Nexmo SIP Trunking Configuration Guide

Mitel MiVoice Business 8.0 PR2 and Mitel Border Gateway 9.4.0.29

July 2017

Contents

1 Introduction

This document is intended for the SIP trunk customer's technical staff and Value Added Retailer (VAR) having installation and operational responsibilities. This configuration guide provides steps for configuring Mitel MiVoice Business version 8.0 PR2 and Mitel Border Gateway 9.4.0.29 to Nexmo SIP Trunking services.

2 SIP Trunking Network Components

The network for the SIP trunk reference configuration shown below is representative of a Mitel MiVoice Business and Mitel MiVoice Border Gateway configuration to Nexmo SIP trunking.

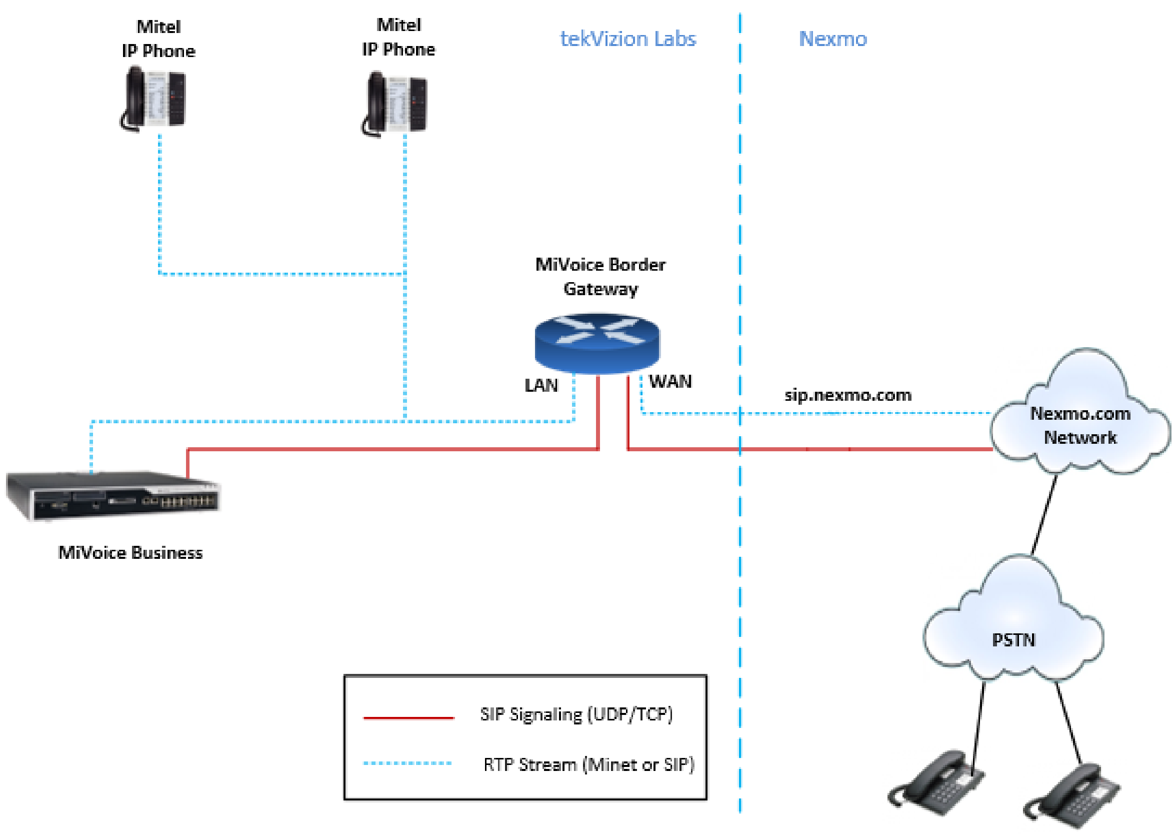


Figure 1: Topology Diagram

2.1 Hardware Components

- MiVoice Business MXE Platform

- VMware Host Version ESXi 5.5.0
- Mitel Minet phones 5360, 5224

2.2 **Software Requirements**

- Mitel MiVoice Business, Version 14.0.0.95, Release 8.0 PR2
- Mitel Border Gateway, Version: 9.4.0.29

3 Features

3.1.1 Features Supported

- Incoming and outgoing off-net calls using G711ULAW & G711ALAW voice codecs
- Calling Line (number) Identification Presentation
- Call Hold and Resume
- Call Transfer (unattended and attended)
- Call Conference
- Call Forward (all, no answer)
- DTMF relay both directions (RFC2833)
- Media flow-through on Mitel Border Gateway

3.1.2 Features Not Supported by PBX

- None

3.1.3 Caveats and Limitations

- None

4 Configuration

4.1 IP Address Worksheet

The specific values listed in the table below and in subsequent sections used in the lab configuration described in this document, and are for **illustrative purposes only**. The customer must obtain and use the values for your deployment.

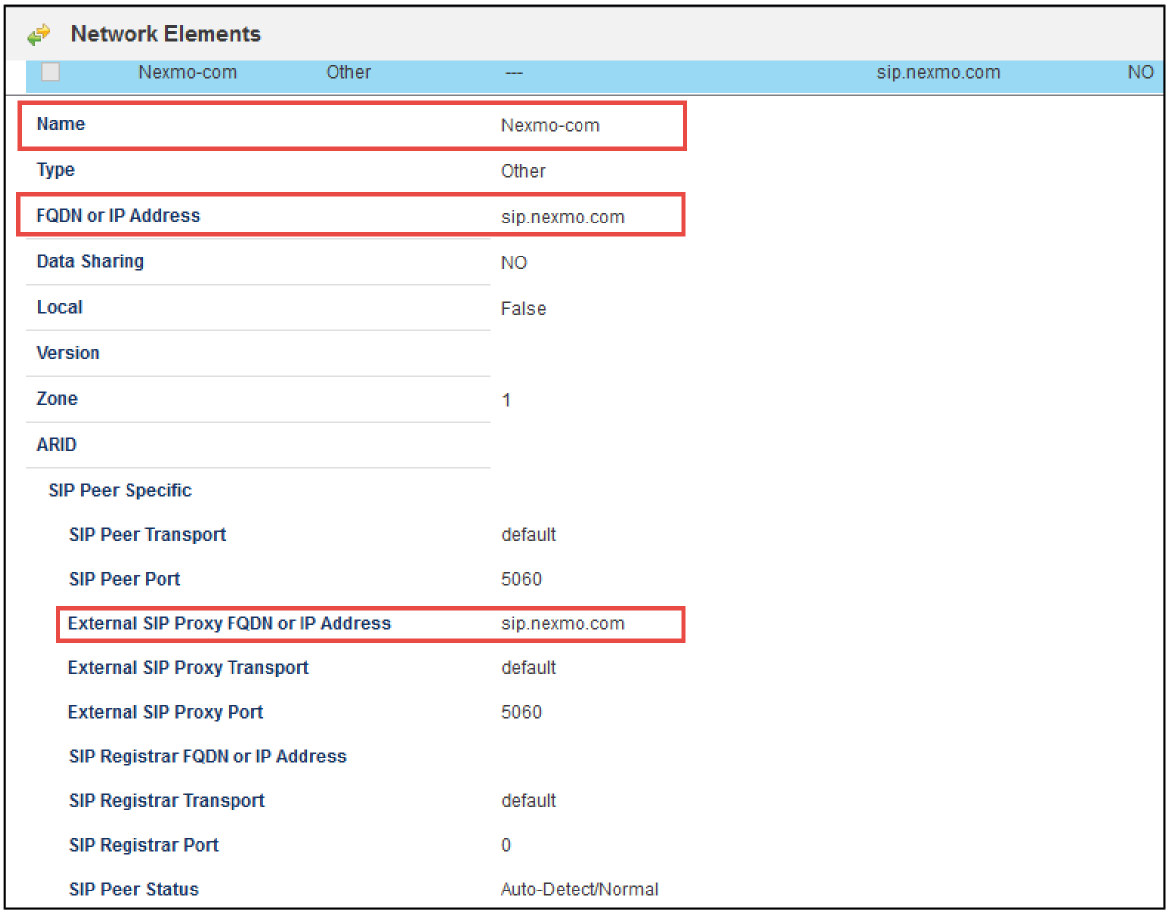
Table 1 – IP Addresses

Component	Lab Value	Customer Value
Mitel Border Gateway		
LAN IP Address	10.64.3.168	
LAN Subnet Mask	255.255.255.0	
WAN IP Address	192.65.79.XXX	
WAN Subnet Mask	255.255.255.128	
Mitel MiVoice Business PBX IP		
System IP Address	10.35.31.85	

4.2 Configuring Mitel MiVoice Business

4.2.1 Network Elements

1. Navigate to **Voice Network → Network Elements**
2. Create network element for Nexmo.com



Network Elements

Nexmo-com	Other	---	sip.nexmo.com	NO

Name	Nexmo-com
Type	Other
FQDN or IP Address	sip.nexmo.com
Data Sharing	NO
Local	False
Version	
Zone	1
ARID	
SIP Peer Specific	
SIP Peer Transport	default
SIP Peer Port	5060
External SIP Proxy FQDN or IP Address	sip.nexmo.com
External SIP Proxy Transport	default
External SIP Proxy Port	5060
SIP Registrar FQDN or IP Address	
SIP Registrar Transport	default
SIP Registrar Port	0
SIP Peer Status	Auto-Detect/Normal

Figure 2: Network Element for Nexmo.com

3. Create network element for Mitel Border Gateway (MBG)

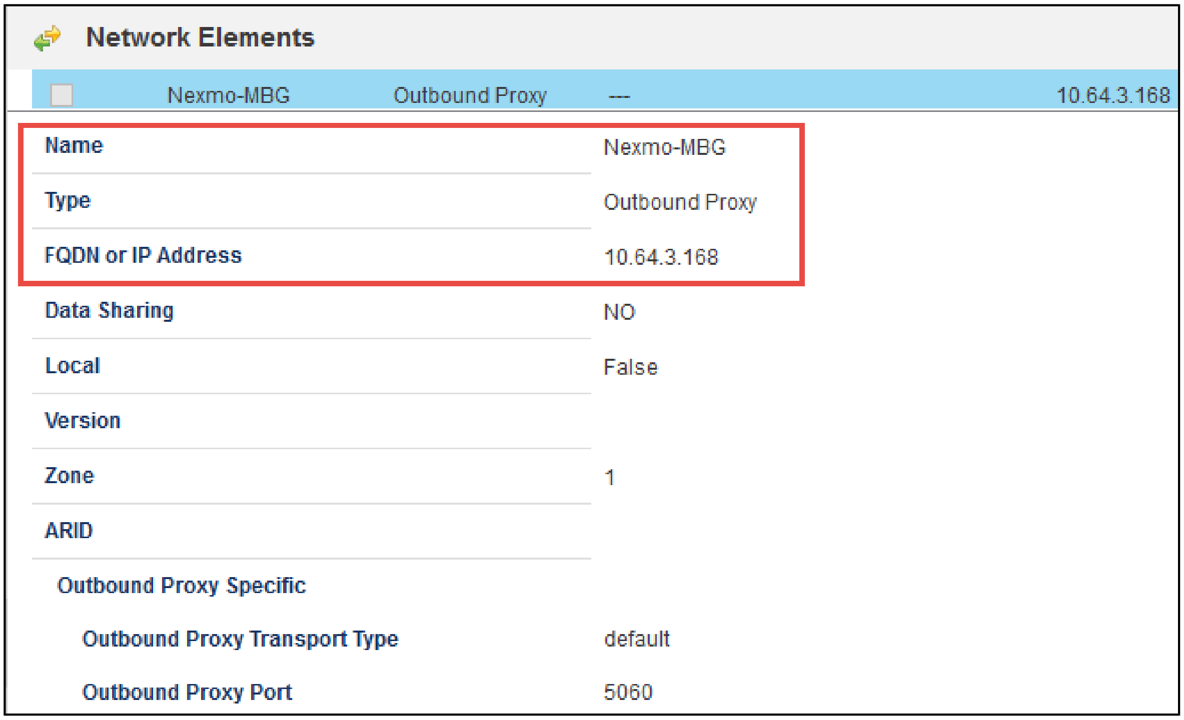


Figure 3: Network Element for Mitel Border Gateway

4.2.2 Mitel Class of Service Parameters

1. Navigate to **System Properties → System Feature Settings → Class of Service Options**
2. Create Class of Service (COS) for Nexmo.com as shown for COS 18 below

