

VOIP
Desktop/Wall

Coin Pay Telephone
User Manual

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First of all, we welcome you to use this VOIP Desktop/Wall Coin PAY PHONE, and thank you for your trust in our products. In order to have a better experience with this VOIP, PSTN or GSM Payphone and to enjoy the professional service, please read the user guide carefully before using it, and keep the manual for later reference.

1. Introduction

This VOIP Desktop/Wall Coin PAY PHONE is our company's latest patent products. This model is manufactured according to VOIP standard and SIP agreement, support the third party voice software package base on SIP standard, SIP equipment, SIP agent, SIP registered server, SIP media network, etc. This model not only have all the traditional payphone functions, but also have network communication function, that is, sending voice information through network. So under the situation of cheap network rate, it can greatly reduce the cost of long distance call, especially the international distance one. This model is the best choice of the long distance call operation, can be used in many public places. The device is specially designed to satisfy different needs of the customers in different environments.

1.1. Product Features



1.2. Hardware Specification

Item		Specification
Power Adapter	Input	110-220V AC
	Output	12V DC 0.5A
Port	WAN	1 10/100Base T RJ-45
	LAN	1 10/100Base T RJ-45
	Lifeline	1 RJ11 for Lifeline (PSTN)
Power Consumption		3.5W/1.8W
Operating Temperature		0~40℃
Relative Humidity		5~95%
Weight		4KG

2. Network feature

2.1. Operation Standard

This model is manufactured follow the VOIP agreement and standard.

- ◆ Support SIP (RFC3261, RFC2543)
- ◆ Support SIP domain
- ◆ Support SIP authentication (none, basic, MD5), domain name parse
- ◆ Register to two SIP servers synchronously, including public server and private server, can make a call by either proxy
- ◆ Support redundant public server, auto swap to backup server when main server has problem
- ◆ Support IAX2
- ◆ Support domain name register and domain name parse
- ◆ Support Voice Codec: G.711A/U, G.726, G.729
- ◆ Support in-band and out-of-band DTMF transmission: DTMF relay, SIP info, RFC2833
- ◆ Support several NAT transversal methods: STUN, CITRON, AVS
- ◆ Support RTP and RTCP

2.2. Network Features

- ◆ Integrate two ports router function, support bridge mode and router mode
- ◆ Support basic NAT and NAPT
- ◆ Support PPPoE for xDSL, auto redial when disconnect
- ◆ Support DHCP client at WAN port
- ◆ Support DHCP server at LAN port
- ◆ Support DNS relay for LAN, can provide DNS service for LAN Network equipment
- ◆ Support several network tools: Ping, trace route and telnet client
- ◆ Have three modes of WAN port IP configuration: Static, DHCP, and PPPoE
- ◆ Provide firewall control for small LAN
- ◆ Provide optional priority level for small LAN
- ◆ Support secondly layer QoS (802.1p)

2.3. Advanced Functions

- ◆ Have three numbers in one phone: 2 VOIP number and 1 PSTN number
- ◆ Switch freely between VOIP and PSTN
- ◆ Use advanced DSP tech to insure high quality voice
- ◆ Use advanced jitter buffer tech to reduce data package delay and lost
- ◆ Support VAD (Voice Activity Detection), CNG (Comfort Noise Generation) and silence suppression
- ◆ Support point-to-point call
- ◆ Support hotline function (off-hook auto dialing)

- ◆ Setup dialing number ending mode
- ◆ Support many countries' standard ring

2.4. Configuration, Management and maintenance

- ◆ Can be configured through web-page and telnet, support IVR and keypad operation (keypad of the phone which is connected to GW2000RSP's phone port)
- ◆ Support HTTP, FTP , TFTP firmware and configuration file upgrading
- ◆ Support safe mode and can upgrade firmware through safe mode
- ◆ MMI access right can be restricted to limited client IPs

