



Avaya Solution & Interoperability Test Lab

Application Notes for configuring Ascom Myco V9.3 with Avaya IP Office Server Edition R10.1 - Issue 1.0

Abstract

These Application Notes describe the configuration steps for provisioning Ascom's Myco smart device to interoperate with Avaya IP Office.

Readers should pay attention to Section 2, in particular the scope of testing as outlined in Section 2.1 as well as any observations noted in Section 2.2, to ensure that their own use cases are adequately covered by this scope and results.

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

1. Introduction

These Application Notes describe the configuration steps for provisioning Ascom Myco wireless smartphones V9.3 to interoperate with Avaya IP Office R10.1. The Avaya IP Office consists of an IP Office Server Edition running on a virtual platform as the primary server with an IP Office IP500 V2 running as the secondary expansion server. Ascom Myco wireless smartphones are configured on Avaya IP Office as SIP users, therefore enabling them to make/receive internal and PSTN/external calls and have other telephony facilities available on Avaya IP Office. The wireless communication is made using a wireless router connected to the same LAN as the Avaya IP Office.

Note: Ascom Myco may be referred to as Myco, Myco handset or Myco smart device throughout this document. These names all refer to the same product, a smart phone that is connected to IP Office by registering as a third-party SIP extension.

The Avaya IP Office solution consists of a primary server and an IP500V2 expansion. Both systems are linked by IP Office Line IP trunks that can enable voice networking across these trunks to form a multi-site network. Each system in the solution automatically learns each other's extension numbers and user names. This allows calls between systems and support for a range of internal call features.

2. General Test Approach and Test Results

The general test approach was to configure Ascom Myco Smartphones to communicate with IP Office as implemented on a customer's premises. The interoperability compliance testing evaluates the ability of the Ascom Myco to make and receive calls to and from Avaya H.323, SIP and Digital deskphones as well as PSTN endpoints. The integrated IP Office Voicemail was used to test for DTMF and Message Waiting Indication (MWI) on the Myco Smartphones. See **Figure 1** for a network diagram. The interoperability compliance test included both feature functionality and serviceability tests.

Note: Ascom Myco handsets were registered to both the Primary and Secondary servers but not simultaneously, i.e., two handsets were registered to the Server Edition primary server and two were registered to the IP 500 V2 expansion.

Note: The Ascom Myco smart device can be set up to use Wi-Fi, GSM or both. For compliance testing only Wi-Fi was used and a wireless router was used to provide a network connection. This wireless router was considered a part of the member's overall solution

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.

Avaya's formal testing and Declaration of Conformity is provided only on the headsets/Smartphones that carry the Avaya brand or logo. Avaya may conduct testing of non-Avaya headset/Smartphone to determine interoperability with Avaya phones. However, Avaya does not conduct the testing of non-Avaya headsets/Smartphones for: Acoustic Pressure, Safety, Hearing Aid Compliance, EMC regulations, or any other tests to ensure conformity with safety, audio quality, long-term reliability or any regulation requirements. As a result, Avaya makes no representations whether a particular non-Avaya headset will work with Avaya's telephones or with a different generation of the same Avaya telephone.

Since there is no industry standard for Smartphone interfaces, different manufacturers utilize different Smartphone/headset interfaces with their telephones. Therefore, any claim made by a headset vendor that its product is compatible with Avaya telephones does not equate to a guarantee that the headset will provide adequate safety protection or audio quality.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with these Application Notes, the interface between Avaya systems and Ascom Myco did not include use of any specific encryption features as requested by Ascom.

Note: Compliance testing was carried out using TCP as the transport for signaling, a selection of basic calls and transfer calls were carried out using UDP.

2.1. Interoperability Compliance Testing

The testing included:

- Registration/Invalid Registration
- Basic calls/PSTN calls
- Blind Transfer, Supervised Transfer and 3 Party Conference
- Feature calls
- Call Alerting and Call Waiting
- Mobile Twinning
- Call forwarding unconditional, no reply, busy
- DTMF support, voicemail and MWI
- Codec support
- Serviceability testing

2.2. Test Results

Tests were performed to ensure full interoperability between Ascom Myco Wireless Smartphones and IP Office. The tests were all functional in nature and performance testing was not included. The following observations were noted during testing.

- For Call Waiting functionality to work on Myco, ensure that the **Call Waiting On** box seen under the Telephony tab of User is checked.
- Ascom Myco does not support local call diversion like Call Forward All, Call Forward Busy and Call Forward No Answer.
- The SIP Expires timer of 180 seconds is hard coded on IP Office.

The following issues were found during compliance testing.

1. **Blind Transfer using Avaya H323**

Myco1 – Avaya H323 (BlindTrn) – Myco2, failure to transfer if both Myco sets are on the same IPO side. The issue presents itself when Avaya cancels the call to Myco2 and then the initial caller (Myco1) tries to setup the new call immediately, Myco2 rejects the new call setup stating ‘486 Busy Here’. Ascom are investigating this issue (MYCO-9213).

2. **Incorrect CLID displayed during ringing on Supervised Transfer**

When Avaya H323 phones make the transfer, the consulting call shows the incorrect number upon ringing; it updates correctly when the call is answered. The ringing number should show the B-party number but it incorrectly shows the A-party number. When Avaya SIP phone makes the transfer the “call back” is to the B-Party, when the Avaya H323 phone makes the transfer the “call back” is to the A-Party. Avaya are investigating this issue.

3. **Incorrect A-party CLID after transfer to Digital on IP500V2**

Myco 4 – Myco 3, supervised trn to Digital. When transferred to the digital set the A party was not updated after transfer. There is no INVITE being sent to the A party (Myco4) with a name update. This is the same for blind and supervised transfers. This behaviour is as per Avaya design.

2.3. Support

Technical support for the Ascom Myco product can be obtained through a local Ascom supplier or Ascom global technical support:

- Email: support@ascom.com or Help desk: +46 31 559450

3. Reference Configuration

Figure 1 illustrates the network topology used during compliance testing. The Avaya solution consists of an IP Office with the Ascom Myco smartphones configured as SIP users. Avaya SIP, H323 and Digital phones were configured on IP Office. PRI and SIP trunks were configured to simulate a connection to the PSTN. A wireless router was connected to the IP network to provide a connection for the Myco smartphones. IP Office Manager was used to manage IP Office.

