Depaepe Telecom

HD2000 IP Phone
User Manual

Made in France

SAS HENRI DEPAEPE

http://www.depaepe.com

Version V2.0

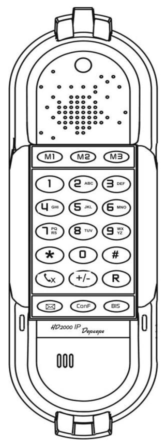
Table of Content

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.

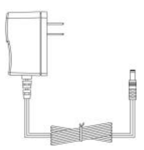
2. WHAT IS IN THE PACKAGE

The HD2000IP package contains:

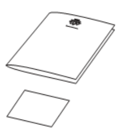


➢One HD2000IP VoIP Phone

➢One power supply (option)



➢Quick installation guide and quick user guide



➢ 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/B, G.726-32, G.722, GSM_FR, iLBC.
- Support standard encryption and authentication (DIGEST using MD5, MD5-sess), AES ?
- Support for Layer 2 QoS (802.1Q/VLAN Tag, 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, HTTP, FTP, HTTPS.
- 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)
PoE Class 4	Power consumption 1,600 mW Current consumption 34 mA
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%

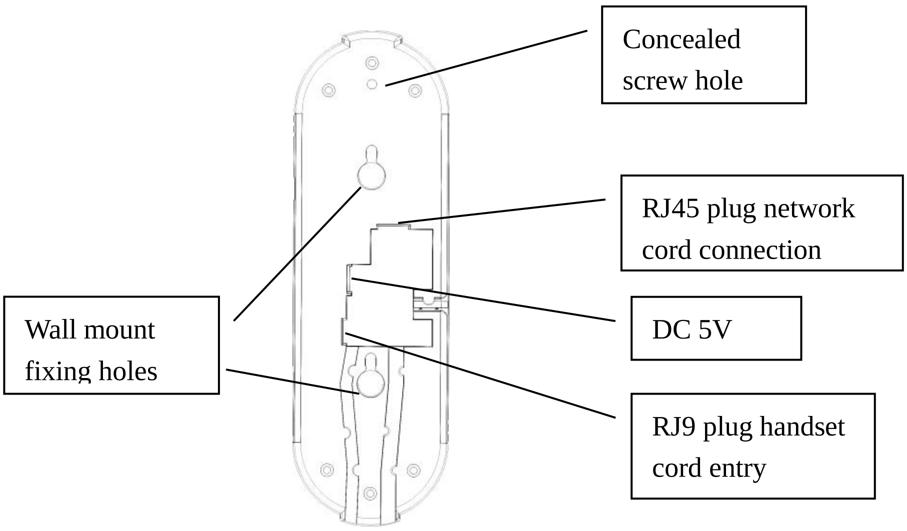
5. INSTALLATIONS

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.

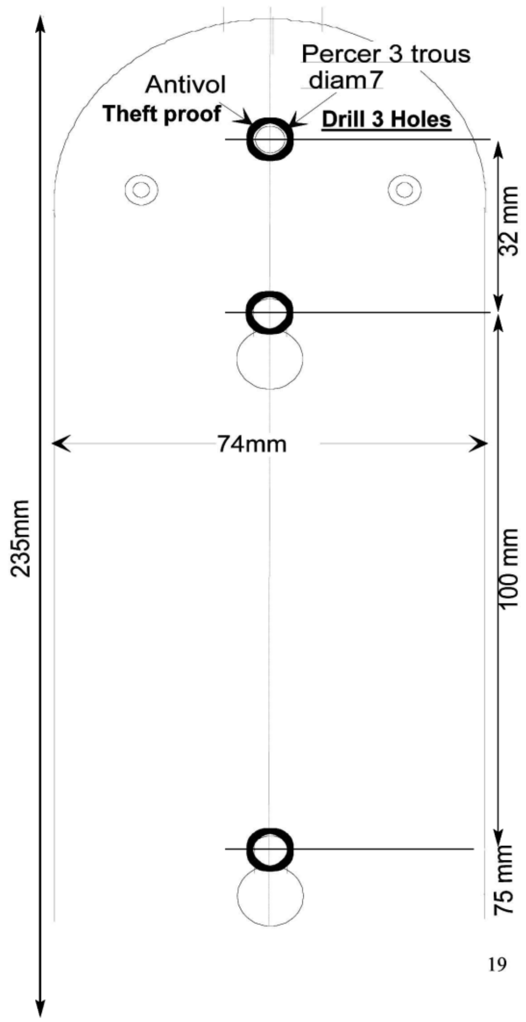
WARRANTY

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 DEPAEPE technologies, contact your DEPAEPE’s Sales and Service Representative for a RMA (Return Materials Authorization) number before you return the product. DEPAEPE reserves the right to remedy warranty policy without prior notification.

5.2. Walls mount installation:

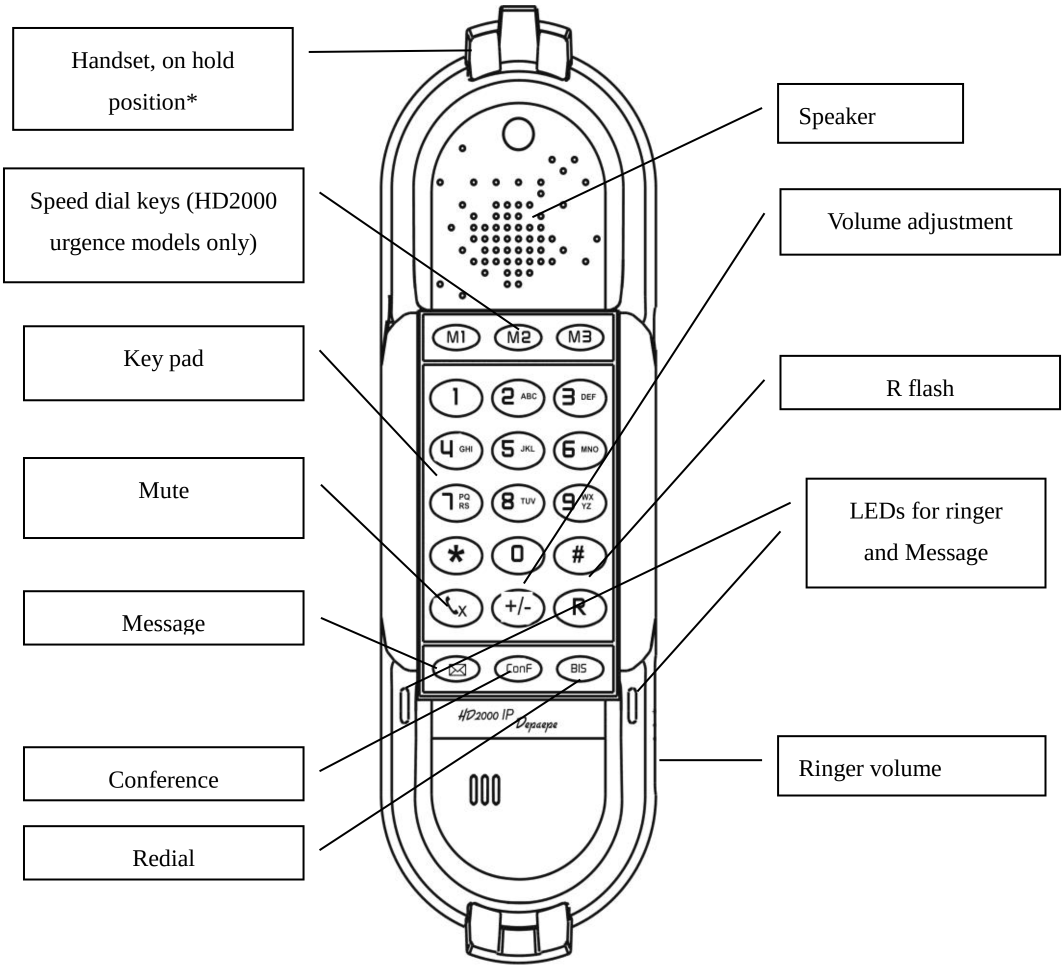
The HD2000 IP comes with a small plastic bag containing 1 thins screw and a foam plug to be used for preventing theft or unauthorized removal, 2 bigger screws for wall mount fixing.

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.

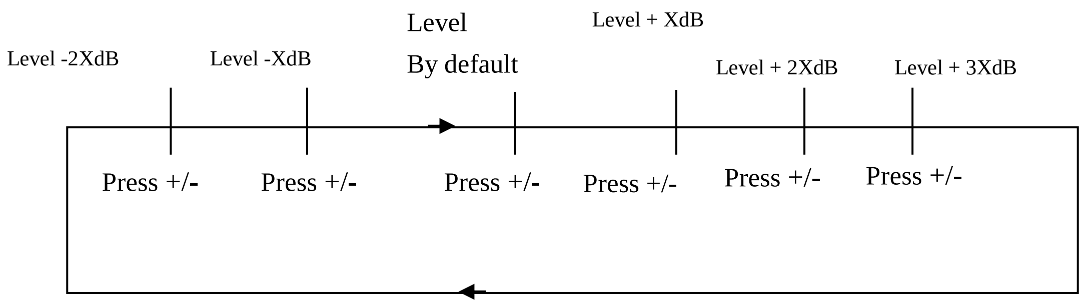
6. Get Familiar with the telephone



*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 (keypad models only)

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 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 Will Say the Following:	
Main Menu	"Enter a Menu Option"	Enter "*" for the next menu option Enter "#" to return to the main menu Enter 01 – 08,12 - 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

08	Your number is	The current number is announced.
12	WAN Port Web Access	Enter “9” to toggle between enable and 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 (keypad models only)

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 (keypad models only)

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 (keypad models only)

7.5.1. Call Features Tables

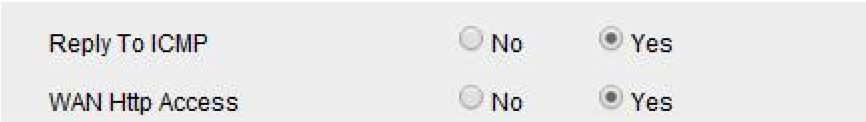
Following table shows the call features of HD2000IP :

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 Firefox browser. Below is a screen shot of the HD2000IP configuration page:



8.1.1. Get the IP address of the HD200IP:

Connect the HD2000 IP to a network via standard Ethernet cable, be 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 HD2000 IP urgency, use a standard network protocol analyzer (for e.g. Wireshark) to eavesdrop 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.

