



Avaya Solution & Interoperability Test Lab

Application Notes for configuring Datatal AB Flexi with Avaya IP Office Server Edition R10.1 - Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Datatal AB Flexi to interoperate correctly with Avaya IP Office Server Edition R10.1.

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

These Application Notes describe the configuration steps required for Datatal AB Flexi to interoperate correctly with Avaya IP Office Server Edition 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. 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 four major products within the same server with shared administration.

- Flexi Tid
- Flexi Presentity
- Flexi CC
- Flexi Wonderphone

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.

Flexi WonderPhone is a softphone with integrated voicemail, presence and contacts. WonderPhone is a separate platform but shares information with Flexi, i.e., currently making it necessary to also have a Flexi system in order for WonderPhone to function. Both platforms can be installed on the same server.

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. For testing with IP Office Server Edition one SIP trunk was configured connecting the Flexi server to the IP Office Server Edition Primary server. Two TAPI connections are required connecting to two separate Datatal Flexi servers. Each of these connections monitors stations on the IP Office Server Edition and the IP Office IP500 V2 separately.

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.

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 Datatal Flexi did not include use of any specific encryption features as requested by Datatal.

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.
- Testing Flexi Wonderphone – Making calls to and from the WonderPhone application.
- Testing Flexi Operator – making calls to and from the Operator.

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 Server Edition running on a virtual platform as the primary server with an IP Office IP500 V2 running as the secondary expansion server. The Datatal Flexi solution has two connections to IP Office, a SIP Trunk connected to the IP Office Server Edition and a Telephony Application Programming Interface (TAPI) connected to both the Server Edition and the IP500V2 which enables Datatal Flexi to control a telephone via IP Office, to act as the Flexi Tid/Contact Center agent.

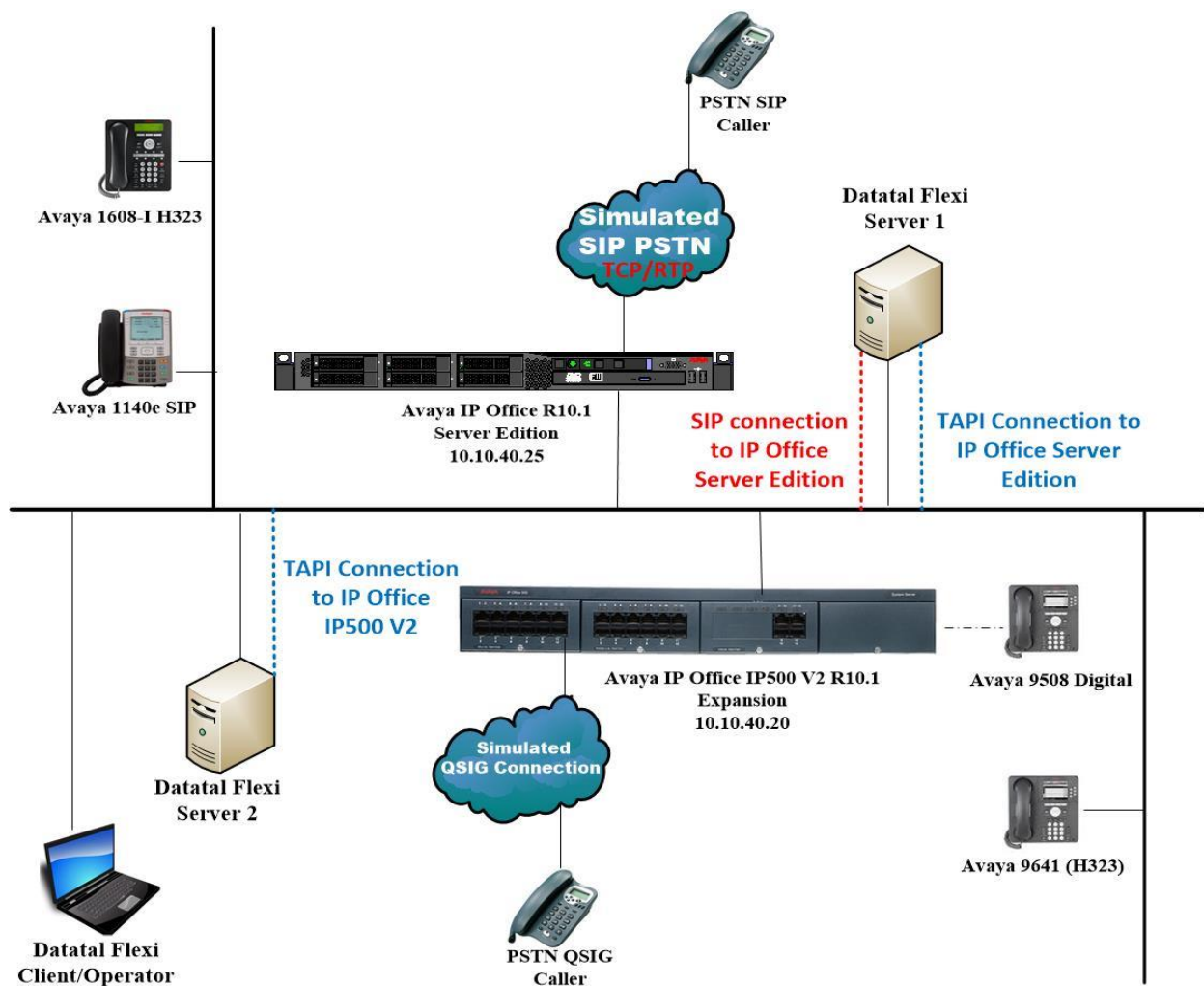


Figure 1: Avaya IP Office and Datatal AB Flexi reference configuration

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya IP Office 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	R1.3.5
Avaya 9641 H323 Deskphone	R6.6115
Avaya 1140e SIP Deskphone	R04.04.28.00
Avaya 9508 Digital Deskphone	V0.6
Datatal Flexi platform running on Microsoft Windows Server 2012 x64 R2	Version 5.12.3

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 describes the details on how to configure both the IP Office Server Edition (Primary) and IP Office IP500 V2 (Expansion) to work with Datatal Flexi. 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**.

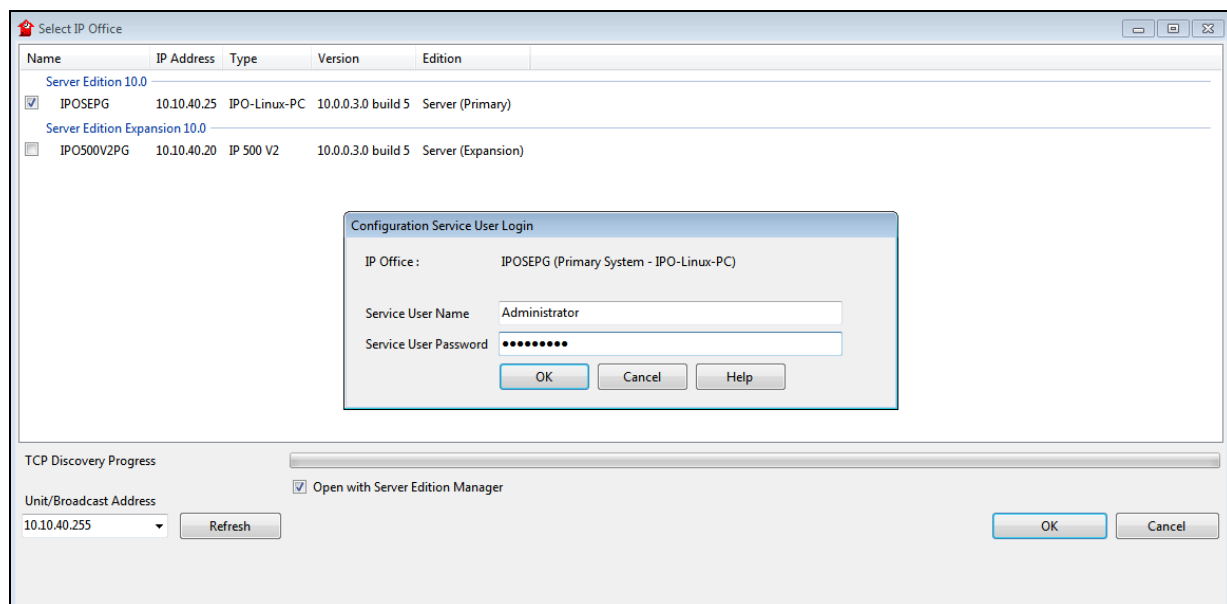
The configuration operations described in this section can be summarized as follows:

- Launch Avaya IP Office Manager (Administration)
- Display LAN Properties
- Create SIP Trunk
- Configure Incoming Call Route
- Create Short Code (Call Routing)
- Create User
- Check Extension Properties
- Save Configuration

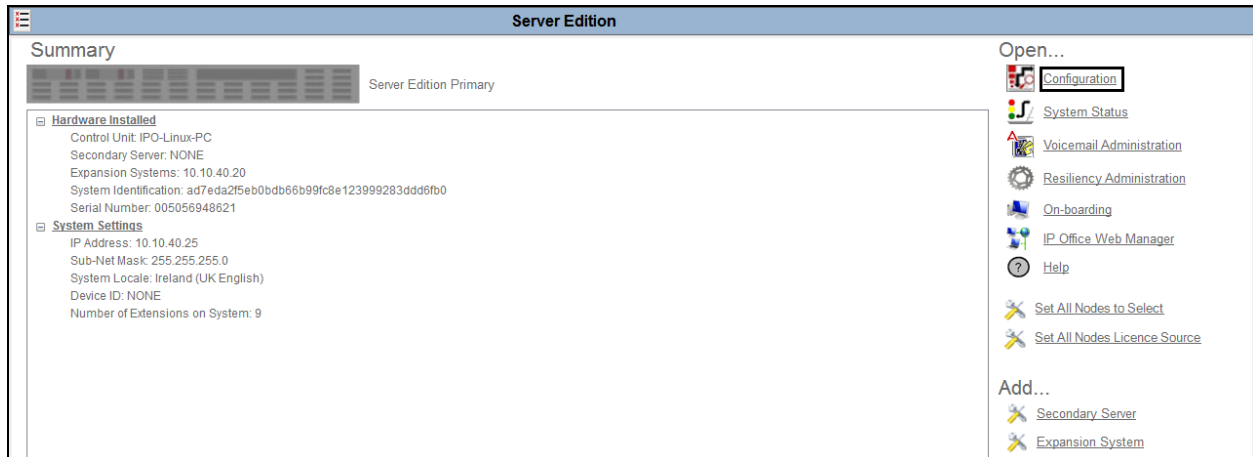
Note: Only the unique prompts are shown in the screen captures below, all other inputs can be left at default.

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). Tick on the Server Edition as shown below and enter the appropriate credentials. Click on the **OK** button.

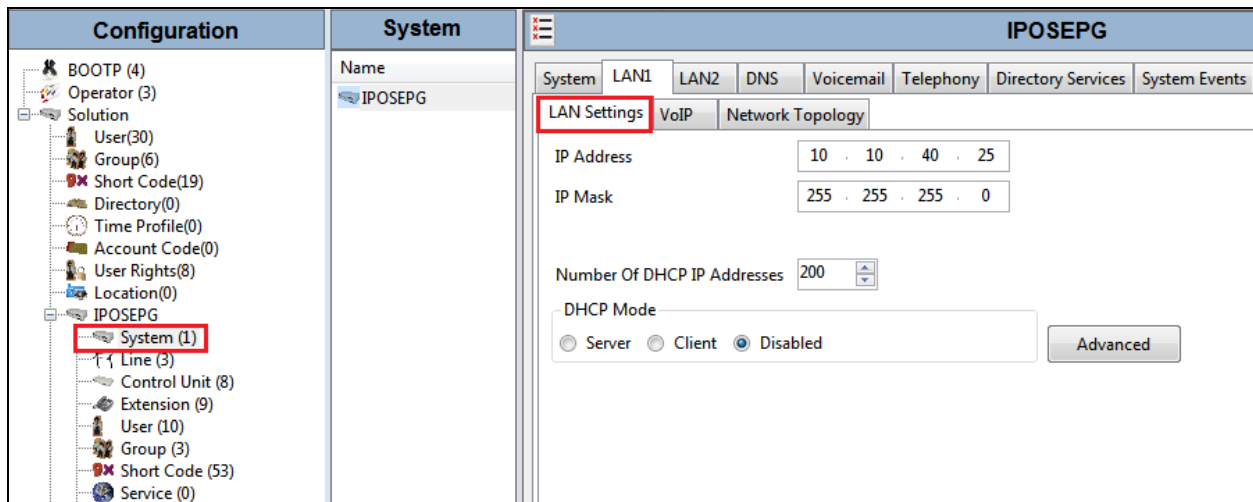


Click on **Configuration** at the top right of the page as shown, to receive the IP Office configuration.



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 the setup in both **Section 6.1** and **Section 6.2**.



