User Manual

SIP-GW3CM
SIP Analog Telephone Adaptor

CATALOG

Key Feature

1. New feature

- Make/Receive both VoIP calls and regular PSTN calls.

- Auto-detection and auto-switch to VoIP or PSTN calls.

- Phone rings for all incoming VoIP and PSTN calls. Support Caller ID/Name display or block.

- Support Call hold, Call waiting, Call forwarding and Call transfer.

- Support three service domain and auto-provision.

2. Network Feature

- Support SIP 2.0 (RFC3261)PPPoE, STUN, DHCP.

- Support in-band and out-of-band DTMF transfer.

- Support automated NAT traversal without manual manipulation of firewall/NAT.

- Support firmware upgrade via TFTP, FTP and HTTP.

3. Voice Feature

- Support VAD,CNG,AEC, PLC,AJB and Volume adjustment. Voice Codec: G.711,G.723.1,G.726,G.729A,G.729B.

4. Feature Description

- Make/Receive both VoIP calls and regular PSTN calls.
- Auto-detection and auto-switch to VoIP or PSTN calls.
- Phone rings for all incoming VoIP and PSTN calls.
- VoIP or PSTN calls selected by phone key.
- Support Caller ID/Name display or block.
- Support call hold, call waiting, call forwarding, and call transfer.
- Dial plans, dial tone, busy tone, ring back tone, alert tone can be set flexible.

5. Management Feature

- Configure GW3CM via web browser or remote.
- Firmware can be upgraded through HTTP, FTP or TFTP.

6. Physical Feature

- Two RJ45 ports: Dual 10M/100M auto-sensing, with router built-in, one for internet, one for PC.
- Two RJ11 Port: One FXS telephone Port and One lifeline for PSTN line.
- LED Indication: There are three LED indicators in the GW3CM to show the Status, and Off-Hook indication.
- Power adaptor: Input: AC 100~240V, Output: DC12V/0.5A.
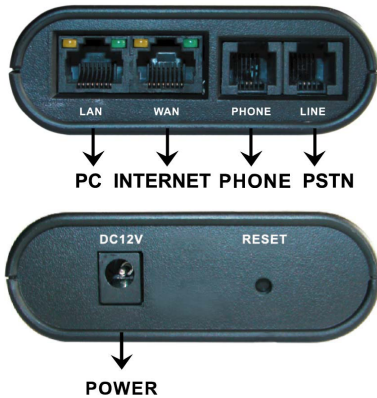- Operating Temperature: 0℃~40℃.

7. Package content

- One GW3CM main body
- One Ethernet cable
- One user Manual
- One Universal Power Adaptor
- One PSTN cable

Quick install

1. Connecting to GW3CM

Please install GW3CM as the connection chart below:

2. Configuring by webpage

2.1 Login the webpage

Login via LAN port

- Connect the PC to LAN port of GW3CM. Default IP address of the LAN port is 192.168.123.1.
- Open web browser and input http://192.168.123.1.
- Enter the account and password (default account is **admin** and password is **admin**).

3. Network Configuration (WAN)

- In Network you can check the Network status, configure the WAN Settings, LAN Settings.
- Network Status: You can check the current Network setting in this page.

Network Status

This page shows current status of network interfaces of the system.

System Up Time:	0 day(s) 2 hour(s) 59 minute(s)
Network Link Up Time:	0 day(s) 2 hour(s) 59 minute(s)
NAT Type:	Symmetric

Interface 0	
Type:	Fixed IP Client
IP:	192.168.1.51
Mask:	255.255.240.0
Gateway:	192.168.1.4
DNS Server 1:	168.95.192.1
DNS Server 2:	168.95.1.1

Interface 1	
Type:	DHCP Server
IP:	192.168.123.1
Mask:	255.255.255.0
Gateway:	192.168.123.1
DNS Server 1:	168.95.192.1
DNS Server 2:	168.95.1.1

- WAN Settings: The WAN port is for you to connect to the ADSL Router, Broadband Router. Also you can use PPPoE to get the WAN IP address from your ISP.

    a) The IP Phone's default setting is NAT mode. If you don't need to use the NAT Mode, you can change into Bridge Mode.

    b) The WAN port default is DHCP Client mode, and you can change the setting to Fixed IP Mode, or PPPoE Mode.

    c) If you change the WAN port setting to Fix IP Mode, then you have to make sure the IP address. Net Mask, Gateway, and DNS setting are suitable in your current network environment.

    d) If you change the WAN port's setting to PPPoE Mode, you have to input a correct username/password to get the IP address from your Internet Service Provider.

