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 4000 V10R0 for voice calls in accordance with the respective test report, dated January 12th, 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 18th, 2021

Andre Bergmann

Director Technology Partner Program

Atos Unify Ready

Trusted partner for your **Digital Journey**

Atos

Test Report of Certification

IP-DECT R11

with

Atos-Unify OpenScape 4000 V10R0

Status: Released
Release Date: 12/1/2021

Contents:

History of Change

Version	Date	Description	Author(s)
1.0	December 16th, 2020	Initial Creation	Michael Korakis Atos Greece SM SA E-mail: **michail.korakis@atos.net**
1.1	December 22nd, 2020	Editorial changes in sub-sections 1.4.1, 1.4.2, 3.6.1 and 4.1.4	Mathew Williams Ascom AG E-mail: **Matthew.Williams@ascom.com**
1.2	December 28nd, 2020	Editorial change in section 4.1	Michael Korakis
1.3	December 30th, 2020	Editorial change in section 1.4	Michael Korakis
2.0	January 12th, 2021	Review	Dimitrios Galanakis 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 4000 V10R0 PBX.

1.1.1 Basis Equipment

Test Equipment:	OpenScape 4000 Server: Simplex / VM – Integrated SoftGate STMI4 Q2324-X500 vHG3500 Q2330-X SLMO24 Q2168-X STMD3 Q2217-X Virtual OpenScape Xpressions server OpenScape Deskphone CP400 HFA & CP600 SIP and OpenStage 80T phones
Software Release:	OpenScape 4000 version: Platform V10 R0.28.2 Assistant V10 R0.28.2 RMX V10 R0.28.6 CSTA V10 R0.28.0 STMI4 loadware version: pzksti40.A8.020 02 vHG3500 loadware version: pzksgw50.A9.007 STMD3 loadware version: pzdstm30 ComWin version: 5.0.137.0 OpenScape Xpressions version: V7 R1 FR5 HF36 CP600 HFA firmware: V1 R4.2.0 HFA 200323 CP400 SIP firmware: V1 R7.8.0 SIP 200626 OpenStage 80T firmware: V2 R1.15.0 TDM VMware ESXi v5.5.0 Build 6480324

1.1.2 IP-DECT

Certification:	Interoperability testing of various call scenarios and features between Ascom IP-DECT handset SIP subscribers registered to OpenScape 4000 and Atos-Unify HFA, SIP & TDM OS4000 stations. Furthermore, calls between PSTN subscribers and Ascom IP-DECT subscribers are, also, tested. Finally, voice codec verification and fault tolerance 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 Holding AG Zugerstrasse 32 CH-6340 Baar Switzerland Tel. +41 41 544 78 00 https://www.ascom.com/
Description:	Ascom DECT handsets are registered on the OpenScape 4000 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 4000 solution. The wireless communication is made using Ascom IP-DECT Base Station connected to the same LAN as OpenScape 4000. The Ascom IP-DECT Base Station operates as a conduit between the OpenScape 4000 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 4000.
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 4000 as SIP stations. OpenScape 4000 provides the VoIP telephony facilities to Ascom DECT handset SIP stations. Additionally, Atos-Unify HFA, SIP and TDM stations are connected to OpenScape 4000, too. OpenScape 4000 is connected via ISDN board (STMD3) to ISDN BRI provider for the PSTN access (OTE ITSP). Voicemail services are provided from an OpenScape Xpressions server connected with a SIP trunk to OpenScape 4000. 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 Atos-Unify voicemail server was used to allow users to leave voicemail messages and to demonstrate Message Waiting Indication and DTMF on the DECT handsets. 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 OS4000.

- OS4000 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:	November 16th, 2020
Test Duration:	November 17th –20th, 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 4000 V10R0 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 OS4000 system, the setting *"Treat rejected calls as"* should be set to *"Busy"* (refer to page 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** scenarios were tested with both local Ascom DECT handset and OS4000 system features.

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

- A SIP station that participates in an OS4000 **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 4-65).

1.4.2 Restrictions

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

  Outgoing caller-ID restricted calls from an Ascom handset are shown in the call list with the format <CLIR PBX feature code> + <called number> (refer to test 3-10 for more details).

- OS4000 **Do Not Disturb** system feature isn't supported for SIP stations, while local Ascom DECT handset feature is working.
- **Call Completion** (Callback) OS4000 feature isn't currently supported for OS4000 SIP users.
- **Call Park** feature isn't supported on SIP stations by OS4000.
- Ascom DECT handsets don't support the display of **Call Pickup Group notifications** with OS4000.
- 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 4-31, 4-34 & 4-36).
- **Call Deflect** isn't supported on Ascom DECT handsets.
- For SIP terminals, **MWI** feature isn't supported by the OS4000.
- Ascom IP-DECT system doesn't support local **3-party Conference** against OS4000.

- Regarding audio **Codecs** support, OS4000 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*

1.5 *Configuration Block Diagram*

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

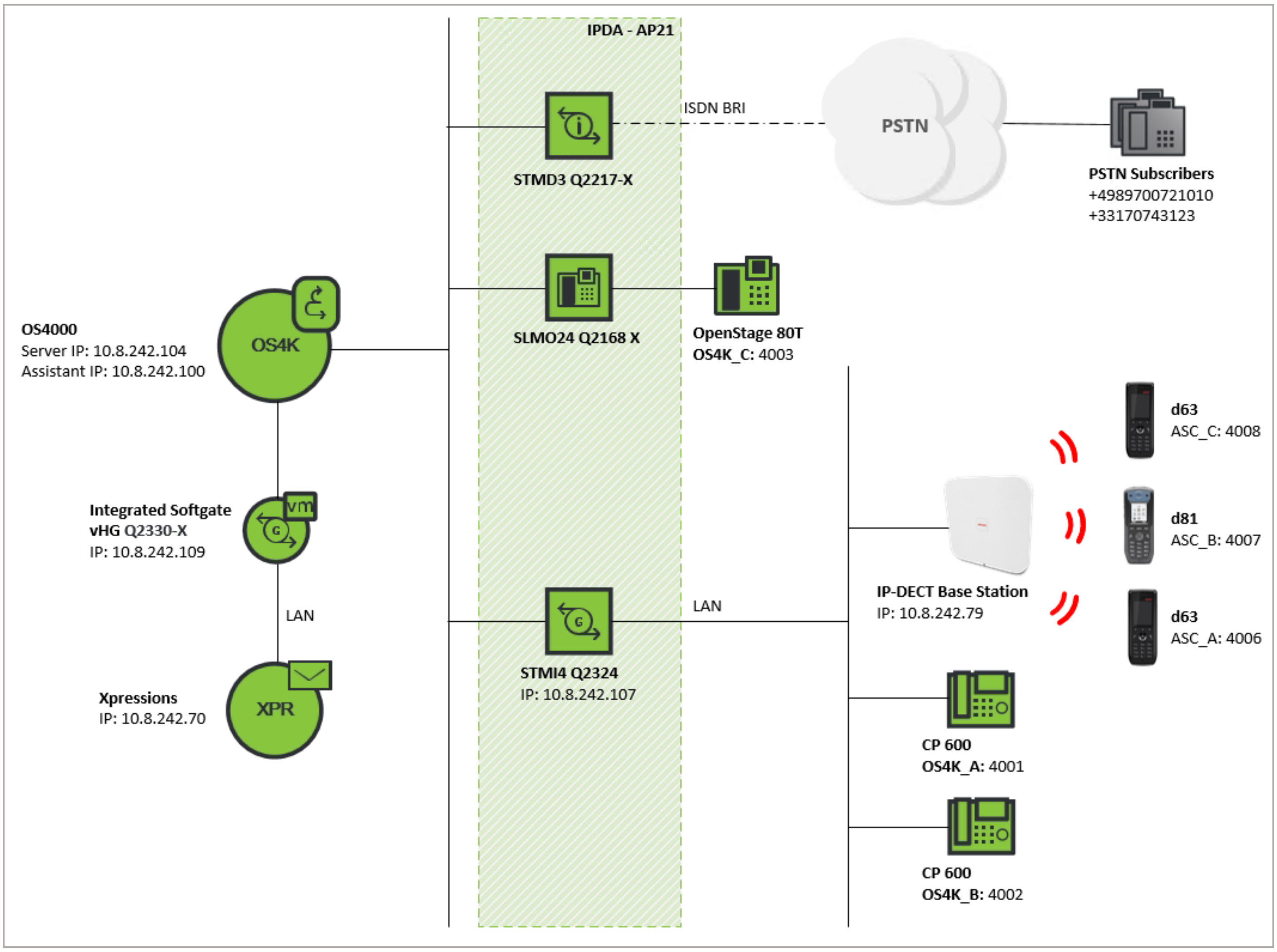


Figure 1: Network Topology Diagram

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 OS4000 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 OS4000 SIP extensions are utilized for Ascom DECT handset SIP stations:

ASC_A = 4006 (d63-Talker handset)

ASC_B = 4007 (d81-Messenger handset)

ASC_C = 4008 (d63-Talker handset).

2.2 Atos-Unify Component Infrastructure

OS4000 is configured for extension dialing between the users. The Office code used is +49 (211) 9598 and the user extensions are using 4 digits (4xxx).

OS4000 is provided with access to PSTN (OTE ISDN BRI provider) via STMD3 board connectivity. OS4000 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 PSTN and then is stripped away by OS4000 configuration before sending the number to PSTN.

In current lab environment there is a single ISDN BRI available to assign to internal OS4000 extensions. Due to that fact in order to reach an internal station from PSTN and execute a specific scenario we have dynamically assigned the public ISDN BRI number to an internal OS4000 extension with e.g.:

```
CHANGE-TDCSU:PEN=1-17-9-0,DGTPR=4001;
```

1-17-9-0 is the STMD3 board PEN and 4001 the extension to redirect the incoming calls from PSTN to an internal extension.

Voicemail services are provided to OS4000 users by an OpenScape Xpressions server connected to OS4000 (vHG board) via SIP trunk. Voicemail callback number is 559 and direct access number is 556.

- OS4000
  - 10.8.242.104 OS4000 Communications server
  - 10.8.242.100 OS4000 Assistant
  - 10.8.242.107 STMI4 Q2324 X500 (IP station registrations)
  - SLMO24 Q2168-X (TDM station connectivity)
  - STMD3 Q2217-X (connection to PSTN)
  - 10.8.242.109 vHG Q2330-X (SIP trunk to Xpressions)
- HFA station (CP600 phone device)
  - OS4K_A = (+49 211 9598) 4001

- SIP station (CP400 phone device)

  OS4K_B = (+49 211 9598) 4002

- TDM station (OS80T phone device)

  OS4K_C = (+49 211 9598) 4003

- OS Xpressions

  10.8.242.70

- PSTN subscribers

  +4989700721010 (landline)

  +33170743123 (landline)

- ISDN BRI (DDI)

  +30 2106203360

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 to OS4000 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 OS4000. 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, call cancelled 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

- Callback No Reply and On Busy

- Call Pickup / Park

- Call Transfer Attended, Semi-Attended and Blind

- Call Deflect

- Call Forwarding Unconditional, Busy and No Reply (both system and local phone features)

- Hunt Group

- Voicemail (DTMF verification) / Message Waiting Indication

- 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. Moreover, the system behavior was observed in codec mismatch scenarios.

Finally, *"Redundancy"* scenarios, focus on the solution behavior in the case of active PBX or Base Station failure and the behavior after the failed components are 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 and a Simplex OS4K systems.

3.1 Registration

3.1.1 OS4000 Registration

No.	Test Procedure	Expected Result	Result	Comments
1-01	• Register Ascom IP-DECT phone to OS4000 without Digest Authentication.	• DECT phone successfully registers with OS4000.	OK	

No.	Test Procedure	Expected Result	Result	Comments
1-02	• Register Ascom IP-DECT phone to OS4000 with Digest Authentication.	• DECT phone successfully registers with OS4000.	OK	
1-03	• Check that Ascom IP-DECT phone may register to OS4000 with either TCP or UDP.	• DECT phone successfully registers with OS4000.	OK	
1-04	• Registration expiration timer at Ascom IP-DECT phone is set below OS4000's corresponding value.	• OS4000 responds with a 423 Interval Too Brief to negotiate the lower re-Registration interval. • Successful registration to OS4000.	OK	Value used on Ascom IP-DECT system was 60secs. The min-expires value suggested by system was 120s.

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	• OS4K_A calls ASC_A. • OS4K_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	• OS4K_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 OS4K_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 OS4K_A. • OS4K_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 directions. • Both ends idle after call clearing.	OK	ASC_A displays external call.
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	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 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. Call clears at about 30 secs.
2-11	• OS4K_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. Call clears at about 15 secs.

No.	Test Procedure	Expected Result	Result	Comments
2-12	• ASC_A calls OS4K_A. • OS4K_A is busy.	• Busy tone/display on calling party. • The call is properly cleared.	OK	
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. 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	• OS4K_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 OS4K_B. • OS4K_C rejects incoming call.	• Busy tone/display on calling party. • The call is properly cleared.	OK	OS4K responds with a 603 Decline. Calls clears at about 15 secs.
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. Call clears at about 3 mins.
2-21	• OS4K_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	IP-DECT base station responds with a 504 Server timeout. Call clears at about 3 mins.
2-22	• ASC_A calls OS4K_A. • OS4K_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	OS4K responds with a 487 Request Terminated. Call clears at about 5 mins.
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	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	• OS4K_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 OS4K_A and hangs up before destination picks up.	• Call is terminated on both sides after hanging up.	OK	
2-28	• PSTN 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 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	ASC_B = switched off, OS4K sends 502 Bad Gateway (Handset displays Hung Up). Call cleared after about 30 secs.
2-31	• ASC_A is unregistered. • OS4K_A calls ASC_A.	• Busy tone/display on calling party. • The call is properly cleared.	OK	
2-32	• OS4K_A is unregistered. • ASC_A calls OS4K_A.	• Busy tone/display on calling party. • The call is properly cleared.	OK	OS4K_A = switched off, OS4K sends 502 Bad Gateway (Handset displays Hung Up). Call cleared after about 30 secs.
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 (Handset displays Temporary Failure and caller gets fast busy for about 5 secs).
2-34	• ASC_A is out of coverage (w/ batt. removed). • OS4K_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 send first 486 BUSY Here and then 503 Service unavailable (OS4K_A displays BUSY and caller gets BUSY for about 12 secs).

