SHORETEL APPLICATION NOTE

for

Ascom IP-DECT

Date:	Dec 12, 2016
App Note Number:	TC-16085
For use with:	Ascom IP-DECT, version 9.0.6
Product:	ShoreTel Connect ONSITE
System:	ST Connect 21.79.9330.0

Contents

ShoreTel tests and validates the interoperability of the Member's solution with ShoreTel's published software interfaces. ShoreTel does not test, nor vouch for the Member's development and/or quality assurance process, nor the overall feature functionality of the Member's solution(s). ShoreTel does not test the Member's solution under load or assess the scalability of the Member's solution. It is the responsibility of the Member to ensure their solution is current with ShoreTel's published interfaces.

The ShoreTel Technical Support organization will provide Customers with support of ShoreTel's published software interfaces. This does not imply any support for the Member's solution directly. Customers or reseller partners will need to work directly with the Member to obtain support for their solution.

Introduction

This document describes the configuration procedures for integrating the Ascom IP-DECT Handsets as SIP extensions on the ShoreTel Connect ONSITE system.

Ascom

Ascom IP-DECT combines Voice over IP with Digitally Enhanced Cordless Telephony (DECT) technology. Ascom IP-DECT a reliable wireless communication solution that offers enterprise-grade telephony, profesional messaging, personal alarm, and positioning over secure dedicated frequency bands. It is developed based on open standards, such as SIP, which maximizes interoperability with leading vendors.

Features

Ascom IP-DECT handsets:

- The Ascom d41 is targeted for users in office environments with a need for a handset with high quality voice and easy access to PBX features.
- The Acom d62 is targeted towards users in medium demanding environments such as hospitals, for users with a need for messaging or alarm functionality.
- The Ascom d81 is the top of the line handset in the Ascom DECT portfolio. It is an extremely robust handset for demanding environments intended for professional users who need to be reachable by voice and messages.
- Professional messaging
- Standards based solution
- Wide range of handsets from office to ruggedized and explosion safe
- Longest industry talk time

Ascom IP-DECT Base Station:

- Dedicated VoIP wireless base station
- Cost efficient mobility solution
- Provides unmatched scalability (1,000 base stations per handover domain and 100,000 users per system)
- Utilizes DECT encryption to prevent eavesdropping
- Utilizes Ethernet backbone for wired infrastructure (shared or dedicated)
- Provides seamless handovers with over-air synchronization

Ascom IP-DECT Gateway:

- With the Ascom IP-DECT gateway, existing DECT systems can be upgraded with IP telephony functionality in a secure radio environment.
- The IP-DECT gateway is compatible with all currently available and previous Ascom legacy DECT base stations.

Technical Support

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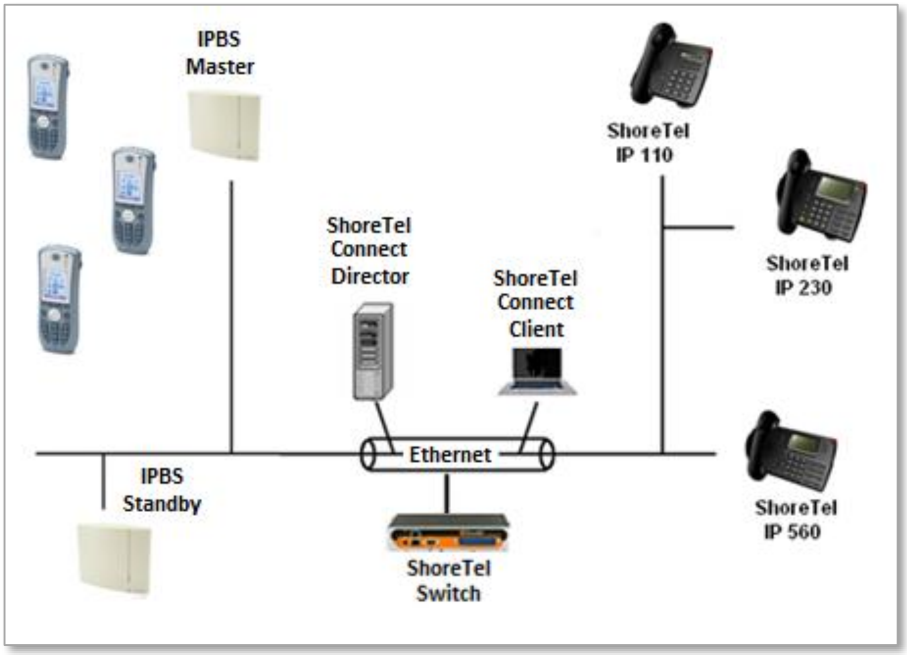
Help desk: +46 31 559450

Network Topology



Test Environment

- ShoreTel Connect ONSITE Server
- ShoreTel Virtual Phone Switch
- ShoreGear Switch
- ShoreTel IP Phones
- Ascom IP-DECT Handsets (9.0.6)

NOTE: This Application Note assumes the setup, configuration and licensing of the Virtual/Physical Switches has already been completed. If you require additional information, please refer to the ShoreTel Connect Onsite Planning and Installation guide at the following location.

[ShoreTel Connect Onsite Planning and Installation Guide]

Special Notes

The following are the caveats and limitations of the Ascom IP-DECT handsets with the ShoreTel Connect ONSITE system.

ShoreTel Extension License

Extension Licenses are required for each Ascom IP-DECT user.

ShoreTel SIP Phone License

Deployment of SIP Extensions require a SIP Phone License. One SIP Phone License is required for each Ascom IP-DECT SIP Extension.

Call Forwarding

When call forwarding is configured on the Ascom IP-DECT handsets, the Ascom IP-DECT uses SIP message “302 Moved Temporarily” to forward the call. Forwarding a call via a 3xx response is not supported by ShoreTel. Call forwarding set in the ShoreTel system is however supported and successfully tested.

NOTE: For additional information on SIP Endpoints with a ShoreTel Connect ONSITE system, please refer to Chapter 19 of the ShoreTel Connect Onsite System Administration Guide.

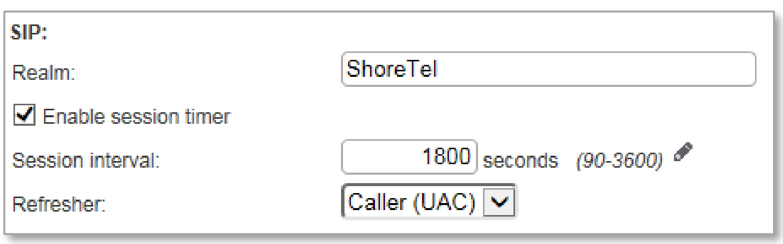
ShoreTel Configuration

This section describes the detailed steps required on the ShoreTel Connect ONSITE system to configure the Ascom IP-DECT handsets as a SIP extension.