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

The screenshot shows the IPOSEPG configuration interface with the VoIP tab active. Key settings include:

- 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**: ☐
- SIP Domain Name**: devconnect.local
- SIP Registrar FQDN**: (empty)
- Layer 4 Protocol**:
 - ☒ **UDP** (UDP Port: 5060)
 - ☒ **TCP** (TCP Port: 5060)
 - ☒ **TLS** (TLS Port: 5061)
- Challenge Expiry Time (secs)**: 10
- RTP**:
 - Port Number Range**: Minimum 40750, Maximum 50750
 - Port Number Range (NAT)**: Minimum 40750, Maximum 50750
 - Enable RTCP Monitoring on Port 5005**: ☒
 - RTCP collector IP address for phones**: 0.0.0.0
 - Keepalives**:
 - Scope**: RTP-RTCP
 - Periodic timeout**: 30
 - Initial keepalives**: Enabled

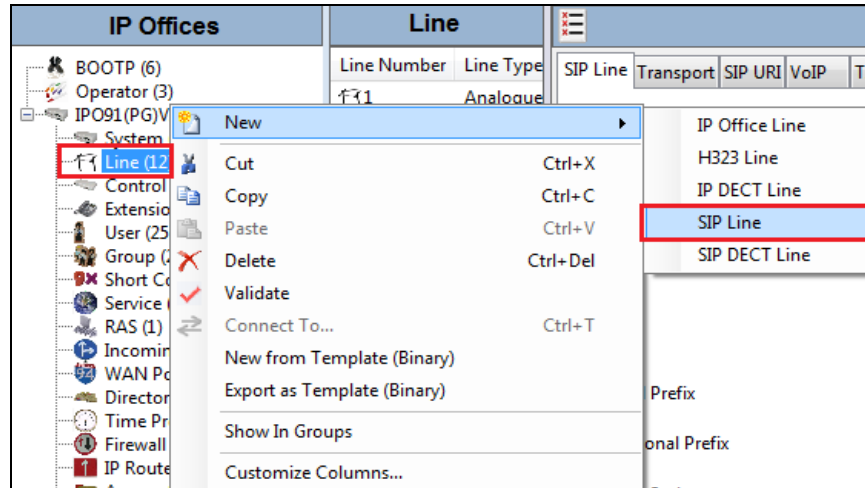
The **Codec** and **DTMF** settings can be changed under the **VoIP** tab as shown below.

The screenshot shows the IPOSEPG configuration interface with the VoIP Security tab active. Key settings include:

- Ignore DTMF Mismatch For Phones**: ☒
- Allow Direct Media Within NAT Location**: ☐
- 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 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 **Auto** 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 21' configuration window. The window has tabs for SIP Line, Transport, SIP URI, VoIP, SIP Credentials, SIP Advanced, and Engineering. The SIP Line tab is active. Fields include: Line Number (21), In Service (checked), ITSP Domain Name (devconnect.local), Check OOS (unchecked), Local Domain Name (empty), URI Type (SIP), Location (Cloud), Session Timers (Refresh Method: Auto, Timer (seconds): On Demand), Prefix (empty), National Prefix (0), International Prefix (00), Country Code (empty), Name Priority (System Default), Description (empty), Redirect and Transfer (Incoming Supervised REFER: Always, Outgoing Supervised REFER: Always, Send 302 Moved Temporarily: unchecked, Outgoing Blind REFER: 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**.

The screenshot shows the 'SIP Line - Line 21' configuration window with the 'Transport' tab selected. The 'ITSP Proxy Address' field contains '10.10.40.120'. Under 'Network Configuration', 'Layer 4 Protocol' is set to 'UDP' and 'Send Port' is '5060'. 'Use Network Topology Info' is set to 'None' and 'Listen Port' is '5060'. 'Explicit DNS Server(s)' are set to '0 . 0 . 0 . 0' and '0 . 0 . 0 . 0'. 'Calls Route via Registrar' is unchecked. 'Separate Registrar' is empty.

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

The screenshot shows the 'SIP Line - Line 21' configuration window with the 'SIP URI' tab selected. The table has columns: URI, Groups, Local URI, Contact, Display Name, Identity, Header, Originator Number, Send Caller ID, Diversion Header, Credential, and Max Calls. On the right, there are 'Add...', 'Remove', and 'Edit...' buttons.

In the subsequent window, enter the following:

- **Local URI** Enter **Auto**
- **Contact** Enter **Auto**
- **Display Name** Enter **Auto**
- **Identity** Select **None** from the dropdown menu
- **Header** Select **P Asserted ID** from the dropdown menu
- **Send Caller Id** Select **None** from the dropdown menu
- **Diversion Header** Select **None** from the dropdown menu
- **Incoming Group** Set this to the line **21**
- **Outgoing Group** Set this to the line **21**
- **Max Sessions** This will be determined by the license for number of SIP trunks available

Click on **OK**, once all is inputted correctly.

Edit URI

Local URI: Auto

Contact: Auto

Display Name: Auto

Identity: None

Header: P Asserted ID

Forwarding And Twinning

Originator Number:

Send Caller Id: None

Diversion Header: None

Registration: 0: <None>

Incoming Group: 21

Outgoing Group: 21

Max Sessions: 10

OK

Cancel

Click on the **VoIP** tab and choose the Codec's that are required and compatible. Tick the **Re-invite Supported** box. **DTMF Support** was set to **RFC 2833/RFC4733** 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 21' configuration window with the 'VoIP' tab selected. The window has a tabbed interface with 'SIP Line', 'Transport', 'SIP URI', 'VoIP', 'SIP Credentials', 'SIP Advanced', and 'Engineering'. The 'VoIP' tab is active, displaying codec selection options. On the left, a 'Codec Selection' dropdown is set to 'System Default'. Below it, there are two lists: 'Unused' (empty) and 'Selected' (containing G.711 ALAW 64K, G.711 ULAW 64K, G.722 64K, and G.729(a) 8K CS-ACELP). Between these lists are buttons for moving items: '>>>', '↑', '<<<', '↓', and '>>>'. To the right of the codec lists are several checkboxes: 'Local Hold Music' (unchecked), 'Re-invite Supported' (checked), 'Codec Lockdown' (unchecked), 'Allow Direct Media Path' (unchecked), 'Force direct media with phones' (unchecked), and 'PRACK/100rel Supported' (unchecked). At the bottom, there are three dropdown menus: 'Fax Transport Support' set to 'None', 'DTMF Support' set to 'RFC2833/RFC4733', and 'Media Security' set to 'Disabled'.

SIP Line - Line 21

SIP Line | Transport | SIP URI | **VoIP** | SIP Credentials | SIP Advanced | Engineering

Codec Selection: System Default

Unused

Selected

- G.711 ALAW 64K
- G.711 ULAW 64K
- G.722 64K
- G.729(a) 8K CS-ACELP

Local Hold Music

☒ Re-invite Supported

☐ Codec Lockdown

☐ Allow Direct Media Path

☐ Force direct media with phones

☐ PRACK/100rel Supported

Fax Transport Support: None

DTMF Support: RFC2833/RFC4733

Media Security: Disabled

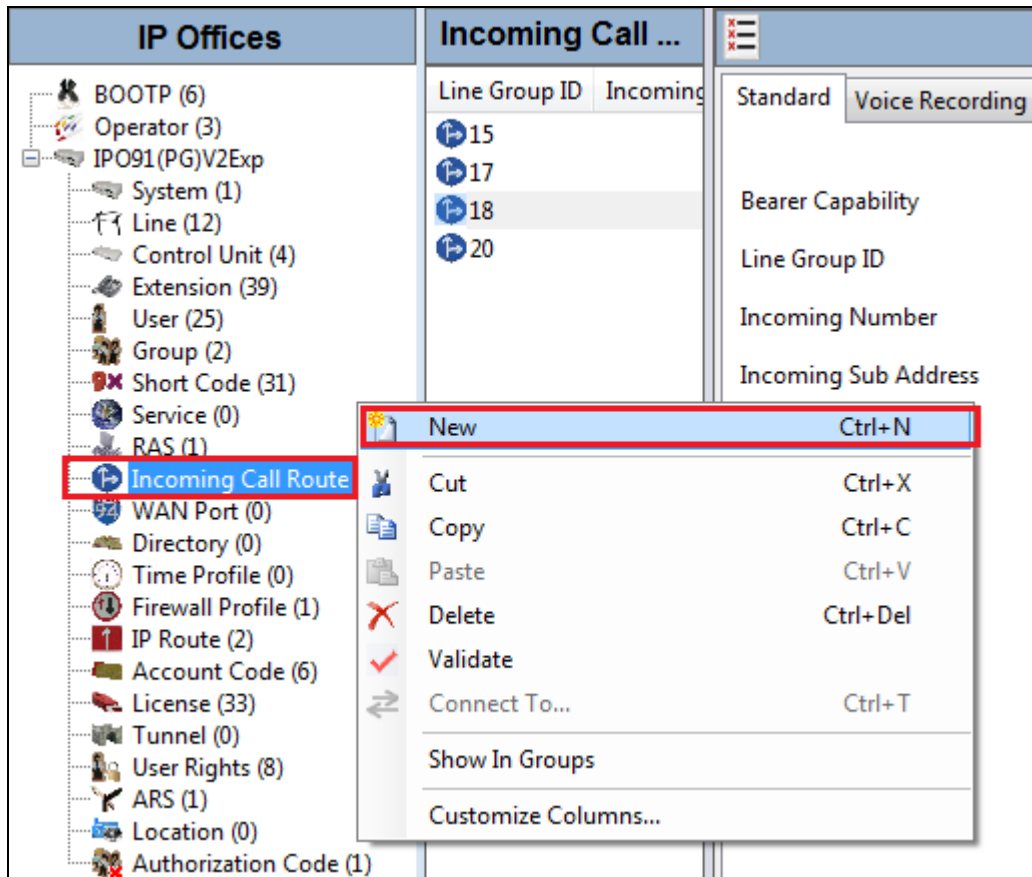
For compliance testing the values under the **SIP Advanced** tab were left as default as shown below.

The screenshot shows the 'SIP Line - Line 21' configuration window with the 'SIP Advanced' tab selected. The window is divided into several sections: Addressing, Identity, Media, and Call Control. The 'Addressing' section includes 'Association Method' (By Source IP address), 'Call Routing Method' (Request URI), and 'Suppress DNS SRV Lookups' (unchecked). The 'Identity' section includes various checkboxes for user identification, with 'Cache Auth Credentials' checked. The 'Media' section includes checkboxes for '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), and 'Indicate HOLD'. The 'Call Control' section includes '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' (Reject Call), 'Suppress Q.850 Reason Header' (unchecked), 'Emulate NOTIFY for REFER' (unchecked), and 'No REFER if using Diversion' (unchecked).

Section	Parameter	Value
Addressing	Association Method	By Source IP address
	Call Routing Method	Request URI
	Suppress DNS SRV Lookups	<input type="checkbox"/>
Identity	Use "phone-context"	<input type="checkbox"/>
	Add user=phone	<input type="checkbox"/>
	Use + for International	<input type="checkbox"/>
	Use PAI for Privacy	<input type="checkbox"/>
	Use Domain for PAI	<input type="checkbox"/>
	Swap From and PAI/Diversion	<input type="checkbox"/>
	Caller ID from From header	<input type="checkbox"/>
	Send From In Clear	<input type="checkbox"/>
	Cache Auth Credentials	<input checked="" type="checkbox"/>
	User-Agent and Server Headers	
Media	Media Connection Preservation	Disabled
	Indicate HOLD	<input type="checkbox"/>
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	Reject Call
	Suppress Q.850 Reason Header	<input type="checkbox"/>
	Emulate NOTIFY for REFER	<input type="checkbox"/>
	No REFER if using Diversion	<input type="checkbox"/>

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 '21' with three tabs: 'Standard', 'Voice Recording', and 'Destinations'. The 'Standard' tab is active. It contains the following fields and values:

Field	Value
Bearer Capability	Any Voice
Line Group ID	21
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.

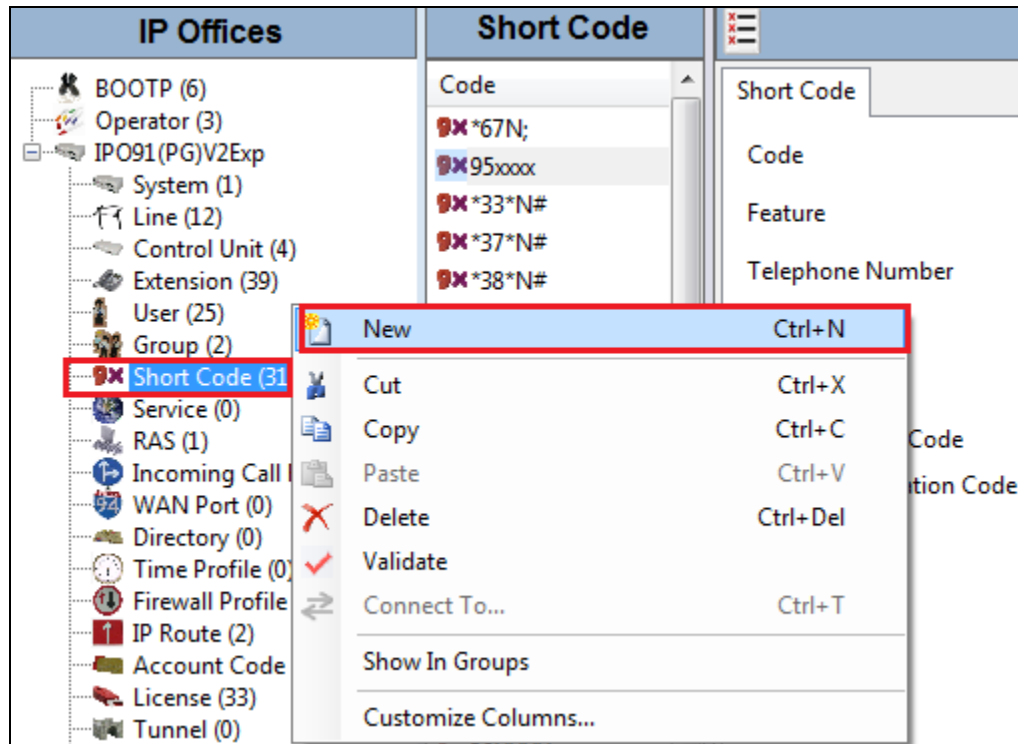
The screenshot shows the same configuration window with the 'Destinations' tab active. It displays a table with three columns: 'TimeProfile', 'Destination', and 'Fallback Extension'. The 'Default Value' row shows a period '.' in the 'Destination' field.