4. Register account (Sip Setting)

- You may get account information from your service provider.

- In Service Domain Function you need to input the account and the related information, please refer to your ISP provider. You can register three SIP account in the GW3CM. You can dial the VoIP phone to your friends via first enable SIP account and receive the phone from these three SIP accounts. For the second phone you can use the same way to register.

- Click Active to enable the Service Domain.

  a) Display Name: input the name you want to display.

  b) User Name: input the User Name get from your ISP.

  c) Register Name: input the Register Name get from your ISP.

  d) Register Password: input the Register Password from your ISP.

  e) Domain Server: input the Domain Server get from your ISP.

  f) Proxy Server: input the Proxy Server get from your ISP.

  g) Outbound Proxy: input the Outbound Proxy get from your ISP. If your ISP does not provide the information, then you can skip this item.

  h) If you have more than one SIP account, you can select the Realm number and following the steps to register to the other ISP.

  i) When you finished the setting, click the Submit button.

  j) If there is nothing need to be changed, please click the Save Change Item in the left side, then click the Save button. The change you made will save into the system and the system will Reboot automatically.

5. The LED of GW3CM

Status's LED	PSTN's LED	VOIP's LED
Unregistered: Off	Offhook: 0.5s Flash	Offhook: 0.5s Flash
Registered: Lighted		
Reboot: Lighted	Reboot: Lighted	Reboot: Lighted

Basic Functions

Note 1:

- When GW3CM is out of power, the RJ11 line jack will act as a pass through jack. The user will be able to use the same analog phone for PSTN calls.
- When GW3CM is not registered to service, Pick up the analog phone and press the number "0*", the RJ 11 line jack will act as a pass through jack. The user will be able to use the same analog phone for PSTN calls.
- When GW3CM registered service, the SIP line jack will act as a pass through jack. The user will be able to use the same analog phone for VoIP calls.

Note 2:

- When you finished the setting, please click the Submit button. If there is nothing need to change, please click the Save Change Item in the left side, then click the Save button. The change you made will save into the system and the system will Reboot automatically.

1. Make Phone Calls

- Make VoIP Calls: Pick up the analog phone, LED "SIP" Wink. There are currently two methods to call out:
  a) Dial the numbers directly and wait for 4 (general) seconds.
  b) Dial the numbers directly, and press # .

Note: You can make a VoIP call only when LED "STATUS" is lighted.

- Make PSTN Calls : Pick up the analog phone, if LED "SIP" wink, Press "0*" to switch to PSTN line. LED "PSTN" wink, and get a PSTN line dial tone, dial the numbers directly.

Note: You can make a PSTN call only when PSTN line is connected.

2. Answer Phone Calls

- Answer VoIP Calls: When somebody calls in from VoIP, the analog phone shows the caller ID. Then pick up the analog phone, LED "SIP" wink, you are in conversation.
- Answer PSTN Calls: When somebody call in from PSTN, the analog phone show the caller ID. Pick up the analog phone, LED "PSTN" and "SIP" wink, you are in conversation.

3. Phone Setting

- In Phone Setting: Include Call Forward, SNTP Settings, Volume Settings, Block Setting, Caller ID, Dial Plan Setting, Flash Time Setting, Call Waiting Setting, and T.38 (FAX) Setting functions.

- Call Forward function: You can setup the phone number you want to forward in this page. There are three type of Forward mode. You can choose All Forward, Busy Forward, and No Answer Forward by click the icon.

  a) All Forward: All incoming calls will forward to the number you choose. You can input the name and the phone number in URL field. If you select this function, then all the incoming calls will directly forward to the speed dial number you choose.

  b) Busy Forward: If you are on the phone, the new incoming call will forward to the number you choose. You can input the name and the phone number in URL field.

  c) No Answer Forward: If you can not answer the phone, the incoming call will forward to the number you choose. You can input the name and the phone number in URL field.