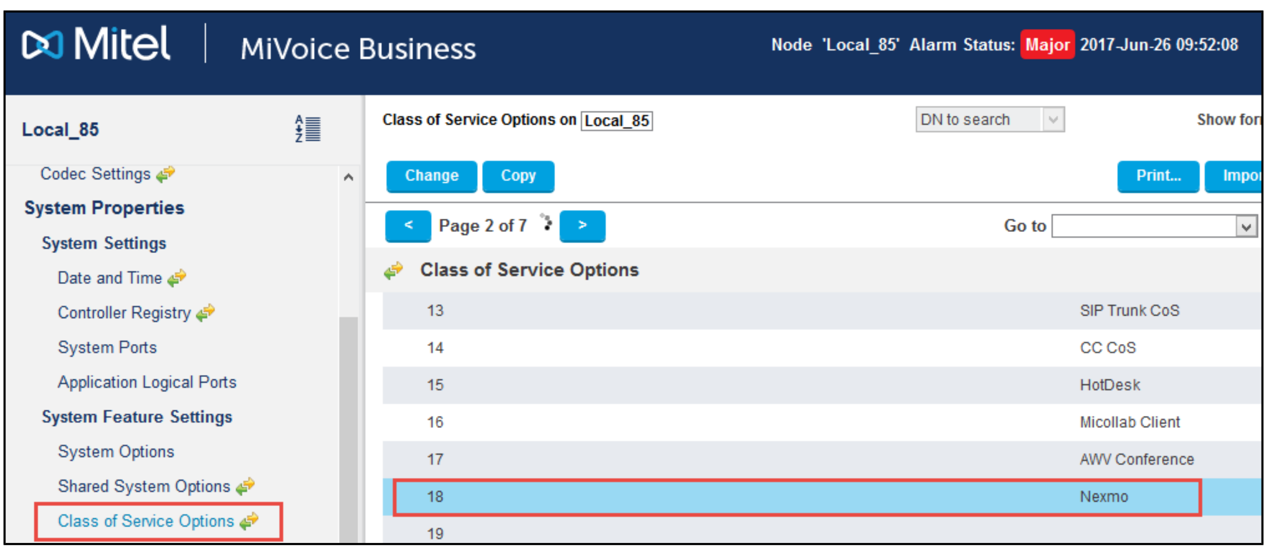
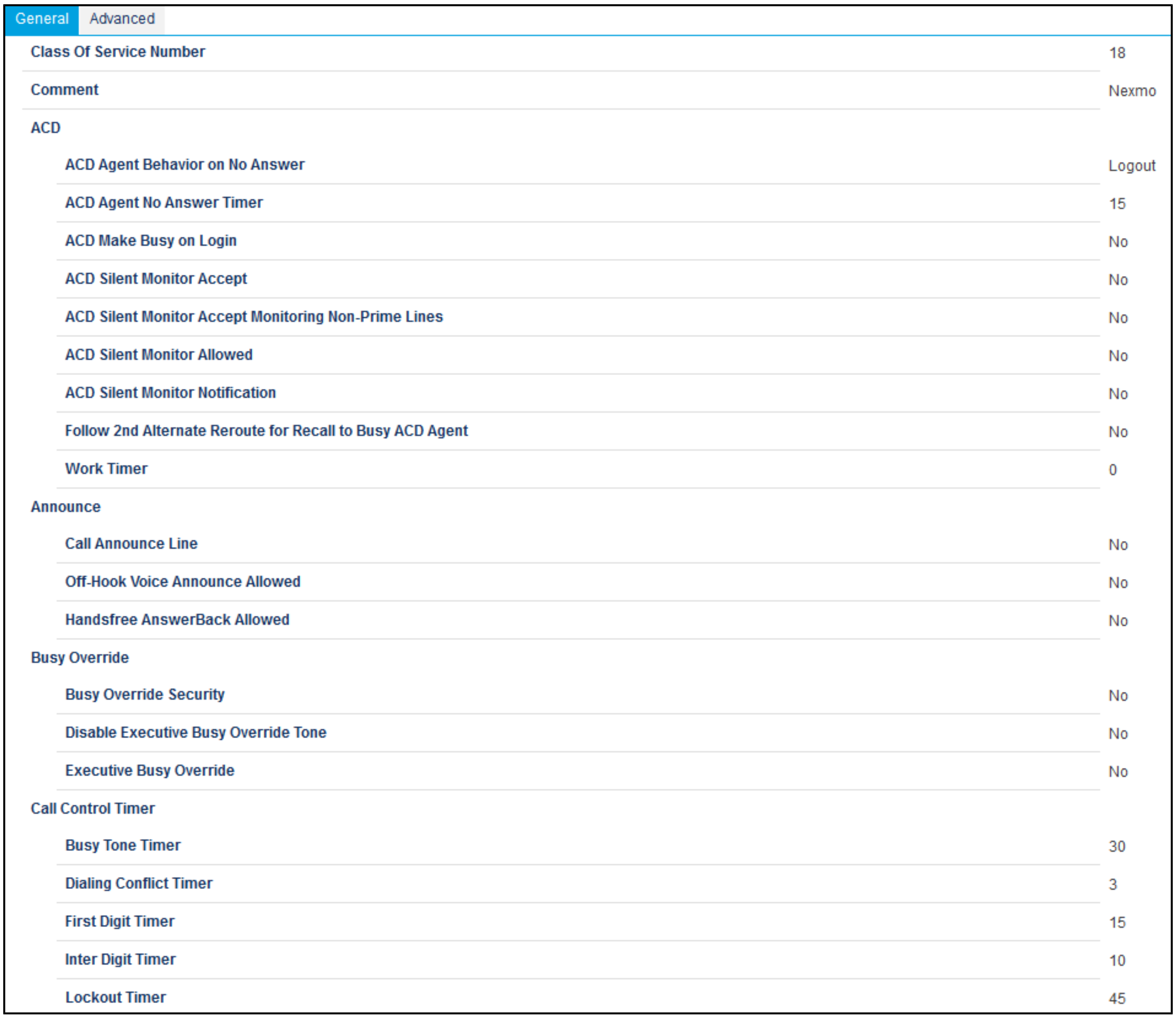


Figure 4: Class of Service for Nexmo SIP Trunk

The configuration changes were only to the Class of Service General tab. The Advanced tab is configure using default settings.

1. Set **Music On Hold on Transfer:** Yes
2. Set **Public Network Access via DPNSS:** Yes
3. The remainder of options are set to default
4. The settings in the **Advanced** Tab are set to default



General | Advanced

Class Of Service Number	18
Comment	Nexmo
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
Dialing Conflict Timer	3
First Digit Timer	15
Inter Digit Timer	10
Lockout Timer	45

Figure 5: Class of Service (General)

Call Duration	
Call Duration	10
Call Duration Forced Cleardown Timer	0
Enable Call Duration Limit on External Calls	No
Enable Call Duration Limit on Internal Calls	No
Call Forwarding/Rerouting	
Call Forward - Delay	0
Call Forward No Answer Timer	15
Call Forward Override	No
Call Forwarding (External Destination)	No
Call Forwarding (Internal Destination)	Yes
Call Forwarding Accept	Yes
Call Reroute after CFFM to Busy Destination	No
Call Forwarding Reminder Ring (CFFM and CFIAH only)	No
Disable Call Reroute Chaining On Diversion	No
Follow Reroute on Disabled Forwarding	No
Group Call Forward Follow Me Accept	No
Group Call Forward Follow Me Allow	No
Third Party Call Forward Follow Me Accept	No
Third Party Call Forward Follow Me Allow	No
Use Held Party Device for Call Re-routing	Yes
Call Hold	
Call Hold	Yes
Call Hold - Retrieve with Hold Key	No
Call Hold Remote Retrieve	Yes

Figure 6: Class of Service Options (General) – Cont.

Call Hold Timer	30
Local Music On Hold source	No
Music on Hold on Transfer	Yes
Use Called Party Call Hold Timer	No
Call Park	
Call Park Timer	180
Call Park-Allowed To Park	No
Call Pickup	
Allow Directed Call Pickup Of Attendant Call	No
Call Pickup Dialed Accept	Yes
Call Pickup Directed Accept	Yes
Call Pickup Display	No
Call Privacy	
Call Privacy	No
Calling Party Name Substitution	No
Name Suppression on outgoing Trunk Call	No
Privacy Released	No
Public Network Identity Provided	No
Call Waiting	
Call Waiting Swap	No
ONS CLASS/CLIP: Visual Call Waiting	Yes
Campon	
Auto Campon Timer	10
Campon Recall Timer	10
Direct Voice Call	
Direct Voice Call - Accept	No

Figure 7: Class of Service Options (General) – Cont.

Direct Voice Call - Allow	No
Direct Voice Call - Maximize Volume	No
Display	
After Answer Display Time	
Calling Name Display - Internal - ONS	Yes
Calling Number Display - Internal - ONS	Yes
Display ANI/DNIS/ISDN Calling/Called Number	No
Display ANI/ISDN Calling Number Only	No
Display Caller ID on multicall/keylines	No
Display Caller ID On Multicall/Keylines Timer	5
Display Caller ID On Single Line Displays For Forwarded Calls	No
Display Dialed Digits during Outgoing Calls	No
Display DNIS/Called Number Before Digit Modification	No
Display DNIS on Key Label	No
Display Held Call ID on Transfer	No
Display Transfer Destination on Recall	No
Hot Desk External User - Display Internal Calling ID	No
Maintain Ringing Party During Recall	No
Non-Prime Public Network Identity	No
Originator's Display Update In Call Forwarding/Rerouting	No
Prefer Call Forwarding/Rerouting Information	No
Prefer Name for Call Information	No
Suppress Delivery of Caller ID Display between Sets	No
Suppress Delivery of Caller ID Display between Sets - Override	No
Suppress Display Of Account Code Numbers	No
Suppress Redial Display	No

Figure 8: Class of Service Options (General) – Cont.

