Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office 500 v2 R9.0 to interoperate with Comdasys Mobile Convergence Solution – Issue 1.0

Abstract

These Application Notes describe the steps to configure SIP trunking between the Comdasys Mobile Convergence Solution and Avaya IP Office 500 v2. The Comdasys Mobile Convergence Solution allows GSM telephones to connect to an Avaya IP Office using wireless networking or data over the cellular network e.g. 3G.

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

The Comdasys Mobile Convergence Solution together with Avaya IP Office 500 v2 allows "dual mode" mobile endpoints to act as local Avaya IP Office extensions. In addition to a GSM interface, such endpoints have a wireless LAN interface and a SIP client. When used within the coverage range of the local wireless LAN, incoming and outgoing calls for these endpoints are made via the mobile endpoint wireless LAN interface. When outside this coverage area, incoming and outgoing calls are made via the GSM network. When mobile endpoints enter or exit the wireless LAN coverage area, calls are "handed over" between the GSM and wireless LAN networks. The Comdasys Mobile Convergence Client needs to be installed on the mobile phone. Placing phone calls and feature invocation are executed transparently for the end-user either in the WIFI or GSM mode.

SIP is a standards-based communications approach designed to provide a common framework to support multimedia communication. In the configuration described in these Application Notes, SIP is used as the signaling protocol between the Avaya components and the Comdasys Mobile Convergence Solution. SIP manages the establishment and termination of connections and the transfer of related information such as the desired codec, calling party identity, etc.

2. General Test Approach and Test Results

The interoperability compliance testing evaluated the ability of the Mobile Convergence Solution to carry out endpoint registration, call routing and call handover. Call handling, feature access and voice quality was performed from the Mobile Convergence Client on the mobile endpoint.

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 following tests were performed as part of the compliance testing. When appropriate, the tests were covered for calls established via the wireless LAN (WLAN) interface and the GSM interface of the client endpoints involved.

- Outgoing/incoming local/cellular call
- Outgoing/incoming local/cellular call rejection
- Outgoing/incoming local/cellular call cancellation
- Call forwarding
- Supervised/blind transfer
- Consultation
- Hold/retrieve
- Manual handover to WLAN
- Automatic handover from GSM/WLAN

- Interruption to Comdasys server LAN interface
- Interruption to Comdasys server power

2.2. Test Results

All functionality and serviceability test cases were completed successfully.

2.3. Support

Support is available via the Comdasys distributor network.

3. Reference Configuration

Figure 1 illustrates the network topology used during compliance testing. The Avaya solution consists of an Avaya IP Office 500 v2 and Avaya Digital endpoints. The Comdasys Mobile Convergence Solution has a SIP trunk to Avaya IP Office for the through-call that is placed to/from the controller when the endpoint is in GSM mode, and also registers as 3rd party SIP extensions to Avaya IP Office. A WIFI network is connected to the IP Office LAN and a GSM network is available.

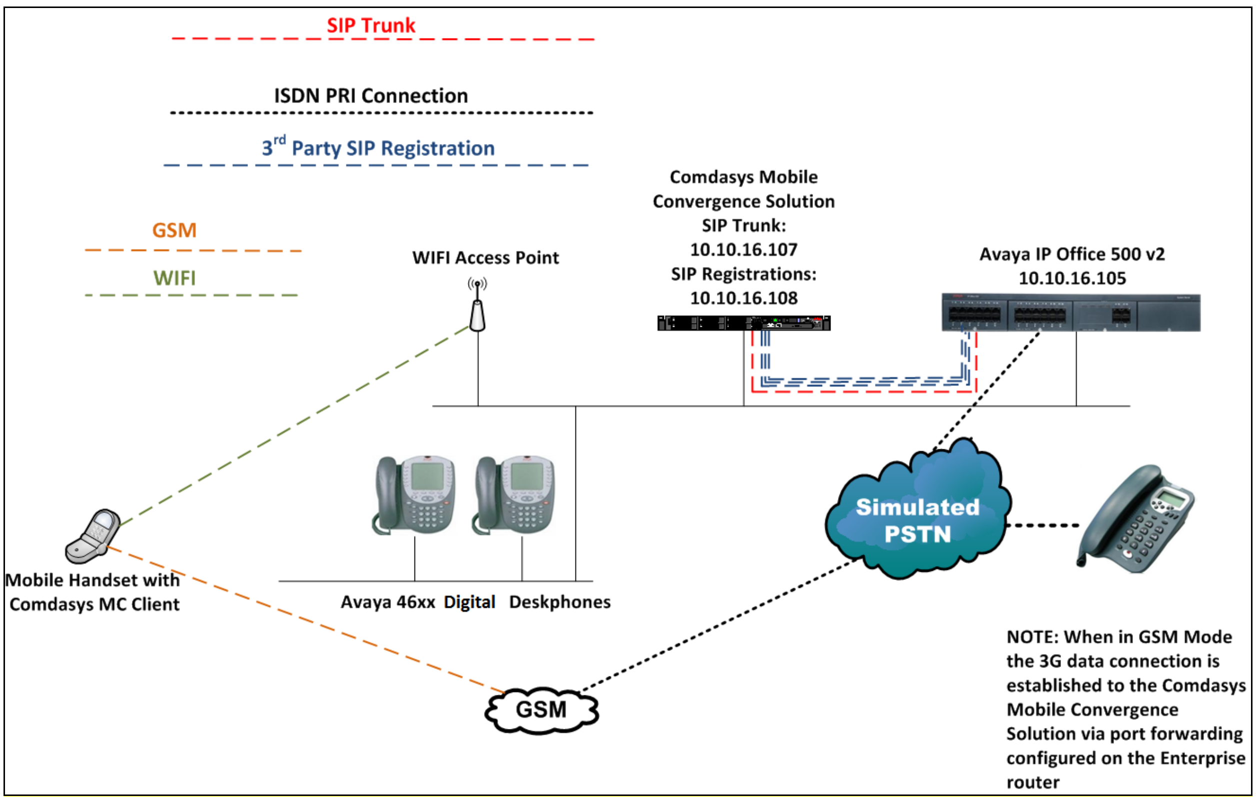


Figure 1: Avaya IP Office 500 v2 with Comdasys Mobile Convergence Solution Configuration

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya IP Office 500 V2	9.0 Build 829
Avaya IP Office Manager	9.0
Avaya 46xx Digital	6.0
Comdasys Mobile Convergence Controller	10684.18.3
Blackberry Curve 8900	V5.0.0.681 Comdasys MC Client 4.1 Build:#2220M
Apple iPhone	iOS 6.0 Comdasys iMC Client v4.1.1 (3260)

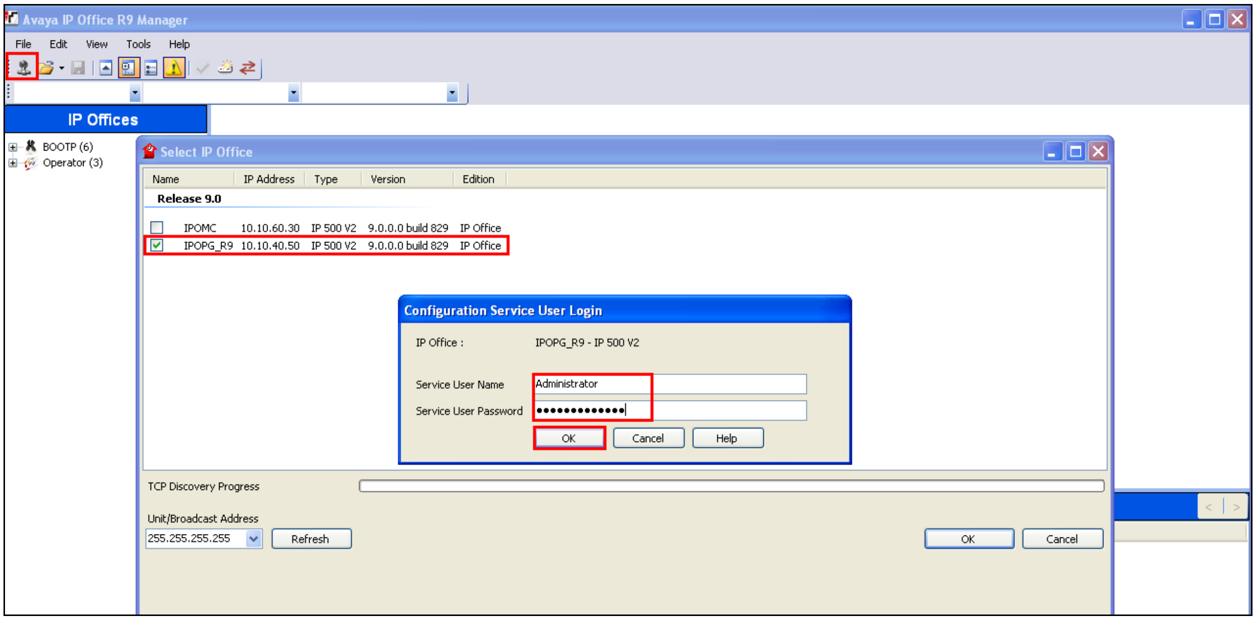
Testing was performed with IP Office 500 v2 R9.0, but it also applies to IP Office Server Edition R9.0. Note that IP Office Server Edition requires an Expansion IP Office 500 v2 R9.0 to support analog or digital endpoints or trunks.

The telephone numbers used for testing are shown in the following table.

Endpoint	Ext	PSTN Number	Station Type
Extn1	201		Digital Endpoint on IPO
Extn2	202		Digital Endpoint on IPO
FMC1	230	00353867818308	SIP Endpoint on IPO associated with Blackberry 8900 with MC Client
FMC2	231	No PSTN number	SIP Endpoint on IPO associated with Apple iPhone with iMC Client
Call through	n/a	0035391482464	Service Access Number on Mobile Convergence Solution

5. Configure Avaya IP Office 500 v2

Avaya IP Office is configured through the Avaya IP Office Manager PC application. From the Avaya IP Office Manager PC, select **Start → Programs → IP Office → Manager** to launch the application. A screen that includes the following is displayed, select the IP Office to be configured and then enter the username and password and click **OK** to continue.



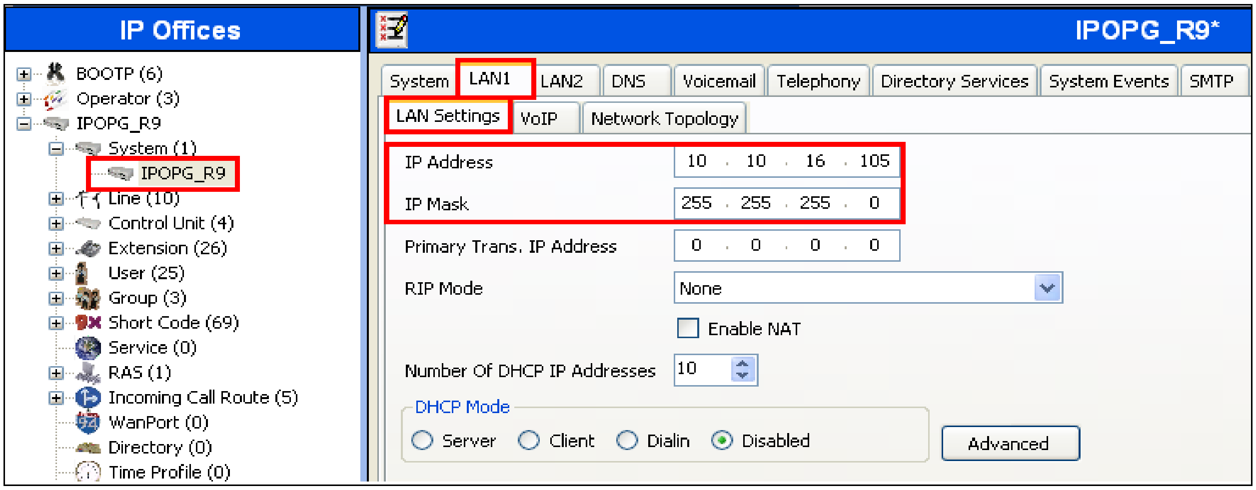
If the above screen does not appear, the configuration may be alternatively opened by navigating to **File → Open Configuration** at the top of the Avaya IP Office Manager window. Select the proper Avaya IP Office system from the pop-up window and log in with the appropriate credentials.

The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this document, the **View** menu was configured to show the Navigation pane on the left side, omit the Group pane in the center, and show the **Details** pane on the right side. Since the Group Pane has been omitted, its content is shown as submenus in the Navigation pane. These panes (Navigation, Group and Details) will be referenced throughout the Avaya IP Office configuration. All licensing and feature configuration that is not directly related to the interface with the Comdasys Mobile Convergence Solution (such as twinning and IP Office Video Softphone support) is assumed to already be in place.

