P R O D U C T   O V E R V I E W

IP Server 900

High-performance IP communications system

Contents

Available information

- **Color brochures**:
  Brochure — ESI # 0450-1318.
  Mini-brochure — ESI # 0450-1319.
  Available for purchase from ESI.
  Downloadable from
  www.esi-estech.com.
- ***User's Guide***: ESI # 0450-1306.
  Downloadable from
  www.esi-estech.com.

Introduction

ESI's **IP Server 900** is a high-performance IP phone system that unites the communication and productivity needs of end users in small-to-medium organizations with one or more sites. The IP Server 900's expandable design integrates built-in capabilities, advanced features, and highly differentiated applications into a product that's as easy-to-use and -manage as it is flexible.

The IP Server 900 fits easily into existing networks and eliminates obsolescence by supporting not only the current lineup of ESI phones and other devices but also an impressive list of legacy ESI equipment.

Customers with growing communications needs will be pleased by how the IP Server 900's capacity expands smoothly and easily to a system maximum of 1,104 communications ports, including up to 860 telephones and up to 508 Esi-Link channels.[1] Expanding the capacity of an IP Server 900 Control Unit is done chiefly through the easy, snap-in installation of compact, low-profile **Resource Modules** — the use of which also enables each IP Server 900 **unit** (Master Control Unit, IP Expansion Unit, or D/A Expansion Unit) to fit into a "1U" opening on a standard equipment rack.

Management of many features and system resources is accomplished through **licensing**. This gives the Reseller much more flexibility in tailoring each IP Server 900 to the specific needs of end users — particularly those who wish to buy only those capabilities and call-handling capacities they truly need now, while retaining the option for easy expansion later.

The base IP Server 900 includes:

- Eight universal IP (UIP) licenses.
- Two ports of voice mail, with a storage capacity of 15 hours.
- Eight (extension) voice mailboxes.
- 1,000 guest/information voice mailboxes.

All ESI phone models, both IP-based and digital, provide advanced ESI features. The supported models are:

- **ESI 60IP Business Phone** and **ESI 60D** (digital) **Business Phone** — The ESI 60IP is available in both Gigabit Ethernet and 10/100 models. Each ESI 60 Business Phone model includes 48 programmable feature keys, visual voice mail, an adjustable backlit display, and a full-duplex, high-definition speakerphone.
- **ESI 40IP Business Phone** and **ESI 40D** (digital) **Business Phone** — Each ESI 40 Business Phone model includes 16 programmable feature keys and an adjustable backlit display.
- **48-Key IP Feature Phone II** and **48-Key Digital Feature Phone** — Each 48-Key Feature Phone model includes 30 programmable feature keys and a large display.
- **ESI 30D** (digital) **Business Phone** — A smaller model intended for lower-traffic users.
- **ESI Cordless Handset II** — Available in digital, local IP, and remote IP models.
- ***VIP 7 Softphone*** — Combines the functionality of a desktop IP phone and the *VIP 7* product in one PC-based phone.

All ESI IP phone models are standards-compliant and operate with the customer's data network to provide highest-quality voice through Quality of Service (QoS) support. In addition, all ESI desktop IP phone models support Power over Ethernet (PoE).

An IP Server 900 provides an ideal, cost-effective upgrade path for several models of ESI systems. See "Migration capability," page 18, for more details.

[1] See "Capacity constraints," page 14.

Advantage summary

Capacities

System

	Master Control Unit	Expansion Units
System ports	264	840
Trunk ports	48	144
T1/PRI Modules (each supporting 24 T1 channels or 23 PRI channels)	2	6
Applications Services	(See next two items)	(See next two items)
Audio recording channels	32	12
Video recording channels	12	12
Station ports[1]	192	672
IP stations	127	381
Digital stations	64	288
Analog stations	32	144
G.729a Resource Module (ESI # 5000-0606) for G.729a codec (each with 48 ports for SIP trunks, remote SIP phones, and Esi-Link)[2]	1	3
Conference ports[3]	64	

Voice mail

	Maximum capacity
On-board integrated auto-attendant/voice mail channels	32
Total voice mailboxes	1,864
User	864
Guest/information	1,000
Group/maximum members	32/64
Cascade notification	20
Q & A mailboxes	20
Broadcast (one to all extensions)	[Yes]
Voice storage (hrs.)	140

[1] See “Capacity constraints,” page 14.
[2] Esi-Link channels are allocated to “reserved” ports; *i.e.*, Esi-Link channels don’t reduce CO or station capacity.
[3] Dynamic assignment allows for unlimited combinations up to the maximum of 16 parties per conference — *e.g.,* 21 three-member conferences, or four four-member conferences in combination with two eight-member conferences. Achieves best audio performance when using digital trunks.

Standard features

- 64 license-controlled universal IP ports for IP stations and devices, SIP stations, SIP trunks, and Esi-Link
- Account codes
- Automatic call distribution (ACD)
- Built-in Network Services Processor (NSP)
- Caller ID key
- Distinctive ring for trunks
- Enhanced Caller ID
- Esi-Dex integrated directories
- Fax tone detection and T.38 fax relay
- Find-me/follow-me
- Flexible numbering plans
- Power over Ethernet support for desktop IP phones
- Recording of calls
- Station redial and callback
- System programming using browser-based *Web ESI System Programmer (WESP)*

Optional features and applications

- Digital, local IP, and remote IP Cordless Handsets
- Dual-configuration desktop IP phones (support local and remote installations)
- Esi-Link IP private networking
- *ESI API*
- ESI Bluetooth Headset Interface
- ESI Presence Management
- ESI Media Management
- ESI Mobile Messaging
- *ESI Salesforce.com Connector*
- “Meet-me” conferencing
- Shared-office tenanting (maximum of eight tenants)
- SIP stations (third-party)
- SIP trunking
- *VIP 7* (*Visually Integrated Phone*)
- *VIP 7 ACD Supervisor* and *VIP 7 ACD Agent*
- *VIP 7 PC Attendant Console*
- *VIP 7 Softphone*

Hardware description

System configuration

The IP Server 900 is a compact, rack-mounted system. The maximum configuration consists of one **Master Control Unit** and three additional units, each of which may be either an **IP Expansion Unit** or a **D/A Expansion Unit**. If desired, the units may be wall-mounted, but rack-mounting is the preferred method of installation.