Call Control Options

This section describes the SIP settings required on the ShoreTel system to work with Ascom IP-DECT handsets.

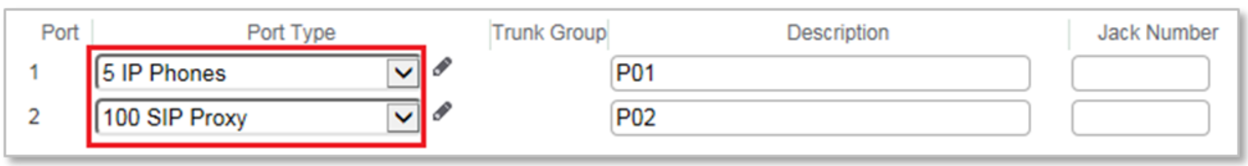
1. Navigate to Administration > Features > Call Control > Options
2. Verify the parameters located under the **SIP** section
3. **Realm**: The realm is used in authenticating all SIP devices. Changing this value will require a reboot of switches serving as SIP extensions. It is not necessary to modify this parameter
4. **Enable SIP Session Timer**: Ensure this parameter is checked
5. **Session interval**: Session interval value indicates the SIP session registration period. There is no need to modify the default value of 1800 seconds.
6. **Refresher**: The refresher setting decides if user agent client or user agent server refreshes the session. There is no need to modify the default value of “Caller (UAC).”
7. Click **SAVE**



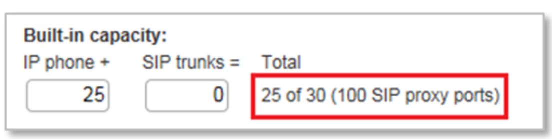
SIP Proxy Settings – Allocating Ports for SIP Extensions

This section describes the Switch configuration required on the ShoreTel system to work with Ascom IP-DECT as a SIP Extension. Depending on the switch type, ShoreTel Voice Switches, and Virtual Phone Switches support variable numbers of SIP Proxies and IP Phones, and can be verified on the Switch Edit page of ShoreTel Connect Director.

ShoreTel ShoreGear Switches with processing resources that support Digital and Analog ports can be reallocated to support 100 SIP Proxies. The ShoreTel Administrator can define one of the "Port Type" settings from the available ports to "100 SIP Proxy", as well as sufficient "IP Phone" ports to support the total number of Ascom IP-DECT users. The following example shows Port allocation designated on a ShoreTel SG-90 for IP Phones and SIP Proxy resources



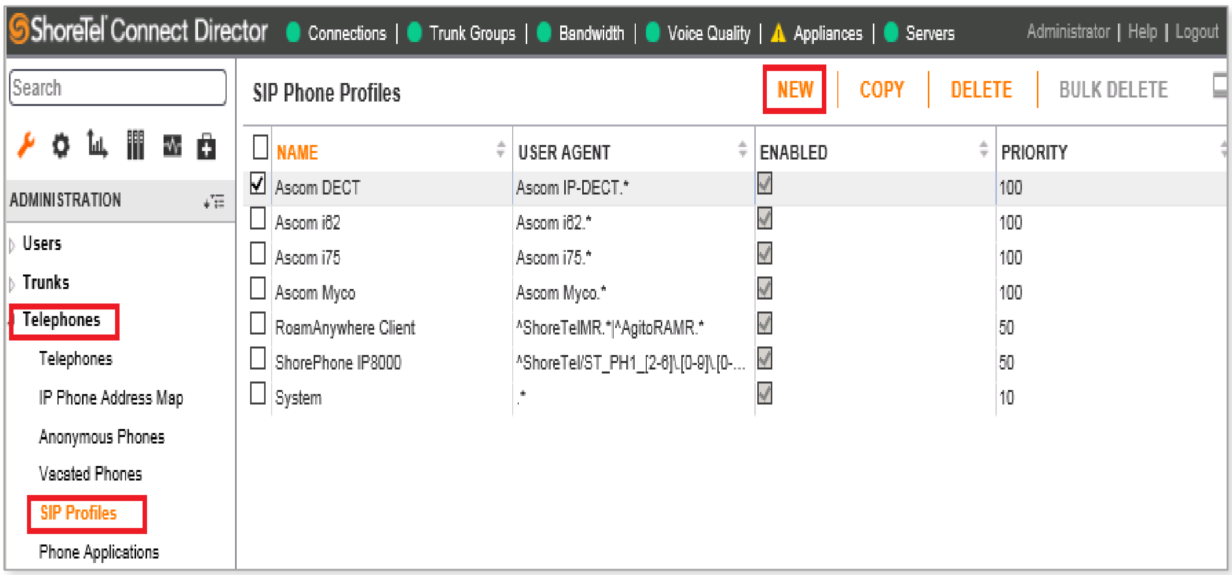
If the ShoreTel ShoreGear Switch that you have selected has "built-in" capacity (i.e., ShoreGear 50/90/220T1/E1, etc.) for IP phones and SIP trunks, you can also remove 5 ports from the total number available to provide the "100 SIP Proxy" configuration necessary. Every 5 ports you remove from the total available will result in "100 SIP Proxy" ports being made available. The following example shows 5 ports removed from total available resulting in 100 SIP Proxy ports being available.



SIP Profile

ShoreTel Connect Director's "Call Control" section contains the "SIP Profiles" option. By default, the Ascom IP-DECT utilize the "_System" profile. In order to optimize the functionality, you will need to add a custom profile. This is accomplished from ShoreTel Connect Director.

