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

ESI-600

System for converged communications

Advanced digital and IP communications platform

The **ESI-600** represents an innovative approach to digital and IP convergence. The science behind the switch is sophisticated in its simplicity: Design a platform with the flexibility to support digital functionality with the ability to be configured as a pure IP-based communications system. It's ideal for any business that wants the familiarity of digital telephony, the benefits of full IP-to-the-desktop, or anything in-between.

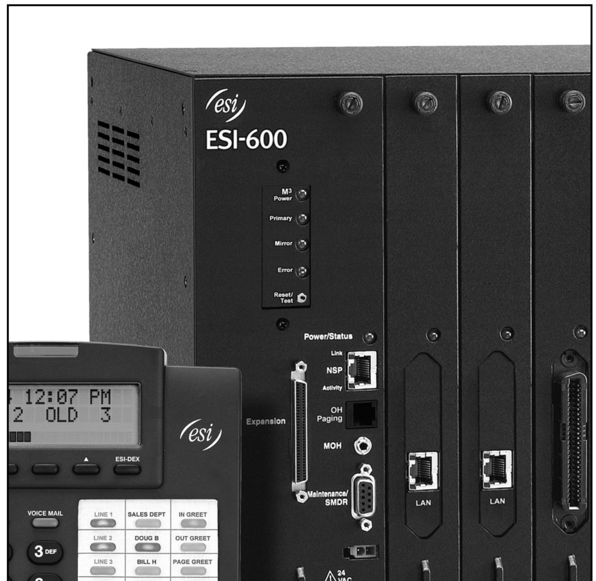
Introduction

The ESI-600 is a third-generation converged business communications platform capable of supporting both traditional digital stations and VoIP technologies. A maximum system capacity of 624 communications ports is supported. The backplane, switching matrix, and main board are designed to allow for a traditional digital installation or a pure-IP configuration in an expanded capacity. The ESI-600's non-blocking architecture increases station capacity to a full complement of 408 telephones.[1]

Many new functions and feature enhancements are introduced with the ESI-600. Additionally, the redesigned architecture of the ESI-600 integrates advanced IP functionalities, such as dedicated IP resources, the ability to support multiple Integrated VoIP Cards (IVCs), up to 48 Esi-Link channels and remotely-installed IP Feature Phones.

In addition to the ESI-600's expanded station capacity, ESI has made significant additions to its family of station instruments. **ESI Cordless Handsets** come in two sizes, each of which has three models to provide more connection choices — Digital, Local IP, and Remote IP. The **48-Key IP Feature Phone II** supports Power over Ethernet (PoE). The optional ***VIP Softphone*** combines the functionality of a 48-Key IP Feature Phone II and the *VIP Professional* product in one PC-based phone.[2] All ESI IP Phones are standards-compliant and operate with the customer's local area network to promote Quality of Service (QoS).

The ESI-600 provides an ideal cost-effective upgrade path for several existing IVX systems[3]. See "Migration capability," page 12, for more details.



Contents

Color brochure available

ESI part # 0450-0900 (Web version downloadable from *www.esi-estech.com/brochures*).

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1 Maximum capacity is achieved in configurations of at least 144 IP stations (installed locally or remotely).
2 See the *VIP Product Overview* (ESI # 0450-0608).
3 IVX E-Class (IVX 128e and IVX 72e) Generation II and IVX X-Class (IVX 128x and 256x).

Advantage summary

Capacities

- Total system ports: 624
- Maximum trunk ports: 168
- Maximum ESI-DLC/E2-DLC12 cards: 6
- Maximum IVCs: 19
- Maximum station ports: 408[1]
  - IP stations: 408
  - Digital stations: 336
  - Analog stations: 188
- Maximum Esi-Link channels (IVC with Esi-Link): 48, up to two IVC Esi-Link cards
- Total conference channels[2]: 64 (dynamic; on-demand): 16 maximum per conference.

> **Note:** Achieves best audio performance when using digital trunks.

