



Nexmo SIP Trunking Configuration Guide

NEC SV9100 version 6.00.50

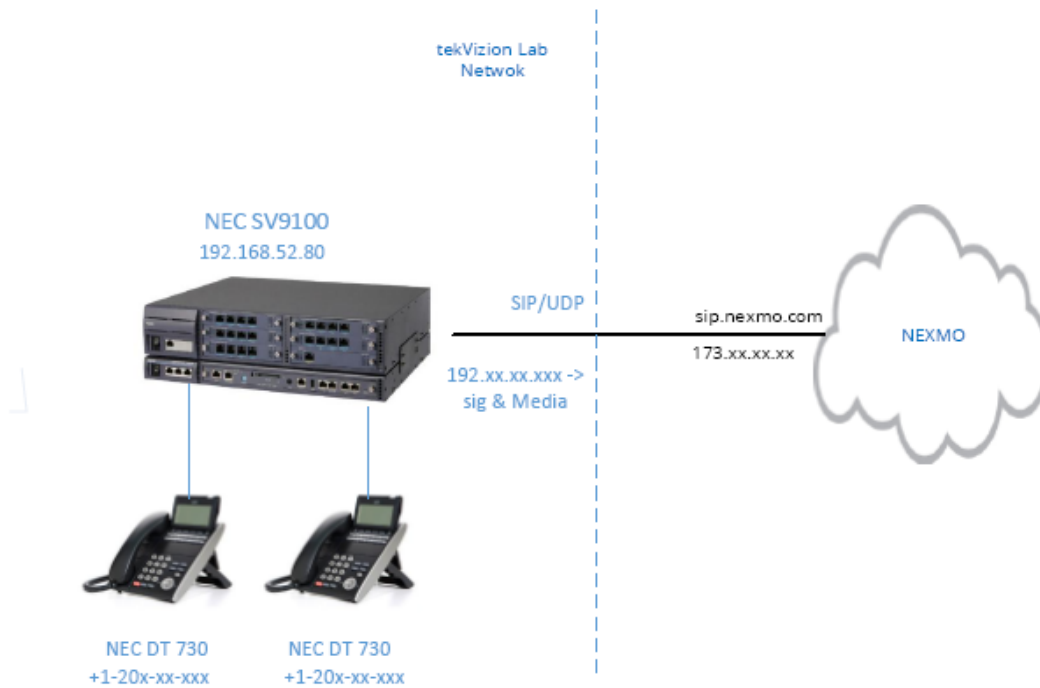
August 2017

1 Audience

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 NEC SV9100 version 6.00.50 with Nexmo SIP Trunking services.

2 SIP Trunking Network Components

The network for the SIP trunk reference configuration is illustrated below and is representative of a NEC SV9100 configuration to Nexmo SIP trunking.



2.1 Network Components

Component	Version	Comments
NEC SV9100	6.00.50	
Cisco IP Phone	Model: CP-7965	This Cisco IP Phone is
	App Load ID: jar45sccp.9-4-	

	2TH1-1.sbn	the PSTN test device
	Boot Load ID: tnp65.9-3-1-CR17.bin	

3 Features

3.1.1 Features Supported

- Incoming and outgoing off-net calls using G711ULAW voice codecs
- Calling Line (number) Identification Presentation
- Calling Line (number) Identification Restriction
- Call hold and resume
- Call transfer (semi-attended and consultative)
- 3 way Conference
- Call forward (All, No answer)
- DTMF relay both directions (RFC2833)
- Media flow-through on NEC SV9100

3.1.2 Features Not Supported by PBX

- None

3.1.3 Features Not Tested

- None

3.1.4 Caveats and Limitations

- When Public DNS is used for resolving sip.nexmo.com, NEC SV9100 receives multiple target address. NEC sends registration to the first target and when it is challenged, it sends with authorization details to the second target. Consequently registration fails. Hence for this testing, a local DNS is used to resolve sip.nexmo.com to one of the intended target IPs and trunk has been registered.
- In the inbound call from Nexmo, the TO header in the INVITE contains sip.nexmo.com instead of the trunk FQDN which is nexmo.tekvizionlabs.com.
- In the outbound call from NEC, the From header in the INVITE contains trunk FQDN (sip.nexmo.com) instead of the PBX IP. It appears to be a design intent of NEC SV9100.
- NEC SV9100 does not appear to support Diversion header. Consequently diversion information is not present in the call forward INVITE from NEC SV9100.
- NEC SV9100 adds +1 to the originating number (From header) in the call forward INVITE if NEC SV9100 is enabled for E164 dialing.
- In a 3 way conference, when PSTN drops out of the conference, the trunks are not released until one of the PBX endpoints disconnect
- No Session Audit message is sent from Nexmo

4 Configuration

4.1 IP Address Worksheet

The specific values listed in the table below and in subsequent sections are 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.

