# Voice over IP: what it is, why people want it, and where it is going

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## Executive summary

VoIP is a set of technologies that enable voice calls to be carried over the Internet (or other networks designed for data), rather than the traditional telephone landline system—the Public Switched Telephone Network, or PSTN. The potential for very low-cost or free voice calls is driving the use of the technology but in the long-term, VoIP is more significant than just free phone calls - it represents a major change in telecommunications. The fact that VoIP transmits voice as digitised packets over the Internet means that it has the potential to converge with other digital technologies, which in turn will result in new services and applications becoming available.

However, the adoption of VoIP is not without complications. The traditional PSTN telephone infrastructure has been built up over the last one hundred years or so and has developed into a robust voice communications system that provides reliability figures of nearly 100%. In contrast, VoIP is a relatively new technology with a fledgling architecture that is built on inherently less reliable data networks. This means that there are therefore justifiable concerns around the extent to which it is deployed.

VoIP has developed considerably in recent years and is gaining widespread public recognition and adoption through consumer solutions such as Skype and BT's strategy of moving to an IP-based network. This adoption is spreading into the F&HE domains, with a number of institutions implementing VoIP, and through work being undertaken by UKERNA.

The technology offers opportunities for the development of new applications and educational services, particularly through the potential for converging voice with other media and data. In the long-term, VoIP is likely to impact on some of the bigger picture developments within further and higher education such as virtual universities, identity management, and integration with enterprise-level services and applications.

# 1. What is VoIP?

## 1.1 An introduction to VoIP

VoIP is a set of technologies that enable voice calls to be carried over the Internet (or other networks designed for data), rather than the traditional telephone landline system—the Public Switched Telephone Network, or PSTN.

The term VoIP was coined by the VoIP Forum which was set up in May 1996 as an industry group concerned with promoting and developing product interoperability and a high quality of service for Internet telephony products[1]. Initially, one of the main drivers in developing VoIP was the potential to cut the cost of telephone calls. Traditional voice calls, running over the PSTN, are made using circuit switching, where a dedicated circuit or channel is set up between two points before the users talk to one another—just like old-fashioned operators, plugging in the wires to connect two callers. The advantage of this is that once the circuit is set up, the call quality is very good, because it is running over a dedicated line. But this type of switching is expensive because the network needs a great deal of (mostly under-used) capacity.

The development of VoIP represents a major change in telecommunications. VoIP uses IP protocols, originally designed for the Internet, to break voice calls up into digital 'packets'. In order for a call to take place the separate packets travel over an IP network and are reassembled at the far end. The breakthrough was in being able to transmit voice calls, which are much more sensitive to any time delays or problems on the network, in the same way as data.

Whereas calls over the PSTN are metered, so the user pays for the amount of time taken by their call, Internet usage is not metered. The user pays a set fee for their Internet service and their VoIP service and can then use the Internet to get free phone calls to other users on the same VoIP service, or pay a small fee to call users on other VoIP services or on the PSTN.

Packetised voice also enables much more efficient use of the network because bandwidth is only used when something is actually being transmitted. Also, the network can handle connections from many applications and many users at the same time, unlike the dedicated circuit-switch approach. This greater efficiency is one of the main reasons that all major carriers, such as BT with its 21CN (21st Century Network) project[2], are changing their own networks so that they are IP-enabled.

## 1.2 How VoIP works

### 1.2.1 The Basics

The basic process involved in a VoIP call is as follows:

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[1] This VoIP Forum (there are many) was subsumed into the International Multimedia Telecommunications Consortium's (IMTC) Session Initiation Protocol Activity Group in October 1996. The SIPAG's website is available at: http://imtc.org/activity_groups/SIP.asp [last accessed 04/09/06].

[2] See: http://www.btplc.com/21cn/ [last accessed 14/09/06].

1. Conversion of the caller's analogue voice signal into a digital format
2. Compression and translation of the digital signal into discrete Internet Protocol packets
3. Transmission of the packets over the Internet or other IP-based network
4. Reverse translation of packets into an analogue voice signal for the call recipient.

The digitisation and transmission of the analogue voice as a stream of packets is carried out over a digital data network that can carry data packets using IP and other, related Internet-related protocols (see section 2.1). This network may be an organisation's internal LAN, a leased network, the PSTN or the open Internet (Gradwell, 2006). The compression process is carried out by a *codec*, a voice-encoding algorithm, which allows the call to be transmitted over the IP network within the network's available bandwidth.

### 1.2.2 What you need to make a VoIP call

To make a VoIP call, the consumer user requires VoIP software and a broadband connection to the Internet. The software will handle the call routing to make sure the call reaches the intended destination as well as providing the codec. The software can be installed on a variety of hardware devices including traditional telephone handsets (using an adaptor that plugs into the telephone[3]) or a PC or wireless device such as a Personal Digital Assistant (PDA). This use of software-enhanced end-user devices is one of the key distinguishing features of VoIP. Whereas the traditional telephone system contains its 'intelligence' within the network, VoIP makes use of the Internet model of intelligence at the *edge* of the network. This is often known as the end-to-end principle.

In order to make a call, an account with a VoIP service provider is also required. Different types of VoIP service are available, including services from traditional telephone carriers such as BT, and from specialised VoIP providers such as US firm Vonage and Luxembourg-based Skype. Some VoIP providers provide support only for PC-to-PC calls, while others provide the ability to make and receive calls from IP-enabled devices to users on the PSTN and on mobile networks.

UK telecoms regulator Ofcom advises UK consumers to carefully check the different services available from VoIP providers, including whether or not the provider offers a backup service to make calls via the PSTN if there is a problem with the broadband connection and offers access to the emergency services.

## 1.3 How VoIP is used, its background and history

Most people are aware of VoIP through the Skype consumer telephone service which has gained large-scale public recognition recently, particularly since its purchase by Ebay. However, VoIP has not suddenly appeared in the last few years as an opportunity afforded by the World Wide Web. Skype is only one particular implementation of VoIP and its related technologies and it is important to understand that VoIP has an important technological

[3] A traditional phone may be used by making use of an Analogue Telephone Adapter (ATA) or alternatively an IP Phone can be used (Ofcom, 2005)

history, intertwined with the telecommunications industry in general, in order to appreciate the complexities of VoIP technologies and applications.

The idea of voice over IP has been discussed since the 1970s but it was the mid-1990s before commercial products became available with the introduction, in 1995, by Israeli company Vocaltec[4], of the first commercial system (Varshney et al., 2002). These early VoIP systems were designed to connect one PC to another and required each PC to have a sound card, speakers, microphone, modem and VoIP software. The software encoded and compressed the voice signal, converting it into IP packets that could be transmitted over the Internet. With this approach, both users used headsets, plugged into their PCs. The calls could only be made between PCs and could not connect to the PSTN network.

