



Nexmo SIP Trunking Configuration Guide

Mitel MiVoice Business 8.0 PR2 and Mitel Border Gateway 9.4.0.29

July 2017

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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.

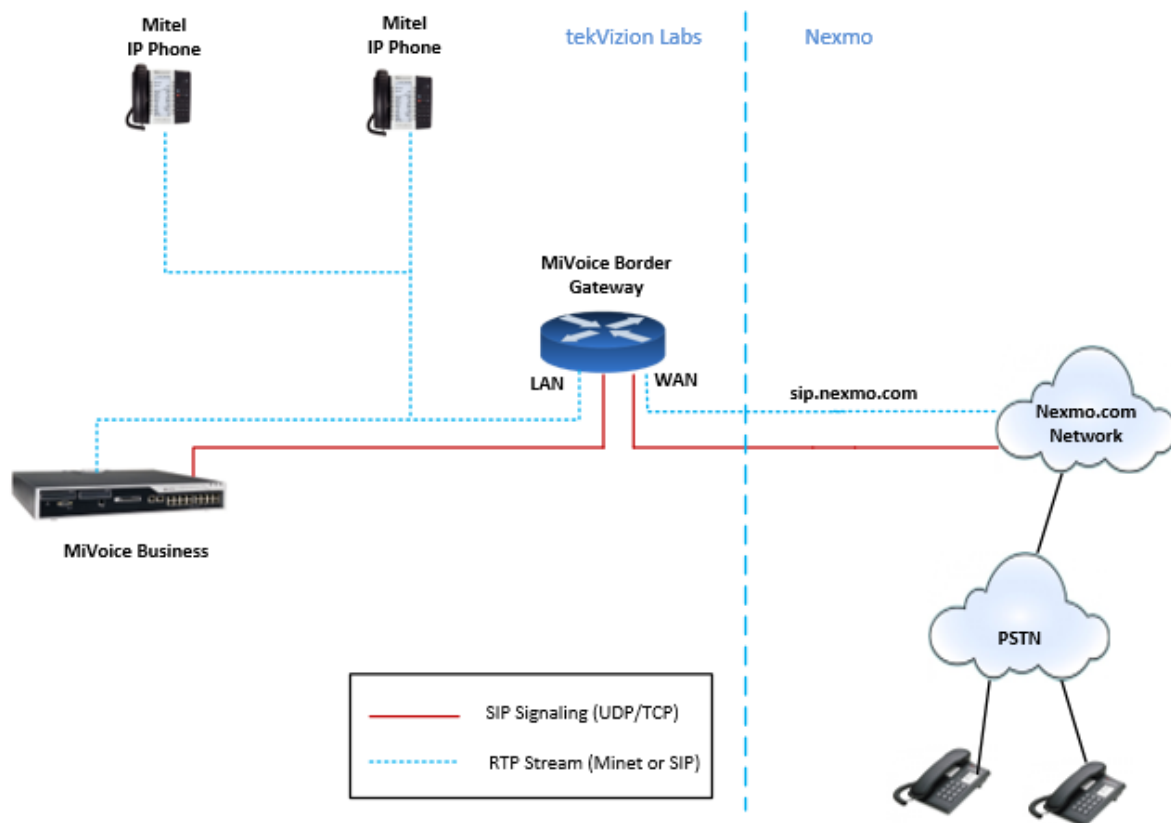


Figure 1: Topology Diagram

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.

