Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office 500 v2 R8.1 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 with the following observations:

• Where FMC1 (iMC Client 1) calls Extn1 (on IPO) which subsequently performs a blind transfer to FMC2 (MC Client 2), upon transfer the call on FMC1 ends and the call on FMC2 continues to ring for around 5 seconds.
• Where FMC1 is configured with Call Forward to any extension, and FMC1 is called the forwarding does not take place. The Mobile Convergence Solution uses the SIP 302 Moved Temporarily method for forwarding which the IP Office does not support when used by SIP users. Only IP Office SIP Trunks support the 302 Moved Temporarily method of forwarding.
• Where Extn1 calls FMC1 which is in GSM mode, the call can only be rejected by the phone OS. The data part of the call is still in progress on the FMC1 iMC client, although Extn1 has been rejected and diverted to cellular voicemail as expected. The call can be ended from either the FMC1 iMC client or Extn1.

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 H323 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
The telephone numbers used for testing are shown in the following table.

<table>
<thead>
<tr>
<th>Endpoint</th>
<th>Ext</th>
<th>PSTN Number</th>
<th>Station Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Extn1</td>
<td>217</td>
<td></td>
<td>H323 Endpoint on IPO</td>
</tr>
<tr>
<td>Extn2</td>
<td>218</td>
<td></td>
<td>H323 Endpoint on IPO</td>
</tr>
<tr>
<td>FMC1</td>
<td>231</td>
<td>00353867818305</td>
<td>SIP Endpoint on IPO associated with Apple iPhone with iMC Client</td>
</tr>
<tr>
<td>FMC2</td>
<td>232</td>
<td>00353867818308</td>
<td>SIP Endpoint on IPO associated with Blackberry 8900 with MC Client</td>
</tr>
<tr>
<td>Call through</td>
<td>n/a</td>
<td>0035391482464</td>
<td>Service Access Number on Mobile Convergence Solution</td>
</tr>
</tbody>
</table>

4. Equipment and Software Validated
The following equipment and software were used for the sample configuration provided:

<table>
<thead>
<tr>
<th>Equipment/Software</th>
<th>Release/Version</th>
</tr>
</thead>
<tbody>
<tr>
<td>Avaya IP Office 500 V2</td>
<td>R8.1 (57)</td>
</tr>
<tr>
<td>Avaya IP Office Manager</td>
<td>10.1 (52)</td>
</tr>
<tr>
<td>Avaya 4621 SW IP (H.323) Telephone</td>
<td>2.9.2</td>
</tr>
<tr>
<td>Avaya 4620 SW IP (H.323) Telephone</td>
<td>2.9.2</td>
</tr>
<tr>
<td>Comdasys Mobile Convergence Controller</td>
<td>10684.16.5</td>
</tr>
<tr>
<td>Blackberry Curve 8900</td>
<td>V5.0.0.681 Comdasys MC Client 3.1.0 Build:#1607M</td>
</tr>
<tr>
<td>Apple iPhone</td>
<td>iOS 6.0 Comdasys iMC Client v3.5.1 (2990)</td>
</tr>
</tbody>
</table>

Testing was performed with IP Office 500 R8.1, but it also applies to IP Office Server Edition R8.1. Note that IP Office Server Edition requires an Expansion IP Office 500 v2 R8.1 to support analog or digital endpoints or trunks.
5. Configure Avaya IP Office 500 v2

All configuration steps for Avaya IP Office were performed using the IP Office Manager application. This application presents the administrator with a hierarchy of icons for configuring various components, as shown below.
5.1. Configure System Settings

Select **System** from the hierarchy 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

![LAN Settings Table]

IP Address: 10.10.16.105
IP Mask: 255.255.255.0
Primary Trans. IP Address: 0.0.0.0
RIP Mode: None
Enable NAT: unchecked
Number Of DHCP IP Addresses: 200
DHCP Mode: Server

