# Application Note

ShoreTel™

ST-0132
February 5, 2007

## Hitachi Wireless IP-5000 Phone (SIP)

**Wireless communication is a powerful solution which provides users the convenience of having their voice features available to them at any time and the freedom and mobility of using these features at any point on the wireless network. This application note provides the details on adding the Hitachi Wireless IP 5000 (SIP) handsets to the ShoreTel IP phone system.**

### Table of Contents



### Overview

This document provides details on the Hitachi Wireless IP-5000 handset and describes how to integrate this handset into the ShoreTel IP Phone System. The document focuses on the configuration procedures needed to set-up the Hitachi handset for the ShoreTel system and the configuration needed on the ShoreTel system to support the Hitachi handset.

**Note:** ShoreTel highly recommends using the Hitachi handset in a "Find Me / Follow Me" type configuration. Currently ShoreTel does not support "SIP Extensions", ShoreTel provides this application note as a work around when used with "SIP Trunks". Please keep in mind by using an "end point" device with "SIP Trunks" will have limitations.

### Document covers the following topics:

- Features and Benefits
- Vendor Overview and Contacts
- Hitachi Product Information
- System Requirements, Certification and Limitations
- ShoreTel Configuration
- Hitachi Configuration

This application note provides the ShoreTel-specific configuration instructions. Important Hitachi configuration steps are defined, however this application note does not cover all the general Hitachi handset installation and setup instructions. For general information, please refer to the Hitachi Wireless IP-5000 handsets product documentation. This application note also does not address detailed WiFi network setup or installation.

## Features and Benefits

The integration of wireless WiFi handsets leverages the power and cost effectiveness of ShoreTel's VoIP system along with the freedom of wireless mobility. The ShoreTel system enables cost effectiveness through the reduced costs of operation and maintenance along with the reduced call costs of VoIP communication.

The use of wireless handsets allows the convenience of placing calls anywhere within the WLAN and provides the users the power of having their calling features available at any time within WLAN. The combination of the ShoreTel's VoIP network and the Hitachi handset is a high value compelling proposition.

## Vendor Overview and Contact

General information can be obtained from ABP Technology (distributor) at the following URL: http://www.abptech.com/mainpages/products/HCL-WirelessIP5000.html

Latest documentation and Firmware can be located at (password proctected): http://www.abptech.com/mainpages/support/hitachi_downloads.html

Support: support@abptech.com

Info: info@abptech.com

Partners: partners@abptech

General Sales Contact:
ABP Technology
1850 Crown Dr. Ste. 1112
Farmers Branch (Dallas metro), TX - 75234
972-831-1600 / 972-831-0280
sales@abptech.com
http://www.abptech.com/

Resellers who want to start selling this solution should contact:
ABP Technology (Hitachi distributor)
www.abptech.com
Norma Adams, VAR coordinator
norma@abptech.com
+1.972.831.1600 x 121

## Hitachi Wireless IP-5000 Information


*Model Number: WLAN IP Phone*
*Part Number: Wireless IP 5000*
*Approximate list price: $355*

Wireless IP-5000 is an advanced wireless (IEEE 802.11b) IP phone that allows users to achieve perfect voice communication over an IEEE 802.11b wireless IP network. Not only does the Wireless IP-5000 support all of the conveniences and functions of a digital phone, it also perfectly integrates all of the merits and features of analog and IP phones over the data communication system.

Wireless IP-5000 provides your own private mobile network for your workplace so you no longer have to miss important calls as you move around your work site. Its compact and light body fits into your pocket, so you can use it anytime and anywhere. It can even be integrated into your existing phone system to provide lower-cost internal and external voice communication.

As voice signals are automatically compressed using audio CODECs, they can be transmitted over a low bandwidth without interference from other data signals. Furthermore, to provide consistent superior voice quality over varying and congested IP network conditions, the Wireless IP-5000 is equipped with advanced Quality of Service (QoS) technologies such as CODEC negotiation, enhanced jitter buffering technology, and packet delay compensation.

The Wireless IP-5000 Wireless VoIP phone comes with an 8 Line LCD Display. This phone is ideal for those who want the power and cost efficiencies of VoIP but also love the freedom of wireless. All the commonly used features needed for day-to-day business are accessible directly from the phone such as:

## Features

Dialed Number Display
Calling Time Display
Caller ID Display
Mute, Hold
Call Waiting Indication
Instant Message & Voicemail Message Indicator
*(Not supported at this time with ShoreTel)*
Phone book
Extension Number Display
Dial Tone
Distinctive Rings
Speed Dial
Redial
Call Pick Up
Call History
Missed calls indicator
Presence
*(Not supported at this time with ShoreTel)*
Short Message (IM: Instant Message)
Stand-by Display
Signal Level/Battery Level
Date/ Day Of The Week
Time
Silent Display/Vibration Display
Alarm Function
PC Connection by USB
Ear Phone/Mic
English /Japanese
User Friendly Graphical Menu

## Technical Specifications

**Hardware Dimensions:** (H x W x T): 127 x 43 x 19.2 [mm]
**Weight:** 102 [gr.]
**Display:** 8 line, 10 digits (2-byte characters)
LCD 128x128 pixel-graphical display with 4 Gray; Dot Size (WxH): 0.18 x 0.22mm; Dot Pitch (WxH): 0.19 x 0.23mm; Active Area (WxH): 24.31 x 29.43mm; FSTN Glass
**Operating Temperature:**
0~45 C / Storage Temperature: -25~70 C
**Buttons:**
Two Soft Key & 4 Directional Stick Key;
Volume Control Key;
Keypad Lock Key;
Send/End/Cancel Key

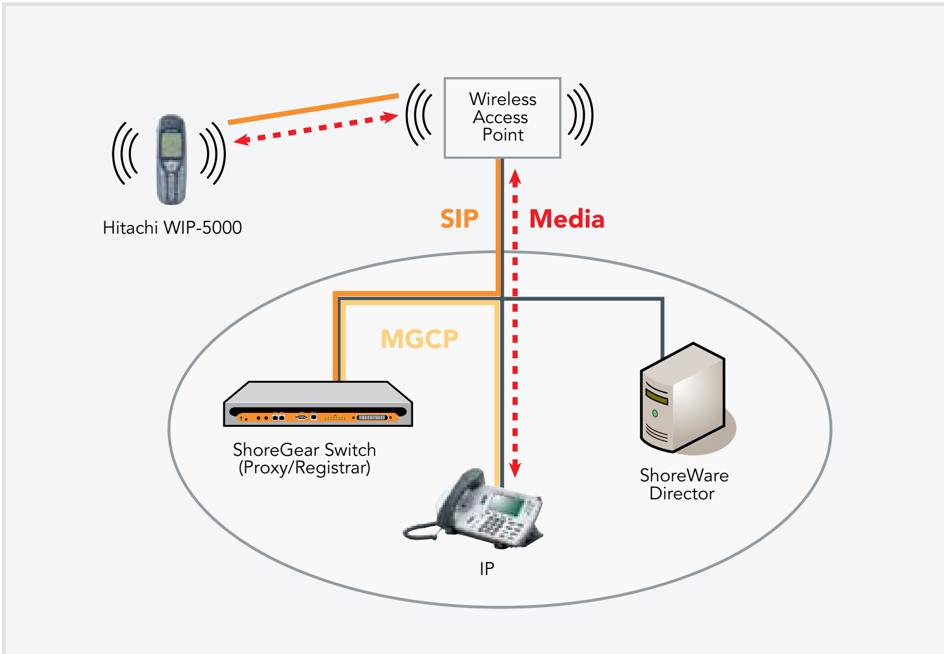
**Wireless LAN**
IEEE 802.11b
CSMA/CA
Direct Sequence Spread Spectrum
11/5.5/2/1Mbps
2.400-2.497GHz
**VoIP**
SIP v2.0
Coding: G.711 u / A-Law / Micro-Law, G.729A
Acoustic Echo Cancellation (to be supported)
**Networking:**
IPv4, TFTP, DNS
DHCP Client
NAT Transversal; Static NAT, uPnP, STUN
**QoS:**
Standard IEEE802.11b
DiffServ/IP Precedence; IEEE 802.11e (to be supported)
**Security:**
WEP (64bit/128bit/256bit)
802.1x (MD5/EAP-TLS)
Secure RTP (to be supported)
WEB based UAM
**Management:**
SNMP
Sys log
HTTP
PING (to check if IP packet has been delivered to a designated address)
Function for searching wireless access points
Function for measuring radio strength
**Standard Accessories:**
Charging Stand
**AC Adapter**
**Battery:** 1,350mA Extended Li-Ion Battery (3.1 / 55 hours)



**Optional Accessories:**
USB cable for PC connection
Wireless IP Manager

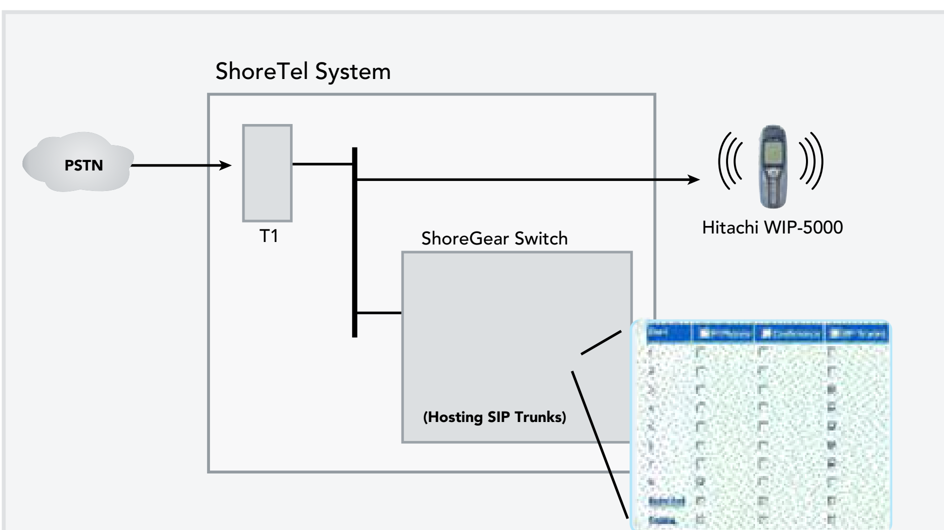

## Architecture Overview

The following is a diagram of the solution architecture showing the integration between the Hitachi Wireless IP-5000 handset and the ShoreTel System:



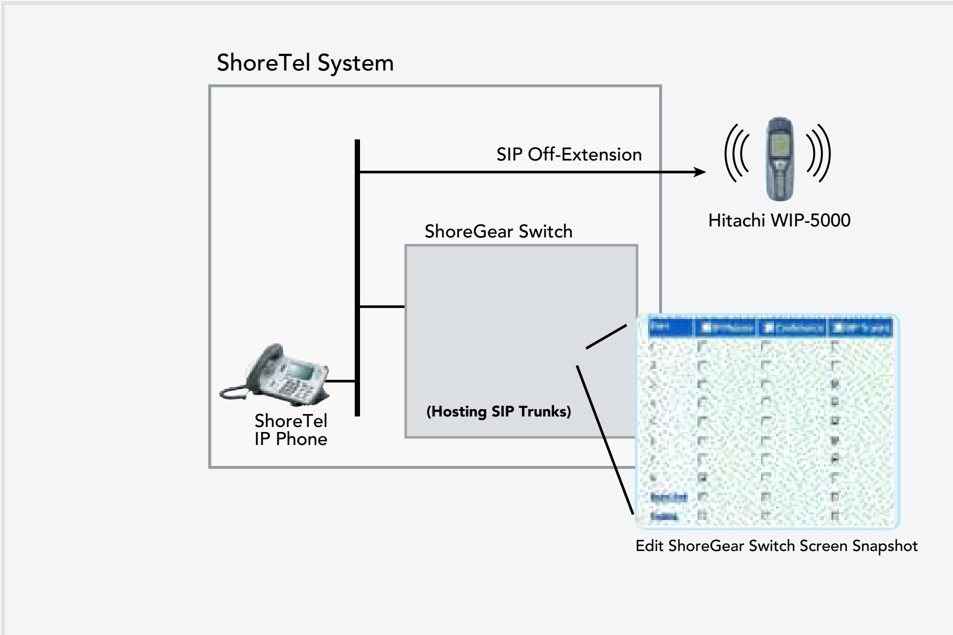

SIP devices are being implemented on the ShoreTel System with SIP trunks. The ShoreTel system will use outbound call routing for the SIP devices. There are various ways to handle the call routing for the SIP devices. The following drawings depict different call routing designs:

1. Route the call with DID (Direct Inward Dial) or DNIS (Dialed Number Identification Service) directly to the Hitachi Wireless IP-5000 extension. In this configuration tandem trunking is enabled and off system extensions are defined in the trunk group (trunk group setup instructions below). Shown in the drawing is a snapshot of the "Edit ShoreGear Switch" screen indicating the selection of SIP trunks.

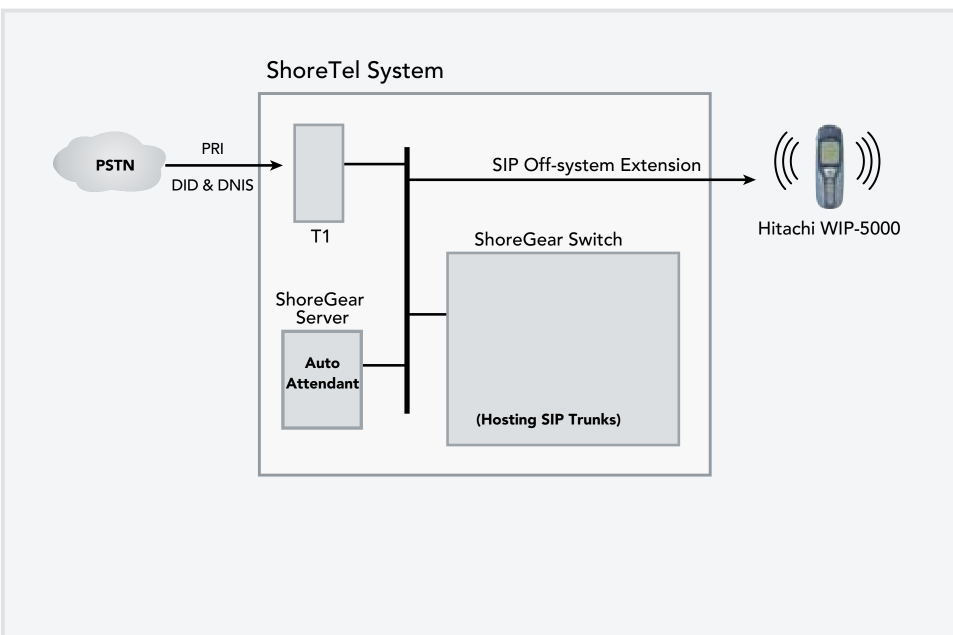



Edit ShoreGear Switch Screen Snapshot

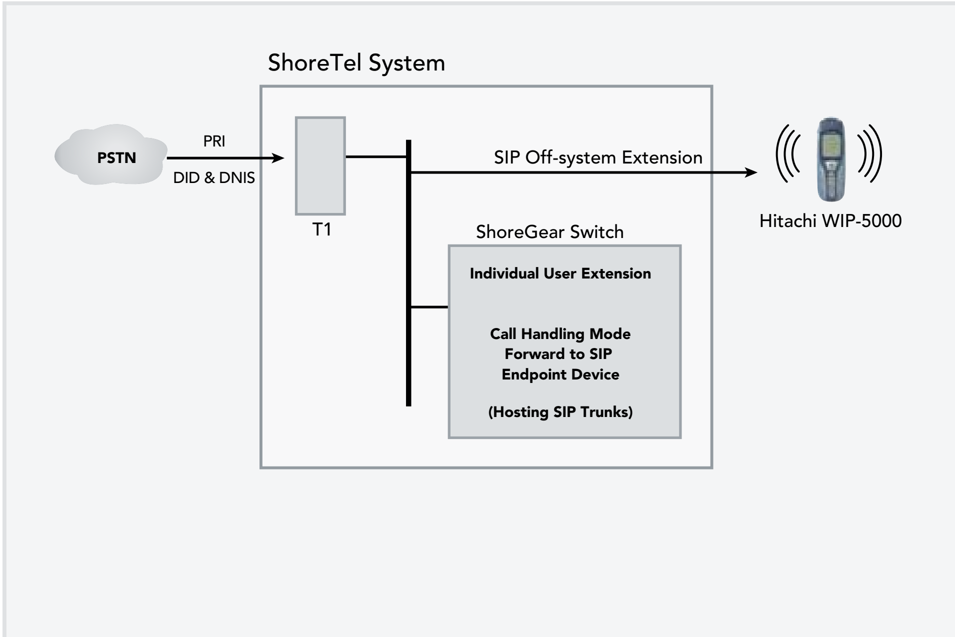
2. Routing an Internally dialed extension to the Hitachi Wireless IP-5000 extension (off system extensions are defined in the trunk group). The selection of SIP trunks is shown again in the snapshot of the "Edit ShoreGear Switch" screen.




3. Routing through the Auto Attendant can be accomplished in two ways.
   a. Program "Multiple digits" field in menu to "Go to extension" operation for routing to Hitachi Wireless IP-5000 extension.
   b. Program digit "0" to "Go to extension" operation for routing to Operator extension. Operator will have to transfer the call to the Hitachi Wireless IP-5000 extension.

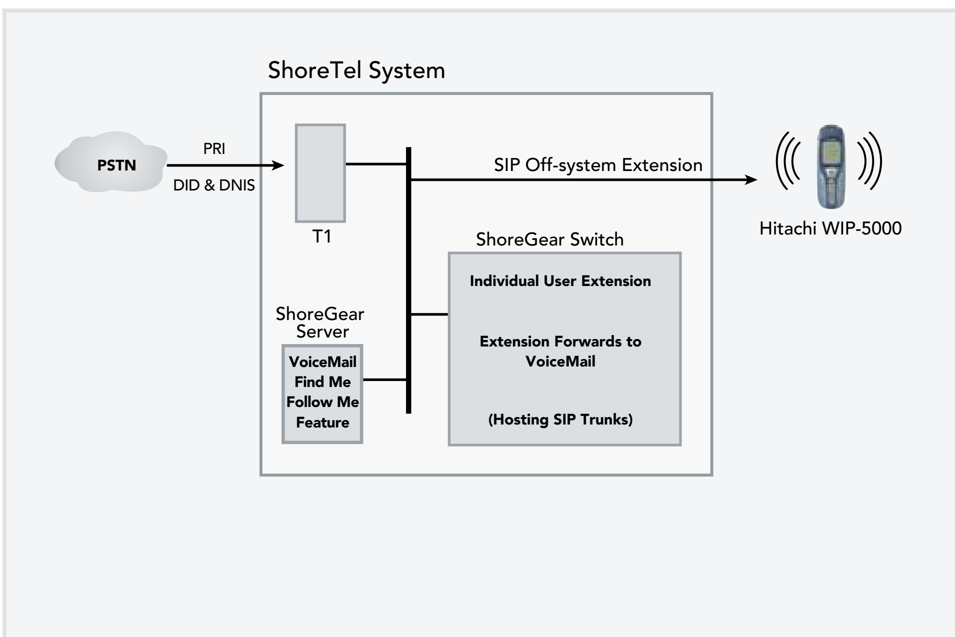


4. Routing through Individual User Extension with the Call Handling Mode set to forward (external destination) enter Hitachi Wireless IP-5000 extension.




5. Routing through Individual User Extension with extension forwarded to Voice Mail. The "Find Me Follow Me" feature of voice mail can be used to forward call (external) to Hitachi Wireless IP-5000 extension.

**Note:** Using this design allows the call to be pulled back to the ShoreTel system should the phone not be answered!



## Requirements, Certification and Limitations

The following requirements are necessary to integrate a Hitachi Wireless IP-5000 handset to the ShoreTel IP Phone system as described in this Application Note.

### ShoreTel Requirements

- ShoreWare Server Software, ShoreTel 6 or higher. Versions prior to this release will not support the Hitachi product.
- ShoreTel SIP Trunk port licenses are required.

### Hitachi Wireless IP-5000 Requirements

- Hitachi Wireless IP-5000 SIP handset – this device should be running the latest firmware (see version support table below).

### WiFi (Wireless) Network Requirements and Recommendations

See application note titled: MobileVoice_AppNote.final.pdf

This application note covers topics such as WiFi, QoS, security and more. The way WiFi is deployed in a infrastructure will have an impact in the voice quality.

