SE30P

IP PBX

Key Features

- PBX functionality with voice mail
- Based on open standards: SIP, Linux, Radius
- Supports up to 30 users in a single box
- 4 FXO ports
- 2 phone bypass ports
- Auto Attendant
- Operator
- Ring group
- Hunt group
- Call detail reports
- Remote users over the internet without the need for VPN or tunneling.
- External users on the PSTN
- T.38 fax transmission and reception
- Multiple languages and worldwide support
- User Web Portal for remote access to voice messages
- Personal call handling rules
- Direct Inward Dial (DID)
- Up to 8 concurrent calls to the IP trunks and 30 concurrent calls with direct media

Overview

The SE30P is a powerful system that simplifies voice communications for all workers in a small office, multiple remote locations, home workers, and workers who are on the road. The SE30P is easy to install, use, and maintain. The embedded system design on the SE30P makes it the most stable system in the industry.

The SE30P is designed specifically to connect to Internet telephony service providers (ITSPs) using SIP. Customers will save money on making phone calls because of the low rates offered by ITSPs. Customers will also save money on infrastructure because the SE30P can be easily integrated into their existing networks.

The system combines the functions of an IP PBX, Internet gateway, network server, and application server. By using standard protocols, it is interoperable with phones, gateways, and devices from other manufacturers.

IP Telephony

The SE30P can interoperate with any SIP compliant phone. Zed-3 has its own range of IP phones that is fully compatible with the SE30P. These include desktop phones as well as a soft phone.

In addition to the soft phone that runs on a PC, Zed-3 provides a soft phone client that runs on any windows based PDA.

The SE30P can connect to up to eight ITSPs (Internet telephony service providers). This permits the use of the Internet for all voice calls without the need to connect to the PSTN. Alternatively, the SE30P can connect to the PSTN through an optional gateway, providing you with the flexibility to route some calls to the PSTN and other calls to the ITSP.

PSTN Interfaces

The SE30P has four FXO ports to connect to analog PSTN lines. They permit the SE30P to obtain telephony service from the traditional network as well as from an ITSP. As such, the SE30P is a perfect solution if you want to take full advantage of VoIP and at the same time have PSTN lines as your main connection or as a backup.

There are two ports to connect to phones. In the event of power failure, these ports are connected to the FXO circuits so that the analog phones can then make and receive calls.

Administration Tool

The SE30P has a HTML based administration UI so that it is not necessary to install any management software on a PC. The administration UI is intuitive and easy to navigate. Customers may access and manage the SE30P from anywhere as long as they have a web browser and access to the Internet.

The administration UI provides various real time information and statistics of the SE30P. These include operation state, system and network information, events and alarms, active phones, active calls, IP Trunks status, and more.

Management of Users

Adding, modifying and deleting a user can be done in a matter of few clicks. Customers can assign different rights and privileges to different users.

Browser based User Portal

The SE30P has an HTML based user portal. Users can log on to the system using any web browser to download, save, and delete their voice messages, change their mail box password, and configure their personal call handling rules.

Dial Plan and Least Cost Routing

The system has a flexible dial plan that allows you to specify the routing of calls based on dialled patterns. Your organization can ensure least cost routing for all calls, regardless of the user or location.

For each dial pattern on the dial plan, you specify the transformed pattern, the primary route, and alternative routes. You can create multiple dial patterns to handle internal calls and external calls. These calls can be routed over the LAN, the WAN, the Internet at an external gateway, or a line interface to the PSTN.

