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|---|---|--------------------------------|
|  |  | Innovation Network App Note |
| | | TPP: 10104 Date: June, 2014 |
| Product: ShoreTel Ascom i62 | | System version: ShoreTel 14.2 |

Abstract

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.

Combining ShoreTel® IP-PBX together with Ascom i62 VoWiFi handsets allows our customers the opportunity to utilize ShoreTel's unique distributed call control architecture and Ascom's rugged, feature-rich wireless i62 handsets.

Table of Contents

| | | | |
|--|-------------------------------------|--|----|
| Overview | 2 | Configuration Overview | 9 |
| Features and Benefits | Error! Bookmark not defined. | ShoreTel Configuration..... | 9 |
| Ascom Overview and Contact..... | 2 | Ascom Configuration..... | 19 |
| Ascom Product Information | 3 | Ascom Troubleshooting..... | 26 |
| Architecture Overview..... | 4 | Ascom Technical Support..... | 26 |
| Requirements, Certification and Limitations..... | 5 | ShoreTel Technical Support | 26 |
| Version Support..... | 5 | Document and Software Copyrights | 27 |
| Table 1: Basic Test Cases..... | 6 | Trademarks | 27 |
| Table 2: Extended Feature Test Cases..... | 7 | Disclaimer..... | 27 |
| Remarks | 8 | Company Information..... | 27 |

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.

Overview

This Application Note describes the configuration process necessary to provide interoperability between ShoreTel Unified Communications solution and Ascom wireless i62 VoWiFi Session Initiation Protocol (SIP) handsets. Specific calling features tested and verified to operate correctly include attended/unattended transfer, conference call participation, conference call add/drop, conference call creation, multiple call appearances, caller ID operation, call forwarding unconditional, call forwarding on busy, call forwarding clear, pick groups, call pickup, bridged appearances, voicemail, MWI, hold and return from hold.

Ascom Overview and Contact

Sales support for the Ascom i62 VoWiFi handset can be obtained through the following:

For local US/Canada:

- Phone: 1-877-71ASCOM or 1-877-712-7266
- Internet: <http://www.ascom.us/us-en/index-us/products-solutions/sales-us.htm> (for your Regional Sales Director)
- Email: techsupport@ascomwireless.com (for Technical support)

For international customers:

- Internet: www.ascom.com/ws and select your country of interest, to find local sales and support contact information.



Ascom Product Information

The Ascom i62 is available in three versions based on license, i62 Talker, i62 Messenger and i62 Protector.



| Handset/Licence | i62 Talker | i62 Messenger | i62 Protector |
|---|------------|---------------|---------------|
| Key features | | | |
| 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 | | | ✓ |

Architecture Overview

The network diagram shown below illustrates the testing environment used for compliance testing. The network consists of: a ShoreTel ShoreWare® Director, a ShoreTel Personal Call Manager, three different models of ShoreTel IP telephones (IP110, IP230, and IP560), three Ascom wireless i62 handsets, one non wireless non IP telephone, and a wireless network infrastructure providing network services such as Dynamic Host Configuration Protocol (DHCP), Trivial File Transfer Protocol (TFTP) and an access point (AP).

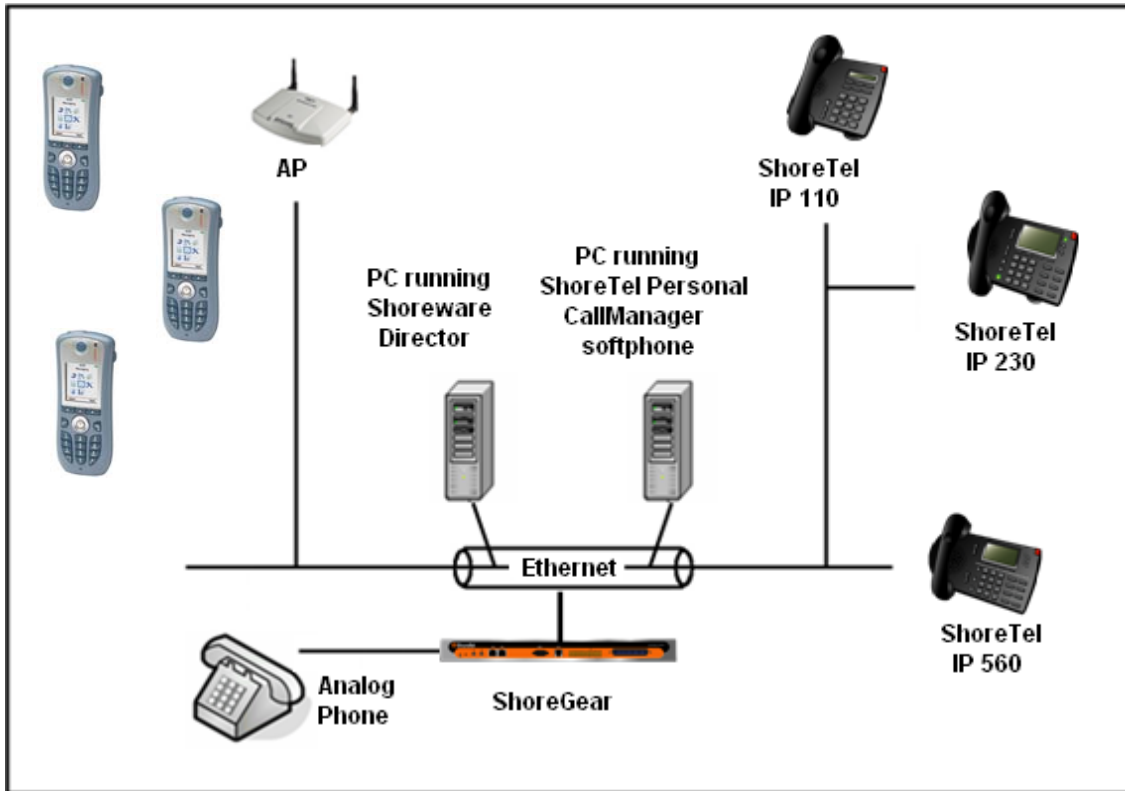


Figure 1 – Testing Environment

Requirements, Certification and Limitations

Ascom Portable Device Manager (PDM) requirements include:

- PC with Windows XP® Professional SP2 or Windows Vista® Business operating system
- Sun Java Runtime Environment (JRE) 6 or higher
- Microsoft Internet Explorer 6.0™ (IE6) or higher
- USB port (USB 1.1 required, USB 2.0 supported)
- Acrobat Reader 4.0 or higher

NOTE: Deployment of Ascom i62 handsets require ShoreTel SIP Phone License(s) (one per Ascom i62 handset) as well as the either the Extension & Mailbox License OR the Extension Only License.

Version Support

| | | Ascom i62 VoWiFi handset |
|------------------|----------------------------|--------------------------|
| | | 5.2.8 |
| ShoreTel Release | 14.2 Build 19.42.2005.0 | ✓ |



Certification Testing Results Summary

Table 1: Basic Test Cases

| ID | Name | Description | Results |
|------|--|---|----------------------|
| 1.1 | 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 | Pass |
| 1.2 | Device reset – idle (for static configurations) | Verify successful re-initialization of device after power loss while device is idle | Pass |
| 1.3 | Device initialization with DHCP | Verify successful startup and initialization of the device up to a READY/IDLE state using DHCP | Pass |
| 1.4 | Device reset – idle (for dynamic configurations) | Verify successful re-initialization of device after power loss while device is idle | Pass |
| 1.5 | Verify Diffserv Code Point support | Verify the ability to set Diffserv Code Point from SIP DUT (device under test) | Not Tested |
| 1.6 | Verify Date and Time Update support | Verify setting of Date and Time Update on SIP DUT | Pass |
| 1.7 | Place call | Verify successful call placement with normal dialing to a variety of terminating phones | Pass |
| 1.8 | Receive call | Verify successful call placement with normal dialing to a variety of terminating phones | Pass |
| 1.9 | CODEC support (DUT to ShoreTel Phone) | Verify successful call connection and audio path using all supported CODECs (G.711-Ulaw and G.729) | Pass |
| 1.10 | CODEC support (DUT to SIP reference) | Verify successful call connection and audio path using all supported CODECs (G.711-Ulaw and G.729) | Pass |
| 1.11 | CODEC negotiation | Verify successful negotiation between devices configured with different default CODECs (G.711-Ulaw and G.729) | Pass |
| 1.12 | Hold DUT to SIP reference | Verify successful hold and resume of connected call | Pass |
| 1.13 | Hold DUT to ShoreTel | Verify successful hold and resume of connected call | Pass |
| 1.14 | Forward | Verify successful forwarding of incoming calls | Pass * |
| 1.15 | Forward from SIP DUT | Verify successful forwarding of incoming calls | Pass * |
| 1.16 | Dual-tone multi-frequency (DTMF) transmission | Verify successful transmission of in-band and out-of-band digits (RFC2833) for calls placed to and from the DUT with a variety of other devices | Pass RFC2833 only |



