AT-530 User Manual

1 AT-530 Features

1.1 Appearance

1.2 Interface

Power: Output Power:12VDC,500mA.
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

1.4 Software

- Support two sip accounts at the same time.
- Redundancies server support.
- NAT, Firewall.
- DHCP client and server.
- Support PPPoE, (used for ADSL, cable modem connecting).
- Support major G7.xxx CODEC.
- VAD,CNG.

- G.168 compliant 32ms echo cancellation
- Tone generation and Local DTMF re-generation according with ITU-T
- E.164 dial plan and customized dial rules
- Hotline.
- Speed Dial
- Call Forward, Call Transfer, 3-way conference calls
- Record
- Caller ID display
- DND(Do Not Disturb),Black List,Limit List
- Upgrade firmware through FTP, TFTP or HTTP,.
- Web management.
- Telnet remote management.
- 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, G7231 5.3/6.3 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

1.6 Compliant Standard

- CE: EN55024,EN55022
- FCC part15
- RoHS

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

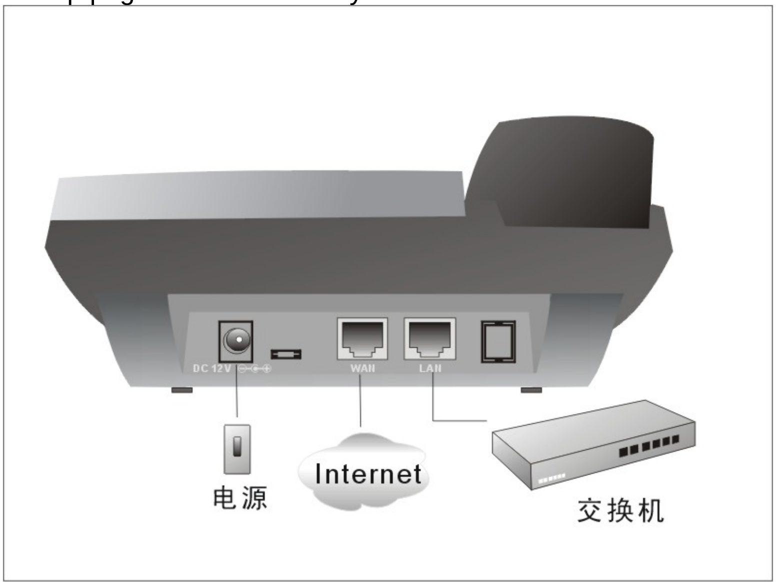
1.8 Package

- Size: 338×220×85mm
- Packing List
  - ✓ AT-530 IP phone
  - ✓ Power adaptor (out put 12v ,500mA)
  - ✓ Manual CD

1.9 Installation

Use ethernet cable to connect AT-530'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 AT-530, the default username and password is admin/admin for administrator and guest/guest for guest.

Set up page for VoIP use only:



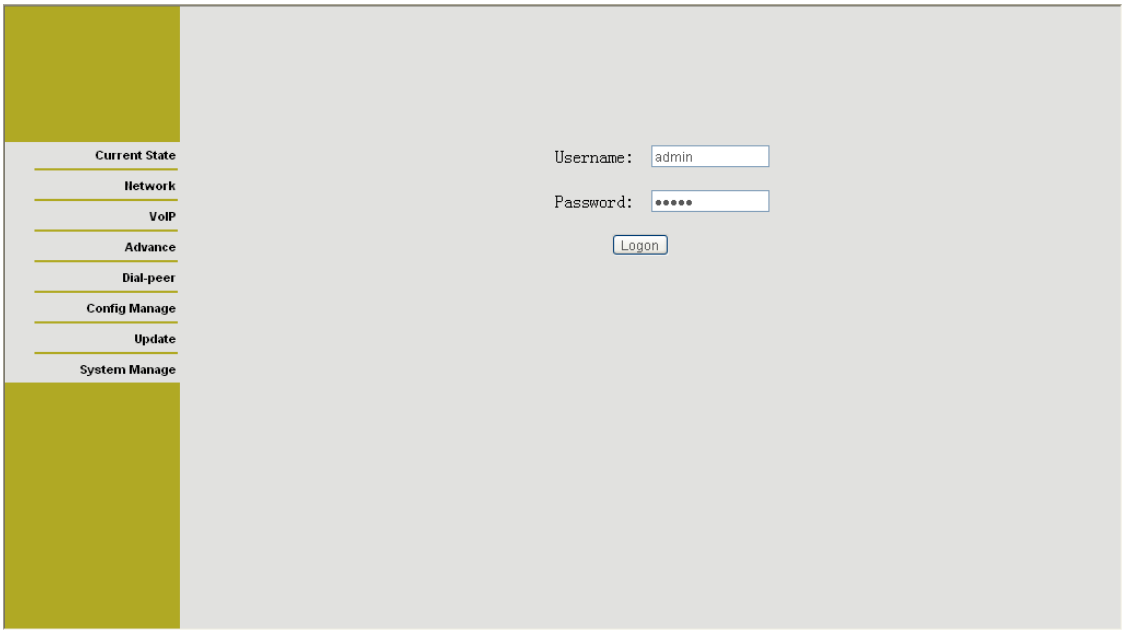
2 Web Configuration

2.1 Access Web setting page

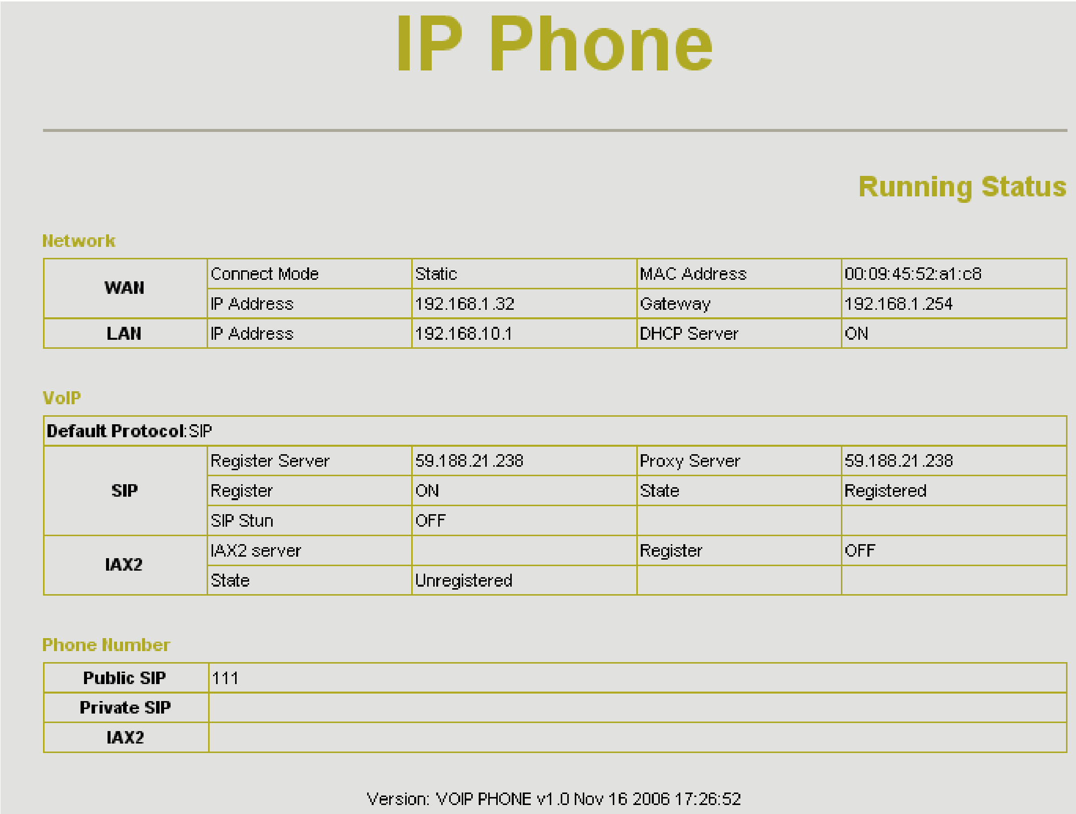
Enter AT-530 IP address in the web browser and press ENTER to go to the log on page, and key in the username and password to access AT-530 setting page.

Default username and password is:

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



2.2 Current state



IP Phone

Running Status

Network

WAN	Connect Mode	Static	MAC Address	00:09:45:52:a1:c8
	IP Address	192.168.1.32	Gateway	192.168.1.254
LAN	IP Address	192.168.10.1	DHCP Server	ON

VoIP

Default Protocol:SIP				
SIP	Register Server	59.188.21.238	Proxy Server	59.188.21.238
	Register	ON	State	Registered
	SIP Stun	OFF		
IAX2	IAX2 server		Register	OFF
	State	Unregistered		

Phone Number

Public SIP	111
Private SIP	
IAX2	

Version: VOIP PHONE v1.0 Nov 16 2006 17:26:52

This page shows AT-530's running state.

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

VoIP part show the working state of VoIP, you can see whether AT-530 has registered the public sip server

Phone Number public sip and private sip 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 AT-530's IP address in the *IP Address*;
  ----Set netmask in the *Netmask* 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, AT-530 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*.
AT-530 will automatically obtain WAN port network information from your ITSP if PPPoE setting and the setup are correct.

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

VoIPon www.voipon.co.uk sales@voipon.co.uk Tel: +44 (0)1245 808195 Fax: +44 (0)1245 600030

2.3.2 LAN Config

Bridge Mode: Enable this option to switch to bridge mode. IP phone 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.

