NEC Business Solutions Ltd

Minimum Requirements for IP Telephony

NEC Business Solutions

Version Report

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Contents

A topical customer question:

What sort of considerations are necessary when implementing Voice over IP (VoIP)?

Implementing Voice over IP involves a number of business and technical decisions. These decisions seem unrelated at first to the initial objective of integrating internal or external voice traffic with existing data infrastructure in an organization. This in itself illustrates part of the issue with this technology... The simple task of adding a VoIP interface to a PABX has an implication for the internal and external data networks throughout the organization. The flexibility VoIP introduces then has financial and business process impacts where ever it is implemented.

The widespread cost impact of a substantial change throughout an organization requires careful examination of reasons for introducing VoIP and IP Telephony. A number of justifications have emerged in support of this change:

- VoIP allows several cost savings to be gained
- VoIP satisfies a new business need
- VoIP gives a competitive advantage to the organization, as one which is able to visibly leverage rapid technology change
- VoIP provides a management tool for driving organizational change

Most organizations have a formally stated or hidden reasons for deploying VoIP which covers at least several of these justifications. For example, a distributed call centre may be much more cost effective, as staff from lower cost regional or international centres may be effectively utilized. This option has staff management issues, as the staff are no longer centralised. Similarly , from a voice perspective, working from home or on the road can be quite practical. The customer ideally has no idea of the customer facing staff's actual location. Finally, suppliers or business partners can become part of the wider Enterprise voice network.

All these options may or may not be desirable from the Enterprise perspective and certainly has wide impacts outside the basic cost and provisioning of a VoIP adapter. Human resources, Unions, liability issues arise to name a few implications.

From a personnel perspective, the introduction of VoIP also involves a complete vocabulary of terms and acronyms with which both business and traditional voice technical staff may be uncomfortable with today. For example, G729 is a voice compression standard, VoIP is used for Voice over IP and QOS for Quality of Services, etc.

The exact network technologies chosen to implement the IP Telephony system with VoIP need consideration. Can the organizations LAN switches support quality VoIP. The type of protocols can also have impact. For example, a common voice over IP protocol called H323 may be used which makes the use of third party "phone" clients possible. These types of non-PABX clients may provide VoIP of a particular quality but may be missing many features required for an Enterprises business, which an IP Telephony system can deliver. This change to reflected in the gradual change from the concept of a PABX which stands for a Private Automatic Branch Exchange to the concept of an Enterprise Communications Platform or ECP.

This choice is one of the differences between simple VoIP and IP Telephony. For a particular task there are a number of other factors that need to be considered in any technology roll-out in today's enterprises. [1]

Other factors that require consideration include the integration with other business processes within the enterprise. The business processes may involve a work flow implemented with workgroup products, integrated with voice mail and messaging. Reliability may be a critical issue. It may be justified to use high availability servers to provide these one or more of the network applications. This type of server application is required to implement a comprehensive VoIP solution in many instances.

The rest of this white-paper will expand the points raised above with some potential solutions under a number of key areas. In the case of VoIP, some of these considerations are:

- **Quality of Service (QOS)**
- **Security**
- **Power**
- **Network Management**
- **Personnel Resourcing**
- **Deployment Plan**

The above items are central to IP Telephony and VoIP projects, however, this is not an exhaustive list.

[1] **Definition:** In this white-paper, a large/medium/small, government/public company or private company will be called an enterprise.

VoIP and Associated Terminology

VoIP

VoIP is the carriage of normal human voice in the format of Internet Protocol (IP). Current data networks can understand and transport the IP packets. This may be privately via Enterprise Intranets or using the public Internet, typically with an encrypted Virtual Private Network (VPN) tunnel for security.

IP Telephony

IP Telephony is the carriage of voice traffic in a similar manner to that provided by our familiar PABX networks of today. The transport technology happens to be IP. It will have full access to the features and facilities various groups of enterprise users have come to expect for today's PABXs. The result is Telephony which must be transparent to the business functions, whether it is delivered via normal means or VoIP.

VoIP is implemented as voice samples or packets to which an addressing structure has been attached. For the technical reader, telephone voice has an audio spectrum of 300Hz-3.8KHz which in digital format becomes a 64 Kbps data stream.

