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 Business V3R0 for voice calls in accordance with the respective test report, dated January 22nd, 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, January 25th, 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

OpenScape Business V3R0

Status: Released
Release Date: 22-01-2021

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History of Change

Version	Date	Description	Author(s)
1.0	January 11th, 2020	Initial Creation	Michael Korakis Atos Greece SM SA E-mail: **michail.korakis@atos.net**
1.1	January 18th, 2020	Editorial changes in sub-sections 1.1.2 & 1.4.3	Mathew Williams Ascom AG E-mail: **Matthew.Williams@ascom.com**
2.0	January 22nd, 2021	Review	Dimitrios Galanakis Atos Greece E-mail: Dimitrios.galanakis@atos.net

1 Overview

1.1 Test Objective

This document describes the results of the testing activities performed for the interoperability between Ascom IP-DECT system and Atos-Unify OpenScape Business V3R0 PBX.

1.1.1 Basis Equipment

Test Equipment:	OpenScape Business X3 (OSBiz X3) OpenScape Deskphone CP400 HFA & CP600 SIP and OpenStage 80T phones
Software Release:	OSBiz X3 server version: osbiz_v3_R0.0.0_157 CP400 HFA firmware: V1 R4.2.0 HFA 200323 CP600 SIP firmware: V1 R7.5.0 SIP 200410 OpenStage 80T firmware: V2 R1.15.1 TDM

1.1.2 IP-DECT

Certification:	Interoperability testing of various call scenarios and features between Ascom IP-DECT handset SIP subscribers registered to OpenScape Business and Atos-Unify HFA, SIP & TDM OSBiz stations. Furthermore, calls between PSTN subscribers and Ascom IP-DECT subscribers are, also, tested. Finally, voice codec verification and DTMF scenarios have been taken into consideration, too.
Test Equipment:	Ascom IP-DECT Base Station (single system) Ascom DECT Handsets d63-Talker Ascom DECT Handset d81-Messenger
Software Release:	Ascom IP-DECT Base Station version: IPBS3[11.1.6], Bootcode[11.1.6], Hardware[IPBS3-A3/1A] Ascom DECT Handset d63-Talker firmware: 2.10.2 Ascom DECT Handset d81-Messenger firmware: 4.12.1
Manufacturer:	Ascom (Sweden) AB Grimbodalen 2 SE-417 49

	Göteborg Sweden Tel. +41 41 544 78 00 https://www.ascom.com/
Description:	Ascom DECT handsets are registered to the OpenScape Business 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 Business. The Ascom IP-DECT Base Station operates as a conduit between the OpenScape Business and the Ascom DECT handsets. After the Ascom DECT wireless handsets register with the Ascom IP-DECT Base Station system, the Base Stations registers the handsets to OpenScape Business.
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 Business as SIP stations. OpenScape Business provides the VoIP telephony facilities to Ascom DECT handset SIP stations. Additionally, Atos-Unify HFA, SIP and TDM stations are connected to OpenScape Business, too. OpenScape Business is connected via ISDN interface to ISDN BRI provider for the PSTN access (OTE ITSP). 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 was to evaluate the ability of the Ascom DECT handsets to make and receive audio calls to and from Atos-Unify HFA, SIP, TDM deskphones and PSTN endpoints. The OSBiz integrated voicemail services were used to allow users to leave voicemail messages and to demonstrate Message Waiting Indication on the DECT handsets. DTMF capabilities were verified with the digit recognition system included in OSBiz' s voicemail and AutoAttendant systems. The interoperability testing activities included both feature functionality and serviceability tests.

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

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

- OSBiz 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.

- OSBiz TCPdump tracing

- Wireshark

1.3 Realization Data

Test Preparation:	November 23rd, 2020
Test Duration:	November 24th – 27th & December 2nd – 3rd, 2020
Test Location:	Atos Greece SM SA CSL Athens lab 455, Irakleiou Avenue 141 22, N. Irakleio, Athens, Greece
Test Personnel:	Atos Greece SM SA, Michael Korakis, michail.korakis@atos.net Ascom AG, Mathew Williams, Matthew.Williams@ascom.com

1.4 Test Result Summary

The certification of Ascom IP-DECT system with OpenScape Business V3R0 PBX is passed with certain restrictions and observations applied. Performance testing was not included.

1.4.1 Remarks

- 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 number, however, will be recorded in the call list.

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

- In **Call Reject** scenarios, Ascom IP-DECT system can be configured to send SIP "486 Busy Here" or SIP "603 Decline" message. For the OSBiz system, the setting *"Treat rejected calls as"* should be 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".

- For **Not Answered** call scenarios, in OSBiz there is no timer until the call is disconnected. The internal caller must clear the call.

- **Call Forwarding** scenarios were tested with both local Ascom DECT handset and OSBiz 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.

  If the CFU is configured from the system (Call destination list - Assignment) and the forwarded to destination is a SIP station, then the latter SIP station doesn't display *"forwarded from"* notification (refer to tests 4-40, 4-42 & 4-43).

- A SIP station that participates in an OSBiz **Conference** created by an HFA station, doesn't display a conference call but a basic call to conference initiator. The SIP server is not aware of the HFA conference and so cannot tell the SIP station that its call is in a conference (refer to test 4-58).

- In an OpenScape Business environment the recommended **DTMF** transport standard is RFC2833/RFC4733. No other method is supported.

1.4.2 Restrictions

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

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

  CID suppression isn't supported in OSBiz for calls between internal stations.

- Ascom DECT handsets don't support the display of **Call Pickup Group notifications** with OSBiz.

- Due to DECT limitations, in **Semi-Attended Call Transfer** scenarios where the transfer action is performed by an Ascom DECT handset, the "transferred to" party displays the "transferred" party on call connected state; on ringing state the "transferred to" party displays the "transferee" (refer to tests 4-26, 4-29 & 4-31).

- **Call Deflect** isn't supported on Ascom DECT handsets.

- OSBiz by design doesn't send the number of unread **Voicemail** messages in NOTIFY, but only if an unread message exists; if there are already unread messages and a new voice message arrives, then the system will detect that there are already unread messages and won't send a NOTIFY again.

  Therefore, no matter how many unread voicemail messages an Ascom handset user has, on the phone 1 unread voicemail message will be displayed.

- Ascom IP-DECT system doesn't support local **3-party Conference** against OSBiz.

- Regarding audio **Codecs** support, OSBiz and Ascom d81 DECT handsets don’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 5-08).

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.

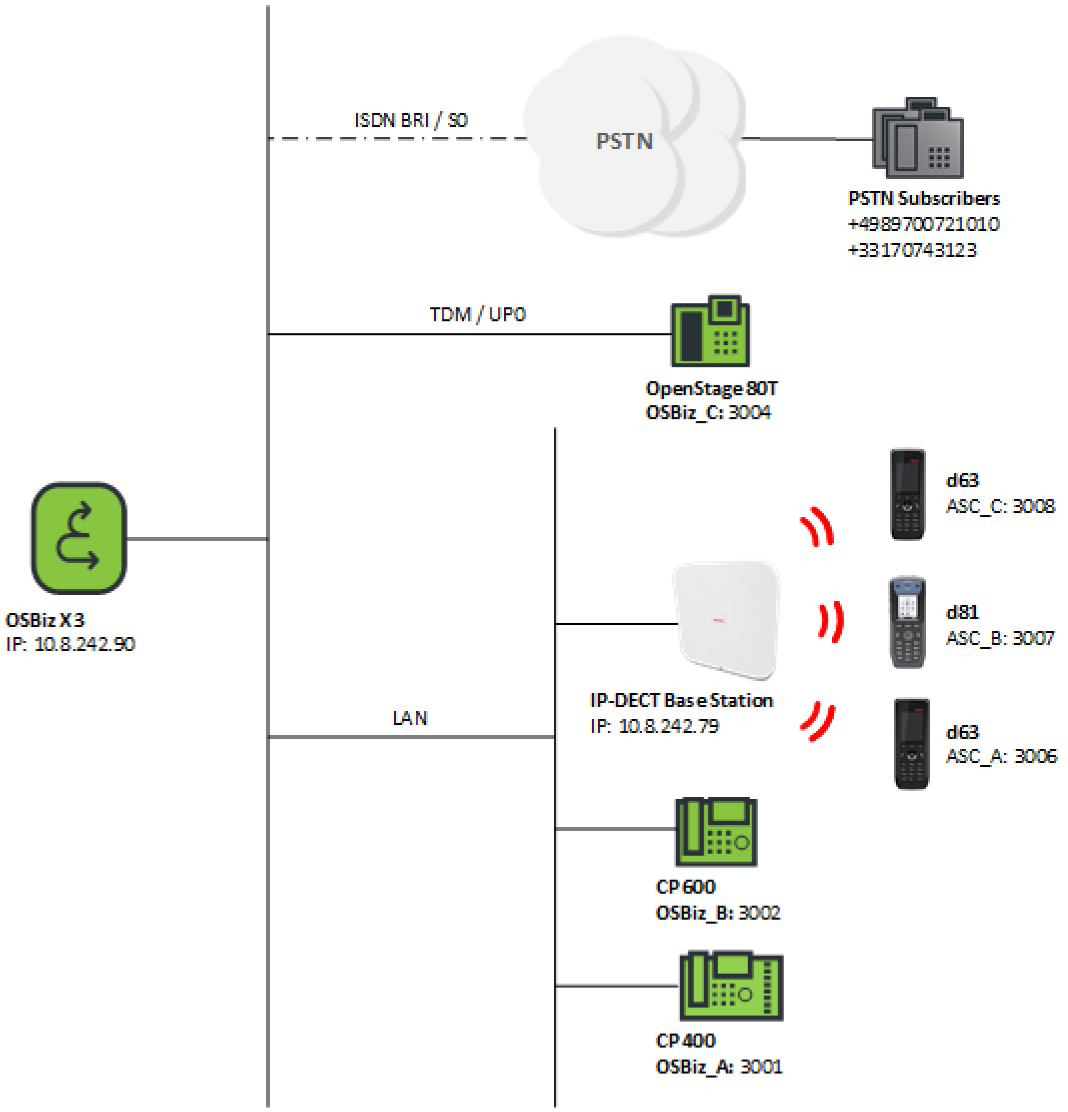


Figure 1: Network Topology Diagram

2 Basic Configuration

2.1 Ascom IP-DECT System

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

The IP addresses of the Master 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 OSBiz SIP extensions are utilized for Ascom DECT handset SIP stations:

ASC_A = 3006 (d63-Talker handset)

ASC_B = 3007 (d81-Messenger handset)

ASC_C = 3008 (d63-Talker handset)

2.2 Atos-Unify Component Infrastructure

OSBiz is configured for 4-digit extension dialing (30xx) between internal subscribers (registered to the same OSBiz).

OSBiz provides access to PSTN (OTE ISDN BRI provider) via the direct connection of ISDN provider DTE equipment to OSBiz ISDN port (S0). OSBiz subscribers dial 0 (ISDN line seizure code) + PSTN number to reach a PSTN subscriber, e.g. 0 2108187986, 0 004989700721010 for national and international calls, correspondingly.