Voice mail

- On-board integrated auto-attendant/voice mail channels: 32 standard
- Total combined voice mailboxes: 1,492
  - Maximum user mailboxes: 408
  - Maximum info/guest mailboxes: Up to 1,000
  - Maximum "special purpose" mailboxes: 84
- Total voice mail storage capacity: 1,200 hours

Standard features

- Account codes
- Automatic call distribution
- Built-in Network Services Processor (NSP)
- Caller ID key
- Distinctive ring for trunks
- Enhanced Caller ID
- Esi-Dex integrated directories
- Fax tone detection
- Flexible numbering plans
- Personal Calling Line ID (CLIP[3]) for DID users
- Recording of calls
- Shared-office tenanting (maximum of eight tenants)
- Station redial and callback

Optional applications

- ESI Presence Management
- Mirrored Memory Module (M3)
- Esi-Link IP private networking
- Power over Ethernet support for IP Feature Phone II
- Dual-configuration 48-key IP Feature Phone II (supports local and remote installations)
- Digital, Local IP, and Remote IP Cordless Handsets
- Third-party SIP stations
- *VIP* (*Visually Integrated Phone*) and *VIP Professional*
- *VIP ACD Supervisor* and *VIP ACD Agent*
- *VIP PC Attendant Console*[4]
- *VIP Softphone*

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1 Maximum capacity is achieved in configurations of at least 144 IP stations (installed locally or remotely).

2 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.

3 Calling Line Identification Presentation modifies the Caller ID sent by a station to reflect the user's assigned DID.

4 Previously called *ESI PC Attendant Console*.

Hardware description

The ESI-600 is a compact, rack-mounted system that consists of a total of four cabinets — the Main Cabinet and up to three Expansion Cabinets. If desired, the cabinets may be wall-mounted, but rack-mounting is the preferred method of installation.

Processing power is provided by a Motorola® ColdFire® MCF-5407 commercial-grade microprocessor, designed specifically for 24/7 operation. This 54 MHz device houses 128 MB of SDRAM for stored program control. It also interfaces with 3 on-board DSPs that manage the HDD controller, inter-card communications, and telephony services, ensuring rapid, dependable communications among all system resources: trunks, digital stations, and IP Phones.

Power provisions

Each cabinet is powered by its own power supply. In rack-mounted installations, a power shelf is available that provides AC connection for each of the four cabinet 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 four. The power shelf is separately fused to protect system components against erratic power fluctuations.

Two additional AC power outlets are available for use by installation and maintenance personnel. Common uses include powering laptop computers and diagnostic equipment, such as a protocol analyzer.

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

Cabinet connection

Connection between cabinets is made through a SCSI (Small Computer System Interface) cable, which is shipped with each expansion cabinet. This SCSI cable extends the motherboard from cabinet to cabinet, creating a common backplane.

Main board

The Main Board houses a built-in Network Services Processor (NSP) for all applications requiring direct connection of the ESI-600 to the customer’s local area network. These applications include SMDR, system programming via TCP/IP, and ESI options such as the *VIP* family of software applications and ESI Presence Management’s *ESI TimeLine* payroll data collection option.

The optional M3 (Mirrored Memory Module) can be integrated to provide a full, real-time back-up of system programming data and voice messages. The M3 is designed with RAID 1 redundancy technology. If the main Hard Disk Driver (HDD) controller senses a drive failure, it will automatically switch to the mirrored drive and continue to run. This switch of drives initiates an audible alarm with a visual LED indication on the front panel of the Base Cabinet.

Fully flexible platform

The ESI-600 offers impressive expansion capabilities. Each ESI-600 cabinet has seven universal card slots. At maximum capacity, the system can grow to 168 CO trunks and 408 stations (digital and IP)[1]. Port cards are mounted onto the ESI-600 card carrier for insertion into the cabinet. This carrier also makes all cards “hot-swappable.”

[1] Maximum capacity is achieved in configurations of at least 144 IP stations (installed locally or remotely).

Features at a glance

The ESI-600 provides the power of a PBX with the ease of use for which every ESI phone system is renowned. Its feature set, capacities and scalability ensure:

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

Integrated voice mail — A full complement of practical voice mail features that are easy to use is standard on every ESI-600 system:

- In addition to its 624 call processing ports, the ESI-600 is configured with a full **32 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).
- **1,200 hours of 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.
  - Nine customizable recordings.
- **New-message notification** can be delivered off-premises to a phone or pager.
- Leaving messages for several 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 ESI-600 provides rich, comprehensive auto attendant features, such as:

- **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.

