VOICE AND DATA INTEGRATION

WHITE PAPER

INTRODUCTION TO VOICE OVER LOCAL AREA NETWORKS

Redefining distribution.

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VOICE AND DATA INTEGRATION

Introduction

Voice and data integration is currently a hot topic within the business community, with its promise of major cost savings and consolidation of data and voice infrastructures. But there remains some scepticism and the old adage ‘if it ain’t broke, don’t fix it’ certainly has a few adherents. However, while PBX based telephone systems have certainly proved their worth in the past, there is growing evidence that the demands of modern business will increasingly necessitate the implementation of integrated voice and data systems. Up to now, small sites were the ones that showed themselves willing to adopt the new standard, but now it is the large enterprise sites with existing PBX based systems that are beginning to be seduced by the significant business and cost benefits that a migration to VoIP technology can bring.

The aim of this paper is to explain how and why this migration can be carried out, and to highlight to the user the myriad of benefits that such a migration entails.

Advantages of Implementing Voice and Data Integration

Before addressing the technical details of packet voice, it is important to put forward the business case for its deployment. The main benefits are best summarised thus:

• Consolidation of voice and data infrastructures, with the inherent cost reductions offered by the consolidation of support teams.

• Freedom from proprietary PBX systems, with their costly upgrade paths and incompatibility with other vendor products. IP telephony offers a vendor independent, universally adopted standards-based solution.

• Cheap long distance calls. All calls previously made between a company’s international offices at expensive international call rates can now be run over a company’s existing data WAN links, resulting in considerable savings.

• Mobility for IP phone users. Since an IP telephone is referenced via its Ethernet MAC address it can plug in anywhere on the network. Unlike traditional phones which have to be connected to specific ports on PBX’s, IP telephones allow for ease of staff and office restructuring as well as facilitating new work practices such as ‘virtual desking’ where staff arrive at work and use any available desk.

However, the greatest benefits will undoubtedly come in the shape of ever more sophisticated multimedia applications. The increasing power of even the most basic desktop PCs to run multimedia applications such as video conferencing and integrated voice/email will drive the continued development of Computer Telephony Integration (CTI) applications such as those deployed by web-enabled call centres.
TECHNOLOGY OVERVIEW

In simple terms, any system which transports voice across a data network employs packet voice technologies. Analogue voice signals are digitised and the resultant digital stream is converted into standard packets. Voice packets appear to a network as 'data' and as such can be treated like any normal data packet i.e. switched across companies’ LANs, routed across WAN links or sent out over the Internet.

From a historical perspective, telephony systems started out using analogue signals to transmit voice from end-to-end. Today, analogue signals are only used from the customer site to the telco’s local exchange, where the signal is then digitised and transmitted throughout the PSTN as a digital signal. Since the human voice frequency range lies below 4kHz and the sampling rate defined by classical theory is twice this frequency, to get an accurate representation one needs to sample an analogue voice signal at 8000 times per second. The samples are then digitised and at 8 bits will provide enough resolution.

64,000 bps is the standard transmission rate for digital voice. Therefore, as a voice signal only requires 64kbps, it is relatively easy to employ Time Division Multiplexing techniques to combine multiple voice channels for transmission over a single high speed digital link – or ‘trunk’ in telco parlance.

This is the standard interface used in digital PBXs and commonly referred to in Europe as an ‘E1’ (2.048 Mbps or 30 voice channels) and in the USA as a ‘T1’ (1.544 Mbps or 24 voice channels). Most offices with digital PBXs employ an E1/T1 trunk to their local exchange.

