TELNET

TelNet Worldwide, Inc.
telnetww.com
1-833-4TELNET

Mitel

Mitel Technical Configuration Notes – HO858

rev. 2018-12-12

Configure MiVoice Business 9.0 for use with TelNet Worldwide SIP Trunking Using MBG

Description:
This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel MiVoice Business to connect to Service Provider TelNet Worldwide SIP Trunking.

Environment:
MiVoice Business 9.0 (9.0.0.184), MiVoice Border Gateway 10.1.0.244, Mitel 69xx Phone 01.04.00.074 and Mitel 68XX Phone 5.1.0.227

6050-10050-123456-R0714-EN

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 MiVoice Business for use with TelNet Worldwide SIP Trunking using MBG

Nov 2018, HO858

Table of Contents

Overview

This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel MiVB to connect to Service Provider TelNet Worldwide SIP Trunking. 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	09-Nov-2018	Initial Interop with Mitel MiVB 9.0 and Service Provider TelNet Worldwide SIP trunk using MBG

Interop Status

The Interop of Service Provider TelNet Worldwide SIP Trunking has been given a Certification status. This service provider or trunking device will be included in the SIP CoE Reference Guide. The status Service Provider TelNet Worldwide SIP Trunking achieved is:

COMPATIBLE	The most common certification which means Service Provider TelNet Worldwide SIP Trunking has been tested and/or validated by the Mitel SIP CoE team. 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 Service Provider TelNet Worldwide SIP Trunking and the MiVoice Business.

Manufacturer	Variant	Software Version
Mitel	MiVoice Business	Release 9.0 (9.0.0.184)
Mitel	MiVoice Border Gateway	10.1.0.244
Mitel	69XX	01.04.00.074
Mitel	68XX	5.1.0.227
BroadSoft	Broadworks	V20SP1
Oracle SBC	NetNet 4500	7.2.0 MR-6 Patch 9

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 TelNet Worldwide and their PSTN gateway, call holding, transferring, conferencing, busy calls, long calls durations, variable codec.	☑
Automatic Call Distribution	Making calls to an ACD environment with RAD treatments, Interflow and Overflow call scenarios and DTMF detection.	☑
Nu-Point Voicemail	Terminating calls to a Nu-Point voicemail boxes and DTMF detection.	☑
Packetization	Forcing the Mitel MiVB to stream RTP packets through its E2T card at different intervals, from 10ms to 40ms	☑
Personal Ring Groups	Receiving calls through Service Provider TelNet Worldwide and their PSTN gateway to a personal ring group. Also moving calls to/from the prime member and group members.	☑
External Hot Desking	Receiving calls through Service Provider TelNet Worldwide and their PSTN gateway to PRG with EHDU. Including moving calls to/from the prime member of the PRG with the EHDU. Also placing calls from the EHDU and using mid call features with EHDU.	☑
Teleworker	Making and receiving a call Service Provider TelNet Worldwide and their PSTN gateway to and from Teleworker extensions.	☑
Video	Making and receiving a call through Service Provider TelNet Worldwide with video capable devices.	✗
Fax	T.38 and G711 Fax Calls	☑
G722/Wideband Codec	Making and receiving a call through Service Provider TelNet Worldwide using G722 codec.	✗
E.164 calling	Make calls using E.164 format	☑

☑ - 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 Service Provider TelNet Worldwide SIP Trunking is connected to the MIVB.

Feature	Problem Description
Packetization	TelNet Worldwide does not support Packetization of 40 MS **Recommendation:** Contact TelNet Worldwide for more information.
TLS	TelNet Worldwide does not support TLS **Recommendation:** Contact TelNet Worldwide for more information.
Codec G722/G722.1	TelNet Worldwide does not support wideband G722/G722.1 Codec. **Recommendation:** Contact TelNet Worldwide for more information.
Video Calls	TelNet Worldwide does not support video calls. **Recommendation:** Contact TelNet Worldwide for more information.
DTMF (INFO)	DTMF via SIP INFO is not Supported by TelNet Worldwide **Recommendation:** Contact TelNet Worldwide for more information.
Blind Transfer	A delay of 2s is noticed for an inbound PSTN call transferred back to another PSTN number. Forcing the p-time value to 20 MS on MiVB reduces the delay

Network Topology

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

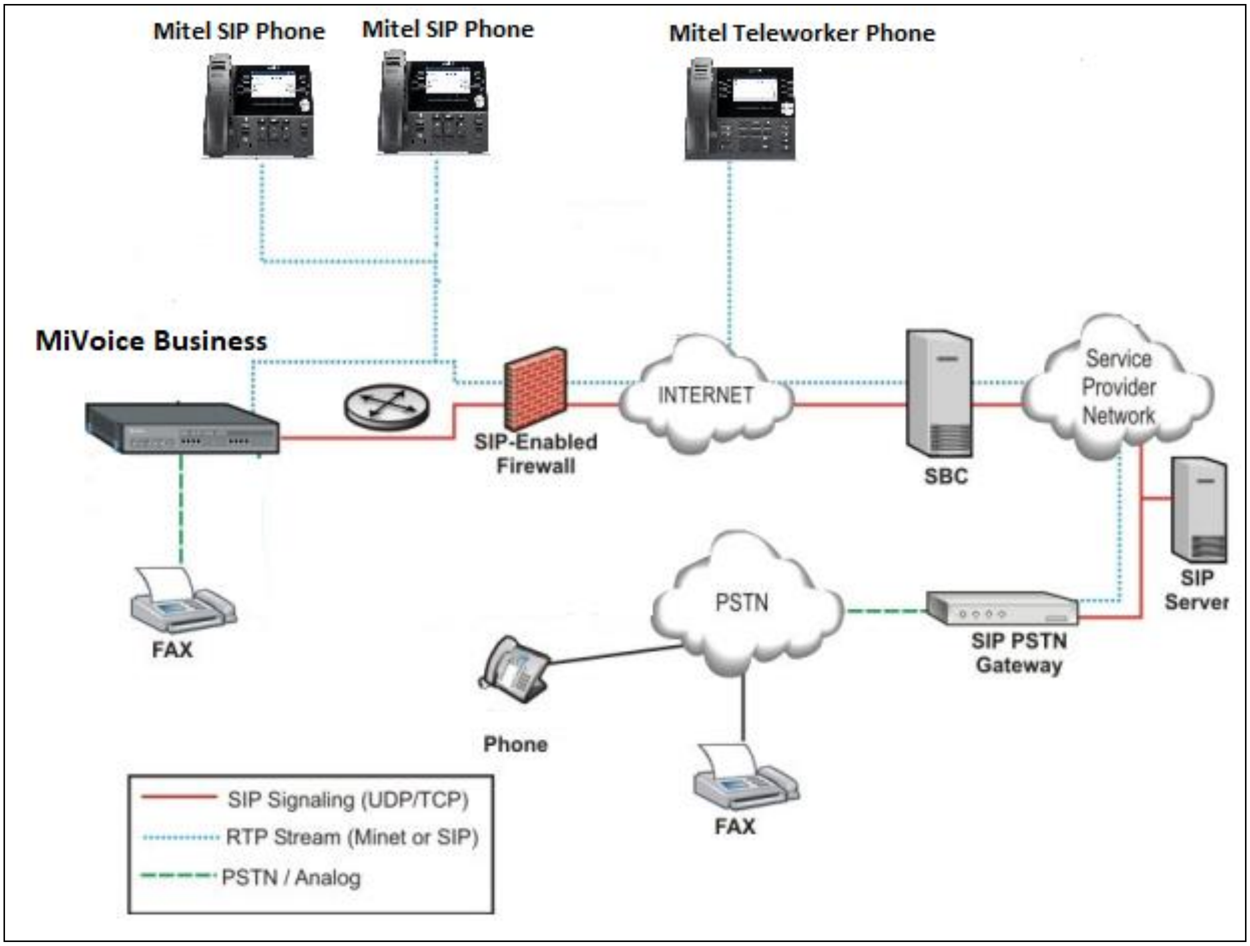


Figure 1 – Network Topology

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 Service Provider TelNet Worldwide SIP Trunking 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 TelNet Worldwide SIP Trunking.

Configuration Template

A configuration template can be found in the same MOL Knowledge Base 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 Service Provider TelNet Worldwide 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.

