OT-ATA200SP+
1 FXS + 1 PSTN SIP/IAX ATA User Manual

OverTek

CONTENT

1 OT-ATA200SP+Features

1.1 Appearance

PWR: power connection status
SYS : server registration status . registered: on , gliting , not-registered , off
WAN: WAN port connectiong , contected , on , offline : off
LAN: LAN port connectiong , contected , on , offline : off
PSTN: light on when PSTN call come in/out
VoIP: light on when VoIP call come in/out , The default status when pick up the call is VoIP call out

1.2 Interface

Power: Output Power:12VDC,500mA.
Port: RJ11 port. Connect to handset or the Lifeline accessory.
WAN: RJ45 port.
LAN: RJ45 port.

1.3 Electricity characteristic

- **Speciality of electric**: output the 12V 500mA DC
- **The network connects**:2 RJ45 connect, a WAN, a LAN
- **FXS**: 2 port

1.4 Software

- Support two sip servers running at the same time.
- Back-up sip server support.
- NAT, Firewall.
- DHCP client and server.
- Support PPPoE, (used for ADSL, cable modem connecting).
- Support major G7.xxx CODEC.
- VAD,CNG.
- G.165 compliant 16ms echo cancellation
- Tone generation and Local DTMF re-generation according with ITU-T
- E.164 dial plan and customized dial rules
- Support Lifeline.
- Hotline.
- Speed Dial
- Call Forward, Call Transfer, 3-way conference calls
- Caller ID display
- DND(Do Not Disturb),Black List,Limit List
- Upgrade firmware through FTP or HTTP.
- Web management.
- Reverse polarity
- Telnet remote management.
- Voice prompt
- adjustable user password and super password

1.5 Standard and Protocols

- IEEE 802.3 /802.3 u 10 Base T / 100Base TX
- PPPoE: PPP Protocol over Ethernet
- DHCP Client and Server: Dynamic Host Configuration Protocol
- G.711 u/a; G729 audio Codec
- SIP RFC3261, RFC 2543
- IAX2
- TCP/IP: Internet transfer and control protocol
- RTP: Real-time Transport Protocol
- RTCP: Real-time Control Protocol
- VAD/CNG save bandwidth
- Telnet: Internet's remote login protocol
- DNS: Domain Name Server
- TFTP: Trivial File Transfer Protocol
- HTTP: Hyper Text Transfer protocol
- FTP: File Transfer protocol
- RFC3362: T.38 protocol

1.6 Compliant Standards

- CE: EN55024,EN55022
- FCC part15
- comply with ROHS in EU
- comply with ROHS in China

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Explanation:

The letter “e” is the first letter of “environment: and “electronic”, The rim is a round with two arrow , stands for recycle. The number 20 stands for the years of environment protection. Please note the years of environment protection is not discarding year nor usage life

1.7 Operating requirement

- Operation temperature: 0 to 40° C (32° to 104° F)
- Storage temperature: -30° to 65° C (-22° to 149° F)
- Humidity: 10 to 90% no dew

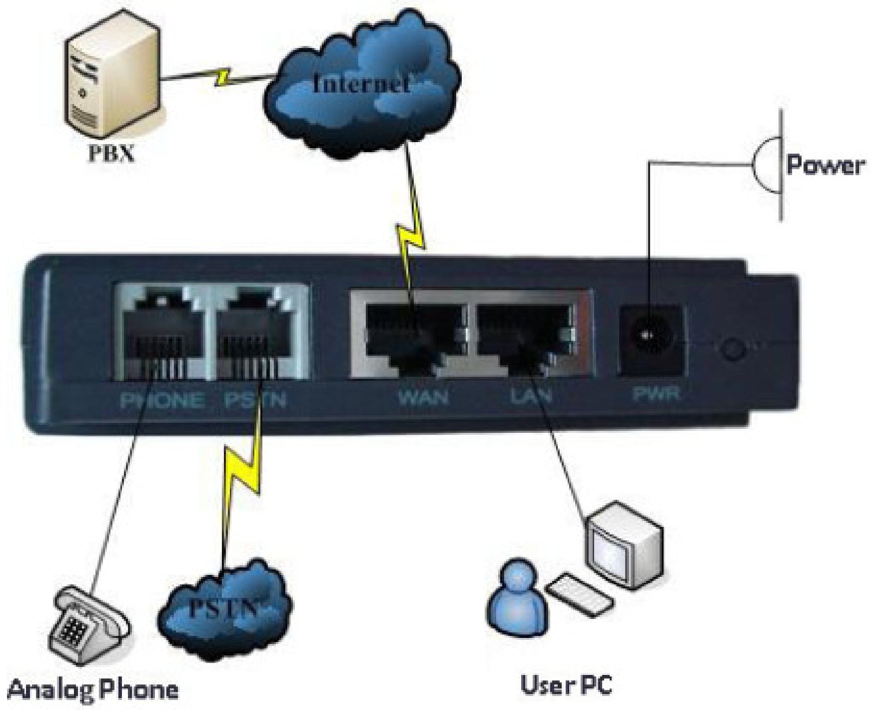
1.8 Package

- Size 128 x 85 x 30 mm
- Packing List
  - ✓ OT-ATA200SP+ ATA
  - ✓ Power adaptor (12v, 500mA)
  - ✓ Manual CD

1.9 Installation

Use Ethernet cable to connect OT-ATA200SP+'s LAN port and your computer. Set your computer's ip to the network 192.168.10.x or using dynamic obtain IP. Open your web browser and key in 192.168.10.1. Then you will see the logon pageof OT-ATA200SP+, the default username and password is **admin/admin** for administrator and **guest/guest** for guest.

Set up page for VoIP use only:



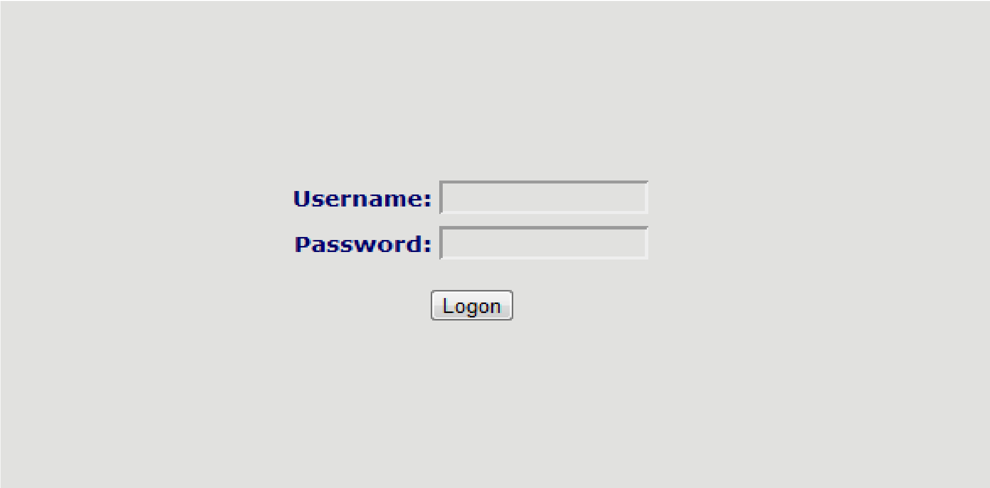
2 Web Configuration

2.1 Access Web setting page

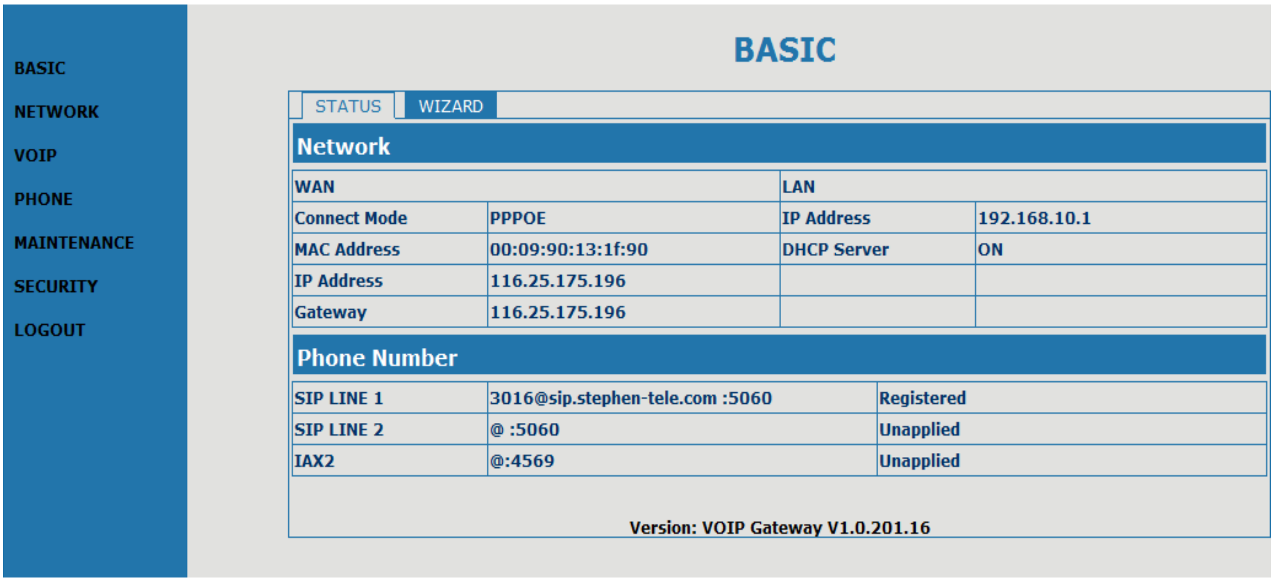
Enter OT-ATA200SP+ IP address in the web browser and press ENTER to go to the log on page, and key in the username and password to access OT-ATA200SP+ setting page.

Default username and password is:

Administrator:	Username: **admin**	password: **admin**
User:	Username: **guest**	Username: **guest**



2.2 Current state



This page shows OT-ATA200SP+'s running state.

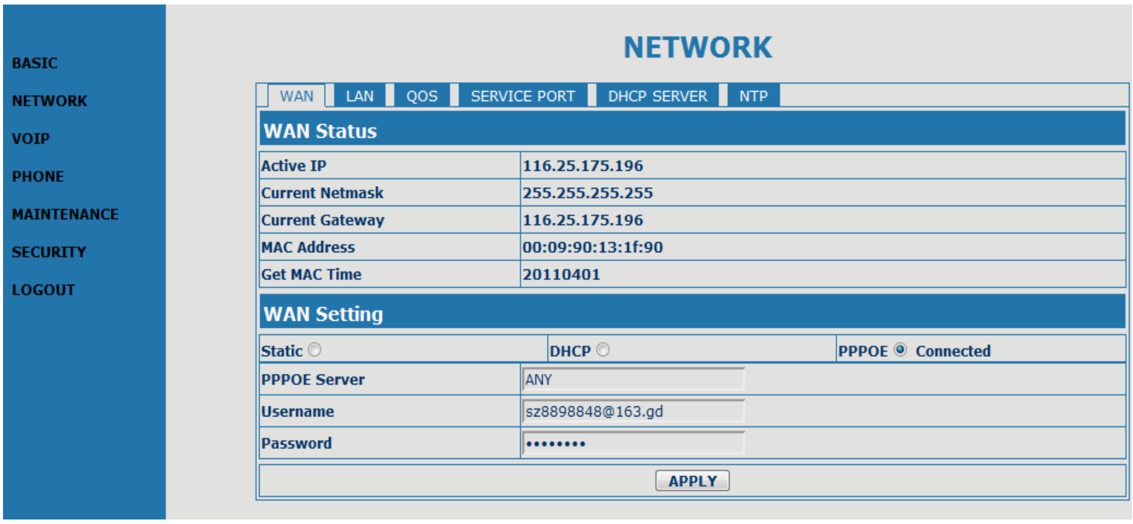
Network： shows the WAN and LAN port connecting state and current settings.