TimeProfile	Destination	Fallback Extension
Default Value	.	

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

5.5. Create Short Code (Call Routing)

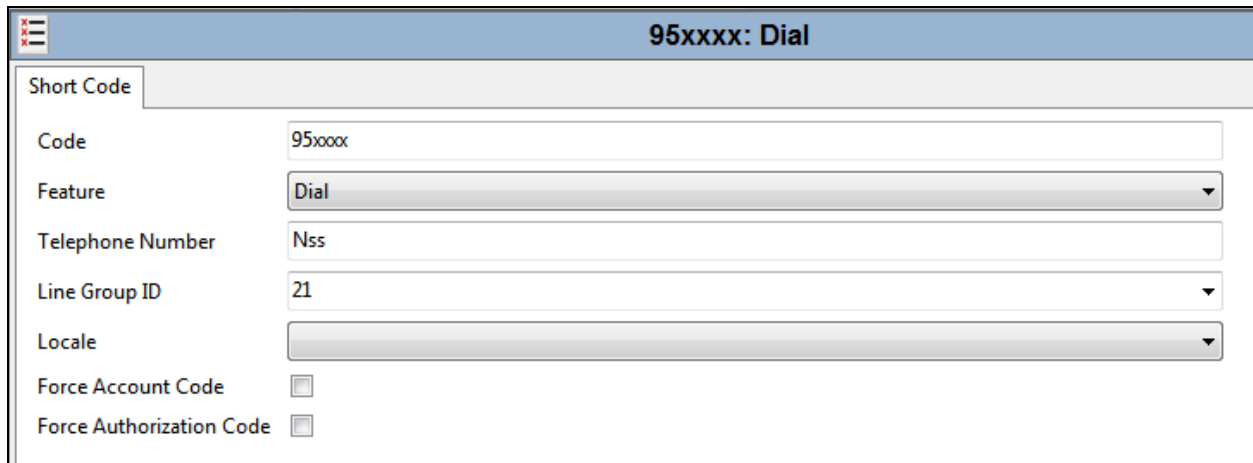
A Short Code needs to be configured on the IP Office to route calls to Flexi server. Right click on **Short Code**, and select **New**.



In the subsequent window, enter the following:

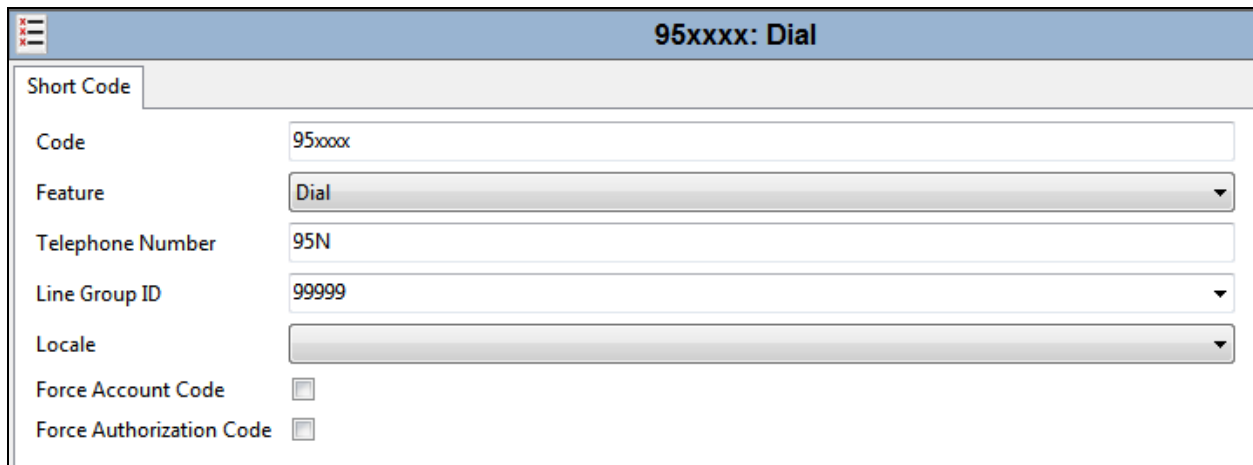
- **Code** Enter the number range that will be routed to Flexi server (during compliance testing, all numbers beginning with 95 were sent to Flexi server, therefore **95xxxx** was entered).
- **Feature** Select **Dial** from the dropdown menu.
- **Telephone Number** Enter **Nss** (Nss will send the originating 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	21
Locale	
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

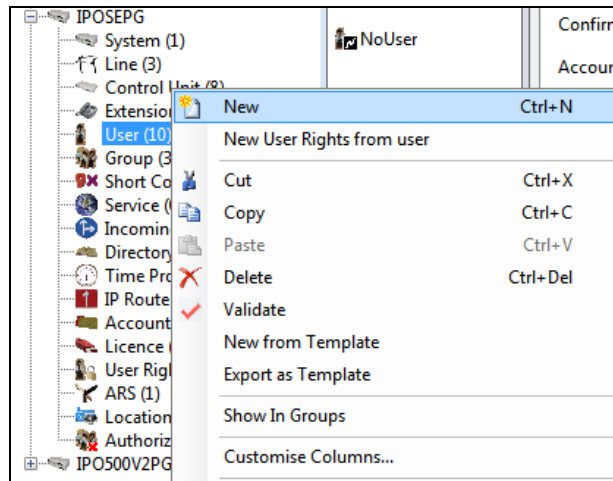
A short code must be created for the IP500 V2 side as well in order to route calls from the IP500 V2 extensions to the Server Edition and then onto the Datatal Flexi server. Like above a new Short Code is created however this time the full number is sent across to the Server Edition. In the example below **Line Group ID 99999** is used to send calls from the IP500 V2 to the Server Edition. **95** plus the number dialled is sent across ensure that this is then used to activate the Short Code that was configured above on the Server Edition.



95xxxx: Dial	
Short Code	
Code	95xxxx
Feature	Dial
Telephone Number	95N
Line Group ID	99999
Locale	
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

5.6. 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 a web-based configuration form for a user. The title bar says '5180: 5180'. The form has tabs: User, Voicemail, DND, ShortCodes, Source Numbers, Telephony, Forwarding, Dial In, Voice Recording, Button Programming, Menu Programming, and Mobility. The 'User' tab is active. The form contains the following fields and options:

- Name: 5180
- Password: [masked with dots]
- Confirm Password: [masked with dots]
- Unique Identity: [empty field]
- Audio Conference PIN: [empty field]
- Confirm Audio Conference PIN: [empty field]
- Account Status: Enabled (dropdown)
- Full Name: DatataISE
- Extension: 5180
- Email Address: [empty field]
- 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: ☐
- Exclude From Directory: ☐

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 and enter the password again for the **Login Code**. Ensure that **Force Login** is ticked.

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 ☐ Force Account Code

Login Idle Period (secs) ☐ Force Authorization Code

Monitor Group <None> ☐ Incoming Call Bar

Coverage Group <None> ☐ Outgoing Call Bar

Status on No-Answer Logged On (No change) ☐ Inhibit Off-Switch Forward/Transfer

Privacy Override Group <None> ☐ Can Intrude

Reset Longest Idle Time ☒ Cannot be Intruded

☒ All Calls ☐ Can Trace Calls

☐ External Incoming ☐ Deny Auto Intercom Calls

Once **OK** is ticked 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 ☐ Force Account Code

Login Idle Per ☐ Force Authorization Code

Monitor Gro ☐ Incoming Call Bar

Coverage Gro ☐ Outgoing Call Bar

Status on No- ☐ Inhibit Off-Switch Forward/Transfer

Privacy Overr ☐ Can Intrude

Reset Longe ☒ Cannot be Intruded

☒ All Calls ☐ Can Trace Calls

☐ External I ☐ Deny Auto Intercom 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

5.7. Check Extension Properties

Once the SIP extension has been successfully created in **Section 5.6**, open the extension configuration, select **Extension** in the left window and select the required extension number. In the main window **3rd Party Auto Answer** must be set to **RFC 5373** for the WonderPhone to work properly. Direct Media Path can be set on/off in the extension properties. This will allow RTP to be sent directly between devices. **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 Datatal.

SIP Extension: 11201 5180

Extn VoIP

IP Address: 0 . 0 . 0 . 0

Codec Selection: System Default

Unused

Selected

- G.711 ALAW 64K
- G.711 ULAW 64K
- G.722 64K
- G.729(a) 8K CS-ACELP

Reserve Licence: None

Fax Transport Support: None

DTMF Support: RFC2833/RFC4733

3rd Party Auto Answer: RFC 5373

Media Security: Same as System (Disabled)

☐ Requires DTMF

☐ Local Hold Music

☒ Re-invite Supported

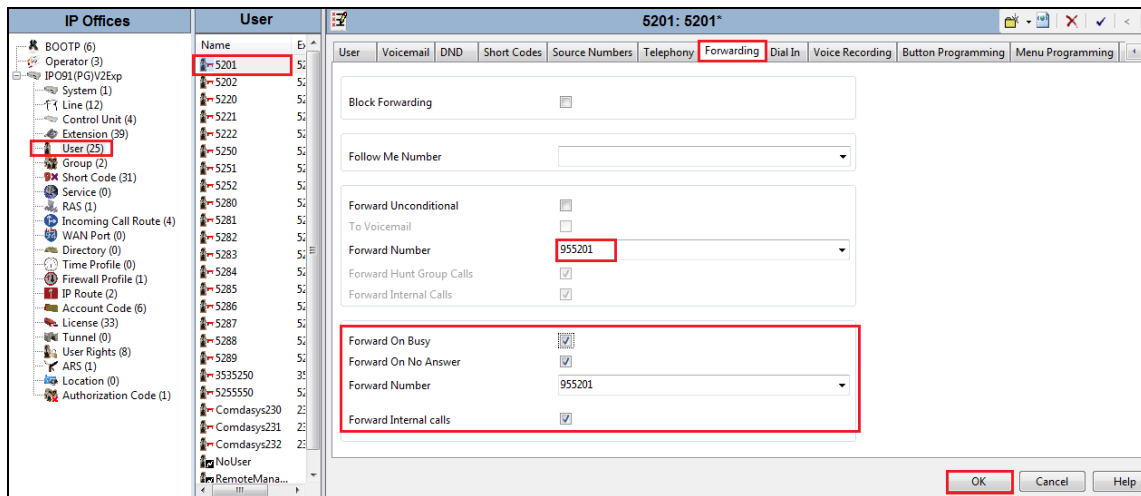
☐ Codec Lockdown

☒ Allow Direct Media Path

5.8. Configure Forwarding

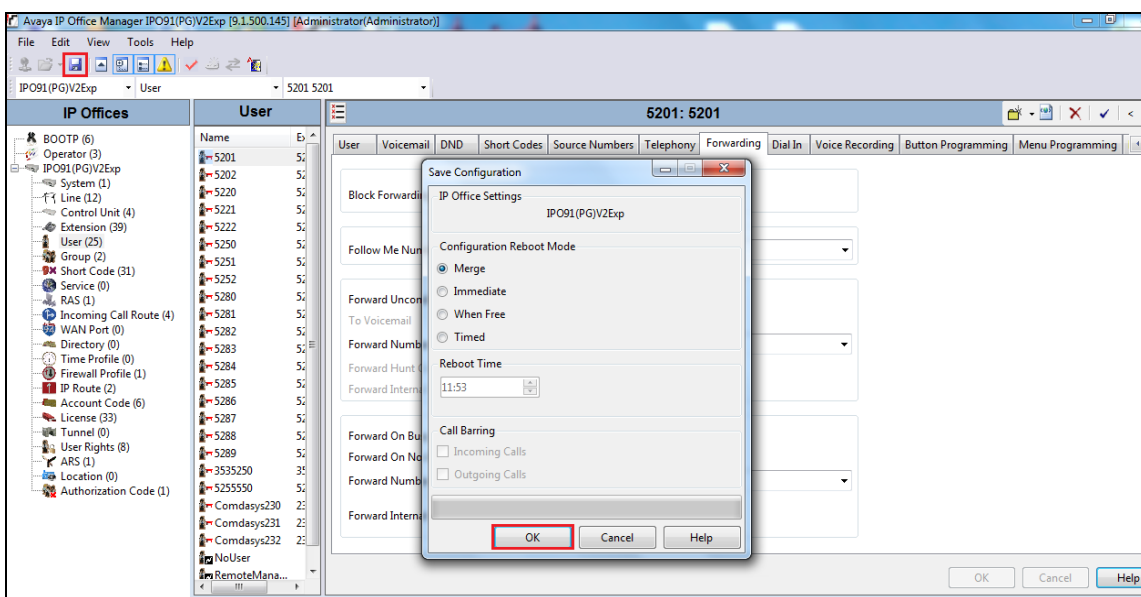
Forward On Busy and **Forward On No Answer** are configured for one of the IP Office users in order to test Flexi Presentity. To configure forwarding click on the **User** and click on the **Forwarding** tab, and in the **Forwarding Number** field enter the Short Code (as configured in **Section 5.5**)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



5.9. 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: Two separate and unique TAPI connections are required one to the IP Office Server Edition and a second to the IP Office IP500 V2 Expansion. The example below shows the connection setup to the IP500 V2.

