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Voice and Data Network of Convergence and the Application of Voice over IP

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Voice and Data Network Convergence and the Application of Voice over IP

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Abstract

This paper looks at emerging technologies for converging voice and data networks and telephony transport over a data network using Internet Protocols. Considered are the benefits and drivers for this convergence. The paper describes these new technologies, how they are being used, and their application to Sandia.

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Introduction

Technological innovations and industry developments in telecommunications have recently generated substantial interest in combining voice and data networks. Numerous articles, presentations, and company announcements [Cas][And][Ren][BCR] have addressed this topic. Manufacturers that have historically produced equipment for either the data or the telephone markets are attempting to broaden their product lines to serve both markets. This paper explores how these developments might apply to Sandia's enterprise operation.

There are several drivers for wanting to combine these networks. They are expected lower cost structure and increased system efficiencies, regulatory changes, the development and maturing of underlying technologies that facilitates service convergence, and the promise of new customer applications that would not otherwise be possible.

Cost/Efficiency --The demand for reduced costs and more leveraged spending is a major driver in considering converged telecommunication networks. Telephone and data networks require large capital outlays to build out cable, switching, and distribution equipment. In addition to the initial infrastructure costs, there are large costs in maintaining the networks and providing ongoing services. Telephone and data networks share many of the same structural elements. Both networks require inter-building cabling, intra-building cabling, switching equipment, reliable power systems, equipment maintenance, and customer services. The two networks require common structural elements, however, the implementations of these elements are unique to each network. This leads one to believe that combining the networks can reduce cost and complexity by removing redundant operations and by reducing the number of unique operations.

New Applications -- User needs for applications that combine voice and data services are emerging. One area of strong interest is in customer relationship management, particularly in call centers or help desks. The ability to aggregate and integrate all aspects of the customer-supplier interaction makes for better and more successful outcomes. Another area is telecommuting or remote access. It is desirable to be able to deliver identical communication services to the end user regardless of location. Voice and data integration allows for more user control over the call process and the ability to handle or screen calls in ways that are not otherwise possible.

Regulatory --National legislation and the move to deregulate the telecommunications industry are leading companies to look at different models for providing telecommunication services. The distinction between local phone, long distance and data service providers is disappearing as companies attempt to provide customers with all services from a single vendor. Traditionally governments and regulatory agencies have placed significantly different excise taxes and access charges on different telecommunication services. These differences have created imbalances in the cost structures for voice and data services with the result that companies have made implementation changes to take advantage of the cost differences. These regulatory factors have an influence on Sandia's public network access.

Technology --Technological change is making it possible to consider alternative methods for deploying telecommunication services. The development and growth of data networks, both Intra-net and Internet, has been to such an extent that the bandwidth requirements for data transport is exceeding that required for voice [Len][Min]. As the available data network bandwidth grows, it becomes possible to include voice traffic without substantially increasing data network bandwidth requirements. Further, digital signal processors have progressed to the stage where they have sufficient processing power at a low cost to allow effective digitization and coding of voice signals for transport over packet and cell networks. These developments are enablers, that allow one to implement converged networks.

The innovation exhibited by and deployment of new services on the Internet is spurring innovation in telephony equipment. Service providers are pushing for the replacement of proprietary telephony equipment with equipment built to open standards, common software tools, and programming interfaces.

Background

Telephone Installation

In the early 1990's, Sandia National Laboratories, SNL, installed a central office class telephone switch at its Albuquerque campus. This telephone system, with a Lucent Technologies 5ESS switch at its core, employs a typical telephone system architecture. A main, centrally located, switch fabric provides all the control and switching functions for the system. Telephones individually attach to the switch via twisted wire copper cables. Because of the large geographic dispersion of phones across SNL technical areas, the 5ESS is configured with optically connected remote modules to place switching equipment closer to telephone users. When a user makes a telephone call, the switch allocates bandwidth and dedicates that bandwidth to that user for the duration of the call. The switch completes all call functions and bandwidth management within its boundaries. Likewise the central switch processors route calls to or from other switches over interconnecting trunks. Sandia's telephone switch, like most telephone switches, is proprietary in design. While there are certain common, standardized interfaces, the switching fabric, software, circuit packs, and other parts of the switch are unique. Figure 1 below graphically illustrates the existing switch layout.

Commercial telephone companies have used the above described system architecture for many years and found that operation of such a system lends itself to certain maintenance and operations activities. SNL has adopted many of these commercial practices for providing telephone services. Day-to-day operation of the system clusters around the operation of the various components of the system -- switching, outside cable plant, inside cable plant,

