Harnessing the power of SIP with the 5E-XC™ High Capacity Switch

Realize the vision for next-generation networking while reducing costs

A solution for implementing new revenue-generating services quickly and affordably. Leverage existing 5ESS® switch platforms with SIP to remove network complexity.

This white paper demonstrates how service providers can capitalize on the potential of Session Initiated Protocol (SIP) networking with Lucent Technologies 5E-XC™ switch. A brief overview of key SIP capabilities illustrates how service providers can apply SIP to enable next-generation services and converge voice onto an IP core network. An overview of the features and benefits of Lucent Technologies 5E-XC™ SIP solutions – for converged services, IP trunking, and advanced VoIP applications – describes how they provide the basis for profitable new service offerings and cost savings in operations and management using the 5E-XC™ switch.

Lucent Technologies
Bell Labs Innovations



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Abstract

A number of important changes are creating significant challenges for service providers. Increasingly, these challenges need to be addressed via a combination of new revenue-generating services and cost reductions. This paper examines the role that the Session Initiation Protocol (SIP) can play in allowing service providers to reach their goals. This paper further describes how the 5E-XC™ High Capacity Switch makes use of those SIP capabilities as building blocks to support new services and to enable service providers to reduce costs. An overview of the key capabilities of SIP is also provided. Also, this paper illustrates how service providers can cost-effectively leverage their existing 5ESS® switches as their networks evolve towards IP.

Introduction – meeting today's challenges

The telecom industry is undergoing significant changes that are creating major challenges for service providers. These changes are primarily the result of:

- Increased competition due to government deregulation initiatives
- Increased popularity of wireless service, which is displacing fixed phone line use
- Increased popularity of broadband services (e.g. DSL and cable) to access the Internet
- An economic downturn which has led to businesses either folding or downsizing, reducing the number of active lines

The changes are, overall, causing a decline in the revenue the service providers derive from traditional phone services. To counter this decline, service providers are looking for ways to:

- Deploy new revenue-generating services
- Retain, win back or attract new customers
- Minimize expenses by leveraging existing infrastructure
- Converge voice and data networks

As a means to achieve a number of these goals, service providers are increasingly looking at packet based technologies. In this paper, we explore how the 5E-XC™ switch takes advantage of one of these technologies, SIP, to enable service providers to achieve these objectives. The 5E-XC™ switch enables service providers to evolve their networks to IP and reduce life cycle costs. It enables them to offer high demand services to all Internet users while minimizing the cost of delivering both traditional and IP based services.

The 5E-XC™ switch enables service providers to evolve their networks to IP and lower life cycle costs.

The 5E-XC™ switch can enable service providers to take advantage of SIP to provide new revenue generating and cost saving solutions without a costly forklift replacement of their installed base.

Besides SIP, it should be noted that a number of other enhancements are being introduced to the 5E-XC™ switch. New 5E-XC™ switch elements and software increase capacity by a factor of almost 300 percent, while at the same time increasing density to drive down equipment footprints and energy consumption. A new Optical Interface Unit (OIU) allows TDM and IP trunking to coexist on the same shelf. The 5E-XC™ SIP-based functionality described in this paper makes extensive re-use of existing 5ESS® infrastructure, including OAM&P capabilities and many of the hardware components already in use for TDM-based networking. The incremental 5E-XC™ switch additions can result in significant cost savings (both in terms of capital as well as operational expenses) as networks migrate from circuit to packet switching and services. And, all this on the switch that continues to deliver industry-leading 99.9999 reliability.

Accelerating business with 5E-XC™ SIP Solutions

In this paper, we will demonstrate how the 5E-XC™ switch can enable service providers to utilize SIP to provide new revenue generating and cost saving solutions without a costly forklift replacement of their installed base. Initial 5E-XC™ SIP-based solutions include the following:

- **Web Portal Solution**: allows service providers to extend new revenue generating services to users of traditional endpoints (POTS or ISDN). These services combine the use of traditional endpoints with data applications provided by servers on the Internet or on an Intranet. The initial focus of this solution is to provide Web-based call management services to traditional endpoint subscribers, enabling new services such as click-to-dial and many others.
- **IP Trunking Solution**: gives service providers the ability to route voice calls across an IP network, and to be able to provide the required interworking between PSTN signaling and SIP required to allow PSTN callers to access advanced services in the IP portion of the network.
- **Advanced VoIP Solution**: offers efficient network utilization for calls (e.g. Pre-paid or Call Centers) that require an application server (e.g. service node). Unlike the Web Portal Solution above, for which the voice can generally remain in the TDM realm, this solution involves the use of voice transported over an IP network.

The next section of this paper provides a brief overview of key SIP capabilities to explain why this technology offers such tremendous potential as a change agent. It will become apparent why Lucent Technologies' 5E-XC™ product development has focused on SIP. It is the most sustainable' path to the future built on today's infrastructure.