Toll bypass

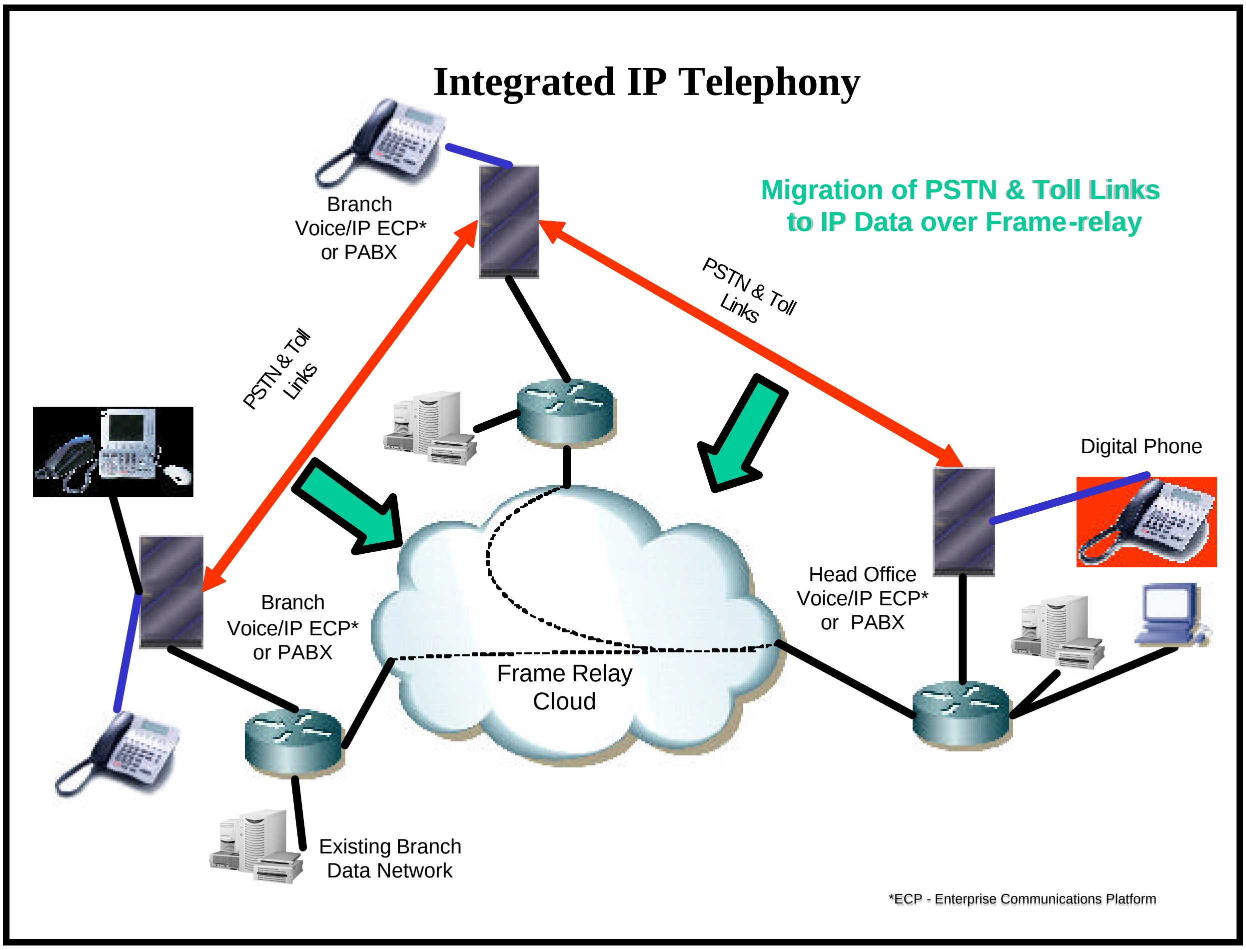
Toll bypass is an early phase of introducing packetised voice into an existing data network. A clearly identifiable cost to any distributed enterprise is the relatively high cost of Wide Area Network (WAN) connections from telecommunications carriers in Australia. These charges have a connection component and data rate or packet rate charging mechanism. Frequently the tariff used for charging for these WAN data services is distance independent. In the case of voice using Telco Toll services for intrastate, interstate or international calls these calls are timed and are NOT distance independent. The financial benefit of consolidating a number of smaller links into one large link is significant. Avoiding Toll charging is nearly always a major cost reduction per unit of voice/data traffic.

It was readily demonstrated a few years ago that the pay back period for an integrated Voice (VoIP) and data link could be as low as sixty days. This clearly attractive option justified many early Toll bypass business cases. Today this approach is still viable, however the rapidly changing mix of Government regulation, competitive data services and the contraction in the Telecommunications market worldwide means these savings are variable. They need regular review in the light of the contracts each Enterprise has with their carrier(s).

Voice Data Convergence.

Voice data convergence can be interpreted in a number of ways. At a basic level it may be taken to mean the carriage of voice and data on the same link. The data may be mixed by using physical services such as frame-relay or Asynchronous Transfer Mode (ATM) or at the packet level by using an IP transport for the data and IP (VoIP) for the voice.

A much more powerful way of thinking about voice/data convergence is when the voice and data streams are considered at an information level rather than just a transport level. Then a business application such as a call center can then focus upon improved customer service rather than purely cost savings. For example, integrating the customer information from the data network directory, billing, complaints systems, etc., with that provided from the telecommunications system can make the enterprise appear more responsive. The PABX can provide caller information such as caller-id which can result in the service representative having the caller's data on screen before answering the call. This can be taken further with full integration to voice and video services supporting customers via the Internet using the Web. To reflect the changing role and structure of a traditional PABX in a IP world, the name "PABX" has itself evolved to "ECP" or Enterprise Communications Platform.



VoIP Implementation Considerations

The implementation of a VoIP solution may have a much wider impact on the enterprise than the immediate technology changes involved. This issue is expanded in the following sections of this paper. Part of the management challenge is a simple request to MIS to reduce Telco recurrent charges may result in a proposal to integrate voice onto the existing data infrastructure. This business case can be approved on its merits and while the recurrent charges for those services are reduced, the enterprise has missed the full benefits of implementing an IP Telephony solution.

An IP Telephony solution must be considered from a whole of enterprise perspective, as it has wide ramifications. The obvious ones like upgrading certain types of infrastructure and changing voice management processes are easy to execute. The underlying implications of increase mobility of staff, the ability to rapidly move business functions across multiple time zones or counties, the concept of "hot-desking" staff, mobile professional workers, security of voice (and data) services in these environments, new Web centric services with voice customer access, etc., really involve major cultural and management change.

VoIP may not be suited for everyone

Today any business case must provide hard savings/benefits to any enterprise and a VoIP implementation must clear this hurdle. Even if the business case is successful on a business case level there are a number of other reasons a transition to VoIP may not be suitable for everyone.

There are many enterprises which may not be willing, able or need to make this paradigm shift. In these organizations the management function may not lend itself to a flat highly mobile flexible professional staff structure. The business groups may require a more rigid command structure with the work flow process highly optimized for the industry or customer/client set. In this situation delaying VoIP is nearly always the least expensive option in the short term. If the existing voice infrastructure is meeting current and near term business needs, then why change? The early-to-near term adopters can face some additional costs. However, if acceleration of change within the enterprise is important to respond to competitive challenge then the delay in introduction of changes like VoIP may be costly to the survival of the enterprise long term.

The early adopters of any change tend to be in the minority. The bulk of the rest of the 5-6 groups of change adopters will move to VoIP once it has become more THE standard, priced at lower cost/commodity level and essentially transparent to the user. Most marketing models show the adoption process taking several years at minimum. An example of VoIP already in this category is the very low cost part of the international toll call market. Calls are presently available to the USA and Europe for of the order of 5-9 cents/minute.

These are implemented underneath by the Telco using the cost savings of VoIP compression but the end user is unaware of the details. The early adopters of VoIP on the Internet had microphones/headsets on their home PCs using IP shareware applications over five years ago. They have now moved onto Video Conferencing a wide range of instant messaging and video network applications. Since deregulation the home telephone user making these low cost international calls via VoIP, uses their home Telecom handset. It is essentially transparent to the phone user and becoming ubiquitous without the bulk of late adopters even knowing.