Table 2: Extended Feature Test Cases

| ID | Name | Description | Notes |
|------|---|--|------------|
| 2.1 | Call waiting | Verify appropriate notification and successful connection of incoming call while busy with another party | Pass |
| 2.2 | Park | Verify successful park and retrieval of connected call | Pass |
| 2.3 | Transfer – blind | Verify successful blind transfer of connected call | Pass |
| 2.4 | Transfer – monitored | Verify successful monitored transfer of connected call | Pass |
| 2.5 | Conference – ad hoc | Verify successful ad hoc conference of three parties | Pass |
| 2.6 | Caller ID | Verify that Caller ID name and number is sent and received from SIP endpoint device | Pass ** |
| 2.7 | 911 | Verify dialing “911” on DUT could connect with “911” services | Not Tested |
| 2.8 | Auto Attendant Menu | Verify that calls are properly terminated on the ShoreTel Auto Attendant menu and that you can transfer to the desired extension. | Pass |
| 2.9 | Auto Attendant Menu “Dial by Name” | Verify that calls are properly terminated on the ShoreTel Auto Attendant menu and that you can transfer to the desired extension using the “Dial by Name” feature. | Pass |
| 2.10 | Auto Attendant Menu checking Voice Mail mailbox | Verify that calls are properly terminated on the ShoreTel Auto Attendant menu and that you can transfer to the Voice Mail Login Extension. | Pass |
| 2.11 | 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. | Pass |
| 2.12 | 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. | Pass |
| 2.13 | Hunt Group Member | Verify that incoming calls to a hunt group can be answered properly when DUT is a member of the hunt group. | Pass |
| 2.14 | Workgroup Agent | Verify that incoming calls to a workgroup can be answered properly when DUT is an agent of the workgroup. | Pass |
| 2.15 | Call Forward – “FindMe” | Verify that calls are forwarded to DUT’s “FindMe” destination. | Pass |
| 2.16 | 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 bridge. | Pass |
| 2.17 | Bridged Call Appearance (BCA) extension | Verify that calls are properly presented to all of the phones that have BCA configured and that the call can be answered, placed on-hold and then transferred. | Pass |

*) Call forwarding was configured from Shoretel GUI (Web Client). Local Call Forward not possible as ShoreTel does not allow 3rd party devices to redirect calls.

**) Caller initiating a call will see only called party’s number and not name. Called party will display the callers name and number. This applies to internal calls.



Remarks

- Call forwarding has to be done via the ShoreTel user interface through the call handling modes.
- The de-registration (Expire=0) that was sent prior to registration after lost WLAN connection is removed from i62 version 2.3.11 and above.
It is still recommended to add the parameter **DelayUnregister=15** to the SIP profile in order to alleviate re-registration issues. Refer to ShoreTel configuration section for details.
- Display information. Caller will see only called party's number.
If A calls B. B will see A's name and number but A will see only B's number.

Configuration Overview

This document describes the major steps needed to configure the ShoreTel system and the Ascom i62 VoWiFi handset for interoperability.

ShoreTel Configuration

This section describes the ShoreTel system configuration to support the Ascom. The section is divided into general system settings and individual user configuration needed to support the Ascom i62 VoWiFi handsets.

ShoreTel System Settings - General

The first settings to address within the ShoreTel system are the general system settings. These configurations include the call control, the switch and the site settings. If these items have already been configured on the system, skip this section and go on to the “ShoreTel System Settings – Individual Users” section below.

Call Control Settings

The Call Control Options within ShoreWare Director may need to be reconfigured. To configure these settings for the ShoreTel system, log into ShoreWare Director and select “**Administration**” “**Call Control**” and then “**Options**” (Figure 2).

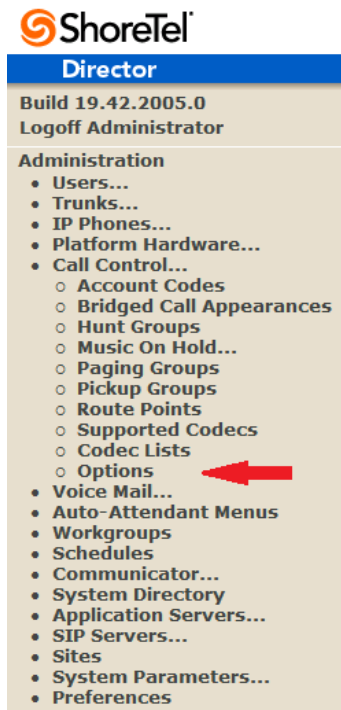


Figure 2 – Administration Call Control/Options

The “Call Control/Options” screen will then appear (Figure 3).

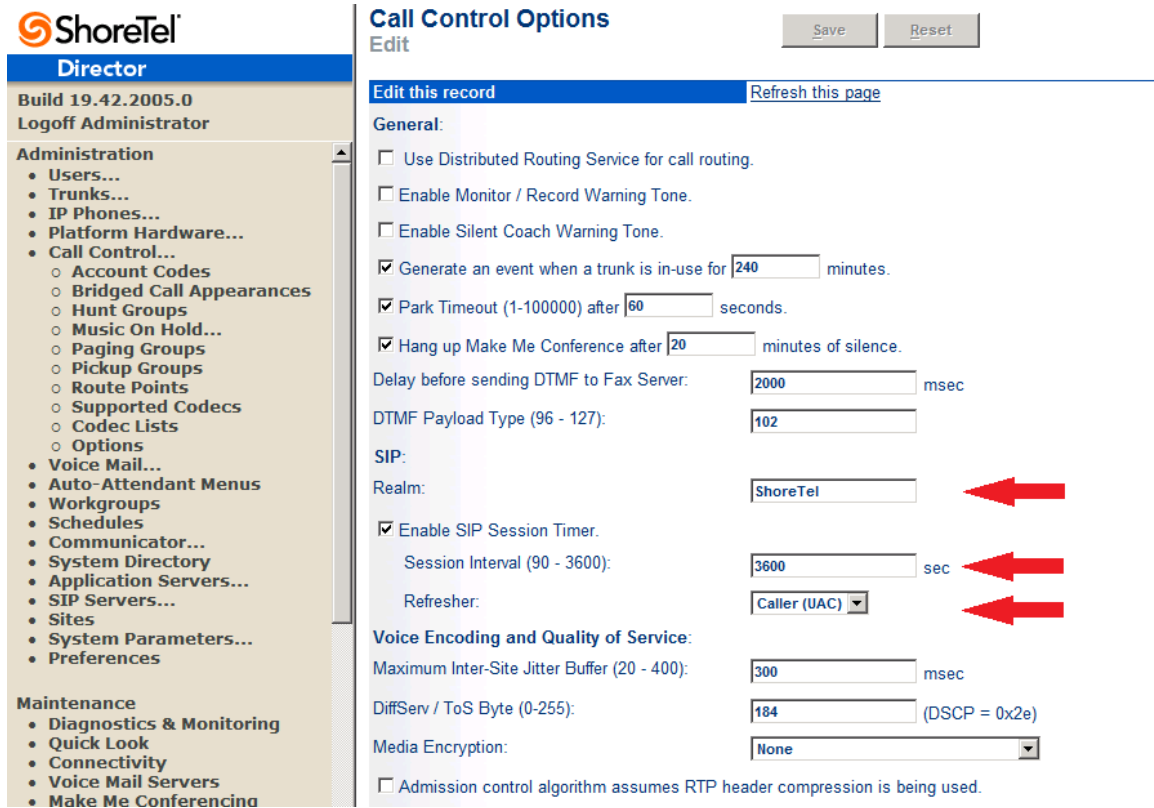


Figure 3 – Call Control/Options Screen

- **Realm:** The realm is used in authenticating all SIP devices. It is typically a description of the computer or system being accessed. Changing this value will require reboot of switches serving as SIP extensions. It is not necessary to modify this parameter to get the i62 VoWiFi handsets functional.
- **SIP session interval:** Session interval value indicates the session (call) “keep alive” period. There is no need to modify the default value of “3600” seconds.
- **SIP session refresher:** The refresher setting decides if user agent client or user agent server refreshes the session. Again, there is no need to modify the default value of “Caller (UAC).” This allows the i62 VoWiFi handset to be in control of the session timer refresh.

