Unify

Atos Unify Ready Technology connectivity certification

The connectivity of

IP-DECT Release 11

developed by Ascom has been certified at the SIP Interface of Atos Unify OpenScape Voice V10 for voice calls in accordance with the respective test report, dated June 1st, 2021.

The test was conducted conforming to DIN EN ISO 9001. This certificate is only valid in conjunction with the full test report and the notes contained therein. **Please consider that the test report only covers the functionality of the interface. The certificate and test report are not good for a statement of end-to-end functionality.**

Munich, June 1st, 2021

Andre Bergmann

Director Technology Partner Program

Atos Unify Ready

Trusted partner for your **Digital Journey**

Atos

Test Report of Certification

Ascom IP-DECT R11

with

Atos-Unify OpenScape Voice V10R1

Status: Released
Release Date: 01-06-2021

Contents:

History of Change

Version	Date	Description	Author(s)
1.0	May 11th, 2021	Initial Creation	Aikaterini Georgaka Atos Greece SM SA Email: **aikaterini.georgaka@atos.net**
1.1	May 12th, 2021	Confirmation of test results and test report.	Michael Korakis Atos Greece SM SA E-mail: **michail.korakis@atos.net**
1.2	May 24th, 2021	Editorial changes in sub-sections 1.4.2, 3.3.5, 3.4.7, 3.4.14, 4.1.6 & 4.1.8	Mathew Williams Ascom AG E-mail: **Matthew.Williams@ascom.com**
2.0	June 1st, 2021	Review	Dimitrios Galanakis Atos Greece E-mail: Dimitrios.galanakis@atos.net

1 Overview

1.1 Test Objective

This document provides the test report and approach used by CSL Athens lab to perform interoperability testing between Ascom IP-DECT system and Atos-Unify OpenScape Voice V10 IP-PBX for the documented product versions. Furthermore, this document provides the test cases, the success criteria, processes and execution steps of testing that was performed.

1.1.1 Basis Equipment

Test Equipment:	Virtual OpenScape Voice co-located cluster Virtual OpenScape CMP / Media server Virtual OpenScape Xpressions Voicemail server Mediatrix 4402plus BRI gateway device OpenScape Deskphone CP400 and CP600
Software Release:	OS Voice version: V10R1.7.5 OS UC version: V10R2.6.2 OS Xpressions version: V7 R1 FR5 HF38 Mediatrix 4402plus firmware: Dgw 44.1.1605 SIP phones firmware: V1 R8.2.1 SIP

1.1.2 Ascom IP-DECT system

Certification:	Ascom IP-DECT system registered to OS Voice V10. Test of various call scenarios and features between SIP, PSTN subscribers using Ascom IP-DECT system. Finally, voice codec and DTMF verification scenarios have been taken into consideration, too.
Test Equipment:	Ascom IP-DECT Base Station (single system) Ascom DECT Handsets d63-Talker
Software Release:	Ascom IP-DECT Base Station version: IPBS3[11.3.4], Bootcode[11.3.4], Hardware[IPBS3-A3/1A] Ascom DECT Handset d63-Talker firmware: 2.11.4
Manufacturer:	Ascom (Sweden) AB Grimbodalen 2 SE-417 49

Description:	Ascom DECT handsets are registered to the OpenScape Voice PBX as SIP users, therefore enabling them to make/receive internal and PSTN/external calls and have full voicemail and other telephony features available by OpenScape Business solution. The wireless communication is made using Ascom IP-DECT Base Station connected to the same LAN as OpenScape Voice. The Ascom IP-DECT Base Station operates as a conduit between the OpenScape Voice and the Ascom DECT handsets. After the Ascom DECT wireless handsets register with the Ascom IP-DECT Base Station system, the Base Station registers the handsets to OpenScape Voice.
Documentation:	Refer to your Ascom supplier for documentation.
Test Network:	Ascom handsets register with Ascom IP-DECT Base Station system and the latter registers the handsets with OpenScape Voice as SIP stations. OpenScape Voice provides the VoIP telephony facilities to Ascom DECT handset SIP stations. Additionally, Atos-Unify SIP are connected to OpenScape Voice, too. OS Voice is connected (SIP) to a Mediatrix 4402plus BRI gateway, that provides access to PSTN (OTE ITSP). Voicemail services are provided by an OS Xpressions server (SIP connectivity with OS Voice). See **fig.1** in section 1.7.
Test Configuration:	Refer to chapters 2 & 4.

1.2 Test Strategy

The main goal of the testing activities is to test the interoperability of Ascom DECT handsets with OS Voice. The testing activities evaluate the ability of the Ascom DECT handsets to make and receive audio calls to and from Atos-Unify SIP desk phones and PSTN endpoints. The Atos-Unify OpenScape Xpressions voicemail server was used to allow users to leave voicemail messages and also to demonstrate Message Waiting Indication and DTMF sending functionalities of the Ascom DECT handset devices.

Within the aforementioned framework, the testing procedure will focus on:

- The SIP interface between Ascom IP-DECT Base Station and OSV.

- OSV call features on DECT handsets.

- The system’s failure/recovery behavior.

1.2.1 Test Intensity

The scope of the tests is to execute / to verify the solution performs within the limits of the system requirements, targeting the end product. To accomplish this, feature and solution-based test cases are created, inspected, and executed under a real system environment (mirroring as close as possible real customer's environment).

Notes:

The testing of the product with regard to compliance to requirements for Product Safety, EMV, Network Access Interfaces and Radiation Protection were not performed.

Unify Software and Solutions GmbH & Co. KG therefore assumes no responsibility for the compliance to these requirements.

1.2.2 *Measuring / Test Instruments*

Tracing and monitoring available on all IP phones, servers, gateways and laptops. No special hardware required.

- Wireshark

1.3 *Realization Data*

Test Preparation:	April 22nd, 2021
Test Duration:	April 23rd – May 7th, 2021
Test Location:	Atos Greece SM SA CSL Athens lab 455, Irakleiou Avenue 141 22, N. Irakleio, Athens, Greece
Test Personnel:	Atos Greece SM SA, Aikaterini Georgaka, aikaterini.georgaka@atos.net Ascom Sweden AB, Matthew Williams, Matthew.Williams@ascom.com

1.4 *Test Result Summary*

The certification of Ascom IP-DECT system with OpenScape Voice V10R1 PBX is passed with certain restrictions and observations applied.

1.4.1 *Remarks*

- **Number Presentation**

  By design Ascom handsets display only the name of the other call participant in calls (on ringing and on connected state) and not both the name and the number.

  The name has preference over the number on the Ascom handset display.

- **Call Reject**

  Ascom IP-DECT system can be configured to send SIP *"486 Busy Here"* or SIP *"603 Decline"* message. For the OSV system in the test environment, the setting *"Treat rejected calls as"* is set to *"Busy"* (refer to sub-section 4.1.5).

  If the Ascom handset receives "603 Decline", then it displays *"Hung up"*. On the other hand, if the Ascom handset receives "486 Busy Here", then it displays *"Busy"*.

- **Call Forwarding**

  Call Forwarding scenarios were tested with both local Ascom DECT handset and OSV system features.

  When executing the forwarding scenarios, make sure that the handset and PBX forwarding feature access codes do not have overlapping digits, because the scenarios will not work correctly.

- **Conference**

  Ascom handset device doesn’t display *"Conference"* call, but R3 is displayed instead.

1.4.2 *Restrictions*

- **Call History**

  Incoming anonymous calls are not displayed in Ascom DECT handset call list.

  If a call from an Ascom DECT handset user is forwarded, then at the call list of the Ascom handset there will be no "forwarded call" information.

- **SIP Session Refresh**

  Ascom IP-DECT Base Station can negotiate that IP-DECT (as UAC) sends SIP Session Refresh messages. However, IP-DECT will never take the initiative re session timers. It is up to the PBX to negotiate this.

  In current testing activities session refresh is performed via OSV with re-INVITE messages.

- **Do Not Disturb**

  For OSV SIP stations Do Not Disturb is supported from local Ascom DECT handset feature only, and not via PBX service code (system feature).

- **Call Hold**

  When an Ascom handset user puts a call on hold, then there is no *"held call"* visual indication on the handset.

  On the other hand, when the call is held by the other call party, then there is a *"held remotely"* visual indication on the Ascom device.

- **Call Park**

  Call Park feature isn't supported with Ascom IP-DECT devices.

- **Call Pick up**

  After the Ascom device picks up a call for another call pick up group member, it does not display any info regarding the group the call picked up for and displays only the connected party.

- **Call Transfer (Semi-Attended)**

  Due to DECT limitations, in semi-attended call transfer scenarios (A->B->C), where the Ascom handset user B is the *"transferee"*, after the transfer action is completed, the *"transferred to"* party C continues to display B on ringing state. After the transferred call is answered, C party displays correctly the *"transferred"* party A.

- **Call Deflect**

  Ascom DECT handsets do not support the feature call deflect.

When an incoming call from an Ascom DECT handset user is deflected, then there isn't *"forwarded from"* display information on the handset.

- **Call Forwarding**

  In call forwarding no reply scenarios (A->B->C), where the Ascom handset is the *"forwarded"* party (A) then on ringing phase it displays the *"forwarding"* party (B), instead of the *"forwarded to"* party (C). A party will display C party after the call is answered.

  See **"Call Deflect"** bullet point for the *"forwarded from"* display indication on DECT handset.

- **Voicemail**

  Ascom handset device doesn't display the number of unread voicemail messages. If an unread message exists and a new voice message arrives then there is no new notification.

- **Codecs**

  Regarding audio codecs support, OSV doesn't use G.722.2 (AMR-WB) codec. On the other hand, Ascom doesn't support G.722 codec.

  If the Ascom IP DECT System receives an INVITE with not supported codec(s) in SDP, only after answering the call, the IP-DECT System replies the call with "503 Service Unavailable - No circuit/channel available". Usually this return code is sent before answering the call on the phone. The phone should not ring before the call is rejected (refer to test case 5-08).

- **TLS/SRTP Calls**

  Ascom handsets don't display any secure call visual indication. The system administrator can see that the call is secure via the web page of the IP-DECT base station. Specifically, under the menu *"Traffic"*, next to the state of the call, a closed padlock icon is shown.

1.4.3 *Known Issues*

None.

1.5 Configuration Block Diagram

The diagram of **figure 1** below displays the logical diagram which is used for the certification project testing.

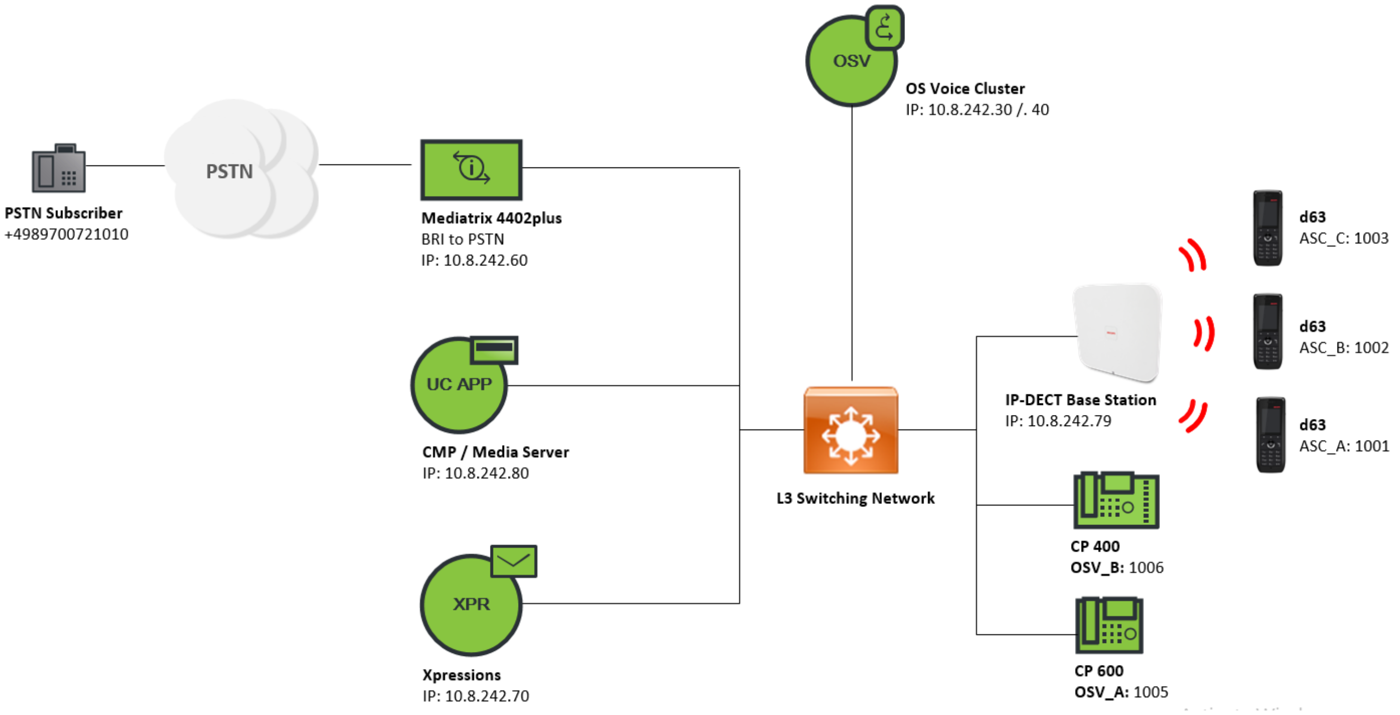


Figure 1: Network Topology Diagram

2 Basic Configuration

2.1 Ascom IP-DECT Base Station and DECT handsets

The Ascom IP-DECT Base Station and Ascom DECT handsets configuration was performed by Ascom personnel for the certification test according to OSV system settings in the certification lab.

The IP address of the Base Station (10.8.242.79) belong to the same subnet as the other Atos-Unify components and the communication is established via corporate LAN.

The following OS Voice SIP extensions are utilized for Ascom DECT handset SIP stations:

ASC_A = 1001 (d63-Talker handset)

ASC_B = 1002 (d63-Talker handset)

ASC_C = 1003 (d63-Talker handset)

2.2 Atos-Unify Component Infrastructure

OS Voice is configured for extension dialing between the users. The Office code used is +30 (210) 200 and the user extensions are using 4 digits (10xx).

OS Voice is provided with access to PSTN (OTE ISDN BRI provider) via a Mediatrix 4402plus BRI gateway (10.8.242.60). OS Voice subscribers dial 9 + PSTN number to reach a public number when making a business call. The digit 9 may be considered a "seizure code" digit which enables call routing to Mediatrix 4402plus and then is stripped away by OS Voice configuration before sending the number to PSTN.

Voicemail services are provided to OS Voice users by an OpenScape Xpressions server (10.8.242.70) connected to OS Voice. OS Voice subscribers may call 302102003001 (direct access number) to access their mailboxes.

- OS Voice IP addresses

  10.8.242.30 node 1

  10.8.242.40 node 2

  10.8.242.36 SIPSM

- OS UC (Media Server, CMP-Management)

  10.8.242.80

- SIP station (CP600 phone device)

  OSV_A = (+30 210 200) 1005

- SIP station (CP400 phone device)

  OSV_B = (+30 210 200) 1006

- OS Xpressions

  10.8.242.70

- PSTN subscribers

  +4989700721010 (landline)

- ISDN BRI for incoming calls from PSTN (MSN)

  +302106203360

3 Test Results in Detail

The testing procedure is grouped in 7 major testing areas; Registration, Basic Calls, Basic Calls Extended, Telephony Features, Audio Features, Redundancy and Encryption.

The *"Registration"* covers Ascom DECT handset registration to OSV with TCP / UDP / TLS transport and with or without digest authentication. In all other testing areas, Ascom IP-DECT handset devices were registered with TCP to OSV.

"Basic Calls" refer to incoming and outgoing call scenarios with local or PSTN subscribers. Additionally, call to a busy subscriber, rejected call, call no reply, call cancelled and call to unavailable Ascom handset user scenarios are, also, checked.

On the other hand, *"Basic Calls Extended"* tests are a special category of *"Basic Calls"* tests where calls from journal (call list), long duration calls, calls to a DND user and calls with caller ID suppression were considered.

The *"Telephony Features"* tests, cover certain IP telephony features like:

- Hold and Retrieve
- Call Consultation / Toggle
- Call Waiting
- Callback No Reply and On Busy
- Call Park
- Simultaneous Ringing
- Serial Ringing
- Call Pickup
- Call Transfer Attended, Semi-Attended and Blind
- Call Deflect
- Call Forwarding Unconditional, Busy and No Reply
- Hunt Group (MLHG)
- Voicemail (DTMF verification) / Message Waiting Indication
- Conference (PBX / local phone)

With the *"Audio Features"* testing area, the codec support is verified along with the station communication capability with the usage of unsupported codecs from the one communication party. Moreover, the system behavior was observed in codec mismatch scenarios. Additionally, DTMF digit recognition is also verified for calls from Ascom IP-DECT users to Xpressions voicemail system.