Forward Setting

You could set the forward number of your phone in this page.

All Forward:	⊙ Off ○ On	
Busy Forward:	⊙ Off ○ On	
No Answer Forward:	⊙ Off ○ On	

	Name	Number or URL
All Fwd No.:		
Busy Fwd No.:		
No Answer Fwd No.:		

No Answer Fwd Time Out: `3` (2~8 Ring)

Submit Reset

- SNTP Setting function: You can setup the primary and second SNTP Server IP Address, to get the date/time information.

SNTP Settings

You could set the SNTP servers in this page.

SNTP:	⊙ On ○ Off
Primary Server:	north-america.pool.ntp.org
Secondary Server:	asia.pool.ntp.org
Time Zone:	GMT + ▼ 08 ▼ : 00 ▼ (hh:mm)
Sync. Time:	1 : 0 : 0 (dd:hh:mm)

Submit Reset

- Volume Setting function: You can setup the Handset Volume and the Handset Gain.

Volume Setting

You could set the volume of your phone in this page.

Handset Volume:	10	(0~12)
Handset Gain:	10	(0~15)

Submit   Reset

a) Handset Volume is to set the volume for you can hear from the handset.

b) Handset Gain is to set the volume send out to the other side's handset.

- DND Setting: You can setup the DND setting to keep the phone silence. You can choose Always Block or Block a period.

   a) Always DND: All incoming calls will be blocked until this feature is disabled.

   b) DND Period: Set a time period and the phone will be blocked during the time period.

   c) When you finished the setting, please click the Submit button.

DND Setting

You could set the do not disturb period of your phone in this page.

DND Always:	○ On ● Off	
DND Period:	○ On ● Off	
From:	00 : 00	(hh:mm)
To:	00 : 00	(hh:mm)

Submit   Reset

- Caller ID function: you can set the device to show Caller ID in your PSTN Phone or IP Phone.

   a) There are four selections of Caller ID. You need to base on your environment to set the Caller ID function for FSK or DTMF. When you change the setting, please double check the PTT setting in Others. You need to choose the correct country code then the Caller ID will be valid.

Caller ID Setting

You could enable/disable the caller ID setting in this page.

Caller ID:	Caller ID after 1st Ring (FSK) ▾
Single Caller ID:	○ Yes ● No
CID Without Time:	○ Yes ● No
CID Type 2:	○ Yes ● No

Submit   Reset

- Dial Plan Setting function: This function is when you input the phone number by the keypad but you don't need to press "#". After time out the system will dial directly.

Dial Plan

You could the set the dial plan in this page.

Drop prefix :	○ Yes ● No
Replace rule 1:	[ ] + [ ]
Drop prefix :	○ Yes ● No
Replace rule 2:	[ ] + [ ]
Drop prefix :	○ Yes ● No
Replace rule 3:	[ ] + [ ]
Drop prefix :	○ Yes ● No
Replace rule 4:	[ ] + [ ]

Dial now:	[ ]
Auto Dial Time:	[5] (3~9 sec)
Use # as send key:	● Yes ○ No
Use * for IP dialing:	● Yes ○ No

Submit   Reset

a）Symbol description:

x or X	0.1.2.3.4.5.6.7.8.9
+	or

b) Drop Prefix:

No: When encountering the accordant rule, a new number will be added in front of the dialing number.

YES: When encountering the accordant rule, a new number will replace the dialing number.

c）Replace rule: Providing the setting number information. 7 digits number is preferred, from (0~9999999) Can be numbers or strings. Maximum length is 8 bytes.

e) Dial Now: Provides the rules for encode and decode. Maximum length is 31 digits number, can be numbers or signs (+, x). (+) means "Or"; (x) means any numbers that is from 0~9. But the first digit cannot be "0". Because 0 cannot judge the rule. So if Dial Now begins with "0", the system cannot work. Maximum length is 124 bytes.

f) Auto Dial Time: Default:5 second. After waiting for a while, but didn't input any number, Auto Dial will run automatically. Time zone: (3~9 sec). . Maximum length is 3 bytes.