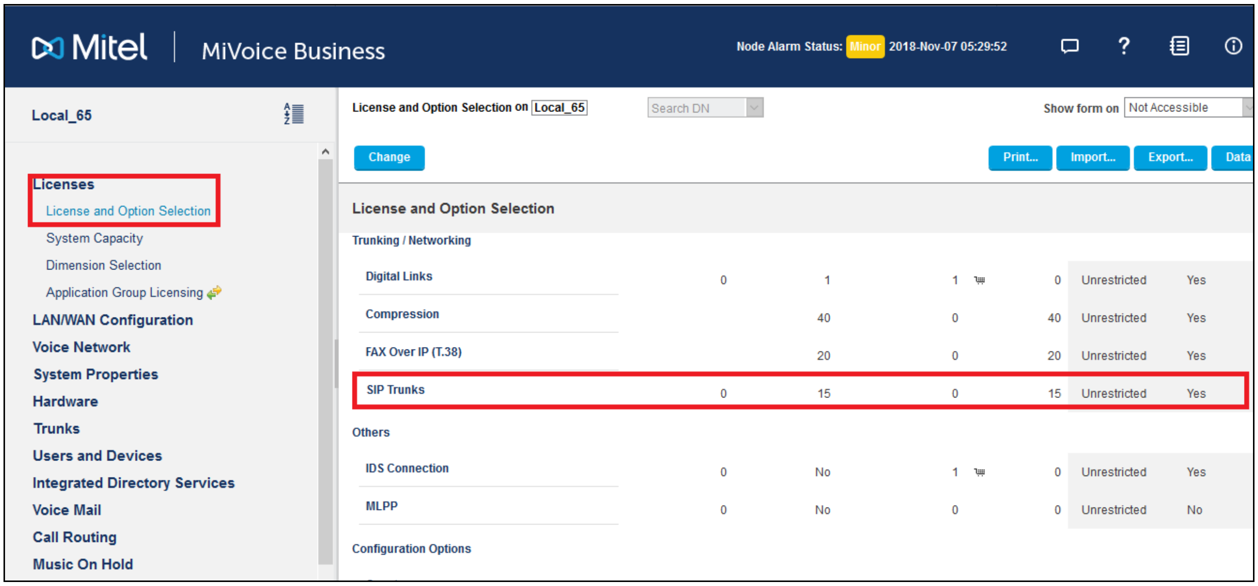


Figure 2 – License and Option Selection

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**

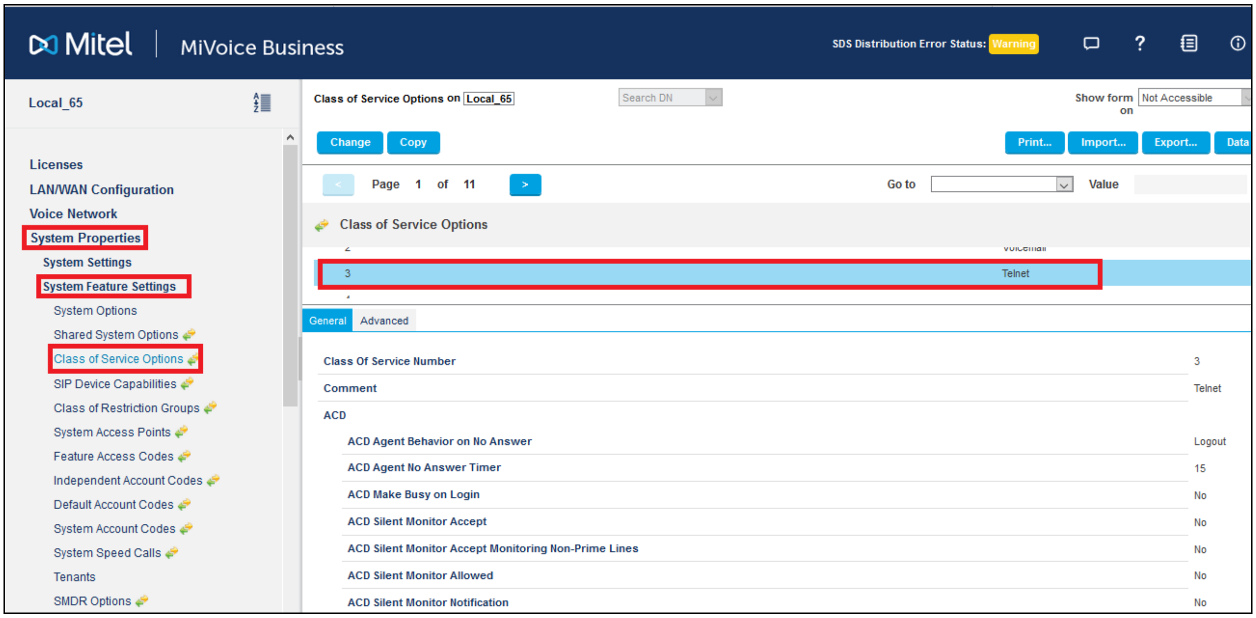


Figure 3 – Class of Service

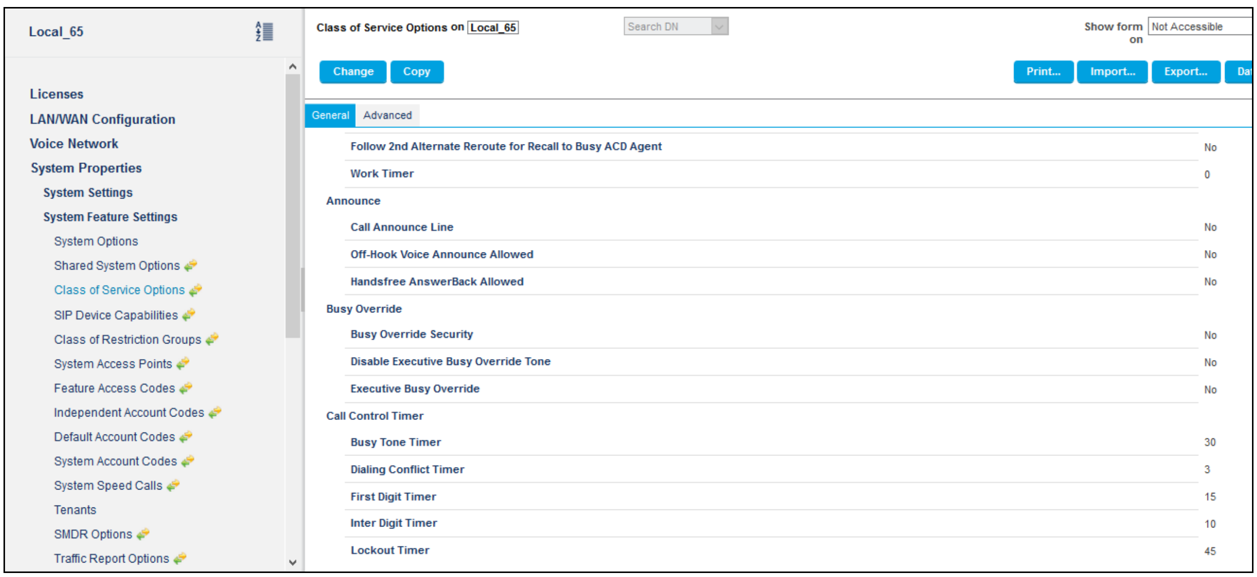


Figure 4 – Class of Service General

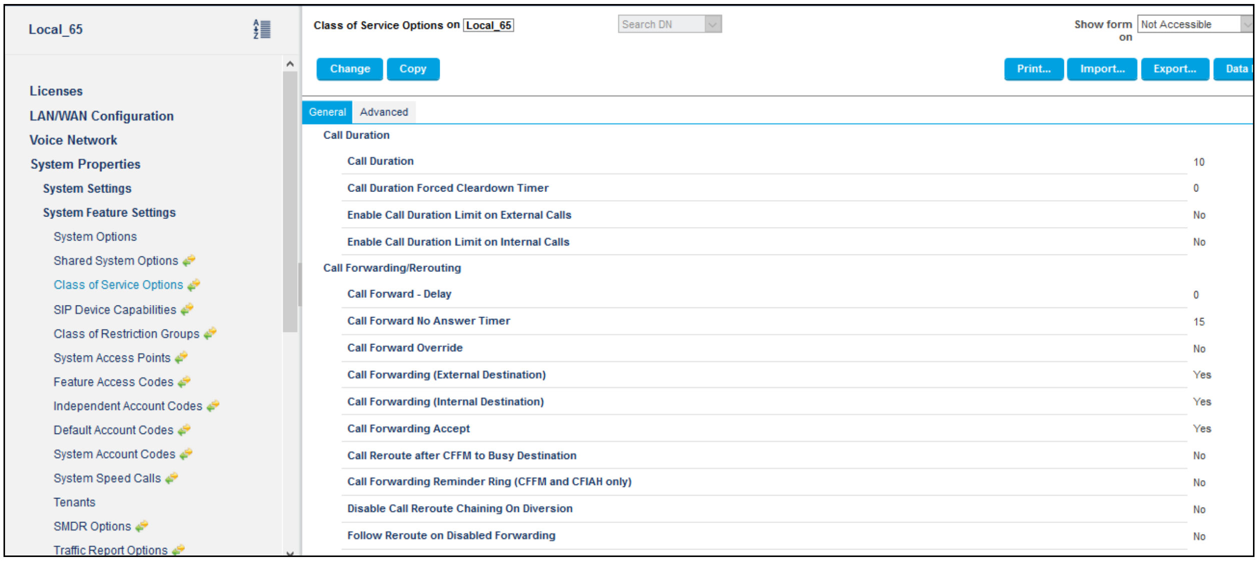


Figure 5 – Class of Service General

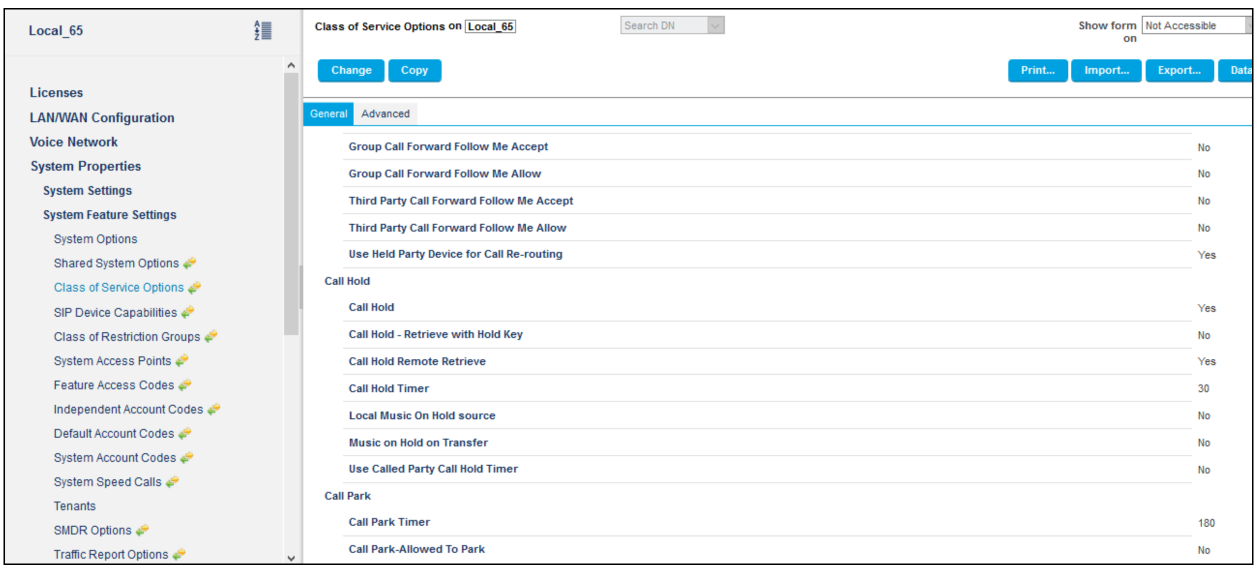


Figure 6 – Class of Service General

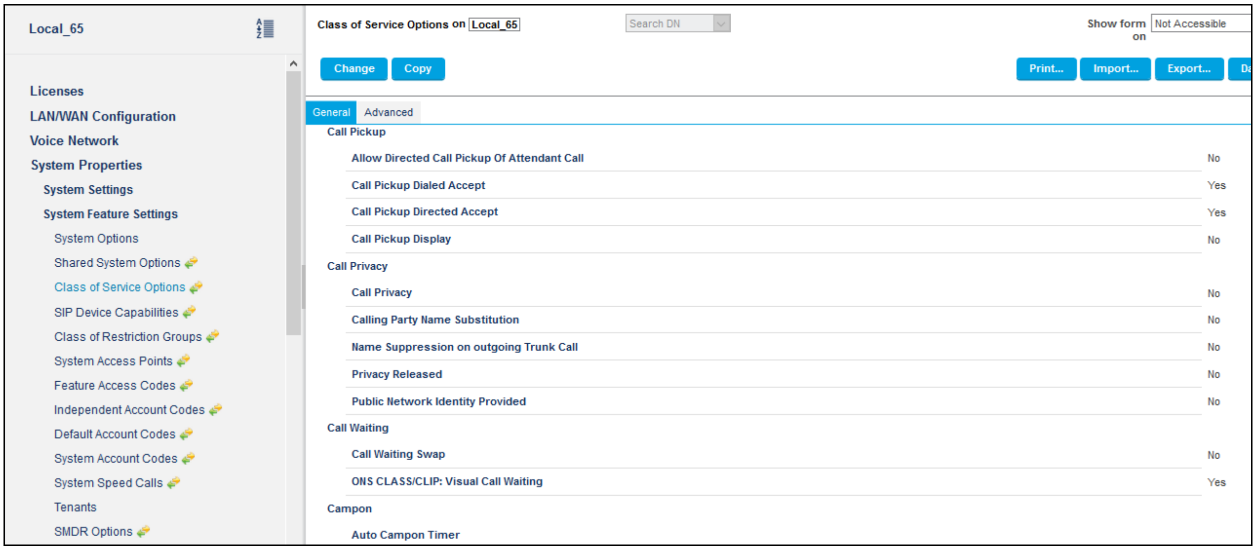


Figure 7 – Class of Service General

