CU3

USB Soft Phone

Basic Features

- Complete IP phone to plug into a USB port
- Requires no additional power adapter
- Includes sound device and earpiece
- Session Initiated Protocol (SIP RFC 3261)
- Caller ID display
- Display of call duration
- Call forward
- Call transfer
- Call hold
- Dynamic selection of codec
- Audio: G.711 A/μ, G729A, G.723.1
- DTMF: Inband, RFC 2833, SIP INFO
- Do Not Disturb (DND)
- User friendly tool tips over items
- Message Waiting Indicator (WMI)
- Call history
- Phone book

Enhanced Features

- Voice recording on demand
- Automatic voice recording
- Network status monitor
- Packet delay detection and monitor
- Packet loss recovery
- Advanced jitter buffer management
- Advanced traversal of NAT and firewall
- Auto answering
- Multiple user profiles
- Three call appearances
- 3 party call conference
- Support for multiple languages: English and Chinese
- Standards Supported

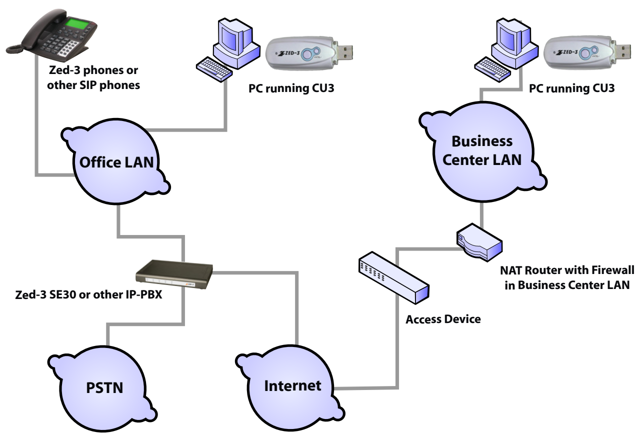
Overview

The CU3 is a complete VoIP telephone. It can plug into most PCs and connect to most IP telephony networks without requiring a sound card on the PC. Further, the software executes directly from the CU3 requiring no installation of software on the PC. This gives a product that is compact to carry, is simple to use, and provides excellent sound quality.

The CU3 is the size of a USB memory stick and is inserted into a free USB port on the PC. The CU3 includes a complete audio system; the earpiece that is provided plugs directly into the CU3 so there is no reliance upon a sound card that may or may not be present in the host computer. By embracing the open standard Session Initiated Protocol (SIP), the CU3 can connect with almost any other soft phone, IP phone, IP phone system, or IP telephony network.

NAT Traversal

The CU3 incorporates the Zed-3 SP3 soft phone. This phone has the advanced ability to traverse any NAT or firewall. This allows the phone to connect to the host system of your choosing from any external network that may include hotels, airports, coffee shops, and book stores with Internet access.



Making and Receiving Calls

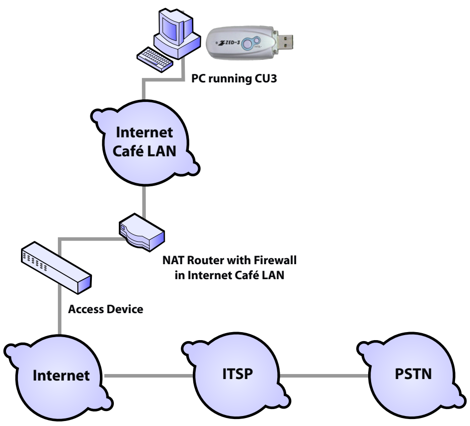
The CU3 has three call appearances, allowing you to maintain three voice conversations simultaneously. Each call appearance is treated as a separate call so it is easy to know which call you are on, and you can switch between calls by a click of a button. The Auto Accept answer feature allows you to answer any incoming call without the need to even touch your computer.

Handling Calls

The CU3 also provide standard PBX functionality: transfer a call, put a call on hold, forward calls, and return a call from the call log.

Voice Call Recording

The SP3 allows you to record your calls when you want or set it up so it will record every call automatically for you. All the recorded messages are saved locally on your computer as a standard Wav file.



Phone Book

The phone book allows you to store phone numbers and information regarding each number. The call log presents a list of the 10 numbers dialed out, received, and received. Each entry in the log displays the date and time stamp, duration, and the caller ID.

Conference

The CU3 supports three-way conference. The conference is easily set up with any combination of inbound and outbound calls. Individual members can join or leave the conference at any time. The conference can be put on hold, allowing the other parties to continue without the host. The host can use a free call appearance to place or answer another call.

USB Memory

The CU3 has additional memory that you can use as a regular storage device. The phone's profiles and phone books are saved in this area, which is about 60 MB.

Minimum Operating Requirement

Operating system: Windows 2000 or XP
Processor speed: 1.5 GHz
Memory: 512 MB RAM
Available disc space: none
IP network connection: 35 kb/s in G.729 or 75 kb/s in G.711

Physical and Environmental

Size: 90 mm (L) x 28 mm (W) x 13 mm (H) (3.5" x 1.1" x 0.5")
Operating temperature: 10°C to 35°C (50°F to 95°F)

Standards Supported

- RFC 2327 – SDP
- RFC 2976 – SIP INFO Method
- RFC 3261 – SIP v2
- RFC 3264 – Offer/Answer model with SDP
- RFC 3265 – SIP-specific Event Notification
- RFC 3515 – SIP REFER Method
- RFC 3515 – A Message Summary and MWI
- RFC 2833 - RTP Payload for DTMF Digits

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