### Certification

## Interoperability Test Program for VoIP Endpoints Test Results Overview

### VoIP Endpoint Product Information

| | |
|---|---|
| **Report Date** | May 22, 2006 |
| **Vendor Name** | ShoreTel |
| **Product Name** | WIP5000 SIP Phone |
| **Product Model** | WIP5000 |
| **Product Release** | Firmware Ver 2.2.1 |
| **Date Range of Tests Performed** | 04/23/06 - 04/26/06 |

## Test Results Overview

This section presents an overview of the results of all the VoIP Endpoint test cases available in this test plan template.

| Passed | ID | Optional? | Name | Description |
|---|---|---|---|---|
| Passed | 1.1 | Mandatory | 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 |
| Passed | 1.2 | Mandatory | Device reset – idle (for static configurations) | Verify successful re-initialization of device after power loss while device is idle |
| Passed | 1.3 | Mandatory | Device initialization with DHCP | Verify successful startup and initialization of the device up to a READY/IDLE state using DHCP |
| Passed | 1.4 | Mandatory | Device reset – idle (for dynamic configurations) | Verify successful re-initialization of device after power loss while device is idle |
| Passed | 1.5 | Mandatory | Verify Diffserv Code Point support | Verify the ability to set Diffserv Code Point from SIP DUT and verify via inspection of packet capture |
| Passed | 1.6 | Optional | Verify Date and Time Update support | Verify setting of Date and Time Update on SIP DUT. Time Zone can be updated by using the web interface |
| Passed | 1.7 | Mandatory | Place call | Verify successful call placement with normal dialing to a variety of terminating phones |
| Passed | 1.8 | Mandatory | Receive call | Verify successful reception of calls with normal dialing from a variety of calling phones |
| Passed | 1.9 | Optional | Place call – re-dial | Verify successful call placement using re-dial to SIP Reference |
| Passed | 1.10 | Optional | Place call – speed dial | Verify successful call placement using programmed speed dial |
| Passed | 1.11 | Mandatory for G.711, Optional for other CODECs | CODEC support – common (from DUT to ShoreTel Phone, REF-x) | Verify successful call connection and audio path using all supported CODECs (G.711-Ulaw and G.729) |
| Passed | 1.12 | Mandatory for G.711, Optional for other CODECs | CODEC support – common (from DUT to SIP Reference Phone, SIP-Ref) | Verify successful call connection and audio path using all supported CODECs (G.711-Ulaw and G.729) |
| Passed | 1.13 | Mandatory (only if more than 1 CODEC is supported) | CODEC support – negotiated | Verify successful negotiation between devices configured with different default CODECs (G.711-Ulaw and G.729) |
| Passed | 1.14 | Mandatory | Hold from DUT to SIP Reference | Verify successful hold and resume of connected call |
| Passed | 1.15 | Mandatory | Hold from DUT to ShoreTel Phone | Verify successful hold and resume of connected call |
| Passed | 1.16 | Mandatory | Forward | Verify successful forwarding of incoming calls |
| Passed | 1.17 | TBD | Forward from SIP DUT | Verify successful forwarding of incoming calls |
| Passed | 1.18 | Optional | Mute | Verify device's mute function |
| Passed | 1.19 | Mandatory | Out-of-band / In-band 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 |
| Passed | 1.20 | Optional | Missed call notification | Verify that device notifies the user about missed calls |
| Passed | 1.21 | Optional | Volume | Verify the device's volume adjustment function |

*Table 1 1: Basic Feature Test Cases*

| Passed | ID | Optional? | Name | Description |
|---|---|---|---|---|
| Passed | 2.1 | Mandatory | Speech quality – Minimal impairment | Verify acceptable voice quality between two parties with minimal network impairment condition |
| Passed | 2.2 | Mandatory | Speech quality – Moderate Impairment | Verify acceptable voice quality between two parties with low-to-moderate artificial network impairment condition |
| Passed | 2.3 | Mandatory | Speech quality – High Impairment | Verify acceptable voice quality between two parties with moderate-to-high artificial network impairment condition |

*Table 1 2: Performance Test Cases*

| Passed | ID | Optional? | Name | Description |
|---|---|---|---|---|
| Passed | 3.1 | Mandatory | Call waiting | Verify appropriate notification and successful connection of incoming call while busy with another party |
| Passed | 3.2 | Mandatory | Park | Verify successful park and retrieval of connected call |
| Passed | 3.3 | Optional | Extended forward | Verify extended call forwarding options – busy forwarding, no-answer forwarding |
| Not Tested | 3.4 | Not Supported | Extended forward from SIP DUT | Verify extended call forwarding options – busy forwarding, no-answer forwarding |
| Passed | 3.5 | Mandatory | Transfer – blind | Verify successful blind transfer of connected call |
| Passed | 3.6 | Mandatory | Transfer – monitored | Verify successful monitored transfer of connected call |
| Failed | 3.7 | Mandatory | Conference – ad hoc | Verify successful ad hoc conference of three parties |
| Passed | 3.8 | Optional | Place call – secondary line | Verify successful call placement using secondary line |
| Passed | 3.9 | Optional | Receive call – secondary line | Verify successful connection of incoming call on secondary line |
| Passed | 3.10 | Optional | Callback | Verify successful connection of a call using the missed-call callback feature of the device |
| Passed | 3.11 | Optional | Headset | Verify the device's support for external headsets (using headsets supplied by the 3P phone vendor) |
| Passed | 3.12 | Optional | Ring selection | Verify the device's ability to change the ring type |
| Passed | 3.13 | Mandatory | Caller ID Name and Number | Verify that Caller ID name and number is sent and received from SIP endpoint device |
| Passed | 3.14 | Mandatory | SIP Device Generates Busy Tone. | Verify that SIP DUT generates busy tone when calling a busy extension |
| N/A | 3.15 | Optional | POTS Analog Gateway supports the transfer operation by "flashing" | Verify that the POTS Analog Gateway can support the transfer operation by "flashing" |
| Passed | 3.16 | TBD | Verify handling of "911" | Verify dialing "911" on DUT could connect with "911" services |
| N/A | 3.17 | N/A | Verify Fax Handling | Verify that fax can be sent and received through DUT |

*Table 1 3: Extended Feature Test Cases*

## Known Limitations

Office Anywhere is not supported with the ShoreTel Release 6.1. A future release will support Office Anywhere.

Currently these are known issues at the time of this release. These issues will be addressed in the future!

## 1-19429394 (Blind Transfer of Hitachi Call from IP Phone Inconsistency)

Blind Transfer isn't currently working in the following scenario, will be resolved in near future: For the case when there is a call up between an IP Phone and a Hitachi SIP Phone, and where the IP phone initiates a Blind Transfer to another IP Phone, the user of the Transferor IP Phone will be able to incorrectly be able to access the call icon to try and bring back the call which at this point will cause the Blind Transfer to fail.

More Details Notes about the problem:

1. This problem is logged as 1-19429394
2. This problem only happens with Hitachi Phone, where IP Phone A tries to blind transfer the Hitachi phone to another IP phone B in it's network/system.
3. After hitting the blind transfer on the IP phone A, ringing will start on IP phone B. At this point in time the IP phone A shows the call as still active, and the user can try to pull the call back up. If the user tries to pull the call back up, the transfer will fail between the Hitachi Phone and IP phone B. This is a bit of a corner case.
4. If we replace the Hitachi phone with another SIP device, or with an IP phone, the call icon on IP Phone A will correctly show that the transfer has completed.

## Interaction with "911" Support

With the release of ShoreTel 6 support for 911 calls is limited. If a 911 call is made whatever number is configured for that SIP device is what will be sent out the local trunk which services 911 calls based on the "Site" of the ShoreGear switch which is associated for the SIP Trunk.

Other items to consider:

- If the SIP device is ONLY configured for example with a four digit extension then that is number which will be sent! Depending how the trunk from the CO is configured it may default to the billing address or something else. Check with your phone company on how 911 calls are handled.
- If the SIP device is configured with a full 10 digit DID then once again that's what will be sent out the local trunk where the ShoreGear Switch is located supporting the SIP Trunk.
- Should the SIP device be in NY and the ShoreGear switch which supports the SIP Trunk for the SIP device in San Jose then the 911 call will go out through whatever 911 trunk is configured for the San Jose site!
- It is recommended 911 is fully tested for based on the design!
- Should it be desired to use 10 digit DID, using "Digit Translation" with "Off System" extensions can be used. Example: Dialing 3510 can translate to 408-331-3510. This is only needed when configuring the "Out Bound" portion of digit translation in the Trunk Group. See the Planning and Installation Guide for information on configuring Digit Translation.

## Version Support

Product certified via the Technology Partner Certification Process for the ShoreTel system. Table below contains the matrix of Hitachi Wireless IP-5000 handset firmware releases certified on the identified ShoreTel software releases.

| | **Hitachi Wireless IP-5000 Handset** |
|---|---|
| **ShoreTel Software Release** | **Firmware Version** |
| | v2.1.11 |
| ShoreTel 6.1 | ✔ |

## Configuration Overview

The following steps are required to configure the Hitachi Wireless IP-5000 handset to work with the ShoreTel system:

1. Configure General ShoreTel system settings
   a. Call Control Options, Site and Switch settings
2. Setup Trunk Groups
3. Setup Individual Trunks and User Groups
4. Configure Hitachi Wireless IP-5000 to function with the ShoreTel system
5. Configure Appropriate Call Routing options

## ShoreTel Configuration

This section describes the ShoreTel system configuration to support the Hitachi WiFi handsets. The section is divided into general system settings and trunk configurations (both group and individual) needed to support the Hitachi Wireless IP-5000 device. Think of Trunk Groups as what rules are applied for in / out bound calls and think of Individual trunks as the channels for the call.

### 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 site and the switch settings. If these items have already been configured on the system, skip this section and go on to the "ShoreTel System Settings – Trunk Groups" section below.

### Call Control Settings

The first settings to configure within ShoreWare Director are the Call Control Options. To configure these settings for the ShoreTel system, log into ShoreWare Director and select "Administration" then "Call Control" followed by "Options" (Figure 1).


*Figure 1 Administration Call Control Options*

The "Call Control Options" screen will then appear (Figure 2).

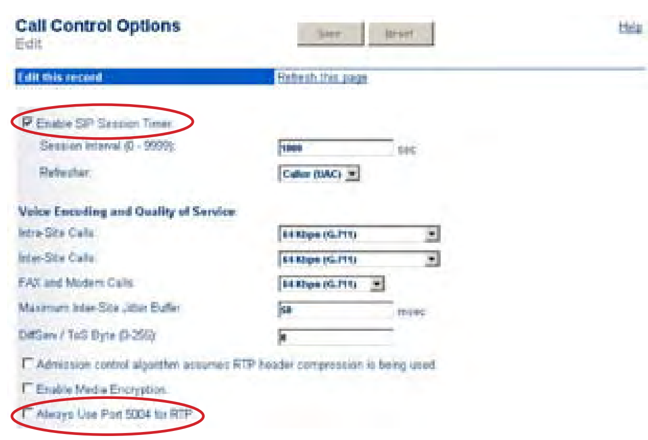



*Figure 2 Call Control Options*

Within the "Call Control Options" screen confirm the appropriate settings are made for the "Enable SIP Session Timer", "Intra-Site Calls", "Inter-Site Calls" and "Always Use Port 5004 for RTP" fields.