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.
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 OS4K_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	OS4K uses SIP UPDATE messages to check if call is alive and all SIP messages got responded correctly from IP-DECT Station.
3-06	• Setup a call between ASC_A and OS4K_A. • The call must last at least 1 hour. • OS4K_A puts the call on mute when call is established. • After 5 mins OS4K_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. *27 - ADND). • 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). OS4K 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. *27 - ADND). • OS4K_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. • OS4K_A calls ASC_A again.	• DND logo disappears from display. • Incoming calls are normally signaled on the phone.	OK	

3.3.4 CLIR

No.	Test Procedure	Expected Result	Result	Comments
3-10	• ASC_A activates Calling Identity Suppression via the CID Suppression feature access code (e.g. *44 - DISUON). • ASC_A calls ASC_B.	• Number and name are not displayed on called device.	OK	ASC_B displays *********** External call. No call is shown in call list. Caller has ASC_B at call list.
3-11	• ASC_A activates Calling Identity Suppression via the CID Suppression feature access code (e.g. *44 - DISUON). • ASC_A calls OS4K_A.	• Number and name are not displayed on called device.	OK	OS4K_A displays NOT AVAILABLE
3-12	• ASC_A activates Calling Identity Suppression via the CID Suppression feature access code (e.g. *44 - DISUON). • ASC_A calls PSTN.	• Number is not displayed on called phone.	OK	When dialing *449<PSTN number>, the called party sees the (ISDN) number of the caller. OS4K sends anonymous in "From" header and "Privacy:id". The OS4K ISDN tracer indicates "Presentation Restricted". It's up to the ISDN provider to have the appropriate configuration to hide or display the number.
3-13	• OS4K_A activates Calling Identity Suppression via the CID Suppression feature access code (e.g. *44 - DISUON). • OS4K_A calls ASC_A.	• Number and name are not displayed on called phone. • Only the dialed digits are displayed on the dialing phone	OK	ASC_A displays *********** External call. No call is shown in call list.
3-14	• PSTN activates identity restriction. • PSTN calls ASC_A.	• Number is not displayed on called phone.	OK	ASC_A displays *********** External call. No call is shown in call list

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.	• On Hold announcement is played to held party. • Held call display on both	OK	On Ascom handset press R for Hold.

No.	Test Procedure	Expected Result	Result	Comments
		parties. • Proper call resume. • No Media delay establishment after resuming the call (media clipping).		ASC_B hears On Hold announcement and displays ON HOLD. ASC_A displays R while call on hold.
4-02	• OS4K_A calls ASC_A and puts the destination on hold.	• On Hold announcement 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	ASC_A displays ON HOLD and hears On Hold announcement.
4-03	• ASC_A calls OS4K_A and puts the destination on hold.	• On Hold announcement 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	OS4K_A displays Held Remotely and hears On Hold announcement. ASC_A displays R.
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. • On Hold announcement 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	• OS4K_A calls ASC_A. • ASC_A makes a consultation call to ASC_B. • ASC_A toggles between calls with OS4K_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. • On Hold announcement 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. OS4K_A displays "Hold Remotely" and hears On Hold announcement.

No.	Test Procedure	Expected Result	Result	Comments
4-06	• ASC_A calls OS4K_A. • OS4K_A makes a consultation call to OS4K_B. • OS4K_A toggles between calls with ASC_A and OS4K_B.	• Ring back tone heard by calling party. • Number presentation on ringing and after call pick up. • No media clipping when connecting both ends. • On Hold announcement 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. • On Hold announcement is played to held party. • Speech path in both directions. • Both ends idle after call clearing.	OK	
4-08	• OS4K_A calls ASC_A. • OS4K_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 OS4K_A and OS4K_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 *Callback (No Reply)*

No.	Test Procedure	Expected Result	Result	Comments
4-09	• ASC_A has the callback feature (ACBK) activated at OS4000. • ASC_A calls ASC_B, but ASC_B doesn't answer. • ASC_A activates callback via phone	• CCNR feature is successfully activated. • No media clipping when connecting both ends. • Speech path in both	NA	Callback isn't supported by OS4K SIP stations.

No.	Test Procedure	Expected Result	Result	Comments
	menu or by dialing the feature access code (e.g., *10). • ASC_B establishes a call with another subscriber and then hangs up. • When ASC_B becomes available, OS4000 calls ASC_A and once ASC_A answers, OS4000 calls ASC_B. • ASC_B answers callback call and connects with ASC_A.	directions. • Both ends idle after call clearing.		
4-10	• OS4K_A has the callback feature (ACBK) activated at OS4000. • OS4K_A calls ASC_A, but ASC_A doesn't answer. • OS4K_A activates callback via phone menu or by dialing the feature access code (e.g., *10). • ASC_A establishes a call with another subscriber and then hangs up. • When ASC_A becomes available, OS4000 calls OS4K_A and once OS4K_A answers, OS4000 calls ASC_A. • ASC_A answers callback call and connects with OS4K_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 (ACBK) activated at OS4000. • ASC_A calls OS4K_A, but OS4K_A doesn't answer. • ASC_A activates callback via phone menu or by dialing the feature access code (e.g., *10). • OS4K_A establishes a call with another subscriber and then hangs up. • When OS4K_A becomes available, after a while, OS4000 calls ASC_A and once ASC_A answers, OS4000 calls OS4K_A. • OS4K_A 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.	NA	Same as 4-09.
4-12	• Repeat scenarios 4-10 & 4-11 but deactivate the callback feature via phone menu or by dialing callback deactivation (DCBK) code (e.g. #10).	• OS4K provides acknowledgement that the callback feature has been cancelled.	NA	

No.	Test Procedure	Expected Result	Result	Comments
		• After the CCNR request has been deactivated, OS4K should not send any notification that the original callback target has become available to receive calls.		

3.4.3 *Callback (On Busy)*

No.	Test Procedure	Expected Result	Result	Comments
4-13	• ASC_A has the feature CCBS activated at OS4000. • ASC_A calls ASC_B, but ASC_B is busy. • ASC_A activates CCBS via phone menu or by dialing access code (e.g., *9). • When ASC_B becomes available, after a while, OS4000 calls ASC_A and once ASC_A answers, OS4000 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.	NA	Same as 4-09.
4-14	• OS4K_A has the feature CCBS activated at OS4000. • OS4K_A calls ASC_A, but ASC_A is busy. • OS4K_A activates CCBS via phone menu or by dialing access code (e.g., *9). • When ASC_A becomes available, after a while, OS4000 calls OS4K_A and once OS4K_A answers, OSV calls ASC_A. • ASC_A answers callback call and connects with OS4K_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 OS4000. • ASC_A calls OS4K_A, but OS4K_A is busy. • ASC_A activates CCBS via phone menu or by dialing the feature access code (e.g., *9). • When OS4K_A becomes available, after a while, OS4000 calls 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.	NA	Same as 4-09.

No.	Test Procedure	Expected Result	Result	Comments
	and once ASC_A answers, OS4000 calls OS4K_A. • OS4K_A answers callback call and connects with ASC_A.			
4-16	• Repeat scenario 4-14 & 4-15, but after called party becomes available, deactivate the callback feature via phone menu or by dialing CCBS deactivation code (e.g. #9).	• CCBS feature is successfully activated. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	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.	• On Hold announcement played on ASC_A while the call is parked. • ASC_C 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 isn't supported by OS4K SIP stations.
4-18	• ASC_A calls OS4K_A. • ASC_A parks the call. • ASC_C retrieves the call from parking lot. • OS4K_A hangs up.	• On Hold announcement played on OS4K_A while the call is parked. • ASC_C 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	Same as 4-17.
4-19	• OS4K_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	• On Hold announcement played on OS4K_A while the call is parked. • ASC_A successfully retrieves parked call. • Number presentation on	NA	Same as 4-17.

No.	Test Procedure	Expected Result	Result	Comments
	answers. • OS4K_A hangs up.	ringing and after call pick up. • Media clipping and speech path after forwarding. • Call is disconnected after hang up.		
4-20	• OS4K_A calls ASC_A. • OS4K_A parks the call. • OS4K_B dials the parking lot number to retrieve the call. • ASC_A hangs up.	• On Hold announcement played on ASC_A while the call is parked. • OS4K_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	Tested with directed call park OS4K feature

3.4.5 *Call Pickup*

No.	Test Procedure	Expected Result	Result	Comments
4-21	• OS4K_B and ASC_B belong to the same Call Pickup group. • ASC_A calls OS4K_B. • ASC_B picks up the call via phone menu or via call pickup access code (e.g. *7) while OS4K_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	No call pickup group notification on Ascom handset. This is not supported towards the OS4K. For OS Voice a special license is required.
4-22	• OS4K_B and ASC_B belong to the same Call Pickup group. • OS4K_A calls ASC_B. • OS4K_B picks up the call via phone menu or via call pick up access code (e.g. *7) while ASC_B is ringing. • OS4K_B hangs up at the end of the communication.	• ASC_B gets a notification • OS4K_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	
4-23	• OSK_A and ASC_A belong to the same Call Pickup group. • PSTN calls OS4K_A. • ASC_A picks up the call via phone menu or via call pick up access code (e.g. *7) while OS4K_A is ringing.	• ASC_A is able to pick up the call. • No media clipping when connecting both ends. • Speech path in both directions.	POK	Same as 4-21.

No.	Test Procedure	Expected Result	Result	Comments
	• ASC_A hangs up at the end of the communication.	• Both ends idle after call clearing.		
4-24	• OS4K_A and ASC_A belong to the same Call Pickup group. • PSTN calls ASC_A. • OS4K_A picks up the call via phone menu or via call pick up access code (e.g. *7) while ASC_A is ringing. • OS4K_A hangs up at the end of the communication.	• OS4K_A is able to pickup the call. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	OK	

3.4.6 *Call Transfer (Attended)*

No.	Test Procedure	Expected Result	Result	Comments
4-25	• 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.	• On Hold announcement 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 R4008 to consult and R4 on Ascom handset to transfer.
4-26	• ASC_A calls OS4K_A. • OS4K_A consults to ASC_B and waits for ASC_B to pick up the call. • OS4K_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.	• On Hold announcement 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	
4-27	• ASC_A calls OS4K_A. • OS4K_A consults to OS4K_B and waits for OS4K_B to pick up the call. • OS4K_A transfers ASC_A towards OS4K_B. • After transfer is completed and communication is established	• On Hold announcement is played to ASC_A while reaching OS4K_B. • No Media clipping and proper speech path after transfer when ASC_A and OS4K_B are connected. • Number presentation on	OK	

No.	Test Procedure	Expected Result	Result	Comments
	between ASC_A and OS4K_B, ASC_A hangs up.	ringing and after call pick up. • Both ends idle after call clearing.		
4-28	• OS4K_A calls ASC_A. • ASC_A puts OS4K_A on hold, calls ASC_B and waits for ASC_B to pick up the call. • ASC_A resumes the call with OS4K_A (ASC_B is placed on hold) and then transfers OS4K_A towards ASC_B. • After transfer is completed and communication is established between OS4K_A and ASC_B, OS4K_A hangs up.	• On Hold announcement is played in all related steps (to OS4K_A while reaching ASC_B / to ASC_B while resuming the call with OS4K_A). • Proper call resume when ASC_A toggles between calls with OS4K_A and ASC_B. • No Media clipping and proper speech path after transfer when OS4K_A and ASC_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	
4-29	• PSTN calls OS4K_A. • OS4K_A consults to ASC_B and waits for ASC_B to pick up the call. • OS4K_A transfers PSTN towards ASC_B. • After transfer is completed and communication is established between PSTN and ASC_B, PSTN hangs up.	• On Hold announcement 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	
4-30	• PSTN calls ASC_A. • ASC_A puts PSTN on hold, calls OS4K_B and waits for OS4K_B to pick up the call. • ASC_A resumes the call with PSTN (OS4K_B is placed on hold) and then transfers PSTN towards OS4K_B. • After transfer is completed and communication is established between PSTN and OS4K_B, PSTN hangs up.	• On Hold announcement is played in all related steps (to PSTN while reaching OS4K_B / to OS4K_B while resuming the call with PSTN). • Proper call resume when ASC_A toggles between calls with PSTN and OS4K_B. • No Media clipping and proper speech path after transfer when PSTN and OS4K_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.		

3.4.7 *Call Transfer (Semi-Attended)*

No.	Test Procedure	Expected Result	Result	Comments
4-31	• 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.	• On Hold announcement 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. This is DECT limitation. Press R4 on ASC_B during ASC_C ringing due to attribute No Transfer on Hang up on IP-DECT base station.
4-32	• ASC_A calls OS4K_A. • OS4K_A consults to ASC_B. • OS4K_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.	• On Hold announcement is played to ASC_A after consultation and before OS4K_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	
4-33	• ASC_A calls OS4K_A. • OS4K_A consults to OS4K_B. • OS4K_A hangs up while OS4K_B is ringing. • ASC_A receives ring back tone. • After communication is established between ASC_A and OS4K_B, ASC_A hangs up.	• On Hold announcement is played to ASC_A after consultation and before OS4K_A hangs up. • No Media clipping and proper speech path after transfer when ASC_A and OS4K_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	ASC_A doesn't receive ringback tone while transfer is in progress but continues to hear On Hold announcement until OS4K_B picks up the call.
4-34	• OS4K_A calls ASC_A. • ASC_A consults to ASC_B.	• On Hold announcement is played to OS4K_A after	POK	Same as 4-31.