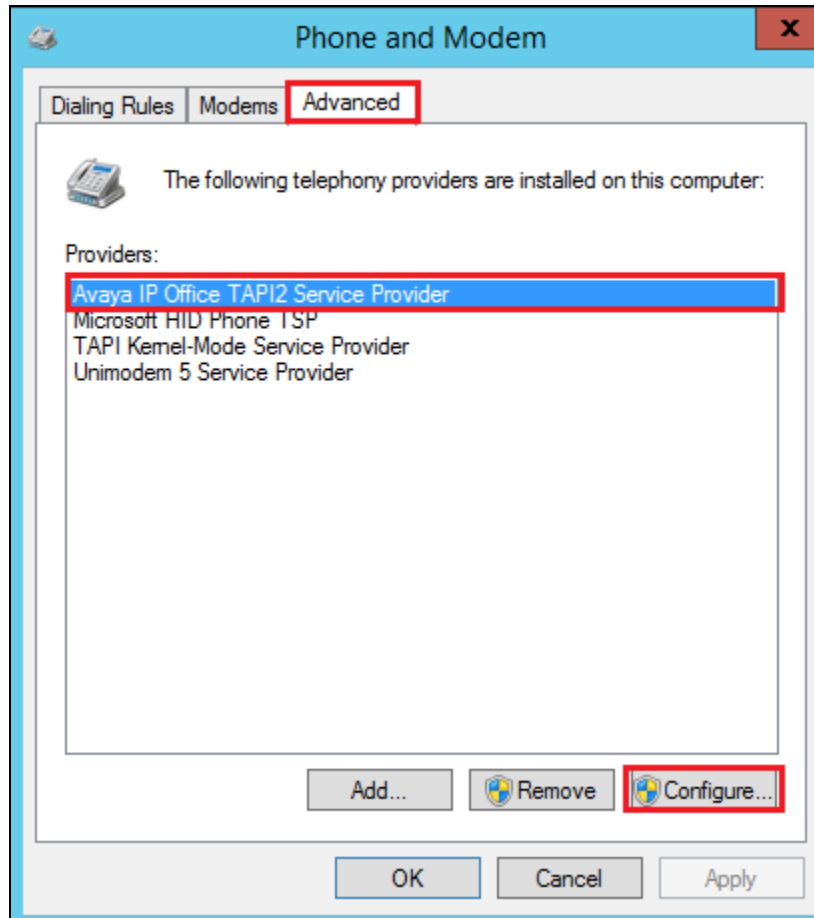
Note: It is important that the TAPI software installation was run as administrator to ensure that the application receives the correct rights to run.

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** tab 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).



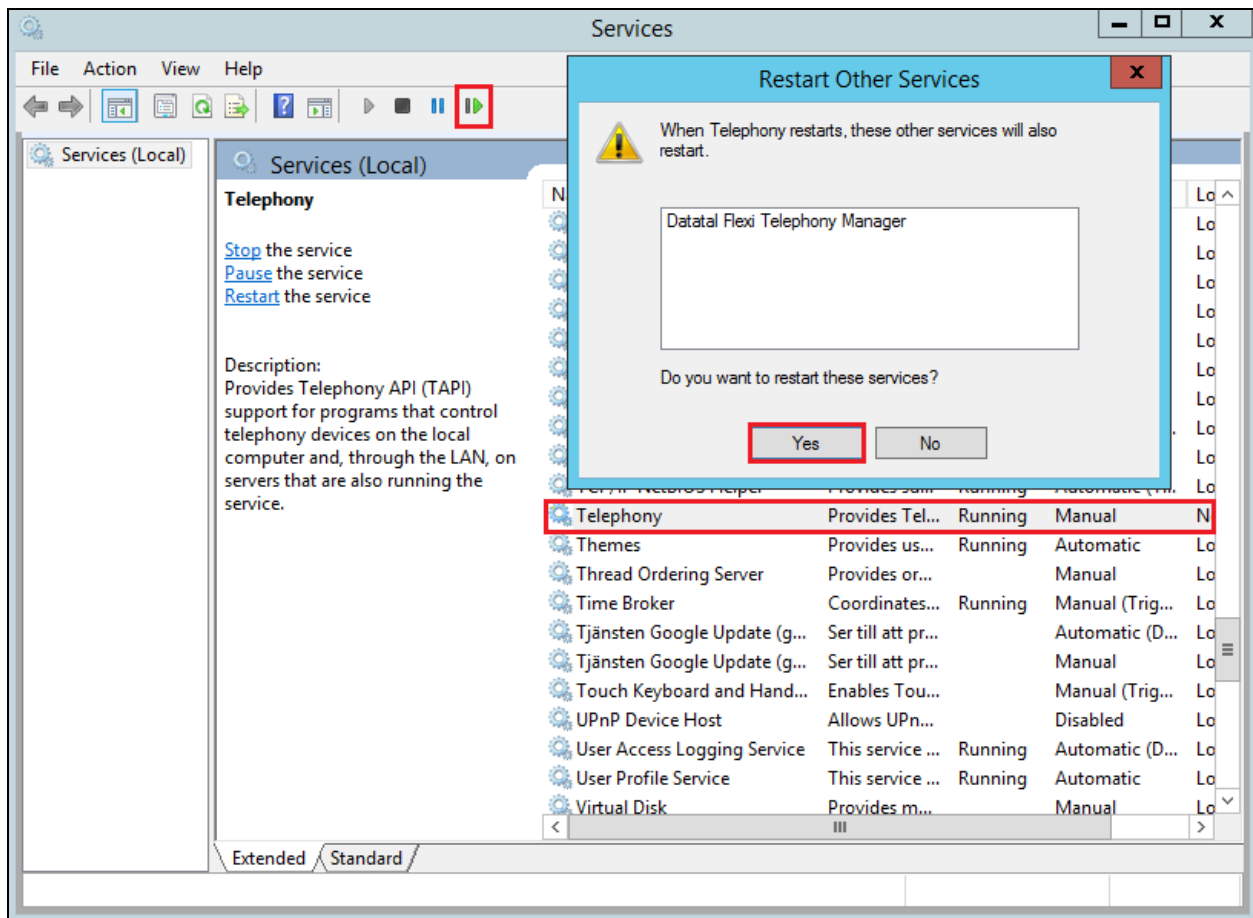
Once the **Avaya TAPI2 Configuration** window opens, enter the following:

- **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 title bar. Inside is a sub-window titled 'Avaya TAPI2 configuration'. The sub-window has a 'Switch IP Address' text box containing '10.10.40.20'. To its right is an 'OK' button highlighted with a red rectangle, and below it is a 'Cancel' button. Below these are two radio buttons: 'Single User' (unselected) and 'Third Party' (selected). Under 'Single User' are 'User Name' and 'User Password' text boxes. Under 'Third Party' is a 'Switch Password' text box containing 'xxxxxxxx', followed by three checkboxes: 'Ex Directory Users' (unchecked), 'WAV Users' (unchecked), and 'ACD Queues' (checked). At the bottom of the sub-window 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.



Go through these same steps on the other Datatal server connecting to the IP Office Server Edition.

6.2. Configure SIP Trunk

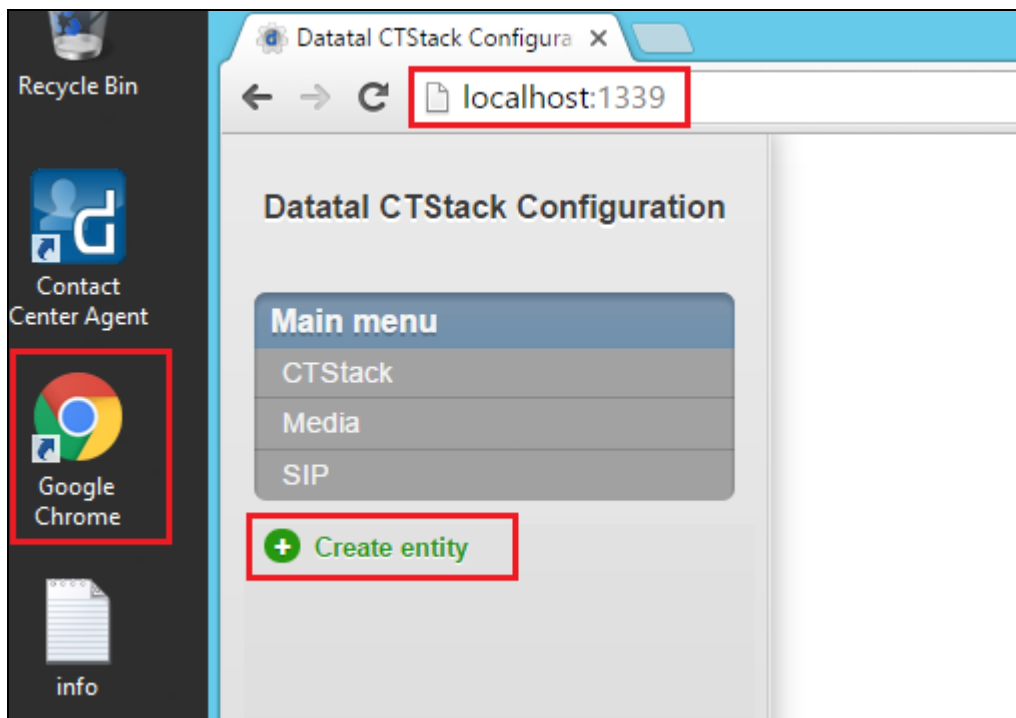
Configuration of the Flexi server is achieved using a web interface. After logging on to the Flexi server, browse to **localhost:1339** using Internet Explorer 10 or higher, Mozilla Firefox or Google Chrome web browsers. The following configuration steps were carried out during compliance testing:

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

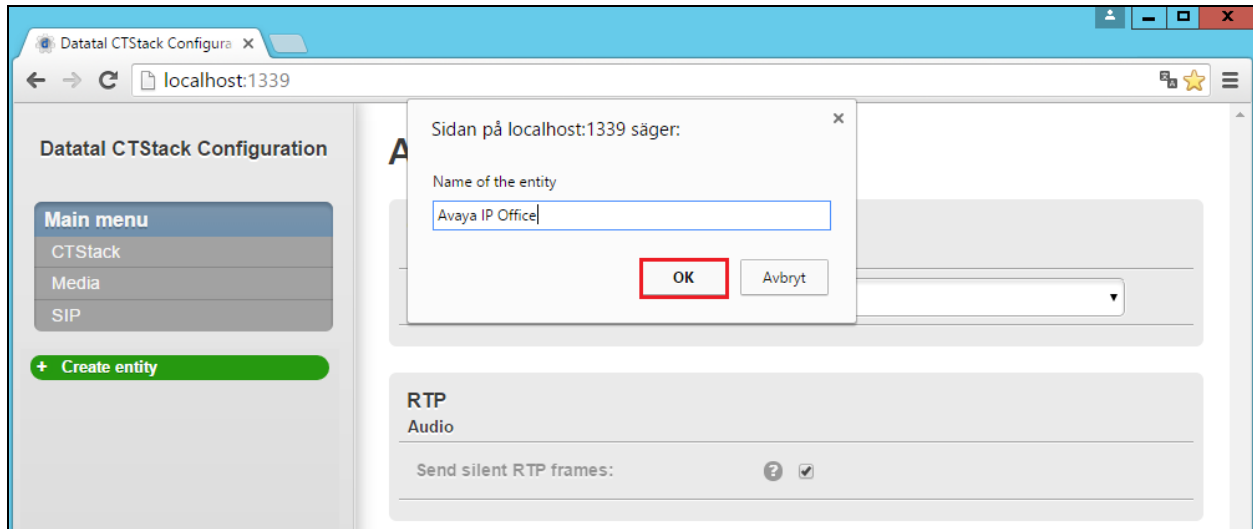
Note: It is implied that the Flexi server is pre-configured including any Licence requirements. Configuration of Flexi Presentity, Flexi CallCenter agents and Flexi Tid agents is outside the scope of these Application Notes.

