

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

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

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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.
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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

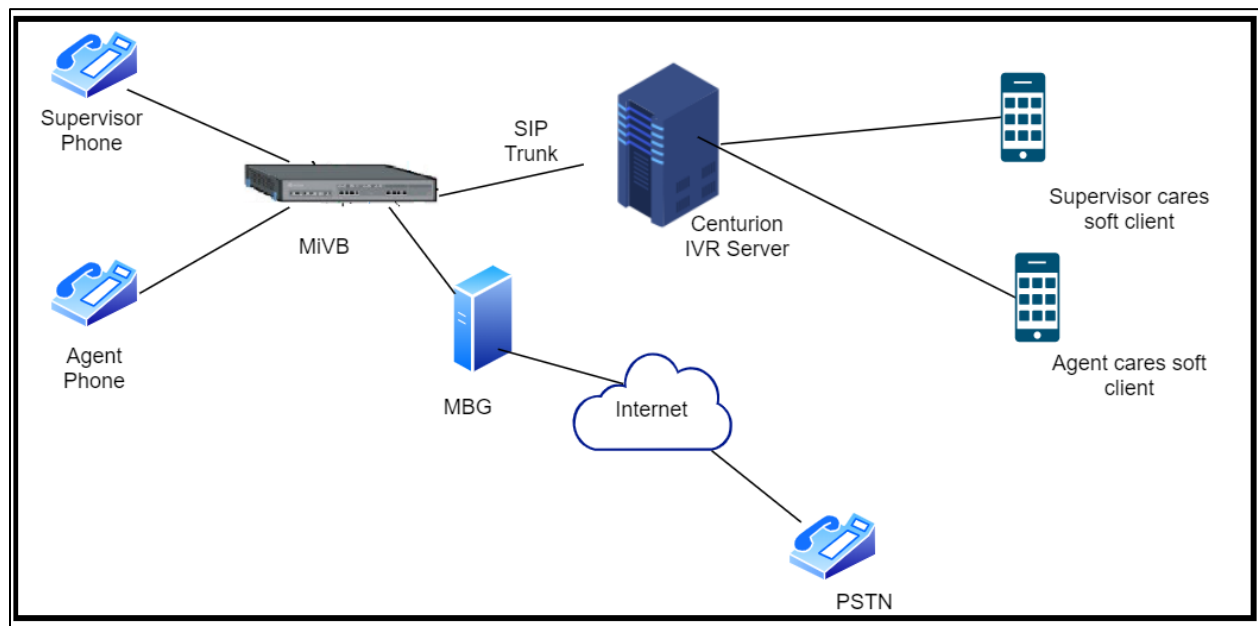


Figure 1 – Network Topology

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

Mitel | MiVoice Business

SOS Distribution Error Status: Warning

Show form on: MN69 (Login Node)

MN69

License and Option Selection on **MN69**

Search CN: [v]

[Change](#) [Print...](#) [Import...](#) [Export...](#) [Data Refresh](#)

Licenses

- License and Option Selection**
- System Capacity
- Dimension Selection
- Application Group Licensing 📄
- LAN/WAN Configuration
- Voice Network
- System Properties
- Hardware
- Trunks
- Users and Devices
- Integrated Directory Services
- Voice Mail
- Call Routing
- Music On Hold
- Emergency Services Management
- Property Management
- Maintenance and Diagnostics

License and Option Selection

Feature	Value	Unit	Limit	Status	Action
Embedded Voice Mail	35	100	0	Unrestricted	Yes
Embedded Voice Mail PMS	0	No	1 %	Unrestricted	Yes
Trunking / Networking					
Digital Links	0	0	2 %	Unrestricted	Yes
Compression		80	0	Unrestricted	Yes
FAX Over IP (T.38)		0	4	Unrestricted	Yes
SIP Trunks	0	50	0	Unrestricted	Yes
Others					
IDC Connection	0	No	1 %	Unrestricted	Yes
MLPP	0	No	0	Unrestricted	No
Configuration Options					
Country	North America				
Extended Agent Skill Group	No				
Maximum Elements per Cluster	30				

Class of Service Assignment

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

Class of Service Options on **MN69**

Search DN

Show form on **MN69 (Login No**

Change

Copy

Print...

Import...

Export...

