# 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 Voice V9 in accordance with the test report dated May 17, 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

# Certification Report

## ASCOM IP-DECT V10

## with

## Unify OpenScape Voice V9



**Customer Solution Lab Athens**

Released Version: **1.1 – 2019-05-17**

For Customer: **ASCOM.**



CSL

Customer Solution Lab

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## *History of Change*

| Version | Date | Description | Name |
|---|---|---|---|
| 0.1 | March 18, 2019 | Initial Creation | Charalambos Charalambidis<br>Atos IT & Tel. Services S.A.<br>: charalambos.charalambidis@atos.net<br>☎: +30 (210) 8189 602 |
| 0.2 | March 29, 2019 | Review from Ascom | Matthew Williams<br>Ascom Wireless Solutions<br>: Matthew.Williams@ascom.com<br>☎: +30 (210) 8189 602 |
| 1.0 | April 1, 2019 | Final Document | Charalambos Charalambidis<br>Atos IT & Tel. Services S.A.<br>: charalambos.charalambidis@atos.net<br>☎: +30 (210) 8189 602 |
| 1.1 | May 17, 2019 | • Edited out SARI number from screenshot at 4.2.7.<br>• Edited out PARK number from screenshot at 4.2.10. | Charalambos Charalambidis |

# 1. Overview

## 1.1. Executive Summary

This document describes the test setup, configuration and test results of the feature test of the Ascom IP- DECT system in an OpenScape V9 environment performed in the Customer Solution Lab in Munich.

The Ascom IP-DECT System comprised the following components during the tests:

- Ascom IP-DECT Base Station Master
- Ascom IP-DECT Base Station Standby
- Ascom IP-DECT Handsets

### 1.1.1. Basis Equipment

| | |
|---|---|
| **Test Equipment:** | Virtual OpenScape Voice co-located cluster<br>Virtual OpenScape CMP / Media server<br>Virtual OpenScape Xpressions Voicemail server<br>Mediatrix 4402plus BRI gateway device<br>OpenScape Desk Phones CP400 & OpenScape Desk Phones CP600 phones |
| **Software Release:** | OS Voice version: V9R3.34.E20<br>OS UC version: V9 R3.0.17<br>OS Xpressions version: V7R1 FR5 HF24<br>Mediatrix 4402plus firmware: Dgw 43.2.1343<br>SIP phones firmware: V1 R5.14.0 SIP<br>VMware ESXi v5.5.0 Build 6480324 |

### 1.1.2. Ascom IP-DECT System

| | |
|---|---|
| **Certification:** | ASCOM IP-Dect system registered to OS Voice V9.<br>Test of various call scenarios and features between SIP, PSTN subscribers using ASCOM IP-Dect system. Voice codec verification and fault tolerance scenarios have been tested too. |
| **Test Equipment:** | 1x Ascom IP-DECT Base Station Master<br>1x Ascom IP-DECT Base Station Standby<br>Ascom DECT Handsets d63<br>Ascom DECT Handset d81 |
| **Software Release:** | Ascom IP-DECT Base Station version: IPBS2[10.2.9], Bootcode[10.2.9], Hardware[IPBS2- |

| | |
|---|---|
| | A3/1B]<br>Ascom DECT Handset d63 firmware: 2.3.10<br>Ascom DECT Handset d81 firmware: 4.7.2 |
| **Manufacturer:** | Ascom Holding AG<br>Zugerstrasse 32<br>CH-6340 Baar<br>Switzerland<br>Tel. +41 41 544 78 00<br>https://www.ascom.com/ |
| **Description:** | Ascom DECT handsets are registered on the OpenScape Voice 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 Voice solution. The wireless communication is made using Ascom IP-DECT Base Stations connected to the same LAN as OpenScape Voice.<br>The Ascom IP-DECT Base Stations operate as a conduit between the OpenScape Voice 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 Voice. |
| **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 Voice as SIP stations.<br>OpenScape Voice provides the VoIP telephony facilities to Ascom DECT handset SIP stations.<br>OpenScape Voice is connected (SIP) to a Mediatrix 4402plus BRI gateway, that provides access to PSTN (OTE ITSP).<br>Voicemail services are provided from an OpenScape Xpressions server connected with a SIP trunk to OpenScape Voice.<br>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 SIP 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 OS8K.

- OS8K 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 GmbH und 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 1st, 2019 |
| Test Duration: | March 4th – March 6th, 2019 & March 14th – March 18th, 2019 |
| Test Location: | Atos IT & Tel. Services SA<br>CSL Athens lab<br><br>455, Irakleiou Avenue<br>141 22, N. Irakleio, Athens, Greece |
| Test Personnel: | Ascom A.G., Mathew Williams<br><br>Atos IT & Tel. Services SA, Charalambos Charalambidis |

## 1.6. Test Result Summary

The certification of Ascom IP-DECT system with OpenScape Voice V9R3 PBX is passed with certain restrictions and observations applied. 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".
- Conflicts between IP Dect and OSV prefixes may cause problems in OSV Feature.
- **Call Forwarding** scenarios were tested with both local Ascom DECT handset and OpenScape Voice 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.
- Unify license is required for Group Pickup and Large Conference feature.

### 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).
- **Call Park** feature isn't supported on Ascom DECT handsets.
- 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".
- **Call Deflect** isn't supported on Ascom DECT handsets.
- Ascom IP-DECT system doesn't support local **3-party Conference**.
- Regarding audio **Codecs** support, OpenScape Voice 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.

## 1.7. Network Diagram

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

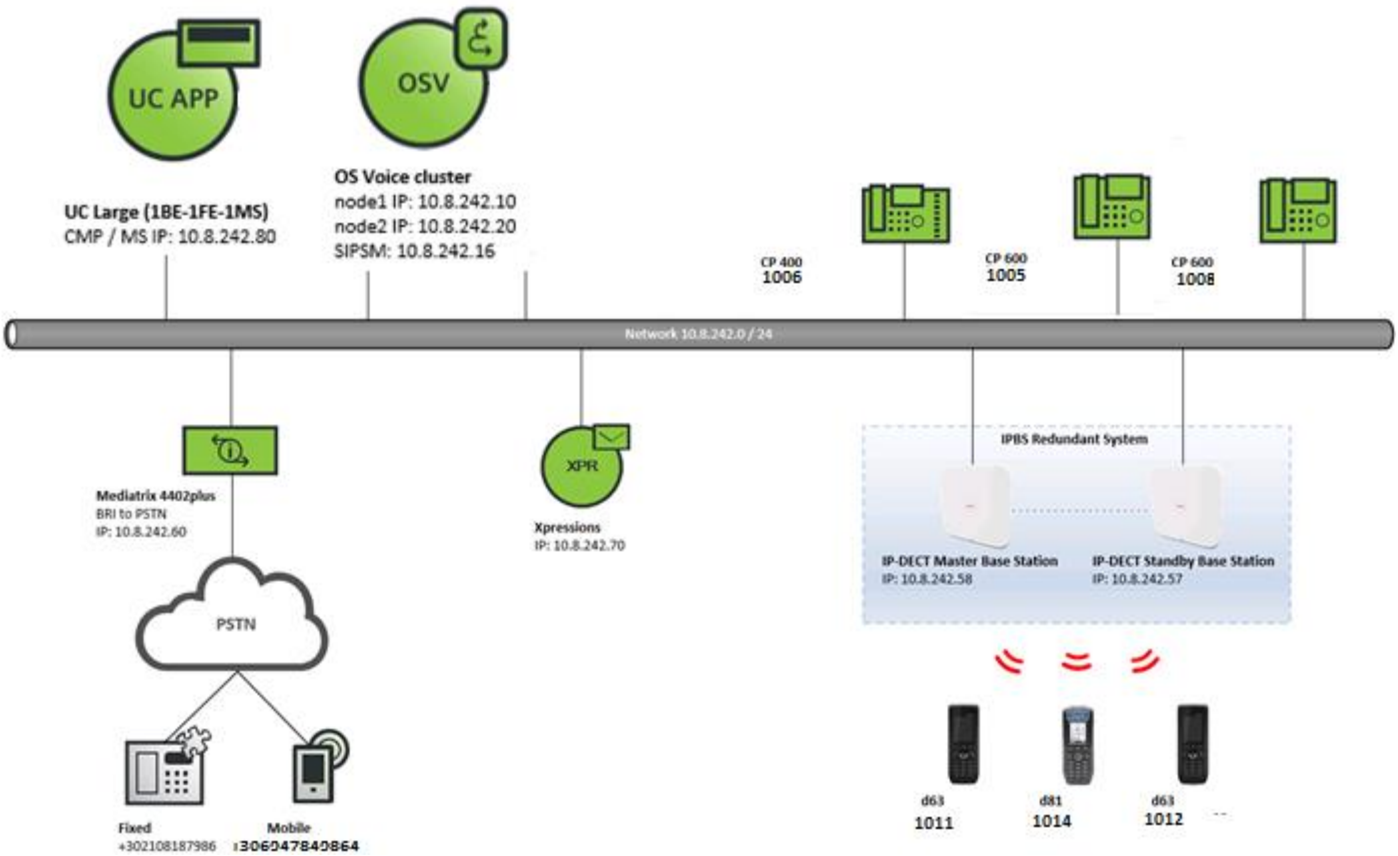


**Figure 1: Logical Topology Diagram**

# 2. Configuration

## 2.1. ASCOM IP Phone

The Ascom IP-DECT Base Stations and Ascom DECT handsets configuration was performed by Ascom personnel for the certification test according to OpenScape Voice 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 OpenScape Voice SIP extensions are utilized:

- Ascom DECT handset SIP stations

  *ASC_A = 1011 (d63 handset)*
  *ASC_B = 1012 (d63 handset)*
  *ASC_C = 1014 (d81 handset)*
  *ASC_D = 1015 (d81 handset)*

## 2.2. Unify Component Infrastructure

OS Voice is configured for extension dialing between the subscribers (SIP and analog). The Office code used is 30 (210) 100 and the subscriber extensions are with 4 digits (10xx). Full DN dialing is also possible (3021010010xx) between subscribers.

OS Voice provides access to PSTN (OTE ISDN BRI provider) via a Mediatrix 4402plus BRI gateway (10.8.242.78). OS Voice subscribers (SIP or analog) 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 OS Voice before sending the number to PSTN.

Voicemail functionality is provided by OS Voice through OpenScape Xpressions server (10.8.242.70). OS Voice subscribers may call 302102003001 (direct access number) to access their mailboxes.

- OS Voice IP addresses

  *10.8.242.10 node 1*
  *10.8.242.20 node 2*
  *10.8.242.16 SIPSM 1*
  *10.8.242.26 SIPSM 2*

- OS UC (Media Server, CMP-Management)

  *10.8.242.80*

- SIP phones

  *1005 OpenScape Desk Phone CP400*
  *1006 OpenScape Desk Phone CP600*
  *1008 OpenScape Desk Phone CP600*

- Voicemail

  302102003000 OS Xpressions system (callback) number

- ISDN BRI number for incoming calls from PSTN

  *302106256856*

- PSTN

  *302108189602 (landline)*
  *306947849864 (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*.

In general, "*Registration"* covers registration scenarios 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 OS8K. Additionally, registration expiration timer was tested, too.

The "*Basic Calls"* area covers 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 scenarios are, also, checked.

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

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

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

The call forwarding scenarios were executed twice; activated via phone or via OS Voice feature.

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.

The *"Failover"* tests cover the redundancy functionality. Tested in Collocated Cluster. Also scenarios with Base Station failure and the behavior after the failed components are back in service was tested as well.

### 3.1.1. Registration

**OSV Registration**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 1-1 | • Register Ascom IP-DECT phone to OpenScape Voice without Digest Authentication. | • DECT phone successfully registers at OpenScape Voice. | OK | |
| 1-2 | • Register Ascom IP-DECT phone to OpenScape Voice with Digest Authentication. | • DECT phone successfully registers at OpenScape Voice. | OK | Registration Takes 30'' |
| 1-3 | • Check that Ascom IP-DECT phone may register to OpenScape Voice with either TCP or UDP. | • DECT phone successfully registers at OpenScape Voice. | OK | |
| 1-4 | • Registration expiration timer at Ascom IP-DECT phone is set below OpenScape Voice's corresponding value. | • OpenScape Voice responds with a 423 Interval Too Brief to negotiate the lower re-Registration interval. | OK | Recommended value: 300 |

### 3.1.2. Basic Call

**Incoming, Outgoing Call**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 2-1 | • ASC_A calls ASC_B.<br>• ASC_B hangs up at the end of the communication. | • Post Dial Delay (PDD).<br>• RingBackTone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• No media clipping when connecting both ends.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | Ascom handsets display only the name in calls (ringing and connected state) and not name + number.<br>Ascom: This is per design (name has preference). The number, however, will be recorded in the call list. |
| 2-2 | • OSV_A calls ASC_A.<br>• OSV_A hangs up at the end of the communication. | • Post Dial Delay (PDD).<br>• RingBackTone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• No media clipping when connecting both ends.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | |
| 2-3 | • OSV_A calls ASC_A.<br>• ASC_A hangs up at the end of the | • Post Dial Delay (PDD).<br>• RingBackTone heard by calling party.<br>• Number presentation on ringing and | OK | |

| | | | | |
|---|---|---|---|---|
| | communication. | after call pick up.<br>• No media clipping when connecting both ends.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | | |
| 2-4 | • ASC_A calls OSV_A.<br>• ASC_A hangs up at the end of the communication. | • Post Dial Delay (PDD).<br>• RingBackTone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• No media clipping when connecting both ends.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | |
| 2-5 | • ASC_A calls OSV_A.<br>• OSV_A hangs up at the end of the communication. | • Post Dial Delay (PDD)<br>• RingBackTone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• No media clipping when connecting both ends.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | |
| 2-6 | • PSTN calls ASC_A.<br>• PSTN hangs up at the end of the communication. | • Post Dial Delay (PDD)<br>• RingBackTone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• No media clipping when connecting both ends.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | ASCOM_A rings as internal call<br>In Invite message with alert info header equals "Bellcore -dlr1" |
| 2-7 | • PSTN calls ASC_A.<br>• ASC_A hangs up at the end of the communication. | • Post Dial Delay (PDD)<br>• RingBackTone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• No media clipping when connecting both ends.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | ASCOM_A rings as internal call<br>In Invite message with alert info header equals "Bellcore -dlr1" |
| 2-8 | • ASC_A calls PSTN.<br>• ASC_A hangs up at the end of the communication. | • Post Dial Delay (PDD)<br>• RingBackTone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• No media clipping when connecting both ends.<br>• Speech path in both directions. | OK | |

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|---|---|---|---|---|
| | | • Both ends idle after call clearing. | | |
| 2-9 | • ASC_A calls PSTN.<br>• PSTN hangs up at the end of the communication. | • Post Dial Delay (PDD)<br>• RingBackTone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• No media clipping when connecting both ends.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | |

**Busy Call**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 2-10 | • ASC_A calls ASC_B.<br>• ASC_B is busy. | • Busy tone/display on calling party.<br>• The call is properly cleared. | OK | 30'' the call to be cleared |
| 2-11 | • OSV_A calls ASC_A.<br>• ASC_A is busy. | • Busy tone/display on calling party.<br>• The call is properly cleared. | OK | |
| 2-12 | • ASC_A calls OSV_A.<br>• OSV_A is busy. | • Busy tone/display on calling party.<br>• The call is properly cleared. | OK | 30'' the call to be cleared |
| 2-13 | • PSTN calls ASC_A.<br>• ASC_A is busy. | • Busy tone/display on calling party.<br>• The call is properly cleared. | OK | |
| 2-14 | • ASC_A calls PSTN.<br>• PSTN is busy. | • Busy tone/display on calling party.<br>• The call is properly cleared. | OK | |
| 2-10 | • ASC_A calls ASC_B.<br>• ASC_B is busy. | • Busy tone/display on calling party.<br>• The call is properly cleared. | OK | 30'' the call to be cleared |
| 2-11 | • OSV_A calls ASC_A.<br>• ASC_A is busy. | • Busy tone/display on calling party.<br>• The call is properly cleared. | OK | |
| 2-12 | • ASC_A calls OSV_A.<br>• OSV_A is busy. | • Busy tone/display on calling party.<br>• The call is properly cleared. | OK | 30'' the call to be cleared |

