

# Snom India IP-PBX

## The complete IP Telephony System

### Component-based approach

The SI IPPBX product makes it possible to choose best-of-class components to set up the corporate communication infrastructure. You may choose the hardware platform of your choice (Windows-based or Linux-based), you may choose your server hardware, your PSTN termination, your handsets and you may integrate new devices like WiFi cell phones into your system.

### Localization and Configuration

The PBX comes with selected pre-recorded languages (e.g. US-English, German, French, Spanish, and Russian); other languages can be added easily. The auto attendant is able to prompt the user to select a language, which makes it possible to use the PBX with dual language requirements.

In order to support easy and safe installation, the PBX includes a TFTP server that generates configuration files for selected user-agents on the fly (e.g. Snom 7.x versions, Cisco phones models 79xx version 8.x, Polycom 2.x versions ). Other user agents can easily be added by simple configuration file templates.

### Security and Monitoring

The PBX protects the caller's privacy by using the latest IETF standards (TLS, SRTP, and SDES). Operators have access to calls by barge-in, teach-mode or listening-in.

Calls can be recorded to the file system or sent to a SIP-based monitoring port. The PBX maintains the security context, so that these features are also available when SRTP is being used.

### Paging and Intercom

The PBX supports these popular functions by using standard SIP and multicast RTP. This makes it possible to implement small paging groups up to large corporate announcement systems. Music on hold can be sourced from files and RTP streaming.

### PBX Functions

The PBX supports hunt groups and waiting queues. All services can be run in day/night mode. If necessary, simple IVR dialogs can be set up and external databases can be queried for routing purposes. A built-in conference mixer is able to host unmanaged conferences. In addition to the built-in mailbox system, external, SIP-based voice mail systems (for example, Microsoft® Exchange 2007) can be integrated.

### NAT and Mini-SBC

The service runs on hosts with multiple IP addresses. For example, several private and public IP addresses can be mixed. A built-in mini-session border controller makes sure that devices behind NAT can be registered to the PBX. This makes it possible to run the service for extensions in the LAN and at home offices.

The PBX also supports the allocation of public IP addresses using STUN. This is necessary for some ITSP in order to present them with a routable address.

## Cell Phone Support

The PBX supports multiple registrations per extensions. The user may decide to ring the cell phone after a certain time when the extension does not pick up. When receiving a voicemail message, the PBX may call the cell phone and read the message out. For international calls, PBX users can call from their cell phones into the PBX to place international calls using the PBX's dial plan and rates. This significantly reduces the communication costs for international oriented companies.

## Management

Management costs make up a large portion of the total cost of ownership when running services like a IP-PBX. The SI IP-PBX significantly compared to other products. Using only the file system makes backup and redundancy easy. Most of the administration can be done from the web interface of the PBX.

A built-in SNMP agent makes it possible to integrate the PBX into the overall corporate monitoring system

## Presence and IM

The PBX supports sharing presence information and instant messaging (IM). This makes it possible to operate these services independently from public vendors. This has the advantage that users do not get distressed by public IM traffic during work hours. A built-in SNMP agent makes it possible to integrate the PBX into the overall corporate monitoring system.

## Busy Lamp and CO-Lines

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## Standard based

- SIP (RFC 3261)
- PRACK (RFC 3262)
- DNS SRV (RFC 3263)
- SIP Offer/Answer with SDP (RFC 3264)
- Subscription for MWI (RFC 3265)
- ISUP to SIP Mapping (RFC 3398)
- SIP extension for IM (RFC 3428)
- SIP Proxy-to-Proxy Extensions (RFC 3603)
- Real-time Transport Protocol (RTP) (RFC 1889)
- Session Description Protocol (SDP) (RFC 2327)
- E.164 (RFC 3824)
- NAPTR (RFC2915),
- rport (RFC3581),
- REFER (RFC3515)
- Indicating User Agent Capabilities in SIP (RFC 3840)
- In-band DTMF/out-of-band DTMF/SIP INFO DTMF
- Draft-levy-sip-diversion-08.txt
- SNMP v2 RFC 1905
- The Transport Layer Security (TLS) Protocol (RFC 2246)
- IPsec RFC 2401
- SSH Protocol (RFC 4251)

# Technical Data

## Main Features

- SIP compliant (RFC 3261)
- Server Based Windows & Linux
- 1000 Users Per Server
- Support for unlimited no of end points via server farms and clustering
- Multiple Redundancy support
- Distributed call processing
- Server Client Architecture
- Scalable to 1 Million BHCC\*
- Web Based Management
- Automatic Bandwidth selection
- Integrated Session border controller
- Built in SNTP server

## Administrative Features

- Multilevel Administration
- Performance Mail
- System Performance Configurable
- Debug Information

## 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

## 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

## Trunking

- B2BUA (IP-Gateway)
- Registration and gateway trunks
- ENUM support
- CO-line emulation
- ANI number presentation (RFC3325)
- DID routing
- Extension-based dial plans

## Codec's, Audio

- G.711 A-law,  $\mu$ -law
- G.729A, G.726
- GSM 6.10 (full rate)
- G.722
- Comfort noise,
- voice activity detection

## Plug and Play

- TFTP, HTTP and HTTPS support
- Time zone provisioning
- PBX provisioning and configuration
- Templates

## Auto Attendant

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

## Conference Subsystem

- N Party Conference\*
- Conference mixer, Barge
- Instant conference
- Conference scheduler with Email Invitation

## Voice Mail System

- Private and shared voicemail
- Voicemail notification through email
- Message Waiting Indication (MWI)
- Voicemail commenting
- Voice Mail Storage\*
- Support for external voicemail system (e.g. Microsoft exchange)

## Hunt Groups

- Day/night mode, holidays
- Web based queue status displays
- 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 and domain level address book
- Address book import

## Presence & 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

\*Depend Upon Server Hardware

For more information, contact

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