5.1. Configure System Settings

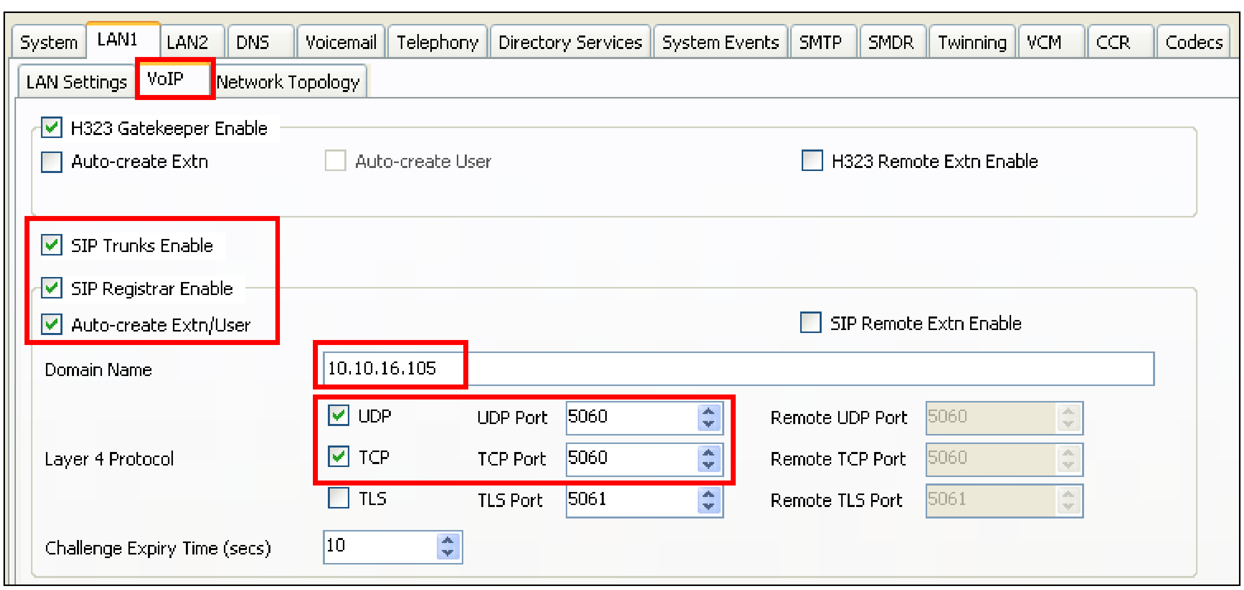
Select **System** from the hierarchy in the left window pane and click the **LAN1** tab. Under the **LAN Settings** sub tab configure the following:

- **IP Address** – configure the IP address for the IP Office
- **IP Mask** – configure the corresponding subnet mask



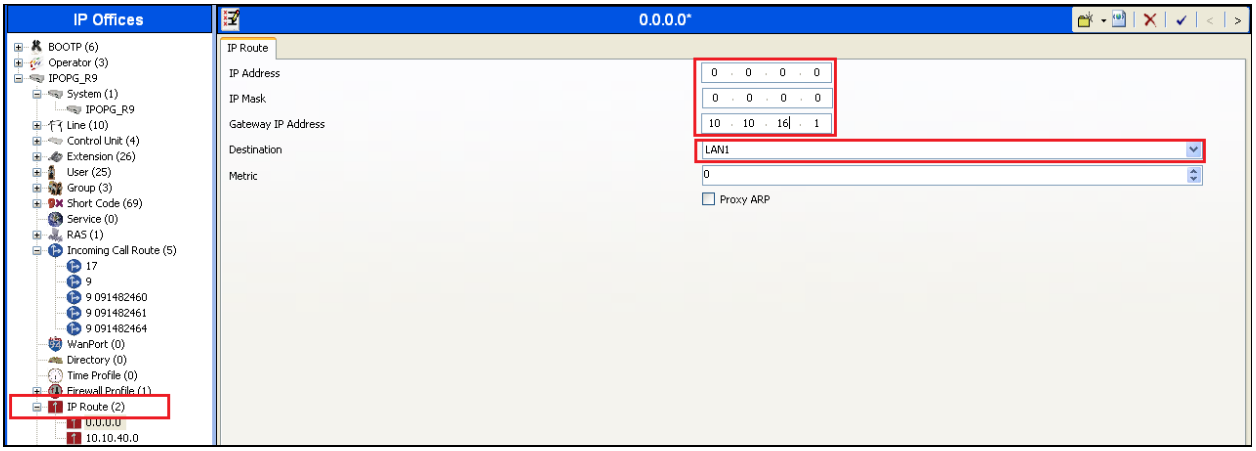
Click the **VoIP** sub tab, and enable the following:

- **SIP Trunks (enable)**
- **SIP Registrar (enable)**
- **Auto-create Extn/User (enable)**
- **Domain Name** – in this case the IP address of IP Office was used
- **UDP Port (enable)** – ensure that **UDP Port** is set to **5060**
- **TCP Port (enable)** – ensure that **TCP Port** is set to **5060**



5.2. Configure Default Route

A default route must be configured for the IP network routing. Click **IP Route** from the left navigation pane and enter the default route information accordingly. In this instance the **Gateway IP Address** is **10.10.16.1** and the **Destination** is **LAN1** which is the IP Office network interface connected to the local network.

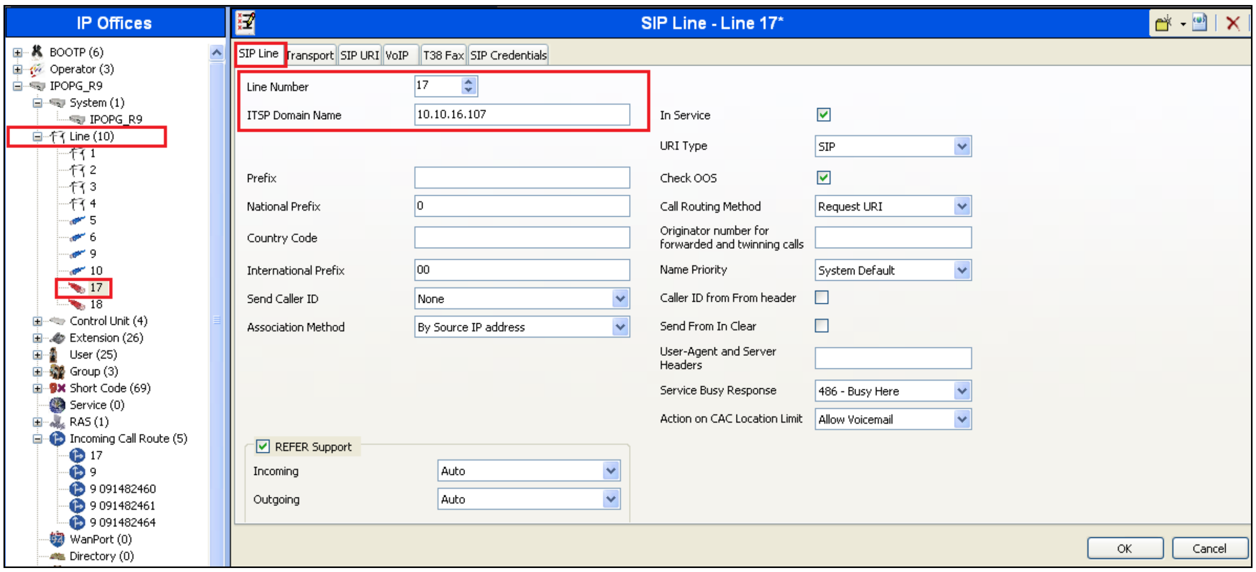


5.3. Create SIP Trunk to Comdasys Mobile Convergence Solution

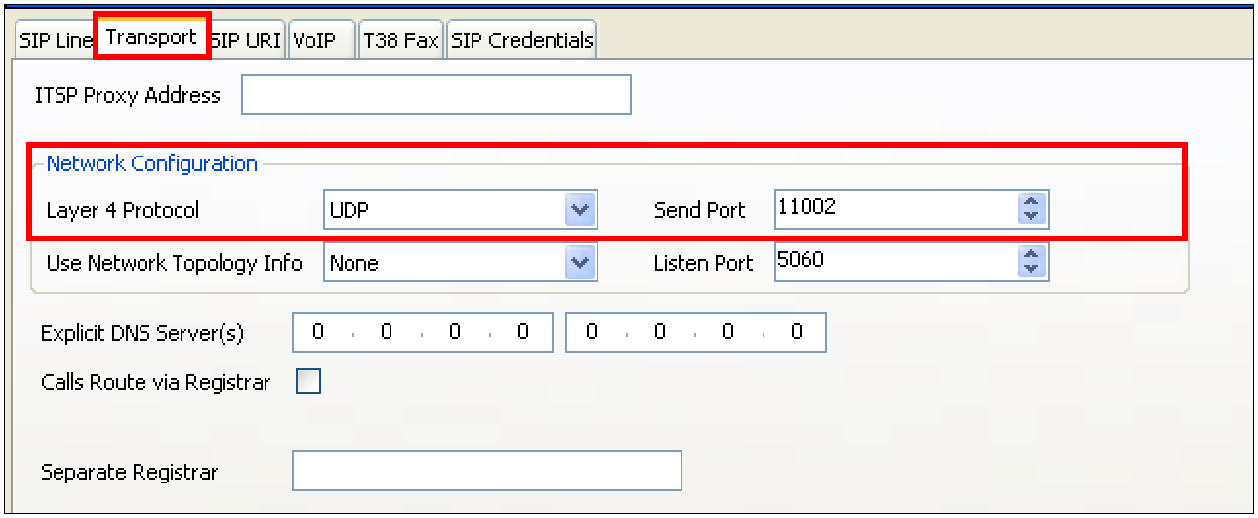
A SIP trunk must be configured between IP Office and the Mobile Convergence Solution for routing calls when the MC endpoints are in GSM mode. Right click on **Line** in the left navigation pane and click **New → SIP Line** (not shown), complete the configuration as follows:

- **Line Number** – configure an appropriate number, this will auto-populate
- **ITSP Domain Name** – enter the IP address of the Mobile Convergence Solution LAN1 interface

Leave other fields as default.



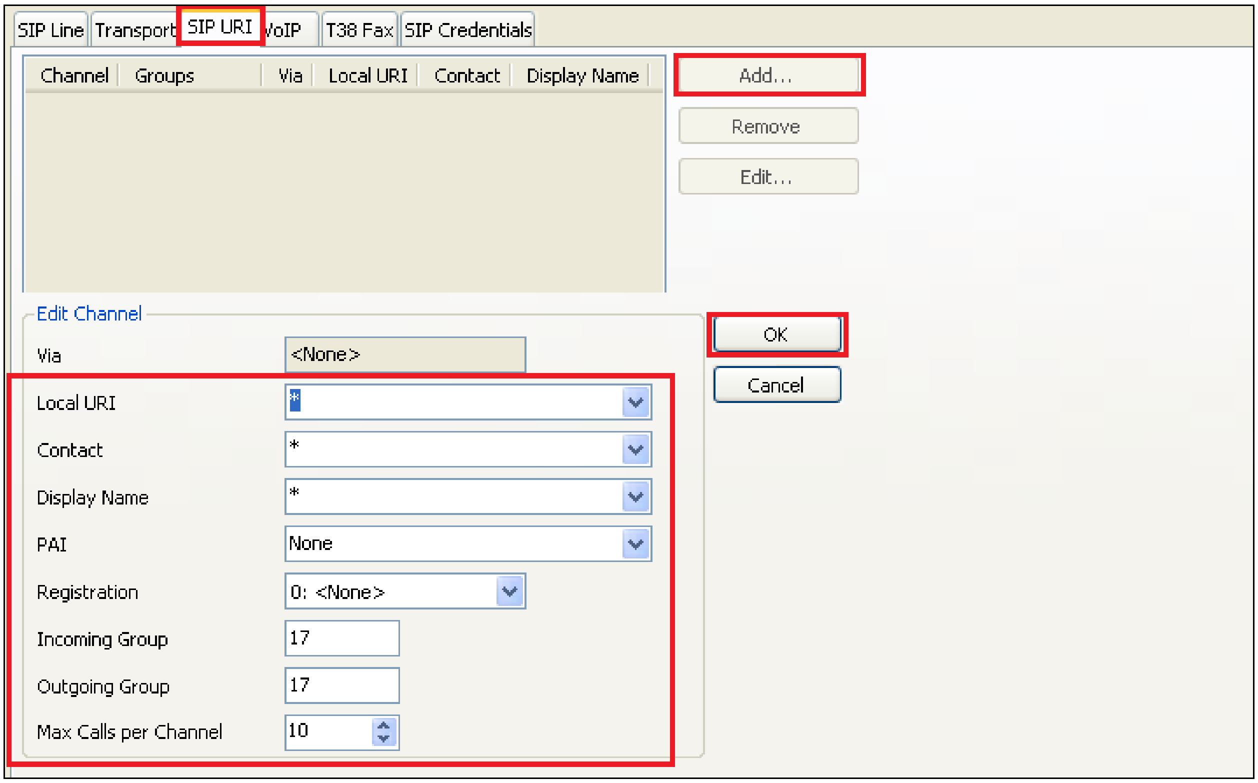
Click on the **Transport** tab and ensure that **UDP** is selected for the **Layer 4 Protocol**. Set the **Send Port** to **11002** and the **Listen Port** to **5060**. Note this information will be used again in **Section 6.6**.



