Mitel Technical Configuration Notes – HO5481

March 3, 2025

MiVB - Configure MiVoice Business 10.2 for use with phonestar* SIP Trunking Without MBG

Description: This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel MiVB to connect to phonestar* SIP Trunking without MBG.

Environment: MiVoice Business 10.2 (10.2.0.54), Mitel 69xx/69xxw MiNET 03.00.00.050 and Mitel 69xx/68xx SIP 6.4.0.3022

NOTICE

TRADEMARKS

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Overview

This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel MiVB to connect to phonestar* SIP Trunking without MBG. 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	February, 2025	Initial interop certification of MiVoice Business 10.2 with phonestar* SIP Trunking without MBG

Interop Status

The Interop of phonestar* SIP Trunking has been given a Certification status. The phonestar* will be included in the Mitel Interoperability Reference Guide (IRG). The status of phonestar* SIP Trunking achieved is:

COMPATIBLE	The most common certification which means phonestar* SIP Trunking 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 phonestar* SIP Trunking and the MiVB without MBG.

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	10.2 (10.2.0.54)	NA
Mitel	69xx/69xxw MiNET	03.00.00.050	NA
Mitel	69xx/68xx SIP	6.4.0.3022	NA

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 through Service Provider phonestar* and their PSTN gateway, call holding, transferring, conferencing, busy calls, long calls durations	☑
Automatic Call Distribution	Making calls to an ACD environment with RAD treatments, Interflow and Overflow call scenarios and DTMF detection	☑
DTMF Signal	Sending DTMF after call setup (i.e. mailbox password)	☑
Personal Ring Groups	Receiving calls through Service Provider phonestar* and their PSTN gateway to a personal ring group. Also moving calls to/from the prime member and group members	☑
Packetization	Forcing the Mitel MiVB to stream RTP packets at different intervals, from 10ms to 30ms	☑
External Hot Desking	Receiving calls through Service Provider phonestar* and their PSTN gateway to PRG with EHDU. Including moving calls to/from the prime member of the PRG with the EHDU	☑
Codec	Different codecs support (G711 and G722)	☑
TLS/SRTP	Making and receiving calls through Service Provider phonestar* via TLS/SRTP	☑
Fax	G711 & T.38 Fax Calls	☑
Video	Making and receiving a call through Service Provider phonestar* with video capable devices	✗
Resiliency	Fail-over and Fail-back scenarios between MiVB, MBG and the service provider phonestar* SBC setup	NA

☑ - No issues found ✗ - Issues found, cannot recommend using ⚠ - Issues found NA – Not Applicable

Device Limitations and Known Issues

This is a list of problems or unsupported features when phonestar* SIP Trunk is connected to the MiVB.

Feature	Problem Description
Video Calls	phonestar* does not support video calling. **Recommendation**: Please contact phonestar* support for updates for supporting of the video calling.
Codec (G729)	phonestar* does not support G729 codec calling. **Recommendation**: Please contact phonestar* support for more information on this.
Resiliency	phonestar* does not support DNS SRV. For redundancy, phonestar* uses a virtual IP concept which failovers to another host. **Recommendation**: Please contact phonestar* support for getting more information on this.
Long Duration Calls	The long-duration calls are failing with BYE from phonestar* side after 30mins when enabling session timers on the MiVB and recommending that to disable session timers on the MiVB in order to avoid issues. **Recommendation**: Please disable session timer (set value to "0") on the MiVB or contact phonestar* support for more information on this.
Supervised Transfer	In case of incoming PSTN call to Mitel Phone (DID) and Mitel Phone doing the supervised transfer the call to another PSTN user scenario is failing and observed two-way audio issues after the call is transferred over G722/Opus codecs. Hence, please restrict the codec to G711 on the MiVB SIP Peer Profile in order to work successfully. **Recommendation**: Please contact phonestar* support for getting more information on this.
Blind Transfer	In case of incoming PSTN call to Mitel Phone (DID) and Mitel Phone doing the blind transfer the call to another PSTN user scenario is failing and observed two-way audio issues after the call is transferred over G722/Opus codecs. Hence, please restrict the codec to G711 on the MiVB SIP Peer Profile in order to work successfully. **Recommendation**: Please contact phonestar* support for getting more information on this.
PRACK	phonestar* does not support PRACK and responding with 420 Bad Extension as Unsupported: 100rel. **Recommendation**: Please contact phonestar* support for getting more information on this.