Highest Priority of Voice Quality: Enable this option to guarantee voice quality. If there is high flux in the LAN port, AT-530 will limit the stream rate.

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, default port is 5060

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 AT-530 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. AT-530 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, AT-530 will periodically detect if the SIP server is available according this setting.

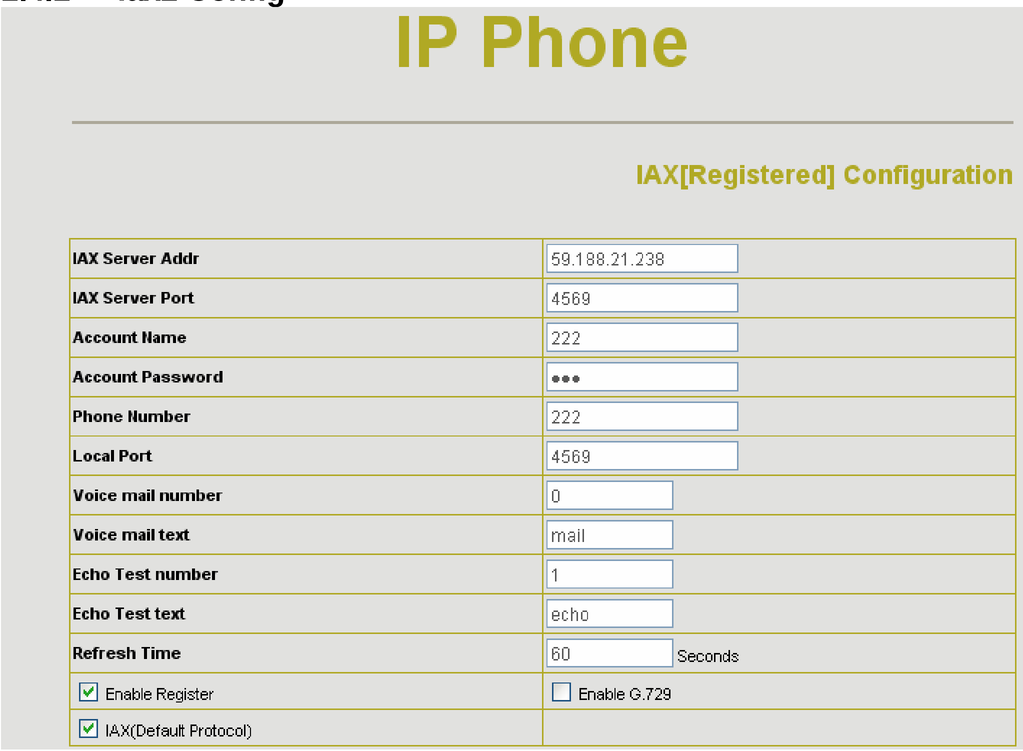
User Agent:

Encrypt Key: The particular service system decrypts of the key , matching with the server Type usage, the key provide by the particular service system supplier, default is empty

Server Type: The particular service system supplier carries out the sign and speeches to

encrypt, default is common
DTMF Mode: DTMF signal sending mode: support RFC2833, DTMF_RELAY (inband audio) and SIP info
RFC Protocol Edition: Current AT-530 SIP version. 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. AT-530 won't sent register info to SIP server if disable register.

2.4.2 Iax2 Config



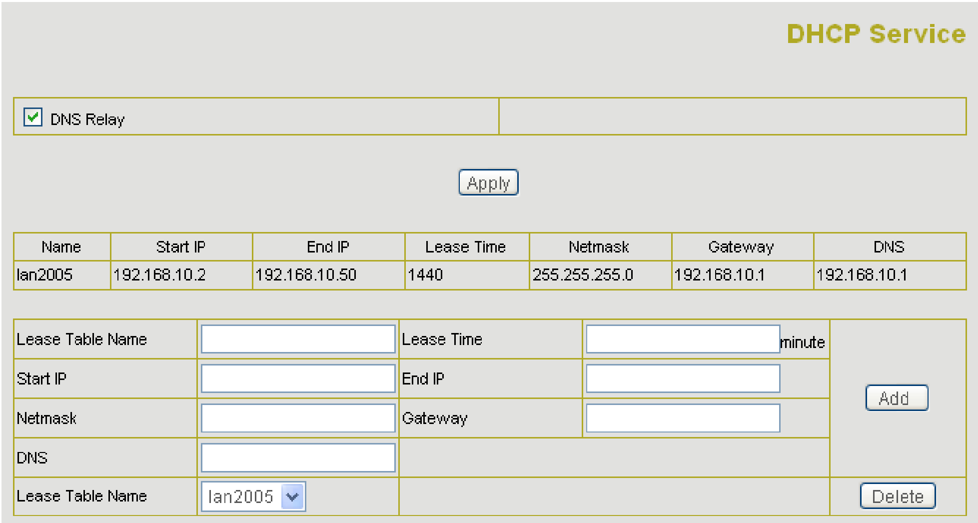
Setting page of public IAX server:
IAX Server Addr: Register address of public IAX server
IAX Server Port: Register port of public IAX server, default port is 4569
Account Name: Username of your SIP account (Always the same as the phone number)
Account Password: Password of your IAX account.
Local port: 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



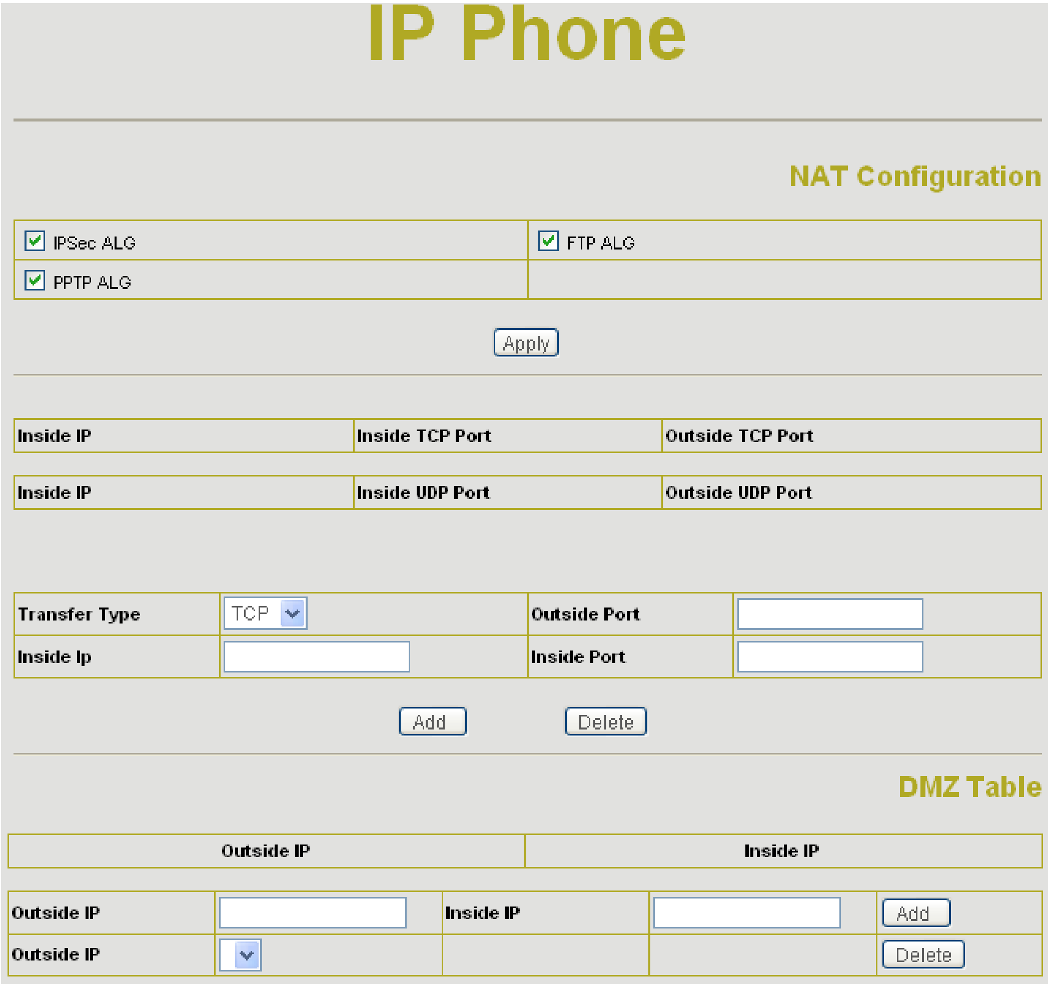
DHCP server manage page.
User may trace and modify DHCP server information in this page.

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 AT-530 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



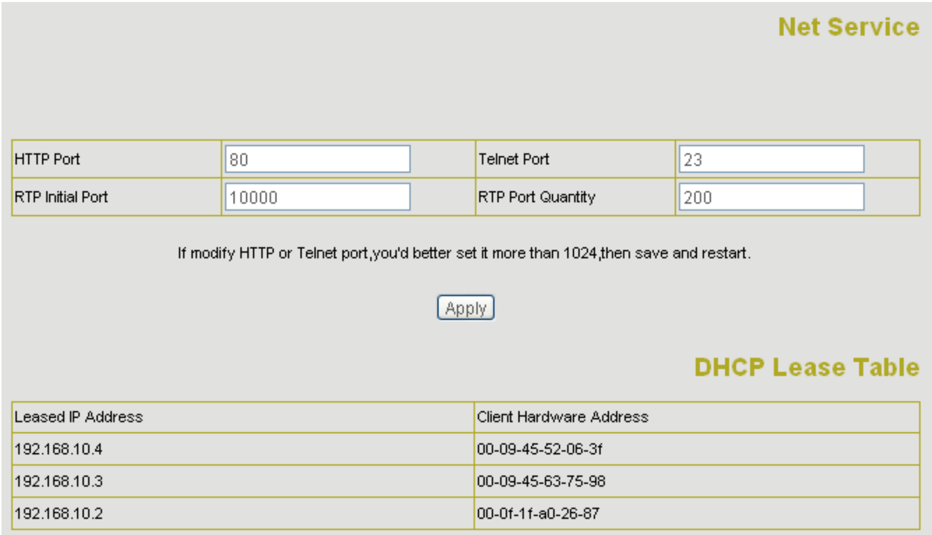
Advance NAT setting. Maximum 10 items for TCP and UDP port mapping.

