

SE 500

IP PBX



Key Features

- PBX functionality with voice mail
- Based on open standards: SIP, Linux, Radius
- Supports up to 500 users in a single box
- 4 Modular slots
- Up to 32 FXO ports
- Up to 32 FXS ports
- Up to 3 T1/E1 ports
- 90 concurrent calls to PSTN or ITSP
- Conference bridge (up to 16 participants)
- ACD (ring groups and hunt groups)
- Auto Attendant and operator
- Direct Inward Dial (DID)
- Inbound call center (ICC) with queuing and call recording
- Call detail reports
- Remote users over the internet without the need for VPN or tunneling
- External users and connectivity to other systems
- T.38 fax transmission and reception
- SEME desktop client that provides access to presence, instant messaging, voice mail, and call handling rules
- Dual hard discs in RAID configuration
- LCD on front panel displays system status
- Multiple languages and worldwide support

Overview

The SE500 is a powerful system that simplifies voice communications for all workers in a small to medium size office, multiple remote locations, home workers, and workers who are on the road. The SE500 is easy to install, use, and maintain. As a product engineered for businesses and enterprises, it is the most stable product in its class in the industry.

The SE500 comes with four modular slots; each can be used with an 8 port FXS card, an 8 port FXO card, or a T1/E1 card. This ensures the SE500 can be fully connected to the PSTN.

The SE500 is also designed to connect to Internet telephony service providers (ITSPs) using SIP. Businesses will save money on making phone calls because of the low rates offered by ITSPs. You will also save money on infrastructure because the SE500 can be easily integrated into their existing networks.

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

IP Telephony

The SE500 can interoperate with SIP compliant phones. Zed-3 has its own range of IP phones that is fully compatible with the SE500. 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 Windows based PDAs. Other phones or PDAs that have Wi-Fi, running the Symbian operating system, also function with the SE500.

The SE500 can connect to up to 64 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 SE500 can connect to the PSTN through its modular cards: FXO and T1/E1.

Global Features

Zed-3 sells and supports its products worldwide, allowing the SE500 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 SE500's administration UI in two different languages: English and Simplified Chinese.

Analog and Digital Interfaces

The SE500 has four modular slots that allow you to install the modular cards you need. You can install an 8 port FXS card for analog phones or fax machines, 8 port FXO card for traditional phone lines, a card with 4 FXS ports and 4 FXO ports, or a T1/E1 card for digital lines. The FXS interfaces can be provisioned as extensions in normal use or assigned a DID number. The card that has both FXS ports and FXO ports passes through the circuits in the event of a power failure. This is useful for applications where it is necessary to have communications when power is lost.

The SE500 supports a total of 90 channels simultaneously to the PSTN. So, you can add four analog cards or three T1/E1 cards to the system.

Administration Tool

The SE500 has an 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. You can access and manage the SE500 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 SE500. These include operation state, system and network information, events and alarms, active phones, active calls, IP Trunks status, and more.

Management of Users

Moving, adding, and changing a user can be done in a matter of few clicks. You can assign different rights and privileges to different users. These rights include Operator, Voicemail, Call Restrictions, and Trunks.

You can create different user groups on the SE500 which can be used for different departments or offices. Each user group has its own rights, privileges, calling restrictions, and prefix number to access the members of the group. These settings are inherited by the users created under this group.

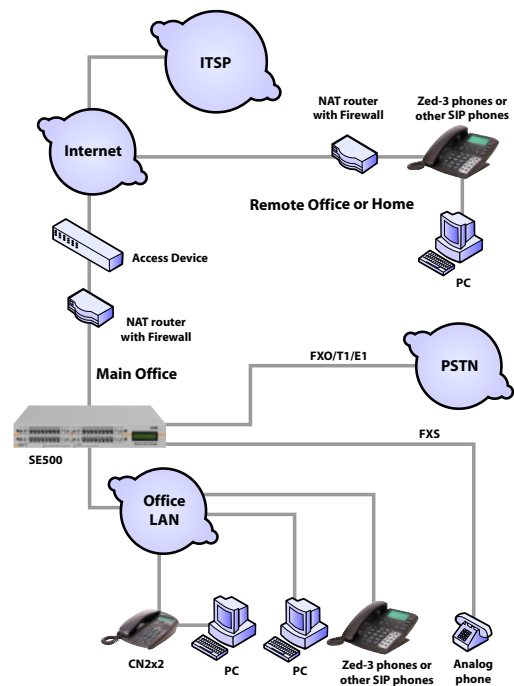
User Portal and SEME

The User Portal provides HTML access to the system. 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.

