Rock Solid · Heart Touching

VoIP Trunk Gateway

VG504HR

The ASUS VG504HR Voice over IP Gateway is a cost-effective IP telephony solution for SMBs, this VoIP trunk gateway provides 4 PSTN ports for connection to 4 telephones and/or fax machines. This allows you to make long-distance phone calls over a data network such as the public Internet.

Advanced features of VG504HR include

- IP/PSTN Dual Access -Incoming and outgoing calls from both IP and PSTN networks use same POTS phone set.
- Dual FXO/FXS interface for connect between PBX and PSTN
- Smart routing select between VoIP and PSTN by prefix number
- Automatic reroute the call via PSTN when VoIP failure with warning sound
- Maintain the call by bypass relay when power failure
- Direct analog bypass route when call between Line and Phone
- Support call waiting notice when VoIP talking

The advantage is obvious: you use existing ISP phone charge to cover long-distance/international calls. Voice over the Internet (VoIP) can be over a DSL or cable modem line. It is an ideal solution to connecting your office to the Internet and making long-distance phone/fax calls.

www.asus.com

Networking

VG504HR

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Connect Port

- 4 x RJ-11 FXS or FXO ports
- 1 x RJ-45 10/100Mbps WAN Ethernet Port
- 3 x RJ-45 10/100Mbps LAN Ethernet Ports

Voice Processing

- Audio CODEC:
  - G.711 A-law / u-law, (64kbps)
  - G.723.1, (5.3 or 6.3kbps)
  - G.726, (ADPCM 40, 32, 24, 16 kbps)
  - G.729a/G.729b, (8kbps)
- Support T.38 FAX Relay (9.6k, 14.4k)
- Carrier tone detection and generation
- Silence suppression and comfort noise
- DTMF In/Out band Relay
- DTMF/ Call progress detection and generation
- Q.931 Fast Start
- Support Caller ID generation and detection
- Support VAD, H.225, H.245, CNG, G.168, Jitter buffer and programmable gain control

Voice Signaling

- Support SIP & H.323 s3 simultaneous VoIP calls
- Support SIP v2 Standard (RFC3261)
  - Outbound Proxy
  - STUN Server
- Register up to 4 Server simultaneously
- Support multiple dialing plan / Call hunting group
- Adaptive Jitter Buffer Function
- Support multiple dialing plan / Call hunting group
- Extensible by external IVR/CDR/Billing server for value-added application
- Support current drop and polarity reversal detection and generation on analog trunk interface
- Selectable group or sequence ring the PBX when VoIP call in
- IP screening table for authorized VoIP call in
- Flexible Routing table and profile

Management

- Web Interface Management
- Support Auto-Provision Server
- Remotely configuration/Upgrade by Web UI or Auto Provision Server.
- Support Call Detailed Records (CDR)
- 1 Reset button for load default ID/Password.
- WAN IP configure can be programmed by IVR via phone set
- Build-in watching dog for auto recovery

Router Functions

- Support static and dynamic IP from DHCP, PPPoE
- Build-in DHCP Server
- Dynamic DNS Support
- Support Network access rules(LAN to WAN & WAN to LAN)
- Self-Protection against DoS Attacks
- Support NAT through function
- Supported Protocol: UDP, TCP, NAT, BOOTP, TFTP, FTP, HTTP, TELNET, IEEE 802.3/ IEEE 802.3u
- Support SNMP, SNTP, HTTP, FTP, NAT, DNS, uPnP, DDNS
- NAT function: Virtual Server, Port mapping, ALG, DMZ, Static routing table
- Firewall option: Client filtering, URL filtering, MAC control, Drop Port scans
- VPN Pass-through (PPTP & IPSEC Pass Through)
- VPN client support! (PPTP & L2TP)
- Support QoS (ToS& DSCP) for VoIP

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