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

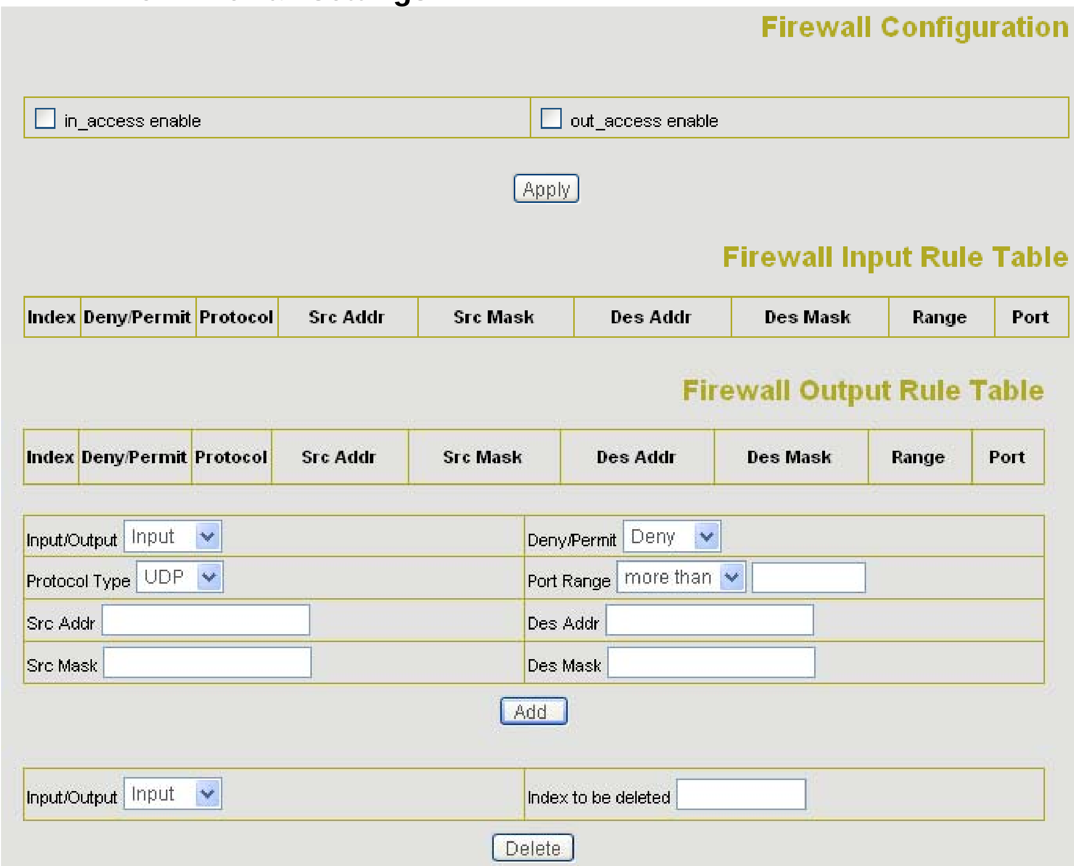
Notice:

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 Firewall settings



Firewall setting page. User may set up firewall to prevent unauthorized Internet users from accessing private networks connected to the Internet (input rule), or prevent unauthorized private network devices to access the internet.

Access list support two type limits: input_access limit or output_access limit. Each type support 10 items maximum.

AT-530 firewall filter is base WAN port. So the source address or input destination address should be WAN port IP address.

Configuration:

in_access enable enable in_access rule

out_access enable enable out_access rule

Input/Output: specify current adding rule is input rule or output rule.

Deny/Permit: specify current adding rule is deny rule or permit rule.

Protocol Type: protocol using in this rule: TCP/IP/ICMP/UDP.

Port Range: port range if this rule

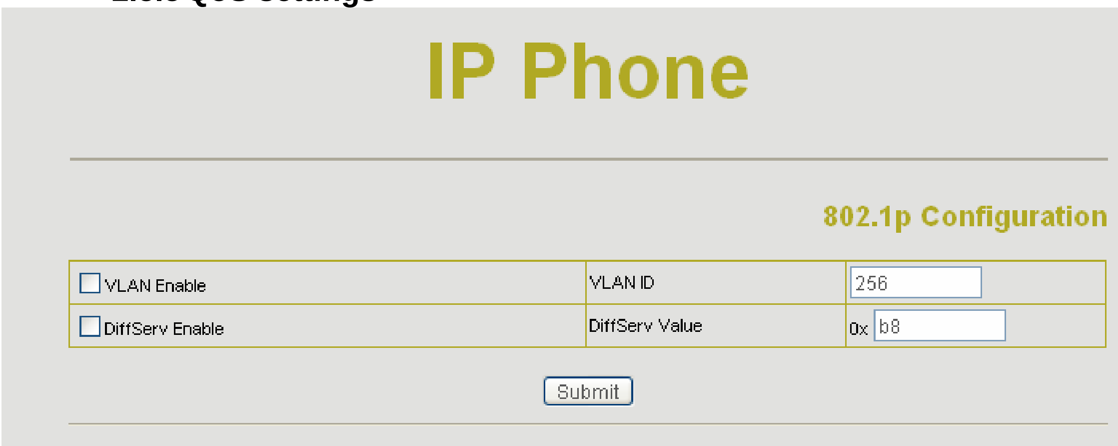
Src Addr: source address. Can be single IP address or network address.

Dest Addr: destination address. Can be IP address or network address.

Src Mask: source address mask. Indicate the source is dedicate IP if set to 255.255.255.255. Otherwise is network ID

Des Mask: Destination address mask. Indicate the source is dedicate IP if set to 255.255.255.255. Otherwise is network ID

2.5.5 QoS settings



AT-530 IP phone implement QoS based on 802.1p, The QoS is used to mark the network communication priority in the data link/MAC sub-layer. AT-530 will sorted the packets using the QoS and sends it to the destination.

VLAN Enable: If enable the VLAN service, the second layer will realize separate voice, signal and data transmission. To realize separate voice and data transmission by dispose for IP precedence of ToS area of voice transmission. To reach upper layer switch or router have priority to transfer voice transmission. (The prerequisite is the upper layer switch or router have to identify ToS area.)

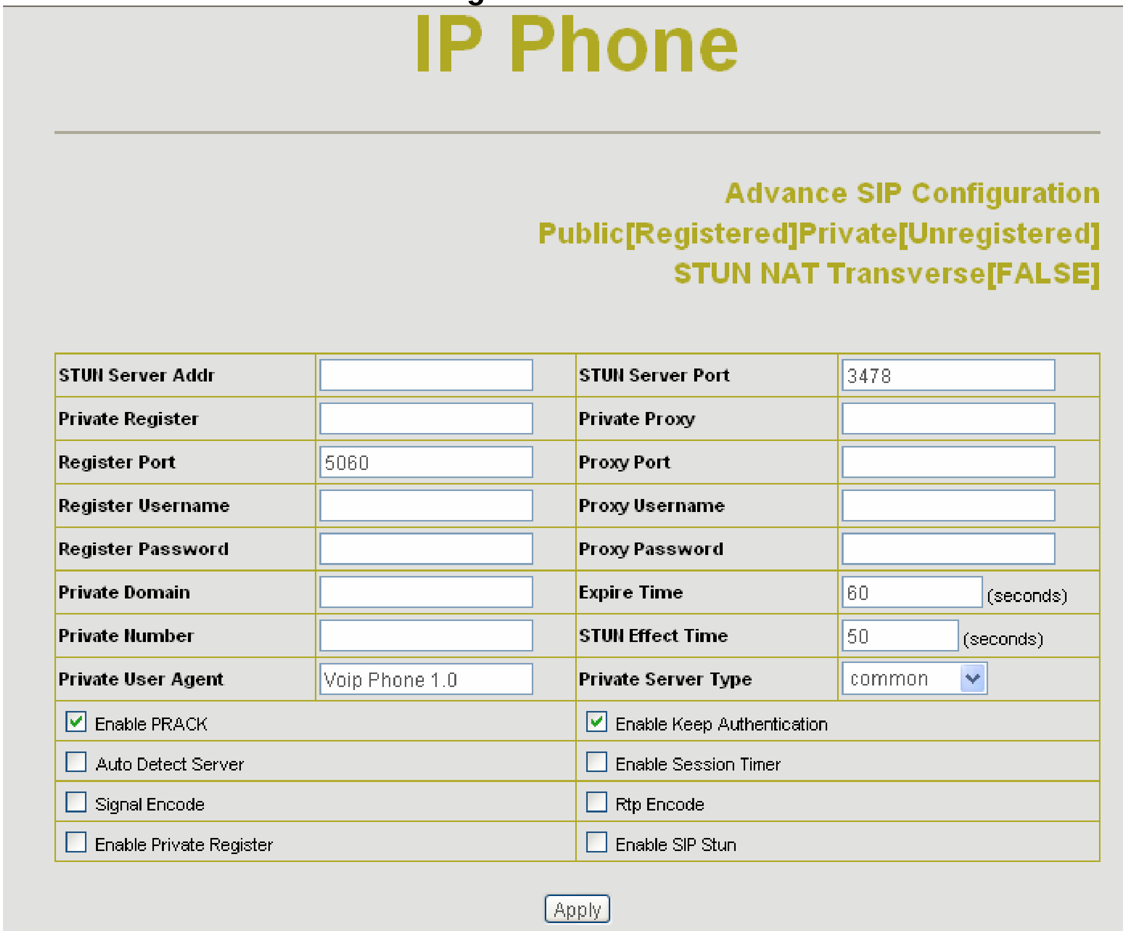
VLAN ID: Dispose VLAN ID is add a Tag header after realize enable the VLAN function. The realized voice packets transfer at the same VLAN. The prerequisite is it must the same as VLAN of upper switch. The value range are 1~4094.

