

AG-198 User Manual

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| | |
|--|----|
| 1 AG-198 Features..... | 4 |
| 1.1 Appearance..... | 4 |
| 1.2 Interface..... | 4 |
| 1.3 Electricity characteristic..... | 5 |
| 1.4 Software..... | 5 |
| 1.5 Standard and Protocols..... | 5 |
| 1.6 Compliant Standards..... | 6 |
| 1.7 Operating requirement..... | 6 |
| 1.8 Package..... | 6 |
| 1.9 Installation..... | 7 |
| 2 Web Configuration..... | 7 |
| 2.1 Access Web setting page..... | 7 |
| 2.2 Current state..... | 8 |
| At the bottom of the page, there shows firmware version..... | 8 |
| Version: VOIP Gateway V1.0.178.16 Jul 12 2011..... | 8 |
| Network..... | 9 |
| 2.2.1 Wan Config..... | 9 |
| 2.2.2 LAN Config..... | 10 |
| 2.3 VoIP..... | 10 |
| 2.4.1 SIP Config..... | 10 |
| 2.4.2 Iax2 Config..... | 13 |
| 2.4 Advance..... | 15 |
| 2.4.1 DHCP Server..... | 15 |
| 2.4.2 NAT..... | 16 |
| 2.4.3 Net Service..... | 17 |
| 2.4.4 QoS settings..... | 18 |
| 2.4.5 Digital Map..... | 19 |
| 2.4.6 Call Service..... | 20 |
| 2.4.7 MMI Filter..... | 21 |
| 2.4.8 Audio Settings..... | 22 |
| 2.4.9 Firewall..... | 23 |
| 2.4.10 VPN..... | 24 |
| Dial-Peer Settings..... | 25 |
| 2.5 Config Manage..... | 26 |
| 2.6 Update..... | 27 |
| 2.7 System Manage..... | 29 |
| 2.7.1 Account Manage..... | 29 |
| 2.7.2 Syslog Config..... | 29 |
| 2.7.3 Time Set..... | 30 |
| 2.7.4 Logout&Reboot..... | 30 |
| 3 IVR..... | 31 |
| POST Mode(safe mode)..... | 31 |
| 4 FAQ..... | 34 |

| | |
|---|----|
| How many SIP servers may AG-198 register simultaneously?..... | 34 |
| How can I know the AG-198's IP address?..... | 34 |
| How to use AG-198's Lifeline function?..... | 34 |
| Why the settings vanish after reboot?..... | 34 |
| How to use the dial rule?..... | 34 |
| How to use speed dial function?..... | 36 |
| How to configure digital map?..... | 36 |
| How to use Call Transfer and 3-way Conference calls?..... | 36 |
| VLAN implement..... | 37 |

1 AG-198 Features

1.1 Appearance



PWR: power connection status

SYS : server registration status . registered: on , gliting , not-registered , off

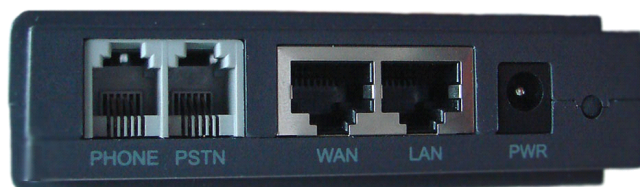
WAN: WAN port connectiong , conctected , on , offline : off

LAN: LAN port connectiong , conctected , on , offline : off

PSTN: light on when PSTN call come in/out

VoIP: light on when VoIP call come in/out. Default to use VOIP to call out.

1.2 Interface



PWR: 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.
- NAT.
- 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
- DTMF of inband audio, RFC2833 and SIP info etc.
- 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 TFTP.
- Auto provision for config file.
- Web management.
- Reverse polarity
- Voice prompt
- adjustable user password and admin 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
- DNS: Domain Name Server
- TFTP: Trivial File Transfer Protocol
- HTTP: Hyper Text Transfer protocol
- FTP: File Transfer protocol
- RFC 3362: T.38 protocol

1.6 Compliant Standards

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



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

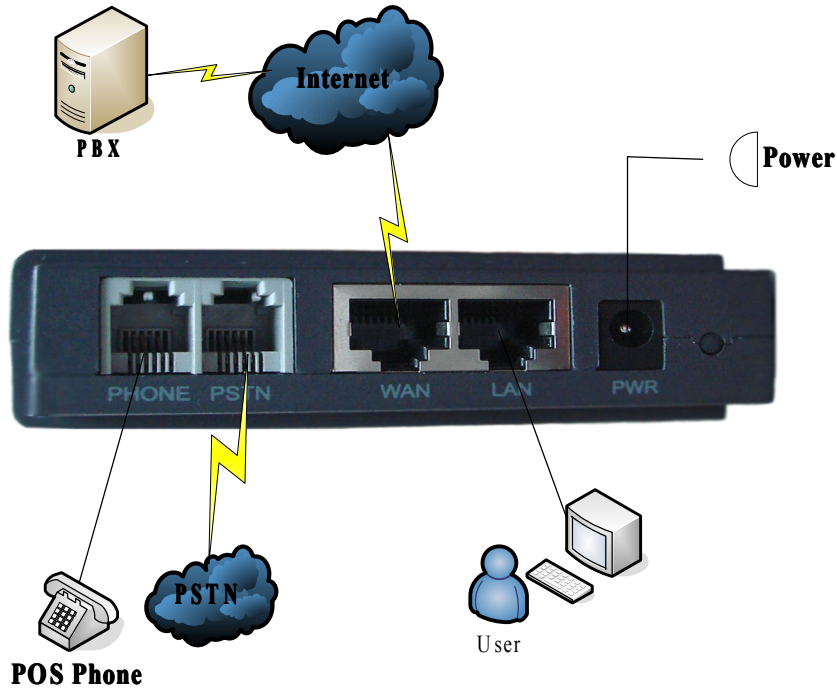
1.8 Package

- Size 128 x 85 x 30 mm
- Packing List
 - ✓ AG-198 gateway
 - ✓ Power adaptor (12v, 500mA)
 - ✓ Manual CD

1.9 Installation

Use Ethernet cable to connect AG-198'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 page of AG-198, 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 AG-198 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 AG-198 setting page.

