Packet Voice Networking

The telephone is the most pervasive of all technology instruments, particularly in business. Every day, businesses make literally thousands of calls, and though the cost of an individual call is often low, the accumulated cost to business is significant.

Introduction

For most companies, a portion of that cost is avoidable. Traditional public voice telephony is a complex tapestry of tariffs and subsidies, often resulting in situations where calling from A to B costs a fraction of the rate from B to A. Companies have long relied on private leased-line networks to bypass public telephone charges, but rates applied to leased lines are also high. Many have looked for alternative strategies.

The search for lower-cost alternatives has never been more pressing. The Gartner Group forecasts:

> WAN usage in Europe will grow 500% but prices will decrease by only 30% resulting in a 300% increase in WAN expenditures… The pressure on the network is building because the two new computing models, Internet/Intranet and Network Computing, both share server centricity and network dependence.
> Source: Gartner Group, 1997

As depicted in Figure 1, there is a confluence of enabling technologies and business needs pressurizing this network tornado.

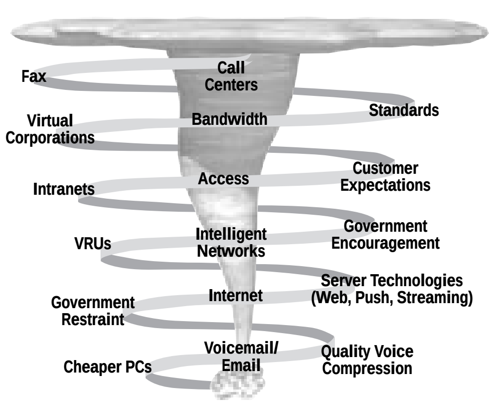


Figure 1: Network Tornado (Adapted from: USA FCC, OPP #29, 1997)

As technologies develop, new needs are uncovered and new solutions become possible. Figure 2 is a closer look at some of the well-known factors feeding this tornado. These charts show dramatic cost reductions and skyrocketing functionality. In the voice arena alone, the price of digital signal processors (DSPs) has fallen dramatically, while the subjective measure of the voice quality generated by the compression algorithms has increased considerably.

All of the above factors increase the pressure in the network tornado. For example, when the MOS scores of compressed voice rise, the amount of bandwidth required to carry high quality voice drops. This has resulted in new technologies like Voice over Frame Relay or IP that enable new business solutions such as web-based telephony and dramatic cost reductions such as toll bypass.

CISCO SYSTEMS

The task at hand is to put these new technologies to work to lower costs and enable new solutions. However, any solution must confront this conundrum: Data networks and telephone networks have been designed with different objectives. A data network is designed for high utilization and is tolerable of small delays. But it is intolerable of errors.

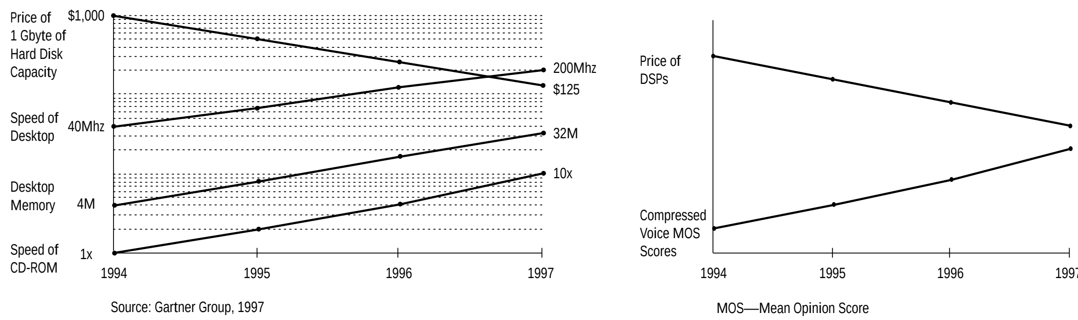


Figure 2: The Pressure Builds

In voice transmission it is delay that is intolerable.

The actual words spoken carry only a part of the meaning. Meaning is also conveyed in inflection, intonation, and pace. A tiny pause has as much meaning as a verbalized part, and its timing must be preserved. Voice networks must be designed to repeat the voice conversation, reliably and in synchronization with the originator's intent.

Networking technologies that are based on packet voice are promising alternatives that can resolve these issues. Most importantly, because packet voice is indistinguishable from data, it can be transported over networks normally reserved for data where costs are often far less.

The Role of This Guide

This guide is intended to help you reduce your voice and data networking expenses while preparing you for the digital tornado. It will help you design packet voice networks that meet your criteria. The first section is a brief review of the characteristics of voice communications and data communications that must be considered when integrating the two types of traffic. The second section is a six-step process for designing integrated voice data networks that ends with real-world examples.

Every network must in some fashion perform addressing, routing, and signaling duties. Addressing is required to identify the calling and the called party. It is also used to associate service classes with the correct party. Routing finds the "best" path from source to destination and moves the information through the network in the most efficient manner as defined by the designer. Signaling alerts end stations and network elements of status and their immediate responsibility to establish a connection.

Because every network, including a packet voice network, has some form of addressing, routing and signaling; both sections are structured around these three topics.

A degree of communications knowledge has been assumed, and the discussion is limited to those areas most pertinent to voice data integration.

Review—Voice Networking

Signaling, addressing, and routing are common to both voice and data networks. Although delay is not normally a consideration in voice networking, it is important enough to introduce the concept first in this section.

Signaling

The purpose of signaling in a voice network is to establish a connection. Signaling occurs at the ingress to the network. It seizes the line, establishes the path across the network, and—at the remote peer—acknowledges the call.

For a telephone call to be completed, several forms of signaling must occur. First, the instant the phone is lifted from the cradle, an "off-hook" signal is sent to the PBX. The PBX responds with a dial tone. Then the telephone sends digits to the PBX.

The exchange that occurs between the PBX and the telephone is known as Station Loop Signaling. (The signaling that originates at the PBX that causes the telephone to ring is also part of Station Loop Signaling.) The transmission of digits from the telephone to the PBX, while part of Station Loop Signaling, is also a form of addressing that will be discussed later.

Once the PBX has received the digits from the telephone, decisions occur at the PBX.

First, is the call local to the PBX? If not, then how is the call best routed? Should the call be placed to the telco central office (CO) or on an internal network to another PBX via a tie line?

In the first case, signaling between the PBX and the CO, the PBX will signal to seize a trunk to the CO. Depending on the facilities, the signaling could be analog or digital. If the facilities are analog, the PBX would use a type of signaling known as E&M.

Assuming the call can be established, similar signaling would then occur at the remote end of the network. The CO seizes a line to the PBX and forwards the digits. The PBX selects the appropriate station, and signals an alert

to the station.

Note that some key systems use other forms of signaling, for example, Foreign Exchange Office (FXO) to Foreign Exchange Station (FXS) signaling for line seizure, rather than E&M. However, the principle is the same.

If the facilities are digital, one of two different methods of signaling can be employed:

• Channel Associated Signaling, (CAS), also known as Robbed Bit Signaling
• Common Channel Signaling, (CCS)

Channel Associated Signaling—In CAS the signaling information is conveyed within the voice channel. Here's how it works: In every sixth frame a bit is stolen from the voice channel to signal information, such as on-hook, off-hook, and so on.

Common Channel Signaling—In this case, you must distinguish T1 from E1.

In North America the standard for digital transmission is T1. T1 uses 24 time slots for a total speed of 1.544 Mbps. In North America CCS, a signaling channel is designated on the T1, and the signaling bits for all the other T1 channels are transmitted across the single CCS channel.

Whereas in Europe and in most other parts of the world, the standard for digital transmission is E1. E1 uses 32 time slots for a total speed of 2.048 Mbps.

The first channel maintains synchronization and passes control information; the 16th channel passes signaling information. Both Channel Associated and Common Channel Signaling use time slot 16. The difference between the two is CCS's use of messages to pass signaling information.

In CAS the signaling is conveyed in channel 16. However, the signaling associated with each channel is maintained via a fixed relationship. CCS eliminates the requirement for the signaling to maintain a fixed relationship with the voice channel. In CCS, signaling is passed in messages between processors that control the terminating switches.

Enterprise PBXs communicate with each other through industry standard protocols such as QSIG and DPNSS, or proprietary protocols such as Siemens CorNet. These specialized protocols enable PBXs to offer enhanced services between sites, such as call redirect and follow-me.

Often, it is impractical to interconnect every PBX. The number of trunks to provide a full mesh can become quite costly. A common alternative is the use of intermediate PBXs, referred to as tandem PBXs. They route signaling requests as well as voice traffic from source to destination PBX.

Signaling System 7—In today's telco network the path for a telephone call is established in a signaling system separate from the transmission path used for the call. Signaling System 7 (SS7) uses out-of-band signaling to establish the appropriate path for the call through the carrier network, before establishing the actual transmission path. Many modern PBXs support the SS7 signaling protocol directly. This enables each PBX to make and process requests from the telco network.

After a call has been established, the transmission path does not change for the duration of the call. Networks with these inherent signaling characteristics are called connection-oriented networks.

To summarize this discussion of signaling thus far, various forms of signaling take place in current voice networks. The discussion has been generally limited to Station Loop Signaling and Trunk Signaling. An integrated voice data network must support the ingress and egress signaling coming from attached telephones, Key Systems, PBXs and telco connections. It should appear as a signaling equivalent to the subscriber lines, tie-lines, or telco connections.

In later sections of this guide the connection-oriented signaling used in telephone networks will be compared with the signaling used in data networks, and then compared with the signaling used in integrated voice data networks.

Addressing

For any telephone network to function, each telephone must be identified by a unique address. Voice addressing relies on a combination of international and national standards, local telephone company practices and internal customer-specific codes.

The International Telecommunications Union ITU-T recommendation E.164 defines the international numbering plan for ISDN. The international telephone service numbering plan is a subset of this numbering plan. Each country's national numbering plan must conform to the E.164 recommendation and work in conjunction with the international numbering plan. Providers of public switched telephone service must ensure their numbering plan aligns with the E164 recommendation and that each of their customers' networks conform.

Alternate numbering schemes are employed by users and PSTN providers for specific reasons. Exceptions to the E.164 recommendation include Carrier Identification Code (CIC), a prefix to select different long distance carriers; prefixes to select tie-lines, trunk groups and WATS lines; and private number plans, such as seven digit dialing. When integrating voice and data networks, each of these numbering plans will need to be considered.

Routing

Routing is closely related to the numbering plan and signaling described above. Routing allows the establishment of a call from the source telephone to the destination telephone. However, most routing is much more sophisticated and enables subscribers to select services, or divert calls from one subscriber to another.

Routing occurs as a result of establishing a set of tables or rules within each switch. As each call arrives, the path to the desired destination and the type of feature services available is derived from these tables or rules. It is important to know just how the routing and the associated features are accomplished in the voice network because these functions may have to be provided in an integrated voice data network.

Delay

Distance is usually the major contributor to delay in existing voice networks. In a phone call to a friend across town, the delay due to distance is imperceptible as the electrical signals travel at the speed of light. In a phone call to someone 6,000 miles away, however, the delay can be noticeable. The time required for the signal carrying voice to travel the distance across the physical network medium is referred to as propagation delay. When distances are short, propagation delay is negligible. As distances increase, delay increases also. The expected propagation delay can be estimated by simply dividing the distance by the speed of light.

In integrated networks, delay can be a voice quality problem. Voice information has a characteristic “timing.” A particular syllable of a word is uttered with an interval of time between it and the following syllable. This tiny pause is as much a part of speech as the verbalized parts, and its timing must be preserved. In traditional voice networks, the voice channel is a synchronized bit stream that preserves timing of all speech elements precisely. However, in data networks, if delay is inserted, due to congestion or handling, the speech becomes corrupted.

Delay is a problem in two ways:

- Delay in an absolute sense can interfere with the by-play of human conversation, the rhythm of inquiry and reply.
- Delay variations, known as “jitter,” can create unexpected pauses between utterances that may impair the intelligibility of the speech itself. Jitter is the more serious problem that packet voice networks must address.

Playout rhythm is corrected in the processing CODEC, while jitter must be addressed in the network.

In the next section handling delay or the delay introduced by network equipment is introduced. You’ll learn that adding equipment for “packetizing” voice, and the delays inherent in the data network, make managing delay a critical factor in integrating voice and data networks.

Review—Data Networking

Data communications, like voice, relies on signaling, addressing and routing to move information from source to destination. In this section a high level overview of data networking will be provided. The section begins with a brief background summary of data networking and connection- oriented and connectionless networks, and then goes on to signaling, addressing and routing.

Time Division Multiplexing—Time Division Multiplexing (TDM) has and will continue to play a major part in the transportation of data. Time Division Multiplexing is heavily used in the networks that are commonly deployed by the telephone companies.

TDM relies on the allocation of time slots on an end-to-end basis. For example, a PCM voice channel requires 64 kbps to be allocated from end-to-end. This bandwidth must be allocated regardless of whether there is an actual phone conversation. As a result of the allocation of time slots, a characteristic common to all TDM multiplexers is that the sum of the inbound ports cannot exceed the outbound trunk to another TDM. This guarantees that connections will have access to the trunk, resulting in very low delay. However there is a cost in utilization which is known as trunk efficiency.

Statistical Networking—By contrast, Statistical Networking relies on the laws of probability for servicing inbound traffic. Unlike TDM, a characteristic of this type of networking is that the sum of the inbound ports routinely exceeds the capacity of the trunk.

Data traffic by nature is very bursty. At any instant in time the average amount of offered traffic may be well below the peak rate. Designing the network to more closely match the average offered traffic ensures that the trunk is more efficiently utilized. However, this efficiency is not without cost.

Data traffic is highly sensitive to data loss. Increasing efficiency elevates the risk of a surge in offered traffic that exceeds the trunk. In this case, there are two options: discard the traffic or buffer it.

Though buffering helps to reduce the potential of discarded data traffic, it, increases by its very nature, the delay of the data. In fact, large amounts of oversubscription and large amounts of buffering can result in long, variable delays.

Another method for improving data transmission efficiency is to modify the packet size. Certain data traffic types, such as file transfer, are more efficiently transported using large packet sizes. Large packet sizes reduce the percentage of overhead in relation to data carried.

Variable packet sizes are ideal for sizing the packet to match the amount of data traffic that needs to be transported. However, fixed size packets result in simpler memory management schemes and are more efficiently processed in hardware. This explains one of the reasons why fixed sized cells are employed in ATM.

Packet efficiency is not the only factor affecting transmission efficiency. Other factors such as congestion have a major and at times larger impact on efficiency. Closed loop congestion management such as that used in the Cisco® StrataCom® products dramatically improves line efficiency.

Most data networking equipment employs some form of statistical networking. Examples include: Ethernet switches, both 10 and 100BaseT, Frame Relay and ATM switches and routers. These devices run the spectrum of fixed versus variable, and small versus large packet sizes. Packet sizing is based on the type of data networking each device is targeting.

Overall, as throughput efficiency is improved it has the result of increasing delay and delay variation. Consequently, techniques must be employed in the integrated voice data network to maintain efficiency while minimizing delay and delay variation.

Signaling

Any discussion of signaling must first note the differences in behavior of connection-oriented versus connectionless networks.

Connection-Oriented Networks—A connection is a logical pathway between the source end station and the destination end station. The term virtual circuit (VC) is often used to describe this type of connection. Examples of virtual circuits include the allocation of time slots in a TDM network and the connection established in an ATM network using permanent virtual circuits or PVCs. To establish a connection, two things are necessary: a route for the connection and the connection requirements.

The connection requirements are a set of requests the end station makes of the network. For example, in an ATM network, a virtual circuit request may specify an average data rate, a maximum data rate, and a loss ratio. The network will attempt to build the specified VC, if the resources are available.

Connectionless Networks—Unlike connection-oriented networks, connectionless networks are not involved in session or connection establishment. Router networks are a classic example of connectionless data networking. Routers accept all traffic and attempt to forward it on a “best effort” basis. The router may prioritize traffic, but it has no prior knowledge of network status.

Connectionless network protocols like RSVP (Resource Reservation Protocol) signal the session requirements to the backbone network, in much the same manner as connection requirements are signaled in a connection-oriented network.

Addressing

The original architects of the Internet designed a protocol and addressing scheme that was abstracted from the underlying technologies. This interoperability let’s users communicate with each other regardless of the network to which they are connected. The TCP/IP addressing scheme and protocol that was developed allows an end station on Ethernet to communicate across X.25 to an end station on Token Ring.

If IP is used as the Layer 3 addressing example, then what stands out is the concept of network and host sections. This address layering increases the flexibility and scalability of data networks.

Examples of layered addressing approaches are available in local area networks as well as many wide area networks. Layered addressing is a much different scheme than that used in voice networking.

Address Resolution

Its important to point out how Layer 3 addresses are resolved to Layer 2 addresses because this concept is foreign to voice networks. The three different methods covered are:

- Broadcast
- Address resolution servers
- Local configuration tables

Broadcast—For networks that share a common media, a simple approach is to use a broadcast. In this case, when a station has the destination's IP address but not its underlying Ethernet address, the station broadcasts a request to everyone on the shared media. All devices receive the message but only that station with the matching IP address responds back to the source with it's Ethernet address.

Address Resolution Server—Broadcast is a very effective mechanism on connectionless, shared media LANs, but not on connection-oriented networks. In the case of ATM, a server is used to resolve the ATM address to an IP address in "Classical IP over ATM." All queries for unknown devices can be sent to the address server and the server will respond with the correct underlying ATM address. After receiving the destination's underlying address from the server, the source communicates directly with the destination.