Flexible conference channels — The ESI-600 reserves **64 channels** for conferencing. These can be dynamically[1] connected in multi-party conversations up to 16 channels per conference. Any combination of conference channels may be joined together, as long as the originating party is an ESI-600 user. All channels reserved for conferencing are dynamically balanced for optimum audio performance.

Shared-office tenanting — Up to eight 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.

ESI Presence Management — RFID scanning technology combines with the ESI-600 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 *ESI TimeLine* software** to track, sort, and prepare employees' attendance data for easy entry into common business payroll software applications.

For more complete details, consult the *ESI Presence Management* Product Overview (ESI document #0450-0794). Resellers may download this document from *www.esiresellers.com* (password required).

[1] Dynamic assignment of the 64 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.

Enhanced automatic call distribution — Manage call overload and increase customer satisfaction with ESI's standard call center feature.[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 any Feature Phone display.
- 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.

Verbal User Guide™ — Users have instant access to assistance in operating their ESI phones and voice mailboxes.[2] By pressing the **HELP** key, 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.

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 ESI-600.
- **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 ESI-600 system equipped with one or more PRI digital trunk circuits have access to this unique feature. Users that forward their calls off-premises are able to view the *original* Caller ID data of incoming forwarded calls.[3]

Personal Caller ID[4] **—** For situations in which the company's leading number identification data may not be the appropriate Caller ID for individual station users, the ESI-600 makes it possible to define a different Caller ID number to be associated with, and sent for, each individual user. This flexibility ensures that the ESI-600 is fully compliant with state and local E-911 requirements[5].

Flexible numbering

Flexible numbering provides the means to assign extensions, mailboxes, and department numbers based on specific customer requirements.
The ESI-600's flexible numbering is separated into three parts:

1. Selecting a numbering plan template;
2. Reassigning ranges of extensions and guest mailboxes (if needed);
3. Reassigning numbers for individual extensions, 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 **number** of digits of extension numbers determines the number of stations that can be connected to the ESI-600. If a three-digit plan is chosen, a maximum of 168 extension numbers are available. In a four-digit plan, the ESI-600 can expand up to its maximum station count of 408.

Range reassignment

Included in *Esi-Access*, flexible number range assignment is used to change the numbers of a block, or range, of extensions or guest mailboxes.[6]

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.

1 The optional ***VIP ACD*** application allows even easier and more substantial management of ACD operations. See the *VIP ACD Product Overview* (ESI # 0450-0988).
2 Not available from an ESI Cordless Handset.
3 For more details about this feature, see the *Intelligent Call Forwarding Feature Overview* (ESI # 0450-0674).

4 Requires the installation of a PRI digital trunk circuit.
5 Check local regulations regarding E-911 compliance.
6 Range assignment of department numbers and special-purpose mailboxes is not supported at initial release of flexible numbering. Function 34 can be used to reassign department numbers.

Number reassignment

The number reassignment function will let the Installer assign new — or reassign existing — numbers for individual extensions, 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 *Esi-Access*.

Esi-Link and selectable numbering

In an Esi-Link network, certain ESI-600 selectable numbering templates can be incompatible with some ESI systems. The following chart lists Esi-Link limitations regarding selectable numbering:

	Templates	
Esi-Linked system	**Three-digit**	**Four-digit**
ESI-600	Yes	Yes
IVX X-Class	Yes	No
IVX E-Class Generation II	Yes	No
IP E-Class Generation II	No	No
IVX E-Class Generation I	No	No
IP E-Class Generation I	No	No
Legacy systems	No	No

Available numbering plan templates

(See page 7.)

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

Dial plan selections: Three-digit

Selection 100 (default)

From	To	Used for
100	267	Extensions
280	299	Departments
300	489	Guest/info mboxes

Selection 200

From	To	Used for
200	367	Extensions
380	399	Departments
100	199	Guest/info mboxes
400	489	Guest/info mboxes

Selection 300

From	To	Used for
300	467	Extensions
470	489	Departments
100	289	Guest/info mboxes

Common to all three-digit dial plan selections

From	To	Used for
0	—	Operator
490	499	Q & A mailboxes
500	—	Broadcast mailbox
501	532	Group mboxes
533	542	Cascade notif. mboxes
600	699	System speed-dial

From	To	Used for
700	709	Esi-Link[1] locations
770	799	Esi-Link locations
71	76	Esi-Link/CO line grps.
8	—	CO line grp.
9	—	CO line grp./ARS
*	—	Call pickup
#	—	Paging

