TELEDEX®

I SERIES VoIP
USER GUIDE

For ND Series, NDC Series, and RediDocks

Important Safety Instructions

When using your telephone equipment, basic safety precautions should always be followed to reduce the risk of fire, electric shock, and injury to persons, including the following:

1. Read and understand all instructions.
2. Follow all warnings and instructions marked on the product.
3. Unplug the product from the wall outlet before cleaning. Do not use liquid cleaner or aerosol cleaners. Use a damp cloth for cleaning.
4. Do not use this product near water—for example, near a bathtub, wash bowl, kitchen sink or laundry tub, in a wet basement, or near a swimming pool.
5. Do not place this product on an unstable cart, stand, or table. The product may fall, causing serious damage to the product.
6. Slots and openings in the cabinet and the back or bottom are provided for ventilation, to protect it from overheating. These openings must not be blocked or covered. The openings should never be blocked by placing the product on the bed, sofa, rug, or any other similar surface. This product should never be place near or over a radiator or heat register. This product should not be placed in a built-in installation unless proper ventilation is provided.
7. Never push objects of any kind into this product through cabinet slots as they may touch dangerous voltage points or short out parts that could result in a risk of fire or electric shock. Never spill liquid of any kind on the product.
8. To reduce the risk of electric shock do not disassemble this product. Take it to a qualified service facility if service or repair work is required. Opening or removing covers may expose you to dangerous voltages or other risks. Incorrect reassembly can cause electric shock when the appliance is subsequently used.
9. Unplug this product from the wall outlet and refer servicing to qualified service personnel under the following conditions:
   - When the power supply cord or plug is damaged or frayed.
   - If liquid has been spilled into the product.
   - If the product has been exposed to rain or water.
   - If the product does not operate normally by following the operating instructions. Adjust only those controls that are covered by the operating instructions, as improper adjustment of other controls may result in damage and may require extensive work by a qualified technician to restore the product to normal operation.
   - If the product has been dropped or the cabinet has been damaged.
   - If the product exhibits a distinct change in performance.
10. Avoid using a telephone (other than a cordless type) during an electrical storm. There may be a remote risk of electric shock from lightning.
11. Do not use the telephone to report a gas leak in the vicinity of the leak.

PLEASE SAVE THESE INSTRUCTIONS.

FCC Interference Information

This device complies with Part 15 of the FCC Rules. Operation is subject to the following two conditions:

1. This device may not cause harmful interference.
2. This device must accept any interference received, including interference that may cause undesired operation.

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation.

If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna for the radio or television that is receiving the interference).
- Reorient or relocate and increase the separation between the telecommunications equipment and receiving antenna.
- Connect the telecommunications equipment into an outlet on a circuit different from that to which the receiving antenna is connected.

FCC RF Radiation Exposure Statement

The installation of the base unit should allow at least 20 centimeters between the base and persons to be in compliance with FCC RF exposure guidelines. For body-worn operation, the portable part (handset) has been tested and meets FCC RF exposure guidelines.

This device must not be co-located or operating in conjunction with any other antenna or transmitter. The changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

Industry of Canada Requirements

Note: This equipment meets the applicable Industry Canada Terminal Equipment Technical Specifications. This is confirmed by the registration number. The abbreviation, IC, before the registration number signifies that registration was performed based on a Declaration of Conformity indicating that Industry Canada technical specifications were met. It does not imply that Industry Canada approved the equipment.

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. The customer should be aware that compliance with the above conditions may not prevent degradation of service in some situations.

Repairs to certified equipment should be coordinated by a representative designated by the supplier. Any repairs or alterations made by a user to this equipment, or equipment malfunctions, may give the telephone communications company cause to request the user to disconnect the equipment.

Users should ensure for their own protection, that the electrical ground connections of the power utility, telephone lines, and internal metallic water pipe system, if present, are connected together. This precaution may be particularly important in rural areas.

Caution: Users should not attempt to make such connections themselves, but should contact the appropriate electric inspection authority, or electrician, as appropriate.

Notice: The Ringer Equivalence Number (REN) assigned to each terminal device provides an indication of the maximum number of terminals allowed to be connected to a telephone interface. The termination on an interface may consist of any combination of devices subject only to the requirement that the sum of the Ringer Equivalence Numbers of all the devices does not exceed 5.

REN: Z

For warranty and service in Canada, please contact:

Williams Telecommunications
5610 Kennedy Road
Mississauga, Ontario, L4Z2A9
Canada
Phone: 905-712-4242
Fax: 905-712-1754

Requirements of Part 15— FCC Rules

Note: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation.

This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

1. Move the telephone away from the receiver.
2. Consult the dealer or an experienced radio/TV technician for help.

Any changes made by the user not approved by the manufacturer can void the user's authority to operate the telephone.

Requirements of Part 68— FCC Rules

This equipment complies with Part 68 of the FCC Rules and the requirements adopted by ACTA. On the bottom of this telephone is a label that contains, among other information, a product identifier in the format US:AAAEQ##TXXXX. If requested, this number must be provided to the telephone company. The USOC Jack for this equipment is RJ11C.

A plug and jack used to connect this equipment to the premises wiring and telephone network must comply with the applicable FCC Part 68 rules and requirements adopted by ACTA. A compliant telephone cord and modular plug are provided with this telephone. It is designed to be connected to a compatible modular jack that is also compliant. See installation instructions for details.

The Ringer Equivalence Number (REN) is used to determine the number of devices that may be connected to a telephone line. Excessive RENs on a telephone line may result in the devices not ringing in response to an incoming call. In most but not all areas, the sum of RENs should not exceed five (5.0). To be certain of the number of devices that may be connected to a line, as determined by the total RENs, contact the local telephone company. For products approved after July 23, 2001, the REN for this product is a part of the product identifier that has the format US:AAAEQ##TXXXX. The digits represented by ## are the REN without a decimal point (e.g., 03 is a REN of 0.3). For earlier products, the REN is separately shown on the label.

If this telephone causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. But if advance notice is not practical, the telephone company will notify the customer as soon as possible. Also, you will be advised of your right to file a complaint with the FCC if you believe it is necessary.

The telephone company may make changes in its facilities, equipment, operations, or procedures that could affect the operation of the equipment. If this happens, the telephone company will provide advance notice in order for you to make the necessary modifications to maintain uninterrupted service.

If trouble is experienced with this equipment, for repair or warranty information, please contact Teledex at (800) 462-9446. If the equipment is causing harm to the telephone network, the telephone company may request that you disconnect the equipment until the problem is resolved.

There are no user-serviceable parts contained in this equipment.

Connection to party line service is subject to state tariffs. Contact the state public utility commission, public service commission, or corporation commission for information.

If your home has specially wired alarm equipment connected to the telephone line, ensure the installation of this product does not disable your alarm equipment. If you have questions about what will disable alarm equipment, consult your telephone company or a qualified installer.

This telephone is hearing aid compatible.

These telephone devices are intended for commercial use only, primarily in hotel guestrooms. They must be used with a PBX (private branch

exchange), and are not intended to be connected directly to a PSTN line (public switched telephone network). There are no user-serviceable parts inside the equipment.

Technical Specifications

This guide covers product made from 2009 forward with a serial number beginning with XX. For all other models please refer to the Teledex website for legacy product documentation.

DIMENSIONS

6" (w) x 9" (l) x 3.375" (h)

WEIGHT

With handset: 1.8 lbs.

POWER ADAPTER

7.5VDC 300mA Adapter (for cordless HDKIT)

9 VDC 300mA (Pass through adapter for cordless telephone)

User Guides

If additional User Guides are needed, please go to www.teledex.com to download the PDF.

Service

When problems arise that cannot be resolved using this or related documents, please go to www.teledex.com/support for information about customer support, technical support, warranty, and product returns.

Statement of Limited Warranty

Teledex product warranty information is available at www.teledex.com.

Converting from Desk to Wall Mount

The I Series phone can be converted for wall mounting applications. The conversion is easiest when the handset and line cords are not connected.

1. Located on the top of the telephone above the speaker grill is the wall/desk mount clip. Remove this clip by firmly pushing upward (towards the hookswitch).
2. Flip the clip over (top to bottom) so that the protruding edge is towards the top of the phone, and reinsert into cutout. This protruding edge will hold the handset.
3. Turn the telephone over so the bottom is up, facing you. Place the telephone on a non-abrasive surface to prevent scratching.
4. Locate and remove the mounting bracket, firmly push back and pull up to remove two of the four retaining tabs.
5. Rotate the mounting bracket 180 degrees clockwise so that the mounting eyelet on the bracket is facing in the same direction as the other mounting eyelet located on the bottom of the telephone.
6. Insert the top two retaining tabs of the mounting brackets into the mounting bracket slots (located near the middle of the telephone). Then firmly push down to insert the retaining tabs on the opposite side of the mounting bracket.
7. Connect a short modular line cord into the jack on the back of the phone (labeled LINE). Route the line cord through the line cord channel. Connect the other end of line cord to the phone jack on the wall mounting plate.
8. Turn the telephone over, and slide the telephone down onto the mounting posts in downward direction. Both eyelets should line up with the mounting posts.
9. Complete the wall mounting by installing the handset and handset cord.