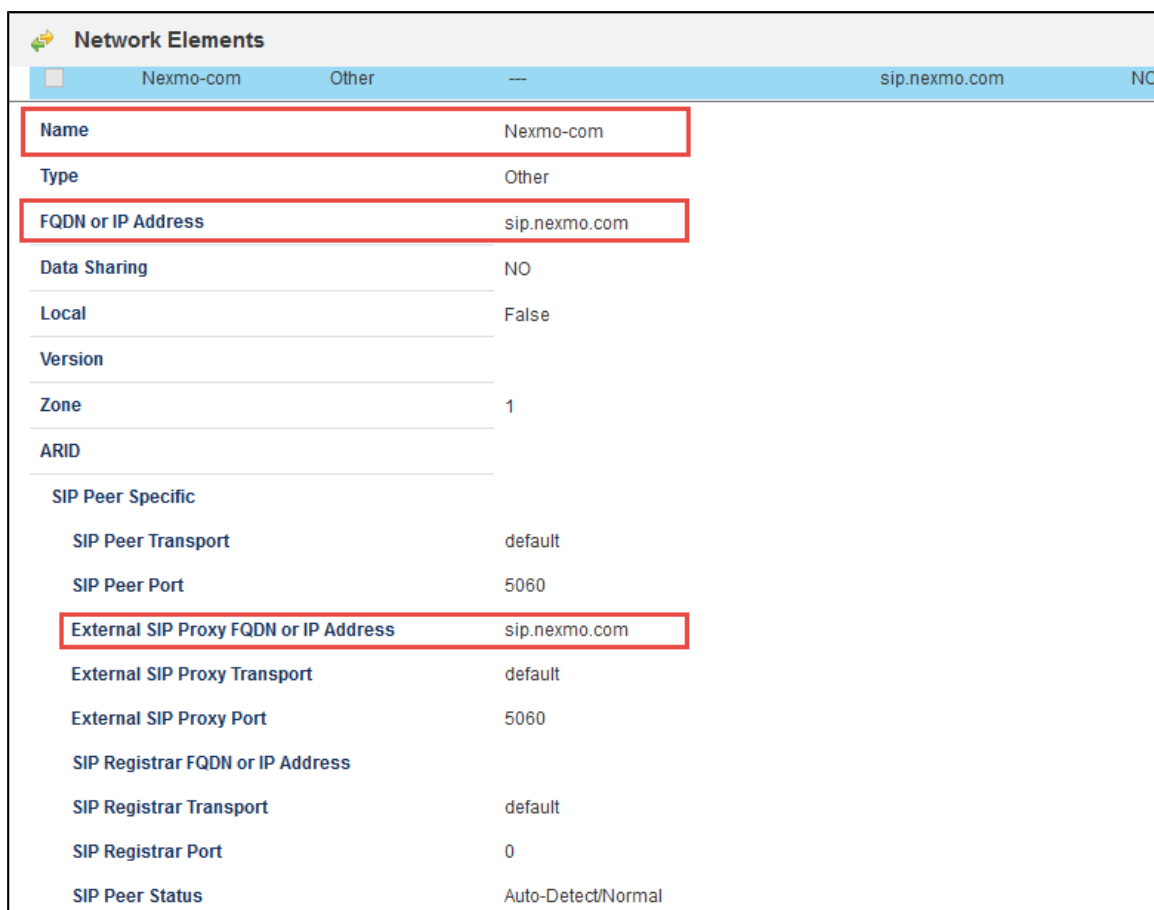
Table 1 – IP Addresses

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)

Network Elements	
<input type="checkbox"/>	Nexmo-MBG Outbound Proxy --- 10.64.3.168
Name	Nexmo-MBG
Type	Outbound Proxy
FQDN or IP Address	10.64.3.168
Data Sharing	NO
Local	False
Version	
Zone	1
ARID	
Outbound Proxy Specific	
Outbound Proxy Transport Type	default
Outbound Proxy Port	5060

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

Mitel

MiVoice Business

Node 'Local_85' Alarm Status: Major 2017-Jun-26 09:52:08

Local_85

Codec Settings

System Properties

System Settings

Date and Time

Controller Registry

System Ports

Application Logical Ports

System Feature Settings

System Options

Shared System Options

Class of Service Options

Class of Service Options on Local_85

DN to search

Show for

Change

Copy

Print...

Import

< Page 2 of 7 >

Go to

Class of Service Options

13	SIP Trunk CoS
14	CC CoS
15	HotDesk
16	Micollab Client
17	AWV Conference
18	Nexmo
19	

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
Resize 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
Ringling	
Delay Ring Timer	10
No Answer Recall Timer	17
Ringling Line Select	No
Ringling 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
<hr/>									
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 configured to route off-net calls between the Mitel Border Gateway (MBG) and Nexmo Network. Calls are configured 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