No.	Test Procedure	Expected Result	Result	Comments
	• ASC_A hangs up while ASC_B is ringing. • OS4K_A receives ring back tone. • After communication is established between OS4K_A and ASC_B, OS4K_A hangs up.	consultation and before ASC_A hangs up. • No Media clipping and proper speech path after transfer when OS4K_A and ASC_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.		
4-35	• PSTN calls OS4K_A. • OS4K_A consults to ASC_B. • OS4K_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.	• On Hold announcement 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	
4-36	• PSTN calls ASC_A. • ASC_A consults to OS4K_B. • ASC_A hangs up while OS4K_B is ringing. • PSTN receives ring back tone. • After communication is established between PSTN and OS4K_B, PSTN hangs up.	• On Hold announcement is played to PSTN after consultation and before ASC_A hangs up. • No Media clipping and proper speech path after transfer when PSTN and OS4K_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	POK	Same as 4-31.

3.4.8 *Call Transfer (Blind)*

No.	Test Procedure	Expected Result	Result	Comments
4-37	• 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	• On Hold announcement to ASC_A after consultation and before blind transfer is completed. • No Media clipping and proper speech path after transfer when ASC_A and	OK	Press RR4008# on Ascom handset.

No.	Test Procedure	Expected Result	Result	Comments
	between ASC_A and ASC_C, ASC_A hangs up.	ASC_C are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.		
4-38	• ASC_A calls OS4K_B. • OS4K_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.	• On Hold announcement 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	ASC_A does not receive ringback tone while transfer is in Progress; it plays On Hold announcement.
4-39	• ASC_A calls OS4K_B. • OS4K_B blind transfers to OS4K_C. • ASC_A receives ring back tone. • After transfer is completed and communication is established between ASC_A and OS4K_C, ASC_A hangs up.	• On Hold announcement to ASC_A after consultation and before blind transfer is completed. • No Media clipping and proper speech path after transfer when ASC_A and OS4K_C are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	Same as 4-38.
4-40	• OS4K_A calls ASC_A. • ASC_A blind transfers to ASC_B. • OS4K_A receives ring back tone. • After transfer is completed and communication is established between OS4K_A and ASC_B, OS4K_A hangs up.	• On Hold announcement to OS4K_A after consultation and before blind transfer is completed. • No Media clipping and proper speech path after transfer when OS4K_A and ASC_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	
4-41	• PSTN calls OS4K_B. • OS4K_B blind transfers to ASC_B. • PSTN receives ring back tone. • After transfer is completed and	• On Hold announcement to PSTN after consultation and before blind transfer is completed.	OK	

No.	Test Procedure	Expected Result	Result	Comments
	communication is established between PSTN and ASC_B, PSTN 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.		
4-42	• PSTN calls ASC_A. • ASC_A blind transfers to OS4K_B. • PSTN receives ring back tone. • After transfer is completed and communication is established between PSTN and OS4K_B, PSTN hangs up.	• On Hold announcement to PSTN after consultation and before blind transfer is completed. • No Media clipping and proper speech path after transfer when PSTN and OS4K_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	

3.4.9 *Call Deflect*

No.	Test Procedure	Expected Result	Result	Comments
4-43	• ASC_A calls ASC_B. • ASC_B doesn't answer and deflects the call to AS_C. • 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.	• 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.	NA	Feature not supported on Ascom DECT handsets.
4-44	• ASC_A calls OS4K_B. • OS4K_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.	OK	

No.	Test Procedure	Expected Result	Result	Comments
		• Both ends idle after call clearing.		

3.4.10 *Call Forwarding (Unconditional)*

No.	Test Procedure	Expected Result	Result	Comments
4-45	• ASC_A calls ASC_B who has CFU activated via phone locally or via CFU feature access code (*24 - AFFWDVCE). • 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*4008# to activate on Ascom handset (deactivate with #21#).
4-46	• OS4K_A calls ASC_A who has CFU activated via phone locally or via CFU feature access code (*24 - AFFWDVCE). • ASC_A forwards the call immediately to OS4K_B. • Leave OS4K_B ring for a few seconds and pick up the call. • After forward is completed and communication is established between OS4K_A and OS4K_B, OS4K_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	• ASC_A calls OS4K_A who has CFU activated via phone locally or via CFU feature access code (*24 - AFFWDVCE). • OS4K_A forwards the call immediately to OS4K_B. • Leave OS4K_B ring for a few seconds and pick up the call. • After forward is completed and communication is established between OS4K_A and OS4K_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.	OK	

No.	Test Procedure	Expected Result	Result	Comments
		• Both ends idle after call clearing.		
4-48	• PSTN calls ASC_A who has CFU activated via phone locally or via CFU feature access code (*24 - AFFWDVCE). • 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	

3.4.11 *Call Forwarding (Busy)*

No.	Test Procedure	Expected Result	Result	Comments
4-49	• ASC_A calls busy subscriber ASC_B who has CFB activated via phone locally or via CFB feature access code (*28). • 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 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 *67*4008# to activate on Ascom handset (deactivate with #67#).
4-50	• OS4K_A calls busy subscriber ASC_A who has CFB activated via phone locally or via CFB feature access code (*28 - AFWDB). • ASC_A doesn't have call waiting activated and forwards the call immediately to OS4K_B. • Leave OS4K_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. • Media clipping and speech path after forwarding.	OK	

No.	Test Procedure	Expected Result	Result	Comments
	between OS4K_A and OS4K_B, OS4K_A hangs up.	• Both ends idle after call clearing.		
4-51	• ASC_A calls busy subscriber OS4K_A who has CFB activated via phone locally or via CFB feature access code (*28 - AFWDB). • OS4K_A doesn't have call waiting activated and forwards the call immediately to OS4K_B. • Leave OS4K_B ring for a few seconds and pick up the call. • After forward is completed and communication is established between ASC_A and OS4K_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	
4-52	• PSTN calls busy subscriber ASC_A who has CFB activated via phone locally or via CFB feature access code (*28 - AFWDB). • 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	

3.4.12 *Call Forwarding (No Reply)*

No.	Test Procedure	Expected Result	Result	Comments
4-53	• ASC_A calls ASC_B who has CFNR activated via phone locally or via CFNR feature access code (*29 - AFWDNA). • 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.	OK	Press *61*4008# activate on DECT handset (deactivate with #61#).

No.	Test Procedure	Expected Result	Result	Comments
		• Both ends idle after call clearing.		
4-54	• OS4K_A calls ASC_A who has CFNR activated via phone locally or via CFNR feature access code (*29). • ASC_A rings and after a while forwards to OS4K_B. • Leave OS4K_B ring for a few seconds and pick up the call. • After forward is completed and communication is established between OS4K_A and OS4K_B, OS4K_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	
4-55	• ASC_A calls OS4K_A who has CFNR activated via phone locally or via CFNR feature access code (*29 - AFWDNA). • OS4K_A rings and after a while forwards to OS4K_B. • Leave OS4K_B ring for a few seconds and pick up the call. • After forward is completed and communication is established between ASC_A and OS4K_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	
4-56	• PSTN calls ASC_A who has CFNR activated via phone locally or via CFNR feature access code (*29 - AFWDNA). • ASC_A rings and after a while forwards 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 *Hunting Group*

No.	Test Procedure	Expected Result	Result	Comments
4-57	• MLHG (Multi Line Hunt Group) is configured with ASC_A, ASC_B and ASC_C. • OS4K_A calls MLHG pilot DN. • 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	
4-58	• MLHG (Multi Line Hunt Group) is configured with ASC_A, ASC_B and ASC_C. • OS4K_A calls MLHG pilot DN. • 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	
4-59	• MLHG (Multi Line Hunt Group) is configured with ASC_A, ASC_B and ASC_C. • PSTN calls MLHG pilot DN. • 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	

3.4.14 *Voicemail*

No.	Test Procedure	Expected Result	Result	Comments
4-60	• OS4K_A and OS4K_B call ASC_A who has call forwarding to voicemail system (Xpressions). • OS4K_A and OS4K_B leave a voicemail message to ASC_A. • ASC_A calls voicemail system to hear unread messages. • ASC_A calls back CP_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.	NA	MWI isn't supported for OS4K SIP stations.

No.	Test Procedure	Expected Result	Result	Comments
		• Both ends idle after call clearing.		
4-61	• After scenario 4-44, 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.	NA	Same as 4-60.
4-62	• 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	

3.4.15 *Conference*

No.	Test Procedure	Expected Result	Result	Comments
4-63	• ASC_A calls OS4K_A and establishes communication. • ASC_A initiates a consultation call to OS4K_B. • After OS4K_B answers, ASC_A initiates a conference. • ASC_A calls ASC_B and ASC_A adds ASC_B to the 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	Ascom IP-DECT system doesn't support 3-party conference against OS4000.
4-64	• OS4K_A calls ASC_A and establishes communication. • OS4K_B calls ASC_A who has call waiting activated. • ASC_A merges calls with OS4K_A and OS4K_B to a conference. • ASC_B calls ASC_A and ASC_A adds ASC_B to the 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 4-63.

No.	Test Procedure	Expected Result	Result	Comments
4-65	• OS4K_A calls ASC_A and establishes communication. • OS4K_A initiates a consultation call to ASC_B. • After ASC_B answers, OS4K_A initiates a conference. • OS4K_A calls OS4K_B and OS4K_A adds OS4K_B to the conference. • OS4K_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 OS4K_A hangs up. • Number(s) presentation after ASC_A hangs up on conference participants.	POK	ASC_A and ASC_B don't display a conference call, but a call to OS4K_A. The SIP server is not aware of the HFA conference and so cannot tell the SIP_A that its call is in a conference.

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 OS4K_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 OS4K_A.	• Connection is established with G.723. • Media clipping and speechpath.	NA	G.722 is not available codec on IP-DECT station
5-03	• IP-DECT base station is configured to have G.729 as first priority. • ASC_A calls OS4K_A.	• Connection is established with G.729. • Media clipping and speechpath.	OK	
5-04	• ASC_A is configured to support G.711 only. • OS4K_A calls ASC_A.	• Connection is established with G.711. • Media clipping and speechpath.	OK	
5-05	• ASC_A is configured to support G.722 only. • OS4K_A calls ASC_A.	• Connection is established with G.723. • Media clipping and speechpath.	NA	G.722 is not available codec on IP-DECT station
5-06	• ASC_A is configured to support G.729 only. • OS4K_A calls ASC_A.	• Connection is established with G.729.	OK	

No.	Test Procedure	Expected Result	Result	Comments
		• Media clipping and speechpath.		
5-07	• 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.	POK	Communication falls back to G.711, because G.722.2 is screened by the system. OS4K doesn't support G.722.2.
5-08	• Make sure there is codec mismatch between ASC_A and OS4K_A. • OS4K_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 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 OS4K_A . • ASC_A calls OS4K_A.	• Call is disconnected. • Call failure indication.	OK	488 Not Acceptable Here from OS4K.

3.6 Redundancy

3.6.1 OS4000 Node Switchover

No.	Test Procedure	Expected Result	Result	Comments
6-01	• Perform a node switchover from node1 to node2 of the OS4000 by putting out of service node1. • Perform an incoming and outgoing call with ASC_A.	• The switchover should not be seen on the IP-DECT handsets. • ASC_A is able to make and receive calls.	NT	Test environment is Simplex.
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.	NT	Same as 6-01.

No.	Test Procedure	Expected Result	Result	Comments
		• ASC_A is able to make and receive calls.		

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.	NT	Single Base Station is used for testing activities.
6-04	• In mirror configuration the function 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.	NT	Same as 6-03.

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	29	29	-	-	-	-
Basic Calls Extended	14	14	-	-	-	-
Telephony Features	65	46	6	-	13	
Audio Features	9	5	2	-	2	
Redundancy	4	-	-	NT	-	-
Sum	**125**	**98**	**8**	**4**	**15**	**-**

4 Configuration Used During Tests

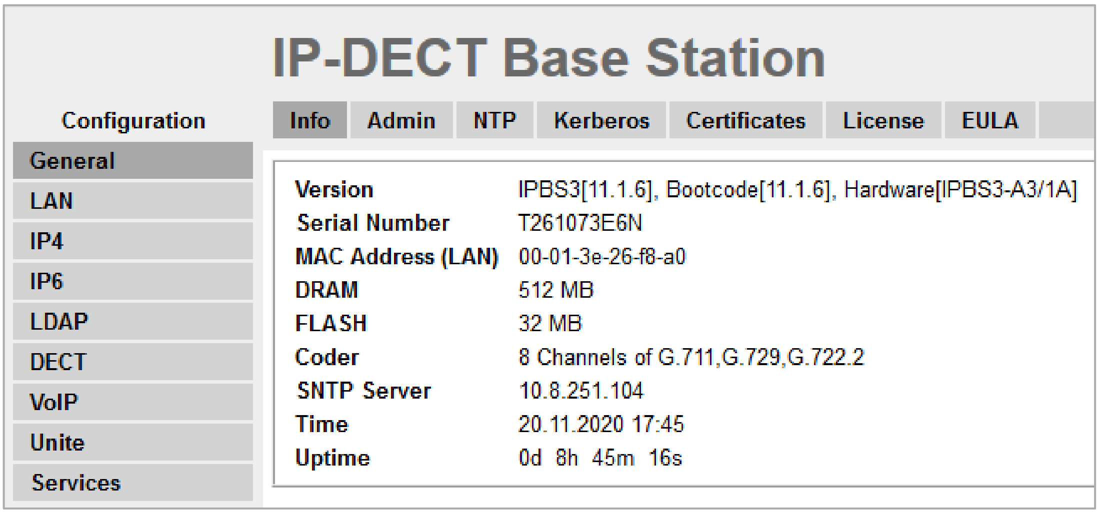
4.1 Ascom IP-DECT Base Stations Configuration

The current section summarizes the example configuration of the Ascom IP-DECT solution for the connection with OS4000, 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.