VoIP： show the default protocol, the working state of SIP and IAX2, you can see whether OT-ATA200SP+ has registered the public sip server and IAX2 server.

Phone Number shows the public sip server, the private sip server and the IAX2 server phone numbers.

2.3 Network

2.3.1 Wan Config



WAN port network setting page.

Support static IP, dynamic obtain IP and PPPoE.

Configure Static IP:
----Enable Static;
----Set OT-ATA200SP+'s IP address in the IP Address;
----Set net mask in the Net mask field;
----Set router IP address in the Gateway;
----DNS Domain:
----Set local DNS server in the Preferred DNS and the Alternate DNS

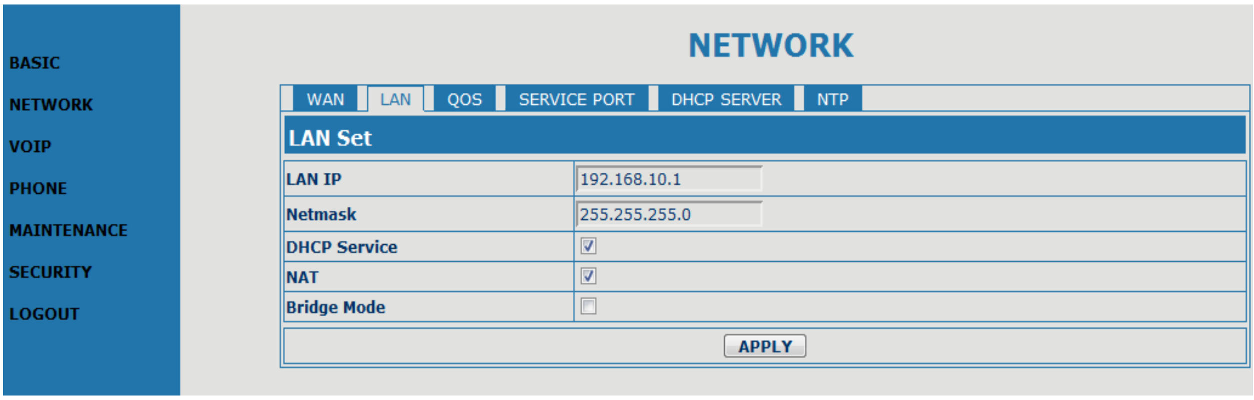
Configure to dynamic obtain IP
----Enable DHCP;
If there is DHCP server in your local network, OT-ATA200SP+ will automatically obtain WAN port network information from your DHCP server.

Configure PPPoE:
----Enable PPPoE
----PPPoE server: Enter "ANY" if no specified from your ITSP.
----Enter PPPoE username and pin in the username and password.

OT-ATA200SP+ will automatically obtain WAN port network information from your ITSP if PPPoE setting and the setup are correct.

Notice: If user accesses the gateway through WAN port. He should use the new IP address to access the gateway when the WAN port address was changed.

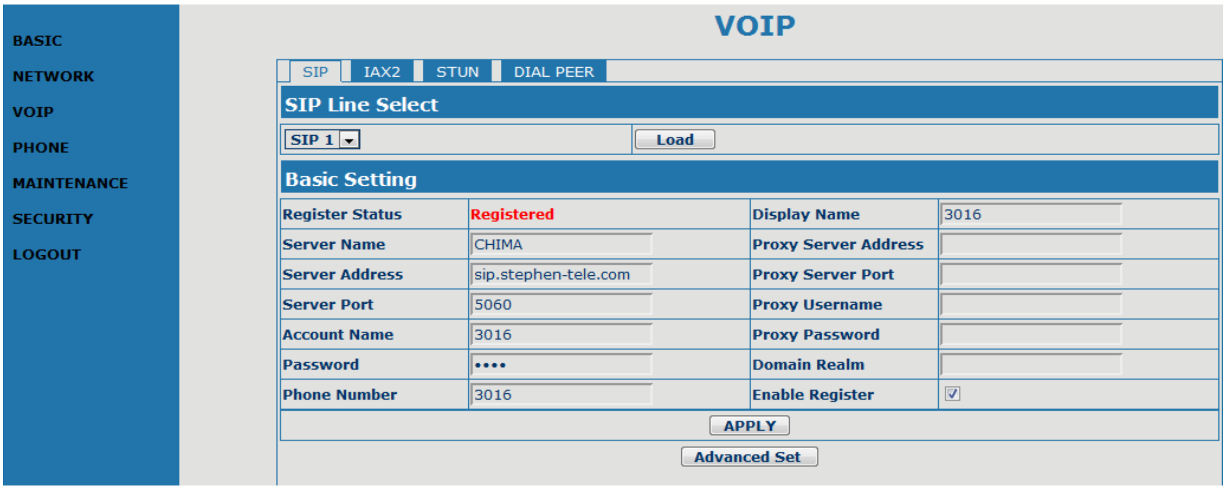
2.3.2 LAN Config



Bridge Mode: Enable this option to switch to bridge mode. Gateway won't assign IP for its LAN port in bridge mode and its LAN and WAN port will be in the same network. (This setting won't take effect unless you save the config and reboot the device)
IP Netmask: Set the IP and Netmask for the LAN
DHCP Server: Enable DHCP service in LAN port
NAT: Enable NAT.

2.4 VOIP

2.4.1 SIP Config

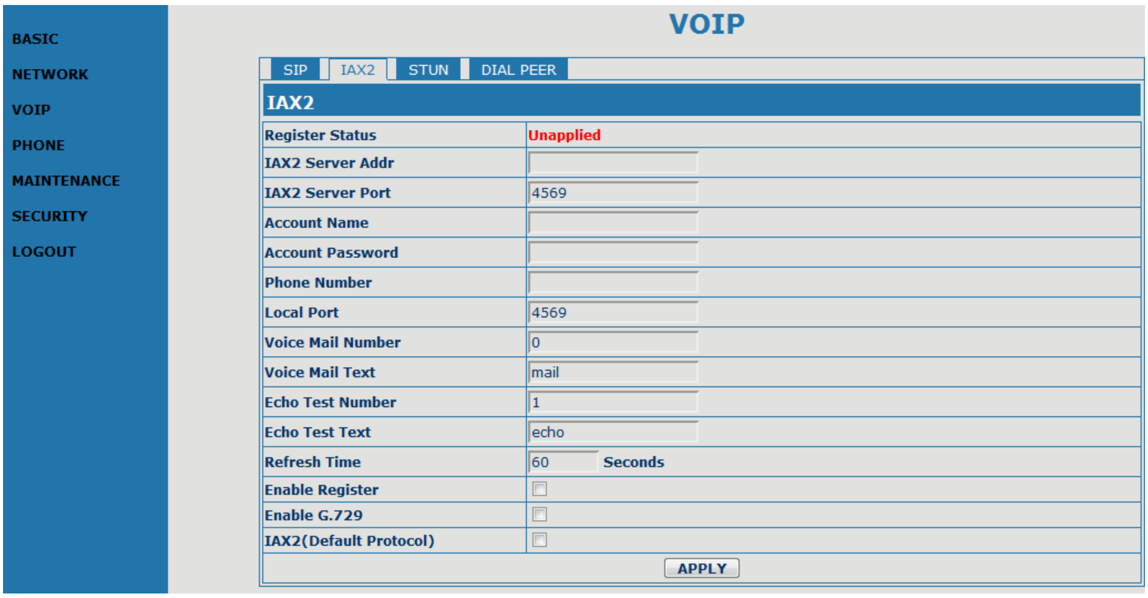


Setting page of public SIP server:

Register Server Addr: Register address of public SIP server
Register Server Port: Register port of public SIP server
Register Username: Username of your SIP account (Always the same as the phone number)
Register Password: Password of your SIP account.
Proxy Server Addr: IP address of proxy SIP server (SIP provider always use the same IP for register server and proxy server, in this case you don't need to configure the proxy server information.)
Proxy Server Port: Signal port of SIP proxy
Proxy Username: proxy server username
Proxy Password: proxy server password

Domain Realm: SIP domain, enter the sip domain if any, otherwise OT-ATA200SP+ will use the proxy server address as sip domain.
Local SIP port: Local SIP register port, default 5060
Phone Number: Phone number of your SIP account
Register Expire Time: register expire time, default is 600 seconds. OT-ATA200SP+ will auto configure this expire time to the server recommended setting if it is different from the SIP server.
Detect Interval Time: Co-work with the Auto Detect Server, if Auto Detect Server is enable, OT-ATA200SP+ will periodically detect if the SIP server is available according this setting.
RFC Protocol Edition: Current OT-ATA200SP+ SIP versions. Set to RFC 2543 if the gate need to communicate to devices (such as CISCO5300) using the SIP 1.0. Default is RFC 3261.
Enable Register: Enable/Disable SIP register. OT-ATA200SP+ won't sent register info to SIP server
DTMF Mode: DTMF signal sending mode: support RFC2833, DTMF_RELAY (inband audio) and SIP info
Auto Detect server: co-work with Server Auto Swap and Detect Interval Time. Enable this option, OT-ATA200SP+ will periodically detect whether the public SIP server is available, if the server is unavailable, the OT-ATA200SP+ will switch to the back-up SIP sever, and continue detecting the public sip server. OT-ATA200SP+ will switch back to the primary SIP server if the **server is available again**.
Server Auto Swap: Please refer to Auto Detect server for detail.
Enable Via rport: config the supporting for RFC 3581
SIP(Default Protocol): Setting for the default protocol of SIP

2.4.2 Iax2 Config



Setting page of public IAX server:
IAX Server Addr: Register address of public IAX server
IAX Server Port: Register port of public IAX server, default port is 4569

Account Name: Username of your SIP account (Always the same as the phone number)
Account Password:Password of your IAX account.
Local port: Signal port of local, default port is 4569

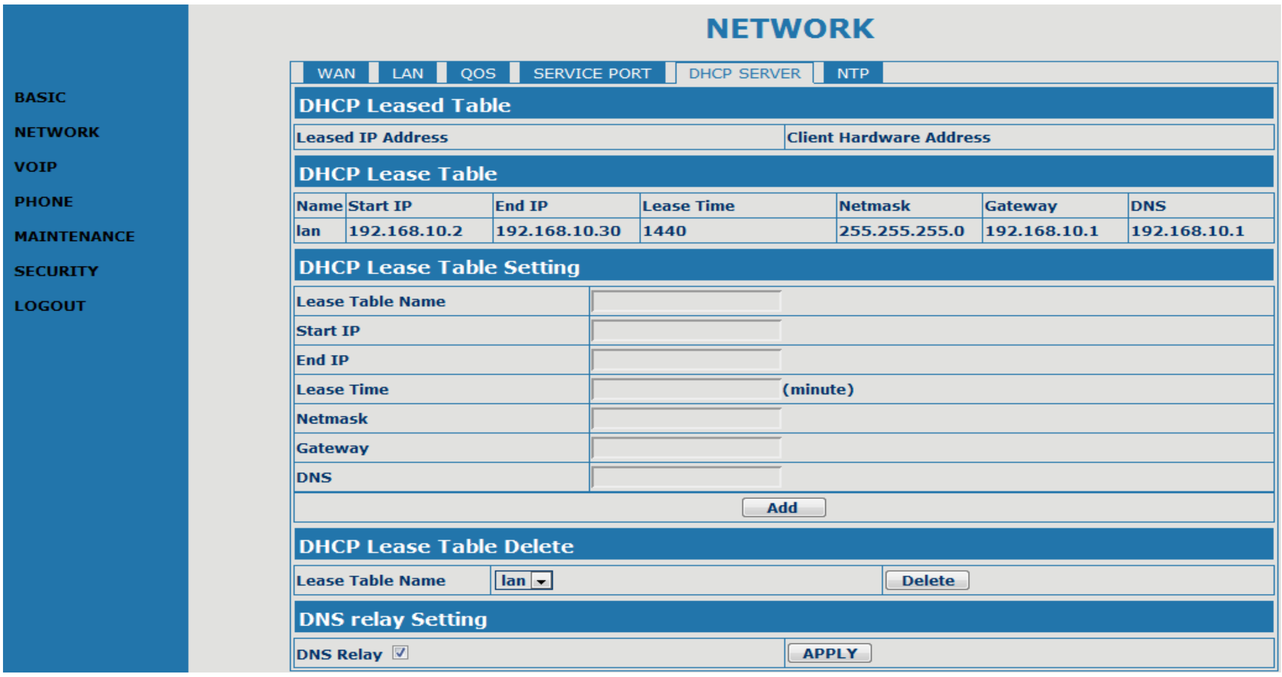
Phone Number: Phone number of your IAX account

Voice mail number: If the IAX support voice mail, but your username of the voice mail is letters which you can not input with the ATA , then you use the number to stand for your username
Voice mail text: if IAX support voice mail, config the domain name of your mail box here.