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

SIP Peer Profile Label Nexmo

Network Element Nexmo-com

Local Account Information

Registration User Name

Address Type FQDN: sip.nexmo.com

Administration Options

Interconnect Restriction 1

Maximum Simultaneous Calls 4

Minimum Reserved Call Licenses 0

Outbound Proxy Server Nexmo-MBG

SMDR Tag 0

Trunk Service 5

Zone 1

Authentication Options

User Name 911236e3

Password *****

Confirm Password *****

Authentication Option for Incoming Calls No Authentication

Subscription User Name

Subscription Password *****

Subscription Confirm Password *****

Figure 13: SIP Peer Profile

7. Select the **Call Routing** tab

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
-------	---------------------	-----------------	-------------	-----------------------------------	--------	-----------------	---------------------	---------------------

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 14: SIP Peer Profile

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

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
-------	--------------	------------------------	-------------	-----------------------------------	--------	-----------------	---------------------	---------------------

Default CPN	
Default CPN Name	
CPN Restriction	No
Public Calling Party Number Passthrough	No
Strip PNI	No
Use Diverting Party Number as Calling Party Number	No
Use Original Calling Party Number If Available	No

Figure 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
<div> Basic Call Routing Calling Line ID SDP Options Signaling and Header Manipulation Timers Key Press Event Outgoing DID Ranges Profile Information </div>						
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	sendrcv
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

SIP Peer Profile on **Local_85** DN to search Show form on Not Accessible

[Add](#) [Change](#) [Delete](#) [Print...](#) [Import...](#) [Export...](#) [Data](#)

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

Keep-Alive (OPTIONS) Period	120
Registration Period	3600
Registration Period Refresh (%)	50
Registration Maximum Timeout	90
Session Timer	90
Session Timer: Local as Refresher	No
Subscription Period	3600
Subscription Period Minimum	300
Subscription Period Refresh (%)	80
Invite Ringing Response Timer	0

Figure 18: SIP Peer Profile – Cont.

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

SIP Peer Profile on **Local_85** DN to search Show form on Not Accessible

[Add](#) [Change](#) [Delete](#) [Print...](#) [Import...](#) [Export...](#) [Data](#)

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

Allow Inc Subscriptions for Local Digit Monitoring	No
Allow Out Subscriptions for Remote Digit Monitoring	No
Force Out Subscriptions for Remote Digit Monitoring	No
Request Outbound Proxy to Handle Out Subscriptions	No
KPML Transport	default
KPML Port	0

Figure 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				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**

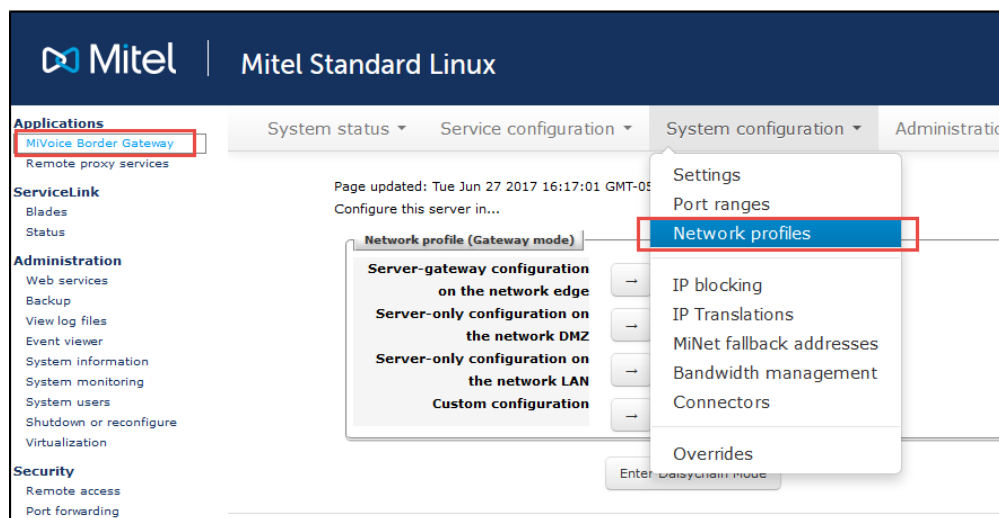


Figure 22: MBG Network Profiles

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)

System status ▾ Service configuration ▾ System configuration ▾ Administration ▾

Page updated: Tue Jun 27 2017 16:18:56 GMT-0500 (Central Standard Time)
Configure this server in...

Network profile (Gateway mode)

Server-gateway configuration on the network edge →

Server-only configuration on the network DMZ →

Server-only configuration on the network LAN →

Custom configuration →

For a server with unique requirements, the above configurations may not be sufficient. A custom configuration making use of manually set streaming addresses may be required.

