Mitel Technical Configuration Notes- HO3860

August 10, 2020

Configure MiVoice Business MiCloud Flex 4.2 for use with NISC IVR

Description: This document provides a reference to Mitel Authorized Solutions providers for configuring the MiVoice Business to connect to the NISC IVR.

Environment: MiCloud Flex 4.2, NISC IVR

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Mitel Technical Configuration Notes – Configure MiVoice Business 9.0 SP1 PR1 and MiVCR for use with NISC IVR.

Mitel Configuration Guide HO3860

July 2020

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Overview

This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel MiVB connect to NISC 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	July 2020	Interop with Mitel MiVB 9.0 SP1 PR1 (9.0.1.23) and NISC IVR.

Interop Status

The Interop of NISC IVR with MiVB has been given a Certification status. This Call recording device will be included in the Mitel Interoperability Reference Guide (IRG). The status Castel Call Recording system achieved is:

COMPATIBLE	The most common certification which means NISC 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 Upload a file from NISC IVR and MiVB.

<u>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 –</u>

Manufacturer	Tested Variants	Software Version	Additional Applicable Variants
Mitel	MiVoice Business	Release 9.0 SP1 PR1 (9.0.1.23)	NA
Mitel	MBG – Teleworker	10.1.0.250	NA
Mitel	69XX SIP and 68XX SIP 69XX MiNET	5.1.0.1024 01.04.00.85	NA
NISC IVR	Version as of July, 2020		

Tested Features

This is an overview of the features tested during the Interop test cycle and not a detailed view of the test cases.

Feature	Feature Description	Issues
Basic Call	Making and receiving a call	☑
Call Hold/Retrieve	Putting a call on hold/retrieve with MOH	☑
Call Transfer	Transferring a call to another destination	☑
Conference	Conferencing multiple calls together	☑
Call Forward	Forwarding calls to another destination using ESM	☑
Teleworker	Mitel remote connectivity with Teleworker	☑
Codec	Making and receiving calls Using G711 Codec	☑
Simultaneous	Making Simultaneous Calls	☑

☑ - No issues found ✗ - Issues found, cannot recommend to use ⚠ - Issues found

Network Topology

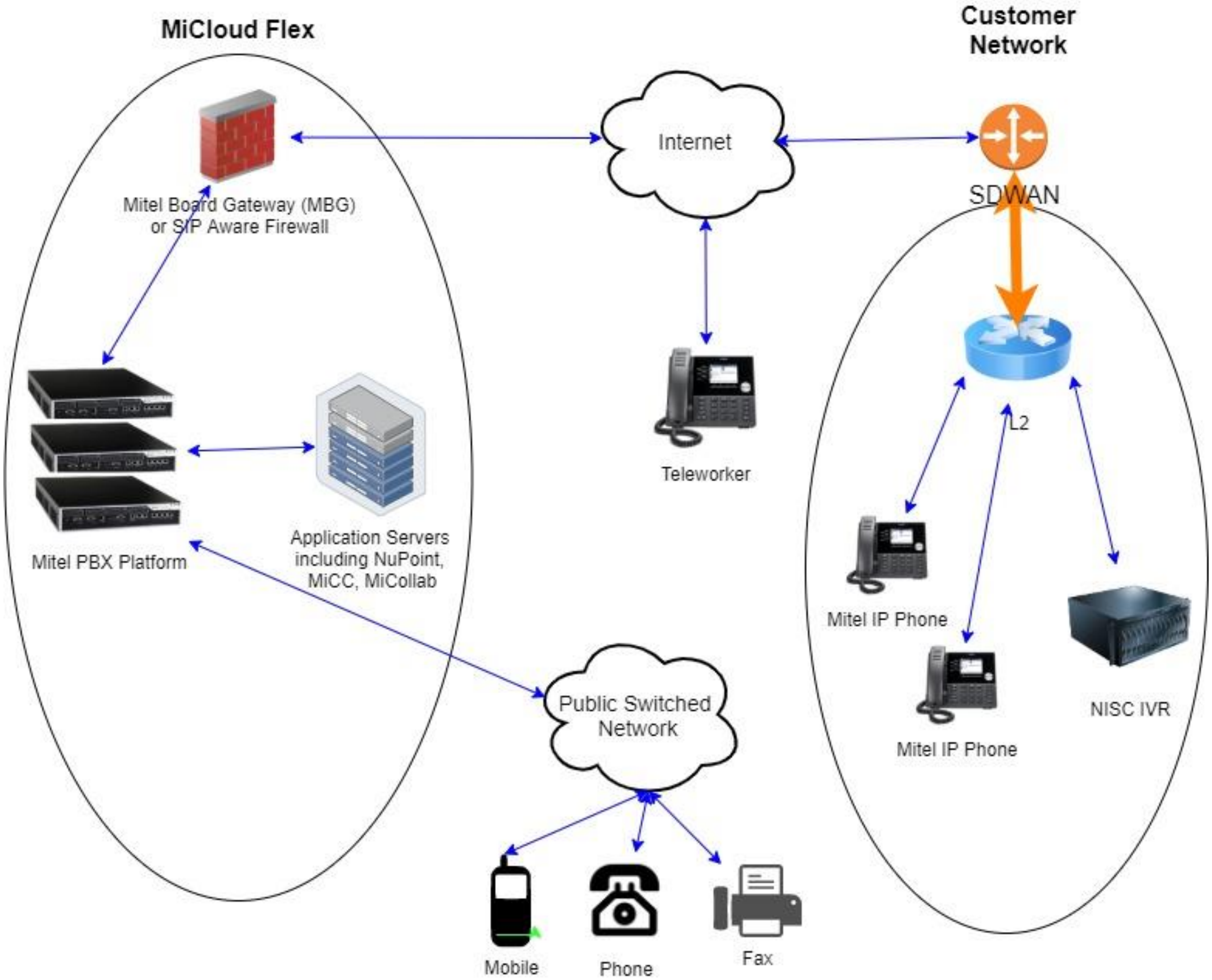


Figure 1 – Network Topology

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 NISC IVR 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.

MiVoice Business Configuration Notes

The following steps show how to program a MiVB to interconnect with Service Provider NISC IVR SIP Trunking.

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 for G729. 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 NISC IVR SIP Trunking. 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.