Call Restrictions

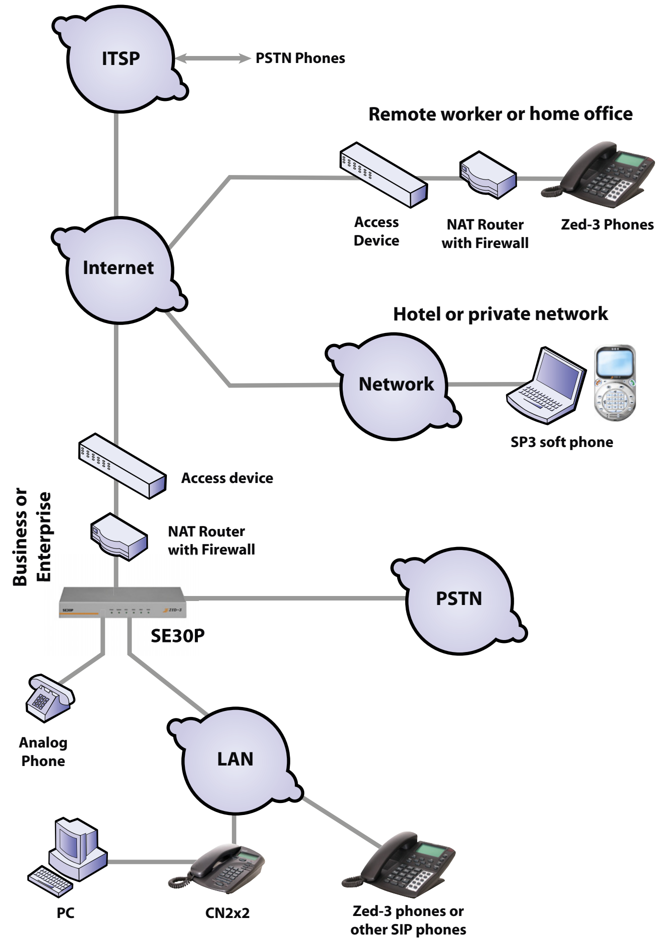
You can restrict calls to specific destinations per user. This allows you to specify who has the rights to make long distance and international calls.

Music on Hold (MoH)

The SE30P can play MoH to callers. This is played from a file on the internal flash memory disc.

Communications with phones off the system

Remote extensions can be achieved easily by setting up call handling rules that route all the calls to a user's home or mobile phone numbers.



Accessibility of Users

The SE30P allows users to be reached even when they are home or travelling. Users can have multiple contact points where they can receive calls. A contact point can be an IP desktop phone or soft phone. An incoming call will alert all contact points simultaneously. Users can also create custom call handling rules that can forward their calls to external numbers based on different conditions.

Remote Users

The SE30P makes it uncomplicated to connect remote phones. Most VoIP solutions today require remote users to use a VPN or other tunneling technologies. With the SE30P, remote users can connect with the system, even when they are behind a NAT, router, or firewall, without the need to use a VPN.

Attendant and Operator

Incoming calls can be routed to an automated attendant (AA), a live attendant, or an operator. The AA or operator can be assigned a direct external number (DID) and an internal extension number. The SE30P allows you to specify working hours and greeting messages that are played during working hours and nonworking hours.

The AA provides the features of dial by name, scheduled greeting messages, and transferring incoming call to different extensions. It is extremely easy to setup and modify.

Hunt Groups

The SE30P can provide multiple hunt groups for an informal call center. You can configure hunt groups for different services within the business. Each group can be assigned a direct inward dial number (DID) in addition to an internal extension. You can assign a user to be an agent for one or more groups, and agents within a group can still make and receive individual calls.

Voice Mail (VM)

Users can access, save, and delete VM by using a phone (internal or external to the SE30P) or by using the Personal Portal. Users can save their voice messages without taking up storage on the SE30P by saving the messages into folders on the local PC. Voice messages are saved as files and can be forwarded to others outside the system through standard means of file sharing such as e-mail and network directories.

The SE30P has a compact flash port that allows customers to install a compact fl ash drive for voice mail storage. A 1 GB compact flash drive will store up to about 1000 minutes of voice messages for the whole system.

Fax Origination and Termination

The SE30P supports fax to devices connected to the analog FXS ports. This can be with fax pass through or with T.38 fax which is supported by most ITSPs.

Data Networking

The SE30P has two Ethernet ports – one connects to the service provider and one connects inside the office.