Switch Settings

When allocating Ports for SIP extensions, these changes are modified by selecting “Administration” “Platform Hardware...” followed by “Voice Switches / Service Appliances...”, then “Primary” in ShoreWare Director (Figure 4).

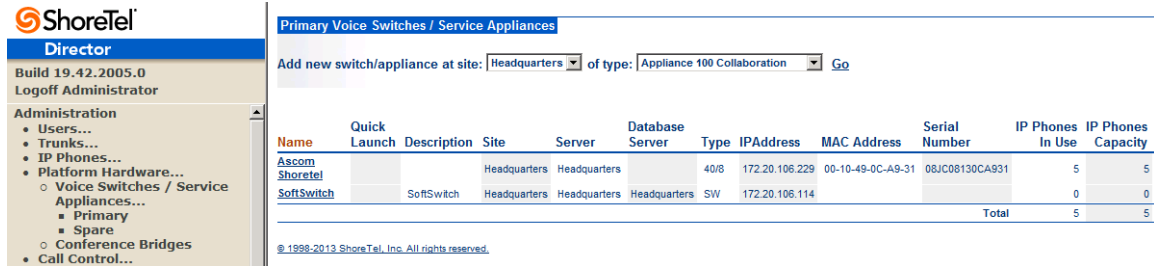


Figure 4 – Administration/Switches

This action brings up the “Switches” screen. From the “Switches” screen, simply select the name of the switch to configure. The “Edit ShoreGear ...Switch” screen will be displayed. Within the “Edit ShoreGear ...Switch” screen, 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 i62 VoWiFi handsets (Figure 5), then save the change.

Note: If your installation requires more than 100 SIP extensions configure the “Port Type” as “100 SIP Proxy” as necessary (i.e., two ports configured for “100 SIP Proxy” will provide 200 SIP extensions). Remember, SIP endpoints also utilize IP Phone Ports.

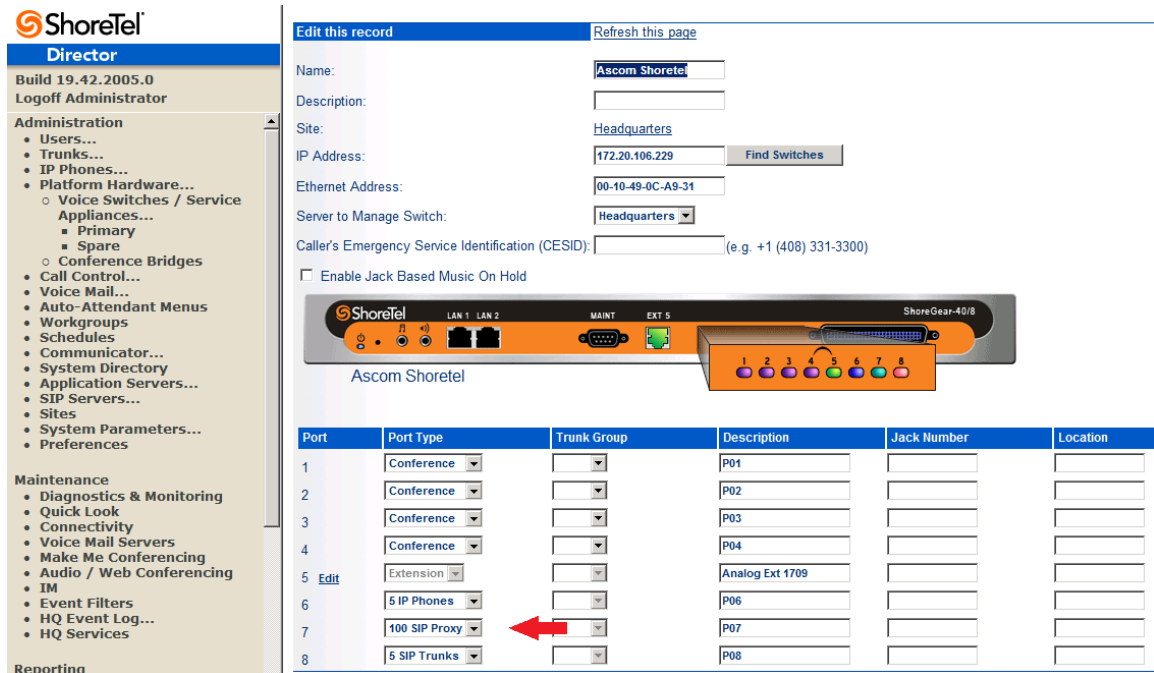


Figure 5 – Edit Switches

If the 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 (**Figure 6**).

Note: Every 5 ports you remove from the total available will result in “100 SIP Proxy” ports being made available.

One dedicated ShoreGear 120 switch can act as a proxy for the entire site and support up to 2400 SIP phones.

Switches
Edit ShoreGear 90 Switch

New Copy Save Delete Reset

Edit this record Refresh this page

Name: HQ-SG90

Description: HQ-SG90

Site: Headquarters

IP Address: 10.23.102.100 Find Switches

Ethernet Address: 00-10-49-07-27-CE


Server to Manage Switch: Headquarters

Caller's Emergency Service Identification (CESID): +1 (509) 921-2221 (e.g. +1 (408) 331-3300)

Built-in Capacity: IP Phone + SIP Trunk = Total
20 + 5 = 25 of 30 (100 SIP proxy ports) ←

Music On Hold Source
Music On Hold Gain (-49 to 13): 5 dB

Use Analog Extension Ports as DID Trunks



The image shows a ShoreTel ShoreGear 90 switch and its port expansion module. The switch is orange and black, with a status indicator, LAN 1, LAN 2, and MAINT ports. The expansion module is also orange and black, with 12 ports labeled 1 through 12.

Figure 6 – ShoreGear Switch Built-in Capacity

Sites Settings

The next settings to address are the administration of sites. These settings are modified under the ShoreWare Director by selecting “Administration” then “Sites” (Figure 7).

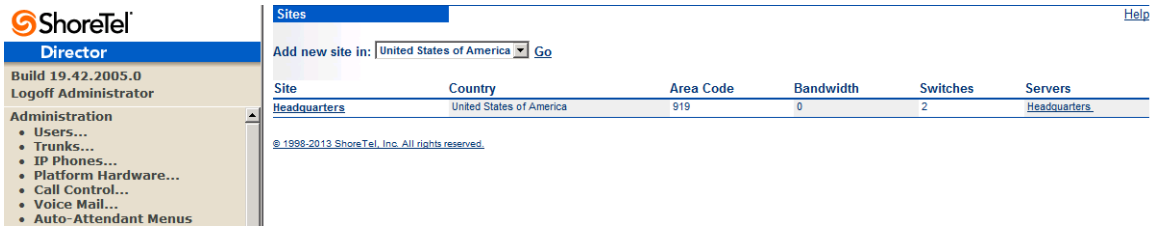


Figure 7 – Administration/Sites

This selection brings up the “Sites” screen. Within the “Sites” screen, select the name of the site to configure. The “Edit Site” screen will then appear. Scroll down to the “SIP Proxy” parameters (Figure 8).