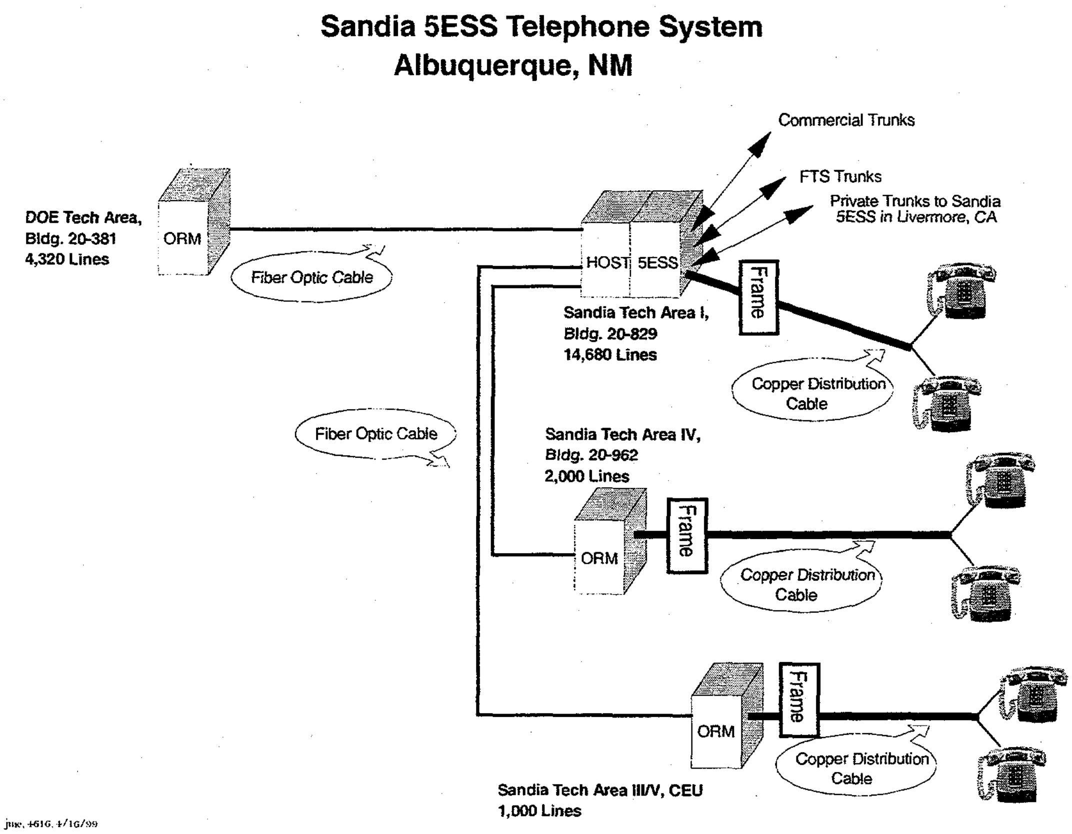

Figure 1 -- Telephone System Installation

move/add/and change activities, customer service requests, and trouble resolutions. Each operational area requires people with different skill sets to perform unique tasks that are necessary to make the system function.

Data Installation

Sandia maintains and operates several data networks that are grouped by security environment. The general network design within each security environment is similar to that of networks in other environments. The networks fall within the Sandia Open Environment, SOE, the Sandia Restricted Environment, SRE, and the Sandia Classified Environment, SCE.

Within each of these environments, Sandia maintains a corporate network. Respectively they are the Sandia Open Network, SON, the Sandia Restricted Network, SRN, and the Sandia Classified Network, SCN. Other departmental or programmatic networks may also exist within these security environments.

Sandia's data networks have evolved throughout the 1990's to support an interconnection of local area networks. The fabric interconnecting the LANs is a geographically distributed collection of switches and routers. There are dozens of autonomous devices that inter-operate with each to form the building blocks of

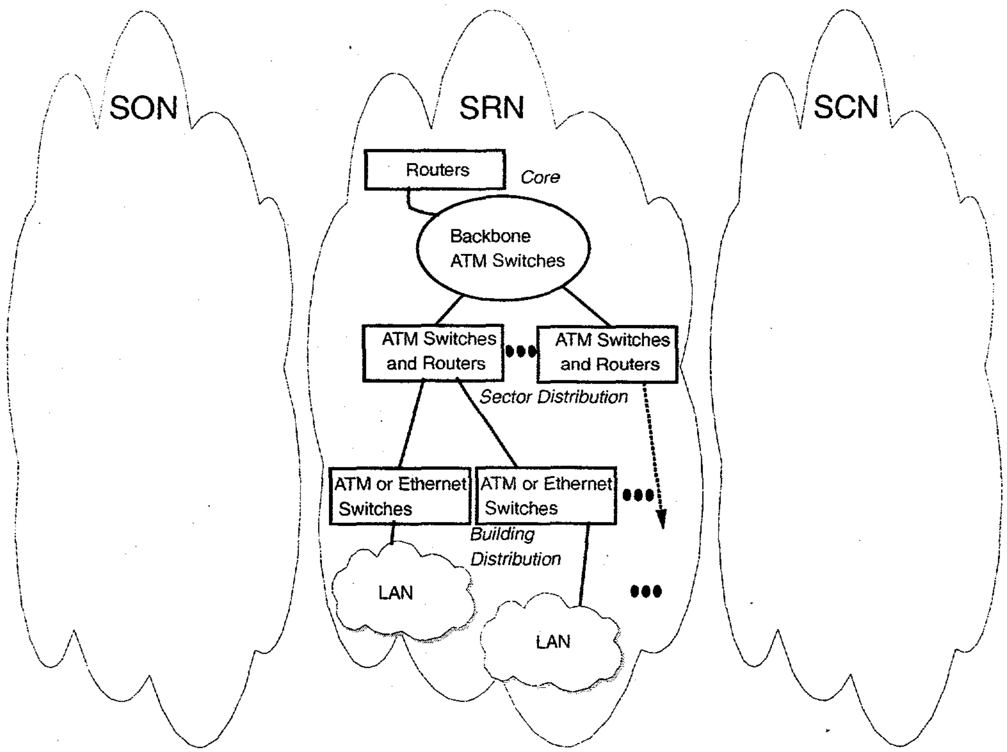

Figure 2 -- Data network installation

the network. Each device executes it own version of firmware, software, and configuration settings. Switches and routers are composed from multiple makes and/or models as necessary to provide the desired functionality.

While the detailed network architecture and interconnection methodology varies from one part of the network to the next, there is a general hierarchical layout to which the network conforms. Figure 2 shows this layout. At the network core is a mesh of asynchronous transfer mode, ATM, switches that transfer traffic between the different geographical sectors of the network Also at the core is a group of routers that provide general, network wide routing functions. A secondary group of ATM switches and routers, located at sector distribution facilities, support the users in the nearby buildings. Within each building are one or more distribution areas with local Ethernet and/or ATM switches that provide that network terminations to the buildings users.

Requirements and Areas of Integration

In considering voice and data network integration, Sandia faces many of the same issues and concerns that face other organizations. These concerns are based in a number of system requirements that the telephone system must address. Requirements derive from service provider and end customer concerns.

From an end customer perspective, the telephone system must promote voice communications within Sandia, and between Sandia and others around the globe. The service must work as expected by the customer. It must meet the customers' reliability expectations. New services and service changes must be delivered within a certain time. The telephone system must provide sufficient features and functionality to make a business difference to Sandia as a whole. A growing requirement is to meet the mobility and flexibility needs of our customers. Customers expect that the service will be available when they want it.

Cost containment or reduction, if possible, is a strong driver of the telephone system and its implementation. However, it is not the only or necessarily the leading requirement. Rather, other customer requirements need to be met at an affordable price. In many respects, cost to provide service is a service provider requirement since the end customer is far removed from the funding process. Other service provider requirements include complexity reduction, flexibility to provide services when and where they are needed both on and off campus, and maintenance of interconnectivity and compatibility with the public network.

