



Avaya Solution & Interoperability Test Lab

Application Notes for configuring Datatal AB Flexi with Avaya IP Office 500 V2 Standalone R10.0 - Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Datatal AB Flexi to interoperate correctly with Avaya IP Office 10.0.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the 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

Datatal Flexi platform is an application platform for telephony and unified communication on the Swedish market, and is also used in some other Nordic countries. Flexi platform includes three major products within the same server with shared administration.

- Flexi Tid.
- Flexi Presentity.
- Flexi CC.

Flexi Tid is a call back module that can handle time bookings. Customers call and book a timeslot for when they will be called back. This application is very useful in the healthcare industry where many incoming calls are received from customers concurrently.

Flexi Presentity is a presence and advanced voicemail module, including a mobile application where an end-user can activate absent states, like 'meeting' or 'lunch' and calling customers will receive a voice prompt that the user is busy in lunch, for instance.

Flexi CC is a call center module for customer services or support units. Incoming calls are queued in Flexi server and when an agent is free and available the call will be transferred. Flexi CC can also handle call back, so that calling customers can schedule a call back.

2. General Test Approach and Test Results

The general test approach was to configure the Flexi server in order to test all three modules. Flexi server utilises both a SIP trunk connection to IP Office in order to route calls and a TAPI connection in order to monitor existing IP Office users.

Flexi Tid makes use of both the SIP trunk and the TAPI connection in order to allow users to dial into a service on Flexi Tid and when the user is free this user can then click to call the customer.

Flexi Presentity makes use of both the SIP trunk and TAPI in order to allow callers route to the Flexi voicemail and the using TAPI to show the status of the IP Office users.

Flexi CC also makes use of both the SIP trunk and TAPI connection, the SIP trunk is used to allow incoming calls queue on the Flexi server and the TAPI connection to determine when an agent is free and available in order to transfer the call.

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

During compliance testing a series of test calls were made in order to test all three modules were functioning exactly as they should. See **Figure 1** for a network diagram. The interoperability compliance test focused on functionality tests, the testing included:

- Verification of connectivity between IP Office and Flexi.
- Testing Flexi CC – Inbound calls to a skillset on Flexi CC.
- Testing Flexi Presentity – Make users absent and divert to voicemail, make inbound calls to that users voicemail.
- Testing Flexi Tid - Inbound calls requiring call back, Flexi Tid agents making outbound calls.

2.2. Test Results

Tests were performed to insure full interoperability of Datatal AB Flexi and Avaya IP Office solution. The tests were all functional in nature and performance testing was not included. All the test cases passed successfully. The following observation was noted.

- When sending a divert to an Avaya IP Office 1140 SIP deskphone the display is not updated with the reason for the diversion only with the diversion and diversion number. The Digital and H323 sets were all updated correctly. This is a known issue with Flexi Presentity.

2.3. Support

Technical support from Datatal AB can be obtained through the following General Technical support contact:

Email: support@datatal.se
Phone: +46498253030

3. Reference Configuration

Figure 1 illustrates the network topology used during compliance testing. The Avaya solution consists of an IP Office 500 V2 which has a SIP Trunk connection to Flexi server. Telephony Application Programming Interface (TAPI) is configured on the Flexi server which enables it to control a telephone via IP Office, to act as the Flexi Tid/Contact Center agent.

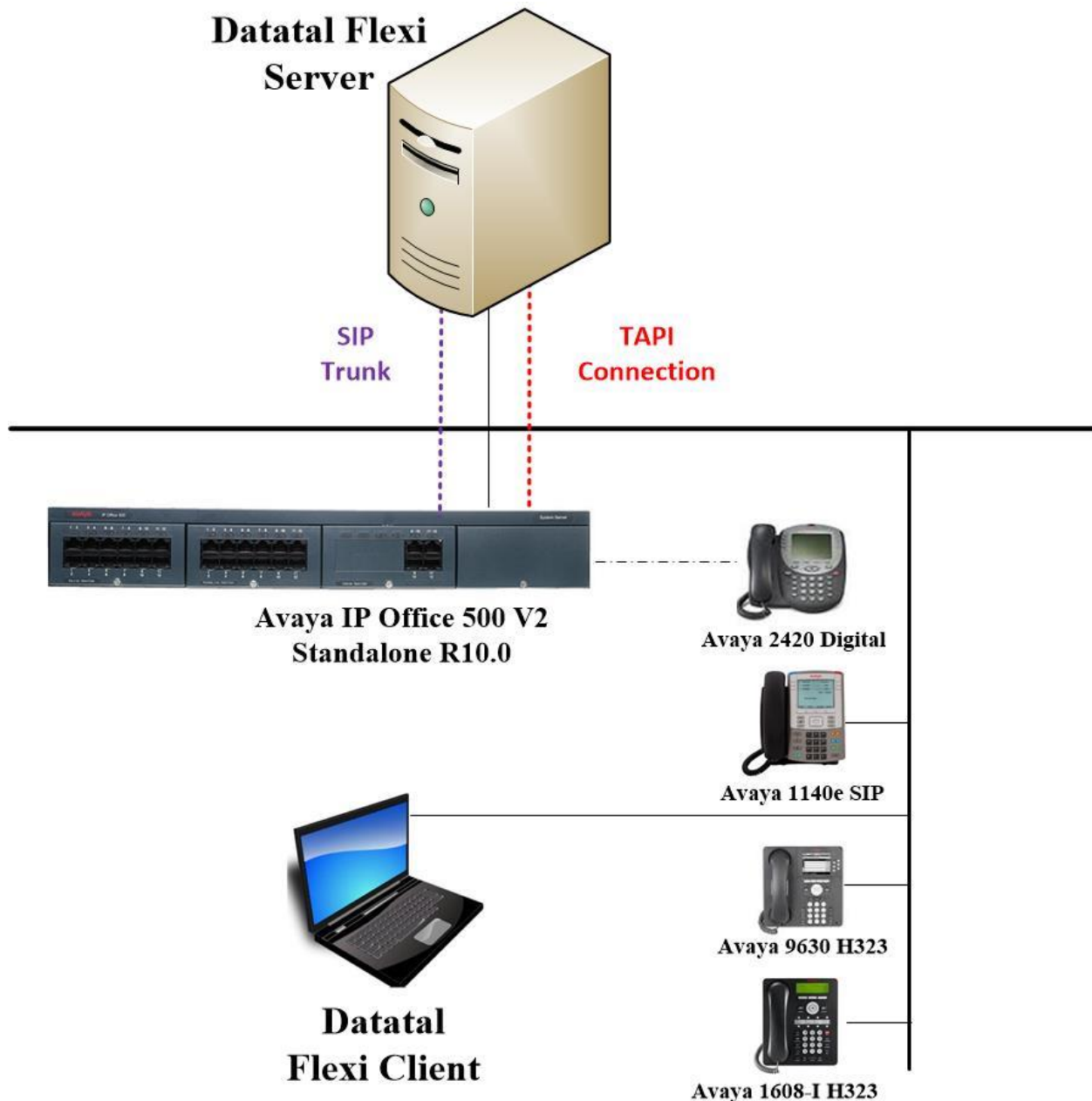


Figure 1: Avaya IP Office and Datatal AB Flexi reference configuration

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	R10.0.2.0 Build 10
Avaya IP Office Manager running on a Windows 7 PC	R10.0.2.0 Build 10
Avaya 9630 H323 Deskphone	R6.4014U
Avaya 1140e SIP Deskphone	R04.04.28.00
Avaya 1608 I Deskphone	H323 1608UA1_350B.bin
Datatall Flexi platform running on Microsoft Windows Server 2012 x64 R2	Version 5.12.1

Note: Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 only.

5. Avaya IP Office Configuration

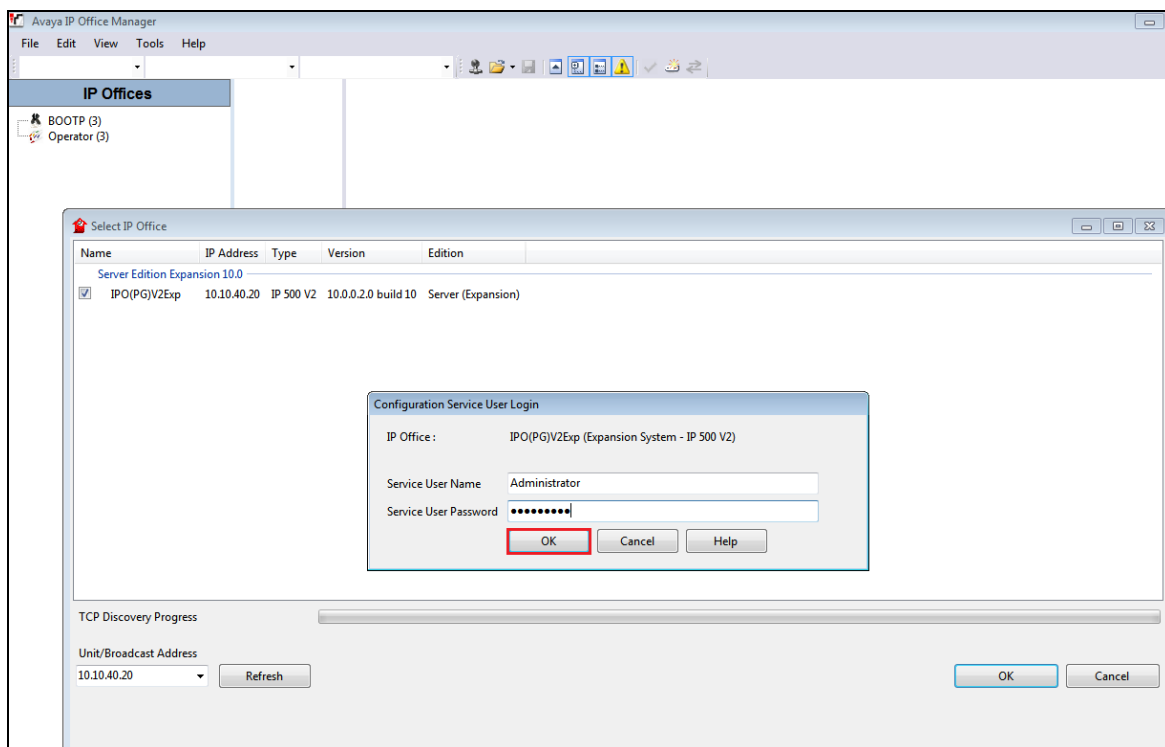
The information provided in this section describes the configuration of Avaya IP Office for this solution. Configuration and verification operations on the Avaya IP Office were all performed using Avaya IP Office Manager. It is implied a working system is already in place with the necessary licensing. 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 SIP Trunk.
- Configure Incoming Call Route.
- Create Short Code (Call Route).
- Configure Forwarding.
- 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). Enter the appropriate credentials and click on the **OK** button 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 in the TAPI setup in **Section 6.1**.

