# SHORETEL APPLICATION NOTE

for

Ascom i62

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|---|---|
| Date: | Dec 12, 2016 |
| App Note Number: | TC-16084 |
| For use with: | Ascom i62, version 5.5.0 |
| Product: | ShoreTel Connect ONSITE |
| System: | ST Connect 21.79.9330.0 |

# Contents

# Introduction

This document describes the configuration procedures for integrating the Ascom i62Handsets as SIP extensions on the ShoreTel Connect Onsite system.

**Ascom**

The Ascom i62 offers a high class telephony, messaging and alarm solution for enterprise business based on the WiFi technology. With offering Voice Over WiFi, only one network is needed to be installed and maintained for all applications running, such as Internet access, e-mail, voice and other business related applications.

The latest 802.11n standard provides the benefits of higher throughput and longer range possibilities which will increase the ability to integrate to other systems and build efficient applications. With the new generation networks and handsets, the capacity and versatility outperforms any other on-site wireless technology.

The Ascom i62 offers a unique management tool with central management concept enabling remote management and SW upgrades of the handsets over the air.

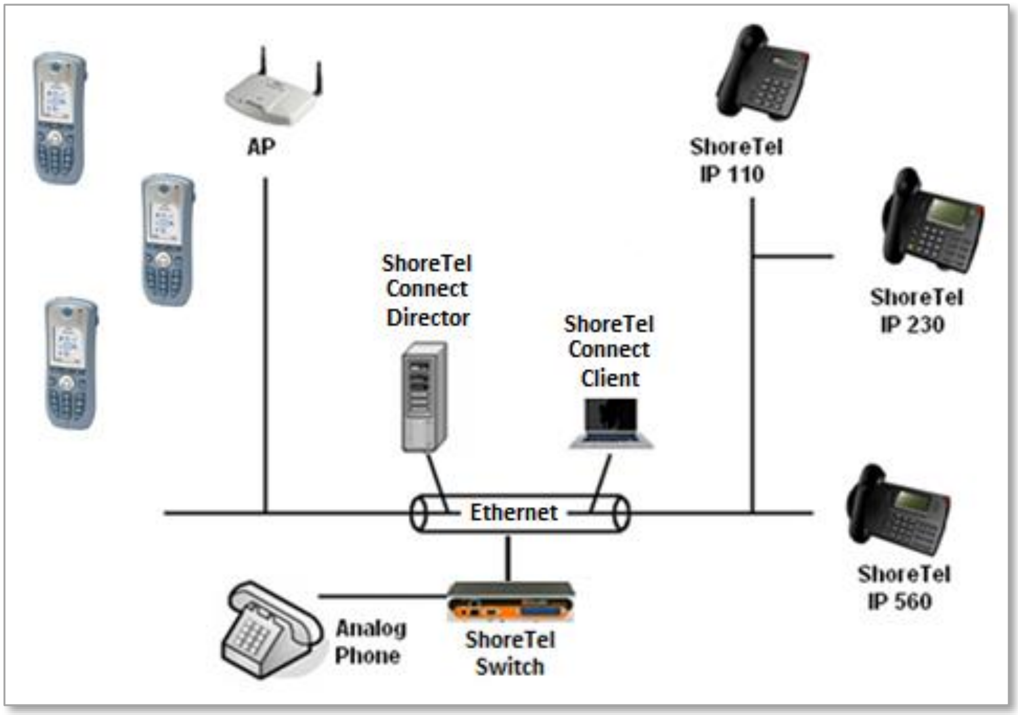
### *Features*

| Key features \ Handset/Licence | i62 Talker | i62 Messenger | i62 Protector |
|---|---|---|---|
| IP44 and possible to disinfect, perfectly suited for healthcare | √ | √ | √ |
| Location capabilities | √ | √ | √ |
| Loud-speaking function | √ | √ | √ |
| Standard headset connector | √ | √ | √ |
| Administrate all handsets centrally over-the-air, no need to collect all handsets for configurations or updates | √ | √ | √ |
| Central phone-book support, always have an up-to-date phone book of all employees and customer contacts | √ | √ | √ |
| Message receipt during active call | | √ | √ |
| Large font option in messages | | √ | √ |
| Remote control functions, e.g. open doors, set process values or ask for medical data | | √ | √ |
| Push-to-talk, PTT, functionality to quickly set up group calls | | √ | √ |
| Color-coded messages | | √ | √ |
| Receive messages with acknowledge and reject options | | √ | √ |
| Ascom Interactive Messaging - receive interactive message with several answer options | | √ | √ |
| Activated alarm button with two different alarm types | | | √ |
| Man-down / no-movement alarm | | | √ |
| Several alarm customization possibilities | | | √ |

### *Technical Support*

Email: support@ascom.com
Help desk: +46 31 559450

## Network Topology




**Test Environment**

- ShoreTel Connect ONSITE Server
- ShoreTel Virtual Phone Switch
- ShoreGear Switch
- ShoreTel IP Phones
- Ascom i62 (5.5.0)

*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 i62 handsets with the ShoreTel Connect ONSITE system.

