Building an enterprise IP telephony network: a technical white paper on the practical deployment of converged solutions

Published: 28 March 2007

Table of contents

List of figures

Introduction

The IP telephony market has reached a level of maturity, in both proven technology and support by carrier networks, which enables organisations and businesses to implement converged solutions with total confidence in the levels of expertise and choice now available to them. Carriers worldwide are migrating their infrastructures to converged networks, with beneficial long-term implications for telecommunications.

IP connection directly into office premises is likely, for example, to become standard in the very near future. In conjunction with the connectivity thereby afforded, converged solutions offer enormous flexibility and scope for integration within existing data systems, databases and other internal in-house standard systems.

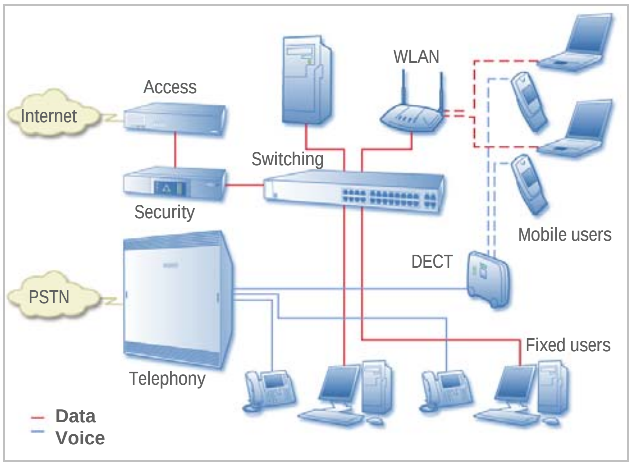
The primary intent of this paper is to acquaint the reader with the process of building and integrating an IP telephony system whilst providing a comprehensive introduction to the proliferation of components available. It also highlights other key considerations, such as how best to manage the project and the migration phase, as well as cultural and organisational areas inextricably linked to the introduction of what will be, to many target users, new and unfamiliar technology.

Building the network - IP telephony core components

This section focuses primarily on the switches and router options that should be considered in the configuration of an IP telephony infrastructure. IP telephony handsets are also covered in some detail.

Figures 1-1 and 1-2 illustrate the basic topological differences between a traditional network and an IP telephony, or converged, solution; showing clearly that a converged solution is a simplification of the traditional network.

Figure 1-1 Standard network environment

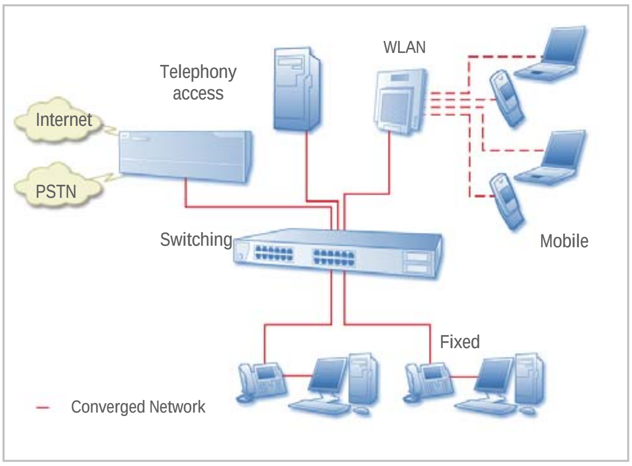


The standard network environment **(Figure 1-1)** comprises a number of key components (desk-tops, notebooks, LAN switches, routers, possibly a VPN, and a firewall) connecting externally. The large, traditional TDM phone switch sits on its own.

Up to now, the two parallel networks have been managed differently, by a telecoms team and an IT team; a logical approach based on the different functions they serve and the variations in technology. This approach, however, is likely to change as systems cross over.

Figure 1-2 demonstrates the simplicity of a converged solution in action. Whilst the number of devices remains constant, two significant benefits result from this system:

Figure 1-2 Converged network environment



Firstly, the number of components required to build the network has been rationalised. This brings with it the subsidiary benefit of reducing the number of manufacturers involved in the system; less maintenance contracts, fewer product-set familiarisation requirements and so on.

Secondly, and more importantly, the duplicate networks for voice and data have been replaced by a simplified, converged, single network.

Whilst the network is easier to support and manage, through the simplification of components and configuration, users will also be able to take advantage of many advanced features, giving them a more sophisticated and multi-functional approach to communications within their business.

Switches and routers

For environments where an IP infrastructure is not currently in place, but where the intention is to invest in new switch equipment in the near future, strong consideration should be given to purchasing devices that will ensure both class of service (CoS) and, ideally, power over Ethernet (PoE). This will establish the foundation for an easy transition to IP telephony when the time comes.

These features are described in more detail below.

Ethernet switches

These can be regarded as the backbone of the system; connecting the network devices to each other. The switches are the key component to integrating both the traditional data systems and the new telephony systems. This compares to a traditional telephone system, in which telephone handsets and computer devices may run over the same structured cabling, but are segregated into two different areas (the telephone switch equipment and the LAN switch equipment) within the patching frames of the data centre.

In a converged environment, everything runs through the Ethernet switches.

Ethernet switch components have a number of features, some of these are essential, some are desirable:

Class of service (CoS)

One of the key requirements, when running voice across a network with an IP telephony solution, is that class of service must be supported at a layer 2 or layer 3 level to ensure that voice quality is maintained, throughout the network, for the duration of any call and for all call traffic.

CoS ensures that the data packets running across the network maintain the same high quality voice calls that users have experienced with a standard phone system.

This includes avoiding issues such as:

- Delays in the traffic being routed to another site
- Jitter
- ‘Voice bounce’

Most of the major manufacturers support a standard known as Auto-CoS. This employs an algorithm which compares fields of packets, so that application data can be identified and allocated a class of service. It can also be assigned to queues of differing priority. Each type of data (e.g., voice, video, thin client, email, Internet traffic) can be assigned generally to one of six classes of service: voice, assured data 1-4 (e.g. video, ERP, CRM, thin client) and general data (e.g. email and Internet). Thus the switches will automatically detect that the voice call has priority across the network above all other traffic types.