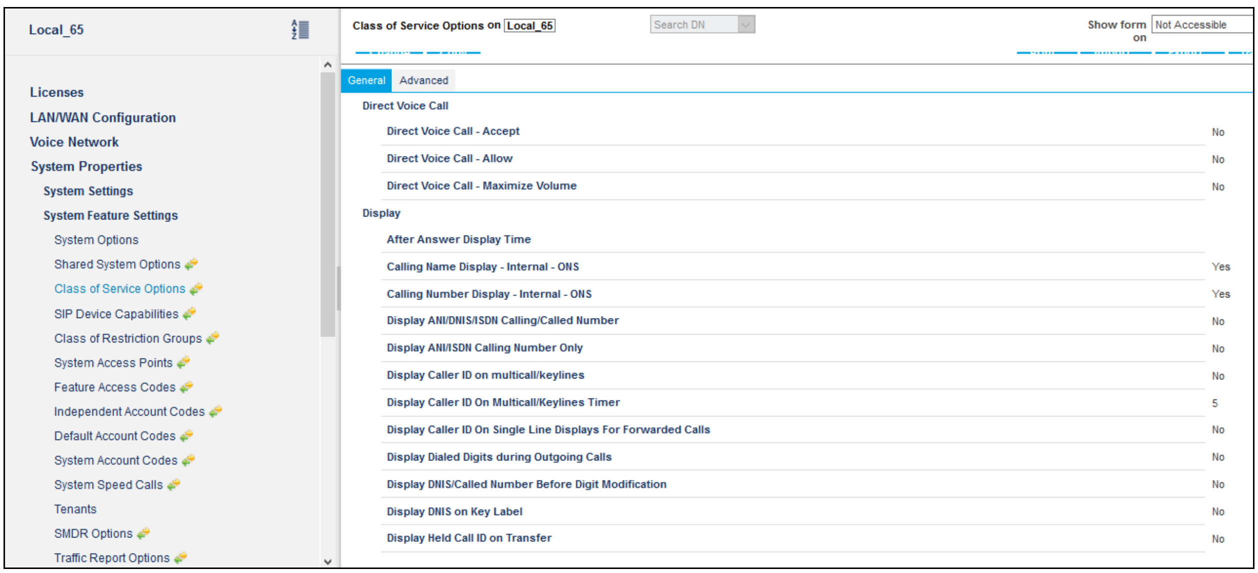


Figure 8 – Class of Service General

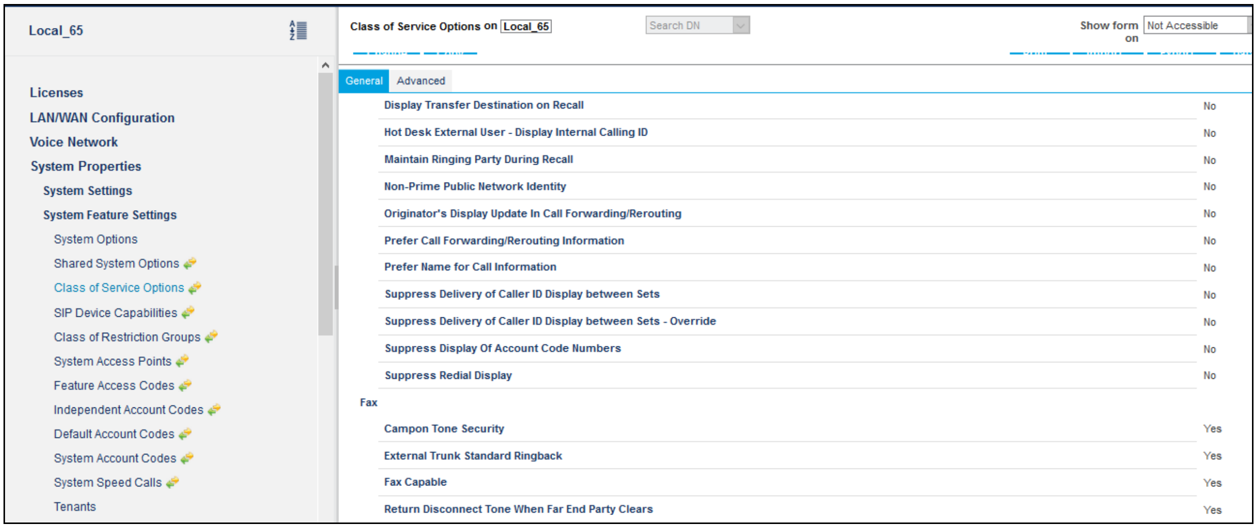


Figure 9 – Class of Service General

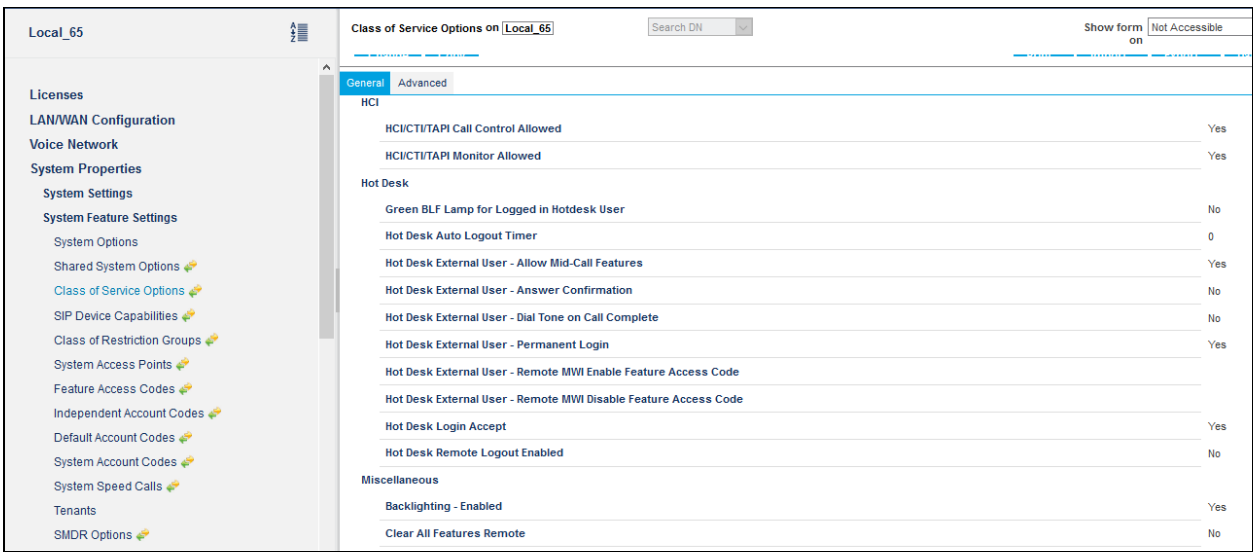


Figure 10 – Class of Service General

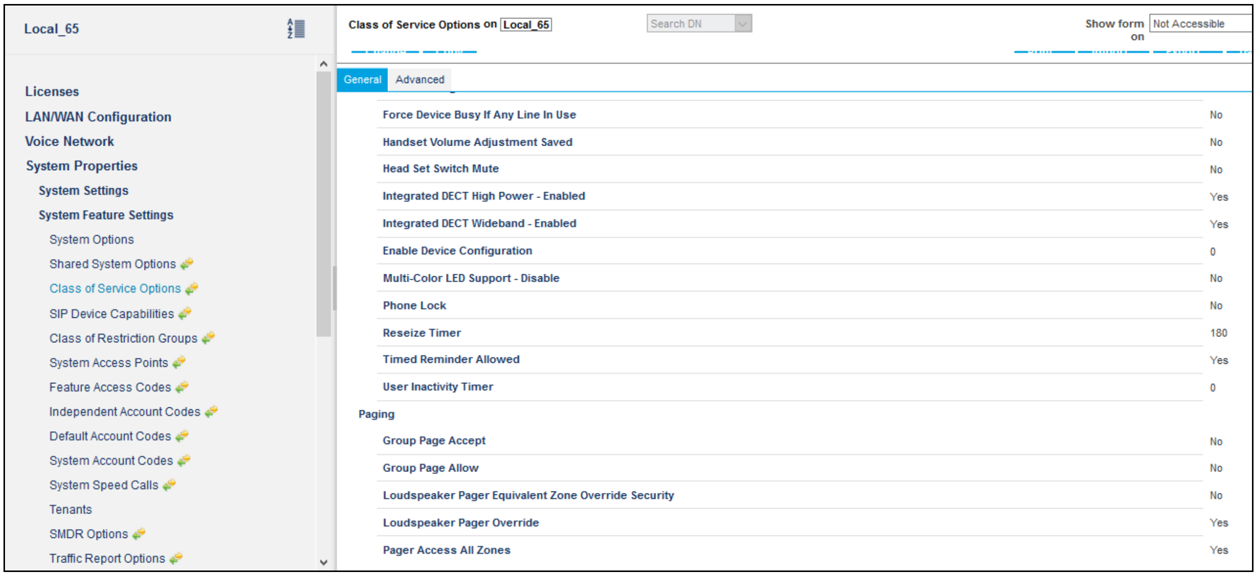


Figure 11 – Class of Service General

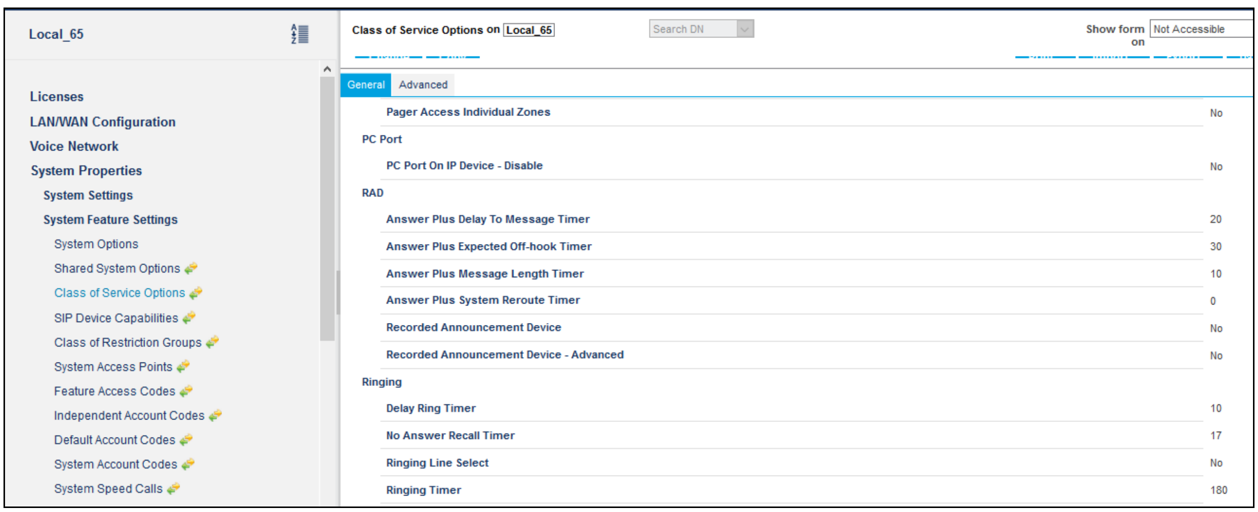


Figure 12 – Class of Service General

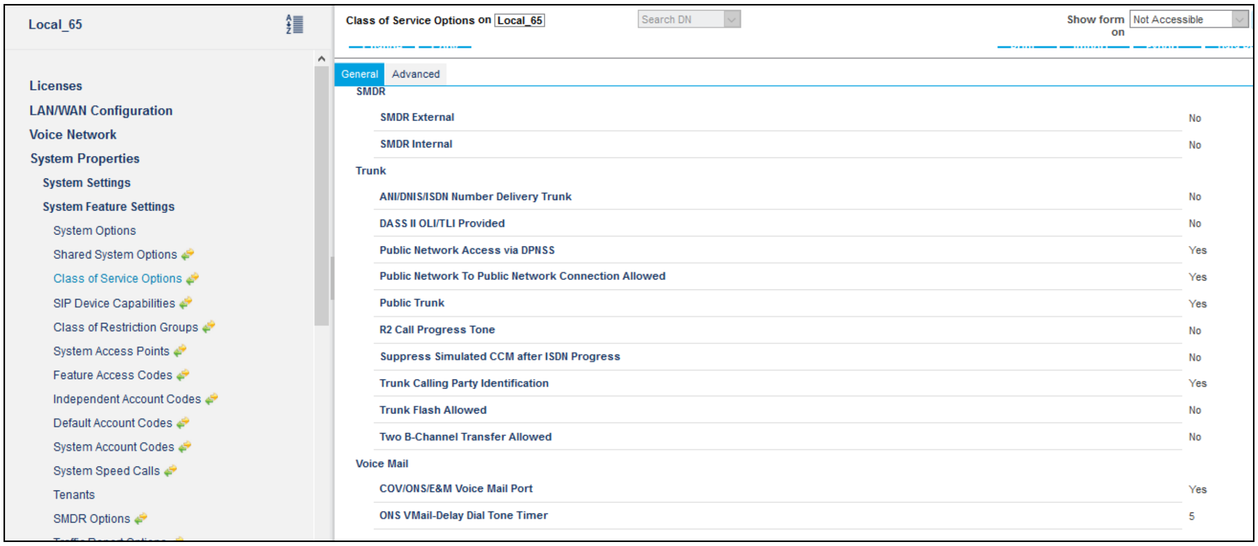


Figure 13 – Class of Service General

Class of Service Advance Tab Configuration Value should be Default. As shown below

