Nexmo SIP Trunking Configuration Guide

NEC SV9100 version 6.00.50

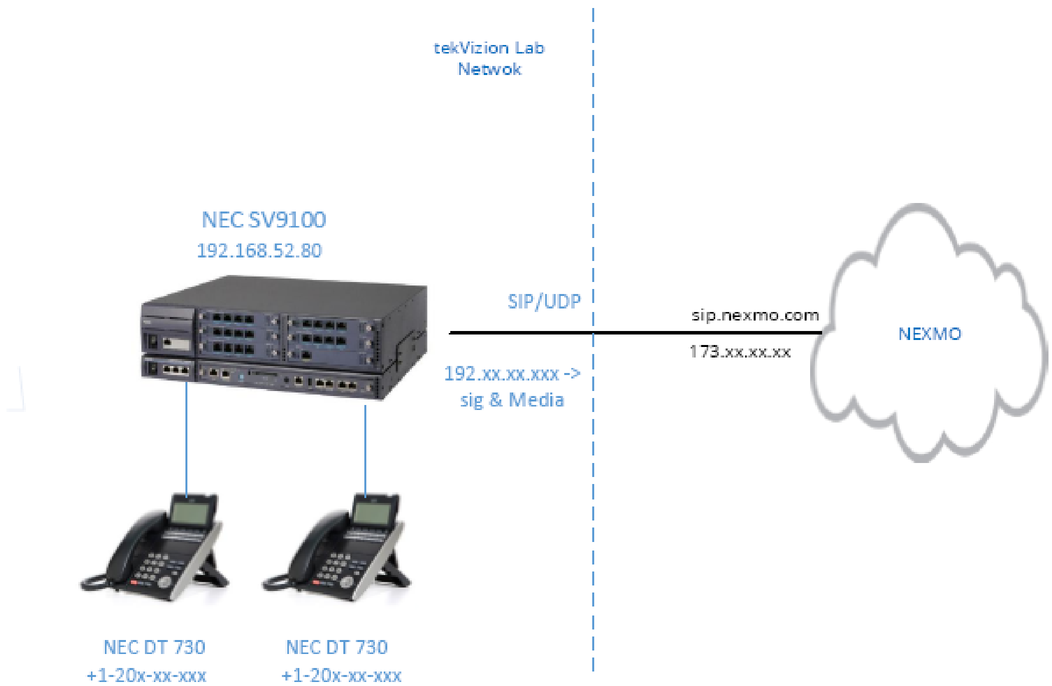
August 2017

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 App Load ID: jar45sccp.9-4-	This Cisco IP Phone is

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

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.

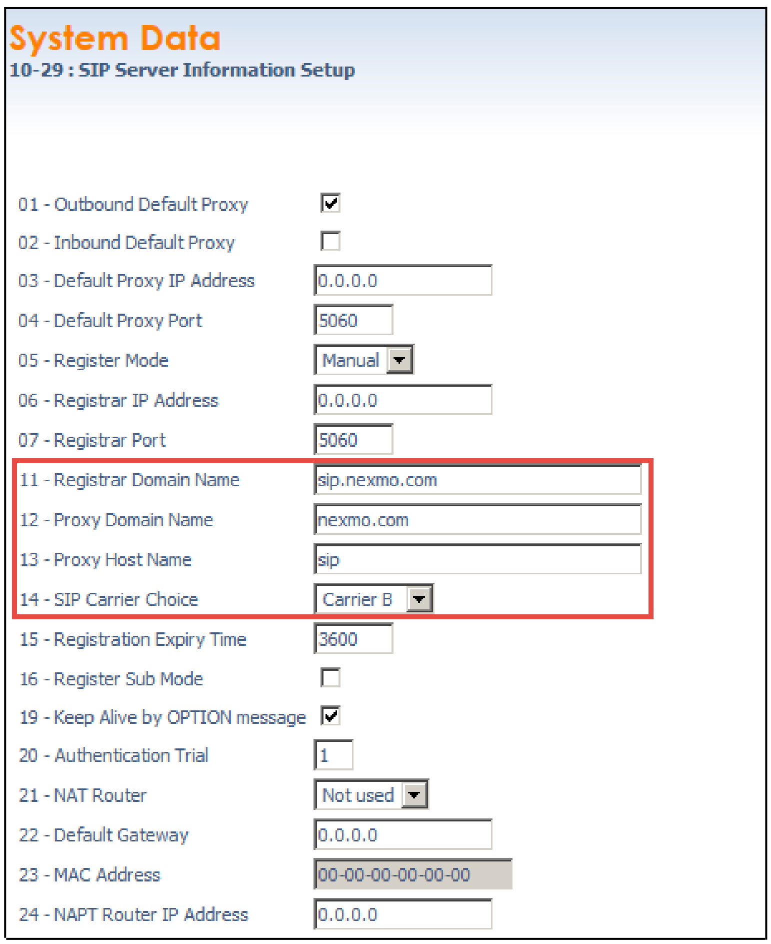
Component	Lab Value	Customer Value
NEC SV9100		
LAN IP Address	192.168.52.80	
LAN Subnet Mask	255.255.255.0	
WAN IP Address (After NATting)	192.xx.xx.xxx	
WAN Subnet Mask	255.255.255.128	

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.