- OSBiz IP address

  10.8.242.90

- OSBiz Clients (directly connected)

  OSBiz_A = 3001 (CP 400 HFA phone device || Fist Name: *OSBiz_A*, Last Name: *HFA*)
  OSBiz_B = 3002 (CP 600 SIP phone device || Fist Name: *OSBiz_B*, Last Name: *SIP*)
  OSBiz_C = 3004 (OpenStage 80T phone device || Fist Name: *OSBiz_C*, Last Name: *TDM*)

- ISDN DDI number for incoming calls from PSTN

  +302106203360 (or 2106203360 national)

- PSTN

  +4989700721010

  +33170743123

Additionally, some other special OSBiz extensions are used for the needs of the testing activities:

- MLHG pilot = 3000
- MULAP Group number = 3100
- Voicemail number = 3300
- AutoAttendant = 3990

3 Test Results in Detail

The testing procedure is grouped in 6 major testing areas; Registration, Basic Calls, Basic Calls Extended, Telephony Features, Audio Features and Redundancy.

The *"Registration"* covers Ascom DECT handset registration with OSBiz with TCP or UDP transport and with or without digest authentication. In the testing activities Ascom DECT handsets were registered with TCP and digest authentication to OSBiz. Additionally, registration expiration timer was tested, too.

"Basic Calls" refer to incoming and outgoing call scenarios with local or PSTN subscribers. Additionally, call to busy subscriber, rejected call, call no reply, cancelled call and call to unavailable DECT 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
- Call Pickup
- Hunt Group
- MULAP Group
- Call Transfer Attended, Semi-Attended and Blind
- Call Deflect
- Call Forwarding Unconditional, Busy and No Reply (both system and local phone features)
- Voicemail (Message Waiting Indication & DTMF)
- Conference (PBX / 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. The exchange of DTMF digits is verified in established calls between Ascom DECT handset users with CP phone users / PSTN subscribers. Additionally, DTMF digit recognition is also verified for calls from Ascom DECT users to OSBiz voicemail and UC AutoAttendant systems.

Finally, *"Redundancy"* scenarios, focus on 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 *"Redundancy"* isn't verified because the test environment was setup with a single IP-DECT base station system.

3.1 Registration

3.1.1 OSBiz Registration

No.	Test Procedure	Expected Result	Result	Comments
1-01	• Register Ascom IP-DECT phone to OSBiz without Digest Authentication.	• DECT phone successfully registers with OSBiz.	OK	

No.	Test Procedure	Expected Result	Result	Comments
1-02	• Register Ascom IP-DECT phone to OSBiz with Digest Authentication.	• DECT phone successfully registers with OSBiz.	OK	
1-03	• Check that Ascom IP-DECT phone may register to OSBiz with either TCP or UDP.	• DECT phone successfully registers with OSBiz.	OK	
1-04	• Registration expiration timer at Ascom IP-DECT phone is set below OSBiz's corresponding value.	• OSBiz responds with a 423 Interval Too Brief to negotiate the lower re-Registration interval. • Successful registration to OSBiz.	OK	Value used on Ascom IP-DECT system was 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	• OSBiz_A calls ASC_A. • OSBiz_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	
2-03	• OSBiz_A calls ASC_A. • ASC_A hangs up at the end of the communication.	• Post Dial Delay (PDD). • Ringback tone heard by calling party.	OK	

No.	Test Procedure	Expected Result	Result	Comments
		• 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.		
2-04	• ASC_A calls OSBiz_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 OSBiz_A. • OSBiz_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	• PSTN01 calls ASC_A. • PSTN01 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	ASC_A displays external call.
2-07	• PSTN01 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	OK	

No.	Test Procedure	Expected Result	Result	Comments
		ringing and after call pick up. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.		
2-08	• ASC_A calls PSTN01. • 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 PSTN01. • PSTN01 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. Call clears at about 30 secs.
2-11	• OSBiz_A calls ASC_A. • ASC_A is busy.	• Busy tone/display on calling party. • The call is properly cleared.	OK	IP-DECT base station responds with a 486 Busy Here.

No.	Test Procedure	Expected Result	Result	Comments
2-12	• ASC_A calls OSBiz_A. • OSBiz_A is busy.	• Busy tone/display on calling party. • The call is properly cleared.	OK	
2-13	• PSTN01 calls ASC_A. • ASC_A is busy.	• Busy tone/display on calling party. • The call is properly cleared.	OK	
2-14	• ASC_A calls PSTN01. • PSTN01 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. Call clears at about 15 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	• OSBiz_A calls ASC_A. • ASC_A rejects incoming call.	• Busy tone/display on calling party. • The call is properly cleared.	OK	Same as 2-15.
2-17	• ASC_A calls OSBiz_C. • OSBiz_C rejects incoming call.	• Busy tone/display on calling party. • The call is properly cleared.	OK	OSBiz responds with a 603 Decline. Call clears at about 35 secs.
2-18	• PSTN01 calls ASC_A. • ASC_A rejects incoming call.	• Busy tone/display on calling party. • The call is properly cleared.	OK	IP-DECT Base station sends 486 Busy Here.
2-19	• ASC_A calls PSTN01. • PSTN01 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. Call clears at about 3 mins by IP-DECT base station.
2-21	• OSBiz_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	Same as 2-20.
2-22	• ASC_A calls OSBiz_A. • OSBiz_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.	NA	OSBiz doesn't have a timer for terminating the call on ringing state. The caller must clear the call.
2-23	• PSTN01 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 PSTN01. • PSTN01 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	Comments
2-25	• ASC_A calls ASC_B and hangs up before destination picks up.	• Call is terminated on both sides after hanging up.	OK	
2-26	• OSBiz_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 OSBiz_A and hangs up before destination picks up.	• Call is terminated on both sides after hanging up.	OK	
2-28	• PSTN01 calls ASC_A and hangs up before destination picks up.	• Call is terminated on both sides after hanging up.	OK	

No.	Test Procedure	Expected Result	Result	Comments
2-29	• ASC_A calls PSTN01 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	ASC_B = switched off, OSBiz sends 486 BUSY HERE (Handset displays BUSY). Call clears after 30 sec.
2-31	• ASC_A is unregistered. • OSBiz_A calls ASC_A.	• Busy tone/display on calling party. • The call is properly cleared.	OK	
2-32	• OSBiz_A is unregistered. • ASC_A calls OSBiz_A.	• Busy tone/display on calling party. • The call is properly cleared.	OK	
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 send 503 Service unavailable, OSBiz sends 486 BUSY HERE; handset display BUSY.
2-34	• ASC_A is out of coverage (w/ batt. removed). • OSBiz_A calls ASC_A.	• Busy tone/display on calling party. • The call is properly cleared.	OK	ASC_A = out of coverage (remove battery while registered), IP-DECT sends 503 Service unavailable.

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, outgoing and missed calls entries.	• Call history properly lists incoming, outgoing and missed calls (internal and external).	OK	Anonymous incoming calls are not displayed in Ascom DECT handset call list.

No.	Test Procedure	Expected Result	Result	Comments
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 OSBiz_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 PSTN01 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	• 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	OK	SIP UPDATE messages are being sent every 15 minutes to check status of the call. All messages are responded correctly.

No.	Test Procedure	Expected Result	Result	Comments
	and then ASC_A mutes for another 5 mins.	endpoints. • Check SIP Session Timer Expiry.		
3-06	• Setup a call between ASC_A and OSBiz_A. • The call must last at least 1 hour. • OSBiz_A puts the call on mute when call is established. • After 5 mins OSBiz_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	Same as 3-05.

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 (e.g. *97). • ASC_A calls ASC_B.	• On softphone display a DND logo is displayed. • Calling party gets busy tone.	OK	Locally via calling *42# (deactivate #42#), handset displays 4007 > Busy, caller gets busy tone (IP-DECT base station send 486 Busy Here). OSBiz DND feature isn't supported for SIP stations.
3-08	• ASC_A has DND activated via menu options or via DND feature code (e.g. *97). • OSBiz_A calls ASC_A.	• On softphone display a DND logo is displayed. • Calling party gets busy tone.	OK	Same as 3-07.
3-09	• After scenario 3-8, ASC_A deactivates DND. • OSBiz_A calls ASC_A again.	• DND logo disappears from display. • Incoming calls are normally signaled on the phone.	OK	

3.3.4 *Calling Identity Suppression (CID)*

No.	Test Procedure	Expected Result	Result	Comments
3-10	• ASC_A activates Calling Identity Suppression via the CID Suppression feature access code (e.g. *86). • ASC_A calls ASC_B.	• Number and name are not displayed on called device.	NA	CID suppression isn't supported in OSBiz for calls between internal stations.

No.	Test Procedure	Expected Result	Result	Comments
3-11	• ASC_A activates Calling Identity Suppression via the CID Suppression feature access code (e.g. *86). • ASC_A calls OSBiz_A.	• Number and name are not displayed on called device.	NA	Same as 3-10.
3-12	• ASC_A activates Calling Identity Suppression via the CID Suppression feature access code (e.g. *86). • ASC_A calls PSTN01.	• Number is not displayed on called phone.	OK	When dialing *449<PSTN01 number>, the called party sees the (ISDN) number of the caller. OSBiz sends anonymous in "From" header and "Privacy:id" to ISDN provider. It's up to the ISDN provider to have the appropriate configuration to hide or display the number.
3-13	• OSBiz_A activates Calling Identity Suppression via the CID Suppression feature access code (e.g. *86). • OSBiz_A calls ASC_A.	• Number and name are not displayed on called phone. • Only the dialed digits are displayed on the dialing phone	NA	Same as 3.10.
3-14	• PSTN01 activates identity restriction. • PSTN01 calls ASC_A.	• Number is not displayed on called phone.	OK	ASC_A display "********* External Call "

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).	OK	
4-02	• OSBiz_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	OK	

No.	Test Procedure	Expected Result	Result	Comments
		establishment after resuming the call (media clipping).		
4-03	• ASC_A calls OSBiz_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	
4-04	• 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.	OK	On Ascom handset, press R2 to toggle between calls and R1 to hang up active call and switch to held call.
4-05	• OSBiz_A calls ASC_A. • ASC_A makes a consultation call to ASC_B. • ASC_A toggles between calls with OSBiz_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 directions. • Both ends idle after call clearing.	OK	
4-06	• ASC_A calls OSBiz_A. • OSBiz_A makes a consultation call to OSBiz_B. • OSBiz_A toggles between calls with ASC_A and OSBiz_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	• Ring back tone heard by calling party.	OK	

No.	Test Procedure	Expected Result	Result	Comments
	ASC_B. • ASC_A toggles between calls with PSTN and ASC_A.	• 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.		
4-08	• OSBiz_A calls ASC_A. • OSBiz_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 OSBiz_A and OSBiz_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	Enable call waiting on Ascom handset with *43#.