A laptop on the network that can access and connect to the Ascom Device Manager is used to configure the Ascom Myco smartphones. The Ascom Unite Connectivity Manager (UniteCM) is the core software platform in the Ascom system.

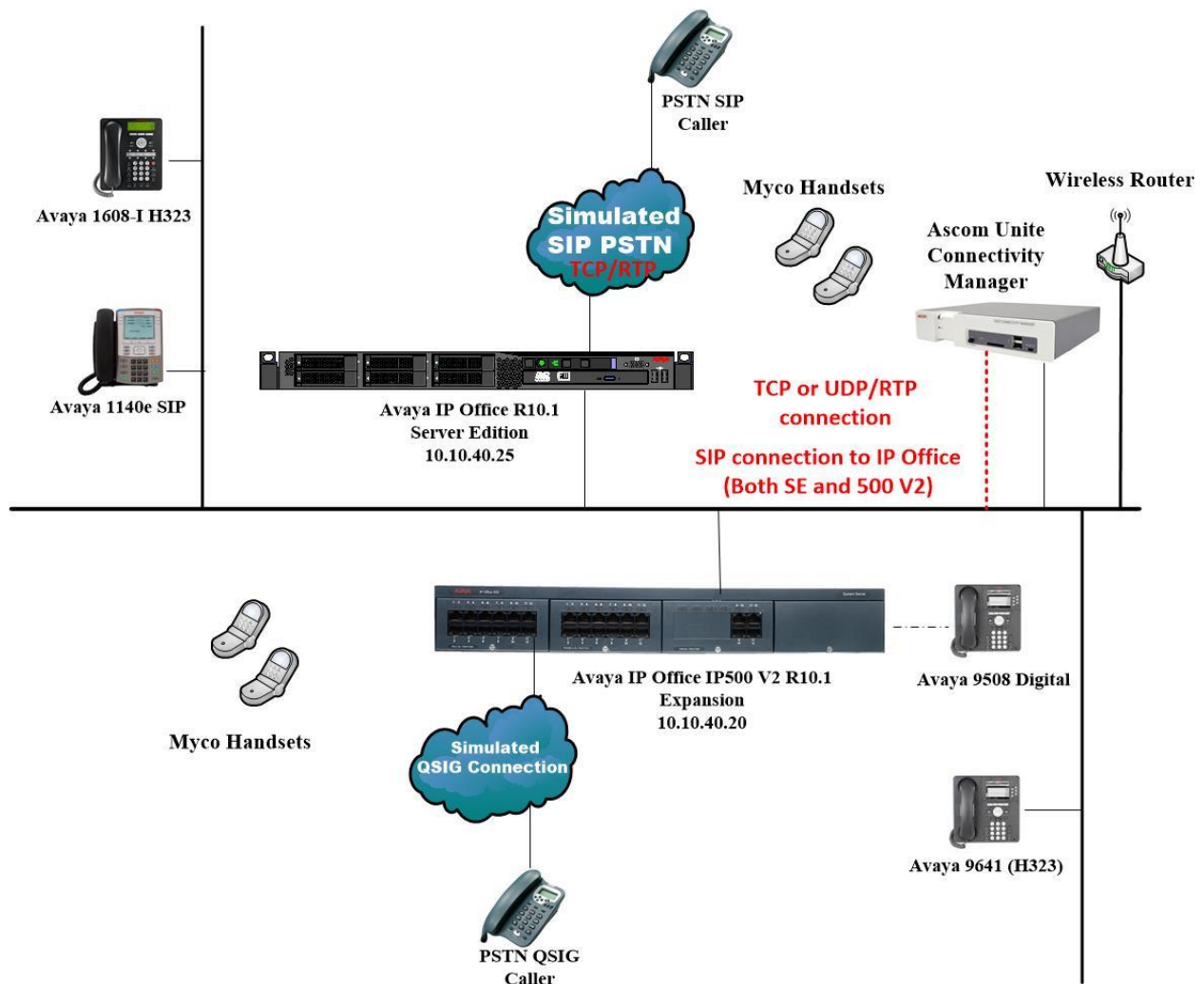


Figure 1: Connection of Ascom Myco with Avaya IP Office Server Edition with IP Office IP500 V2 expansion

4. Equipment and Software Validated

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

| Equipment/Software | Release/Version |
|--|------------------------------------|
| Avaya IP Office Server Edition running on a Virtual Platform | R10.1.0.0 Build 237 |
| Avaya IP Office 500 V2 | R10.1.0.0 Build 237 |
| Avaya IP Office Manager running on a Windows 7 PC | R10.1.0.0 Build 237 |
| Avaya 1608-I H323 Deskphone | 1608UA1_350B.bin |
| Avaya 9641G H323 Deskphone | R6.6115 |
| Avaya 1140e SIP Deskphone | R04.04.28.00 |
| Avaya 9508 Digital Deskphone | V0.6 |
| Ascom Myco Smart Device | Myco 1 & 2, v9.3 (SIP App v2.1) |
| Ascom Device Manager running on Ascom UniteCM | Unite DM/CM v5.8.1 |

Note: Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with IP Office Server Edition in all configurations.