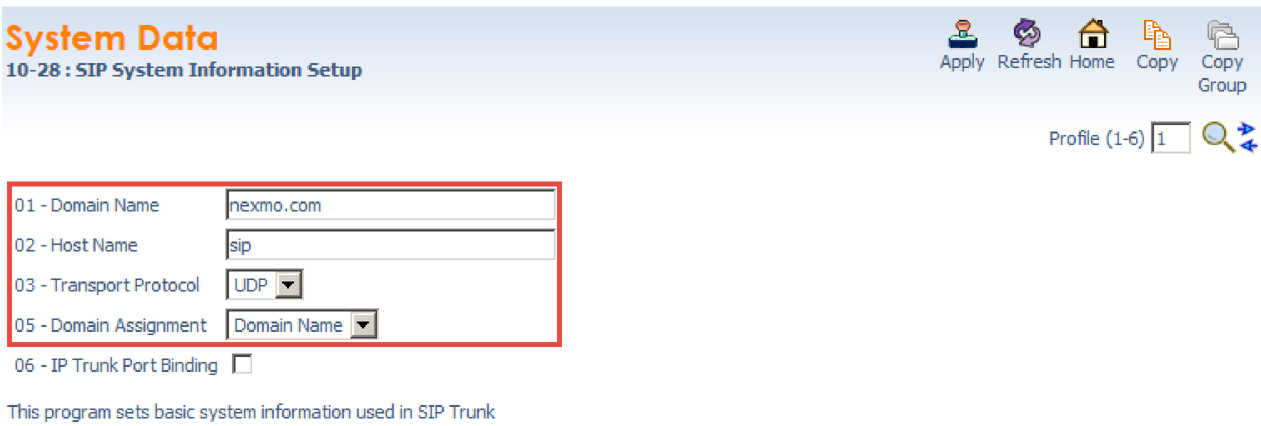
4.2.1 SIP Server Information Setup

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



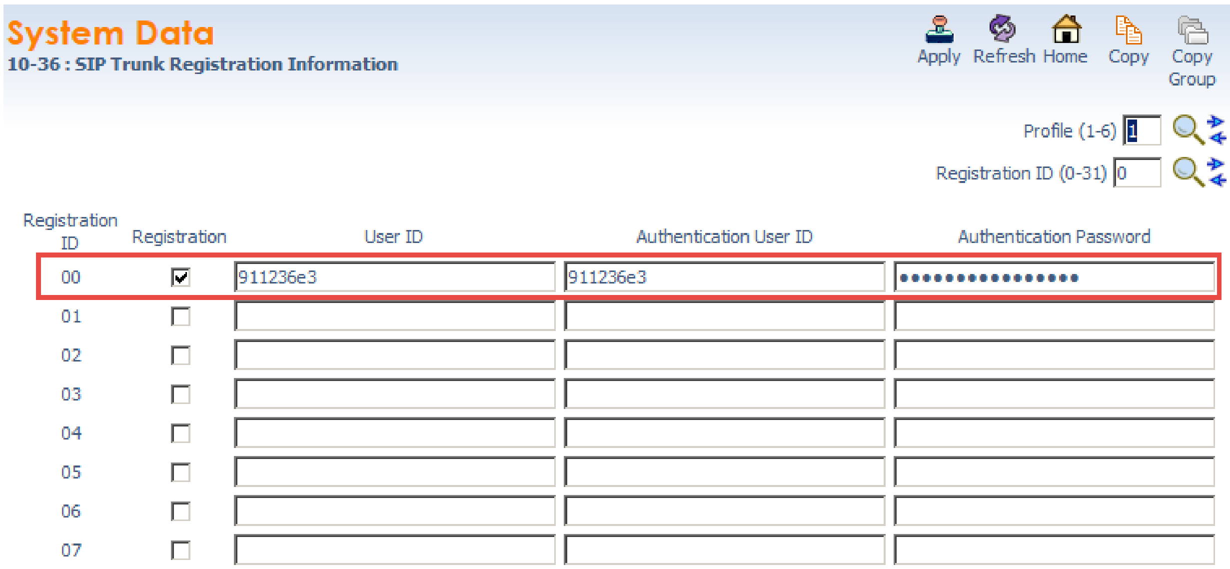
4.2.2 SIP System Information Setup

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



4.2.3 SIP Trunk Registration Information

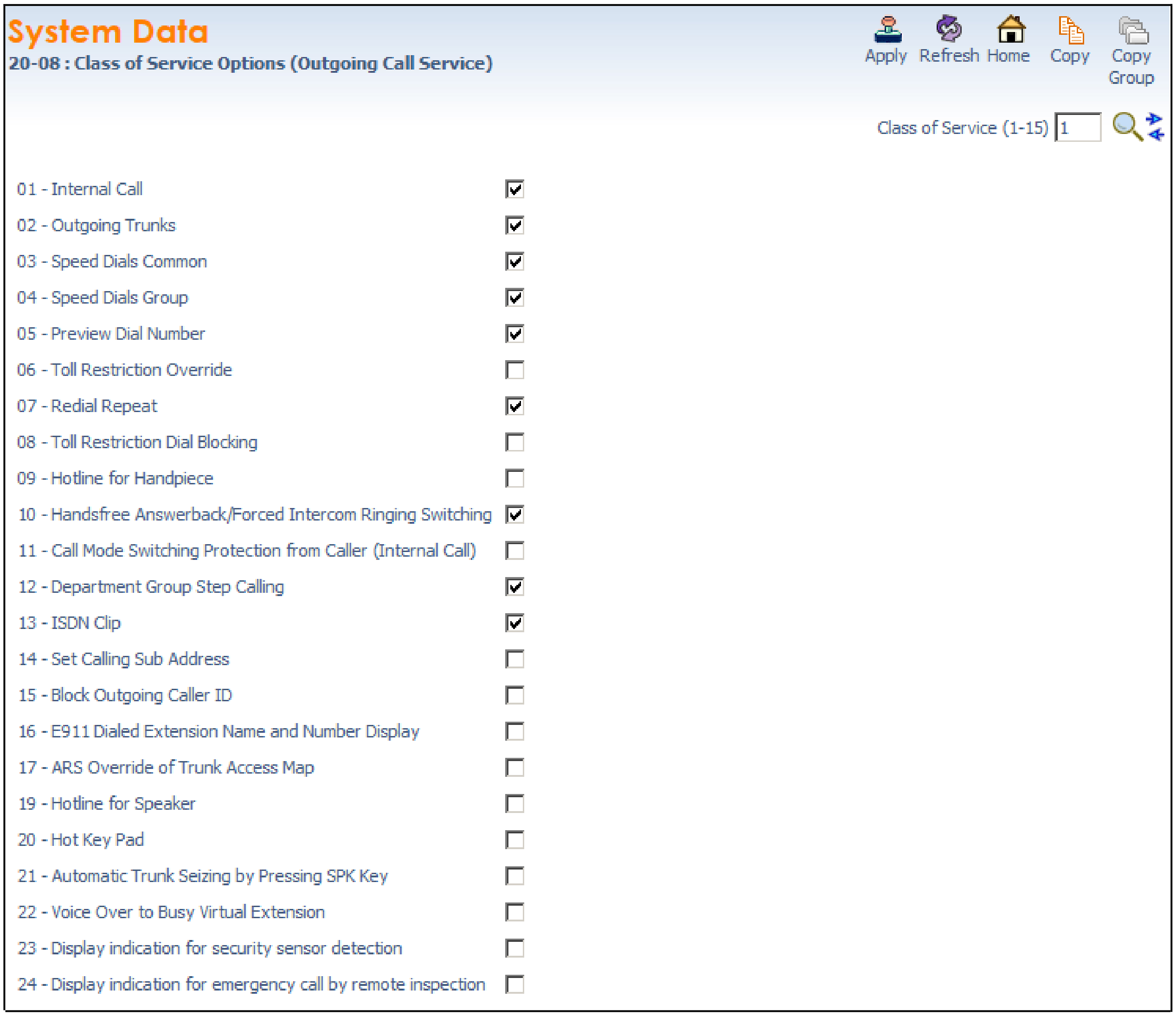
1. Navigate to **10-XX:** System Configuration
2. Click **10-36:** SIP Trunk Registration Information
3. Check **Registration**
4. Enter **User ID:** 911236e3 (Provided by Nexmo for this particular testing)
5. Enter **Authentication User ID:** 911236e3 (Provided by Nexmo for this particular testing)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