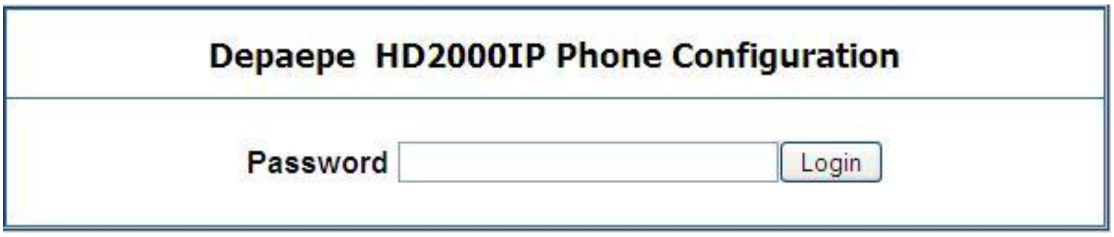
8.1.2. Accessing the Web Configuration

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. Login, Passwords

Password is case sensitive and all Depaepe devices come with factory default password as indicated below:

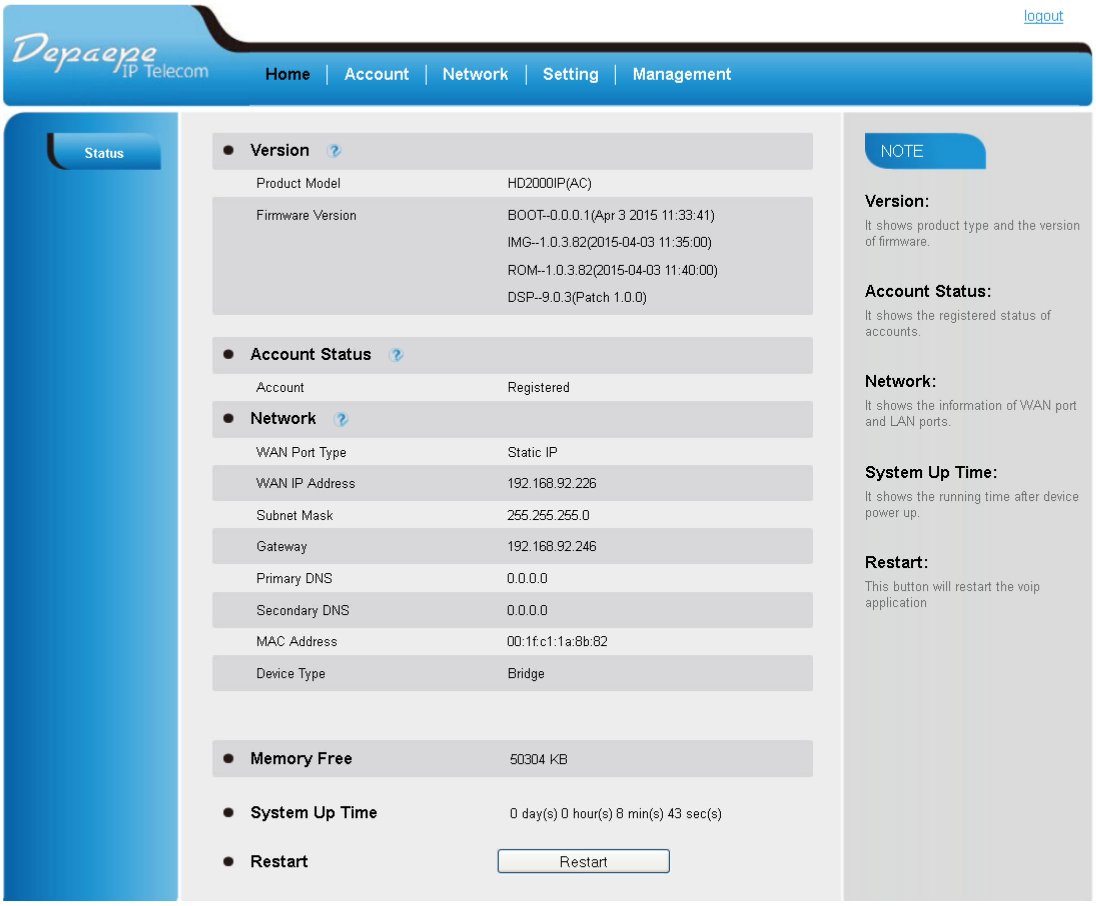
Advanced User Password for access to Super User Options: admin

End User Password for access to Basic User Options: 1234

8.1.5. Configuration Options and Explanations

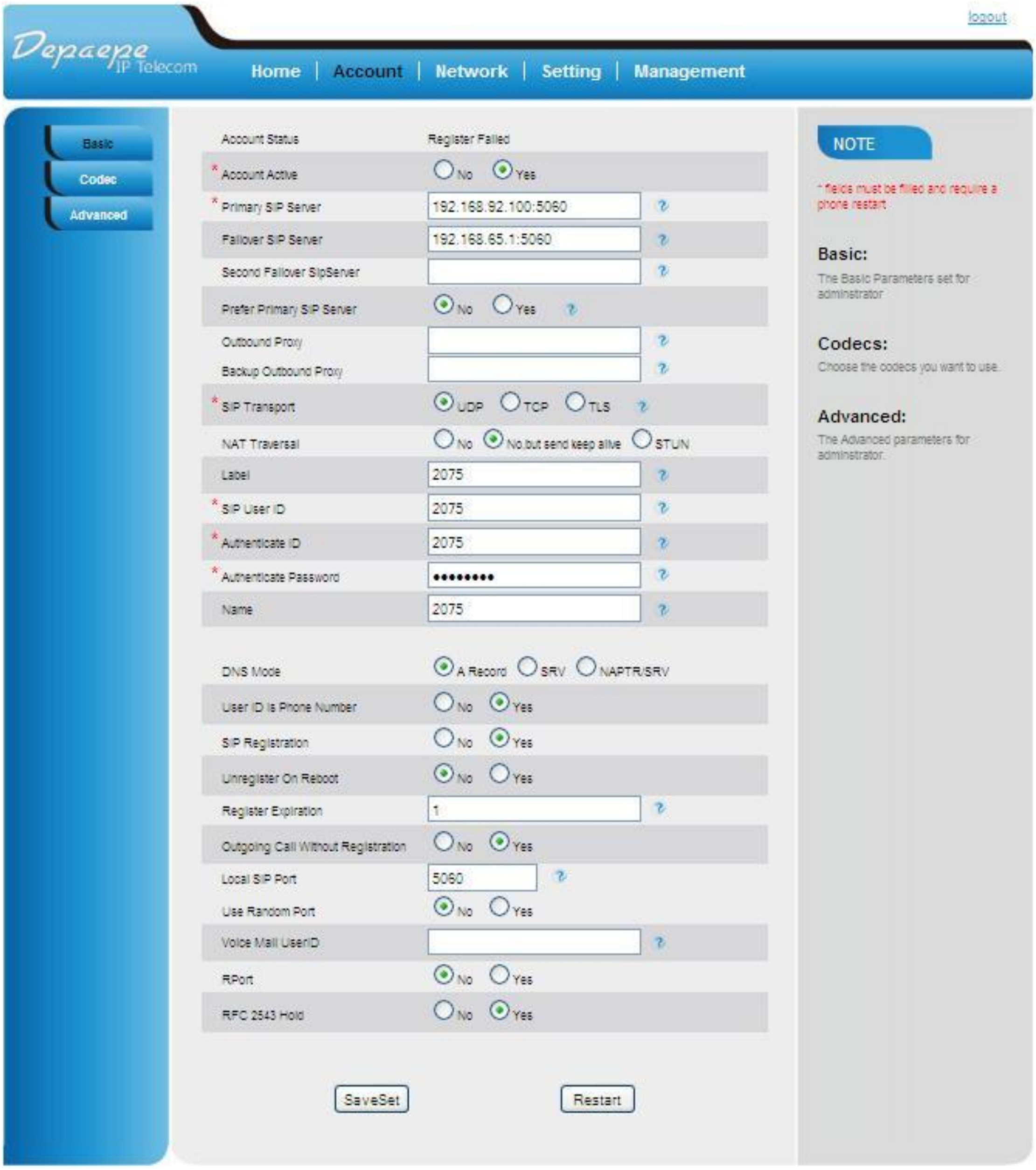
After a correct password is entered in the login screen, the embedded web server inside the HD2000IP will show the configuration page, which is explained in details below:

8.1.5.1 Device Status



Home – Status	
Options	**Meaning**
	Version
Product Model	Contains the product model info.
Firmware Version	•**Boot:** Booting code version number. • **IMG:** This is the main software (firmware) release number, always used to identify the software (firmware) system of the phone. • **ROM:** This is the main software (firmware) release number, always used to identify the software (firmware) system of the phone. • **DSP:** Powerful Digital Signal Processing (DSP) technology to ensure superior audio quality.
	Account Status
Account	Indicates whether account are registered to the related SIP server(s).
	Network
WAN Port Type	This field shows type of IP address of HD2000IP.
WAN IP Adess	This field shows IP address of HD2000IP.
Subnet Mask	This field shows Subnet Mask address of HD2000IP.
Gateway	This field shows Gateway address of HD2000IP.
Primary DNS	This field shows Primary DNS address of HD2000IP.
Secondary DNS	This field shows Secondary address of HD2000IP.
MAC Address	This field shows MAC address of HD2000IP.
Device type	This field shows device type : bridge or routeur.
Memory Free	Administrator information
System Up Time	Shows system up time since the last reboot.
Restart	Restart This button will restart the voip application.