Processing

Processing power is provided by a fourth-generation Motorola® ColdFire® commercial-grade microprocessor running at 266 MHz, designed specifically for 24/7 operation. This device houses 1 GB of external DDR2 memory for stored program control. The system also features an ARM9 core processor running at 400 MHz and with 1 GB external DDR2 memory; and three Spartan-6 FPGAs. On-board, third-generation DSPs manage inter-module communications and telephony services, ensuring rapid, dependable communications among all system resources: trunks, IP stations, and digital stations.

Power provisions

Each IP Server 900 unit is powered by its own power supply. In rack-mounted installations, a power shelf is available that provides AC connection for each of the unit power supplies. This reduces the number of AC power outlets needed to one, instead of one per power supply. When a UPS system is installed, only one connection to the UPS must be made from the system, rather than one from each of the units. The power shelf is separately fused to protect system components against erratic power fluctuations.

Each unit has a grounding lug and solder terminal for the connection of a ground wire. It is highly recommended that all units be grounded to a common grounding point by “pig-tailing” the ground wire from one unit to the one below it.

Connection between units

Connection between IP Server 900 units is made through an RJ-45 cable, which is shipped with each IP Expansion Unit or D/A Expansion Unit. This RJ-45 cable (maximum length of three meters) extends the motherboard from unit to unit, creating a common backplane.

Motherboard

The IP Server 900 motherboard houses a built-in Network Services Processor (NSP) for all applications requiring direct connection of the IP Server 900 to the customer’s data network. These applications include SMDR, system programming via a Web browser, and ESI options.

Fully flexible platform

The IP Server 900 offers impressive expansion capabilities, using a variety of unique **Resource Modules** that are easily snapped into place for greater ease of maintenance.

For system capacities, refer to “Advantage summary,” page 3.

Features at a glance

The IP Server 900 combines the power of a PBX with the ease of use for which every ESI system is renowned. Its feature set, capacities and scalability ensure:

- Availability of all features, functionalities and tools ESI offers to increase the productivity of an enterprise.
- Expansion ability to meet shifting demands of business growth.

Universal IP ports

Eight **universal IP ports** are prelicensed with the IP Server 900 Master Control Unit, and licenses for more may be purchased. These ports may be used for:

- ESI IP stations and other devices.
- SIP stations.
- SIP trunking.
- Esi-Link channels.

Integrated voice mail

The IP Server 900 boasts a full complement of practical, easy-to-use voice mail features, many standard while others are added by licensing as the customer requires:

- In addition to its call processing ports, the IP Server 900 is configured with **built-in voice mail channels**. There is no need to balance voice mail needs at the expense of a customer's call-handling requirements.
- Voice mail and other message storage are recorded at the **highest grade of voice quality** (64-Kbit/second sampling).
- **Substantial voice message storage** ensures ample capacity for all mailbox users, including the needs of users enabled with the optional auto-record feature.
- Support for 12 **message-on-hold recordings** — Three pre-recorded tracks and nine customizable recordings.
- **New-message notification** can be delivered off-premises to a phone or pager.
- Leaving messages for up to 65 mailboxes at once is easy, using ESI's unique **Quick Groups**™ feature.
- ESI's **Quick Move**™ function enables conversations to be recorded directly into another user's mailbox. At the mailbox user's option, urgent messages can be treated with priority and delivered first, instead of on a "first in-first out" basis.
- Several **different mailbox types** — including group, broadcast, information, cascade paging, Q & A, and guest mailboxes — support a wide range of customer applications.
- Callers forwarded to user or guest voice mailboxes can reach the called individual at a designated **off-premises "reach me"** number.
- Each user mailbox is equipped with a **Message Recycle Bin** that remembers, and can restore, the 10 most recently deleted messages.
- One or more stations may have a programmable **Virtual Mailbox Key**™ on their stations to allow easy monitoring of a second mailbox.

Auto attendant

The IP Server 900 provides rich, comprehensive auto attendant features:

- **100 branches (six levels deep)** to permit the design of a more natural, caller-friendly answering environment, including a company directory.
- **Virtually unlimited call routing**, including off-premises transfer.
- **Three-character dial-by-name** for callers to search through the auto attendant directory and all Esi-Dex directories to find the desired name.

Automatic call distribution

Manage call overload and increase customer satisfaction with **automatic call distribution (ACD)** that is useful to a business of any size.[1] ACD ensures that:

- Calls are prioritized and routed within designated departments for **quickest possible call handling**.
- Managers and agents receive **up-to-the-second information** on queues and wait times via the displays on most ESI desktop phones supported by the IP Server 900.
- Supervisors have access to agents' ACD call activity to **more effectively evaluate** call traffic and agent performance.
- A **separate hold recall timer** is provided for ACD agents, further ensuring that customer care is enhanced.
- Agents may log into two separate ACD departments simultaneously, with departmental prioritization.

Flexible conference channels

The IP Server 900 reserves channels for conferencing. These can be dynamically[2] connected in multi-party conversations of up to 16 channels per conference. Any combination of conference channels may be joined together, as long as the originating party is an IP Server 900 user. All channels reserved for conferencing are dynamically balanced for optimum audio performance.

Find-me/follow-me

The **find-me/follow-me** feature simplifies communications by allowing a user to give his/her business contacts just one phone number, to which the IP Server 900 can direct all the user's incoming calls. For more details on this feature, see the *Find-me/Follow-me Feature Overview* (ESI # 0450-1360).

"Meet-me" conferencing

"Meet-me" conferencing makes it easy to set up a conference call on the IP Server 900. Each person who calls a pre-established number at a specified time is automatically added to the call. This can reduce or eliminate the customer's need for third-party conferencing services. The 31 channels available for "meet-me" conferencing are in addition to the regular system conferencing channels. For more details on this feature, see the *"Meet-me" Conferencing Feature Overview* (ESI # 0450-1367).

