



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Datatal AB Flexi with Avaya IP Office Server Edition R11.0 - Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Datatal AB Flexi to interoperate with Avaya IP Office Server Edition R11.0.

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

These Application Notes describe the configuration steps required for Datatal AB Flexi to interoperate correctly with Avaya IP Office Server Edition R11.0. 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 AB Flexi platform is an application platform for telephony and unified communication on the Swedish market and is also used in some other Nordic countries. Datatal AB Flexi platform includes three major modules 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. Customer calls are routed to the SIP Trunk to the Flexi server to book a timeslot for when they will be called back or leave a message. User can monitor queue for voice message and play back. Flexi Tid uses TAPI to make call back to agents on timeslot and calls customer before transferring to agents via SIP REFER method.

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 Presentity makes use of TAPI to activate call forwarding so that calls to user can be forwarded to Flex Presentity for coverage to Flexi voicemail. Message Waiting Lamp is also activated and de-activated through TAPI. The Flexi Presentity Web application is used as a browser interface where the employee can:

- See the presence of their colleagues
- Set their absence status on / off
- Edit their own information
- View call log (dialed, received, unanswered)

Flexi Operator which is part of the Flexi Presentity module provides the operator function and can even view or modify employee information and their presence status. Each operator can perform call control function on the phone assigned to via TAPI.

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 via SIP REFER method. Flexi CC can also handle call back, so that calling customers can schedule a call back. Flexi CC 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 connections are used to determine when an agent is free and available in order to transfer the call.

Flexi Wonderphone is a SIP softphone and a separate application all together. It integrates with IP Office as a SIP user with access to voicemail, presence and contacts. Wonderphone is a separate platform but shares information with Flexi, i.e. currently making it necessary to also have a Datatal AB Flexi system in order for Wonderphone to function. Both 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 and the softphone. 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 AB Flexi servers. Each of these connections monitors extensions on the IP Office Server Edition and the IP Office IP500 V2 separately. In this compliance testing, the Datatal Client/Operator PC is used for the other TAPI connection. So, the main server is the master Flexi and the other is a slave that connects to the master to update on the TAPI status.

Each major module is tested on the feature interaction with IP Office which included use of Flexi Operator as client and Flexi Wonderphone as softphone.

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 AB 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 major 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 with queuing, transfer and call back scenario.
- Testing Flexi Presentity – Make users absent and divert to voicemail, make inbound calls to that user's voicemail and verifying message indication lamp.
- 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 including call transfer, conference, mobile twinning and setting presence status.
- Testing Flexi Operator – Making calls to and from the Operator including call transfer (supervised and blind).

The serviceability test includes rebooting the Flexi server and then check for connectivity.

2.2. Test Results

Tests were performed to ensure 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.

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 AB Flexi solution has three connections to IP Office, a SIP Trunk connected to the IP Office Server Edition and Telephony Application Programming Interface (TAPI) connected to both the Server Edition and the IP500V2. The TAPI connected to IP500V2 function as a slave and will update the master TAPI connected to the IP Office Server Edition. Users were created on both the primary server and IP Office IP500 V2 as in the diagram.

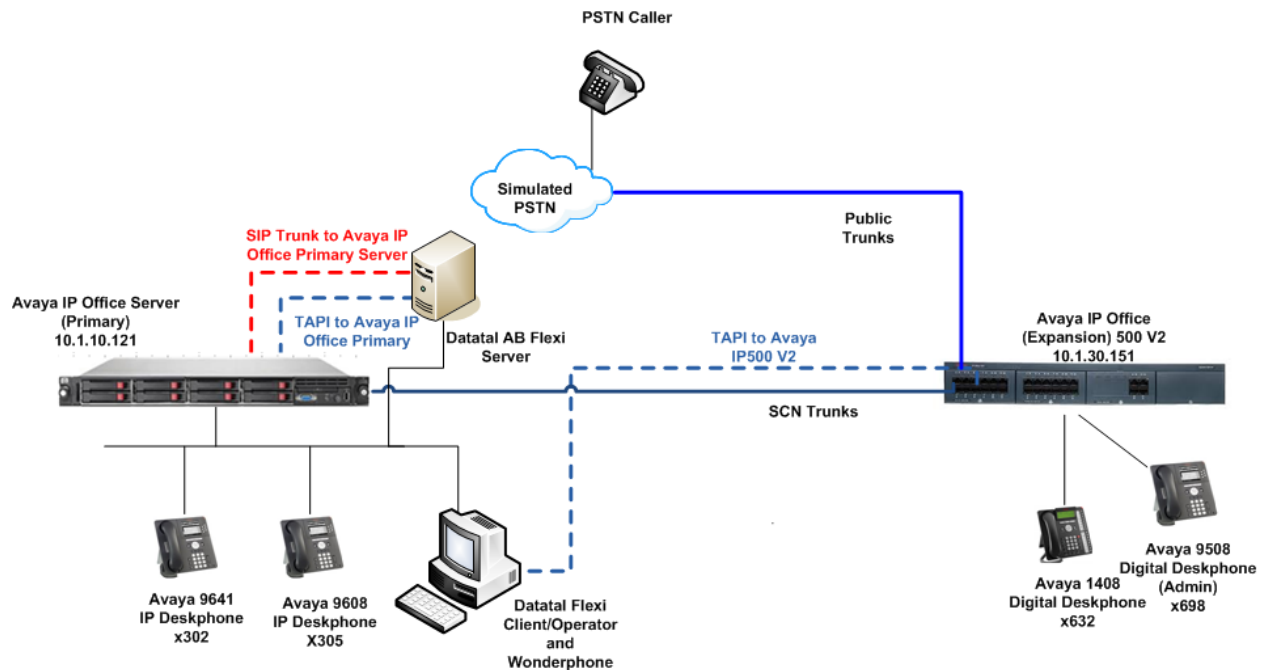


Figure 1: Test Configuration

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	R11.0.0.1.0 build 8
Avaya IP Office 500 V2	R11.0.0.1.0 build 8
Avaya IP Office Manager running on a Windows 10 PC	R11.0.0.1.0 build 8
Avaya 9641 H323 Deskphone	R6.6.6.04
Avaya 9608 H323 Deskphone	R6.6.6.04
Avaya 1408 Digital Deskphone	R48 (vintage 16)
Avaya 9508 Digital Deskphone	R60
Datatal AB Flexi platform running on Microsoft Windows Server 2016 x64	Version 5.12.8
Datatal Flexi Contact Center Agent running on Windows 10	Version 5.12.8
Datatal Flexi Operator running on Windows 10	Version 6.0.0.94
Flexi Wonderphone running on Windows 10	2.0

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 IP Office 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 AB 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**.

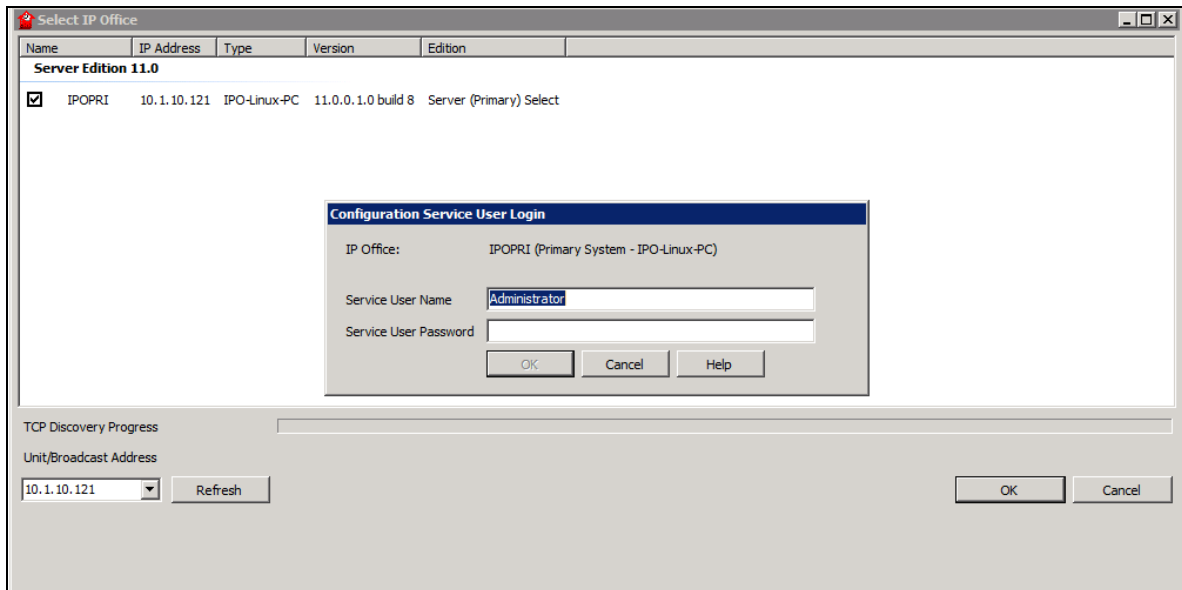
The configuration operations described in this section can be summarized as follows:

- Launch Avaya IP Office Manager (Administration)
- Display LAN properties
- Verify IP Office licenses and activate TAPI
- Create SIP Trunk
- Configure Call Route to Flexi server
- Create Short Code (Call Routing)
- Create Extension for Flexi Wonderphone
- Create User for Flexi Wonderphone
- 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). Tick on the Server Edition as shown below and enter the appropriate credentials. Click on the **OK** button.

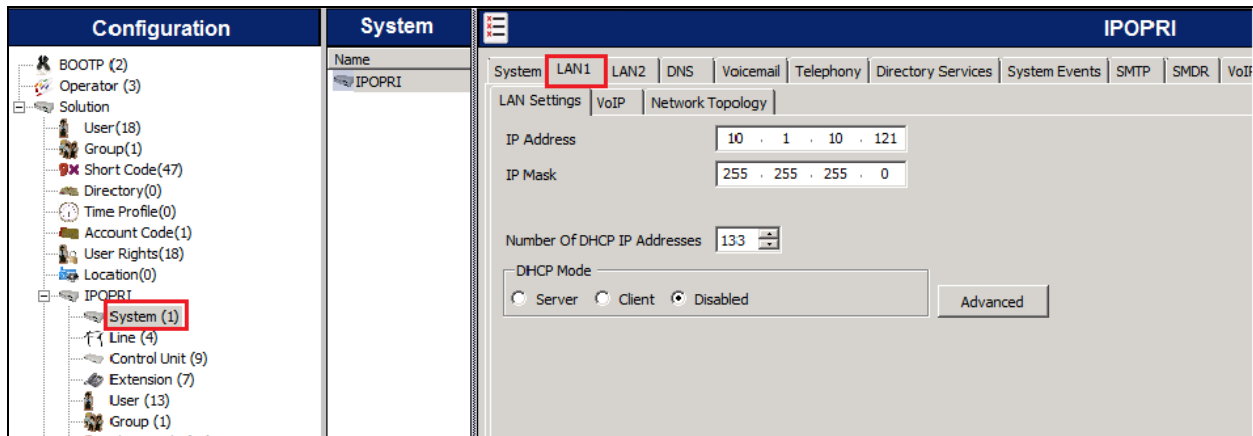


Click on **Configuration** at the top right of the page, as shown, to receive the IP Office configuration.

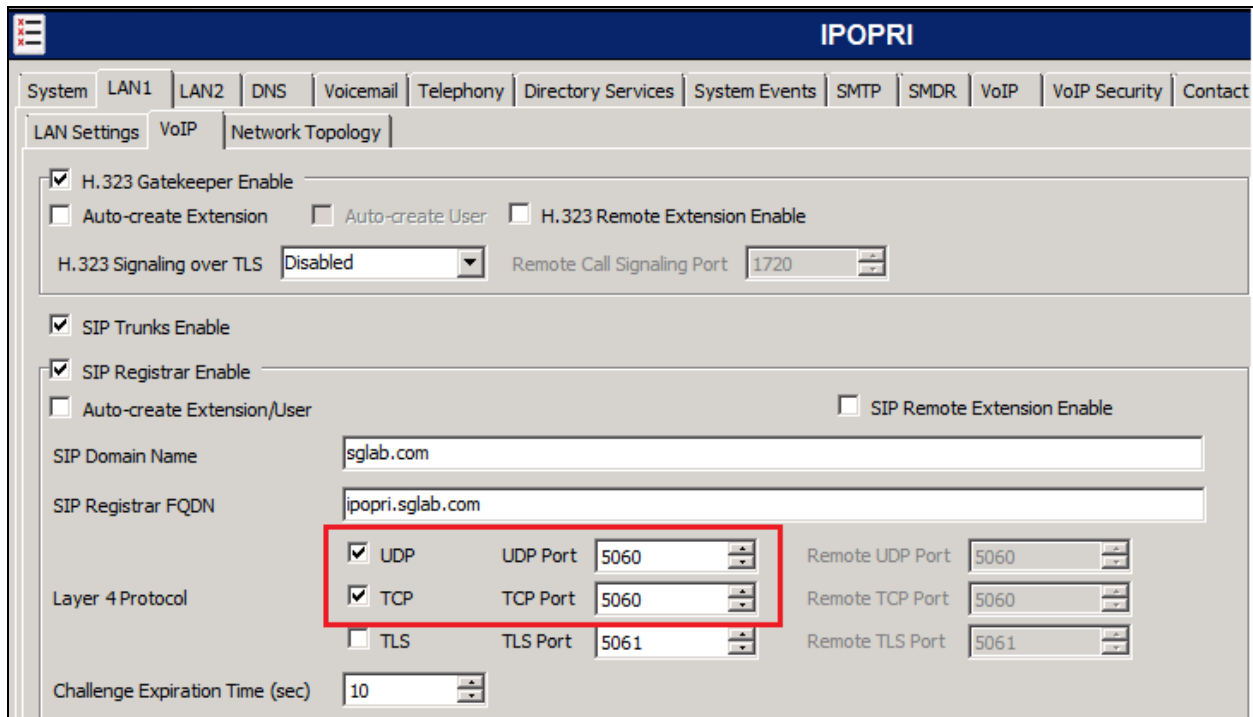


5.2. Display LAN Properties

From the left window navigate to **System (1)** 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.



