Commission for
**Communications Regulation**

Briefing Note

# Voice over Internet Protocol (VoIP)

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| **Document No:** | **03/21** |
| **Date:** | **18 February 2003** |

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# Contents

# 1 Foreword

This Briefing Note has been prepared as a follow-up to the 'Innovation in Communications – Planning for the Future' symposium held by the ODTR[1] in June 2002. Both the symposium and the Briefing Note series come under our Forward-looking Programme, and primarily deal with technology developments in the telecommunications sector.

The topic of this paper is Voice over Internet Protocol (VoIP). VoIP is a technique that allows telecommunications network operators to carry voice services using IP networks, including the Internet, instead of traditional circuit switched networks. The advantages of this include increased efficiencies in operational costs which arise as a result of the combination of voice and data onto a single network[2]. We expect these will be translated into lower costs for users of telephony services in Ireland. Furthermore, the use of IP technology enables further development of new value added voice services such as unified messaging (UM) and greater integration with Internet and e-Commerce applications.

Reliable VoIP systems are currently available and are beginning to be implemented in networks in Ireland and internationally. This technology is already being deployed in access networks (i.e. the 'last mile to customers) over, for example, DSL, cable and FWA, as well as in international and national core infrastructure. In Ireland VoIP has already been implemented in a number of private networks allowing companies to use their private data networks for voice services also.

As our current telecommunications networks evolve towards Next Generation Networks voice telephony will remain an important service and will evolve with the networks, both in terms of how it is delivered and ways in which it can be utilised. The more general move towards IP and Next Generation Networks will help to bring about more flexible networks offering users more choices. VoIP and other IP based technologies and services are developing rapidly, and the migration to IP based networks is likely to lead to many changes to the current telecommunications market. It is vital that as these changes take place we continue to promote and maintain competition in the market place.

**Etain Doyle**

**Chairperson,**

**Commission for Communications Regulation.**

[1] Office of the Director of Telecommunications Regulation – The ODTR was replaced by ComReg in December 2002.

[2] See our earlier Briefing Note on Next Generation Networks (NGNs) www.comreg.ie/_fileupload/publications/comreg0188.pdf

## 2 Comments on this Briefing Note

We welcome any comments or views on this Briefing Note and these should be sent to:

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to arrive on or before 5.30pm on Friday 28th March, 2003.

In submitting comments, respondents are requested to reference the relevant section of this document. Responses will be available for inspection by the public on request. Where elements of any response are deemed confidential, these should be clearly identified and placed in a separate annex to the main document.

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# 3 Introduction

This Briefing Note is intended to raise awareness of technology developments in the area of Voice over Internet Protocol (VoIP). It is targeted primarily at non-technical readers with some background and technical awareness of the telecommunications industry.

## 3.1 What is VoIP?

Using Internet Protocol (IP) technology and IP networks to deliver voice services is known as Voice over IP (VoIP). This is different from how the majority of voice services are currently delivered on operators' circuit switched networks (i.e. PSTN, ISDN)[3]. The use of IP technology equips operators with a flexible means to create and deliver new voice related services and applications in addition to basic voice. Also, from the network operator's perspective it can be more efficient to use IP to carry their telephony traffic than circuit switched technology, particularly when combined with their other data traffic. This increased efficiency results in lower operational costs which should translate into lower call costs for end users.

VoIP, also known as 'IP Telephony', refers to the general case of using IP technology to carry voice traffic[4]. A sub-set of this is 'Internet Telephony', which refers to voice calls on the public Internet. There are three basic ways that users could avail of VoIP:

**Phone-to-phone:** users' telephones are connected to an IP network through special equipment (known as gateways) that converts the telephony signals to IP and vice versa.

**PC-to-PC[5]:** users make voice calls through their PCs, typically over the public Internet. In this case each of the users must be running VoIP communications software to facilitate their call[6].

**Phone-to-PC:** A user's telephone is connected to an IP network such as the Internet through a gateway, enabling them to carry out a voice call with a PC user directly connected to the IP network.

See Annex 8.1 for diagrams illustrating these alternative scenarios.

## 3.2 Why VoIP?

There are two key reasons why VoIP could be preferable to traditional circuit switched voice:

- Its ability to support new voice related applications and services

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[3] In a circuit switched network a dedicated link is set up between the parties involved for the duration of a call, whereas IP networks (which are packet based networks) divide up the data into small pieces and send out each one individually to find their own way to their destination. This is analogous to sending a group of people between two towns on a train (circuit switched – dedicated link) or sending each one to the same destination in a different taxi via different routes (packet switched – no dedicated link or route). If popular direct routes become congested, packets will be diverted via circuitous routes, to the extent that communications originating and terminating in Ireland, for example, may be routed through other countries.

