HD2000 IP Phone User Manual

Table of Content

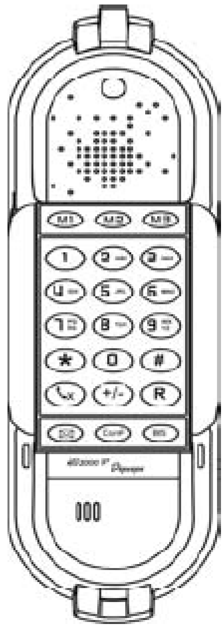
1. WELCOME

The IP PHONE HD2000IP is an internet telephone set that features superb audio quality, rich functionalities, high level of integration, and compactness. By converting analog voice for transmission over the internet, the IP Phone HD2000IP allows users with broadband internet connections to make calls to and from anywhere in the world. The IP PHONE HD2000IP is fully compatible with SIP industry standard and can interoperate with many other SIP compliant devices and software on the market.

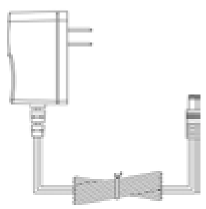
2. WHAT IS IN THE PACKAGE

The HD2000IP package contains:

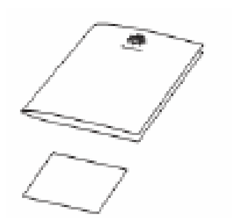
- One HD2000IP VoIP Phone



- One power supply (option)



- User manual



- 1 thin screw, 2 bigger screws, 1 foam.

3. Key Features

- Support SIP 2.0 (RFC 3261), TCP/UDP/IP, RTP/RTCP, HTTP, ICMP, ARP/RARP, DNS, DHCP, NTP, PPPoE, STUN, UPNP,TFTP, etc.
- Powerful Digital Signal Processing (DSP) technology to ensure superior audio quality
- Support various codecs including G.711 (PCM a-law and u-law), G.723.1 , G.729A, G.726.
- Support standard encryption and authentication (DIGEST using MD5, MD5-sess)
- Support for Layer 2 (802.1Q VLAN, 802.1p) and Layer 3 QoS (ToS, DiffServ, MPLS)
- Support Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise Generation), Acoustic Echo Cancellation (AEC) with Acoustic Gain Control(AGC) for speakerphone mode.
- Support automated provisioning for mass deployment ,RTP and TLS (pending)for security protection
- Support automated NAT traversal without manual manipulation of firewall/NAT
- Support Hold, Transfer, Forward, 3-way Conference, in-band and out-of-band DTMF, Call Waiting, Call Log, Off-hook Auto Dial, Auto Answer, Downloadable Ringtones, SMS,Direct IP Call, Intercom, Paging, Pick up.
- Support syslog, full duplex hands-free speakerphone with advanced acoustic echo cancellation, redial, volume control, voice mail with indicator, downloadable ring tones.
- Provide easy configuration through manual operation (phone keypad), Web interface or automated centralized configuration file via TFTP or HTTP
- Support 6 dedicated function keys: Mute, 3 level volume key, Flash (Hold), Message, Conference, Redial.
- Support 3 levels ringer volume (high/middle/off for HD2000IP with keypad, high/middle/low for HD2000IP urgence)

4. Hardware specification

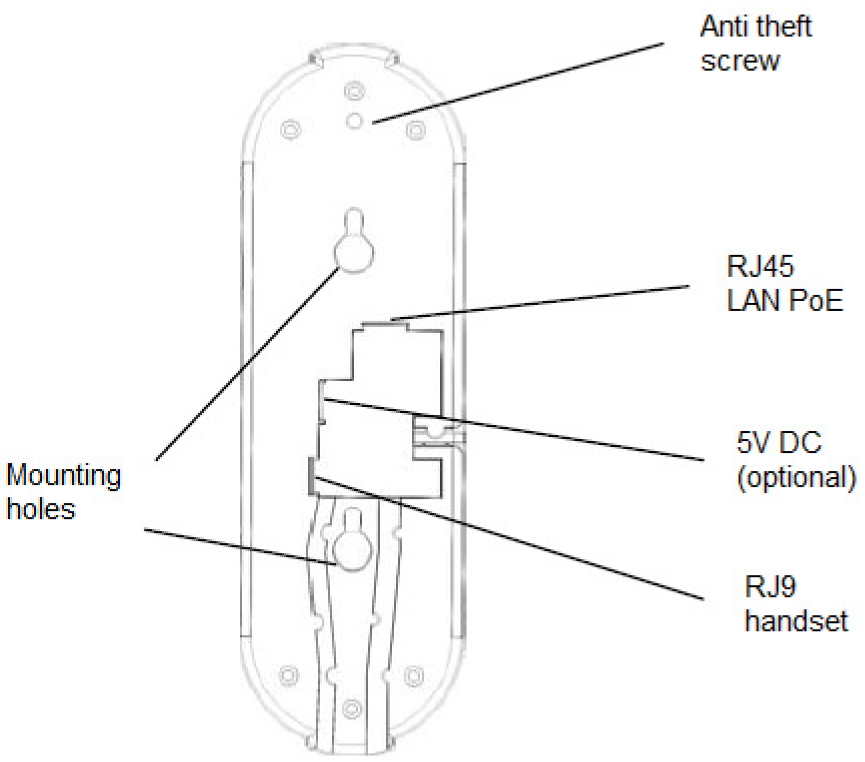
Model	HD2000IP
LAN interface	1x RJ45 100Base-T(POE supported)
Power supply (optional)	Input: 100-240VAC 50-60 Hz Output: +5VDC, 1200mA CE/FCC/UL certification
Dimension	215 x 165 x 70 mm (L x W x H)
Weight	0.9kg
Temperature	40 – 130 F 5 – 45 C
Humidity	10 - 90%

5. INSTALLATION

5.1. Power and LAN connection

Following are the steps to install a HD2000IP:

- Connect Ethernet cable from back of the phone (LAN Port) to a PoE port of switch or router.
- If you don't have PoE switch or router, please use power adapter (optional) into back of the phone and connect it to a power outlet.



Power Jack	5V DC power port
LAN	10/100Mbps RJ45 port for LAN
Handset Jack	RJ9 port connect to handset

SAFETY COMPLIANCES

The HD2000IP phone complies with FCC/CE and various safety standards. The HD2000IP power adaptor (optional) is compliant with these standards. Only use the HD2000IP power adaptor provided by Depaepe. The manufacturer's warranty does not cover damages to the phone caused by unsupported power adaptors.

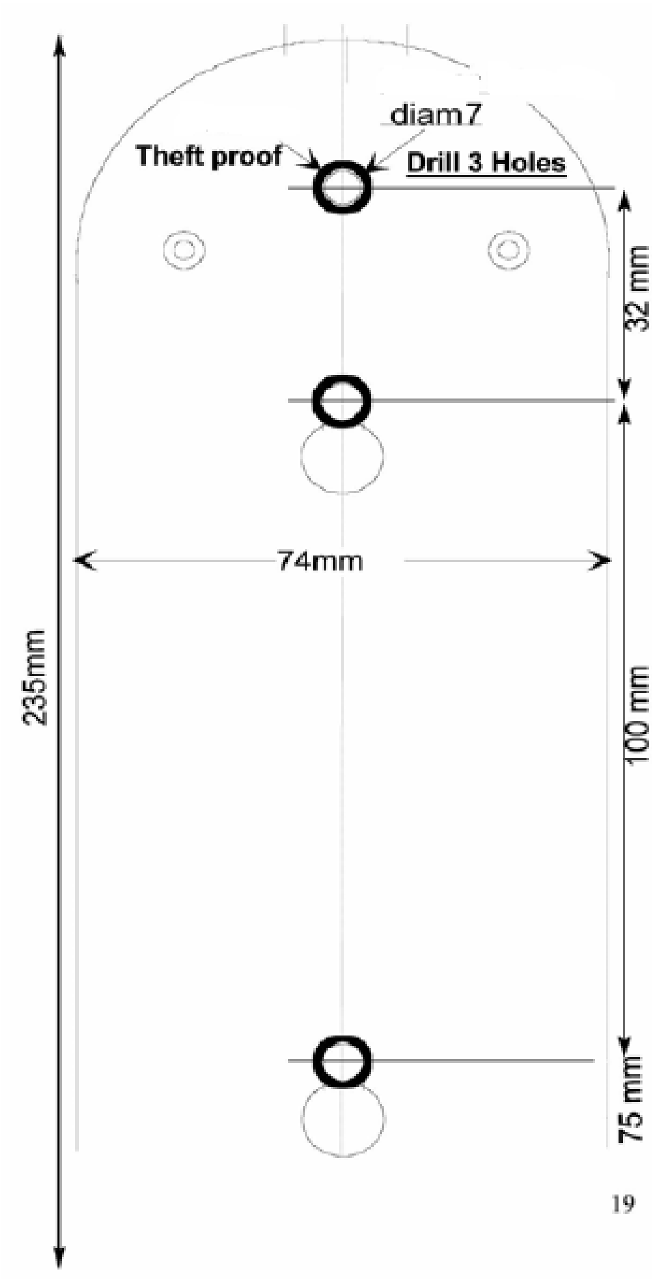
WARRANTY

If you purchased your HD2000IP from a reseller, please contact the company where you purchased your phone for replacement, repair or refund. If you purchased the product directly from Diamond Telecom Products, contact your Sales and Service Representative for a RMA (Return Materials Authorization) number before you return the product. Diamond Telecom Products reserves the right to remedy warranty policy without prior notification.

5.2. Walls mount installation:

The HD2000 IP comes with a small plastic bag containing 1 thins screw and a foam plug to be used for preventing theft or unauthorized removal, 2 bigger screws for wall mount fixing.

