



## **Avaya Solution & Interoperability Test Lab**

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# **Application Notes for Configuring 2N<sup>®</sup> VoiceBlue Next 1.17 with Avaya Aura<sup>®</sup> Communication Manager 6.3 Using a SIP Trunk - Issue 1.0**

### **Abstract**

These Application Notes describe the configuration steps required for Avaya Aura<sup>®</sup> Communication Manager 6.3 with 2N<sup>®</sup> VoiceBlue Next using a SIP trunk.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

## 1. Introduction

2N<sup>®</sup> VoiceBlue Next from 2N Telekomunikace is a GSM/UMTS gateway using a VoIP interface. The 2N<sup>®</sup> VoiceBlue Next is designed to maximize cost savings on GSM/UMTS calls. This is ensured by an automated saving solution able to route calls according to time, pre-code or free minutes. A Callback service is able to reduce roaming fees on trips abroad. The PBX can be configured to use the 2N<sup>®</sup> VoiceBlue Next to route calls when the primary PSTN access (i.e. PRI or SIP) is not available.

## 2. General Test Approach and Test Results

The general test approach was to configure the 2N<sup>®</sup> VoiceBlue Next (2N VoiceBlue) to communicate with the Avaya Aura<sup>®</sup> Communication Manager (Communication Manager) as implemented on a customer's premises. Testing focused on verifying that calls could be routed via the 2N VoiceBlue from the communication Manager to the Public switched Telephone Network (PSTN) and vice versa. Various call scenarios were performed to simulate real call types as would be observed on a customer premises including the use of H323 and SIP telephones on the Communication Manager. See Figure 1 for a network diagram. The interoperability compliance test included both feature functionality and serviceability tests.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

### 2.1. Interoperability Compliance Testing

The testing included:

- Verification of the Module status of the 2N VoiceBlue
- Calls to the PSTN from Communication Manager, including Busy numbers
- Calls to Communication Manager from the PSTN, including Busy numbers
- Ensuring proper disconnects
- Observing DTMF
- Hold/Unhold
- Long Call Duration (one hour +)
- Restarting the 2N VoiceBlue

### 2.2. Test Results

Tests were performed to insure interoperability between 2N<sup>®</sup> VoiceBlue Next and Communication Manager. The tests were all functional in nature and performance testing was not included. All the test cases passed successfully.

### 2.3. Support

Technical support from 2N Telekomunikace can be obtained through the following:

Phone: +420 261 301 111

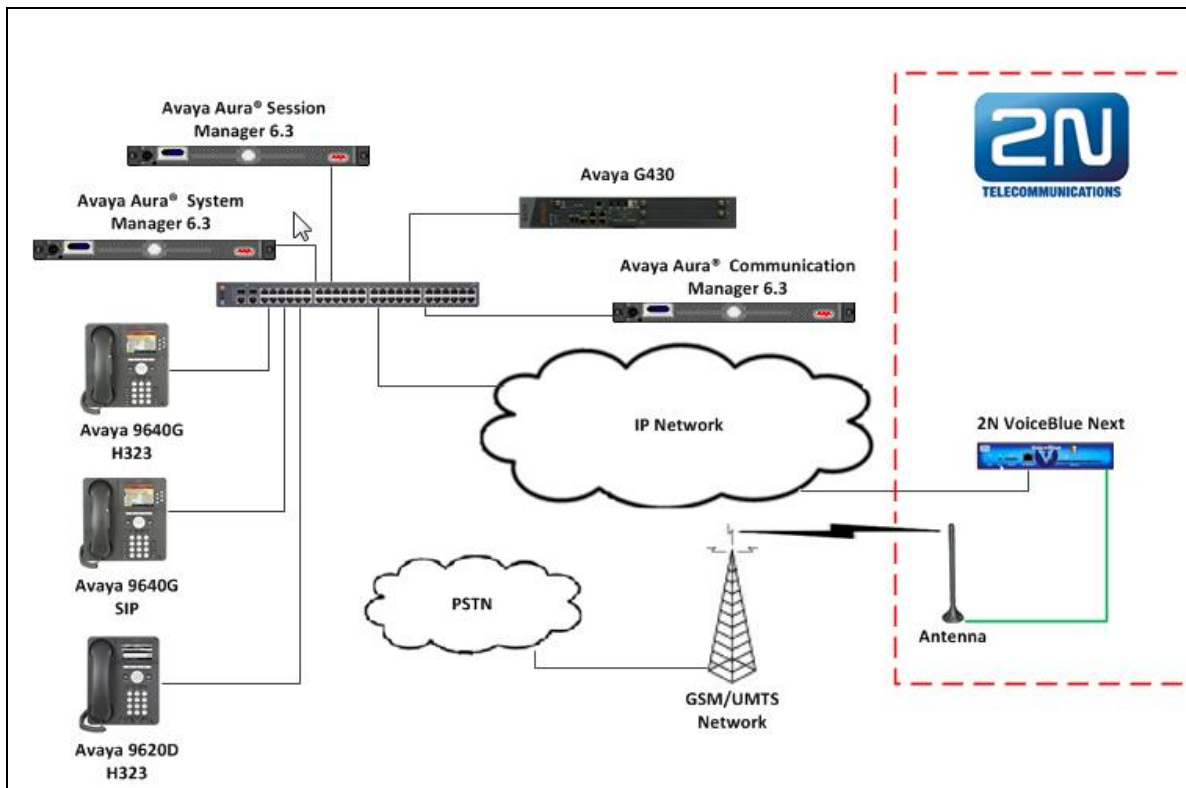
E-mail: sales@2n.cz

Web: <http://support.2n.cz/> (Account required)

### 3. Reference Configuration

**Figure 1** illustrates the network topology used during compliance testing. The Avaya solution consists of a Communication Manager, System Manager, Session Manager and an Avaya G430 Gateway. The Communication Manager is configured to communicate to the 2N VoiceBlue via a SIP Trunk configured on the Session Manager. An Ethernet cable is connected from a 10/100 BaseT port on the 2N VoiceBlue to the IP Network. A variety of Avaya 96XX H323 and SIP Desk phones were used to generate inbound/outbound calls via the 2N VoiceBlue from/to the GSM/UMTS Service Provider. The System Manager was used to administer the Session Manager which had the SIP Trunk configured and the SIP Desk Phone registered.

**Note:** A number of Subscriber Identity Module's (SIM's) from the GSM/UMTS Service Provider is inserted in the 2N VoiceBlue. During compliance only one SIM was used, therefore only one inbound/outbound call could be made simultaneously. Also an antenna was connected to the 2N VoiceBlue.



**Figure 1:** Avaya Aura<sup>®</sup> Communication Manager and 2N Telekomunikace Configuration

## 4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Avaya Equipment	Software / Firmware Version
Avaya Aura <sup>®</sup> Communication Manager	R6.3 Build R016x.03.0.124.0 Update 03.0.124.0-20850
Avaya Aura <sup>®</sup> System Manager	R6.3 Build 6.3.0.8.5682-6.3.8.1814 Update 6.3.3.5.1719
Avaya Aura <sup>®</sup> Session Manager	R6.3 Build 6.3.3.0.633004
Avaya 96xx Deskphones	
- H323 9620D	3.101S
- H323 9640G	3.105S
- SIP 9640D	2.6.10.1
2N Telekomunikace Equipment	Software / Firmware Version
2N <sup>®</sup> VoiceBlue Next	Firmware version 1.17.0.20.2 Bootware version 1.20

## 5. Configure Avaya Aura<sup>®</sup> Communication Manager

Configuration and verification operations on the Communication Manager illustrated in this section were all performed using Communication Manager's SAT administration interface. The information provided in this section describes the configuration of the Communication Manager for this solution. It is implied a working system is already in place including a Node for the Session manager and procr. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 10**. The configuration operations described in this section can be summarized as follows: (Note: during Compliance Testing all inputs not highlighted in Bold were left as Default)

- Configure signaling-group
- Configure Trunk Group

## 5.1. Configure signaling-group

A signalling group is required before a trunk-group can be configured. Use the **add signaling-group** command followed by next available signaling-group number to configure the following:

- **Group Type:** Enter **sip**
- **Transport Method:** Enter **tcp**
- **Near-end Node Name:** Enter **procr**
- **Far-end Node Name:** Enter **SM63RPSIG** (this is the Node of the Session Manager)
- **Near-end Listen Port:** Enter **5060**
- **Far-end Listen Port:** Enter **5060**
- **Far-end Domain:** Enter **devconnect.local** (This is the domain configured on the Session Manager)

```
add signaling-group 1                               Page 1 of 2
                                                    SIGNALING GROUP
Group Number: 1                                   Group Type: sip
IMS Enabled? n                                   Transport Method: tcp
  Q-SIP? n
  IP Video? n                                     Enforce SIPS URI for SRTP? y
Peer Detection Enabled? y Peer Server: SM
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n

Near-end Node Name: procr                         Far-end Node Name: SM63RPSIG
Near-end Listen Port: 5060                       Far-end Listen Port: 5060
                                                    Far-end Network Region: 1

Far-end Domain: devconnect.local

Incoming Dialog Loopbacks: eliminate             Bypass If IP Threshold Exceeded? n
  DTMF over IP: rtp-payload                       RFC 3389 Comfort Noise? n
Session Establishment Timer(min): 3              Direct IP-IP Audio Connections? y
  Enable Layer 3 Test? y                         IP Audio Hairpinning? n
                                                    Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n         Alternate Route Timer(sec): 6
```

