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The Definitive Guide™ To

Successful Deployment of VoIP and IP Telephony

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Chapter 8: Sweating the Details and Assuring Success

In 1926 self-help author Robert Collier wrote *"The great successful people of the world have used their imaginations, they think ahead and create their mental picture, and then go to work materializing that picture in all its details, filling in here, adding a little there, altering this a bit and that bit, but steadily building, steadily building."* We will likely find no better theme as we enter the final chapter of this guide. We started by imagining what could be:

- A new world of telephony where dial tone was only the beginning
- A new world of telephony that provides a cornerstone for the carrier triple play, Unified Communications, and Multimedia, Internet, Communications, and Entertainment (MICE) systems.
- A new world of productivity and communications enabled by the Internet Protocol, SIP, and related protocols and technologies.

We then filled in the details, planned our approach, vetted our designs, and developed a strategy for continually refreshing our system and improving, enhancing, and optimizing it throughout its, hopefully long, life cycle. All along the way we strove to keep the best parts of the system we are replacing and to add new capabilities as yet undreamed of when that early system was architected and built.

In this chapter, we will discuss many of the details required to ensure success in this new system. Some of the points have been covered elsewhere; others have not. We will also provide a comprehensive checklist for the implementation of a modern, packet-based telephony system that can be used to assure that all the critical details, large and small, are covered.

Legal and Regulatory Issues

Technology has always been ahead of the law. It is the nature of technology, and of the law. The gun was invented before there were gun laws. The car was invented before there were automobile regulations. Nuclear power plants existed before there were regulatory controls. In each of these cases, the technology existed before the needed legal and regulatory framework was in place. In each case, however, there was also a body of law that could readily be applied to the new situation. Though not an ideal fit and not completely regulating all aspects of the new technology, it existed, nonetheless. For instance, gun laws contain details regulating specific aspects of guns but prior to law regarding murder, self defense and other areas of gun-related behavior had been around for millennia. The same is true of VoIP and IPT.

There is a body of telecom regulation still on the books from earlier times, much of which is being used as a stop-gap measure until more specific regulation of packet-based, as opposed to circuit-based, services comes along. Some of the earlier regulation actually appears as though it may be applied to packet-based systems on a more permanent basis. These issues and their impact on the VoIP and IPT initiatives of all enterprises and organizations warrant the greatest consideration to assure the overall success of any packet voice initiative. Or stated in a more negative way, if these aspects of implementation are not considered, they could cause implementation of the new system to be stopped dead in its tracks.

General Considerations

The first question to ask might be "Do specific regulations apply to my organization?" A better question, however, might be "Does specific regulation apply to my organization directly or might it in some way impact me indirectly?" The answers to both questions will vary depending upon the role of your organization. If your organization is the end-user organization implementing VoIP or IPT, then one set of regulations may apply to you. If your organization is a service provider offering packet telephony services commercially in a country, to businesses and/or consumers, then your organization may be subject to a different set of regulations. There is also a third situation that may or may not be worthy of consideration: that is, if a person is an individual user of a consumer, or residential VoIP service that is used for business and reimbursed by the company.

In most cases, if you are creating and sending your own packets, there is no specific regulation on your activities, though there may be an indirect impact on you if you are using the services of a carrier or service provider and they run afoul of the law and their service is suspended or shut-down, thereby impacting the voice traffic they are carrying for you.

The major categories of issues that must be considered are:

- Countrywide bans of VoIP services or service providers
- Penalties for non-compliance
- Emergency services
- Security and wiretapping/call recording
- Records retention
- Taxes and fees

Each will be considered in broad terms here. It is your responsibility to research the specific countries in which you will be operating to determine the present state of regulation in that country and to plan accordingly, including continuing to use traditional circuit telephony until regulations change or become clear.

Countrywide Bans of VoIP Services or Service Providers

Numerous countries in the Middle East and some in Asia and South America have either fully or partially banned VoIP services. Often they have banned VoIP services from any provider except the registered telephone companies, many of which are fully or partially state-owned Postal, Telephone & Telegraph (PTT) organizations. In some cases, a limited set of outside providers is allowed to operate. The most commonly blocked VoIP service seems to be the one whose technology is most adept at evading being blocked: Skype.

The impact on a multi-country enterprise is that in order to provide packet-based services across the entire user base, possibly incorporating suppliers and customers, a patchwork quilt of providers and services may need to be stitched together. Alternatively, a service-neutral Virtual Private Network (VPN) service will need to be employed that operates in all countries that require it and that will transparently and securely move IP-based voice packets. In many cases, this approach will introduce a level of complexity and cost that many organizations are trying to avoid through the adoption of VoIP. And, in some cases, if the VoIP traffic is not completely new traffic but rather traffic that previously moved across traditional telephone lines, its absence will be missed and it might raise a red flag for authorities to determine where their previous telephony network revenue is going. It is worth noting that many governments around the world view the settlements on their traditional telephony services as an important source of hard currency.

Penalties for Non-Compliance

There are instances where penalties ranging from fines to incarceration have been levied for both private use of VoIP and the offering of VoIP as a service to the public. It is very important to understand the penalties that are possible and to err on the side of compliance. One does not need to go as far as Namibia or Vietnam to find examples of penalties: there are plenty in the U.S. that have had an indirect impact on companies using the services of certain providers, including services intended as residential services that are used for small office/home office connectivity and have caused service interruptions, delays in bringing services to new areas, and increases in the cost, and therefore price, for the service.

Emergency Services

Most industrialized nations have a system for summoning police, fire, and emergency medical services with a single nationwide number, such as 911 in the U.S. and Canada, and 999 in the United Kingdom. On the private enterprise side, it is important to assure that key information such as the location of the party calling for help—for instance, their office building, floor and area on the floor—be transmitted to emergency call centers as opposed to the billing location or the location where the voice switch is located. It is also important for all employees and any other persons in a facility to know exactly how to dial to emergency services. Should they dial 9-1-1? Or 9 for an outside line and then 9-1-1? The legal liabilities are beyond the scope of this chapter, but this is a very serious area in its own right that deserves full consideration, implementation, and testing prior to actual release of VoIP or IPT systems for general use. The law has generally found companies liable for non-working or inaccurate emergency calling systems. It is also worth noting that most companies' present systems also are not set up properly. Companies often do not think about this until the system is pressed into service and it does not work. Many companies also rely on cell phones for such services, which have their own set of problems and issues, especially inside buildings.

From the standpoint of service providers, there are country-by-country regulations on their obligations to provide emergency services. Companies should ensure that any service providers used fully comply with all regulations and the systems should be tested.

Security and Wiretapping/Call Recording

Security and wiretapping/call recording are issues that have different dimensions if they are considered in an enterprise or service provider environment. In the enterprise environment, the enterprise has a need to monitor and assess what their employees and/or contractors are doing, the nature of their communications and the applicability of organizational policy and applicable law to those activities. Certain industries, such as the financial industry, also have particular requirements to record calls to verify certain transactions and not just "for training purposes."

For the service provider, call recording reaches into the realm of "legal intercept" referred to commonly as wiretapping. Different countries have different regulations but generally there is a trend to apply traditional wiretapping laws to packet-based services. In general, service providers must comply and may not inform the subscribers of the ongoing investigation. This is especially true in organized crime and terrorism cases. In the United States, the appropriate law is the Communications Assistance to Law Enforcement Act (CALEA), which has recently been expanded to include packet-based services.