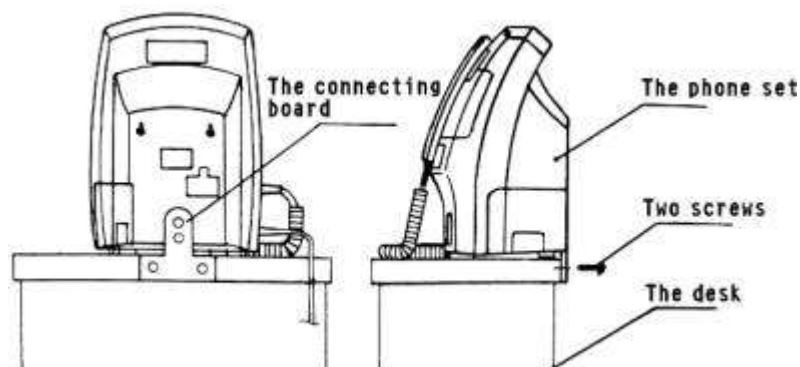
3. Using Directions

3.1. Installation

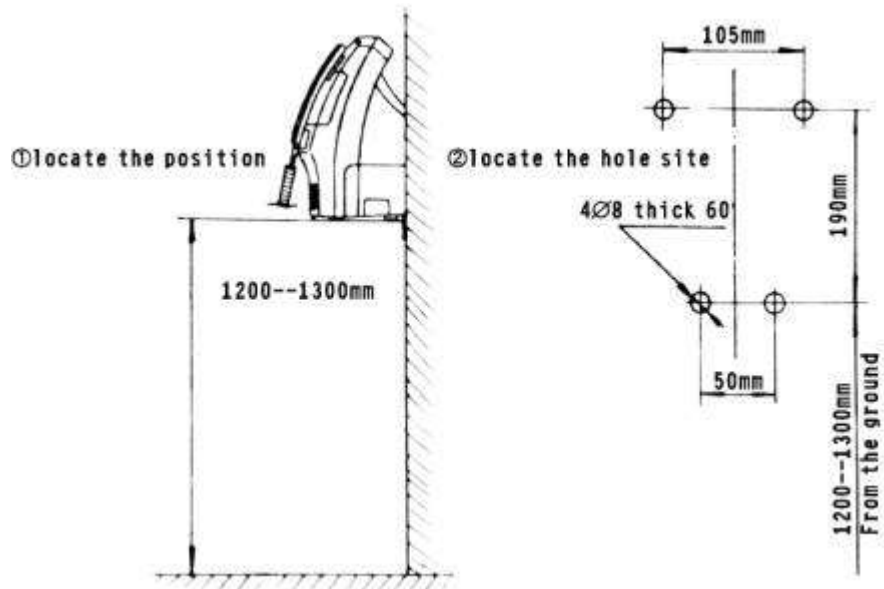
3.1.1 External Parts Installation



1. Desktop installation: using the accessories to fix the payphone to the desk or just put it on the desk.

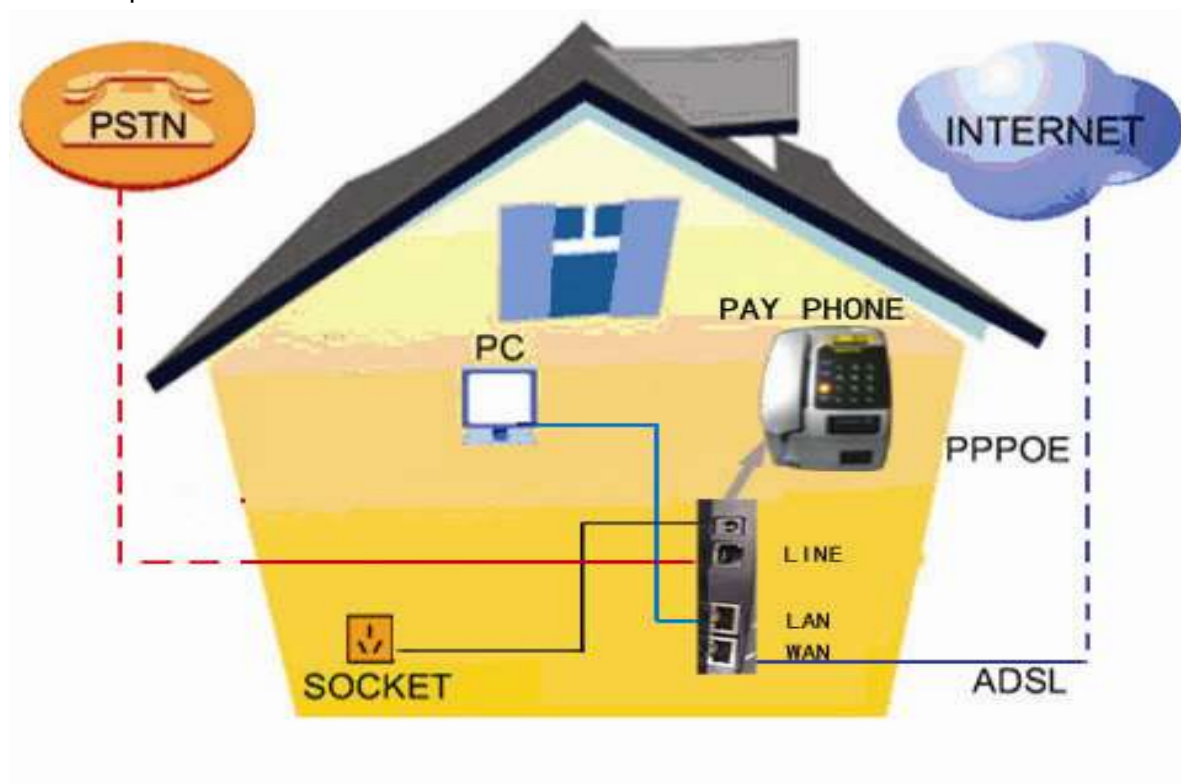


2. Wall mounting installation:



3.1.2. Equipment Connection

- ◆ One piece of HT8868V0IP PAYPHONE
- ◆ Standard RJ45 network cable
- ◆ Original power adapter
- ◆ Computer



- 1) Connect an Ethernet cable between "LAN" port and your PC (PC is only used for

configure the gateway especially during the first time using the gateway. PC is not required to set up for making a call).

- 2) Insert the power adapter's (output DC 12V) plug into the gateway power jack and insert the other end of the adaptor into well grounded power outlet.
- 3) Insert an Ethernet cable into "WAN" port of the phone and make sure another end of the cable is connected to a network (the wide area network or your local area network).
- 4) Connect PSTN phone line to "LINE" port of gateway as the lifeline. (optional)
Note: This model should only be operated with the universal power adapter provided together with the model. Damages to the model caused by using other unsupported power adapters would not be covered by the manufacturer's warranty.