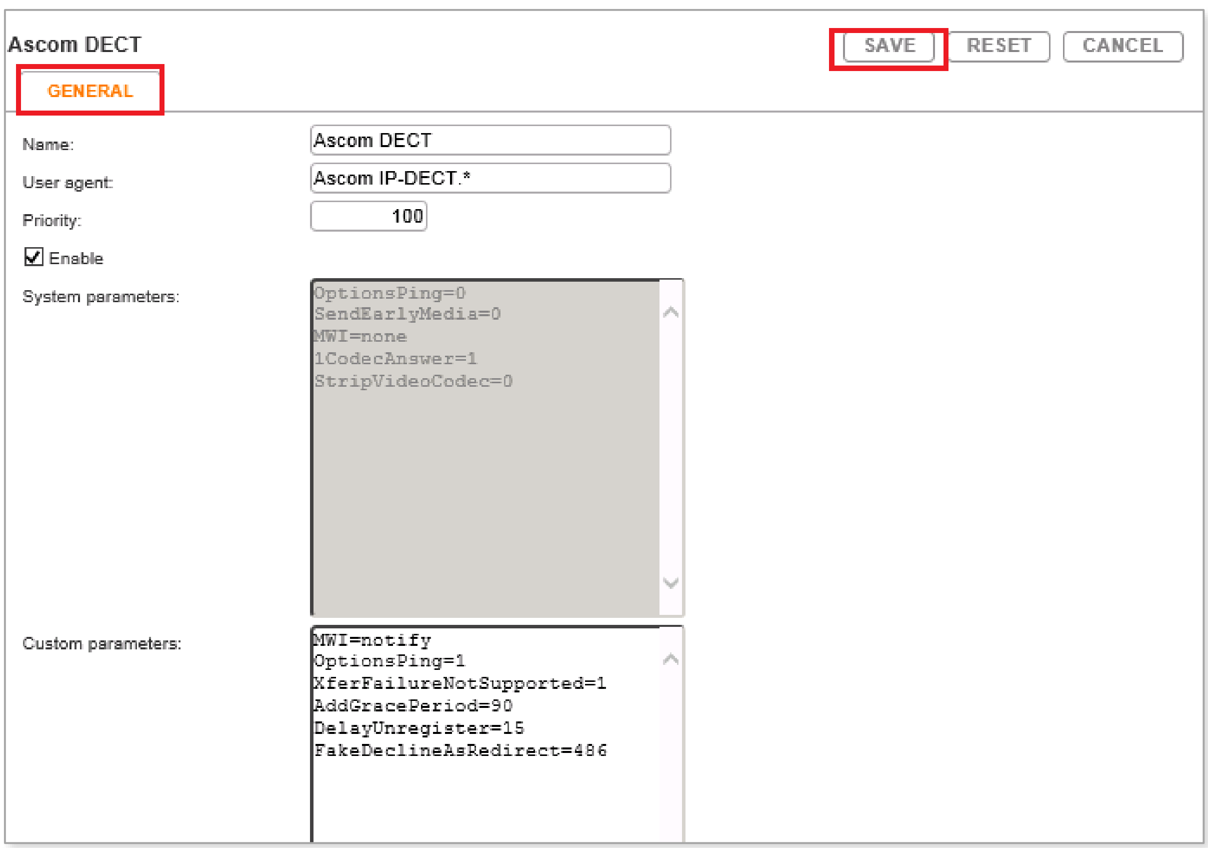
1. Navigate to Administration > Telephones > SIP Profiles
2. Click **New**, to create a new SIP Profile.



3. In the General Tab, define a **Name:** for the entry, and be sure to define an appropriate name.
4. For the parameter **User agent:**, enter “Ascom IP-DECT.*” (without quotes)
5. The parameter “**Priority:**” defaults to 100, no change is required.
6. Enable the profile by checking (enabling) the **Enable** option.
7. In the “**Custom Parameters:**” options, add the following entries:

OptionsPing=1
MWI=notify
FakeDeclineAsRedirect=486
XferFailureNotSupported=1
AddGracePeriod=90
DelayUnregister=15

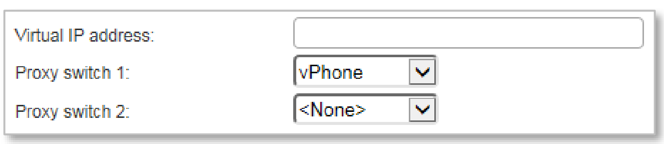
8. Click **SAVE**



Site Settings

The next settings to address are the administration of Sites. The ShoreTel Administrator can designate up to two Proxy switches per site for redundancy and reliability: one switch is assigned as the primary Proxy server, and the other switch acts as the backup Proxy server in case the primary fails. A Virtual IP Address is the IP Address of the switch that is configured as the SIP Proxy server for the Site. The Virtual IP Address must be static. If you choose not to define a "Virtual IP Address," you can only define one proxy switch, and there will be no redundancy or failover capabilities. The switches available in the "Proxy Switch 1 / 2" will only be shown if proxy resources have been enabled on the switch. This is accomplished from ShoreTel Connect Director.

1. Navigate to Administration > System > Sites
2. Select the name of the Site in which SIP Proxies will be assigned
3. In the General Tab, set **Proxy switch 1:** Select the ShoreTel switch configured with SIP Proxies for the Site
4. Click **SAVE**



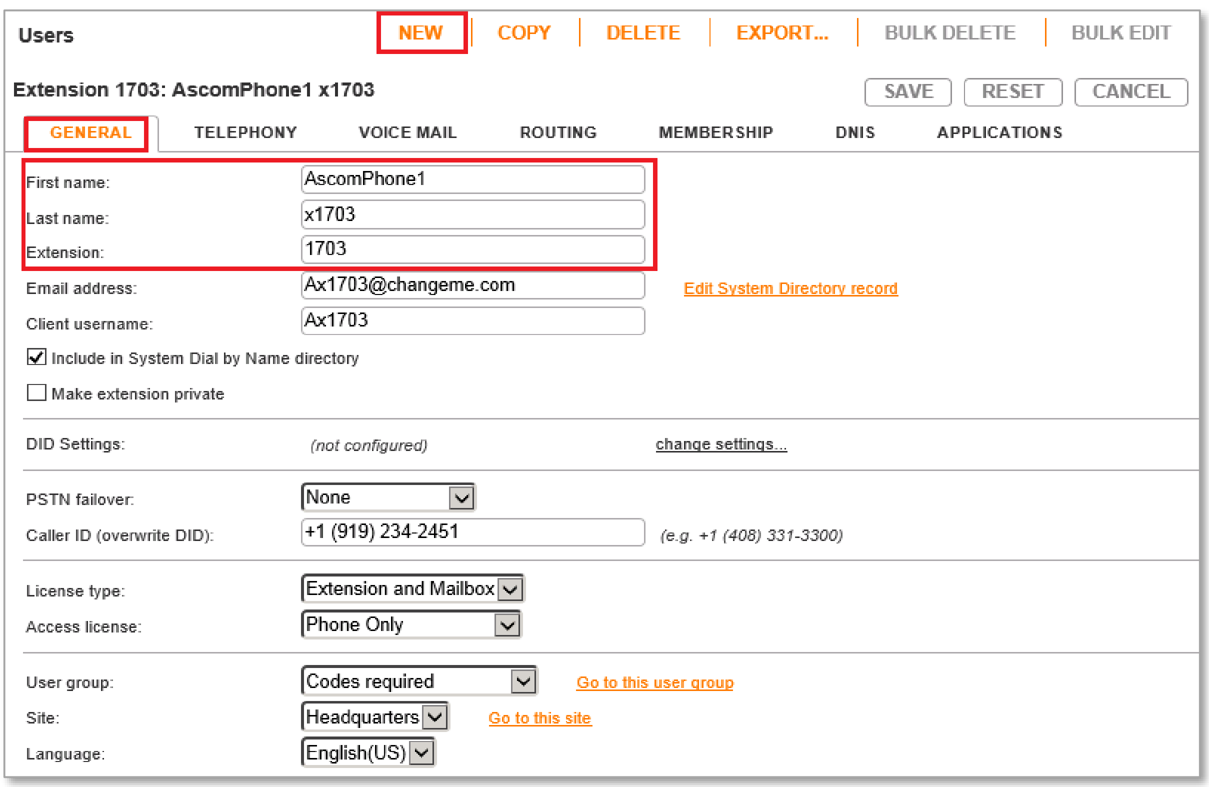
NOTE: Once the ShoreTel switch has been selected to support SIP Proxies, please note the IP Address of the switch as it will be used later in the configuration of the IP-DECT base station.