The *"Redundancy"* scenarios aim to verify the solution behavior in the case of active Base Station failure and the behavior after the failed component is back in service. However, in current certification activities Ascom Base Station *"Redundancy"* isn't verified because the test environment was setup with a single IP-DECT base station system. On the other hand, OSV node active-standby failover is tested.

Finally, the "*Encryption"* tests, focus on the security functionality and cover registration and basic calls using TLS/SRTP.

3.1 Registration

3.1.1 OSV Registration

No.	Test Procedure	Expected Result	Result	Comments
1-01	• Register Ascom IP-DECT phone to OSV without Digest Authentication.	• DECT phone successfully registers with OSV.	OK	
1-02	• Register Ascom IP-DECT phone to OSV with Digest Authentication.	• DECT phone successfully registers with OSV.	OK	
1-03	• Check that Ascom IP-DECT phone may register to OSV with either TCP or UDP.	• DECT phone successfully registers with OSV.	OK	
1-04	• Registration expiration timer at Ascom IP-DECT phone is set below OSV's corresponding value.	• OSV responds with a 423 Interval Too Brief to negotiate the lower re-Registration interval. • Successful registration to OSV.	OK	Value used on Ascom IP-DECT system is 60secs.

3.2 Basic Calls

3.2.1 Incoming / Outgoing Call

No.	Test Procedure	Expected Result	Result	Comments
2-01	• ASC_A calls ASC_B. • ASC_B hangs up at the end of the communication.	• Post Dial Delay (PDD). • Ringback tone heard by calling party. • Number presentation on ringing and after call pick up. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	OK	
2-02	• OSV_A calls ASC_A. • OSV_A hangs up at the end of the communication.	• Post Dial Delay (PDD). • Ringback tone heard by calling party. • Number presentation on ringing and after call pick up. • No media clipping when connecting both ends.	OK	

No.	Test Procedure	Expected Result	Result	Comments
		• Speech path in both directions. • Both ends idle after call clearing.		
2-03	• OSV_A calls ASC_A. • ASC_A hangs up at the end of the communication.	• Post Dial Delay (PDD). • Ringback tone heard by calling party. • Number presentation on ringing and after call pick up. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	OK	
2-04	• ASC_A calls OSV_A. • ASC_A hangs up at the end of the communication.	• Post Dial Delay (PDD). • Ringback tone heard by calling party. • Number presentation on ringing and after call pick up. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	OK	
2-05	• ASC_A calls OSV_A. • OSV_A hangs up at the end of the communication.	• Post Dial Delay (PDD). • Ringback tone heard by calling party. • Number presentation on ringing and after call pick up. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	OK	
2-06	• PSTN calls ASC_A. • PSTN hangs up at the end of the communication.	• Post Dial Delay (PDD). • Ringback tone heard by calling party. • Number presentation on ringing and after call pick up. • No media clipping when connecting both ends. • Speech path in both	OK	The display on Ascom device is "External Call" and the number.

No.	Test Procedure	Expected Result	Result	Comments
		directions. • Both ends idle after call clearing.		
2-07	• PSTN calls ASC_A. • ASC_A hangs up at the end of the communication.	• Post Dial Delay (PDD). • Ringback tone heard by calling party. • Number presentation on ringing and after call pick up. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	OK	
2-08	• ASC_A calls PSTN. • ASC_A hangs up at the end of the communication.	• Post Dial Delay (PDD). • Ringback tone heard by calling party. • Number presentation on ringing and after call pick up. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	OK	
2-09	• ASC_A calls PSTN. • PSTN hangs up at the end of the communication.	• Post Dial Delay (PDD). • Ringback tone heard by calling party. • Number presentation on ringing and after call pick up. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	OK	

3.2.2 *Busy Call*

No.	Test Procedure	Expected Result	Result	Comments
2-10	• ASC_A calls ASC_B. • ASC_B is busy.	• Busy tone/display on calling party. • The call is properly cleared.	OK	IP-DECT base station responds with a "486 Busy Here" SIP message.

No.	Test Procedure	Expected Result	Result	Comments
2-11	• OSV_A calls ASC_A. • ASC_A is busy.	• Busy tone/display on calling party. • The call is properly cleared.	OK	
2-12	• ASC_A calls OSV_A. • OSV_A is busy.	• Busy tone/display on calling party. • The call is properly cleared.	OK	OSV responds with a "486 Busy Here" SIP message.
2-13	• PSTN calls ASC_A. • ASC_A is busy.	• Busy tone/display on calling party. • The call is properly cleared.	OK	
2-14	• ASC_A calls PSTN. • PSTN is busy.	• Busy tone/display on calling party. • The call is properly cleared.	OK	

3.2.3 *Rejected Call*

No.	Test Procedure	Expected Result	Result	Comments
2-15	• ASC_A calls ASC_B. • ASC_B rejects incoming call.	• Busy tone/display on calling party. • The call is properly cleared.	OK	IP-DECT base station responds with a "486 Busy Here" SIP message. Call clears at about 30 secs. IP-DECT base station can be configured to send "603 Decline". With 486 Ascom handset displays busy, whereas with 603 the handset displays hung up.
2-16	• OSV_A calls ASC_A. • ASC_A rejects incoming call.	• Busy tone/display on calling party. • The call is properly cleared.	OK	
2-17	• ASC_A calls OSV_A. • OSV_A rejects incoming call.	• Busy tone/display on calling party. • The call is properly cleared.	OK	OSV responds with a SIP "603 Decline" message.
2-18	• PSTN calls ASC_A. • ASC_A rejects incoming call.	• Busy tone/display on calling party. • The call is properly cleared.	OK	
2-19	• ASC_A calls PSTN. • PSTN rejects incoming call.	• Busy tone/display on calling party. • The call is properly cleared.	OK	

3.2.4 *Not Answered Call*

No.	Test Procedure	Expected Result	Result	Comments
2-20	• ASC_A calls ASC_B. • ASC_B doesn't answer. Let the call ring until disconnection.	• Calling party gets announcement or specific tone. • Check timer after which the call gets disconnected.	OK	IP-DECT base station responds with a "504 Server timeout". The call is terminated after approximately 3 min.
2-21	• OSV_A calls ASC_A. • ASC_A doesn't answer. Let the call ring until disconnection.	• Calling party gets announcement or specific tone. • Check timer after which the call gets disconnected.	OK	
2-22	• ASC_A calls OSV_A. • OSV_A doesn't answer. Let the call ring until disconnection.	• Calling party gets announcement or specific tone. • Check timer after which the call gets disconnected.	OK	OSV sends a SIP "487 Request Terminated" message. The call is terminated after approximately 3 min.
2-23	• PSTN calls ASC_A. • ASC_A doesn't answer. Let the call ring until disconnection.	• Calling party gets announcement or specific tone. • Check timer after which the call gets disconnected.	OK	
2-24	• ASC_A calls PSTN. • PSTN doesn't answer. Let the call ring until disconnection.	• Calling party gets announcement or specific tone. • Check timer after which the call gets disconnected.	OK	

3.2.5 *Call Cancellation*

No.	Test Procedure	Expected Result	Result	A
2-25	• ASC_A calls ASC_B and hangs up before destination picks up.	• Call is terminated on both sides after hanging up.	OK	Ascom base station sends a SIP "CANCEL" message to OSV.
2-26	• OSV_A calls ASC_A and hangs up before destination picks up.	• Call is terminated on both sides after hanging up.	OK	
2-27	• ASC_A calls OSV_A and hangs up before destination picks up.	• Call is terminated on both sides after hanging up.	OK	

No.	Test Procedure	Expected Result	Result	A
2-28	• PSTN calls ASC_A and hangs up before destination picks up.	• Call is terminated on both sides after hanging up.	OK	
2-29	• ASC_A calls PSTN and hangs up before destination picks up.	• Call is terminated on both sides after hanging up.	OK	

3.2.6 *Call to Unavailable*

No.	Test Procedure	Expected Result	Result	Comments
2-30	• ASC_B is unregistered. • ASC_A calls ASC_B.	• Busy tone/display on calling party. • The call is properly cleared.	OK	The ASC_B is switched off. The message is heard immediately "The user is not available, try later" and a busy tone is heard. A "487 Request Terminated" SIP message is sent.
2-31	• ASC_A is unregistered. • OSV_A calls ASC_A.	• Busy tone/display on calling party. • The call is properly cleared.	OK	
2-32	• OSV_A is unregistered. • ASC_A calls OSV_A.	• Busy tone/display on calling party. • The call is properly cleared.	OK	A "487 Request Terminated" SIP message is sent.
2-33	• ASC_B is out of coverage (w/ batt. removed). • ASC_A calls ASC_B.	• Busy tone/display on calling party. • The call is properly cleared.	OK	ASC_B = out of coverage (remove battery while registered), IP-DECT sends "503 Service Unavailable", OSV sends "486 Busy Here", OSV sends a "487 Request Terminated" SIP message; handset displays BUSY.
2-34	• ASC_A is out of coverage (w/ batt. removed). • OSV_A calls ASC_A.	• Busy tone/display on calling party. • The call is properly cleared.	OK	

3.3 Basic Calls Extended

3.3.1 Call History

No.	Test Procedure	Expected Result	Result	Comments
3-01	• Check at ASC_A's phone device call history, the incoming, the outgoing and the missed calls entries.	• Call history properly lists incoming, the outgoing and the missed calls (internal and external).	OK	Anonymous incoming calls are not displayed in Ascom DECT handset call list.
3-02	• ASC_A initiates an outgoing call to ASC_B from call history.	• Post Dial Delay (PDD) • Ringback tone heard by calling party. • Number presentation on ringing and after call pick up. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	OK	
3-03	• ASC_A initiates an outgoing call to OSV_A from call history.	• Post Dial Delay (PDD) • Ringback tone heard by calling party. • Number presentation on ringing and after call pick up. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	OK	
3-04	• ASC_A initiates an outgoing call to PSTN from call history.	• Post Dial Delay (PDD) • Ringback tone heard by calling party. • Number presentation on ringing and after call pick up. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	OK	

3.3.2 *Long Duration Call*

No.	Test Procedure	Expected Result	Result	Comments
3-05	• Setup a call between ASC_A and ASC_B (with AMR-WB). • The call must last at least 1 hour. • ASC_B puts the call on mute when call is established. • After 5 mins ASC_B unmutes call and then ASC_A mutes for another 5 mins.	• Check that call lasts for at least 1h. • When MUTE function is activated during at least 5 minutes, check that calls are still active when MUTE function is deactivated on endpoints. • Check SIP Session Timer Expiry.	OK	SIP re-INVITE messages are being sent every 15 minutes from OSV to check status of the call. All messages are responded correctly by IP-DECT Base station.
3-06	• Setup a call between ASC_A and OSV_A. • The call must last at least 1 hour. • OSV_A puts the call on mute when call is established. • After 5 mins OSV_A unmutes call and then ASC_A mutes for another 5 mins.	• Check that call lasts for at least 1h. • When MUTE function is activated during at least 5 minutes, check that calls are still active when MUTE function is deactivated on endpoints. • Check SIP Session Timer Expiry.	OK	

3.3.3 *Do-Not-Disturb*

No.	Test Procedure	Expected Result	Result	Comments
3-07	• ASC_B has DND activated via menu options or via DND feature code. • ASC_A calls ASC_B.	• On phone display a DND logo is displayed. • Calling party gets busy tone.	OK	Locally via calling *42# (deactivate #42#), handset displays "302102001001 Busy", caller gets busy tone (IP-DECT base station sends "486 Busy Here").
3-08	• ASC_A has DND activated via menu options or via DND feature code. • OSV_A calls ASC_A.	• On phone display a DND logo is displayed. • Calling party gets busy tone.	OK	
3-09	• After scenario 3-08, ASC_A deactivates DND. • OSV_A calls ASC_A again.	• DND logo disappears from display. • Incoming calls are normally signaled on the phone.	OK	

3.3.4 *Name and Number Presentation (CLIP)*

No.	Test Procedure	Expected Result	Result	Comments
3-10	• Initiate a call from ASC_A with name configured to OSV_A.	• Number and name are displayed on devices.	OK	The number on Ascom devices isn't displayed, just the name.
3-11	• Initiate a call from ASC_A without name configured to OSV_A.	• Number and NOT the name are displayed on devices.	OK	BG group name is displayed instead.
3-12	• Initiate a call from OSV_A with name configured to ASC_A.	• Number and name are displayed on device.	OK	
3-13	• Initiate a call from OSV_A without name configured to ASC_A.	• Number and NOT the name are displayed on devices.	OK	

3.3.5 *Calling Identity Suppression (CID)*

No.	Test Procedure	Expected Result	Result	Comments
3-14	• ASC_A activates Calling Identity Suppression via the CID Suppression feature access code. • ASC_A calls ASC_B.	• Number and name are not displayed on called device.	OK	ASC_B displays "***********" on ringing state and when the call is established. No call is shown in call list.
3-15	• ASC_A activates Calling Identity Suppression via the CID Suppression feature access code. • ASC_A calls OSV_A.	• Number and name are not displayed on called device.	OK	
3-16	• ASC_A activates Calling Identity Suppression via the CID Suppression feature access code. • ASC_A calls PSTN.	• Number is not displayed on called phone.	OK	
3-17	• OSV_A activates Calling Identity Suppression via the CID Suppression feature access code. • OSV_A calls ASC_A.	• Number and name are not displayed on called phone.	OK	
3-18	• PSTN activates identity restriction. • PSTN calls ASC_A.	• Number is not displayed on called phone.	OK	

3.4 Telephony Features

3.4.1 Hold, Consultation, Toggle, Call Waiting

No.	Test Procedure	Expected Result	Result	Comments
4-01	• ASC_A calls ASC_B and puts the destination on hold.	• MOH is played to held party. • Held call display on both parties. • Proper call resume. • No Media delay establishment after resuming the call (media clipping).	POK	R is pressed to put "on hold" the called user. ASC_A has no "held call" display.
4-02	• OSV_A calls ASC_A and puts the destination on hold.	• MOH is played to held party. • Held call display on both parties. • Proper call resume. • No Media delay establishment after resuming the call (media clipping).	OK	
4-03	• ASC_A calls OSV_A and puts the destination on hold.	• MOH is played to held party. • Held call display on both parties. • Proper call resume. • No Media delay establishment after resuming the call (media clipping).	POK	Same as 4-01.
4-04	• ASC_A calls ASC_B. • ASC_B makes a consultation call to ASC_C. • ASC_B toggles between calls with ASC_A and ASC_C.	• Ring back tone heard by calling party. • Number presentation on ringing and after call pick up. • No media clipping when connecting both ends. • MOH is played to held party. • Speech path in both directions. • Both ends idle after call clearing.	POK	Press R to consult to the third user, R2 to toggle between calls and R1 to end the active call and continue with the held one. Same as 4-01.
4-05	• OSV_A calls ASC_A. • ASC_A makes a consultation call to ASC_B. • ASC_A toggles between calls with OSV_A and ASC_B.	• Ring back tone heard by calling party. • Number presentation on ringing and after call pick up. • No media clipping when connecting both ends. • MOH is played to held party. • Speech path in both	POK	Same as 4-04.

No.	Test Procedure	Expected Result	Result	Comments
		directions. • Both ends idle after call clearing.		
4-06	• ASC_A calls OSV_A. • OSV_A makes a consultation call to OSV_B. • OSV_A toggles between calls with ASC_A and OSV_B.	• Ring back tone heard by calling party. • Number presentation on ringing and after call pick up. • No media clipping when connecting both ends. • MOH is played to held party. • Speech path in both directions. • Both ends idle after call clearing.	OK	
4-07	• PSTN calls ASC_A. • ASC_A makes a consultation call to ASC_B. • ASC_A toggles between calls with PSTN and ASC_A.	• Ring back tone heard by calling party. • Number presentation on ringing and after call pick up. • No media clipping when connecting both ends. • MOH is played to held party. • Speech path in both directions. • Both ends idle after call clearing.	POK	
4-08	• OSV_A calls ASC_A. • OSV_B calls the same, ASC_A. • Call waiting is being signaled on ASC_A. • ASC_A accepts the second incoming call and toggles between calls with OSV_A and OSV_B.	• Ring back tone heard by calling party. • Number presentation on ringing and after call pick up. • No media clipping when connecting both ends. • MOH is played to held party. • Speech path in both directions. • Both ends idle after call clearing.	POK	Enable call waiting on Ascom handset with *43#. Use of R2 to accept the second call and at the same time putting on hold the first call. For toggling between calls again R2 is used. R1 is used to end the active call and continue with the held one. Deactivation of the feature #43#. Same as 4-01 for "held call" indication.

