LessWires Advanced IP Soft-PBX System

Our IP soft-PBX is a complete communications platform. In addition to having PBX functionality, the system is a soft-switch, a protocol gateway, a media server, and a VoIP gateway. Fully compatible with a wide range of IP and analog protocols and codecs, it translates between them on the fly. It also offers the type of features one would expect of a large proprietary PBX system such as Voicemail, Conference Bridging, Call Queuing, Music on Hold, Interactive Voice Response (IVR) and Call Detail Records to name a few. For a complete feature list, please refer to the Detailed Features section later on.

The key benefits of such system are the flexibility that it enjoys (based on the fact that it runs on Linux and that it is in software) and the tremendous savings in cost compared to legacy PBX solutions. It actually runs on Red Hat Enterprise Linux 4. That means it comes on top of a rock-solid, secure and reliable Operating System. Being implemented in software also means that it can be easily customized and enhanced in functionality.

At LinuxWorld 2004 Expo, a leading open-source advocate, Jon Hall president of Linux International, has predicted that VoIP services based on open-source software will generate more business than the Linux operating system itself. Today, we already started witnessing a boom in the VoIP services business driven mainly by the lower Total Cost of Ownership (TCO) and ongoing savings in the communications expenses.

The system also comes handy with a very intuitive web interface to provision and monitor the system. The web interface is divided into 3 main parts:

- Voicemail web interface: where users can access their voicemails (and settings) and call recordings
- Operator Panel: where the user gets a live view of the system (extensions, queues, trunks) and can even connect calls with the click of a mouse
- Management Portal: which is a very easy-to-use interface to setup the system, provision trucks, extensions and routes and create CDR reports

Among other supported VoIP protocols, our system can talk Cisco Skinny (SCCP). That means it acts like Cisco CallManager for your existing/new Cisco IP phones only without the added software cost and the extra required phone licenses. This allows you to get all the benefits and features of the Cisco IP phones included in the PBX system for no extra charge.

We're eager to talk to you and discuss your needs. Please contact us if this system interests you.

Thank you.

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Detailed Features

Our IP soft-PBX solution offers a rich and flexible feature set. It offers both classical PBX functionality and advanced features, and interoperates with traditional standards-based telephony systems and Voice over IP systems.

Call Features:

- ADSI On-Screen Menu System
- Alarm Receiver
- Append Message
- Authentication
- Automated Attendant
- Blacklists
- Blind Transfer
- Call Detail Records
- Call Forward on Busy
- Call Forward on No Answer
- Call Forward Variable
- Call Monitoring
- Call Parking
- Call Queuing
- Call Recording
- Call Retrieval
- Call Routing (DID & ANI)
- Call Snooping
- Call Transfer
- Call Waiting
- Caller ID
- Caller ID Blocking
- Caller ID on Call Waiting
- Calling Cards
- Conference Bridging
- Database Store / Retrieve
- Database Integration
- Dial by Name
- Direct Inward System Access
- Distinctive Ring
- Distributed Universal Number Discovery (DUNDi)
- Do Not Disturb
- E911

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- ENUM
- Fax Transmit and Receive
- Flexible Extension Logic
- Interactive Directory Listing
- Interactive Voice Response (IVR)
- Local and Remote Call Agents
- Macros
- Music On Hold
- Music On Transfer
  - Flexible Mp3-based System
  - Random or Linear Play
  - Volume Control
- Predictive Dialer
- Privacy
- Open Settlement Protocol (OSP)
- Overhead Paging
- Protocol Conversion
- Remote Call Pickup
- Remote Office Support
- Roaming Extensions
- Route by Caller ID
- SMS Messaging
- Spell / Say
- Streaming Media Access
- Supervised Transfer
- Talk Detection
- Text-to-Speech (via a 3rd party package)
- Three-way Calling
- Time and Date
- Transcoding
- Trunking
- VoIP Gateways
- Voicemail
  - Visual Indicator for Message Waiting
  - Stutter Dialtone for Message Waiting
  - Voicemail to email
  - Voicemail Groups
  - Web Voicemail Interface

Scalability:

- TDMoE (Time Division Multiplex over Ethernet)
  - Allows direct connection of multiple PBX's
  - Zero latency
  - Uses commodity Ethernet hardware
- Voice-over IP
  - Allows for integration of physically separate installations
  - Uses commonly deployed data connections
  - Allows a unified dialplan across multiple offices

Codecs:

- ADPCM
- G.711 (A-Law & µ-Law)
- G.723.1 (pass through)
- G.726
- G.729 (though purchase of commercial license is required)
- GSM
- iLBC
- Linear
- LPC-10
- Speex

Protocols:

- IAX
- H.323
- SIP (Session Initiation Protocol)
- MGCP (Media Gateway Control Protocol)
- SCCP (Cisco Skinny)

Traditional Telephony Interoperability

- E&M
- E&M Wink
- Feature Group D
- FXS
- FXO
- GR-303
- Loopstart
- Groundstart
- Kewlstart
- MF and DTMF support
- Robbed-bit Signaling (RBS) Types

PRI Protocols

- 4ESS
- BRI (ISDN4Linux)
- DMS100
- EuroISDN
- Lucent 5E
- National ISDN2
- NFAS

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