Power over Ethernet (PoE)

With switches it is possible to incorporate the ability to deliver power over Ethernet to handsets deployed across the network. PoE technology transmits electrical power, along with data, to remote devices over standard twisted-pair cables in an Ethernet network. This removes the requirement to add extra electrical services to new devices, as well as providing enhanced business continuity capabilities.

Most IP telephony handsets have a built-in switch enabling a single, structured cable (CAT 5, CAT 5e OR CAT6) to be run into the handset and from the handset onto the PC or notebook interconnecting with the device. The voice and standard data is separated using VLANs by the switch.

In the absence of a PoE function, a separate power supply needs to be added locally to each handset which means more structured cabling and, potentially, more power points.

Inline power injectors (IPI)

Inline power injectors come in two forms:

***Single unit*.** Often used when installing small numbers of WiFi access points (APs). The single unit comprises a standard switch plus a cable running into the IPI, which resides in the data centre. A physical plug connects to it to inject the power into the structured cabling system, which is then routed on to the device at the other end , providing the power through the cable. This is very useful when installing IP CCTV and WiFi APs into difficult locations, such as roof spaces, where power circuits are not generally located.

***Rack-mounted unit*s.** These reside in the data cabinet so that the switches tier into the larger, 24-port IPI, for example, and again on to the network devices at the end of the cable.

The downside with both solutions is the potential jumble of cables in the data centre.
It is also important to appreciate the cost differential between power injectors and a PoE switch; the POE switch can be more than ten times the cost of an IPI.

Virtual LAN (VLAN)

An extremely useful feature to have on the Ethernet switch is the ability to support a virtual LAN. This enables flexibility and security in the configuration of the network.

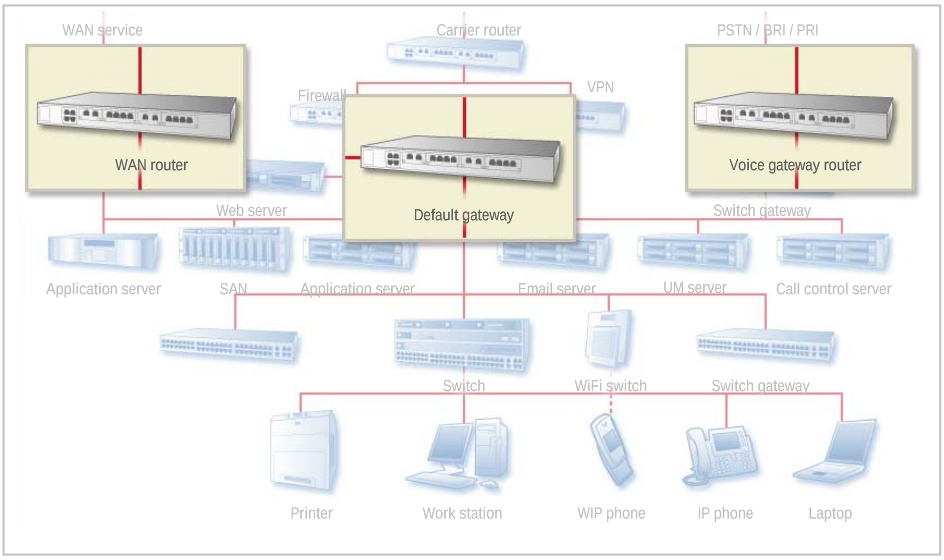
Some companies prefer to run their IP telephony solution over a different subnet to other data systems. This offers benefits in both management and ease of deployment but, given that each handset requires its own IP address, it can also place a significant load on the existing IP schema. Rather than re-designing the entire network from scratch, to accommodate IP telephony, a separate subnet can be created purely for the phone devices by using a virtual LAN.

An IP telephony system can be run over a standard switch, but quality of the calls will be affected and IP handsets will be more difficult to manage. It may also be undesirable to have large amounts of cable added to desk areas which may already be cluttered with cables.

Routers

Routers are the primary devices used to direct traffic outside the local area network. **Figure 1-3** shows three routers, but the number used is largely dependant on the complexity of the network and/or size of the environment.

Figure 1-3 Router deployment



Multiple routers can also be deployed so that in the event of a system failure, another device can take over the function of the primary unit using HSRP protocols. The primary router in figure above serves as the default gateway, or the master device, controlling routing of the IP traffic in a Layer 2 switching environment locally but also, via three potentially different routes within an IP telephony system (PRI for external calls, Internet for SIP calls and the WAN for interoffice calls). The LAN can also be routed at the switch level if Layer 3 switches are implemented. Here the IP routing occurs at the switch and port level and would not require the default gateway.

Voice gateway router

This is the device that will typically connect the PSTN, ISDN 2 (Basic Rate), or ISDN 30 (Primary Rate) voice lines to the LAN. It is possible, however, that such connections, for voice traffic to the public network, will change as BT's 21st century network (21 CN) evolves and as other key carriers provide new services to businesses using IP and SIP.

Figure 1-3 also shows a WAN router breaking out into a multi-protocol label switching network (MPLS) or a public Internet VPN. MPLS WANs that support CoS are preferable for running voice over WANs to ensure voice quality and availability.

Public Internet VPN traffic does not necessarily support the class of service standards required over the routers and therefore affect voice call quality and availability. As discussed with the switches, CoS is required to maintain voice quality across the network until it is handed off to any gateway beyond the control of the internal telephony system, such as the public dial network. Within the routing environment, consideration may also be given to integrated service routers or modular routers, available from most major manufacturers to connect multiple networks to a single device.