#### ShoreTel Extension License

Extension Licenses are required for each Ascom i62 user.

#### ShoreTel SIP Phone License

Deployment of SIP Extensions require a SIP Phone License. One SIP Phone License is required for each Ascom i62 SIP Extension.

#### Call Forwarding

When call forwarding is configured on the Ascom i62 handsets, the Ascom i62 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.*

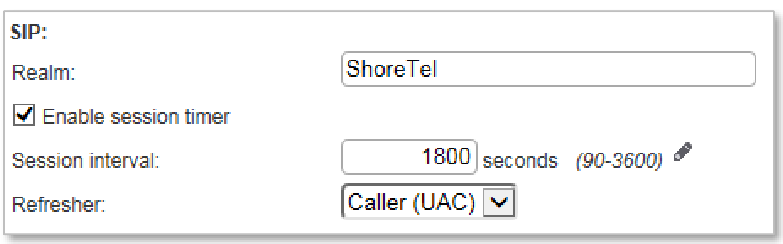
## ShoreTel Configuration

This section describes the detailed steps required on the ShoreTel Connect ONSITE system to configure the Ascom i62 handsets as a SIP extension.

**Call Control Options**

This section describes the SIP settings required on the ShoreTel system to work with Ascom i62 handsets.

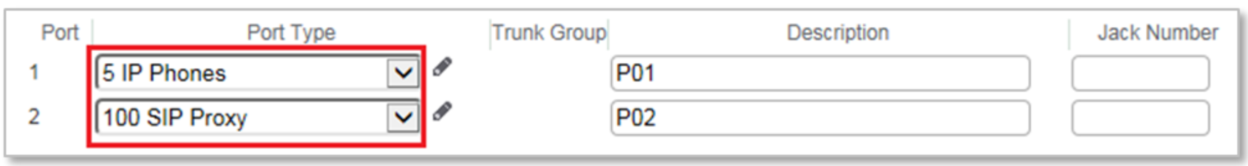
1. Navigate to Administration > Features > Call Control > Options
2. Verify the parameters located under the **SIP** section
3. **Realm**: The realm is used in authenticating all SIP devices. Changing this value will require a reboot of switches serving as SIP extensions. It is not necessary to modify 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**



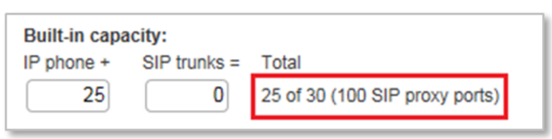
### SIP Proxy Settings – Allocating Ports for SIP Extensions

This section describes the Switch configuration required on the ShoreTel system to work with Ascom i62 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 i62 users. The following example shows Port allocation designated on a ShoreTel SG-90 for IP Phones and SIP Proxy resources