3.1.3. Led Indicator

WAN: WAN port indicator.

Light on — the cable has been plugged well with the WAN port of the payphone.

LAN: LAN port indicator.

Light on — the cable has been plugged well with the LAN port of the payphone.

PHONE: Phone indicator

Light on — handset off hook

Light off — handset on hook

REG: Register indicator

Light on — Register successfully;

Light off — Register function disabled;

Light flash — logging on to SIP server

POWER: Power supply indicator

Light on — power good.

3.2. Network Setup

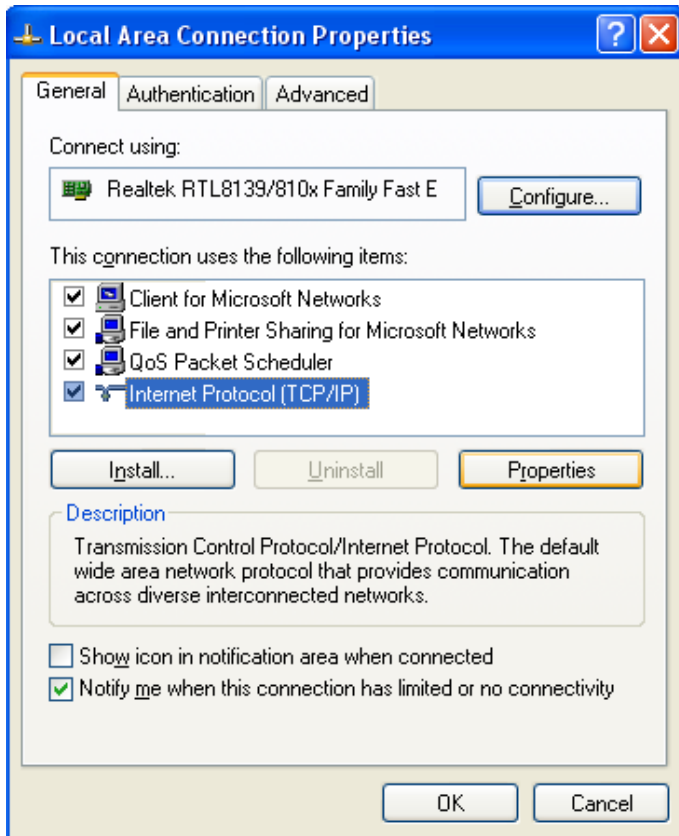
3.2.1. WEB Configuration

WEB Configuration is completed by the payphone, computer and internet.

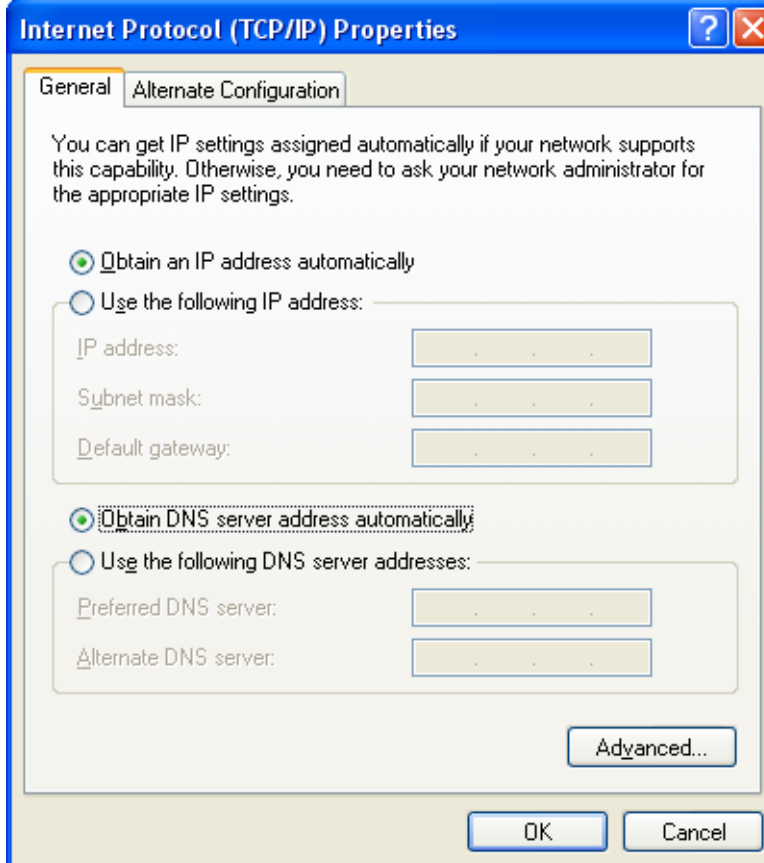
3.2.1.1. IP Inspection

Connect a PC to LAN port of the payphone. Set the IP obtain way on dynamic. Enter "ipconfig/all" in order field. The default address of checking PC is the payphone LAN port's IP. Following is the steps:

Open the window of "local area connection properties" of your PC.