The first step is to ensure the box is checked for the "Enable SIP Session Timer". Next the Session Interval Timer needs to be set. The recommended setting for "Session Interval" is 1800 seconds. The last item to select is the appropriate refresher (from the pull down menu) for the SIP Session Timer. The "Refresher" field will be set either to "Caller (UAC)" [User Agent Client] or to "Callee (UAS)" [User Agent Server]. If the "Refresher" field is set to "Caller (UAC)" the Caller's device will be in control of the session timer refresh.

The next settings to verify are the "Intra-Site Calls" and the "Inter-Site Calls" settings under the" Voice Encoding and Quality of Service" prompt. For the Intra-Site Calls verify the desired audio bandwidth is selected for the CODEC for calls within the system. The settings should then be confirmed for the desired audio bandwidth for the CODEC for Inter-Site calls (calls between sites).

**Note:** SIP uses both G.711 and G.729 CODECs. The CODEC setting will be negotiated to the highest CODEC supported.

Un-checking the box for "Always Use Port 5004 for RTP" is required for implementing SIP on the ShoreTel system. For SIP configurations, Dynamic UDP must be used for RTP Traffic. Note: If the box is unchecked MGCP will no longer use UDP port 5004; MGCP and SIP traffic will use dynamic UDP ports. It is also important to note that this "one time" setting requires a system (IP phones, ShoreGear Switches & Servers) reboot to take effect.

## 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 3).


*Figure 3 Administration Site*

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. The only change required to the "Edit Site" screen is to the "Admission Control Bandwidth" field (Figure 4).


*Figure 4 Admission Control Bandwidth*

The Admission Control Bandwidth defines the bandwidth available to and from the site. This is important as SIP devices will be counted against the site bandwidth. See the *ShoreTel Planning and Installation Guide* for more information. When a call is "Intra-site" it will not count against the "Admission Control Bandwidth" unless "Teleworker" is checked in the "Trunk Group Page".

## Switch Settings - Allocating Ports for SIP Trunks

The final general settings to input are the ShoreGear switch settings. These changes are modified by selecting "Administration" then "Switches" in ShoreWare Director (Figure 5).


*Figure 5 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 select the desired number of SIP trunks from the ports available (Figure 6).

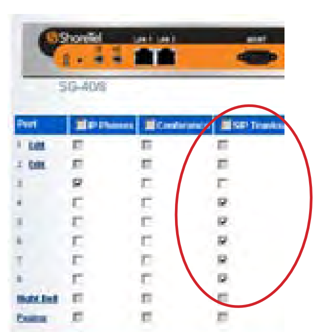


*Figure 6 ShoreGear Switch Setting*

Each Check box designated as a SIP trunk enables the support for 5 individual trunks.

## ShoreTel System Settings - Trunk Group

ShoreTel Trunk Groups support both Dynamic and Static SIP end point Individual Trunks.

**Note:** A ShoreGear switch can only support one Trunk Group with Dynamic IP addressing.

In trunk planning a couple of things need to be considered.

1. Are the SIP devices using DHCP or Static IP?
2. Are the SIP devices endpoints (like ATAs, Conference Phone or WiFi handset) or non-endpoint devices like an ISP?

If the SIP Trunk Groups have already been configured on the system, skip down to the "ShoreTel System Settings - Individual Trunks" section. The settings for Trunk Groups are changed by selecting "Administration" then "Trunks" followed by "Trunk Groups" within ShoreWare Director (Figure 7).

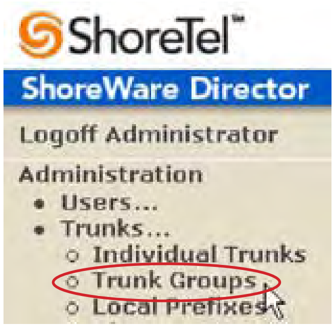



*Figure 7 Administration Trunk Groups*

This selection brings up the "Trunk Groups" screen (Figure 8).

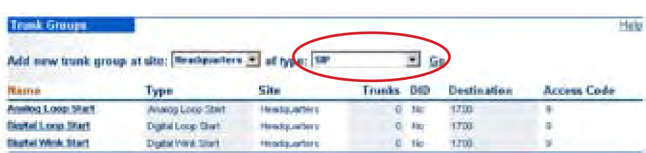



*Figure 8 Trunk Groups Settings*

From the pull down menus on the "Trunk Groups" screen select the "site" desired and then select "SIP" and click on the "Go" link. The "Edit SIP Trunk Group" screen" will appear (Figure 9).


*Figure 9 SIP Trunk Group Settings*

Determine whether the trunks need to be configured as inter-site trunks (trunks between sites) or intra-site trunks (trunks within a site). In the "Edit SIP Trunks Group" screen, input a name for the trunk group. When naming the trunk group it is recommended to include the trunk type and the extension or extension range of the trunk group being established. Having this detail in the name is a convenient way to recall the trunk type and extension range. In the example in Figure 9 the name "SIP x3000-3019 Intra-site" has been created. This name indicates that a SIP Intra-site trunk with twenty off system extensions in the range of 3000-3019 has been created.

**Pick up here** - The next step is to choose the setting of the "Teleworkers" check box.

- By NOT checking Teleworker no bandwidth usage will count against the site and the Intra-Site codec will be used when calls are made Intra-Site.
- If the Teleworker checkbox "is" checked then the call will be counted against the allocation of bandwidth and the Inter-Site codec will be used.
- Should Teleworker not be checked and the call goes from San Jose (location of the SIP Trunk) to another site in NY then Inter-Site will apply.
- Note: Location of the SIP Trunk NOT the SIP device is what's used to determine the location of the call.

The next item on this screen is the "Enable Digest Authentication" field. This is a SIP feature that allows a user ID and password to be established for authentication. If this feature is configured for the Trunk Group the same settings must be configured on the SIP endpoint devices.

The “Enable SIP Info for G.711 DTMF Signaling” box should not be checked.  Enabling SIP info is currently only used with SIP tie trunks between ShoreTel systems.

The next item to change for the SIP trunk groups is the handling of the inbound trunk configuration (Figure 10).


*Figure 10  Inbound Trunk Configuration*

For the Inbound Trunk Settings ensure the “Number of Digits from CO” is correct.  Select the “DNIS” (Dialed Number Identification Service) box to create a DNIS to extension mapping.  Select the “DID” (Direct Inward Dial) box to input a DID range.  The next step is to make sure the “Extension” box is checked along with the “Translation Table” button along with the translation table from the pull down menu. The last step for the Inbound Trunk configuration is to select the “Tandem Trunking” box if needed.  Selecting this box will allow trunking between trunk groups. The “Destination:” contains the number for the Auto Attendant.  Trunk calls not routed by previous entries under the Inbound Trunk routing will be routed to the Auto Attendant.

The next item to change for the SIP trunk groups is the handling outbound trunk configuration (Figure 11).

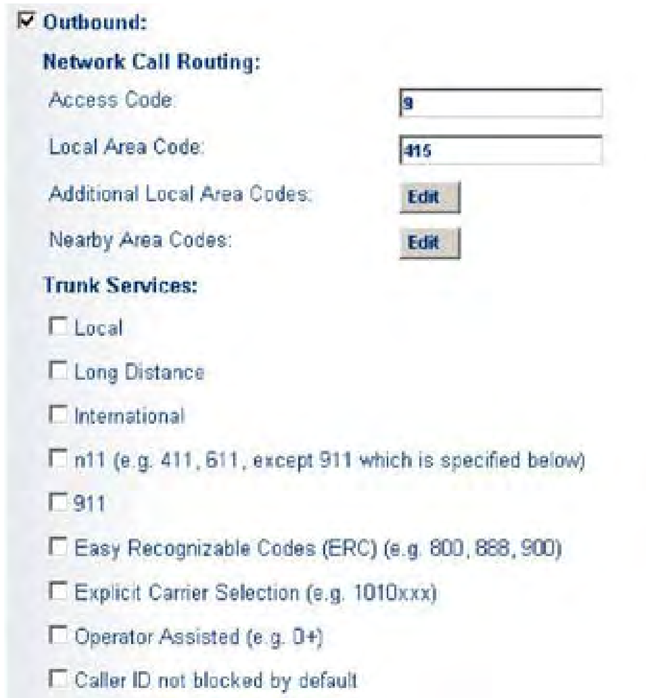



*Figure 11 Outbound Trunk Configuration*

For the Outbound Trunk Configuration set the “Access Code” and “Local Area Code” fields for the system.  For the “Trunk Services” boxes, typically none of these will be selected.

**Note:** Remember, these are calls being sent from the ShoreTel system to the SIP device.

After these settings are made press the “Save” button to input the changes.

The next item to change is under "Trunk Digit Manipulation" (Figure 12).

Typically when sending digits to the "Hitachi" device it is not needed to check the below three check boxes for "Remove leading 1 from 1+10D", "Remove leading 1 for Local Area Codes" and "Dial 7 digits for Local Area Code'. Nothing should need done for the "Local Prefixes" drop down.


*Figure 12 Off System Extension*

Selecting the "Edit" button for the Off System Extensions will bring up the"Off System Extension Ranges" dialog box (Figure 13).

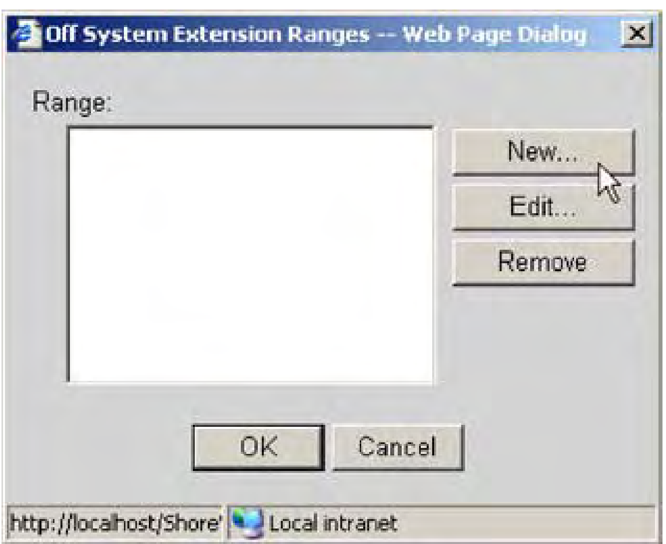



*Figure 13 Off System Extension Ranges*

Selecting the "New" button will bring up the "New Range" dialog box (Figure 14).

