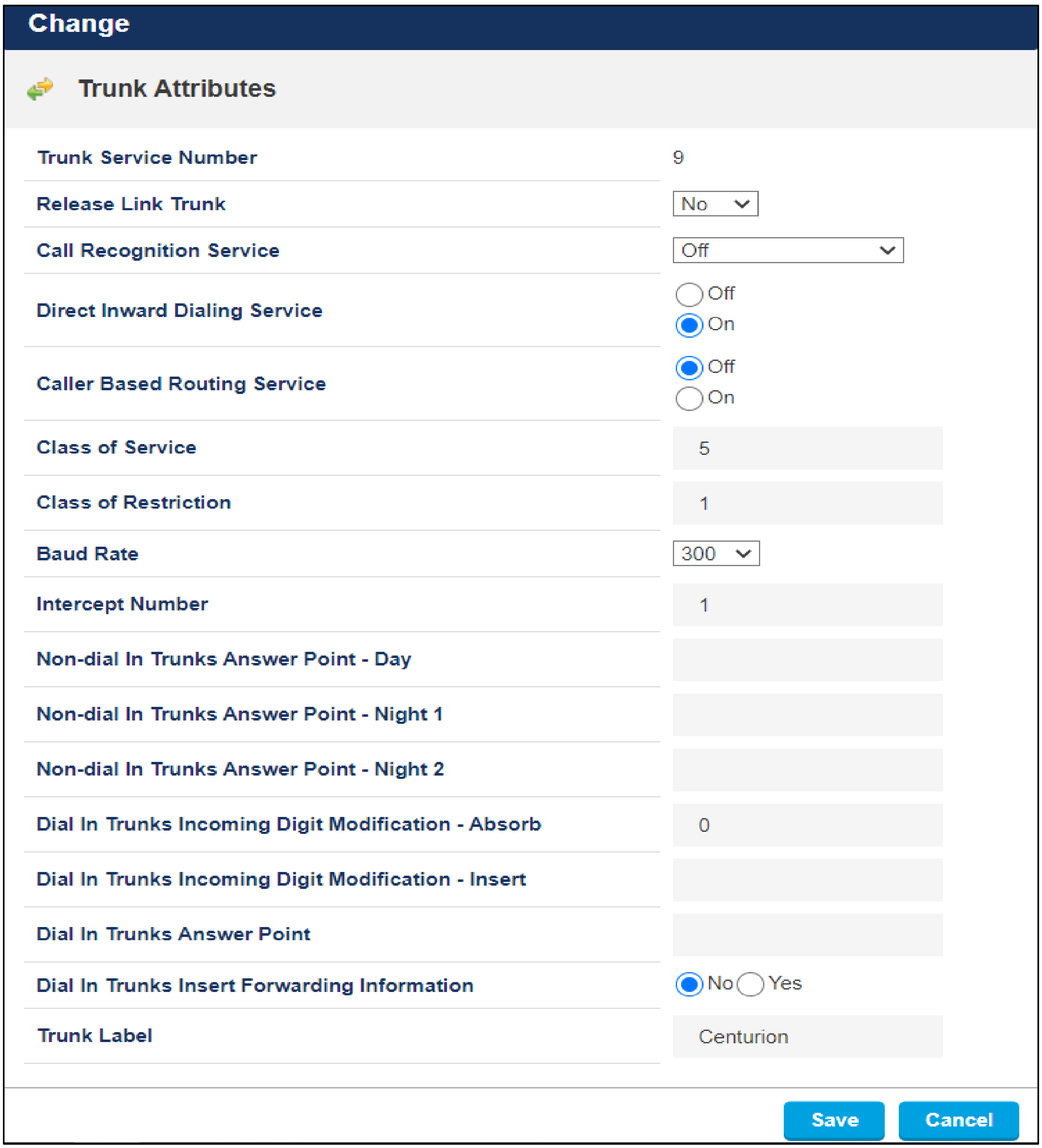


Figure 4 – Network Element Assignment

Trunk Attributes

Use Trunk Attributes form to configure Trunk Service Number. In this example, the Trunk Service Number 8 will be used to direct in this example. Please refer to the Mitel MiVB System Administration documentation for further programming information.



