



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.

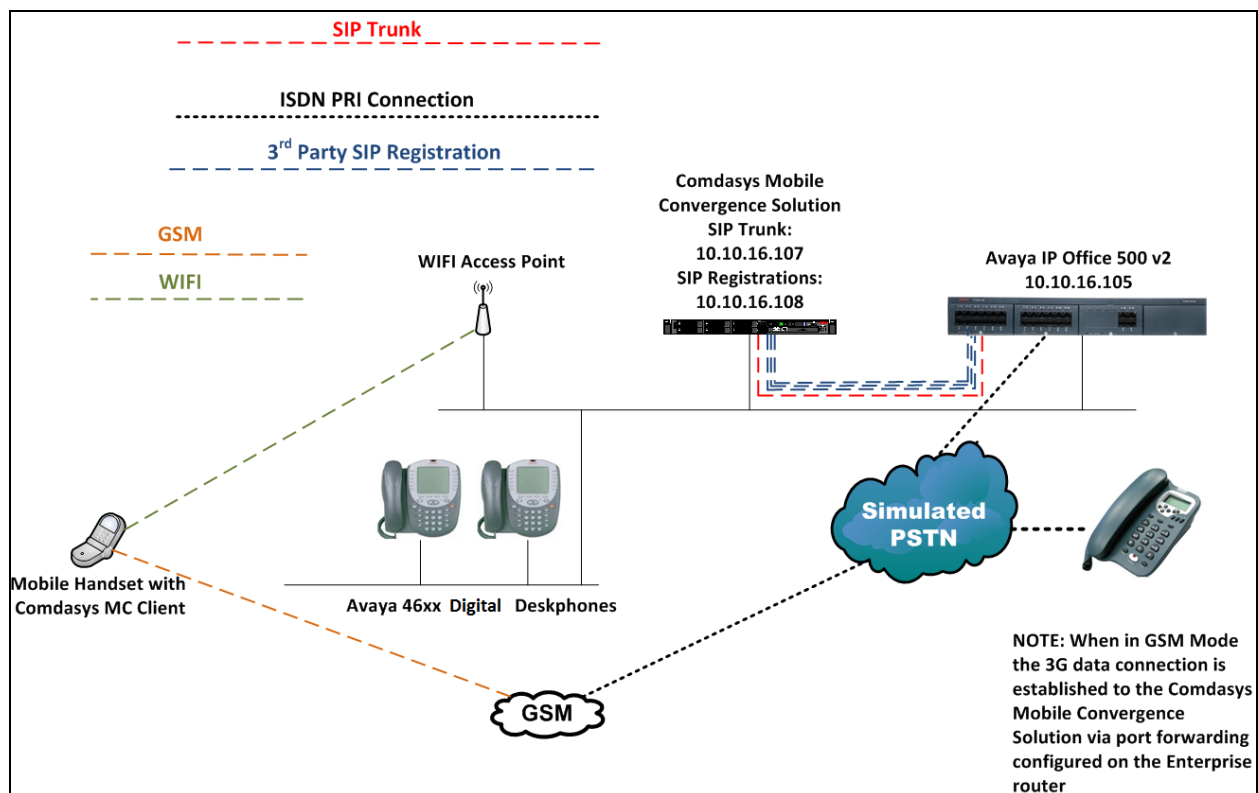


Figure 1: Avaya IP Office 500 v2 with Comdasys Mobile Convergence Solution Configuration

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)

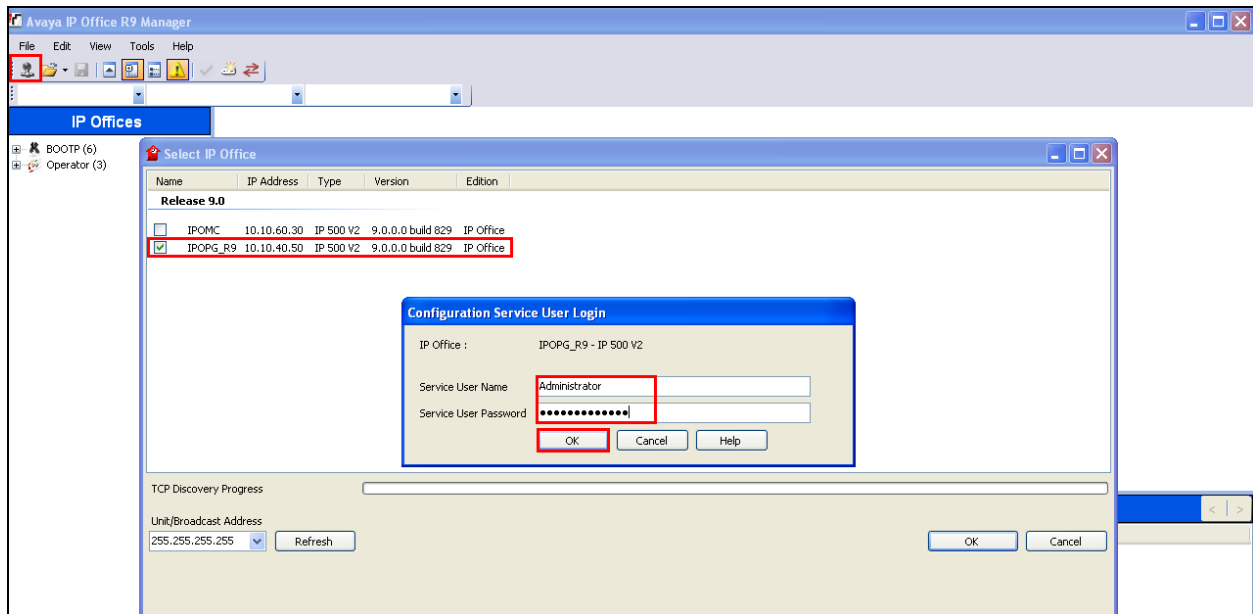
Testing was performed with IP Office 500 v2 R9.0, but it also applies to IP Office Server Edition R9.0. Note that IP Office Server Edition requires an Expansion IP Office 500 v2 R9.0 to support analog or digital endpoints or trunks.

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

The screenshot shows the IP Office configuration interface. On the left, the 'IP Offices' hierarchy is displayed, with 'IPOP_R9' selected. The main pane shows the 'LAN1' tab selected, with the 'LAN Settings' sub-tab active. The 'IP Address' is set to '10 . 10 . 16 . 105' and the 'IP Mask' is set to '255 . 255 . 255 . 0'. Other settings include 'Primary Trans. IP Address' (0 . 0 . 0 . 0), 'RIP Mode' (None), 'Enable NAT' (unchecked), and 'Number Of DHCP IP Addresses' (10). The 'DHCP Mode' is set to 'Disabled'.

