انتقال خط گیتوی گرند استریم HT-503

تهیه شده توسط

سینا پروانه

www.Kavatelecom.com/blog

www.VoIPshop.ir

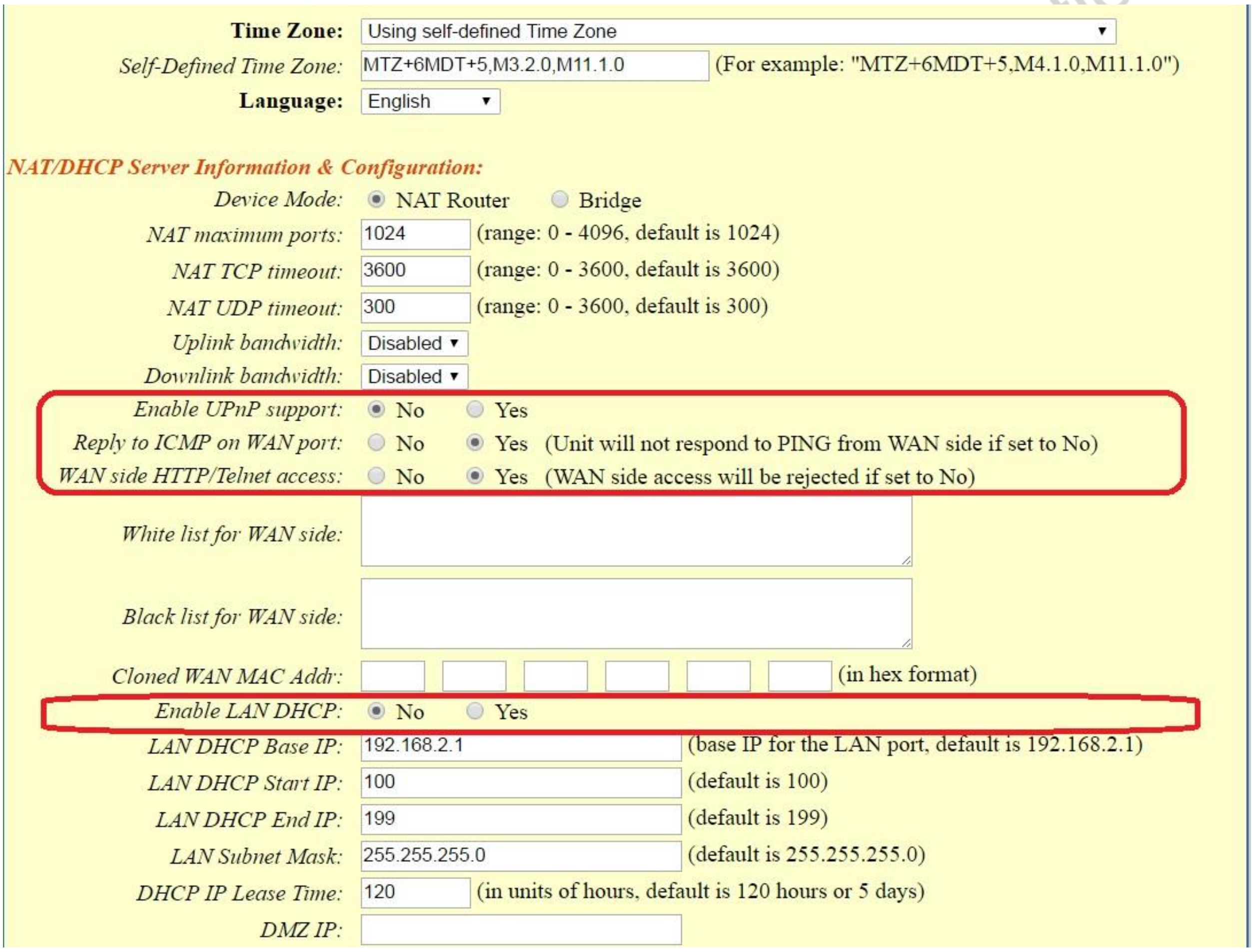
جهت دسترسی به پنل وب این دستگاه طبق مراحل فوق عمل نمایید :

