Grandstream IP Phones with Tpad

The following instructions detail how to configure the Grandstream Budgetone VoIP IP phones with Tpad. The configuration guide is applicable to other IP Phones in the Grandstream range.

Configuration through the Web Browser

The Grandstream series IP phone has an embedded Web server that can be used to configure the phone. It has embedded HTML pages that allow a user to configure the IP phone through a Web browser such as Microsoft's IE or Mozilla's Firefox.

Access the Web Configuration Menu

The Grandstream IP Phone Web Configuration Pages can be accessed by inputting the phones IP into the browser's URL address field like:

http://192.168.002.001

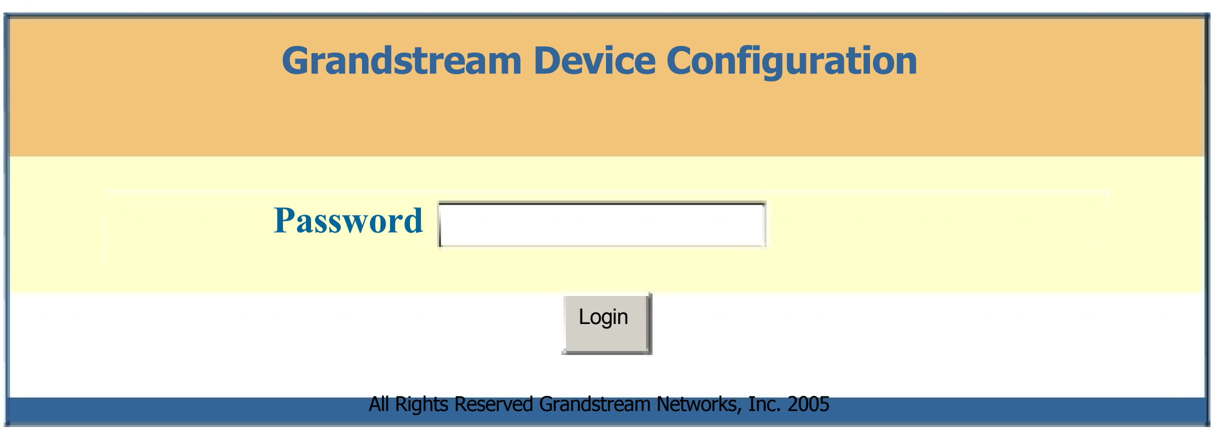
There are two ways to retrieve this IP address from the phone:

1) When the phone is *off-hook or in speakerphone* mode, simply press *MENU* button. *(This is most common way to get the IP address of the phone)*
2) When the phone is on-hook, press *MENU* button and then the browsing arrow keys to "**`[2] IP Addr`**", press *MENU* again.

NOTE:

- *To type IP address into browser to bring up the configuration pages, please strip out the leading "0. e.g.: if the IP address is: 192.168.002.001, please type in: 192.168.2.1.*

Once the correct IP address of the phone is input into browser and "Enter" key pressed, the web log in page will come up like following:



The password is case sensitive. The factory default password for Administrator is “admin”.

