



Avaya Solution & Interoperability Test Lab

Application Notes for configuring Capita Secure Information Solutions DS3000 with Avaya IP Office R10.0 using SIP Trunks - Issue 1.0

Abstract

These Application Notes describe the configuration steps for provisioning Capita Secure Information Solutions DS3000 to interoperate with Avaya IP Office R10.0.

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

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

1. Introduction

These Application Notes describe the configuration steps for provisioning DS3000 from Capita Secure Information Solutions to interoperate with Avaya IP Office 500 V2 Standalone R10.0 using SIP trunks to make calls between the DS3000 and the Avaya IP Office. The DS3000 is an Integrated Communication Control System that is used by emergency service customers for answering 999/112 calls and then from the same application using radio communication (TETRA digital radio or analogue PMR) to pass details to mobile resources.

As a radio dispatch deployment with basic PTN/PSTN the DS3000 acts as an end Private Branch Exchange (PBX) and performs call prioritisation and distribution to DS3000 operators as defined by the profile in which they have logged in to the DS3000 application. In this type of configuration the DS3000 has one primary connection to the Avaya IP Office, a SIP connection over SIP trunks. The DS3000 supports basic call control including hold and transfer.

2. General Test Approach and Test Results

The interoperability compliance testing evaluates the ability of the DS3000 application to make and receive calls to and from IP Office endpoints. All calls destined for the DS3000 both locally and from the PSTN are routed to the DS3000 over SIP trunks between the DS3000 and IP Office.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The interoperability compliance testing focuses on various technical testing scenarios to verify the usage of DS3000 with the Avaya solution. In addition, serviceability tests were also performed to assess the reliability and accuracy of the joint solution. The testing focused on the following types of calls:

- **Calls to IP Office Endpoints** – Ensure that calls can be made to IP Office extensions from the DS3000.
- **Calls to DS3000 Operators**– Ensure that calls can be made to the DS3000 operators from IP Office extensions.
- **Calls to PSTN from DS3000 Operators** – Ensure that calls can be made from the DS3000 to PSTN across the SIP trunk through IP Office.
- **Calls from PSTN into DS3000 Operators** – Ensure that calls can be made to the DS3000 from the PSTN by calling into IP Office and across the SIP trunk to the DS3000.
- **Hold/transfer and conference functionality**– Verify that calls can be placed on hold and transferred and conferenced.

- **Caller information is preserved on all calls to/from DS3000** – Ensure that the correct CLID information is preserved.
- **Serviceability testing** – Verify the behaviour of DS3000 application under different simulated LAN failure conditions on the Avaya platform.

Note: All test cases were performed using the following set types; see **Section 4** for more details.

- Avaya 2420 Digital deskphone.
- Avaya 1140 SIP deskphone
- Avaya 9630 H323 deskphone
- Avaya 1616-I H323 deskphone

2.2. Test Results

All test cases passed with the following observations.

1. DTMF must be set to “Inband” on the IP Office setting for DTMF in order for DTMF to work.
2. When transferring to an IP Office set that is CFNA to voicemail- when the operator then hears VM and decides to go back to the original caller – the call to the voicemail appears not to drop and that call remains up. A CANCEL or BYE is not sent by the DS3000. This is because the DS3000 does not send a BYE during a “recall” once the call has been answered. Once the call is dropped by the operator the original PSTN caller is then transferred to the Voicemail.

The following issue was found:

3. The CLID on the DS3000 is not updated after supervised transfer. The DS3000 is either not updating its own GUI correctly or it is not taking the updates for the display information from the “Contact” or the “P-Asserted-Identity” headers.

2.3. Support

Support from Avaya is available by visiting the website <http://support.avaya.com> and a list of product documentation can be found in **Section 9** of these Application Notes. Technical support for the Capita DS3000 product can be obtained as follows.

- Tel : + 44 (0) 8456 041999
- Email: csis.info@capita.co.uk

3. Reference Configuration

Figure 1 shows the setup for compliance testing Capita's DS3000 with IP Office using SIP signalling over SIP trunks to pass callers from IP Office extensions to DS3000 Operators.

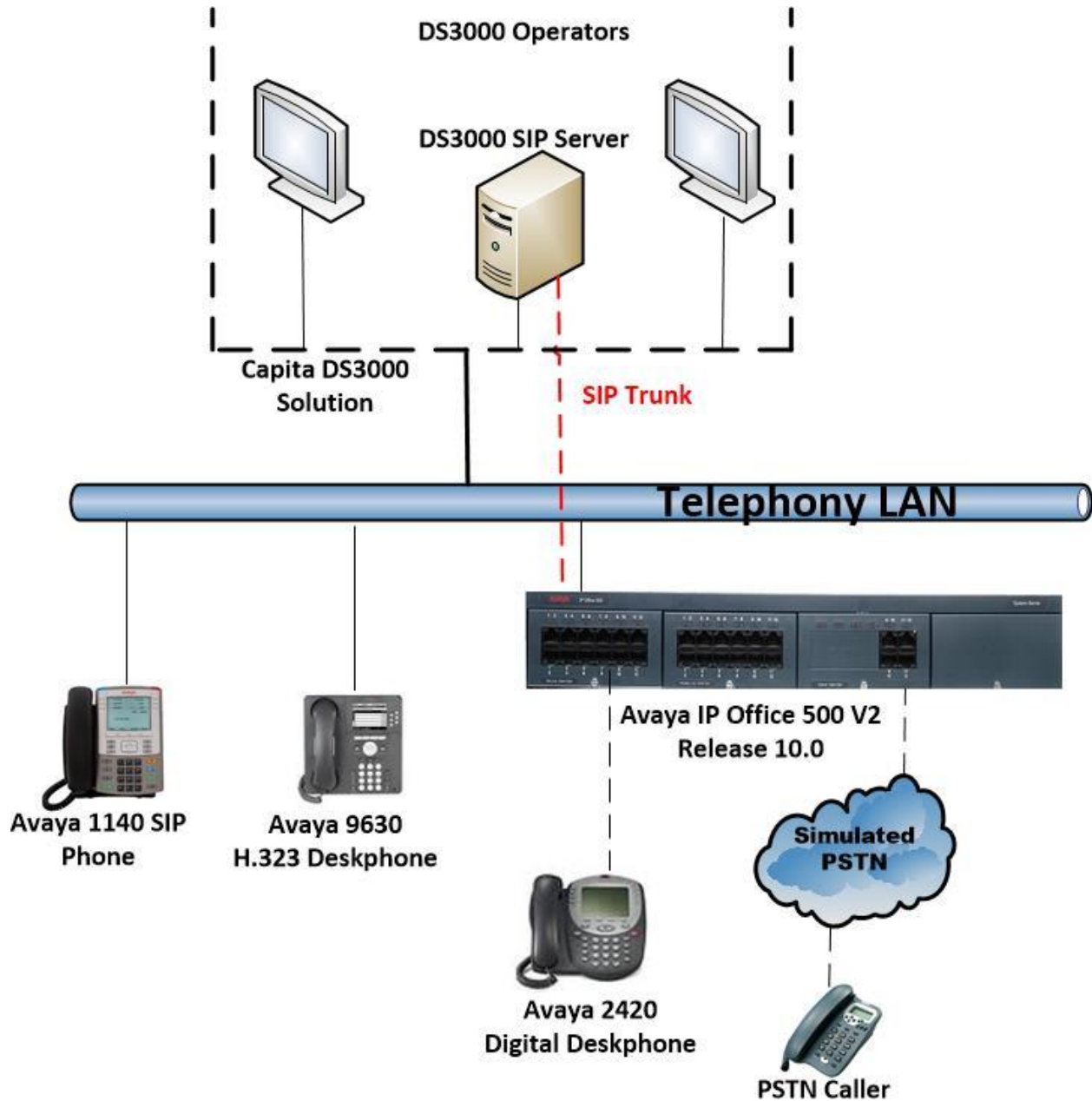


Figure 1: Connection of Capita DS3000 with Avaya IP Office R10.0

4. Equipment and Software Validated

The following equipment and software were used for the compliance test.

Device Description	Versions Tested
Avaya IP Office 500 V2 Standalone	R10.0.0.2.0 Build 10
Avaya IP Office Manager running on a Windows 7 PC	R10.0.0.2.0 Build 10
Avaya 1616-I H323 Deskphone	1608UA1_350B.bin
Avaya 9630 H323 Deskphone	Release s3.186a
Avaya 1140e SIP Deskphone	R04.04.28.00
Avaya 2420 Digital Deskphone	V6.0
Capita DS3000 Solution Kit (DSX Converged Versions 2017 R1 and later) - Aculab Dual Redundant SIP Server	Release 33x Series V6.6.4