Both the voice and data networks attempt to meet customer needs for robust communication, information sharing, and information dissemination. Each network evolved at different times with different technologies to meet distinct modes of human communication. Increasingly individuals wish to communicate knowledge in more depth using a variety of information modes. The manner in which people wish to communicate and use information technology fosters the integration of distinct voice and data networks. There are benefits to integrating the technologies upon which these services are built. However the greatest benefit to the customer will come from applications that integrate different communication services to improve the ways in which people communicate or conduct business.

Voice and data networking evolved at different times and use different technologies, however, they share common functional elements which lend themselves to integration. Common themes between the two networks include switching/electronic systems, distribution including the cable plant, customer premises equipment, and operational processes and procedures. Network integration may occur at any or all of these subsystems. The benefits of integration and the extent to which it occurs is dependent on requirements and the drivers for integrating the systems.

Functional Integration

Synergistic Services

It is possible to integrate voice and data network services without actually integrating the underlying networks. As an example of this type of integration, consider WEB dialing. With this service, a customer is able to use browser software on his or her network connected computer to locate someone's phone number from an electronic directory. The customer is able to place a phone call to that person through the selection of a hyperlink tied to that phone number. The customer benefits from a streamlined process of locating a phone number and then placing a call to that number. Other examples of functional integration include data network based facsimile transfer, data network delivery and handling of voice mail, and call center applications.

In many cases, material that is faxed between locations is generated on computers connected to a data network. To transmit the material, it first has to be printed, then the printed material carried to a facsimile machine and transmitted. On the remote end, the material may be converted to an electronic format that is manipulated on a computer. In this process the telephone network provides a standardized, ubiquitous means for moving information. The availability of a ubiquitous data network provides a more attractive alternative for accomplishing some tasks. Facsimile is a case in point. In many cases at SNL, customers have replaced facsimile transfers within the company with E-mail. When interacting with entities outside of the company, E-mail is not always an option in place of facsimile.

Voice mail has evolved to the point where it provides a wide range of features for manipulating, listening to, and handling voice messages. With the Interactive Voice Response, IVR, features of the voice mail system it is possible to access data stored on corporate computers. The voice mail system is an information store as well as it provides the customer the ability to use a telephone set as an access device to retrieve information from corporate databases. In some cases, the customer may find that a computer provides a more functional interface for handling voice information. The voice mail and E-mail systems are convergence points for the integration of information and the ability to access that information with different tools depending on the customers' preferences.

Each of these applications makes use of synergistic functions of the voice and data networks. The strength of each network is leveraged to provide the customer a service that is more functional and easier to use than what could be provided by each network alone. Consider the utility of an application that combines voice messages, textual E-mail messages, video, as well as other data formats for communicating and storing information. When customers interact with each other, it usually occurs at several levels. A phone conversation may occur; data files may be exchanged through E-mail; the customers may meet face to face. Ultimately customers will expect a communication system that allows them to access, retrieve, and manipulate information of different types through a variety of interfaces.

Operations

In running separate voice and data networks, many of the operational aspects of the two networks are duplicated. Eliminating this duplication can provide service delivery efficiencies and cost reductions. The operation of each network requires engineering support, installation and maintenance crews, procedures and processes for delivering services, documentation systems to track customer orders and the state of the system, and mechanisms for service restoration in the event of an outage.

Because of the history of the two networks, customers have different expectations from each network. The voice network is expected to be very reliable and always available to a larger extent than the data network. The data network is expected to be more innovative in adapting to customer requirements and available technology. These different expectations lead to different attitudes in the manner in which each network is implemented and operated. Over time, customer expectations for each network will converge, and the operation of each network will need to adapt.

Voice over IP Telephony

Voice transport over the IP data protocol, VoIP, has garnered abundant interest lately. There are several reasons for this, but the main ones are the desire to reduce telephone system installation and operating costs and the push to move telephony away from proprietary architectures towards more open standards. It is important to note that VoIP is not the same thing as transporting voice over the internet. VoIP and the internet merely use the same under-lying communication protocols. VoIP systems in a business environment are built on private networks, or inter-nets, to ensure service quality.

While the overall market penetration of VoIP systems remains small, the technology deployment growth as a percentage is high [Duk]. In early 1999, there were few companies and relatively few products in the VoIP realm. By early 2000, all of the major telecommunication equipment manufactures either had VoIP products available or were in later stages of development. Also, investors have formed many new companies specifically to address the VoIP market.

There are several areas in which the technology is deployed and used in industry today. The applications include switch to switch trunking, network based private branch exchanges, PBX, call center applications, and mobile telephony. Each area of use targets different customer requirements. VoIP as it has arisen in the market takes two forms. PBX to PBX gateways and network based "PBXs."

Gateways

The first use of VoIP has been for switch to switch trunking. Telephone trunks are the interconnecting pathways that allow customers served by one telephone switch to communicate with customers in the public switched network or served by other switches. Traditionally trunks are analog lines or an aggregation of individual voice channels over a T1 carrier facility. In the VoIP implementation, the interconnecting analog or T1 trunks are replaced with an IP network. Encoder/decoder units, or gateways, act as the interface between the analog and T1 interfaces on the telephone switch and the IP network. Many of the current generation gateways use Ethernet or ATM interfaces to attach to the IP network. See Figure 3(a).

