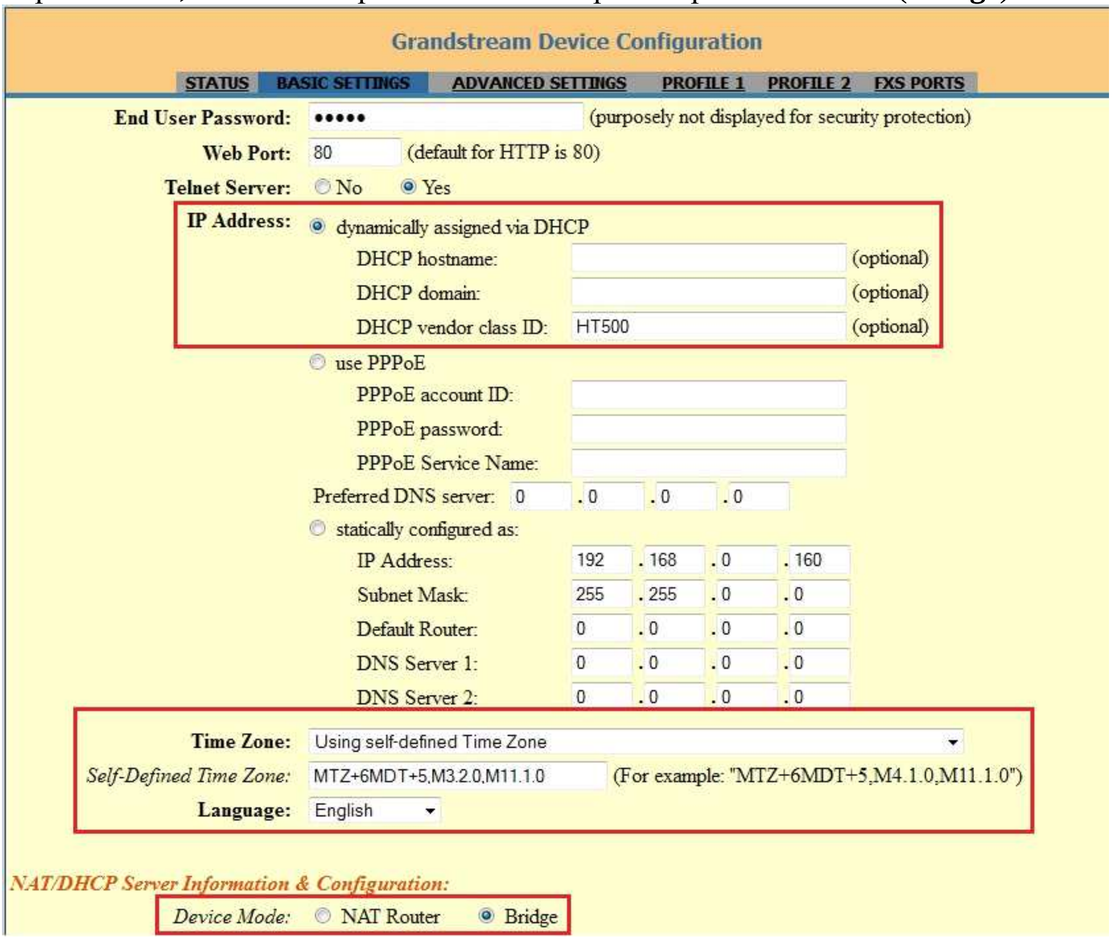
Grandstream GXW4008 - аналоговый шлюз на 8 FXS портов для подключения абонентских устройств (аналоговые телефоны и факсы).
Данная инструкция актуальна для моделей **Grandstream GXW4008** и **Grandstream GXW4004**.

1. Подключите к FXS порту шлюза телефонный аппарат. Сам шлюз подключите к сети через **WAN порт**. На телефонном аппарате наберите *****129**, чтобы **разрешить доступ к WEB интерфейсу шлюза через WAN порт**, а затем ***** 999** для перезагрузки шлюза. Чтобы узнать IP адрес WAN порта, наберите на телефонном аппарате *****02**. Шлюз проговорит IP адрес WAN порта. Теперь вы можете подключиться к WEB интерфейсу шлюза через WAN порт.