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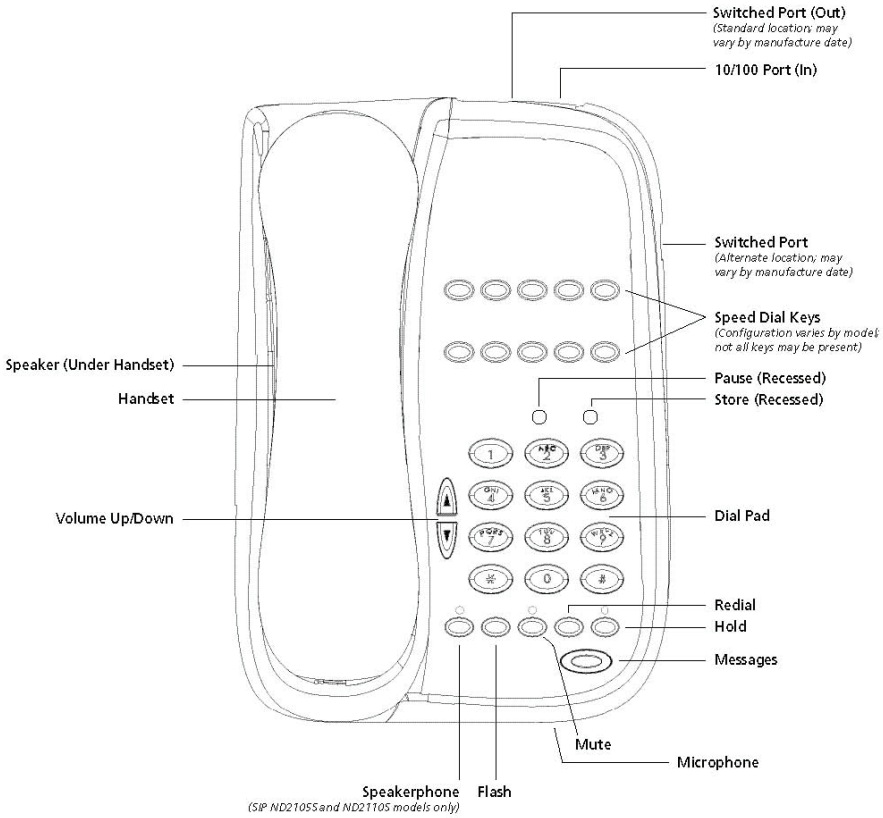
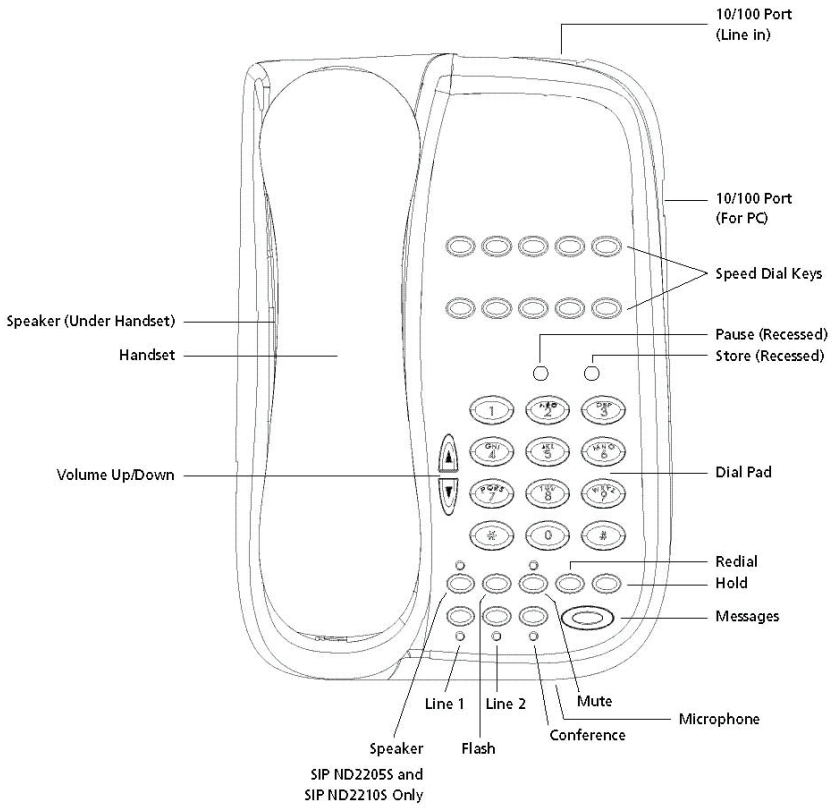
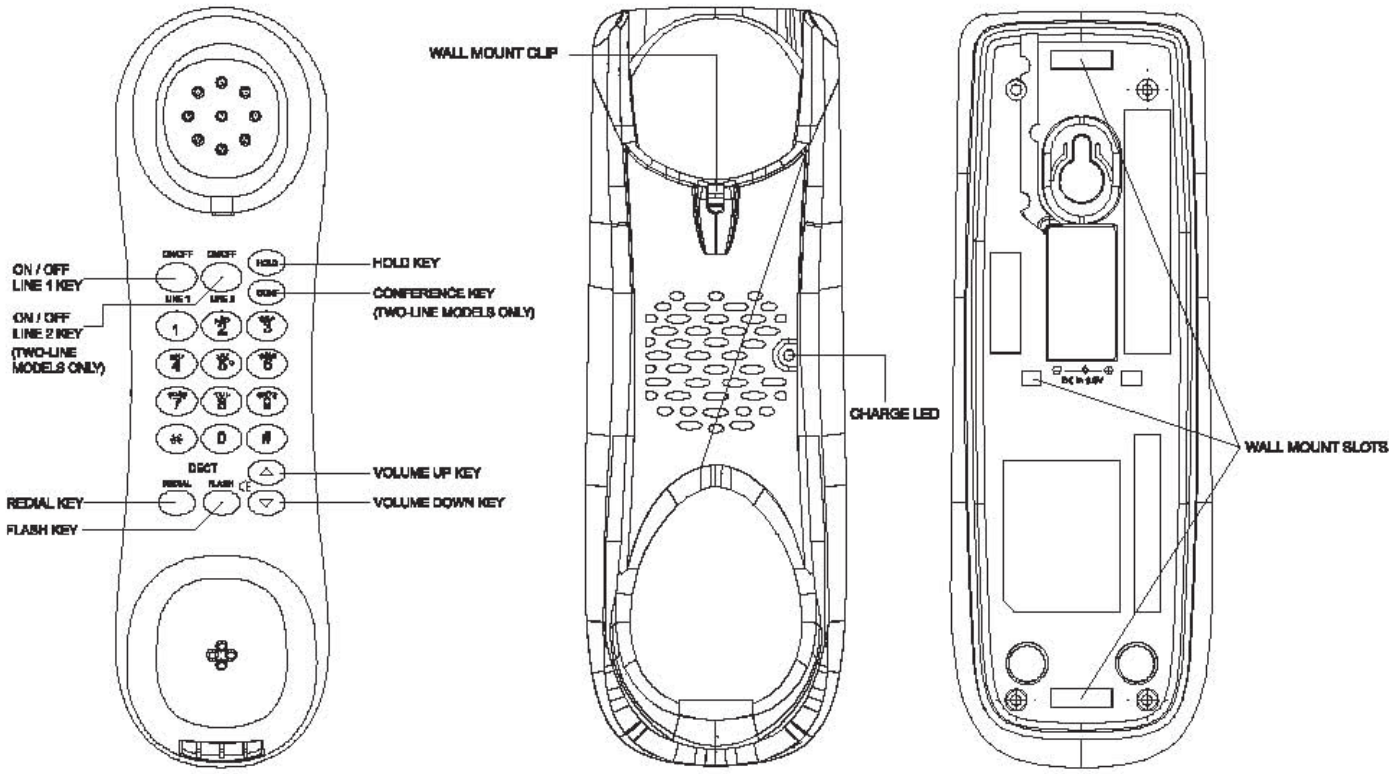
I Series ND and NDC Series 1-Line VoIP Phone Map



I Series ND and NDC Series 2-Line VoIP Phone Map



I Series RediDock VoIP Phone Map



Functions

- Support failover connection to a redundant SIP server
- Support NAT (Network Address Translation), Firewall
- Support DHCP for phone to accept IP via WAN port or assign IP address via LAN port
- Support PPPoE (used when connecting the ADSL or cable modem)
- Update the firmware and/or configuration file program through HTTP, FTP, and TFTP
- Hold function
- Hotline function (dial a specific number as soon as going off-hook or set a delay before dialing)
- Speed-dial
- Call-forward, Three-way conference call (in select 2-line models)
- Caller ID display
- DND (Do Not Disturb), Black List, Limit List
- Auto-answer
- Program settings through a standard web browser
- Remote management functionality via Telnet
- Classification management for common user's password and superuser's password

Standards and Protocols

- IEEE 802.3 / 802.3 u 10 Base T / 100 Base TX
- PPPoE
- DHCP Client and Server
- Support G.711a/u, G729, G7231 5.3/6.3 audio Codec
- RFC 3261, RFC 2543
- Support IAX2
- TCP/IP Internet transfer and control protocol (Transport Control Protocol/Internet Protocol)
- RTP Real-time Transport Protocol
- RTCP Real-time Control Protocol
- VAD/CNG Save bandwidth (Voice Activity Detection)
- Telnet Remote login protocol
- DNS Domain Name Server
- TFTP Trivial File Transfer Protocol

1. Introduction

This is the User Guide for the I Series VoIP phone. This phone is a standard SIP (Session Initiation Protocol) phone, and some configuration is necessary before it is ready for use. This manual will illustrate how to set up the phone through keyboard and Web User Interface (UI).

1.1 Overview of Hardware

1.1.1

The two RJ-45 network interfaces support 10/100M Ethernet. The default WAN interface is a DHCP Client. The user connects the WAN interface to a Power Over Ethernet (POE) enabled switch to obtain power and an IP address. (Alternatively, the phone can be assigned a static IP address.) The LAN port is commonly used for bridging a guest computer to the network. You can use the default administrator's user name "admin" and password "admin" to login and setup the device's configuration.

1.1.2

Only the WAN interface supports POE (802.3af).

