GW6200

USER'S MANUAL

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1. Introduction

Congratulate on becoming the owner of the GW6200.Your LAN (local area network) and WAN (wide area network) will now be able to access the Internet using your high-speed Ethernet connection. This Administrator Manual will show you how to install and set up your GW6200.

The IG (Intelligent Gateway) provides a variety of functions for users and can support a wide range of application in a complicated environment. Therefore, it needs Administrator to plan, install and set the whole environment of VoIP communication. And the knowledge of modern network and Internet technologies are essential for administrator.

Router Connection Diagram

Interfaces

- WAN Interface: 10/100Base-T Ethernet port
- CO Interface: 3 FXO (Loop Start, for PSTN)
- Analog Device Interface: 1 FXS (for analog Telephone or FAX)
- LAN Interface: 4 Ethernet (10/100Base-T)
- Built-in 802.11g Wi-Fi access point

Terminals

- 1 Analog Terminal
- 24 IP Stations (Wired/Wi-Fi IP Phone)

In principle, the Administration is required to do the following things:

    (1) To understand the architecture, resources, and devices of whole environment which will be involved with the VoIP communications.

    (2) To build a common setting file for most users.

(3) To configure each phone and install them into the network.

(4) To configure each interfaces and install them into IG.

(5) And to solve the problems what users encounter during operation.

2. Getting to Know the Router

2.1 Specifications

Feature	Additional Info	
Main Processor	240MHZ MIPS32 CPU	
Processor SDRAM	External 64MB	
Processor Flash ROM	External 8 MB	
Supplementary Processor	1x DSP AudioCode AC482-C (4 channel/each)	
System Flash (Voice Mail)	128MB	
Switch controller	Broadcom BCM5325E	
System Power:	15V/0.8A	
Standards	IEEE802.11g,IEEE802.11b,IEEE802.3	
Ports	1× WAN，4× LAN，1× FXS，3× FXO	
Button	Reset, Power on/off	
Cabling	RJ-45,RJ-11	
LEDs	POWER, WIRELESS, 3× LINE, TEL, 4× ETHERNET, WAN	
EMI/EMC:	FCC Part15&68 ClassB / CE Certificated	
Environmental	Operating Temp	.0ºC to40ºC (32ºF to 104ºF)
	Storage Temp.	-20ºC to 70ºC (-4ºF to 158ºF)
	Operating Humidity	10% to 80% relative humidity, Non-Condensing
	Storage Humidity	10% to 90% Non-Condensing

2.2 Front Panel
The front panel contains several LEDs that indicate the status of the unit.
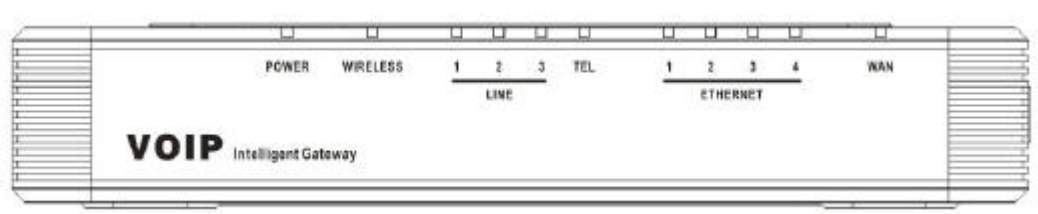
Figure 2-1. Front Panel of GW6200

LED Name	Color	Status	Description
POWER	Blue	On	Power on.
		Off	Power off.
WIRELESS	Blue	On	Wireless LAN initialization success.
		Off	Wireless LAN is not present.
LINE (1-3)	Blue	Off	PSTN Line is idle.
		On	PSTN Line is busy.
TEL	Blue	Off	Phone is idle.
		On	Phone is busy.
ETHERNET(1-4)	Blue	On	LAN connected.
		Off	LAN network connection not present.
		Flashing	LAN activity present (traffic in either direction).
INTERNET	Blue	On	WAN connected.
		Off	WAN network connection not present.
		Flashing	WAN activity present (traffic in either direction).