Records Retention

As with Instant Messages and emails there is an increasingly strong preference both for and against retention of voice packet traffic. On the "for" side are certain regulations, such as certain provisions of the Sarbanes-Oxley law and SEC regulations in the US, that currently require retention of IM and email and may be broadened to include voice packets. On the "against" side is an increasing number of cases in which corporate or government wrong-doing has been able to be proven by the use of records which, if they did not exist, could not have been used for this purpose. Retention of voice packets is an area of current legal interest and one that will continue to be refined in the future. It is also an area that must be considered in any corporate policy.

Taxes and Fees

One of the biggest benefits of a move from traditional telephony to VoIP was that VoIP was free of much of the regulation and nearly all of the fees traditionally levied against phone companies and the services they provide. This was translated into lower prices and was responsible for much of the initial interest in VoIP. As governments around the world become aware of VoIP as a source of revenue, from the local government to the national level, taxes and fees are being dreamed up for VoIP. Absent strong precedent many taxing authorities may, at least for some initial period of time, be able to levy unbelievably high fees, such as the city of Baltimore has done, with a $3.50 per number tax on VoIP subscribers. This will impact service providers directly and enterprises indirectly due to the service providers passing on the costs of fees and compliance.

Country-Specific Regulatory Issues

It is imperative that you check current regulations in every country in which you operate to assure that you are meeting all legal and regulatory requirements. In addition to national requirements there may be state or provincial requirements and even requirements imposed by local governments that can harm your packet-based voice initiative.

Contractual Issues

Many organizations, especially large ones, although the same may be true for small to medium ones with especially high call volumes to specific countries, have specific contracts, often called tariffs with the previously popular "Tariff 12" being a good example - in place with traditional telephony carriers. The tariffs are filed with the FCC or other applicable authority in other countries than the US and spell out the pricing and contract conditions, and often have minimum numbers of minutes that must be paid for whether delivered by the carrier or not. Before calculating the "savings" which can be realized by a move to non-traditional telephony organizations should carefully review and understand the conditions of the current contracts. In many cases there are guaranteed minimum numbers of minutes, or payments, or both. Moving minutes to a VPN or another telephony service, even of the same carrier or service provider, may not satisfy the minimum service requirements and payment must be made for the minimum traffic levels whether the traffic is transported by the carrier or not. Many organizations have been on the path to implementing voice packet services and have found themselves in the position of having to delay implementation until the end of a multiyear contract for traditional telephony services.

Patent Issues

It is said that history repeats itself and so it does more often than we might realize. When statements are made that VoIP and IPT technologies are “mature” and “ready for prime time” one must look no further than the early days of traditional telephony to see evidence that, in fact, VoIP and IPT are in *their* earliest days. If one were to read the newspapers from the ‘70s, that is to say the *1870s*, not the *1970s*, one would find that patent disputes are common in the early days of the commercialization of any new technology and packet telephony is no different. The 1870s saw bitter legal battles between competitors for the claims to telephony patents. There are many examples just in the lawsuits against Alexander Graham Bell from a variety of individuals including Antonio Meucci and Elisha Gray and the problems Bell had due to his patents being filed in America before the were filed in the UK. History is repeating itself and this is why the final topic in this section is the business impact of the increasing reliance on patent filings and the enforcement of patents as a business strategy on VoIP, IPT and related Unified Communications projects.

Once the dust settled in the telephone industry, solid patents and standards became the bedrock of reliable, interoperable global phone systems. All things considered there were really very few variations in phone systems globally, compared with their data counterparts and almost none whatsoever considering the number of variations that could exist. That having been said, there are still hundreds of important variations from country to country and beyond the national variations there are even manufacturer-specific variations. But, even with all of the variations on the themes underlying issues of patent protection, once settled, became a non-issue throughout the remainder of the life-cycle of the traditional telephony system.

Because VoIP, IPT, Unified Communications, Internet Multimedia Subsystem (IMS) and related systems are in their commercial infancy, patent issues remain an important consideration for any organization implementing VoIP and IPT systems and/or services. In fact, these are the two categories that must be considered: systems and services. If a company is purchasing systems then they must be cognizant of the fact that the systems they are purchasing, and the underlying hardware and/or software, even if purchased from a historically reliable manufacturer may contain technologies that might later be proven to be infringing on patents that the manufacturer does not own or have license to use. Likewise, when purchasing services the purchaser must be aware that the services may be based upon patents that the service provider does not own or have license to use. What is the “bottom line”? The bottom line is that while most legal outcomes do not require that technology purchased be given back they do make acquisition of subsequent products more expensive or impossible and may cause the suspension of services.

While there is really no analysis or report on what the risks are this is still an area that must be considered and, more than anything else, will impact the decision of companies to adopt VoIP and IPT until many of the early patent issues have been decided. The alternative? Rely on extending current technologies and only using VoIP and IPT technologies where they are strategically vital in the mean time.

Technical Issues

Even though many technical issues have been addressed in other areas of this eBook they are being re-emphasized here because they often get lost in the background noise but have confounded a proper implementation of VoIP and IPT in many organizations. These topics are presented in no particular order of importance and if properly addressed during the implementation phase none will be particularly thorny but if left unaddressed can become very serious impediments to productivity.

Voice Band Data

There is a general expectation that telephony systems can handle a variety of Voice Band Data (VBD) options such as:

- FAX
- Modems
- Dual Tone Multi-Frequency (DTMF, also known as touch tone and used to access voicemail, bank balances, flight information and other services)
- TTY/TDD devices for the deaf
- Alarm systems
- Other telemetry systems

There is, however, no set of standards for supporting these capabilities within VoIP and IPT systems and each manufacturer and service provider will offer their own set of options.

If an organization implements VoIP using Ethernet-based devices, and a standard Ethernet physical interface such as the ubiquitous eight wire RJ45 modular connector then the expectation of VBD support usually goes away right along with the traditional phone jack. This is because the Ethernet physical connector is different from the four or two wire phone jack and there is no place to plug in the fax, modem, TTY terminal or alarm system. If, on the other hand, Internet Protocol Telephony is implemented using some sort of analog terminal adapter or gateway that connects via a standard phone plug then the expectation that the system can handle VBD usually still exists.

If VBD is to be handled properly in the new system then proper arrangements need to be made. If the new system is not intended to provide support or VBD then proper training and support needs to be provided to deal with the lack of the aforementioned capabilities or to provide workarounds. If, for instance, faxes are to be supported then the new system must either use 64Kbps Pulse Code Modulation (PCM) or, possibly, 32Kbps ADPCM voice coding which requires additional bandwidth for all calls. If a lower bit rate codec is to be used for normal voice calls then the system must be capable of detecting the presence of the fax signal and switching over to PCM or ADPCM or employing FAX T.38 forwarding, fax servers or similar capabilities.

Numbering Plans

VoIP and IPT do not use numbering plans or phone numbers, per se, but they can, and do, have the capability of mimicking phone numbers, at least for an interim period of time. Many in the IP standards bodies have been chiding the current purveyors of VoIP services claiming that they don't provider "real VoIP" or "real SIP implementation" because they still use phone numbers. The use of phone numbers, at least in an alphanumeric form as part of a SIP User Resource Identifier (URI), will be critical until the migration is made from the older telephony networks and, perhaps, that older telephony network may still be needed for some time to come for its ability to locate subscribers and route calls to them as this is a scalability and interoperability capability as yet to be proven by SIP networks.