The screenshot shows the IP Office configuration interface. On the left, the 'System' tree is expanded, and 'System (1)' is selected. The main window displays the 'LAN1' tab under the 'IPO91(PG)V2Exp' system. The 'LAN Settings' sub-tab is active, showing the following configuration:

- IP Address: 10 . 10 . 40 . 20
- IP Mask: 255 . 255 . 255 . 0
- Primary Trans. IP Address: 10 . 10 . 40 . 1
- RIP Mode: None
- Enable NAT: ☐
- Number Of DHCP IP Addresses: 10
- DHCP Mode: ☐ Server ☐ Client ☐ Dialin ☒ Disabled
- Advanced:

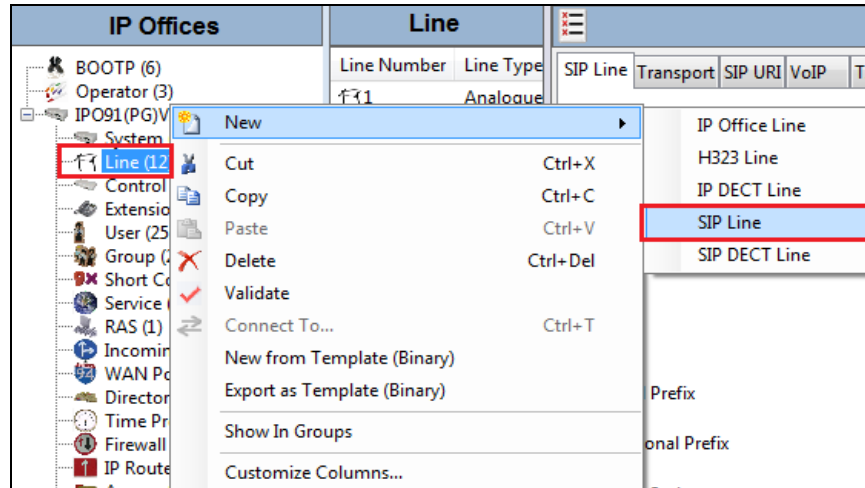
Select the **VoIP** tab and note the following highlighted below. These settings will be required in the setup of the SIP trunk on the Flexi server in **Section 6.2**.

The screenshot shows the IP Office configuration interface with the 'VoIP' tab selected under the 'LAN1' tab. The following settings are highlighted:

- ☒ H323 Gatekeeper Enable
- ☐ Auto-create Extn
- ☐ Auto-create User
- ☐ H323 Remote Extn Enable
- Remote Call Signalling Port: 1720
- ☒ SIP Trunks Enable
- ☒ SIP Registrar Enable
- ☐ Auto-create Extn/User
- ☐ SIP Remote Extn Enable
- Domain Name: devconnect.local
- Layer 4 Protocol: ☒ UDP, UDP Port: 5060, Remote UDP Port: 5060
- ☒ TCP, TCP Port: 5060, Remote TCP Port: 5060
- ☐ TLS, TLS Port: 5061, Remote TLS Port: 5061
- Challenge Expiry Time (secs): 10
- RTP: ☐ Port Number Range

5.3. Create SIP Trunk

To create the SIP trunk from the IP Office to the Datatal Flexi server, navigate to **System** and right click on **Line** followed by **New** → **SIP Line**.



In the subsequent **SIP Line** window, enter the following in the **SIP Line** tab.

- **ITSP Domain Name** Enter the telephony domain name.
- **Refresh Method** Select **Reinvite** from the dropdown menu.
- **REFER and Transfer** Select **Always** both the **Incoming** and **Outgoing** dropdown boxes.

Note: **Line number** is chosen and defaults were used for the remaining fields.

A screenshot of the 'SIP Line - Line 22' configuration window. The 'SIP Line' tab is selected. The window contains various fields and checkboxes for configuring the SIP line. The 'Line Number' is set to 22. The 'ITSP Domain Name' is 'devconnect.local'. The 'Local Domain Name' is empty. The 'URI Type' is 'SIP'. The 'Location' is 'Cloud'. The 'Prefix' is empty. The 'National Prefix' is '0'. The 'International Prefix' is '00'. The 'Country Code' is empty. The 'Name Priority' is 'System Default'. The 'Description' is empty. The 'In Service' checkbox is checked. The 'Check OOS' checkbox is checked. The 'Refresh Method' is 'Reinvite'. The 'Timer (seconds)' is 'On Demand'. The 'Incoming Supervised REFER' is 'Always'. The 'Outgoing Supervised REFER' is 'Always'. The 'Send 302 Moved Temporarily' checkbox is unchecked. The 'Outgoing Blind REFER' checkbox is unchecked.

Click on the **Transport** tab enter the IP address of the Flexi Server in the **ITSP Proxy Address** field. **Layer 4 Protocol** was set to **UDP** and **Port 5060** was used as this will be referenced again in **Section 6.2**.

SIP Line - Line 22

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

ITSP Proxy Address 10.10.40.145

Network Configuration

Layer 4 Protocol UDP Send Port 5060

Use Network Topology Info None Listen Port 5060

Explicit DNS Server(s) 0 . 0 . 0 . 0 0 . 0 . 0 . 0

Calls Route via Registrar ☒

Separate Registrar

In the **SIP URI** tab click on the **Add** button.

SIP Line - Line 18

SIP Line Transport SIP URI VoIP T38 Fax SIP Credentials SIP Advanced Engineering

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls

Add... Remove Edit...

In the subsequent window, enter the following:

- **Local URI** Enter *
- **Contact** Enter *
- **Display Name** Enter *
- **Identity** Select **None** from the dropdown menu.
- **Header** Select **P Asserted ID** from the dropdown menu.
- **Send Caller Id** Select **Diversion Header** from the dropdown menu.
- **Diversion Header** Select **None** from the dropdown menu.

SIP Line - Line 22

SIP Line Transport SIP URI VoIP T38 Fax SIP Credentials SIP Advanced Engineering

URI	Groups	Local URI	Contact	Display Name	Identity	Header	Originator Number	Send Caller ID	Diversion Header	Credential	M
1	22 22	Auto	Auto	Auto	None	PAI		Diversion	None	0: <Non...	10

Add... Remove Edit...

Edit URI

Local URI *

Contact *

Display Name *

Identity

Identity None

Header P Asserted ID

Forwarding And Twinning

Originator Number

Send Caller Id Diversion Header

Diversion Header None

OK Cancel

Enter the following:

- **Incoming Group** Enter the SIP trunk number.
- **Outgoing Group** Enter the SIP trunk number.
- **Max Sessions** Enter the amount of trunks to be created.

Click the **OK** button.

The screenshot shows the 'SIP Line - Line 22' configuration window with the 'SIP URI' tab selected. The 'Edit URI' section contains fields for 'Local URI', 'Contact', and 'Display Name', all set to '*'. The 'Identity' section has 'Identity' set to 'None' and 'Header' set to 'P Asserted ID'. The 'Forwarding And Twinning' section has 'Originator Number' empty and 'Send Caller Id' set to 'Diversion Header'. The 'Diversion Header' is set to 'None'. The 'Registration' is set to '0: <None>'. The 'Incoming Group' and 'Outgoing Group' are both set to '22'. The 'Max Sessions' is set to '10'. The 'OK' button is highlighted with a red box.

Click on the **VoIP** tab and selecting **Custom** for the **Codec Selection** and choose the Codec's that are required and compatible. Tick the **Re-invite Supported** and the **Codec Lockdown** boxes. **DTMF Support** was set to **RFC 2833** for compliance testing but this may differ on a customer site. Click the **OK** button once everything is set correctly (not shown).

The screenshot shows the 'SIP Line - Line 22' configuration window with the 'VoIP' tab selected. The 'Codec Selection' is set to 'Custom'. The 'Unused' list contains 'G.711 ULAW 64K', 'G.722 64K', and 'G.723.1 6K3 MP-MLQ'. The 'Selected' list contains 'G.711 ALAW 64K' and 'G.729(a) 8K CS-ACELP'. The 'Fax Transport Support' is set to 'None'. The 'DTMF Support' is set to 'RFC2833'. The 'Media Security' is set to 'Media Security Features Disabled'. The 'Re-invite Supported' and 'Codec Lockdown' checkboxes are checked. The 'OK' button is highlighted with a red box.

For compliance testing the values under the **SIP Advanced** tab were left as default as shown below.

The screenshot shows the 'SIP Line - Line 22' configuration window with the 'SIP Advanced' tab selected. The window is divided into several sections: Addressing, Identity, Media, and Call Control. The 'SIP Advanced' tab is highlighted with a red box in the top navigation bar.