**Rejected Call**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 2-15 | • ASC_A calls ASC_B.<br>• ASC_B rejects incoming call. | • Busy tone/display on calling party.<br>• The call is properly cleared. | OK | Busy Tone - 30'' the call to be cleared<br>We can modify from ASCOM to send decline instead of Busy) |
| 2-16 | • OSV_A calls ASC_A.<br>• ASC_A rejects incoming call. | • Busy tone/display on calling party.<br>• The call is properly cleared. | OK | |
| 2-17 | • ASC_A calls OSV_A.<br>• OSV_A rejects incoming call. | • Busy tone/display on calling party.<br>• The call is properly cleared. | POK | Hung Up on Display.<br>(Works as Designed) |
| 2-18 | • PSTN calls ASC_A.<br>• ASC_A rejects incoming call. | • Busy tone/display on calling party.<br>• The call is properly cleared. | OK | |
| 2-19 | • ASC_A calls PSTN.<br>• PSTN rejects incoming call. | • Busy tone/display on calling party.<br>• The call is properly cleared. | OK | |

**Not Answered Call**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 2-20 | • ASC_A calls ASC_B.<br>• ASC_B doesn't answer. Let the call ring until disconnection. | • Calling party gets announcement or specific tone.<br>• Check timer after which the call gets disconnected. | OK | It takes 3' for the call to be cleared. |
| 2-21 | • OSV_A calls ASC_A.<br>• OSV_A doesn't answer. Let the call ring until disconnection. | • Calling party gets announcement or specific tone.<br>• Check timer after which the call gets disconnected. | OK | It takes 3' for the call to be cleared. |
| 2-22 | • ASC_A calls OSV_A.<br>• OSV_A doesn't answer. Let the call ring until disconnection. | • Calling party gets announcement or specific tone.<br>• Check timer after which the call gets disconnected. | OK | It takes 3' for the call to be cleared. |
| 2-23 | • PSTN calls ASC_A.<br>• ASC_A doesn't answer. Let the call ring until disconnection. | • Calling party gets announcement or specific tone.<br>• Check timer after which the call gets disconnected. | OK | It takes 3' for the call to be cleared. |
| 2-24 | • ASC_A calls PSTN.<br>• PSTN doesn't answer. Let the call ring until disconnection. | • Calling party gets announcement or specific tone.<br>• Check timer after which the call gets disconnected. | OK | 1' and 30" - Not reachable was displayed in ASCOM device. |

**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 | Get a missed call notification on ASCOM Device |
| 2-26 | • OSV_A calls ASC_A and hangs up before destination picks up. | • Call is terminated on both sides after hanging up. | OK | Get a missed call notification on ASCOM Device |
| 2-27 | • ASC_A calls OSV_A and hangs up before destination picks up. | • Call is terminated on both sides after hanging up. | OK | Get a missed call notification on CP Device |
| 2-28 | • PSTN calls ASC_A and hangs up before destination picks up. | • Call is terminated on both sides after hanging up. | OK | Get a missed call notification on ASCOM Device |
| 2-29 | • ASC_A calls PSTN and hangs up before destination picks up. | • Call is terminated on both sides after hanging up. | OK | Get a missed call notification on Device |

**Call To Unavailable**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 2-30 | • ASC_B is unregistered.<br>• ASC_A calls ASC_B. | • Busy tone/display on calling party.<br>• The call is properly cleared. | OK | "Hung Up" Tone |
| 2-31 | • ASC_A is unregistered.<br>• OSV_A calls ASC_A. | • Busy tone/display on calling party.<br>• The call is properly cleared. | OK | |
| 2-32 | • OSV_A is unregistered.<br>• ASC_A calls OSV_A. | • Busy tone/display on calling party.<br>• The call is properly cleared. | OK | The call is properly cleared after 30 seconds. |
| 2-33 | • ASC_B is out of coverage (w/ batt. removed).<br>• ASC_A calls ASC_B. | • Busy tone/display on calling party.<br>• The call is properly cleared. | OK | The call is properly cleared after 15 seconds. |
| 2-34 | • ASC_B is out of coverage (w/ batt. removed).<br>• OSV_A calls ASC_B. | • Busy tone/display on calling party.<br>• The call is properly cleared. | OK | |

### 3.1.3. Basic Call 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). | OK | Anonymous calls are not displayed in ascom handset call list. |
| 3-2 | • ASC_A initiates an outgoing call to ASC_B from call history. | • Post Dial Delay (PDD)<br>• RingBackTone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• No media clipping when connecting both ends.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | |
| 3-3 | • ASC_A initiates an outgoing call to OSV_A from call history. | • Post Dial Delay (PDD)<br>• RingBackTone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• No media clipping when connecting both ends.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | |
| 3-4 | • ASC_A initiates an outgoing call to PSTN from call history. | • Post Dial Delay (PDD)<br>• RingBackTone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• No media clipping when connecting both ends.<br>• Speech path in both directions.<br>• 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).<br>• The call must last at least 1 hour.<br>• ASC_B puts the call on mute when call is established. | • Check that call lasts for at least 1h.<br>• When MUTE function is activated during at least 5 minutes, check that calls are still active when MUTE function is deactivated on endpoints. | OK | Max Time: 12h (Configurable from OSV) |

| | | | | |
|---|---|---|---|---|
| | • After 5 mins ASC_B unmutes call and then ASC_A mutes for another 5 mins. | • Check SIP Session Timer Expiry. | | |
| 3-6 | • Setup a call between ASC_A and OSV_A.<br>• The call must last at least 1 hour.<br>• OSV_A puts the call on mute when call is established.<br>• After 5 mins OSV_A unmutes call and then ASC_A mutes for another 5 mins. | • Check that call lasts for at least 1h.<br>• When MUTE function is activated during at least 5 minutes, check that calls are still active when MUTE function is deactivated on endpoints.<br>• Check SIP Session Timer Expiry. | OK | Max Time: 12h (Configurable from OSV) |

**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. *9).<br>• ASC_A calls ASC_B. | • On softphone display a DND logo is displayed.<br>• Calling party gets busy tone. | OK | Activated via PBX there is no display in ASCOM device that DND is activated<br>Activate DND via ASCOM locally there is a Busy notification in the display and the user gets "Busy message". |
| 3-8 | • ASC_B has DND activated via menu options or via DND feature code (e.g. *9).<br>• OSV_A calls ASC_B. | • On softphone display a DND logo is displayed.<br>• Calling party gets busy tone. | OK | Activated via PBX there is no display in ASCOM device that DND is activated<br>Activate DND via ASCOM locally there is a Busy notification in the display and the user gets "Busy message". |
| 3-9 | • After scenario 3-8, ASC_A deactivates DND.<br>• OSV_A calls ASC_A again. | • DND logo disappears from display.<br>• Incoming calls are normally signalled on the phone. | OK | |

**Clir**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 3-9 | • ASC_A activates Calling Identity Suppression via the CID Suppression feature access code (e.g. *51).<br>• ASC_A calls ASC_B. | • Number and name are not displayed on called device. | OK | xxxxxxxxxxxx was displayed in ASCOM device. |
| 3-10 | • ASC_A activates Calling Identity Suppression via the CID Suppression feature access code (e.g. *51).<br>• ASC_A calls OSV_A. | • Number and name are not displayed on called device. | OK | |
| 3-11 | • ASC_A activates Calling Identity Suppression via the CID Suppression feature access code (e.g. *51).<br>• ASC_A calls CELL. | • Number is not displayed on called phone. | NA | Not Supported |
| 3-12 | • OSV_A activates Calling Identity Suppression via the CID Suppression feature access code (e.g. *51).<br>• OSV_A calls SC_A. | • Number and name are not displayed on called phone.<br>• Only the dialed digits are displayed on the dialing phone | OK | ASC_A displays *********** External call. No call is shown in call list. |
| 3-13 | • CELL activates identity restriction.<br>• CELL calls ASC_A. | • Number is not displayed on called phone. | OK | ASC_A displays *********** External call. No call is shown in call list |

### 3.1.4. Telephony Features

**Hold, Consultation, Toggle Key, Call Waiting**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 4-1 | • ASC_A calls ASC_B and puts the destination on hold. | • Music on Hold is played to held party.<br>• Held call display on both parties.<br>• Proper call resume.<br>• No Media delay establishment after resuming the call (media clipping). | OK | "On Hold" message was shown on ASCOM |
| 4-2 | • OSV_A calls ASC_A and puts the destination on hold. | • Music on Hold is played to held party.<br>• Held call display on both parties.<br>• Proper call resume.<br>• No Media delay establishment after resuming the call (media clipping). | OK | "On Hold" message was shown on ASCOM |
| 4-3 | • ASC_A calls OSV_A and puts the destination on hold. | • Music on Hold is played to held party.<br>• Held call display on both parties.<br>• Proper call resume.<br>• No Media delay establishment after resuming the call (media clipping). | OK | Name updates correct and "On Hold" message was shown on ASCOM. |
| 4-4 | • ASC_A calls ASC_B.<br>• ASC_B makes a consultation call to ASC_C.<br>• ASC_B toggles between calls with ASC_A and ASC_C. | • Ring back tone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• No media clipping when connecting both ends.<br>• MOH is played to held party.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | On ascom handset, press R2 to toggle between calls and R1 to hang up active call and switch to held call.<br>Name updates correct and "On Hold" message was shown on ASCOM. |
| 4-5 | • OSV_A calls ASC_A.<br>• OSV_A makes a consultation call to ASC_B.<br>• OSV_A toggles between calls with ASC_A and ASC_B. | • Ring back tone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• No media clipping when connecting both ends.<br>• MOH is played to held party.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | Name updates correct and "On Hold" message was shown on ASCOM. |
| 4-6 | • ASC_A calls OSV_A.<br>• ASC_A makes a consultation call to | • Ring back tone heard by calling party. | OK | Name updates correct and "On Hold" message |

| | | | | |
|---|---|---|---|---|
| | OSV_B.<br>• ASC_A toggles between calls with OSV_A and OSV_B. | • Number presentation on ringing and after call pick up.<br>• No media clipping when connecting both ends.<br>• MOH is played to held party.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | | was shown on ASCOM. |
| 4-7 | • PSTN calls ASC_A.<br>• ASC_A makes a consultation call to ASC_B.<br>• ASC_A toggles between calls with PSTN and ASC_A. | • Ring back tone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• No media clipping when connecting both ends.<br>• MOH is played to held party.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | |
| 4-8 | • OSV_A calls ASC_A.<br>• OSV_B calls the same, ASC_A.<br>• Call waiting is being signalled on ASC_A.<br>• ASC_A accepts the second incoming call and toggles between calls with OSV_A and OSV_B. | • Ring back tone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• No media clipping when connecting both ends.<br>• MOH is played to held party.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | |

**Callback (No Reply)**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 4-9 | • ASC_A has the feature CCNR activated at OpenScape Voice.<br>• ASC_A calls ASC_B, but ASC_B doesn't answer.<br>• ASC_A activates CCNR via phone menu or by dialing the feature access code (e.g., *6).<br>• ASC_B establishes a call with another subscriber and then hangs up.<br>• When ASC_B becomes available, OpenScape Voice calls ASC_A and once ASC_A answers, OpenScape Voice calls ASC_B.<br>• ASC_B answers callback call and connects with ASC_A. | • CCNR feature is successfully activated.<br>• No media clipping when connecting both ends.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | |

| | | | | |
|---|---|---|---|---|
| 4-10 | • OSV_A has the feature CCNR activated at OpenScape Voice.<br>• OSV_A calls ASC_A, but ASC_A doesn't answer.<br>• OSV_A activates CCNR via phone menu or by dialing the feature access code (e.g., *6).<br>• ASC_A establishes a call with another subscriber and then hangs up.<br>• When ASC_A becomes available, OpenScape Voice calls OSV_A and once OSV_A answers, OpenScape Voice calls ASC_A.<br>• ASC_A answers callback call and connects with OSV_A. | • CCNR feature is successfully activated.<br>• No media clipping when connecting both ends.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | |
| 4-11 | • ASC_A has the feature CCNR activated at OpenScape Voice.<br>• ASC_A calls OSV_A, but OSV_A doesn't answer.<br>• ASC_A activates CCNR via phone menu or by dialing the feature access code (e.g., *6).<br>• OSV_A establishes a call with another subscriber and then hangs up.<br>• When OSV_A becomes available, after a while, OpenScape Voice calls ASC_A and once ASC_A answers, OpenScape Voice calls OSV_A.<br>• OSV_A answers callback call and connects with ASC_A. | • CCNR feature is successfully activated.<br>• No media clipping when connecting both ends.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | |
| 4-12 | • Repeat scenarios 4-10 & 4-11, but deactivate the callback feature via phone menu or by dialing CCNR deactivation code (e.g. #6). | • OSV provides acknowledgement that the callback feature has been cancelled.<br>• After the CCNR request has been deactivated, OSV should not send any notification that the original callback target has become available to receive calls. | OK | |

**Callback (On Busy)**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 4-13 | • ASC_A has the feature CCBS activated at OpenScape Voice.<br>• ASC_A calls ASC_B, but ASC_B is busy.<br>• ASC_A activates CCBS via phone menu or by dialing access code (e.g., *6).<br>• When ASC_B becomes available, after a while, OpenScape Voice calls ASC_C and once ASC_C answers, OpenScape Voice calls ASC_A.<br>• ASC_B answers callback call and connects with ASC_A. | • CCBS feature is successfully activated.<br>• No media clipping when connecting both ends.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | |
| 4-14 | • OSV_A has the feature CCBS activated at OpenScape Voice.<br>• OSV_A calls ASC_A, but ASC_A is busy.<br>• OSV_A activates CCBS via phone menu or by dialing access code (e.g., *6).<br>• When ASC_A becomes available, after a while, OpenScape Voice calls OSV_A and once OSV_A answers, OSV calls ASC_A.<br>• ASC_A answers callback call and connects with OSV_A. | • CCBS feature is successfully activated.<br>• No media clipping when connecting both ends.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | |
| 4-15 | • ASC_A has the feature CCBS activated at OpenScape Voice.<br>• ASC_A calls OSV_A, but OSV_A is busy.<br>• ASC_A activates CCBS via phone menu or by dialing the feature access code (e.g., *6).<br>• When OSV_A becomes available, after a while, OpenScape Voice calls ASC_A and once ASC_A answers, OpenScape Voice calls OSV_A.<br>• OSV_A answers callback call and connects with ASC_A. | • CCBS feature is successfully activated.<br>• No media clipping when connecting both ends.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | |
| 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 | • CCBS feature is successfully activated.<br>• No media clipping when connecting both ends. | OK | |

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|---|---|---|---|---|
| | deactivation code (e.g. #6). | • Speech path in both directions.<br>• Both ends idle after call clearing. | | |

**Call Park**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 4-17 | • ASC_A calls ASC_B.<br>• ASC_B parks the call.<br>• ASC_C dials the parking lot number to retrieve the call.<br>• ASC_A hangs up. | • Music on hold played on ASC_A while the call is parked.<br>• ASC_C successfully retrieves parked call.<br>• Number presentation on ringing and after call pick up.<br>• Media clipping and speech path after forwarding.<br>• Call is disconnected after hang up. | NOK | ASCOM cannot park a call. The call returns. Limitation: cannot transfer in early media state.<br>Jira Ticket was opened against OSV (OSV-11723) |
| 4-18 | • ASC_A calls OSV_A.<br>• OSV_A parks the call.<br>• ASC_B retrieves the call from parking lot.<br>• OSV_A hangs up. | • Music on hold played on OSV_A while the call is parked.<br>• ASC_C successfully retrieves parked call.<br>• Number presentation on ringing and after call pick up.<br>• Media clipping and speech path after forwarding.<br>• Call is disconnected after hang up. | OK | |
| 4-19 | • ASC_A calls OSV_A.<br>• OSV_A parks the call.<br>• ASC_A does not retrieve the call for more tha180 seconds.<br>• ASC_A gets an incoming call to retrieve the parked call and SC_A answers.<br>• OSV_A hangs up. | • Music on hold played on OSV_A while the call is parked.<br>• ASC_A successfully retrieves parked call.<br>• Number presentation on ringing and after call pick up.<br>• Media clipping and speech path after forwarding.<br>• Call is disconnected after hang up. | OK | |
| 4-20 | • OSV_A calls ASC_A.<br>• OSV_A parks the call.<br>• OSV_B dials the parking lot number to retrieve the call.<br>• ASC_A hangs up. | • Music on hold played on ASC_A while the call is parked.<br>• OSV_B successfully retrieves parked call.<br>• Number presentation on ringing and after call pick up.<br>• Media clipping and speech path after forwarding.<br>• Call is disconnected after hang up. | OK | |