Local Configuration Tables—Simple networks can avoid either of the previous approaches. Each end station can be configured with the proper Layer 3 and Layer 2 mappings.

Private voice networks, in contrast, typically don't need to resolve addresses.

Address Abstraction

Two benefits of Layer 3 abstraction of addressing from the physical layer are innovations known as:

- Dynamic Host Configuration Protocol (DHCP)
- Domain Name System (DNS)

Dynamic Host Configuration Protocol (DHCP) eases the burden of maintaining and administering IP addresses. A server assigns IP addresses to devices on an as-needed basis. This eliminates the need to statically configure an IP address in the end stations. Also, a pool of addresses can be shared among a greater number of end stations, reducing the required number of IP addresses.

The Domain Name System (DNS) eliminates the need to know the IP address of the destination. Instead, an easy to remember naming convention is used. The DNS server resolves the IP address based on the name in "jsmith@cisco.com" format. After the DNS provides the IP address, the end stations communicate with each other directly.

Both mechanisms, DHCP and DNS, are part of the effort to add mobility and directory name services to the entire enterprise. Adding voice to an IP network will undoubtedly have to be compatible with these addressing mechanisms.

Routing

Route determination is the process of finding the best path from source end station to destination end station. Routing is used at both Layer 2 and at Layer 3 protocols. Either static tables pre-programmed in each switch or a dynamic routing protocol implemented throughout the network can be used to determine the route.

As a point of contrast, it is typical for a PBX in a private network to employ statically configured routing tables. Consequently, the routing decision is made during the call request at the first PBX.

Packet Voice Network Design—A Six Step Process

In designing an integrated voice and data network one must be alert to the similarities between data and voice networks. Overall, both voice and data networks are trying to establish an end-to-end session between two users. This is why the concepts of signaling, address interpretation, and routing are similar.

The challenge in designing integrated voice and data networks is understanding just how these elements are reconciled in the same network. Delay and delay variation in the integrated network—and how to reduce their impact—as both delay-sensitive voice and delay-insensitive data traffic are interwoven into the same network fabric are discussed in later sections.

We shall also discuss delay and delay variation in the integrated network—and how to reduce their impact—as both delay-sensitive voice and delay-insensitive data traffic are interwoven into the same network fabric.

A point worth reiterating, all "voice" traffic is not necessarily delay sensitive. For example, fax and voice mail may not impose the same "real-time" restrictions as natural voice conversations. Adding fax and voice mail may be justification alone for supporting "voice" over data networks.

Here are the six steps that will be followed:

1. Network Audit
2. Network Objectives
3. Technology and Services Review
4. Technical Guidelines
5. Capacity Planning
6. Financial Analysis

The sequence begins with an evaluation of the current network, then helps set objectives and goals, evaluates available technologies, provides a technical design engineered for the unique characteristics of voice communications, and ends with a financial analysis.

Step 1—Network Audit

The first step in designing a network is to take stock of what currently exists. Review the existing equipment and evaluate its capabilities and operating costs. Determine existing facility costs and whether the networks will meet planned voice and data needs. Identify up coming projects that will require the network and determine their impact on the network as much as possible.

What has been the service quality of both voice and data? Does it need improving? Finally, a traffic study may be necessary to look at current traffic patterns. Perhaps some links can be removed, while others need increasing.

Step 2—Network Objectives

Once the baseline is established, the next step is to set the integrated network objectives. First, determine the dominant traffic type the integrated network is expected to support. Also, consider how closely voice and data functionality will be tied together. These appraisals will help in the selection of the appropriate technologies. Setting voice quality objectives will establish your organization's acceptable delay and compression limits.

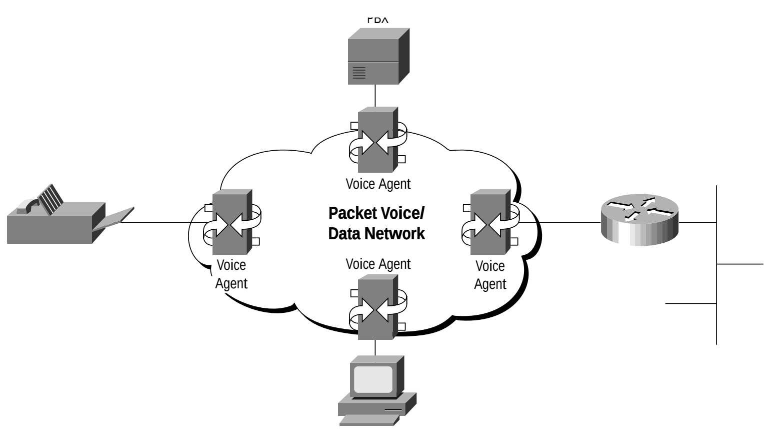
Determine the voice traffic load the network can absorb and still meet the baseline data networking requirements. In setting financial objectives determine the appropriate ROI or payback period.

Step 3—Technology & Services Review

Step 3 is to evaluate available technologies and services and select the model and technology that best meet the objectives set in Step 2.

Figure 3: Packet Voice Model

All packet voice systems follow a common model, as shown in Figure 3. The packet voice transport network, which may be based on IP, Frame Relay, or Asynchronous Transfer Mode (ATM), forms the traditional "cloud." At the edges of this network are devices or components that can be called "voice agents." It is the mission of these devices to change the voice information from its traditional telephony form to a form suitable for packet transmission. The network then forwards the packet data to a voice agent serving the destination or called party.



Integrating voice and data networks should include an evaluation of these three technologies:

- Voice over ATM (VoATM)
- Voice over Frame Relay (VoFR)
- Voice over IP (VIP)

In the next section each of these will be reviewed. First, here's a look at the transport versus translational model for integrating voice and data.

There are two basic models for integrating voice over data, Transport versus Translate. Transport is the transparent support of voice over the existing data network. Simulation of tie lines over ATM using circuit emulation is a good example.

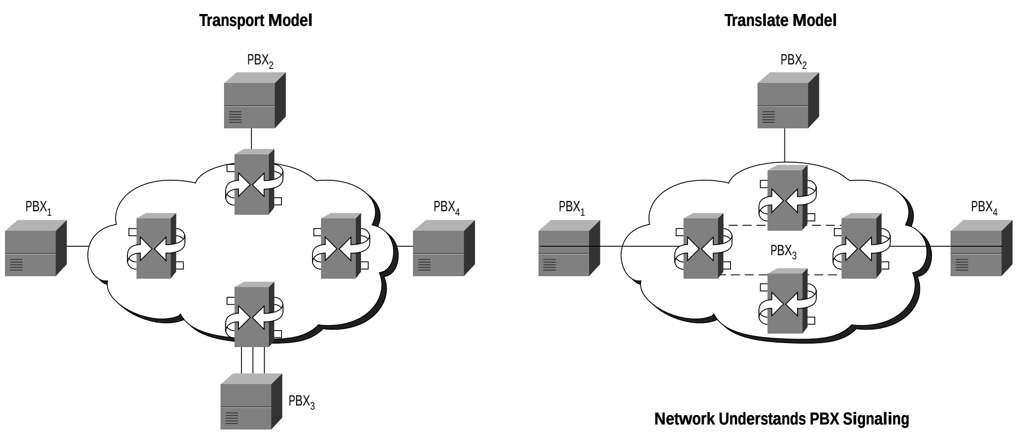
Translate is the translation of traditional voice functions by the data infrastructure. An example would be the interpretation of voice signaling and the creation of SVCs within ATM. Translate networking is more complex than transport networking and its implementation is a current topic within many of the standards committees today.

Select the appropriate model based on availability, cost and technical considerations. Voice over ATM, FR and IP are covered next.

Voice over ATM

The ATM Forum and the ITU have specified different classes of services to represent different possible traffic types.

Figure 4: Connectivity/Signalling Models



Designed primarily for voice communications, Constant Bit Rate (CBR) and Variable Bit Rate (VBR) classes have provisions for passing real-time traffic and are suitable for guaranteeing a certain level of service. CBR, in particular, allows the amount of bandwidth, end-to-end delay, and delay variation to be specified during the call setup.

Designed principally for bursty traffic, Unspecified Bit Rate (UBR) and Available Bit Rate (ABR) are more suitable for data applications. UBR, in particular, makes no guarantees about the delivery of the data traffic.

The method of transporting voice channels through an ATM network is dependent upon the nature of the traffic. Different ATM adaptation types have been developed for different traffic types, each with its benefits and detriments. ATM Adaptation Layer 1 (AAL1) is the most common adaptation layer used with CBR services.

Unstructured AAL1 takes a continuous bit stream and places it within ATM cells. This is a common method of supporting a full E1 byte stream from end-to-end. The problem with this approach is that a full E1 may be sent, regardless of the actual amount of voice channels in use.

Structured AAL1 contains a pointer in the payload that allows the DS0 structure to be maintained in subsequent cells. This allows network efficiencies to be gained by not using bandwidth for unused DS0s.

The remapping option allows the ATM network to terminate structured AAL1 cells and remap DS0s to the proper destinations. This eliminates the need for PVCs between every possible source-destination combination. The major difference from the above approach is that a PVC is not built across the network from edge to edge.

VoATM Signaling

This diagram describes the transport method, in which voice signaling is carried through the network transparently. PVCs are created for both signaling and voice transport. First, a signaling message is carried transparently over the signaling PVC from end station to end station. Second, coordination between the end systems allow the selection of a PVC to carry the voice communication between end stations.

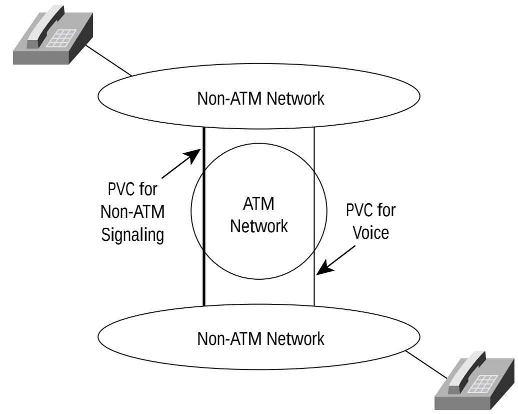


Figure 5: VoATM Signaling, Transport Model

At no time is the ATM network participating in the interpretation of the signaling that takes place between end stations. However, as a value-added feature, some products are capable of understanding CAS signaling and can prevent the sending of empty voice cells when the end stations are on-hook.

Figure 6 shows the translate model. In this model the ATM network interprets the signaling from both non-ATM and ATM network devices. PVCs are created between the end stations and the ATM network. This contrast with the previous model, in which the PVCs were carried transparently across the network.

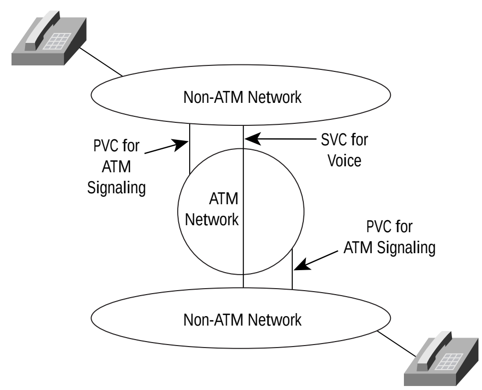


Figure 6: VoATM Signaling, Translate Model

A signaling request from an end station causes the ATM network to create a switch virtual circuit (SVC) with the appropriate QoS to the desired end station. The creation of an SVC versus the prior establishment of PVCs is clearly more advantageous from three aspects:

- SVCs are more efficient users of bandwidth.
- QoS for connections do not need to be constant as with PVCs.
- The ability to switch calls within the network, can lead to the elimination of the Tandem PBX and potentially the edge PBX.

Cisco has developed a similar function, referred to as Voice Network Switching (VNS) that performs the translate function.

VoATM Addressing

ATM standards support both a private and public addressing scheme. Both schemes are 20 bytes in length.

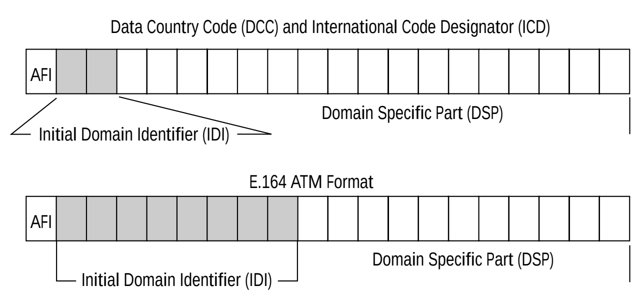


Figure 7: VoATM Addressing

The AFI or Authority and Format Identifier, identifies the particular addressing format employed. There are three identifiers currently specified: Data Country Code (DCC), International Code Designator (ICD) and E.164. Each is administered by a standards body. The second part of the address is the Initial Domain Identifier. This address will uniquely identify the customer’s network. The E164 scheme has a longer IDI that corresponds to the 15-digit ISDN network number. The final portion, the Domain Specific Part, or DSP, identifies logical groupings and ATM end stations.

In a transport model you don't need to be aware of the underlying addressing used by the voice network. However, in the translate model, the ability to communicate from a non-ATM network device to an ATM network device implies a level of address mapping. Fortunately, ATM supports the E.164 addressing scheme; the same scheme is employed by telephone networks throughout the world.

VoATM Routing

ATM uses a private network to network interface (PNNI), a hierarchical, link-state, routing protocol that is scalable for global usage. In addition to determining reachability and routing within an ATM network, it is also capable of call setup.

A VC call request causes a connection with certain QoS requirements to be requested through the ATM network. The route through the network is determined by the source ATM switch based on what it determines is the best path through the network based on the PNNI protocol and the QoS request. Each switch along the path is checked to see if it has the appropriate resources for the connection.

Once the connection is established, voice traffic will flow between end stations as if a leased line existed between the two. This specification spells out routing in private networks. Within carrier networks the switch to switch protocol is B-ICI. Current research and development of integrated non-ATM and ATM routing will yield new capabilities to build translate level voice and ATM networks

VoATM and Delay

ATM has several mechanisms for controlling delay and delay variation.

The QoS capabilities of ATM allow the specific request of constant bit rate traffic with bandwidth and delay variation guarantees. The use of VC queues allows each traffic stream to be treated uniquely. In the case of voice traffic, priority can be given for its transmission. The use of small fixed size cells reduces queuing delay and the delay variation associated with variable sized packets.

Voice over Frame Relay

VoFR Signaling

Historically, Frame Relay call setup has been proprietary by vendor. This has meant product from different vendors would not interoperate. Frame Relay Forum FRF.11 establishes a standard for call setup, coding types, and packet formats for voice over Frame Relay, and will provide the basis for interoperability between vendors in the future.

VoFR Addressing

Address mapping is handled through static tables; dialed digits mapped to specific PVCs. How voice is routed depends upon which routing protocol is chosen to establish PVCs and the hardware used in the Frame Relay network. Routing can be based on bandwidth limits, hops, delay, or some combination, but most routing implementations are based on maximizing bandwidth utilization.

The two extremes for designing a VoFR network are:

- A full mesh of voice and data PVCs to minimize the number of network transit hops and maximize the ability to establish different qualities of service. A network designed in this fashion minimizes delay and improves voice quality, but exemplifies the highest network cost.
- Most Frame Relay providers charge based on the number of PVCs used. To reduce costs, both data and voice segments can be configured to use the same PVC; thereby, reducing the number of PVCs required. In this design, the central site switch re-routes voice calls. This design has the potential problem of creating a transit hop when voice needs to go from one remote to another remote office. However, it avoids the compression-decompression that occurs when using a tandem PBX.

A number of mechanisms can minimize delay and delay variation on a Frame Relay network. The presence of long data frames on a low-speed Frame Relay link can cause unacceptable delays for time-sensitive voice frames. To reduce this problem, some vendors implement smaller frame sizes to help reduce delay and delay variation. FRF.12 proposes an industry standard approach to do this, so products from different vendors will be able to interoperate and consumers will know what type of voice quality they can expect.

Methods for prioritizing voice frames over data frames also help reduce delay and delay variation. This, and the use of smaller frame sizes, are vendor-specific implementations. To ensure voice quality, the Committed Information Rate (CIR) on each PVC should be set to ensure that voice frames are not discarded.

Frame Relay is quite common and comparatively affordable. It is also becoming widely available throughout the world. Frame Relay is an interface specification, whereas, ATM and TCP/IP are architectural specifications. Consequently, Frame Relay will likely be used solely as a transport mechanism.

Future Frame Relay networks will provide SVC signaling for call set up, and may also allow Frame Relay DTEs to request a quality of service for a call. This will enhance voice over Frame Relay quality in the future. Cisco has enhanced the LMI and the congestion notification methods between Cisco Frame Relay switches and routers to improve VoFR functionality.

Voice over IP

In this last section of Step 3, Voice over IP signaling, routing and addressing are explored.

VoIP Signaling

VoIP signaling has three distinct areas: signaling from the PBX to the router, signaling between routers, and signaling from the router to the PBX. The corporate intranet appears as a trunk line to the PBX, which will signal the corporate intranet to seize a trunk. Signaling from the PBX to the intranet may be any of the common signaling methods used to seize a trunk line, such as FXS or E&M signaling. In the future, digital signaling such as CCS or QSIG will become available. The PBX then forwards the dialed digits to the router in the same manner the digits would be forwarded to a telco switch.