## 5.2. Configure Trunk Group

This section describes the Trunk Group configuration used during compliance. Use the **add trunk-group** command followed by next available Group number and configure the following:  
**Page 1**

- Group Type: Enter **sip**
- Group Name: Enter an informative name for the trunk i.e. **To SM6.3 SIP**
- TAC: Enter a TAC number i.e. **701**
- Service Type: Enter **public-ntwrk**
- Signaling Group: Enter **1** (This is the Signaling Group configured in **Section 5.1**)

```
add trunk-group 1                                     Page 1 of 21
                                                    TRUNK GROUP
Group Number: 1                                     Group Type: sip          CDR Reports: y
  Group Name: To SM6.3 SIP                          COR: 1                  TN: 1          TAC: 701
  Direction: two-way                               Outgoing Display? n
  Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: public-ntwrk                         Auth Code? n
                                                    Member Assignment Method: auto
                                                    Signaling Group: 1
                                                    Number of Members: 30
```

## 6. Configuring Avaya Aura® Session Manager

A number of configurations are required to enable the Session Manager to route calls between the Communication Manager and 2N VoiceBlue. All configurations of the Session Manager are performed using System Manager. The configuration operations described in this section can be summarized as follows:

- Logging on to System Manager
- Create 2N VoiceBlue Next as a SIP Entity
- Create an Entity Link for 2N VoiceBlue Next
- Create a Routing Policy for 2N VoiceBlue Next
- Create a Dial Pattern for 2N VoiceBlue Next

**Note:** It is implied a working system is already in place. During Compliance testing a SIP Entity and an Entity Link for the Communication Manager were created. Also a Routing Policy and a Dial Pattern to route calls to the Communication Manager were created and are outside the scope of this Application Note.

### 6.1. Logging on to Avaya Aura® System Manager

Log on by accessing the browser-based GUI of System Manager, using the URL “http://<fqdn>/SMGR” or “http://<ip-address>/SMGR”, where:

“<fqdn> is the fully qualified domain name of the Avaya Aura® System Manager or the “<ipaddress>” is the IP address of Avaya Aura® System Manager.

Once the System Manager Web page opens Log in with the appropriate credentials and click on the Log on button.

AVAYA Avaya Aura® System Manager 6.3

Home / Log On

Log On

Recommended access to System Manager is via FQDN.  
[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

Use the "Change Password" hyperlink on this page to change the password manually, and then login.

Also note that single sign-on between servers in the environment is not supported.

User ID:

Password:

Log On Cancel

Change Password

Supported Browsers: Internet Explorer 8.x, 9.x or 10.x or Firefox 15.0, 16.0 or 17.0.

## 6.2. Create 2N<sup>®</sup> VoiceBlue Next as a SIP Entity

Once logged in select **Routing** from under the **Elements** column.

The screenshot displays the Avaya Aura System Manager 6.3 web interface. At the top left is the AVAYA logo. The page title is "Avaya Aura® System Manager 6.3". On the top right, it shows "Last Logged on at November 7, 2013 11:24 AM" and links for "Help | About | Change Password | Log off admin". The main content area is divided into three columns: "Users", "Elements", and "Services".

- Users**
  - Administrators: Manage Administrative Users
  - Directory Synchronization: Synchronize users with the enterprise directory
  - Groups & Roles: Manage groups, roles and assign roles to users
  - User Management: Manage users, shared user resources and provision users
- Elements**
  - Communication Manager: Manage Communication Manager 5.2 and higher elements
  - Communication Server 1000: Manage Communication Server 1000 elements
  - Conferencing: Manage Conferencing Multimedia Server objects
  - IP Office: Manage IP Office elements
  - Meeting Exchange: Manage Meeting Exchange and Avaya Aura Conferencing 6.0 elements
  - Messaging: Manage Avaya Aura Messaging, Communication Manager Messaging, and Modular Messaging
  - Presence: Presence
  - Routing**: Session Manager Routing Administration (highlighted with a red box)
  - Session Manager: Session Manager Administration, Status, Maintenance and Performance Management
- Services**
  - Backup and Restore: Backup and restore System Manager database
  - Bulk Import and Export: Manage Bulk Import and Export of Users, User Global Settings, Roles, Elements and others
  - Configurations: Manage system wide configurations
  - Events: Manage alarms, view and harvest logs
  - Geographic Redundancy: Manage Geographic Redundancy
  - Inventory: Manage, discover, and navigate to elements
  - Licenses: View and configure licenses
  - Replication: Track data replication nodes, repair replication nodes
  - Scheduler: Schedule, track, cancel, update and delete jobs
  - Security: Manage Security Certificates
  - Shutdown: Shutdown System Manager Gracefully
  - Software Management: Upgrade and Patch Management for Communication Manager devices and IP Office
  - Templates: Manage Templates for Communication Manager, Messaging System and IP Office elements



A SIP Entity must be added for the 2N VoiceBlue. To add a SIP Entity, select **SIP Entities** on the left panel menu and then click on the **New** button (not shown).