3.4.2 *Callback (No Reply)*

No.	Test Procedure	Expected Result	Result	Comments
4-09	• ASC_A has the callback feature activated at OSV. • ASC_A calls ASC_B, but OSV_B doesn't answer. • ASC_A activates callback via phone menu or by dialing the feature access code. • ASC_B establishes a call with another subscriber and then hangs up. • When ASC_B becomes available, OSV calls ASC_A and once ASC_A answers, OSV calls ASC_B. • ASC_B answers callback call and connects with ASC_A.	• CCNR feature is successfully activated. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	OK	ASC_A handset displays "Callback" and the number of ASC_B.
4-10	• OSV_A has the callback feature activated at OSV. • OSV_A calls ASC_A, but ASC_A doesn't answer. • OSV_A activates callback via phone menu or by dialing the feature access code. • ASC_A establishes a call with another subscriber and then hangs up. • When ASC_A becomes available, OSV calls OSV_A and once OSV_A answers, OSV calls ASC_A. • ASC_A answers callback call and connects with OSV_A.	• CCNR feature is successfully activated. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	OK	
4-11	• ASC_A has the callback feature activated at OSV. • ASC_A calls OSV_A, but OSV_A doesn't answer. • ASC_A activates callback via phone menu or by dialing the feature access code. • OSV_A establishes a call with another subscriber and then hangs up. • When OSV_A becomes available, after a while, OSV calls ASC_A and once ASC_A answers, OSV calls OSV_A.	• CCNR feature is successfully activated. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	OK	Same as 4-09.

No.	Test Procedure	Expected Result	Result	Comments
	• OSV_A answers callback call and connects with ASC_A.			
4-12	• Repeat scenarios 4-10 & 4-11 but deactivate the callback feature via phone menu or by dialing callback deactivation code.	• OSV provides acknowledgement that the callback feature has been cancelled. • After the CCNR request has been deactivated, OSV should not send any notification that the original callback target has become available to receive calls.	OK	

3.4.3 *Callback (On Busy)*

No.	Test Procedure	Expected Result	Result	Comments
4-13	• ASC_A has the feature CCBS activated at OSV. • ASC_A calls ASC_B, but ASC_B is busy. • ASC_A activates CCBS via phone menu or by dialing access code. • When ASC_B becomes available, after a while, OSV calls ASC_A and once ASC_A answers, OSV calls ASC_B. • ASC_B answers callback call and connects with ASC_A.	• CCBS feature is successfully activated. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	OK	Same as 4-09.
4-14	• OSV_A has the feature CCBS activated at OSV. • OSV_A calls ASC_A, but ASC_A is busy. • OSV_A activates CCBS via phone menu or by dialing access code. • When ASC_A becomes available, after a while, OSV calls OSV_A and once OSV_A answers, OSV calls ASC_A. • ASC_A answers callback call and connects with OSV_A.	• CCBS feature is successfully activated. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	OK	
4-15	• ASC_A has the feature CCBS activated at OSV. • ASC_A calls OSV_A, but OSV_A is busy.	• CCBS feature is successfully activated. • No media clipping when connecting both ends.	OK	

No.	Test Procedure	Expected Result	Result	Comments
	• ASC_A activates CCBS via phone menu or by dialing the feature access code. • When OSV_A becomes available, after a while, OSV calls ASC_A and once ASC_A answers, OSV calls OSV_A. • OSV_A answers callback call and connects with ASC_A.	• Speech path in both directions. • Both ends idle after call clearing.		
4-16	• Repeat scenarios 4-14 & 4-15 but deactivate the callback feature via phone menu or by dialing callback deactivation code while the busy subscriber is still busy.	• OSV provides acknowledgement that the callback feature has been cancelled. • After the CCBS request has been deactivated, OSV should not send any notification that the original callback target has become available to receive calls.	OK	

3.4.4 *Call Park*

No.	Test Procedure	Expected Result	Result	Comments
4-17	• ASC_A calls ASC_B. • ASC_B parks the call. • ASC_C dials the parking lot number to retrieve the call. • ASC_A hangs up.	• MOH played on ASC_A while the call is parked. • OSV_A successfully retrieves parked call. • Number presentation on ringing and after call pick up. • Media clipping and speech path after forwarding. • Call is disconnected after hang up.	NA	Call Park feature isn't supported with Ascom IP-DECT devices.
4-18	• ASC_A calls OSV_A. • ASC_A parks the call. • ASC_C retrieves the call from parking lot. • OSV_A hangs up.	• MOH played on OSV_A while the call is parked. • OSV_B successfully retrieves parked call. • Number presentation on ringing and after call pick up. • Media clipping and speech path after forwarding. • Call is disconnected after hang up.	NA	

No.	Test Procedure	Expected Result	Result	Comments
4-19	• OSV_A calls ASC_A. • ASC_A parks the call. • ASC_A does not retrieve the call for more than 60 seconds. • ASC_A gets an incoming call to retrieve the parked call and SC_A answers. • OSV_A hangs up.	• MOH played on OSV_A while the call is parked. • ASC_A successfully retrieves parked call. • Number presentation on ringing and after call pick up. • Media clipping and speech path after forwarding. • Call is disconnected after hang up.	NA	
4-20	• OSV_A calls ASC_A. • OSV_A parks the call. • OSV_B dials the parking lot number to retrieve the call. • ASC_A hangs up.	• MOH played on ASC_A while the call is parked. • OSV_B successfully retrieves parked call. • Number presentation on ringing and after call pick up. • Media clipping and speech path after forwarding. • Call is disconnected after hang up.	OK	

3.4.5 *Simultaneous Ringing*

No.	Test Procedure	Expected Result	Result	Comments
4-21	• ASC_A and ASC_B belong to the same simultaneous ringing group. • OSV_A calls ASC_A. • ASC_B answers the call. • OSV_A hangs up at the end of the communication.	• All phones from the group are ringing and every ringing phone can answer the call. • The phone number of calling and called phone is displayed. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	OK	ASC_B shows that the call was for the ASC_A.
4-22	• OSV_A and ASC_A belong to the same simultaneous ringing group. • PSTN calls OSV_A. • ASC_A answers the call. • ASC_A hangs up at the end of the communication.	• All phones from the group are ringing and every ringing phone can answer the call. • The phone number of calling and called phone is displayed. • No media clipping when connecting both ends. • Speech path in both directions.	OK	

No.	Test Procedure	Expected Result	Result	Comments
		• Both ends idle after call clearing.		
4-23	• From scenario 4-21, ASC_A switches off simultaneous ringing function with the feature access code. • OSV_A calls ASC_A. • ASC_A switches on simultaneous ringing function with the feature access code. • OSV_A calls ASC_A.	• Successful activation / deactivation of simultaneous ringing with feature access code.	OK	

3.4.6 *Serial Ringing*

No.	Test Procedure	Expected Result	Result	Comments
4-24	• ASC_A and ASC_B belong to the same serial ringing group. • OSV_A calls ASC_A. • ASC_A doesn't answer and ASC_B starts ringing. • ASC_B picks up the call. • OSV_A hangs up at the end of the communication.	• All phones from the group are ringing successive and every ringing phone can answer the call. • The phone number of calling and called phone is displayed. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	OK	Same as 4-21.
4-25	• OSV_A and ASC_A belong to the same serial ringing group. • PSTN calls OSV_A. • OSV_A doesn't answer and ASC_A starts ringing. • ASC_A picks up the call. • ASC_A hangs up at the end of the communication.	• All phones from the group are ringing successive and every ringing phone can answer the call. • The phone number of calling and called phone is displayed. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	OK	
4-26	• From scenario 4-24, ASC_A switches off serial ringing function with the feature access code. • OSV_A calls ASC_A. • ASC_A switches on simultaneous ringing function with the feature	• Successful activation / deactivation of serial ringing with feature access code.	OK	

No.	Test Procedure	Expected Result	Result	Comments
	access code. • OSV_A calls ASC_A.			

3.4.7 *Call Pickup*

No.	Test Procedure	Expected Result	Result	Comments
4-27	• OSV_B and ASC_B belong to the same Call Pickup group. • ASC_A calls OSV_B. • ASC_B picks up the call via phone menu or via call pickup access code while OSV_B is ringing. • ASC_B hangs up at the end of the communication.	• ASC_B gets a notification • ASC_B is able to pick up the call. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	POK	ASC_B receives Call Pickup Notification and is able to pick up the call via handset button. On connected phase ASC_B doesn't display any info for call group picked up call, but only the connected party.
4-28	• OSV_B and ASC_B belong to the same Call Pickup group. • OSV_A calls ASC_B. • OSV_B picks up the call via phone menu or via call pick up access code, while ASC_B is ringing. • OSV_B hangs up at the end of the communication.	• ASC_B gets a notification • OSV_B is able to pick up the call. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	OK	
4-29	• OSV_A and ASC_A belong to the same Call Pickup group. • PSTN calls OSV_A. • ASC_A picks up the call via phone menu or via call pick up access code while OSV_A is ringing. • ASC_A hangs up at the end of the communication.	• ASC_A is able to pick up the call. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	POK	Same as 4-27.
4-30	• OSV_A and ASC_A belong to the same Call Pickup group. • PSTN calls ASC_A. • OSV_A picks up the call via phone menu or via call pick up access code while ASC_A is ringing. • OSV_A hangs up at the end of the communication.	• OSV_A is able to pick up the call. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	OK	

3.4.8 *Call Transfer (Attended)*

No.	Test Procedure	Expected Result	Result	Comments
4-31	• ASC_A calls ASC_B. • ASC_B consults to ASC_C and waits for ASC_C to pick up the call. • ASC_B transfers ASC_A towards ASC_C. • After transfer is completed and communication is established between ASC_A and ASC_C, ASC_A hangs up.	• MOH is played to ASC_A while reaching ASC_C. • No Media clipping and proper speech path after transfer when ASC_A and ASC_C are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	ASC_B consults by pressing R and transfers the call by pressing R4.
4-32	• ASC_A calls OSV_A. • OSV_A consults to ASC_B and waits for ASC_B to pick up the call. • OSV_A transfers ASC_A towards ASC_B. • After transfer is completed and communication is established between ASC_A and ASC_B, ASC_A hangs up.	• MOH is played to ASC_A while reaching ASC_B. • No Media clipping and proper speech path after transfer when ASC_A and ASC_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	
4-33	• ASC_A calls OSV_A. • OSV_A consults to OSV_B and waits for OSV_B to pick up the call. • OSV_A transfers ASC_A towards OSV_B. • After transfer is completed and communication is established between ASC_A and OSV_B, ASC_A hangs up.	• MOH is played to ASC_A while reaching OSV_B. • No Media clipping and proper speech path after transfer when ASC_A and OSV_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	
4-34	• OSV_A calls ASC_A. • ASC_A puts OSV_A on hold, calls ASC_B and waits for ASC_B to pick up the call. • ASC_A resumes the call with OSV_A (ASC_B is placed on hold) and then transfers OSV_A towards ASC_B. • After transfer is completed and communication is established between OSV_A and ASC_B, OSV_A hangs up.	• MOH is played in all related steps (to OSV_A while reaching ASC_B / to ASC_B while resuming the call with OSV_A). • Proper call resume when ASC_A toggles between calls with OSV_A and ASC_B. • No Media clipping and proper speech path after transfer when OSV_A and ASC_B are connected. • Number presentation on ringing and after call pick up.	OK	

No.	Test Procedure	Expected Result	Result	Comments
		• Both ends idle after call clearing.		
4-35	• PSTN calls OSV_A. • OSV_A consults to ASC_A and waits for ASC_A to pick up the call. • OSV_A transfers PSTN towards ASC_A. • After transfer is completed and communication is established between PSTN and ASC_A, PSTN hangs up.	• MOH is played to PSTN while reaching ASC_B. • No Media clipping and proper speech path after transfer when ASC_B picks up the call. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	
4-36	• PSTN calls ASC_A. • ASC_A puts PSTN on hold, calls OSV_B and waits for OSV_B to pick up the call. • ASC_A resumes the call with PSTN (OSV_B is placed on hold) and then transfers PSTN towards OSV_B. • After transfer is completed and communication is established between PSTN and OSV_B, PSTN hangs up.	• MOH is played in all related steps (to PSTN while reaching OSV_B / to OSV_B while resuming the call with PSTN). • Proper call resume when ASC_A toggles between calls with PSTN and OSV_B. • No Media clipping and proper speech path after transfer when PSTN and OSV_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	

3.4.9 *Call Transfer (Semi-Attended)*

No.	Test Procedure	Expected Result	Result	Comments
4-37	• ASC_A calls ASC_B. • ASC_B consults to ASC_C. • ASC_B hangs up while ASC_C is ringing. • ASC_A receives ring back tone. • After communication is established between ASC_A and ASC_C, ASC_A hangs up.	• MOH is played to ASC_A after consultation and before ASC_B hangs up. • No Media clipping and proper speech path after transfer when ASC_A and ASC_C are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	POK	On ringing state ASC_C continues to display ASC_B and ASC_A is on hold status (no ringback tone). ASC_C is updated on the display with ASC_A after accepting the call. Due to attribute "Transfer on Hangup" is disabled in base station, user must press R4 while ASC_C is ringing for transfer to take place.

No.	Test Procedure	Expected Result	Result	Comments
4-38	• ASC_A calls OSV_A. • OSV_A consults to ASC_B. • OSV_A hangs up while ASC_B is ringing. • ASC_A receives ring back tone. • After communication is established between ASC_A and ASC_B, ASC_A hangs up.	• MOH is played to ASC_A after consultation and before OSV_A hangs up. • No Media clipping and proper speech path after transfer when ASC_A and ASC_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	
4-39	• ASC_A calls OSV_A. • OSV_A consults to OSV_B. • OSV_A hangs up while OSV_B is ringing. • ASC_A receives ring back tone. • After communication is established between ASC_A and OSV_B, ASC_A hangs up.	• MOH is played to ASC_A after consultation and before OSV_A hangs up. • No Media clipping and proper speech path after transfer when ASC_A and OSV_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	
4-40	• OSV_A calls ASC_A. • ASC_A consults to ASC_B. • ASC_A hangs up while ASC_B is ringing. • OSV_A receives ring back tone. • After communication is established between OSV_A and ASC_B, OSV_A hangs up.	• MOH is played to OSV_A after consultation and before ASC_A hangs up. • No Media clipping and proper speech path after transfer when OSV_A and ASC_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	POK	Same as 4-37.
4-41	• PSTN calls OSV_A. • OSV_A consults to ASC_A. • OSV_A hangs up while ASC_A is ringing. • PSTN receives ring back tone. • After communication is established between PSTN and ASC_A, PSTN hangs up.	• MOH is played to PSTN after consultation and before ASC_A hangs up. • No Media clipping and proper speech path after transfer when PSTN and ASC_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	

No.	Test Procedure	Expected Result	Result	Comments
4-42	• PSTN calls ASC_A. • ASC_A consults to OSV_B. • ASC_A hangs up while OSV_B is ringing. • PSTN receives ring back tone. • After communication is established between PSTN and OSV_B, PSTN hangs up.	• MOH is played to PSTN after consultation and before ASC_A hangs up. • No Media clipping and proper speech path after transfer when PSTN and OSV_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	POK	See 4-37.

3.4.10 *Call Transfer (Blind)*

No.	Test Procedure	Expected Result	Result	Comments
4-43	• ASC_A calls ASC_B. • ASC_B blind transfers to ASC_C. • ASC_A receives ring back tone. • After transfer is completed and communication is established between ASC_A and ASC_C, ASC_A hangs up.	• MOH to ASC_A after consultation and before blind transfer is completed. • No Media clipping and proper speech path after transfer when ASC_A and ASC_C are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	Press RR to blind transfer.
4-44	• ASC_A calls OSV_A. • OSV_A blind transfers to ASC_B. • ASC_A receives ring back tone. • After transfer is completed and communication is established between ASC_A and ASC_B, ASC_A hangs up.	• MOH to ASC_A after consultation and before blind transfer is completed. • No Media clipping and proper speech path after transfer when ASC_A and ASC_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	
4-45	• ASC_A calls OSV_A. • OSV_A blind transfers to OSV_B. • ASC_A receives ring back tone. • After transfer is completed and communication is established between ASC_A and OSV_B, ASC_A hangs up.	• MOH to ASC_A after consultation and before blind transfer is completed. • No Media clipping and proper speech path after transfer when ASC_A and OSV_B are connected.	OK	

No.	Test Procedure	Expected Result	Result	Comments
		• Number presentation on ringing and after call pick up. • Both ends idle after call clearing.		
4-46	• OSV_A calls ASC_A. • ASC_A blind transfers to ASC_B. • OSV_A receives ring back tone. • After transfer is completed and communication is established between OSV_A and ASC_B, OSV_A hangs up.	• MOH to OSV_A after consultation and before blind transfer is completed. • No Media clipping and proper speech path after transfer when OSV_A and ASC_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	
4-47	• PSTN calls OSV_A. • OSV_A blind transfers to ASC_A. • PSTN receives ring back tone. • After transfer is completed and communication is established between PSTN and ASC_A, PSTN hangs up.	• MOH to PSTN after consultation and before blind transfer is completed. • No Media clipping and proper speech path after transfer when PSTN and ASC_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	
4-48	• PSTN calls ASC_A. • ASC_A blind transfers to OSV_B. • PSTN receives ring back tone. • After transfer is completed and communication is established between PSTN and OSV_B, PSTN hangs up.	• MOH to PSTN after consultation and before blind transfer is completed. • No Media clipping and proper speech path after transfer when PSTN and OSV_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	

3.4.11 *Call Deflect*

No.	Test Procedure	Expected Result	Result	Comments
4-49	• ASC_A calls ASC_B. • ASC_B doesn't answer and deflects the call to AS_C.	• Calling party receives forwarding call display notification.	NA	Feature not supported on Ascom DECT handsets.