The digitising function described above is referred to as codec (COding/DECoding), and the different codec standards are given G.7xx numbers by the International Telecommunications Union (ITU). G.711 is the standard 64kbps telephony codec but there are of course various compression schemes which will reduce the number of bps required, thus enabling the use of lower bandwidth links to carry voice traffic. These various compression schemes are also described by G.7xx standards and are summarised in the table below:

<table>
<thead>
<tr>
<th>Code Type</th>
<th>Bit Rate</th>
<th>Processor Usage</th>
<th>Voice Quality</th>
<th>Delay</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.711</td>
<td>64</td>
<td>—</td>
<td>Very good</td>
<td>Negligible</td>
</tr>
<tr>
<td>G.726</td>
<td>40/32/24/16</td>
<td>8MIPS</td>
<td>Good (40) to poor (16)</td>
<td>Very low</td>
</tr>
<tr>
<td>G.729</td>
<td>8</td>
<td>30MIPS</td>
<td>Good</td>
<td>Low</td>
</tr>
<tr>
<td>G.729A</td>
<td>8</td>
<td>20MIPS</td>
<td>Fair</td>
<td>Low</td>
</tr>
<tr>
<td>G.723</td>
<td>6.4/5.3</td>
<td>20MIPS</td>
<td>Good (6.4) to fair (5.3)</td>
<td>High</td>
</tr>
<tr>
<td>G.723</td>
<td>1 6.4/5.3</td>
<td>20MIPS</td>
<td>Good to fair</td>
<td>High</td>
</tr>
<tr>
<td>G.728</td>
<td>16</td>
<td>40MIPS</td>
<td>Good</td>
<td>Low</td>
</tr>
</tbody>
</table>

Without going into too great a detail, what can be deduced from this table is that although you get reduced bit rates when using compression schemes, they can result in inferior voice quality and increased processor usage. In general however, this trade-off between voice quality and bandwidth savings is tolerated as compression schemes offer considerable bandwidth optimisation and cost savings.

Clearly, digitised voice signals can be treated in the same way as digital data and converted to a packet for transmission over a packet switching network. But, before moving onto discussing the major packet voice technology – Voice over IP (VoIP), it is worth looking briefly at the two other prominent technologies:

- Voice over Frame Relay (VoFR)
- Voice over ATM (VoA)
VOICE OVER FRAME RELAY

Frame Relay (FR) is a packet switched WAN protocol widely employed today to interconnect company LANs. FR is based on the older X25 WAN technology but due to improvements in digital line quality has removed the requirement for error protection/correction and therefore offers much improved speed and efficiency over X25. Companies can purchase FR services from providers for point-to-point or point-to-multipoint connections and it offers viable alternatives to expensive leased line services. VoFR’s real function is the transmission of voice over WAN links and hence does not scale to the desktop. Nevertheless, many enterprises employ VoFR for the savings made on long distance branch-to-branch calls. Historically, VoFR solutions have been proprietary by vendor, but the establishment of the FRF.11 standard for call setup, coding types and packet formats will allow future interoperability between vendors’ products.

VOICE OVER ATM

ATM is a highspeed backbone technology which offers inherent superior Quality of Service (QOS) features and is ideally suited for transmission of voice and video. As with VoFR, VoA is primarily for transmission over WAN links and does not scale to the desktop.

VOICE OVER IP

Internet Protocol (IP) is the most widely used protocol today and is ubiquitous throughout LANs, campus networks, enterprise intranets and the Internet. Its popularity makes IP the unifying protocol for telephony solutions. Companies with existing LAN/WAN infrastructures running IP will find it easy to implement VoIP. Suitable solutions may scale from a purely internal IP telephony system right through to enterprise-wide systems employing WAN links.

As IP is a connectionless protocol it normally works in conjunction with the connection oriented Transmission Control Protocol (TCP) to ensure a guaranteed delivery service. However, although this works smoothly with data – any packet not delivered is simply re-transmitted – it will not work with real-time applications such as voice, because any word received out of sequence within the structure of a sentence will result in a garbled message.

Consequently, a new standard was required to cope with the real-time video and voice applications which have become increasingly popular. The H.323 standard is one such standard and provides a foundation for audio, video and data communications across IP based networks. By conforming to H.323 standards, multimedia products and applications from different vendors can interoperate across IP based networks, including the Internet. A comparison between the ITU H.323 Standard and the ISO Protocol Layers is shown below:

<table>
<thead>
<tr>
<th>ISO Protocol Layer</th>
<th>ITU H.323 Standard</th>
</tr>
</thead>
<tbody>
<tr>
<td>Presentation/Session</td>
<td>H.323, SIP, H.245, H.225, RTCP</td>
</tr>
<tr>
<td>Transport</td>
<td>RTP, UDP</td>
</tr>
<tr>
<td>Network</td>
<td>IP, RSVP, WFQ</td>
</tr>
<tr>
<td>Data Link</td>
<td>PPP, Frame, ATM etc</td>
</tr>
</tbody>
</table>

VoIP uses the ITU H.323 Standard as opposed to the traditional ISO Protocol Layer (OSI 7 Layer model), and although VoIP uses TCP to carry the signalling channels, the real-time audio streams deploy H.323’s Real Time Protocol (RTP). RTP uses the connectionless UDP protocol as a transport mode since it has lower delay than TCP and, as previously explained, retransmissions are pointless. VoIP is the most popular implementation of packet voice and it is the increasing prevalence of H.323 based desktop applications that will drive it to even greater acceptance. It also constitutes the primary focus of this paper.
Quality of Service

A typical definition of Quality of Service (QoS) is: ‘a mechanism for defining absolute and relative network performance requirements for the various streams of traffic on a network’.

It is of paramount importance to VoIP that QoS can be guaranteed from end-to-end. The two main network problems that affect voice quality are delay and delay variation, or ‘jitter’. Delay can cause the receiver to start to talk before the sender is finished and can also exacerbate the problem of echo, while ‘jitter’ causes gaps in the speech pattern. Delay is inherent in data networks where it has no real impact, but with voice it is necessary that QoS features are implemented right across the IP network – in the LAN and across the WAN for example. After all, a chain is only as strong as its weakest link and there would be no use in implementing advanced QoS features on WAN routers if the LAN switches offer no QoS support.

QoS on the LAN

LAN switches are inherently fast devices but congestion can still occur, for instance when two traffic streams compete for a single output port or when the speed of the incoming traffic exceeds the transmission rate of the outgoing port. In these situations the switch must be able to buffer some traffic, while transmitting the rest. Obviously if there is a mixture of voice and data, priority needs to be given to voice – the IEEE 802.1p standard was developed to address this issue. Since a voice packet is encapsulated within an Ethernet frame when transmitted across the Layer 2 LAN switching infrastructure, it is important that all switches support the 802.1p standard. An Ethernet frame has no defined priority field, but the 802.1p standard specifies the use of 3 bits from within the VLAN identification header for prioritisation. 3 bits will provide 8 levels of priority (on a level of 0 to 7) and Priority 6 is used for voice. This ensures voice traffic will receive priority over any data packets.

QoS on the WAN

Since WAN links are generally low bandwidth (typically in the range of 64Kbps to 2 Mbps) when compared to Gigabit/Fast Ethernet LANs, there is added importance in ensuring QoS for voice traffic across these often congested links. The main QoS implementations for WAN links are IP Precedence, Queuing, RSVP and RTP Header Compression.

IP Precedence – An IP packet has a Type of Service (ToS) field in which 3 bits are used for defining six classes of service. With IP telephones, precedence level 5 is given to voice packets which ensures priority over other (data) packets.

Queuing – Since a router will receive a mix of voice and data packets on its LAN interface, it must be able to provide a means for time sensitive traffic such as voice to get across the WAN link while slowing the flow of other traffic i.e. data. Queuing provides such a means: Priority Queuing will always forward voice traffic before data while Weighted Fair Queuing (WFQ) will ensure that bandwidth is shared between voice and data packets with priority given to voice – based upon its detection of the IP precedence bit in the voice packets. If WFQ is employed it is recommended that fragmentation is used to break data packets into smaller fragments to avoid a voice packet becoming queued behind a larger data packet.

RSVP – Resource Reservation Protocol (RSVP) allows applications to dynamically reserve network bandwidth and thereby request a specific QoS for a data flow. RSVP provides for an end-to-end guarantee of reserved bandwidth so all devices along the route (hosts and routers) must support RSVP.