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**

The screenshot shows the IP Office configuration interface with the 'VoIP' sub-tab selected. The 'H323 Gatekeeper Enable' checkbox is checked. The 'SIP Trunks Enable', 'SIP Registrar Enable', and 'Auto-create Extn/User' checkboxes are checked. The 'Domain Name' is set to '10.10.16.105'. The 'Layer 4 Protocol' section shows 'UDP' and 'TCP' checked, with 'UDP Port' and 'TCP Port' both set to '5060'. The 'Challenge Expiry Time (secs)' is set to '10'.

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.

The screenshot displays the IP Office configuration interface. On the left, the 'IP Offices' navigation pane shows a tree structure with 'IP Route (2)' highlighted. The main configuration area is titled 'IP Route' and shows the following fields:

- IP Address:** 0 . 0 . 0 . 0
- IP Mask:** 0 . 0 . 0 . 0
- Gateway IP Address:** 10 . 10 . 16 . 1
- Destination:** LAN1
- Metric:** 0
- ☐ Proxy ARP

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.

The screenshot shows the 'SIP Line - Line 17' configuration window in IP Office. The left navigation pane shows the 'Line (10)' folder selected. The main window has tabs for 'SIP Line', 'transport', 'SIP URI', 'VoIP', 'T38 Fax', and 'SIP Credentials'. The 'SIP Line' tab is active. The 'Line Number' is set to 17 and the 'ITSP Domain Name' is set to 10.10.16.107. Other fields include Prefix, National Prefix (0), Country Code, International Prefix (00), Send Caller ID (None), and Association Method (By Source IP address). On the right, there are checkboxes for 'In Service' (checked), 'Check OOS' (checked), and 'Call Routing Method' (Request URI). There are also dropdowns for 'URI Type' (SIP), 'Name Priority' (System Default), 'Service Busy Response' (486 - Busy Here), and 'Action on CAC Location Limit' (Allow Voicemail). At the bottom, there is a 'REFER Support' section with 'Incoming' and 'Outgoing' set to 'Auto'. The window has 'OK' and 'Cancel' buttons at the bottom right.

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**.

SIP Line **Transport** SIP URI VoIP T38 Fax SIP Credentials

ITSP Proxy Address

Network Configuration

Layer 4 Protocol Send Port

Use Network Topology Info Listen Port

Explicit DNS Server(s)

Calls Route via Registrar ☐

Separate Registrar

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.

The screenshot shows a configuration window with tabs: SIP Line, Transport, SIP URI, VoIP, T38 Fax, and SIP Credentials. The 'SIP URI' tab is selected. Below the tabs is a table with columns: Channel, Groups, Via, Local URI, Contact, and Display Name. To the right of the table are buttons: Add..., Remove, and Edit... The 'Add...' button is highlighted with a red box. Below the table is the 'Edit Channel' section. It contains the following fields: Via (set to <None>), Local URI (set to *), Contact (set to *), Display Name (set to *), PAI (set to None), Registration (set to 0: <None>), Incoming Group (set to 17), Outgoing Group (set to 17), and Max Calls per Channel (set to 10). The 'Edit Channel' section is highlighted with a red box. To the right of the 'Edit Channel' section are buttons: OK and Cancel. The 'OK' button is highlighted with a red box.

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

User	Voicemail	DND	ShortCodes	Source Numbers	Telephony	Forwarding	Dial In	Voice Recording
Name	Comdasys230							
Password	****							
Confirm Password	****							
Full Name	Comdasys230							
Extension	230							
Locale	<div></div>							
Priority	5							
System Phone Rights	None							
Profile	Basic User							
<input type="checkbox"/> Receptionist								
<input type="checkbox"/> Enable Softphone								
<input type="checkbox"/> Enable one-X Portal Services								
<input type="checkbox"/> Enable one-X TeleCommuter								
<input type="checkbox"/> Enable Remote Worker								
<input type="checkbox"/> Enable Flare								
Flare Mode								Standalone

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

The screenshot shows the 'Voicemail' tab selected in a user configuration window. The 'Voicemail On' checkbox is checked and highlighted with a red box. Other options include 'Voicemail Help' (checked), 'Voicemail Ringback' (unchecked), 'Voicemail Email Reading' (unchecked), and 'UMS Web Services' (unchecked). There are input fields for 'Voicemail Code', 'Confirm Voicemail Code', and 'Voicemail Email'. Below these, there are radio buttons for 'Off', 'Copy', 'Forward', and 'Alert'. At the bottom, there is a 'DTMF Breakout' section with three dropdown menus for 'Reception / Breakout (DTMF *0)', 'Breakout (DTMF *2)', and 'Breakout (DTMF *3)', all set to 'System Default ()'.

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

The screenshot shows the 'Telephony' tab selected, with the 'Call Settings' sub-tab active. The 'Call Waiting On' checkbox is checked and highlighted with a red box. Other options include 'Answer Call Waiting On Hold' (checked), 'Busy On Held' (unchecked), and 'Offhook Station' (unchecked). There are several input fields: 'Outside Call Sequence' (Default Ring), 'Inside Call Sequence' (Default Ring), 'Ringback Sequence' (Default Ring), 'No Answer Time (secs)' (25, highlighted with a red box), 'Wrap-up Time (secs)' (2), 'Transfer Return Time (secs)' (Off), and 'Call Cost Mark-Up' (100).

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

The screenshot shows the 'Supervisor Settings' tab in the Avaya IP Office Manager. The 'Login Code' field is highlighted with a red box and contains '****'. Other fields include 'Login Idle Period (secs)', 'Monitor Group' (set to '<None>'), 'Coverage Group' (set to '<None>'), and 'Status on No-Answer' (set to 'Logged On (No change)'). There are radio buttons for 'All Calls' (selected) and 'External Incoming'. A section for 'Reset Longest Idle Time' is also present. On the right, there are several checkboxes: 'Force Login', 'Force Account Code', 'Outgoing Call Bar', 'Inhibit Off-Switch Forward/Transfer', 'Can Intrude', 'Cannot be Intruded' (checked), 'Can Trace Calls', 'CCR Agent', 'Automatic After Call Work', and 'Deny Auto Intercom Calls'. The 'After Call Work Time (secs)' is set to 'System Default (10)'.

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.

The screenshot shows a dialog box titled 'Avaya IP Office Manager'. It asks 'Would you like a new VoIP extension created with this number?'. There are three radio button options: 'None', 'H323 Extension', and 'SIP Extension'. The 'SIP Extension' option is selected and highlighted with a red box. At the bottom, there is an 'OK' button, also highlighted with a red box.

