Unify Ready

Technology connectivity certification

The connectivity of

ASCOM IP-DECT V10

developed by Ascom Wireless Solutions has been certified at the open SIP interface of OpenScape 4000 V8 in accordance with the test report dated May 16, 2019.

Ascom Wireless Solutions is now entitled to label the above mentioned product with the Unify Ready emblem.

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, May 20th 2019

Luzia Stephan

Director Technology Partner Program

unify READY

unify
atos collaboration solutions

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Certification Test Report

Ascom IP-DECT

with

Unify OpenScape 4000 V8R2

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Customer Solution Lab Athens

Release Version: **1.2 – 2019-05-16**

For Customer: **Ascom A.G.**

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CSL Customer Solution Lab

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History of Change

Version	Date	Description	Name
0.1	March 21st, 2019	Initial Creation	Michael Korakis Atos IT & Tel. Services S.A. E-mail: **michail.korakis@atos.net** Phone: +30 (210) 8187 986
0.2	April 1st, 2019	• Title for paragraph 1.6.3 is changed from “Problems” to “Known Issues”. • At testcase 1-4 comments, the sentence: “The min-expires value suggested by system was 120s” is added to the comments.	Mathew Williams Ascom. A.G E-Mail: **Matthew.Williams@ascom.com** Phone: +46 31 55 93 58
1.0	April 22nd, 2019	Release	Dimitrios Galanakis Atos S.A. E-mail: **dimitrios.galanakis@atos.net** Phone: +30 (210) 8187 680
1.1	May 15th, 2019	Wrong screenshot replaced, at 4.1.9 paragraph	Michael Korakis
1.2	May 16th, 2019	• Edited out SARI number from screenshot at 4.1.7. • Edited out PARK number from screenshot at 4.1.10.	Michael Korakis

1. Overview

1.1. Executive Summary

This document describes the results of the testing activities performed for the interoperability between Ascom IP-DECT system and Unify OpenScape 4000 V8R2 PBX.

1.1.1. Basis Equipment

Test Equipment:	OpenScape 4000 Server: Simplex (DSCXL2), AP21 (NCUI2+) STMI2 Q2316 X STMI4 Q2324 X500 SLMO24 Q2168 X Virtual OpenScape Xpressions server Mediatrix 4402plus BRI gateway device OpenScape Deskphone CP400 HFA & CP600 SIP and OpenStage 80T phones
Software Releases:	OpenScape 4000 version: Platform V8 R2.22.1 Assistant V8 R2.22.1 RMX V8 R2.22.7 CSTA V8 R2.22.1 STMI2 loadware version: pzksti40.A7.002-008 STMI4 loadware version: pzksti40.A7.002-008 SLMO24 loadware version: pzdsmo10 ComWin version: 5.0.135.0 OpenScape Xpressions version: V7 R1 FR5 HF24 Mediatrix 4402plus firmware: Dgw 43.2.1343 CP400 HFA firmware: V1 R2.4.0 HFA CP600 SIP firmware: V1 R5.14.0 SIP OpenStage 80T firmware: V2 R1.15.0 TDM VMware ESXi v5.5.0 Build 6480324

1.1.2. Ascom IP-DECT

Certification:	Interoperability testing of various call scenarios and features between Ascom DECT handset SIP subscribers registered to OpenScape 4000 and Unify HFA, SIP & TDM OS4000 stations. Furthermore, calls between PSTN subscribers and Ascom 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 Stations (active and no-active redundant system) Ascom DECT Handsets d63 Ascom DECT Handset d81
Software Releases:	Ascom IP-DECT Base Station version: IPBS2[10.2.9], Bootcode[10.2.9], Hardware[IPBS2-A3/1B] Ascom DECT Handset d63 firmware: 2.3.10

	Ascom DECT Handset d81 firmware: 4.7.2
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 Stations connected to the same LAN as OpenScape 4000. The Ascom IP-DECT Base Stations operate as a conduit between the OpenScape 4000 and the Ascom IP-DECT handsets. After the Ascom IP-DECT wireless handsets register with the Ascom IP-DECT base Station redundant system, the Base Stations register the handsets to OpenScape 4000.
Documentation:	Refer to your Ascom supplier for documentation.
Test Network:	Ascom handsets register with Ascom IP-DECT Base Station redundant 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, Unify HFA, SIP and TDM stations are connected on OpenScape 4000, too. OpenScape 4000 is connected (SIP) to a Mediatrix 4402plus BRI gateway, that provides access to PSTN (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 Unify HFA, SIP, TDM deskphones and PSTN endpoints. The 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.

1.3. 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.

1.4. Measuring / Test Instruments

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

- Wireshark.

1.5. Realization Data

Test Preparation:	March 5th, 2019
Test Duration:	March 6th –13th, 2019
Test Location:	Atos IT & Tel. Services SA CSL Athens lab 455, Irakleiou Avenue 141 22, N. Irakleio, Athens, Greece
Test Personnel:	Ascom A.G., Mathew Williams, Matthew.Williams@ascom.com Atos IT & Tel. Services S.A., Michael Korakis, michail.korakis@atos.net

1.6. Test Result Summary

The certification of Ascom IP-DECT system with OpenScape 4000 V8R2 PBX is passed with certain restrictions and observations applied. Additionally, there is an issue pending for resolution. Performance testing was not included.

1.6.1. 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.

  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.

1.6.2. 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.
- A SIP station that participates in a 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).
- Regarding audio **Codecs** support, OS4000 and Ascom d81 DECT handsets don't use G.722.2 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 5-8).

1.6.3. Known Issues

- In Call Deflect and Call Forward No Reply scenarios when the original caller is a SIP station, then the caller display isn't correctly updated on ringing state after Call Deflect / CFNR action; the SIP caller displays the initial call party on ringing and not the Call Deflect / CFNR destination, whereas the SIP caller display is correctly updated when the call is answered by the Call Deflect / CFNR destination.

  The issue is identified from Unify Dev and planned to be fixed in OS4000 V10R0 release (refer to tests 4-44 & 4-53 || Unify JiRa ticket nr: OSFOURK-7353).

1.7. Network Diagram

The diagram of figure 1 below illustrates the logical network topology used during the certification project testing.

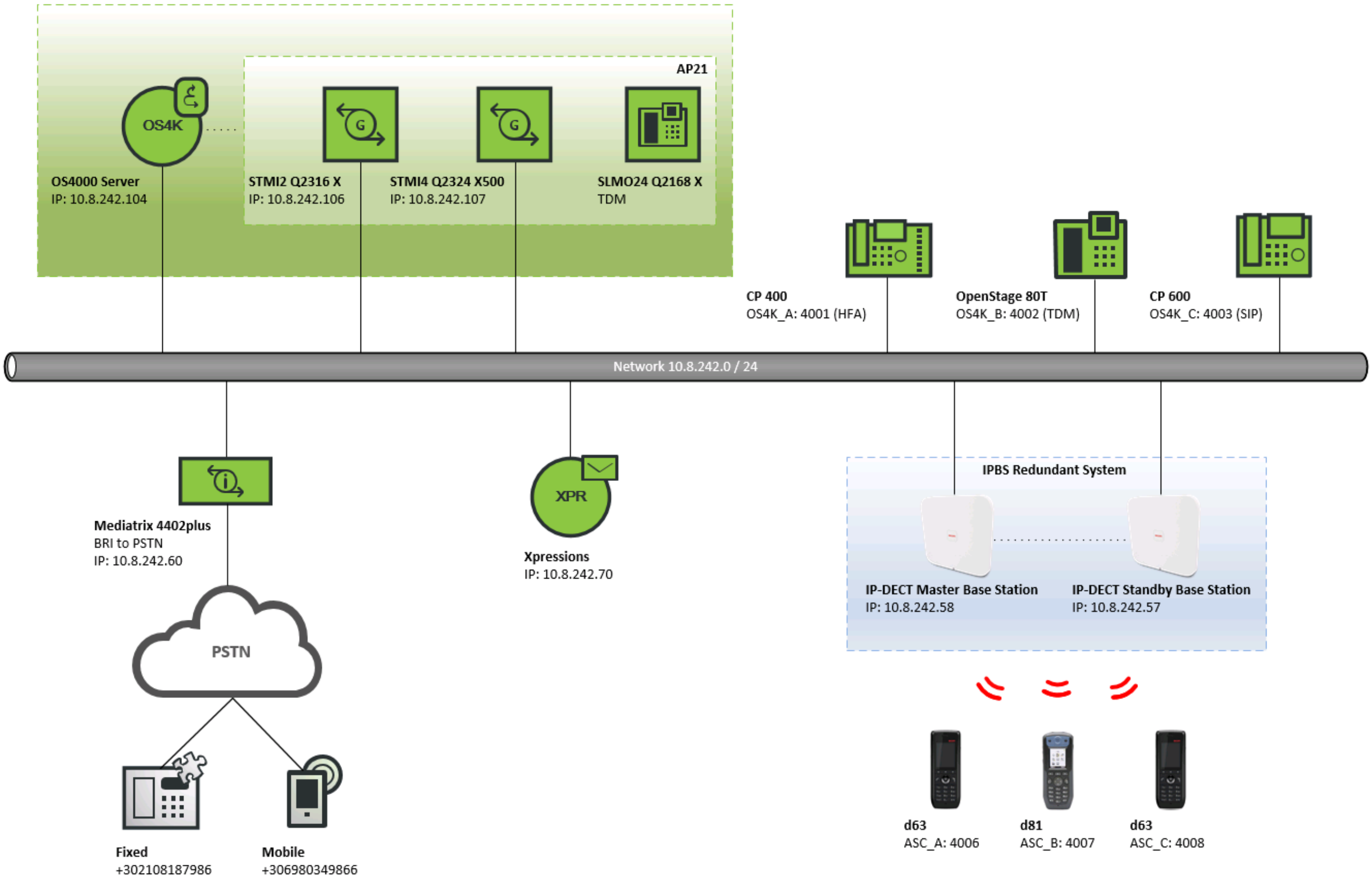


Figure 1: Ascom IP-DECT with OS4000 reference topology

2. 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 and Standby Base Stations (Master = 10.8.242.57, Standby = 10.8.242.58) belong to the same subnet as the other Unify components and the communication is established via corporate LAN.

The following OS4000 SIP extensions are utilized:

- Ascom DECT handset SIP stations

  ASC_A = 4006 (d63 handset)
  ASC_B = 4007 (d81 handset)
  ASC_C = 4008 (d63 handset)

2.2. 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 a Mediatrix 4402plus BRI gateway. OS4000 subscribers dial 9 + number to reach a PSTN subscriber. The digit 9 may be considered a “seizure code” digit which enables call routing to Mediatrix 4402plus and then is stripped away by OS4000 before sending the number to PSTN.

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

Due to existing lab setup STMI2 card was used as a wildcard HG card; it was either connected to Xpressions server or to Mediatrix device.

An MLHG group is configured with the pilot number is 800.

- OS4000

  10.8.242.104 OS4000 Communications server
  10.8.242.206 STMI2 Q2316 X (SIP trunk to Xpressions or Mediatrix 4402plus)
  10.8.242.107 STMI4 Q2324 X500 (Station registrations)
  SLMO24 Q2168 X (TDM station connectivity)

- HFA station

  OS4K_A = 4001 (CP400 phone device)

- TDM station

  OS4K_B = 4002 (OpenStage 80T phone device)

- SIP station

  OS4K_C = 4003 (CP600 phone device)

- OS Xpressions

  10.8.242.70

- Mediatrix 4402plus

  10.8.242.60

- ISDN BRI number for incoming calls from PSTN

  302106256856

- PSTN test numbers

  PSTN = 302108187986 (landline)
  CELL = 3069803498966 (mobile)

3. Test Results in Detail

3.1. Test cases

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 *"Call 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, *"Fault Tolerance"* scenarios, tests 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.