SEME (pronounced *see me*) is a desktop client software that runs under Windows 2000 and XP. It provides access to the SE500's services and resources to permit users to send and receive instant messages (IMs) to other users, inform other users about their presence, use address books, and manage voice mail messages and settings.

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 SE500 can play MoH to callers. This is played from a file on the internal hard disc or streamed from an external source (such as a web site).

Accessibility of Users

The SE500 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 SE500 makes it uncomplicated to connect remote phones. Most VoIP solutions today require remote users to use a VPN or other tunneling technologies. With the SE500, remote users can connect with the system, even when they are behind a NAT, router, or firewall, without the need to use a VPN.

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.

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 SE500 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.

Operators can use software that runs under windows to manage incoming calls. This console software is more effective than the legacy busy lamp field (BLF) array of buttons and lights.

ACD (Ring Groups and Hunt Groups)

The SE500 provides multiple ACD (Automatic Call Distribution) groups for basic call center needs. Each group can be assigned a direct inward dial number (DID) in addition to an internal extension. Call distribution strategies include ring all and round robin are available. 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.

Conference Bridge

The SE500 provides a conference bridge supports up to ten participants in a conference session. Each participant can join or leave the conference at any time. The conference can be put on hold, allowing the other parties to continue without the host.

Inbound Call Center (ICC)

The ICC supports up to 90 agents or supervisors in one or more groups. Agents may belong to multiple groups and there may be multiple supervisors in each group. The ICC option provides queuing with different messages played to the caller at different times in the queue. Callers may exit the queue by leaving a voice mail.

Calls can be recorded (using the external Zed-3 CR90 call recorder) and later retrieved. The ICC captures statistics on the performance of the group such as wait time and response time by agents.

Voice Mail (VM)

Users can access, save, and delete VM by using a phone (internal or external to the SE500) or by using the Personal Portal or SEME. Users can save their voice messages without taking up storage on the SE500 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.

Users can configure the system to send an e-mail to them to alert them that a new voice mail is received. The voice mail can be optionally attached as a WAV file.

Voice mail is stored on the hard disc. This provides about 1,000 hours of storage.

Fax Origination and Termination

The SE500 supports fax to devices connected to the analog FXS ports. The system can also connect to an ITSP with T.38 which is supported by most ITSPs.

Data Networking

The SE500 has a single Ethernet port to connect to your network. This would typically be placed on the DMZ of your firewall.

The port 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 SE500 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 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 SE500 supports DDNS to permit external users to access the system even when the IP address is not static.

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 SE500 fully supports IEEE 802.1q VLAN tagging and IEEE 802.1p QoS. At the IP layer, the SE500 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 SE500 provides comprehensive CDR for reconciliation of billing and tracking of system usage. The SE500 supports RADIUS client, which can be used to interact with a RADIUS server for authentication. Using this feature, you can easily integrate the SE500 with any billing system that support RADIUS.

Diagnostics and SNMP

The system has an LCD on the front panel. This shows the status of the SE500 and its IP address. This enables rapid deployment and troubleshooting.

The SE500 provides Ping, Traceroute, and DNS Lookup for network diagnostic. These tools will help you identify most networking problems. To identify call setup problems, the SE500 also provides SIP Route Trace diagnostic tool. The SE500 supports Trap in SNMP. This allows you to be notified immediately if the SE500 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 and ICC
- MWI Support
- CDR
- Conference bridge (up to 16 participants)

Capacity

- One auto attendant
- Up to 32 FXS ports
- Up to 32 FXO ports
- Up to 3 T1/E1 ports
- Up to 500 users configurable
- Up to 500 internal calls
- Up to 90 external calls
- Up to 1000 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: HTML browser, telnet
- English and Chinese
- Online Upgrade
- Configuration Backup and Restore
- NTP, HTTP
- 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

- Reset Button
- Four Modular slots
- LCD on front panel
- Fan cooled

Network Interface

- One 10/100 Mb/s Ethernet port

Power

- Input: 100~240 Vac @ 47~63 Hz
- Power: 125 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: 7 kg (15 lb). Shipping weight 8 kg (18 lb)

Size: 440 mm (W) x 450 mm (D) x 70 mm (H) (17" x 18" x 3")

Mount: Standard 19" rack from front, or rear; 2 RU (89 mm)

Safety: FCC Part 68

EMI: FCC Part 15A

RoHS: Compliant



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