**Note:** A SIP Entity was already configured for the Communication Manager and was called **CM63**.

Enter the following for 2N VoiceBlue SIP Entity:

Under **General:**

- **Name** Enter an informative name (e.g., **2N VoiceBlue**)
- **FQDN or IP Address** enter the IP address of the signalling interface of the 2N VoiceBlue
- **Type** Select **SIP Trunk** from the dropdown box
- **Location** Select **location** from the dropdown box, during compliance testing DevConnectRP was used.
- **Time Zone** Select Time Zone for this location from the dropdown box
- **SIP Timer** Enter **4**

Once the correct information is entered click the **Commit** Button.

**Note:** During compliance testing **Adaptation** was left blank.

The screenshot displays the Avaya Aura System Manager 6.3 interface. The left-hand navigation pane shows a tree structure with 'Routing' selected, and 'SIP Entities' highlighted. The main content area is titled 'SIP Entity Details' and includes a 'General' tab. The configuration fields are as follows:

- Name:** 2N VoiceBlue
- FQDN or IP Address:** 10.10.16.226
- Type:** SIP Trunk
- Notes:** (empty)
- Adaptation:** (empty)
- Location:** DevConnectRP
- Time Zone:** Europe/Dublin
- Override Port & Transport with DNS SRV:** (unchecked)
- SIP Timer B/F (in seconds):** 4
- Credential name:** (empty)
- Call Detail Recording:** egress

Buttons for 'Commit' and 'Cancel' are visible at the top right of the configuration area.

### 6.3. Create an Entity Link for 2N<sup>®</sup> VoiceBlue Next

The SIP trunk between the Session Manager and the 2N VoiceBlue requires an Entity Link. To add an Entity Link, select **Entity Links** on the left panel menu and click on the **New** button (Not shown) Enter the following:

- **Name** An informative name, (e.g. **VoiceBlue\_5060\_TCP**).
- **SIP Entity 1** Select **SM63** from the **SIP Entity 1** dropdown box.
- **Protocol** Select **TCP** from the Protocol drop down box
- **Port** Enter **5060**
- **SIP Entity 2** Select **2N VoiceBlue** from the **SIP Entity 2** dropdown box (configured in **Section 6.2**)
- **Port** Enter **5060** as the Port.
- **Connection Policy** Check the **Trusted** check box.

Click **Commit** to save changes. The following screen shows the Entity Links used.

Avaya Aura® System Manager 6.3

Last Logged on at November 7, 2013 11:24 AM  
Help | About | Change Password | Log off admin

Routing \* Home

Home / Elements / Routing / Entity Links

Entity Links   [Help ?](#)

1 Item Refresh Filter: Enable

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service	Notes
<input type="checkbox"/>	*VoiceBlue_5060_TCP	*SM63	TCP	*5060	*2N VoiceBlue	*5060	trusted	<input type="checkbox"/>	

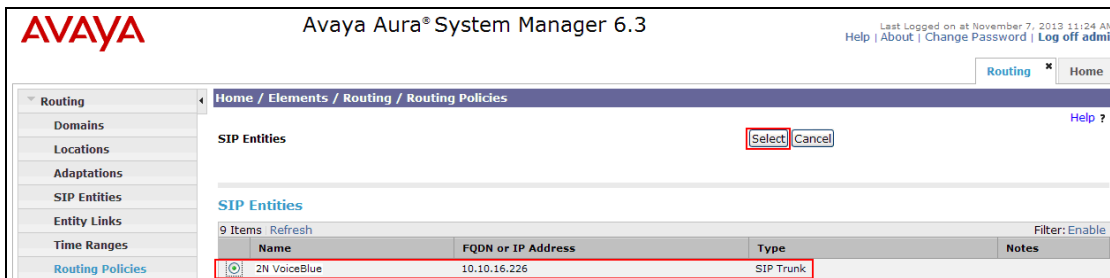
Select : All, None

## 6.4. Create a Routing Policy for 2N<sup>®</sup> VoiceBlue Next

Create routing policies to direct calls to the 2N VoiceBlue via the Session Manager. To add a routing policy, select **Routing Policies** on the left panel menu and then click on the **New** button (not shown). In **Routing Policy Details** enter an informative name in the **Name** field (example, **To GSM**) and enter **0** in the **Retries** field. In **SIP Entity as Destination**, click the **Select** button. A Routing Policy was also configured to direct calls to the Communication Manager, but is outside the scope of this Application Note.