Within the router the Dial Plan Mapper maps the dialed digits to an IP address and signals a Q.931 Call Establishment Request to the remote peer that is indicated by the IP address. Meanwhile, the control channel is used to set up the Real Time Protocol (RTP) audio streams, and the RSVP protocol is used to request a guaranteed quality of service.

When the remote router receives the Q.931 call request it signals a line seizure to the PBX. After the PBX acknowledges, the router forwards the dialed digits to the PBX, and signals a call acknowledgment to the originating router.

In connectionless network architectures like IP, the responsibility for session establishment and signaling resides in the end stations. To successfully emulate voice services across an IP network, enhancements to the signaling stacks are required.

For example, an H.323 agent is added to the router for standards-based support of the audio and signaling streams. The Q.931 protocol is used for call establishment and tear down between H.323 agents or end stations. RTCP, the Real Time Control Protocol, is used to establish the audio channels themselves. A reliable session-oriented protocol, TCP, is deployed between end stations to carry the signaling channels. RTP, the Real Time Transport Protocol, which is built on top of UDP, is used for transport of the real-time audio stream. RTP uses UDP as a transport mechanism because it has lower delay than TCP, and because actual voice traffic, unlike data traffic or signaling, tolerates low levels of loss and cannot effectively exploit retransmission.

Table 1 depicts the relationship between the ISO reference model and the protocols used in IP voice agents.

Table 1 ISO Reference Model and H.323 Standards

ISO Protocol Layer	ITU H.323 Standard
Presentation	G.711,G.729, G.729a, etc.
Session	H.323, H.245, H.225, RTCP
Transport	RTP,UDP
Network	IP, RSVP, WFQ
Link	RFC1717(PPP/ML), Frame, ATM, etc.

VoIP Addressing

In an existing corporate intranet, an IP addressing plan will be in place. To the IP numbering scheme, the voice interfaces will appear as additional IP hosts, either as an extension of the existing scheme, or with new IP addresses.

Translation of dial digits from the PBX to an IP host address is performed by the dial plan mapper. The destination telephone number, or some portion of the number, will be mapped to the destination IP address. When the number is received from the PBX, the router compares the number to those mapped in the router table. If a match is found, the call is routed to the IP host. After the connection is established, the corporate intranet connection is transparent to the subscriber.

VoIP Routing

One of the strengths of IP is the maturity and sophistication of its routing protocols. A modern routing protocol, such as EIGRP, is able to take delay into consideration in calculating the best path. These are also fast converging routing protocols, which allows voice traffic to take advantage of the self-healing capabilities of IP networks. Advanced features, such as policy routing and access lists, make it possible to create highly sophisticated and secure routing schemes for voice traffic.

RSVP can be automatically invoked by Cisco's VoIP gateways to insure that voice traffic is able to use the best path through the network. This can include segments of arbitrary media, such as switched LANs or ATM networks. Some of the most interesting developments in IP routing is the development of Tag Switching and other IP switching disciplines. Tag Switching provides a way of extending IP routing, policy, and RSVP functionality over ATM and other high-speed transports. Another benefit of Tag Switching is its traffic engineering capabilities, which are needed for the efficient use of network resources. Traffic engineering can be used to shift traffic load based on different predicates, such as time of day.

The next section explores developments in IP that will control network delay and delay variation.

VoIP and Delay

Routers and specifically IP networks offer some unique challenges in controlling delay and delay variation. Traditionally, IP traffic has been treated as "best effort," meaning that incoming IP traffic is allowed to be transmitted on a first-come, first-served basis. Packets have been variable in nature, allowing large file transfers to take advantage of the efficiency associated with larger packet sizes. These characteristics have contributed to large delays and large delay variations in packet delivery. However, recent efforts have been made through standards and Cisco's own unique efforts to support traffic that is more sensitive to delay and delay variation. Resource Reservation Protocol or RSVP allows us to reserve resources in the network by the end station. This allows us to allocate queues for different types of traffic, helping us to reduce delay and delay variation inherent in current IP networks.

The second part of supporting delay-sensitive voice traffic is to provide a means of prioritizing the traffic within the router network. RFC 1717 breaks down large packets into smaller packets at the link layer. This reduces the problem of queuing delay and delay variation by limiting the amount of time a voice packet must wait in order to gain access to the trunk.

Weighted Fair Queuing or priority queuing allows the network to put different traffic types into specific QoS queues. This is designed to prioritize the transmittal of voice traffic over data traffic. This reduces the potential of queuing delay.

These next points are raised to explain some of the developments taking place within the IP community. One of the issues with regard to the deployment of IP was the different underlying Layer 2 protocols and the need to provide some means of address resolution. Address resolution can be statically defined within a table, employ some form of broadcast or use a central address resolution server.

Another development within the IP community, explained earlier, is the use of DHCP and DNS to provide a level of abstraction. DHCP allows you to ignore your IP address; whereas, DNS allows you to ignore the address of the person or thing you wish to contact. Perhaps similar mechanisms will help evolve the telephone from a physical to a logical device.

Finally, one of the “hot” topics is the use of an enterprise directory service to define identities and policies. This is a level of abstraction beyond IP addressing and it will be interesting how this could be potentially tied to voice communications.

Packet Voice Summary

Here is a review of the different technology alternatives.

ATM is connection-oriented. It was designed to handle time-sensitive traffic, such as voice. Its signaling, addressing and routing allows you to build a network that follows the translation model. The routing function in particular is quite robust, allowing you to build connections based on meeting a certain delay and delay variation.

Frame Relay has provisions for specifying voice within the type field. It is also relatively inexpensive and quite common in many parts of the world. Frame Relay services may provide SVCs and support QoS in the future, but its lack of sophisticated signaling, addressing and routing functionality will likely prevent you from moving beyond the transport model towards building the translation model.

IP is connectionless; developments in areas of prioritization, resource reservation, and packet fragmentation are all relatively new. IP like ATM has robust signaling, addressing and routing functionality, which makes the translation model a possibility. Other compelling incentives for IP are its integration with current data applications and its ubiquity.

After evaluating the available voice over data network technologies and services, the next section covers the salient technical design guidelines that should be considered.

Step 4—Technical Guidelines

This section will review factors that could potentially impact voice quality and provide voice quality guidelines. Further information is also provided on how much delay can be expected.

Coding and voice compression methods are the first factors that could potentially affect voice quality. The term coding refers to the entire process of converting between an analog voice signal and its digital counterpart. Pulse Code Modulation or PCM is the standard for representing voice as a 64 kbps bit stream.

Compression is the method of reducing the amount of digital information below the traditional 64 kbps. Advances in technology have greatly improved the quality of compressed voice and have resulted in a spectrum of compression algorithms. Multiple conversions from analog to digital or changes in compression schemes can impair the quality of the original voice signal.

Two common network characteristics that affect quality are: delay and delay variation, or more often referred to as “digital jitter.” Delay can cause two potential impairments to speech. First, long delays in conversation cause the receiver to start to talk before the sender is finished. Second, delay exacerbates the problem of echo, which is the reflection of the original signal back to the sender. Echo is indiscernible under low delay conditions. It is noticeable to the point of distraction when the delay becomes too great. Digital jitter causes gaps in the speech pattern that cause the quality of voice to be “jerky.” Line quality also affects voice quality, but is outside the scope of this paper.

When dealing with voice compression, a trade-off occurs between the level of voice quality delivered and the amount of bandwidth savings achieved. Through the use of voice compression and, therefore, the optimization of bandwidth, significant monthly cost savings are possible.

Pulse Code Modulation (PCM) refers to the toll quality voice expected from the public switched telephone network (PSTN). PCM runs at 64 kbps, provides no compression, and therefore no opportunity for bandwidth savings.

Adaptive Differential Pulse Code Modulation (ADPCM) provides three different levels of compression. The quality change is virtually imperceptible compared to 64 Kb PCM. Some fidelity is lost as the compression increases. Depending on traffic mix, cost savings generally run 25% for 32 Kb ADPCM, 30% for 24 Kb ADPCM, and 35% for 16 Kb ADPCM.

Low-Delay Code-Excited Linear-Prediction is known as LD CELP. The CELP algorithm models the human voice. Depending on traffic mix, cost savings may be up to 35% for 16 Kb LDCELP.

Conjugate-Structure Algebraic-Code-Excited Linear-Prediction, abbreviated as CS-ACELP, provides 8 times the bandwidth savings over PCM and, of course, 4 times that of 32 Kb ADPCM. CS-ACELP is a more recently developed algorithm which is modeled after the human voice and delivers quality comparable to LDCELP and 24 Kb ADPCM. Again, depending on traffic mix, cost savings generally run 40% for 8 Kb CS-ACELP.

Table 2 lists the various compression methods and their associated quality scores.

Table 2 Compression Methods

Compression Method	ITU Standard	Data Rate	MOS Score	Delay
PCM	G.711	64 kbps	4.4	0.75 ms
ADPCM	G.726	32 kbps	4.2	1 ms
LD-CELP	G.728	16 kbps	4.2	3-5 ms
CS-ACELP	G.729	8 kbps	4.2	10 ms
CS-ACELP	G.729a	8 kbps	4.2	10 ms

MOS, or mean opinion score, is a widely used subjective measure of voice quality. Scores of 4 to 5 are deemed toll quality, 3 to 4 communication quality and less than 3, synthetic quality. Table 2 shows today's MOS for varying compression algorithms such as ADPCM and CS-ACELP. These high MOS scores are a result of improvements in the algorithms along with the dramatic increases in the power of the Digital Signal Processors (DSPs). This table makes clear the opportunity that now exists to integrate voice and data networks while maintaining high voice quality. Note the trade-off between high mean opinion voice quality scores and the higher delay. When designing networks, this must be balanced to ensure overall voice quality.

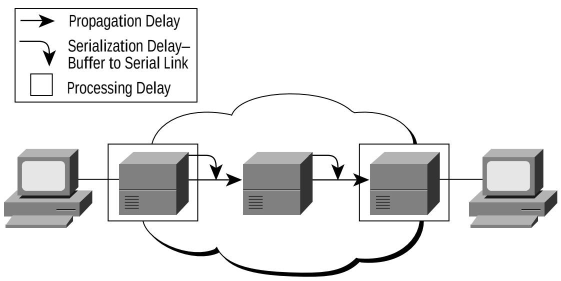
In Table 3 is a summary of the ITU's recommendations for voice delay guidelines. Delays below 150 milliseconds (ms) are considered acceptable for most applications. Delays ranging from 150 - 400 ms are also acceptable subject to current voice quality. For example, a 200 ms delay from Chicago to NYC is unacceptable, given experiences with public networks. On the other hand, a 200 ms delay from Chicago to Singapore will be acceptable given current conditions. Furthermore, higher delays may be acceptable if cost savings are taken into account.

Table 3 ITU Delay Recommendations

One Way Delay	Description
0-150	Acceptable for most user applications.
150-400	Acceptable provided that administrations are aware of the transmission time impact on the tansmission quality of user applications.
400+	Unacceptable for general network planning purposes; however, it is recognized that in some exceptional cases this limit will be exceeded.

Based on the above, reasonable quality guidelines for compression and delay can be established.

Next, the components of delay will be explored, first fixed delay components, then variable delay factors.



The delay components in Figure 8 are fixed in nature and add very little to delay variation.

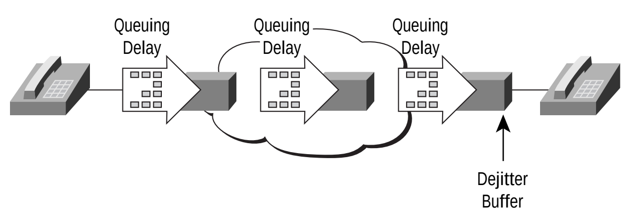
First, propagation delay is based on the total distance between source and destination. As a planning number, 6 microseconds per kilometer can be used.

Serialization is the process of placing bits on the circuit. The higher the circuit speed, the less time it takes to place the bits on the circuit. So, the higher the speed the less serialization delay. For example, it takes 125 microseconds to place one byte on a 64 Kb circuit. The same byte placed on an OC-3 circuit will take 0.05 microseconds.

Processing delays can be broken down as follows: coding, compression, decompression and decoding delay will be based on the algorithm employed. These functions can be performed in either hardware or software. Using specialized hardware such as DSPs and devoting processing power

will dramatically improve the quality and reduce the delay associated with different voice or video compression schemes. Current voice over Internet products should not be confused with the practicality of carrying voice over IP.

Packetization delay is the process of holding the digital voice samples for placement into the payload until enough samples are collected to fill the packet or cell payload. To reduce excessive packetization delay associated with some compression schemes, partial packets could be sent.

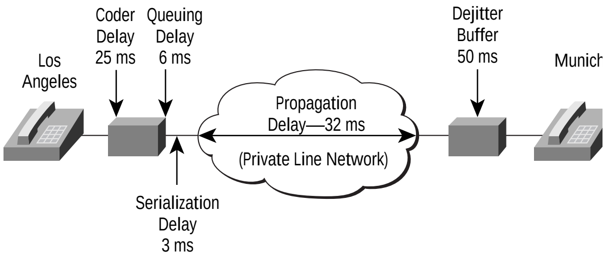


The delay components depicted in Figure 9 are variable in nature and result in higher delay variation; they are also to some degree more controllable.

Queuing delay is the delay caused by waiting on other packets to be serviced first on the trunk. At any point in time a voice packet may be waiting in queue a variable amount of time while awaiting access to the trunk. This waiting time is statistically based on the arrival of traffic; hence, when a node has more input sources, trunk contention is more likely. Waiting time is also based on the size of the packet currently being serviced.

Dejitter buffers are used at the receiving end to smooth out the delay variability and to allow for decoding and decompression. They also help on the first talk spurt to provide smooth playback of the voice. Setting these buffers too low will cause overflows and loss of data. Setting them too high will cause excessive delay. In effect, a dejitter buffer reduces or eliminates delay variation by converting it to constant delay.

Given a general understanding of the fixed and variable delay components, the delay budget can be calculated. The delay budget is the amount of delay permissible in the planned network while still meeting voice quality objectives.



	Fixed Delay	Variable Delay
Coder Delay G.729 (5 ms Look-ahead)	5 ms	
Coder Delay G.729 (10 ms per frame)	20 ms	
Packetization Delay—Included in Coder Delay		
Queuing Delay 64 kbps Trunk		6 ms
Serialization Delay 64 kbps Trunk	3 ms	
Propagation Delay (Private Lines)	32 ms	
Network Delay (e.g., Public Frame Relay Svc)		
Dejitter Buffer	50 ms	
Total	**110 ms**	

In Example 1, a delay budget of 200 ms will be used. Note that this example is of a private Frame Relay network running over leased lines. For a public service, such as public Frame Relay, the service provider's network delay figures should be used in the delay budget.

Frames will contain two 10 byte packets. The coder delay for G.729 voice compression is an initial 5 ms for a look-ahead, plus 20 ms for the two 10 byte frames.

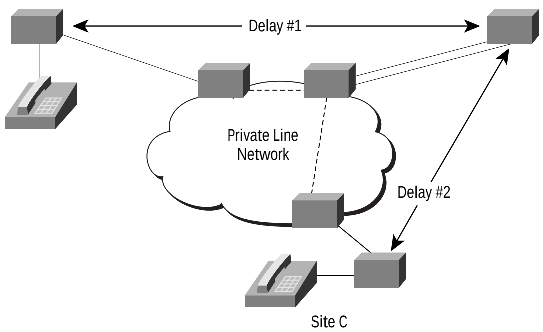
Packetization delay is the rate at which a packet is filled. This is typically governed by the speed at which voice samples are played out. Standard voice is transmitted at 64 kbps or one byte every 125 microseconds. The delay for this is included in the coder delay.

Queuing delay is variable, and with a 64 kbps trunk will equal 3 ms per 20 byte packet already in queue, assuming two voice packets in queue for a total queuing delay of 6 ms. However, this assumption will be variable, and is also cumulative based on the number of devices in the network.

Serialization delay is the time it takes to play out a 20 byte packet onto the 64 kbps trunk. The approximate propagation delay on a 64 kbps private line from Los Angles to Munich is 32 ms.

Finally, the dejitter buffer was set to twice the coder delay as a rule of thumb. This accounts for the decoding and decompression delay as well as any variable delays in the network. The variable delay in Example 1 is only 6 ms, but a public packet network such as Frame Relay or the Internet can have much higher variable delays. The size of the dejitter buffer should be configurable.

The total delay in this example is 110 ms. This is well within the delay guidelines set forth by the ITU and the target planning number of 200 ms.



	Fixed Delay	Variable Delay
DELAY #1 Total	110 ms	
DELAY #2		
Coder Delay G.729	25 ms	
Packetization Delay (Included in Coder Delay)		
Queuing Delay 2 Mbps Trunk		0.2 ms
Serialization Delay 2 Mbps Trunk	0.1 ms	
Propagation Delay (Private Lines)	5 ms	
Dejitter Buffer	50 ms	
Delay #2 Total	80 ms	
Total Delay	**190 ms**	

This next example, depicted in Figure 11, shows the delay effects of switching voice calls through a central site tandem PBX. Again, this example is of a private Frame Relay network running over leased lines. The speed of the Site A to Site B link is 64 kbps. The one from Site B to Site C is E1. The first hop is a duplicate of the first example. So the total for Delay #1 is 110 ms.

For Delay #2, the numbers are calculated in the same manner and total 80 ms. The delay is less because the distance is shorter from Site B to Site C, and because the speed of the private line is 2 Mbps versus 64 kbps. The total for both hops equals 190 ms. This is in the "acceptable provided administrations are aware" category as appears in the previous ITU recommendation chart and meets the planning delay budget of 200 ms.