3.4.2 *Call Pickup Group*

No.	Test Procedure	Expected Result	Result	Comments
4-09	• OSBiz_B and ASC_B belong to the same Call Pickup group. • ASC_A calls OSBiz_B. • ASC_B picks up the call via phone menu or via call pickup access code (e.g. *57) while OSBiz_B is ringing. • ASC_B hangs up at the end of the communication.	• ASC_B gets a notification • ASC_B can 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 does not receive Call Pickup Notification but was able to retrieve the call via *57 Pickup Access Code.
4-10	• OSBiz_B and ASC_B belong to the same Call Pickup group. • OSBiz_A calls ASC_B. • OSBiz_B picks up the call via phone menu or via call pick up access code (e.g. *57) while ASC_B is ringing. • OSBiz_B hangs up at the end of the communication.	• ASC_B gets a notification • OSBiz_B can pick up the call. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	OK	
4-11	• OSBiz_A and ASC_A belong to the same Call Pickup group. • PSTN calls OSBiz_A.	• ASC_A can pick up the call. • No media clipping when connecting both ends.	POK	ASC_A does not receive Call Pickup Notification but was able to retrieve the

No.	Test Procedure	Expected Result	Result	Comments
	• ASC_A picks up the call via phone menu or via call pick up access code (e.g. *57) while OSBiz_A is ringing. • ASC_A hangs up at the end of the communication.	• Speech path in both directions. • Both ends idle after call clearing.		call via *57 Pickup Access Code
4-12	• OSBiz_A and ASC_A belong to the same Call Pickup group. • PSTN calls ASC_A. • OSBiz_A picks up the call via phone menu or via call pick up access code (e.g. *57) while ASC_A is ringing. • OSBiz_A hangs up at the end of the communication.	• OSBiz_A can pick up the call. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	OK	

3.4.3 *Hunt Group*

No.	Test Procedure	Expected Result	Result	Comments
4-13	• Hunt Group is configured with ASC_A, ASC_B and ASC_C. • OSBiz_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	
4-14	• Hunt Group is configured with ASC_A, ASC_B and ASC_C. • OSBiz_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	
4-15	• 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	

No.	Test Procedure	Expected Result	Result	Comments
4-16	• Hunt Group is configured with ASC_A, OSBiz_A and ASC_C. • ASC_A dials #85 and leaves the group. • ASC_B calls the hunt group number. • ASC_A dials *85 and rejoins the group. • ASC_B calls the hunt group number again. • ASC_A rings and answers. • Call is cleared at the end of the communication.	• ASC_A successfully leaves the HG after dialing #85 and incoming calls to the HG group number aren't signaled on ASC_A device. • ASC_A successfully rejoins the HG after dialing *85 and incoming calls to the HG group number are now signaled on ASC_A device. • Both ends idle after call clearing.	OK	

3.4.4 *MULAP Group*

No.	Test Procedure	Expected Result	Result	Comments
4-17	• ASC_A, OSBiz_A and ASC_C are members of the same MULAP group. • OSBiz_B calls MULAP number. • All MULAP group members are ringing. • ASC_A 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	
4-18	• ASC_A, OSBiz_A and ASC_C are members of the same MULAP group. • ASC_B calls MULAP number. • All MULAP group members are ringing. • ASC_A 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	
4-19	• ASC_A, OSBiz_A and ASC_C are members of the same MULAP group. • PSTN calls MULAP number. • All MULAP group members are ringing. • ASC_A 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	

3.4.5 *Call Transfer (Attended)*

No.	Test Procedure	Expected Result	Result	Comments
4-20	• 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	Press R3008 to consult and R4 on Ascom handset to transfer.
4-21	• ASC_A calls OSBiz_A. • OSBiz_A consults to ASC_B and waits for ASC_B to pick up the call. • OSBiz_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	
4-22	• ASC_A calls OSBiz_A. • OSBiz_A consults to OSBiz_B and waits for OSBiz_B to pick up the call. • OSBiz_A transfers ASC_A towards OSBiz_B. • After transfer is completed and communication is established between ASC_A and OSBiz_B, ASC_A hangs up.	• MOH is played to ASC_A while reaching OSBiz_B. • No Media clipping and proper speech path after transfer when ASC_A and OSBiz_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	
4-23	• OSBiz_A calls ASC_A. • ASC_A puts OSBiz_A on hold, calls ASC_B and waits for ASC_B to pick up the call. • ASC_A resumes the call with OSBiz_A (ASC_B is placed on hold) and then transfers OSBiz_A towards ASC_B. • After transfer is completed and communication is established between OSBiz_A and ASC_B, OSBiz_A hangs up.	• MOH is played in all related steps (to OSBiz_A while reaching ASC_B / to ASC_B while resuming the call with OSBiz_A). • Proper call resume when ASC_A toggles between calls with OSBiz_A and ASC_B. • No Media clipping and proper speech path after transfer when OSBiz_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.		
4-24	• PSTN calls OSBiz_A. • OSBiz_A consults to ASC_B and waits for ASC_B to pick up the call. • OSBiz_A transfers PSTN towards ASC_B. • After transfer is completed and communication is established between PSTN and ASC_B, 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	
4-25	• PSTN calls ASC_A. • ASC_A puts PSTN on hold, calls OSBiz_B and waits for OSBiz_B to pick up the call. • ASC_A resumes the call with PSTN (OSBiz_B is placed on hold) and then transfers PSTN towards OSBiz_B. • After transfer is completed and communication is established between PSTN and OSBiz_B, PSTN hangs up.	• MOH is played in all related steps (to PSTN while reaching OSBiz_B / to OSBiz_B while resuming the call with PSTN). • Proper call resume when ASC_A toggles between calls with PSTN and OSBiz_B. • No Media clipping and proper speech path after transfer when PSTN and OSBiz_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	

3.4.6 *Call Transfer (Semi-Attended)*

No.	Test Procedure	Expected Result	Result	Comments
4-26	• 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	ASC_C is updated on the display with ASC_A after accepting the call. On ringing state ASC_C continues to display ASC_B. 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
4-27	• ASC_A calls OSBiz_A. • OSBiz_A consults to ASC_B. • OSBiz_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 OSBiz_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	
4-28	• ASC_A calls OSBiz_A. • OSBiz_A consults to OSBiz_B. • OSBiz_A hangs up while OSBiz_B is ringing. • ASC_A receives ring back tone. • After communication is established between ASC_A and OSBiz_B, ASC_A hangs up.	• MOH is played to ASC_A after consultation and before OSBiz_A hangs up. • No Media clipping and proper speech path after transfer when ASC_A and OSBiz_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	
4-29	• OSBiz_A calls ASC_A. • ASC_A consults to ASC_B. • ASC_A hangs up while ASC_B is ringing. • OSBiz_A receives ring back tone. • After communication is established between OSBiz_A and ASC_B, OSBiz_A hangs up.	• MOH is played to OSBiz_A after consultation and before ASC_A hangs up. • No Media clipping and proper speech path after transfer when OSBiz_A and ASC_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	POK	Same as 4.26.
4-30	• PSTN calls OSBiz_A. • OSBiz_A consults to ASC_B. • OSBiz_A hangs up while ASC_B is ringing. • PSTN receives ring back tone. • After communication is established between PSTN and ASC_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 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
4-31	• PSTN calls ASC_A. • ASC_A consults to OSBiz_B. • ASC_A hangs up while OSBiz_B is ringing. • PSTN receives ring back tone. • After communication is established between PSTN and OSBiz_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 OSBiz_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	POK	Same as 4.26

3.4.7 *Call Transfer (Blind)*

No.	Test Procedure	Expected Result	Result	Comments
4-32	• 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 RR3008# on Ascom handset.
4-33	• ASC_A calls OSBiz_B. • OSBiz_B 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	
4-34	• ASC_A calls OSBiz_B. • OSBiz_B blind transfers to OSBiz_C. • ASC_A receives ring back tone. • After transfer is completed and communication is established between ASC_A and OSBiz_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 OSBiz_C 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.		
4-35	• OSBiz_A calls ASC_A. • ASC_A blind transfers to ASC_B. • OSBiz_A receives ring back tone. • After transfer is completed and communication is established between OSBiz_A and ASC_B, OSBiz_A hangs up.	• MOH to OSBiz_A after consultation and before blind transfer is completed. • No Media clipping and proper speech path after transfer when OSBiz_A and ASC_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	
4-36	• PSTN calls OSBiz_C. • OSBiz_C blind transfers to ASC_B. • PSTN receives ring back tone. • After transfer is completed and communication is established between PSTN and ASC_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 ASC_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	
4-37	• PSTN calls ASC_A. • ASC_A blind transfers to OSBiz_B. • PSTN receives ring back tone. • After transfer is completed and communication is established between PSTN and OSBiz_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 OSBiz_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	

3.4.8 *Call Deflect*

No.	Test Procedure	Expected Result	Result	Comments
4-38	• 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 deflect 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.		
4-39	• ASC_A calls OSBiz_B. • OSBiz_B 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.	OK	

3.4.9 *Call Forwarding (Unconditional)*

No.	Test Procedure	Expected Result	Result	Comments
4-40	• 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.	OK	Press, *21*3008# to activate on ascom handset (deactivate with #21#). The local phone feature access code should not overlap with the PBX feature access code, because they will conflict and not work correctly. When CFU enabled from OSBiz system, ASC_C does not display CF display notification.
4-41	• OSBiz_A calls ASC_A who has CFU activated via phone locally or via the system. • ASC_A forwards the call immediately to OSBiz_B. • Leave OSBiz_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	Same as 4.40.

No.	Test Procedure	Expected Result	Result	Comments
	between OSBiz_A and OSBiz_B, OSBiz_A hangs up.	• Media clipping and speech path after forwarding. • Both ends idle after call clearing.		
4-42	• ASC_A calls OSBiz_A who has CFU activated via phone locally or via the system. • OSBiz_A forwards the call immediately to OSBiz_B. • Leave OSBiz_B ring for a few seconds and pick up the call. • After forward is completed and communication is established between OSBiz_A and OSBiz_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.	OK	When CFU enabled from OSBiz system, OSBiz_B does not display CF display notification.
4-43	• 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	Same as 4.42.

3.4.10 *Call Forwarding (Busy)*

No.	Test Procedure	Expected Result	Result	Comments
4-44	• 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.	OK	Press *67*3008# to activate on Ascom handset (deactivate with #67#).

No.	Test Procedure	Expected Result	Result	Comments
	between ASC_A and ASC_C, ASC_A hangs up.	• Both ends idle after call clearing.		
4-45	• OSBiz_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 OSBiz_B. • Leave OSBiz_B ring for a few seconds and pick up the call. • After forward is completed and communication is established between OSBiz_A and OSBiz_B, OSBiz_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	
4-46	• ASC_A calls busy subscriber OSBiz_A who has CFB activated via phone locally or via the system. • OSBiz_A doesn't have call waiting activated and forwards the call immediately to OSBiz_B. • Leave OSBiz_B ring for a few seconds and pick up the call. • After forward is completed and communication is established between ASC_A and OSBiz_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.	OK	
4-47	• 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	

3.4.11 *Call Forwarding (No Reply)*

No.	Test Procedure	Expected Result	Result	Comments
4-48	• 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.	OK	*61*3008# activate on DECT handset (deactivate with #61#).
4-49	• OSBiz_A calls ASC_A who has CFNR activated via phone locally or via the system. • ASC_A rings and after a while forwards to OSBiz_B. • Leave OSBiz_B ring for a few seconds and pick up the call. • After forward is completed and communication is established between OSBiz_A and OSBiz_B, OSBiz_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	
4-50	• ASC_A calls OSBiz_A who has CFNR activated via phone locally or via the system. • OSBiz_A rings and after a while forwards to OSBiz_B. • Leave OSBiz_B ring for a few seconds and pick up the call. • After forward is completed and communication is established between ASC_A and OSBiz_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.	OK	
4-51	• 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.		