Echo test number: If the platform support echo test , and the number is test form , the config the test number to replace the text format The echo test is to test the woring status of terminals and platform
Echo test text: echo test number in text format
Refresh time: IAX refresh time
Enable Register: enable or disable register
IAX(Default Protocol): Set IAX 2 as the default protocol , if not the system will choose SIP as default
Enable G.729: Using G.729 speech coding mandatory consultations

2.5 Advance

2.5.1 DHCP Server



DHCP server manage page.
User may trace and modify DHCP server information in this page.
Update Mode: Using DHCP updated model ,None expressed are not updated, Update firmware update firmware is used to DHCP. Update file is used to configure DHCP updated configuration files.

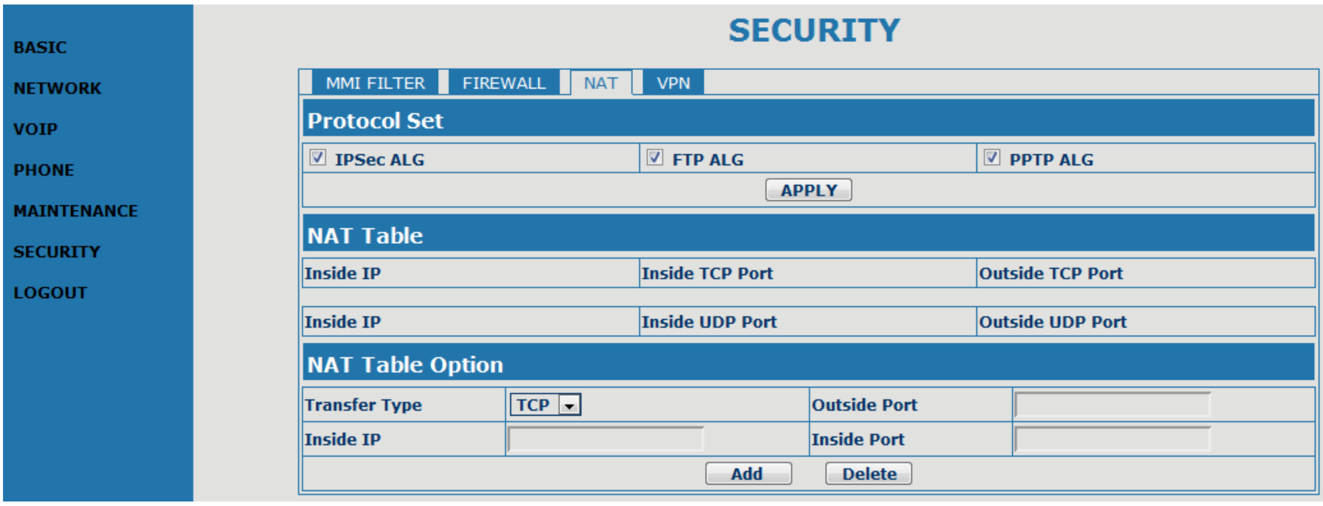
Tftp Server: Addresses using TFTP server upgrade.

DNS Relay: enable DNS relay function.
User may use below setting to add a new lease table.
Lease Table Name: Lease table name.
Lease Time: DHCP server lease time.
Start IP: Start IP of lease table.
End IP: End IP of lease table. Network device connecting to the OT-ATA200SP+ LAN port can dynamic obtain the

IP in the range between start IP and end IP.
Netmask: Netmask of lease table.
Gateway: Default gateway of lease table
DNS: default DNS server of lease table.

Notice: This setting won't take effect unless you save the config and reboot the device

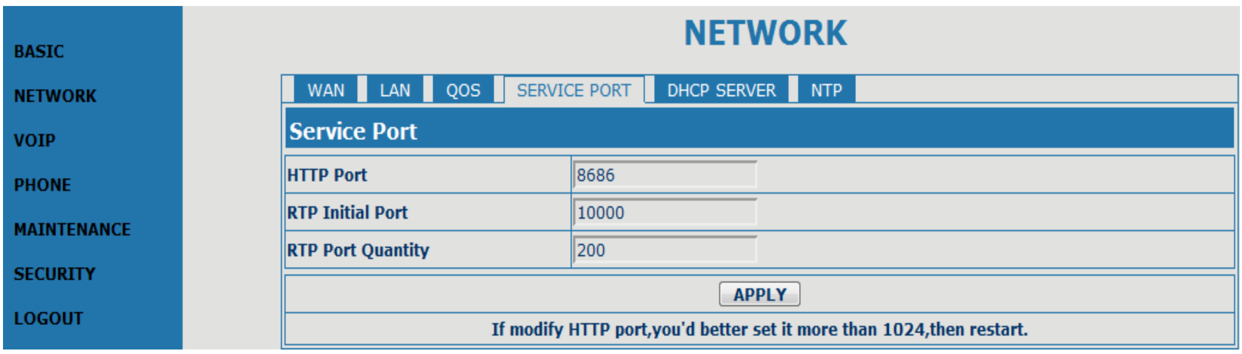
2.5.2 NAT



Advance NAT setting. Maximum 10 items for TCP and UDP port mapping.

IPSec ALG: Enable/Disable IPSec ALG;
FTP ALG: Enable/Disable FTP ALG;
PPTP ALG: Enable/Disable PPTP ALG;
Transfer Type: Transfer type using port mapping.
Inside IP: LAN device IP for port mapping.
Inside Port: LAN device port for port mapping.
Outside Port: WAN port for port mapping.
Click **Add** to add new port mapping item and **Delete** to delete current port mapping item.

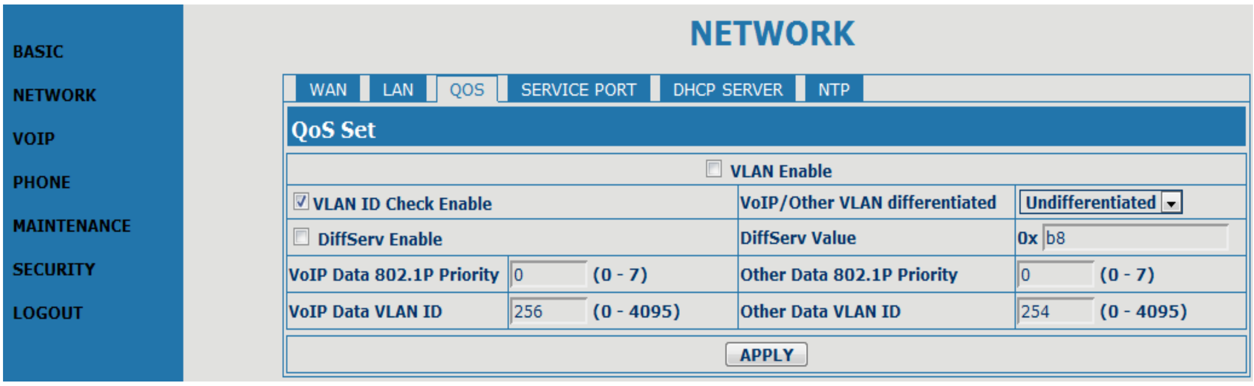
2.5.3 Net Service



HTTP Port: configure HTTP transfer port, default is 80.User may change this port to enhance system's security. When this
port is changed, please use http://xxx.xxx.xxx.xxx:xxxx/ to reconnect.
Telnet Port: configure telnet transfer port, default is 23.

RTP Initial Port: RTP initial port.
RTP Port Quantity: Maximum RTP port quantity, default is 200
Notice:
Settings in this page won't take effect unless save and reboot the device.
If you need to change telnet port or HTTP port, please use the port greater than 1024, because ports under 1024 is system remain ports.
HTTP service if HTTP is set to 0.

2.5.4 QoS settings



OT-ATA200SP+ implement QoS based on 802.1p, The QoS is used to mark the network communication priority in the data link/MAC sub-layer. OT-ATA200SP+ will sorted the packets using the QoS and sends it to the destination.

1. Voice 802.1p Priority --- Configure the priority of the voice packets in 802.1p protocol.
2. VLAN Enable --- Disable/Enable VLAN function
3. Voice VLAN ID --- configure the Voice/signaling VLAN ID
4. DiffServ Enable --- Disable/Enable Diffserv service

5. DiffServ Value --- Configure Diffserv parameter. The value range : value range ：

0x28,0x30,0x38,0x48,0x50,0x58,0x68,0x70,0x78,0x88,0x90,0x98,0xb8.default is 0xb8 ,oxb8 stands for best fast transmission; 28-30 is guaThrantee for the transmission priority for the 1st rank , 48-58 is guarantee for the transmission priority for the 2nd rank, 68-78 is guarantee for the transmission priority for the 3rd rank, 88-98 is guarantee for the transmission priority for the 4th rank.

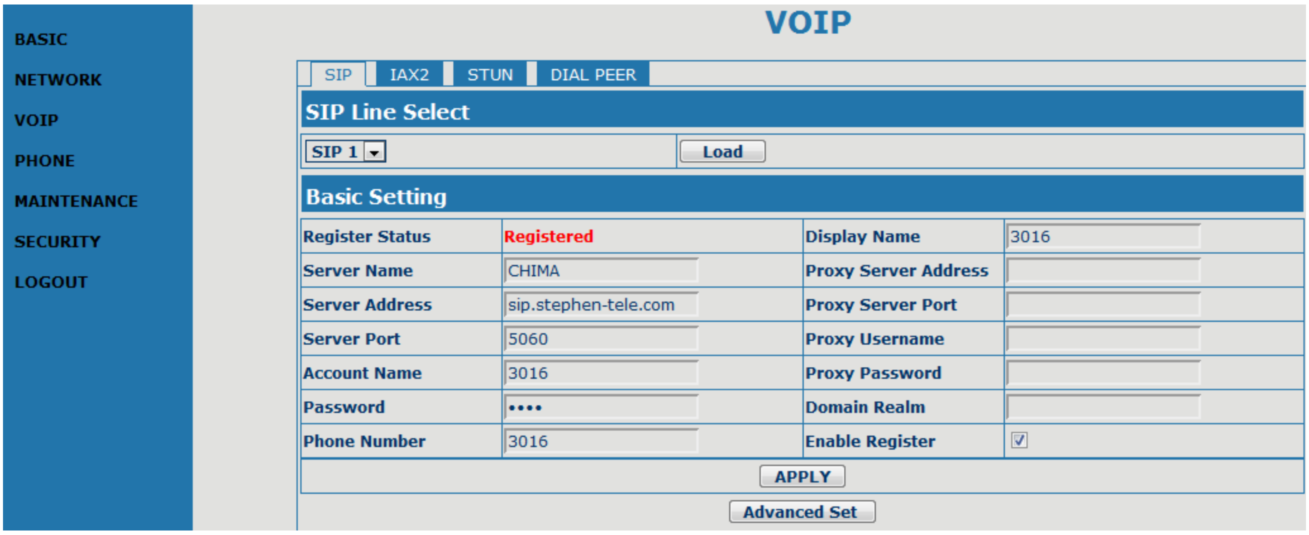
6. Data VLAN ID--- Assign VLAN id for data stream.