Although a VoIP solution today may be perceived as new and possibly more expensive, over time the older generations of voice equipment will become unsupported and increasingly expensive to maintain. At this point in the migration to VoIP, and eventually full IP Telephony, will become essentially compulsory as the alternatives cease to be marketed. There are many parallels to this process, which may take 7-15 years. An example is the Beta/VHS migration. It took less than 10 years for Beta to become essentially non-existent from being market leader.

Costs – infrastructure, training resources, power.

Once a decision to adopt VoIP or a full IP Telephony solution is made then the business case will have identified all the related costs and benefits. Each of the major cost areas a dealt with in subsequent sections of this paper. A key item is the infrastructure itself. Depending upon the extent of the VoIP implementation this may include:

- the upgrade on Local Area Network (LAN) components,
- Wide Area Network components (WAN),
- consolidation of Telco WAN links,
- addition of Wireless LANs,
- new handsets,
- PABX upgrades,
- the adoption of a Security Policy
- Expanded Network Management,
- Power supply upgrades (Including UPS),
- and most importantly the upgrade of staff skills with training to support the above systems.

The cost of VoIP is depends rather on what quality of voice one is trying to achieve. For a business conferencing system very high quality is essential. By comparison, a metallic sounding voice is considered quite acceptable for battle field VoIP, as long as the data is compressed as much as possible. A smaller data rate decreases the chance vital military orders are lost on the noisy radio links on a battlefield and makes the signal harder to find and jam. Quality of Service or QOS is thus an important consideration and will be covered in the next section.

QOS

It is not recommended to implement a VoIP network without first assessing the Quality of Service or QOS capability of the underlying enterprise network. This clearly is an issue on low bandwidth lines but surprisingly, can also be a major concern in the internal LAN backbones which have dramatically higher bandwidths than WAN links. The old adage of "just throw more bandwidth at it" is not applicable in a VoIP application nor in other time sensitive applications such as Video dstribution. In order to clearly understand QOS as it is applied to voice, including Voice over IP (VoIP), it is essential to understand voice quality in a general sense.

What is Quality Voice?

A distinguishing characteristic of a voice conversation is that it is real time compared to most data traffic which is not time critical. The human ear perceives a quality conversation to be that of two people talking at a normal interpersonal distance (for a given culture or race) and with nominal background noise. The acoustic hardness of the surroundings is important for an indoor conversation as there should be little, if any, room echo. The next question is how do you know when you have "Quality Voice"? Traditionally, you ask people. The result of a formal survey is a Mean Opinion Score (MOS).

> **Mean Opinion Score (MOS) on a scale of 0-5. A MOS of 5 is our reference conversation above. A MOS of 4.1 is "TOLL/PABX" quality (64Kbps G711), a MOS of 3.92 is "standard" VoIP at 8 Kbps (G.729 CS-ACELP), a MOS of below 3 is unacceptable, and finally a MOS of below 1 is unintelligible.**

Data QOS and Response Time

A gradual approach is generally acceptable when transitioning to VoIP. The objective is to integrate voice with minimal disruption to your existing data network. For most data networks measures of quality are normally related to end-user's perception of application response time and throughput. This is best done as an internal trial (non-customer facing) network with minimal risk. The use of compressed VoIP on trunk or WAN links has been used because of the clear cost savings. This only tests the transport aspects of VoIP which is only a small portion of IP Telephony. Typically a trial of IP Telephony is tried on the internal company LAN. This is where the more subtle QOS issues begin to arise.

LAN VoIP

On a LAN there are several sources of other traffic which can impact the QOS of the small but real time voice packets. The concern with many LAN networks is that they are based on using larger and larger pipes to handle mixtures of packet sizes, typically from 64 to 1500 Bytes[2] of ever increasing volume. Typical LAN switches have as single large buffer on each port, of the

[2] 1 Byte =8 bits

order of 180 K Bytes. A file transfer will fill the buffer with many of the large packets thus delaying the smaller, time critical, VoIP packets. A switch on the network edge needs the addition of a separate buffer for the voice packets. It also needs the ability to segregate the VoIP packets into this higher priority VoIP buffer.

As most Enterprises have moved to switched networks, the contention of protocols such as Ethernet has been largely resolved. The older LAN Hubs which share the LAN capacity are quite unsuitable for VoIP. This leaves the normal traffic of network broadcasts of various types, multicasts of items multimedia such as video and music, large data file transfers, etc. This data traffic is in contention with the voice traffic (VoIP). A common solution is to use separate virtual LANs or VLANs to segregate the voice traffic from the data traffic. This voice traffic can be then tagged and prioritised so that it will be given similar priority treatment, as it moves to the destination phone elsewhere in the network

VoIP Packet Sizing

Clearly, even an uncompressed VoIP stream of a single traditional digital voice using the TOLL/PABX standard of 64Kbps is small in comparison to a 10 MBps or 100 Mbps Ethernet link (see side box about digitizing the Human Voice). With IP addresses added, the data rate for a VoIP call reaches a relatively small 80 Kbps.

WAN VoIP

A compressed voice stream is more suitable for a WAN link. There is generally no need to introduce the additional delay of the voice compression, if the call is generated and terminated within an enterprise LAN network. The popular compressed voice G729 Codec produces an 8Kbps data stream. If the IP address overhead is also compressed the total reaches about 14Kbps per voice call. There are options of implementing the compression in the PABX/ECP[3] or externally in the edge routers. This choice is generally made according to which is the lower cost solution once all technical and QOS constraints have been met. Like all things in the technical world, this compression comes at some cost. The quality drops as discussed above (MOS score), it costs processor power (and $) for each Digital Signal Processor (DSP) used in the compression, and each DSP

Digitizing the Human Voice

The human voice is sampled 8000 times per second. Each sample is an 8 bit quantity shaped to match the human voice with an A-law or mu-law filter. These samples are grouped in small batches for transmission on the network. The industry uses either 50 batches or 25 batches per second. The difficulty lies in the need to have 50 batches per second of 1600 bits (200 bytes) each arrive regularly (every 20mS), in sequence, with none lost. If these relatively small packets get behind a big 1500 Byte packet in a queue somewhere in a LAN switch, or worse, clocked onto a slow WAN line, of say 128 Kbps, then all sorts of variable delays (known as Jitter) start being introduced to our voice stream.