5. Avaya IP Office Configuration

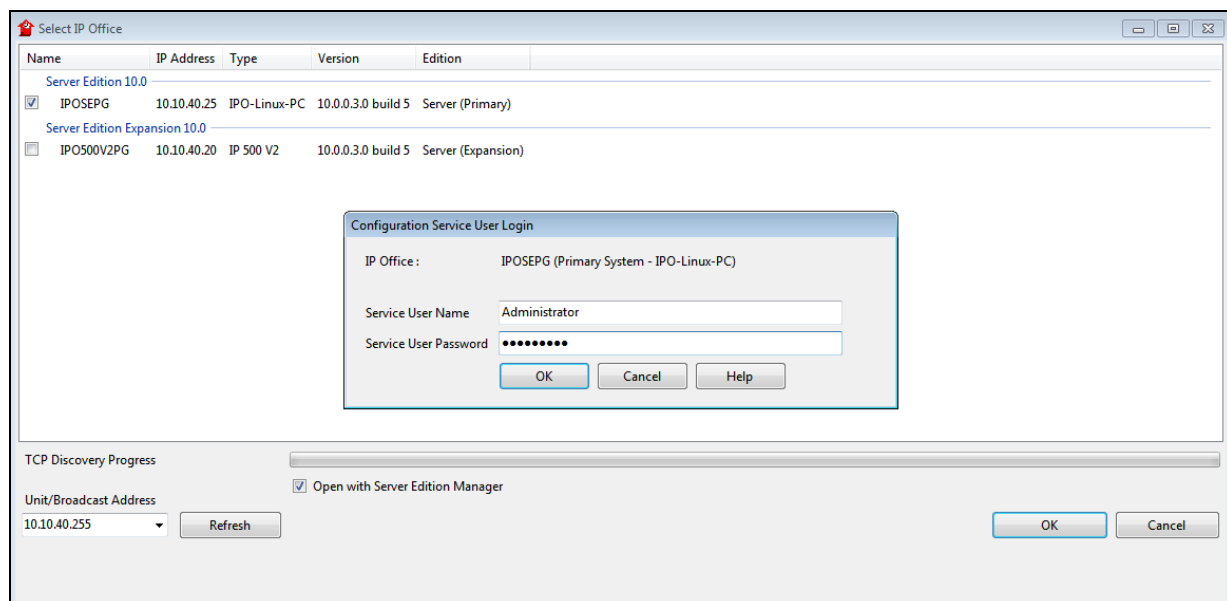
The document assumes that Avaya IP Office Server Edition has been installed and configured to work with an IP500 V2 expansion. This section only describes the details on how to configure both the IP Office Server Edition (Primary) and IP Office IP500 V2 (Expansion) to work with Ascom Myco. Configuration and verification operations on the Avaya IP Office illustrated in this section were all performed using Avaya IP Office Manager. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 9**. The configuration operations described in this section can be summarized as follows:

- Launch Avaya IP Office Manager (Administration)
- Display LAN Properties
- Create User
- Check Extension Properties
- Save Configuration

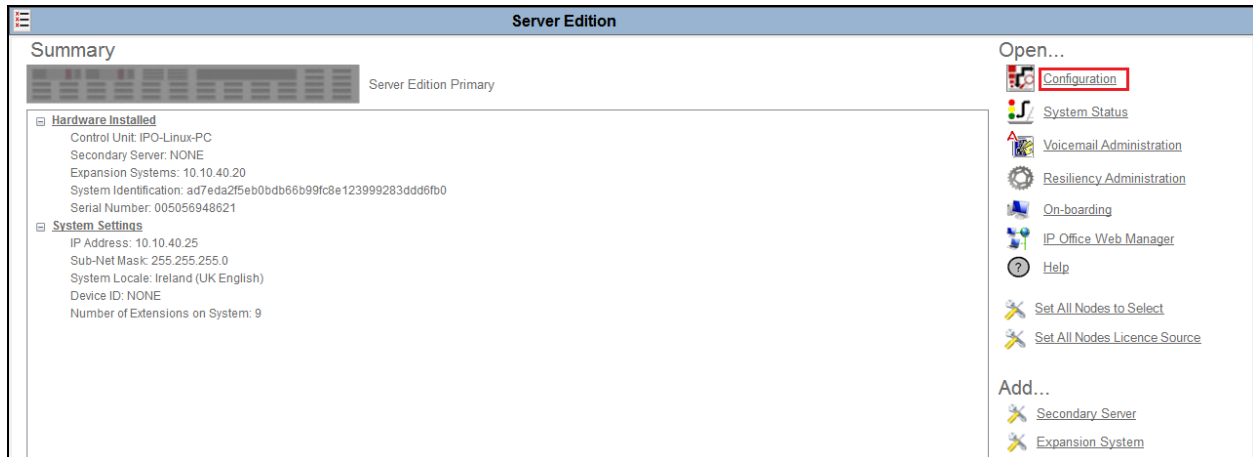
Note: Only the unique prompts are shown in the screen captures below, all other inputs can be left at default.

5.1. Launch Avaya IP Office Manager (Administration)

From the IP Office Manager PC, click **Start → Programs → IP Office → Manager** to launch the Manager application (not shown). Select the required Server Edition as shown below and enter the appropriate credentials. Click on the **OK** button.

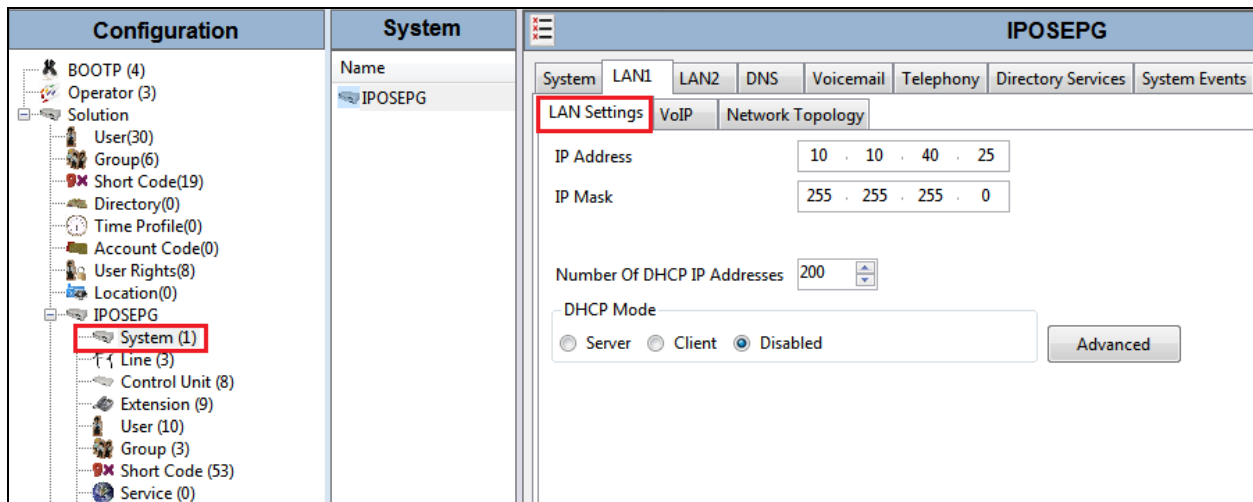


Click on **Configuration** at the top right of the page, as shown, to receive the IP Office configuration.