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

5.3. Verify IP Office Licenses and Activate TAPI

The CTI Link Pro license is required for Datatal AB Flexi server to connect to IP Office via TAPI. Scroll down the left pane and select **License → CTI Link Pro**, to display the **CTI Link Pro** and **SIP Trunk Channels** in the bottom pane. Verify that the **License Status** is “Valid” for **CTI Link Pro** and sufficient licenses for **SIP Trunk Channels**. Ensure that the expansion server also has a valid CTI Link Pro license.

License Remote Server				
Select Licensing Valid				
Feature	Instances	Status	Expiration Date	Source
Receptionist	10	Valid	Never	PLDS Nodal
Additional Voicemail Pro Ports	252	Valid	Never	PLDS Nodal
VMPro Recordings Administrators	1	Valid	Never	PLDS Nodal
Office Worker	1000	Valid	Never	PLDS Nodal
VMPro TTS Professional	40	Valid	Never	PLDS Nodal
Power User	1000	Valid	Never	PLDS Nodal
Avaya IP endpoints	1000	Valid	Never	PLDS Nodal
SIP Trunk Channels	256	Valid	Never	PLDS Nodal
IP500 Universal PRI (Additional cha...	100	Obsolete	Never	PLDS Nodal
CTI Link Pro	1	Valid	Never	PLDS Nodal
Wave User	16	Obsolete	Never	PLDS Nodal
3rd Party IP Endpoints	1000	Valid	Never	PLDS Nodal
Server Edition	150	Valid	Never	PLDS Nodal
UMS Web Services	1000	Valid	Never	PLDS Nodal

Select **File → Advanced → Security Settings** and ensure that TAPI is enabled on both the primary and expansion server.

System: IPOPRI

System Details | Unsecured Interfaces | Certificates

System Password: [password field] [Change]

Voicemail Password: [password field] [Change] [Warning Icon]

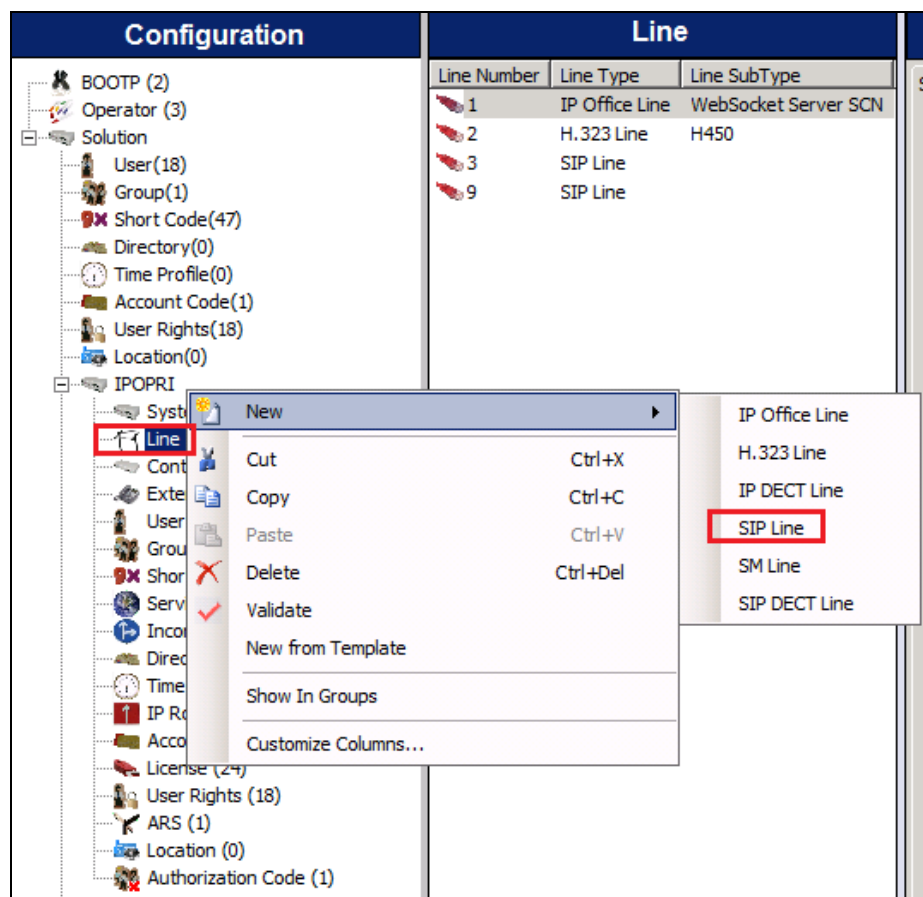
Monitor Password: [password field] [Change] ☐ Use Service User Credentials

Application Controls

TFTP Server	<input checked="" type="checkbox"/> [Warning Icon]	DevLink	<input checked="" type="checkbox"/> [Warning Icon]	TAPI/DevLink3	<input checked="" type="checkbox"/> [Warning Icon]
TFTP Directory Read	<input checked="" type="checkbox"/> [Warning Icon]			HTTP Directory Read	<input checked="" type="checkbox"/> [Warning Icon]
TFTP Voicemail	<input checked="" type="checkbox"/> [Warning Icon]			HTTP Directory Write	<input checked="" type="checkbox"/> [Warning Icon]

5.4. Create SIP Trunk

To create the SIP trunk from the IP Office to the Datatal AB 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. Here IP address of the Flexi server is being used.
- **Refresh Method** Select **Auto** from the dropdown menu.
- **Redirect and Transfer** Select **Always** for both the **Incoming** and **Outgoing** dropdown boxes.

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

SIP Line - Line 9

SIP Line | Transport | Call Details | VoIP | SIP Credentials | SIP Advanced | Engineering

Line Number: 9

ITSP Domain Name: 10.1.10.125

Local Domain Name:

URI Type: SIP URI

Location: Cloud

Prefix:

National Prefix: 0

International Prefix: 00

Country Code:

Name Priority: System Default

Description:

In Service: ☒

Check OOS: ☐

Session Timers

Refresh Method: Auto

Timer (sec): On Demand

Redirect and Transfer

Incoming Supervised REFER: Always

Outgoing Supervised REFER: Always

Send 302 Moved Temporarily: ☐

Outgoing Blind REFER: ☐

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 9' configuration window with the 'Transport' tab selected. The 'ITSP Proxy Address' field is set to '10.1.10.125'. Under the 'Network Configuration' section, the 'Layer 4 Protocol' is set to 'UDP', and both the 'Send Port' and 'Listen Port' are set to '5060'. The 'Use Network Topology Info' is set to 'None'. The 'Explicit DNS Server(s)' are set to '0 . 0 . 0 . 0'. The 'Calls Route via Registrar' checkbox is checked. The 'Separate Registrar' field is empty.

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

The screenshot shows the 'SIP Line - Line 9' configuration window with the 'SIP URI' tab selected. The table below shows the columns for SIP URIs: URI, Groups, Credential, Local URI, Contact, P Asserted ID, P Preferred ID, Diversion Header, and Remote Party ID. The 'Add...' button is highlighted in the bottom right corner.

URI	Groups	Credential	Local URI	Contact	P Asserted ID	P Preferred ID	Diversion Header	Remote Party ID
-----	--------	------------	-----------	---------	---------------	----------------	------------------	-----------------

In the subsequent window, enter the following:

- **Incoming Group** Set this to the available incoming trunk group.
- **Outgoing Group** Set this to the available outgoing trunk group.
- **Max Sessions** This will be determined by the license for number of SIP trunks available.
- **P Asserted ID** Tick

Leave the default settings for the remaining fields. Click on **OK** to save (not shown).

SIP Line - 9 | Call Details | SIP URI

New URI

Incoming Group: 20 Max Sessions: 50

Outgoing Group: 20

Credentials: 0: <None>

	Display	Content
Local URI	Auto	Auto
Contact	Auto	Auto
P Asserted ID	<input checked="" type="checkbox"/> Auto	Auto
P Preferred ID	<input type="checkbox"/> None	None
Diversion Header	<input type="checkbox"/> None	None
Remote Party ID	<input type="checkbox"/> None	None

Field meaning

Outgoing Calls	Forwarding/Twinning	Incoming Calls
Caller	Original Caller	Called
Caller	Original Caller	Called
Caller	Original Caller	Called
None	None	None
None	None	None
None	None	None

Click on the **VoIP** tab and **Codec Selection** is shown as “System Default” as in **Section 5.1**. If the system default codec is not suitable, select “Custom” from the drop down menu beside **Codec Selection** (not shown). 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. Tick the **Codec Lockdown** to enforce codec support. Click the **OK** button at bottom (not shown).

The screenshot shows the 'SIP Line - Line 9' configuration window with the 'VoIP' tab selected. The '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, and G.729(a) 8K CS-ACELP). To the right, there are checkboxes for 'Local Hold Music' (unchecked), 'Re-invite Supported' (checked), 'Codec Lockdown' (checked), 'Allow Direct Media Path' (unchecked), 'Force direct media with phones' (unchecked), and 'PRACK/100rel Supported' (unchecked). At the bottom, there are dropdown menus for 'Fax Transport Support' (set to 'None'), 'DTMF Support' (set to 'RFC2833/RFC4733'), and 'Media Security' (set to 'Disabled'). Red boxes highlight the 'VoIP' tab, the 'Re-invite Supported' and 'Codec Lockdown' checkboxes, and the 'DTMF Support' dropdown.

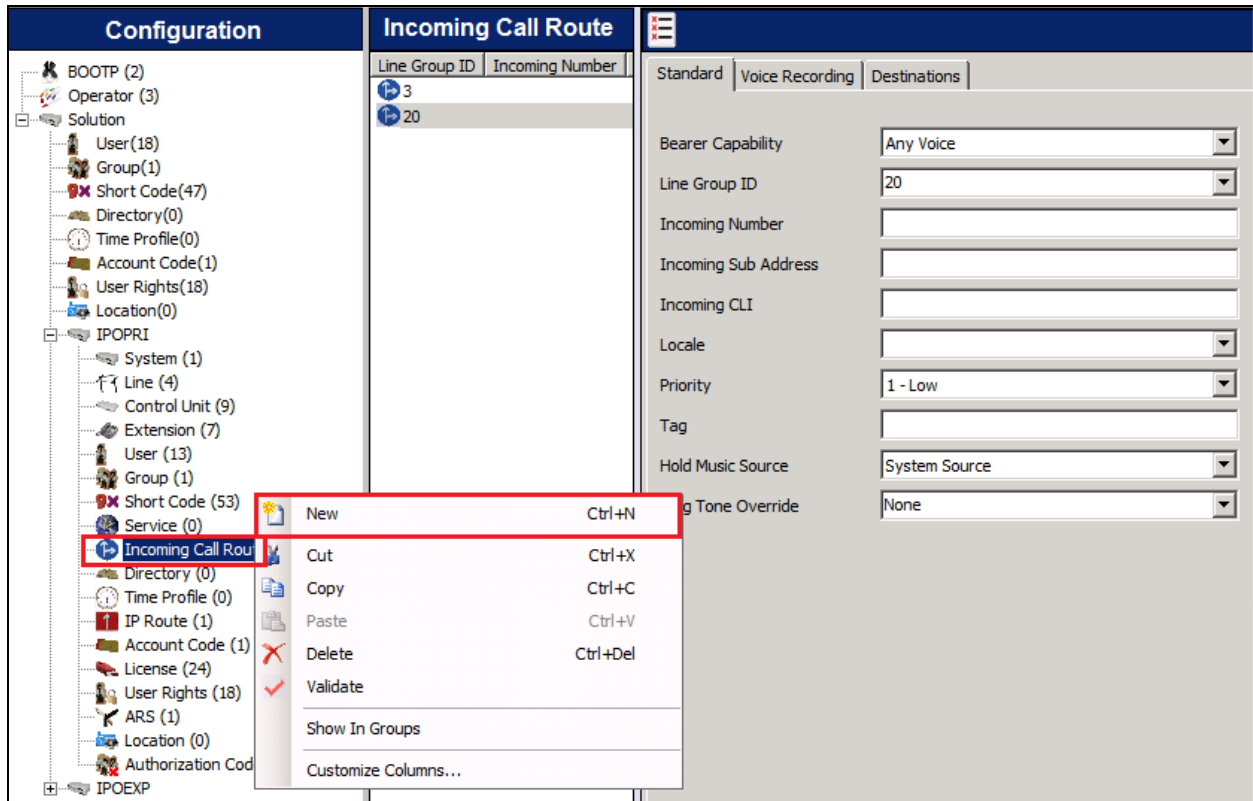
For compliance testing the values under the **SIP Advanced** tab were left as default as shown below.

