Ping Communication
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Bad sound kills conversations

NetPhone Adapter 201E

Upgrade your conversations by better sound through our new generation IP-phone adapter.

During the past 5 years it has become more and more popular to try out IP-telephony, due to both the flexibility and the low cost of usage.

To avoid bad sound experiences, as on some low cost adapters, this unit is using the new generation technology to enhance sound quality, avoid delays and to ensure you get the best sound in your phone set, ever.

Technically this has been made possible by using the best components available, in addition to two new technologies called StreamEngine™ and NetEQ™.

StreamEngine™ ensures that the outgoing Voice/datastream is being prioritized so there will be no interruptions in the sound.

Global IP Solutions (GIPS) VoiceEngine uses advanced, patented algorithms to ensure that, even in the most challenging network conditions, the incoming data stream generates the highest quality conversations.

Thereby you get not only a superior hardware quality but also two technologies giving the best out- and ingoing data & voice stream, so you will probably have the best sound experience ever on a phone line.

Advantages:

- Makes your normal analog phone work like you know it now, just using your internet connection.
- No phone line subscription needed as this uses your Internet connection.
- High Quality Sound by using the Ubicom StreamEngine™ and Global IP Solutions (GIPS) VoiceEngine technology, probably even better than your traditional telephone.
- Cheaper rates from your own VoIP Service Provider, compared to a fixed land line.
- Not just one phone line but two telephone plugs with two different phone numbers, if you want.
- "Micro-Switchboard" enabling you to transfer calls from one phone to the other and to dial internally between two phone lines.
- No need to manually set it up. If supported by your operator this can be handled automatically using the auto configuration utility.
- Option to keep your current phone number.

ping communication

IP Telephony

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Switch / LAN

Technical Specifications

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Background:

People having tried voice-over IP phone services might have suffered from poor voice quality, delays, voice-overlap, and dropped calls. Most users believe this is caused by the Internet, but actually some problems are caused by your home network and its configuration. The main bottleneck in a voice call on the Internet is your home Internet connection. Most users have limited bandwidth on the Internet connection, compared to the home network itself. This creates a natural bottleneck.

This bandwidth is shared between all network activities, such as voice, e-mail, web browsing, and file-sharing (P2P). Therefore it usually causes problems on the network. Since this is data, it will not really be shown and is usually not a big problem, but when using voice (phone calls) and gaming etc. it will result in poorer quality and performance and drop outs. Then it's suddenly becoming vital how to handle the data.

Usually there are problems with both in- and outgoing data, creating delays and interruptions in the conversations. The Pingcom Phone adapter provides solutions to both of these issues.

IP Telephony:

The Pingcom NetPhone adapter uses StreamEngine™, which will automatically detect VoIP traffic and ensure safe delivery to the Internet. The Pingcom NetPhone adapter also includes GIPS NetEQ™, an advanced jitter buffer and packet loss concealment that results in dramatic improvements in sound quality for the incoming VoIP stream.

NetEQ™ maintains very high voice quality even under high packet loss conditions. Priorities of these voice applications may be assigned automatically or may be strictly provisioned by the service provider.

Ease of use:

The Pingcom NetPhone adapter can be remotely managed by the service provider. It supports the TR-069 standard from DSL Forum, working together with any standard TR-069 ACS device configuration system. It includes provisioning of the gateway and the VoIP clients as well as supporting TR-111 for proxying other TR-069 compliant devices through the router.

The Pingcom NetPhone adapter is designed for performance and QoS handling, and it is prepared for remote monitoring in real-time. It uses the Owera OPP protocol, a protocol providing realtime bi-directional connectivity to the Pingcom NetPhone adapter's system and QoS engine.

The Owera OPP protocol will automatically traverse additional routers ahead and establish the desired connection. The OPP client is a highly efficient Remote Procedure Call (RPC) engine that can run monitoring algorithms for VoIP quality, online gaming conditions, or simply log the results from the StreamEngine™ QoS engine to the service provider's management system. The OPP client can also be used for network diagnostics to help troubleshoot undesired situations.

Data

General

- Ambient temperature: < 45 Celsius
- Power Consumption < 6W
- Boot time (from power ON to SIP registered): <5 sek

Router features

- 1 lan port.
- Security—Password protected administration, Admin and
- User Access Authority, HTTP/HTTPS-support
- DNS Relay Static or WAN Assigned DNS Servers
- DHCP Server
- Port Range / Service Filtering
- WAN IP Address– DHCP, PPPoE, PPTP, L2TP*, BigPond
- MTU Setting
- MAC Address Filtering
- Event Logging
- Severities; 1K Entries
- Dynamic DNS
- Current Time & Daylight Savings Time Adjustment, Network Time Protocol
- DHCP Client
- QoS (Quality of Service), WISH, GIPS
- StreamEngine™, ToS/CoS, DiffServ, 802.1p *
- 1 x 10/100 Mb Ethernet WAN
- 1 x 10/100 Mb Ethernet LAN
- LED's (power, LAN, WAN, 2FXS, Status)

VOIP features

- SIPv2 – Session Initiation Protocol ver.2 (RFC3261
- Re-register/Re-invite, Digest authentication with MD5, redundant proxy, SIP Option, Alert-Info header
- SIP/SDP RFC compliancy
- 2 x POTS (FXS) for analog phone or fax
- Hosted PBX control & DTMF
- In-band/Out-of-band, SIP Info, SIP Subscribe-Notify
- NAT traversal with STUN (RFC3489) or Symmetric response Routing (RFC3581)
- Voice Algorithms—G.711 (A/u-law), G.729AB (optional: G.723.1, G.726 (16/24/32/48 kbps)
- Ring Tones—progress, dial, busy, network busy, call wait, ring back, call hold,flash, power ring tone (SLIC)
- Global IP Solutions (GIPS) VoiceEngine featuring NetEQ - advanced jitter buffer and packet loss concealment module, VAD - Voice Activity Detection, CNG - Comfort Noise Generation, LEC-Line Echo Cancellation G.168), Attenuation / Gain Adjustments, Dynamic volume control Phone user services—T.38 fax, G.711 fax pass through,
- Dial Plan, Redial, CLIP/CLIP2. generation, 2 separate lines,
- conferencing, 3 way call (CO mixed), Call waiting support
- Data Networking—MAC address (IEEE 802.3), IPv4, ARP,
- DHCP Client, ICMP, TCP, UDP, RTP, RTCP, PPPoE, TOS, DiffServ
- Local or central call handling
- Emergency call handling
- Calling/called party control
- Audible message feedback SIP
- ALG enable/disable

Provisioning

- Configuration and reporting integrates to xAPS provisioning system
- Provisioning via TFTP, TR-069, OPP or customized
- Configuring via web interface, adaptable look and feel
- Secure (Authenticated & Encrypted)
- Manageable (User Groups, Monitoring, Reporting)
- Easy configuration web wizard, Web based interface—HTTP v1.1

IP Telephony

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