3.1.1. Registration

Test Name	Test Description	Functional Checks	Result	Comments
1-1	• Register Ascom IP-DECT phone to OS4000 without Digest Authentication.	• DECT phone successfully registers with OS4000.	OK	
1-2	• Register Ascom IP-DECT phone to OS4000 with Digest Authentication.	• DECT phone successfully registers with OS4000.	OK	
1-3	• Check that Ascom IP-DECT phone may register to OS4000 with either TCP or UDP.	• DECT phone successfully registers with OS4000.	OK	
1-4	• 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.

3.1.2. Basic Calls

Incoming / Outgoing Call

Test Name	Test Description	Functional Checks	Result	Comments
2-1	• 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	
2-2	• 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	
2-3	• 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. • 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	
2-4	• 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	
2-5	• 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	
2-6	• 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.	OK	ASC_A displays external call.

Test Name	Test Description	Functional Checks	Result	Comments
		• Both ends idle after call clearing.		
2-7	• PSTN calls ASC_A. • ASC_A hangs up at the end of the communication.	• Post Dial Delay (PDD) • Ringback tone heard by calling party. • Number presentation on ringing and after call pick up. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	OK	
2-8	• 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	
2-9	• 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	

Busy Call

Test Name	Test Description	Functional Checks	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.
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 CELL. • CELL is busy.	• Busy tone/display on calling party. • The call is properly cleared.	OK	

Rejected Call

Test Name	Test Description	Functional Checks	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 BS responds with a 486 Busy Here. Call clears at about 15 secs.

Test Name	Test Description	Functional Checks	Result	Comments
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_C. • OS4K_C rejects incoming call.	• Busy tone/display on calling party. • The call is properly cleared.	OK	OS4K responds with a 603 Decline. Call 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	

Not Answered Call

Test Name	Test Description	Functional Checks	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	Same as 2-20.
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 CELL. • CELL 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	

Call Cancellation

Test Name	Test Description	Functional Checks	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	
2-29	• ASC_A calls PSTN and hangs up before destination picks up.	• Call is terminated on both sides after hanging up.	OK	

Call to Unavailable

Test Name	Test Description	Functional Checks	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).
2-31	• ASC_A is unregistered. • OS4K_A calls ASC_A.	• Busy tone/display on calling party. • The call is properly cleared.	OK	ASC_A = switched off, OS4K sends 502 Bad Gateway (Currently not accessible, call clears after about 12 secs).
2-32	• OS4K_A is unregistered. • ASC_A calls OS4K_A.	• Busy tone/display on calling party. • The call is properly cleared.	OK	480 Temporarily not available by OS4K, caller clears after about 40 secs. Caller displays Not reachable.
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 503 Service unavailable (phone displays Not Possible and caller gets fast busy for about 12 secs).

3.1.3. Basic Calls Extended

Call History

Test Name	Test Description	Functional Checks	Result	Comments
3-1	• 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).	POK	Anonymous incoming calls are not displayed in Ascom DECT handset call list.
3-2	• 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	
3-3	• 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	

Test Name	Test Description	Functional Checks	Result	Comments
3-4	• 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	

Long Duration Call

Test Name	Test Description	Functional Checks	Result	Comments
3-5	• 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	
3-6	• 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	

Do-Not-Disturb

Test Name	Test Description	Functional Checks	Result	Comments
3-7	• 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).
3-8	• 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 3-7.
3-9	• 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	

CLIR

Test Name	Test Description	Functional Checks	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. Caller has *444007 at call list.

Test Name	Test Description	Functional Checks	Result	Comments
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 displays *XXX* NoNumberProvided.
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.	NA	When dialing *449<PSTN number>, the called party sees the (ISDN) number of the caller. OS4K sends anonymous in "From" header and "Privacy:id". At mediatrix syslog traces, the "Call Router" message to the provider includes name = Anonymous. 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.
3-14	• CELL activates identity restriction. • CELL calls ASC_A.	• Number is not displayed on called phone.	OK	ASC_A displays *********** External call.

3.1.4. Telephony Features

Hold, Consultation, Toggle, Call Waiting

Test Name	Test Description	Functional Checks	Result	Comments
4-1	• ASC_A calls ASC_B 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	
4-2	• 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	
4-3	• 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	
4-4	• 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.	OK	On Ascom handset, press R2 to toggle between calls and R1 to hang up active call and switch to held call.

Test Name	Test Description	Functional Checks	Result	Comments
		• On Hold announcement is played to held party. • Speech path in both directions. • Both ends idle after call clearing.		
4-5	• 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	
4-6	• 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	
4-7	• 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	
4-8	• 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#.

Callback (No Reply)

Test Name	Test Description	Functional Checks	Result	Comments
4-9	• 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 menu or by dialing the feature access code (e.g., *10). • ASC_B establishes a call with another subscriber and then hangs up.	• CCNR feature is successfully activated. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	NA	Callback isn't supported by OS4K SIP stations.

Test Name	Test Description	Functional Checks	Result	Comments
	• 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.			
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 4-9.
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. • After the CCNR request has been deactivated, OS4K should not send any notification that the original callback target has become available to receive calls.	OK	

Callback (On Busy)

Test Name	Test Description	Functional Checks	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,	• 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 4-9.

Test Name	Test Description	Functional Checks	Result	Comments
	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.			
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 and once ASC_A answers, OS4000 calls OS4K_A. • OS4K_A 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 4-9.
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	

Call Park

Test Name	Test Description	Functional Checks	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 hanging 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	NA	Same as 4-17.

Test Name	Test Description	Functional Checks	Result	Comments
		after call pick up. • Media clipping and speech path after forwarding. • Call is disconnected after hanging up.		
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 answers. • OS4K_A hangs up.	• On Hold announcement played on OS4K_A while the call is parked. • ASC_A successfully retrieves parked call. • Number presentation on ringing and after call pick up. • Media clipping and speech path after forwarding. • Call is disconnected after hanging up.	NA	Same as 4-17.
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 hanging up.	OK	

Call Pickup

Test Name	Test Description	Functional Checks	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	Call pickup group notification on Ascom handset 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 hangs up at the end of the communication.	• ASC_A is able to pick up the call. • No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.	POK	Same as 4-21.
4-24	• OS4K_A and ASC_A belong to the	• OS4K_A is able to pick up the call.	OK	

Test Name	Test Description	Functional Checks	Result	Comments
	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.	• No media clipping when connecting both ends. • Speech path in both directions. • Both ends idle after call clearing.		

Call Transfer (Attended)

Test Name	Test Description	Functional Checks	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 between ASC_A and OS4K_B, ASC_A hangs up.	• 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 ringing and after call pick up. • Both ends idle after call clearing.	OK	
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	• On Hold announcement is played to PSTN while reaching ASC_B. • No Media clipping and proper speech path after transfer when	OK	

Test Name	Test Description	Functional Checks	Result	Comments
	ASC_B. • After transfer is completed and communication is established between PSTN and ASC_B, PSTN hangs up.	ASC_B picks up the call. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.		
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. • Both ends idle after call clearing.	OK	

Call Transfer (Semi-Attended)

Test Name	Test Description	Functional Checks	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 due to DECT handset limitation.
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	
4-34	• OS4K_A calls ASC_A. • ASC_A consults to ASC_B. • 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,	• On Hold announcement is played to OS4K_A after 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	POK	Same as 4-31.

Test Name	Test Description	Functional Checks	Result	Comments
	OS4K_A hangs up.	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.

Call Transfer (Blind)

Test Name	Test Description	Functional Checks	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 between ASC_A and ASC_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 ASC_C are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	Press RR4008# on Ascom handset.
4-38	• ASC_A calls OS4K_C. • OS4K_C 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	
4-39	• ASC_A calls OS4K_C. • OS4K_C blind transfers to OS4K_B. • ASC_A receives ring back tone. • After transfer is completed and communication is established between ASC_A and OS4K_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 OS4K_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	
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	• On Hold announcement to OS4K_A after consultation and before blind transfer is completed. • No Media clipping and proper	OK	

Test Name	Test Description	Functional Checks	Result	Comments
	communication is established between OS4K_A and ASC_B, OS4K_A hangs up.	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.		
4-41	• PSTN calls OS4K_C. • OS4K_C blind transfers to ASC_B. • PSTN receives ring back tone. • After transfer is completed and communication is established between PSTN and ASC_B, PSTN hangs up.	• 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 ASC_B are connected. • Number presentation on ringing and after call pick up. • Both ends idle after call clearing.	OK	
4-42	• PSTN calls ASC_A. • ASC_A consults to OS4K_B. • ASC_A performs blind transfer in menu. • 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	

Call Deflect

Test Name	Test Description	Functional Checks	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 call 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_C. • OS4K_C doesn't answer and deflects the call to ASC_B. • Leave ASC_B ring for a few seconds and pick up the call. • After call deflect is completed and communication is established between ASC_A and ASC_B, ASC_A hangs up.	• Calling party receives forwarding call display notification. • Ring back tone heard by calling party. • Number presentation on ringing and after call pick up. • Media clipping and speech path after forwarding. • Both ends idle after call clearing.	POK	ASC_A continues to display OS4K_C on ringing after call deflect. After ASC_B answers, ASC_A is correctly updated and displays ASC_B (Unify Jira ticket: OSFOURK-7353).

Call Forwarding (Unconditional)

Test Name	Test Description	Functional Checks	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.	• Call forwarding info is displayed on the phone. • Calling party receives forwarding call display notification. • Ring back tone heard by calling party.	OK	Press, *21*4008 to activate on Ascom handset (deactivate with #21#).

Test Name	Test Description	Functional Checks	Result	Comments
	• 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.	• Number presentation on ringing and after call pick up. • Media clipping and speech path after forwarding. • Both ends idle after call clearing.		
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. • Both ends idle after call clearing.	OK	
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	

Call Forwarding (Busy)

Test Name	Test Description	Functional Checks	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	• Call forwarding info is displayed on the phone. • Calling party receives forwarding call display notification. • Ring back tone heard by calling party. • Number presentation on ringing and after call pick up. • Media clipping and speech path after forwarding.	OK	Press *67*4008# to activate on Ascom handset (deactivate with #67#).

Test Name	Test Description	Functional Checks	Result	Comments
	communication is established between ASC_A and ASC_C, ASC_A hangs up.	• Both ends idle after call clearing.		
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 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	
4-51	• 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	

Call Forwarding (No Reply)

Test Name	Test Description	Functional Checks	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	• Call forwarding info is displayed on the phone. • Calling party receives forwarding call display notification. • Ring back tone heard by calling party. • Number presentation on ringing and after call pick up. • Media clipping and speech path after forwarding.	POK	*61*4008# activate on DECT handset (deactivate with #61#). ASC_C is updated on the display with ASC_A after accepting the call. On ringing state ASC_A continues to display ASC_B (Unify Jira ticket: OSFOURK-7353).

Test Name	Test Description	Functional Checks	Result	Comments
	between ASC_A and ASC_B, ASC_A hangs up.	• 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.	POK	Same behavior as 4-53.
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	

MLHG

Test Name	Test Description	Functional Checks	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.	• 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.	OK	

Test Name	Test Description	Functional Checks	Result	Comments
	• Call is cleared at the end of the communication.	• Both ends idle after call clearing.		
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	

Voicemail

Test Name	Test Description	Functional Checks	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. • Both ends idle after call clearing.	NA	MWI isn't supported for OS4K SIP stations.
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 4-61.
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	

Conference

Test Name	Test Description	Functional Checks	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.	• Media clipping and proper speechpath after conferencing participants in the conference. • Number(s) presentation on	NA	Same as 4-63.

Test Name	Test Description	Functional Checks	Result	Comments
	• 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.	conference participants. • Conference participants are still on the conference after ASC_A hangs up. • Number(s) presentation after ASC_A hangs up on conference participants.		
4-65	• 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.