[4] VoIP is a type of Voice over Packet, which is a more generic term that includes other packet based technologies (see footnote 2 above) such as Asynchronous Transfer Mode (ATM) and Frame Relay (FR).

[5] In this context a PC refers to any device capable of running VoIP applications, including laptop computers, Personal Digital Assistants (PDA), and advanced mobile handsets.

[6] PC Internet Telephony software typically presents users with a virtual representation of a telephone on their computer screens. This is often known as a Soft Phone.

- Its potential to deliver cost savings for operators and users

### *3.2.1 New Applications and Services*

A wide range of new voice based applications are being developed using IP technologies. Some of these applications are described below.

**Unified Messaging, Instant messaging:** Unified messaging puts various different messaging formats into a single mail box (e.g. voice, fax, email, text). The different formats can then be accessed via computer, or telephone (using text to speech converters). This type of service is particularly useful to mobile users who can listen to their email messages while on the move. Another potential application could be where a user is alerted to an incoming voice call, but is unable to answer it (e.g. when in a meeting). They could type in a short message and have it instantly sent to the caller, as text or converted to speech, either postponing the conversation or advising them to use text based communications also. Instant messaging is where users, once they have discovered that they are on-line at the same time, can send messages to each other in real time. Typically these are text messages (e.g. email), but can also include voice services (e.g. AOL, MSN, Yahoo[7]).

**Voice Recognition/Interactive Voice Response (IVR):** With voice recognition software a user's verbal commands can be interpreted by a computer and used to carry out actions such as web browsing (hands free/eyes free). Voice Mark-up Language (VML) is a technique being used to make web style content accessible to voice users.

**Find-me-follow-me, Personal Virtual Assistant (PVA):** A personal virtual assistant can manage a user's calls so that they can be routed directly to the most convenient terminal (e.g. office phone, mobile phone, home phone, hotel phone). The PVA can track down which number to contact the user on at any given time. Other location based services could be integrated with voice services also (e.g. voice based directions for mobile users).

**Web-telephony integration:** The integration of telephony into web sites could enable users to select 'click to talk' functions on a web page, initiating a voice call to a customer support centre or sales representative for example. VoIP could also be integrated with web-based e-commerce applications.

**IP second voice line:** VoIP can be offered by ISPs as a low cost second line for voice calls. Users may be willing to accept occasional lapses in call quality for a low cost second voice line (see Section 4.1).

**PC conferencing with file sharing:** The integration of voice and Internet technology through VoIP, could for example allow voice users to transfer files to other called parties during a voice call, thereby enhancing the interactivity. Video conferencing facilities could also be added.

A key characteristic of these new user features and applications should be more intuitive and easier operation for users. Downloadable user profiles would enable a user to get the same features and functions on any IP phone as they have on their own phone by 'logging-on', in the same way as they can log-on to an Internet email service. It is likely that users will be encouraged to increase their use of basic telephony services as new features become available and are adopted.

It should be noted that although some of these applications could potentially be implemented using Advanced Intelligent Networks (AIN[8]) technology on circuit

[7] www.aim.com , http://messenger.msn.com/ , http://messenger.yahoo.com/

[8] AIN is a way for circuit switched network operators to create and deliver new services on their networks. AIN services can be interoperated with Internet services using protocols from the IETF such as PINT and SPIRITS.

switched networks, more flexible implementation and greater integration with Internet applications and services is possible using IP based solutions.  IP can potentially enable new service creation tools, with web-based interfaces, allowing operators to reduce the cost and time taken to deploy new functions and applications on their networks[9].  Furthermore, IP-based voice solutions may appear to be more 'future-proof' to operators or corporate customers when compared with upgrades to traditional voice solutions.

### *3.2.2 Cost Savings*

Telecommunications network operators typically operate separate networks to carry their voice traffic (circuit switched networks) and their data and Internet traffic (packet switch/IP networks).  Using VoIP technology, operators could also carry their voice traffic on their IP networks, reducing the operational costs associated with two distinct networks (see Section 5.1).  These combined networks are known as Next Generation Networks (see also Next Generation Networks Briefing Note[10]).  Furthermore, modern VoIP techniques allow greater degrees of compression of voice traffic than existing PSTN[11], thereby allowing transmission capacity to be used more efficiently (see section 5.1).  Similar cost savings can be made by operators of private corporate telecommunications networks, both in local area networks (LANs) and wide area networks (WANs).  Although operators would need to invest in new IP based equipment to avail of these efficiencies, migration strategies that enable them to continue using existing circuit switch infrastructure in conjunction with IP networks are generally employed[12].