Modular routers

Modular routers provide flexibility. As the devices or requirements change, extra modules or additional connectivity will not necessitate having to replace the entire unit or re-design the network. For example, consider a small office with ISDN 2 break-out for its voice network, which then requires ISDN Primary Rate module, as the office or the business grows. By simply replacing the ISDN 2 module with a Primary Rate module, or placing a Primary Rate module in parallel, the business can effect a smooth migration without massive disruption to the configuration of the network, or considerable down-time.

Another useful feature of modular routing equipment, particularly for smaller branch offices or remote offices, is that they can actually run the telephony system themselves. Rather than having a separate PBX or a call control server, in an IP telephony environment the phone system resides within the router. The router has a static image of the master phone system sent to it. In Cisco's technology this is known as SRST (survival remote site telephony).   If the system loses connectivity to the master site through, for example, the wide area network, or if connectivity to the master network is lost, then the site can continue to run independently using the routers configured with SRST.

It also enables resilience to be built into the network. In the case, for example, of a head office default connection being across a wide area network from an SRST router, where there might also be a local break-out to an ISDN dial network for local calls, the router would automatically break-out through the public network if the WAN was not available.

This resilience means that remote offices do not have to be dependent on the central site and that failures at the central site no longer automatically disable the remote sites.

Call control servers

Figure 1-4 Call control server deployment

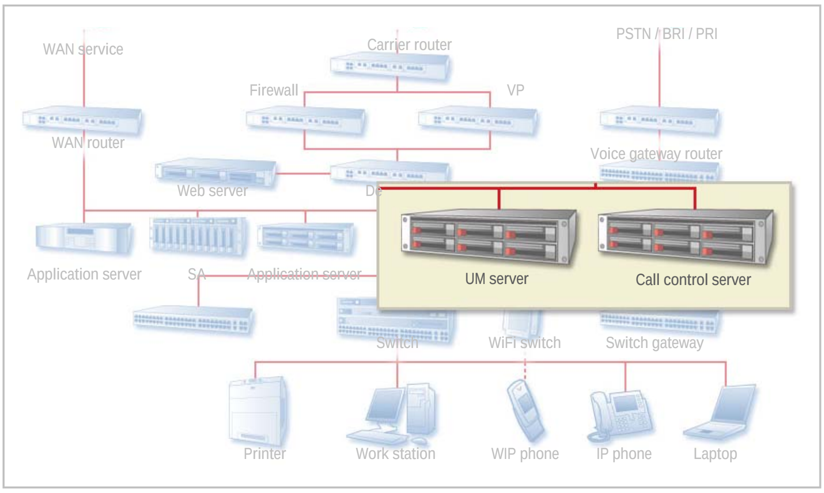


Figure 1-4 shows two units, but the number deployed will, as in the case of routers, be dictated by size of environment, complexity and also level of reliance required. The first unit is the call control server itself, the nearest equivalent to a traditional PBX.

This is, effectively, a server that sits on the network and holds the dial plans and the extensions, and could also be the interface with the directory service using an LDAP compliant structure such as Microsoft's Active Directory. LDAP enables users to have the same log-on and authentication for their handsets as they have on their standard PC.

It also enables softphone users (see ‘IP telephony handsets’) to have mobile connectivity through the network into the central IT systems, gaining access with their same user name and password.

All major manufacturers provide rack-mounted servers for installation within data centres. Some manufacturers actually use branded products from, for example, the HP and IBM server ranges. Most also use a bespoke Linux operating system within the server.

The call control server is the main device that telephony administrators or IT and telecoms managers will be using on a daily basis to configure and add servers, or effect moves and changes. Because the system is often integrated into the LDAP-compliant**,** or Microsoft Active Directory system, this can also be managed centrally using standard domain control tools.

The other server in **Figure 1-4** is a unified messaging server (UM). This facilitates the introduction of extra features such as voice-mail into e-mail, or e-mail into voice-mail or perhaps integration with other systems within the network, such as screen pop-ups to desktops and CTI interfaces. This enables integration of existing data systems, databases and other internal in-house standard systems with the telephony device and is one of the most significant benefits of a converged system over a traditional PBX.

IP telephony handsets

Handsets are probably the largest component, in terms of sheer numbers, involved in the deployment of an IP telephony system and the most significant element of cost. An extremely wide range of options is available.

Figure 1-5 Handset options

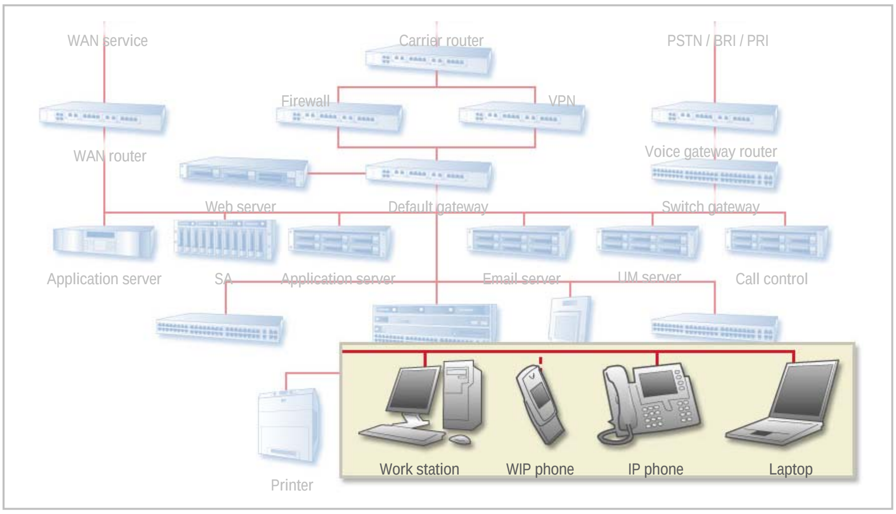


Figure 1-5 shows three handset options:

- Standard handsets, similar in appearance to those used in a traditional telephone system
- Softphones; a piece of software that runs on any standard system and interconnects with the telephone systems, across the network through the desktop’s operating system
- WiFi wireless IP handsets**.** An interesting area, particularly as the mobile industry moves on with mobile networks and plans to integrate with WiFi technology as well as the standard GPRS 3G mobile networks in use today

The IP telephony handset, unlike the traditional telephony handset, is effectively a mini PC. The market offers an extremely wide range of handsets and the 'rule of thumb' is that the more expensive the handset, the larger the screen will become, it may become colour, or indeed, at the top end of the range, will become a colour touch-screen, with a number of short dial buttons improvements, and possibly video. The basic functionality, however, is the same throughout the range.