5.2. Display LAN Properties

From the left window navigate to **System** as shown and in the main window click on the **LAN1** tab and within that tab select the **LAN Settings** tab. The **IP Address** of the IP Office is shown and this will be required for setup in **Section 6.1**.



Within the **LAN1** tab, click on the **VoIP** tab. Ensure that **TCP** and **UDP** boxes are checked and that port **5060** is being used. During compliance testing **RTP-RTCP Keepalives** were set to **30secs**.

The screenshot shows the 'IPOSEPG' configuration window with the 'VoIP' tab selected. The 'LAN Settings' sub-tab is active. Key configurations include:

- H323 Gatekeeper Enable:** Checked.
- SIP Trunks Enable:** Checked.
- SIP Registrar Enable:** Checked.
- UDP and TCP:** Both are checked and highlighted with a red box. Their ports are set to 5060.
- TLS:** Checked, with port 5061.
- Challenge Expiry Time (secs):** 10.
- RTP Port Number Range:** Minimum 40750, Maximum 50750.
- Enable RTCP Monitoring on Port 5005:** Checked.
- RTCP collector IP address for phones:** 0.0.0.0.
- Keepalives Scope:** RTP-RTCP.
- Periodic timeout:** 30.
- Initial keepalives:** Enabled.

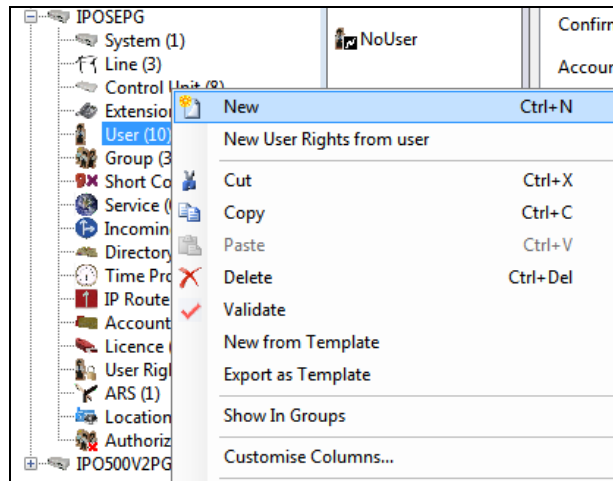
The Codec and DTMF settings can be changed under the **VoIP** tab as shown below.

The screenshot shows the 'VoIP' tab in the 'IPOSEPG' configuration window. Key settings include:

- Ignore DTMF Mismatch For Phones:** Checked.
- Allow Direct Media Within NAT Location:** Unchecked.
- RFC2833 Default Payload:** 101.
- Available Codecs:**
 - ☒ G.711 ULAW 64K
 - ☒ G.711 ALAW 64K
 - ☒ G.722 64K
 - ☒ G.729(a) 8K CS-ACELP
- Default Codec Selection:**
 - Unused:** (Empty list)
 - Selected:**
 - G.711 ALAW 64K
 - G.711 ULAW 64K
 - G.722 64K
 - G.729(a) 8K CS-ACELP

5.3. Create a new User

From the left window, right click on **User** and select **New**.



In the **User** tab add a **Name** and **Password** along with the **Extension**.

A screenshot of the '5180: 5180' user configuration page in the IPOSEPG interface. The page has a tabbed header with 'User' selected. The form contains the following fields and options:

- Name: 5180
- Password: masked with dots
- Confirm Password: masked with dots
- Unique Identity: empty
- Audio Conference PIN: empty
- Confirm Audio Conference PIN: empty
- Account Status: Enabled (dropdown)
- Full Name: MYCO SE 5180
- Extension: 5180
- Email Address: empty
- Locale: empty (dropdown)
- Priority: 5 (dropdown)
- System Phone Rights: None (dropdown)
- Profile: Basic User (dropdown)
- Receptionist: ☐
- Enable Softphone: ☐
- Enable one-X Portal Services: ☐
- Enable one-X TeleCommuter: ☐
- Enable Remote Worker: ☒
- Enable Communicator: ☒
- Enable Mobile VoIP Client: ☐
- Send Mobility Email: ☐
- Web Collaboration: ☐

Under the **Voicemail** tab, **Voicemail On** can be selected in order to provide voicemail to this user/extension.

5180: 5180

User Voicemail DND ShortCodes Source Numbers Telephony Forwarding Dial In Voice Recording Button Programming

Voicemail Code: ••••

Confirm Voicemail Code: ••••

Voicemail Email:

☒ Voicemail On

☐ Voicemail Help

☐ Voicemail Ringback

☐ Voicemail Email Reading

☐ UMS Web Services

☐ Enable GMAIL API

Voicemail Email:
☒ Off ☐ Copy ☐ Forward ☐ Alert

DTMF Breakout

Reception / Breakout (DTMF 0): System Default ()

Breakout (DTMF 2): System Default ()

Breakout (DTMF 3): System Default ()

Under the **Telephony** tab and **Call Settings** tab, **Call Waiting On** can be turned on/off depending on what is required by the user.

5180: 5180*

User Voicemail DND ShortCodes Source Numbers Telephony Forwarding Dial In Voice Recording Button Programming

Call Settings Supervisor Settings Multi-line Options Call Log TUI

Outside Call Sequence: Default Ring

Inside Call Sequence: Default Ring

Ringback Sequence: Default Ring

No Answer Time (secs): System Default (15)

Wrap-up Time (secs): 2

Transfer Return Time (secs): Off

Call Cost Mark-Up: 100

Advertise Callee State To Internal Callers: System Default (Off)

☒ Call Waiting On

☒ Answer Call Waiting On Hold

☐ Busy On Held

☐ Offhook Station

Under **Supervisor Settings** tab enter the password again for the **Login Code**. Ensure that **Force Login** box is checked.