In parallel, from the 1970s onwards, traditional telecommunications carrier companies were developing new systems that introduced IP-enabling software for traditional telephony equipment. Human speech is an analogue wave signal and historically, voice telephone calls had been made over networks using analogue circuits which provided a temporary end-to-end connection, through the network, for each call (Sherburne and Fitzgerald, 2004). This is known as circuit switched, and builds on the original phone network of local telephone exchanges, in which wires between households were literally connected together for the duration of the call by a telephone operator. The companies that provided these services were often public agencies, usually part of a country's post office service, and such networks became known as Post Office Telephone Systems (POTS), sometimes also referred to (post privatisation) as Plain Old Telephone Systems. The Public Switched Telephone Network was the name given to the overall network created by telephone companies.

Between the 1950s and the 1990s these analogue systems were replaced by digital networks and telephone exchanges which made use of high-speed leased lines (known as T1 lines) and modern digital computer technology in the telephone exchanges, and digital signalling protocols such as ISDN between exchanges. However, these newer systems still relied on the circuit switch concept for end-to-end connection so for the consumer, things remained analogue since, by and large, the connection between the local exchange and the household remained a simple copper wire.

In the 1990s, with the Internet and Web boom, telephone equipment manufacturers and telecoms companies also began to make increasing use of the idea of transmitting digital information between exchanges through IP-based packets. This was in part driven by the lower costs associated with transmitting voice calls in this manner, as bandwidth use is more efficient[5]. From the mid-1990s onwards, telephone equipment manufacturers added IP capabilities to their existing PBX telephony switches[6] and, more recently, software has been developed to enable users to plug a VoIP adaptor into their traditional telephone. In this way, VoIP calls can start and end on the PSTN, but are then routed, via a software gateway, over the Internet.

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[4] www.vocaltec.com

[5] "as much as half that of a traditional circuit-switched network (such as the PSTN)" (Varshney et al., 2004, p.90)

[6] Private Branch Exchange (PBX) – private, local networks and exchanges implemented within an institution or company, often referred to simply as *switches.* These exchanges operated, in effect, as intelligent networks as they were equipped with additional functionality, not available on the PSTN. This functionality included, for example, sophisticated call routing, conference calling, support for pager systems, premium number barring etc.

This history means that VoIP is operating in a heterogeneous environment that extends way beyond the Internet. Voice calls need to have the potential to be carried over a variety of different networks including local networks, PBXs, PSTN and the Internet. Advances in VoIP technology mean PC telephony software is available from many software developers. Gateway servers with voice-processing cards are also available, to act as an interface between the Internet and the PSTN, enabling users to make calls either from their PCs, or from an IP phone, into the traditional telephone networks. Calls can also be made using IP handsets, which look similar to traditional phones, but which are plugged into an IP-based network rather than into the traditional telephony network, and have more features and capabilities than traditional telephones. The result is that there are now a number of ways in which VoIP can be implemented:

- PC to PC. Both the caller and recipient use headsets plugged into their PC.
- PC to PSTN. Only the caller uses a headset. The recipient receives the call in the traditional way.
- PSTN to PSTN. The caller uses an IP adaptor on their traditional telephone and the call is received on a traditional phone. But the call travels over an IP network.
- IP phone to PSTN. The caller uses an IP phone, and the call transfers from the IP network to the telephone network via a gateway.
- IP phone to IP phone. The call travels over an end-to-end IP network.

It should be noted that there is confusion amongst communications professionals and industry commentators as to the use of terms like “VoIP”, “Internet Telephony” and “IP Telephony'. In this report we shall use the term VoIP to refer to the set of technologies that allow voice to be transported over an IP infrastructure (in effect, an IP-enabled PSTN) and the term IP telephony (IPT) to refer to VoIP technologies that also incorporate and build on the more advanced functionality provided by the old PBX systems (see footnote six).

**1.4 The scope of this report**

This report will provide a basic introduction to VoIP and its associated technologies and standards, and will explain the issues with regard to VoIP’s use in higher and further education. It will also outline some of the factors that need to be considered when thinking about implementing VoIP, such as call quality, peer-to-peer network problems, packet discrimination, wireless VoIP, directories and addressing. Issues regarding the perceived pros and cons of using VoIP in an educational setting will be explored, such as the concern about use of Skype.

The report will also outline in brief the developments to date that have been initiated by UKERNA with regard to VoIP. It will outline in more detail some of the interesting emerging VoIP applications and services and highlight potential educational uses, including real-life examples from HE, which will help illuminate the technology’s potential for convergence between voice communication, multimedia and data. It will also speculate on the potential future development of VoIP and its associated technologies over the next five to ten years, and the ways in which these are likely to impact on some of the bigger picture developments such as virtual universities, identity management, and integration with enterprise-level services and applications.

## 2. Technology, standards and challenges

As already mentioned, the traditional PSTN telephone infrastructure has been built up over the last one hundred years or so and has developed into a highly reliable voice communications system. In contrast, VoIP is a relatively new technology with a fledgling architecture that is built on inherently less reliable data networks. This means that there are therefore justifiable concerns around the extent to which it is deployed. The purpose of this section is to examine the underlying protocols and technologies used in VoIP and to discuss the potential challenges inherent in their deployment.

### 2.1 Protocols

In common with other modern telecommunications systems, VoIP is structured around two key components: the *bearer* (the actual voice being sent over the network) and the *signalling* (additional messaging necessary to control and handle other call elements such as the dialled digits for the destination). Both these components make use of a variety of standards and protocols.

**Table 1: VoIP Protocol stack and comparison with the OSI model**

| VoIP Protocol | Layer | Common Internet Equivalent | OSI Model |
|---|---|---|---|
| VoIP Appl., SIP | 7 | HTTP | Application |
| H.323 | 6 | | Presentation |
| RTP, RTCP | 5 | SSL | Session |
| UDP | 4 | TCP | Transport |
| IP | 3 | IP | Network |
| Data | 2 | Ethernet | Data |
| Physical | 1 | 100-Base T | Physical |

The fundamental building block for a VoIP call is, as with other Internet-based applications, the layered/tiered architecture of the Internet, which is outlined in table 1. This shows how layers of protocols build on top of each other to produce applications that can be delivered and handled over the Internet's physical structure. Lower layers of the VoIP version of this stack share considerable commonality with other Internet-based protocol stacks, such as the Web, whereas higher layers, which are abstractly ‘closer’ to what the end-user experiences, differ. Lower layers concentrate on the physical transmission and delivery of data, higher levels on tasks such as controlling the communication (e.g. signalling).

Table 1 demonstrates how VoIP shares the lower, physical and data layers, based on technologies such as Ethernet and WiFi, with other Internet-based applications such as the

Web and FTP. The network layer is, again in common with other Internet applications, handled by the Internet Protocol (IP) which handles the basic transmission of packets (also known as datagrams) of data. VoIP systems build on this network layer not by using the more common Transmission Control Protocol (TCP) but by making use of User Datagram Protocol (UDP), and Real-Time Transport Protocol (RTP) as a transport layer[7] (Goode, 2002; Varshney et al., 2002).