Component	Lab Value	Customer Value
NEC SV9100		
LAN IP Address	192.168.52.80	
LAN Subnet Mask	255.255.255.0	
WAN IP Address (After NATting)	192.xx.xx.xxx	
WAN Subnet Mask	255.255.255.128	

4.2 Configuring NEC SV9100

This section describes NEC SV9100 configuration. A direct SIP trunk is established between NEC SV9100 and Nexmo. There is no PBX level NATing done.

4.2.1 SIP Server Information Setup

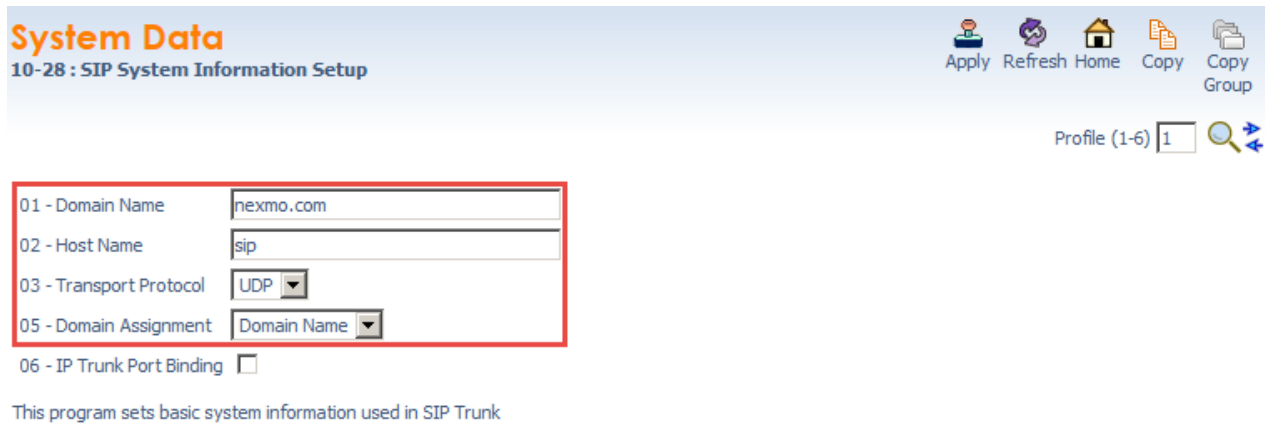
1. Navigate to **10-XX: System Configuration**
2. Click **10-29: SIP Server Information Setup**
3. Enter **Registrar Domain Name:** sip.nexmo.com
4. Enter **Proxy Domain Name:** nexmo.com
5. Enter **Proxy Host Name:** sip
6. Select **SIP Carrier Choice:** Carrier B

System Data
10-29 : SIP Server Information Setup

01 - Outbound Default Proxy	<input checked="" type="checkbox"/>
02 - Inbound Default Proxy	<input type="checkbox"/>
03 - Default Proxy IP Address	0.0.0.0
04 - Default Proxy Port	5060
05 - Register Mode	Manual
06 - Registrar IP Address	0.0.0.0
07 - Registrar Port	5060
11 - Registrar Domain Name	sip.nexmo.com
12 - Proxy Domain Name	nexmo.com
13 - Proxy Host Name	sip
14 - SIP Carrier Choice	Carrier B
15 - Registration Expiry Time	3600
16 - Register Sub Mode	<input type="checkbox"/>
19 - Keep Alive by OPTION message	<input checked="" type="checkbox"/>
20 - Authentication Trial	1
21 - NAT Router	Not used
22 - Default Gateway	0.0.0.0
23 - MAC Address	00-00-00-00-00-00
24 - NAPT Router IP Address	0.0.0.0

4.2.2 SIP System Information Setup

1. Navigate to **10-XX: System Configuration**
2. Click **10-28: SIP System Information Setup**
3. Enter **Domain Name:** nexmo.com
4. Enter **Host Name:** sip
5. Select **Transport Protocol:** UDP