2. Подключитесь к шлюзу по IP адресу, узнанному в п.1 **Password - admin**. Перейдите на вкладку **Basic Settings**. Тут можно установить IP адрес для WAN порта шлюза, выставить временной пояс и режим работы шлюза (**Bridge**).



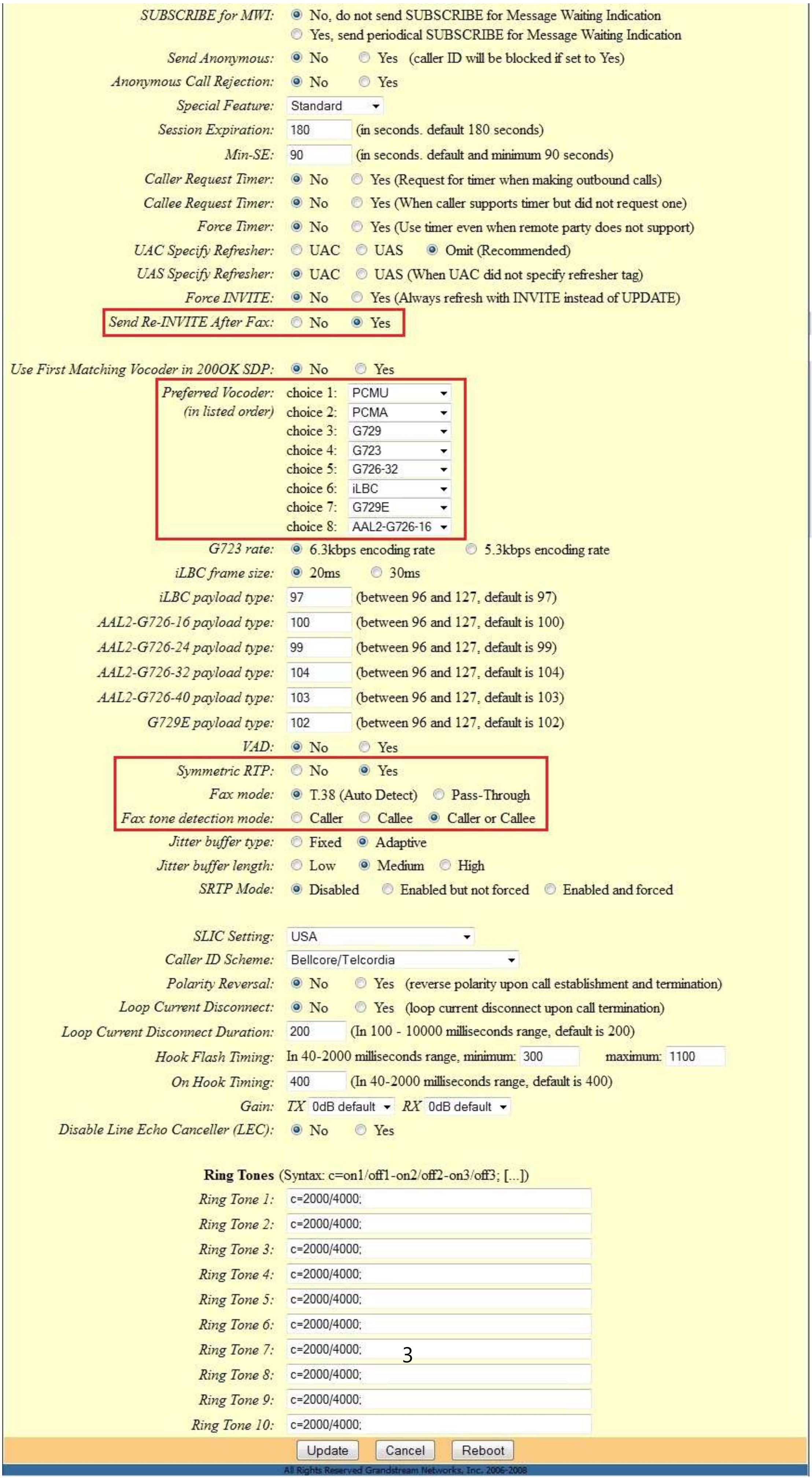
Нажмите внизу Update и Reboot.

3. В разделе **Profile1** заполните поля согласно рисунку.