6.2.1. Configure entity for Avaya IP Office

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

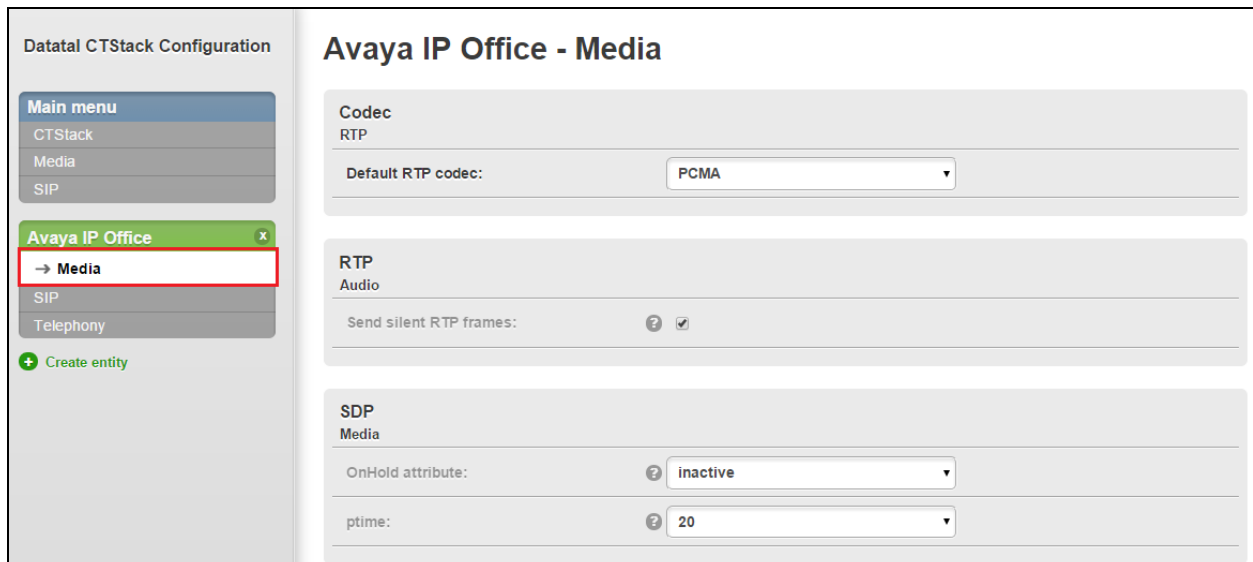


Once the new frame opens enter an informative name in the **Name of the entity** box, **Avaya IP Office** was used during compliance testing. Click the **OK** button to save.



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 'datatal ab' logo. 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). An 'Activate Windows' watermark is visible in the bottom right corner.

Dialogs	
Always create early dialogs:	<input type="checkbox"/>
Retry-After 4xx:	25
Use OPTIONS for keep-alive:	<input type="checkbox"/>

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 right, there is a watermark for 'Activate Windows'.

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

1 change(s) pending

Commit Revert

datatal ab

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

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).

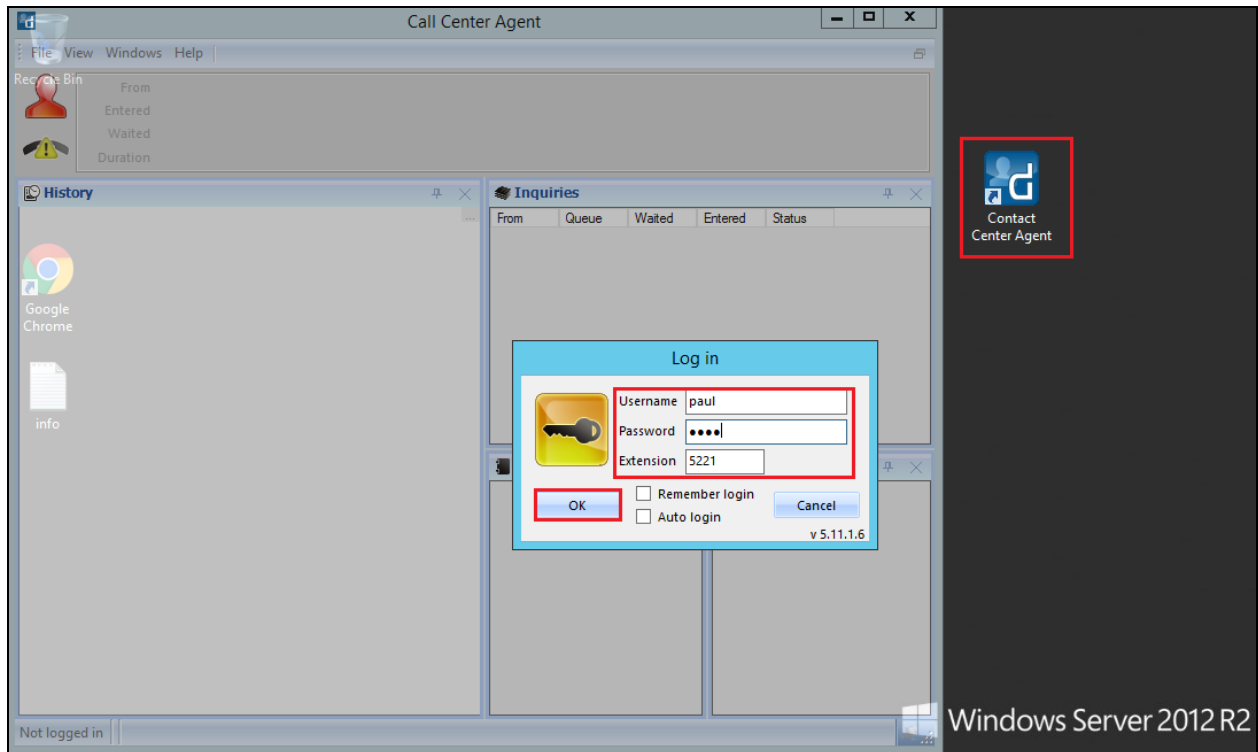
The screenshot displays the 'Datatal CTStack Configuration' web interface. On the left is a sidebar with a 'Main menu' containing 'CTStack', 'Media', and 'SIP'. Below this is a section for 'Avaya IP Office' with sub-items 'Media', 'SIP', and '→ Telephony' (which is highlighted with a red box). A 'Create entity' button is also present. At the bottom of the sidebar, it shows '1 change(s) pending' and 'Commit' and 'Revert' buttons. The main content area is titled 'SIP' and contains several configuration fields: 'Description' (string), 'INVITE expires' (25), 'Lines' (10), 'Address' (5250), 'Default domain' (devconnect.local), 'Default SIP URI host' (10.10.40.25), 'Default SIP URI port' (5060), and 'Name' (DevConnect). Below these is the 'Profile' section, which is highlighted with a red box. It contains 'Apply' (set to 'Avaya IPO (trunk)' via a dropdown), 'Current' (None), and 'Trunk mode' (checked). A 'Commit' button is visible at the bottom right of the 'Profile' section. The bottom of the interface shows the 'datatal ab' logo.

7. Verification Steps

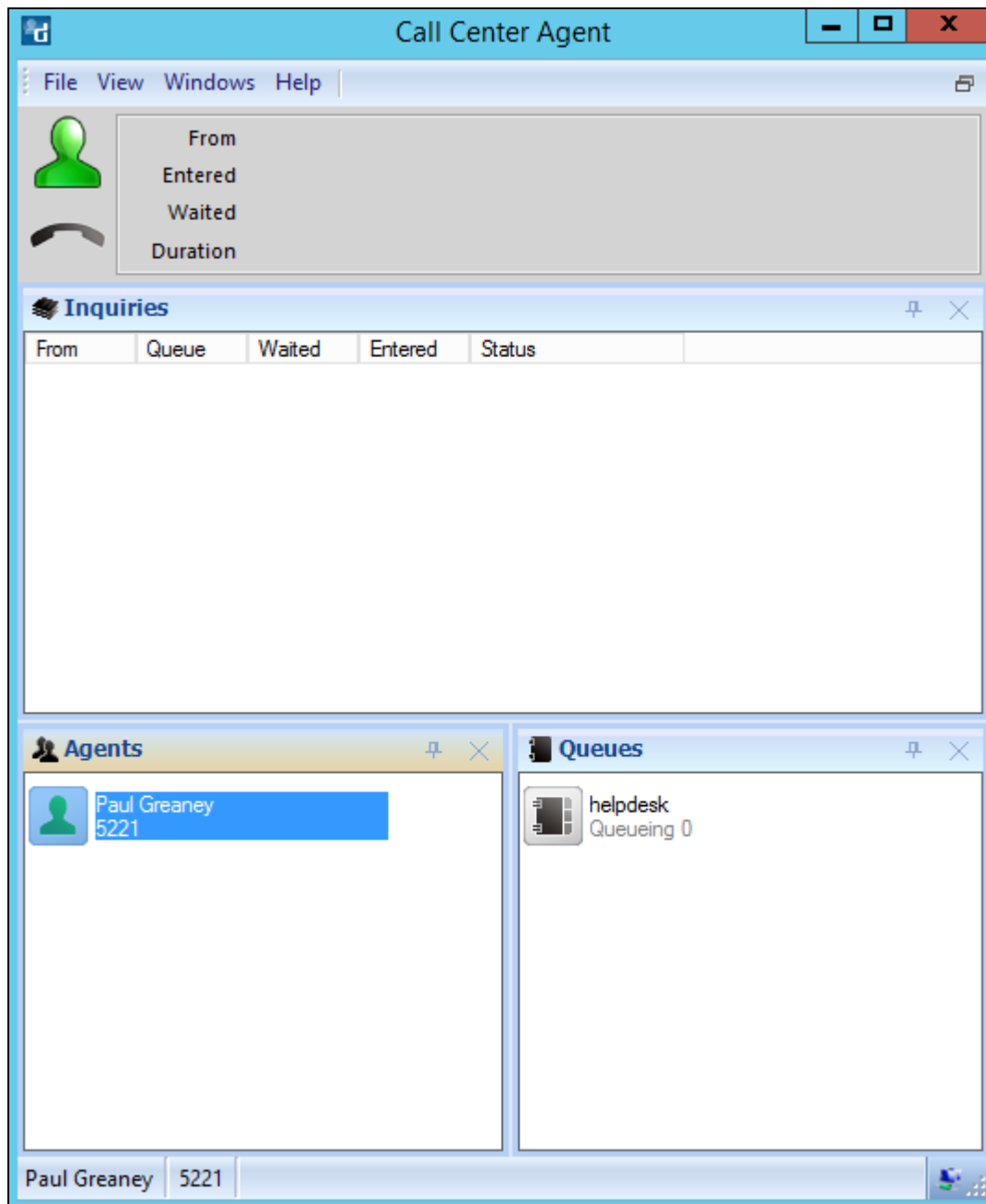
This section provides the tests that can be performed to verify correct configuration of the Avaya IP Office and Datatal AB Flexi.

7.1. Verify Flexi CC

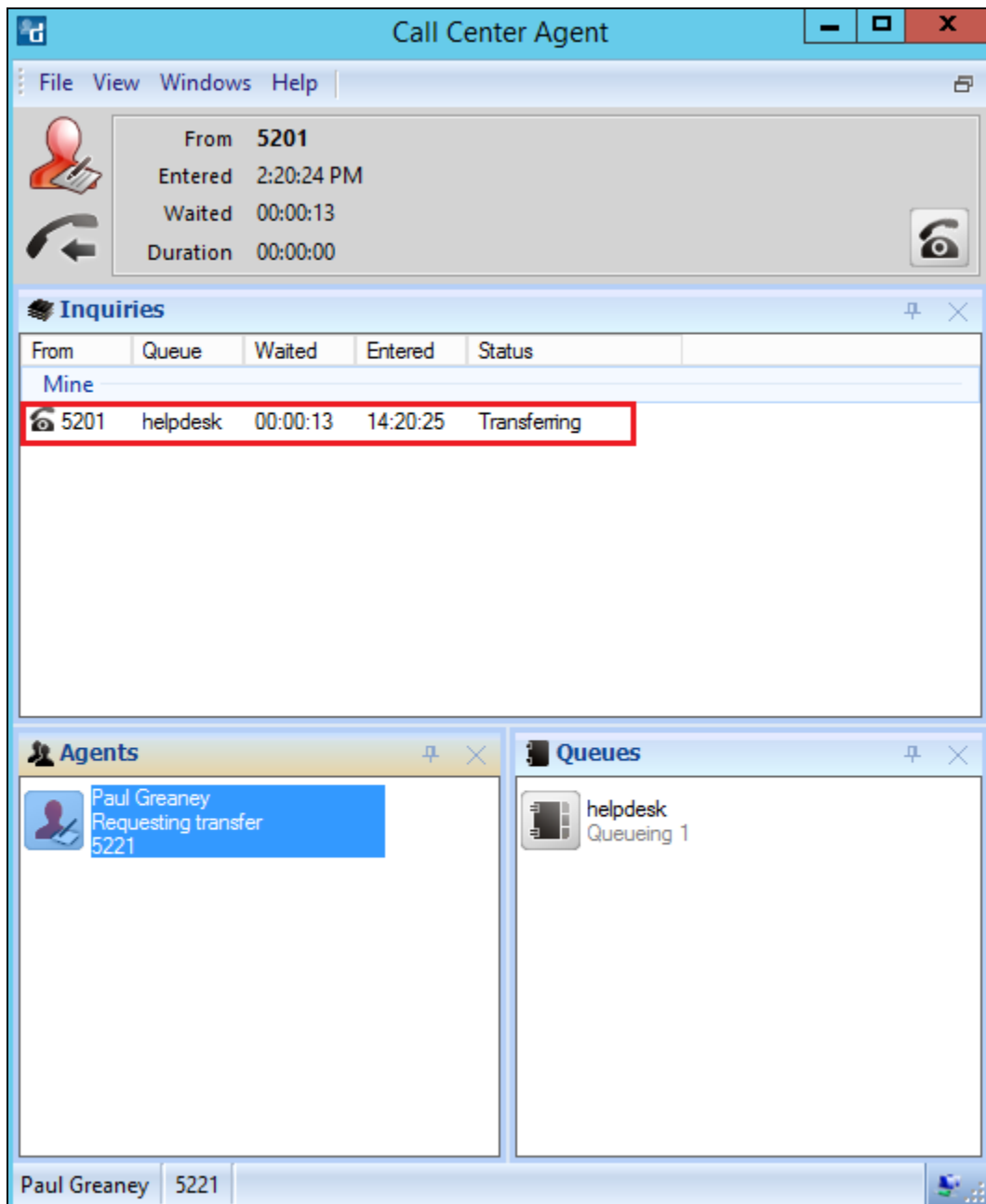
Using the shortcut on the desktop open **Contact Center Agent**, enter the appropriate credentials and click on **OK**.