Softphones are slightly different in that they have the advantage of offering the top-end phone functionality as standard and are therefore possibly the best phone available for the lowest outlay.

If the softphone has a drawback, it is that it may require a head set plugged into the PC, e.g. a blue tooth unit.

A good example of the potential of IP telephony handsets is 'The connected campus' (Figure 1-6).

Figure 1-6 The connected campus

Having recently invested in upgrading its campus sites to attract both the best students and the best lecturers, an American university decided to deploy a structured cabling system within its halls of residence and throughout the campus.

The initial objective was to allow students to connect their own PCs and devices to the network. The university also decided to provide students with their own telephones for calls within the campus, as well as to talk to the administration or facilities managers should they have any problems. However, with the installation of structured cabling, and the deployment of an IP telephony system, the university decided to explore how IP telephony could be used differently from a standard TDM. Their innovative ideas are a perfect illustration of the possibilities that exist with IP telephony.

The first fairly basic idea was to apply the same student ID and authorisation system used for accessing the IT network and buildings around the campus to the phone system. That way the system administrator could always know who was logged on, where calls needed to be directed and, from a security perspective, ensure that the phone user had authority to access the system. Additionally, for cross charging and billing on external calls, costs could be correctly assigned to an individual's account.

Once the university was able to identify who was actually logged on to the phone, it started to deliver personalised information to the students. For example the student's timetable would be displayed and updated daily on the phone and if he or she needed to know where a lecture room was, this information could also be accessed through the phone.

The university then approached local businesses to see if they might be interested in supporting the development and the deployment of the solution throughout the university.

A well-known pizza company offered to sponsor the roll-out as long as it could have the first short dial button on the phone. The user could then very easily dial through to order a pizza and because they were logged on to the phone, the pizza company could automatically identify the student along with their exact location in the halls of residence, ensuring the quickest possible pizza delivery and payment.

Over time, the pizza company was able to build a database of student preferences and buying patterns and began to use 'ticker-tape' phone feeds to advertise special offers. Ultimately the system was developed so that students could actually order the pizza through the phone handset without speaking to a member of staff at all.

Other developments included sponsorship by a local cinema complex, which advertised its films and enabled reservations to be made through the phone, and billed to a central user account. The university also benefited by centrally routing all public calls outside the network using least cost routing which meant that not only did campus phone users benefit from cheaper calls but the university also took a slice of the revenue helping finance the system.

Wireless IP phones

The technology most likely to make major advances over the next few years will be in the field of wireless IP phones. The key performance difference currently between wireless IP phones and DECT phones is their physical roaming distance. Wireless IP telephony allows for cheaper installation and offers increased functionality over DECT, however, the different requirements for signal strength to support wireless IP telephony can result in connectivity service fluctuations as the user roams through the building; it is typically around two-thirds of the distance available from a standard DECT connection. If an existing wireless network exists the access points will need to be re-positioned and re-configured to support voice traffic.

Wireless IP phones arouse great interest for two reasons:

Firstly, with the continuing deployment of WiFi access points throughout most urban areas, as well as the future potential of WiMAX (should the standards ever become fully ratified and deployed in the UK) the mobile industry is starting to introduce mobile phones with WiFi, 3G and GPRS connectivity – so that a call can be placed or received over any type of wireless network.

Secondly, WiFi compliance means that when the caller is in an area that picks up a suitable wireless access point, the network will automatically hand the call across to the WiFi network, radio the call across the fixed connection, albeit with its higher data speeds, and hand back to the central call control server to negotiate the calls. Effectively it negotiates a call in the background across the GPRS or 3G network and hands the call back to the mobile provider.

Video telephony

Video telephony is available across a number of IP telephony solutions. This should not be confused with video conferencing, but is rather the ability for a user to have a personal video telephone call with another user. Cisco views this functionality as the standard, and envisages a not-too distant future in which video telephony will be widely adopted. According to the Cisco view, a user with a camera-enabled softphone and a user with an IP telephony handset with camera, or video, feature, will automatically be able to share a video session.

Video is a useful feature for home and remote workers, although it must be borne in mind that it requires significantly higher bandwidth than standard voice calls.

Once again, class of service must be factored in for video telephony. A specific class of service parameter for video is required to ensure that frames-per-second are maintained and also that suitable bandwidth is allocated for the call.

Considerations for selecting handsets

As established in the preceding section, there is a wide choice of handset options available. It is therefore important to keep a few things in mind when selecting handsets:

- Do not over specify requirements for the default unit since costs could mount up considerably across the whole implementation.
- Decide how the users work and how they will best interact with the solution. Softphones may appear to hold some allure but many users prefer a physical handset; something they can pick up and feel, with the reassurance of buttons they can press. To address this need, whether it is perceived or actual, softphones can be provided with handsets that plug into the desk top. This then raises the question would it be better to provide a standard handset in the first place?
- Wireless IP handsets provide a perfect communications solution for large open-plan areas, such as warehouses or office complexes. For the solution to be effective, however, access points will be required to cover the radial area as well as having the

ability to hand calls off from one cell to another. They should not, therefore, be configured as stand alone access points but instead as unified roaming and load balanced zones able to support the call throughout the complex.

The optimum solution to handset requirements within any organisation is to identify all of the different user requirements and establish a procurement plan based on the best combination of these needs.