Voice VPNs

Many organizations, from multi-branch single-city banks to multinational manufacturers and consulting firms have long ago established voice Virtual Private Networks (VPNs) with their own specialized dialing plans and associated volume pricing agreements. The particulars of the use of the specialized dialing plans has become so engrained in these organizations that any productivity gains associated with a shift to packet telephony would be more than offset for many years to come by making changes to the dialing plans. Employees know each other's extension numbers: clients and suppliers know Direct Inward Dialing (DID) numbers based upon those extensions, employees know specific internal numbers such as Human Resources and Security, directories are published, phone numbers are part of documentation and often posted on telephones, bulletin boards and web sites. In short, the entire ability of a company to communicate via voice, both internally and with the outside world, is based upon previously established phone numbers and associated dialing plans. For these reasons it is imperative that the prior dialing plan and associated phone numbers are maintained. It is also imperative, to the extent possible, that the special characteristics associated with the old system, such as the exact sound of inside and outside dial tone, the delay that occurs after dialing the digit (0? 8? 9? 7?) used to get an outside line and all other characteristics be kept unchanged.

E.164 Compliance

E.164 is the global ITU (http://www.itu.org) standard for telephone numbers. E.164 provides an international numbering plan for public telephone systems in which each assigned number contains a country code (CC), a national destination code (NDC), and a subscriber number (SN). There can be up to 15 digits in an E.164 number and each E.164 number is unique worldwide. With up to 15 digits possible in a number, there are 100 trillion possible E.164 phone numbers, more than 10,000 for every human being on earth. This makes it possible, in theory, to direct-dial from any conventional phone to any other conventional phone in the world by inputting no more than 15 single digits. The hierarchical nature of E.164 as well as the number and use of digits is based largely on work done in the 1880s and 1890s by Alexander Graham Bell and outlined in his famous letter to the board of directors of the Bell Telephone Company in 1894. This same letter described the use of telephone exchanges and foretold of automated switching, something for which Dr. Bell could see the need but could neither describe nor produce.

Maintaining strict compliance with E.164 standards globally at every gateway between the packet world and the circuit world is critical in maintaining the bridge between the two, and assuring interoperability as long as the bridge between the two worlds of telephony is still needed, which is to say, well into the foreseeable future.

URIs/URLs

The Internet uses Uniform Resource Locators (URLs) such as www.prognosis.com to locate hosts that have some sort of geographic alignment. SIP uses a very similar addressing construct, the Uniform Resource Identifier (URI). A SIP URI identifies a communications resource. Like all URIs, SIP URIs may be placed in web pages, email messages, or printed literature. They contain sufficient information to initiate and maintain a communication session with the resource.

SIP Uniform Resource Identifiers

The "sip:" scheme follows the guidelines in Internet Engineering Task Force (IETC) Request for Comment (RFC) 2396 - Uniform Resource Identifiers (URI): Generic Syntax. They use a form similar to the mailto URL, allowing the specification of SIP request-header fields and the SIP message-body. This makes it possible to specify the subject, media type, or urgency of sessions initiated by using a URI on a Web page or in an email message. In general, the form of a SIP URI, is

sip:user:password@host:port;uri-parameters?headers

- In a SIP URI user is the identifier of a particular resource at the host being addressed. The term "host" in this context frequently refers to a domain.
- The "userinfo" of a URI consists of this user field, the password field
- The @ sign following the userinfo fields. The userinfo part of a URI is optional and *may* be absent when the destination host does not have a notion of users or when the host itself is the resource being identified.
- If the @ sign is present in a SIP URI, the user field *must not* be empty.
- If the host being addressed can process telephone numbers, for instance it is an Internet telephony gateway, a telephone subscriber field may be used to populate the user field."
- Password is optional and is a password associated with the user. While the SIP URI syntax allows this field to be present, its use is optional and not recommended, because the passing of authentication information in clear text (such as URIs) has proven to be a security risk in almost every case where it has been used.
- The host field provides the identifier for the SIP resource. The host part contains either a fully-qualified domain name or numeric IPv4 or IPv6 address. Using the fully-qualified domain name form, such as phn.company.com, is recommended whenever possible.
- In addition to the host name a port field may be provided and, if provided, designates the port number where the request is to be sent.

If a telephone number is to be transmitted it is converted into a SIP URI of the form:

sip:nnnnn@domain;user=phone or sip:nnnnn@host:5060;user=phone

- The "user=phone" parameter is a hint that the part to the left of the '@' sign is actually a phone number, in case there are SIP users whose names happen to consist of all digits. Typically a proxy server for the domain will use a dial plan to resolve this into a real destination.

ENUM and Other Numbering Issues

The ITU and the Internet Engineering Task Force (IETF) are currently working on a plan called ENUM that will expand E.164 to encompass both traditional analog phones and digital devices, including computers and other devices on the Internet. All types of communications devices—including analog phones and fax machines, digital phones and fax machines, wireless (cellular) phones, pagers, digital modems, digital video terminals, and VoIP devices -- will have unique E.164 addresses with direct dialing possible from any device to any other. Considerations for ENUM and extensions to corporate directories using Light weight Directory Access Protocol (LDAP) and Domain Name Service (DNS) systems must be considered as a part of any comprehensive implementation.

Interworking Issues with IPT Service Providers

Invariably a packet telephony project of any size will need to connect to IP Telephony service providers for either connectivity between geographically distant internal locations, often including small office/home office sites, or customer or supplier sites but usually all of the above. There are several options and issues that must be considered, all of which have analogs in the earlier phone system.

The New Demark

The issue of where the responsibility of the "phone company" ends and the responsibility of the customer begins has been the subject of very specific regulation over the years and the issue is very clear. The point at which the "hand off" occurs and responsibility shifts is called the point of demarcation, or "demark". With individual voice circuits the point of demarcation is marked at a wire connecting a punch-down block owned by the phone company (normally attached to a piece of plywood on the user premise) to another punch-down block (on the same or a different piece of plywood on the user premise). In the case of T1 systems the demark is a phone company or customer owned T1 DSU/CSU device. There are other examples, but the point is that the stage (or place) where the hand-off occurs is very specific in traditional systems.

In emerging VoIP and IPT systems the point of demarcation is usually as specific because it is the same carrier, or type of carrier, providing the wires but often a different service provider using those wires to provide the service. This is why a fresh interest must be taken in clearly defining customer responsibilities and service provider responsibilities.

End-to-End Service Management

An increasingly important issue that impacts all aspects of multimedia, whether the service is provided by an internal or external organization, is "end-to-end" service management. In the past "end-to-end" has been defined more as provider edge to provider edge, not including the local access links on the end, unless equipment was co-located in a service provider Point-of-Presence (POP) or carrier Central Office (CO). That definition was expanded by managed service providers to include the local access links and the managed customer premise equipment, such as the router, but did not include the actual service that used the underlying connections nor did it include things such as servers and client end-systems.