1) Drill 2 holes as shown on the wall mounting layout (see the next drawing). Install the 2 bigger screws in those holes on the wall.



2) If the telephone must be secured against thefts or unauthorized removal, drill a third hole as shown on the wall mounting layout.

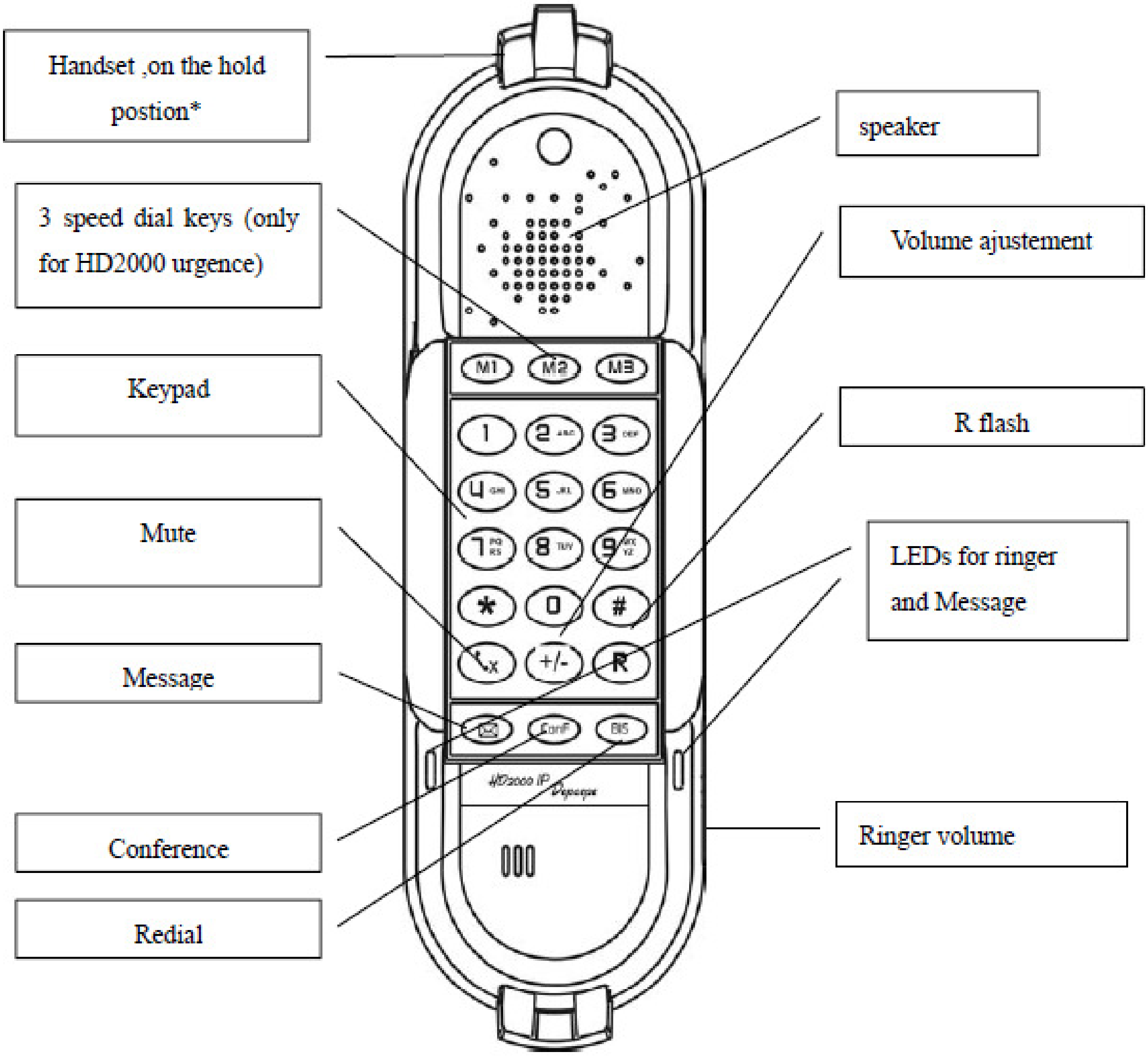
3) Check the LAN or/and Power connection.

4) Align the 2 slots at the base of the HD2000IP in front of the 2 screws and pull down.

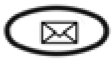
5) If needed, install the third screw though the hole located above the telephone (see the page

6) and hide it with the foam plug.

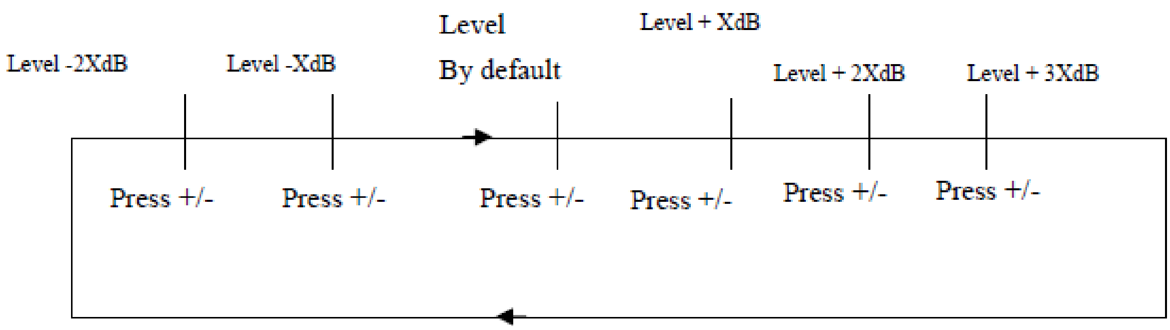
6. Get Familiar with the telephone



*Handset on hold position: This feature can be used to secure the handset while waiting for someone to ‘come to the phone” without going back to on-hook condition.

Keypad Buttons	Keypad Buttons Definitions
0 - 9, *(.), #	To input: numbers, *(.), #
	Stop voice to receiver (Mute key).
	Enter to retrieve voice mails or other messages
+/- **	control the earpiece (Handset) volume, the last state is memorized, when you press the key, you increase the volume up to the maximum step, a new pressing goes to the first step
R	Switch the call (Transfer key)
ConF	Conference call for three sides
#	Press # button to send a call immediately before "no key entry timeout" value Expires
Bis	redial the last number dialed
Speed dial keys	Short cut of register call

**Note:



7. BASIC OPERATIONS

7.1. Get Familiar with Voice menu:

HD2000IP has stored a voice prompt menu for quick access to settings and simple configuration. You can enter this voice prompt menu one ways

- Pick up the receiver of the analog telephone and press "***"

A voice will say, "Enter the new option." At this point, you can select from the following menu voice prompt options to begin using the HD2000IP:

Menu	Voice prompt	
Main Menu	"Enter a Menu Option"	Enter "*" for the next menu option Enter "#" to return to the main menu Enter 01 – 07,12 - 17, 47, 86 or 99 Menu option
01	"DHCP Mode", "Static IP Mode"	Enter '9' to toggle the selection If user selects "Static IP Mode", user needs configure all the IP address information through menu 02 to 05. If user selects "Dynamic IP Mode", the device will retrieve all IP address information from DHCP server automatically when user reboots the device.
02	"IP Address " + IP address	The current WAN IP address is announced. If in Static IP Mode, enter 12-digit new IP address like 192168000123.
03	"Subnet " + IP address	Same as Menu option 02
04	"Gateway " + IP address	Same as Menu option 02
05	"DNS Server " + IP address	Same as Menu option 02
06	"MAC Address"	Announces the Mac address of the unit.
07	Preferred Vocoder	Enter "9" to go to the next selection in the list: PCM U PCM A G-726 G-723 G-729
12	WAN Port Web Access	Enter "9" to toggle between enable / disable

13	Firmware Server IP Address	Announces current Firmware Server IP address. Enter 12 digit new IP address.
14	Configuration Server IP Address	Announces current Config Server Path IP address. Enter 12 digit new IP address.
15	Upgrade Protocol	Upgrade protocol for firmware and configuration update. Enter “9” to toggle between TFTP and HTTP
16	Firmware Version	Firmware version information.
17	Firmware Upgrade	Firmware upgrade mode. Enter “9” to rotate among the following three options: 1. always check 2. check when pre/suffix changes 3. never upgrade
47	“Direct IP Calling”	Enter the target IP address to make a direct IP call, after dial tone. (See “Make a Direct IP Call”.)
99	“RESET”	Enter “9” to reboot the device; or Enter MAC address to restore factory default setting (See Restore Factory Default Setting section)
	“Invalid Entry”	Automatically returns to Main Menu

Other Menu Prompt Features:

- “*” shifts down to the next menu option
- “#” returns to the main menu
- “9” functions as the ENTER key in many cases to confirm an option
- All entered digit sequences have known lengths - 2 digits for menu option and 12 digits for IP address. Once all of the digits are collected, the input will be processed.
- Incorrect keyed entry cannot be deleted or undone. The HD2000IP will prompt you to start Over by telling you that you made an error.

7.2. Make a Phone call

7.2.1. Completing Calls

There are two ways to complete a call:

- **DIAL:** To make a phone call.

• Take Handset off-hook
• The phone will have a dial tone.
• Enter the phone number
• Waiting for 4 seconds or press the # key
Note: 1) The value ‘no key entry time out’ by default is 4 seconds, you can change it.
2) You can also modify the dial plan for send a call immediately.

- **REDIAL:** To redial the last dialed phone number.

• Take Handset off-hook
• press Bis button.