No.	Test Procedure	Expected Result	Result	Comments
	• Leave ASC_C ring for a few seconds and pick up the call. • After deflection is completed and communication is established between ASC_A and ASC_C, ASC_A hangs up.	• Ring back tone heard by calling party. • Number presentation on ringing and after call pick up. • Media clipping and speech path after forwarding. • Both ends idle after call clearing.		
4-50	• ASC_A calls OSV_A. • OSV_A doesn't answer and deflects the call to ASC_B. • Leave ASC_B ring for a few seconds and pick up the call. • After deflect is completed and communication is established between ASC_A and ASC_B, ASC_A hangs up.	• Calling party receives forwarding call display notification. • Ring back tone heard by calling party. • Number presentation on ringing and after call pick up. • Media clipping and speech path after forwarding. • Both ends idle after call clearing.	POK	ASC_B displays that the call is forwarded from OSV_A. ASC_A doesn't display "forwarded from" information.

3.4.12 *Call Forwarding (Unconditional)*

No.	Test Procedure	Expected Result	Result	Comments
4-51	• ASC_A calls ASC_B who has CFU activated via phone locally or via the system. • ASC_B forwards the call immediately to ASC_C. • Leave ASC_C ring for a few seconds and pick up the call. • After forward is completed and communication is established between ASC_A and ASC_C, ASC_A hangs up.	• Call forwarding info is displayed on the phone. • Calling party receives forwarding call display notification. • Ring back tone heard by calling party. • Number presentation on ringing and after call pick up. • Media clipping and speech path after forwarding. • Both ends idle after call clearing.	POK	ASC_A doesn't display "forwarded from" information.
4-52	• OSV_A calls ASC_A who has CFU activated via phone locally or via the system. • ASC_A forwards the call immediately to OSV_B. • Leave OSV_B ring for a few seconds and pick up the call. • After forward is completed and communication is established	• Call forwarding info is displayed on the phone. • Calling party receives forwarding call display notification. • Ring back tone heard by calling party. • Number presentation on ringing and after call pick up.	OK	

No.	Test Procedure	Expected Result	Result	Comments
	between OSV_A and OSV_B, OSV_A hangs up.	• Media clipping and speech path after forwarding. • Both ends idle after call clearing.		
4-53	• ASC_A calls OSV_A who has CFU activated via phone locally or via the system. • OSV_A forwards the call immediately to OSV_B. • Leave OSV_B ring for a few seconds and pick up the call. • After forward is completed and communication is established between ASC_A and OSV_B, ASC_A hangs up.	• Call forwarding info is displayed on the phone. • Calling party receives forwarding call display notification. • Ring back tone heard by calling party. • Number presentation on ringing and after call pick up. • Media clipping and speech path after forwarding. • Both ends idle after call clearing.	POK	Same as 4-51.
4-54	• PSTN calls ASC_A who has CFU activated via phone locally or via the system. • ASC_A forwards the call immediately to ASC_B. • Leave ASC_B ring for a few seconds and pick up the call. • After forward is completed and communication is established between PSTN and ASC_B, PSTN hangs up.	• Call forwarding info is displayed on the phone. • Calling party receives forwarding call display notification. • Ring back tone heard by calling party. • Number presentation on ringing and after call pick up. • Media clipping and speech path after forwarding. • Both ends idle after call clearing.	OK	

3.4.13 *Call Forwarding (Busy)*

No.	Test Procedure	Expected Result	Result	Comments
4-55	• ASC_A calls busy subscriber ASC_B who has CFB activated via phone locally or via the system. • ASC_B doesn't have call waiting activated and forwards the call immediately to ASC_C. • Leave ASC_C ring for a few seconds and pick up the call. • After forward is completed and communication is established	• Call forwarding info is displayed on the phone. • Calling party receives forwarding call display notification. • Ring back tone heard by calling party. • Number presentation on ringing and after call pick up. • Media clipping and speech path after forwarding.	POK	Same as 4-51.

No.	Test Procedure	Expected Result	Result	Comments
	between ASC_A and ASC_C, ASC_A hangs up.	• Both ends idle after call clearing.		
4-56	• OSV_A calls busy subscriber ASC_A who has CFB activated via phone locally or via the system. • ASC_A doesn't have call waiting activated and forwards the call immediately to OSV_B. • Leave OSV_B ring for a few seconds and pick up the call. • After forward is completed and communication is established between OSV_A and OSV_B, OSV_A hangs up.	• Call forwarding info is displayed on the phone. • Calling party receives forwarding call display notification. • Ring back tone heard by calling party. • Number presentation on ringing and after call pick up. • Media clipping and speech path after forwarding. • Both ends idle after call clearing.	OK	
4-57	• ASC_A calls busy subscriber OSV_A who has CFB activated via phone locally or via the system. • OSV_A doesn't have call waiting activated and forwards the call immediately to OSV_B. • Leave OSV_B ring for a few seconds and pick up the call. • After forward is completed and communication is established between ASC_A and OSV_B, ASC_A hangs up.	• Call forwarding info is displayed on the phone. • Calling party receives forwarding call display notification. • Ring back tone heard by calling party. • Number presentation on ringing and after call pick up. • Media clipping and speech path after forwarding. • Both ends idle after call clearing.	POK	Same as 4-51.
4-58	• PSTN calls busy subscriber ASC_A who has CFB activated via phone locally or via the system. • ASC_A doesn't have call waiting activated and forwards the call immediately to ASC_B. • Leave ASC_B ring for a few seconds and pick up the call. • After forward is completed and communication is established between PSTN and ASC_B, PSTN hangs up..	• Call forwarding info is displayed on the phone. • Calling party receives forwarding call display notification. • Ring back tone heard by calling party. • Number presentation on ringing and after call pick up. • Media clipping and speech path after forwarding. • Both ends idle after call clearing.	OK	

3.4.14 *Call Forwarding (No Reply)*

No.	Test Procedure	Expected Result	Result	Comments
4-59	• ASC_A calls ASC_B who has CFNR activated via phone locally or via the system. • ASC_B rings and after a while forwards to ASC_C. • Leave ASC_C ring for a few seconds and pick up the call. • After forward is completed and communication is established between ASC_A and ASC_B, ASC_A hangs up.	• Call forwarding info is displayed on the phone. • Calling party receives forwarding call display notification. • Ring back tone heard by calling party. • Number presentation on ringing and after call pick up. • Media clipping and speech path after forwarding. • Both ends idle after call clearing.	POK	After the call is forwarded, ASC_A displays ASC_B instead of ASC_C. When the call is answered ASC_A correctly shows ASC_C. Regarding "forwarded from" visual indication, see 4-51.
4-60	• OSV_A calls ASC_A who has CFNR activated via phone locally or via the system. • ASC_A rings and after a while forwards to OSV_B. • Leave OSV_B ring for a few seconds and pick up the call. • After forward is completed and communication is established between OSV_A and OSV_B, OSV_A hangs up.	• Call forwarding info is displayed on the phone. • Calling party receives forwarding call display notification. • Ring back tone heard by calling party. • Number presentation on ringing and after call pick up. • Media clipping and speech path after forwarding. • Both ends idle after call clearing.	OK	
4-61	• ASC_A calls OSV_A who has CFNR activated via phone locally or via the system. • OSV_A rings and after a while forwards to OSV_B. • Leave OSV_B ring for a few seconds and pick up the call. • After forward is completed and communication is established between ASC_A and OSV_B, ASC_A hangs up.	• Call forwarding info is displayed on the phone. • Calling party receives forwarding call display notification. • Ring back tone heard by calling party. • Number presentation on ringing and after call pick up. • Media clipping and speech path after forwarding. • Both ends idle after call clearing.	POK	Same as 4-59.
4-62	• PSTN calls ASC_A who has CFNR activated via phone locally or via the system. • ASC_A rings and after a while forwards to ASC_B.	• Call forwarding info is displayed on the phone. • Calling party receives forwarding call display notification.	OK	

No.	Test Procedure	Expected Result	Result	Comments
	• Leave ASC_B ring for a few seconds and pick up the call. • After forward is completed and communication is established between PSTN and ASC_B, PSTN hangs up.	• Ring back tone heard by calling party. • Number presentation on ringing and after call pick up. • Media clipping and speech path after forwarding. • Both ends idle after call clearing.		

3.4.15 *Hunt Group*

No.	Test Procedure	Expected Result	Result	Comments
4-63	• Hunt Group is configured with ASC_A, ASC_B and ASC_C. • OSV_A calls hunt group number. • ASC_A rings and answers. • Call is cleared at the end of the communication.	• Post Dial Delay (PDD). • Ring back tone heard by calling party. • Number presentation on ringing and after call pick up. • Media clipping and speech path after forwarding. • Both ends idle after call clearing.	OK	
4-64	• Hunt Group is configured with ASC_A, ASC_B and ASC_C. • OSV_A calls hunt group number. • ASC_A rings, but no answer. • ASC_B rings, but no answer. • ASC_C rings and answers. • Call is cleared at the end of the communication.	• Post Dial Delay (PDD). • Ring back tone heard by calling party. • Number presentation on ringing and after call pick up. • Media clipping and speech path after forwarding. • Both ends idle after call clearing.	OK	
4-65	• Hunt Group is configured with ASC_A, ASC_B and ASC_C. • PSTN calls hunt group number. • ASC_A rings, but no answer. • ASC_B rings, but no answer. • ASC_C rings and answers. • Call is cleared at the end of the communication.	• Post Dial Delay (PDD). • Ring back tone heard by calling party. • Number presentation on ringing and after call pick up. • Media clipping and speech path after forwarding. • Both ends idle after call clearing.	OK	

3.4.16 Voicemail

No.	Test Procedure	Expected Result	Result	Comments
4-66	• OSV_A and OSV_B call ASC_A who has call forwarding to voicemail system (Xpressions). • OSV_A and OSV_B leave a voicemail message to ASC_A. • ASC_A calls voicemail system to hear unread messages. • ASC_A calls back OSV_A from Xpressions.	• Proper MWI indication with the correct number of messages on phone. • MWI is retained after IP-DECT base station restart. • Voicemail messages are properly heard. • Successful call establishment and proper speech path. • Both ends idle after call clearing.	POK	The number of the unread voice messages isn't displayed on the handset, but only that unread messages exist.
4-67	• After scenario 4-66, ASC_A deletes voicemail messages one by one (logout from Xpressions menu and the login again).	• Number of voicemail messages is correctly displayed after one message deletion. • MWI disappears from the phone after all voicemail messages are deleted.	POK	When all unread messages are deleted then the MWI display notification disappears from the handset. See, also, 4-66.
4-68	• ASC_A calls voicemail system to hear messages and perform various actions using the appropriate DTMF options.	• DTMF digits are properly recognized from voicemail system. • Login possible to personal voicemail box.	OK	

3.4.17 Conference

No.	Test Procedure	Expected Result	Result	Comments
4-69	• ASC_A calls OSV_A and establishes communication. • ASC_A initiates a consultation call to ASC_B. • After ASC_B answers, ASC_A initiates a local conference. • ASC_A (conference initiator) hangs up.	• Media clipping and proper speechpath after conferencing participants in the conference. • Number(s) presentation on conference participants. • Conference participants are still on the conference after ASC_A hangs up. • Number(s) presentation after ASC_A hangs up on conference participants.	OK	Press R3 to establish the conference. ASC_A displays R3.
4-70	• ASC_A calls PSTN and establishes communication.	• Media clipping and proper speechpath after conferencing	OK	Same as 4-69.

No.	Test Procedure	Expected Result	Result	Comments
	• ASC_A initiates a consultation call to ASC_B. • After ASC_B answers, ASC_A initiates a local conference. • ASC_A (conference initiator) hangs up.	participants in the conference. • Number(s) presentation on conference participants. • Conference participants are still on the conference after ASC_A hangs up. • Number(s) presentation after ASC_A hangs up on conference participants.		
4-71	• OSV_A calls ASC_A and establishes communication. • OSV_B calls ASC_A who has call waiting activated. • ASC_A merges calls with OSV_A and OSV_B to a local conference. • ASC_A (conference initiator) hangs up.	• Media clipping and proper speechpath after conferencing participants in the conference. • Number(s) presentation on conference participants. • Conference participants are still on the conference after ASC_A hangs up. • Number(s) presentation after ASC_A hangs up on conference participants.	OK	Same as 4-69.
4-72	• OSV_A calls ASC_A and establishes communication. • OSV_A initiates a consultation call to ASC_B. • After ASC_B answers, OSV_A initiates a system controlled conference. • OSV_A calls OSV_B and OSV_A adds OSV_B to the conference. • OSV_A (conference initiator) hangs up.	• Media clipping and proper speechpath after conferencing participants in the conference. • Number(s) presentation on conference participants. • Conference participants are still on the conference after OSV_A hangs up. • Number(s) presentation after ASC_A hangs up on conference participants.	OK	

3.5 Audio Features

3.5.1 Codecs

No.	Test Procedure	Expected Result	Result	Comments
5-01	• IP-DECT base station is configured to have G.711 as first priority. • ASC_A calls OSV_A.	• Connection is established with G.711. • Media clipping and speechpath.	OK	
5-02	• IP-DECT base station is configured to have G.722 as first priority. • ASC_A calls OSV_A.	• Connection is established with G.722. • Media clipping and speechpath.	NA	Ascom IP-DECT doesn't support G.722.
5-03	• IP-DECT base station is configured to have G.722.2 (AMR-WB), as first priority. • ASC_A calls ASC_B.	• Connection is established with G.722.2. • Media clipping and speechpath.	OK	
5-04	• IP-DECT base station is configured to have G.729, as first priority. • ASC_A calls OSV_A.	• Connection is established with G.729. • Media clipping and speechpath.	OK	
5-05	• ASC_A is configured to support G.711 only. • OSV_A calls ASC_A.	• Connection is established with G.711. • Media clipping and speechpath.	OK	
5-06	• ASC_A is configured to support G.722 only. • OSV_A calls ASC_A.	• Connection is established with G.722. • Media clipping and speechpath.	NA	Same as 5-02.
5-07	• ASC_A is configured to support G.729 only. • OSV_A calls ASC_A.	• Connection is established with G.729. • Media clipping and speechpath.	OK	
5-08	• Make sure there is codec mismatch between ASC_A and OSV_A . • OSV_A calls ASC_A.	• Call is disconnected. • Call failure indication.	POK	If the Ascom IP-DECT System receives an INVITE with not supported codec(s) in SDP, only after answering the call, the IP DECT System replies the call with "503 Service Unavailable". Usually this return code is sent before answering the call on the

No.	Test Procedure	Expected Result	Result	Comments
				phone. The phone should not ring before the call is rejected.
5-09	• Make sure there is codec mismatch between ASC_A and OSV_A . • ASC_A calls OSV_A.	• Call is disconnected. • Call failure indication.	OK	CP phone sends a "500 Internal Server Error" and OSV sends "487 Request Terminated".