The screenshot shows the 'SIP Line - Line 9' 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-agent and server headers, and 'Send Location Info' (Never). 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' (unchecked). The Call Control section includes 'Call Initiation Timeout (s)' (4), 'Call Queuing Timeout (mins)' (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' (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"/>
	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	
Send Location Info	Never	
Add UUI header	<input type="checkbox"/>	
Add UUI header to redirected calls	<input type="checkbox"/>	
Media	Allow Empty INVITE	<input type="checkbox"/>
	Send Empty re-INVITE	<input type="checkbox"/>
	Allow To Tag Change	<input type="checkbox"/>
	P-Early-Media Support	None
	Send SilenceSupp=Off	<input type="checkbox"/>
	Force Early Direct Media	<input type="checkbox"/>
	Media Connection Preservation	Disabled
	Indicate HOLD	<input type="checkbox"/>
Call Control	Call Initiation Timeout (s)	4
	Call Queuing Timeout (mins)	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	<input type="checkbox"/>
	Emulate NOTIFY for REFER	<input type="checkbox"/>
	No REFER if using Diversion	<input type="checkbox"/>

5.5. Configure Call Route to Flexi Server

The call route is configured for routing diverted calls to Flexi Server. To configure the Incoming Call Route, navigate to Primary Server **IPOPRI** 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 '20' with three tabs: 'Standard', 'Voice Recording', and 'Destinations'. The 'Standard' tab is active. It contains the following fields:

Bearer Capability	Any Voice
Line Group ID	20
Incoming Number	
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None

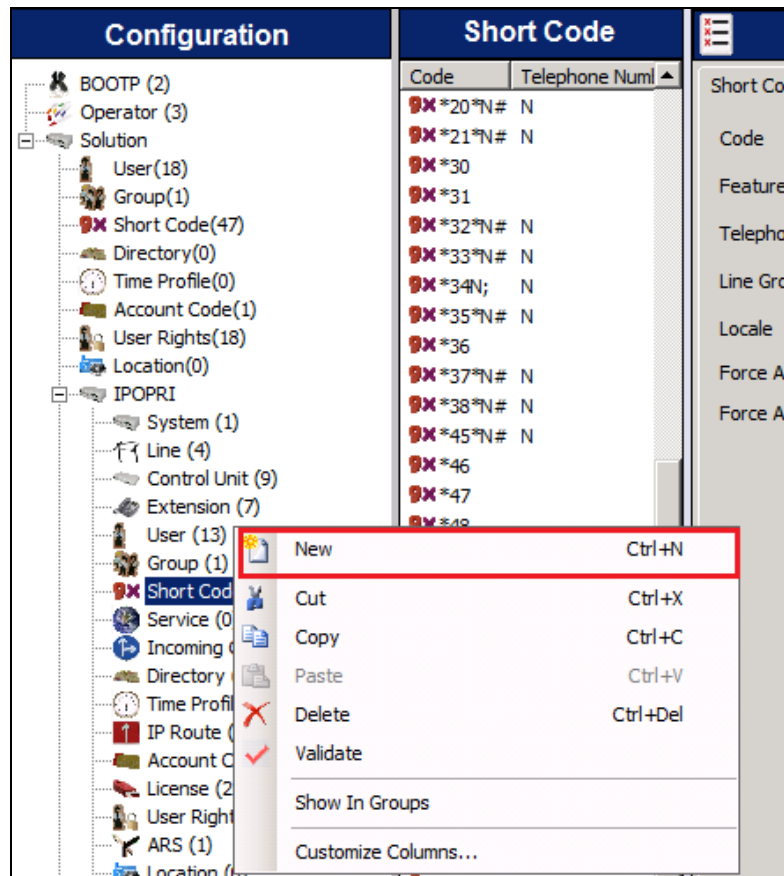
In the **Destinations** tab, enter a . (full stop/period) in the **Destination** field. Click on the **OK** button (not shown).

The screenshot shows the 'Destinations' tab of the configuration window titled '20'. It contains a table with the following data:

TimeProfile	Destination	Fallback Extension
Default Value	.	

5.6. Create Short Code (Call Routing)

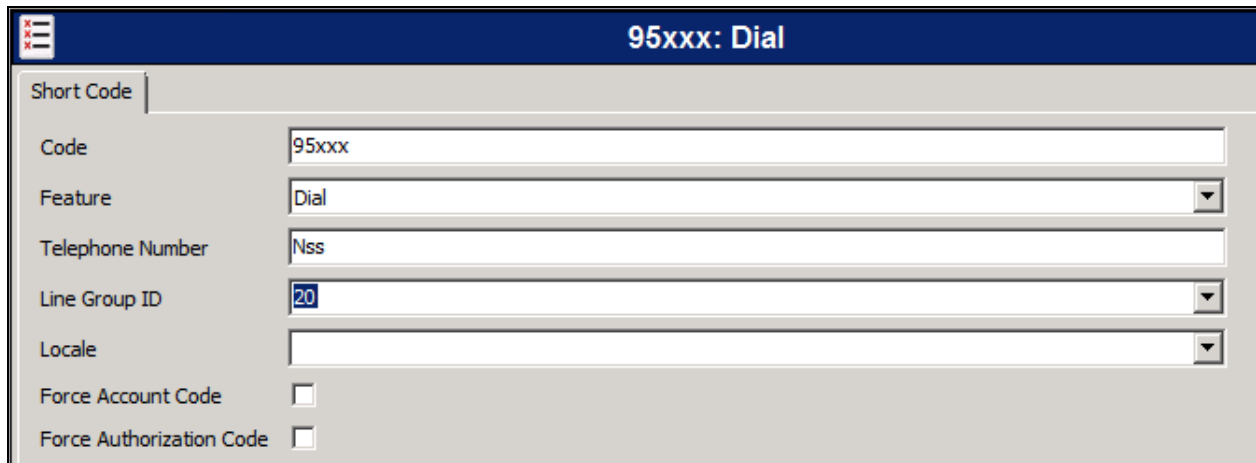
A Short Code needs to be configured on the IP Office to route calls to Flexi server. Call to users who are busy or not answered are directed to Flexi server using this method. Right click on **Short Code** and select **New**.



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 **95xxx** 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 5.3**.

Click the **OK** button at bottom (not shown).



95xxx: Dial

Short Code

Code: 95xxx

Feature: Dial

Telephone Number: Nss

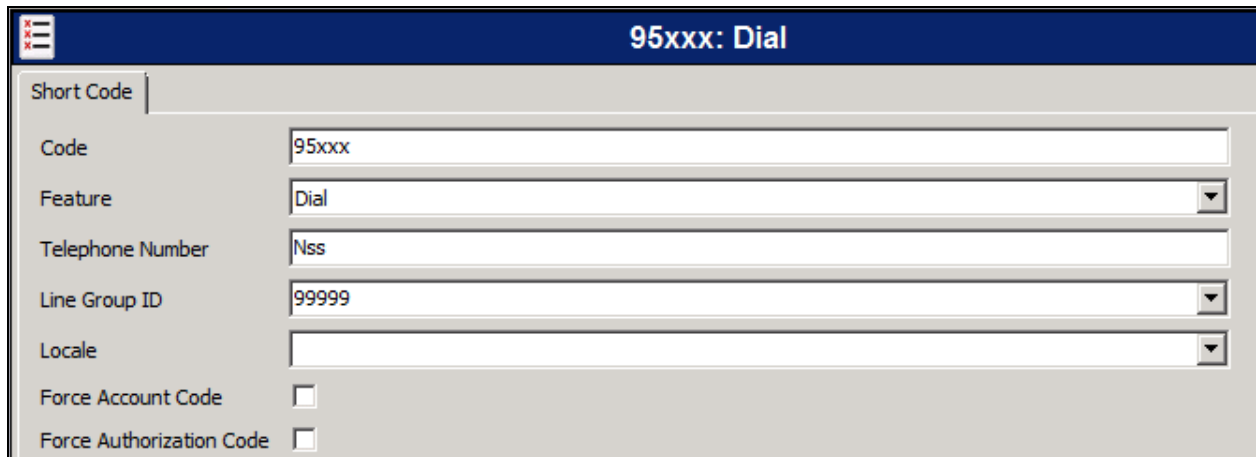
Line Group ID: 20

Locale:

Force Account Code: ☐

Force Authorization Code: ☐

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 AB 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. 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.



95xxx: Dial

Short Code

Code: 95xxx

Feature: Dial

Telephone Number: Nss

Line Group ID: 99999

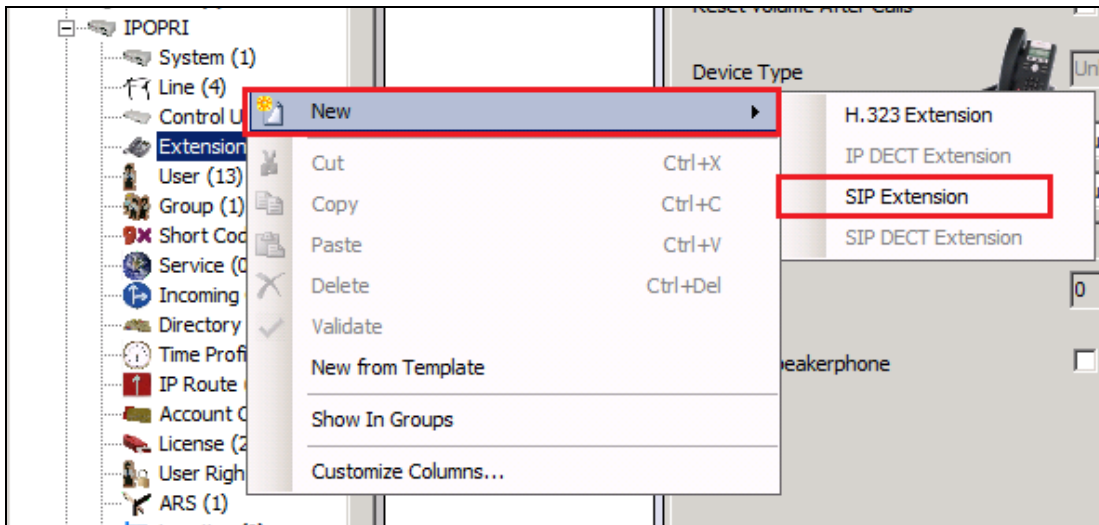
Locale:

Force Account Code: ☐

Force Authorization Code: ☐

5.7. Create Extension for Flexi Wonderphone

From the left window, right click on **Extension** and select **New** → **SIP Extension**.



Enter the **Base Extension** and **Phone Password** and confirm password.

A screenshot of the 'SIP Extension: 11211 309' configuration form. The form has a blue header bar with the title 'SIP Extension: 11211 309'. Below the header, there are two tabs: 'Extension' and 'VoIP'. The 'Extension' tab is selected. The form contains the following fields and options:

- Extension ID: 11211
- Base Extension: 309
- Phone Password: masked with dots
- Confirm Phone Password: masked with dots
- Caller Display Type: On (dropdown menu)
- Reset Volume After Calls: unchecked checkbox
- Device Type: Unknown SIP device (text field with a phone icon)
- Location: Automatic (dropdown menu)
- Fallback As Remote Worker: Auto (dropdown menu)
- Module: 0 (text field)
- Port: 0 (text field)
- Disable Speakerphone: unchecked checkbox

Click on the **VoIP** tab, **3rd Party Auto Answer** must be set to **answer-after** 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 set as below.

SIP Extension: 11211 309

Extension VoIP

IP Address: 0 . 0 . 0 . 0

Codec Selection: System Default

Unused: [Empty Box]

Selected: G.711 ALAW 64K, G.711 ULAW 64K, G.729(a) 8K CS-ACELP

Reserve License: Reserve 3rd party IP endpoint license

Fax Transport Support: None

DTMF Support: RFC2833/RFC4733

3rd Party Auto Answer: answer-after

Media Security: Same as System (Disabled)

☐ Requires DTMF

☐ Local Hold Music

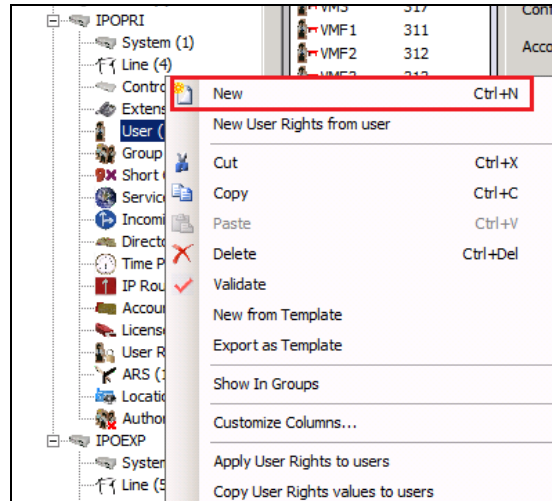
☒ Re-invite Supported

☐ Codec Lockdown

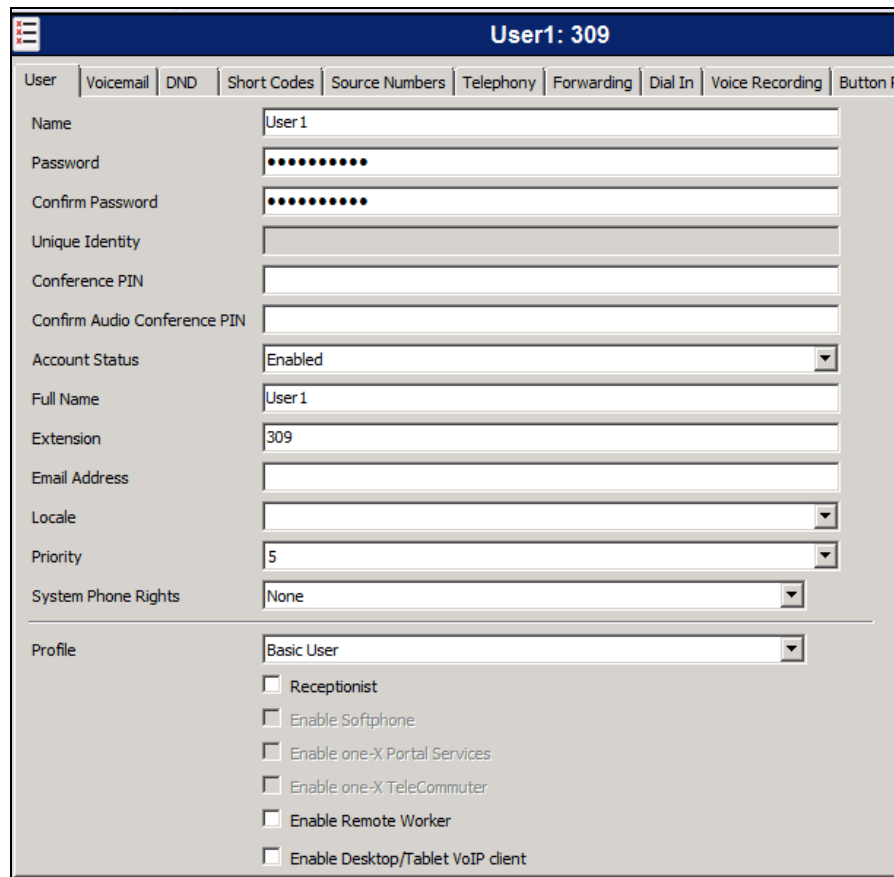
☒ Allow Direct Media Path

5.8. Create a New User for Flexi Wonderphone

New user is created for the Flexi Wonderphone. From the left window, right click on **User** and select **New**.



