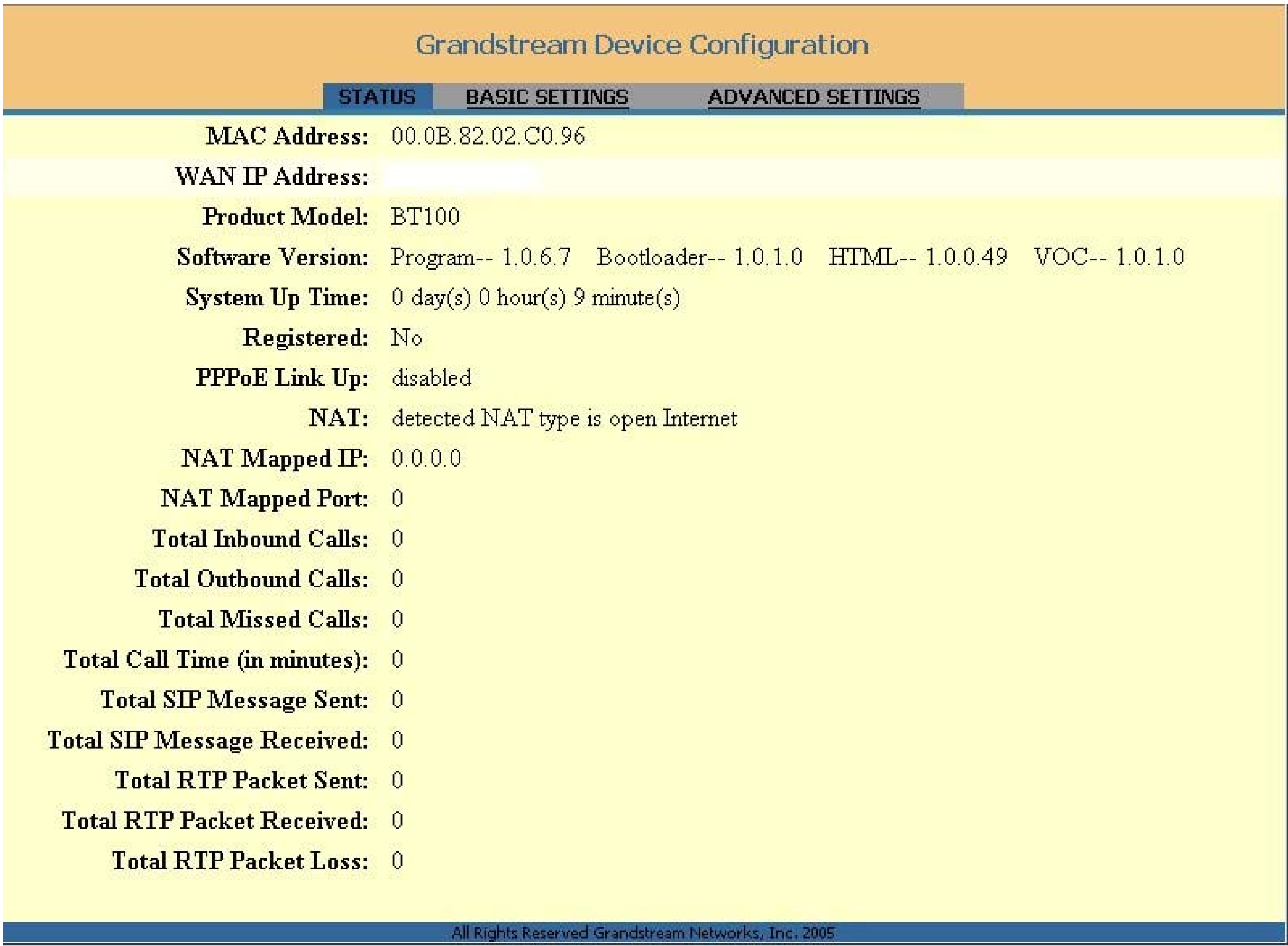
Telefon se konfiguruje přes www rozhraní. Po zapojení podle ukázkového zapojení zvedněte sluchátko a stiskněte tlačítko menu na telefonu. Na disleji se zobrazí IP adresa telefonu. Tuto adresu zadejte do www prohlížeče na počítači. (co tři číslice zadejte "." např.:192.168.1.123). Původní heslo je "admin".