Default username and password is:

| | | |
|-----------------------|------------------------|------------------------|
| Administrator: | Username: admin | password: admin |
| User: | Username: guest | Username: guest |

The screenshot shows the login interface with the following fields and button:

- Username:
- Password:
- Logon button

2.2 Current state

The screenshot displays the ATCOM VoIP Gateway web interface. The left sidebar contains a navigation menu with the following items: Current State (selected), Network, VoIP, SIP Config, IAX2 Config, Advance, DHCP Server, NAT, Net Service, QOS, Digital Map, Call Service, MMI Filter, Audio Settings, firewall, VPN, Dial-Peer, Config Manage, Update, and System Manage. The main content area is titled 'VoIP Gateway' and 'Running Status'. It features two tables:

| Network | | Running Status | | |
|---------|--------------|----------------|-------------|-------------------|
| WAN | Connect Mode | DHCP | MAC Address | 00:09:45:5a:62:32 |
| | IP Address | 192.168.1.230 | Gateway | 192.168.1.254 |
| LAN | IP Address | 192.168.10.1 | DHCP Server | ON |

| Phone Number | | |
|--------------|-------------------------|------------|
| SIP LINE 1 | 8151@192.168.1.203:5060 | Registered |
| Phone Number | | |
| SIP LINE 2 | 8008@192.168.1.203:5060 | Registered |
| IAX2 | 6009@192.168.1.99:4569 | Registered |

At the bottom of the page, the firmware version is displayed: Version: VOIP Gateway V1.0.178.16 Jul 12 2011

This page shows AG-198'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 AG-198 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.

At the bottom of the page, there shows firmware version.

Version: VOIP Gateway V1.0.178.16 Jul 12 2011

Network

2.2.1 Wan Config

| WAN Status | |
|-----------------|-------------------|
| Active IP | 192.168.0.102 |
| Current Netmask | 255.255.255.0 |
| Current Gateway | 192.168.0.1 |
| MAC Address | 00:09:45:5a:62:32 |
| Get MAC Time | 20110601 |

| WAN Setting | | |
|---|---------------------------------------|-----------------------------|
| Static <input type="radio"/> | DHCP <input checked="" type="radio"/> | PPPOE <input type="radio"/> |
| Static IP Address | 192.168.1.179 | |
| Netmask | 255.255.255.0 | |
| Gateway | 192.168.1.254 | |
| DNS Domain | | |
| Primary DNS | 202.96.134.133 | |
| Alter DNS | 202.96.128.68 | |
| <input checked="" type="checkbox"/> Obtain DNS server automatically | | |
| PPPOE Server | ANY | |
| Username | user123 | |
| Password | ***** | |

WAN support static IP, dynamic obtain IP and PPPoE. Default to use DHCP.

- Configure Static IP:
 - Enable Static;
 - Set AG-198'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, AG-198 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.
 - AG-198 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 AG-198 when the WAN port address was changed.

2.2.2 LAN Config

IP , Netmask: Set the IP and Netmask for the LAN

DHCP Server: Enable DHCP service in LAN port

NAT: Enable NAT.

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)

2.3 VoIP

2.4.1 SIP Config

➤ **SIP Line Select:**

AG-198 support 2 sip line. You can select SIP1 or SIP2 in the drop down box, and press Load to switch.

➤ **Basic Setting:**

- ✓ Register Status: Shows whether the sip account registered successfully
- ✓ Server Name: Keep it as blank if no specific

- ✓ Register Server Addr: Register address of public SIP server
- ✓ Register Server Port: Register port of public SIP server
- ✓ Account Name: Username of your SIP account
- ✓ Password: Password of your SIP account
- ✓ Phone number: Set phone number the same as Account Name
- ✓ Display Name: Name displayed as Caller ID. If server has set the Caller ID, your Caller ID will be the one which set by server not by phone.
- ✓ 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 AG-198 will use the proxy server address as sip domain
- ✓ Enable Register: Enable register or disable register
- **Advanced Set:**

Advanced Set

| | | | |
|----------------------------|-------------------------------------|--------------------------|--------------------------|
| Register Expire Time | 60 seconds | Forward Type | Off |
| NAT Keep Alive Interval | 60 seconds | Forward Phone Number | |
| User Agent | Voip Phone 1.0 | Server Type | COMMON |
| DTMF Mode | DTMF_RELAY | Subscribe Expire Time | 300 seconds |
| Media Key | | RFC Protocol Edition | RFC3261 |
| Local Port | 5060 | Transport Protocol | UDP |
| RFC Privacy Edition | NONE | MWI Number | |
| Transfer Expire Time | 0 seconds | Enable DNS SRV | <input type="checkbox"/> |
| Enable Keep Authentication | <input type="checkbox"/> | Enable Subscribe | <input type="checkbox"/> |
| NAT Keep Alive | <input type="checkbox"/> | Rtp Encode | <input type="checkbox"/> |
| Enable Via rport | <input type="checkbox"/> | Enable Session Timer | <input type="checkbox"/> |
| Enable PRACK | <input type="checkbox"/> | Answer With Single Codec | <input type="checkbox"/> |
| Long Contact | <input type="checkbox"/> | Auto TCP | <input type="checkbox"/> |
| Enable URI Convert | <input checked="" type="checkbox"/> | Enable Strict Proxy | <input type="checkbox"/> |
| Dial Without Register | <input type="checkbox"/> | Enable GRUU | <input type="checkbox"/> |
| Ban Anonymous Call | <input type="checkbox"/> | Enable Displayname Quote | <input type="checkbox"/> |

APPLY

- ✓ Register Expire Time: Register expire time, default is 60 seconds. AT-640 will auto configure this expire time to the server recommended setting if it is different from the SIP server.
- ✓ NAT Keep Alive Interval: Set examining interval of server, default is 60 seconds. If the phone has enabled Nat Keep Alive, the phone will detect server after interval to check if the server has response.
- ✓ User Agent: Set the user agent if have, the default is VoIP Phone 1.0.
- ✓ DTMF Mode: Select DTMF sending mode, there are three modes:

- DTMF_RELAY
- DTMF_RFC2833
- DTMF_SIP_INFO
- Different VoIP Service providers may provide different modes.
- ✓ Media Key: voice stream encryption Key.
- ✓ Local Port: Local SIP signal port, default as 5060.
- ✓ RFC Privacy Edition: Set privacy protocol.
- ✓ Transfer Expire Time: Transfer expire time set.
- ✓ Enable Keep Authentication: Enable/Disable Keep Authentication System will take the last authentication field which is passed the authentication by server to the request packet. It will decrease the server's repeat authorization work, if it is enable.
- ✓ NAT Keep Alive: If the phone has enabled Nat Keep Alive, the phone will detect server after interval to check if the server has response.
- ✓ Enable Via rport: Enable/Disable system to support RFC3581. Via rport is special way to realize SIP NAT.
- ✓ Enable PRACK: Enable or disable SIP PRACK function, suggest use the default config.
- ✓ Long Contact: Set more parameters in contact field.
- ✓ Enable URI Convert: convert # into %23 when sending URI.
- ✓ Dial Without Register: Set call out by proxy without registration.
- ✓ Ban Anonymous Call: Set to ban Anonymous Call.
- ✓ Forward Type: Select call forward mode, the default is Off .
- Off: Close down calling forward.
- Busy: If the phone is busy, incoming calls will be forwarded to the appointed phone.
- No answer: If there is no answer, incoming calls will be forwarded to the appointed phone.
- Always: Incoming calls will be forwarded to the appoint phone directly.
The phone will prompt the incoming while doing forward.
- ✓ Forward Phone Number: Appoint your forward phone number.
- ✓ Server Type: Select the special type of server which is encrypted, or has some unique requirements or call flows.
- ✓ Subscribe Expire Time: Overtime of resending subscribe packet. Suggest using the default config.
- ✓ RFC Protocol Edition: Select SIP protocol version to adapt for the SIP server which uses the same version as you select. For example, if the server is CISCO5300, you need to change to RFC2543; else phone may not cancel call normally. System uses RFC3261 as default.
- ✓ Transport Protocol: Set transport protocols, TCP or UDP.
- ✓ MWI Number: Set SIP1 voicemail Number.
- ✓ Enable DNS SRV: Enable DNS
- ✓ Enable Subscribe: Enable Subscribe: Overtime of resending subscribe packet. Suggest using the default config.
- ✓ Rtp Encode: enable voice data encryption.
- ✓ Enable Session Timer: enable rfc4028 to refresh the SIP sessions.

- ✓ Answer With Single Codec: only answer the call with a certain Codec.
- ✓ Auto TCP: enable TCP transmission protocol when the length of message exceed 1300 byte
- ✓ Enable Strict Proxy: Support the special SIP server-when phone receives the packets sent from server, phone will use the source IP address, not the address in via field.
- ✓ Enable GRUU: Set to support GRUU.
- ✓ Enable Displayname Quote: Set to make quotation mark to display name as the phone sends out signal, in order to be compatible with server.

2.4.2 IAX2 Config

| Parameter | Value |
|------------------------|-------------------------------------|
| Register Status | Registered |
| IAX2 Server Addr | 192.168.1.99 |
| IAX2 Server Port | 4569 |
| Account Name | 6009 |
| Account Password | •••• |
| Phone Number | 6009 |
| Local Port | 4569 |
| Voice Mail Number | 0 |
| Voice Mail Text | mail |
| Echo Test Number | 1 |
| Echo Test Text | echo |
| Refresh Time | 60 Seconds |
| Enable Register | <input checked="" type="checkbox"/> |
| Enable G.729 | <input type="checkbox"/> |
| IAX2(Default Protocol) | <input type="checkbox"/> |

- IAX2 Server Addr:** Register address of public IAX server
- IAX2 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 working 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

2.4 Advance

2.4.1 DHCP Server

ATCOM

Current State

Network

VoIP

Advance

- DHCP Server
- NAT
- Net Service
- QOS
- Digital Map
- Call Service
- MMI Filter
- Audio Settings
- firewall
- VPN

Dial-Peer

Config Manage

Update

System Manage

DHCP Leased Table

| Leased IP Address | Client Hardware Address |
|-------------------|-------------------------|
| lan | |

DHCP Lease Table

| Name | Start IP | End IP | Lease Time | Netmask | Gateway | DNS |
|------|--------------|---------------|------------|---------------|--------------|--------------|
| lan | 192.168.10.2 | 192.168.10.33 | 1440 | 255.255.255.0 | 192.168.10.1 | 192.168.10.1 |

DHCP Lease Table Setting

| | |
|------------------|-------------------------------|
| Lease Table Name | <input type="text"/> |
| Start IP | <input type="text"/> |
| End IP | <input type="text"/> |
| Lease Time | <input type="text"/> (minute) |
| Netmask | <input type="text"/> |
| Gateway | <input type="text"/> |
| DNS | <input type="text"/> |

DHCP Lease Table Delete

| | | |
|------------------|-------|---------------------------------------|
| Lease Table Name | lan ▼ | <input type="button" value="Delete"/> |
|------------------|-------|---------------------------------------|

DNS relay Setting

| | |
|---|--------------------------------------|
| DNS Relay <input checked="" type="checkbox"/> | <input type="button" value="APPLY"/> |
|---|--------------------------------------|

User may trace and modify DHCP server information in this page.

DHCP Leased Table: show existed DHCP lease table.

DHCP Lease Table setting: User may use below setting to add a new lease table.

- Lease Table Name: Lease table name.
- Start IP: Start IP of lease table.
- End IP: End IP of lease table. Network device connecting to the AG-198 LAN port can dynamic obtain the IP in the range between start IP and end IP.
- Lease Time: DHCP server lease time.
- Netmask: Netmask of lease table.
- Gateway: Default gateway of lease table.
- DNS: default DNS server of lease table.

DHCP Lease Table Delete: Delete existed lease table.

DNS Relay: enable DNS relay function.

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

2.4.2 NAT

VoIP Gateway

NAT Configuration

IPsec ALG FTP ALG

PPTP ALG

Apply

| | | |
|-----------|-----------------|------------------|
| Inside IP | Inside TCP Port | Outside TCP Port |
|-----------|-----------------|------------------|

| | | |
|-----------|-----------------|------------------|
| Inside IP | Inside UDP Port | Outside UDP Port |
|-----------|-----------------|------------------|

Transfer Type: TCP Inside IP:

Inside Port: Outside Port:

Add Delete

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.