Addressing

- Association Method: By Source IP address
- Call Routing Method: Request URI
- Suppress DNS SRV Lookups: ☐

Identity

- Use "phone-context": ☐
- Add user=phone: ☐
- Use + for International: ☐
- Use PAI for Privacy: ☐
- Use Domain for PAI: ☐
- Swap From and PAI/Diversion: ☐
- Caller ID from From header: ☐
- Send From In Clear: ☐
- Cache Auth Credentials: ☒
- User-Agent and Server Headers:
- Send Location Info: Never

Media

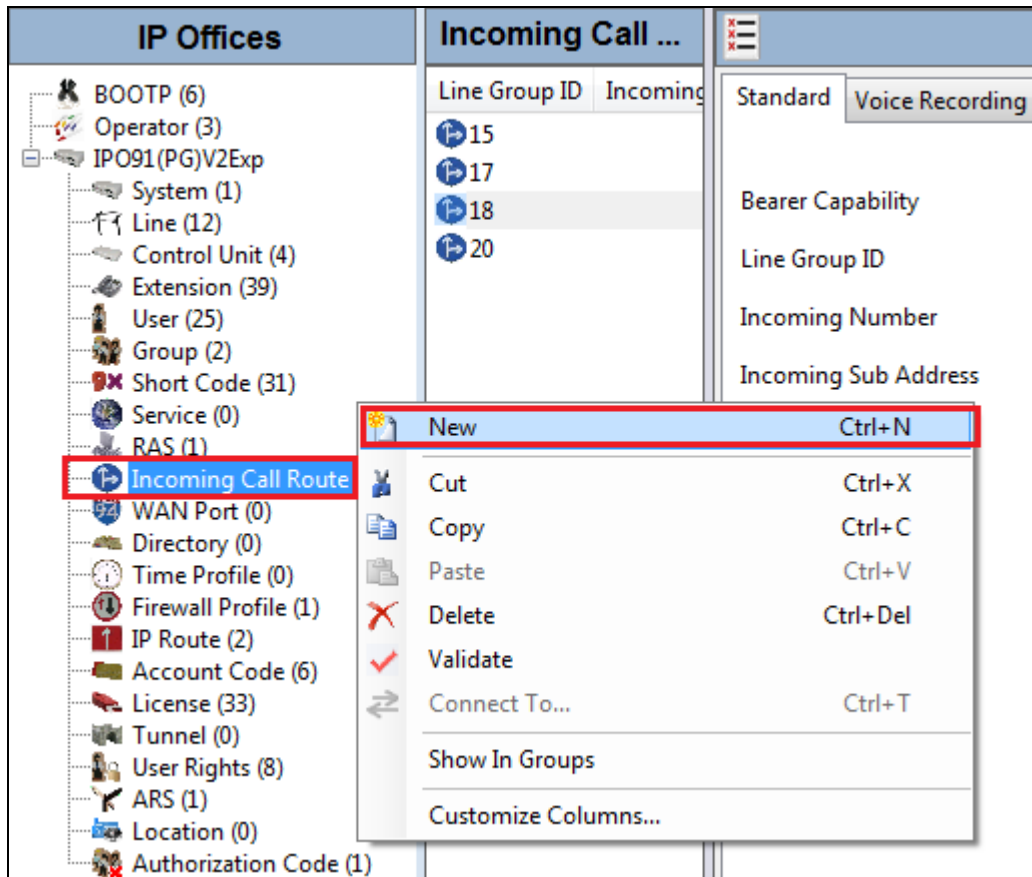
- Allow Empty INVITE: ☐
- Send Empty re-INVITE: ☐
- Allow To Tag Change: ☐
- P-Early-Media Support: None
- Send SilenceSupp=Off: ☐
- Force Early Direct Media: ☐
- Media Connection Preservation: Disabled
- Indicate HOLD: ☐

Call Control

- Call Initiation Timeout (s): 4
- Call Queuing Timeout (m): 5
- Service Busy Response: 486 - Busy Here
- on No User Responding Send: 408-Request Timeout
- Action on CAC Location Limit: Allow Voicemail
- Suppress Q.850 Reason Header: ☐
- Emulate NOTIFY for REFER: ☐
- No REFER if using Diversion: ☐

5.4. Configure Incoming Call Route

To configure the Incoming Call Route, navigate to **System** and right click on **Incoming Call Route** followed by **New**.



In the subsequent window, enter the following in the **Standard** tab.

- **Line Group ID** Enter the Incoming Group number as used in **Section 5.3**.

Defaults were used for the remaining fields.

The screenshot shows a configuration window titled '22' with three tabs: 'Standard', 'Voice Recording', and 'Destinations'. The 'Standard' tab is selected and highlighted with a red box. It contains the following fields:

Field	Value
Bearer Capability	Any Voice
Line Group ID	22
Incoming Number	
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None

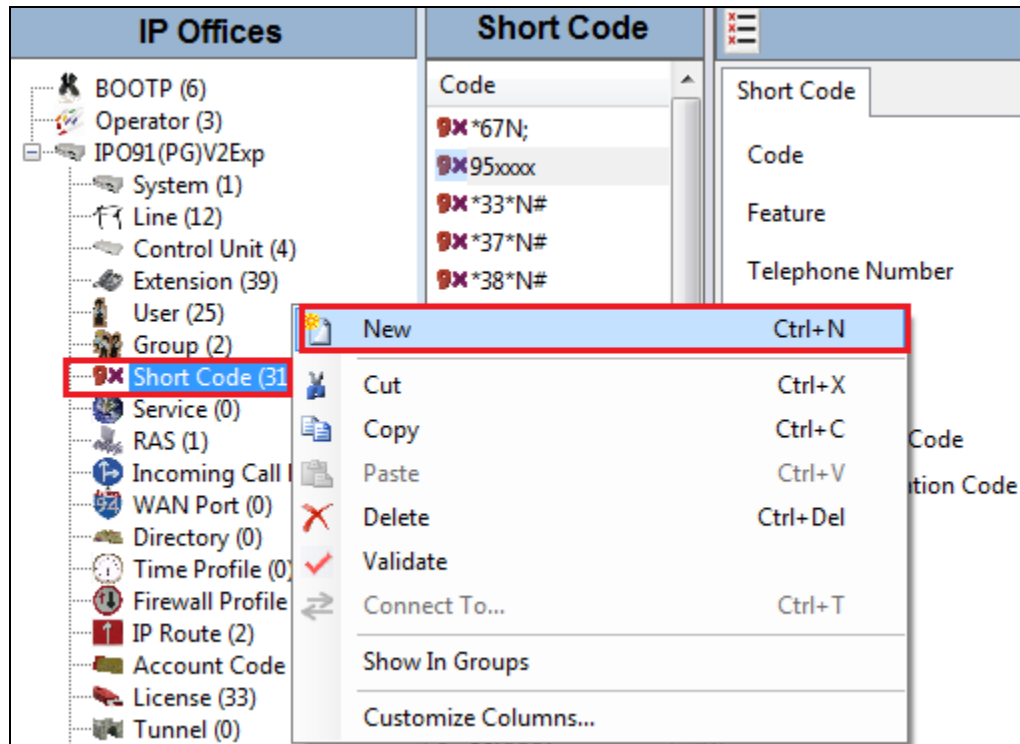
In the **Destinations** tab, enter a . (full stop/period) in the **Destination** field. Click on the **OK** button.

The screenshot shows the same configuration window with the 'Destinations' tab selected and highlighted with a red box. The 'Destination' field is also highlighted with a red box and contains a period (.). The 'Fallback Extension' field is empty. The 'OK' button is highlighted with a red box.

TimeProfile	Destination	Fallback Extension
Default Value	.	

5.5. Create Short Code (Route Calls)

A Short Code needs to be configured on the IP Office to route calls to Flexi server. Right click on **Short Code**, and select **New**.



In the subsequent window, enter the following:

- **Code** Enter the number range that will be routed to Flexi server (during compliance testing, all numbers beginning with 95 were sent to Flexi server, therefore **95xxxx** was entered).
- **Feature** Select **Dial** from the dropdown menu.
- **Telephone Number** Enter **Nss** (Nss will send the originating calling parties caller ID).
- **Group Line ID** Enter the Incoming Group number as used in **Section 5.4**.

Click the **OK** button.