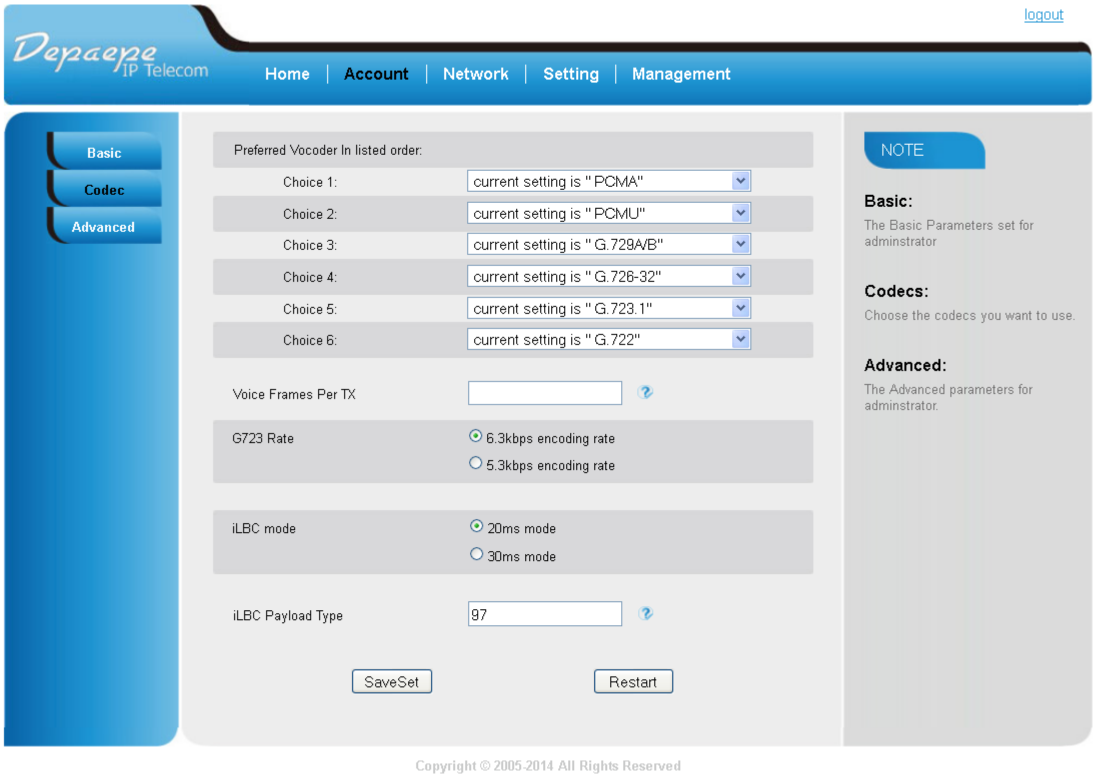
8.1.5.2 Account



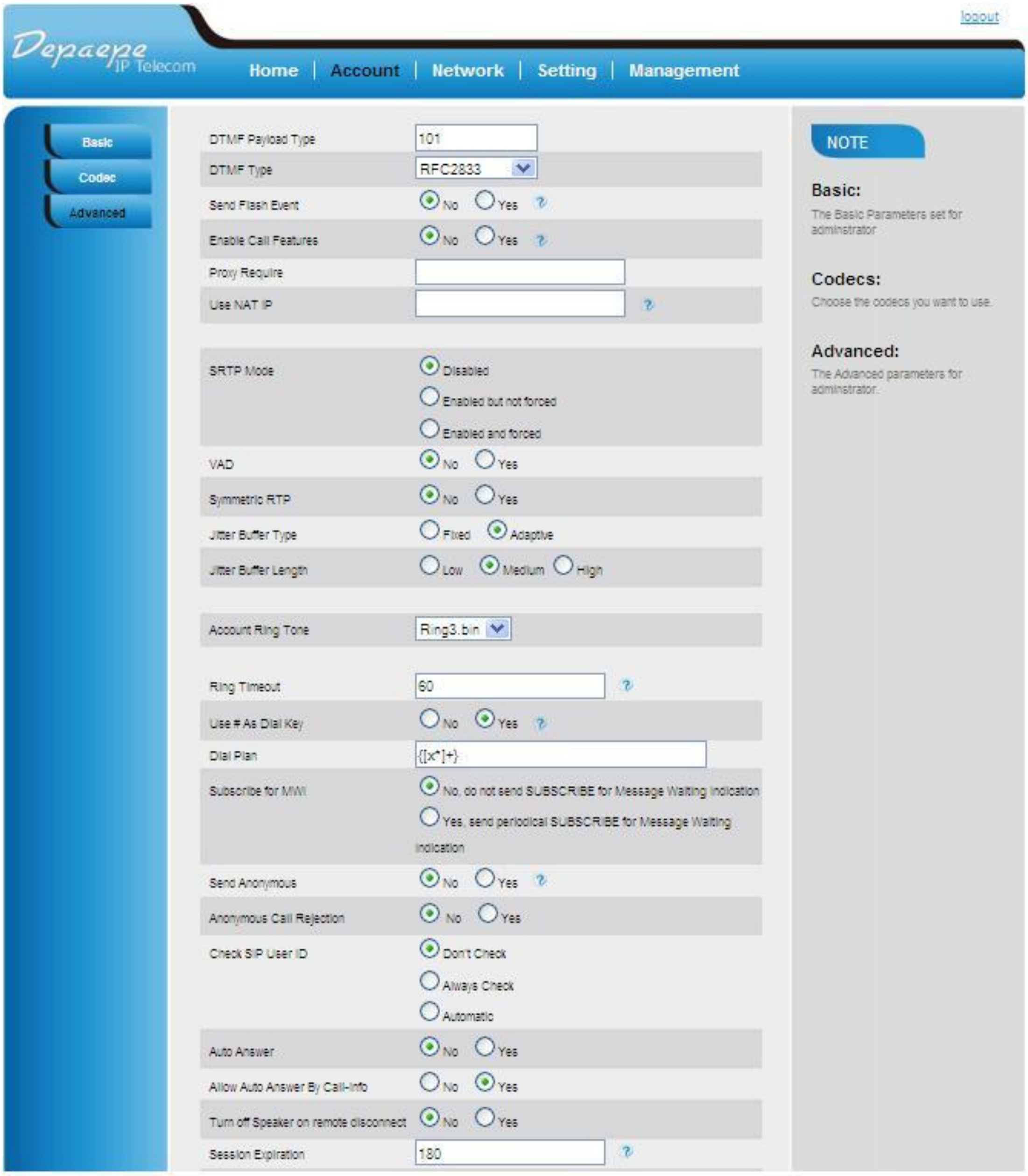
Account – Basic	
Options	**Meaning**
Account Status	Indicates whether accounts are registered to the related SIP server(s).
Account Active	When set to Yes the FXS port is activated Sip.mycompany.com or IP Adress.
Primary SIP Server	This field contains the URI string or the IP address (and port, if different from 5060) of the SIP primary 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
Failover SIP server	This field contains the URI string or the IP address (and port, if different from 5060) of the SIP failover 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 Optional, used when primary server no response.
Second Failover Sip Server	This field contains the URI string or the IP address (and port, if different from 5060) of the SIP Second failover 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 Optional, used when failover SIP Server no response.
Prefer Primary SIP Server	Optional, used when failover SIP Server no response. Yes – will register to primary server if failover registration expires.
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. e.g., proxy.myprovider.com? or IP Address, if any.
Backup Outbound Proxy	This is usually Set as IP Address
SIP Transport	Default is UDP.
NAT Transersal	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.

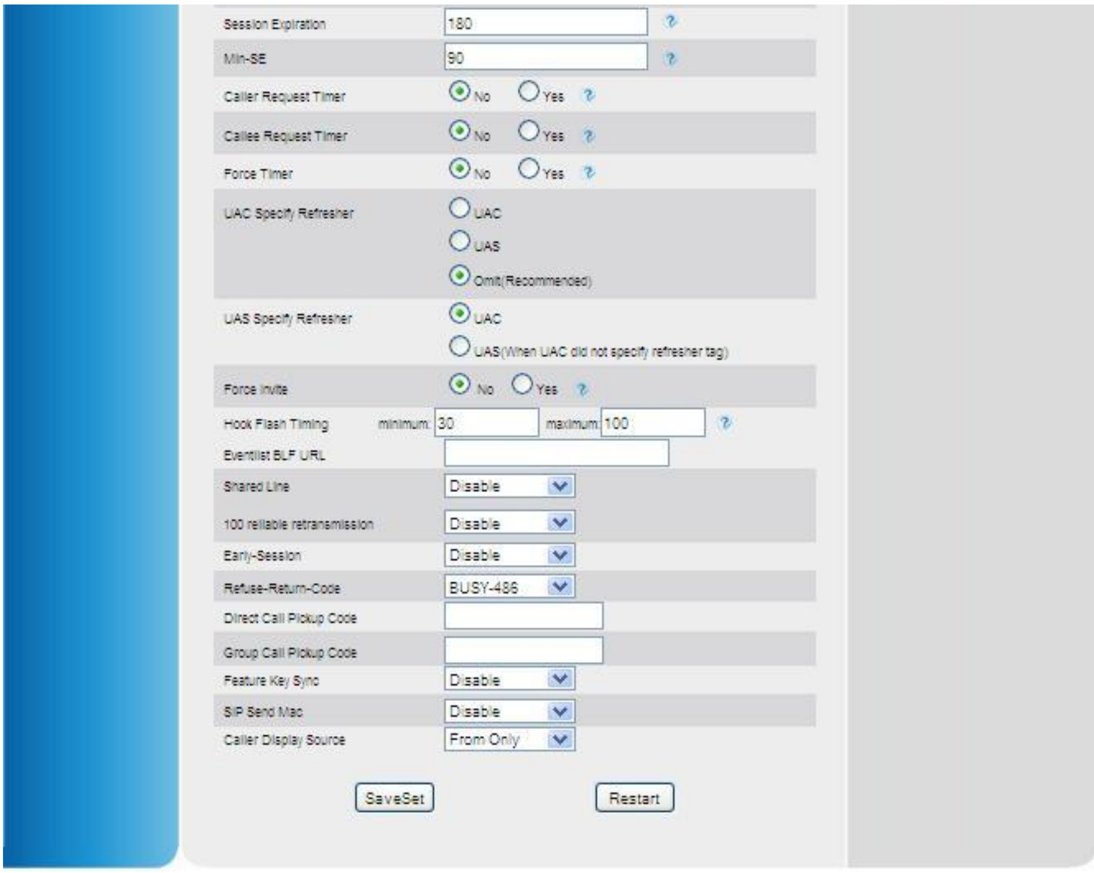
Label	The name show on the LCD of the current device
SIP User ID	SIP service subscriber's User ID
Authenticate ID	SIP service subscriber's Authenticate ID. Can be identical to or different from SIP User ID
Authenticate Password	SIP service subscriber's account password
Name	SIP service subscriber's Label which will be used for Caller ID display.
DNS Mode	Default is ARecord. If select SRV 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.
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.
Voice Mail User ID	The number for check voice mail.
RPort	RPort in RFC 3581.

RFC 2543 Hold	By default is Yes. Enable if Yes or disable if No : Hold



Account – Codec	
Options	**Meaning**
Preferred Vocader In listed order	The HD2000IP supports up to 6 different Vocoder types including G.711 A-/U-law， G.723.1, G.726, G.729A/B, G.722, iLBC, GSM_FR . 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 6”.
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.
PTime(ms)	By default 20ms. G723 voice frame contains 30ms of audio.
G723 Rate	This defines the encoding rate for G723 vocoder. By default, 6.3kbps rate is chosen.
iLBC mode	Mode 20ms or 30ms default 20ms.
iLBC Payload Type	*from 96 to 127, default is 97*


logout
Depaepe IP Telecom
Home
Account
Network
Setting
Management
Basic
Codec
Advanced
DTMF Payload Type
101
DTMF Type
RFC2833
Send Flash Event
No
Yes
Enable Call Features
No
Yes
Proxy Require
Use NAT IP
SRTP Mode
Disabled
Enabled but not forced
Enabled and forced
VAD
No
Yes
Symmetric RTP
No
Yes
Jitter Buffer Type
Fixed
Adaptive
Jitter Buffer Length
Low
Medium
High
Account Ring Tone
Ring3.bin
Ring Timeout
60
Use # As Dial Key
No
Yes
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
Anonymous Call Rejection
No
Yes
Check SIP User ID
Don't Check
Always Check
Automatic
Auto Answer
No
Yes
Allow Auto Answer By Call-Info
No
Yes
Turn off Speaker on remote disconnect
No
Yes
Session Expiration
180
NOTE
Basic:
The Basic Parameters set for administrator
Codecs:
Choose the codecs you want to use.
Advanced:
The Advanced parameters for administrator.