3.4.12 *Voicemail*

No.	Test Procedure	Expected Result	Result	Comments
4-52	• OSBiz_A and OSBiz_B call ASC_A and leave a voicemail message.	• Proper MWI indication with the correct number of new messages on phone. • MWI is retained after Ascom phone restart.	OK	
4-53	• After scenario 4-52, ASC_A deletes or hears voicemail messages one by one.	• Voicemail messages are properly heard. • Number of voicemail messages is correctly displayed after one message is listened or deleted. • MWI disappears from the phone after all voicemail messages are listened or deleted.	OK	
4-54	• ASC_A calls voicemail system to hear messages and perform various actions using the appropriate DTMF options. • ASC_A calls back OS4K_A from voicemail system.	• DTMF digits are properly recognized from voicemail system. • Login possible to personal voicemail box.	OK	

3.4.13 *Conference*

No.	Test Procedure	Expected Result	Result	Comments
4-55	• ASC_A calls OSBiz_A and establishes communication. • ASC_A initiates a consultation call to ASC_B. • After ASC_B answers, ASC_A initiates a local conference.	• Media clipping and proper speechpath after conferencing participants in the conference. • Number(s) presentation on conference participants.	NA	Ascom IP-DECT system doesn't support 3-party conference against OSBiz.

No.	Test Procedure	Expected Result	Result	Comments
	• ASC_A (conference initiator) hangs up.	• Conference participants are still on the conference after ASC_A hangs up. • Number(s) presentation after ASC_A hangs up on conference participants.		
4-56	• ASC_A calls PSTN 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.	NA	Same as 4.55.
4-57	• OSBiz_A calls ASC_A and establishes communication. • OSBiz_B calls ASC_A who has call waiting activated. • ASC_A merges calls with OSBiz_A and OSBiz_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.	NA	Same 4.55.
4-58	• OSBiz_A calls ASC_A and establishes communication. • OSBiz_A initiates a consultation call to ASC_B. • After ASC_B answers, OSBiz_A initiates a system controlled conference. • OSBiz_A calls OSBiz_B and OSBiz_A adds OSBiz_B to the conference. • OSBiz_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 OSBiz_A hangs up. • Number(s) presentation after ASC_A hangs up on conference participants.	OK	On SIP stations the system conference is being displayed as a basic call.

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 OSBiz_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 OSBiz_A.	• Connection is established with G.722. • Media clipping and speechpath.	NA	Ascom 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_C.	• Connection is established with G.722.2. • Media clipping and speechpath.	NA	Communication falls back to G.711, because G.722.2 is screened by the system. OSBiz doesn't support G.722.2.
5-04	• IP-DECT base station is configured to have G.729 as first priority. • ASC_A calls OSBiz_A.	• Connection is established with G.729. • Media clipping and speechpath.	OK	
5-05	• ASC_A is configured to support G.711 only. • OSBiz_A calls ASC_A.	• Connection is established with G.711. • Media clipping and speechpath.	OK	
5-06	• ASC_A is configured to support G.723 only. • OSBiz_A calls ASC_A.	• Connection is established with G.723. • Media clipping and speechpath.	NA	Ascom and OSBiz do not support G.723.
5-07	• ASC_A is configured to support G.729 only. • OSBiz_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 OSBiz_A. • OSBiz_A calls ASC_A.	• Call is disconnected. • Call failure indication.	POK	503 Service unavailable from IP-DECT base station. 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

No.	Test Procedure	Expected Result	Result	Comments
				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.
5-09	• Make sure there is codec mismatch between ASC_A and OSBiz_A. • ASC_A calls OSBiz_B.	• Call is disconnected. • Call failure indication.	OK	488 Not Acceptable Here from OSBiz.

3.5.2 *DTMF*

No.	Test Procedure	Expected Result	Result	Comments
5-10	• ASC_A calls OSBiz_A. • ASC_A sends DTMF 1234567890*# to OSBiz_A and then OSBiz_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 OpenScape Business UC Auto Attendant and sends DTMF tones.	• DTMF digits are properly recognized by UC Auto Attendant.	OK	AutoAttendant number = 3990

3.6 Redundancy

3.6.1 *IP-DECT Base Station Switchover*

No.	Test Procedure	Expected Result	Result	Comments
6-01	• 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.	NT	Single Base Station is used for testing activities.

No.	Test Procedure	Expected Result	Result	Comments
6-02	• In mirror configuration the functions do 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.	NT	Same as 6-01.

Abbreviations:

OK	Test case executed successfully
POK	Test case executed with PARTIAL success
NOK	Test case executed NOT successfully
NA	Test case not applicable
NT	Test case not tested

3.7 Summary

Test results aggregated:

Category	Planned	OK	POK	NOK	NA	NT
Registration	4	4	-	-	-	-
Basic Calls	34	33	-	-	1	-
Basic Calls Extended	14	11	-	-	3	-
Telephony Features	58	49	5	-	4	
Audio Features	12	8	1	-	3	
Redundancy	2	-	-	-	-	2
Sum	**124**	**105**	**6**	**-**	**11**	**2**

4 Configuration Used During Tests

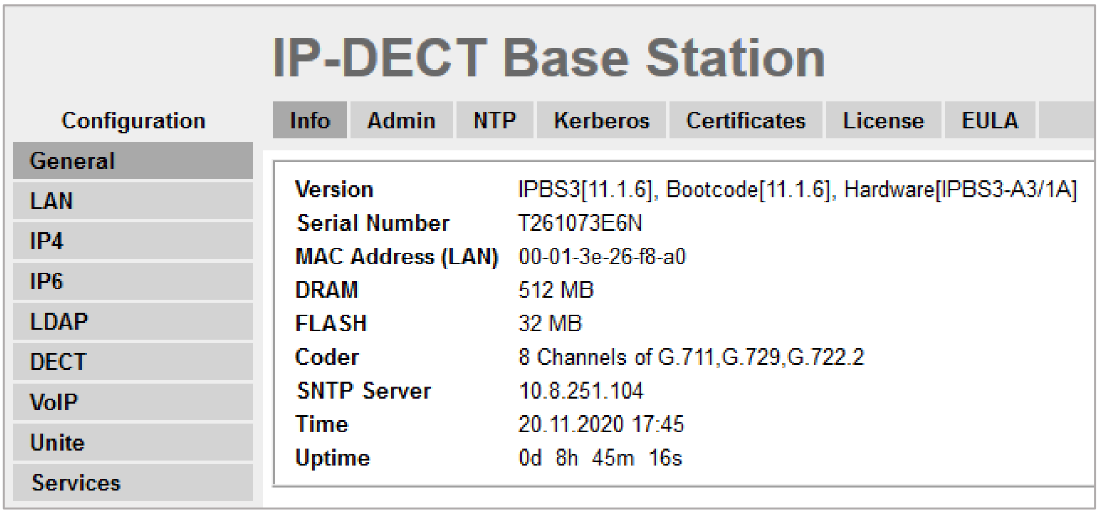
4.1 Ascom IP-DECT Base Stations Configuration

The current section summarizes the example configuration of the Ascom IP-DECT solution for the connection with OSBiz, 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.

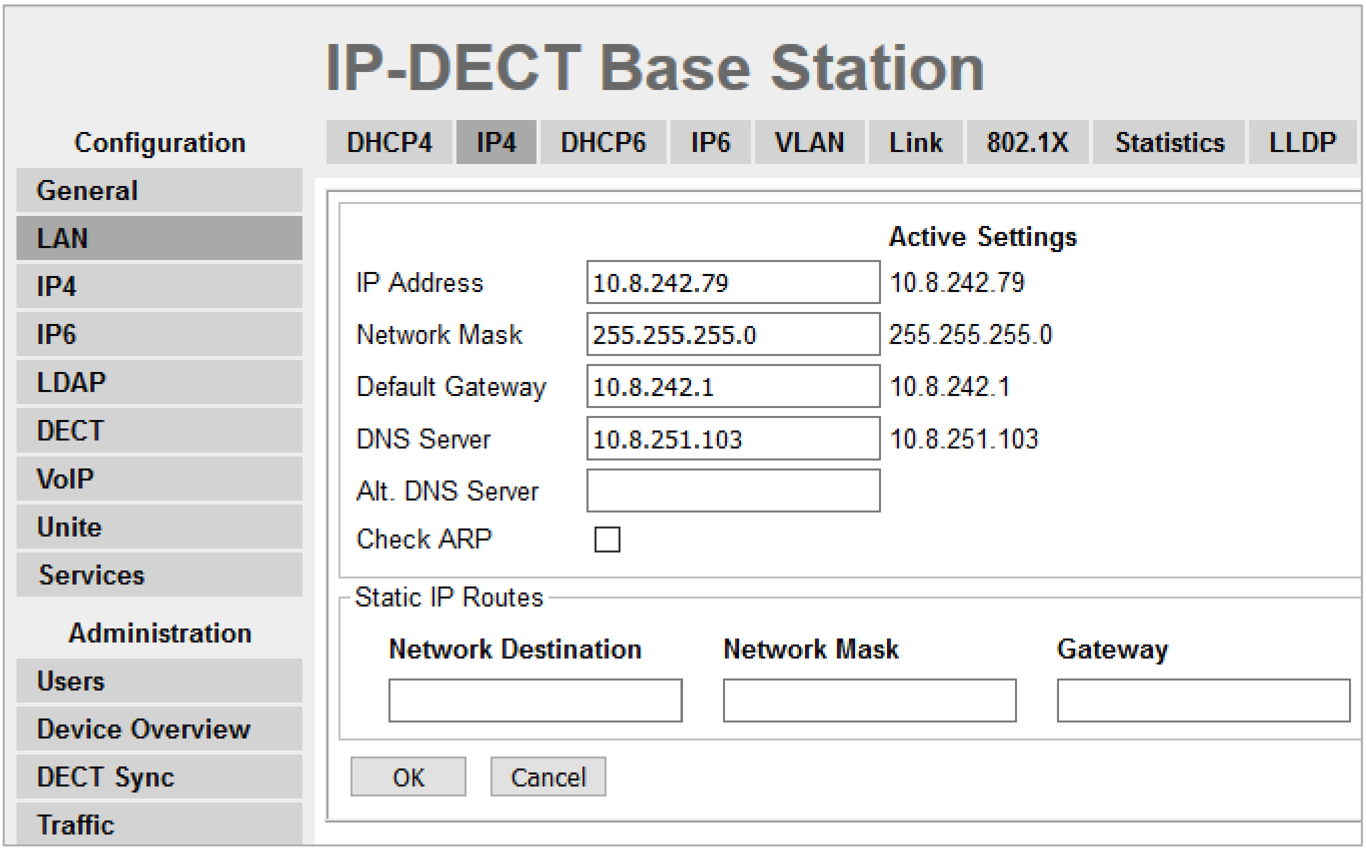
Click the System administration link and enter the appropriate credentials to access the Ascom IP-DECT Base Station and then click on **[OK]**.