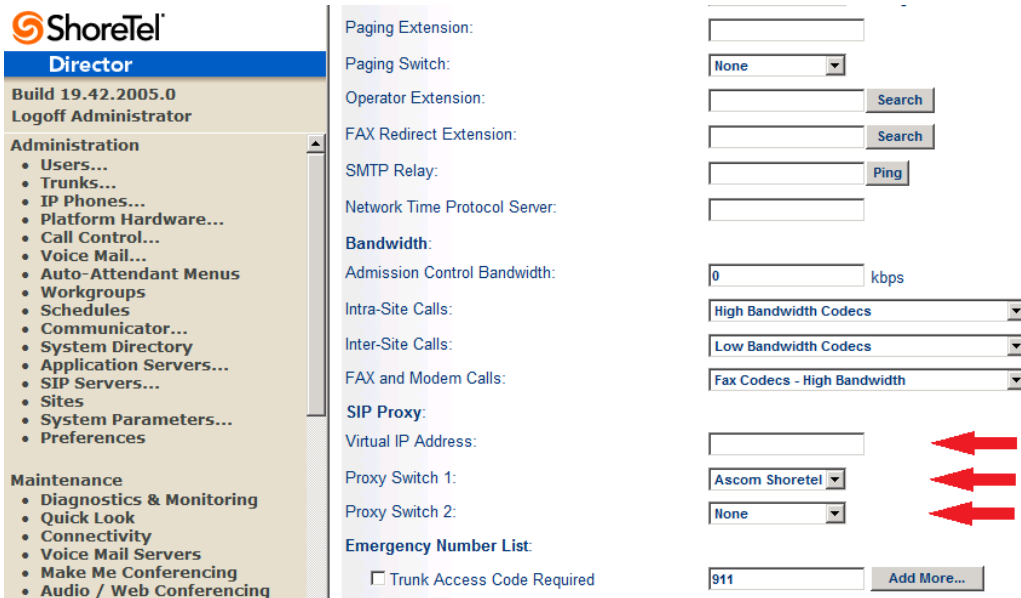


Figure 8 – Site Screen SIP Proxies

The “Virtual IP Address” parameter is a new configuration parameter beginning with ShoreTel 8. This “Virtual IP Address” is an IP address that can be moved to a different switch during a failure. For each site that supports SIP extensions, one “Virtual IP Address” is defined that will act as the SIP Proxy for the site. This IP address must be unique and static.

The ShoreTel server will assign this “Virtual IP Address” to the ShoreGear that is configured as SIP proxy for the site. Two ShoreGear switches can be configured as SIP proxy servers for redundancy and reliability purposes. If the primary proxy server goes down, the other proxy switch will take over the “Virtual IP Address.” Due to this “Virtual IP Address” mechanism, SIP phones will not know if the proxy switch goes off-line.

Note: If you choose not to define a “Virtual IP Address,” you can only define one proxy switch, and there is 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.

The Admission Control Bandwidth defines the bandwidth available to and from the site. This is important as SIP endpoints may be counted against the site bandwidth. See the ShoreTel Planning and Installation Guide for more information about this.

By default, ShoreTel 14.x has 12 built-in codecs, these codecs can be grouped as “Codec Lists” and defined in the sites page for “Inter-site” and “Intra-site” calls. See ShoreTel’s Administration Guide for more information. The default settings will work properly with the Ascom i62 VoWiFi handsets.

Creating SIP Extension

You need to create a user extension for the Ascom i62 VoWiFi handset. This is accomplished from ShoreWare Director by selecting “Administration” followed by “Users...” then “Individual Users” This action will bring up the “Individual Users” screen at the top of the page. To the right of “Add new user at site:” select the site you wish to create the user in (from the drop down menu), and select “Go” (Figure 9).

The screenshot shows the 'Individual Users' page in the ShoreTel Director interface. The left sidebar contains a navigation menu with 'Administration' expanded to 'Users...' and 'Individual Users' selected. The main content area features a header with the ShoreTel logo and 'Director' title, and a sub-header 'Individual Users' with a 'Help' link. Below the header, there is a form to 'Add new user at site:' with a dropdown menu set to 'Headquarters' and a 'Go' button. A 'Show page:' dropdown is set to '1: Analog - ShoreTel230', and there are navigation icons and a '6 Records 25 per page' indicator. The main content is a table with the following data:

| First Name | Last Name | Site | User Group | Access License | Extension | Mailbox | Switch | Port | Status |
|-------------|-----------|--------------|----------------|----------------|-----------|---------|----------------|-------------------|----------|
| Analog | Ext 1709 | Headquarters | Executives | Personal | 1709 | 1709 | Ascom Shoretel | 5 | Home |
| AscomPhone1 | x1703 | Headquarters | Codes required | Personal | 1703 | 1703 | SoftSwitch | | Assigned |
| AscomPhone2 | x1704 | Headquarters | Executives | Personal | 1704 | 1704 | SoftSwitch | | Assigned |
| AscomPhone3 | x1705 | Headquarters | Executives | Personal | 1705 | 1705 | SoftSwitch | | Assigned |
| ShoreTel230 | One | Headquarters | Executives | Personal | 1701 | 1701 | Ascom Shoretel | 00-10-49-0A-F5-90 | Home |
| ShoreTel230 | Two | Headquarters | Executives | Personal | 1702 | 1702 | Ascom Shoretel | 00-10-49-0A-F5-8F | Home |

At the bottom of the page, there is a copyright notice: © 1998-2013 ShoreTel, Inc. All rights reserved.

Figure 9 – Individual Users Settings

This action brings up the “Users” “Edit Users” screen (**Figure 10**).

Figure 10 – Adding/Editing Users

Define the “**First Name**” and “**Last Name**” as you deem appropriate. ShoreWare Director will auto-assign the next available “**Number**” (i.e. extension), but you can modify it to any available extension. Define the “**License Type**” and “**Access License**” type as needed; in this example we chose “**Extension and Mailbox**” although it’s not necessary to have a mailbox, and “**Professional**” for “**Access License**”. Define the proper “**User Group**” and set the “**Primary Phone Port**” to “**Any IP Phone**” , the Primary Phone Port will automatically update once the Ascom i62 handset registers to the ShoreTel system.

Note: If you configured the “**License Type**” for “**Extension-Only**,” you cannot select “**Any IP Phone**” but instead must set the “**Home Port**” for the “**SoftSwitch**” selection. Save your changes, then scroll down to the “**SIP Password:**” section (**Figure 11**).

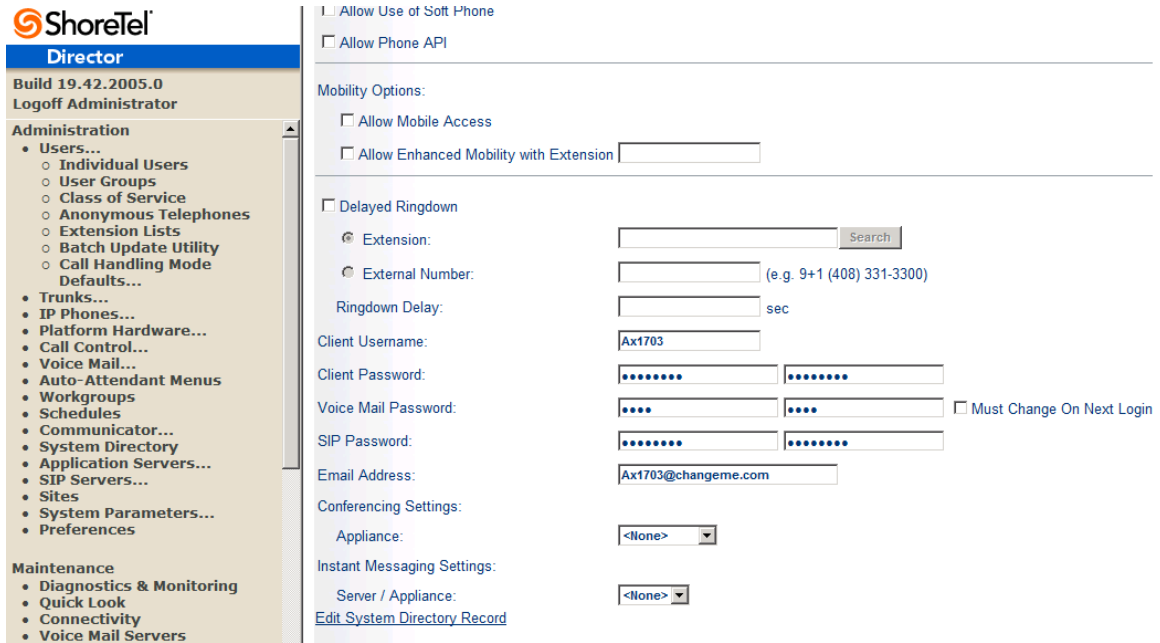


Figure 11 – Individual User SIP Settings

There is no default “**SIP Password**” it is masked with the appearance that there is, but don’t be confused to think that there’s a default password. You can modify it to any value you wish, but be certain to note what you changed it to, as you will need it when configuring the i62 VoWiFi handset parameters. **Save** your changes.