Configure a User as a SIP Extension

This section describes the steps required to configure a User to use the Ascom IP-DECT handsets as a SIP Extension.

1. Navigate to Administration > Users > Users
2. Click **New**, to create a new user
3. Define the **First name:** and **Last name:** Enter the appropriate user information
4. Define an **Extension:** ShoreTel Director will automatically assign the next available extension number, but it can also be modified to any available extension number
5. Define the **License type:** and **Access license:** In our example we chose “Extension and Mailbox”, although it is not necessary to have a mailbox with the Ascom IP-DECT handsets, and “Connect Client” for Access license

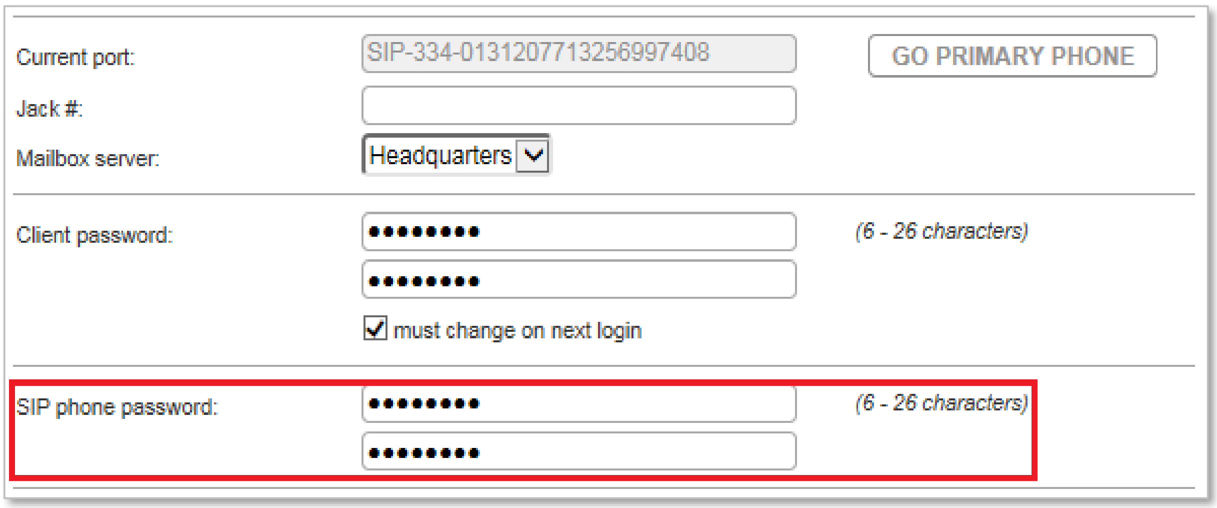
NOTE: If the “License type” is configured as “Extension-Only”, then “Any IP Phone” cannot be selected, but instead must be set to “SoftSwitch”.



6. Define a **SIP phone password:** There is no default SIP phone password configured, it is masked with the appearance that there is a default password, and must be defined by the ShoreTel Director Administrator. Make certain to type the password in both fields.

NOTE: Please note the "SIP phone password" configured for the user as it will be used later in the Ascom IP-DECT configuration.