Consumers could experience significant cost savings using Internet Telephony – between 30% and 50% according to an ITU study[13] – particularly in countries where there is little competition in the international voice market.  However, at present due to quality of service limitations on the Internet (see Section 4.1), Internet Telephony users may experience lower levels of quality than they might expect from a PSTN connection.

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[9] Open access to network interfaces (Application Programming Interfaces) such as OSA/Parlay and JAIN are important for encouraging new application and service creation from third party developers.  Open interfaces can also allow applications to be used across AIN and IP based networks.

[10] www.comreg.ie/_fileupload/publications/comreg0188.pdf

[11] Modern coding techniques that can be applied to VoIP signals can deliver speech at data rates as low as 2.4kbit/s (compared to 64kbit/s on the PSTN).  Although such techniques carry penalties in terms of sound quality, delay, and equipment complexity, more efficient transmission of good quality voice can be achieved.

[12] Some of these migration strategies involve consolidation of network infrastructure, reducing the number of elements in the network, thus reducing capital invest going forward (e.g. Telecom Italia).

[13] 'IP Telephony, Report by the group of experts on Internet Protocol (IP) Telephony', ITU-D, March 2002, p70

# 4 Current Limitations of VoIP

## 4.1 Quality of Service

Early implementations of VoIP often provided poorer quality voice services[14] than those typical of traditional circuit switched networks. VoIP over the public Internet or other IP networks not specifically adapted for good quality voice could suffer from delays (often referred to as latency) and general poor speech quality. For these reasons voice services could not always be guaranteed to achieve the minimum service levels stipulated for circuit switched networks.

Unlike many forms of data, voice services need to be delivered in real-time and therefore are not tolerant of the delays that can be a characteristic of IP networks. Unless special precautions are taken, an IP network can re-route data or simply discard packets of information when it experiences congestion, requiring them to be re-sent a short time later. If the discarded data segments are parts of a voice signal the users will experience poor quality speech[15].

A problem with VoIP potentially arises when calls must traverse more than one operators' network for completion. Whereas an operator can control the quality of a call on its own network, the operator may not have control over how the call will be treated on other operators' networks. Special service level agreements (SLAs) can however be developed between IP network operators to help address this problem.

Looking ahead however, it is likely that software development and other innovations will overcome such difficulties, and that packet based networks will increasingly be used to implement more choice for users with respect to quality of service. In some cases users may wish to avail of different levels of service for different situations. For example a high quality and highly reliable service could be selected for an important business call, or a lower or variable quality of service for an unimportant social call, with pricing varying according to the quality demanded. The concept of 'quality on demand' could potentially emerge as an important feature in future telecommunications networks. IP is also likely to help facilitate innovative and flexible billing solutions.

## 4.2 Security

In IP networks, much of the control is generally done close to the end users rather than at the centre of the network. Therefore, VoIP security must also, for the most part, be deployed in the end user equipment, and in the case of Internet telephony responsibility for security lies entirely with the end users. In the case of managed IP networks (i.e. an IP network that is controlled by a network operator) the operator can control the security protocols and standards used (e.g. the IPSec protocol[16]).

## 4.3 Overcoming VoIP Limitations

Where an IP network is within the control of a network operator – a managed IP network – the network can be physically scaled in such a way as to avoid the congestion that causes poor quality voice (i.e. add more capacity[17]). Alternatively,

[14] Quality in this context refers to general clarity of the voice signal and the continuation of a voice call until terminated by a user (i.e. not 'cut-off').

[15] Delay, Packet Loss, Jitter (packets take different routes and arrive out of sequence), and Echo are the main problems that can reduce quality in IP networks.

[16] The IPSec protocol is an extension of the IP protocol that can be added to provide security for VoIP. IPSec allows corporate users to avail of secure VoIP services over Virtual Private Networks (VPNs).

[17] IP was originally designed to operate with limited resources.

operators can add traffic management facilities that use techniques such as allowing capacity to be reserved[18] or for particular traffic to be prioritised in a network[19], ensuring that voice signals get preferential treatment. Another approach is for operators to create IP networks within their existing data networks[20]. Multi-Protocol Label Switching (MPLS) can also help to ensure good quality voice communications across IP networks.