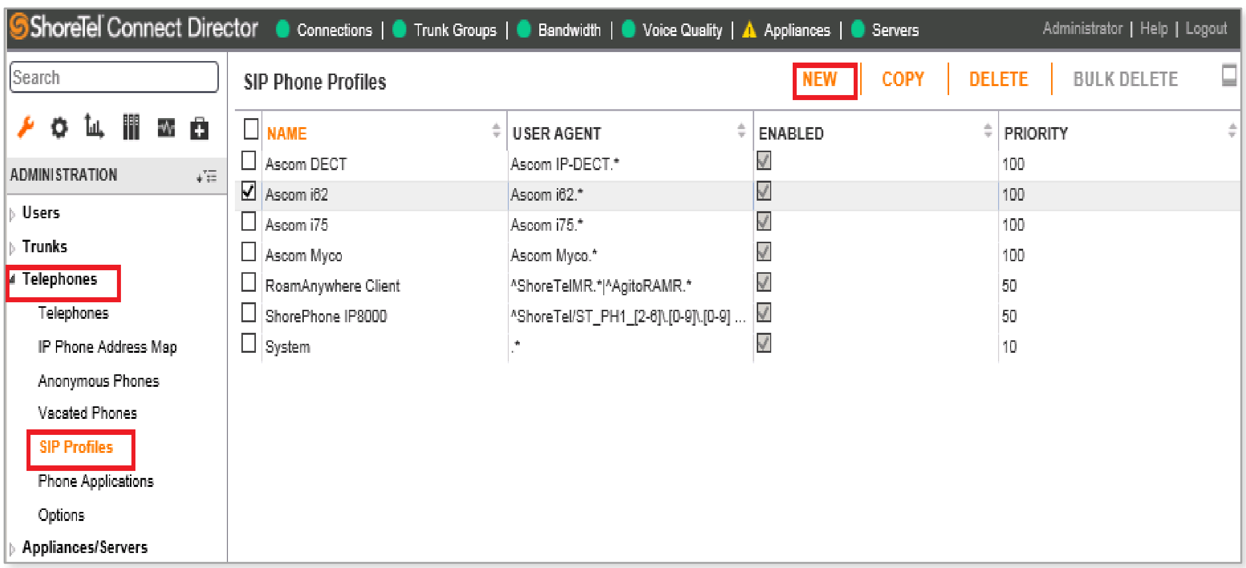

If the ShoreTel ShoreGear Switch that you have selected has “built-in” capacity (i.e., ShoreGear 50/90/220T1/E1, etc.) for IP phones and SIP trunks, you can also remove 5 ports from the total number available to provide the “100 SIP Proxy” configuration necessary. Every 5 ports you remove from the total 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 i62 VoWiFi handsets 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.

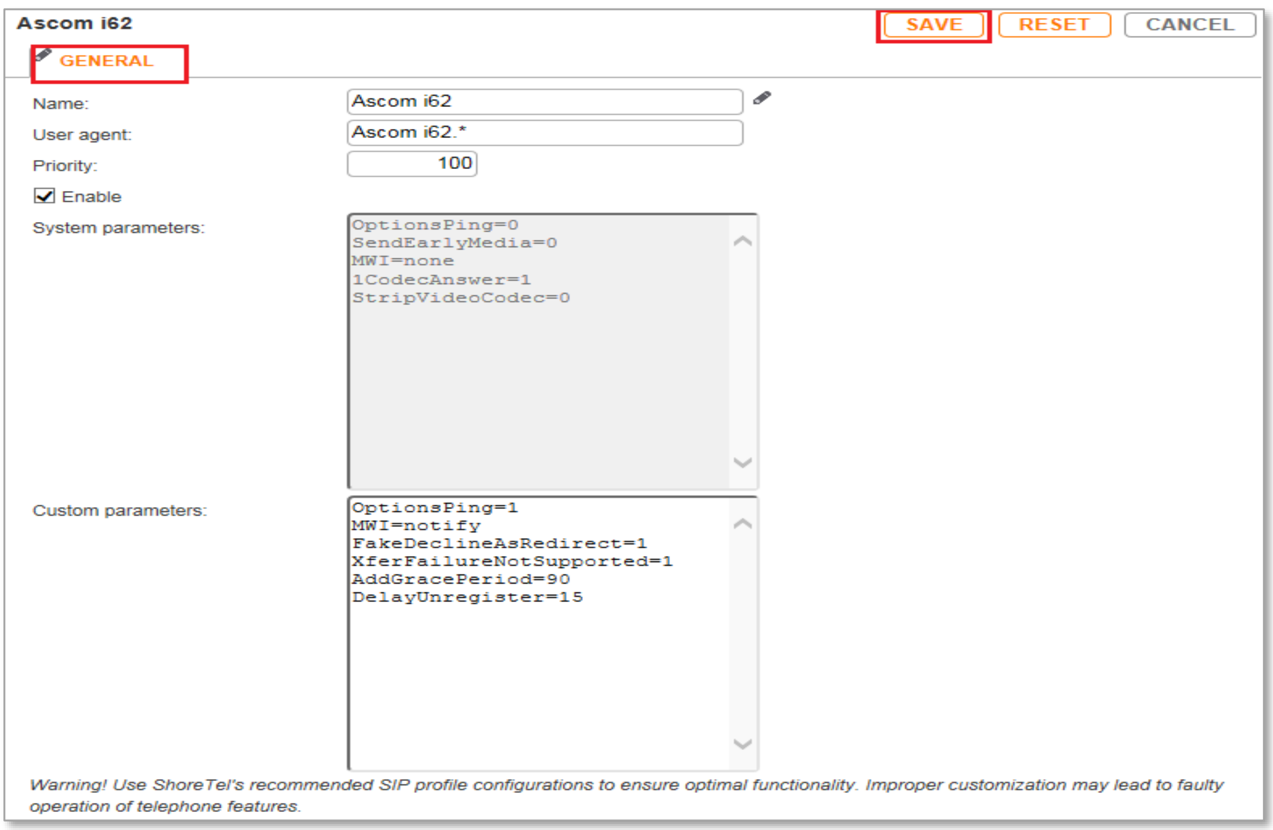
1. Navigate to Administration > Telephones > SIP Profiles
2. Click **New**, to create a new SIP Profile