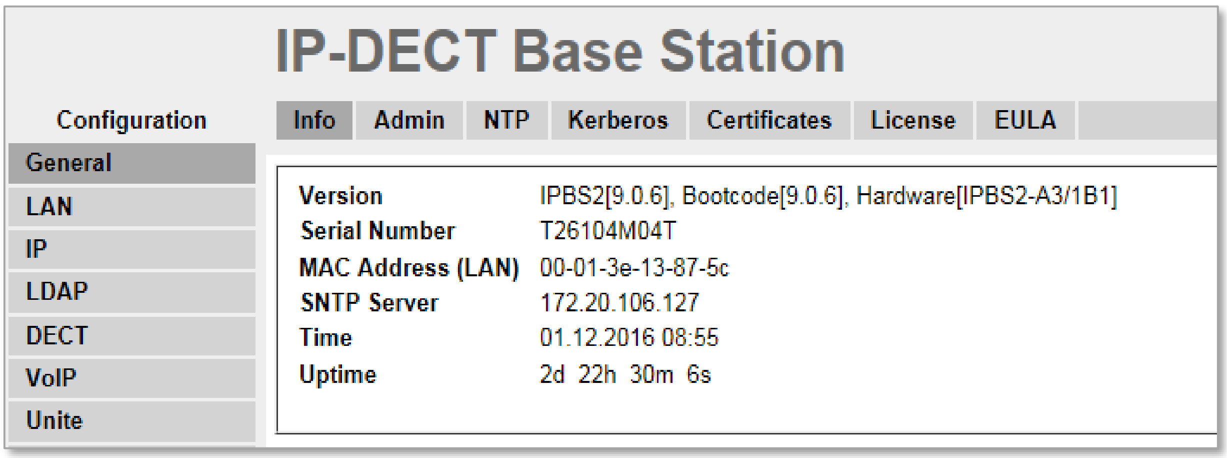
7. Click **SAVE**



Ascom Configuration

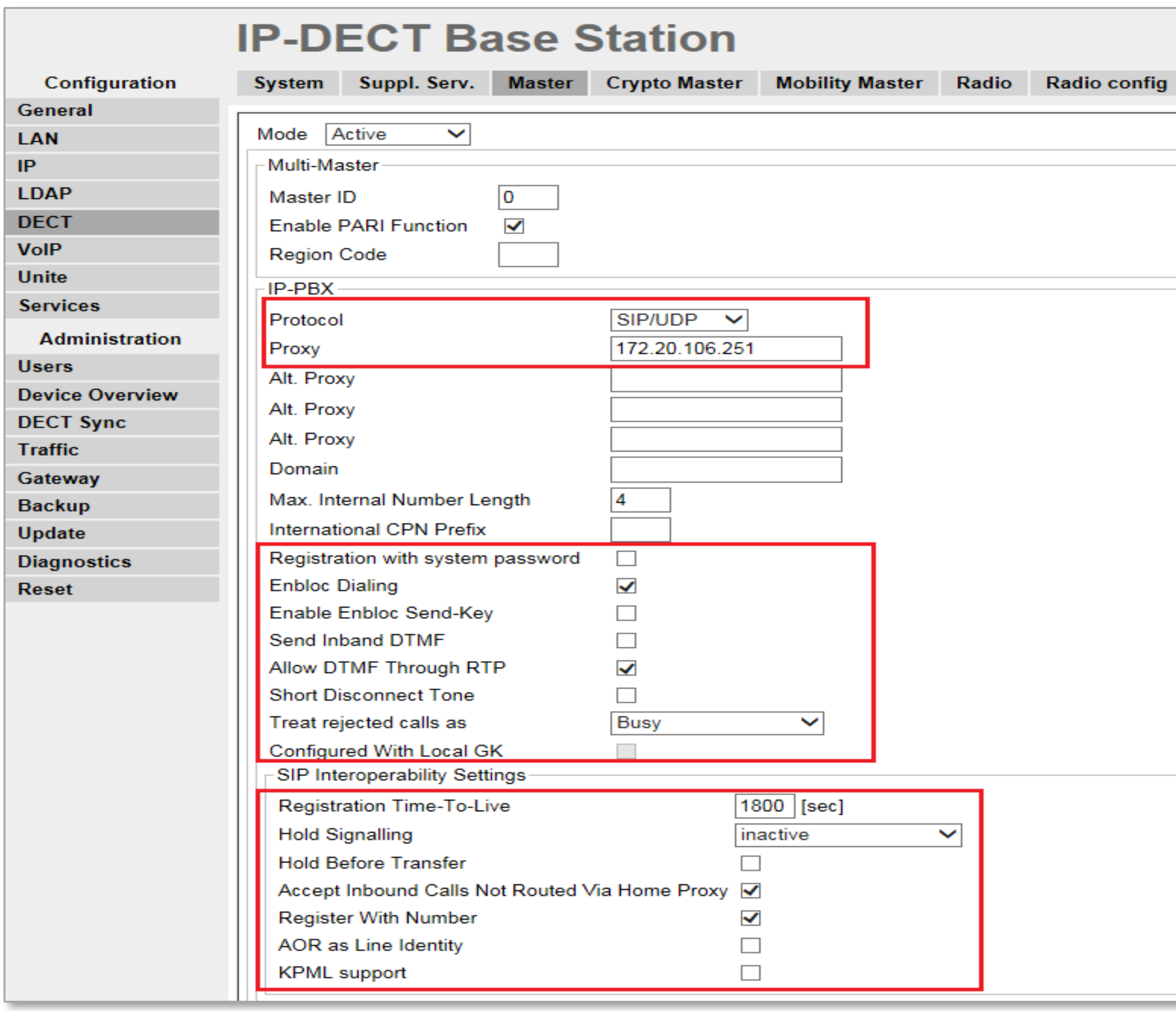
The following configuration steps detail the configuration process used to configure an Ascom IP-DECT Base Station in Master mode but the same steps are applicable also for the IP-DECT Gateway.

1. The user is presented with the General Info frame where the system information for the Ascom IP-DECT Base Station is displayed.



2. Navigate to the DECT Master frame by clicking DECT and then clicking Master.

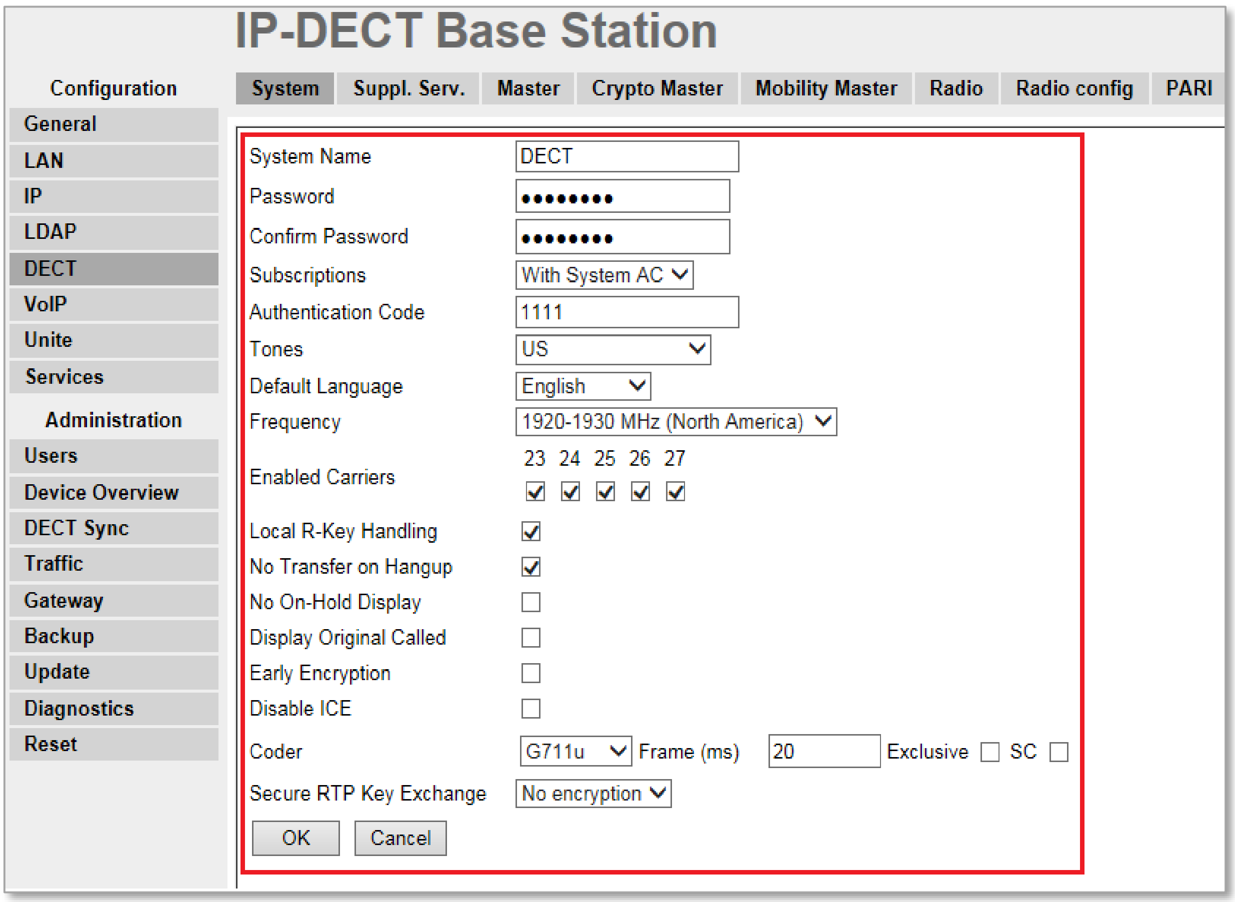
   Use the IP-PBX, Protocol drop-down list to set the protocol to “SIP”. The IP-PBX Proxy is set to the IP address of the ShoreGear Switch that you enabled SIP Proxy ports. Checking the Enbloc Dialing box will allow for post dialing. You should also enable (check) the following parameters: Allow DTMF through RTP, accept inbound calls not routed via home proxy and Register with number. We also recommend that you configure Registration time-to-live to a value of 1800.