Higher up the stack, the session, presentation and application layers are used to handle the signalling required for a telecommunications system. These protocols control the communication between the two end points of a call e.g. the call set-up and signalling necessary to transmit traffic over an IP network. There are two key signalling standards: H.323 and the Session Initiation Protocol (SIP):

- H.323. This is the protocol ratified in 1996 by the International Telecommunication Union[8], and is a well-established protocol for handling voice, video and data conferencing over packet-based networks (Mehta and Udani, 2001). It follows many of the message sequencing conventions established by PSTN signalling and therefore allows considerable scope for interoperability with existing PSTN systems. There is also an OpenH323 project, which aims to create a fully featured, interoperable, Open Source implementation of the ITU standard[9].

- SIP is a text-based, open protocol, ratified by the Internet Engineering Task Force (IETF), for telephone calls over IP[10] and has recently gained rapid uptake in spite of the fact that H.323 is the more established protocol. SIP does a similar job to H.323 (Packetizer, 2006), but was specifically designed for the Internet (Mehta and Udani, 2001). It takes a more modular approach than H.323, is similar to the HTTP Web protocol, and sets up a "session" over the Internet (Goode, 2002). The content of SIP-based sessions can range from a basic telephone call to complex, multi-party, mixed media sessions, so SIP enables services to be created that combine elements from telephony and other Web-based applications, such as email, messaging and video streaming (and hence encourages convergence)[11].

## 2.2 Codecs

Prior to the transmission of a voice call across an IP-based network a person's voice (which is an analogue sound wave) must be converted to a digital form and encoded. A certain amount of data compression can also take place in order to save bandwidth during the subsequent transmission. On receipt of the voice data at the other end this process must be reversed. A number of different voice-encoding algorithms are used (codecs) which have been

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[7] The normal TCP/IP combination is not suitable for real-time situations such as voice calls because it allows for the retransmission of lost packets and therefore introduces delays. UDP does not support re-transmission and so conserves bandwidth (at the expense of some packet loss). RTP builds on UDP by providing additional information such as packet sequence numbering and time stamping (which help in the rebuilding process from individual packets that takes place at the receiving end).

[8] http://www.itu.int/rec/T-REC-H.323/en [last accessed 12/09/06].

[9] http://www.openh323.org [last accessed 12/09/06].

[10] http://www.ietf.org/html.charters/sip-charter.html [last accessed 12/09/06].

[11] Further detail on SIP can be found on p. 4 of Ahuja, S. and Ensor, J. *VoIP: What is it good for?* ACM Queue.

standardized by the ITU as a series of recommendations known as the G-series (Sherburne and Fitzgerald, 2004). The common ones are G.711, which is in widespread use in the telecommunications industry within PSTN networks, and G.729[12]. Codecs differ in the algorithms they use for sampling the analogue voice wave and the sophistication of the compression used. This in turn determines the amount of digital bandwidth required for the encoded sample. G.711, for example, requires a relatively higher bandwidth (of 64Kbps which in practice translates to 90Kbps in an actual VoIP implementation) whereas G.729 operates at 8Kbps (Nooning, 2005). However, ultimately there is a trade-off between the sophistication of the algorithms, the amount of bandwidth required and the quality of the voice signal received.

### 2.3 Factors to be considered when implementing VoIP

The traditional PSTN telephone infrastructure has been built up over the last one hundred years or so and has developed into a highly reliable voice communications system and provides reliability figures of nearly 100%[13] (Chong and Matthews, 2004). It has its own power supply so that in case of power loss (e.g. in the event of a fire), the telephone will still work. In contrast, VoIP is a relatively new technology with a fledgling architecture which is built on inherently less reliable data networks (Street, 1999) and there are therefore justifiable concerns over such things as:

- voice quality
- reliability of service
- access to 999 services. This is not a legal requirement in the UK at present, but UK telecommunications regulator Ofcom is encouraging VoIP providers to offer access to 999 services.
- directories and addressing

Within H&FE there are also additional concerns surrounding the use of peer-to-peer applications, mainly to download music and games, that have been further exacerbated by the use of Skype — the consumer, peer-to-peer VoIP system.

In the UK, the main networking service for F&HE institutions, as well as for research bodies and specialist colleges, is the JANET network, run by UKERNA. Traditionally, voice services within JANET-connected organisations have been managed and implemented locally, with external connectivity provided by telecommunications companies. However, as VoIP develops, many institutions, and UKERNA itself, have been looking at the potential of providing IP-based voice services.

In response to this growing interest from the education community in using JANET to carry voice traffic, UKERNA set up a Voice Advisory Group (VAG) early in 2005. The group is considering how VoIP will fit in to an overall strategy for voice services and the application services support requirements on JANET. A requirements analysis was initiated in August 2005 and resulted in a strategy for the support of voice on JANET (UKERNA, 2006a). The

[12] A more detailed list of common codecs can be found at: http://www.tech-pro.net/voice-over-ip.html. The full list of G-series ITU recommendations is available online at: http://www.itu.int/rec/T-REC-G/en [last accessed 12/09/06].

[13] Actually 99.999 percent, known as the "five nines" of reliability within the industry.

Voice Advisory Group is working on such aspects as directories and addressing, security and integration of VoIP with IP-based videoconferencing.

### 2.3.1 Voice Quality

A key consideration with VoIP is the real-time nature of voice. When voice is transferred as data, problems on the network will immediately affect the quality and reliability of the call and these can lead to callers missing part of the conversation, having echoes on the line, or getting poor sound quality—the so-called 'Dalek effect'.

According to Roberts (2005) and Chong and Matthews (2004) the main technical issues for voice services over an IP network are:

- Latency – delays in packet delivery
- Jitter – caused by *variations* in the delay of packet delivery (i.e. variations in the latency)[14]
- Packet loss – packets are lost during transmission or simply arrive too late to be used. Alternatively, the network actually 'drops' packets during periods of network congestion.

Of these, latency is the most important as it directly affects voice quality. Latency is the time delay that occurs during the process of transporting the voice call packets over the network. This latency is composed of two kinds of delay – propagation delay (the time taken for the packets to travel along copper wires or fibre-optic cables) and handling delay (the process of digitizing, placing the data in packets and the passage through hardware devices such as routers) (Chong and Matthews, 2004). The lower the latency the more 'natural' the conversation sounds (Mehta and Udani, 2001).

The latency in an IP system is influenced by a number of variables including the level of traffic on the network, the packet size and the number of routers and gateways the packets must pass through (Varshney et al., 2002). When delay in the network exceeds about 150 milliseconds the natural flow of conversation is interrupted (Ahuja, 2004). Jitter is slightly less of an issue for voice calls, but when consecutive voice packets don't arrive at evenly spaced intervals, the result can be a 'distortion of sound, which can make the speaker unintelligible if the distortion is severe' (Chong and Matthews, 2004, p. 107).

Packet loss occurs when packets are lost during transmission or simply arrive too late to be used. Transmission of data (such as a webpage) makes use of the TCP/IP protocol suite which allows for retransmission of missing packets, but VoIP, which uses UDP, does not allow retransmission and the missing packets are simply left out of the call. Such loss causes voice clipping and skips (Goode, 2002). This is less of a problem than latency or jitter, since the codecs used in voice processing can cope with a certain amount: up to 1% is usually undetectable, more than 3% is the maximum permitted within industry standards (Chong and Matthews, 2004) whilst 10% or more will not be tolerated by the listener (Roberts, 2005).