1.2 Overview of Software

Network Protocol
• SIP v1 (RFC2543) • SIP v2 (RFC3261) • IP/TCP/UDP/RTP/RTCP • IP/ICMP/ARP/RARP/SNTP • TFTP Client/DHCP Client/PPPOE Client • Telnet/HTTP Server • DNS Clients
Codec
• G.711: 64K bit/s (PCM) • G.723.1: 63k/5.3k bit/s • G.726: 16k/24k/32k/40k bit/s (ADPCM) • G.729A: 8k bit/s (CS-ACELP) • G.729B: Adds VAD & CNG to G.729
Voice Quality
• VAD: Voice Activity Detection • CNG: Comfortable Noise Generator • LEC: Line Echo Canceller • Packet Loss Compensation • Adaptive Jitter Buffer
Call Function
• Call Hold • Call Forward • Call Waiting • Three-way Conference
Tone
• Ring Tone • Dial Tone • Ring Back Tone • Busy Tone
Phone Function
• Volume Adjustment • Phone Book • Speed Dial Key
IP Assignment
• IP (Static IP) • DHCP • PPPoE
Security
• HTTP 1.1 Basic/Digest Authentication Web Setup • MD5 for SIP Authentication (RFC2069/ RFC2617)
QoS
• QoS Field
NAT Traversal
• STUN
Configuration
• Web Browser • Console/Telnet • Keypad
DTMF
• DTMF RELAY • DTMF SIP Info • DTMF RFC 2833
SIP Server
• Support two SIP servers working at the same time • Provide a backup SIP server
Firmware Upgrade
• TFTP • HTTP • FTP

2. Handset Keys for I Series

2.1 Function Table of Keystrokes

Name	Status	Function/Display
Hold	Call	Hold on
MWI	On-Hook	Press flashing light to retrieve messages
Mute	Press	Mute voice path
Redial	Call/ Dialing	Call the number dialed last time
Spkr		Speaker
Volume +	Call	Increase the volume
Volume –	Call	Decrease the volume
M1–M10	On-Hook	10 speed dial numbers
1	Dialing	"1"
2	Dialing	"2"
3	Dialing	"3"
4	Dialing	"4"
5	Dialing	"5"
6	Dialing	"6"
7	Dialing	"7"
8	Dialing	"8"
9	Dialing	"9"
0	Dialing	"0"
*	Dialing	"*"
#	Dialing	It can be regarded as the first number being dialed out or the end mark for ending number

2.1.1 MWI—MESSAGE WAITING INDICATION

The Messages key serves as an indicator that messages are present as well as a speed dial key to dial in and retrieve those messages. If messages are present, press the flashing red MWI key labeled Messages to go off-hook and dial the programmed voicemail number. First, program the speed dial to the correct voice mail port of your PBX in the Web UI (found under Advanced -> Memory Key -> MWI Number).

2.1.2 CALL VOLUME CONTROL

Press "Volume +" to increase the volume and "Volume –" to decrease the volume.

2.1.3 HOLD FUNCTION

Hold the current line's call.

2.1.4 REDIAL FUNCTION

Redial the last number dialed.

2.1.5 SPEED DIAL KEYS

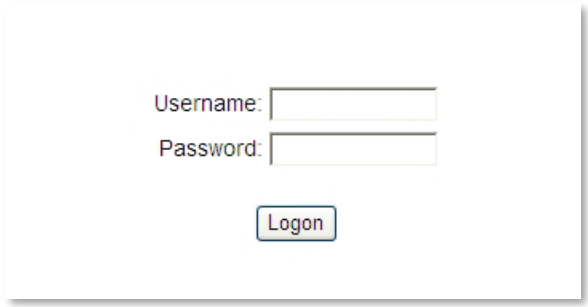
The I Series phone has 10 speed dial keys programmable through the Web UI under Advanced -> Memory Key.

3. Web Browser User Interface to Configure the Phone

In most cases, the phone will obtain an IP address from the local DHCP server. You may find this IP address from the phone by using a simple diagnostic code. On the phone's dialpad, press * * 4 7 # (* * I P #). The phone will read out the IP address (and display it on the LCD if the phone has a screen). To access the phone, it is necessary that your computer's IP is on the same network with the phone IP, so make sure it gets a compatible IP address within the same network. Then open a web browser, input the phone's IP address in the address field, and you will enter the web-based User Interface pages of the I Series phone.

3.1 Login

The default Username and Password are admin/admin and guest/guest.



3.2 Current Status

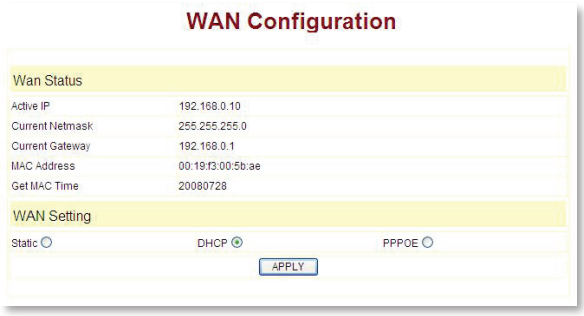
The Current Status page shows the working state of the VoIP phone. The network part shows the connection state of the WAN interface and the LAN interface as well as the network setting (DHCP or Static): the MAC address of the phone and the Firmware version. Here you can also see whether the phone is registered to the IP/PBX server or not. The Phone Number section shows the telephone numbers in SIP Line 1 and SIP Line 2. (SIP Line 2 is commonly used as a failover to a second IP/PBX in case of emergency.)

3.3 Network

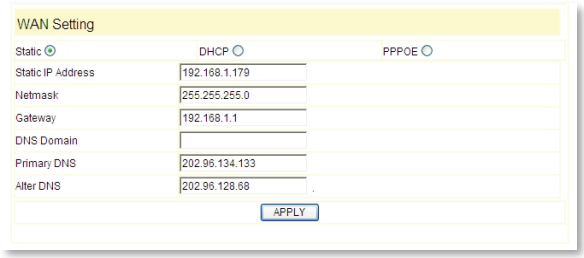
3.3.1 WAN CONFIGURATION

WAN PORT NETWORK SETTING PAGE

Supports Static IP, DHCP (Dynamic Host Control Protocol), and PPPoE.



CONFIGURE STATIC IP



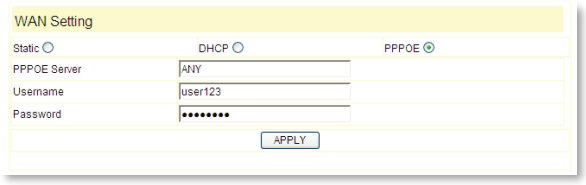
- Enable Static
- Set I Series' IP address in the IP Address field
- Set the Subnet mask in the Netmask field
- Set router IP address in the Gateway field
- DNS Domain
- Set local DNS server IP addresses in the Primary DNS and the secondary IP address in the Alternate DNS field

Configure to dynamically obtain the IP.

Enable DHCP.

If there is a DHCP server in your local network, the I Series phone will automatically obtain the WAN port network information from your DHCP server.

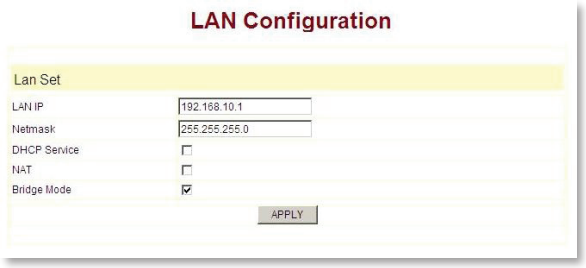
CONFIGURE PPPOE



- Enable PPPoE
- PPPoE Server: Enter "ANY" if not specified from your ITSP
- Enter PPPoE username and pin in the Username and Password entries

The I Series phone will automatically obtain the WAN port network information from your ITSP if the PPPoE setting and the setup are correct. Note: If a user accesses the IP phone through a WAN port he or she should use the new IP address to access the IP phone when the WAN port address was changed.

3.3.2 LAN CONFIGURATION



LAN IP: This is where you can set the IP address of the phone (at its LAN port). This is seldom used because in most environments Bridge Mode is enabled, causing the LAN port to act as a bridged extension of the WAN port.

Netmask: Set the IP Subnet mask for the LAN.

DHCP Service: Enable DHCP service on the LAN port. After the user changes LAN IP, the phone will automatically modify the DHCP Lease Table and save the configuration according to the IP and netmask information. The DHCP server configuration won't take effect until you reboot the device.

NAT: Enable Network Address Translation (NAT) for navigating through firewalls.

Bridge Mode: Enable this option to switch to Bridge Mode. An IP phone won't assign an internally generated IP for its LAN port in Bridge Mode. Most hotel properties will bridge the LAN and WAN port, but separate the Guest network from the Hotel's IP/PBX by implementing VLANs (Virtual LAN's). (This setting won't take effect unless you save the configuration and reboot the device.)

3.4 VoIP

3.4.1 SIP CONFIGURATION

SETTING PAGE OF PUBLIC SIP SERVER

Server Address: The IP address of the SIP server that the SIP phone will attempt to register to.

Server Port: The Internet Socket Port number of the SIP registration server. The default port is 5060.

Account Name: Username of your SIP account. (Usually the same as the phone number.)

Password: Password of your SIP account.

Phone Number: Phone number of your SIP account.

Enable Register: Enable/Disable SIP register. The I Series phone won't send registration information to the SIP server if the register is disabled.

Display Name: This field will display the User Agent (UA) in the header.

Proxy Server Address: IP address of proxy SIP server. (A SIP provider usually uses the same IP for the register server and the proxy server. In this case you don't need to configure the proxy server information.)

Proxy Server Port: Signal port of the SIP proxy.

Proxy Username: Proxy server username.