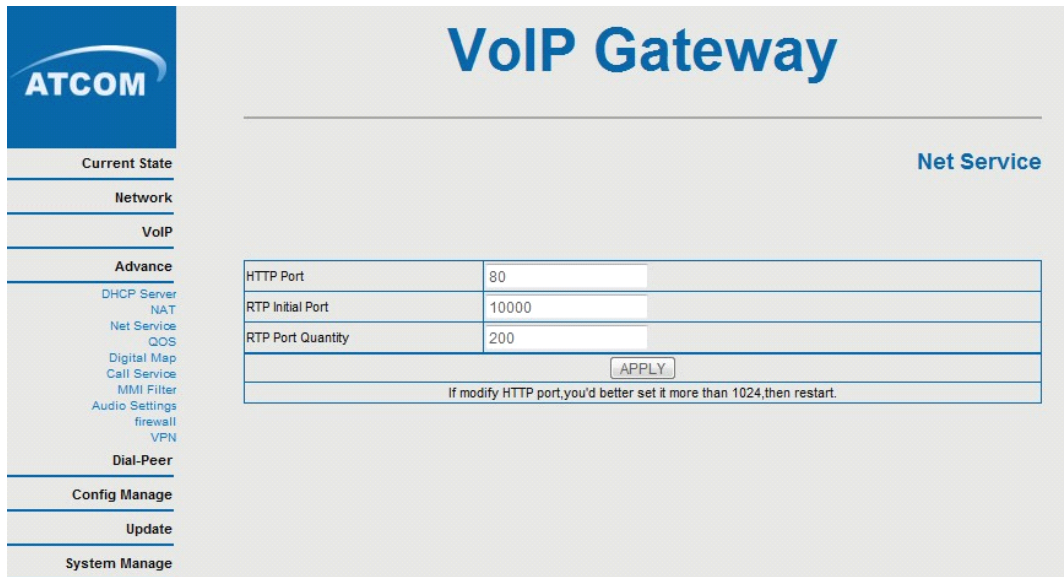
Inside IP: LAN device IP for port mapping.

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.

2.4.3 Net Service



| VoIP Gateway | |
|--|-------|
| Net Service | |
| HTTP Port | 80 |
| RTP Initial Port | 10000 |
| RTP Port Quantity | 200 |
| <input type="button" value="APPLY"/> | |
| If modify HTTP port, you'd better set it more than 1024, then restart. | |

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.

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.

2.4.4 QoS settings

| QoS Configuration | | | |
|--|--------------------------------|----------------------------|----------------|
| <input type="checkbox"/> VLAN Enable | | | |
| <input checked="" type="checkbox"/> VLAN ID Check Enable | VoIP/Other VLAN differentiated | Undifferentiated ▼ | |
| <input type="checkbox"/> DiffServ Enable | DiffServ Value | 0x b8 | |
| VoIP Data 802.1P Priority | 0 (0 - 7) | Other Data 802.1P Priority | 0 (0 - 7) |
| VoIP Data VLAN ID | 256 (0 - 4095) | Other Data VLAN ID | 254 (0 - 4095) |

AG-198 implement QoS based on 802.1p, The QoS is used to mark the network communication priority in the data link/MAC sub-layer. AG-198 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 ,0xb8 stands for best fast transmission; 28-30 is guarantee 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

2.4.5 Digital Map

The screenshot displays the 'Digital Map Configuration' page in the VoIP Gateway web interface. On the left is a navigation sidebar with categories: Current State, Network, VoIP, Advance, Dial-Peer, Config Manage, Update, and System Manage. The 'Advance' section is expanded, showing sub-items like DHCP Server, NAT, Net Service, QOS, SIP, Digital Map (selected), Call Service, MMI Filter, Audio Settings, and VPN. The main content area is titled 'VoIP Gateway' and 'Digital Map Configuration'. It features a configuration box with the following settings: 'End with #' is checked, 'Fixed Length' is set to 11, and 'Time out' is set to 5 seconds. An 'Apply' button is present. Below the configuration box is the 'Digital Map Table', which contains a list of prefix numbers: *, 8[3-8]XXXXX, 89XXXX, 6567, 78XXXT2, and 5[3,7,9]XXXXX. At the bottom of the table, there are input fields for 'Prefix Number' and 'Prefix to be deleted', along with 'Add' and 'Delete' buttons.

Digit map is a set of rules to determine when the user has finished dialing.

AG-198 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. AG-198 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.

2.4.6 Call Service

The screenshot displays the 'Call Service' configuration interface. On the left is a navigation menu. The main area is titled 'VoIP Gateway' and 'Call Service'. It features three main sections:

- Call Service Setting:** A table with the following rows:

| | | | |
|-----------------------|-------------------------------------|---------------------|--|
| Hot Line | <input type="text"/> | Warm Line Time | <input type="text" value="0"/> (0~9 seconds) |
| P2P IP Prefix | <input type="text"/> | No Answer Time | <input type="text" value="20"/> (0~60 seconds) |
| Do Not Disturb | <input type="checkbox"/> | Accept Any Call | <input checked="" type="checkbox"/> |
| Enable Call Transfer | <input checked="" type="checkbox"/> | Ban Outgoing | <input type="checkbox"/> |
| Enable Three Way Call | <input checked="" type="checkbox"/> | Enable Call Waiting | <input checked="" type="checkbox"/> |

 Below the table is an 'APPLY' button.
- Black List:** A section with a title 'Black List', an input field, an 'Add' button, a dropdown arrow, and a 'Delete' button.
- Limit List:** A section with a title 'Limit List', an input field, an 'Add' button, a dropdown arrow, and a 'Delete' button.

User configure the value add service such as hotline, call forward, call transfer, 3-way conference call .etc in this page

Call Service Setting:

- Hotline: configure hotline number. AG-198 will immediately dials this number after hook-off if it is set.
- Warm Line Time: configure warm line time, if you pick up the phone, the number will be send out after the time.
- P2P IP Prefix: Set ip prefix. For example, you set prefix as 192.168.1.,then you can dial #200# to call the phone which ip is 192.168.1.200.
- No Answer Time: no answer call forward time setting
- Do Not Disturb: DND, enable this option to refuse any calls.
- Accept Any Call: If this option is disable, AG-198 refuse the incoming call when the called number is different from AG-198's phone number.
- Enable Call Transfer: Please refer to value add service for detail.
- Ban Outgoing: Enable this to ban outgoing calls.
- Enable Three Way Call: Please refer to value add service for detail.
- Enable Call Waiting: Enable/disable Call Waiting

Black List: incoming call in these phone numbers will be refused.