This last example shows the impact of using a tandem PBX at Site B. If you were to switch the voice without having to break it out at the tandem PBX, the aggregate delay could be reduced by 75 ms, and the total delay would have diminished to 115 ms. Note the delay associated with the tandem PBX itself has not been included.

The prior section provides guidelines to help determine requirements, a method for calculating your delay budget, and technical guidelines to be used when designing an integrated voice data network. As in all designs, a balance must be struck between quality and costs. Given the large MOS quality improvements in today's compression techniques, this balance is easier than ever to strike. The guidelines can be summarized as follows:

Technical guidelines summary
- Balance voice quality, delay and bandwidth
- Determine acceptable delay and delay variation thresholds
- Calculate delay for the chosen model
- Avoid tandem (or multiple) conversions

This next section will touch on capacity planning for the integrated voice data network.

Step 5—Capacity Planning

Line trunk provisioning is establishing the number of trunks from the PBX to the integrated voice data network. After establishing the number of trunks, the next step is to translate that number to the required network bandwidth.

The correct number of PBX or key system trunks will be determined by traffic volume and flow, the selected grade of service or blocking factor and, other network-specific objectives. Some organizations will simply move trunks from the current network to the integrated network. Others will take this opportunity to update the traffic engineering information and conduct a traffic study. Either approach can work and is very familiar territory for voice engineering professionals.

The use of the transport or translational model can have an impact on the number of trunks simulated by the network. The transport network model matches a virtual connection for a tie line on a one-for-one basis. From a voice engineer's standpoint, nothing has changed. However, the translate model uses the network to simulate a tandem PBX, thereby, potentially reducing the number of trunks required. Routing of calls over specific trunk groups in which one group could be the integrated voice data network is left to the voice engineer. One engineer may choose to use the network as the first option while another may use it as the last choice.

Based on the proposed network design and the required number of trunks between locations, the required bandwidth can be calculated. Bandwidth calculations should take into account compression, overhead, and utilization. Each of these will vary, depending on which voice over data technology is chosen. Then calculate the delay matrix between locations and confirm that the delay calculations meet the delay budget requirements. If they do not, adjust the bandwidth, or select a different voice over data technology.

Step 6—Financial Analysis

The objectives have been set, the packet voice technology or technologies chosen, the capacity planning completed and the trunks sized properly to support the additional traffic within the delay budget. Now, we must ask: is the network cost justified?

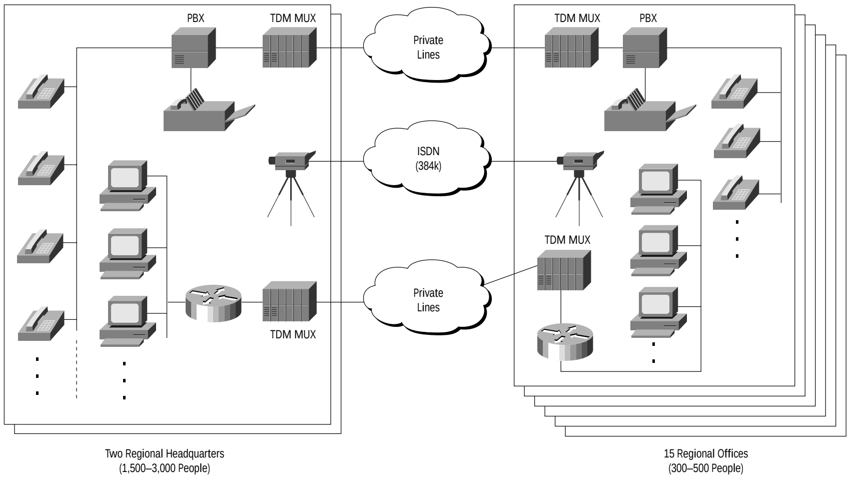
Three different case studies are presented in this section to illustrate the financial benefits of the various solutions. The cases include a large global organization using a private cell-based network, a medium size regional company using a public Frame Relay service, and a small international firm using a private line network with routers.

Each case is built using standard voice engineering principles and actual PSTN rates and public Frame Relay or private line costs. Because these costs differ around the world, Case 2 and Case 3 are duplicated for different regions. The traffic volume, patterns and assumptions have been held constant. Only the regional PSTN, Frame Relay and/or private line prices change within each case. Therefore, rather than reiterate the same background information for each region, each case is presented once in its entirety. Then the other versions are summarized in Appendix 1. The cases described are:

Case 1:A division of a large global organization using a private cell-based network
Case 2:A medium-size Pan-European company based in Brussels using a public Frame Relay service
Case 2A:The same firm based in Singapore using a Pan-Asian public Frame Relay service (refer to Appendix 1)
Case 3:A small international firm based in Paris using a private network with routers and leased lines
Case 3A:The same firm based in San Jose California, USA (refer to Appendix 1)
Case 3B:The same firm based in Sao Paulo, Brazil (refer to Appendix 1)
Case 3C:The same firm based in Singapore, (refer to Appendix 1)

CASE 1

Geographic Scope: Europe, North America
Headquarters: Amsterdam, The Netherlands
Network: Private ATM Network



The division of this global firm has two large offices and 15 somewhat smaller offices connected via a combination of E1, T1 and fractional private lines. For historical reasons, they have two separate, private, TDM-based networks, one transporting data traffic and the other transporting voice traffic. Additionally, a relatively new requirement, conference room video conferencing, has been implemented using the public ISDN network at 384 kbps rates. The original design is shown in Figure 12.

The objective is to reduce costs by merging the three networks onto one multiservice network as efficiently and cost effectively as possible while maintaining voice quality. A second goal is to reduce the reliance on the PBXs for tandem switching. This would avoid the decompression/recompression cycle associated with the tandem PBX thereby allowing higher voice compression ratios and resulting in further reductions in bandwidth expenses.

After reviewing the different voice over data alternatives, an ATM network was selected because of the large capacity requirements, the video conferencing needs, and the intent for the network to participate in voice switching.

Maintaining separate networks is very costly, particularly, when the networks are very large. Focusing only on bandwidth costs, Tables 4 and 5 show the voice network and data network expenses, respectively. The costs of the ISDN network were not captured for this firm. Consequently, those savings are not shown in this case, although the redesigned network was sized to carry the video traffic.

Table 4 Case 1 Voice Backbone Configuration and Expense

Source	Destination	Qty	Speed	Monthly Expense
London	Manchester	1	FT768K	$4,460
Paris	Lyon	1	FT384K	$6,516
Frankfurt	Cologne	1	F-1920	$10,314
Frankfurt	Munich	1	E1	$9,156
Milan	Rome	1	FT512K	$13,338
London	Frankfurt	1	T1	$40,452
London	Philadelphia	1	E1	$47,950
London	Toronto	1	FT512K	$21,822
London	Dublin	1	FT256K	$8,276
Amsterdam	Paris	1	FT512K	$21,672
Amsterdam	Zurich	1	FT512K	$17,179
Amsterdam	Philadelphia	1	F-198	$48,581
Amsterdam	Rome	1	FT768K	32,837
Amsterdam	Manchester	1	FT768K	$22,374
Amsterdam	Cologne	1	FT768K	$19,228
Amsterdam	Budapest	1	FT512K	$29,001
Amsterdam	Oslo	1	FT384K	$12,573
Frankfurt	Milan	1	FT768K	$30,967
Frankfurt	Copenhagen	1	FT384K	$17,377
Frankfurt	Zurich	1	FT384K	$17,362
Frankfurt	Lyon	1	FT512K	$20,259
Copenhagen	Oslo	1	FT256K	$5,275

Source	Destination	Qty	Speed	Monthly Expense
Philadelphia	Toronto	1	FT512K	$8,829
Manchester	Dublin	1	E1	$22,093
Budapest	Munich	1	FT768K	$19,228
TOTAL	-	25	-	$507,119

Table 5 Case 1 Data Backbone Configuration and Expense

Source	Destination	Qty	Speed	Monthly Expense
Cologne	Frankfurt	1	F-1920	$10,314
Cologne	Munich	1	F-1920	$15,158
Paris	Lyon	1	FT256K	$4,117
Frankfurt	Munich	1	E1	$9,156
Milan	Rome	1	FT256K	$8,927
Amsterdam	Philadelphia	1	F-1984	$38,581
Amsterdam	London	1	FT512K	$17,281
Amsterdam	Copenhagen	1	FT256K	$10,130
Amsterdam	Zurich	1	FT512K	$17,179
Amsterdam	Philadelphia	1	F-1984	$48,581
Amsterdam	Rome	1	FT256K	$15,785
Amsterdam	Manchester	1	FT512K	$17,281
Amsterdam	Budapest	1	FT512K	$29,001
Amsterdam	Oslo	1	FT256K	$8,867
Philadelphia	Toronto	1	T1	$12,277
Cologne	Paris	1	FT256K	$12,269
Cologne	Zurich	1	F-1920	$41,382
Cologne	Lyon	1	FT256K	$12,269
London	Dublin	1	FT256K	$8,276
Milan	Munich	1	FT256K	$18,017
Copenhagen	Oslo	1	FT256K	$5,275
Philadelphia	Toronto	1	T1	$12,277
Manchester	Dublin	1	E1	$22,093
Budapest	Munich	1	FT768K	$19,228

Source	Destination	Qty	Speed	Monthly Expense
TOTAL	-	24	-	$413,721

By redesigning the network, the different types of traffic can be consolidated onto one redundant infrastructure based on ATM cell switching.

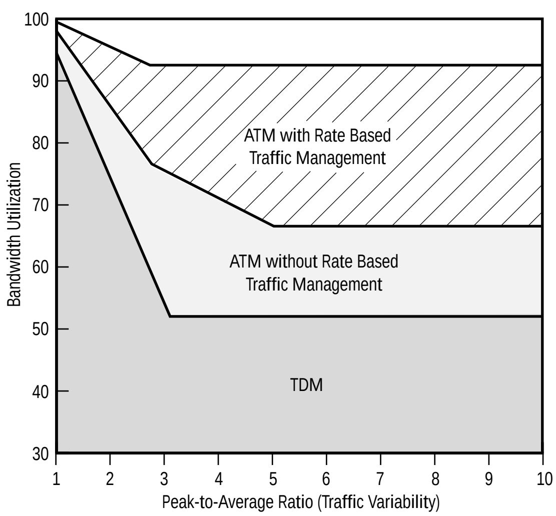
Network Redesign

The redesign takes advantage of the efficiencies an ATM network affords. Plus, voice compression and closed-loop congestion traffic management are implemented to maximize bandwidth utilization. Other techniques are also employed such as Voice Activity Detection (VAD) to further increase utilization. Given the efficiency gained by deploying this network, voice traffic essentially rides for free.

Numerous port types and speeds are now supported across the network, providing flexibility to deploy new applications to users quickly. The next section looks more closely at some of the new functions of the integrated network.

Customer Premises Equipment

In each office, a Cisco Stratacom IGX switch was installed and connected to the video conferencing equipment, the PBX and the local area network.



The bandwidth utilization advantages that cell switched networks have over TDM networks are well known. TDM use only 50-60% of the line bandwidth. ATM alone increases that to about 65%. But ATM with rate-based traffic management achieves utilizations over 90%. This is illustrated in Figure 13.

Rate-based traffic management is a function of the Cisco Stratacom switches. The switch monitors the network for congestion every 40 milliseconds and adjusts the data admission rate to a speed the network can support. This capability helps prevent network congestion from occurring, and serves to minimize cell discards and retransmissions. Rate-based traffic management ensures optimal bandwidth utilization.

Typically, up to 60% of an average 64 kbps voice channel is silence, as we listen and pause between sentences. By implementing Voice Activity Detection (VAD) to detect the presence of speech, this bandwidth is recaptured for other use. If a cell does not contain speech, it is not transmitted through the network.

Financial Analysis

The capital costs for the redesigned ATM network are $1,205,000 including equipment installation. Table 6 shows the monthly bandwidth costs for the redesigned network.

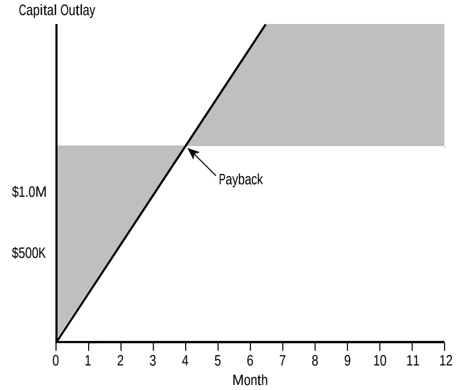
Table 6 Case 1 Redesigned ATM Network Expenses

Source	Destination	Qty	Speed	Monthly Expense
Paris	Lyon	1	FT768K	$11,835
Frankfurt	Munich	1	F-1920	$12,776

Source	Destination	Qty	Speed	Monthly Expense
Milan	Rome	1	FT384K	$19,124
Amsterdam	Philadelphia	1	F-1984	$48,581
Amsterdam	Zurich	1	F-1984	$44,952
Amsterdam	Rome	1	FT512K	$35,993
Amsterdam	Manchester	1	FT768K	$22,374
Amsterdam	Budapest	1	F-1024	$23,456
Amsterdam	Oslo	1	F-1984	$31,357
Philadelphia	Toronto	1	T1	$12,277
Cologne	London	1	F-1024	$38,073
Cologne	Paris	1	F-1024	$32,831
Cologne	Milan	1	FT512K	$32,082
Cologne	Copenhagen	1	F-1920	$41,578
Cologne	Zurich	1	F-1920	$41,382
London	Toronto	1	T1	$44,090
London	Dublin	1	FT768K	$16,412
Frankfurt	Lyon	1	FT768K	$27,054
Copenhagen	Oslo	1	T1	$17,280
Manchester	Dublin	1	E1	$22,093
Budapest	Munich	1	F-1920	$44,680
TOTAL	21	-	-	$620,280

By saving $300,000 per month, this firm will pay for the equipment in about four months. This fact is illustrated in Figure 14.

Figure 14: Case 1 Financial Summary



By redesigning the network to carry voice, video and data, and implementing special functions in the network, this firm reduced their costs by more than 30% while preparing for future growth. By combining the networks, the original annual network costs dropped from $11 million dollars to about $7.5 million.

Voice Network Switching

Figure 15 illustrates a traditional network where a PBX is used to route voice traffic through a TDM network. This approach is inefficient because statically defined routes must be used. It also requires more PBX and network interface hardware and may in fact require a dedicated PBX solely for switching voice calls. Also, voice compression alternatives are limited because the tandem PBX introduces a decompression/recompression cycle that negatively impacts voice quality. This limits the amount of bandwidth cost savings that are achievable

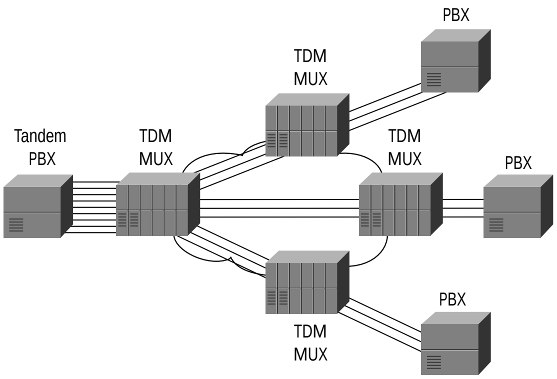


Figure 15: Case 1 Tandem PBX.

As shown in Figure 16, Cisco Stratacom's Voice Network Switching could be implemented on the new ATM network to reduce the reliance on tandem PBXs. This dramatically reduces PBX hardware requirements, simplifies PBX network topology, and increases the dollar amount of bandwidth savings. It also maintains voice quality and PBX feature transparency throughout the network. If VNS is implemented in this network an additional $42,000 could be saved.

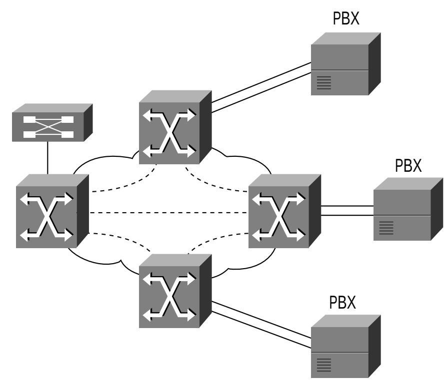


Figure 16: Case 1 Voice Network Switching

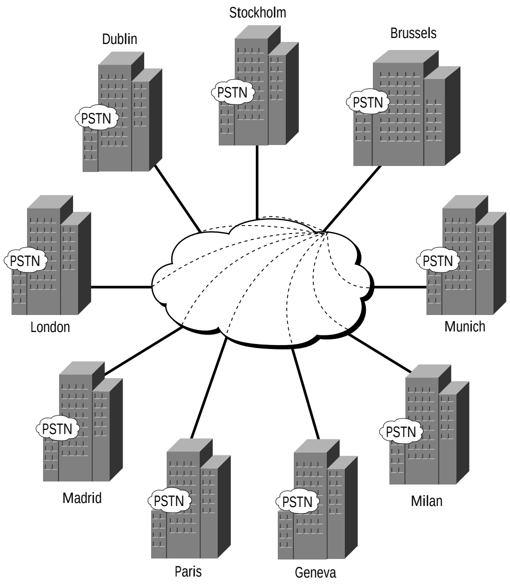
CASE 2

Geographic Scope: Pan-European

Headquarters: Brussels, Belgium

Network: Public Frame Relay Service

This medium-size firm has its headquarters in Brussels. Its eight branch offices are in Stockholm, Milan, London, Madrid, Paris, Munich, Geneva, and Dublin. Each branch has between 45 and 70 people. Figure 17 shows the Frame Relay network.