Change

Trunk Attributes

Trunk Service Number	9
Release Link Trunk	No
Call Recognition Service	Off
Direct Inward Dialing Service	Off / On (selected: On)
Caller Based Routing Service	Off / On (selected: 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	
Dial In Trunks Answer Point	
Dial In Trunks Insert Forwarding Information	No / Yes (selected: No)
Trunk Label	Centurion

Save Cancel

Figure 5 – 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 MiVoice Business Platform. SIP Peer profiles provision SIP trunks with local account information, outbound proxy server information, SIP trunk policies, Calling Line ID parameters, and authentication information.

The SIP Peer Profile should be configured with the following options.

- SIP Peer Profile Label – Enter a name for the SIP peer profile Enghouse
- Network Element – Centurion
- Address Type – Enter a site-specific value for the IP address - 192.168.10.69
- Interconnect Restriction – Enter a site-specific value - 1
- Maximum Simultaneous Calls – The maximum number of SIP trunks - 10
- Trunk Service – Select the applicable value - 9
- Enable Special Re-Invite Collision Handling – Yes
- Route Call Using to Header – Yes
- Enable Mitel Proprietary SDP – No
- Force sending SDP in initial Invite message – Yes
- Prevent the Use of IP Address 0.0.0.0 in SDP Messages – Yes
- Suppress Use of SDP Inactive Media Streams – Yes
- Disable Reliable Provisional Responses – Yes
- Ignore Incoming Loose Routing Indication – Yes
- Use P-Asserted Identity Header – Yes
- Require Reliable Provisional Responses on Outgoing Calls – No