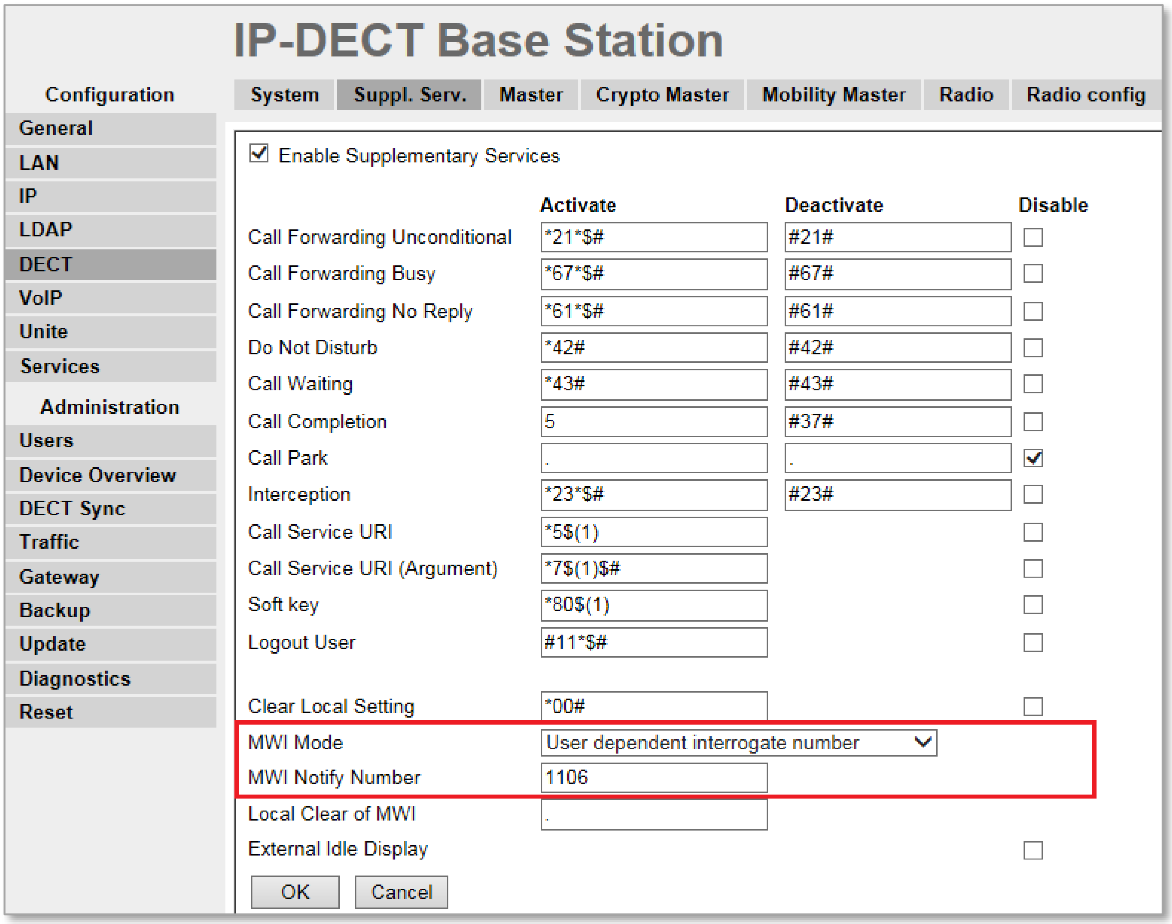
3. Navigate to the DECT System frame by clicking DECT and then clicking System.

   Select Local R-Key handling and No Transfer on Hangup.

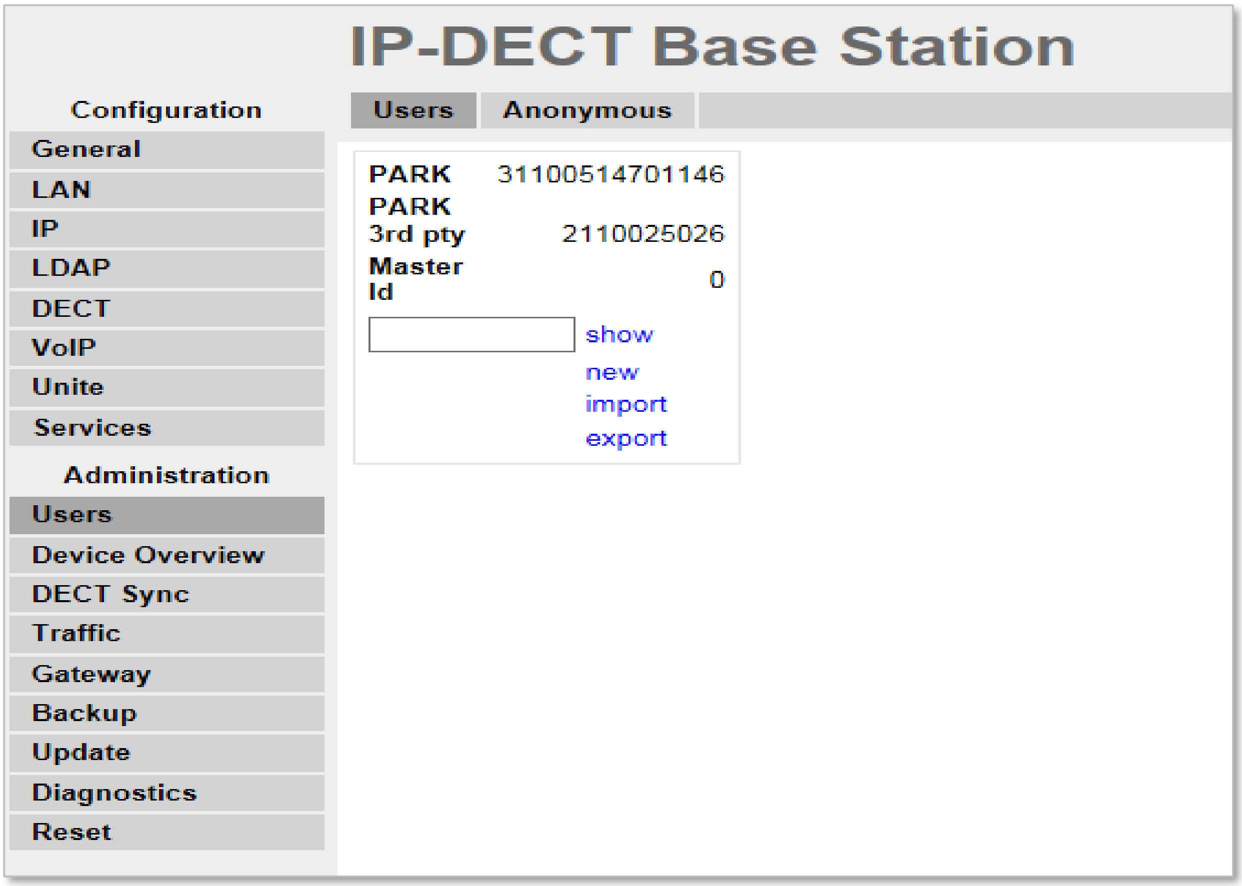
   Use the drop-down list for Tones and Frequency select your geographical region. The Use the drop-down list for Coder and select “G711u” and set Frame (ms) to 20.



