Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Session Manager R6.2 and Avaya Aura® Communication Manager R6.2 to interoperate with 2N Telekomunikace Helios IP - Issue 1.0

Abstract

These Application Notes describe the configuration steps required for 2N Telekomunikace Helios IP to interoperate with Avaya Aura® Session Manager R6.2 and Avaya Aura® Communication Manager R6.2. The 2N Telekomunikace Helios IP is a door communicator that supports both voice and video transmission using the Session Initiation Protocol (SIP).

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

These Application Notes describe the configuration steps required for 2N Telekomunikace Helios IP to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. The 2N Telekomunikace Helios IP is a door communicator that supports both voice and video transmission using the Session Initiation Protocol (SIP), in addition to being a door entry device with its keyboard or card reader. In the compliance testing, the 2N Telekomunikace Helios IP was set up as a SIP user on Avaya Aura® Session Manager and underwent testing of various call scenarios with other Avaya telephones.

2. General Test Approach and Test Results

The general test approach was to place calls to and from Helios IP and exercise basic telephone operations. For serviceability testing, failures such as cable pulls and hardware resets were performed.

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 interoperability compliance test included feature and serviceability testing. The feature testing was to verify that:

- Helios IP successfully registers with Session Manager.
- Helios IP successfully establishes audio calls with Avaya H.323, SIP and digital endpoints registered to Session Manager and Communication Manager.
- Helios IP successfully establishes audio calls with PSTN.
- Helios IP successfully establishes video calls with one-X Communicator registered to.Communication Manager and Flare device registered to Session Manager.
- Helios IP successfully negotiates the appropriate audio codec.
- Helios IP successfully negotiates the appropriate video codec.
- DTMF tones could be passed successfully to Communication Manager Messaging.
- Helios IP successfully calls multiple destinations using a Sequential Hunt Group.
- Helios IP successfully streams video to a PC running Helios IP Eye when calling phones without video capabilities.
- Use of feature access codes for call pickup.
- Correct handling of forwarded calls, cover paths and cover answer groups.

The serviceability testing focused on verifying the ability of Helios IP to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet cable to the devices and denying service on Session Manager.

2.2. Test Results

All test cases passed. As Helios IP was not designed to be a desk phone, the following features were not supported:

- Handling multiple calls.
- Call hold and un-hold.
- Call park and un-park.
- Call transfer and conference.

2.3. Support

Technical support on 2N Telekomunikace Helios IP can be obtained through the following:

- **Phone:** +420 261 301 111
- **Web:** http://www.2n.cz/en/support/

3. Reference Configuration

Figure 1 illustrates a test configuration that was used to compliance test the interoperability of Helios IP (HIP) with Session Manager and System Manager. The configuration consists of Communication Manager configured with Communication Manager Messaging, System Manager and Session Manager. Communication Manager has connections to one-X Communicator (H323), 9630 IP (H323) deskphone and 2420 Digital Telephone. Session Manager has SIP registrations with Flare, Helios IP and 9630 IP (SIP) deskphone. An ISDN-PRI trunk connects Communication Manager to the PSTN. The Helios IP Eye application is also installed on a PC to receive the video streaming from Helios IP.

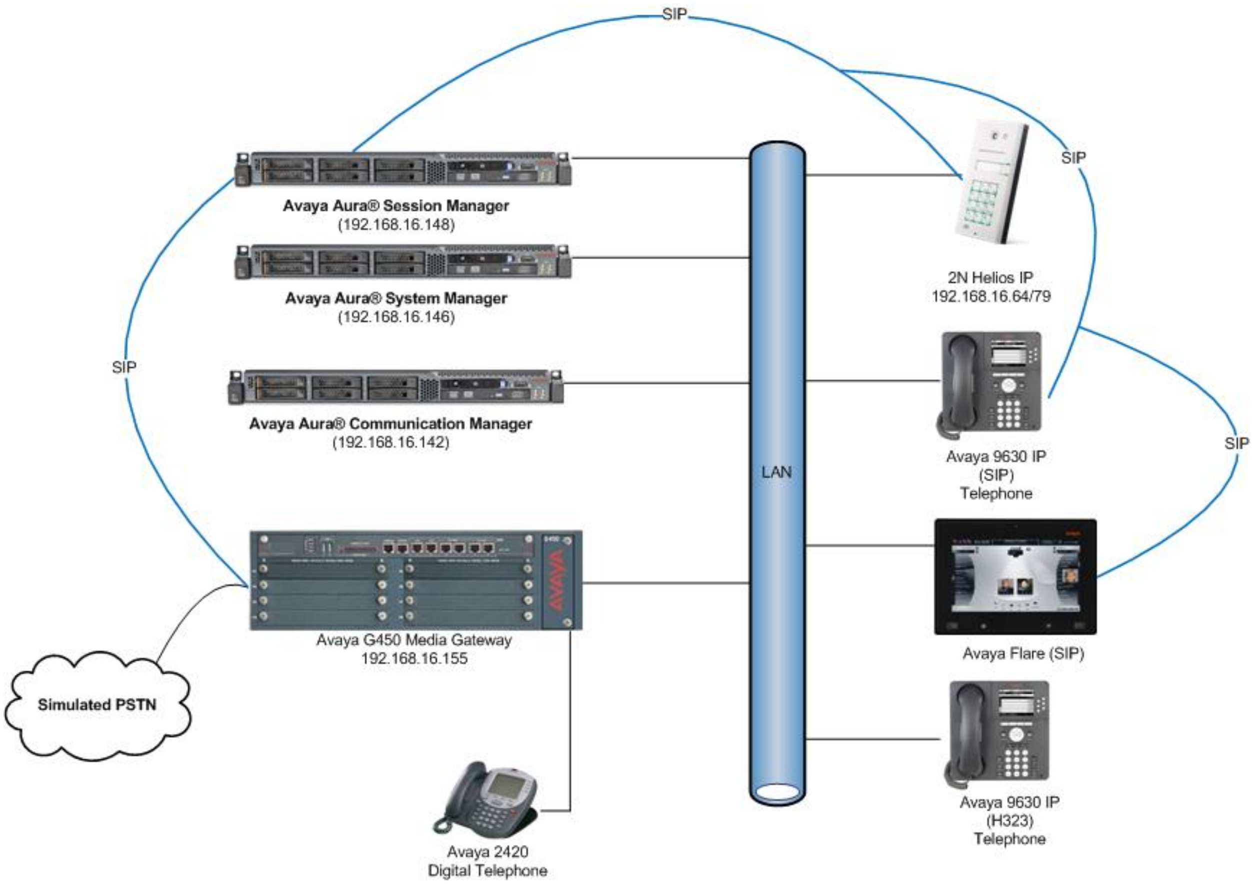


Figure 1: Avaya Aura® Session Manager and Avaya Aura® Communication Manager with 2N Telekomunikace Helios IP Configuration

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Session Manager running on Avaya S8800 Server	R6.2 SP1
Avaya Aura® System Manager running on Avaya S8800 Server	R6.2 SP1
Avaya Aura® Communication Manager running on Avaya S8800 Server	R6.2 SP1
Avaya 9630 IP Telephone	3.1 SP4 (H.323) 2.6.7 (SIP
Avaya Flare/A175 Desktop Video Device	1.1.0
Avaya 2420 Digital Telephone	N.A.
Avaya one-X® Communicator	3.1 SP3 Patch 3
2N Telekomunikace Helios IP Tested models: 9137111CKU (1 button + camera + keypad)	Software version: 1.15.3.308.6 Bootloader version: 1.4.0.6.0 Hardware version: 535v5

5. Configure Avaya Aura® Communication Manager

The configuration changes in this section for Communication Manager are performed through the Site Administration tool and via the System Manager web interface. Except where stated, the parameters in all steps are the default settings and are supplied for reference. For all other provisioning information such as provisioning of the trunks, call coverage, extensions, and voicemail, please refer to the Avaya product documentation in **Section 9**.