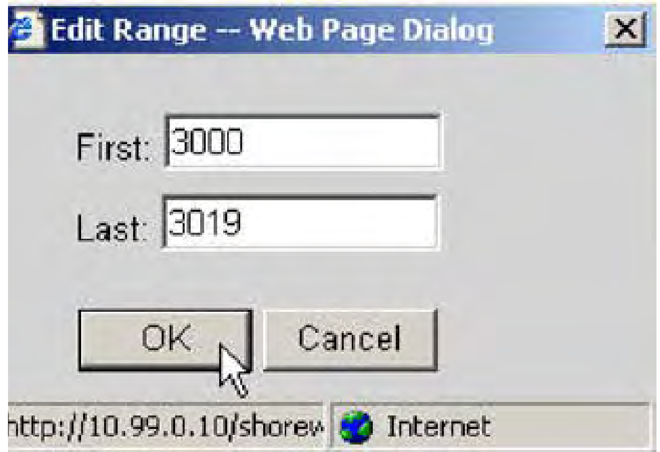



*Figure 14 New Range Settings*

Input a new range of off system extensions for the SIP devices and click "OK". For this example the extension range 3000-3019 has been input, which corresponds to the naming convention used in the previous example. This completes the settings needed to set up the SIP Trunk Groups on the ShoreTel system.

**Note:** It is important to note that only one dynamic trunk group can be set up per ShoreGear switch. If another dynamic trunk group is desired another ShoreGear switch will be needed.

When it is desired to translate an off system extension to a 10 digit DID "Digit Translation must be used. For example, when a call is placed to a Hitachi phone by dialing ext. 3000 but the Hitachi device is configured with 4083313000 then Digit Translation needs to be used. Main reason for doing this is when it's necessary for calls made from the Hitachi device to present a 10 digit number to the person being called. It can also be needed for 911 reasons (see above section which covers 911 for more information).

## ShoreTel System Settings - Individual Trunks

This section covers the configuration of the individual trunks. Select “Administration” then “Trunks” followed by “Individual Trunks” to configure the individual trunks (Figure 15).


*Figure 15 Individual Trunks*

The “Trunks by Group” screen, used to change the individual trunks settings, then appears (Figure 16).

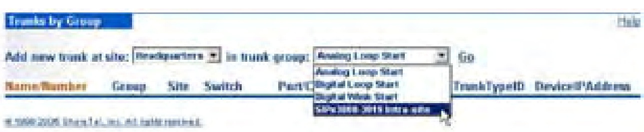


*Figure 16 Trunks by Group*

Select the site for the new individual trunk(s) to be added and select the appropriate trunk group from the pull down menu. In this example the site is “Headquarters” and the trunk group is SIP x3020 Intra-site”. Click on the “Go” button to bring up the “Edit Trunk” screen (Figure 17).

*Figure 17 Edit Trunks Screen for Individual Trunks*

From the individual trunks “Edit Trunk” screen, input a name for the individual trunks, select the appropriate switch, select the SIP trunk type and input the number of trunks. When selecting a name, the recommendation is to name the individual trunks the same as the name of the trunk group so that the trunk type and extension range can easily be tracked. Select the switch upon which the individual trunk will be created. For the SIP Trunk Type decide whether the trunks are to be configured as dynamic or static. Dynamic trunks are typically configured for endpoint devices like wireless handsets, conference phone or Integrated Access Devices (IADs). For the “SIP Trunk Type” field select either “Dynamic” or for a static configuration select “Use IP Address” button and input an IP address. In this example a dynamic SIP trunk type has been chosen. The last step is to select the number of individual trunks desired. In the example 20 trunks were chosen, which matches the naming convention used. Once these changes are complete, select the “Save” button to create the list of individual trunks (Figure 18).

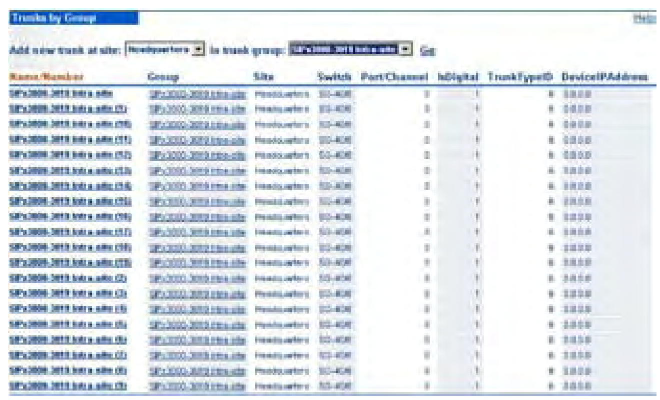


*Figure 18 Trunks by Group*

**Note:** Individual SIP trunks cannot span networks. SIP trunks can only terminate on the switch selected. There is no failover to another switch.

After setting up the trunk groups and individual trunks refer to the ShoreTel Product Installation Guide to make the appropriate changes for the User Group settings. This completes the settings for the ShoreTel system side.

# Hitachi Wireless IP-5000 Configuration

For certain portions of the Hitachi Wireless IP-5000 configuration it may be necessary to refer to the documentation provided for the Hitachi wireless handset. The document "WirelessIP5000_Administrator_manual.pdf "can be obtained from the "Vendor Overview and Contact" section of this document.

The Hitachi handsets can be configured manually or by using the user.ini (Advanced feature) configuration file (Example in Appendix A). The configuration examples in this document are based on the manual configuration of the Hitachi handsets.

**Note:** The Hitachi handset can not be configured 100% via the handset. It does require a few configurations which can only be changed through the web interface for the handset or the user.ini file.

## Hitachi Handset Connection to WiFi Network

Refer to the documentation to perform the necessary wireless network configuration. Consult with the system administrator, if necessary, to setup the handsets to operate on the wireless network.

## Retrieving Handset IP Address

Before connecting to the handsets web interface, obtain the handsets IP Address. Perform the following steps on the Hitachi handset to obtain the IP address:

1. Press the Menu softkey
2. Select item #5 - Setup
3. Select Item #6 - Information
4. Select item #1 - TCP
5. Write down the information displayed in the "IP Address" field ______________________________
6. Push the "END" key on the phone to exit out of the menus.

## Connecting to the Handset Web Interface

Access the "web interface" for the Hitachi handset by opening a web browser and entering the IP address (previously obtained) for the handset as follows:

http://<IP address>:8080
Example: http://10.0.1.51:8080

## Manual Web Interface Configuration

Once the IP address has been input in the web browser a "User ID" and "Password" prompt will be presented. The default "User ID" is "admin" and the default "Password" is "000000" (6 zero's), the "Wireless IP-5000 Web Configuration Tool" screen for the Hitachi handset will appear (Figure 17).

| Software Specification | |
|---|---|
| Model | WirelessIP5000 |
| Software Version | v2.1.11 |
| IP Address | 10.9.0.103 |
| Netmask | 255.255.255.0 |
| Gateway | 10.9.0.1 |
| MAC Address | 00:03:2A:00:70:A5 |

*Figure 17 WirelessIP5000 Web Configuration Tool*

# Wireless IP5000 Web Configuration Tool

There are four main sections to address when using the web configuration tool. These sections are:

1. Configuration
2. System Setup
3. Network Setup
4. Download Configuration File

This documentation will cover items needing configured or validated for use with the ShoreTel system only. For other areas of configuration please consult the Hitachi documentation. For example this document will not go over how to configure the WiFi interface to communicate with the network. This document will however tell the user how to configure DTMF tones to work with the ShoreTel system.

## Software Version Verification

Confirm the appropriate version of software is used to begin the configuration process by checking the software version detailed under the "Software Specification" heading of the "Wireless IP-5000 Web Configuration Tool" screen (Figure 17).

Verify the version shown is "v2.1.11" or newer. If the firmware is not an appropriate version, see the "Load and Upgrade" option (for firmware update) under the "System Setup" section, consult the Hitachi documentation for further information.

The Hitachi phone can be configured in three primary way's.

1. Via the UI on the actual phone (not covered in this document).
2. Via a web browser interface (as shown in this document) or
3. Via a "User.ini" file which is uploaded into the phone example config provided in appendix A (details not covered in this document, refer to the Hitachi documentation for more information).

## Web Based Configuration

To change the configuration, select the "Configuration" link from the "Main Page" screen.

The "System Configuration" window will appear (Figure 18).

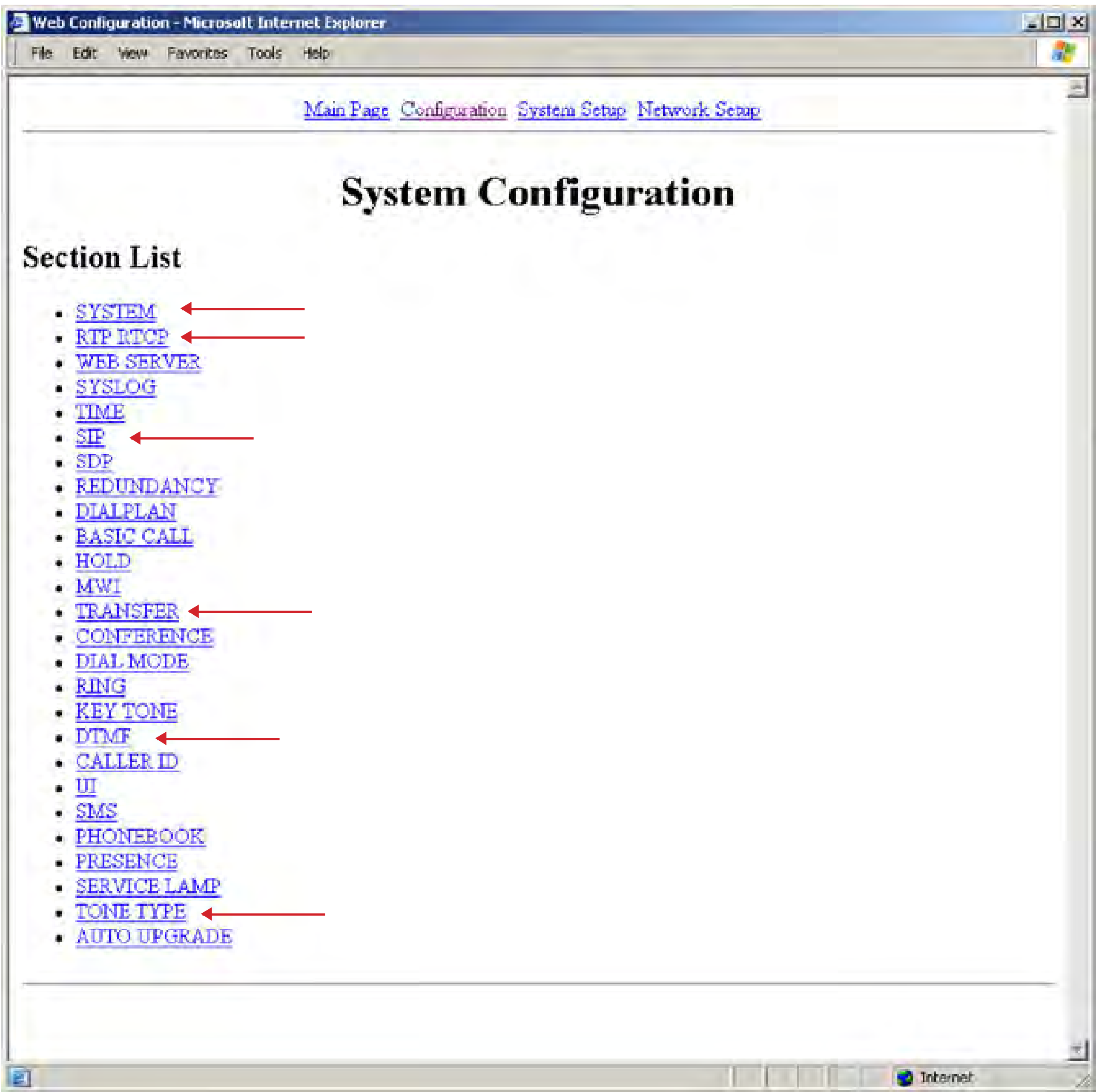



*Figure 18 System Configuration*

Within the "System Configuration" ShoreTel only documents items that pertain to the ShoreTel system. Consult the Hitachi documentation for configuration of other features in the Hitachi WiFi handset that don't pertain to the ShoreTel system. Changes need to be made to the following areas under "System Configuration", they are: System, RTP RTCP, SIP, Transfer, Conference, DTMF and Tone Type.