5E-XC™ Switch Solution	Service Provider Benefits
Web Portal Solution	New revenues Customer retention Minimal cost
IP Trunking Solution	Reduced costs
Advanced VoIP Solution	New revenues Customer retention Reduced costs

Enhancing network value with the building blocks of SIP

Lucent believes that SIP can be a significant tool for providing next-generation services and a critical enabler for circuit-to-packet convergence. SIP supports a number of capabilities that add versatility to the protocol, allowing for a wide range of uses in the network, such as an interface to application services and trunking of PSTN calls, etc., to ultimately simplify operations. Key SIP capabilities include:

- **Establishing and controlling IP media streams –** SIP enables networks to establish and terminate multimedia sessions (both two-party and conferences) over IP networks. The use of flat IP networks offer potential cost savings. Additionally, the support of multimedia sessions will enable many new services in the future – enhancing the revenue potential per subscriber.
- **Modifying existing media streams –** SIP allows networks to modify existing media streams during a call. An important aspect of this capability is the ability to "redirect" or "pivot" an existing media stream from one destination to another. This capability enables optimization of the media stream paths after the media server is no longer needed.
- **Proxy routing –** SIP supports the use of network elements called proxy servers, which can route SIP requests from one SIP user agent to another. This allows user agents on networks to send a request without knowing the exact location of the user agent to which it is sending the request. When SIP is used for IP trunking applications, proxy servers can reduce the amount of routing information that needs to be provisioned on each of the switches serving as PSTN gateways.
- **Redirection –** SIP supports the ability to have the originator of a SIP request be informed of the current location of the user agent to which the SIP request is intended, so that the originator itself can then send the SIP request directly to the intended user agent. The redirection capability can reduce the amount of routing information that each switch needs to maintain.
- **Ability to carry Multipurpose Internet Mail Extensions (MIME) content –** SIP signaling can carry just about any type of information within its messages, as long as the format of that information comes in the form of a valid MIME. Many future applications will undoubtedly make use of this capability.
- **Interworking with PSTN signaling –** SIP standards have been enhanced to better support interworking with the PSTN, including the ability to encapsulate ISUP messages within SIP, along with the ability to support the mapping of ISUP parameters to/from SIP headers. Networks can easily support the tandeming of PSTN calls across a SIP network, and IP endpoints can originate or receive calls to/from PSTN subscribers.

SIP can be a significant tool for providing next generation services and a critical enabler for IP convergence.

The 5E-XC™ Web Portal Solution drives $400M in new revenue from the Centrex, Enterprise, DSL, and Dial-up markets in an example network.

- **Event notification –** SIP supports a framework by which a SIP entity can request another SIP entity to send a notification when particular events have occurred. Extensions (called "event packages") can be defined, on top of this framework, to specify the specific events that can be subscribed to. The ability of a node in the network to notify another node of the occurrence of certain events will make it easier to enable new services in the future.
- **Call Referral –** SIP supports a mechanism that allows a SIP party to request another SIP party to establish a call to a third party. This mechanism supports call transfers by SIP endpoints. It also allows for application servers to drop themselves out of a call after they complete performing their service.

Creating new opportunities with 5E-XC™ SIP Solutions

As noted earlier, Lucent Technologies' current 5E-XC™ SIP Solutions support Web Portal Solutions, IP Trunking Solutions, and Advanced VoIP Solutions. In order to ensure that these would mesh with service providers' migration strategies and integrate with their operations, Lucent developed them with the following guiding principles in mind:

- To allow service providers to leverage existing investments in 5ESS® equipment and features
- To extend new revenue generating services to traditional CPE (POTS, ISDN), thereby avoiding unnecessary near term investments in media gateways and call controllers as well as leveraging existing operations infrastructure (network, staff and procedures)
- To apply the same processes to ensure high reliability of the SIP-enabled capabilities as have been applied to all 5ESS® switches' TDM-based features in the past
- To make SIP interfaces on the 5E-XC™ switches open by utilizing industry standards (IETF, ITU, ANSI) to guide SIP protocol implementation

By 2007, Probe Research expects between 15% and 20% of Internet users to also subscribe to enhanced calling or messaging services, representing a worldwide (consumer) market of $14.8B. Probe also estimates the worldwide enterprise market for enhanced services at $4.9B.

Web simplicity for all subscribers

The primary benefit of the 5E-XC™ Web Portal Solution is to supplement existing, traditional, features on the 5ESS® switches with services provided by application servers on an Intranet or the Internet.

Application overview

The initial focus of the Web Portal Solution will enable traditional subscribers (i.e., subscribers that make use of traditional phones such as analog or ISDN handsets) to be able to access the same type of Web-based call management services that are already available to users of IP endpoints. For this solution, the 5E-XC™ switch supports a SIP-based interface to an application server with Web portal capabilities, such as the Enhanced Business Services (EBS) application server. Note, however, that this interface is used for signaling purposes only. The transport of voice between the 5E-XC™ switch and the Web Portal App Server is not required. This configuration is shown in *Figure 1*.