By compressing the encoded voice signal before transmitting it, switch to switch data flow can be significantly reduced. Companies have made use of this process to reduce the total voice and data bandwidth between two locations. In instances where separate voice and data transport facilities interconnect two

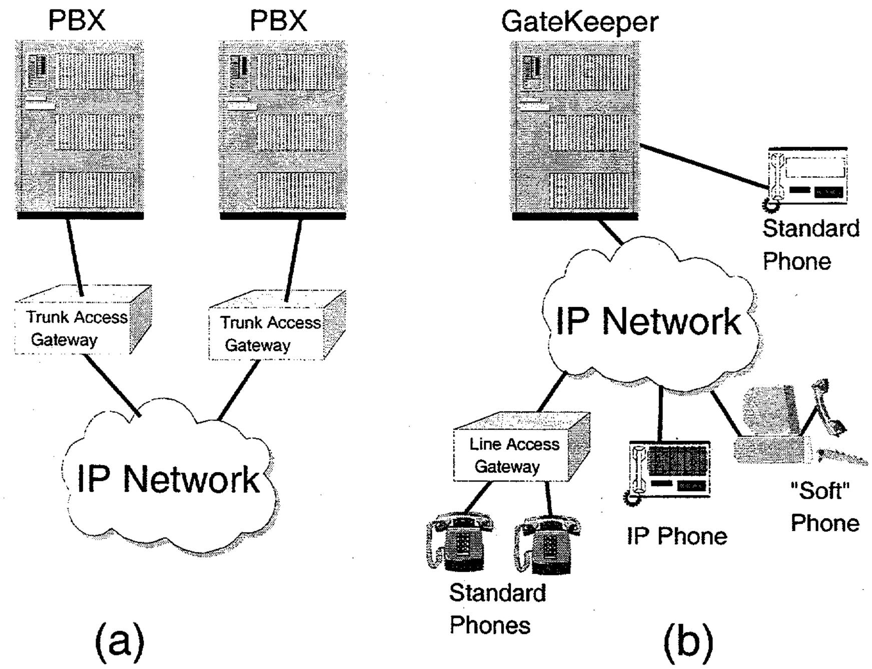

Figure 3 -- -- VoIP System Implementations. (a) Switch to Switch Trunk Access Gateways, (b) VoIP PBX with Gatekeeper, Line Access Gateways, and VoIP Phones.

locations and there is excess data bandwidth, the use of VoIP gateways for the voice interconnect may make it possible to eliminate the separate voice transport facilities. In this event, there would be a corresponding cost decrease. Also reducing the amount and variety of equipment that a company needs to maintain and support between locations can reduce costs. Organizations that have significant voice and data traffic to fixed foreign locations have made extensive use of VoIP gateways to those locations.

New telecommunication service providers are designing and building networks from the ground up based completely on IP packet transport. Level (3) Communications is one such company. The operating premise of these companies is that they can build out IP networks that are simpler and less expensive to operate for providing both voice and data services. As these carriers deploy their networks more widely, VoIP gateway access to the public switched network will become more common.

"LAN" PBXs

Another major area of VoIP technology development and implementation has been in the deployment of "Network" PBXs. See Figure 3(b). In this application, the IP network is the switching fabric for voice services. Other than for adaptation devices to provide interconnectivity to the PSTN, there are no circuit switched elements in the switching fabric. A network attached server, called a Gatekeeper, provides the services necessary for establishing a telephone call. It performs telephone number to IP address resolution. The gatekeeper also provides the commonly expected telephone features such as call forwarding, call transfer, and call conferencing, as well as others. Network PBXs offer abundant possibilities for computer control of telephone call routing, telephone call features, and voice/data application integration. Typically once a telephone call is established, the gatekeeper is no longer required to maintain the connection.

The customer equipment also directly attaches to the IP network and uses it for voice transport. Customer equipment may take on a number of formats. At the most basic level may be an adapter, or an analog access gateway, that translates the interface of a standard telephone set to a network attached interface. The adapter provides the means for encoding and decoding the analog signal from the telephone instrument into IP packets. The adapter may be capable of terminating several to many analog telephone sets. A variation on this theme is what is called a "soft IP phone." With a soft phone, a personal computer or workstation executes software that emulates the function of an access gateway. The software replaces the telephone set with a computer interface that emulates buttons and controls of a real telephone. The computer's sound card provides the physical mechanism for encoding and decoding the voice signal. Perhaps more significant is the development of IP phones. In essence an IP phone is nothing more and nothing less than a specialized, single purpose computer. To the casual observer it looks like any other telephone instrument with a body, a keypad, and a handset. The network interface is Ethernet, either 10 Base-T or 100 Base-T. The IP phone makes it possible to completely combine voice and data infrastructures to the customer desk.

Encoding

The transport of voice over an IP network is accomplished through a series of encoding and packaging steps. The general process is as such. A transducer converts the voice acoustical wave into an analog electrical signal. Typically, VoIP equipment limits the bandwidth of the signal to 3.5kHz, as does a traditional telephone switch. The encoder, or codec, then samples the signal at discrete time intervals, usually 8k samples/second and represents each sample with 7 or 8 bits of data. Note that each sample represents 0.125 ms of the voice signal. Hence the voice signal is correspondingly encoded into a 56 or 64 kbit/s data stream in each direction. This encoding process is called pulse code modulation, PCM, and is specified in ITU standard G.711.

Modern digital telephone switches, including the 5ESS, use G.711 signal encoding for voice calls as well. The difference between a "circuit switched" telephone switch, the 5ESS, and a VoIP system is the manner in which the system transports the data stream from the speaker to the listener. In a circuit switched system, the switch establishes a dedicated channel or circuit between two telephones during calling setup. If a channel is available, the switch constantly transports data at a rate of 56 or 64 kbit/s between the two telephones. In the event that the switch can not

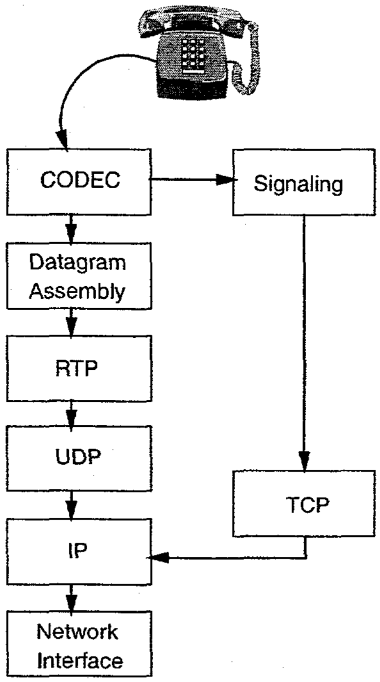

Figure 4 -- Voice Encoding and Encapsulation.

establish a channel between the two telephones, then the switch presents the caller a busy signal. In a circuit switched system, once a channel is established then there is a guaranteed fixed bandwidth, fixed delay, data path between the two end points.