4.1.1 *General Configuration of IP-DECT Base Station*



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

4.1.2 *IP-DECT Base Station LAN IP*



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

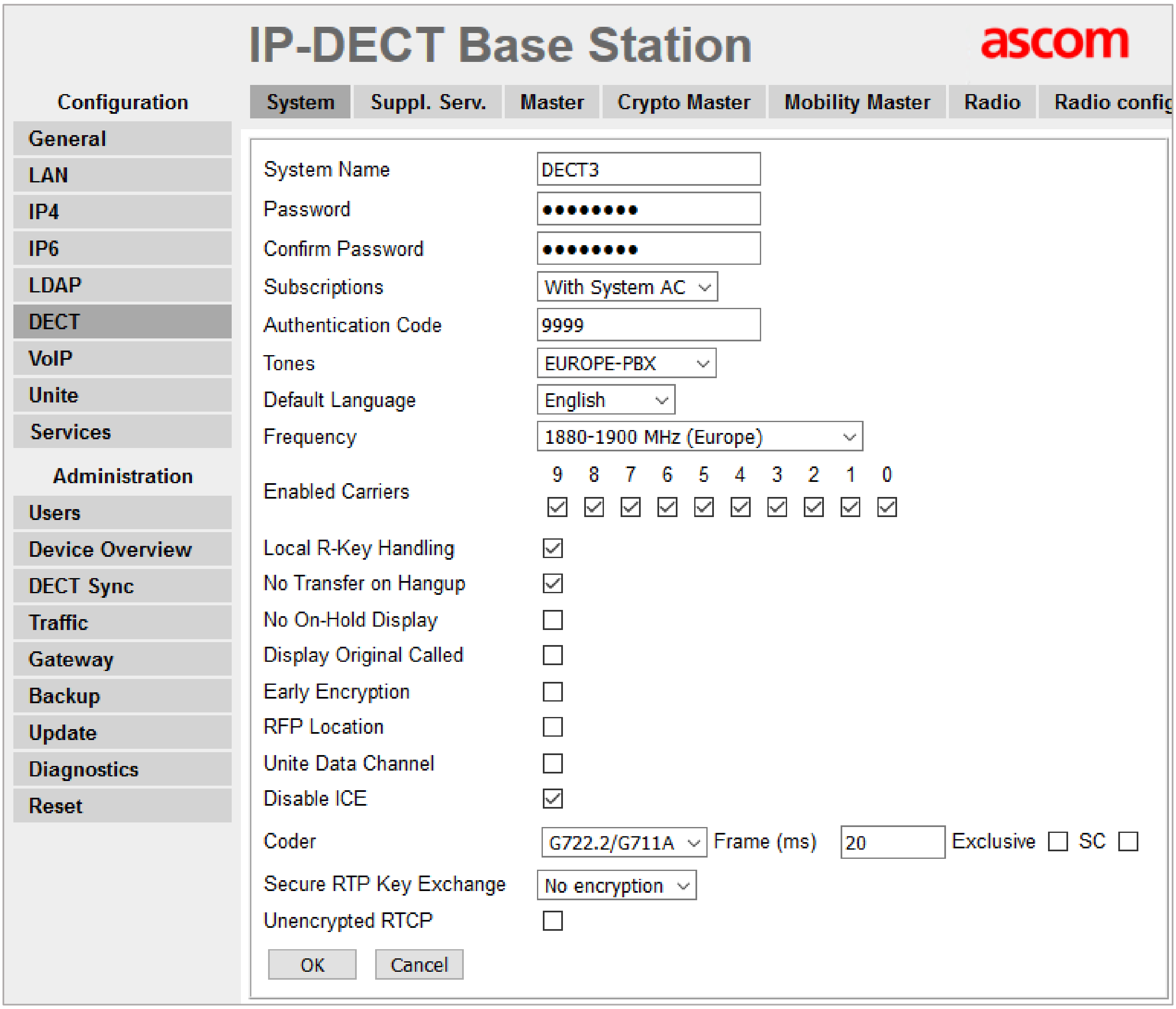
- **IP Address** 10.8.242.79
- **Network Mask** 255.255.255.0

- **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:

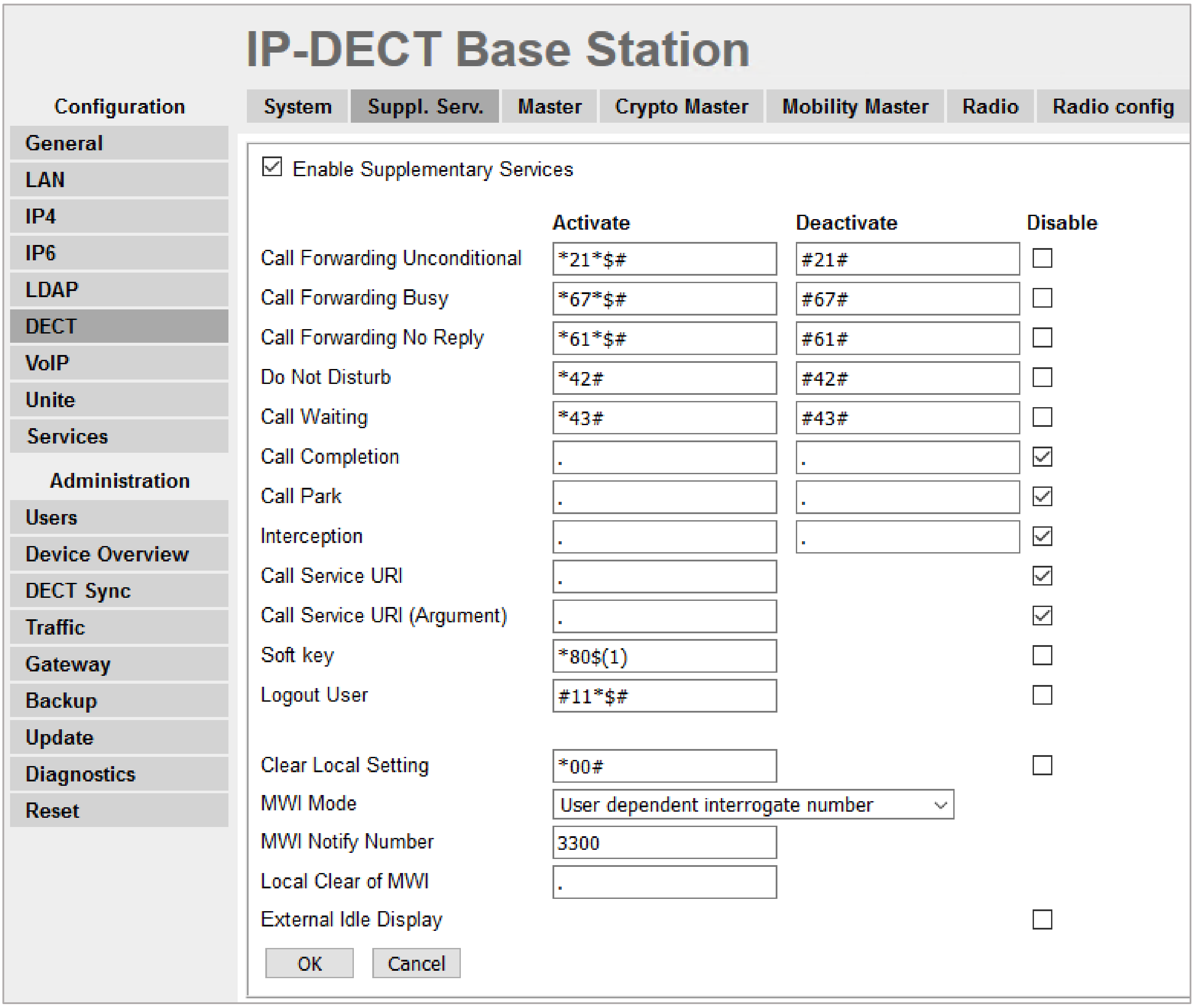
- **System Name** Enter the System Name as previously configured
- **Password** Enter the Password as previously configured
- **Confirm Password** Confirm the Password
- **Subscriptions** Select “With System AC” from the dropdown box

- **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) = 20)

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

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

4.1.4 *IP-DECT Suppl. Serv.*



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

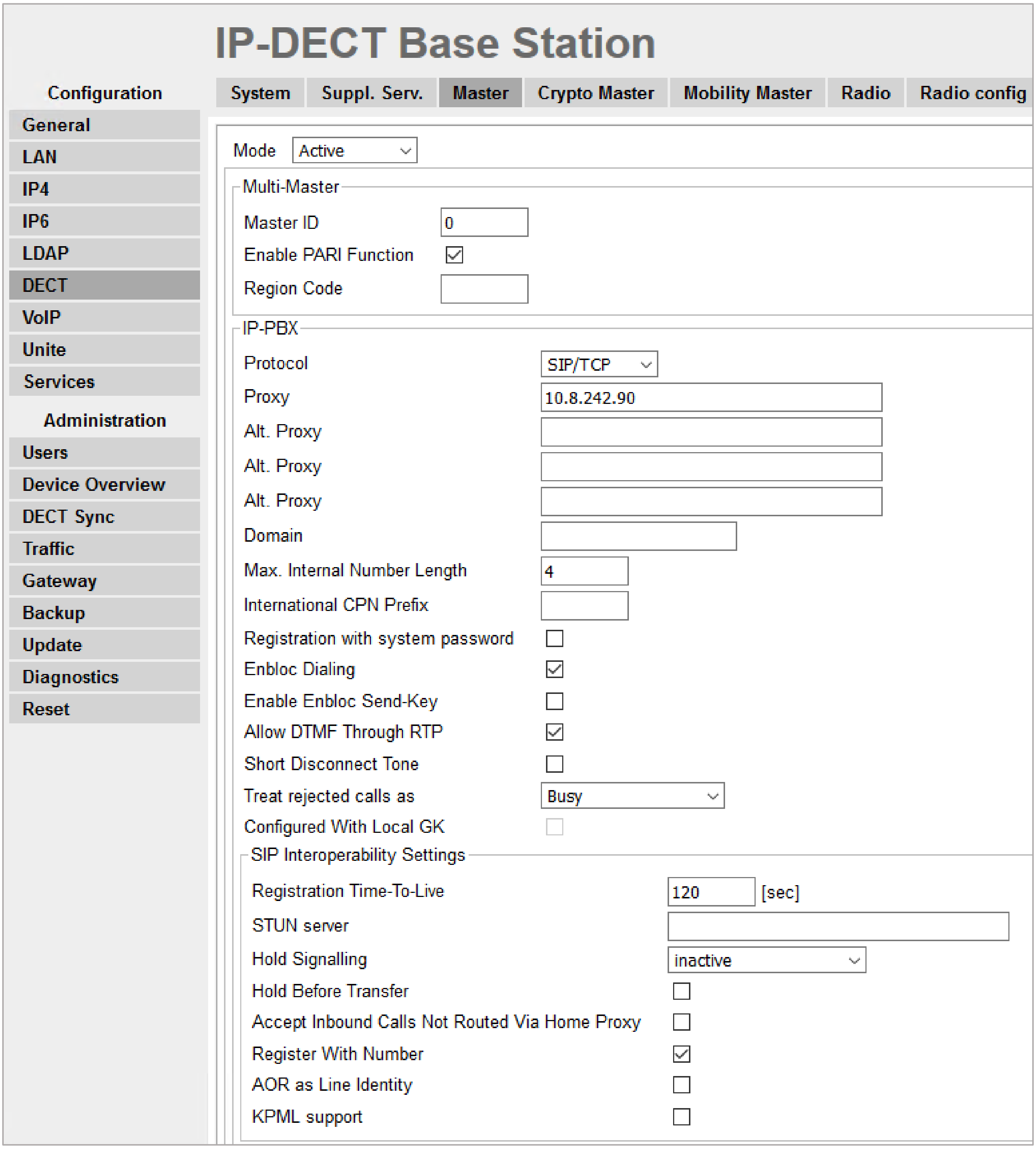
- **Enable Supplementary Services check box** Checked
- **Call Forwarding Unconditional** Activate = *21*$#, Deactivate = #21#
- **Call Forwarding Busy** Activate = *67*$#, Deactivate = #67#
- **Call Forwarding No Reply** Activate = *61*$#, Deactivate = #61#
- **Do not Disturb** Activate = *42*$#, Deactivate = #42#
- **Call waiting** Activate = *43*$#, Deactivate = #43#
- **Soft Key** Activate = *80$(1)
- **Logout User** Activate = #11*$#
- **Clear Local Settings** Activate = *00#
- **MWI Mode** User dependent interrogate number
- **MWI Notify Number** 3300 (voicemail number as configured in OSBiz)

Disable the other handset local features / functions which are either not supported towards OS4000 or not included in the test plan.