**Call Transfer (Attended)**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 4-21 | • ASC_A calls ASC_B.<br>• ASC_B consults to ASC_C and waits for ASC_C to pickup the call.<br>• ASC_B transfers ASC_A towards ASC_C.<br>• After transfer is completed and communication is established between ASC_A and ASC_C, ASC_A hangs up. | • MOH played to ASC_B while reaching ASC_C.<br>• No Media clipping and proper speech path after transfer when ASC_B and ASC_C are connected.<br>• Number presentation on ringing and after call pick up.<br>• Both ends idle after call clearing. | OK | Name updates correct. |
| 4-22 | • OSV_A calls ASC_A.<br>• ASC_A consults to ASC_B and waits for ASC_B to pickup the call.<br>• ASC_A transfers OSV_A towards ASC_B.<br>• After transfer is completed and communication is established between OSV_A and ASC_B, OSV_A hangs up. | • MOH played to OSV_A while reaching ASC_B.<br>• No Media clipping and proper speech path after transfer when OSV_A and ASC_B are connected.<br>• Number presentation on ringing and after call pick up.<br>• Both ends idle after call clearing. | OK | Name updates correct. |
| 4-23 | • ASC_A calls OSV_A.<br>• ASC_A consults to OSV_B and waits for OSV_B to pickup the call.<br>• ASC_A transfers OSV_A towards OSV_B.<br>• After transfer is completed and communication is established between OSV_A and OSV_B, OSV_A hangs up. | • MOH played to OSV_A while reaching OSV_B.<br>• No Media clipping and proper speech path after transfer when OSV_A and OSV_B are connected.<br>• Number presentation on ringing and after call pick up.<br>• Both ends idle after call clearing. | OK | Name updates correct. |
| 4-24 | • ASC_A calls OSV_A .<br>• OSV_A puts ASC_A on hold, calls ASC_B and waits for ASC_B to pickup the call.<br>• ASC_A resumes the call with OSV_A (ASC_B is placed on hold) and then transfers OSV_A towards ASC_B.<br>• After transfer is completed and communication is established between OSV_A and ASC_B, OSV_A hangs up. | • MOH is played in all related steps (to OSV_A while reaching ASC_B / to ASC_B while resuming the call with OSV_A).<br>• Proper call resume when ASC_A toggles between calls with OSV_A and ASC_B.<br>• No Media clipping and proper speech path after transfer when OSV_A and ASC_B are connected.<br>• Number presentation on ringing and after call pick up.<br>• Both ends idle after call clearing. | OK | |
| 4-25 | • PSTN calls ASC_A.<br>• ASC_A consults to ASC_B and waits | • MOH played to PSTN while reaching ASC_B. | OK | |

| | | | | |
|---|---|---|---|---|
| | for ASC_B to pickup the call.<br>• ASC_A transfers PSTN towards ASC_B.<br>• After transfer is completed and communication is established between PSTN and ASC_B, PSTN hangs up. | • No Media clipping and proper speech path after transfer when ASC_B picks up the call.<br>• Number presentation on ringing and after call pick up.<br>• Both ends idle after call clearing. | | |
| 4-26 | • PSTN calls ASC_A.<br>• ASC_A consults to OSV_B and waits for OSV_B to pickup the call.<br>• ASC_A transfers PSTN towards OSV_B.<br>• After transfer is completed and communication is established between PSTN and OSV_B, PSTN hangs up. | • MOH is played in all related steps (to PSTN while reaching OSV_B / to OSV_B while resuming the call with PSTN).<br>• Proper call resume when ASC_A toggles between calls with PSTN and OSV_B.<br>• No Media clipping and proper speech path after transfer when PSTN and OSV_B are connected.<br>• Number presentation on ringing and after call pick up.<br>• Both ends idle after call clearing. | OK | |

**Call Transfer (Semi-Attended)**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 4-27 | • ASC_A calls ASC_B.<br>• ASC_B consults to ASC_C<br>• ASC_A hangs up while ASC_C is ringing.<br>• ASC_B receives ring back tone.<br>• After communication is established between ASC_B and ASC_C, ASC_B hangs up. | • MOH played to ASC_B after consultation and before ASC_A hangs up.<br>• No Media clipping and proper speech path after transfer when ASC_B and ASC_C are connected.<br>• Number presentation on ringing and after call pick up.<br>• Both ends idle after call clearing. | OK | ASC_C is updated on the display with ASC_A after accepting the call. |
| 4-28 | • OSV_A calls ASC_A.<br>• ASC_A consults to ASC_B<br>• ASC_A hangs up while ASC_B is ringing.<br>• OSV_A receives ring back tone.<br>• After communication is established between OSV_A and ASC_B, OSV_A hangs up. | • MOH played to OSV_A after consultation and before ASC_A hangs up.<br>• No Media clipping and proper speech path after transfer when OSV_A and ASC_B are connected.<br>• Number presentation on ringing and after call pick up.<br>• Both ends idle after call clearing. | OK | ASC_C is updated on the display with ASC_A after accepting the call. |
| 4-29 | • OSV_A calls ASC_A.<br>• ASC_A consults to OSV_B<br>• ASC_A hangs up while OSV_B is ringing. | • MOH played to OSV_A after consultation and before ASC_A hangs up.<br>• No Media clipping and proper | OK | ASC_C is updated on the display with ASC_A after accepting the call. |

| | | | | |
|---|---|---|---|---|
| | • OSV_A receives ring back tone.<br>• After communication is established between OSV_A and OSV_B, OSV_A hangs up. | speech path after transfer when OSV_A and OSV_B are connected.<br>• Number presentation on ringing and after call pick up.<br>• Both ends idle after call clearing. | | |
| 4-30 | • OSV_A calls ASC_A.<br>• ASC_A consults to ASC_B<br>• ASC_A hangs up while ASC_B is ringing.<br>• OSV_A receives ring back tone.<br>• After communication is established between OSV_A and ASC_B, OSV_A hangs up. | • MOH played to OSV_A after consultation and before ASC_A hangs up.<br>• No Media clipping and proper speech path after transfer when OSV_A and ASC_B are connected.<br>• Number presentation on ringing and after call pick up.<br>• Both ends idle after call clearing. | OK | ASC_C is updated on the display with ASC_A after accepting the call. |
| 4-31 | • PSTN calls ASC_A.<br>• ASC_A consults to ASC_B.<br>• ASC_A hangs up while ASC_B is ringing.<br>• PSTN receives ring back tone.<br>• After communication is established between PSTN and ASC_B, PSTN hangs up. | • MOH played to PSTN after consultation and before ASC_A hangs up.<br>• No Media clipping and proper speech path after transfer when PSTN and ASC_B are connected.<br>• Number presentation on ringing and after call pick up.<br>• Both ends idle after call clearing. | OK | ASC_C is updated on the display with ASC_A after accepting the call. |
| 4-32 | • PSTN calls ASC_A.<br>• ASC_A consults to OSV_B.<br>• ASC_A hangs up while OSV_B is ringing.<br>• PSTN receives ring back tone.<br>• After communication is established between PSTN and OSV_B, OSV_A hangs up. | • MOH played to PSTN after consultation and before ASC_A hangs up.<br>• No Media clipping and proper speech path after transfer when PSTN and OSV_B are connected.<br>• Number presentation on ringing and after call pick up.<br>• Both ends idle after call clearing. | OK | ASC_C is updated on the display with ASC_A after accepting the call. |

**Call Transfer (Blind)**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 4-33 | • ASC_A calls ASC_B.<br>• ASC_A Transfers (Blind) to ASCOM_C.<br>• ASC_B receives ring back tone.<br>• After transfer is completed and communication is established between ASC_B and ASC_C, ASC_B hangs up. | • MOH played to ASC_B after consultation and before blind transfer is completed.<br>• No Media clipping and proper speech path after transfer when ASC_B and ASC_C are connected.<br>• Number presentation on ringing and after call pick up.<br>• Both ends idle after call clearing. | OK | Name updates correct. |

| | | | | |
|---|---|---|---|---|
| 4-34 | • OSV_A calls ASC_A.<br>• ASC_A Transfers (Blind) to ASCOM_C.<br>• OSV_A receives ring back tone.<br>• After transfer is completed and communication is established between OSV_A and ASC_B, OSV_A hangs up. | • MOH played to OSV_A after consultation and before blind transfer is completed.<br>• No Media clipping and proper speech path after transfer when OSV_A and ASC_B are connected.<br>• Number presentation on ringing and after call pick up.<br>• Both ends idle after call clearing. | OK | Name updates correct. |
| 4-35 | • OSV_A calls ASC_A.<br>• ASC_A Transfers (Blind) to OSV_B.<br>• OSV_A receives ring back tone.<br>• After transfer is completed and communication is established between OSV_A and OSV_B, OSV_A hangs up. | • MOH played to OSV_A after consultation and before blind transfer is completed.<br>• No Media clipping and proper speech path after transfer when OSV_A and OSV_B are connected.<br>• Number presentation on ringing and after call pick up.<br>• Both ends idle after call clearing. | OK | Name updates correct. |
| 4-36 | • ASC_A calls OSV_A.<br>• OSV_A Transfers (Blind) to ASC_B.<br>• ASC_B receives ring back tone.<br>• After transfer is completed and communication is established between ASC_A and ASC_B, ASC_A hangs up. | • MOH played to OSV_A after consultation and before blind transfer is completed.<br>• No Media clipping and proper speech path after transfer when OSV_A and ASC_B are connected.<br>• Number presentation on ringing and after call pick up.<br>• Both ends idle after call clearing. | OK | Name updates correct. |
| 4-37 | • PSTN calls ASC_A.<br>• ASC_A consults to ASC_B.<br>• ASC_A selects blind transfer in menu.<br>• PSTN receives ring back tone.<br>• After transfer is completed and communication is established between PSTN and ASC_B, PSTN hangs up. | • MOH played to PSTN after consultation and before blind transfer is completed.<br>• No Media clipping and proper speech path after transfer when PSTN and ASC_B are connected.<br>• Number presentation on ringing and after call pick up.<br>• Both ends idle after call clearing. | OK | Name updates correct. |
| 4-38 | • PSTN calls ASC_A.<br>• ASC_A consults to OSV_B.<br>• ASC_A selects blind transfer in menu.<br>• PSTN receives ring back tone.<br>• After transfer is completed and communication is established between PSTN and OSV_B, PSTN | • MOH played to PSTN after consultation and before blind transfer is completed.<br>• No Media clipping and proper speech path after transfer when PSTN and OSV_B are connected.<br>• Number presentation on ringing and after call pick up. | OK | |

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| | hangs up. | • Both ends idle after call clearing. | | |

**Call Deflect**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 4-36 | • ASC_A calls ASC_B.<br>• ASC_B doesn't answer and deflects the call to AS_C.<br>• Leave ASC_C ring for a few seconds and pick up the call.<br>• After deflect is completed and communication is established between ASC_A and ASC_C, ASC_A hangs up. | • Calling party receives forwarding call display notification.<br>• Ring back tone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• Media clipping and speech path after forwarding.<br>• Both ends idle after call clearing. | NA | **Not Supported** |
| 4-37 | • ASC_A calls OSV_A.<br>• OSV_A doesn't answer and deflects the call to ASC_B.<br>• Leave ASC_B ring for a few seconds and pick up the call.<br>• After deflect is completed and communication is established between ASC_A and ASC_B, ASC_A hangs up. | • Calling party receives forwarding call display notification.<br>• Ring back tone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• Media clipping and speech path after forwarding.<br>• Both ends idle after call clearing. | OK | ASC_B displays "ASC_A > from OSV_A"<br>ASC_C is updated on the display with ASC_A after accepting the call. |

**Call Forwarding (Unconditional)**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 4-38 | • ASC_A calls ASC_B who has CFU activated via phone locally or via CFU feature access code (*40).<br>• ASC_B forwards the call immediately to ASC_C.<br>• Leave ASC_C ring for a few seconds and pick up the call.<br>• After forward is completed and communication is established between ASC_A and ASC_C, ASC_A hangs up. | • Call forwarding info is displayed on the phone.<br>• Calling party receives forwarding call display notification.<br>• Ring back tone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• Media clipping and speech path after forwarding.<br>• Both ends idle after call clearing. | OK | Activated the feature via PBX there is |
| 4-39 | • OSV_A calls ASC_A who has CFU activated via phone locally or via CFU feature access code (*40).<br>• ASC_A forwards the call immediately to OSV_B.<br>• Leave OSV_B ring for a few seconds and pick up the call. | • Call forwarding info is displayed on the phone.<br>• Calling party receives forwarding call display notification.<br>• Ring back tone heard by calling party.<br>• Number presentation on ringing and | OK | Activated the feature via PBX there is |

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|---|---|---|---|---|
| | • After forward is completed and communication is established between OSV_A and OSV_B, OSV_A hangs up. | after call pick up.<br>• Media clipping and speech path after forwarding.<br>• Both ends idle after call clearing. | | |
| 4-40 | • ASC_A calls OSV_A who has CFU activated via phone locally or via CFU feature access code (*40).<br>• OSV_A forwards the call immediately to OSV_B.<br>• Leave OSV_B ring for a few seconds and pick up the call.<br>• After forward is completed and communication is established between OSV_A and OSV_B, ASC_A hangs up. | • Call forwarding info is displayed on the phone.<br>• Calling party receives forwarding call display notification.<br>• Ring back tone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• Media clipping and speech path after forwarding.<br>• Both ends idle after call clearing. | POK | Activated the feature via PBX there is<br>In the call list displays the forwarded number. Not the initially. (OSV_B) |
| 4-41 | • PSTN calls ASC_A who has CFU activated via phone locally or via CFU feature access code (*40).<br>• ASC_A forwards the call immediately to ASC_B.<br>• Leave ASC_B ring for a few seconds and pick up the call.<br>• After forward is completed and communication is established between PSTN and ASC_B, PSTN hangs up. | • Call forwarding info is displayed on the phone.<br>• Calling party receives forwarding call display notification.<br>• Ring back tone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• Media clipping and speech path after forwarding.<br>• Both ends idle after call clearing. | OK | Activated the feature via PBX there is |

**Call Forwarding (Busy)**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 4-42 | • ASC_C calls busy subscriber ASC_A who has CFB activated via phone locally or via CFB feature access code (*41).<br>• ASC_A doesn't have call waiting activated and forwards the call immediately to ASC_B.<br>• Leave ASC_B ring for a few seconds and pick up the call.<br>• After forward is completed and communication is established between ASC_C and ASC_B, ASC_C hangs up. | • Call forwarding info is displayed on the phone.<br>• Calling party receives forwarding call display notification.<br>• Ring back tone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• Media clipping and speech path after forwarding.<br>• Both ends idle after call clearing. | OK | The display is correct - but there is a limitation for number of characters can be displayed. |
| 4-43 | • OSV_A calls busy subscriber ASC_A | • Call forwarding info is displayed on | OK | |

| | | | | |
|---|---|---|---|---|
| | who has CFB activated via phone locally or via CFB feature access code (*41).<br>• ASC_A doesn't have call waiting activated and forwards the call immediately to OSV_B.<br>• Leave OSV_B ring for a few seconds and pick up the call.<br>• After forward is completed and communication is established between OSV_A and OSV_B, OSV_A hangs up. | the phone.<br>• Calling party receives forwarding call display notification.<br>• Ring back tone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• Media clipping and speech path after forwarding.<br>• Both ends idle after call clearing. | | |
| 4-44 | • ASC_A calls busy subscriber OSV_A who has CFB activated via phone locally or via CFB feature access code (*41).<br>• OSV_A doesn't have call waiting activated and forwards the call immediately to OSV_B.<br>• Leave OSV_B ring for a few seconds and pick up the call.<br>• After forward is completed and communication is established between ASC_A and OSV_B, ASC_A hangs up. | • Call forwarding info is displayed on the phone.<br>• Calling party receives forwarding call display notification.<br>• Ring back tone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• Media clipping and speech path after forwarding.<br>• Both ends idle after call clearing. | OK | Display name on ASCOM_A updated correctly but cannot see that the call was redirected. |
| 4-45 | • PSTN calls busy subscriber ASC_A who has CFB activated via phone locally or via CFB feature access code (*41).<br>• ASC_A doesn't have call waiting activated and forwards the call immediately to ASC_B.<br>• Leave ASC_B ring for a few seconds and pick up the call.<br>• After forward is completed and communication is established between PSTN and ASC_B, PSTN hangs up. | • Call forwarding info is displayed on the phone.<br>• Calling party receives forwarding call display notification.<br>• Ring back tone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• Media clipping and speech path after forwarding.<br>• Both ends idle after call clearing. | OK | |