In a VoIP system, there is no explicitly dedicated channel between the two end points. Rather, the system encapsulates the encoded voice signal into a datagram, inserts the datagram into a network packet, and combines the packet with those from other sources and sinks, and then transports the packet from the speaker to the listener. Because there is no predefined bandwidth for a VoIP call channel, it is possible to use codecs of various bandwidths. In fact, this is one manner in which it is possible for VoIP systems to reduce telephone operating costs. Speech signals contain much redundancy. By removing this redundancy, it is possible to compress the bandwidth requirements needed to transport the voice signal. Also many times during a telephone conversation there are periods when there are no utterances between the two callers. By not encoding these silent periods, it is possible to further reduce the required bandwidth for the call. The ITU has define several widely used compressing codecs, including G.726, G.728, G.729/A, G.723.1. Each of these codecs are "lossy" and make a tradeoff between voice quality and bandwidth.

Figure 4 shows the network packet creation process. After the codec samples the voice signal, the packet engine collects several voice samples into a datagram. Different implementations vary the size of the datagram to strike a balance between signal delay and packet overhead. A datagram consisting of 160 samples, 20 ms of the voice signal, is about optimum. The better implementations allow the user to set the datagram size to suite the application. The packet engine then wraps the datagram within several protocol layers including the real time transport protocol, RTP, the user datagram protocol, UDP, and IP before transmitting it through the network interface. RTP is the Internet-standard protocol for the transport of real-time data, including audio and video. It is used ,for media-on-demand as well as interactive services such as Internet telephony. RTP consists of a data and a control part. The latter is called RTCP. The data part of RTP is a thin protocol that provides support timing reconstruction, loss detection, security and content identification. With the protocol overhead, a VoIP call using the G.711 codec requires 70 to 90 kbit/s of bandwidth for current VoIP systems. When using compressing codecs, the bandwidth requirement is between 20 and 60 kbit/s.

The end-to-end delivery delay of VoIP packets is critical to the quality of the call. Through studies and operational experience, service providers have found that round trip voice delays of more than 100 ms are discernable to callers, and delays of more than 500 ms are unbearable. A 100 ms round trip delay dictates that the end-to-end delivery delay is only 50 ms. To maintain the quality of the call and assuming a 20 ms loading delay for the datagram, then a delay budget of 30 ms is available for router queuing and network interface emissions delays. This delay budget imposes a limit of at most two router hops within a typical network design. In addition to overall end-to-end delay, end-to-end delay variance is critical to call quality. Unlike a data connection, voice is a time sensitive application. Datagrams that arrive late at the destination are unusable. To compensate for the delay variance in datagram delivery, VoIP systems employ a jitter buffer which also contributes to the end-to-end delay. Discussions of network design for VoIP applications often mention network design for Quality of Service, QoS. In this context, QoS usually only refers to the guaranteed availability of a specified bandwidth for the voice call within the network. While this is a necessary condition it is not sufficient for maintaining call quality. The overall network design must also consider the delay requirements for VoIP applications.

VoIP Standards

VoIP is a new technology, and as with any technology it is still evolving. The very earliest systems were completely based on proprietary implementations. As demand in the systems grew and more players entered the market, there has been a shift to developing and implementing more to standards. The earliest and still the most widely used standard for VoIP call control is based on the International Telecommunications Union H.320 standard. H.320 is actually a set of specifications for video conferencing over a defined, fixed bandwidth infrastructure such as ISDN. IP telephony uses a subset, H.323, of the total specification. The H.323 specification specifically covers video conferencing over the IP protocol. Since telephony only covers the voice component of teleconferencing some of the H.323 protocol specification is irrelevant to telephony. H.323 is a stable, established protocol. Important also is the fact that H.323 grew out the ITU, an international standards body on telecommunications, essentially the same organization that develops standards for the existing public telephone network. Because it was initially developed to support calls over the switched public network, H.323's call control functions, which are the same as for an ISDN call, are more complex than what is absolutely required for service over an IP network. This complexity has led to the development of a new set of protocols called the Session Interface Protocol, SIP [Sch]. SIP attacks the problem of providing VoIP from a "net-head" perspective.

The Internet Engineering Task Force, IETF, is the sponsor and developer of SIP. SIP uses a simplified message passing service, similar to electronic mail, for establishing voice calls. The protocol is new and still evolving. Some telecom equipment companies have started to develop equipment, telephones, and software utilizing SIP over the last 6 months to a year. We can expect that these two competing protocol sets will evolve, and that the two standards bodies will jockey for position in defining VoIP standards.

Lucent 7 R/E Switch

Market interest is driving Lucent, as well as other equipment vendors, to develop VoIP systems. System development is occurring on small to large carrier class switches. As part of these developments and to keep the switch current, Lucent is adding hardware and features to the 5ESS switch to handle VoIP and ATM telephony and to generally add new functionality. Along with these changes, Lucent is renaming the 5ESS switch to be the 7R/E switch. The 7R/E design does not radically depart from that of the 5ESS, nor does it necessarily obsolete any of the 5ESS's main subsystems. Rather the 7R/E represents an evolutionary path for maintaining the large capital investment in a central office class switch.

Lucent has announced new 7R/E elements to support VoIP, and they are projecting delivery of these elements in the second half of calendar year 2001. These elements include a Packet Switch Module, PSM, a Call Feature Server, CFS, and the Packet Interworking Gateway, IWG. The CFS performs the function of a VoIP gatekeeper. It provides call control functions and implements telephony features now performed by the 5ESS administrative module. The CFS processes calls for both the 5ESS and 7R/E peripherals and adds packet call control to the 5ESS. The IWG performs the role of a VoIP gateway as well as providing an internal bridge for bearer channel traffic between the circuit and packet switched elements of the 7R/E. The IWG is also capable of processing and handling ATM traffic which provides an alternate interface for connecting to the public network or to another private telephone switch. The PSM provides the basic fabric for interconnecting the IWG, the CFS and ATM/IP network interfaces. It also provides the fabric for attaching line and trunk access gateways.

The 7R/E will use the H.323 protocol to implement VoIP in the CFS. The customer interface is provided through one or more mechanisms. Also under development by Lucent is an H.323, network attached telephone. This same phone is intended to work with the Avaya, formerly Lucent, Definity line of PBXs. They will be customizable to either of the switches. Note that while other manufacturers are also producing H.323 phones, these instruments are all largely proprietary and do not inter-operate. Other customer interfaces include standard H.323 soft phone software. The most commonly available package is NetMeeting by Microsoft. Lastly, line access gateways to the 7R/E will provide support to terminate standard telephone instruments. These gateways can be remotely placed at the customer location and interconnected to the 7R/E through an IP network.