7.2.2. Quick IP Call Mode

Direct IP calling allows two phones to talk to each other in an ad hoc fashion without a SIP proxy. VoIP calls can be made between two phones if:

• Both phones have public IP addresses, or
• Both phones are on a same LAN/VPN using private or public IP addresses, or
• Both phones can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ)

The HD2000IP also supports Quick IP call mode. This enables the phone to make direct IP-calls, using only the last few digits (last octet) of the target phone’s IP-number.
This is possible only if both phones are in under the same LAN/VPN. This simulates a PBX function using the CMSA/CD without a SIP server. Controlled static IP usage is recommended.

For example:

192.168.0.2 calling 192.168.0.3 -- dial *473 follow by #
192.168.0.2 calling 192.168.0.23 -- dial *4723 follow by #
192.168.0.2 calling 192.168.0.123 -- dial *47123 follow by #
192.168.0.2: dial *473 and *4703 and *47003 results in the same call -- call 192.168.0.3

NOTE: If you have a SIP Server configured, a Direct IP-IP still works. If you are using STUN, the Direct IPIP call will also use STUN. Configure the “Use Random Port” to “NO” when completing Direct IP calls.

7.3. ANSWERING PHONE CALLS

7.3.1. Receiving Calls

1. Incoming single call: Phone rings with selected ring-tone. Answer call by taking Handset.

2. Incoming multiple calls: When another call comes in while having an active call, the phone will produce a Call Waiting tone (stutter tone). Answer the incoming call by pressing the "R" key. The current active call will be put on hold.

7.4. PHONE FUNCTIONS DURING A PHONE CALL

7.4.1. Call Hold

While in conversation, pressing the "R" button will put the remote end on hold. Pressing the "R" button again will release the previously Hold state and resume the bi-directional media.

7.4.2. Call Waiting and Call Flashing

If call waiting feature is enabled, while the user is in a conversation, he will hear a special stutter tone if there is another incoming call. User then can press R button to put the current call party on hold automatically and switch to the other call. Pressing flash button toggles between the two active calls.

7.4.3. Call Transfer

HD2000IP supports both blind and attended call transfer. Each is easy to use. Use blind transfer if you want to transfer a call without speaking with someone first; use attended transfer if you want to speak with the someone prior to transferring call.

7.4.3.1 Blind Transfer

Transfer an active call to a third party without announcement.
Press the R button and wait for a dial tone. Dial the third party's phone number followed by the # button. Hang up to transfer the call
NOTE: The "Enable Call Feature" must be configured to "Yes" in the web configuration page to enable this feature.

7.4.3.2 Attended Transfer

Transfer an active call to a third party with attended.
Press R button and make a call and automatically place the ACTIVE call on HOLD. Once the call is established, hang up to transfer the call.
NOTE: To transfer calls across SIP domains, SIP service providers must support transfer across SIP domains.

7.4.4. Conference Call

HD2000IP phone supports 3-way conference.

Assuming that call party A and B are in conversation. A wants to bring C in a conference:

- A presses the “R” button to get a dial tone and put B on hold
- A dials C’s number then “SEND” key to make the call
- If C answers the call, then A presses “CONF” button to bring B, C in the conference.
- If C does not answer the call, A can press R back to talk to B.

NOTE:

- During the conference, if B or C drops the call, the remaining two parties can still talk. However, if A the conference initiator hangs up, all calls will be terminated.

7.4.5. Mute incoming calls

Press the Mute button to enable/disable muting the microphone.

7.4.6. Voice Messages (Message Waiting Indicator)

A blinking red MWI (Message Waiting Indicator) indicates a message is waiting. Press the MSG button to retrieve the message. An IVR (Interactive Voice Response) will prompt the user through the process of message retrieval.

NOTE: Account requires a voicemail portal number to be configured in the “voicemail user id” field.

7.5. CALL FEATURES

7.5.1. Call Features Tables

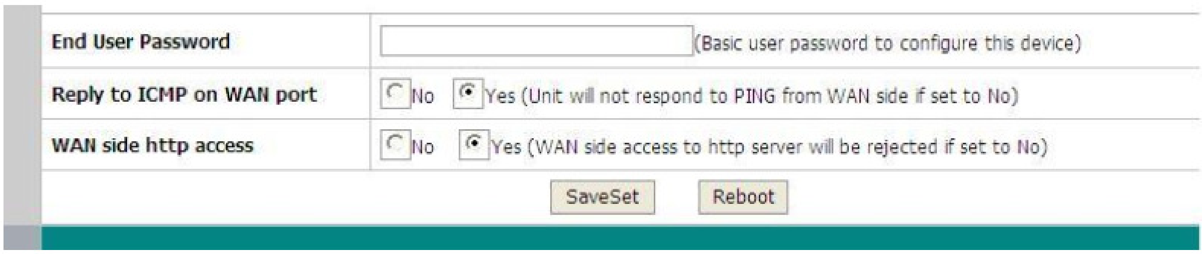
Following table shows the call features of HD2000IP :

Key	Call Features
*30	Block Caller ID (for all subsequent calls)
*31	Send Caller ID (for all subsequent calls)
*67	Block Caller ID (per call)
*82	Send Caller ID (per call)
*50	Disable Call Waiting (for all subsequent calls)
*51	Enable Call Waiting (for all subsequent calls)
*70	Disable Call Waiting. (Per Call)
*71	Enable Call Waiting (Per Call)
*72	Unconditional Call Forward. To use this feature, dial “*72” and get the dial tone. Then dial the forward number and “#” for a dial tone, then hang up.
*73	Cancel Unconditional Call Forward. To cancel “Unconditional Call Forward”, dial “*73” and get the dial tone, then hang up.
*90	Busy Call Forward. To use this feature, dial “*90” and get the dial tone. Then dial the forward number and “#” for a dial tone, then hang up.
*91	Cancel Busy Call Forward. To cancel “Busy Call Forward”, dial “*91” and get the dial tone, then hang up.
*92	Delayed Call Forward. To use this feature, dial “*92” and get the dial tone. Dial the forward number and “#” for a dial tone and then hang up.
*93	Cancel Delayed Call Forward. To cancel this feature, dial “*93”, get the dial tone, and then hang up.
Flash/Hook	Call waiting indication. When in conversation without an incoming call, this action will switch to a new channel to make a new call.

8. CONFIGURATION GUIDE

8.1. Configuring HD2000IP using Web Browser

HD2000IP has embedded Web server and HTML pages that allow users to configure the HD2000IP through an easy-to-use Web browser interface such as Microsoft's Internet Explorer or Netscape browser. Below is a screen shot of the HD2000IP configuration page:



8.1.1. Get the IP address of the HD2000IP:

Connect the HD2000IP to a network via standard Ethernet cable, default the HD2000IP is in DHCP mode.

If it is the HD2000IP with keypad, use the voice menu to get the IP address of the HD2000IP (see the 7.1 get familiar with voice menu)

If you have the HD2000IP urgence, use a standard network protocol analyzer (for e.g. Wireshark) to check the IP address allocated to the base unit by the DHCP server.

You can also contact with your administrator to get the IP address allocated to HD2000IP by DHCP server.

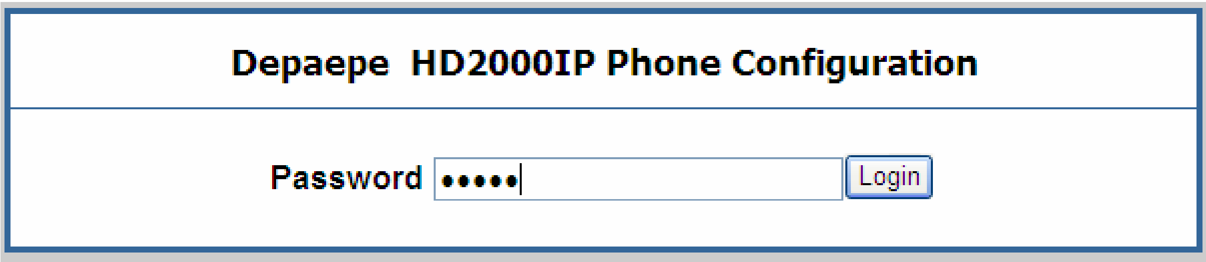
8.1.2. Accessing the Web Configuration

The HD2000IP configuration page can be accessed via your web browser by entering the WAN IP address: **http://yourip's**

Be sure that your PC is connected to the same VLan with the HD2000IP.

8.1.3. User Programming and Configuration

From your web browser, the HD2000IP will show the following login screen:



Enter the password and click on the "Login" button

8.1.4. Passwords

Passwords are case sensitive and the devices come with factory default passwords as indicated below:

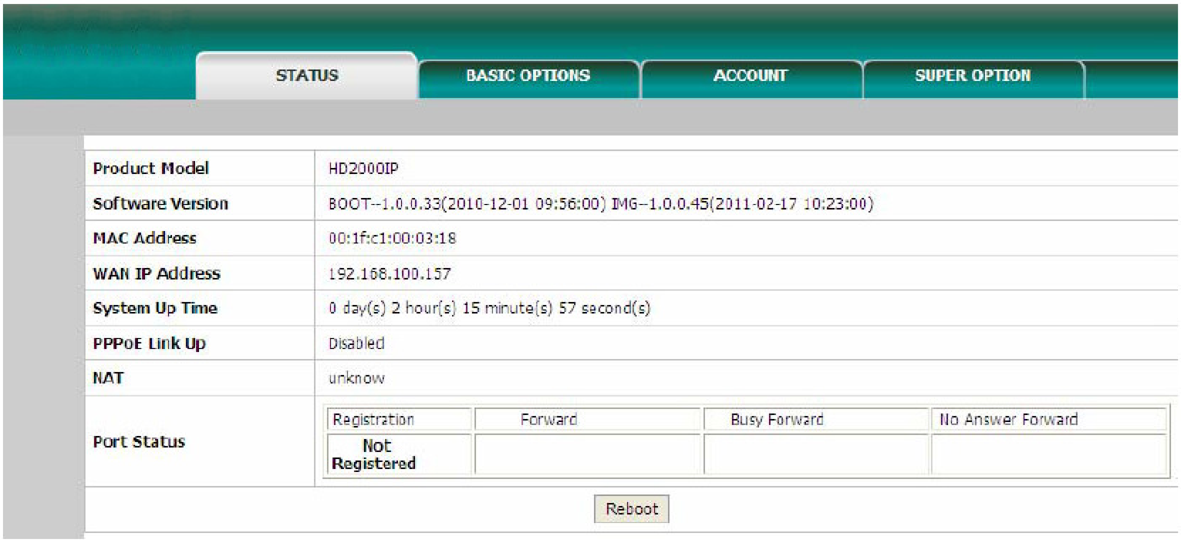
Advanced User Password for access to Super User Options: admin