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 i62.*” (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=1
XferFailureNotSupported=1
AddGracePeriod=90
DelayUnregister=15

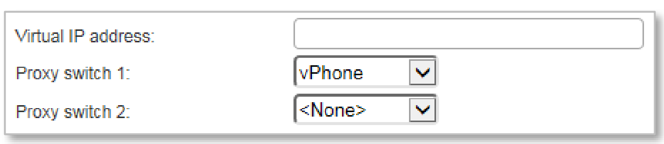
8. Click **SAVE**



**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**



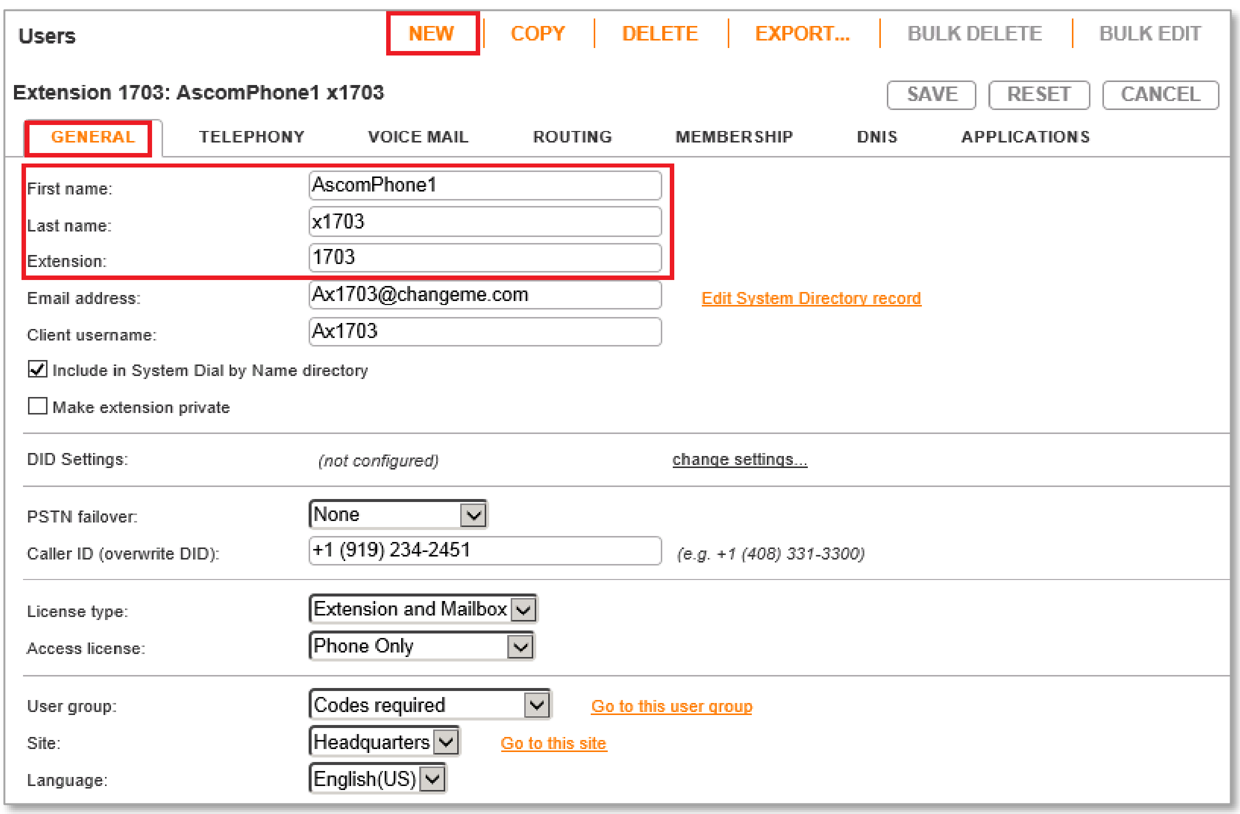

*NOTE: Once the ShoreTel switch has been selected to support SIP Proxies, please note the IP Address of the switch as it will be used later in the Ascom i62 Device Manager under the VoIP/SIP configuration.*