The procedures fall into the following areas:

- Configure Signalling Group
- Configure IP Codec Set
- Configure SIP User
- Configure Endpoints for IP Video

5.1. Configure Signaling Group

It is assumed a trunk and signaling group are configured between Communication Manager and Session Manager. Enter the command **change sig x** where **x** is the relevant signaling group number. Ensure the following settings are set in order to enable video for the endpoints:

- **IP Video?**: Set to **y**
- **Initial IP-IP Direct Media?**: Set to **n**

```
change signaling-group 4                                          Page   1 of   2
                                SIGNALING GROUP

 Group Number: 4              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? y          Priority Video? n        Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM



   Near-end Node Name: procr                 Far-end Node Name: sm62sigint
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region:

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

5.2. Configure IP Codec Set

The IP Codec set must be configured with the codecs for use by IP endpoints and video must be enabled. Enter the command **change ip-codec-set x** where **x** is the relevant codec set and set the **Audio Codec** to be used on **Page 1**. In the example below, codecs **G.711A, G.711MU and G.729** are configured.

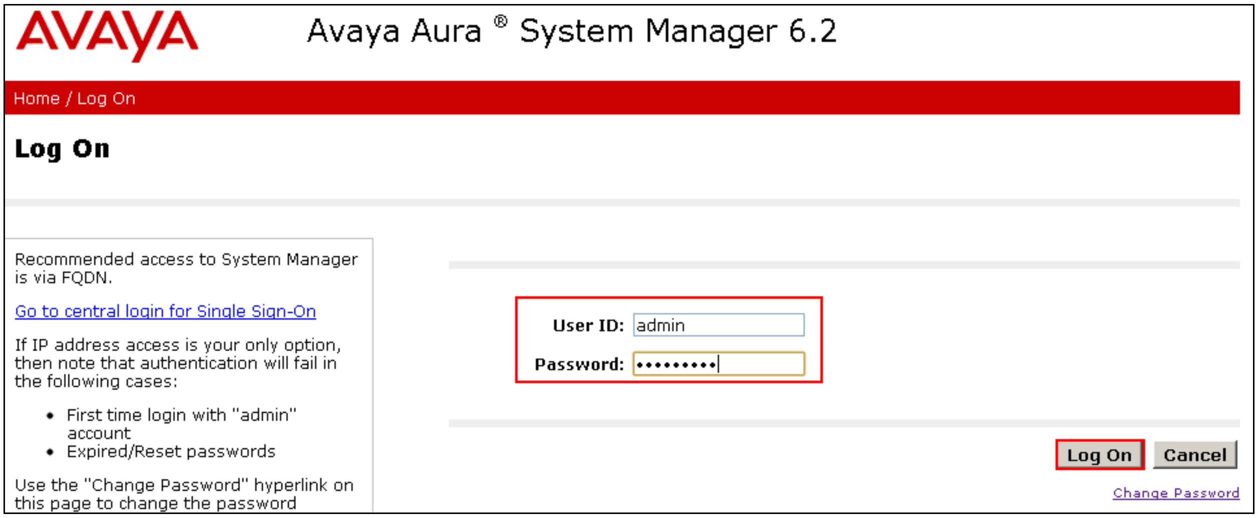
```
change ip-codec-set 1                                         Page   1 of   2

                          IP Codec Set

    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711A            n          2        20
 2: G.711MU           n          2        20
 3: G.729             n          2        20
 4:
 5:
 6:
 7:
```

Continue to **Page 2**, ensure **Allow Direct-IP Multimedia** is set to **y** and **Maximum Call Rate for Direct-IP Multimedia** is set to **576:Kbits**.

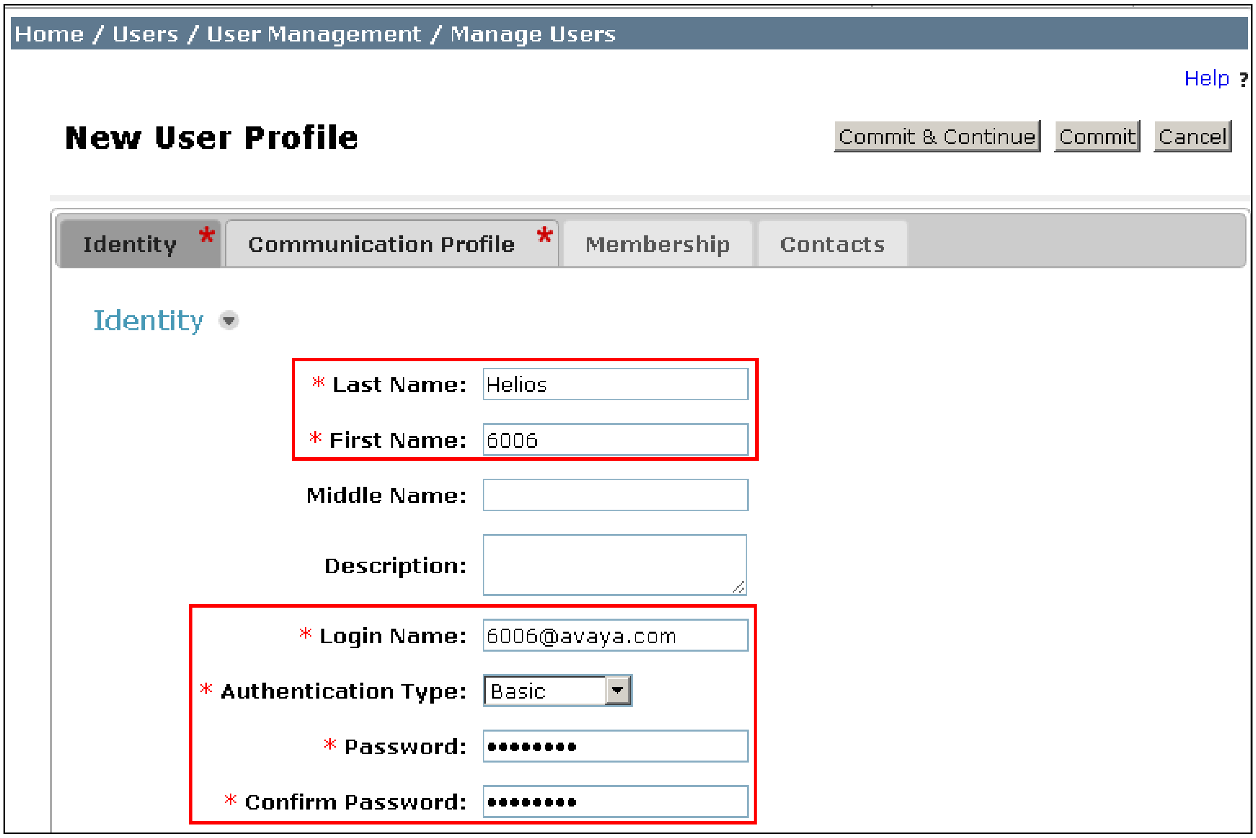
```
change ip-codec-set 1                                         Page   2 of   2

                          IP Codec Set

                              Allow Direct-IP Multimedia? y
              Maximum Call Rate for Direct-IP Multimedia:   576:Kbits
     Maximum Call Rate for Priority Direct-IP Multimedia:   576:Kbits