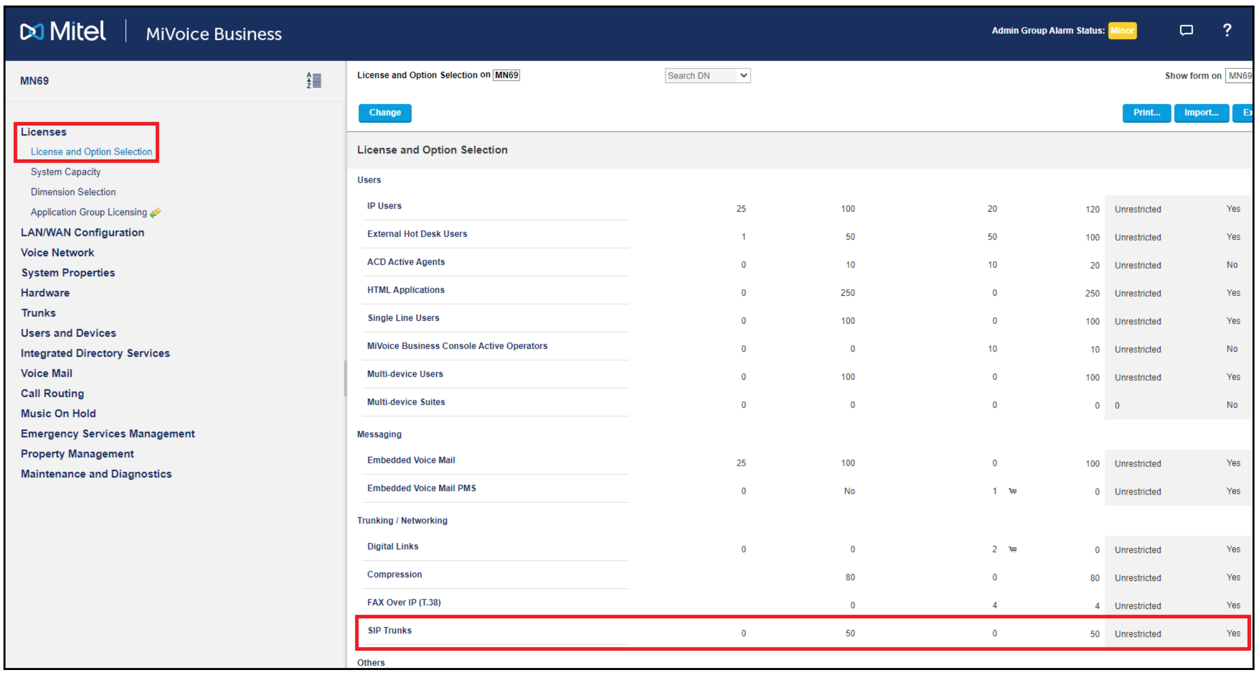


Figure 2 – License and Option Selection

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

•

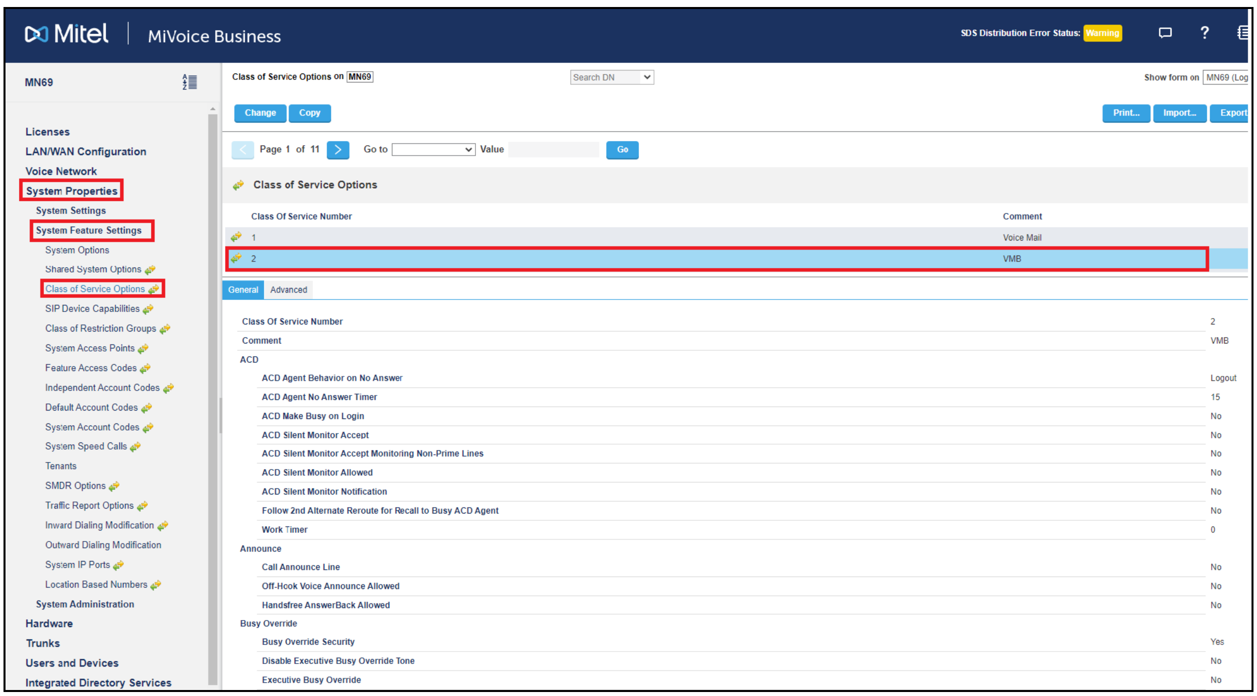
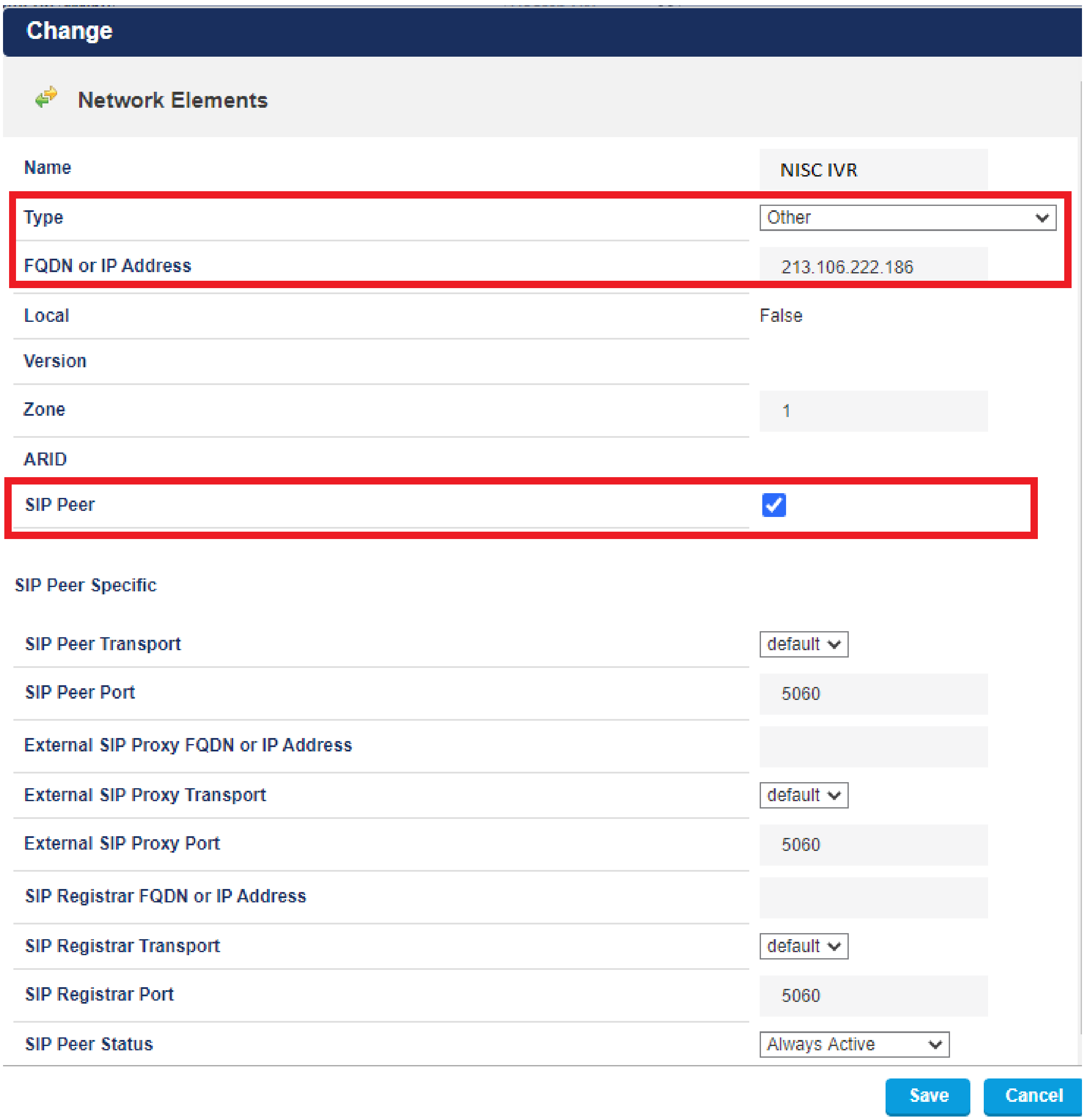


Figure 3 – Class of Service

Network Element Assignment

Create a network element for Service Provider NISC IVR SIP Trunking. In this example, the soft switch is reachable by an IP Address and is defined as “NISC IVR“ 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 NISC IVR 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.