7. Data 802.1P Priority --- Configure the priority of the data packets (non-voice/signaling data) in 802.1p protocol.

8. Data/Voice DiffServ differentiated --- undifferentiated for Date and voice VLAN is not distinction VLAN tag, Tag differentiated for Date and Voice VLAN is distinction VLAN tag, Date untagged for Date VLAN is distinction VLAN tag

Please refer to VLAN implement for detail.

2.5.5 Advance SIP settings



This page is used to set the private sip server, stun server, and back up sip server information.
STUN Server setting:
STUN Server Addr: configure stun server address;
STUN Server Port: configure stun server port default 3478
STUN Effect Time: stun detect NAT type circle, unit: minute.
Enable SIP STUN: enable/disable stun.
Please refer to sip conf for the setting for how to set the public alter server.
Private Server Type: The particular Private service system supplier carries out the sign and speeches to encrypt, default is common. User can register two sip servers: public sip server and private sip server. these two sip servers are independent from each other and running in the same time.

For how to configure private sip server. Please refer to sip configuration.

2.5.6 Digital Map

Digit map is a set of rules to determine when the user has finished dialing.
OT-ATA200SP+ support below digital map:
Digital Map is based on some rules to judge when user end their dialing and send the number to the server.
OT-ATA200SP+ support following digital map:

----End With “#”: Use # as the end of dialing.
----Fixed Length: When the length of the dialing match, the call will be sent.
----Timeout: Specify the timeout of the last dial digit. The call will be sent after timeout
----Prefix: User define digital map:

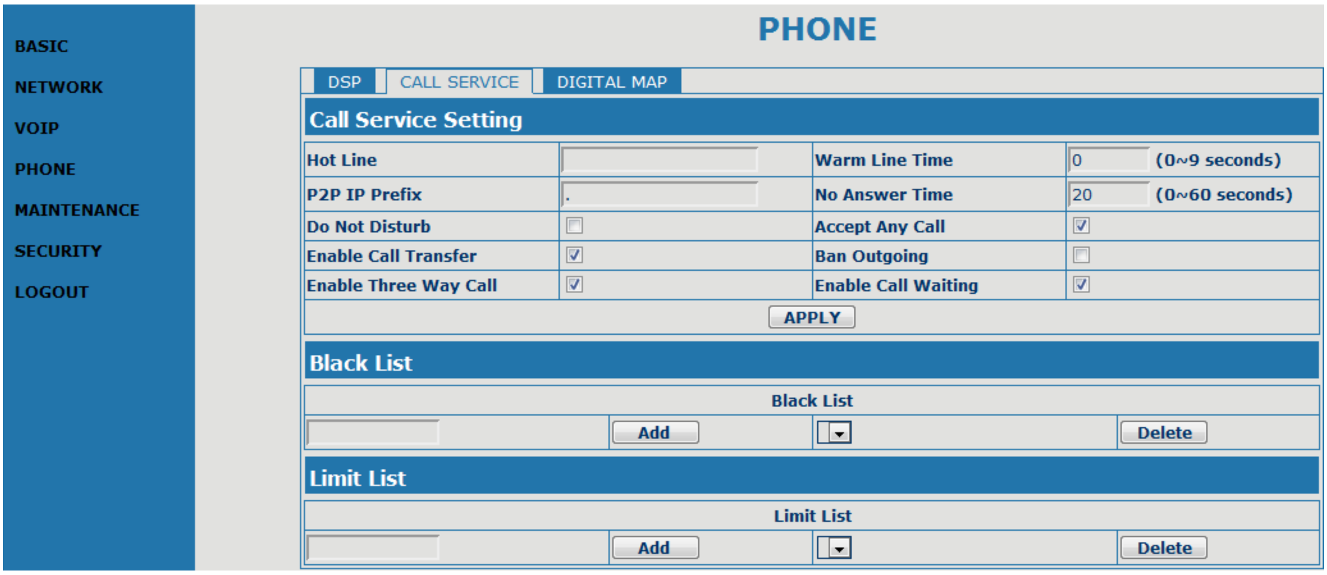
[ ] represents the range of digit, can be a range such as [1-4], or use comma such as [1,3,5], or use a list such as [234]

x represents any one digit between 0~9

Tn represents the last digit timeout. n represents the time from 0~9 second, it is necessary. Tn must be the last two
digit in the entry. If Tn is not included in the entry, we use T0 as default, it means system will sent the number immediately
if the number matches the entry.

Example:
[1-8]xxx All number from 1000 to 89999 will be sent immediately.
9xxxxxxx 8 digits numbers begin with 9 will be sent immediately.
911 Number 911 will be sent will be immediately
99xT4 3 digits numbers begin with 99 with be sent after four seconds.

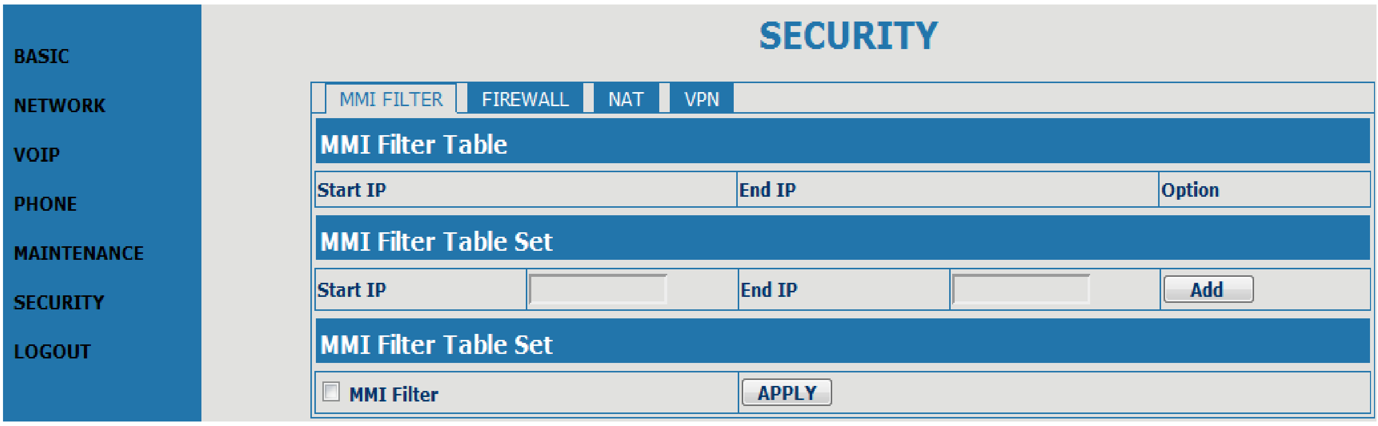
2.5.7 Call Service Settings



User configure the value add service such as hotline, call forward, call transfer, 3-way conference call .etc in this page
Hotline: configure hotline number. OT-ATA200SP+ immediately dials this number after hook-off if it is set.
Call Forward: Please refer to value add service for detail.
No Disturb: DND, do not disturb, enable this option to refuse any calls.
Ban Outgoing: Enable this to ban outgoing calls.
Enable Call Transfer: Please refer to value add service for detail.
Enable Three Way Call: Please refer to value add service for detail.
Enable Call Waiting: Enable/disable Call Waiting
Accept Any Call: If this option is disable, OT-ATA200SP+ refuse the incoming call when the called number is different from OT-ATA200SP+’s phone number.

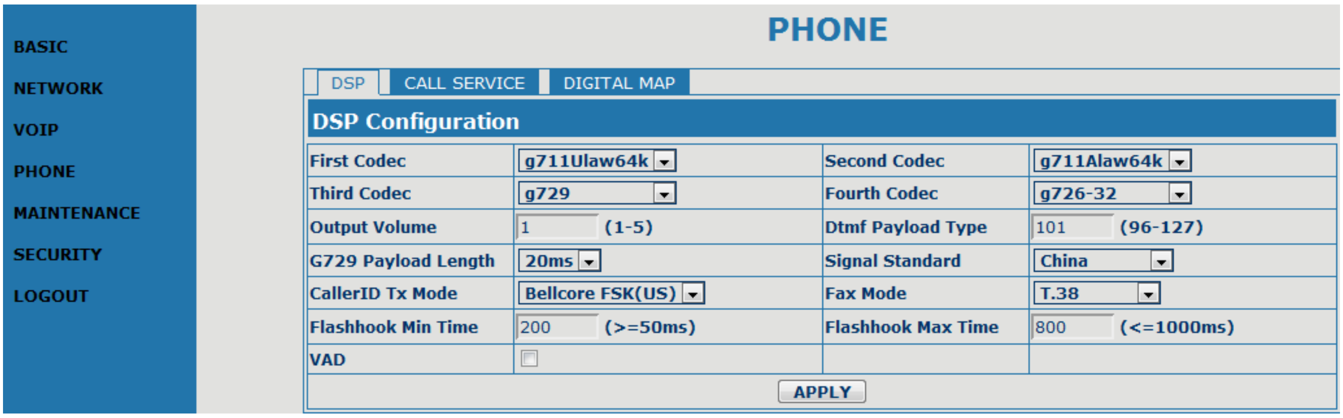
No Answer Time: no answer call forward time setting.
Black List: incoming call in these phone numbers will be refused.
Limit List: outgoing calls with these phone numbers will be refused

2.5.8 MMI Filter



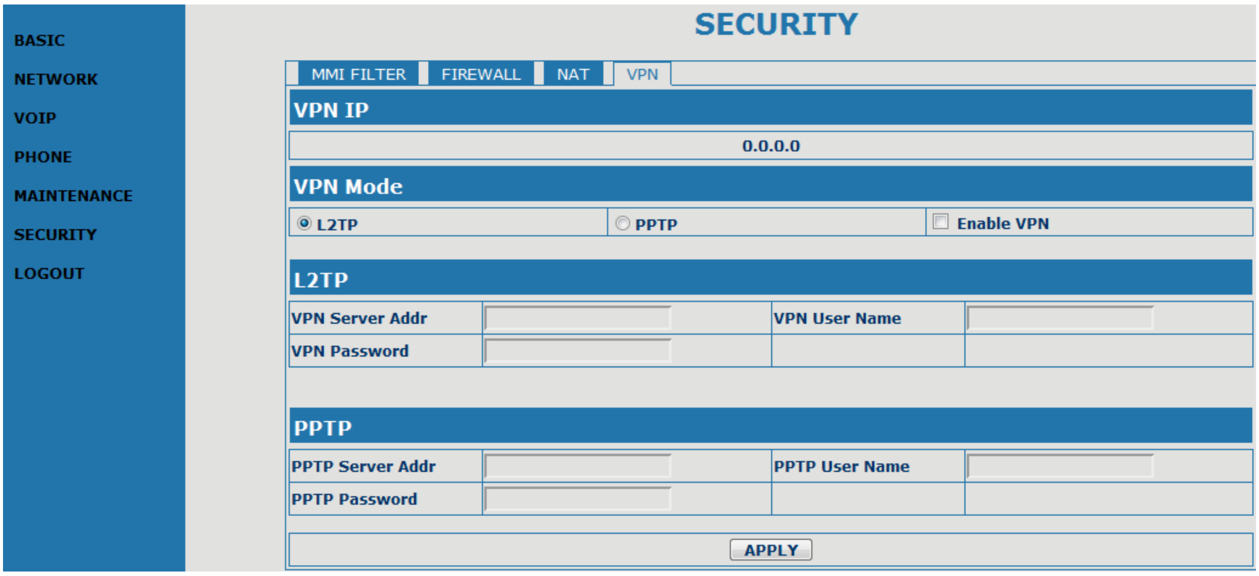
MMI filter is used to make access limit to OT-ATA200SP+ Gateway.
When MMI filter is enable. Only IP address within the start IP and end IP can access OT-ATA200SP+ gateway.