Network Topology

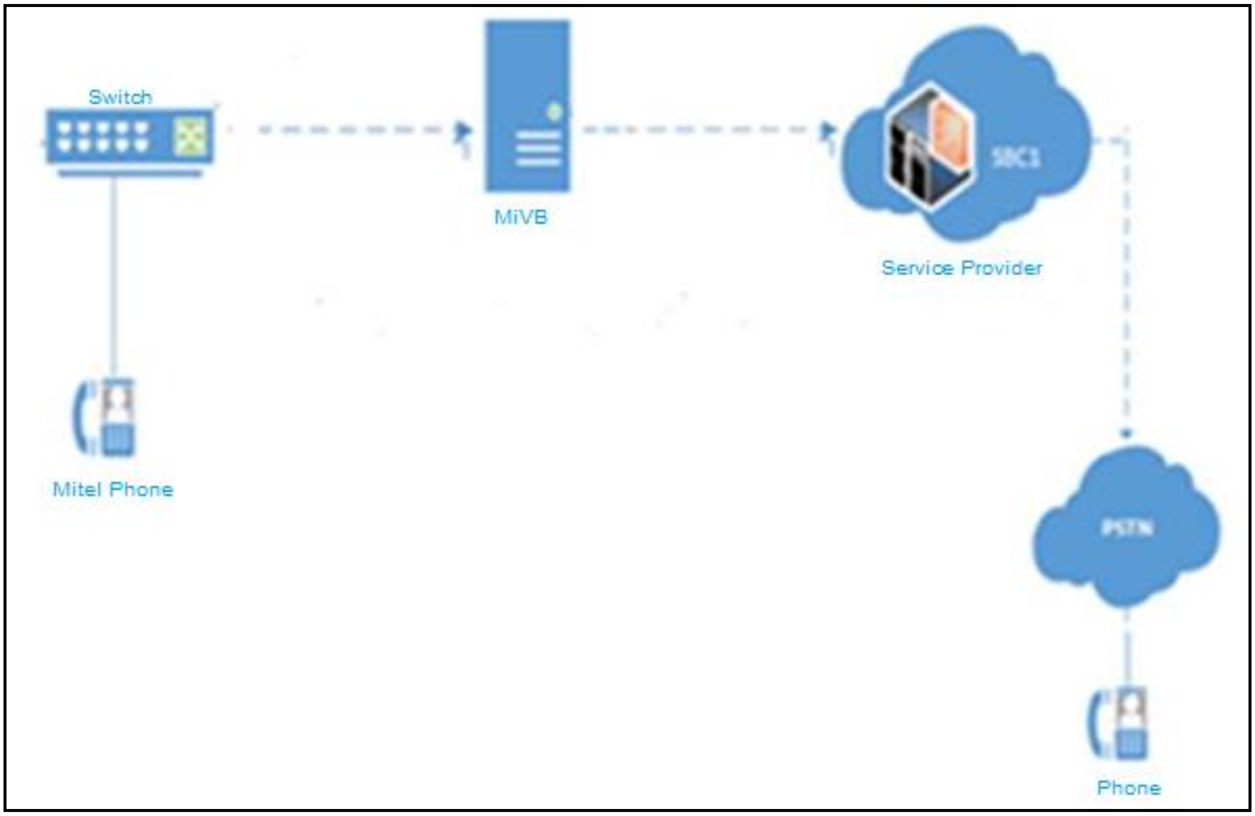


Figure 1 – Network Topology

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 phonestar* SIP Trunking 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 phonestar* with or without an asterisk SIP Trunking without MBG.

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 phonestar* with or without an asterisk 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.