Click the **SIP URI** tab and click **Add**, configure as follows:

- **Local URI** – set to *
- **Contact** – set to *
- **Display Name** – set to *
- **Incoming Group** – set to the **Line Number** configured in the previous screen
- **Outgoing Group** – set to the **Line Number** configured in the previous screen

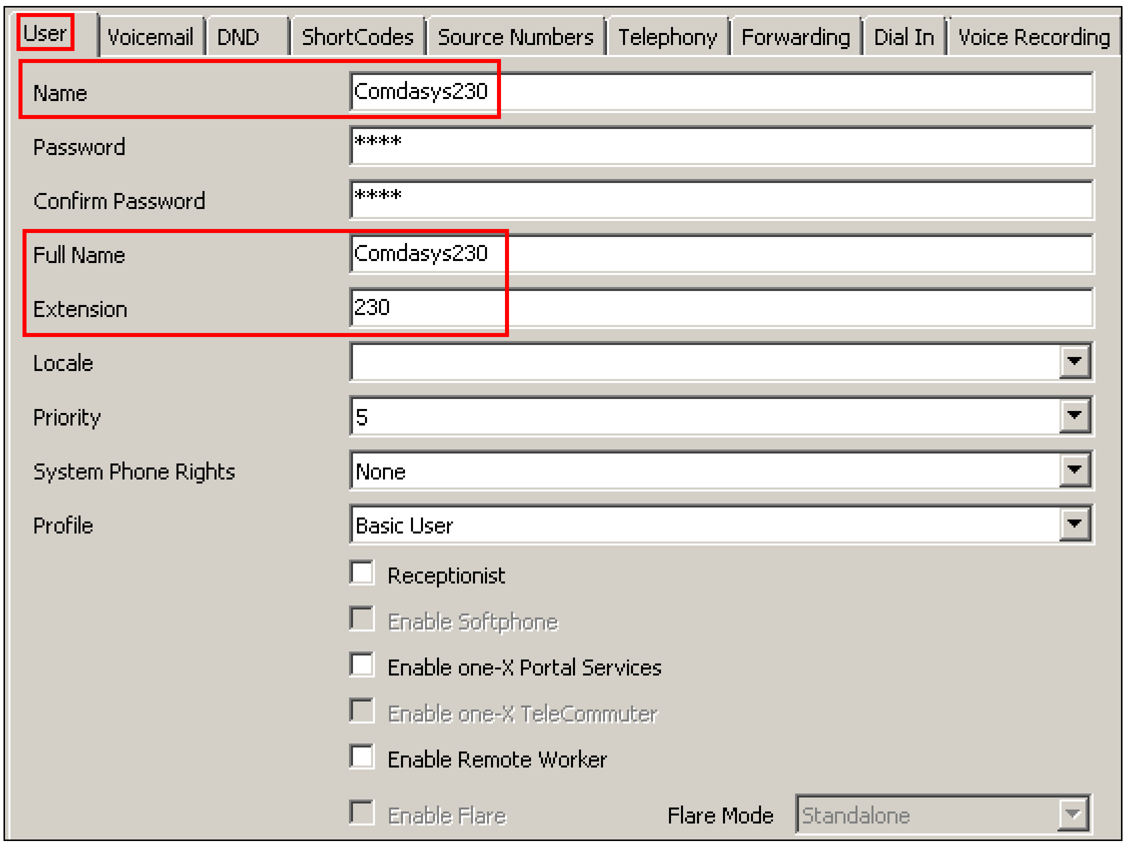
Click **OK** when done.



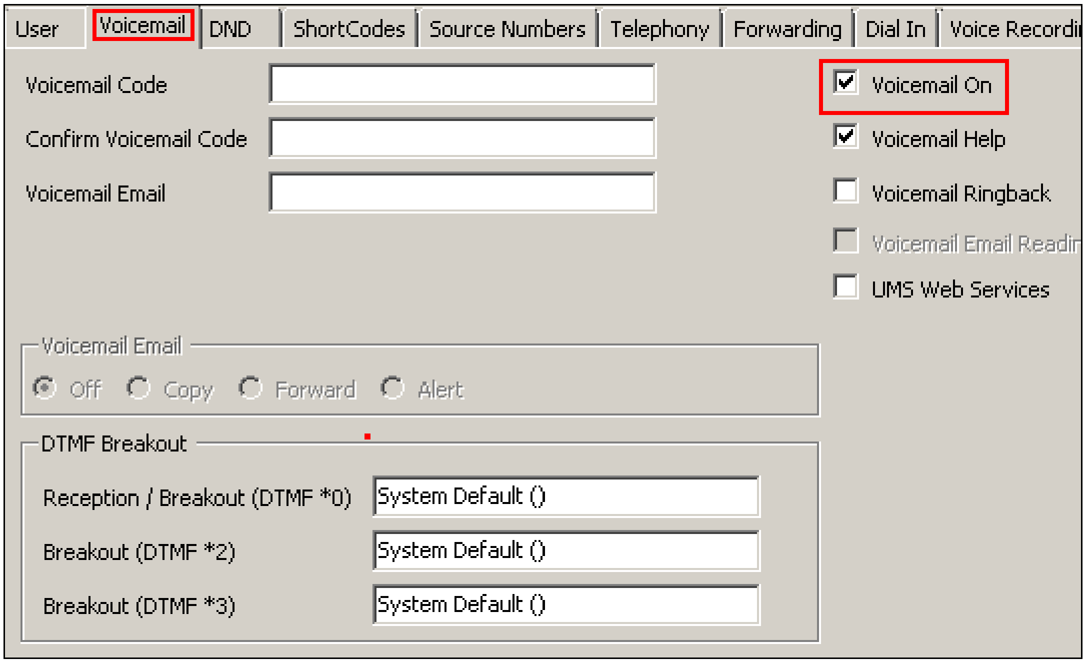
5.4. Configure SIP Endpoint

SIP users must be configured on IP Office which the Mobile Convergence Solution will use to register with IP Office. Right click **User** in the left navigation pane and click **New.** Configure as follows:

- **Name** – enter an identifying name
- **Full Name** – enter an identifying name
- **Extension** – enter a valid extension number

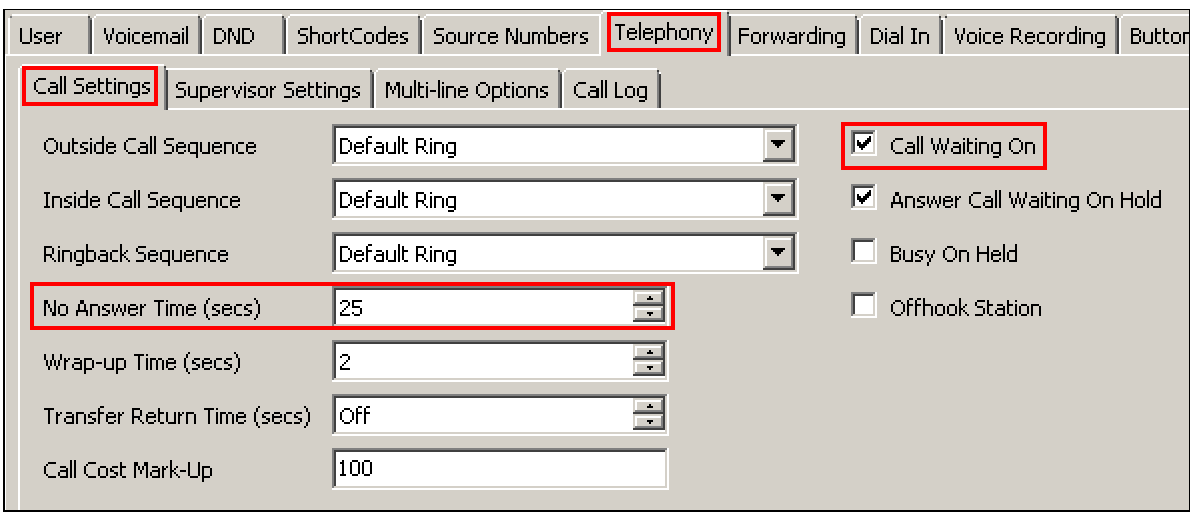


Click on the Voicemail tab and check the **Voicemail On** box to enable voicemail for this user.

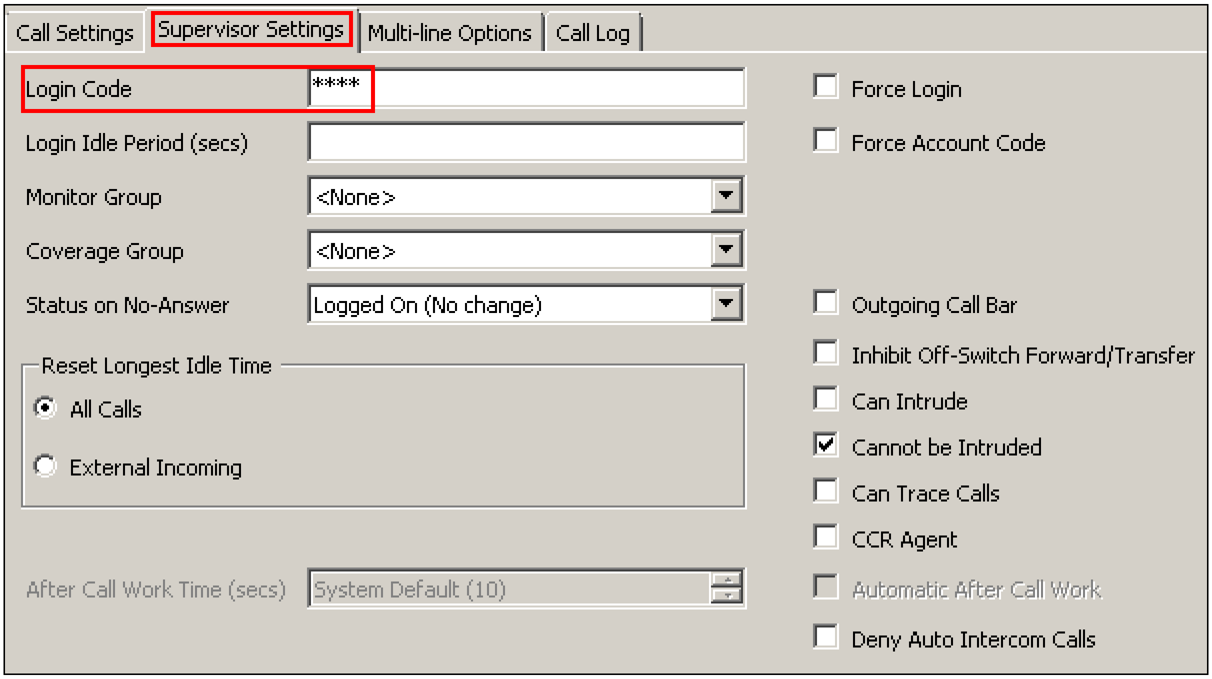


Click the **Telephony** tab and select the **Call Settings** sub tab and configure as follows:

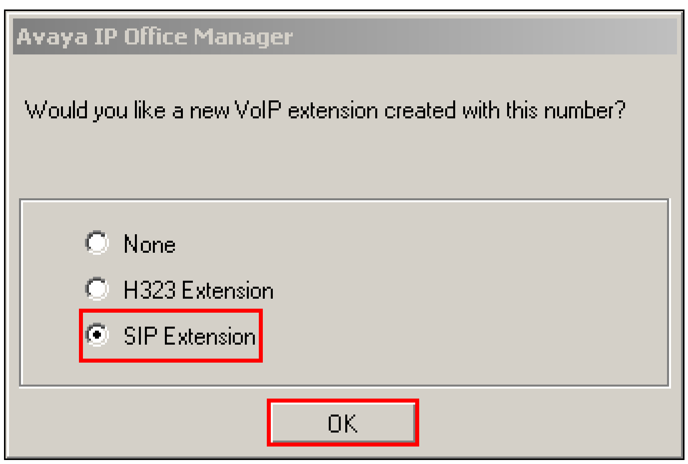
- **Call Waiting On** – check the box to enable. This is essential for support of REFER SIP messaging
- **No Answer Time (secs)** – increase the delay before forwarding on no-answer or busy or voicemail coverage. This is necessary when the MC client is GSM mode due to the delay inherent on the cellular networks



Click on the **Supervisor Settings** sub tab and configure the **Login Code**, this is used when registering the SIP endpoint.



Click **OK** (not shown) to commit and select **SIP Extension** from the screen which appears in order to create a corresponding extension for this user.

