CT-812M

VoIP Gateway

Features

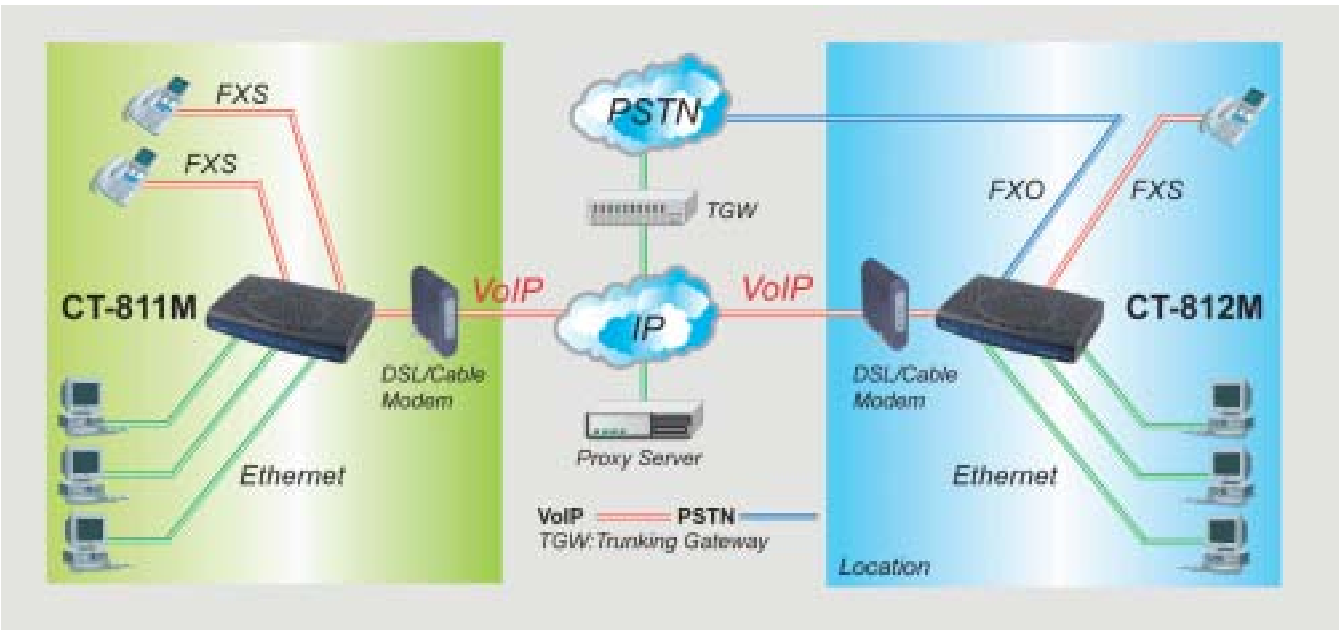
- Supports Voice over IP (VoIP)
- Uses a standard Telephone to make Internet calls free of charge
- Use just one IP address to access the Internet over your entire network
- Supports Caller ID
- Supports silence suppression
- Configurable through your Networked PC's Web Browser
- Remote administration and remote firmware upgrades over the internet
- Supports PPTP, L2TP and IpSec Pass-Through
- Supports life line for Emergency call
- Internal multi-port switch dramatically speeds up your gaming and multimedia connections

Description

Comtrend's CT-812M is a powerful VoIP Gateway, providing predictable, real-time, toll-quality voice over the Internet. The CT-812M is designed for residential and business users. It connects to an external Cable/DSL modem or internal NAT environment to access the broadband service.

Standard telephones connect to the FXS port (RJ-11 telephone jack) on the back of the CT-812M, allowing calls to be routed to anywhere in the world – significantly reducing or eliminating long distance charges. Your PCs connected to the CT-812M don't even have to be turned on to make calls. The CT-812M also has an RJ-11 port (LINE interface) for PSTN service. The CT-812M supports policy based QoS on layer 3, which provides high quality voice service. The CT-812M solves your entire network and telephony needs in one integrated unit, which reduces space requirements, and cost of hardware and cabling. This makes the CT-812M the most cost effective solution for your application.

Applications Diagram



CT-812M VoIP Gateway

Specifications

WAN Interface

Ethernet x 1	IEEE 802.3 10/100 Base-T, Auto-crossing

LAN Interface

Ethernet x 3	IEEE 802.3 10/100 Base-T, Auto-crossing

Analog Interface

- FXS x 1
- FXO x 1

Management

- Telnet, Web-based management, Configuration backup and restoration
- Software upgrade via TFTP client or FTP server

Routing Functions

- Static route, NAT/PAT, DHCP Server/DHCP Relay, DNS, ARP

Security Functions

Authentication protocols	PAP, CHAP,
VPN features	PPTP/L2TP/IpSec pass-through

QoS

- L3 policy based QoS
- Rate limit
- ToS

Voice Functions

SIP	RFC 3261	Caller ID Restriction	Yes
CODEC	G.711, G.729ab	Caller ID Presentation	Yes
RTP(RFC1889)	Yes	Call Transfer	Yes
SDP(RFC2327)	Yes	Call Waiting	Yes
AEC(Acoustic Echo cancellation)	G.168	Call Hold/Resume	Yes
VAD/Silence suppression	Yes	Call Forward	Yes
DTMF detection/generation	Yes	CFU (Call Forward Unconditional)	Yes
CNG(Comfort Noise Generation)	Yes	Dial Plan	Yes
Out-band DTMF(RFC2833)	Yes	Phone Book	Yes
Jitter Buffer	Yes	STUN (RFC3489)	Yes
Outbound Proxy (SIP)	Yes	Call Switch	Yes
Lifeline (CT-812M)	Yes	Ring Back Tone	Yes
*T.38	Yes	Busy Tone/ Dial Tone	Yes
Fax pass-through	Yes	Direct IP Peer-to-Peer Call	Yes

LED

- Power, WAN, LAN x3, ALARM, PHONE, LINE

Power Supply

External power adapter (input)	110 Vac or 220 Vac 50~60Hz
External power adapter (output)	12Vdc/1.5A

Environmental Conditions

Operating temperature	0 ~ 50 degrees Celsius
Relative humidity	5 ~ 90% (non-condensing)

Dimensions

- 205 mm (W) x 47 mm (H) x 145 mm (D)

Note* 1.Specifications are subject to change without notice.

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