Fax	
Campon Tone Security	No
External Trunk Standard Ringback	No
Fax Capable	No
Return Disconnect Tone When Far End Party Clears	No
HCI	
HCI/CTI/TAPI Call Control Allowed	Yes
HCI/CTI/TAPI Monitor Allowed	Yes
Hot Desk	
Green BLF Lamp for Logged in Hotdesk User	No
Hot Desk External User - Allow Mid-Call Features	Yes
Hot Desk External User - Answer Confirmation	Yes
Hot Desk External User - Dial Tone on Call Complete	Yes
Hot Desk External User - Permanent Login	No
Hot Desk External User - Remote MWI Enable Feature Access Code	
Hot Desk External User - Remote MWI Disable Feature Access Code	
Hot Desk Login Accept	Yes
Hot Desk Remote Logout Enabled	No
Miscellaneous	
Backlighting - Enabled	Yes
Clear All Features Remote	No
Enbloc Dialing - Enabled	No
Force Device Busy If Any Line In Use	No
Handset Volume Adjustment Saved	No
Head Set Switch Mute	No
Long Key Press Timer	0

Figure 9: Class of Service Options (General) – Cont.

Multi-Color LED Support - Disable	No
Phone Lock	No
Reseize Timer	180
Timed Reminder Allowed	Yes
User Inactivity Timer	0
Paging	
Group Page Accept	No
Group Page Allow	No
Loudspeaker Pager Equivalent Zone Override Security	No
Loudspeaker Pager Override	Yes
Pager Access All Zones	Yes
Pager Access Individual Zones	No
PC Port	
PC Port On IP Device - Disable	No
RAD	
Answer Plus Delay To Message Timer	20
Answer Plus Expected Off-hook Timer	30
Answer Plus Message Length Timer	10
Answer Plus System Reroute Timer	0
Recorded Announcement Device	No
Recorded Announcement Device - Advanced	No
Ringing	
Delay Ring Timer	10
No Answer Recall Timer	17
Ringing Line Select	No
Ringing Timer	180

Figure 10: Class of Service Options (General) – Cont.

SMDR	
SMDR External	No
SMDR Internal	No
Trunk	
ANI/DNIS/ISDN Number Delivery Trunk	No
DASS II OLI/TLI Provided	No
Public Network Access via DPNSS	Yes
Public Network To Public Network Connection Allowed	No
Public Trunk	No
R2 Call Progress Tone	No
Suppress Simulated CCM after ISDN Progress	No
Trunk Calling Party Identification	Yes
Trunk Flash Allowed	No
Two B-Channel Transfer Allowed	No
Voice Mail	
COV/ONS/E&M Voice Mail Port	No
ONS VMail-Delay Dial Tone Timer	5

Figure 11: Class of Service Options (General) – Cont.

4.2.3 Trunk Attributes

Navigation: **Trunks → Trunk Attributes**

1. Set **Direct Inward Dialing Service**= On
2. Set **Class of Service**= 18 as our example was used
3. Set **Dial In Trunks Incoming Digit Modification–Absorb** = 1
4. Set **Trunk Label**= Nexmo as our example was used
5. The remaining attributes are set to default as shown below

Trunk Attributes

5	No	Off	On	18	1	9600	1	Nexmo

Trunk Service Number	5
Release Link Trunk	No
Call Recognition Service	Off
Direct Inward Dialing Service	On
Class of Service	18
Class of Restriction	1
Baud Rate	9600
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	1
Dial In Trunks Incoming Digit Modification - Insert	
Dial In Trunks Answer Point	
Dial In Trunks Insert Forwarding Information	No
Trunk Label	Nexmo

Figure 12: Trunk Attributes

4.2.4 SIP Peer Profile for Nexmo

SIP trunks are configure to route off-net calls between the Mitel Border Gateway (MBG) and Nexmo Network. Calls are configure to route via Mitel MBG to sip.nexmo.com. The firewall was open for the list of IPs in the portal provided by Nexmo.

1. Navigate to **Trunks → SIP → SIP Peer Profile → Basic** tab
2. Set **SIP Peer Profile Label**: *Nexmo* is used as an example
3. Set **Network Element**: Nexmo-com in this example
4. Set **Address Type:** FQDN: sip.nexmo.com
5. Administration Options:
    - Set **Maximum Simultaneous Calls**: 4 is used as an example
    - Set **Outbound Proxy Server:** Nexmo-MBG
    - Set **Trunk Service**: 5 is used as an example
    - Set **Zone**: 1 is used as an example
6. Authentication Options:
    - Set **User Name**: Enter the user name provided by Nexmo.com
    - Set **Password:** Enter the password provided by Nexmo.com