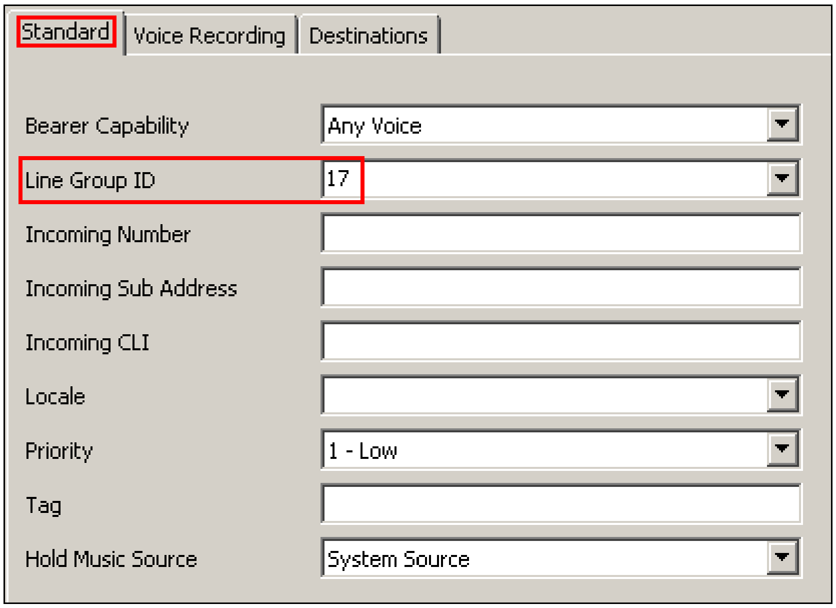


5.5. Configure Incoming Call Route from Comdasys Mobile Convergence Solution to Avaya IP Office

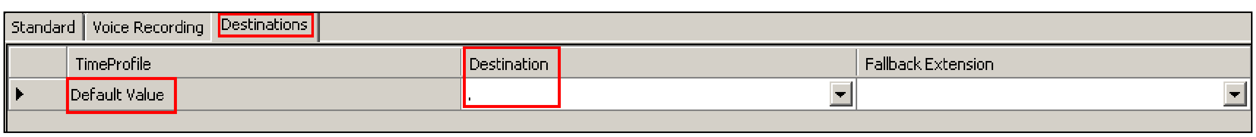
An incoming call route must be configured to route incoming calls from the Mobile Convergence Solution through IP Office. Right click **Incoming Call Route** from the left navigation pane and click **New** (not shown). Configure as follows:

- **Line Group ID** – enter the **Line Number** configured in **Section 5.3**

Leave all other settings as default.



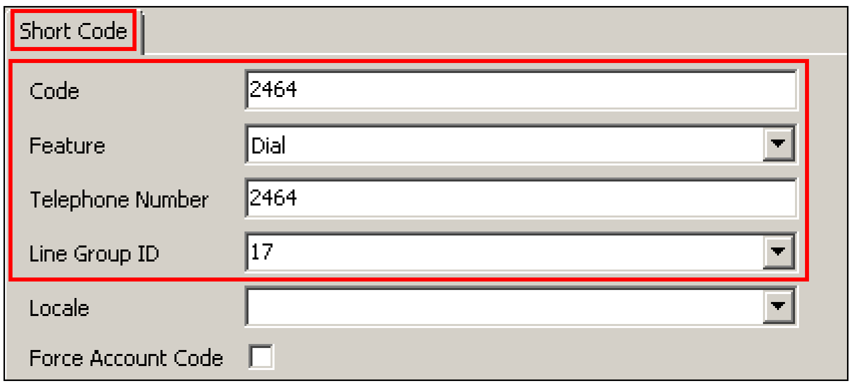
Click the **Destinations** tab and set the **Default Value** with a **Destination** of "**.**". This will route all dialed strings from the Mobile Convergence Solution with no change.



5.6. Configure Short Code for Call Through Feature

The Mobile Convergence Solution uses the call through feature when the MC clients are in GSM Mode. The MC client dials the call through number whereby the call is answered and handled by the Comdasys Mobile Convergence Solution. Right click on **Short Code** in the left navigation pane and click **New** and configure as follows:

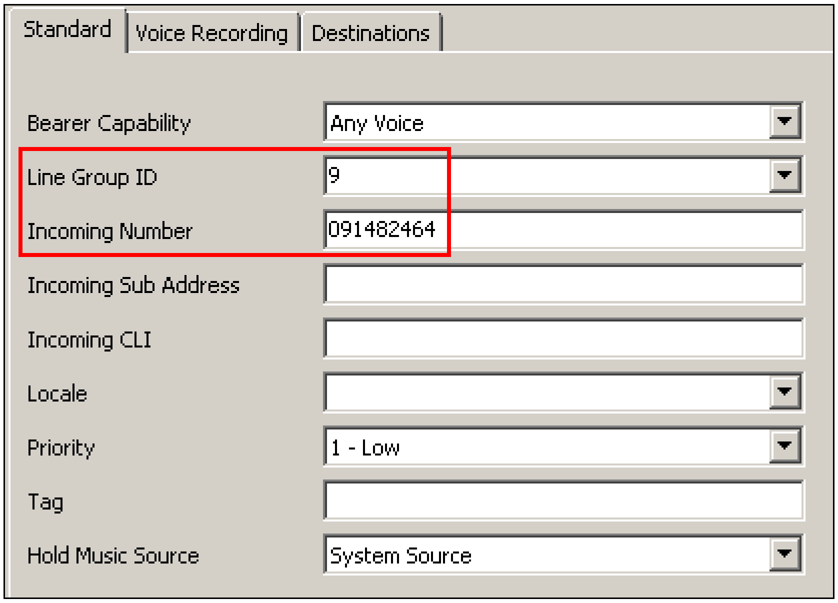
- **Code** – enter an appropriate short code
- **Feature** – select **Dial** from the drop down lists
- **Telephone Number** – configure this as the number to be presented to the Mobile Convergence Solution
- **Line Group ID** – select the **Line Number** configured in **Section 5.3**



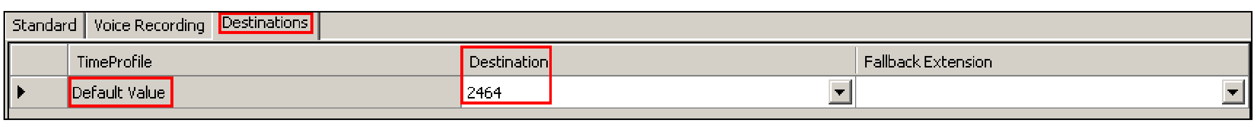
5.7. Configure Incoming Call Route for Call Through

Due to the variety of configurations possible when configuring a PSTN connection with IP Office details of the PSTN configuration are not detailed. For reference, the **Line Number** for the PSTN trunk is **9**. Right click **Incoming Call Route** from the left navigation pane and click **New** (not shown). Configure as follows:

- **Line Group ID** – enter the PSTN **Line Number** in this case **9**
- **Incoming Number** – enter the incoming number assigned to the Call Through feature



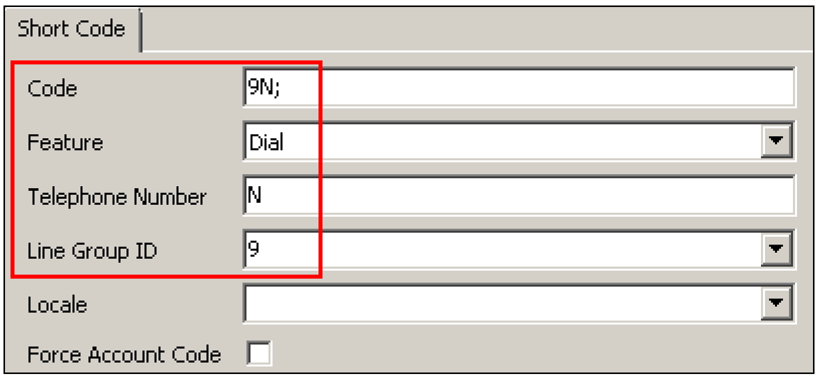
Click the **Destinations** tab and set the **Default Value** with a **Destination** of the short code configured in **Section 5.6** – this will route the incoming number to the short code destination.



5.8. Configure Short Code for PSTN Access

A short code must be configured to access the PSTN, in this case 9 is used as a prefix for all external calls. Right click on **Short Code** in the left navigation pane and click **New** and configure as follows:

- **Code** – enter an appropriate short code
- **Feature** – select **Dial** from the drop down list
- **Telephone Number** – configure **N** to define the number to send to the PSTN line.
- **Line Group ID** – select the PSTN **Line Number**, in this case **9**

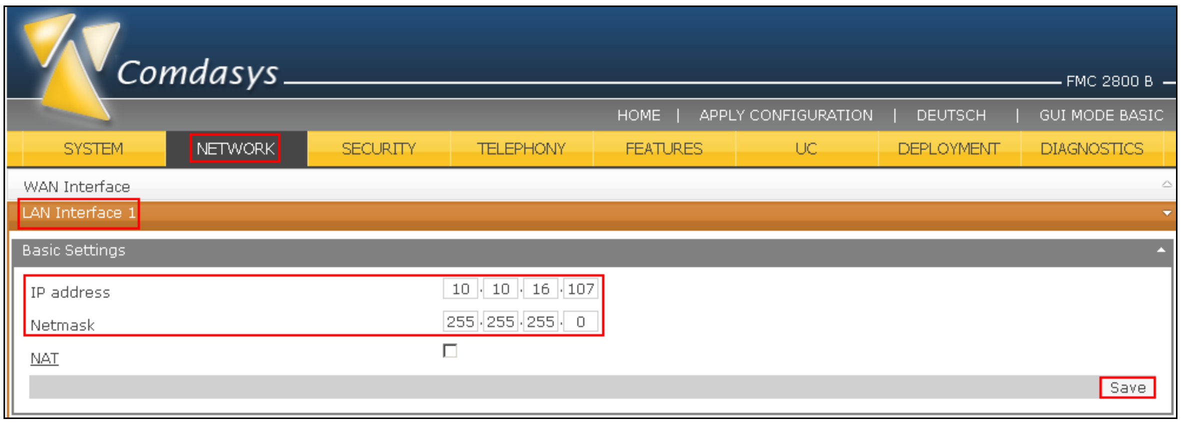


6. Configure Comdasys Mobile Convergence Solution

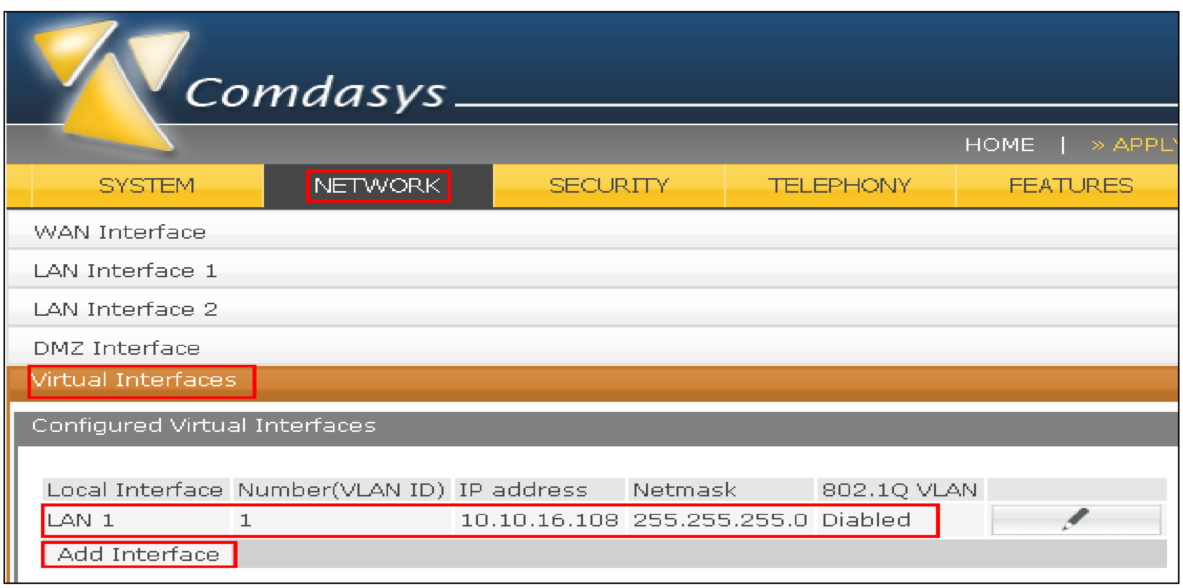
These Application Notes assume a Mobile Convergence Controller is supplied by Comdasys. All administration of the Mobile Convergence Controller is performed through its web interface. Login to the Mobile Convergence Controller web interface using its IP address, in this case https://10.10.16.107/. Enter the appropriate credentials and log on.

6.1. Administer LAN Interfaces

Two IP addresses on the LAN interface are required, one for the SIP Trunk connection to IP Office and another for the SIP user registrations. Click **NETWORK → LAN Interface 1** and enter a valid **IP address** and **Netmask**, click **Save** when complete.