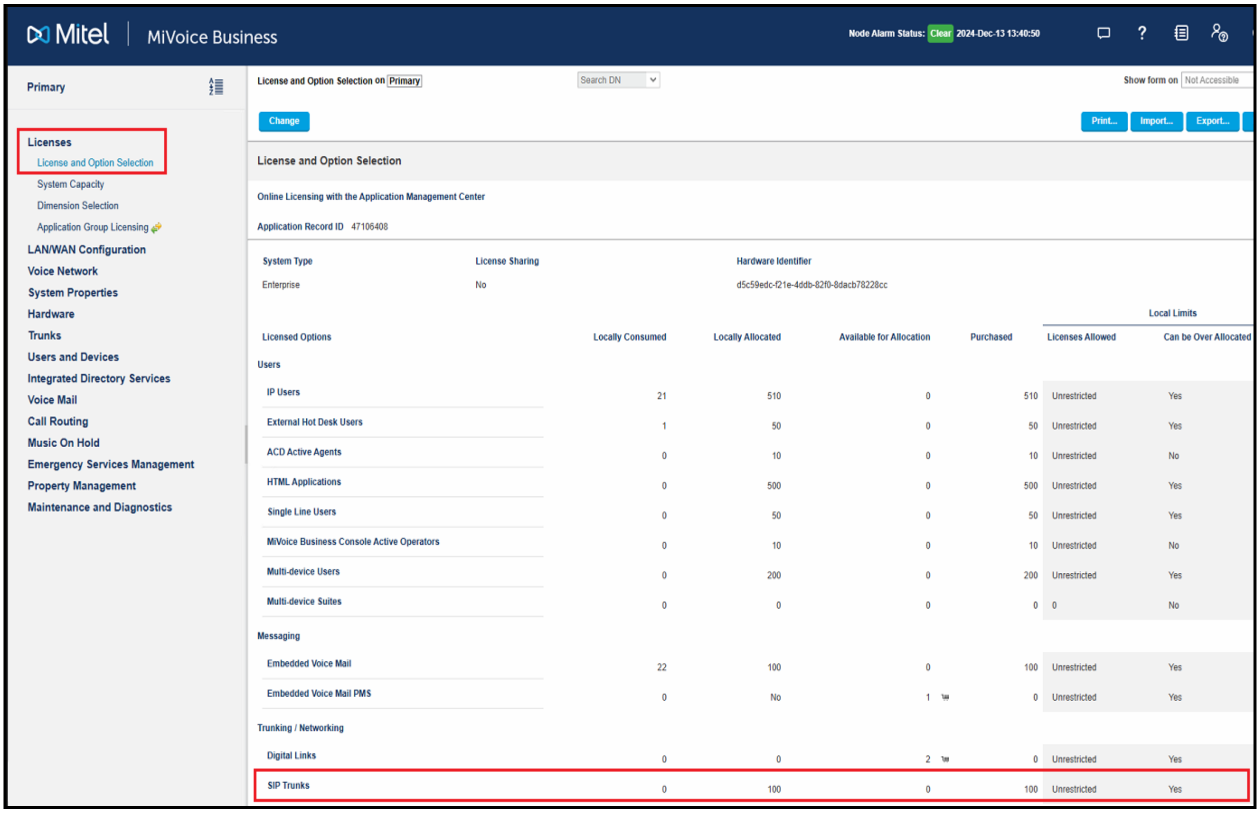


Figure 2 – License and Option Selection

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 Capable set to Yes

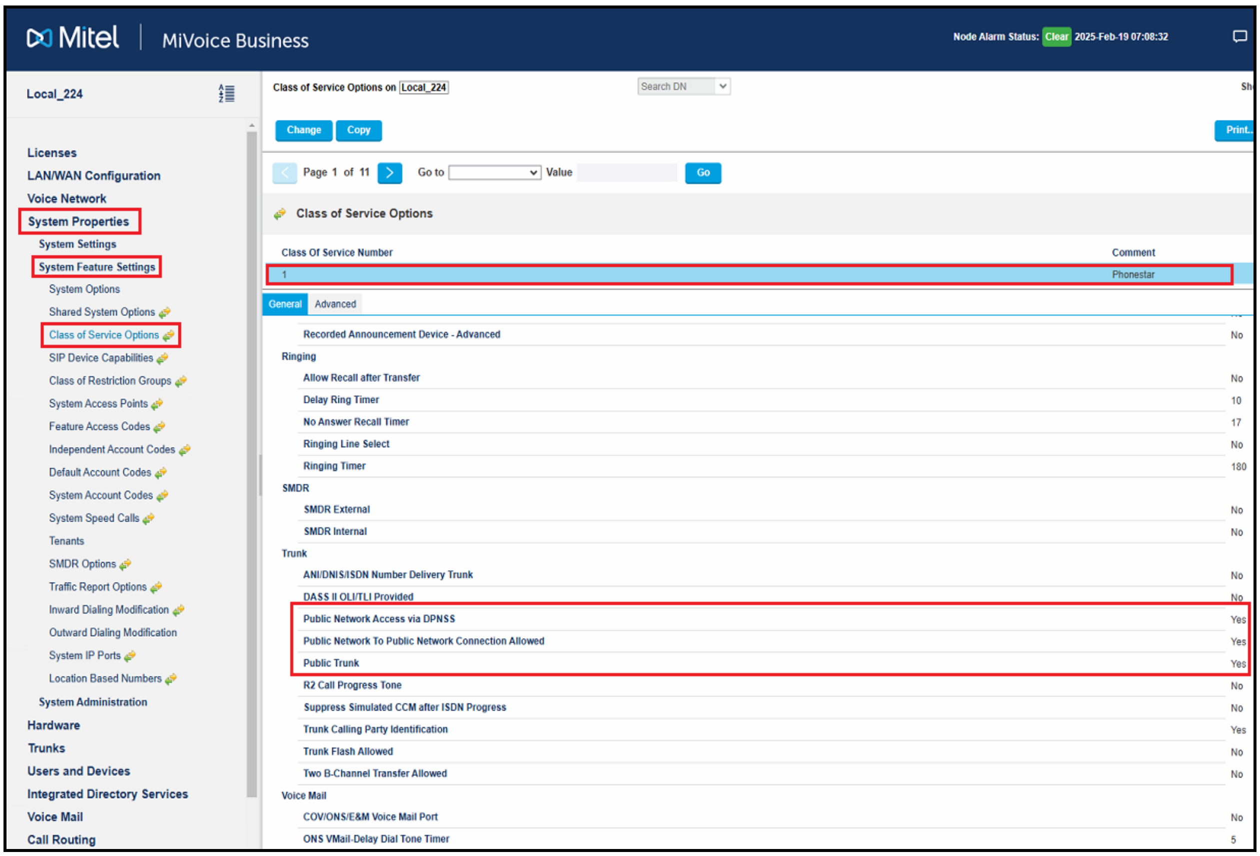


Figure 3 – Class of Service

Network Element Assignment

Create a network element for Service Provider phonestar* with or without an asterisk SIP Trunking. In this example, the soft switch is reachable by an IP Address/FQDN and is defined as “phonestar“ 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 service provider phonestar* 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.