2.5.9 Audio Settings



CODEC: select the prefer CODEC; support ulaw, alaw and G729
Signal Standard: Support CHINA, Japan and USA standard
Input Volume: Handset in volume.
Output Volume: Handset out volume.
G729 Payload Length: G729 payload length
VAD: Enable/disable Voice Activity Detection
FAX Mode: select the FAX Mode

2.5.10 VPN



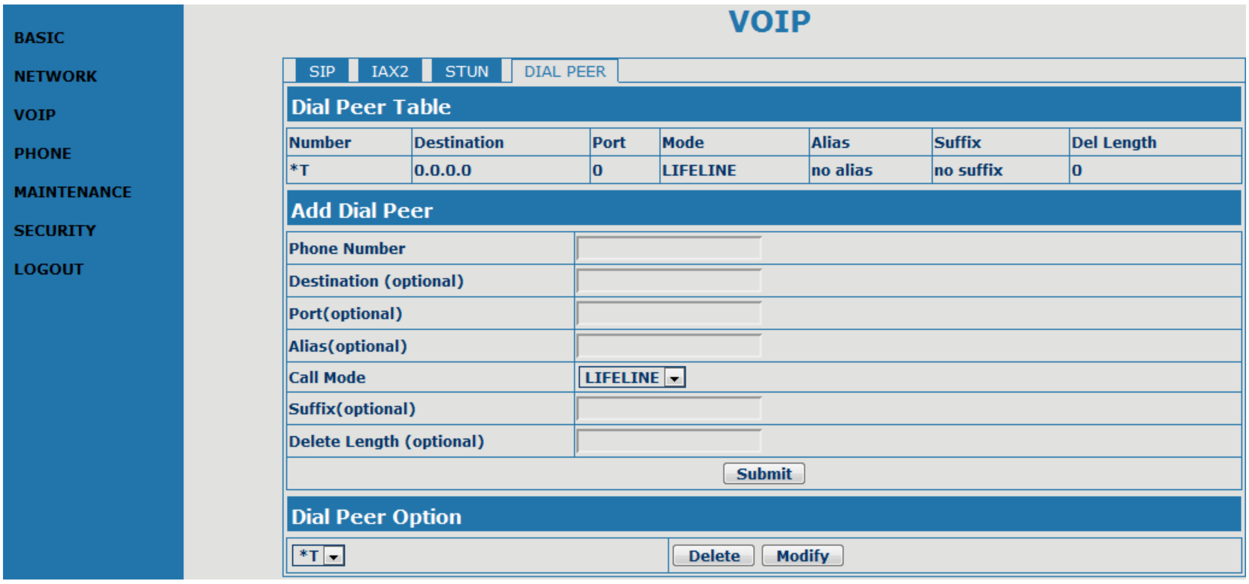
this page is VPN setting page , the Gateway support the VPN with UDP and L2TP protocol .The parameters is as below

VPN IP: After VPN registered successfully, VPN server will give an IP aggress to the terminal . If there is a IP address shown on terminal (except for 0.0.0.0) ,it means your VPN has registered UDP Tunnel

VPN Server Addr: register to the address of VPN server.
VPN Server Port: Register to the port of VPN server
Server Group ID: the group ID of UDP VPN
Server Area Code: the are code of VPN server
L2TP VPN Server Addr: register to the address of VPN server
VPN User Name: L2TP VPN username
VPN Password: L2TP VPN password

UDPTunnel: use the UDP to visit VPN
L2TP: use the L2TP to visit VPN
Enable VPN: Enable the VPN server, you must choose UDP or L2TP type in advance

2.6 Dial-Peer Settings



Please refer to "how to use dial rule?" for detail.

2.6.1 Config Manage

Save Config: save current settings.
Clear Config: restore to default settings.

Notice: clear config in admin mode, all settings restores to factory default; clear config in guest modem, all settings except
sip, advance sip restore to factory default.

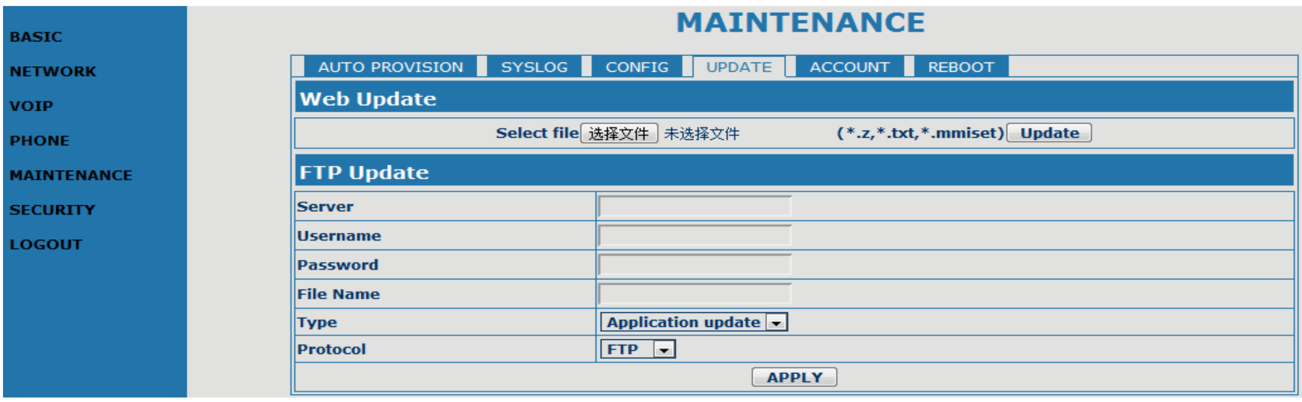
2.7 Update

2.8.1 Web Update:

update gateway's settings or firmware. Firmware file is .dlf extension when configure file is .cfg extension, OT-ATA200SP+ will auto select configure update or firmware update according the extension.

2.8.2 FTP Update:

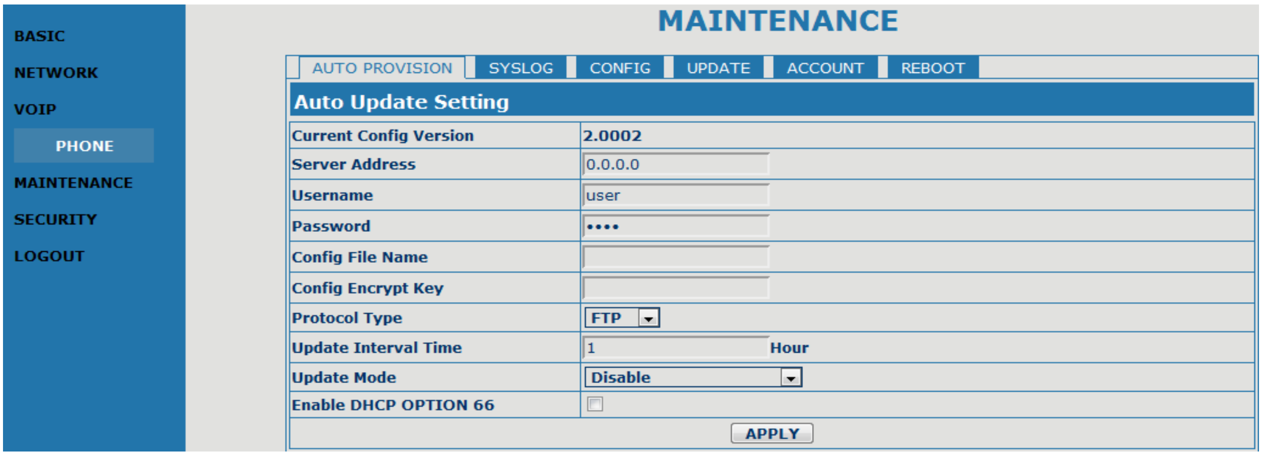
back up the configure file to FTP or TFTP server. Or auto update configure file from your auto update server. Back up configure file to your FTP/TFTP server.



* configure use .cfg extension.

2.8.3 Auto update:

OT-ATA200SP+ gateway support FTP and TFTP auto update. The gateway will auto obtain the configure file from your update server if configured. To obtain the original configure file, you can use the FTP/TFTP back up as describe above. Configure file using module structure, user may remain the concerned modules and remove other modules. Put the configure file in the root directory of update serve when finish editing.



Current Version: the system will display the current version number.
Server Address: FTP/TFTP server address
Username: FTP server user name
Password: FTP server password
Config File Name: The name of configuration file
Config Encrypt Key: The encrypt key of confirmation file
Protocol Type: The protocol type that used for upgrading
Update Interval Time: The interval time that the terminals search for new configuration file.
Update Mode: auto provision mode; Disable: not auto update,
Update after reboot:auto update after reboot,
Update at time interval:auto update after a certain time

Configure file version was in the <<VOIP CONFIG FILE>> and <GLOBLE CONFIG MODULE> ConfFile Version
For instance:

Gateway original version is:
<<VOIP CONFIG FILE>>Version:1.0000

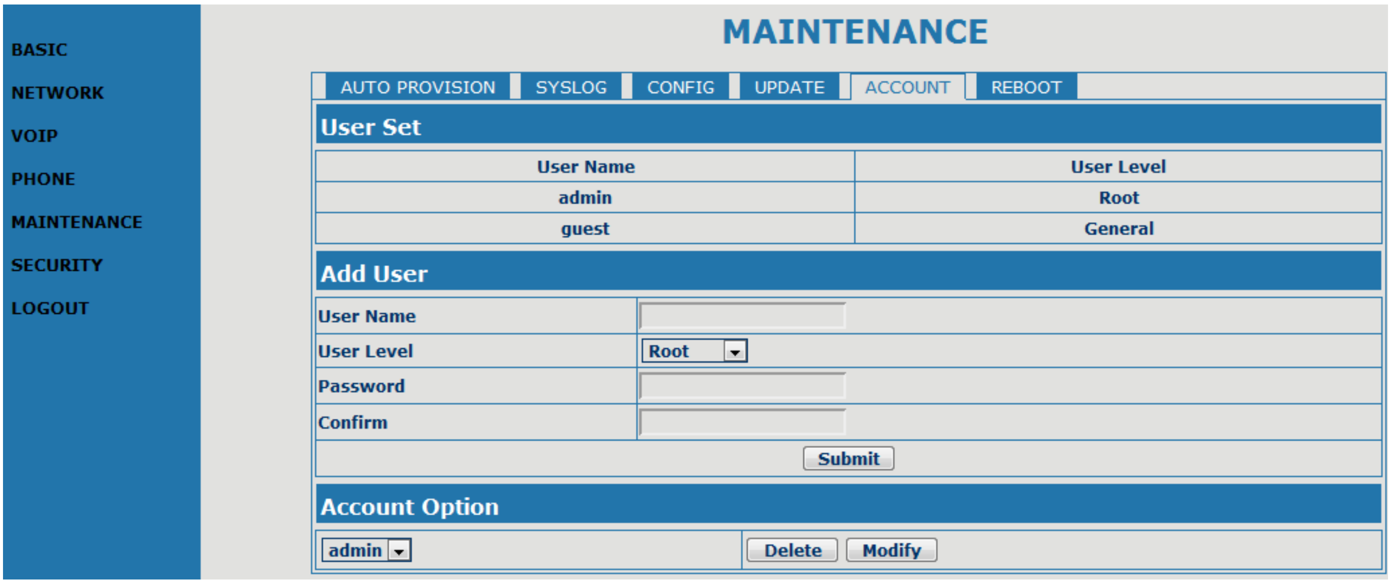
<GLOBLE CONFIG MODULE> ConfFile Version ： 6