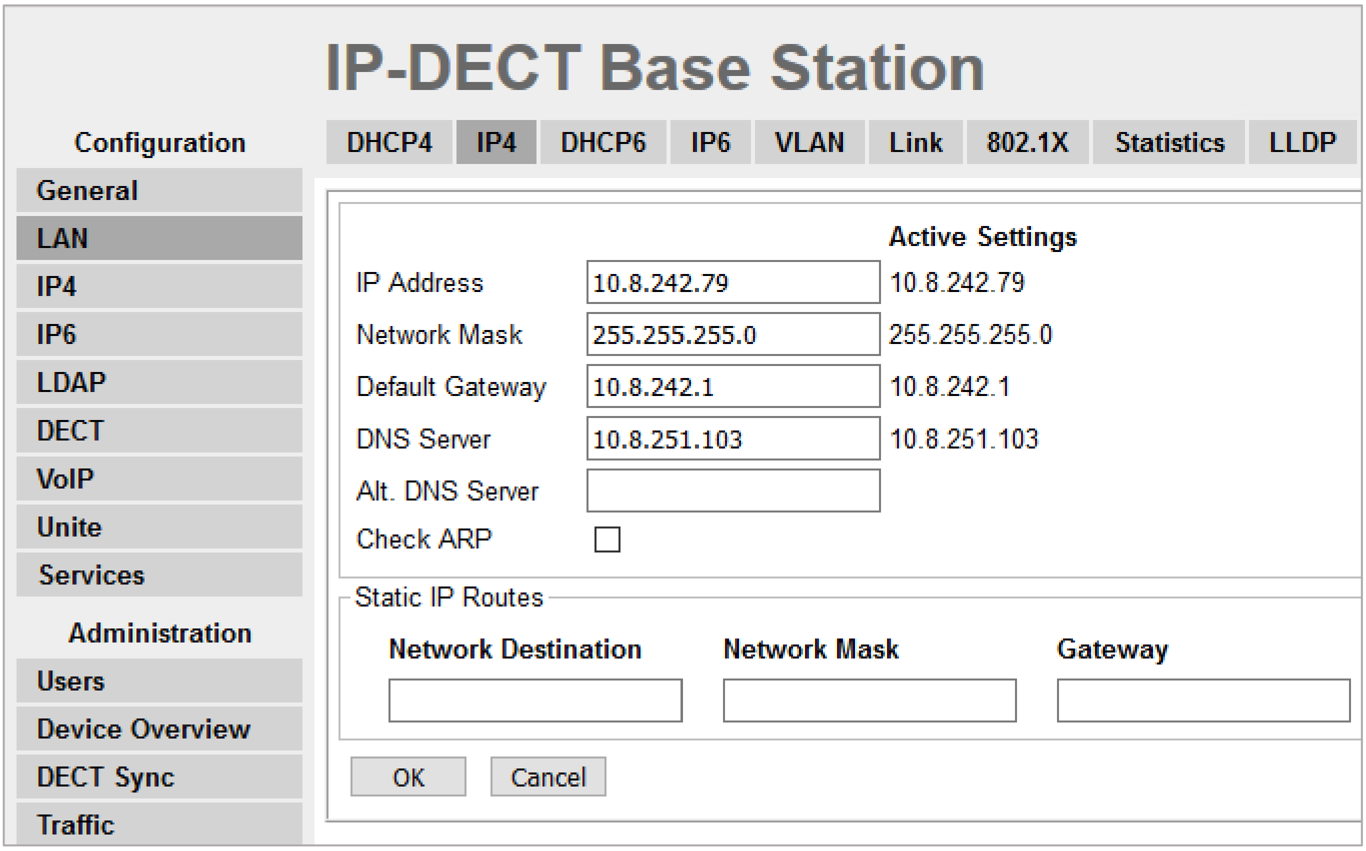
Click the System administration link and enter the appropriate credentials to access the Ascom IP-DECT Base Station and then click on **[OK]**.

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:

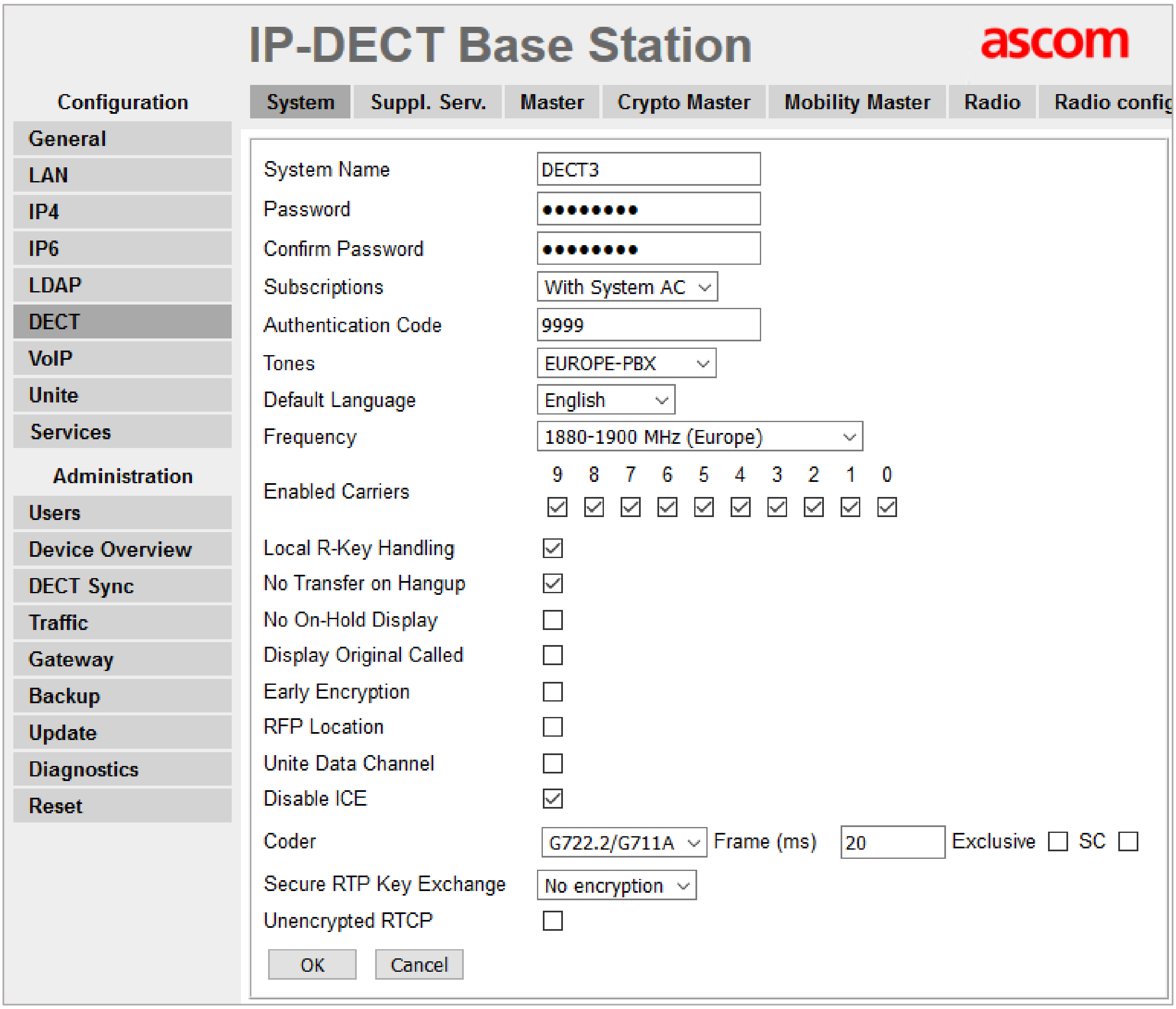
- **IP Address** 10.8.242.79
- **Network Mask** 255.255.255.0

- **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:

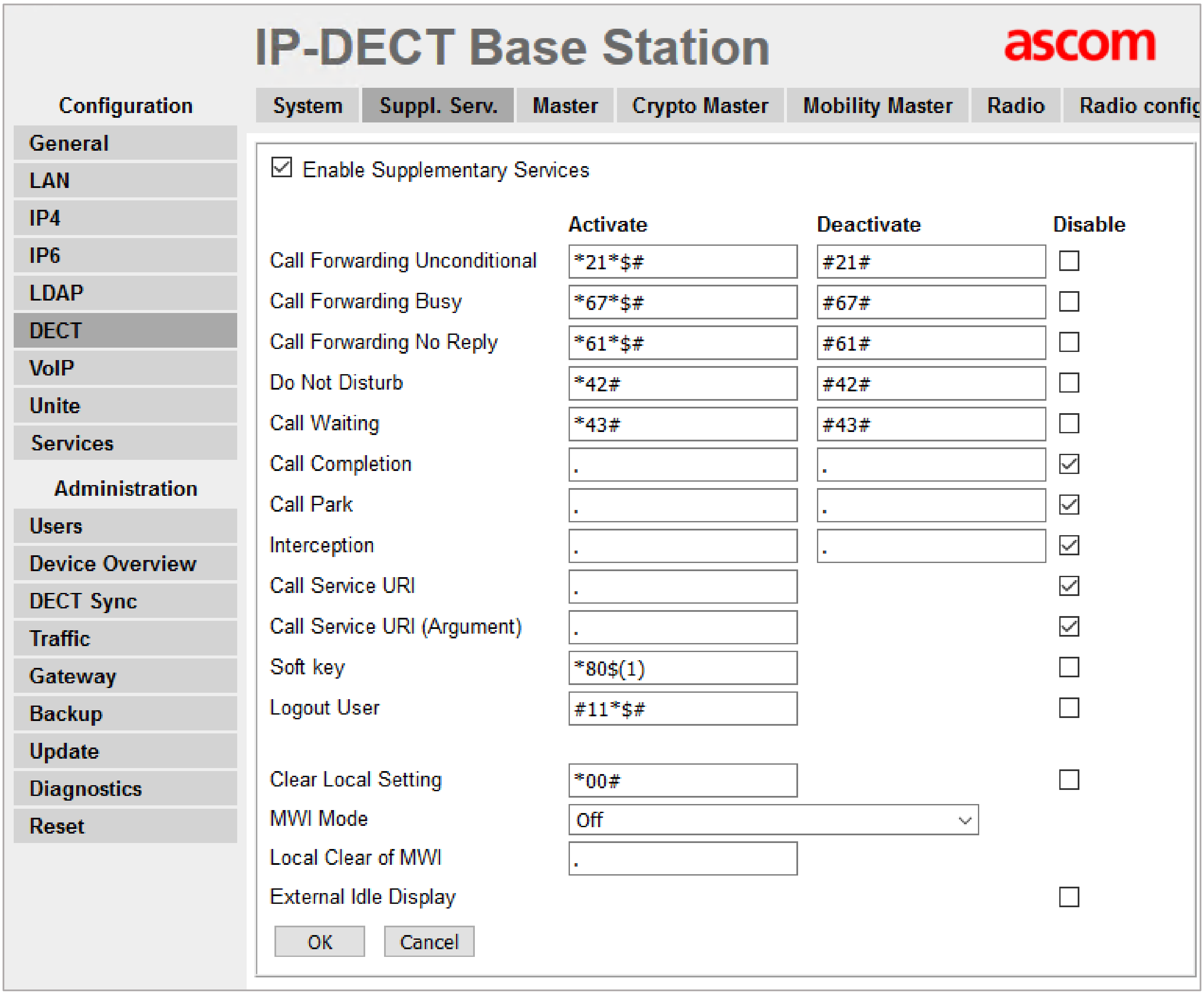
- **System Name** Enter the System Name as previously configured
- **Password** Enter the Password as previously configured
- **Confirm Password** Confirm the Password
- **Subscriptions** Select "With System AC" from the dropdown box

- **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) = 30)

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

Note: It is recommended that the PBX (HG board) 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:

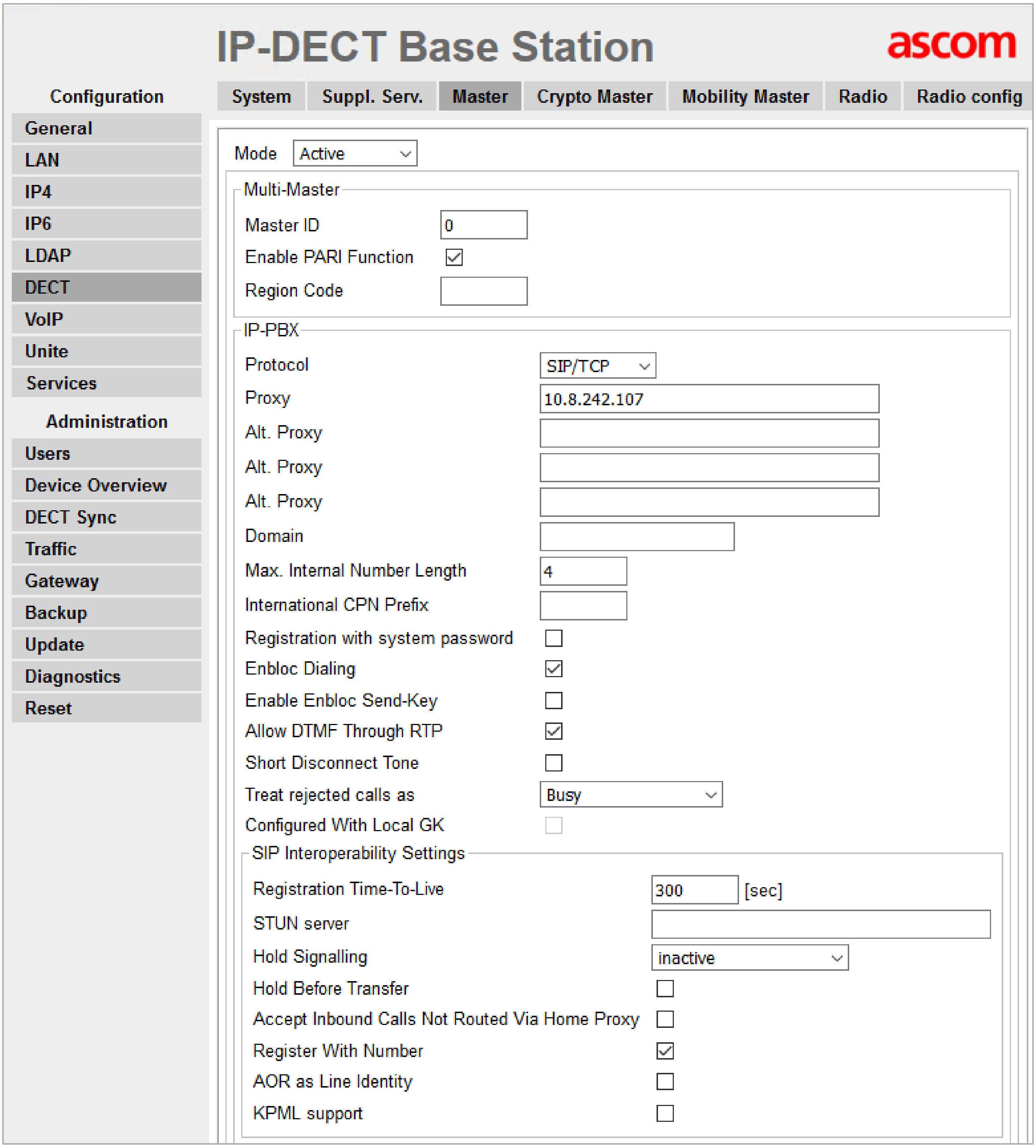
- **Enable Supplementary Services check box** Checked
- **Call Forwarding Unconditional** Activate = *21*$#, Deactivate = #21#
- **Call Forwarding Busy** Activate = *67*$#, Deactivate = #67#
- **Call Forwarding No Reply** Activate = *61*$#, Deactivate = #61#
- **Do not Disturb** Activate = *42*$#, Deactivate = #42#
- **Call waiting** Activate = *43*$#, Deactivate = #43#
- **Soft Key** Activate = *80$(1)
- **Logout User** Activate = #11*$#
- **Clear Local Settings** Activate = *00#
- **MWI Mode** Off (MWI isn't supported on SIP terminals by OS4000)

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*

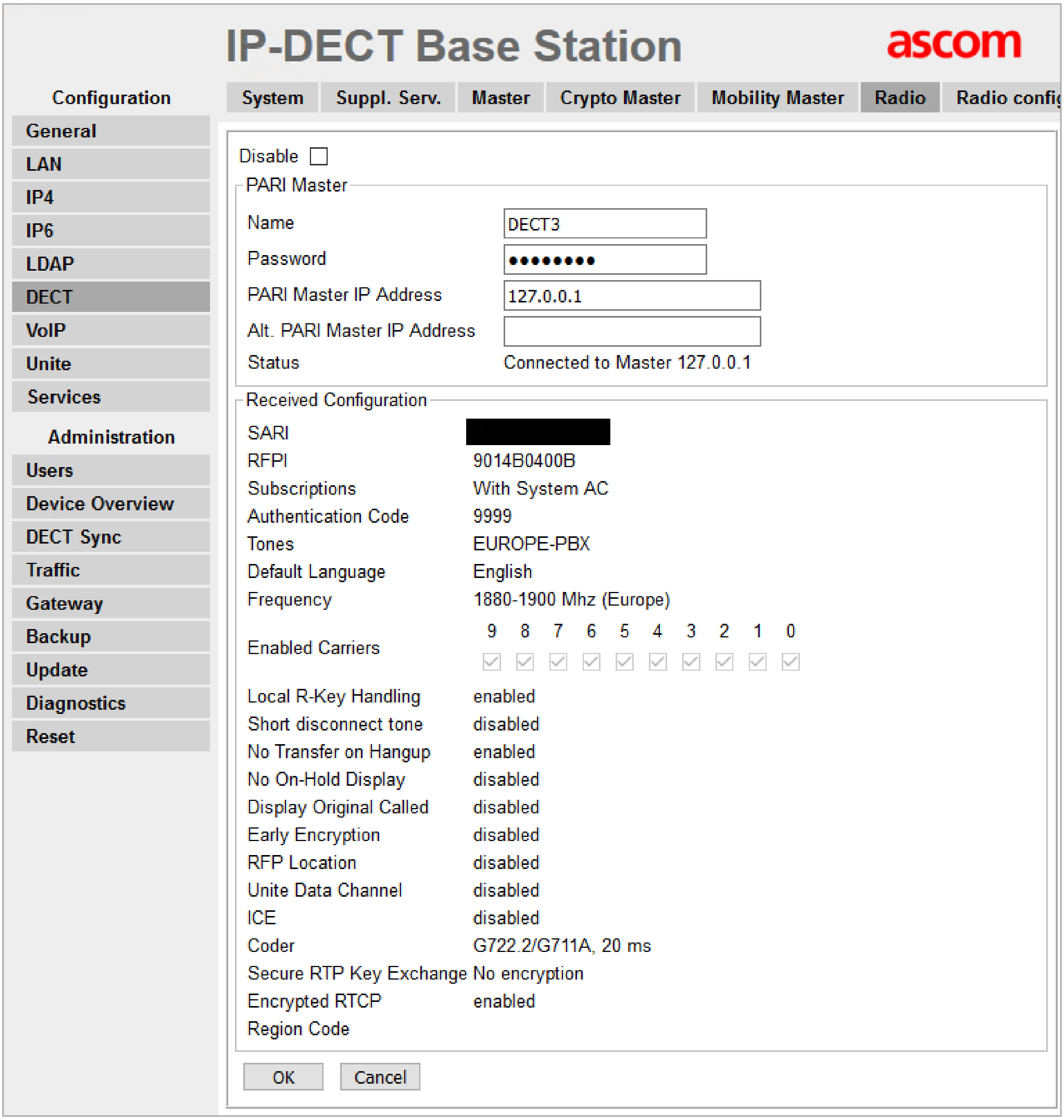


Navigate to **DECT**, click on **Master** and enter the following:

▪ **Mode**	Select Active from the dropdown box
▪ **Enable PARI Function check box**	Checked
▪ **Protocol**	Select SIP/TCP from the dropdown box
▪ **Proxy**	STMI4 IP: 10.8.242.107
▪ **Enbloc Dialling 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*

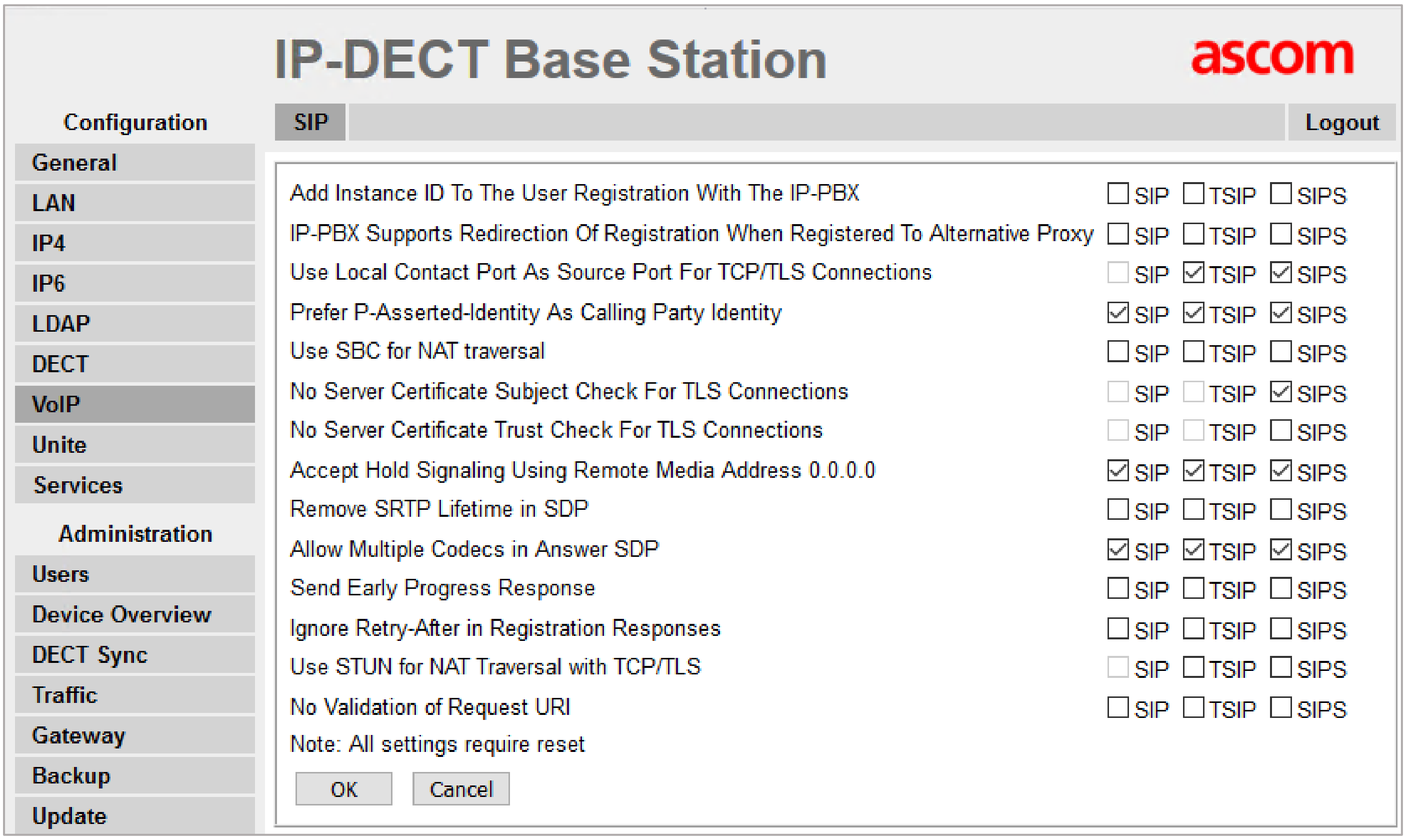


Navigate to **Radio** and select the Admin tab and enter the following:

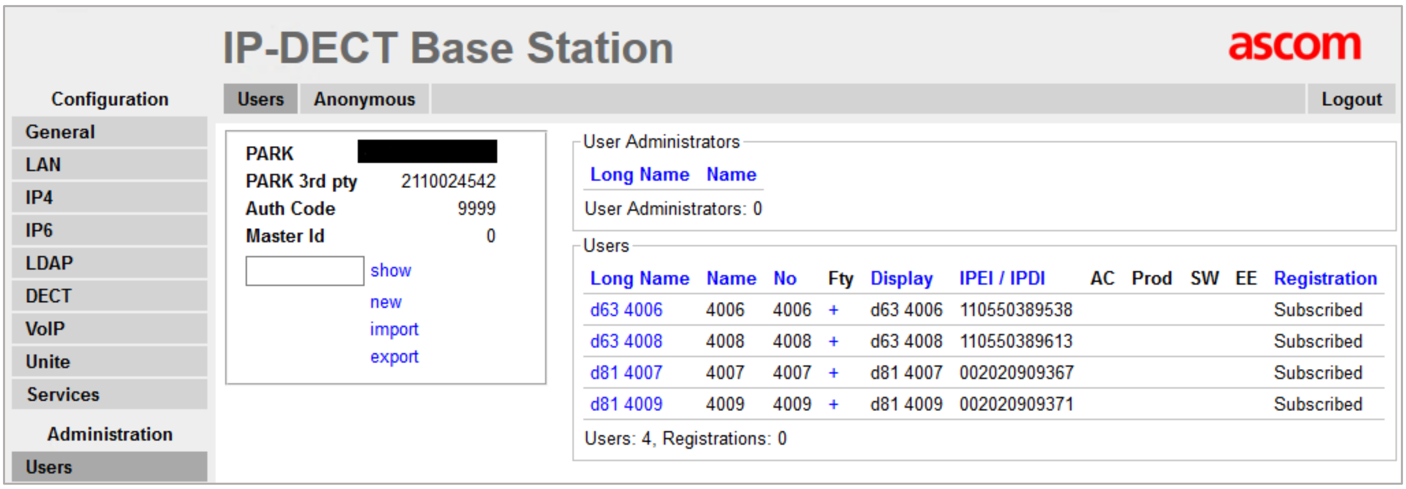
- **Device Name** Enter the name for the PARI Master in the *"Name"* text field
- **Password** Enter the password for the PARI Master in the *"Password"* text field
- **PARI Master IP Address** 127.0.0.1 (single base station)