5180: 5180*

User Voicemail DND ShortCodes Source Numbers **Telephony** Forwarding Dial In Voice Recording Button Programming

Call Settings **Supervisor Settings** Multi-line Options Call Log TUI

Login Code: •••• ☒ Force Login

Confirm Login Code: ••••

Login Idle Period (secs):

Monitor Group: <None>

Coverage Group: <None>

Status on No-Answer: Logged On (No change)

Privacy Override Group: <None>

Reset Longest Idle Time:

☒ All Calls ☐ External Incoming

☐ Force Account Code

☐ Force Authorization Code

☐ Incoming Call Bar

☐ Outgoing Call Bar

☐ Inhibit Off-Switch Forward/Transfer

☐ Can Intrude

☒ Cannot be Intruded

☐ Can Trace Calls

☐ Deny Auto Intercom Calls

Once **OK** is clicked at the bottom of the screen a new window should appear asking to create a new extension. Select **SIP Extension** as is shown below.

Note: If the system is not setup to auto-create extensions then a new extension can be added by right-clicking on Extension on the left window and selecting New, (not shown).

<User:0>: *

User Voicemail DND ShortCodes Source Numbers **Telephony** Forwarding Dial In Voice Recording Button Programming Menu Programming

Call Settings Supervisor Settings Multi-line Options Call Log TUI

Login Code: •••• ☒ Force Login

Confirm Login Code: ••••

Login Idle Period (secs):

Monitor Group: <None>

Coverage Group: <None>

Status on No-Answer: Logged On (No change)

Privacy Override Group: <None>

Reset Longest Idle Time:

☒ All Calls ☐ External Incoming

☐ Force Account Code

☐ Force Authorization Code

☐ Incoming Call Bar

☐ Outgoing Call Bar

☐ Inhibit Off-Switch Forward/Transfer

☐ Can Intrude

☒ Cannot be Intruded

☐ Can Trace Calls

Avaya IP Office Manager

Would you like a new VoIP extension created with this number?

☐ None

☐ H323 Extension

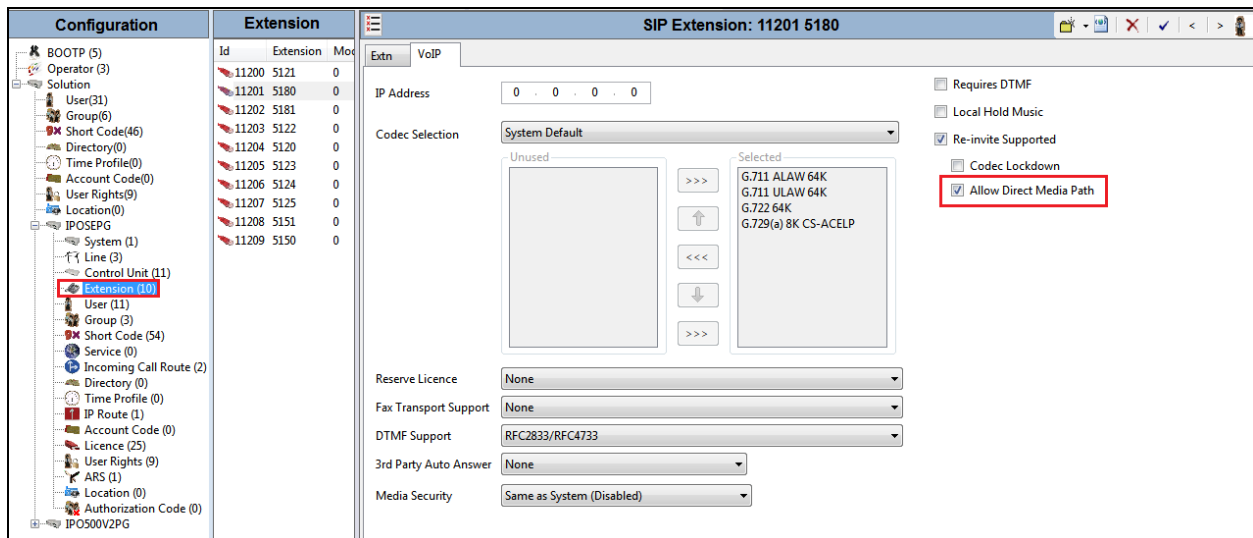
☒ SIP Extension

OK

OK Cancel Help

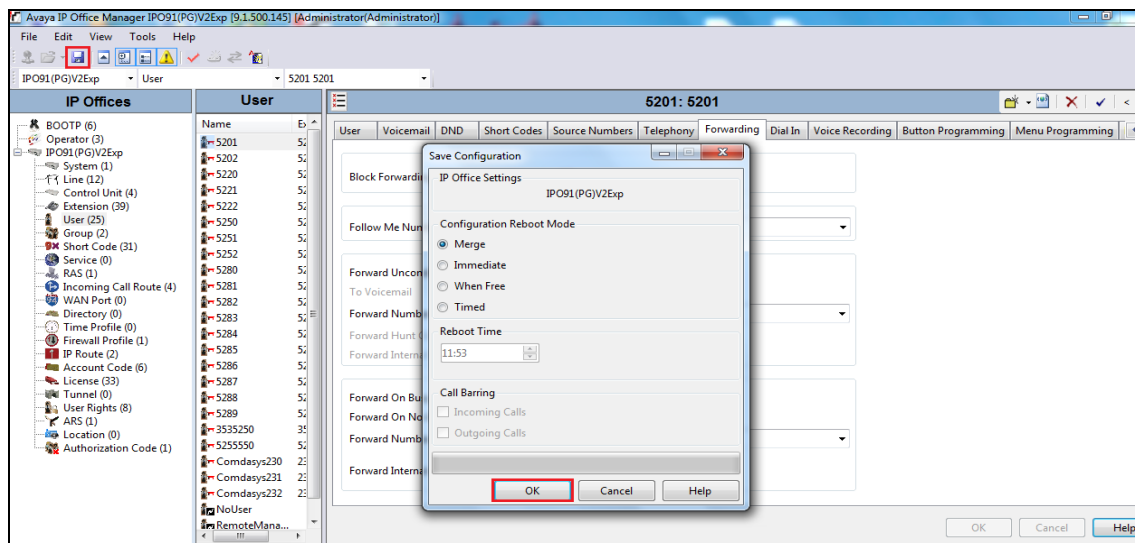
5.4. Check Extension Properties

Direct Media Path can be set on/off in the extension properties. This will allow RTP to be sent directly between devices. Once the SIP extension has been successfully created in **Section 5.3**, open the extension configuration to check to see if Allow Direct Signalling is selected. Select **Extension** in the left window and select the required extension number. In the main window under **VoIP** tab, **Allow Direct Media Path** can be checked or unchecked as shown below. Other settings such as **DTMF Support** and **Codec Selection** are possible to change here as well again if required by Ascom.