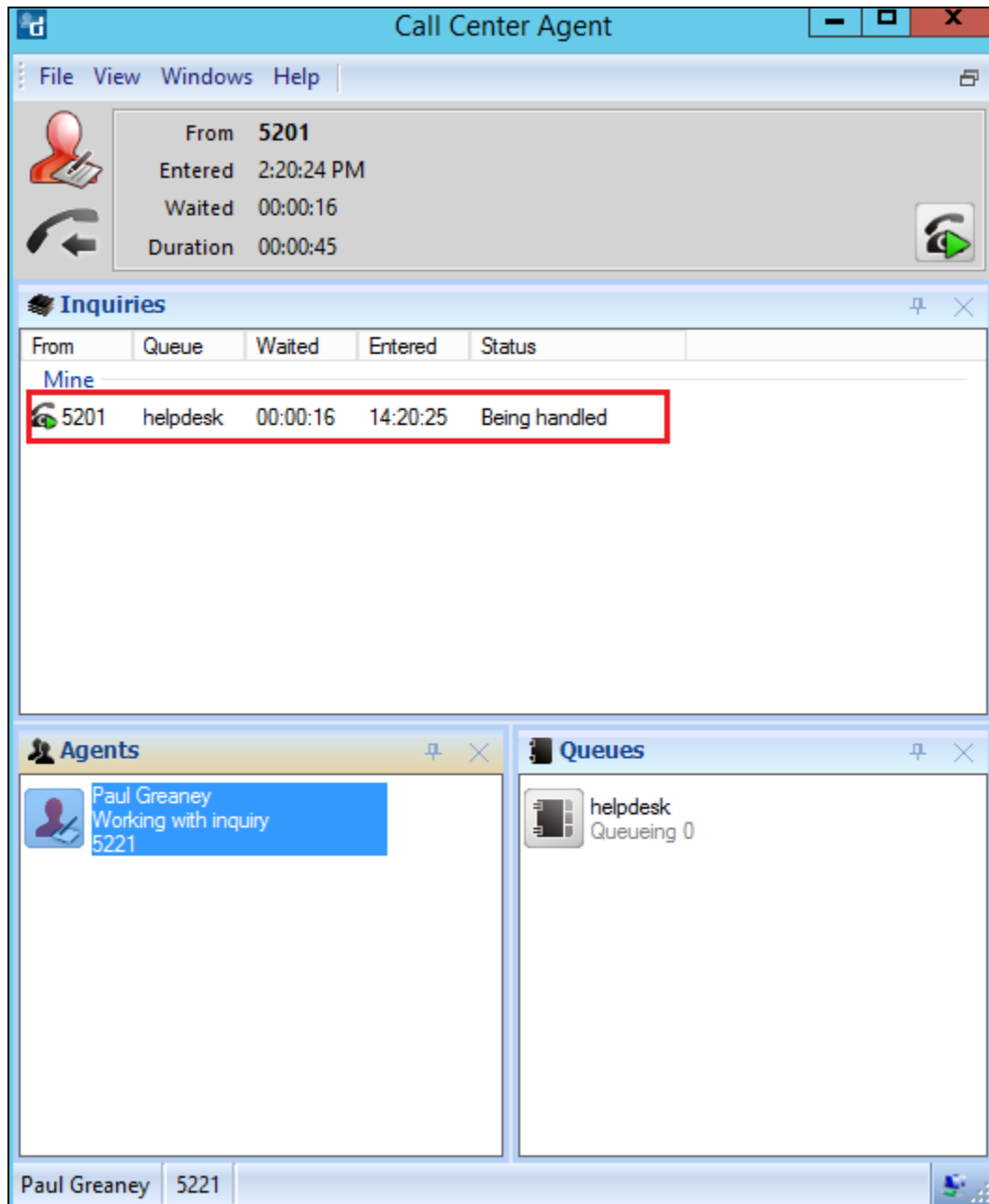
Once logged in the following screen shows the status of the agent and the queue associated with the agent.



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.

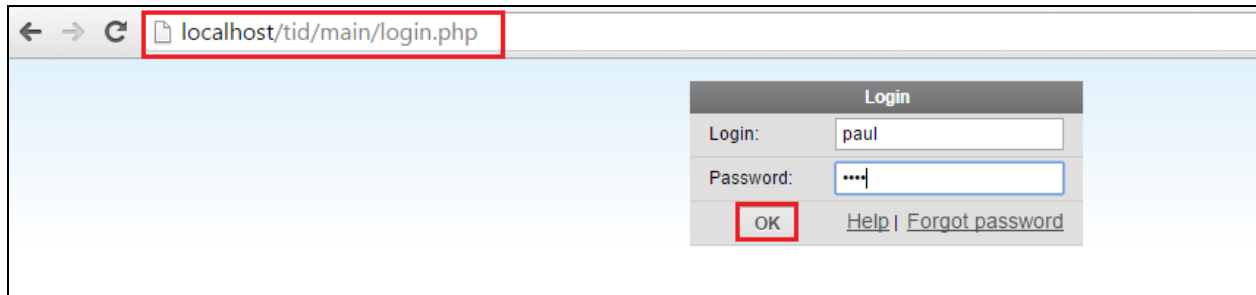


Once the call is answered this is reflected on the desktop as shown below.



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

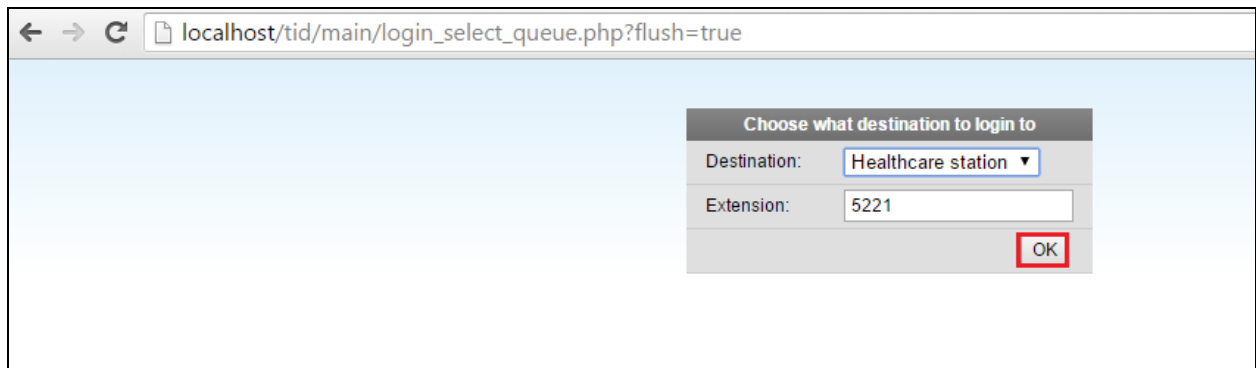
Login

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

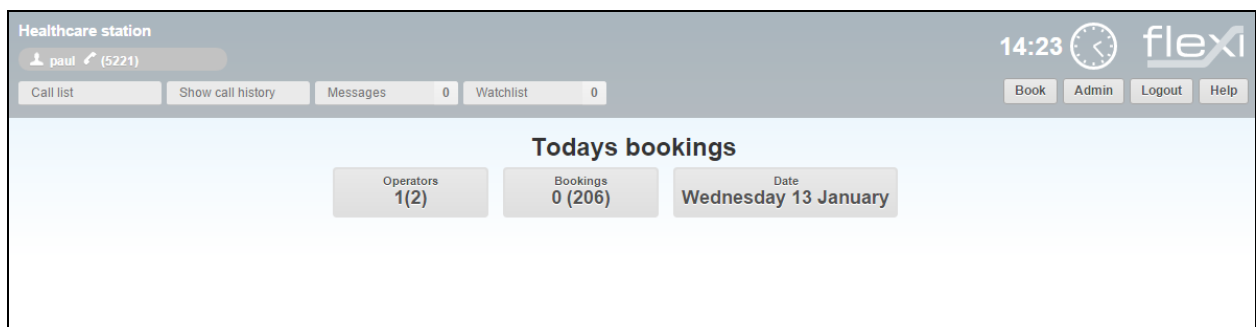
Choose what destination to login to

Destination: Healthcare station ▼

Extension: 5221

OK

The following screen is displayed once logged in correctly.



Healthcare station

paul (5221)

14:23 flexi

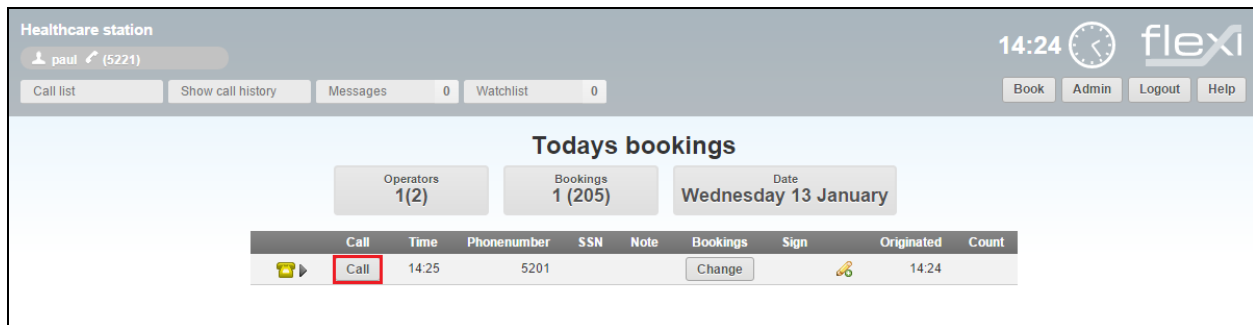
Call list Show call history Messages 0 Watchlist 0

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.





Healthcare station

14:24 flexi

Call list Show call history Messages 0 Watchlist 0 Book Admin Logout Help

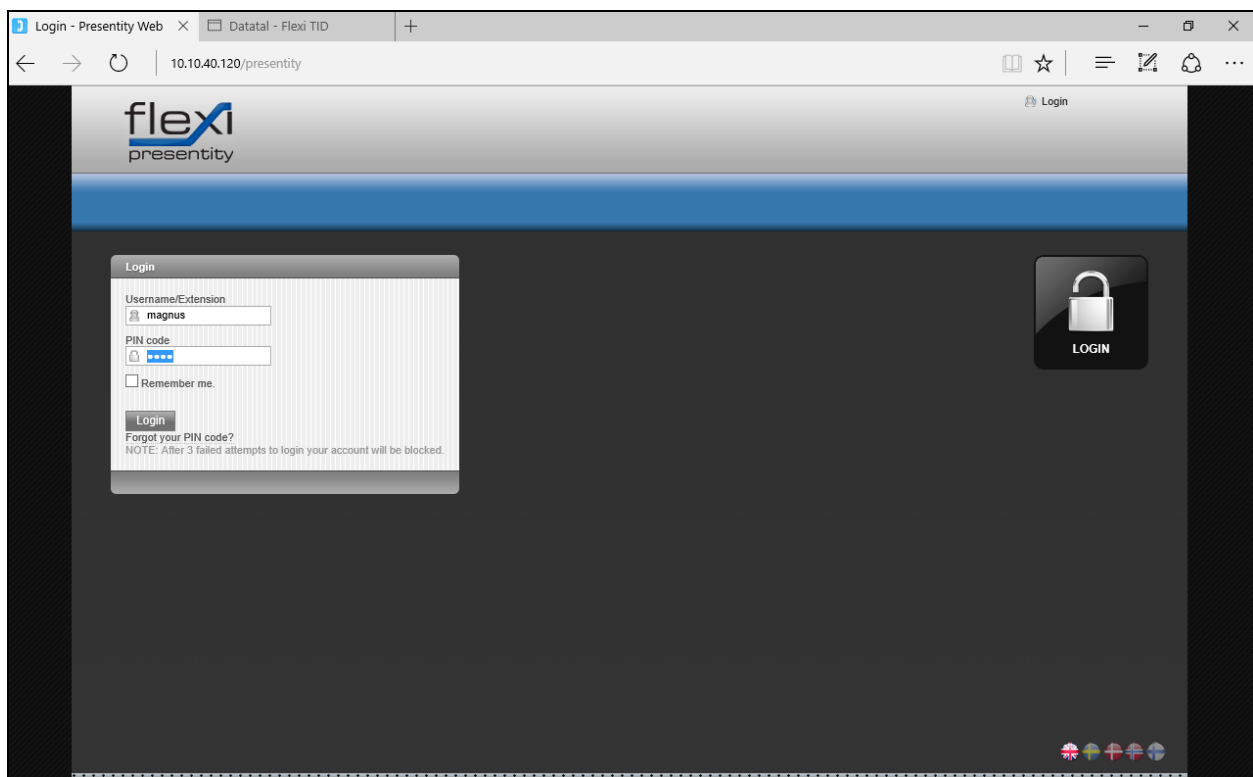
Today's bookings

Operators 1(2) Bookings 1 (205) Date Wednesday 13 January

Call	Time	Phonenumber	SSN	Note	Bookings	Sign	Originated	Count
 Call	14:25	5201			Change		14:24	

7.3. Verify Flexi Presentity

Open a web browser and navigate to the Flexi Presentity server as shown below
<http://<server>/presentity>. Enter the appropriate credentials and click on **Login**.