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

5. Configure Avaya IP Office

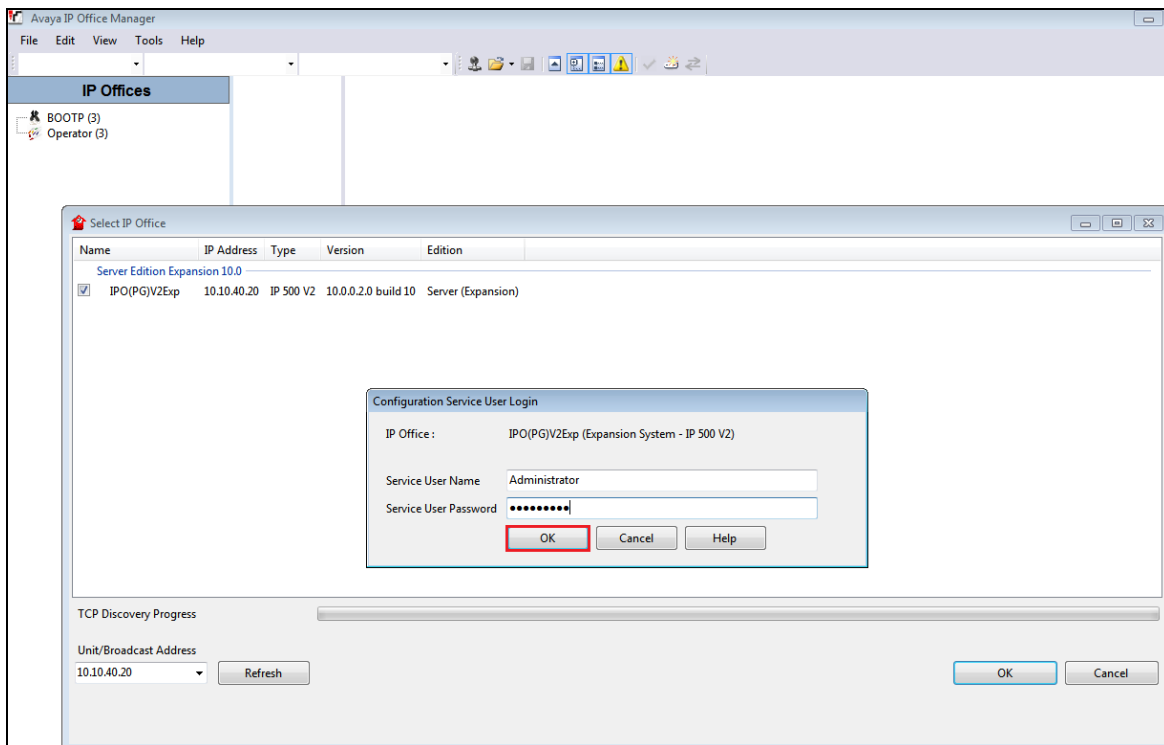
It is assumed that a fully functioning IP office is in place with the necessary licensing. The configuration and verification operations illustrated in this section were all performed using Avaya IP Office Manager. The information provided in this section describes the configuration of IP Office for this solution. 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
- Display LAN Configuration
- Configure SIP Trunks
- Configure Short Codes
- Save Configuration

Note: The configuration of PSTN trunks and routes are outside the scope of these Application Notes.

5.1. Launch Avaya IP Office Manager (Administration)

From the IP Office Manager PC, click **Start → Programs → IP Office → Manager** to launch the Manager application (not shown). Enter the appropriate credentials and click on the **OK** button to receive the IP Office configuration.



5.2. Display LAN Configuration

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

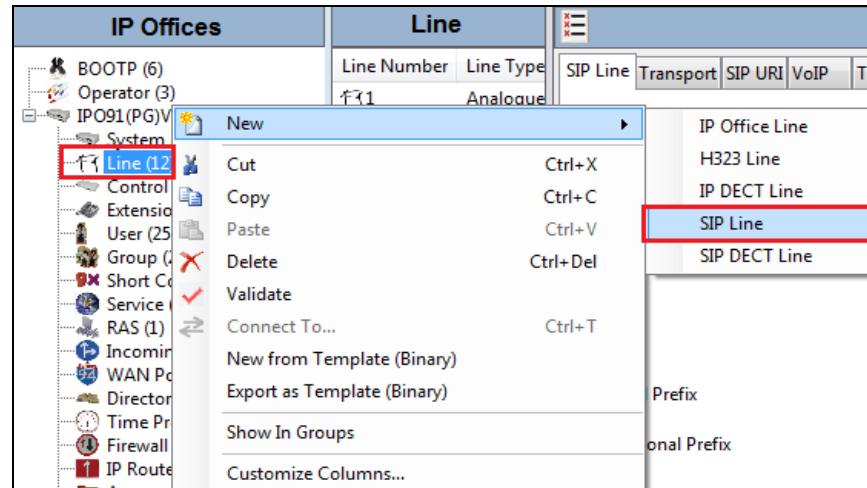
The screenshot shows the IP Office configuration interface. On the left, the 'IP Offices' tree has 'System (1)' selected under 'IPO91(PG)V2Exp'. The main window has the 'System' tab selected, and within it, the 'LAN1' sub-tab is active. The 'LAN Settings' sub-tab is also selected. The 'IP Address' field is highlighted with a red box and contains the value '10 . 10 . 40 . 20'. Other fields include 'IP Mask' (255 . 255 . 255 . 0), 'Primary Trans. IP Address' (10 . 10 . 40 . 1), 'RIP Mode' (None), 'Enable NAT' (unchecked), and 'Number Of DHCP IP Addresses' (10). The 'DHCP Mode' is set to 'Disabled'.

Select the **VoIP** tab and note the following highlighted below. These settings will be required in the setup of the SIP trunk on the DS3000 server in **Section 6.2**.

The screenshot shows the IP Office configuration interface with the 'VoIP' tab selected. The 'SIP Trunks Enable' checkbox is checked and highlighted with a red box. Below it, the 'Domain Name' field is highlighted and contains the value 'devconnect.local'. The 'Layer 4 Protocol' section has 'UDP' and 'TCP' checkboxes checked and highlighted with a red box. The 'UDP Port' and 'TCP Port' are both set to '5060'. Other settings include 'H323 Gatekeeper Enable' (checked), 'Auto-create Extn' (unchecked), 'Auto-create User' (unchecked), 'H323 Remote Extn Enable' (unchecked), 'Remote Call Signalling Port' (1720), 'SIP Registrar Enable' (checked), 'Auto-create Extn/User' (unchecked), 'SIP Remote Extn Enable' (unchecked), 'Challenge Expiry Time (secs)' (10), and 'RTP Port Number Range'.

5.3. Create SIP Trunk

To create the SIP trunk from the IP Office to the DS3000 server, navigate to **System** and right click on **Line** followed by **New** → **SIP Line**.



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

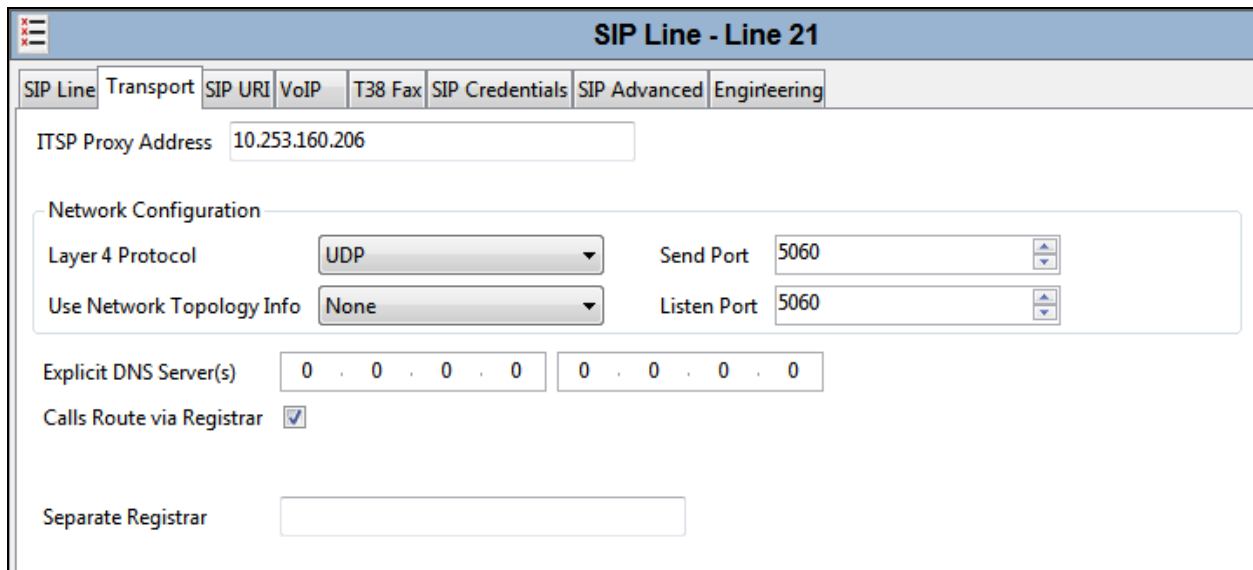
- **ITSP Domain Name** Enter the telephony domain name.
- **Refresh Method** Select **Reinvite** from the dropdown menu.
- **REFER and Transfer** Select **Always** both the **Incoming** and **Outgoing** dropdown boxes.

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

A screenshot of the 'SIP Line - Line 21' configuration window. The 'SIP Line' tab is selected. The window contains various fields and sections:

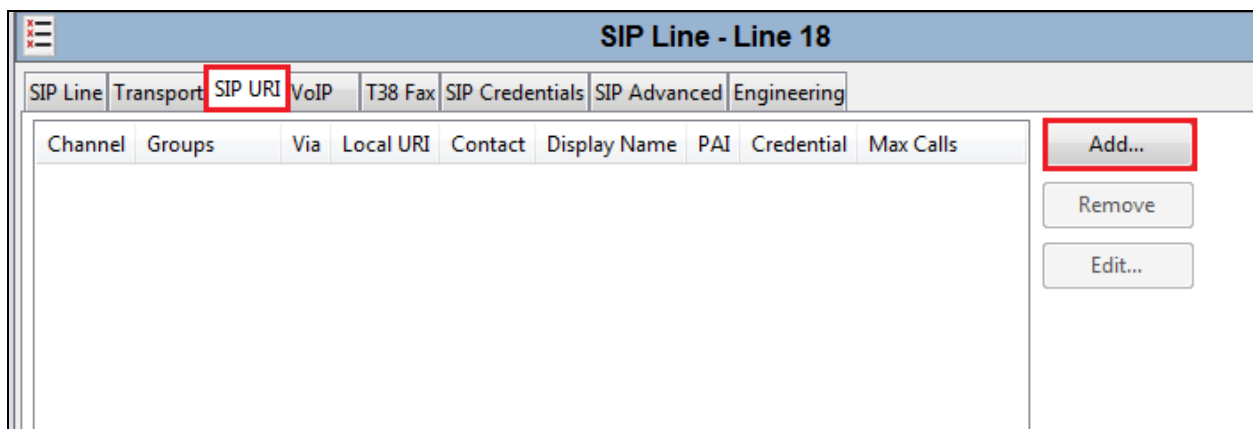
- Line Number:** 21
- ITSP Domain Name:** 10.253.160.206
- Local Domain Name:** (empty)
- URI Type:** SIP
- Location:** Cloud
- Session Timers:**
 - Refresh Method:** Auto
 - Timer (seconds):** On Demand
- Prefix:** (empty)
- National Prefix:** 0
- International Prefix:** 00
- Country Code:** (empty)
- Name Priority:** System Default
- Description:** (empty)
- Redirect and Transfer:**
 - Incoming Supervised REFER:** Auto
 - Outgoing Supervised REFER:** Auto
 - 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. The 'Transport' tab is selected. The 'ITSP Proxy Address' field contains '10.253.160.206'. Under 'Network Configuration', 'Layer 4 Protocol' is set to 'UDP', '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 checked. The 'Separate Registrar' field is empty.