**Call Forwarding (No Reply)**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 4-46 | • ASC_C calls ASC_A who has CFNR activated via phone locally or via CFB feature access code (*42).<br>• ASC_A forwards the call to ASC_B.<br>• Leave ASC_B ring for a few seconds and pick up the call.<br>• After forward is completed and communication is established between ASC_C and ASC_B, ASC_C hangs up. | • Call forwarding info is displayed on the phone.<br>• Calling party receives forwarding call display notification.<br>• Ring back tone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• Media clipping and speech path after forwarding.<br>• Both ends idle after call clearing. | OK | The display is correct - but there is a limitation for number of characters can be displayed. |
| 4-47 | • OSV_A calls ASC_A who has CFNR activated via phone locally or via CFB feature access code (*42).<br>• ASC_A forwards the call to OSV_B.<br>• Leave OSV_B ring for a few seconds and pick up the call.<br>• After forward is completed and communication is established between OSV_A and OSV_B, OSV_A hangs up. | • Call forwarding info is displayed on the phone.<br>• Calling party receives forwarding call display notification.<br>• Ring back tone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• Media clipping and speech path after forwarding.<br>• Both ends idle after call clearing. | OK | |
| 4-48 | • ASC_A calls OSV_A who has CFNR activated via phone locally or via CFB feature access code (*42).<br>• OSV_A forwards the call to OSV_B.<br>• Leave OSV_B ring for a few seconds and pick up the call.<br>• After forward is completed and communication is established between ASC_A and OSV_B, ASC_A hangs up. | • Call forwarding info is displayed on the phone.<br>• Calling party receives forwarding call display notification.<br>• Ring back tone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• Media clipping and speech path after forwarding.<br>• Both ends idle after call clearing. | OK | Display name on ASCOM_A updated correctly but cannot see that the call was redirected. |
| 4-49 | • PSTN calls ASC_A who has CFNR activated via phone locally or via CFNR feature access code (*42).<br>• ASC_A rings and after a while forwards to SC_B.<br>• Leave ASC_B ring for a few seconds and pick up the call.<br>• After forward is completed and communication is established between PSTN and ASC_B, PSTN | • Call forwarding info is displayed on the phone.<br>• Calling party receives forwarding call display notification.<br>• Ring back tone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• Media clipping and speech path after forwarding. | OK | |

| | | | | |
|---|---|---|---|---|
| | hangs up. | • Both ends idle after call clearing. | | |

**MLHG**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 4-50 | • MLHG (Multi Line Hunt Group) is configured with ASC_A, ASC_B and ASC_C.<br>• OSV_A calls MLHG pilot DN.<br>• ASC_A rings and answers.<br>• Call is cleared at the end of the communication. | • Post Dial Delay (PDD).<br>• Ring back tone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• Media clipping and speech path after forwarding.<br>• Both ends idle after call clearing. | OK | |
| 4-51 | • MLHG (Multi Line Hunt Group) is configured with ASC_A, ASC_B and ASC_C.<br>• OSV_A calls MLHG pilot DN.<br>• ASC_A rings, but no answer.<br>• ASC_B rings, but no answer.<br>• ASC_C rings and answers.<br>• Call is cleared at the end of the communication. | • Post Dial Delay (PDD).<br>• Ring back tone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• Media clipping and speech path after forwarding.<br>• Both ends idle after call clearing. | OK | |
| 4-52 | • MLHG (Multi Line Hunt Group) is configured with ASC_A, ASC_B and ASC_C.<br>• PSTN calls MLHG pilot DN.<br>• ASC_A rings, but no answer.<br>• ASC_B rings, but no answer.<br>• ASC_C rings and answers.<br>• Call is cleared at the end of the communication. | • Post Dial Delay (PDD).<br>• Ring back tone heard by calling party.<br>• Number presentation on ringing and after call pick up.<br>• Media clipping and speech path after forwarding.<br>• Both ends idle after call clearing. | OK | |

**Voicemail**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 4-53 | • OSV_A and OSV_B call ASC_A who has call forwarding to voicemail system (Xpressions).<br>• OSV_A and OSV_B leave a voicemail message to ASC_A.<br>• ASC_A calls voicemail system to hear unread messages.<br>• ASC_A calls back CP_A from Xpressions. | • Proper MWI indication with the correct number of messages on phone.<br>• MWI is retained after IP-DECT base station restart.<br>• Voicemail messages are properly heard.<br>• Successful call establishment and proper speech path. | OK | Only one icon for more than one messages. |

| | | | | |
|---|---|---|---|---|
| | | • Both ends idle after call clearing. | | |
| 4-54 | • 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.<br>• MWI disappears from the phone after all voicemail messages are deleted. | OK | |

**Conference**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 4-55 | • ASC_A calls OSV_A and establishes communication.<br>• ASC_A initiates a consultation call to OSV_B.<br>• After OSV_B answers, ASC_A initiates a PBX conference.<br>• ASC_A calls ASC_B and ASC_A adds ASC_B to the conference.<br>• ASC_A calls ASC_C and ASC_A adds ASC_C to the conference.<br>• ASC_A calls OSV_C and ASC_A adds OSV_C to the conference.<br>• ASC_A (conference initiator) hangs up. | • Media clipping and proper speechpath after conferencing participants in the conference.<br>• Number(s) presentation on conference participants.<br>• Conference participants are still on the conference after ASC_A hangs up.<br>• Number(s) presentation after ASC_A hangs up on conference participants. | OK | A special license is required.<br>Conference is initiated via entering R3. The same for adding new participants. |
| 4-56 | • OSV_A calls ASC_A and establishes communication.<br>• OSV_B calls ASC_A who has call waiting activated.<br>• ASC_A merges calls with OSV_A and OSV_B to a large conference.<br>• ASC_B calls ASC_A and ASC_A adds ASC_B to the conference.<br>• ASC_C calls ASC_A and ASC_A adds ASC_C to the conference.<br>• ASC_A (conference initiator) hangs up. | • Media clipping and proper speechpath after conferencing participants in the conference.<br>• Number(s) presentation on conference participants.<br>• Conference participants are still on the conference after ASC_A hangs up.<br>• Number(s) presentation after ASC_A hangs up on conference participants. | OK | A special license is required.<br>ASCOM User should press R2 to accept the call and R3 to add the new participant in the conference |
| 4-57 | • ASC_A isn't configured with OpenScape Voice feature.<br>• ASC_A calls OSV_A and establishes communication.<br>• ASC_A initiates a consultation call to OSV_B.<br>• After OSV_B answers, ASC_A | • Media clipping and proper speechpath after conferencing participants in the conference.<br>• Number(s) presentation on conference participants.<br>• Number(s) presentation after ASC_A hangs up on conference participants. | NA | Local 3rd party conference is not supported by ASCOM |

| | | | | |
|---|---|---|---|---|
| | initiates a local conference.<br>• ASC_A (conference initiator) hangs up. | | | |

**Simultaneous Ringing**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 4-58 | • Activate Simultaneous Ringing on ASCOM_A<br>• ASCOM_A, ASCOM_B and OSV_C belong to the same simultaneous ringing group.<br>• OSV_A calls ASCOM_A.<br>• ASCOM_B answers the call.<br>• OSV_A hangs up at the end of the communication. | • All phones from the group are ringing and every ringing phone can answer the call.<br>• The phone number of calling and called phone is displayed.<br>• No media clipping when connecting both ends.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | If called users (ASCOM_A) declines the call all members stop ringing.<br><br>If other user declines the call , all other members still ringing. |
| 4-59 | • OS_A and ASCOM_A belong to the same simultaneous ringing group.<br>• PSTN calls OS_A.<br>• ASCOM_A answers the call.<br>• ASCOM_A hangs up at the end of the communication. | • All phones from the group are ringing and every ringing phone can answer the call.<br>• The phone number of calling and called phone is displayed.<br>• No media clipping when connecting both ends.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | |
| 4-60 | • From scenario 4-58, ASCOM_A switches off simultaneous ringing function with the feature access code (e.g. #10).<br>• OS_A calls ASCOM_A.<br>• ASCOM_A switches on simultaneous ringing function with the feature access code (e.g. *10).<br>• OS_A calls ASCOM_A. | • Successful activation / deactivation of simultaneous ringing with feature access code. | OK | |

**Serial Ringing**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 4-61 | • ASCOM_A,  ASCOM_B and OSV_B belong to the same serial ringing group.<br>• OSV_A calls ASCOM_A.<br>• ASCOM_A doesn't answer and ASCOM_B starts ringing.<br>• ASCOM_B picks up the call.<br>• OSV_A hangs up at the end of the communication. | • All phones from the group are ringing successive and every ringing phone can answer the call.<br>• The phone number of calling and called phone is displayed.<br>• No media clipping when connecting both ends.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | |
| 4-62 | • OS_A and ASCOM_A belong to the same serial ringing group.<br>• PSTN calls OS_A.<br>• OS_A doesn't answer and ASCOM_A starts ringing.<br>• ASCOM_A picks up the call.<br>• ASCOM_A_A hangs up at the end of the communication. | • All phones from the group are ringing successive and every ringing phone can answer the call.<br>• The phone number of calling and called phone is displayed.<br>• No media clipping when connecting both ends.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | |
| 4-63 | • From scenario 4-14, ASCOM_A switches off serial ringing function with the feature access code (e.g. *11 and then digit 3).<br>• OS_A calls ASCOM_A.<br>• ASCOM_A switches on simultaneous ringing function with the feature access code (e.g. *11 and then digit 3).<br>• OS_A calls ASCOM_A. | • Successful activation / deactivation of serial ringing with feature access code. | OK | |

**Call Pickup**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 4-64 | • OS_B and ASCOM_B belong to the same Call Pickup group.<br>• ASCOM_A calls OS_B.<br>• ASCOM_B picks up the call via phone menu or via call pickup access code (e.g. *7) while OS_B is ringing.<br>• ASCOM_B hangs up at the end of the communication. | • ASCOM_B gets a notification<br>• ASCOM_B is able to pick up the call.<br>• No media clipping when connecting both ends.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | A special OSV license is required. |
| 4-65 | • OS_B and ASCOM_B belong to the same Call Pickup group.<br>• OS_A calls ASCOM_B.<br>• OS_B picks up the call via phone menu or via call pick up access code (e.g. *7) while ASCOM_B is ringing.<br>• OS_B hangs up at the end of the communication. | • ASCOM_B gets a notification<br>• OS_B is able to pick up the call.<br>• No media clipping when connecting both ends.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | |
| 4-66 | • OS_A and ASCOM_A belong to the same Call Pickup group.<br>• PSTN calls OS_A.<br>• ASCOM_A picks up the call via phone menu or via call pick up access code (e.g. *7) while OS_B is ringing.<br>• ASCOM_B hangs up at the end of the communication. | • ASCOM_B is able to pick up the call.<br>• No media clipping when connecting both ends.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | |
| 4-67 | • OS_A and ASCOM_A belong to the same Call Pickup group.<br>• PSTN calls ASCOM_A.<br>• OS_A picks up the call via phone menu or via call pick up access code (e.g. *7) while ASCOM_B is ringing.<br>• OS_A hangs up at the end of the communication. | • OS_B is able to pickup the call.<br>• No media clipping when connecting both ends.<br>• Speech path in both directions.<br>• Both ends idle after call clearing. | OK | |

### 3.1.5. Audio Features

**Codecs**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 5-1 | • ASC_A is configured to have G.711a-law as first priority.<br>• ASC_A calls OSV_A. | • Connection is established with G.711a-law.<br>• Media clipping and speechpath. | OK | |
| 5-2 | • ASC_A is configured to have G.711u-law as first priority.<br>• ASC_A calls OSV_B. | • Connection is established with G.711u-law.<br>• Media clipping and speechpath. | OK | |
| 5-3 | • ASC_A is configured to have G.722 as first priority.<br>• ASC_A calls ASC_C. | • Connection is established with G.722.<br>• Media clipping and speechpath. | NA | ASCOM Uses different G.722. |
| 5-4 | • ASC_A is configured to have G.729 as first priority.<br>• ASC_A calls OSV_A. | • Connection is established with G.729.<br>• Media clipping and speechpath. | OK | |
| 5-5 | • ASC_A is configured to support G.711 only.<br>• OSV_A calls ASC_A. | • Connection is established with G.711.<br>• Media clipping and speechpath. | OK | |
| 5-6 | • ASC_A is configured to support G.722 only.<br>• OSV_A calls ASC_A. | • Connection is established with G.722.<br>• Media clipping and speechpath. | NA | ASCOM Uses different G.722 |
| 5-7 | • ASC_A is configured to support G.729 only.<br>• OSV_A calls ASC_A. | • Connection is established with G.729.<br>• Media clipping and speechpath. | OK | |
| 5-8 | • ASC_A is configured to support G.722.2 (AMR-WB), while ASC_B doesn't have G.722.2 (AMR-WB) as first priority.<br>• ASC_A calls ASC_B. | • Connection is established with G.722.2.<br>• Media clipping and speechpath. | OK | |
| 5-9 | • Make sure there is codec mismatch between ASC_A (G.711) and OSV_A (G.729) .<br>• OSV_A calls ASC_A. | • Call is disconnected.<br>• Call failure indication. | POK | ASCOM device rings, after picking up the phone, the call was dropped. The phone should not ring before the call is rejected. |
| 5-10 | • Make sure there is codec mismatch between ASC_A (G711) and OSV_A (G729) .<br>• ASC_A calls OSV_A. | • Call is disconnected.<br>• Call failure indication. | OK | |

**DTMF**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 5-11 | • OSV_A calls ASC_A.<br>• OSV_A sends DTMF 1234567890*# to ASC_A then ASC_A does the same towards OSV_A. | • DTMF digits are heard by the called party.<br>• DTMF digits are heard by the calling party. | OK | |
| 5-12 | • ASC_A calls OSV_A.<br>• ASC_A sends DTMF 1234567890*# to OSV_A then OSV_A does the same towards ASC_A. | • DTMF digits are heard by the called party.<br>• DTMF digits are heard by the calling party. | OK | |
| 5-13 | • 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.<br>• Login possible to personal voicemail box. | OK | |

### 3.1.6. Redundancy

**OS Voice Node Switchover**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 6-1 | • Perform a node switchover from node1 to node2 of the OpenScape Voice by putting out of service node1.<br>• Perform an incoming and outgoing call with ASC_A. | • The switchover should not be seen on the IP-DECT handsets.<br>• ASC_A is able to make and receive calls. | OK | |
| 6-2 | • The node1 is put back into service.<br>• Perform an incoming and outgoing call to ASC_A. | • The switchover should not be seen on the IP-DECT handsets.<br>• ASC_A is able to make and receive calls. | OK | |

**IP-DECT Base Station Switchover**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 6-3 | • The master base station fails.<br>• After switchover to the second base station, ASC_A calls ASC_B.<br>• 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.<br>• Calls are possible after switchover. | POK | 1. Calls are possible after new registration<br>2. No speechpath to Active calls<br>Note: No DNS was used from IP-Dect side |
| 6-4 | • In mirror configuration the functions do not return to Master automatically.<br>• After scenario 6-3, switch back manually to the first base station via GUI.<br>• After switchover to the first base station, ASC_A calls ASC_B.<br>• 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.<br>• Calls are possible after switchover. | OK | Active calls were dropped. |

**Voicemail**

| Test Name | Test Description | Functional Checks | Result | Comments |
|---|---|---|---|---|
| 6-5 | • OSV_A call ASC_A who has call forwarding to voicemail system (Xpressions).<br>• OSV_A leaves a voicemail message to ASC_A.<br>• Restart IP-DECT Station<br>• ASC_A calls voicemail system to hear unread messages.<br>• ASC_A clears the message | • Proper MWI indication with the correct number of messages on phone.<br>• MWI is retained after IP-DECT base station restart.<br>• Voicemail messages are properly heard.<br>• Successful call establishment and proper speech path.<br>• Both ends idle after call clearing. | 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:**

Customer Solution Lab

| Test Group | OK | POK | NOK | NA | NT | Total |
|---|---|---|---|---|---|---|
| 1. Registration | 4 | 0 | 0 | 0 | 0 | 4 |
| 2. Basic Calls | 33 | 1 | 0 | 0 | 0 | 34 |
| 3. Basic Calls Extended | 13 | 0 | 0 | 1 | 0 | 14 |
| 4. Telephony Features | 66 | 1 | 1 | 2 | 0 | 70 |
| 5. Audio Features | 10 | 1 | 0 | 2 | 0 | 13 |
| 6. Failover | 4 | 1 | 0 | 0 | 0 | 5 |
| **Sum** | **130** | **4** | **1** | **5** | **0** | **140** |
| **Percentage** | **92,9%** | **2,9%** | **0,7%** | **3,6%** | **0,0%** | **100,0%** |