The screenshot displays the Avaya Aura System Manager 6.3 interface. The top navigation bar includes the Avaya logo, the title 'Avaya Aura System Manager 6.3', and user information: 'Last Logged on at November 7, 2013 11:24 AM', 'Help | About | Change Password | Log off admin'. The breadcrumb trail is 'Home / Elements / Routing / Routing Policies'. The left sidebar contains a menu with 'Routing Policies' selected. The main content area is titled 'Routing Policy Details' and includes 'Commit' and 'Cancel' buttons. Under the 'General' tab, the 'Name' field is 'To GSM', 'Disabled' is unchecked, and 'Retries' is '0'. The 'SIP Entity as Destination' section has a 'Select' button highlighted. A 'Notes' field is also present.

Once the **SIP Entity** List screen opens, check the **2N VoiceBlue** radio button. Click on the **Select** button to confirm the chosen options and then return to the Routing Policies Details screen and select the **Commit** button (Not shown) to save.



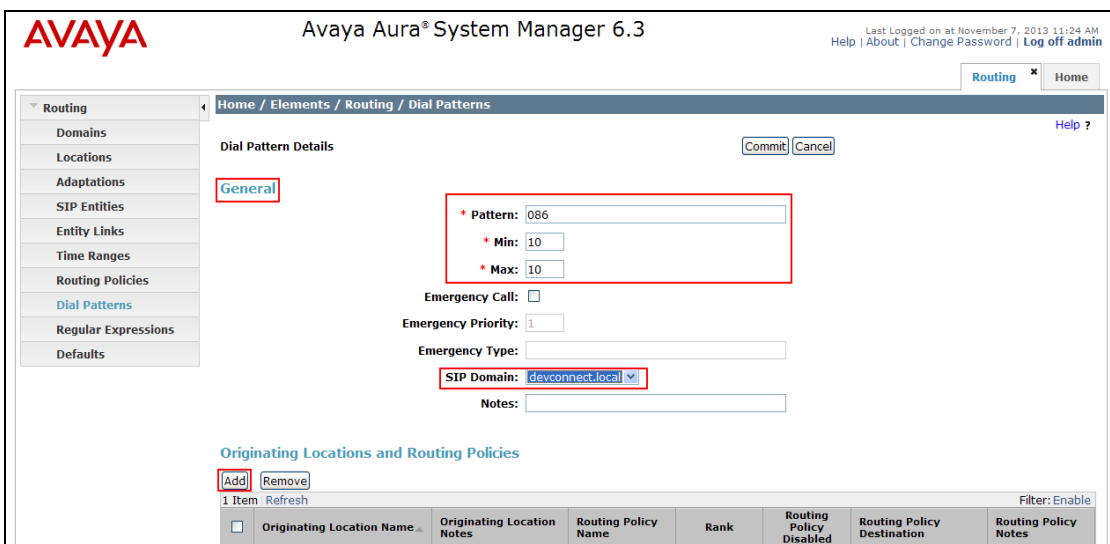
## 6.5. Create a Dial Pattern for 2N<sup>®</sup>VoiceBlue Next

A dial pattern must be created on the Session Manager to route calls to and from the 2N VoiceBlue. During compliance testing a number of patterns were used. The example below only shows 086, the remaining numbers are created the same way. To configure the Dial Pattern to route calls to the VoiceBlue Gateway, select **Dial Patterns** on the left panel menu and then click on the **New** button (not shown). A Dial Pattern was also configured to route calls to the Communication Manager, but is outside the scope of this Application Note.

Under **General** carry out the following for each number

- **Pattern** Enter 086
- **Min** Enter 10 as the minimum length of dialed number
- **Max** Enter 10 as the maximum length of dialed number
- **SIP Domain** Select **devconnect.local** from the drop down box

Click the **Add** button in **Originating Locations and Routing Policies**.