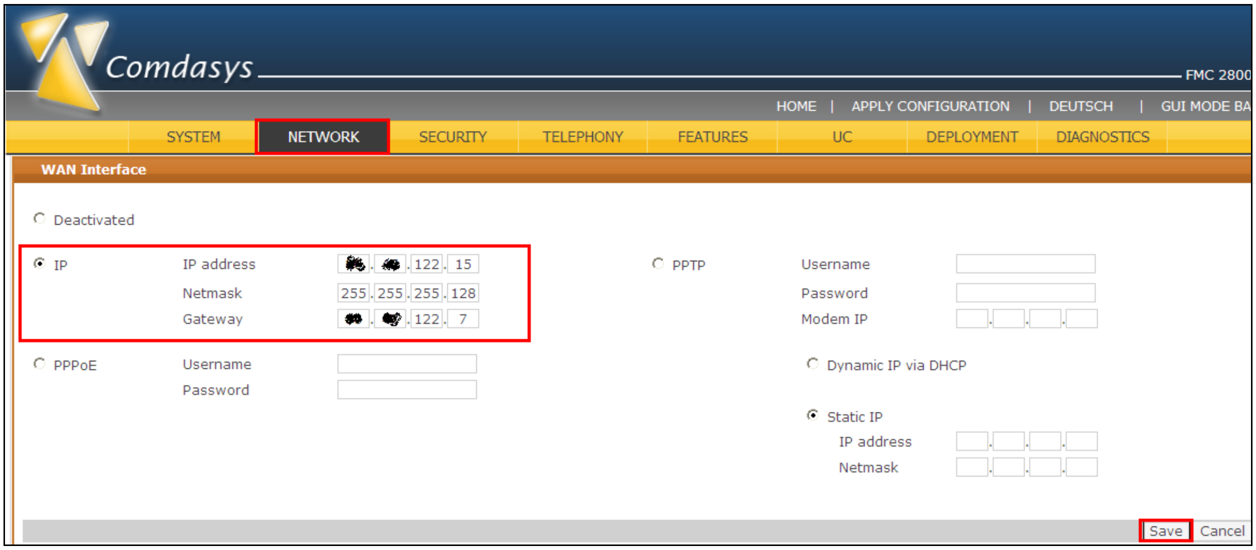


Click **NETWORK → Virtual Interfaces → Add Interface**, select **LAN1** from the drop down list, enter a valid **VLAN ID, IP address** and **Netmask** and click on  to commit (not shown). The screen below will be displayed.



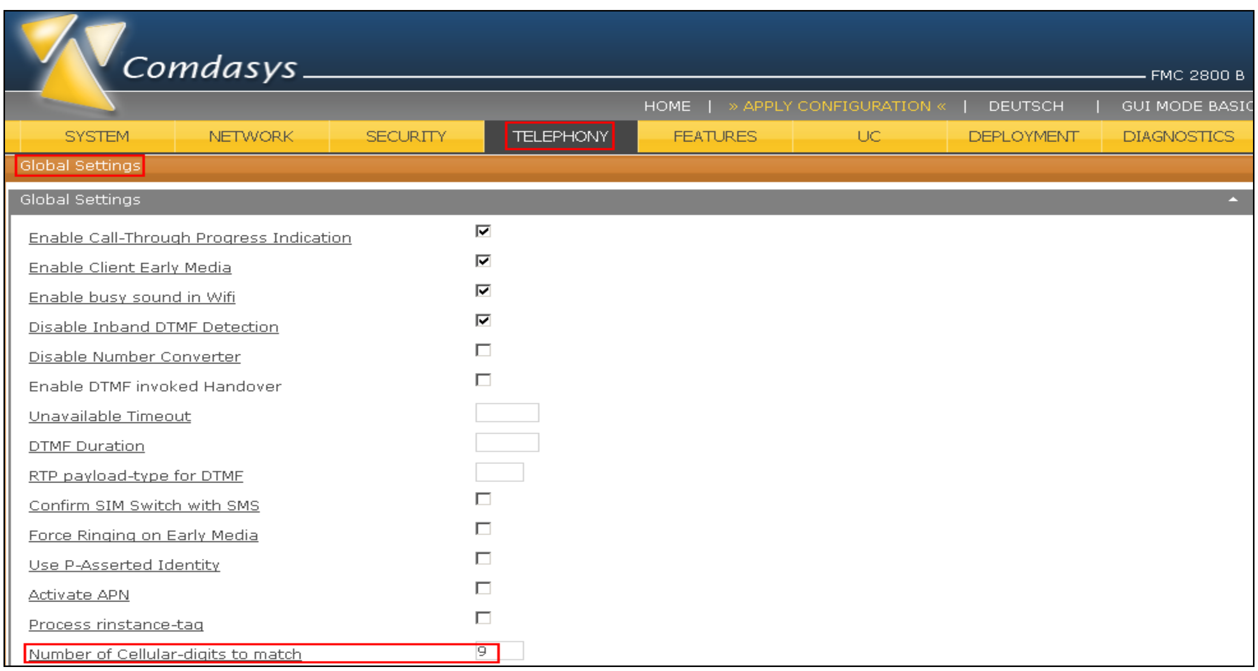
6.2. Configure WAN Interface

Click on the **NETWORK** tab to setup the WAN IP as indicated on the screenshot below. Enter a suitable external **IP Address**, **Netmask** and **Gateway** and click on **Save** once finished.



6.3. Configure Global Settings

Click **TELEPHONY → Global Settings** to setup the global settings as indicated on the screenshot below. For details explaining the options, consult the Mobile Convergence administrator's manual. Of particular importance is the **Number of Cellular-digits to match** field which is required for succesful routing of calls when the MC client is in GSM mode.

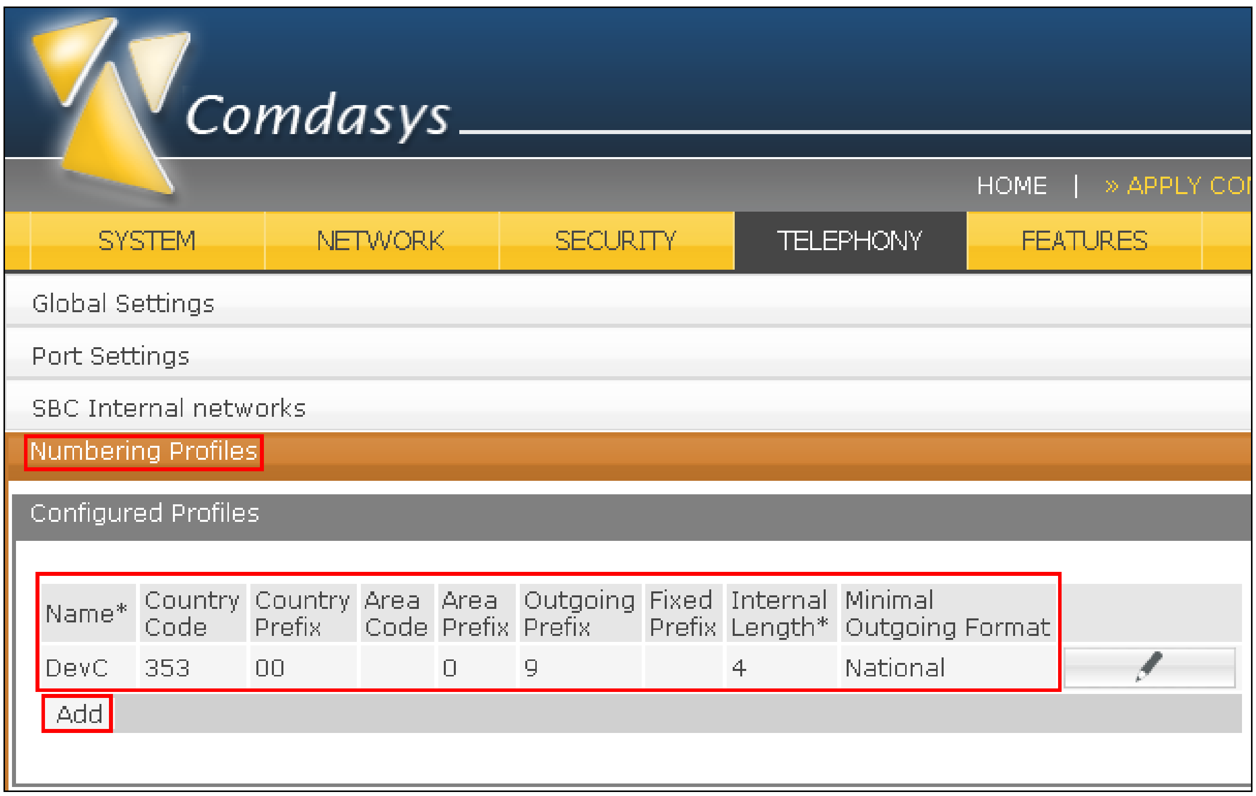


6.4. Configure Numbering Profiles

Numbering profiles are configured according to the country of implementation. Click **TELEPHONY → Numbering Profiles → Add** and enter the following:

- **Name** - to identify the location
- **Country Code** – enter the international country code
- **Country Prefix** – enter the prefix used for dialing international numbers
- **Area Prefix** – enter the prefix used for dialing the local area
- **Outgoing Prefix** – enter the prefix used to access the PSTN, in this case **9**.
- **Minimal Outgoing Format** – set to **National**

Click on  to commit (not shown).

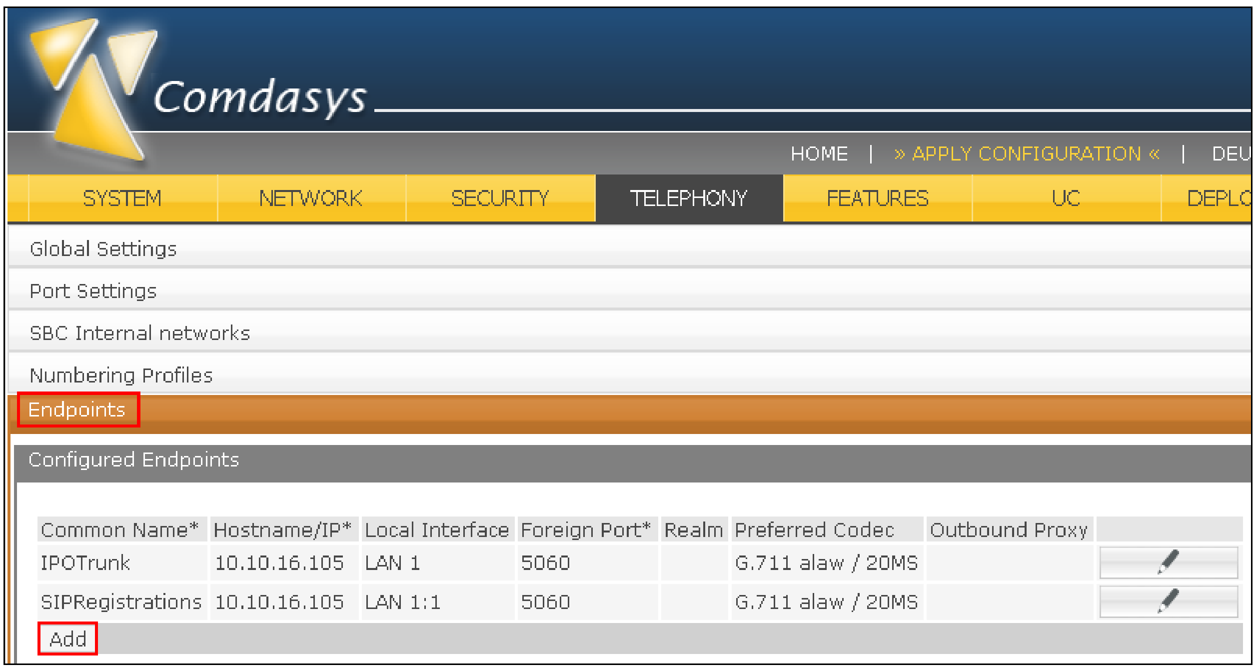


6.5. Configure Endpoints

Endpoints must be configured on the Mobile Convergence Controller. One endpoint must be configured for both the SIP trunk and the SIP user registrations. Click **TELEPHONY → Endpoints → Add** and configure as follows:

- **Common Name** – assign a name to identify this endpoint
- **Hostname/IP** – enter the IP Address of the IP Office
- **Foreign Port** – enter the port configured in **Section 5.1** under the **SIP Registrar** sub tab
- **Prefered Codec** – choose a prefered codec

For the SIP trunk the **Local Interface** must be configured as **LAN 1**, for the SIP user registrations, the **Local Interface** must be set as the virtual interface configured in **Section 6.1** in this case denoted as **LAN 1:1**. Click on ✓ to commit (not shown).