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



User1: 309	
Name	User1
Password
Confirm Password
Unique Identity	
Conference PIN	
Confirm Audio Conference PIN	
Account Status	Enabled
Full Name	User1
Extension	309
Email Address	
Locale	
Priority	5
System Phone Rights	None
Profile	Basic User
<input type="checkbox"/> Receptionist	
<input type="checkbox"/> Enable Softphone	
<input type="checkbox"/> Enable one-X Portal Services	
<input type="checkbox"/> Enable one-X TeleCommuter	
<input type="checkbox"/> Enable Remote Worker	
<input type="checkbox"/> Enable Desktop/Tablet VoIP client	

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

The screenshot shows the Avaya User Configuration interface for User1: 309*. The 'Telephony' tab is selected, and the 'Call Settings' sub-tab is active. The 'Call Waiting On' checkbox is checked and highlighted with a red box.

Setting	Value	Option
Outside Call Sequence	Default Ring	<input checked="" type="checkbox"/> Call Waiting On
Inside Call Sequence	Default Ring	<input checked="" type="checkbox"/> Answer Call Waiting On Hold
Ringback Sequence	Default Ring	<input type="checkbox"/> Busy On Held
No Answer Time (sec)	15	<input type="checkbox"/> Off-hook Station
Wrap-Up Time (sec)	2	
Transfer Return Time (sec)	Off	
Call Cost Mark-Up	100	
Advertise Callee State To Internal Callers	System Default (Off)	

5.9. Configure Forwarding

Users who use Flexi Presentity needs to configure forwarding in order for call to be diverted. **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 prefix (as configured in **Section 5.6**) followed by the extension used by this user (example **302**).

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.

Repeat the same setting for users in the expansion server that requires Flexi Presentity.

The screenshot displays the IP Office configuration interface. On the left, the 'Configuration' tree shows 'User (13)' selected. The 'User' list in the center shows 'User2' with extension 302. The right pane shows the 'Forwarding' tab for 'User2: 302'. The 'Forwarding' tab is highlighted with a red box. The 'Forward On Busy' and 'Forward On No Answer' checkboxes are checked, and the 'Forward Number' is set to 95302. The 'Forwarding Number' field is also highlighted with a red box.

Configuration	User	User2: 302
BOOTP (3)	Name	User
Operator (3)	Extensio	Voice
Solution		DND
User (18)	Agent2 305	Short Codes
Group(1)	NoUser	Source Numbers
Short Code(47)	Room 1 - 1 301	Telephony
Directory(0)	Test1 304	Forwarding
Time Profile(0)	Test2 388	Dial In
Account Code(1)	User1 309	
User Rights(18)	User2 302	
Location(0)	VM1 315	
IPOPRI	VM2 316	
System (1)	VM3 317	
Line (4)	VMF1 311	
Control Unit (9)	VMF2 312	
Extension (7)	VMF3 313	
User (13)		
Group (1)		
Short Code (6)		
Service (0)		
Incoming Call Route (2)		
IP Route (1)		
License (24)		
ARS (1)		
Location (0)		

5.10. 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 follow the prompt 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 connections. The Avaya IP Office TAPI driver is installed and configured on the Flexi servers. The SIP Trunk is configured using a web GUI by opening a browser session to the Flexi server.

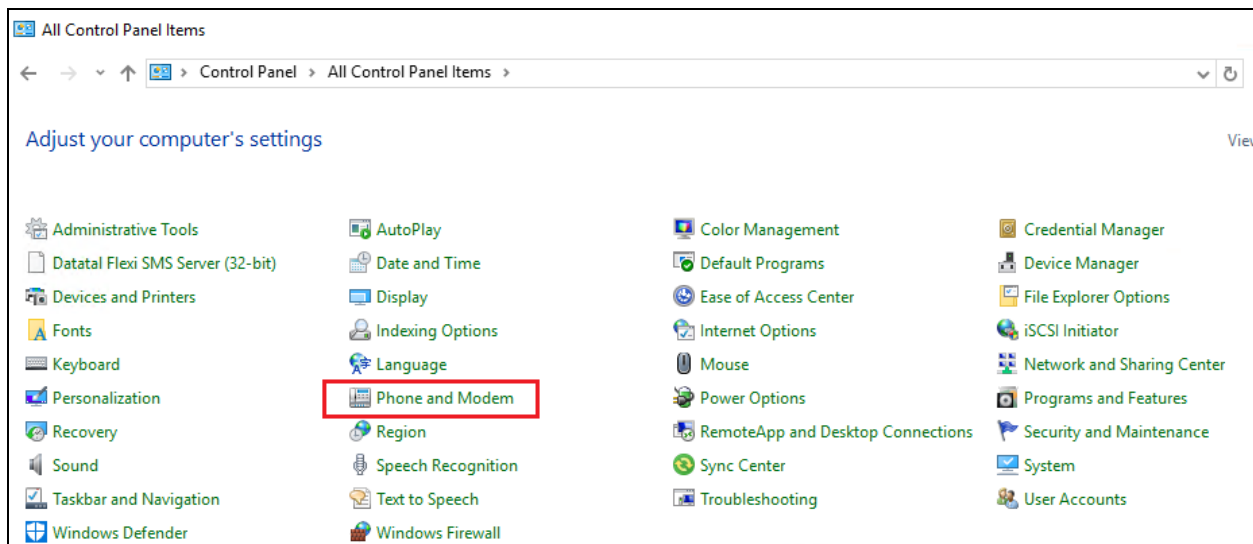
6.1. Configure Avaya IP Office TAPI

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 IP Office Server Edition.

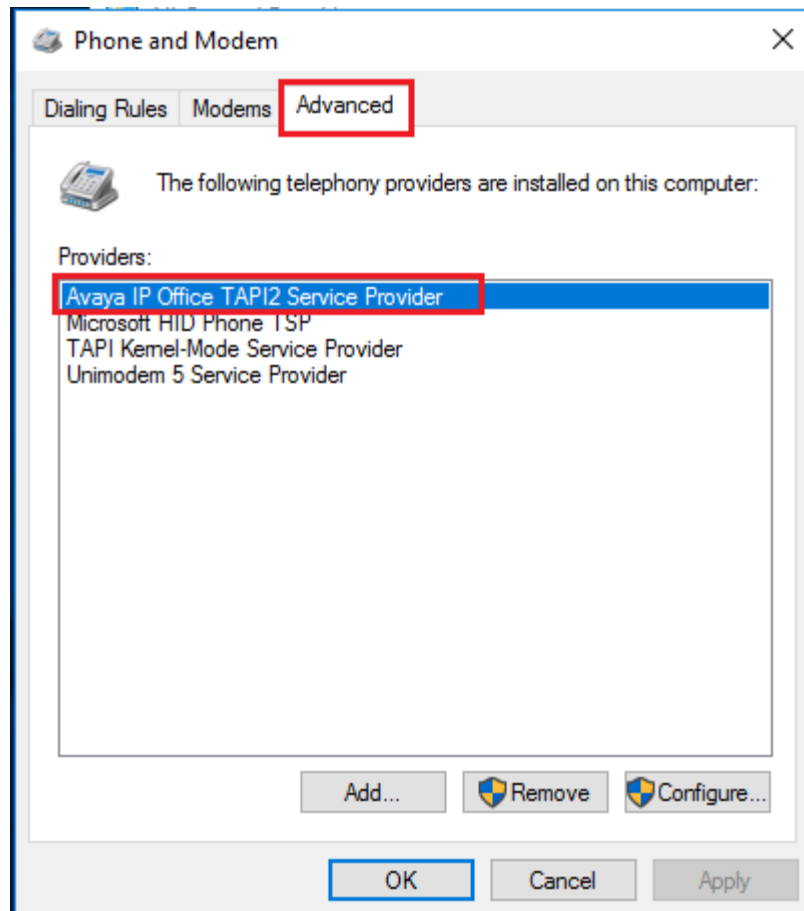
Note: It is important that the TAPI software installation was run as administrator to ensure that the application receives the correct rights to run.

From the Windows 2016 Server, right click **Start** → **Control Panel** → **Phone and Modem**.



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

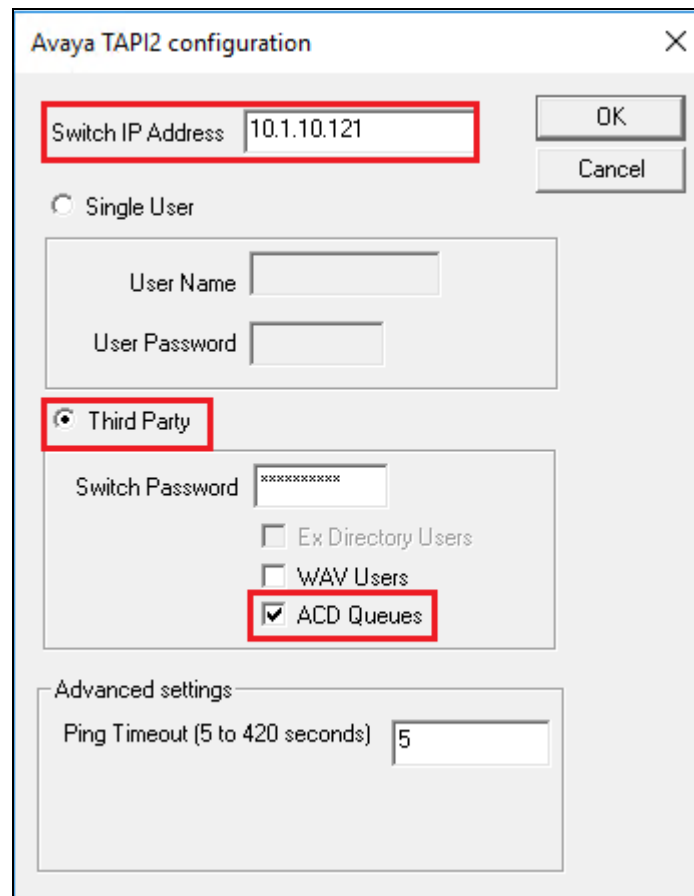
Note: Enter any appropriate dealing rules in the **Dialing Rules** tab as required (not shown).



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

- **Switch IP Address** Enter the IP address of the pertinent 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 image shows the 'Avaya TAPI2 configuration' dialog box. It has a title bar with a close button (X). The main area contains several fields and options. At the top, there is a 'Switch IP Address' field with the value '10.1.10.121'. Below this are two radio buttons: 'Single User' and 'Third Party'. The 'Third Party' radio button is selected. Below the radio buttons are two groups of fields. The first group contains 'User Name' and 'User Password' fields. The second group contains 'Switch Password' (masked with 'x'), 'Ex Directory Users' (unchecked), 'WAV Users' (unchecked), and 'ACD Queues' (checked). At the bottom, there is an 'Advanced settings' section with a 'Ping Timeout (5 to 420 seconds)' field set to '5'. 'OK' and 'Cancel' buttons are located in the top right corner.