The majority of branch calls are between branch employees and local outside customers. Only about 20% of the total call volume is between headquarters and branch employees. Although the branch-to-headquarters traffic volume is lower, it is also the most expensive on a per-minute basis because it is billed at international rates. As a result of these high rates, this firm pays more than $50,000 per month or about $600,000 per year for international long distance services.

Voice Network

Voice services are provided by small PBXs connected over the public switched telephone network (PSTN). To ensure the financial analysis is conservative, it is assumed that the firm's call volume is large enough to obtain a VPN contract from a carrier at about a 15% discount from standard PSTN rates.

Each individual in the branch offices spends about one and a half hours per day communicating via telephone or fax. A bit more than a quarter of an hour of this traffic is back and forth between the branch and headquarters. Table 7 shows the potential on-net voice and fax traffic volume and expense.

Location	Purpose	Number of People	Average Minutes per Person per Day	On-Net % to HQ	Workdays per Month	Total Minutes per Person per Month	Total Minutes per Office per Month	Cost per Minute	Monthly Cost per Office
Brussels	Headquarters								
Stockholm	Branch	45	90	20%	21.67	390	17,553	$0.39	$6,846
Milan	Branch	55	90	20%	21.67	390	21,453	$0.34	$7,294
London	Branch	50	90	20%	21.67	390	19,503	$0.23	$4,486
Madrid	Branch	40	90	20%	21.67	390	15,602	$0.39	$6,085
Paris	Branch	70	90	20%	21.6	390	27,304	$0.26	$7,099
Munich	Branch	70	90	20%	21.67	390	27,304	$0.26	$7,099
Geneva	Branch	65	90	20%	21.67	390	25,354	$0.35	$8,874
Dublin	Branch	45	90	20%	21.67	390	17,553	$0.36	$6,319

Location	Purpose	Number of People	Average Minutes per Person per Day	On-Net % to HQ	Workdays per Month	Total Minutes per Person per Month	Total Minutes per Office per Month	Cost per Minute	Monthly Cost per Office
Total	-	-	-	-	-	-	171,626	-	$54,101
Assumptions **1. This is the average cost per minute and assumes 50% of calls are to HQ and 50% from HQ.** **2. Cost of a voice call based on carrier quote, off-net pricing with discount.**									

Data Network

The firm is using a public Frame Relay service. The topology is illustrated in Figure 17. The exact configuration of the network and the on going monthly expenses, or run-rate, are shown in Table 8. Note there is a single PVCs between each branch and headquarters.

Table 8 Case 2 Initial Network Configuration

Location	Purpose	Access Line Speed	Initial Port Speed	Initial PVC CIR	Frame Relay Charges
Brussels	Headquarters	E-1	768	-	$2,800
Stockholm	Branch	E-1	128	64	$2,690
Milan	Branch	E-1	128	64	$3,010
London	Branch	E-1	128	64	$3,010
Madrid	Branch	E-1	128	64	$3,170
Paris	Branch	E-1	128	64	$3,010
Munich	Branch	E-1	128	64	$3,010
Geneva	Branch	E-1	128	64	$3,110
Dublin	Branch	E-1	128	64	$3,110
Total	-	-	-	-	$26,920
The Port cost and the PVC cost are included in the Frame Relay Charges column.					

Any of the packet voice technologies could be used in this case to build a multiservice network. However, given their current infrastructure and familiarity with Frame Relay, a Voice over FR network is chosen.

Network Redesign

The objective is to redesign the data network to support the added voice traffic while not adversely affecting its performance.

The first step is to determine the additional bandwidth required on the data network to support the voice and fax traffic. This can be done in one of two ways. The best way is to collect traffic information from both the key system or PBX and the router, and graphically add the voice and data traffic together to see how often the combined voice and data traffic exceeds the current available bandwidth. However, this kind of traffic information is often unavailable or hard to obtain. If this information is not available, the easiest method to determine the proper upgrade is to estimate whether (and how much) extra bandwidth is required, provision that, and then add the voice traffic to the data stream while tracking two measures of performance: user reported voice quality, and data latency. If either performance measure appears to be suffering, then more bandwidth should be added.

Often, data and voice traffic will peak at different times during the day. Consequently, the data network will frequently benefit from the added bandwidth.

The network was redesigned considering the following assumptions.

- The total voice and fax call volume per person equals about 90 minutes per day.
- The call volume between headquarters and branch personnel represents about 20% of the total call volume or about 1/4 hour.
- A busy hour loading factor of 17% is appropriate.
- The Cisco voice compression module uses 8 kbps plus 3 kbps overhead (11 kbps per voice channel); thus, each 64 kbps trunk supports 5 voice channels.
- The branch PBXs require additional trunk modules.

The amount of voice and fax traffic at each branch office that would optimally be diverted from the PSTN to the multiservice network is determined using the above assumptions and the appropriate traffic engineering tables, given the desired P grade of service. This firm chose a P.05 grade of service. Table 9 summarizes the calculations and provides the number of PBX trunks that would be required at each site.

Number of Users	Number of Sites	Hours per Day on Phone	Minutes per Day	Busy Hour 17%	Minutes per Busy Hour	% of Traffic to HQ	Total Erlangs to HQ	Required Lines for.05 Blocking Probability
40	1	1.5	3,600	0.17	612	20%	2.04	6
45	2	1.5	4,050	0.17	689	20%	2.30	6
50	1	1.5	4,500	0.17	765	20%	2.55	6
55	1	1.5	4,950	0.17	842	20%	2.81	7
60	0	0	-	0	-	0%	0.00	0
65	1	1.5	5,850	0.17	995	20%	3.32	8
70	2	1.5	6,300	0.17	1,071	20%	3.57	8
-						Total Erlangs for HQ	22.44	31

The next step is to determine the additional bandwidth required on the Frame Relay network to support the voice and fax traffic. As indicated in the assumptions, each additional 64 kbps of bandwidth will provide a minimum of 5 voice channels.

Network Upgrades

Each branch office was originally linked to the Frame Relay network via a 128 kbps Frame Relay port (using an E1 access line), with the PVC CIR set to 64 kbps. This was more than sufficient to handle the original data traffic. However, it was not enough to carry the added compressed voice traffic as well, and the bandwidth at each location had to be upgraded. Rather than conduct traffic studies to determine the minimum upgrades each location would require, it was felt that upgrading all locations to the same high bandwidth level would provide a conservative approach, and leave room for future growth in either voice or data traffic.

The largest branch offices, containing 70 people, were reengineered to direct a maximum of 8 voice lines of traffic over the Frame Relay network. At its maximum, this data stream could reach 88 kbps (since each voice line can load a maximum of 11 kbps). The branch port speeds were doubled from 128 kbps to 256 kbps, far more than the maximum added voice load, to provide improved performance (decrease buffer delay at the port) for the more sensitive voice traffic, and to leave room for future growth. The PVC CIR was increased from 64 to 128 kbps to ensure that delay due to vendor network congestion would be minimized (although as always with a Frame Relay network, close watch must be kept on latency measurements to see that the PVC CIR is set properly).

This added Frame Relay bandwidth is also certain to improve current data traffic performance. The voice traffic, even at it's maximum, still takes up less bandwidth than the new doubled port speed delivers. And voice traffic, being exceptionally variable over time, leaves the new doubled Frame Relay pipe open for more and faster data traffic for most of the day. While the voice traffic at its peak adds an extra 88 kbps load to the Frame Relay circuit, on average it adds less than 30 kbps of data throughout the day. The rest of the added 128 kbps bandwidth is available for improved data performance.

If more savings is required, the smaller branch locations' circuit and port speed could be upgraded in a smaller increment. Half the branches under consideration needed only 5 or 6 voice circuits to be added to the data stream; adding an additional 64 kbps to the port size (upgrading from 128 kbps to 192 kbps) would have been more than sufficient.

As long as the business case could support it, however, it was preferred to upgrade the network once, and then reduce bandwidth, and expenses, over time as experience is gained with the new traffic patterns.

Customer Premises Equipment

The Cisco Multiservice Access Concentrator, the MC3800, was installed in the branches and the headquarters. As can be seen in Table 10, the port speeds and the CIR of the PVCs were increased to accommodate the voice traffic. Adding a separate PVC for the voice was unnecessary because the MC3800 prioritizes the traffic on a single PVC. This capability avoided the cost of another PVC.

Each branch office connected the appropriate number of PBX trunks to the Cisco MC3800. The PBX can direct approximately 95% of traffic destined for headquarters preferentially to one of the trunks connected to the Cisco MC3800. The remaining overflow traffic, an estimated 5%, is directed to the PSTN.

Using G.729 encoding, the MC3800 compresses each 64 kbps voice channel to a data stream of approximately 11 kbps (8 kbps or less plus 3 kbps overhead) and forwards the compressed voice traffic over the Frame Relay network. The 11 kbps is a conservative measure because it does not include the benefits of the silence suppression techniques implemented in the MC3800.

The MC3800 is capable of supporting up to 24 channels of 8 kbps compressed voice that can be transported over public or private Frame Relay, ATM or leased line networks.

A number of different network designs are available to this firm. One option is to add a second PVC to transport the voice traffic. This would avoid to some degree the potential contention between the two different traffic types. However, it would also result in higher expenses because public Frame Relay providers commonly charge for each PVC. A less expensive approach would be to utilize the same PVC for both data and voice traffic. This design is available when using the Cisco MC3800 because it can prioritize the delay-sensitive traffic, voice, while ensuring the delay-insensitive traffic, data, is also serviced properly.

The Cisco Internetwork Operating System (Cisco IOS™) technology provides voice switching and advanced call management. For example, the headquarters MC3800 can be configured to switch calls between branches. Although the call volume may be low, the per minute cost, in this case, is very high. By off-loading this tandeming function from the PBX, the delay budget is reduced and the quality maintained by avoiding the compression/decompression cycle required by the PBX. This additional savings was not included in the following financial analysis.

The MC3800 can support E1/T1 ATM services via a software change as these services become available. Plus, the CBR and VBR support in the unit prepares this firm for video conferencing when it becomes required.

The costs and savings resulting from the network redesign are summarized in the next section.

Financial Analysis

The cost to upgrade the network to support the additional voice traffic is shown in Table 10.

Location	Purpose	Access Line Speed	Initial Port Speed	Initial PVC CI	Frame Relay Charges	Upgraded Port Speed	Upgraded PVC CIR	Frame Relay Charges	Cost to Upgrade
Brussels	Headquarters	E-1	768		$2,800	2 x 1024	-	$14,300	$11,500
Stockholm	Branch	E-1	128	64	$2,690	256	128	$4,958	$2,268
Milan	Branch	E-1	128	64	$3,010	256	128	$6,238	$3,228
London	Branch	E-1	128	64	$3,010	256	128	$4,958	$1,948
Madrid	Branch	E-1	128	64	$3,170	256	128	$6,238	$3,068
Paris	Branch	E-1	128	64	$3,010	256	128	$4,958	$1,948
Munich	Branch	E-1	128	64	$3,01	256	128	$4,958	$1,948
Geneva	Branch	E-1	128	64	$3,110	256	128	$6,238	$3,128
Dublin	Branch	E-1	128	64	$3,110	256	128	$6,238	$3,128
Total	-				$26,920	-		$59,084	$32,164
The cost of the second E1 access circuit in Brussels is included in the upgrade cost.									

The capital costs for the branches and the headquarters are shown in Table 11. The required additional bandwidth indicated in the preceding table costs $32,000 per month. Comparing the savings to these additional expenses, Table 12 illustrates the net monthly savings

Table 11 Case 2 Capital Costs

Cisco MC3800s	$72,000
Branch PBX trunk modules	$5,400
Number required	8
Total	$43,200
HQ PBX trunk module	$12,300
Number required	1
Total	$12,300
Total Capital Cost	$127,500

Table 12 Case 2 Savings and Payback

Monthly PSTN voice expenses	$54,101
Multiply by 95% (P.05)	95%
Monthly PSTN voice savings	$51,396
Required added bandwidth	$(32,164)
Net total monthly savings	$19,232
Net annual savings	$230,787
Total capital costs	$127,500
Installation	$20,000
Total Capital Costs	$147,500
Payback Period (Months)	7.7

This firm saves $230,000 per year by moving on-net voice traffic onto the Frame Relay network. The payback period is under eight months. Figure 18 summarizes the important numbers.

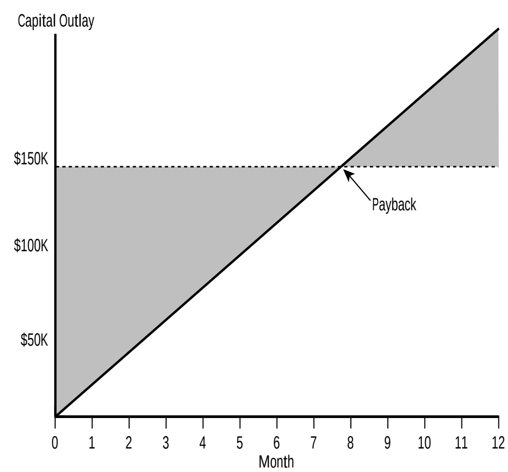


Figure 18 Case 2 Financial Summary

Cost per Minute

An organization's voice cost per minute is a common expense measurement. Table 7 provides the details required to calculate the original cost per minute. The total on-net minutes per month for all offices equals 171,626 minutes and the total monthly cost is $54,101. Dividing total cost by total minutes yields an average cost per minute of $0.32 for all calls between headquarters and the branches. To carry 95% of this traffic over the data network, that network must be upgraded at a cost of $32,164 per month. Ninety-five percent of 171,626 equals 163,044 minutes. Dividing the upgrade costs by this amount yields an average cost per minute of $0.20. In this case, the average cost per minute was reduced from $0.32 to $0.20, a 38% reduction.

In Appendix 1, this case is modified to illustrate the solution in Asia. The modified case is labeled Case 2A.

CASE 3

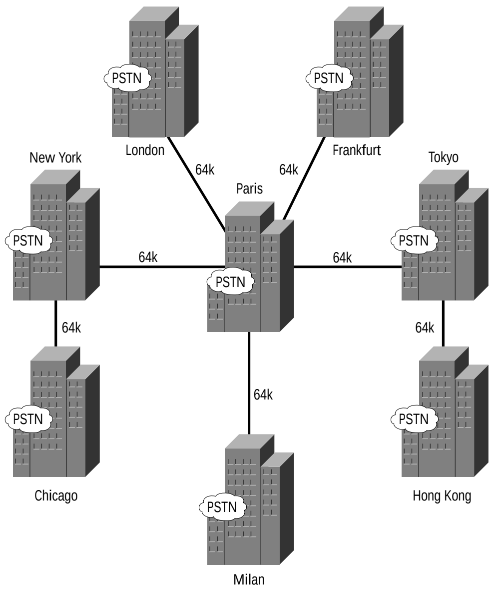
Geographic Scope: International

Headquarters: Paris, France

Network: Private Lines

With its headquarters in Paris, this firm has seven branch offices. There are about fifteen people in the branch offices except for London and New York where there are approximately 45. The network topology and leased-line circuit speeds are shown in Figure 19.

Figure 19 Case 3 Initial Network Topology



Most branch calls are between branch employees and customers in the local area. Calls between headquarters and branch employees account for only 20% of the total call volume. Although the branch-to-headquarters traffic volume is in the minority, it is also the most expensive on a per-minute basis because it is billed at international rates. Consequently, for international long distance services, this firm pays approximately $38,000 per month, or about $456,000 annually.

Voice Network

Key systems and small PBXs connected over the public switched telephone network (PSTN) provide voice services. In the analysis that follows it is assumed that the firm generates enough call volume to obtain a VPN contract from a carrier at about a 15% discount from standard PSTN rates. This is to ensure the financial analysis is conservative.

Headquarters is connected to the PSTN by a number of E1 circuits. Voice traffic between headquarters and branches is carried by one E1.

In the branch offices, each individual spends approximately two and a half hours each day communicating via telephone or fax. About 20% of this traffic is between the branch and headquarters.

Table 13 shows the potential on-net voice and fax traffic volume and expense.

Location	Purpose	Number of People	Average minutes per Person per Day	On-Net % to HQ	Workdays per Month	Total Minutes per Person per Month	Total Minutes per Office per month	Cost per Minute(1)	Monthly Cost per Office
Paris	Headquarters	-							
Frankfurt	Branch	15	150	20%	21.67	650	9,752	$0.25	$2,438
Milan	Branch	15	150	20%	21.67	650	9,752	$0.27	$2,633
London	Branch	45	150	20%	21.67	650	29,255	$0.23	$6,729
New York	Branch	45	150	20%	21.67	650	29,255	$0.33	$9,508
Chicago	Branch	15	150	20%	21.67	650	9,752	$0.33	$3,169
Tokyo	Branch	15	150	20%	21.67	650	9,752	$0.70	$6,826
Hong Kong	Branch	15	150	20%	21.67	650	9,752	$0.78	$7,606
Total	107,267	$38,908	-						
Assumptions 1. This is the average of the cost per minute in each direction and assumes 50% of calls are to HQ and 50% from HQ. 2. Cost of a voice call based on carrier quote with customer discount.									