DiffServ Enable: If enable the VLAN service, it indicates use DSCP mode to realize three layers QoS. This moment, the DSCP of SIP signals which between IP Phone and MGC. It will use Class Selector 5 (The value is 0xA0). And the DSCP of mediums information (In RTP packets) would be used the values of DiffServ Value field.

DiffServ Value: The 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 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.

2.5.6 Advance SIP settings



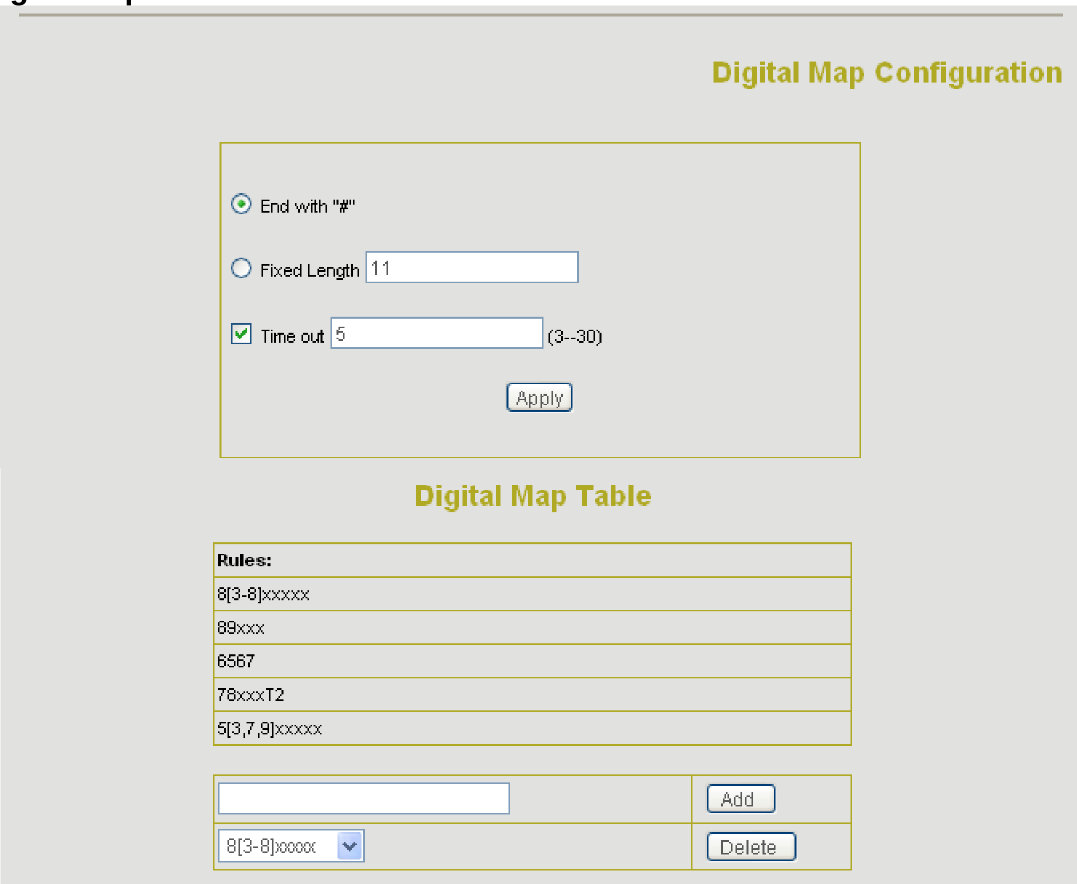
This page is used to set the private sip server, stun server, and back up sip server information.

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.
Enable PRACK: Whether to make gateway or IP phone support Prack function in SIP , we suggest you keeping default setting
Enable Keep Authentication: registering signal together with the authentication information. If enable it, the server will confirm the registering and send back the confirmation massage directly instead of requesting the terminals to send authentication information if needed.
Auto Detect Server: Whether to enable the function of auto detecting the server. With this function your ATA and IP phone will send information to auto detect the server at every period of time. If find the server is not available it will try to register the server again.
Enable Session Timer: Whether to enable te RFC4028
Signal Encode: Wherther to enable the signal encrypt
Rtp Encode: Whether to enable the voice encrypt
Enable Private Register: Whether to enable the second SIP Server to register
Please refer to **SIP_Config** for the setting for how to set the public alter server.
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_Config**

Digital Map



Digit map is a set of rules to determine when the user has finished dialing.
AG-188 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-188 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]

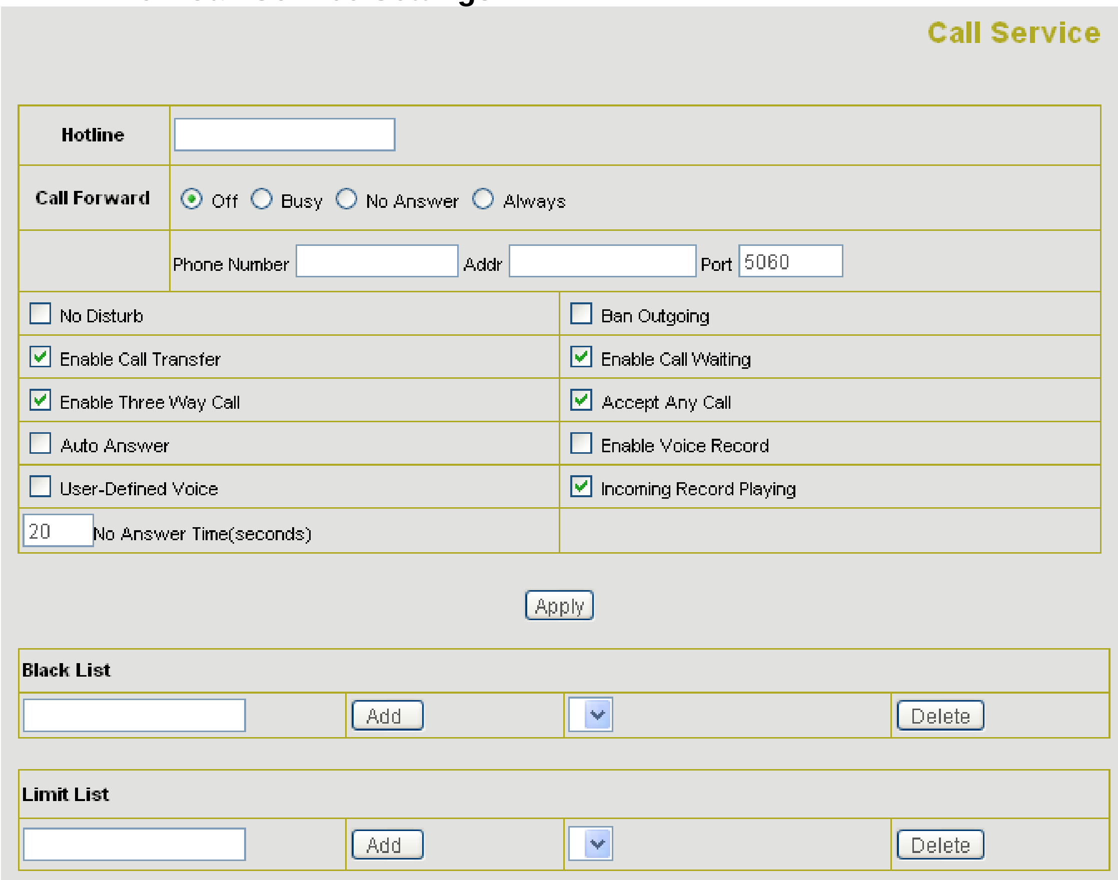
x   represents any one digit between 0~9

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. AT-530 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, AT-530 refuse the incoming call when the called number is different from AT-530's phone number.

No Answer Time: no answer call forward time setting.

Auto Answer: Enable/disable auto answer function.

Enable Voice Record: Enable/disable answering machine function. Please refer to **Record Function** for detail.