Avaya TAPI2 configuration

Switch IP Address 10.1.10.121

OK

Cancel

☐ Single User

User Name

User Password

☒ Third Party

Switch Password

☐ Ex Directory Users

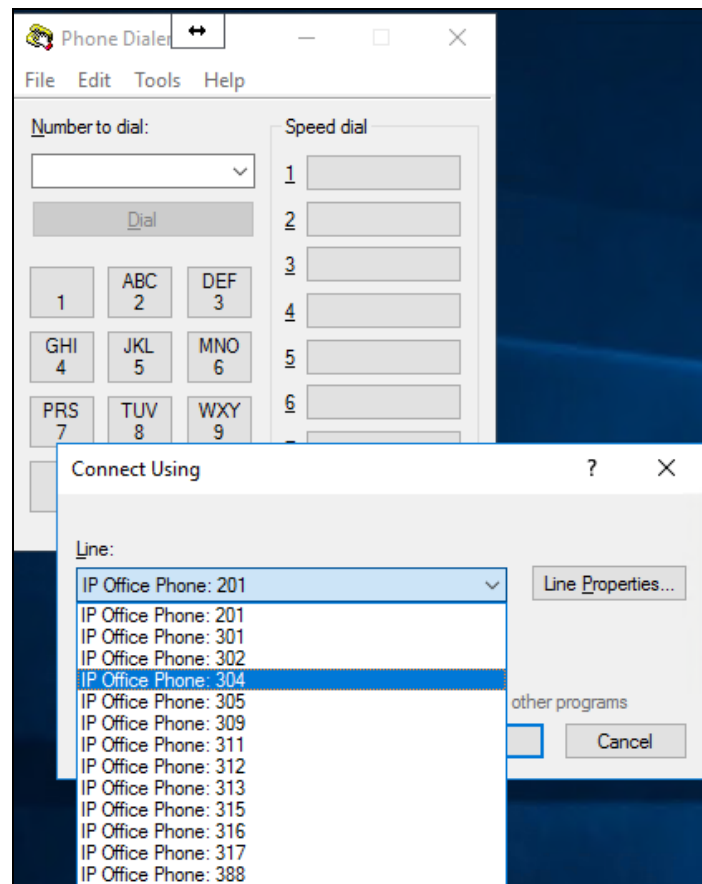
☐ WAV Users

☒ ACD Queues

Advanced settings

Ping Timeout (5 to 420 seconds) 5

Once TAPI is configured, restart the server. After reboot, test the TAPI connection by **Start → Run**. Enter **dialer** under line, you should now see your extension in IP Office as shown below. If not, check system password in IP Office.



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

6.2. Configure SIP Trunk

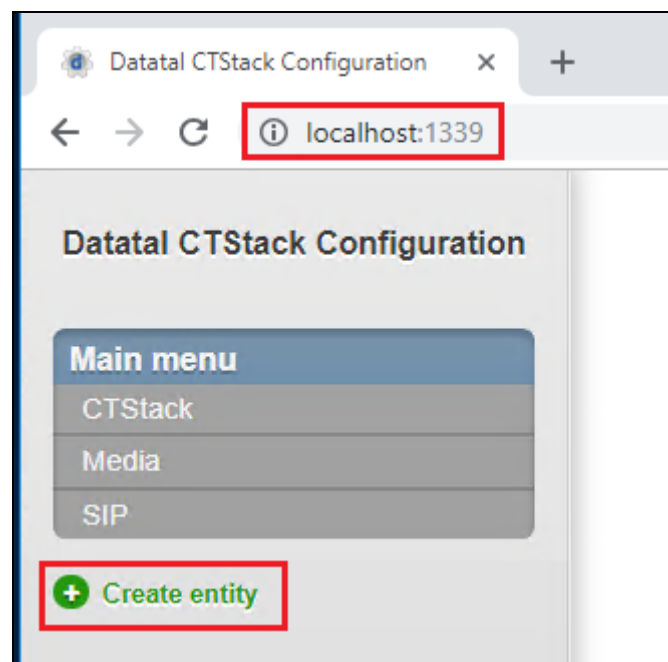
Configuration of the Flexi server is achieved using a web interface. After logging on to the Flexi server, browse to **localhost:1339** using Internet Explorer 10 or higher, Mozilla Firefox or Google Chrome web browsers. The following configuration steps were carried out during compliance testing:

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

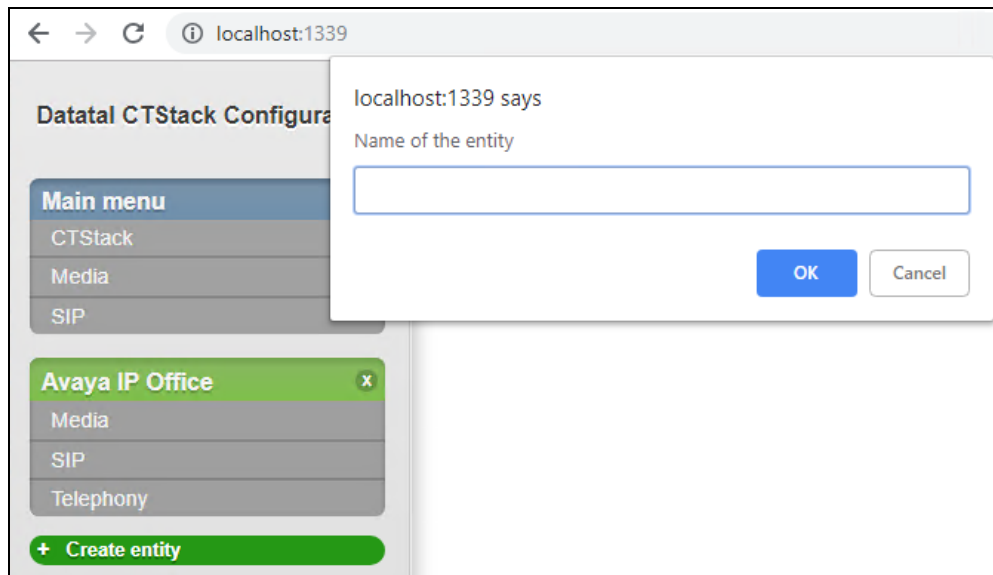
Note: It is implied that the Flexi server is pre-configured including any Licence requirements. Configuration of Flexi Presentity, Flexi CC agents and Flexi Tid agents is outside the scope of these Application Notes.

6.2.1. Configure Entity for Avaya IP Office

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

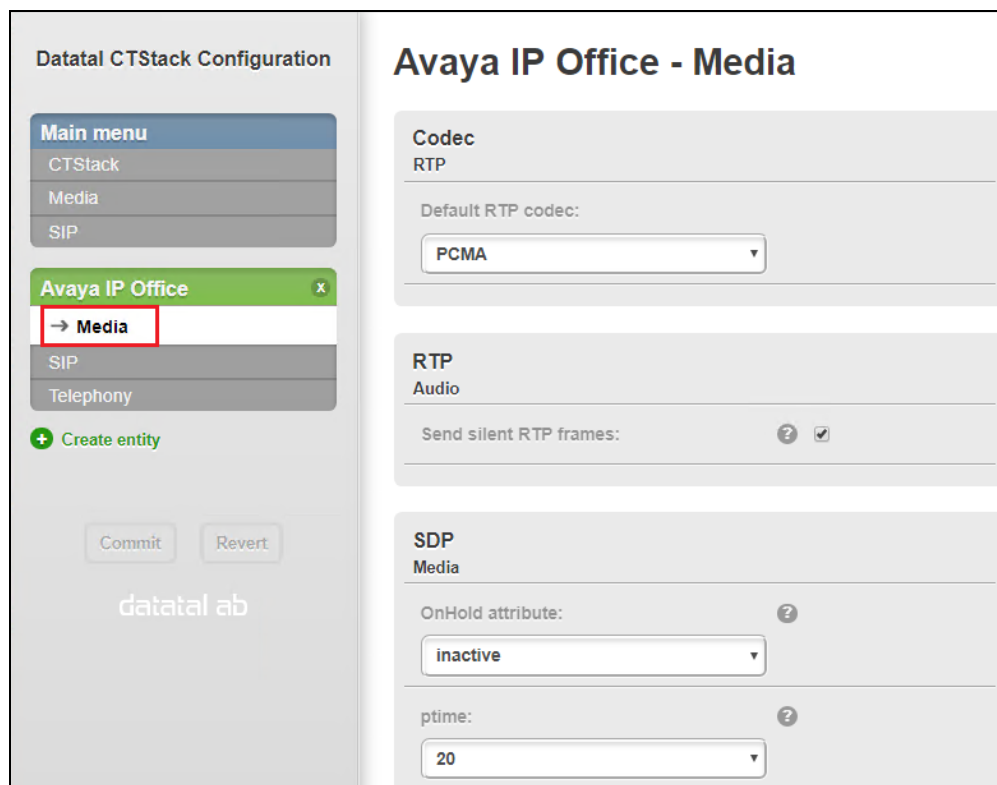


Once the new frame opens enter an informative name in the **Name of the entity** box, **Avaya IP Office** was used during compliance testing. Click the **OK** button to save.



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**. Retain all default values shown below.

The screenshot displays the 'Datatal CTStack Configuration' interface. On the left sidebar, under 'Main menu', there are links for 'CTStack', 'Media', and 'SIP'. The 'SIP' link is highlighted with a red box. Below this, there is a section for 'Avaya IP Office' with a sub-menu containing 'Media', '→ SIP' (highlighted with a red box), and 'Telephony'. A green '+ Create entity' button is located below the sub-menu. At the bottom of the sidebar are 'Commit' and 'Revert' buttons, and the 'datatal ab' logo.

The main content area is titled 'Avaya IP Office - SIP'. It contains several configuration sections:

- Dialogs**
 - Always create early dialogs: ☐
 - Retry-After 4xx:
 - Use OPTIONS for keep-alive: ☐
- Inbound**
 - Request-URI user as ANI: ☐
 - Swap CmpNum on deflection: ☒
 - Use Flexi TID ListenExtension: ☐
- Outbound**
 - 'Privacy' header value:
 - Set 'Diversion' header on MakeCall: ☒
 - Set 'History-Info' header on MakeCall: ☐
 - Use 'P-Asserted-Identity': ☒

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 shows the 'Datatal CTStack Configuration' interface. On the left is a 'Main menu' with options: CTStack, Media, SIP, and Avaya IP Office (selected). Below the menu is a '+ Create entity' button. The right pane is titled 'Transfer' and contains the following settings:

Setting	Help (?)	Value
Hangup leg A on supervised 180/183:	?	<input checked="" type="checkbox"/>
Hangup leg A on supervised 200:	?	<input checked="" type="checkbox"/>
Park other calls on MakeCall:	?	<input type="checkbox"/>
Play 'ring' at other calls on MakeCall:	?	<input checked="" type="checkbox"/>
Terminate local call transfer on INVITE:	?	<input type="checkbox"/>
Treat BYE as transfer success:	?	<input type="checkbox"/>
Use 'Remote-Target' in 'Refer-To':	?	<input checked="" type="checkbox"/>
Wait for park complete on MakeCall:	?	<input checked="" type="checkbox"/>

A red rectangular box highlights the 'Park other calls on MakeCall' and 'Play 'ring' at other calls on MakeCall' rows.

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 **TCP** 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).

The screenshot displays the 'Datatal CTStack Configuration' web interface. On the left sidebar, the 'Main menu' includes 'CTStack', 'Media', and 'SIP'. Below it, the 'Avaya IP Office' section shows 'Media', 'SIP' (highlighted with a right arrow), and 'Telephony'. A '+ Create entity' link and a '1 change(s) pending' status are also visible, along with 'Commit' and 'Revert' buttons. The 'Commit' button is highlighted with a red rectangle. The main content area is divided into sections: 'Registrations' (with an empty list and 'ADD', 'EDIT', 'REMOVE' buttons), 'SIP Dialogs' (with a 'Use 'From' header:' checkbox checked and highlighted with a red rectangle), 'RFC 3325' (with a 'P-*-Identity mode:' dropdown set to 'Both'), and 'Transport' (with a 'Transport:' dropdown set to 'TCP' and highlighted with a red rectangle). The 'datatal ab' logo is at the bottom left of the sidebar.

6.2.4. Configure Telephony

To configure Telephony, click on **Telephony** for the IP Office configured in **Section 5**.
Configure the following:

- **Lines**
The lines limit the number of calls routed from IP Office to Flexi server via the SIP Trunk. Enter the number of SIP lines that Flexi is licensed for. This must tally with the license purchased.
- **Address**
Enter the Flexi queue number (100 was used during compliance testing) and calls were routed via 95xxx short code.
- **Default Domain**
Enter the telephony domain. This was left blank.
- **Default SIP URI host**
Enter the IP address of the IP Office as per **Section 5.1**.
- **Default SIP URI port**
Enter the UDP port number configured in **Section 5.1**.
- **Name**
Enter an informative name for the Flexi Server (e.g. **Flexi**).
- **Apply**
Select **Avaya IPO (Trunk)** from the dropdown box.
- **Trunk Mode**
Check the check box.