                    Mode                Redundancy
    FAX             t.38-standard        0
    Modem           off                  0
    TDD/TTY         US                   3
    Clear-channel   n                    0
```

5.3. Configure SIP User

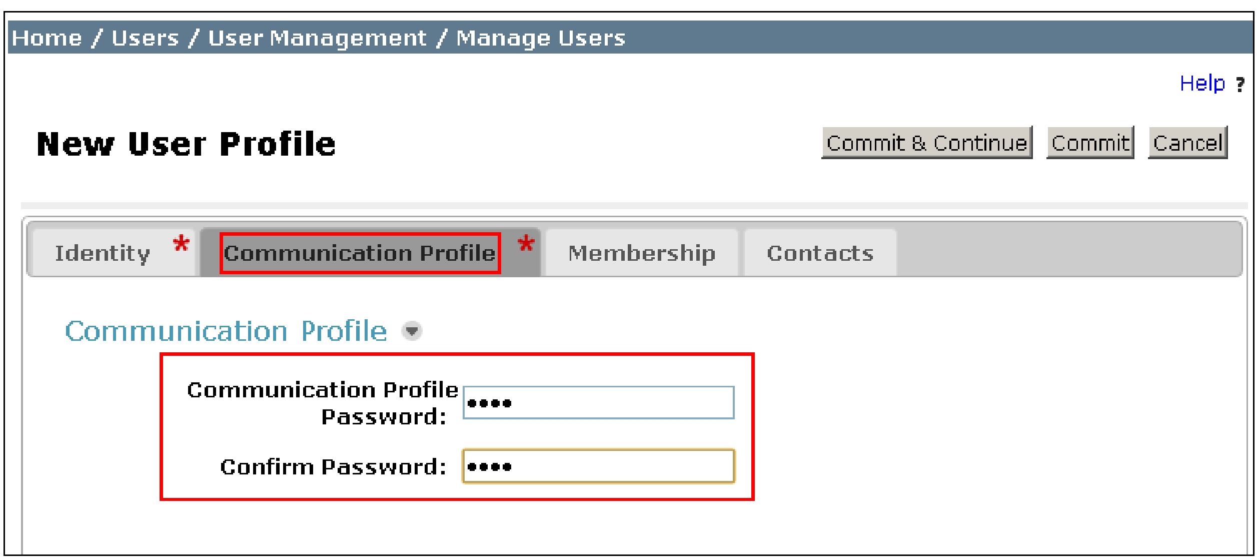
A SIP user must be added for each HIP endpoint required. Navigate to the System Manager web interface, in this case **https://192.168.16.146/SMGR** and login with the relevant credentials.



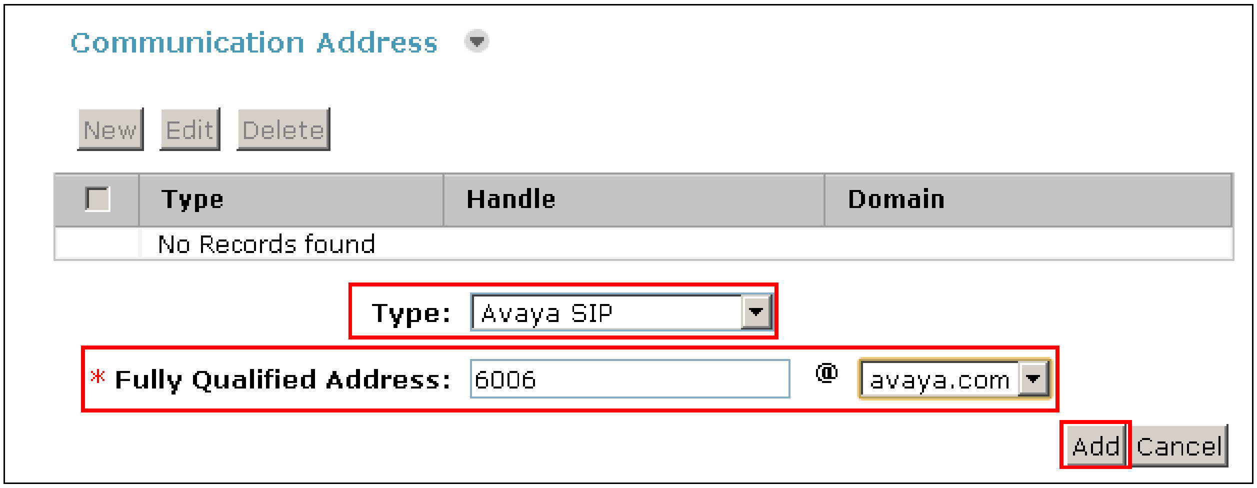
Navigate to **User Management → Manage Users → New** (not shown) and enter an identifying **Last Name** and **First Name**, an appropriate **Login Name**, set **Authentication Type** to **Basic** and administer a password in the **Password** and **Confirm Password** fields.



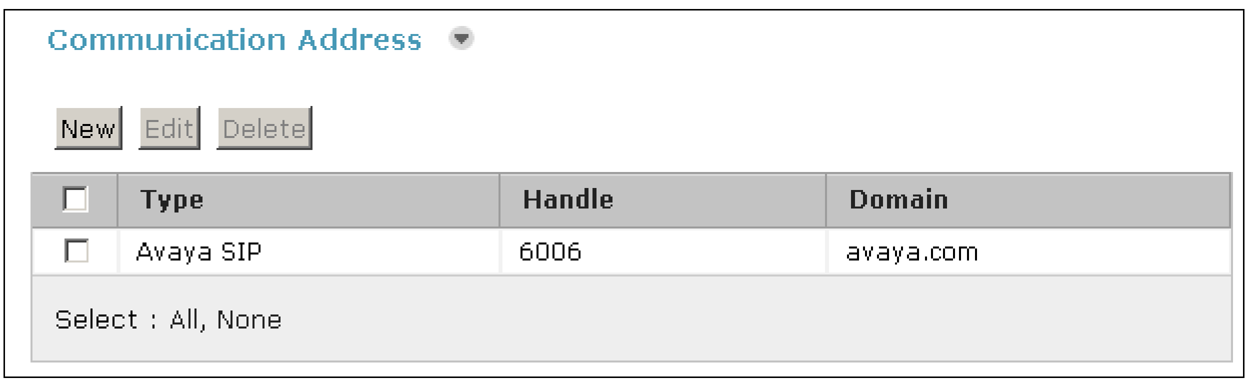
Click on the **Communication Profile** tab and enter and confirm a **Communication Profile Password** used when logging in the SIP endpoint.



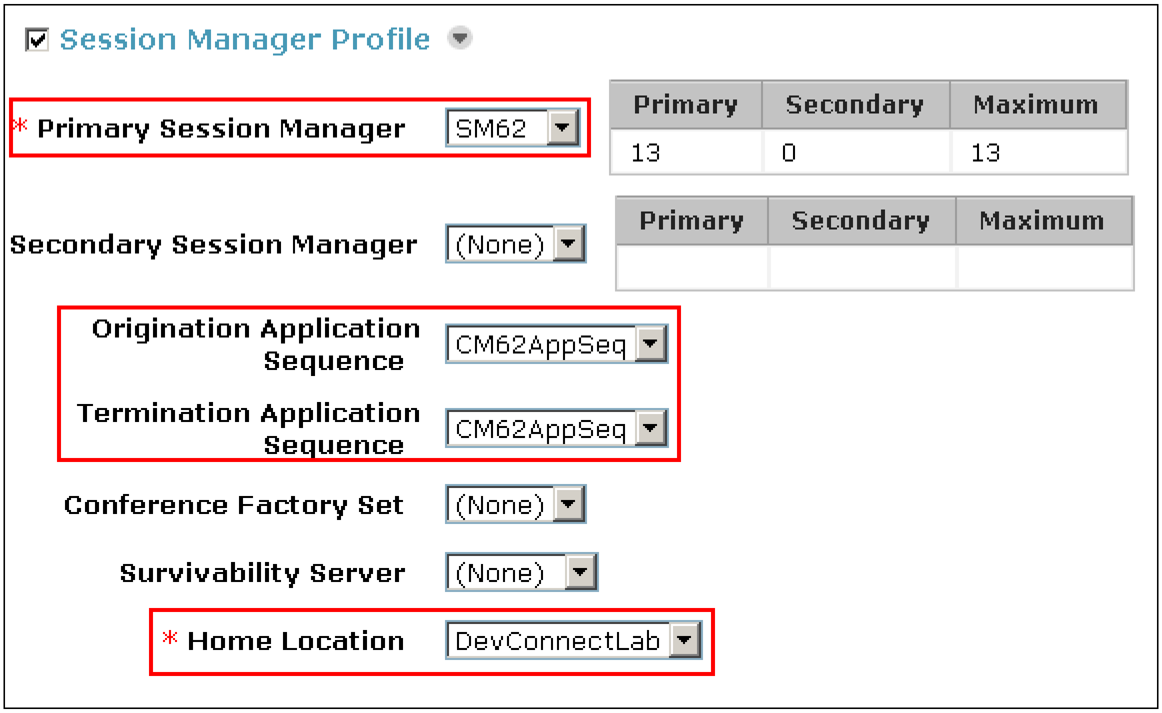
On the same page, scroll down and under **Communication Address** click **New**, select **Avaya SIP** from the **Type** drop down box and enter the **Fully Qualified Address** of the new SIP user. Click **Add** when done.



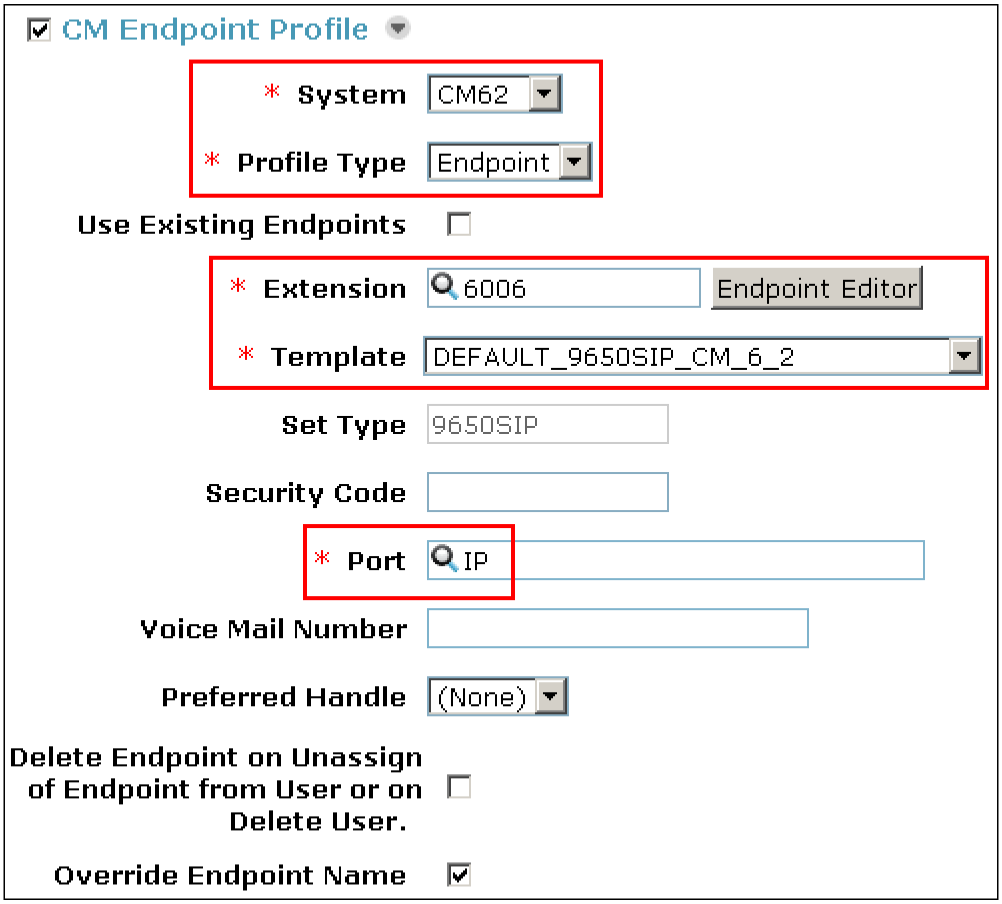
The Communication Address will now appear added.