Characteristics of G729

The steady G729 data stream of 10 Byte packets every 20 mS or 20 Byte packets every 40 mS (10x50x8=8000 bps) By comparison, the data packets are up to 1500 Bytes.

[3] ECP – Enterprise Communications Platform

introduces an additional delay of about 15mS for G729. For high cost WAN connections the link capacity saved is generally considered worth accepting these tradeoffs.

Solution Using Traffic Separation.

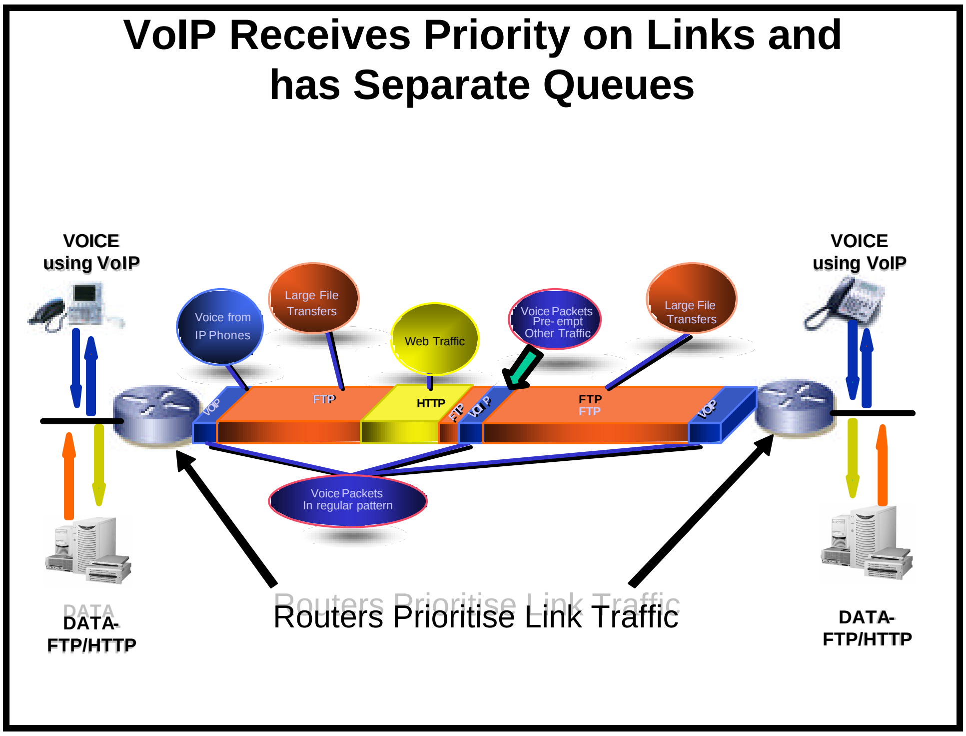
VoIP is composed of many small packets which may still get jammed behind the larger pieces of data on the LAN or WAN. This is rather like a sports car being stuck behind a long line of semi-trailers on the motorway. A solution to the truck problem, used in the United Kingdom, is they are not allowed out into the fast lane on motorways. (... something Australia could adopt.)

For our VoIP quality requirements all the technology fixes or "knobs" the engineers have to resolve the dilemma where small packets get jammed behind larger packets basically come down to the same solution.

The Enterprise network must be optimised for VoIP. This is applicable to both wide area and local networks. The problems are slightly different, as are the solutions.

The goal is the same: namely to maintain the voice quality our customers have come to expect from traditional voice services such as Telco phone switches or PABX switches.

The benefit is the cost savings to be made from integration of Voice applications onto a single converged infrastructure. The business case savings are covered in an associated white paper.



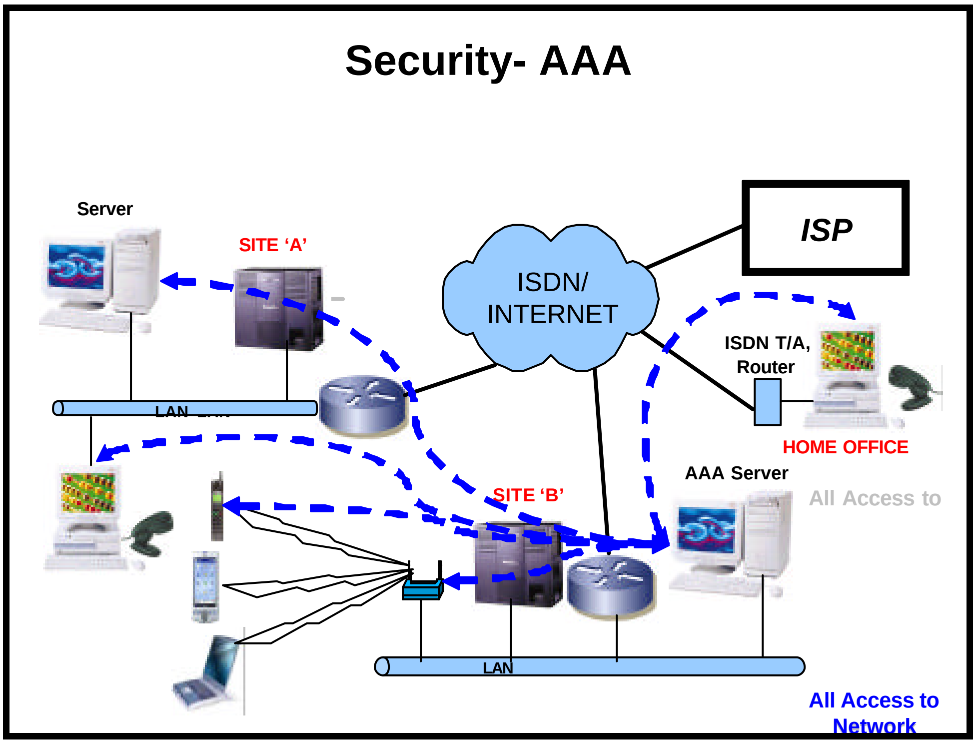
Security

A Voice over IP solution is no more or less secure than a traditional voice call. The only thing that changes is the method of attack if a third party wishes to tap or record conversations or even impersonate a caller. The VoIP system actually can be seen as allowing some advantages over a traditional voice network. It can make use of all the security mechanisms previously developed for securing our existing data traffic. Existing data is for a large part already running in an IP centric world. Within IP there are several variations in the approach to providing security.