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.

The screenshot shows the 'Standard' tab of the Incoming Call Route configuration window. The 'Line Group ID' field is highlighted with a red box and contains the value '17'. Other fields include 'Bearer Capability' (Any Voice), 'Incoming Number', 'Incoming Sub Address', 'Incoming CLI', 'Locale', 'Priority' (1 - Low), 'Tag', and 'Hold Music Source' (System Source).

Field	Value
Bearer Capability	Any Voice
Line Group ID	17
Incoming Number	
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source

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.

The screenshot shows the 'Destinations' tab of the Incoming Call Route configuration window. The 'Default Value' field is highlighted with a red box and contains the value “.”. The 'Destination' field is also highlighted with a red box and contains the value “.”. The 'Fallback Extension' field is empty.

Field	Value
TimeProfile	
Default Value	.”
Destination	.”
Fallback Extension	

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**

Short Code	
Code	2464
Feature	Dial
Telephone Number	2464
Line Group ID	17
Locale	
Force Account Code	<input type="checkbox"/>

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

Standard	Voice Recording	Destinations
Bearer Capability	Any Voice	
Line Group ID	9	
Incoming Number	091482464	
Incoming Sub Address		
Incoming CLI		
Locale		
Priority	1 - Low	
Tag		
Hold Music Source	System Source	

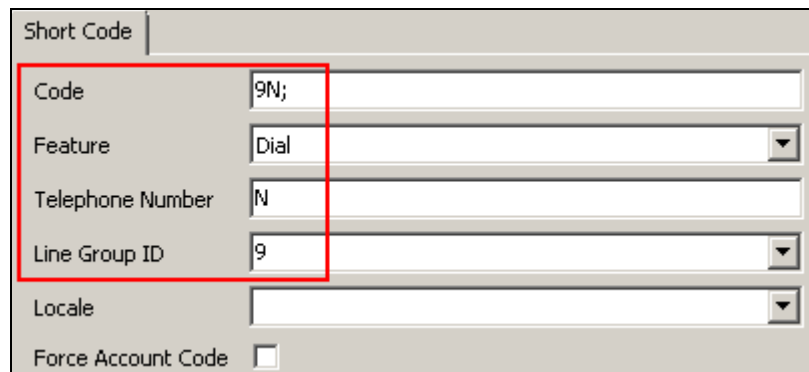
Click the **Destinations** tab and set the **Default Value** with a **Destination** of the short code configured in **Section 5.6** – this will route the incoming number to the short code destination.

Standard	Voice Recording	Destinations
TimeProfile	Destination	Fallback Extension
Default Value	2464	

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**



The screenshot shows a configuration form titled "Short Code". It contains several fields: "Code" with the value "9N;", "Feature" with a dropdown menu showing "Dial", "Telephone Number" with the value "N", "Line Group ID" with a dropdown menu showing "9", "Locale" with a dropdown menu, and "Force Account Code" with an unchecked checkbox. A red rectangular box highlights the "Code", "Feature", "Telephone Number", and "Line Group ID" fields.

Short Code	
Code	9N;
Feature	Dial
Telephone Number	N
Line Group ID	9
Locale	
Force Account Code	<input type="checkbox"/>

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.



The screenshot shows the Comdasys web interface. The 'NETWORK' tab is selected. Under 'LAN Interface 1', the 'Basic Settings' section is expanded. The 'IP address' field is set to 10.10.16.107 and the 'Netmask' field is set to 255.255.255.0. A 'Save' button is located at the bottom right of the configuration area.

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.



The screenshot shows the Comdasys web interface. The 'NETWORK' tab is selected. Under 'Virtual Interfaces', the 'Configured Virtual Interfaces' section is expanded. A table lists the configured interfaces:

Local Interface	Number(VLAN ID)	IP address	Netmask	802.1Q VLAN
LAN 1	1	10.10.16.108	255.255.255.0	Disabled

An 'Add Interface' button is located at the bottom of the table.

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.

Comdasys FMC 2800 B

HOME | APPLY CONFIGURATION | DEUTSCH | GUI MODE BASIC

SYSTEM **NETWORK** SECURITY TELEPHONY FEATURES UC DEPLOYMENT DIAGNOSTICS

WAN Interface

☐ Deactivated

☒ IP

IP address: 122.15
Netmask: 255.255.255.128
Gateway: 122.7

☐ PPTP

Username:
Password:
Modem IP:

☐ PPPoE

Username:
Password:

☐ Dynamic IP via DHCP

☒ Static IP

IP address:
Netmask:

Save Cancel

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 successful routing of calls when the MC client is in GSM mode.