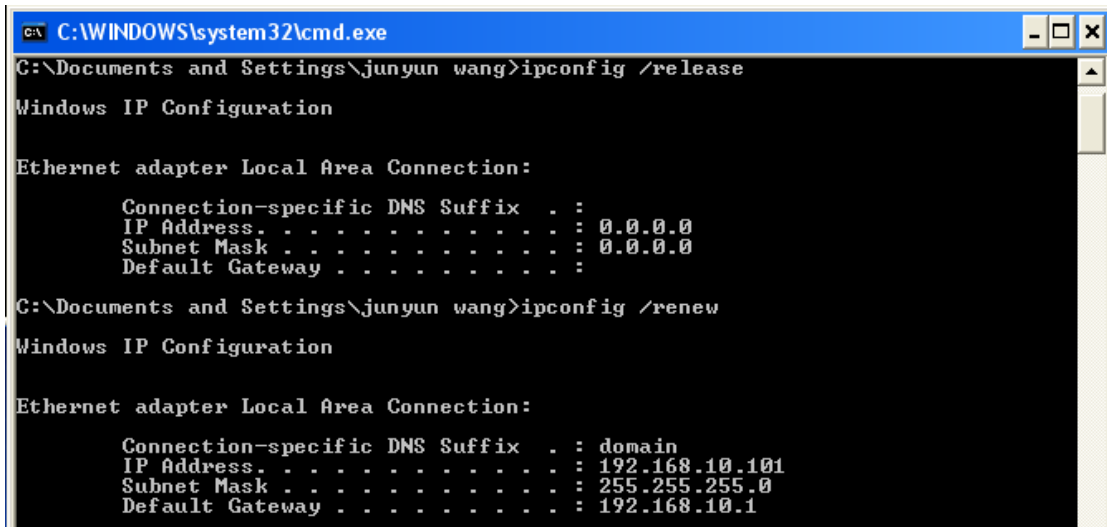
Choose "Internet protocol (TCP/IP)" and press "properties" button.



Setup as above and press "OK" button.

Input "cmd" when run mode, press "enter".

Input "ipconfig /all", can check. Default gateway is the payphone's LAN IP.



```
ca C:\WINDOWS\system32\cmd.exe
C:\Documents and Settings\junyun wang>ipconfig /release
Windows IP Configuration

Ethernet adapter Local Area Connection:

    Connection-specific DNS Suffix  . : 
    IP Address . . . . . : 0.0.0.0
    Subnet Mask . . . . . : 0.0.0.0
    Default Gateway . . . . . : 

C:\Documents and Settings\junyun wang>ipconfig /renew
Windows IP Configuration

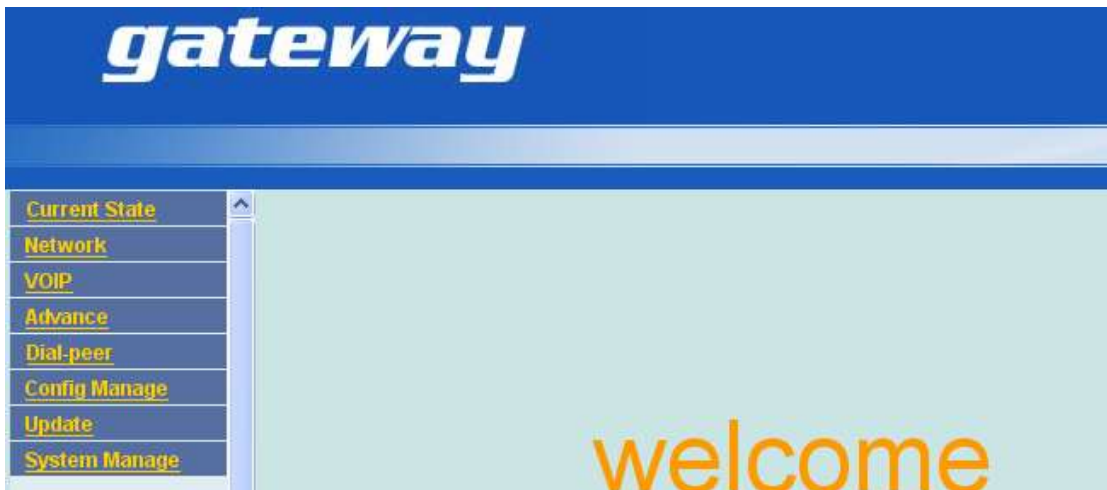
Ethernet adapter Local Area Connection:

    Connection-specific DNS Suffix  . : domain
    IP Address . . . . . : 192.168.10.101
    Subnet Mask . . . . . : 255.255.255.0
    Default Gateway . . . . . : 192.168.10.1
```

You can see the Default Gateway of your PC is 192.168.10.1, so the LAN port IP address of payphone is 192.168.10.1 (the default IP address on LAN port of payphone) . After getting the IP address, input "http://192.168.10.1" on the web browser and press "enter" to configure the payphone.

3.2.1.2. WEB Page Configuration

The web configuration interface mainly consists of the configuration menu on the left and configuration Interface on the right, enter the corresponding configuration interface to configure by selecting the menu, see as below:



Before using this payphone, the main menu you need configure is Network (configure the IP, default route, DNS), VOIP (configure SIP server and the number).

If the menu has submenu, press it to see all the parts. Press once to open the menu, and press twice to close the menu.

