



# SHORETEL APPLICATION NOTE

for

Ascom i62

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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

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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 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

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

Technical Support

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## **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:	
Realm:	ShoreTel
Enable session timer	
Session interval:	1800 seconds (90-3600) 🖋
Refresher:	Caller (UAC)

#### 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

Port	Port Type	-	Trunk Group	Description	Jack Number
1	5 IP Phones 🗸	ø		P01	
2	100 SIP Proxy	ø		P02	

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.

Built-in capa	city:	
IP phone +	SIP trunks =	Total
25	0	25 of 30 (100 SIP proxy ports)

#### **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

ShoreTel Connect Direc	tor 🛛 Connections   🔵 Trunk Groups   🦲 Bandwidth   🌑 Voice Quali	ty   🚹 Appliances   🧧 Servers	Administrator   Help   Logout
Search	SIP Phone Profiles	NEW COPY DEI	LETE BULK DELETE
	□ NAME   USER AGENT  Ascom DECT  Ascom IP-DECT.*		PRIORITY
Users	✓ Ascom i62     Ascom i62.*     Ascom i75     Ascom i75.*		100
b Trunks 4 Telephones	Ascom Myco     Ascom Myco.*     RoamAnywhere Client     Ascom Myco.*		100
Telephones	ShorePhone IP8000         ^ShoreTel/ST_PH1_[2-6]\[0-9]\[0           swttom         •	-9] 🗹	50
Anonymous Phones	- System .		
Vacated Phones SIP Profiles			
Phone Applications			
Appliances/Servers			

- 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

Ascom i62		SAVE RESET CANC
Name:	Ascom i62	8
User agent:	Ascom i62.*	
Priority:	100	
Enable		
System parameters:	OptionsPing=0 SendEarlyMedia=0 MWI=none 1CodecAnswer=1 StripVideoCodec=0	
Custom parameters:	OptionsPing=1	~
Gustom parameters.	MWI=notify FakeDeclineAsRedirect=1 XferFailureNotSupported=1 AddGracePeriod=90 DelayUnregister=15	
		Ť.
Warning! Use ShoreTel's rec operation of telephone featur	ommended SIP profile configurations to ensure op res.	ptimal functionality. Improper customization may lead to faul

#### 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

Virtual IP address:	
Proxy switch 1:	vPhone 🗸
Proxy switch 2:	<none></none>

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
- 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".*

Users	NEW COPY DEL	ETE EXPORT BULK DELETE BULK EDIT
Extension 1703: AscomPhone	e1 x1703	SAVE RESET CANCEL
GENERAL TELEPHONY	VOICE MAIL ROUTING	MEMBERSHIP DNIS APPLICATIONS
First name:	AscomPhone1	
Last name:	x1703	
Extension:	1703	
Email address:	Ax1703@changeme.com	Edit System Directory record
Client username:	Ax1703	]
✓ Include in System Dial by Name	directory	
Make extension private		
DID Settings:	(not configured)	change settings
PSTN failover:	None	
Caller ID (overwrite DID):	+1 (919) 234-2451	(e.g. +1 (408) 331-3300)
License type:	Extension and Mailbox 🗸	
Access license:	Phone Only	
User group:	Codes required Go to this	s user group
Site:	Headquarters Go to this site	
Language:	English(US)	

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

	•••••	]
SIP phone password:	•••••	) (6 - 26 characters)
	must change on next login	
	•••••	]
Client password:	•••••	(6 - 26 characters)
Mailbox server:	Headquarters	
Jack #:		]
Current port:	SIP-334-0131207713256997408	GO PRIMARY PHONE

## Ascom Configuration

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

 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".

👸 Edit parameters	for 1703	×
Device type:	i62 Protector	
Parameter definition:	: 14.330	
Network     General     Network     Network     Network C     Network     Network     Network     Network     Network C	Name         Network name         DHCP mode         802.11 protocol         SSID         Security mode         WPA-PSK passphrase         Voice power save mode         802.11a/n channels         Advanced: 802.11 channels         World mode regulatory domain         Transmission power         IP DSCP for voice         IP DSCP for signaling         TSPEC Call Admission Control         Transmit gratuitous ARP         Deauthenticate on roam         Roaming methodology         Maximum transfer unit         Aruba 800 controller compability         Check IP connectivity after roaming	Value       2         On       2         802.11a/n       2         AWSVOIP       2         WPA-PSK & WPA2-PSK       2         **********       2         U-APSD       2         Non DFS       2         World mode (802.11d)       2         Automatic       2         0x2E (46) - Expedited Forwarding       2         0x1A (26) - Assured Forwarding 31       2         Off       2         No       2
		OK Cancel

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.

🔋 Edit parameters f	for 1703		×
Device type:	i62 Protector		
Parameter definition:	14.330		
Network     General     Network A     Network A     Network C     Networ C     Networ C     Networ C     Networ C     Networ C     Networ	Name Replace Call Rejected with User Busy VoIP protocol Codec configuration Codec packetization time configuration Offer Secure RTP Internal call number length Endpoint number Endpoint ID	Value           Yes           SIP           G.711 u-law           20           No           0           1703           1703	
			OK Cancel

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

Edit parameters	for 1703	ascom	×
Device type:	i62 Protector		
Image: Service	Name SIP Transport Outbound proxy mode Primary SIP proxy Secondary SIP proxy Listening port SIP proxy ID SIP proxy password STUN server address Send DTMF using RFC 2833 or SIP INFO Hold type Registration identity Authentication identity Call forward locally MOH locally Hold on Transfer Direct signaling SIP Register Expiration SIP Message behavior ICE negotiation	Value           UDP           No           172.20.106.251           0.0.0           5060           **********           0.0.0.0           RFC2833           Inactive           Endpoint number           Endpoint number           No           Yes           No           Yes           1800           Ignore           No	2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2
		ОК	Cancel

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.

Edit parameters for 1703	3	Step Description	x
Device type: i62 Pro	tector	]	
Parameter definition: 14.330		]	
Image: Second	Name Message Centre number Voice mail number Voice mail call clears MWI	Value 1106 1106 No	2 2 2
		[	OK Cancel

## Summary of Tests and Results

N/S = Not Supported N/T = Not TestedN/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 Multiline not supp placement using secondary line	
2.9	N/A	Receive call – secondary line	Verify successful connection of Multiline not sup incoming call on secondary line	
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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