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



The screenshot shows the 'SIP Line - Line 18' configuration window. The 'SIP URI' tab is selected and highlighted with a red box. Below the tab is a table with columns: Channel, Groups, Via, Local URI, Contact, Display Name, PAI, Credential, and Max Calls. To the right of the table are three buttons: 'Add...' (highlighted with a red box), 'Remove', and 'Edit...'.

In the subsequent window, enter the following:

- **Local URI** Enter *
- **Contact** Enter *
- **Display Name** Enter *
- **Identity** Select **None** from the dropdown menu.
- **Header** Select **P Asserted ID** from the dropdown menu.
- **Send Caller Id** Select **Diversion Header** from the dropdown menu.
- **Diversion Header** Select **None** from the dropdown menu.
- **Incoming Group** Enter the SIP trunk number.
- **Outgoing Group** Enter the SIP trunk number.
- **Max Sessions** Enter the amount of trunks to be created (not shown below but at bottom of the screen).

Click the **OK** button.

SIP Line - Line 21

URI	Groups	Local URI	Contact	Display Name	Identity	Header	Originator Number	Send Caller ID	Diversion Header	Credential	Max Sessions
1	21	21	Auto	Auto	Auto	None	PAI	PAI	None	0: <Non...	10

Edit URI

Local URI: Auto

Contact: Auto

Display Name: Auto

Identity: None

Header: P Asserted ID

Forwarding And Twinning

Originator Number:

Send Caller Id: P Asserted ID

Diversion Header: None

Registration: 0: <None>

Incoming Group: 21

Outgoing Group: 21

Buttons: Add..., Remove, Edit..., OK, Cancel

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

SIP Line - Line 21

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

Codec Selection: Custom

Unused:

- G.711 ULAW 64K
- G.722 64K
- G.729(a) 8K CS-ACELP
- G.723.1 6K3 MP-MLQ

Selected:

- G.711 ALAW 64K

Fax Transport Support: None

DTMF Support: Inband

Media Security: Media Security Features Disabled

☐ VoIP Silence Suppression
☐ Local Hold Music
☐ Re-invite Supported
☐ Codec Lockdown
☐ Allow Direct Media Path
 ☐ Force direct media with phones
☒ PRACK/100rel Supported
☐ G.711 Fax ECAN

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

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

Addressing

Association Method: By Source IP address

Call Routing Method: Request URI

Suppress DNS SRV Lookups: ☐

Identity

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

Media

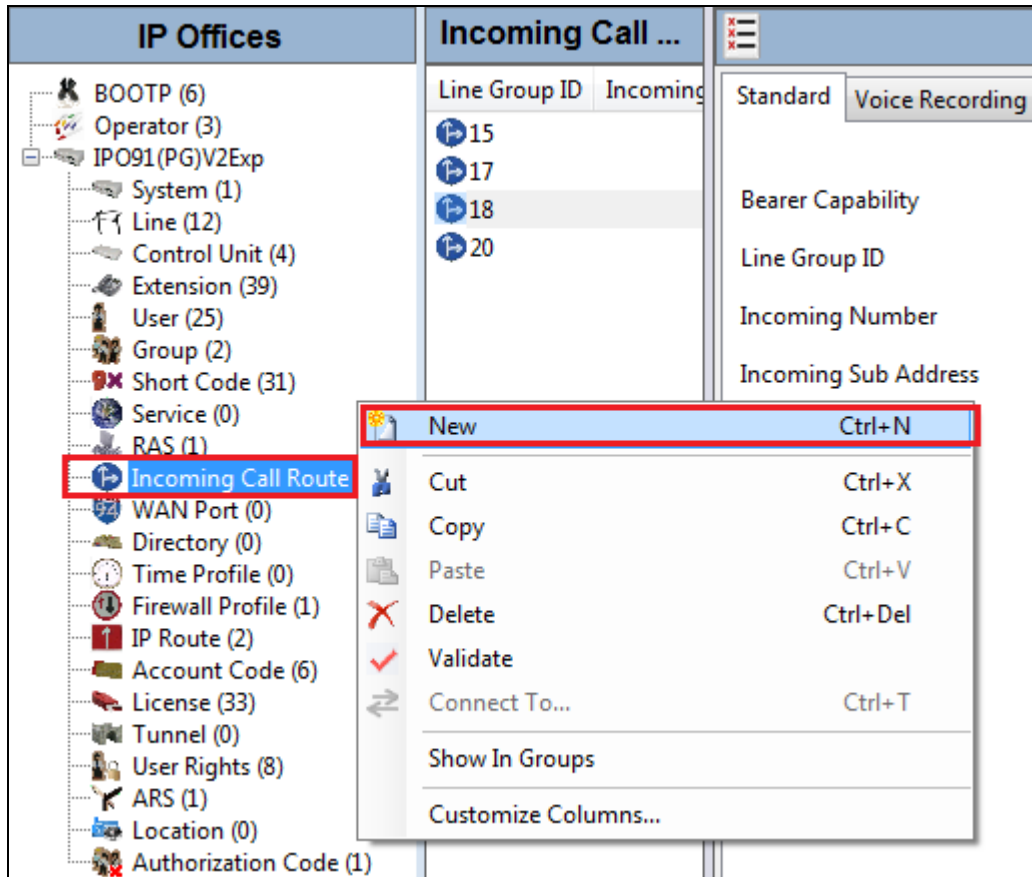
☐ Allow Empty INVITE
☐ Send Empty re-INVITE
☐ Allow To Tag Change
 P-Early-Media Support: None
☐ Send SilenceSupp=Off
☐ Force Early Direct Media
 Media Connection Preservation: Disabled
☐ Indicate HOLD

Call Control

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

5.4. Configure Incoming Call Route

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



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

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

Defaults were used for the remaining fields.

The screenshot shows a configuration window titled '21' with three tabs: 'Standard', 'Voice Recording', and 'Destinations'. The 'Standard' tab is active. It contains the following fields:

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.

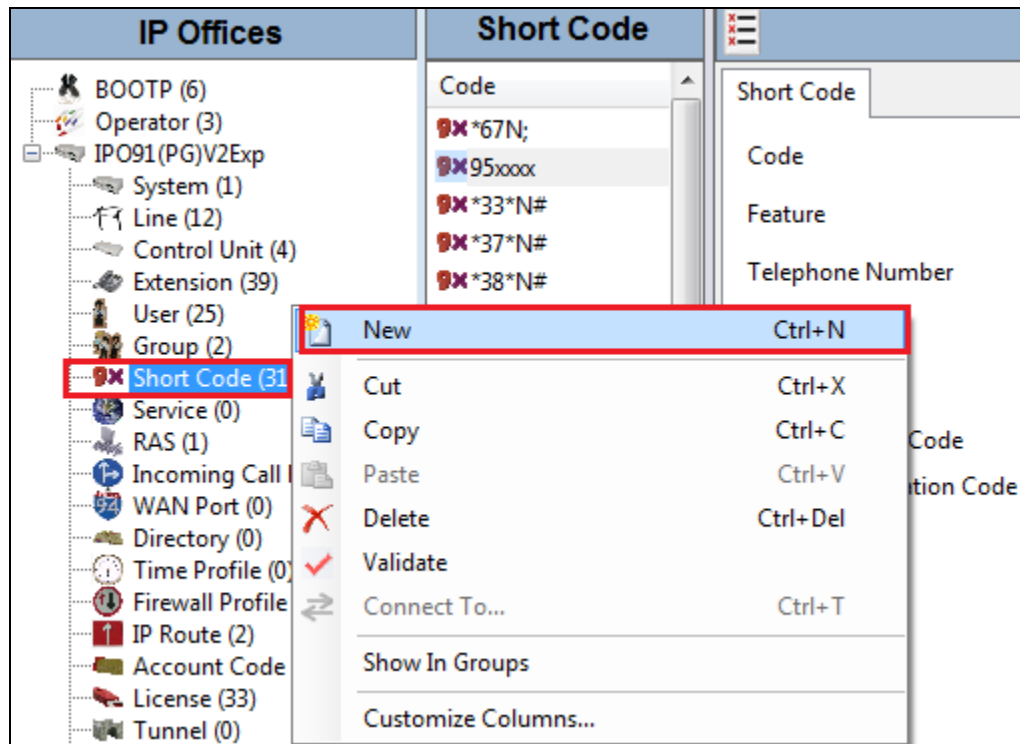
The screenshot shows the same configuration window with the 'Destinations' tab active. It contains a table with the following columns: 'TimeProfile', 'Destination', and 'Fallback Extension'.

TimeProfile	Destination	Fallback Extension
Default Value	.	

At the bottom right, there are three buttons: 'OK', 'Cancel', and 'Help'. The 'OK' button is highlighted with a red box.