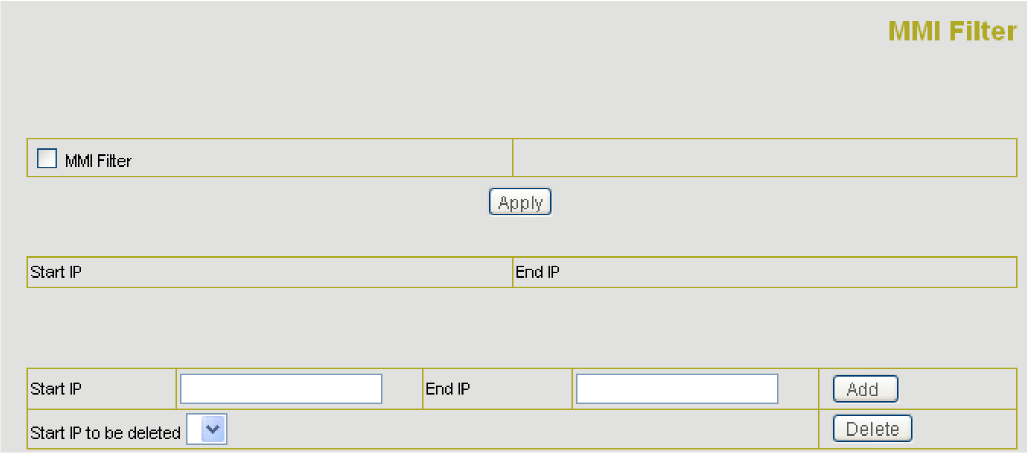
User-defined Voice: Use customized greeting message.

Incoming Record Playing: simultaneously play the message when recording.

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

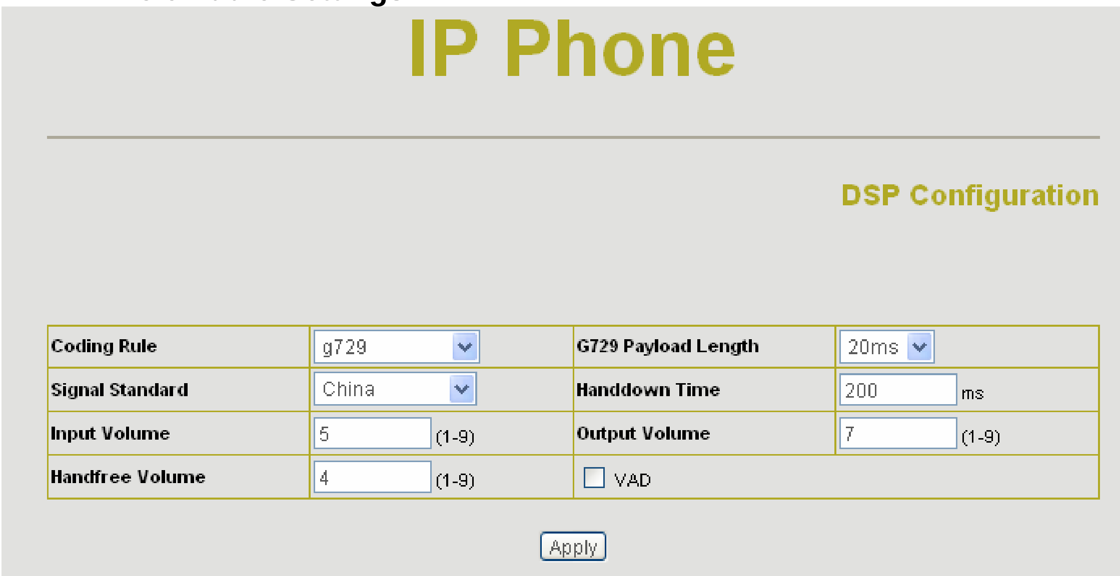
Limit List: outgoing calls with these phone numbers will be refused

2.5.8 MMI Filter



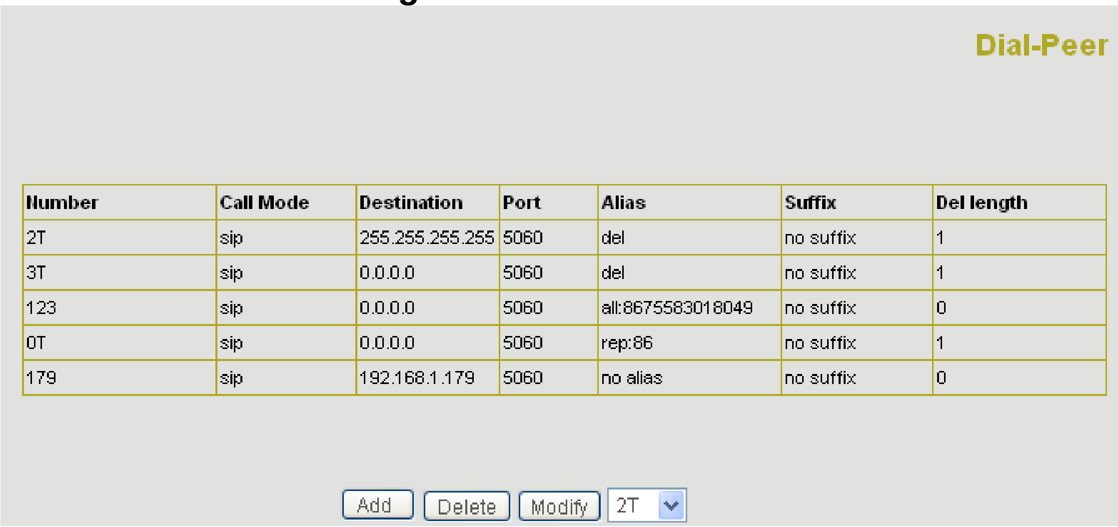
MMI filter is used to make access limit to AT-530 IP phone.
When MMI filter is enable. Only IP address within the *start IP* and *end IP* can access AT-530 IP phone.

2.5.9 Audio Settings



CODEC: select the prefer CODEC; support ulaw, alaw,G729 and G7231 5.3/6.3
Signal Standard: Signal standard for different area.
Input Volume: Handset in volume.
Output Volume: Handset out volume.
Handfree Volume: Hand free volume
Handdown Time: hand down detect time.
G729 Payload Length: G729 payload length
VAD: Enable/disable Voice Activity Detection

2.6 Dial-Peer Settings



Number	Call Mode	Destination	Port	Alias	Suffix	Del length
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

Please refer to How to use dial rule for detail.

2.7 Config Manage

Save Config: save current settings.
Clear Config: restore to default settings.
Backup Config: Backup the config file, via point the right key of mouse-→ save target as….-→will pop a save window, then type the config file name in the File name(the file type is text file)

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.

2.8 Update

Web Update:

Update IP phone's settings or firmware. Firmware file is .z extension when configure file is .cfg extension, AT-530 will auto select configure update or firmware update according the extension.

TFTP/FTP Update:

upload/download the configure file with FTP or TFTP server. or download firmware from FTP or TFTP server

Back up configure file to your FTP/TFTP server.

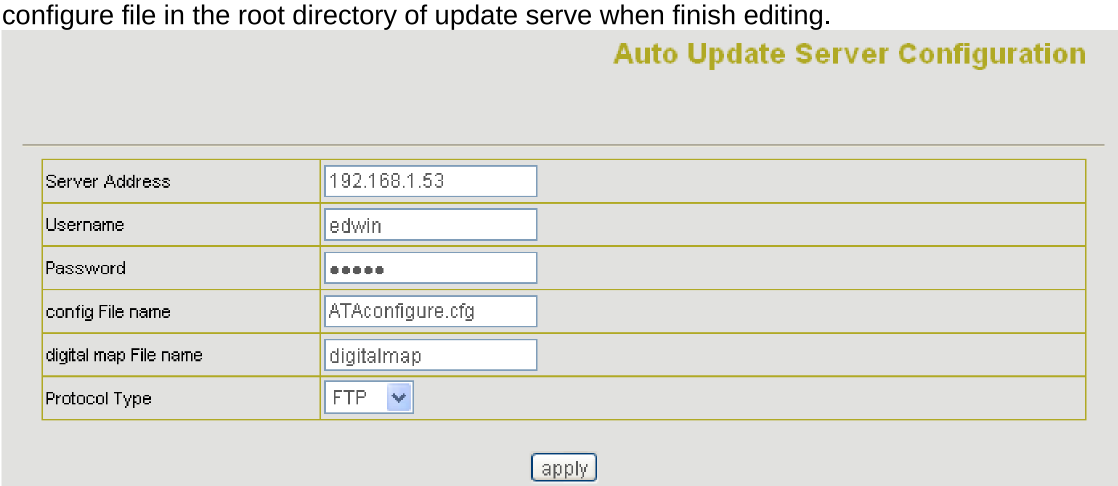
configure use .cfg extension.

The Type includes two parts of config file export and config file import

Config file export:export the config file

Config file import:import the config file

Auto update: AT-530 IP phone 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.



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

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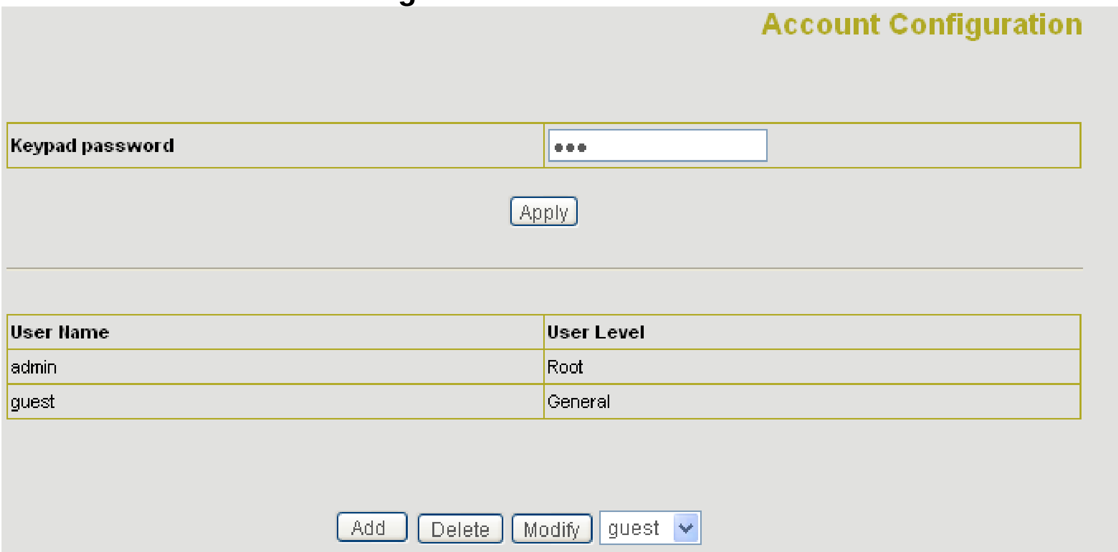
2.9 System Manage

2.9.1 Account Manage



Set web access account or keypad password of AT-530.