End User Password for access to Basic User Options: 1234

8.1.5. Configuration Options and Explanations

After a correct password is entered in the login screen, the embedded web server inside the HD2000IP will show the configuration page, which is explained in details below:

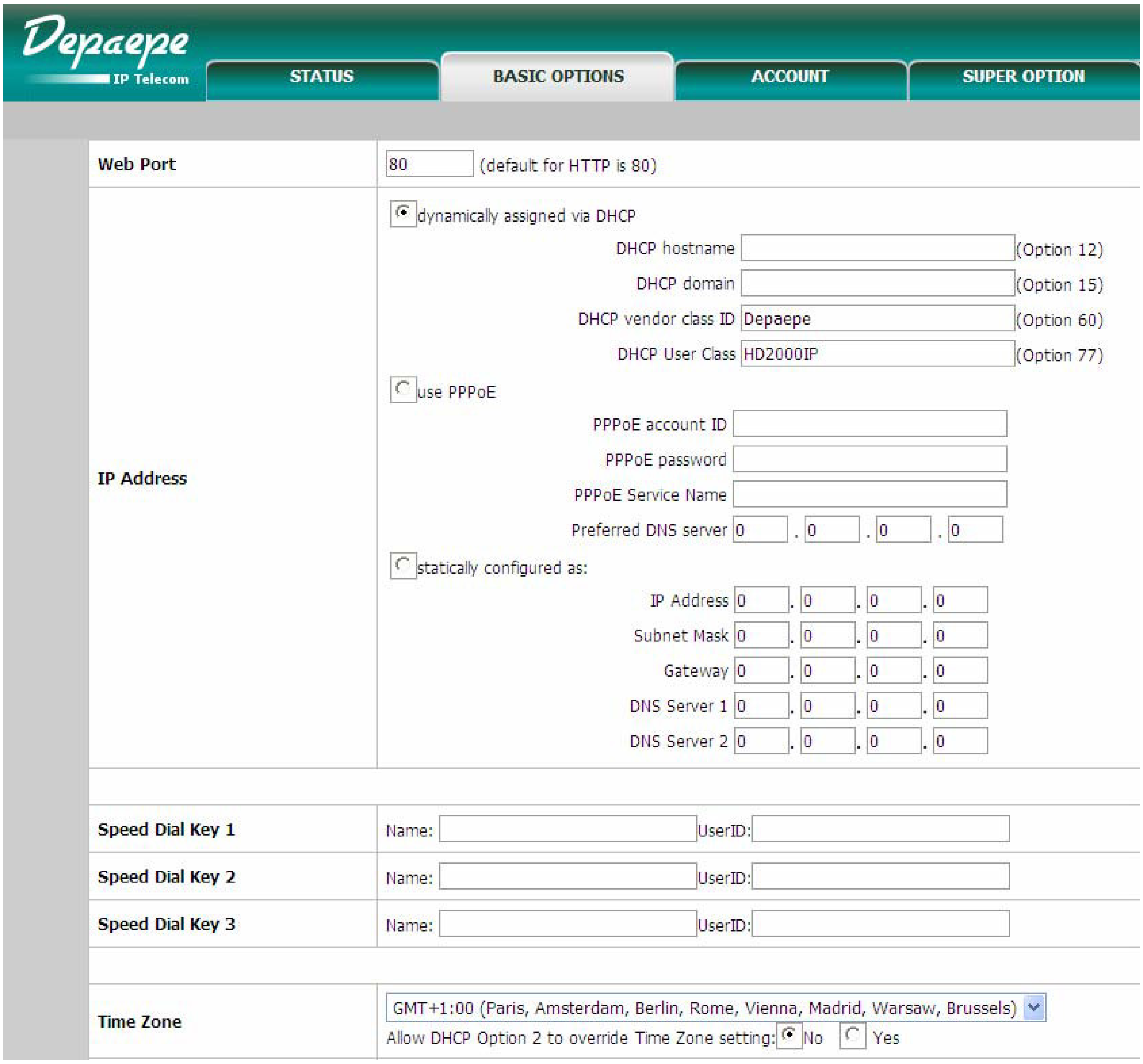
8.1.5.1 Device Status



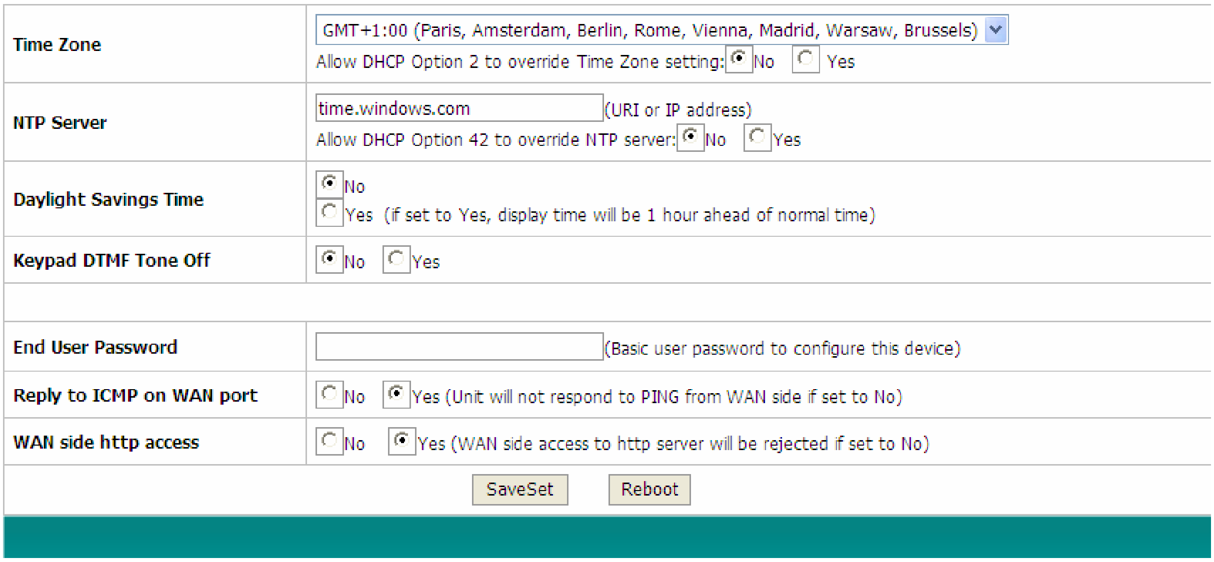
Device Status explained

Options	Meaning
Product Model	Contains the product model info
Software Version	• **Program:** This is the main software (firmware) release number, always used to identify the software (firmware) system of the phone. • **Boot:** Booting code version number
MAC Address	The device ID, in HEX format. This is a very important ID for ISP troubleshooting.
WAN IP Address	This field shows IP address of HD2000IP.
System Up Time	Shows system up time since the last reboot.
PPPoE Link Up	Indicates whether the PPPoE connection is enabled (connected to a modem).
NAT	Indicates the type of NAT connection used by the HD2000IP via its WAN port. Based on STUN protocol.
Port Status	Shows several information regarding the individual FXS ports. **Registration** **Forward** **Busy Forward** **No answer Forward** Indicates whether account are registered to the related SIP server(s).