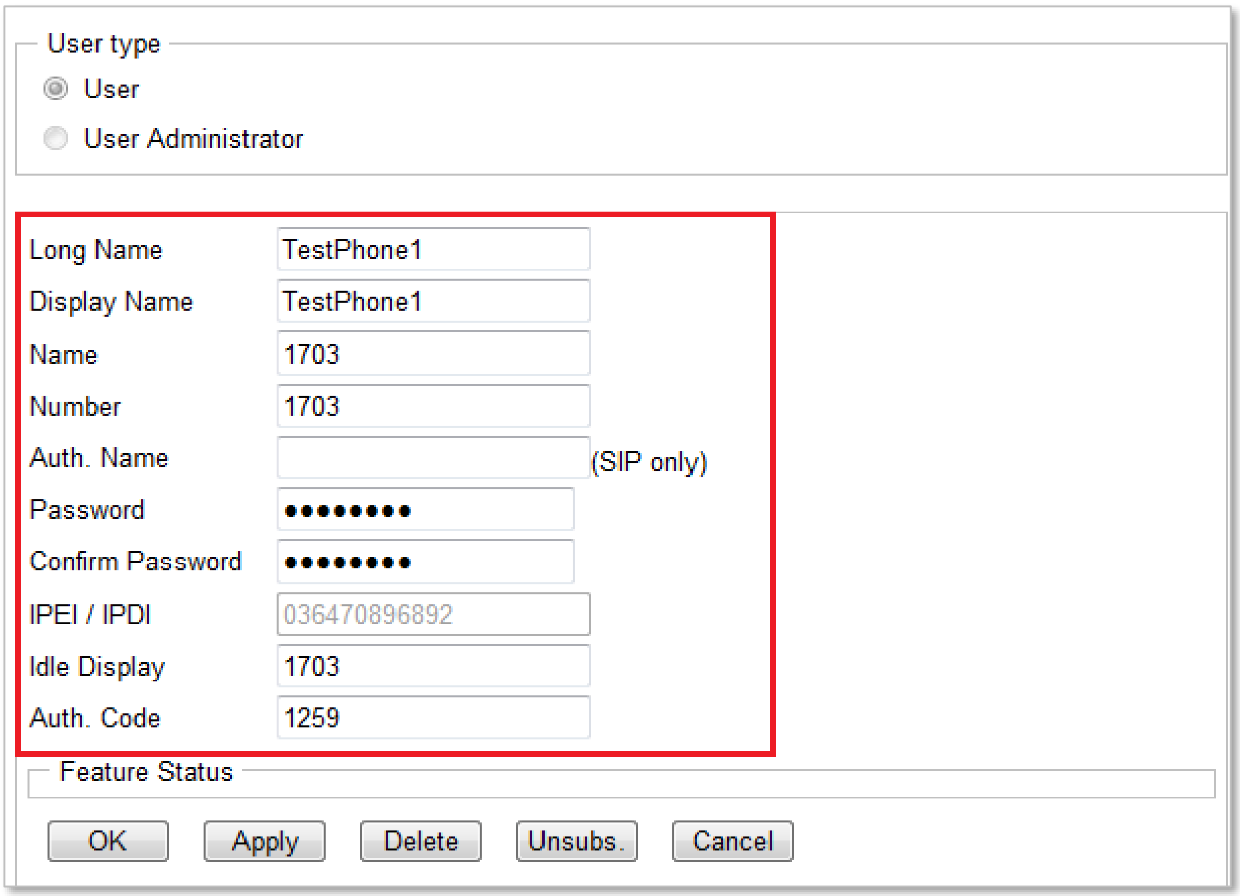
4. Navigate to the DECT Suppl. Serv. frame by clicking DECT and then clicking Suppl. Serv. Check the Enable Supplementary Services check box. Enter the extension used for Voice Mail in the MWI notify No. field. Click OK when finished.



5. Navigate to the Users frame by clicking Users and then clicking Users. Click new to provision a new user account. The PARK code is displayed. This value is needed when programming Ascom DECT handsets. The PARK code is similar to an SSID in an 802.11 wireless environment.



6. The user is presented with the Edit User web page. Long Name and Name can be any descriptive name that identifies this user. The Number field is the extension assigned to this user. The Password field is the password used to register with the ShoreTel IP-PBX. The box below Password is to confirm the password and the value entered for the Password field must be entered here. Display Text is the text string that will be displayed on the LCD screen of the Ascom DECT Handset. For additional information regarding IPEI, Auth Code and how to register the handset towards the base station please refer to Ascom's installation manual.