Session Expiration
180
Min-SE
90
Caller Request Timer
No
Yes
Callee Request Timer
No
Yes
Force Timer
No
Yes
UAC Specify Refresher
UAC
UAS
Omit(Recommended)
UAS Specify Refresher
UAC
UAS(When UAC did not specify refresher tag)
Force Invite
No
Yes
Hook Flash Timing
minimum: 30
maximum: 100
Eventlist BLF URL
Shared Line
Disable
100 reliable retransmission
Disable
Early-Session
Disable
Refuse-Return-Code
BUSY-486
Direct Call Pickup Code
Group Call Pickup Code
Feature Key Sync
Disable
SIP Send Mac
Disable
Caller Display Source
From Only
SaveSet
Restart

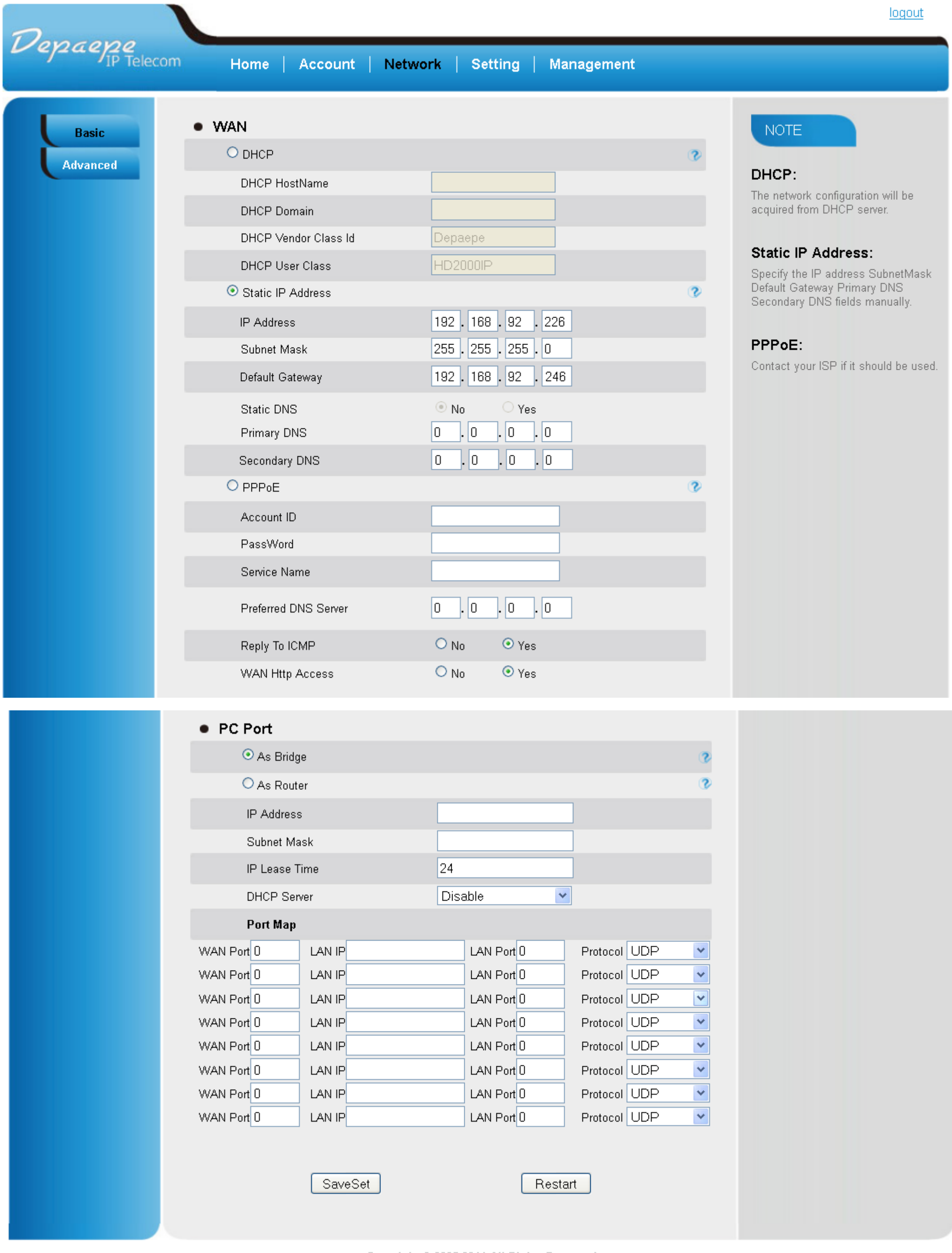
Account -Advanced	
Options	**Meaning**
DTMF Payload Type	This parameter sets the payload type for DTMF using RFC2833
DTMF Type	Send DTMF as in-audio, According to RFC 2833 or SIP INFO message.
Send Flash event	This parameter allows users to control whether to send an SIP NOTIFY message indicating the Flash event, or just to switch to the voice channel when users press the Flash key.
Enable Call Feature	Default is No. If set to Yes, Call Forwarding & Do-Not-Disturb are supported locally (see P.17) *if Yes, call features using star codes will be supported locally*
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.
SRTP Mode	Secure Real-Time Transport Protocol (SRTP) encrypts the RTP during VoIP phone.
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 Lengh	Select Low, Medium or High based on network conditions.
Account Ring Tone	There are **8** uniquely defined ring tones.
Ring Timeout	Incoming call will stop ringing when not picked up given a specific period of time.
Use # As Dial Key	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.
Dial Plan	**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+}
Subscribe for MWI	Default is No. When set to "Yes" a SUBSCRIBE for Message Waiting Indication will be sent periodically.
Send Anonymous	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.
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	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.
Turn off Speaker on remote disconnect	When the remote user hang up, the phone will disable the speaker.
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.

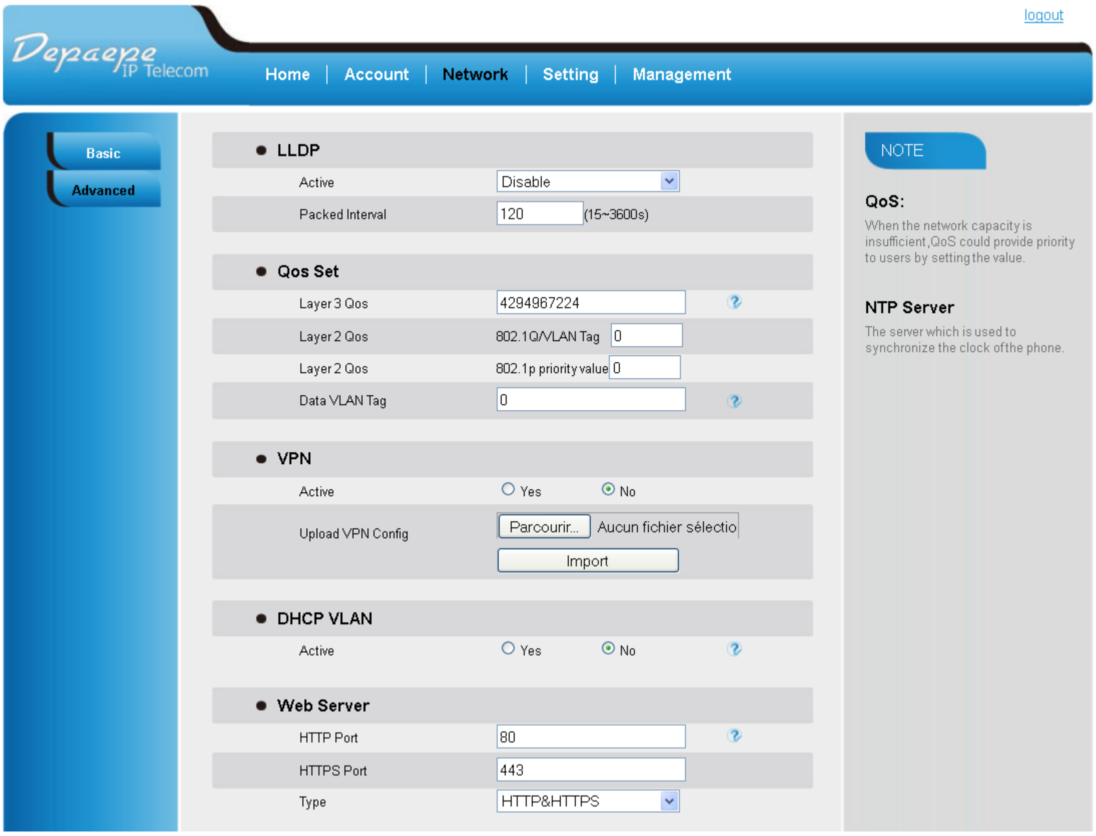
Min-SE	Defines the minimum session expiration (in seconds). Default is **90 seconds**.
Caller Request Timer	If set to “Yes”, the phone will use session timer when it makes outbound calls if remote party supports session timer.
Callee Request Timer	If selecting “Yes”, the phone will use session timer 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
Eventlist BLF URL	Used for BLF list, phone will get BLF INFO from the URL
Shared Line	Used for Broadsoft server, two or more device use a same account, when one of these devices activate a line, other device will display this line is occupied.
100 reliable retransmission	This feature often work together with “Early-Session”. And need server support, when phone send this INFO to server, it will receive the ring tone from server.
Early-Session	
Refuse-Return-Code	This setting decide the response when you reject a call automatically or manually, phone will reject as Busy here (Phone is busy can’t answer), or Not found, or Temporarily unavailable.
Direct Call Pickup Code	
Group Call Pickup Code	
Feature Key Sync	This option used for phone and server synchronization status. For example, when you press DND, server will get the INFO make the

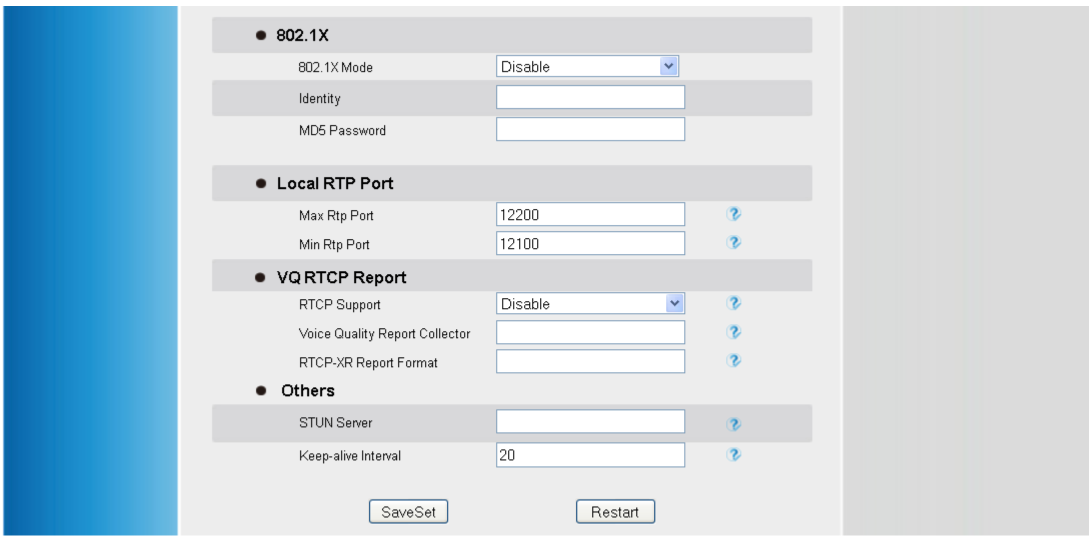
	phone status on the server to DND.
SIP Send Mac	
Caller Display Source	Setting caller name display source, contain “FROM”, “PAI-FROM”, “PAI-RPID-FROM”, “RPID-PAI-FROM”, “RPID-FROM”