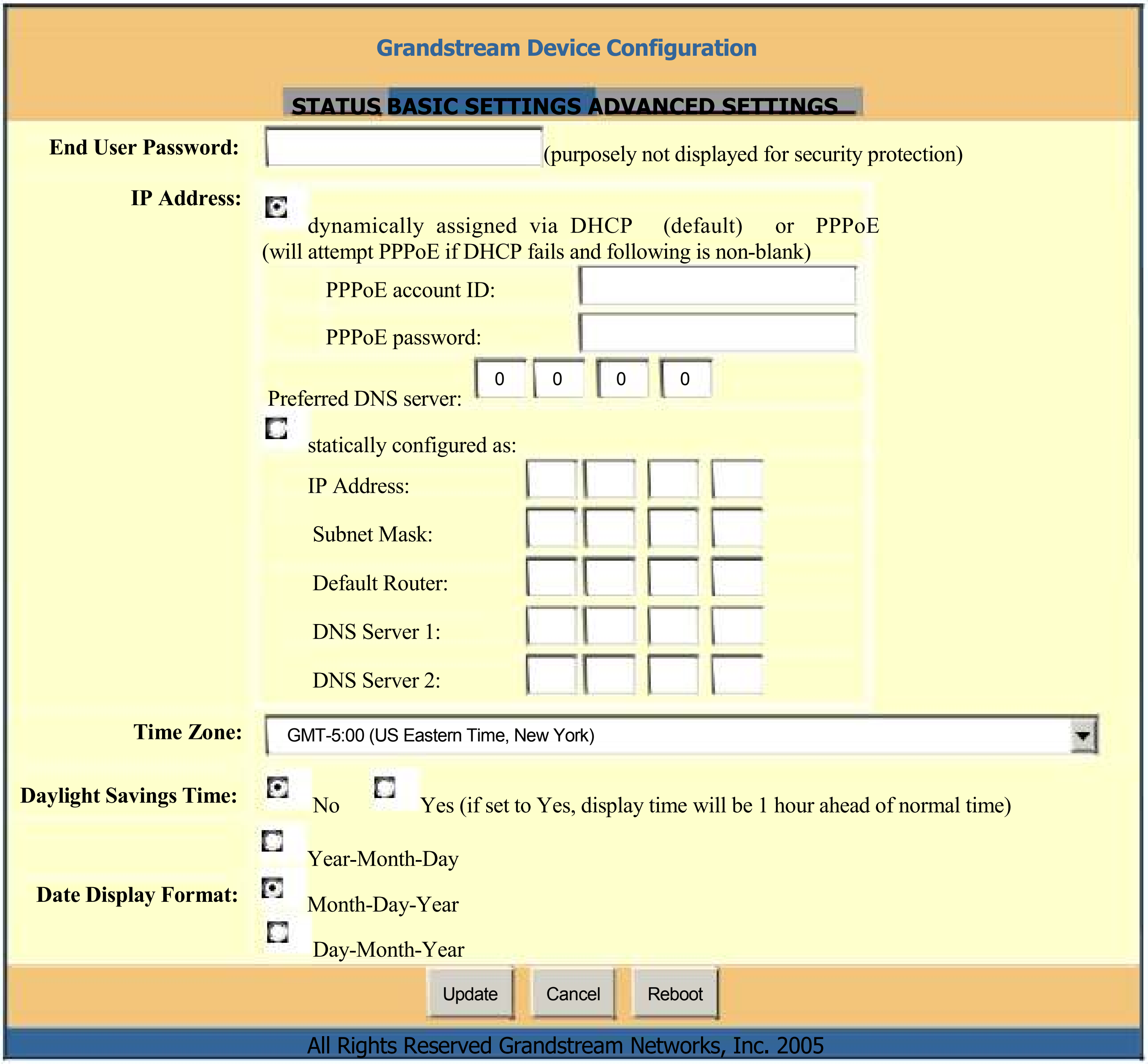
Configuration Menus

After inputting the password into the login screen, the embedded Web server of the IP phone will respond with the Configuration Page screens, which are explained in detail below.

Status Page:

This page is just for information.

Basic Settings: Tpad settings are in RED TEXT



Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS

End User Password: (purposely not displayed for security protection)

IP Address:

dynamically assigned via DHCP (default) or PPPoE
(will attempt PPPoE if DHCP fails and following is non-blank)

PPPoE account ID:

PPPoE password:

Preferred DNS server: 0 0 0 0

statically configured as:

IP Address:

Subnet Mask:

Default Router:

DNS Server 1:

DNS Server 2:

Time Zone: GMT-5:00 (US Eastern Time, New York)

Daylight Savings Time: No Yes (if set to Yes, display time will be 1 hour ahead of normal time)

Date Display Format:

Year-Month-Day

Month-Day-Year

Day-Month-Year

Update Cancel Reboot

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End user Password	This contains the password for end user to access the Web Configuration Menu. This field is case sensitive and maximum length is 25 characters. The default end user password is ***"123***". For Administrator the default password is "***admin"***
IP Address	There are 2 modes under which the IP phone can operate: - If DHCP mode is enabled, then all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory) and the IP phone will acquire its IP address from the first DHCP server it discovers on the LAN it attaches to. - If Static IP mode is selected, the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (mandatory), DNS Server 2 (optional) fields need to be configured. These fields are set to zero by default. DHCP mode is recommended.
Time zone	Displayed date/time will be adjusted according to the specified time zone.
Daylight savings time	Default NO. If set to Yes, then the displayed time will be 1 hour ahead of normal time.
Date displayformat	This parameter controls the date display format.