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

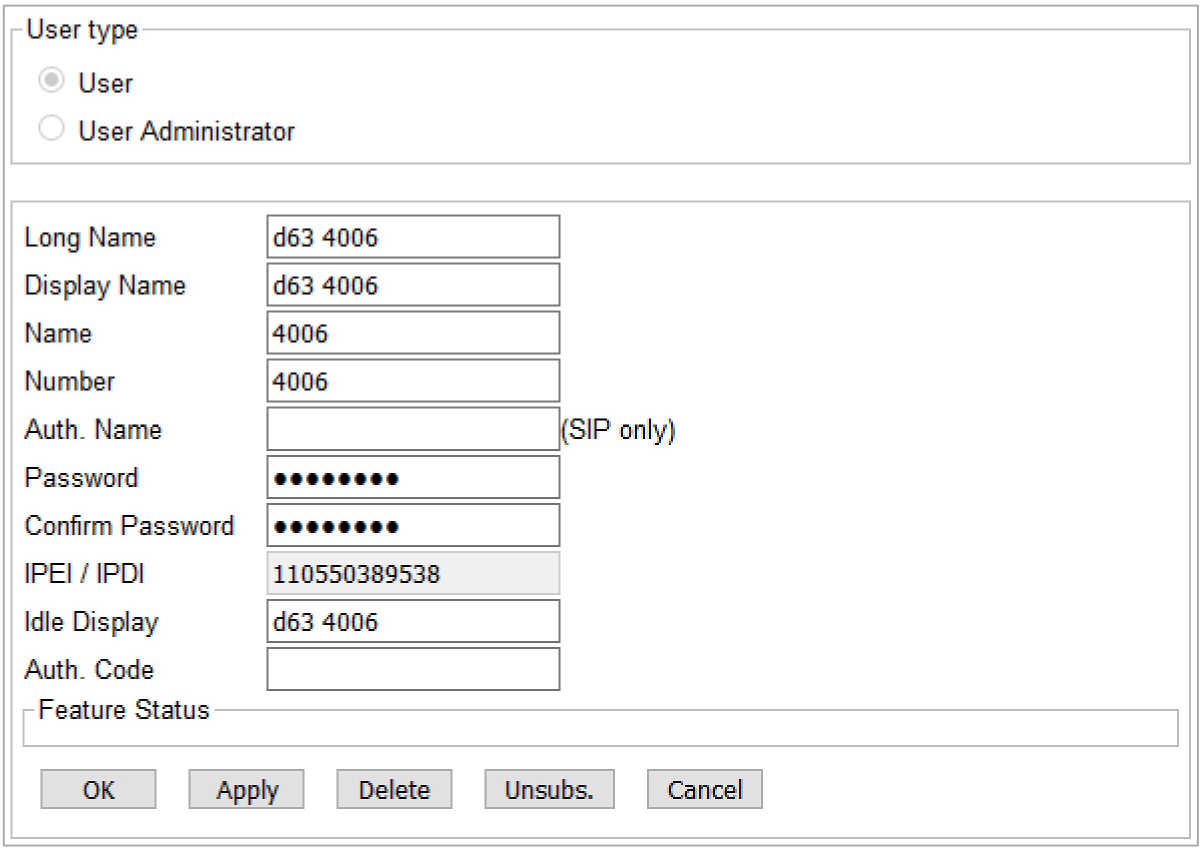
4.1.7 *VoIP*



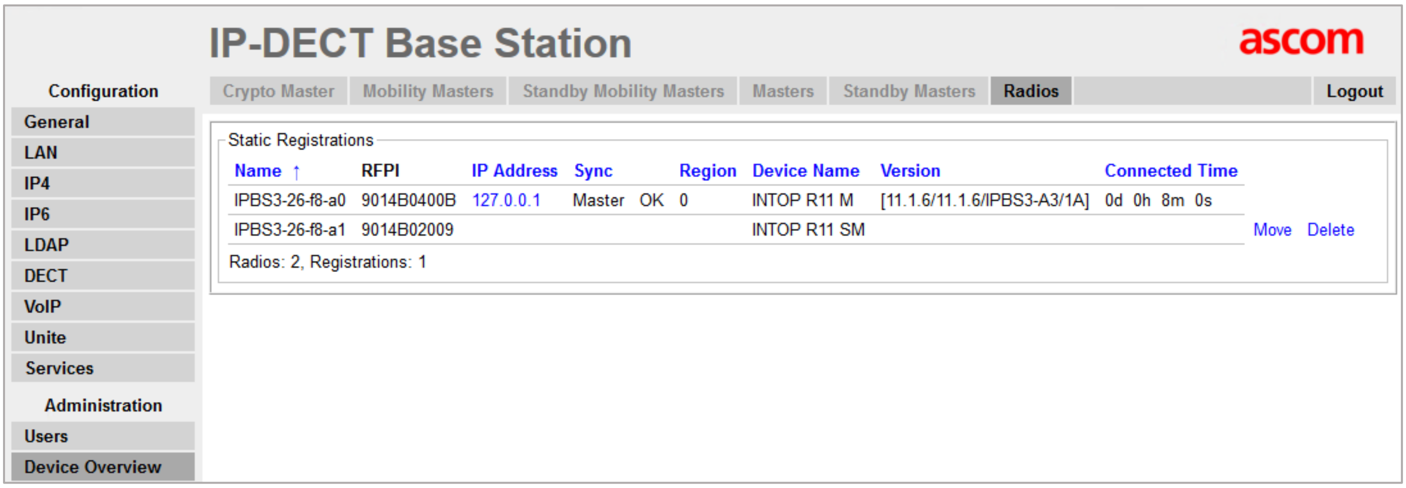
4.1.8 *IP-DECT Users*



Click on e.g. user **d63 4006**



4.1.9 *Device Overview*



4.2 OpenScape 4000 Configuration

For the needs of the current certification project, certain configuration at OS4000 should be performed. Routine OS4000 configuration is omitted for simplicity reasons.

4.2.1 AMO Configuration for DECT SIP Users

Add DECT SIP numbers in dial plan:

```
ADD-WABE:CD=4006&&4008,DAR=STN,CHECK=N;
```

Add DECT SIP extensions:

```
ADD-SBCSU:STNO=4006,OPT=OPTI,CONN=IP2,PEN=1-17-2-7,DVCFIG=UFIP,TSI=1,COS1=1,COS2=1,LCOSV1=1,LCOSV2=1,
LCOSD1=1,LCOSD2=1,DPLN=0,ITR=0,SSTNO=N,COSX=0,SPDI=0,IDCR=N,STD=110,INS=Y,ALARMNO=0,RCBKB=N,RCBKNA=N,CBKBMAX=
5,HEADSET=N,HSKEY=NORMAL,CBKNAMB=Y,TEXTSEL=ENGLISH,HMUSIC=0,PMIDX=1,COMGRP=0,IPPASSW="123456",USRID="4006",SE
CZON="OS4K";
```

```
ADD-SBCSU:STNO=4007,OPT=OPTI,CONN=IP2,PEN=1-17-2-8,DVCFIG=UFIP,TSI=1,COS1=1,COS2=1,LCOSV1=1,LCOSV2=1,
LCOSD1=1,LCOSD2=1,DPLN=0,ITR=0,SSTNO=N,COSX=0,SPDI=0,IDCR=N,STD=110,INS=Y,ALARMNO=0,RCBKB=N,RCBKNA=N,CBKBMAX=
5,HEADSET=N,HSKEY=NORMAL,CBKNAMB=Y,TEXTSEL=ENGLISH,HMUSIC=0,PMIDX=1,COMGRP=0,IPPASSW="123456",USRID="4007",SE
CZON="OS4K";
```

```
ADD-SBCSU:STNO=4008,OPT=OPTI,CONN=IP2,PEN=1-17-2-9,DVCFIG=UFIP,TSI=1,COS1=1,COS2=1,LCOSV1=1,LCOSV2=1,
LCOSD1=1,LCOSD2=1,DPLN=0,ITR=0,SSTNO=N,COSX=0,SPDI=0,IDCR=N,STD=110,INS=Y,ALARMNO=0,RCBKB=N,RCBKNA=N,CBKBMAX=
5,HEADSET=N,HSKEY=NORMAL,CBKNAMB=Y,TEXTSEL=ENGLISH,HMUSIC=0,PMIDX=1,COMGRP=0,IPPASSW="123456",USRID="4008",SE
CZON="OS4K";
```

4.2.2 AMO Configuration for Xpressions SIP Trunk

Add function block for vHG board:

```
ADD-BFDAT:FCTBLK=6,FUNCTION=HG3550,BRDBCHL=BCHL120,ATTR=SOCO;
```

```
CHANGE-BFDAT:CONFIG=CONT,FCTBLK=6,FUNCTION=HG3550,LINECNT=4,UNITS=3;
```

```
CHANGE-BFDAT:CONFIG=OK,FCTBLK=6,ANSW=YES;
```

Add vHG board:

```
ADD-BCSU:MTYPE=IPGW,LTG=1,LTU=99,SLOT=2,PARTNO="Q2330-X",FCTID=1,LWVAR="0",FCTBLK=6,BCHL3550=30,ALARMNO=0,
IPMODE=IPV4,DHCPV4=NO,DHCPV6=NO;
```

Add IP address for STMI2 board:

```
ADD-CGWB:LTU=99,SLOT=2,SMODE=NORMAL,IPADR=10.8.242.109,NETMASK=255.255.255.0;
```

```
CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=2,TYPE=GLOBIF,PATTERN=213,VLAN=NO,VLANID=0,DEFRT=10.8.242.1,TRPRSIP=30,
TRPRSIPQ=0,TRPRH323=0,TPRH323A=0,TLSP=4061,DNSIPADR=10.8.251.103,DNSIPAD2=0.0.0.0,USEWANIF=NO,WPUBIP=0.0.0.0,
SIPTCPP=5060,SIPTLSP=5061;
```

```
CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=2,TYPE=SERVIF,LOGINTRM="TRM",PASSW="HICOM";
```

```
CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=2,TYPE=ASC,UDPPRTLO=29100,UDPPRTHI=30099,TOSPL=184,TOSSIGNL=104,T38FAX=YES,
RFCFMOIP=NO,RFCDTMF=YES,REDRFCTN=YES,PRIO=PRIO1,CODEC=G711A,VAD=NO,RTP=30;
```

```
CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=2,TYPE=ASC,PRIO=PRIO2,CODEC=G729A,VAD=NO,RTP=20;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=2,TYPE=ASC,PRIO=PRIO3,CODEC=NONE,VAD=NO,RTP=30;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=2,TYPE=ASC,PRIO=PRIO4,CODEC=NONE,VAD=NO,RTP=20;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=2,TYPE=ASC,PRIO=PRIO5,CODEC=NONE,VAD=NO,RTP=20;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=2,TYPE=ASC,PRIO=PRIO6,CODEC=NONE,VAD=NO,RTP=20;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=2,TYPE=ASC,PRIO=PRIO7,CODEC=G729AB,VAD=YES,RTP=20;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=2,TYPE=ASC,PRIO=PRIO8,CODEC=G722,VAD=NO,RTP=20;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=2,TYPE=ASC,PRIO=PRIO9,CODEC=OPUS,VAD=NO,RTP=20;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=2,TYPE=MGNTDATA,MGNTIP=10.8.242.100,MGNTPN=8000,BUSIP=10.8.242.100,
BUSPN=443,UIMODE=CLASSIC;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=2,TYPE=DMCDATA,DMCCONN=0,SMP=YES,SMP4OSV=NO;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=2,TYPE=WBMDATA,LOGINWBM="HP4K-DEVEL",ROLE=ENGR;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=2,TYPE=WBMDATA,LOGINWBM="HP4K-SU",ROLE=SU;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=2,TYPE=WBMDATA,LOGINWBM="HP4K-ADMIN",ROLE=ADMIN;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=2,TYPE=WBMDATA,LOGINWBM="HP4K-READER",ROLE=READONLY;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=2,TYPE=GWDATA,GWID1="PRIMARYRASMANAGERID";

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=2,TYPE=SIPTRERH,GWAUTREQ=NO;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=2,TYPE=SIPTRSSA,SIPREG=NO,REGIP1=0.0.0.0,PORTTCP1=5060,PORTTLS1=5061,
REGTIME=300,REGIP2=0.0.0.0,PORTTCP2=5060,PORTTLS2=5061;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=2,TYPE=DLSDATA,DLSIPADR=10.6.25.5,DLSPORT=18443,DLSACPAS=YES;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=2,TYPE=JB,AVGDLYV=40,MAXDLYV=120,MINDLYV=20,PACKLOSS=4,AVGDLYD=60,
MAXDLYD=200,JBMODE=2;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=2,TYPE=IPCONF,IPMODE=IPV4,DHCPV4=NO,DHCPV6=NO;

CHANGE-CGWB:MTYPE=CGW,LTU=99,SLOT=2,TYPE=MANLANIF,MIPADR=0.0.0.0,MNETMASK=0.0.0.0,MVLAN=NO,MVLANID=0,
MDEFRT=0.0.0.0;
```

Add Bundle:

```
ADD-BUEND:TGRP=1,NAME="XPRESSIONS",NO=30,TRACENO=0,ACDTHRH=*,PRIONO=2,TDDRFLAG=OFF,GDTRRULE=0,ACDPMGRP=0,
CHARCON=NEUTRAL;
```

Add Class of Parameter:

```
ADD-COP:COPNO=1,PAR=ANS&L3AR&IMEX,TRK=TA,TOLL=TA;

CHANGE-COP:COPNO=1,COPTYPE=COPADD,DEV=INDEP,INFO="IP TR";
```

Add Class of Trunk:

```
ADD-COT:COTNO=1,PAR=RCL&IIDL&IVAC&IBSY&INAU&ITB&IDND&IFR&ANS&CHRT&AEOD&CEBC&COTN&IEVT&IDIS&BSHT&LWNC&
INDG&NLCR&ICZL&ABNA&ABPD&WAAN&IONS&NLRD&NPIS&ANNC;
```

Add Class of Service:

```
ADD-COSSU:NEWCOS=1,INFO="";
CHANGE-COSSU:TYPE=COS,COS=1,AVCE=TA&TNOTCR&CDRINT&COSXCD&MB&DATA&CFNR&VCE;
CHANGE-COSSU:TYPE=COS,COS=1,AVCE=RSVLN&DICT&SPKR&FWDNWK&TTT&MSN&CFB&MULTRA;
CHANGE-COSSU:TYPE=COS,COS=1,AVCE=FWDEXT&CCBS&CW;
CHANGE-COSSU:TYPE=COS,COS=1,AFAX=TA&TNOTCR;
CHANGE-COSSU:TYPE=COS,COS=1,ADTE=TA&TNOTCR&CDRINT&BASIC&MSN&MULTRA;
```