Connecting TDM devices to an IP telephony network

Some organisations will need to connect standard, traditional analogue TDM-type devices into the IP telephony network. These may include fax machines, modems, or traditional conference phones such as the 'star' conference phone.

IP-equivalent units can be expensive, but a solution to the problem, for organisations that may not yet be ready for the requisite investment, exists in the form of an analogue telephony adaptor (ATA).

An ATA will convert the IP telephony signal into a standard analogue signal and allow the traditional devices to connect to the IP telephony solution.

One drawback of ATA devices is that many will only support a maximum baud rate of 96 bps. The implication is, therefore, that some fax machines either have to be adjusted or actually may not be compatible with an ATA. A careful calculation has to be made of the cost of analogue telephony adaptors, as well as the manpower required to work traditional devices into the network; compare this to installation of a separate PSTN line that allows users to connect to the outside world independently of the IP telephony solution.

Legal requirements

Health and safety considerations must also be taken into account. In the UK, for example, a separate wall mounted PSTN device may be required, in locations such as lifts or other worker environments. This ensures that in the event of a power failure or emergency situation workers will still be able to make calls because the PSTN line in the UK provide power down to the handset, where as an IP telephony handset does not.

Integration – IP telephony interfaces and interdependencies

Having reviewed the components required to build an IP telephony network, this section discusses the integration of these devices and how they need to be configured to interact in a converged environment**.**

Local area network (LAN)

In deploying an IP telephony system, each handset (assuming that handsets are chosen over softphones) requires an individual IP address. Depending on the existing network subnet**,** and the existing network configuration , a virtual LAN could be deployed to run a separate IP subnet specifically for the telephony system.

It can sometimes be necessary to re-design the entire IP subnetting structure locally and in all remote offices to support additional IP devices on the system.

Deployment of handsets to the desktop

If running a structured cabling system, consider the switch feature on the back of the phone which will enable connection of desktop devices into each handset.

If this route is taken, it may be desirable to install power over Ethernet or inline power switching, so that the handsets do not require separate power supplies creating the clutter of extra cables. To run inline power or PoE, structured cabling systems should be based on Cat 5, Cat 5e or Cat 6. (Cat 3 standard does not support the PoE wire-pair required to inject the power into the cabling).

For the deployment of wireless mobile handsets across a network, using WiFi IP phones, existing access points may need re-tuning and re-positioning, or additional access points may be required, in order to ensure sufficient coverage.

Some access points on the market have the ability to automatically re-tune one another, minimising signal overlap. These are beneficial in campus sites, with multiple access points, but must be configured as load balanced wireless zones, allowing the handsets to hand off from one to another as the user roams.

At this juncture it is appropriate to reiterate the importance of ensuring class of service across the network, from the handset through the cabling system into the switches, from the switches into the routers and then on to their final destination; and that switches should support AutoCos, so that when a voice call is made, that data is handled differently across the LAN (ensuring no jitter and no delay).

Call control servers

Also important with the LAN is the integration of call control servers, and other devices required to interface with the existing server infrastructure. These can be integrated with Microsoft Active Directory or an LDAP-compliant structure, allowing phone systems to be integrated with a standard authentication and directory management system.

When new users are created, it would not be necessary to enter and configure them separately for both the server environment and the traditional TDM application; all user information would be entered in a single place.

When the system is configured, consideration should be given to the introduction of features such as unified messaging, voicemail into email, and email into voicemail along with text-to-speech conversion.

Voice messages can be quite large data files. This obviously increases if video messages are also recorded. To avoid problems with email environments, such as Microsoft Exchange or IBM Domino, adequate disk capacity should be allowed for extra storage, in conjunction with adequate mail box usage policies to ensure that users receiving voice mails have enough storage to retain messages within their in-box. In addition, the data usage increase should be taken into account for existing data back up and archiving systems.

Deploying an IP telephony system is essentially the same as adding a new application to the network and does not necessarily mean replacing the existing network. Certain components are likely to require modification, upgrade or re-development to support the IP telephony system but it would be judicious not to overlook the capabilities of the existing system.

Deployment methodology

For manageability, wherever possible deployment of an IP telephony system should be phased. A suggested approach for each phase would be:

Phase 1: Installation and configuration of the core infrastructure, the switches and the routers; ensuring that the network can support the IP schema and CoS required.
Phase 2: Replacement of the traditional TDM phone system with the IP telephony solution, ensuring that it at least performs to the standard of the existing system, whilst providing added benefits; this phase could, for example, include unified messaging.
Phase 3: Introduction of advanced features once the system is established and stable; this phase could include screen popping, integration to data bases and applications, SIP and CTI.

Wide area network (WAN)

The WAN is where, potentially, control of the network can be lost. When a call is handed off to the public network, a PSTN or ISDN interface, management and control of the call across the network falls to the carrier.

Whilst this is mostly a reliable route, building some resilience into the configuration can provide the ability to route calls out of other offices, should one site have a problem.

Resilience can be built into the network components, the public network and the WAN.

With the WAN itself, class of service standard needs to be maintained across data links to remote offices. If an MPLS network is being used to connect offices, CoS data parameters should also be run across the network to support the voice data. This should not be an encapsulated or a pseudo CoS, but a true CoS service, throughout the network, ensuring that voice calls receive priority and are handled differently to data calls using the bandwidth available.

The exercise of reviewing the relationship between the infrastructure and the WAN has additional benefits in terms of the opportunity to review current external connectivity, both to the public network and on the WAN.

Included in such a review should be the type of circuits, coming into a building, provided by the carrier. It is sometimes the case that lines can be eliminated. Conversely, in the absence of such a review, rental could continue to be paid for redundant lines for many years to come. The IP telephony solution will allow consolidation of individual separate services into, perhaps, a single primary rate ISDN line for multiple functions.

With new WAN technology, many MPLS networks are considerably lower cost than a frame-relay network, or most traditional networks for inter-office connectivity. They frequently provide more bandwidth, with CoS, for an equivalent or lower cost.