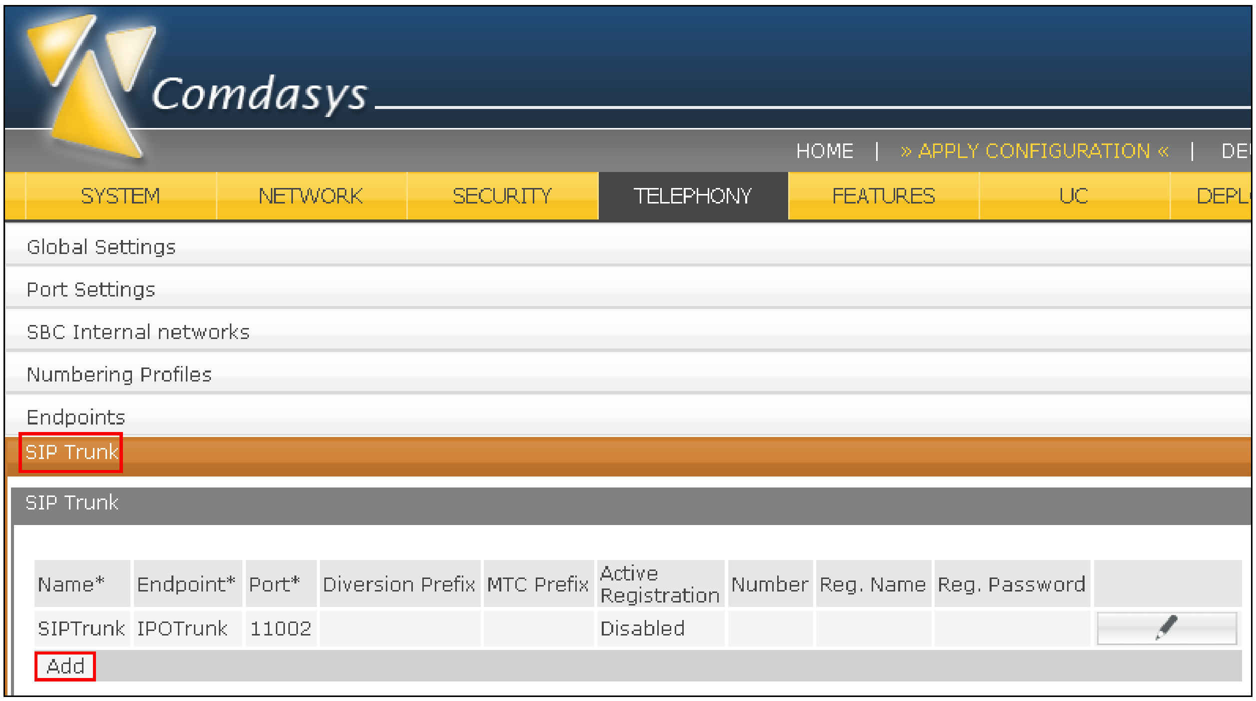


6.6. Configure SIP Trunk

The SIP Trunk will be used whenever the Mobile Convergence Controller needs to call a registered MC client which is not connected in Wi-Fi mode. Click **TELEPHONY → SIP Trunk → Add** and configure as follows:

- **Name** - assign an identifying name
- **Endpoint** – configure the trunk endpoint configured in **Section 6.5**
- **Port** – choose a local port on the Mobile Convergence Controller side for establishing the SIP trunk. Note this port was also configured in **Section 5.3**

Click on  to commit (not shown).

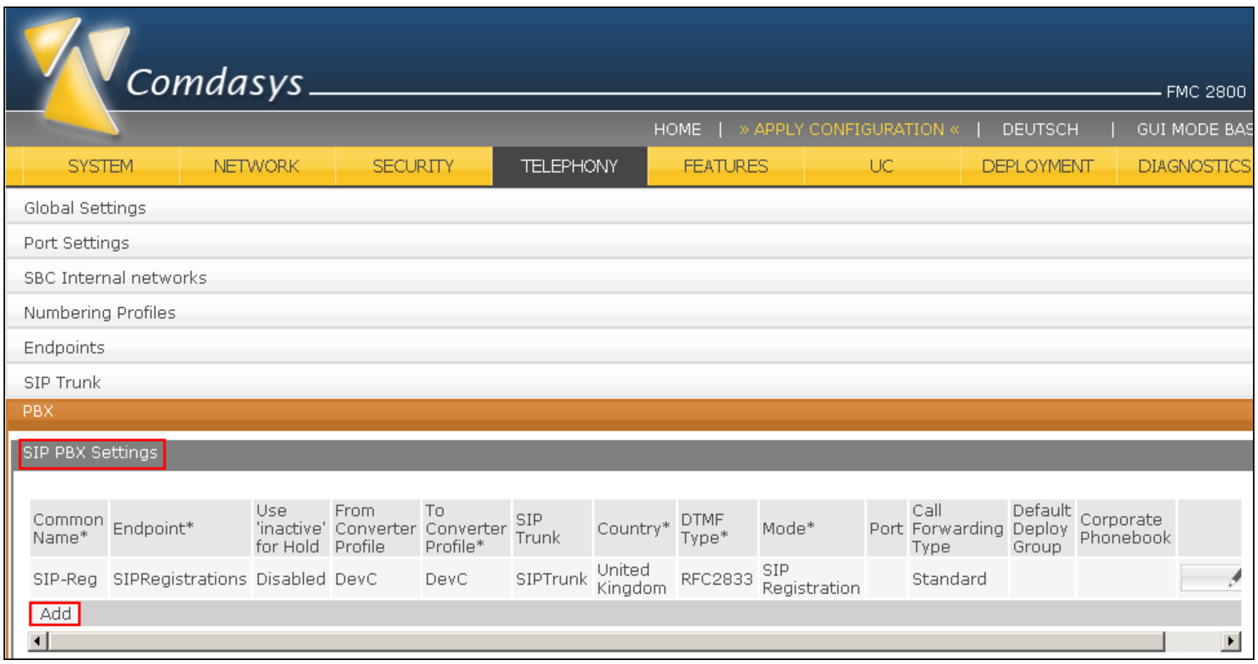


6.7. Configure PBX Profile

A new PBX must be added in order for SIP registrations to be made from the Mobile Convergence Controller. Click **TELEPHONY → PBX → Add** and configure as follows:

- **Common Name** – assign an identifying name
- **Endpoint** - select the user registration endpoint configured in **Section 6.5**
- **From Converter Profile** - select the numbering profile configured in **Section 6.4**
- **SIP Trunk** – configure the SIP trunk endpoint configured in **Section 6.6**
- **Country** – configure in accordance with the country of implementation
- **Mode** - choose **SIP Registration**
- **Call Forwarding Type** – select **Standard** (this uses the 302 Moved Temporarily feature). It is recommended however that **Trunk** is used for interoperability with IP Office

Click on  to commit (not shown).

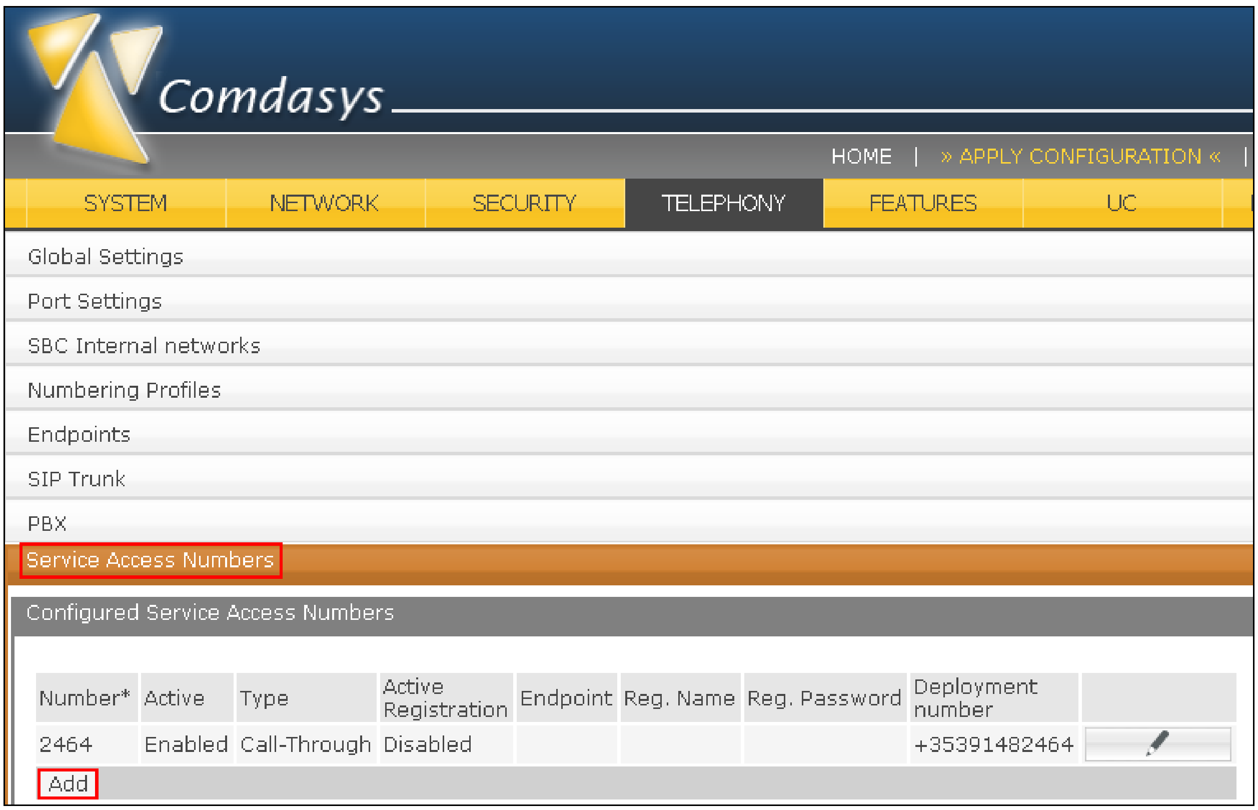


6.8. Configure Service Access Number

The call through feature is mandatory and is configured as a Service Access Number. This single number will be shared by all users to access the call through service. Click **TELEPHONY → Service Access Numbers → Add** and configure as follows:

- **Number** – enter the call through number as it is presented from IP Office, this is the short code defined in **Section 5.6**
- **Enabled** – check the box to enable the feature
- **Type** – set to **Call-Through**
- **Deployment Number** – configure the DDI for the call through feature in its international format

Click on  to commit (not shown).

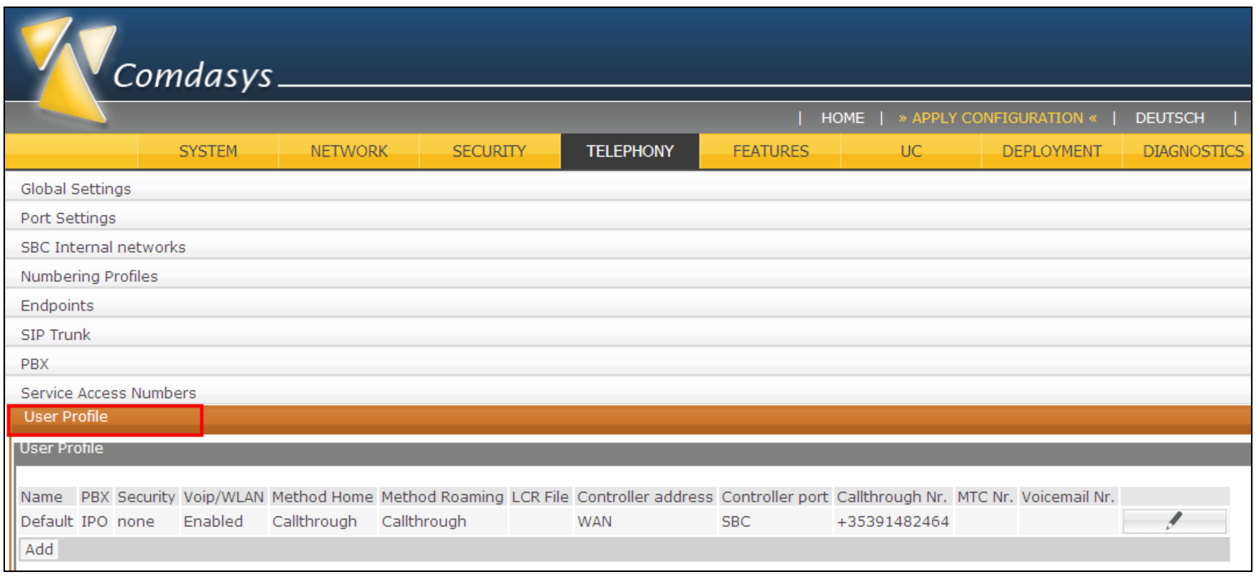


6.9. Configure User Profile

A user profile is added in order to have different user groups. Click **TELEPHONY → User Profile→ Add** and configure as follows:

- **PBX** – **IPO** (PBX type is IP Office)
- **Callthrough Nr.** – number created in **Section 6.8**
- **Controller address –** Select **WAN** or Public IP Address
- The remaining fields are customer specific

Click on  to commit (not shown).

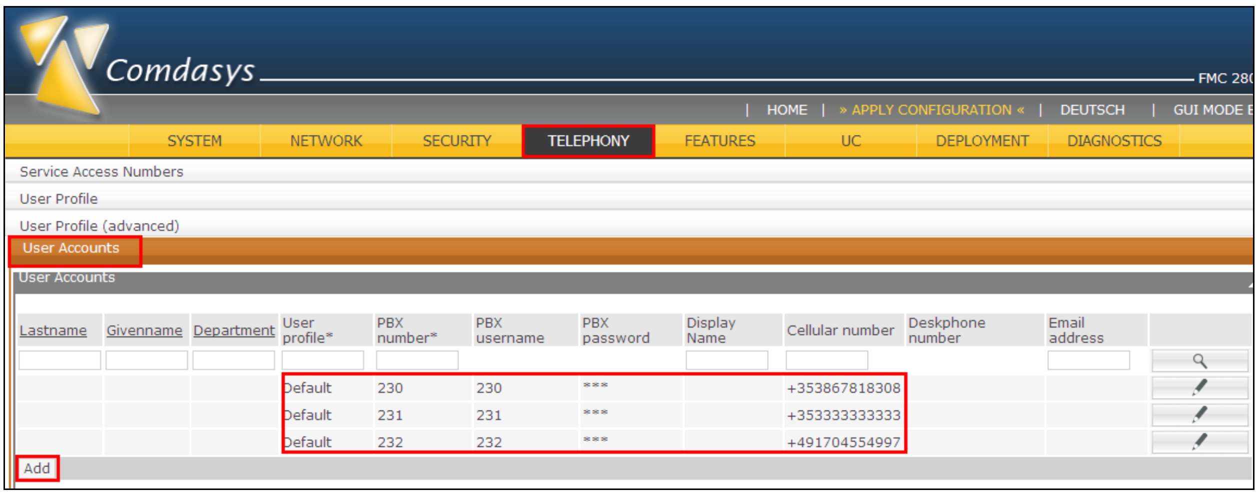


6.10. Configure User Accounts

Each Mobile Convergence Controller user requires a user account. Click **TELEPHONY → User Accounts → Add** and configure as follows:

- **PBX Number** – enter a User Extension number as configured in **Section 5.4**
- **PBX Password** – enter the corresponding password configured in **Section 5.4** under the **Supervisor Settings** sub tab
- **PBX Username** – enter an identifying user name
- **Cellular Number** – enter the corresponding cell phone number in the format as shown below

Click on  to commit (not shown).