3.1.5. Audio Features

Codecs

Test Name	Test Description	Functional Checks	Result	Comments
5-1	• 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	
5-2	• IP-DECT base station is configured to have G.723 as first priority. • ASC_A calls OS4K_A.	• Connection is established with G.723. • Media clipping and speechpath.	OK	
5-3	• 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	
5-4	• 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	
5-5	• ASC_A is configured to support G.723 only. • OS4K_A calls ASC_A.	• Connection is established with G.723. • Media clipping and speechpath.	OK	
5-6	• ASC_A is configured to support G.729 only. • OS4K_A calls ASC_A.	• Connection is established with G.729. • Media clipping and speechpath.	OK	
5-7	• IP-DECT base station is configured to have G.722.2 (AMR-WB), as first priority. • ASC_A calls ASC_C.	• Connection is established with G.722.2. • Media clipping and speechpath.	NA	Communication falls back to G.711, because G.722.2 is screened by the system. OS4K doesn't support G.722.2.
5-8	• 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

Test Name	Test Description	Functional Checks	Result	Comments
				the call with "503 Service Unavailable - No circuit/channel available".
5-9	• 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.

3.1.6. Redundancy

OS4000 Node Switchover

Test Name	Test Description	Functional Checks	Result	Comments
6-1	• 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.	NA	Test environment is Simplex.
6-2	• The node1 is put back into service. • Perform an incoming and outgoing call to ASC_A.	• The switchover should not be seen on the IP-DECT handsets. • ASC_A is able to make and receive calls.	NA	Test environment is Simplex.

IP-DECT Base Station Switchover

Test Name	Test Description	Functional Checks	Result	Comments
6-3	• 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.	OK	
6-4	• In mirror configuration the functions do not return to Master automatically. • After scenario 6-3, while ASC_A is in a call with ASC_B, switch back manually to the first base station via GUI. • After switchover to the first base station, ASC_A calls ASC_B. • Call is cleared at the end of the communication.	• The redundant mirror Base Station becomes the Master and takes over the Handsets and the SIP-Registration. • Calls are possible after switchover.	OK	

3.2. Summary

Explanation of acronyms:

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

Test results aggregated:

Category	Planned	OK	POK	NOK	NA	NT
Registration	4	4	-	-	-	-
Basic Calls	29	29	-	-	-	-
Basic Calls Extended	14	12	1	-	1	-
Telephony Features	65	44	9	-	12	-
Audio Features	9	7	1	-	1	-
Redundancy	4	2	-	-	2	-
Totals	**125**	**98**	**11**	**0**	**16**	**0**

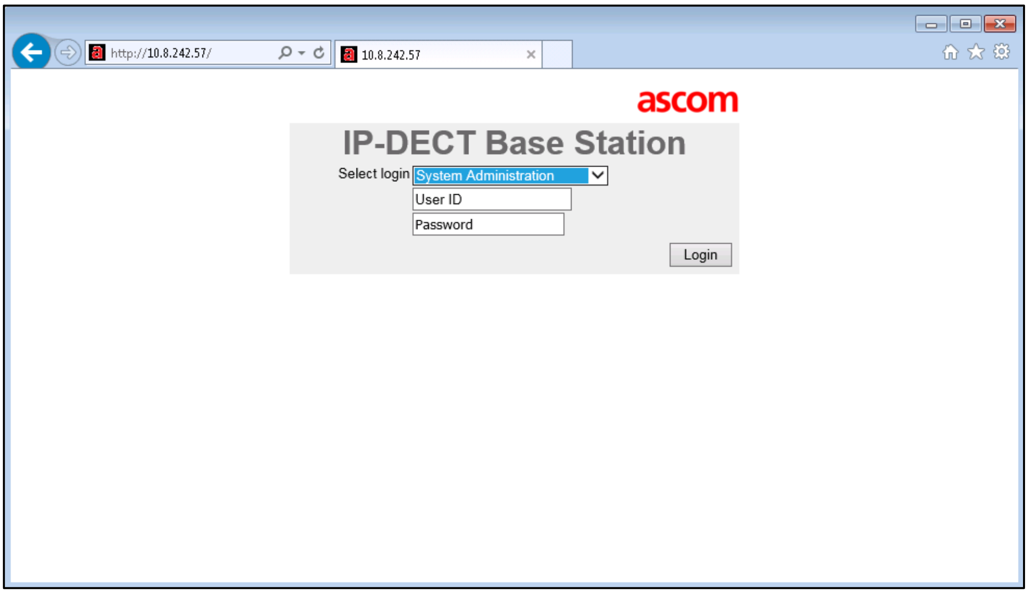
4. Configuration Data

4.1. Ascom IP-DECT Base Stations Configuration (Mirror Mode)

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. The Ascom wireless IP-DECT Base Stations are configured in a Master/Standby mode to provide redundancy. The following configuration steps detail the configuration process used to configure an Ascom wireless IP-DECT Base Station in Master mode only.

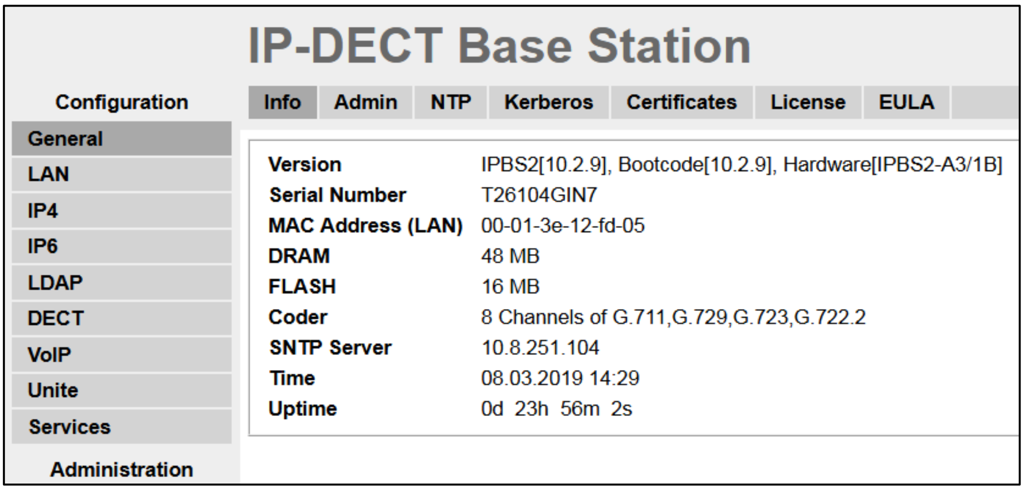
Default or non-project specific Ascom IP-DECT system configuration will not be referenced to in subsequent paragraphs.

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



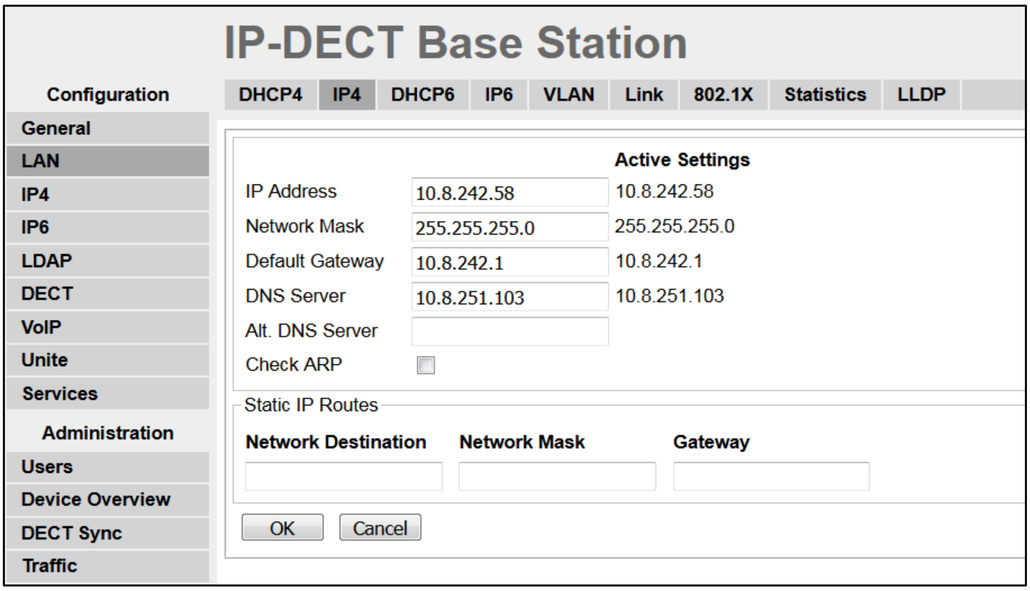
Click the System administration link and enter the appropriate credentials to access the Ascom IP-DECT Base Station and then click on **OK**.

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:

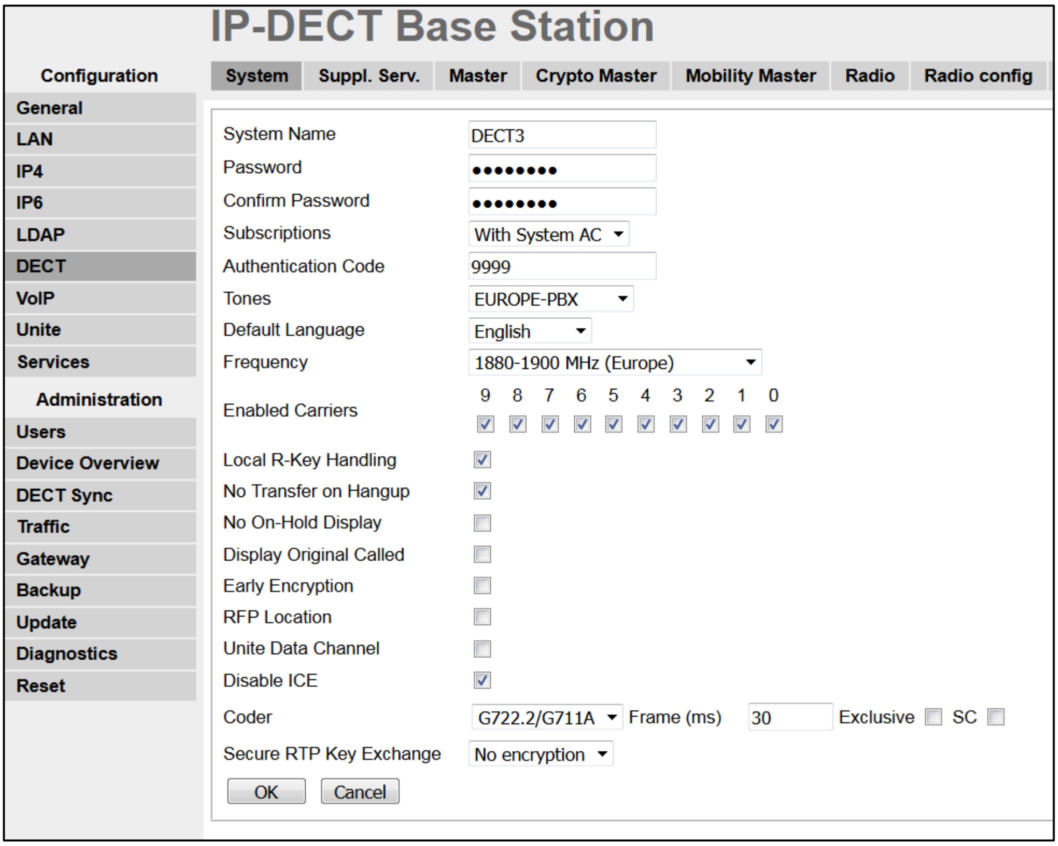
- **IP Address** 10.8.242.58
- **Network Mask** 255.255.255.0
- **Default Gateway** 10.8.242.1
- **DNS Server** 10.8.251.103

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

Note 1: Master IP-DECT Base Station with IP: 10.8.242.57 was shut down at the time the above screenshot was taken.

Note 2: DNS Server isn't used during testing activities because IPs instead FQDNs were utilized in the test environment setup.