Limit List: outgoing calls with these phone numbers will be refused

2.4.7 MMI Filter

VoIP Gateway

MMI Filter

MMI Filter

Start IP End IP

Start IP End IP

Start IP to be deleted

MMI filter is used to make access limit to AG-198 Gateway.

When MMI filter is enable. Only IP address within the start IP and end IP can access AG-198 gateway.

2.4.8 Audio Settings

| | | | |
|---------------------|--------------------------|--------------------|----------------|
| First Codec | g711Ulaw64k | Second Codec | g711Alaw64k |
| Third Codec | g729 | Fourth Codec | g726-32 |
| Output Volume | 1 (1-5) | Dtmf Payload Type | 101 (96-127) |
| G729 Payload Length | 10ms | Signal Standard | China |
| CallerID Tx Mode | Bellcore FSK(US) | Fax Mode | T.38 |
| Flashhook Min Time | 200 (>=50ms) | Flashhook Max Time | 800 (<=1000ms) |
| VAD | <input type="checkbox"/> | | |

- CODEC: select the prefer CODEC; support G711A, G711U, G729 and G726-32
- Output Volume: Handset out volume
- Dtmf Payload Type: Choose dtmf payload type
- G729 Payload Length: G729 payload length
- Signal Standard: Support Australia, Brazil, China, Czech, Israel, Japan, South Korea, UK and USA standard
- CallerID Tx Mode: Support Bellcore FSK(US), Brazil DTMF(BR), ETSI FSK(UK) and ETSI DTMF(UK)
- FAX Mode: Support T.38, BYPASS-A and BYPASS-U
- Flashhook Min Time: Set the minimum Flashhook time
- Flashhook Max Time: Set the maximum Flashhook time
- VAD: Enable/disable Voice Activity Detection

2.4.9 Firewall

The screenshot shows the 'Firewall Configuration' page in the VoIP Gateway interface. On the left is a sidebar with the ATCOM logo and a menu with categories: Current State, Network, VoIP, Advance, Dial-Peer, Config Manage, Update, and System Manage. The main content area has the title 'VoIP Gateway' and 'Firewall Configuration'. At the top, there are two checkboxes: 'In_access Enable' and 'Out_access Enable', with an 'APPLY' button below them. Below this are two tables: 'Firewall Input Rule Table' and 'Firewall Output Rule Table', both with columns for Index, Deny/Permit, Protocol, Src Addr, Src Mask, Src Port Range, Des Addr, Des Mask, and Des Port Range. Underneath is the 'Firewall Set' section, which includes a table with columns for Input/Output, Deny/Permit, Protocol Type, Src Addr, Src Mask, Src Port Range, Des Addr, Des Mask, and Des Port Range, and an 'Add' button. At the bottom is the 'Rule Delete' section, with a table for Input/Output, Index To Be Deleted, and a 'Delete' button.

- In_access Enable: Select it to Enable in_ access rule.
- Out_access enable: Select it to Enable out_ access rule.
- Firewall Input Rule Table: Firewall input rule
- Firewall Output Rule Table: Firewall output rule
- Input/output: Specify current adding rule by selecting input rule or output rule.
- Deny/Permit: Specify current adding rule by selecting Deny rule or Permit rule.
- Protocol Type: Filter protocol type. You can select TCP, UDP, ICMP, or IP.
- Port Range: Set the filter Port range.
- Src Addr: Set source address. It can be single IP address, network address, complete address 0.0.0.0, or network address similar to *.*.*.0.
- Dest Addr: Set the destination address. It can be IP address, network address, complete address 0.0.0.0, or network address similar to *.*.*.*.
- Src Mask: Set the source address' mask. For example, 255.255.255.255 means just point to one host; 255.255.255.0 means point to a network which network ID is C type.
- Des Mask: Set the destination address' mask. For example, 255.255.255.255 means just point to one host; if set to 255.255.255.0 means point to a network which network ID is C type.

2.4.10 VPN

The screenshot shows a web interface for configuring a VPN tunnel. The title is "VPN Tunnel". The form includes the following sections:

- VPN IP:** A text input field containing "0.0.0.0".
- VPN Mode:** A section with a radio button selected for "L2TP" and an unchecked checkbox for "Enable VPN".
- L2TP VPN Settings:** A table with four rows and two columns:

| | | | |
|-----------------|----------------------|---------------|----------------------|
| VPN Server Addr | <input type="text"/> | VPN User Name | <input type="text"/> |
| VPN Password | <input type="text"/> | | |
| | | | |
| | | | |

An "APPLY" button is located at the bottom center of the form.

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


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

VPN Mode: Only support L2TP VPN(without IPSec) and whether to enable VPN.

L2TP VPN Settings:

- VPN Server Addr: register to the address of VPN server
- VPN User Name: L2TP VPN username
- VPN Password: L2TP VPN password

Dial-Peer Settings



VoIP Gateway

Current State

Network

VoIP

Advance

- DHCP Server
- NAT
- Net Service
- QOS
- Digital Map
- Call Service
- MMI Filter
- Audio Settings
- firewall
- VPN

Dial-Peer

Config Manage

Update

System Manage

Dial-Peer

| Number | Destination | Port | Mode | Alias | Suffix | Del Length |
|--------|-------------|------|----------|----------|-----------|------------|
| *T | 0.0.0.0 | 0 | LIFELINE | no alias | no suffix | 0 |

| | |
|--------------------------|----------------------|
| Phone Number | <input type="text"/> |
| Destination (optional) | <input type="text"/> |
| Port(optional) | <input type="text"/> |
| Alias(optional) | <input type="text"/> |
| Call Mode | LIFELINE ▼ |
| Suffix(optional) | <input type="text"/> |
| Delete Length (optional) | <input type="text"/> |

| | |
|------|---|
| *T ▼ | <input type="button" value="Delete"/> <input type="button" value="Modify"/> |
|------|---|

Please refer to [“how to use dial rule?”](#) for detail.