This short description of the 7R/E provides no more than an overview of the direction of the Lucent product. Product delivery is still a year or two away before first delivery. As the VoIP technology progresses in the standards bodies and in the market place, one can expect the 7R/E to evolve.

VoIP At Sandia

There are a number of factors that come into play in the decision to deploy VoIP at Sandia. Included among these are the maturity of the technology, cost, reliability, performance, operations, security, and scalability. Also in many cases there are several viable technologies that can address a given problem or task. VoIP competes with these alternate technologies, some of which are more mature and tested. VoIP provides solution sets to address different telephony applications. Some of these solution sets may be more appropriate for use than others. For each application, one has to consider the alternative options on a case by case basis.

When considering VoIP deployment, it will rarely involve an all or nothing decision. What is more likely is that it will coexist with other technologies including the existing circuit switched voice system, wireless, SONET transport, and others.

Security --

From a security view point Sandia's operating environment is unique compared to many companies. DOE and government orders and the presence of classified information and processing systems on site present obstacles to completely combining voice and data traffic. By definition general telephone traffic is unclassified and has the same security profile as the Internet. In the public switched telephone network, PSTN, calls can and do originate and terminate anywhere in the world. A VoIP system at Sandia would reside in the open environment and likely as a network separate from the Sandia Open Network, SON. From an implementation perspective, the Sandia Restricted Network, SRN, is the most suitable environment for VoIP traffic. Of all the networks at Sandia, the SRN is the most pervasive, and it provides a desirable level of protection and threat isolation. However, general deployment of VoIP on the SRN may require that the SRN assume some of the risks inherent with interconnection to the public switched network.

It should also be noted that telephony by its nature needs to be ubiquitous to make it the most useful. This drives the need to deploy it into classified computing areas where it may present a security risk. While Sandia's networks cover a large portion of the population and work spaces, no single network reaches everyone. This report does not delve into all of the security ramifications of VoIP on Sandia's existing networks other than to recognize that security is an issue.

Scalability --

The majority of PBX systems that companies install are relatively small [Sul][BCR], less than 200 lines. This has dictated the initial size of VoIP PBX systems that vendors have introduced. These systems have been small, tailored for a small business or building, and have been designed for an organization to quickly install a combined voice and data network. This situation is changing. Vendors are beginning to target larger enterprise and carrier class installations. Systems from Lucent, Nortel, Siemens, and Alcatel fit into this category.

For a VoIP system, scalability must cover the aspects of system size and capacity, system distribution, operations and feature set complexity. System size and capacity refer to the configured lines and the call volume. System distribution is the physical location and dispersion of the system. Operations and feature set scalability allow the system to offer the functionality that customers want in a manner that can be delivered and maintained.

There have been discussions at Sandia about the feasibility of implementing the voice system with a complete VoIP solution. By any measure Sandia's PBX, the 5ESS, is a large installation. Any VoIP PBX that would replace the Sandia 5ESS circuit switch functionality would need to accommodate Sandia's size, distribution, and complexity. The following addresses some of the basic system sizing that would be necessary to implement such a system. As noted earlier, VoIP, does not represent an all or nothing solution. There may be cases where VoIP is applied more selectively. In these instances, the system size would need to accommodate the particular application.

Figure 5 and Table 1 show measures of Sandia's 5ESS usage. Data such as this provides the basis for sizing the components of a VoIP system. The number of telephone lines in use, or the line count, provides an indication of the

Line Count	19,840	Active
Average Daily Call Completions	117,000	Calls
Average Call Length	5.5	Minutes /Call
Maximum Call Rate	26,000	Calls/Hour

Table 1 --5ESS Telephone Switch Usage.

amount of network equipment that is needed to support telephone services. There is a direct correlation between the line count and the number of hubs, routers, and gateways needed to support the service. It also correlates to the number of resources needed to perform IP address assignment and configuration. The call completion rates provide indicators of the amount of telephone traffic that the network will need to transport. It also correlates to the transaction rate that the gatekeeper will need to process and the rate of domain name service resolutions that are required to support the application.

One can estimate the aggregated traffic that a VoIP application would generate. The traffic on any one hub, router or network component may vary widely depending on traffic patterns. Many times telephone calls are localized to a physical region such as a building so that traffic is also localized. However, one can not always assume localization and one must expect that a significant amount of the traffic will transverse the network backbone. By referring to Figure 5, the maximum call rate is about