SIP Profiles

ShoreWare Director’s “Call Control...” section contains an “SIP Profiles” option. ShoreTel 12.1 comes standard with a “_System” and “_ShorePhoneIP8000” SIP profiles (they cannot be deleted - only disabled). 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 ShoreWare Director by selecting “**Administration**” followed by “**IP Phones...**” then “**SIP Profiles**” This action brings up the “SIP Profiles” screen. At the top of the page, below the “SIP Profiles List” , select the “**New...**” radio button, as shown in **Figure 12**.



Figure 12 – SIP Profiles

This action brings up the “Edit SIP Profile” screen, Figure 13.

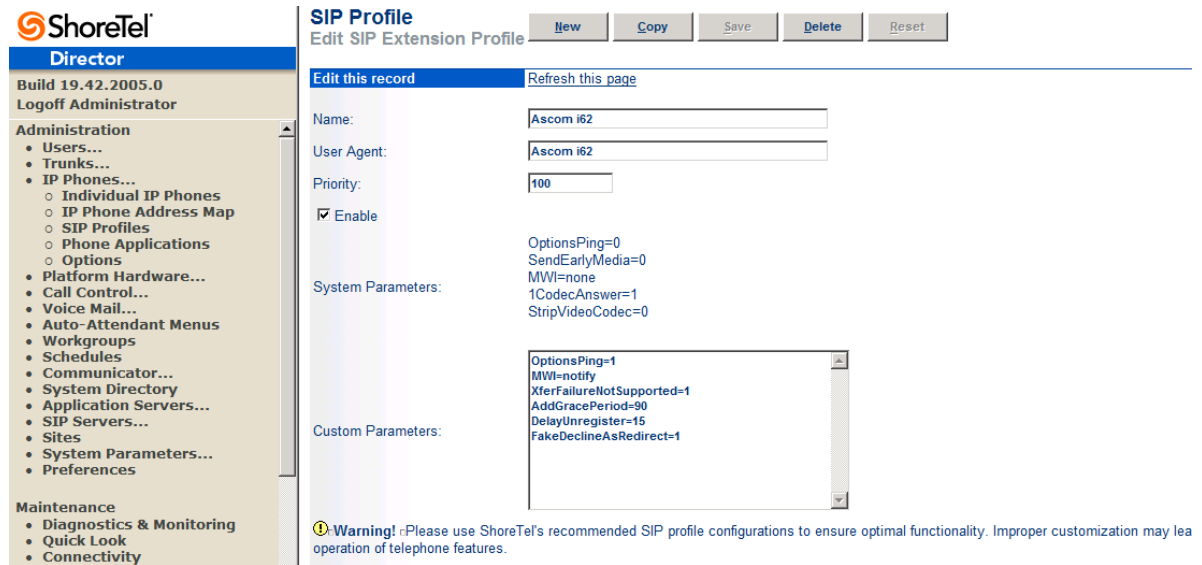


Figure 13 – Edit SIP Profile

Define a “**Name:**” for the entry, and be sure to define an appropriate name. For the “**User Agent:**” option, enter “Ascom i62.*” (without quotes); the “**Priority:**” defaults to 100, no change is required. Enable the profile by checking (enabling) the “**Enable**” option. In the “**Custom Parameters:**” options, add the following entries:

```
OptionsPing=0
MWI=notify
FakeDeclineAsRedirect=1
XferFailureNotSupported=1
AddGracePeriod=90
DelayUnregister=15
```

Save the changes.

Note: Please do not disable any of the default SIP profiles. In case there are issues with the custom profile defined, disabling the system profiles may cause the Ascom i62VoWiFi handsets to not be added to the ShoreTel system. Refer to the ShoreTel’s Planning and Installation Guide for more information.

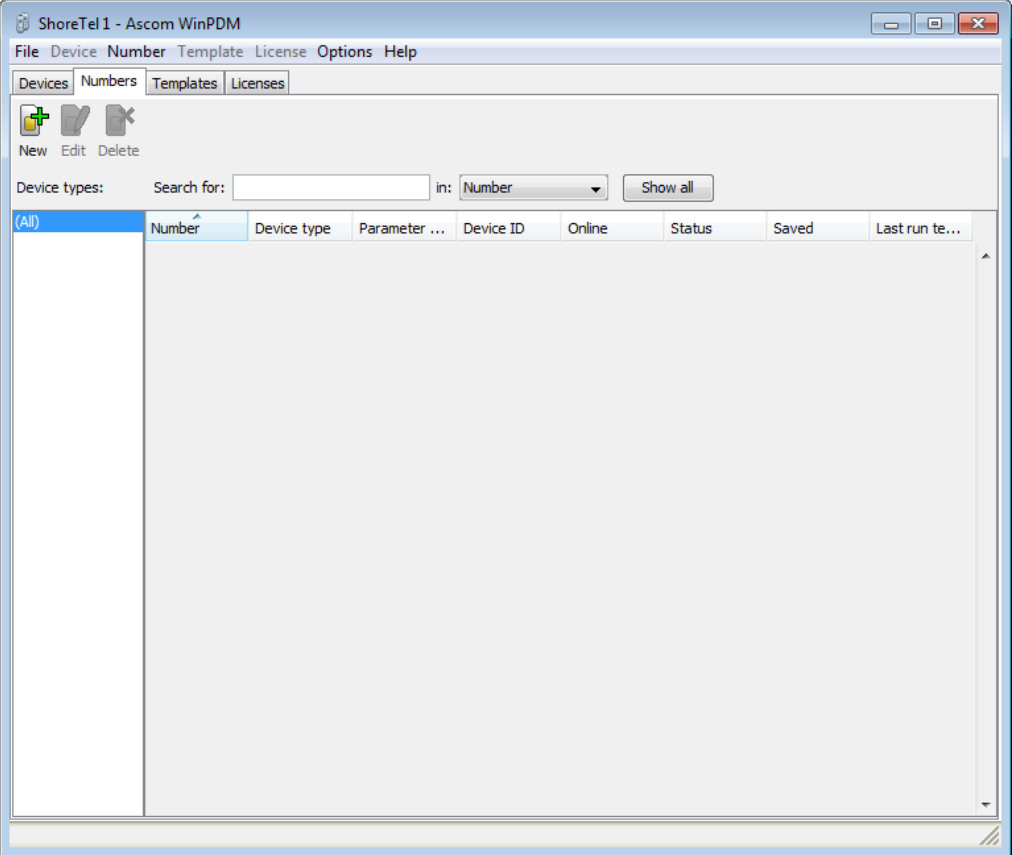
IP address Phone Map

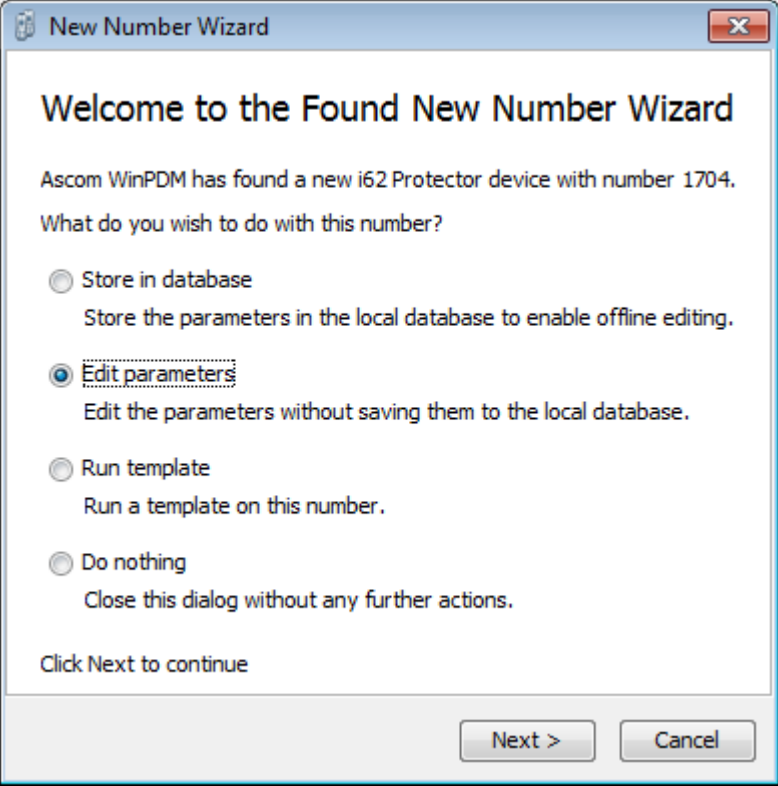
If you plan on adding Ascom i62 VoWiFi handsets at a different site, you will need to create an “IP Address Phone Map”. Create an “IP Address Phone Map”. You can do so via ShoreWare Director, navigating to the “**Administration**” “**IP Phones...**” “**IP Address Phone Map**” screen, then adding an entry for the desired site, with the IP address range of the i62 VoWiFi handsets. For more information on creating sites and adding switches, please refer to the ShoreTel Planning and Installation Guide.