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

# 4. Configuration Data

## 4.1. OpenScape Voice Configuration

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

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

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

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

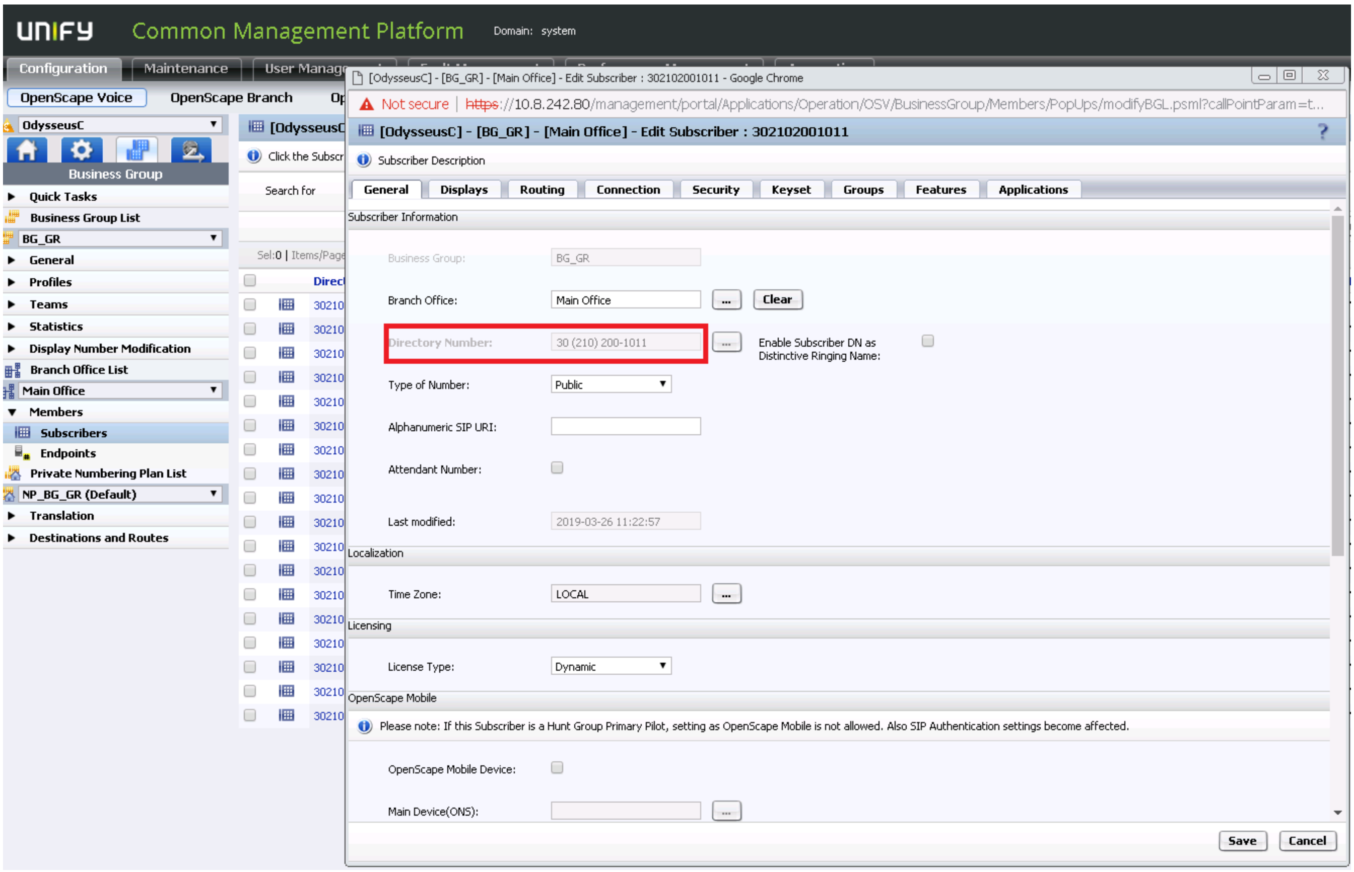
At the following paragraphs an example configuration is presented.

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

#### *4.1.1.1. OS Voice Subscriber Creation*

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

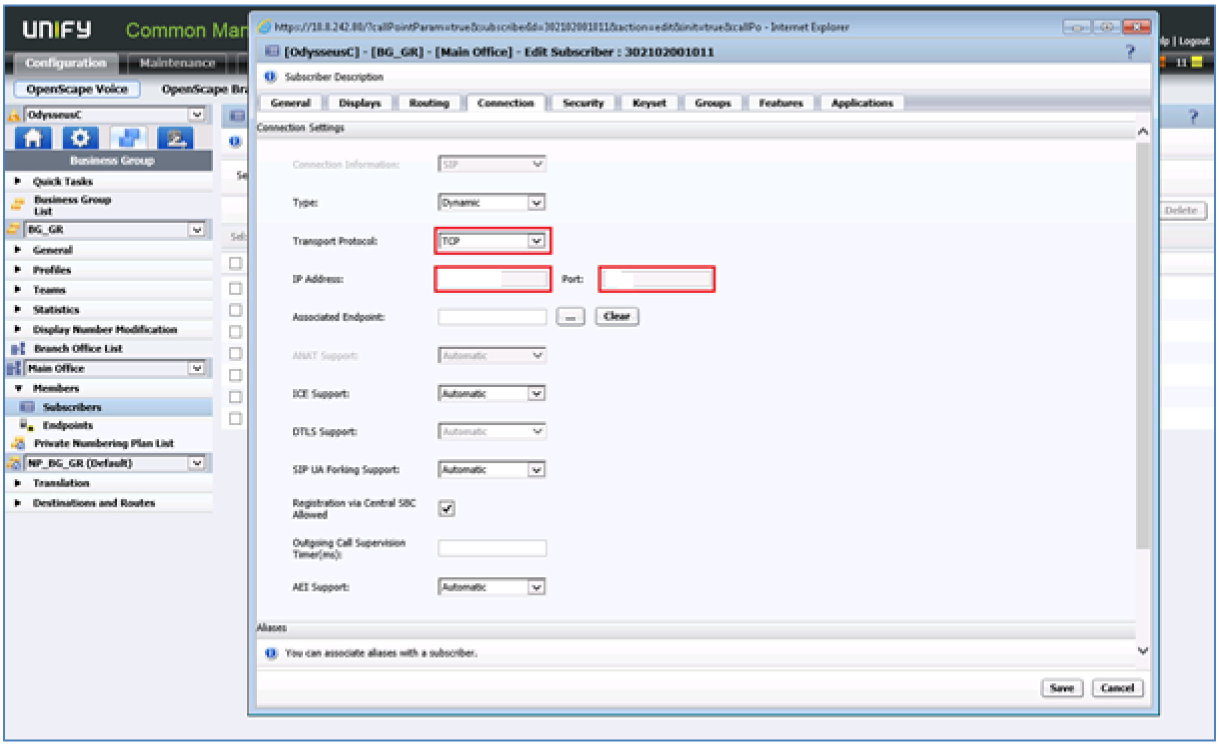
Go to **Configuration** >> **OpenScape Voice** >> **Business Group** >> **Members** >> **Subscribers**




Click on [Add & General].

Select the appropriate available **Directory Number** from the list for the analog user, e.g. 30 (210) 100-1011.

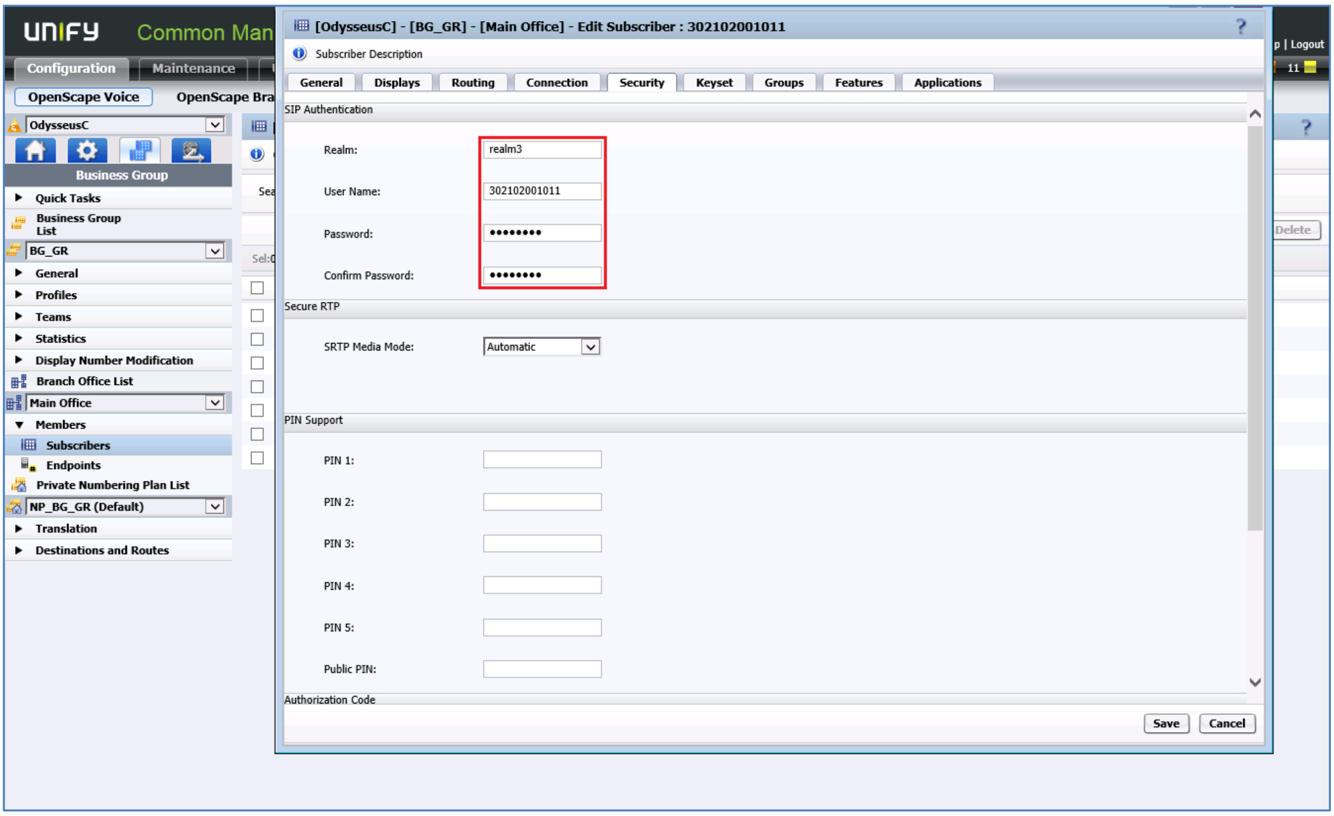
Click on [Connection]



Select the appropriate value for **Transport Protocol**, i.e. TCP.

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

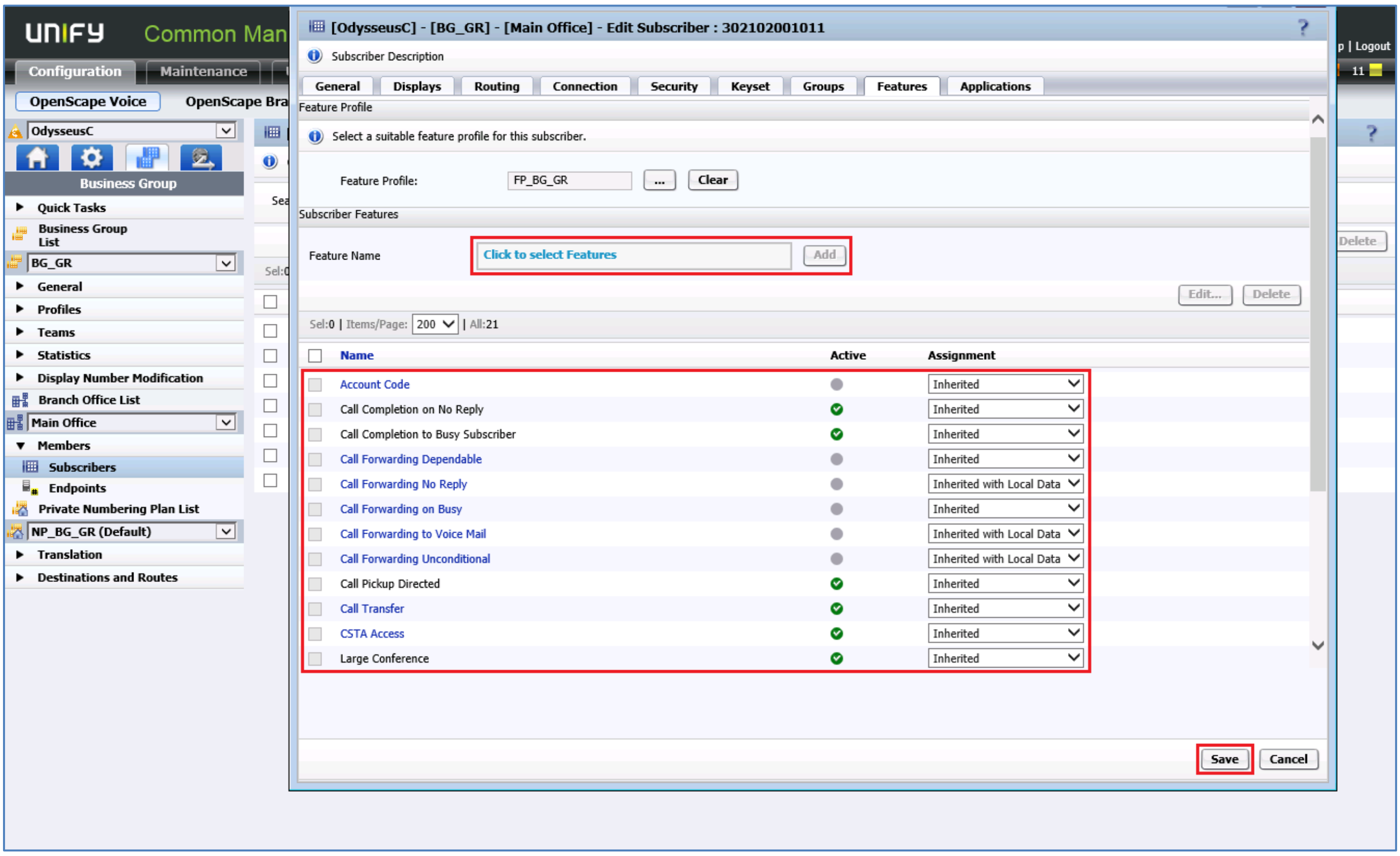
Click on [Security].




Hereby, the authentication of a subscriber to OS Voice is determined. In current environment, **Digest Authentication** is required at OS Voice, therefore **Security** tab needs to be populated.

Configure **Realm** value (e.g. realm3) an appropriate subscriber **User Name** (e.g. 302102001011) and **Password**. These credentials will be used for OSV end user authentication.

Click on [Features].




The administrator may add the required IP telephony features for the subscriber by **Feature Name** menu.

Click on [Save].

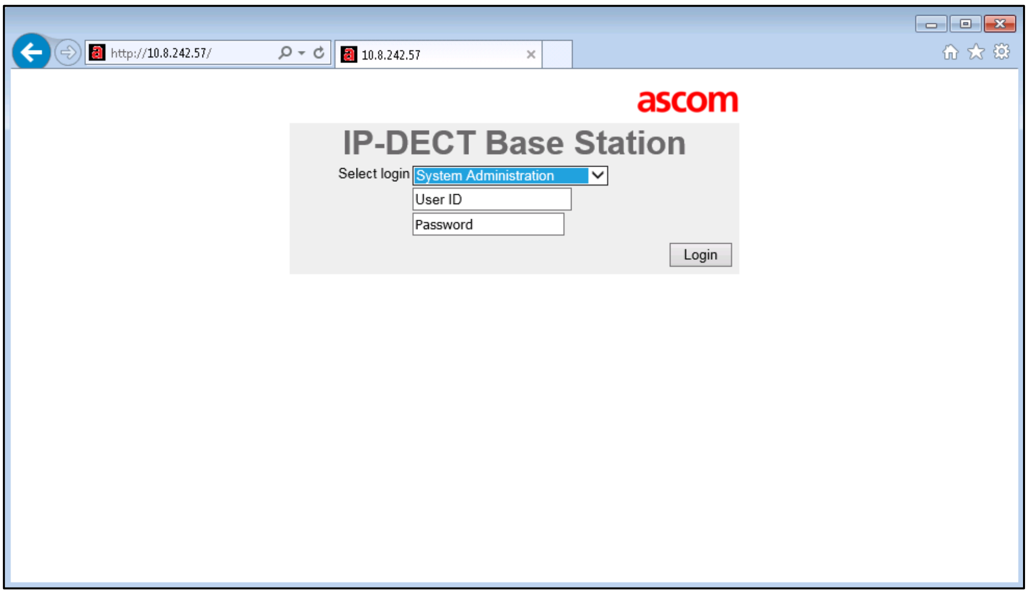
Repeat the same for all subscribers.