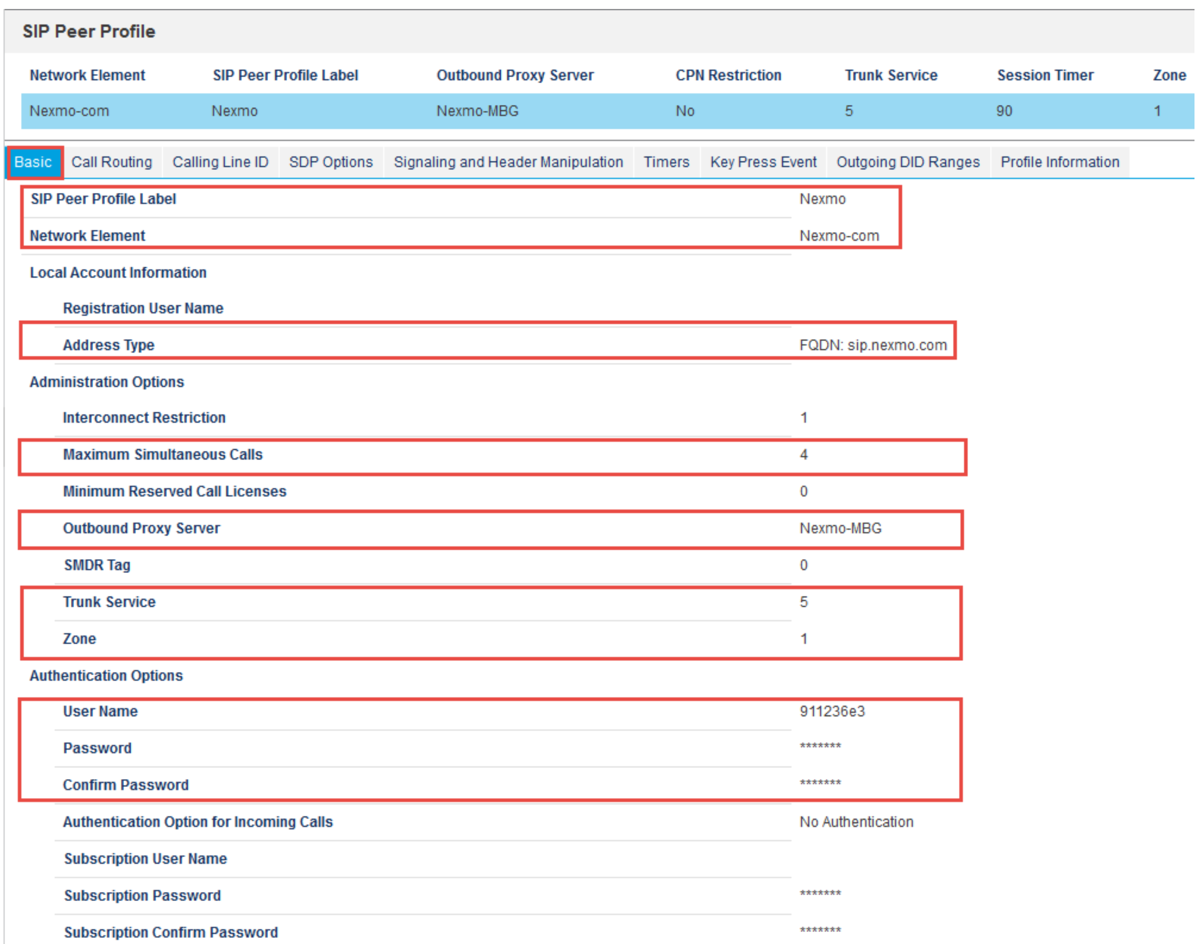


Figure 13: SIP Peer Profile

7. Select the **Call Routing** tab

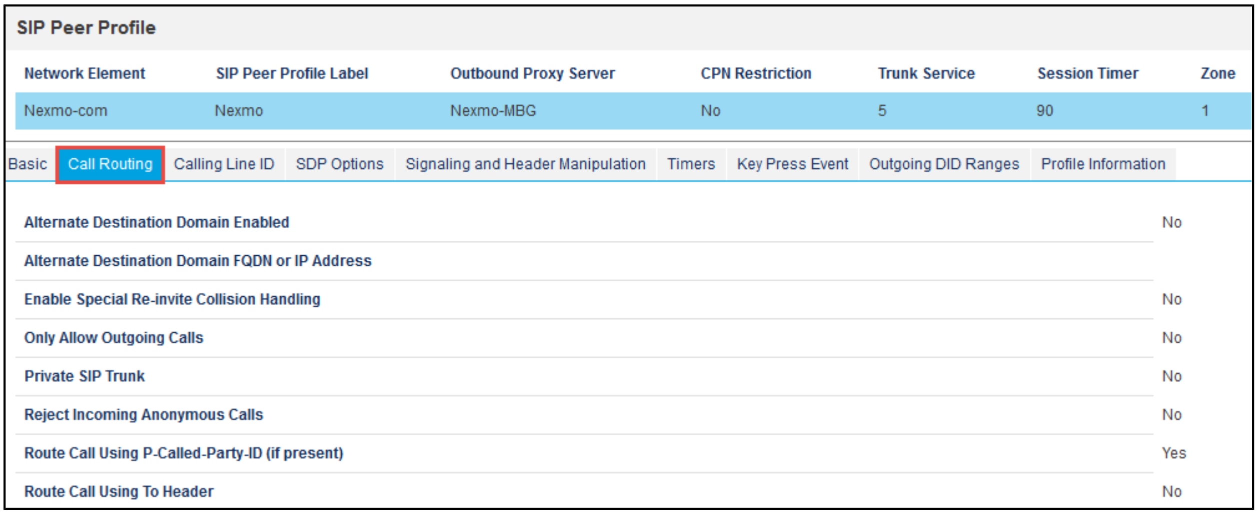


Figure 14: SIP Peer Profile

8. Select the **Calling Line ID** tab

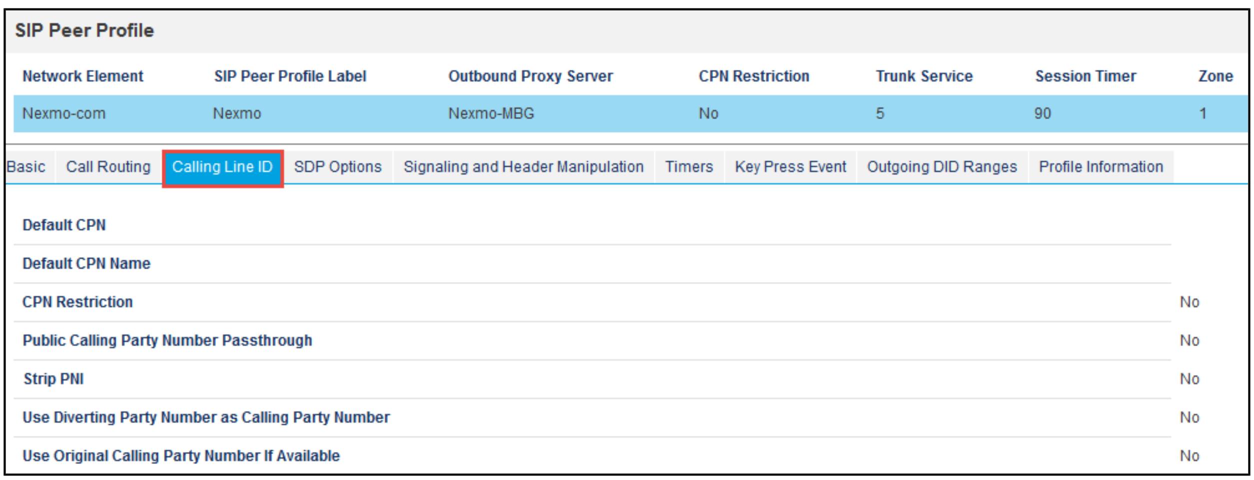
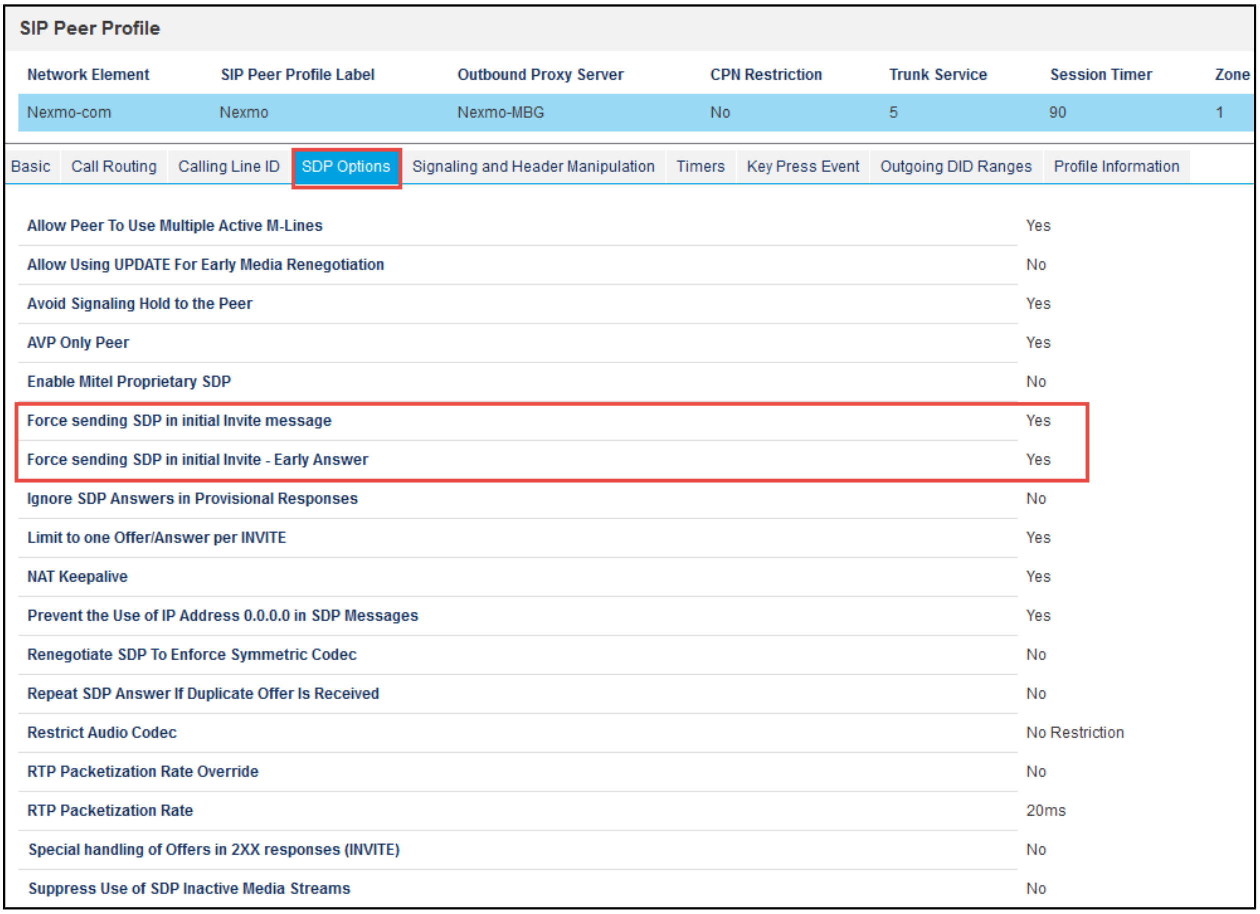


Figure 15: SIP Peer Profile – Cont.

9. Select the **SDP Options** tab
10. Set **Force sending SDP in initial invite message:** Yes
11. Set **Force sending SDP in initial invite – Early Answer:** Yes



SIP Peer Profile

Network Element	SIP Peer Profile Label	Outbound Proxy Server	CPN Restriction	Trunk Service	Session Timer	Zone
Nexmo-com	Nexmo	Nexmo-MBG	No	5	90	1

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	Yes
Ignore SDP Answers in Provisional Responses	No
Limit to one Offer/Answer per INVITE	Yes
NAT Keepalive	Yes
Prevent the Use of IP Address 0.0.0.0 in SDP Messages	Yes
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	No

Figure 16: SIP Peer Profile – Cont.

12. Select the **Signaling and Header Manipulation** tab

13. Set **E.164: Enable sending '+':** Yes
14. Set **E.164: Add '+' if digit length > N digits:** 10
15. Set **E.164: Do not add '+'to Called Party:** Yes
16. Set **Use P-Early-Media Header:** sendrcv

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
Domain for Trunk Context	
E.164: Enable sending '+'	Yes
E.164: Add '+' if digit length > N digits	10
E.164: Do not add '+' to Emergency Called Party	No
E.164: Do not add '+' to Called Party	Yes
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
Multilingual Name Display	No
Only use SDP to decide 180 or 183	Yes
Prefer From Header for Caller ID	No
Require Reliable Provisional Responses on Outgoing Calls	Yes
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	sendrecv
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 17: SIP Peer Profile – Cont.

17. Select the **Timers** tab

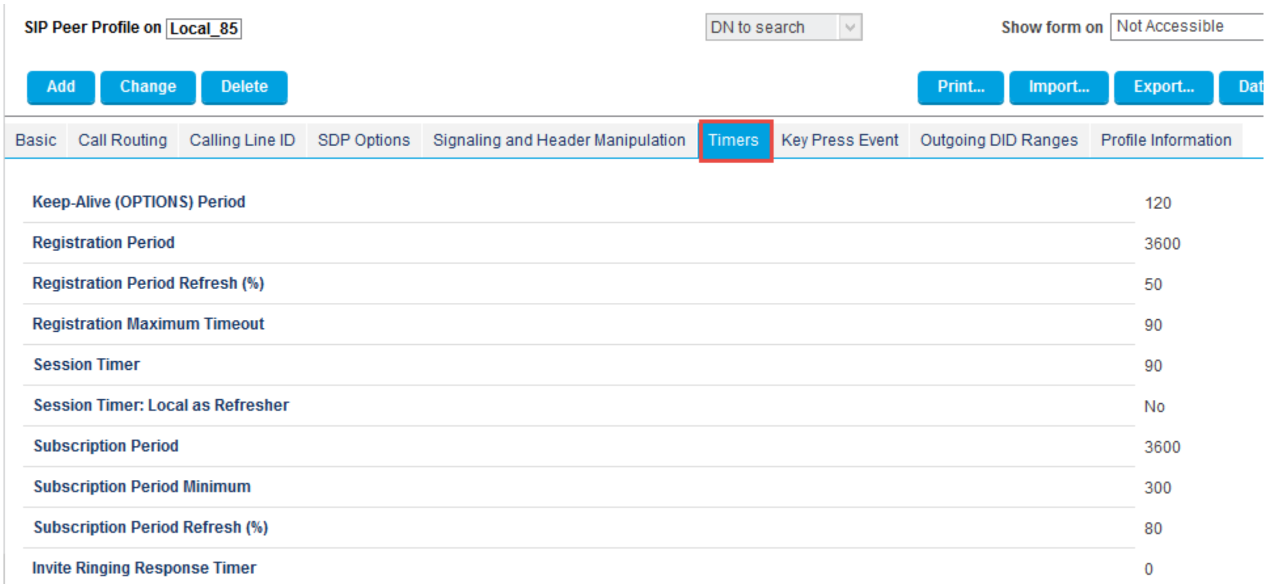


Figure 18: SIP Peer Profile – Cont.

18. Select the **Key Press Event** tab

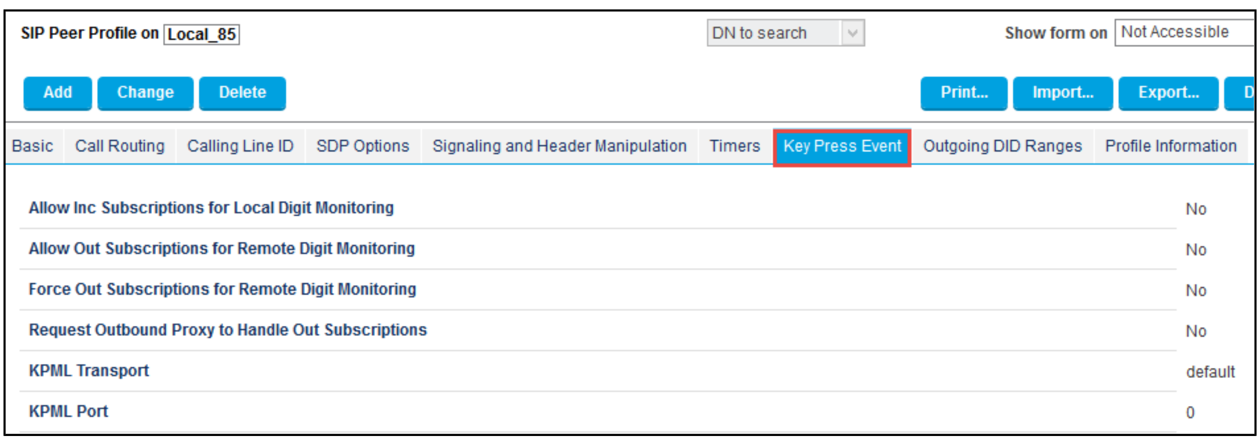


Figure 19: SIP Peer Profile – Cont.

4.2.5 Dial Plan

1. Navigate to **Call Routing → ARS Routes**
2. An ARS Route is configured for Nexmo trunk routing

ARS Routes

Route Number	Routing Medium	Trunk Group Number	SIP Peer Profile	PBX Number / Cluster Element ID	COR Group Number	Digit Modification Number	Digits Before Outpulsing	Route Type	Compression
1	SIP Trunk		[redacted]		1	1		Non-verified Account	Off
2					1	1			Off
3					1	1			Off
4					1	1			Off
5					1	1			Off
6					1	1			Off
7	SIP Trunk		Nexmo		1	5			Off
8					1	1			Off

Figure 20: ARS Route List

3. Navigate to **Call Routing → ARS Digits Dialed**
4. Prefix digits are configure to route calls to PSTN and International networks

ARS Digits Dialed

Digits Dialed	Number of Digits to Follow	Termination Type	Termination Number
7	10	Route	6
71	10	Route	6
8	11	Route	5
9	11	Route	7
9011	10	Route	7

Figure 21: ARS Digits Dialed

4.3 **Configuring Mitel Border Gateway (MBG)**

4.3.1 Network Profile

Configure the LAN and WAN IP addresses on the MBG. The IP addresses shown are for illustration only, the actual IP address can vary.