8.1.5.3 Network



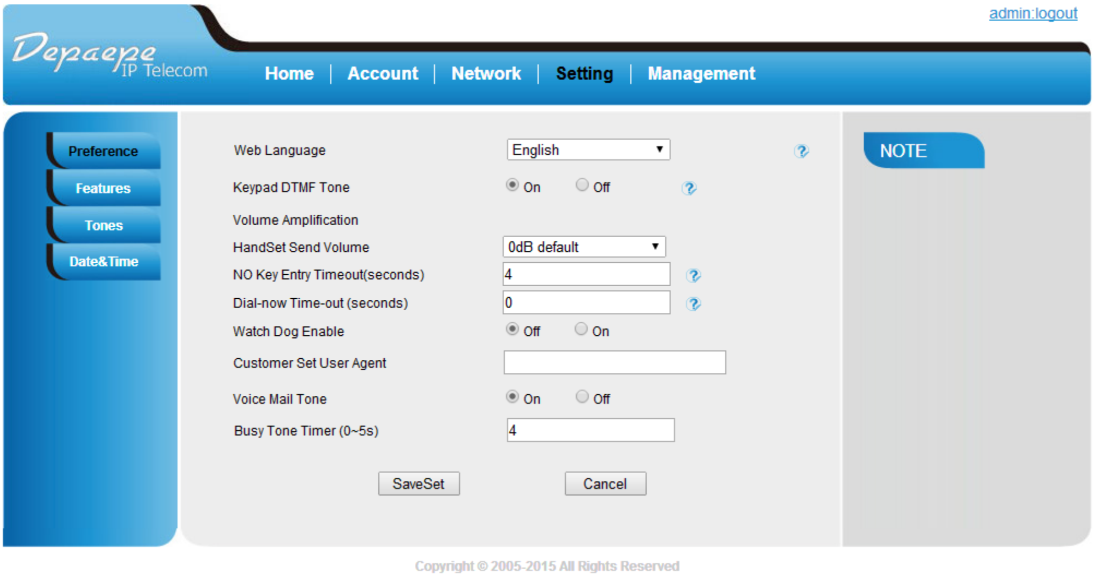
Network – Basic	
Options	**Meaning**
WAN	
WAN	There HD2000IP operates in two modes: 1. **DHCP mode**: all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory.) The HD2000IP acquires its IP address from the first DHCP server it discovers on its LAN. 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 Gateway address, Primary DNS , Secondary DNS. These fields are set to zero by default. 3. PPPoE mode :
Reply To ICMP	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 Http Access	If this parameter is set to “No”, the HTML configuration update via WAN port is disabled.
PC port	
PC port	As Bridge As Router IP address Subnet Mask IP Lease Time DHCP Server : Port Map :


logout
Home | Account | Network | Setting | Management
Basic
Advanced
LLDP
Active
Disable
Packed Interval
120
(15~3600s)
Qos Set
Layer3 Qos
4294967224
Layer2 Qos
802.1Q/VLAN Tag
0
Layer2 Qos
802.1p priority value
0
Data VLAN Tag
0
VPN
Active
Yes
No
Upload VPN Config
Parcourir...
Aucun fichier sélectio
Import
DHCP VLAN
Active
Yes
No
Web Server
HTTP Port
80
HTTPS Port
443
Type
HTTP&HTTPS
NOTE
QoS:
When the network capacity is insufficient,QoS could provide priority to users by setting the value.
NTP Server
The server which is used to synchronize the clock ofthe phone.

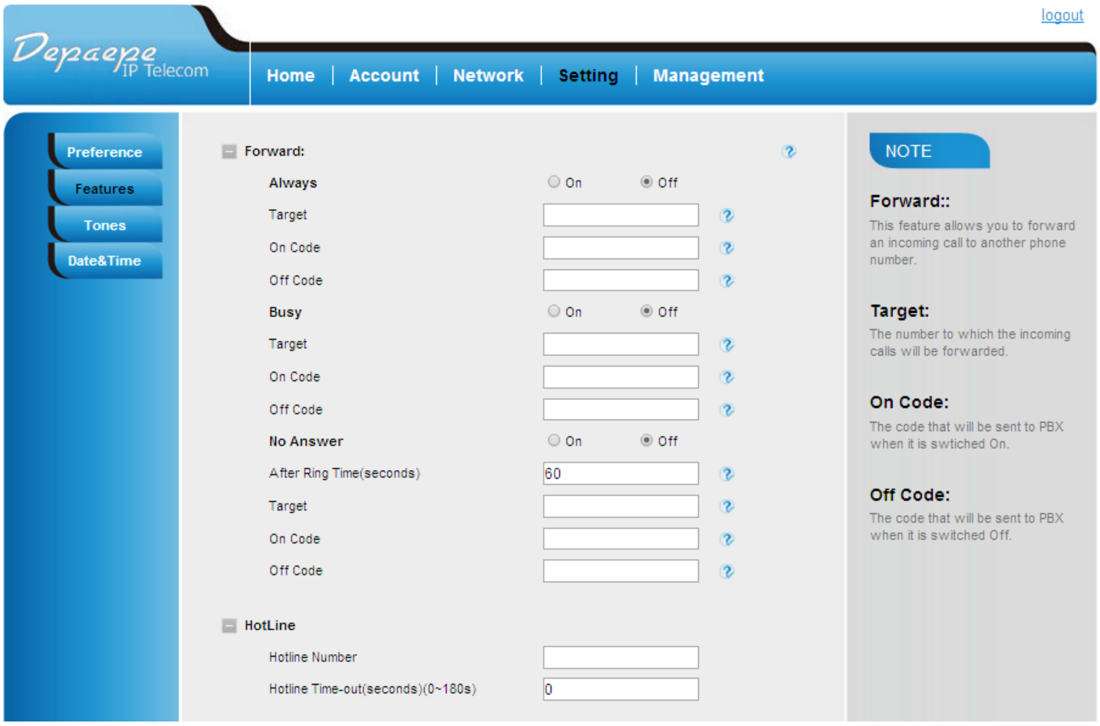

802.1X
802.1X Mode
Disable
Identity
MD5 Password
Local RTP Port
Max Rtp Port
12200
Min Rtp Port
12100
VQ RTCP Report
RTCP Support
Disable
Voice Quality Report Collector
RTCP-XR Report Format
Others
STUN Server
Keep-alive Interval
20
SaveSet
Restart

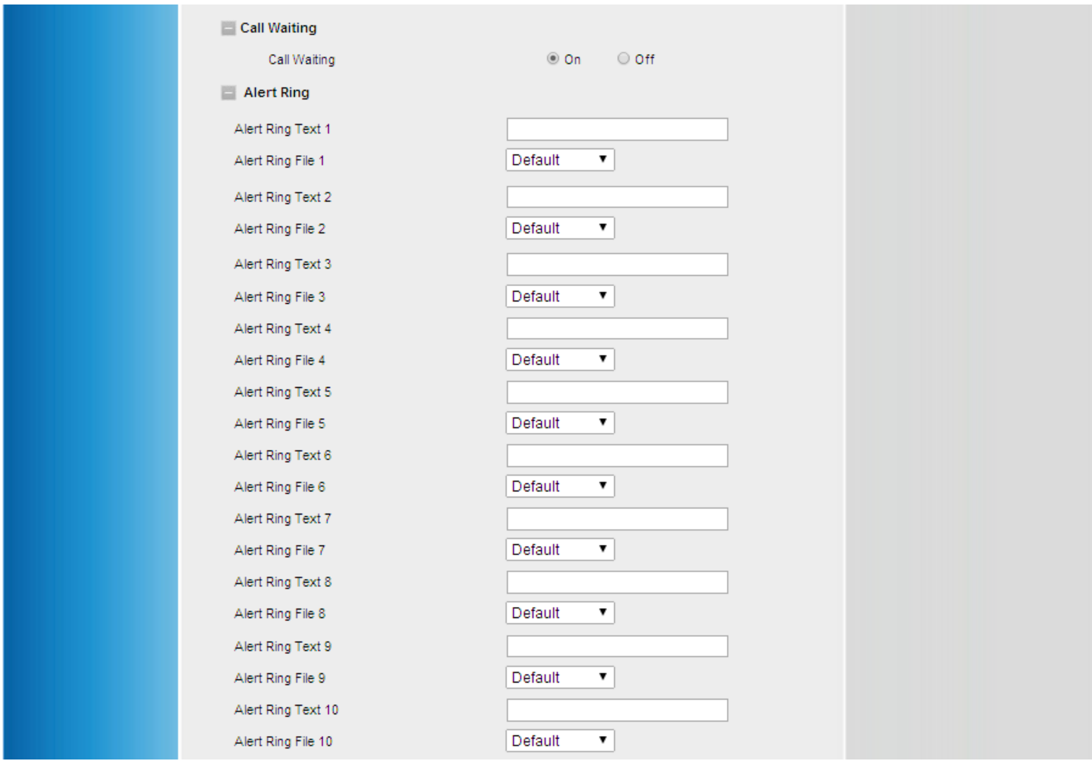
Network -Advanced	
Options	**Meaning**
LLDP	Active : select the desired value from the pull-down list of Active. Packed interval : enter the desired time(in seconds) in the Packet interval (1~3600s) field.
Qos Set	Layer 3 Qos : This field defines the layer 3 QoS parameter which can be the value used for IP Precedence or Diff-Serv or MPLS. Default value is **48**. Layer 2 Qos : Value used for layer 2 VLAN tag. Default setting is **blank.** Data VLAN Tag : Valid only when bridge mode.
VPN	Active : Upload VPN Config :
DHCP VLAN	Active : yes or no (DHCP option 132)
Web Server	HTTP Port : default for http is 80. HTTPS Port : Type :
802.1X	802.1X Mode : Identily : MD5 Password :
Local RTP Port	Max Rtp Port : The Largest RTP Port : Should be large than (Min-RTP-Port +100) and less than 65535. Min Rtp Port : The Lowest RTP Port : Port>= 1024
VQ RTCP Report	RTCP Support : If Enable, Phone will statistic analysis RTP Info, Then Report to Collector if Need. Voice Quality Report Collector : The RTCP Collector Info, e.g : sip:account@sample.com:8765. RTCP-XR-report Format : The Format of Reported PUBLISH Info
Others	STUN Server : URI or IP:port Keep-alive interval : default 20 seconds.