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

- **SIP Trunks Enable**
- **SIP Registrar Enable**

<table>
<thead>
<tr>
<th>VoIP Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td><img src="image_url" alt="Image of VoIP settings" /></td>
</tr>
</tbody>
</table>

- **H.323 Gatekeeper Enable**
- **SIP Trunks Enable**
- **SIP Registrar Enable**

<table>
<thead>
<tr>
<th>RTP Port Number Range</th>
</tr>
</thead>
<tbody>
<tr>
<td>Port Range (Minimum)</td>
</tr>
<tr>
<td>Port Range (Maximum)</td>
</tr>
</tbody>
</table>

**DiffServ Settings**

- 88 DSCP (Hex)
- 46 DSCP

**DHCP Settings**

- Primary Site Specific Option Number (SSON) 176
Click the **SIP Registrar** sub tab and configure the following:
- **Domain Name** – in this case the IP address of IP Office was used
- **Layer 4 Protocol** – ensure that *UDP* is an option
- **UDP Port** – must be configured as **5060**
- **Auto-create Extn/User** – place a tick in the check box

![SIP Registrar Configuration](image)

Click the **Voicemail** tab and configure voicemail accordingly. In this case IP Office **Embedded Voicemail** was configured.

![Voicemail Configuration](image)
5.2. Configure Default Route

A default route must be configured for the IP network routing. Click IP Route from the hierarchy 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.

![IP Route Configuration Table]

- **IP Address:** 0.0.0.0.0
- **IP Mask:** 0.0.0.0
- **Gateway IP Address:** 10.10.16.1
- **Destination:** LAN1
- **Metric:** 0
- **Proxy ARP:** unchecked
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 hierarchy 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.

<table>
<thead>
<tr>
<th>SIP Line</th>
<th>Transport</th>
<th>SIP URI</th>
<th>VoIP</th>
<th>T38 Fax</th>
<th>SIP Credentials</th>
</tr>
</thead>
<tbody>
<tr>
<td>Line Number</td>
<td>17</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>ITSP Domain Name</td>
<td>10.10.16.107</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Prefix</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>National Prefix</td>
<td>0</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Country Code</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>International Prefix</td>
<td>00</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Call Routing Method</td>
<td>Request URI</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>In Service</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Use Tel URI</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Check OOS</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Name Priority</td>
<td>System Default</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Caller ID from From header</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Send From In Clear</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>User-Agent and Server Headers</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Send Caller ID</td>
<td>None</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Association Method</td>
<td>By Source IP address</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>REFER Support</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
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 hierarchy and click New. Configure as follows:

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

![User Configuration Table]

**Example Configuration:**

- **Name**: Comdasys230
- **Password**: ******
- **Confirm Password**: ******
- **Full Name**: Comdasys230
- **Extension**: 230
- **Profile**: Basic User
- **System Phone Rights**: None
- **Locale**: 
- **Priority**: 5
- **Flare Mode**: Standalone

*Note: The table above shows a sample configuration with the highlighted fields filled in.*
Click on the Voicemail tab and check the **Voicemail On** box to enable voicemail for this user.

![Voicemail Settings Table]

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.

![Telephony Call Settings Table]
Click on the **Supervisor Settings** sub tab and configure the **Login Code**, this is used when registering the SIP endpoint.

![Login Code Configuration](image)

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.

![SIP Extension Selection](image)
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 hierarchy 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 by and handled accordingly by the Comdasys Mobile Convergence Solution. Right click on Short Code in the hierarchy 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 Configuration](image)
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 hierarchy 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

<table>
<thead>
<tr>
<th>Standard</th>
<th>Voice Recording</th>
<th>Destinations</th>
</tr>
</thead>
<tbody>
<tr>
<td>Bearer Capability</td>
<td>Any Voice</td>
<td></td>
</tr>
<tr>
<td>Line Group ID</td>
<td>9</td>
<td></td>
</tr>
<tr>
<td>Incoming Number</td>
<td>091482464</td>
<td></td>
</tr>
<tr>
<td>Incoming Sub Address</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Incoming CLI</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Locale</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Priority</td>
<td>1 - Low</td>
<td></td>
</tr>
<tr>
<td>Tag</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Hold Music Source</td>
<td>System Source</td>
<td></td>
</tr>
</tbody>
</table>

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 hierarchy 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 sent to line.
- **Line Group ID** – select the PSTN Line Number, in this case 9

![Short Code Configuration](image)

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.