5.5. Create Short Code (Route Calls)

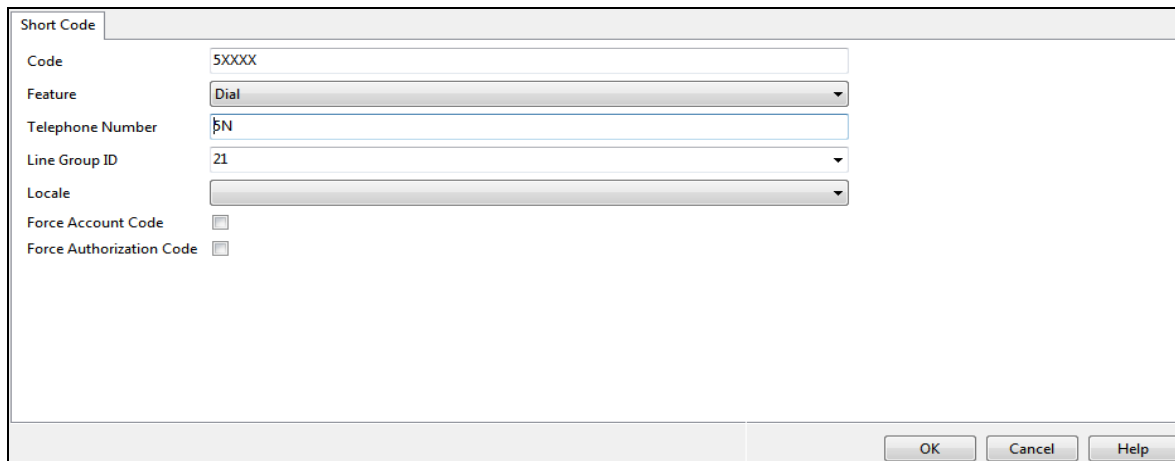
A Short Code needs to be configured on the IP Office to route calls to Flexi server. Right click on **Short Code**, and select **New**.



In the subsequent window, enter the following:

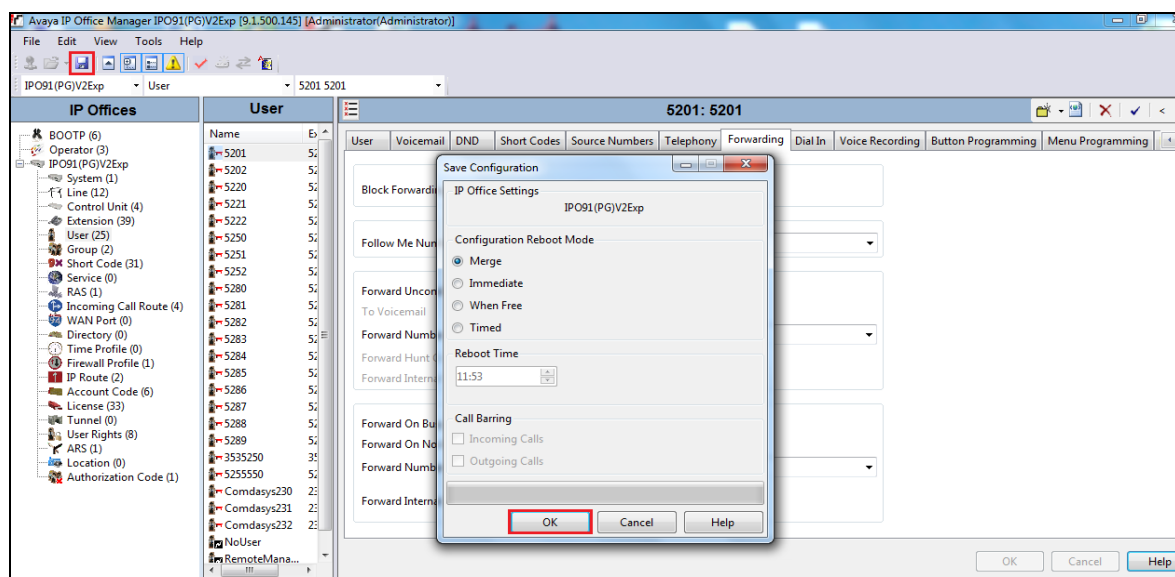
- **Code** Enter the number range that will be routed to Flexi server (during compliance testing, all numbers beginning with 5 that were 5 digits in length were sent to DS3000 server, therefore **5XXXX** was entered).
- **Feature** Select **Dial** from the dropdown menu.
- **Telephone Number** **5N** which is 5 plus the numbers entered after 5.
- **Group Line ID** Enter the Incoming Group number as used in **Section 5.4**.

Click the **OK** button.



5.6. 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.



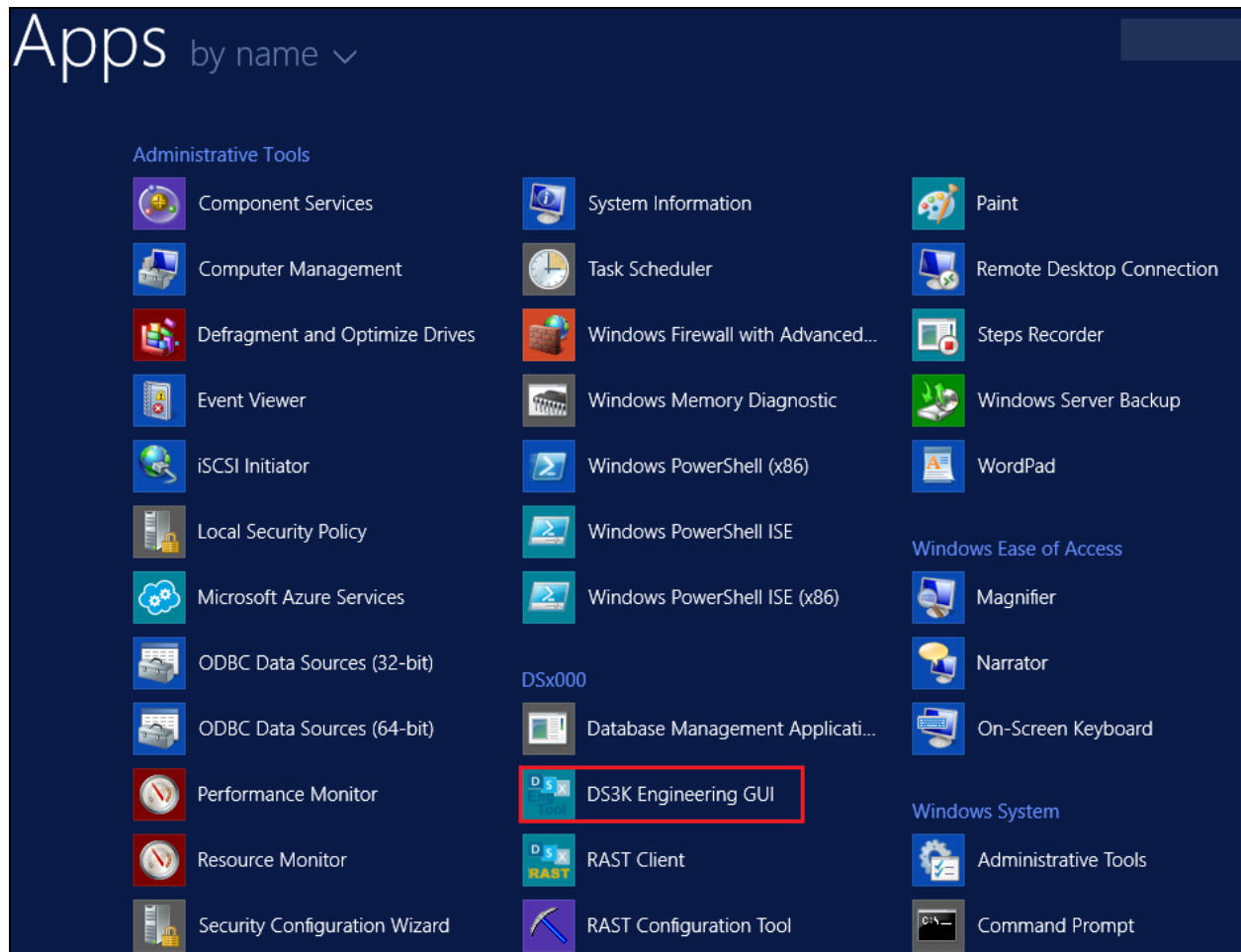
6. Configure Capita Secure Information Solutions DS3000

The following sections describe the steps required to configure the DS3000 application in order to connect successfully with IP Office using SIP trunks.

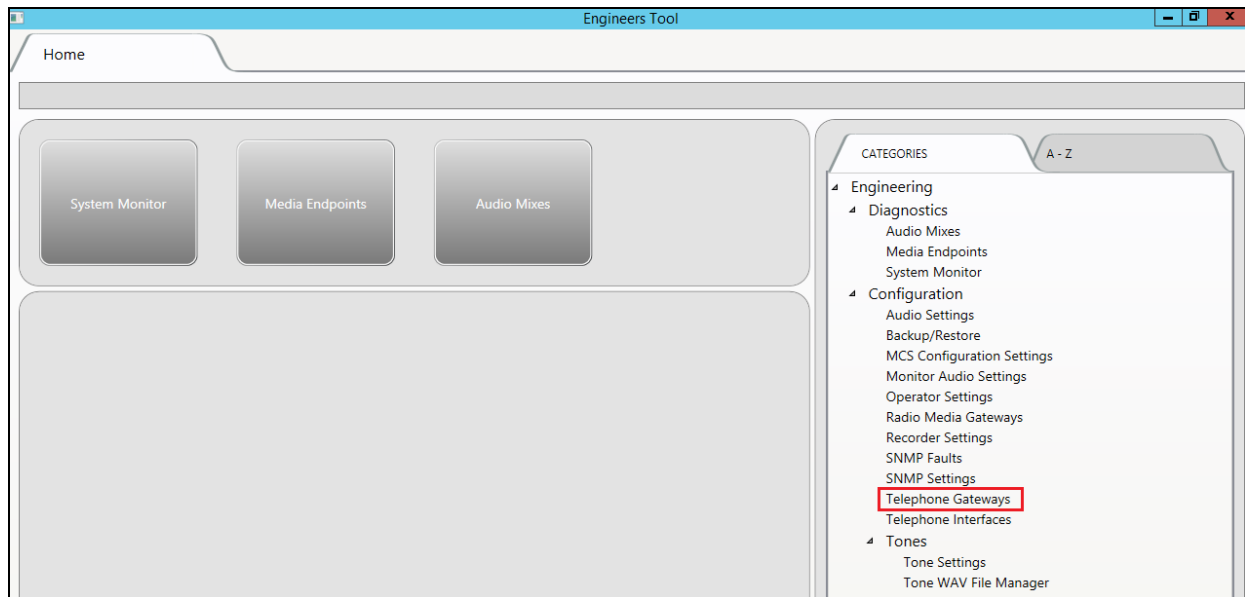
6.1. Configure DS3000 connection to Avaya IP Office

The configuration for the connection to IP Office is performed on the DS3000 FCS machine.