Grandstream Device Configuration

STATUS | BASIC SETTINGS | ADVANCED SETTINGS | PROFILE 1 | PROFILE 2 | FXS PORTS

Profile Active:	○ No ◉ Yes
Primary SIP Server:	voip.uiscom.ru:9060 (e.g., sip.mycompany.com, or IP address)
Failover SIP Server:	(Optional, used when primary server no response)
Prefer Primary SIP Server:	◉ No ○ Yes (yes - will register to Primary Server if Failover registration expires)
Outbound Proxy:	(e.g., proxy.myprovider.com, or IP address, if any)
SIP transport:	◉ UDP ○ TCP ○ TLS (default is UDP)
NAT Traversal (STUN):	○ No ◉ No, but send keep-alive ○ Yes
DNS Mode:	◉ A Record ○ SRV ○ NAPTR/SRV
User ID is phone number:	◉ No ○ Yes
SIP Registration:	○ No ◉ Yes
Unregister On Reboot:	○ No ◉ Yes
Outgoing Call without Registration:	◉ No ○ Yes
Register Expiration:	10 (in minutes. default 1 hour, max 45 days)
SIP Registration Failure Retry Wait Time:	20 (in seconds. Between 1-3600, default is 20)
local SIP port:	5060 (default is 5060 for UDP and TCP; 5061 for TLS)
local RTP port:	5004 (1024-65535, default 5004)
Use random port:	◉ No ○ Yes
Refer-To Use Target Contact:	◉ No ○ Yes
Transfer on Conference Hangup:	◉ No ○ Yes
Remove OBP from Route Header:	◉ No ○ Yes
Support SIP Instance ID:	○ No ◉ Yes
Validate Incoming SIP Message:	◉ No ○ Yes
Check SIP User ID for incoming INVITE:	◉ No ○ Yes (no direct IP calling if Yes)
Allow Incoming SIP Messages from SIP Proxy Only:	◉ No ○ Yes (no direct IP calling if Yes)
SIP T1 Timeout:	0.5 sec
SIP T2 Interval:	4 sec
DTMF Payload Type:	101
Preferred DTMF method: (in listed order)	Priority 1: RFC2833 Priority 2: SIP INFO Priority 3: In-audio
Send Hook Flash Event:	◉ No ○ Yes (Hook-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)
Proxy-Require:	
Use NAT IP:	(used in SIP/SDP message if specified)
Distinctive Ring Tone:	Ring Tone 1 used if incoming caller ID is Ring Tone 1 used if incoming caller ID is Ring Tone 1 used if incoming caller ID is
Disable Call-Waiting:	○ No ◉ Yes
Disable Call-Waiting Caller ID:	○ No ◉ Yes
Disable Call-Waiting Tone:	○ No ◉ Yes
Disable Reminder Ring for On-Hold Call:	◉ No ○ Yes
Disable Visual MWI:	◉ No ○ Yes
Ring Timeout:	60 (10-300, default is 60 seconds)
Hunting Group Ring Timeout:	20 (5-300, default is 20 seconds)
Delayed Call Forward Wait Time:	20 (Allowed range 1-120, in seconds.)
No Key Entry Timeout:	4 (in seconds, default is 4 seconds)
Early Dial:	◉ No ○ Yes (use "Yes" only if proxy supports 484 response)
Dial Plan Prefix:	(this prefix string is added to each dialed number)
Use # as Dial Key:	○ No ◉ Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)
Dial Plan:	{ x+ | *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
Special Feature:	Standard
Session Expiration:	180 (in seconds. default 180 seconds)
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)
Send Re-INVITE After Fax:	○ No ◉ Yes
Use First Matching Vocoder in 200OK SDP:	◉ No ○ Yes
Preferred Vocoder: (in listed order)	choice 1: PCMU choice 2: PCMA choice 3: G729 choice 4: G723 choice 5: G726-32 choice 6: iLBC choice 7: G729E choice 8: AAL2-G726-16
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)
AAL2-G726-16 payload type:	100 (between 96 and 127, default is 100)
AAL2-G726-24 payload type:	99 (between 96 and 127, default is 99)
AAL2-G726-32 payload type:	104 (between 96 and 127, default is 104)
AAL2-G726-40 payload type:	103 (between 96 and 127, default is 103)
G729E payload type:	102 (between 96 and 127, default is 102)
VAD:	◉ No ○ Yes
Symmetric RTP:	○ No ◉ Yes
Fax mode:	◉ T.38 (Auto Detect) ○ Pass-Through
Fax tone detection mode:	○ Caller ○ Callee ◉ Caller or Callee
Jitter buffer type:	○ Fixed ◉ Adaptive
Jitter buffer length:	○ Low ◉ Medium ○ High
SRTP Mode:	◉ Disabled ○ Enabled but not forced ○ Enabled and forced
SLIC Setting:	USA
Caller ID Scheme:	Bellcore/Telcordia
Polarity Reversal:	◉ No ○ Yes (reverse polarity upon call establishment and termination)
Loop Current Disconnect:	◉ No ○ Yes (loop current disconnect upon call termination)
Loop Current Disconnect Duration:	200 (In 100 - 10000 milliseconds range, default is 200)
Hook Flash Timing:	In 40-2000 milliseconds range, minimum: 300 maximum: 1100
On Hook Timing:	400 (In 40-2000 milliseconds range, default is 400)
Gain:	*TX* 0dB default *RX* 0dB default
Disable Line Echo Canceller (LEC):	◉ No ○ Yes
Ring Tones	(Syntax: c=on1/off1-on2/off2-on3/off3; [...])
Ring Tone 1:	c=2000/4000;
Ring Tone 2:	c=2000/4000;
Ring Tone 3:	c=2000/4000;
Ring Tone 4:	c=2000/4000;
Ring Tone 5:	c=2000/4000;
Ring Tone 6:	c=2000/4000;
Ring Tone 7:	c=2000/4000;
Ring Tone 8:	c=2000/4000;
Ring Tone 9:	c=2000/4000;
Ring Tone 10:	c=2000/4000;