Network Element Assignment (Proxy)

In addition, depending in 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 MiVB will require knowledge of this by programming the Proxy as a network element then referencing this proxy in the SIP Peer profile assignment (later in this document).

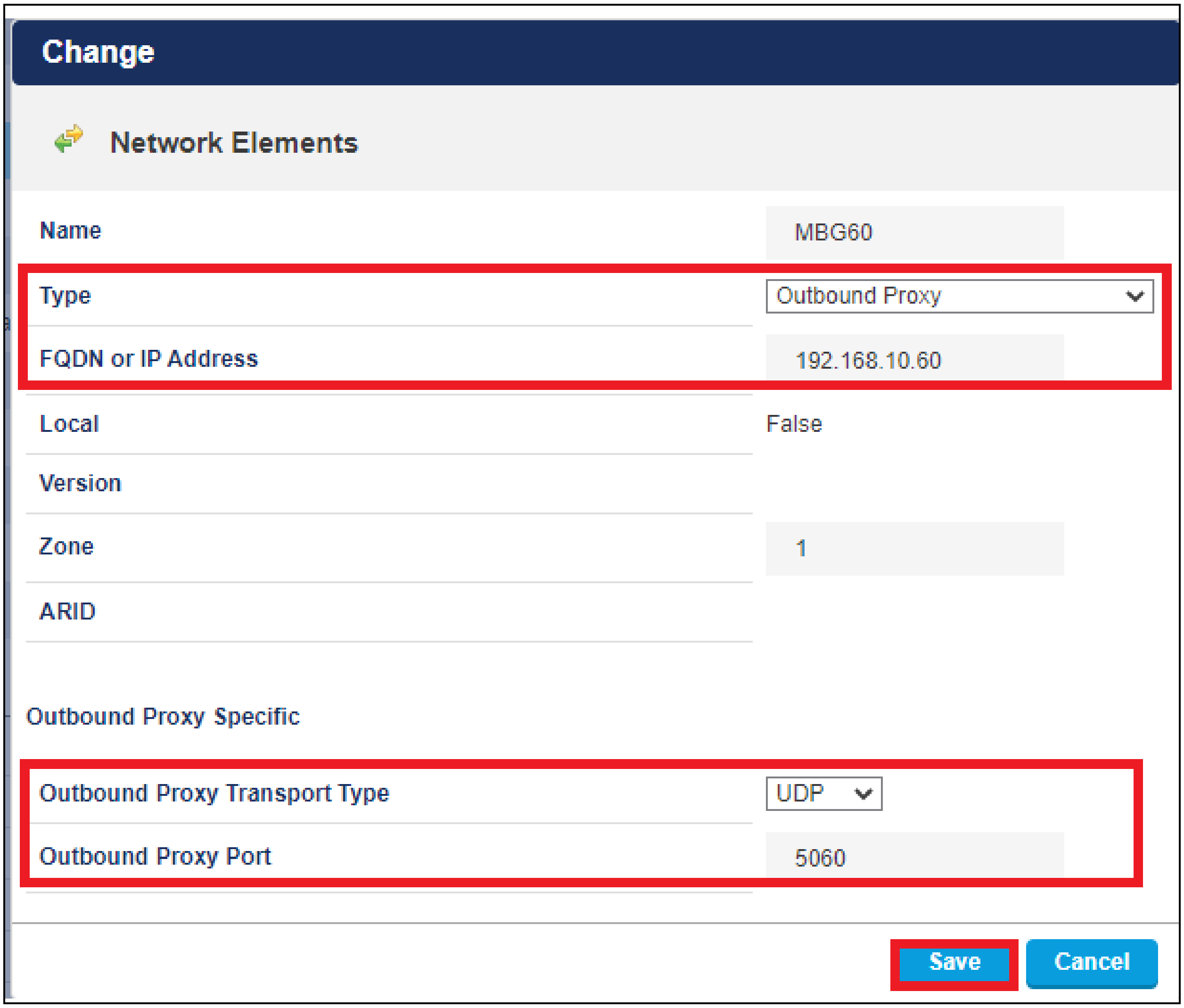


Figure 5 – Network Element Assignment (Proxy)

Trunk Attributes

This is configured in the Trunk Attributes form. In this example the Trunk Attributes is defined for Trunk Service Number 1 for NISC IVR which will be used to direct incoming calls to an answer point in the Mitel MiVB.

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 MiVB will absorb the first 8 digits of the DID number from Service Provider NISC IVR leaving 4 digits for the MiVB to translate and ring the remaining 4-digit extension. For example, Service Provider NISC IVR delivers 4411-8337-4140 through the SIP trunk to the MiVB. The MiVB will absorb the first 8 digits

(44118337) leaving the MiVB to ring extension 4140. Extension 4140 must be programmed as a valid dialable number in the MiVB. Please refer to the Mitel MiVB System Administration documentation for further programming information.

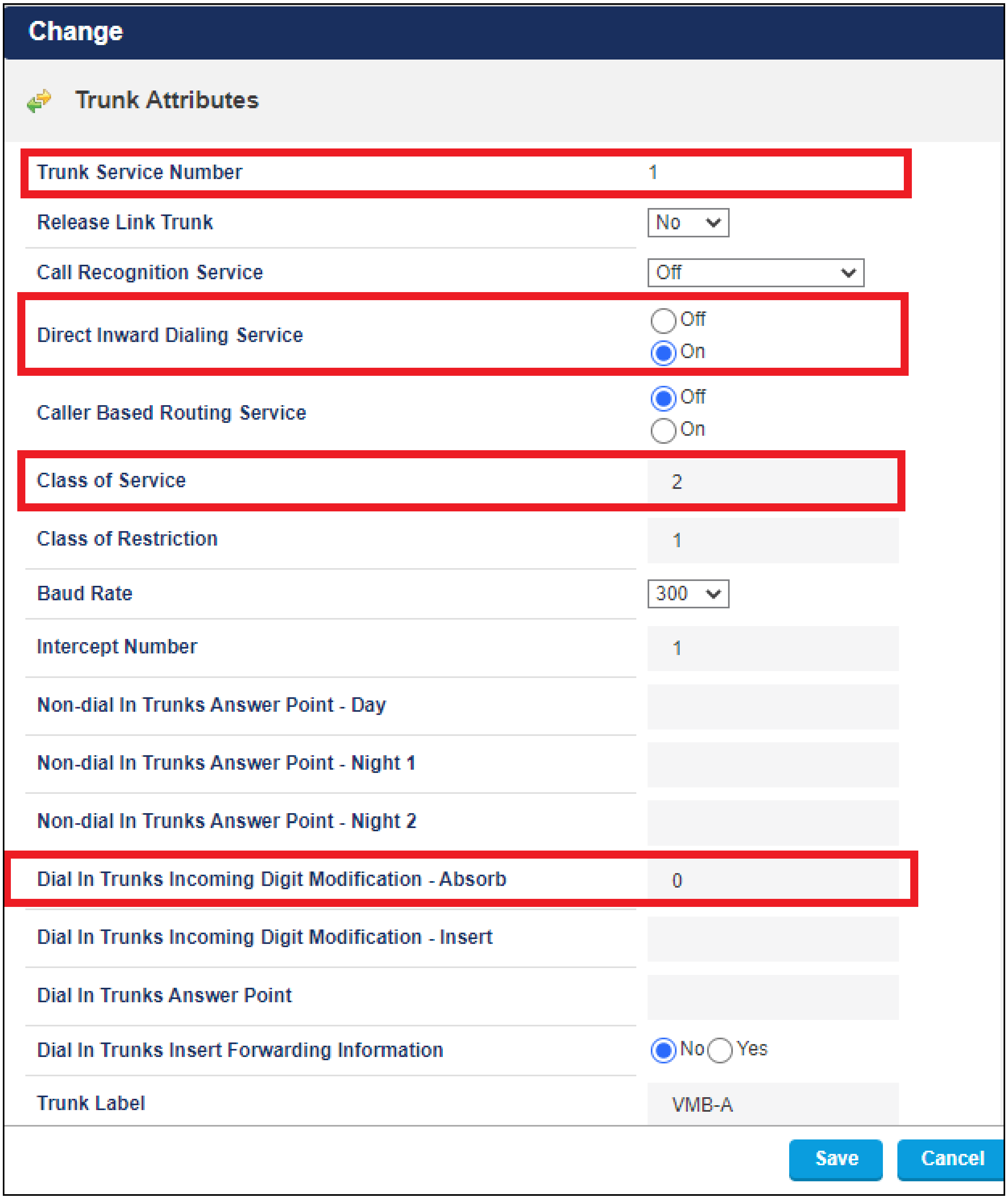

Change
Trunk Attributes
Trunk Service Number
1
Release Link Trunk
No
Call Recognition Service
Off
Direct Inward Dialing Service
Off
On
Caller Based Routing Service
Off
On
Class of Service
2
Class of Restriction
1
Baud Rate
300
Intercept Number
1
Non-dial In Trunks Answer Point - Day
Non-dial In Trunks 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
Trunk Label
VMB-A
Save
Cancel

