

Настройка Grandstream HT502

Grandstream HT502 - аналоговый шлюз на 2 FXS порта для подключения абонентских устройств (аналоговые телефоны и факсы).

Данная инструкция актуальна для моделей **Grandstream HT502** и **Grandstream HT702**.

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, узнаанному 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***).

* Режим работы доступен только для модели HT502.

Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS FXS PORT1 FXS PORT2

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

NAT maximum ports: 1024 (range: 0 - 4096, default is 1024)

Нажмите внизу Update и Reboot.

3. В разделе **FXS Port 1** заполните поля согласно рисунку*.

*Поле Fax Tone Detection Mode доступно только для модели HT502.

Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS **FXS PORT1** FXS PORT2

Account 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

SIP User ID: sip:логин (the user part of an SIP address)

Authenticate ID: sip:логин (can be identical to or different from SIP User ID)

Authenticate Password: ***** (purposely not displayed for security protection)

Name: sip:логин (optional, e.g., John Doe)

DNS Mode: A Record SRV NAPTR/SRV

Tel URI: Disabled

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

Enable Ring-Transfer: No (RFC5589 Semi-Attended Transfer) Yes

Disable Bellcore Style 3-Way Conference: No Yes (Using star code *23 for 3-way conference)

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

Disable DTMF Negotiation: No (negotiate with peer) Yes (use above DTMF order without negotiation)

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)

Offhook Auto-Dial: (User ID/extension to dial automatically when offhook)

Proxy-Require:

Use NAT IP: (used in SIP/SDP message if specified)

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

Distinctive Ring Tone:

Disable Call-Waiting: No Yes

Disable Call-Waiting Caller ID: No Yes

Disable Call-Waiting Tone: No Yes

Disable Receiver Offhook Tone: No Yes (ROH tone will not be played after offhook for 60 seconds)
 Disable Reminder Ring for On-Hold Call: No Yes
 Disable Visual MWI: No Yes
 Ring Timeout: 60 (10-300, default is 60 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
 Enable Silence Detection for Fax Disconnect: No Yes
 Enable 100rel: 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
 Voice Frames per TX: 2 (default 2, from 1 to 4 for G711/G726/G729)
 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)
 G726-32 Payload Type: 112 (between 96 and 127, default is 112)
 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

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).

4. Нажмите внизу Update и Reboot. После этого зайдите в раздел Status и обратите внимание, зарегистрировались ли порты шлюза на сервере регистрации. На рисунке видно, что Port Status 1 - Registered.

Grandstream Device Configuration

STATUS
BASIC SETTINGS
ADVANCED SETTINGS
FXS PORT1
FXS PORT2

MAC Address: WAN-- 00:0B:82:3A:41:D1 LAN-- 00:0B:82:3A:41:D0 (**Device MAC**)
WAN IP Address: 192.168.12.184
Product Model: HT-502 V1.2A
Software Version: Program-- 1.0.5.10 Bootloader-- 1.0.0.15 Core-- 1.0.5.9 Base-- 1.0.5.10
System Up Time: 08:29:50 up 0 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	Not Registered	No			

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Если Вам необходимо настроить две линии, возвращаетесь к пункту 3 и настраиваете **FXS Port2**.

На этом настройка голосового шлюза завершена.