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[14] An in-depth discussion of jitter can be found at: http://www.voiptroubleshooter.com/indepth/jittersources.html [last accessed 12/09/06].

#### 2.3.1.1 Quality of Service (QoS)

Quality of Service is the main approach used to maintain an adequate standard of voice quality. It is not a standard or protocol, but 'simply a generic industry term for outlining technologies, standards, and strategies to provide for network quality' (Intel, 2006, p.2). QoS puts in place technological mechanisms to reserve dedicated bandwidth, manage network congestion and prioritise packets which have real-time sensitive data (Mehat and Udani, 2001). The latter, known as traffic shaping or Differentiated Services, relies on the IP packet including information on what type of service is being transported and the capability of network equipment to offer priority to certain types or classes of such data. This is fairly uncontroversial within a well-managed private LAN or PABX network, but when carried out on the open Internet the potential use of such prioritization has caused controversy (see section 2.3.3).

JANET is running a project to look at QoS development and the first phase of the project concluded that voice and video traffic can benefit from IP QoS by receiving priority treatment over the network. Phase 2 of the QoS development project is looking at the feasibility of implementing a QoS service across JANET.

### 2.3.2 Peer-to-peer networking (e.g. Napster, Kazaa, Skype)

The type of VoIP implementation most well known to consumers and students uses client-based, peer-to-peer (P2P) technologies such as Napster and Skype (Jones, 2005). P2P systems are far more decentralised than other networking or computer systems, with little or no reliance on the idea of clients being served by a central server. Clients (individual computers or telephony end points) act as nodes and are as likely to be ‘providing’ resources to other nodes (its peers) as to be consuming them from other nodes.

It is fairly well known within higher and further education that such client-based P2P applications can cause problems on IP networks because they make it much harder to manage the use of bandwidth and to exercise some level of control over the use of the network. There are particular problems with applications such as Kazaa and Napster, which are used to download music, films and games and the challenge is growing with the further development of P2P applications such as BitTorrent and Groove Networks’ Virtual Office, and the increasing popularity of instant messaging (IM) which is also a P2P application (see section 3.2 for more on IM). The use of Skype within college networks has raised similar concerns.

Many institutions see Skype as a good thing as it enables overseas students to keep in touch with their families at low cost and, as a consumer technology, has encouraged renewed interest in the potential for new developments in distance learning. However, because of the way it works, Skype is also considered by many to pose potential security threats, as well as possibly creating a major overhead on the network over which it is running.

Skype works by forming an *ad hoc* decentralised network of ordinary nodes and super nodes. The only centralised service is the Skype login server that stores user login and password details: the handling of voice calls is undertaken entirely by the decentralised network of nodes and super nodes. In order to make a telephone call, ordinary nodes must connect to an available super node (Baset and Schulzrinne, 2004). In this arrangement ‘any node with a public IP address having sufficient CPU, memory and network bandwidth is a candidate to become a super node’ and a Skype client “cannot prevent itself from becoming a super node” (Baset and Schulzrinne, 2004, pp.1 and 2).

This obviously has potential implications for the node that becomes a supernode with regard to the consumption of network resources, and this has been one of the concerns within UK HE institutions. UKERNA noted in March 2006 (2006b) that 'Networks with super-nodes may experience large flows of in-bound and outbound traffic that have no connection with any local user' (p. 2). Skype acknowledges that its software depends on the willingness of users to contribute in what it calls 'a minor way' to the network itself, but says it has engineered its system so that users who have become super nodes will not be able to notice any decrease in the performance of their computers because the data bandwidth and computing power usage is minimal (Dudman, 2006).

In addition to the concerns over bandwidth usage it is worth remarking that Skype differs from some of the other VoIP services in that it is based on the company's own, proprietary protocols, rather than the standard SIP protocol and one implication of this is that Skype users can only connect with other Skype users (Jones, 2005). UKERNA notes that Skype is not the only IP telephony system and that 'Alternatives that are standards-based may prove easier to manage and provide a more predictable service' (UKERNA, 2006b, p.4). For example, VoIP systems such as the Gizmo Project[15] use the SIP protocol and can connect effortlessly to multiple VoIP networks and SIP-based PBX systems.

**Case Study: Brunel University**

Brunel University is one HE institution that has been trialling different approaches to managing P2P traffic, including Skype calls. It had a number of concerns about the possible impact of uncontrolled and ubiquitous P2P applications and wanted to take precautions both to protect its bandwidth and to ensure the use of Skype, in particular, would not expose the university to potential security vulnerabilities (Dudman, 2006).

Initially, the university network team attempted to block Skype traffic. They used the access control list to block direct access by Skype to the network. However, it was discovered that it was very difficult to block Skype completely, because of the software's ability to find alternative routes into and across a network. There were also concerns within the university that blocking Skype would cut off useful applications, particularly the ability of overseas students to make free phone calls to friends and family in other countries.

As a consequence, Brunel has now changed its policy. It is using a proprietary software package, Packetshaper to partition Skype traffic and the university's network team is now monitoring the impact of Skype on the university network. Since this policy has been implemented, the maximum level of Skype traffic that has been registered has been about 1Mb, which has not breached the limit set.

A similar approach has been taken at Manchester University, which was concerned about unsanctioned downloads, which it estimated was taking up nearly 70% of its available bandwidth. Manchester has not banned use of Skype, which it regards as useful P2P traffic (Thomas, 2006).

[15] http://www.gizmoproject.com/ [last accessed 12/09/06].

A further consideration with regard to Skype is that it is considered by many to pose a potential security threat because of the way it works. Skype traffic is encrypted and uses a random combination of IP addresses and ports. This means it is hard to detect Skype calls because they run through their own, encrypted 'tunnel' over a network (UKERNA, 2006b) and there is concern that this is a possible way in which viruses or other problems could be introduced into a network, without being easily detected (Blackwell, 2005).

It is possible to block unauthorised Skype traffic. Blocking specific types of traffic over an IP network is usually done by blocking ports or denying access to specific IP addresses. But Skype traffic, because of the way it travels over the network and in particular its use of random combinations of IP addresses and ports, causes problems for traditional port blocking filters.

Skype traffic can be identified, and therefore blocked, by investigating the headers of every IP packet crossing the network. The challenge is to do this quickly, so other network services are not affected. VoIP blocking is often a function added to existing network or security management software, such as Narus's IP Platform, Verso Technologies' NetSpective 2.0, and SonicWALL's enterprise appliances. Other systems able to help manage and control IP networks include Ellacoya's IP Service Control System, Sandvine Broadband Network Management, and software from P-Cube, now owned by Cisco.

A proxy appliance, widely used to apply controls to Web traffic, can also be used to block specified unwanted traffic, including voice calls, if necessary. Deep Packet Inspection[16] is another approach, developed from firewall technology, that can also assist in intrusion detection and prevention.