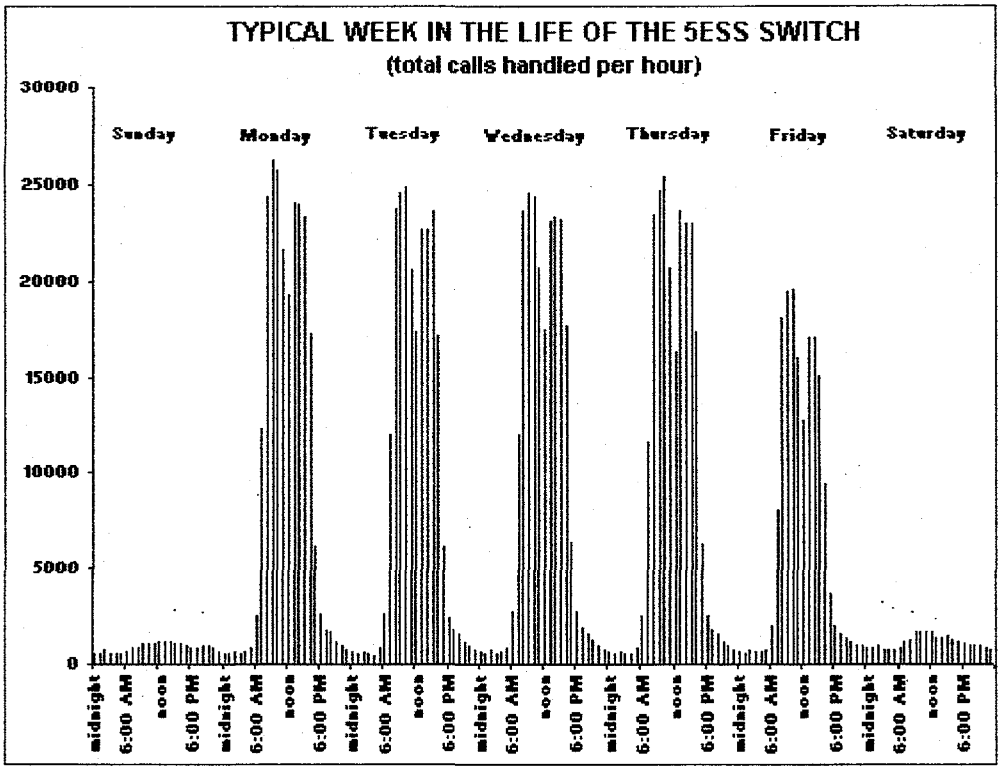

Figure 5 -- 5ESS Weekly Traffic Pattern

26,000 calls per hour. A typical call will last about 5.5 minutes on average. From these statistics we can calculate the aggregate simplex traffic to be 190.7Mbits/s. Total traffic would be twice this rate to accommodate the full duplex nature of a phone call. This rate assumes that the system uses the standard G.711 codec which has an average transport rate of 80 kbit/s with available VoIP systems. The aggregate traffic is calculated as follows.

Traffic Rate=$C_{\text{Rate}}E_{\text{BitRate}}C_{\text{Length}}$, where

C_{Rate}, the call rate (in Calls/Hour),
E_{BitRate}, the bit rate per call (in b/S),
C_{Length}, the call length (in S)

Availability/Reliability --

Often times when people talk about telephone service they refer to it as a lifeline service. By this they mean that the service is always available and that people rely upon it for emergency communications. We have grown to expect that telephone service is always available regardless of external events. To this end commercial equipment vendors have designed multiple layers of redundancy into their systems. Service providers, including Sandia, also go to great lengths to design the telephone service so that it is available and that it performs as expected. In contrast, customers generally do not expect the same level of service availability from data networks and service providers do not attempt to provide the same level of service from the onset of the network design. This is not to say that customers don't want high levels of service availability, just that they do not expect that it will happen. This dissimilarity in service levels presents a dilemma to designers in trying to provide telephony services over a data network service.

The deployment of VoIP services on Sandia data networks will require that the data networks are specifically designed and operated for availability. Some aspects of the data network that may introduce obstacles to making the system available include equipment reliability, configuration management, complexity, system design, service maturity, and customer expectations.

Telephone services provided through digital switches such as the 5ESS represent a very optimized and vertical application. The 5ESS is designed to execute a single application in a particular manner -- the delivery of voice calls and features through circuit switched channels. This design minimizes system complexity and makes it easier to manage, and correspondingly more reliable. By contrast the data networks that Sandia implements are more complex and serve a number of applications. This complexity does not preclude the system from functioning at expected service levels, but it makes it harder to operate them so that they do perform. The data networks transport various protocols for various applications that require different system services. Applications require that several key services all function simultaneously for the application to run. In most data network designs there are a wide variety of routing engines, hubs, and interfaces. VoIP layers telephony services on top of data network services; the components of both of these services have to perform at a very high level to achieve the desired overall system availability. Minimization of data network system complexity is integral to achieving the desired VoIP service availability.

A telephony system, whether VoIP or otherwise, is a collection of subsystems, and the system availability is directly related to the availability of the subsystems. The availability of the 5ESS switch is specified at 99.9999%, or out of service for 31.53 seconds per year. However, the availability of an individual phone line to a customer is less than this amount because of failures in cable, power, or other factors. To meet the expected system availability, the system designer must design the overall system with high availability subsystems and with subsystems redundancy. Sandia's data networks are currently not designed to this criteria. Equipment such as routers and hubs do not have active redundant boards or components. One area of focus in the data network is design for loss of commercial power. The network is not specifically designed to survive a commercial power outage or the intentional disabling of power to buildings at specified times. For VoIP to be successful, the data network and any VoIP specific services must be designed from the beginning for system availability.

Customer expectations play a large role in data network availability. Traditionally, customers have not demanded the same level of availability from data networks and services as they have from the telephone network. As a result, operational and design practices have emerged that hamper system availability. It is not uncommon, to see customers or services providers at

Sandia make data system changes during periods when it can be reasonably expected to affect service or to make implementation changes that are likely to degrade availability. When data service outages do occur, the perceived cost of the outage is not placed as high as the perceived cost of telephone network outages. Data network services are still evolving more quickly than telephony services. As a result, customer demand for faster or different network interfaces and the demand for the new features have shifted the service delivery emphasis to performance at the expense of availability. These differences are not inherently wrong, however they present a problem in trying to provide an expected level of telephony service through the data network.

Recommendations

One condition that arises in deciding to deploy a VoIP telephone system at Sandia, is that we are trying to compare a very mature system with well known functionality and costs to one that is much less so. VoIP involves a rapidly evolving set of products, with emerging functionality, and with less well understood cost components. To date we have not found documentation regarding a large, distributed enterprise like Sandia entirely transitioning to VoIP. However, companies have successfully deployed VoIP for specific applications where VoIP provided advantage.

We have repeatable field data that demonstrates that the 5ESS and our data networks offer two distinctly different levels of availability. Additional probing of other aspects of availability such as the number of simultaneously out-of-service devices or ports needs to be compared. We designed the telephone system to a specification that puts a limit on that number, but we do not in general design our data networks as such. We regularly design the data networks to fill all available ports and limit the total number of chassis in the field in an effort to minimize the cost/port. We would need to redesign our data networks to accommodate reducing the breadth of outages that a single component failure would induce. To do so would most certainly affect the cost and management of our data network, potentially to the point of negating any cost savings of migrating the voice and data services together.

VoIP is currently receiving a lot of attention in the marketplace which is one driver for this discussion. When considering changes to Sandia's telephone system with respect to VoIP, the discussion should not be centered on implementing a system that is entirely or largely built using packet switching technology as opposed to the current circuit switched implementation. Rather, a coexistence of the two technologies is more likely. VoIP is still evolving with respect to standards and product development and at this time is immature.