3.2.1.3. User Login

In order to browse and configure this payphone, you need to login first. Administrator account: the default username and password are all "admin", this user can configure the system.



After inputting username and password, press "Enter" to enter the page.

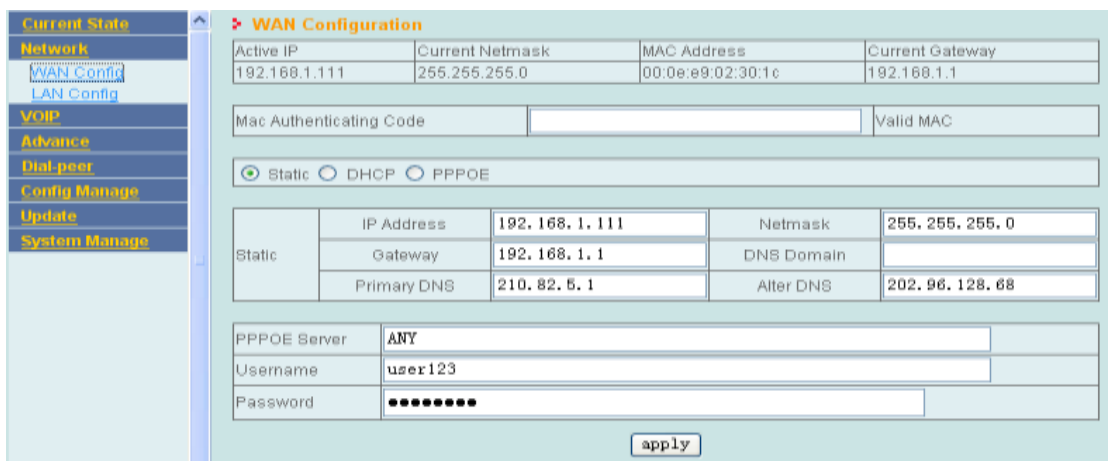
Note: After you finish configuring each page please don't forget to click the "apply" button at the bottom of the page! Also please go to Config Manage->Save Config and click on "save" to save all your configuration!

3.2.1.4. Network Configuration

Note: after configuring the IP of WAN correctly, the payphone can connect Internet.

If your LAN router opens DHCP service, payphone WAN IP will be gotten dynamically, no need any other configuration. (the default configuration for WAN is DHCP client mode, the default LAN IP is 192.168.10.1);

If your LAN router does not open DHCP service, you need configure the WAN port to static IP (if this payphone does not work as router, you don't need any configuration).



Active IP	Current Netmask	MAC Address	Current Gateway
192.168.1.111	255.255.255.0	00:0e:e9:02:30:1c	192.168.1.1

Mac Authenticating Code: Valid MAC:

Static DHCP PPPOE

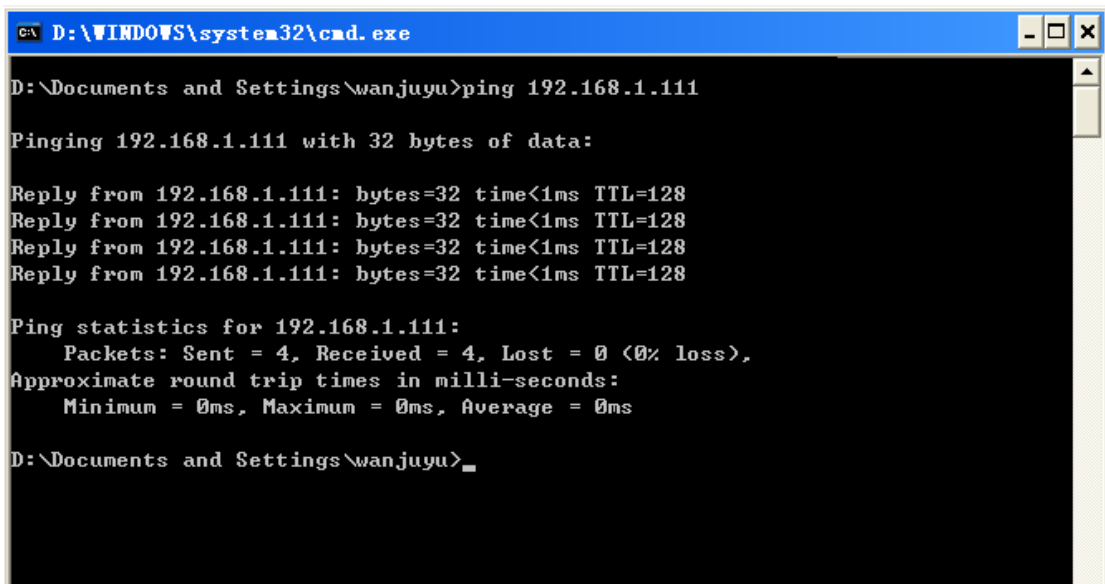
Static	IP Address	Netmask
	192.168.1.111	255.255.255.0
	Gateway	192.168.1.1
	Primary DNS	210.82.5.1
	Alter DNS	202.96.128.68

PPPOE Server: ANY

Username: user123

Password: [masked]

When configuring static IP, the static IP must be on the same IP segment with the IP of LAN router. After configuring, you should check if the network is working: open the run mode, and input "cmd", press "Enter" button. Input "ping 192.168.1.111" and press "enter" (change the IP address "192.168.1.111" to the actual IP address of the payphone if your payphone is configured with an IP address other than this).