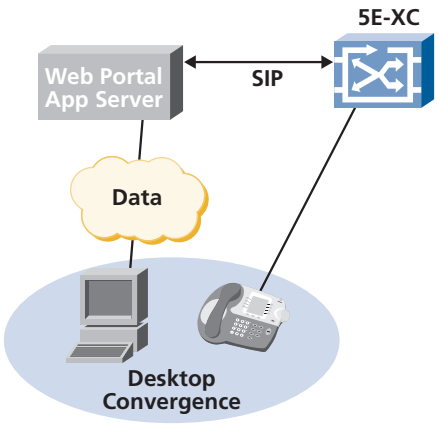


Figure 1 5E-XC™ switch interfaces via SIP to a Web portal application server to provide desktop convergence of telephony and the personal computer.

The applications that support new, enhanced services to subscribers via a Web portal may be divided into three groups: Internet Call Management, Unified Communications, and Personal Information Manager. These are described below.

Internet Call Management applications

These applications support a variety of services or features that simplify end user communications. They all make use of a Web portal principally to initiate call related requests, such as a request to establish a call, place a call on hold, or to add a party to a conference. In addition, subscribers also have the ability to use the Web interface to modify feature-related information on the 5E-XC™ switch, such as Call Forwarding numbers or Call Waiting activation/deactivation.

Examples of Web-based services supported as part of Internet Call Management include:

- **Click To Dial –** initiating a call via the Web
- **Click to Transfer –** transferring a call via the Web
- **Click to Add/Conference –** adding parties to a conference via the Web
- **AnyDial –** dialing any number found on a Web page
- **Speed Calling –** maintaining an easily accessible list of numbers that can be used by the subscriber to place calls via the Web

To support Internet Call Management types of services, the 5E-XC™ switch is notified by the Web portal via the SIP interface of the requests being made by the subscriber. The 5E-XC™ switch will then act to fulfill the Web-initiated requests.

The 5E-XC™ Web Portal Solution leverages three key SIP capabilities:

1. *Redirection*
2. *Ability to carry MIME content*
3. *Event notification*

Unified Communication applications

These applications support a variety of services or features, which ensure realtime communications, regardless of subscriber location or availability. They all enable subscribers to finely control how incoming calls to their 5E-XC™ switch's line are to be handled. The subscriber can either provide specific instructions pertaining to the handling of a call, such as "forward the call to this number", or it can allow the Web portal application server to decide how to handle a call based on information the system may have available, such as presence information.

One important example of the Unified Communication type of service is:

Find-me/Follow-me – this allows the subscriber to specify a set of numbers (such as work phone, home phone, cellular number) which incoming callers can be routed (usually in a sequential manner). Different call-handling treatments may be specified based on caller ID, day-of-week, and other factors.

For example, a subscriber may specify that certain calls, such as calls from important customers, be forwarded to his/her boss whenever the subscriber is not available.

For this type of service, the 5E-XC™ switch issues a notification to the Web portal application server (via the SIP interface) upon receiving an incoming call for a Web portal subscriber. It then waits for further instructions from the application server as to how to proceed with the call. The application server will be expected, based on rules indicated by the subscriber, to provide information to the 5E-XC™ switch regarding the further handling of the call.

Personal Information Manager applications

These applications support a variety of services or features, which enhance subscriber productivity. They all enable the user to gain access to needed information via a single web portal.

Following are examples of this type of service provider services or features:

- **Call Logging/Call Notes** – allows the subscriber to view information about both incoming and outgoing calls (including missed calls)
- **Presence** – allows the subscriber to view information about the availability of others
- **Message Waiting** – provides the subscriber with a message-waiting indication via the Web
- **Instant Messaging**
- **Group Calendars**
- **Personal address book Wireless PDA access**

- **Unified messaging (Voice Mail, Email, Faxes, SMS)**

To support the first three Personal Information Manager services, the 5E-XC™ switch reports information, via the SIP interface, to the application server about the converged services subscriber, such as call-related information or changes in message waiting status.

Figure 2 shows an example of a Web-based user interface provided to 5E-XC™ switch subscribers. The Enhanced Business Services (EBS) application server provides the Web page shown.