Dial plan selections: Four-digit

Selection 1000

From	To	Used for
1000	1407	Extensions
1408	1471	Departments
3000	3999	Guest/info mboxes
4000	—	Broadcast mailbox
4001	4032	Group mboxes
4040	4059	Q & A mboxes
4060	4079	Cascade notif. mboxes
6000	6999	System speed-dial

Selection 2000

From	To	Used for
2000	2407	Extensions
2408	2471	Departments
3000	3999	Guest/info mboxes
4000	—	Broadcast mailbox
4001	4032	Group mboxes
4040	4059	Q & A mboxes
4060	4079	Cascade notif. mboxes
6000	6999	System speed-dial

Selection 3000

From	To	Used for
3000	3407	Extensions
3408	3471	Departments
2000	2999	Guest/info mboxes
4000	—	Broadcast mailbox
4001	4032	Group mboxes
4040	4059	Q & A mboxes
4060	4079	Cascade notif. mboxes
6000	6999	System speed-dial

Selection 4000

From	To	Used for
4000	4407	Extensions
4408	4471	Departments
3000	3999	Guest/info mboxes
2000	—	Broadcast mailbox
2001	2032	Group mboxes
2040	2059	Q & A mboxes
2060	2079	Cascade notif. mboxes
6000	6999	System speed-dial

Selection 5000

From	To	Used for
5000	5407	Extensions
5408	5471	Departments
3000	3999	Guest/info mboxes
4000	—	Broadcast mailbox
4001	4032	Group mboxes
4040	4059	Q & A mboxes
4060	4079	Cascade notif. mboxes
6000	6999	System speed-dial

Selection 6000

From	To	Used for
6000	6407	Extensions
6408	6471	Departments
3000	3999	Guest/info mboxes
4000	—	Broadcast mailbox
4001	4032	Group mboxes
4040	4059	Q & A mboxes
4060	4079	Cascade notif. mboxes
2000	2999	System speed-dial

Common to all four-digit dial plan selections

From	To	Used for
0	—	Operator
71	76	CO line grps. or Esi-Link loc. prefixes
700	709	Esi-Link locations
770	799	Esi-Link locations

From	To	Used for
8	—	CO line grp.
9	—	CO line grp./ARS
*	—	Call pickup
#	—	Paging

1 See "Function 8: IP PBX programming" in the *ESI-600 Installation Manual*.

Optional *VIP* PC applications

ESI's optional ***VIP*** takes the power of *Microsoft® Outlook®* and adds a missing critical application: control of phone calls, faxes, and voice mail messages.

VIP is offered in several configurations: the basic *VIP*, ***VIP Professional***, ***VIP PC Attendant Console***, ***VIP ACD Supervisor***, ***VIP ACD Agent***, and ***VIP Softphone***.

VIP PC Attendant Console, *VIP ACD Supervisor*, and *VIP Softphone* are sold in **single-seat** licenses. *VIP Professional* and *VIP ACD Agent* licenses can be combined to add up to seats in quantities of two, five, 25, 100, and unlimited. The basic *VIP* is sold separately in seats of two, five, 25, 100, and unlimited.

The maximum number of seats for *VIP PC Attendant Console* and *VIP ACD Supervisor* on an ESI-600 is eight seats each. The maximum number of *VIP Softphone* licenses is dependent upon the available remote IP channels provided with an installed IVC.

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

The ESI-600 supports **integration with third-party fax server applications** in systems equipped with a digital PRI circuit. Each *VIP* user receives his faxes in that user's own *Outlook* Inbox as an attachment. Faxes can be printed, archived, or attached to any e-mail for forwarding. This improves office efficiency, and ensures that sensitive faxed information is not lying in a common fax machine's tray.

All *VIP* users can:

- Manage voice mail, e-mail and faxes from the *Outlook* Inbox.
- Organize all contacts in one convenient list.
- Control the ESI Feature Phone from a desktop PC.
- Capture all inbound and outgoing calls in historical log files.
- Program the phone with just a few mouse-clicks.

Users of *VIP Professional*, *VIP PC Attendant Console*, *VIP ACD Supervisor*, *VIP ACD Agent*, and *VIP Softphone* receive additional benefits:

- An **enhanced graphical user interface** (GUI) to further increase user efficiency.
- **Text-messaging** to provide a quick method of communication between users of these applications.
- An **auto-record** feature, so select users (up to 16) never again miss recording important calls.