Datatall CTStack Configuration

Main menu

- CTStack
- Media
- SIP

Avaya IP Office x

- Media
- SIP
- Telephony

+ Create entity

1 change(s) pending

Commit **Revert**

datatall ab

Avaya IP Office - Telephony

Line configuration

Standard

BlindCall source mode: ? All

Description: ? string

INVITE expires: ? 27

Lines: ? 20

Use Flexi database: ? ☒

SIP

Address

Address: ? 100

Default domain: ? string

Default SIP URI host: ? 10.1.10.121

Default SIP URI port: ? 5060

Name: ? Flexi

Profile

Apply: ? None

Current: ? Avaya IPO (trunk)

Trunk

Trunk mode: ? ☒

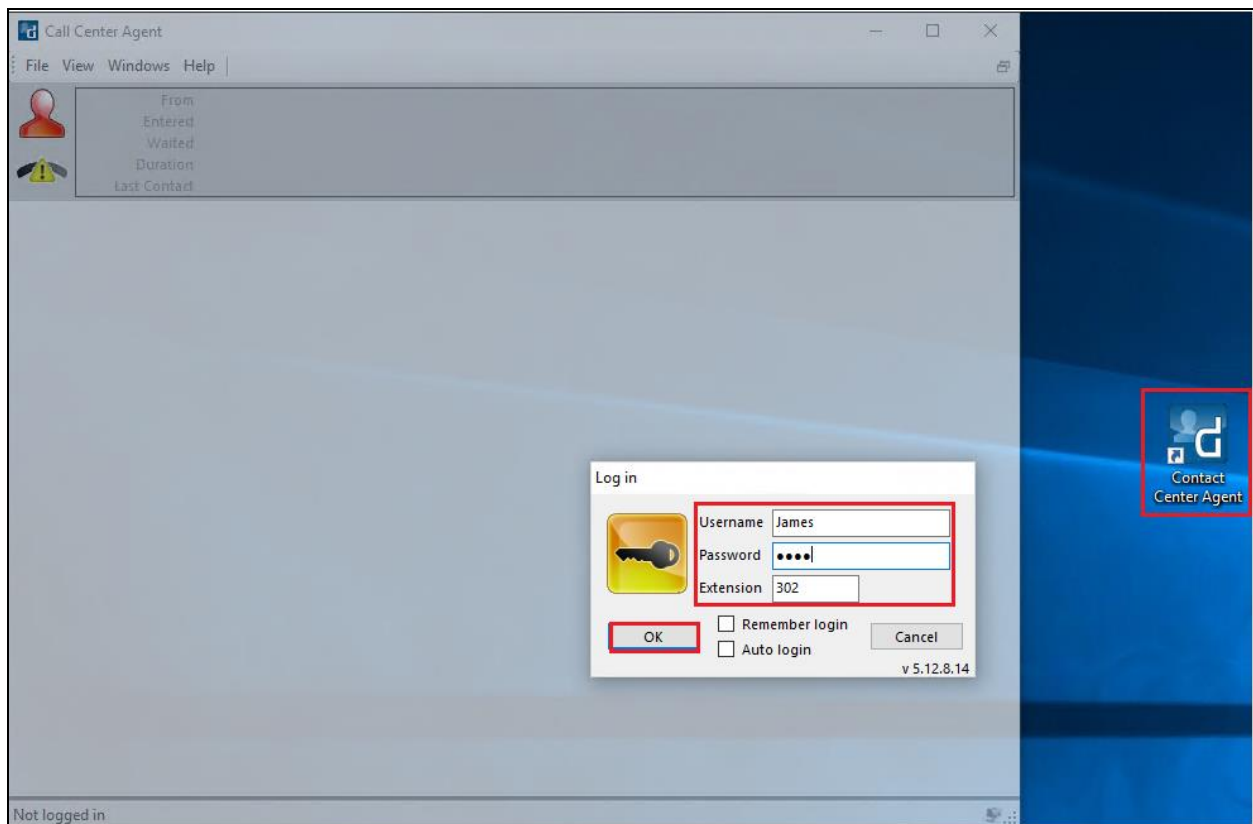
Defaults were used for the remaining fields. Click on the **Commit** button. When the **Commit** dialog window opens click on **Commit changes now** button (not shown).

7. Verification Steps

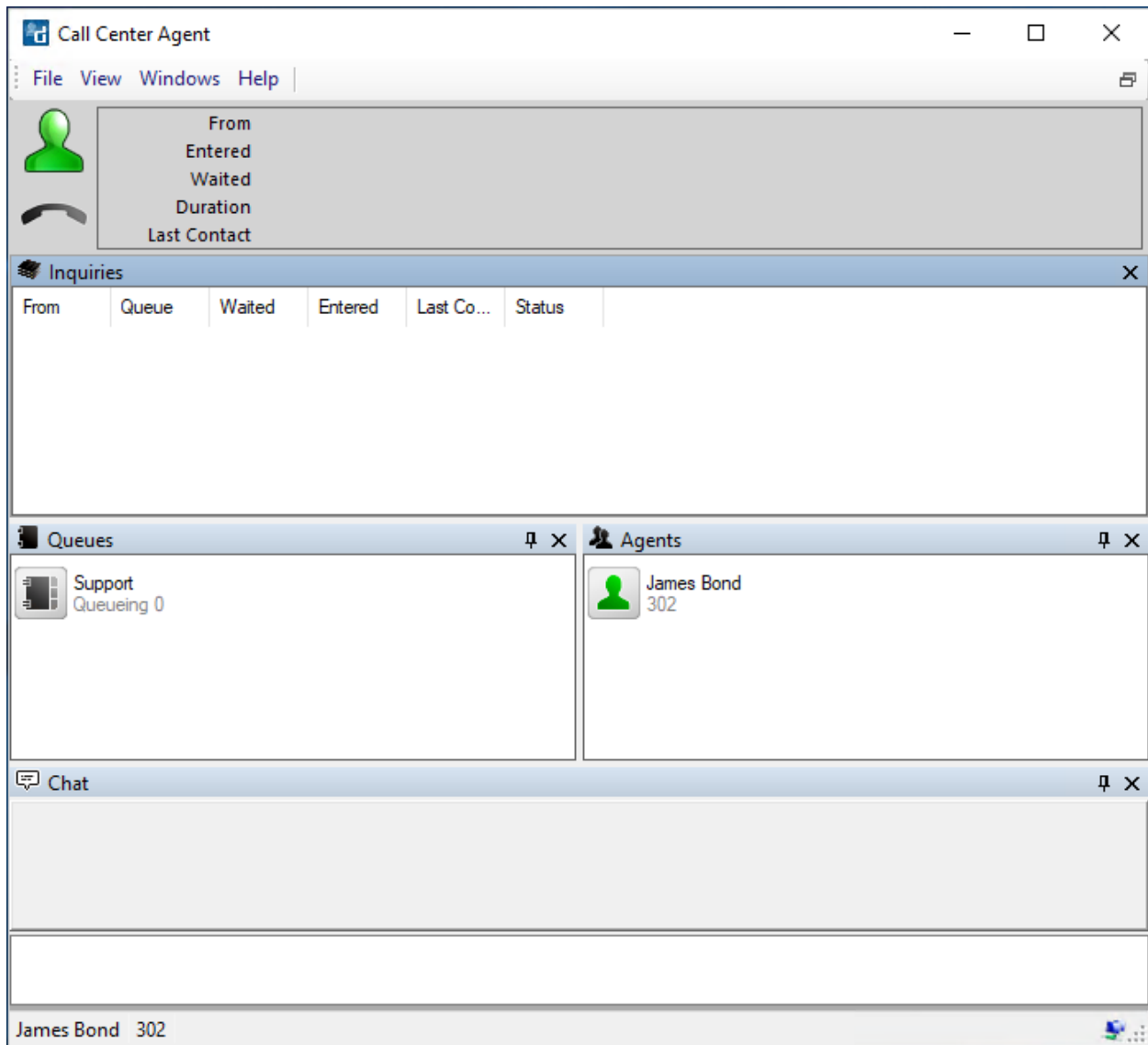
This section provides the tests that can be performed to verify correct configuration of the Avaya IP Office and Datatal AB Flexi. Repeat similar steps for verifying the expansion agents of which screenshots will not be shown.

7.1. Verify Flexi CC

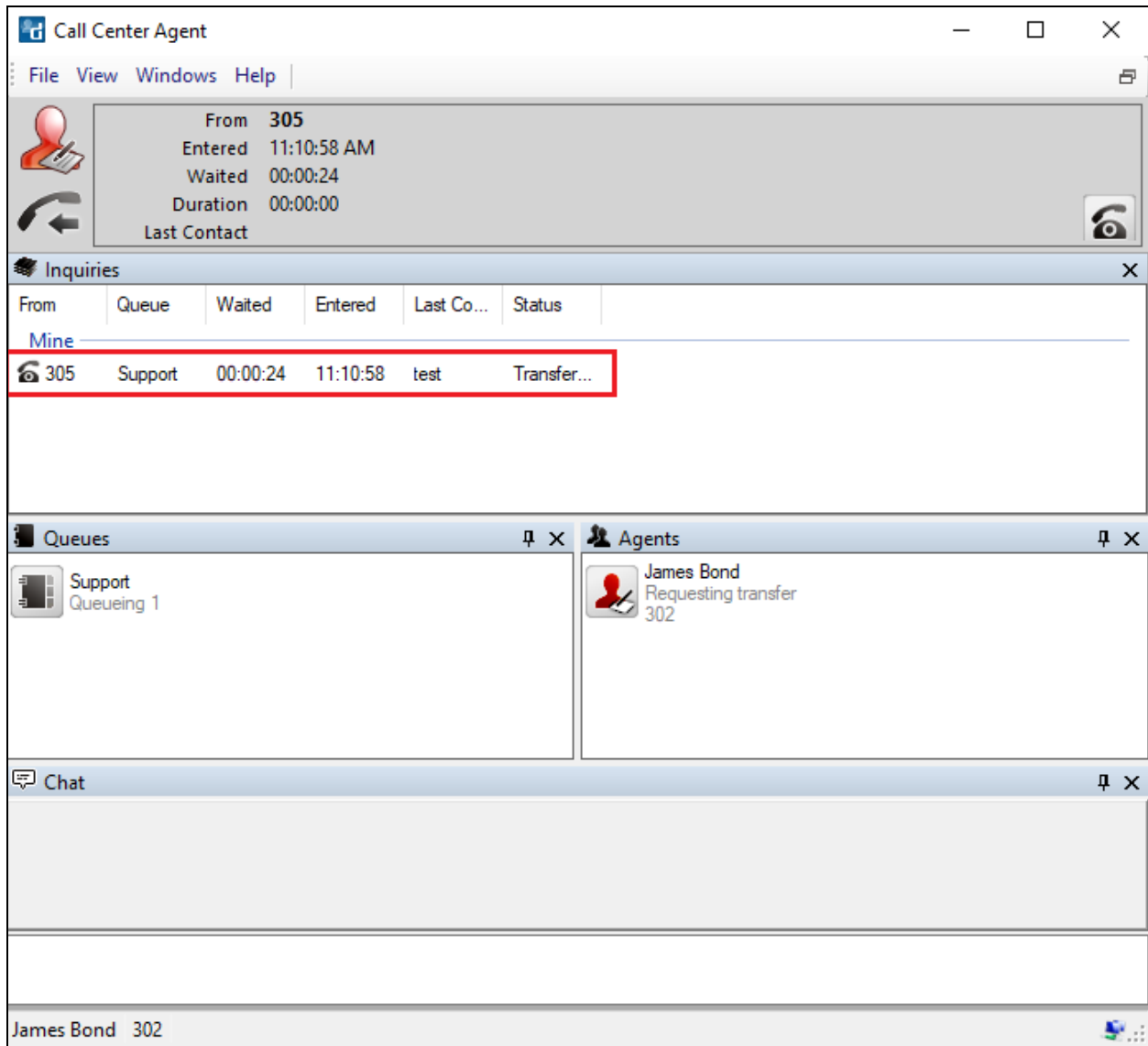
Using the shortcut on the desktop open **Contact Center Agent**, enter the appropriate credentials and click on **OK**. Below is an example of an agent user with extension 302.



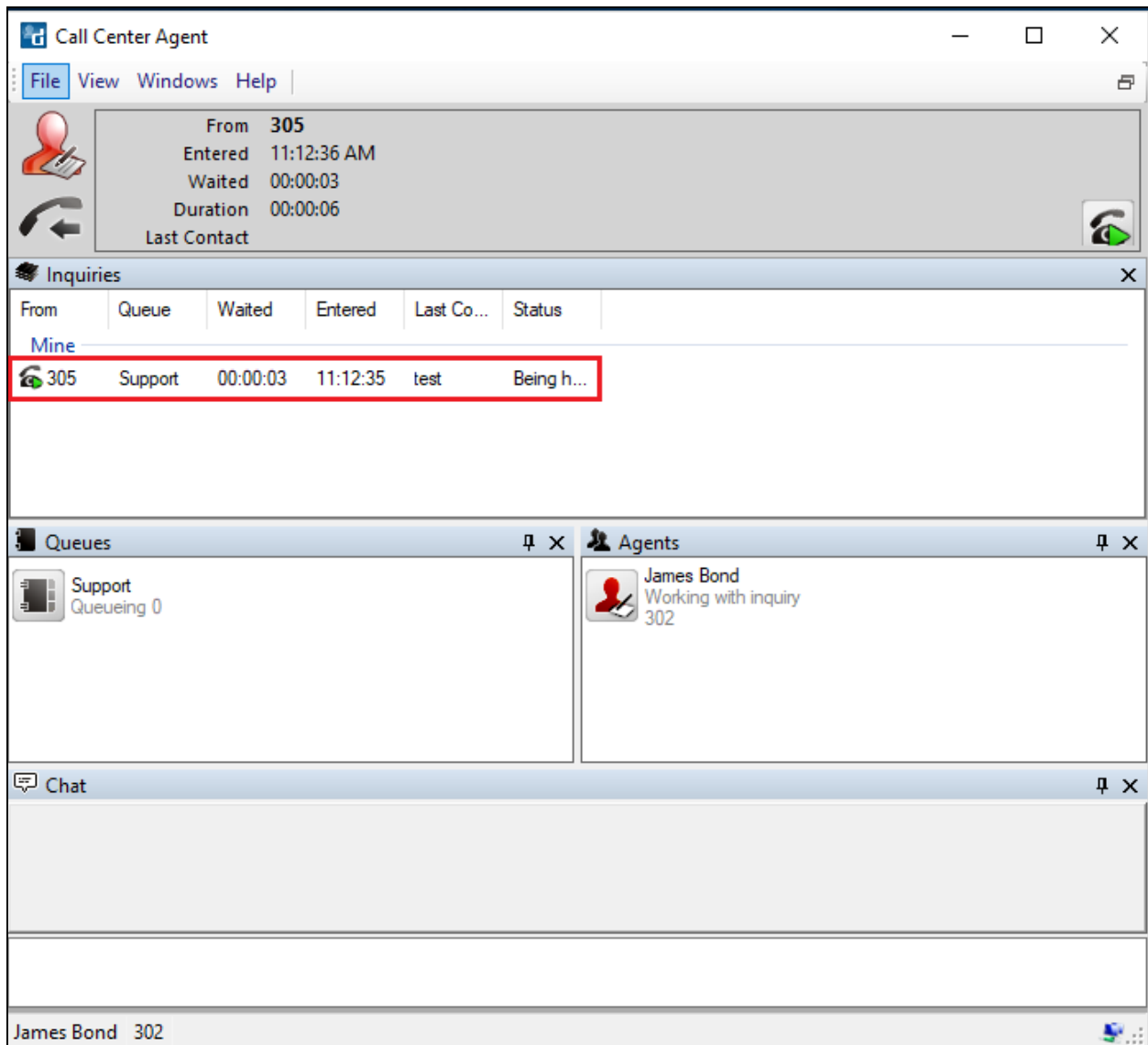
Once logged in the following screen shows the status of the agent and the queue associated with the agent.