4.1.3. DECT System



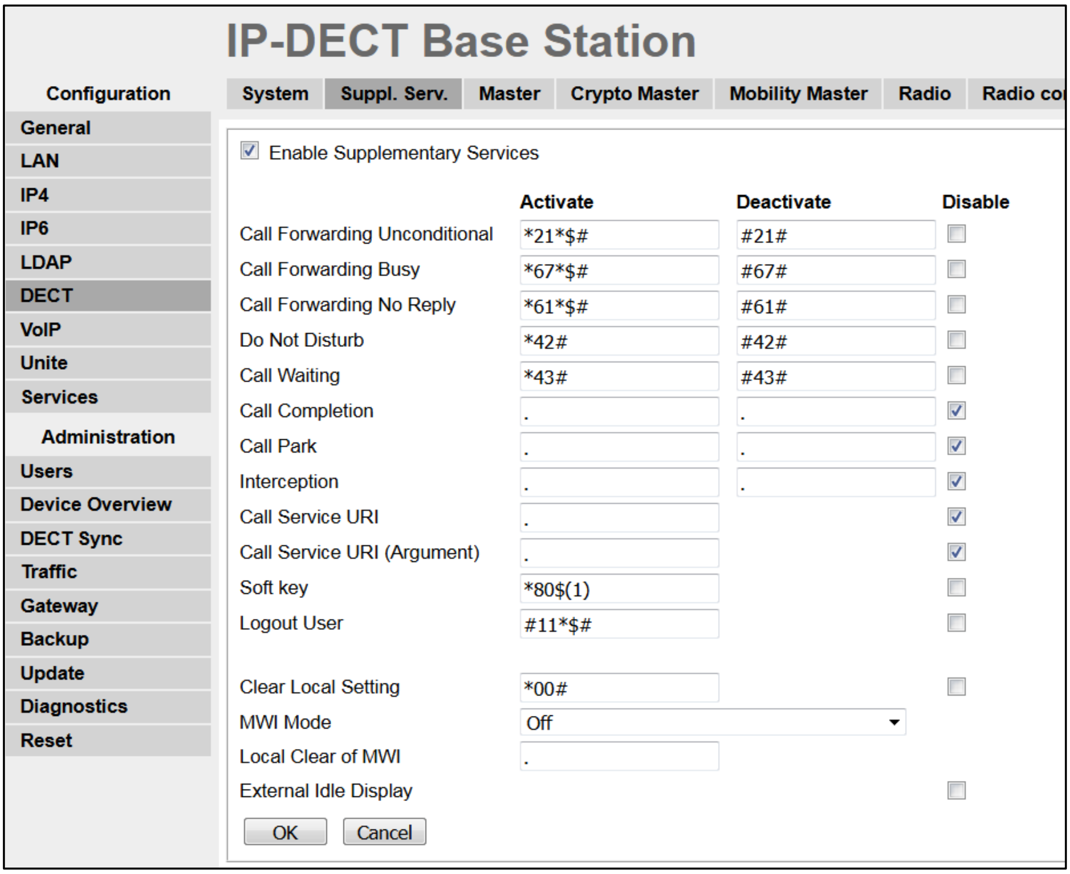
Click on the System tab and enter the following:

- **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
- **Coder** Select the required Coder from the Coder dropdown box (Coder = G722.2/G711A, Frame (ms) = 30)

Click the **OK** button to continue.

Note: During testing the *"ptime"* was changed to 30ms, while the default value is 20ms. It is recommended that the PBX 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:

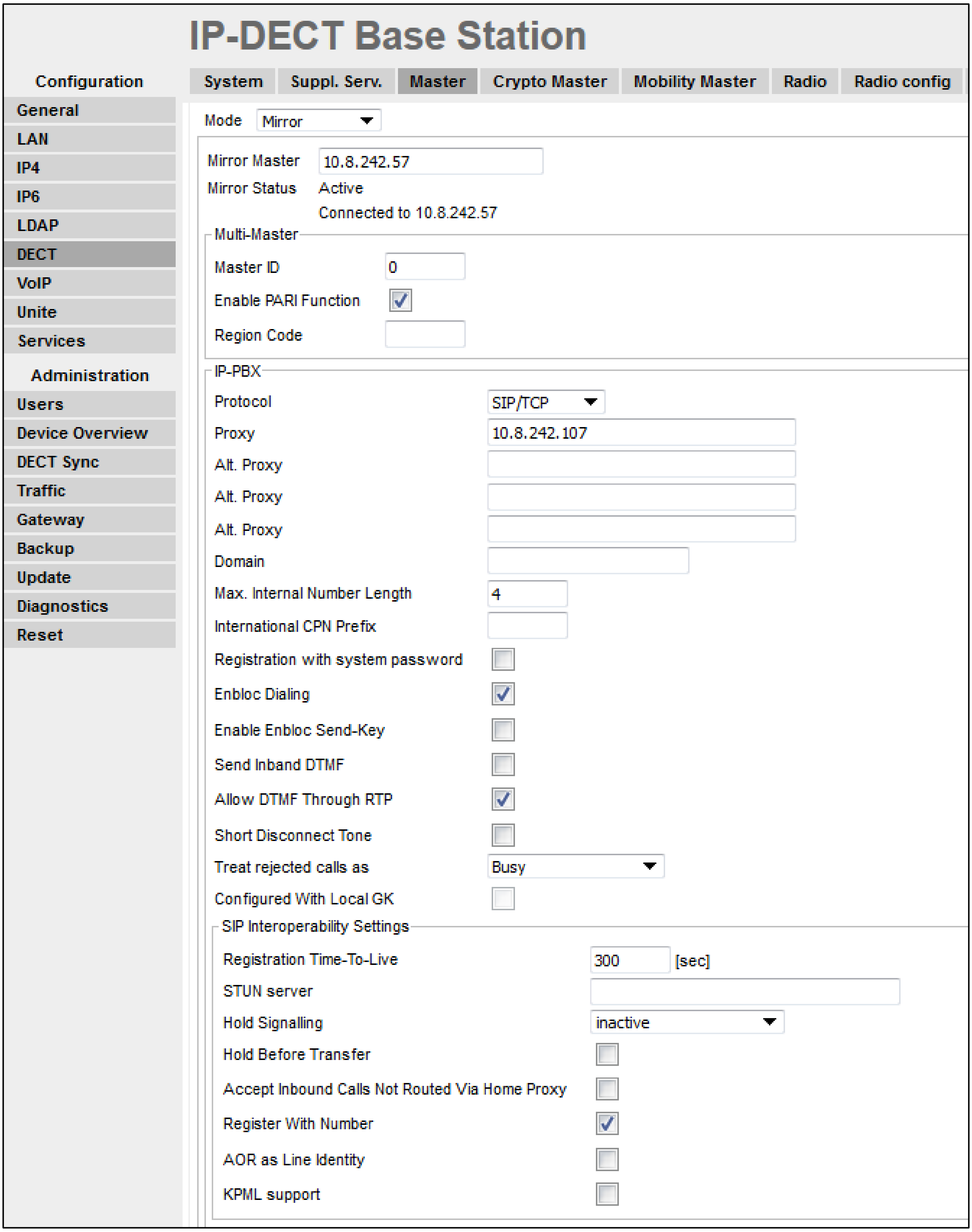
- **Enable Supplementary Services check box** Checked
- **Call Forwarding Unconditional** Activate = *21*$#, Deactivate = #21#
- **Call Forwarding Busy** Activate = *67*$#, Deactivate = #67#
- **Call Forwarding No Reply** Activate = *61*$#, Deactivate = #61#
- **Do not Disturb** Activate = *42*$#, Deactivate = #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.

4.1.5. IP-DECT Master – Active Base Station

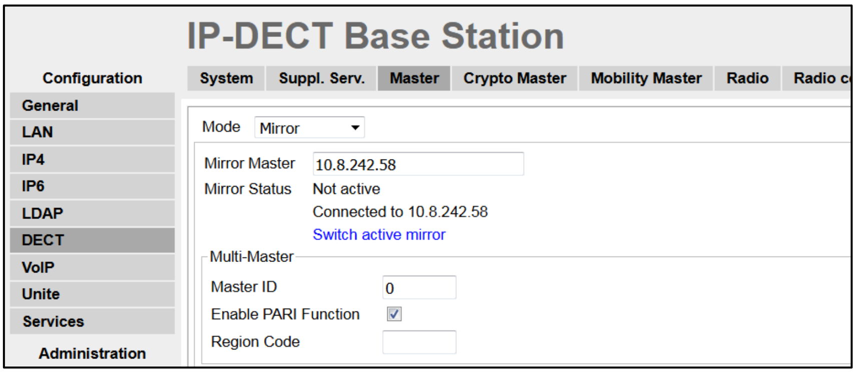


Navigate to **DECT**, click on **Master** and enter the following:

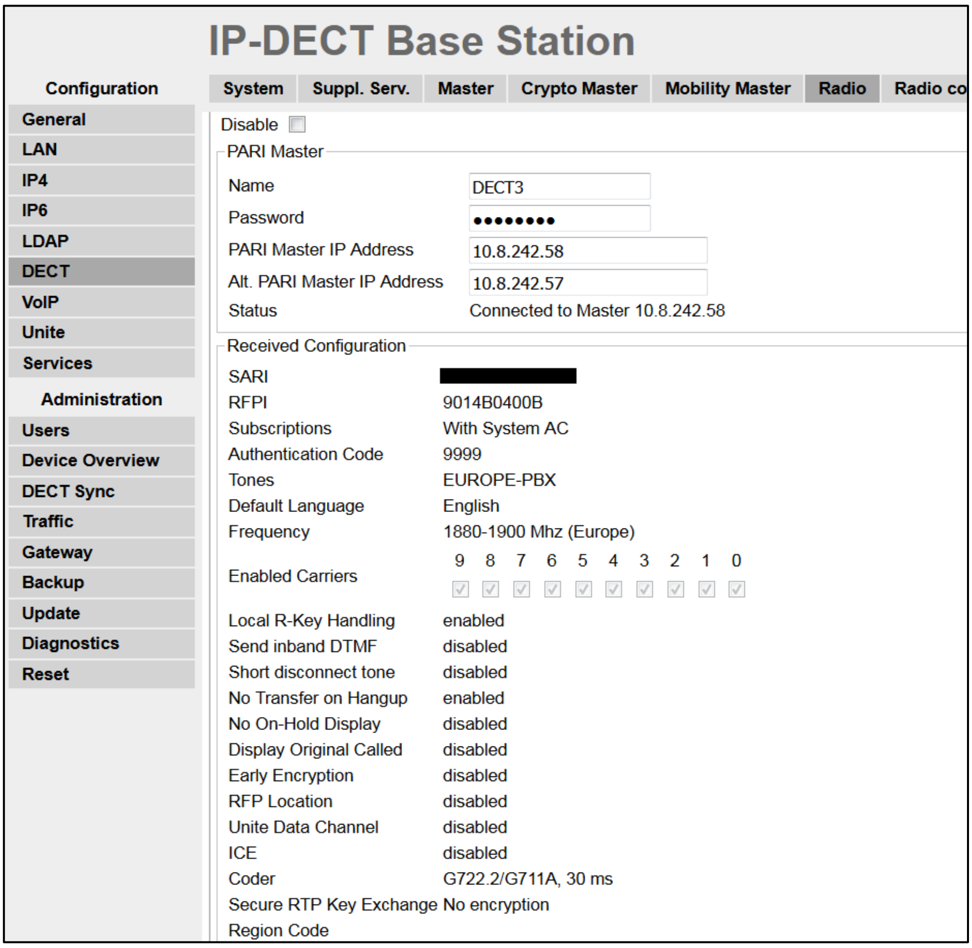
▪ **Mode**	Select Mirror from the dropdown box
▪ **Mirror Master IP address**	Mirrored Master base station IP: 10.8.242.57
▪ **Enable PARI Function check box**	Checked
▪ **Protocol**	Select SIP/TCP from the dropdown box
▪ **Proxy**	STMI4 IP: 10.8.242.107
▪ **Enbloc Dialling check box**	Checked
▪ **Allow DTMF through RTP check box**	Checked
▪ **Registration Time-To-Live**	300

Click the OK button to continue (not shown).