Shared-office tenanting

With **shared-office tenanting**, businesses can **share a common telephone system** while maintaining a true separation of various system resources, facilities, and features:

- Private or dedicated outside lines by line groups.
- Distinctive incoming ring assignments per tenant.
- Separate auto attendant greetings and branches.
- Individual "dial **0**" operators, music-on-hold sources, and paging zones.
- Unique day/night modes of operation.

Verbal User Guide™

With the **Verbal User Guide**, users have instant access to assistance in operating their ESI phones and voice mailboxes.[3] By pressing the **PROG/HELP** key (or **HELP** key on a 48-Key Feature Phone), the user is presented with extensive spoken and displayed prompts to assist with phone operation, voice mail features, programming instructions, and more. System Administrators and Reseller technicians can also use the Verbal User Guide to prompt them through infrequently used programming changes.

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1 The optional ***VIP 7 ACD*** application allows even easier and more substantial management of ACD operations.

2 Dynamic assignment of the conference channels allows for any combination of members (up to the maximum of 16) per conference — *e.g.*, ten four-member conferences and three eight-member conferences can take place simultaneously.

3 Not available from an ESI Cordless Handset.

Esi-Dex™

Locating and calling hundreds of frequently dialed phone numbers is easy when using ESI's **Esi-Dex** speed-dialing feature. Up to four separate lists ("Dexes") are available:

- **Station Dex** — All extensions within the system.
- **Personal Dex** — All speed-dial entries programmed by each individual user.
- **System Dex** — All speed-dial entries stored system-wide.
- **Location Dex** (available when Esi-Link is installed) — Lists all dial access codes associated with each location within an Esi-Link private network.

Saving numbers to the Personal Dex is just as easy. When Caller ID is presented with an incoming call or a voice mail message, one touch of the **ESI-DEX** key stores the provided number for future use.

Intelligent Call Forwarding™

Users of an IP Server 900 equipped with one or more PRI digital trunk circuits have access to ESI's unique **Intelligent Call Forwarding** feature. Users who forward their calls off-premises are able to view the *original* Caller ID data of incoming forwarded calls.[1]

Personal Caller ID

For situations in which the company's leading number identification data may not be the appropriate Caller ID for individual station users, the **Personal Caller ID**[2] feature makes it possible to define a different Caller ID number to be associated with, and sent for, each individual user. This feature provides E-911 support.[3]

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1 For more details about this feature, see the *Intelligent Call Forwarding Feature Overview* (ESI # 0450-0674).
2 Requires the installation of a PRI digital trunk circuit.
3 Check local regulations regarding E-911 compliance.

Flexible numbering

Flexible numbering provides the means to assign extensions, mailboxes, and department numbers based on specific customer requirements. The IP Server 900's flexible numbering is separated into three parts:

1. Selecting a numbering plan template;
2. Reassigning ranges of extensions, speed-dial numbers, and guest mailboxes (if needed);
3. Reassigning numbers for individual extensions, speed-dial numbers, guest mailboxes, and departments.

Selectable numbering plans

The selectable numbering plan template is the basis for flexible numbering assignment. When a numbering template is selected, all extensions, mailboxes, departments, and other system features are automatically assigned with the numbering plan of that template. Choosing the template that is closest to the customer's existing configuration greatly simplifies, or even eliminates the need for, number reassignment.

The choice of numbering plan determines how many extension numbers are available on a fully equipped IP Server 900:

- **Four-digit plan** — 864 extension numbers.
- **Three-digit plan** — 168 extension numbers.

Range reassignment

Flexible number range assignment is used to change the numbers of a block, or range, of extensions, speed-dial numbers, guest mailboxes, or departments.

The flexible numbering plan is very useful in matching station extension numbers with blocks of DID numbers assigned by the telephone company. If a customer already has an extension number directory assigned and does not want to change it, the flexible dialing plan will also accommodate this request.

Number reassignment

The number reassignment function will let the Installer assign new — or reassign existing — numbers for individual extensions, speed-dial numbers, departments, and mailboxes.

Station move

Station move is used by the Installer or System Administrator to move, or exchange, extension numbers and other station information between extensions of the same station type.[1] Programmable feature keys, personal greetings, voice mail messages, and other station information are automatically and instantly exchanged between the two stations when station move is done.

The Installer can use a separate programming function for flexible reassignment of station and department numbers through *Web ESI System Programmer (WESP)*.

Esi-Link and selectable numbering

In an Esi-Link network, certain IP Server 900 selectable numbering templates can be incompatible with some ESI systems. For additional details, refer to the *Esi-Link Product Overview* (ESI # 0450-0214).

Available numbering plan templates

To view the available numbering plan templates, refer to the *Flexible Numbering Feature Overview* (ESI # 0450-0952).

[1] Such stations must be like types — *e.g.*, digital phone to digital phone, IP phone to IP phone, or analog extension to analog extension.

Maintenance and updates

Certified ESI Resellers can perform system maintenance via the LAN/WAN or direct connection.

Authorized personnel can also use the convenient **browser-based** *Web ESI System Programmer* (see "System programming," page 19) to manage system settings. System updates are easily accomplished through software downloads. ESI systems are fully self-contained, for higher reliability and more security.

SNMP support

The IP Server 900 supports **Simple Network Management Protocol (SNMP)**, which allows monitoring devices via IP. For more information about the system's SNMP support, see the *SNMP Feature Overview* (ESI # 0450-1356).

Fax over IP

The IP Server 900 supports the industry-standard **T.38** protocol to ensure that faxes into and out of the system can be processed as reliably over IP as was traditionally the case over conventional telephony. For more information about the IP Server 900's **fax over IP** capabilities, see the *Fax Over IP Feature Overview* (ESI # 0450-1355).

Fax over e-mail

The IP Server 900's **fax over e-mail** feature automatically converts incoming faxes to documents in the almost universally used Portable Document Format (PDF), and e-mails them to a pre-defined address where they can be viewed, printed, archived, or forwarded as necessary. For more information about the IP Server 900's fax over e-mail capabilities, see the *Fax Over E-mail Feature Overview* (ESI # 0450-1354).

API support

The IP Server 900 provides a Microsoft® TAPI 2.x-standard **application programming interface (API)**, which allows qualified developers to integrate third-party solutions, such as Salesforce®.com. For more details about the ESI API, see the *ESI API Product Overview* (ESI # 0450-1353).