User may edit the configure file version to:
<<VOIP CONFIG FILE>>Version:1.0007

<GLOBLE CONFIG MODULE> ConfFile Version ： 7

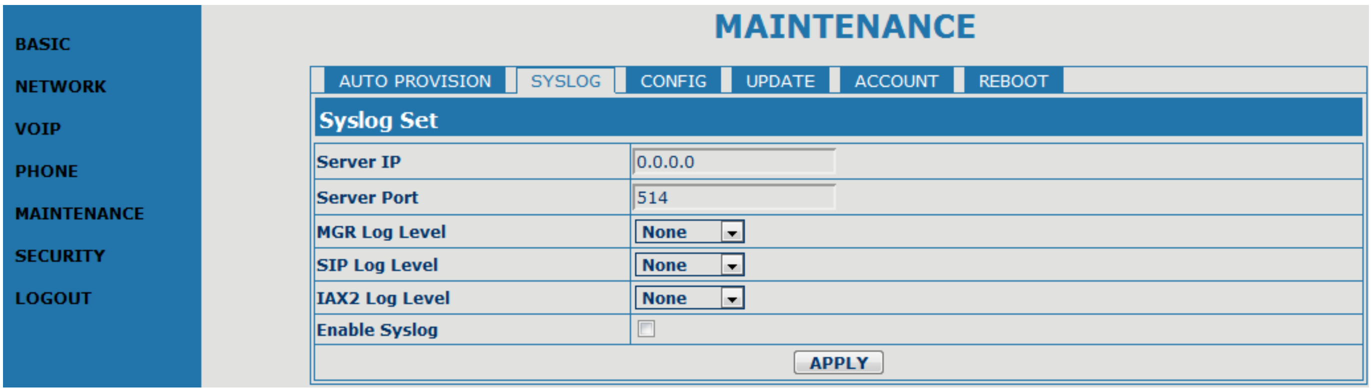
2.8 System Manage

2.8.1 Account Manage



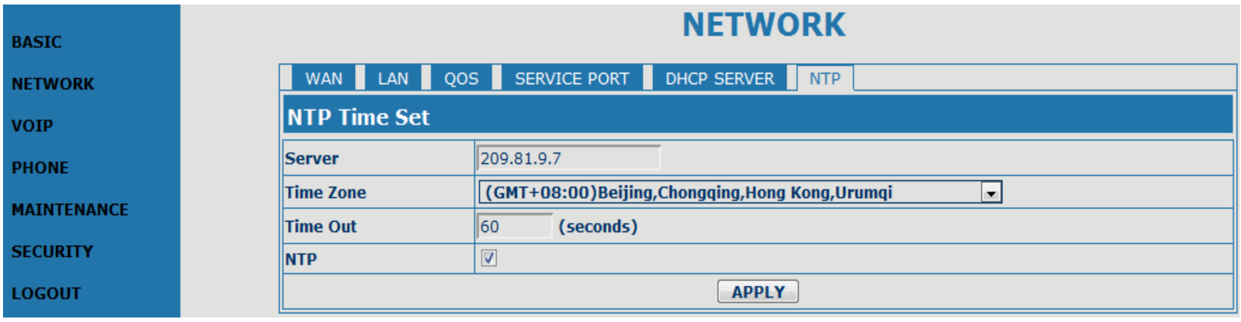
Set web access account or keypad password of OT-ATA200SP+.

2.8.2 Syslog Config:



Server IP: set the syslog server address
Server Port: set the syslog server port
MGR Log Level: set the MGR log level
SIP Log Level: set the SIP log level
IAX2 Log Level: set the IAX2 log level
Please click "apply" after setting

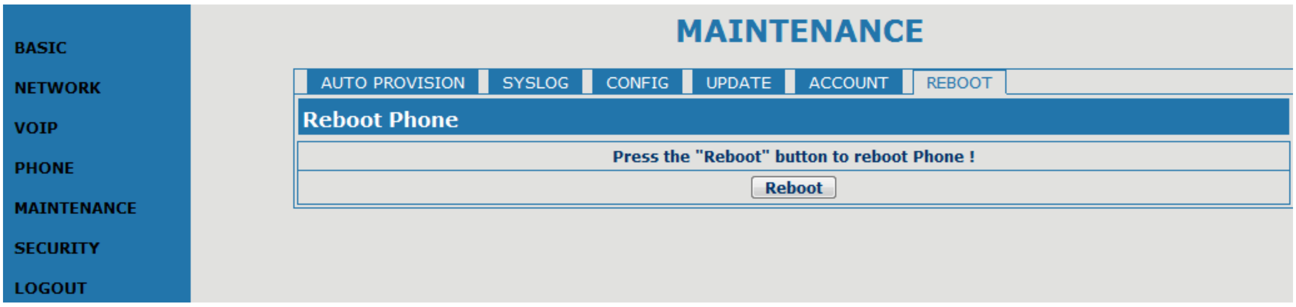
2.8.3 Time Set:



Server: type the ip address of time server
Time zone: select correct time zone in list box
Timeout: longest response time for SNTP
Manual Time set: The time setting
Daylight: Daylight Saving time

2.8.4 Logout &Reboot

Reboot Gateway, some setting needs to reboot to make it works. Please always save config before reboot, otherwise the setting will return to previous setting.



3 IVR setting

User may pre-config OT-ATA200SP+ gateway using a normal phone connecting to OT-ATA200SP+. please refer the below
command:
Notice: all command below can be end with # to speed response.

"#****" /*reboot gateway*/
"#*000" /*clear settings*/
"#*100" /*set the IP type to static ip */
"#*101" /*set IP type to DHCP */
"#*102" /*set IP type to PPPoE*/

"#*111" /*prompt gateway ip*/
"#*222" /* prompt phone number*/

Below setting need reboot to take effect
"#*103" /*change to bridge mode*/
"#*104" /*change to router mode*/

"#*50192.168.1.117" set WAN port IP address
"#*51192.168.1.1" set default gateway IP
"#*52202.112.10.37" set dns server
"#*53255.255.255.0" set netmask, use 255.255.255.0 if no be set

4 Telnet Console

4.1 Introduce

4.1.1 Basic structure

User may use telnet command to access and manage gateway.

OT-ATA200SP+ adopts tree structure for telnet. Every node contains its sub-nodes or local command. User can type “help” or “?” whenever to see sub-nodes and all local command under current node.

Besides local command, there are some global commands can be used in each node.

4.1.2 Basic command

Logout: exit telnet mode.

Write: save current settings.

Type sub-nodes name in current node to switch to sub-node.

Type “!” or “exit” in current node to return to parent-node.

Type “help” or “?” can see all sub-nodes and all local command under current node, every help item has comments such as <command> or <node> to distinguish sub-nodes and local command. Type “help” or “?” in command can see all parameters using in this command.

When typing node name or command, user no need to key the full name, use TAB button will make it more efficient. There are two types in command parameters: optional and required. “required” parameter use “-” as prefix and “optional” use “_” as prefix. User may type “-” or “_” then press TAB button for complementarily.

4.2 Global Command

Global command is available under all nodes, OT-ATA200SP+ support following commands:

Command	**Function**	**Example**
exit	Return to parent-node	#exit
logout	Exit	#logout
ping	Ping command, use to check network	#ping www.stephen-tele.com
write	Save setting to flash	#write

4.3 Tree Structure

4.3.1 Debug (Level 0~7)

path: <debug>#

show debug setting ---show

[disable]enable debug all modules ---[no] all xxx

[disable]enable debug app module ---[no] app xxx

[disable]enable debug cdr module ---[no] cdr xxx

[disable]enable debug sip module ---[no] sip xxx

[disable]enable debug tel module ---[no] tel xxx

[disable]enable debug dsp module ---[no] dsp xxx

4.3.2 reload
usage: #reload
Reboot system

4.3.3 show system running info
➢ basic
path: <show>#
show network status
Example: #<show>#basic

Show ip packets Stat.
Example:#<show>#ip ip

Show RTP packets Stat.
Example:#<show>#ip rtp

Show TCP packets Stat.
Example:#<show>#ip tcp

Show UDP packets Stat.
Example:#<show>#ip udp
➢ memory

path: <show>#
show gateway memory
Example:#<show>#memory
➢ nat
path: <show>#
show NAT information
Example:#<show>#nat
➢ uptime
path: <show>#
show running time
Example:#<show># uptime
➢ version
path: <show>#
show gateway version
Example:#<show># version

4.3.4 telnet and logout
Usage: #telnet –target -port
Login:xxx
Password:xxx
#
#logout

4.3.5 tracert trace network path info
usage: #tracert –host
Example:#tracert www.google.com

5 Network Diagnosis

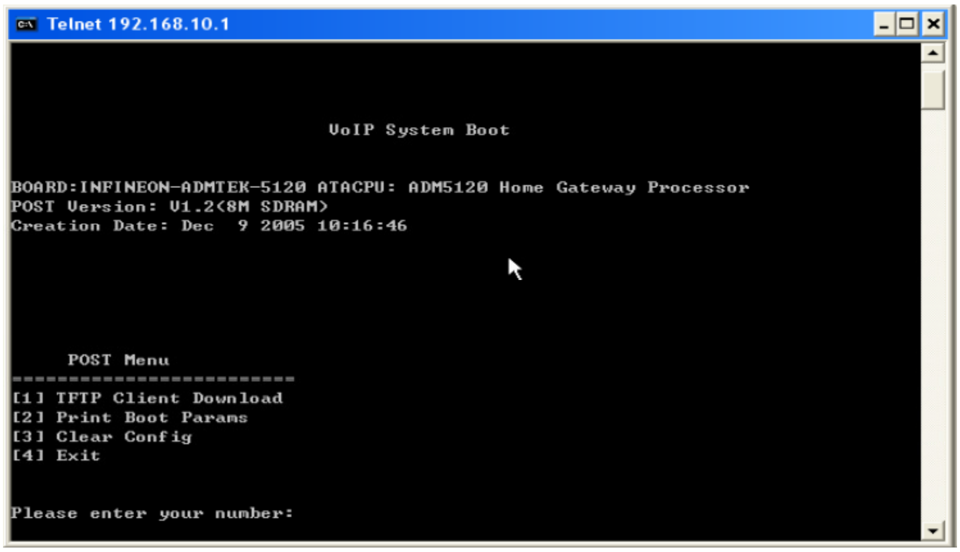
There are some telnet commands for checking your network. Now Listing below for your information

Command	Function	Example
ping	Check if the destination is accessible	#ping www.stephen-tele.com
tracert	Show network path info	#tracert www.stephen-tele.com
Show basic	Show network settings	#show basic
Show ip route	Show route table	#show ip route
Show ip arp	Show arp table	#show ip arp
telnet	Telnet to another device	#telnet 192.168.1.2

6 Restore to factory default

#setdefault clear gateway settings expect network part
#setdefault all clear all settings.

7 POST Mode(safe mode)



OT-ATA200SP+ provide safe mode. When there is booting problem because of setting problem or firmware problem. User can restore the factory setting or upgrade to a new firmware to solve this problem.

How to enter safe mode?
In the OT-ATA200SP+ booting procedure, it use the static ip 192.168.1.179 (WAN port IP) for a short time, user can telnet to this ip address in this occasion to enter the save mode.(remember to change your PC into the network 192.168.1.xx) Then user can according the guide in post mode to clear the settings or upgrade the firmware.

8 FAQ

How many SIP servers may OT-ATA200SP+ register simultaneously?

OT-ATA200SP+ support 2 SIP servers and 1 IAX server. The Default server is SIP . If you want to use the IAX server you must set IAX as default protocol in the IAX config page. IAX and SIP can register simultaneously but not work simultaneously. If you set 2 SIP servers in the SIP setting page , you can choose the route (server) by dialing plan which is edited by you. Please see"How to use the dial rule?" for detail.

How can I know the OT-ATA200SP+'s IP address?

Pick up the handset and then dial "#*111#", and the OT-ATA200SP+ will promote you its IP address.

How to use OT-ATA200SP+'s Lifeline function?