Comdasys FMC 2800 B

HOME | » APPLY CONFIGURATION « | DEUTSCH | GUI MODE BASIC

SYSTEM NETWORK SECURITY **TELEPHONY** FEATURES UC DEPLOYMENT DIAGNOSTICS

Global Settings


Global Settings

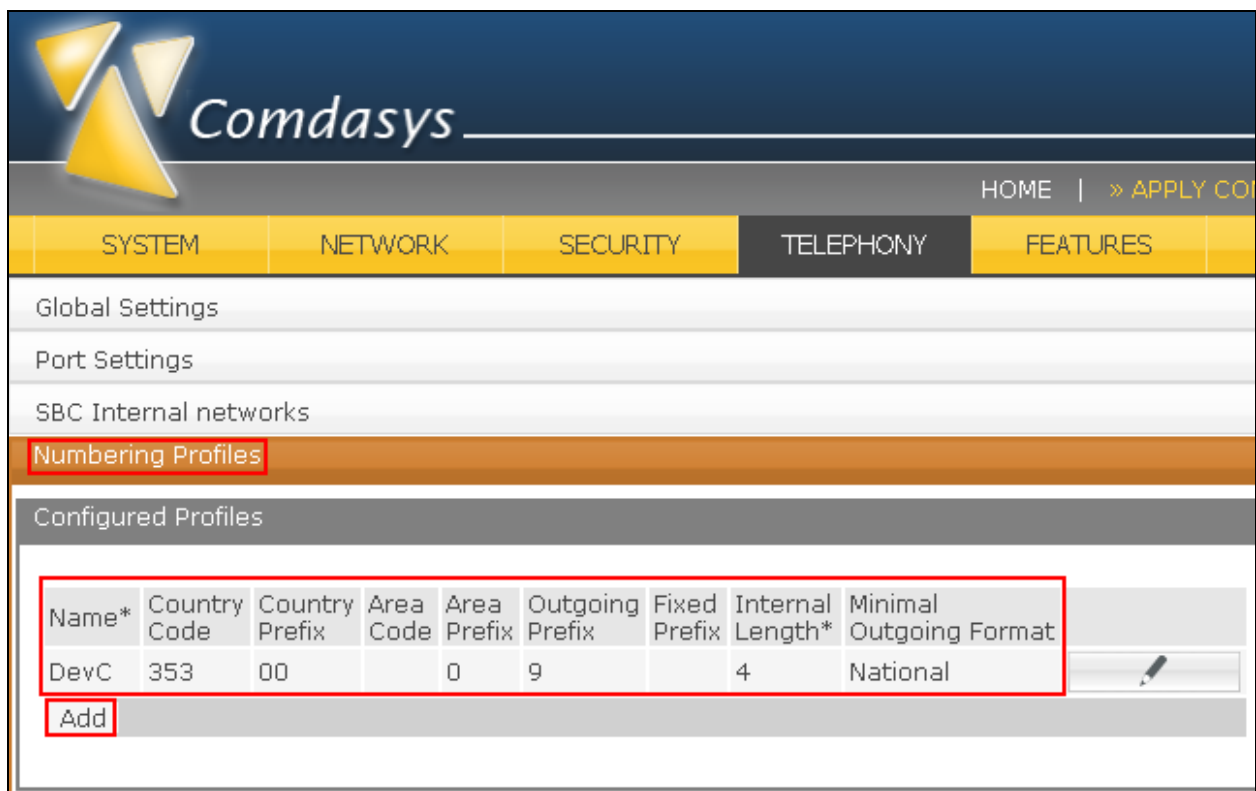
[Enable Call-Through Progress Indication](#) ☒
[Enable Client Early Media](#) ☒
[Enable busy sound in Wifi](#) ☒
[Disable Inband DTMF Detection](#) ☒
[Disable Number Converter](#) ☐
Enable DTMF invoked Handover ☐
[Unavailable Timeout](#)
[DTMF Duration](#)
[RTP payload-type for DTMF](#)
[Confirm SIM Switch with SMS](#) ☐
[Force Ringing on Early Media](#) ☐
[Use P-Asserted Identity](#) ☐
[Activate APN](#) ☐
[Process rininstance-tag](#) ☐
[Number of Cellular-digits to match](#)

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).



The screenshot shows the Comdasys web interface. The top navigation bar includes 'HOME' and '» APPLY COI'. Below this is a menu with 'SYSTEM', 'NETWORK', 'SECURITY', 'TELEPHONY', and 'FEATURES'. The 'TELEPHONY' section is active, showing a list of settings: 'Global Settings', 'Port Settings', 'SBC Internal networks', and 'Numbering Profiles' (which is highlighted with a red box). Under 'Numbering Profiles', there is a section for 'Configured Profiles' containing a table with one row of data. The table has columns for Name*, Country Code, Country Prefix, Area Code, Area Prefix, Outgoing Prefix, Fixed Prefix, Internal Length*, and Minimal Outgoing Format. The data row shows 'DevC', '353', '00', an empty cell for Area Code, '0' for Area Prefix, '9' for Outgoing Prefix, an empty cell for Fixed Prefix, '4' for Internal Length*, and 'National' for Minimal Outgoing Format. Below the table is an 'Add' button (highlighted with a red box) and a pencil icon for editing.


Name*	Country Code	Country Prefix	Area Code	Area Prefix	Outgoing Prefix	Fixed Prefix	Internal Length*	Minimal Outgoing Format
DevC	353	00		0	9		4	National

[Add](#)

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
- **Preferred Codec** – choose a preferred 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).



Comdasys

HOME | > APPLY CONFIGURATION < | DEU

SYSTEM NETWORK SECURITY **TELEPHONY** FEATURES UC DEPLO

Global Settings

Port Settings

SBC Internal networks

Numbering Profiles

Endpoints


Configured Endpoints

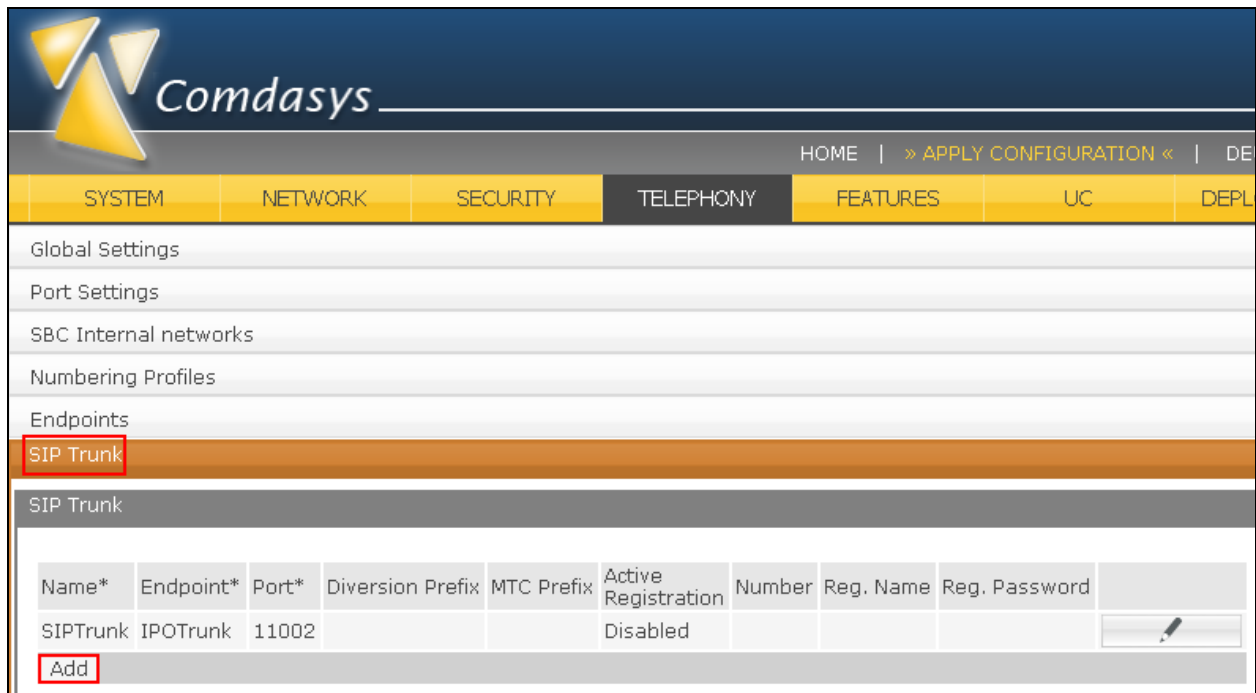
Common Name*	Hostname/IP*	Local Interface	Foreign Port*	Realm	Preferred Codec	Outbound Proxy	
IPOTrunk	10.10.16.105	LAN 1	5060		G.711 alaw / 20MS		
SIPRegistrations	10.10.16.105	LAN 1:1	5060		G.711 alaw / 20MS		
Add							

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).