Optional ESI Presence Management

ESI Presence Management — RFID scanning technology combines with the IP Server 900 to offer an innovation in presence status, call control, entrance security, and documented tracking of users' work hours and attendance history. Highlighted benefits of ESI Presence Management include:

- **Remote entry control** with **built-in doorphone**.
- **Access control** through the use of authorized electronic keys (key fobs or scan cards).
- **Presence indication** to show "in" and "out" status of employees on programmed DSS keys.
- **Personal Call Routing** to modify the behavior of a station when the user is scanned in or out.
- **Optional third-party software**[1] to track, sort, and prepare employees' attendance data for easy entry into common business payroll software applications.

For more complete details about ESI Presence Management, consult the *ESI Presence Management Product Overview* (ESI # 0450-0794).

Optional ESI Media Management

ESI Media Management is a hardware/software combination which provides audio and video monitoring directly through the IP Server 900. These advanced capabilities help customers reduce many of the inherent risks in their organization. Additionally, ESI Media Management serves as an "all-in-one" solution by eliminating the need to install and manage multiple systems from various vendors.

Using the IP Server 900's built-in functionality[2], ESI Media Management collects and stores not only recordings of selected phone calls (call logging) but also video camera recordings, detailed call activity (SMDR), and building access events from across the customer's facility. ESI Media Management gives customers the flexibility to decide who is authorized to access the stored information, so there's no need to worry that information is getting into the wrong hands.

Here are just a few of the benefits ESI Media Management provides:

- Recording of all calls to and from employees for improved customer service and quality control.
- Capture and review of video from around customer facilities using standard video cameras.
- Use of live video to improve facility monitoring and enhance access control.
- Review of system-wide building access events and call detail records for employees.
- Quick location of a collection of related events using simple search criteria.

For more complete details about ESI Media Management, consult the *ESI Media Management Product Overview* (ESI # 0450-1238).

[1] *WaspTime* software is not sold by ESI but, rather, is available for direct purchase from the manufacturer, Wasp Barcode Technologies (*www.waspbarcode.com*).
[2] This functionality is similar to what is provided on a compatible ESI Communications Server by an optional Applications Services Card (ASC).

Optional ESI Mobile Messaging

ESI Mobile Messaging combines the advanced capabilities of the IP Server 900 with the convenience of users' existing e-mail accounts. When one receives a **message** (a voice mail or a recording) at an extension or guest mailbox, the person also receives an e-mailed notification to which a .WAV of the message may be attached. The notification's header contains information about the message — the Caller ID name and number, as well as the call's date, time, and duration. ESI Mobile Messaging also allows users to quickly do these (and more):

- **Listen to a message on one's PC or "smartphone"** — Play back a message on one's PC or "smartphone" by simply double-clicking the attachment.
- **Share messages** — Forward important messages to interested individuals, even if not on the user's system.
- **Choose which messages to handle and how to manage them** — A user with numerous messages can directly access any message right away.
- **Remotely manage messages** — A user can manage messages using Web mail from: a home PC or laptop; a personal (or alternate) e-mail account; or a "smartphone."
- **Store important messages** — Save a message attachment to a hard drive or USB Flash® drive.

> **Note:** Some of these capabilities require activation in user programming, as explained in this feature's *Installation Guide* (ESI # 0450-1231).

For more complete details about ESI Mobile Messaging, consult the *ESI Mobile Messaging Feature Overview* (ESI # 0450-1243).

Optional ESI Bluetooth Headset Interface

The **ESI Bluetooth Headset Interface** is an add-on device which integrates directly with an ESI phone (digital or IP), allowing users to "pair" a standard Bluetooth headset.[1] Once connected in this fashion, the headset user may answer, originate, and terminate calls seamlessly, using the key on the Bluetooth headset. The ESI Bluetooth Headset Interface maintains all headset capabilities available on the IP Server 900, but frees users from traditional and costly wired headsets and handset lifters. For more complete details, consult the *ESI Bluetooth Voice Integration Product Overview* (ESI # 0450-1173).[2]

Optional *ESI Salesforce.com Connector*

The ***ESI Salesforce.com Connector*** allows the IP Server 900's ESI API to connect to the popular Salesforce.com® customer relationship management (CRM) tool. This allows a business that uses Salesforce.com to make calls directly from the Salesforce.com Web interface and receive inbound screen "pops" from its Salesforce.com contact list. For more information on the *ESI Salesforce.com Connector*, see the *ESI Salesforce.com Connector Feature Overview* (ESI # 0450-1361).

[1] Bluetooth headsets sold separately (not available from ESI).
[2] The IP Server 900 does not support ESI Cellular Management, which is also mentioned in the *ESI Bluetooth Voice Integration Product Overview*.

Optional *VIP 7* PC applications

ESI's ***VIP 7*** (*VIP* stands for *Visually Integrated Phone*) works with the advanced capabilities of the IP Server 900 to enhance day-to-day communication — including the ability to control calls and organize voice mail and contacts. *VIP 7* captures and catalogs details about every call for better management. In addition, *VIP 7* makes it easy to program the phone with just a few mouse clicks. The familiar *Windows*® graphical user interface is intuitive and easy to learn, requiring minimal training. Each *VIP 7* application is a fully standalone application. *VIP* is offered in several configurations: the basic *VIP 7*, ***VIP 7 PC Attendant Console***, ***VIP 7 ACD Supervisor***, ***VIP 7 ACD Agent***, and ***VIP 7 Softphone***. Licenses for *VIP 7, VIP 7 ACD Agent*, and *VIP 7 Softphone* are sold in four-packs and 16-packs; *VIP 7 PC Attendant Console* and *VIP 7 ACD Supervisor* are sold in **single-seat** licenses. Up to eight seats for *VIP 7 PC Attendant Console* and *VIP 7 ACD Supervisor* can be installed on an IP Server 900. The maximum number of *VIP 7 Softphone* licenses is dependent upon the IP channels on the IP Server 900, which itself depends on the number of IP Resource Modules installed in the Master Control Unit and any IP Expansion Units.