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

Note: If the use of digits *80 causes a numbering plan collision, **Soft Key** can also be **disabled**.

4.1.5 *IP-DECT Master*

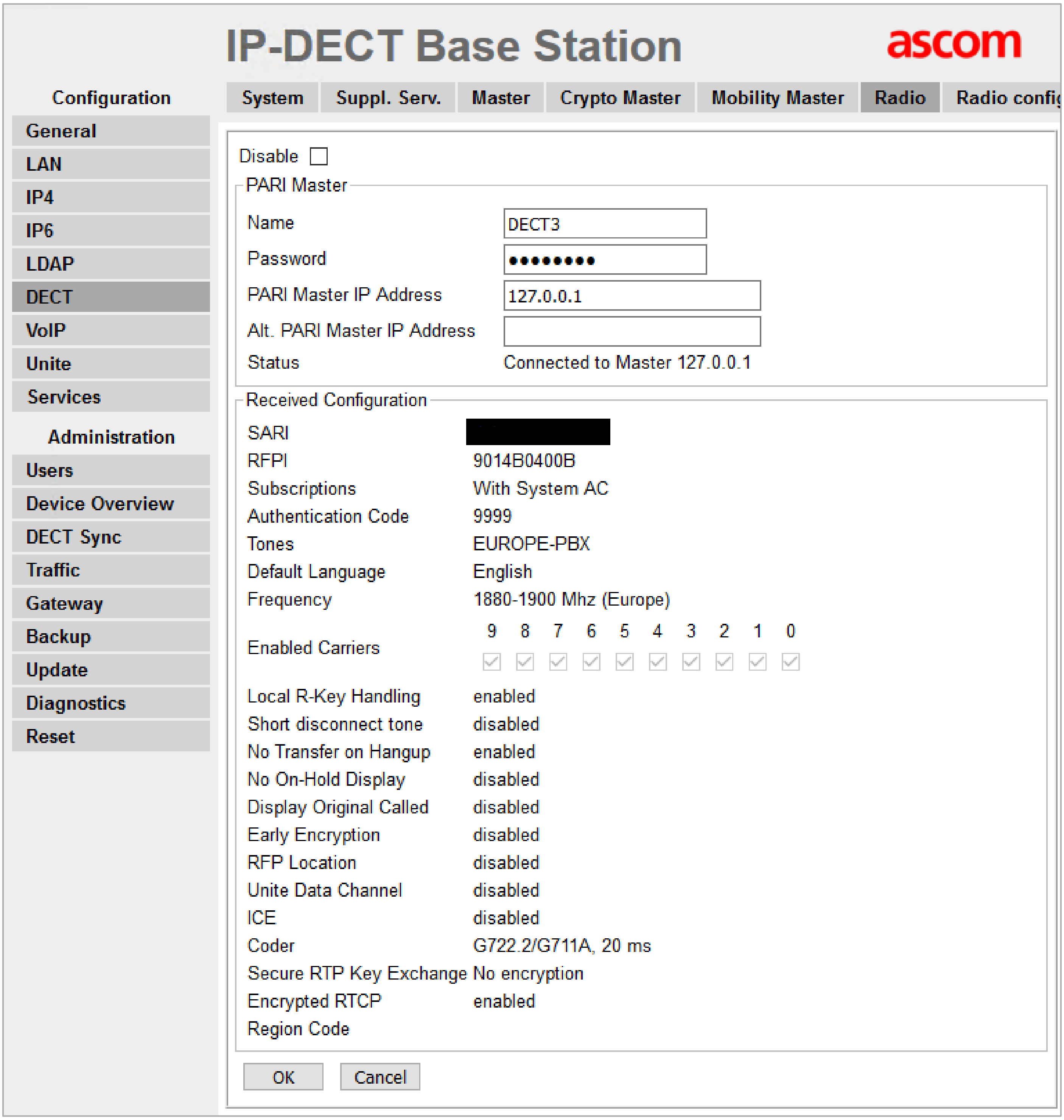


Navigate to **DECT**, click on **Master** and enter the following:

- **Mode** Select Active from the dropdown box
- **Enable PARI Function check box** Checked
- **Protocol** Select SIP/TCP from the dropdown box
- **Proxy** 10.8.242.90 (OSBiz IP)
- **Enbloc Dialing check box** Checked
- **Allow DTMF through RTP check box** Checked
- **Registration Time-To-Live** 120

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

4.1.6 *DECT Radio*

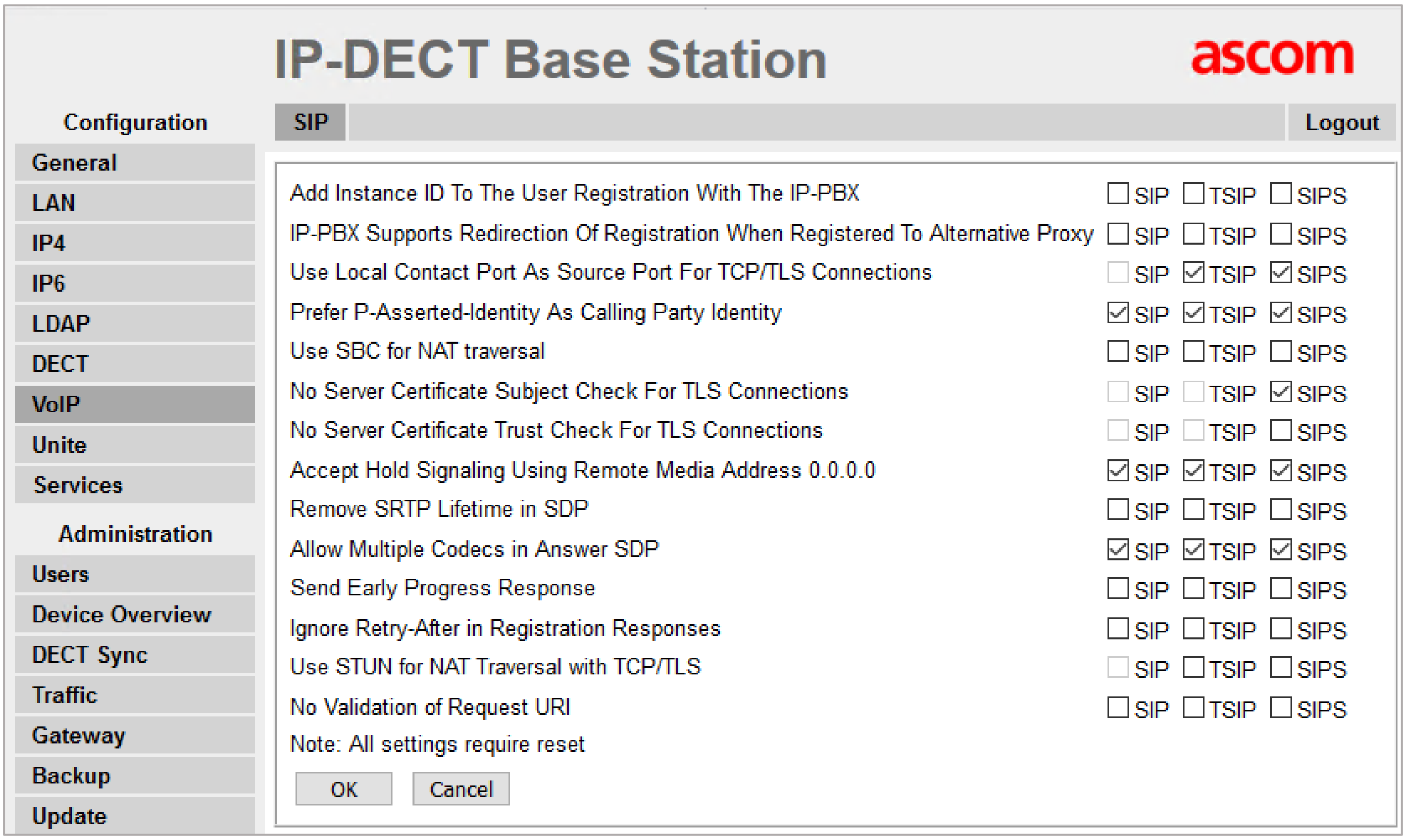


Navigate to **Radio** and select the Admin tab and enter the following:

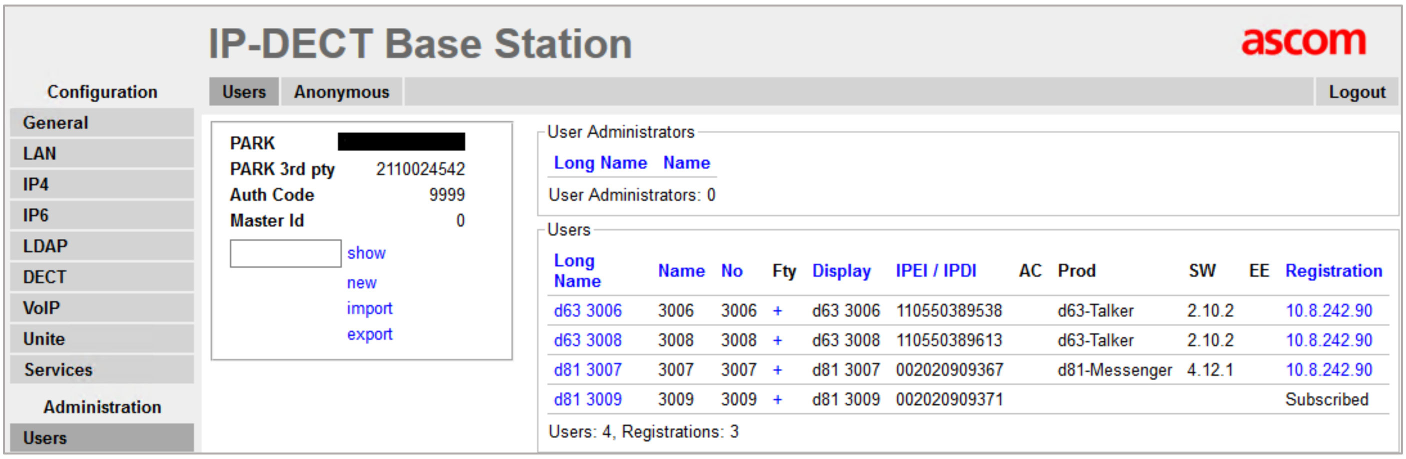
- **Device Name** Enter the name for the PARI Master in the *"Name"* text field
- **Password** Enter the password for the PARI Master in the *"Password"* text field
- **PARI Master IP Address** 127.0.0.1 (single base station)