### Configure a User as a SIP Extension

This section describes the steps required to configure a User to use the Ascom i62 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 Connect 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 i62 handsets, and “Connect Client” for Access license

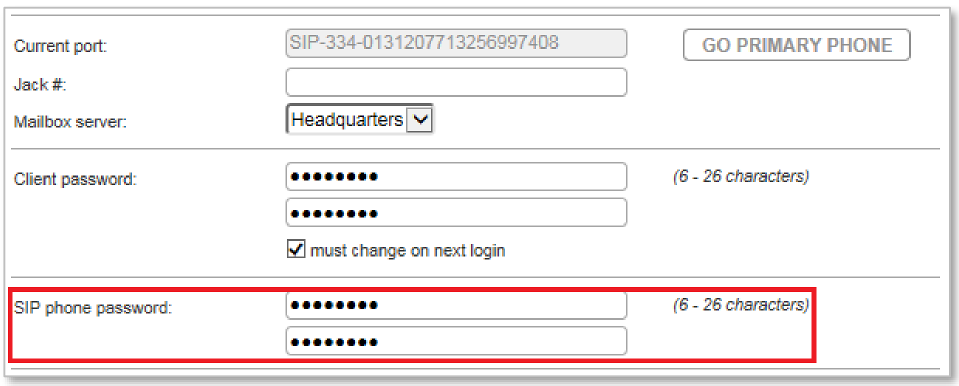
*NOTE: If the “License type” is configured as “Extension-Only”, then “Any IP Phone” cannot be selected, but instead must be set to “SoftSwitch”.*



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 i62 Device Manager under the VoIP/SIP configuration.*

7. Click **SAVE**



# Ascom Configuration

The following steps detail the configuration process for the Ascom i62 VoWiFi handset using the Device Manager.

1. Navigate to the "System -> A" configuration page by clicking System and then A. Configure the following parameters. These settings should be repeated for each Ascom i62 VoWiFi handset being provisioned. The ESSID field value must match the ESSID value specified in the AP.

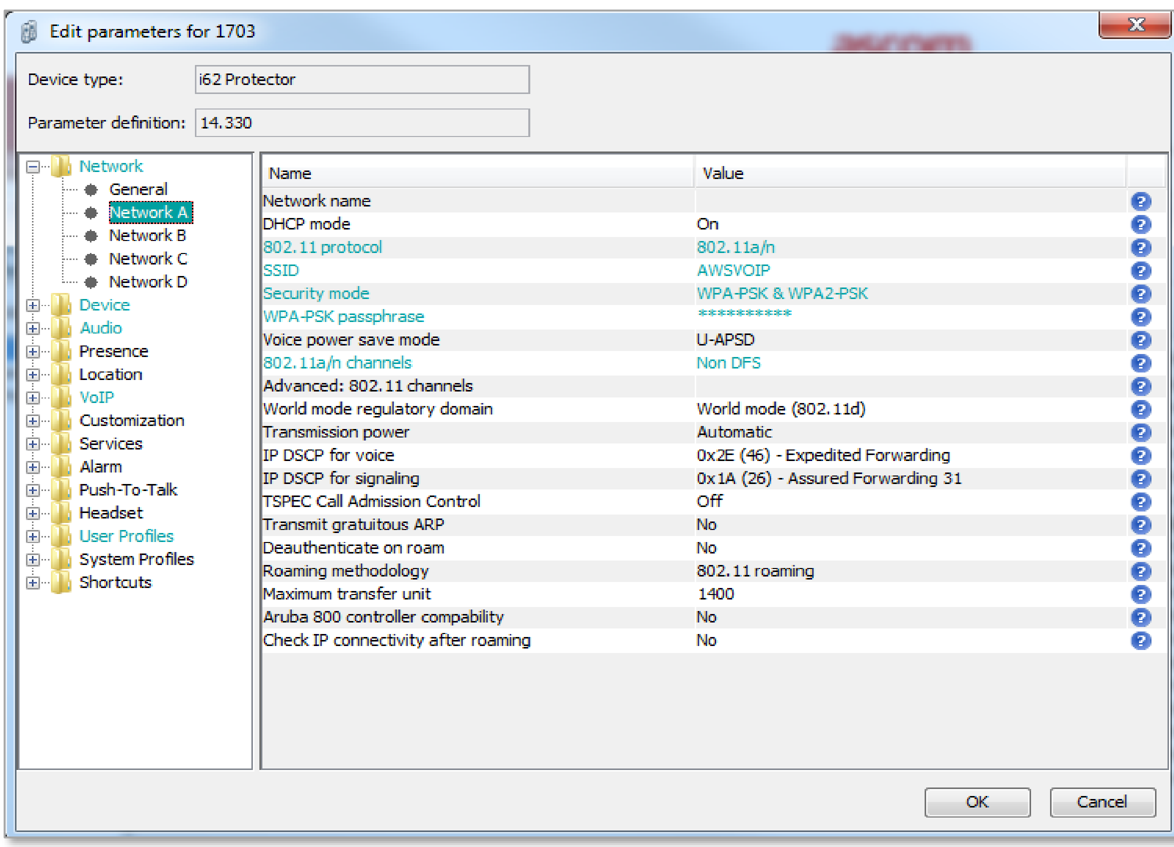
   Note: Below is a typical configuration utilizing. Different Security modes might be used.