If "ping" showed it is ok, that means the network is connected. And then only need configure SIP account.

3.2.1.5. SIP Server configuration

User can configure specific parameter of SIP protocol on this page.

Current State	SIP[Registered] Configuration	
Network	Register Server Addr	202. 56 . 120. 165
VOIP	Register Server Port	5060
SIP Config	Register Username	5943234
Advance	Register Password	••••••••
Dial-peer	Domain Realm	sip-domain.com
Config Manage	Phone Number	7943235
Update	Detect Interval Time	600 seconds
System Manage	DTMF Mode	DTMF_SIP_INFO
	Encrypt Key	
	<input checked="" type="checkbox"/> Enable PRACK	<input type="checkbox"/> Signal Encrypt
	<input type="checkbox"/> Auto Detct Server	<input type="checkbox"/> RTP Encrypt
	<input checked="" type="checkbox"/> Enable Keep Authentication	<input type="checkbox"/> Enable Session Timer
	<input checked="" type="checkbox"/> Enable Register	
	<input type="button" value="apply"/>	

Configuration Explanation:

SIP[Registered] Configuration It shows SIP register state; if register successfully, it will show Registered in the square bracket, otherwise show Unregistered.

* Register Server Addr 192.168.1.1 it shows the configured IP address for SIP register server. (Domain Name or IP address)

Register Server Port it shows the configured signal port for SIP register server. Normally its 5060.

* Register Username it shows the configured SIP register account name (usually it is the same with the phone number that configured. When some special SIP servers have different phone number and account name, here need configure the account name).

* Register Password it shows the configured password of SIP register account.

Proxy Server Addr it shows the configured IP address for proxy server (usually SIP service provider provides user with service of proxy server and register server which have the same configuration, so the configuration of proxy server usually is the same with that of register server, but if the configurations of them are different (such as different IP address), then each server's configuration should be modified separately).

Proxy Server Port it shows the configured signal port for SIP proxy server.

Proxy Username it shows the configured account name for proxy server.

Proxy Password it shows the configured password for proxy server.

* Phone Number it shows the configured register phone number.

* Domain Realm it shows the configured SIP register domain realm. it must be accordant with the register server address. If the SIP terminal port is not default 5060, the back should be need add the detail port number. e.g.: sipdomain.com:6058.

Local SIP Port it shows the configured local SIP signal port, the default is 5060 (this port will go into effect immediately, the SIP call will use the modified port for communication after modification).

Register Expire Time seconds it shows the configured expire time of SIP server register, the default is 60 seconds. If the expire time that server requires is more or less than that configured by the payphone, the payphone can automatically modify it to the recommended time limit and register.

Detect Interval Time seconds it shows the configured detection interval time of the server, if the payphone enables SIP detection server function, the payphone will detect once for whether the server has response every detection internal time.

Enable Register it shows the configured enable/disable register.

RFC Protocol Edition it shows the configured using protocol version. When the payphone needs to communicate with the payphone using SIP 1.0 such as CISCO 5300, the RFC2543 configuration is necessary, then the communication will be

normal. The default is using RFC3261;

it shows the configured DTMF sending mode; there are three modes for sending mode: the different service provider may provide the different mode.

Auto Detct Server it shows the configured automatic detection server of the payphone.

Note: After configuring the network and VoIP settings, if the register is successfully, the REG light will be on (if the light is twinkling, that means the register is not successful). At this time, the user should be able to make VOIP calls through the register and proxy. Must pay more attention, dialing number is ended by #.

3.2.1.6. Save and Clear the configuration

Enter into WEB page, "Config Manage->Save Config" to save the configuration; Enter into WEB page, "Config Manage->Clear Config" to clear the configuration, and come back to default configuration.

3.2.1.7. Reboot the gateway

Enter into WEB page, "System Manage->REBOOT" to restart the gateway;

Note: if the gateway always has no answer, please cut off the power supply and restart the gateway.

3.2.2. Factory Default Configuration

- ◆ Default DHCP mode to get the IP address for WAN port, if switch to Static mode, the WAN IP address is 192.168.1.179, the LAN IP address is 192.168.10.1, the default DHCP service and NAT service for LAN is opened.
- ◆ Default Protocol is SIP, default SIP port is 5060.
- ◆ Default HTTP port is 80, default Telnet port is 23.
- ◆ Default ending sign for dialing number is #.
- ◆ Default user accounts are admin and guest.

3.3. Rate and Function Setup

Rate and Function Setup can be completed on the payphone (no need to connect the pc, only using adapter).

3.3.1. Definition and Function Setup

0 Clear the memory:

#*ABCD Setting the condition of the line

D stands for the rate for incoming calls: To continue the talk

after the free time limit, the unit will collect coins according to the relevant rate set in C; 1=#1,2=#2, 7=#7

C stand for the free time limit :C=1means 1 min.