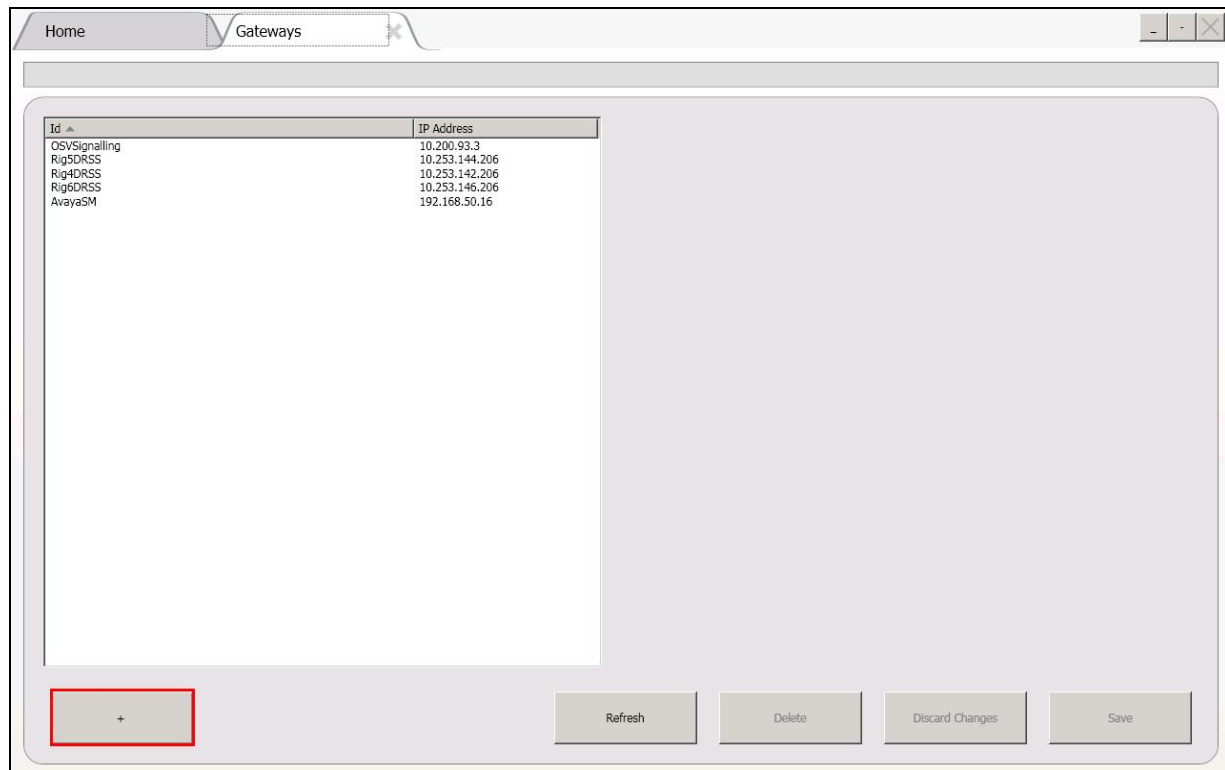
Log into **DS3K Engineering GUI** as shown below on the DS3000 FCS Server.



Once logged in the following screen appears. Select **Telephone Gateways** in the right column, highlighted below.



The **Gateways** tab is opened. Select the + icon at the bottom left of the screen.



Fill in the **Gateway Id** and **IP Address** information. This will be the IP address of the IP Office. Click on **Save** once this is done.

Id	IP Address
RG6TMG01	10.253.146.111
RG6TMG02	10.253.146.113
RG6TMG03	10.253.146.115
AudioCodes	10.253.146.235
AudioCodes8FXO	10.253.146.236
Callvision	10.253.146.233
RG4DRSS	10.253.142.206
AudioCodesM1000	10.253.146.110
Asterisk	10.253.100.99
AACC6	10.12.17.14
SIPP	10.253.146.206
RG6CME	10.253.146.230
AudioCodesM2000	10.253.160.125
AvayaSessionMan	10.10.40.12
AvayaIpOffice	10.10.40.20

Gateway Id: AvayaIpOffice

Address: 10.10.40.20

Buttons: Refresh, Delete, Save (highlighted), Discard Changes

Click on the **Home** tab and select **Telephone Interfaces** in the right column as highlighted below.

Home | Gateways

System Monitor | Media Endpoints | Audio Mixes

CATEGORIES | A - Z

- Engineering
 - Diagnostics
 - Audio Mixes
 - Media Endpoints
 - System Monitor
 - Configuration
 - Audio Settings
 - Backup/Restore
 - MCS Configuration Settings
 - Monitor Audio Settings
 - Operator Settings
 - Radio Media Gateways
 - Recorder Settings
 - SNMP Faults
 - SNMP Settings
 - Telephone Gateways
 - Telephone Interfaces (highlighted)
 - Tones
 - Tone Settings
 - Tone WAV File Manager

The **Telephone Interfaces** tab is opened. Select the + icon at the bottom left of the screen to add a new Telephone interface.

Telephone Interface types: ☒ Normal ☒ Inter Site Enter search string here...

Id	Gateway	Interface Number	Type	Group	Start Line	No of Lines	Card Number
AvayaSM	AvayaSM	2	SIP		31	30	N/A
OSV/SIP	OSV/Signalling	1	SIP		1	30	N/A
RG4	Rig4DRSS	3	SIP		61	10	N/A
RG6	Rig6DRSS	4	SIP		71	10	N/A

Refresh Delete Discard Changes Save

All the information in the right column must be filled in. The screen below shows the information used during compliance testing. Click on **Save** at the bottom right of the screen once all the information has been entered correctly. Set the **Operator ringing tone generation to generate only when there is no early media**, this will provide ringtone when there is no early media on the PBX.

Telephone Interface types: ☒ Normal ☒ Inter Site Enter search string here...

Id	Gateway	Interface Number	Type	Group	Start Line	No of Lines	Card Number
Asterisk Dir	Asterisk	8	SIP		181	30	N/A
TI 06-02 FXO	AudioCodes8FXO	2	Analog	06	403	4	6
TI 06-01 FXO	AudioCodes8FXO	1	Analog	06	399	4	5
TMG03 Analogue	AudioCodesM1000	3	Analog	04	391	4	4
AACCAgent	AudioCodesM2000	5	DPNSS	03	1	30	N/A
Avaya	AvayaSessionMan	3	SIP		91	30	3
Centricity	Callvision	7	SIP		211	30	N/A
Rig4	RG4DRSS	4	SIP		121	10	N/A
TMG04	RG6CME	2	SIP		61	30	N/A
TMG01	RG6TMG01	0	DPNSS	01	151	30	4
TMG02	RG6TMG02	1	DPNSS	02	31	30	N/A
TMG03 ISDN	RG6TMG03	4	ISDN	05	395	4	4
RG6SIP	SIPP	9	SIP		241	30	N/A

Telephone Interface Id: Avaya

Gateway: AvayaOffice

Interface Number: 3

Type: SIP

Group:

Start Line Number: 91

Number of Lines: 30

Operator ringing tone generation: Generate only when there is no early media

Monitor Interface: ☒

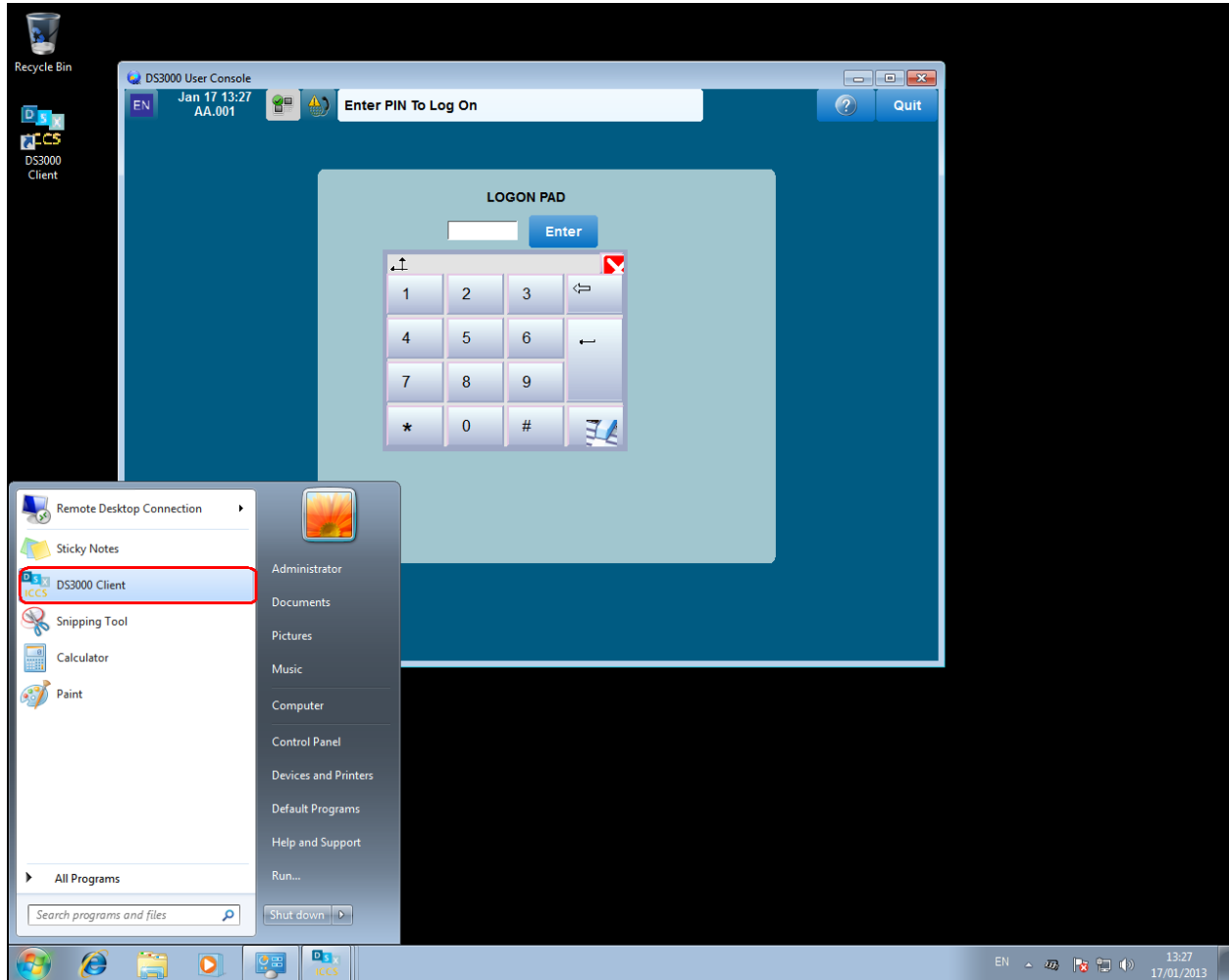
GSIC Number for error reporting: 3

Inter Site: ☐

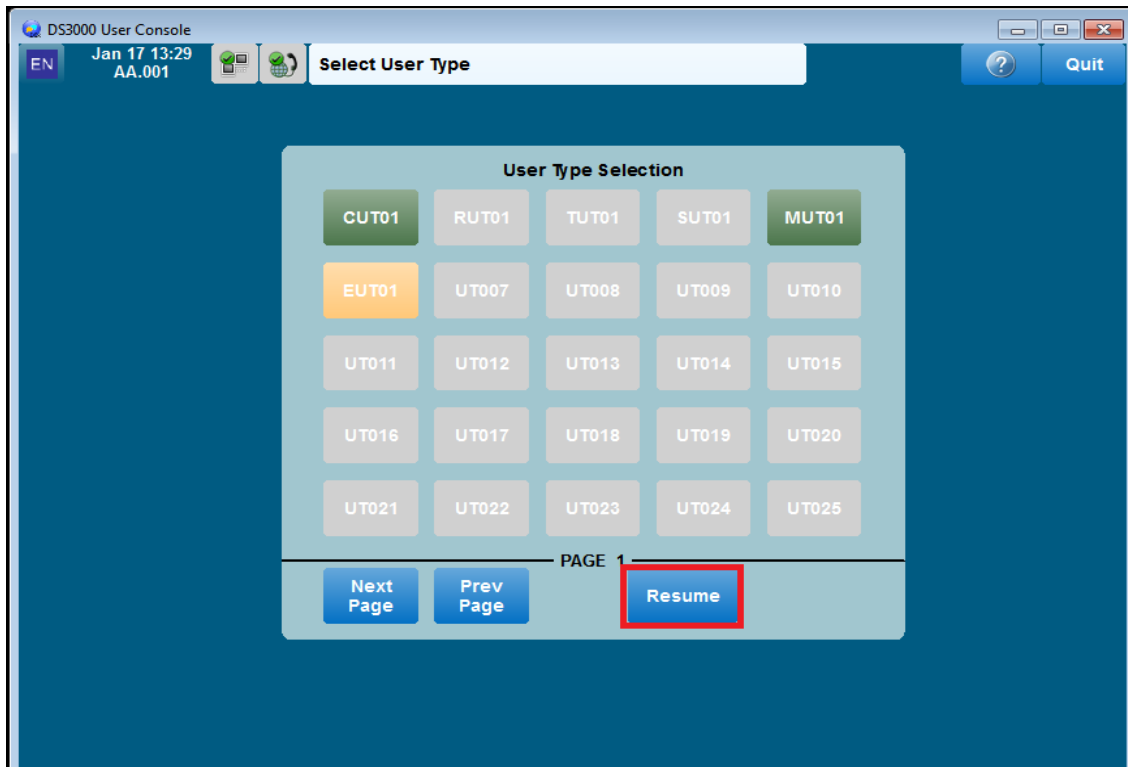
Refresh Delete Discard Changes Save

6.2. Configure the DS3000 extension numbers

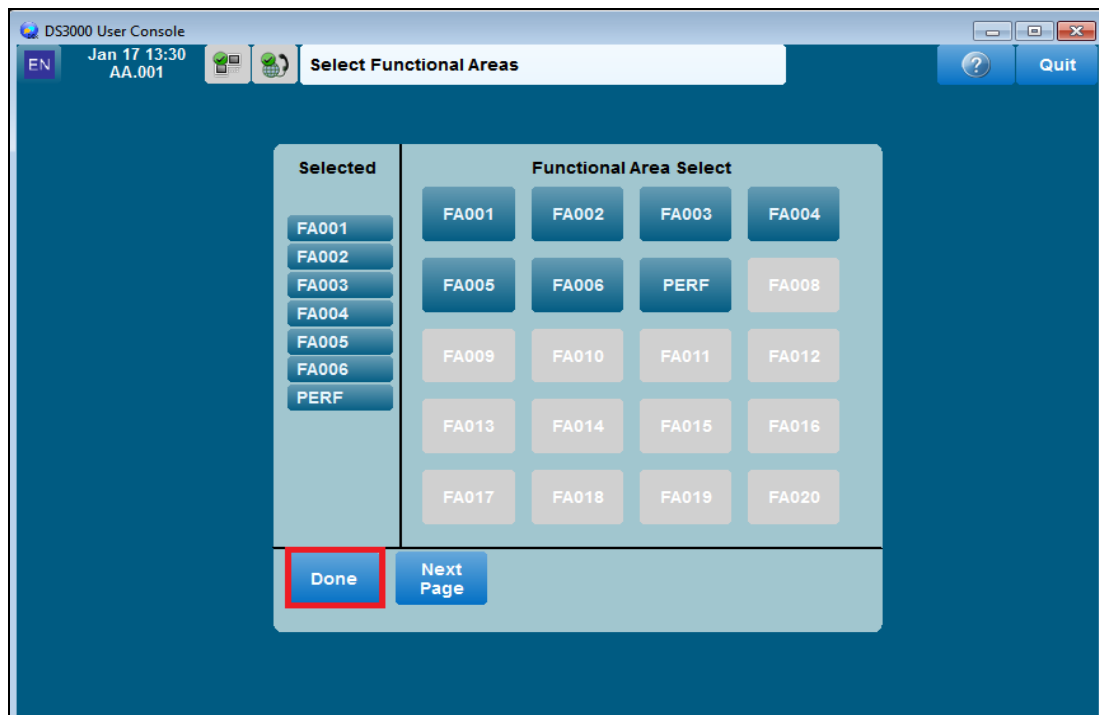
Open the **DS3000 Client** on the DS3000 Client machine. Enter the correct credentials on the **LOGON PAD**.



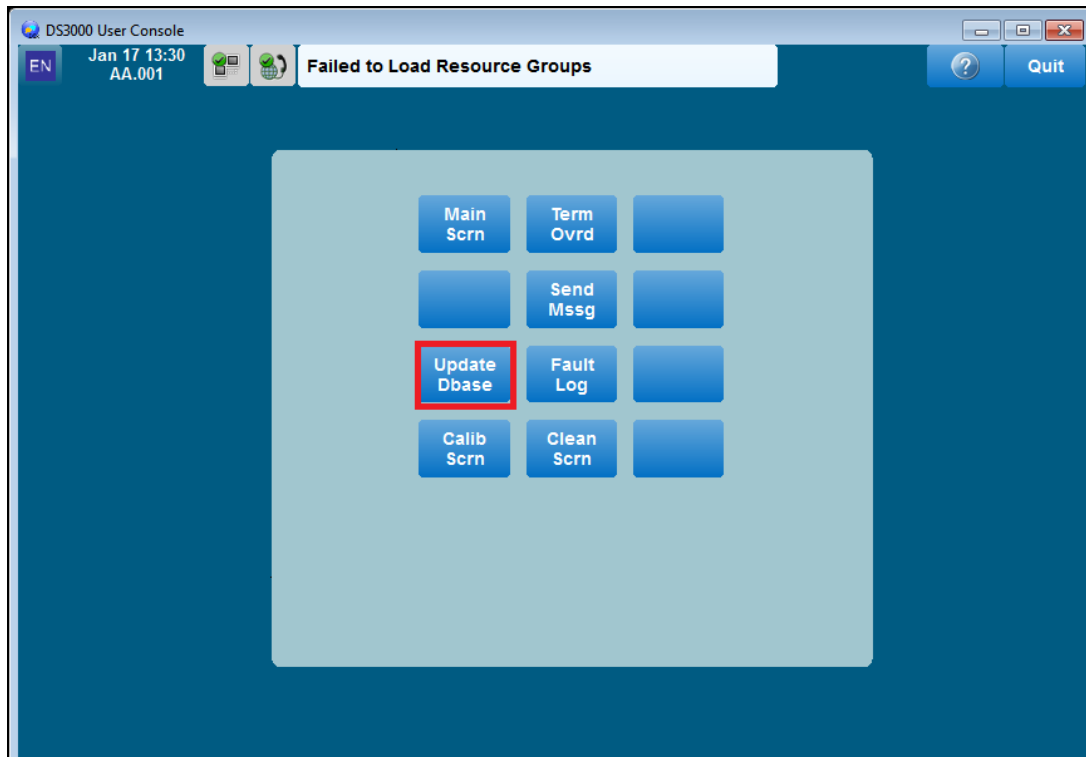
Click on **Resume** at the bottom of the screen as highlight.