1. Navigate to **Applications → MiVoice Border Gateway → System Configuration → Network Profiles**

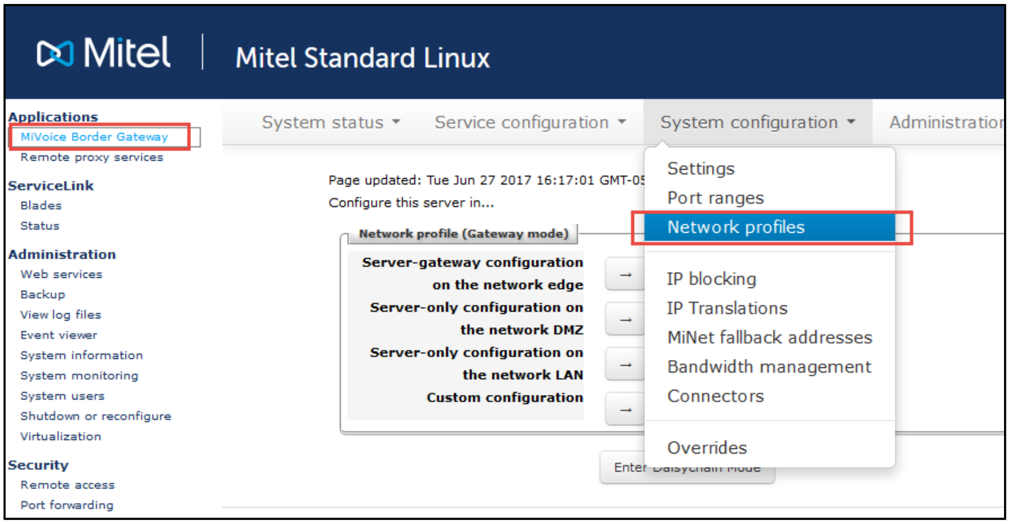


Figure 22: MBG Network Profiles

2. Navigate to **System Configuration → Network profiles → Custom Configuration**
3. Set **LAN IP address to ICP**: 10.64.3.168 is used in this example
4. Set **WAN IP address**: 192.65.79.xxx (as provided by your network administrator)

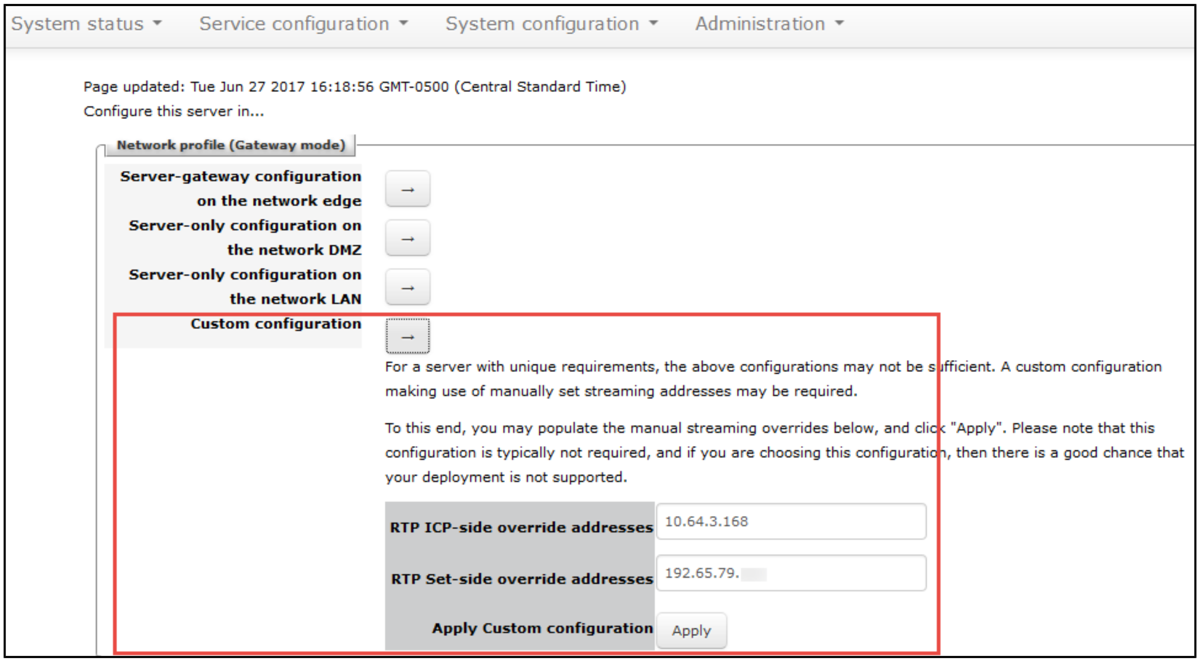


Figure 23: Network Profiles Configuration

4.3.2 ICP Configuration

Configuration for MiVoice Business ICP

1. Navigate to **Service Configuration → ICPs**

ICP Information

Default for MiNet	Default for SIP	Name	Hostname or IP address	Type	Installer password	SIP capabilities	Indirect call recording capable
◉	◉	10.35.31.85	10.35.31.85	MiVoice Business		UDP	✘

Figure 24: ICP Configuration

4.3.3 SIP Trunk Information

The SIP trunks were configure as follows to meet the ITSP requirements. The ITSP required outgoing calls to route to the FQDN sip.nexmo.com. The FQDN resolved to a single IP address.

Inbound calls routing to the MBG utilized different IP addresses that were not resolved by FQDN sip.nexmo.com. Individual trunks had to be configure for each IP address used by the ITSP. Otherwise, the MBG does not know about these IP addresses and calls are not trusted.

- Outbound calls utilized the ITSP FQDN sip.nexmo.com to route calls
- Inbound trunks were configure for calls to be trusted into the MBG. The following IP addresses were configured as separate trunks as shown below.

1. Navigate to **Service Configuration → SIP Trunking**

SIP trunk information

+

Enabled	Name	Remote endpoint	Number of routing rules	PRACK support	Remote RTP framesize (ms)	RTP address override	Local streaming
✓	Nexmo_Outbound	sip.nexmo.com : 5060	2	Use master setting	20		False
✓	nexmo_inbound1	174.37.245.[redacted] : 5060	2	Use master setting	0		False
✓	nexmo_inbound2	5.10.112.[redacted] : 5060	2	Use master setting	0		False
✓	nexmo_inbound3	5.10.112.[redacted] : 5060	2	Use master setting	0		False
✓	nexmo_inbound4	119.81.44.[redacted] : 5060	2	Use master setting	0		False
✓	nexmo_inbound5	119.81.44.[redacted] : 5060	2	Use master setting	0		False

Figure 25: SIP Trunk Service Configuration

4.3.3.1 Outbound Trunk Configuration

Nexmo_Outbound Trunk (Outbound trunk) details shown below.

1. Set **Name:** Nexmo_Outbound used in this example
2. Set **Remote trunk endpoint address:** sip.nexmo.com
3. Set **Remote RTP framesize(ms):** 20ms
4. Set **Authentication username, password:** (see ITSP for this information)
5. Click on **Enabled checkbox** to enable trunk

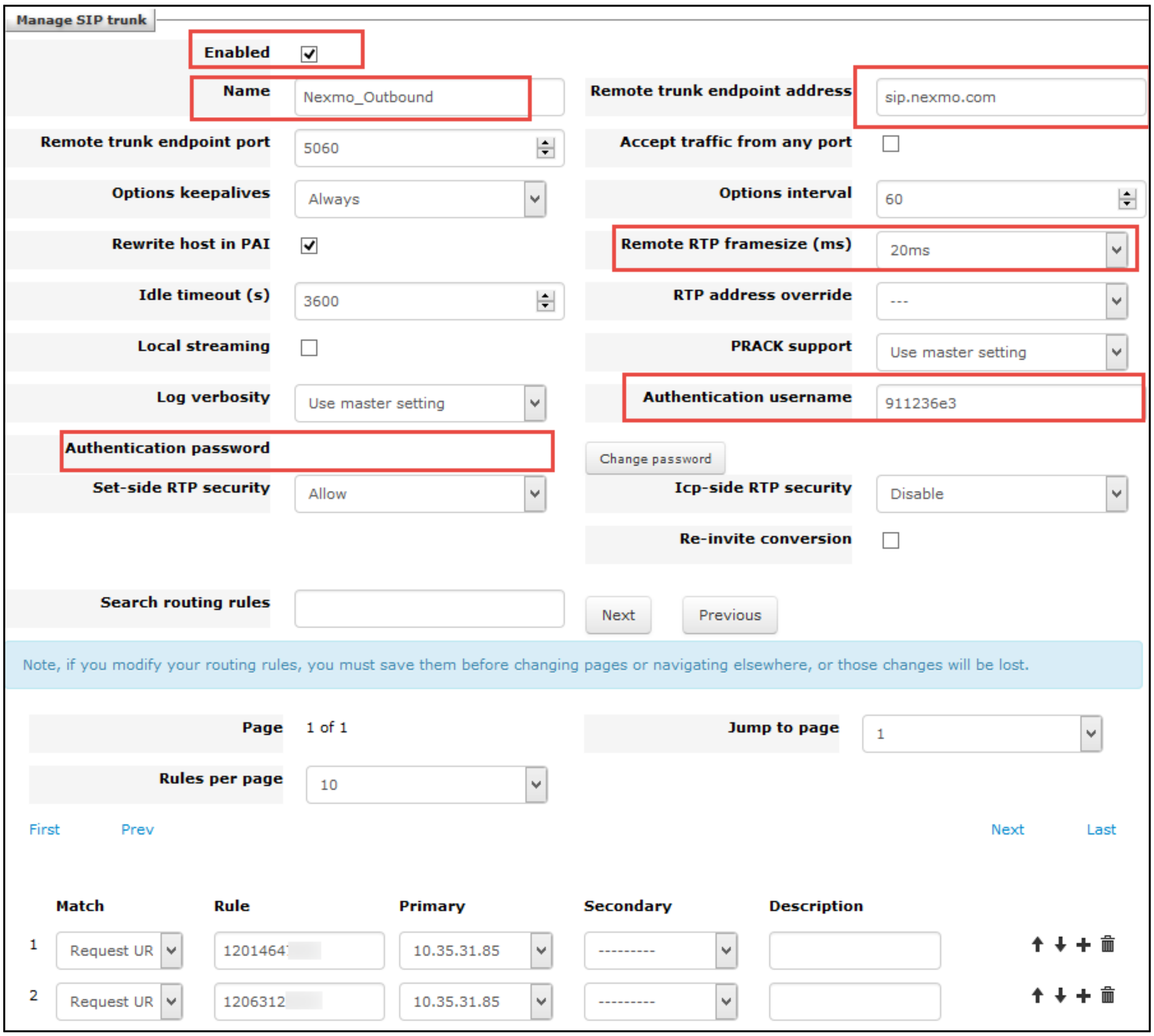


Figure 26: Outbound Trunk Configuration

4.3.3.2 Inbound Trunk Configuration

Inbound calls routing to the MBG from the ITSP arrived from different IP addresses and trunks had to be configure for each IP address used by the ITSP. Otherwise, the MBG does not know about these IP addresses and calls are not trusted.