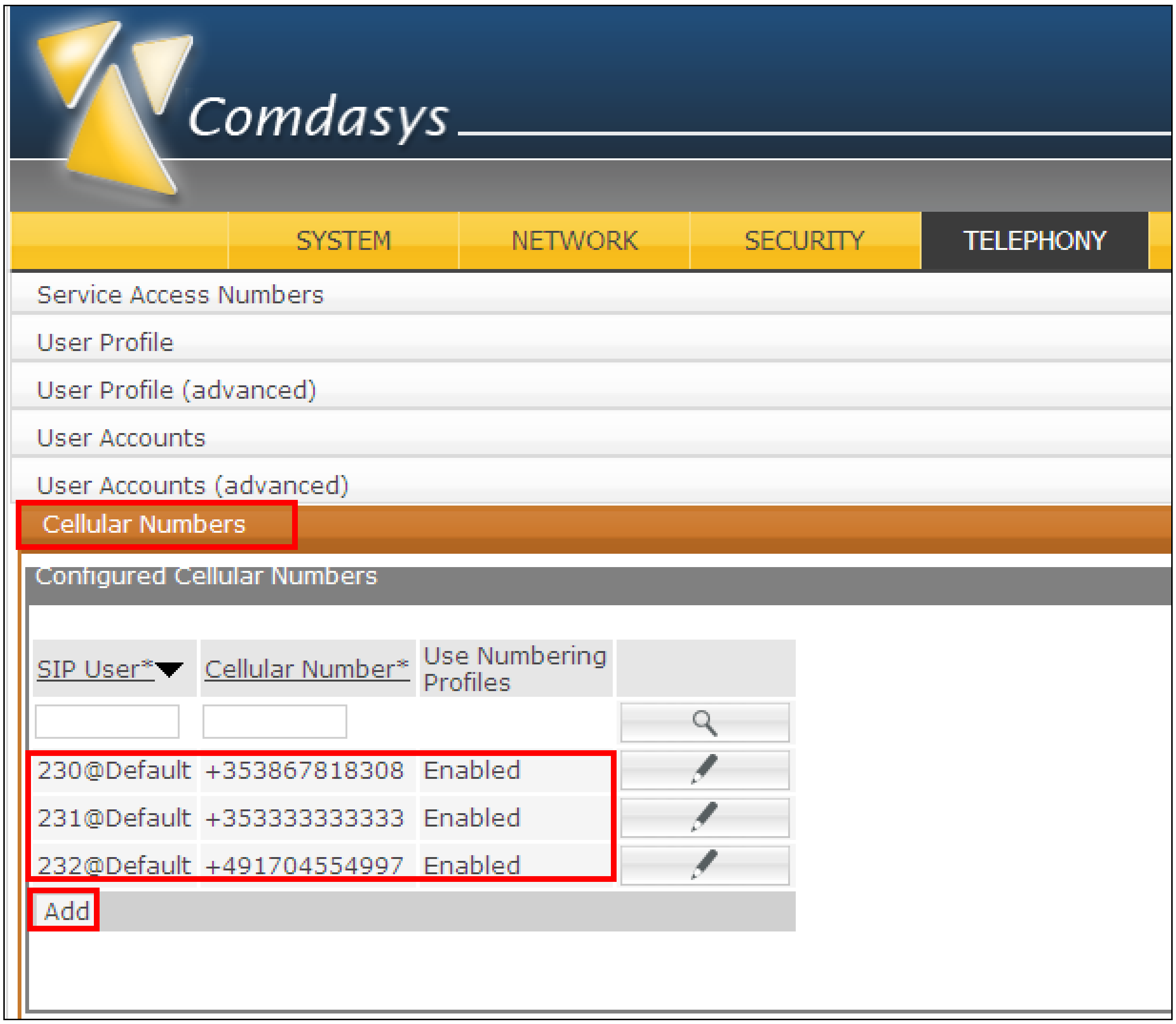


6.11. Configure Cellular Numbers

Each mobile phone number should be configured in international E.164 format with a leading plus sign. Click **TELEPHONY → Cellular Numbers → Add** and configure as follows:

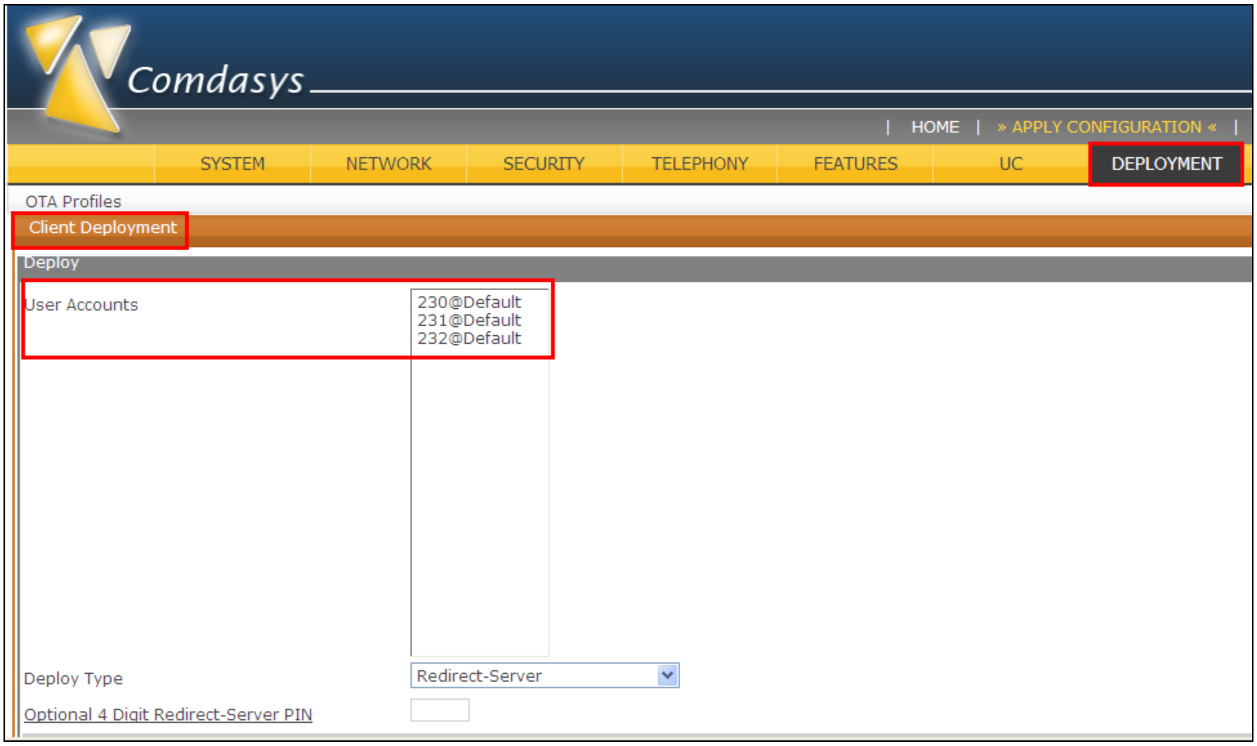
- **SIP User** - select a User Account configured in **Section 6.12**
- **Use Numbering Profiles** – check the tick box to enable this option. Whenever the controller needs to call one of the mobile phones it converts the number according to the rules defined in **Section 6.4**. A user might have more the one mobile number, however only one can be active at the same time

Click on  to commit (not shown).



6.12. Deploy Configuration to MC Clients

The configuration is now ready to be pushed to the MC client. This is achieved by pushing to a public deployment server. When the MC client logs in they connect to the public deployment server to obtain their configuration. Click **DEPLOYMENT → Client Deployment → Deploy** and select the relevant **User Accounts**, choose the **Redirect Server** as the **Deploy Type** and click **Send** (not shown).

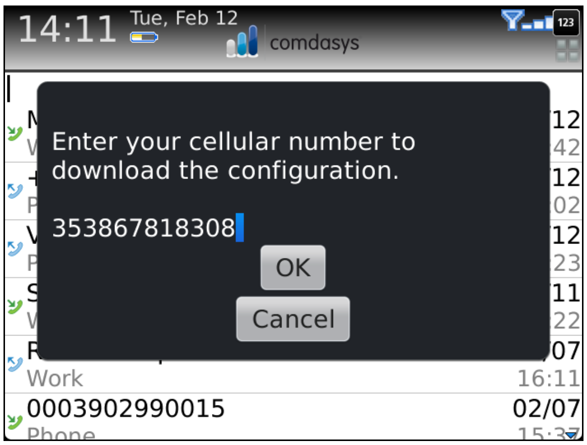


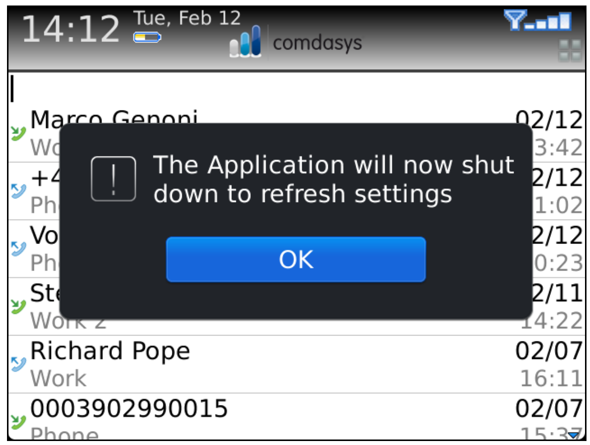
7. Configure Comdasys Mobile Convergence Client

The setup of the MC Clients is not part of this document and might differ depending on the used phone platform. However, the mandatory settings for Blackberry user, extension 230 are described below. The Client can be installed from the various application stores accessible on the phone. Search for “MC Client” or “iMC Client” on the iPhone.

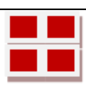
After first startup of the application enter the mobile phone number related to the handset, the redirect configuration will be downloaded from the provisioning server deployed by the Mobile Convergence Controller as described in **Section 6.12**.

Please note that the phone requires internet access for contacting the provisioning server.





All Clients will show a registraion status, indicated by one of the following symbols:

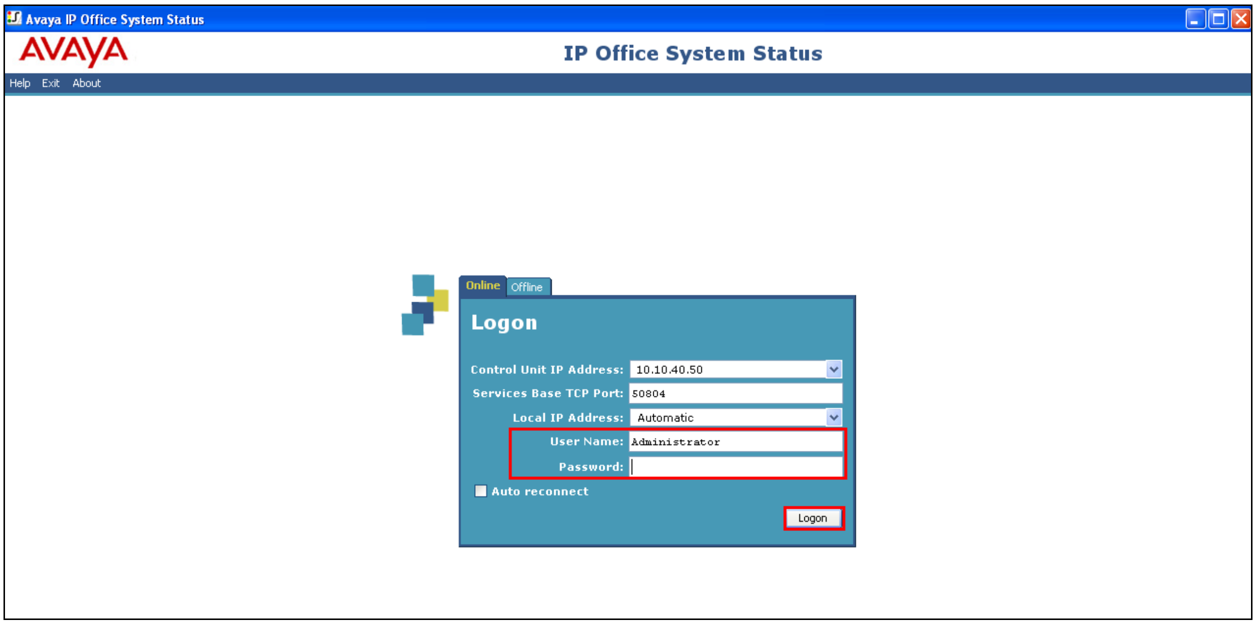
	Successful registered for VoIP - calls routed via WIFI or 3G Data
	Successful registered “InfoMode” - calls routed via cellular network
	Connected to network / registration in progress
	Registration failed / connection refused
	Offline - no WIFI / packet data connection active / available, calling possible via cellular but without features