Advanced Settings:

The Tpad settings are in RED TEXT

Grandstream Device Configuration

ADVANCED SETTINGS

Admin Password:		(purposely not displayed for security protection)
SIP Server:	sip.tpad.com	(Tpad SIP Server)
Outbound Proxy:	sip.tpad.com	(Tpad SIP proxy server)
SIP User ID:	1123444	(Your TPAD phone number)
Authenticate ID:	1123444	(Your TPAD phone number)
Authenticate Password:	********	(Your TPAD password)
Name:	Tom Smith	(optional, e.g., John Doe)

Advanced Options:

Preferred Vocoder: (in listed order)	choice 1:	G.729A /B
	choice 2:	iLBC
	choice 3:	G.723.1
	choice 4:	PCMU
	choice 5:	PCMA
	choice 6:	G.726-32
	choice 7:	G.722 (w ide band)
	choice 8:	PCMA

G723 rate:	(•) 6.3kbps encoding rate ( ) 5.3kbps encoding rate
iLBC frame size:	( ) 20ms (•) 30ms
iLBC payload type:	97 (between 96 and 127, default is 97)
Silence Suppression:	(•) No ( ) Yes
Voice Frames per TX:	2 (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)
Layer 3 QoS:	48 (Diff-Serv or Precedence value)
Layer 2 QoS:	802.1Q/VLAN Tag 0 802.1p priority value 0 (0-7)
Allow incoming SIP messages from SIP proxy only:	(•) No ( ) Yes

Use DNS SRV:	No Yes
User ID is phone number:	No Yes
SIP Registration:	Yes No
Unregister On Reboot:	Yes No
Register Expiration:	900 (in seconds. default 1 hour, max 45 days)
Early Dial:	No Yes (use "Yes" only if proxy supports 484 response)
Allow outgoing call without Registration:	No Yes
Dial Plan Prefix:	(this prefix string is added to each dialed number)
No Key Entry Timeout:	4 (in seconds, default is 4 seconds)
Use # as Dial Key:	No Yes (if set to Yes, "#" will function as the Dial key)
local SIP port:	5060 (default 5060)
local RTP port:	5004 (1024-65535, default 5004)
Use random port:	No Yes
NAT Traversal:	No Yes, STUN server is: (URI or IP:port)
keep-alive interval:	20 (in seconds, default 20 seconds)
Use NAT IP	(if specified, this IP address is used in SIP/SDP message)
Proxy-Require:	(if specified, the content will appear in Proxy-Require header)
Voice Mail UserID:	(User ID/extension for 3rd party voice mail system)
SUBSCRIBE for MWI:	No, do not send SUBSCRIBE for Message Waiting Indication Yes, send periodical SUBSCRIBE for Message Waiting Indication
Auto Answer:	No Yes
Offhook Auto-Dial:	(User ID/extension to dial automatically when offhook)
Enable Call Features:	No Yes (if Yes, Call Forwarding & Call-Waiting-Disable are supported locally)
Disable Call-Waiting:	No Yes

Send DTMF:	☐ in-audio ☑ via RTP (RFC2833) ☐ via SIP INFO
DTMF Payload Type:	101
Send Flash Event:	(•) No ( ) Yes (Flash will be sent as a DTMF event if set to Yes)
Onhook Threshold:	800 ms
NTP Server:	time.nist.gov (URI or IP address)
Default Ring Tone:	( ) system ring tone ( ) custom ring tone 1, used if incoming caller ID is (•) custom ring tone 2, used if incoming caller ID is 1001 ( ) custom ring tone 3, used if incoming caller ID is
Send Anonymous:	(•) No ( ) Yes (caller ID will be blocked if set to Yes)
Anonymous Method:	(•) Use From Header ( ) Use Privacy Header
Time to ring:	60 seconds
Special Feature:	Standard
Syslog Server:	
Syslog Level:	NONE
Firmware Upgrade and Provisioning:	Upgrade Via ( ) TFTP (•) HTTP Firmware Server Path: fm.grandstream.com/gs Config Server Path: Firmware File Prefix: Firmware File Postfix: Config File Prefix: Config File Postfix: Automatic Upgrade: (•) No ( ) Yes, check for upgrade every 10080 minutes (default 7 days) (•) Always Check for New Firmware ( ) Check New Firmware only when F/W pre/suffix changes ( ) Always Skip the Firmware Check