VoIP and IPT services should be managed - and key metrics such as Quality of Service (QoS) and Quality of [user] Experience (QoE) measured—from the user phone device on one end to the user phone device on the other end. Monitoring, measurement and management of all intermediate pieces from which the end-to-end call is created remains very important but it is the actual service itself that should be measured and managed. It is also critical to view VoIP and IPT traffic in its context as part of the bigger picture of multimedia traffic.

Tasks

Even back when IP networks were "data only" there were many "flavors" of data - browser traffic, file transfer protocol, terminal emulation, email—all of which have their own performance requirements Assessing IP performance has become even more complex with the addition of two more distinct types of traffic, voice and video, to the multimedia mix. All of the tasks required of today's IP multimedia networks—functions referred to as applications in an environment often called the single network voice, data, video "triple play"—are easier to deal with if we place them into one of three "loss" categories and cross reference those with two delay-related categories, understanding that these categories are relative to each other and not to some standard baseline. It is in the dynamic prioritization of each type of traffic in the network where we achieve true quality of service that matches, or doesn't match, the needs of each type of traffic.

As shown in Table 8.1 applications on a multimedia network can be broadly divided into those that are less sensitive to delay and delay variation and those that are more sensitive. We can further divide those two categories into low, medium and high in terms of the loss that is relatively acceptable for each application. If we were considering the underlying protocols we would actually place the applications into only two categories, those that use the reliable, error-checking, retransmit-on-loss connection-oriented TCP protocol or the connection-less User Datagram Protocol (UDP), often referred to by this author as the "Unreliable Datagram Protocol". In this case there would be no loss in the TCP examples, but possible loss in the UDP examples, but we are looking at this from the user's perspective and will view the "Acceptable Loss" based on its impact on the total user experience, often referred to as QoE, or Quality of Experience.

	Acceptable Loss – User's Perspective		
	Low	Medium	High
Low Delay/Delay Variation Sensitivity	Browser, Telnet	File Transfer	eMail
Delay & Delay Variation Sensitive	Telepresence, Instant Messaging	IPTV	Real-time Voice/IP Telephony

Table 8.1: Acceptable Multimedia Loss vs. Delay.

These relationships also imply a certain prioritization relative to Quality of Experience (QoE) which comes into play if packets are competing for network resources. For instance, from the table we can rightly imply that if an email packet were competing with a real-time voice (VoIP) packet for network resources that the VoIP packet would get the resources. Likewise if a device such as a router had to discard a packet and a VoIP packet were competing with a browser packet that the browser packet would win because the discarding of a browser packet would cause the retransmission of the packet, placing a further demand on the network to do the retransmission. Quite the opposite is true of voice. Voice will sustain a 3-5% loss before the quality drop is noticeable and can randomly lose up to around 10% of voice samples before intelligibility begins to decline substantially. Considering these dynamic trade-offs, based upon the demand on the network at a specific moment in time, is an important a part of assessing the performance of your multimedia IP network.

If this explanation seems to run counter to traditional thinking, based upon the well-known characteristics of IP, TCP and UDP you are quite right. The underlying protocols were designed for a uni-media, not multi-media, world and the shift in network operations is a function of changes in the software code in routers and switches that lie in the path between one communicating system and another, not in a fundamental re-write of TCP or UDP.

Tools

A specific, finite list of characteristics may be ascribed to certain types of packets either as they are formed in their respective end systems, as they enter the network, or at any other point where they are handled by systems capable of such discrimination and traffic marking. Each of the combination of characteristics can be matched to a specific designation, or class of service, with the intention that traffic so-marked will achieve the corresponding promised quality of service. Any given network may or may not implement all of the classes of service or may have some special hybrid.

Constant Bit Rate (CBR) promises service most like a dedicated circuit. It promises a constant bit rate and near zero loss. Real-Time (RT) and near-Real-Time (nRT) are two flavors of Variable Bit Rate (VBR) classes of service that promise performance below that of a guaranteed circuit but better than the best effort objective of historical IP networks. Available Bit Rate (ABR), which promises a modest "best effort" service with a minimal assured throughput, even in times of congestion, and Unspecified Bit Rate (UBR), which makes no guarantees whatsoever, in most cases, round out the classes of service.

By aligning the needs of each of the applications to specific classes of service it is possible to provide the full range of tools needed to provide the mixed quality-of-service needed for multimedia traffic and to request the network to provide for the specific needs of each.

Differentiation and Outcomes

The key to success is to properly differentiate and then to aim for the best but plan for the worst. In today's multi-terabit networks it is easy to picture an optical highway system so empty that every packet can receive first class treatment. While this may be an accurate high level view it neglects two important facts: the first is that one must actually be able to get on the highway and the second is that regardless of the low and declining cost per bit per second, businesses are always driven to save money and often sacrifice performance to do so.

In this frame of reference it is easy to see that trade-offs are made in how the carrier or service provider's bandwidth is carved up and how it is paid for: regardless of low price there is still a cost. The key is to properly differentiate traffic and not to over-provision or under-provision the system relative to the need.

Table 8.2 shows which class of service is ordinarily applied to each sample application. In a perfect world—and it does happen in the networking world, especially at off-peak times—all traffic is marked according to what class of service it is promised, but all packets, regardless of marking, traverse the network in near-wire-speed times and none of them are delayed or discarded. In that case the differentiation and class-of-service marking of packets is inconsequential and it could even be argued adds to delay and overhead. So, why do it? Because the network is not a perfect place, especially at peak times, and the marking tells the network how to resolve conflicts for resources when they ultimately arise in the shared environment of the IP network.

Class of Service	Sample Applications
Constant Bit Rate (CBR)	Streaming Video, Circuit Emulation
Real-Time (RT)	Voice over IP, IP Telephony, Telepresence, Video Conferencing
near-Real-Time (nRT)	Browser, Real-time File Transfer, Instant Messaging
Available Bit Rate (ABR)	Email, Background File Transfer
Unspecified Bit Rate (UBR)	Server Synchronization, Back-ups

Table 8.2: Class of Service (CoS) Examples.

Monitoring and Testing

Each of the classes of service will have some associated metrics that can be measured and interpreted relative to its impact on the application. The metrics that are normally gathered and used are packet loss, delay, delay variation (jitter) and service availability. Constant Bit Rate, for instance, has a packet loss near zero, delay near wire speed (meaning that distance is a more important factor than the performance of intermediate pieces of equipment), delay variation near zero and a 99.99+% availability. On the other end of the spectrum is Unspecified Bit Rate, with no guarantees at all, in most cases. To judge both is easy, and fair. A packet loss of a whopping 2.6% for CBR is considered to be abysmal performance and most likely subject to a financial penalty against the carrier. A loss of only 2.6%, especially during peak traffic times, is probably pretty good for Unspecified Bit Rate. The key is to have the target metrics, and penalties, spelled out in advance and to monitor performance relative to the agreed upon targets. This is best done in the Service Level Agreement and monitored via a combined program of intrusive testing and passive monitoring.

SIP-T

Traditional telephone networks, as shown in Figure 1, employ two distinct parallel networks to create, facilitate and delete end-to-end telephone connections. The network interconnecting the ISDN phone on the left via discrete circuits running through the grey cloud in the middle to the analog phone on the right is only the network that carries the bits between the two phones.