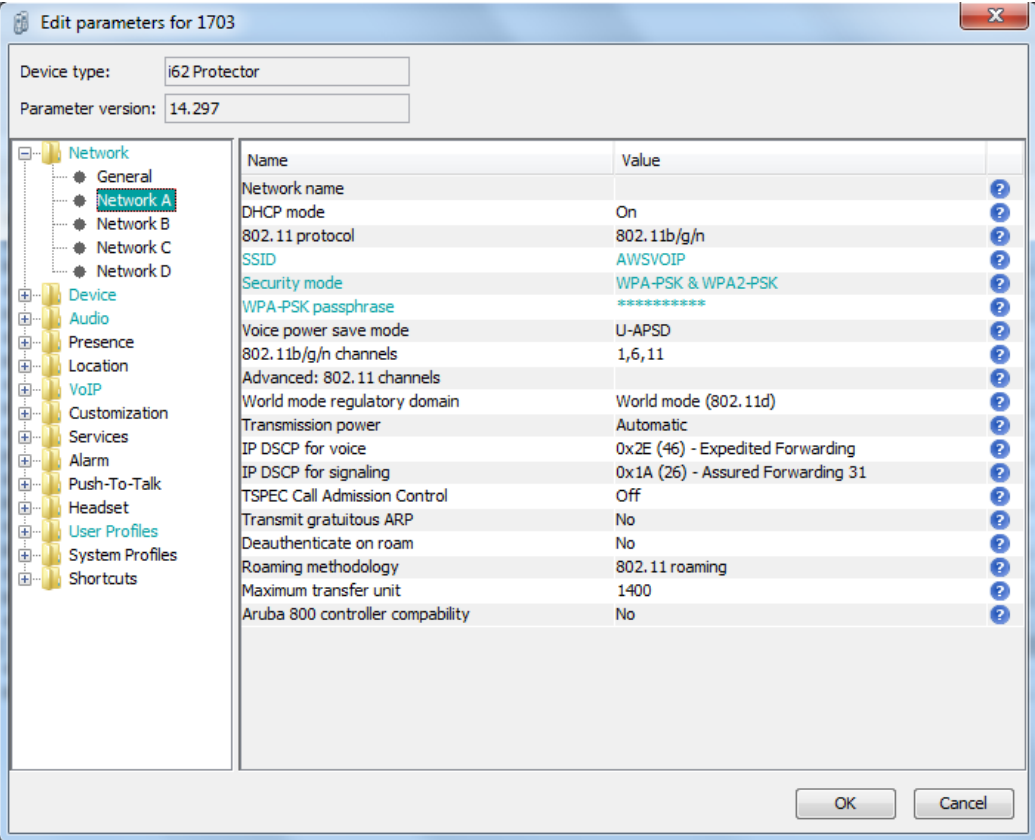
This completes all of the ShoreTel configuration parameters necessary to install the Ascom i62 VoWiFi handsets.

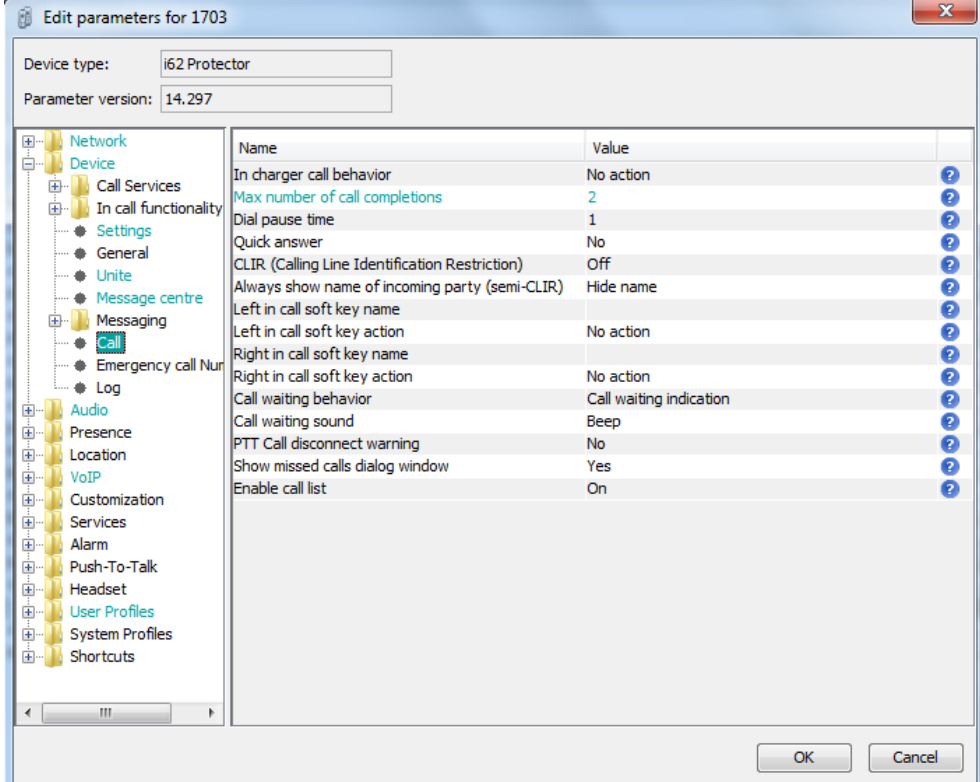
Ascom Configuration

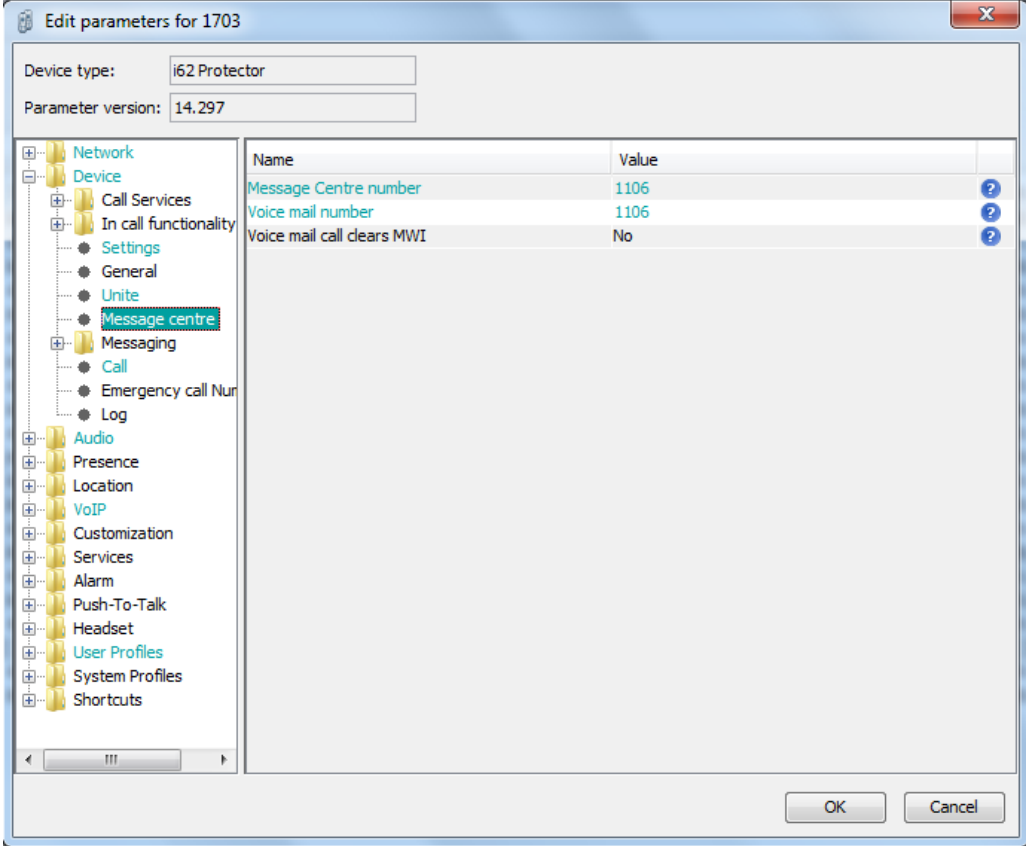
The following steps detail the configuration process for the Ascom i62 VoWiFi handset using the Ascom Portable Device Manager (PDM) Windows-based application.