System Data
10-28 : SIP System Information Setup

Apply Refresh Home Copy Copy Group

Profile (1-6) 1

01 - Domain Name	<input type="text" value="nexmo.com"/>
02 - Host Name	<input type="text" value="sip"/>
03 - Transport Protocol	<input type="text" value="UDP"/>
05 - Domain Assignment	<input type="text" value="Domain Name"/>
06 - IP Trunk Port Binding	<input type="checkbox"/>

This program sets basic system information used in SIP Trunk

4.2.3 SIP Trunk Registration Information

1. Navigate to **10-XX: System Configuration**
2. Click **10-36: SIP Trunk Registration Information**
3. Check **Registration**
4. Enter **User ID:** 911236e3 (Provided by Nexmo for this particular testing)
5. Enter **Authentication User ID:** 911236e3 (Provided by Nexmo for this particular testing)
6. Enter **Authentication Password**

System Data

10-36 : SIP Trunk Registration Information



Profile (1-6)

Registration ID (0-31)

Registration ID	Registration	User ID	Authentication User ID	Authentication Password
00	<input checked="" type="checkbox"/>	911236e3	911236e3
01	<input type="checkbox"/>			
02	<input type="checkbox"/>			
03	<input type="checkbox"/>			
04	<input type="checkbox"/>			
05	<input type="checkbox"/>			
06	<input type="checkbox"/>			
07	<input type="checkbox"/>			



4.2.4 Class of Service Options (Outgoing Call Service)

1. Navigate to **20-XX**: System Options
2. Click **20-08**: Class of Service Options (Outgoing Call Service)

The Class of Service Options are configured as below

System Data
20-08 : Class of Service Options (Outgoing Call Service)

Apply Refresh Home Copy Copy Group

Class of Service (1-15)  

01 - Internal Call	<input checked="" type="checkbox"/>
02 - Outgoing Trunks	<input checked="" type="checkbox"/>
03 - Speed Dials Common	<input checked="" type="checkbox"/>
04 - Speed Dials Group	<input checked="" type="checkbox"/>
05 - Preview Dial Number	<input checked="" type="checkbox"/>
06 - Toll Restriction Override	<input type="checkbox"/>
07 - Redial Repeat	<input checked="" type="checkbox"/>
08 - Toll Restriction Dial Blocking	<input type="checkbox"/>
09 - Hotline for Handpiece	<input type="checkbox"/>
10 - Handsfree Answerback/Forced Intercom Ringing Switching	<input checked="" type="checkbox"/>
11 - Call Mode Switching Protection from Caller (Internal Call)	<input type="checkbox"/>
12 - Department Group Step Calling	<input checked="" type="checkbox"/>
13 - ISDN Clip	<input checked="" type="checkbox"/>
14 - Set Calling Sub Address	<input type="checkbox"/>
15 - Block Outgoing Caller ID	<input type="checkbox"/>
16 - E911 Dialed Extension Name and Number Display	<input type="checkbox"/>
17 - ARS Override of Trunk Access Map	<input type="checkbox"/>
19 - Hotline for Speaker	<input type="checkbox"/>
20 - Hot Key Pad	<input type="checkbox"/>
21 - Automatic Trunk Seizing by Pressing SPK Key	<input type="checkbox"/>
22 - Voice Over to Busy Virtual Extension	<input type="checkbox"/>
23 - Display indication for security sensor detection	<input type="checkbox"/>
24 - Display indication for emergency call by remote inspection	<input type="checkbox"/>

4.2.5 IP Trunk Party Calling Party Number Setup for Extensions

1. Navigate to **20-XX: System Options**
2. Click **21-19: IP Trunk (SIP) Calling Party Number Setup for Extension**
3. Enter the **Calling Party Number** (DID) against the respective **ICM Extension** (For e.g. in this test setup ICM Extensions 109 and 111 are used. The respective DIDs are entered against them)

System Data

21-19 : IP Trunk (SIP) Calling Party Number Setup for Extension



ICM Extension

Profile

ICM Extension	Calling Party Number	ICM Extension	Calling Party Number
101	<input type="text"/>	109	<input type="text" value="200040000"/>
102	<input type="text"/>	110	<input type="text"/>
103	<input type="text"/>	111	<input type="text" value="200012010"/>
104	<input type="text"/>	112	<input type="text"/>
105	<input type="text"/>	113	<input type="text"/>
106	<input type="text"/>	114	<input type="text"/>
107	<input type="text"/>	201	<input type="text"/>
108	<input type="text"/>	202	<input type="text"/>

Valid characters are 0-9, *, #

Use Program 21-19: IP (SIP) Trunk Calling Party Number Setup for Extensions to allow for the Calling Party Number to be displayed for IP extensions when the VoIP feature is used.