Figure 6 – Trunk Attributes for both VMB trunks

SIP Peer Profile

The recommended connectivity via SIP Trunking does not require additional physical interfaces. IP/Ethernet connectivity is part of the base MiVB 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 "NISC IVR" Network Element.

Registration Username: The Mitel MiVB does not support Bulk Registration; therefore, trunks will have to be registered individually. Enter the DIDs assigned by Service Provider NISC IVR. Enter one or more numbers. The field has a maximum of 60 characters. The maximum number of digits per number is 26. You can enter a mix of ranges and single numbers (for example, "441183374140, 441183374141, 441183374142, 441183374143"). Use a comma to separate telephone numbers and ranges. Use a dash (-) to indicate a range of telephone numbers. The first and last characters cannot be a comma or a dash.

Address Type: Select IP address.

Outbound Proxy Server: Select the Network Element previously configured for the Outbound Proxy Server.

Calling Line ID: The default CPN 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 Service Provider NISC IVR.** 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 Service Provider NISC IVR.

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

SIP Peer Profile Parameter	Default	Required Setting
Basic		
SIP Peer Profile Label	Blank	IVR
Network Element	Blank	IVR
Basic - Local Account Information		
Registration User Name	Blank	default
Address Type	FQDN	default
Basic - Administration Options		
Interconnect Restriction	1	default
Maximum Simultaneous Calls	2000	<trunk License>
Minimum Reserved Call Licenses	0	default
Outbound Proxy Server		
SMDR Tag	Blank	<customer requirement>
Trunk Service	1	<as desired programming>
Zone	1	default
Basic - Authentication Options		
User Name	Blank	default
Password	Blank	default
Confirm Password	Blank	default
Authentication Option for Incoming Calls	No Authentication	default
Subscription User Name	*******	default
Subscription Password	*******	default
Subscription Confirm Password	*******	default
Gateway Options		default
Digital Trunk Licenses		default
Maximum Digital/Analog Channels		default
Call Routing		
Alternate Destination Domain Enabled	No	default
Alternate Destination Domain FQDN or IP Address	Blank	default
Enable Special Re-invite Collision Handling	No	default
Only Allow Outgoing Calls	No	default
Private SIP Trunk	No	default
Reject Incoming Anonymous Calls	No	default
Route Call Using P-Called-Party-ID (if present)	Yes	default
Route Call Using To Header	No	default
Calling Line ID		
Default CPN	Blank	default
Default CPN Name	Blank	default
CPN Restriction	No	default

Override From Header with Default CPN	No	default
Public Calling Party Number Passthrough	No	default
Strip PNI	No	default
Use Diverting Party Number as Calling Party Number	No	default
Use Original Calling Party Number If Available	No	default
SDP Options		
Allow Peer to Use Multiple Active M-Lines	No	default
Allow Using UPDATE For Early Media Renegotiation	No	default
Avoid Signaling Hold to the Peer	Yes	default
AVP Only Peer	Yes	default
Enable Mitel Proprietary SDP	No	default
Force sending SDP in initial Invite message	No	Yes
Force sending SDP in Initial Invite - Early Answer	No	default
Ignore SDP Answers in Provisional Responses	No	default
IP Media Default	ipv4	default
Limit to one Offer/Answer per INVITE	Yes	default
NAT Keepalive	Yes	default
Prevent Codec Selection on Answer	No	default
Prevent the Use of IP Address 0.0.0.0 in SDP Messages	Yes	default
Reject Call without telephone-event payload	No	default
Renegotiate SDP To Enforce Symmetric Codec	No	default
Repeat SDP Answer If Duplicate Offer Is Received	No	default
Restrict Audio Codec	No Restriction	default
RTP Packetization Rate Override	No	default
RTP Packetization Rate	20 ms	default
Special handling of Offers in 2XX responses (INVITE)	No	default
Suppress Use of SDP Inactive Media Streams	No	default
Signaling and Header Manipulation		
Trunk Group Label	Blank	default
Allow Display Update	No	default
Build Contact Using Request URI Address	No	default
De-register Using Contact Address not "*"	Yes	default
Disable Reliable Provisional Responses	Yes	Yes
Disable Use of User-Agent and Server Headers	No	default
Discard Received P-Asserted-Identity Headers	No	default
Domain for Trunk Context	Blank	default
E.164: Enable sending '+'	No	default
E.164: Add '+' if digit length > N digits	0	default
E.164: Do not add '+' to Emergency Called Party	No	default
E.164: Do not add '+' to Called Party	No	default
Force Max-Forward: 70 on Outgoing Calls	No	default

If TLS use 'sips:' Scheme	No	default
Ignore Incoming Loose Routing Indication	No	default
Multilingual Name Display	No	default
Only use SDP to decide 180 or 183	Yes	default
Include Diversion Header for EHDU	No	default
Mode for Out-of-Band DTMF	RFC 4733 DTMF	default
Prefer From Header for Caller ID	No	default
Q.850 Reason Headers	No	default
Require Reliable Provisional Responses on Outgoing Calls	Yes	No
Signal Privacy (if enabled) on Emergency Calls	No	default
Suppress Incoming Name	No	default
Suppress Redirection Headers	No	default
Use Fixed Retry Time for 491	No	default
Use Privacy: none	No	default
Use P-Asserted Identity Header	Yes	default
Use P-Asserted Identity for Billing	No	default
Use P-Call-Leg-ID Header	No	default
Use P-Early-Media Header	No	default
Use P-Preferred Identity Header	No	default
Use Restricted Character Set for Authentication	No	default
Use To Address in From Header on Outgoing Calls	No	default
Use user=phone	No	default
Use user=phone for Diversion Header	No	default
Timers		
Invite Ringing Response Timer	0 (disabled)	default
Keep-Alive (OPTIONS) Period	120	default
Registration Period	3600	default
Registration Period Refresh (%)	50	default
Registration Maximum Timeout	90	default
Session Timer	90	default
Session Timer: Local as Refresher	No	default
Subscription Period	3600	default
Subscription Period Minimum	300	default
Subscription Period Refresh (%)	80	default
Key Press Event Options		
Allow Inc Subscriptions for Local Digit Monitoring	No	default
Allow Out Subscriptions for Remote Digit Monitoring	No	default
Force Out Subscriptions for Remote Digit Monitoring	No	default
Request Outbound Proxy to Handle Out Subscriptions	No	default
KPML Transport	Default (UDP)	default

KPML Port	0	default
Outgoing DID Ranges		
Add Member		
Delete Member		
Index		
DID Range	Blank	
CPN Substitution	Blank	
Profile Information		
Creator	Local	
Date Created	MM DD YY	
Created with Version		
Service Provider	Blank	
Vendor Notes	Blank	

Figure 15 – SIP Peer Profile Assignment- Profile Information

SIP Peer Profile Assignment by Incoming DID

This form is used to associate DID range numbers from Service Provider NISC IVR SIP trunk to a particular SIP Peer profile. The configured settings here help matching the incoming DID numbers with the SIP Peer Profile when call is arriving from anonymous caller.