Continue to scroll down on the same page, enter the **Primary Session Manager**, **Origination Application Sequence**, **Termination Application Sequence** and **Home Location** relevant to the implementation.

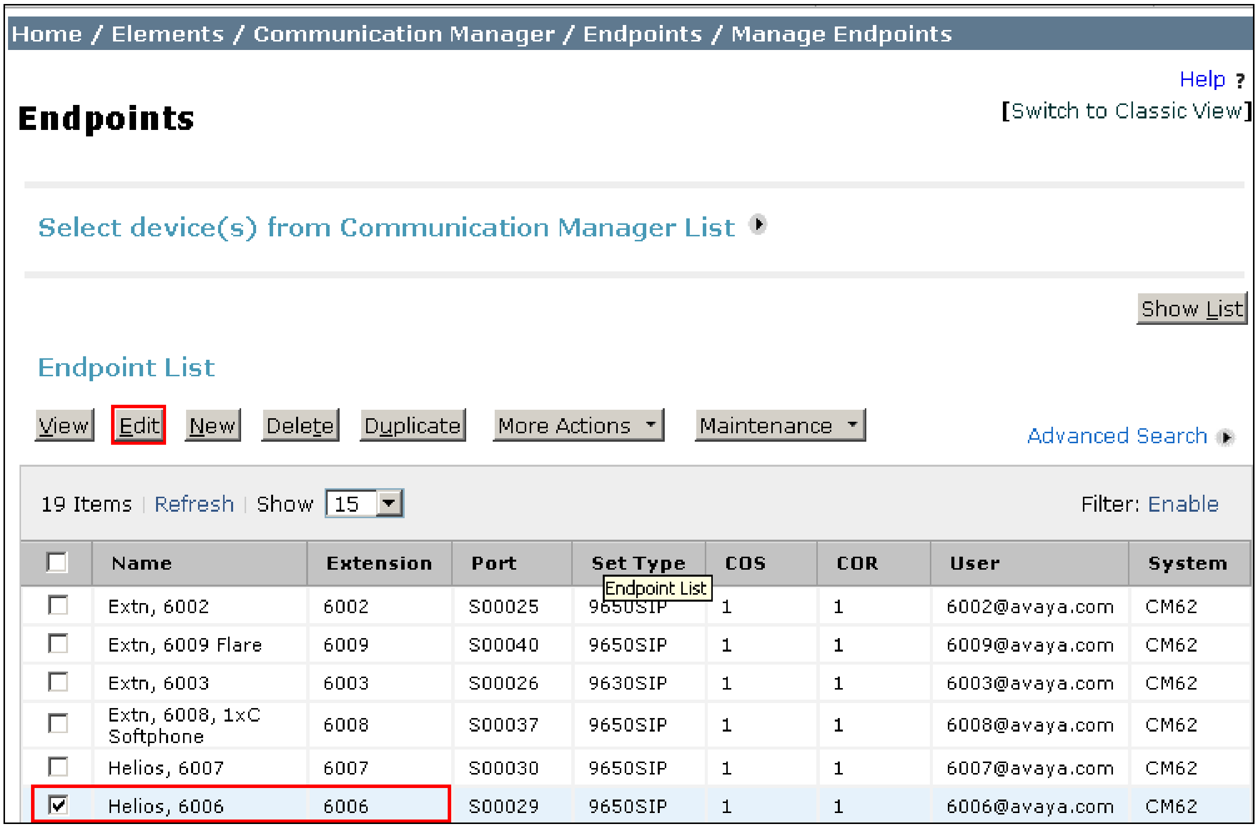


Scroll down to the page to the **CM Endpoint Profile** section. Select the Communication Manager instance from the **System** drop down box, select **Endpoint** as the **Profile Type**, enter the **Extension** number to be used, select **DEFAULT_9650SIP_CM_6_2** as the **Template** and ensure **Port** is set to **IP**; click **Commit** (not shown) when done. Repeat this for every SIP extension required.



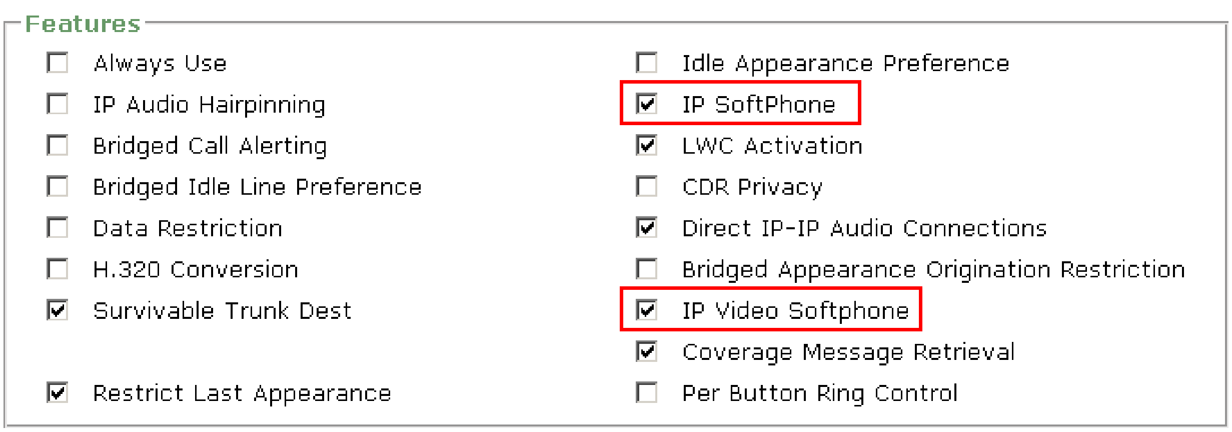
5.4. Configure Endpoints for IP Video

In order to use IP video the corresponding endpoints must be configured with the appropriate feature. From the System Manager web interface (not shown) navigate to **Communication Manager → Endpoints → Manage Endpoints** and select the box next to the endpoint on which IP Video is required and click **Edit**.



Click the **Feature** Options tab (not shown), scroll down to the **Features** section and place a tick in the **IP Video** box; click **Commit** when done.

Where administering a user for One-X Communicator, ensure that a tick is placed in **IP Softphone** and **IP Video Softphone** tick boxes as shown below.



6. Configure 2N Helios IP

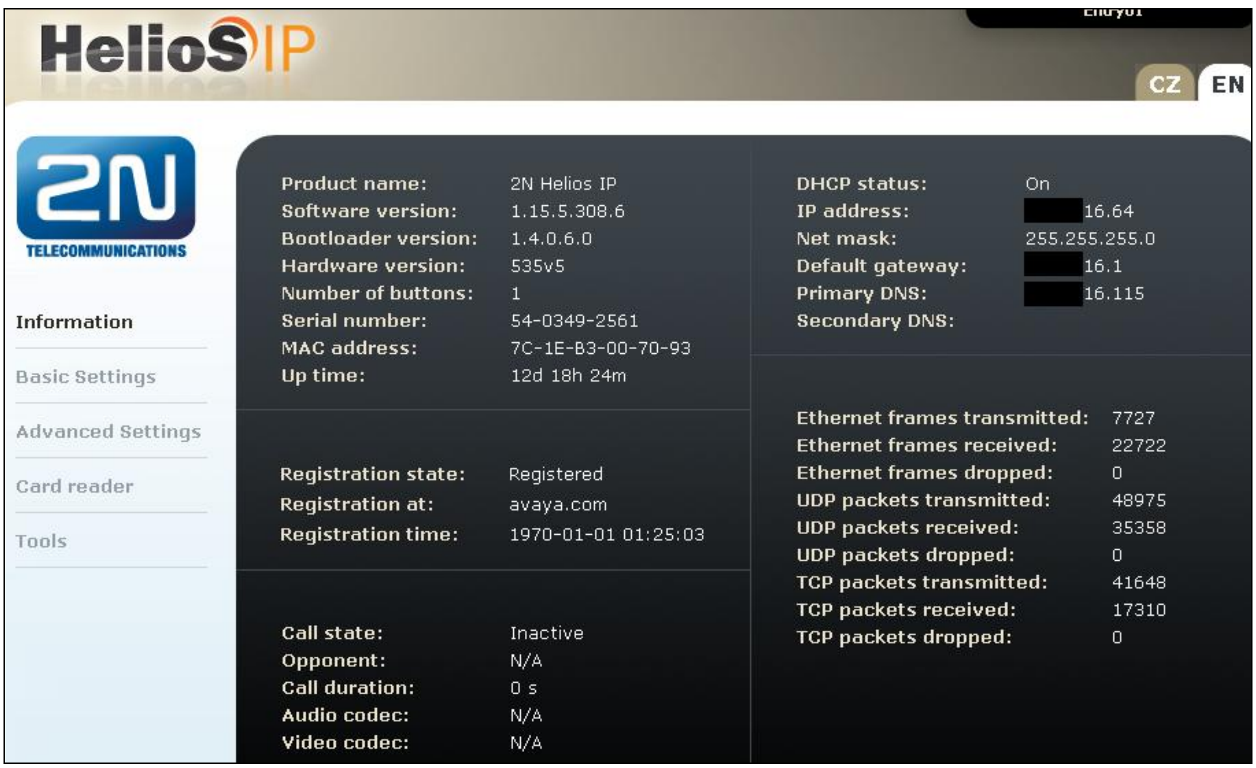