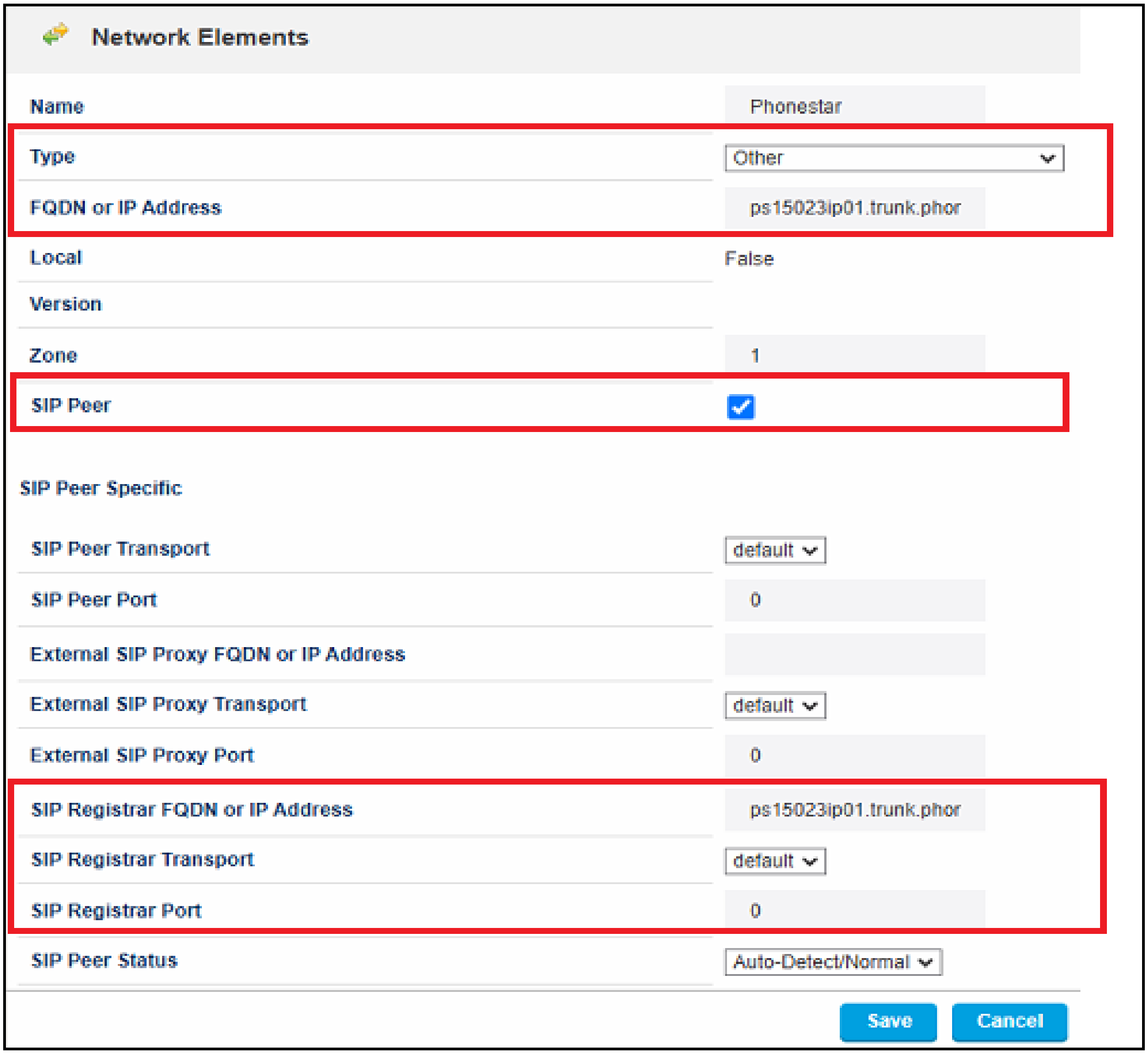


Figure 4 – Network Element Assignment

Trunk Attributes

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

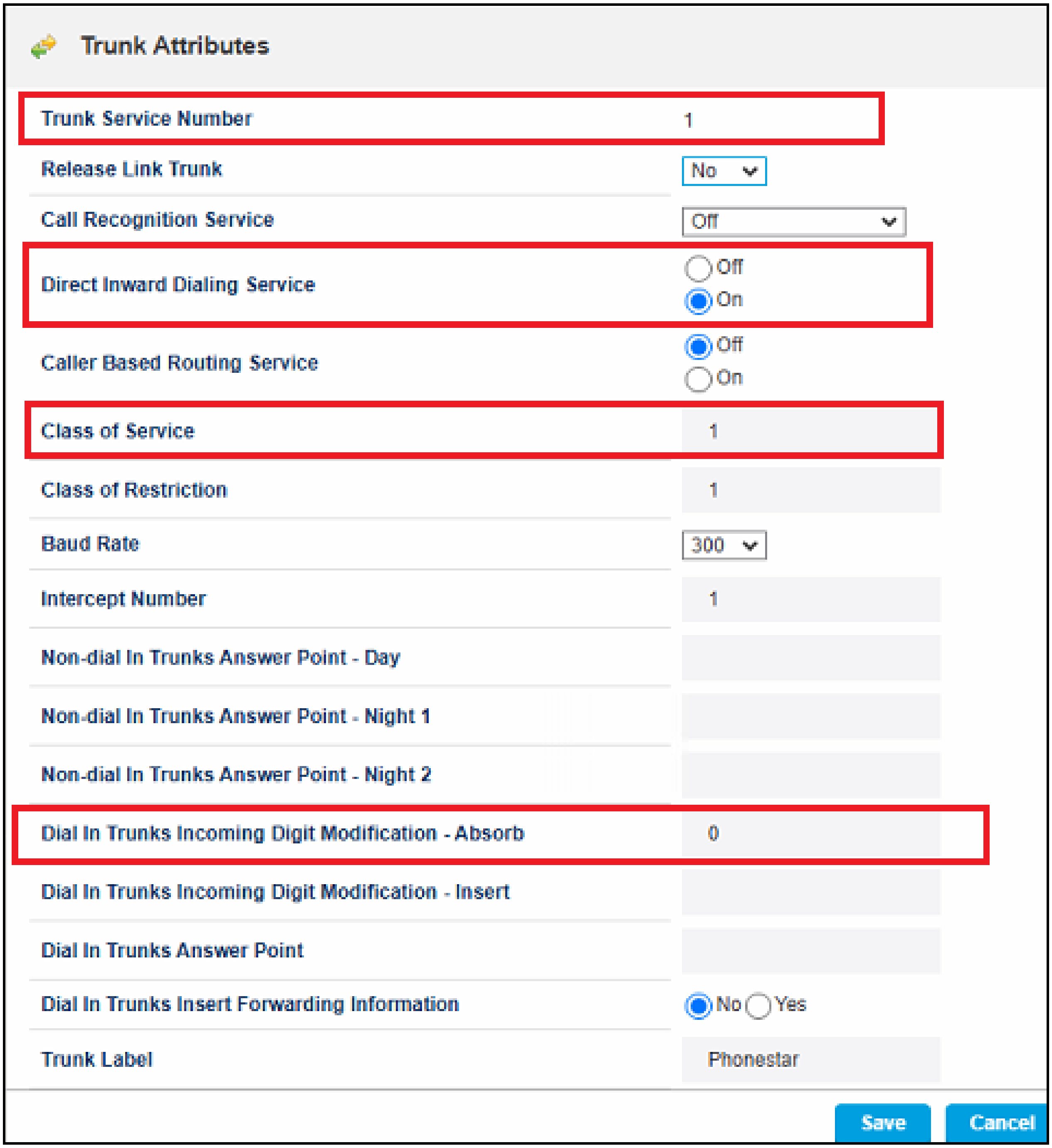


Figure 5 – Trunk Attributes

SIP Peer Profile

The recommended connectivity via SIP trunking does not require additional physical interfaces. IP/Ethernet connectivity is a 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 "phonestar" Network Element.

Registration User Name: Provide SIP trunk user name that given by phonestar.

Address Type: Select IP address.

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. 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 phonestar*.