Table 1 summarises some of the advantages and drawbacks of IP based versus circuit switched voice.

| | **Circuit Switched (PSTN)** | **Internet Protocol (IP)** |
|---|---|---|
| **Quality of Service** | Dedicated connections guarantee a level of service during a call – if sufficient capacity is not available the call cannot not be initiated | Need extra capacity, or additional features to guarantee quality (i.e. equipment with quality-management support)<br><br>Other levels of service may also be selectable – e.g. low cost lower quality voice |
| **Efficiency (cost)** | Separate networks for voice and data<br><br>A fixed amount of capacity is reserved for each voice call. | Combined network for voice and data – lower operational costs<br><br>Packet-based services (e.g. IP) can be readily combined<br><br>Increased compression of voice – less capacity is needed |
| **Security** | Controlled in the network by the operator | Controlled by operators in managed public and private networks<br><br>Controlled locally by users in Internet telephony |
| **Features** | New features can be added using Advanced Intelligent Network (AIN) technology | New features can be added more quickly and directly by the end-user using web based interfaces<br><br>More flexible implementation of new features<br><br>Greater integration with Internet applications |

Table 1. – Comparison of circuit switched and IP based voice services.

[18] e.g. Resource Reservation Protocol (RSVP), Integrated Services (IntServ) – see Annex 8.2

[19] e.g. Differentiated Services (DiffServ) – see Annex 8.2

[20] This means that IP packets of data are carried by the ATM or Frame Relay networks. This is analogous to a car-ferry transporting cars across the sea i.e. one type of vehicle transporting another over unsuitable terrain.

# 5 Implementation of VoIP

It is already evident that the migration of network operators to VoIP and all-IP networks in general, will not occur overnight. Despite the potential operational cost savings associated with VoIP, many operators have made substantial investments in their existing circuit switched voice networks, from which they generally seek to leverage as much value as they can. Therefore many early implementations of VoIP are operated in conjunction with circuit switched networks, indicating that change will occur in incrementally. For example, operators investing in new network equipment will choose elements that are compatible with both their mature circuit switched networks and their developing IP networks. During the migration to IP networks many of the traffic management aspects and features of circuit switched networks are likely to be maintained, albeit in packet switched forms (e.g. ATM, MPLS).

## 5.1 Implementation Issues

VoIP is capable of being implemented by operators of public, private, corporate, national, international, fixed and mobile networks alike.

**Core Network –** For many larger and incumbent operators initial implementations of VoIP technology will be in the high capacity core sections of their networks used to link large towns and cities[21]. In this portion of the network operational cost saving is the main driver as operators can gain from combining their data and voice traffic onto a single IP network core[22]. For example, Telecom Italia recently announced that it is moving its voice traffic onto an IP network for core national transmission[23].

**Access Networks[24] –** Closer to the end users, public network operators may choose to implement VoIP to save on operational costs, but they may also do it to enable the deployment of new voice services such as web-telephony integration (see Section 3.2.1), with a view to creating new revenue generating opportunities. In the access network VoIP is emerging over DSL (VoDSL)[25], over cable (VoCable), and over fixed wireless access (FWA)[26] – see Table 2. Network operators can choose how far they want to push their VoIP into their access networks. For example a DSL service provider could use IP for voice traffic from the core network as far as their DSLAMs, delivering normal PSTN voice over the local loop. They could also send VoIP as far as a gateway on the customer premises (i.e. a Residential Gateway/Integrated Access Device), or they could send it all the way to IP telephones. In the case of PC users, VoIP software and a speaker and microphone would be needed. Telephone users may need an IP-phone or an IP adapter for their existing telephones.

**Corporate networks –** Corporate users are proving to be early adopters of VoIP technology. They can benefit from the operational cost savings of combining their

[21] Core network links can be considered as equivalent to motorways in a national road network – i.e. carrying large volumes of traffic between central areas.

[22] According to a recent ITU report 'IP Telephony Report by the Group of Experts on IP Telephony', March 2002, VoIP can save between 40 - 60% in transmission costs through greater compression.

[23] 'Telecom Italia leads Europe in VoIP push', CWI Online, October 2002.

[24] This is the section of a network that links users to the telecommunications operator (i.e. the cables coming into a user's home).