95xxxx: Dial*

Short Code

Code: 95xxxx

Feature: Dial

Telephone Number: Nss

Line Group ID: 22

Locale:

Force Account Code: ☐

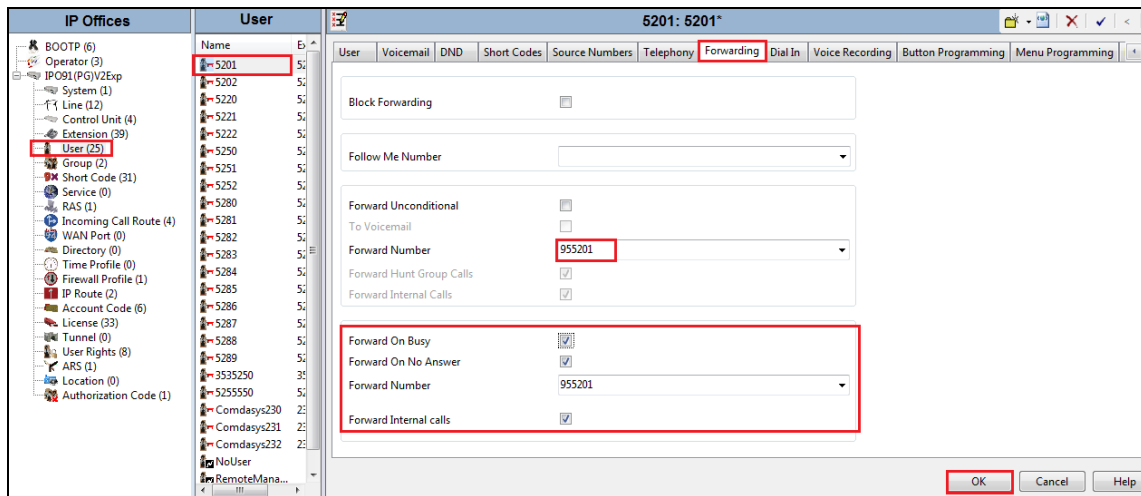
Force Authorization Code: ☐

OK Cancel Help

5.6. Configure Forwarding

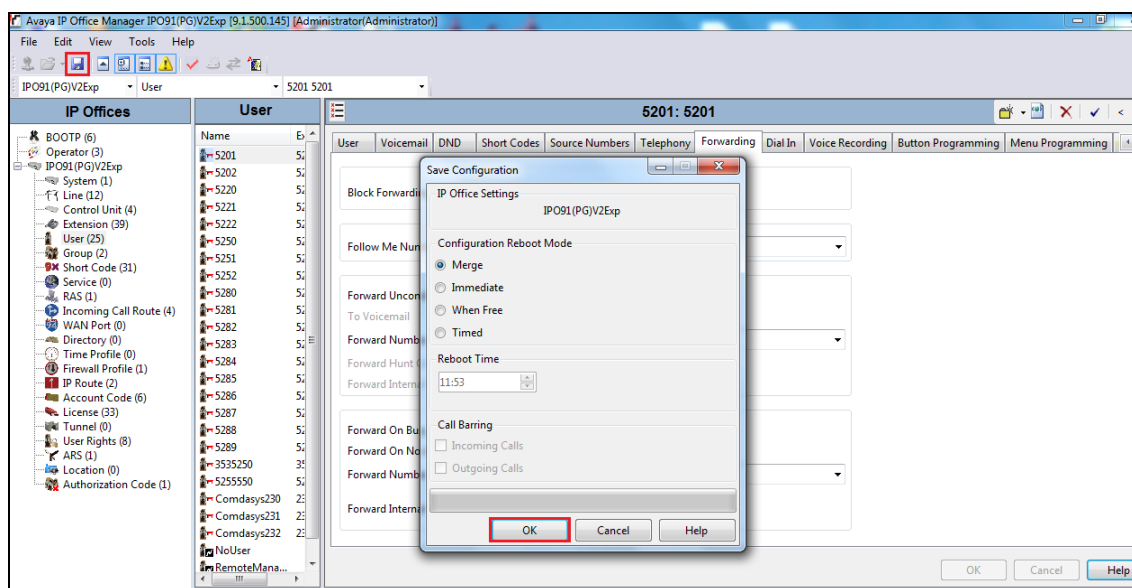
Forward On Busy and **Forward On No Answer** are configured for one of the IP Office users in order to test Flexi Presentity. To configure forwarding click on the **User** and click on the **Forwarding** tab, and in the **Forwarding Number** field enter the Short Code (as configured in **Section 5.5**) followed by the extension used by this user (example **5201**).

To set **Forward On Busy** and **Forward On No Answer** ensure that both of these fields are ticked as shown below and click the **OK** button.



5.7. 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 Datatal AB Flexi

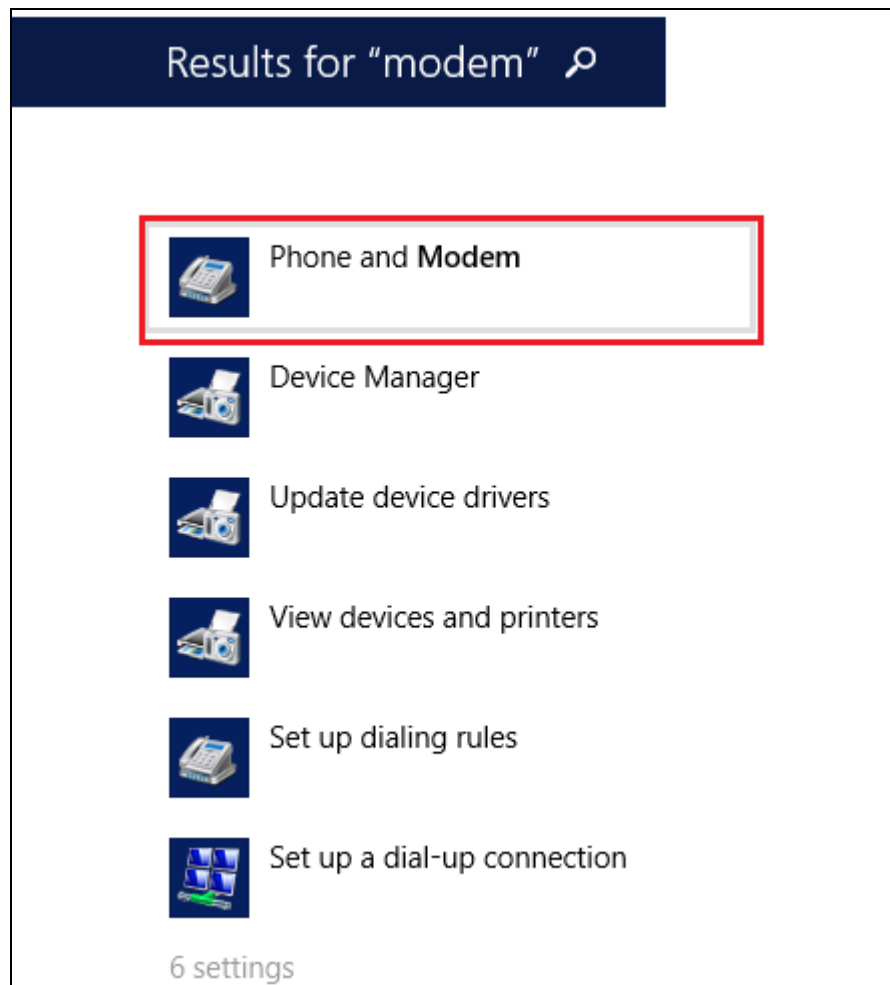
Configuration of the Flexi server consists of two specific parts, the SIP trunk and the TAPI connection. The Avaya IP Office TAPI driver is installed and configured on the Flexi server. The SIP Trunk is configured using a web GUI by opening a browser session to the Flexi server.

6.1. Configure Avaya IP Office TAPI

The Avaya IP Office TAPI is required so as to allow certain features of Flexi to interoperate with IP Office. It is implied that the TAPI software is already installed.

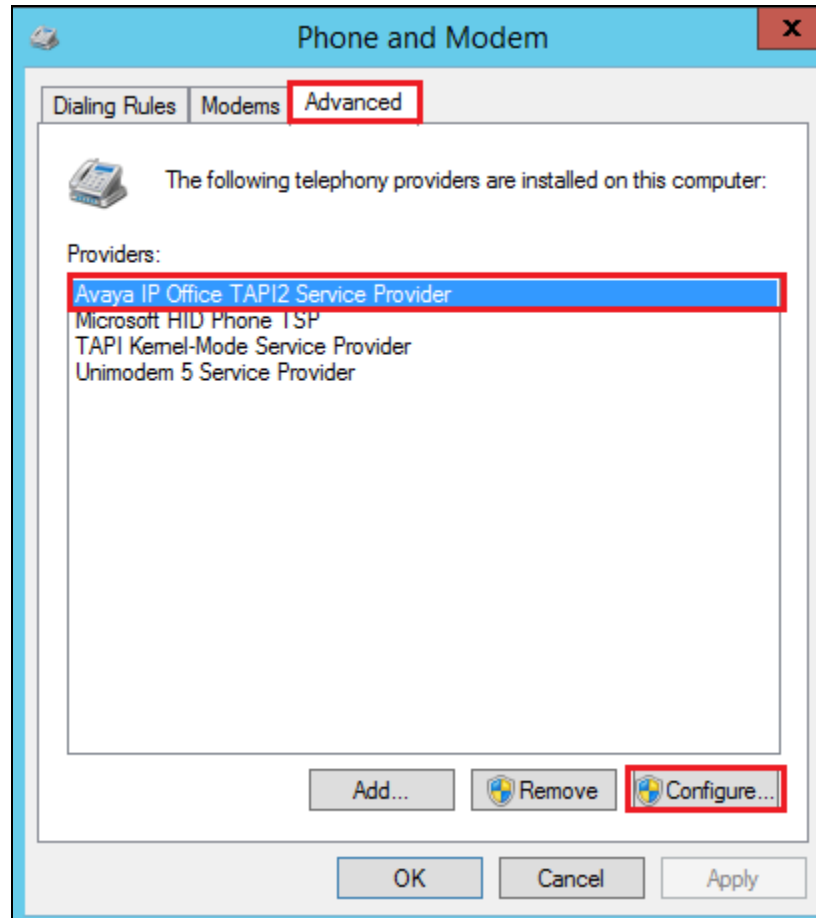
Note: It is important that the TAPI software installation was run as administrator to ensure that the application receives the correct rights to run.

From the Windows 2012 Server search for “**modem**” and the following window should appear, double click on **Phone and Modem** as shown below.



Select the **Advanced** tab. Once the **Advanced** tap opens, select **Avaya IP Office TAPI2 Service Provider** and click on the **Configure** button.

Note: Enter any appropriate dealing rules in the **Dialing Rules** tab as required (not shown).