There are specific operational instances where VoIP has made a difference to other companies. In instances where Sandia's situation is similar, VoIP could possibly provide cost and operational benefits. Organizations that have had to support different switching systems at multiple locations have been able to reduce cost and operating effort by deploying VoIP. At Sandia we have a similar situation in the way we provide services to offsite locations such as the Research Park complex, the BDM building, and Sandia's Carlsbad office, as well as other potential locations. In each of these instances, we extend both data network services and services from the 5ESS. At Carlsbad we also have to support a different PBX; support that could be absorbed into a central switch. The other major area where companies have deployed VoIP is in call center applications. In these settings the call center operators need close integration between voice and data services and have found that the delivery of voice through the computer to improve their operations. As VoIP technology evolves, its range of practical applicability to Sandia may increase.

When trying to evaluate cost advantages to using VoIP, one must be careful to optimize for overall system costs and not just subsystem cost benefits. It has been proposed that VoIP could result in cost savings in operating and maintaining Sandia's outside cable plant and duct system. Sandia's outside cable plant is more than 40 years old in some locations, and it needs to be replaced [ARC]. When considering up front capital expenditures, VoIP appears to be less expensive. However, this does not present a view of the entire life cycle cost. Voice transport via VoIP is much more complex than that over copper cable, and it is reasonable to expect that it will be more expensive to operate. There is also a wide disparity in the life span of the two systems which will drive replacement costs in the future.

It is possible to integrate voice and data network services without actually integrating the underlying networks. As an example of this type of integration, consider the WEB Dialing application. With this service, a customer is able to use browser software on his or her network-connected computer to locate someone's phone number from an electronic directory. The customer is then able to place a phone call to that person through the selection of a hyperlink tied to that phone number. The customer benefits from a streamlined process of locating a phone number and then placing a call to that number. Other examples of functional integration include data network based facsimile transfer, voicemail based facsimile receipt, data network delivery and handling of voice. Each of these applications makes use of synergistic functions of the voice and data networks. The strength of each network is leveraged to provide the customer a service that is more functional and easier to use than what could be provided by each network alone. Sandia should continue to look for and implement these synergistic opportunities.

Sandia's aging inter-building cable system [ARC] and its Move, Add, Change, MAC, work load are two significant cost areas. While VoIP may provide one method for reducing these costs, other technology alternatives are also available. In particular SONET distribution coupled with building located, subscriber line distribution equipment is a well known and mature technology to address inter-building cabling. Wireless technology also presents a viable alternative to reduce cable, duct system and MAC costs. Wireless also enables user mobility. While this paper predominately looks at

VoIP, VoIP as well as other available technologies should be considered.

The following initiatives should be considered to leverage the existing infrastructures while taking advantage of evolving technologies.

- Continue to integrate best of both voice and data work processes into synergistic systems.
- Continue to consolidate operations such as Customer Service Representatives, Trouble Desk, and other processes to support moves/adds/changes.
- Continue funding VoIP initiatives to remain abreast of the technology and its direction. At the same time consider and fund other complementary technologies such as wireless.
- Monitor closely the evolution of Lucent's 7R/E switch
- Continue to monitor the marketplace for case studies or product launches that suite large, physically diverse enterprises like Sandia.
- When making changes or additions to the data network, design in availability and performance levels to accommodate VoIP.
- Support long term infrastructure improvements such as the Exterior Communications Infrastructure Modernization to insure that no matter which path current technology takes, the inside and outside plant infrastructure remains flexible and viable enough to provide whatever telecommunication technology emerges.

Summary

Technological innovations and industry developments in telecommunications have recently generated substantial interest in combining voice, video, and data networks. There are several drivers for wanting to combine these networks. They are expected lower cost structure and increased system efficiencies, regulatory changes, the development and maturing of underlying technologies that facilitates service convergence, and the promise of new customer applications that would not otherwise be possible.

In considering voice and data network integration there are a number of system requirements that the telephone system must address. Requirements derive from service provider and end customer concerns.

From an end customer perspective, the telephone system must promote voice communications, work as expected, meet the reliability expectations, and new services and service changes must be delivered within a certain time. Cost containment or reduction if possible is a strong driver of the telephone system and its implementation. However, it is not the only or necessarily the leading requirement. Rather, other customer requirements need to be met at an affordable price. In many respects, cost to provide service is a service provider requirement since the end customer is far removed from the funding process in Sandia's current funding structure. Other service provider requirements include complexity reduction, flexibility to provide services when and where they are needed both on and off campus, and maintenance of interconnectivity and compatibility with the public network. A growing requirement is to meet the mobility and flexibility needs of our customers.

It is possible to integrate voice and data network services without actually integrating the underlying networks. Examples of this type of integration include Web dialing, data network fax delivery, unified mail messaging and interactive voice response services. Integration can also occur at the operational level. Each network requires engineering support, installation and maintenance crews, procedures and processes for delivering services, documentation systems to track customer orders and the state of the system, and mechanisms for service restoration in the event of an outage.

VoIP, or the transport of voice communications over the Internet Protocol, manifests itself in a few basic applications. The applications include switch to switch trunking, network based private branch exchanges call center applications, and mobile telephony. VoIP as it has arisen in the market takes two forms. PBX to PBX gateways and network based "PBXs." VoIP is an evolving technology. It is based on both proprietary and international standards body specifications. Specifications that are still in flux. VoIP also makes specific demands on the underlying network to maintain service levels. There are a number of factors that come into play in the decision to deploy VoIP at Sandia. Included among these are the maturity of the technology, cost, reliability, performance, operations, security, and scalability. Also in many cases there are several viable technologies that can address a given problem or task.

One condition that arises in deciding to deploy a VoIP telephone system at Sandia, is that we are trying to compare a very mature system with well known functionality and costs to one that is much less so. VoIP involves a rapidly evolving set of products, with emerging functionality, and with less well understood cost components. To date we have not found documentation regarding a large, distributed enterprise like Sandia entirely transitioning to VoIP. However, companies have successfully deployed VoIP for specific applications where VoIP provided advantage. Sandia can leverage it voice and data networks by continuing to integrate the best voice and data work processes and operations, investigating VoIP products such as the Lucent 7R/E, making data network changes to facilitate VoIP, and investing in the infrastructure.

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