2.9.2 Phone Book:

User may set contacts in this page, and the contacts will be saved in the memory. Then using the Pbook, Vol+,Vol-,Menu/OK and Exit keys to choose your friend in the contacts and then press # to call out.

2.9.3 Syslog Config:

2.9.4 Time Set:

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

2.9.5 Reboot:

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

3 Use keypad configure AT-530 IP phone

3.1 Keypad function

User can configure AT-530 through its keypad. List below is the keypad function

Keypad	**Mode**	**Function/Display**
Idle mode	----	show current time
Sysinfo	Idle mode	circularly show phone number,wan ip, gateway info
Menu/OK	Idle mode	enter config mode, default password 123
	config mode	confirm or enter sub-menu
Exit	config mode	exit
Up	Calling mode	volume up (Max:9)
	config mode	Page up
Down	Calling mode	volume down (Min:1)
	config mode	Page down
Del	Calling mode	Delete digits
	config mode	Delete digits
Mute	Calling mode	Mute
Out call	Idle mode	Outgoing call menu
In call	Idle mode	Incoming call menu
Record	Idle mode	Enter record menu, usage refer FAQ
Pbook	Idle mode	Enter Phone book set up
Handfree	Calling mode	Handfree
0 – 9	Calling mode	Digits 0~9
	config mode	Hit quickly to switch between numeric or alphabetic
*	Calling mode	Use in 3-way conference call.
	config mode	Use as “.” In the ip address setting
#	Calling mode	Use as end key of dialing or the dial number
Hold	Calling mode	Hold, detail refer value add service
FWD	Calling mode	Transfer, detail refer value add service
Redial	Calling mode	Redial key
Send	Calling mode	call key
No.1~No.9	Idle mode	Speed dial key

3.2 Keypad Menu

User may use **SET, Menu/ok, Exit, Vol+**, **Vol-** to config AT-530 detail setting. Press **Menu/ok** to enter config mode, and the default password is 123.

Below list the keypad menu of AT-530

AT-530 Keypad Menu			
Level 1	Level 2	Level 3	Level 4
Network	LAN	Bridge Mode	
		IP	
		Netmask	
		DHCP Server	
		NAT	Switch
			FTPalg
			IPSec alg
			PPTPalg
	WAN	Status	
		Static Net	1. IP
			2. NetMask
			3. Gateway
			4. DNS
			5. DNS2
		PPPoE	User name
			Password
		QoS	
Call Feature	Phone-number		
		Public SIP	
		Private SIP	
	Limit-List	Current	
		ADD	
		DEL	
	Black-List	Current	
		ADD	
		DEL	
	FastCall		
	Three Call		
	Call-Transfer		
	Call-Waiting		
	Call-Forward	Condition	
		SIP	Transfer Num

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			Transfer IP
			Port
	Dial-Rule	End With #	
		Fixed Length	Switch
			Length
SIP	Reg Status	Public Reg	
		Private Reg	
	Detect-server		
	Dtmf-mode		
	Interval-time		
	Swap-server		
	RFC-version		
	Signal-Port		
	Stun	Switch	
		Addr	
		Port	
		Expire Time	
DSP	Codec		
	Handdown-time		
	Dtmf-Volume		
	Input-volume		
	Output-Volume		
Other Setting	Syslog	Switch	
		Server-IP	
		Server-Port	
4. System	1. Save		
	2. Reboot		
	3. Set Default		

4 Telnet Console

4.1 Introduce

4.1.1 Basic structure

User may use telnet command to access and manage IP phone.

AT-530 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, AT-530 support following commands:

Command	Function	Example
chinese	Set to Chinese UI	#chinese
clear	Clear telnet screen	#clear
english	Set to English UI	#english
exit	Return to parent-node	#exit
help	Show help info Show sub-nodes and local command	1. #help ping 2. #help
history	Show command history	#history
logout	Exit	#logout
ping	Ping command, use to check network,	#ping www.google.com
tree	Print tree structure of current command	#tree
who	Show current user	#who
write	Save setting to flash	#write

5 Tree Structure

5.1.1 account

path: <account>#

[stop]start Syslog ---syslog [no] start
Configure Syslog server address and port ---syslog server –ip x.x.x.x _port xxx
Example: #<config-account-syslog>#server –ip 202.112.20.10
Show syslog settings ---syslog show
Show all account settings ---show

5.1.2 config

- accesslist firewall config

path: <config-accesslist>#

add firewall rule ---entry –I/O xxx –P/D xxx –proto xxx –srcaddr x.x.x.x –srcmask x.x.x.x–desaddr x.x.x.x –desmask x.x.x.x –portrange xxx –portnum xxx
Example:<config-accesslist>#entry –I/O input –P/D deny –proto udp –straddr 202.112.10.1 –srcmask 255.255.255.0 –desaddr 210.25.132.1 –desmask 255.255.255.0 –portrange neq –portnum 5060
delete firewall rule ---no entry –I/O xxx –index xxx
Example :<config-accesslist>#no entry –I/O input –index 1
Show firewall settings ---show
[disable] enable input filter ---[no]in-access
[disable] enable output filter ---[no]out-access

- DHCP

path: <config-dhcp>#

add DHCP rule ---entry –name xxx –startip x.x.x.x –endip x.x.x.x –netmask x.x.x.x –gateway x.x.x.x –dnsserver x.x.x.x _time xxx
Example:<config-dhcp>#entry –name lan2004 –startip 192.168.1.2 –endip 192.168.1.254 –netmask 255.255.255.0 –gateway 192.168.1.1 –dnsserver 192.168.10.18
delete DHCP rule ---no entry –name xxx
Example: <config-dhcp>#no entry –name lan2004
Show DHCP settings ---show
[disable]enable DNS-relay ---[no]dns-relay

- dialrule

path: <config-dialrule>#

[disable] enable End with # ---[no]endchar
Set end with fix length ---fixlen xxx
Disable end with fix length ---no fixlen
Set timeout to send ---timeout-send xxx
Disable timeout to send ---no timeout-send
Add digital map ---entry –prefix xxx –length xxx
Example: <config-dialrule>#entry –prefix 010 –length 11
Delete digital map rule ---no entry –prefix xxx
Example: <config-dialrule>#no entry –prefix 010
Show current digital map ---show

- LAN interface settings

path: <config-interface-fastethernet-lan>#

[disable]enable bridge mode ---[no]bridgemode
[disable]enable DHCP service ---[no]dhcp-server
[disable]enable NAT ---[no]nat

Show current DHCP rules ---dhcpshow
Show LAN port IP address ---ipshow
Show NAT info ---natshow
Change LAN port IP address ---ip –addr x.x.x.x –mask x.x.x.x
Example:<config-interface-fastethernet-lan>#ip –addr 192.168.1.10 –mask 255.255.255.0

- WAN interface settings

path: <config-interface-fastethernet-wan>#
[disable]enable dhcp client ---[no]dhcp
[disable]enable pppoe ---[no]pppoe
[disable]enable QOS ---[no]qos
Set default gateway IP ---gateway x.x.x.x
Clear default gateway IP ---no gateway
Set WAN port IP address ---ip –address x.x.x.x -mask x.x.x.x
Example:<config-interface-fastethernet-wan>#ip –addr 202.112.241.100 –mask 255.255.255.0
You need to reconnect if the WAN port has been changed.
Show WAN port settings ---show

- MMI Filter

path: <config-mmifilter>#
add filter rule ---entry –start x.x.x.x –end x.x.x.x
Example:<config-mmifilter>#entry –start 202.112.20.1 –end 202.112.20.255
Delete filter rule ---no entry –start x.x.x.x
Example:<config-mmifilter>#no entry –start 202.112.20.1
Show filter rule ---show
[disable]enable MMI filter ---[no]start-filter

- NAT settings

path: <config-nat>#
[disable]enable ftp alg ---[no]ftpalg
[disable]enable ipsec alg ---[no]ipsecalg
[disable]enable pptp alg ---[no]pptpalg
Add TCP mapping rule ---tcp-entry –ip x.x.x.x –lanport xxx –wanport xxx
Example:<config-nat>#tcp-entry –ip 192.168.1.5 –lanport 1720 –wanport 1000
Delete TCP mapping rule ---no entry –ip x.x.x.x –lanport xxx –wanport xxx
Example:<config-nat>#no tcp-entry –ip 192.168.1.5 –lanport 5060 –wanport 1000
Add UDP mapping rule ---udp-entry –ip x.x.x.x –lanport xxx –wanport xxx
Delete UDP mapping rule ---no udp-entry –ip x.x.x.x –lanport xxx –wanport xxx
Show NAT info ---show

