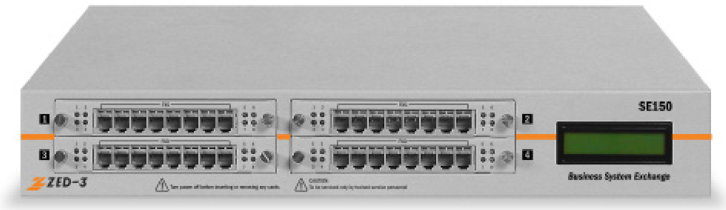
ZED-3

SE150

IP PBX



Key Features

- PBX functionality with voice mail
- Gateway functionality for SIP to PSTN
- Based on open standards: SIP, Linux, Radius
- Supports up to 150 users in a single box
- 4 Modular slots
- Up to 32 FXO ports
- Up to 32 FXS ports
- Up to 2 T1/E1 ports
- Conference bridge (up to five participants)
- ACD (ring groups and hunt groups)
- Auto Attendant
- Operator
- Call detail reports
- Remote users over the internet without the need for VPN or tunneling
- External users
- T.38 fax transmission and reception
- User Web Portal for remote access to voice messages
- Personal call handling rules
- Direct Inward Dial (DID)
- 30 concurrent calls to PSTN or ITSP
- Multiple languages and worldwide support

Overview

The SE150 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 SE150 is easy to install, use, and maintain. The embedded system design on the SE150 makes it the most stable system in the industry.

The SE150 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 SE150 can be fully connected to the PSTN.

The SE150 is also designed 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 SE150 can be easily integrated into their existing networks.

The system combines the functions of an IP PBX, Internet gateway, 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 SE150 can interoperate with any SIP compliant phone. Zed-3 has its own range of IP phones that is fully compatible with the SE150. 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 SE150 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 SE150 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 SE150 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 SE150's administration UI in two different languages: English and Simplified Chinese.

Analog and Digital Interfaces

The SE150 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, 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 SE150 supports a total of 30 channels simultaneously to the PSTN. So, you can add four analog cards or a T1/E1 card to the system.

VoIP Gateway

The SE150 can be used as a gateway to translate between SIP on the IP side and PSTN interfaces on the other side. The SE150 is designed for wire line, cable, and broadband access and can be used by enterprises and service providers.

Enterprises can benefit from a variety of valuable applications such as the connectivity of remote offices and consolidation of long distance traffic. A service provider can deploy numerous applications such as SIP trunking, wholesale VoIP termination, calling cards, tandem switching, and least cost routing. In these applications, the SE150 provides you with a cost effective deployment of next generation networks. Based on Zed-3's reliable SIP technology the SE150 can aid integration in any standard network environment to provide connectivity for SIP to SIP, TDM to TDM, and SIP to TDM.

Administration Tool

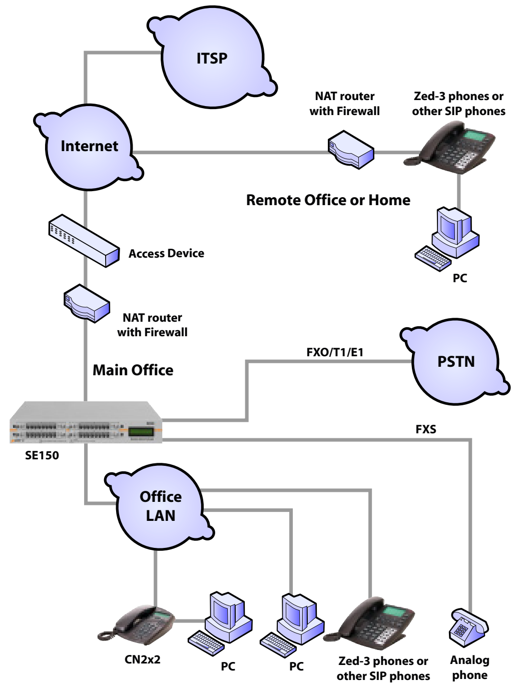
The SE150 has an HTML based administration UI so that it is not necessary to install any management software on a PC. The administration UI is intuitive and easy to navigate. Customers may access and manage the SE150 from anywhere as long as they have a web browser and access to the Internet. The administration UI provides various real time information and statistics of the SE150. 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. These rights include Operator, Voicemail, Call Restrictions, and Trunks.

Browser based User Portal

The SE150 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 SE150 can play MoH to callers. This is played from a file on the internal flash memory disc.

Accessibility of Users

The SE150 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 SE150 makes it uncomplicated to connect remote phones. Most VoIP solutions today require remote users to use a VPN or other tunneling technologies. With the SE150, 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 SE150 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.

ACD (Ring Groups and Hunt Groups)

The SE150 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 SE150 provides a conference bridge supports up to five 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.

Voice Mail (VM)

Users can access, save, and delete VM by using a phone (internal or external to the SE150) or by using the Personal Portal. Users can save their voice messages without taking up storage on the SE150 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 SE150 has a compact flash port that allows customers to install a compact flash drive for voice mail storage. A 2 GB compact flash drive will store up to about 2000 minutes of voice messages for the whole system.

Fax Origination and Termination

The SE150 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 SE150 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 SE150 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 SE150 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 SE150 fully supports IEEE 802.1q VLAN tagging and IEEE 802.1p QoS. At the IP layer, the SE150 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 SE150 provides comprehensive CDR for reconciliation of billing and tracking of system usage. The SE150 supports RADIUS client, which can be used to interact with a RADIUS server for authentication. Using this feature, you can easily integrate the SE150 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 SE150 and its IP address. This enables rapid deployment and troubleshooting.

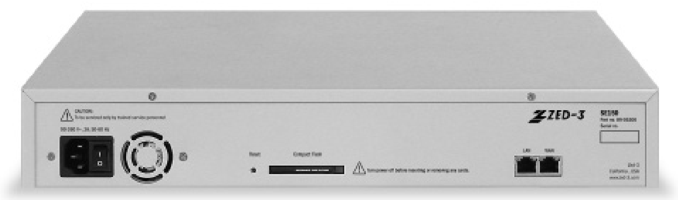
The SE150 provides Ping, Traceroute, and DNS Lookup for network diagnostic. These tools will help you identify most networking problems. To identify call setup problems, the SE150 also provides SIP Route Trace diagnostic tool. The SE150 supports SNMP. This powerful feature allows you to make various traps to the SE150 using the SNMP.

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
- MWI Support
- CDR
- Conference bridge (up to five participants)

Capacity
- One auto attendant
- Up to 32 FXS ports
- Up to 32 FXO ports
- Up to 2 T1/E1 ports
- Up to 150 users configurable
- Up to 150 internal calls
- Up to 30 external calls
- Up to 8 GB compact flash

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
- 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 Interfaces
- One 10/100 WAN port
- One 10/100 LAN port

Power
- Input: 100~240 Vac @ 47~63 Hz
- Power: 70 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: 5 kg (11 lb). Shipping weight 7 kg (15 lb)
Size: 440 mm (W) x 300 mm (D) x 70 mm (H) (17" x 12" 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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