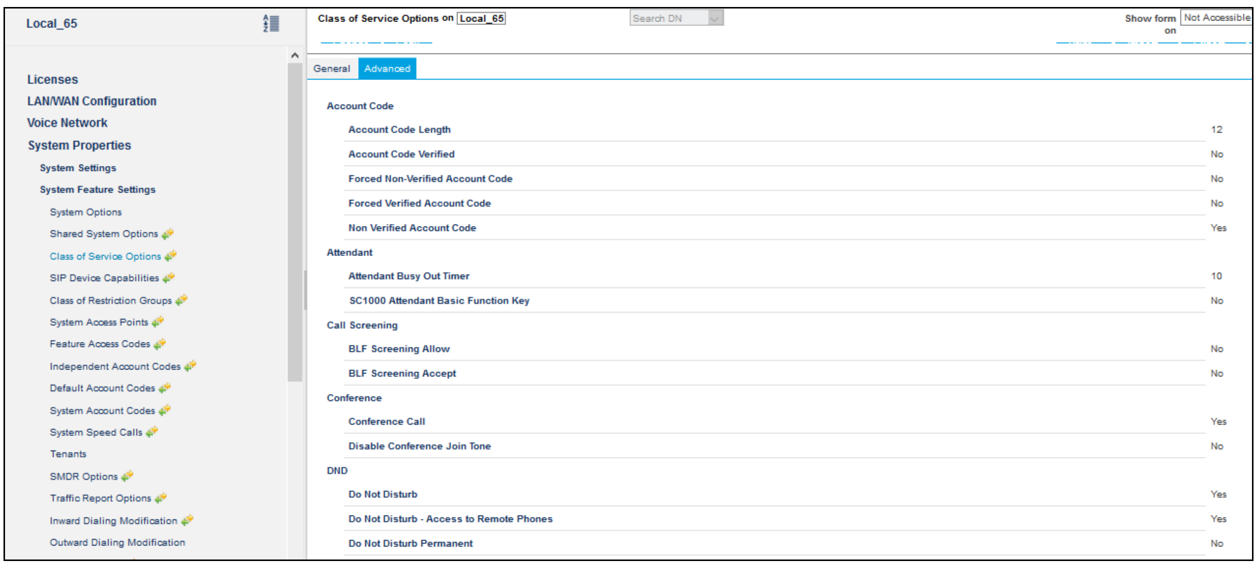


Figure 14 – Class of Service Advance

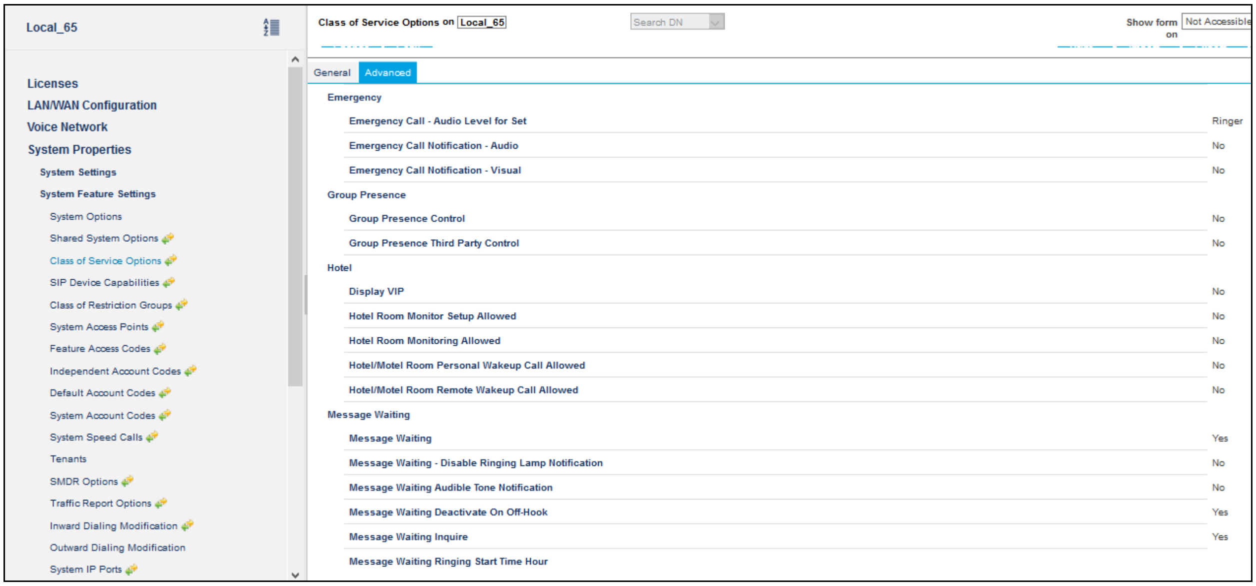


Figure 15 – Class of Service Advance

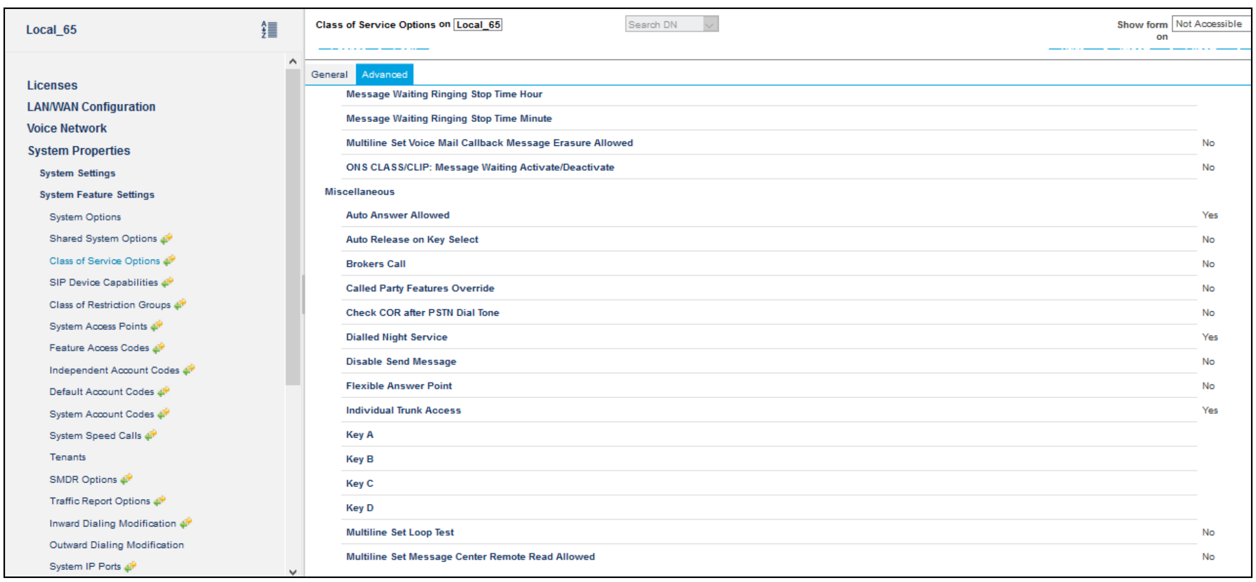


Figure 16 – Class of Service Advance

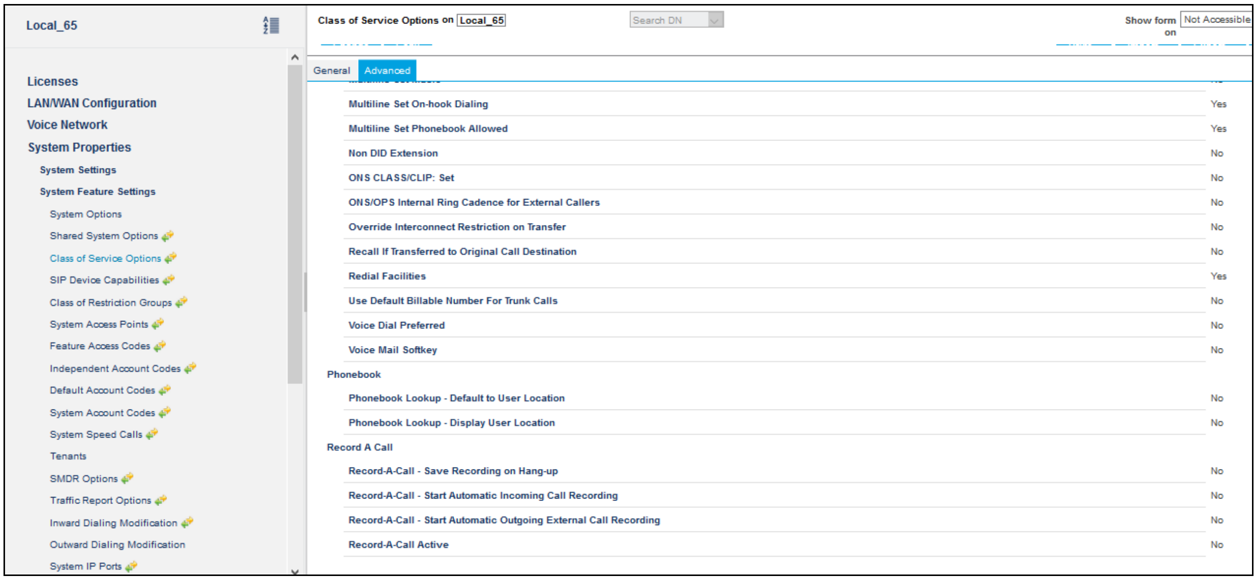
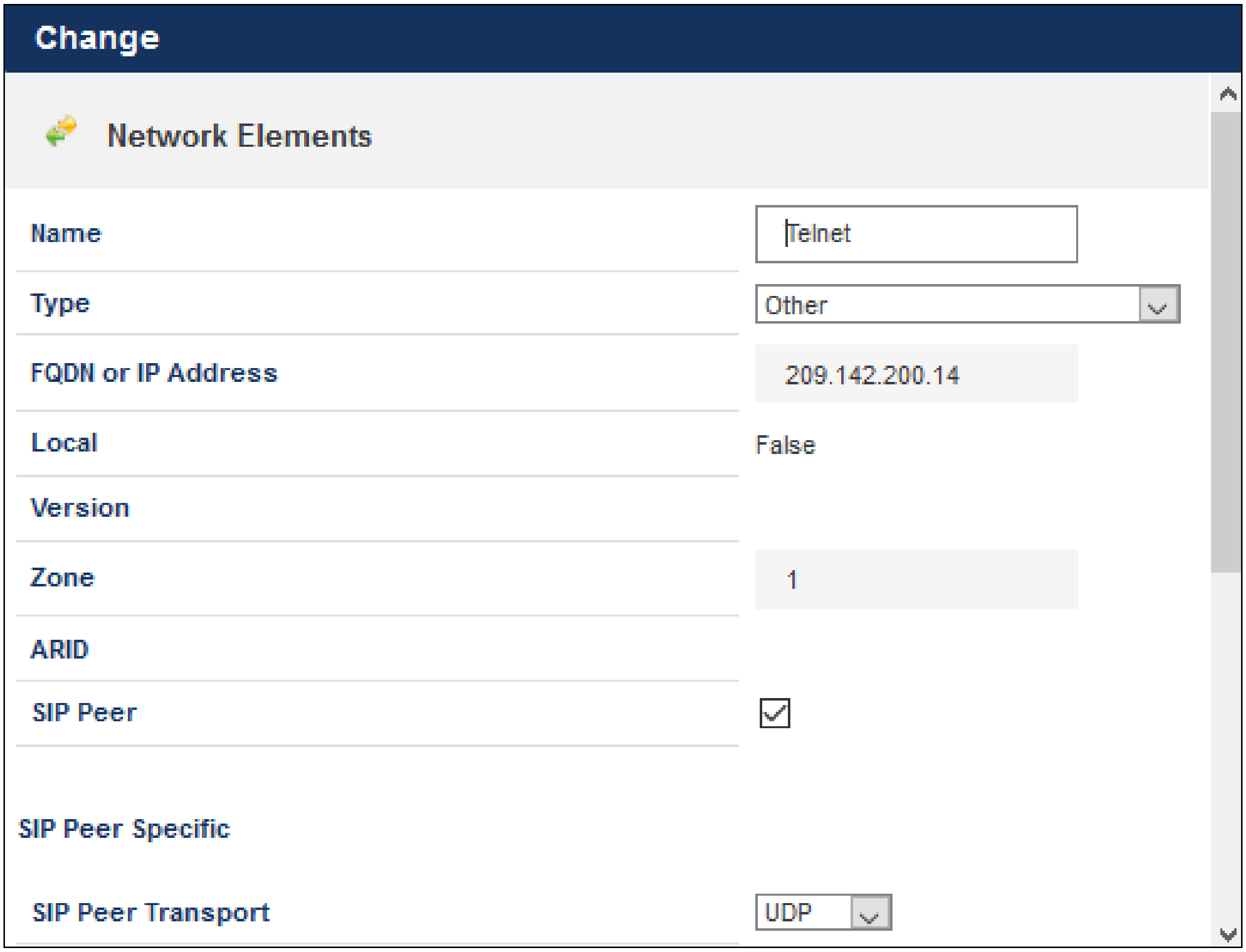


Figure 17 – Class of Service Advance

Network Element Assignment

Create a network element for Service Provider TelNet Worldwide SIP Trunking. In this example, the soft switch is reachable by an IP Address and is defined as “Service Provider TelNet Worldwide” in the network element assignment form. **The FQDN or IP addresses of the SIP Peer (Network Element), the External SIP Proxy are provided by your service provider.**

If your service provider 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. Set the transport to UDP and port to 5060.


Change
Network Elements
Name
Telnet
Type
Other
FQDN or IP Address
209.142.200.14
Local
False
Version
Zone
1
ARID
SIP Peer
SIP Peer Specific
SIP Peer Transport
UDP

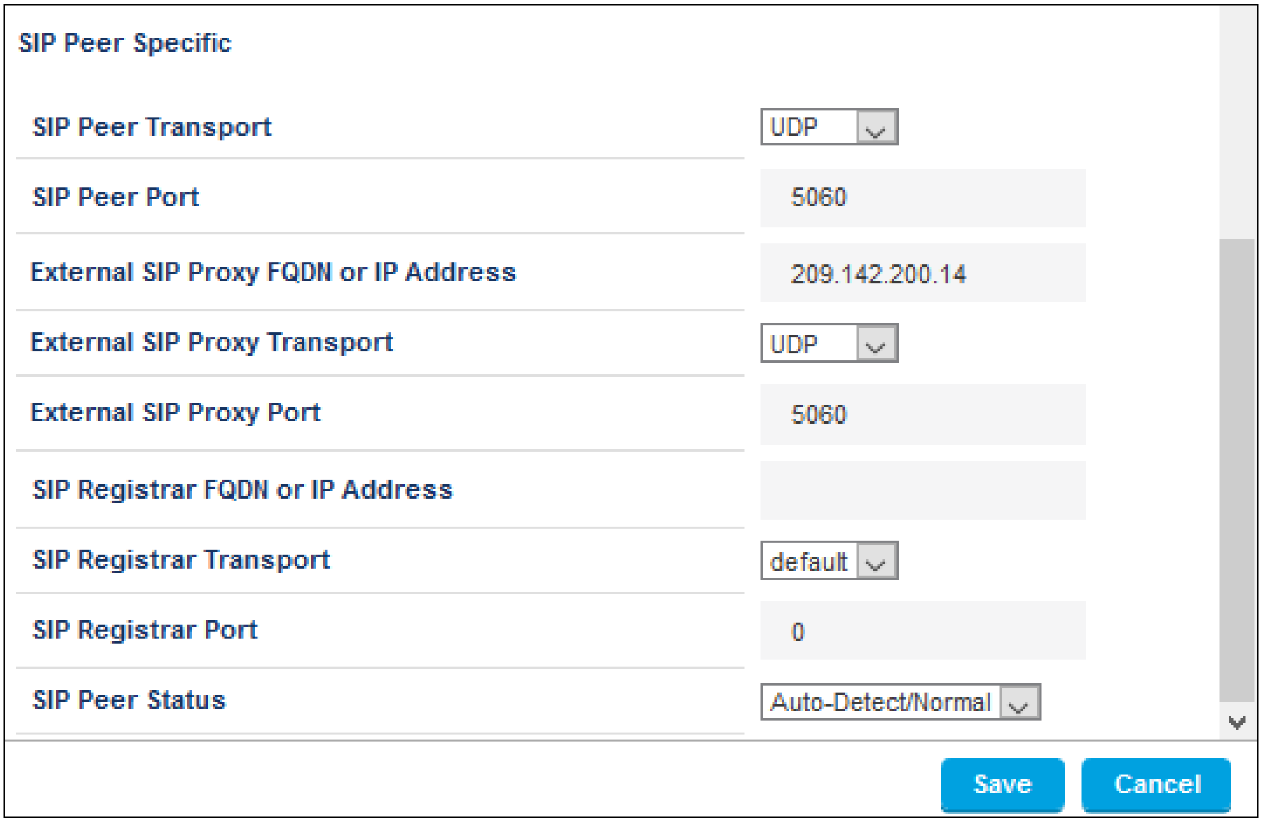


Figure 18 – Network Element Assignment

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).

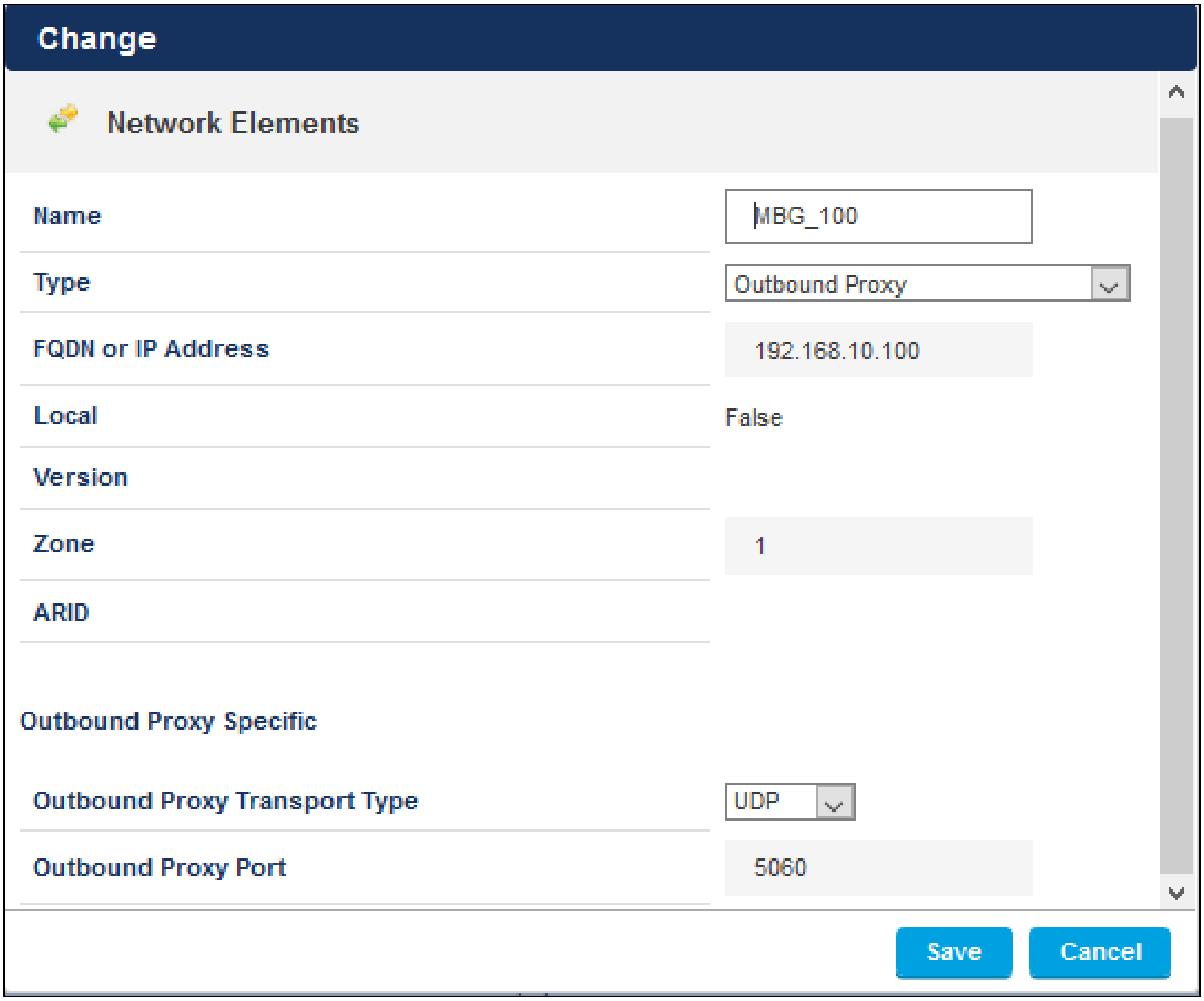
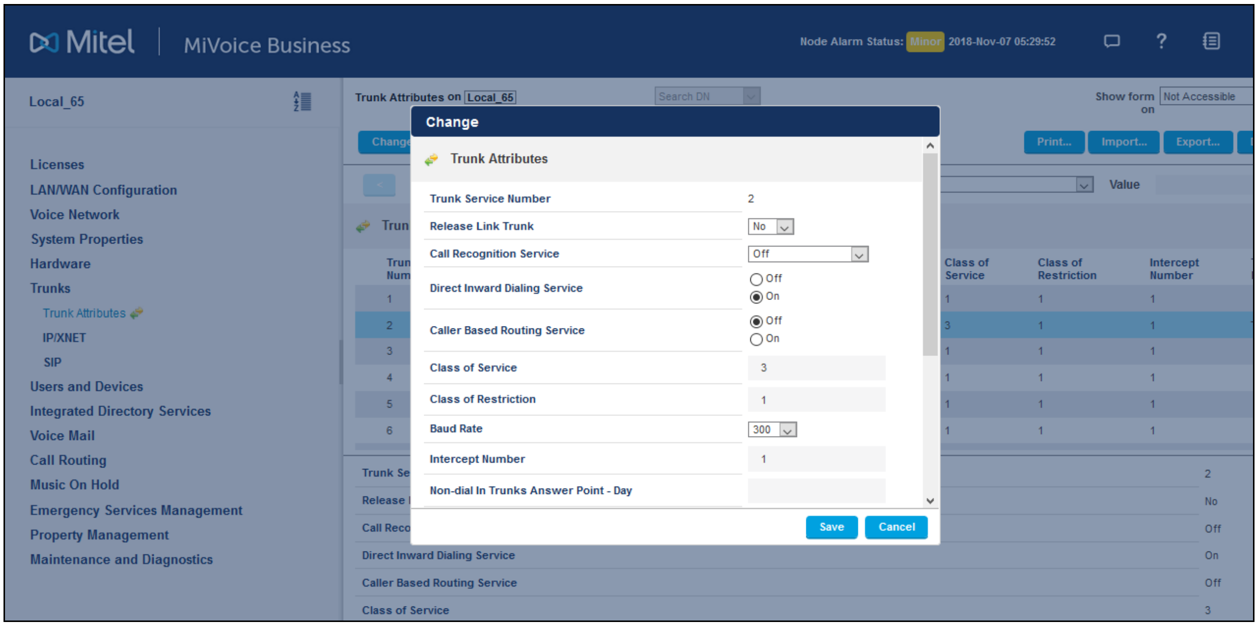