Virtual private network (VPN)

CoS must be supported across the WAN (for voice quality, signalling and prioritisation reasons, as previously discussed) but across the public Internet VPN this is not possible. Whilst the calls will work, there is no guarantee of call quality. Where it is required that permanent sites have high call volumes, it is advisable to use a private network that does have the CoS.

The cost implications of providing a suitable WAN, once viewed as prohibitively expensive, are now considerably lower. The situation is further improved because many carriers now provide CoS across their networks either as standard, or as a very low-cost option.

These observations having been made, it should be stated that the Internet VPN route works in certain situations, albeit, in a slightly less controlled or guaranteed manner. This is generally used for home workers using softphones or IP phones

Directory services and applications

Deployment of an IP telephony system involves the introduction of additional devices into the network. Whilst it may appear obvious, this fact is often overlooked, despite its implications for small data centres or operations with confined physical storage space. Most TDM systems tend to be wall-mounted or stand-alone, whereas many of the new devices are rack-mounted.

Attention should therefore be given to the capacity and space within the cabinet to maintain and store this equipment as well as to its power requirements, integration with Active Directory or an LDAP compliant structure, and the ongoing support and management of the devices.

If unified messaging, voice mail into email and/or email into voicemail are then added, disk capacity, both for storage and back-up, as well as software restrictions, may also require re-assessment.

It could be the case that such considerations dictate a re-design, to avoid physical and/or data management restrictions hampering the installation and configuration of the system.

IP telephony - planning, support and management

This paper has so far covered the requisite components and their integration with a live environment. In reality, the project phase of deploying an IP telephony solution is a short period compared to planning period, as well as supporting and maintaining the system throughout its life.

Support and management

If the intention is that the IP telephony solution will be integrated with Microsoft Active Directory or an LDAP compliant structure this means that it may cross boundaries traditionally separated within the organisation; IT and telecoms. Responsibilities within the organisation may well need to be clarified and agreed.

A cultural change within the organisation, in the area of user support, is implicit in these areas of operational 'overlap'.

Also implicit will be a very real need to evolve and enhance the telephony devices to support the IT infrastructure.

With the integration of the two systems (telephony and IT), phone faults may ultimately be diagnosed and rectified through the data system, by IT rather than telecoms personnel. Fault diagnostics are thus simplified and will no longer require personal attention at the physical desk or handset level.

Also simplified will be the ability to undertake moves, adds and changes (MACs). The integration with Active Directory or LDAP services, means that these changes will be implemented by the server team, rather than a specialist in TDM management.
IP telephony could, therefore, lead to the absorption of the telecoms' role within the IT department, which will, in itself, require some retraining of the IT team to acquire the additional skill sets.

There are two key areas to focus on for support and management of an IP telephony solution:

- i) Cross-training of skill sets between the telecoms and IT teams to share the knowledge for support and management of the systems deployed and reduce reliance on one or two individuals. Due to integration between the devices, this approach will also broaden the understanding of where a fault may occur, or how a particular system may need to be configured.
- ii) Fault procedures. Because the troubleshooting and/or monitoring of the network could be automated, fault-finding is easier than for a traditional system. However, the procedure required to rectify a fault may not be as simple as in the past, where previously the solution may just have been removing the handset and plugging in a replacement. With the IP system the problem could be related to the LAN, the routers, the class of service running across the network or an IP subnet, rather than a basic malfunction in a physical device.

In terms of fault procedures there are two important factors to be aware of:

1. Firstly, with wide area network routers, in the case of a managed network, it can sometimes take two weeks for a change control request to be accepted. This could be a request for an adjustment on the network, such as expanding the class of service, re-routing, or the way in which the system is configured to support the IP telephony network. The lead-times involved when a third party has to effect these changes can be inconvenient. Where possible look for unmanaged MPLS networks where the user or a trusted 3rd party can make changes quickly and efficiently. This

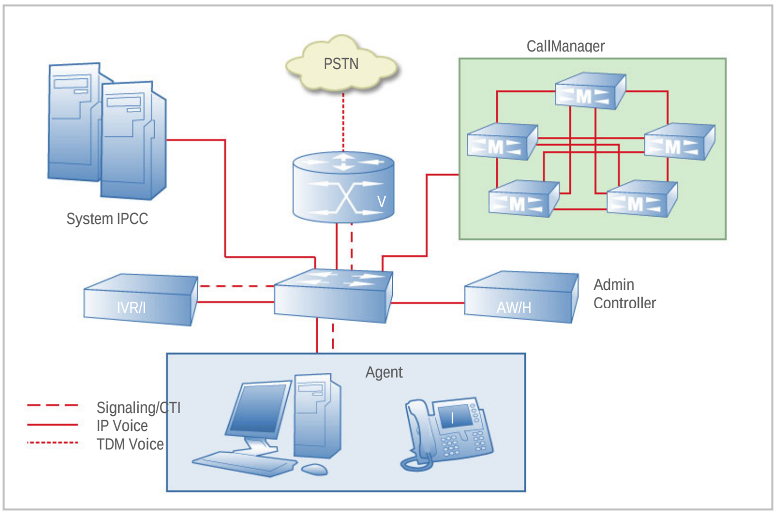
also provides access for monitoring the service and visibility to ensure the carrier is delivering the service that has been contracted.

2. Secondly, depending on the system being used, tracing the root cause of a problem can involve a relatively wide-spread investigation that can also be time-consuming. It could actually be a more generic infrastructure problem or, indeed, in the control of a third party; the bottom line is that any network device could malfunction to impact voice calls.

IP telephony planning

This section reviews how the IP telephony can be deployed in different environments and looks at some of the options for adopting the technology according to number of users, how they work, number of sites and site inter-connection. Figure 1.7 illustrates the main configuration for a standard single-office set-up; an extremely simple solution.