3.5.2 *DTMF*

No.	Test Procedure	Expected Result	Result	Comments
5-10	• ASC_A calls OSV_A. • ASC_A sends DTMF 1234567890*# to OSV_A and then OSV_A does the same towards ASC_A.	• DTMF digits are received bothways.	OK	
5-11	• ASC_A calls PSTN. • ASC_A sends DTMF 1234567890*# to PSTN and then PSTN does the same towards ASC_A.	• DTMF digits are received bothways.	OK	
5-12	• ASC_A calls Xpressions and sends DTMF tones.	• DTMF digits are properly recognized by Xpressions.	OK	

3.6 Redundancy

3.6.1 *OSV Failure*

No.	Test Procedure	Expected Result	Result	Comments
6-01	• Node 1 is the active node. • Shutdown node1 or perform command srxctrl 3 4. • Start node 1 or perform command srxctrl 4 4.	• The switchover should not be seen on the IP-DECT handsets. • ASC_A is able to make and receive calls.	OK	
6-02	• The node1 is put back into service. • Perform an incoming and outgoing call to ASC_A.	• The switchover should not be seen on the IP-DECT handsets. • ASC_A is able to make and receive calls.	OK	

3.6.2 IP-DECT Base Station Switchover

No.	Test Procedure	Expected Result	Result	Comments
6-03	• ASC_A is in a call with ASC_B. • The master base station fails and active call is dropped. • After switchover to the second base station, ASC_A calls ASC_B again. • Call is cleared at the end of the communication.	• The redundant mirror Base Station becomes the Master and takes over the Handsets and the SIP-Registration. • Calls are possible after switchover.	NA	Single Base Station is used for testing activities.
6-04	• In mirror configuration the functions does not return to Master automatically. • After scenario 6-3, while ASC_A is in a call with ASC_B, switch back manually to the first base station via GUI. • After switchover to the first base station, ASC_A calls ASC_B. • Call is cleared at the end of the communication.	• The redundant mirror Base Station becomes the Master and takes over the Handsets and the SIP-Registration. • Calls are possible after switchover.	NA	Same as 6-03

3.7 Encryption

3.7.1 Registration - TLS /SRTP

No.	Test Procedure	Expected Result	Result	Comments
7-01	• Register ASC IP-DECT phone to OSV without Digest Authentication and TLS.	• ASC phone successfully registers with OSV.	OK	
7-02	• Register ASC phone to OSV with Digest Authentication and TLS.	• ASC phone successfully registers with OSV.	OK	

3.7.2 Basic Calls - TLS / SRTP

No.	Test Procedure	Expected Result	Result	Comments
7-03	• ASC_A and ASC_B are registered with OSV with TLS and are configured to use secure media (SRTP). • ASC_A calls ASC_B.	• Post Dial Delay (PDD). • Ringback tone heard by calling party. • Number presentation on ringing and after call pick up. • Secure call display on the	POK	There isn't any display of the secure call on the Ascom handsets. The administrator is able to see that the call is secure on the IP-DECT base station

No.	Test Procedure	Expected Result	Result	Comments
		phone devices • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.		web page under "Trafffic". On the "state" of the calls there is a closed padlock icon.
7-04	• OSV_A and ASC_A are registered with OSV with TLS and are configured to use secure media (SRTP). • OSV_A calls ASC_A.	• Post Dial Delay (PDD). • Ringback tone heard by calling party. • Number presentation on ringing and after call pick up. • Secure call display on the phone devices • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	POK	Same as 7-03.
7-05	• OSV_A and ASC_A are registered with OSV with TLS and are configured to use secure media (SRTP). • ASC_A calls OSV_A.	• Post Dial Delay (PDD). • Ringback tone heard by calling party. • Number presentation on ringing and after call pick up. • Secure call display on the phone devices • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	POK	Same as 7-03.

3.8 Summary

Test results aggregated:

Category	Planned	OK	POK	NOK	NA	NT
Registration	**4**	4	-	-	-	-
Basic Calls	**34**	34	-	-	-	-
Basic Calls Extended	**18**	18	-	-	-	-
Telephony Features	**72**	48	20	-	4	-
Audio Features	**12**	9	1	-	2	-
Redundancy	**4**	2	-	-	2	
Encryption	**5**	2	3	-	-	-
Sum	**149**	**117**	**24**	**-**	**8**	**-**

4 Configuration Used During Tests

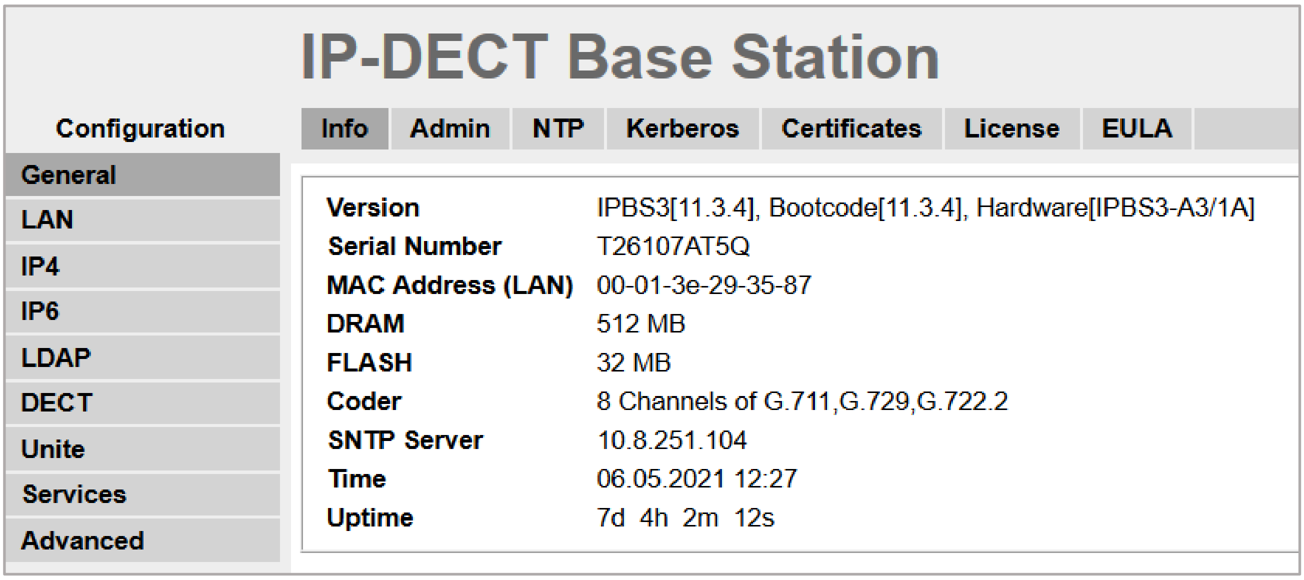
4.1 Ascom IP-DECT Base Stations Configuration

The current section summarizes the example configuration of the Ascom IP-DECT solution for the connection with OSV V10, which is used for the current testing activities. Default or non-project specific Ascom IP-DECT system configuration will not be referenced to in subsequent paragraphs.

To configure the IP-DECT Base Station, access a web browser and enter the IP address of the Base Station as the URL.

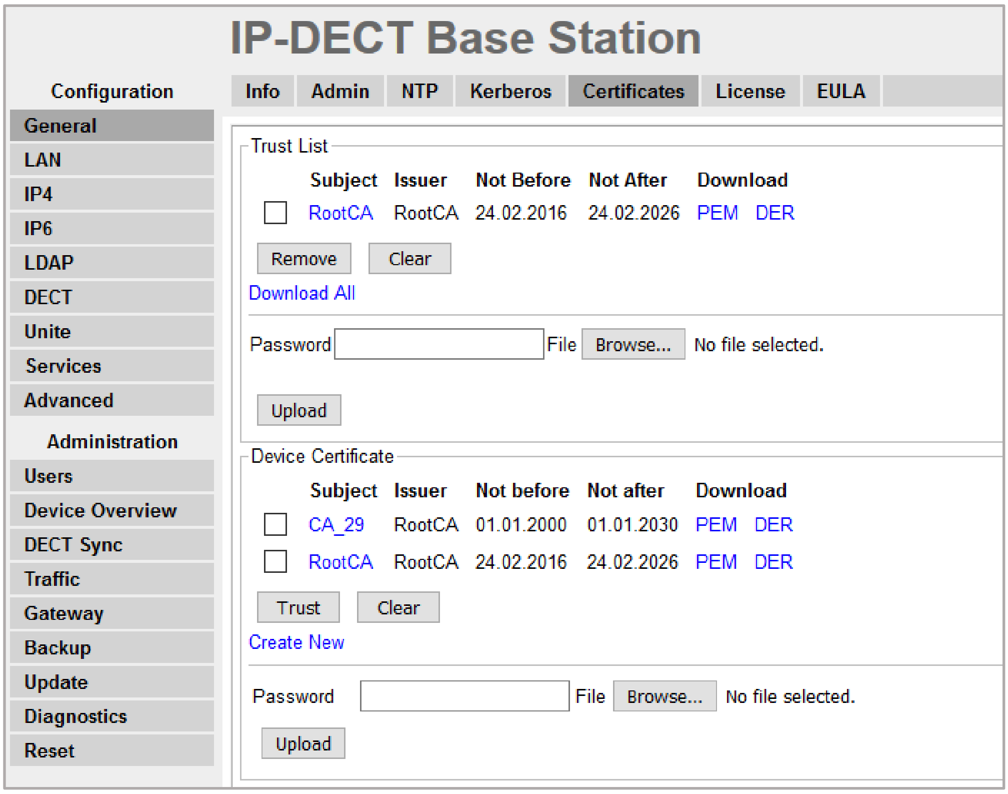
Click the System administration link and enter the appropriate credentials to access the Ascom IP-DECT Base Station and then click on **[OK]**.

4.1.1 *General Configuration of IP-DECT Base Station*



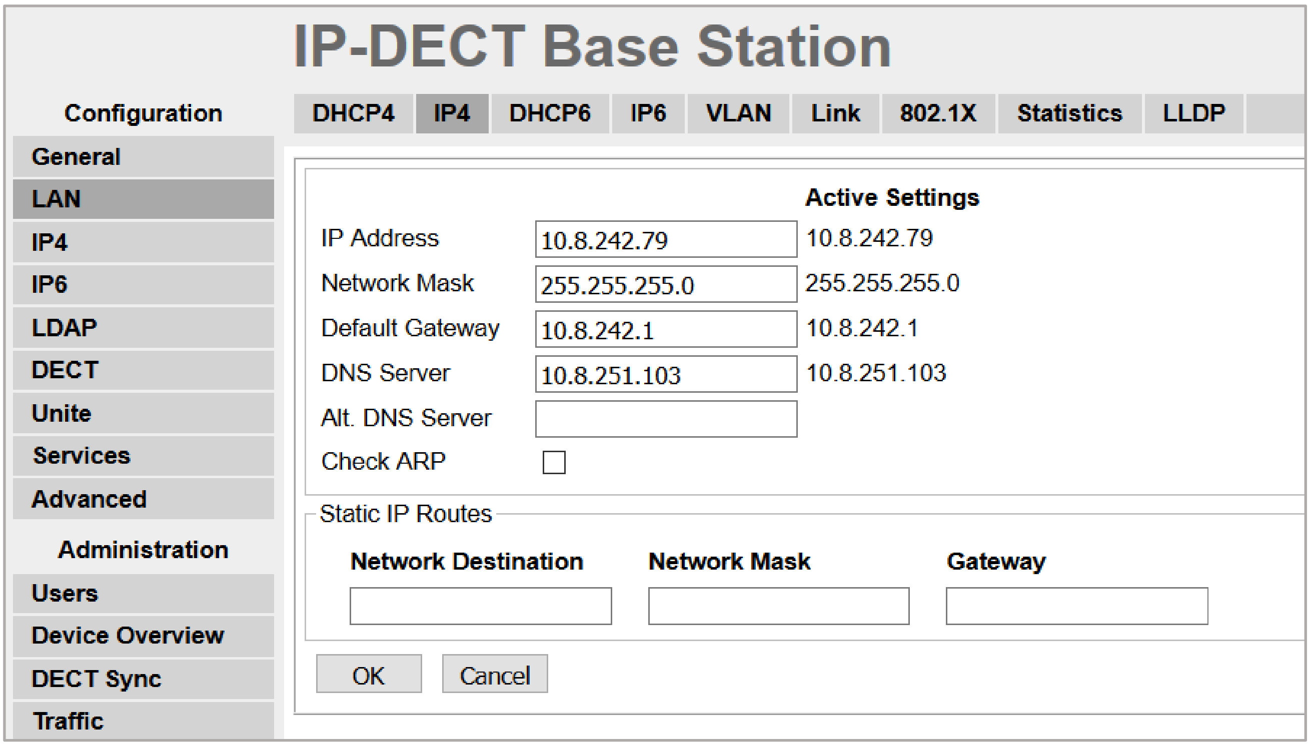
The configuration of NTP server (10.8.251.104) is strongly recommended.

For TLS registration of the handsets to OSV the root CA certificate needs to be uploaded to IP-DECT base Station. Export, the OSV root.pem certificate file from OSV SLES directory: /usr/local/ssl/certs.



Go to **"General"**, select the **"Certificates"** tab and upload the certificate by selecting it from the *"Browse"* button, on the *"Create New"* area.

4.1.2 *IP-DECT Base Station LAN IP*



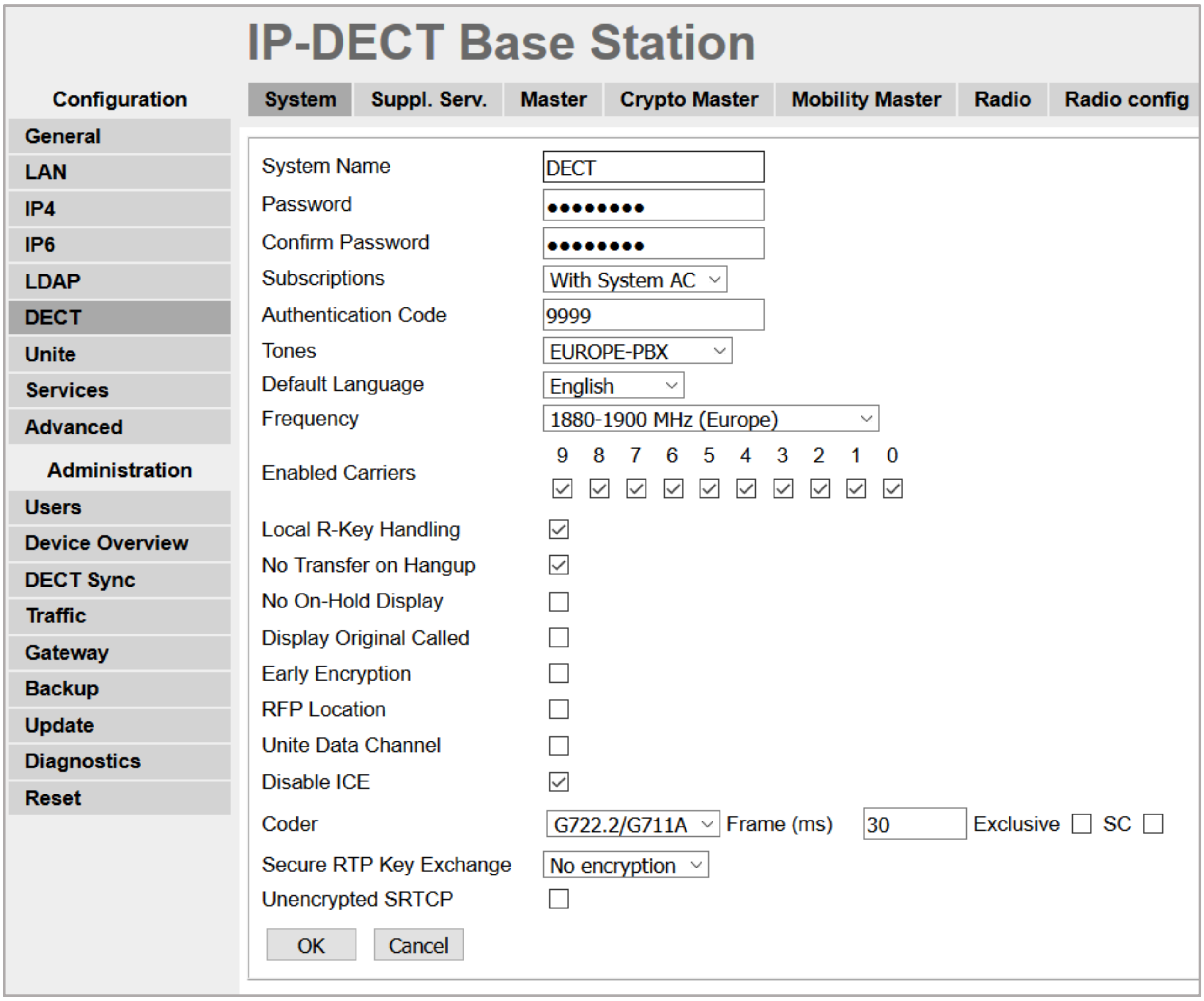
Navigate to **"LAN"** and select the **"IP"** tab and enter the following:

- **IP Address** 10.8.242.79
- **Network Mask** 255.255.255.0
- **Default Gateway** 10.8.242.1
- **DNS Server** 10.8.251.103

Click on the **[OK]** Button to save.

Note: DNS Server isn't used during testing activities because IPs instead FQDNs were utilized in the test environment setup.