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

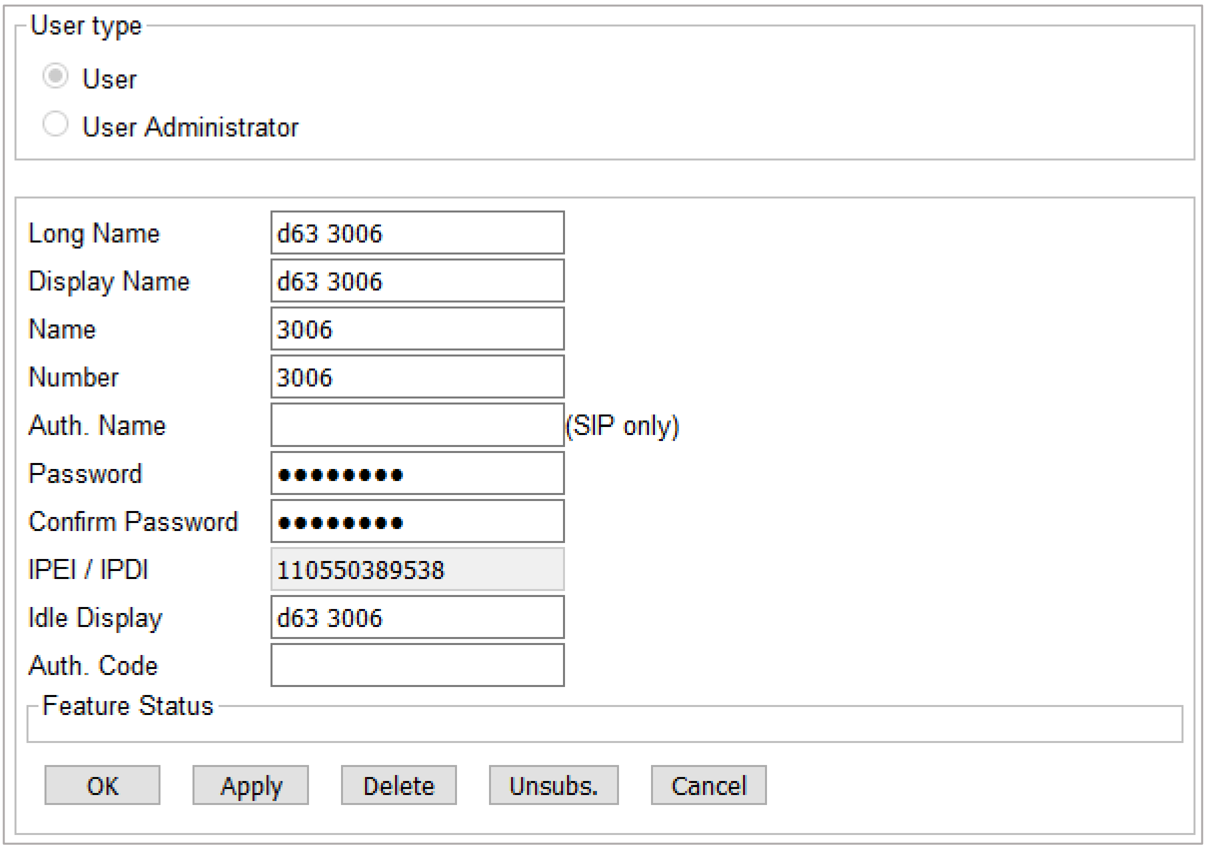
4.1.7 *VoIP*



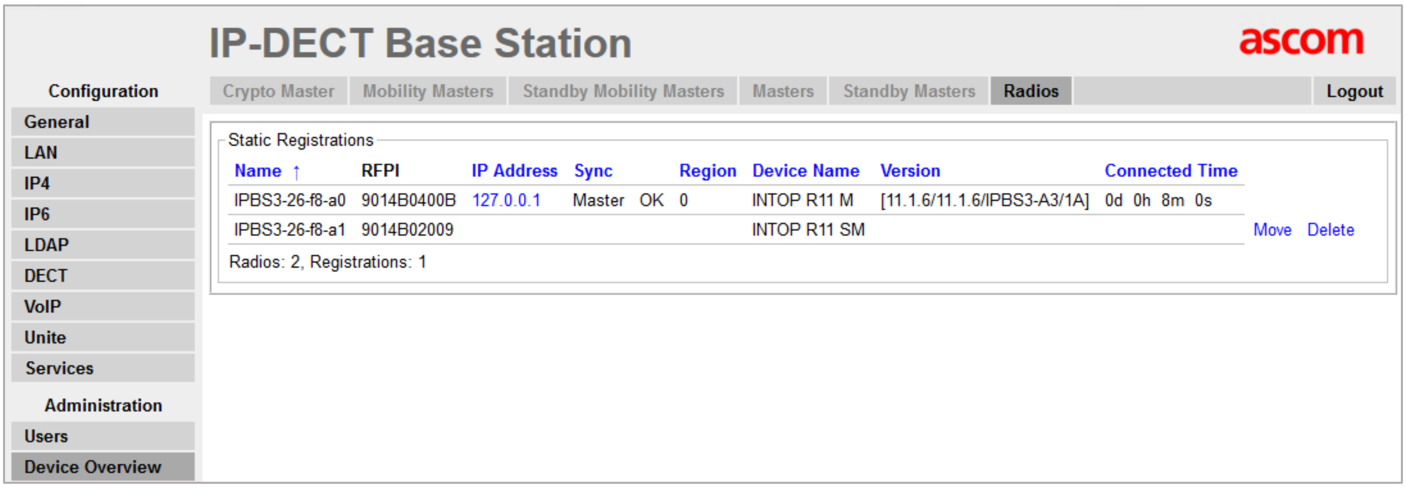
4.1.8 *IP-DECT Users*



Click on e.g. user **d63 3006**



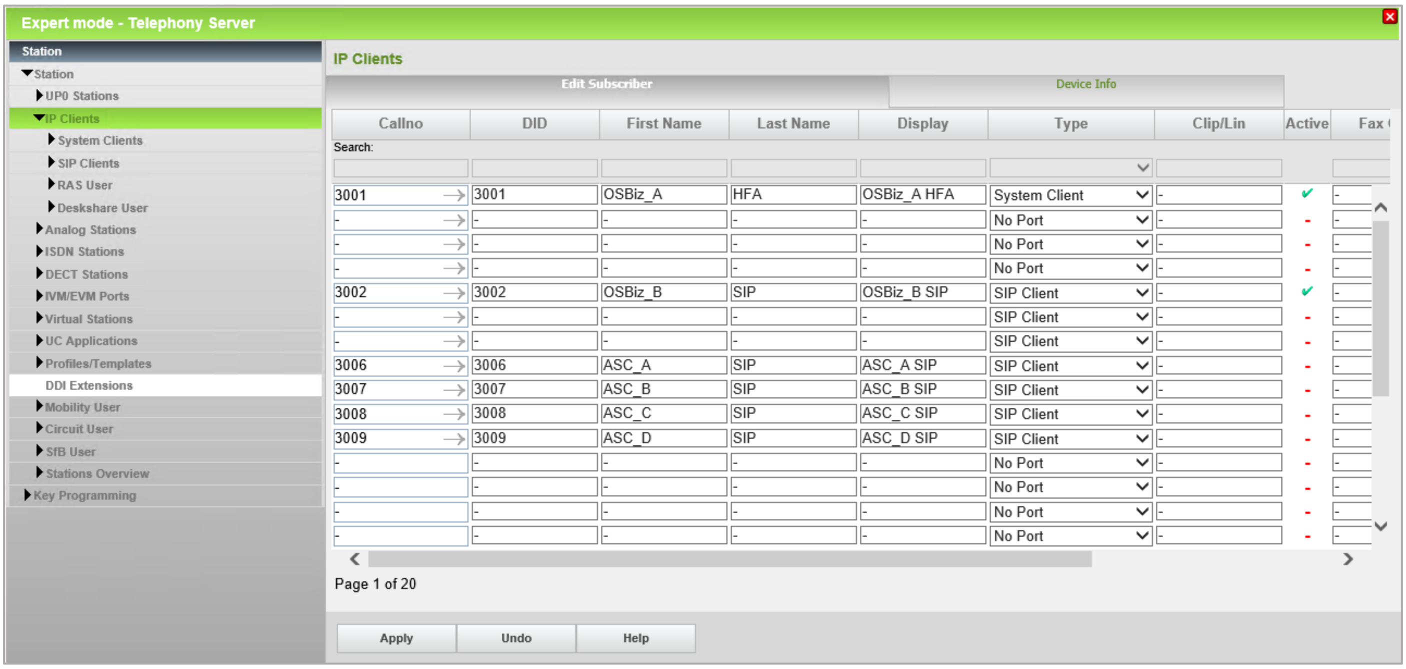
4.1.9 *Device Overview*



4.2 OpenScape Business Configuration

For the needs of the current certification project, certain configuration at OSBiz should be performed. Routine OSBiz configuration, is out of scope for the current project and thus will be omitted.

4.2.1 IP Clients



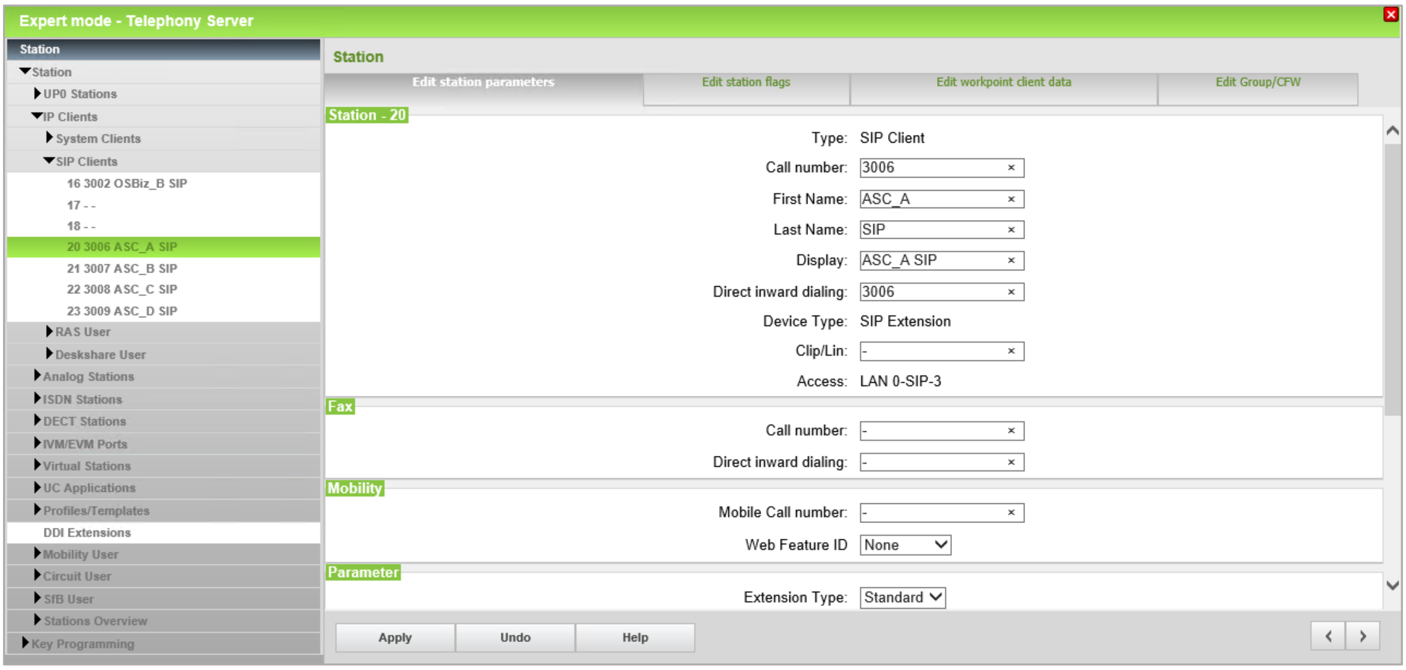
Go to **OSBiz Assistant >> Expert mode >> Telephony Server >> Station >> IP Clients.**

