

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, узанному 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**).

The screenshot shows the 'Grandstream Device Configuration' web interface, specifically the 'BASIC SETTINGS' tab. The interface is yellow with a blue header. The 'End User Password' field is masked with dots. The 'Web Port' is set to 80. The 'Telnet Server' is set to 'Yes'. The 'IP Address' section is highlighted with a red box and contains the following options: 'dynamically assigned via DHCP' (selected), 'DHCP hostname' (optional), 'DHCP domain' (optional), and 'DHCP vendor class ID' (HT500, optional). Below this are options for 'use PPPoE' and 'statically configured as'. The 'Time Zone' is set to 'Using self-defined Time Zone', and the 'Self-Defined Time Zone' is 'MTZ+6MDT+5.M3.2.0.M11.1.0'. The 'Language' is set to 'English'. At the bottom, the 'NAT/DHCP Server Information & Configuration' section is highlighted with a red box, showing 'Device Mode' set to 'Bridge'.

Field	Value	Notes
End User Password	•••••	(purposely not displayed for security protection)
Web Port	80	(default for HTTP is 80)
Telnet Server	<input checked="" type="radio"/> Yes	
IP Address	<input checked="" type="radio"/> dynamically assigned via DHCP	
DHCP hostname		(optional)
DHCP domain		(optional)
DHCP vendor class ID	HT500	(optional)
use PPPoE	<input type="radio"/>	
PPPoE account ID		
PPPoE password		
PPPoE Service Name		
Preferred DNS server	0 . 0 . 0 . 0	
statically configured as	<input type="radio"/>	
IP Address	192 . 168 . 0 . 160	
Subnet Mask	255 . 255 . 0 . 0	
Default Router	0 . 0 . 0 . 0	
DNS Server 1	0 . 0 . 0 . 0	
DNS Server 2	0 . 0 . 0 . 0	
Time Zone	Using self-defined Time Zone	
Self-Defined Time Zone	MTZ+6MDT+5.M3.2.0.M11.1.0	(For example: "MTZ+6MDT+5.M4.1.0.M11.1.0")
Language	English	
Device Mode	<input checked="" type="radio"/> 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:**  (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:**  (in minutes, default 1 hour, max 45 days)

**SIP Registration Failure Retry Wait Time:**  (in seconds, Between 1-3600, default is 20)

**local SIP port:**  (default is 5060 for UDP and TCP; 5061 for TLS)

**local RTP port:**  (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:**

**SIP T2 Interval:**

**DTMF Payload Type:**

**Preferred DTMF method:** Priority 1:  Priority 2:  Priority 3:

**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:**  (10-300, default is 60 seconds)

**Hunting Group Ring Timeout:**  (5-300, default is 20 seconds)

**Delayed Call Forward Wait Time:**  (Allowed range 1-120, in seconds.)

**No Key Entry Timeout:**  (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:**

*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 2000K 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; 3

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.

The screenshot shows the 'Grandstream Device Configuration' web interface, specifically the 'FXS PORTS' tab. The 'User Settings' section contains a table with 8 rows for SIP users. The first four rows are populated with 'sip\_логин 1' through '4' for both User ID and Authenticate ID, with masked passwords. The 'Name' field also contains the corresponding login names. The 'Profile ID' is set to 'Profile 1' and 'Hunting Group' to 'None' for all. The second table below, 'Map to FXO Port#', shows port 1 mapped to FXO port 1, with other ports 2-8 also mapped to 1. At the bottom, there are 'Update', 'Cancel', and 'Reboot' buttons.

Port#	SIP User ID	Authenticate ID	Password	Name	Profile ID	Hunting Group
1	sip_логин 1	sip_логин 1	●●●●●●●●	sip_логин 1	Profile 1	None
2	sip_логин 2	sip_логин 2	●●●●●●●●	sip_логин 2	Profile 1	None
3	sip_логин 3	sip_логин 3	●●●●●●●●	sip_логин 3	Profile 1	None
4	sip_логин 4	sip_логин 4	●●●●●●●●	sip_логин 4	Profile 1	None
5					Profile 1	None
6					Profile 1	None
7					Profile 1	None
8					Profile 1	None

  

Port#	Offhook Auto-dial	Map to FXO Port#	Map to FXO Gateway IP
1		1	
2		1	
3		1	
4		1	
5		1	
6		1	
7		1	
8		1	

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.

**Grandstream Device Configuration**

STATUS
BASIC SETTINGS
ADVANCED SETTINGS
PROFILE 1
PROFILE 2
FXS PORTS

**MAC Address:** 00:0B:82:2F:08:D5  
**WAN IP Address:** 192.168.12.196  
**Product Model:** GXW-4008 V1.5A  
**Software Version:** Program-- 1.0.3.10 Bootloader-- 1.0.0.9 Core-- 1.0.3.6 Base-- 1.0.3.8  
**System Up Time:** 02:54:55 up 5 min  
**PPPoE Link Up:** Disabled  
**NAT:**

**Port Status:**

Port	Hook	Registration	DND	Forward	Busy Forward	Delayed Forward
FXS 1	On Hook	Registered	No			
FXS 2	On Hook	Registered	No			
FXS 3	On Hook	Not Registered	No			
FXS 4	On Hook	Not Registered	No			
FXS 5	On Hook	Not Registered	No			
FXS 6	On Hook	Not Registered	No			
FXS 7	On Hook	Not Registered	No			
FXS 8	On Hook	Not Registered	No			

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На этом настройка голосового шлюза завершена.