4.1.3 *DECT System*

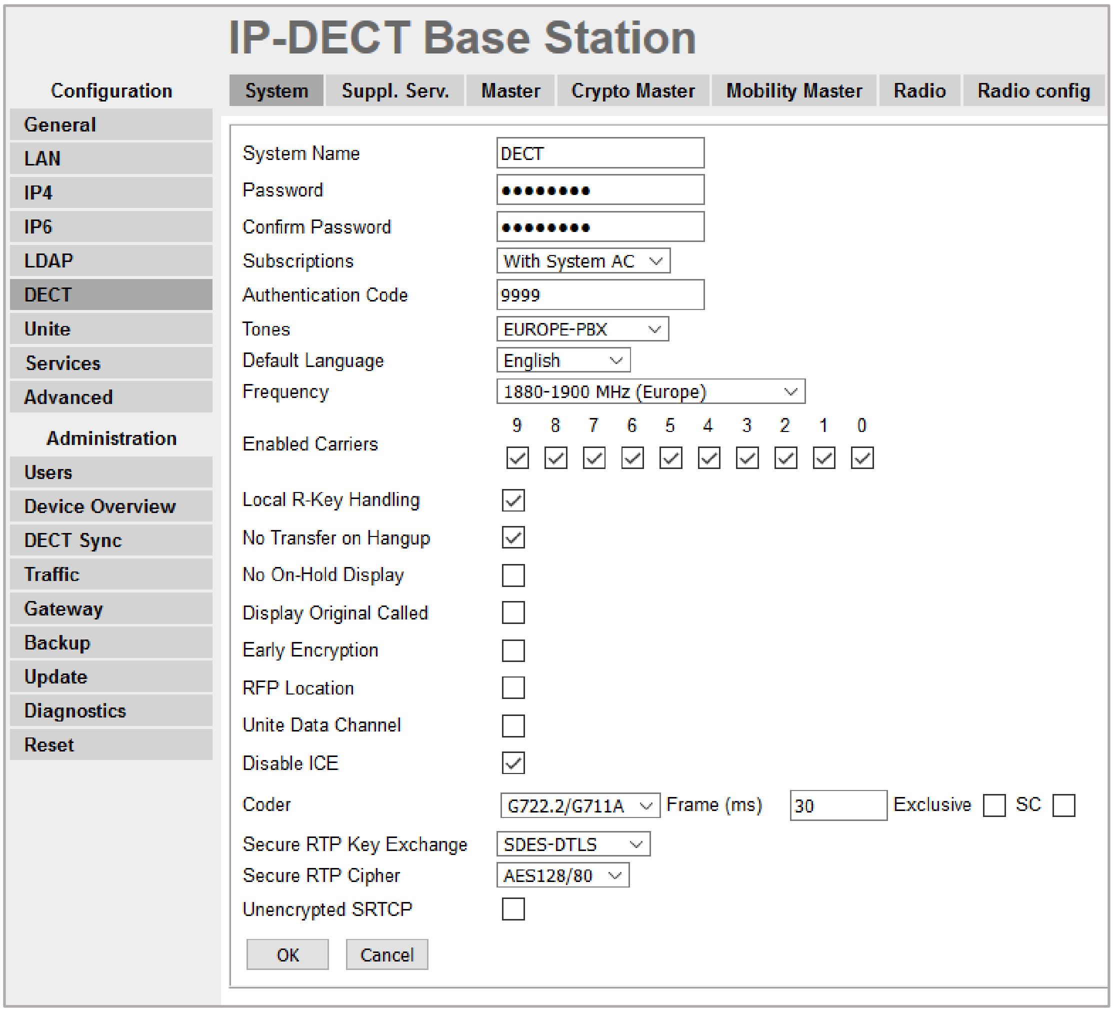


Click on the **"System"** tab and enter the following:

- **System Name** Enter the System Name as previously configured
- **Password** Enter the Password as previously configured
- **Confirm Password** Confirm the Password
- **Subscriptions** Select "With System AC" from the dropdown box
- **Authentication Code** Enter the appropriate DECT handset Login code
- **Tones** Select the location where the IP-DECT system is located (EUROPE-PBX)
- **Default Language** Select the required Language from the dropdown box (English)
- **Frequency** Select the required Frequency from the dropdown box (1800-1900 MHz (Europe))
- **Enabled** Select the number of Carriers required
- **Local R-Key Handling Box** Checked
- **No Transfer on Hangup** Checked (R4 key procedure is used on DECT handset, instead)
- **Disable ICE** Checked (used for NAT)
- **Coder** Select the required Coder from the Coder dropdown box (Coder = G722.2/G711A, Frame (ms) = 30)

Click the **[OK]** button to continue.

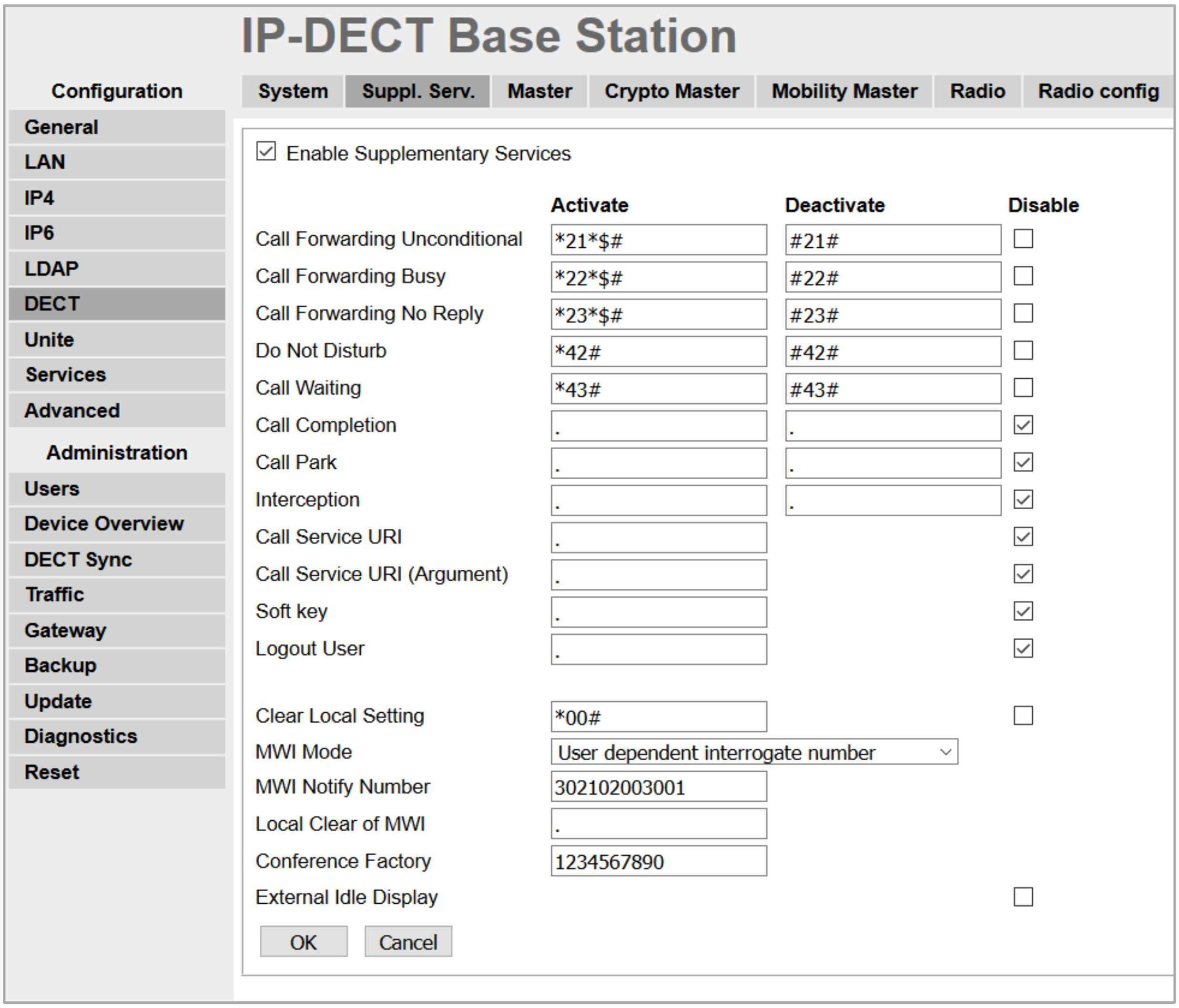
Note: It is recommended that the PBX (Codec Parameters) and the Base Station to have identical *"ptime"* values.



For secure media configuration, when TLS registration is required, on the **"System"** tab enter the following:

- **Secure RTP Key Exchange** Select "SDES-DTLS" from the dropdown box
- **Secure RTP Cipher** Select "AES128/80" from the dropdown box

4.1.4 *IP-DECT Supp. Serv.*



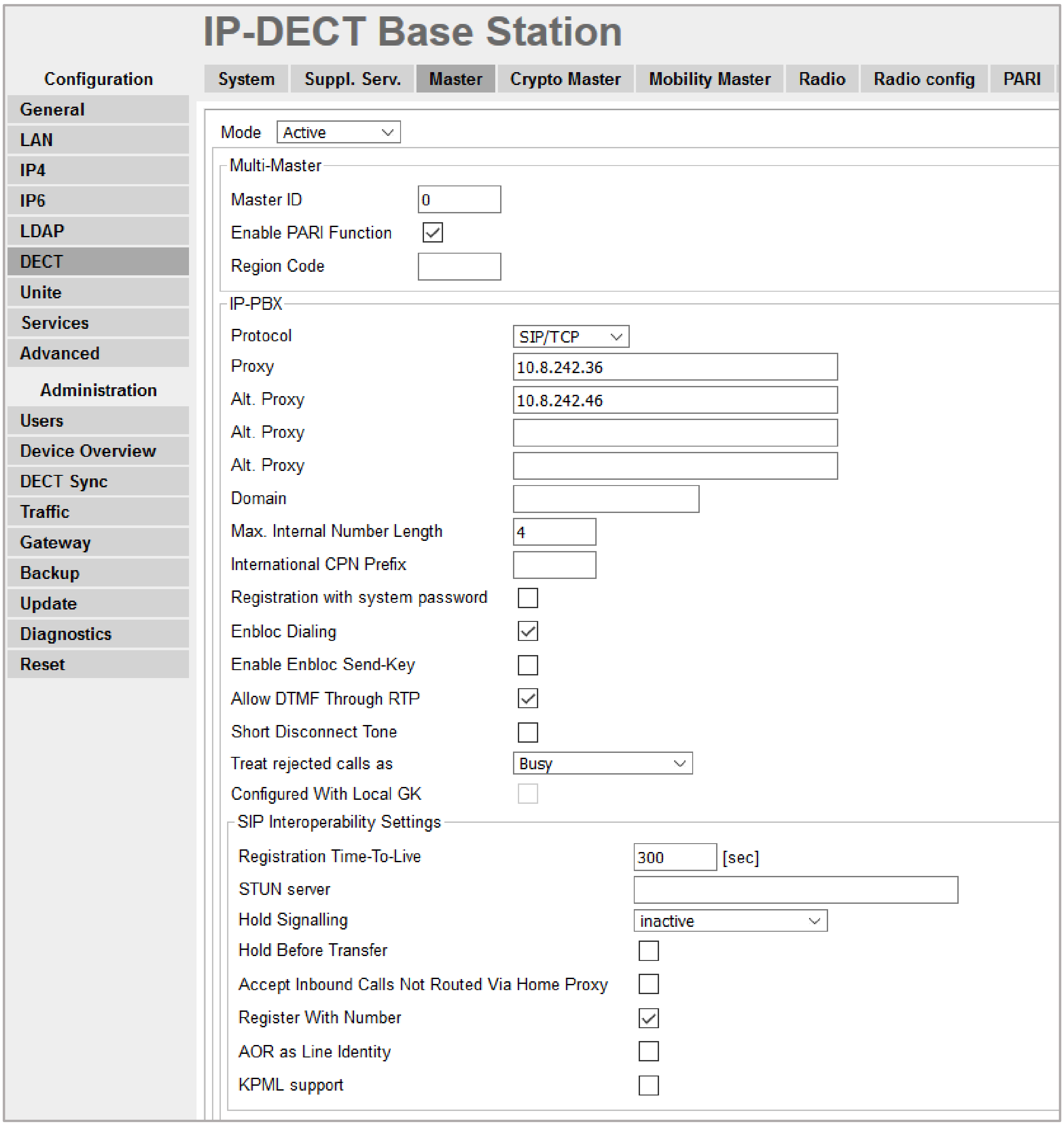
Click on the **"Suppl. Serv."** tab and enter the following:

- **Enable Supplementary Services check box** — Checked
- **Call Forwarding Unconditional** — Activate = *21*$#, Deactivate = #21#
- **Call Forwarding Busy** — Activate = *22*$#, Deactivate = #22#
- **Call Forwarding No Reply** — Activate = *23*$#, Deactivate = #23#
- **Do not Disturb** — Activate = *42*$#, Deactivate = #42#
- **Call waiting** — Activate = *43*$#, Deactivate = #43#
- **Clear Local Settings** — Activate = *00#
- **MWI Mode** — User dependent interrogate number
- **MWI Notify Number** — 3002102003001 (voicemail number as configured in OSV)

Disable the other handset local features / functions which are either not supported towards OSV or not included in the test plan.

Click the **[OK]** button to continue.

4.1.5 *IP-DECT Master*

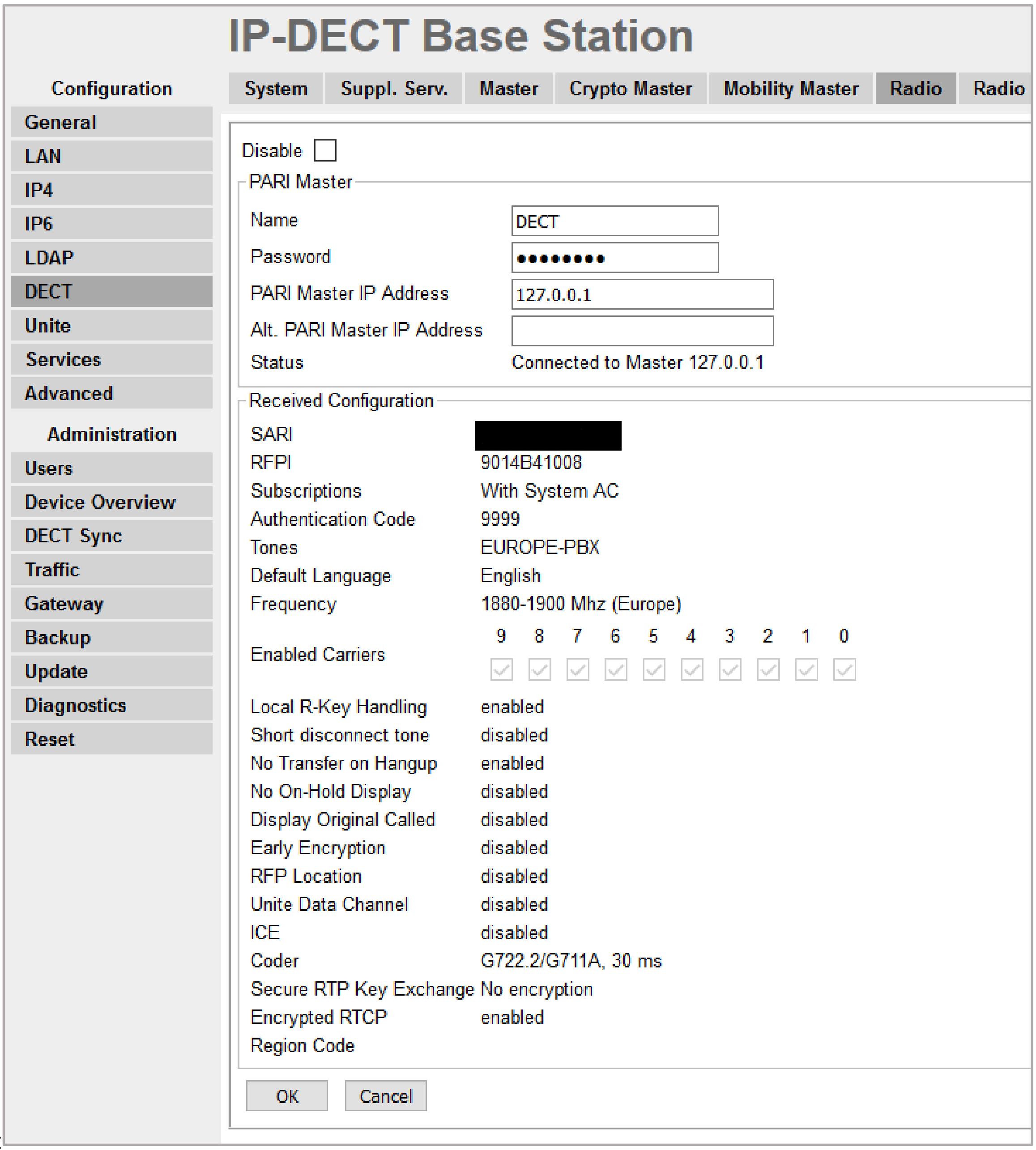


Navigate to **"DECT"**, click on **"Master"** and enter the following:

- **Mode** — Select Active from the dropdown box
- **Enable PARI Function check box** — Checked
- **Protocol** — Select SIP/TCP from the dropdown box or SIP/TLS in case of TLS registration required
- **Proxy** — 10.8.242.36 (OSV SIPSM)
- **Enbloc Dialing check box** — Checked
- **Allow DTMF through RTP check box** — Checked
- **Registration Time-To-Live** — 300

Click the **[OK]** button to continue (not shown).