5.5. Save Configuration

Once all the configurations have been made it must be saved to IP Office. Click on the **Save** icon at the top of the screen and the following window appears, click on **OK** to commit the changes to memory.

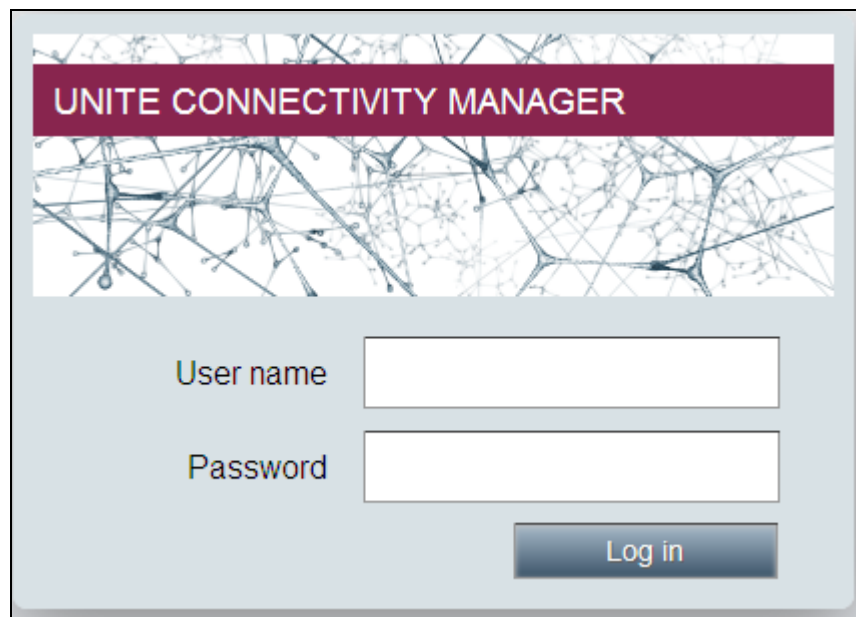


6. Configure Ascom Myco Wireless Smartphones

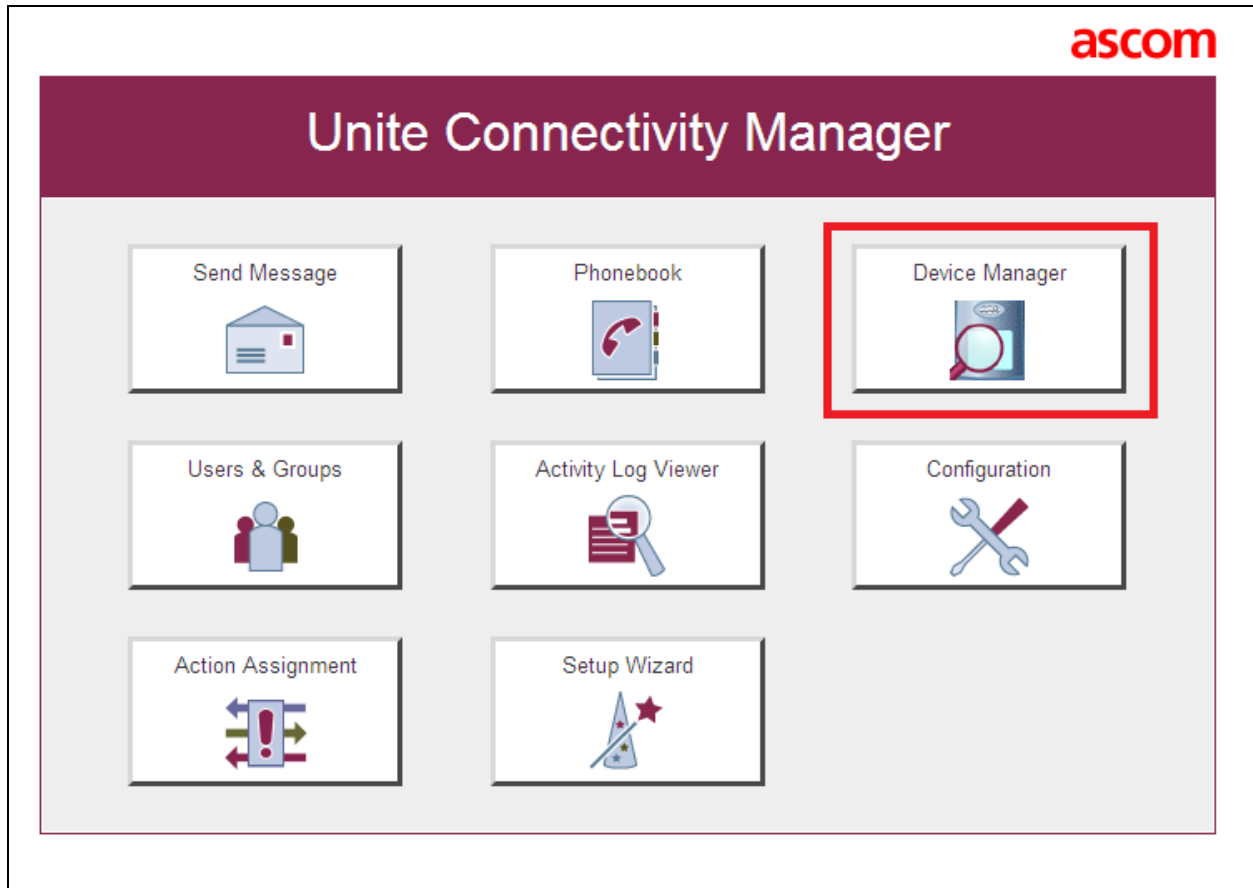
This section describes how to access and configure the Ascom Myco Wifi Smartphone via the Device Manager. It is implied that the Wifi network has been configured and operational and the Ascom UniteCM box has an IP address assigned.