Figure 19 – Network Element Assignment (Proxy)

Trunk Attributes

This is configured in the Trunk Attributes form. In this example the Trunk Attributes is defined for Trunk Service Number 2 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 6 digits of the DID number from Service Provider TelNet Worldwide leaving 4 digits for the MiVB to translate and ring the remaining 4-digit extension. For example, Service Provider TelNet Worldwide delivers 248-498-1136 through the SIP trunk to the MiVB. The MiVB will absorb the first 6 digits (248498) leaving the MiVB to ring extension 5000. Extension 5000 must be programmed as a valid dialable number in the MiVB. Please refer to the Mitel MiVB System Administration documentation for further programming information.



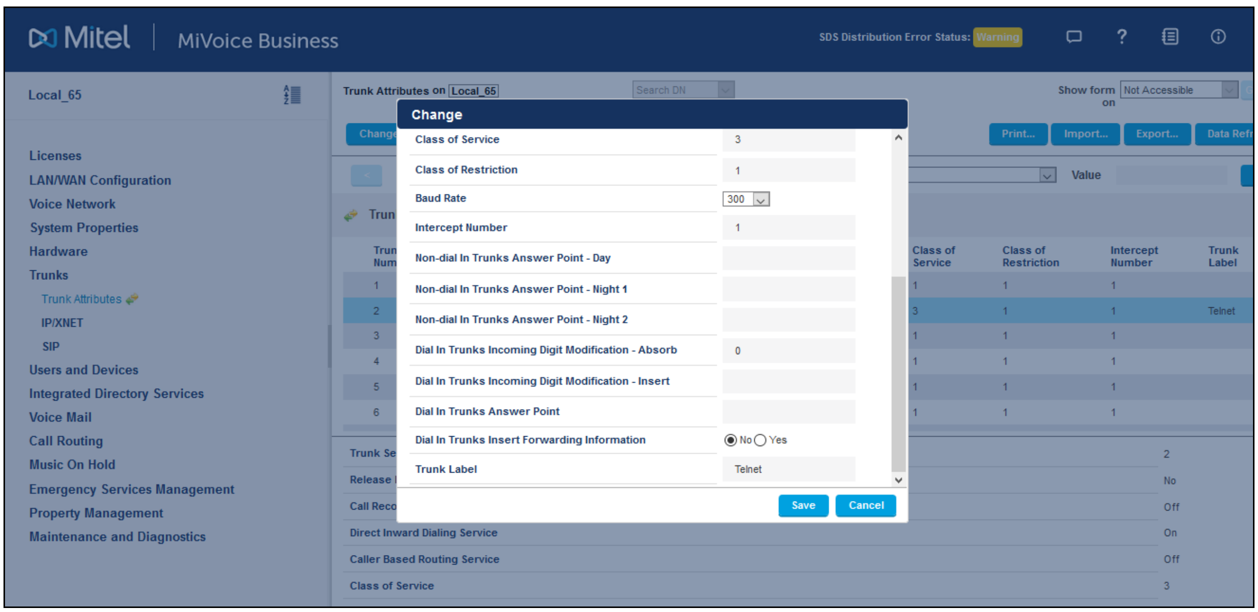


Figure 20 – Trunk Attributes

SIP Peer Profile

The recommended connectivity via SIP Trunking does not require additional physical interfaces. IP/Ethernet connectivity is part of the base 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 "Service Provider TelNet Worldwide" Network Element.

Registration User Name: The Mitel MiVB does not support Bulk Registration; therefore, trunks will have to be registered individually. Enter the DIDs assigned by Service Provider TelNet Worldwide. 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, "6135554000-6135554400, 6135554500"). 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.

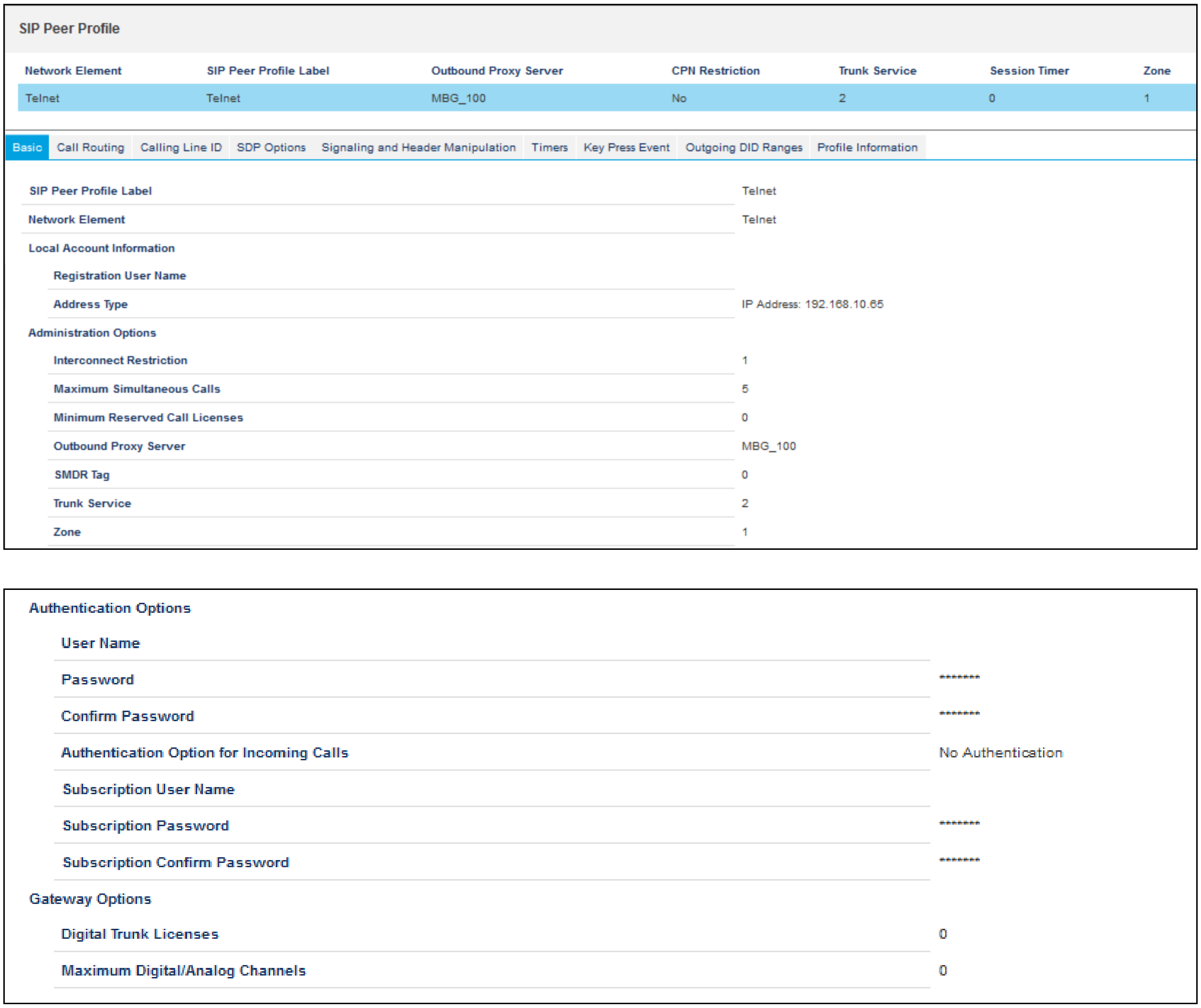
Calling Line ID: The default CPN is applied to all calls unless there is a match in the "Outgoing DID Ranges" of the SIP Peer Profile. **This number will be provided by Service Provider TelNet Worldwide.** Do not use a Default CPN if you want public numbers to be preserved through the SIP interface. 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 TelNet Worldwide.

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



SIP Peer Profile

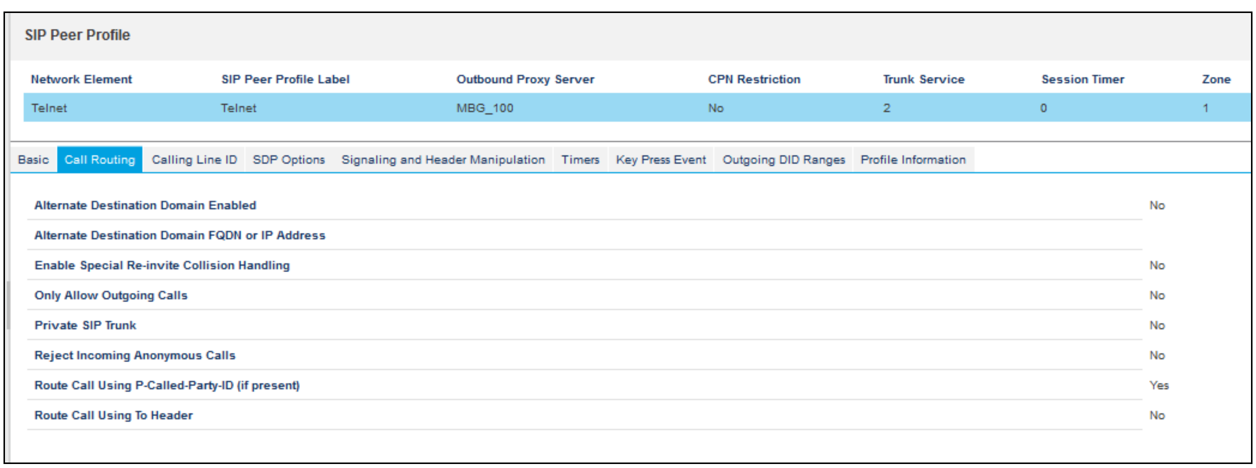
Network Element	SIP Peer Profile Label	Outbound Proxy Server	CPN Restriction	Trunk Service	Session Timer	Zone
Telnet	Telnet	MBG_100	No	2	0	1

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	Telnet
Network Element	Telnet
Local Account Information	
Registration User Name	
Address Type	IP Address: 192.168.10.65
Administration Options	
Interconnect Restriction	1
Maximum Simultaneous Calls	5
Minimum Reserved Call Licenses	0
Outbound Proxy Server	MBG_100
SMDR Tag	0
Trunk Service	2
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 21 – SIP Peer Profile Assignment- Basic



SIP Peer Profile

Network Element	SIP Peer Profile Label	Outbound Proxy Server	CPN Restriction	Trunk Service	Session Timer	Zone
Telnet	Telnet	MBG_100	No	2	0	1

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 22 – SIP Peer Profile Assignment- Call Routing