Example:

*xx	If matched with one of *00,*01….*99 then will send number out
#xx	If matched with one of #00,#01….#99 then will send number out
10x	If matched with one of 100,101….109 then will send number out
11x	If matched with one of 110,111….119 then will send number out
Xxxxxxxx	If dial with 8 digits then send number out

- Flash Time Setting function: This function is for you to set the time you press the Hook to represent the Flash function.

Flash Time Setting

You could set the flash time in this page.

Flash Signal Detect (MAX):	60	x 10 ms (4~255)
Flash Signal Detect (MIN):	7	x 10 ms (7~12)

Submit   Reset

- Call Waiting Setting function: You can Enable/Disable the Call waiting function, when you are talking with someone, there is a new incoming call, and you will hear the call waiting tone.

Call Waiting Setting

You could enable/disable the call waiting setting in this page.

Call Waiting: ⊙ On   ○ Off

Submit   Reset

- T.38 Setting function: You can Enable/Disable the T.38 function.

T.38 (FAX) Setting

You could enable/disable the FAX function in this page.

T.38 (FAX):	⊙ On ○ Off	
T.38 Port:	20000	(1024~65533)

Submit   Reset

4. IVR Interface for The GW3CM

You can use the PSTN phone to configure the GW3CM. Please follow the instruction to configure your terminal adaptor.

- Please make sure the GW3CM is connected with Phone, Ethernet cable and PSTN Line.
- If the GW3CM is not registered to the SIP Proxy Server, it will switch to PSTN after 30 seconds. Also if Ethernet or Power fails, the GW3CM will switch to PSTN mode.
- The IVR default setting is disabled, if you want to use IVR please input #190# first.
- If the GW3CM is connected to PSTN, then you can use**"0*"** to switch from VoIP to the PSTN line. If you want to change to VoIP again, you have to "on-hook" and "off-hook "again.

Group	IVR Action	IVR Menu Choice	Parameter(s)	Notes
Function	enable call waiting	**#138#**	None	Enable Call waiting
Function	disable call waiting	**#139#**	None	Disable call waiting
Function	unlock keypad	**#190#**	None	You have to unlock keypad first, and then you can change the setting by keypad.
Function	lock keypad	**#191#**	None	Lock keypad.
Function	Reboot	**#195#**	None	The system will reboot automatically.
Function	Factory Reset	**#198#**	None	System will automatically Reboot and restore to default settings. WARNING: ALL "User-Changeable" NONDEFAULT SETTINGS WILL BE LOST! This will include network and service provider data.
Info	Check IP Address	**#120#**	None	IVR will report the LAN port IP address
Info	Check IP Type	**#121#**	None	IVR will report the WAN Port DHCP is enabled or disabled.
Info	Check the Phone Number	**#122#**	None	IVR will report current in use VoIP number
Info	Check Network Mask	**#123#**	None	IVR will report the WAN Port network mask
Info	Check Gateway IP Address	**#124#**	None	IVR will report the WAN Port gateway IP address
Info	Check Primary DNS Server Setting	**#125#**	None	IVR will report the WAN Port Primary DNS server IP Address.

Info	Check IP Address	**#126#**	None	IVR will report the WAN port IP address
Info	Check Firmware Version	**#128#**	None	IVR will report the firmware version
Setting	Set DHCP client	**#111#**	None	The system will change the WAN port to DHCP Client type
Setting	Set Static IP Address	**#112xxx*xxx*xxx*xxx#**	Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	WAN port DHCP Client will be disabled and WAN port will change to the Static IP type. Set WAN port IP Address
Setting	Set Network Mask	**#113xxx*xxx*xxx*xxx#**	Enter value using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	Must set Static IP first. Set WAN port Network Mask
Setting	Set Gateway IP Address	**#114xxx*xxx*xxx*xxx#**	Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	Must set Static IP first. Set WAN port Gateway IP Address
Setting	Set Primary DNS Server	**#115xxx*xxx*xxx*xxx#**	Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	Must set Static IP first. Set WAN port Primary DNS Server IP Address

Setting	Set Codec	**#130+[1-8]#**	1:G.711 u-Law, 2: G.711 a-Law, 3: G.723.1, 4: G.729a, 5: G.726 16K, 6: G.726 24K, 7: G.726 32K, 8: G.726 40K,	You can set the codec you want to the first priority.
Setting	Set Handset Gain	**#131+[00~15]#**	Handset Gain from 0~15	You can set the Handset gain to proper value, default is 10
Setting	Set Handset Volume	**#132+[00~12]#**	Handset Volume from 0~12	You can set the Handset volume to proper value, default is 10
Setting	Auto configure mode	**#137+[0~2]#**	0: Disable, 1: TFTP mode, 2: FTP mode	You can set the Auto configuration mode, 0: Disable, 1: use TFTP Server, 2: user FTP Server

5. Phone Book

Phone Book

You could add/delete items in current phone book.