Note: The Wireless router and Ascom UniteCM configuration are outside the scope of these Application Notes.

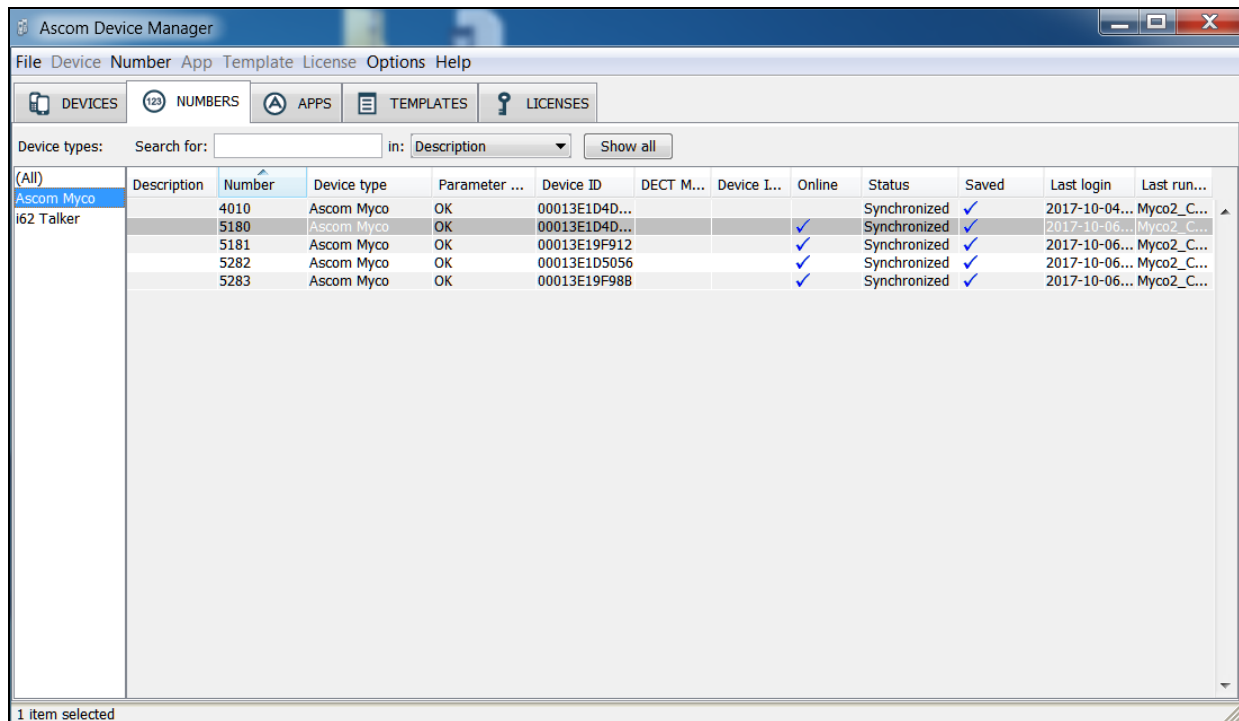
Access the UniteCM box by typing the URL, `http://<ip address>` in a web browser (not shown). Screen below shows the login screen. Enter the required credentials in the **User name** and **Password** fields and click on **Log in**.

The image shows a web-based login interface for the 'UNITE CONNECTIVITY MANAGER'. At the top, there is a dark red banner with the text 'UNITE CONNECTIVITY MANAGER' in white, set against a background of a complex network diagram. Below the banner, there are two input fields: one for 'User name' and one for 'Password'. To the right of the 'Password' field is a blue 'Log in' button with white text.

The main screen of Unite Connectivity Manager is seen as shown below. Click on the Device Manager application.



The **Ascom Device Manager** screen is seen as shown below. In the example below, a device with number **5180** is discovered. Double click on this number.



A close up of the same screen shown above shows that **5180** was selected.

| | | | | | | |
|-------------|---------|-------------|---------------|---------------|-----------|----------|
| 123 | NUMBERS | A | APPS | TEMPLATES | KEY | LICENSES |
| Search for: | | in: | Description | Show all | | |
| Description | Number | Device type | Parameter ... | Device ID | DECT M... | |
| | 4010 | Ascom Myco | OK | 00013E1D4D... | | |
| | 5180 | Ascom Myco | OK | 00013E1D4D... | | |
| | 5181 | Ascom Myco | OK | 00013E19F912 | | |
| | 5282 | Ascom Myco | OK | 00013E1D5056 | | |
| | 5283 | Ascom Myco | OK | 00013E19F988 | | |

The **Edit parameters for 5180** screen is displayed as shown below. Click on **Ascom VoIP** that is seen on the left hand side and configure the following values.

- **SIP Transport** Set to either **TCP** or **UDP** (for compliance testing TCP was selected as shown below)
- **Primary SIP Proxy** IP address of the IP Office to which this user is to be registered against
- **Listening Port** **5060**
- **SIP Register Expiration** **180** (This value is null and void as this is hardcoded to 180 by IP Office)
- **Endpoint ID** This is the extension number
- **Password** The password from the supervisor settings tab assigned to the endpoint in **Section 5.3**
- **Codec configuration** This will depend on the country (**G711 A-law** was used for compliance testing)
- **DTMF Type** **RFC 2833** is chosen
- **Replace Call Reject w/ User Busy** Defines whether calls are rejected using “603 Decline” or “486 Busy Here” messages

Retain default values for all other fields.

Device type:

| Name | Value |
|--------------------------------------|-------------|
| SIP Transport | TCP |
| Primary SIP proxy | 10.10.40.25 |
| Secondary SIP proxy | |
| Listening port | 5060 |
| SIP proxy ID | |
| SIP Register expiration | 180 |
| Endpoint ID | 5180 |
| Password | ***** |
| VoIP phone number | |
| Codec configuration | G711 A-law |
| Secure RTP mode | Off |
| DTMF type | RFC 2833 |
| Hold type | inactive |
| MOH locally | Yes |
| Direct signaling | No |
| Active mode during call | No |
| Replace Call Rejected with User Busy | Yes |
| Enable overlap dialing | No |

OK Cancel

The following step is optional. From the same screen as above, click on **Display** and configure the **Owner ID/name** field with the directory number configured, in this case **5180** as shown below. Retain default values for all other fields and click on **OK** to complete the configuration.