8.1.5.2 Basic Options

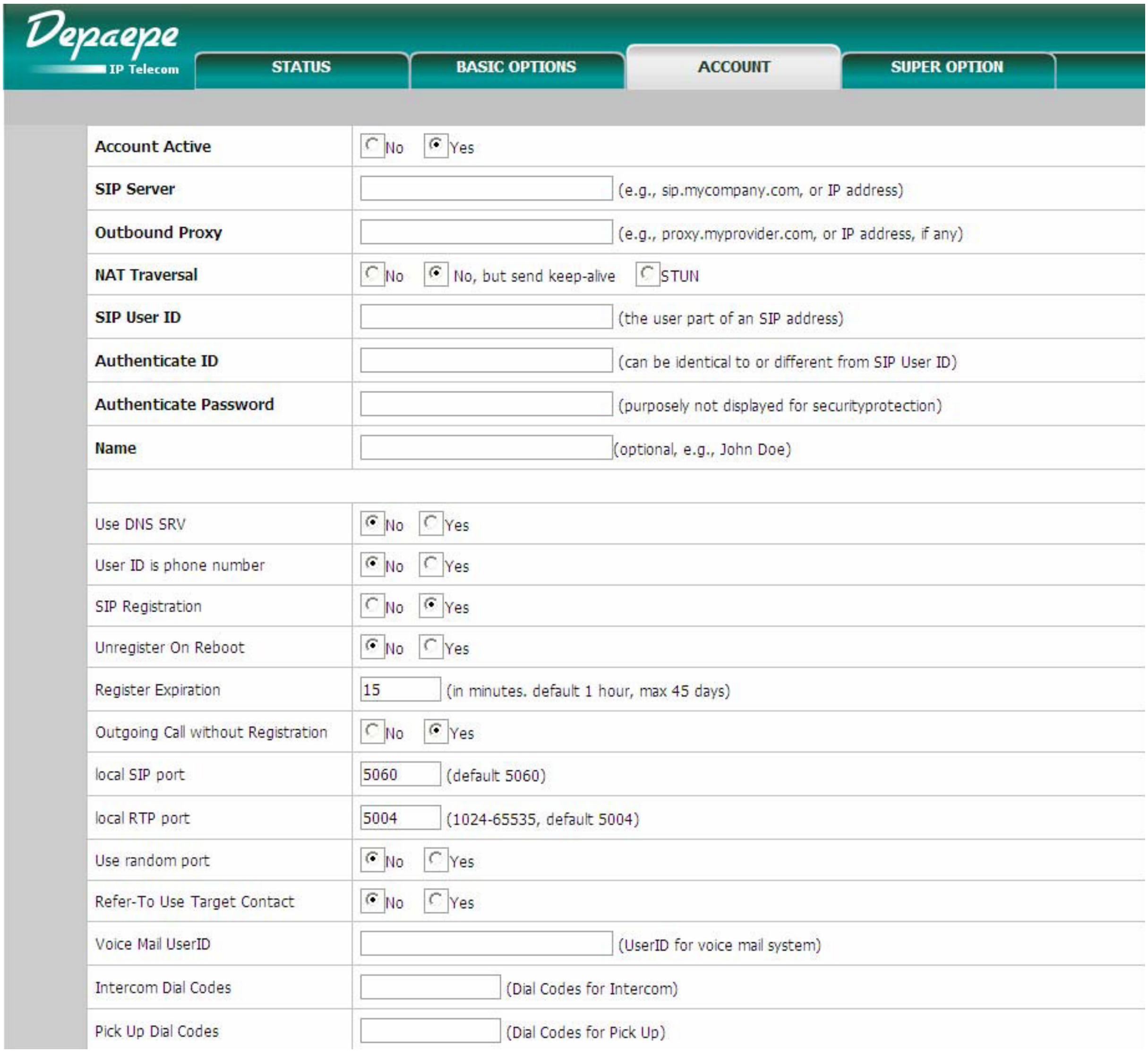


BASIC OPTIONS SETTING	
Options	Meaning
Web Port	Default is 80.
IP Address	There HD2000IP operates in two modes: 1. **DHCP mode**: all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory.) The HD2000IP acquires its IP address from the first DHCP server it discovers on its LAN. The DHCP option is reserved for NAT router mode. To use the PPPoE feature, set the PPPoE account settings. The HD2000IP establishes a PPPoE session if any of the PPPoE fields are set. 2. **Static IP mode**: configure all of the following fields: IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary). These fields are set to zero by default.
Speed Dial Key1~3	HD2000IP has defined 3 speed dial keys. After you program numbers for this key. You can touch a speed dial key and then the call will be originated.



BASIC OPTIONS SETTING	
Options	Meaning
Time Zone	This parameter controls the date/time display according to the specified time zone.
NTP server	URI or IP address of the NTP (Network Time Protocol) server. Used by the phone to synchronize the date and time.
Daylight Savings Time	This parameter controls time displayed in daylight savings time. If set to “Yes”, then the displayed time will be 1 hour ahead of normal time. The “Optional Rule” is configured to automatically adjust the Daylight Savings Time (DST) based on the rule set in this field. Rule Syntax: • start-time; end-time; saving • Both start-time and end-time have the same syntax: month, day, weekday, hour, minute • month: 1,2,3,..,12 (for Jan, Feb, .., Dec) • day: [+|-]1,2,3,..,31 • weekday: 1, 2, 3, .., 7 (for Mon, Tue, .., Sun), or 0 which means the daylight saving rule is not based on week days but based on the day of the month. • hour: hour (0-23), minute: minute (0-59) If “weekday” is 0, it means the date to start or end daylight saving is at exactly the given date. In that case, the “day” value must not be negative. If “weekday” is not zero and “day” is positive, then the daylight saving starts on the first “day” the iteration of the weekday (e.g.: 1st Sunday, 3rd Tuesday etc). If “weekday” is not zero and “day” is negative, then the daylight saving starts on

	the last “day” the iteration of the weekday (e.g.: last Sunday, 3rd last Tuesday etc). The saving is in the unit of minutes. The saving time may also be preceded by a negative (-) sign if subtraction is desired instead of addition. The default value is set for US, the “Automatic Daylight Saving Time Rule” shall be set to “3,2,7,2,0;11,1,7,2,0;60” Examples US/Canada where daylight saving time is applicable: 03,02,7,02,00;11,1,7,02,00;60 This means the daylight saving time starts from the second Sunday of March at 2AM and ends the first Sunday of November at 2AM. The saving is 60 minutes.
Keypad DTMF Tone off	Enable or disable the sound when press a key
End User Password	This contains the password to access the Web Configuration Menu. This field is case sensitive.
Reply to ICMP on WAN port	If set to “Yes”, the HD2000IP will respond to the PING command from other computers, but it also is vulnerable to the DOS attack. Default is **No**.
Wan Side Http Access	If this parameter is set to “No”, the HTML configuration update via WAN port is disabled.



Depaepe IP Telecom | STATUS | BASIC OPTIONS | ACCOUNT | SUPER OPTION

Account Active	○ No ◉ Yes
SIP Server	(e.g., sip.mycompany.com, or IP address)
Outbound Proxy	(e.g., proxy.myprovider.com, or IP address, if any)
NAT Traversal	○ No ◉ No, but send keep-alive ○ STUN
SIP User ID	(the user part of an SIP address)
Authenticate ID	(can be identical to or different from SIP User ID)
Authenticate Password	(purposely not displayed for securityprotection)
Name	(optional, e.g., John Doe)
Use DNS SRV	◉ No ○ Yes
User ID is phone number	◉ No ○ Yes
SIP Registration	○ No ◉ Yes
Unregister On Reboot	◉ No ○ Yes
Register Expiration	15 (in minutes. default 1 hour, max 45 days)
Outgoing Call without Registration	○ No ◉ Yes
local SIP port	5060 (default 5060)
local RTP port	5004 (1024-65535, default 5004)
Use random port	◉ No ○ Yes
Refer-To Use Target Contact	◉ No ○ Yes
Voice Mail UserID	(UserID for voice mail system)
Intercom Dial Codes	(Dial Codes for Intercom)
Pick Up Dial Codes	(Dial Codes for Pick Up)

Account	
Settings Options	**Meaning**
Account active	When set to Yes the FXS port is activated
SIP Server	This field contains the URI string or the IP address (and port, if different from 5060) of the SIP proxy server. e.g., the following are some valid examples: sip.my-voip-provider.com, or sip:my-company-sip-server.com, or 192.168.1.200:5066
Outbound Proxy	This field contains the URI string or the IP address (and port, if different from 5060) of the outbound proxy. If there is no outbound proxy, this field SHOULD be left blank. If not blank, all outgoing requests will be sent to this outbound proxy.
NAT Traversal	This parameter defines whether or not the HD2000IP NAT traversal mechanism is activated. If activated (by choosing "Yes") and a STUN server is also specified, then the HD2000IP performs according to the STUN client specification. Using this mode, the embedded STUN client will detect if and what type of firewall/NAT is being used. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the HD2000IP will use its mapped public IP address and port in all of its SIP and SDP messages. If the NAT Traversal field is set to "Yes" *with no specified STUN server*, the HD2000IP will periodically (every 20 seconds or so) send a blank UDP packet no payload data) to the SIP server to keep the "hole" on the NAT open.
SIP User ID	SIP service subscriber's User ID
Authenticate ID	SIP service subscriber's Authenticate ID. Can be identical to or different from SIP User ID
Authenticate Password	SIP service subscriber's account password
Name	SIP service subscriber's name which will be used for Caller ID display
Use DNS SRV	Default is No. If set to Yes the client will use DNS SRV for server lookup
User ID is Phone Number	If the HD2000IP has an assigned PSTN telephone number, this field should be set to "Yes". Otherwise, set it to "No". If "Yes" is set, a "user=phone" parameter will be attached to the "From" header in SIP request
SIP Registration	This parameter controls whether the HD2000IP needs to send REGISTER messages to the proxy server. The default setting is "Yes".
Unregister on Reboot	Default is "No." If set to "Yes", then the SIP user will be unregistered on reboot.
Register Expiration	This parameter allows the user to specify the time frequency (in

	minutes) the HD2000IP refreshes its registration with the specified registrar. The default interval is 60 minutes (or 1 hour). The maximum interval is 65535 minutes (about 45 days).
Outgoing call without Registration	Default is **No**. If set to “Yes,” user can place outgoing calls even when not registered (if allowed by ITSP) but is unable to receive incoming calls.
Local SIP port	This parameter defines the local SIP port the HD2000IP will listen and transmit. The default value for FXS port is 5060. The default value for FXO port is 5062
Local RTP port	This parameter defines the local RTP-RTCP port pair the HD2000IP will listen and transmit. It is the base RTP port for channel 0. When configured, channel 0 will use this port _value for RTP and the port_value+1 for its RTCP; channel 1 will use port_value+2 for RTP and port_value+3 for its RTCP. The default value for FXS port is 5004. The default value for FXO port is 5008.
Use Random Port	This parameter, when set to Yes, will force random generation of both the local SIP and RTP ports. This is usually necessary when multiple HD2000IP are behind the same NAT.
Refer-To Use Target Contact	Default is NO. If set to YES, then for Attended Transfer, the “Refer-To” header uses the transferred target’s Contact header information.
Voice Mail user ID	The number for check voice mail
Intercom Dial codes	Dial this code for intercom feature
Pickup Dialcodes	Dial this code for pickup feature