4.2.4 Class of Service Options (Outgoing Call Service)

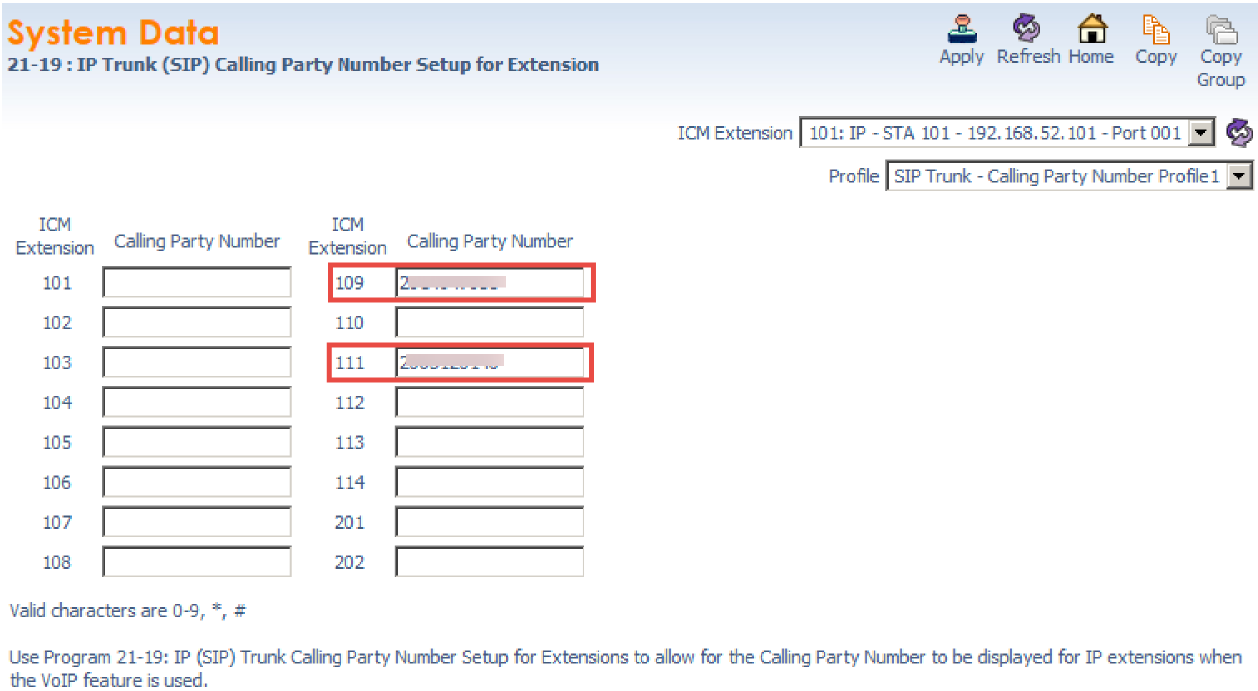
1. Navigate to **20-XX:** System Options
2. Click **20-08:** Class of Service Options (Outgoing Call Service)

The Class of Service Options are configured as below



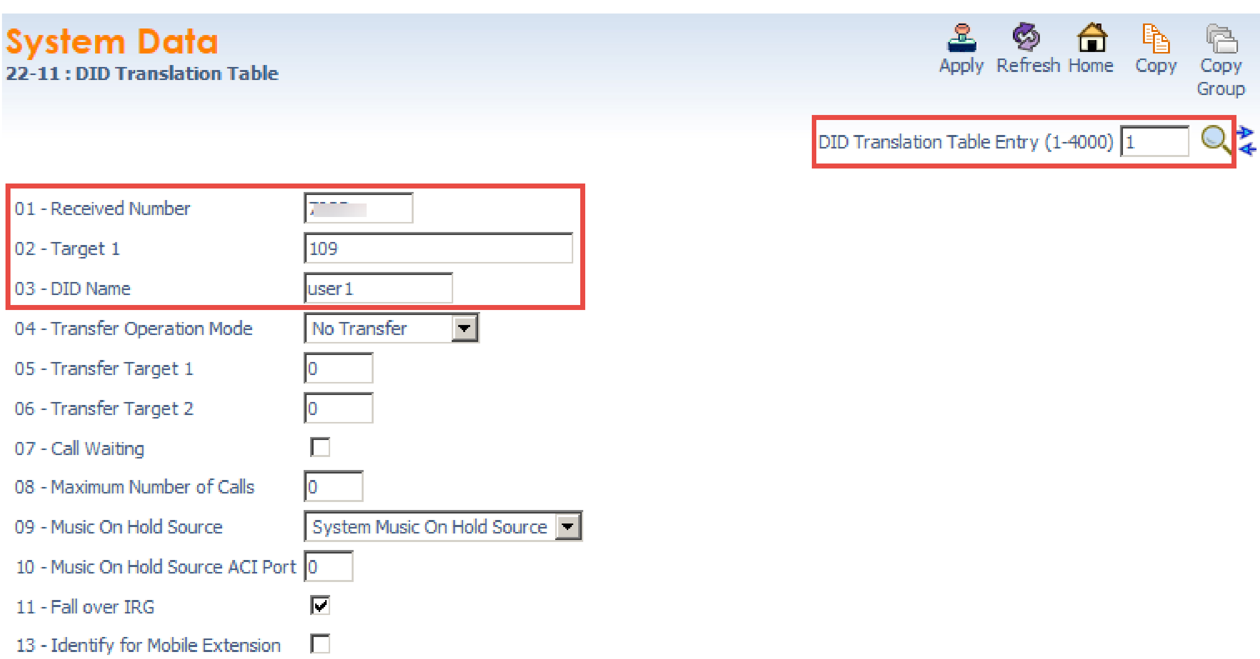
4.2.5 IP Trunk Party Calling Party Number Setup for Extensions