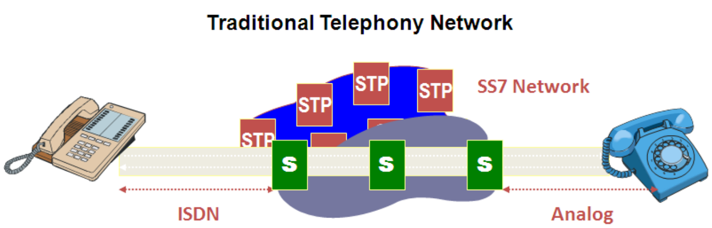

Figure 8.1: Traditional telephony architecture.

The blue cloud in the back, in this case marked “SS7 Network” because it facilitates the signaling system used in North America, is a parallel control network comprised of highly reliable dual-home packet switches that does the set-up and tear-down of the circuits. Without the secondary signaling network there would be no calls. This is quite different in IP-based networks. IP-based networks are characterized by the fact that the same network, the IP-based Internet, intranets or VPNs, carries both the content and the signaling traffic. One aspect of IP networks that has concerned telephony engineers from the beginning, is that while their SS7 networks provide highly reliable out-of-band signaling to insure the integrity of the network and minimize risks to network reliability and, therefore revenue, IP networks started out as “best effort” with control and user traffic competing for shared resources on an even footing. This, fortunately, has changed for a variety of reasons and control traffic can receive priority treatment.

It is important to consider the signaling needs of the network during design and implementation. The Session Initiation Protocol (SIP) is the most likely choice for voice sessions in the network and, therefore, SIP-T, Session Initiation Protocol for Telephones, will be the most likely choice for signaling, call set-up and tear-down. SIP-T provides a framework for the integration of legacy telephony signaling into SIP messages. SIP-T accomplishes this using two techniques known as ‘encapsulation' and 'translation'. At a SIP gateway, SS7 messages are encapsulated within SIP in order that information necessary for services is not discarded in the SIP request. However, intermediaries like proxy servers that make routing decisions for SIP requests cannot be expected to understand SS7’s particular signaling protocol called the ISDN-User Part (ISUP), so simultaneously, some critical information is translated from an ISUP message into the corresponding SIP headers in order to determine how the SIP request will be routed.

While pure SIP has all the requisite instruments for the establishment and termination of calls, it does not have any baseline mechanism to carry any mid-call information (such as the ISUP INFormation/Information Requst (INF/INR) query) along the SIP signaling path during the session. This mid-call information does not result in any change in the state of SIP calls or the parameters of the sessions that SIP initiates. A provision to transmit such optional application-layer information is also needed.

Electrical Power

AC wall power or Power over Ethernet (PoE) must be provided to power most IP phones. In many cases consideration of power is not a part of network design and results in substantial additional costs and implementation delays.

Business Issues

Are packet-based telephony systems less expensive than their circuit-based counterparts? Is it possible to tell? Are the systems so different that it becomes an apples vs. watermelon comparison? It is most likely to be true that IPT systems, which use a new engine to deliver traditional voice services will be less expensive, possibly 30 to 60% less expensive, than their traditional telephony counterparts. Most organizations, however, eschew a strict replacement approach for one of replacing dial tone and then adding new capabilities, placing them squarely in the VoIP realm, regardless of their choice of underlying technology. With VoIP the organization adds additional capability but also brings additional training requirements and budgetary and staffing needs.

Budgets

Packet-based telephony systems are different from their circuit counter-parts in many important ways that have an impact on budgets. For instance, hardware becomes obsolete much more quickly and refresh or replacement must be budgeted. In the old telephony networks it was possible to have a standard touch-tone phone in service for a decade or two. Not so with today's IP phones, palm devices and wireless phones: the technologies and capabilities are changing too rapidly. A reasonable refresh cycle is twelve to eighteen months depending on your industry.

Software and services are also areas that are quite different than in the old telephony days. IP phones often have software licenses or costs for specific features that were either non-existent or associated with the phone switch or PBX in traditional systems. These items must be managed and budgeted.

Staff

One of the biggest misconceptions in the early days of IP telephony was that voice was "just another IP application" and that it could be managed successfully by the existing staff with existing tools. That is a misconception that is now understood to be wildly wrong. What many organizations are now coming to grips with is that they must keep at least some of the former telephony staff members. Ideally from this process a "multi-tech" with the requisite skills to handle voice, data and video issues will emerge but wide-spread availability of such super-techs is still some way off. Organizations should guard against skills attrition in their staffing efforts and assure that they are providing both training of new staff and cross-training of existing staff in all aspects of voice, data and video.

Repositioning/Redeployment of Old Equipment

While some parts of the organization may be moving rapidly to deploy packet technology it may be difficult or impossible for other areas to do so and it might make very good sense to reposition or redeploy equipment that is being removed from one area of the organization to another area. What if one division has chosen not to make a move to packet telephony at this time? They may still need to upgrade their old key system to handle more phone lines. Or, what if one location is in an area where regulations forbid VoIP or there is a minimum number of minutes required on a tariff that can't be met if traffic is shifted to VoIP before the end of the contract period? These are great places to reposition older equipment.

Secondary Market for Old Equipment

For a variety of reasons, the most important of which is manufacturer discontinuation of specific lines of traditional telephony equipment there is a thriving secondary market for older telephony equipment. The secondary market should always be considered vs discarding older equipment.

Donation of Old Equipment

In the early days of packet telephony, when the direction was less certain, donation of old equipment to educational or non-profit organizations for their own internal use or training purposes made sense and the donations were welcome. The farther we go toward world packet domination the less true this is and the more that donations of this type are seen as liabilities. It is still possible to donate old equipment but the context needs to change: donations to manufacturers, carriers or other museums are becoming more important, especially for pieces of equipment that are particularly unique or have an interesting role they played in a historical event. Organizations should not underestimate the value of their old, outdated equipment and should always look at the value of donation of that equipment as opposed to discarding the equipment.

Revisiting TCO and ROI Models for your organization

In Chapter 1 we described several different approaches to determining Total Cost of Ownership (TCO) and Return on Investment (RoI) models. Your approach might exactly parallel one of the suggested methodologies, might closely resemble one or two or might be completely different. In any case once the new system has been installed and stabilized, and once a sufficiently large number of users have been using the system to get some meaningful numbers it is time to revisit your TCO and RoI models and see how you've done.

Is the Total Cost of Ownership lower or higher than originally estimated? Was the investment returned in the time frame estimated or does it appear as though it will be? Will it take longer? Less time? Why or why not? Many organizations never go back and recheck the initial assumptions but it is important to do so. The first reason might be for "bragging rights" to a properly performed prognostication or, in any case, maybe some original assumptions were inaccurate or never made it into the final design. Regardless, it is a valuable and important exercise to revisit the original planning exercises and clear up any discrepancies.

Security, Privacy, and Safety

Generally speaking because VoIP and IPT operate over IP networks they are subject to the same security vulnerabilities and issues as any IP service. However, to the extent that the calls may traverse traditional phone systems and networks they are subject to the security vulnerabilities and issues of traditional telephone networks as well. For this reason a plan must be put into place to assure the integrity, security and privacy of all phone calls regardless of the system over which they are connected. This would also be a good place to discuss wireless vs. wireline issues as well. A complete and comprehensive security analysis at each design step followed by security assessments of the actual system once it is installed and periodically thereafter are critical to the operation of the system.