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

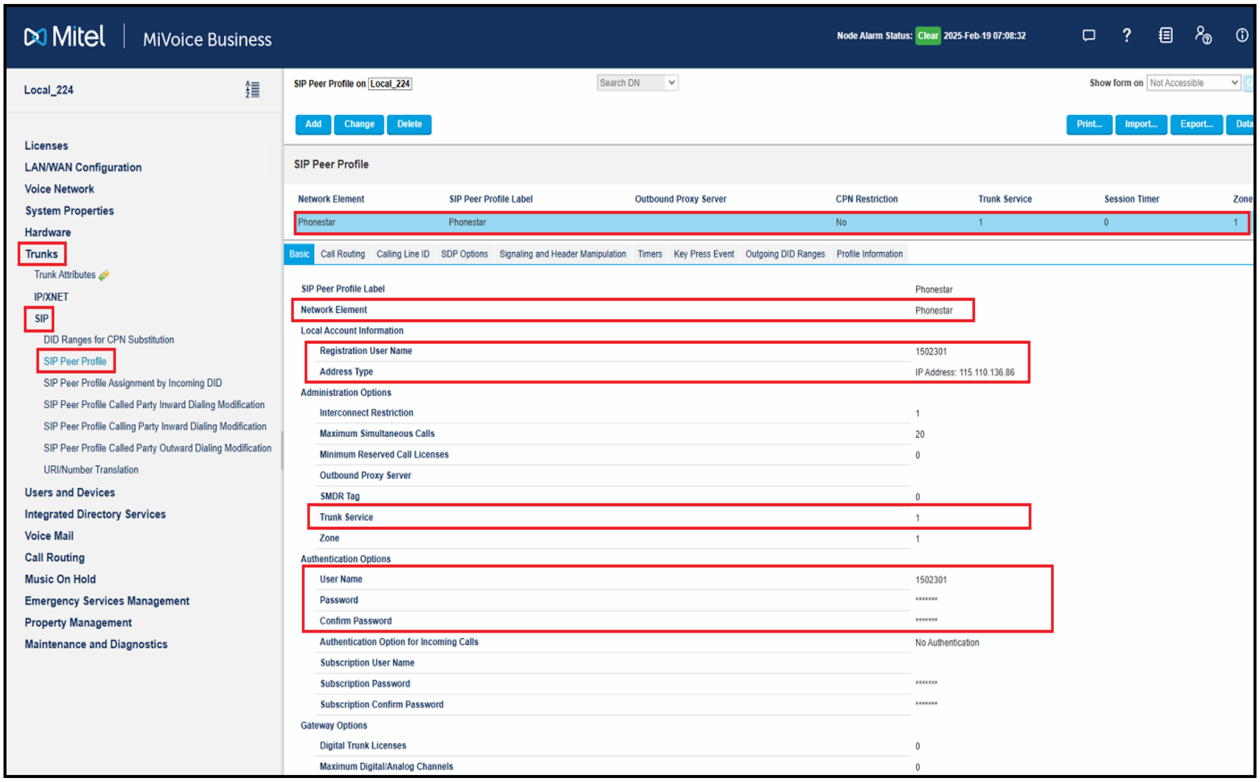


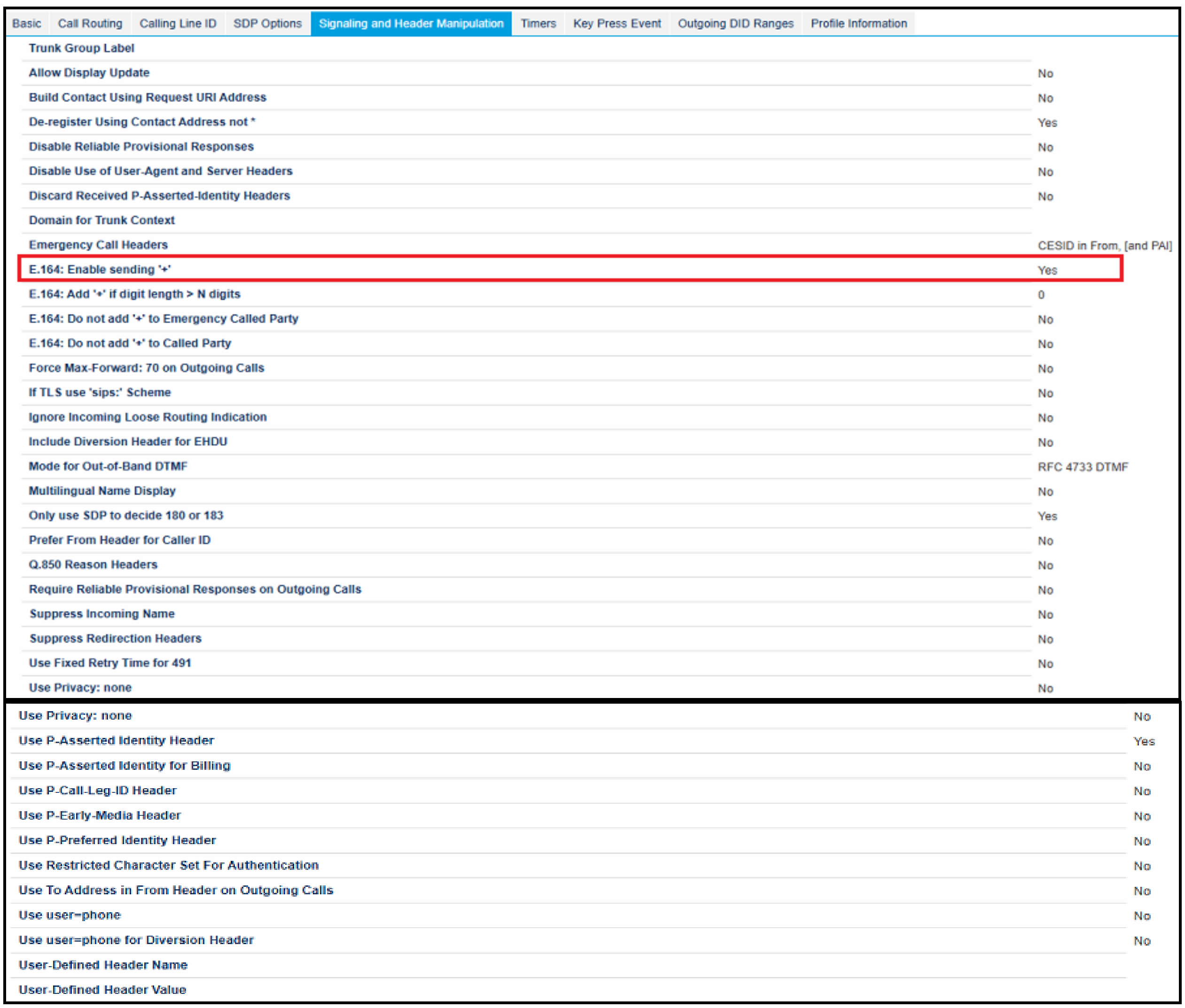
Figure 6 – SIP Peer Profile Assignment - Basic

Figure 7 – SIP Peer Profile Assignment - Call Routing

Figure 8 – SIP Peer Profile Assignment - Calling Line ID

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	Yes
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



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	
Emergency Call Headers	CESID in From, [and PAI]
E.164: Enable sending '+'	Yes
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
Prefer From Header for Caller ID	No
Q.850 Reason Headers	No
Require Reliable Provisional Responses on Outgoing Calls	No
Suppress Incoming Name	No
Suppress Redirection Headers	No
Use Fixed Retry Time for 491	No
Use Privacy: none	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
User-Defined Header Name	
User-Defined Header Value	

