COMTREND

CT-811M

VoIP Gateway

Features

- Supports Voice over IP (VoIP)
- Use 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- hrough
- Internal multi-port switch dramatically speeds up your gaming and multimedia connections

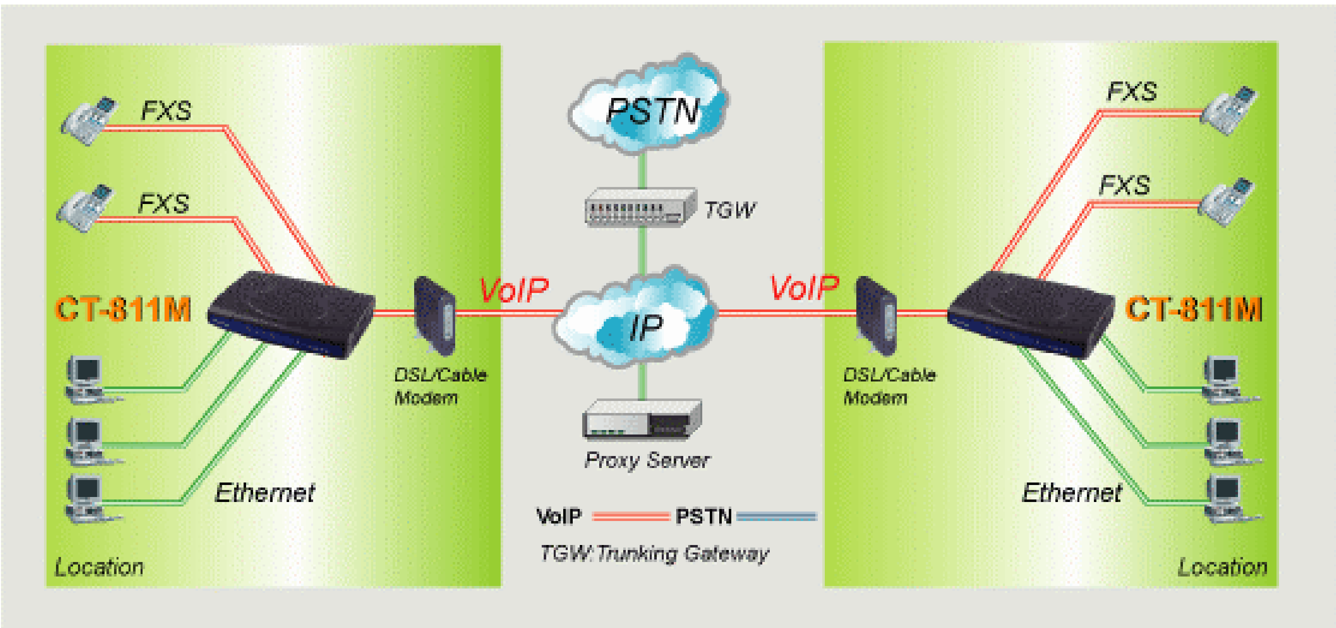
Description

Comtrend's CT-811M is a powerful VoIP Gateway, providing predictable, real-time, toll-quality voice over the Internet. The CT-811M is designed for residential and business users. It can connect to an external Cable/DSL modem or internal NAT environment to access the broadband service.

Standard telephone(s) connect to the FXS port(s) (RJ-11 telephone jack) on the back of the CT-811M, allowing calls to be routed to anywhere in the world – significantly reducing or eliminating long distance charges. Your PCs connected to the CT-811M don't even have to be turned on to make calls.

The CT-811M supports policy based QoS on layer 3, which provides high quality voice service. The CT-811M solves your entire network and telephony needs in one integrated unit, reduces space requirements, and the cost of hardware and cabling. This makes the CT-811M the most cost effective solution for your application.

Applications Diagram



CT-811M 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 2

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
Multi-user	up to 2 SIP accounts	Ring Back Tone	Yes
Direct IP Peer-to-Peer Call	Yes	Busy Tone/ Dial Tone	Yes

LED

Power, WAN, LAN 1, LAN 2, LAN 3, PHONE 1, PHONE 2

Power Supply

External power adapter (input)	110VAC or 220VAC 50~60Hz
External power adapter (output)	12VDC/1A

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: Specifications are subject to change without notice.

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