2.3 Rear Panel

The rear panel contains the ports for the unit's data and power connections.
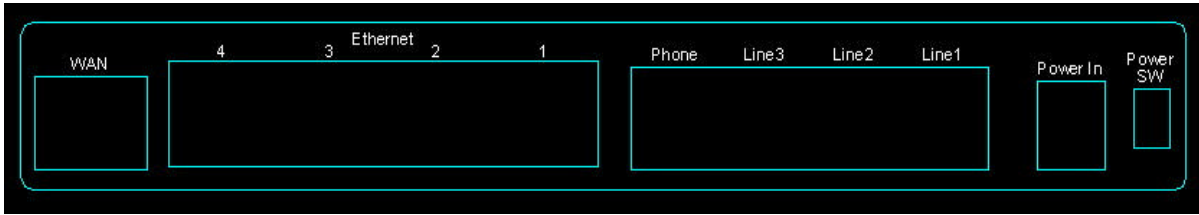
Figure 2-2. Rear Panel of GW6200

Label	Function
WAN	RJ-45 connector: Connects the device to your cable modem, or to your ADSL Modem, It's to connect to the remote network
LAN1-4	RJ-45 connector: Connects the device to your PC's Ethernet port, or to the uplink port on your LAN's hub,
TEL1	RJ-11 connector: Connects the device to your analog phone,
Line 1-3	RJ-11 connector: Connects the device to your PSTN lines
Power	Connects to the supplied power converter cable
On/Off	Switches the device on and off

3. System Features

3.1 Answering Position

For incoming calls from PSTN (FXO ports), system provides the flexibility to ring to different destinations: Auto Attendant, Operator or Extension, according to the system service mode (day or night modes). For Extension, system provides the ability to have the calls to ring to one extension or one hunt group.

It is also for those incoming VoIP calls from the registered ISP (Uplink Server) to be routed with the same method.

3.2 Auto Attendant & Voice Mail

The IG Auto Attendant can greatly enhance business office productivity by providing either a full-duty automated attendant to handle all incoming system calls or part-duty automated attendant to handle overflow traffic.

The Automated Attendant & Voice Mail Module provides 128MB of programmable voice announcements. It provides four ports to handle four (4) simultaneous callers. It provides the following functions:

- Auto Attendant Functions
  It provides an incoming caller with a customized welcome greeting that describes the actions available to an incoming caller:
  - Provide four kinds of main greeting messages: Day, Night, Noon, and Holiday greeting messages.
  - Dial an extension number or operator code to reach the appreciate destination.
  - Press a digit to leave a message in a particular mailbox.
  - Press a digit to make an outside call via the specified check.
  - Provide the phone to forward the call to its voicemail box.
  - Provide the phone to set Direct Call Forward, Busy Call Forward or No Answer Call Forward. It will forward the incoming trunk call to another extension.
  - Provide the phone to set Do Not Disturb to stop the incoming line call from ringing.
- Voice Mail Functions
  All mailbox users have access to these mailbox features
  - Delete, save, or skip messages.
  - Send messages to other mailbox.
  - Receive message information indicating the date, time, and sender information, if available.
  - Change personal greeting, and password.
  - Playback controls when reviewing messages.
  - Record a temporary greeting.
  - Send a notification via email when getting a leaving message.
  - Add attached WAV file in notifying email for leaved message.
  - The maximum recording length for each call is 180 seconds.
  - The all extension users share the same VM size. If the all extensions exist (25 extensions), the maximum total length for each user is about 10 minutes.

■ For each leaving message, it's saved 1 - 7 days
● Voice Messages
■ Provide two languages service for the all voice files.
■ Administrators can record the all voice messages by themselves.
■ Administrators can update, backup or delete the all voice messages on the PC.

3.3 Basic Call

To make an intercom call, dial a Station number (IP Terminal, POTS) or a Voice Mail number.