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

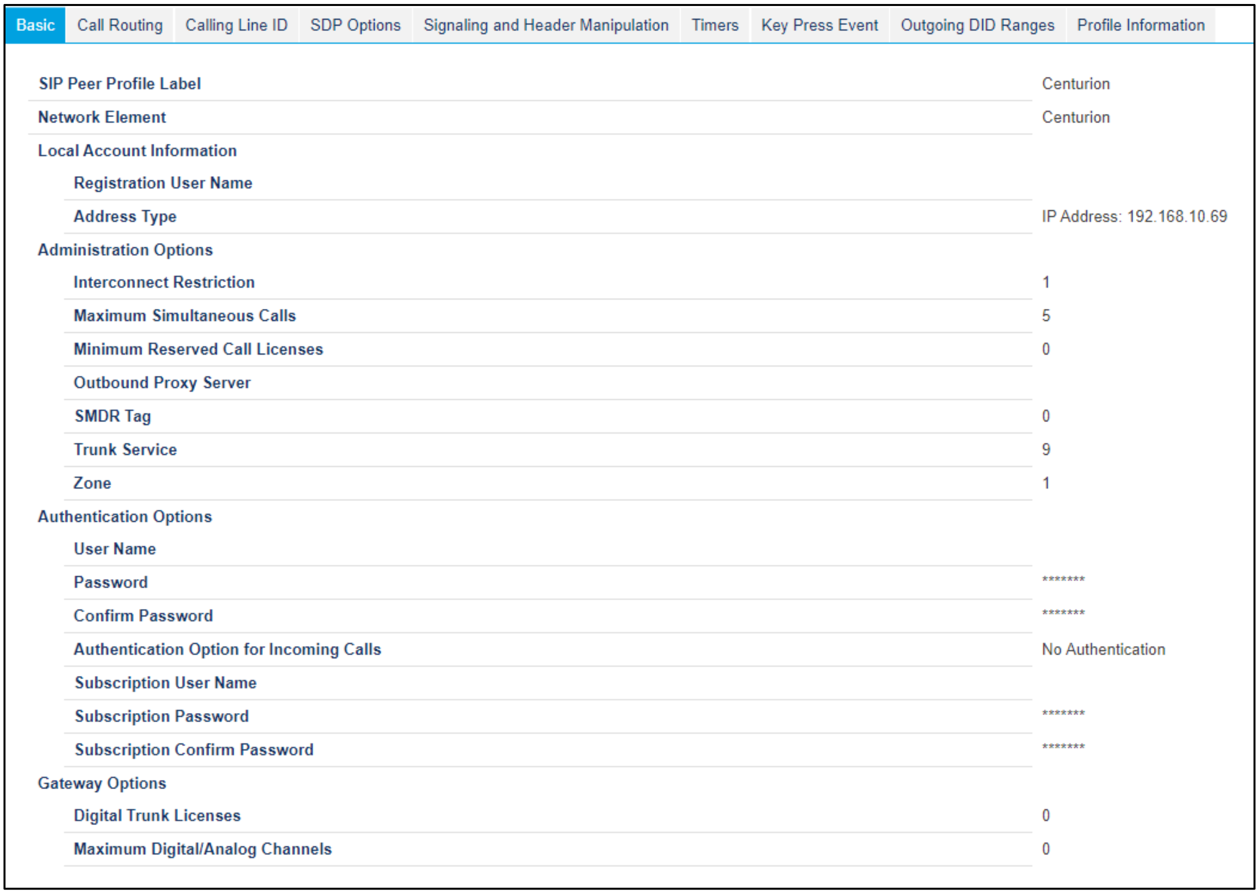


Figure 6 – SIP Peer Profile Assignment- Basic

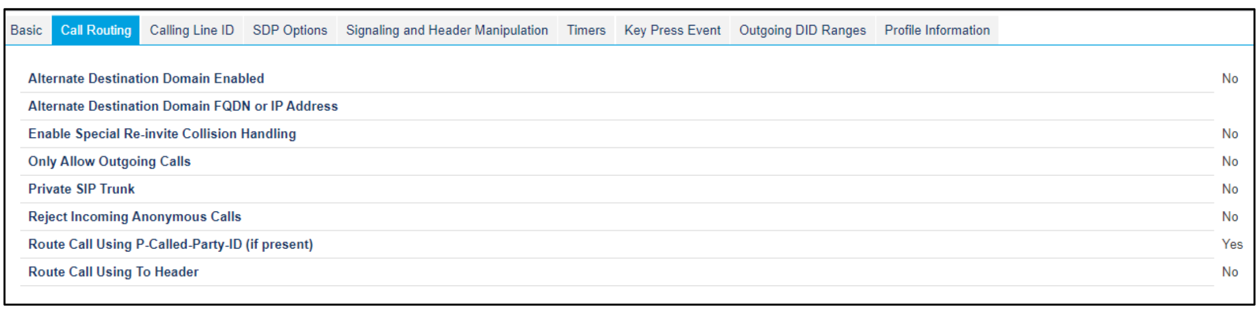


Figure 7 – SIP Peer Profile Assignment- Call Routing

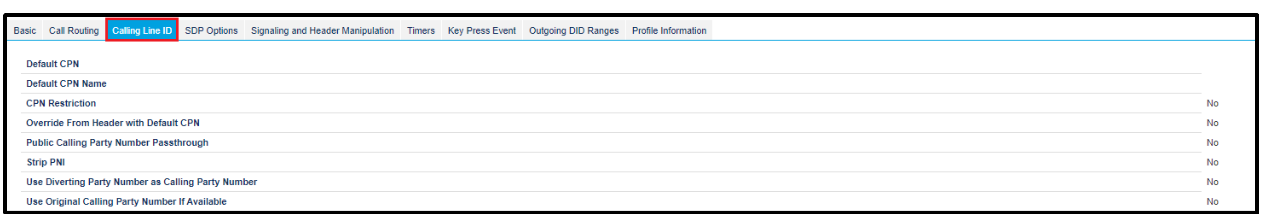


Figure 8 – SIP Peer Profile Assignment- Calling Line ID

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
AVP Only 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 Answers in Provisional Responses	No
IP Media Default	ipv4
Limit to one Offer/Answer per INVITE	Yes
NAT Keepalive	Yes
Prevent Codec Selection on Answer	No
Prevent the Use of IP Address 0.0.0.0 in SDP Messages	Yes
Reject Call without telephone-event payload	No
Renegotiate SDP To Enforce Symmetric Codec	No
Repeat SDP Answer If Duplicate Offer Is Received	No
Restrict Audio Codec	No Restriction
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	Yes

Figure 9 – SIP Peer Profile Assignment- SDP Options

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
Discard Received P-Asserted-Identity Headers	No
Domain for Trunk Context	
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
Include Diversion Header for EHDU	No
Mode for Out-of-Band DTMF	RFC 4733 DTMF
Multilingual Name Display	No
Only use SDP to decide 180 or 183	Yes