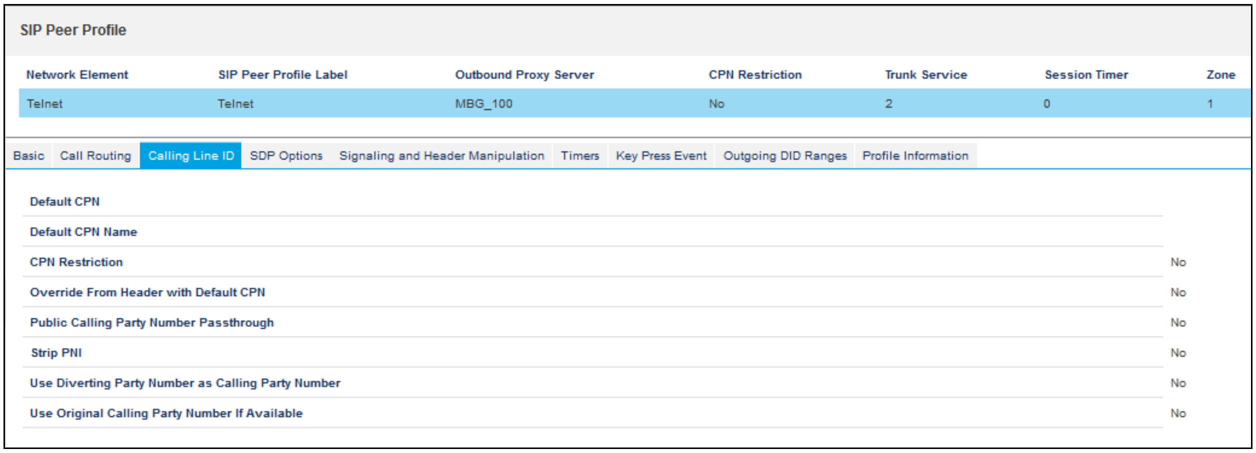


Figure 23 – SIP Peer Profile Assignment- Calling Line ID

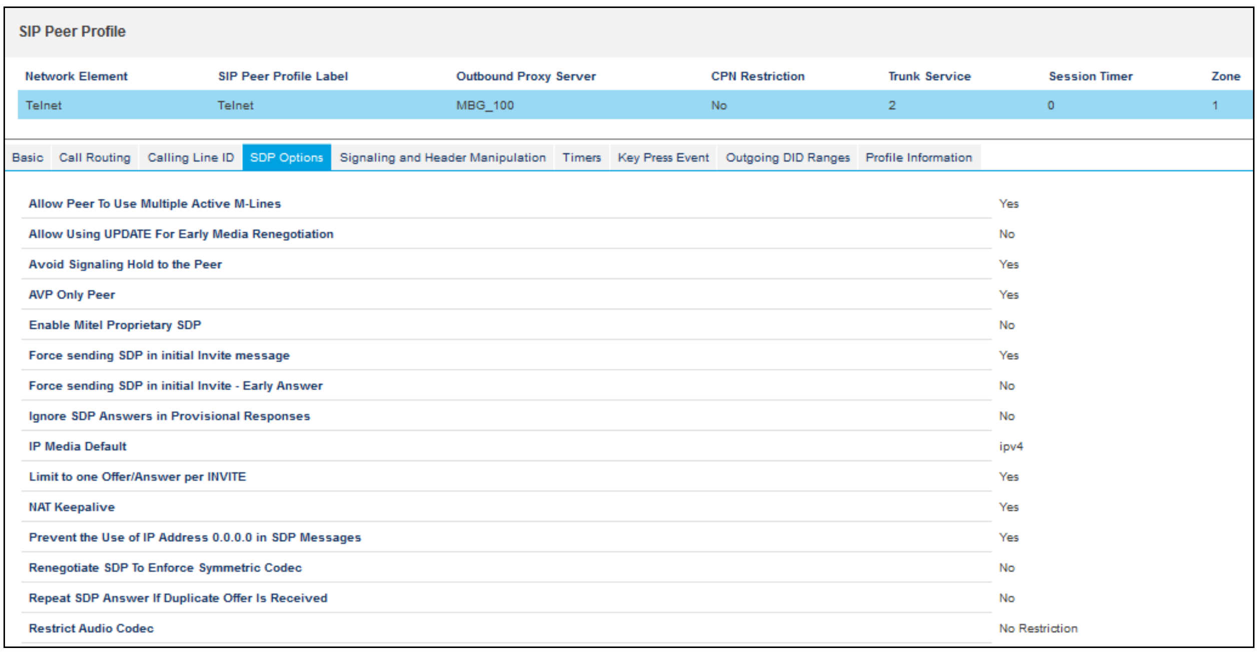


Figure 24 – SIP Peer Profile Assignment- SDP Options

SIP Peer Profile

Network Element	SIP Peer Profile Label	Outbound Proxy Server	CPN Restriction	Trunk Service	Session Timer	Zone
Telnet	Telnet	MBG_100	No	2	0	1

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	Yes
Disable Use of User-Agent and Server 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

SIP Peer Profile

Network Element	SIP Peer Profile Label	Outbound Proxy Server	CPN Restriction	Trunk Service	Session Timer	Zone
Telnet	Telnet	MBG_100	No	2	0	1

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

Only use SDP to decide 180 or 183	Yes
Prefer From Header for Caller ID	No
Require Reliable Provisional Responses on Outgoing Calls	No
Signal Privacy (if enabled) on Emergency Calls	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 25 – SIP Peer Profile Assignment- Signaling and Header Manipulation

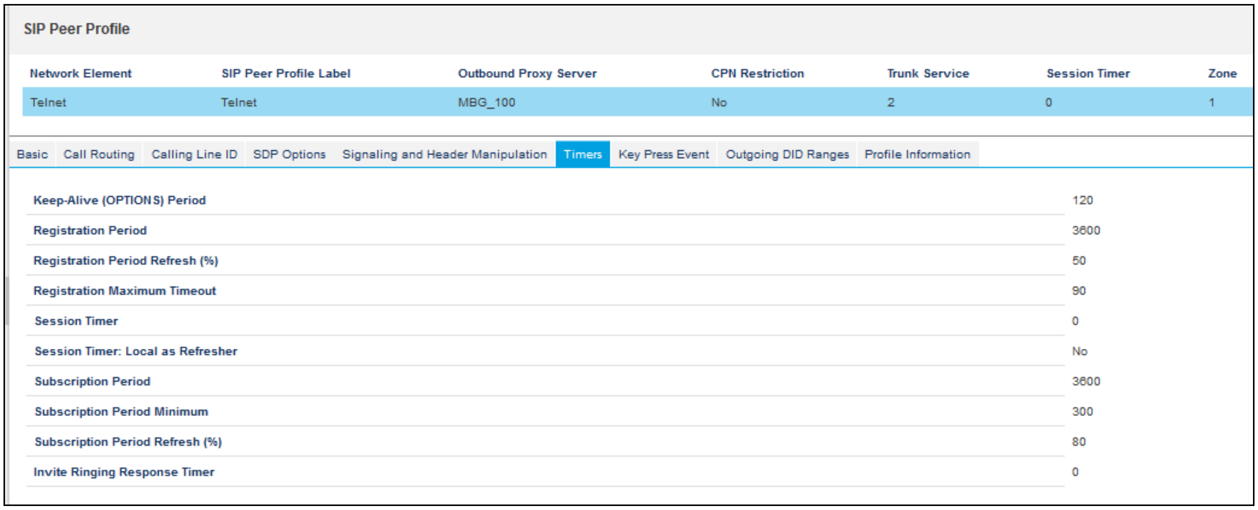


Figure 26 – SIP Peer Profile Assignment- Timers

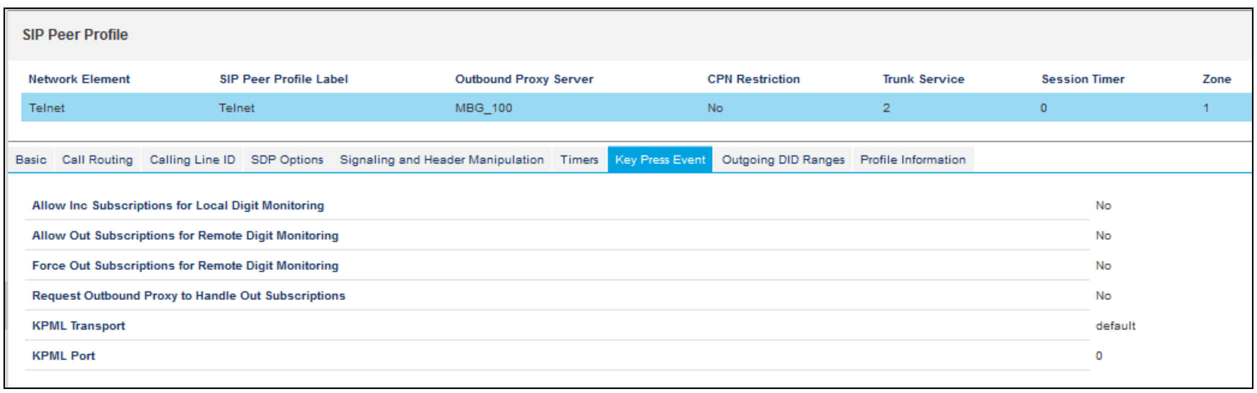


Figure 27 – SIP Peer Profile Assignment- Key Press Event

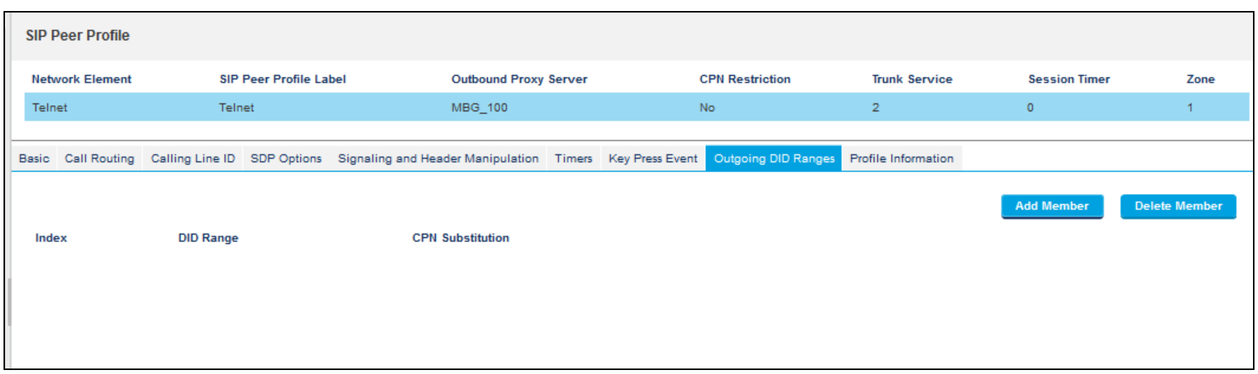


Figure 28 – SIP Peer Profile Assignment- Outgoing DID Ranges

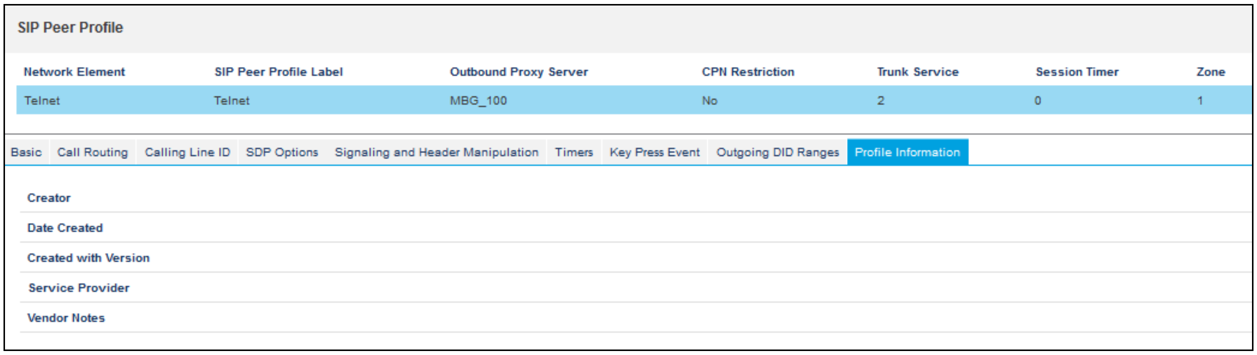


Figure 29 – SIP Peer Profile Assignment- Profile Information

SIP Peer Profile Assignment by Incoming DID

This form is used to associate DID range numbers from Service Provider TelNet Worldwide SIP trunk to a particular SIP Peer profile. The configured here settings 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, "33970008470-33970008472, 33970008475"). 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.

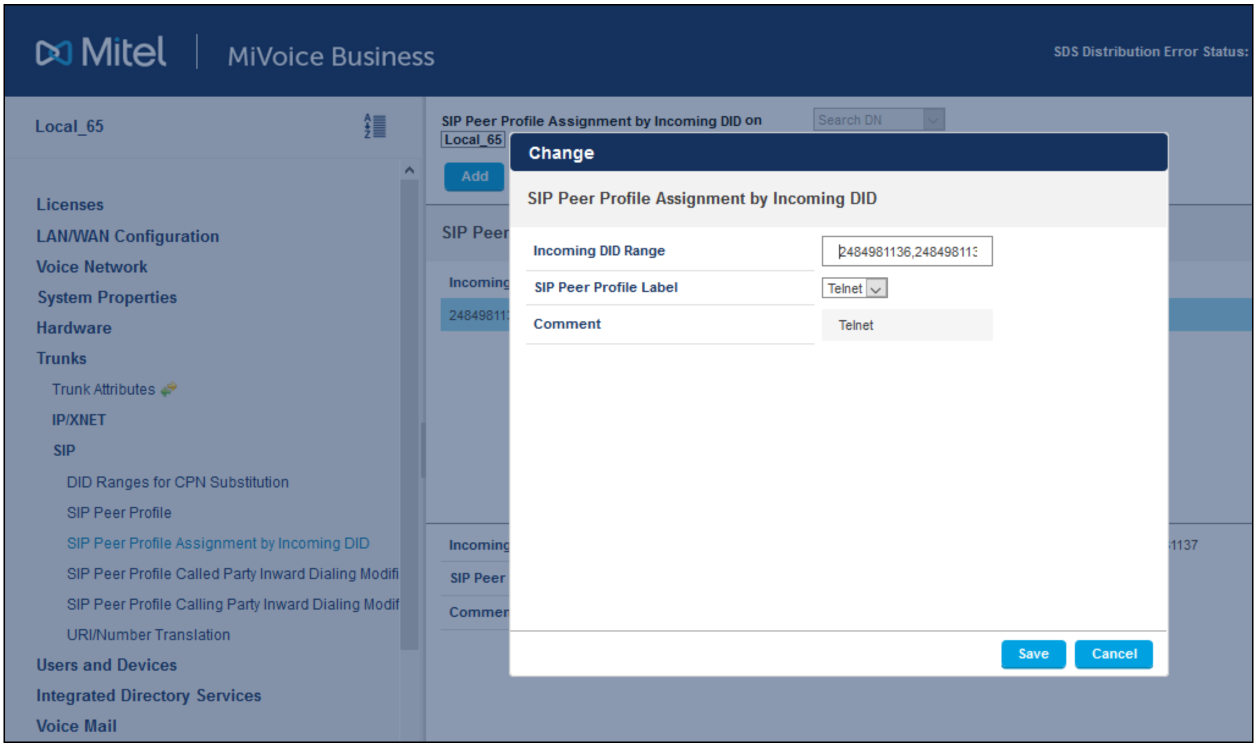


Figure 30 – SIP Peer Profile Assignment by Incoming DID

ARS Digit Modification Plans

Ensure that Digit Modification for outgoing calls on the SIP trunk to Service Provider TelNet Worldwide absorbs or inject 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).