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

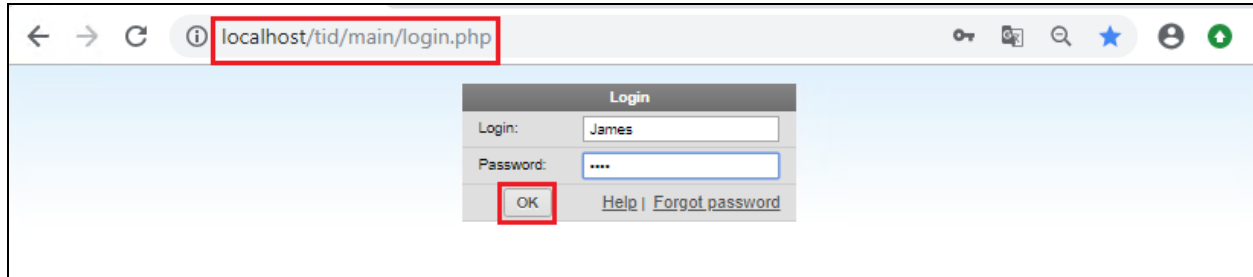


Once the call is answered this is reflected on the desktop as shown below.

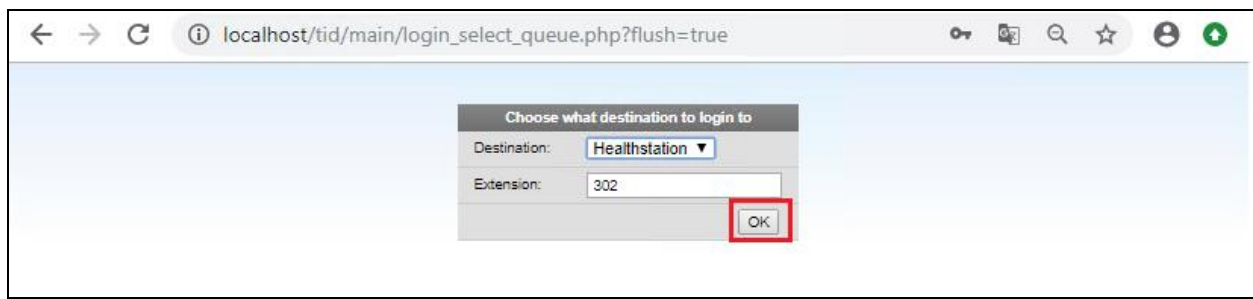


7.2. Verify Flexi Tid

Open a web session to Flexi Tid. Enter the appropriate credentials of an agent logging into the Tid queue and click on **OK** to log in.



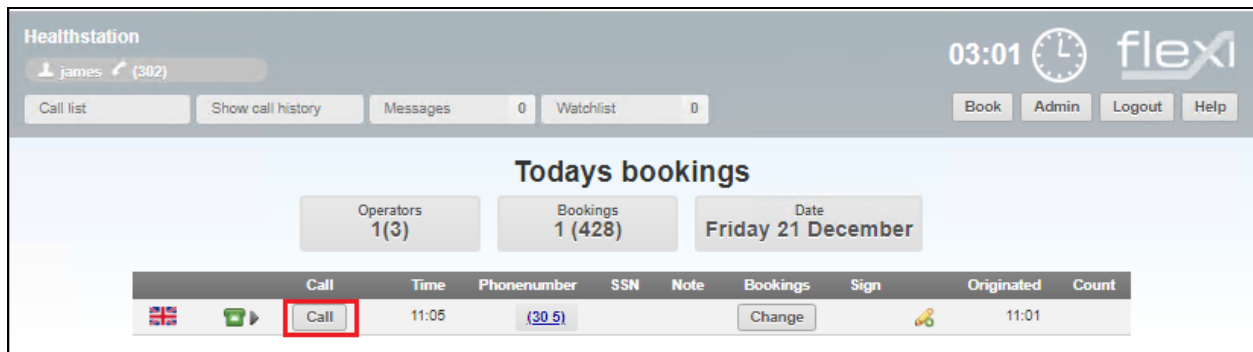
Log in to the correct extension and queue and click on **OK**.



The following screen is displayed once logged in correctly.

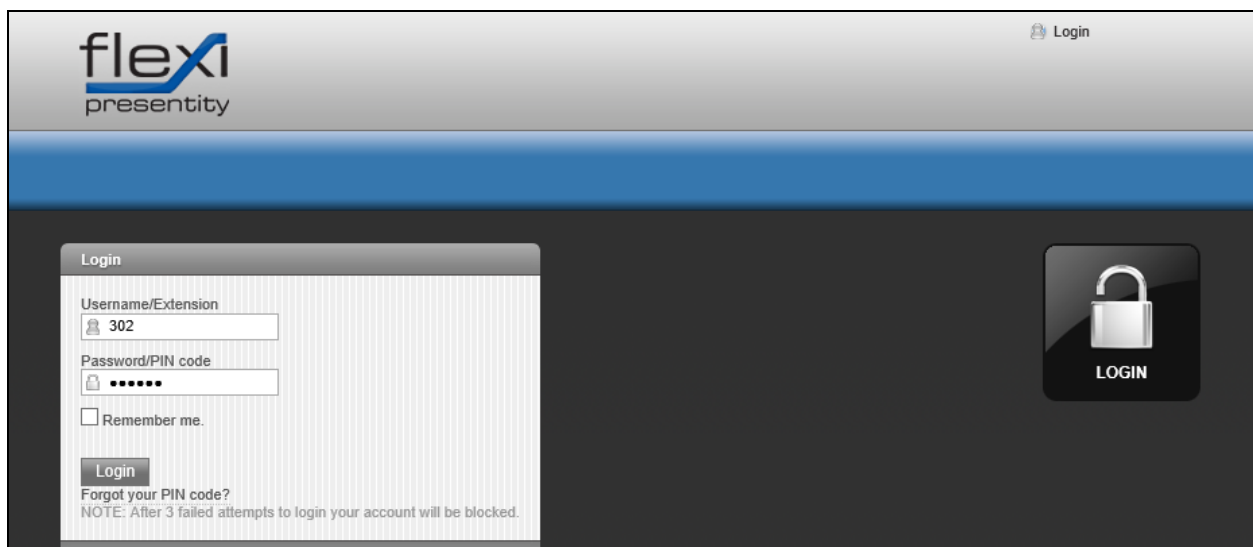


Make a call to the Flexi Tid queue number and request a call back. The following screen is then updated to show that a new call is ready for call back. By clicking on **Call** the phone call to the **Phonenumber** is initiated. Ensure the agent desk phone and called number is connected.



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**.



Once logged in the extension can be diverted as shown below. The extension **302** is diverted to voicemail from **10:45** on the **21st of December** to **10:50** on the **22nd of December**. Make subsequent call to verify diversion is successful and leave a voicemail message. Verify MWI is activated. Retrieve voicemail and verify MWI is de-activated.

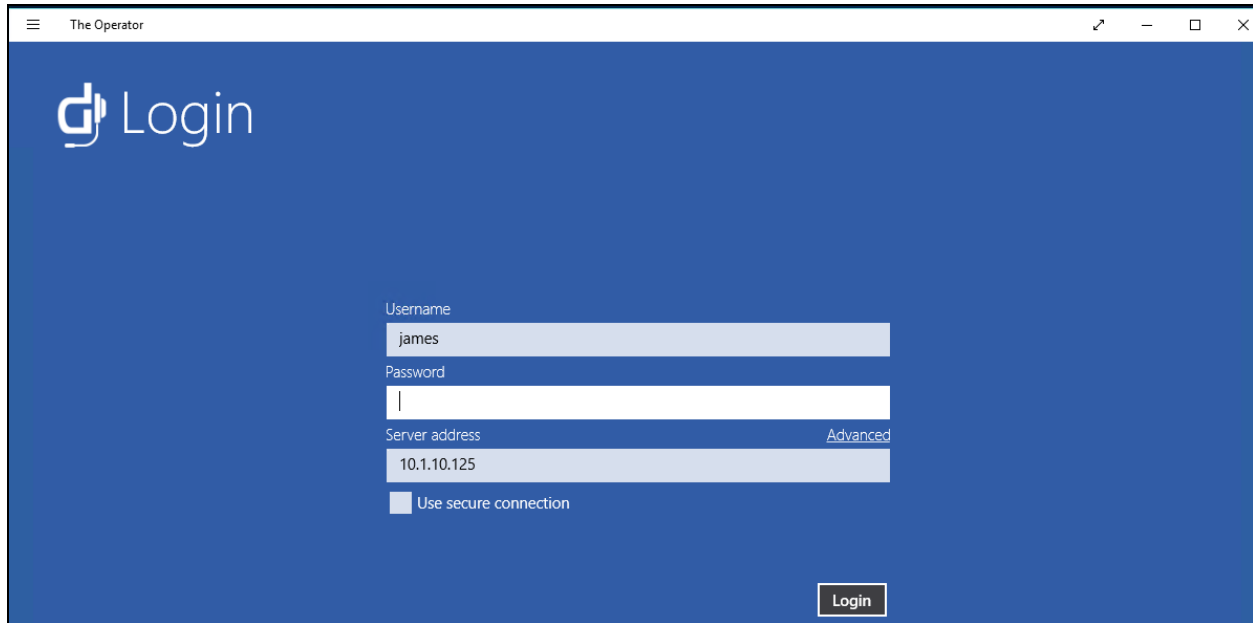
The screenshot displays the 'flexi presentity' web interface. The top navigation bar includes links for DIVERT, CALLS, COLLEAGUES, SETTINGS, and ADD-ONS. A search bar is located on the right. The main content area is divided into several sections:

- User to divert:** A dropdown menu showing 'James Bond (302)'.
- Divert:** A configuration panel for 'James Bond (302)'. It includes a 'Diversion' dropdown set to 'Business trip', 'From' time set to 10:45 on 2018-12-21, 'To' time set to 10:50 on 2018-12-22, a checked 'Divert to voicemail' option, and a 'Voicemail' input field. A 'Save' button is at the bottom.
- Hotkeys:** A panel with a '+ Add / Edit' button.
- Active interceptions:** A table with columns 'Diversion', 'From', and 'Until'.
- Future Interceptions:** A table with columns 'Diversion', 'From', and 'Until'.

A 'DIVERT' button with a handset icon is visible on the right side of the interface.