4.2.6 DID Translation Table

1. Navigate to **22-XX: Incoming**
2. Click **22-11: DID Translation Table**
3. Select a **DID Translation Entry** (e.g. 1 and 2)
4. Enter **Received Number** as the last 4 digits of the DID
5. Enter **Target 1: ICM Extension** (e.g. 109 and 111)

System Data

22-11 : DID Translation Table





DID Translation Table Entry (1-4000)

01 - Received Number	<input type="text" value="2000"/>
02 - Target 1	<input type="text" value="109"/>
03 - DID Name	<input type="text" value="user 1"/>
04 - Transfer Operation Mode	<input type="text" value="No Transfer"/>
05 - Transfer Target 1	<input type="text" value="0"/>
06 - Transfer Target 2	<input type="text" value="0"/>
07 - Call Waiting	<input type="checkbox"/>
08 - Maximum Number of Calls	<input type="text" value="0"/>
09 - Music On Hold Source	<input type="text" value="System Music On Hold Source"/>
10 - Music On Hold Source ACI Port	<input type="text" value="0"/>
11 - Fall over IRG	<input checked="" type="checkbox"/>
13 - Identify for Mobile Extension	<input type="checkbox"/>

System Data

22-11 : DID Translation Table

Apply Refresh Home Copy Copy Group

DID Translation Table Entry (1-4000)  

01 - Received Number	<input type="text" value="00000"/>
02 - Target 1	<input type="text" value="111"/>
03 - DID Name	<input type="text" value="user2"/>
04 - Transfer Operation Mode	<input type="text" value="No Transfer"/>
05 - Transfer Target 1	<input type="text" value="0"/>
06 - Transfer Target 2	<input type="text" value="0"/>
07 - Call Waiting	<input type="checkbox"/>
08 - Maximum Number of Calls	<input type="text" value="0"/>
09 - Music On Hold Source	<input type="text" value="System Music On Hold Source"/>
10 - Music On Hold Source ACI Port	<input type="text" value="0"/>
11 - Fall over IRG	<input checked="" type="checkbox"/>
13 - Identify for Mobile Extension	<input type="checkbox"/>

4.2.7 SIP Trunk Basic Setup

1. Navigate to **84-XX: VoIP Hardware Setup**
2. Click **84-14: SIP Trunk Basic Information Setup**
3. Select **Incoming/Outgoing SIP Trunk for E.164: Mode 1**

System Data

84-14 : SIP Trunk Basic Information Setup



Profile (1-6)

06 - SIP Trunk Port	<input type="text" value="5060"/>
07 - Session Timer Value	<input type="text" value="1800"/>
08 - Minimum Session Timer Value	<input type="text" value="1800"/>
09 - Called Party Info	<input type="text" value="Request URI"/>
10 - URL Type	<input type="text" value="SIP-URL"/>
11 - URL/To HeaderSetting Information	<input type="text" value="Proxy Server Domain"/>
13 - Incoming/Outgoing SIP Trunk for E.164	<input type="text" value="Mode 1"/>
15 - 100rel Settings	<input type="text" value="Use Default Setting"/>
16 - SIP Trunk SIP-URI E.164 Incoming Mode	<input type="text" value="Mode 1"/>
17 - Call Forward Moved Temporarily Support	<input type="text" value="Disabled"/>
18 - Keep Alive by OPTION Interval Timer	<input type="text" value="180"/>
19 - Keep Alive by OPTION Fail Limit	<input type="text" value="1"/>
20 - Option Keep Alive User ID	<input type="text" value="ping"/>
21 - SIP Trunk TLS Port Number	<input type="text" value="5061"/>
22 - TLS Certificate	<input type="text"/>

Use Program 84-14: SIP Trunk Basic Information Setup to define the basic setup for SIP trunks.