Summary of Tests and Results

N/S = Not Supported   N/T= Not Tested   N/A= Not Applicable

Basic Feature Test Cases

ID	Result	Name	Description	Notes
1.1	PASS	Device initialization with static IP address	Verify successful startup and initialization of the device up to a READY/IDLE state using a static IP address	
1.2	PASS	Device reset – idle (for static configurations)	Verify successful re-initialization of device after power loss while device is idle	
1.3	PASS	Device initialization with DHCP	Verify successful startup and initialization of the device up to a READY/IDLE state using DHCP	
1.4	PASS	Device reset – idle (for dynamic configurations)	Verify successful re-initialization of device after power loss while device is idle	
1.5	PASS	Verify Diffserv Code Point support	Verify the ability to set Diffserv Code Point from SIP DUT (device under test)	
1.6	PASS	Verify Date and Time Update support	Verify setting of Date and Time Update on SIP DUT	
1.7	PASS	Place call	Verify successful call placement with normal dialing to a variety of terminating phones	
1.8	PASS	Receive call	Verify successful call placement with normal dialing to a variety of terminating phones	
1.9	PASS	Place call - redial	Verify successful call placement using re-dial to SIP Reference	
1.10	PASS	Place call – speed dial	Verify successful call placement using programmed speed dial	

ID	Result	Name	Description	Notes
1.11	PASS	CODEC support (DUT to ShoreTel Phone)	Verify successful call connection and audio path using all supported CODECs (G.711-Ulaw and G.729)	
1.12	PASS	CODEC support (DUT to SIP reference)	Verify successful call connection and audio path using all supported CODECs (G.711-Ulaw and G.729)	
1.13	PASS	CODEC negotiation	Verify successful negotiation between devices configured with different default CODECs (G.711-Ulaw and G.729)	
1.14	PASS	Hold DUT to SIP reference	Verify successful hold and resume of connected call	
1.15	PASS	Hold DUT to ShoreTel	Verify successful hold and resume of connected call	
1.16	PASS	Forward	Verify successful forwarding of incoming calls	Local forward not supported
1.17	PASS	Forward from SIP DUT	Verify successful forwarding of incoming calls	
1.18	PASS	Mute	Verify device’s mute function	
1.19	PASS	Out-of-band DTMF Transmission	Verify successful transmission of out-of- band digits (RFC2833) for calls placed to and from the DUT with a variety of other devices	
1.20	PASS	Missed call notification	Verify that device notifies the user about missed calls	
1.21	PASS	Volume	Verify the device’s volume adjustment function	

ID	Result	Name	Description	Notes
2.1	PASS	Call waiting	Verify appropriate notification and successful connection of incoming call while busy with another party	
2.2	N/A	Park	Verify successful park and retrieval of connected call	
2.3	PASS	Extended forward	Verify extended call forwarding options – busy forwarding, ring no answer forwarding	Local forwarding not supported
2.4	PASS	Extended forward from SIP DUT	Verify extended call forwarding options – busy forwarding, ring no answer forwarding	Local forwarding not supported
2.5	PASS	Transfer – blind	Verify successful blind transfer of connected call	
2.6	PASS	Transfer – monitored	Verify successful monitored transfer of connected call	
2.7	N/A	Conference – ad hoc	Verify successful ad hoc conference of three parties	Not supported by Ascom
2.8	N/A	Place call – secondary line	Verify successful call placement using secondary line	Multiline not supported
2.9	N/A	Receive call – secondary line	Verify successful connection of incoming call on secondary line	Multiline not supported
2.10	PASS	Callback	Verify successful connection of a call using the missed- call callback feature of the device	
2.11	PASS	Headset	Verify the device's support for external headsets (using headsets supplied by the 3P phone vendor)	
2.12	PASS	Ring selection	Verify the device's ability to change the ring type	
2.13	PASS	Caller ID	Verify that Caller ID name and number is sent and received from SIP endpoint device	

ID	Result	Name	Description	Notes
2.14	PASS	SIP Device Generates Busy Tone	Verify that SIP DUT generates busy tone when calling a busy extension	
2.15	Not tested	POTS Analog Gateway supports the transfer operation by “flashing”	Verify that the POTS Analog Gateway can support the transfer operation by “flashing”	
2.16	NOT TESTED	911	Verify dialing “911” on DUT could connect with “911” services	
2.17	N/A	Fax Handling	Verify that fax can be sent and received through DUT	
2.18	PASS	Auto Attendant Menu	Verify that DUT can initate calls properly to a ShoreTel Auto Attendant menu and that you can transfer to the desired extension.	
2.19	PASS	Auto Attendant Menu “Dial by Name”	Verify that DUT can initiate calls properly to a ShoreTel Auto Attendant menu and that you can transfer to the desired extension using the “Dial by Name” feature.	
2.20	PASS	Auto Attendant Menu checking Voice Mail mailbox	Verify that DUT can initiate calls properly to a ShoreTel Auto Attendant menu and that you can transfer to the Voice Mail Login Extension.	
2.21	PASS	Initiate call to a Hunt Group	Initiate a call from DUT and verify that calls route to the proper Hunt Group and are answered by an available hunt group member with audio in both directions using G.729 and G.711 codecs.	

ID	Result	Name	Description	Notes
2.22	PASS	Initiate call to a Workgroup	Initiate a call from DUT and verify that calls route to the proper Workgroup and are answered successfully by an available workgroup agent with audio in both directions using G.729 and G.711 codecs.	
2.23	PASS	Hunt Group Member	Verify that incoming calls to a hunt group can be answered properly when DUT is a member of the hunt group.	
2.24	PASS	Workgroup Agent	Verify that incoming calls to a workgroup can be answered properly when DUT is an agent of the workgroup.	
2.25	PASS	Call Forward – “FindMe”	Verify that calls are forwarded to DUT’s “FindMe” destination. Verify that DUT works properly when it’s a “FindMe” destination	
2.26	NOT TESTED	ShoreTel Converged Conferencing Server	Verify that calls are properly forwarded to the ShoreTel Converged Conferencing Server and it properly accepts the access code and you’re able to participate in the conference.	
2.27	PASS	Bridged Call Appearance (BCA) extension	Verify that DUT can initiate calls properly to a BCA extension and the call is presented to all of the phones that have BCA configured. Verify that the call can be answered, placed on-hold and then transferred.	
2.28	PASS	Additional Phones (Simulring)	Verify that calls ring simultaneously on DUT and ShoreTel IP Phone	

Conclusion

Ascom IP-DECT was successfully validated and approved with ShoreTel Connect ONSITE.

Additional Resources

[ShoreTel Connect ONSITE System Administration Guide]

[ShoreTel Connect ONSITE Planning and Installation Guide]

Version	Date	Contributor	Content
1.0	November 2016	J.Rodriguez	Original App Note
1.1	January 2017	K.Magnus Olsson	Feedback Incorporated

ShoreTel. Brilliantly simple business communications.

ShoreTel, Inc. (NASDAQ: SHOR) is a leading provider of brilliantly simple IP phone systems and unified communications solutions powering today's always-on workforce. Its flexible communications solutions for on-premises, cloud and hybrid environments eliminate complexity, reduce costs and improve productivity.

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