To this end, you may populate the manual streaming overrides below, and click "Apply". Please note that this configuration is typically not required, and if you are choosing this configuration, then there is a good chance that your deployment is not supported.

RTP ICP-side override addresses: 10.64.3.168

RTP Set-side override addresses: 192.65.79.

Apply Custom configuration Apply

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
<input checked="" type="radio"/>	<input checked="" type="radio"/>	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 configured 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 configured 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 configured 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. : 5060	2	Use master setting	0		False
✓	nexmo_inbound2	5.10.112. : 5060	2	Use master setting	0		False
✓	nexmo_inbound3	5.10.112. : 5060	2	Use master setting	0		False
✓	nexmo_inbound4	119.81.44. : 5060	2	Use master setting	0		False
✓	nexmo_inbound5	119.81.44. : 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

Manage SIP trunk

Enabled ☒

Name Nexmo_Outbound

Remote trunk endpoint address sip.nexmo.com

Remote trunk endpoint port 5060

Options keepalives Always

Rewrite host in PAI ☒

Idle timeout (s) 3600

Local streaming ☐

Log verbosity Use master setting

Authentication password

Set-side RTP security Allow

Search routing rules

Remote RTP framesize (ms) 20ms

RTP address override ---

PRACK support Use master setting

Authentication username 911236e3

Change password

Icp-side RTP security Disable

Re-invite conversion ☐

Next Previous

Note, if you modify your routing rules, you must save them before changing pages or navigating elsewhere, or those changes will be lost.

Page 1 of 1

Jump to page 1

Rules per page 10

First Prev Next Last

	Match	Rule	Primary	Secondary	Description	
1	Request UR	1201464	10.35.31.85	-----		↑ ↓ + ✕
2	Request UR	1206312	10.35.31.85	-----		↑ ↓ + ✕

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

Manage SIP trunk

Enabled☒

Name

nexmo_inbound1

Remote trunk endpoint address

174.37.245.

Remote trunk endpoint port

5060

Options keepalives

Always

Rewrite host in PAI☒

Idle timeout (s)

3600

Local streaming☐

Log verbosity

Use master setting

Authentication password

Set-side RTP security

Allow

Accept traffic from any port☐

Options interval

60

Remote RTP framesize (ms)

Auto

RTP address override

PRACK support

Use master setting

Authentication username

Confirm authentication password

Icp-side RTP security

Disable

Re-invite conversion☐

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Note, if you modify your routing rules, you must save them before changing pages or navigating elsewhere, or those changes will be lost.

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	Match	Rule	Primary	Secondary	Description	
1	Request UR	1201464	10.35.31.85	-----		↑ ↓ + ✖
2	Request UR	1206312	10.35.31.85	-----		↑ ↓ + ✖

Figure 27: Inbound Trunk 1 Configuration

26

Inbound Trunk Configuration for DID 5.10.112.xxx

Manage SIP trunk

Enabled ☒

Name nexmo_inbound2

Remote trunk endpoint address 5.10.112.

Remote trunk endpoint port 5060

Accept traffic from any port ☐

Options keepalives Always

Options interval 60

Rewrite host in PAI ☒

Remote RTP framesize (ms) Auto

Idle timeout (s) 3600

RTP address override ---

Local streaming ☐

PRACK support Use master setting

Log verbosity Use master setting

Authentication username

Authentication password

Confirm authentication password

Set-side RTP security Allow

Icp-side RTP security Disable

Re-invite conversion ☐

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Note, if you modify your routing rules, you must save them before changing pages or navigating elsewhere, or those changes will be lost.

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	Match	Rule	Primary	Secondary	Description	
1	Request UR	12014647035	10.35.31.85	-----		↑ ↓ + ✕
2	Request UR	12063120140	10.35.31.85	-----		↑ ↓ + ✕

Figure 28: Inbound Trunk 2 Configuration

Inbound Trunk Configuration for DID 5.10.112.xxx.

Manage SIP trunk

Enabled ☒

Name nexmo_inbound3

Remote trunk endpoint address 5.10.112.