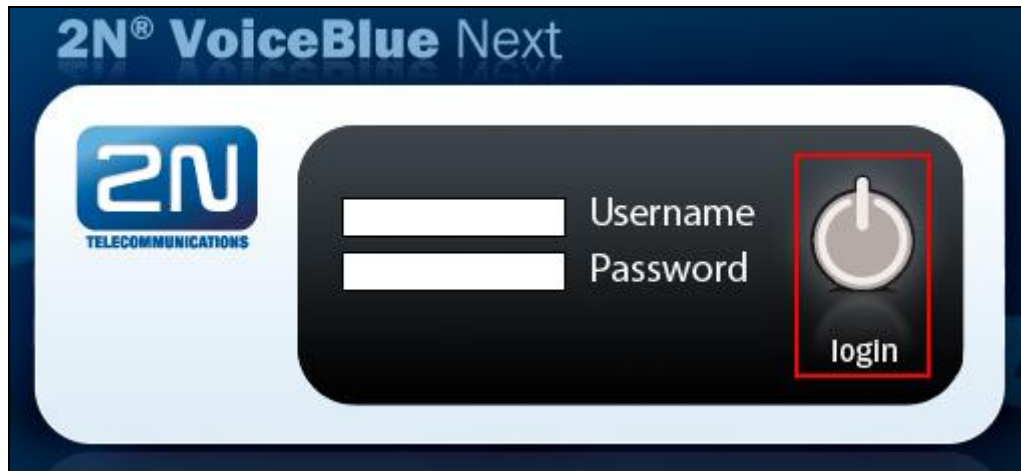
In **Originating Location** check the **DevConnectRP** check box. Under **Routing Policies** check the **To GSM** check box. Click on the **Select** button to confirm the chosen options and then be returned to the Dial Pattern screen (shown previously), select the **Commit** button to save.

The screenshot shows the Avaya Aura System Manager 6.3 interface. The breadcrumb navigation is Home / Elements / Routing / Dial Patterns. The 'Originating Location' section has a 'Select' button highlighted. Below it, the 'Originating Location' section has a checkbox for 'Apply The Selected Routing Policies to All Originating Locations' which is unchecked. A table with 1 item is shown, with 'DevConnectRP' checked. The 'Routing Policies' section has a table with 8 items. The 'To GSM' policy is checked.

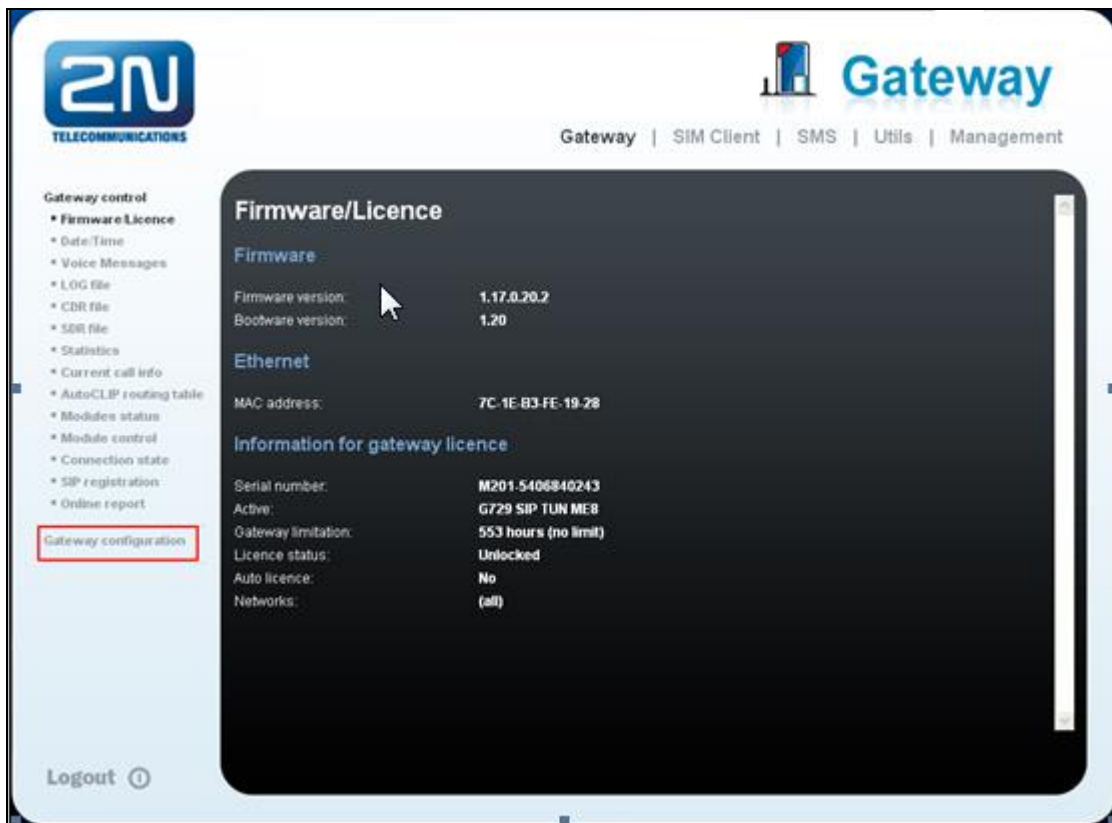
Name	Disabled	Destination	Notes
<input type="checkbox"/> CS1KPG1	<input type="checkbox"/>	CS1KPG1	
<input type="checkbox"/> NRS	<input type="checkbox"/>	NRS	
<input type="checkbox"/> TOAAC63SIP	<input type="checkbox"/>	AACC63CMSIP	
<input type="checkbox"/> ToCM6.3	<input type="checkbox"/>	CM63	
<input type="checkbox"/> ToCS1KRP	<input type="checkbox"/>	CS1kRP	
<input checked="" type="checkbox"/> To GSM	<input type="checkbox"/>	2N StarGate	