4.1.6. IP-DECT Master – No Active Base Station



4.1.7. DECT Radio – Active Base Station



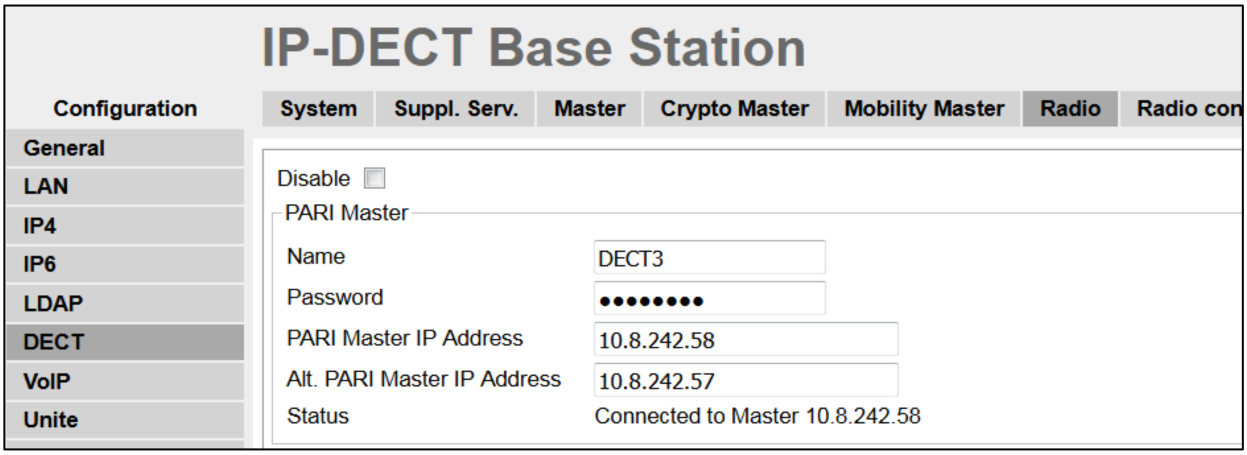
Navigate to **Radio** and select the Admin tab and enter the following:

- **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** — 10.8.242.58 (this base station became master after switchover test 6-3). If this device is the PARI Master, we could enter 127.0.0.1
- **Alt. PARI Master IP Address** — 10.8.242.57. If this is the Standby PARI Master, we could enter 127.0.0.1

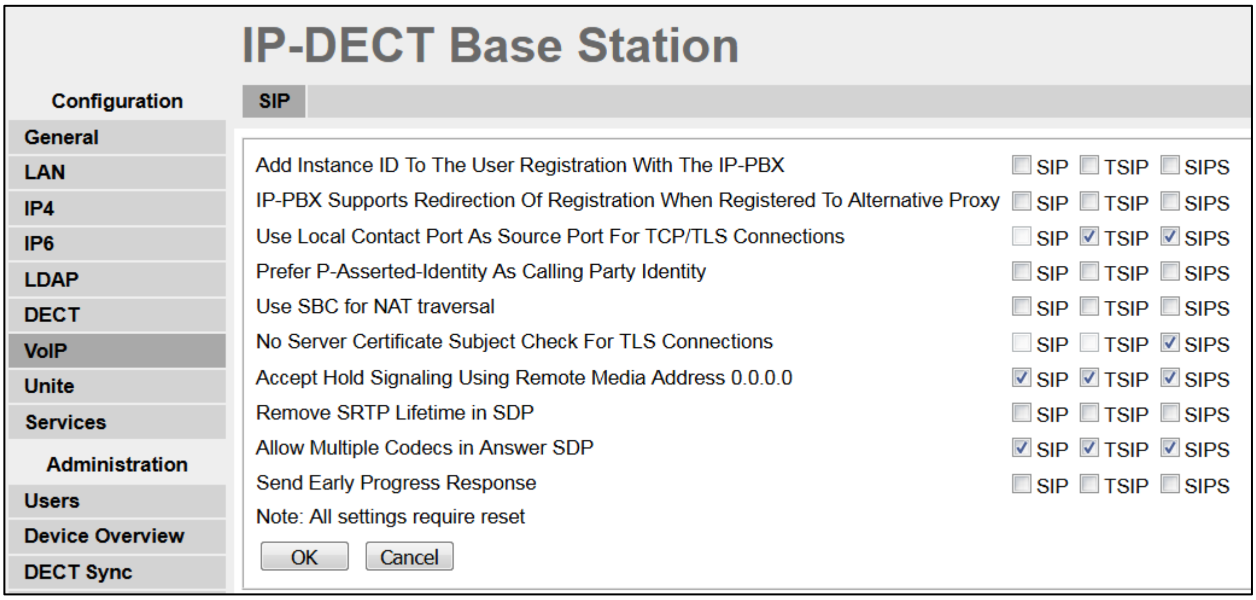
Click on the **OK** Button to save.

Note: It is recommended to use static IP addresses for mirrored base stations.

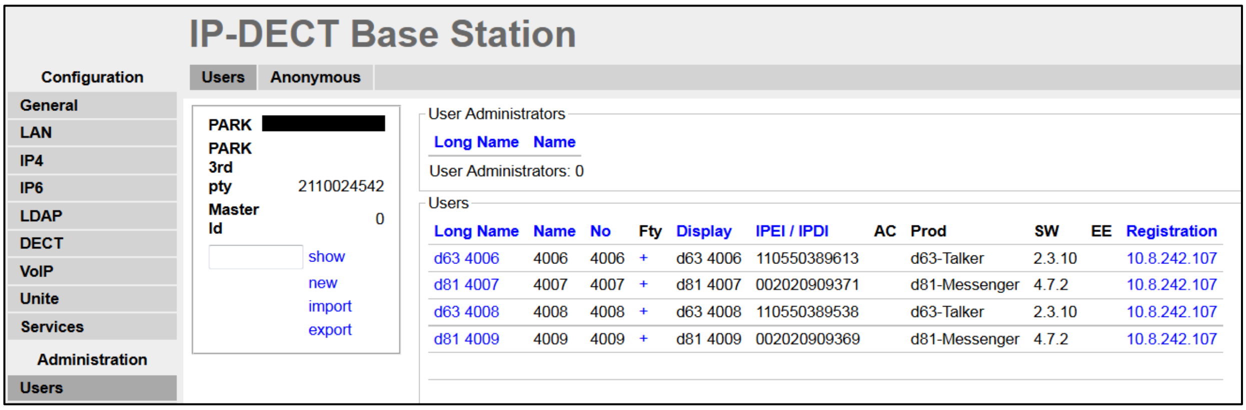
4.1.8. DECT Radio – No Active Base Station



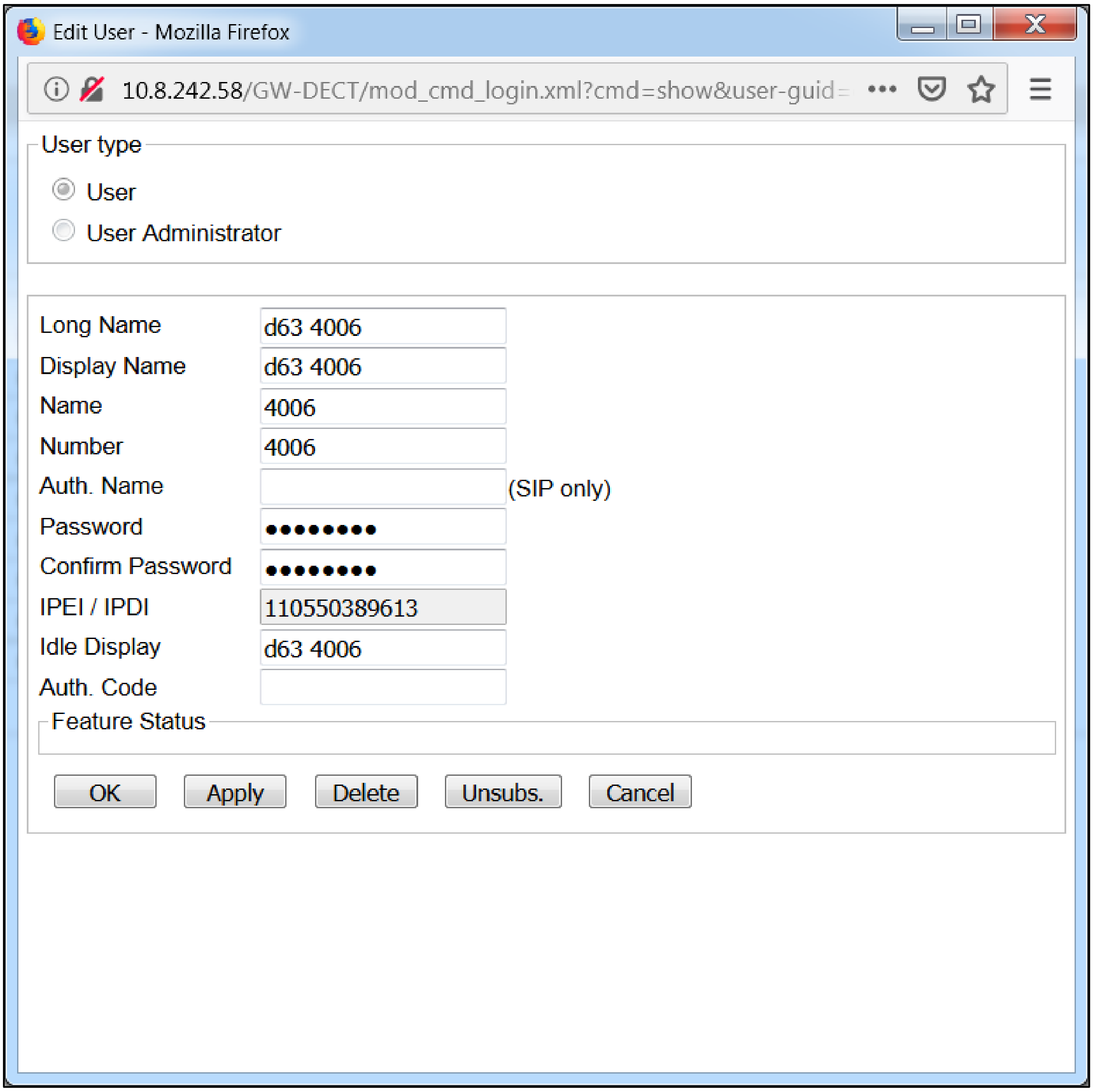
4.1.9. VoIP



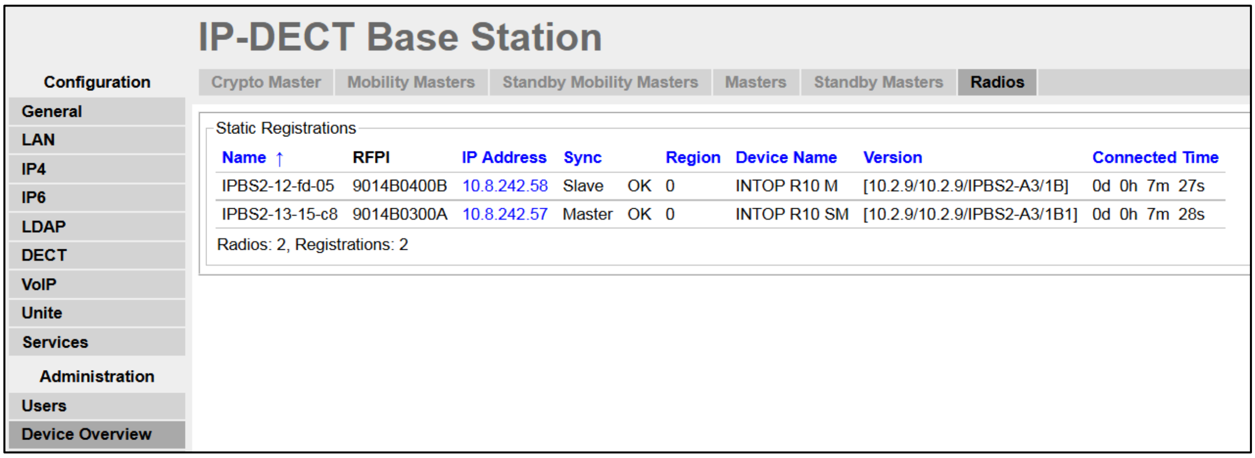
4.1.10. IP-DECT Users



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



4.1.11. 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",SECZON="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",SECZON="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",SECZON="OS4K";
```

4.2.2. AMO Configuration for Xpressions, Mediatrix 4402plus SIP Trunks

The *"Common"* section below describes the general configuration of the STMI2 board for both Xpressions & Mediatrix 4402plus SIP trunks.

================================Common Configuration=======================================

Add function block for STMI2 board:

```
ADD-BFDAT:FCTBLK=3,FUNCTION=HG3550,BRDBCHL=BCHL60;
```

Add STMI2 board:

```
ADD-BCSU:MTYPE=IPGW,LTG=1,LTU=17,SLOT=1,PARTNO="Q2316-X",FCTID=1,LWVAR="0",FCTBLK=3,BCHL3550=60,ALARMNO=0,IPMODE=IPV4,DHCPV4=NO,DHCPV6=NO;
```

Add IP address for STMI2 board:

```
ADD-CGWB:LTU=17,SLOT=1,SMODE=NORMAL,IPADR=10.8.242.106,NETMASK=255.255.255.0;

CHANGE-CGWB:MTYPE=CGW,LTU=17,SLOT=1,TYPE=GLOBIF,PATTERN=213,VLAN=NO,VLANID=0,
    DEFRT=10.8.242.1,BITRATE=AUTONEG,TRPRSIP=30,TRPRSIPQ=0,TRPRH323=0,TPRH323A=0,T
    LSP=4061,DNSIPADR=0.0.0.0,DNSIPAD2=0.0.0.0,USEWANIF=NO,WPUBIP=0.0.0.0,SIPTCPP=
    5060,SIPTLSP=5061;

CHANGE-CGWB:MTYPE=CGW,LTU=17,SLOT=1,TYPE=SERVIF,LOGINTRM="TRM",PASSW="HICOM";

CHANGE-CGWB:MTYPE=CGW,LTU=17,SLOT=1,TYPE=ASC,UDPPRTLO=29100,UDPPRTHI=30099,
    TOSPL=184,TOSSIGNL=104,T38FAX=YES,RFCFMOIP=NO,RFCDTMF=YES,REDRFCTN=YES,PRIO=PR
    IO1,CODEC=G711A,VAD=NO,RTP=30;

CHANGE-CGWB:MTYPE=CGW,LTU=17,SLOT=1,TYPE=ASC,PRIO=PRIO2,CODEC=G729A,VAD=NO,RTP=20;

CHANGE-CGWB:MTYPE=CGW,LTU=17,SLOT=1,TYPE=ASC,PRIO=PRIO3,CODEC=G723,VAD=NO,RTP=30;

CHANGE-CGWB:MTYPE=CGW,LTU=17,SLOT=1,TYPE=ASC,PRIO=PRIO4,CODEC=NONE,VAD=NO,RTP=20;

CHANGE-CGWB:MTYPE=CGW,LTU=17,SLOT=1,TYPE=ASC,PRIO=PRIO5,CODEC=NONE,VAD=NO,RTP=20;

CHANGE-CGWB:MTYPE=CGW,LTU=17,SLOT=1,TYPE=ASC,PRIO=PRIO6,CODEC=NONE,VAD=NO,RTP=20;

CHANGE-CGWB:MTYPE=CGW,LTU=17,SLOT=1,TYPE=ASC,PRIO=PRIO7,CODEC=G729AB,VAD=YES,RTP=20;

CHANGE-CGWB:MTYPE=CGW,LTU=17,SLOT=1,TYPE=ASC,PRIO=PRIO8,CODEC=NONE,VAD=NO,RTP=20;

CHANGE-CGWB:MTYPE=CGW,LTU=17,SLOT=1,TYPE=ASC,PRIO=PRIO9,CODEC=NONE,VAD=NO,RTP=20;

CHANGE-CGWB:MTYPE=CGW,LTU=17,SLOT=1,TYPE=GKDATA,PRIGKPN=1719,
    PRIGKID1="PRIMARYRASMANAGERID",SECGKPN=1719,SECGKID1="SECONDARYRASMANAGERID",T
    IMTLIVE=120;

CHANGE-CGWB:MTYPE=CGW,LTU=17,SLOT=1,TYPE=MGNTDATA,MGNTPN=8000,BUSPN=443,UIMODE=CLASSIC;

CHANGE-CGWB:MTYPE=CGW,LTU=17,SLOT=1,TYPE=DMCDATA,DMCCONN=0,SMP=YES,SMP4OSV=NO;

CHANGE-CGWB:MTYPE=CGW,LTU=17,SLOT=1,TYPE=WBMDATA,LOGINWBM="HP4K-DEVEL",ROLE=ENGR;

CHANGE-CGWB:MTYPE=CGW,LTU=17,SLOT=1,TYPE=WBMDATA,LOGINWBM="HP4K-SU",ROLE=SU;

CHANGE-CGWB:MTYPE=CGW,LTU=17,SLOT=1,TYPE=WBMDATA,LOGINWBM="HP4K-ADMIN",ROLE=ADMIN;

CHANGE-CGWB:MTYPE=CGW,LTU=17,SLOT=1,TYPE=WBMDATA,LOGINWBM="HP4K-READER",ROLE=READONLY;

CHANGE-CGWB:MTYPE=CGW,LTU=17,SLOT=1,TYPE=GWDATA,GWID1="PRIMARYRASMANAGERID";

CHANGE-CGWB:MTYPE=CGW,LTU=17,SLOT=1,TYPE=H235DATA,SECSUBS=NO,SECTRNK=NO,
    GLOBID1="h235securedgateway",TIMEWIN=100,GLOBPW=242-191-30-119-188-83-173-161-
    43-0-70-36-218-74-169-221-78-102-174-170;

CHANGE-CGWB:MTYPE=CGW,LTU=17,SLOT=1,TYPE=SIPTRERH,GWAUTREQ=NO;

CHANGE-CGWB:MTYPE=CGW,LTU=17,SLOT=1,TYPE=DLSDATA,DLSIPADR=10.6.25.5,DLSPORT=18443,DLSACPAS=YES;

CHANGE-CGWB:MTYPE=CGW,LTU=17,SLOT=1,TYPE=JB,AVGDLYV=40,MAXDLYV=120,MINDLYV=20,
    PACKLOSS=4,AVGDLYD=60,MAXDLYD=200,JBMODE=2;

CHANGE-CGWB:MTYPE=CGW,LTU=17,SLOT=1,TYPE=MANLANIF,MIPADR=0.0.0.0,MNETMASK=0.0.0.0,
    MVLAN=NO,MVLANID=0,MDEFRT=0.0.0.0;
```

Add Class Of Parameter:

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

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&DSDL&NPIS&ANNC;
```

Add Class Of Service:

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

```
CHANGE-COSSU:TYPE=COS,COS=1,AVCE=TA&TNOTCR&COSXCD&DATA&CFNR&VCE&RSVLN&DICT;
```

```
CHANGE-COSSU:TYPE=COS,COS=1,AVCE=SPKR&FWDNWK&TTT&MSN&CFB&MULTRA&FWDEXT&CCBS;
```

```
CHANGE-COSSU:TYPE=COS,COS=1,AVCE=CW;
```

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

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

==================================Xpressions Configuration===

Add Bundle:

```
ADD-BUEND:TGRP=1,NAME="XPRESSIONS
    ",NO=16,TRACENO=0,ACDTHRH=*,PRIONO=2,TDDRFLAG=OFF,GDTRRULE=0,ACDPMGRP=0,CHARCO
    N=NEUTRAL;
```

Add SIP trunk:

```
ADD-TDCSU:OPT=NEW,PEN=1-17-001-0,COTNO=1,COPNO=1,DPLN=0,ITR=0,COS=1,LCOSV=1,LCOSD=1,CCT="XPR
    ",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,C
    ARRIER=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,SECL
    EVEL=TRADITIO,HMUSIC=0,CALLTIM=60,WARNTIM=60,DEV=HG3550IP,BCHAN=1&&10,BCNEG=N,
    BCGR=1,LWPP=0,LWLT=0,LWPS=0,LWR1=0,LWR2=1,DMCALLWD=Y;
```

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,INF
    O="",PDNNO=0,CHARCON=NEUTRAL,CONFTONE=NO,RERINGRP=NO,NOPRCFWD=NO,NITO=NO,CLNAM
    EDL=NO,FWDSWTCH=NO,LINFEMER=NO,NOINTRTE=NO;
```

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;
```

===============================Mediatrix 4402plus Configuration====================================

Add Bundle:

```
ADD-BUEND:TGRP=3,NAME="MEDIATRIX4402 OTE
    ",NO=30,TRACENO=0,ACDTHRH=*,PRIONO=2,TDDRFLAG=OFF,GDTRRULE=0,ACDPMGRP=0,CHARCO
    N=NEUTRAL;
```

Add SIP trunk:

```
ADD-TDCSU:OPT=NEW,PEN=1-17-001-1,COTNO=1,COPNO=1,DPLN=0,ITR=0,COS=1,LCOSV=1,LCOSD=1,CCT="MED4402 OTE
          ",DESTNO=0,PROTVAR=ECMAV2,SEGMENT=8,DEDSVC=NONE,TRTBL=GDTR,SIDANI=N,ATNTYP=CO,
          CBMATTR=NONE,TCHARG=N,SUPPRESS=0,ISDNIP=00,ISDNNP=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=3,SRCHM
          ODE=DSC,INS=Y,SECLEVEL=TRADITIO,HMUSIC=0,CALLTIM=60,WARNTIM=60,DEV=HG3550CO,BC
          HAN=1&&15,BCNEG=N,BCGR=1,LWPP=0,LWLT=0,LWPS=0,LWR1=0,LWR2=0,DMCALLWD=N;
```

Add SIP trunk destination:

```
ADD-RICHT:MODE=LRTENEW,LRTE=6,LSVC=ALL,NAME="MED4402-OTE     ",TGRP=3,DNNO=1-1-
          122,ROUTOPT=NO,DTMFCNV=FIX,DTMFTEXT="",DTMFPULS=PP80,ROUTATT=NO,EMCYRTT=NO,INF
          O="",PDNNO=0,CHARCON=NEUTRAL,CONFTONE=NO,RERINGRP=NO,NOPRCFWD=NO,NITO=NO,CLNAM
          EDL=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=3,ODR=10,LAUTH=1,CARRIER=1,ZONE=EMPTY,LATTR=WCHREG,VCCYC=4;
```

Add digits to dial plan:

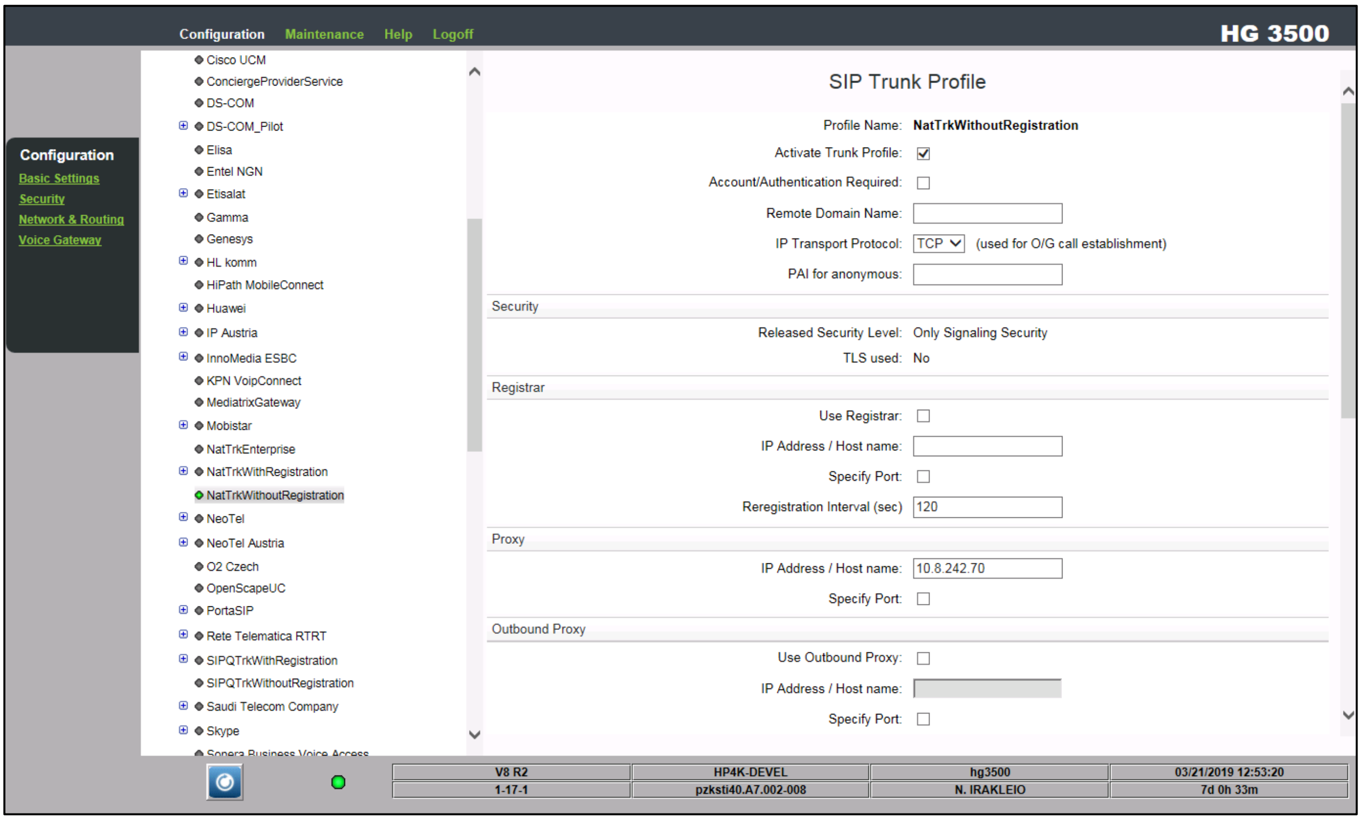
```
ADD-WABE:CD=9,DAR=CO,CHECK=N;
```

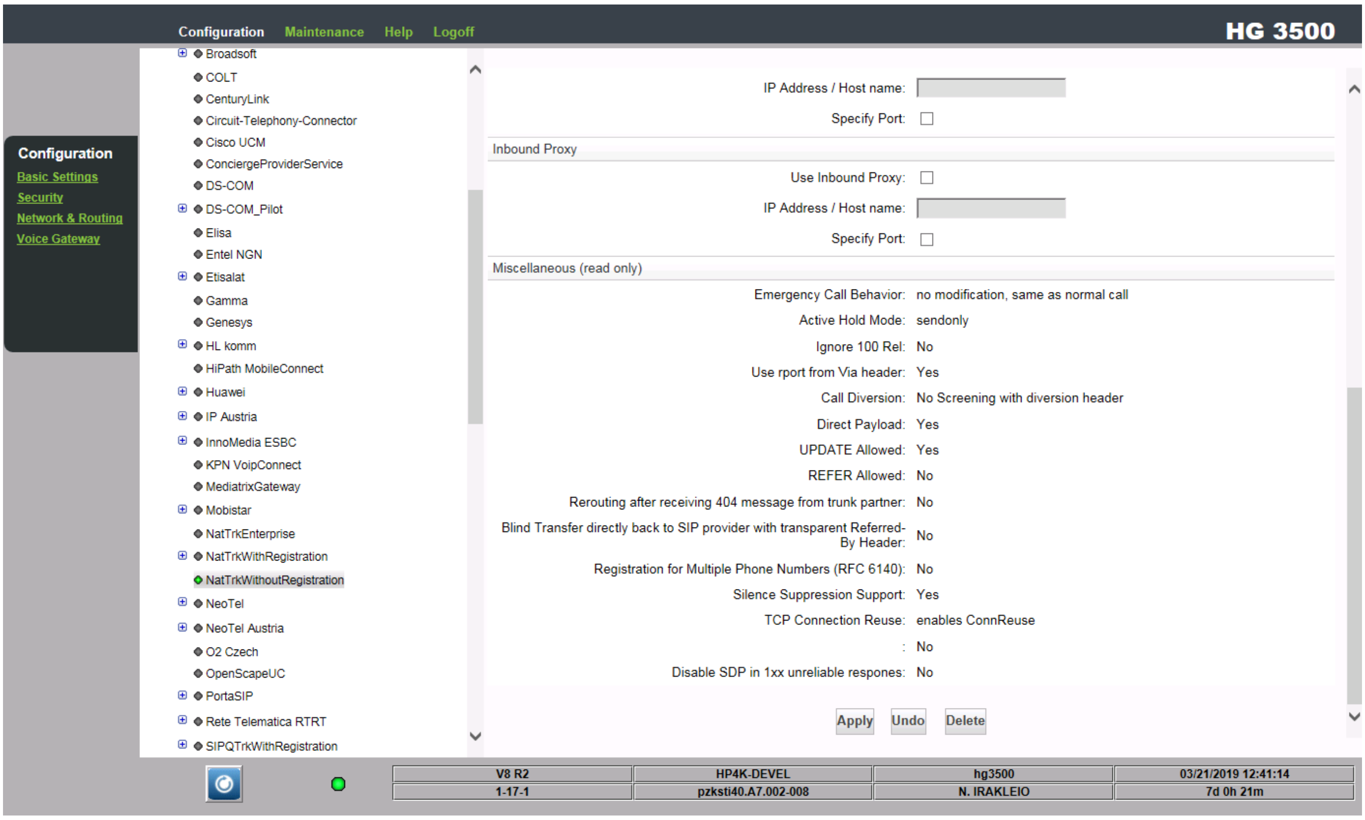
Add Administration LCR – Dialplan:

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

4.2.3. WBM Configuration for Xpressions, Mediatrix 4402plus SIP Trunks

As mentioned in section 2.2, the STMI2 board is either connected to Xpressions or to Mediatrix 4402plus.





Go to STMI2 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.

Note: When STMI2 is used for Mediatrix 4402plus, configure **Proxy – IP Address / Host Name** = 10.8.242.60.

4.2.4. OS4000 Dialing Plan

Global OS4000 system dialing plans and feature access codes:

```
DISPLAY-WABE:TYPE=GEN;
```