Figure 10 – SIP Peer Profile Assignment- SDP Options

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

Include Diversion Header for EHDU	No
Mode for Out-of-Band DTMF	RFC 4733 DTMF
Multilingual Name Display	No
Only use SDP to decide 180 or 183	Yes
Prefer From Header for Caller ID	No
Q.850 Reason Headers	No
Require Reliable Provisional Responses on Outgoing Calls	No
Signal Privacy (if enabled) on Emergency Calls	No
Suppress Incoming Name	No
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-Call-Leg-ID Header	No
Use P-Early-Media Header	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
Use user=phone for Diversion Header	No

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

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	90
Session Timer: Local as Refresher	No
Subscription Period	3600
Subscription Period Minimum	300
Subscription Period Refresh (%)	80
Invite Ringing Response Timer	0

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

Figure 13 – SIP Peer Profile Assignment- Key Press Event

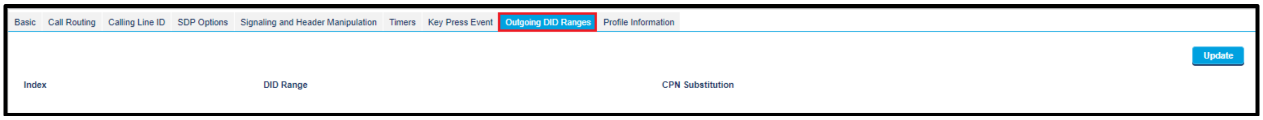


Figure 14 – SIP Peer Profile Assignment- Outgoing DID Ranges

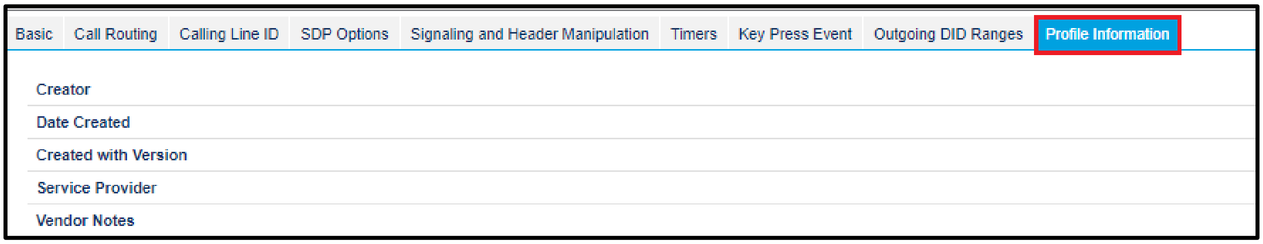


Figure 15 – SIP Peer Profile Assignment- Profile Information

ARS Digit Modification Plans

Ensure that Digit Modification for outgoing calls on the SIP trunk to Centurion CARES IVR absorbs or injects additional digits according to your dialling plan. In this example, we will be absorbing no digits.

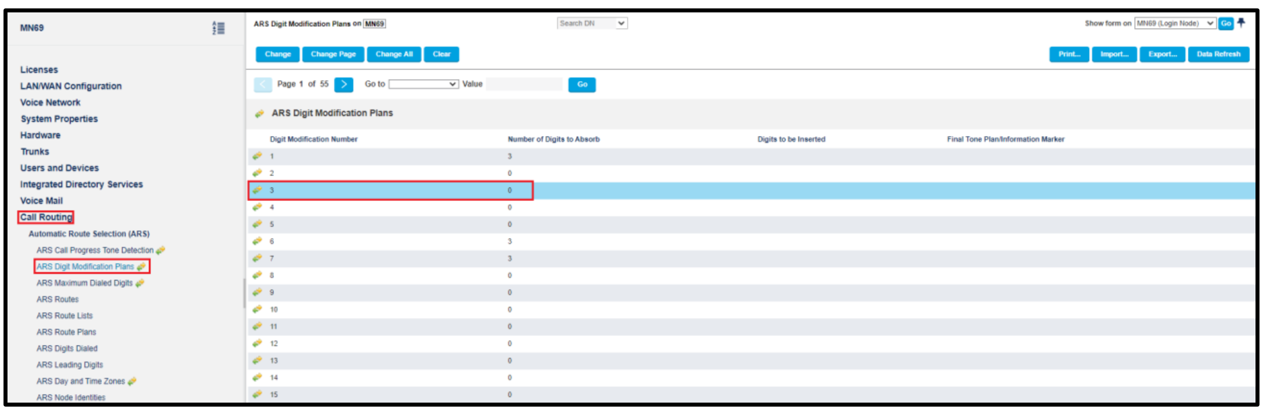


Figure 16 – Digit Modification Assignment

ARS Routes

Create a route for SIP Trunks connecting a trunk to Centurion CARES IVR. In this example, the SIP trunk is assigned to Route Number 10. Choose SIP Trunk as a routing medium and choose the SIP Peer Profile and Digit Modification entry created earlier.