## 4.2. 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 OpenScape Voice, 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.

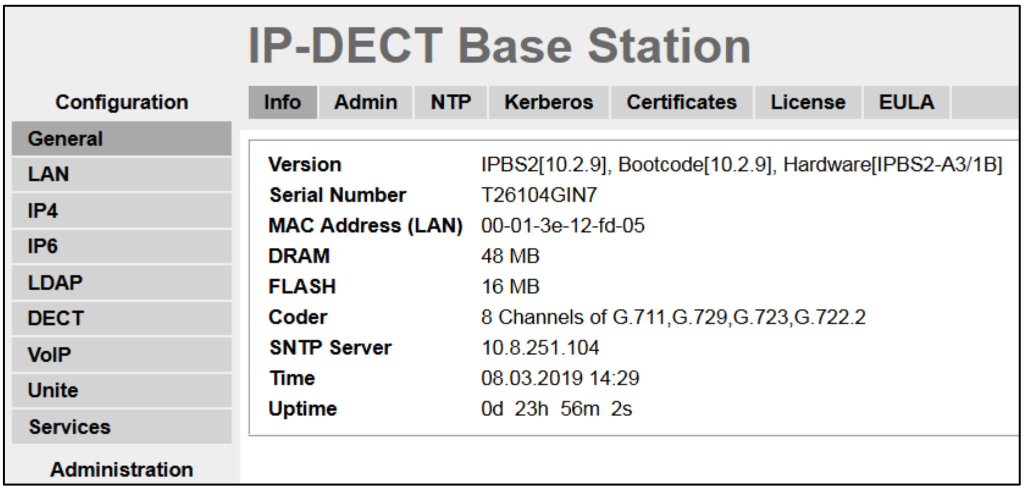
Default or non-project specific Ascom IP-DECT system configuration will not be referenced to in subsequent paragraphs.

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



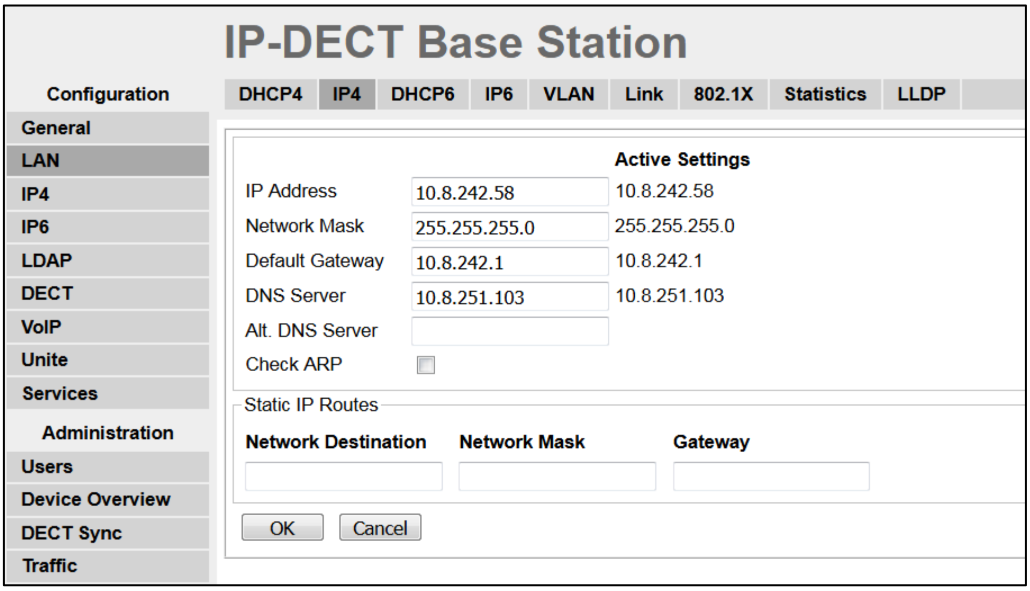

Click the System administration link and enter the appropriate credentials to access the Ascom IP-DECT Base Station and then click on **OK**.

### 4.2.1. General Configuration of IP-DECT Base Station




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

### 4.2.2. IP-DECT Base Station LAN IP




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

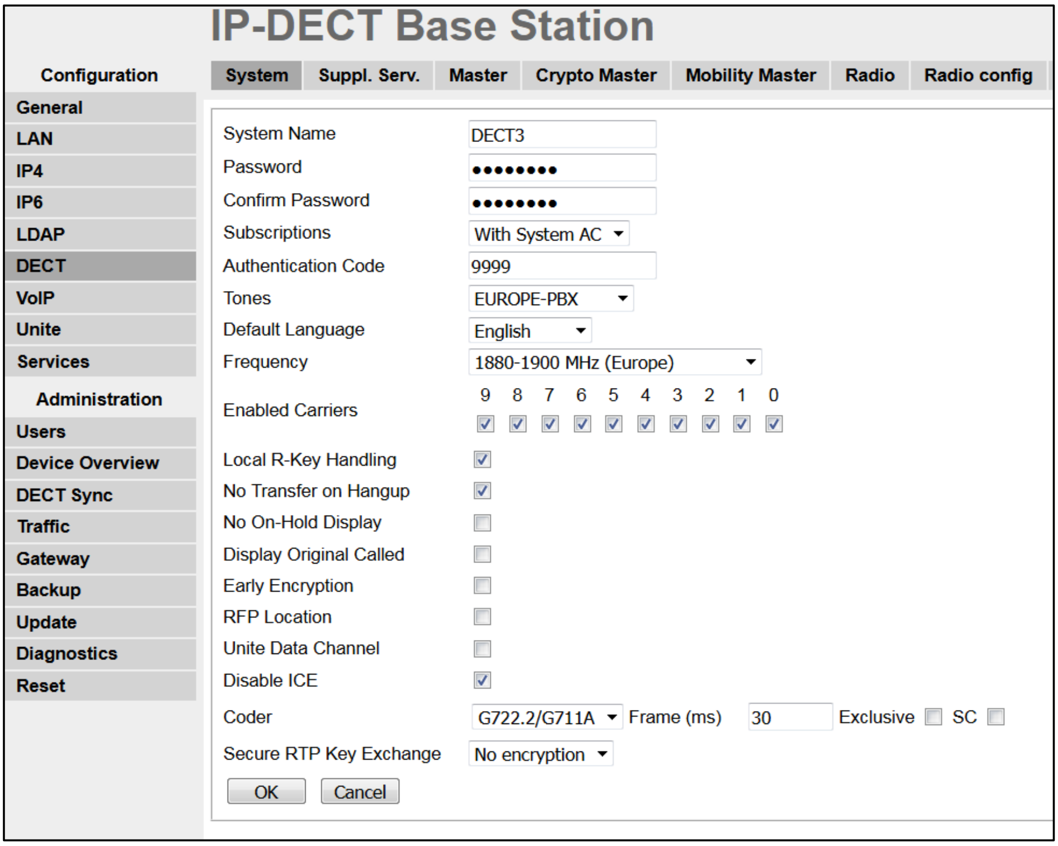
- **IP Address** 10.8.242.58
- **Network Mask** 255.255.255.0
- **Default Gateway** 10.8.242.1
- **DNS Server** 10.8.251.103

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.

### 4.2.3. DECT System



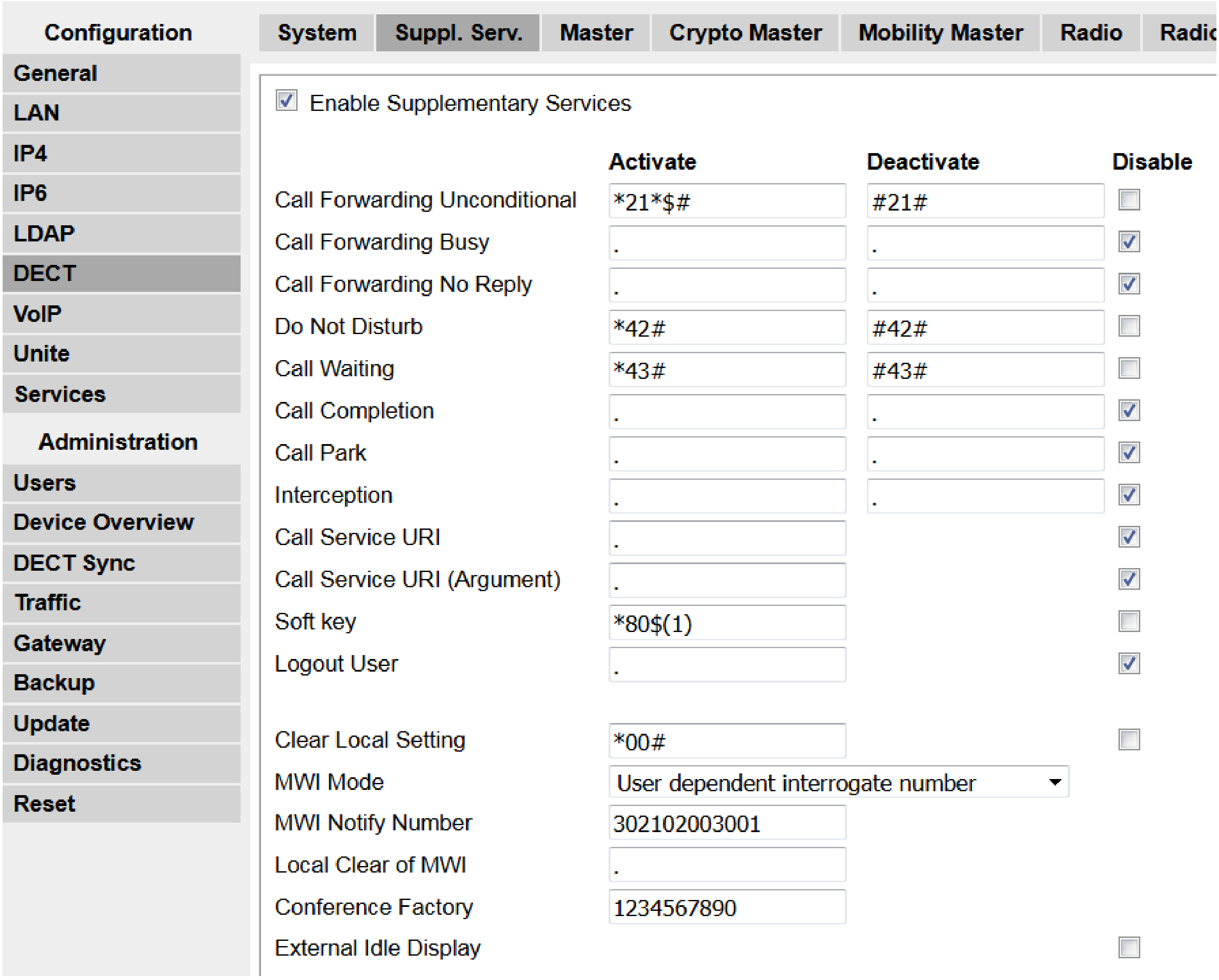

Click on the System tab and enter the following:

- **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.

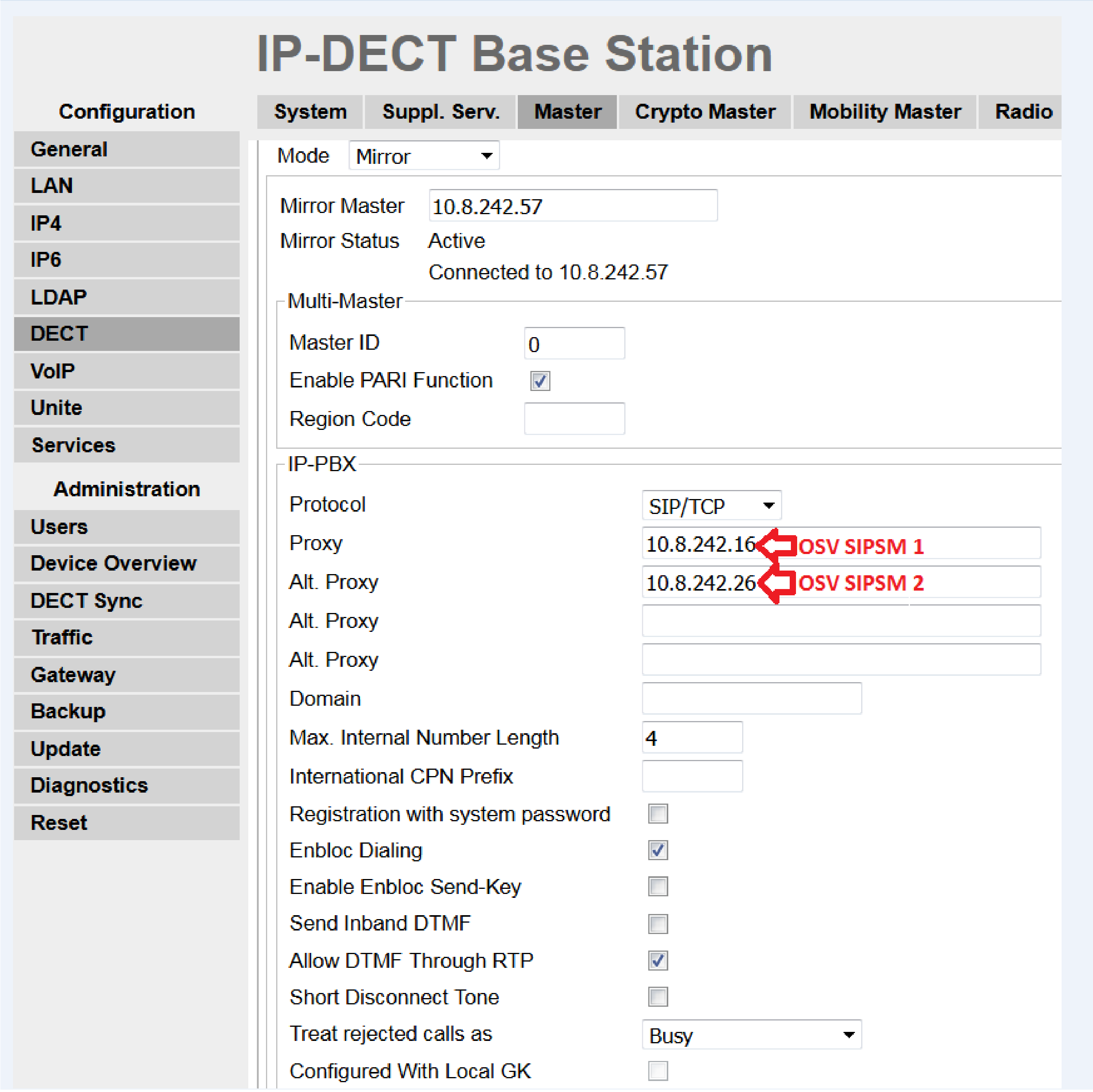
### 4.2.4. IP-DECT Suppl. Serv.