Phone Book Page: page 1

Phone	Name	Number or URL	Select
0			☐
1			☐
2			☐
3			☐
4			☐
5			☐
6			☐
7			☐
8			☐
9			☐

Delete Selected   Delete All   Reset

Add New Phone

Position: _____ (0~139)
Name: _____
Number or URL: _____

Add Phone   Reset

- The Phone Book section contains Speed Dial Settings. You can setup the Speed Dial number. If you want to use Speed Dial please dial the speed dial number (from 0~9) and press "#".

- In Speed Dial setting function: You can add/delete Speed Dial number. The maximum speed dial list is 10.

6. Network Items

- LAN Settings: In this page you can configure the IP Phone LAN port's setting.

  a) The LAN port's default IP address is 192.168.123.1, Net Mask is 255.255.255.0., and DHCP Server enabled. The start IP address if 150, end IP address is 200. It is not necessary to change the LAN settings.

  b) You can connect your PC to the LAN port, set your PC as DHCP Client mode, and then you can get IP address from the GW3CM.

LAN Settings

You could configure the LAN settings in this page.

LAN Setting	
IP:	192.168.123.1
Mask:	255.255.255.0
MAC:	00156510f2a4

DHCP Server	
DHCP Server:	⦿ On ○ Off
Start IP:	150
End IP:	200
Lease Time:	1 : 0 (dd:hh)

Submit   Reset

● DDNS Setting: You can configure the DDNS setting in this page. You need to have the DDNS account and input the information properly. You can have a DDNS account with a public IP address then others can call you via the DDNS account. But now most of the VoIP applications work with SIP Proxy Server.

DDNS Settings

You could set the configuration of DDNS in this page.

DDNS:	○ On ⊙ Off
Host Name:	
User Name:	
Password:	
E-mail Address:	
DDNS Server:	
DDNS Server List:	User Input
Type:	dyndns
Wild Card:	on
BACKMX:	○ On ⊙ Off
Off Line:	○ On ⊙ Off

Submit   Reset

● VLAN Setting: You can set the VLAN setting in this page. There are two parts in this page. First one is to set the packets related to the GW3CM, and the second part is if you use the VLAN setting in the NAT Mode.

a) There are two kinds of destination packets getting from the WAN port of GW3CM, one kind of packet will go to the GW3CM, and the other will go through the LAN port to the PC.

b) VLAN Packets: if you enable the first VLAN Packets and set the VID, User Priority, and CFI, then all the incoming packets will be checked with the IP Address and the VID.

c) VID: You can follow your service provider to set your VID.

d) User Priority: Usually this will be defined by your service provider.

e) CFI: Canonical Format Indicator is always set to zero for Ethernet switches. CFI is used for compatibility reason between type of Ethernet network and Token Ring network. If a frame receives at an Ethernet port has a CFI set to 1, then the frame should not be forwarded as it is to an untagged port.

f) When you enable the first VLAN Packets and set the VID, User Priority, and CFI, then all the incoming packets with the GW3CM's IP address and the same VID will be accepted by the GW3CM. If the incoming packets with the GW3CM's IP address but the VID is different then the

packets will be discarding by the GW3CM. The Other incoming packets with different IP address will go through the LAN port to the PC.

g) NAT VLAN Setting: When you set your device in NAT mode, the GW3CM can help you to filter the wrong incoming packets. You can separate the other device connected behind the GW3CM into 4 VLAN group. You can set different VID for these 4 groups. When the incoming packets go through the GW3CM's WAN port then the GW3CM will check the VID, if the packets is not going to the GW3CM (with the GW3CM's IP address and the correct VID), and the VID is not the VID you set, then the packets will be discarded by the GW3CM.