2.5 Config Manage

The screenshot shows the ATCOM VoIP Gateway web interface. On the left is a navigation menu with the ATCOM logo at the top. The menu items are: Current State, Network, VoIP, Advance (with sub-items: DHCP Server, NAT, Net Service, QOS, Digital Map, Call Service, MMI Filter, Audio Settings, firewall, VPN), Dial-Peer, Config Manage (highlighted), Update, and System Manage. The main content area is titled 'VoIP Gateway' and 'Config Manage'. It contains three sections: 'Save Config' with a 'Save' button and the instruction 'Press the "Save" button to save the configuration files!'; 'Backup Config' with a 'Right Click here to Save as Config File (.txt)' link and the instruction 'Save all Network and VoIP settings.'; and 'Clear Config' with a 'Clear' button and the instruction 'Press the "Clear" button to Clear the configuration files!'.

Save Config: save current settings.

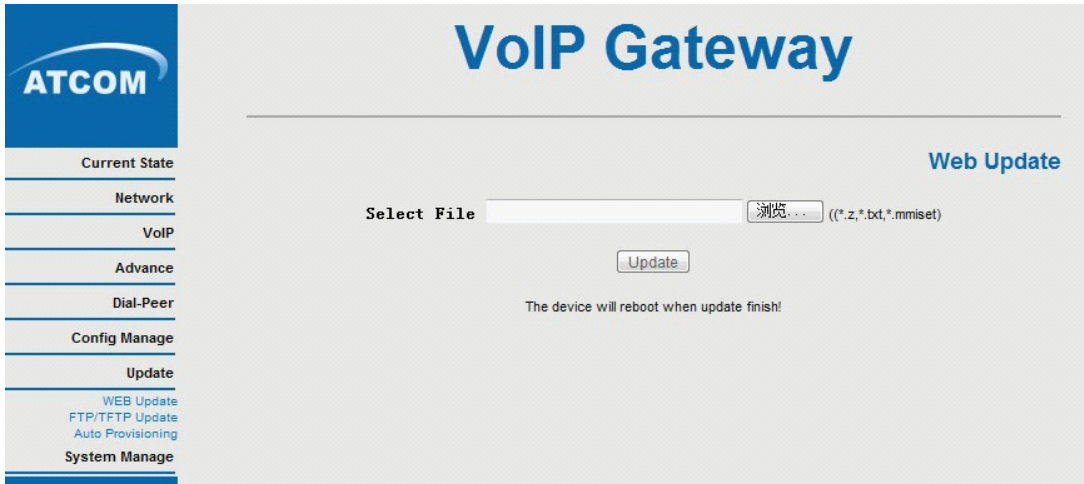
Backup Config: left click,you will see the config.txt file. Right click it you can save the file.

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.

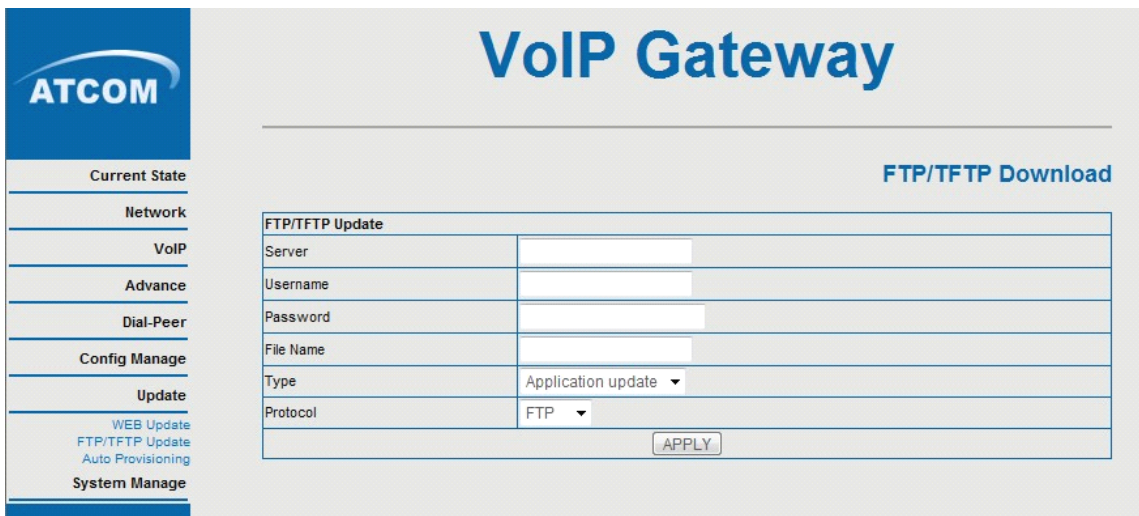
2.6 Update

2.8.1 Web Update:



update gateway's settings or firmware. Firmware file is .z extension when configure file is .txt extension, AG-198 will auto select configure update or firmware update according the extension.

2.8.2 FTP/TFTP Update:



Update the firmware or config.txt file by FTP/TFTP, or export config.txt file to your computer by FPT/TFTP.

2.8.3 Auto update:

AG-198 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 | 2.0001 |
| Server Address | <input type="text" value="0.0.0.0"/> |
| Username | <input type="text" value="user"/> |
| Password | <input type="password" value="....."/> |
| Config File Name | <input type="text"/> |
| Config Encrypt Key | <input type="text"/> |
| Protocol Type | FTP |
| Update Interval Time | <input type="text" value="1"/> Hour |
| Update Mode | Disable |

- 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 update interval time

2.7 System Manage

2.7.1 Account Manage

| User Name | User Level |
|-----------|------------|
| admin | Root |
| guest | General |

| | |
|---------------------------------------|----------------------|
| Add User | |
| User Name | <input type="text"/> |
| User Level | Root ▼ |
| Password | <input type="text"/> |
| Confirm | <input type="text"/> |
| <input type="button" value="Submit"/> | |

| | |
|----------------|---|
| Account Option | <input type="button" value="Delete"/> <input type="button" value="Modify"/> |
| admin ▼ | |

There are two User Level:

- Root: as administrator, can configure all the items
- General: as guest, can only check the status and change network.