The following steps detail the configuration for Helios IP using the Web Interface. The steps include the following areas:

- Launch Web Interface
- Administer SIP Settings
- Administer Codecs
- Configure Quick Dialling Buttons
- Configure Miscellaneous Settings

The factory default setting for DHCP is on. Prior to configuration, follow the procedures in **Section 9 Reference [2]** to obtain the IP address of Helios IP**.**

6.1. Launch Web Interface

Access the Helios IP web interface, enter **http://<ipaddress>** in an Internet browser window, where **<ipaddress>** is the IP address of Helios IP. Log in with the appropriate credentials. The **Helios IP Information** screen is shown.



6.2. Administer SIP Settings

Select **Advanced Settings → SIP Settings** (not shown) from the left menu. In the **User settings** section, configure the following:

- **Display name**: Enter the desired name.
- **User ID**: Enter a user extension administered from **Section 5.3**.
- **Domain**: Enter the SIP Domain of the Session Manager.
- **Use auth ID**: Select **Yes**.
- **Auth ID**: Enter a user extension administered from **Section 5.3**.
- **Password**: Enter the **Communication Profile Password** from **Section 5.3**.

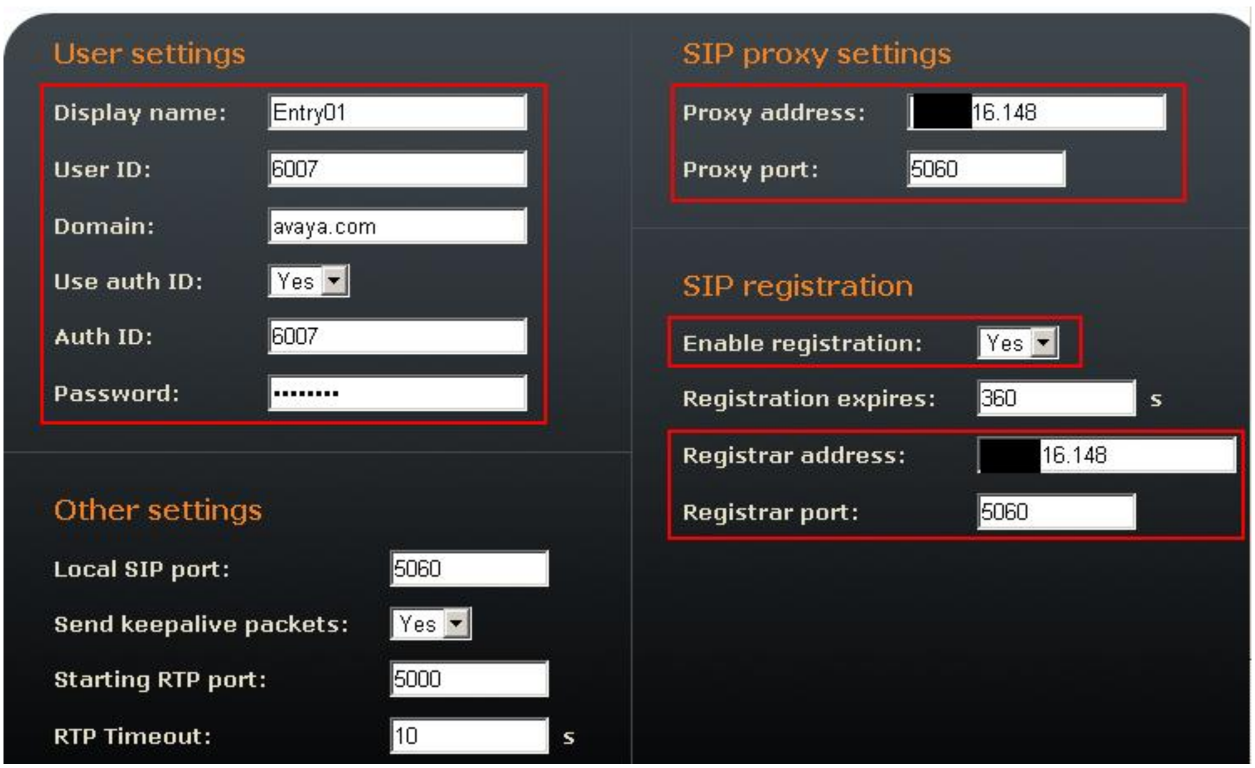
In the **SIP proxy settings** section, configure the following:

- **Proxy address**: Enter the IP address of Session Manager.