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

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

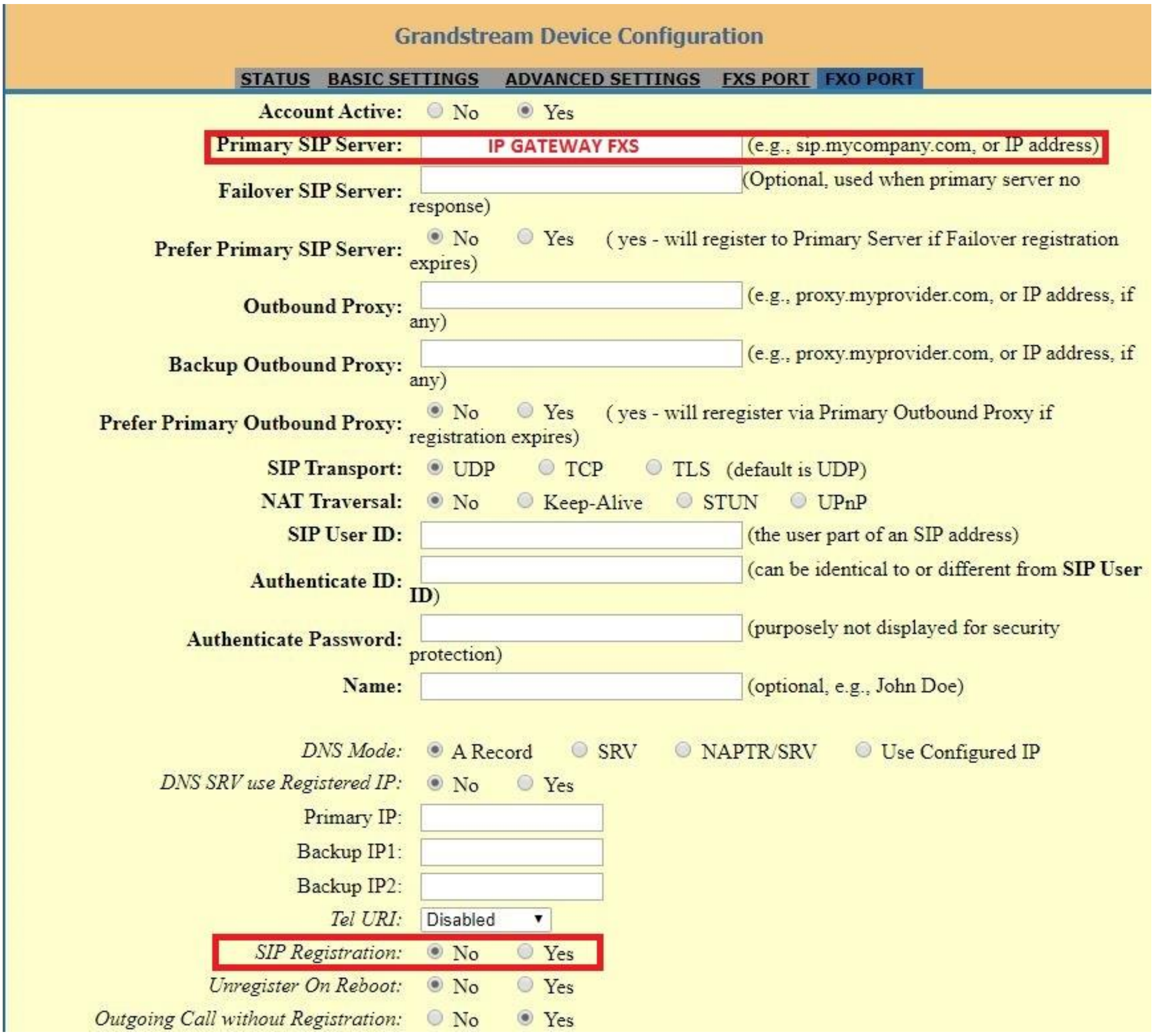
رمز عبور admin میباشد .