Enter one or more telephone numbers. The maximum number of digits per telephone number is 26. You can enter a mix of ranges and single numbers (for example, " 441183374140-441183374149"). The entire field width is limited to 60 characters.

Use a comma to separate telephone numbers and ranges. Use a dash (-) to indicate a range of telephone numbers. The first and last characters cannot be a comma or a dash. If the numbers do not fit within the 60 characters maximum, you can create a new entry for the same profile.

Use a '*' to reduce the number of entries that need to be programmed. This is a type of "prefix identifier" and cannot be used as a range with '-'. For example, the string "11*" would be used to associate a peer with any number in the range from 110 up to the maximum digits per telephone number (In this case, 11999999999999999999999999.) Note that the string "11" by itself would not count as a match, as the '*' represents 1 or more digits.

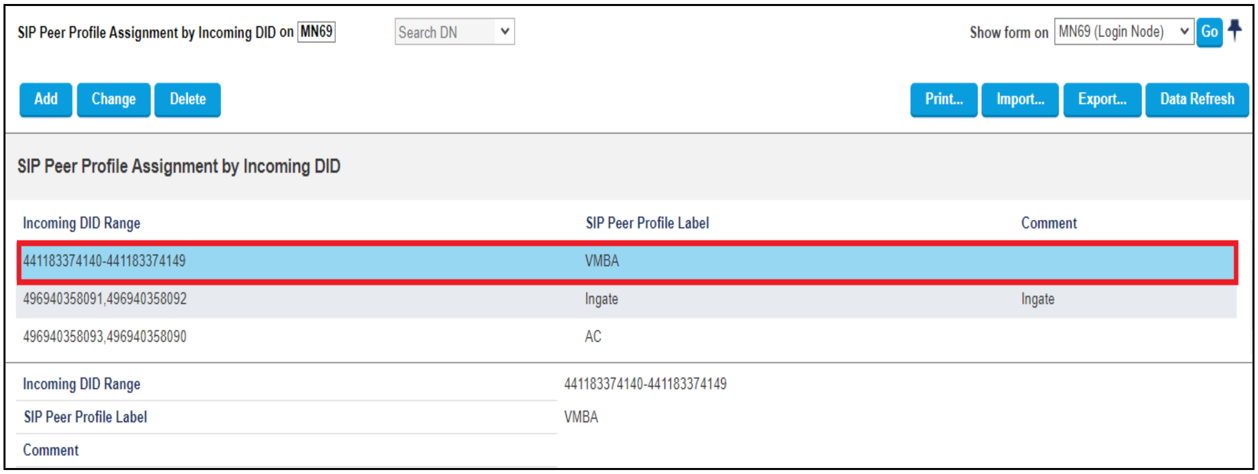


Figure 16 – SIP Peer Profile Assignment by Incoming DID

ARS Digit Modification Plans

Ensure that Digit Modification for outgoing calls on the SIP trunk to Service Provider NISC IVR absorbs or injects additional digits according to your dialling plan. In this example, we will be absorbing 3 digits (in this case will be 456 to dial out).

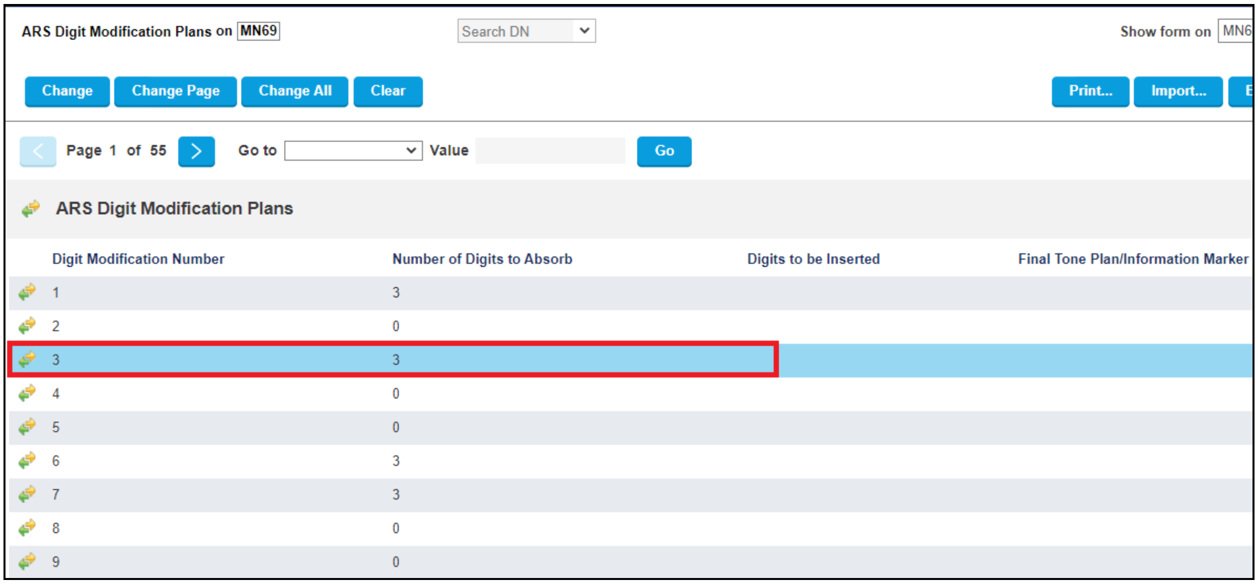


Figure 17 – Digit Modification Assignment for both VMB trunks

Ensure that Digit Modification for outgoing calls on the SIP trunk to Service Provider NISC IVR absorbs or injects additional digits according to your dialling plan. In this example, we will be absorbing 0 digits for Emergency calls (in this case will directly dial 112 or 999 or 18000).

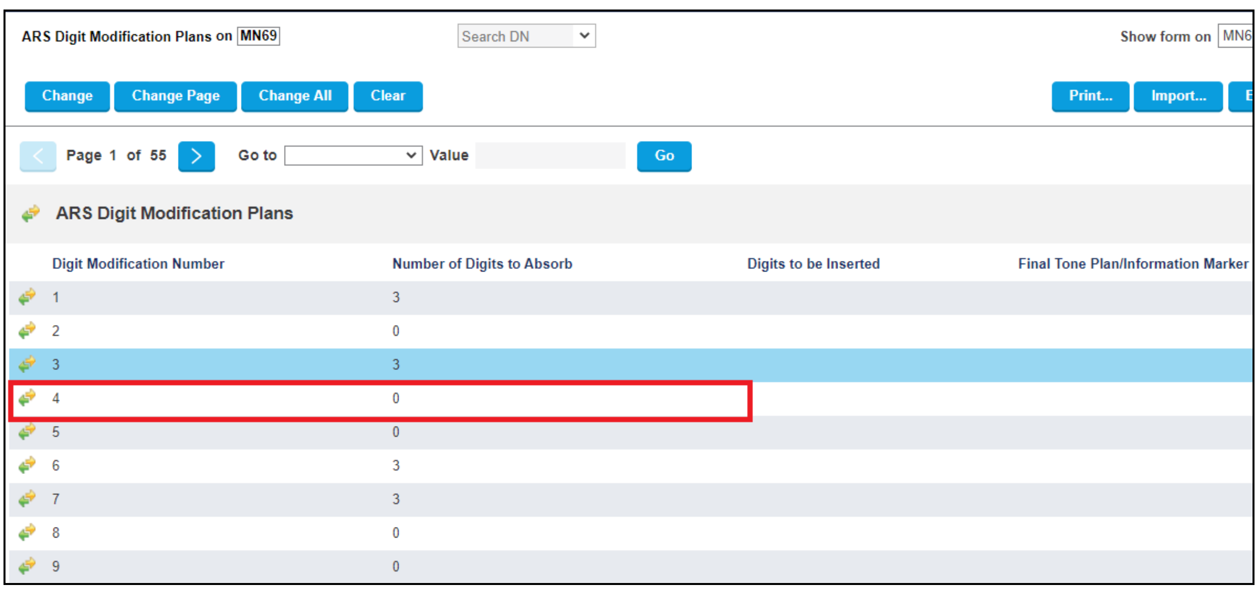


Figure 17 (a)– Digit Modification Assignment for Emergency Calls

ARS Routes

Create a route for both SIP Trunks connecting a trunk to Service Provider NISC IVR In this example, VMB-A SIP trunk is associated with Route Number 1 and VMB-B SIP trunk is associated with Route Number 2. In this example 3 is associated with emergency route. Choose SIP Trunk as a routing medium and choose the SIP Peer Profile and Digit Modification entry created earlier.