Data Network

An eight-node data network, leased by the firm, utilizes routers, and is hubbed out of their Paris headquarters. Two of the seven branch locations connect through other branches to reach the headquarters location. This arrangement was set up to hold down leased line costs. At headquarters an E1 handles the aggregate data circuits (320k) from all the branches.

Network Redesign

It is essential that redesign of the data network to support the added voice traffic be accomplished without adversely affecting performance. The plan is to have the PSTN cost reductions pay for the redesign. Though any of the packet voice technologies could be used to build this multiservice network, given the firm's infrastructure and expertise in IP, a Voice-over-IP network is chosen. Considering the assumptions, the following redesign is conservative.

First, the additional bandwidth required on the data network to support the voice and fax traffic is determined. As indicated in Case 2, there are two approaches to doing this. The best way is to collect traffic information from both the key system or PBX and the router, and then graphically add the voice and data traffic together. This enables one to see how often the combined voice and data traffic would exceed the available bandwidth. But this kind of traffic information is often unavailable. If this information is not available, the easiest method to establish the proper upgrade would be to estimate whether (and how much) extra bandwidth would be required, and provision that amount. Then the voice traffic can be added to the data stream while tracking two measures of performance: user reported voice quality and data latency. If either performance measure appears to be insufficient, then more bandwidth should be added.

Data and voice traffic frequently peak at different times in the day. The data network will frequently benefit from the added bandwidth.

In redesigning the network the following assumptions were used:
- There are approximately 15 people per small branch, 45 per large branch.
- The bidirectional voice and fax call volume totals about 2.5 hours per person per day per branch.
- About 20% of the total call volume is between headquarters and each branch location.
- A busy hour loading factor of 17% is appropriate.
- The Cisco voice compression module uses 8 kbps, plus 1 kbps overhead per call. It was assumed that a 64 kbps trunk circuit supports 5 calls, rather than 7. This is a conservative estimate.
- In the small branches, one key system trunk module would be required, whereas two cards would be necessary for the large branches.

Using the above assumptions and the following calculations, the amount of voice and fax traffic at each branch office that would optimally be diverted from the PSTN to the multiservice network is as follows:
- 2.5 hours call volume per user per day X 15 users = 37.5 hours daily call volume per office
- 37.5 hours X 60 minutes per hour = 2,250 minutes per day
- 2,250 minutes X 17% (busy hour load) = 382.5 minutes per busy hour
- 382.5 minutes per busy hour X 1 Erlang/60 minutes per busy hour = 6.375 Erlangs
- 6.375 Erlangs X 20% of traffic to headquarters = 1.275 Erlangs volume proposed

To determine the appropriate number of trunks required to carry the traffic, traffic engineering tables are consulted next, given the desired P grade of service. This firm chose a P.05 grade of service. Shown in Table 14 are the applicable sections of the Erlang C tables:

Table 14 Case 3 Erlang C Table

Blocking Probability (Grade of Service)	Small Branch Traffic to HQ (1.275 Erlangs)	Large Branch Traffic to HQ (3.825 Erlangs)
P = 0.01	5 trunks	10 trunks
P = 0.05	4 trunks	8 trunks
P = 0.10	3 trunks	7 trunks
P = 0.20	3 trunks	6 trunks

Using the calculated Erlangs and Table 14 it turns out that four trunks are required in the smaller offices and eight trunks in the larger ones. In the following table the calculations and trunking requirements are summarized, along with the figures for the larger branch office.

The London and Frankfurt private lines were increased from 64 kbps to 128 kbps to support the added voice traffic based on the above conclusions. Since the circuit from New York to Paris must carry, at a maximum, the 4 compressed voice streams from Chicago and the 8 compressed voice streams from New York, it is increased to 192 kbps. At 9 kbps apiece, this added traffic at its maximum would be 12 voice streams x 9 kbps/voice streams = 108 kbps. This still leaves 84 kbps left over for the data traffic when all voice circuits are busy. Since most of the time some of the voice circuits would be unused, there would often be more bandwidth available for data traffic, and the added bandwidth would provide enhanced data performance.

Site	Number of Users	Call Volume per Day	Minutes per Day	Busy Hour (17%)	Minutes per Busy Hour	% of Traffic to HQ	Total Erlangs to HQ	Required Trunks for 0.05 Blocking Probability
Large	45	2.5	6,750	0.17	1,147.5	20%	3.825	8
Small	15	2.5	2,250	0.17	382.5	20%	1.275	4
-	-	-	-	-	-	2 Large	7.65	-
-	-	-	-	-	-	5 Small	6.375	-
-	-	-	-	-	-	Total Erlangs for HQ	14.025	23

The Chicago-to-New York circuit was not increased beyond its original 64 kbps. Even at its maximum, the added Chicago voice traffic can add no more than 36 kbps of traffic onto the data pipe, throttling the data traffic down to a minimum of 28 kbps. Like Chicago, many locations will find that their data traffic can tolerate the added delay created by this occasional reduction in bandwidth. The peak hours for voice calls frequently differ from the peak data transfer times, and the two sets of traffic interfere with each other only rarely, as mentioned previously.

The Hong Kong-to-Tokyo link did not require an upgrade from its original 64 kbps as is the case for the Chicago-to-New York link. The combined voice traffic from Tokyo to Paris (4 channels from Hong Kong, 4 more from Tokyo) required a maximum of 72 kbps (and usually used less); so, only the Tokyo circuit was increased to 128 kbps. This leaves a minimum of 56 kbps for data traffic, and during less than peak voice calling moments, it offers more than the original 64 kbps for improved data performance. The redesigned network appears in Figure 20.

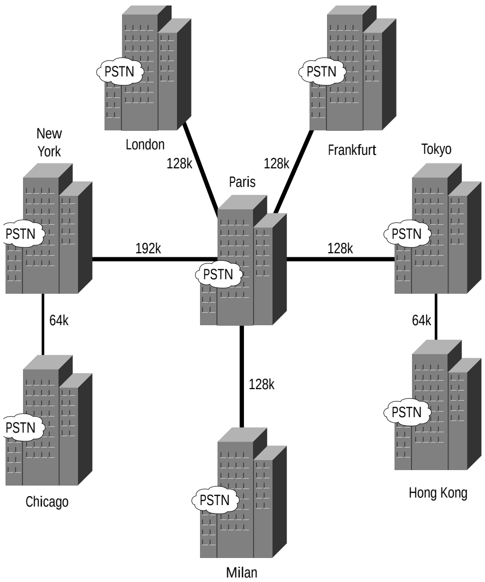


Figure 20 Case 3 Redesigned Topology

The additional bandwidth is not free. Shown in the following table are the incremental expense details. The complete financial picture is analyzed in the following section

Paris to:	Original 64 kbps (Monthly Cost)	Redesigned 128 kbps (Monthly Cost)	Redesigned 192 kbps (Monthly Cost)	Incremental Cost (Monthly Cost)
Frankfurt	$5,200	$7800	-	$2,600
Milan	$7,000	$12,600	-	$5,600
London	$4,100	$7,300	-	$3,200
New York	$4,800	-	$9,400	$4,600
Tokyo	$10,400	$16,450	-	$6,050
Chicago to NY	$800	-	-	-
Hong Kong to Tokyo	$5,500	-	-	-
TOTAL	-			$22,050

Customer Premises Equipment

A Cisco 3620 router was installed in the smaller branches and a Cisco 3640 router was put in place in the two larger branches. At each of the smaller branch office, four key system FXO trunks were connected to the Cisco 3620 router (eight trunks to the Cisco 3640 in the larger branch locations). The reprogrammed key system would then direct approximately 95% of traffic destined for headquarters preferentially to one of the trunks connected to the Cisco router. The remaining overflow on-net voice traffic, an estimated 5%, is directed to the PSTN.

The leased lines terminate at the Paris Headquarters, where the voice channels are decompressed and then routed to the headquarters PBX. The Paris headquarters can remove one of the PSTN E1 access lines because 23 channels (nearly a full E1 worth of traffic) have been removed from the PSTN and are now sent over the router network.

The Cisco 3600s transport the voice calls over the IP network compressed at 8 kbps, excluding overhead, using the G.729 CS-ACELP algorithm. Each voice channel utilizes a dedicated digital signal processor (DSP) to perform encoding and compression. The design of the 3600 and the dedicated DSPs enables the high performance and low latency which assure very high voice quality.

To reliably deliver the real-time voice traffic, Cisco's Internetworking Operating System (IOS) employs a number of techniques. Resource Reservation Protocol (RSVP) reserves bandwidth when the remote phone number is dialed. Compressed Real Time Protocol (CRTP) compresses the overall header, thereby keeping both overhead low and payload throughput high.

It turns out that approximately 50% of normal voice conversation is silence. By not transmitting this silence, bandwidth is available for the data traffic. The Cisco IOS software uses sophisticated, silence-suppression techniques to realize this savings. So that the receiver remains assured that the call remains connected, comfort noise is generated locally.

In the future, this firm's networking plans include a possible migration to Frame Relay and telecommuting support. Commuting to work can be very difficult in Hong Kong and New York. Therefore, the Cisco 3600 routers will eventually be used as access servers for employees who will telecommute from their homes. If the firm decides to transition to Frame Relay, the 3600 supports that protocol.

Note that not all key systems provide automatic route selection for preferential voice traffic switching. If plans call for this type of configuration, contact the key system or PBX vendor to confirm that this capability exists.

In the next section the costs and savings resulting from the network redesign are summarized.

Financial Analysis

Shown in Table 17 are the capital costs for the branches and the headquarters. The required additional bandwidth indicated in the preceding section costs $22,050 per month. Comparing the savings to these additional expenses, the net monthly savings are illustrated in Table 18.

Table 17 Case 3 Capital Costs

Cisco 3620 &3640 routers	$84,000
Branch key system modules	$700
Number required (1/small branch; 2/large branch)	9
Total	$6,300
HQ PBX trunk modules	$5,418
Number required	1
Total	$5,418
Total Capital Cost	$95,718

Table 18 Case 3 Savings and Payback

Monthly PSTN voice expenses	$38,908
Multiply by 95% (P0.05)	95%
Monthly PSTN voice savings	$36,963
E1 removed from HQ to PSTN	$5,840
Required added WAN links	$(22,050)
Net total monthly savings	20,753
Net total annual savings	$249,037
Capital costs	$95,718
Installation (Estimate)	$15,000
Total Capital Costs	$110,718
Payback Period (Months)	5.3

This firm saves nearly $250,000 per year by moving their internal voice traffic onto their router backbone. The payback period is just over five months. The important numbers are summarized in Figure 21.

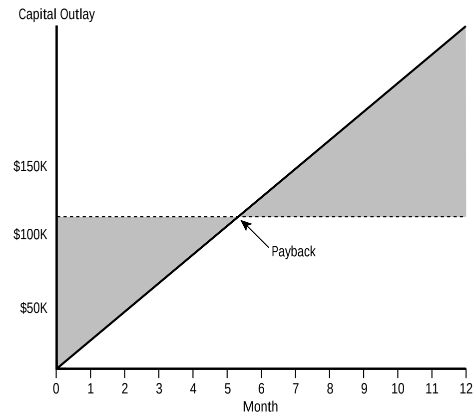


Figure 21 Case 3 Financial Summary

Cost per Minute

An organization's voice cost per minute is a common expense measurement. The details required to calculate the original cost per minute are given in Table 13. The total on-net minutes per month for all offices equals 107,270 minutes and the total monthly cost is $38,908. Dividing $38,908 by 107,270 yields an average cost per minute of $0.36 for all calls between headquarters and the branches. To carry 95% of this traffic over the data network, that network must be upgraded at a cost of $22,050 per month. Ninety-five percent of 107,270 equals 101,906 minutes. Dividing the upgrade costs by the carried minutes yields an average cost per minute of $0.22. In this case, the firm was able to reduce their average cost per minute from $0.36 to $0.22, a 40% reduction.

In Appendix 1, this case is modified to illustrate the solution in North America, South America and Asia. The modified cases are labeled Case 3A, 3B and 3C.

This concludes Step 6, the Financial Analysis. The first five steps suggested a process for designing integrated voice data networks that meet voice quality guidelines. This final step has demonstrated the financial attractiveness of these networks. Packet voice networks will help you prepare for the coming tornado of new business requirements.

You are encouraged to review the cases in Appendix 1 to see how these solutions apply around the world.

APPENDIX 1

Cases 2 and 3 are repeated in this appendix to show the applicability of the technologies and solutions for different geographical regions. Refer to the original case in the body of this document for complete descriptions of each case.

CASE 2A

Geographic Scope:	Pan Asian
Headquarters:	Singapore
Network:	Public Frame Relay Service

The headquarters is now located in Singapore and the eight offices are in Jakarta, Manila, Melbourne, Sydney, Tokyo, Osaka, Taipei and Hong Kong. Each branch has between 40 and 60 people. The Frame Relay network topology is shown in Figure 22.

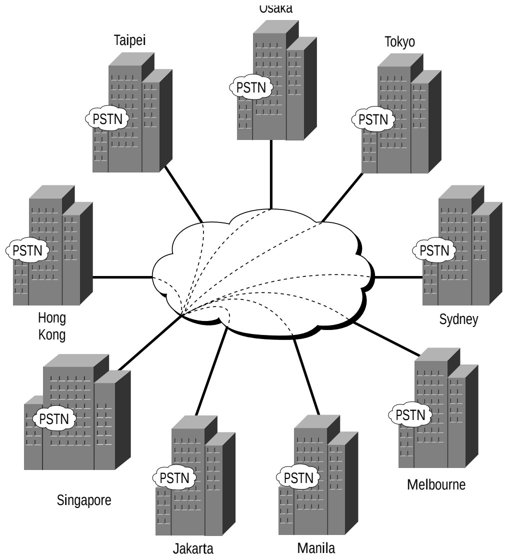


Figure 22 Case 2A Initial Network Topology

As in the original case, Case 2, the majority of branch calls are between branch employees and local outside customers with only about 20% between headquarters and branch employees. As a result of the high international PSTN rates, the cost for long distance service is about $138,000 per month or more than $1,600,000 per year.

Voice Network

Small PBXs connected over the public switched telephone network (PSTN) provide voice services. A 15% discount from standard PSTN rates is assumed. Table 19 shows the potential on-net voice and fax traffic volume and expense.

Data Network

The firm is using a public Frame Relay service, the topology of which is illustrated in Figure 22. The public Frame Relay service configuration and the on-going monthly expenses, or run-rate, are shown in Table 20. Note there is a single PVC between each branch and headquarters.

Table 19: Case 2A PSTN Volume and Expenses

Location	Purpose	Number of People	Average Minutes per Person per Day	On-Net % to HQ	Workdays per Month	Total Minutes per Person per Month	Total Minutes per Office per Month	Cost per Minute	Monthly Cost per Office
Singapore	Headquarters	-							
Hong Kong	Branch	45	90	20%	21.67	390	17,553	$0.89	$15,622
Tokyo	Branch	55	90	20%	21.67	390	21,453	$0.78	$16,626
Osaka	Branch	45	90	20%	21.67	390	17,553	$0.78	$13,603
Jakarta	Branch	40	90	20%	21.67	390	15,602	$1.23	$19,113
Manila	Branch	45	90	20%	21.67	390	17,553	$1.53	$26,768
Taipei	Branch	40	90	20%	21.67	390	15,602	$0.93	$14,432
Sydney	Branch	65	90	20%	21.67	390	25,354	$0.75	$19,015
Melbourne	Branch	45	90	20%	21.67	390	17,553	$0.75	$13,165
Total	-						148,223	$138,345	-

Assumptions
1. This is the average cost per minute and assumes 50% of calls are to HQ and 50% from HQ.
2. Cost of a voice call: Based on Singapore Telecom and other PTT pricing, with estimated small customer discount.

Network Redesign

The objective is to redesign the data network to support the added voice traffic while not adversely affecting its performance. The redesign is paid for by the PSTN cost reductions. Given the current infrastructure and familiarity with Frame Relay, a Voice over FR network is chosen.

The additional bandwidth requirements are determined as in Case 2 using the same assumptions repeated here.

- The total voice and fax call volume per person equals about 90 minutes per day.
- The call volume between headquarters and branch personnel represents about 20% of the total call volume or about 1/4 hour.
- A busy hour loading factor of 17% is appropriate.
- The Cisco voice compression module uses 8 kbps plus 3 kbps overhead (11 kbps per voice channel); thus, each 64 kbps trunk supports 5 voice channels.
- The branch PBXs require additional trunk modules.

Table 20: Case 2A Initial Network Configuration

Location	Purpose	Access Line Speed	Initial Port Speed	Initial PVC CIR	Frame Relay Charges
Singapore	Headquarters	E-1	768	-	$1,800
Hong Kong	Branch	E-1	128	64	$5,557
Tokyo	Branch	E-1	128	64	$3,453
Osaka	Branch	E-1	128	64	$3,453
Jakarta	Branch	E-1	128	64	$12,088
Manila	Branch	E-1	128	64	$12,968
Taipe	Branch	E-1	128	64	$4,474
Sydney	Branch	E-1	128	64	$5,600
Melbourne	Branch	E-1	28	64	$5,600
Total	-				$54,993
Footnote: The Port charge and the PVC charge are included in the Frame Relay Charges column.					

The amount of voice and fax traffic at each branch office that would optimally be diverted from the PSTN to the multiservice network is determined using the above assumptions and the appropriate traffic engineering tables given the desired P grade of service. This firm chose a P.05 grade of service. Table 21 summarizes the calculations and provides the number of PBX trunks that would be required at each site. The slight difference between Case 2 and Case 2A is a result of fewer personnel at the branch locations.