The familiar *Windows*® graphical user interface is intuitive and easy to learn, requiring minimal training. With *VIP 7*, the user handles incoming and outgoing calls, manages contacts, and organizes voice mail, all on one's PC. Voice mail messages or personal recordings may be saved as .WAV files.

A *VIP 7* user can:

- Manage voice mail messages and call recordings.
- Organize all contacts in one convenient list.
- Control the ESI phone from a desktop PC.
- Capture all inbound and outgoing calls in historical log files.
- Program the phone with just a few mouse-clicks.
- Use instant messaging to provide a quick method of communication between users of *VIP 7* applications.

VIP 7 PC Attendant Console provides superior call handling abilities for busy attendants:

- **Incoming Calls** and **Holding Calls** displays that show calls in the attendant queue, calls that were re-routed to the operator, and system-wide recalling held calls.
- A **400-button Virtual Button Window** for single-click access to stations, departments, speed-dial numbers, and mailboxes, as well as many of the system features which can be assigned to programmable feature keys.

VIP 7 ACD Supervisor's benefits include:

- A **real-time status display** of departmental performance, including service level.
- A **view of agent status** — logged in, logged out, wrap, DND, off-hook, and off-premises.[1]
- Access to **six departmental reports**.[2]

VIP 7 ACD Agent gives each ACD agent:

- A **view of agent status** — DND, off-hook, and off-premises.[1]
- **Log-on, log-off, and wrap control** for up to two departments, directly from the PC.

The *VIP 7 Softphone* user[3] benefits from:

- **Combined operation** of *VIP 7* features and an **IP phone** resident within the PC.
- **Local or remote** operation.

Unlike many messaging offerings, *VIP 7* does **not** require installation of a *Microsoft Exchange*® server. This puts a powerful call and message management tool within financial reach for even smaller businesses.

For more details on the *VIP 7* family of applications, visit *www.esi-estech.com* or see the *VIP 7 Product Overview* (ESI # 0450-1340).

Note: The older *VIP* and *VIP SE* applications are incompatible with the IP Server 900.

1 Off-premises indication requires optional ESI Presence Management (see the *ESI Presence Management Product Overview*, ESI #0450-0794).
2 Requires the third-party *Crystal Reports* application.
3 *VIP Softphone* requires an available IP port and universal IP license, as well as use of a USB headset.

IP telecommunications capabilities

The IP Server 900 architecture, which provides a standard 64 universal IP ports[1] and can allow more through expansion, provides a robust infrastructure for both LAN[2]-based IP telephony and remote IP applications.

Standards-based design

The IP Server 900's IP capabilities are supported by **compliance with major industry standards**. ESI employs all applicable standards to ensure that, regardless of location, an IP Server 900's IP users experience the best communications quality.

- User Datagram Protocol (UDP).
- Layer 3 QoS support via DiffServ (Differentiated Services).
- Voice compression methods of G.711 (for locally installed IP stations), G.726 (for remotely installed IP stations and *VIP Softphone)*, and the optional G.729 (for Esi-Link connectivity).
- 802.3 100Base-TX Ethernet interfaces.
- Layer 2 Quality of Service (QoS) support through compliance with 802.1p for voice packet prioritization and 802.1q for VLAN (Layer 2) support.
- 802.3af Power over Ethernet.
- Dynamic Host Configuration Protocol (DHCP) for IP address conservation within a customer's LAN.
- Session Initiated Protocol (SIP) to support local SIP-compliant third-party IP telephones.
- T.38 fax relay.

Capacity constraints

The IP Server 900's full station capacity can be reached **either** (a.) with **all** extensions installed as IP stations **or** (b.) when at least 696 of the installed stations are IP instruments.

The IP Server 900 supports ESI's Power Over Ethernet (PoE) IP phones installed locally or remotely, in any combination.

If any IP Server 900 Master Control Unit or Expansion Unit fails, only the IP stations assigned to that Unit will go off-line.

[1] Licenses for the first eight ports are included with purchase of the Master Control Unit.
[2] Local area network.

Esi-Link private IP networking capabilities

The IP Server 900 supports up to 508 Esi-Link channels (in multiple ways, depending on system configuration), thus allowing up to 508 simultaneous VoIP connections between systems over an Esi-Link network. Esi-Link allows up to 100 individual sites to be connected together via a customer's WAN[1] or the Internet.

For further details, see the *Esi-Link Product Overview* (ESI document #0450-0214).

IP station sets

The IP Server 900 supports several types of IP telephones:

- **Desktop IP phones** — The **ESI 60IP Business Phone**, **ESI 40IP Business Phone**, and **48-Key IP Feature Phone II** can be installed in-house on the customer's network, or remotely wherever a broadband connection to the Internet is available. There is a two-port Ethernet switch built into each of these desktop IP phones. This provides a single Ethernet connection to the network for both the customer's IP phone and his office computer. Support for **Quality of Service** (see "Quality of Service (QoS) support," page 16) is critical in this type of installation, to ensure that there is no loss of audio or dropped voice packets during large data downloads.
  The phone includes built-in Power over Ethernet (PoE) capabilities for those customers whose LAN employs powered switches. In cases where the customer does **not** have PoE switches installed, the optional 48VDC adapter is used to provide operating power to the phone.
  When connected to the IP Server 900, an ESI desktop IP phone can optionally utilize DHCP to obtain an IP address from the customer's LAN. If the customer's LAN does *not* support DHCP, a static IP address will automatically be assigned by the system.
  An ESI desktop IP phone may also be installed outside the confines of the customer's LAN. When installed remotely, the phone uses the higher compression rate of G.726 to maximize voice quality. A remote location might include a remote facility, home office, or any other location where broadband Internet access is available. Remote IP users are connected directly to the system, and operate as if they were on-premises.
- The **ESI Local IP Cordless Handset II**[2] provides connection of the customer's LAN to the phone's base station. Users of Local IP Cordless Handsets II are free to move throughout their facility while staying in touch with customers and co-workers.
- For remote teleworkers, ESI also offers the **Remote IP Cordless Handset II**.[2] This phone connects like a "wired" Remote IP Phone, and can be installed anywhere broadband Internet access is available. The teleworker's home phone line can be connected into the Remote IP Cordless Handset II's base station.
- The optional ***VIP 7 Softphone*** combines the functionality of an ESI desktop IP phone and the *VIP 7* product in one PC-based phone. For more information about this product, see also "Optional *VIP 7* PC applications," beginning on page 13, as well as the *VIP 7 Product Overview* (ESI # 0450-1340).
- ESI additionally supports **SIP-compliant hardware endpoints** — *i.e.*, SIP "phones." However, due to limitations with SIP itself, not all of the ESI feature set is available via a compatible SIP phone.