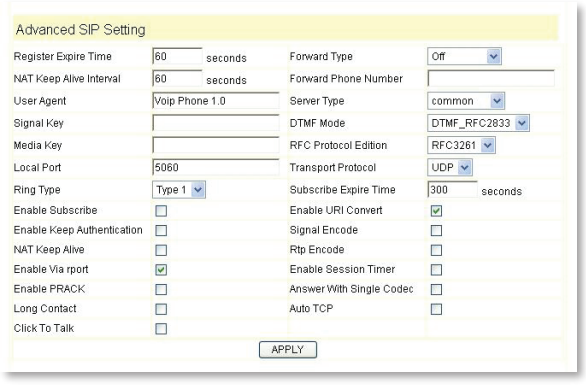
Proxy Password: Proxy server password.

Domain Realm: SIP domain—enter the SIP domain if any. Otherwise, the I Series phone will use the proxy server address as the SIP domain.

Enable Message Waiting: The configuration allows or forbids Message Waiting.

ADVANCED SIP SETTING

Advanced SIP Settings rarely need to be changed. Most of the settings should be left at Default, and many are not applicable to hospitality phones. Two of the settings you may (in rare instances) need to change are DTMF Mode and Forward Type (however, forwarding in hospitality is usually handled by the PBX).



Register Expire Time: Registration expiration time. The default is 60 seconds. The I Series phone will automatically configure this expire time to the server recommended setting if it is different from the SIP server.

Forward Type: Call forwarding has three types: Always, Busy, and No Answer. You may select the type of forwarding here, and then you must complete the configuration by Disabling Call Waiting (Found in Advanced -> Call Service).

No answer: If no answer after a set period of time (No Answer Time is found in Advanced -> Call Service) it will forward to the appointed phone.

Always: The caller is always forwarded to the appointed phone.

NAT Keep Alive Interval: The Network Address Translation (NAT) is the process of modifying IP address information in IP packet headers while in transit across a traffic routing device. Keep Alive is a method of maintaining that the service remains active.

Forward Phone Number: The designated phone number calls will forward to.

User Agent: The User Agent (UA) is an identifier for the phone and is part of the SIP Header used with every packet.

Server Type: The particular IP/PBX system supplier carries out the sign and speeches to encrypt. The default is COMMON and should always be used.

Signal Key: Not applicable in Hospitality environments.

DTMF Mode: DTMF (Dual Tone Multi-Frequency) signal sending mode: supports RFC2833, DTMF_ RELAY (inband audio), and SIP information where DTMF is sent as a SIP message. (DTMF is the acronym for the tones that are send from the number dialpad.)

DTMF SIP INFO Mode: DTMF Tones can be sent as SIP messages as well as audio streamed tones. SIP INFO mode has two differentiations in this drop-down menu. The default of "Send 10/11" is recognized by

most IP/PBXs, but some vendors (Avaya CM) recognized “Send */#”.

Media Key: Not applicable in Hospitality environments.

RFC Protocol Edition: SIP versions are identified by the IETF (Internet Engineering Task Force). SIP version 1 is RFC 2543. The default is the newer SIP 2 RFC 3261. An example of when you might revert back to RFC2543 is if the gate needs to communicate to older devices (such as CISCO5300) using the SIP 1.0 (RFC2543 protocol). Default is RFC 3261.

Local Port: The Standard Internet Port for SIP signaling is 5060. However, sometimes these port numbers are changed to enhance security. They would be changed in Basic Settings as well as in Advanced Settings.

Transport Protocol: Default is UDP (User Datagram Protocol) and should be used instead of the TCP (Transport Control Protocol) option.

Ring Type: The alerting ring that the phone presents to the called party is set here. Default is the USA conventional ring Type 1.

Subscribe Expire Time: Message Waiting Indication is an event package that must be subscribed to. The default is 300 seconds to check and re-subscribe with the server.

Enable Subscribe: Enable the subscription to the voice mail service.

Enable URI Convert: Default is enabled/checked. A Uniform Resource Identifier (URI) is a string of characters used to identify a name or a resource. Such identification enables interaction with representations of the resource over a network (typically the World Wide Web) using specific protocols. Leave this parameter selected.

Enable Keep Authentication: Not applicable in Hospitality environments. The default is unchecked.

Signal Encode: Not applicable in Hospitality environments. The default is unchecked.

NAT Keep Alive: Usually not applicable in Hospitality environments. The default is unchecked. “Keep Alive” is a term used to describe the act of keeping a persistent connection. Ports are opened and closed in IP communications. A keep-alive prompt/command ensures the connection is NOT dropped, but is instead kept open. When the client sends another request, it uses the same connection. This will continue until either the client or the server decides that the conversation is over, and one of them drops the connection.

Rtp Encode: Not applicable in Hospitality environments. Video H.273 encode related.

Enable Via Rport: Default is enabled/checked. This checkbox enables or disables rport parameter for the Via header field. The default setting allows a client to request that the server send the response back to the source IP address and port from which the request originated. However, in some environments it might be desired to switch this parameter off.

Enable Session Timer: Default is unchecked. Not applicable in Hospitality environments.

Enable PRACK: Default is unchecked/disabled. SIP PRACK (Provisional Acknowledgement) is a way to enable reliability for SIP 1xx messages like 180 ringing and 183 session in progress. PRACK messages flow from calling party to called party. PRACK reliability has been found to be important in several cases, including interoperability scenarios with the PSTN and Proxy servers with SIP v1 RFC2543. The default is DISABLED.

Answer with Single Codec: Not applicable in Hospitality environments. The default is unchecked/disabled.

Long Contact: Not applicable in Hospitality environments. The default is unchecked/disabled.

Auto TCP: Not Applicable in Hospitality environments. The default is unchecked/disabled. Auto TCP refers to TCP Auto-tuning. TCP tuning techniques adjust the network congestion

avoidance parameters of TCP connections over high-bandwidth, high-latency networks.

Click to Talk: Not applicable in Hospitality environments. The default is unchecked/disabled.

3.5 Advance

3.5.1 DHCP SERVER

DHCP SERVER MANAGE PAGE

User may trace and modify DHCP server information on this page.

DHCP Lease Table: Displays the IP-MAC corresponding table that the server distributed.

Lease Table Name: Lease table name.

Start IP: The starting IP of the lease table or DHCP Scope.

End IP: The ending IP of the lease table or DHCP Scope. Any network device connecting to the I Series phone's LAN port can dynamically obtain the IP range between the start IP and the end IP.

Lease Time: DHCP server lease time defining how long the IP address handed out to the network device is valid.

Netmask: Subnet mask of the lease table.

Gateway: Default gateway of the lease table.

DNS: Default DNS (Domain Name Server) of the lease table.

DNS Relay: Enables the Domain Name Server relay function.

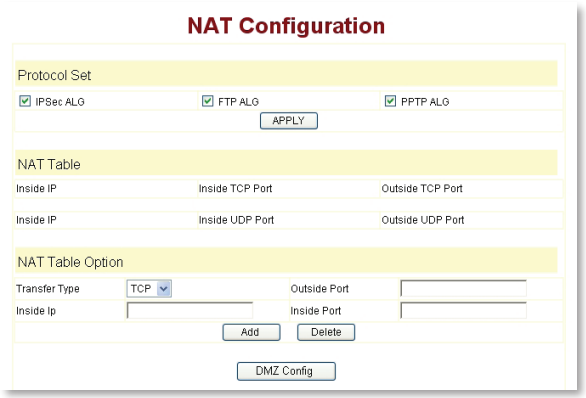
DHCP Lease Table: Shows the IP-MAC corresponding table assigned by the DHCP server.

Note: This setting won't take effect unless you save the configuration and reboot the device.

3.5.2 NAT CONFIGURATION

ADVANCE NETWORK ADDRESS TRANSLATION (NAT) SETTING

Maximum of 10 items for TCP and UDP port mapping.



IPSec ALG: Enable/Disable IPSec ALG.

FTP ALG: Enable/Disable FTP ALG.

PPTP ALG: Enable/Disable PPTP ALG.

Transfer Type: Transfer type using port mapping.

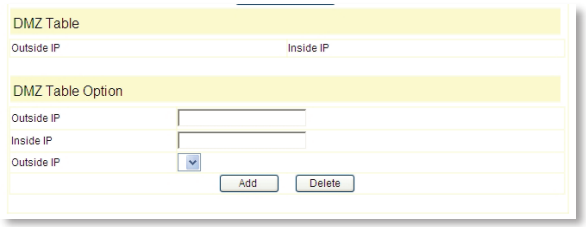
Outside Port: WAN port for port mapping.

Inside IP: LAN device IP for port mapping.

Inside Port: LAN device port for port mapping.

Click Add to add new port mapping item and Delete to delete current port mapping item.

DMZ CONFIGURATION



The purpose of a DMZ is to add an additional layer of security to an organization's Local Area

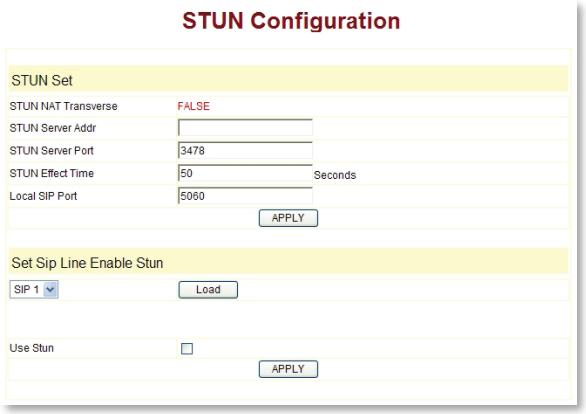
Network (LAN). An external attacker only has access to equipment in the DMZ, rather than any other part of the network. The name is derived from the term "demilitarized zone," an area between nation states in which military action is not permitted.

3.5.3 STUN CONFIGURATION

This page is used to set the private SIP server, STUN server, and back up SIP server information.

STUN SERVER SETTINGS

STUN means Simple Traversal of UDP through NATs (Network Address Translation). It is a protocol used for assisting devices behind a NAT firewall or router with their packet routing. Used to penetrate through NAT, start by configuring IP and port of STUN server (default is 3478) and Enabling SIP Stun. This should enable routing of SIP UDP packets from the phone through a NAT firewall. STUN only supports three NAT methodologies: FULL CONE, restricted, and port restricted.