RTP Header Compression – On low speed serial links (< 2 Mbps), RTP header compression can be used to reduce network overhead. This will compress the RTP/UDP/IP header from typically 40 bytes down to 2-5 bytes. This will result in significant bandwidth savings because, as often occurs with VoIP, the actual ‘payload’ or data packet within the header can be quite small, with the header itself consuming a significant portion of the bandwidth.
COMPONENTS OF A VOIP SYSTEM

Now that the fundamental technologies employed in VOIP have been outlined, one needs to consider the physical implementation. In other words, the components that go to make up the system. These are as follows:

The IP Telephone

The IP telephone is a new device that causes some confusion amongst people used to traditional phones. Although the device resembles a normal phone which traditionally connects to a proprietary PBX port, the IP phone connects into an Ethernet LAN in the same way as a desktop PC does. As IP phones have a unique Ethernet MAC address (just like a standard NIC in a desktop PC), after the initial configuration they can be plugged in and used anywhere on the LAN. Some IP phones have a built-in hub which allows connection of the LAN cable into the phone, and then cascading via a patch cable connection to the PC.

IP phones usually require an external power supply, but some vendors are addressing this issue by developing LAN switches which allow DC voltage supplies to be carried along existing (unused) wires within the Ethernet UTP cables, thereby removing the necessity to have external PSs cluttering the desktop.

In fact, there is no actual need for a physical telephone at all. Softphone is a software simulation of a telephone which runs on the users PC. With a sound card and connected microphone/headset the PC can provide all the functionality of a real phone while enabling multimedia communication.

The LAN Switch

An existing component of the user’s LAN, this switch has added importance by keeping delay to a minimum – vital where VoIP is concerned. Only LAN switch connections should be used with IP telephones. Shared media devices such as Ethernet hubs should not be used as the collisions inherent within such devices can introduce unacceptable delays. It is also paramount that the switch has QoS support (such as the 802.1p standard where a number of bits within the Ethernet frame are used for prioritisation) to enable voice packets to receive priority switching across the LAN.

Redundancy is also an important issue that needs to be considered. In an IP telephone-only environment, any failure of a LAN switch or an inter-switch link could result in the loss of phone connectivity, with potentially dire consequences for a company. As a result, the LAN infrastructure should feature redundant inter-switch links and even redundant switches. On larger campus sites one should also consider implementing layer 3 switching in the backbone in order to provide subnetting and broadcast containment.

The IP Router

As well as being existing components of the user’s WAN infrastructure, routers are used for data transmission across company WAN links, commonly in the guise of either leased lines, ISDN or Frame Relay connections. When packet voice is introduced on these WAN links it may be necessary to increase the bandwidth of the leased line or the CIR (committed information rate) in order that the Frame Relay connection can cope with the increased traffic. When looking to ensure prioritisation for voice packets over data, it also becomes vital to consider the QoS features offered by the router.
The IP PBX
Also known as the Service Control Unit (SCU) or the Call Manager, this is essentially the heart of the VoIP system and performs all the functions of a traditional PBX. These functions include call switching and administration as well as ‘gatekeeper’ functionality, such as translating between telephone numbers and IP addresses, call signal processing, and call establishment & management. It will also run core voice applications such as voice mail, auto attendant and web-based call centre applications. The IP PBX can be implemented in either hardware or software. Some vendors offer software implementations which will run on standard platforms such as Windows NT servers, whilst hardware implementations are vendor specific; usually taking the form of a chassis based system, with a modular call processor and a trunk gateway. Hard drives are usually incorporated to provide voice mail capability.

The PSTN Gateway
This allows translation between the IP network and the PSTN or, in other words, between the packet switching domain and the circuit switching domain. This is essential because it is obviously still necessary to route calls originating from IP phones externally to the PSTN. Since gateways can interface to existing PBXs this means that existing legacy equipment can be retained to co-exist alongside the IP telephony system. Gateways can be standalone external devices, modules for routers, or incorporated within vendors’ chassis-based IP PBXs.