The four areas of security that will need to be addressed are:

- Privacy
- Operational continuity
- System integrity
- Theft of service

Privacy assures that information being transmitted remains between the communicating parties and cannot be "overheard" by any third party. Privacy can be accomplished through a variety of means, the most effective of which is end-to-end encryption. End-to-end encryption scrambles the signal at one end-user device and unscrambles it at the other end. End-to-end encryption assures that the message is as secure as possible from one end to the other.

Operational continuity is the process of protecting a system from issues such as Denial of Service (DoS) attacks and assuring that calls can be processed under any foreseeable circumstance. Denial of Service attacks that are beginning to emerge in VoIP and IPT systems include restarting call servers so that the traffic generated by all of the connections attempting to re-establish themselves brings the system down.

System Integrity problems include situations where hackers can masquerade as internal callers or spoof IDs and phone numbers as well as assuring the integrity of call logs and other tools that can be used after-the-fact for their forensic value.

Theft of service, often called 'toll fraud" in the traditional phone network has almost disappeared as an issue from the lists of most managers of packet voice systems but shouldn't. In the past the number one impact of toll fraud was the high cost. The cost of stolen minutes had to be paid to the telephone company and this had a direct financial impact. Packet-based systems are so inexpensive, by comparison, and are often flat-rate so the financial impact is not so noticeable. Theft of service, or unauthorized use by employees, steals time from the job and often has other implications such as setting up an organization for prosecution if their systems are being used by hackers, criminals or terrorists.

Emergency Calls—911, 999, 112, and So On

In addition to the regulatory issues discussed earlier it is imperative that an organization fully understand the liabilities associated with not providing proper emergency calling. Not only are the lives of employees, contractors and visitors placed at risk with a non-functional emergency calling system property can be placed at risk if it is not possible to summon fire fighters in case of a fire in the building. As stated earlier it is imperative that emergency calling procedures are well documented and tested to assure they will operate properly when needed and summon assistance to the proper location.

Telephony Migration Checklist

The telephony migration checklist provided at the end of this chapter offers a single list that can be used to double check all of the key aspects of your migration from traditional telephony to IP-based communications.

Summary

The successful implementation of VoIP and IP Telephony is a topic that is broad and deep all at once. In this eight chapter e-book we have made every attempt to touch on all of the issues that you might encounter and to delve deeper into those that are likely to be of greatest importance. We have provided templates and checklists, guidance and "war stories". With all of this input and your own experiences to guide you it is still impossible to get everything exactly right on the first go but with careful planning and implementation it will be possible to get closer to perfection than would ever have been possible by just "taking a stab at it".

If we have provided some valuable insights, if we have shown some possible booby-traps and some practical work-arounds and saved you time, aggravation and money we have accomplished what we set out to accomplish at the beginning. Best of luck to you as you pioneer a new way of communicating and find the best path forward for your organization and its users.

Telephony Migration Checklist

Justification

- _ Return on Investment (RoI)
- _ Return on Effort (RoE)
- _ Total Cost of Ownership
- _ Strategic capabilities
- _ Manufacturer discontinuation of older systems

Expectation Management

- _ Users
  - _ Different is not bad
  - _ Provides all important functions of the current system
  - _ Will provide important new capabilities.
- _ Management
  - _ Total CAPEX cost may be the same or higher
  - _ Total OPEX cost may be the same or higher
  - _ Must assure the continuity of voice communications
  - _ Acquire advanced capabilities of strategic benefit
  - _ VoIP is not “free voice”
  - _ VoIP is not “just another application.”

Budgeting and Planning

- _ Reuse/Redeployment of existing assets
- _ Acquisition
  - _ Products
  - _ Services
  - _ Managed services and outsourcing
- _ Support
  - _ Internal Help Desk
  - _External Help Desk

Service Level Agreements (SLA)

- _ Parameters
  - _ Availability
  - _ QoS: Packet loss
  - _ QoS: Delay
  - _ QoS: Delay Variation
  - _ QoE: MOS
  - _ QoE: E-Model/R value
  - _ QoE: ITU metrics

_ Classes of Service (CoS)
    _ Differentiation
    _ Prioritization
    _ Queue management
    _ Per CoS shaping and policing
_ Penalties

SLAs, Monitoring and Management

_ Pre-deployment simulation and modeling
_ Manufacturer-specific monitoring and management
_ Real-time business views
    _ Call detail records
    _ Calls in progress
_ Delay-to-dial-tone rates
_ Gateway channel utilization and loading
_ Real-time call monitoring
_ Phone and multi-media device availability and monitoring
_ Successful vs. failed call completion rates
_ Poorly performing components
_ Service level breaches and SLA compliance
_ Real-time interface to Manager of Managers (such as HP OpenView)
_ Summary and exception reporting
_ Utilization trends over time
_ Managed devices by company, department, and location
_ Asset tracking
_ Capacity planning
    _ Incoming and outgoing calls
    _ Loading by dial plan, routing rules and gateway.
    _ Bandwidth utilization
    _ Delay and delay variation (jitter)
    _ Packet loss
    _ Route patterns, utilization, and availability
_ Customer Network Management (CNM)
_ MSP provided tools
_ Enterprise Tools
    _ Manufacturer provided tools
    _ Third-party tools

Assessing Urgency and Prioritizing Migration

_ No flash cut
_ Assess urgency
_ Prioritize migration order

- Which office or region has outgrown, or is closest to outgrowing their existing system?
- Which region has bandwidth or IP network infrastructure good enough to carry voice, and other resources to accommodate the new telephony upgrades?
- Which region has the most technically savvy support staff?
- Which region was involved with early testing and/or staging and is, therefore, most familiar with the new system?
- Which office or region is closest, because the earliest migrations are always learning experiences and should be expected to go slower and be more problematic?

Migration

_ Systems Integrator vs. Do It Yourself
_ Managed Service vs. Do It Yourself

Planning & Assessment

_ Inventory of existing capabilities
_ Network infrastructure

- Identify circuits used for wide area networking and Internet connectivity
- Inventory the network hardware—routers, switches, cabling. Don't overlook elements like router operating system version. This is a good time to get your routed network all on a common foundation.

_ Current voice services and equipment

- Document any inbound and outbound call center environments
- List the details of hunt groups, call pickup groups and automatic all distribution groups.

_ Current business data applications

- What servers are in place today? What's changing or evolving in parallel? Identify any linkages between your existing business applications and voice services. If you have a CRM system in place, this piece will be absolutely crucial.

_ Security posture

- Inventory your security environment early to ensure there are no surprises later on.

_ Existing voice needs—The calling patterns today
_Call Detail Records (CDRs)

_ Network Readiness

- Estimate the bandwidth needed to support a known number of phone lines
- Calculate the bandwidth required to support busy-hour call volumes
- Determine how many paths might be needed across a wide area network (WAN)
- Delay budget is important and includes all sources of fixed delay. Delay less than 150ms one-way is ideal; >300ms will impact voice QoE. Delay should be specified in the SLA.
- Delay variation (aka jitter) is a bigger problem than delay. Delay is fixed and easier for the listener to adjust to. Delay variation is difficult to adjust for and causes anxiety and stress on the part of listener.
- The human ear can adjust to packet loss up to about 10 percent. Packet loss >1.5 to 3 percent can have impact on voice quality. Packet loss impact can be reduced by shortened voice samples and fewer samples per RTP packet.
- Network availability directly impacts QoE. No dial tone is a BIG problem and causes loss of confidence in the system.