Select **Done** at the bottom of the screen as highlighted.



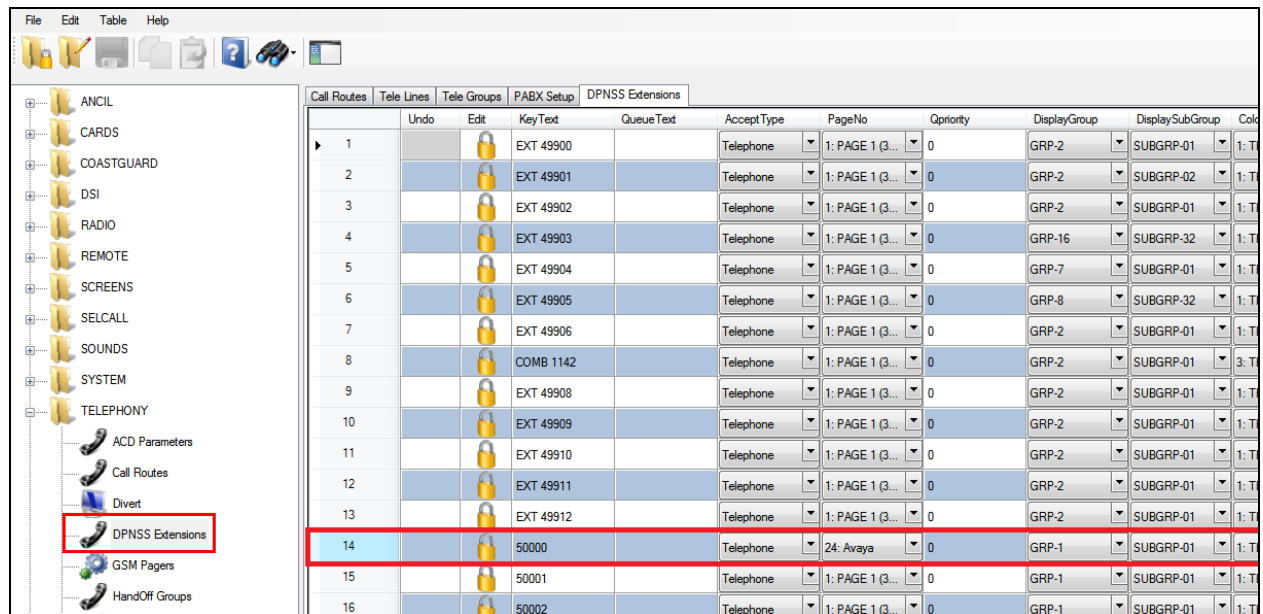
Click on the **UpdateDbase** button highlighted.



Click on the **Call Routes** icon highlighted in the left window. The highlighted row in the right window shows that when 3xxx is dialed that Interface 2 is used. Note: The interface numbers are as defined by the configuration entered in **Section 6.1**.

Call Routes											
	Undo	Edit	DigitsCompare	Leng	Digits	InsertLength	RoutingDigits	GSIC(0)	GSIC(1)	GSIC(2)	GS
1			1	0	0			255	255	255	255
2			1	1	0			0	255	255	255
3			1	2	0			0	255	255	255
4			1	3	0			6	255	255	255
5			1	4	0			1	255	255	255
6			1	5	0			1	255	255	255
7			1	6	0			1	255	255	255
8			1	7	0			1	255	255	255
9			1	8	0			255	255	255	255
10			1	9	0			1	255	255	255
11			2	01	0			255	255	255	255
12			2	22	0			255	255	255	255
13			2	21	0			255	255	255	255
14			2	31	0			255	255	255	255
15			2	4444444	0			255	255	255	255
16			2	51	0			255	255	255	255

Select **DPNSS Extensions** in the left column highlighted. Note the entry highlighted is for the DS3000 Extension **50000**. Ensure **Accept Type** is set to **Telephone**.



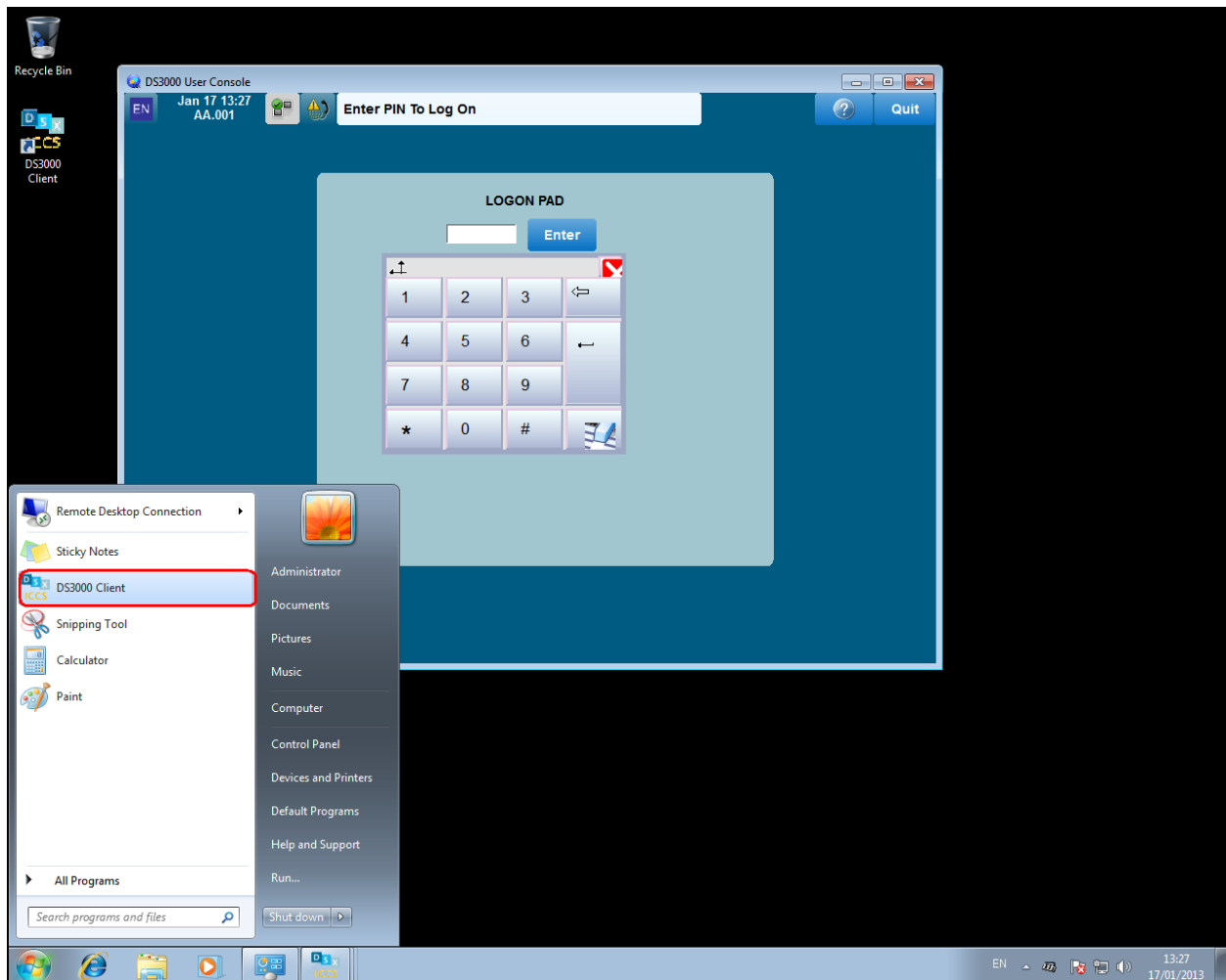
	Undo	Edit	KeyText	QueueText	Accept Type	PageNo	Priority	DisplayGroup	DisplaySubGroup	Col
1			EXT 49900		Telephone	1: PAGE 1 (3...	0	GRP-2	SUBGRP-01	1: T
2			EXT 49901		Telephone	1: PAGE 1 (3...	0	GRP-2	SUBGRP-02	1: T
3			EXT 49902		Telephone	1: PAGE 1 (3...	0	GRP-2	SUBGRP-01	1: T
4			EXT 49903		Telephone	1: PAGE 1 (3...	0	GRP-16	SUBGRP-32	1: T
5			EXT 49904		Telephone	1: PAGE 1 (3...	0	GRP-7	SUBGRP-01	1: T
6			EXT 49905		Telephone	1: PAGE 1 (3...	0	GRP-8	SUBGRP-32	1: T
7			EXT 49906		Telephone	1: PAGE 1 (3...	0	GRP-2	SUBGRP-01	1: T
8			COMB 1142		Telephone	1: PAGE 1 (3...	0	GRP-2	SUBGRP-01	3: T
9			EXT 49908		Telephone	1: PAGE 1 (3...	0	GRP-2	SUBGRP-01	1: T
10			EXT 49909		Telephone	1: PAGE 1 (3...	0	GRP-2	SUBGRP-01	1: T
11			EXT 49910		Telephone	1: PAGE 1 (3...	0	GRP-2	SUBGRP-01	1: T
12			EXT 49911		Telephone	1: PAGE 1 (3...	0	GRP-2	SUBGRP-01	1: T
13			EXT 49912		Telephone	1: PAGE 1 (3...	0	GRP-2	SUBGRP-01	1: T
14			50000		Telephone	24: Avaya	0	GRP-1	SUBGRP-01	1: T
15			50001		Telephone	1: PAGE 1 (3...	0	GRP-1	SUBGRP-01	1: T
16			50002		Telephone	1: PAGE 1 (3...	0	GRP-1	SUBGRP-01	1: T