- **Proxy port**: Enter 5060 (default).

In the **SIP registration** section, configure the following:

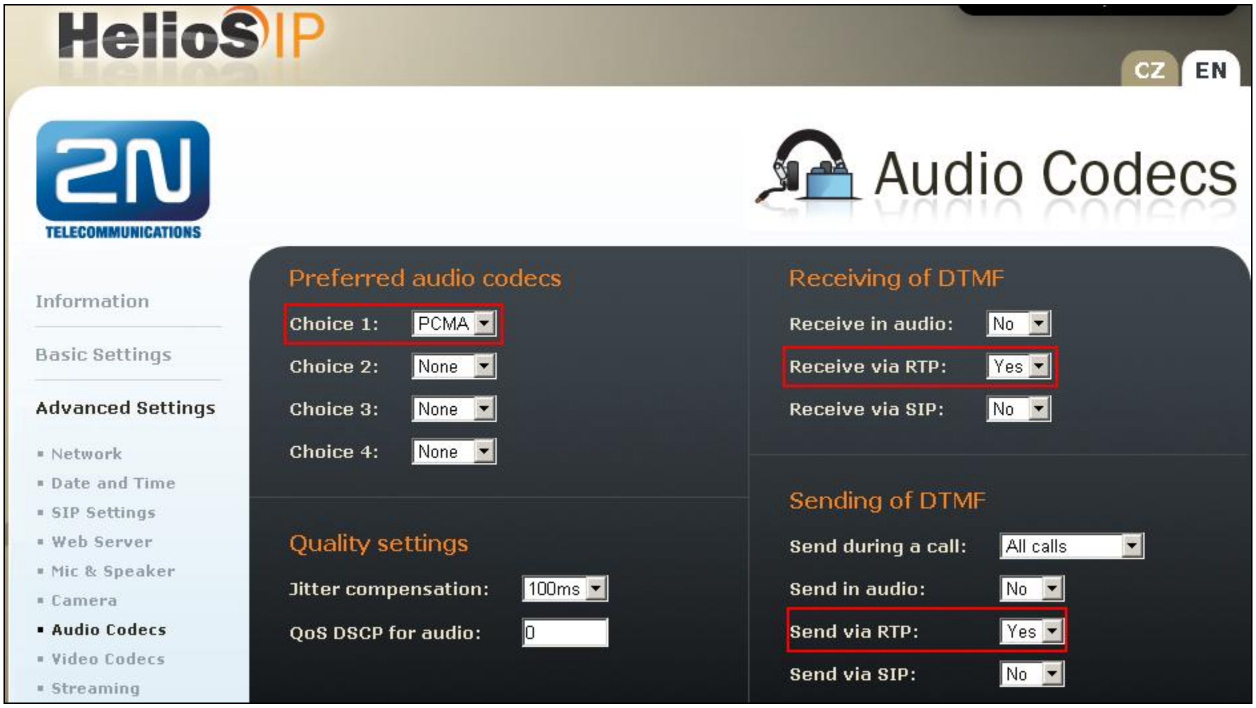
- **Register Helios IP**: Select **Yes**.
- **Registrar address**: Enter the IP address of Session Manager.
- **Registrar port**: Enter 5060 (default).

Retain the default values for the remaining fields. Click the disk icon (not shown) to save when done.

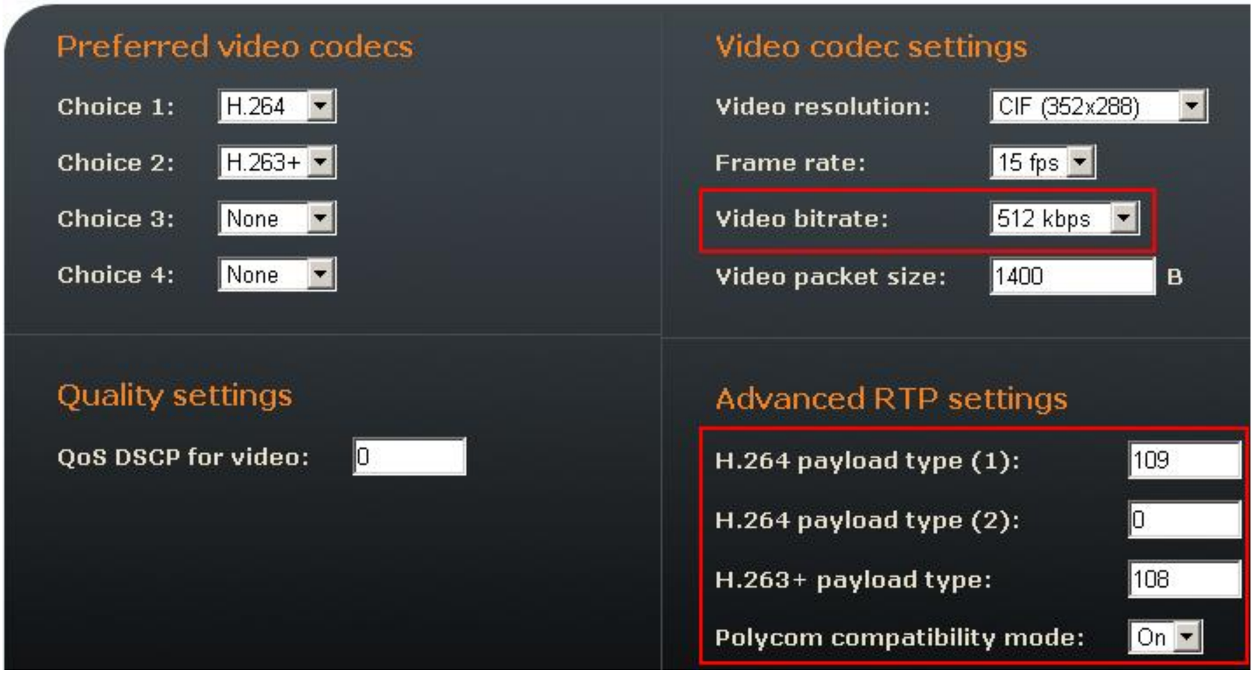


6.3. Administer Codecs

Click **Advanced Settings → Audio Codecs** from the left menu to configure the audio codecs. In the **Preferred audio codecs** section, enable and prioritize the codecs as per requirement. To enable **DTMF** using RFC2833, set **Receive via RTP** and **Send via RTP** to **Yes**. Click the save icon (not shown) when done.



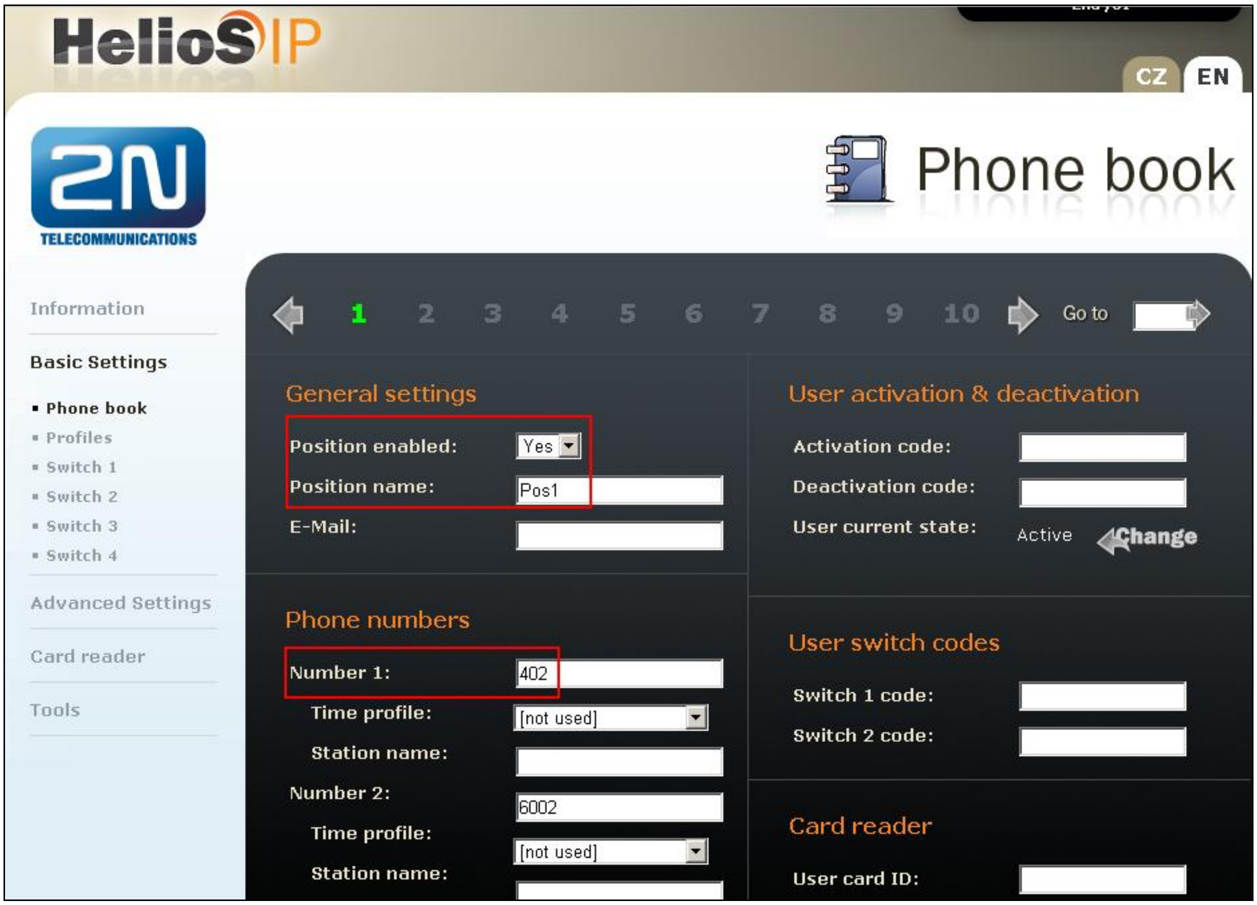
Select **Advanced Settings → Video Codecs** from the left menu (shown above) to configure the video codecs. In the **Preferred video codecs** section, enable and prioritize the codecs as per requirement. Ensure the **Video Bitrate** is less than or equal to the **Maximum Call Rate for Direct-IP Multimedia** configured in **Section 5.2**, set the **H.264 payload type (1)**, **H.264 payload type (2)** and **H.263+ payload type** as shown. Ensure **Polycom compatibility mode** is set to **On**.



6.4. Configure Quick Dialling Buttons