## System

In the system configuration page for English input the value of "1" (Figure 19). See Hitachi documentation other language value options.

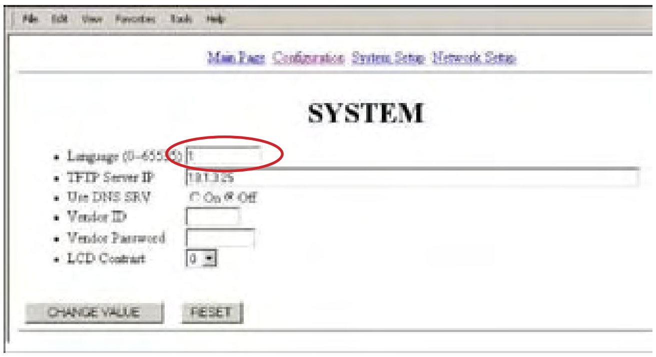



*Figure 19 Choosing the language*

## RTP RTCP

Configure the first RTCP value to "off" (Figure 20).

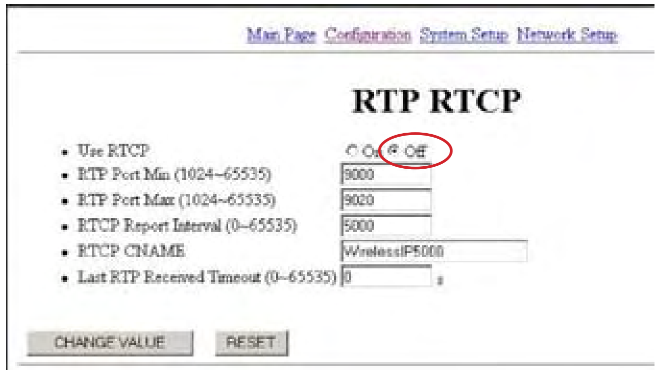



*Figure 20 Turn RTCP off*

## SIP Configuration

From the SIP page all default values should be fine, just make sure that the local port is set to "5060" as shown in Figure 21.

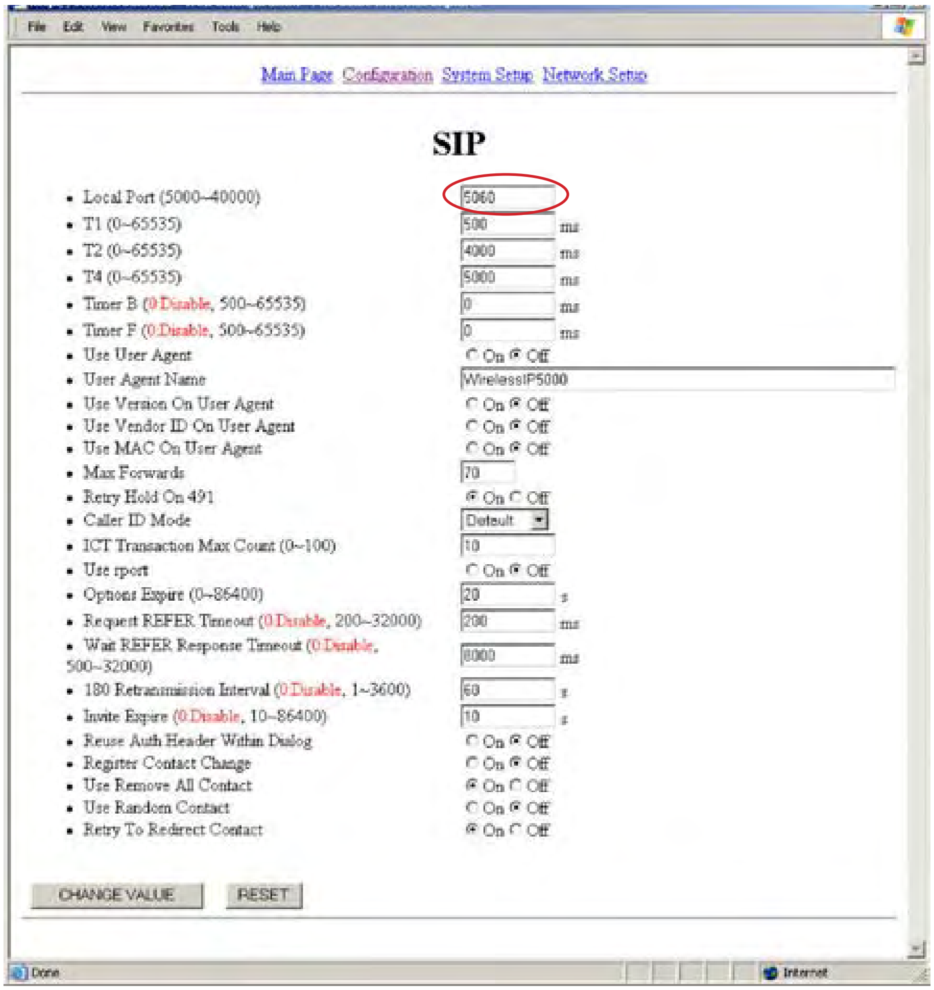



*Figure 21 Confirm "Local Port 5060"*

## Transfer

In order for "Transfer" to work correctly it's important all the values are set correctly. Confirm the following is set: Transfer Target hold "On", Consultation Transfer "Off" and all others should be marked "On" (Figure 22).

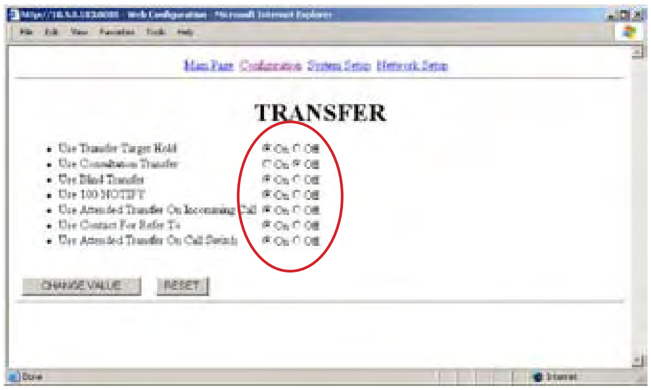



*Figure 22 Transfer*

## Conference

Conference should be set to "off" under "Use Conference" (Figure 23).

**Note:** From the user.ini file (Appendix A) insure the value "Use_Consultation_Transfer" is set to "0". Example: Use_Consultation_Transfer=0.

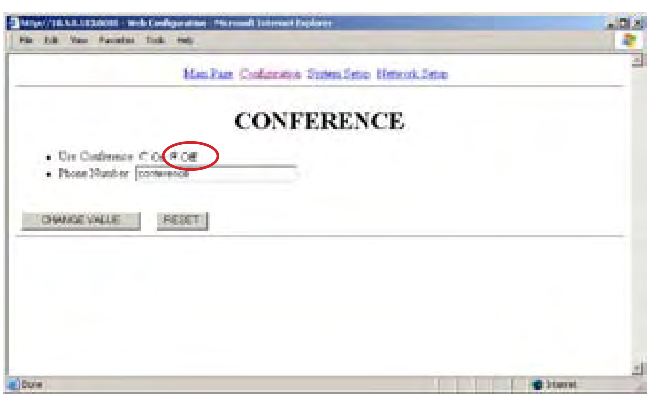



*Figure 23 Conference*

## DTMF

DTMF must be set correctly and tested to insure proper DTMF functionality. Without DTMF gaining access to Voice Mail or navigating an Auto Attendant may not work correctly. Confirm the following values are configured. Mode = RFT2833, Duration = 100 ms, RFC 2833 Volume = 10, RFC2833 Payload Type = 102 and Enable Auto DTMF Mode = "ON" (Figure 24).

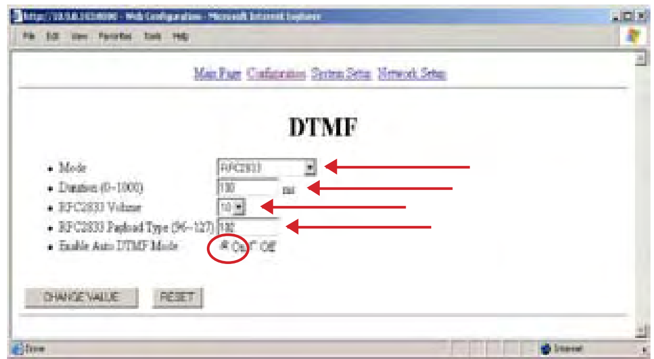



*Figure 24 DTMF Values*

## Tone Type

Another important section that needs configured is the Tone Type. Make sure Dial Tone Type On Idle = CDT, Dial Tone Type on Hold = Silence and Send Dial Tone Type = SDT (Figure 25).

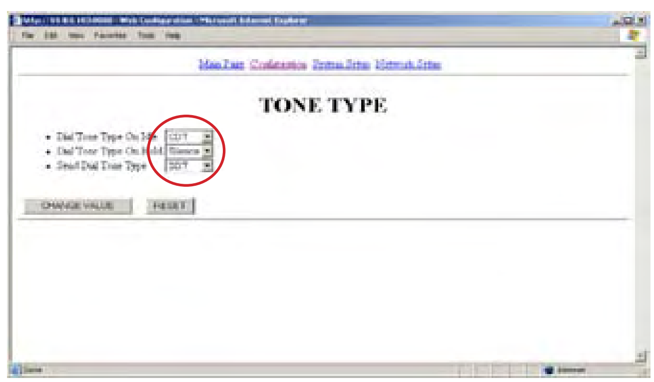



*Figure 25 Tone Type*

## System Setup

"System Setup" section of the Hitachi web interface consist of three areas "Load & Upgrade, Change Password and WebServer Stop" (Figure 26). Load and upgrade provides the ability to install new firmware or configurations onto the Hitachi phone. Change Password will do just that… Web-Server Stop will disable the web interface used to access the phone. If by chance this is disabled it will be necessary to use the actual phones UI to re-enable the phones web interface. Consult the Hitachi documentation for additional information.