Login - Presentity Web x Datatal - Flexi TID +

10.10.40.120/presentity

flexi presentity

Login

Username/Extension
magnus

PIN code

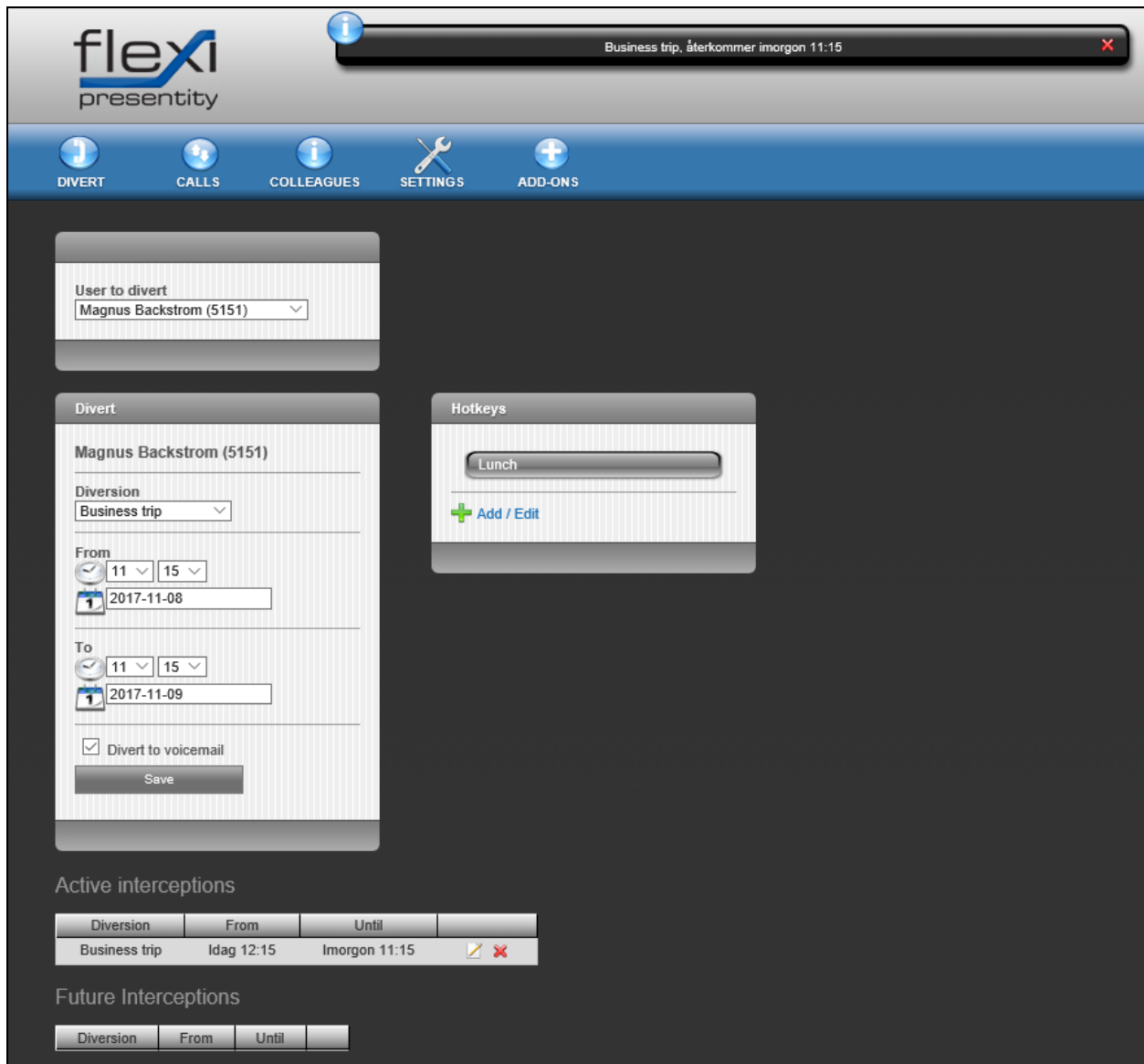
☐ Remember me.

Login

Forgot your PIN code?
NOTE: After 3 failed attempts to login your account will be blocked.

LOGIN

Once logged in the extension can be diverted as shown below. The extension **5151** is diverted to voicemail from **11:15** on the 8th of November to **11:15** on the 9th of November.



The screenshot shows the flexi presentity web interface. At the top, there is a notification bar that says "Business trip, återkommer imorgon 11:15". Below this is a navigation bar with icons for DIVERT, CALLS, COLLEAGUES, SETTINGS, and ADD-ONS. The main content area is divided into several sections:

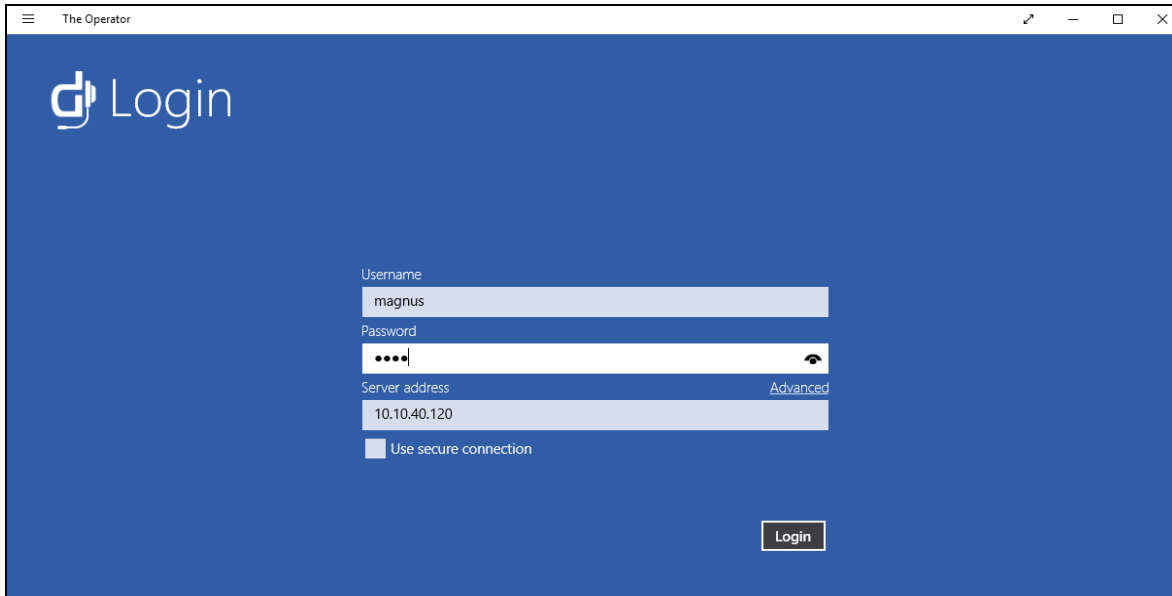
- User to divert:** A dropdown menu showing "Magnus Backstrom (5151)".
- Divert:** A section for configuring diversion for "Magnus Backstrom (5151)". It includes a "Diversion" dropdown set to "Business trip", "From" and "To" time and date pickers (11:15 on 2017-11-08 and 2017-11-09), and a checkbox for "Divert to voicemail". A "Save" button is at the bottom.
- Hotkeys:** A section with a "Lunch" button and a "+ Add / Edit" link.
- Active interceptions:** A table showing the current diversion settings.
- Future Interceptions:** A table for scheduling future diversions.

Diversion	From	Until	
Business trip	Idag 12:15	Imorgon 11:15	

Diversion	From	Until	

7.4. Verify Flexi Operator

Open Flexi Operator from the client PC as shown below, enter the appropriate credentials and the correct **Server address** and click on **Login**.



The Operator

Login

Username
magnus

Password
.....

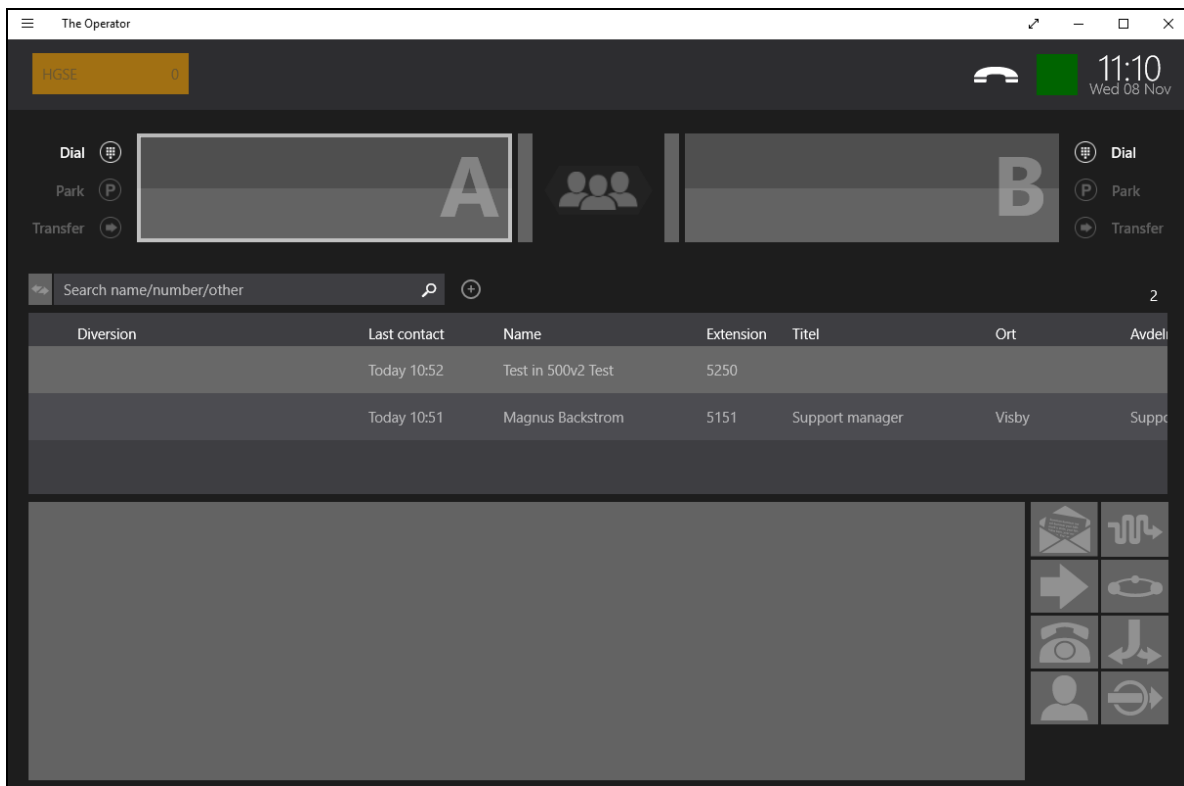
Server address
10.10.40.120

Advanced

☐ Use secure connection

Login

Upon login the following screen is shown.



