



ATA (Analog Telephone Adapter) Configuration Guide

You can use your DynaSky VoIP at home or anywhere else on your landline, and enjoy features such as free in-Net calling, voicemail and unlimited call receiving. Find ATAs at any Target, Best Buy, Amazon.com, and you're connected.

Here we'll use the [Grandstream HandyTone 286 ATA](#)™ as an example and provide step-by-step instructions on how to set it up on your landline phone.



Please have on hand:

- ATA
- Ethernet cable (correctly connected to your router)
- Landline phone (functioning)
- Landline phone cable
- PC/laptop with Internet access

Step One: Connecting the ATA

1. Power off router and modem
2. Take the Ethernet cable and connect to RJ45 port, make sure the other end is correctly connected to your router
3. Unplug your phone cable from its wall socket. Then unplug from landline phone, connect instead to the ATA RJ11 port
4. Power on the router and modem. Wait for the process to be completed, until all the lights are green
5. Plug both the ATA and landline into a power socket
6. Turn on your computer. Make sure the ATA is fully powered on, this may take 5 minutes

Step Two: Confirm ATA IP Address



1. On your landline phone, press “***” then dial “02.” The system will announce your IP address.
2. On your computer, open a browser and go to “http://<IP Address>” (it would look like: http:// 192.168.0.1/)
3. You will be prompted to enter password “admin.” Click on “Login”
4. You can now begin configuration.

Step Three: Basic Settings

1. IP Address – The IP address of the ATA as confirmed in Step Two.
2. Subnet Mask – It’s always 255.255.255.0
3. Default Router – By default, D-link router IP address is 192.168.0.1 (which is what I’m using) as opposed to Linksys router default IP of 192.168.1.1. Check the documentation of your router to find the default IP.
4. DNS server 1: DNS server address can be found on the status screen of your router or you can always check with your ISP (Internet Service Provider). DNS server 2: Same as DNS server 1.
5. Time Zone: Change it to your current time zone. Otherwise, choosing “dynamically assigned via DHCP” will work for most people.

The screenshot shows the 'Grandstream Device Configuration' web interface. The 'BASIC SETTINGS' tab is selected. The 'End User Password' field is empty. The 'IP Address' section has two radio buttons: 'dynamically assigned via DHCP (default) or PPPoE' (selected) and 'statically configured as:'. Under 'statically configured as:', there are input fields for IP Address (192.168.0.102), Subnet Mask (255.255.255.0), Default Router (192.168.0.1), DNS Server 1 (199.234.69.158), and DNS Server 2 (199.234.65.250). The 'Preferred DNS server' field is set to 0.0.0.0. The 'Time Zone' dropdown is set to 'GMT-8:00 (US Pacific Time, Los Angeles)'. The 'Daylight Savings Time' section has 'No' selected. At the bottom, there are 'Update', 'Cancel', and 'Reboot' buttons. A footer note reads 'All Rights Reserved Grandstream Networks, Inc. 2005'.

Step Four: Advanced Settings

1. SIP server: voip.dynasky.com
2. Outbound Proxy: voip.dynasky.com
3. SIP user ID: dynasky username-voip.dynasky.com
4. Authenticate ID: dynasky username-voip.dynasky.com



5. Authenticate Password: dynasky softphone password

Some advanced settings highlights:

6. Preferred Vocoder (Codec): Dynasky supports G711.u, also known as PCMU, so PCMU should be on Choice 1.
7. User ID is phone number: No. Dynasky doesn't use VoIP number as a username.
8. SIP Registration: Yes.
9. Unregister on reboot: Yes. Sometimes the changes won't take effect if the ATA doesn't unregister first before rebooting.
10. Send DTMF: via RTP (RFC2833)
11. DTMF Payload Type: 101 Click on update then reboot after entering the details.

Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS

Admin Password: (purposely not displayed for security protection)
SIP Server: (e.g., sip.mycompany.com, or IP address)
Outbound Proxy: (e.g., proxy.myprovider.com, or IP address, if any)
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 security protection)
Name: (optional, e.g., John Doe)

Advanced Options:

Preferred Vocoder: (in listed order)
 choice 1:
 choice 2:
 choice 3:
 choice 4:
 choice 5:
 choice 6:
 choice 7:

G723 rate: 6.3kbps encoding rate 5.3kbps encoding rate
iLBC frame size: 20ms 30ms
iLBC payload type: (between 96 and 127, default is 97)
Silence Suppression: No Yes
Voice Frames per TX: (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)
Fax Mode: T.38 (Auto Detect) Pass-Through
Layer 3 QoS: (Diff-Serv or Precedence value)
Layer 2 QoS: 802.1Q/VLAN Tag 802.1p priority value (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: (in seconds, default 1 hour, max 45 days)
Early Dial: No Yes (use "Yes" only if proxy supports 484 response)
Dial Plan Prefix: (this prefix string is added to each dialed number)
No Key Entry Timeout: (in seconds, default is 4 seconds)
Use # as Dial Key: No Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)

local SIP port: (default 5060)
local RTP port: (1024-65535, default 5004)
Use random port: No Yes
NAT Traversal: No
 Yes, STUN server is: (URI or IP:port)

keep-alive interval: (in seconds, default 20 seconds)
Use NAT IP: (used in SIP/SDP message if specified)
Proxy-Require:

Firmware Upgrade:
 Via TFTP Server . . .
 Via HTTP Server
 Automatic HTTP Upgrade:
 No Yes, check for upgrade every minutes (default 7 days)

SUBSCRIBE for MWI: No, do not send SUBSCRIBE for Message Waiting Indication
 Yes, send periodical SUBSCRIBE for Message Waiting Indication

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:

Send Flash Event: No Yes (Flash will be sent as a DTMF event if set to Yes)
FXS Impedance:

Caller ID Scheme:

Onhook Voltage:

Polarity Reversal: No Yes (reverse polarity upon call establishment and termination)

NTP Server: (URI or IP address)

Send Anonymous: No Yes (caller ID will be blocked if set to Yes)

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:

Update Cancel Reboot



After the reboot, check the status to see if you are registered with DynaSky. If not, then review the details above and make sure everything is configured accordingly. If registered, you are now ready to make DynaSky VoIP calls on your landline. To call a US number, just dial the area code, destination number then press #. For international calls, dial 011, followed by the country code, area code then phone number.

Grandstream Device Configuration	
STATUS	BASIC SETTINGS
MAC Address:	00.0B.82.01.06.32
WAN IP Address:	192.168.0.102
Product Model:	HT286
Software Version:	Program-- 1.0.7.19 Bootloader-- 1.0.8.11 HTML-- 1.0.7.11 VOC-- 1.0.0.10
System Up Time:	0 day(s) 0 hour(s) 19 minute(s)
Registered:	Yes
PPPoE Link Up:	disabled
NAT:	
NAT Mapped IP:	0.0.0.0
NAT Mapped Port:	0
Total Inbound Calls:	2
Total Outbound Calls:	1
Total Missed Calls:	0
Total Call Time (in minutes):	4
Total SIP Message Sent:	70
Total SIP Message Received:	65
Total RTP Packet Sent:	7956
Total RTP Packet Received:	3705
Total RTP Packet Loss:	146

All Rights Reserved Grandstream Networks, Inc. 2005