7. SIP Settings

● In SIP Settings you can setup the Service Domain, Port Settings, Codec Settings, Codec ID Setting, RTP Setting, RPort Setting and Other Settings. If the VoIP service is provided by ISP, you need to setup the related information correctly then you can register to the SIP Proxy Server correctly.

● Service Domain Function(see section 2.4)

● Port Settings: You can setup the SIP and RTP port number in this page. Each ISP provider has different SIP/RTP port setting, please refer to the ISP to setup the port number correctly.

● Codec Settings: You can setup the Codec priority, RTP packet length, and VAD function in this page. You need to follow the ISP suggestions to set up these items.

Codec Settings

You could set the codec settings in this page.

Codec Priority	
Codec Priority 1:	G.711 u-law
Codec Priority 2:	G.711 a-law
Codec Priority 3:	G.723
Codec Priority 4:	G.729
Codec Priority 5:	G.726 - 16
Codec Priority 6:	G.726 - 24
Codec Priority 7:	G.726 - 32
Codec Priority 8:	G.726 - 40
Codec Priority 9:	GSM

RTP Packet Length	
G.711 & G.729:	20 ms
G.723:	30 ms

G.723 5.3K	
G.723 5.3K:	○ On ● Off

Voice VAD	
Voice VAD:	○ On ● Off

Submit Reset

- Codec ID Setting: Sometimes 2 VoIP devices with different Codec ID will cause the interoperability issue. If you are talking with others and come up against the problem, you may ask the other one what kind of Codec ID he uses, and then you can change your Codec ID.

Codec ID Setting

You could set the value of Codec ID in this page.

Codec Type	ID		Default Value
G726-16 ID:	23	(95~255)	☑ 23
G726-24 ID:	22	(95~255)	☑ 22
G726-32 ID:	2	(95~255)	☑ 2
G726-40 ID:	21	(95~255)	☑ 21
RFC 2833 ID:	101	(95~255)	☑ 101

Submit Reset

- DTMF Setting: You can setup the RFC2833 Out-Band DTMF, Inband DTMF and Send DTMF SIP Info in this page. To change this setting, please follow your ISP information.

DTMF Setting

You could set the DTMF setting in this page.

- ● RFC 2833
- ○ Inband DTMF
- ○ Send DTMF SIP Info

Submit Reset

- RPort Function: You can Enable/Disable the RPort in this page. To change this setting, please follow your ISP information.

RPort Setting

You could enable/disable the RPort setting in this page.

RPort: ● On ○ Off

Submit Reset

- Other Settings: You can setup the Hold by RFC, Voice/SIP QoS and SIP expire time in this page. To change these settings please follow your ISP information. The QoS setting is to set the voice packets' priority. If you set the value higher than 0, then the voice packets will get higher priority to the Internet. But the QoS function still needs to cooperate with

other Internet devices.

Other Settings

You could set other settings in this page.

Hold by RFC:	○ On ● Off
Voice QoS (Diff-Serv):	40 (0~63)
SIP QoS (Diff-Serv):	40 (0~63)
SIP Expire Time:	300 (15~86400 sec)
Use DNS SRV:	○ On ● Off
Send Keep Alives Packet:	○ On ● Off
Keep Alives Period:	60 (15~250 sec)
Jitter Buffer:	1 (0~250 packets)

Submit   Reset

8. NAT Trans

- In NAT Trans. You can setup STUN function. These functions can help your GW3CM working properly behind NAT.
- STUN Setting: You can Enable/Disable the STUN and set STUN Server IP address in this page. This function can help your GW3CM working properly behind NAT. To change these settings please follow your ISP information.

STUN Setting

You could set the IP of STUN server in this page.

STUN:	○ On ● Off
STUN Server:	stun.xten.com
STUN Port:	3478 (80~65535)

Submit   Reset

9. Save Change

- In Save Change you can save the changes you have done. If you want to use new settings in GW3CM, You have to click the Save button. After you click the Save button, the GW3CM will automatically restart and the new setting will be valid.

Save Changes

You have to save changes to effect them.

Save Changes: Save

Y-GW3CM-Ver.1.3

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