Shown below are configuration used for the inbound Trunks. The same configuration is required for each inbound IP address as provided by the ITSP.

1. **Enabled:** Check to enable trunk
2. Set the **Name:** Nexmo_inbound1 used in this example
3. Set the **Remote trunk endpoint address:** Enter the IP address provided by ITSP
4. Added **Rules for DIDs to Match Request URI** as shown

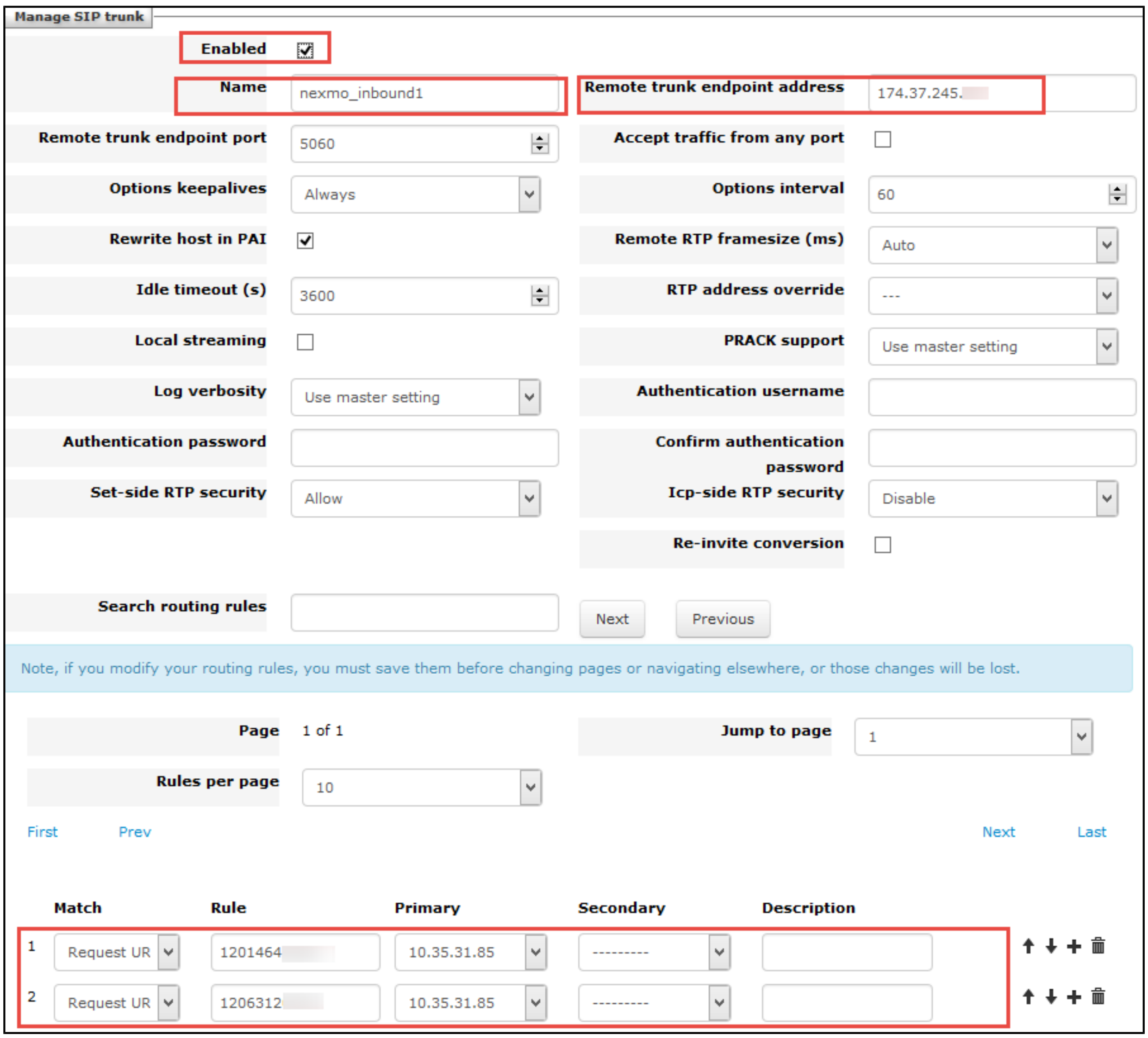


Figure 27: Inbound Trunk 1 Configuration

Inbound Trunk Configuration for DID 5.10.112.xxx

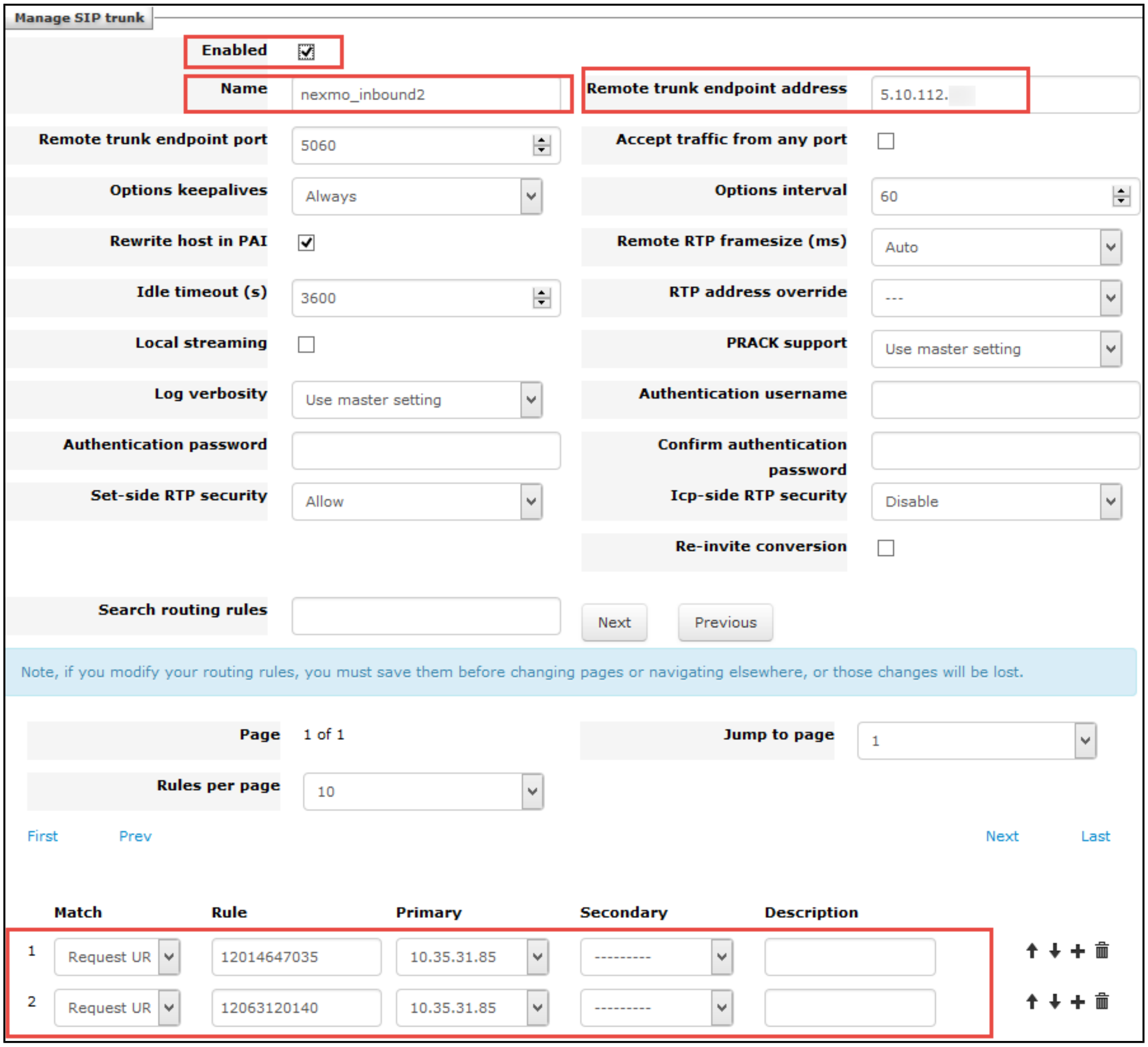


Figure 28: Inbound Trunk 2 Configuration

Inbound Trunk Configuration for DID 5.10.112.xxx.

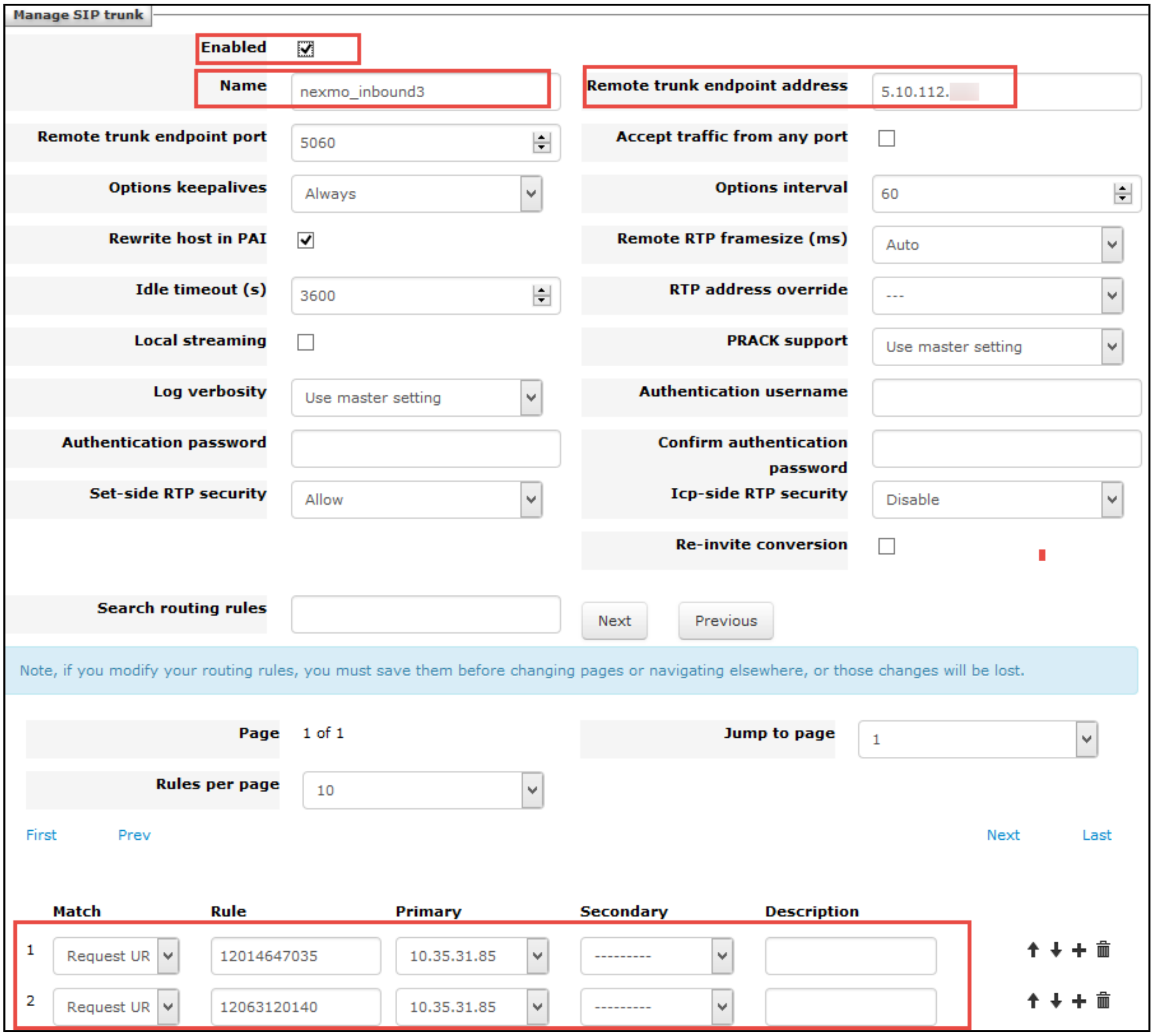


Figure 29: Inbound Trunk 3 Configuration

Inbound Trunk Configuration for DID 119.81.44.xxx.