Figure 1-7 Single site deployment



Whilst most manufacturers of IP telephony equipment will make available a cut-down version of the main telephony system solution for single-site SME organisations, in more complex environments the network follows one of two primary routes.

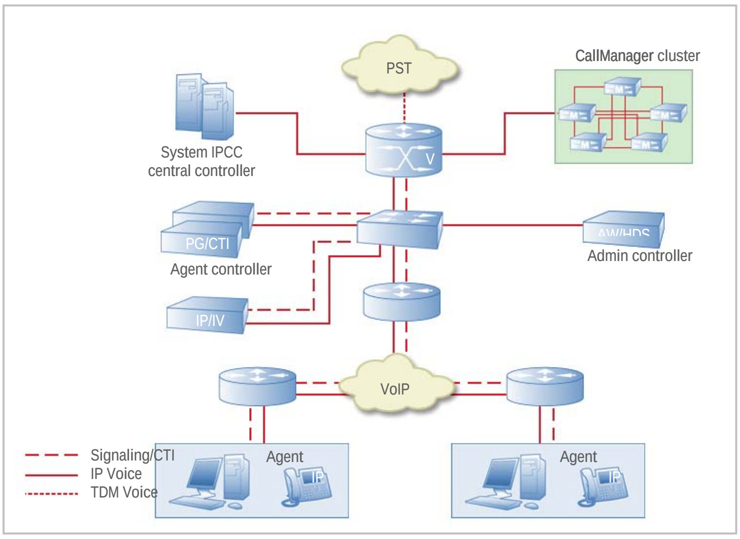
Route one: multi-site planning. A typical set-up is most likely a head office serving as the data centre and housing the bulk of the data equipment. Remote offices or users connect to a centralised network (Figure 1-8). This model requires a minimal amount of physical equipment at remote sites or locations.

In this situation, investment is primarily at one location, the call control server, perhaps, in a centralised data centre. This may entail clustered servers; routers and switches to provide system resilience .

Configuration at the remote sites will be basic; possibly a single router such as the Cisco ISR router using the SRST feature to enable the site to work independently in case of WAN or central systems failure, although the site will still be an extension of the central office.

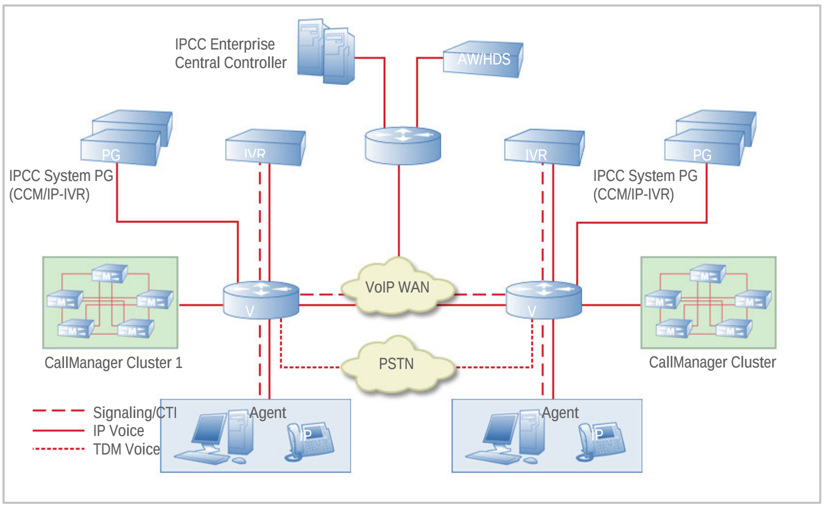
This means that any changes are centralised and administrated at the head office location and then replicated to the remote offices

Figure 1-8 Centralised multi-site deployment



Route two: **multiple central sites**. This route (Figure 1-9) is often selected either for reasons of disaster recovery, or for organisations with offices located through different time zones; the classic scenario being the 'follow-the-sun' network configuration, including, for example, offices in Central Europe, North America, and the Asia Pacific region.

Figure 1-9 Decentralised multi-site deployment



Networks such as these may require multiple call control servers that act as a geographically diverse cluster.

The primary difference over route one is that centralised administration and control will be regional. This could entail configuring the rule sets to transfer between call control servers whilst retaining some localised configuration specific to one area where, for example, another area has no need to share certain information.

In this scenario, much more resilience can be built into the configuration of both the IP telephony network and the external connectivity.

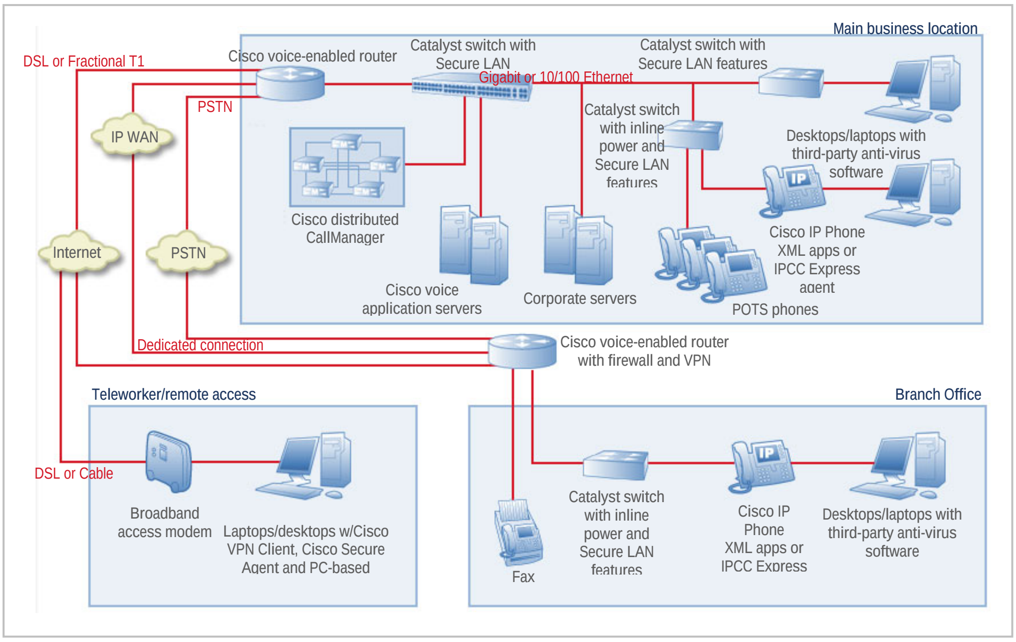
If there were an organisation comprising a London and Paris office, for example; when a Bank Holiday falls in Paris, users can still dial the international call code or the local call for the Paris office and the call control server can be configured to route the incoming PSTN and ISDN voice traffic automatically across the wide area network to the UK system. The system can also be configured to identify, for the benefit of the agents answering the phone, that the call has been re-directed from Paris. The call can then be dealt with accordingly, perhaps by bi-lingual staff.