The issue with voice is that the security comes at a cost. That cost has one particular dimension of importance to time critical voice, the delay. Encryption takes time per packet and therefore adds to the delay across the network. Authentication takes time and the data authentication methods require facilities not present in traditional handsets and ask the user to interact in a non-phone manner e.g. logging on before dialling. Before looking at some of the security implications for a network upon which we can run a voice call via VoIP, it is worth reviewing the concepts of electronic signatures and Authentication, Authorization and Accounting, or AAA, as it commonly referred to in the data world.

Electronic Signature and Validation plus AAA Services

Electronic Signature and Validation is included in current AAA authentication services implementations. A discussion of this area first requires a definition of what AAA actually means. In communications terms, AAA stands for Authentication, Authorization and Accounting. The underlying concept is that in order to provide controlled access to all sorts of resources there needs to be a structured approach which spans each of these areas. The requirements in this area have been evolving over time. For example, the recent emphasis on cost control, outsourcing and charge-back to end-user departments has increased the need to accurately measure the time and type of resources end-users have access to, once their identity has been established, and after their right to access that type of resource verified. The reporting and billing functions frequently receive more attention today than a purely security conscious policy would proscribe.



Summary of AAA

Authentication

Authentication techniques range from a simple user-id/password, user-id/single use passwords (e.g. SecureID), and to the much more specific identification procedures such as thumb prints/retina scans etc, based on types of biometric data.

Authorization

Authorization is the process by which users are assigned access to specific resources within the Enterprise.

Accounting

Accounting for access to resources was initially envisaged on the basis of tracking who did what to whom, including billing of transactions or use.
Electronic Signatures and Validation
A popular recent advance is the use electronic identification facilities. The sort of approach can take a number of forms. The most common is the use of a series of electronic keys or certificates to identify individuals, end devices such as PCs or network facilities such as encrypted tunnels and VPNs.

Electronic Signatures and AAA for VoIP

Today the standard electronic phone does not employ the whole concept of AAA directly but already has some elements. Most proprietary digital phones for example, have to be defined to the PABX before calls may be placed. Certain services (like applications in a data context) require authorization, such as STD or Toll dialling which may be restricted or barred until a PIN or access code is entered. Specifically PABXs and main Telco exchanges have

historically had very strong call accounting for recording usage charges etc. All the same, these facilities are generally quite proprietary. The AAA used in data networks is generally based on notional standards with some proprietary extensions, e.g. Radius authentication servers, certificates, etc.

Phone location

The location of the phone is an important security and safety issue. If the phone is authenticated correctly, but is not located on the correct part of the network, it may be now be able to access resources not available in the original location. The phone user may have avoided authorization by moving to where it is now connected.

The issue with VoIP handsets is that their very advantage of total portability is actually their greatest difficulty. They are not restricted to one specifically define physical port, unless this is a specifically selected restriction in the design of the network.

This portability has some unintended consequences, as it is a legal requirement in many parts of the world to be able to define a handset location from an emergency services perspective. The 911 legislation in the USA is a prime example of this requirement. The addition of Wireless LANs (as discussed in a following section) makes the roaming aspect of an IP handset even more difficult to track and resolve. It is worth remembering that the VoIP handset may be a PDA or laptop with a phone client. In this case it is possible to identify the device with a certificate loaded on the local storage. In the case of may VoIP handsets, there is frequently no provision for storing an electronic certificate and no certificate distribution mechanism in the phone or indeed the network voice application or PABX. A smart voice product designer of VoIP network products would certainly be keen to see such a facility in their VoIP handsets, PDAs, Laptops and Voice Network Servers, ... let us call them ECPs[4] i.e updated "PABXs".

Voice Path Security.

The prevention of eavesdropping on voice conversations is considered mandatory or at least desirable. This is particularly true in sensitive areas such as financial/commercial or Government security. The choice of algorithm is partly driven by the packet delay considerations and is based upon a balance between the security requirement, the algorithmic efficiency and the memory and processor present in the VoIP handset. If one makes a commitment to a certificate based architecture, then the phone is clearly identified. The certificate may be used to create an encryption key to secure the voice path.

Infrastructure

The sort of control offered by implementing AAA is not without a cost penalty. The very need to introduce a consistent scaleable implementation across the

[4] ECP – Enterprise Communications Platform

Enterprise necessitates a centralization of the control functions. This introduces some single point of failure into the overall network. Normal redundancy features must be designed into the network and good engineering practices employed to deliver a consistent and pleasant end-user environment.

Project planning, change control, service level agreements are the sort of management tools which must be applied.
Complications with the process are:

- **The need to have a central certificate server**
- **The authorization process typically makes use of a Radius Server function (Instead of Radius some implementations use Kerberos)**
- **Network devices become clients of the Radius Server**
- **The Radius Server may make access to resources via a centralized service such as directories based on LDAP**
- **The directory in turn uses the Certificates to implement two way authentication**
- **Normal backup/UPS/Physical Access control of Network Servers**
- **Management resource to issue and track certificates**
- **A reliable and secure distribution method for the certificates**
- **Recovery procedures in the event an certificate is corrupted or destroyed**

Example:

A typical problem which may arise is the lack of access to the certificate whilst in a remote location. If the certificate is portable and being used to authenticate a logon to a secure web site this still creates a difficulty when using it from a customer site or a coffee shop. Firstly there may be no way to access or load a certificate in the remote web terminal, and secondly the wisdom of leaving copies of your certificate on PCs scattered around the internet is questionable. A typical response is to use your laptop (with certificate installed). This sounds like good idea until you break your LAN cable or some other part of the Laptop while on a business trip to Sydney or overseas. The business impact of failures in parts of the electronic signature implementation must be part of the feasibility/risk analysis of the business case supporting the improved security they can offer. This comes at a cost and may not fit into some parts of the Security Policy adopted by the enterprise regularly, etc. The encryption (VPN) and authentication techniques discussed above are applicable to the other VOIP WAN transmission technologies mentioned in previous sections. No single technique provides a complete solution and the rapid advances in security attack tools and defensive measures means an ongoing and flexible security policy (documented) is essential. This will involve proactive self attack and intrusion detection tools as well.

Power

A number of manufacturers have announced support for the IEEE standard for in-line power for Ethernet attached IP handsets. The IEEE 802.3af standard is presently in draft format and is estimated to be completed and formally accepted by the end of 2002.