UKERNA (2006b) has issued a policy document on Skype and JANET, recommending that JANET-connected organisations should consider how they manage Skype in order to protect their own network availability and those of others, to ensure filters to hostile and inappropriate material remain effective and to comply with the JANET acceptable use policy.

### 2.3.3 Net Neutrality and packet shaping

At present, the Internet is an end-to-end network, over which, in theory, any traffic can flow freely, and no one application or use is prioritised over another. The network carriers and ISPs are, in effect, neutral as to the traffic they carry — known, unsurprisingly, as network neutrality. Recently there has been considerable controversy, particularly in the US, over plans to alter this. Major US telecommunications carriers such as AT&T and Verizon want to prioritise some services and potentially charge different rates for premium services. At the moment there is nothing to stop them from doing this (other than the general ethos of the Internet) so objectors have called for regulation of the telecommunications carriers to ensure network neutrality (Stern, 2006). The key issue is whether prioritising newer services such as VoIP and Video on Demand, which are more sensitive to network problems, will lead to a 'two-tier' Internet, with some services running more slowly, and less reliably, than others. This is the policy context within which the rapid rise of VoIP is operating.

[16] For a technical explanation of Deep Packet Inspection see Porter, 2005.

There are three main ways to prioritise Internet traffic, a process known as traffic management, traffic shaping or packet shaping, because it is all about prioritising specific types of traffic and specific packets of content:

- **Best effort**. All packets treated on a first in, first out basis. When there is unmanageable congestion, all types of packets can be dropped
- **Needs-based discrimination**. Packets are treated as first in, first out until there is congestion, at which time certain packets are given priority
- **Active discrimination**. Packets are examined as they enter the router and are prioritised even when there is sufficient bandwidth.

Packet-shaping policies are important in the development of IP-based voice services because they can provide a framework in which to prioritise voice applications, which are highly sensitive to any network disruption. There is concern that when such traffic-shaping policies are imposed by network operators or ISPs they could lead to packet discrimination, with content from some independent providers being discriminated against or excluded altogether, leading to the end of network neutrality. Interested readers should see Felten's (2006) excellent overview of the technical issues in relation to the policy and legal issues.

There have already been instances in which providers have expressed alarm over the potential impact of network discrimination. In 2005, US VoIP provider Vonage complained to the Federal Communications Commission that competitors were blocking the use of its service and as a consequence, one local US ISP was investigated by the FCC. Jeffrey Citron, CEO of Vonage, has expressed his concern that there is no protection for the Internet against packet discrimination and his worry is that network operators may disrupt customers' access to certain services or hinder the quality of their customers' broadband to give priority to other applications (Pulver, 2005).

The US Congress is debating net neutrality as part of its reform of existing US telecommunications laws through the Communications, Consumers' Choice and Broadband Deployment Act 2006, which was considered by the Senate commerce committee in June 2006[17].

The interpretation of this situation is different in the UK. In its consultation document on the UK VoIP market, telecommunications regulator Ofcom (2006) acknowledged concern by VoIP service providers that their ability to provide a reliable service could be impinged by ISPs selectively degrading or blocking their VoIP traffic, but concluded that the competitive nature of the ISP market in the UK is likely to deter VoIP blocking, since customers finding they were suffering from packet degradation would be able to change supplier.

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[17] http://commerce.senate.gov/public/index.cfm?FuseAction=Hearings.Hearing&Hearing_ID=1778 [last accessed 14/09/06]

In other countries, some telecoms providers and ISPs are believed to be using VoIP-blocking software in order to protect their revenue by preventing free Internet-based traffic from running across their networks. In Saudi Arabia, for instance, national carrier Saudi Telecom is using Narus software to block VoIP calls.

The JANET voice strategy requirements analysis indicates that VoIP can operate successfully over IP networks as they are today on a 'best effort' basis. This is the level of service currently provided by JANET, where no guarantee is made about how a packet will be treated *en route* to its destination or even that it will get there at all. However, although the best effort' level of service can support VoIP, UKERNA found that this may be inappropriate for a centrally-supported and co-ordinated voice service on JANET, which would require a high level of voice call quality. The Internet by its nature is an uncertain environment and therefore the experience of a voice call can be mixed.

### 2.3.4 Directories and addressing

VoIP numbering systems are not standardised in the same way as telephone numbers on the PSTN, where numbers are allocated using the global E.164 numbering system[18] which provides globally unique, language-independent identifiers for resources on public telecommunications networks. VoIP identifiers, on the other hand, are based on IP addresses and there is no widely adopted standard for these identifiers similar to the E.164 scheme. There are different possible approaches to this issue and, at present, it is not clear which of the schemes is the most applicable, or how to create a centralised database of VoIP numbers or addresses. It is one of the issues that UKERNA is investigating through the remit of the JANET Voice Advisory Group.

One potential scheme is the allocation of an E.164 number that can be used for VoIP as well as for PSTN calls and in the UK, non-geographic VoIP numbers have been allocated a 056 prefix by the telecoms regulator Ofcom (Wearden, 2004). An alternative method is to map standard telephone numbers onto an IP environment. The IETF's ENUM group[19] is working on such a system and it describes itself as the bridge between the switched telephony network and the Internet[20]. To date, ENUM has got as far as the definition of an architecture and a protocol, RFC 3761, that translates E.164 numbers into IP addressing schemes. In the UK, Neustar and Nominet are both potential hosters of an ENUM service.

However, there are potential privacy problems with any globally unique identification numbering system, which have been openly acknowledged[21]. The convenience of using a single number to contact another person means that these systems are also of huge potential interest to marketers and spammers and would be a prime target for the data mining of personal contact information.

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[18] The international public telecommunication numbering plan, available online at: http://www.itu.int/rec/T-REC-E.164/en [last accessed 12/09/06].

[19] http://www.ietf.cnri.reston.va.us/internet-drafts/draft-ietf-enum-infrastructure-00.txt [last accessed 12/09/06].

[20] How ENUM works: http://www.eurescom.de/message/messageSep2004/ENUM_The_bridge_between_telephony_and_Internet.asp [last accessed 12/09/06].

[21] http://www.nominet.org.uk/tech/enum/ [last accessed 14/09/06].

On behalf of the UK H&FE community, UKERNA is following a three-pronged strategy of monitoring Ofcom's 056 numbering scheme, trialling a central ENUM register, and building on the JANET-developed Global Addressing Scheme, a dial plan, based on the E.164 standard, which was originally developed for Video over IP technology (UKERNA, 2006a).

# 3. Why people want it

*'Voice/data convergence based on IP telephony and VoIP will be under way in more than 95 percent of major companies by 2010. Convergence will drive additional classes of communications-enabled business applications and cause the greatest upheaval in the telecommunications industry since its inception.'*

Gartner, 12th May, 2005[22].

## 3.1 Network convergence

Recently, large numbers of organisations, public and private, have begun evaluating IP technologies as they believe that IP-based systems offer increased reliability and fault-tolerance. Forrester estimates that more than 50% of European enterprises are evaluating or piloting VoIP, IP PBXs and other services, and that within a few years around 75% of phone calls will made by over IP networks (Mohamed, 2006).