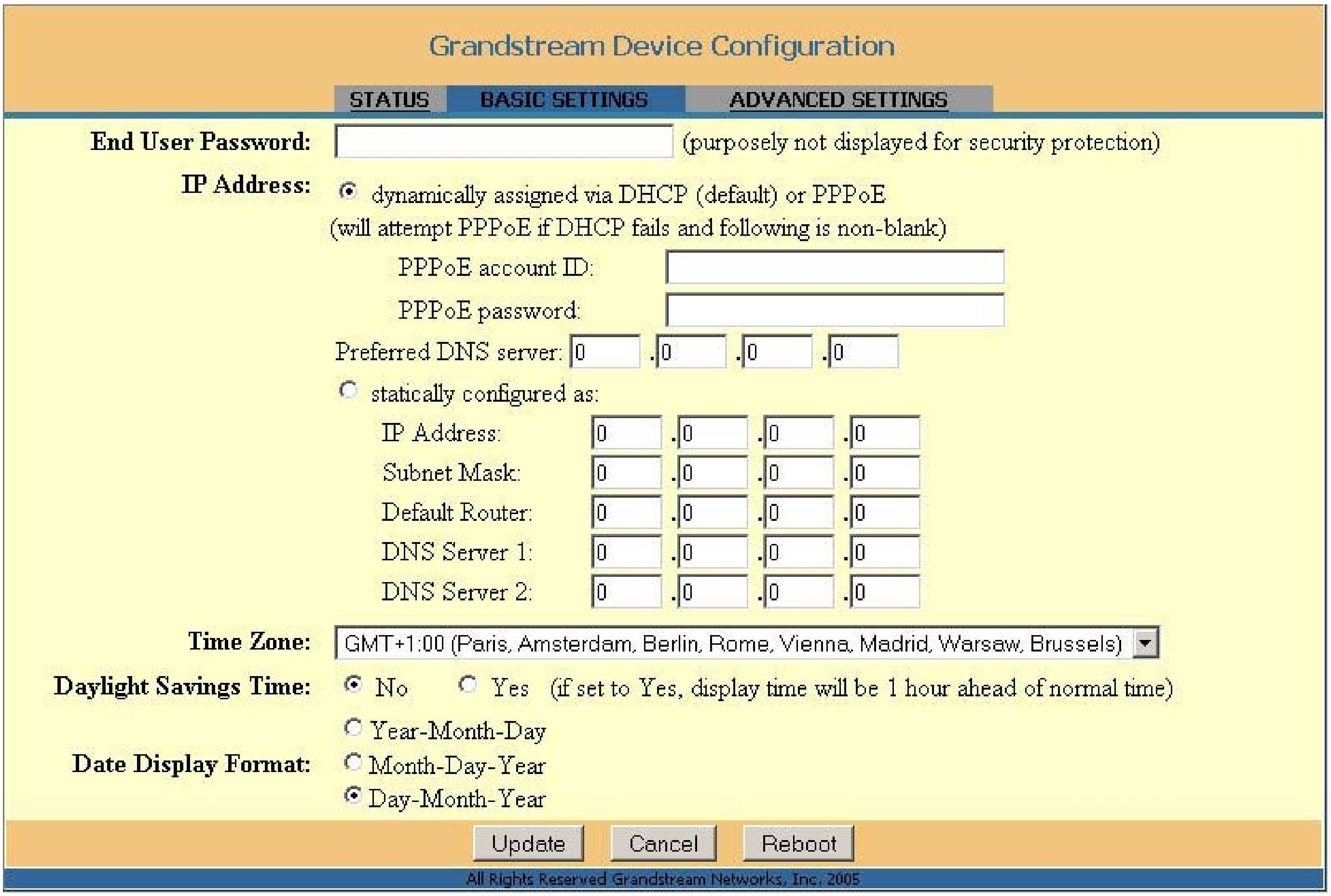
Sip Server a Outbound Proxy: zjistíte na http://www.802.cz/zjisteni-sip-serveru/



Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS

MAC Address:	00.0B.82.02.C0.96
WAN IP Address:	
Product Model:	BT100
Software Version:	Program-- 1.0.6.7 Bootloader-- 1.0.1.0 HTML-- 1.0.0.49 VOC-- 1.0.1.0
System Up Time:	0 day(s) 0 hour(s) 9 minute(s)
Registered:	No
PPPoE Link Up:	disabled
NAT:	detected NAT type is open Internet
NAT Mapped IP:	0.0.0.0
NAT Mapped Port:	0
Total Inbound Calls:	0
Total Outbound Calls:	0
Total Missed Calls:	0
Total Call Time (in minutes):	0
Total SIP Message Sent:	0
Total SIP Message Received:	0
Total RTP Packet Sent:	0
Total RTP Packet Received:	0
Total RTP Packet Loss:	0


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: 0 .0 .0 .0
Subnet Mask: 0 .0 .0 .0
Default Router: 0 .0 .0 .0
DNS Server 1: 0 .0 .0 .0
DNS Server 2: 0 .0 .0 .0

Time Zone: GMT+1:00 (Paris, Amsterdam, Berlin, Rome, Vienna, Madrid, Warsaw, Brussels)

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

Grandstream Device Configuration

STATUS | BASIC SETTINGS | ADVANCED SETTINGS

Admin Password:		(purposely not displayed for security protection)
SIP Server:	podle vašeho TO	(e.g., sip.mycompany.com, or IP address)
Outbound Proxy:		(e.g., proxy.myprovider.com, or IP address, if any)
SIP User ID:	vaše telefonní číslo	(the user part of an SIP address)
Authenticate ID:	vaše telefonní číslo	(can be identical to or different from **SIP User ID**)
Authenticate Password:	vaše 802.cz heslo	(purposely not displayed for security protection)
Name:		(optional, e.g., John Doe)

Advanced Options:

Preferred Vocoder: *(in listed order)*	choice 1:	current setting is " PCMU"
	choice 2:	current setting is " PCMA"
	choice 3:	current setting is " G723"
	choice 4:	current setting is " G729"
	choice 5:	current setting is " G726-32"
	choice 6:	current setting is " G728"
	choice 7:	current setting is " iLBC"
	choice 8:	current setting is " G722"
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)
Use DNS SRV:	◉ No	○ Yes
User ID is phone number:	◉ No	○ Yes
SIP Registration:	◉ Yes	○ No
Unregister On Reboot:	○ Yes	◉ No
Register Expiration:	10	(in minutes. 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:	4 (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:	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)
Firmware Upgrade:	○ Via TFTP Server 168 . 75 . 215 . 188 ◉ Via HTTP Server fm.grandstream.com/gs Automatic HTTP Upgrade: ◉ No ○ Yes, check for upgrade every 7 days (default 7 days)
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)
NTP Server:	**212.71.150.2** (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 ○ custom ring tone 3, used if incoming caller ID is
Send Anonymous:	◉ No ○ Yes (caller ID will be blocked if set to Yes)
Lock keypad update:	◉ No ○ Yes (configuration update via keypad is disabled if set to Yes)
Syslog Server:	
Syslog Level:	current setting is "NONE"

Update Cancel Reboot