DTMF Payload Type	101
DTMF in Audio	No Yes
DTMF via RFC2833	No Yes
DTMF via SIP INFO	No Yes
Send Flash Event	No Yes (Flash will be sent as a DTMF event if set to Yes)
Enable Call Features	No Yes (if Yes, call features using star codes will be supported locally)
Offhook Auto-Dial	(User ID/extension to dial automatically when offhook)
Proxy-Require	
Use NAT IP	(used in SIP/SDP message if specified)
Preferred Codecs (in listed order)	choice 1: current setting is " PCMU" choice 2: current setting is " PCMA" choice 3: current setting is " G.729A/B" choice 4: current setting is " G.726-32" choice 5: current setting is " G.723.1" choice 6: current setting is " PCMU"
Voice Frames per TX	2 (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)
G723 rate	6.3kbps encoding rate 5.3kbps encoding rate
VAD	No Yes
Symmetric RTP	No Yes
Jitter Buffer Type	Fixed Adaptive
Jitter Buffer Length	Low Medium High

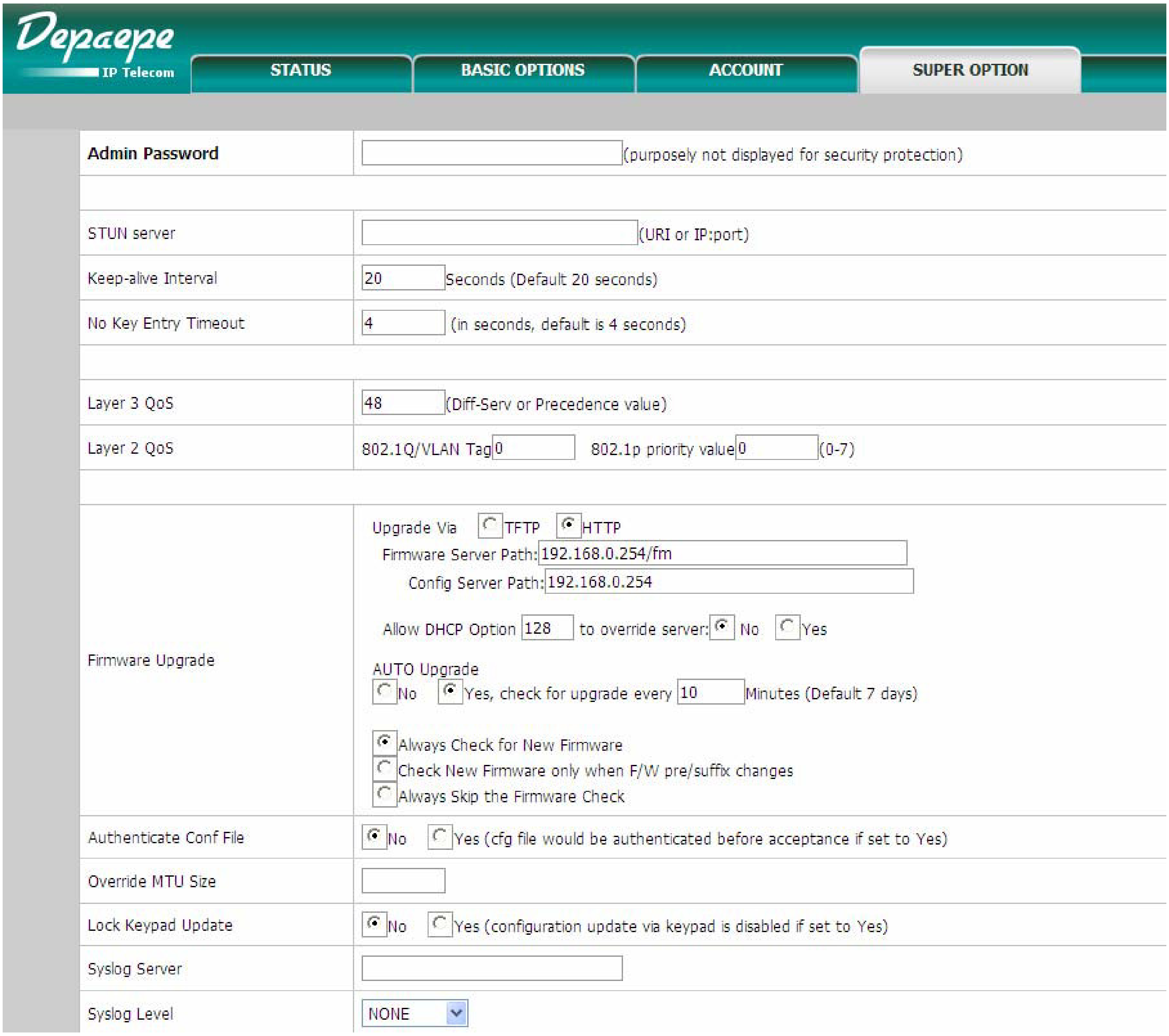
DTMF Payload Type	This parameter sets the payload type for DTMF using RFC2833
DTMF in Audio	Send DTMF as inband (in-audio).
DTMF via RFC2833	Send DTMF via RTP (According to RFC 2833).
DTMF via SIP INFO	Send DTMF via SIP INFO message.
Send Flash Event	This parameter allows users to control whether to send an SIP NOTIFY message indicating the Flash event, or just to switch to the voice channel when users press the Flash key.
Enable Call Features	Default is No. If set to Yes, Call Forwarding & Do-Not-Disturb are supported locally (see P.17)
Offhook Auto-Dial	This parameter allows users to configure a User ID or extension number to be automatically dialed upon offhook. Please note that only the user part of a SIP address needs to be entered here. The HD2000IP will automatically append the “@” and the host portion of the corresponding SIP address.
Proxy-Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
USE NAT IP	NAT IP address used in SIP/SDP message. Default is blank.
Preferred Codecs	The HD2000IP supports up to 7 different Vocoder types including G.711 A-/U-law_G.723.1, G.726, G.728, G.729A/B, iLBC. Depending on the product model, some of these Vocoders may not be provided in standard release. Users can configure Vocoders in a preference list that will be included with the same preference order in SDP message. The first Vocoder in this list can be entered by choosing the appropriate option in “Choice 1”. Similarly, the last Vocoder in this list can be entered by choosing the appropriate option in “Choice 7”.

Voice Frames per TX	This field contains the number of voice frames to be transmitted in a single packet. When setting this value, the user should be aware of the requested packet time (used in SDP message) as a result of configuring this parameter. This parameter is associated with the first vocoder in the above vocoder Preference List or the actual used payload type negotiated between the 2 conversation parties at run time. e.g., if the first vocoder is configured as G723 and the "Voice Frames per TX" is set to be 2, then the "ptime" value in the SDP message of an INVITE request will be 60ms because each G723 voice frame contains 30ms of audio. Similarly, if this field is set to be 2 and if the first vocoder chosen is G729 or G711 or G726, then the "ptime" value in the SDP message of an INVITE request will be 20ms. If the configured voice frames per TX exceeds the maximum allowed value, the HD2000IP will use and save the maximum allowed value for the corresponding first vocoder choice. The maximum value for PCM is 10(x10ms) frames; for G726, it is 20 (x10ms) frames; for G723, it is 32 (x30ms) frames; for G729/G728, 64 (x10ms) and 64 (x2.5ms) frames respectively.
G723 Rate	This defines the encoding rate for G723 vocoder. By default, 6.3kbps rate is chosen.
VAD	Default is **No**. VAD allows detecting the absence of audio and conserve bandwidth by preventing the transmission of "silent packets" over the network.
Symmetric RTP	Default is **No**. When set to Yes the device will change the destination to send RTP packets to the source IP address and port of the inbound RTP packet last received by the device.
Jitter Buffer Type	Select either Fixed or Adaptive based on network conditions.
Jitter Buffer Length	Select Low, Medium or High based on network conditions.

Account Ring Tone	○ system ring tone ○ custom ring tone 1 ○ custom ring tone 2 ◉ custom ring tone 3
Disable Call-Waiting	◉ No ○ Yes
Disable Call-Waiting Tone	◉ No ○ Yes
Ring Timeout	60 (10-300 seconds, default is 60 seconds)
Use # as Dial Key	○ No ◉ Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)
Dial Plan	{[x*]+}
SUBSCRIBE for MWI	○ No, do not send SUBSCRIBE for Message Waiting Indication ◉ Yes, send periodical SUBSCRIBE for Message Waiting Indication
Send Anonymous	◉ No ○ Yes (caller ID will be blocked if set to Yes)
Anonymous Call Rejection	◉ No ○ Yes
Check SIP User ID for incoming INVITE	◉ No ○ Yes
Auto Answer	◉ No ○ Yes
Allow Auto Answer by Call-Info	○ No ◉ Yes
Session Expiration	180 (in seconds. default 180 seconds)
Turn off speaker on remote disconnect	○ No ◉ Yes
Min-SE	90 (in seconds. default and minimum 90 seconds)
Caller Request Timer	◉ No ○ Yes (Request for timer when making outbound calls)
Callee Request Timer	◉ No ○ Yes (When caller supports timer but did not request one)
Force Timer	◉ No ○ Yes (Use timer even when remote party does not support)
UAC Specify Refresher	○ UAC ○ UAS ◉ Omit (Recommended)
UAS Specify Refresher	◉ UAC ○ UAS (When UAC did not specify refresher tag)
Force INVITE	◉ No ○ Yes (Always refresh with INVITE instead of UPDATE)
Hook Flash Timing	minimum: 30 maximum: 100 (Note: In 50-1200 milliseconds range)