STUN Server Addr: Configures the STUN server address.

STUN Server Port: Configures the STUN server port. The default is 3478.

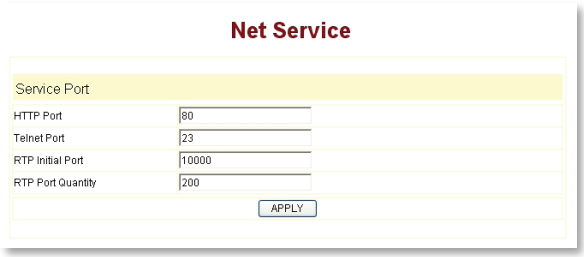
STUN Effect Time: STUN detects the NAT type circle; unit: minute.

Local SIP Port: The SIP port of this phone.

Load: Load the choices of the SIP line.

Use STUN: The STUN setting that allows (enables) or forbids use.

3.5.4 NET SERVICE



HTTP Port: Configures the Hyper Text Transfer Port; the default HTTP port is 80. User may change this port to enhance the system's security (the number must be between 1024 and 65000). For example, if you were to change to port 34687, then when this port is changed, you would use a generic IP address of format http://xxx.xxx.xxx.xxx:34687/ to reconnect.

Telnet Port: Configures the Telnet transfer port; the default is 23.

RTP Initial Port: Real-time Transport Protocol initial port. RTP is the standard packet format for delivering audio and video over IP networks.

RTP Port Quantity: Maximum number of unpriveliged UDP ports to be used for delivering the RTP packets; the default is 200.

Note: Settings on this page won't take effect unless you save and reboot the device.

If you need to change the Telnet port or HTTP port, use a port greater than 1024, because ports under 1024 are reserved ports.

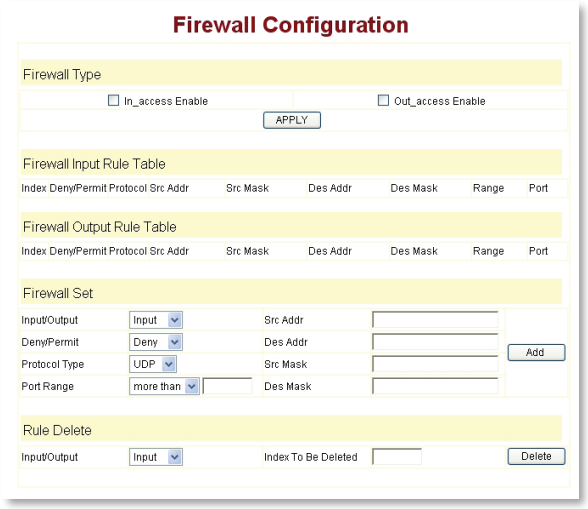
3.5.5 FIREWALL CONFIGURATION

Firewall Settings Page: User may set up a firewall to prevent unauthorized Internet users from accessing private networks connected to the Internet (input rule), or prevent unauthorized private network devices from accessing the Internet.

Access List Supports Two Type Limits: The input_access limit or output_access limit. Each type supports 10 items maximum.

I Series firewall filter is the base WAN port. So the source address or input destination address should be the WAN port IP address.

FIREWALL CONFIGURATION



In_access Enable: Enables the in_access rule.

Out_access Enable: Enables the out_access rule.

Input/Output: Select rule type input rule or output rule.

Deny/Permit: Select rule type deny rule or permit rule.

Protocol Type: Protocol used in this rule: TCP/IP/ICMP/UDP.

Port Range: Port range selections are more than, less than, equal, or not equal.

Src Addr: Source address. It can be a single IP address or a network address range.

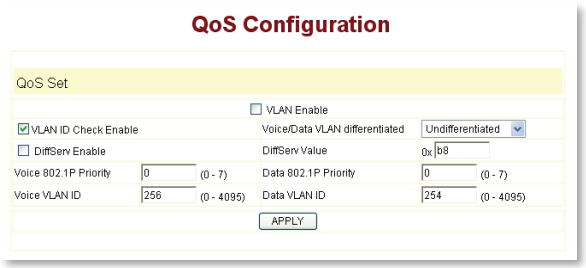
Des Addr: Destination address. It can be an IP address or a network address range.

Src Mask: Source address mask. Indicates the source is dedicated IP if it is set to 255.255.255.255. Otherwise, this is the network ID.

Des Mask: Destination address mask. Indicates the source is dedicated IP if it is set to 255.255.255.255. Otherwise, this is the network ID.

3.5.6 VLAN CONFIGURATION/QOS CONFIGURATION

The I Series phones implement Virtual LANs and Quality of Service (QoS) within this section of the menu. The QoS is based on the 802.1p protocol. The QoS is used to mark the network communication priority in the data link/MAC sub-layer. An I Series phone will sort the packets using QoS and send it to the destination.



VLAN Enable: If enabled the traffic over the WAN port (and the bridged LAN port) will separate voice, signaling, and data transmission. This is an essential security feature in hospitality environments to separate voice on one Virtual LAN and guest data traffic on a second Virtual LAN.

VLAN ID Check Enable: VLAN ID is the identification of the Virtual Local Area Networks.

DiffServ Enable: If enabled the VLAN service indicates to use the DSCP mode (DiffServ Code Point) to differentiate three layers of QoS (Quality of Service).

802.1P Priority: QoS (Quality of Service) is based on the 802.1p protocol. QoS is used to tag the packets with a Priority number (0–7 with 7 being highest priority and 0 being lowest). Usually voice packets will be tagged in order to ensure that voice quality will be preserved. This tagging happens in the Layer 2 data link/MAC sub-layer. An I Series phone will sort the packets using QoS and send them to the destination.

Voice VLAN ID: Is used for establishing a secure separate Virtual LAN for Voice signaling and traffic to the voice server. The value ranges are 0–4095. Default value is 256.

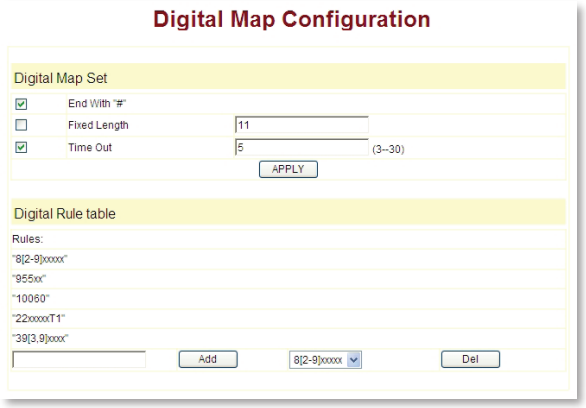
IMPORTANT: Voice/Data VLAN differentiated IS ALWAYS SET TO "Undifferentiated" in the drop down menu.

DiffServ Value: The value range: 0x28,0x30, 0x38,0x48,0x50,0x58,0x68,0x70,0x78,0x88,0x90,0x 98,0xb8. (Default is 0xb8.) 0xb8 stands for best fast transmission. 28–38 is for the transmission priority of the first rank. 48–58 is for the transmission priority of the second rank. 68–78 is for the transmission priority of the third rank. 88–98 is for the transmission priority of the fourth rank.

Data 802.1P Priority: QoS (Quality of Service is a 3-bit field called the Priority Code Point (PCP) within an Ethernet frame header when using VLAN tagged frames, as defined by IEEE 802.1Q. It specifies a priority value of between 0 and 7 inclusive that can be used by QoS disciplines to differentiate traffic. (0 is lowest priority traffic and 7 is highest priority traffic.)

Data VLAN ID: Is used for establishing a secure separate Virtual LAN for Guest Data traffic to the Internet. The value ranges are 0–4095. Default Value is 254.

3.5.7 DIGITAL MAP CONFIGURATION



The digital map is a set of rules to determine when the user has finished dialing.

I Series phones support the following digital map:

A digital map is based on rules judging when the user stops dialing and when to send the string of numbers to the server. I Series phones support the following digital map:

- End With "#": Use # as the end of dialing.
- Fixed Length: When the length of the dialing matches, the call will be sent.
- Timeout: Specify the timeout of the last digit dialed. The call will be sent after the timeout.

[ ] represents the range of digits used. It can be a range such as [1–4] or it can use commas such as [1,3,5] or it can use a list such as [234].

x represents any one digit between 0–9.

Tn represents the last digit for a timeout. It is necessary for n to represent the time from 0–9 seconds. Tn must be the last alpha and digit in the entry. If Tn is not included in the entry, we use T0 as default: which means the system will send the number immediately if the number matches the entry.

EXAMPLES:

8[2-8]xxxxx All numbers from 8200000 to 8899999 will be sent immediately.

955xx5 Six–digit numbers that begin with 9 will be sent immediately.

10060 The number 10060 will be sent immediately.

22xxxxx T1 Seven-digit numbers that begin with 22 will be sent after one second.

39[3,9]xxxx Seven-digit numbers beginning with 393 or 399 will be sent immediately.

BEST PRACTICES: DIAL PLAN PATTERNS

Below are some examples of best practices. These examples will help those in a hospitality environment to set up a useable dial plan.

The Dial Plan (Digital Map) is a set of patterns. As the user dials, as soon as the digits entered match one of the patterns, dialing is considered complete and the call is initiated.