- Netservice

path: <config-netservice>#
Set DNS address ---dns -ip x.x.x.x _domain xxx
Example:<config-netservice>#dns –ip 202.112.10.36 _domain voip.com
Set alternate DNS address ---alterdns -ip x.x.x.x _domain xxx
Set hostname ---hostname xxx
Set http access port ---http-port xxx

Show http access setting ---http-port
Set telnet access port ---telnet-port xxx
Show telnet access port ---telnet-port
Set RTP initial port and quantity ---media-port –startport xxx –number xxxx
Example:<config-netservice>#media-port –startport 10000 –number 200
Add route rule ---route –gateway x.x.x.x –addr x.x.x.x –mask x.x.x.x
Example: Arcihfone<config-netservice>#route –gateway 202.112.10.1 –addr 202.112.210.1 –mask 255.255.255.0
Delete route rule ---no route –gateway x.x.x.x –addr x.x.x.x –mask x.x.x.x
Show route info ---route
Show netservice info ---show

- Dial-peer settings

path: <config-pbook>#
[disable]enable calling through GK and proxy ---[no]enableGKandProxy
Add number-IP bond entry ---entry –number xxx –ip x.x.x.x –protocol xxx
Example:<config-pbook>#entry –number 100 –ip 202.112.20.100 –protocol sip

Add number-IP bond and add prefix to the dial number
---entry –number xxx –ip x.x.x.x –protocol xxx _add xxx
Example:<config-pbook>#entry –number 100 –ip 202.112.20.100 –protocol sip _add 123(dial 100 and will send 123100 according this rule)

Add number-IP bond and replace the destination with another number
---entry –number xxx –ip x.x.x.x –protocol xxx _all xxx
Example:<config-pbook>#entry –number 100 –ip 202.112.20.100 –protocol sip _all 123(user dial 100 and gateway will sent 100 instead)

Add number-IP bond and delete the prefix of the destination number
---entry –number xxx –ip x.x.x.x –protocol xxx _del xxx
Example:<config-pbook>#entry –number 1234 –ip 202.112.20.100 –protocol sip _del 2 (dial 1234 will send 34 instead)

Add number-IP bond and replace the prefix with another number
---entry –number xxx –ip x.x.x.x –protocol xxx _rep xxx _length xxx
Example:<config-pbook>#entry –number 1234 –ip 202.112.20.100 –protocol sip _rep 567 _length 2(dial 1234 will send 56734)

Delete dial-peer entry ---no entry –number xxx
Show current dial-peer rules ---show
Set default voip protocol ---default-protocol xxx

- Port settings

path: <config-port># 或<config-port X>#
set accecp relay mode ---accept-relay xxx
set callerid mode ---callerid xxx
disable callerid ---no callerid
config call forward ---callforward –conditon xxx –number xxx –ip xxx –port xxx –protocol xxx
Example:<config-port 0>#callforward –condition busy –number 100 –ip 202.112.10.100 -port 5060 –protocol sip

Disable call forward ---no callforward
[disable]enable call transfer ---[no]calltransfer
[disable]enable call waiting ---[no]callwaiting
Set prefer codec ---codec xxx
Set DTMF gain ---dtmfvolume xxx
Set black list ---in-limit xxx
Show black list ---in-limit
Set input volume ---input xxx
Set outgoing limit list ---out-limit xxx
Show outgoing limit list ---out-limit
Set output volume ---output xxx
[disable]enable outgoing limit ---[no]shutdown out
[disable]enable black list ---[no]shutdown in
[disable]enable outgoing limit and black list ---[no]shutdown
[disable]enable 3-way conference ---[no]threetalk
Show port settings ---show

- PPPoE settings

path: <config-pppoe>#
PPPoE account settings ---auth –user xxx -password xxx
Example:<config-pppoe>#auth –user aaa –password 123456
[disable]enable service settings ---[no]service xxx
Show pppoe settings ---show

- QOS settings

path: <config-qos>#
[delete]add QoS table entry --- [no]entry –addr x.x.x.x –mask x.x.x.x
Example:<config-qos>#entry –addr 202.112.10.1 –mask 255.255.255.0
[disable]enable include QOS table ---[no]include
Show QoS settings ---show

- SIP settings

path: <config-sip>#
[disable]enable registration ---[no] register
[disable]enable auto detect server ---[no] detect-server
Set sip domain ---default-domain xxx
Set DTMF mode ---dtmf-mode xxx
Set auto detect interval time ---interval-time xxx
Set RFC edition ---rfc-version xxx
[disable]enable auto swap server --- [no]swap-server
Set sip account ---number-password –number xxx –password xxx
Set local SIP signal port --- signalport xxx
Set proxy server ---server proxy -ip x.x.x.x _port xxx _user xxx _password xxx
Example:<config-sip-server># proxy ip 210.25.23.22 _port 5060 _user aaa _password 123456
Set register server info ---server register -ip x.x.x.x _port xxx –user xxx _password xxx
Set alter proxy info ---alter-server proxy –ip x.x.x.x _port xxx _user xxx _password xxx

Set alter server info ---alter-server register –ip x.x.x.x _port xxx _user xxx _password xxx
[disable]enable stun server ---stun [no]enable
Set stun detecting interval time ---stun interval-time xxx
Set stun server ip and port ---stun –ip x.x.x.x –port xxx
Show current sip info ---show

- User management

path: <config-user>#
Change user right. ---access –user xxx –access xxx
Example:<config-user>#access –user aaa –access 7
Change user password ---password –user xxx
Add new user ---entry –user xxx –access xxx
Example:<config-user>#entry –user abc –access 7
Delete user entry ---no entry –user xxx
Show current sip info ---show

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5.1.3 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

5.1.4 download configure to flash

usage: #download tftp –ip x.x.x.x –file xxx
#download ftp –user xxx –password xxx –ip x.x.x.x –file xxx
Example: #download ftp –user abc –password 123 –ip 202.112.20.15 –file AG188.cfg

5.1.5 password

usage: #password
Enter new password:xxx
Confirm new password:xxx

5.1.6 reload

usage: #reload
Reboot system

5.1.7 show system running info

- accesslist

path: <show>#
show: accesslist (firewall) settings
Example: #<show>#accesslist

- basic

path: <show>#
show network status
Example: #<show>#basic

- call

path: <show>#
show current call info
Example: #<show>#call active

- capability

path: <show>#
show CODEC capability
Example: #<show>#capability

- debugging

path: <show>#
show debug info
Example:#<show>#debugging

- dhcp-server

path: <show>#
show LAN status and DHCP server info
Example:#<show># dhcp-server

- dial-rule

path: <show>#
show digital-map info
Example:#<show># dial-rule

- interface

path: <show>#
show LAN info
Example:#<show>#interface fastethernet lan
show WAN info
Example:#<show>#interface fastethernet wan

- ip

path: <show>#
show arp table info
Example:#<show>#ip arp

Show DNS server info
Example:#<show>#ip dns

Show netstate info
Example:#<show>#ip netstat

Show route info
Example:#<show>#ip route

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

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

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 IP phone memory
Example:#<show>#memory

- nat

path: <show>#
show NAT information
Example:#<show>#nat

- port

path: <show>#
show caller-ID info
Example:#<show>#port callerID

show dsp info
Example:#<show>#port dsp

show hotline info
Example:#<show>#port hotline

show black list info
Example:#<show>#port in-limit

show outgoing limit info
Example:#<show>#port out-limit

show current phone number
Example:#<show>#port number

show current port status
Example:#<show>#port status

- PPPoE

path: <show>#
show PPPoE info
Example:#<show># pppoe

- qos

path: <show>#
show QoS table info
Example:#<show>#qos

- sip

path: <show>#
show sip info
Example:#<show>#sip

- udptunnel

path: <show>#
show UDP tunnel info
Example:#<show># udptunnel

- uptime

path: <show>#
show running time
Example:#<show># uptime

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- version

path: <show>#
show IP phone version
Example:#<show># version

5.1.8 telnet and logout

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

5.1.9 timesettings

path: <time>#
---manualset –year xxx –month xxx –day xxx –hour xxx –minute xxx –second xxx
Example:<time>#manulset –year 2004 –month 10 –day 1 –hour 8 –minitute 30 –second 0

[disable]enable SNTP server	---sntp [no] start
Set SNTP IP address	---sntp server x.x.x.x
Set SNTP server timeout	---sntp timeout xxx
Set timezone (-12~+12)	---sntp zone xxx
Show SNTP info	---sntp show
Show current time	---print

5.1.10 tracert trace network path info

usage: #tracert –host
Example:#tracert ‼ HYPERLINK "http://www.google.com" ¶ www.google.com┴

5.1.11 update IP phone

usage: # update ftp –user xxx –password xxx –ip x.x.x.x –file xxx
update tftp –ip x.x.x.x –file xxx
Example:# update ftp –user abc –password 123 –ip 202.112.20.15 –file AG188.dlf

5.1.12 upload configure file

usage: # upload ftp –user xxx –password xxx –ip x.x.x.x –file xxx
upload tftp –ip x.x.x.x –file xxx