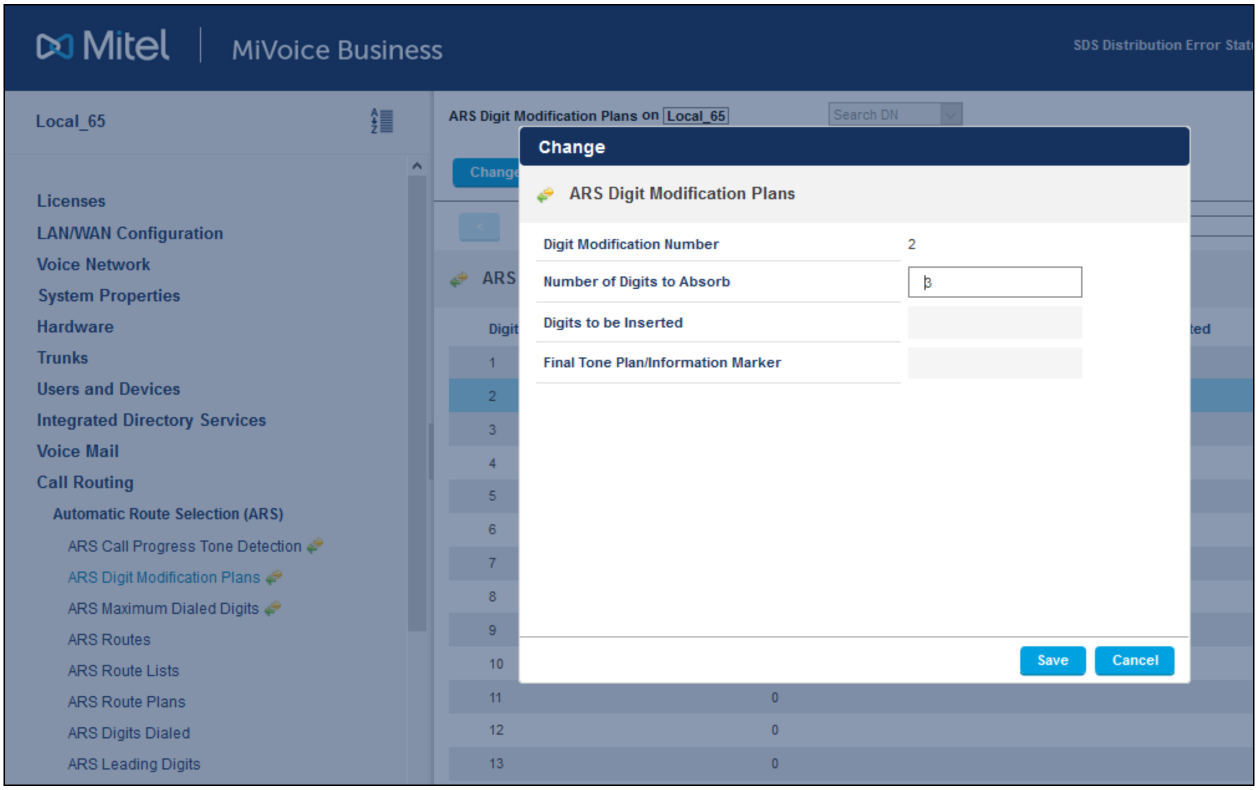


Figure 31 – Digit Modification Assignment

ARS Routes

Create a route for SIP Trunks connecting a trunk to Service Provider TelNet Worldwide. In this example, the SIP trunk is assigned to Route Number 2. Choose SIP Trunk as a routing medium and choose the SIP Peer Profile and Digit Modification entry created earlier.

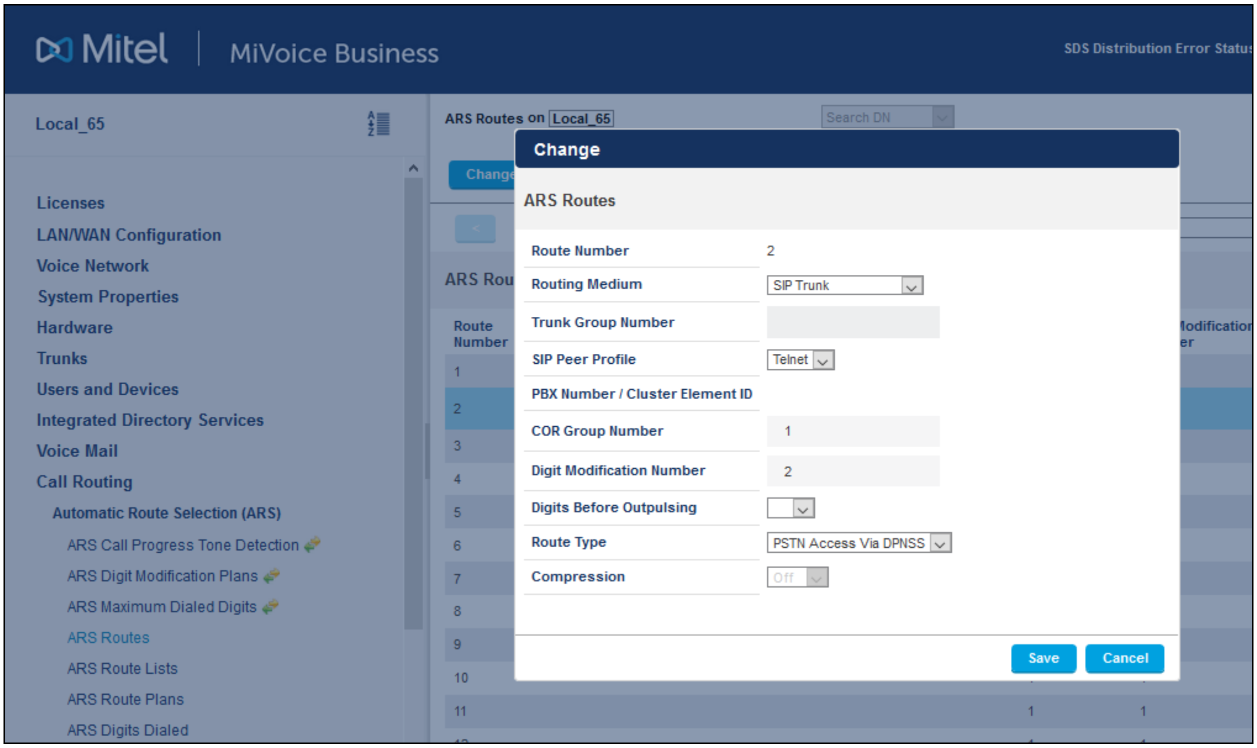


Figure 32 – SIP Trunk Route Assignment

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 456, the call will be routed to Service Provider TelNet Worldwide (i.e. Route 2).

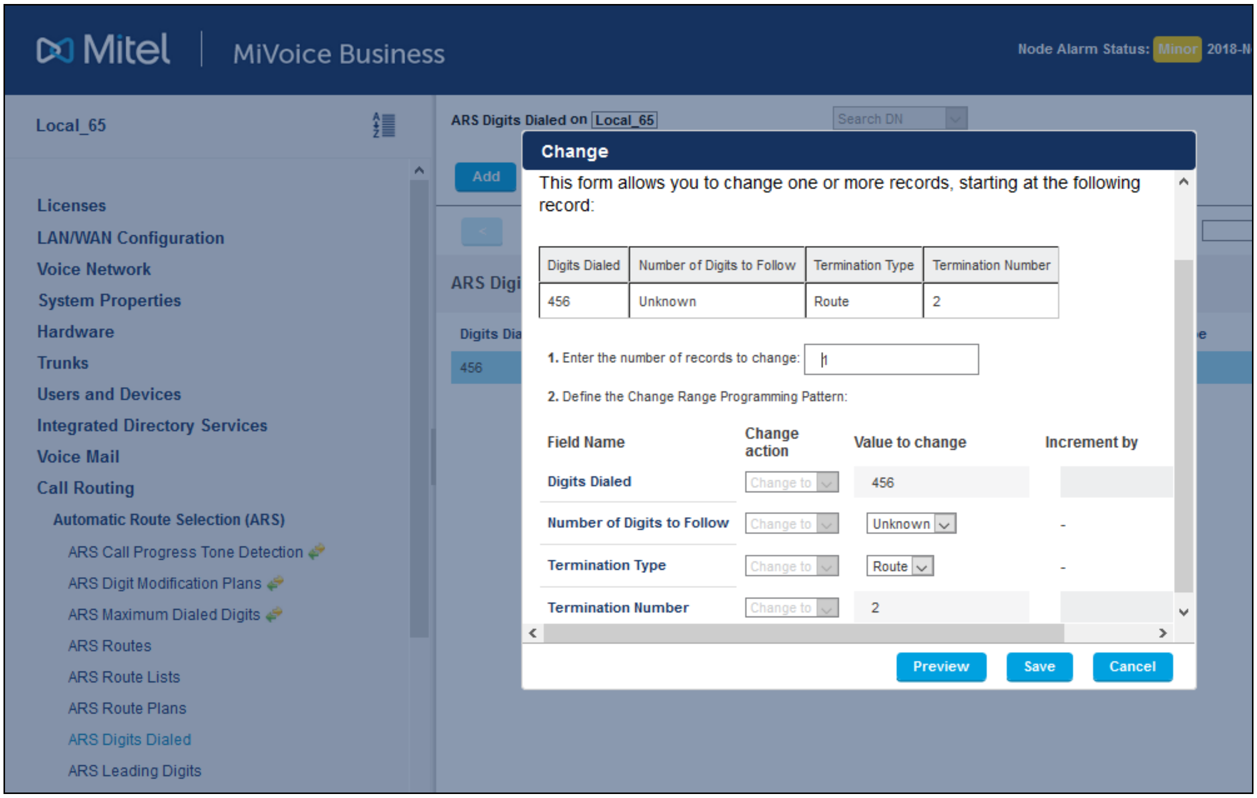


Figure 33 – ARS Digit Dialed Assignment

T.38 Fax Configuration

Service Provider TelNet Worldwide 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).

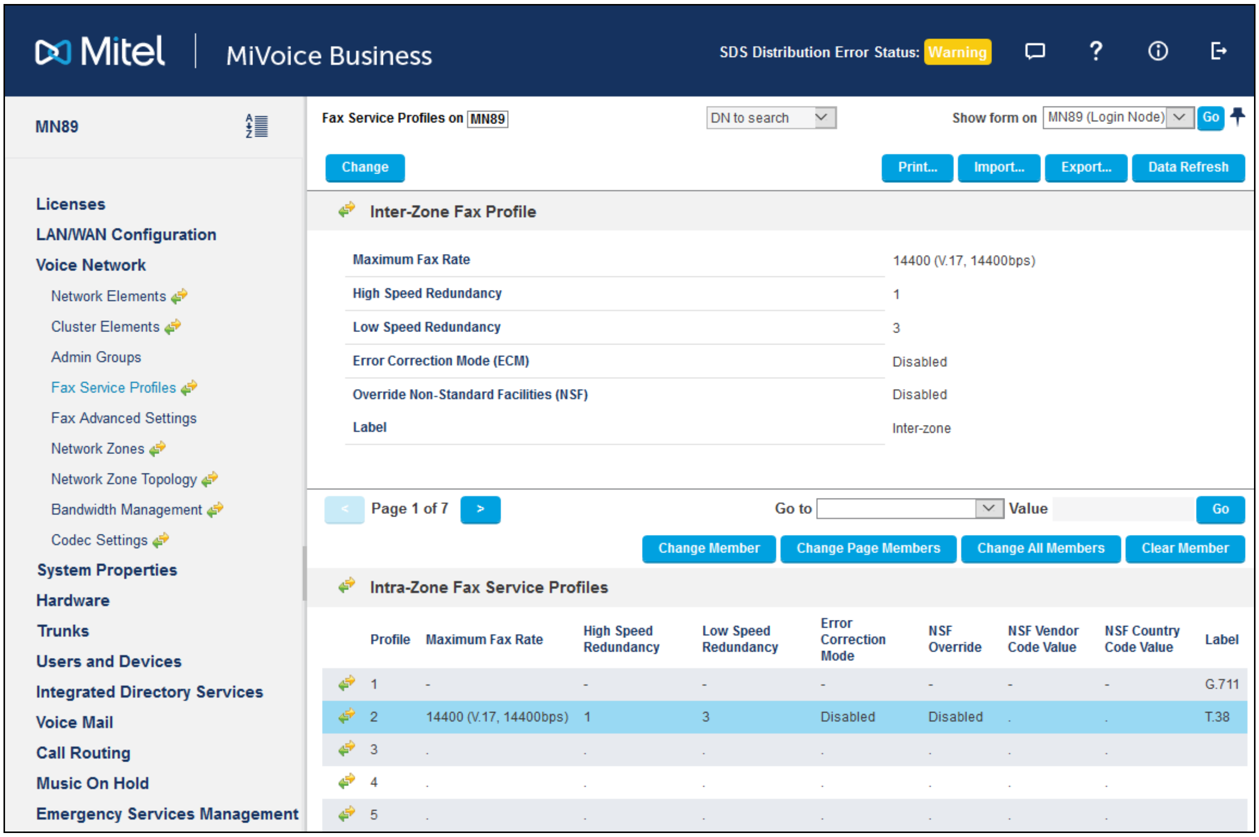


Figure 34 - Fax Configuration

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 TelNet Worldwide uses the Intra-zone FAX Profile 2 for T.38 FAX.