To make an outgoing call, dial a phone number. System chooses a line (PSTN line, IP trunk, another GW6200) via Call Routing Table to dial out. If it includes "*" in the phone number, and the call is dialed through PSTN Trunk, it will be as a one second Pause time.

To make an outgoing call, dial a PSTN, an IP Trunk or a Trunk Group number first. After hearing dial tone, dial the phone number.

3.4 Call Abandon

For every PSTN call, system provides the facility (Loop Break, Polarity Reversal, Release Tone) to monitor the call status. If the remote party hangs up, the ongoing call must be terminated.

3.5 Call Operator (Call Attendant)

One primary operator may be assigned in the system. The standard IP phone will serve as the operator telephone. When assigned as operator, this extension supports general system functions.

While the internal extension dials Operator Directory Number ("0" at default), or the outside party dials "0" when Auto Attendant plays the Welcome message, these calls will be stored into Operator Queue. The Operator is First-In-First-Out to service these calls. At the waiting time, the calling party will be on Music-On-Hold state.

3.6 Call Routing

The Call Routing feature automatically routes outgoing calls using the most appropriate route. The appropriate route is determined based on the number dialed. If necessary, GW6200 can automatically modify the dialed number by deleting and inserting digits.

The call routing destination is a PSTN line, an IP line, a line group or another GW6200.

3.7 Caller ID

GW6200 provides the ability to detect the calling party identification provided by CO via PSTN lines or by Uplink Server via IP trunks. This data when received by the telephone carrier will be displayed on all ringing IP phones or Caller ID equipped Single Line Telephone.

3.8 Class Of Service (COS)

GW6200 provides eight (8) Classes of Service (COS) for assignment of outside line dialing-privileges. Each extension may be assigned one Day-COS and one Night-COS. The Extension COS is primarily used for restriction and control of long distance dialing. COS 0 is the highest priority. COS 7 is the lowest one.

3.9 Daylight Saving Time

Daylight Saving Time (DST) feature supports auto adjustment for daylight saving time.

3.10 Direct Inward System Access (DISA) via Auto Attendant

The current PSTN/IP lines are all DISA lines. While ringing to AA, the outside callers have direct access to extensions, PSTN/IP lines, Call Routing and Trunk Groups.

3.11 Dynamic DNS

GW6200 support Dynamic DNS to see the DDNS host name as GW6200's WAN IP address in SIP application. Let the GW6200 have a static domain name for dynamic IP address.

Before using the feature, you must register domains on DynDNS.org or TZO.

3.12 Extension Password

All extensions of the IG system have an associated User Password. Password length is system programmable from 0 to 24 characters.

All extensions of the IG system also have an associated User Password. The Password is applied to Voice Mail service, and some system feature settings (Phone Lock, COS Following).

3.13 Fax/Modem

GW6200 supports FAX/modem tone detection and auto-fallback to G.711.

3.14 Flash - Analog Port (SLT) Flash Recognition

Flash is the momentary operation of the hook-switch at the analog device, which can be deciphered by the IG system in such a way that the previous call in progress is held, or placed in a status of transfer awaiting further instructions from the user.

3.15 Hunt Group

Incoming Trunk call can ring all Hunt Group member extensions simultaneously (All Ring). GW6200 supports 4 Hunt Groups. Each Hunt Group can have up to 24 members.

Incoming Trunk calls ring all member extensions simultaneously. If more than one call rings at the same time, the first agent to go off hook will be connected to the call that has been ringing the longest.

For unanswered Trunk call, it will be forwarded to a Reroute destination. The Reroute destination can be Auto Attendant or the first member's Voice Mailbox.

3.16 IG Expansion

While dialing a number, it may route to another GW6200 through Call Routing Table. It allows the user to make a direct call to an extension in another GW6200. It also allows the user to share the PSTN or IP Trunks in another GW6200 to make an outside call.