*Figure 26 System Setup*

## Network Setup

This is a very important section when it comes to configuring the Hitachi WiFi phone for use with the ShoreTel system. In this section the SIP Registration (Figure 27) needs to be focused on most for configuration of the SIP Proxy / Register and User Account information. The Network Config. section is standard network setup configuration, one thing to note in this section is the ability to add "multiple" Network Configurations and the setting of an Outbound Proxy. This is important when moving for example from a "work" network to a "home" network. Most networks can very from network to network in the configuration. Example could be as simple as the WEP key or SSID used by each network might be different. Then when the user is using the Hitachi WiFi phone they can simply either have the phone auto find or manually select the network of choice. All other sections see the Hitachi documentation for additional information.

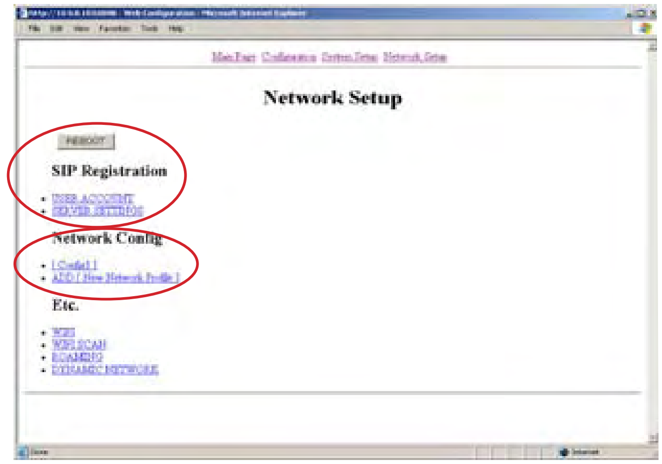



*Figure 27 Network Setup*

## SIP Registration

Configure both the User Account information as well as the Server Settings section.

## User Account

In the user account four main fields exist (Figure 28). First is the Display Name, the display name will be displayed on the phone when supported. Then the Phone Number field, this is the extension or DID assigned to this phone. User ID and User Password are only needed when "Digest Authentication" has been configured for the trunk group on the ShoreTel system. During initial testing we recommend leaving this field blank, getting your audio working then adding this to the phone. Should a problem arise this makes troubleshooting much easier! URL Scheme set this value for "SIP".

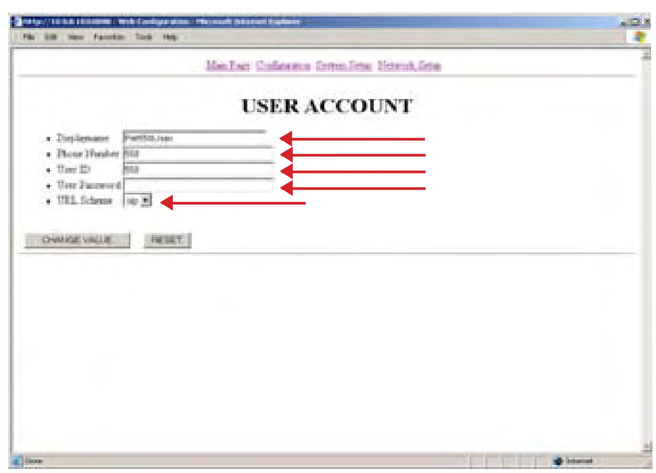



*Figure 28 User Account*

## Server Settings

Server settings section (Figure 29) of the configuration is another important configuration area. From this screen the SIP Proxy, SIP Registrar (optional 2nd) and Registrar Expire are configured. With regards to Domain Realm and Register Retry Backoff Interval settings just leave defaults or refer to the Hitachi documentation.

Settings:

- 1st Proxy set to IP Address of ShoreGear switch with associated Individual SIP Trunks.
- 1st Registrar set to IP Address of ShoreGear switch with associated Individual SIP Trunks.
- Register Expire default is 3600. Network Profile

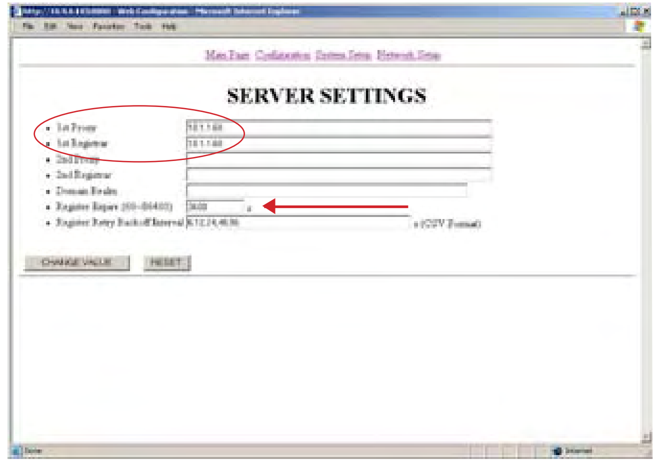



*Figure 29 Server Settings*

## Network Profile

As with many sections of this document this too is an important section as this page controls how the device is able to join a particular network. This document will touch on a couple key areas that need consideration or configuration. For information on other sections please refer to the Hitachi documentation.

Configuration sections (Figure 30):

- Name: Provide a WiFi network name that is meaningful especially if there will be more then one.
- SIP Outbound Proxy: This parameter is typically used when the Hitachi is outside the LAN of where the ShoreGear switch is located with Individual Trunk for SIP. For example… take the Hitachi from the office and go home (or hotel etc… with WiFi), once on that network the Hitachi won't know how to communicate with the ShoreGear switch (SIP Proxy and Registrar). If the company has a box such as an "inGate" SIParator (think of this as a LAN to WAN proxy) connected back at the office where the ShoreGear switch is located it will proxy the SIP traffic to the appropriate ShoreGear switch.

**Note:** inGate (http://www.ingate.com) application note and certification are in progress at the time of this writing.

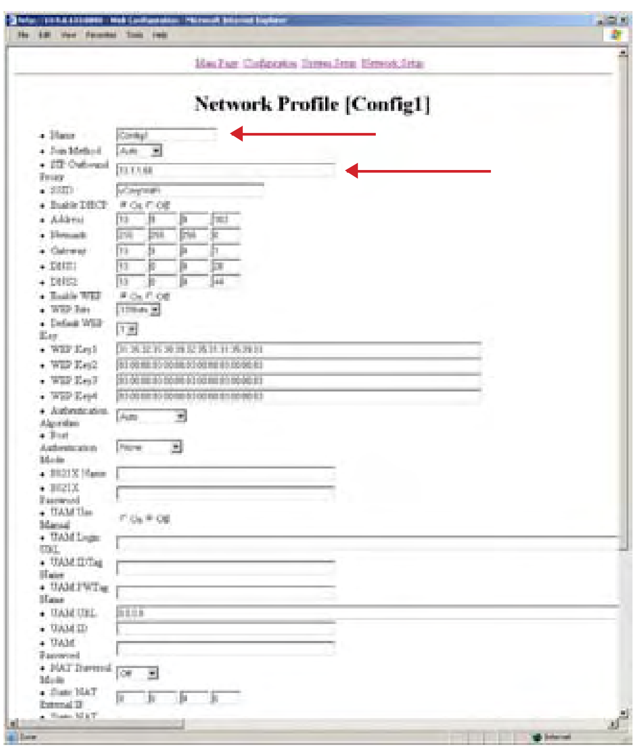



*Figure 30 Network Profile*

# Appendix "A" - User.ini example

The below configuration is an example of a user.ini file from a Hitachi WiFi phone running firmware 2.1.11

```
[SYSTEM]
Language=1
Admin_Password=
TFTP_Server_IP=10.1.3.25
Use_DNS_SRV=0
Vendor_ID=
Vendor_Password=
LCD_Contrast=0

[RTP_RTCP]
Use_RTCP=0
RTP_Port_Min=9000
RTP_Port_Max=9020
RTCP_Report_Interval=5000
RTCP_CNAME=WirelessIP5000
Last_RTP_Received_Timeout=0

[WEB_SERVER]
Use_WEB_Server=1

[SYSLOG]
Mode=0
Server_IP=0.0.0.0
Server_Port=514
Use_MACIP_Header=1

[TIME]
Enable_NTP=1
Date_Format=0
Time_Format=1
Time_Zone=-08:00
Enable_Daylight_Saving_Time=1
DST_Start_Month=3
DST_Start_Day=29
DST_Start_Hour=2
DST_Start_Min=0
DST_End_Month=10
DST_End_Day=29
DST_End_Hour=2
DST_End_Min=0
NTP_Refresh_Interval=7200
NTP_Server1=10.5.0.20
NTP_Server2=0.0.0.0

[SIP]
Local_Port=5060
T1=500
T2=4000
T4=5000
Timer_B=0
Timer_F=0
Use_User_Agent=0
User_Agent_Name=WirelessIP5000
Use_Version_On_User_Agent=0
Use_Vendor_ID_On_User_Agent=0
Use_MAC_On_User_Agent=0
Max_Forwards=70
Retry_Hold_On_491=1
Caller_ID_Mode=0
ICT_Transaction_Max_Count=10
Use_rport=0
Options_Expire=20
Request_REFER_Timeout=200
Wait_REFER_Response_Timeout=8000
180_Retransmission_Interval=60
Invite_Expire=10
Reuse_Auth_Header_Within_Dialog=0
Register_Contact_Change=0
Use_Remove_All_Contact=1
Use_Random_Contact=0
Retry_To_Redirect_Contact=1

[SDP]
Use_Increase_session_id=0
Use_Increase_version=1
Modified_Session_Detection=0
Session_Name=A_converstion
[USER_ACCOUNT]
Displayname=Pat550User
Phone_Number=550
User_ID=550
User_Password=
URL_Scheme=0

[SERVER_SETTINGS]
1st_Proxy=10.1.1.68
1st_Registrar=10.1.1.68
2nd_Proxy=
2nd_Registrar=
Domain_Realm=
Register_Expire=3600
Register_Retry_Backoff_Interval=6,12,24,48,96

[REDUNDANCY]
Mode=0
Request_Timeout=4000
Use_Fixed_Primary_Server=1
Use_DNS_Additional_Records=1

[DIALPLAN]
External_Prefix=0
Caller_ID_On_Prefix=186
Caller_ID_Off_Prefix=184

[BASIC_CALL]
Busy_Tone_Count=10
Call_Waiting_Tone_Count=1
Reject_Hold_Request_On_Hold=0
Reject_Hold_Request_On_2Calls=0
Block_Request_Hold_On_Holded=0
```