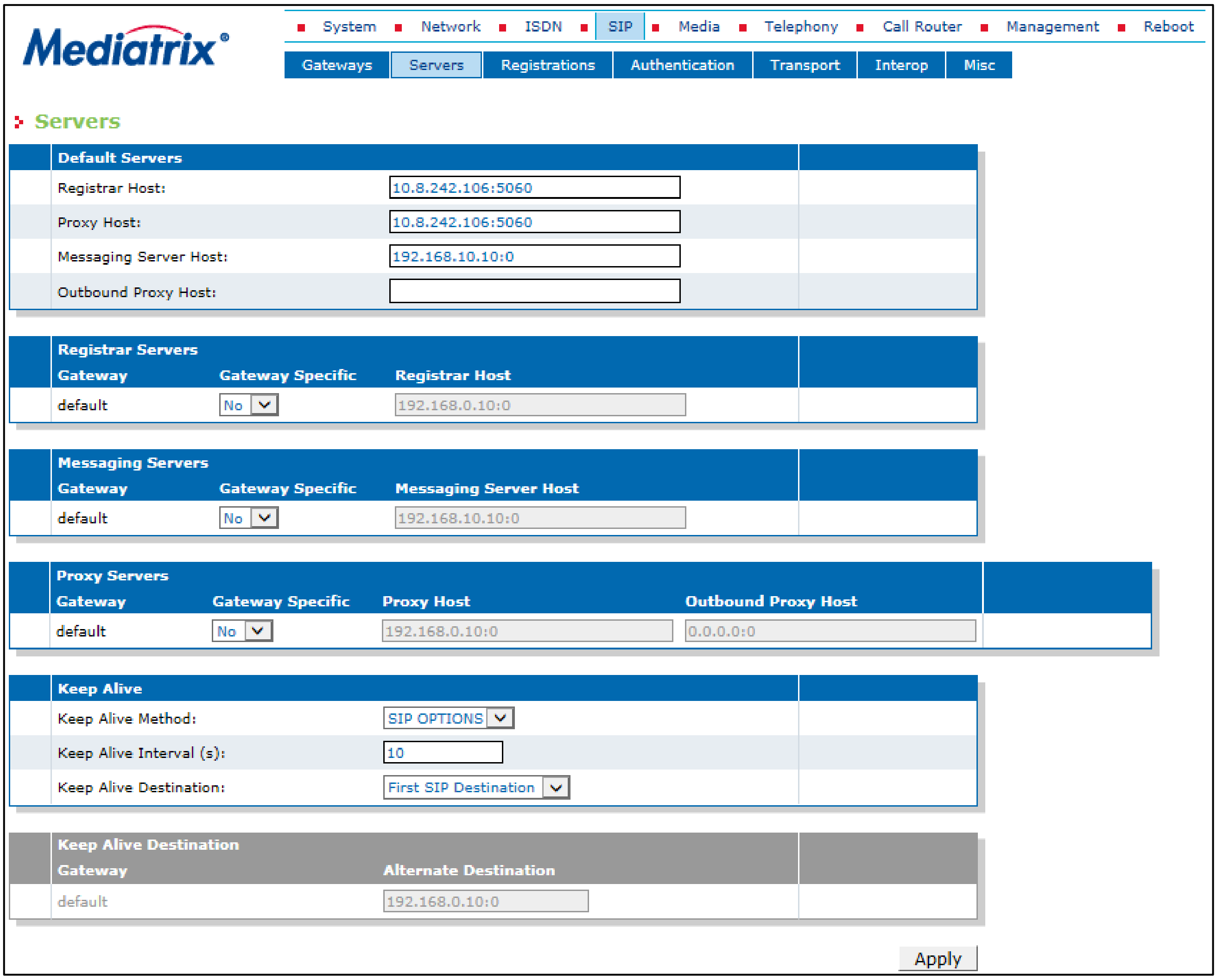
```
 ------------------------------------------------------------------------------
 | DIGIT INTERPRETATION                  VALID FOR ALL DIAL PLANS             |
 ------------------------------------------------------------------------------
 |                  |    CALL PROGRESS STATE      |NODE/DIGIT| 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  |                  |
 | 2                |. .**** ..*** **... ..... .*| CO       |                  |
 | 3580             |. .**** ***** **... ..... .*| STN      | R                |
 |                  |                            |          |DESTNO    0       |
 |                  |                            |          |DNNO     1-  1-100*|
 | 3581    - 3582   |. .**** ***** **... ..... .*| STN      |                  |
 |                  |                            |          |DESTNO    0       |
 |                  |                            |          |DNNO     1-  1-100*|
 | 3583             |. .**** ***** **... ..... .*| STN      | R                |
 |                  |                            |          |DESTNO    0       |
 |                  |                            |          |DNNO     1-  1-100*|
 | 3584             |. .**** ***** **... ..... .*| STN      |                  |
 |                  |                            |          |DESTNO    0       |
 |                  |                            |          |DNNO     1-  1-100*|
 ------------------------------------------------------------------------------
 | 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       |
-----------------------------------------------------------------------------
| 3585   - 3589   |. .**** ***** **... ..... .*| STN      | R                |
|                 |                            |          |DESTNO   0        |
|                 |                            |          |DNNO    1-  1-100*|
| 3590   - 3591   |. .**** ***** **... ..... .*| STN      |                  |
|                 |                            |          |DESTNO   0        |
|                 |                            |          |DNNO    1-  1-100*|
| 3592   - 3595   |. .**** ***** **... ..... .*| STN      | R                |
|                 |                            |          |DESTNO   0        |
|                 |                            |          |DNNO    1-  1-100*|
| 4000   - 4009   |. .**** ***** **... ..... .*| STN      |                  |
|                 |                            |          |DESTNO   0        |
|                 |                            |          |DNNO    1-  1-100*|
| 4010   - 4011   |. .**** ***** **... ..... .*| STN      | R                |
|                 |                            |          |DESTNO   0        |
|                 |                            |          |DNNO    1-  1-100*|
-----------------------------------------------------------------------------
| 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       |
-----------------------------------------------------------------------------
| 4012            |. .**** ***** **... ..... .*| STN      | R                |
|                 |                            |          |DESTNO  10        |
|                 |                            |          |DNNO    0-   0-  0 |
|                 |                            |          |PDNNO   0-   0-  0 |
| 556    - 559    |. .**** ***** **... ..... .*| TIE      |                  |
| 800             |. .**** ***** **... ..... .*| HUNT     |                  |
| 9               |. .**** ..*** **... ..... .*| CO       |                  |
| *10             |. *.... ..*.. ....* ..... .*| ACBK     |                  |
| *12             |. ....* ..*.. ..... ..... .*| VOICECAL |                  |
| *13             |. .**.. ..**. ..... ..... ..| SELFPA   |                  |
| *14             |. .**.. ..**. ..... ..... ..| SELFPA   |                  |
| *15             |. ....* ..**. ..... ..... .*| RECONNPA |                  |
| *20             |. .**** ***** **..* ..... ..| AFWDREM  |                  |
|                 |                            |          |CFREMVAR CFU      |
|                 |                            |          |CFREMSE  VOICE    |
| *21             |. ....* ..... ....* ..... ..| AFWDVCE  |                  |
| *22             |. ....* ..... ....* ..... ..| AFWDDTE  |                  |
-----------------------------------------------------------------------------
| 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       |
-----------------------------------------------------------------------------
| *23             |. ....* ..... ....* ..... ..| AFWDFAX  |                  |
| *24             |. ....* ..... ....* ..... ..| AFFWDVCE |                  |
| *25             |. ....* ..... ....* ..... ..| AFFWDDTE |                  |
| *26             |. ....* ..... ....* ..... ..| AFFWDFAX |                  |
| *27             |. ....* ..... ..... ..... ..| ADND     |                  |
| *28             |. ....* ..... ....* ..... ..| AFWDB    |                  |
| *29             |. ....* ..... ....* ..... ..| AFWDNA   |                  |
| *31             |. .**** ..**. ..... .*... .*| APIN1    |                  |
| *33             |. .**.* *.*** **... ..... ..| FAX      |                  |
| *34             |. .**** ..**. ..... .*... .*| APIN4    |                  |
| *40             |. .**** ***** **..* ..... ..| FWDREM   |                  |
|                 |                            |          |CFREMVAR CFU      |
|                 |                            |          |CFREMSE  VOICE    |
| *41             |. *...* ..**. ..... ..... .*| BROADCST |                  |
| *42             |. ....* ..**. ..... ..... ..| HOLD     |                  |
| *43             |. ....* ..... ..... ..... ..| MHFALGON |                  |
| *44             |. .**** *.*** **... ..... .*| DISUON   |                  |
-----------------------------------------------------------------------------
| 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       |
-----------------------------------------------------------------------------
| *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   |                  |
| #26                |. ....* ..... ..... ..... ..| DFFWDFAX   |                  |
| #27                |. ....* ..... ..... ..... ..| DDND       |                  |
| #31                |. .**** ..**. ..... .*... .*| DPIN       |                  |
------------------------------------------------------------------------------
| 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       |
------------------------------------------------------------------------------
| #43                |. ....* ..... ..... ..... ..| MHFALGOF |                  |
| ##1                |. *...* ..... ..... ..... ..| KNOVR    |                  |
| ##2                |. *...* ..**. ..... ..... ..| EOVR     |                  |
| ##3                |. *..** ..**. ..... ..... ..| RELEASE  |                  |
------------------------------------------------------------------------------
```

4.3. Mediatrix 4402plus Configuration

The configuration of Mediatrix 4402plus is performed via the device's WBM. Any typical configuration like e.g. call routing to and from OS4000 and for ISDN provider is out of scope and therefore omitted.

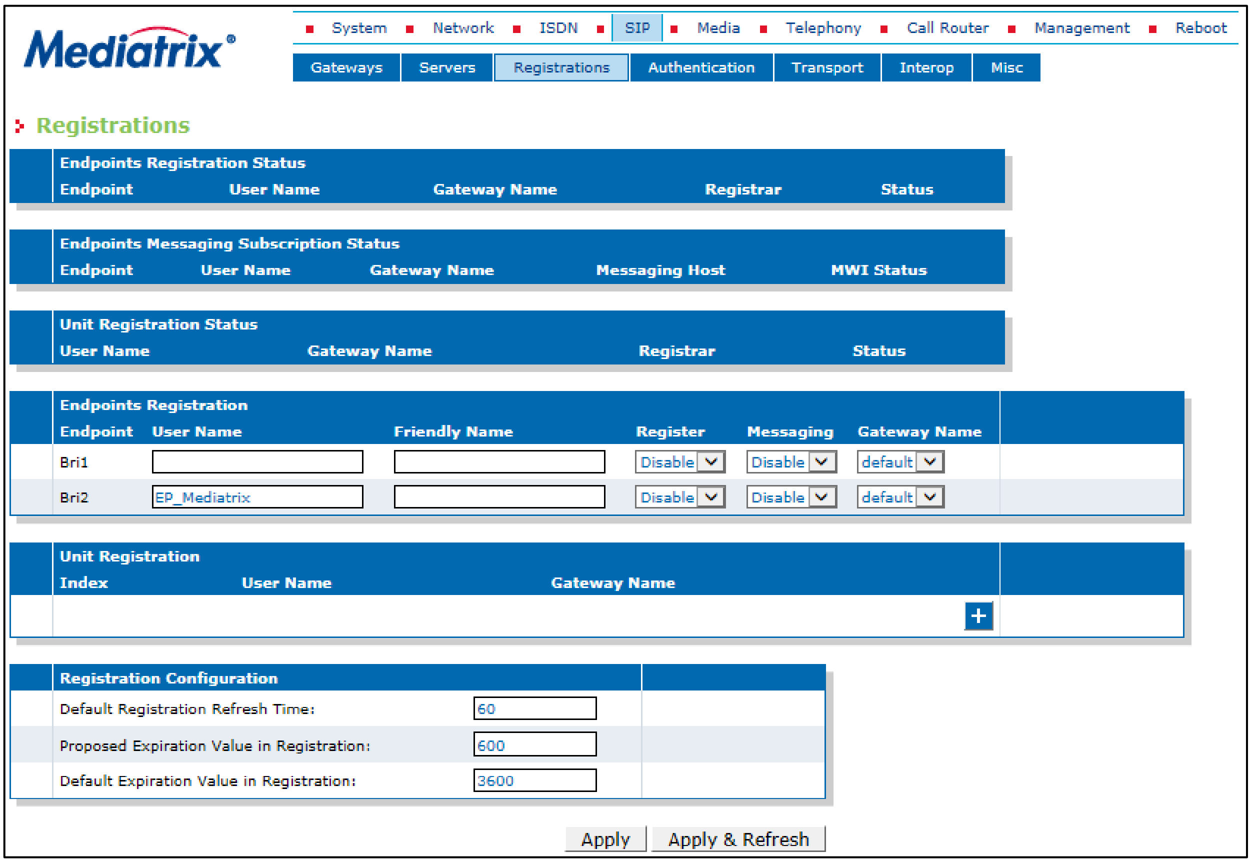


Navigate to **SIP** >> **Servers.**

Configure the value 10.8.242.106:5060 (OS4000 STMI2 board & non-secure port) for **Registrar Host** and **Proxy Host**.

The SIP trunk connection between OS4000 and Mediatrix 4402plus has been configured as TCP for the needs of the current project.

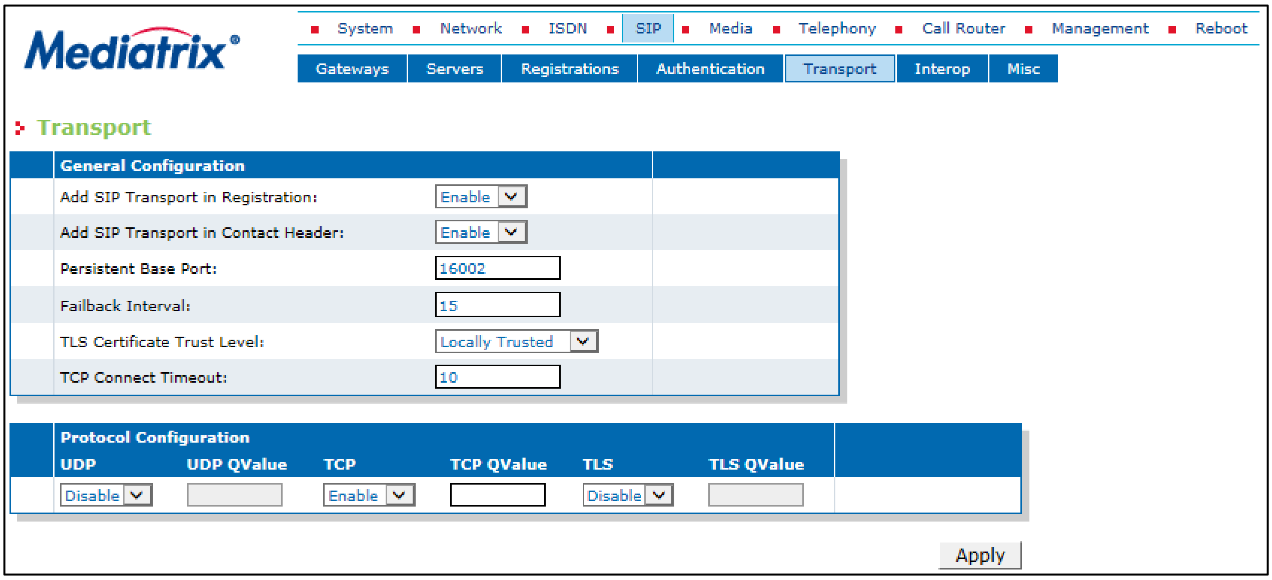
Click on **Apply** button to save the settings.



Go to **SIP** >> **Registrations**

Since Mediatrix 4402plus is connected to ISDN provider via devices Bri2 port, set an endpoint User Name, e.g. **EP_Mediatrix** and for registration to OS4000 set **Register** = Disable (no registration required as displayed in 4.2.2 for the Mediatrix SIP Profile in OS4000 STMI2 board web assistant configuration).

Click on **Apply & Refresh** button.



Navigate to **SIP** >> **Transport** and for TCP connection to OS4000, select **TCP** = Enable from the dropdown box.

Click on **Apply** button.

5. Confirmation

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

Athens, March 21st, 2019	Valid until <date>
Michael Korakis Atos IT & Tel. Services S.A.	(Name, function and signature of authorized person)

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