   SSID: AWSVOIP

   Security mode: WPA2-PSK

   IP DSCP for voice "0x2e (46) – Expedited Forwarding"

   IP DSCP for signaling "0x1A (26) – Assured Forwarding 31".



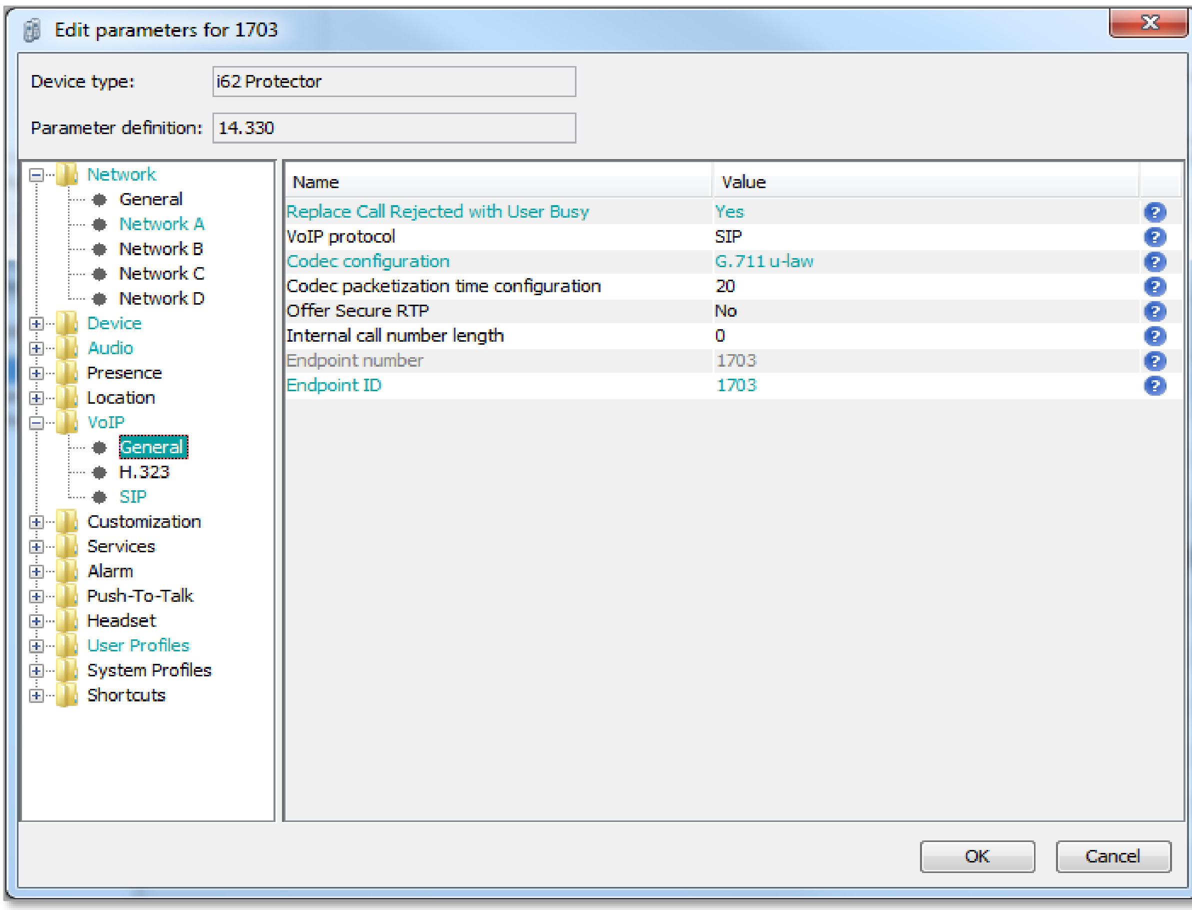
2. Navigate to the "VoIP/General" configuration page by clicking VoIP and then General. Configure the following parameters.