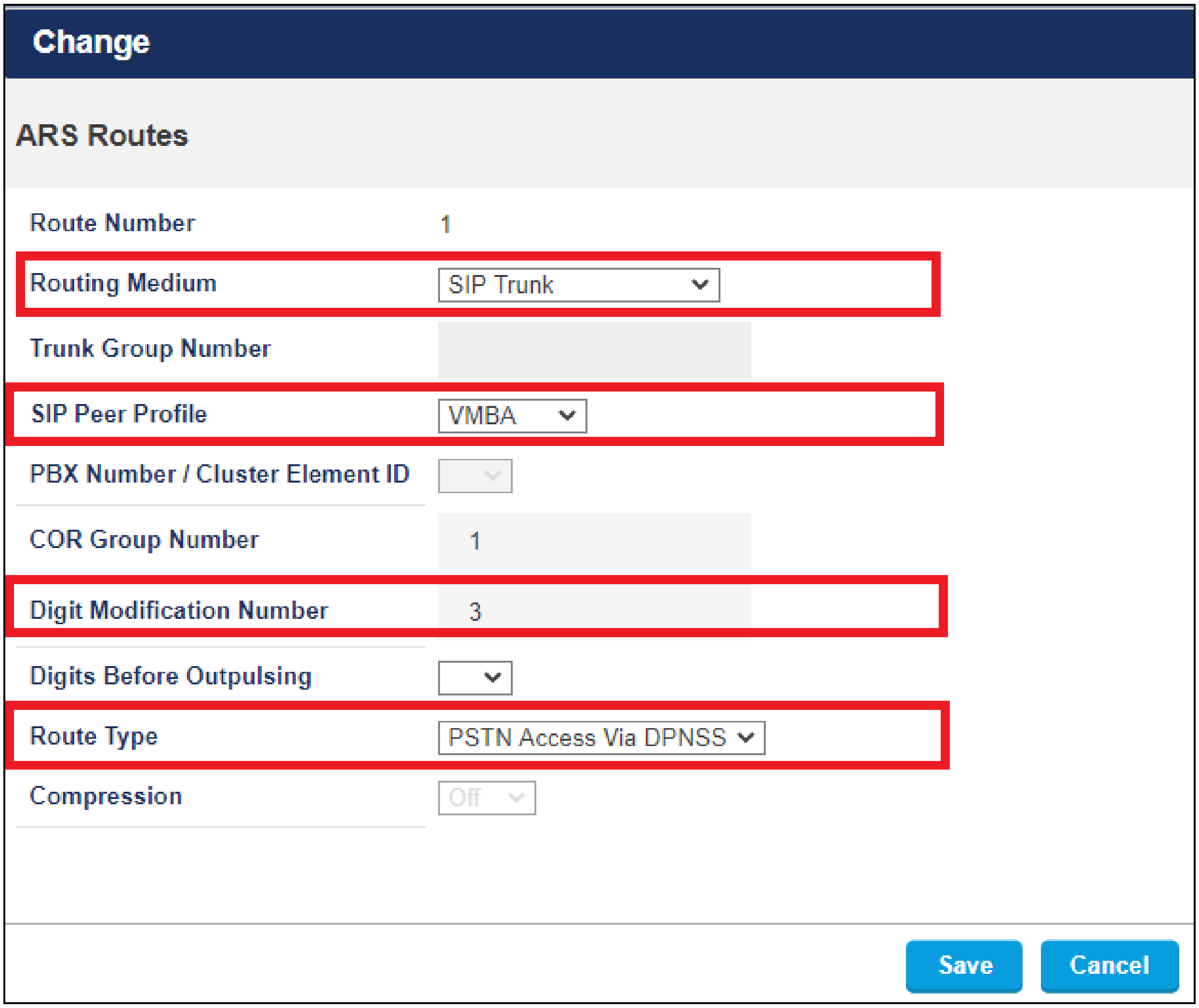


Figure 18 – SIP Trunk Route Assignment for VMB-A

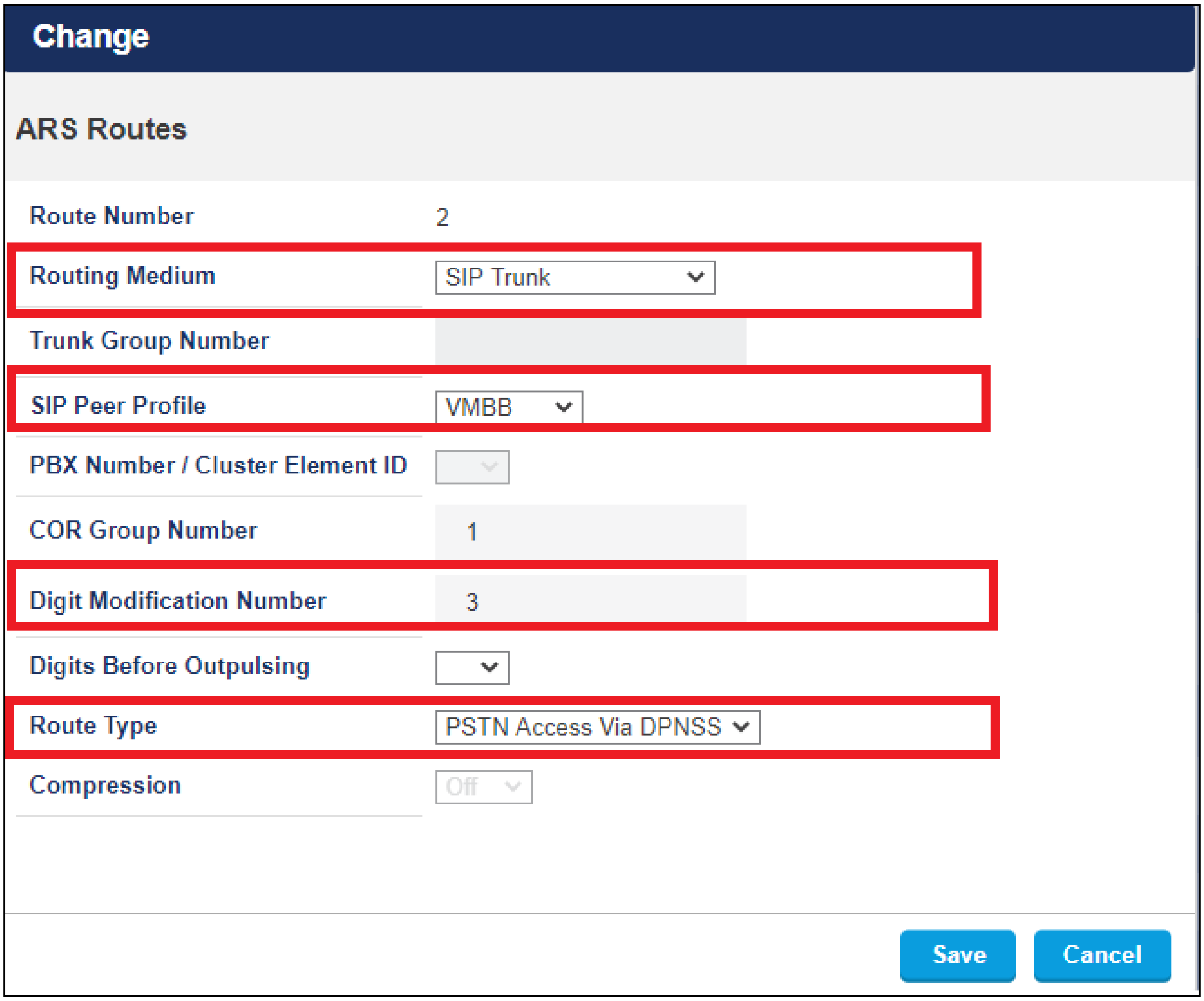


Figure 18 (a) – SIP Trunk Route Assignment for VMB-B

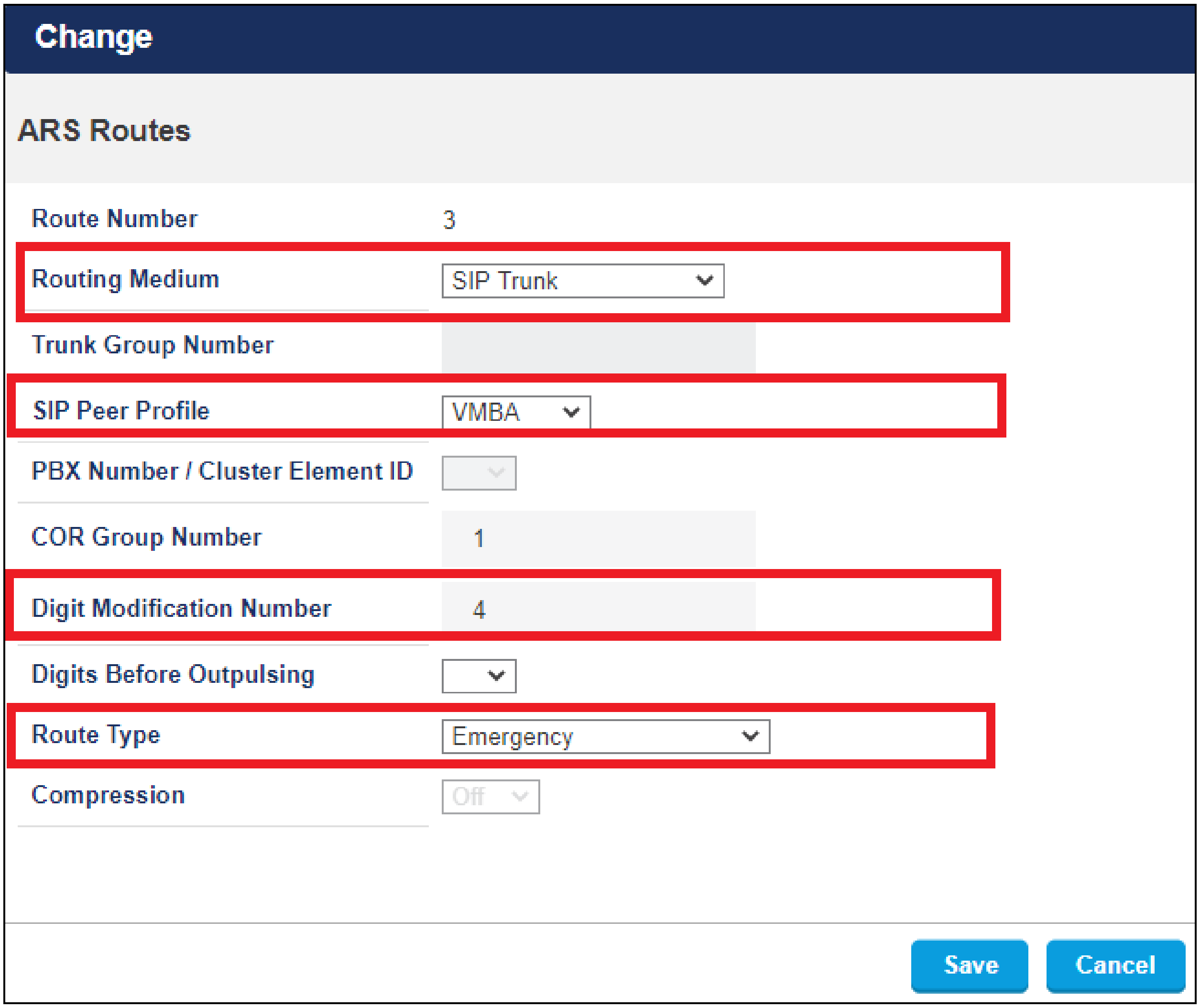