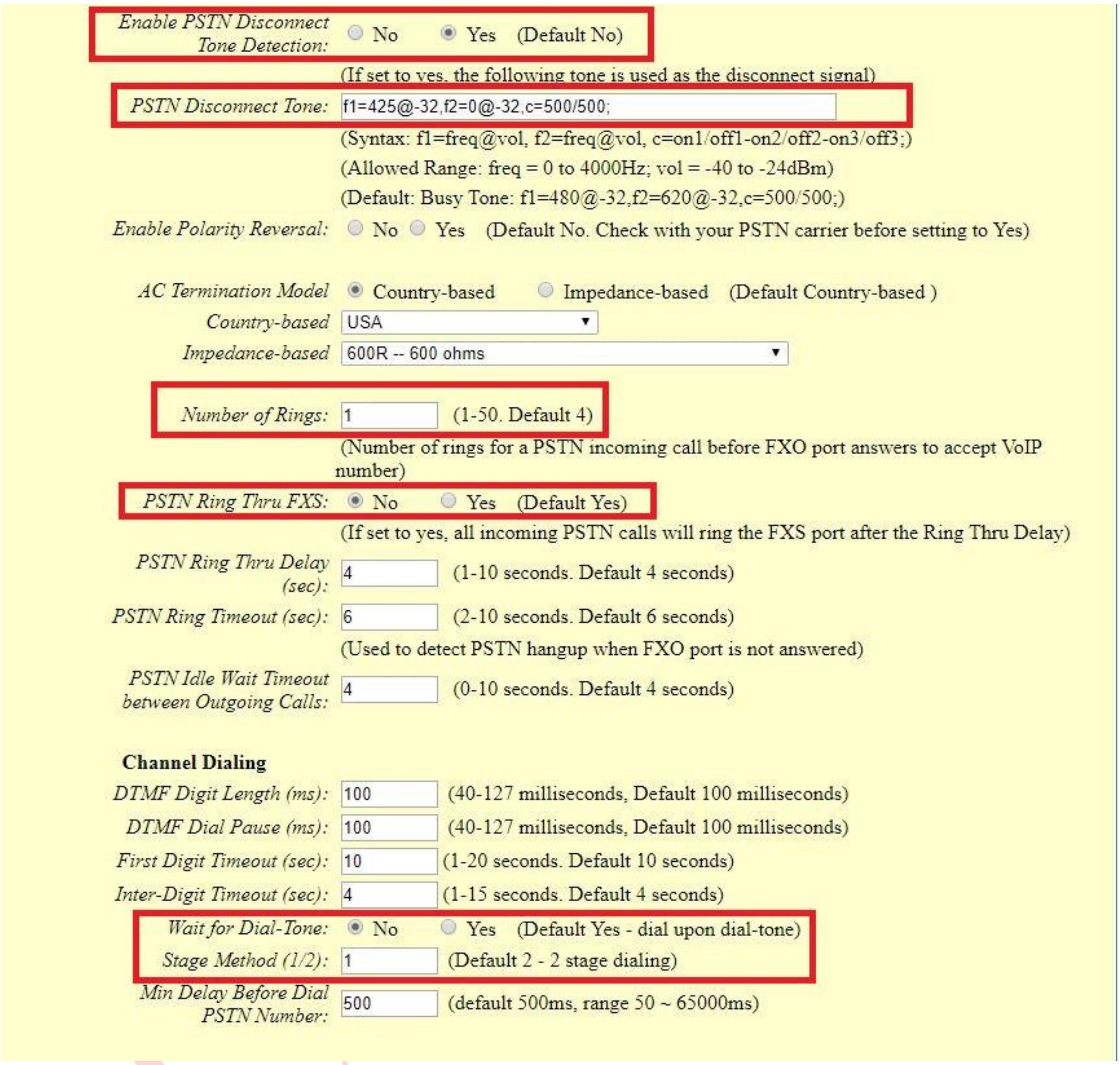
در صورتی که میخواهید از ATA در شبکه خود استفاده نمایید جهت فعال کردن پورت Wan ، در سربرگ Basic Settings طبق تصویر فوق عمل نمایید :



توجه : لازم به ذکر است سناریو انتقال خطوت 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





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: f1=425@-32,f2=0@-32,c=500/500;

(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: USA

Impedance-based: 600R -- 600 ohms

Number of Rings: 1 (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): 4 (1-10 seconds. Default 4 seconds)

PSTN Ring Timeout (sec): 6 (2-10 seconds. Default 6 seconds)

(Used to detect PSTN hangup when FXO port is not answered)

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

Channel Dialing

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

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

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

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

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

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

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

PSTN: number)

Unconditional Call Forward to VOIP: User ID 200 Any Number @ Sip Server IP GATEWAY FXO : Sip Destination Port 5060

