D-Link

SAVE MONEY USING INTERNET PHONE

2-PORT VOIP TELEPHONE ADAPTER

2 INTERNET PHONE CONNECTIONS

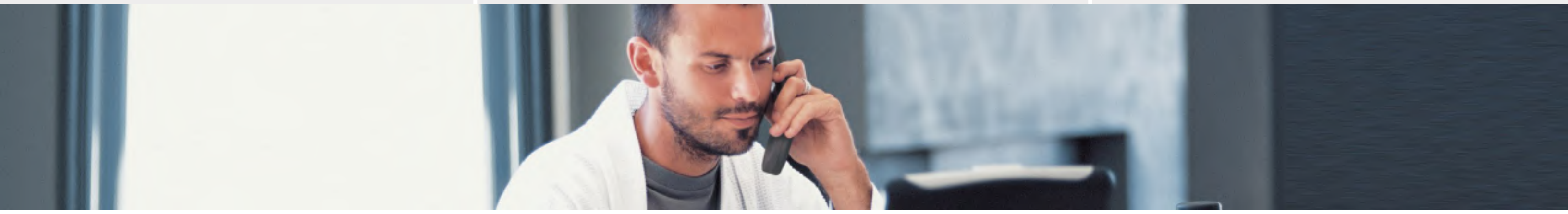
2 FXS ports connect analog phone to Internet to allow you to make inexpensive Internet phone calls

SUPPORTS MANY FEATURES

Support for call transfer, caller ID display, 3-way conference, phone book to make dialing out and answering calls more convenient

TOTAL SECURITY & QoS

Firewall and voice VLAN protection, Priority Queues for smooth voice and streaming multimedia over Internet



INTERNET PHONE MADE FOR HOME & SOHO

The D-Link DVG-5102S 2-port VoIP Telephone Adapter (TA) allows you to take advantage of your DSL/cable modem connection to make inexpensive Internet phone calls. It combines the industry's latest Voice over IP network technology with advanced communication features, and is compatible with industry wise phone service. With 2 FXS phone ports, this VoIP TA connects you to an ordinary phone set to let you make Internet phone calls.

SUPERIOR VOICE QUALITY

The DVG-5102S incorporates Quality of Service (QoS) to ensure that voice received through the Internet is the same as or even surpasses that received on the ordinary phone. It supports many useful functions such as call transfer, caller ID display, 3-way conference, phone book, speed dialing and hot lines to make it convenient to dial out or answer phone calls.

COMPLETE SECURITY

The DVG-5102S supports voice VLAN to isolate your voice communication so it cannot be tapped over the network. It also provides various types of DOS protection in an attempt to make computer resources unavailable to its intended users.

BROADBAND

DVG-5102S

D-Link®

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WHAT THIS PRODUCT DOES

Connect the DVG-5102S to up to two ordinary phone sets and make phone calls anywhere in the world using the Internet. This VoIP TA lets two people make Internet phone calls at the same time. Furthermore, it provides convenient Interactive Voice Response functions. Users are able to get query and setup the device with a phone set without turning on the PC.

CLEAR, SMOOTH VOICE OVER THE INTERNET

This VoIP TA lets you allocate network resources while guaranteeing Quality of Service (QoS). Network bandwidth management delivers smooth and clear voice communication over the Internet while increasing productivity and efficiency by tailoring your system to specific demands such as time-sensitive VoIP and multimedia applications.

TECHNICAL SPECIFICATIONS

IP Network Specifications

WAN: Static IP, PPPoE, DHCP, PPTP and L2TP
NAT Functions
Support up to 250 Clients
Port Forwarding (Virtual Servers) DMZ
Support IPv4, IPv6(optional)
QoS Support:
WAN: DiffServ, IP Precedence, Priority Queue
Rate Control, 802.1Q (VLAN Tagging),
802.1p (Priority Tag) LAN: Rate Limit
DDNS Support
Network Protocol Support:
IP, TCP, UDP, TFTP, FTP, RTP, RTCP, ARP, ICMP, NTP,DHCP, STUN Client, HTTP, HTTPS, DNS, DNS SRV, Telnet, UPnP, IGMP, IGMP snooping, IGMP proxy, RTSP ALG, SIP ALG