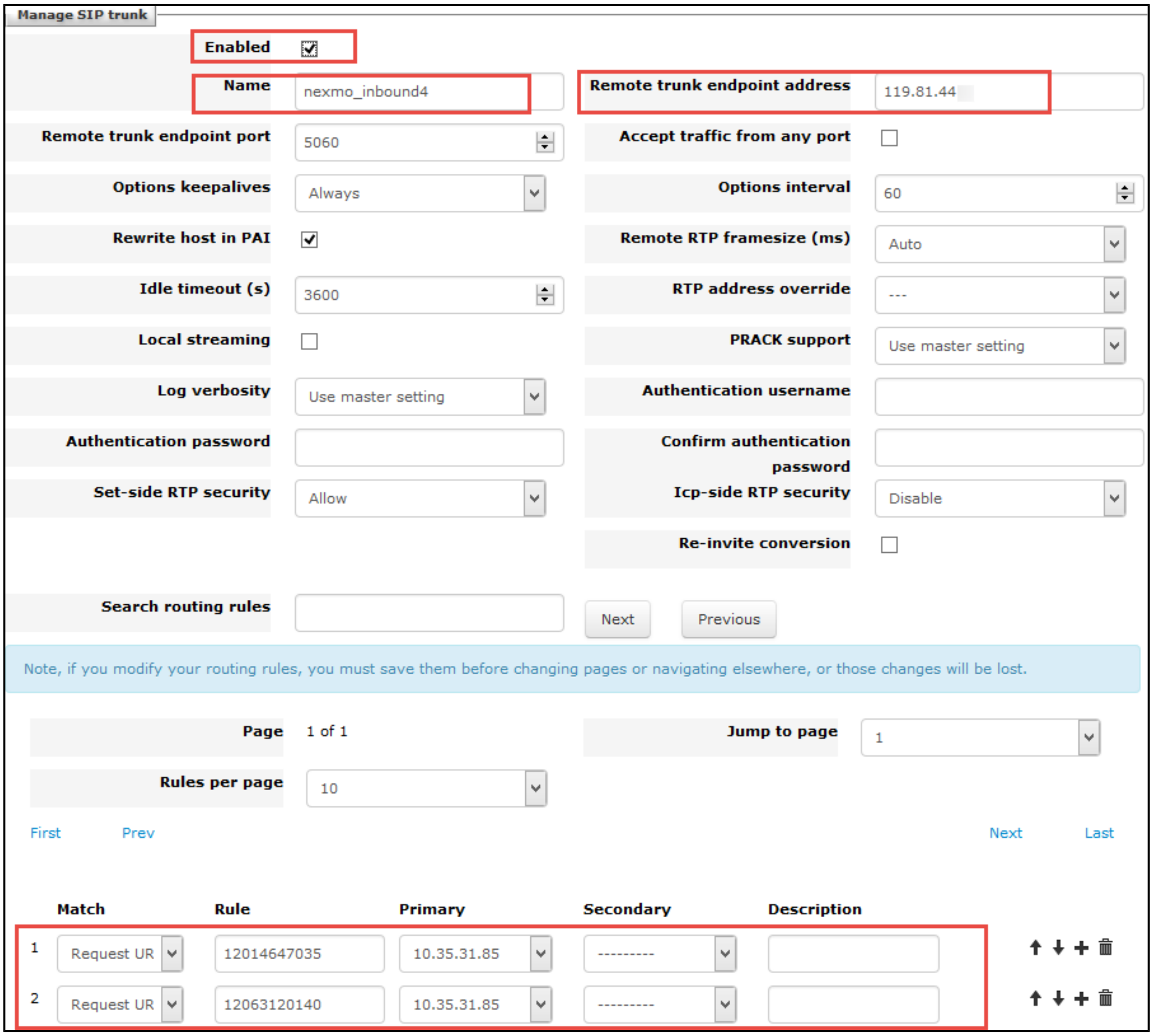


Figure 30: Inbound Trunk 4 Configuration

Inbound Trunk Configuration for DID 119.81.44.xxx.

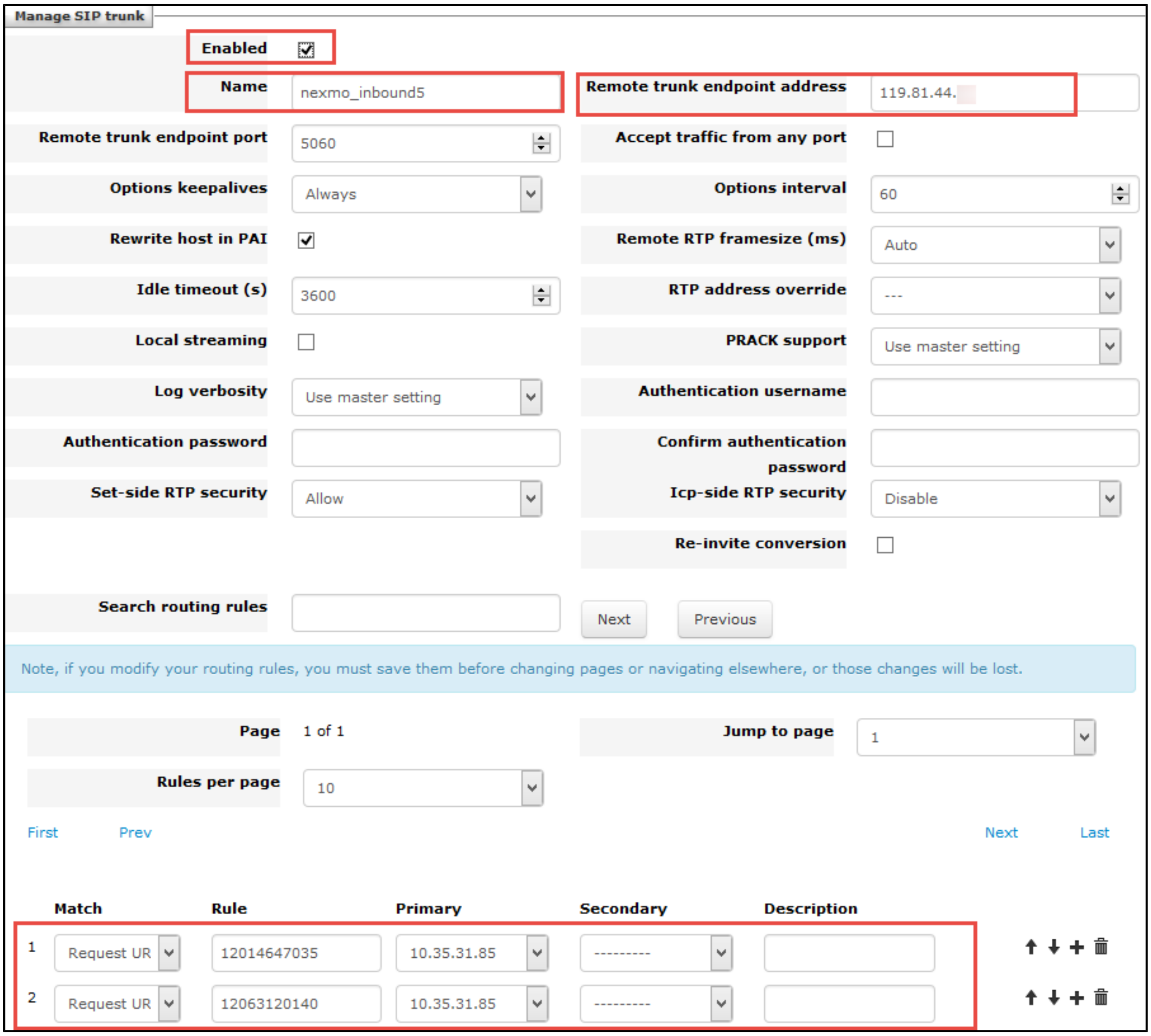


Figure 31: Inbound Trunk 5 Configuration

4.3.4 System Configuration Settings

This configuration did not change for this ITSP but shown for informational purposes.