4.2.8 IP Trunk Basic Setup

1. Navigate to **14-XX: Trunk Setup**
2. Click **14-01: Trunk Basic Setup**
3. Check **Trunk to Trunk Outgoing CallerID Through Mode**

System Data

14-01 : Trunk Basic Setup

Trunk 001: SIP

- 01 - Trunk Name
- 02 - Transmit Gain Level
- 03 - Receive Gain Level
- 04 - Conference and Transfer Calls Transmit Gain Level
- 05 - Conference and Transfer Calls Receive Gain Level
- 06 - SMDR Print-out
- 07 - Outgoing Calls
- 08 - Toll Restriction
- 09 - Private Line
- 10 - Outgoing Call DTMF Tone
- 11 - Account Code Requirement
- 13 - Trunk to Trunk Transfer
- 14 - Long Conversation Cut-off
- 15 - Long Conversation Alarm before Cut-off
- 16 - Long-time Holding Forced Release
- 17 - Trunk to Trunk Long Conversation Warning Tone
- 18 - Warning Beep Tone Signaling
- 19 - Privacy Mode Toggle Option
- 20 - Block Outgoing Caller ID
- 21 - Caller ID Block Code
- 22 - Caller ID to Voice Mail
- 23 - Least Cost Routing
- 24 - Trunk to Trunk Outgoing Caller ID Through Mode

- 25 - Continued/Discontinued Trunk to Trunk Conversation
- 26 - Automatic Trunk to Trunk Transfer Mode
- 27 - Caller ID Refuse
- 28 - Conversation Recording Destination for Extension
- 30 - Flexible Ringing by Caller ID
- 32 - Anti-trombone Function
- 33 - VM00 Trunk Receive Gain
- 35 - Large LED Illumination Setup(Trunk Incoming)
- 36 - Calling Party Name notification
- 38 - Outgoing CLI selection
- 39 - CLI composition
- 40 - ISDN Queue announcement connect mode
- 41 - Incoming Caller Name Usage
- 46 - Collect Call Blocking
- 47 - DTMF Receiver Type

4.2.9 Location Setup

1. Navigate to **10-XX: System Configuration**
2. Click **10-02: Location Setup**
3. Enter **Country Code: 1**
4. Enter **Caller ID Edit Code: 9**

System Data

10-02 : Location Setup

Apply Refresh Home Copy Copy Group

01 - Country Code	<input type="text" value="1"/>
02 - International Access Code	<input type="text"/>
03 - Caller ID Edit Code	<input type="text" value="9"/>
04 - Area Code	<input type="text"/>
05 - Trunk Access Code	<input type="text"/>

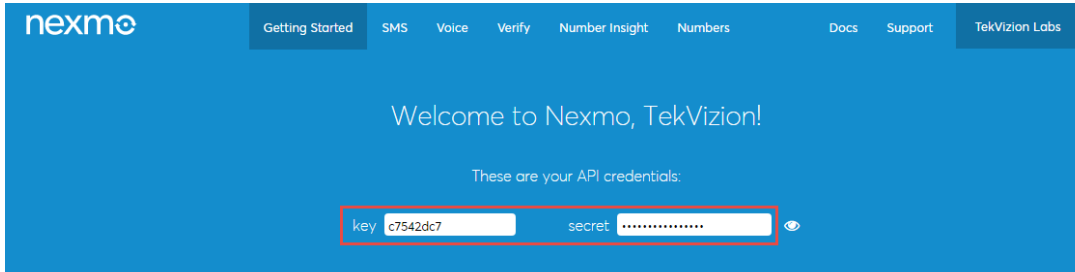
Valid characters are 0-9, *, #

Use Program 10-02: Location Setup to define the location of the installed system.