3.17 IP Trunk

GW6200 can register up to 8 SIP Uplink Servers. The extensions may make a call to the users of the Uplink Servers, or any user in the world through the Uplink Servers.
- SIP message, including INVITE, re-INVITE, ACK, CANCEL, OPTIONS, BYE,

REGISTER, INFO, REFER, SUSCRIBE/NOTIFY and REPLACE messages.
- SIP Outbound Proxy, SIP Proxy and Registrar
- Auto-Registration when power-on or period
- Session Timer support
- Support IP address, domain name, user name, display name for SIP URL
- STUN

3.18 Make an Outside Call

System provides three methods to make an outside call.
- The extension user dials the phone number directly. System automatically routes the outgoing call using the least expensive line via Call Routing Table.
- The extension user dials a Line number or a Line Group number first. After hearing the dial tone, the user dials the phone number.
- The extension presses a Line key first. After hearing the dial tone, the user dials the phone number.

3.19 Message Waiting Indication

It's a Voice Mail feature. When somebody leave messages, the router will inform the phones, and phones' LCD will display new voice mails information, and its led will flash accordingly.

3.20 Music On Hold

Any PSTN/IP line calls which placed on hold will give music to the other external party.

3.21 Night Switching

GW6200 provide Day and night settings in service mode page. And during different time, Trunk incoming call will be forwarded to different extensions according to the settings.

3.22 Numbering Plan

The Numbering Plan refers to the structure of dialed access to the various resources that are part of the system. GW6200 also allows for a very flexible configuration numbering for the various system resources.

3.23 Paging(All/Group)

A Page can be initiated from any extension in the system. Dialing a Paging Group Directory number allows an extension to broadcast a page to all assigned members of the selected paging group.

3.24 PSTN Backup

In case of power failure, GW6200 automatically switches the first PSTN line to the Single-line analog phone. The other PSTN lines are not supported.

3.25 Registration

The IG combines Proxy and Registrar servers in its application. For a Registrar server, it acts as the front end to the location service for a domain, reading and writing mappings based on the contents of REGISTER requests. The location service is then typically consulted by a Proxy server.

3.26 Remote Management via PC

Programming the IG system database is possible via PC. IG build-in a web server for remote administration.

3.27 System Speed Dial

GW6200 stores frequently dialed numbers. These Speed Dial Numbers are accessed for dialing by the associated Speed Dial Directory Numbers. The Speed Dial Directory Numbers are assigned in Numbering Table.

GW6200 stores up to 100 Speed Dial Numbers. In the Speed Dial Number, it's allowed to store Phone number up to 20 digits in length. Call Restriction and Call Routing are applied and extensions may utilize only those numbers allowed based on their extension COS.

3.28 System Time & Date

The IG system provides a built-in time clock to track System Time for reference in certain features such as System Night Service Mode Change. This clock has the ability to automatically adjust with network NTP server through Internet.

3.29 Toll Restriction

GW6200 provides sophisticated monitoring of digits dialed on PSTN/IP Trunks. If a digit or range of digits dialed on a Trunk line is inconsistent with the dialing extension's COS, the call is denied. This calling COS criteria can be applied to local calls, long distance calls, and specific numbers that are considered allowed in areas where other numbers may be restricted.

3.30 Transfer

Transfer is used to deliver calls at your extension to another extension. It means that calls can be routed to IG system destinations: an extension or an outside phone number.

3.31 Trunk Group

System assigns each PSTN Trunk and IP Trunk to a specific Trunk Group. Each Trunk can be assigned to only one Trunk Group. The Trunk group assignment is used for Trunk pool access.

If setting some PSTN trunks and some IP trunks into the same Trunk Group, the trunk access depends on the access priority.

System provides up to 4 Trunk Groups. All PSTN Trunks and IP Trunks are assigned to default Trunk Group 1 and PSTN Trunks has higher access priority.

System provides up to 4 Trunk Groups. All PSTN Trunks are assigned to Trunk Group 1 and all IP Trunks are assigned to Trunk Group 2.

3.32 Web Access Control

It has System and Phone web access level.