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

- **Incoming Calls** and **Holding Calls** displays showing calls in the attendant queue, calls that were re-routed to the operator, and system-wide recalling held calls.
- A **200-button Virtual Button Window** for single-click access to stations, departments, speed-dial numbers, and features.

VIP 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 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 Softphone* user benefits from:

- **Combined operation** of *VIP Professional*-level features and an **IP Phone** resident within the PC.
- **Local or remote** operation.

Unlike many unified messaging offerings, *VIP* 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, consult the following documents, which Resellers may download from *www.esiresellers.com* (password required):

- VIP Product Overview
  - ESI #0450-0608
  - (Contains details concerning basic *VIP*, *VIP Professional*, and *VIP Softphone*).
- *VIP PC Attendant Console Product Overview*
  - ESI #0450-0914
- *VIP ACD Supervisor/ACD Agent Product Overview*
  - ESI #0450-0988

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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.

IP telecommunications capabilities

The ESI-600 architecture provides a robust infrastructure for both LAN-based IP telephony and remote IP applications.

Standards-based design

The ESI-600's IP capabilities are supported by **compliance with major industry standards**. ESI employs all applicable standards to ensure that, regardless of location, ESI-600 IP users experience the best audio quality.

- User Datagram Protocol (UDP).
- Layer 3 QoS support via DiffServ (Differentiated Services).
- Packet compression levels of G.711 (for locally installed IP stations), G.726 (for remotely installed IP stations and *VIP Softphone)*, and 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 SIP-compliant third-party IP telephones.

Multiple IVCs supported

The ESI-600's full station capacity of 408 phones can be met when at least 144 of the installed stations are IP instruments. Up to 17 Integrated VoIP Cards (IVCs) may be installed, each supporting 24 IP stations. The IVC supports ESI's Power Over Ethernet (PoE) IP Phones installed locally or remotely, in any combination. The primary IVC manages the successful negotiation for "registration" of each IP Feature Phone.

If any IVC fails, only the 24 IP stations assigned to that card will go off-line.

Esi-Link private IP networking capabilities

The IVC Esi-Link card is reserved for the support of either eight or 24 Esi-Link channels. With the Esi-Link IP networking option, up to 100 individual sites may be connected together via a customer's WAN or the Internet.

Up to two IVC Esi-Link cards may be installed in an ESI-600, providing up to 48 simultaneous VoIP connections between systems.

For further details, see the *Esi-Link Product Overview* (ESI document #0450-0214). ESI-trained Resellers may download this document from *www.esiresellers.com/docs* (password required).

IP station sets

The ESI-600 supports several types of IP telephones:

- The **48-Key IP Feature Phone II** can be installed in-house on the customer's LAN, or remotely wherever a broadband connection to the Internet is available. There is a two-port Ethernet switch built into the IP Feature Phone II. This provides a single Ethernet connection to the LAN for both the customer's IP Phone and his office computer. Support of Quality of Service 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 those 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 ESI-600 system, the 48-Key IP Feature Phone II 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.

  The 48-Key IP Feature Phone II 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**[1] provides connection of the customer's LAN to the phone's base station. Users of the Local IP Cordless Handset 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**.[1] 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's base station.
- The optional ***VIP Softphone*** combines the functionality of a 48-Key IP Feature Phone II and the *VIP Professional* product in one PC-based phone. For more information about this product, see also "Optional *VIP* PC applications," beginning on page 8, as well as the *VIP Product Overview* (ESI # 0450-0608).

(Continued)

1 See the *ESI Cordless Handsets Product Overview* (ESI # 0450-0840).

- 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 SIP phone. For a complete list of ESI features supported via SIP, see *Technical Update* #242 ("Installing a SIP phone on an ESI-600"), available from *ww.esiresellers.com/tech* (password required).

  The following SIP hardware endpoints have been tested with the ESI-600:

  – Aastra 9133i
  – Polycom Soundpoint IP 301
  – Polycom Soundpoint IP 501

> **Note:** Each compatible ESI IP Phone [IP Feature Phone II, IP Cordless Handset (Local or Remote), or *VIP Softphone*] or SIP endpoint requires an available IVC port and the activation in the system of either a local or remote license before the IVC will connect to the IP Phone. When an IP Phone is programmed in the system, a license is consumed.