Number of Users	Number of Sites on Phone	Hours per day	Minutes per day	Busy hour (17%)	Minutes per Busy Hour	% of Traffic to HQ	Total Erlangs to HQ	Required Lines for.05 Blocking Probability
40	2	1.5	3,600	0.17	612	20%	2.04	6
45	4	1.5	4,050	0.17	689	20%	2.30	6
50	0	0	-	0	-	0%	0.00	0
55	1	1.5	4,950	0.17	842	20%	2.81	7

Number of Users	Number of Sites on Phone	Hours per day	Minutes per day	Busy hour (17%)	Minutes per Busy Hour	% of Traffic to HQ	Total Erlangs to HQ	Required Lines for.05 Blocking Probability
60	0	0	-	0	-	0%	0.00	0
65	1	1.5	5,850	0.17	995	20%	3.32	7
70	0	0	-	0	-	0%	0.00	0
-						Total Erlangs for HQ	19.38	28

The next step is to determine the additional bandwidth required on the Frame Relay network to support the voice and fax traffic. As indicated in the assumptions, each additional 64 kbps of bandwidth will provide for a minimum of 5 voice channels.

Network Upgrades

As in Case 2, all locations were upgraded to the same high bandwidth level to transport the added compressed voice traffic and to leave room for future growth in either voice or data traffic. This follows a conservative approach.

Refer to Case 2 for a complete discussion about the network upgrade.

Customer Premises Equipment

The Cisco Multiservice Access Concentrator, the MC3800, was installed in the branches and the headquarters. As can be seen in Table 22, the port speeds and the CIR of the PVCs were increased to accommodate the voice traffic. Adding a separate PVC for the voice was unnecessary because the MC3800 prioritizes the traffic on a single PVC. This capability avoided the cost of another PVC.

Each branch office connected the appropriate number of PBX trunks to the Cisco MC3800. The PBX can direct approximately 95% of traffic destined for headquarters preferentially to one of the trunks connected to the Cisco MC3800. The remaining overflow traffic, an estimated 5%, is directed to the PSTN.

Using G.729 encoding, the Cisco MC3800 compresses each 64 kbps voice channel to a data stream of approximately 11 kbps (8 kbps or less plus 3 kbps overhead) and forwards the compressed voice traffic over the Frame Relay network. The 11 kbps is a conservative measure because it does not include the benefits of the silence suppression techniques implemented in the MC3800.

The MC3800 is capable of supporting up to 24 channels of 8 kbps compressed voice that can be transported over public or private Frame Relay, ATM or leased line networks.

A number of different network designs are available to this firm. One option is to add a second PVC to transport the voice traffic. This would avoid to some degree the potential contention between the two different traffic types. However, it would also result in higher expenses because public Frame Relay providers commonly charge for each PVC. A less expensive approach would be to utilize the same PVC for both data and voice traffic. This design is available when using the Cisco MC3800 because it can prioritize the delay-sensitive traffic, voice, while ensuring the delay-insensitive traffic, data, is also serviced properly.

The Cisco Internetwork Operating System (Cisco IOS™) technology provides voice switching and advanced call management. For example, the headquarters MC3800 can be configured to switch calls between branches. Although the call volume may be low, the per minute cost, in this case, is very high. By off-loading this tandeming function from the PBX, the delay budget is reduced and the quality maintained by avoiding the compression/decompression cycle required by the PBX. This additional savings was not included in the following financial analysis.

The MC3800 can support E1/T1 ATM services via a software change as these services become available. Plus, the CBR and VBR support in the unit prepares this firm for video conferencing when it becomes required.

The costs and savings resulting from the network redesign are summarized in the next section.

Financial Analysis

The cost to upgrade the network to support the additional voice traffic is shown in Table 22.

Table 22: Case 2A Upgrade Expenses

Location	Purpose	Access Line Speed	Initial Port Speed	Initial PVC CIR	Frame Relay Charges	Upgraded Port Speed	Upgraded PVC CIR	Frame Relay Charges	Cost to Upgrade
Singapore	Headquarters	E-1	768	-	$1,800	2 x E1	-	$6,886	$5,086
Hong Kong	Branch	E-1	128	64	$5,557	256	128	$8,423	$2,866
Tokyo	Branch	E-1	128	64	$3,453	256	128	$5,803	$2,350
Osaka	Branch	E-1	128	64	$3,453	256	128	$5,803	$2,350
Djakarta	Branch	E-1	128	64	$12,088	256	128	$16,800	$4,712
Manila	Branch	E-1	128	64	$12,968	256	128	$17,624	$4,656
Taipei	Branch	E-1	128	64	$4,474	256	128	$6,809	$2,335
Sydney	Branch	E-1	128	64	$5,600	256	128	$9,199	$3,599
Melbourne	Branch	E-1	128	64	$5,600	256	128	$9,199	$3,599
Total	$54,993	$86,546	$31,553	-					
The cost of the second E1 access circuit at headquarters is included in the upgrade cost.									

The capital costs for the branches and the headquarters are shown in Table 11 of Case 2. The required additional bandwidth indicated in the preceding table costs $31,553 per month. Comparing the savings to these additional expenses, Table 23 illustrates the net monthly savings.

Table 23: Case 2A Savings and Payback

Monthly PSTN voice expense	$138,345
Multiply by 95% (P0.05)	95%
Monthly PSTN voice savings	$131,427
Required added bandwidth	$(31,553)
Net total monthly savings	$99,874
Net total annual savings	$1,198,492
Capital costs	$127,500
Installation (Estimate)	$20,000

Total Capital Costs	$147,500
Payback Period (Months)	1.5

This firm saves about $1,200,000 per year by moving internal voice traffic onto their Frame Relay network. The payback period is under two months. Figure 23 summarizes the important numbers.

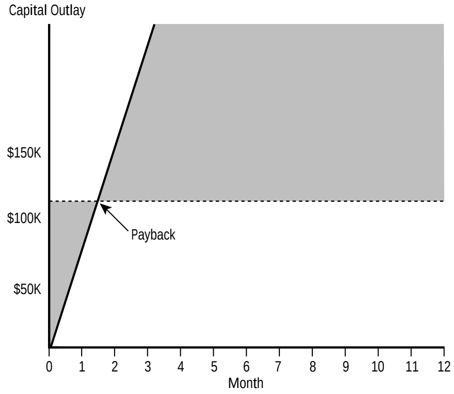


Figure 23 Case 2A Financial Summary

Cost per Minute

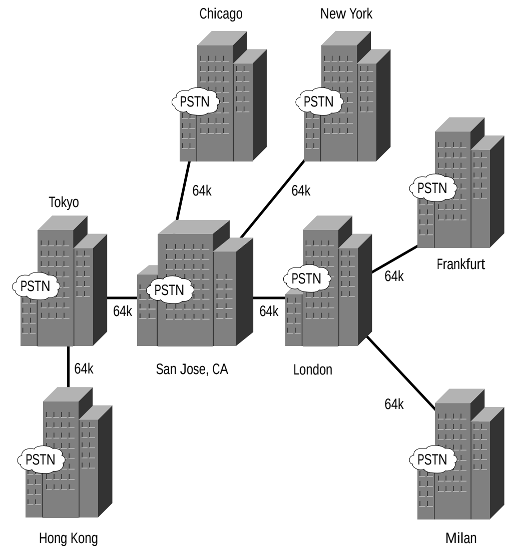
An organization's voice cost per minute is a common expense measurement. Table 19 provides the details required to calculate the original cost per minute. The total on-net minutes per month for all offices equals 148,223 minutes and the total monthly cost is $138,345. Dividing total cost by total minutes yields an average cost per minute of $0.93 for all calls between headquarters and the branches. To carry 95% of this traffic over the data network, that network must be upgraded at a cost of $31,553 per month. Ninety-five percent of 148,223 equals 140,811 minutes. Dividing the upgrade costs by this figure yields an average cost per minute of $0.22. In this case, the firm was able to reduce their average cost per minute from $0.93 to $0.22, a 75% reduction.

CASE 3A

Geographic Scope	International
Headquarters:	San Jose, California
Network:	Private Lines

In this case, the firm has headquarters in San Jose, California and branch offices in Hong Kong, Tokyo, Chicago, New York, London, Milan, and Frankfurt. Here, the on-net calling is primarily between the United States and international locations. The branch sizes and the call volume assumptions are unchanged from Case 3 in order to highlight the applicability of the solutions. Figure 24 shows the network topology and leased line circuit speeds.

Figure 24 Case 3A Initial Network Topology



Although the assumptions are unchanged from Case 3, the international long distance costs are approximately $32,000 per month or about $390,000 per annum. This is approximately 16% less than in Case 3. Of course, this is because of the lower PSTN rates.

Voice Network

Voice services are the same as in Case 3 and there is also a VPN contract from a carrier at about a 15% discount from standard PSTN rates.

Each individual in the branch offices spends about 2 hours per day communicating via telephone or fax. Approximately 20% of this traffic is back and forth between the branch and headquarters.

Table 24 shows the potential on-net voice and fax traffic volume and expense.

Location	Purpose	Number of People	Average Minutes per Person per Day	On-Net % to HQ	Workdays per Month	Total Minutes per Person per Month	Total Minutes per Office per Month	Cost per Minute(1)	Monthly Cost per Office
San Jose, CA	Headquarters	-							
Frankfurt	Branch	15	150	20%	21.67	650	9,752	$0.54	$5,266
Milan	Branch	15	150	20%	21.67	650	9,752	$0.48	$4,681
London	Branch	45	150	20%	21.67	650	29,255	$0.29	$8,484
New York	Branch	45	150	20%	21.67	650	29,255	$0.07	$2,048
Chicago	Branch	15	150	20%	21.67	650	9,752	$0.07	$683
Tokyo	Branch	15	150	20%	21.67	650	9,752	$0.52	$5,071
Hong Kong	Branch	15	150	20%	21.67	650	9,752	$0.63	$6,095
Total	-						107,267	-	$32,326
Assumptions **1. This is the average cost per minute and assumes 50% of calls are to HQ and 50% from HQ.** **2. Cost of a voice call based on carrier quote off-net pricing with customer discount.**									

Data Network

An eight-node data network, leased by the firm, utilizes routers, and is hubbed out of their San Jose headquarters. A number of branches connect through other branches to reach the headquarters location. This arrangement was set up to hold down leased line costs. The existing network can be seen in Figure 24.

Network Redesign

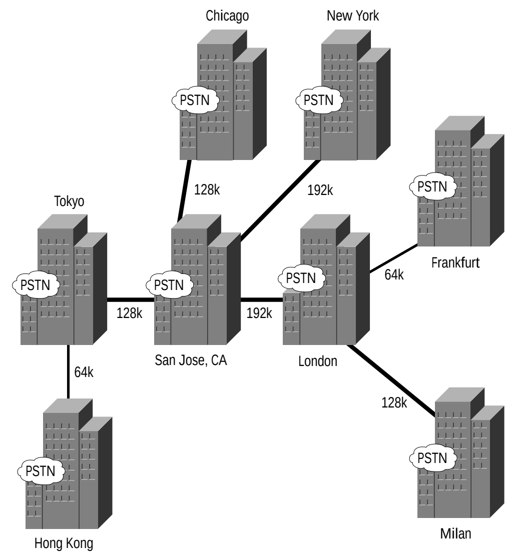
The objective is to redesign the data network to support the added voice traffic while not adversely affecting its performance. The redesign is paid for by the PSTN cost reductions. A Voice over IP network is chosen. The following redesign is conservative considering the following assumptions.

- The additional bandwidth required was estimated following the steps indicated in Case 3.
- The network was redesigned using the same assumptions as in Case 3.

Traffic engineering tables are consulted next to determine the appropriate number of trunks required to carry the traffic, given the desired P grade of service. This firm chose a P.05 grade of service. The amount of voice and fax traffic at each branch office that would optimally be diverted from the PSTN to the multiservice network is determined using the above assumptions and the tables.

The calculations and trunking requirements are summarized in Table 15 in Case 3.

Figure 25 Case 3A Redesigned Topology



Particular circuit speeds were increased to accommodate the added traffic. The reasoning is the same here as it was in Case 3. Refer to Case 3 for a complete discussion of the reasoning behind the speed increases.

The additional bandwidth is not free. The following table furnishes the incremental expense details. The following section analyzes the complete financial picture.

Customer Premises Equipment

A Cisco 3620 router was installed in the smaller branches and a Cisco 3640 router was installed in the two larger branches. Each smaller branch office connected four key system FXO trunks to the Cisco 3620 router (eight trunks to the Cisco 3640 in the larger branch locations). The reprogrammed key system would then direct approximately 95% of traffic destined for headquarters preferentially to one of the trunks connected to the Cisco router. The remaining overflow on-net voice traffic, an estimated 5%, is directed to the PSTN.

Table 25: Case 3A Upgrade Expenses

San Jose to:	Original 64 kbps (Monthly Cost)	Redesigned 128 kbps (Monthly Cost)	Redesigned 192 kbps (Monthly Cost)	Incremental Cost (Monthly Cost)
Tokyo	$8,700	$13,400	-	$4,700
Tokyo to Hong Kong	$5,500	-	-	-
Chicago	$1,250	$2,050	-	$800
New York	$1,400	-	$3,100	$1,700
London	$6,400	-	$13,400	$7,000
London to Milan	$5,100	$8,150	-	$3,050
London to Frankfurt	$4,250	-	-	-
TOTAL	$17,250	-	-	-

The leased lines terminate at the San Jose Headquarters, where the voice channels are decompressed, and routed to the headquarters PBX. Because 23 channels have been removed from the PSTN and are now sent over the router network, the San Jose headquarters can remove one of the PSTN T1 access lines.

As explained in Case 3 the Cisco 3600 enables the high performance and low latency which result in very high voice quality.

This firm is planning a possible migration to Frame Relay and support for telecommuting. In Hong Kong and New York, where commuting to work can be very difficult, the Cisco 3600 will eventually be used as access servers for employees telecommuting from their homes. If the firm decides to transition to Frame Relay the Cisco 3600 can support that protocol.

It should be noted that not all key systems provide automatic route selection for preferential voice traffic switching. If considering this type of configuration, contact the key system or PBX vendor to insure this capability exists in the equipment.

The costs and savings resulting from the network redesign are summarized in the next section.

Financial Analysis

The capital costs for the branches and the headquarters are shown in Table 17 in Case 3. The required additional bandwidth indicated in the preceding section costs $17,250 per month. Comparing the savings to these additional expenses, Table 26 illustrates the net monthly savings.

Table 26: Case 3A Savings and Payback

Monthly PSTN voice expenses	$32,326
Multiply by 95% (P0.05)	95%
Monthly PSTN voice savings	$30,710
T1 removed from HQ to PSTN	$950
Required added WAN links:	$(17,250)
Net total monthly savings	$14,410
Net total annual savings	$172,919
Capital costs	$95,718
Installation (Estimate)	$15,000
Total Capital Costs	$110,718
Payback Period (Months)	7.7

By moving their internal voice traffic onto their router backbone, this firm saves nearly $175,000 per year and the payback period is less than eight months. Figure 26 summarizes the important numbers.

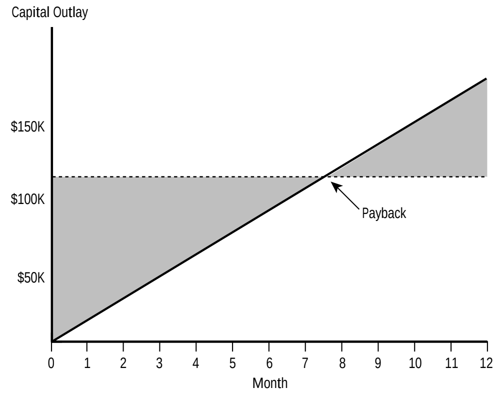


Figure 26 Case 3A Financial Summary

Cost per Minute

Table 24 provides the details required to calculate the original cost per minute. The total on-net minutes per month for all offices equals 107,267 minutes and the total monthly cost is $32,326. Dividing $32,326 by 107,267 yields an average cost per minute of $0.30 for all calls between headquarters and the branches. To carry 95% of this traffic over the data network, that network must be upgraded at a cost of $17,250 per month. Ninety-five percent of 107,267 equals 101,904 minutes. Dividing the monthly upgrade costs by the carried minutes yields an average cost per minute of $0.17. In this case, the firm was able to reduce their average cost per minute from $0.30 to $0.17; a 43% reduction.

CASE 3B

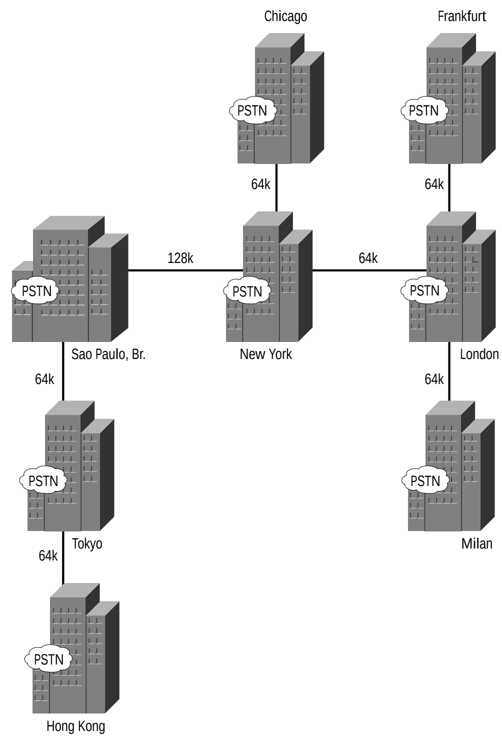
Geographic Scope: International

Headquarters: Sao Paulo, Brazil

Network: Private Lines

In Case 3B, the headquarters of the firm is in Sao Paulo, Brazil and branch offices are located in Frankfurt, Milan, London, New York, Chicago, Tokyo, and Hong Kong. Here, the on-net calling is primarily between Brazil and international locations. The branch sizes and the call volume assumptions are unchanged from Case 3 in order to highlight the applicability of the solutions.