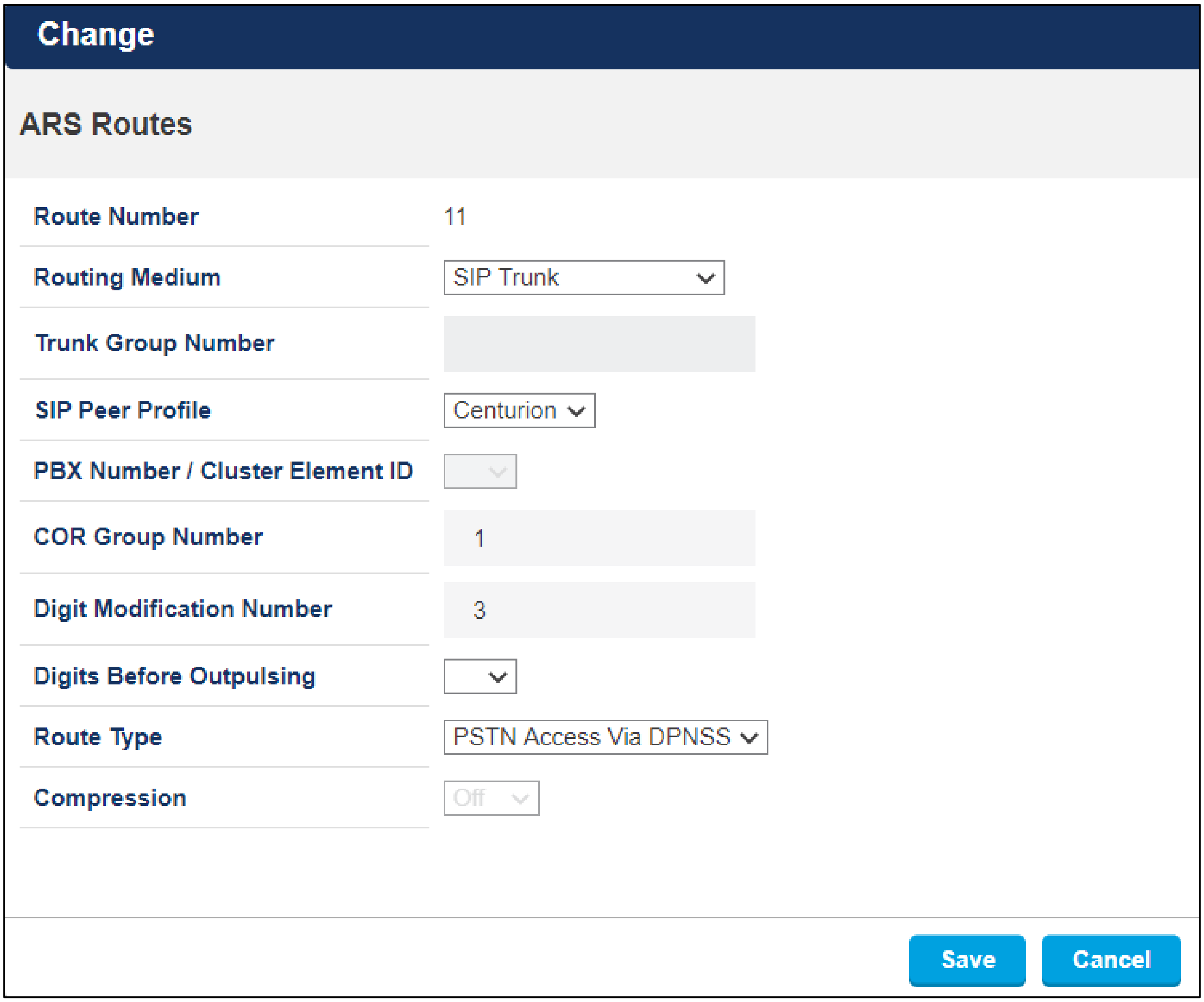


Figure 17 – SIP Trunk Route Assignment

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 4443000, the call will be routed to Centurion CARES IVR and 4443001, the call will be routed to Centurion CARES Contact Center IVR