[25] VoDSL can be delivered in a loop emulation configuration, or in an end-to-end voice over IP configuration. See also Future Digital Subscriber Line (DSL) Technology – Briefing Note : www.comreg.ie/_fileupload/publications/comreg0301.pdf

[26] VoIP is also available on some WLAN products and would be well-suited to Optical Access networks – see WLAN and Optical Access Briefing Notes: www.comreg.ie/_fileupload/publications/comreg0216.pdf , www.comreg.ie/_fileupload/publications/comreg0229.pdf .

voice and data networks, both in individual offices and between offices. The additional features enabled by VoIP (see Section 3.2.1) and potential future proofing are also attractive to corporate users. In some cases the PBX (private branch exchange) or LAN/WAN provider may provide a migration path to VoIP, minimising the investment in new equipment and the associated disruption caused (e.g. telephone handsets may not need to be replaced). An IP PBX (iPBX) is a PBX that uses VoIP. The more the trend towards VoIP in corporate networks grows, the stronger is the case for a move to wholesale IP in public networks, to link the corporate 'islands' of VoIP without the need for special gateways.

**Mobile Networks –**IP has been chosen by the 3GPP[27] as a core technology of future third generation mobile networks. This means that VoIP will be used at least in the core segments of these networks before eventually migrating onto user handsets[28]. Mobile networks could potentially be migrated to VoIP more quickly than fixed line networks as they are typically configured with fewer local exchanges (i.e. Mobile switching Centres, Serving GPRS Nodes), resulting in less equipment to be upgraded[29]. A number of cordless telephone systems utilise VoIP.

**Internet –** New Internet applications are likely to continue to emerge that integrate voice and web-based applications (e.g. Instant Messaging, Unified Messaging). Growth in the use of the public Internet for large volumes of voice traffic is likely to lead to demand for extra bandwidth to avoid congestion, which would otherwise reduce overall quality of service (for voice and data).

**Intranet/LAN/WAN –** Intra office communications was one of the first places that VoIP was deployed. In this case companies can save on IT costs by combining their voice and data traffic onto a single IP network.

**Numbering –** the problem of connecting traditional telephone users with ordinary telephone numbers (know as E.164 numbers) to VoIP users with IP addresses can be approached in different ways. In some cases a gateway (see Section 3.1) may convert between IP addresses and E.164 numbers. ENUM is a way to convert between E.164 telephone numbers and URIs (universal resource identifiers) developed by the IETF and supported by the ITU[30]. Several European countries have consulted on ENUM and it is currently being investigated or trialled in some cases (http://www.eto.dk/numbering/enum.htm ). Comreg will address ENUM in a forthcoming paper.

An important issue related to numbering in a VoIP environment is the inevitable global shortage of IP addresses. An updated version of the IP addressing scheme used, known as IPv6, could eliminate this problem if widely implemented (see IPv6 Briefing Note[31]).

**Billing Systems –** When connected to traditional PSTN networks, VoIP systems may need to support the same level of information that is available in call detail records (CDRs) for billing purposes (i.e. PSTN operators will want to settle interconnection accounts based on CDRs). Solutions to this problem are being developed by an industry initiative called IPDR.org[32] (IP detail records), and early solutions are currently available in some VoIP products. Traditional operators may also require VoIP

[27] http://www.3gpp.org –A collaborative body including ETSI (European Telecommunications Standards Institute)

[28] Some mobile devices that currently support WLANs as well as mobile standards may be able to deliver VoIP over the WLAN section.

[29] Probe Research – Voice over packet markets: Wireless voice over packet – When, where and why?, March 2002.

[30] The ENUM equivalent of the E.164 number '+353 1 8049600' might be '0.0.6.9.4.0.8.1.3.5.3.enum.arpa' . Further details are available at http://www.itu.int/ITU-T/worksem/enum/index.html

[31] www.comreg.ie/_fileupload/publications/comreg0263.pdf

[32] www.ipdr.org

systems to determine whether a call originated on its own network or on another network (on-net/off-net), to allow charging to be applied accordingly.

## 5.2 VoIP in Ireland

VoIP is being increasingly deployed in Ireland. International IP telephony services are available to users in Ireland through services such as ePhone[33]. Major VoIP wholesalers such as ITXC and iBasis have gateways installed in Ireland to pass locally dialled calls onto international IP networks. Probe Research[34] estimated that approximately 3% of Irish incoming and outgoing international voice traffic was carried over IP in 2002. eircom began using the Net2Phone VoIP service for some international traffic in June 2000[35].

