

# CT-801 VoIP Gateway

#### Features

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



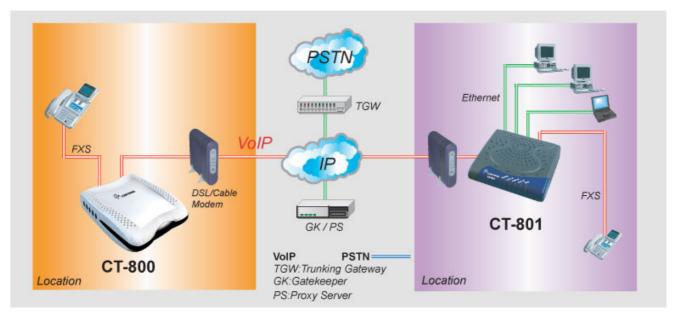
CT-801

### Description

Comtrend's CT-801 is a powerful VoIP Gateway, providing predictable, real-time, toll-quality voice over the Internet. The CT-801 is designed for residential and business users. It connects to an external Cable/DSL modem or internal NAT environment to access the broadband service.

An standard telephone connects to the RJ-11 port (telephone jack) on the back of the CT-801, allowing calls to be routed to anywhere in the world – significantly reducing or eliminating long distance charges. Your PCs connected to the CT-801 don't even have to be turned on to make calls. The CT-801 supports policy based QoS on layer 3, which provides high quality voice service. The CT-801 solves your entire network and telephony needs in one integrated unit, which reduce space requirements, and cost of hardware and cabling. This makes the CT-801 the most cost effective solution for your application.

#### **Applications Diagram**



## CT-801 VoIP Gateway

### Specifications

WAN Interface				
Ethernet x 1	IEEE 802.3 10	0/100 Base-T, Auto-crossing		
LAN Interface				
Ethernet x 3	IEEE 802.3 10	0/100 Base-T, Auto-crossing		
Analog Interface				
FXS x 1				
Management				
Telnet, Web-based management, Conf	iguration backup and re	estoration		
Software upgrade via TFTP client or F	TP server			
Routing Functions				
Static route, NAT/PAT, DHCP Server	/DHCP Relay, DNS, A	ARP		
Security Functions				
Authentication protocols PAP, CHAP,				
VPN features PPTP/L2TP/IpSec pass-through				
QoS				
L3 policy based QoS				
Rate limit				
ToS				
Voice Functions		Caller ID Presentation	Yes	
SIP RFC 3261		Caller ID Restriction	Yes	
*H.323 H.323, H.225, H.245		Call Transfer	Yes	
Codec G.711, G.723.1, G.729ab		Direct IP Peer-to-Peer Call	Yes	
RTP(RFC1889) Yes		Call Waiting	Yes	
SDP(RFC2327) Yes		Call Hold/Resume	Yes	
AEC(Acoustic Echo cancellation) G.168		Call Forward	Yes	
VAD/Silence suppression Yes		CFU (Call Forward Unconditional)	) Yes	
DTMF detection/generation Yes		CFB (Call Forward Busy)	Yes	
CNG(Comfort Noise Generation) Yes		Phone Book	Yes	
Out-band DTMF(RFC2833) Yes		STUN(RFC3489)	Yes	
Jitter Buffer Yes		Call Switch	Yes	
Outbound Proxy(SIP) Yes		Ring Back Tone	Yes	
		Busy Tone/ Dial Tone	Yes	
LED				
Power, WAN, LAN 1x, LAN 2x, L	AN 3V ALARM PHO	NE		
· · · · ·				
Power Supply	110.1/ 02			
External power adapter (input)		20 Vac 50~60Hz		
External power adapter (output)	) 12Vdc/1.5A			
Environmental Conditions				
Operating temperature 0 ~ 50 degrees				
Relative humidity	<u> </u>	5 ~ 90% (non-condensing)		

#### **Dimensions**

150 mm (W) x 42 mm (H) x 140 mm (D)

Note\* 1.Mark with \*, Please check with Comtrend representatives for its availability.

2. Specifications are subject to change without notice.





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