8. Verification Steps

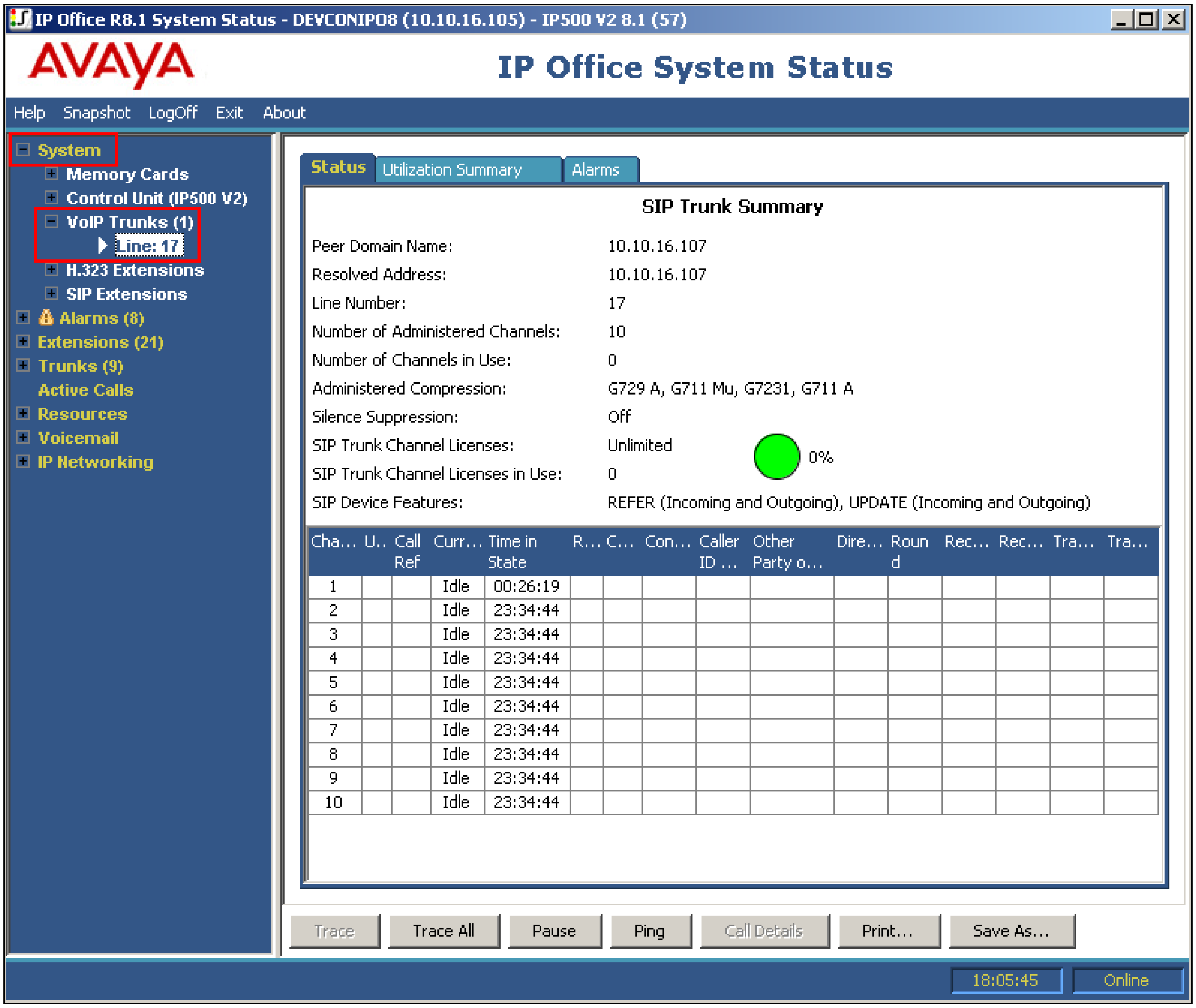
This section provides the tests that can be performed to verify correct configuration of Avaya and Mobile Convergence Controller solution.

8.1. Verify Avaya IP Office SIP Trunk

Open the **IP Office System Status** application and enter the proper credentials and click **Logon** as shown below.

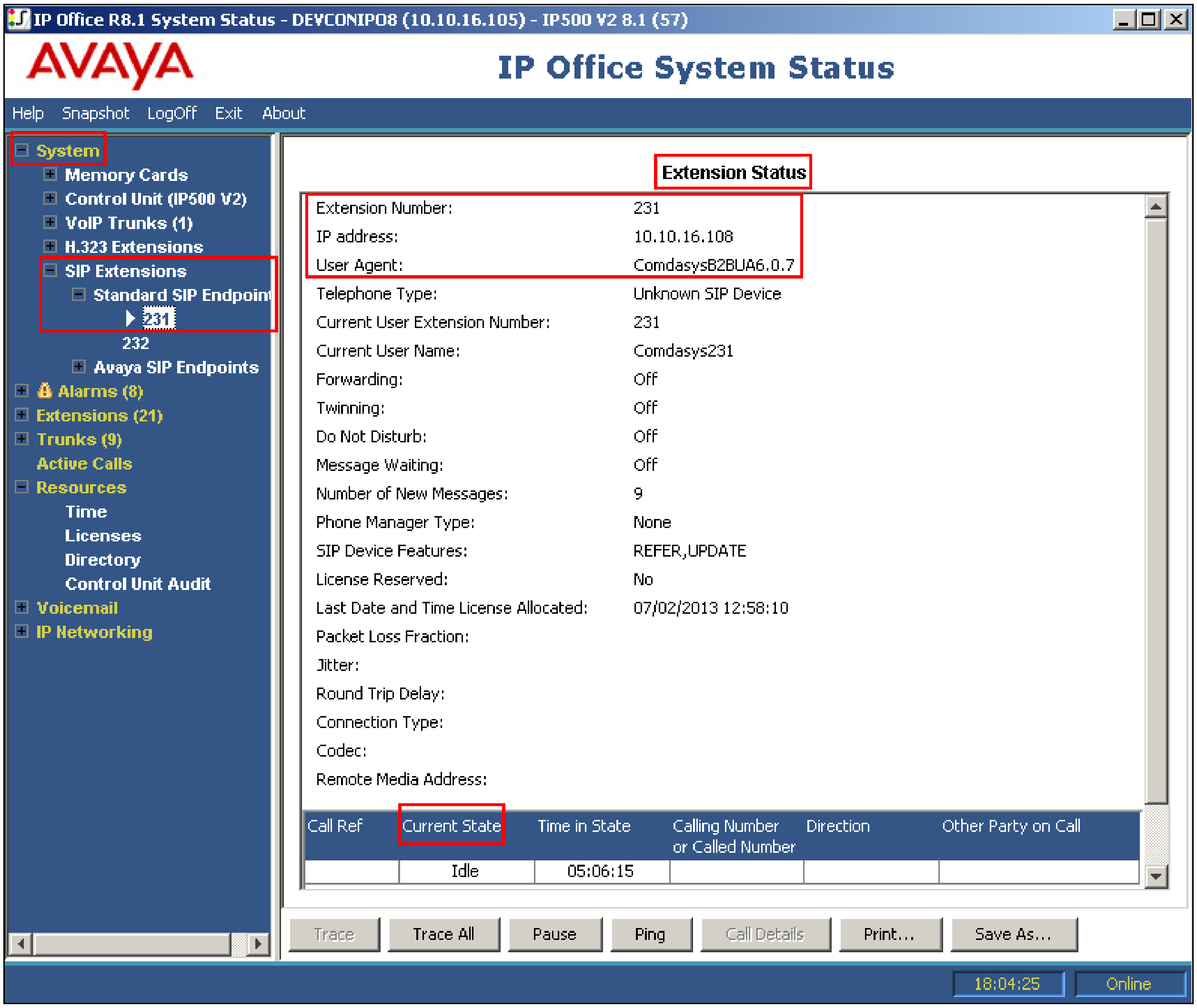


Click **System → VoIP Trunks** and select the appropriate SIP trunk. Confirm the channel status accurately represents activity on the trunk and that there are no alarms.



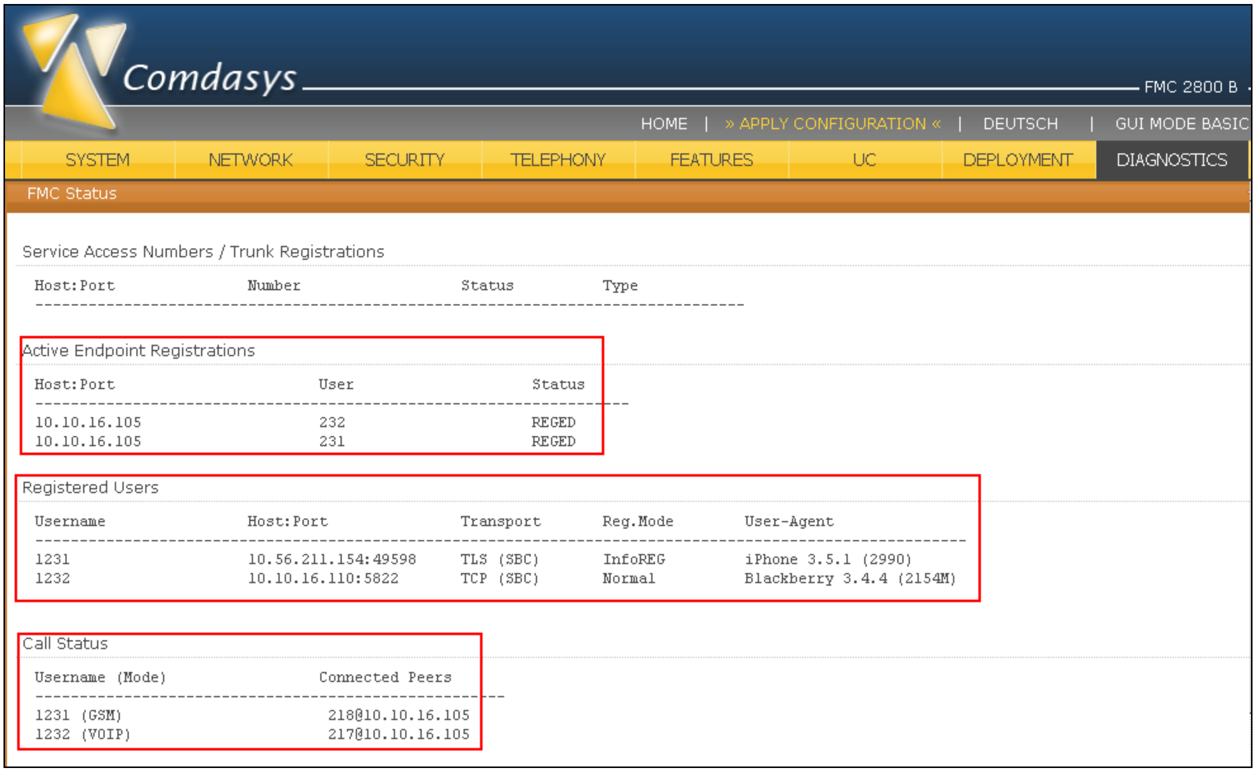
8.2. Verify Avaya IP Office SIP User Registrations

In the IP Office System Status application, click **System** → **SIP Extensions** → **Standard SIP Endpoints** and select the relevant SIP endpoint. Verify the **Extension Status** is as expected and the **Current State** is correct.



8.3. Verify Comdasys Mobile Convergence Controller Active Endpoint Registrations, Registered Users and VoIP/GSM Call Status

Click on **DIAGNOSTICS**. Confirm **Active Endpoint Registrations** match the SIP Users added on IP Office, **Registered Users** match the Users administered on Mobile Convergence Controller, and **Call Status** match **GSM** and **VoIP** delivered calls.



9. Conclusion

These Application Notes describe the configuration steps required for the Comdasys Mobile Convergence Solution to successfully interoperate with Avaya IP Office 500 v2 R9.0. All functionality cases were completed successfully.

10. Additional References

This section references documentation relevant to these Application Notes. In general, Avaya product documentation is available at http://support.avaya.com.

[1] *IP Office 9.0 IP500/IP500 V2 Installation*, Document Number 15-601042, Issue 27m, July 2, 2013.
[2] *IP Office Release 9.0 Manager 9.0*, Document Number 15-601011, Issue 29u, April 5, 2013.
[3] *IP Office System Status Application*, Document Number 15-601758, Issue 07a, November 26, 2012.
[4] *IP Office System Monitor*, Document Number 15-601019, Issue 03c, March 1, 2013

Comdasys product documentation is available at http://www.comdasys.com. The following documents were downloaded and used during the compliance testing.

[5] Comdasys Mobile Convergence Administrator Manual
http://ftp.comdasys.com/pub/documentation/FMC_series/
[6] Comdasys Mobile Convergence Client Manuals:
http://ftp.comdasys.com/pub/documentation/MC_Client/

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.