OT-ATA200SP+ supports Lifeline function, you can use the same handset to place PSTN and VoIP calls. First, you need to set up the Lifeline with the accessory send with the OT-ATA200SP+, connect this accessory to OT-ATA200SP+'s FXS port, and then connect the handset to the accessory's phone port, connect the landline to the accessory's line port. You can receive PSNT and VoIP calls simply with configuration. To place the PSTN call, you need to set up as follow:

----Add a new dial rule in the Dial-Peer setting: set the phone number to *T, and choose the Lifeline as the Call mode.----Add new Digital map item in the Advance _Digital Map: set Prefix Number to and *, and the length to 1. Then when you want to place a PSTN calls, you can first press * to switch to the PSTN line and then place your call as you normal do.

Why the settings vanish after reboot?

Please go to Config Manage_Save Config to save your setting always

How to use the dial rule?

OT-ATA200SP+ provide flexible dial rule, with different dial-rule configure, user can easily implement the following function:

----Replace, delete or add prefix of the dial number.
----Make direct IP to IP call
----Place the call to different SIP server according the prefix.
----Make PSTN calls use Lifeline function (Please refer "How can use the Lifeline function of OT-ATA200SP+?").
You can click "Add" to add a new dial rule. Below is the detail setting of the dial-rule:

Phone Number: The Number suit for this dial rule, cam be set as full match or prefix match. Full match means that if the number user dialed is completely the same as this number, the call will use this dial-rule. Prefix match means that if prefix of the number that the user dials is the same as the prefix, the call will use this dial-rule, to distinguish from the full match case, you need to add "T" after the prefix number in the phone number setting.

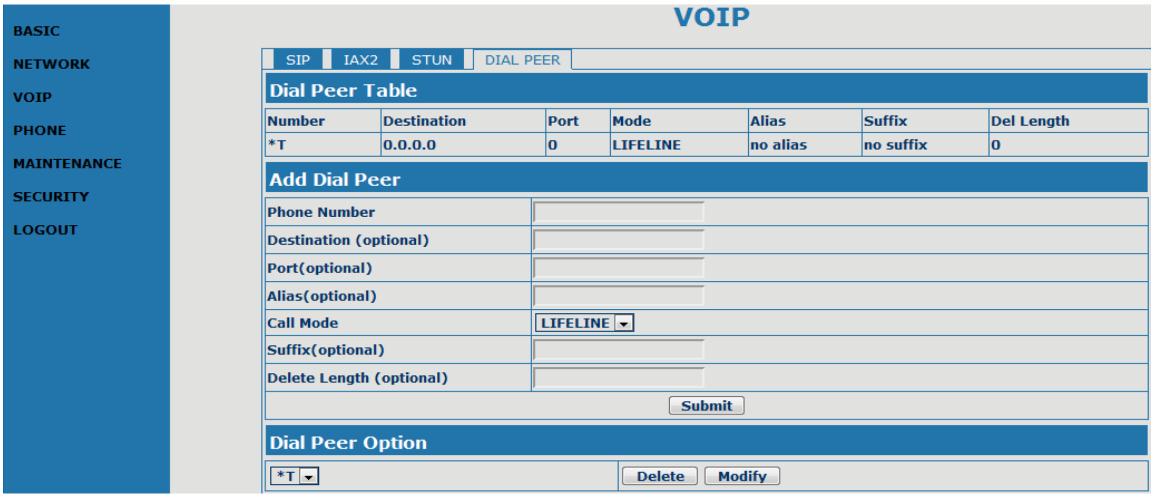
Call Mode: support SIP and Lifeline,SIP means the call will use sip protocol，Lifeline means the call will use the PSTN line.

Destination (optional): call destination, can be IP or domain. Default is 0.0.0.0, in this case the call will be routed to the Public SIP server. If you set the destination to 255.255.255.255, then the call will be routed to the private SIP server. Also you can key other address here to make direct IP calls
Port (optional): Configure the port of the destination, default is 5060
Alias (optional): Set up the Alias. We support four Alias as below. Alias need to co-work with the Del Length:

- add:xxx, add prefix to the phone number, can set to reduce the dial length.

- all: xxx, replace the phone number with the xxx, can use as speed dial function.
- del, delete the first N numbers. N is set in the Del Length
- rep:xxx,

replace the first N numbers. N is set in the Del Length. For Example: Use wants to place a call 8610-62281493, then you can set the phone number in the dial rule as 010T, and set the Alias as rep:8610, and set the Del Length to 3. Then all calls begin with 010 will be changed to 8610 xxxxxxxx.
Suffix (optional): Configure suffix, show no suffix if not set Instance:

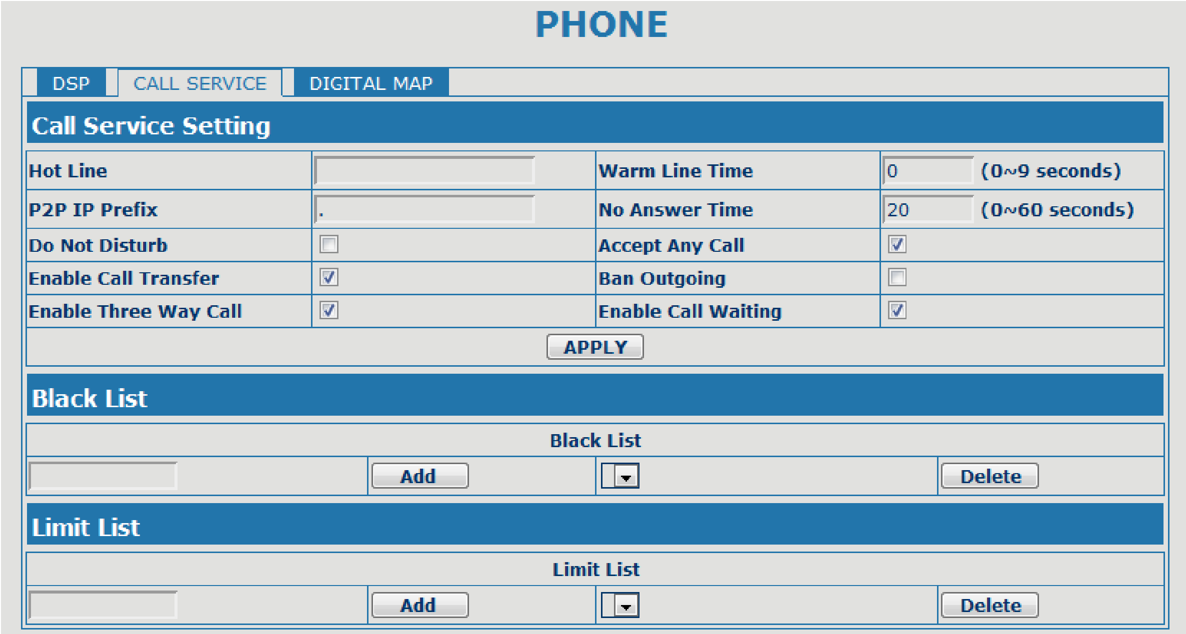


2T rule: If the call starts with 2, the first 2 will be deleted, and the rest number will be sent to private server. **3T rule:** If the call starts with 3, the first 3 will be deleted, and the rest number with be sent to public server. **123 rule:** Dial 123 and will send 8675583018049 to your server. Used as speed dial function.
0T rule: If the calls is begin with 0, the first 0 will be replace by 86. Means that if you dial 075583018049 and OT-ATA200SP+ will send 8675583018049 to your server.
***T rule:** Dial the * and the line with switch to PSTN. Note that you need to set another rule "Prefix Number: *; Length: 1" in the Digital Map. (Refer "How to use OT-ATA200SP+'s Lifeline Function?")
179 rule: when you dial 179 , the call with send to 192.168.1.179, suit for LAN application without set up a sip server.

How to use speed dial function?
Please refer to "How to use dial rule?".

How to configure digital map?
Please refer digit map settings.

How to use Call Forward, Call Transfer and 3-way Conference calls?
User may set up the configuration in the Call Service page to use these value add service.



- Call Forward:

----Forward when busy: select Busy in the Call Forward Field, and Key in the destination phone number in the Forward Number. If some one calls you when you having a call, the caller will be forwarded to the destination number.

----Forward no answer: Select No Answer in the Call Forward Field, and Key in the destination phone number in the Forward Number, fill the time in the No Answer Time. If some one calls you and no one answer the caller during the No Answer Time, the call will be forward to the destination number.

----Forward Always: Select Always in the Call Forward Field, and Key in the destination phone number in the Forward Number, then any one calls this gateway will be forward to the destination number.

- Call Transfer:

Check the Enable Call Transfer.
If A is the OT-ATA200SP+ user, and B calls and talking with A through VoIP. A can **press the Hook-Flash** to hold the call with B, and then press * and then **enter C's number**. B will be transferred to C and can talk with C.

- 3-Way Conference Calls

Check Enable Three Way Call
Only sip protocol support this function .Assume A is the OT-ATA200SP+ user, and B calls and talking with A through VoIP. A can **press Hook-Flash** to hold the call with B, then **enter C's number** to talk with C, and then **press Hook-Flash** again switch back to user B , then A can press * to make 3-way conference calls.

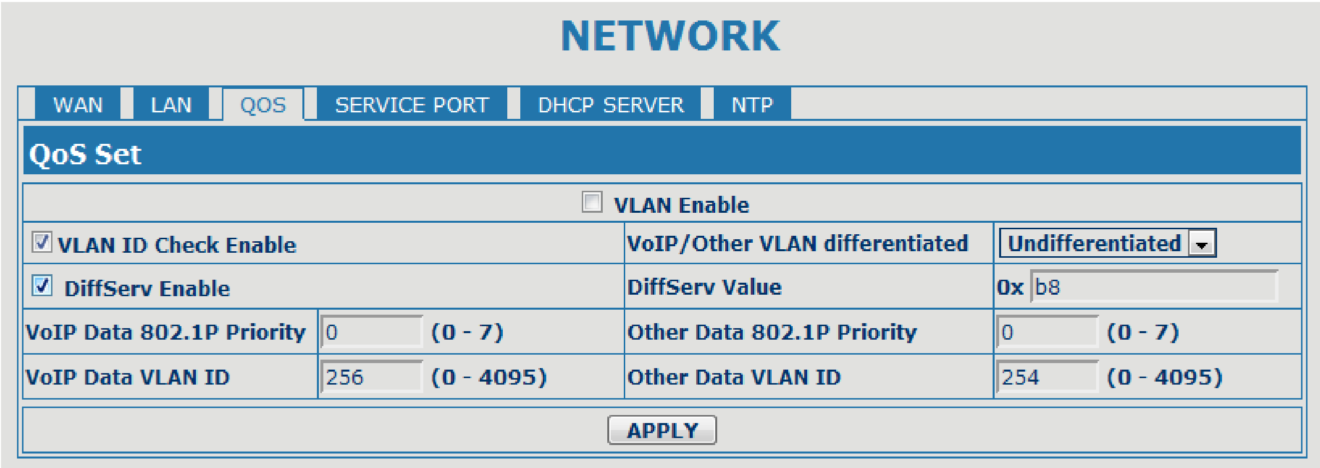
Notice: A can press **Hook-Flash** to switch between B and C. or press # to cancel the current call and switch to the other user.