A quick search on the web leads to the IEEE summary page. http://www.ieee802.org/3/af/index.html. The relevant standard is P802.3AF "Draft 3.2 Supplement to CSMA/CD access method and physical layer specifications - Data Terminal Equipment (DTE) Power via Media Dependent Interface (MDI)" 2002.

There are two variations of power feeding on the Cat 5 cable using differing combinations of pairs of wires. Both standards have found support on major switch vendors and suppliers of other enterprise LAN equipment such as Wireless LAN access points, IP telephone handsets, etc. Some devices support both wiring options. The exact requirements need to be established in your situation. The other important factor is the maximum load supplied per port. The draft standard specifies 15 watts.

There are two methods of adding the inline power feature. The first is to supply an inline pass-through panel, which inserts the inline power for an existing Ethernet switches. The second is to add it to the hardware present in the Ethernet switch. This almost always requires a replacement switch in the case of stackable switches. In the case of a chassis based Ethernet switch, selected modules can be added or changed. There is a power supply limitation on some designs. Adding several modules in increments of 24 ports, each needing up to 15 watts of additional power per port, may overwhelm the original power supply.

Another option is a small power insertion device with a mains feed plug-pack and RJ45 in & out tap-in block is suitable for single devices such as a WiFi Wireless LAN Access Point. This device is frequently ceiling mounted and an available power point may be difficult to supply. The inline power may only be justified for a single port on a multipoint LAN switch connected to a particular Access Point device. It would be uneconomic to upgrade a complete switch blade of perhaps twenty-four ports to inline power.

UPS Power Requirements

In many parts of the world it is a requirement that certain phone systems must be kept up in the event of a mains power failure. In the case of VoIP implemented to the desktop, the power for the handset may be coming from the LAN switch with in-line power. If the handset must remain powered for legal or practical reasons, an appropriately dimensioned Un-interruptible Power Supply or UPS must be provisioned for all switches and router components. The same considerations must be given to any mission critical network appliances such as message systems, ECPs[5], PABXs and wireless LANs, including authentication servers.

[5] ECP – Enterprise Communications Platform

Network Management

Network management spans a wide range of tasks ranging from management of the end devices in a network, through to the devices or applications providing the end user service. It really requires and end-to-end view of the tasks and is not restricted to just a Voice or Data component in the case of a successful VoIP deployment.

Voice Components.

PABX devices have their own proprietary command line interfaces which are generally considered unfriendly for end users and take significant engineering training and expertise. Specifically there never has been a wide acceptance of SNMP as a management protocol and even the Teleco oriented CMIP has received minimal support on smaller switch products such as PABXs.

Major PABX vendors are moving to increase the IP orientation of the PABX management facilities with action to make PABX alarms, etc., visible via a Web interface. Many vendors are supporting the SNMP network management protocol on latest PABX or ECP[6] offerings.

Major manufacturers are also including remote access to the PABX management console via a minimum of a "telnet" like facility or client application, again via IP. The PABX console access allows corrective actions to be executed in response to any alarm conditions gathered from the SNMP monitoring. Some implementations use an installed application to provide access, others rely upon telnet like facilities.

Another popular network management access technique is to use a standard Web browser to access a custom web server built into the PABX. Some VoIP implementations have created Java based applets which enable ready configuration of their VoIP services. These have the advantage of being usable from any web browser. They can be dynamically loaded via IP and thus do not require permanent installation on the managing personal computer or Unix workstation. On the client side, vendors have started creating web based interfaces for the IP handsets. This type of tool enables end users to modify their own features such as programmable buttons without assistance from a support group, hence reducing the TCO (Total Cost of Ownership).

Adds, Moves and Changes

A major issue in larger organizations is the cost of adds, moves and changes. Integration with appropriate corporate directories is a direction which is already implemented to a large degree some PABX suppliers. The objective is to empower the end user to update their profiles within the corporate

[6] ECP – Enterprise Communications Platform

security guidelines and improve end user productivity when trying to search for and call other parts of their organization, both within and outside the enterprise.

Most implementations allow users to initiate calls by selection of a person or business function in the directory and then clicking on a call button or double clicking the web entry. If the default action is to call the selected entry the call is initiated and your phone also initiates a form of ringing. The user picks up to complete the call. The directory management functions may be initiated from the browser on a PC, laptop, PDA or mini browser built into the phone. The phone handset may be implemented as a physical phone, a soft phone on a PC or PDA (Personal Digital Assistant).

Voice Quality Management

Voice quality for a VoIP implementation is no less important than on a conventional PABX network. Various voice specific tools are available to monitor voice traffic on voice enabled switches and routers. These range from console interface where the router can be interrogated for lost or dropped packets through to graphical interfaces which display voice quality. These can be stand alone applications or Java based with a Web interface into a Web server which is focal point for network alarms. There is a major trend to integrate the voice and data management functions on a single management platform for larger installations.

There are several large players in the network and systems management toolset arena and a number of suppliers of Network Operation Centre (NOC) packaged services. The other aspect to consider with VoIP implementations is the perception that phones work 7/24, 365 days a year. Voice systems, including VoIP, have stricter availability expectations than data networks have historically delivered. Many customers have found implementing such support internally is very expensive. The NOC chosen must provide this type of service for conventional voice and data networks. The NOC may be based in-house resources or can be outsourced to several leading suppliers of management services. Irrespective of the method of supply, the NOC selected needs to use a specialised set of tools to manage both voice and data. If the customer selects 7/24 service via a support contract, the NOC will normally need to provide this option across a wide range of multi-vendor equipment. VoIP management is best seen as an augmentation of an existing voice and data management facilities.

