PRODUCT OVERVIEW

Remote IP Feature Phone

48-key • For E-Class (IVX®/IP) and IVX X-Class

The “virtual extension” that goes nearly everywhere

ESI’s 48-key **Remote IP Feature Phone**, a special version of the IP Feature Phone, uses state-of-the-art VoIP technology to operate at almost any outlying location that has a suitable broadband data connection. This fully functional, off-site extension of an ESI IP-enabled phone system provides the same business phone features and functions of the highly featured, in-office ESI Feature Phone.

The Remote Phone is based on a principle at the heart of VoIP technology — once voice is converted to IP data packets, it can travel almost anywhere that data can: from desktop to desktop over the LAN; between offices over a WAN; or to remote sites connected via a dedicated broadband connection (or, in some cases, the public Internet).

While identical in appearance to ESI’s 48-Key Feature Phone sets, the Remote Phone contains voice compression circuitry that reduces its bandwidth requirements by approximately 80%, allowing it to function more efficiently over a broadband connection. Once installed at the remote site, it operates like other extensions of the host ESI system. The Remote Phone is transparent to an outside caller or other extension user, has full access to voice mail, can initiate a page, and can perform virtually all other advanced station features.

ESI’s Remote Phone technology is the ideal solution for today’s off-site teleworkers, an executive desiring a fully functional “office extension” in his home, or for very small branch offices or warehouses requiring only one or two phones but needing full access to the main office’s phone system’s voice and message capabilities.

The main office and the remote site require a high-speed broadband connection that meets the minimum technical requirements necessary to achieve acceptable quality of service (QoS). The ESI host phone system — IVX® X-Class, IVX E-Class, or IP E-Class, located in the main office — must be equipped with the appropriate card to support the voice data compression required by the Remote Phones. (See “ESI’s IP-capable phone systems,” page 2.)

Since the Remote Phone functions as an extension of the IP phone system, “local” dial tone and 911 emergency access for the Remote Phone will be made from the location of the IP phone system, not the location of the Remote Phone.

Quality of Service (QoS) is an important consideration for business phone systems. In Remote Phone applications, the carrier, broadband access and other external factors will greatly affect QoS. Many of these QoS issues can be minimized by following ESI’s installation guidelines regarding WAN provider selection, bandwidth, and latency.

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ESI's IP-capable phone systems

Four ESI phone systems can be configured to support Remote IP Feature Phones by adding optional IP expansion cards.

IP E-Class: The All-In-One IP PBX

The **IP E-Class** systems — **IP 200e** and **IP 40e** — are advanced IP network-based business telephone systems tailored for small to midsize businesses. The IP 200e and IP 40e reside on the office's LAN routing internal voice and message traffic as data, using true one-wire connectivity and packetized voice to all feature phone extensions. Although both models offer similar functionality, the IP 200e contains 192 call processing ports and 10 card slots, while the IP 40e contains 70 call processing ports and two card slots. Remote Phone support on IP E-Class requires installation of the **Remote Network Card**, which plugs onto the system's built-in Local Network Card.

IVX X- Class and IVX E-Class: The All-In-One Digital Phone Systems

ESI's **IVX** digital phone systems, of which IVX X-Class is the top of the line, provide high-quality traditional telephony capability within the office. An IVX X-Class or IVX E-Class system also offers the unique advantage of providing IP capability, with the installation of an optional **IVX VoIP Card (IVC)** in one of the system's available card slots.

Remote network channels

Adding the appropriate card to each compatible ESI system gives it the software and voice compression necessary to support Remote Phones. Depending on the card, the system may support three, 12, or 24[1] **remote network channels**.

Each remote network channel handles one active bi-directional conversation, compressing voice data from 64 kilobits per second (Kbps) to 8 Kbps in each direction using G.729 compression algorithms. Separate, dedicated data channels carry station data between the host system and Remote Phones providing display updates and lamp appearances. The compressed channels are "pooled" channels shared on a first-come-first-served basis. Therefore, more Remote Phones can be deployed than there are channels on IP E-Class (IVX X-Class and IVX E-Class are limited to 12 IP Phones each, local and/or Remote — see the **note** under "IVX VoIP Card," next column). A system programming function can be used to limit the number of allowable channels, if needed to manage bandwidth consumption.

[1] The maximum 24 network channels are available for both Remote Phone and Esi-Link (multi-site) use, but no IVX system supports more than 12 Remote Phones.