4.1.6 *DECT Radio*



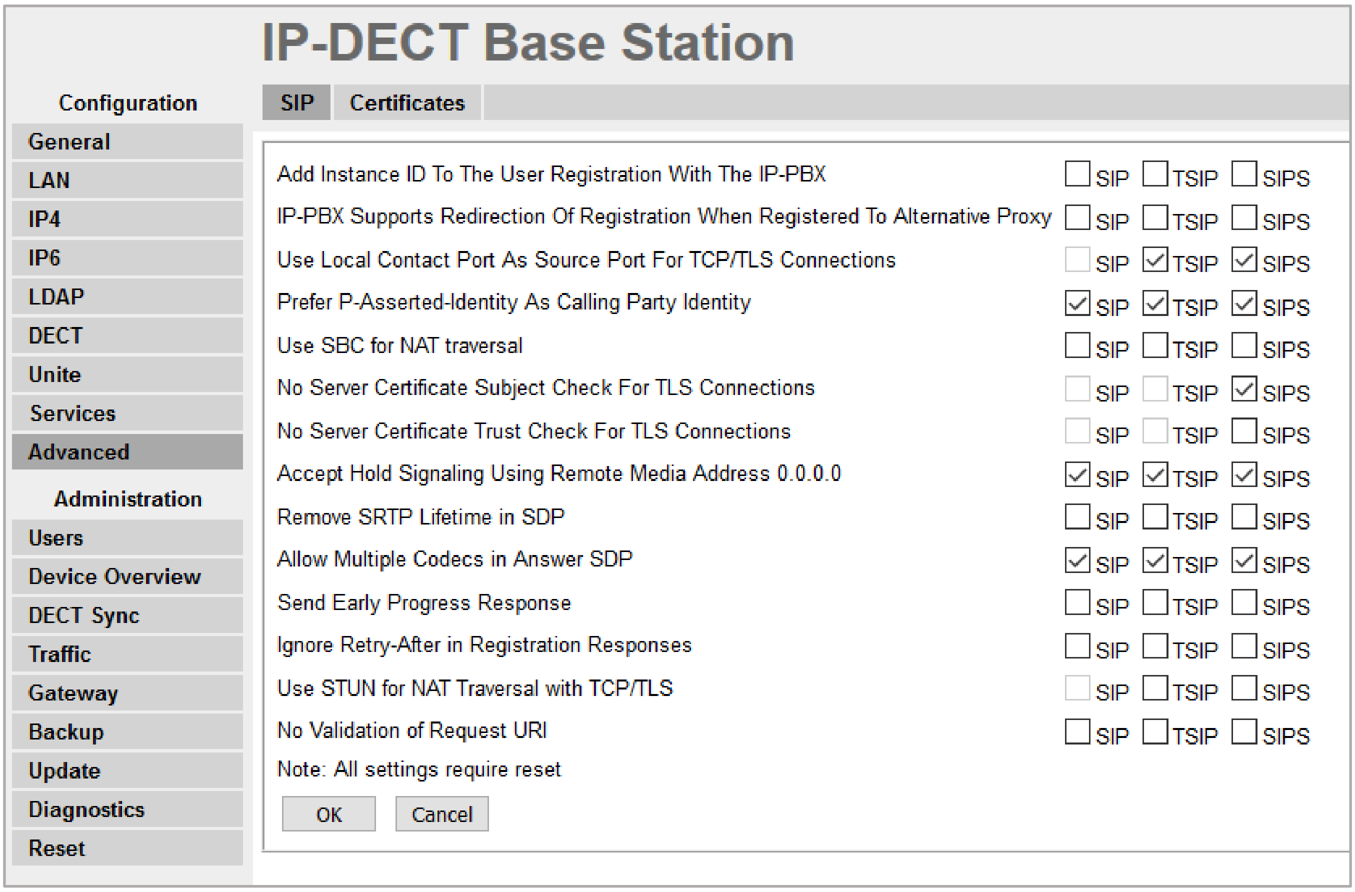
t

Navigate to **"Radio"** and select the Admin tab and enter the following:

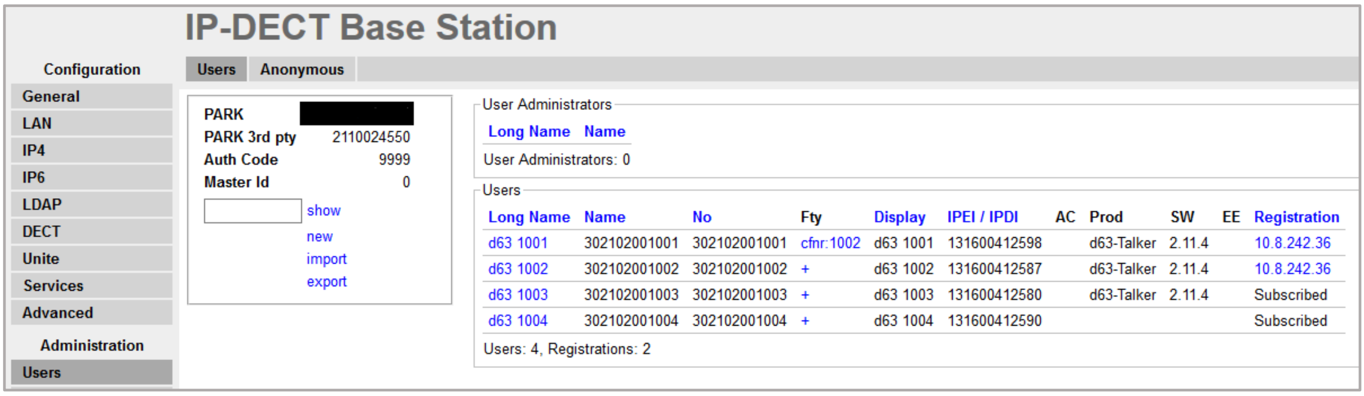
- **Device Name** Enter the name for the PARI Master in the ***"Name"*** text field
- **Password** Enter the password for the PARI Master in the *"Password"* text field
- **PARI Master IP Address** 127.0.0.1 (single base station)

Click on the **[OK]** Button to save.

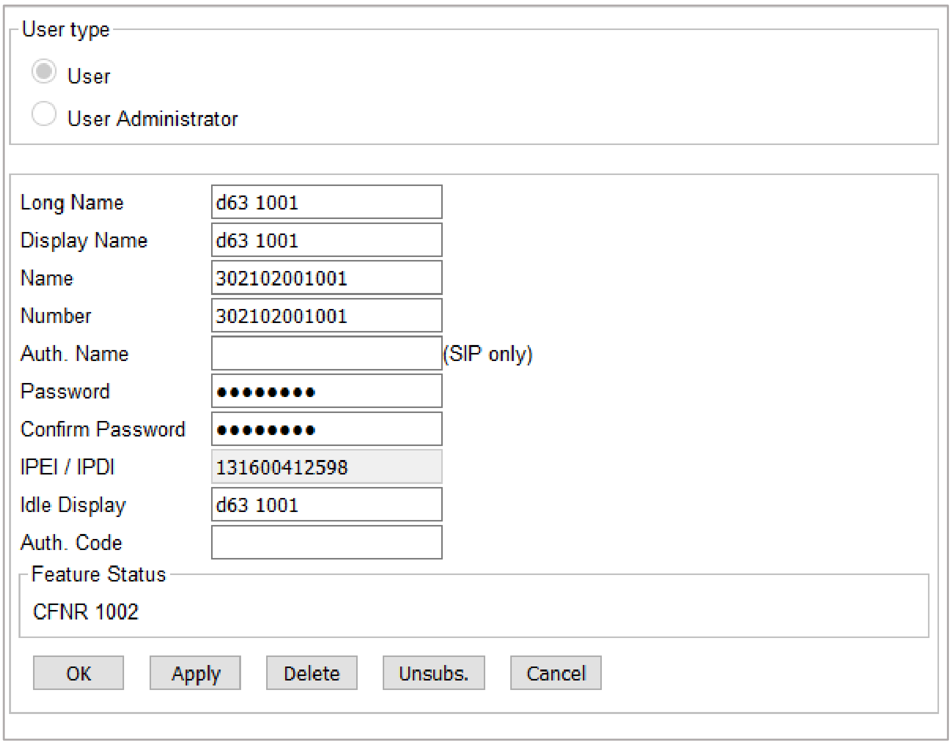
4.1.7 *Advanced SIP*



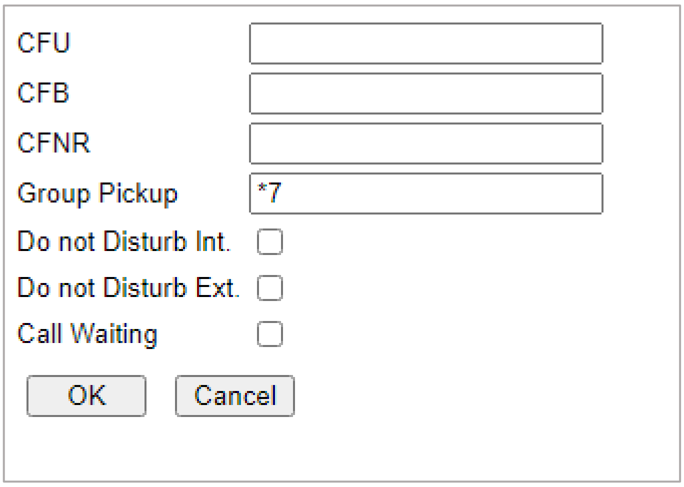
4.1.8 *DECT Radio*



Click on e.g. user **"d63 1001"** to view digest authentication and local phone feature status data.

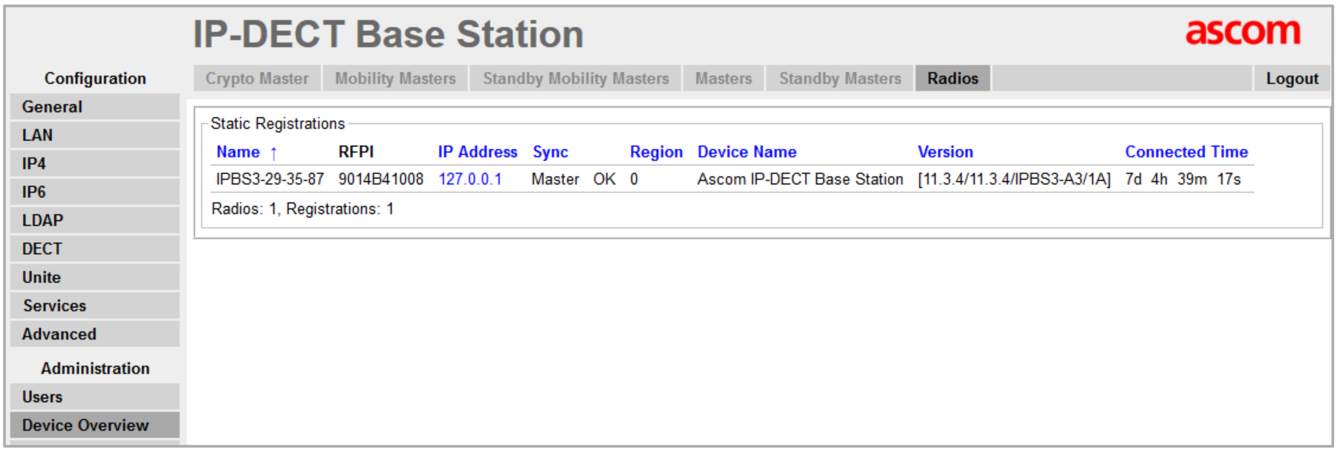


By clicking on the link in **"Fty"** column for a particular subscriber, the local phone feature configuration screen appears.



In the example above the **"Group Pickup"** feature access code is setup (e.g. "*7"), as it is configured in OSV.

4.1.9 *Device Overview*



4.2 OpenScape Voice Configuration

For the needs of the current certification project, at OS Voice the following connections are configured:

- An SIP trunk connection to Mediatrix 4402plus that offers connectivity to ISDN (BRI) provider (OTE).

- A SIP trunk connection to Xpressions server for voicemail services.

Additionally, at OS Voice telephony subscribers are created, which are allocated with various IP telephony features. These subscribers are the users of the SIP and Ascom phone devices which are used in current project's test plan.

At the following paragraphs an example configuration is presented.

Note, that the configuration of the Xpressions server and the SIP phone devices doesn't include any project specific configuration, thus for simplicity reasons will be excluded from current report. SIP trunk and number translation configuration for Mediatrix 4402plus and Xpressions and other common OS Voice configuration is out of scope of the current project and will be omitted, too.

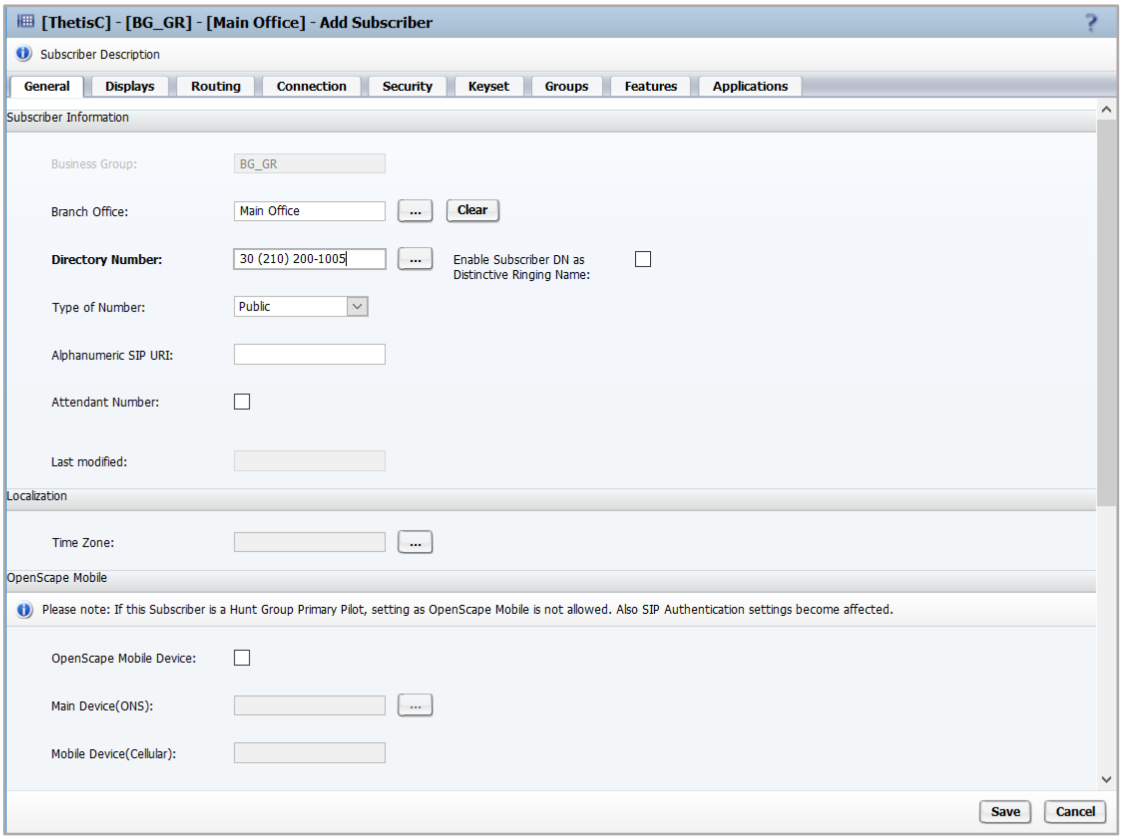
4.2.1 *OS Voice Subscriber Creation*

The subscribers which are related to Ascom handset numbers are created at OS Voice. The process is described in this paragraph.

Go to **Configuration** >> **OpenScape Voice** >> **Business Group** >> **Members** >> **Subscribers**

Click on **[Add]** button.

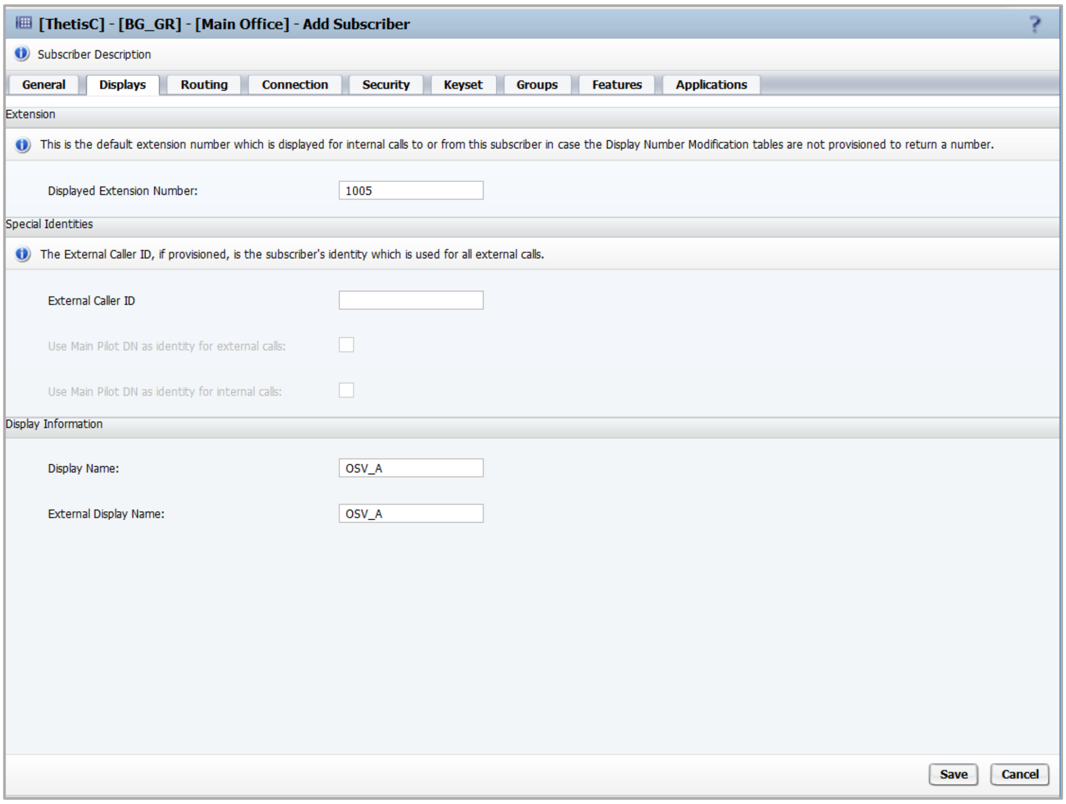
Go to **"General"** tab.