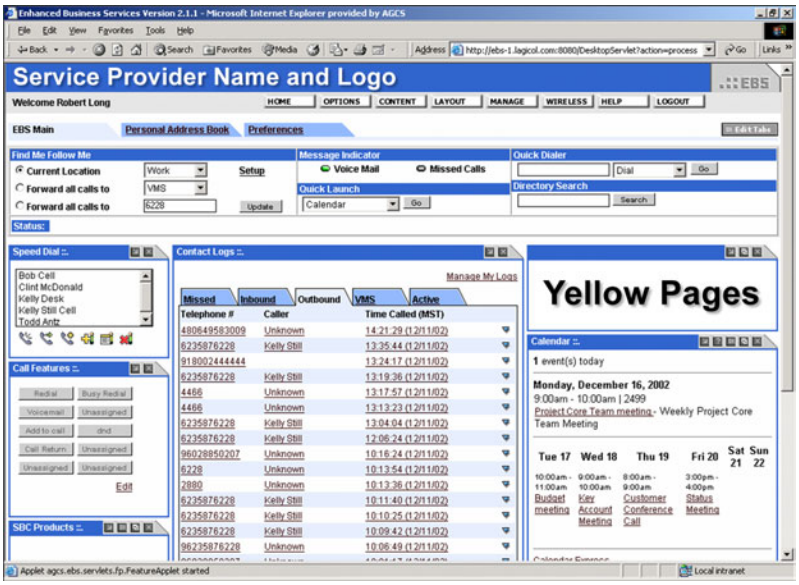


Figure 2 Example of Web-based user interface provided by EBS to converged services subscribers

Benefits of the 5E-XC™ Web Portal Solution

With the 5E-XC™ Web Portal Solution, service providers can look forward to obtaining the following benefits:

- Easier access via the Web portal resulting in increased revenues from features already existing on the 5E-XC™ switch
- Extension of new revenue generating services to traditional endpoints thus,
  - – Avoiding cost of CPE change
  - – Avoiding cost of media gateway investment
- Increase of revenues from existing customers by selling new Web-based features (see *Figure 3)*
- New and attractive Web-based features making it easier to retain existing customers
- Leveraging embedded base (both switching equipment and operations) thus minimizing CAPEX and OPEX costs associated with new features
- Increased ability to attract new business customers who wish to make use of the Web-based call management capabilities without the expense of converting to IP phones

The 5E-XC™ Trunking Solutions offer the potential for significant service provider cost savings compared to overlay network solutions. Overlay networks not only double operating expenses in power, cooling and real estate costs on a per trunk basis, they also complicate existing service operations.

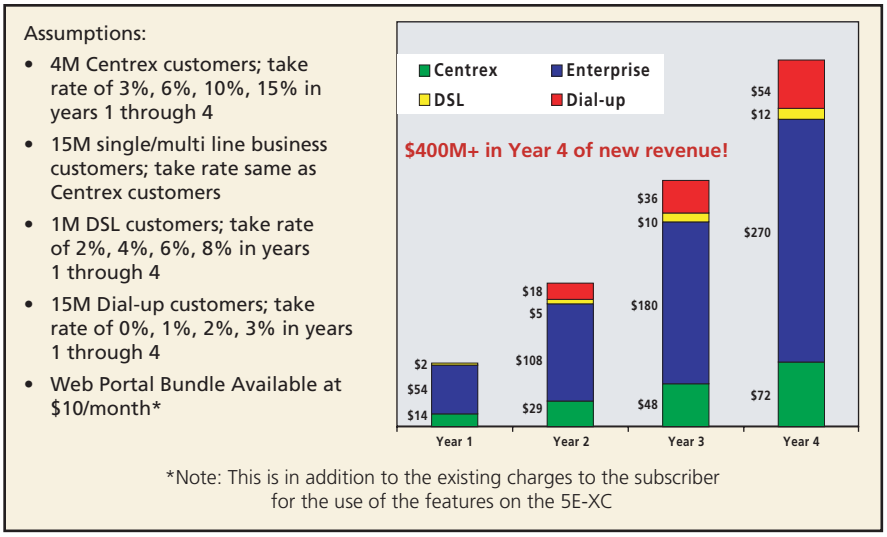


Figure 3 Web Portal Revenue Example

Flatten networks with IP Trunking Solution

The 5E-XC™ IP Trunking Solution allows service providers to make use of an IP network for routing voice calls across their network. The IP Trunking Solution also provides the capabilities necessary to facilitate the introduction of advanced VoIP applications. All of the functionality required for this solution is available with the 5E-XC™ switch.

Application overview

The IP trunking-related functions provided by the 5E-XC™ switch are:

- Establishing calls using SIP (with encapsulated ISUP)[1]
- Interworking SIP signaling with ISUP, PRI, and MF signaling
- Interconnecting the associated RTP media streams with TDM interfaces
- Interacting with SIP redirection servers and proxies in the SIP network for the routing of calls

A basic IP Trunking configuration involving two 5E-XC™ switches are shown in *Figure 4*.

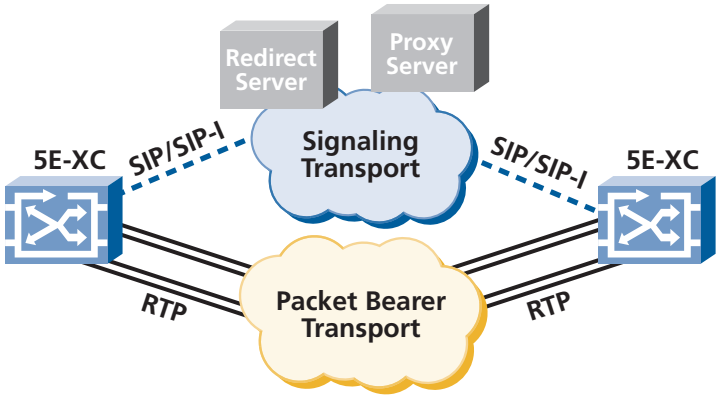


Figure 4 5E-XC™ switch support for IP trunking using SIP

1 The 5E-XC™ switch is making use of the ITU and ANSI specifications related to interworking between existing PSTN signaling and SIP. The ITU and ANSI specifications use the term *SIP-I* to refer to SIP signaling carrying encapsulated ISUP, instead of the term *SIP-T* used in some IETF RFCs.