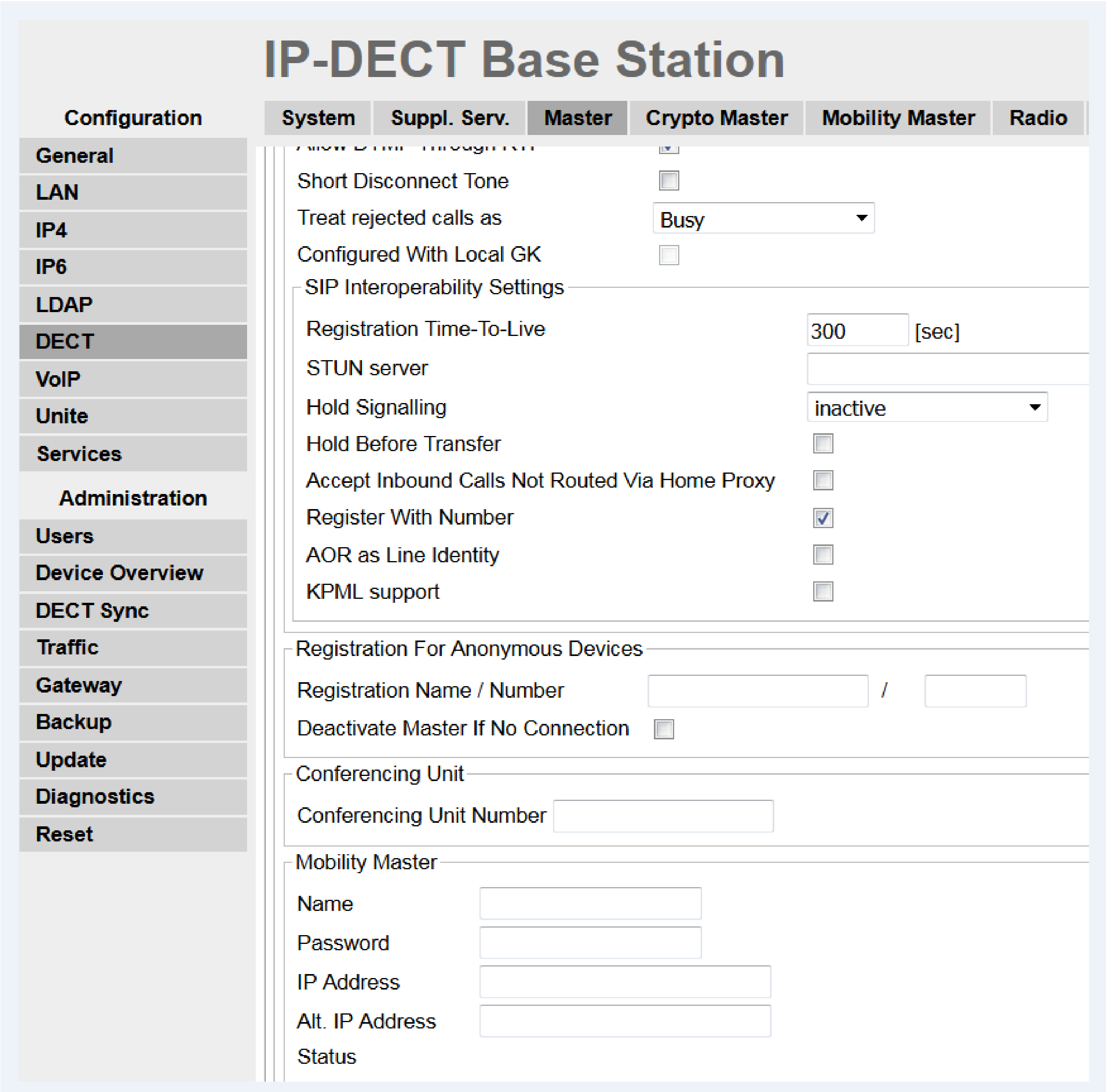



Important Notes:

1. In the Conference Factory field is specified the Conference Factory Feature Access Code specified in Voice to enable an Ascom user to initiate a Large Conference handled on the Media Server attached to Voice.
2. Unify license is required for Group Pickup and Large Conference features.
3. Make sure that Feature Access Code under this tab do not conflict with those on the IP-PBX.

### 4.2.5. IP-DECT Master – Active Base Station



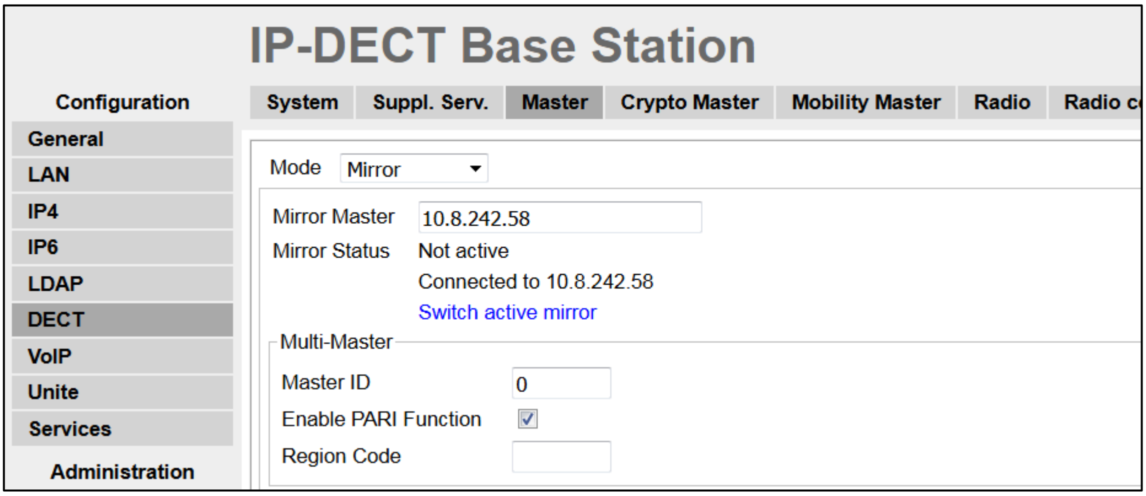



Navigate to **DECT**, click on **Master** and enter the following:

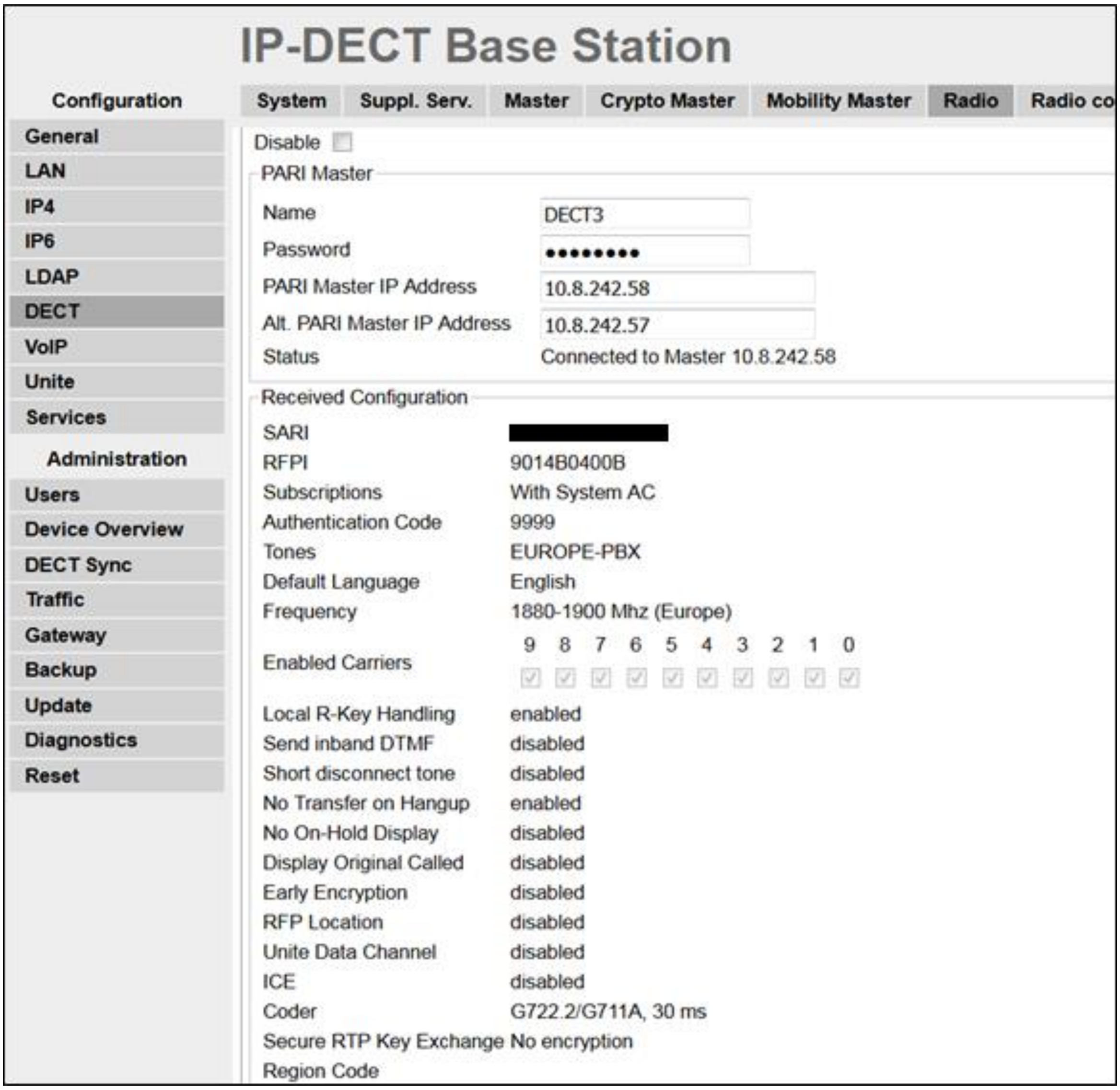
- **Mode** — Select Mirror from the dropdown box
- **Mirror Master IP address** — Mirrored Master base station IP: 10.8.242.57
- **Enable PARI Function check box** — Checked
- **Protocol** — Select SIP/TCP from the dropdown box
- **Proxy** — OSV SIPSM 1: 10.8.242.16
- **Alt Proxy** — OSV SIPSM 2: 10.8.242.26
- **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).

### 4.2.6. IP-DECT Master – No Active Base Station




### 4.2.7. DECT Radio – Active Base Station



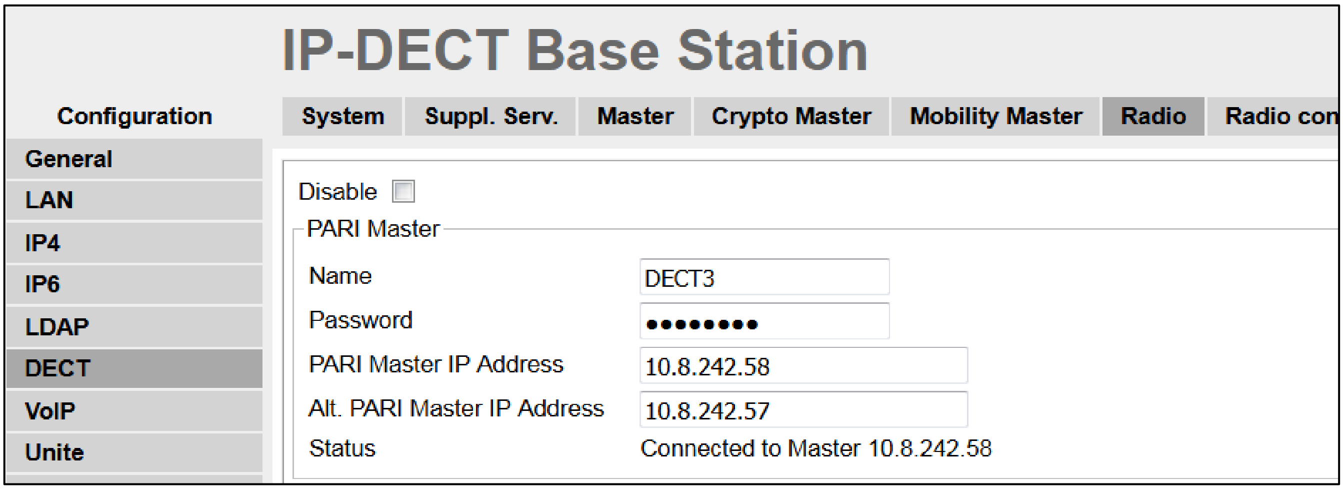
Navigate to **Radio** and select the Admin tab and enter the following:

- **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

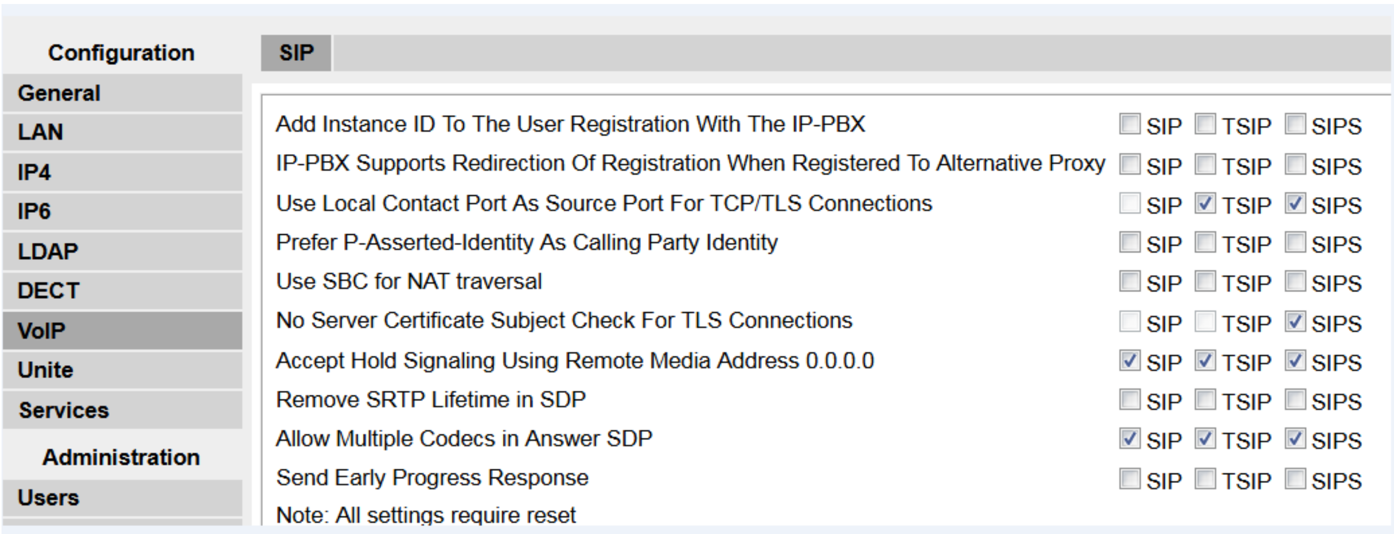
Click on the **OK** Button to save.

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

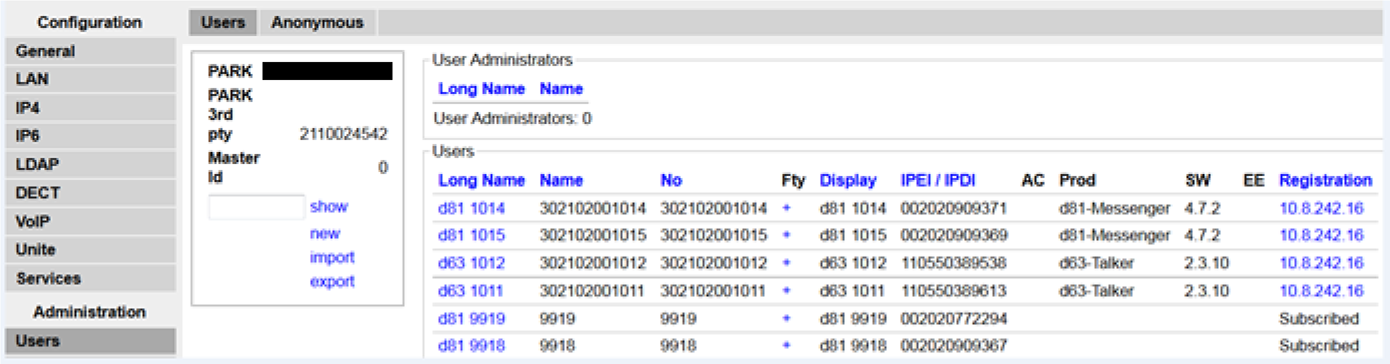
### 4.2.8. DECT Radio – No Active Base Station




### 4.2.9. VoIP



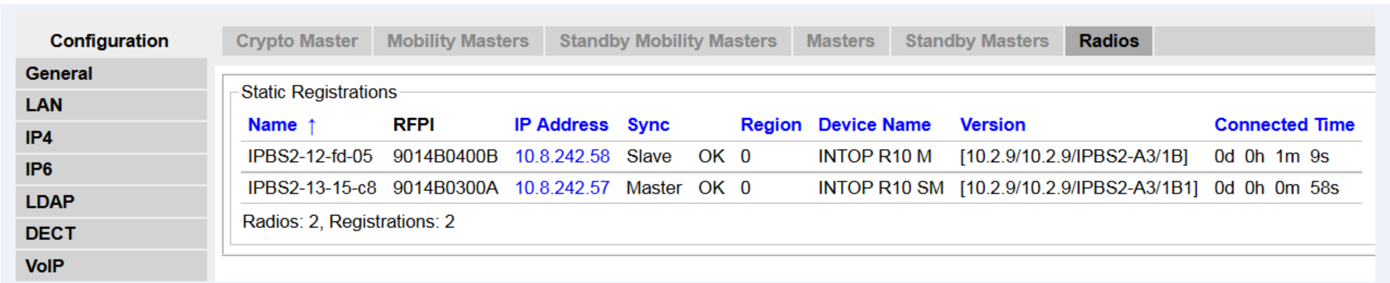
## 4.2.10. IP-DECT Users




To enable an Ascom user to use the Voice feature *Call Pickup Group* the Call Pickup Feature Access Code specified in Voice must be configured by clicking on + in the Fty column for the user:

The Voice Feature Access Code for Group Pickup must be configured here.