Firmware Key:	(in Hexadecimal Representation)
Authenticate Conf File:	No Yes (cfg file would be authenticated before acceptance if set to Yes)
Lock keypad update:	No Yes (configuration update via keypad is disabled if set to Yes)
Allow conf SIP Account in Basic Settings:	No Yes
Override MTU Size:	0

Update   Cancel   Reboot

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Admin Password	Administrator password. Leave default
SIP Server	Enter sip.tpad.com
Outbound Proxy	Enter sip.tpad.com
SIP User ID	Enter your TPAD phone number e.g. 1123444
Authentication ID	Enter your TPAD phone number e.g. 1123444
Authenticate Password	Enter your TPAD password
Preferred Vocoder	The BudgeTone IP phone supports up to 8 different codec types including G711-ulaw (PCMU), G711-alaw (PCMA), G723, G729A, G726-32 (ADPCM), G722, G728 and iLBC. Leave as Default.
G723 Rate:	Encoding rate for G723 codec. Leave default, 6.3kbps rate is set.

iLBC frame size	iLBC packet frame size. Leave Default is 20ms.
iLBC payload type	Payload type for iLBC. Leave Default value is 97.
Silence Suppression	This controls the silence suppression/VAD feature of G723 and G729. If set to "Yes", when a silence is detected, small quantity of VAD packets (instead of audio packets) will be sent during the period of no talking. Set to "No", this feature is disabled.
Voice Frames per	Leave as Default
Layer 3 QoS	Default value is 48.
Layer 2 QoS	Default setting is blank or "0"
Allow incoming SIP messages from SIP proxy only	Default is No.

Use DNS SRV	Default is No. If set to Yes, the phone will use DNS SRV configured to lookup for the server
Use ID is phone number	If "Yes" is set, a "user=phone" parameter will be attached to the "From" header in SIP request, which will be processed by supported SIP proxy. Default is No.
SIP registration	This parameter controls whether the BudgeTone phone needs to send REGISTER messages to the proxy server. The default setting is "Yes".
Unregister On Reboot	Default is No.
Register Expiration	The default interval is 3600 seconds (or 1 hour).
Early Dial	Default setting is No.
Allow outgoing call without Registration	Default is No.
Dial Plan Prefix	Sets the prefix added to each dialed number. If configured, the prefix will be added to EVERY number input. Default is none.
No key Entry Timeout	Default is 4 seconds. User can short or extend that depends on digits dialed habit
Use # as Dial Key	Default is NO.
Local SIP port	The default value is 5060.

Local RTP port	The default value is 5004.
Use Random port	Default is No. If set to Yes, the device will pick randomly generated SIP and RTP ports. This is usually necessary and useful when multiple IP Phones are behind the same full cone NAT router.
NAT Traversal	Default is NO. It should be set to YES if the device is behind NAT router.
Keep alive interval	Default is 20 seconds.
Use NAT IP	Default is blank.
Proxy-Require	Default is blank.
Voice Mail User ID	Default is blank.
Subscribe for MWI	Default is No.
Auto Answer	Default is No.
Offhook Auto-Dial	Default is blank.
Enable call features	Default is Yes.
Disable Call Waiting	Default is No.

Send DTMF	Set as RTP (RFC2833)
DTMF Payload Type	Default is 101
Send Flash Event	Default is NO.
Onhook Threshold	Default is 800ms.
NTP server	Leave as Default.
Default Ring Tone	Default set as "system ring tone"
Send Anonymous	Default is NO.
Anonymous Method	Set to "Use From Header".
Time to ring	Default is 60 seconds
Special Feature	Default is Standard.
Syslog Server	Default is None.

Click **UPDATE** at the bottom of the configuration page then click to view your changes and then click the **REBOOT** on the phone menu screen. Make sure you click **UPDATE** before clicking **REBOOT** to make sure your changes are implemented correctly.