IVX VoIP Card

The **IVX VoIP Card (IVC)** provides connectivity from the IVX X-Class or IVX E-Class phone system to the LAN. Through an RJ45 connector, the IVC can support both 10 and 100 Base-T Ethernet allowing deployment of IP Feature Phones locally across the LAN. It also supports up to 12 local IP Feature Phone extensions or up to 12 Remote Phone extensions, as well as Esi-Link.

> **Note:** Since an IVX E-Class system's LNC is installed in a standard card slot, it must conform to the extensions normally assigned to that slot. Therefore, the maximum number of extension numbers that can be assigned as local or remote IP Phones is 12. However, IP E-Class supports up to 96 IP Feature Phone extensions (IP 200e; up to 28 on IP 40e) in any combination, local IP and Remote IP.

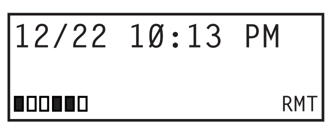
Remote Phone/Esi-Link

For each IP-compatible ESI system, the appropriate add-on card provides the functionality to support both Remote Phones and **Esi-Link** multi-site interconnectivity. With Esi-Link, multiple systems can be connected, sharing call data and voice traffic over the WAN. Remote Phones are designed to serve single users scattered at different locations away from the office, while Esi-Link ties together multiple offices, each with its own IP E-Class or IVX E-Class system. If both applications are used, the remote network channels will be shared for conversations between systems or to Remote Phones. For more information on Esi-Link, see the *Esi-Link Product Overview* (ESI item #0450-0214).

Remote Phone operation

The Remote Phone functions identically to the in-house versions with these exceptions:

- The bottom line of the IP Remote Phone's display shows real-time channel usage rather than CO line usage, keeping the user aware of channel availability before he/she initiates a call.



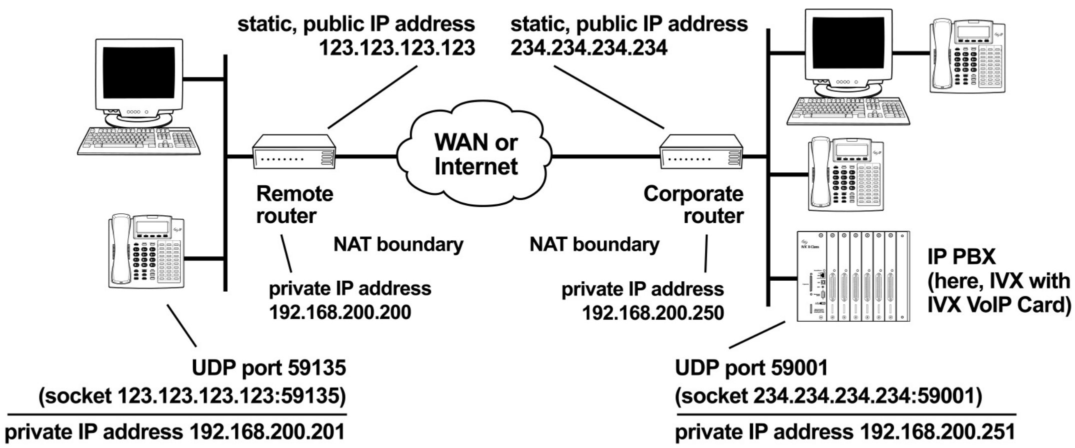
- When a user dials **9** to make an outside call, the channel-availability display will switch to the standard line-availability display showing CO line usage for the system.
- If a call is attempted when no remote channels are available, the display will prompt the user to queue for an available channel or they can simply choose to try the call later.
- Remote Phones will not receive pages originating from other stations.

Installation considerations

The end-to-end data pipeline

The elements of a Remote Phone installation are: the LAN at the remote site, the connectivity from the remote site to the carrier, the connectivity to the main office, and finally the LAN within the main office. Quality of service (QoS) will be no better than that allowed by the "weakest link" of these elements of the pipeline. Pre-installation evaluation should, therefore begin with an end-to-end assessment of all components.

Current industry WAN IP capabilities do not provide for automatic priority of voice traffic over data traffic. Therefore, a burst of data activity on a small data pipe — while insignificant for normal data applications — is likely to affect voice quality. Therefore allow enough bandwidth for both data and voice to coexist. ESI recommends that at least 44 Kbps of bandwidth be allocated for each channel supported at a site – above and beyond the peak data requirement.