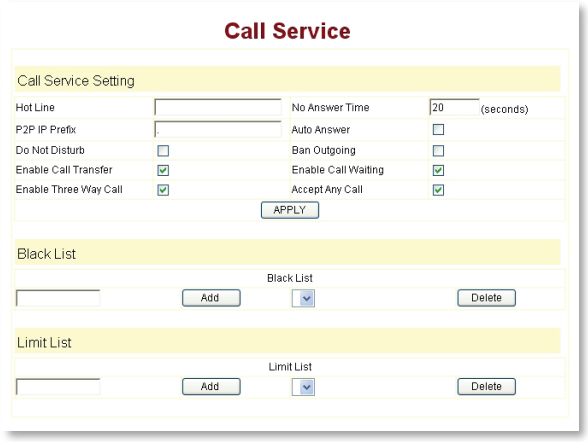
ELEMENTS IN A DIAL PLAN PATTERN:

X	Matches any dialed digit, #, or *
0–9, *, #	Matches the digit
[<digits>]	Matches any one of the digits between brackets
,	Play a secondary dial tone if the patter up to this point has been matched, e.g. "9,1xxxxxxxxx" means play a dial tone after a 9 is entered.
Tn	Additional digits may be dialed, but the pattern will be matched when n seconds elapse with no further dialing

IN THE U.S., A RECOMMENDED MINIMUM DIAL PLAN INCLUDES:

Item1 Rule	:0	
Item2 Rule	:911	#Emergency Number
Item3 Rule	:9911	#Emergency Number
Item4 Rule	:[2–7]xxx	#as needed to cover internal numbers
Item5 Rule	:91xxxxxxxxxx	#as needed to cover external numbers

3.5.8 CALL SERVICE



The user can also configure Call Service options such as a hotline, call forward, call transfer, call waiting, 3-way conference call, auto-answer, etc., on this page.

Hot Line: Configures the hotline number. If this option is set, I Series immediately dials this number when the handset is picked up, or the speaker key is pressed, taking the phone off-hook.

No Answer Time: The no answer/call forward time setting. Default is 20 seconds before a call forward.

P2P IP Prefix (Peer-to-Peer IP Prefix): Not applicable for Hospitality environments. Sets the Prefix for Peer-to-Peer IP calls. For example, if you want to dial the number 192.168.1.119, you define the P2P IP Prefix as "192.168.1.". So you dial only 119 to reach the number 192.168.1.119. Default is ".". If there is no "." set, the result disables the dialing IP.

Auto Answer: Enable/disable the auto-answer function.

Do Not Disturb: DND (Do Not Disturb) enables this option to refuse all calls.

Ban Outgoing: Enabling this bans outgoing calls.

Enable Call Transfer: This is used in business sets to FWD (Forward) or transfer a call. It is not enabled in hospitality sets.

Enable Call Waiting: Enable/disable the call waiting option. (It is also necessary to disable Call Waiting when enabling Call Forwarding under VOIP -> SIP Configuration -> Advanced Set.)

Enable Three Way Call: Default is enabled, allowing for conference calling.

Accept Any Call: Default is enabled. If this option is disabled, the I Series phone refuses the incoming call when the called number is different from I Series phone number.

Black List: Any incoming call from these phone numbers will be refused.

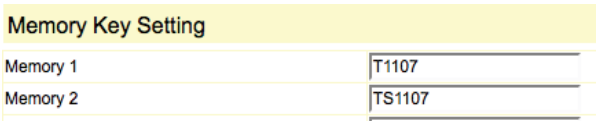
Limit List: Any outgoing calls using these phone numbers will be refused.

3.5.9 MEMORY KEY PROGRAMMING

This page layout shows the phone numbers used by the speed dial keys. There are several special functions that can be programmed into the memory keys, such as Transfer, Transfer-Silent, and Call-Park/Retrieve.

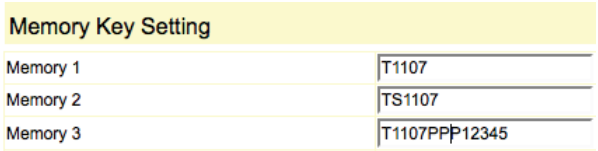
3.5.9.1 MEMORY KEY SPECIAL PROGRAMMING—TRANSFER, TRANSFER SILENT, AND 1-SECOND PAUSE

Transfer occurs when there is an active/live call in progress. If no call is active, the 'T' is ignored, and the memory key will just call the programmed number (1107). In the settings below, Memory 1 (T1107) will transfer to extension 1107 when pressed. The user will hear the DTMF tones as they are transferred to 1107.



In the settings above, Memory 2 (TS1107) will transfer "silently" to extension 1107. 'Silent' means the user will not hear the DTMF tones.

In the settings below in Memory 3, the transfer to 1107 will occur, after the connection is established, a pause will happen then digits are sent. P equals a 1-second pause, so PPP equals a 3-second pause, then the digits 12345 are sent.

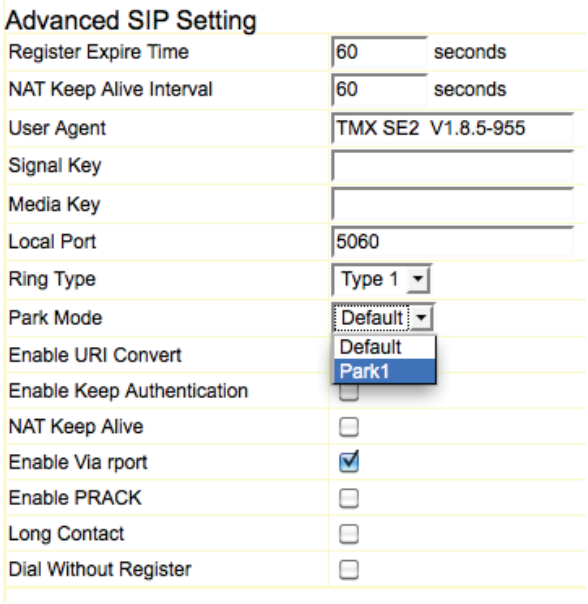


3.5.9.2 CALL-PARK/RETRIEVE

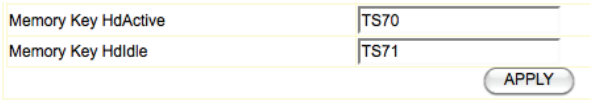
Call-Park/Retrieve allows the phone's user to send a call to a 'Park' location (setup within the PBX), where another user may then 'Retrieve' the call from a different phone.

It is programmed as follows in two steps:

1. Under SIP Config -> Advanced Set you will find Park Mode. Default is Off, and Park 1 is enabled or On.



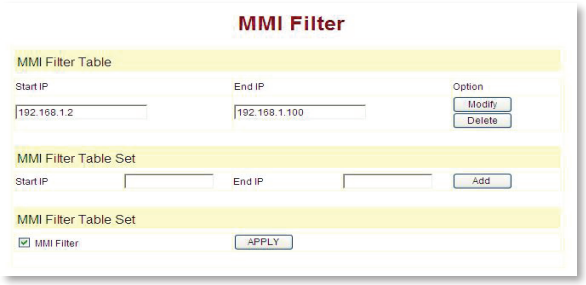
2. Under Advanced -> Memory Key, program the Memory Key HdActive to Transfer Silent the call to the Park Location (70 in the example below)—for example, the code used would then be TS70.



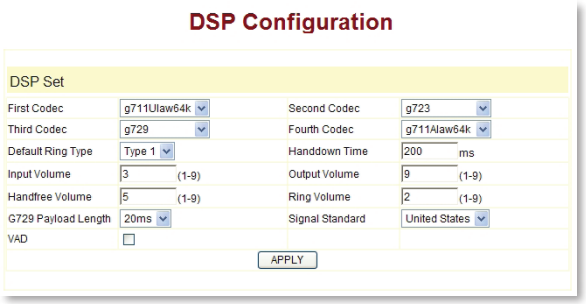
Also program Memory Key HdIdle to Retrieve from the Parked Location, in the above case TS71 is used. The Park and Retrieve numbers (70 and 71) are arbitrary, and subject to the programmer of the PBX. They are used here as an example only.

3.5.10 MMI FILTER

The MMI filter is used limit access to the I Series phone. When an MMI filter is enabled, only the IP addresses within the start IP and end IP can access that I Series phone.



3.5.11 DSP CONFIGURATION/AUDIO SETTINGS



CODEC (First, Second, Third, and Fourth): Select the preferred CODEC. Supports uLAW, aLAW, G729, and G723 The CODEC (Coder/Decoder) compression algorithms are negotiated with the IP/PBX server, and negotiation attempts will happen in the order seen here. The First CODEC will have priority over the others listed in descending order when attempting to negotiate recognized compression algorithms with the server.

Default Ring Type: This is the alerting "Ring" tone presented to the user when the phone is ringing.

Handdown Time: Handset down detect time. The amount of time before the handset goes on-hook.

Input Volume: Handset in volume (receiver).

Output Volume: Handset out volume (speaker).

Handfree Volume: The speaker phone volume.

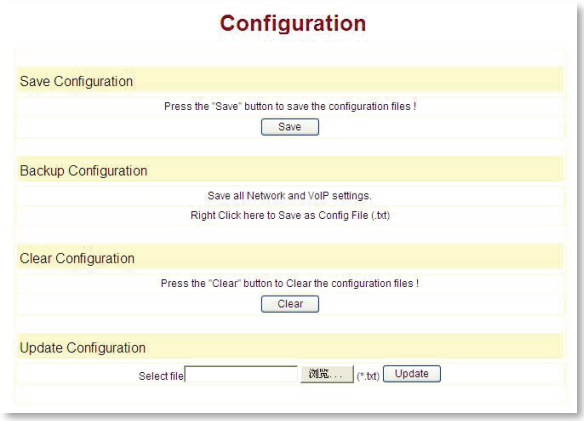
Ring Volume: The volume of the alerting ring as heard by the called party.

G729 Payload Length: G729 payload length. G.729 is an audio data compression algorithm for voice that compresses digital voice in packets of 10 milliseconds duration. It is officially described as Coding of speech at 8 kbit/s. The payload is the user data being transmitted, and the length is measured in milliseconds.