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6 Network Diagnosis

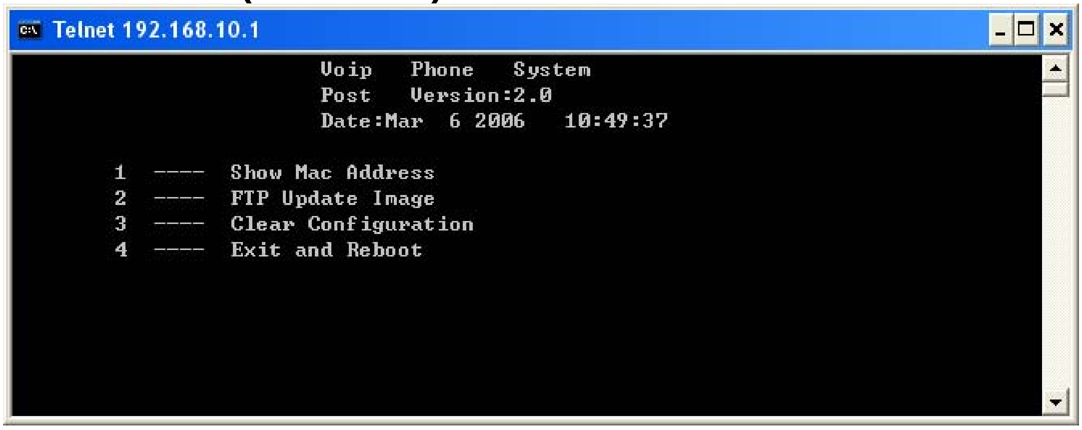
There are some telnet commands for checking your network. Now Listing below for your information

Command	Function	Example
ping	Check if the destination is accessible	#ping www.google.com
tracert	Show network path info	#tracert www.google.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
show ip netstat	Netstat programe	#show ip netstat
telnet	Telnet to another device	#telnet 192.168.1.2

7 Restore to factory default

#setdefault clear IP phone settings expect network part
#setdefault all clear all settings.

8 POST Mode(safe mode)



AT-530 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?
There will be a schedule bar in the AT-530 booting procedure, press # key within the first 5 seconds, then the phone will go to POST mode. It has a default ip 192.168.10.1 in POST mode. User may change the PC's IP address to 192.168.10.xx and telnet to 192.168.10.1 to access the IP phone in POST mode.
User can accord the guide in post mode to clear the settings or upgrade the firmware.

9 FAQ

9.1 How many servers may AT-530 register simultaneously?

AT-530 is able to register two SIP servers simultaneously, and redundancy servers. User can configure the dial peer to route calls between these servers. Please refer “How to use the dial rule?” for detail.

9.2 Why the settings vanish after reboot?

Please go to Config Manage→Save Config to save your setting always.

9.3 How to use the dial rule?

AT-530 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 servers according the prefix.

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..

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 in SIP

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:

Dial-Peer

Number	Call Mode	Destination	Port	Alias	Suffix	Del length
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

Add Delete Modify 2T

2T rule: If the call starts with 2, the first 2 will be deleted, and the rest number will be sent to private SIP server.
3T rule: If the call starts with 3, the first 3 will be deleted, and the rest number with be sent to public SIP 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 AG-188 will send 8675583018049 to your server.
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.

9.4 How to use speed dial function?

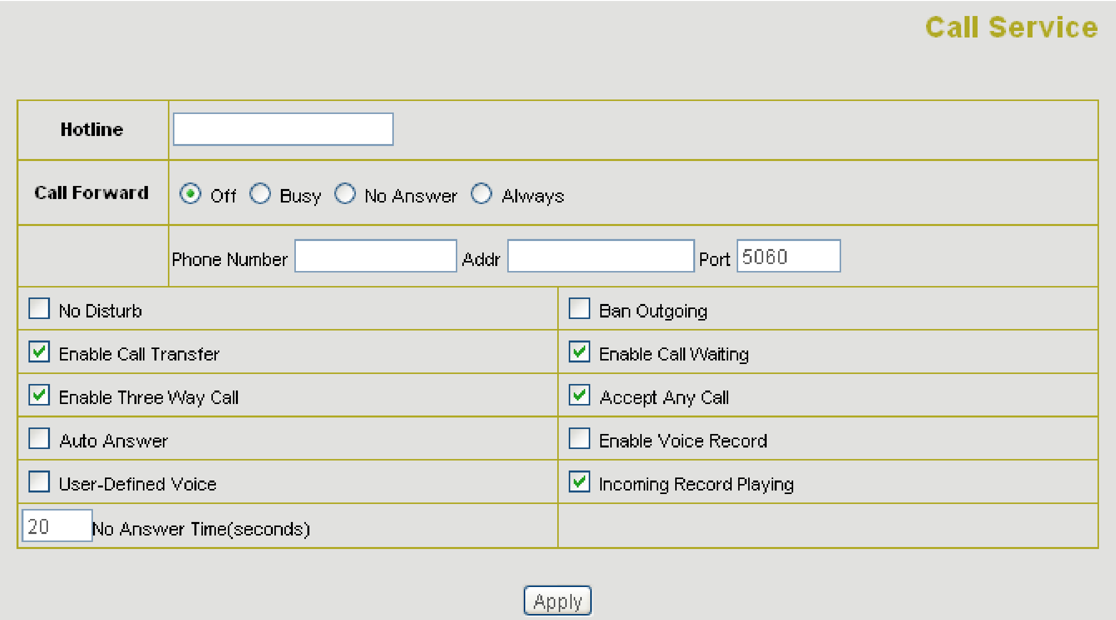
There are 9 speed dial keys in the AT-530 panel, Usage:
Set speed dial number: press the speed key and enter the speed dial number and then press Menu/OK key to save the setting.
Pick up the handset and press the speed dial key to dial the pre-define number.

9.5 How to configure digital map?

Please refer the digit_map.

9.6 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 call this gateway will be forward to the destination number.

➢ Call Transfer:

Check the *Enable Call Transfer*.

Unattended transfer:

If A is the AT-530 user, and B calls and talking with A through VoIP. A can **press FWD button** to hold the call with B, and then **enter C's number**. B will be transferred to C and can talk with C.

Attended transfer:

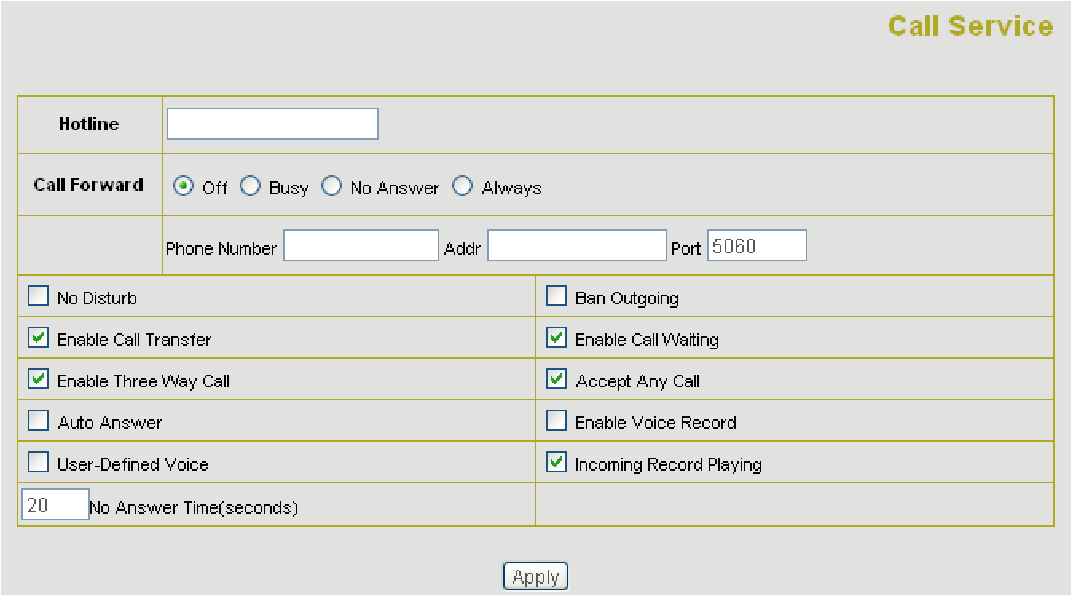
If A is the AT-530 user, and B calls and talking with A through VoIP. A can **press Hold button** to hold the call with B, and then **enter C's number** to talk will C. and press **Hold** to switch back to A, and then press **FWD** key , B will be transferred to C and can talk with C.

➢ 3-Way Conference Calls

Check Enable Three Way Call

Assume A is the AG-530 user, and B calls and talking with A through VoIP. A can **press FWD button** to hold the call with B, then **enter *** and then **enter C's number** to talk with C, and then **press * button** again to make 3-way conference calls.

9.7 How to use the record function?



AT-530 provides record function. With this function, user may record three VoIP message and one local message.

Active answering machine:

Select **"Enable Voice Record"** to active answering machine, and config **No Answer Time.** If there is an incoming call and no one answer the call. After timeout, AT-530 will auto answer this call and ask the caller to leave message.

Incoming Record Playing: play the message when recording.

User-Defined Voice: Use customizes greeting voice for answering machine.

Notice: AT-530 supports three message maximum, each message can be 90 seconds. Answering will be deactivated if the message numbers is 3.

Record local message:

User may use local message to leave message to other local users.

Please refer the **Record** button function as below:

Record Function		
Level1	Level2	Description
Received	New	New message info
	Old	Old message info
	Record	Enable/disable answering machine
	Playing	Enable/disable Incoming Record Playing
Local	Play	Play local message
	Rec	Record local message
User define	Switch	Enable/disable customize greeting message
	Play	Play customize greeting message
	Rec	Record customize greeting message

9.8 How to use set the IP type via keypad?

In the idle mode, user may us the keypad to set the IP type as the below procedure:
Keep pressing the button 1 for changing to static mode.
Keep pressing the button 2 for changing to DHCP mode.
Keep pressing the button 3 for changing to PPPoE mode.