<table>
<tr><td>Account Ring Tone</td><td>There are 4 uniquely defined ring tones:
• One (1) System Ring Tone: when selected, all calls will ring with system ring tone.
• Three (3) Customer Ring Tones: when selected, incoming calls from designated account will play selected ring tone.</td></tr>
<tr><td>Disable Call -Waiting</td><td>Default is No.</td></tr>
<tr><td>Disable Call –Waiting Tone</td><td>Default is No. This is to disable the stutter Call Waiting Tone when a Call Waiting call arrives. The CWCID will still be displayed.</td></tr>
<tr><td>Ring Timeout</td><td>Incoming call will stop ringing when not picked up given a specific period of time.</td></tr>
<tr><td>Use # as Send Key</td><td>This parameter allows users to configure the “#” key to be used as the “Send” (or “Dial”) key. If set to “Yes”, pressing this key will immediately trigger the sending of dialed string collected so far. In this case, this key is essentially equivalent to the “(Re)Dial” key. If set to “No”, this “#” key will then be included as part of the dial string to be sent out.</td></tr>
<tr><td>Dial Plan</td><td>Dial Plan Rules:-
1. Accept Digits: 1,2,3,4,5,6,7,8,9,0
2. Grammar: x - any digit from 0-9;
a. xx+ - at least 2 digit number;
b. ^ - exclude;
c. [3-5] - any digit of 3, 4, or 5;
d. [147] - any digit 1, 4, or 7;
e. <2=011> - replace digit 2 with 011 when dialing
• Example 1: {[369]11 | 1617xxxxxxx} –
Allow 311, 611, 911, and any 10 digit numbers of leading digits 1617
• Example 2: {^1900x+ | <=1617>xxxxxxx} –
Block any number of leading digits 1900 and add prefix 1617 for Any dialed 7 digit numbers
• Example 3: {1xxx[2-9]xxxxxx | <2=011>x+} –
Allow any length of number with leading digit 2 and 10 digit-numbers of leading digit 1 and leading exchange number between 2 and 9; If leading digit is 2, replace leading digit 2 with 011 before dialing
3. Default: Outgoing - {x+}</td></tr>
<tr><td>Subscribe for MWI</td><td>Default is No. When set to “Yes” a SUBSCRIBE for Message Waiting Indication will be sent periodically.</td></tr>
<tr><td>Send Anonymous</td><td>If this parameter is set to “Yes”, the “From” header in outgoing INVITE message will be set to anonymous, essentially blocking the Caller ID from displaying.</td></tr>
</table>

Anonymous Call Rejection	Default is **No**. If set to Yes, incoming calls with anonymous Caller ID will be rejected with 486 Busy message.
Check SIP User ID for Incoming INVITE	When the phone receive INVITE, at first it will check if it is right the 'To' in the message.
Auto answer	Enable or disable the 'auto answer' feature
Allow Auto answer by Call-Info	Enable or disable 'auto answer' when some call use Call-info to activate this feature
Session Expiration	The SIP Session Timer extension enables SIP sessions to be periodically "refreshed" via a SIP request (UPDATE, or re-INVITE. Once the session intervaexpires, if there is no refresh via a UPDATE or re-INVITE message, the session is terminated. Session Expiration is the time (in seconds) at which the session is considered timed out, provided no successful session refresh transaction occurs beforehand. The default value is 180 seconds.
Turn off speaker on remote disconnect	When the remote user hang up, the phone will disable the speaker.
Min-SE	Defines the minimum session expiration (in seconds). Default is 90 seconds.
Caller Request Timer	If set to "Yes", the phone will use session timer when it makes outbound calls if remote party supports session timer.
Callee Request Timer	If selecting "Yes", the phone will use session timer when it receives inbound calls with session timer request.
Force Timer	If set to "Yes", the phone will use session timer even if the remote party does not support this feature. If set to "No", the session timer is enabled only when the remote party supports this feature. To turn off Session Timer, select "No" for Caller Request Timer, Callee Request Timer, and Force Timer.
UAC Specify Refresher	As a Caller, select UAC to use the phone as the refresher, or UAS to use the Callee or proxy server as the refresher.
UAS Specify Refresher	As a Callee, select UAC to use caller or proxy server as the refresher, or UAS to use the phone as the refresher.
Force Invite	Session Timer can be refreshed using INVITE method or UPDATE method. Select "Yes" to use INVITE method to refresh the session timer.
Hook flash Timing	The time break for hook flash


Depaepe IP Telecom
STATUS
BASIC OPTIONS
ACCOUNT
SUPER OPTION
Admin Password
(purposely not displayed for security protection)
STUN server
(URI or IP:port)
Keep-alive Interval
20
Seconds (Default 20 seconds)
No Key Entry Timeout
4
(in seconds, default is 4 seconds)
Layer 3 QoS
48
(Diff-Serv or Precedence value)
Layer 2 QoS
802.1Q/VLAN Tag 0
802.1p priority value 0
(0-7)
Firmware Upgrade
Upgrade Via TFTP HTTP
Firmware Server Path: 192.168.0.254/fm
Config Server Path: 192.168.0.254
Allow DHCP Option 128 to override server: No Yes
AUTO Upgrade
No Yes, check for upgrade every 10 Minutes (Default 7 days)
Always Check for New Firmware
Check New Firmware only when F/W pre/suffix changes
Always Skip the Firmware Check
Authenticate Conf File
No Yes (cfg file would be authenticated before acceptance if set to Yes)
Override MTU Size
Lock Keypad Update
No Yes (configuration update via keypad is disabled if set to Yes)
Syslog Server
Syslog Level
NONE

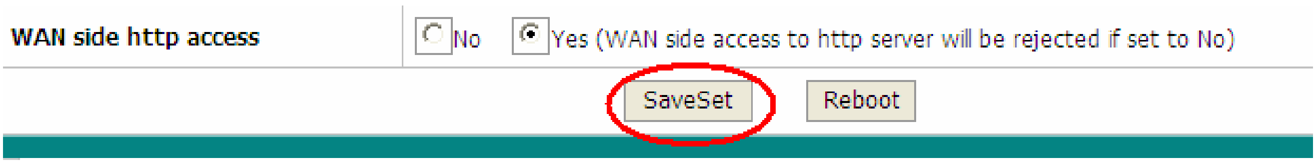
Super Options	
Setting options	**Meaning**
Admin Password	This contains the password to access the Advanced Web Configuration page. This field is case sensitive. Only the administrator can configure the “Advanced Settings” page. Password field is purposely left blank for security reasons after clicking update and saved. The maximum password length is 26 characters,only digit or letter.
STUN server is	IP address or Domain name of the STUN server.
Keep-alive interval	This parameter specifies how often the HD2000IP sends a blank UDP packet to the SIP server in order to keep the “hole” on the NAT open. Default is 20 seconds. Minimum value is 20 seconds.
No Key Entry Timeout	Default is 4 seconds.
Layer3 Qos	This field defines the layer 3 QoS parameter which can be the value used for IP Precedence or Diff-Serv or MPLS. Default value is **48**.
Layer2 Qos_VOIP_	Value used for layer 2 VLAN tag. Default setting is **blank**
Layer2 Qos _PC_	Layer 2 QoS settings for LAN port device traffic. Default setting is blank. VLAN supported equipment is required if user needs to change these settings.
Firmware Upgrade	Support firmware upgrade via TFTP or HTTP, Support for Authenticating configuration file before accepting changes User specific URL for configuration file and firmware files
Authenticate Conf File	Default is “No”. If set to “Yes”, configuration file would be authenticated before acceptance. End user should use default setting.
Override MTU Size	
Lock Keypad Update	If set to “Yes”, the configuration update via keypad is disabled.
Syslog Sever	The IP address or URL of System log server. This feature is especially useful for the ITSP (Internet Telephone Service Provider)
Syslog level	Select the HD2000IP to report the log level. Default is NONE. The level is one of DEBUG, INFO, WARNING or ERROR. Syslog messages are sent based on the following events: 1. product model/version on boot up (INFO level) 2. NAT related info (INFO level) 3. sent or received SIP message (DEBUG level) 4. SIP message summary (INFO level) 5. inbound and outbound calls (INFO level) 6. registration status change (INFO level) 7. negotiated codec (INFO level) 8. Ethernet link up (INFO level) 9. SLIC chip exception (WARNING and ERROR levels) 10. memory exception (ERROR level) The Syslog uses USER facility. In addition to standard Syslog payload, it contains the following components: GS_LOG: [device MAC address][error code] error message *Example*: May 19 02:40:38 192.168.1.14 GS_LOG: [00:0b:82:00:a1:be][000] Ethernet link is up

Download Device Configuration:	Download
Reset To Factory Settings	Reset To Factory Settings
Distinctive Ring Tone	Custom ring tone 1, used if incoming caller ID is Custom ring tone 2, used if incoming caller ID is Custom ring tone 3, used if incoming caller ID is
System Ring Tone	f1=440@-13,f2=480@-13,c=2000/ (default: f1=440,f2=480,c=200/400;)
Call Progress Tones	**Syntax: f1=freq@vol, f2=freq@vol, c=on1/off1-on2/off2-on3/off3; [...]** **Note: freq: 0 - 4000Hz; vol: -30 - 0dBm** Dial Tone f1=350@-13,f2=440@-13,c=0/0; Ringback Tone f1=440@-19,f2=480@-19,c=2000/4000; Busy Tone f1=480@-24,f2=620@-24,c=500/500; Reorder Tone f1=480@-24,f2=620@-24,c=250/250; Confirmation Tone f1=350@-11,f2=440@-11,c=100/100-100/100-100/100 Call Waiting Tone f1=440@-13,c=300/10000-300/10000-0/0;
	SaveSet Reboot
Restore Configuration	Parcourir... Restore Configuration