Select **Basic Settings → Phone book** from the left menu and select one of the positions (e.g. 1 to 10 as shown below) to configure it. The position number corresponds to the Quick Dialling Button on the Helios IP. For example, the following shows the configuration for Position 1.

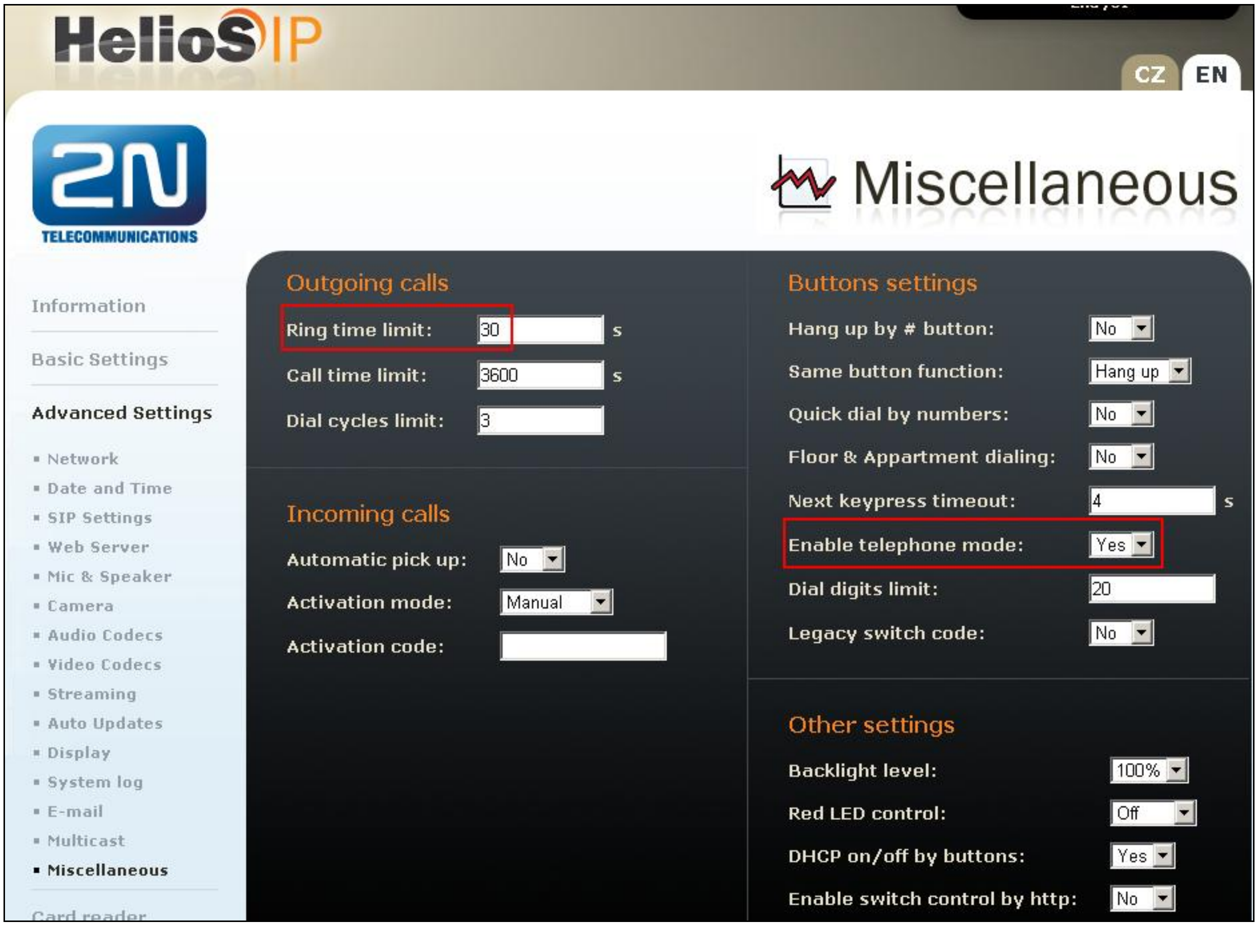
- **Position enabled**: Select **Yes**.
- **Position name**: Enter a descriptive name.
- **Number 1**: Enter the number to call when the button is pressed, for example a virtual station routing to a cover path and cover answer group.



For the description and usage of all other fields on the above page, e.g. door-lock codes, activation codes, refer to **Section 9 Reference [2]**.

6.5. Configure Miscellaneous Settings

Select **Advanced Settings → Miscellaneous** from the left menu. To allow Helios IP to ring all the users in a cover path, configure the **Ring time limit** with a value that is equal to or greater than the **No Answer Time (secs)** value multipled by the number of users in a cover path. Optionally, set **Enable telephone mode** to **Yes** to allow Helios IP to call any number using the keypad.



7. Verification Steps

This section provides the tests that can be performed to verify correct configuration of Session Manager and Helios IP.

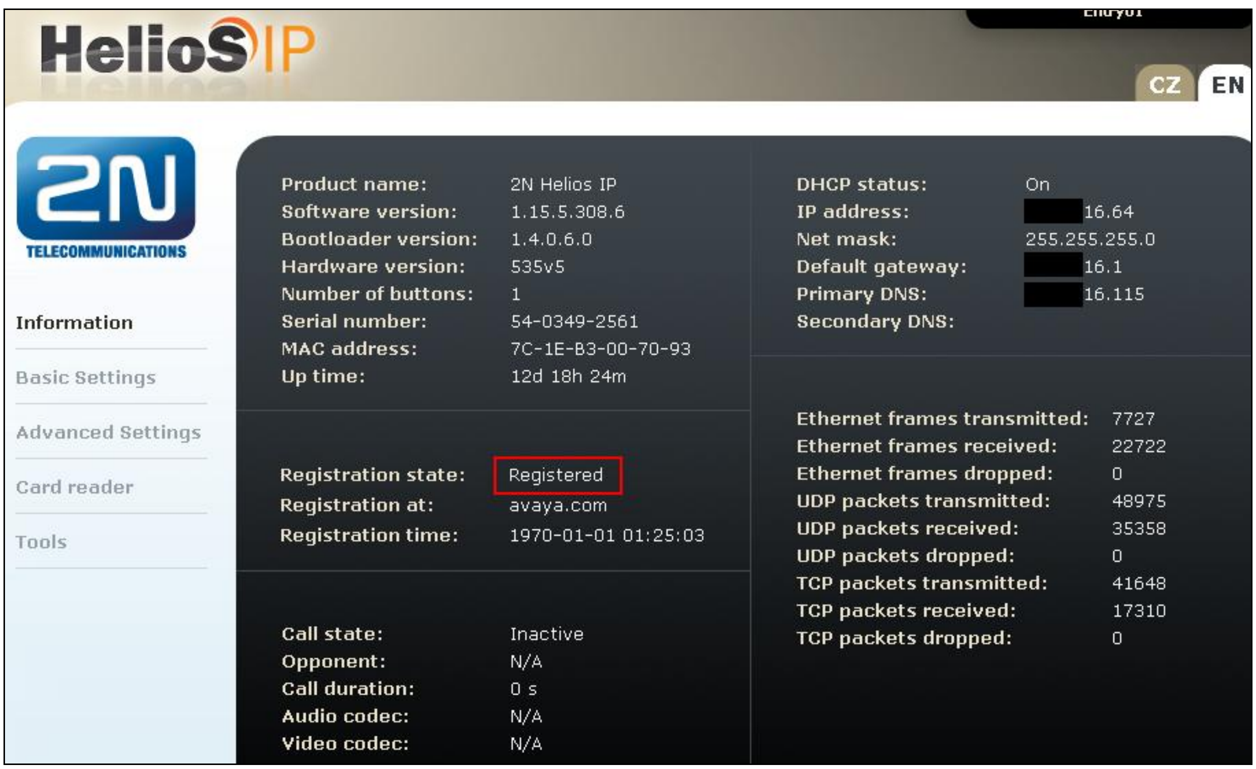
7.1. Verify Session Manager SIP Registration

From the System Manager web interface click **Session Manager → System Status → User Registrations.** Verify that Helios IP endpoints are successfully registered as shown below.

☐	Details	Address	Login Name	First Name	Last Name	Location	IP Address	AST Device	Registered		
									Prim	Sec	Surv
☐	▶ Show	6007@avaya.com	6007@avaya.com	6007	Helios	DevConnectLab	10.10.16.64:5060	☐	☑ (AC)	☐	☐

7.2. Verify 2N Helios IP

From the Helios IP web interface, select **Information** from the left menu. Verify that the **Registration state** shows **Registered**. Place a call to another phone to verify basic call operation.



8. Conclusion

These Application Notes describe the configuration steps required for configuring 2N Telekomunikace Helios IP to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. All feature and serviceability tests were completed successfully with observations made in **Section 2.2**.

9. Additional References

This section references the Avaya and 2N product documentation that are relevant to these Application Notes.

The following Avaya product documentation can be found at http://support.avaya.com.

[1] Administering Avaya Aura® Communication Manager, Doc ID 03-300509, Release 6.2

The 2N Helios IP documentation can be found at http://www.2n.cz/en/products/communicators/doors/helios-ip/downloads/.

[2] *2N® Helios IP Configuration manual version 1.15*, March 2012.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.