Integrated Systems
Many of the major networking vendors are now offering integrated IP telephony systems, comprising a chassis-based system housing the IP PBX and PSTN gateway and router, which are also platforms for the voice mail, auto attendant and web-based voice/data converged applications. Hard drives are usually incorporated to provide voice mail capability. These systems are usually bundled with IP phones making them an attractive proposition to customers who need a packaged solution to their VoIP requirements.
**CTI APPLICATIONS**

**Customer Relationship Management (CRM)**

As previously discussed, it is not just cost savings on long distance calls that will drive acceptance of IP telephony systems but also the establishment of web enabled call centres via CTI applications. With the explosive growth of the Internet, companies are increasingly looking to utilise the web to enable transactions and provide customer service. A typical example of this would involve an interested customer accessing a company’s web site and entering details for information about a product. Entering their phone number initiates a company call-back, the call centre agent receives a message on their PC to contact the prospective (or existing) customer which they do using a SoftPhone on their PC. As soon as the number is dialled, the PC displays all the customers details on-screen that are automatically retrieved from their database. The whole process is automatic and greatly improves the efficiency of the call centre agent who doesn’t waste valuable time searching for the relevant information. Such applications are generically described as Customer Relationship Management (CRM) software.

**Unified Messaging**

Unified Messaging integrates voice mail, email and fax mail into a single application suite. Remote workers can use their notebook computers to dial into their company network and download their emails. These emails can also contain voice mail attachments (WAV files) which the person simply clicks on to hear re-played. This system can also allow a worker sitting in an airport lounge to retrieve their emails over their mobile phone via text-to-speech conversion carried out at the company headquarters.
Migration will be the most common scenario because unless the installation is on a ‘greenfield’ location, most sites will already have an existing traditional telephone infrastructure. It is alongside the incumbent infrastructure that the IP telephony system must be integrated. A phased migration is recommended as it allows for IT staff to learn VoIP technology at their own pace and businesses can save on long distance phone charges. The stages of implementation are shown below:

**Stage 1: Testing Voice Transmission over Existing WAN Links**

This can be accomplished with existing analogue phones, PBX and routers. The primary requirement is a gateway to the IP network which will be in the form of either a module for the router or an external standalone device. The gateway will provide interconnectivity between the phone/PBX system and the router and will provide functionalities such as sampling, digitisation and packetisation. Initially two sites should be selected with just one or two phones at each location. Network managers are then able to analyse WAN traffic to assess the impact upon bandwidth utilisation, while an incremental addition of more phones would allow the network manager to predict any need for additional bandwidth capacity. A financial analysis could also be carried out to compare savings made on these inter-site calls.

With the WAN links successfully transmitting voice traffic between sites, it is possible for the IT staff to implement QoS features on the routers at both ends to test the impact of heavy data traffic upon voice quality. QoS features such as priority queuing and RTP header compression are standard features on most routers and are easily configured to enable experimentation. At this stage voice and data would only be combined over the WAN links.

**Stage 2: Incorporating IP Telephones**

With the LAN suitably prepared for VoIP (hubs having been replaced by 802.1p compliant switches in the wiring closet) IP telephones would now be connected to the LAN, along with the IP PBXs at both sites. No routing changes are required and external phone calls can still be switched via the existing PBX. LAN traffic can be analysed and with the addition of more IP phones, engineers can now look at planning the phase out of all remaining analogue/digital phones at the site.

**Stage 3: Removal of the PBX**

With the migration to IP telephones completed, the traditional PBX is removed and replaced with an E1/T1 trunk from the gateway to the Local Exchange.
CONCLUSIONS

This paper is intended to present an overview of voice and data integration, and specifically the compelling arguments for a move towards VoIP and the technologies involved in its implementation.

It is widely recognised that an increasing number of companies with multiple locations will begin moving from their conventional phone systems to IP telephony systems within the next five years. Recent studies have shown that 10% of major enterprises have already begun experimenting with and implementing IP telephony technology.

In the face of the rapid development of VoIP technology, traditional PBX vendors and established telcos have been quick to counter the perceived threat to their business. Reduced rates on long distance calls are being aggressively marketed and PBX vendors have begun introducing packet switching gateway add-ons to their products.

Clearly, it is unlikely that the traditional PBX market is going to disappear overnight. Nevertheless, the innovative nature of packet voice technology and its practical business advantages have made VoIP a very credible alternative. Indeed with all of the major networking vendors now offering VoIP products, both as standalone systems and as add-on modules to their existing router products, we can safely say that the era of packet voice has truly arrived!
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