This is the first stage of the creation of the 'converged network' in which a single network replaces the current set-up of twin, separate networks of voice (PBX) and data (LAN). This provides some immediate benefits such as increased reliability and reduced total cost of ownership, as well as additional functionality such as portability — in which incoming phone calls are automatically routed to the VoIP phone, regardless of where it is connected on the network (also known as ‘find-me, follow-me’). In parallel with this development, networks are extending their reach through the use of wireless technologies such as WiFi and WiMax.

## 3.2 Application convergence

Organisations are carrying out network convergence in order to reduce costs and maintenance and to build capacity for a second stage of convergence. This second stage will see the convergence of different applications between, for example, voice and e-mail, Instant Messaging and video. The use of IP and the integration of voice and other data applications will allow VoIP to offer features that improve on those offered by traditional PBX systems.

### 3.2.1 VoIP and Instant Messaging (IM).

Instant Messaging is growing fast. Research firm Radicati Group forecasts there will be more than 1billion IM accounts in use by 2009. Kerner (2005) points out that a good number of these accounts will have direct access to VoIP within the same timeframe, which could have a major impact on both IM and VoIP use and developments. It has been possible to use voice within IM for some time, but many IM suppliers, including Microsoft and AOL, are now looking to enhance their existing IM software with improved VoIP capabilities, which Kerner says may well challenge existing VoIP providers such as Skype. A key IM standard is Jabber, an open protocol, based on XML, which used for both IM and presence services.

**Presence**

Traditional telephones provide some information about the status of the device that is being called: the call can be connected if there is a dialling tone, while an engaged tone means the call cannot go through. But presence services, which originated with the development of

[22] Available online at: http://www.gartner.com/DisplayDocument?doc_cd=125868 [last accessed 14/09/06].

Instant Messaging, provide far more information and, as distinct from traditional telephony, provide information about the status of the user, rather than the device (Vogiazou, 2002).

Presence allows a caller to see, in advance, who is available and the most appropriate way to contact them. Typical presence information, as used in IM, could include status messages such as ‘free for chat’, ‘away’, ‘do not disturb’ and ‘out to lunch’. It should be noted that this type of status can be set automatically – for instance, the ‘away’ status may come up if the user’s keyboard has not been used for a set period of time. Presence services are seen as part of a wider approach to unified communications; they enable a caller to check, for instance, whether the user’s mobile phone is on, or check their online calendar system to see whether they are in a meeting.

The IETF has an Instant Messaging and Presence Protocol (IMPP) working group, focusing on the application of SIP to IM and presence services[23]. The IETF has also worked on expanding the functionality of presence services, through definition of a Rich Presence Information Data (RPID) format, which would include information about what a person is doing, where they are located, and in what type of location, such as an office or school; and details of how best to communicate with the user (Schulzrinne, 2003). Some in the industry[24] believe presence is a key aspect of evolving voice services, because the ability to update information about a user’s availability means more intelligent call routing (and therefore efficient use of bandwidth) can take place.

### 3.2.2 VoIP and multimedia

Television and film are increasingly being digitised, which means they can be delivered as data through IP-based networks (e.g. IPTV). This combination of telephony, data and video is often referred to as ‘triple play’ and as voice traffic is also moved to IP-based networks there will be opportunities for merging voice applications with those based around other media.

**Persistence**

One of the attributes of SIP-based VoIP sessions is that they can be long-lived, or persistent, which enables enhanced multimedia communications involving voice, video and data without losing the original call (Ahuja, 2004). This has enabled the development of applications in which users interact through voice, video and data during multimedia conferences and exchange private and public messages. They can create and access stored data in a shared repository and users can drop in and out of calls. For instance, a voice session could be changed to a video session and a whiteboard could then be added.

These changes in session types can be automated, so that the SIP-based system checks if a remote endpoint supports a specific multimedia type. Users can switch between the most appropriate medium for particular communications, without losing the existing connection. This is powerful technology. SIP servers can be used for persistent multimedia conferences, such as a ‘push to talk’ audio conference, in which only one user at a time is permitted to talk, making it easy to keep track of what is being said, and by whom. The server can notify existing conference members when new members join while also managing the multimedia

[23] http://www.ietf.org/html.charters/simple-charter.html [last accessed 14/09/06].

[24] See www.sipcenter.com

session parameters (Kapoor, 2006). A persistent session can therefore provide the focus point for a long-term group project and also creates an environment for a series of chat sessions. In a SIP environment, SIP addresses will become increasingly important and it is possible that we will start to see these appearing on business cards and email footers in the near future (Dobbetsteijn, 2006).

### 3.2.3 VoIP and Web.

As a voice technology, VoIP is intimately linked to developments in automatic voice processing and recognition systems. Such systems work through a combination of telephony, speech recognition, speech synthesis and digitized audio and are becoming increasingly common in call centre activities, automated booking systems (e.g. cinema tickets) and information lines (such as weather reports or train times). A recent trend in such systems is a convergence with Web-based systems to enable what IEEE Computer magazine calls the 'Voice Web' (Srinivasan and Brown, 2002). A key technology in the development of this convergence is VoiceXML, designed to bring together such interactive voice response (IVR) applications and the Web. In short, it allows Web applications to be accessed via the telephone by providing a mark-up language for creating voice user interfaces that facilitate voice-based 'dialogues'[25] [26]. It allows the developer of a voice interface to have complete control of the spoken dialogue between the automatic system and the person calling on the telephone. The application or IVR prompts the user and, in turn, the user responds (verbally). At each stage the VoiceXML document outlines the actions to be performed, the responses expected and the automatic vocal prompts and responses[27].

A comparison with HTML and the Web helps. Instead of accessing information from a PC using a Web browser, such voice applications allow access from the telephone with a voice interpreter, which handles VoiceXML, running on the telephony server (VoiceXML Forum, 2006).

In a typical example, outlined by Danielsen (2000), a user wishes to find out the current price of a company on the stock market. He dials an automatic stock checking service and the call is routed to an IVR, which has a VoiceXML client. This platform translates the phone number dialled to a URI and the client places an HTTP request to the URI which responds with a file that contains VoiceXML instructions for conducting a dialogue with the caller. It makes use of speech recognition and/or touchtone for input, and pre-recorded audio and text-to-speech synthesis for output (VoiceXML Forum, 2006)[28].

This technology has uses in education with Kondratova and Goldfarb (2006) giving examples of access to learning materials for visually impaired learners, IVR systems for audio clip retrieval from archives based on a student's verbal request and improving mobile learning applications on mobile phones.

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[25] http://www.w3.org/TR/voicexml20/#dml1.1 [last accessed 14/09/06].

[26] *Voice-enabling your Website*, IBM Tutorial: http://www-128.ibm.com/developerworks/websphere/library/techarticles/0111_kemble/0111_kemble.html [last accessed 14/09/06].

[27] A tutorial at: http://www.w3.org/Voice/Guide/ provides further information. [last accessed 12/09/06].