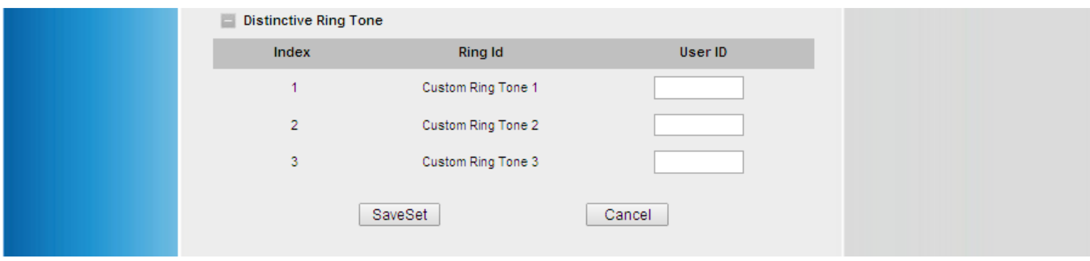
8.1.5.4 Setting



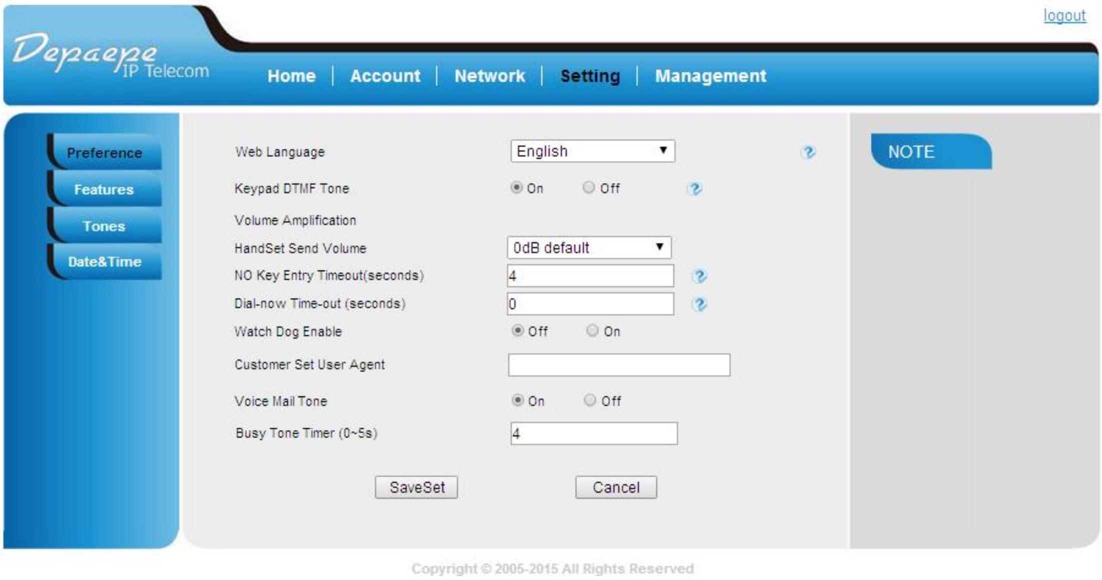
Setting -Preference	
Options	**Meaning**
Web Language	Select the displayed language for web page
Keypad DTMF Tone	Open or close keypad DTMF Tone
Volume Amplification	
HandSet Send Volume	
No Key Entry Timeout(seconds)	In seconds, 0 means never timeout, default is 0 second.
Dial now Timeout(seconds)	In seconds, Interval for DialNow, default is 0 second.
Watch Dog Enable	Prevent phone freeze, so that it can be automatically reset when it run error.
Customer Set User Agent	When phone send SIP data it will make User Agent in the packet, default use its model info.
Voice Mail Tone	
Busy Tone Timer	


logout
Home | Account | Network | Setting | Management
Preference
Features
Tones
Date&Time
Forward:
Always On Off
Target
On Code
Off Code
Busy On Off
Target
On Code
Off Code
No Answer On Off
After Ring Time(seconds) 60
Target
On Code
Off Code
HotLine
Hotline Number
Hotline Time-out(seconds)(0~180s) 0
NOTE
Forward::
This feature allows you to forward an incoming call to another phone number.
Target:
The number to which the incoming calls will be forwarded.
On Code:
The code that will be sent to PBX when it is swtiched On.
Off Code:
The code that will be sent to PBX when it is switched Off.

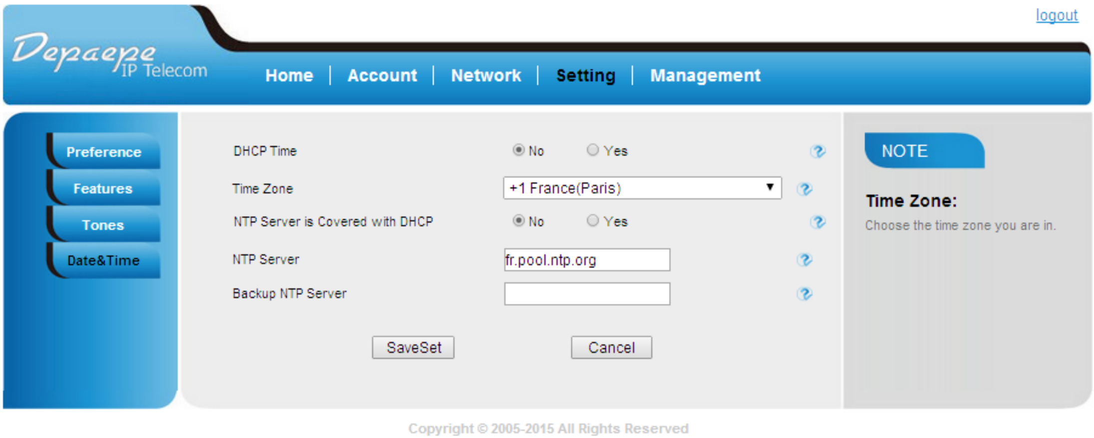

Call Waiting
Call Waiting On Off
Alert Ring
Alert Ring Text 1
Alert Ring File 1 Default
Alert Ring Text 2
Alert Ring File 2 Default
Alert Ring Text 3
Alert Ring File 3 Default
Alert Ring Text 4
Alert Ring File 4 Default
Alert Ring Text 5
Alert Ring File 5 Default
Alert Ring Text 6
Alert Ring File 6 Default
Alert Ring Text 7
Alert Ring File 7 Default
Alert Ring Text 8
Alert Ring File 8 Default
Alert Ring Text 9
Alert Ring File 9 Default
Alert Ring Text 10
Alert Ring File 10 Default


Distinctive Ring Tone
Index Ring Id User ID
1 Custom Ring Tone 1
2 Custom Ring Tone 2
3 Custom Ring Tone 3
SaveSet
Cancel

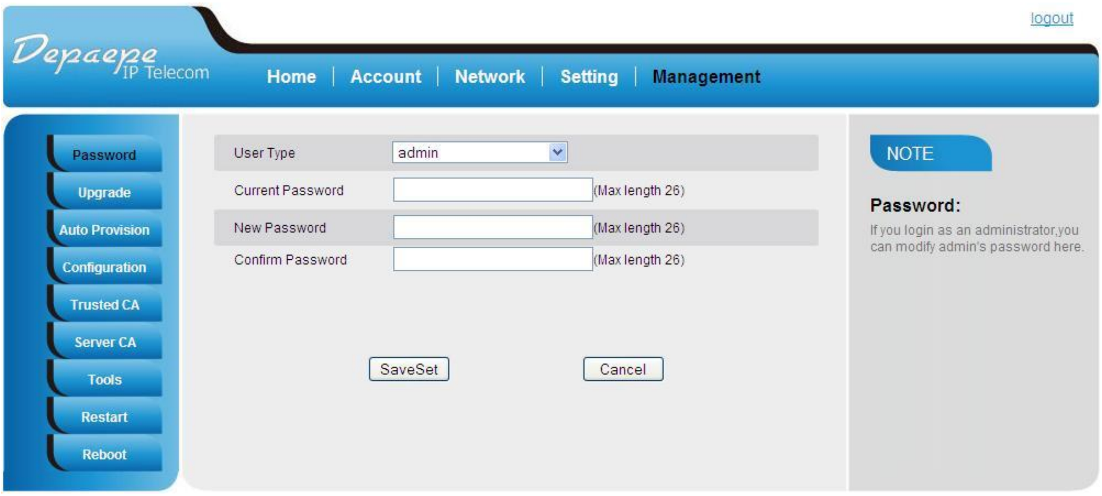
Setting - Features	
Options	**Meaning**
Forward	
Always	Target : Target number for transfer. On Code : The feature code to enable all incoming calls forward, the phone will send the feature code directly to open all incoming calls forward. Off Code : The feature code to disable all incoming calls forward, the phone will send the feature code directly to close all incoming calls forward.
Busy	Target : transferred to target number. On Code : The feature code to enable busy call forward, the phone will send the feature code directly to open busy call forward. Off Code : The feature code to disable busy call forward, the phone will send the feature code directly to close busy call forward.
No Answer	After Ring Time(seconds) : waiting time for NoAnswer forward. Target : number for transfer. On Code : The feature code to enable no answer call forward, the phone will send the feature code directly to open no answer call forward. Off Code : The feature code to disable no answer call forward, the phone will send the feature code directly to close no answer call forward.
HotLine	
HotLine Number	
Call Waiting	
Call Waiting	Enable if On or Disable if Off call waiting.
Alert Ring	
Alert Ring Text 1~10 Alert Ring File 1~10	These two option work together, you can set phone ring the appointed tone when received the Alert Info.
Speed Dial	
M1 to M4	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.
Distinctive Ring Tone	
User ID	



Setting - Tones	
Options	**Meaning**
Select Country	
Dial Tone, Ringback Tone, Busy Tone, Reorder Tone, Confirmation Tone, Call Waiting Tone	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.

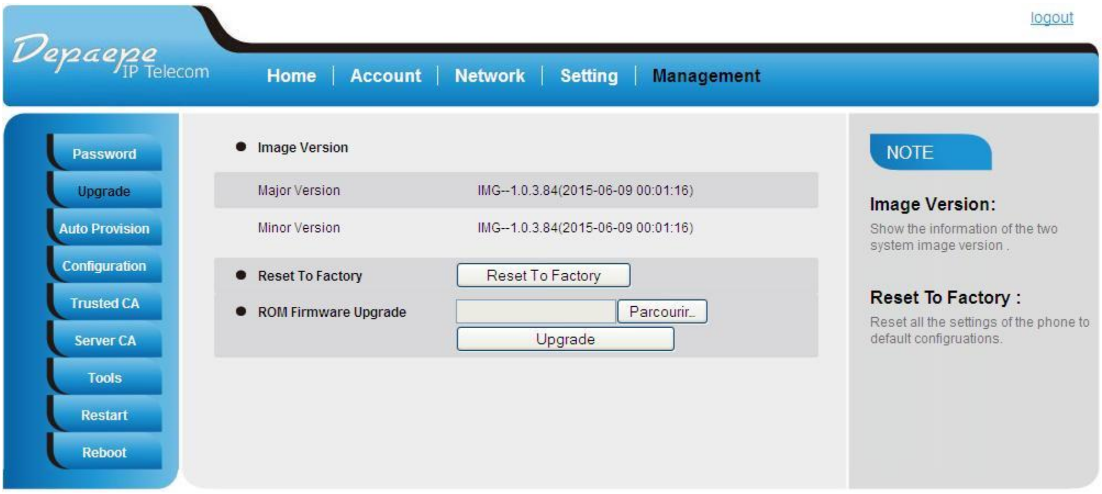


Setting - Date&Time	
Options	**Meaning**
DHCP Time	
Time Zone	Current local time in cities worldwide
NTP Server is Covered with DHCP	Allow DHCP Option 42 to override NTP server
NTP Server	Address of NTP server
Backup NTP Server	