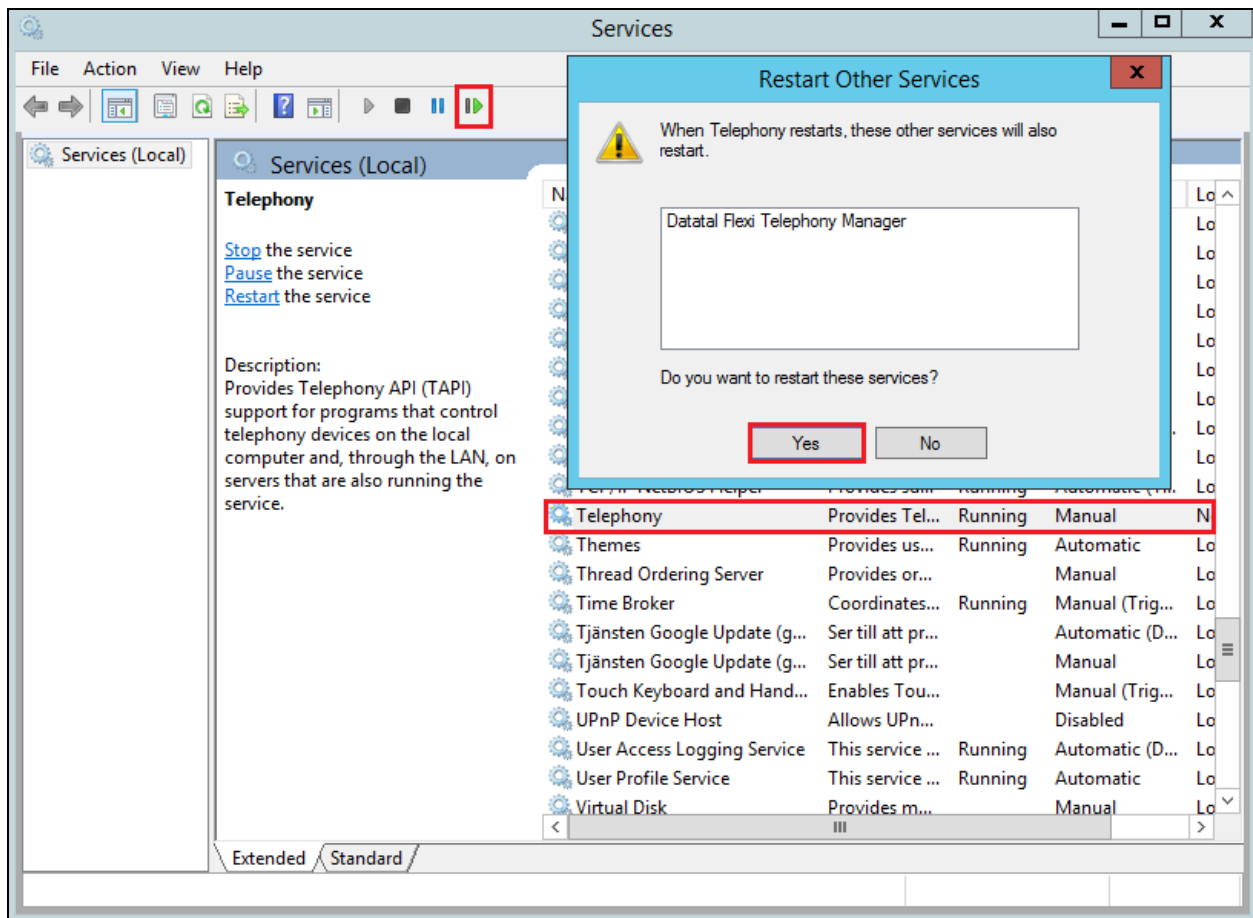
Once the **Avaya TAPI2 Configuration** window opens, enter the following:

- **Switch IP address** Enter the IP address of the IP Office.
- **Third Party** Click on the Radio button.
- **Switch Password** Enter the password of the IP Office System User.
- **ACD Queues** Click on the check box.

Click the **OK** button.

The screenshot shows a 'Phone and Modem' window with a sub-dialog 'Avaya TAPI2 configuration'. The sub-dialog has a title bar with a close button. Inside, there's a 'Switch IP Address' text box containing '10.10.40.20'. To its right is an 'OK' button (highlighted with a red rectangle) and a 'Cancel' button. Below these are two radio buttons: 'Single User' (unselected) and 'Third Party' (selected). Under 'Third Party', there's a 'Switch Password' text box with 'xxxxxxxx' and three checkboxes: 'Ex Directory Users' (unchecked), 'WAV Users' (unchecked), and 'ACD Queues' (checked). At the bottom of the sub-dialog are 'Add...', 'Remove', and 'Configure...' buttons. At the bottom of the main window are 'OK', 'Cancel', and 'Apply' buttons.

Once TAPI is configured, restart the **Telephony** service, restarting any other service that may need to be restarted also.



6.2. Configure SIP Trunk

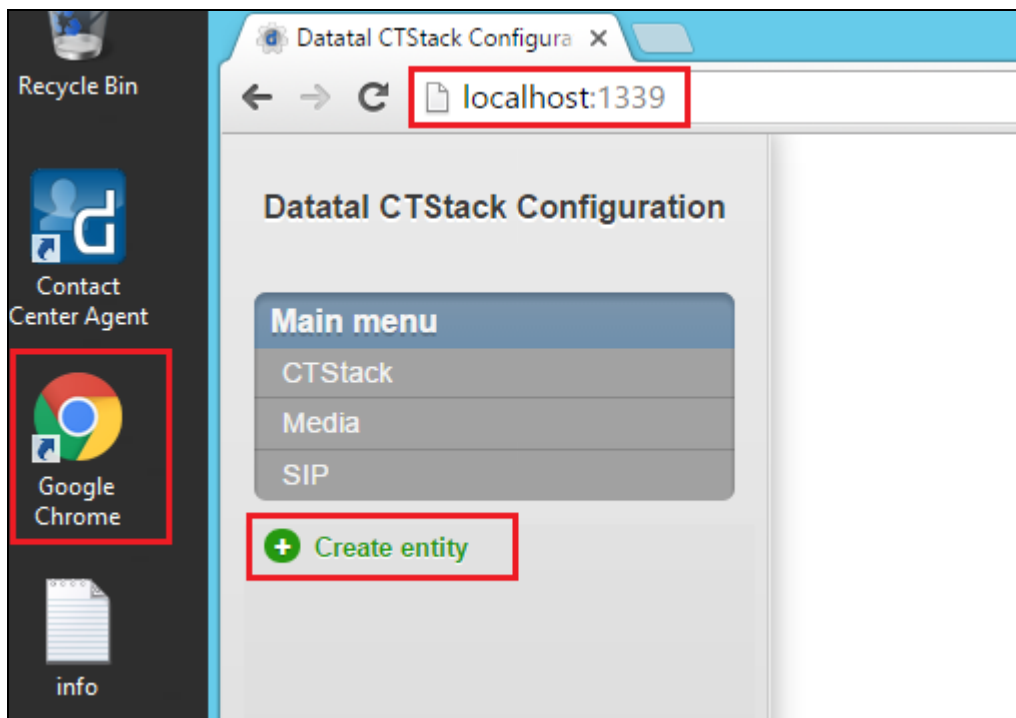
Configuration of the Flexi server is achieved using a web interface. After logging on to the Flexi server, browse to **localhost:1339** using Internet Explorer 10 or higher, Mozilla Firefox or Google Chrome web browsers. The following configuration steps were carried out during compliance testing:

- Configure entity for Avaya IP Office.
- Configure Media.
- Configure SIP.
- Configure Telephony.

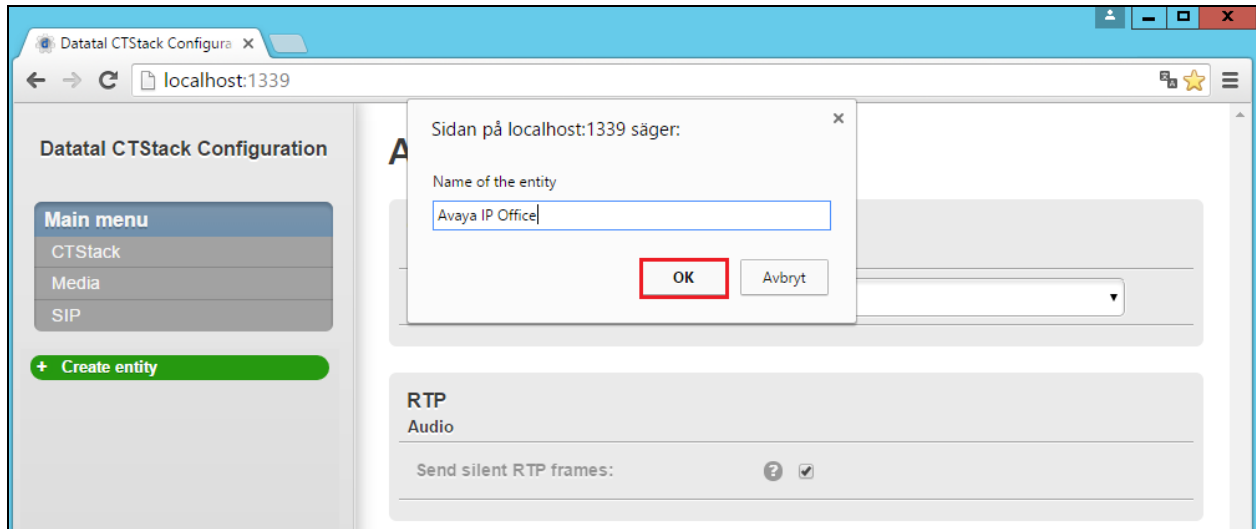
Note: It is implied that the Flexi server is pre-configured including any Licence requirements. Configuration of Flexi Presentity, Flexi CallCenter agents and Flexi Tid agents is outside the scope of these Application Notes.

6.2.1. Configure entity for Avaya IP Office

Once the web page opens, select **Create entity**.

