

## Настройка Grandstream HT702

Grandstream HT702 - 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:login (the user part of an SIP address)

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

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

Name: sip:login (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.