[28] An introductory tutorial is available at: http://www.voicexml.org/tutorials/intro2.html [last accessed 12/09/06].

## 3.3 Wireless VoIP

Development of wireless VoIP has been rapid in the past few years. The main focus for development has been in the area of WiFi, which is based on the IEEE 802.11 wireless standard (Lozano-Gendreau et al., 2006), although IP-based telephony is also possible over other wireless systems, such as WiMax, based on 802.16 (Robinson, 2005). The main benefit, apart from replacing expensive mobile phone calls with cost-effective Internet-based calls, is that with wireless VoIP the user, once logged into the network, can access their usual phone extension, with all its usual functions and features, regardless of their actual location.

There are a number of ways to run wireless VoIP. A softphone (i.e. a *soft*ware tele*phone*) can be loaded onto a laptop and used in a wireless hotspot. This is already a well-established technology and softphones are also available for handheld computers and PDAs and some trials of mobile learning using VoIP with PDAs have taken place (Hogan, 2006). In addition, WiFi phones are now being developed which are similar to mobile phones, but use a wireless Internet connection. One recently released example is Netgear's SPH101 Skype WiFi phone[29] launched in January 2006. Dual-function handsets, able to switch between mobile networks and WiFi networks, are also being developed and are likely to be available by the end of 2006 (Rosmarin, 2006)[30]. This is a fast-growing area of wireless voice development, based on the interface and integration of mobile networks, including the existing GSM network, with 802.11 wireless phones. BT's Fusion project is another example in this area, offering 'intelligent' services that switch calls from the mobile network to a BT broadband network when the phone is being used at home.

There are still considerable technical challenges in implementing Voice over WiFi, including concerns about security, battery life in WiFi handsets and call quality. Wireless networks allocate bandwidth according to which devices are nearest to the WLAN access points, which can cause problems for voice call quality although some suppliers, such as Meru Networks, are developing systems that allocate bandwidth equally from the access points and can prioritise voice traffic, using QoS.

Several HE institutions have implemented wireless VoIP systems, including Loughborough College, which in 2001 installed a solution involving two wireless bridges, based on the 802.11b standard and providing sufficient bandwidth for 30 PCs and VoIP services (Wincott et al., 2005).

[29] http://www.netgear.com/products/details/SPH101.php [last accessed 14/09/06].
[30] The formal standard involved is: Generic Access (GA) to the A/Gb interface; Mobile GA interface layer 3 specification

## 4. VoIP in Further and Higher Education

Several UK F&HE institutions have already implemented VoIP and the development of the Skype peer-to-peer system has driven a high level of consumer – and therefore student – awareness of the technology. Many institutions are also either considering or already implementing wireless networks, and running VoIP services over wireless networking is a key focus for its further development. A number of institutions have begun to investigate the integration of VoIP within their existing telephone systems or even the replacement of existing PSTN voice systems with IP-based technologies, particularly on new-build campuses. One example is New College Durham, which has installed an integrated IP-based network to deliver video, audio and data services, and is in the process of migrating applications from an array of separate networks over to a single, integrated network (Telindus, 2005).

Brunel University provides an example of an HE institution that has implemented IP telephony on top of its existing data network. Brunel has implemented a VoIP system and now has 2,500 IP telephones on its network. It has also implemented an IP-based centralised messaging system, using voicemail and unified messaging software supplied by US manufacturer Cisco (Cisco, 2005).

Many JANET-connected organisations operate VoIP systems and one potential use of VoIP would be to provide inter-organisation VoIP connectivity across JANET, as a direct replacement for normal trunk telephony links. Providing a full-scale PSTN service is likely to be beyond the scope of VoIP on JANET, but it may be possible to provide interconnections with other voice telephony providers, using VoIP. This is being investigated by UKERNA.

UKERNA will also be running a trial of an IP conference bridge, likely to span both PSTN and VoIP users. Where possible, UKERNA intends to ensure that voice services are interoperable with the existing JANET videoconferencing service infrastructure. It has also undertaken development work to look at requirements for presence and instant messaging.

### 4.1 Collaborative and distance learning

E-learning or Web-based learning is a subject of considerable debate. Card et al. (2006) argue that podcasting and VoIP technologies offer the return of voice to distance learning, in both synchronous and asynchronous forms, and that this allows instructors and students greater social presence. It is argued that one of the benefits of social presence is that it allows online learners, through the use of highly responsive tools, to create themselves as ‘real’ and project their personal characteristics. Previously, argues Card, attempts to introduce voice into distance learning systems have failed due to both technical complexity and pedantic methodology and that podcasting and VoIP may be able to contribute to overcoming these hurdles.

Foreman (2003) says telephone-based conversations are highly effective and superior to both email and chat for organising small-team distance learning experiences and plays to a skill set we have all developed as members of a telephonic culture. But long-distance phone conferencing over the PSTN is expensive, whereas VoIP can provide a cost-effective way to include telephone-based conversations in the learning environment. Foreman’s students have used a shared application, MindManager, together with either phone conferencing or VoIP, from Groove. The students work online, with a shared view of MindManager maps and an audio connection to orient themselves to problem solving and task analysis and organisation.

Foreman recommends that an instructor wanting to learn more about synchronous learning work could start with Microsoft NetMeeting, a free software tool that provides Web-based VoIP and application sharing and is relatively easy to use. This can provide a foundation from which to move on to more complex team learning efforts.

There are a number of collaborative projects that use Internet-based voice. One example is FlashMeeting[31], a project of the Centre for New Media, part of the Open University's Knowledge Media Institute (Kmi). FlashMeeting uses VoIP technology to create an audio-visual instant messaging system, to support small groups of distance learners and provide an alternative to face-to-face tutorials. The aim of FlashMeeting and its sister project Hexagon, is to provide easy-to-use multimedia applications that support collaborative distance learning. Hexagon is more suitable for student support, or to enable faculty colleagues to keep in touch over specific projects.

Also in development, and now being used by a small number of universities, is the Marratech conferencing software, developed in Sweden by the company of the same time. It is mainly aimed at use by a small number of simultaneous users, is cross platform and is being trialled by the Monash University of Australia[32]

Students at Pepperdine University, California, used Skype as a communication device to facilitate peer-to-peer interaction to correct misunderstandings over text-based communication, particularly in situations where it was clear that the context was not being fully understood. They set up conferences for up to eight people and found the software to be a valuable tool and a catalyst for the promotion of the virtual 'real' personas of the doctoral students involved (Card et al., 2006).

### 4.2 New services for learners with disabilities

VoIP systems enable those with disabilities to access their message through voice, audio or a combination of both. Hearing-impaired people can place or receive calls from their computer without the need for a legacy TTY device. With VoIP, hearing-impaired learners or staff members can read their voicemail from their email program, in much less time than it takes with a TTY, while sight-impaired users can use IP-based phones to hear audible caller IT, missed-call log and line status. In addition, VoIP may, in the near future, provide an effective medium for incorporating audio, text and video, allowing the creation of integrated communication tools that feature speech, text, language translation, captioning, speech recognition and speech synthesis from text. In her review of three products with advanced accessibility features for inclusive distance learning, Schwartz (2004) says development of these types of products indicates the progress that is being made towards truly inclusive communication interfaces.