The screenshot shows the Comdasys web interface. The top navigation bar includes links for HOME, » APPLY CONFIGURATION «, and DE. Below this is a menu with tabs for SYSTEM, NETWORK, SECURITY, TELEPHONY, FEATURES, UC, and DEPL. The TELEPHONY tab is selected. Under TELEPHONY, there are links for Global Settings, Port Settings, SBC Internal networks, Numbering Profiles, Endpoints, and SIP Trunk. The SIP Trunk link is highlighted with a red box. Below the SIP Trunk link, there is a table with columns: Name*, Endpoint*, Port*, Diversion Prefix, MTC Prefix, Active Registration, Number, Reg. Name, and Reg. Password. The table contains one row with the values: SIPTrunk, IPOTrunk, 11002, , , Disabled, , , and . Below the table, there is an 'Add' button highlighted with a red box.


Name*	Endpoint*	Port*	Diversion Prefix	MTC Prefix	Active Registration	Number	Reg. Name	Reg. Password
SIPTrunk	IPOTrunk	11002			Disabled			

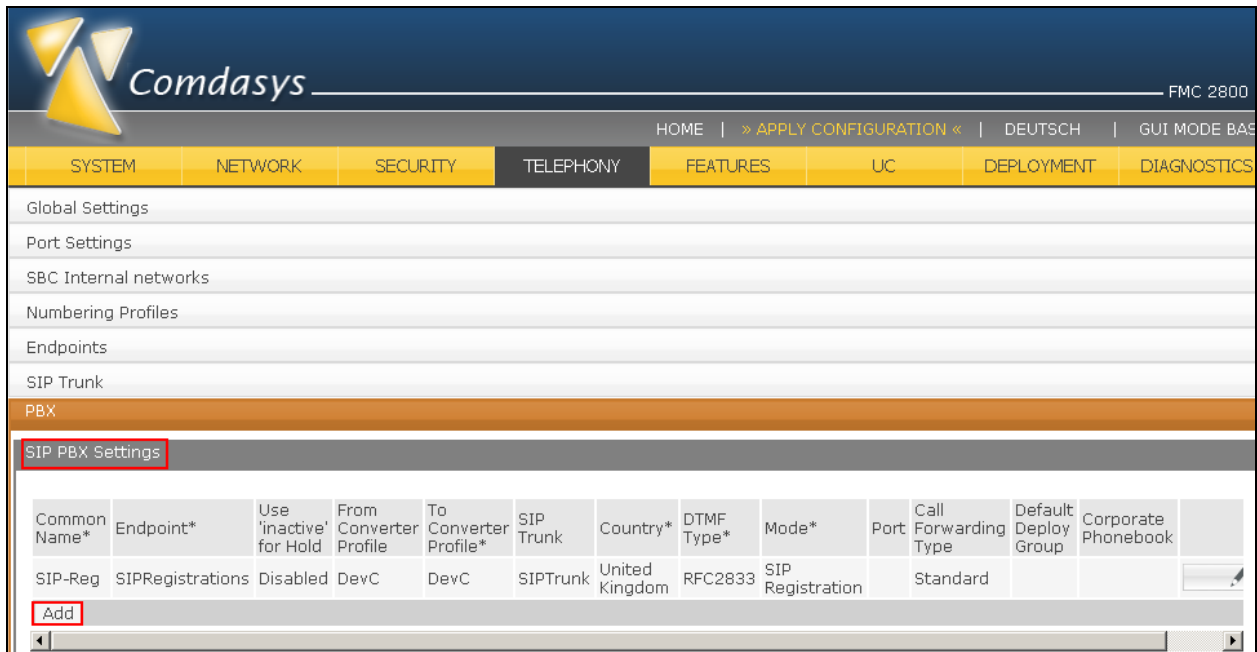
[Add](#)

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).



The screenshot shows the Comdasys FMC 2800 web interface. The top navigation bar includes the Comdasys logo, the text 'FMC 2800', and links for 'HOME', '» APPLY CONFIGURATION «', 'DEUTSCH', and 'GUI MODE BAS'. Below this is a main menu with tabs for 'SYSTEM', 'NETWORK', 'SECURITY', 'TELEPHONY', 'FEATURES', 'UC', 'DEPLOYMENT', and 'DIAGNOSTICS'. The 'TELEPHONY' tab is selected, and a sidebar on the left lists various settings: 'Global Settings', 'Port Settings', 'SBC Internal networks', 'Numbering Profiles', 'Endpoints', 'SIP Trunk', and 'PBX'. The 'PBX' section is highlighted in orange. Under 'PBX', there is a 'SIP PBX Settings' section with a table of existing profiles and an 'Add' button.


Common Name*	Endpoint*	Use 'inactive' for Hold	From Converter Profile	To Converter Profile*	SIP Trunk	Country*	DTMF Type*	Mode*	Port	Call Forwarding Type	Default Deploy Group	Corporate Phonebook
SIP-Reg	SIPRegistrations	Disabled	DevC	DevC	SIPTrunk	United Kingdom	RFC2833	SIP Registration		Standard		

Add

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).




The screenshot shows the Comdasys web interface. At the top is the Comdasys logo. Below it is a navigation bar with tabs: SYSTEM, NETWORK, SECURITY, TELEPHONY, FEATURES, and UC. The TELEPHONY tab is selected. Under the TELEPHONY tab, there is a list of sub-items: Global Settings, Port Settings, SBC Internal networks, Numbering Profiles, Endpoints, SIP Trunk, PBX, and Service Access Numbers. The Service Access Numbers item is highlighted with a red box. Below this, there is a section titled 'Configured Service Access Numbers' which contains a table with the following data:

Number*	Active	Type	Active Registration	Endpoint	Reg. Name	Reg. Password	Deployment number	
2464	Enabled	Call-Through	Disabled				+35391482464	
Add								

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).