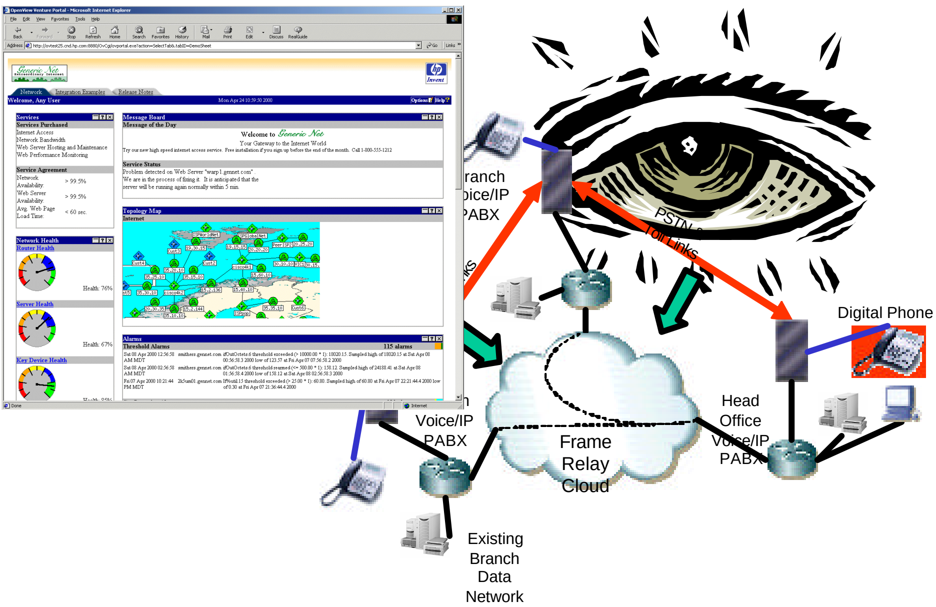
Data components.

The open nature and degree of flexibility of data network management tools based upon SNMP protocols is well known. It is generally accepted that product specific tools be employed to manage smaller installations. For example, many network products have a built-in Web server which can be enabled and then accessed via a Web Browser. Some products include basic tools in the form of a Windows application or Java applet. Others make this type of network element management available though a cut down network manager running on Windows. This is usually a subset of the fully version

which is usually multi-platform. These managers are generally more than pure network managers. They have systems management, security management and other extensions. They also provide the ability to correlate different type of faults and the ability to automate the management response via scripting tools and perhaps expert system engines.

The final element of the picture is that individual vendor products are frequently run as subsystems of a full management suite. These provide an umbrella into which the disparate tools of several suppliers may be integrated. It has been found that this type of network management should be implemented with any sizeable enterprise installation approaching 100 network elements or greater. This is subject to a rigorous cost benefit analysis of the benefits available. As with the voice management discussion above, many customers are struggling with the difficulties of obtaining, training and retaining the high level technical staff required. This has lead to a degree of specialist outsourcing of part or all of the Voice/Data Network Management function to third party network management companies.

Network Management: a Window on the Voice/Data Network.



Personnel Resourcing

The support issues of a VoIP architecture in any enterprise is similar to that of any new technology introduction to the corporate environment. Essential components are as follows:

- **Obtain an executive sponsor, preferably in one of the key business groups who will benefit most from VoIP introduction.**
- **Ensure the IT executive is behind the VoIP program with BOTH MIS and Voice group support.**
- **Obtain budget support to enable and evaluation programme with a trial or prototype project.**
- **Support the project with initially contractors/project managers mixed with a several key members of your IT group who will migrate to run the VoIP architecture.**
- **Implement training of these IT staff and others on the key technologies required for a successful rollout. This will include key potential infrastructure suppliers, major VoIP vendors, general VoIP technology training.**
- **Separately survey the enterprise network, particularly, LAN, WAN, Firewalls, security policy. This technical information needs to be coupled with future business and growth plans to ensure both tactical and strategic views are covered in the VoIP migration plan which will emerge.**
- **Map the network considerations to the present network architecture and vendor(s). If a mismatch is detected select a new network vendor.**
- **Map the Voice Network considerations to the present voice network architecture and vendor(s). If a mismatch is detected select a new PABX vendor. In this case, the migration to VoIP must be a clear part of the existing vendors product plan.**
- **Encourage key support staff to take certification in the products selected.**
- **For example, if a specific vendor(s) switches are selected then the related product related certification is appropriate. Note that this is not just general class room training and normally has an externally moderated certification test(s). These certifications expire over a two to three period, so it is worth checking contractors or suppliers hold *current* certificates.**
- **For the directory and security work a certification on the operating system of choice plus security specialization is appropriate.**
- **Augment these staff with selected contractors for specialist installs e.g. a high level network architect or installer may be appropriate at times but difficult to retain as a permanent employee.**

The other approach is to outsource the whole project or at least the installation component. It is worth ensuring any outsourcing organization or supplier specializes in the VoIP/network integration area. They also need to

have the resources to operate in all your locations within Australia or overseas.

A final solution which has been very successful in the area of traditional voice is to purchase the VoIP service as just that… a IP phone service priced on a per port basis. This is otherwise known as a “managed service” in the PABX world. The pricing for this approach depends on the level of service required. Examples are business day only, 7/24 with SLA in place, full onsite support with Network Operations Centre after hours, etc.

The business case for each organization drives which of these approaches are more appropriate in each case.

Deployment Plan

This part of this white paper is predicated on a suitable business case for implementing VoIP. This business case has been covered in a separate whitepaper. This business case has to be accepted and the strategy to implement VoIP must have received senior management approval, to a least the level of a significant field trial.

The first step in deployment of VoIP to an Enterprise network is a consulting task with a field survey of target end user business needs and the existing data/voice network. Once these have been documented and the tactical or strategic objectives of the field trial or full implementation defined, it is a relatively simple business of engineering a suitable network infrastructure to meet the delta between these two positions.

The existing PABX products need to be augmented, upgraded or replaced with VoIP enabled PABXs i.e. VoIP network application servers called ECPs[7], if this is the way you would prefer to view them. Existing telephone handsets need to be upgraded to VoIP capability and any new VoIP handsets acquired.