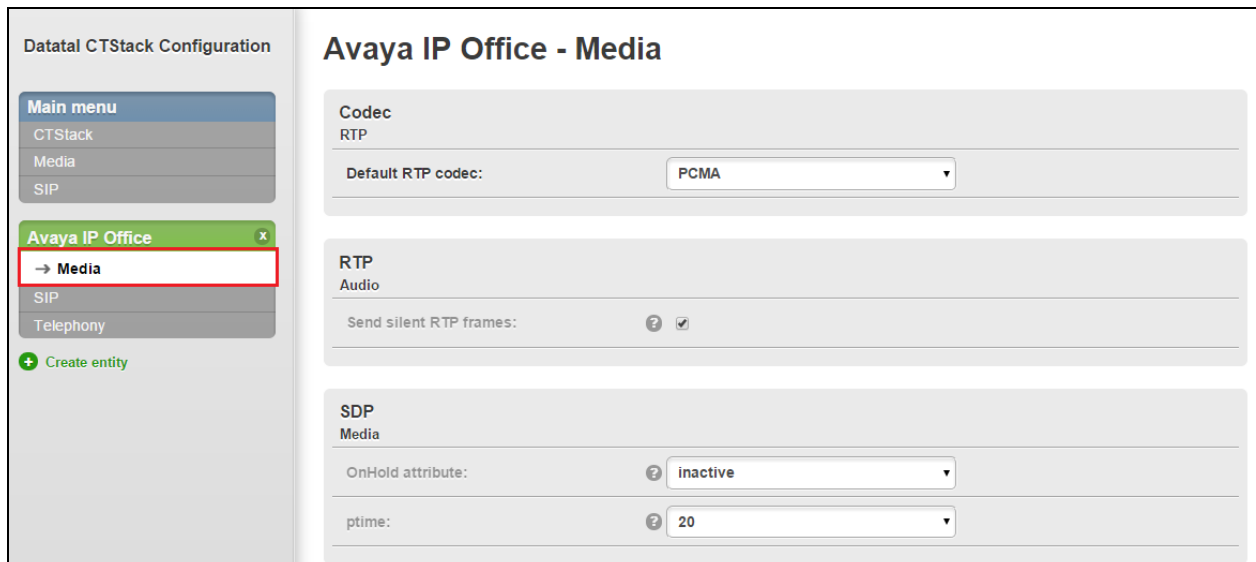


Once the new frame opens enter an informative name in the **Name of the entity** box, **Avaya IP Office** was used during compliance testing. Click the **OK** button to save.



6.2.2. Configure Media

The following were set for **Media** for compliance testing.



6.2.3. Configure SIP

After the entity is created the SIP configuration is required. Select **SIP** for the IP Office configured in **Section 5**.

The screenshot shows the 'Avaya IP Office - SIP' configuration page. On the left is a sidebar with a 'Main menu' containing 'CTStack', 'Media', and 'SIP'. Below this is a section for 'Avaya IP Office' with a sub-menu containing 'Media', '→ SIP' (highlighted with a red box), and 'Telephony'. At the bottom of the sidebar are 'Commit' and 'Revert' buttons, and a 'Create entity' button. The main area is titled 'Avaya IP Office - SIP' and contains several sections: 'Dialogs' with 'Always create early dialogs:' (checkbox) and 'Retry-After 4xx:' (text input '25'); 'Inbound' with 'Use Flexi TID ListenExtension:' (checkbox); 'Outbound' with 'Privacy' header value (text input 'none'), 'Set Diversion' header on MakeCall (checkbox checked), 'Set History-Info' header on MakeCall (checkbox), and 'Use P-Asserted-Identity' (checkbox checked); and 'Transfer' with 'Hangup leg A on supervised 180/183:' (checkbox checked) and 'Hangup leg A on supervised 200:' (checkbox). A watermark 'Activate Windows' is visible in the bottom right.

Section	Configuration Item	Value / Status
Dialogs	Always create early dialogs:	<input type="checkbox"/>
	Retry-After 4xx:	25
Inbound	Use Flexi TID ListenExtension:	<input type="checkbox"/>
Outbound	'Privacy' header value:	none
	Set 'Diversion' header on MakeCall:	<input checked="" type="checkbox"/>
	Set 'History-Info' header on MakeCall:	<input type="checkbox"/>
	Use 'P-Asserted-Identity':	<input checked="" type="checkbox"/>
Transfer	Hangup leg A on supervised 180/183:	<input checked="" type="checkbox"/>
	Hangup leg A on supervised 200:	<input type="checkbox"/>

On the **SIP** page (**Transfer** section) configure the following:

- **Park other calls on MakeCall** Uncheck the check box
- **Play 'ring' at other calls on MakeCall** Check the check box

Default values were used for the remaining fields.

The screenshot displays the 'Datatal CTStack Configuration' web interface. On the left is a sidebar with a 'Main menu' containing 'CTStack', 'Media', and 'SIP'. The 'SIP' item is highlighted with a red box and a red arrow. Below the menu is a 'Create entity' button. The main content area is titled 'Transfer' and contains several configuration options, each with a help icon and a checkbox:

- Hangup leg A on supervised 180/183: ☒
- Hangup leg A on supervised 200: ☐
- Park other calls on MakeCall: ☐ (highlighted with a red box)
- Play 'ring' at other calls on MakeCall: ☒ (highlighted with a red box)
- Terminate local call transfer on INVITE: ☐
- Treat BYE as transfer success: ☐
- Use 'Remote-Target' in 'Refer-To': ☒
- Wait for park complete on MakeCall: ☒

Below the 'Transfer' section is a 'Registrations' section with a 'Users' tab. It features a 'Registrations:' label, a help icon, an empty list box, and three buttons: 'ADD' (green), 'EDIT' (blue), and 'REMOVE' (red). At the bottom of the interface are 'Commit' and 'Revert' buttons, and a 'datatal ab' logo.

Scroll down to **Dialogs** using the vertical scroll bar on the right side of the page to the **SIP** section and check the **Use 'from' header** check box, and select **UDP** from the **Transport** dropdown box. Defaults were used for the remaining fields. Click on the **Commit** button. When the **Commit** dialog window opens click on **Commit changes now** button (not shown).

Avaya IP Office

Media

SIP

Telephony

+ Create entity

Wait for park complete on MakeCall: ? ☒

Registrations

Users

Registrations: ?

ADD

EDIT

REMOVE

SIP

Dialogs

Use 'From' header: ? ☒

RFC 3325

P-Identity mode: ? Both

Transport

Transport: ? UDP

1 change(s) pending

Commit Revert

datatal ab

6.2.4. Configure Telephony

To configure Telephony click on **Telephony** for the IP Office configured in **Section 5**.

Configure the following:

- **Lines** Enter the number of SIP lines that Flexi is licensed for.
- **Address** Enter the Flexi queue number (5250 was used during compliance testing).
- **Default Domain** Enter the telephony domain as per **Section 5.2**.
- **Default SIP URI host** Enter the IP address of the IP Office as per **Section 5.2**.
- **Default SIP URI port** **Enter the UDP port number configured in Section 5.2.**
- **Name** Enter an informative name for the Flexi Server (e.g. **DevConnect**).

Scroll down to the **Profile** section and enter the following:

- **Apply** Select **Avaya IPO (Trunk)** from the dropdown box.
- **Trunk Mode** Check the check box.

Defaults were used for the remaining fields. Click on the **Commit** button. When the **Commit** dialog window opens click on **Commit changes now** button (not shown).