Add digital trunk circuits:

```
ADD-TDCSU:OPT=NEW,PEN=1-99-002-0,COTNO=1,COPNO=1,DPLN=0,ITR=0,COS=1,LCOSV=1,LCOSD=1,CCT="XPRESSIONS
",DESTNO=0,PROTVAR=ECMAV2,SEGMENT=8,DEDSVC=NONE,TRTBL=GDTR,SIDANI=N,ATNTYP=TIE,CBMATTR=NONE,TCHARG=N,SUPPRESS
=0,TRACOUNT=30,SATCOUNT=MANY,ALARMNO=0,FIDX=1,CARRIER=1,ZONE=EMPTY,COTX=1,FWDX=5,CHIMAP=N,UUSCCX=16,UUSCCY=8,
FNIDX=0,NWMUXTIM=10,SRCGRP=5,CLASSMRK=EC&G711&G729AOPT,TCCID="",TGRP=1,SRCHMODE=DSC,INS=Y,SECLEVEL=TRADITIO,H
MUSIC=0,CALLTIM=60,WARNTIM=60,DEV=HG3550IP,BCHAN=1&&10,BCNEG=N,BCGR=1,LWPP=0,LWLT=0,LWPS=0,LWR1=0,LWR2=1,DMCA
LLWD=Y,GWPROT=NONE;
```

Add SIP trunk destination:

```
ADD-RICHT:MODE=LRTENEW,LRTE=1,LSVC=ALL,NAME="XPR",TGRP=1,DNNO=1-1-121,ROUTOPT=NO,DTMFCNV=FIX,
DTMFTEXT="",DTMFPULS=PP80,ROUTATT=NO,EMCYRTT=NO,INFO="",PDNNO=0,CHARCON=NEUTRAL,CONFTONE=NO,RERINGRP=NO,NOPRC
FWD=NO,NITO=NO,CLNAMEDL=NO,FWDSWTCH=NO,LINFEMER=NO,NOINTRTE=NO;
ADD-RICHT:MODE=PM,IDX=1,SAN=492119598559,NAME="XPR",STYPE=XPRESION;
```

Add LCR outdial rule:

```
ADD-LODR:ODR=1,CMD=OUTPULSE,DGTS=492119598;
ADD-LODR:ODR=1,CMD=ECHO,FIELD=1;
ADD-LODR:ODR=1,CMD=NPI,NPI=ISDN,TON=INTERNAT;
ADD-LODR:ODR=1,CMD=END;
```

Add Administration LCR:

```
ADD-LDAT:LROUTE=1,LSVC=ALL,LVAL=1,TGRP=1,ODR=1,LAUTH=1,CARRIER=1,ZONE=EMPTY,LATTR=WCHREG,VCCYC=4;
```

Add digits to dial plan:

```
ADD-WABE:CD=556&&559,DAR=TIE,CHECK=N;
```

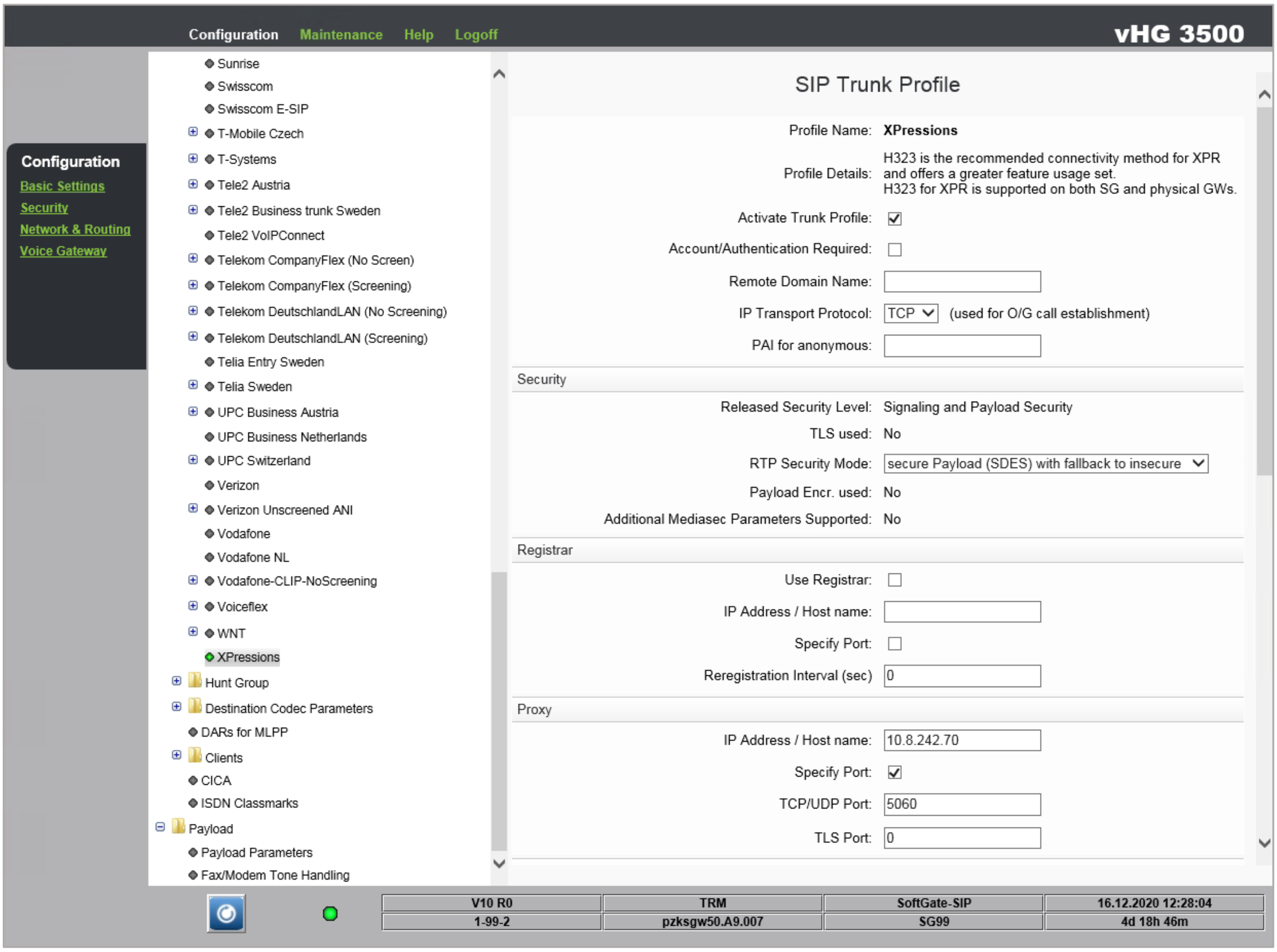
Add Administration LCR – Dialplan:

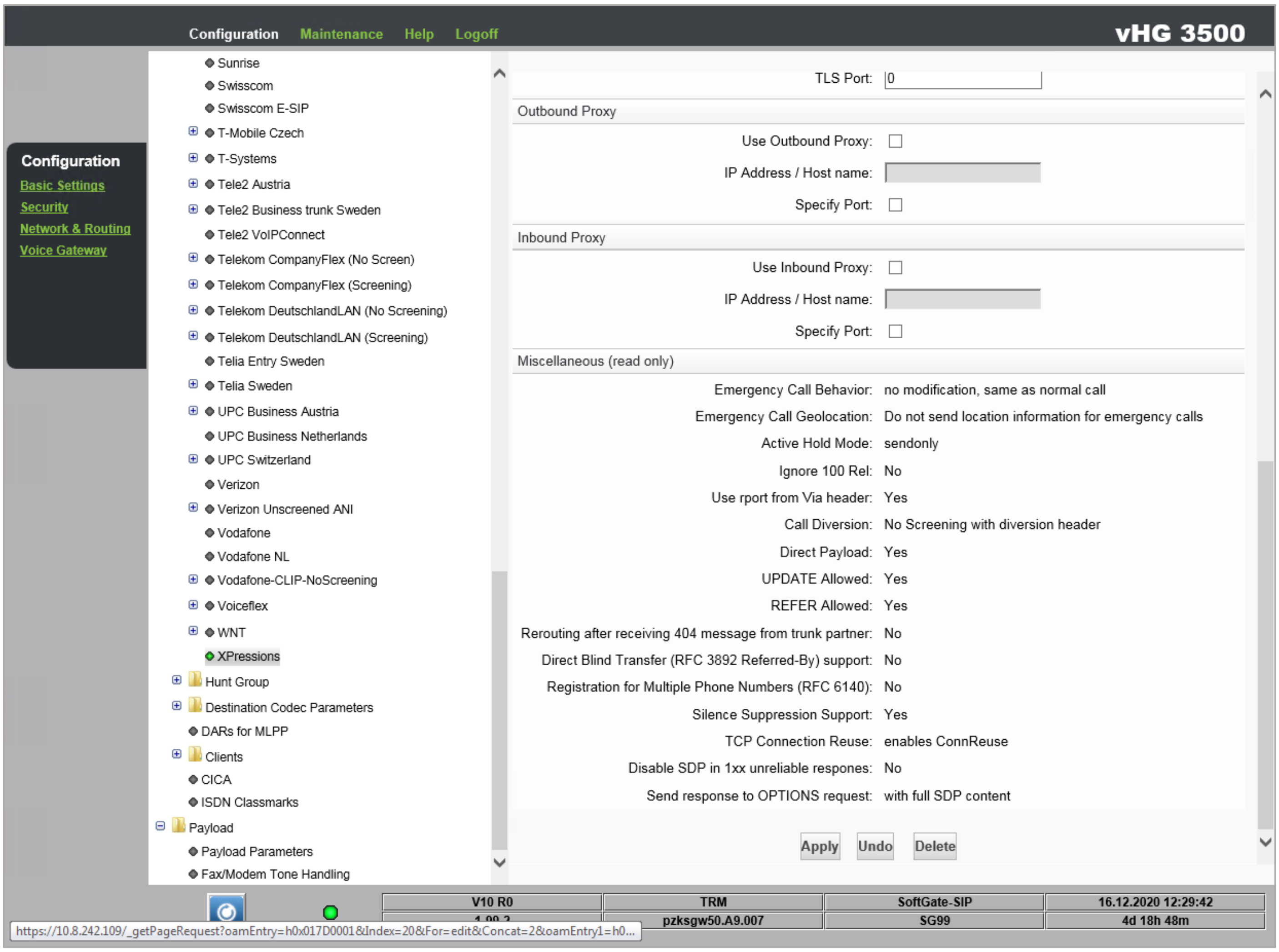
```
ADD-LDPLN:LCRCONF=LCRPATT,DIPLNUM=0,LDP="556",DPLN=0&1&2&3&4&5&6&7&8&9&10&11&12&13&14&15,LROUTE=1,LAUTH=1,
PINDP=N;
ADD-LDPLN:LCRCONF=LCRPATT,DIPLNUM=0,LDP="557",DPLN=0&1&2&3&4&5&6&7&8&9&10&11&12&13&14&15,LROUTE=1,LAUTH=1,
PINDP=N;
```

```
ADD-LDPLN:LCRCONF=LCRPATT,DIPLNUM=0,LDP="558",DPLN=0&1&2&3&4&5&6&7&8&9&10&11&12&13&14&15,LROUTE=1,LAUTH=1,
PINDP=N;

ADD-LDPLN:LCRCONF=LCRPATT,DIPLNUM=0,LDP="559",DPLN=0&1&2&3&4&5&6&7&8&9&10&11&12&13&14&15,LROUTE=1,LAUTH=1,
PINDP=N;
```

4.2.3 *WBM Configuration for Xpressions SIP Trunk*





Go to vHG WBM and select **Configuration >> Voice Gateway >> SIP Trunk Profiles >> NatTrkWithoutRegistration** and enter the following:

- **Activate Trunk Profile check box** Checked
- **IP Transport Protocol** TCP
- **Proxy - IP Address / Host Name** 10.8.242.70

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

4.2.4 *AMO Configuration for ISDN Provider Connection*

STMD3 board:

```
DISPLAY-SDSU:TYPE=PEN,LEVEL=PER3,LTG=1,LTU=17,SLOT=9;
H500:  AMO SDSU  STARTED

  LTG1  (PERIPHERY)
  ------
  MOUNTING LOCATION    MODULE NAME     BDL BD(#=ACT)   STATUS
  **   .LTG 1.LTU17.009 STMD3           ?   Q2217-X
             CCT  LINE          STNO  SI BUS TYPE
             000  1563                  PP NW          UNACA
               MULTLINE 3. . . . . . . . . . . . . . .UNACH
```

```
      001 NETWORK SUBUNIT . TMD CO ISDN      UNACH/CP
      (ALT_ROUT: N)       (S0COD    )
        LINE: 1563 STNO:               SI:
          001 . . . . . . . . TMD CO ISDN    UNACH/CP
      002 NETWORK SUBUNIT . TMD CO ISDN      UNACH/CP
      (ALT_ROUT: N)       (S0COD    )
        LINE: 1563 STNO:               SI:
          001 . . . . . . . . TMD CO ISDN    UNACH/CP
    CCT  LINE        STNO  SI BUS TYPE
    001  1564                                NOGEN
    002  1565                                NOGEN
    003  1566                                NOGEN
    004  1567                                NOGEN
    005  1568                                NOGEN
    006  1569                                NOGEN
    007  1570                                NOGEN
```

Add Bundle:

ADD-BUEND:TGRP=7,NAME="CO S0 STMD",NO=8,TRACENO=0,ACDTHRH=*,PRIONO=1,TDDRFLAG=ON,GDTRRULE=0,ACDPMGRP=0,

CHARCON=NEUTRAL;

Add Class of Parameter:

ADD-COP:COPNO=7,PAR=DTMF&COCN&NO1A&L3AR&TIM1&NSDL&TIM3&IDP2&DTM1,TRK=TA,TOLL=TA;

Add Class of Trunk:

ADD-COT:COTNO=7,PAR=RCL&INAU&ANS&CEOC&CEBC&CBBN&COTN&BSHT&BLOC&LWNC&INDG&NLCR&CECO&TSCS&DFNN&IONS&NLRD&
AOCC&NTON;