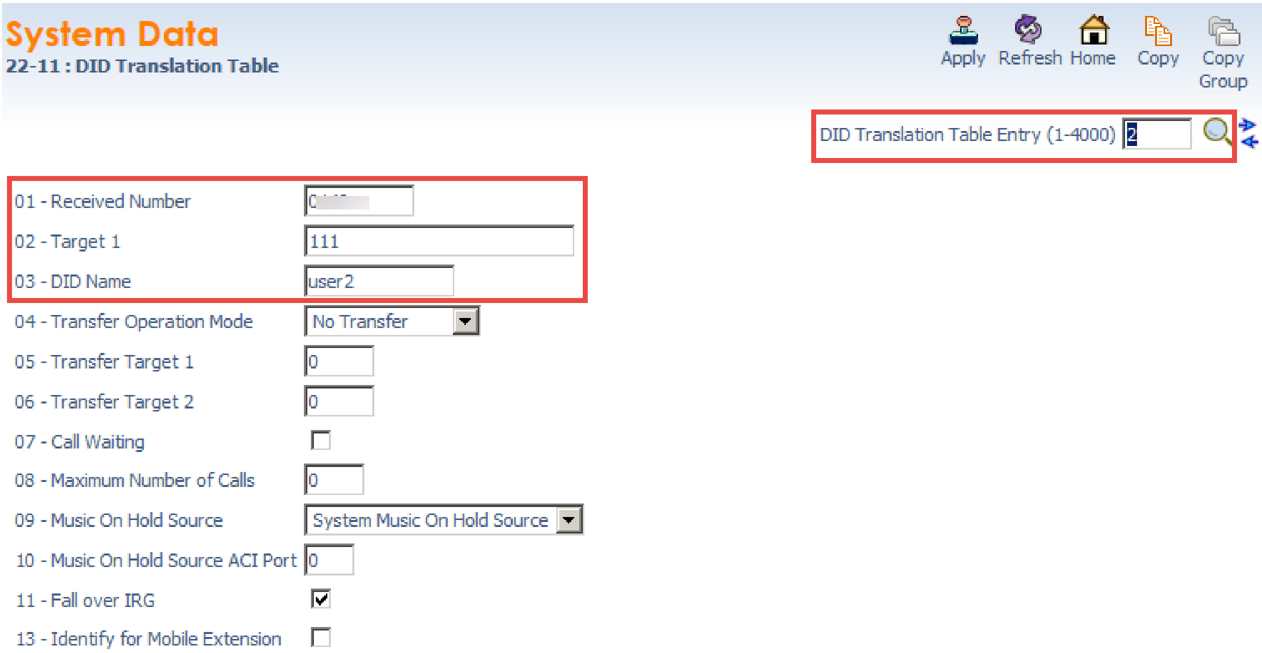
1. Navigate to **20-XX:** System Options
2. Click **21-19:** IP Trunk (SIP) Calling Party Number Setup for Extension
3. Enter the **Calling Party Number** (DID) against the respective **ICM Extension** (For e.g. in this test setup ICM Extensions 109 and 111 are used. The respective DIDs are entered against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



4.2.6 DID Translation Table

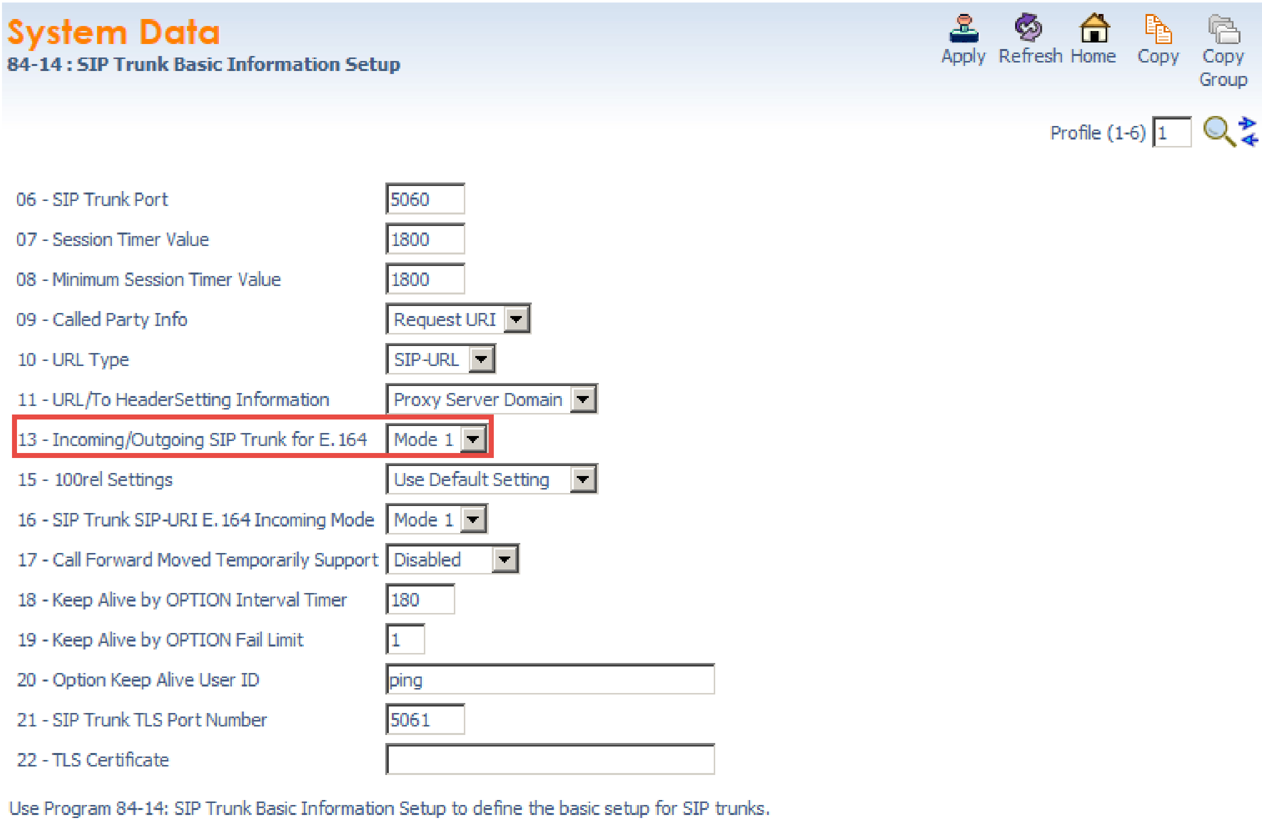
1. Navigate to **22-XX:** Incoming
2. Click **22-11:** DID Translation Table
3. Select a **DID Translation Entry** (e.g. 1 and 2)
4. Enter **Received Number** as the last 4 digits of the DID
5. Enter **Target 1:** ICM Extension (e.g. 109 and 111)