2.7.2 Syslog Config:

| | |
|--|--------------------------------------|
| Server IP | <input type="text" value="0.0.0.0"/> |
| Server Port | <input type="text" value="514"/> |
| MGR Log Level | None ▼ |
| SIP Log Level | None ▼ |
| IAX2 Log Level | None ▼ |
| <input type="checkbox"/> Syslog Enable | |
| <input type="button" value="Apply"/> | |

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

2.7.3 Time Set:

| NTP Time set | |
|--------------|---|
| server | 209.81.9.7 |
| timezone | (GMT+08:00)Beijing,Chongqing,Hong Kong,Urumqi |
| timeout | 60 (seconds) |
| NTP | <input checked="" type="checkbox"/> |

Server: type the ip address of time server

Timezone: select correct time zone in list box

Timeout: longest response time for SNTP

Manual Timeset: The time setting

Daylight: Daylight Saving time

NTP: enable NTP time server

2.7.4 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.

The screenshot displays the ATCOM VoIP Gateway web interface. On the left is a blue sidebar menu with the ATCOM logo at the top. The menu items are: Current State, Network, VoIP, Advance, Dial-Peer, Config Manage, Update, System Manage, Account Management, Syslog Config, Time Config, and Logout & Reboot. The main content area has a light gray background with the title 'VoIP Gateway' in large blue font. Below the title is a horizontal line. On the right side of the main area, the text 'Reboot System' is displayed in blue. In the center, there are two instructions: 'Press the "Reboot" button to reboot Phone !' followed by a 'Reboot' button, and 'Press the "Logout" button to Logout IP Phone' followed by a 'Logout' button.

3 IVR

User may pre-config AG-198 gateway using a normal phone connecting to AG-198. 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*/

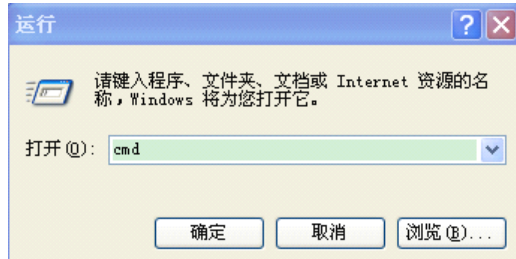
"#*50192.168.1.117" set WAN port IP address
"#*51192.168.1.1" set default gateway IP
"#*52202.112.10.37" set dns server
"#*53255.255.255.0" set netmask, use 255.255.255.0 if no be set
```

POST Mode(safe mode)

AG-198 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.

STEPS:

1. Download nc folder and save it on your computer
2. Connect your PC to the Lan port of AG-198, AG198 LAN ip is 192.168.10.1, and set your computer ip as 192.168.10.X
3. Power up AG-198 and long-press "reset" key (beside PWR) immediately for 10s.
4. Run CMD on your computer



5. Enter the folder which you save nc, for example I save it in disk E, so type e: and enter nc, connect nc with command "nc -u -v -s hostip -p 6666 -n atalp 6666". For example, my computer ip is 192.168.10.31, and press enter several times until you can see POST Menu.

```

c:\ 命令提示符 - nc -u -v -s 192.168.10.31 -p 6666 -n 192.168.10.1 6666
Microsoft Windows XP [版本 5.1.2600]
(C) 版权所有 1985-2001 Microsoft Corp.

C:\Documents and Settings\Administrator>e:

E:\>cd nc

E:\nc>nc -u -v -s 192.168.10.31 -p 6666 -n 192.168.10.1 6666
<UNKNOWN> [192.168.10.1] 6666 (?) open

Unknown command

      POST Menu
=====
[1] TFTP Client Download
[2] Show current MAC address
[3] Clear Config
[4] Exit

Please enter your number:

```

Then if you want to update firmware, you can act as following:

```

c:\ 命令提示符 - nc -u -v -s 192.168.10.31 -p 6666 -n 192.168.10.1 6666

E:\>cd nc

E:\nc>nc -u -v -s 192.168.10.31 -p 6666 -n 192.168.10.1 6666
<UNKNOWN> [192.168.10.1] 6666 (?) open

Unknown command

      POST Menu
=====
[1] TFTP Client Download
[2] Show current MAC address
[3] Clear Config
[4] Exit