_ CODEC Selection
    _ Voice Coding & Quality
    _ Bandwidth Use
        _ Constant Bit Rate
        _ Variable Bit Rate
        _ Silence Suppression / Voice Activity Detection
_ Gateways & Signaling
    _ Trunking Gateways
    _ Signaling
_ Echo Cancellation
_ Assessment Tools & Modeling Process
    _ Traffic collection from existing IP network
    _ Call traffic collection from existing telephony network
    _ Import of Call Detail Records (CDRs) from telephony network/bill
    _ Off-line simulation of VoP/VoIP network impact
    _Simulated VoP/VoIP traffic insertion into live network
    _ Real-time assessment of VoP/VoIP traffic impact
    _ Real-time tuning of VoP/VoIP parameters
    _ Manufacturer-specific parameters, metrics and analysis
    _ Call Server, IP-PBX or Switch Manufacturer-provided tool
    _ Simulate complex traffic patterns
    _ Modeling/sizing of softswitches, IP PBX and IP Centrex
    _ Modeling/sizing of gateways
    _Transfer pre-deployment assessment info to NMS

_ Model different codecs
    _ Pulse Code Modulation (G.711/ PCM)
    _ Adaptive Differential Pulse Code Modulation (G.721/G.726/G.727 ADPCM)
    _ Linear Predictive (LP) codecs: G.729, G.729a or G.723.1, G.726
    _ Other Proprietary Linear Predictive (LP) codecs
    _ Wideband codecs
_ Different voice sample sizes
_ Different packet fills
_ Voice activity detection/silence suppression
_ VoIP w/SIP
_ VoIP w/H.323
_ Voice over Frame Relay (VoFR)
_ Voice over ATM (VoATM)
_ Voice over DSL (VoDSL)
_ Modeling/sizing VoP gateway performance
_ Performance through multiple gateways
_ Delay
_ Delay Variation / Jitter
_ Service Level Agreement Classes of Service
_ Multimedia VoIP/VoP calls with video
_ Impact of VoIP/VoP on data and video performance
_ Impact of data and video performance on VoP/VoIP
_ Performance of multimedia sessions/applications
_ Simulate combined multimedia nets (i.e. different divs or depts)
_ Simulate multimedia MPLS VPN
_ Simulate multimedia Metro or Wide Area Ethernet VPN
_ Simulate multimedia VLANs
_ Forecast Mean Opinion Score (MOS) using G.107 eModel
_ Forecast Mean Opinion Score (MOS) using proprietary method
_ Forecast Delay
_ Forecast Delay Variation / Jitter
_ Forecast VoP Service availability
_ Model voice-only QoS/QoE
_ Model non-voice multimedia QoS/QoE (i.e. video)
_ Model combined multimedia QoS/QoE
_ Model packet loss and impact
_ Draft "Buy vs. Build" reports to support VoP/VoIP approach
_ Cost/price analysis for design option
_ Reports/tables/charts showing voice quality metrics vs costs
_ Draft Return on Investment (RoI) Calculations
_ Draft Total Cost of Ownership (TCO) Calculations
_ Draft Cost/Minute or Cost/User breakdown
_ Draft Cost/Minute or Cost/User analysis by SLA category
_ SLA compliance and out-of-compliance reports

Design and Pre-Deployment Testing

_ The Team
  _ Current IP Network Technical
  _ Current IP Network Management
  _ Current Voice Technical
  _ Current Voice Management
  _ Temp/Contract Multi-Technical
  _ Current Multi-Technical (If they exist)
  _ New Multi-Technical
  _ IT/Network Management
  _ Consultant
  _ Manufacturer Professional Services
  _ Internet Service Provider
  _ Data Systems Integrator
  _ Voice Systems Integrator
  _ Telephony Service Provider
_ Service/Feature Support
  _ Needed telephony features
  _ Voice Band Data (FAX, Modem)
  _ DTMF Support
_ SLA Compliance Modeling and Prediction
  _ Predict service availability for voice/telephony services
  _ Predict MOS for voice/telephony services
  _ Predict service availability for data services
  _ Predict service availability for video/IPTV services
  _ Predict Time-to-Dial Tone (TDTT) for voice/telephony services
  _ Predict Call Set-Up Time for voice/telephony services
  _ Predict Call Tear-Down Time for voice/telephony services
  _ Predict delay for voice/telephony services
  _ Predict delay for data services
  _ Predict delay for video/IPTV services
  _ Predict delay variation for voice/telephony services
  _ Predict delay variation for data services
  _ Predict delay variation for video/IPTV services
  _ Predict packet loss for voice/telephony services
  _ Predict packet loss for data services
  _ Predict packet loss for video/IPTV services
  _ Distance sensitivity for delay and delay variation
  _ Predict high/low performance for mix traffic loads
  _ Predict Mean-Time-To-Repair
  _ Predict Mean-Time-Between-Failure
  _ Compare different SLA Scenarios
  _ Model Network-Only SLA Compliance
  _ Model Network+Access SLA Compliance

_ Model End System-to-End System SLA Compliance

_ Component Simulation

_Simulate IPT Phone Performance

- All codecs
- Compression
- Encryption
- Silence Suppression/Voice Activity Detection
- Multi-Line
- Multi-Media
- Ethernet-10M,100M,1G,WiFi

_ Simulate VoIP SoftPhone Performance

- All codecs
- Compression
- Encryption
- Silence Suppression/Voice Activity Detection
- Multi-Line
- Multi-Media
- Ethernet-10M,100M,1G,WiFi

_ Simulate Analog Terminal Adapter Performance

- All codecs
- Compression
- Encryption
- Silence Suppression/Voice Activity Detection
- Multi-Line
- Multi-Media
- Ethernet-10M,100M,1G,WiFi

_ Simulate SoftSwitch/IP PBX Performance

- Simultaneous Calls
- Busy Hour Call Attempts
- Max number of stations

_ Simulate Gateway/BGC Performance

- Total Simultaneous Calls
- Busy Hour Call Attempts
- Security Filtering/Monitoring

_ Power & Power Back-Up

- _ QoS / QoE Forecasting
  - _ Forecast IPT Mean Opinion Score (MOS) using G.107 E-Model
  - _ Forecast Data QoS parameters
  - _ Forecast Video QoS
  - _ Simulate Differentiated Services QoS (DiffServ)
  - _ Simulate 802.1p/q VLAN Prioritization
  - _ Simulate Congestion Avoidance (RED, WRED, Flow-based WRED)
  - _ Simulate QoS Packet Marking
  - _ Simulate QoS Congestion Management (WFQ,CBWFQ, FIFO,LLQ, PQ)
  - _ Simulate RTP Priority Mechanisms (IP & FR)
  - _ Simulate RTCP Feedback Loops
  - _ Simulate NetFlow application performance
  - _ Simulate impact of access bandwidth differences
  - _ Simulate impact of backbone bandwidth differences
- _ Validation of Design
- _ Design Tuning
- _ Test Bed Architecture and Proof of Concept