![LAN Interface Configuration](image)
Click **NETWORK \(\rightarrow\) Virtual Interfaces \(\rightarrow\) Add Interface**, select **LAN1** from the drop down list, enter a valid **VLAN ID**, **IP address** and **Netmask** and click on **✓** to commit. The screen below will be displayed.
6.2. 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 administrators 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.
### 6.3. Configure SIP Options

The Mobile Convergence Controller is configured with a public IP address for access by MC clients in GSM mode over a 3G data connection. In this case the enterprise firewall is configured to port forward to relevant ports to the Mobile Convergence Controller. Click **TELEPHONY** → **Global Settings** → **SIP Options** and enter the **External IP for NAT** according to the implementation.

![Comdasys SIP Options Configuration](image)

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Registration Timeout Interval</td>
<td></td>
</tr>
<tr>
<td>Registration Retry Interval</td>
<td></td>
</tr>
<tr>
<td><strong>External IP for NAT</strong></td>
<td>86.47.122.3</td>
</tr>
<tr>
<td>Attempt Near-End NAT Pinhole</td>
<td>☑</td>
</tr>
<tr>
<td>TCP keepalive Interval</td>
<td></td>
</tr>
<tr>
<td>SBC call admission control</td>
<td></td>
</tr>
<tr>
<td>Time interval for registrations</td>
<td></td>
</tr>
<tr>
<td>Max value of simultaneous registrations</td>
<td></td>
</tr>
</tbody>
</table>
6.4. Configure Port Settings
Click TELEPHONY → Port Settings → SBC to configure the Receiving TLS port the Mobile Convergence Controller will use for TLS SIP registrations. It is occasionally necessary to change this where cellular providers block SIP traffic on the traditional SIP TLS 5061 port. This is the port which will be used by the MC client to register with the Mobile Convergence Controller when in GSM mode.

![Port Settings Configuration](image-url)
6.5. Configure SBC Internal Networks

The internal network subnet must be configured to define which networks should/shouldn’t be processed by the SBC feature of the Mobile Convergence Controller. Click TELEPHONY → SBC Internal networks → Add and enter the internal subnet and mask, click on ☑️ to commit, as shown below.
6.6. 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, as shown below.
6.7. 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 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.
6.8. 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 6.7

**Port** – choose a local port on the Mobile Convergence Controller side for establishing the SIP trunk. Click on ✅ to commit.

![Comdasys](image)
6.9. 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 6.7
- **From Converter Profile** - select the numbering profile configured in Section 6.6
- **SIP Trunk** – configure the SIP trunk endpoint configured in Section 6.7
- **Country** – configure accordingly
- **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.
6.10. 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.
6.11. Configure SIP Registrations

Each Mobile Convergence Controller user requires a SIP registration. Click TELEPHONY → Registrations → 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
- **PBX** – select a PBX Profile as configured in Section 6.9

Click on **✓** to commit. Perform this task for each user required.
6.12. Configure SIP User Accounts

In addition to the IP Office user, each MC client must use a different account to register to the Mobile Convergence Controller. It can be any username (numbers and/or letters). It will never show up on the PBX side. In this instance, the IP Office SIP User extension number was prefixed with a 1. Click TELEPHONY → User Accounts → Add and configure as follows:

- **SIP Number** – enter an appropriate SIP extension number
- **SIP User Password** – enter an appropriate password
- **GSM Number** – enter the corresponding cell phone number
- **Registrations** – enter the IP Office user to map to

Click on **✔** to commit. Perform this task for each user required.
6.13. Configure User Account Settings

Click on TELEPHONY → User Account Settings. Click ‍ to edit the relevant user and enable Call Waiting and Subscribe MWI, to enable multiple simultaneous calls to be delivered to the MC client and IP Office voicemail message waiting indication on the MC client.