Navigation: System Configuration → Settings

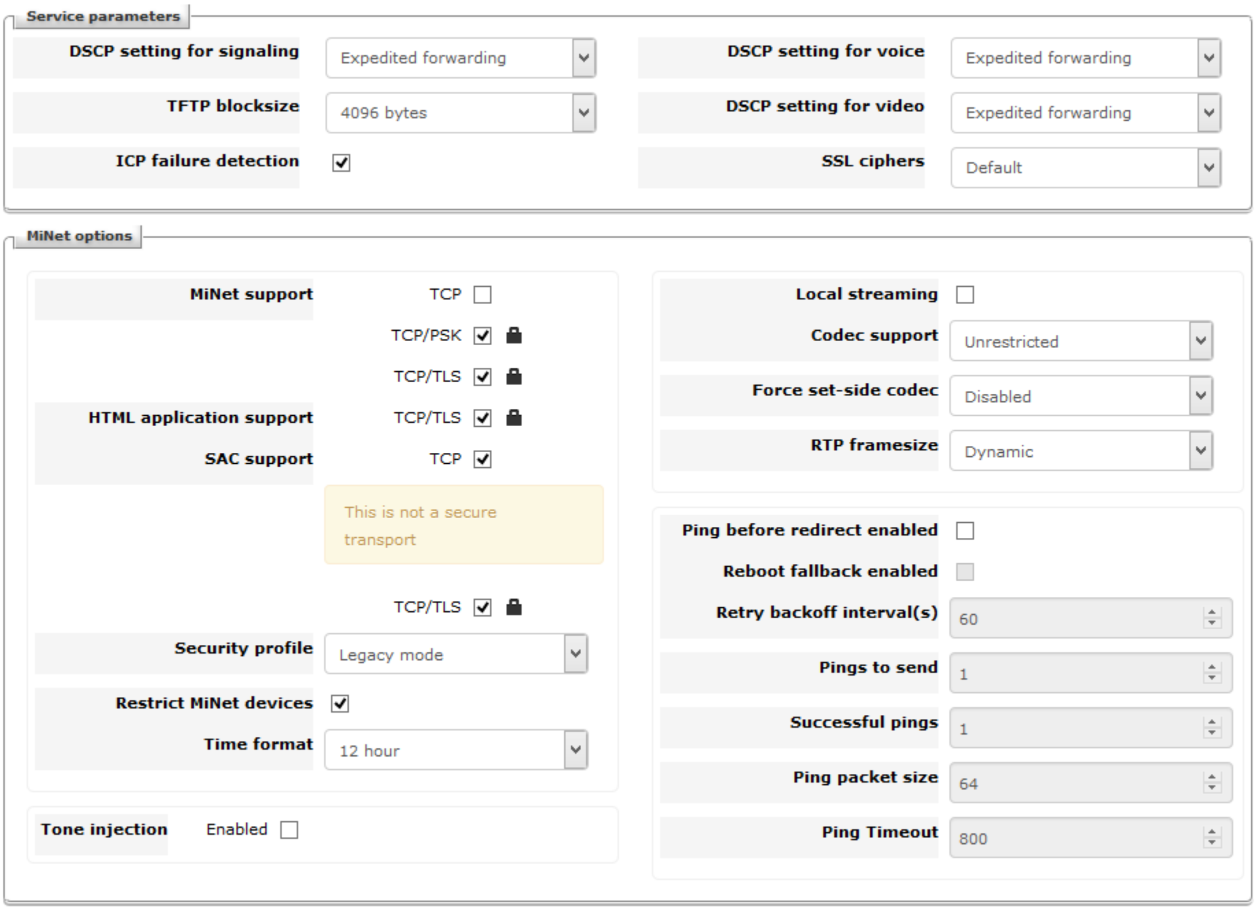


Figure 32: System Configuration

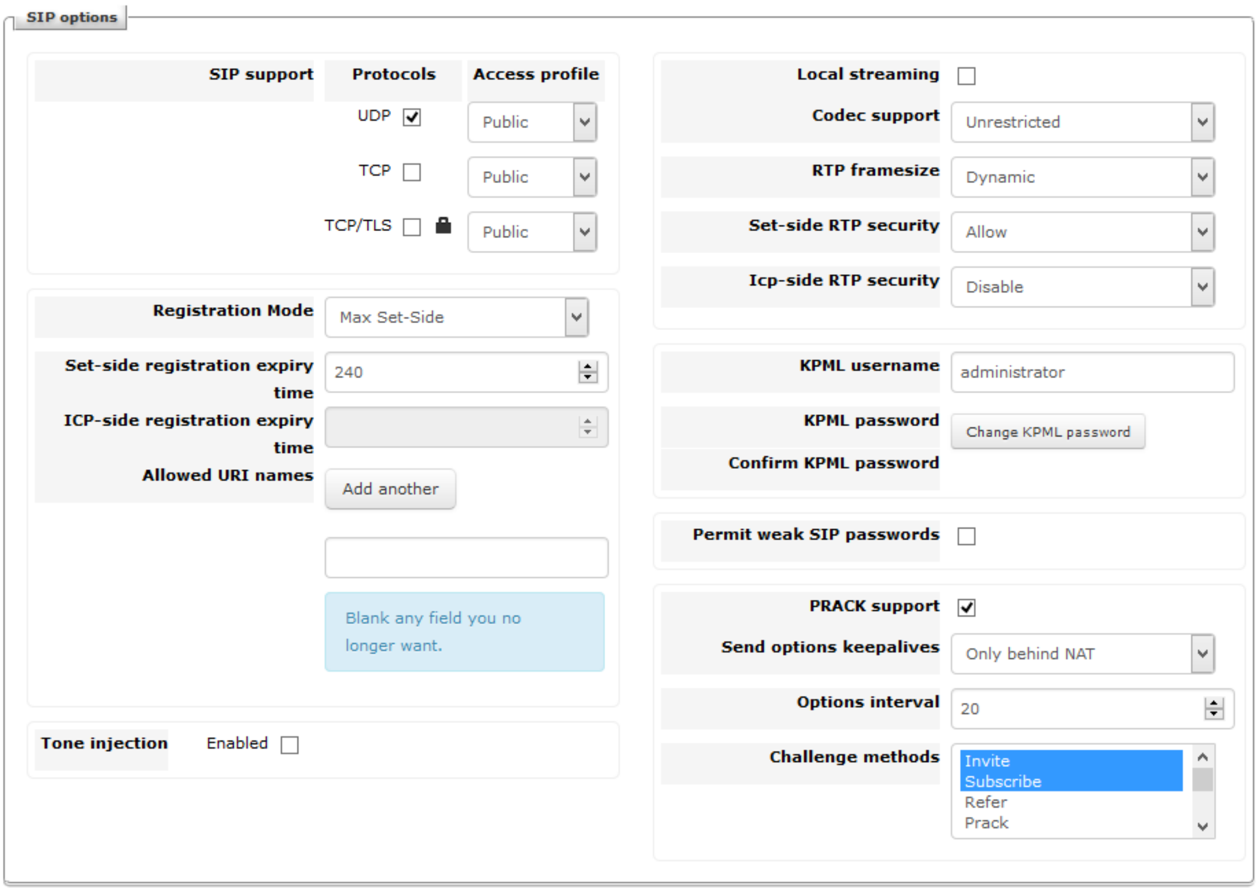


Figure 33: System Configuration – Cont.

4.4 Configure Numbers in Nexmo Account

1. Login to the Nexmo account using the credentials provided at the time of registration. A **Key** and **Secret** is display on the dashboard and can be use as the username and password for Registration SIP Trunks.

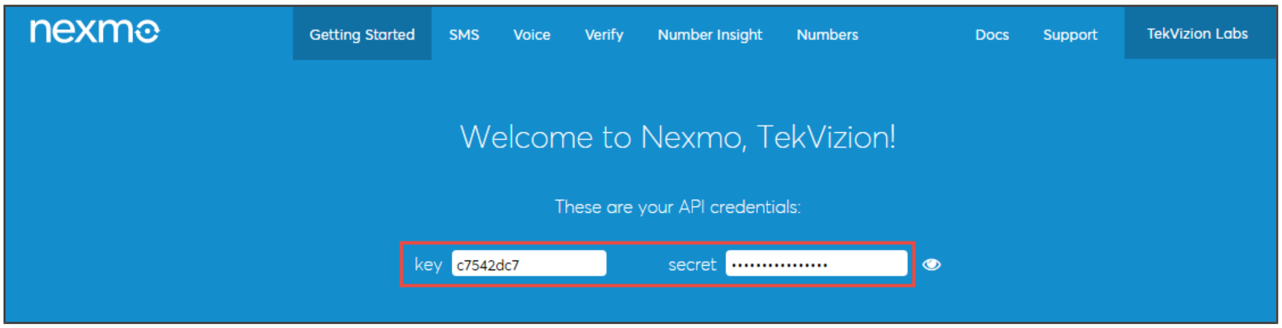


Figure 34: Nexmo Dashboard

2. In order to provide the URL to which the call has to be routed from Nexmo, navigate to the **Numbers** tab
3. Click **Edit** against each number as shown below

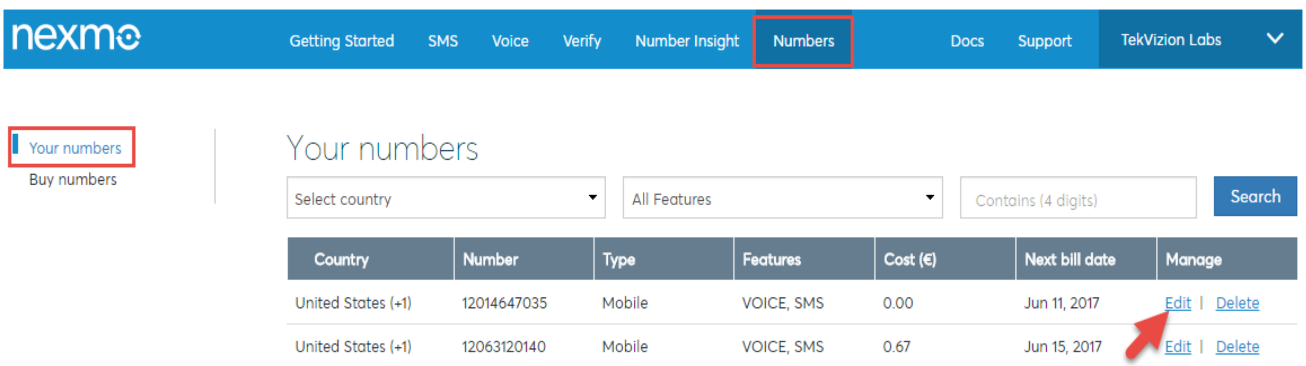


Figure 35: Nexmo DID Numbers

4. A pop-up will be displayed
5. Select the "**Forward to**" and provide the URL to which the calls route
6. Click **Update** to save the changes

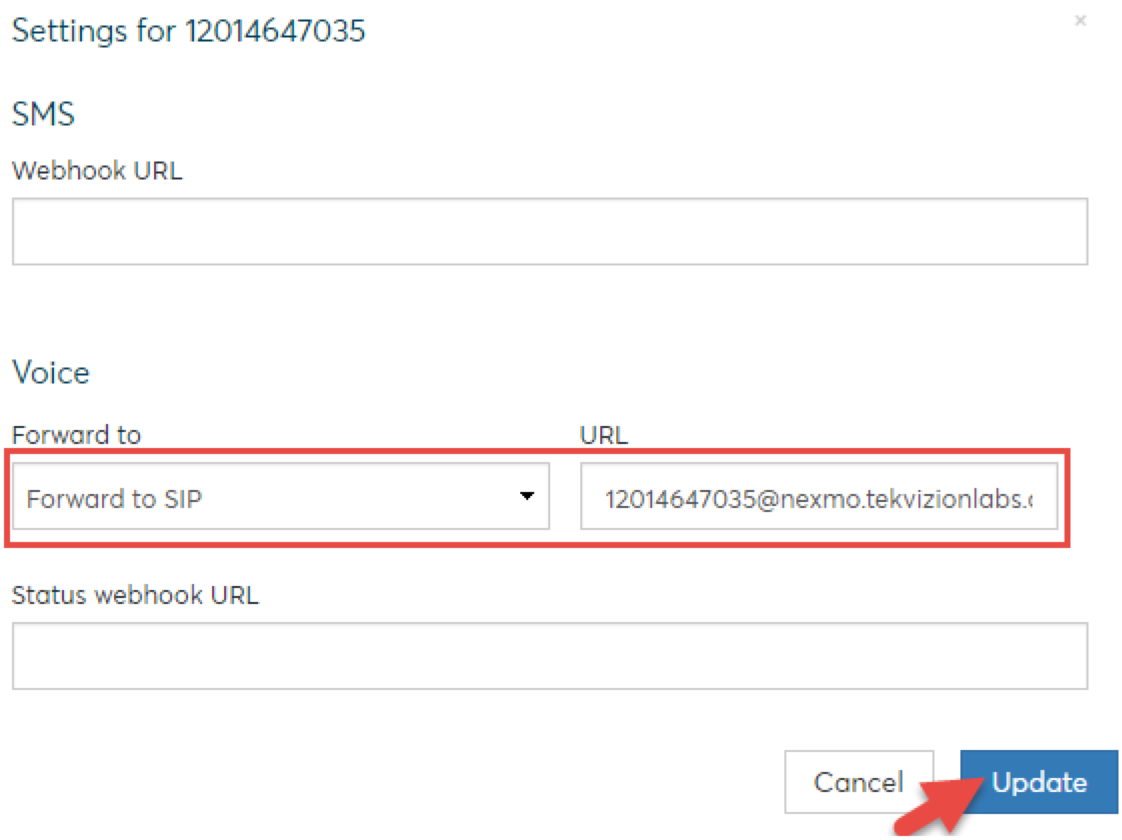


Figure 36: Your Numbers – Cont.