The Operator

HGSE 0

11:10 Wed 08 Nov

Dial
Park
Transfer

A

B

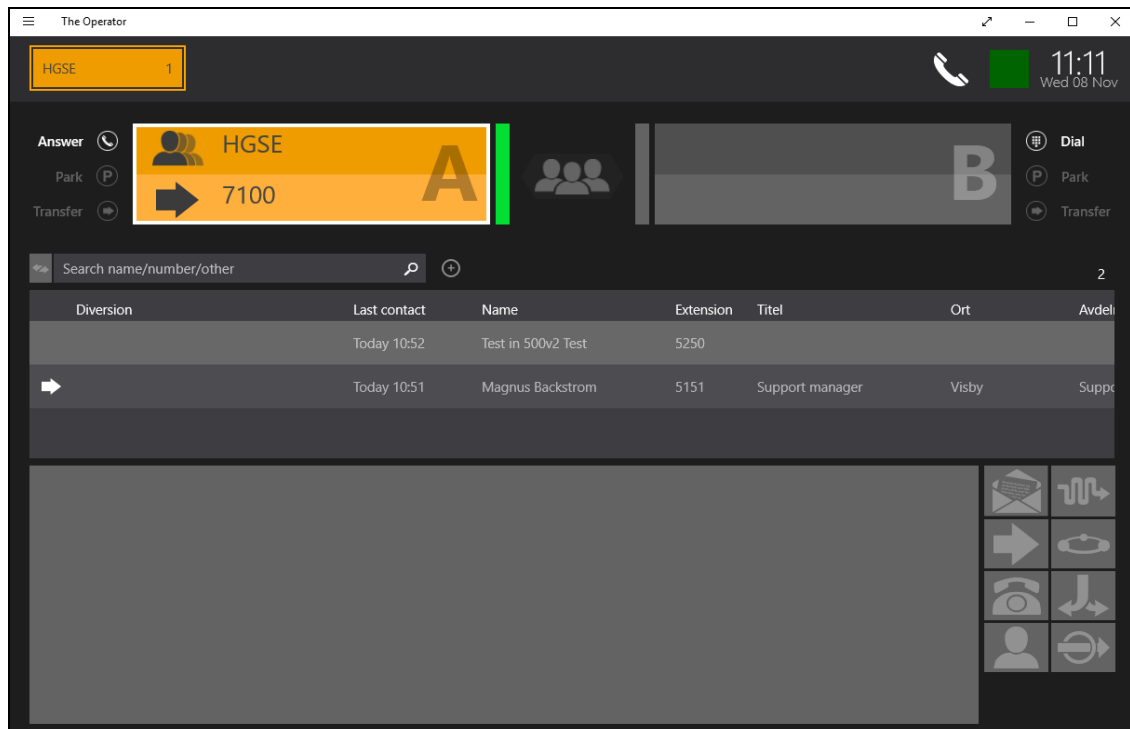
Search name/number/other

Diversion	Last contact	Name	Extension	Titel	Ort	Avdel
	Today 10:52	Test in 500v2 Test	5250			
	Today 10:51	Magnus Backstrom	5151	Support manager	Visby	Supp

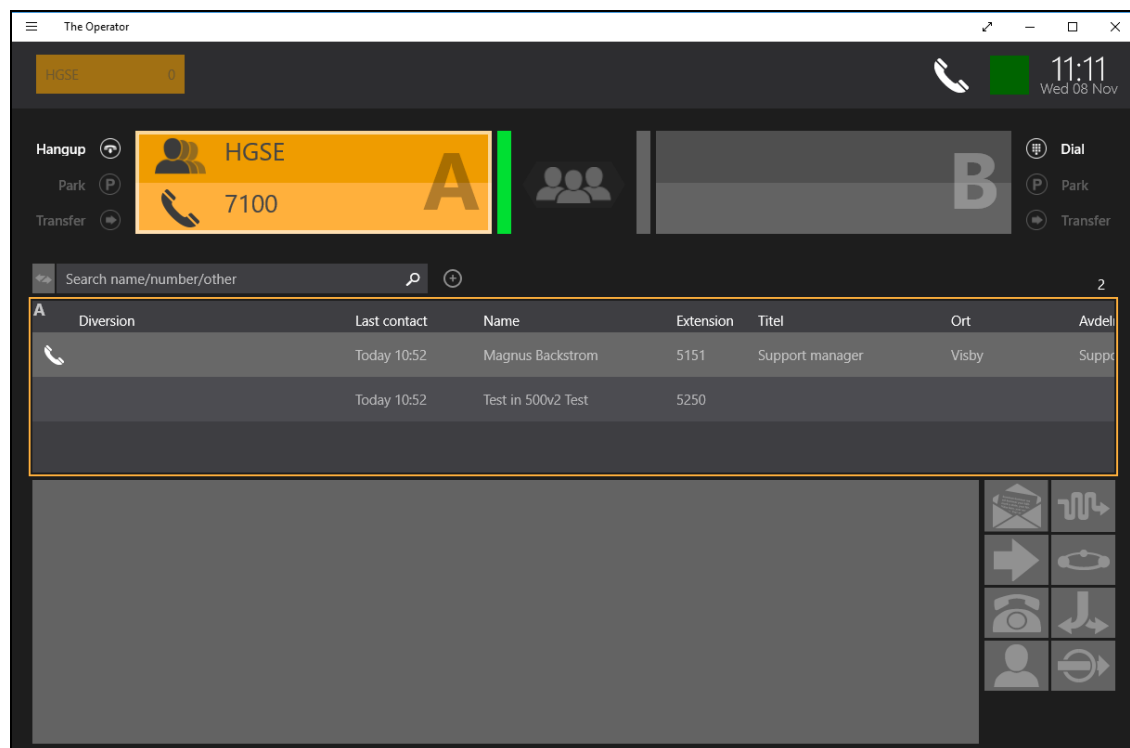
2

Icons: Envelope, Wavy arrow, Right arrow, Left arrow, Phone, Person, etc.

A call from 7100 is presented to the operator as shown, the answer button can be pressed to answer the call.

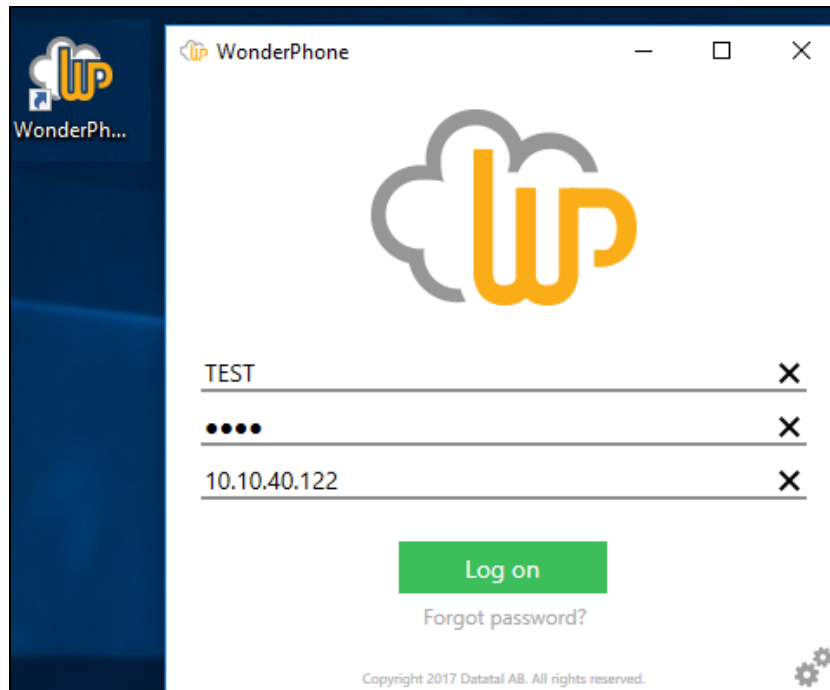


Once the call is answered it can be transferred or hung up as required.

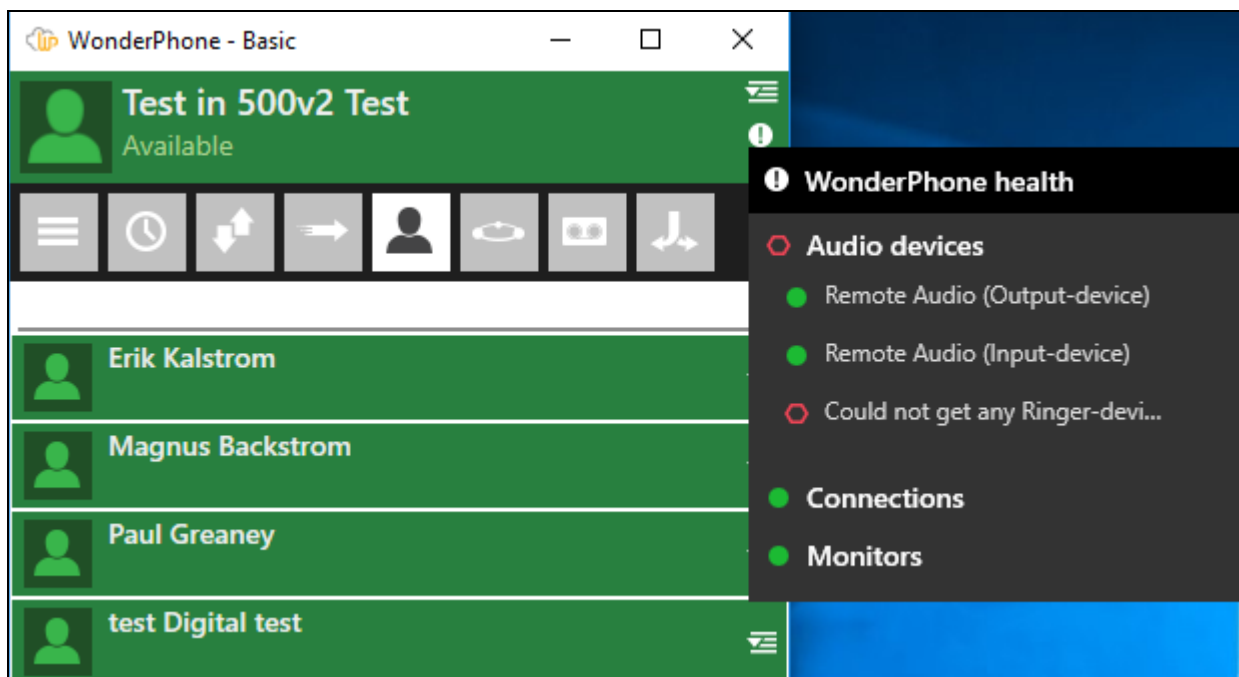


7.5. Verify Flexi Wonderphone

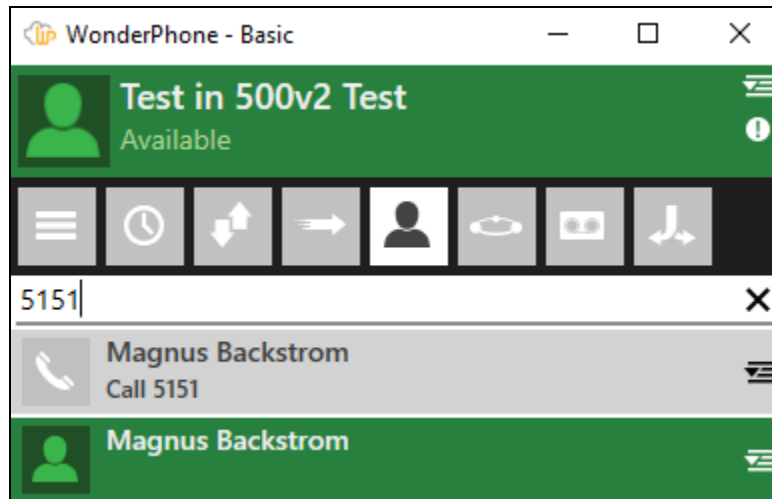
Open the **WonderPhone** application and enter the appropriate credentials and click on **Log on**.



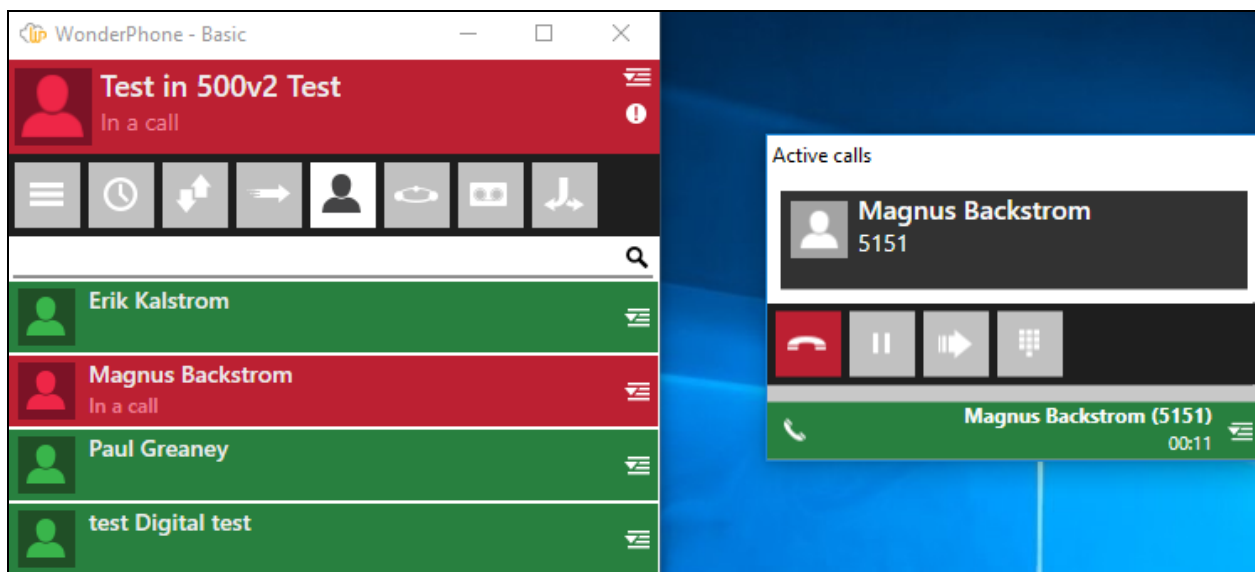
Once logged in the **!** icon can be pressed and this will show the status of the connection as shown below.



A call can be made to **5151** as shown below.



A second window is opened showing the **Active** calls.



8. Conclusion

These Application Notes describe the required configuration steps necessary Datatal AB Flexi to interoperate with Avaya IP Office Server Edition R10.1. 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>

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