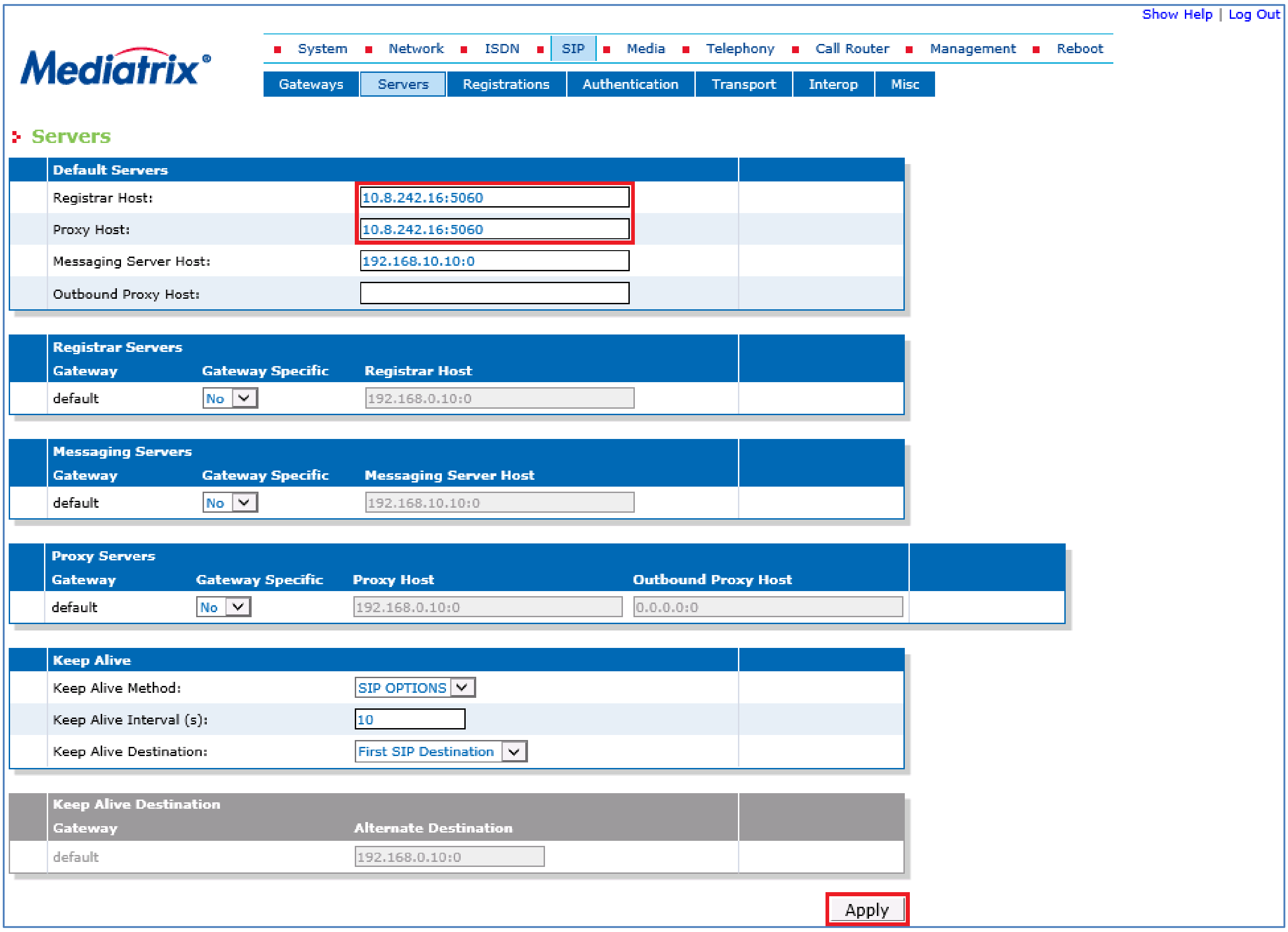
## 4.2.11. Device Overview



## 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 OS Voice and for ISDN provider is out of scope and therefore omitted.

1. Go to **SIP** >> **Servers**

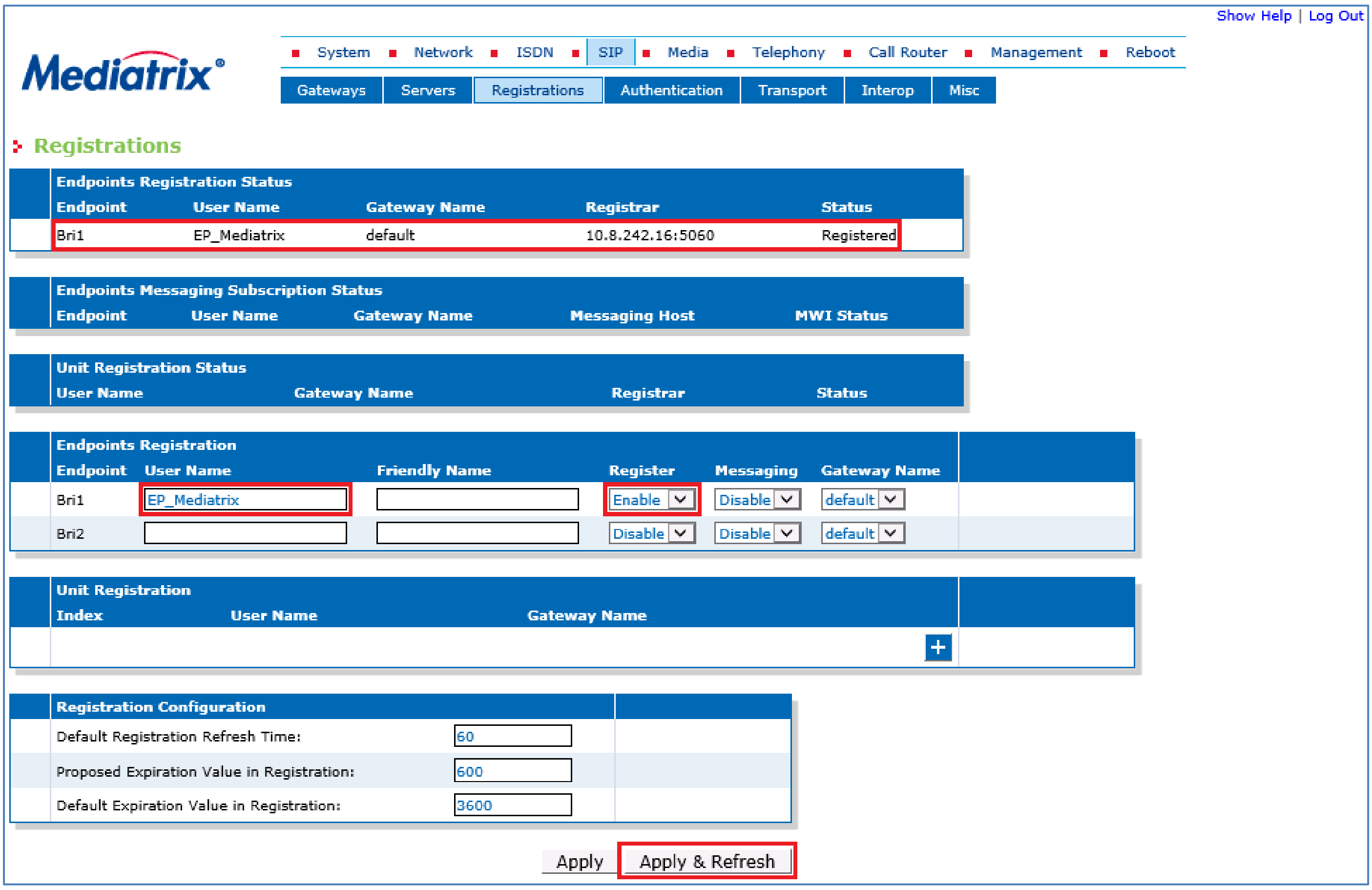



Configure the value 10.8.242.16:5060 (OS Voice SIPSM & non-secure port) for Registrar Host and Proxy Host.

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

Click on [Apply].

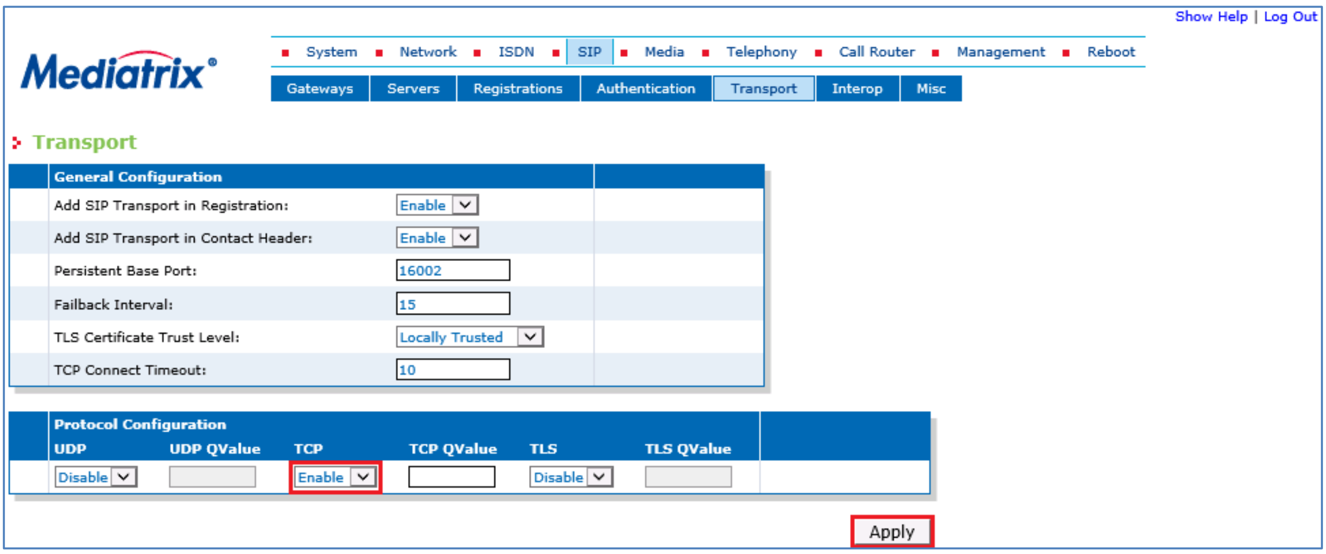
2. Go to **SIP** >> **Registrations**




Since Mediatrix 4402plus is connected to ISDN provider via devices Bri1 port, set an endpoint User Name, e.g. EP_Mediatrix and for registration to OS Voice set Register = Enable.

Click on [Apply & Refresh].

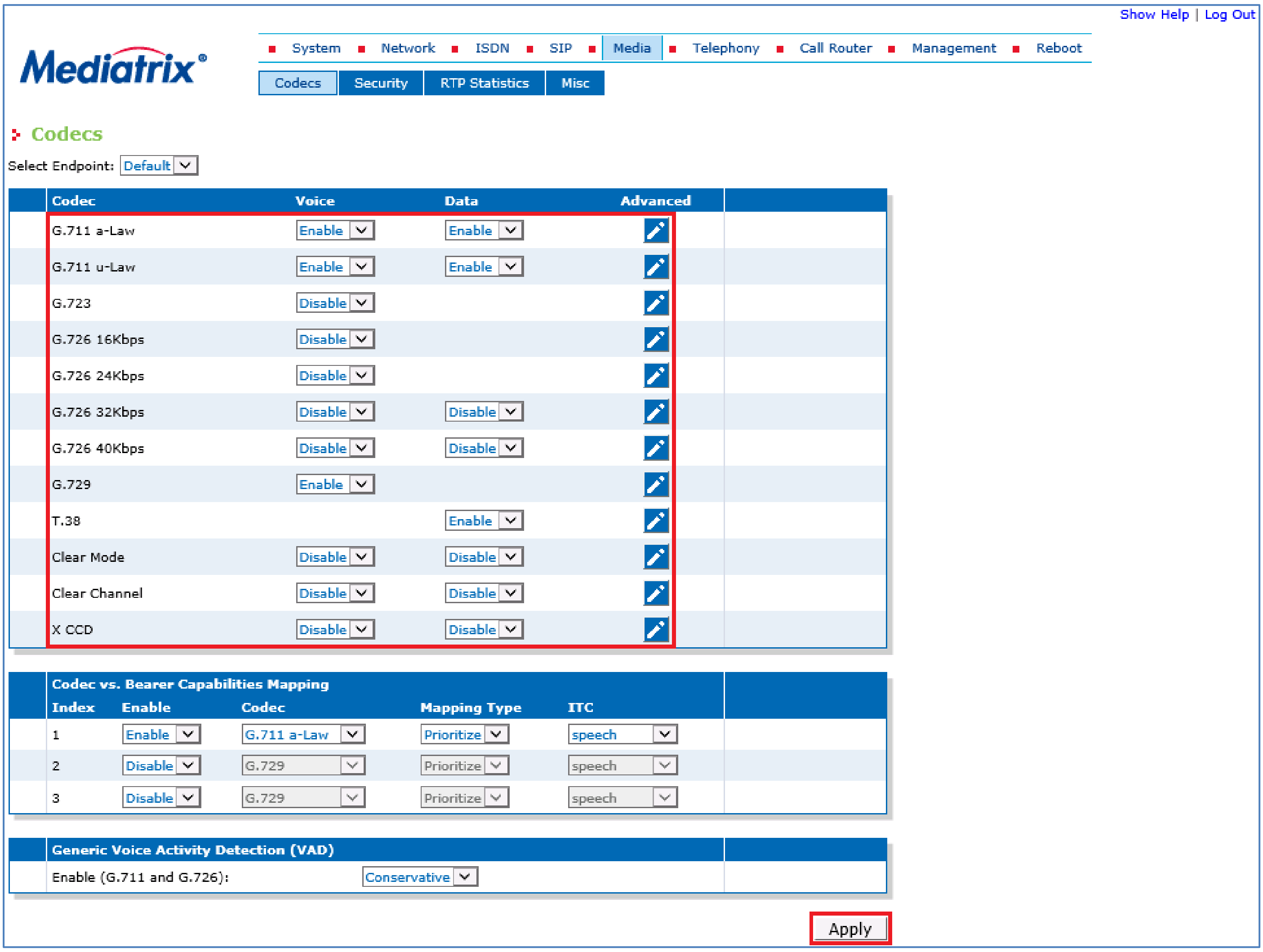
3. Go to **SIP** >> **Transport**



For TCP connection to OS Voice, **TCP** = Enable.

Click on [Apply].

4. Go to **Media** >> **Codecs**




During audio or fax calls between analog subscribers (physically connected to MP-1288) and PSTN subscribers (via Mediatrix 4402plus), media traffic is exchanged between MP-1288 and Mediatrix 4402plus.

Select the desired **Codec**(s) according to test scenario needs for **Voice** (audio calls) and **Data** (fax calls) and set the value Enable.

Click on [Apply].

# 5. Confirmation

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

| <Location, Date > | Valid until <date> |
|---|---|
| (Name, function and signature of authorized person) | (Name, function and signature of authorized person) |

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Unify products represent a strong heritage of technology innovation, reliability and flexibility. Their award-winning intuitive user experience can be delivered through almost any device and in any combination of cloud or on-premise deployment. Augmented by Atos' secure digital platforms, vertical solutions and transformation services, they set the global standard for a rich and reliable collaboration experience that empowers teams to deliver extraordinary results.

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