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 6.6. A user might have more than one mobile number, however only one can be active at the same time.
6.15. Configure Profile List
An OTA Profile List must be configured on the Mobile Convergence Controller, click
DELPLOYMENT (not shown) → Feature/LCR/OTA Profile Lists → OTA Profile List →
Add and enter an identifying Name. Click on ✔ to commit.
6.16. Configure OTA Profile

OTA profiles are configured for both Default (TCP) and TLS connections to the Mobile Convergence Controller. Consult Comdasys documentation for more information. These profiles will be assigned to a User Group in the next Section.
6.17. Deploy User Groups

User groups are configured to define a set of common features and assigned to users in the next Section. Click **DEPLOYMENT ➔ Deploy User Groups ➔ Add** and configure as follows:

- **User group** – enter an identifying name for the user group
- **Controller Address** – select the IP address configured in **Section 6.3**
- **Port** – select the appropriate port as required, consult Comdasys documentation for more information
- **Callthrough number** – enter the DDI configured for the call-through feature
- **OTA settings** – select the appropriate profile configured in **Section 6.16**

Click on ✔ to commit.
6.18. Configure User Assignment

Click **DEPLOYMENT → Client Deployment → User Assignment**. The User Assignment is configured to associate the **User Group** and **Deskphone Number** with the SIP user configured for the MC client. Configure a **Deskphone Number**. This is the IP Office extension that is dialed when the MC client performs the Deskphone Handoff feature.

<table>
<thead>
<tr>
<th>User</th>
<th>User Group</th>
<th>Deskphone Number</th>
<th>Force Config Download</th>
</tr>
</thead>
<tbody>
<tr>
<td>1230</td>
<td>TLS</td>
<td></td>
<td>No</td>
</tr>
<tr>
<td>1231</td>
<td>TLS</td>
<td>218</td>
<td>No</td>
</tr>
<tr>
<td>1232</td>
<td>Default</td>
<td></td>
<td>No</td>
</tr>
</tbody>
</table>
6.19. 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 1321 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 6.19.

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

![Application Screenshots]

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

<table>
<thead>
<tr>
<th>Symbol</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>🟢🟢🟢</td>
<td>Successful registered for VOIP - calls routed via WIFI or 3G Data</td>
</tr>
<tr>
<td>🟢🟢</td>
<td>Successful registered “InfoMode” - calls routed via cellular network</td>
</tr>
<tr>
<td>🟢🟢</td>
<td>Connected to network / registration in progress</td>
</tr>
<tr>
<td>🟢🔴</td>
<td>Registration failed / connection refused</td>
</tr>
<tr>
<td>🟢⚪️</td>
<td>Offline - no WIFI / packet data connection active / available, calling possible via cellular but without features</td>
</tr>
</tbody>
</table>
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
Using the IP Office System Status Application, 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

Using 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.

![IP Office System Status](image)

**Extension Status**
- **Extension Number**: 231
- **IP address**: 10.10.16.103
- **User Agent**: ComdasysIPO8.6.0.7
- **Telephone Type**: Unknown SIP Device
- **Current User Extension Number**: 231
- **Current User Name**: ComdasysIPO
- **Forwarding**: Off
- **T mourning**: Off
- **Do Not Disturb**: Off
- **Message Waiting**: Off
- **Number of New Messages**: 9
- **Application**: 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**: Idemp
- **Time In State**: 05:06:15
- **Calling Number or Called Number**: 0
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 Mobile Convergence Controller Active Endpoint Registrations](image)

**9. Conclusion**

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

**10. Additional References**

Product documentation for Avaya products may be found at [http://support.avaya.com](http://support.avaya.com)


[2] Comdasys Mobile Convergence Client Manuals:
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