Carriers ("the cloud")

Often, the Remote Phone application will be added to a remote site that has an existing data carrier connection, and the presumption is likely to be that the current carrier will be adequate for voice data — which might or might not be correct. Again, allow 44 Kps per channel above peak data requirements. Following is an overview of these carrier types:

Circuit-switched dedicated WAN

The ideal solution to the bandwidth and contention problem is a traditional, dedicated switched-circuit connection — such as T-1 or frame relay — with routers for WAN connectivity. While not cost-effective for a single Remote IP Feature Phone application, this end-to-end connection might be practical for locations with several Remote Phones, or in a situation where large volumes of data transfer justify the expense of a dedicated line.

Managed network

Providers with national backbones, such as Verizon and Sprint, offer managed networks. These enhanced networks often guarantee performance levels for voice and other real-time applications. Pricing levels and availability vary widely.

VPNs

Virtual Private Networks (VPN) use encryption techniques to provide a secure connection across the Internet. Encryption is not necessary for a Remote Phone application and, in fact, may increase latency. However, some VPNs guarantee optimum latency levels by reducing the number of hops through the Internet.

Public Internet

The quality of the connection over the Internet can be affected by many factors including: number of hops, different network backbones, and time-of-day traffic. Using the same Internet Service Provider (ISP) at both ends, or at least ISPs who use the same backbone, can minimize hops. For example, if the main office is using Verizon, which carries its inter-regional traffic over Level 3's fiber backbone, find a local ISP that also uses Level 3's backbone. By avoiding the latency caused by jumping from one backbone carrier to another, latency can be reduced.

Connectivity ("the last mile")

If the carrier used is a dedicated circuit or managed network, the carrier will provide the connectivity end-to-end. However if the Internet is used, then the last mile of connectivity must be provisioned in conjunction with an ISP. Following is an overview of some of these connection options:

DSL

The minimum available DSL bandwidth is usually 160 Kbps. Although this speed is the most widely available form of broadband access, the distance from the user's Central Office may limit bandwidth. When provisioning DSL, ensure that both downstream and upstream speeds are 160 Kbps or better.

Cable modem

Cable modems offer most of the same advantages as DSL, including big inbound and outbound pipes at a low cost. Unlike DSL, more users on the same cable link will reduce bandwidth. Normally, however, cable modems are unavailable to business offices. As they provide only an Ethernet connection and no security or private addressing, cable modems almost always require some other device to sit between them and the network.

ISDN

ISDN-128 (the Basic Rate Interface) is available almost anywhere through the telco's ISDN service. If the site requires more than one channel, 128K of bandwidth may not be sufficient. The next step up is to provision a primary rate interface (PRI) span, which may be cost-prohibitive in some locations.

Dial-up ISDN linked through an ISP might be suitable if the ISP/carrier does not auto-disconnect the line for inactivity. Note that some NAT[1] routers will continuously reconnect which may negate this shortcoming. Contact ESI Tech Support before using dial-up ISDN. Per-minute tariffs may also apply on dial-up ISDN.

Wireless broadband

Two types of wireless broadband connections are available: symmetric-return and no-return (or low-return). While both have high-speed downstream pipes, symmetric-return systems provide upstream bandwidth of 128 Kbps or greater. No-return systems, however, rely on dial-up to provide the upstream path and should not be used.

Dial-up

There are simply too many limitations with 56K or slower dial up connections, making them unsuitable for Remote Phone applications. ESI Tech Support therefore cannot provide assistance for dial-up connections.

LAN requirements

Remote location

Even if the site has only one Remote Phone, it will require a LAN for connectivity. The Remote Phone connects through an RJ-45 connector on the bottom of the phone to either a 10 or 100 Base-T Ethernet connection to the router for connectivity to the WAN or Internet.

If the installation consists of only one Remote Phone, then the gateway could be the device supplied by the provider (*i.e.*, the cable modem or DSL modem) if the ISP provides a public IP address. If one or more additional devices are to be used, such as PCs or additional Remote Phones, then additional public IP addresses might be obtainable from the ISP. More likely, a NAT router will be required to convert from a single static, public IP address (assigned to the site by the ISP), to the required private IP addresses used by each device behind the router.