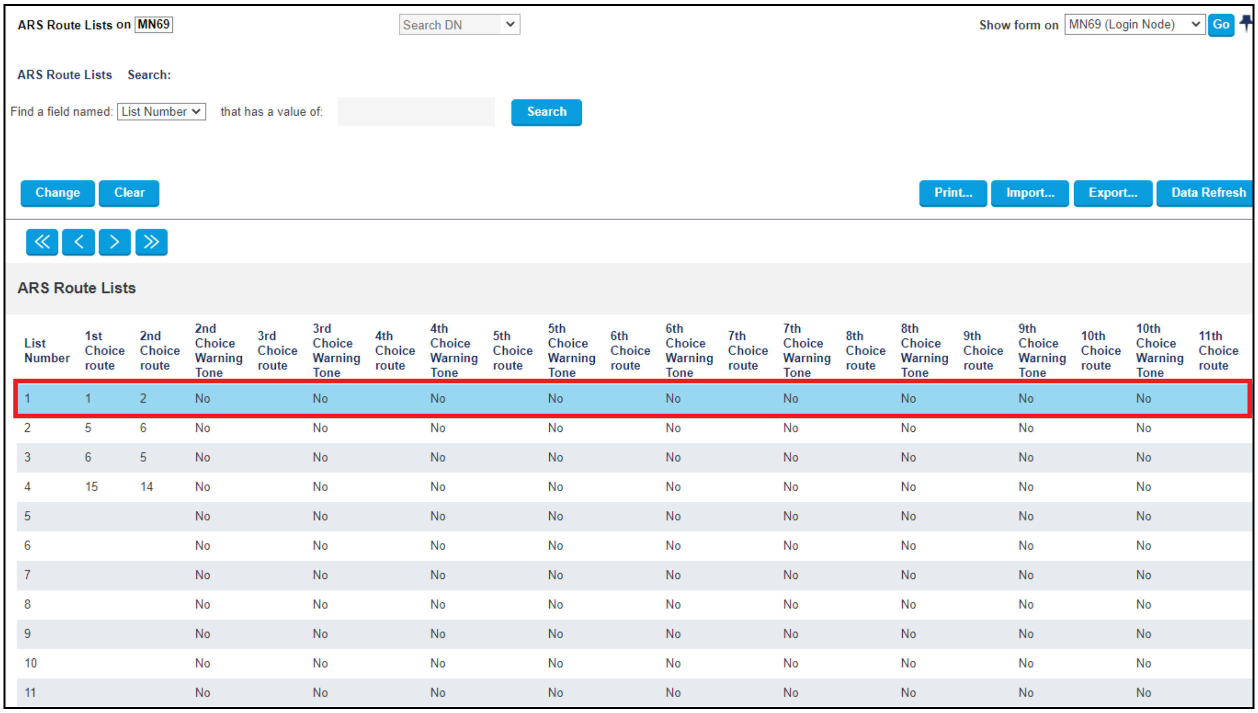


Figure 18 (b) – SIP Trunk Route Assignment for Emergency Calls

ARS Route Lists

The ARS Route Lists form provides a list of trunk routes (created in 1) to call according to customer requirements and cost.