Implementation and Migration

- _ Planning
  - _ Plan Validation

    Visualization

    Role Play

    Naysayer Validation

    Non-Expert Validation

    Red Teams
- _ Implementation Plan
- _ Training Plan(s)
- _ Test/Acceptance Plan(s)
- _ Migration Plan(s)
- _ On-Going Operations Plan(s)
- _ Contingency Plan
- _ Business/System Continuity Plan
- _ Escalation Procedure(s)
- _ Security Issues
  - _ Liaison with Security Department
  - _ Firewalls & Proxy Server Issues
- _ IP Addresses & Port Numbers
- _ NAT, Internal and External IP Addresses
- _ Internal & External Phone number mapping
- _ Purchasing & Project Management
  - _ Establish Site Profiles
  - _ Rough-Order-of-Magnitude Pricing
  - _ Management Review and Site Prioritization

    _ Initial Migration/Green Field Plan & Renegotiation of Favorable Contracts
    _ Joint Migration/Green Field Planning Effort
    _ Contracts & Delivery
    _ Surplus Equipment Management Effort
    _ Inventory & Fixed Asset Accounting & Disposition of Old Equipment
_ Implementation of Changes to Business Operations
    _ Training
    _ Dealing with User Reluctance
    _ Standardization
    _ Impact Assessment
    _ Changes to Processes
    _ Staff Implications
_ Review of Standards Status and Maturity
_ Develop Private Net, IP VPN and Carrier/PTT Strategies
_ Private Backbone Design/Upsize for Voice Traffic and Backbone Re-engineering
_ Establish Common Standards and Test Plan
_ Establish Private Net, IP VPN and Carrier/PTT Demo/Test Capabilities
_ Training/Staging for Migration Teams
_ Training for Support Teams
_ Review of Planning and New Telephony Project Roll-Outs
_ Legal
    _ Closing contracts for old telephony system
    _ New contracts
    _ Compliance with National Law & Telco/PTT Regulation
_ Security & Privacy Issues
_ Carrier/PTT IP Bypass & Local Tail Drop-Off
_ PBX and Gateway Issues
_ Contact Centers / Call Centers
    _ Special Considerations & Contingency Planning
    _ Phased Implementation & Call-taker Training

Ongoing Operations

_ Review the original plans, goals and objectives

Are the original goals and objectives still valid?

Have technologies changed significantly?

Are standards more mature, and does it matter?

Have laws or regulations in any of the areas in which we operate changed in any important ways?

Have our operating procedures or business environment evolved in a way that impacts our new telephony system?

_ Growth Management
  _ Fixed Assets and Equipment Budgets
  _ Network Bandwidth: Access and Backbone
  _ Shared Resources: Switches, Routers, Call Servers, IP PBXs, Gateways
  _ Numbers and Licenses
  _ Record Keeping & Documentation
_ Fixed Asset Accounting
_ Shipping / Warehousing / Tracking
_ Repair / Refurbishment / Return to Service / End-of-Life

Optimization

_ Constant Testing and Monitoring
_ Exception Reporting and Filtering
_ "What If" Scenarios and Contingency Plans
_ Trending and Capacity Planning
  _ Bandwidth
  _ Ports and Lines
  _ Phone Numbers and Numbering Plan
  _ Grooming
  _ VLAN Capacity
  _ VPN Capacity
  _ Infrastructure
_ Hardware
  _ Repair
  _ Obsolescence / Refresh
_ Software
  _ Maintenance
  _ Enhancements
  _ Upgrades
  _ Patches & Security Audit

_ SLA Improvements
  _ Delay
    _ Call Set-up
    _ During Call
    _ Call Tear Down / Release
  _ Delay Variation
  _ Sample and Packet Loss
  _ Availability
    _ Telephony Service Availability
    _ Grade of Service (GoS) Measurement
    _ Busy Hour Call Attempts (BHCAs)
    _ Busy Hour Call Completions (BHCCs)
  _ Price and Cost Optimization
    _ Hardware Costs
    _ Service and Support Costs
    _ In-Sourcing vs. Out-Sourcing
  _ IP Contact Center Optimization

Operations & Security

_ Tracking of SLA parameters: packet loss
_ Tracking of SLA parameters: delay
_ Tracking of SLA parameters: delay variation/jitter
_ Tracking of SLA parameters: availability
_ Tracking of SLA parameters: user profiles
_ Tracking of SLA parameters: voice quality metrics
_ Tracking of SLA parameters: MTBF
_ Tracking of SLA parameters: MTTR
_ Tracking of SLA parameters: Bandwidth per call
_ SLA Compliance Exception Reporting
_ Tracking of protocols used
_ Tracking of services and applications used
_ Tracking of out-of-service time for circuits
_ Tracking of out-of-service time for components
_Availability tracking for TDM circuits
_ Security events reporting: encryption and key management
_ Security events reporting: tunnels and secure tunnels
_ Security events reporting: user access info & violations
_ Security events reporting: DoS attempts
_ Security events reporting: firewall performance/blocked intrusions
_ Security events reporting: SW release levels & patch history
_ Security events reporting: Access Retries
_ Security events reporting: Session Hijacking Attempts
_ Security events reporting: Eavesdropping Attempts

Administration

_ Track/Audit service provider call detail records (CDR)
_ Track/Audit bandwidth and resource usage by user and user profile
_ Track/Audit minutes of use by user and user profile
_ Track/Audit applications by user and user profile
_ Track/Audit features and services used by user and user profile
_ Reconcile Telco Carrier Billing
_ Reconcile Managed Service Provider Billing
_ Reconcile Service Provider Billing
_ Automate SLA Compliance Penalties
_ Report Calculated MOS by user and user profile
_ Correlate MOS with SLA compliance
_ Report costs per user and user profile

Provisioning

_ Remote configuration of hard phones
_ Remote configuration of soft phones
_ Remote configuration of call servers
_ Remote configuration of IP PBXs
_ Remote configuration of gateways and Session Border Controllers
_ Implement Secure Shell (SSH)
_ Supports multiple levels of service technician log-in
_ Tracking of moves/adds/changes by service tech
_ Automated fall-back to last known good configuration
_ Automated bulk configuration using profiles and templates
_ Pre-configuration of IP Phones and ATAs prior to shipping
_ Tracking of user changes
_ Support for Tripwire or similar anti-tampering functionality
_ Other configuration stabilization and anti-tampering
_ Support for secure FTP
_ Support of Trivial File Transfer Protocol for config download
_ Support of anonymous File Transfer Protocol for config download
_ Support of Telnet for remote configuration

Maintenance

_ Track moves/adds/changes
_ Track software levels, patches and upgrades
_ Track firmware levels, patches and upgrades
_ Track/Audit IP hard phones and ATAs
_ Track/Audit IP soft phones
_ Track/Audit IP PBXs
_ Track/Audit IP gateways and Session Border Controllers (SBC)
_ Track/Audit services and features used
_ Testing of Services and Features
_ Remote Testing of Services and Features
_ Automated Testing of Services and Features
_ Hardware preventative maintenance
_ Spare inventory management
_ Maintenance contract tracking
_ Warranty tracking
_ Service and Trouble Ticket Tracking
_ Trouble Ticket Trends and Statistics
_ Automated/Scripted Root Cause Analysis (RCA)
_ Escalation Procedures