Figure 10 – SIP Peer Profile Assignment - Signaling and Header Manipulation

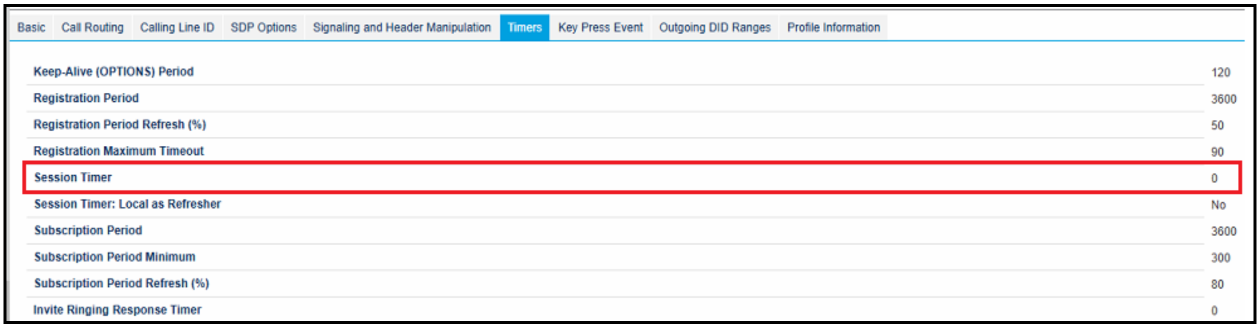


Figure 11 – SIP Peer Profile Assignment – Timers

Figure 12 – SIP Peer Profile Assignment - Key Press Event

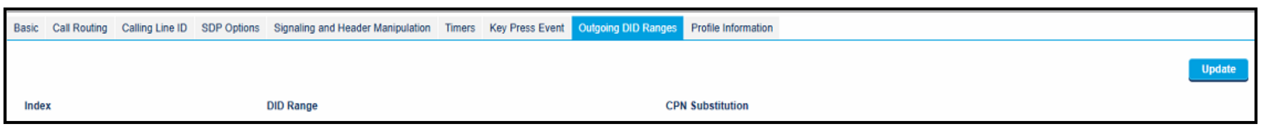


Figure 13 – SIP Peer Profile Assignment - Outgoing DID Ranges

Figure 14 – 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 phonestar* with or without an asterisk 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, "41445896887,41916896751"). 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.

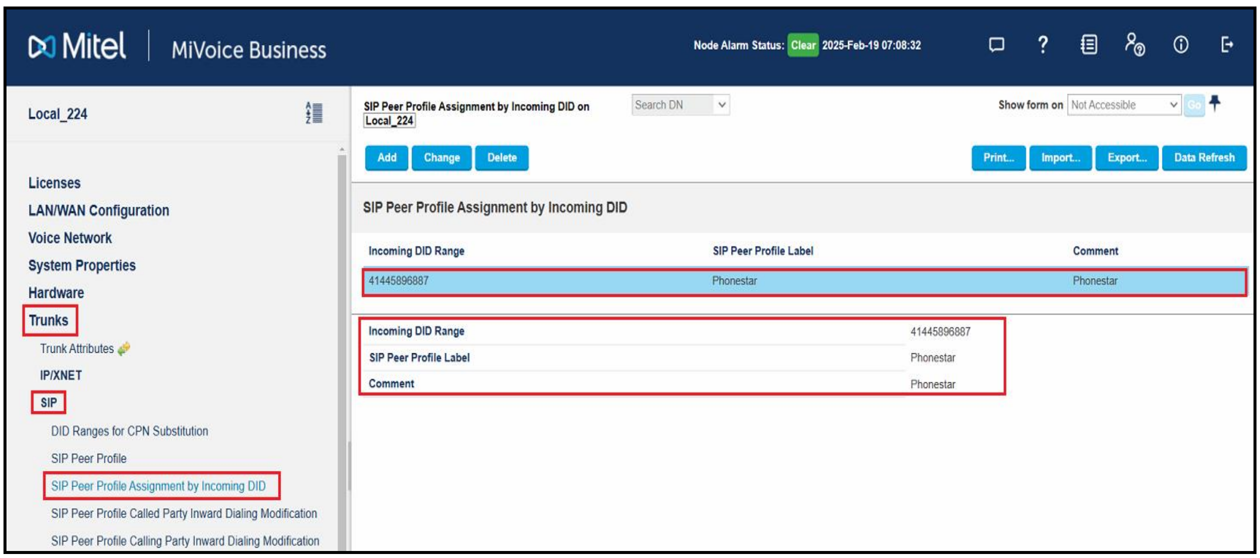


Figure 15 – SIP Peer Profile Assignment by Incoming DID

ARS Digit Modification Plans

Ensure that Digit Modification for outgoing calls on the SIP trunk to phonestar* SIP Trunking absorbs or injects additional digits according to your dialling plan. In this example, we will be absorbing 3 digits.

