

ارتباطات هوشمند **Kava**

انتقال خط گیتوی گرند استریم HT-503



تهیه شده توسط

سینا پروانه

www.Kavatelecom.com/blog

www.VoIPshop.ir

ارتباطات هوشمند **KavaTelecom.com** <http://www.KavaTelecom.com>

تلفن : 021-4956 - فاکس: ۴۱۷۸۷۹۸

جهت دسترسی به پنل وب این دستگاه طبق مراحل فوق عمل نمایید :

کامپیوتر خود را مستقیماً به پورت LAN دستگاه متصل نموده و شبکه خود را در رنج آی پی ۱۹۲،۱۶۸،۲،۰ تنظیم نمایید.

جهت ورود به صفحه تنظیمات ATA از آی پی 192.168.2.1 استفاده کنید .

رمز عبور admin میباشد .

در صورتی که میخواهید از ATA در شبکه خود استفاده نمایید جهت فعال کردن پورت Wan ، در سربرگ Basic

Settings طبق تصویر فوق عمل نمایید :

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)
NAT TCP timeout: 3600 (range: 0 - 3600, default is 3600)
NAT UDP timeout: 300 (range: 0 - 3600, default is 300)
Uplink bandwidth: Disabled
Downlink bandwidth: Disabled

Enable UPnP support: No Yes
Reply to ICMP on WAN port: No Yes (Unit will not respond to PING from WAN side if set to No)
WAN side HTTP/Telnet access: No Yes (WAN side access will be rejected if set to No)

White list for WAN side:
Black list for WAN side:

Cloned WAN MAC Addr: [][][][][][] (in hex format)

Enable LAN DHCP: No Yes

LAN DHCP Base IP: 192.168.2.1 (base IP for the LAN port, default is 192.168.2.1)
LAN DHCP Start IP: 100 (default is 100)
LAN DHCP End IP: 199 (default is 199)
LAN Subnet Mask: 255.255.255.0 (default is 255.255.255.0)
DHCP IP Lease Time: 120 (in units of hours, default is 120 hours or 5 days)
DMZ IP:

توجه: لازم به ذکر است سناریو انتقال خطوط تلفن از نقطه مبدا به نقطه مقصد از طریق گتوی FXO در مبدا و گیتوی FXS در مقصد برقرار میگردد که در این سناریو از ۲ گیتوی مدل ۵۰۳ جهت پیاده سازی استفاده میشود.

گیتوی FXO

Grandstream Device Configuration

STATUS
BASIC SETTINGS
ADVANCED SETTINGS
FXS PORT
FXO PORT

Account Active: No Yes

Primary SIP Server: IP GATEWAY FXS (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)

Backup Outbound Proxy: (e.g., proxy.myprovider.com, or IP address, if any)

Prefer Primary Outbound Proxy: No Yes (yes - will reregister via Primary Outbound Proxy if registration expires)

SIP Transport: UDP TCP TLS (default is UDP)

NAT Traversal: No Keep-Alive STUN UPnP

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

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

Authenticate Password: (purposely not displayed for security protection)

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

DNS Mode: A Record SRV NAPTR/SRV Use Configured IP

DNS SRV use Registered IP: No Yes

Primary IP:

Backup IP1:

Backup IP2:

Tel URI: Disabled

SIP Registration: No Yes

Unregister On Reboot: No Yes

Outgoing Call without Registration: No Yes

Enable PSTN Disconnect Tone Detection: No Yes (Default No)

(If set to yes, the following tone is used as the disconnect signal)

PSTN Disconnect Tone:

(Syntax: f1=freq@vol, f2=freq@vol, c=on1/off1-on2/off2-on3/off3;)
 (Allowed Range: freq = 0 to 4000Hz; vol = -40 to -24dBm)
 (Default: Busy Tone: f1=480@-32,f2=620@-32,c=500/500;)

Enable Polarity Reversal: No Yes (Default No. Check with your PSTN carrier before setting to Yes)

AC Termination Model Country-based Impedance-based (Default Country-based)

Country-based:

Impedance-based:

Number of Rings: (1-50. Default 4)

(Number of rings for a PSTN incoming call before FXO port answers to accept VoIP number)

PSTN Ring Thru FXS: No Yes (Default Yes)

(If set to yes, all incoming PSTN calls will ring the FXS port after the Ring Thru Delay)

PSTN Ring Thru Delay (sec): (1-10 seconds. Default 4 seconds)

PSTN Ring Timeout (sec): (2-10 seconds. Default 6 seconds)
 (Used to detect PSTN hangup when FXO port is not answered)

PSTN Idle Wait Timeout between Outgoing Calls: (0-10 seconds. Default 4 seconds)

Channel Dialing

DTMF Digit Length (ms): (40-127 milliseconds, Default 100 milliseconds)

DTMF Dial Pause (ms): (40-127 milliseconds, Default 100 milliseconds)

First Digit Timeout (sec): (1-20 seconds. Default 10 seconds)

Inter-Digit Timeout (sec): (1-15 seconds. Default 4 seconds)

Wait for Dial-Tone: No Yes (Default Yes - dial upon dial-tone)

Stage Method (1/2): (Default 2 - 2 stage dialing)

Min Delay Before Dial PSTN Number: (default 500ms, range 50 ~ 65000ms)

PSTN: number)

Unconditional Call Forward to VOIP: **Any Number**

User ID: **Any Number**

Sip Server:

Sip Destination Port:

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گیتوی FXS

Grandstream Device Configuration

STATUS
BASIC SETTINGS
ADVANCED SETTINGS
FXS PORT
FXO PORT

Account 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)

Backup Outbound Proxy: (e.g., proxy.myprovider.com, or IP address, if any)

Prefer Primary Outbound Proxy: No Yes (yes - will reregister via Primary Outbound Proxy if registration expires)

SIP Transport: UDP TCP TLS (default is UDP)

NAT Traversal: No Keep-Alive STUN UPnP

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

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

Authenticate Password: (purposely not displayed for security protection)

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

DNS Mode: A Record SRV NAPTR/SRV Use Configured IP

DNS SRV use Registered IP: No Yes

Primary IP:

Backup IP1:

Backup IP2:

Tel URI:

SIP Registration: No Yes

Unregister On Reboot: No Yes

Outgoing Call without Registration: No Yes