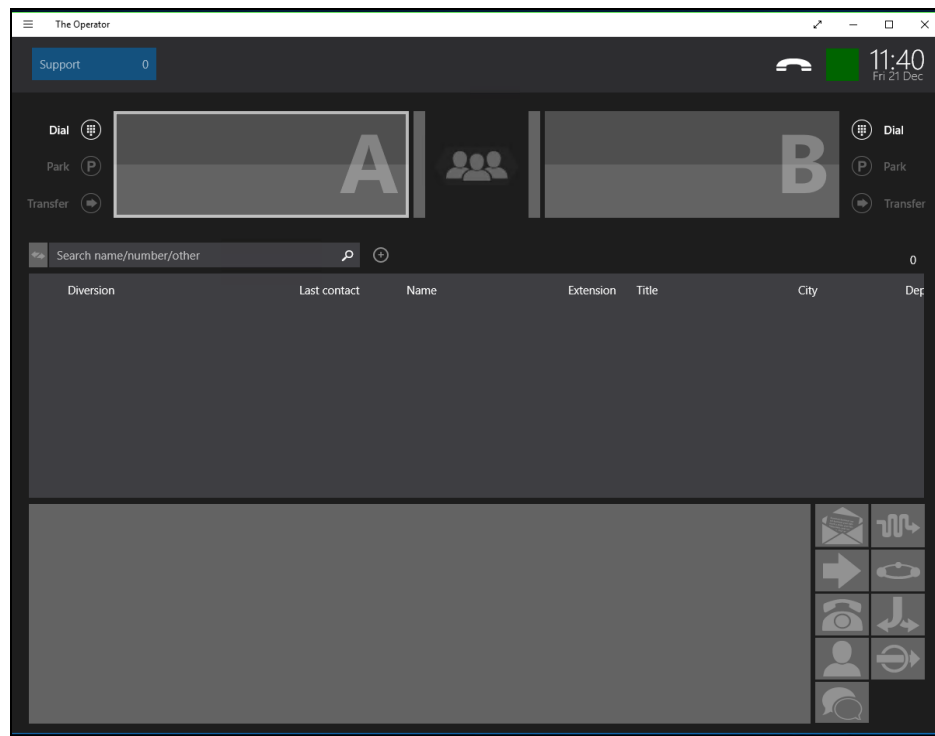
7.4. Verify Flexi Operator

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



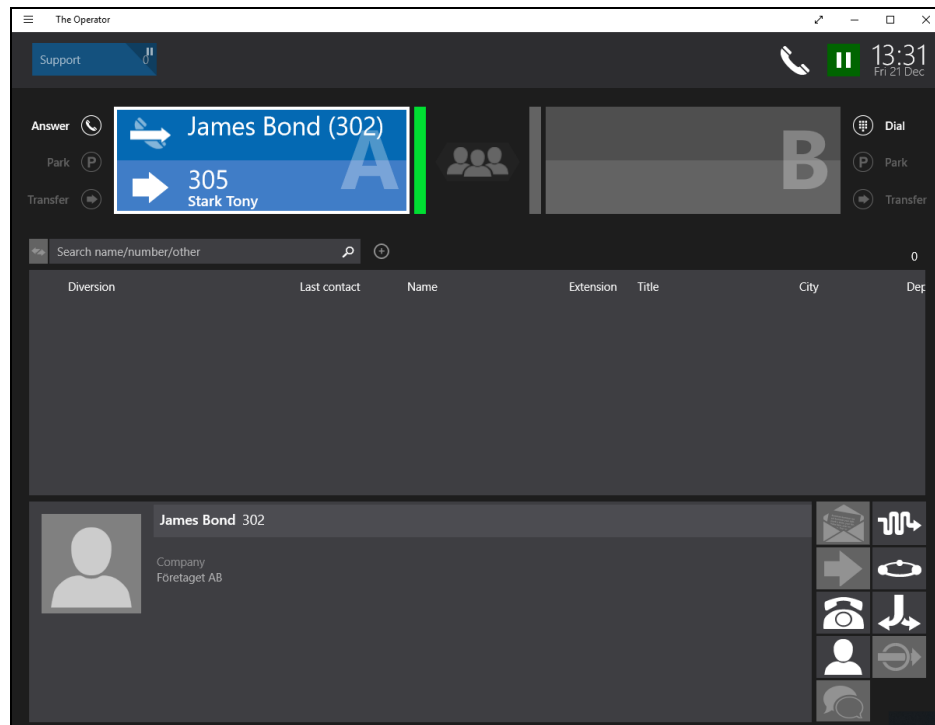
The screenshot shows the Flexi Operator login interface. The window title is "The Operator". The background is a solid blue color. In the top left corner, there is a white logo consisting of a stylized 'G' followed by the word "Login". The login form is centered and contains the following fields: "Username" with the value "james", "Password" which is empty, "Server address" with the value "10.1.10.125", and a checkbox labeled "Use secure connection". A "Login" button is located at the bottom right of the form. The "Server address" field has a link labeled "Advanced" to its right.

Upon login the following screen is shown.

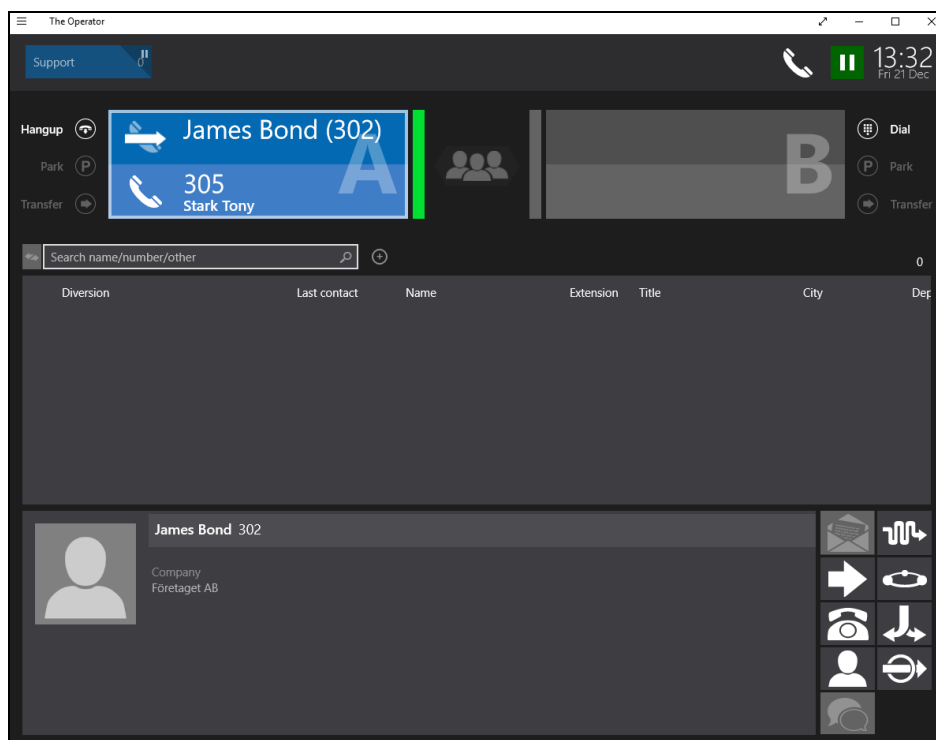


The screenshot shows the Flexi Operator main interface after login. The window title is "The Operator". The interface is dark-themed. At the top left, there is a "Support" counter showing "0". At the top right, there is a clock showing "11:40" and "Fri 21 Dec". Below the clock, there are two large grey boxes labeled "A" and "B". Below these boxes, there is a search bar with the placeholder text "Search name/number/other" and a magnifying glass icon. Below the search bar, there is a table with the following columns: "Diversion", "Last contact", "Name", "Extension", "Title", "City", and "Def". The table is currently empty. On the right side of the interface, there is a vertical sidebar containing several icons for dialing, parking, and transferring calls.

A call from 305 is presented to the operator as shown, the answer button can be pressed to answer the call.

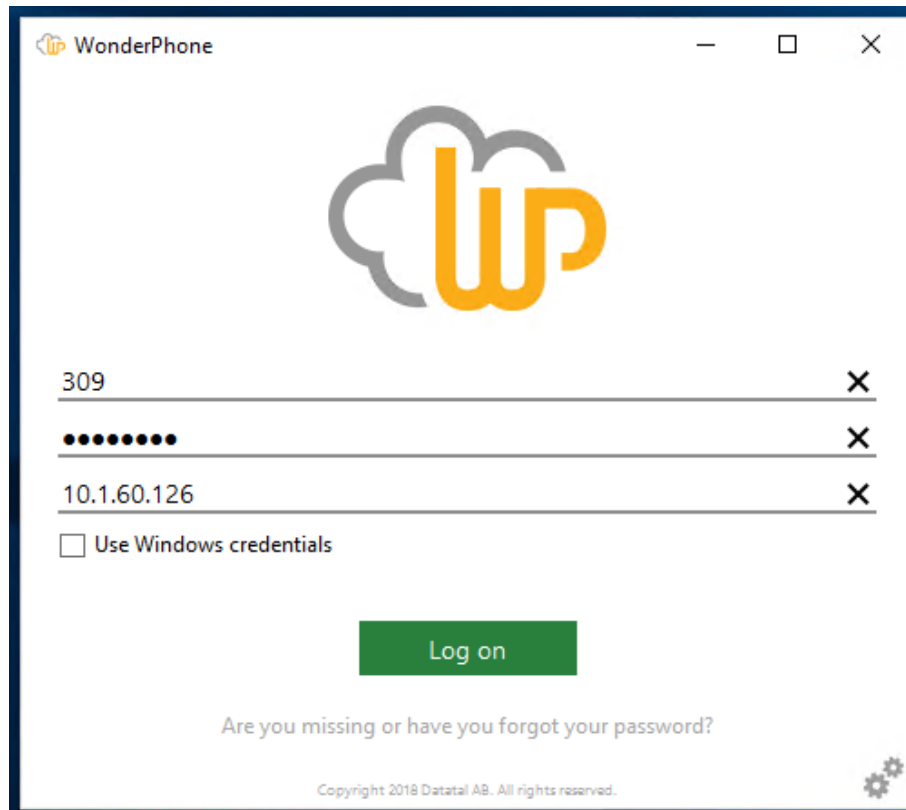


Once the call is answered it can be transferred or hung up as required.



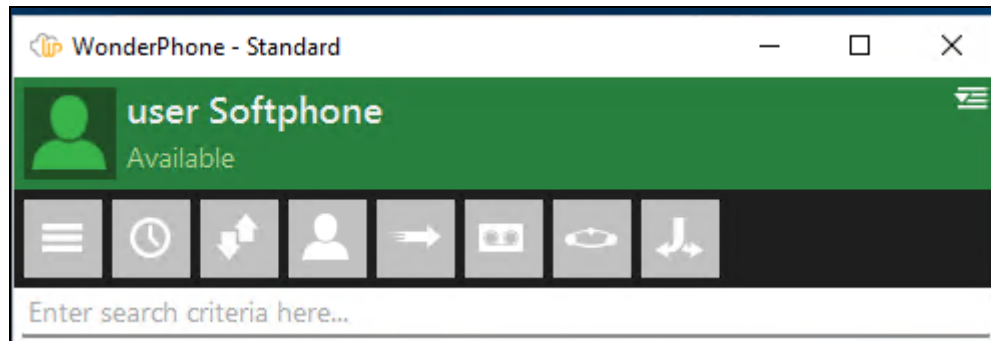
7.5. Verify Flexi Wonderphone

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

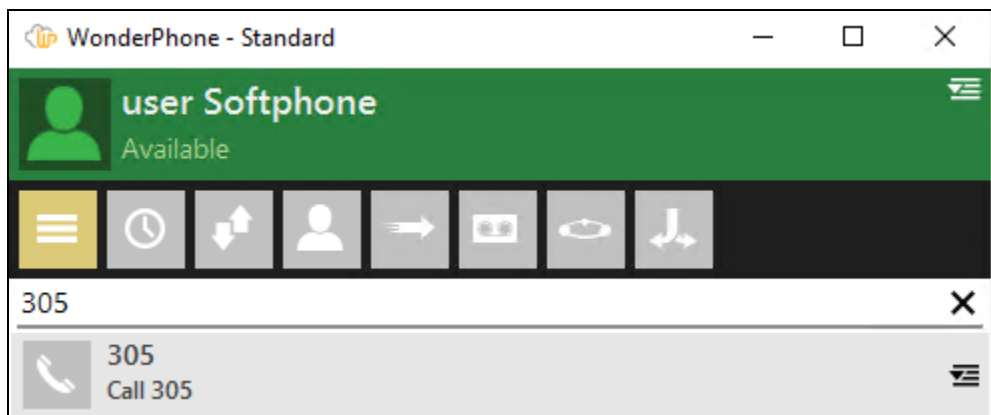


The screenshot shows the WonderPhone application window. The title bar reads "WonderPhone" with standard window controls. The main area features a large logo consisting of a grey cloud and an orange "wp" monogram. Below the logo are three input fields, each with a clear button (X) on the right: the first contains "309", the second contains masked characters "••••••••", and the third contains "10.1.60.126". Below these fields is a checkbox labeled "Use Windows credentials". A green "Log on" button is centered below the checkbox. At the bottom, there is a link "Are you missing or have you forgot your password?" and a footer with "Copyright 2018 Datatal AB. All rights reserved." and a gear icon.

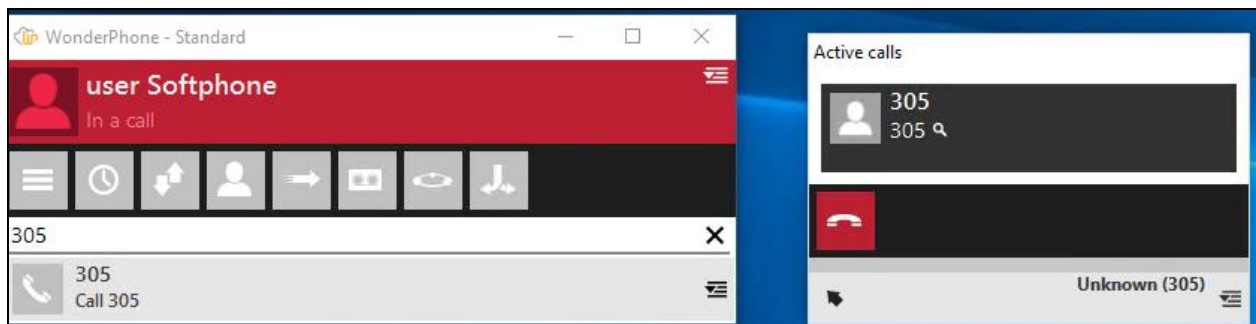
Once logged in, the user logged in status is shown below the name i.e., **Available** in this case.



A call can be made to **305** 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 R11.0. 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] *Administering Avaya IP Office Platform™ with Manager*, Release 11.0, Issue 17a, August 2018
- [2] *Deploying Avaya IP Office Platform Servers as Virtual Machines*, Document 15-1601011 Issue 06e, September 2018

Product documentation for Flexi can be obtained from Datatal AB.

- [1] *Setting for Avaya IPO and Datatal Flexi*
- [2] *Installation manual Datatal Flexi server*

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