C=9 means 9 min

AB stand for outgoing codes for PABX connection AB=*x means one digit outgoing code for PABX, AB=xx means two digit outgoing code for PABX)

#8ABCDE Setting length of dialed number
DE means length of dialed number
ABC means 3 digit called office code (telephone number prefix)

Remark: At most 15 groups of number can be input. A number covered by another should be input after the number number#88##13, #89**13 and #8***15,you should input #8***15 first, then input#88**13and#89**13)

9ABCDEFGH Setting the password for inquiring the record of coins collected

ABCDEFGH means password only for inquiring the record of coins collected (Please do not input "*" and "#" in the password).

ABCDEFGH Setting show the telephone number
ABCDEFGH means any eight-digit telephone number, if less than eight, add * to the end.

nABCR

R means coin amount to be inserted. Use 50 as coin unit, then 1=50, 2=100

ABC means time(unit:1Second) 001=1Second, 999=999Second

n means rate type: n=1 is auto-pager

n=2~7 is rate type

NUMBER GROUPS & RELEVANT FEE RATES SETTING

* 3ABCR Setting 3 digit called office code (telephone number prefix) and the relevant rate

R means relevant rate: 1=#1, 2=#2,...7=#7; 0=prepaid call, use rate #1 to control; *=free call, #=forbidden call

ABC means 3 digit called office code/telephone number prefix (*used to represent 0-9)

*4ABCDR Setting four-digit called office code and the relevant rate

R means relevant rate: 1=#1, 2=#2,...7=#7, 0=prepaid call,
use rate #1 to control. *=free call, #=forbidden call
ABCD means 4 digit called office code/telephone number prefix (*used to
represent 0-9)

*5ABCDEF Setting five-digit called office code and the
relevant rate

R means relevant rate: 1=#1, 2=#2, ...7=#7, 0=prepaid call,
use rate #1 to control. *=free call,
#=forbidden call

ABCDE means 5 digit called office code/telephone number prefix (* used to
represent 0-9)

*6 ABCDEF Setting six-digit called office code and the
relevant rate

R means relevant rate: 1=#1, 2=#2,...7=#7, 0=prepaid call,
use rate #1 to control. *=free call,
#=forbidden call

ABCDEF means 6 digit called office code/telephone number prefix (* Used to
represent 0-9)

*7ABCDEFGR Setting 7digit called office code and the
relevant rate

R means relevant rate: 1→#1, 2=#2,...7=#7, 0=prepaid call,
use rate #1 to control. *=free call,
#=forbidden call

ABCDEFG means 7 digit called office code/telephone number prefix (* used to
represent 0-9)

*8 ABCDEFGHR Setting 8digit called office code and the
relevant rate

R means relevant rate: 1=#1, 2=#2,...7=#7, 0=prepaid call,
use rate #1 to control. *=free call,
#=forbidden call

ABCDEFGH means 8 digit called office code/telephone number prefix (* used to
represent 0-9)

*9 ABCDEFGHIR Setting 9 digit called office code and the
relevant rate

R means relevant rate: 1=#1, 2=#2,...7=#7, 0=prepaid call,
use rate #1 to control. *=free call,
#=forbidden call

ABCDEFGH I means 9 digit called office code/telephone number prefix(* used to represent 0-9)

Note: Since * can represent one of the characters of 0 to 9, when entering the numbers, if a number group included by another group has different rate, the setting of it should be input precedently.

3.3.2. Setup Example

A customer wants to set the following items:

1. Free time limit for incoming calls (CO line) = 3 minutes. Charge rate to continue the talk after the free time limit is 50 per 180 seconds.
2. Owner code for free call: 09874526.
3. The user telephone's number: 8904558.
4. Block number 2xx 18 digits, 3xx 19 digits, 08x 13 digits , xxx 17 digits and 0xxx 20 digits
5. Free call: 110,104,119,118,181,101,131,103,113
6. Barred call: 160,168,98*
7. Auto-pager 127,988,95*; charge rate = 100 per 80 seconds
8. Local call: 2XX, 3XX, 9XX; charge rate = 100 per 180 seconds.
9. Mobile call: 010X, 012X, 013X, 016X, 019X, 015X; charge rate=150 per 180 seconds.
10. Long distance call 02XXX, 03XXX, 04XXX, 06XXX, 07XXX, charge rate=200 per 60 seconds.

Please input according to the following steps:

1. Connect the telephone to the line or a testing instrument.
2. Take out the coin box.
3. Put the DIP switch 1、2 to ON position.
4. Pick up the handset, press # and * at the same time. LCD display P on the left side.
5. # 0 clear EEPROM
6. # 909874526 to set free call password 09874526.
7. ##8904558* to set own telephone number 8904558.
8. #8***17,#82**18 ,#83**19,#80**20,#808*13
9. #***32 to set Co .line free time limit of incoming calls 3 min and charge rate of continue talk 50/180 seconds.
10. *3 110*, *3 104*, *3 119*,*3 118*,*3 181*, *3 101*, *3 131*,*3 103*,*3113*, Free call.
11. *3 160#,*3 168#,*3 96*#, Barred call
12. #10801 Rate 2 =50/80 seconds.
13. *3 1271,*3 9881,*3 95#1, auto charge
14. #21801 Rate 2 =50/180 seconds.
15. *3 2**2, *3 3**2,*3 9**2. Local call number and rate.