<

Page 1 of 11

>

Go to

Value

Go

Class of Service Options

3

Centurion

General

Advanced

Class Of Service Number	3
Comment	Centurion
ACD	
ACD Agent Behavior on No Answer	Logout
ACD Agent No Answer Timer	15
ACD Make Busy on Login	No
ACD Silent Monitor Accept	No
ACD Silent Monitor Accept Monitoring Non-Prime Lines	No
ACD Silent Monitor Allowed	No
ACD Silent Monitor Notification	No
Follow 2nd Alternate Reroute for Recall to Busy ACD Agent	No
Work Timer	0
Announce	
Call Announce Line	No
Off-Hook Voice Announce Allowed	No
Handsfree AnswerBack Allowed	No
Busy Override	
Busy Override Security	No
Disable Executive Busy Override Tone	No
Executive Busy Override	No
Call Control Timer	
Busy Tone Timer	30


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.

Change


Network Elements

Name	Centurion
Type	Other
FQDN or IP Address	192.168.10.16
Local	False
Version	
Zone	1
ARID	
SIP Peer	<input checked="" type="checkbox"/>
SIP Peer Specific	
SIP Peer Transport	default
SIP Peer Port	0
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	Always Active

Save


Cancel

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	<input type="radio"/> Off <input checked="" type="radio"/> On
Caller Based Routing Service	<input checked="" type="radio"/> Off <input type="radio"/> On
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	<input checked="" type="radio"/> No <input type="radio"/> Yes
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.

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Outgoing DID Ranges	Profile Information
SIP Peer Profile Label								Centurion
Network Element								Centurion
Local Account Information								
Registration User Name								
Address Type								IP Address: 192.168.10.69
Administration Options								
Interconnect Restriction								1
Maximum Simultaneous Calls								5
Minimum Reserved Call Licenses								0
Outbound Proxy Server								
SMDR Tag								0
Trunk Service								9
Zone								1
Authentication Options								
User Name								
Password								*****
Confirm Password								*****
Authentication Option for Incoming Calls								No Authentication
Subscription User Name								
Subscription Password								*****
Subscription Confirm Password								*****
Gateway Options								
Digital Trunk Licenses								0
Maximum Digital/Analog Channels								0

Figure 6 – SIP Peer Profile Assignment- Basic

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	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 P-Called-Party-ID (if present)								Yes
Route Call Using To Header								No

Figure 7 – SIP Peer Profile Assignment- Call Routing

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Outgoing DID Ranges	Profile Information
Default CPN								
Default CPN Name								
CPN Restriction								No
Override From Header with Default CPN								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

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

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Outgoing DID Ranges	Profile Information
<div>Index</div> <div>DID Range</div> <div>CPN Substitution</div>								<div>Update</div>

Figure 14 – SIP Peer Profile Assignment- Outgoing DID Ranges

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Outgoing DID Ranges	Profile Information
<div>Creator</div> <div>Date Created</div> <div>Created with Version</div> <div>Service Provider</div> <div>Vendor Notes</div>								

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.

MN69

Licenses

LAN/WAN Configuration

Voice Network

System Properties

Hardware

Trunks

Users and Devices

Integrated Directory Services

Voice Mail

Call Routing

Automatic Route Selection (ARS)

ARS Call Progress Tone Detection

ARS Digit Modification Plans

ARS Maximum Dialed Digits

ARS Routes

ARS Route Lists

ARS Route Plans

ARS Digits Dialed

ARS Leading Digits

ARS Day and Time Zones

ARS Verification Times

ARS Digit Modification Plans on MN69

Change

Change Page

Change All

Clear

Show form on: MN69 (Login Node)

Go

Print...

Import...

Export...

Data Refresh

Page 1 of 55

Go to

Value

Go

ARS Digit Modification Plans

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

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.

Change

ARS Routes

Route Number	11
Routing Medium	SIP Trunk
Trunk Group Number	
SIP Peer Profile	Centurion
PBX Number / Cluster Element ID	
COR Group Number	1
Digit Modification Number	3
Digits Before Outpulsing	
Route Type	PSTN Access Via DPNSS
Compression	Off

Save

Cancel

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

Change

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
444	Unknown	Route	10

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	<input type="text" value="Change to"/>	<input type="text" value="444"/>	<input type="text"/>
Number of Digits to Follow	<input type="text" value="Change to"/>	<input type="text" value="Unknown"/>	-
Termination Type	<input type="text" value="Change to"/>	<input type="text" value="Route"/>	-
Termination Number	<input type="text" value="Change to"/>	<input type="text" value="10"/>	<input type="text"/>

Figure 18 – ARS Digit Dialed Assignment

Change

This form allows you to change one or more records, starting at the following record:

DID Number	Primary Node Id (PNI)	Destination Number	DID Type
14695651607		1593001	Standard DID

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
DID Number	Change to ▾	14695651607	
Primary Node Id (PNI)	Change to ▾		
Destination Number	Change to ▾	4443001	
DID Type	Change to ▾	<input checked="" type="radio"/> Standard DID <input type="radio"/> Emergency DID	-

Preview

Save

Cancel

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