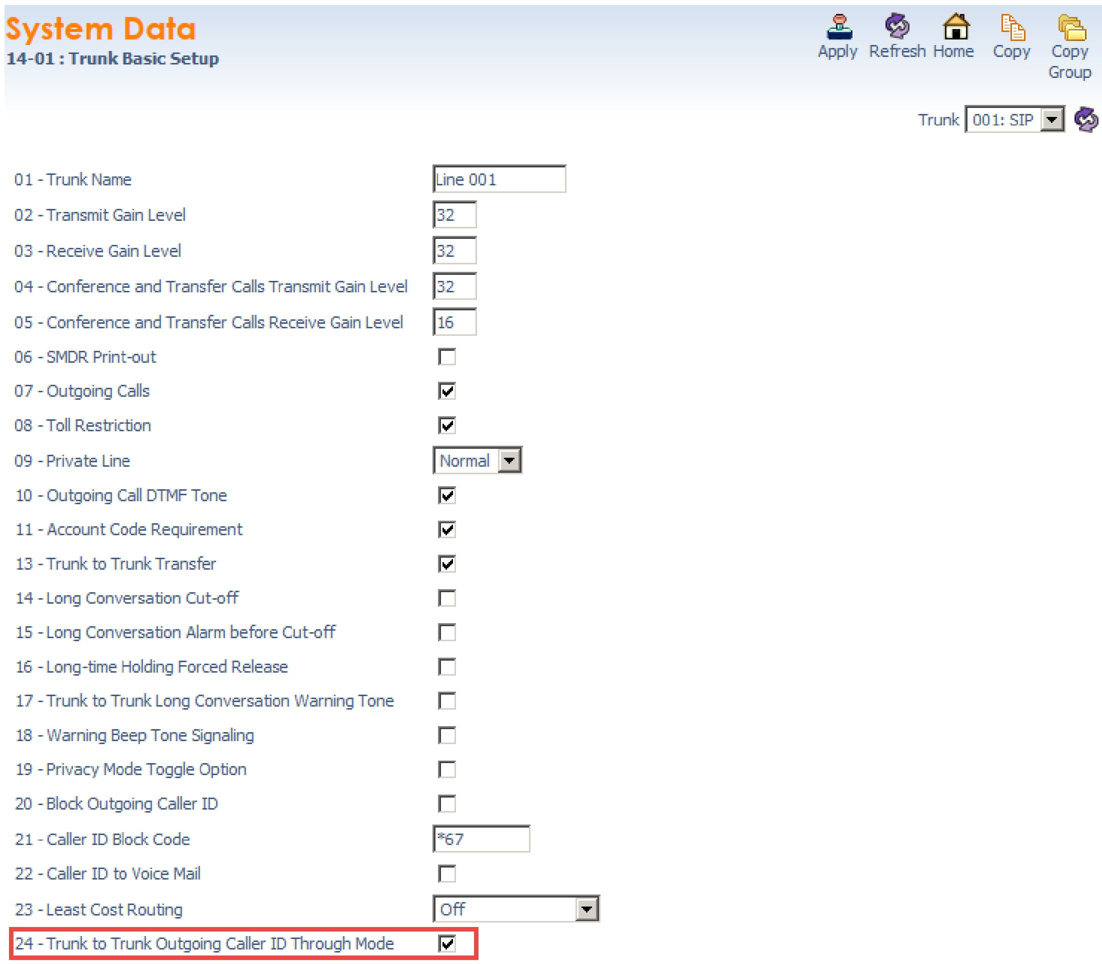
4.2.7 SIP Trunk Basic Setup

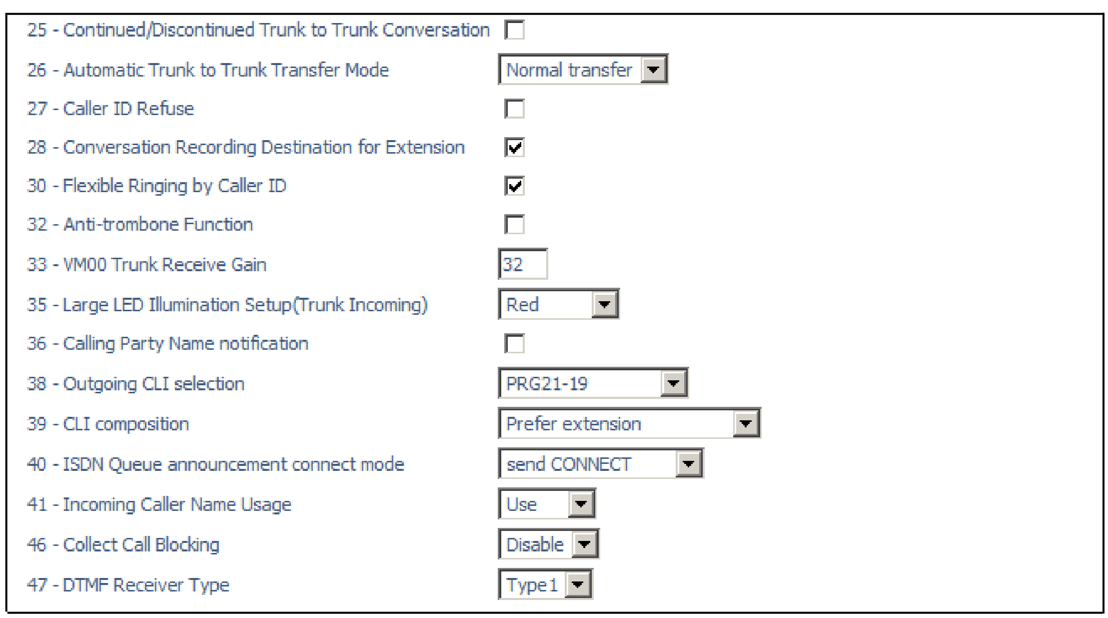
1. Navigate to **84-XX:** VoIP Hardware Setup
2. Click **84-14:** SIP Trunk Basic Information Setup
3. Select **Incoming/Outgoing SIP Trunk for E.164**: Mode 1



4.2.8 IP Trunk Basic Setup

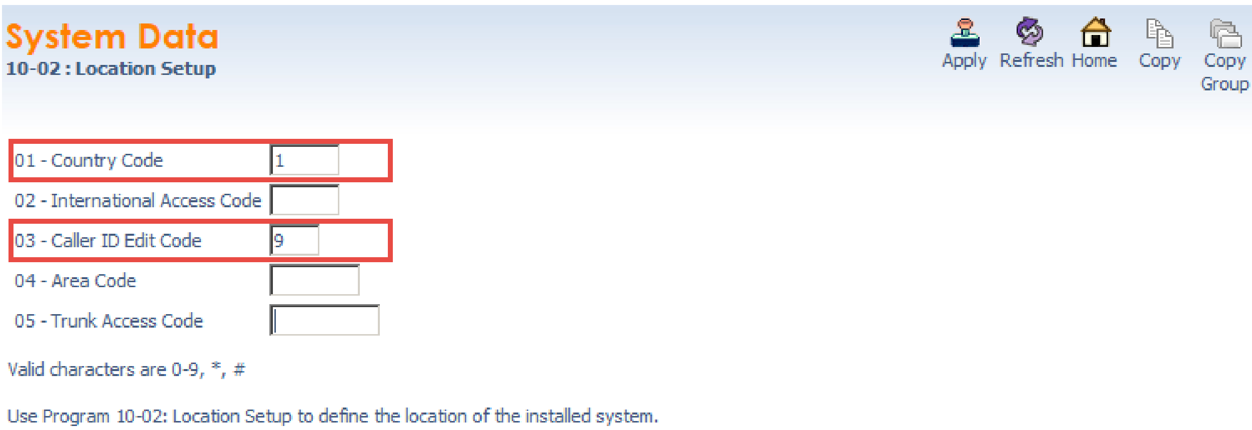
1. Navigate to **14-XX:** Trunk Setup
2. Click **14-01:** Trunk Basic Setup
3. Check **Trunk to Trunk Outgoing CallerID Through Mode**


System Data
14-01 : Trunk Basic Setup
Apply
Refresh
Home
Copy
Copy Group
Trunk 001: SIP
01 - Trunk Name
Line 001
02 - Transmit Gain Level
32
03 - Receive Gain Level
32
04 - Conference and Transfer Calls Transmit Gain Level
32
05 - Conference and Transfer Calls Receive Gain Level
16
06 - SMDR Print-out
07 - Outgoing Calls
08 - Toll Restriction
09 - Private Line
Normal
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
*67
22 - Caller ID to Voice Mail
23 - Least Cost Routing
Off
24 - Trunk to Trunk Outgoing Caller ID Through Mode