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.

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.

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.

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 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.

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 or other Esi-Link system), 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 48-Key IP Feature Phone II and IVCs: G.711, G.726, and G.729. 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. Each 48-Key IP Feature Phone II has built-in G.711 and **G.726** codecs. Additionally, each channel of the IVC has dedicated G.711 and G.726 codecs for conversion between unlike compression standards. This conversion ability of the IVC allows intelligible audio between remotely-installed and locally-installed IP Phones.

Calls to or from a remotely-installed IP Phone use standard compression rates of G.726 (calls to/from the IVC) and **G.729** (calls to/from Esi-Link channels). This reduces latency in the IP conversation and minimizes dropped or lost packets. Each of the 24 channels on the IVC has a dedicated G.726 codec to support the connection of remotely installed IP Phones. The **Esi-Link IVC** is equipped with 24 dedicated G.729 codecs. By dedicating codecs on each available IVC and Esi-Link IVC channel, an IP Phone or Esi-Link user will never be denied the ability to place or receive a call due to a lack of a codec.

Power over Ethernet (PoE)

ESI's 48-Key IP Feature Phone II 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. The 48-Key IP Feature Phone II can also be powered by using the optional 48VDC adapter. There are many benefits to designing an IP telephony application with Power over Ethernet 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.

Migration capability

For customers that outgrow their existing IVX E-Class (Generation II) or X-Class systems, the ESI-600 provides the perfect **upgrade** path. Most station equipment currently installed on the E-Class or X-Class system may be reinstalled on the ESI-600.

The ESI-600 supports a wide range of port cards; **however**, pre-Generation II legacy cards (see the table *below*) cannot be used on this platform, because they lack the processing power and memory storage required for proper operation in a large system with a heavier traffic load:

Cards NOT supported by the ESI-600

Port Card	Part No.
612	5000-0104
684	5000-0160
D12	5000-0135
A12	5000-0160
LNC	5000-0149
DLC12	5000-0157
IVC	5000-0318

Generally, all Generation II port cards **are** fully supported by the ESI-600.

No migration path is available for customers of the legacy IVX 128 and IP E-Class series systems.

Specifications and requirements

Capacities

The ESI-600 represents expanded station capacities far beyond that of any other ESI platform. The maximum number of station ports possible in the ESI-600 is 408.[1]

The IVC supports 24 channels to which local IP Phones or remote IP Phones may be connected. This is **double** the station capacity of any port card that supports digital phones. Therefore, the maximum station capacity can be achieved when at least 144 IP stations are installed (local or remote).

The maximum trunk capacity of the ESI-600 is 168. Up to 144 of the 168 allowable trunks may be digital (T1 and/or PRI), connected to the system via the E2-DLC12 card or the ESI-DLC card. A maximum of six digital trunk cards may be installed in the system.

A maximum of 48 Esi-Link channels may be configured in an ESI-600 system. This maximum is achieved by installing up to two Esi-Link IVCs in the system. Esi-Link channels do not detract from the number of available station or trunk ports.

System components

The ESI-600 is comprised of one base cabinet, with the capability of adding up to 3 expansion cabinets. Each cabinet has its own power supply unit to support the port cards inserted in each. Cabinets are connected together through front-mounted cables.

The base cabinet holds the main board, which controls all call control and switching within the ESI-600. The main board also contains these integrated connectors and components:

- **Memory Module** — A hard drive with improved performance that contains all system programming and configuration data, and pre-loaded voice prompts. The Memory Module provides 1,200 hours of 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 jack provides a 10/100Base-T connection to a site's LAN.[2] In its basic configuration, the ESI-600 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 *VIP* and ESI Presence Management's *ESI TimeLine* time and attendance management application.
- **Optional M3 memory back-up** — Using RAID[3]-1 hard drive technology, the Mirrored Memory Module (M3) maintains system operation on a separate disk drive in the event of a hard drive failure. M3 is required when redundancy of system programming, speed-dial entries, and voice mail messages and prompts is desired.
- **On-board MOH and overhead paging inputs** — Connection of ancillary equipment is easy using the built-in jacks on the front plate of the Main Board.
- **Serial port** — SMDR call detail data is output from this port. Technicians connect their laptop computers to this port to perform on-site programming.