   Replace Call Rejected with User Busy: Enable. If this value is not set correctly, certain calling features such as transfer will not operate properly.

   VoIP protocol "SIP"

   Codec configuration "G.711 u-law"

   Endpoint number – This is the extension associated with the Ascom i62 VoWiFi handset being provisioned. This setting should be repeated for each Ascom i62 VoWiFi handset being provisioned.



3. Navigate to the “VoIP / SIP” configuration page by clicking VoIP and then SIP. Configure the following information and then click OK. The SIP proxy password field must match the Media Server Extension password configured on ShoreTel IP-PBX. Once the information has been configured, the PDM reports the information as ***********. After clicking OK, pick up the telephone from the PDM cradle in order to reboot the handset and activate the new configuration.

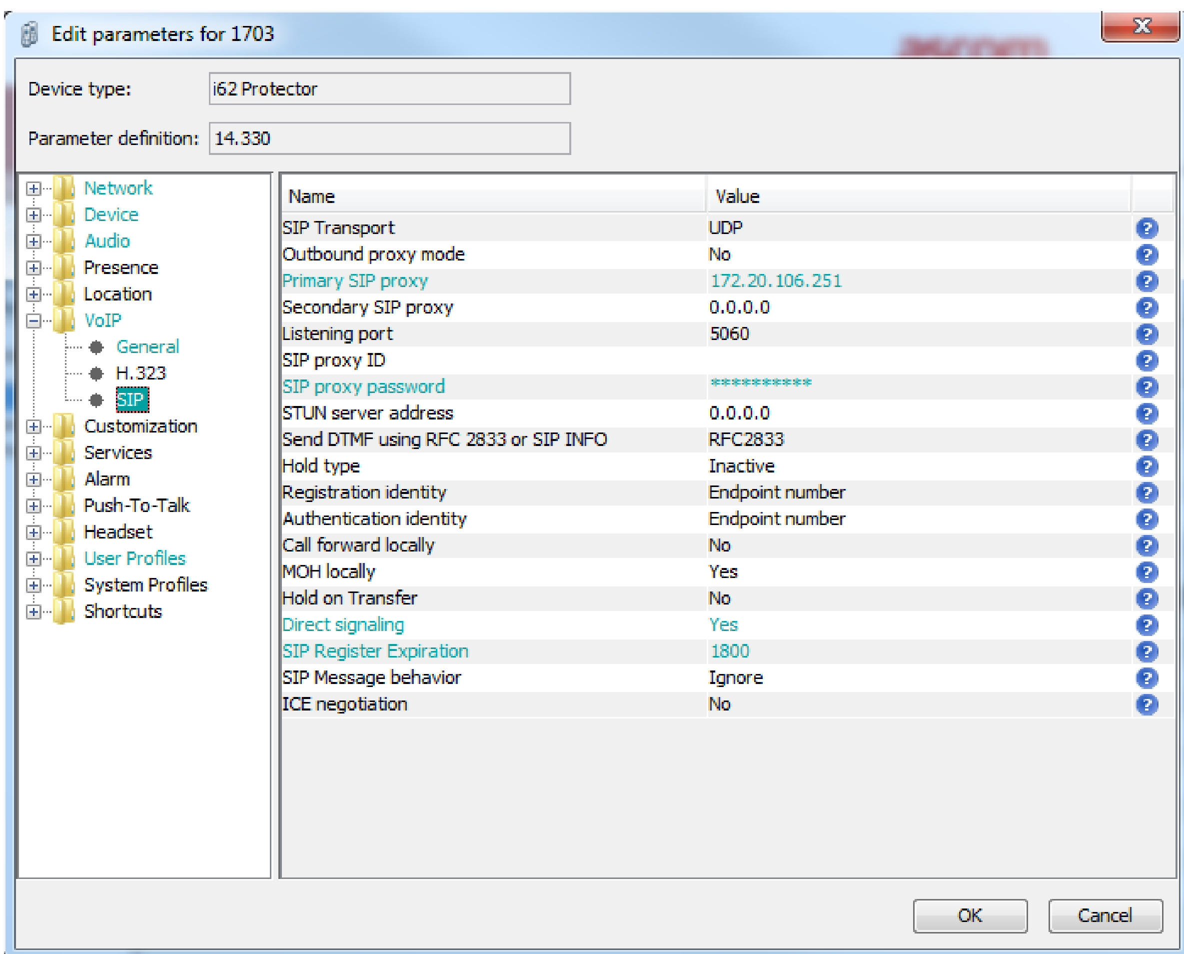
   The following screen shot shows:

   SIP proxy IP address: “172.20.106.251”

   SIP proxy password: <Set Password>

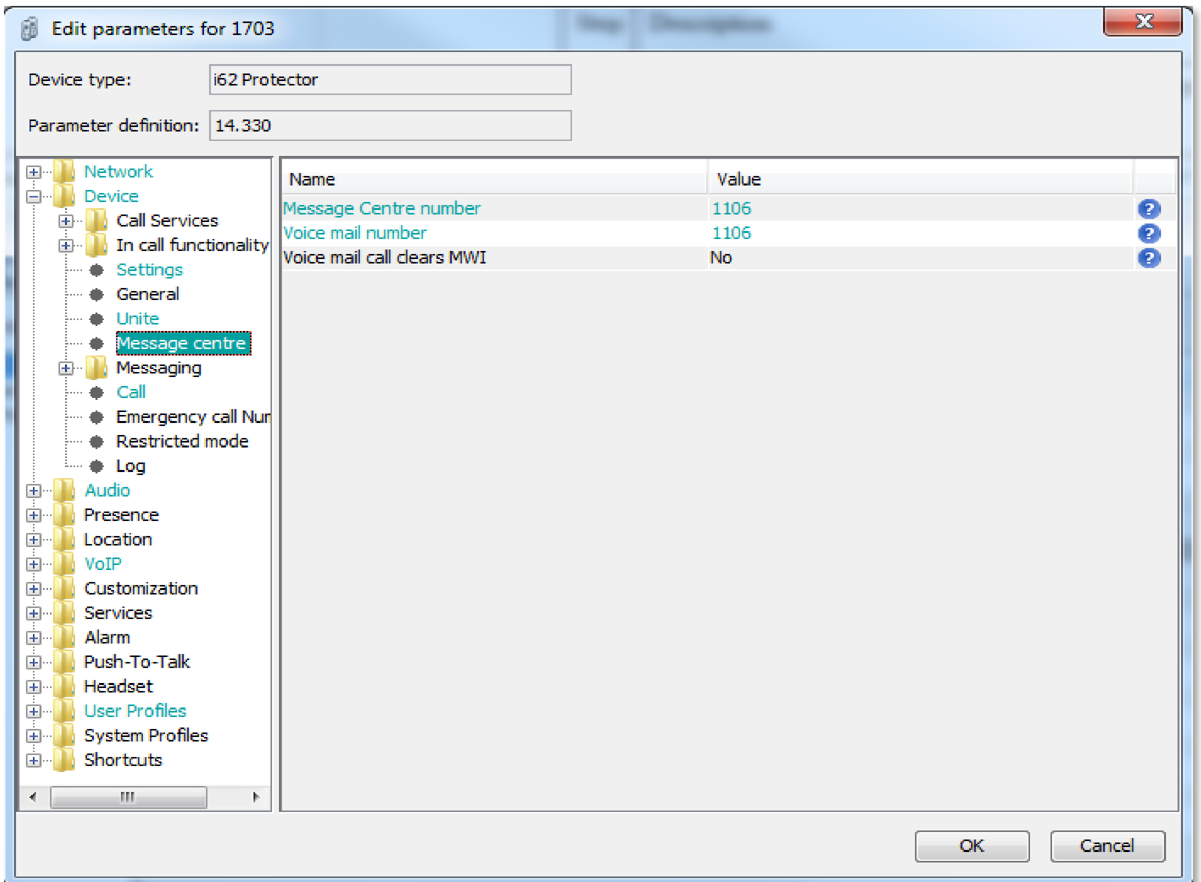
   Direct signaling: Enabled

   SIP Registration Expiration: 1800



4. Navigate to the “Device -> Message Centre” configuration page by clicking Device and then Message Centre.

   Enter the number to the Voice Mail at both Message centre number and at Voice mail number. Voice mail number will speed dial the specified VM number when long pressing button no 1.



# 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 | PASS | Conference – ad hoc | Verify successful ad hoc conference of three parties | |
| 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 | Replace Call Rejected with User Busy needs to be enabled in i62 |
| 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 i62 handsets were 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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