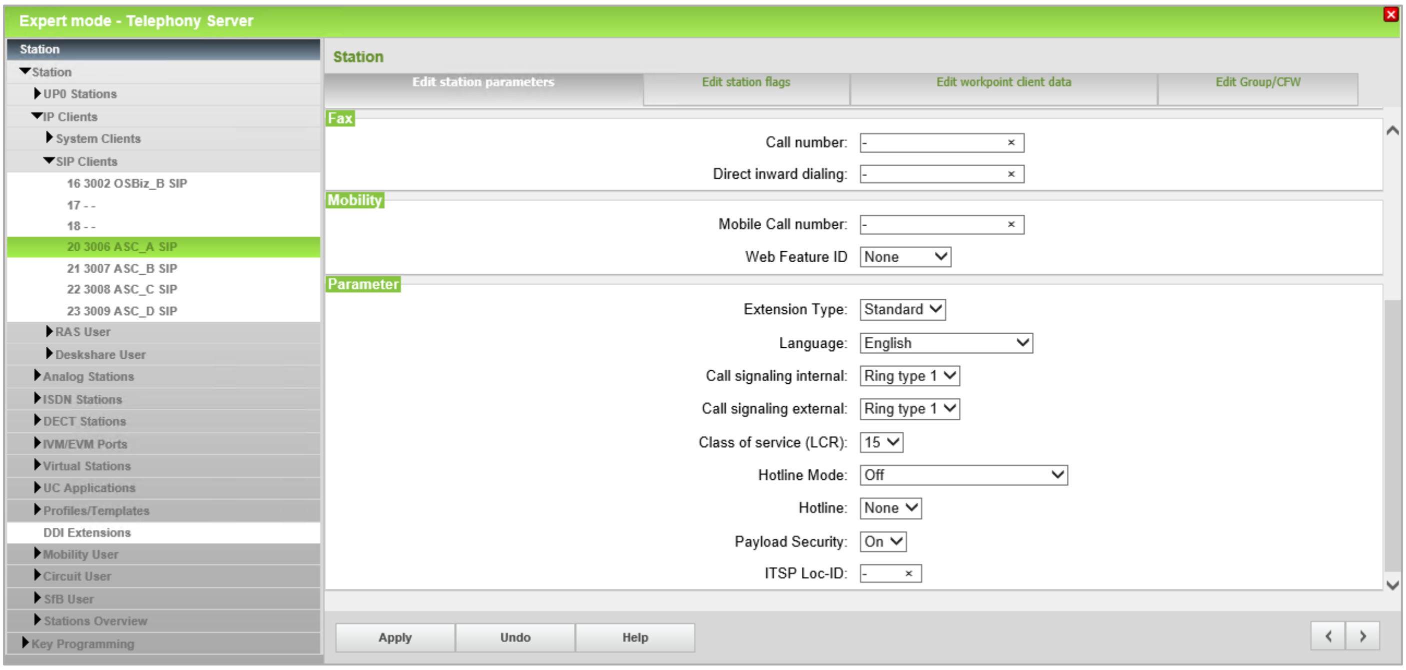
Find an empty line and enter the desired extension numbers, e.g.:

- **Callno:** 3006
- **DID:** 3006 (if incoming calls from PSTN are required to ring on 3006 station then the ISDN DID number is entered, e.g. 2106203360; the DID number formatting depends on the configuration of the ISDN provider in OSBiz, see also 2.2)
- **First Name:** ASC_A (friendly name)
- **Last Name:** SIP (friendly name)
- **Display:** SIP, ASC_A (autocompleted when first name and last name are entered)
- **Type:** SIP Client

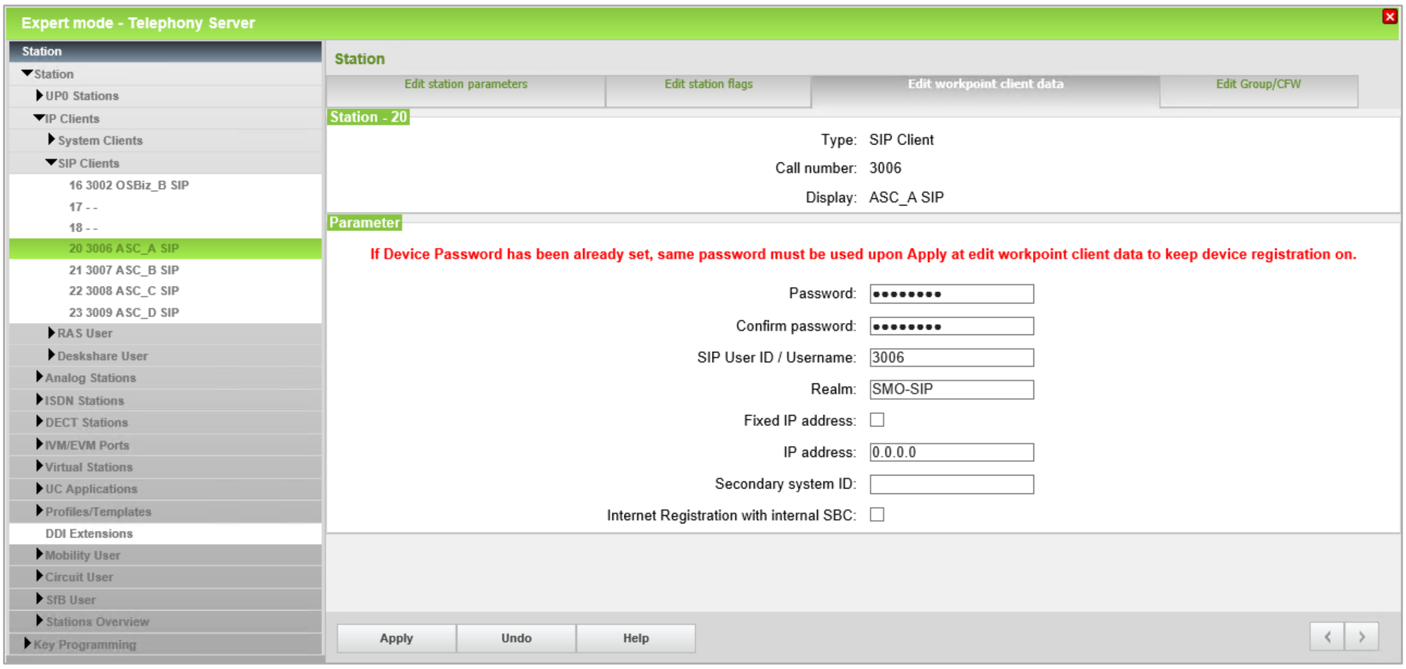
Click on **[Apply]**.

Repeat the same steps for the rest of the test Ascom handset users, i.e. 3007, 3008.





Navigate to **OSBiz Assistant >> Expert mode >> Telephony Server >> Station >> IP Clients >> SIP Clients**. The newly created SIP stations are shown. By clicking on the corresponding client and then clicking on various tabs certain parameters may be configured.



If digest authentication is required for the Ascom handset users to register to OSBiz, click on **"Edit workpoint client data"** and enter the following:

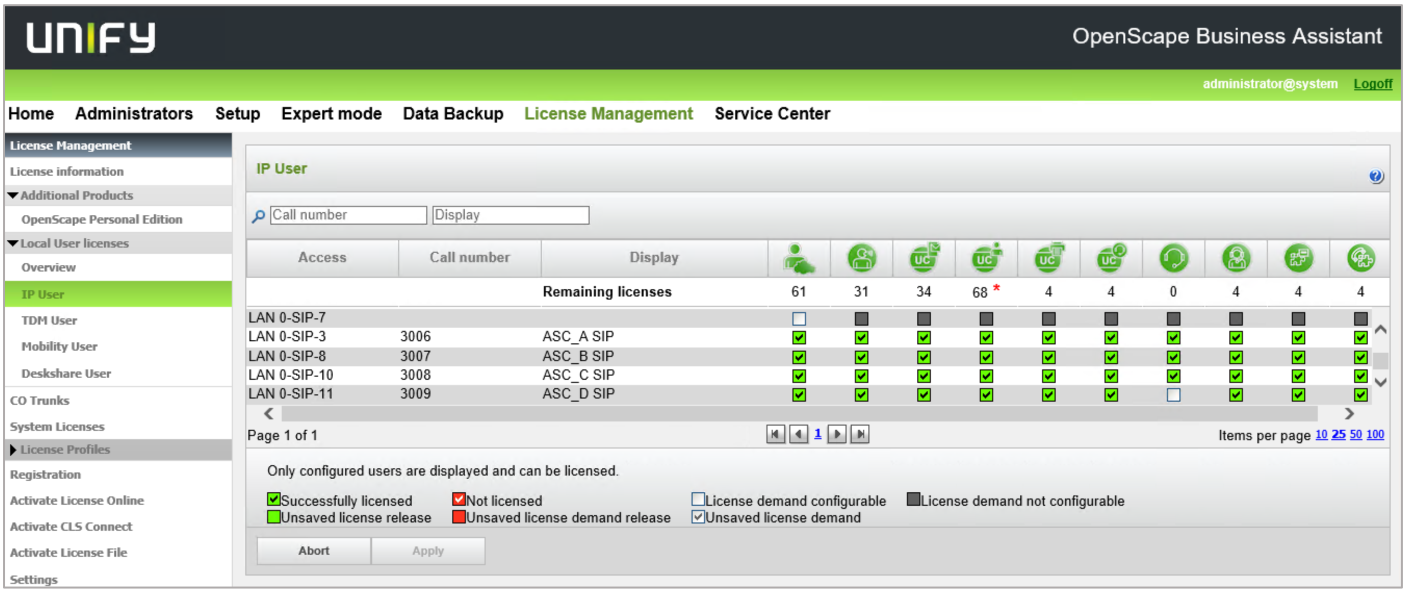
- **Password:** enter an appropriate passphrase.
- **Confirm password:** repeat passphrase.
- **SIP User ID / Username:** 3006 (usually the station number).
- **Realm:** SMO-SIP.

Click on **[Apply]**.

If required, repeat the same steps for the rest (i.e. 3007, 3008).

4.2.2 *IP User Licensing*

After the Gigaset M3 SIP stations are created, the next step is to assign them the proper licensing in order to use OSBiz facilities.



Access the page **OSBiz Assistant >> License Management >> Local User licenses >> IP User** and tick the appropriate check boxes to activate the corresponding functionalities (OpenScape Business IP User is at least required for basic telephony access rights).

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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Ascom AG	Atos Greece SM SA