



VoicePath VOIP6900 Intelligent Multifunction Gateway

CyberPath VoicePath VOIP6900 allows you to create a PBX that rivals the features and functionality of traditional telephony switches. VOIP6900 is cost-effective, low maintenance, and flexible enough to handle all voice and data networking. VOIP6900 can replace an existing switch or complement a PBX by adding VoIP, voicemail, conferencing and many other capabilities. VOIP6900 interoperates with most standards-based IP telephone handsets and software. Analog phones and ADSI screen phones are also supported.

In additionally, VOIP6900 has following functions as well:

- SIP Server: support SIP registration
- DHCP Server
- PSTN Gateway
- E1/T1 and FXS/FSO interface: can be easily connected with POTs phone, PSTN, and traditional PBX.

Capabilities

- Server-based System with high performance main board and CPU with 512MB RAM expandable to 1GB RAM.
- External interfaces
 - 2 RJ45 10/100/1000 Mbps Ethernet ports
 - 1 to 16 T1 or E1 ISDN/PRI interfaces to Public Telephone Network.
 - 1 to 16 FXO/FXS interface to PSTN or POTS
- Support Robbed Bit Signaling (RBS) and Primary Rate ISDN (PRI) protocol
- VoIP Gateway for originating/terminating phone calls to/from any VoIP Gateway, SIP or H.323 Devices and Applications.
- VOIP6900 to VOIP6900 interconnections with super high performance and lowest IP bandwidth utilization
- Support up to 800+ concurrent phone calls
- Support G.711, G.723, G.729 and GSM Codecs
- Intelligent Call Routing Algorithm and Session Control.
- Web-based configuration tool that could be easily integrated with Service Provider's OSS software

Features

- **IP PBX Features**
 - Voice Prompts and Directory Lookup
 - Music on Hold; Music on Transfer
 - Call Waiting
 - Caller ID, Caller ID Blocking, and Caller ID on Call Waiting
 - Call Forward on Busy; Call Forward on No Answer; Call Forward Variable
 - Blind Call Transfer and Monitored Call Transfer
 - Call Parking, Call Retrieval
 - Remote Call Pickup
 - Do Not Disturb
 - Block Calls from Telemarketing
 - Three-way calling
- **Conference Call**
 - Support reserved or non-reserved conferencing
 - Support up to 100 conference rooms. Each conference can have up to 100 people
 - Support multiple conferences with multiple channels at the same time
 - Conference room recording and playback in WAV, MP3 or GSM formats
- **Voicemail Features**
 - Voicemail box for each user. Total recording time up to 400 hours
 - Message indicators
 - Shutter dial tone on traditional phones, or
 - Message Indicator LED on some supported devices, like Cisco 7960 IP phones.
 - Message notification
 - Email notification or,
 - Email notification with .wav file attachment
 - Voicemail retrieval/handling through the phone interface
 - User login with password protection.
 - Voice prompt for each operation.
 - Play, repeat, previous, next, delete and save
 - Message forwarding to another user
- **IVR (Interactive Voice Response) system**
 - DTMF detection
 - Flexible configuration and recording tools
 - Multiple level voice menus and digit read back