Benefits of the 5E-XC™ IP Trunking Solution[2]

IP trunking can provide service providers with significant CAPEX savings because it eliminates the need for investment in media gateways and the softswitches to control trunking of voice calls across an IP network using conventional 5ESS® switches. Further, the resulting network will be simpler to manage than one involving the additional gateway and softswitch network elements.

IP trunking allows IP interface bandwidth on the 5E-XC™ switches to be used for any outgoing or incoming VoIP call on the switch, regardless of which neighboring switch is connected on the far end of the call. This contrasts with TDM, in which calls established to/from a neighboring switch might only use the TDM trunks configured to that switch. This characteristic of IP trunking can save an estimated 10- 30 percent in trunking resources.

SIP's redirection and proxy routing capabilities can enable routing within the IP network to be controlled from a relatively few elements in the network. The provisioning activities necessary to establish the routing information within the 5E-XC™ switches supporting IP trunking can be reduced significantly. Network changes that impact routing of calls across the IP network can generally be done without modifying the routing information in each of the 5E-XC™ switches in the network. This can result in considerable OPEX savings.

Cost efficiencies with Advanced VoIP Solution

The 5E-XC™ Advanced VoIP Solution allows service providers to quickly deploy powerful services using SIP on application servers located anywhere in their network. By building on top of IP trunking functionality, the 5E-XC™ switches can provide all of the functionality needed to bring application servers into a call via its built-in call routing capabilities. Alternatively, the 5E-XC™ switch can communicate via a service broker (a network entity responsible for determining whether any services need to be applied to a call), which can then bring in the appropriate application server to the call.

Expanded functionalities

In addition to the functions provided for the IP trunking solution, the 5E-XC™ switch makes available the following functions as part of its Advanced VoIP Solution:

- Allows application servers to modify existing VoIP media streams for redirect/pivot purposes
- Allows application servers to subscribe to DTMF events that may occur in the 5E-XC™ switch on a particular call, and to notify the application server
- Allows an application server to drop out of a call completely once it completes providing its service
- Supports "early media" calls. These are calls in which a connection needs to be established between a caller and a network-based application server before an "answer" is generated for the call. This capability is useful when, for example, a prompt-and-collect interaction between a caller and an application server needs to take place before billing is to start for the call.

2 A more detailed discussion of some of these benefits can be found in the companion white paper *Internet Protocol (IP) Trunking on Optical interface Unit (OIU)*.

The 5E-XC™ IP Trunking Solution leverages five key SIP capabilities:

1. *Establishing and controlling IP media streams*
2. *PSTN networking*
3. *Proxy routing*
4. *Redirection*
5. *Ability to carry MIME content*

- Allows application servers to initiate calls towards the 5E-XC™ switch to bring in additional participants to a call.

Figure 5 shows a basic configuration in which the 5E-XC™ switch communicates directly with an application server in the network.

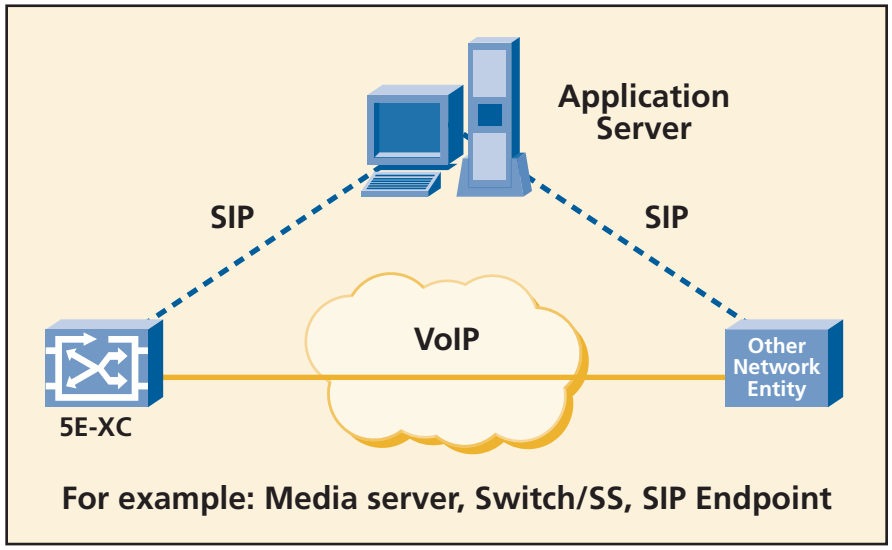


Figure 5 5E-XC™ SIP-based interface to a network-based application server

The 5E-XC™ Advanced VoIP Applications leverage five key SIP capabilities:

1. *Establishing and controlling IP media streams*
2. *Modification of existing media streams*
3. *Ability to carry MIME content*
4. *Event notification*
5. *Call referral*

The following examples illustrate how the capabilities provided as part of the Advanced VoIP Applications solution can help reduce costs associated with providing services by removing complexity from interworking. Keep in mind that other services provided in TDM networks, such as operator services and network-based call centers will benefit similarly from 5E-XC™ SIP Solutions.

Generic 5E-XC™ SIP VoIP call routing

In this example, the 5E-XC™ switch takes an incoming call coming in from a PRI interface or from the SS7 network, interworks the call with SIP, and routes that call (via the use of the dialed digits, or via instructions from an SCP) towards an application server in the SIP portion of the network.

Note that the 5E-XC™ switch could first route the call to an entity that performs a service broker type of function, which would then pass the call on to the application server. The application server could then perform service-specific actions (such as a prompt-and-collect interaction with the caller) to determine how to then proceed with the call. The end result could be a request from the application server to the 5E-XC™ switch to, for example, pivot the media stream to another switch in the network or to a SIP endpoint (e.g. an operator), or to notify the application server of any digits dialed by the caller.

Privacy Screening Service

The privacy screening service takes incoming calls with either an unavailable or blocked caller ID and routes them to a service node

Today, this happens according to the following sequence (see *Figure 6*):

1. Call arrives at the local switch serving the privacy screening subscriber
2. The local switch interacts with an SCP
3. Switch is instructed to route the call towards the privacy screening service node.
4. The service node collects the caller's name
5. The service node then establishes a call to the called party (i.e. the privacy screening subscriber)
6. The service node plays the name of the caller and prompts the subscriber for acceptance
7. The service node drops out of the call (via PRI 2B-channel transfer)

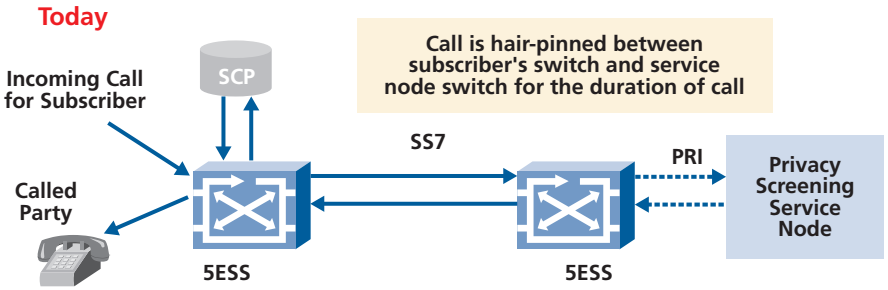
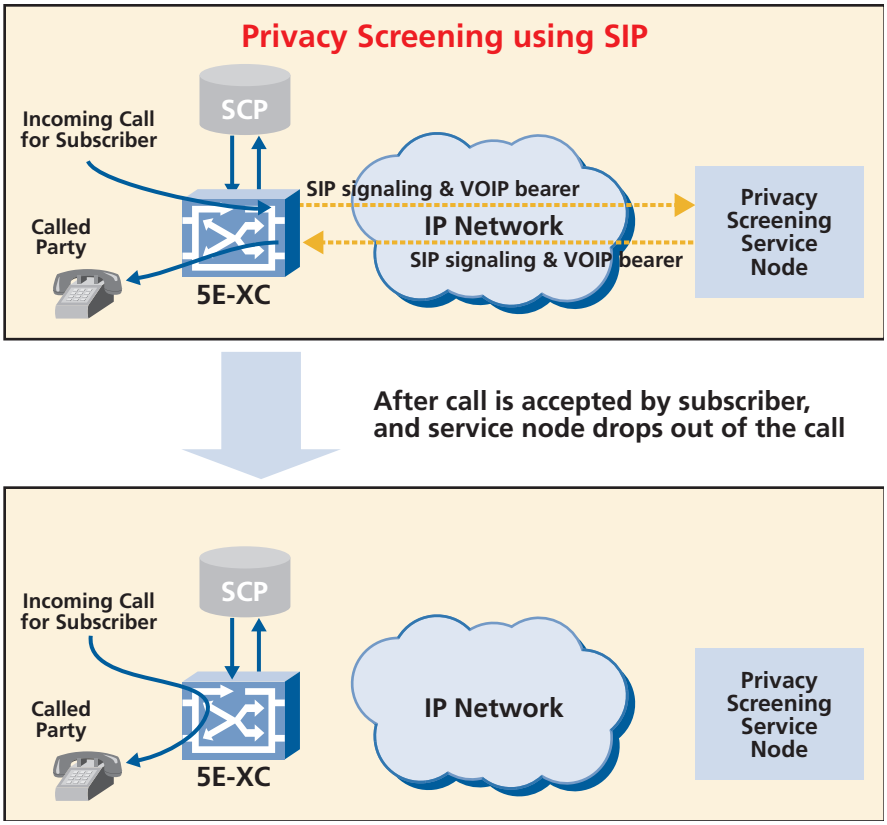


Figure 6 Example of typical Privacy Screening call today

The problem with this scenario is that for the entire call the call remains hair-pinned between the subscriber's switch and the service node switch, increasing the cost of providing the service.