25 - Continued/Discontinued Trunk to Trunk Conversation
26 - Automatic Trunk to Trunk Transfer Mode
Normal transfer
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
32
35 - Large LED Illumination Setup(Trunk Incoming)
Red
36 - Calling Party Name notification
38 - Outgoing CLI selection
PRG21-19
39 - CLI composition
Prefer extension
40 - ISDN Queue announcement connect mode
send CONNECT
41 - Incoming Caller Name Usage
Use
46 - Collect Call Blocking
Disable
47 - DTMF Receiver Type
Type1

4.2.9 Location Setup

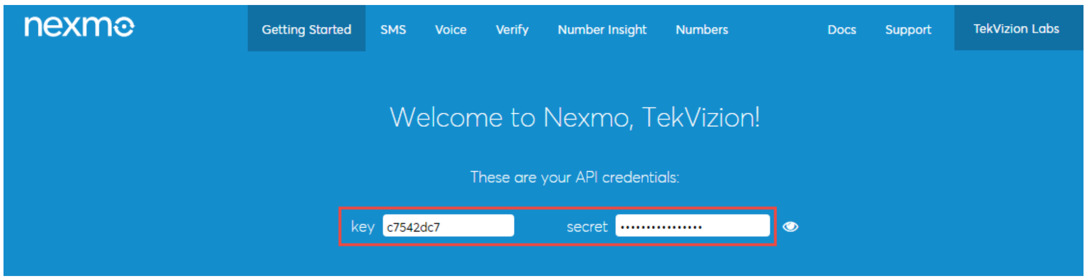
1. Navigate to **10-XX:** System Configuration
2. Click **10-02:** Location Setup
3. Enter **Country Code: 1**
4. Enter **Caller ID Edit Code: 9**



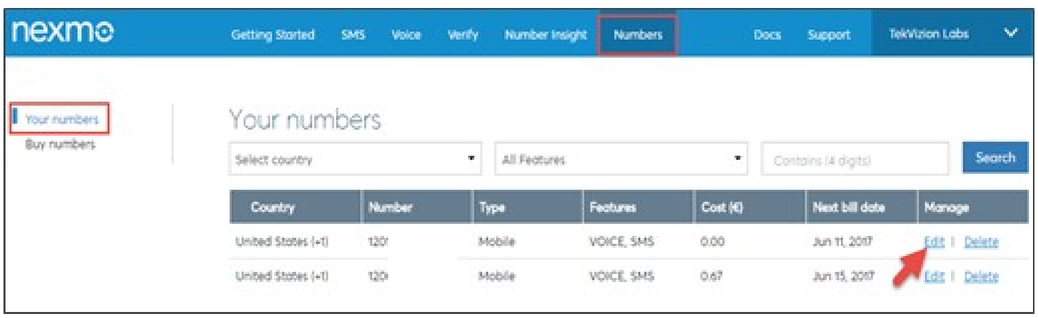
4.3 Nexmo Configuration

4.3.1 Configure Numbers in Nexmo Account

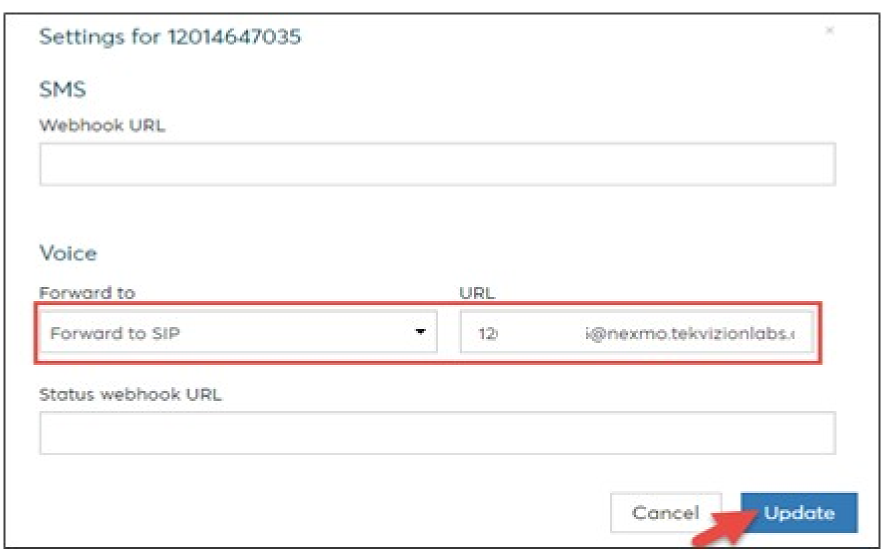
1. Login to the Nexmo account using the credentials provided at the time of registration. A **Key** and **Secret** will be displayed on the dashboard and this can be used as the username and password for Registration SIP Trunks.



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	