



117588, Москва, ул. Ясногорская. д.5, стр. 1 (а/я 63)
ООО «НОВОСИСТЕМ», тел. +7 (495) 989-51-51
info@comagic.ru, www.comagic.ru
ИНН 7710311878, КПП 772801001, ОГРН 1037739054682

Настройка Grandstream GXW4004

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, указанному 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).

Grandstream Device Configuration

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

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

Web Port: 80 (default for HTTP is 80)

Telnet Server: No Yes

IP Address: dynamically assigned via DHCP

DHCP hostname: (optional)

DHCP domain: (optional)

DHCP vendor class ID: HT500 (optional)

use PPPoE

PPPoE account ID:

PPPoE password:

PPPoE Service Name:

Preferred DNS server: 0 . 0 . 0 . 0

statically configured as:

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

NAT/DHCP Server Information & Configuration:

Device Mode: NAT Router 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:

(in listed order) 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 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 Schema: 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 КАЖДОЙ линии КАЖДОГО 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 with the 'FXS PORTS' tab selected. The 'User Settings' section contains a table for configuring SIP users on ports 1 through 8. The first four rows are populated with 'sip логин 1' through 'sip логин 4' for both SIP User ID and Authenticate ID, with passwords masked by dots. The Name field also contains the respective SIP login. Profile ID is set to 'Profile 1' and Hunting Group to 'None'. Below this is a table for mapping ports to FXO ports and gateway IPs, with '1' entered in the 'Map to FXO Port#' column for all ports. 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	

Update Cancel Reboot

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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			

На этом настройка голосового шлюза завершена.