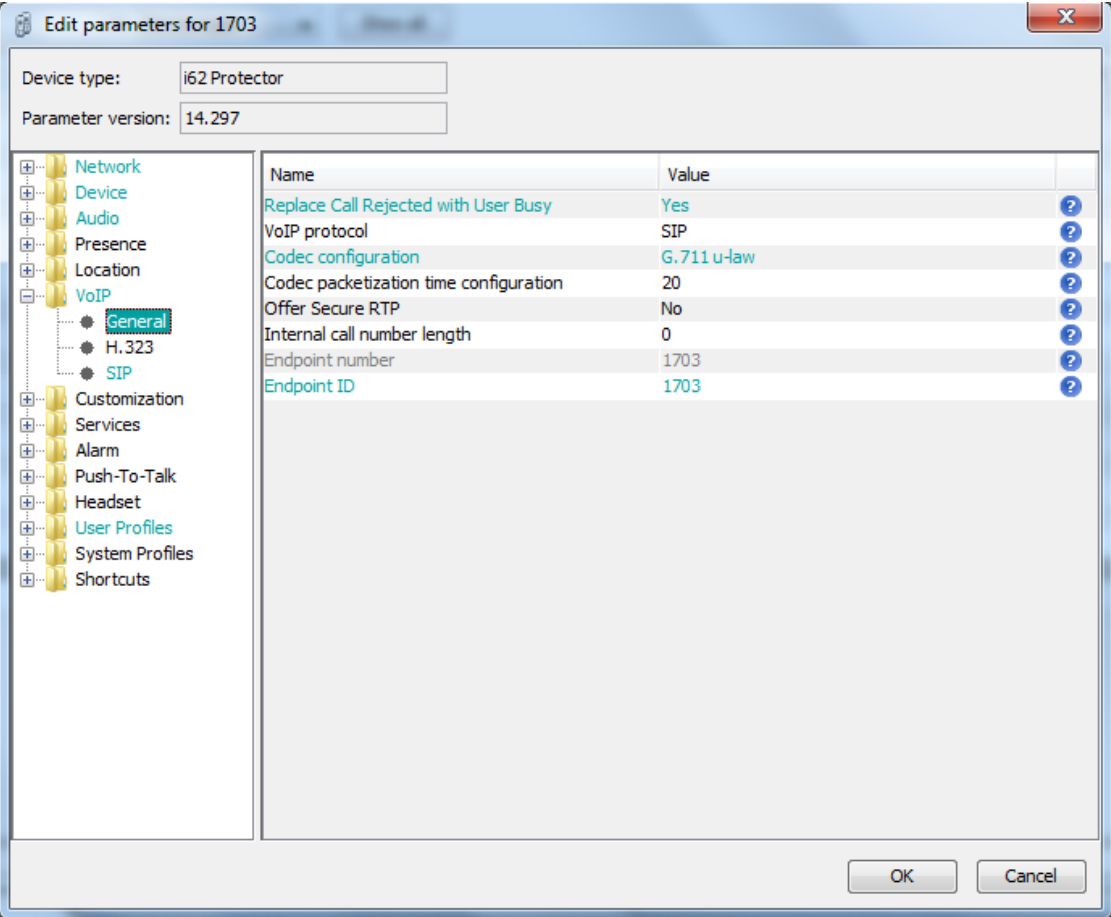
| Step | Description |
|------|---|
| 1 | <p>Launch the PDM application from the computer that has the application installed and has the PDM cradle physically attached via a USB cable. Before the user is presented with the following screen, a login is required. See Section 10 [3] for administration and configuration information on the PDM.</p> <p>After the user has logged onto the PDM, the following screen is displayed showing the devices found in the database. Since no devices have been plugged into the PDM, none are shown at this time.</p>  |

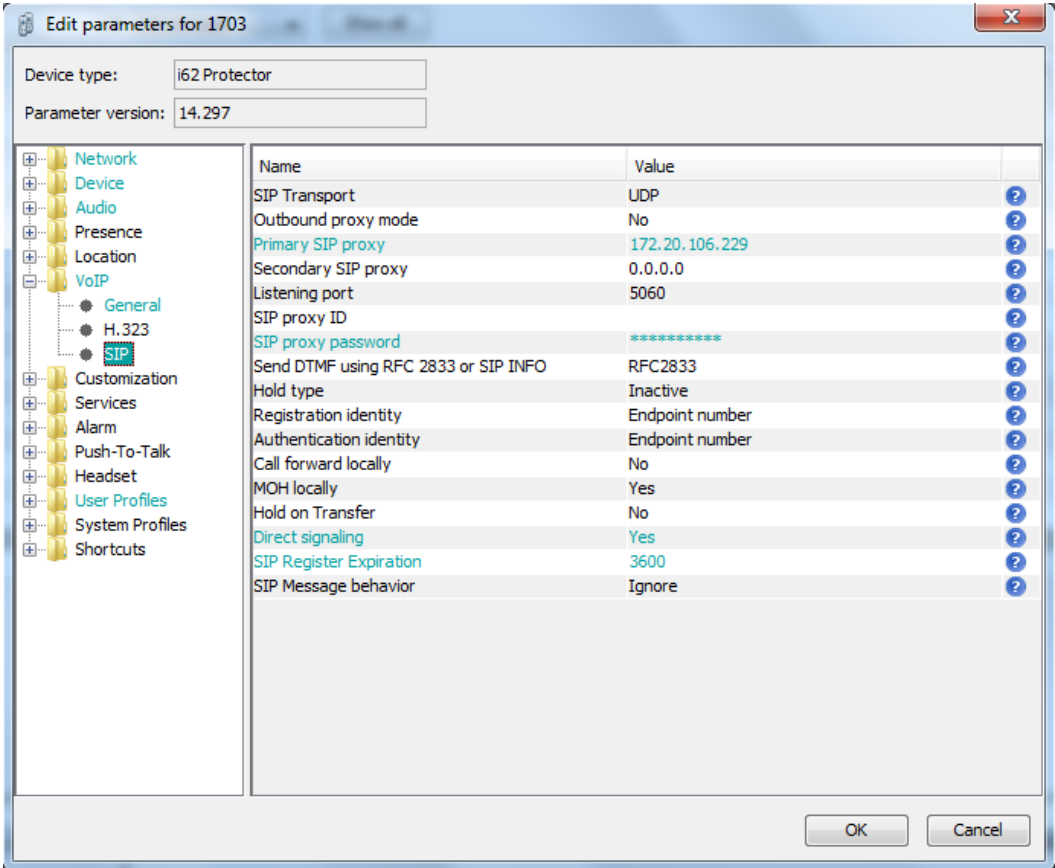
| Step | Description |
|------|---|
| 2 | <p>Once an Ascom i62 portable handset is placed into the cradle, the PDM recognizes the telephone and cross references the database of telephones. If the telephone is not found in the database, the PDM prompts the user to save the new telephone to the database. Click the radio button labeled Edit parameters and then click Next.</p>  |

| Step | Description |
|------|--|
| 3 | <p>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.</p> <p>Note: Below is a typical configuration utilizing. Different Security modes might be used.</p> <p>SSID: AWSVOIP2</p> <p>Security mode: WPA2-PSK</p> <p>IP DSCP for voice “0x2e (46) – Expedited Forwarding”</p> <p>IP DSCP for signaling “0x1A (26) – Assured Forwarding 31”</p>  |