Figure 27 Case 3B Initial Network Topology



Voice Network

Voice services are the same as in Case 3, and there is also a VPN contract from a carrier at about a 15% discount from standard PSTN rates.

Each individual in the branch offices spends about 2 hours per day communicating via telephone or fax. Approximately 20% of this traffic is back and forth between the branch and headquarters.

Table 27 shows the potential on-net voice and fax traffic volume and expense.

Data Network

An eight-node data network, leased by the firm, utilizes routers, and is hubbed out of their San Jose headquarters. A number of branches connect through others to reach the headquarters location. This arrangement was set up to hold down leased line costs.

Network Redesign

The objective is to redesign the data network to support the added voice traffic while not adversely affecting its performance. The redesign is paid for by the PSTN cost reductions. A Voice over IP network is chosen. The following redesign is conservative considering the following assumptions.

- The additional bandwidth required was estimated following the steps indicated in Case 3.
- The network was redesigned using the same assumptions as in Case 3.

Table 27: Case 3B PSTN Volume and Expenses

Location	Purpose	Number of People	Average Minutes per Person per Day	On-Net % to HQ	Workdays per Month	Total Minutes per Person per Month	Total Minutes per Office per Month	Cost per Minute(1)	Monthly Cost per Office
Brazil	Headquarters	-							
Frankfurt	Branch	15	150	20%	21.67	650	9,752	$1.24	$12,043
Milan	Branch	15	150	20%	21.67	650	9,752	$1.56	$15,164
London	Branch	45	150	20%	21.67	650	29,255	$1.45	$42,273
New York	Branch	45	150	20%	21.6	650	29,255	$0.66	$19,162
Chicago	Branch	15	150	20%	21.67	650	9,752	$0.66	$6,387
Tokyo	Branch	15	150	20%	21.67	650	9,752	$1.12	$10,873
Hong Kong	Branch	15	150	20%	21.67	650	9,752	$1.46	$14,237
Total	-						107,267	-	$120,138

Assumptions
1. This is the average cost per minute and assumes 50% of calls are to HQ and 50% from HQ.
2. Cost of a voice call based on carrier quote off-net pricing with customer discount, and with discounted PTT prices.

Traffic engineering tables are consulted next to determine the appropriate number of trunks required to carry the traffic, given the desired P grade of service. This firm chose a P.05 grade of service. The amount of voice and fax traffic at each branch office that would optimally be diverted from the PSTN to the multiservice network is determined using the above assumptions and the tables.

The calculations and trunking requirements are summarized in Table 15 in Case 3.

Particular circuit speeds were increased to accommodate the added traffic. The reasoning is the same here as it was in Case 3. Refer to Case 3 for a complete discussion of the reasoning behind the speed increases.

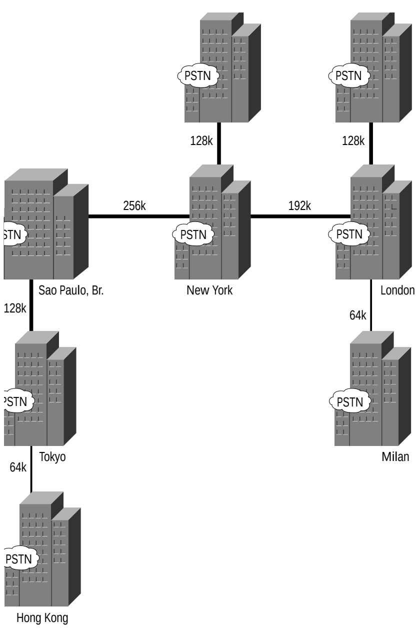


Figure 28 Case 3B Redesigned Topology

Customer Premises Equipment

A Cisco 3620 router was installed in the smaller branches and a Cisco 3640 router was installed in the two larger branches. Each smaller branch office connected four key system FXO trunks to the Cisco 3620 router (eight trunks to the Cisco 3640 in the larger branch locations). The reprogrammed key system would then direct approximately 95% of traffic destined for headquarters preferentially to one of the trunks connected to the Cisco router. The remaining overflow on-net voice traffic, an estimated 5%, is directed to the PSTN.

The leased lines terminate at the San Jose Headquarters, where the voice channels are decompressed, and routed to the headquarters PBX. Because 23 channels have been removed from the PSTN and are now sent over the router network, the San Jose headquarters can remove one of the PSTN T1 access lines.

As explained in Case 3 the Cisco 3600 enables the high performance and low latency which result in very high voice quality.

This firm is planning a possible migration to Frame Relay and support for telecommuting. In Hong Kong and New York, where commuting to work can be very difficult, the Cisco 3600 will eventually be used as access servers for employees telecommuting from their homes. If the firm decides to transition to Frame Relay the Cisco 3600 can support that protocol.

It should be noted that not all key systems provide automatic route selection for preferential voice traffic switching. If considering this type of configuration, contact the key system or PBX vendor to insure this capability exists in the equipment.

The costs and savings resulting from the network redesign are summarized in the next section.

Financial Analysis

The next table shows the monthly incremental cost to upgrade the leased lines.

Table 28: Case 3B Upgrade Expenses

Sao Pablo to:	Original 64 kbps (Monthly Cost)	Redesigned 128 kbps (Monthly Cost)	Redesigned 192/256 kbps (Monthly Cost)	Incremental Cost (Monthly Cost)
Tokyo	$12,500	$20,400	-	$7,900
Tokyo to Hong Kong	$5,500	-	-	-
New York	-	$8750 (1)	$15,800 (1)	$7,050
New York to Chicago	$1,400	$2,300	-	$900
New York to London	$6,200	-	$13,000(2)	$6,800
London to Milan	$5,100	-	-	-
London to Frankfurt	$4,250	$7,600	-	$3,350
TOTAL	$26,000	-	-	-
(1) The speed of the circuit from New York to Sao Paulo was upgraded from 128 kbps to 256 kbps. (2) The speed of the circuit from New York to London was upgraded from 64 kbps to 192 kbps.				

The capital costs for the branches and the headquarters are shown in Table17 in Case 3. The required additional bandwidth indicated in the preceding section costs $26,000 per month. Comparing the savings to these additional expenses, Table 29 illustrates the net monthly savings.

Table 29: Case 3B Savings and Payback

Monthly PSTN voice expenses	$120,138
Multiply by 95% (P0.05)	95%
Monthly PSTN voice savings	$114,131
E1 removed from HQ to PSTN	$9,300
Required added WAN links	$(26,000)
Net total Monthly Savings	$97,431
Net total Annual Savings	$1,169,173

Capital costs	$95,718
Installation (Estimate)	$15,000
Total Capital Costs	$110,718
Payback Period (Months)	1.1

By moving their on-net voice traffic onto their router backbone this firm saves over $1,000,000 per year. The payback period is less than two months. The numbers are summarized in Figure 29.

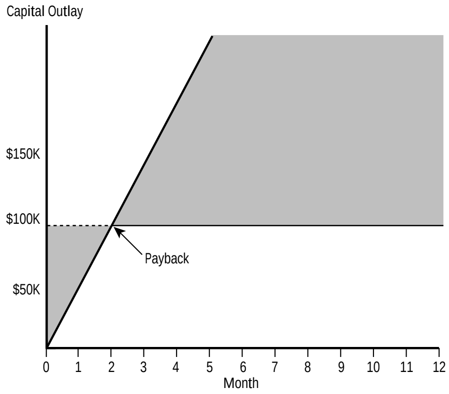


Figure 29 Case 3B Financial Summary

Cost per Minute

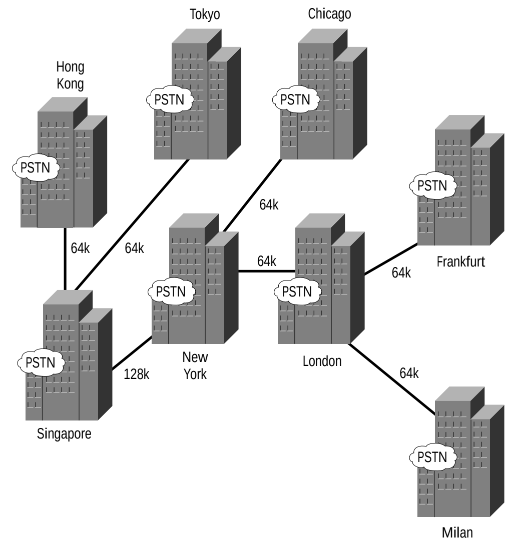
Table 27 provides the details required to calculate the original cost per minute. The total on-net minutes per month for all offices equals 107,267 minutes and the total monthly cost is $120,138. Dividing $120,138 by 107,267 yields an average cost per minute of $1.12 for all calls between headquarters and the branches. To carry 95% of this traffic over the data network, that network must be upgraded at a cost of $26,000 per month. Ninety-five percent of 107,267 equals 101,904 minutes. Dividing the monthly upgrade costs by the carried minutes yields an average cost per minute of $0.24. In this case, the firm was able to reduce their average cost per minute from $1.12 to $0.24; a 79% reduction.

CASE 3C

Geographic Scope:	International
Headquarters:	Singapore
Network:	Private Lines

In Case 3C, the company headquarters are in Singapore and branch offices are in Hong Kong, Tokyo, Chicago, New York, London, Milan, and Frankfurt. Here, the on-net calling is primarily between the Singapore and the international locations. The branch sizes and the call volume assumptions are unchanged from Case 3.

Figure 30 Case 3C Initial Network Topology



Original Design

Voice services are the same as in Case 3 and there is also a VPN contract from a carrier at about a 15% discount from standard PSTN rates.

Each individual in the branch offices spends about 2 hours per day communicating via telephone or fax. Approximately 20% of this traffic is back and forth between the branch and headquarters.

Data Network

The data network is based in Singapore and uses routers. The existing network can be seen in Figure 30.

Network Redesign

The objective is to redesign the data network to support the added voice traffic while not adversely affecting its performance. The redesign is paid for by the PSTN cost reductions. A Voice over IP network is chosen. The following redesign is conservative considering the following assumptions.

- The additional bandwidth required was estimated following the steps indicated in Case 3.
- The network was redesigned using the same assumptions as in Case 3.

Table 30 shows the potential on-net voice and fax traffic volume and expense.

Location	Purpose	Number of People	Average Minutes per Person per Day	On-Net % to HQ	Workdays per Month	Total Minutes per Person per Month	Total Minutes per Office per Month	Cost per Minute(1)	Monthly Cost per Office
Singapore	Headquarters	-							
Frankfurt	Branch	15	150	20%	21.67	650	9,752	$1.40	$13,652
Milan	Branch	15	150	20%	21.67	650	9,752	$1.74	$16,968
London	Branch	45	150	20%	21.67	650	29,255	$1.04	$30,278
New York	Branch	45	150	20%	21.67	650	29,255	$0.56	$16,383
Chicago	Branch	15	150	20%	21.67	650	9,752	$0.56	$5,461
Tokyo	Branch	15	150	20%	21.67	650	9,752	$0.78	$7,557
Hong Kong	Branch	15	150	20%	21.67	650	9,752	$0.89	$8,679
Total	-						107,267	-	$98,978

Assumptions
1. This is the average cost per minute and assumes 50% of calls are to HQ and 50% from HQ.
2. Cost of a voice call based on carrier quote off-net pricing with customer discount, and with discounted PTT prices.

Traffic engineering tables are consulted next to determine the appropriate number of trunks required to carry the traffic, given the desired P grade of service. This firm chose a P.05 grade of service. The amount of voice and fax traffic at each branch office that would optimally be diverted from the PSTN to the multiservice network is determined using the above assumptions and the tables.

The calculations and trunking requirements are summarized in Table 15 in Case 3.

Certain circuit speeds were increased to accommodate the added traffic. The reasoning is the same here as it was in Case 3. Refer to Case 3 for a complete discussion of the reasoning behind the speed increases.

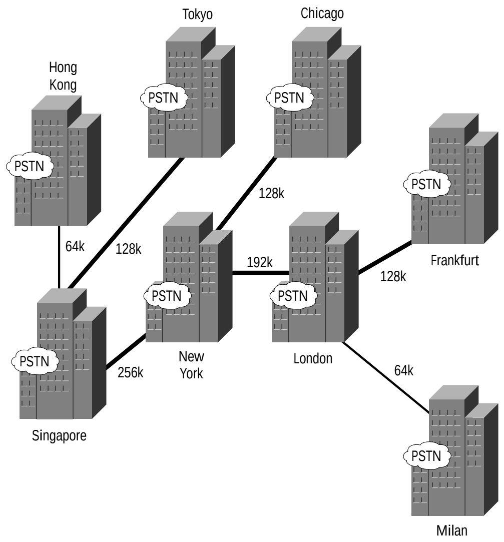


Figure 31 Case 3C Redesigned Topology

Customer Premises Equipment

A Cisco 3620 router was installed in the smaller branches and a Cisco 3640 router was installed in the two larger branches. Each smaller branch office connected four key system FXO trunks to the Cisco 3620 router (eight trunks to the Cisco 3640 in the larger branch locations). The reprogrammed key system would then direct approximately 95% of traffic destined for headquarters preferentially to one of the trunks connected to the Cisco router. The remaining overflow on-net voice traffic, an estimated 5%, is directed to the PSTN.

The leased lines terminate at the San Jose Headquarters, where the voice channels are decompressed, and routed to the headquarters PBX. Because 23 channels have been removed from the PSTN and are now sent over the router network, the San Jose headquarters can remove one of the PSTN T1 access lines.

As explained in Case 3 the Cisco 3600 enables the high performance and low latency which result in very high voice quality.

This firm is planning a possible migration to Frame Relay and support for telecommuting. In Hong Kong and New York, where commuting to work can be very difficult, the Cisco 3600 will eventually be used as access servers for employees telecommuting from their homes. If the firm decides to transition to Frame Relay the Cisco 3600 can support that protocol.

It should be noted that not all key systems provide automatic route selection for preferential voice traffic switching. If considering this type of configuration, contact the key system or PBX vendor to insure this capability exists in the equipment.

The costs and savings resulting from the network redesign are summarized in the next section.

Financial Analysis

The next table shows the monthly incremental cost to upgrade the leased lines.

Table 31: Case 3B Upgrade Expenses

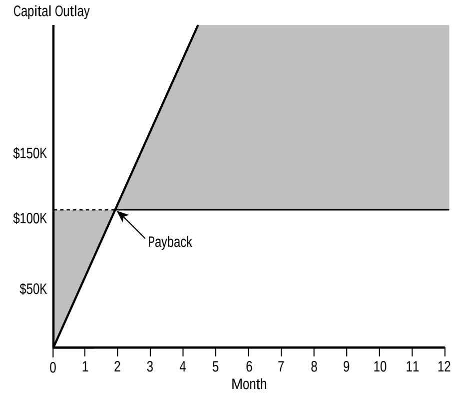
Singapore to:	Original 64 kbps (Monthly Cost)	Redesigned 128 kbps (Monthly Cost)	Redesigned 192/256 kbps (Monthly Cost)	Incremental Cost (Monthly Cost)
Tokyo	$3,400	$5900	-	$2,500
Hong Kong	$2,200	-	-	-
New York	-	$15,700(1)	$22,000(1)	$6,300
New York to Chicago	$1,400	-	-	-
New York to London	$6,200	-	$13,000(2)	$6,800
London to Milan	$5,100	-	-	-

Singapore to:	Original 64 kbps (Monthly Cost)	Redesigned 128 kbps (Monthly Cost)	Redesigned 192/256 kbps (Monthly Cost)	Incremental Cost (Monthly Cost)
London to Frankfurt	$4,250	$7,600	-	$3,350
-	-	-	-	-
TOTAL	$19,850	-	-	-
Footnotes **(1) The speed of the circuit from New York to Singapore was upgraded from 128 kbps to 256 kbps.** **(2) The speed of the circuit from New York to London was upgraded from 64 kbps to 192 kbps.**				

The capital costs for the branches and the headquarters are shown in Table 17 in Case 3. The required additional bandwidth indicated in the preceding section costs $19,850 per month. Comparing the savings to these additional expenses, Table 31 illustrates the net monthly savings.

By moving their internal voice traffic onto their router backbone this firm saves over $900,000 per year. The payback period is less than two months. Figure 32 summarizes the financial numbers.

Figure 32 Case 3C Financial Summary



Cost per Minute

Table 30 provides the details required to calculate the original cost per minute. The total on-net minutes per month for all offices equals 107,267 minutes and the total monthly cost is $98,978. Dividing $98,978 by 107,267 yields an average cost per minute of $0.92 for all calls between headquarters and the branches. To carry 95% of this traffic over the data network, that network must be upgraded at a cost of $19,850 per month. Ninety-five percent of 107,267 equals 101,904 minutes. Dividing the monthly upgrade costs by the that figure yields an average cost per minute of $0.19. In this case, the firm was able to reduce their average cost per minute from $0.92 to $0.19; a 79% reduction.

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