If the ISP does not supply a public IP address and/or uses PPPoE, a NAT router will be required to negotiate addressing with the ISP. ESI recommends using a Linksys® Etherfast™ Cable/DSL Router (or similar NAT-capable router) for these applications.

Since Remote Phones use UDP/IP protocol rather than TCP/IP protocol, the NAT router software must be able to direct data through an installer definable UDP port to the private IP address that it has assigned to a particular Remote Phone.

[1] *Network address translation*.

Main office

The corporate LAN has the IP phone system on one node and might have multiple IP Feature Phones or data devices on the others. It must have a router with wide-area capability to connect to the WAN or Internet.

The minimum requirements for the main office LAN are basically the same as for the remote site: the wide-area connection must be an always-on connection of at least 128 Kbps and the router must have a static public address on its WAN interface. If the corporate LAN is using Network Address Translation (NAT), then the NAT router must be able to forward specific, user-defined UDP ports to an individual IP address.

It must be determined that both the remote site's gateway device and the corporate-side gateway device (*i.e.*, the routers) are capable of forwarding specific UDP ports to specific IP addresses.

The installer will likely need to contact the user's ISP to gather the necessary WAN information and make arrangements with the ISP's tech support staff to configure the router. To establish telnet access into the remote site router, the installer must know the correct telnet commands and capabilities. A visit to the remote site, or (if available) Internet access allowing a telnet into the remote router, might be necessary. For a remote visit, a variety of cables and connectors should be prepared in order to connect to the router's console. The installer should be familiar with the configuration of many different models of routers.

Remote IP Feature Phone installations

The on-line **Network Qualification Checklist** outlines the information to be gathered to determine whether a site is suitable for installation of the Remote IP Feature Phone (as well as Esi-Link). The Checklist's location on the **password-protected** ESI Resellers' Web site is:

http://www.esiresellers.com/netcheck

Programming

The Remote Phone and the main office system will be located on different LANs; therefore the programmer must program the Remote Phone's IP addresses into the host system and vice versa.

In system programming, the installer designates individual extensions as either "local" or "remote." The "remote" extensions will require the additional addressing information (gathered on the *Network Qualification Checklist for Remote IP Feature Phone Installations*).

Once the programming is complete, the installer attaches each Remote Phone to the cabinet's Ethernet LAN or to the PC-based *Esi-Address* and assigns its extension number. The system will automatically download into the phone's non-volatile memory its required addressing information. The Remote Phone is then ready for installation at the remote location.

The remote site's gateway device must be programmed to open the UDP port to the phone and forward it to the phone's IP address. When first connected to the remote site's LAN, the phone will automatically contact the main office system, log itself in, and be available for use.

Relocating Remote Phone

Although the Remote Phones can be moved within the remote site's LAN, they cannot be moved to a different remote site without reprogramming. You must first **either** (a) use *Esi-Address* to download to Remote Phones the new location's addressing information **or** (b) return the Remote Phones to the home site and reprogram them through the IP system.

Network security considerations

Most networks contain security provisions. In order for the Remote Phone to communicate with the host system, the network administrator must create voice communication “tunnels” through that security. This can be done by designating a UDP port through which all inbound traffic must flow, and then assigning or forwarding that port to a specific device or IP address.

Both the host system and the Remote Phone must have a UDP port exclusively assigned to each of them. Each also contains a specific IP address and the public IP address of the remote gateway device. Devices behind the NAT router use these private IP addresses, while devices on the public side of the NAT router use the socket (created by the outbound session).

As an example of communication between the devices, when the host system sends voice data to a specific Remote Phone’s socket, the remote router receives it and only passes the communication through security to the designated Remote Phone’s port.

The router at the remote site must be capable of opening a UDP port tunnel through its security and forwarding the port to a particular IP address. ESI recommends that the Remote Phone be placed outside the firewall to minimize the latency such devices may introduce. The Remote IP Feature Phone is not a security risk to the remote network’s data devices, so it does not need firewall protection.

User expectations

If the Remote Phone application requires near-toll-grade voice quality, a managed network or private intranet is strongly recommended. When using the Internet, varying degrees of latency and lost packets will exist making consistent voice grade QoS more difficult to achieve. It is critical that the prospective user be aware of the potential for variations in QoS. With proper network management, these adverse quality issues can be minimized.

There will be situations where the user must adjust his expectations to the realities of today’s technology. The advantages of the Remote Phone coupled with the intended usage will need to be balanced by the user against the probable QoS deliverable by the available connection.