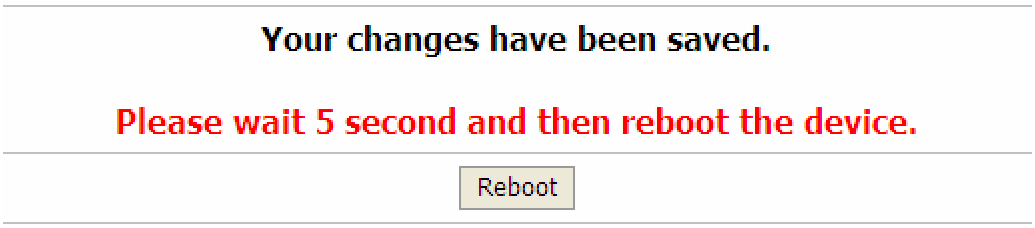
Download Device Configuration	Used to download the change you have made.
Reset To factory settings	Restore the phone to factory's configuration.
Distinctive Ring Tone	Caller ID must be configured. Select a Distinctive Ring Tone 1 through 3 for a particular Caller ID. The HD2000IP will ONLY use selected ring tones for particular Caller IDs. For all other calls, the HD2000IP will use System Ring Tone. When selected and no Caller ID is configured, the selected ring tone will be used for all incoming calls.
System Ring Tone	System ring tone. Default is North American standard. Adjust system ring tone frequencies and cadences based on local telecom standard.
Call Progress Tones	Using these settings, users can configure ring or tone frequencies based on parameters from local telecom. By default, they are set to North American standard. Frequencies should be configured with known values to avoid uncomfortable high pitch sounds. **Syntax**: f1=val,f2=val[,c=on1/off1[-on2/off2[-on3/off3]]]; (Frequencies are in Hz and cadence on and off are in 10ms) ON is the period of ringing ("On time" in 'ms') while OFF is the period of silence. In order to set a continuous ring, OFF should be zero. Otherwise it will ring ON ms and a pause of OFF ms and then repeat the pattern. Up to three cadences are supported.
Restore Configuration	User can restore the before configuration from the configuration file saved at local pc

8.1.6. Saving the Configuration Changes

Once a change is made, users should click on the "SaveSet" button in the Configuration page, as follow:



The HD2000IP will then display the following screen to confirm that the changes have been saved. Please allow 5 seconds before rebooting the device.



8.1.7. Rebooting the HD2000IP

You can reboot the HD2000IP by clicking on the "Reboot" button after each update to the configuration page. Alternatively, you can reboot by unplugging the power supply of the HD2000IP and then powering it on again. If your HD2000IP ever becomes "stuck" or un-responsive, you can unplug the power supply to reboot it. Frequent rebooting by unplugging the power supply is not recommended and should not be necessary.

8.1.8. Configuration through a Central Server

HD2000IP devices can be automatically configured from a central provisioning system.

When HD2000IP boots up, it will send TFTP or HTTP request to download configuration files. There are two configuration files, one is “cfg.txt” and the other is“cfg001fc1xxxxxx”, where “001fc1xxxxxx” is the MAC address of the HD2000IP.

For more information regarding configuration file format, please refer to the related technical documentation.

The configuration file can be downloaded via TFTP or HTTP from the central server. A service provider or an enterprise with large deployment of HD2000IPs can easily manage the configuration and service provisioning of individual devices remotely and automatically from a central server. The central provisioning system uses enhanced (NAT friendly) TFTP or HTTP (thus no NAT issues) and other communication protocols to communicate with each individual HD2000IP for firmware upgrade, etc.

About DHCP option supported

At present, HD2000IP support DHCP options, 2/12/15/42/43/60/66/128/150

1. Option 2--Time Offset
Basic Option->Time Zone: Allow DHCP Option 2 to override Time Zone setting:

2. Option 12--Host Name.
Basic Option->dynamically assigned via DHCP: DHCP hostname:

3. Option 15--Domain Name.
Basic Option->dynamically assigned via DHCP: DHCP domain:

4. Option 60--Class-identifier.
Basic Option->dynamically assigned via DHCP: DHCP vendor class ID:

5. Option 43--Vendor specific information.
Basic Option->dynamically assigned via DHCP: DHCP vendor specific information:

6. Option 42--NTP servers.
SUPER OPTIONS->NTP Server:Allow DHCP Option 42 to override NTP server

Note bellow,
SUPER OPTIONS->Firmware Upgrade and Provisioning:Allow DHCP Option, If you fill in 66, mean DHCP option 66; fill in 128, mean DHCP option 128; fill in 150, mean DHCP option 150,

7.Option 66--TFTP server name(if you select SUPER OPTION->Upgrade Via->TFTP), HTTP server name(if you select SUPER OPTION->Upgrade Via->HTTP)

8. Option 128--TFPT Server IP address. (if you select SUPER OPTION->Upgrade Via->TFTP),
HTTP Server IP address (if you select SUPER OPTION->Upgrade Via->HTTP)

9. Option 150--TFTP server address. (if you select SUPER OPTION->Upgrade Via->TFTP), HTTP server address (if you select SUPER OPTION->Upgrade Via->HTTP)

9. SOFTWARE UPGRADE

To upgrade software, HD2000IP can be configured with a TFTP server where the new code image is located. The TFTP upgrade can work in either static IP or DHCP mode using private or public IP address. It is recommended to set the TFTP server address in either a public IP address or on the same LAN with the HD2000IP.

There are two ways to set up the TFTP server to upgrade the firmware, namely through voice menu prompt or via the HD2000IP's Web configuration interface. To configure the TFTP server via voice prompt, follow section 8.1, once set up the TFTP IP address, power cycle the HD2000IP, the firmware will be fetched once the HD2000IP boots up.

To configure the TFTP server via the Web configuration interface, open up your browser to point at the IP address of the HD2000IP. Input the admin password to enter the configuration screen. From there, enter the TFTP server address in the designated field towards the bottom of the configuration screen.

Once the TFTP server is configured, please power cycle the HD2000IP.

TFTP process may take as long as 1 to 2 minutes over the Internet, or just 20+ seconds if it is performed on a LAN. Users are recommended to conduct TFTP upgrade in a controlled LAN environment if possible. For those who do not have a local TFTP server, DEPAEPE provides a NAT-friendly TFTP server on the public Internet for firmware upgrade. Please check the Service section of DEPAEPE's Web site to obtain this TFTP server's IP address.

NOTES:
When DEPAEPE IP Phone boot up, it will send TFTP or HTTP request to download configuration files, there are two configuration files, one is "cfg.txt" and the other is "cfg001fc1xxxxxx", where "001fc1xxxxxx" is the MAC address of the HD2000IP . These two files are for initial automatically provisioning purpose only, for normal TFTP or HTTP firmware upgrade, the following error messages in a TFTP or HTTP server log can be ignored.

10. RESTORE TO FACTORY DEFAULT SETTINGS

Warning:
Restoring to the factory default settings will delete all configuration information of the device.

Steps to follow in restoring to factory default settings by keypad:

a) Press "****" for voice prompt.

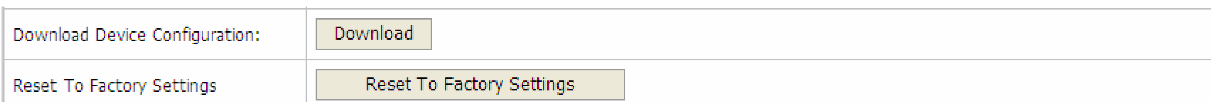
b) Enter "99" and then you will hear the voice prompt "Reset".

c) Enter the number "862584658050". A "click" sound will be heard.

d) Wait for 15 seconds.

The device is now restored to the factory default setting.

You can also reset the phone via web page. Enter in the super option, and click the 'Reset to factory setting' button. Then the device will be restored and reboot.



Bay Area Compliance Labs.Corp.

Bay Area Compliance Laboratories Corp. (Shenzhen)
6/F, the 3rd Phase of Wanli Industrial Building ShiHua Road,
FuTian Free Trade Zone, Shenzhen, Guangdong, China
www.baclcorp.com

ATTESTATION OF CONFORMITY

Date of Issue: 2010-11-30 Attestation Number: RSH101022001

Bay Area Compliance Laboratories Corp.(Shenzhen) hereby declares that testing has been completed and reports have been generated for;

Trade Name:	**DEPAEPE**
Product:	**HD2000IP**
Model Number:	**HD2000IP**
Applicant:	**HENRI DEPAEPE SAS** 75/77 rue du pre brochet 95110 Sannois

That this product has been assessed and found to comply against the following Standards;

EMC:	**EN 55022:2006 + A1:2007** **EN 55024:1998 + A1:2001 + A2:2003** **EN 61000-3-2:2006** **EN 61000-3-3:2008**
LVD:	**EN 60950-1:2006 + A11:2009**

Compliance to those standards are required for the following Directives;
Directive 2004/108/EC Electromagnetic Compatibility
Directive 2006/95/EC Low Voltage

Application of the CE Mark is permitted only after all applicable requirements are met in accordance with the European Union Rules, including the manufacturer's issuance of a "Declaration of Conformity." additional guidelines can be found at :
EMC Directive website: http://ec.europa.eu/enterprise/electr_equipment/emc/index.htm,
LVD Directive website: http://ec.europa.eu/enterprise/electr_equipment/lv/index.htm.
This attestation is specific to the standard(s) stated above and compliance to additional standards and/or directives may be required.

BAY AREA COMPLIANCE LABORATORIES CORP. INCORPORATED NOV. 12, 1996 CALIFORNIA

Attestation by: John Chan
Certification Manager

Signature

2010-11-30
Date

CI012-E