With the 5E-XC™ switch, in contrast, this happens according to the following sequence (see *Figure 7*):

1. Call arrives at the local switch serving the privacy screening subscriber
2. The local switch interacts with an SCP
3. Switch is instructed to route the call towards the privacy screening service node via SIP over IP network
4. A VoIP connection is established between the subscriber's switch and the service node
5. The service node collects the caller's name
6. The service node then establishes a call via SIP to the called party (i.e. the privacy manager subscriber)
7. The service node plays the name of the caller and prompts the subscriber for acceptance
8. The service node drops itself out of the call (via SIP REFER method)



SIP-enabling privacy screening services can save up to 75% of trunk terminations.

Figure 7 Example of Privacy Screening service using SIP

The savings associated with trunk terminations, as compared to TDM, for SIP-enablement of the privacy screening service has been estimated at 75 percent, while overall savings has been estimated at 22 percent.

Pre-Paid Card Calling Service

Today a typical pre-paid card calling scenario proceeds as follows (as illustrated in *Figure 8*):

1. A pre-paid card calling user places a call to a service node that supports the pre-paid card service
2. The service node prompts the user for the number of the intended called party along with information about the pre-paid card being used
3. Service node establishes a call towards the called party. The service node remains in the path of the call for its entire duration.
4. The service node listens for DTMF tones that would indicate that the caller wants to originate another call

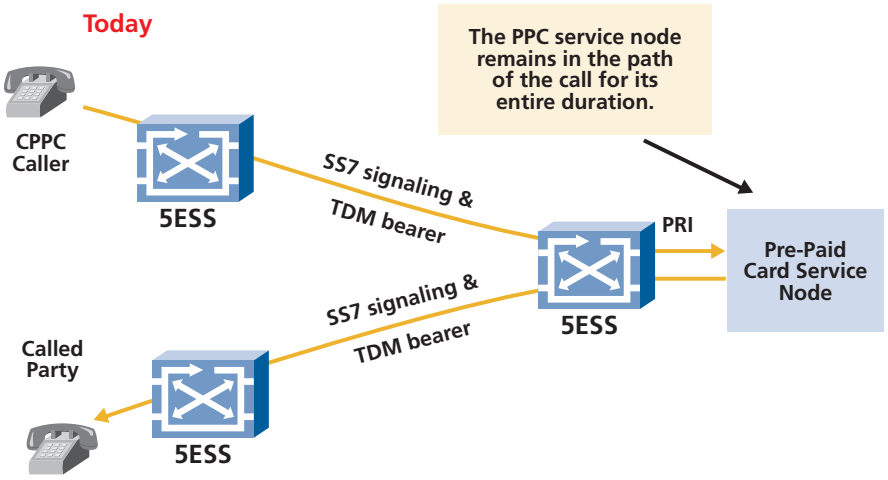


Figure 8 Example of typical Pre-Paid Card call today

The problem with this scenario is that it keeps the service node in the path for the entire duration of the call, resulting in inefficient use of network resources.

Compare this to a typical scenario in which SIP is used to provide the pre-paid card calling service (as illustrated in *Figure 9*):

1. A pre-paid card calling user initiates a call to an Application Server that supports the pre-paid card service
2. The call is routed to an application network in the SIP network. The bearer is connected to a media server controlled by the AS.
3. The media server prompts the user for the number of the intended called party (along with information about the pre-paid card being used). This information is passed on to the AS.
4. The AS establishes a call towards the called party. The AS uses SIP re-INVITE message to make the connection between the caller and called party more efficient.
5. The AS requests the originating gateway switch to report any DTMF tones it receives from the caller (to be able to detect re-origination attempts on the part of the caller). The bearer retains its efficient path for the duration of the call.