VLAN implement

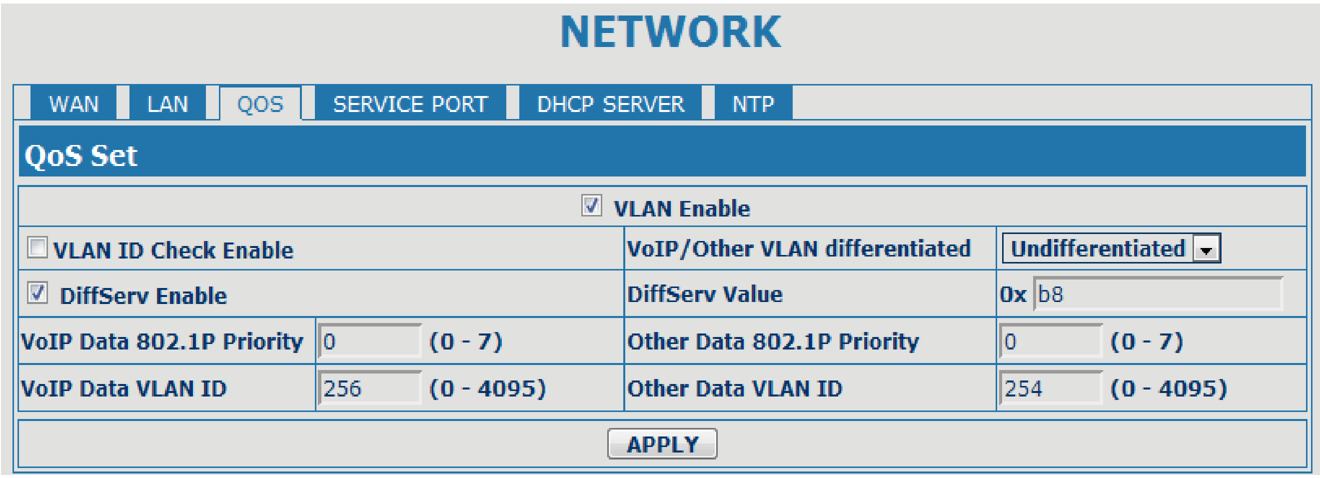
OT-ATA200SP+ support rich 802.1Q/P protocol and Diffserv configuration. Through its flexible VLAN function, you can set the voice/signaling and data packets in different VLAN via different VLAN id.

Different implement of VLAN function:

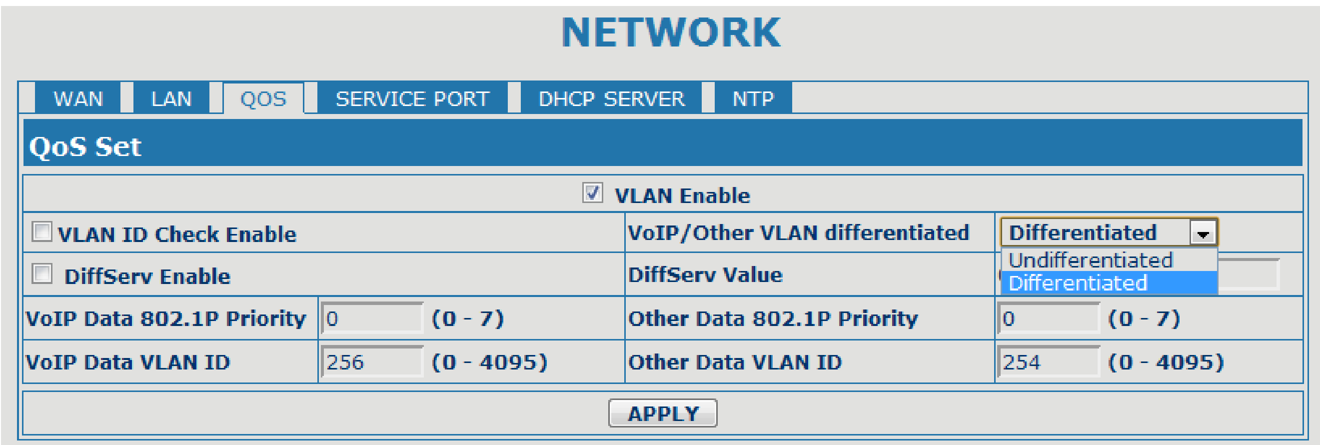
1: if "Data/Voice VLAN differentiated" is undifferentiated. Device will set the same vlan ID for voice and data. As show below



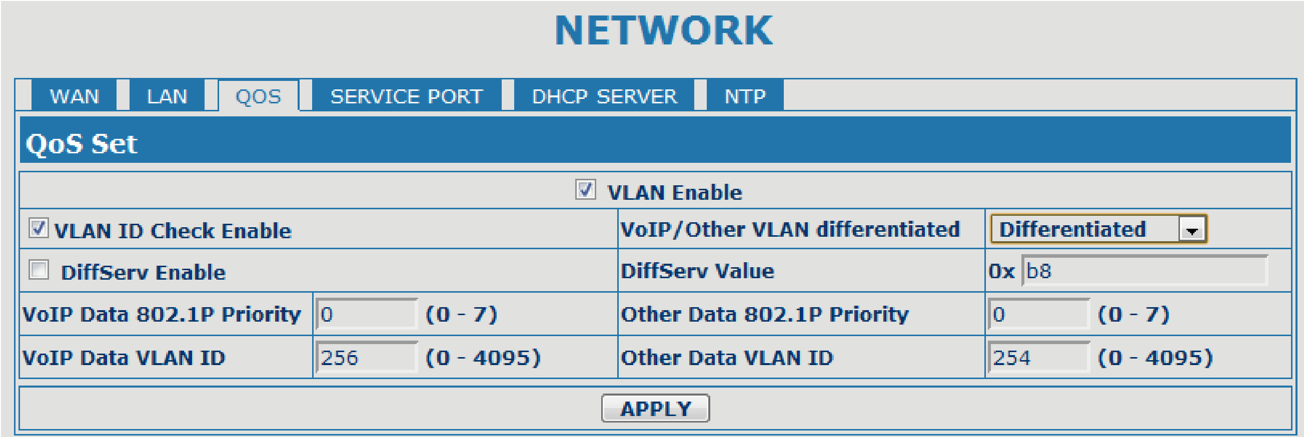
Or



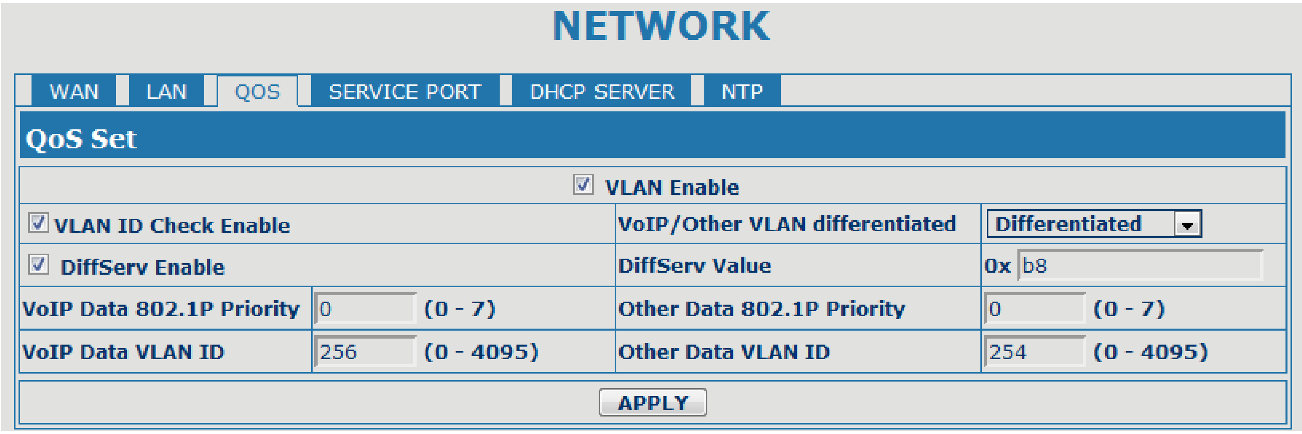
2. if "Data/Voice VLAN differentiated" is Tag differentiated but the DiffServ is disable. Device won't distinguish the voice, signaling and data stream. It will add the same data vlan id to them. As below:



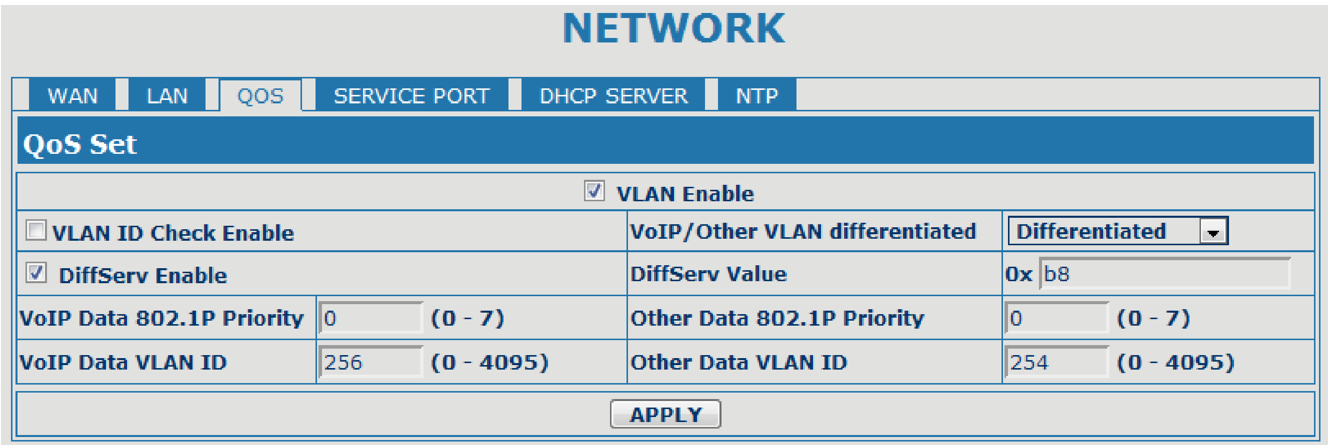
Or



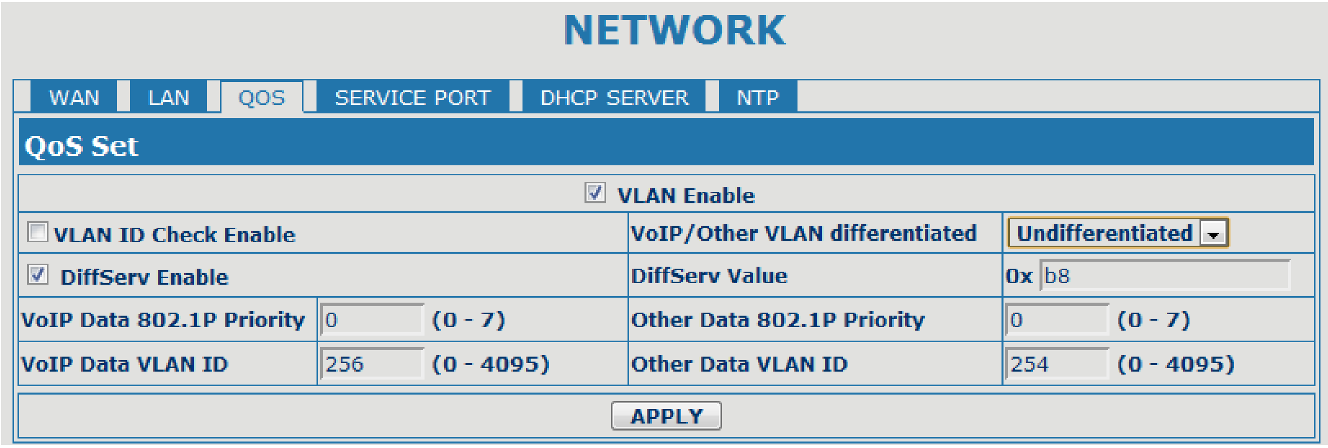
3. if "Data/Voice VLAN differentiated" is Tag differentiated and diffServ are both enable. Then device will distinguish the voice, signaling and data stream to VLAN ID setting. As below:



Or



4. if "Data/Voice VLAN differentiated" is Date untaged and diffServ are both enable. Then device will undistinguish the date to VLAN ID setting. As below:



5. if VLAN is disable. Device won't add any vlan ID to the stream. In this case, if the Diffserv is enable, the DiffServ value response to the voice/signaling stream.

6. When VLAN function is enable. If "VLAN ID check" is enable, OT-ATA200SP+ will have strict requirement on the VLAN, it won't handle any packets with different VLAN ID. If "VLAN ID check" is disable, OT-ATA200SP+ will handle the packets even from different vlan ID. Please notice that VLAN ID check is enable in default.

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