

pbxnsip

"The voice communications platform of choice."

Communications Server 410

The pbxnsip CS 410

The small office appliance from pbxnsip integrates analog telephone lines within the pbxnsip software. It alleviates the need to run the software on a separate PC reducing the power consumption and administrative effort. Using solid-state hardware minimizes the risk associated with hard disks and CPU fans.

It is the perfect solution for small to medium sized businesses (SMB) that want to take advantage of VoIP's benefits and have the option to keep their existing phone lines. For example, local calls can be terminated through the FXO lines while international calls can be sent through the Internet.

Modern System Design

The CS 410 is built on a System on Chip (SoC) design that integrates all essential PBX functions one chip. This reduces the hardware costs and increases the system stability. Using standard memory like NAND-Flash makes it possible to store large amounts of data in the system. A digital signal processor echo cancellation and gain control on the PSTN side.

Hardware and Interconnectivity

The pbxnsip CS 410 contains 4 integrated FXO ports to connect to existing PSTN trunks and the system also supports IP trunks to connect to Internet Telephony Service Providers that support SIP.

Key Benefits

- Built-in 4 Port FXO PSTN Gateway
- 10 Extension Support (CS 425: 25 Extensions)
- Paging and Music on Hold audio connectors
- Full PBX features including:
 - Voicemail, Auto Attendant, Intercom, Paging, Conference Server, Hot Desking, Agent Groups
 - Cell Phone Integration, find me/follow me
 - SIP Trunking, NAT support
 - Full version 2.0 feature set

The CS 410 comes with two Ethernet ports. This makes it possible to use a private IP address in the LAN and at the same time connect phones in remote locations to the public IP address on the second Ethernet interface.

Full Feature Set

The system includes the features from the other pbxnsip PBX editions. This includes standard features like voicemail, auto attendant or conferencing, but also advanced features like call barge in or cell phone integration.

Popular CPE PBX Features

The CS 410 supports music on hold input from a regular audio jack input. Music on hold can be provided from CD and MP3 players as well as local radio stations. The system has a paging output connector. This makes it possible to use existing overhead paging infrastructure. In the LAN the system can also send RTP multicast traffic and use multicast-enabled devices for office audio paging.

Plug and Play

The CS 410 IP PBX supports plug and play of popular IP phones like Polycom, snom, Aastra, and Linksys. The PBX itself can be managed through the web interface and remotely through SSH. It also supports SNMP for remote quality supervision with standard network management tools.



The complete Internet Telephony System

- Plug & Play and Remote Management
- Integrates with Microsoft SIP-compatible products
- Full Security Support (TLS/SRTP)
- HTTP-based management and full OS access
- Now with 1 GB of built-in flash memory
- Low power consumption, fanless operation (less than 7 watts)

Technical Specifications of the pbxnsip CS 410/425

Main Features

- SIP compliant
- Built-in 4 Port FXO PSTN Gateway
- 2 x RJ-45 Network Interfaces
- Music In and Paging Out audio connectors
- Small size and fan less operation
- T.38 (FAX) relay

Accounts

- Auto Attendant
- Extensions
- Paging Groups
- Hunt Groups
- Agent Groups
- Outbound Call Authentication
- IVR Nodes
- Service Flags

Security Features

- TLS and SRTP support
- HTTPS web interface
- Secure provisioning
- Password and PIN per extension
- SSH access to the system

Mobility Support

- Call forking to cell phone
- Voicemail triggers call to cell phone
- Inbound call cell phone detection
- Camp on from cell phone
- Hot desking support

Call Supervision

- Call barge in
- Training mode
- Listen in
- Permission and license-based access

Plug and Play

- TFTP, HTTP and HTTPS support
- One-shot password provisioning
- Time zone provisioning
- PBX provisioning and configuration templates

Trunking

- B2BUA (IP-Gateway) architecture
- Registration and gateway trunks
- ENUM support
- CO-line emulation
- ANI number presentation
- DID routing
- Extension-based dial plans

MoH, Paging and Intercom

- Multiple MoH sources (RTP, File, audio input)
- Multiple audio paging output (audio output, RTP multicast)
- Intercom through star code

Auto Attendant

- Dual language support
- Prerecorded standard destinations
- Day/night mode, holidays
- Dial by name
- Anonymous call intercept
- Black and white list management
- Camp on

Conference Subsystem

- Conference mixer
- Instant conference
- Conference scheduler with Email invitation

Voicemail System

- Private and shared voicemail
- Voicemail notification through email
- Message Waiting Indication (MWI)
- Voicemail commenting
- Support for external voicemail system (e.g. Microsoft Exchange™)

Waiting Queues

- Up to ten announcements
- MoH mixing with announcements
- Agent recovery time
- Call pickup from queue
- Call escalation
- Day/night mode, holidays
- Web based queue status display

Hunt Groups

- Serial and parallel search
- Day/night mode, holidays
- Distinctive ringing
- Custom IVR recordings and routing decisions (IVR Node)

Call Redirection and Treatment

- Do not disturb
- Redirect on busy, timeout, always
- Multiple registrations per extension
- Call park, call pickup, call retrieve
- Star code-based transfer
- Last call return, redial
- Caller-ID blocking

Address Book

- Personal/domain level address book
- Address book import

Presence and Instant Messaging

- Presence agent for presence information
- Instant Messaging support
- Support for dialog state (BLF) System Management
- Loadable SSL/TLS certificate
- Performance monitoring and load protection
- SNMP agent
- CDR export through SOAP interface
- Build-in session border functionality for remote offices

Languages

- Multiple interface languages
- Multiple audio languages
- Multiple simultaneous time zones