[1] Maximum capacity is achieved in configurations of at least 144 IP stations (installed locally or remotely).

[2] Local area network.

[3] *RAID* means *Redundant Array of Independent Drives*.

Port cards supported

The ESI-600 supports all Generation II “E2” cards that are common to ESI’s X-Class and E-Class platforms. The ESI-600 port card adapter must be used when inserting these cards into the ESI-600’s backplane. This card carrier ensures that the cards mounted onto it can be inserted and removed under full system power.

The following chart indicates the types of port cards that are supported by the ESI-600, as well as the capacities of each:

	Ports				
Card	**COs**	**Digital or IP stns.**	**Analog stns.**	**Esi-Link chs.**	**Max. per system**
E2-684	6	8	4	—	**28**
E2-612	6	12	—	—	**28**
E2-A12	—	—	12	—	**15**[1]
E2-DLC12	24 (T1) or 23B +1D (PRI)	12	—	—	**6**
ESI-6ALC	6	—	—	—	**28**
ESI-DLC	24 (T1) or 23B +1D (PRI)	—	—	—	**6**
IVC 12-station	—	12 (IP)[2]	—	—	**17**
IVC 24-station	—	24 (IP)[2]	—	—	
Esi-Link IVC 8-channel	—	—	—	8	**2**
Esi-Link IVC 24-channel	—	—	—	24	

Power consumption

The ESI-600’s Base Cabinet and three Expansion Cabinets are each powered by their own separately fused power transformer. For rack-mounted systems, a power shelf is available onto which all power transformers may be mounted so that there is only one power cable required for connection to a commercial AC power outlet or UPS system.

At full capacity with a full complement of stations, the ESI-600 draws a maximum of 720 watts. Since each cabinet 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 installation environments where insufficient space surrounding the system and mounting rack is permitted, the power shelf may be mounted at the top of the rack (above the Base Cabinet) 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 ESI-600 system, 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 ESI-600 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%.

[1] A maximum of 188 analog station ports (combination of E2-684 and E2-A12) are supported by the ESI-600.
[2] Combination of local and remotely installed IP Phones.

FCC regulatory information

The ESI-600 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 ESI-600 and all associated ESI telephone station equipment meet all FCC requirements for hearing-aid compatibility.

NTS Test Report B5317 includes all testing procedures and satisfactory results data. The FCC number for the ESI-600 is *1T1MF08B33727*, with a ringer equivalency of *0.8*.

Power over Ethernet compatibility

With the inclusion of PoE support for 48-Key IP Feature Phones II on the customer's in-house LAN, 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

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

Glossary

CLIP — Calling Line Identification Presentation; the standard maintained by the ESI-600 in which the calling party from the system presents a modified Caller ID string to the outside party to which he is calling. The purpose of this is to ensure that the ESI-600 party is sending the true Caller ID for his location, not necessarily the Caller ID associated with the system's primary incoming line number. This standard is required to meet E-911 requirements set forth by Public Utility Commission boards of municipalities and states.

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

HDD — Hard Disk Drive; the device on which the system's operating software program, and voice mail prompts and messages are stored.

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

ICC — Inter-card communication; describes the method in which cards within a cabinet, as well as multiple card cabinets, communicate with each other.

NSP — Network Services Processor; the ESI device, mounted on the Main Board, that provides for an Ethernet connection between the ESI-600 system and the customer's local area network (LAN). Multiple applications may run concurrently over the NSP connection: *VIP*, ESI Presence Management optional *ESI TimeLine* payroll application, 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.

RAID1 — Redundant array of independent drives.

VoIP — Voice over Internet Protocol.

About ESI

ESI (Estech Systems, Inc.) is a privately held corporation based in Plano, Texas. Founded in 1987, ESI specializes in telephone systems for the small to mid-size business. Since its days as a small start-up, ESI has enjoyed exceptional stability and growth while maintaining its dedication to small company values — including the need to take care of the most important part of the equation: your business.

ESI pioneered the all-in-one telephone and voice mail system. The original IVX, introduced in 1996, represented a radical breakthrough in system design: the inclusion of a full suite of features within a single integrated telephone design.

Committed to excellence, ESI is an ISO-9001-2000 certified company — assuring that quality is fundamental.