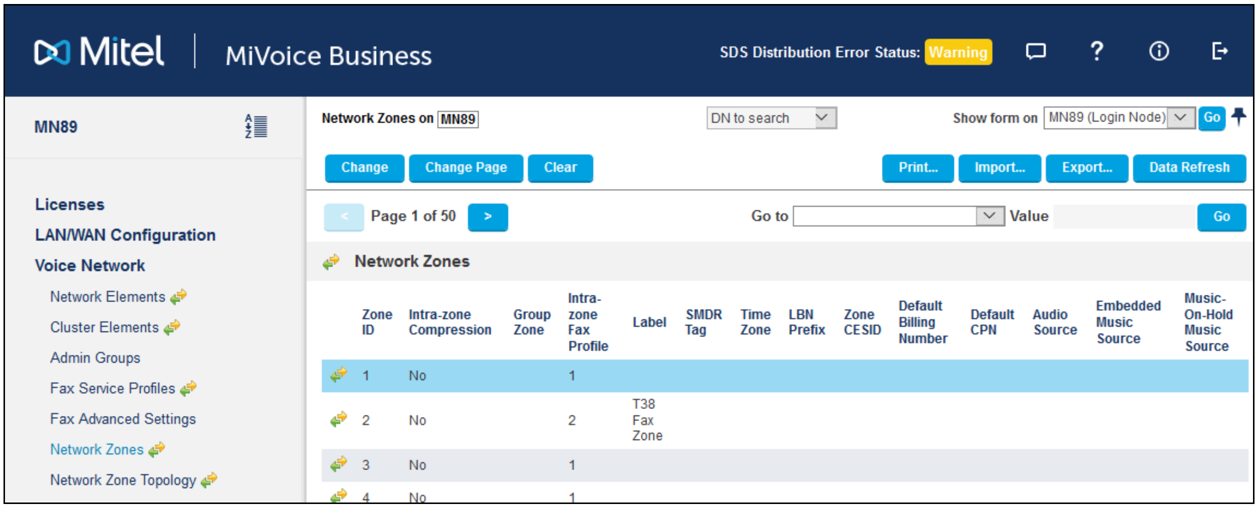


Figure 35 – Zone Assignment

MiVoice Border Gateway Configuration Notes (Optional)

When configuring MiVoice Border Gateway (MBG), you need to identify the working MiVB ICP where to forward SIP messages to and then to configure the SIP trunk.

To do this:

- Login to MBG and click **MiVoice Border Gateway**
- In right pane, click **Service Configuration** tab and then **ICPs** (see Figure 36 for details)

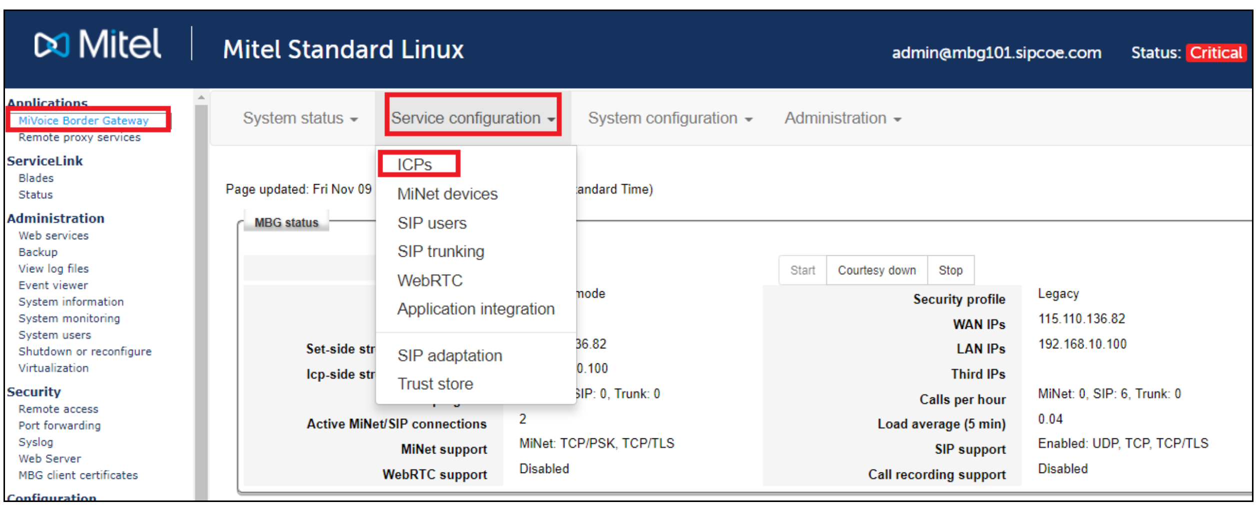


Figure 36 – MBG's Configuration page

- On **ICPs** page, ensure that the "working" MiVB is configured. If needed, click **Add ICP** link and add a new Mitel switch.
- Click **Update** button

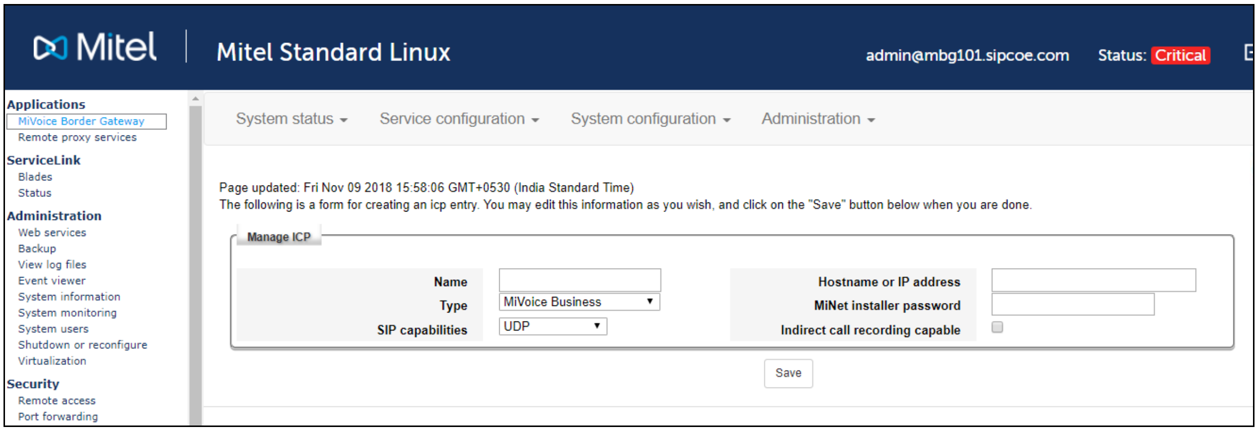


Figure 37 – ICP configuration page

To add a new SIP trunk:

- Click **Service Configuration** tab and then click **SIP trunking** as shown below
- Click **Add a SIP trunk** link (see Figure 38)

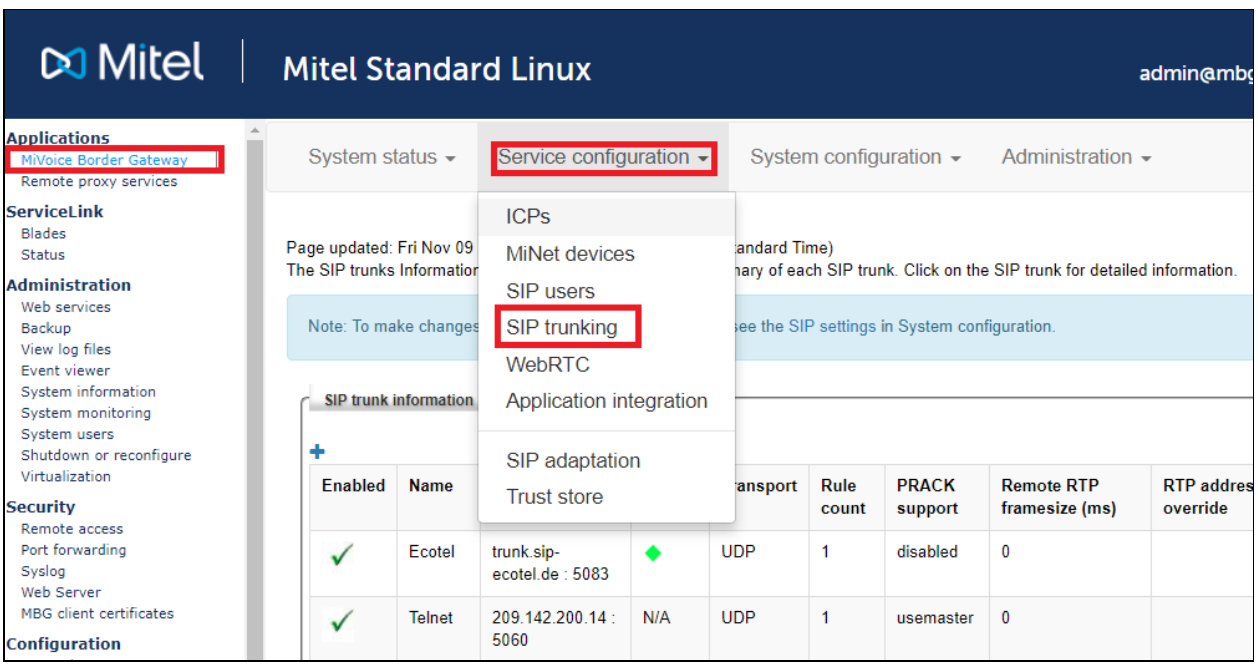


Figure 38 – SIP Trunking configuration page

Enter the SIP trunk's details as shown in Figure 39:

Name – is the name of the trunk

Remote trunk endpoint address – the public IP address of the provider's switch or gateway (this address should be given to you by the provider, e.g. Service Provider TelNet Worldwide).

Local/Remote RTP frame size (ms) – is the packetization rate you want to set on this trunk
PRACK – Use master setting.
Routing rule one – it allows routing of any digits to the selected MiVB
The rest of the settings are optional and could be configured if required.
Click **Save** button

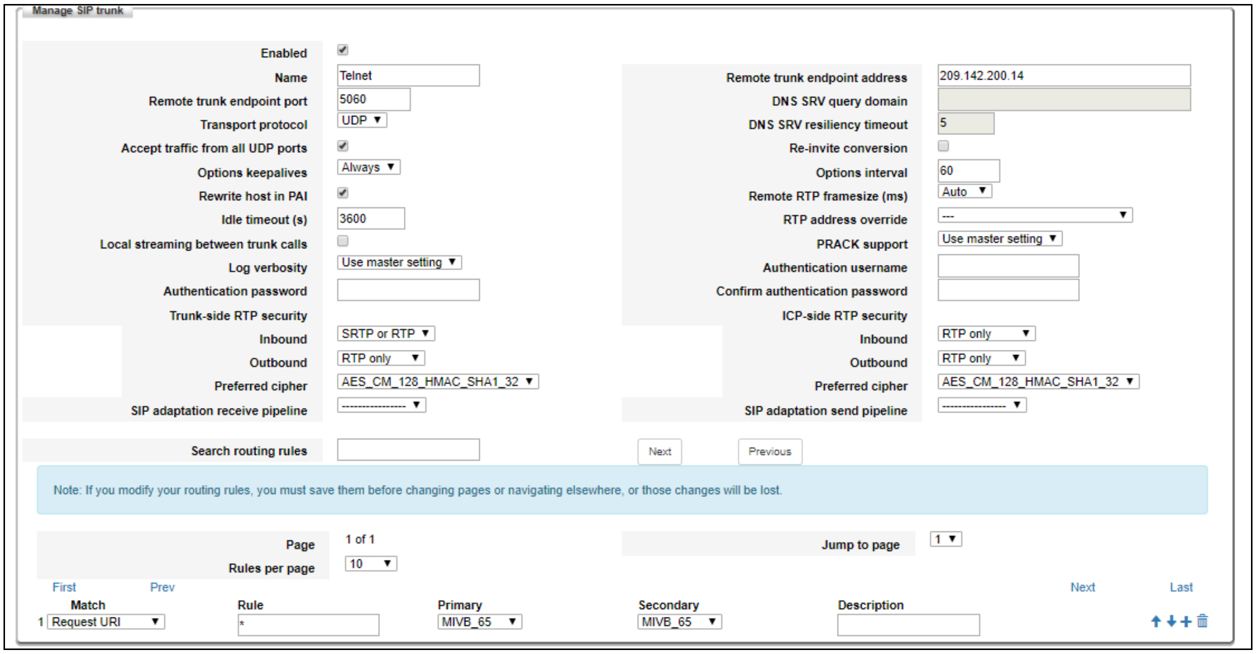


Figure 39 – SIP Trunk configuration settings

Check status: click **System status** tab and then click **SIP Trunks**

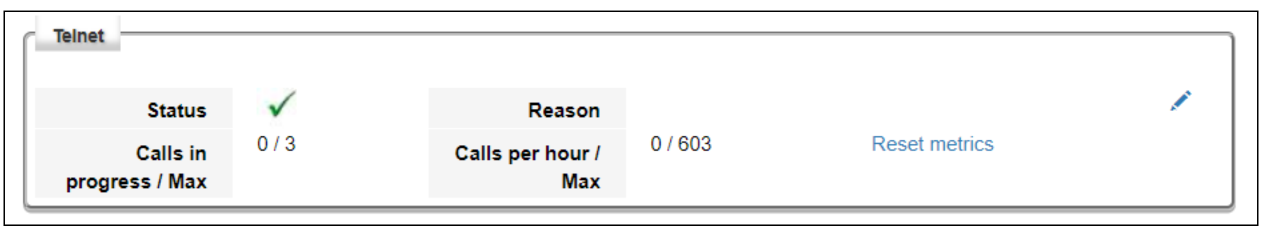


Figure 40 – SIP Trunk Status