Session_Expire=90
Ringing_Timeout=180
Update_RBT=0
Dial_Sending_Timeout=30000
Check_Alias_In_Call_List=1
Remove_DASH_On_Alias=1
Use_Silent_Packet_On_Mute=0
[HOLD]
Mode=0
Use_Local_Hold_Tone=1
RTP_Hold_Multiframe=40

[MWI]
Use_MWI=1
Use_Subscribe=1
Subscribe_Server=
Subscribe_Expire=3600

[TRANSFER]
Use_Transfer_Target_Hold=1
Use_Consultation_Transfer=0
Use_Blind_Transfer=1
Use_100_NOTIFY=1
Use_Attended_Transfer_On_Incomming_Call=1
Use_Contact_For_Refer_To=1
Use_Attended_Transfer_On_Call_Switch=1

[CONFERENCE]
Use_Conference=0
Phone_Number=conference

[DIAL_MODE]
DTMF_Input_Timeout=20
UnHold_Indication_Timeout=30
Overlap_Send_Timeout=4
Use_Overlap_Send_on_Hold=0

[RING]
Use_Ring=1
External1_Ring_Type=11
External1_Mode=0
External1_Led=1
External2_Ring_Type=4
External2_Mode=0
External2_Led=1
Internal1_Ring_Type=12
Internal1_Mode=0
Internal1_Led=2
Internal2_Ring_Type=6
Internal2_Mode=0
Internal2_Led=2
Message_Mode=0
Message_Ring_Type=1
Message_Ring_Volume=3
ID_String_External1=
ID_String_External2=
ID_String_Internal1=
ID_String_Internal2=
ID_String_Silence=

[KEY_TONE]
Key_Tone_Volume=4

[WIFI]
Data_Packet_TxRate=0
Dot11_Packet_TxRate=0
Use_PowerSave=1
PowerSave_Wakeup_Period=28
PowerSave_Awake_Interval=400
Domain=0
Use_Statistics_Window=0
Statistics_Window_Update_Interval=10
Use_Meru_Extension=1
Short_Retry_Limit=7
Long_Retry_Limit=7
Preamble_Mode=0
RTS_Threshold=2347
Default_Listen_Interval=3
RxLevel_Association_Threshold=-69
Use_PowerSave_Monitor=1
Use_Supported_Rates_By_AP=0

[WIFI_SCAN]
Scan_Count=1
Scan_Channel_List=1,2,3,4,5,6,7,8,9,10,11,12,13,14

[ROAMING]
Try_RxLevel=-79
PreRoaming_Enable_RxLevel=-59
Try_Over_TxError_Count=10
Try_Over_RxError_Count=10
Level_Diff_Higher_Than_Curr_Site=10
Use_Refresh_PreRoaming=1
Enable_PreRoaming_On_Association=0
PreRoaming_Mode=0
PreRoaming_Refresh_Interval=0

[NETWORK1]
Name=Config1
Enable=1
Join_Method=0
SIP_Outbound_Proxy=10.1.1.68
SSID=vCorpWiFi
Enable_DHCP=1
Address=10.9.0.103
Netmask=255.255.255.0
Gateway=10.9.0.1
DNS1=10.0.0.20
DNS2=10.0.0.44
Enable_WEP=1
WEP_Bits=1
Default_WEP_Key=1
WEP_Key1=31:35:32:35:38:39:32:35:31:31:35:39:33
WEP_Key2=00:00:00:00:00:00:00:00:00:00:00:00:00
WEP_Key3=00:00:00:00:00:00:00:00:00:00:00:00:00
WEP_Key4=00:00:00:00:00:00:00:00:00:00:00:00:00
Authentication_Algorithm=0
Post_Authentication_Mode=0
8021X_Name=
8021X_Password=
UAM_Use_Manual=0

UAM_Login_URL=
UAM_IDTag_Name=
UAM_PWTag_Name=
UAM_URL=0.0.0.0
UAM_ID=
UAM_Password=
NAT_Traversal_Mode=0
Static_NAT_External_IP=0.0.0.0
Static_NAT_Start_Port=0
STUN_Server=
STUN_Port=3478
DiffServ_Signal=0
DiffServ_Media=0
Jitter_Buffer_Size=60
Payload_Type=18,0,8
Multiframe=2,2,2

[NETWORK2]
Name=Config2
Enable=0
Join_Method=0
SIP_Outbound_Proxy=
SSID=
Enable_DHCP=1
Address=0.0.0.0
Netmask=255.255.255.0
Gateway=0.0.0.0
DNS1=0.0.0.0
DNS2=0.0.0.0
Enable_WEP=0
WEP_Bits=0
Default_WEP_Key=1
WEP_Key1=
WEP_Key2=
WEP_Key3=
WEP_Key4=
Authentication_Algorithm=0
Post_Authentication_Mode=0
8021X_Name=
8021X_Password=
UAM_Use_Manual=0
UAM_Login_URL=
UAM_IDTag_Name=
UAM_PWTag_Name=
UAM_URL=0.0.0.0
UAM_ID=
UAM_Password=
NAT_Traversal_Mode=0
Static_NAT_External_IP=0.0.0.0
Static_NAT_Start_Port=0
STUN_Server=
STUN_Port=3478
DiffServ_Signal=0
DiffServ_Media=0
Jitter_Buffer_Size=60
Payload_Type=0,8,18
Multiframe=2,2,2

[NETWORK3]
Name=Config3
Enable=0
Join_Method=0
SIP_Outbound_Proxy=
SSID=
Enable_DHCP=1
Address=0.0.0.0
Netmask=255.255.255.0
Gateway=0.0.0.0
DNS1=0.0.0.0
DNS2=0.0.0.0
Enable_WEP=0
WEP_Bits=0
Default_WEP_Key=1
WEP_Key1=
WEP_Key2=
WEP_Key3=
WEP_Key4=
Authentication_Algorithm=0
Post_Authentication_Mode=0
8021X_Name=
8021X_Password=
UAM_Use_Manual=0
UAM_Login_URL=
UAM_IDTag_Name=
UAM_PWTag_Name=
UAM_URL=0.0.0.0
UAM_ID=
UAM_Password=
NAT_Traversal_Mode=0
Static_NAT_External_IP=0.0.0.0
Static_NAT_Start_Port=0
STUN_Server=
STUN_Port=3478
DiffServ_Signal=0
DiffServ_Media=0
Jitter_Buffer_Size=60
Payload_Type=0,8,18
Multiframe=2,2,2

[NETWORK4]
Name=Config4
Enable=0
Join_Method=0
SIP_Outbound_Proxy=
SSID=
Enable_DHCP=1
Address=0.0.0.0
Netmask=255.255.255.0
Gateway=0.0.0.0
DNS1=0.0.0.0
DNS2=0.0.0.0
Enable_WEP=0
WEP_Bits=0
Default_WEP_Key=1
WEP_Key1=
WEP_Key2=
WEP_Key3=
WEP_Key4=
Authentication_Algorithm=0
Post_Authentication_Mode=0
8021X_Name=
8021X_Password=

```
UAM_Use_Manual=0
UAM_Login_URL=
UAM_IDTag_Name=
UAM_PWTag_Name=
UAM_URL=0.0.0.0
UAM_ID=
UAM_Password=
NAT_Traversal_Mode=0
Static_NAT_External_IP=0.0.0.0
Static_NAT_Start_Port=0
STUN_Server=
STUN_Port=3478
DiffServ_Signal=0
DiffServ_Media=0
Jitter_Buffer_Size=60
Payload_Type=0,8,18
Multiframe=2,2,2

[NETWORK5]
Name=Config5
Enable=0
Join_Method=0
SIP_Outbound_Proxy=
SSID=
Enable_DHCP=1
Address=0.0.0.0
Netmask=255.255.255.0
Gateway=0.0.0.0
DNS1=0.0.0.0
DNS2=0.0.0.0
Enable_WEP=0
WEP_Bits=0
Default_WEP_Key=1
WEP_Key1=
WEP_Key2=
WEP_Key3=
WEP_Key4=
Authentication_Algorithm=0
Post_Authentication_Mode=0
8021X_Name=
8021X_Password=
UAM_Use_Manual=0
UAM_Login_URL=
UAM_IDTag_Name=
UAM_PWTag_Name=
UAM_URL=0.0.0.0
UAM_ID=
UAM_Password=
NAT_Traversal_Mode=0
Static_NAT_External_IP=0.0.0.0
Static_NAT_Start_Port=0
STUN_Server=
STUN_Port=3478
DiffServ_Signal=0
DiffServ_Media=0
Jitter_Buffer_Size=60
Payload_Type=0,8,18
Multiframe=2,2,2

[DTMF]
Mode=2
Duration=100
RFC2833_Volume=10
RFC2833_Payload_Type=101
Enable_Auto_DTMF_Mode=1

[CALLER_ID]
Use_Caller_ID_OnOff=1
Enable_Caller_ID=1
Anonymous_Displayname=Anonymous
Use_Update_Caller_ID=0
Hide_Displayname=0
Update_Caller_ID_After_Transfer=1

[UI]
Dial_Number_Font_Size=1
Enable_Key_Lock=0
Enable_Key_Lock_Password=1
Enable_Mannermode=0
Enable_Error_Indication=1
Use_Advanced_Rate_Set=0
Network_Setup_Menu_Location=1
SIP_Menu_Location=1
[SMS]
Use_SMS=1
Message_Server=

[PHONEBOOK]
Use_Index_Sending=1
[PRESENCE]
Use_Presence=1
Enable_Online_Ring=1
Online_Ring_Type=3
Online_Ring_Mode=3
Subscribe_Expire=600
Presence_Server=

[SERVICE_LAMP]
Enable_Service_Lamp=1
Red_Alert_Antenna_Level=0
Indication_Interval=10
Indication_Mode=0
 [DYNAMIC_NETWORK]
Backoff_Interval=4,8,16,32,64
DHCP_Verify_Count=3
DHCP_Verify_Interval=3
8021X_Bind_Timeout=15

[TONE_TYPE]
Dial_Tone_Type_On_Idle=2
Dial_Tone_Type_On_Hold=0
Send_Dial_Tone_Type=1

[AUTO_UPGRADE]
Enable=0
Time=0
Repeat=0
```

## Record of Change

This application note is subject to change. Updates and corrections are always welcome. Please submit any updates or corrections to info@shoretel.com.

| Issue | Author | Reason for Change | Date |
|---|---|---|---|
| 2.0 | J. Casselman | Edit | February 5, 2007 |
| 1.0 | J. Casselman | Initial Release | May 24, 2006 |

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