Were there to be a fault at either location, the call control routers and servers can be configured to route the call to the other location.

Depending on cross-charging and internal policies, there are also enhanced functionalities that enable the system to take full advantage of the wide area network. When international dialling codes are inserted by users the codes can be recognised by the call control servers and the traffic routed out across the wide area network to break out locally, incurring local rather than international call costs.

Telephone systems will often be more complex than these scenarios, particularly when mobile and remote workers are taken into account (Figure 2-0). A branch office may have a small basic router with either SRST functionality or, alternatively, a stand-alone router with IP devices directly connected.

Figure 2-0 Centralised site with home and mobile workers



Permanent home workers who might be equipped with a basic wireless router connecting them via a VPN through the public Internet, or, increasingly, through a dedicated private network.

With MPLS-type wide-area connectivity available now over DSL, and class of service becoming increasingly cost effective, this latter route is a much more viable option for 10 home workers and above.

Project scope

As discussed earlier in this document, the best approach for deploying an IP telephony system is to phase its introduction not only to rectify 'teething' problems as they occur, but also to familiarise users without overwhelming them. IP telephony affects all staff, the project therefore needs to implemented with full support.

It is important to note however that this approach will impact the project timetable, procurement and funding plans.

The return on investment from an IP telephony solution can often be made up of a wide-ranging number of components. It is often difficult to find one single compelling reason to justify the cost. Reasons become self-evident, however, by undertaking a process of gradual investment in the infrastructure, enabling the business to put in place the hardware and devices that will eventually be required to support the system.

Project ownership

Deciding where project ownership falls, both during deployment and for the life of an IP telephony system is a delicate issue that will involve personnel across functions, disciplines and levels, including the users themselves, whose adoption is critical if the business or organisation is going to benefit from the system. Additionally, the deployment project team needs to involve the telecoms department, the IT department, and possibly even the security department where such a function exists.

Manufacturer/product selection

Most major manufacturers no longer produce or support traditional TDM systems, choosing instead to satisfy the enormous demand for IP-based solutions. This implies limitless choice, but can create difficult decisions in manufacturer and equipment selection.
Purchasing decisions can be based on numerous independent research studies available, such as those produced by the Gartner Group, but should obviously be tempered by the functionality required of the system, which, in turn, relates to the business processes currently or planned to be in place.

Migration options

Breaking the project down into phases significantly eases the transition from existing to new systems. Such an approach will not preclude the need to pay considerable attention to detailed aspects of the migration which can sometimes be overlooked in pursuance of the overall strategic goal. IP schema, CoS, VLANs, authentication and migration of the handsets are all examples of such essential detail.

During the implementation phase a decision may be made to run two handsets concurrently on each desk. This can be inconvenient and is not always entirely practical but can lead to a smoother implementation. A more convenient, though riskier, approach is the 'big bang' whereby staff leave the office one day with the old system still operable and return the next to find the new system installed. This approach is not generally recommended for larger sites.

Running parallel systems, during a transition phase, is often preferable for multiple office networks where certain sites move to the new systems whilst others retain the old system. Integration between the two then becomes the biggest issue and it is absolutely essential not to 'over-engineer' the new system to be completely compatible with the old. Such functionality can require considerable investment which might best be avoided by a full migration.

Some basic functionality, such as call transferring and inline extensions, is, however, relatively straightforward to implement.

Internal/external resource

This may vary depending on each element of the project; system design, manufacturer research and selection, deployment, and ongoing support of the system and third-line support to the internal team. Decisions on which resource to utilise will depend on skill sets and experience, and deployment time-frame.

External resource is often called upon to provide consultancy on the project or simply for additional hand-on experience to assist the internal resource.

Training

Training is sometimes viewed as the process of educating users on how to use the new system. With IP telephony, however, training is much broader and should involve cross-training the telecoms team on IT and the IT team on telecoms enabling both to administer, manage and support the system on an ongoing basis. They will need to accommodate a wide range of users coming into the system for a variety of different needs and functions.

Conclusion

Building an IP telephony system should not be regarded as a one-off project. Reviewing the current telephony structure will often reveal areas where savings can be made and will certainly serve to create a more complete understanding of the needs of different user groups even within a single organisation.

IP telephony offers new and exciting opportunities to many organisations, but if the technology and practices are not understood and adopted by users it does little to advance communications.

Patience, training and a full grasp of the cultural shift which may be necessary will, however, enable the organisation to explore possibilities, functionalities and, ultimately, efficiencies simply not achievable with traditional telephony.

About SAS

The SAS Group is a major UK systems integrator specialising in IP infrastructures and communications. The company provides consultancy and implementation services to design, build and manage converged IP networks for companies of all sizes, enabling them to benefit from the cost and operational efficiencies associated with unified communications. Since its inception in 1989, the company has successfully delivered more than 2000 highly commended solutions, to a broad spectrum of clients including many renowned brands, such as Coca Cola Enterprises, The Body Shop International and Millennium and Copthorne Hotels.

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