Datatal CTStack Configuration

Main menu

- CTStack
- Media
- SIP

Avaya IP Office x

- Media
- SIP
- **Telephony**

+ Create entity

1 change(s) pending

Commit **Revert**

datatal ab

Description: ? string

INVITE expires: ? 25

Lines: ? 10

SIP

Address

Address: ? 5250

Default domain: ? devconnect.local

Default SIP URI host: ? 10.10.40.20

Default SIP URI port: ? 5060

Name: ? DevConnect

Profile

Apply: ? Avaya IPO (trunk) ▼

Current: ? None

Trunk

Trunk mode: ? ☒

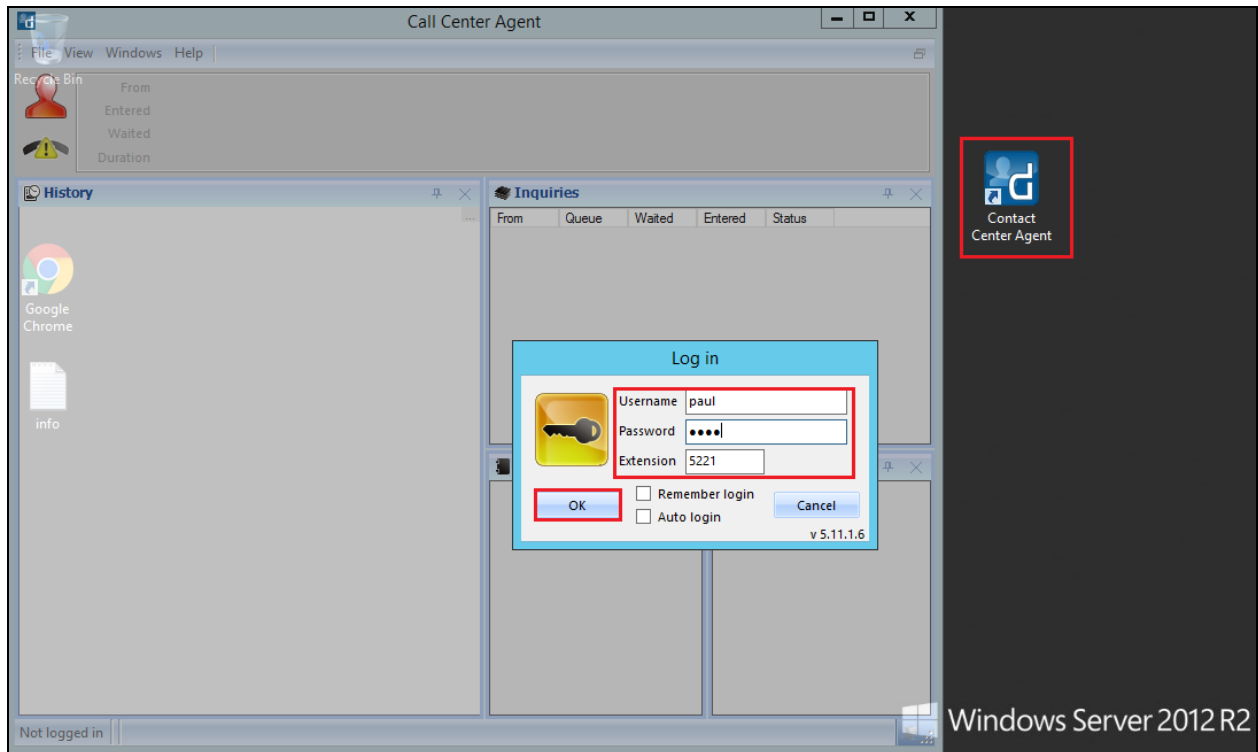
Activat
Go to Sys
Windows

7. Verification Steps

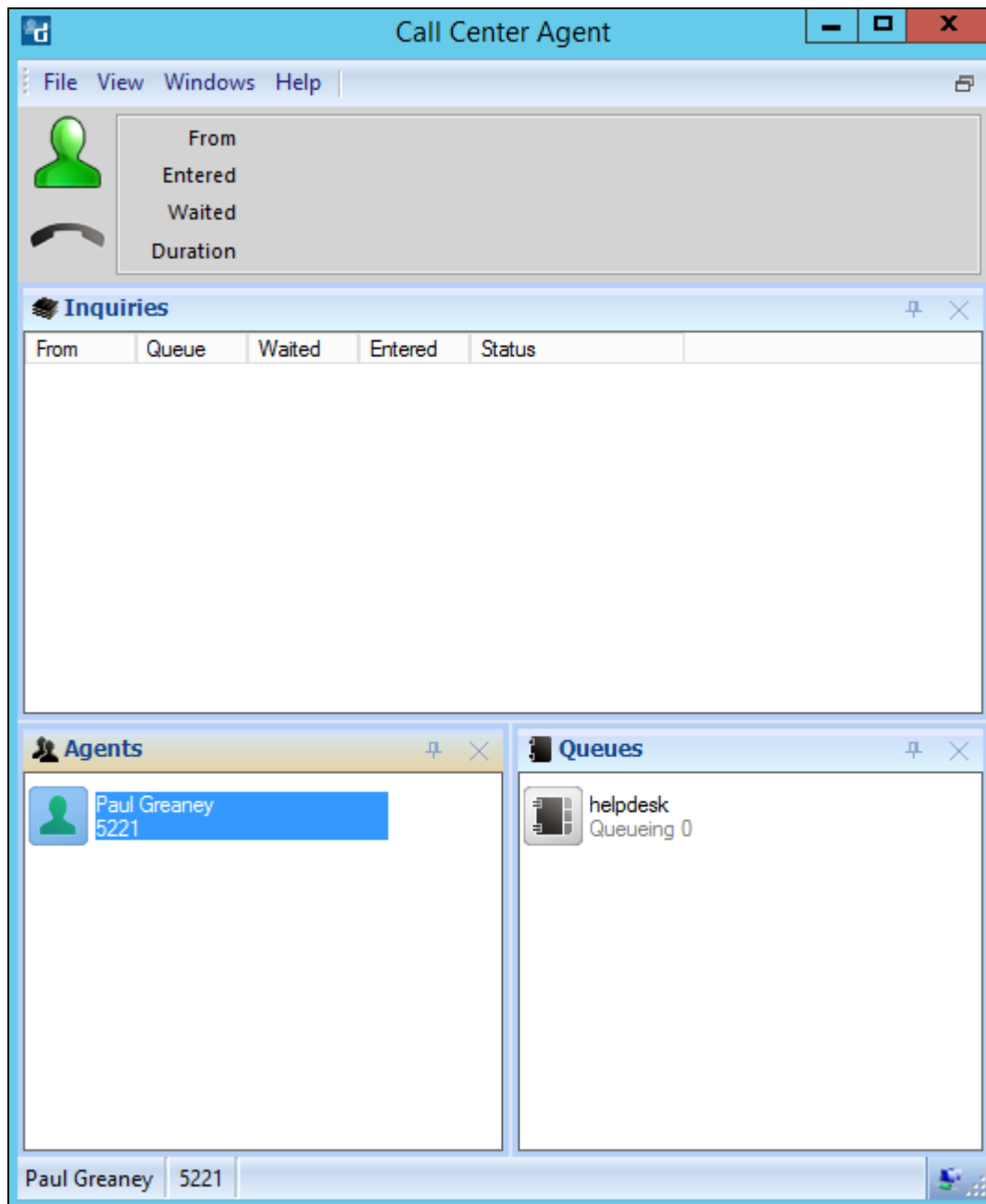
This section provides the tests that can be performed to verify correct configuration of the Avaya IP Office and Datatal AB Flexi.

7.1. Verify Flexi CC

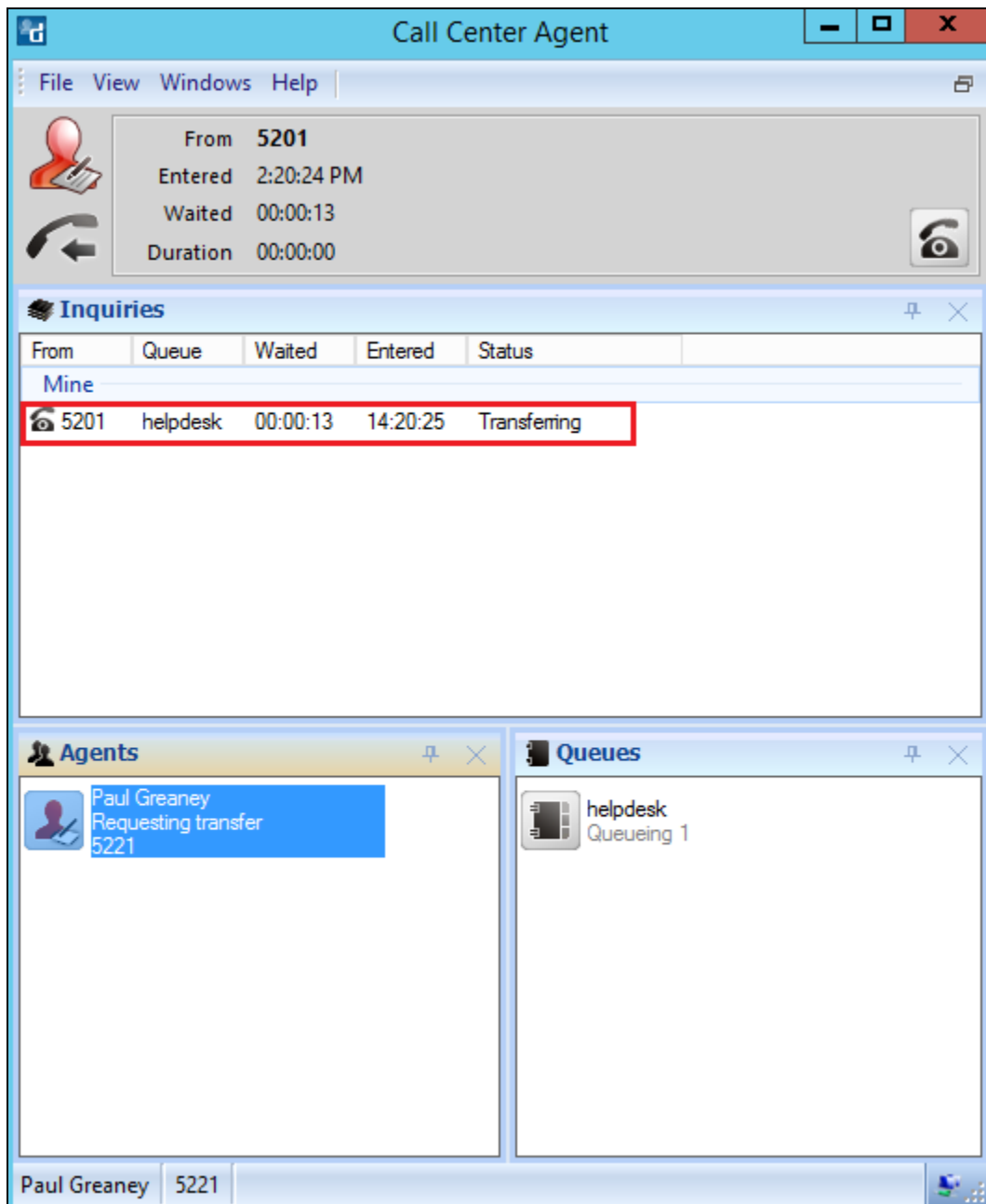
Using the shortcut on the desktop open **Contact Center Agent**, enter the appropriate credentials and click on **OK**.