Signal Standard: Signal standard for different areas—this refers to the alerting ring tone that the calling party hears (ringing in their ear) as they call another party.

VAD: Enable/disable the VAD (Voice Activity Detection) option—VAD can facilitate speech processing, and can also be used to deactivate some processes during non-speech section of an audio session. It can avoid unnecessary coding/transmission of silence packets in Voice over Internet Protocol applications, saving on computation and on network bandwidth.

3.6 Configuration Management



Save Configuration: Saves the current settings.

Backup Configuration: Backup the configuration file by using the right key of mouse to click on save target as. A save window appears. Type the configuration file name in the File name area. (The file type is a text file.)

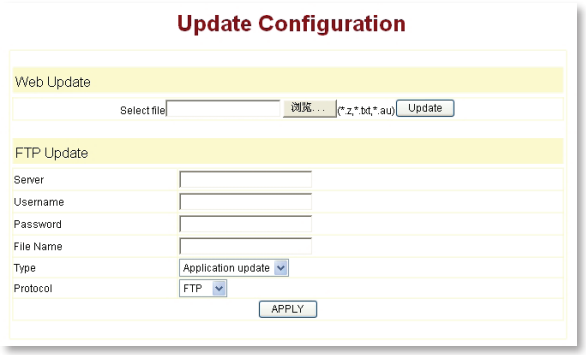
Clear Configuration: Restores to the default settings.

Update Configuration: Update the current configuration through configuration files.

Note: Clearing the configuration in admin mode restores all settings to factory default. Clearing the configuration in guest modem restores all settings to factory default except SIP and advanced SIP.

3.7 Update Firmware

3.7.1 UPDATE CONFIGURATION



Web Update: Updates the application or configuration files of the phone. The telephone's firmware is in a .z format and the configuration files are in a .txt format.

Click the "Browse" button (left of Update) to open the firmware (.z) file or the configuration file xxx.3300IP.txt), then click on the "Update" button. After the upgrade, the I Series phone will automatically restart.

FTP Update: Upload or download the configuration file with an FTP or TFTP server. Or download firmware from an FTP or TFTP server. Back up the configuration file to your FTP/TFTP server. The configuration file uses a .txt extension.

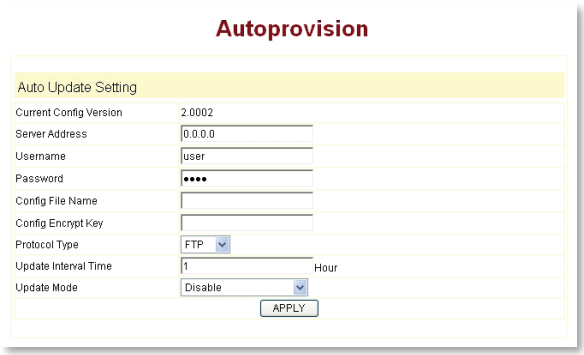
The "Type" includes two parts: a configuration file export and a configuration file import.

Configuration File Export: Exports the configuration file.

Configuration File Import: Imports the configuration file.

The I Series phone supports FTP and TFTP auto updating. The gateway will automatically obtain the configuration file from your updated server if it is configured. To obtain the original configuration file, you can use the FTP/TFTP back up as described above. Put the configuration file in the root directory of the updated server when finished editing.

3.7.2 AUTOPROVISION FIRMWARE UPDATE



Current Config Version: The system displays the current version number.

Server Address: The FTP/TFTP server address.

Username: The FTP server username.

Password: The FTP server password.

Config File Name: The name of the configuration file.

Config Encrypt Key: The encryption key of the confirmation file.

Protocol Type: The protocol type used for upgrading.

Update Interval Time: The interval time that the terminals search for a new configuration file.

Update Mode: The auto provision mode. Disable: no automatic update. Update after reboot: automatically updates after a reboot. Update at time interval: automatically updates after a certain time.

Configuration file version is found in the top section of the .txt file in the <<VOIP CONFIG FILE>> Version 1.0007 and <GLOBAL CONFIG MODULE> ConfFile Version.

For instance:

Gateway original version is:

<<VOIP CONFIG FILE>>Version: 1.0000
<GLOBAL CONFIG MODULE> ConfFile Version: 6

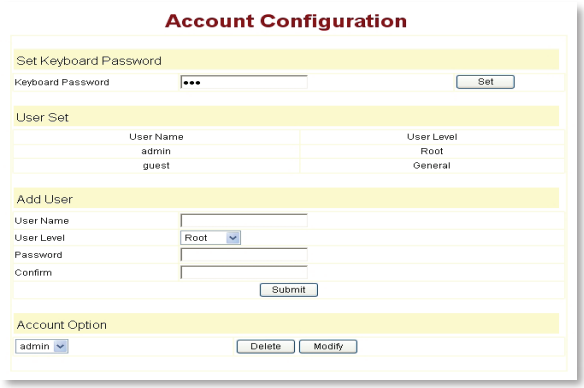
User may edit the configure file version to:

<<VOIP CONFIG FILE>>Version: 1.0007
<GLOBAL CONFIG MODULE> ConfFile Version: 7

3.8 System Manage

3.8.1 ACCOUNT MANAGE

Set the Web access account or keypad password for the I Series phone.



3.8.2 SYSLOG CONFIGURATION

Sets the system log.

Server IP: Sets the Syslog server address.

Server Port: Sets the Syslog server port.

MGR Log Level: Sets the MGR log level.

SIP Log Level: Sets the SIP log level.

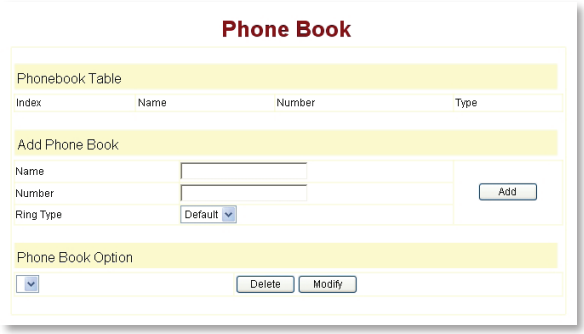
IAX2 Log Level: Sets the IAX2 log level.

Enable Syslog: Turns the system log information capture on. This information is highly technical and we recommend using the free software Wireshark for capturing network data packets instead of Syslog. Warning: if the power is cycled on the phone, Syslog is set back to its default "off" state.

Click "Apply" after setting this up.

3.8.3 PHONE BOOK

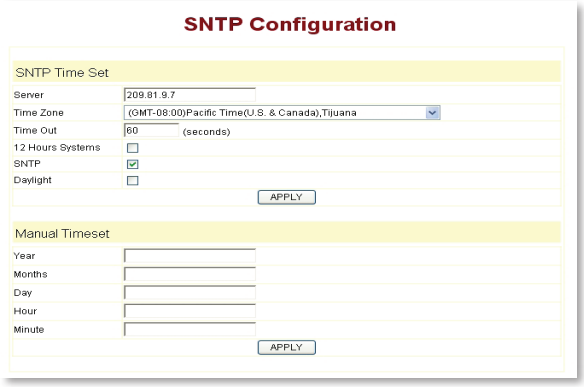
Cetis VoIP phones have a standard "base" firmware that is used across several models. Not all of the models have the same functionality, but the firmware may still show and be present in the User Interface (UI). Some functions may be inappropriate for hospitality environments, or not show up in a phone without a screen, but useful in commercial applications or display sets. Some of those examples are below.



Phone Book is not used in Hospitality environments, but remains in the universal firmware set for commercial IP phones.

3.8.4 TIME SET—VIA SIMPLE NETWORK TIME PROTOCOL (SNTP)

This page configures setting the time on the phone.



Server: Enter the IP address of the SNTP time server.

Time Zone: Select the correct time zone from the list in the box.

Time Out: The longest response time for SNTP server.

12 Hours Format: Select the 12 hours format.

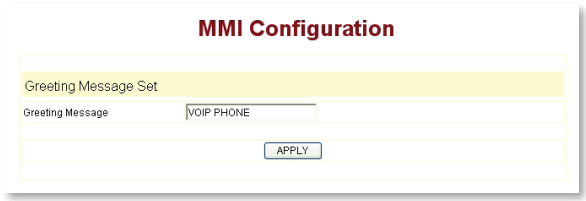
SNTP: Select the SNTP server.

Daylight: Whether Daylight Saving Time is being used.

Manual Configuration: Set up the date and time here instead of using a time server.

3.8.5 MMI CONFIGURATION

Set the greeting information on the LCD display through the Man-Machine-Interface (MMI).



3.8.6 LOGOUT AND REBOOT SYSTEM

Logout: Logout the Web entry.

Reboot Phone: Logout and reboot the phone. Whenever a user modifies a configuration parameter of the phone that will take effect after the phone is rebooted, the user enters into this screen and clicks "Reboot". The phone reboots automatically.

Note: About rebooting an IP phone: some settings require a reboot in order for the new settings to work. Always save the configuration before the reboot, otherwise the settings will return to the previous settings.

4. Operating Method for Dialing

4.1 How to Dial an IP Phone

You can make a call after your phone is properly set up. Please confirm whether all the network wires are connected correctly, and the phone credentials have been added (under VOIP section) to register successfully to the IP/PBX server.

To make a call, dial the number and then press "#".

You can find the phone's IP address by dialing * * 4(I) 7(P) #.

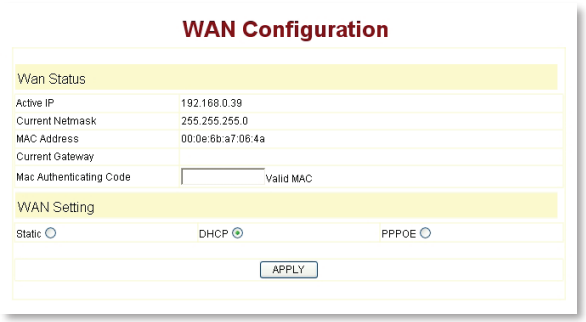
Input the IP address of the I Series phone in the browser, press the Enter key, and then you can visit the settings layout of the I Series phone. The administrator's account is admin/admin; the common user account is guest/guest.