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[31] http://flashmeeting.open.ac.uk/index.html [last accessed 12/09/06].

[32] http://www.infotech.monash.edu.au/itsupport/marratech/#tips [last accessed 12/09/06].

### 4.3 Virtual/remote laboratories and telemedicine

VoIP technologies (based around the H.323 protocol) are being used in advanced conferencing applications within remote and virtual laboratories, in which researchers share a collaborative space although they are not all physically present (e.g. for e-science and telemedicine applications).

Such 'laboratories' have been facilitated by the development of high-bandwidth data networks, high quality display systems, sophisticated visualisation technologies, Internet Grid technologies and web-based video conferencing systems. They are a further example of the move towards the integration of voice, video, data and Web-based technologies[33].

[33] A full discussion of this is beyond the scope of this report, but interested readers are referred to Multi-Site Videoconferencing for the UK e-Science Programme: A Roadmap for the Future of Videoconferencing within e-Science (http://www.nesc.ac.uk/technical_papers/UKeS-2002-04.html)

## 5. Future developments

In the future, VoIP will increasingly integrate with business and education applications, to the point where it becomes essentially 'invisible'. This will result in a change in how we see the process of making a voice call – instead of only using voice to communicate with other people we will also be using voice to communicate with services. In order for this integration to take place, especially with business services, VoIP will have to integrate closely with applications built using Service-Oriented Architectures (SOA) (Ascierto, 2005).

### 5.1 VoIP and Identity

The service-based approach to ICT systems means that identity management will become increasingly important. In the service-based approach online tasks, such as access to content, are carried out on behalf of a user (a person with an identity). Increasingly, these services are spread across a number of entities (companies, libraries, educational institutions etc.) and identity management is being carried out through a decentralised architecture which manages and asserts the user's identity using different, often remote identity management services that operate using some level of agreement or trust arrangements (federated identity). The success of these identity management services is due, in large part, to the facility for advanced Single Sign On (SSO)[34].

As telecommunications, through VoIP, become closely integrated with other data and Web-based technologies the identity infrastructure associated with telecommunications will need to be absorbed. Work is ongoing in this regard through the ENUM system and work on SIP. In the US, the Internet2 consortium is experimenting with the convergence of communication identities within education. Their SIP.edu Working Group is looking at the use of email addresses for voice and multimedia communications[35].

### 5.2 Virtual universities

Over the years there has been discussion of the development of virtual universities (Twigg and Oblinger, 1996) in which services will be provided on-demand through the network, and the location, governance or ownership of the institution will be increasingly transparent. VoIP is one of the technologies that is likely to contribute to the development of a more virtual university environment.

Katz (2005) states that virtual universities and those able to blend the physical and the virtual will be better placed to handle increasing numbers of students. He says the ongoing improvements in technological capabilities, together with investment in network capacity and performance, may render existing debates about virtual, distance, hybrid and face-to-face education meaningless. He believes network-mediated learning opportunities will disrupt higher education’s tradition market segmentation, pricing and branding and that the focus of institutions, teachers and learners will shift, from where learning takes place, towards the achievement of social and educational outcomes.

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[34] Single Sign-On (SSO) –the user's ability to authenticate with one system entity and have that authentication honoured by other system entities (Shibboleth is an example of an SSO used within higher and further education).

[35] http://www.internet2.edu/sip.edu/ [last accessed 14/09/06].

Katz goes on to argue that VoIP is one of the technologies that are helping to blur the lines between real and virtual in the context of learning and scholarship, by making collaborative working environments increasingly ‘human’. He states that convergence, in the end, is less a technical exercise than a social one, promising technology-mediated collaboration and community. Katz cites Mark Clark, of the University of Manchester, who says the trend is towards compound documents that incorporate image, data, text and voice annotation. Email is likely to give way to increased use of collaborative working environments for document development, analysis, editing and even drafting.

However, the management, deployment and integration of converged technologies into a cohesive service environment and ultimately into a rich collaborative environment is likely to demand considerable attention in the future. There are important pedagogical issues for the production and assessment of the kind of compound documents cited above not least concerns about plagiarism, which is becoming an increasingly significant problem. It may even be necessary for the F&HE community to take a more pro-active approach and become more actively involved in the development of such technologies, in order to make sure that they are pedagogically 'fit for purpose'.

### 5.3 Pay-for services

Integrated triple play IP networks offer the potential capabilities for F&HE institutions to offer a much wider range of services to students. These could range from entertainment services, including digital broadcast television and video on demand, data services, learning assistance, distance learning and video lectures on demand. These converged services are moving into the mainstream: Google’s proposed GoogleTalk service, for instance, which is being beta tested, links IM contacts with VoIP contacts.

In 2005, San Jose State University implemented this type of triple play network. It has installed an integrated VoIP and video over IP network from US supplier IPlay3. One of its aims in implementing this network is to create a platform for future services such as distance learning and video calling (Iplay3, 2005).

## Conclusion

Many UK F&HE institutions are already looking closely at the possible use of VoIP within their overall telecoms and data networking infrastructures and policies. Although many of these institutions are developing their voice services in an independent manner, the role of UKERNA will be of increasing importance in the next few years.

Implementation of VoIP is growing rapidly, in both education and other sectors. While the initial impetus towards voice services over IP networks may have been driven by a desire to reduce costs, the move into IP-based services is now beginning to open up considerable potential within the learning and administrative environments of F&HE institutions. Some VoIP providers see this as a major opportunity for F&HE institutions to gain valuable revenue by selling services. Those within the institutions may be more interested in the potential of converged IP-based networks to enhance and extend the learning environment.

There will also be further convergence of media, data, voice and video. In order to achieve these further levels of convergence, interoperability and continued work on standards will be critical. SIP is emerging as the major protocol to underpin IP-based advanced services.

Another key driver will be the development of wireless VoIP, which will provide students and staff with different ways to access learning environments while integrated services will open up the possibilities of ubiquitous computing. In future, learning may take place in a shared virtual world of social computing, similar to an online, multiplayer game, and VoIP will be one of the components of such an environment.

However, it should not be forgotten that there are a number of technical issues that need to be considered in implementing VoIP. UKERNA's Voice Advisory Group is working on such aspects as directories and addressing, security and integration of VoIP with IP-based videoconferencing. Many F&HE institutions have already begun, or are considering, some form of VoIP implementation and may now want to look both at the work being done by UKERNA in this area and at the bigger picture of possibilities for the use of VoIP within the learning and teaching environment.

### About the author

Jane Dudman has been writing about technology and telecoms for more than 15 years. She is a former news editor of Computing and was subsequently editor of Communications Management magazine. She writes widely on technical topics for both specialist and more general publications and recently contributed an article on VoIP blocking to the Guardian's 'Technology' supplement. She also writes regularly about the uptake and development of ICT within the UK education sector for many publications, including Computer Weekly and Government Computing.

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