| Step | Description | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
|--|--|------|-------|--------------------------|-----------|--------------------------------|---|-----------------|---|--------------|----|--|-----|--|-----------|----------------------------|--|------------------------------|-----------|-----------------------------|--|-------------------------------|-----------|-----------------------|-------------------------|--------------------|------|-----------------------------|----|---------------------------------|-----|------------------|----|
| 4 | <p data-bbox="349 346 1347 378">Navigate to the “Device -> General” configuration page by clicking Device and then Call.</p>  <table border="1" data-bbox="576 535 1315 1123"> <thead> <tr> <th>Name</th> <th>Value</th> </tr> </thead> <tbody> <tr><td>In charger call behavior</td><td>No action</td></tr> <tr><td>Max number of call completions</td><td>2</td></tr> <tr><td>Dial pause time</td><td>1</td></tr> <tr><td>Quick answer</td><td>No</td></tr> <tr><td>CLIR (Calling Line Identification Restriction)</td><td>Off</td></tr> <tr><td>Always show name of incoming party (semi-CLIR)</td><td>Hide name</td></tr> <tr><td>Left in call soft key name</td><td></td></tr> <tr><td>Left in call soft key action</td><td>No action</td></tr> <tr><td>Right in call soft key name</td><td></td></tr> <tr><td>Right in call soft key action</td><td>No action</td></tr> <tr><td>Call waiting behavior</td><td>Call waiting indication</td></tr> <tr><td>Call waiting sound</td><td>Beep</td></tr> <tr><td>PTT Call disconnect warning</td><td>No</td></tr> <tr><td>Show missed calls dialog window</td><td>Yes</td></tr> <tr><td>Enable call list</td><td>On</td></tr> </tbody> </table> | Name | Value | In charger call behavior | No action | Max number of call completions | 2 | Dial pause time | 1 | Quick answer | No | CLIR (Calling Line Identification Restriction) | Off | Always show name of incoming party (semi-CLIR) | Hide name | Left in call soft key name | | Left in call soft key action | No action | Right in call soft key name | | Right in call soft key action | No action | Call waiting behavior | Call waiting indication | Call waiting sound | Beep | PTT Call disconnect warning | No | Show missed calls dialog window | Yes | Enable call list | On |
| Name | Value | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| In charger call behavior | No action | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Max number of call completions | 2 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Dial pause time | 1 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Quick answer | No | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| CLIR (Calling Line Identification Restriction) | Off | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Always show name of incoming party (semi-CLIR) | Hide name | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Left in call soft key name | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Left in call soft key action | No action | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Right in call soft key name | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Right in call soft key action | No action | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Call waiting behavior | Call waiting indication | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Call waiting sound | Beep | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| PTT Call disconnect warning | No | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Show missed calls dialog window | Yes | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Enable call list | On | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |

| Step | Description | | | | | | | | |
|----------------------------|---|------|-------|-----------------------|------|-------------------|------|----------------------------|----|
| 5 | <p>Navigate to the “Device -> Message Centre” configuration page by clicking Device and then Message Centre.</p> <p>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.</p>  <table border="1" data-bbox="594 646 1360 1276"> <thead> <tr> <th>Name</th> <th>Value</th> </tr> </thead> <tbody> <tr> <td>Message Centre number</td> <td>1106</td> </tr> <tr> <td>Voice mail number</td> <td>1106</td> </tr> <tr> <td>Voice mail call clears MWI</td> <td>No</td> </tr> </tbody> </table> | Name | Value | Message Centre number | 1106 | Voice mail number | 1106 | Voice mail call clears MWI | No |
| Name | Value | | | | | | | | |
| Message Centre number | 1106 | | | | | | | | |
| Voice mail number | 1106 | | | | | | | | |
| Voice mail call clears MWI | No | | | | | | | | |

| Step | Description |
|------|---|
| 6 | <p>Navigate to the “VoIP/General” configuration page by clicking VoIP and then General. Configure the following parameters.</p> <p>Replace Call Rejected with User Busy: Enable. If this value is not set correctly, certain calling features such as transfer will not operate properly.</p> <p>VoIP protocol “SIP”</p> <p>Coder configuration “G.711 u-law”</p> <p>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.</p>  |

| Step | Description | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
|--------------------------------------|---|------|-------|---------------|-----|---------------------|----|-------------------|----------------|---------------------|---------|----------------|------|--------------|--|--------------------|-------|--------------------------------------|---------|-----------|----------|-----------------------|-----------------|-------------------------|-----------------|----------------------|----|-------------|-----|------------------|----|------------------|-----|-------------------------|------|----------------------|--------|
| 7 | <p>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.</p> <p>The following screen shot shows:</p> <p>SIP proxy IP address “172.20.106.229”</p> <p>SIP proxy password “*****”</p> <p>Direct signaling: Enabled</p> <p>SIP Registration Expiration: 3600</p>  <table border="1" data-bbox="350 800 1398 1663"> <caption>Edit parameters for 1703</caption> <thead> <tr> <th>Name</th> <th>Value</th> </tr> </thead> <tbody> <tr><td>SIP Transport</td><td>UDP</td></tr> <tr><td>Outbound proxy mode</td><td>No</td></tr> <tr><td>Primary SIP proxy</td><td>172.20.106.229</td></tr> <tr><td>Secondary SIP proxy</td><td>0.0.0.0</td></tr> <tr><td>Listening port</td><td>5060</td></tr> <tr><td>SIP proxy ID</td><td></td></tr> <tr><td>SIP proxy password</td><td>*****</td></tr> <tr><td>Send DTMF using RFC 2833 or SIP INFO</td><td>RFC2833</td></tr> <tr><td>Hold type</td><td>Inactive</td></tr> <tr><td>Registration identity</td><td>Endpoint number</td></tr> <tr><td>Authentication identity</td><td>Endpoint number</td></tr> <tr><td>Call forward locally</td><td>No</td></tr> <tr><td>MOH locally</td><td>Yes</td></tr> <tr><td>Hold on Transfer</td><td>No</td></tr> <tr><td>Direct signaling</td><td>Yes</td></tr> <tr><td>SIP Register Expiration</td><td>3600</td></tr> <tr><td>SIP Message behavior</td><td>Ignore</td></tr> </tbody> </table> | Name | Value | SIP Transport | UDP | Outbound proxy mode | No | Primary SIP proxy | 172.20.106.229 | Secondary SIP proxy | 0.0.0.0 | Listening port | 5060 | SIP proxy ID | | SIP proxy password | ***** | Send DTMF using RFC 2833 or SIP INFO | RFC2833 | Hold type | Inactive | Registration identity | Endpoint number | Authentication identity | Endpoint number | Call forward locally | No | MOH locally | Yes | Hold on Transfer | No | Direct signaling | Yes | SIP Register Expiration | 3600 | SIP Message behavior | Ignore |
| Name | Value | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| SIP Transport | UDP | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Outbound proxy mode | No | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Primary SIP proxy | 172.20.106.229 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Secondary SIP proxy | 0.0.0.0 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Listening port | 5060 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| SIP proxy ID | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| SIP proxy password | ***** | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Send DTMF using RFC 2833 or SIP INFO | RFC2833 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Hold type | Inactive | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Registration identity | Endpoint number | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Authentication identity | Endpoint number | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Call forward locally | No | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| MOH locally | Yes | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Hold on Transfer | No | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Direct signaling | Yes | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| SIP Register Expiration | 3600 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| SIP Message behavior | Ignore | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |

Ascom Troubleshooting

For troubleshooting of the Ascom i62 handset, see the following Ascom documentation:

User Manual – Ascom i62 VoWiFi Handset TD92597GB

Configuration Manual – Ascom i62 VoWiFi Handset 92675GB

Ascom Technical Support

Technical support for the Ascom i62 VoWiFi handset can be obtained through the following:

For local US/Canada support:

- **Phone:** 1-877-71ASC0M or 1-877-712-7266
- **Email:** techsupport@ascomwireless.com (for Technical support)

For world wide support:

- **Phone:** 46 31 55 9450
- **Email:** support@ascom.se (for Technical support)

For international customer:

- **Internet:** www.ascom.com/ws and select your country of interest, to find local sales and support contact information.

ShoreTel Technical Support

ShoreTel technical support can be obtained through the following:

- **Phone:** +1 800 742-2348
- **Web:** www.support.shoretel.com



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