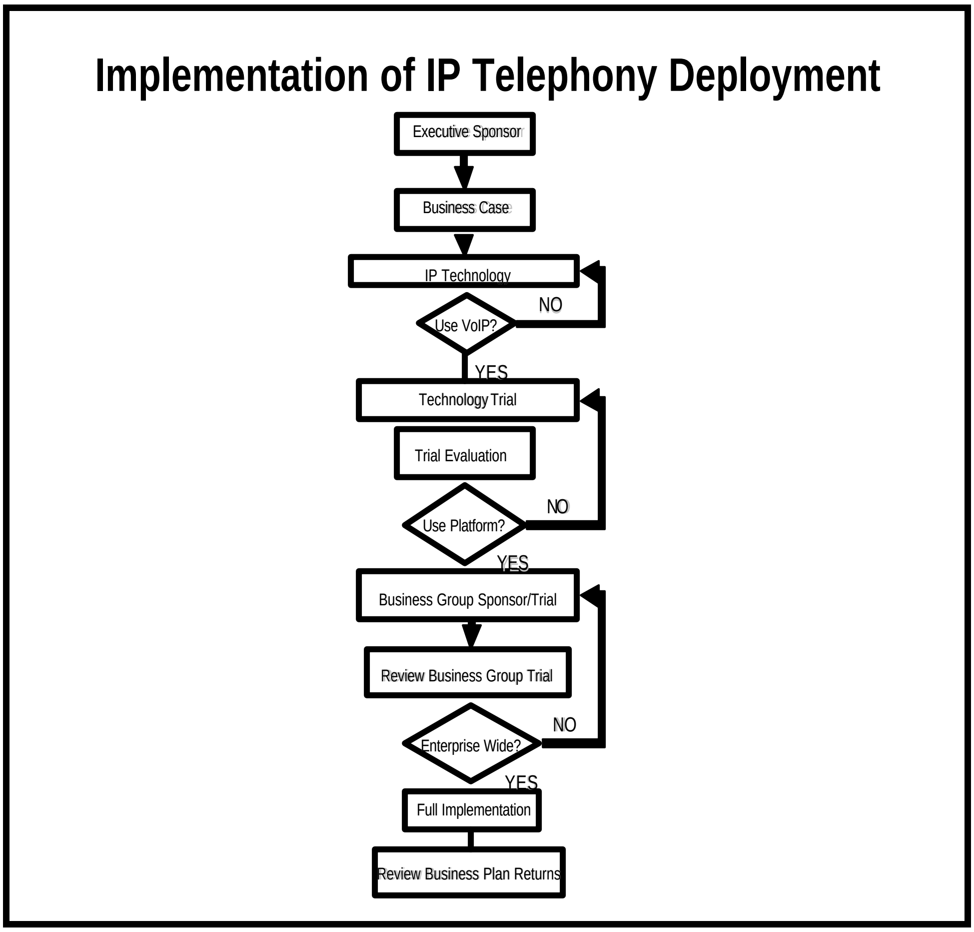
Project Management

The management of an implementation of VoIP across multiple sites, possibly several states and internationally requires a strong and skilful project manager(s) to drive a successful conclusion. Key to any new technology introduction is the careful management of user expectations, clear guidelines regarding seamless cutovers plus the education of, and communications with, all parties involved.

Data Network Upgrade

Typically the underlying enterprise data network will have significant gaps between what is necessary to achieve the QOS objectives for VoIP and those of the existing data applications needs. This generally will require a major upgrade to key LAN switching elements to support multiple queues to support the VoIP QOS. A key benefit of VoIP business cases involves the reduction of existing Voice WAN links. This derives from the reduction of the number of circuits, each of which has a periodic connection charge. The compression usually implemented allows a bandwidth reduction in addition. These cost savings partly justify an enterprise implementing VoIP. As the VoIP traffic is added to the existing data traffic the circuit data rates will have to be increased to some degree. Existing service level agreements (SLA) for data traffic must still be honored and the existing traffic may have timing characteristics which prevent the new VoIP traffic from dominating the revised link capacity. Examples of this are various types of bridged traffic historically used to access mainframe or midrange minicomputers. End users are used to their remote telnet sessions or web access responding within a specific time. If these session freeze when all phone calls are using the same link, this may also limit to what degree the VoIP traffic can dominate individual WAN links.

[7] ECP – Enterprise Communications Platform



Implementation of Deployment

The implementation of an IP telephony network exploiting VoIP is not a single defined process. It rather consists of a series potential steps. The exact implementation will vary greatly from enterprise to enterprise. To establish what will work in you particular organization, some form of consultancy is an appropriate first step. This should be able to define a suitable plan to introduce VoIP, once a suitable business case has been accepted. Some of the steps which can be employed are contained in the following checklist:

- **Obtain executive sponsor**
- **Form technical/business group advisory committee**
- **Commission an appropriate business case**
- **Commission an implementation plan to deliver the benefits revealed by the business case.**
- **This will cover the timescales, technical staff/end user training and project management to meet the projected milestones outlined in the following.**
- **Run a technology evaluation trail to select VoIP technology**
- **Review/select VoIP technologies**
- **Implementation technology approved in small trial**
- **Review enduser/technical team feedback**
- **Extend trial to major business group with best payback**

- **Review business group feedback and technical scaling issues which may have been discovered**
- **Approval of full implementation**
- **Post implementation review**
- **Resolve outstanding issues**
- **Review business case and measured returns have matched predictions on quarterly basis for first year since going live.**

The exact steps and number of iterations required to achieve this for each enterprise depends on the size of the organization, the scale of the VoIP project planned, the timescale required and the degree of risk each enterprise is willing to accept in moving to obtain the benefits of IP Telephony. The result is the implementation of a new generation of voice product called an ECP – Enterprise Communications Platform.

Conclusion

VoIP is only part of the story of a voice solution implemented in an IP environment. It is really only the transport portion and misses the application layer considerations of an IP Telephony solution. The user of a voice solution cares little for the devices and network beyond the handset features they see and use. Similarly they judge the quality of the voice network relative to their current experience both at home and work. If the quality of the voice experience does not meet these expectations, then any voice solution including VoIP is doomed to failure.

The successful creation of a converged voice and data network in any enterprise requires a careful balance of the considerations mentioned in the body of this paper. Chief amongst those is the business case justifying the direct financial cost balanced against the flexibility an IP Telephony solution will bring. If these benefits and some of the cultural changes they introduce to an enterprise do not match the cultural direction of the organization then an IP Telephony solution may not be appropriate. Some cost savings of voice compression using VoIP on expensive WAN links may be the final solution implemented. If a full IP Telephony solution is adopted, then careful analysis and project implementation is required. Careful consideration of the complexity of this transition to IP Telephony is difficult to achieve without appropriate consultation with suppliers. This will hopefully lead to the formation a partnership with your key specialist vendor(s). Quality, security, power and availability, network management and staff training must all be implemented to ensure the full benefits of an IP Telephony solution are realized on your Enterprise Communications Platform (ECP).