- System's Web Page Access Control
  Allow only one user can login system's Web page at the same time. One can't login system's Web page when there is another user has login already until the user logout or has no-action in 3 minutes.

- IP Phone's Web Page Access Control
  Allow the users access internal IP phone's Web page. There are two methods to follow.
  - Login system's Web page. In "Registered Phone" Page, all registered phones will be listed with their IP links. Click the link you can enter IP phone's Web page.
  - Use voicemail username and password of IP phone when login system's Web page. It will give a page list all registered phones with their IP links.

3.33 Wizard Setup

Wizard Setup allows system administrator to select the appropriate mode and configure the corresponding settings step by step to create a basic GW6200 operation.

The following eight operation setting items are supported

WAN Setting

LAN Setting

Wireless Basic

Wireless Security

Internet Time

Numbering Plan

IP Trunk

Call Routing Table

Appendix : Product Summary

TCP/IP Protocols

- ❑ IP Protocol (RFC 791)
- ❑ ARP (RFC 826) / RARP (RFC 903)
- ❑ ICMP (RFC792)
- ❑ TCP (RFC 793)
- ❑ UDP (RFC 768)
- ❑ SNTP
- ❑ DNS
- ❑ HTTP
- ❑ Telnet
- ❑ TFTP
- ❑ RTP
- ❑ Static Routing
- ❑ NAT with ALGs

IP Address Assignment

- ❑ Static
- ❑ Dynamic
- ❑ Subnet Mask
- ❑ PPPoE Client (RFC 2516)
- ❑ Primary and Secondary DNS
- ❑ DHCP Server (RFC 2131-2132)
- ❑ DHCP Client (RFC 2132)

Routing

- ❑ RIP v1/v2
- ❑ Static routing
- ❑ DHCP Server/Relay/Client
- ❑ DNS Relay
- ❑ NAT/NAPT
- ❑ IGMP Proxy

Virtual Server

- ❑ Virtual Server
- ❑ Port Trigger

QoS

- ❑ IP ToS function (RFC 1349)
- ❑ Priority queues for upstream traffic based on ToS field.

❑   IP Precedence

VoIP Protocols

❑   SIP (RFC 3261)

❑   SDP(RFC2327) t

❑   Real Time Protocol (RTP ; RFC 1889)

❑   MD5 (RFC3261 HTTP) digest authentication

❑   G.168 Echo Cancellation

❑   Voice Codec: G.711, G.729a and G.726

❑   Support FAX/modem tone detection and auto-fallback to G.711

IP Trunk

Support up to 8 IP Trunk, which can register to ITSP carrier.

❑   SIP message, including INVITE, re-INVITE, ACK, CANCEL, OPTIONS, BYE, REGISTER, INFO, REFER, SUSCRIBE/NOTIFY and REPLACE messages.

❑   SIP Outbound Proxy, SIP Proxy and Registrar

❑   Auto-Registration when power-on or period

❑   Session Timer support

❑   Support IP address, domain name, user name, display name for SIP URL

Digital Audio

❑   Codec:
    G.711 a-law/µ-law 64Kbps,
    G.729A/B (8Kbps),
    G.726

❑   SIP Call Offer /Answer: Codec auto capacity exchange

❑   Echo Cancellation: G.168 for each voice line

❑   Silence Detection/Suppression

❑   Comfort Noise Generation

❑   Adaptive jitter buffer

❑   Different frame size support (10,20,30,40, 50, and 60ms)

❑   Packet loss concealment

❑   Out-band (RFC2833) and In-band DTMF

Security

❑   Password protected system management

❑   User authentication for PPP (PAP/CHAP/MSCHAP)

❑   Firewall

❑   Packet Filtering

❑   Access Control List

❑   Wireless Security:
    Support WEP (64, 128-bit) encryption
    802.1x and WPA/WAP2 authentication
    MAC Address-based access control

WDS support

Configuration Management

- ❑ LAN/WAN management via Telnet interface or Web-based browser interface
- ❑ Firmware upgrade available by HTTP
- ❑ Status display and event report from Web-based management
- ❑ Settings Save and Restore
- ❑ Reset to factory default