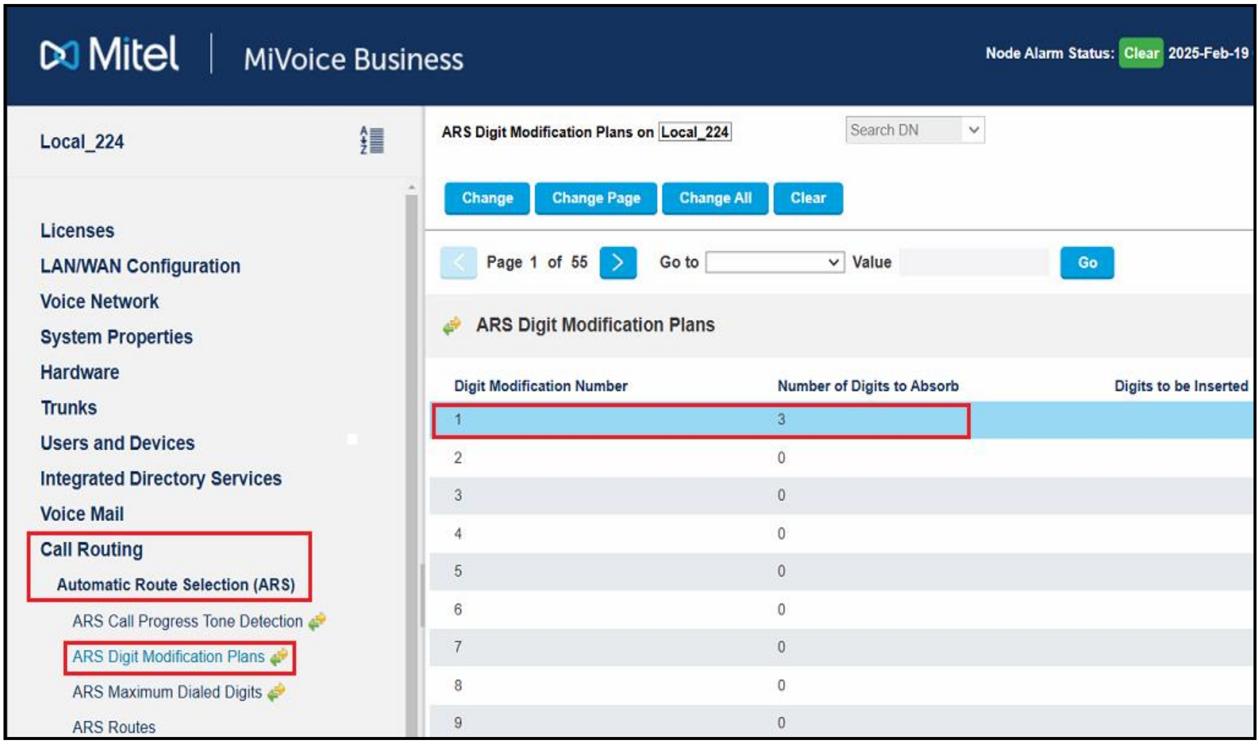


Figure 16 – Digit Modification Assignment

ARS Routes

Create a route for SIP trunks connecting a trunk to phonestar* with or without an asterisk SIP Trunking. In this example, the SIP trunk is assigned to Route Number **1**. Choose SIP Trunk as a routing medium and choose the SIP Peer Profile and Digit Modification entry created earlier.

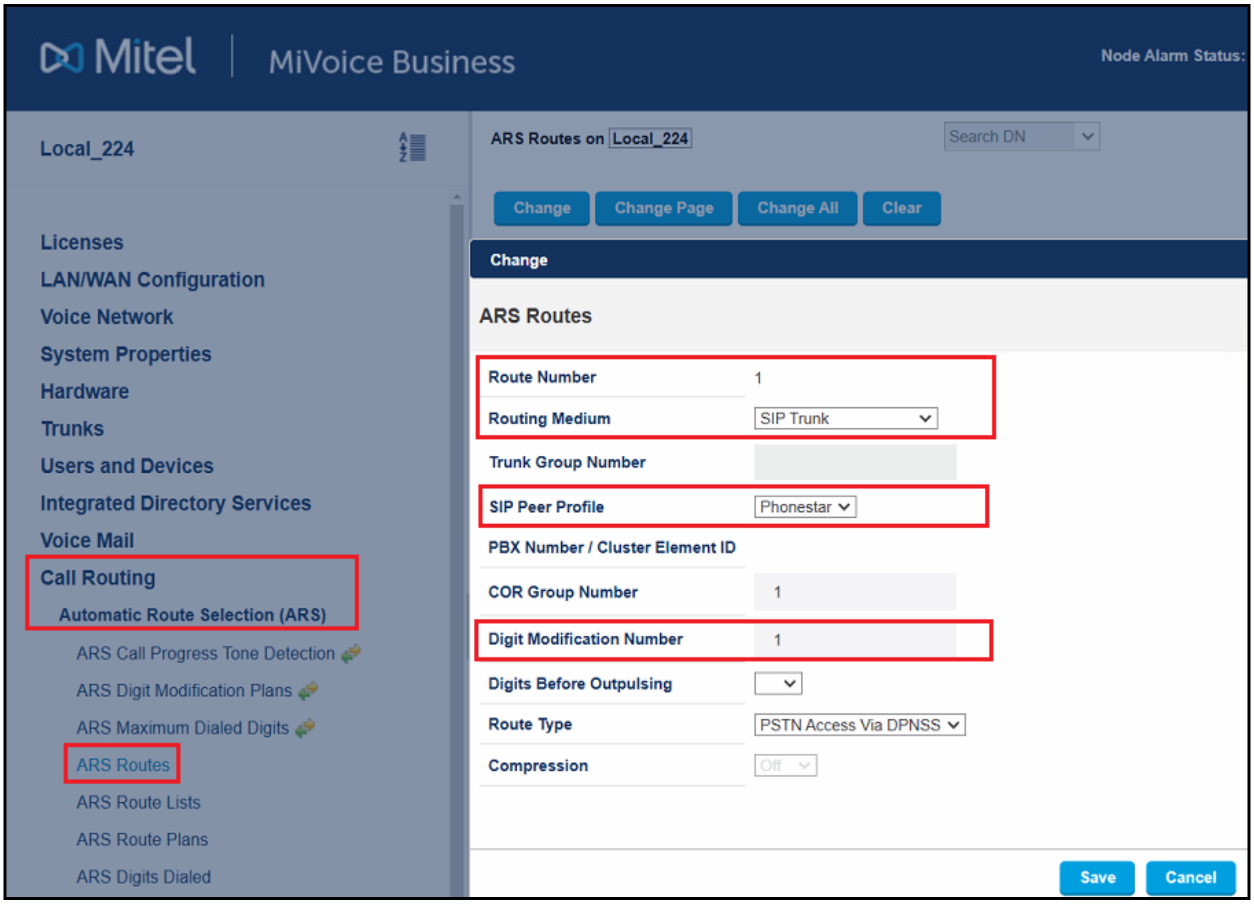


Figure 17 – SIP Trunk Route Assignment

ARS Digits Dialed

ARS initiates the routing of SIP trunk calls when certain digits are dialed from a station. In this example, when a user dials 111918123347168, the call will be routed to phonestar* SIP Trunking.