ARS Route Lists on MN69

Search DN

Show form on MN69 (Login Node) Go

ARS Route Lists Search:

Find a field named: List Number that has a value of: Search

Change Clear Print... Import... Export... Data Refresh

ARS Route Lists

List Number	1st Choice route	2nd Choice route	2nd Choice Warning Tone	3rd Choice route	3rd Choice Warning Tone	4th Choice route	4th Choice Warning Tone	5th Choice route	5th Choice Warning Tone	6th Choice route	6th Choice Warning Tone	7th Choice route	7th Choice Warning Tone	8th Choice route	8th Choice Warning Tone	9th Choice route	9th Choice Warning Tone	10th Choice route	10th Choice Warning Tone	11th Choice route
1	1	2	No		No		No		No		No		No		No		No		No	
2	5	6	No		No		No		No		No		No		No		No		No	
3	6	5	No		No		No		No		No		No		No		No		No	
4	15	14	No		No		No		No		No		No		No		No		No	
5			No		No		No		No		No		No		No		No		No	
6			No		No		No		No		No		No		No		No		No	
7			No		No		No		No		No		No		No		No		No	
8			No		No		No		No		No		No		No		No		No	
9			No		No		No		No		No		No		No		No		No	
10			No		No		No		No		No		No		No		No		No	
11			No		No		No		No		No		No		No		No		No	

Figure 19 - ARS Route Lists Form

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 456, the call will be routed to VMB-A trunk (or to VMB-B trunk in case of failover).

(Route List 1)

ARS Digits Dialed on MN69 | Search DN | Show form on MN69 (Login Node) Go

Add | Change | Delete | Print... | Import... | Export... | Data Refres

Page 1 of 1 | Go to | Value | Go

ARS Digits Dialed

Digits Dialed	Number of Digits to Follow	Termination Type	Termination Number
112	Unknown	Route	3
123	Unknown	List	2
18000	Unknown	Route	3
456	Unknown	List	1
457	Unknown	Route	7
555	Unknown	Route	4
789	Unknown	List	3

Figure 20 – ARS Digit Dialed Assignment

ARS initiates the routing of trunk calls when certain digits are dialed from a station. In this example, when a user dials 112 or 999 or 18000, the call will be routed to Service Provider NISC IVR.

ARS Digits Dialed on MN69 | Search DN | Show form on MN69 (Login No

Add | Change | Delete | Print... | Import... | Export...

Page 1 of 1 | Go to | Value | Go

ARS Digits Dialed

Digits Dialed	Number of Digits to Follow	Termination Type	Termination Number
112	Unknown	Route	3
123	Unknown	List	2
18000	Unknown	Route	3
456	Unknown	List	1
457	Unknown	Route	7
555	Unknown	Route	4
789	Unknown	List	3
888	Unknown	List	4
911	Unknown	Route	3
999	Unknown	Route	3

Figure 20(a) – ARS Digit Dialed Assignment for Emergency Calls

Note – This configuration is just a reference where the customer can always change the configuration according to their requirements. If any customization is required, "the customer should refer to the product guides, which provides detailed information".

T.38 Fax Configuration

Service Provider NISC IVR 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).

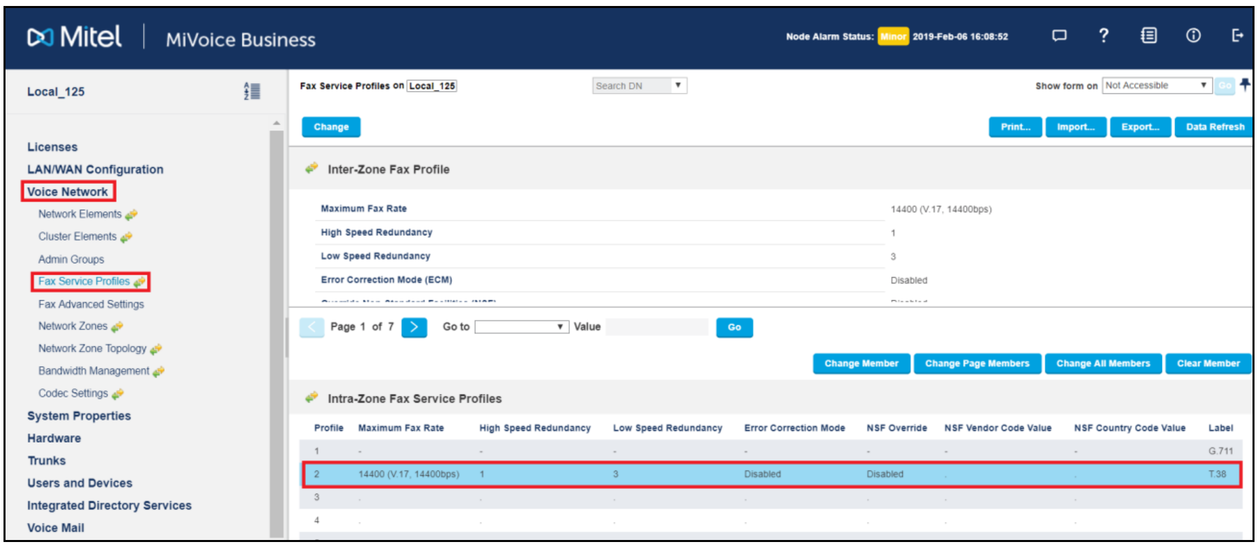


Figure 21 - Fax Configuration

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”. Service Provider NISC IVR uses the Intra-zone FAX Profile 2.

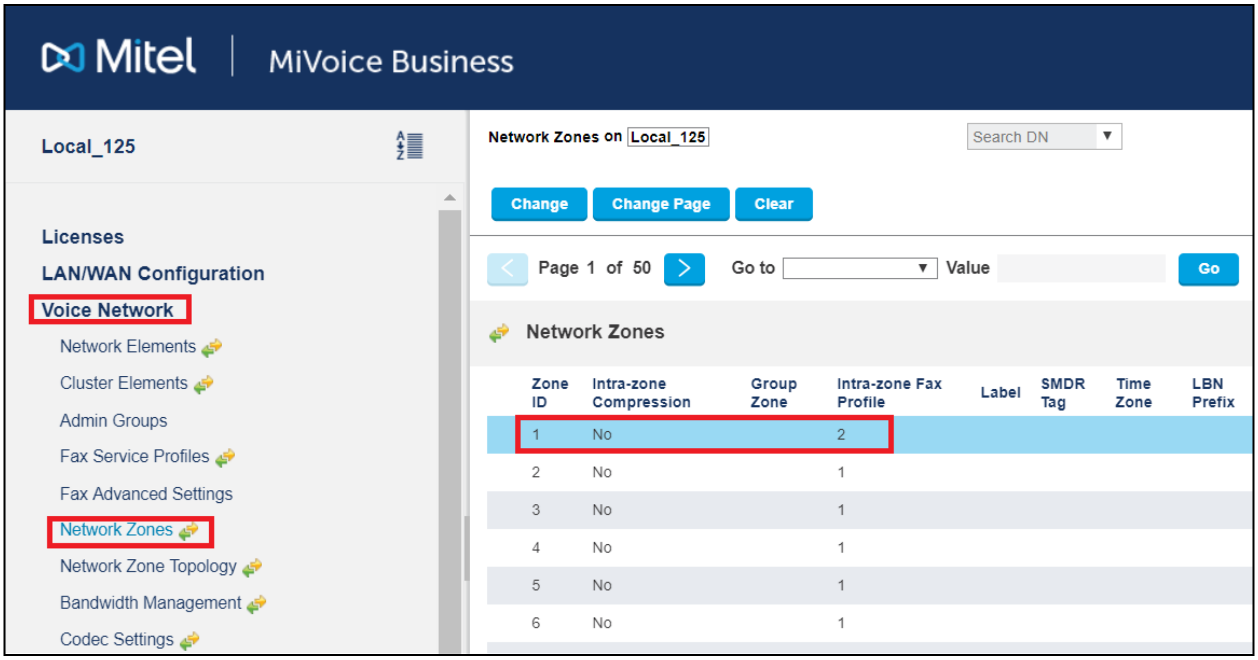


Figure 22 – Zone Assignment

Glossary

MiVoice Business	MiVB
MiVoice Border Gateway	MBG
MiVoice Call Recorder	MiVCR
Mitel Solutions Alliance	MSA
Knowledge Management System	KMS
Interoperability Reference Guide	IRG
Not Applicable	NA
Secure Recording Connector	SRC