Remote trunk endpoint port 5060

Options keepalives Always

Rewrite host in PAI ☒

Idle timeout (s) 3600

Local streaming ☐

Log verbosity Use master setting

Authentication password

Set-side RTP security Allow

Accept traffic from any port ☐

Options interval 60

Remote RTP framesize (ms) Auto

RTP address override ---

PRACK support Use master setting

Authentication username

Confirm authentication password

Icp-side RTP security Disable

Re-invite conversion ☐

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Note, if you modify your routing rules, you must save them before changing pages or navigating elsewhere, or those changes will be lost.

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	Match	Rule	Primary	Secondary	Description	
1	Request UR	12014647035	10.35.31.85	-----		↑ ↓ + ✕
2	Request UR	12063120140	10.35.31.85	-----		↑ ↓ + ✕

Figure 29: Inbound Trunk 3 Configuration

Inbound Trunk Configuration for DID 119.81.44.xxx.

Manage SIP trunk

Enabled
☒

Name
nexmo_inbound4

Remote trunk endpoint address
119.81.44

Remote trunk endpoint port
5060

Options keepalives
Always

Rewrite host in PAI
☒

Idle timeout (s)
3600

Local streaming
☐

Log verbosity
Use master setting

Authentication password

Set-side RTP security
Allow

Accept traffic from any port
☐

Options interval
60

Remote RTP framesize (ms)
Auto

RTP address override
...

PRACK support
Use master setting

Authentication username

Confirm authentication password

Icp-side RTP security
Disable

Re-invite conversion
☐

Search routing rules

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Note, if you modify your routing rules, you must save them before changing pages or navigating elsewhere, or those changes will be lost.

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	Match	Rule	Primary	Secondary	Description	
1	Request UR	12014647035	10.35.31.85	-----		↑ ↓ + ✕
2	Request UR	12063120140	10.35.31.85	-----		↑ ↓ + ✕

Figure 30: Inbound Trunk 4 Configuration

Inbound Trunk Configuration for DID 119.81.44.xxx.

Manage SIP trunk

Enabled
☒

Name
nexmo_inbound5

Remote trunk endpoint address
119.81.44.

Remote trunk endpoint port
5060

Options keepalives
Always

Rewrite host in PAI
☒

Idle timeout (s)
3600

Local streaming
☐

Log verbosity
Use master setting

Authentication password

Set-side RTP security
Allow

Search routing rules

Accept traffic from any port
☐

Options interval
60

Remote RTP framesize (ms)
Auto

RTP address override
...

PRACK support
Use master setting

Authentication username

Confirm authentication password

Icp-side RTP security
Disable

Re-invite conversion
☐

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Note, if you modify your routing rules, you must save them before changing pages or navigating elsewhere, or those changes will be lost.

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Match	Rule	Primary	Secondary	Description		
1	Request UR <div> Request UR </div>	12014647035	10.35.31.85 <div> 10.35.31.85 </div>	----- <div> ----- </div>		<div> ↑ ↓ + </div> <div> </div>
2	Request UR <div> Request UR </div>	12063120140	10.35.31.85 <div> 10.35.31.85 </div>	----- <div> ----- </div>		<div> ↑ ↓ + </div> <div> </div>

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

The screenshot displays the 'System Configuration Settings' interface, divided into two main sections: 'Service parameters' and 'MiNet options'.

Service parameters

- DSCP setting for signaling:** Expedited forwarding (dropdown)
- TFTP blocksize:** 4096 bytes (dropdown)
- ICP failure detection:** ☒
- DSCP setting for voice:** Expedited forwarding (dropdown)
- DSCP setting for video:** Expedited forwarding (dropdown)
- SSL ciphers:** Default (dropdown)

MiNet options

MiNet support

- TCP: ☐
- TCP/PSK: ☒ (locked)
- TCP/TLS: ☒ (locked)

HTML application support

- TCP/TLS: ☒ (locked)

SAC support

- TCP: ☒

Security profile: Legacy mode (dropdown)

Restrict MiNet devices: ☒

Time format: 12 hour (dropdown)

Tone injection: Enabled ☐

Local streaming: ☐

Codec support: Unrestricted (dropdown)

Force set-side codec: Disabled (dropdown)

RTP framesize: Dynamic (dropdown)

Ping before redirect enabled: ☐

Reboot fallback enabled: ☐

Retry backoff interval(s): 60 (spinner)

Pings to send: 1 (spinner)

Successful pings: 1 (spinner)

Ping packet size: 64 (spinner)

Ping Timeout: 800 (spinner)

A yellow warning box states: "This is not a secure transport".