Radio - WLAN

- ❑ Stansard : IEEE 802.1g and 802.11b
- ❑ Media Access Control : CSMA/CA with ACK
- ❑ Modulation : OFDM/CCK
- ❑ Frequency Range (Range depends on different country)
- ❑ Output Power : 15dBm (typical)
- ❑ Sensitivity : -67 (54Mbps) / -83 (11Mbps) dBm (typical)
- ❑ Data Rate : 54, 48, 36, 24, 18, 12, 11, 6, 5.5, 2, 1Mbps, auto-fallback

Remote Diagnostic

- ❑ Syslog
- ❑ Device Diagnostic, Enable remote test following:
  Test the connection to your local network,
  The connection to your Internet service provider,
  The status of PSTN Line,
  The status of IP Trunk Line,
  The status of VAA Line,
  The status of FXS
  The registered status of SIP Phones

Physical Interfaces

- ❑ One 10/100BaseT Internet port (RJ-45) for WAN interface
- ❑ Four 10/100BaseT Internet port (RJ-45) for LAN interface
- ❑ One Telephone interface (RJ-11)
- ❑ Three PSTN Line interface
- ❑ One PSTN Backup line

LED Status

LED Name	Color	Status	Description
POWER	Blue	On	Power on.
		Off	Power off.
WIRELESS	Blue	On	Wireless LAN initialization success.
		Off	Wireless LAN is not present.
LINE (1-3)	Blue	Off	PSTN Line is idle.
		On	PSTN Line is busy.
TEL	Blue	Off	Phone is idle.
		On	Phone is busy.
ETHERNET(1-4)	Blue	On	LAN connected.

		Off	LAN network connection not present.
		Flashing	LAN activity present (traffic in either direction).
INTERNET	Blue	On	WAN connected.

Power Requirement

- ❑ Input : Voltage Range 120VAC/60HZ or 220VAC/50HZ
- ❑ Output : Linear 15V DC / 0.8A, 2.1mm barrel jack

Operating Environment

- ❑ Temperature : 0~40^0 C
- ❑ Humidity : 10 to 90%, non-condensing

Physical Specification

- ❑ Dimension : 192(W) x 280(L) x 32(D) (mm)
- ❑ Notice : The changes or modifications not expressly approved by the party
- ❑
- ❑ responsible for compliance could void the user's authority to operate the equipment.
- ❑
- ❑ IMPORTANT NOTE: To comply with the FCC RF exposure compliance requirements,
- ❑
- ❑ the antenna(s) used for this transmitter must be installed to provide a separation
- ❑
- ❑ distance of at least 20 cm from all persons and must not be co-located or operating in
- ❑
- ❑ conjunction with any other antenna or transmitter. No change to the antenna or the
- ❑
- ❑ device is permitted. Any change to the antenna or the device could result in the
- ❑
- ❑ device exceeding the RF exposure requirements and void user's authority to operate
- ❑
- ❑ the device.
- ❑ FCC INFORMATIONThe Federal Communication Commission Radio Frequency InterferenceStatement includes the following paragraph:The equipment has been tested and found to comply with the limits for a ClassB Digital Device, pursuant to part 15 of the FCC Rules. These limits are designedto provide reasonable protection against harmful interference in a residentialinstallation. This equipment generates, uses and can radiate radio frequencyenergy and, if not installed and used in accordance with the instruction, may causeharmful interference to radio communication. However, there is no grantee thatinterference will not occur in a particular installation. If this equipment dose causeharmful interference to radio or television reception, which can be determined by turning the equipment off and on , the user is encouraged to try to correct the interference by one or more of the following measures:--Reorient or relocate the receiving antenna.--Increase the separation between the equipment and receiver.--Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.--Consult the dealer or an experienced radio/TV technician for help.The user should not modify or change this equipment without written approvalform TECOM CO., LTD. Modification could void authority to use this equipment.