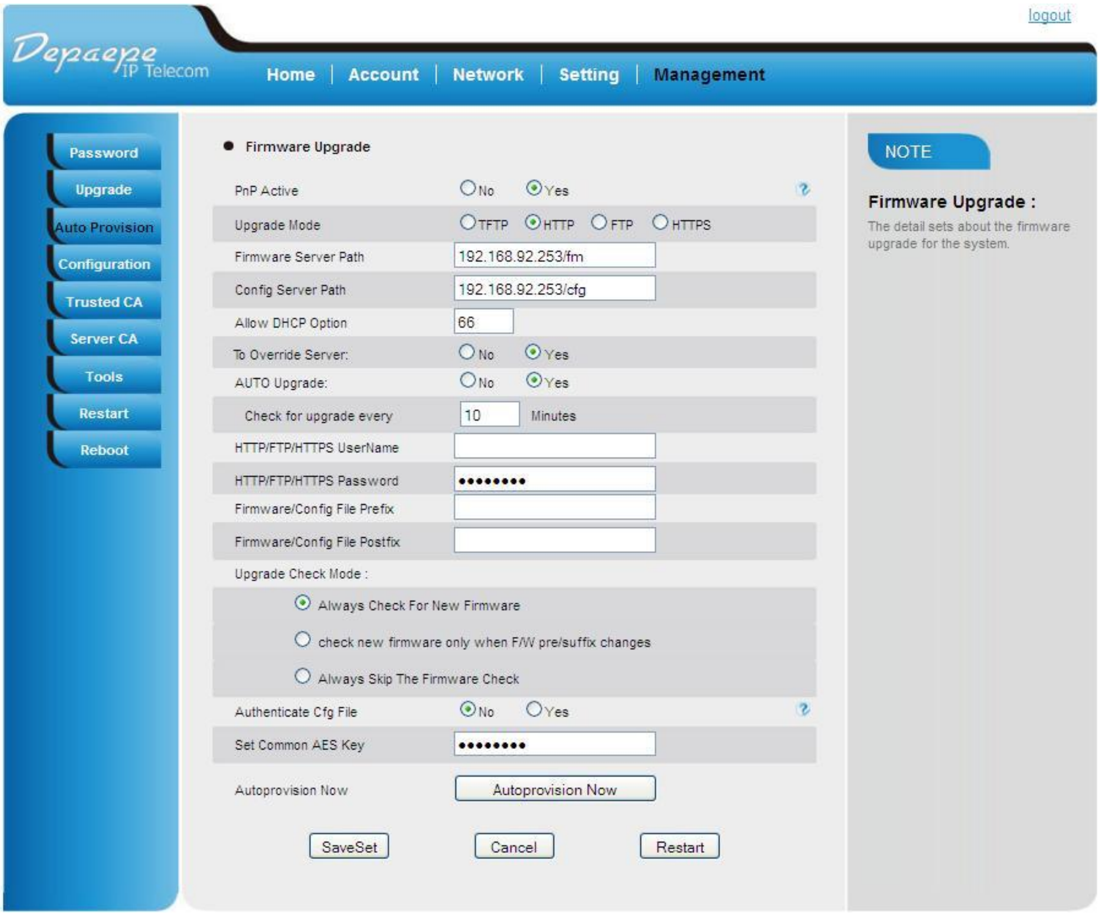


Management -Password	
Options	**Meaning**
User Type	Choose between : **Admin**: The administrator access level. With this access level, all configurations on interface. The authentication identity for this access level is admin. And the default password is admin. **Var:** The value-added reseller access level. Generally, with this access level, most on the web user interface and phone user interface can be read and written. The authentication identity

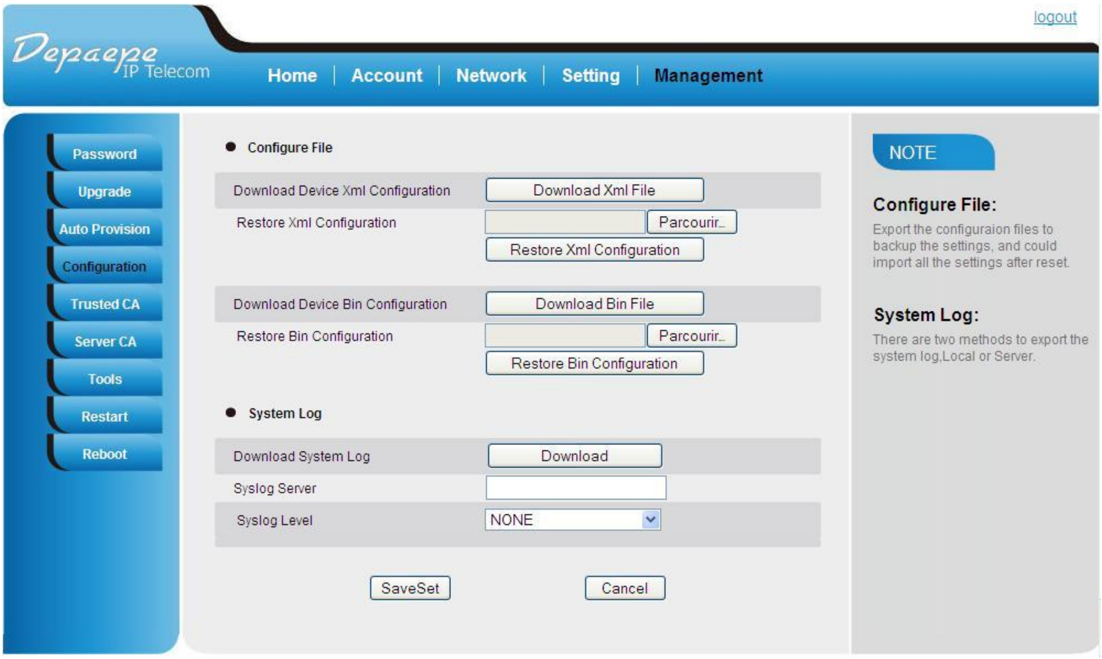
	for this access level is var. And the default password is 1234. **User**: The end user access level. Generally, only a few configurations are allowed to be written and read for access user. The authentication identity for this access level is user. And the default password is 1234.
Current Password	
New password	
Confirm Password	



Management - Upgrade	
Options	**Meaning**
Major Version	Major Version of the firmware.
Minor Version	Minor Version of the firmware.
Reset To Factory	Reset factory setting.
ROM Firmware Upgrade	

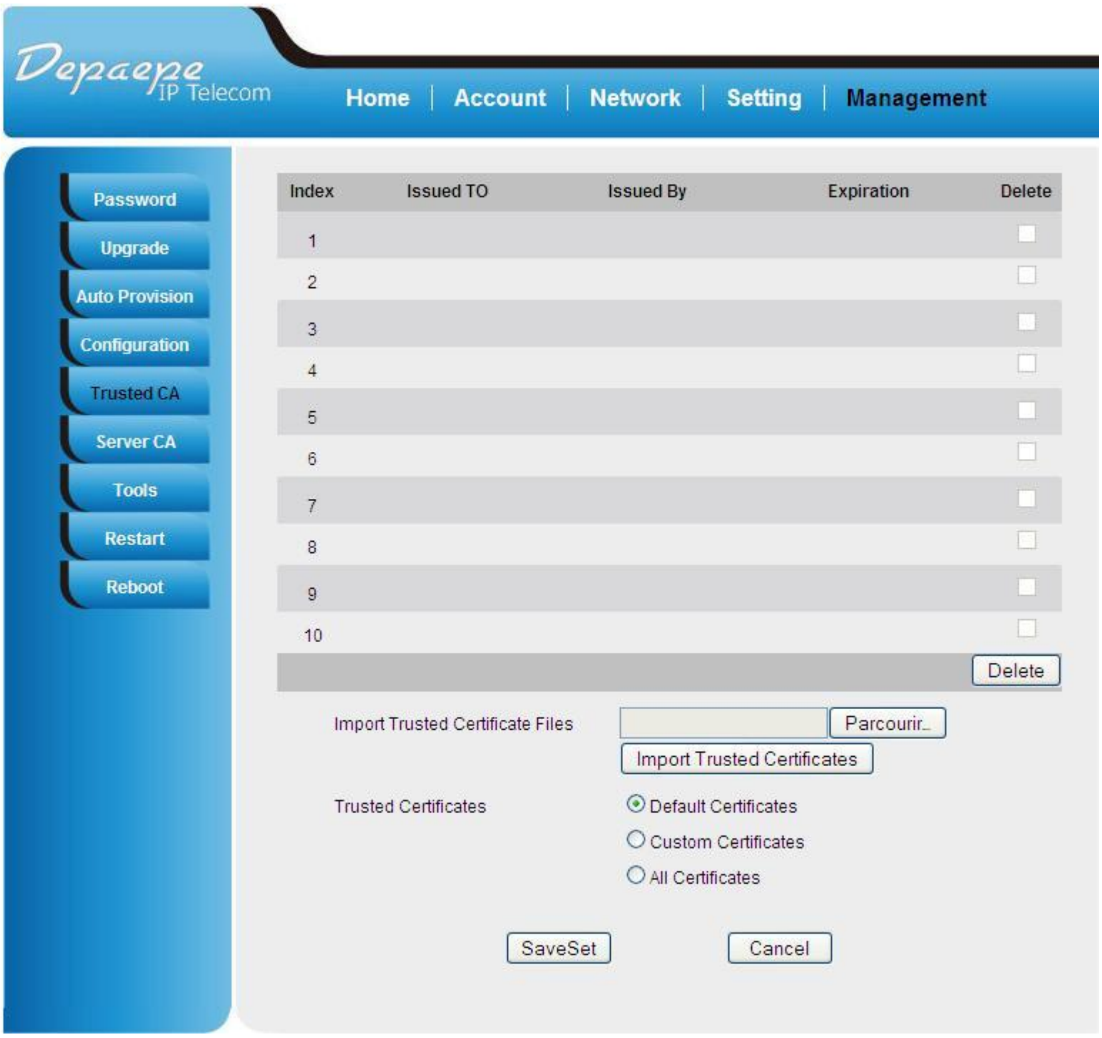

logout
Home | Account | Network | Setting | Management
Password
Upgrade
Auto Provision
Configuration
Trusted CA
Server CA
Tools
Restart
Reboot
Firmware Upgrade
PnP Active No Yes
Upgrade Mode TFTP HTTP FTP HTTPS
Firmware Server Path 192.168.92.253/fm
Config Server Path 192.168.92.253/cfg
Allow DHCP Option 66
To Override Server: No Yes
AUTO Upgrade: No Yes
Check for upgrade every 10 Minutes
HTTP/FTP/HTTPS UserName
HTTP/FTP/HTTPS Password
Firmware/Config File Prefix
Firmware/Config File Postfix
Upgrade Check Mode :
Always Check For New Firmware
check new firmware only when F/W pre/suffix changes
Always Skip The Firmware Check
Authenticate Cfg File No Yes
Set Common AES Key
Autoprovision Now
Autoprovision Now
SaveSet
Cancel
Restart
NOTE
Firmware Upgrade :
The detail sets about the firmware upgrade for the system.