Please enter your number: 1
1
Please Input new ones - to discard
Input server IP <192.168.10.2> ==:192.168.10.31
192.168.10.31
Input Target filename <1980526.z> ==:198.z

```


4 FAQ

How many SIP servers may AG-198 register simultaneously?

AG-198 support 2 SIP servers and a 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 AG-198's IP address?

Pick up the handset and then dial “#*111#”, and the AG-198 will promote you its IP address.

How to use AG-198's Lifeline function?

AG-198 supports Lifeline function, you can use the same handset to place PSTN and VoIP calls. First, you need to connect PSTN line to AG198's PSTN port,connect a analog phone to Phone port. Then you can receive voip call and pstn call, and call default by sip. If you want to call by pstn, you can dail * to switch to pstn mode.

If you want to replace the * rule, you need to set up as follow:

----Delete * rule in Digital Map(Advance->Digital Map), and add another rule,for example:1

----Delete * rule in Dial Peer and add a new dial rule: set the phone number as 1, and choose the Lifeline as the Call mode.Then apply.

Then when you want to place a PSTN calls, you can first press 1 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?

AG-198 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 AG-198?”](#)).

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, can 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, IAX2 and Lifeline, SIP means the call will use sip protocol, IAX2 means the call will use iax2 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:

| Number | Call Mode | Destination | Port | Alias | Suffix | Del Length |
|--------|-----------|-----------------|------|-------------------|-----------|------------|
| *T | lifeline | 0.0.0.0 | 0 | no alias | no suffix | 0 |
| 2T | sip | 255.255.255.255 | 5060 | del | no suffix | 1 |
| 3T | sip | 0.0.0.0 | 5060 | del | no suffix | 1 |
| 123 | sip | 0.0.0.0 | 5060 | all:8675583018049 | no suffix | 0 |
| 0T | sip | 0.0.0.0 | 5060 | rep:86 | no suffix | 1 |
| 179 | sip | 192.168.1.179 | 5060 | no alias | no suffix | 0 |

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 will 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 call starts with 0, the first 0 will be replaced by 86. Means that if you dial 075583018049 and AG-198 will send 8675583018049 to your server.

***T rule:** Dial the * and the line will switch to PSTN. Note that you need to set another rule "Prefix Number: *; Length: 1" in the Digital Map. (Refer "How to use AG-198's Lifeline")

Function?)

179 rule: when you dial 179 , the call will 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 Transfer and 3-way Conference calls?

User may set up the configuration in the *Call Service* page to use these value add service.

| Call Service Setting | | | |
|--------------------------------------|-------------------------------------|---------------------|--|
| Hot Line | <input type="text"/> | Warm Line Time | <input type="text" value="0"/> (0-9 seconds) |
| P2P IP Prefix | <input type="text"/> | No Answer Time | <input type="text" value="20"/> (0-60 seconds) |
| Do Not Disturb | <input type="checkbox"/> | Accept Any Call | <input checked="" type="checkbox"/> |
| Enable Call Transfer | <input checked="" type="checkbox"/> | Ban Outgoing | <input type="checkbox"/> |
| Enable Three Way Call | <input checked="" type="checkbox"/> | Enable Call Waiting | <input checked="" type="checkbox"/> |
| <input type="button" value="APPLY"/> | | | |

| Black List | | | |
|----------------------|------------------------------------|---|---------------------------------------|
| <input type="text"/> | <input type="button" value="Add"/> | <input type="text" value="Black List"/> | <input type="button" value="Delete"/> |

| Limit List | | | |
|----------------------|------------------------------------|---|---------------------------------------|
| <input type="text"/> | <input type="button" value="Add"/> | <input type="text" value="Limit List"/> | <input type="button" value="Delete"/> |

➤ Call Transfer:

Check the *Enable Call Transfer*.

If A is the AG-198 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 the *Enable Three Way Call*

Only sip protocol support this function .Assume A is the AG-198 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

AG-198 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

VoIP Gateway

QoS Configuration

<

| | | | |
|--|--------------------------------|----------------------|----------------|
| <input type="checkbox"/> VLAN Enable | | | |
| <input checked="" type="checkbox"/> VLAN ID Check Enable | Voice/Data VLAN differentiated | Undifferentiated | |
| <input checked="" type="checkbox"/> DiffServ Enable | DiffServ Value | 0x b8 | |
| Voice VLAN ID | 256 (0 - 4095) | Data VLAN ID | 254 (0 - 4095) |
| Voice 802.1P Priority | 0 (0 - 7) | Data 802.1P Priority | 0 (0 - 7) |

Or

VoIP Gateway

QoS Configuration

<

| | | | |
|---|--------------------------------|----------------------|----------------|
| <input checked="" type="checkbox"/> VLAN Enable | | | |
| <input type="checkbox"/> VLAN ID Check Enable | Voice/Data VLAN differentiated | Undifferentiated | |
| <input checked="" type="checkbox"/> DiffServ Enable | DiffServ Value | 0x b8 | |
| Voice VLAN ID | 256 (0 - 4095) | Data VLAN ID | 254 (0 - 4095) |
| Voice 802.1P Priority | 0 (0 - 7) | Data 802.1P Priority | 0 (0 - 7) |

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:

VoIP Gateway

QoS Configuration

<

| | | | |
|---|----------------|--------------------------------|----------------------|
| <input checked="" type="checkbox"/> VLAN Enable | | | |
| <input type="checkbox"/> VLAN ID Check Enable | | Voice/Data VLAN differentiated | Tag differentiated ▾ |
| <input type="checkbox"/> DiffServ Enable | | DiffServ Value | 0x b8 |
| Voice VLAN ID | 256 (0 - 4095) | Data VLAN ID | 254 (0 - 4095) |
| Voice 802.1P Priority | 0 (0 - 7) | Data 802.1P Priority | 0 (0 - 7) |

Submit

Or

VoIP Gateway

QoS Configuration

<

| | | | |
|--|----------------|--------------------------------|----------------------|
| <input checked="" type="checkbox"/> VLAN Enable | | | |
| <input checked="" type="checkbox"/> VLAN ID Check Enable | | Voice/Data VLAN differentiated | Tag differentiated ▾ |
| <input type="checkbox"/> DiffServ Enable | | DiffServ Value | 0x b8 |
| Voice VLAN ID | 256 (0 - 4095) | Data VLAN ID | 254 (0 - 4095) |
| Voice 802.1P Priority | 0 (0 - 7) | Data 802.1P Priority | 0 (0 - 7) |

Submit

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:

VoIP Gateway

QoS Configuration

<

| | | | |
|--|----------------|--------------------------------|----------------------|
| <input checked="" type="checkbox"/> VLAN Enable | | | |
| <input checked="" type="checkbox"/> VLAN ID Check Enable | | Voice/Data VLAN differentiated | Tag differentiated ▾ |
| <input checked="" type="checkbox"/> DiffServ Enable | | DiffServ Value | 0x b8 |
| Voice VLAN ID | 256 (0 - 4095) | Data VLAN ID | 254 (0 - 4095) |
| Voice 802.1P Priority | 0 (0 - 7) | Data 802.1P Priority | 0 (0 - 7) |

Submit

Or

VoIP Gateway

QoS Configuration

<

| | | | |
|---|----------------|--------------------------------|----------------------|
| <input checked="" type="checkbox"/> VLAN Enable | | | |
| <input type="checkbox"/> VLAN ID Check Enable | | Voice/Data VLAN differentiated | Tag differentiated ▾ |
| <input checked="" type="checkbox"/> DiffServ Enable | | DiffServ Value | 0x b8 |
| Voice VLAN ID | 256 (0 - 4095) | Data VLAN ID | 254 (0 - 4095) |
| Voice 802.1P Priority | 0 (0 - 7) | Data 802.1P Priority | 0 (0 - 7) |

4. if “Data/Voice VLAN differentiated” is Data untagged and diffServ are both enable. Then device will undistinguish the data to VLAN ID setting. As below:

VoIP Gateway

QoS Configuration

<

| | | | |
|---|----------------|--------------------------------|-----------------|
| <input checked="" type="checkbox"/> VLAN Enable | | | |
| <input type="checkbox"/> VLAN ID Check Enable | | Voice/Data VLAN differentiated | data Untagged ▾ |
| <input checked="" type="checkbox"/> DiffServ Enable | | DiffServ Value | 0x b8 |
| Voice VLAN ID | 256 (0 - 4095) | Data VLAN ID | 254 (0 - 4095) |
| Voice 802.1P Priority | 0 (0 - 7) | Data 802.1P Priority | 0 (0 - 7) |

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, AG-198 will have strict requirement on the VLAN, it won't handle any packets with different VLAN ID. If “VLAN ID check” is disable, AG-198 will handle the packets even from different vlan ID. Please notice that VLAN ID check is enable in default.