> **Note:** Each compatible ESI IP phone (including *VIP 7 Softphone*) or SIP endpoint requires an available IP port and universal IP license.

1 Wide area network.
2 See the *ESI Cordless Handsets II Product Overview* (ESI # 0450-1228).

Quality of Service (QoS) support

Quality of Service is an important component in any converged or pure-IP telephone system. It increases the likelihood for IP voice communications to be clear, and free of dropped calls and delayed audio.

QoS is defined as providing the means for specific data streams in a network to be prioritized over other types of traffic. In the case of a voice over IP application, the IP packets carrying the voice conversation are given priority over data packets. When using the built-in two-port data switch to connect the IP Phone and customer's computer to the same Ethernet port, it is highly advisable for the customer's network to support QoS so that large downloads do not affect the quality of voice communications to the IP phone.

Benefits of QoS

Networks that are designed to support QoS are best suited for IP deployment since quality of voice is judged by the end-to-end experience of the user. It is not sufficient for ESI's IP applications to support QoS if all network components used in the transport of voice over the customer's LAN are not properly configured for QoS support.

The benefits of end-to-end QoS in any IP telephony application are many, and when absent, quite noticeable to the user:

- Available bandwidth is optimized by ensuring that voice packets are sent and delivered at a higher priority than "regular" data traffic on the LAN. This may allow the customer to delay upgrading the speed of transmission of his network. He may be able to defer this expense until other applications are added or IT changes in the business dictate it is necessary.
- The quality of the IP conversation is improved by ensuring that voice packets are delivered and "reassembled" at the other end of the conversation in order. This eliminates garbled conversation, hollowness, and noticeable gaps in speech.
- Unlike data packets, voice packets cannot be resent if they are dropped. Jitter[1] is reduced for voice packets by QoS. This improves the likelihood that all voice packets will not be dropped before being delivered at the other end of the IP conversation, as happens when the amount of jitter of a packet exceeds an acceptable level.
- The latency with which voice packets are delivered is minimized in a network employing QoS. This results in more natural-sounding speech patterns for both sides of an IP conversation.

802.1p and 802.1q standards for VLANs

Virtual LANs (VLANs) provide a method of separating data streams to make a local area network appear to be two or more networks. A VLAN is likely to be implemented in a business where IP telephony is heavily used. The VLAN segregates the voice packets onto their own network to prevent the degradation of voice quality, loss of packets, and late delivery of voice packets (latency).

Two standards are concerned with VLAN. Both are required to be supported in order to adequately support VLAN operation. These are:

- **802.1p** — Provides for the prioritization of voice packets. This standard establishes eight levels of priority, *0* through *7*, with *7* being the highest priority. Level 7 is reserved for those applications and packets that are considered network-critical. Levels 5 and 6 identify packets that are delay-sensitive. Priority levels below 5 are used for "loss-eligible" data, meaning that if a packet is lost and must be retransmitted, nothing is affected. This is not the case with voice, where if a packet is lost, portions of words will be missing or unintelligible. ESI defines its prioritization field at 5.
- **802.1q** — Dictates how the prioritization level (or "tag") is attached to each packet. Without this tagging of voice packets, prioritization would not be possible because there would be no differentiation between types of packets.

By compliance to the 802.1p and 802.1q standards, ESI's local IP phones have built-in prioritization to simplify managing traffic and QoS over a LAN.

[1] The variation from packet-to-packet in transit time, expressed in milliseconds. For a more detailed explanation, see the *Esi-Link Product Overview* (ESI # 0450-0214).

Differentiated Services (DiffServ)

This standard is primarily used with remote IP phones and Esi-Link installations in a WAN environment. This protocol allows IP voice packets to be prioritized over data transmission in LAN/WAN environments whose routers provide prioritization. As with all QoS provisioning within a LAN or WAN, the network components, such as routers and switches, must be able to support, and be configured for Quality of Service.

Some Internet connections may not support DiffServ. Contact the customer's ISP to determine whether it supports DiffServ.

Dedicated voice over IP resources

A **codec** is used to take the analog spoken voice, en**co**de it as an IP packet so it can be compressed and transmitted as a "data" packet. When received by another IP device (IP phone, SIP phone, or another system connected via Esi-Link), the IP packet is **dec**oded so that it is converted back into analog voice. Communication via IP is not possible without codecs.

Three types of industry-standard codecs are used by ESI's current IP phones and the IP Server 900: G.711, G.726, and the optional G.729a. This refers to the amount of compression that a voice packet will undergo when being converted into an IP packet.

G.711 is the non-compressed standard from which all other compression standards are established. IP phones that are locally installed use G.711.[1] Each ESI desktop IP phone has built-in G.711 and **G.726** codecs. Additionally, each IP channel of the IP Server 900 has dedicated G.711 and G.726 codecs for conversion between unlike compression standards. This conversion ability allows intelligible audio between remotely-installed and locally-installed IP phones.

Calls to or from a remotely-installed IP phone use the standard compression rate of G.726 for calls to/from the IP Server 900 (while **G.729a** is optionally used for Esi-Link, SIP stations, and SIP trunks). This reduces latency in the IP conversation and minimizes dropped or lost packets. Each of the IP channels on the IP Server 900 has a dedicated G.726 codec to support the connection of remotely installed IP phones. The **G.729a Resource Module** (ESI # 5000-0606) is equipped with 48 dedicated G.729 codecs. By dedicating codecs on each available IP port and Esi-Link channel, an IP phone or Esi-Link user will never be denied the ability to place or receive a call due to the lack of a codec.

Notes: Any SIP station can use G.711 or G.726 (as well as G.729a if installed).

SIP trunks can use G.711 (or G.729a if installed).

Esi-Link uses G.711 between IP Server 900 installations and G.726 to communicate with other ESI systems; it also can use G.729a, if installed, for either.