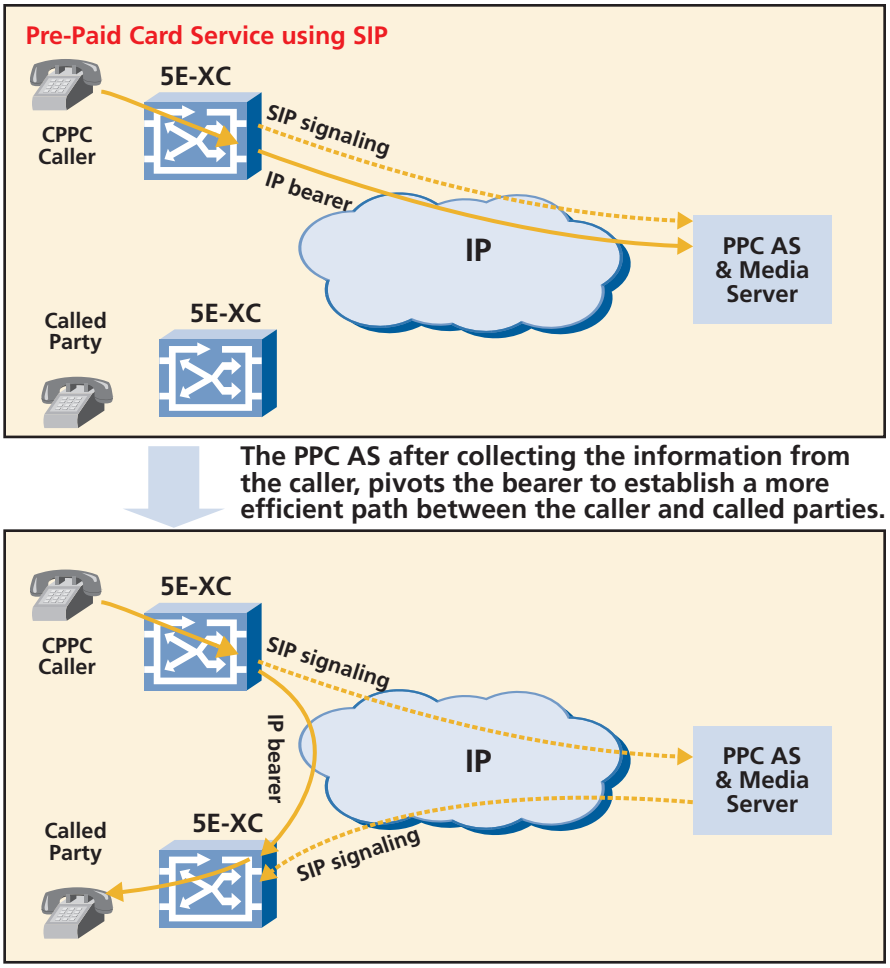


Figure 9 Example of Pre-paid Card service using SIP

Lucent estimates that SIP-enabled pre-paid card service can save service providers, as compared to TDM, up to 75 percent of the trunk termination costs, and about 40 percent of the overall costs.

Benefits of the 5E-XC™ Advanced VoIP Application Solution

The 5E-XC™ Advanced VoIP solution can:

- Reduce CAPEX and OPEX associated with more efficient network utilization (reduce or eliminate trunking hairpins)
- Reduce service node CAPEX by minimizing bearer termination duration
- Increase revenues by identifying new services that now have positive business cases

Conclusion

The 5E-XC™ SIP solutions can help make valuable new SIP-enabled services available to customers while keeping the associated impact to networks to a minimum through simplification and reuse of the 5ESS® switch and system operations infrastructure.

Lucent believes that SIP will continue to offer important advantages as a protocol for new and evolving networks. SIP provides an unequaled set of powerful capabilities that Lucent will apply as building blocks for new services and as a means to reduce the costs associated with existing services. A significant number of application server vendors are currently making plans to support services based on SIP, offering additional potential services that can be leveraged. In addition, the versatility of SIP that enables it to be applied in a wide variety of ways, such as between PSTN gateways, to IP endpoints, to IP PBXs, and to application servers, will make it easier to keep the number of protocols in networks to a minimum, thus helping to further reduce operational expenses.

Summary

The table below summarizes the 5E-XC™ SIP Solutions, together with lists of key SIP capabilities related to each solution and the benefits that can be derived from each solution.

5E-XC™ Switch Solution	Key SIP Capabilities	Service Provider Benefits
Web Portal Solution	Redirection Ability to carry MIME content Event notification	New revenues Customer retention Minimal cost
IP Trunking Solution	Establishing and controlling of IP media streams Proxy routing Redirection Ability to carry MIME content Interworking with PSTN signaling	Reduced costs
Advanced VoIP Solution	Establishing and controlling of IP media streams Modifying of existing media streams Ability to carry MIME content Event notification Call referral	New revenues Customer retention Reduced costs

Acronyms

AS	Application Server
CAPEX	Capital Expenses
CPE	Customer Premise Equipment
HTTP	Hypertext Transfer Protocol
IP	Internet Protocol
MIME	Multipurpose Internet Mail Extensions
OPEX	Operational Expenses
OS	Operating System
OIU	Optical Interface Unit
PBX	Private Branch Exchange
PPC	Pre-Paid Card
PSTN	Public Switched Telephone Network
RFC	Request For Comments
RTP	Real-time Transport Protocol
SCP	Service Control Point
SIP	Session Initiation Protocol
TDM	Time Division Multiplexing
VoIP	Voice Over Internet Protocol
WAP	Wireless Application Protocol

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