Comdasys

HOME | > APPLY CONFIGURATION < | DEUTSCH

SYSTEM NETWORK SECURITY **TELEPHONY** FEATURES UC DEPLOYMENT DIAGNOSTICS

Global Settings

Port Settings

SBC Internal networks

Numbering Profiles

Endpoints

SIP Trunk

PBX

Service Access Numbers

User Profile


User Profile

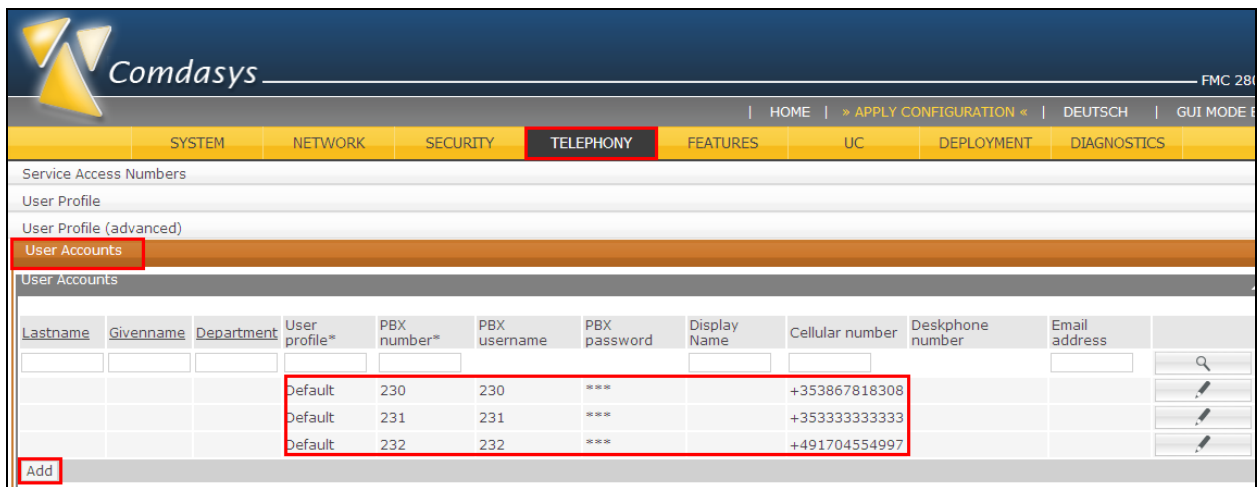
Name	PBX	Security	Voip/WLAN	Method Home	Method Roaming	LCR File	Controller address	Controller port	Callthrough Nr.	MTC Nr.	Voicemail Nr.	
Default	IPO	none	Enabled	Callthrough	Callthrough		WAN	SBC	+35391482464			
Add												

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).



Comdasys FMC 280

HOME » APPLY CONFIGURATION « DEUTSCH GUI MODE

SYSTEM NETWORK SECURITY **TELEPHONY** FEATURES UC DEPLOYMENT DIAGNOSTICS




Service Access Numbers

User Profile

User Profile (advanced)

User Accounts

User Accounts


Lastname	Givenname	Department	User profile*	PBX number*	PBX username	PBX password	Display Name	Cellular number	Deskphone number	Email address	
			Default	230	230	***		+353867818308			
			Default	231	231	***		+353333333333			
			Default	232	232	***		+491704554997			

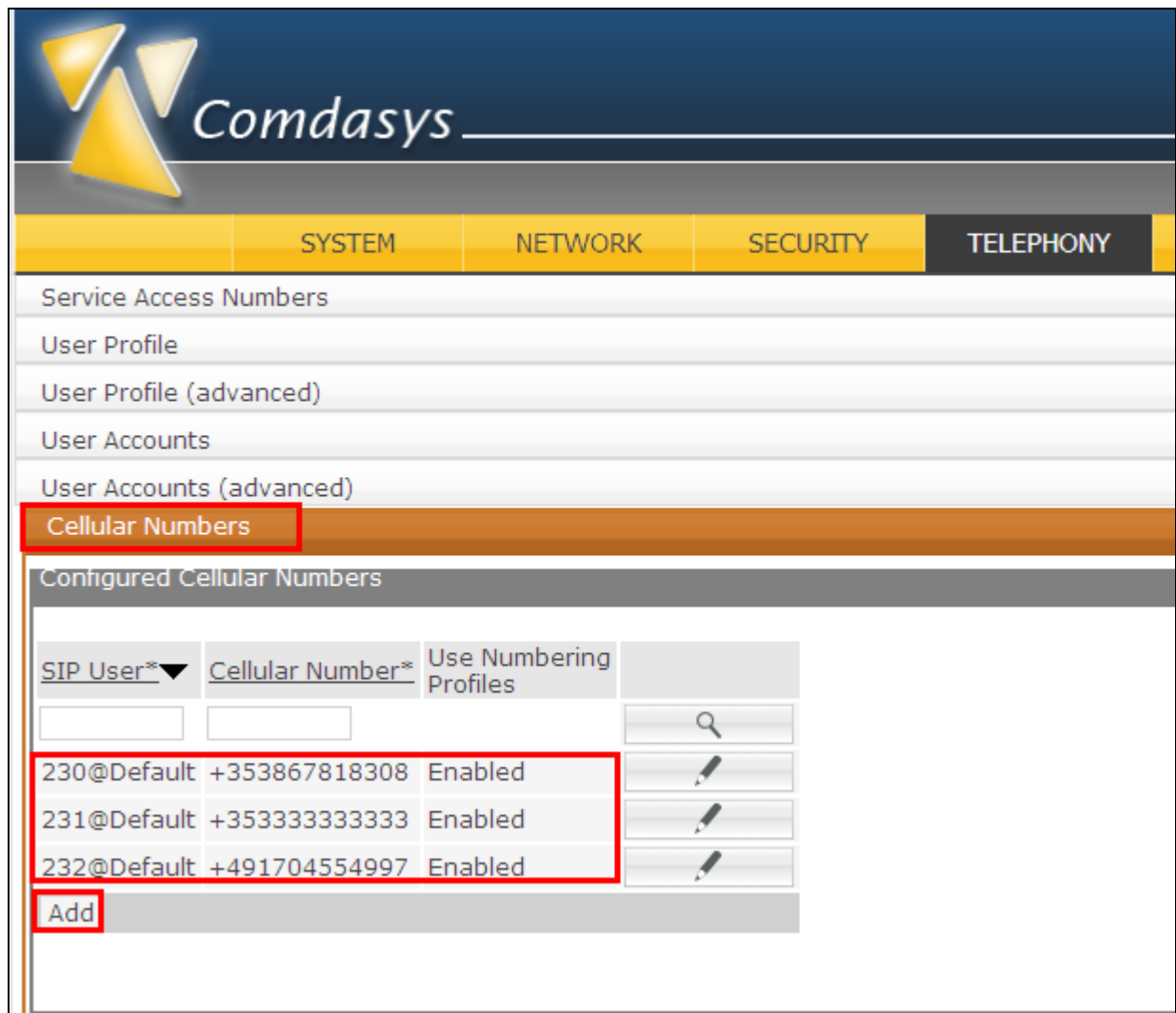
Add




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).