The screenshot shows the 'Edit parameters for 5180' window. The 'Device type' is set to 'Ascom Myco'. The left sidebar shows a tree view with 'Display' selected. The main table lists the following parameters:

| Name | Value |
|------------------------------|-----------|
| *Font size | Normal |
| *Owner ID/name | 5180 |
| *Sleep timeout | 1 minute |
| *Rotate top display | Off |
| Top display alert formatting | Automatic |

At the bottom right, there are 'OK' and 'Cancel' buttons.

The **Voicemail number** is set as ***66**, this is a short code on IP Office that was used to call to Voicemail. This may be different depending on the system.

The screenshot shows the 'Edit parameters for 5180' window. The 'Device type' is set to 'Ascom Myco'. The left sidebar shows a tree view with 'Voicemail' selected under 'Telephony'. The main table lists the following parameters:

| Name | Value |
|---------------------------|-------|
| Message centre number | 5180 |
| Voicemail number | *66 |
| Voicemail call clears MWI | No |
| Cellular voicemail number | |

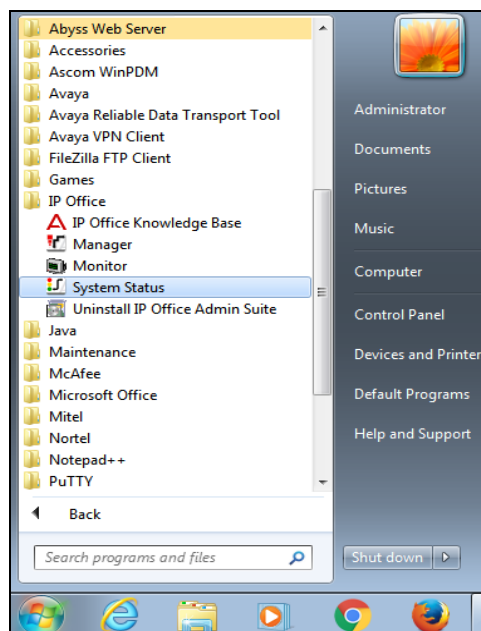
At the bottom right, there are 'OK' and 'Cancel' buttons.

7. Verification Steps

This section provides the tests that can be performed to verify correct configuration of the IP Office and Ascom solution. The best way to verify compliance is to make calls to/from the Ascom Myco handsets. Once calls can be made/received then this will show that the Myco handset is registered correctly with the IP Office and is capable of performing as designed. For further information on registration and call status the following section will be useful.

7.1. Verify the Ascom Myco Wireless Smartphone status

To verify the 'connection type' and the 'media security' IP Office System Status can be used to monitor each handset including the Myco handsets. Open **IP Office System Status** as shown below.



Connect to the required IP Office and enter the appropriate credentials then click on **Logon**.



Place a call to one of the Myco handsets and select the handset as shown below. Information on the call and the connection is displayed in the main window.

Note: The information shown below is intended as an example of what such a call should look like.

The screenshot shows the 'Avaya IP Office System Status' window. The left sidebar contains a tree view with categories like System, Alarms, Extensions, Trunks, Active Calls, Resources, Voicemail, IP Networking, and Locations. The 'Extensions' category is expanded, and extension 5180 is selected. The main pane displays the 'Extension Status' for 5180, including details like IP address, location, registrar, and various call settings. At the bottom, a table shows the current state of the extension.

| Call Ref | Current State | Time in State | Calling Number or Called Number | Direction | Other Party on Call |
|----------|---------------|---------------|---------------------------------|-----------|---------------------|
| 128 | Connected | 00:05:30 | 5181 | Incoming | Extn 5181, 5181 |

Information on the **Media Stream** and the **Layer 4 Protocol** are shown as well as the **Connection Type**. The display below shows a **Direct Media** call using **RTP** and **TCP**.

This image provides a detailed view of the 'Extension Status' for extension 5180. The 'Media Stream' is set to RTP, the 'Layer 4 Protocol' is set to TCP, and the 'Connection Type' is 'Direct Media'. The 'Codec' is 'G711 A' and the 'Remote Media Address' is '10.10.40.206'. These three items are highlighted with red boxes to emphasize the call configuration.

| Extension Status | |
|---------------------------------------|--|
| Extension Number: | 5180 |
| IP address: | 10.10.40.206 |
| Standard Location: | None |
| Registrar: | Primary |
| Telephone Type: | Unknown SIP Device |
| User Agent: | NEC_IP_DECT_GW_G566/4920b655_01.14.00.01 |
| Media Stream: | RTP |
| Layer 4 Protocol: | TCP |
| Current User Extension Number: | 5180 |
| Current User Name: | 5180 |
| Forwarding: | Off |
| Twinning: | Off |
| Do Not Disturb: | Off |
| Message Waiting: | Off |
| Number of New Messages: | 0 |
| Phone Manager Type: | None |
| SIP Device Features: | REFER,UPDATE |
| License Reserved: | No |
| Last Date and Time License Allocated: | 02/06/2017 11:01:55 |
| DTMF Required: | No |
| Packet Loss Fraction: | |
| Jitter: | |
| Round Trip Delay: | |
| Connection Type: | Direct Media |
| Codec: | G711 A |
| Remote Media Address: | 10.10.40.206 |

8. Conclusion

These Application Notes describe the configuration steps required for Ascom's Myco Smart Device V9.3 to successfully interoperate with Avaya IP Office Server Edition and IP Office IP500 V2 Expansion R10.1 by registering the Myco handsets with IP Office as SIP phones. Please refer to **Section 2.2** for test results and observations.

9. Additional References

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

- [1] Administering Avaya IP Office™ Platform with Manager, Release 10, Issue 1.
- [2] Avaya IP Office™ Platform Documentation Catalog Release 10, Document number 16-604278 Issue 1
- [3] Avaya IP Office™ Platform 10. Deploying Avaya IP Office™ Platform IP500 V2, Document number 15-601042.

Product Documentation for Ascom Products can be obtained from an Ascom supplier or may be accessed at <https://www.ascom-ws.com/AscomPartnerWeb/Templates/WebLogin.aspx> (login required).

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