[1] Local and remote installations of *VIP Softphone* use G.726.

Power over Ethernet (PoE)

Each ESI desktop IP phone complies with the IEEE 802.3af standard for powering devices over a customer's existing local area network. This enhancement requires the customer to install the appropriate PoE network components, such as switches and routers. An ESI desktop IP phone can also be powered by using its optional 48VDC adapter. There are many benefits to designing an IP telephony application with PoE capabilities:

- By using the local area network to power the IP phones, a consistent voltage is provided to all phones without the fluctuations that frequently occur in commercial office buildings.
- Since all power is provided from one location, a single UPS system can be used to protect the IP phones from power surges, brown-outs, and other electrical anomalies.
- Powering the IP phones via the customer's LAN saves the cost and inconvenience of providing a fused power strip at each IP phone placement.

ESI has tested several Power over Ethernet devices for compatibility with its PoE IP phones:

- Cisco Catalyst 3560 24-port 10/100T PoE
- Adtran Netvanta 1224 PoE
- 3Com Superstack 3 4400 switch power
- 3Com PW130

In addition, the following mid-span Power over Ethernet devices have been tested:

- 3Com 3CNJPSE24 24-port Midspan Solution
- D-link DWL-P1012 12-port PoE Midspan

ESI VoIP Network Assessment

Due to the impact that network performance has on the quality of VoIP communications, **ESI VoIP Network Assessment** is highly recommended, particularly prior to any installation of VoIP equipment. This assessment involves placing an *appliance* on each node of a network which is to be used for carrying VoIP traffic. (Depending upon an installation's number of planned sites, as well as its particular VoIP applications, the use of more appliances may be required. This will be determined prior to the assessment.) For a specific time period, these appliances will send communications simulating the amount of VoIP traffic that is expected for the network. The result will be a detailed report that indicates the network either is ready for the traffic or has performance issues, such as unacceptable levels of latency or jitter, which should be addressed prior to the installation.

For more information about ESI VoIP Network Assessment, please contact your ESI sales representative.

Migration capability

For customers who outgrow their existing ESI systems, an IP Server 900 provides the perfect **upgrade** path. Most ESI station equipment currently installed on other ESI systems, including the legacy IVX® E-Class and X-Class systems, may be reinstalled on an IP Server 900:

ESI station equipment	Supported?
ESI 60IP	Yes
ESI 60D	Yes
ESI 40IP	Yes
ESI 40D	Yes
ESI 30D	Yes
48-Key IP Feature Phone II	Yes
48-Key Digital Feature Phone	Yes
ESI Digital Cordless Handset II	Yes
ESI Local IP Cordless Handset II	Yes
ESI Remote IP Cordless Handset II	Yes
60-Key Expansion Console	Yes

ESI station equipment	Supported?
60-Key Second Expansion Console	Yes
VIP 7 Softphone (software installation)	Yes
48-Key [local] IP Feature Phone	No
48-Key Remote IP Feature Phone	No
ESI Digital Cordless Handset I	No
ESI Local IP Cordless Handset I	No
ESI Remote IP Cordless Handset I	No
24-Key Digital Feature Phone	Yes
12-Key Digital Feature Phone	Yes
16-Key Digital Feature Phone	No
16-Key [local] IP Feature Phone	No
16-Key Remote IP Feature Phone	No

The IP Server 900 does not accept port cards from any other ESI systems. Similarly, IP Server 900 modules are incompatible with any other ESI systems.

System programming

Programming the IP Server 900 is greatly simplified through use of ***Web ESI System Programmer (WESP)***. This browser-based tool provides Installers and System Administrators the ability to easily review and modify the programming on the IP Server 900.

WESP enhances communication between the system and the programming application. Installers and System Administrators can navigate through an easy-to-use “tree” menu to access programming functions. The intuitive graphical user interface (GUI) makes learning the tool as simple as it is to use, resulting in a shortened training time for new technicians and System Administrators.

This application’s dependable backup-and-restore functionality retains programming, recorded custom prompts, and Caller ID information — giving you peace-of-mind when unforeseen circumstances occur.

The following are built into *WESP*:

- Programming of all phone system functions
- Send/Receive of programming
- Error-checking to prevent common mistakes
- Instant feedback informs whether data was successfully sent, and provides warnings and alerts about potential problems
- Import of system software

See also the *IP Server 900 Web ESI System Programmer Feature Overview* (ESI # 0450-1358).

Specifications and requirements

Capacities

Note: Refer also to "Capacities" in the "Advantage Summary" (page 3).

Because it accepts both IP and digital stations, and due to the more flexible configurations this capability allows, the IP Server 900 supports up to **864 stations** when configured appropriately.[1] The maximum station capacity can be achieved with maximum use of IP stations.

The IP Server 900 supports up to 192 trunks. These can be conventional trunks, SIP trunks, and/or digital trunks (T1/PRI).

Out of the box, the IP Server 900 supports eight universal IP ports, which can be any combination of Esi-Link channels, IP stations, SIP trunks, or SIP stations. Esi-Link channels do not detract from the number of available station or trunk ports.

System components

The IP Server 900 is comprised of one Master Control Unit, with the capability of adding one to three Expansion Units, each of which may be either a **D/A Expansion Unit** or an **IP Expansion Unit**. While either of these Expansion Units allows for expansion through the addition of **Resource Modules** installed in **carriers** (see "Modules and carriers," page 21), the IP Expansion Unit also possesses additional Applications Services capabilities beyond what are available in the Master Control Unit.

Each IP Server 900 unit has its own power supply unit to support installed modules. Units are connected together through rear-mounted RJ-45 cables (maximum length of each cable is three meters).

The Master Control Unit holds the motherboard, which controls all call control and switching within the IP Server 900. The motherboard also contains these integrated connectors and components:

- **Memory Module** — A CompactFlash drive that contains all system programming and configuration data, and pre-loaded voice prompts. Each Memory Module provides voice storage at 64 kilobits/second — the industry's highest-quality sampling rate.
- **Network Services Processor (NSP)** — The NSP consists of a dedicated Motorola® ColdFire® processor and Ethernet port. The front-panel RJ-45 LAN jack provides an auto-sensing 10/100/1000Base-T connection to a site's LAN.[2] In its basic configuration, the NSP provides remote access via Ethernet and the Internet for system programming and maintenance. The NSP is required for all LAN-based options, such as the various *VIP 7* applications.
- **On-board MOH and overhead paging inputs** — Connection of ancillary equipment is easy using the system's built-in jacks.