Update Apply Cancel Reboot

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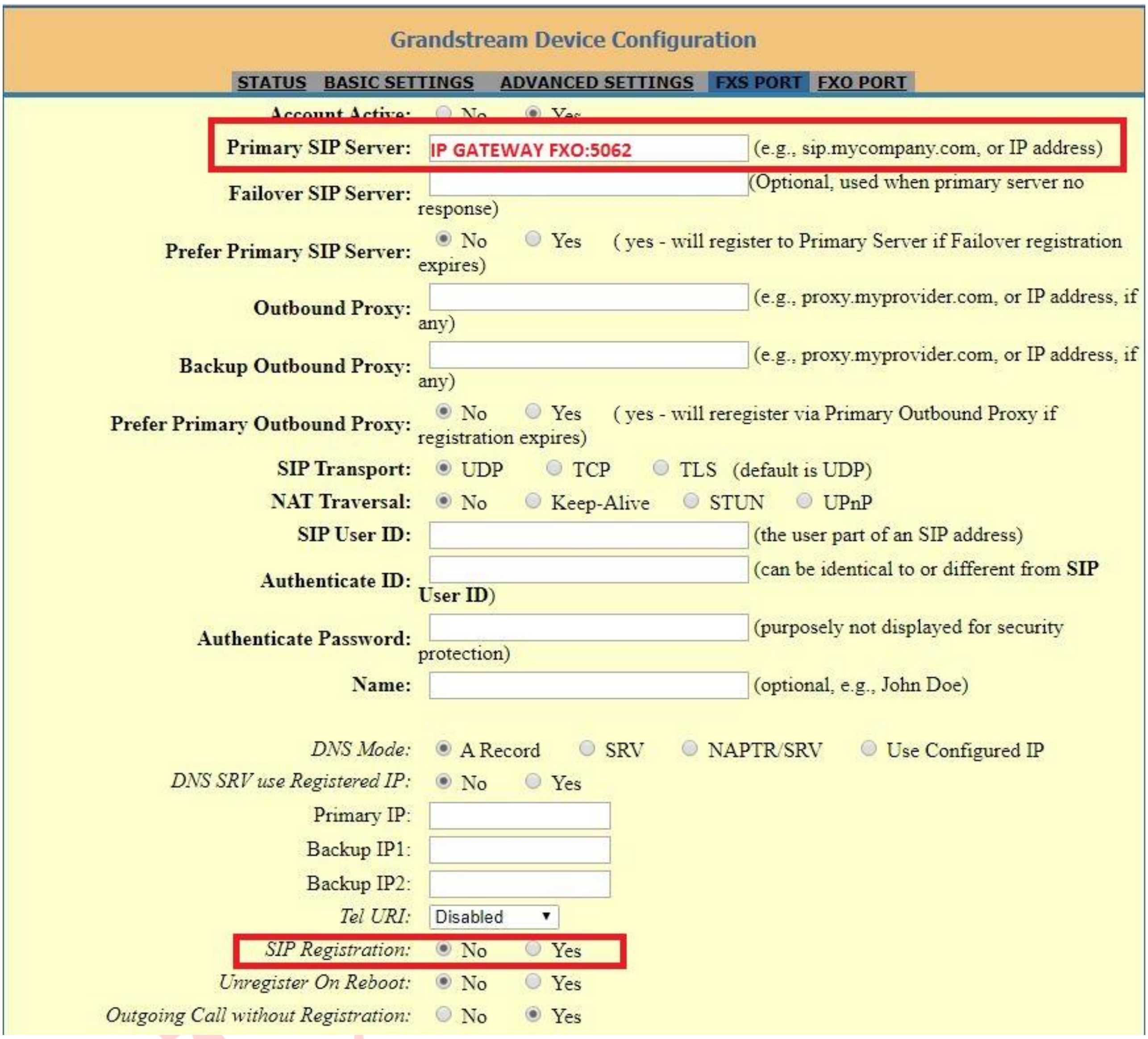
گیتوی FXS


Grandstream Device Configuration
STATUS
BASIC SETTINGS
ADVANCED SETTINGS
FXS PORT
FXO PORT
Account Active: No Yes
Primary SIP Server: IP GATEWAY FXO:5062 (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