Voice Features

+ G.722 64kbps, G.711 a/μ-law, G.729A, G.726, G.723.1 GSM 6.10 Full Rate, iLBC 13.3 kbps
+ DTMF Detection and Generation
+ Silence Suppression & Detection
+ Comfort Noise Generation (CNG)
+ Voice Activity Detection (VAD)
+ Echo Cancellation (G.168)
+ Dynamic Jitter Buffer
+ Call progress tone detection (FXO) and generation (FXS)
+ Programmable Gain Control
+ Inbuilt Local Mixer ITU-T V.152 Voice-band Data over IP Networks

Device Management

+ Web Based Configuration
+ Telnet command line interface (CLI)
+ IVR Configuration
+ FTP / TFTP / HTTP Software Upgrade
+ Configuration Backup and Restore
+ Factory Defaults
+ TR-069, TR-098, TR-104
+ TR-111 part I & II (DHCP option 125)
+ DHCP option 43, 60 Auto Provisioning
+ SNMP V3/ V2c/ V1(optional)

SIP Account Management

+ By port registration
+ By device registration (share account)
+ Mixed mode
  (Hunt number for inbound, by port number for outbound)
+ Invite with Challenge
+ IP Address or Domain Name registration
+ Support RFC3986 SIP URI format

SIP Method Support

+ ACK, BYE, CANCEL, INFO, INVITE, MESSAGE, NOTIFY,
+ OPTIONS, PING, PRACK, PUBLISH, REFER,
+ REGISTER, SUBSCRIBE, UPDATE

SIP Call Features

+ Peer to Peer Call
+ Call Hold / Retrieve
+ Call Waiting
+ Call Pick Up
+ Call Park / Retrieve (SIP Server Required)
+ Call Forward - unconditional, busy, no answer
+ Call Transfer - attended, unattended
+ Do Not Disturb
+ Speed Dialing
+ Repeat Dialing
+ Three-way Calling
+ MWI (RFC-3842)
+ Hot Line and Warm Line

SIP Call Management

+ Support Outbound Proxy
+ Support SIP Compact Form
+ SIP Registration Failover
+ Group Hunting
+ P-Asserted-Identity per RFC3325
+ Privacy Mechanism per RFC3323
+ Session Timers (Update / Re-invite)
+ DNS SRV Support
+ Call Types: Voice / Modem / FAX
+ User Programmable Dial Plan Support
+ Automatic Calling Number Manipulation
+ CDR by RADIUS client
+ Manual Peer Table (for P2P calls)
+ E.164 Numbering, ENUM support

Telephony Specifications

+ Configurable Payphone charging pulse interval by SIP OPTION
+ In-Band DTMF, Out-of-Band DTMF Relay (RFC2833 or SIP INFO)
+ DTMF / PULSE Dial
+ Caller ID Generation / Detection: DTMF
  FSK-Bellcore Type 1 & 2
  FSK-ETSI Type 1 & 2
  FSK-NTT
+ FSK: Calling Name, Number, Date and Time, VMWI
+ FXS metering pulse options: Polarity Reversal
  12kHz calling tone
  16kHz calling tone
+ Configurable Payphone charging signal interval by SIP OPTION
+ Polarity Reversal Detection (FXO) and Generation (FXS)
+ T.30 FAX passthrough, T.38 Real Time FAX Relay
+ Call Feature enable/disable via phoneset

Security Specifications

+ DIGEST Authentication
+ MD5 Encryption
+ DoS Protection
+ Caller Filter by IP address
+ IP Filter
+ SIP/TLS and sRTP(optional)

Physical

+ WAN : 1 x 100 baseTx, auto cross-over, auto speed negotiation, RJ-45 connector
+ L A N : 1 x 100 baseTx, auto cross-over, auto speed negotiation, RJ-45 connector
+ Telephone : RJ11
+ Factory default
+ Reset button
+ Power jack

CE FC

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