## 7. 2N<sup>®</sup> VoiceBlue Next configuration

To access the 2N VoiceBlue, open a web browser and navigate to **http://<IP address of the 2N VoiceBlue>** and log in using the appropriate credentials and click on the **login** button.



Once logged in, click on **Gateway configuration**.



When the **System parameters** page opens, enter the following:  
Using the scroll bar on the right side of the page scroll down to **Others** section.



During compliance testing the SIM did not require a pin, therefore the **PIN:** box was left empty. Enter # in the **End of dialling** box. Click on the **Save Settings** icon on the bottom of the page to save the configuration. Select **VoIP parameters** to continue with the VoiceBlue configuration.

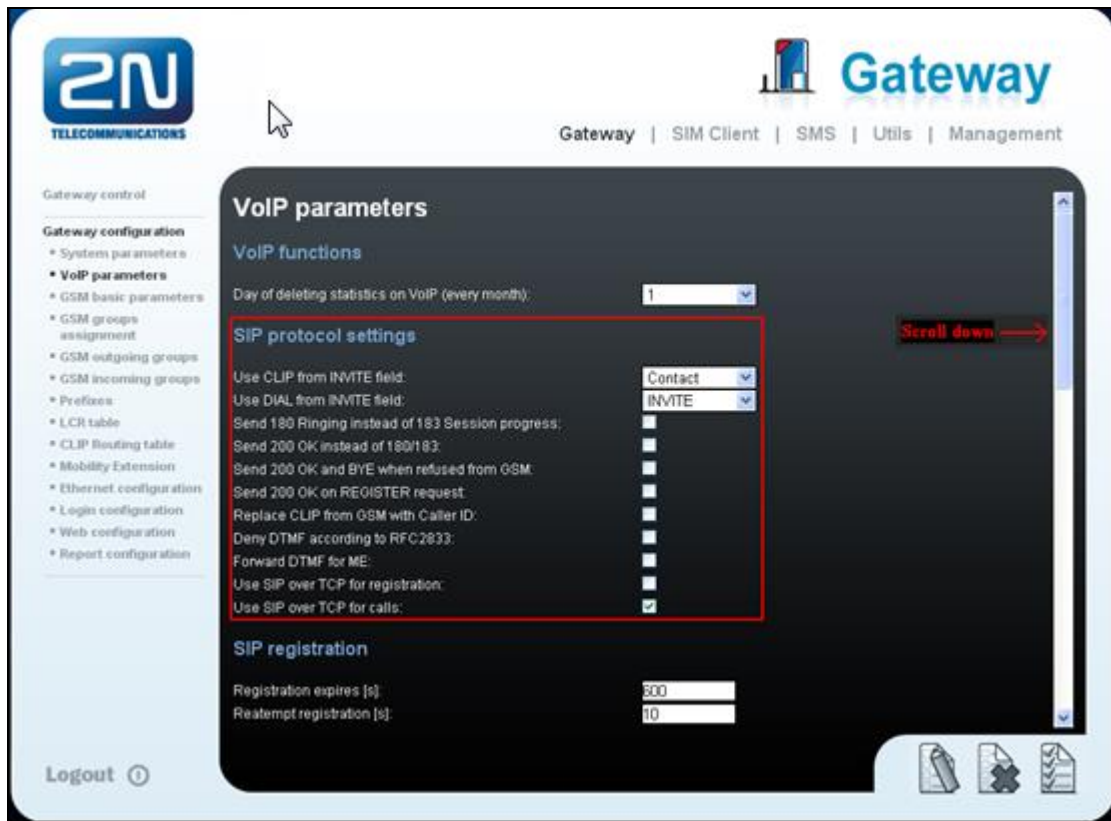




When the **VoIP parameters** page opens, enter the following in the **SIP protocol settings** section:

- **Use CLIP from INVITE field**      Select **Contact** from the dropdown box
- **Use DIAL from INVITE field**      Select **INVITE** from the dropdown box
- **USE SIP over TCP for calls** Check the Check box

Using the scroll bar on the right side of the page, scroll down to the **Codec Priority** section.