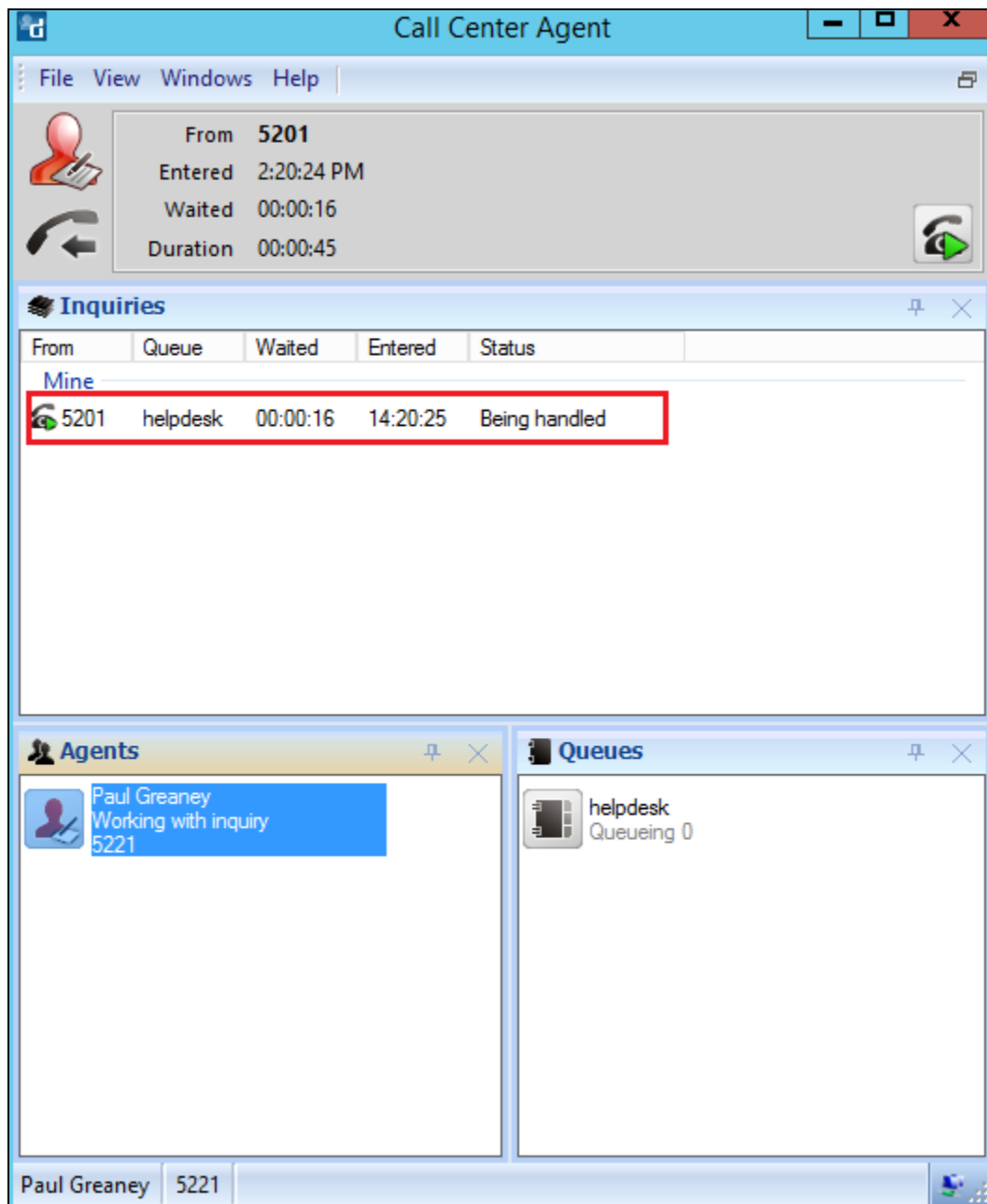
Once logged in the following screen shows the status of the agent and the queue associated with the agent.



Make a call to the Flexi CC queue. The agent's status has now changed and a call is seen incoming to the queue and **Transferring** to the agent who is free to take the call.

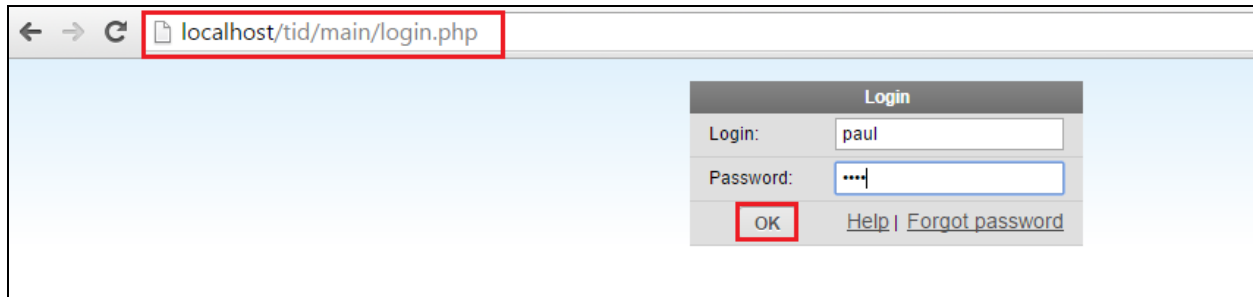


Once the call is answered this is reflected on the desktop as shown below.



7.2. Verify Flexi Tid

Open a web session to Flexi Tid. Enter the appropriate credentials and click on **OK** to log in.



localhost/tid/main/login.php

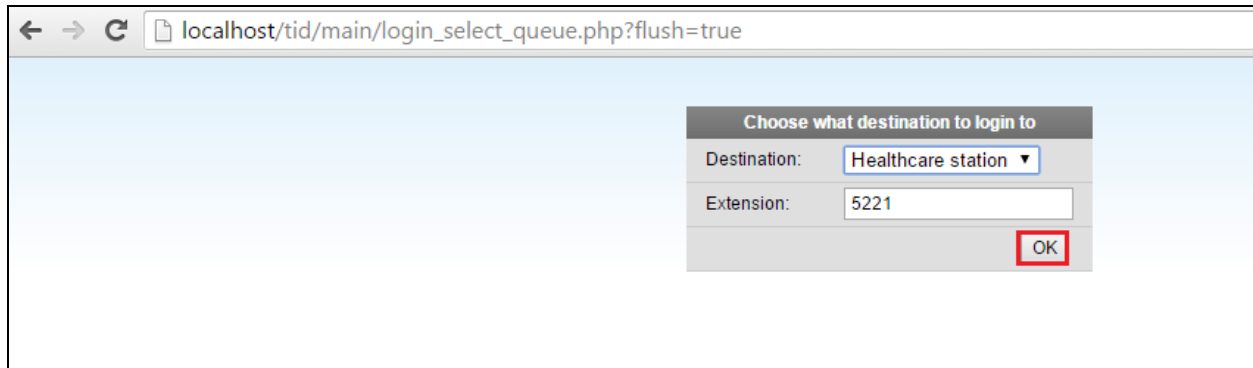
Login

Login: paul

Password:

OK Help | Forgot password

Log in to the correct extension and queue and click on **OK**.



localhost/tid/main/login_select_queue.php?flush=true

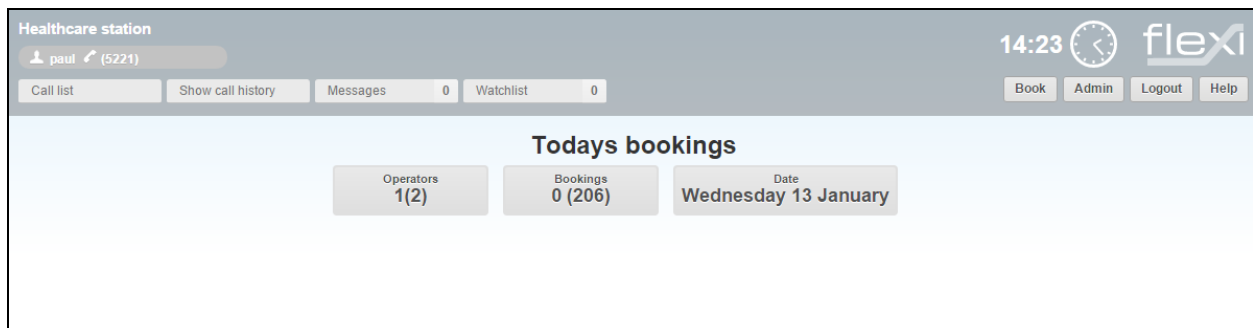
Choose what destination to login to

Destination: Healthcare station ▼

Extension: 5221

OK

The following screen is displayed once logged in correctly.



Healthcare station

paul (5221)

14:23 flexi

Call list Show call history Messages 0 Watchlist 0

Book Admin Logout Help

Todays bookings

Operators 1(2) Bookings 0 (206) Date Wednesday 13 January

Make a call to the Flexi Tid queue number and request a call back. The following screen is then updated to show that a new call is ready for call back. By clicking on **Call** the phone call to the **Phonenumber** is initiated. Ensure the agent desk phone and called number is connected.

The screenshot shows the Flexi Tid interface for a healthcare station. The header includes the station name, user 'paul (5221)', and the time '14:24'. Below the header, there are buttons for 'Call list', 'Show call history', 'Messages', and 'Watchlist'. The main section displays 'Today's bookings' for Wednesday 13 January. It shows 'Operators 1(2)', 'Bookings 1 (205)', and a table with columns: Call, Time, Phonenumber, SSN, Note, Bookings, Sign, Originated, and Count. The 'Call' button in the table is highlighted with a red box.

8. Conclusion

These Application Notes describe the required configuration steps necessary Datatal AB Flexi to interoperate with Avaya IP Office 500 V2 Standalone R10.0. All test cases passed successfully with observations noted in **Section 2.2**.

9. Additional References

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

Product documentation for Avaya products may be found at <http://support.avaya.com>

[1] *Avaya IP Office R10.0 Manager 10.0, Document Number 15-601011*

[2] *Avaya IP Office R10.0 Doc library*

Product Documentation for Flexi can be obtained from Datatal AB at: <http://www.datatal.se>

©2017 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.