SIP User*	Cellular Number*	Use Numbering Profiles	
230@Default	+353867818308	Enabled	
231@Default	+353333333333	Enabled	
232@Default	+491704554997	Enabled	

Add

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).

The screenshot shows the Comdasys web interface. At the top is the Comdasys logo. Below it is a navigation bar with tabs: SYSTEM, NETWORK, SECURITY, TELEPHONY, FEATURES, UC, and DEPLOYMENT. The DEPLOYMENT tab is selected. Under DEPLOYMENT, there is a sub-menu with 'OTA Profiles' and 'Client Deployment'. 'Client Deployment' is selected. Below this is a 'Deploy' section. It contains a table with 'User Accounts' and a list of accounts: 230@Default, 231@Default, and 232@Default. Below the table is a 'Deploy Type' dropdown menu set to 'Redirect-Server'. At the bottom, there is a field for 'Optional 4 Digit Redirect-Server PIN'.

User Accounts
230@Default
231@Default
232@Default

Deploy Type: Redirect-Server

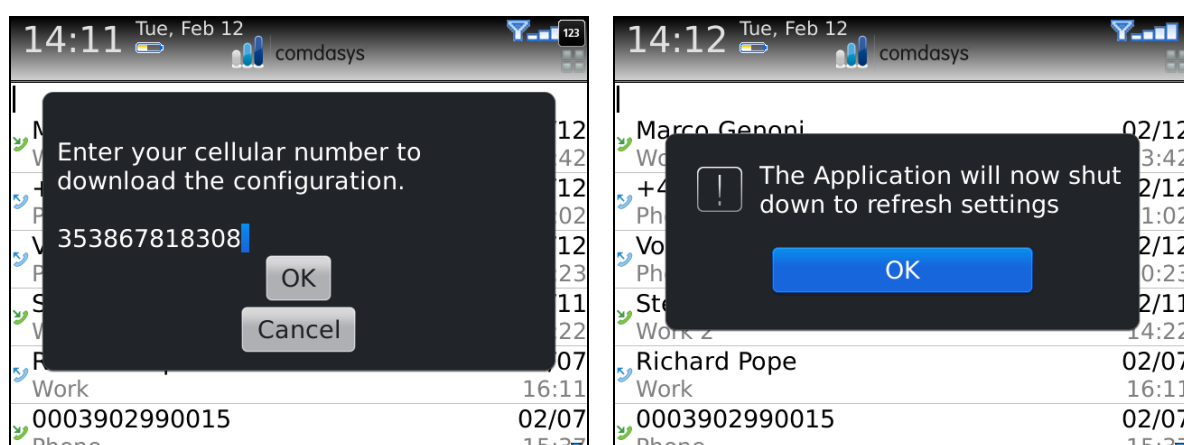
Optional 4 Digit Redirect-Server PIN:

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:

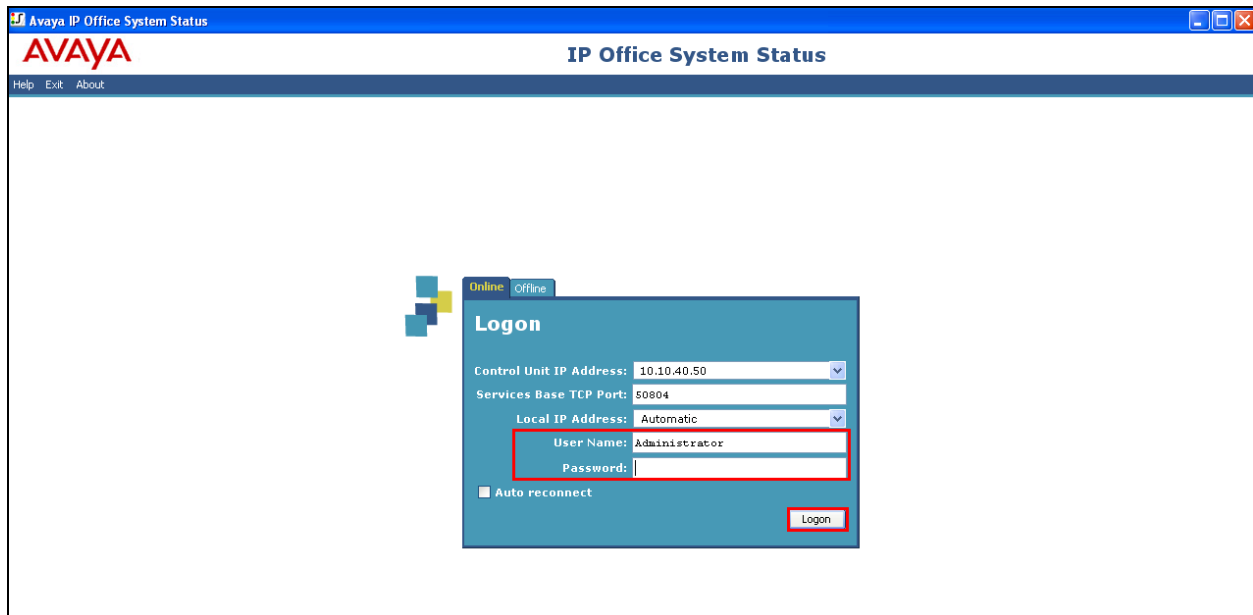
	Successful registered for VoIP - calls routed via WIFI or 3G Data
	Successful registered “InfoMode” - calls routed via cellular network
	Connected to network / registration in progress
	Registration failed / connection refused
	Offline - no WIFI / packet data connection active / available, calling possible 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.

The screenshot shows the Avaya IP Office System Status window. The left sidebar has a tree view with 'System' expanded, and 'VoIP Trunks (1)' selected, with 'Line: 17' highlighted. The main area shows the 'SIP Trunk Summary' for Line 17. The summary includes fields for Peer Domain Name, Resolved Address, Line Number, Number of Administered Channels, Number of Channels in Use, Administered Compression, Silence Suppression, SIP Trunk Channel Licenses, SIP Trunk Channel Licenses in Use, and SIP Device Features. A green circle indicates 0% utilization. Below the summary is a table with 10 rows and 14 columns. The table headers are: Cha..., U., Call Ref, Curr..., Time in State, R..., C..., Con..., Caller ID ..., Other Party o..., Dire..., Roun d, Rec..., Rec..., Tra..., Tra... The table contains 10 rows of data, all with 'Idle' state and '23:34:44' time.