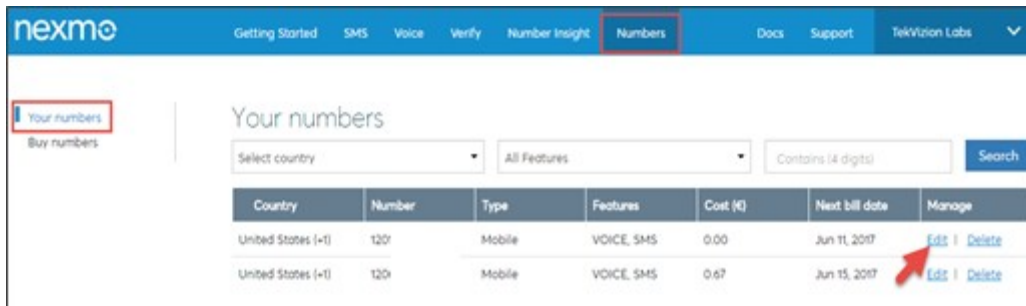
4.3 Nexmo Configuration

4.3.1 Configure Numbers in Nexmo Account

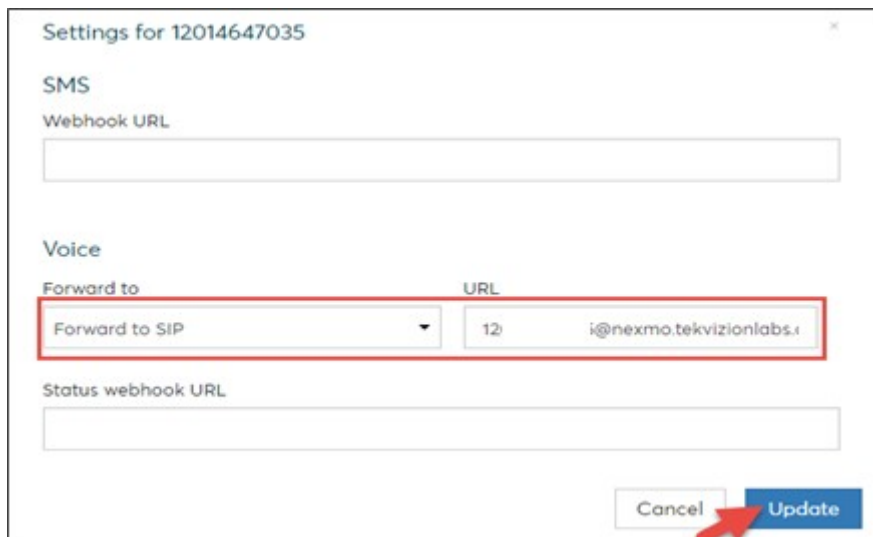
1. Login to the Nexmo account using the credentials provided at the time of registration. A **Key** and **Secret** will be displayed on the dashboard and this can be used as the username and password for Registration SIP Trunks.



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



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



5 Summary of Tests and Results

N/S = Not Supported N/T= Not Tested N/A= Not Applicable

Test Case	Test Case Description	Result	Notes
1	Calling Party Disconnects Before Answer	PASS	When the call comes from Nexmo, the TO header in the INVITE contains sip.nexmo.com instead of the trunk FQDN which is nexmo.tekvizionlabs.com.
2	Calling Party Disconnects After Answer	PASS	
3	Called Party Disconnects After Answer	PASS	
4	Three Way Calling	PASS	In a 3 way conference, when PSTN drops out of the conference, the trunks are not released until one of the PBX endpoints disconnect.
5	Calling Party Presentation Restricted	PASS	
6	Calling Party Disconnect Before Answer	PASS	When NEC initiates a call, the FROM header in the INVITE contains trunk FQDN (sip.nexmo.com) instead of the PBX IP. It appears to be a design intent of NEC SV9100.
7	Calling Party Disconnects after Answer	PASS	
8	Called Party Disconnects after Answer	PASS	
9	Calling Party Receives Busy	PASS	
10	International Outbound Dialing	FAIL	With E164 dialing enabled, NEC adds +1 with international dialing also. Call fails henceforth.
11	Outbound Call Forward Always	PASS	NEC SV9100 does not appear to support Diversion header. Consequently, diversion information is not present in the call forward INVITE from NEC SV9100. NEC SV9100 adds +1 to the originating number (From header) in the call forward INVITE if NEC SV9100 is enabled for E164 dialing
12	Outbound Call Forward Not Available (Ring No Answer)	PASS	Same as 11
13	Outbound Consultative Call Transfer	PASS	
14	Outbound Semi-Attended/Blind Call Transfer	PASS	

15	Outbound Call Hold	PASS	
16	Terminate Early Media Outbound Call Before Answer	PASS	
17	Early Media Forward Call	PASS	
18	Outbound, Wait for Session Audit	PASS	No Session Audit message is sent by Nexmo
19	Inbound, Wait for Session Audit	PASS	
20	Outbound DTMF (RTPevent)	PASS	
21	Inbound DTMF(RTPevent)	PASS	