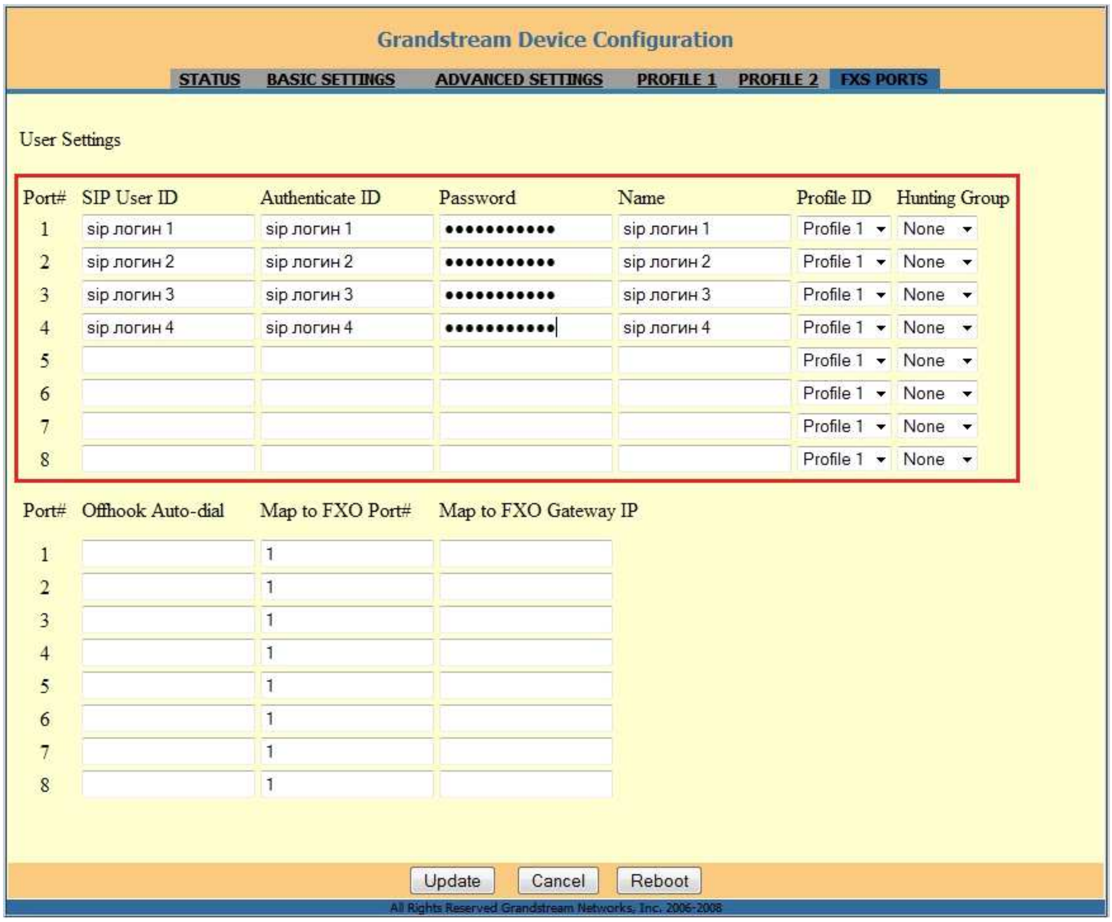
Update Cancel Reboot

Поле, помеченное на рисунке звездочкой:

Если Вы используете более одного голосового шлюза, обратите внимание, чтобы на **КАЖОЙ линии КАЖОГО шлюза** данный порт был разным (Вы можете использовать любой порт из диапазона 5060-5960).

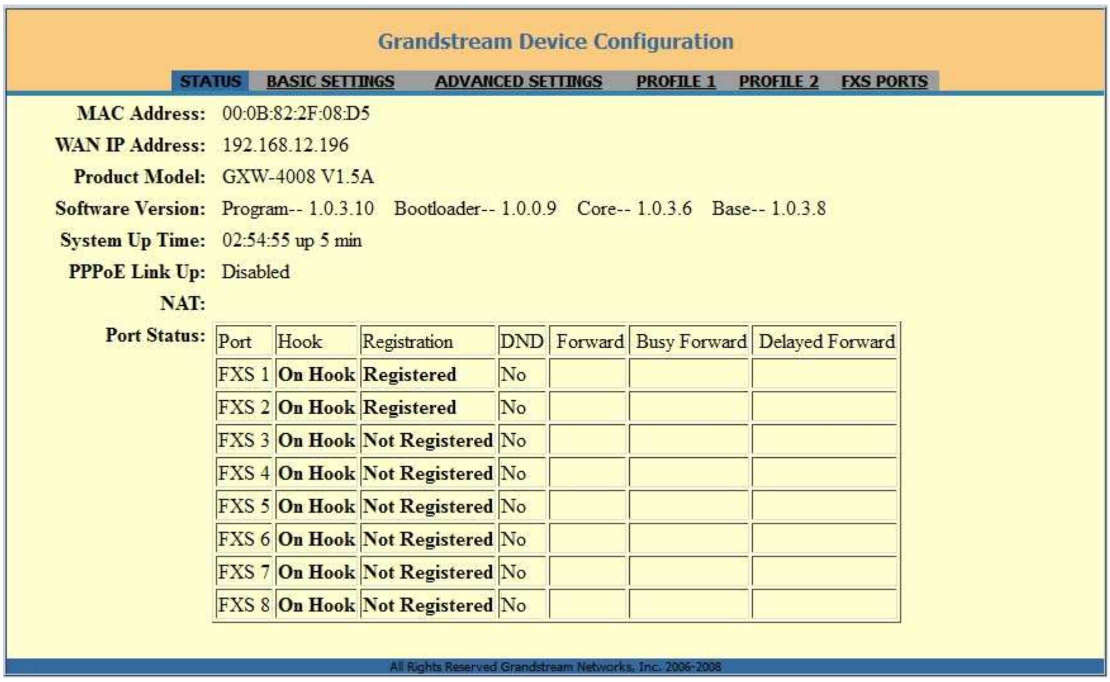
Нажмите внизу Update и Reboot.

4. В разделе **FXS Ports** введите логины и пароли от sip линий.



Нажмите внизу Update и Reboot.

5. Зайдите в раздел Status и обратите внимание, зарегистрировались ли порты шлюза на сервере регистрации. На рисунке видно, что Port Status 1, 2 - Registered.



На этом настройка голосового шлюза завершена.