[1] See "Capacity constraints," page 14.
[2] Local area network.

Resource Modules and carriers

The IP Server 900 supports the items briefly described below. They are for use on **only** the IP Server 900. Each module uses one expansion slot on the Control Unit on which it is installed.

> **Note:** For each IP module, the quantity of IP stations is a combination of locally and remotely installed IP phones.

- **4-FXO Module (ESI # 5000-0603)** — Provides connectivity for four FXO loop-start CO lines. Depending on system configuration, 4-FXO Modules can be installed to support a maximum of 168 FXO ports.
- **4-FXS Module (ESI # 5000-0604)** — Provides connectivity for four analog station ports. Depending on system configuration, 4-FXS Modules can be installed to support a maximum of 176 FXS ports.
- **D8 Module (ESI # 5000-0605)** — Provides connectivity for eight digital station ports. Depending on system configuration, D8 Modules may be installed to support a maximum of 352 digital station ports. Each installed D8 Module will provide station connectivity in addition to the available IP stations which may be connected to the IP Server 900.
- **T1/PRI Module (ESI # 5000-0611)** — Provides connectivity for one T1 or PRI connection. Each IP Server 900 unit can support two T1/PRI Modules, up to a maximum of eight on the system. Therefore, depending on system configuration, T1/PRI Modules may be installed to support a maximum of **(a.)** 192 T1 channels or **(b.)** 184 PRI “B” channels and eight PRI “D” channels.
- **UIP Resource Module (ESI # 5000-0607)** — Provides an additional 64 ports of connectivity for ESI IP stations as well as SIP endpoint, SIP trunking, and Esi-Link channels. Supports the G.726 and G.711 codecs. Each IP Expansion Unit supports a maximum of 127 IP resource ports, for a system maximum of 508 IP resource ports.
- **G.729a Resource Module (ESI # 5000-0606)** — Supports up to 48 G.729a channels for SIP trunking, SIP endpoints, or Esi-Link channels. Each system Control Unit can accept one G.729a Resource Module, for a system maximum of 192 G.729a ports.
- **IP 900 D/A Carrier Card (ESI # 5000-0602)** — Provides connectivity for up to four **D/A Resource Modules** (4-FXO, 4-FXS, D8, or T1/PRI). Contains eight RJ-45 jacks, each allowing the output of four ports.

These modules cannot be hot-swapped (*i.e.*, replaced while the system is powered-up); it is necessary to power down all units on the system before one or more modules can be installed or replaced.

> **Note:** See also the *IP Server 900 Hardware Installation Manual* (ESI # 0450-1305).

Power consumption

The power consumption of the IP Server 900 when fully loaded — one Master Control Unit and three Expansion Units — is 127 watts;' if the installation is only the Master Control Unit with no Expansion Units, the power consumption is 30 watts. Each unit is powered by its own separately fused power transformer. For rack-mounted systems, a power shelf is available onto which all power transformers may be mounted so only one power cable is required for connection to a commercial AC power outlet or UPS system.

Since each IP Server 900 unit has its own distributed power, the heat dissipation of each power "brick" is well within the environmental range for proper operation of all system components. In an installation environment with insufficient space surrounding the system and mounting rack, the power shelf may be mounted at the top of the rack (above the Master Control Unit) so that the power bricks can utilize convection cooling as a means of dissipating any potential build-up of heat.

Environmental considerations

When planning the installation of the IP Server 900, observe good common sense by providing a dry, clean and accessible area.

If the equipment is to be rack-mounted, ensure that there is adequate room for a standard 19˝ rack. If wall-mounting is planned, ensure that all power cords have ready availability to a 110 VAC power outlet. For optimum performance, ensure that the system is located no further than 1,000 feet from the farthest station location.

The IP Server 900 is tolerant of broad ranges in environmental characteristics:

- Ambient room temperature should fall within the range of 40°–80° F.
- Relative humidity in the room should not exceed 90%.

FCC regulatory information

The IP Server 900 has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 and Part 68 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the system is operated in a commercial environment.

The IP Server 900 and all associated ESI telephone station equipment meet all FCC requirements for hearing-aid compatibility.

The FCC number for the IP Server 900 is *1T1MF08B33727*, with a ringer equivalency of *0.8*.

Glossary

Codec — The device required to *encode* analog spoken voice into IP packets for transmission through a VoIP network. The encoded voice is *decoded* at the receiving end, converting voice into an analog component.

FXO — Foreign Exchange Office. See *http://en.wikipedia.org/wiki/Foreign_exchange_office*.

FXS — Foreign Exchange Subscriber. See *http://en.wikipedia.org/wiki/Foreign_exchange_service_(telecommunications)*.

IEEE — Institute of Electrical Engineers; the professional organization that establishes standards for, among others, the telecommunications industry.

NSP — Network Services Processor; the ESI device, mounted on the Main Board, that provides for an Ethernet connection between the IP Server 900 and the customer's local area network (LAN). Multiple applications may run concurrently over the NSP connection, such as *VIP* and remote Internet programming.

PoE — Power over Ethernet; this IEEE standard (802.3af) defines the method of injecting power over a customer's local area network cabling infrastructure to operate TCP/IP devices at the Ethernet port. ESI uses this method, in conjunction with a customer-provided power switch, to operate its PoE local IP phones.

RF — Radio frequency.

RFID — Radio frequency identification.

VoIP — Voice over Internet Protocol.

About ESI

ESI (Estech Systems, Inc.) designs and manufactures high-performance phone systems for businesses and organizations. ESI uses advanced technology to design IP and digital communications systems that integrate built-in capabilities, advanced features, and highly differentiated applications into flexible products that are easy to use and keep employees productive. ESI has sold over 250,000 business communications systems through hundreds of factory-trained Certified Resellers. Founded in 1987, ESI is a privately held corporation with headquarters in Plano, Texas.