Management - Auto Provision	
Options	**Meaning**
PnP Active	The request to the server to obtain a support URL for upgrade.
Upgrade Mode	Type of the provider : TFTP/HTTP/FTP/HTTPS
Firmware Server Path	needs to be set to be a valid URL of a provider server, server name can be in either FQDN or IP address format. Below are examples of some valid URLs. e.g. firmware.mycompany.com:5688/Depaepe e.g. www.mycompany.com:5688/fm/Depaepe e.g. 218.2.83.110 By default address_ip_server_provider/fm
Config Server Path	needs to be set to be a valid URL of a provider server, server name can be in either FQDN or IP address format. Below are examples of some valid URLs. e.g. firmware.mycompany.com:5688/Depaepe e.g. www.mycompany.com:5688/cfg/Depaepe e.g. 218.2.83.110 By default : address_ip_server_provider/cfg
Allow DHCP Option	Allow DHCP Option (128 or 150 or 66).
To Override Server	
Auto Upgrade	
Check for upgrade every	Check for upgrade every ? Minutes.
HTTP/FTP/HTTPS UserName	HTTP/FTP/HTTPS UserName
HTTP/FTP/HTTPS Password	HTTP/FTP/HTTPS Password.
Firmware/Config File Prefix	Firmware Prefix allows device to download the firmware name with the matching Prefix.
Firmware/Config File Postfix	Firmware Postfix allows device to download the firmware name with the matching Postfix.
Upgrade Check Mode	
Authenticate Cfg File	Enable if Yes or disable if No Cfg File.
Set Common AES Key	The XML configuration file could be encrypted in AES-128-CBC algorithm. The encryption password is defined in P8631 (Management->Auto Provision->Set Common AES Key) of the configuration file. The Password length is from 1-16, and password must be[0-9,A-F].
Autoprovision Now	

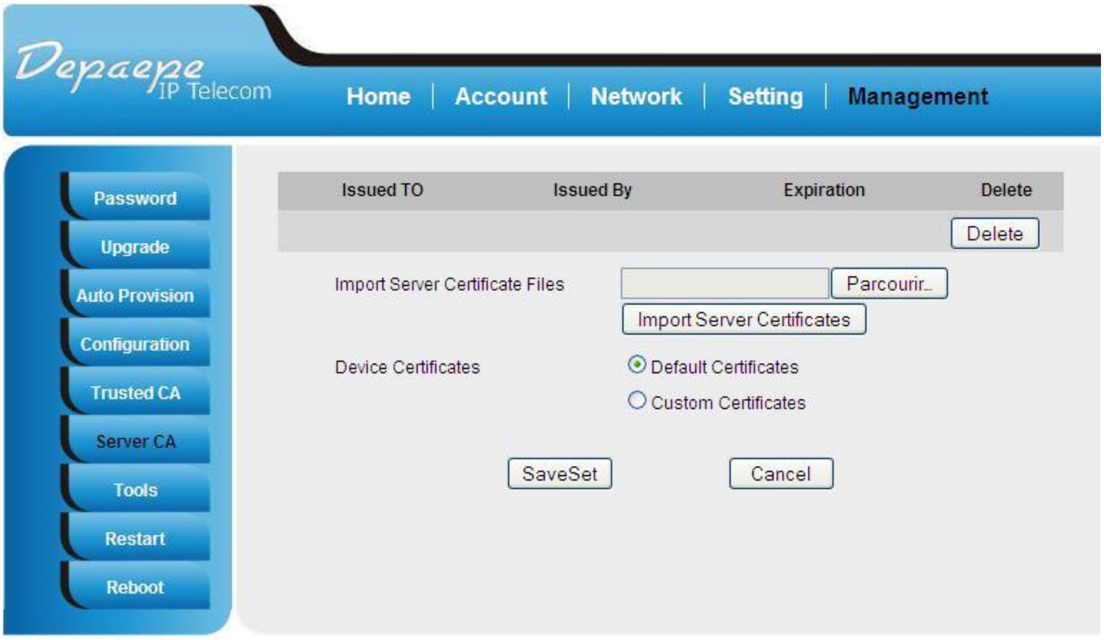


Management- Configuration	
Options	**Meaning**
	Configure File
Dowload Device Xml Configuration	Export the XML configuration files to backup the settings.
Restore Xml Configuration	Import the XML configuration files to restore the settings.
Dowload Device Xml Configuration	Export the BIN configuration files to backup the settings.
Restore Xml Configuration	Import the BIN configuration files to restore the settings.
	System Log
Dowload System Log	
Syslog Server	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)

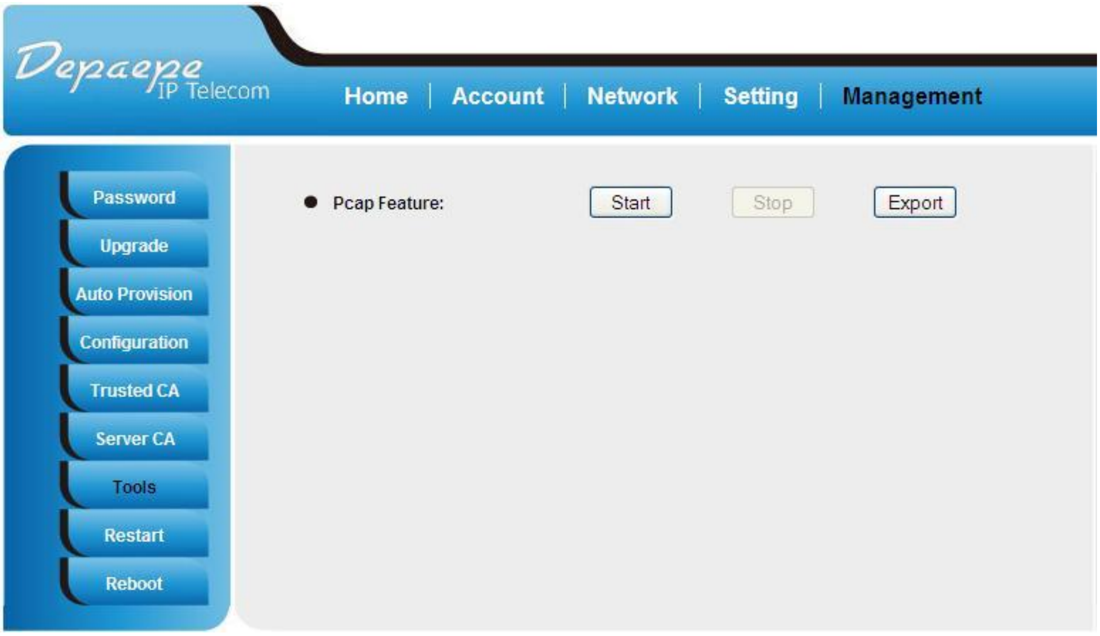
	9. SLIC chip exception (WARNING and ERROR levels) 10. memory exception (ERROR level) The Syslog uses USER facility. In addition to standard Syslog payload, it contains the following components: GS_LOG: [device MAC address][error code] error message *Example*: May 19 02:40:38 192.168.1.14 GS_LOG: [00:0b:82:00:a1:be][000] Ethernet link is up



Management - Trusted CA	
Options	**Meaning**
Import Trusted Certificates Files	
Trusted Certificates	



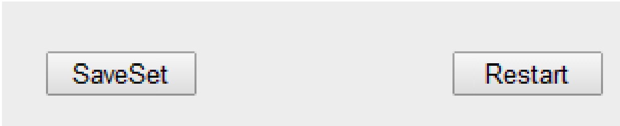
Management - Server CA	
Options	**Meaning**
Import Trusted Certificates Files	
Trusted Certificates	



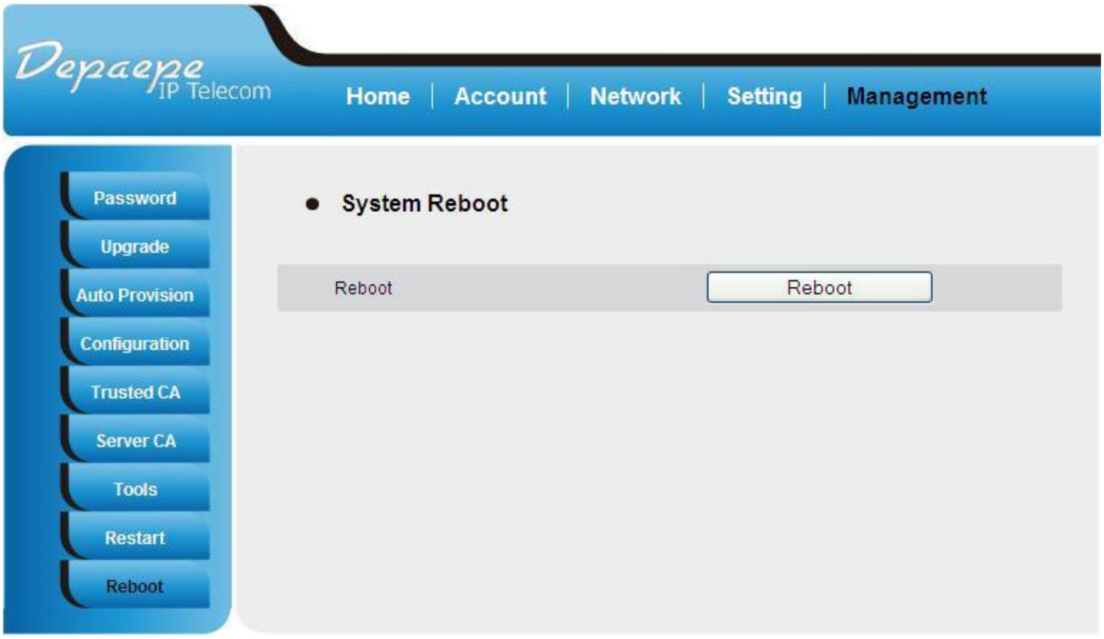
Management— Tools	
Options	**Meaning**
Pcap Feature	

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:



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”.

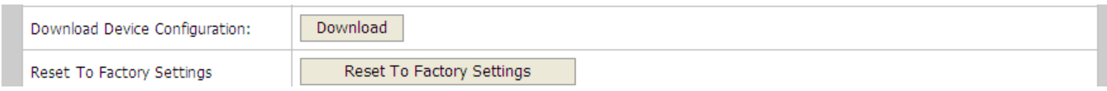
c) Enter the number “862584658050”. A "click" sound will be heard.

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.



Restoring to factory default settings by Reset key:
this key is located behind the base of the unit, , at the level of the lower wall mount fixing hole, a hole in the plastic case allows to reach it.
Press about 20 seconds the reset key, then disconnect and reconnect the Ethernet cable (PoE) or the power supply, 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

Depaepe se réserve le droit de modifier cette notice à tout moment et sans préavis.

SAS HENRI DEPAEPE
75-77 rue du Pré Brochet
95110 SANNOIS
Ph: +33 (0) 1 30 25 81 60
Fax: +33 (0) 1 39 98 61 24
www.depaepe.com

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