-
16. #30603 Rate 3 =150/60 seconds
 17. *4010*3, *4012*3, *4013*3, *4016*3, *4019*3, *4015*3, Mobile numbers and rate..
 18. 15.#40604 Rate 4 =200/60 seconds.
 19. *502***4, *503***4, *504***4, *506***4, *5 07***4.
 20. Put the DIP switch 1、 2、 to OFF position
 21. Hand up the handset.
 22. Put back the coin box and lock it.

3.4. Using Steps

After completing the above setup, you can take off the pc connection and dial the phone according to the following steps:

- 1、 Pick up the handset (the LCD displays 0 on the left side)
- 2、 Wait for the dial tone, put in sufficient coins(the LCD displays the total value needed)
- 3、 Dial the number (for numbers more than seven digits, only the last seven digits would be displayed).
- 4、 Press the TALK button to enable the talk and collect the coins(the LCD displays the value needed for extending the talk and time limit counting down).
- 5、 When hearing the time limit warning tone, put in more sufficient coins to continue the call (the LCD displays the value needed on the left side). Otherwise the call will be terminated when the time limit reaches. Hang up the handset after finishing the talk (the LCD returns to displaying the clock).
- 6、 Dial auto calling number. After dialing the first three numbers, (dial local call number and long distance call number to limited digit) press "talk", then continue dialing the rest of the numbers.
- 7、 When the line is busy or no one answers the call, hang up the handset to withdraw the coin (the LCD returns to the clock).
- 8、 When the line is busy or no one answers the call, press the [REDIAL] key to dial the number again without withdrawing the coin (the LCD displays the number redialed). Please DO NOT hang up the handset if you want to redial without withdrawing the coins.
- 9、 When the line is busy or dial the wrong number, press "redial" to dial the correct number without withdrawing the coin (the LCD displays the number dialed).
- 10、 Password dialing: Lift the handset, press key # and enter the eight digits password, wait one second and you would hear the dial tone , then dial the number to call (This function is only for the phone owner).
- 11、 Free Time limit for incoming calls: After incoming calls are answered, there would be a certain free time limit(programmable).(LCD will display the value needed for extending the talk and time limit counting down). After hearing the time limit warning tone, insert sufficient coins to continue the call. Otherwise the call will be terminated when time limit reaches.

3.5. Clock Adjustment

Hang up the handset and take out the coin box.

-
1. Put the DIP switch 3, 4, 5 to ON position.
 2. Press the [REDIAL] key to select hour or minute.
 3. Press the [FLASH] key to adjust the timing.
 4. Press the [REDIAL] key again to confirm the setting.
 5. Put the slide switch 3, 4, 5 to OFF position to finish setting.
 6. Put the coin box back and lock it.

4. Troubleshooting

- 1) Lift the handset but no dial tone can be heard.

Check if the handset cord is mal-connected or the receiver is out of order.

- 2) When lift the handset, the LCD displays nothing.

Check if the tie line of the LCD is loose or broken.

- 3) Lift the handset but can not dial the number.

Check the connection of the tie line of the dial keypad and if the character keys are gripped.

- 4) The line is cut off soon after dialing.

Check if the keyboard connection line is short circuit or open circuit.

- 5) The line is cut off after the coin is collected and dialing the third digit.

Check if the coins are enough and the rate is set correctly.

- 6) After the line is connected and the coin is collected, the called party can't hear the caller's voice.

Check if there is any fault with the coin detection circuit or mouthpiece.

- 7) After the line is connected and the coin is collected, the line is cut off at once.

Check if the rate is set correctly. (The talking time limit may be set as 0 second.)

- 8) After the handset is hanged up, the LCD doesn't display real time clock.

Check if the hook switch is gripped and the time setting DIP switch is on "OFF" position.

- 9) When dialing the LCD flicker or when hang up the hook the LCD doesn't display.

Check if the dry battery is used up or unconnected with the phone.

- 10) The phone can't assign WAN or LAN static IP.

When the phone is in the NAT mode (no bridge mode), please don't assign the WAN and LAN IP with the same net segment e.g., if the LAN IP address is 192.168.1. x, don't set the WAN address in the same segment of 192.168.1.x.

- 11) Send a busy tone in the process of call.

The phone can detect if the network is normal or not, if the phone is not connecting to the network, the phone will send a busy tone automatically to remind the user. User can check the network is normal or not, ensure the cable is connected with the phone or not.

12) After I configure the WAN static IP, why does it return to the primary IP when confirming the modified local IP

It's likely to use the same net segment in WAN and LAN. Please try to change the LAN IP address.

13) Why can't I log in to gateway by Telnet?

It's likely to use the private address in the phone and PC, but the two IP addresses aren't in the same net segment. e.g. the phone uses the address 192.168.1.x, the PC uses the address 192.168.10.180, try to add a IP 192.168.1. xx in PC.

14) After configuring as the manual, but still can't dial normally?

Check the network, Telnet the phone, then use Ping or tracet command to access the exterior network to try. After the phone connected well, the fault operation and pause the upgrade maybe caused the phone dead, or can not log in the WEB page, or can not hear the dialing tone when picking up the receiver?

First, please check the network and the phone connection. If the network is connected correctly, but the problem still exist, we can diagnose the phone is dead because the error software, try to reboot the phone. If the problem is existing, at last only restart the phone to enter the POST mode: telnet 192.168.10.1 to enter the POST mode in 5 seconds after restarting the phone, then enter TFTP to upgrade and recovery according to the message.

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