IP Office System Status

Help Snapshot LogOff Exit About

System

- Memory Cards
- Control Unit (IP500 V2)
- VoIP Trunks (1)**
 - Line: 17**
- H.323 Extensions
- SIP Extensions
- Alarms (8)
- Extensions (21)
- Trunks (9)
- Active Calls
- Resources
- Voicemail
- IP Networking

SIP Trunk Summary

Peer Domain Name: 10.10.16.107
Resolved Address: 10.10.16.107
Line Number: 17
Number of Administered Channels: 10
Number of Channels in Use: 0
Administered Compression: G729 A, G711 Mu, G7231, G711 A
Silence Suppression: Off
SIP Trunk Channel Licenses: Unlimited
SIP Trunk Channel Licenses in Use: 0
SIP Device Features: REFER (Incoming and Outgoing), UPDATE (Incoming and Outgoing)

0%

Cha...	U.	Call Ref	Curr...	Time in State	R...	C...	Con...	Caller ID ...	Other Party o...	Dire...	Roun d	Rec...	Rec...	Tra...	Tra...
1			Idle	00:26:19											
2			Idle	23:34:44											
3			Idle	23:34:44											
4			Idle	23:34:44											
5			Idle	23:34:44											
6			Idle	23:34:44											
7			Idle	23:34:44											
8			Idle	23:34:44											
9			Idle	23:34:44											
10			Idle	23:34:44											

Trace Trace All Pause Ping Call Details Print... Save As...

18:05:45 Online

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.

The screenshot displays the Avaya IP Office System Status application. The left-hand navigation pane shows the 'System' menu expanded, with 'SIP Extensions' and 'Standard SIP Endpoints' selected. The extension number '231' is highlighted. The main pane shows the 'Extension Status' for extension 231, with various fields and a table at the bottom.

Extension Status

Extension Number:	231
IP address:	10.10.16.108
User Agent:	ComdasysB2BUA6.0.7
Telephone Type:	Unknown SIP Device
Current User Extension Number:	231
Current User Name:	Comdasys231
Forwarding:	Off
Twinning:	Off
Do Not Disturb:	Off
Message Waiting:	Off
Number of New Messages:	9
Phone Manager Type:	None
SIP Device Features:	REFER,UPDATE
License Reserved:	No
Last Date and Time License Allocated:	07/02/2013 12:58:10
Packet Loss Fraction:	
Jitter:	
Round Trip Delay:	
Connection Type:	
Codec:	
Remote Media Address:	

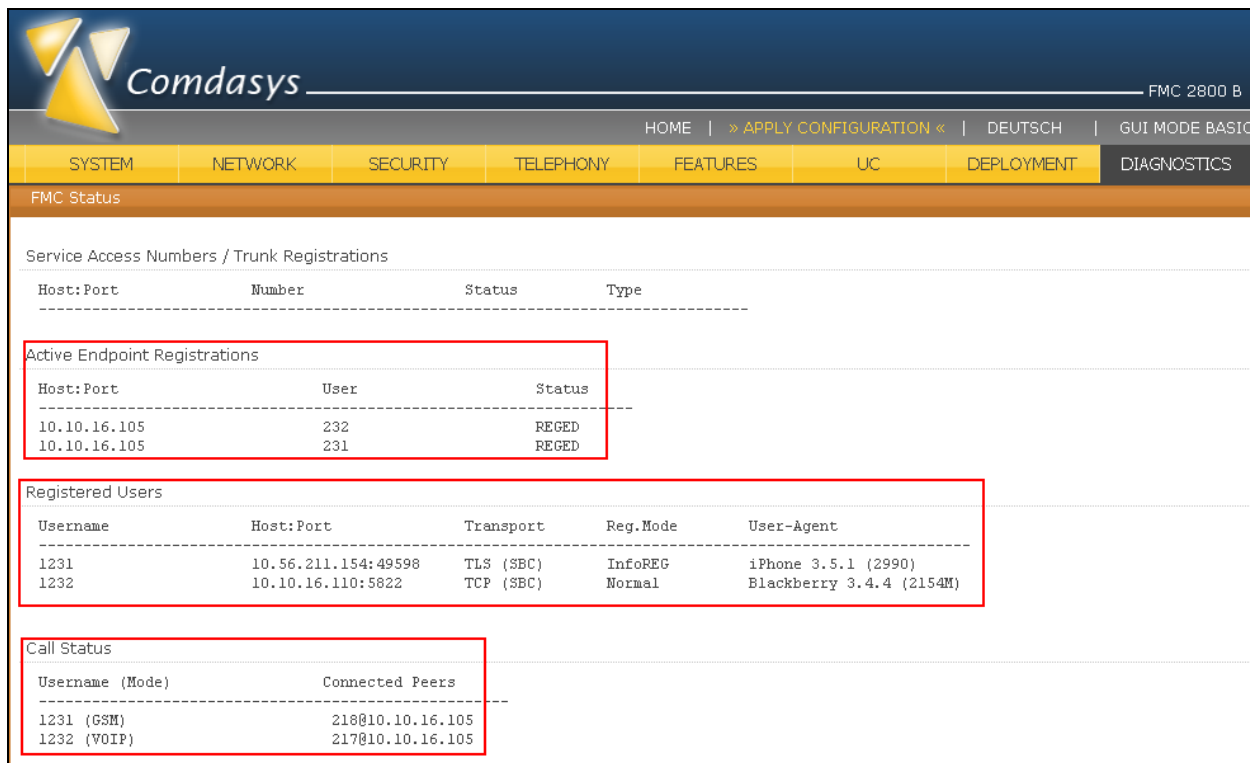
Call Ref	Current State	Time in State	Calling Number or Called Number	Direction	Other Party on Call
	Idle	05:06:15			

Buttons: Trace, Trace All, Pause, Ping, Call Details, Print..., Save As...

18:04:25 Online

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.



Comdasys FMC 2800 B

HOME | > APPLY CONFIGURATION < | DEUTSCH | GUI MODE BASIC

SYSTEM NETWORK SECURITY TELEPHONY FEATURES UC DEPLOYMENT **DIAGNOSTICS**

FMC Status

Service Access Numbers / Trunk Registrations

Host:Port	Number	Status	Type

Active Endpoint Registrations

Host:Port	User	Status

10.10.16.105	232	REGED
10.10.16.105	231	REGED

Registered Users

Username	Host:Port	Transport	Reg.Mode	User-Agent

1231	10.56.211.154:49598	TLS (SBC)	InfoREG	iPhone 3.5.1 (2990)
1232	10.10.16.110:5822	TCP (SBC)	Normal	Blackberry 3.4.4 (2154M)

Call Status

Username (Mode)	Connected Peers

1231 (GSM)	218@10.10.16.105
1232 (VOIP)	217@10.10.16.105

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/

©2014 Avaya Inc. All Rights Reserved.

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

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.