As an example, a Remote Phone–to-CO line call (likely with a customer) will deliver QoS directly associated with that Remote Phone’s connection, whereas a Remote Phone intercom call to another Remote Phone (likely to be internal) will, in fact, double the likelihood for a noticeable reduction in voice quality.

Another variable which may affect Remote Phone audio quality is the performance of the CO lines accessed by Remote Phone users for outside calls. Due to network delays within a packetized phone system, the CO line connection, whether digital or analog, has the potential to cause echo. To compensate for this, the IP Series and IVX 128 Plus phone systems each include an ITU-standard G.168 echo canceller; it will adapt automatically to various CO line conditions, eliminating or significantly reducing echo. However, in some areas, certain CO line conditions may cause degraded performance of the echo canceller (and of the packetized phone system’s audio performance in general), even if such line conditions have not previously caused problems of this nature with traditional phone systems.

VIP compatibility

ESI’s *VIP* unified communications software can be used with the Remote Phone. For more information, refer to the *VIP and NSP Advanced Options Guide* (ESI # 0450-0667).

Remote Phone network/voice-quality Issues

In general, to achieve optimum voice QoS network performance should be within the following targets: latency of 150ms delay or less in each direction, packet loss of 5% or less, jitter less than 10ms, and no more than 10 hops.

Bandwidth

Currently, minimal QoS capabilities exist for IP WANs and the Internet. As a result, “bursty” data traffic and the type of connectivity chosen will greatly affect the quality of a conversation when using a Remote IP Feature Phone. The greater the amount of bandwidth provided, the lesser the opportunity or duration of voice interruption from bursts of data. Hence, a DSL connection at 600 Kbps will provide less voice interference than a 128 Kps ISDN connection.

Latency

Latency or delay in speech is a necessary component of IP telephony. Generally, latency of 150 milliseconds (ms) or below in each direction yields good voice quality, 150 ms to 250 ms is marginal, and over 250 ms is considered poor. Extreme latency will result in QoS similar to a very poor satellite conversation and will be unacceptable.

Packetizing, compression and system processing add a necessary amount of latency to IP calls. In addition, the WAN and Internet will add latency. Because the Internet is dynamic, the number of hops made between sites will contribute a wide variance of latency. Using the same ISP at both ends, combined with the best possible connection speed, will minimize the delay. ESI recommends working closely with the ISP in designing the WAN routing paths between the Remote Phone site and the host system location to minimize "hops" and ensure as fast and uninterrupted flow of data as possible.

Multiple Remote Phones at a single location

A remote office site with multiple Remote Phones must allocate enough bandwidth for each potential simultaneous conversation. All conversations, including intercom calls and voice-mail retrieval, etc., use bandwidth back to the main office. To allow for contingencies, determine data requirements and then plan for an additional 44 Kbps per phone.

If multiple Remote Phones are being considered for a site, ESI's *Esi-Link* multi-site application may be a preferable solution, instead of Remote Phones. *Esi-Link* technology allows each location to have its own ESI IP phone system, providing local loop and access to 911 emergency service, while using VoIP to "link" each office for IP-based inter-office telephony.

CO line quality

Another variable which may affect Remote Phone audio quality is the performance of the CO lines accessed by Remote Phone users for outside calls. Due to network delays within a packetized phone system, the CO line connection, whether digital or analog, has the potential to cause echo. To compensate for this, the E-Class phone systems include an ITU-standard G.168 echo canceller; it will adapt automatically to various CO line conditions, eliminating or significantly reducing echo. However, certain CO line conditions in some areas may cause degraded performance of the echo canceller (and of the packetized phone system's audio performance in general), even if such line conditions haven't previously caused problems with traditional phone systems.

911 considerations

Internal and outside dial tone on a Remote Phone is provided by the main office system. Therefore, dialing 911 at a Remote Phone will guide emergency help to the main office where the CO lines are terminated, not to the remote site where the Remote Phone is located. For this reason, each remote site must have a local telephone line available for 911 emergencies.

> **Note:** For more complete details on the Remote IP Feature Phone, consult the *Remote IP Feature Phone Installation Manual* (ESI document #0450-0450). ESI-trained Resellers may download this document and any others mentioned herein from *www.esiresellers.com* (password required).

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-certified company — assuring that quality is fundamental.