7. Verification Steps

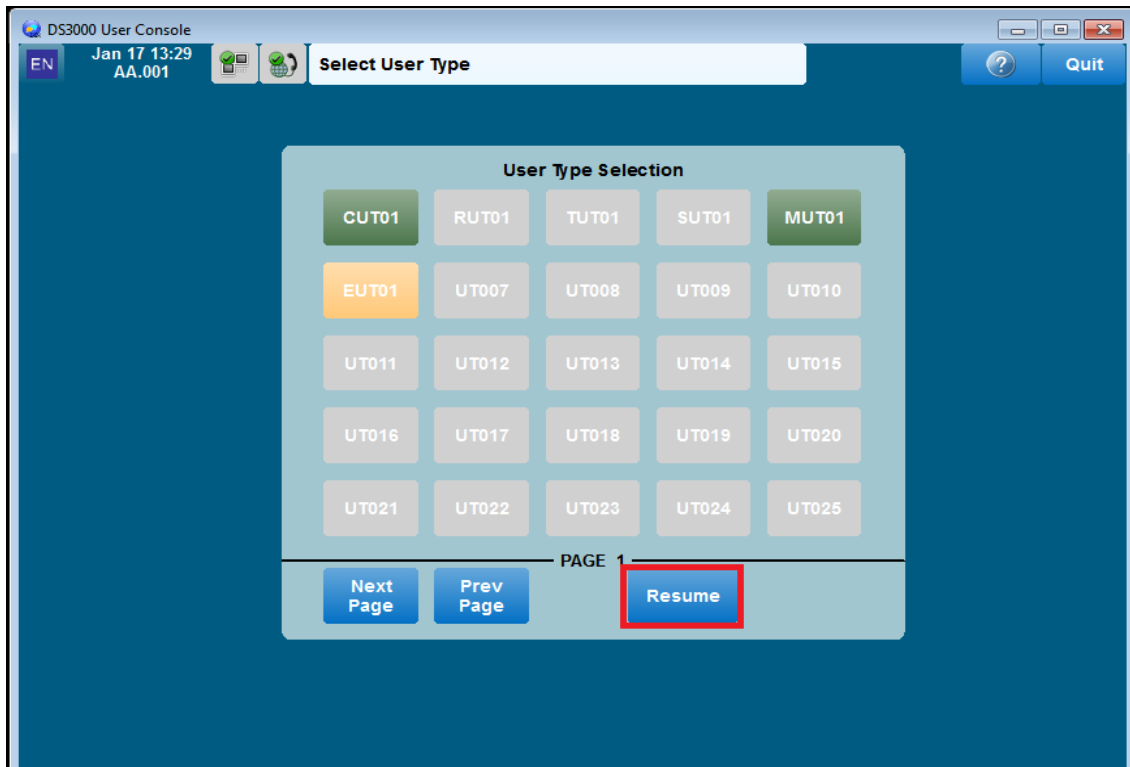
The following step can be taken to ensure that the connection between Capita's DS3000 solution and the Avaya solution is configured correctly. Make a call to and from the DS3000 and verify that the caller can be heard.

7.1. Verify that calls can be made to the DS3000

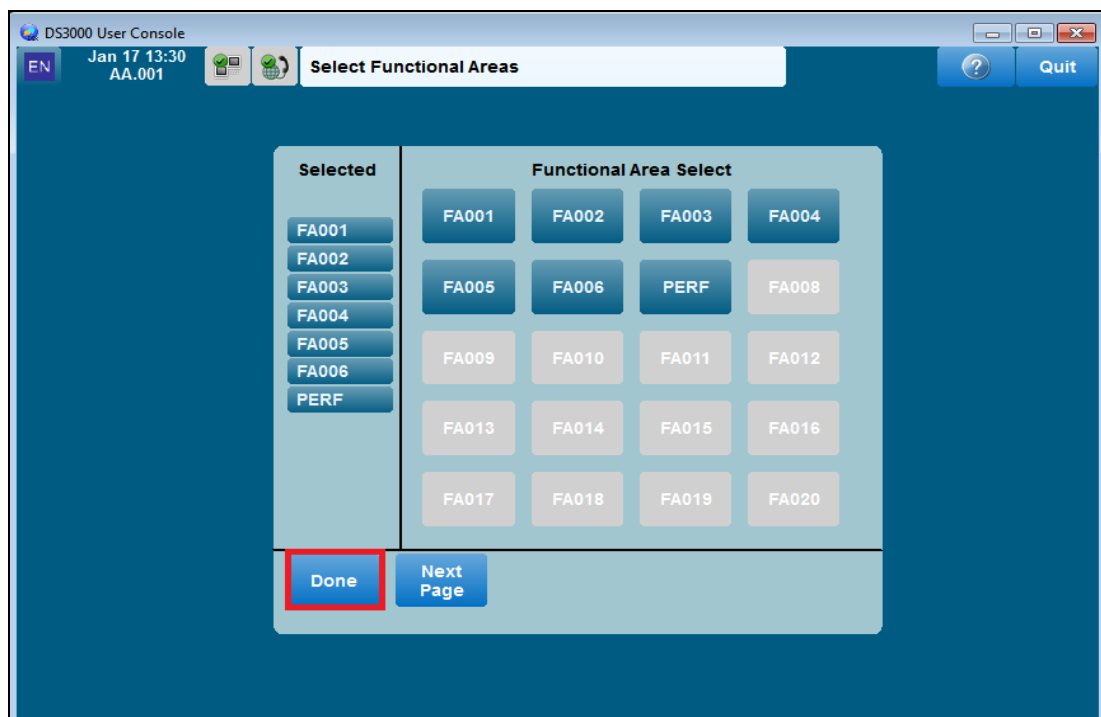
Open the **DS3000 Client** on the DS3000 Client machine. Enter the correct credentials on the **LOGON PAD**.



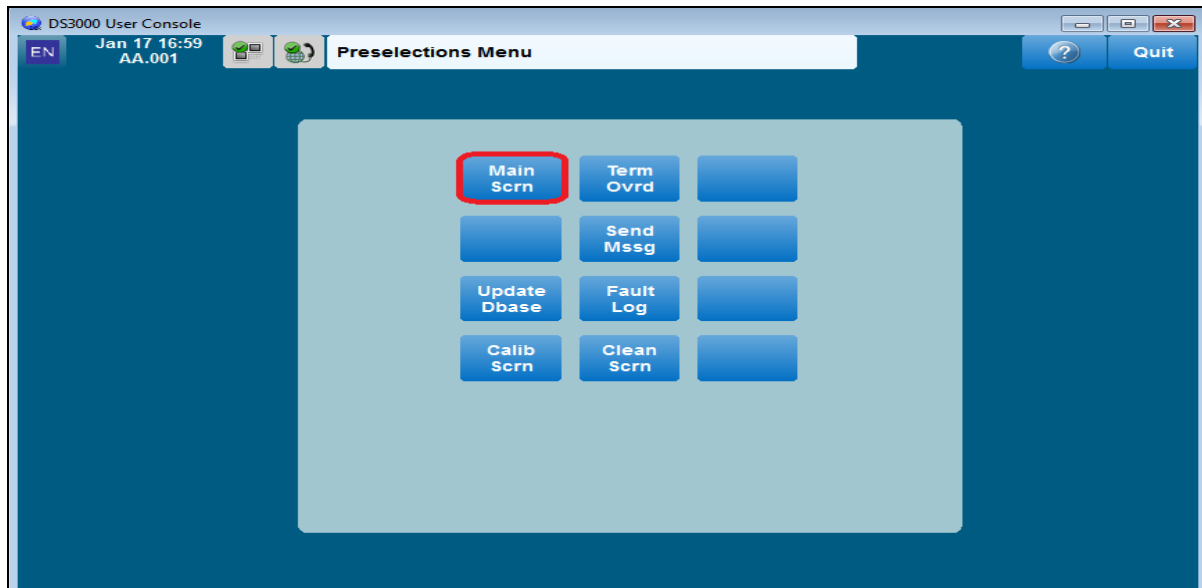
Click on **Resume** at the bottom of the screen as highlight.



Select **Done** at the bottom of the screen as highlighted.



Click on the **Main Scrn** button highlighted below.



Once a call is presented to the DS3000 the following screen should appear. Click on the **Take Call** button on the bottom right of the screen to take the call.



8. Conclusion

These Application Notes describe the configuration steps required for DS3000 from Capita Secure Information Solutions to successfully interoperate with Avaya IP Office R10.0. Please refer to **Section 2.2** for test results and observations.

9. Additional References

This section references documentation relevant to these Application Notes. The Avaya product documentation is available at <http://support.avaya.com>, where the following documents can be obtained.

- [1] *IP Office 10.0 IP500/IP500 V2 Installation*, Document Number 15-601042, Issue 27m, July 2, 2013.
- [2] *IP Office Release 10.0 Manager 10.0*, Document Number 15-601011, Issue 29u, April 5, 2013.
- [3] *IP Office System Status Application*, Document Number 15-601758, Issue 07a, November 26, 2012.
- [4] *IP Office System Monitor*, Document Number 15-601019, Issue 03c, March 1, 2013

Product documentation for DS3000 can be requested from Capita or may be downloaded from <http://www.capitasecureinformationsolutions.co.uk>

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