Enter on the **"Directory Number"** the appropriate available Directory Number from the list for the analog user, e.g. 30 (210) 200-1005.

Click on **[Save]**.

Click on **"Display"** tab.

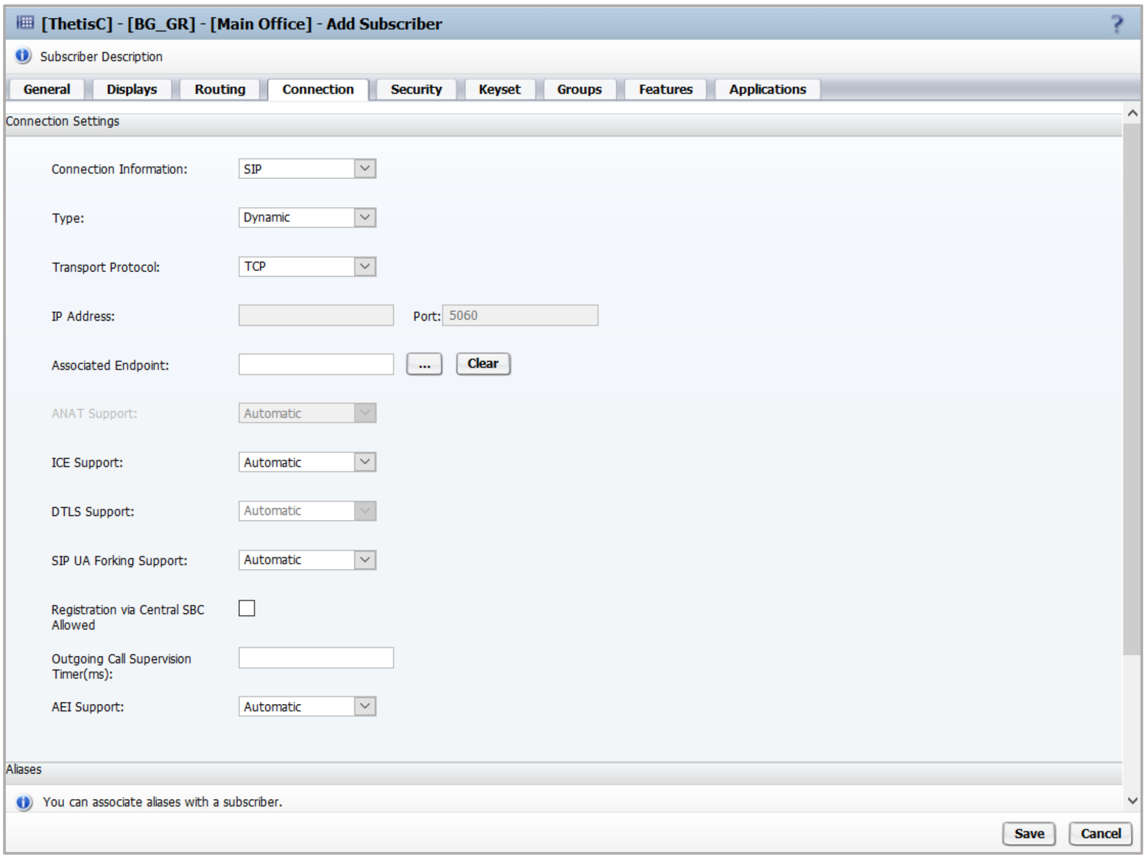


Enter the following:

- **Displayed Extension Number:** 1005 (holds the default value of the extension number that is displayed for incoming and outgoing internal calls of this subscriber when the Display Number Modification tables are not configured to return a number)
- **Display Name:** OSV_A(the name entered here is used as internal display name)
- **External Display Name:** OSV_A (the External Display Name can be a name or a number. It is displayed when the subscriber makes an external call)

Click on **[Save]**.

Click on **"Connection"** tab.

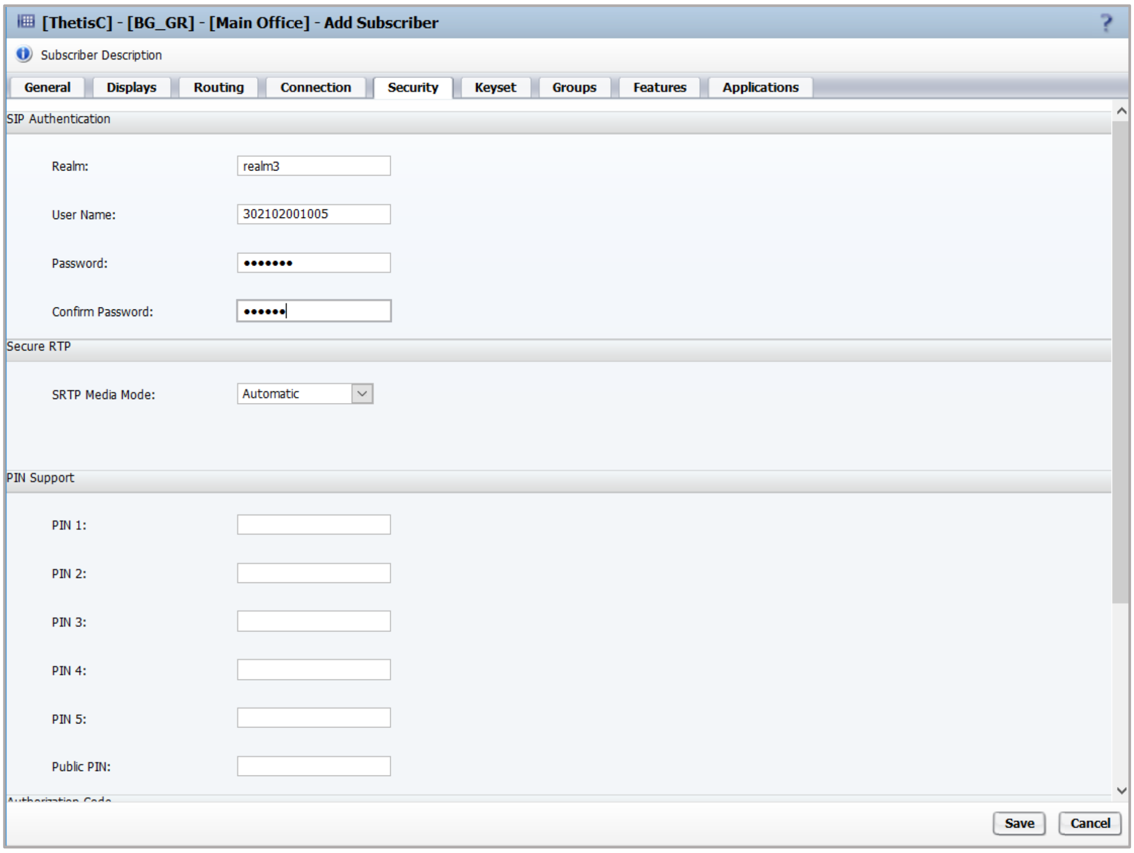


On the field **"Transport Protocol"** fill the value of the protocol to be used, e.g. "TCP" (or "TLS")

This value is the transport protocol for the connection of an SIP phone to OS Voice.

Click on **[Save]**.

Click on **"Security"** tab to configure the subscriber's **digest authentication** data.



Enter the following:

- **Realm:** realm3
- **User Name:** 302102001005
- **Password:** <passphrase>
- **Confirm Password:** <passphrase>

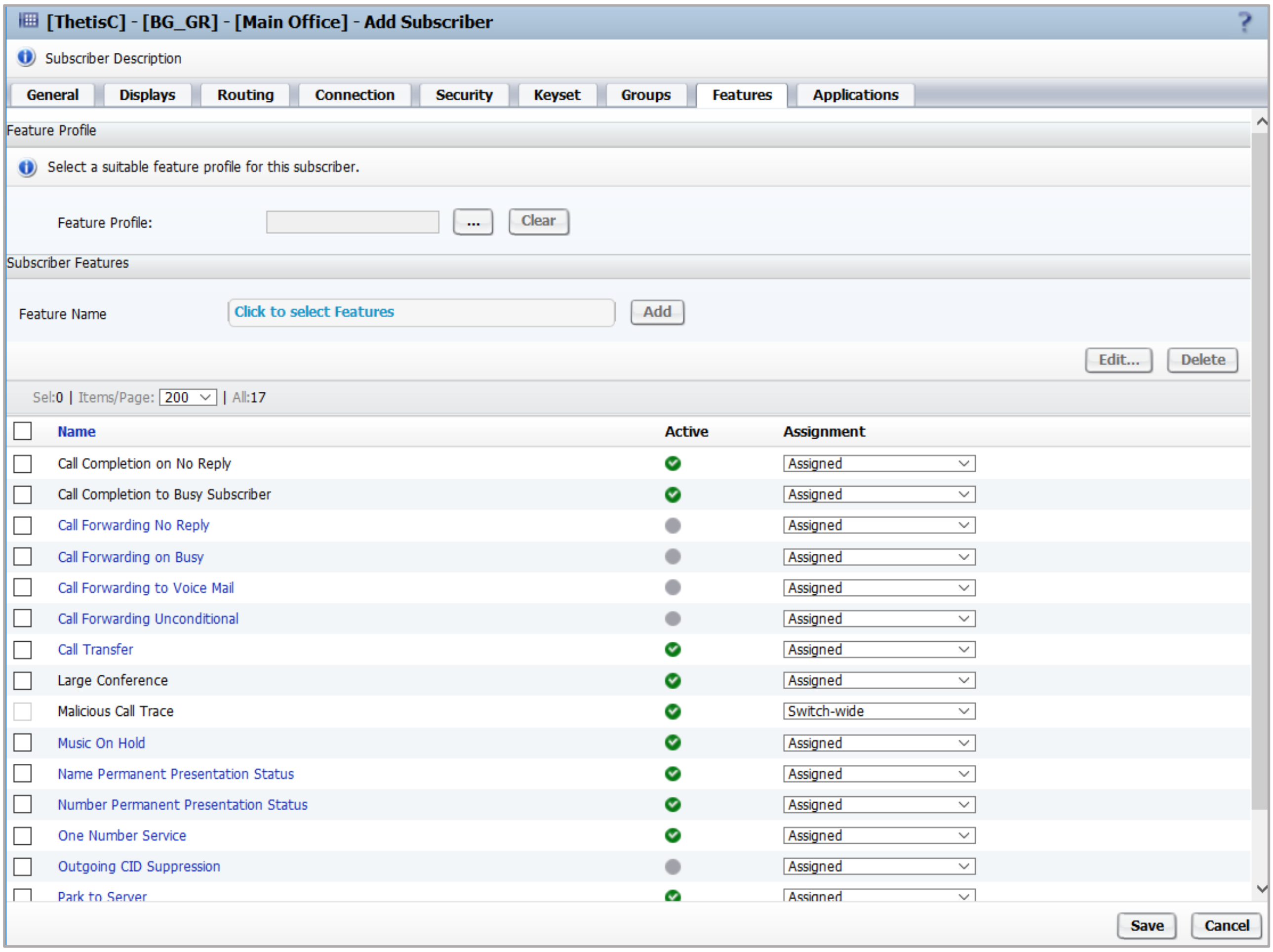
Click on **[Save]**.

Click on **"Features"** tab.

Click on **"Click to select Features"** to add subscriber OSV telephony features by selecting the appropriate feature from the **"Feature Name"** drop down list.

After selecting, click on **"Add".**

As an example, from current testing environment:



Click on **[Save]**.

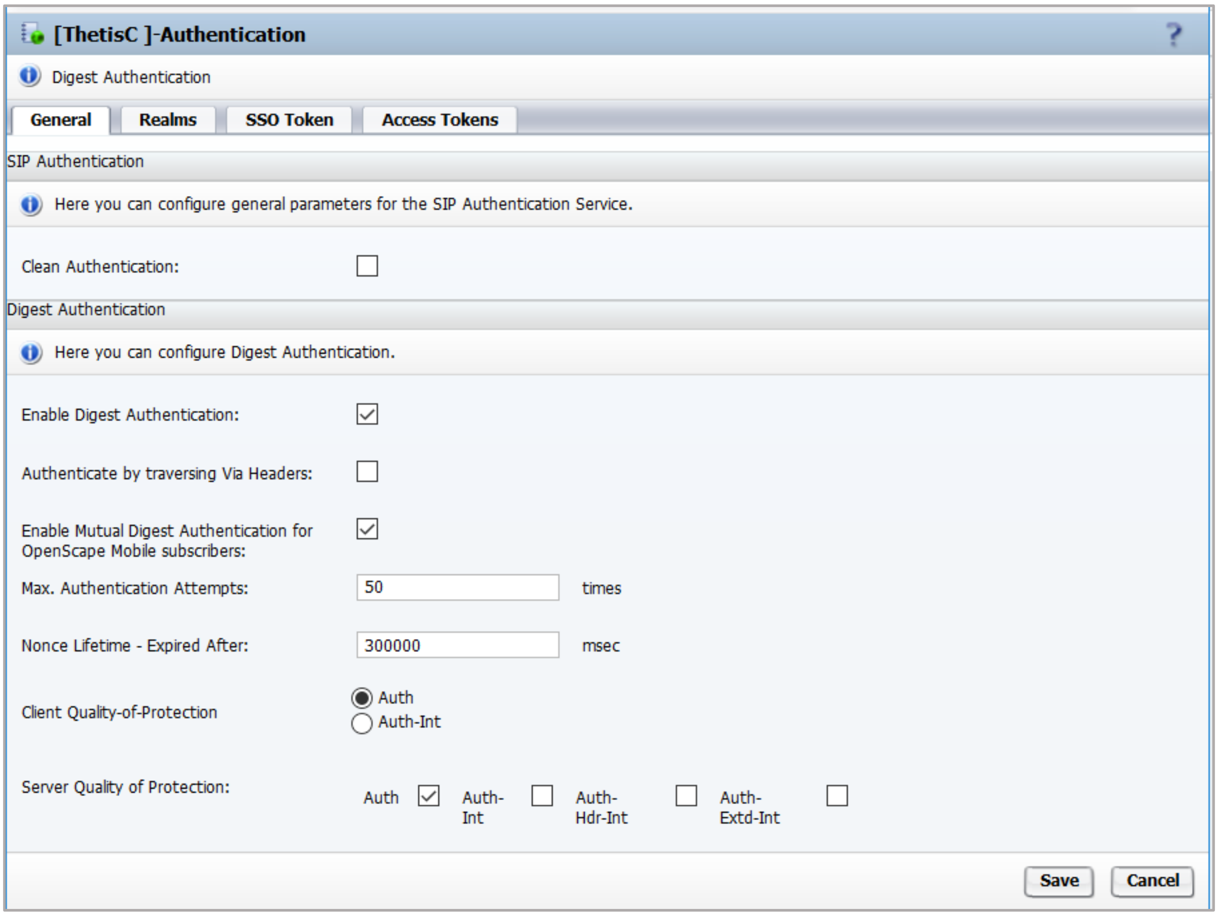
Repeat the same for all the subscribers used in the testing activities (i.e. 30 (210) 200-1001, 30 (210) 200-1002, 30 (210) 200-1003, 30 (210) 200-1006).

4.2.1 *OS Voice Digest Authentication*

In case digest authentication is required for the OSV subscriber, besides the OSV subscriber digest authentication configuration (see sub-section 4.2.1), at OSV a certain flag needs to be activated.

Navigate to **Configuration** >> **OpenScape Voice** >> **Administration** >> **Signaling Management** >> **Authentication.**

The field **"Enable Digest Authentication"** should be checked as shown below:



Click on **[Save]**.

5 Confirmation

Testing personnel confirms that the test cases in chapter 3 were performed and that the results were as described in this document.

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