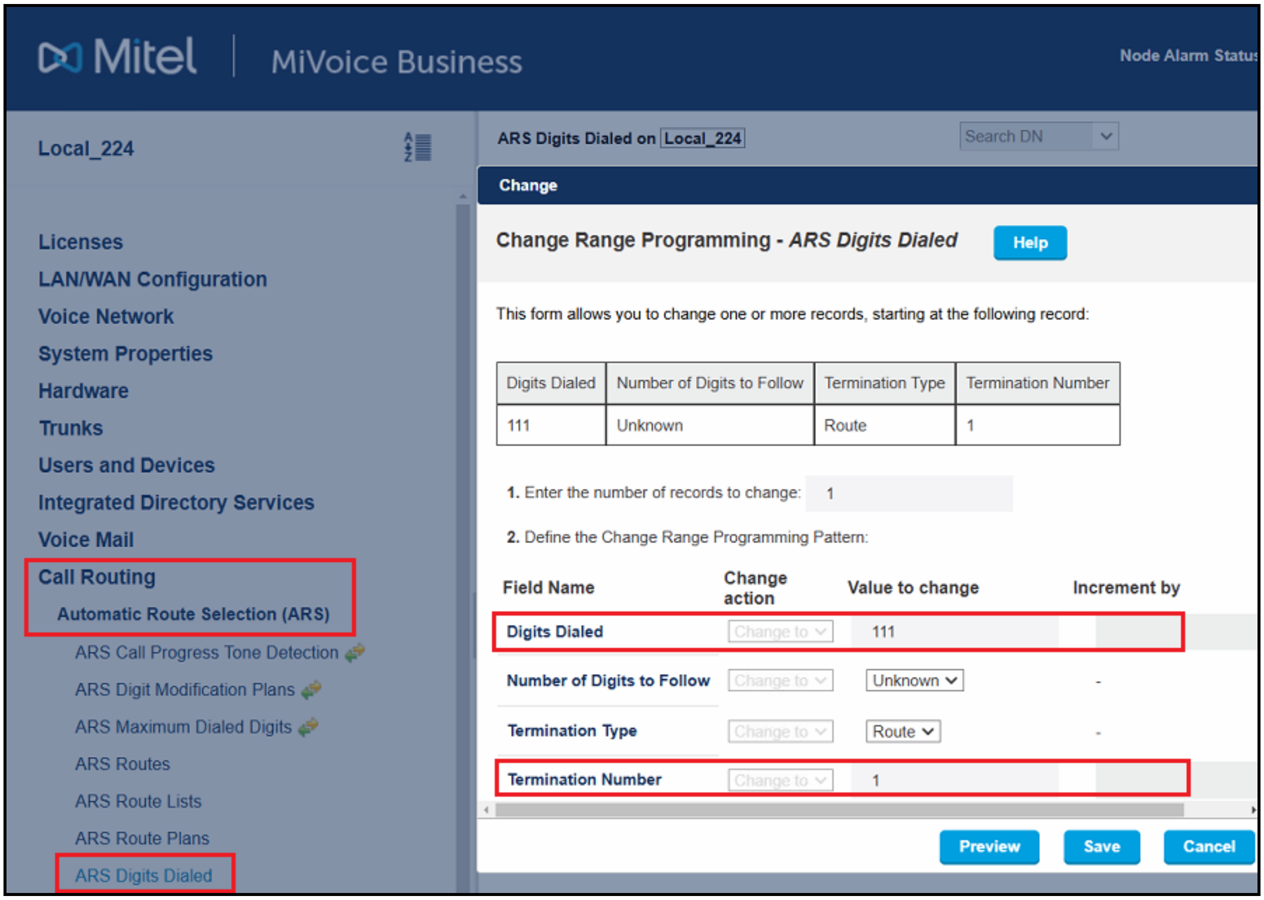


Figure 18 – ARS Digit Dialed Assignment

<u>**Note**</u> – This configuration is just a reference. Customers 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.

Fax Configuration

Service Provider phonestar* with or without an asterisk 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).

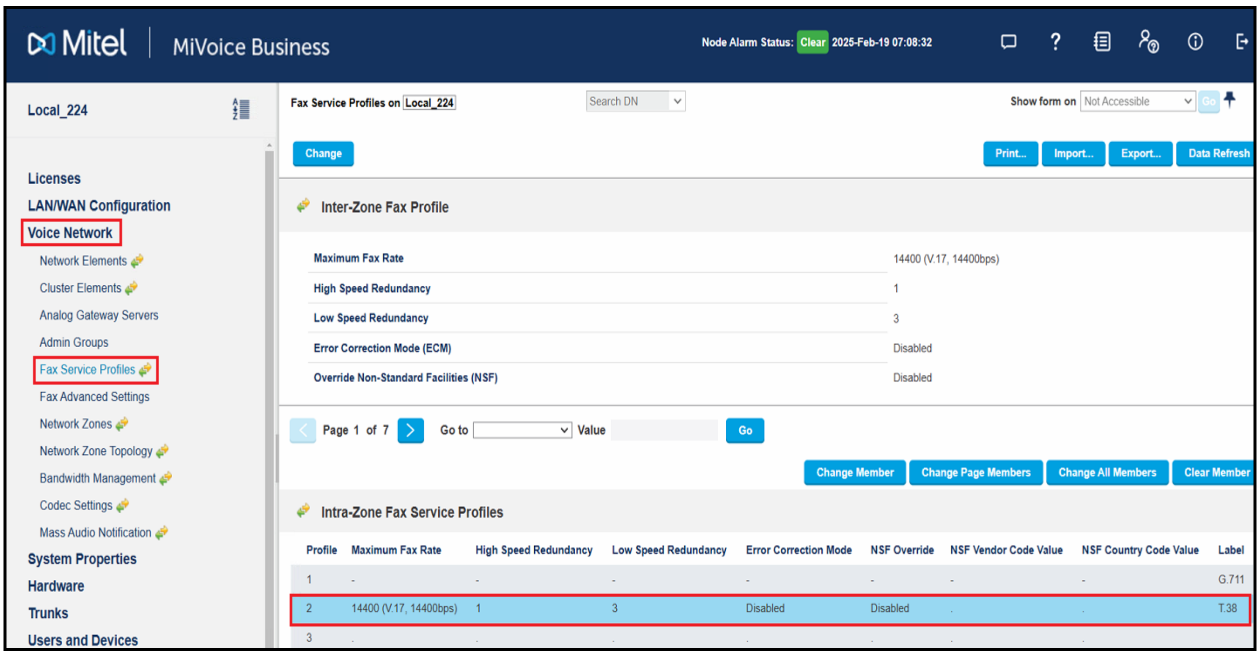


Figure 19 - Fax Configuration

Glossary

MiVoice Business	MiVB
MiNET Interface	MiNET
Mitel Solutions Alliance	MSA
Knowledge Management System	KMS
Class of Service	COS
Automatic Route Selection	ARS
Personal Ring Group	PRG
External Hot Desk User	EHDU
Automatic Call Distribution	ACD