If case of PSTN user need to reach IVR through SIP Trunk, then we need map 4443001 and 4443000 to existing DID

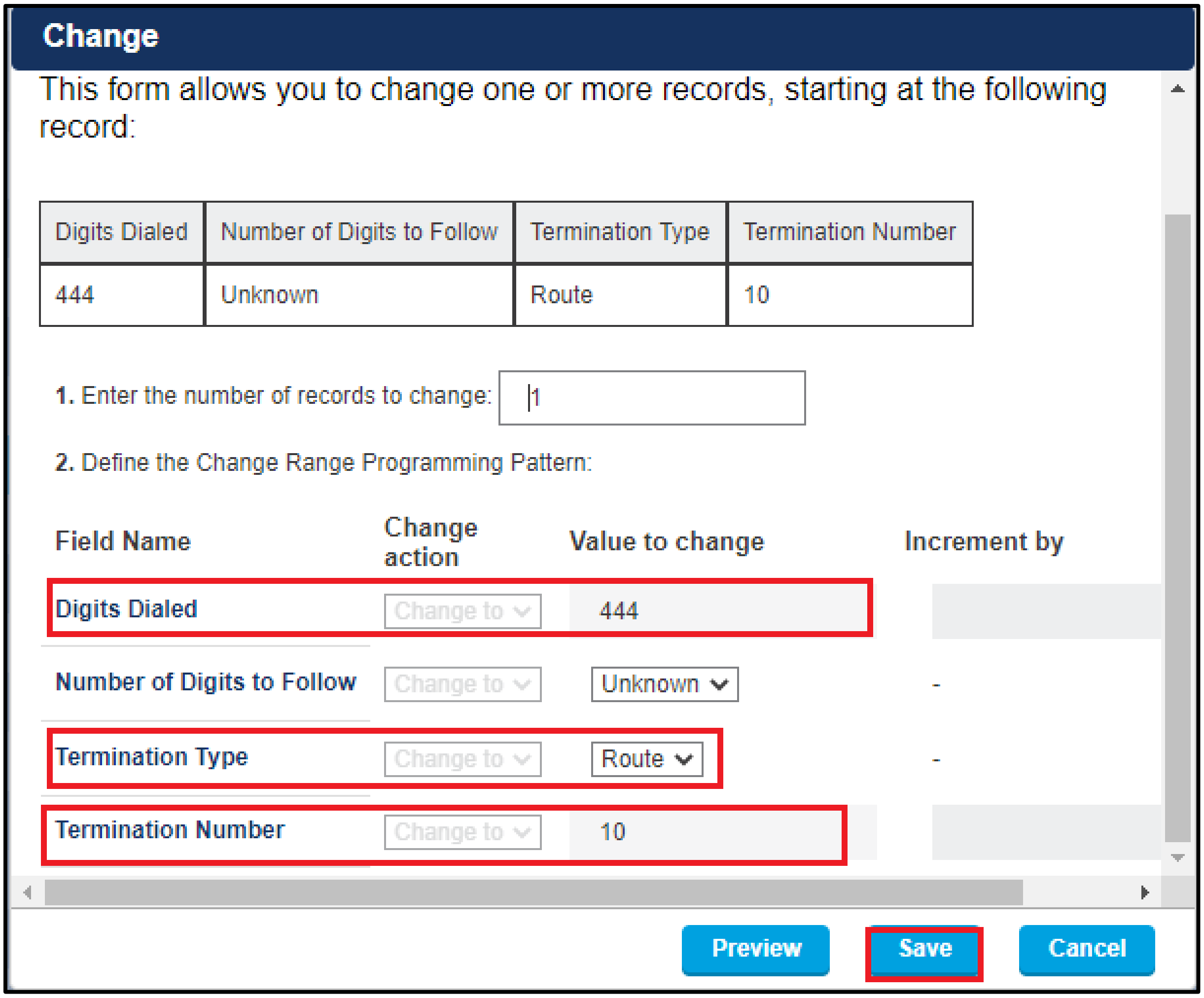


Figure 18 – ARS Digit Dialed Assignment

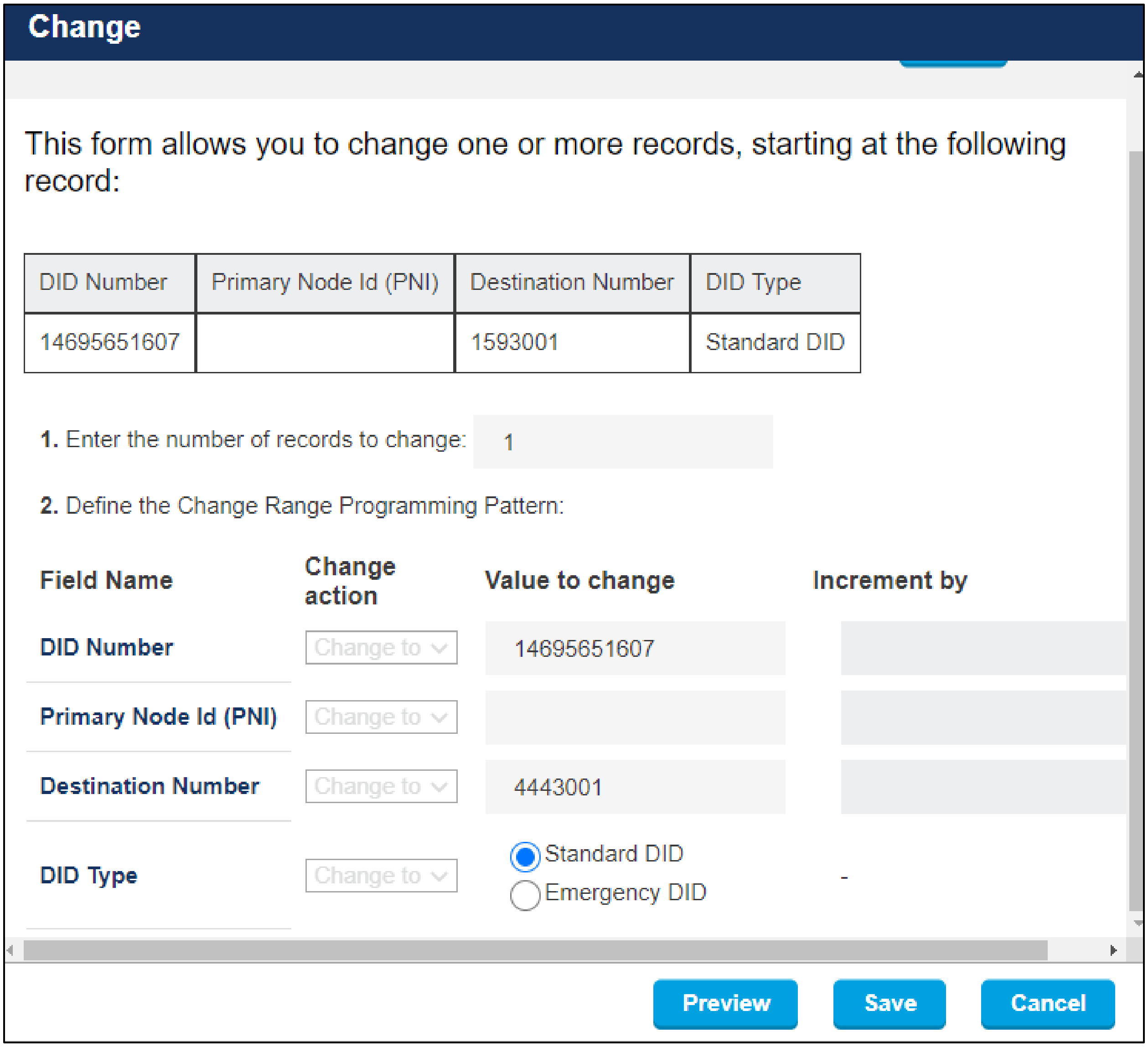


Figure 19 – DID Mapping

Note:

Please Contact Centurion for Configuration. In Lab environment configuration was done by Centurion Team

Glossary

MiVoice Business	MiVB
MiNET Interface	MiNET
Mitel Solutions Alliance	MSA
Knowledge Management System	KMS
Class of Service	COS
Automatic Route Selection	ARS