In the **Codec priority** section, enter the following:

- **Priority 1** Select **G711a (8)** from the dropdown box
- **Priority 2** Select **G711u (0)** from the dropdown box
- **Priority 3** Select **G729 (18)** from the dropdown box

Using the scroll bar on the right side of the page, scroll down to the **IP addresses** section.

The screenshot shows the 2N Gateway configuration interface. The top left features the 2N TELECOMMUNICATIONS logo. The top right has the Gateway logo and navigation links: Gateway | SIM Client | SMS | Utils | Management. A left sidebar lists 'Gateway control' options, with 'VoIP parameters' selected. The main content area is divided into two sections: 'Codec priority' and 'IP addresses'. The 'Codec priority' section has three dropdown menus: Priority 1 (G711a (8)), Priority 2 (G711u (0)), and Priority 3 (G729 (18)). The 'IP addresses' section contains several rows of input fields for IP addresses and ports, with 'Set default port' buttons. A red arrow points to the scroll bar on the right side of the main content area.

Section	Field	Value	Port	Action
Codec priority	Priority 1:	G711a (8)		
	Priority 2:	G711u (0)		
	Priority 3:	G729 (18)		
IP addresses	SIP proxy (IP->GSM):	10.10.16.214	5060	Set default port
	- alternative 1:	0.0.0.0		
	- alternative 2:	0.0.0.0		
	- alternative 3:	0.0.0.0		
	- alternative 4:	0.0.0.0		
	- alternative 5:	0.0.0.0		
	- alternative 6:	0.0.0.0		
	- alternative 7:	0.0.0.0		
	- alternative 8:	0.0.0.0		
	- no reply on other IP addresses requests:			
SIP proxy (GSM->IP):	10.10.16.214	5060	Set default port	
SIP registrar:	0.0.0.0	5060	Set default port	
NAT firewall:	0.0.0.0			
STUN server:	0.0.0.0	3478	Set default port	
Next STUN server request (60-6553, 0=off) [s]:	600			

In the **IP addresses** section, enter the following:

- **SIP proxy (IP->GSM)** Enter the IP address of the Session Manager and Port number
- **SIP proxy (GSM->IP)** Enter the IP address of the Session Manager and Port number
- Ring tone to VoIP Select **From GSM** from the dropdown box

Click on the **Save Settings** icon on the bottom of the page to save the configuration.



## 8. Verification Steps

This section provides tests that can be performed to verify correct configuration of the Avaya and 2N VoiceBlue solution.

### 8.1. Verify 2N<sup>®</sup>VoiceBlue Next activity

Log in to the Gateway and click on **Modules status** and verify that something similar to the screen shot below is seen. During compliance testing, only one SIM was used. The following screen shows that module 01 was used and has signal strength of -85 dBm.



### 8.2. Verify Calls via the 2N<sup>®</sup>VoiceBlue Next

Place a call from a Desk Phone on the Communication Manager via the 2N VoiceBlue to an external number, ensure the call is completed and verify two-way audio is heard.

Place a call from an external number via the 2N VoiceBlue to a Desk Phone on the Communication Manager, ensure the call is completed and verify two-way audio is heard.

## 9. Conclusion

A full and comprehensive set of feature functional test cases were performed during Compliance testing. 2N<sup>®</sup> VoiceBlue Next is considered compliant with Avaya Aura<sup>®</sup> Communication Manager 6.3. All test cases have passed and met the objectives outlined in **Section 2.1**.

## 10. Additional References

These documents form part of the Avaya official technical reference documentation suite. Further information may be had from <http://support.avaya.com> or from your Avaya representative.

- [1] *Administering Avaya Aura<sup>®</sup> Communication Manager, Release 6.3, October 2013, Document Number 03-300509, Issue 9.0.*
- [2] *Avaya Aura<sup>®</sup> Communication Manager Feature Description and Implementation, Release 6.3, May 2013, Document Number 555-245-205, Issue 10.0.*
- [3] *Administering Avaya Aura<sup>®</sup> Session Manager, Release 6.3, Issue 3 October 2013*
- [4] *Administering Avaya Aura<sup>®</sup> System Manager, Release 6.3, Issue 3, October, 2013*

Product Documentation for VoiceBlue can be obtained from 2N Telekomunikace at <http://www.2n.cz/>

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