The WAN port connects directly behind the broadband IAD (integrated access device) and provides termination of traffic and address translation. The translation comprises NAT (network address translation) for Internet traffic and ALG (application layer gateway) for SIP traffic. The SE30P accomplishes much more than simple ALG by allowing flexible selection of how direct media is handled to ensure simple integration in a variety of network configurations.

The WAN port can have a fixed IP address, an IP address received from the ISP using DHCP, or use PPPoE to obtain all information. Default and static routes can be provisioned to control the proper routing of voice and data traffic.

The LAN port connects directly to devices or an Ethernet switch. Over this port, the SE30P can provide DHCP service.

VLAN and QoS Support

Most IP networks existing today support VLAN and QoS. Having VLAN and QoS can increase the voice quality of phone calls. The SE30P fully supports IEEE 802.1q VLAN tagging and IEEE 802.1p QoS. At the IP layer, the SE30P support the ability to mark the lower six bits of the IP QoS byte with the various differentiated service code point (DSCP) markings.

Billing and Call Detail Records (CDR)

The SE30P provides comprehensive CDR for reconciliation of billing and tracking of system usage.

The SE30P supports RADIUS client, which can be used to interact with a RADIUS server for authentication. Using this feature, you can easily integrate the SE30P with any billing system that support RADIUS.

Global Features

Zed-3 sells and supports its products worldwide, allowing the SE30P to be readily deployed in one or more countries. The system supports telephony protocols for many countries so it can connect directly to the local PSTN. You can navigate the SE30P's administration UI and user portal in three different languages: English, Simplified Chinese, and Korean.

Diagnostics and SNMP

The SE30P provides Ping, Traceroute, and DNS Lookup for network diagnostic. These tools will help you identify most networking problems. To identify call setup problems, the SE30P also provides SIP Route Trace diagnostic tool.

The SE30P supports Trap in SNMP. This allows you to be notified immediately if the SE30P is experiencing any problems.

Specifications

PBX Features

- Auto Attendant
- Live Attendant and Operator
- Caller ID
- Call Transfer
- Call Forward
- Call Waiting
- Call Restriction
- Call Hold
- Call Park
- Call Pick up
- Do Not Disturb
- DID
- ACD (Ring Groups and Hunt Groups)
- MWI Support
- CDR

Capacity

- Up to 30 users configurable
- Up to 30 Internal Calls
- Up to 8 External Calls
- Up to 20 hours of voice mail

Voice Quality Protection

- Dynamic Jitter Buffering
- VAD
- CNG
- VLAN tagging
- DiffServ
- Packet Loss Compensation
- Echo Cancellation
- Voice Packet Prioritization

Codec

- Audio: A-law and μ-law G.711, G.729A, G.723.1
- Fax: T.38

DTMF

- In-band
- RFC2833
- SIP Info

System Features

- Management Interface: Web
- English and Chinese Languages
- Online Upgrade
- Configuration Backup and Restore
- NTP
- Pre-set Progress Tones for Multiple Regions
- Least Cost Routing
- RADIUS client
- Diagnostic Tools: Ping, Tracer, DNS Lookup, SIP
- Route Trace
- Comprehensive Alarm and Event Records
- Access Control List (ACL)
- Personal Call Handling Rules

Hardware Specifications

- Five LED Indicators
- Reset Button
- Four FXO Interfaces
- Two Phone Bypass Ports
- CF memory socket

Network Interfaces

- One 10/100 Mb/s WAN port
- One 10/100 Mb/s LAN port
- DHCP server

Power

- External ac to dc power adapter
- Input: 100~240 Vac @ 47~63 Hz
- Power: 20 W

Physical and Environmental

Operating temperature: 10°C to 40°C (50°F to 104°F)
Storage temperature: 0°C to 50°C (32°F to 122°F)
Weight: 1.5 kg (3.3 lb). Shipping weight 2 kg (4.5 lb)
Size: 260 mm (W) x 160 mm (D) x 35 mm (H) (10.2" x 6.2" x 1.4")
Safety: FCC Part 68
EMI: FCC Part 15A
RoHS: Compliant

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