Add Class of Service:

ADD-COSSU:NEWCOS=1,INFO="";

CHANGE-COSSU:TYPE=COS,COS=1,AVCE=TA&TNOTCR&CDRINT&COSXCD&MB&DATA&CFNR&VCE;

CHANGE-COSSU:TYPE=COS,COS=1,AVCE=RSVLN&DICT&SPKR&FWDNWK&TTT&MSN&CFB&MULTRA;

CHANGE-COSSU:TYPE=COS,COS=1,AVCE=FWDEXT&CCBS&CW;

CHANGE-COSSU:TYPE=COS,COS=1,AFAX=TA&TNOTCR;

CHANGE-COSSU:TYPE=COS,COS=1,ADTE=TA&TNOTCR&CDRINT&BASIC&MSN&MULTRA;

Add digital trunk circuits:

ADD-TDCSU:OPT=NEW,PEN=1-17-009-0,COTNO=7,COPNO=7,DPLN=0,ITR=0,COS=1,LCOSV=1,LCOSD=1,CCT="CO S0",DESTNO=0,
PROTVAR=ETSI,SEGMENT=1,DEDSVC=NONE,TRTBL=GDTR,SIDANI=N,ATNTYP=CO,CBMATTR=NONE,TCHARG=N,SUPPRESS=0,DGTPR=3582,
ISDNIP=900,ISDNNP=0,TRACOUNT=31,SATCOUNT=MANY,NNO=1-10-999,ALARMNO=2,FIDX=1,CARRIER=1,ZONE=EMPTY,COTX=7,
FWDX=5,CHIMAP=N,UUSCCX=16,UUSCCY=8,FNIDX=1,NWMUXTIM=10,SRCGRP=17,CLASSMRK=EC&G711&G729AOPT,TCCID="",TGRP=7,SR
CHMODE=CIR,INS=Y,SECLEVEL=TRADITIO,HMUSIC=0,CALLTIM=60,WARNTIM=60,DEV=S0COD,PERMACT1=Y,PERMACT2=Y,TEIVERIF=N,
FIXEDTEI=0,CNTRNR=0,BCNEG=N;

Add destination:

```
ADD-RICHT:MODE=LRTENEW,LRTE=5,LSVC=ALL,NAME="OTE_BRI",TGRP=7,DNNO=1-1-122,ROUTOPT=NO,DTMFCNV=FIX,DTMFTEXT="",
DTMFPULS=PP80,ROUTATT=NO,EMCYRTT=NO,INFO="",PDNNO=0,CHARCON=NEUTRAL,CONFTONE=NO,RERINGRP=NO,NOPRCFWD=NO,NITO=
NO,CLNAMEDL=NO,FWDSWTCH=NO,LINFEMER=NO,NOINTRTE=NO;
```

Add LCR outdial rule:

```
ADD-LODR:ODR=10,CMD=NPI,NPI=UNKNOWN,TON=UNKNOWN;

ADD-LODR:ODR=10,CMD=ECHO,FIELD=2;

ADD-LODR:ODR=10,CMD=END;
```

Add Administration LCR:

```
ADD-LDAT:LROUTE=5,LSVC=ALL,LVAL=1,TGRP=7,ODR=10,LAUTH=1,CARRIER=1,ZONE=EMPTY,LATTR=WCHREG&PUBNUM,VCCYC=4;

ADD-LDAT:LROUTE=5,LSVC=ALL,LVAL=2,TGRP=7,ODR=10,LAUTH=1,CARRIER=1,ZONE=EMPTY,LATTR=WCHREG&PUBNUM,VCCYC=4;
```

Add digits to dial plan:

```
ADD-WABE:CD=9,DAR=CO,CHECK=N;
```

Add Administration LCR – Dialplan:

```
ADD-LDPLN:LCRCONF=LCRPATT,DIPLNUM=0,LDP="9"-"Z",DPLN=0&1&2&3&4&5&6&7&8&9&10&11&12&13&14&15,
LROUTE=5,LAUTH=1,PINDP=N;
```

4.2.5 *OS4000 Dialing Plan*

Global OS4000 system dialing plans and feature access codes:

```
DISPLAY-WABE:TYPE=GEN;

H500:  AMO WABE  STARTED
  ------------------------------------------------------------------------------
  | DIGIT INTERPRETATION                  VALID FOR ALL DIAL PLANS              |
  ------------------------------------------------------------------------------
  |                 |    CALL PROGRESS STATE     |NODE/DIGIT| RESERVED/CONVERT |
  |      CODE       |            1 11111 11112 22| ANALYSIS | DNI/ADD-INFO/UFIP|
  |                 |0 12345 67890 12345 67890 12|  RESULT  | *=OWN NODE       |
  ------------------------------------------------------------------------------
  | 0               |. .**** ..*** **... ..... .*| CO       |                  |
  | 101             |. .**** ***** **... ..... .*| OWNNODE  |                  |
  | 13              |* ..... ..... ..... ..... ..| NETRTE   |                  |
  | 17              |* ..... ..... ..... ..... ..| NETRTE   |                  |
  | 2               |. .**** ..*** **... ..... .*| CO       |                  |
  | 323             |. .**** ***** **... ..... .*| TIE      |                  |
  | 3580            |. .**** ***** **... ..... .*| STN      | R                |
  |                 |                            |          |DESTNO    0       |
  |                 |                            |          |DNNO     1- 30-300*|
  | 3581    - 3584  |. .**** ***** **... ..... .*| STN      |                  |
  |                 |                            |          |DESTNO    0       |
  |                 |                            |          |DNNO     1- 30-300*|
  | 3585            |. .**** ***** **... ..... .*| STN      | R                |
  |                 |                            |          |DESTNO    0       |
  |                 |                            |          |DNNO     1- 30-300*|
  ------------------------------------------------------------------------------
  | DIGIT INTERPRETATION                  VALID FOR ALL DIAL PLANS              |
  ------------------------------------------------------------------------------
```

```
|                 |    CALL PROGRESS STATE     |NODE/DIGIT| RESERVED/CONVERT  |
|      CODE       |            1 11111 11112 22| ANALYSIS | DNI/ADD-INFO/UFIP |
|                 |0 12345 67890 12345 67890 12|  RESULT  | *=OWN NODE        |
-------------------------------------------------------------------------------
| 3586            |. .**** ***** **... ..... .*| STN      |                   |
|                 |                            |          |DESTNO   0         |
|                 |                            |          |DNNO     1- 30-300*|
| 3587    - 3595  |. .**** ***** **... ..... .*| STN      | R                 |
|                 |                            |          |DESTNO   0         |
|                 |                            |          |DNNO     1- 30-300*|
| 4000    - 4009  |. .**** ***** **... ..... .*| STN      |                   |
|                 |                            |          |DESTNO   0         |
|                 |                            |          |DNNO     1- 30-300*|
| 4010    - 4019  |. .**** ***** **... ..... .*| STN      |                   |
|                 |                            |          |DESTNO  17         |
|                 |                            |          |DNNO     1-  1-127 |
|                 |                            |          |PDNNO    1-  1-127 |
| 492             |. .**** ***** **... ..... .*| TIE      |                   |
| 556     - 559   |. .**** ***** **... ..... .*| TIE      |                   |
| 60      - 62    |. .**** ***** **... ..... .*| TIE      |                   |
-------------------------------------------------------------------------------
| DIGIT INTERPRETATION                  VALID FOR ALL DIAL PLANS              |
-------------------------------------------------------------------------------
|                 |    CALL PROGRESS STATE     |NODE/DIGIT| RESERVED/CONVERT  |
|      CODE       |            1 11111 11112 22| ANALYSIS | DNI/ADD-INFO/UFIP |
|                 |0 12345 67890 12345 67890 12|  RESULT  | *=OWN NODE        |
-------------------------------------------------------------------------------
| 630     - 631   |. .**** ***** **... ..... .*| STN      |                   |
|                 |                            |          |DESTNO  13         |
|                 |                            |          |DNNO     1-  1-123 |
|                 |                            |          |PDNNO    1-  1-123 |
| 64              |. .**** ***** **... ..... .*| TIE      |                   |
| 65              |. .**** ..**. ..... .*... .*| APIN4    |                   |
| 66              |. .**** ..*** **... ..... .*| CO       |                   |
| 800             |. .**** ***** **... ..... .*| HUNT     |                   |
| 888     - 889   |. .**** ***** **... ..... .*| STN      |                   |
|                 |                            |          |DESTNO   0         |
|                 |                            |          |DNNO     1- 30-300*|
| 9               |. .**** ..*** **... ..... .*| CO       |                   |
| *10             |. *.... ..*.. ....* ..... .*| ACBK     |                   |
| *12             |. ....* ..*.. ..... ..... .*| VOICECAL |                   |
| *13             |. .**.* ..**. ..... ..... ..| SELFPA   |                   |
| *14             |. .**.* ..**. ..... ..... ..| SELFPA   |                   |
| *15             |. ....* ..**. ..... ..... .*| RECONNPA |                   |
-------------------------------------------------------------------------------
| DIGIT INTERPRETATION                  VALID FOR ALL DIAL PLANS              |
-------------------------------------------------------------------------------
|                 |    CALL PROGRESS STATE     |NODE/DIGIT| RESERVED/CONVERT  |
|      CODE       |            1 11111 11112 22| ANALYSIS | DNI/ADD-INFO/UFIP |
|                 |0 12345 67890 12345 67890 12|  RESULT  | *=OWN NODE        |
-------------------------------------------------------------------------------
| *20             |. .**** ***** **..* ..... ..| AFWDREM  |                   |
|                 |                            |          |CFREMVAR CFU       |
|                 |                            |          |CFREMSE  VOICE     |
| *21             |. ....* ..... ....* ..... ..| AFWDVCE  |                   |
| *22             |. ....* ..... ....* ..... ..| AFWDDTE  |                   |
| *23             |. ....* ..... ....* ..... ..| AFWDFAX  |                   |
| *24             |. ....* ..... ....* ..... ..| AFFWDVCE |                   |
| *25             |. ....* ..... ....* ..... ..| AFFWDDTE |                   |
| *26             |. ....* ..... ....* ..... ..| AFFWDFAX |                   |
| *27             |. ....* ..... ..... ..... ..| ADND     |                   |
| *28             |. ....* ..... ....* ..... ..| AFWDB    |                   |
| *29             |. ....* ..... ....* ..... ..| AFWDNA   |                   |
| *33             |. .**.* *.*** **... ..... ..| FAX      |                   |
| *40             |. .**** ***** **..* ..... ..| FWDREM   |                   |
|                 |                            |          |CFREMVAR CFU       |
|                 |                            |          |CFREMSE  VOICE     |
| *41             |. *...* ..**. ..... ..... .*| BROADCST |                   |
-------------------------------------------------------------------------------
| DIGIT INTERPRETATION                  VALID FOR ALL DIAL PLANS              |
-------------------------------------------------------------------------------
```

```
|                  |    CALL PROGRESS STATE     |NODE/DIGIT| RESERVED/CONVERT |
|      CODE        |            1 11111 11112 22| ANALYSIS | DNI/ADD-INFO/UFIP|
|                  |0 12345 67890 12345 67890 12|  RESULT  | *=OWN NODE       |
-----------------------------------------------------------------------------
| *42              |. ....* ..**. ..... ..... ..| HOLD     |                  |
| *43              |. ....* ..... ..... ..... ..| MHFALGON |                  |
| *44              |. .**** *.*** **... ..... .*| DISUON   |                  |
| *50     - *59    |. .**** ..**. ..... ..... ..| PARK     |                  |
| *6               |. .**** ..**. ..... ..... .*| PUDIR    |                  |
| *7               |. ....* ..... ..... ..... ..| PU       |   UFIP           |
| *8               |. *.... ..... ..... ..... ..| KNOVR    |                  |
| *9               |. ..... ..... ..... ....* ..| CONSKY   |                  |
| #10              |. ....* ..... ..... ..... ..| DCBK     |                  |
| #20              |. .**** ***** **..* ..... ..| DFWDREM  |                  |
|                  |                            |          |CFREMVAR CFU      |
|                  |                            |          |CFREMSE  VOICE    |
| #21              |. ....* ..... ..... ..... ..| DFWDVCE  |                  |
| #22              |. ....* ..... ..... ..... ..| DFWDDTE  |                  |
| #23              |. ....* ..... ..... ..... ..| DFWDFAX  |                  |
| #24              |. ....* ..... ..... ..... ..| DFFWDVCE |                  |
| #25              |. ....* ..... ..... ..... ..| DFFWDDTE |                  |
-----------------------------------------------------------------------------
| DIGIT INTERPRETATION                VALID FOR ALL DIAL PLANS              |
-----------------------------------------------------------------------------
|                  |    CALL PROGRESS STATE     |NODE/DIGIT| RESERVED/CONVERT |
|      CODE        |            1 11111 11112 22| ANALYSIS | DNI/ADD-INFO/UFIP|
|                  |0 12345 67890 12345 67890 12|  RESULT  | *=OWN NODE       |
-----------------------------------------------------------------------------
| #26              |. ....* ..... ..... ..... ..| DFFWDFAX |                  |
| #27              |. ....* ..... ..... ..... ..| DDND     |                  |
| #31              |. .**** ..**. ..... .*... .*| DPIN     |                  |
| #43              |. ....* ..... ..... ..... ..| MHFALGOF |                  |
| ##1              |. *...* ..... ..... ..... ..| KNOVR    |                  |
| ##2              |. *...* ..**. ..... ..... ..| EOVR     |                  |
| ##3              |. *..** ..**. ..... ..... ..| RELEASE  |                  |
-----------------------------------------------------------------------------

AMO-WABE -111      DIALLING PLANS, FEATURE ACCESS CODES
DISPLAY COMPLETED;
```

5 Confirmation

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

Mathew Williams	Michael Korakis
Ascom AG	Atos Greece SM SA