Figure 32: System Configuration

SIP options

SIP support

Protocols

UDP ☒

TCP ☐

TCP/TLS ☐

Access profile

Public

Public

Public

Registration Mode

Max Set-Side

Set-side registration expiry time

240

ICP-side registration expiry time

Allowed URI names

Add another

Blank any field you no longer want.

Tone injection

Enabled ☐

Local streaming ☐

Codec support

Unrestricted

RTP framesize

Dynamic

Set-side RTP security

Allow

Icp-side RTP security

Disable

KPML username

administrator

KPML password

Change KPML password

Confirm KPML password

Permit weak SIP passwords ☐

PRACK support ☒

Send options keepalives

Only behind NAT

Options interval

20

Challenge methods

Invite

Subscribe

Refer

Prack

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.

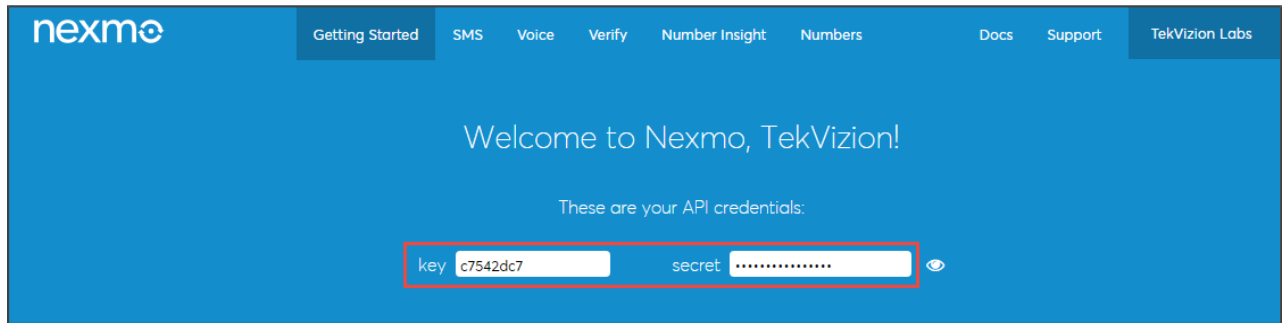


Figure 34: Nexmo Dashboard

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

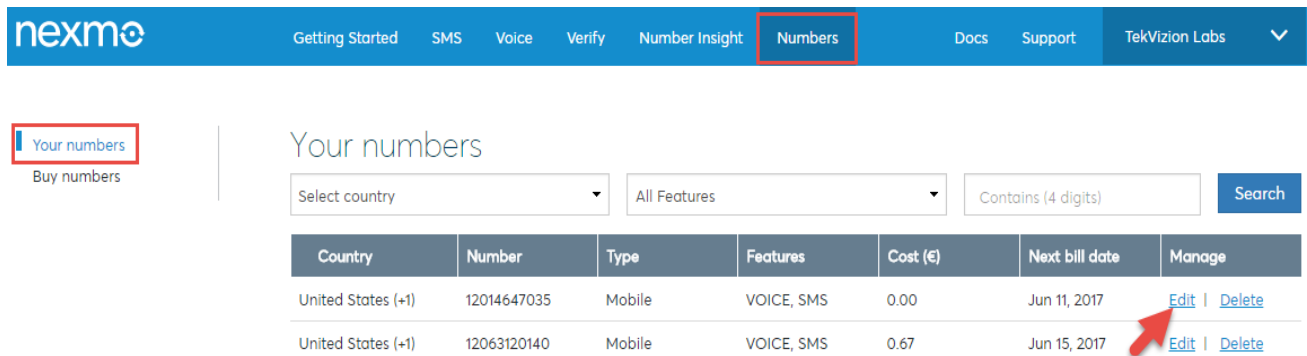


Figure 35: Nexmo DID Numbers

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

Settings for 12014647035 ✕

SMS

Webhook URL

Voice

Forward to	URL
Forward to SIP ▼	12014647035@nexmo.tekvizionlabs.␣

Status webhook URL

Cancel Update




Figure 36: Your Numbers – Cont.