4.2 Set Up the Phone to Register to a SIP Server

4.2.1 SET THE WAN INTERFACE

Here are the connection ways of entering the Network -> WAN Configuration layout in the phone's net port:

The I Series phone can be connected to the Internet by using the static IP, DHCP IP, or PPPoE dialing.



CONFIGURE STATIC IP

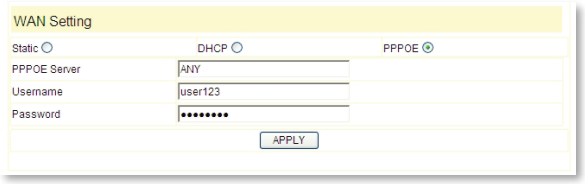
- Choose Static
- Fill in the IP address of the I Series phone in the Static IP Address field
- Fill in the subnet mask in Netmask
- Fill in the router address or up Gateway address in the Gateway
- Fill in the local DNS server address in the Primary DNS and Alter DNS respectively

USE THE CONFIGURATION TO DYNAMICALLY OBTAIN IP TO GET THE IP ADDRESS

- Choose the DHCP option

Now, if the network has a DHCP server, then the I Series phone will receive the IP Address, Netmask, Gateway, Primary DNS, and Alter DNS from this DHCP server automatically.

USE PPPOE DIALING TO CONNECT THE INTERNET



- Choose the PPPoE option
- Fill in the account and the password that PPPoE dialed in the PPPoE Username and Password areas

So the I Series phone could connect to the Internet through PPPoE dialing and automatically receive the IP Address, Netmask, Gateway, Primary DNS, and Alter DNS information, and so on.

4.2.2 SIP CONFIGURATION

Enter into the VoIP -> SIP Configuration window to set up the SIP account information:

Server Address: This is the IP address or FQDN (Fully Qualified Domain Name) of the IP/PBX SIP Registration server.

Server Port: This is the Internet Socket Port for SIP information. The common UDP port for a SIP server is 5060. Default port is 5060.

Account Name: Username of your SIP account. (Often the same as the phone number.)

Password: Password of your SIP account.

Phone Number: Phone number of your SIP account.

Enable Register: Choose Enable Register to allow the phone to begin the SIP registration Call/Response process.

Display Name: This field will display the User Agent (UA) in the header.

Proxy Server Address: (usually left blank)—IP address of proxy SIP server. (A SIP provider usually uses the same IP for the register server and the proxy server. In this case you don't need to configure the proxy server information.)

Proxy Server Port: (usually left blank) Signal port of the SIP proxy.

Proxy Username: (usually left blank) Proxy server username.

Proxy Password: (usually left blank) Proxy server password.

Domain Realm: (often left blank) SIP domain—enter the SIP domain if any. Otherwise, the I Series phone will use the proxy server address as the SIP domain.

Enable Message Waiting: (usually enabled)—The configuration allows or forbids Message Waiting.

Note: Choose Save Configuration in the Configuration Management window after setting up the new information or the existing setting information will be reset when the phone is rebooted.

Reference—Quick Keys

The Teledex family of SIP phones has implemented an innovative way to determine various states of the SIP phone and the network it is attached to via the phone's dial-pad. Press the * key twice in succession followed by a series of numbers shown below (the numbers are generally matched to the command's corresponding letters).

Many of these features are newly added in Firmware 1.8.0 and higher.

Show IP Address—**47# (**IP#)
Description: Keying **47# causes the phone to read out the IP address currently assigned. If the phone has an LCD panel, it should be displayed there. If the phone has no display panel, or the address cannot be displayed, it should be read out audibly on the speaker (or through the handset if it is off-hook).

Show VLAN—**85# (**VL#)
Description: When this is entered the phone responds with the Voice VLAN ID. If the VLAN is disabled "Zero" to indicate "off" is spoken.

Set VLAN—**87# (**87*password*VLANid#)
Description: This allows the user to set the VLAN ID. The Keypad Password is also required. As an example: to set the VLAN ID when the password is 9382 and the VLAN ID is 30, enter "**87*123*301#". When a VLAN ID is set, the phone should set VLAN Enabled and the Voice VLAN ID. If necessary the LAN should be set to Bridge Mode also.

Show Firmware Version—**39# (**FW#)
Description: When **39# is entered, the phone will display or audibly respond with the firmware version number such as "1.7.240.235".

Show TFTP Server Address—**83# (**TF#)
Description: When entered, the phone responds with the IP address of the TFTP server. This should be the TFTP server address actually being used. It might come from the configuration file, option 66, option 125, or option 43.

Re-Program New ConfigID—
77*password*ConfigID# (RP*xxx*yyyy#)
Description: After entering **77*, the user must enter the Keypad Password followed by a * then the ConfigID of the new configuration file, followed by #. For example, if the password in the existing configuration on the phone is 123, and the new configuration file is 1234.3300ip.txt, then the user would enter **77*123*1234#. The user should NOT be prompted for the password. The phone will then download the new configuration file 1234.3300ip.txt.

Reset Phone to Factory Default—
33*<password># = (DE*123#)
Description: After entering **33*123#, the phone will reset to the factory default, which implies that there is no current programming information, so the phone will prompt for a ConfigID at reboot.

Re-boot Phone (without reset)—**72# (**RB#)
Description: After entering 72#, the phone will reboot.

SIP TeleWorker Configuration File Download –
**89*<tftp server IP address>*<ConfigID>#
Description: To download a configuration file for a phone remotely use this **TW code to designate a TFTP server IP address and ConfigID to download.

Reference—SIP Quick Start to Register a Single Phone

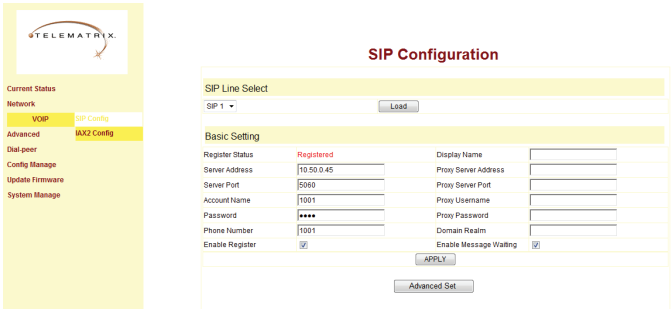
The Cetis, Inc., SIP phones are SIP-only endpoints powered only by POE/802.3af and will acquire an IP address when connected to a DHCP server. Begin by plugging your POE Ethernet cable into the WAN port. To retrieve an IP address, press * * 4 7 # (* * I P #). The phone's IP address will be displayed on the LCD or spoken by audio through the speaker. Point your Web browser to that IP address. You will be presented with a login screen.

Login = admin Password = admin

This is the Default Current Status Page

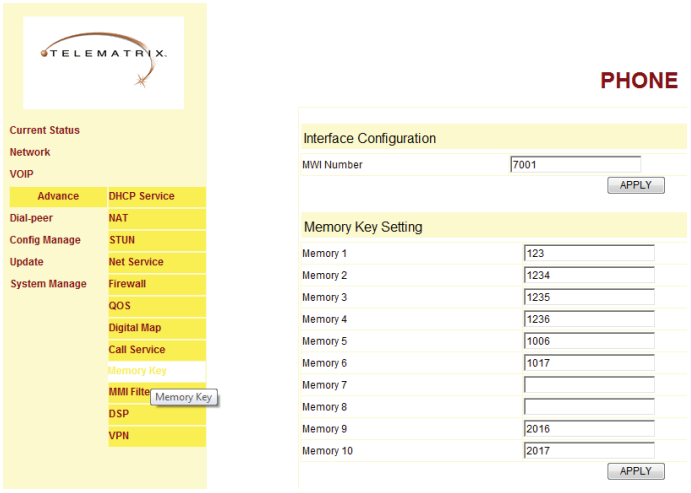
The scope of this document is to provide the basic screenshots necessary to quickly configure the guest phone for basic registration and show the user where to program the voice mail retrieval keys and the speed dials. In our example, the SIP phone is programmed as extension 1001. The firmware version of the phone is circled in red.

Program Your Connection Under the VoIP -> SIP Configuration Screen



The Server Address of the IP/PBX, Extension Number/Account Name, Phone Number, Display Name, Enable Register, and Enable Message Waiting indicators are all programmed here.

Speed Dial and OneTouch Message Keys Are Programmed Under Advance -> Memory Key



MWI Number is the field for programming the red Message Waiting Indicator OneTouch.

In this example, it is programmed to dial number 7001 on the IP/PBX for retrieving voice mail messages.

Reset to Factory Default

To reset the phone to the factory default settings, press "#" during the startup procedure. (At power up you can see a black progress bar.) The phone will enter into "post" mode, then input * # 1 6 8. Then you will see "clearing conf" on the screen, next you see "conf reset". Now you have a reset to the default settings. Lastly, cycle power (remove and replace WAN power cable) to restart the phone with factory default settings.

Reference—IP DECT Handset/Base AutoSync Registration

The IP-DECT cordless hospitality phones have a base station that is powered via POE 802.3af. The max power consumption is 3.5W, thus falling into the POE class spectrum of Class 2. The auxiliary handset stations are powered via standard electrical plugs.

AutoSync Handset Registration

Handsets can be registered most simply by performing AutoSync, which is to place the handset into the cradle and the pairing will happen automatically.

Register Up To 4 Additional Handsets

It is possible to register up to 4 additional remote handset kits to a single base station. Repeat the process above for three more handsets (LEAVE BATTERIES IN ALL HANDSETS).

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