A number of private networks are being developed in Ireland that employ VoIP technology. The National Software Centre in Cork has deployed a VoIP network to serve over 1500 users[36]. VoIP technology is also being deployed for communications in the LUAS transport network in Dublin[37].

## 5.3 International VoIP Implementations

### *5.3.1 Implementations of VoIP*

VoIP is being implemented in telecommunications networks worldwide, both in countries with developed and developing telecommunications markets. In Europe VoIP has been implemented by operators in the core (e.g. Telecom Italia) and access networks (see Table 2).

| Country | Operator | Access |
|---|---|---|
| Belgium | Telenet | Cable |
| Finland | Sonera | Cable |
| | Sonera Entrum | DSL |
| Germany | Callahan | Cable |
| | ISH (NRW) | Cable |
| | KPN | DSL |
| | QSC AG | DSL |
| | PrimaCom | Cable |
| Iceland | IMC WorldCell | Wireless |
| Netherlands | UPC Nederland | Cable |
| | PrimaCom | Cable |
| | Versatel | DSL |
| Spain | ONO | Cable |
| UK | Telewest | Cable |

Table 2: Some European operators who are offering or trialling VoIP services[38] (Source: Probe Research).

[33] www.ePhone.ie

[34] www.proberesearch.com

[35] http://web.net2phone.com/about/press/releases/06_27_2000.asp

[36] Provided by Nortel - http://www.nortelnetworks.com/corporate/global/emea/ireland/nsc.html

[37] Provided by Cable and Wireless - http://www.techcentral.ie/techcentral/corporate_it/it_in_action/luas_goes_online_with_voip.xml

[38] Source: Probe Research

In Europe most VoIP deployments in access networks (see Section 5.1) are on cable networks, whereas in the US VoIP is mainly provided on DSL. Over the last two years a number of European cable companies (see Table 2 above) with a large number of existing voice subscribers have announced that they are up-grading their networks to handle VoIP along with broadband Internet and digital TV offerings (voice, video, Internet/data – 'triple play')[39]. Delays in standardisation for cable access networks, and some financial difficulties for the companies involved, have helped to slow the roll-out of VoIP on cable networks. The German cable operator ISH currently has 2000 VoIP customers, and plans on expanding this in the future once their economic situation improves. Typically, as with other direct access operators, cable operators would migrate to VoIP in the core sections of their networks before the access segments.

In China[40] (including Hong Kong) there has been widespread deployment of VoIP. Other countries with advanced broadband access such as Singapore and South Korea also have successful IP networks.

### *5.3.2 International Operators*

International telecommunications operators such as BT, Cable & Wireless, Equant, Genuity, and WorldCom have all employed VoIP in their international networks. In some cases VoIP is used on these types of networks for transit between different countries (e.g. to carry VoIP traffic between public or private voice operators), and in other cases it is offered as a service to corporate customers, typically as part of an IP Virtual Private Network (IP VPN).

## 5.4 VoIP Equipment Manufacturers

Major telecommunications equipment manufacturers such as Alcatel, Avaya, Ericsson, Lucent, Nokia, Nortel, and Siemens are continuing to develop VoIP products for telecommunications operators offering both public and private voice services and for corporate markets seeking internal office solutions (i.e. iPBX). Earlier products were typically focused on one of the main VoIP signalling protocols (i.e. ITU H.323, or IETF SIP) and manufacturers often developed their own proprietary standards or proprietary versions of existing standards. However, current products can typically handle multiple different protocols. VoIP products have been and are being developed to allow operators to continue to use existing equipment installed in their networks, to facilitate a more gradual migration to fully VoIP networks with a view to phasing investment.

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[39] Callahan - http://www.nortelnetworks.com/corporate/news/newsreleases/2001c/09_18_01_callahan.html

ISH - http://www.juniper.net/company/presscenter/pr/2002/pr-020123.html

PrimaCom - http://www.tellabs.com/news/02news/nr031402b.shtml

Telenet - http://www.telenet.be/bedrijfsinfo/persberichten/archief/publicatie_17sept2001.php

[40] e.g. China Unicom. See also ITU, *"Summary: IP Telephony in Practice"*

# 6 Market Development and Regulatory Issues

Network operators, public and private, are typically attracted to adopting VoIP on their networks for two main reasons: operational cost savings, and new revenue opportunities through value added voice services. However, there are numerous other factors (e.g. economic, commercial, regulatory), which vary from operator to operator and from county to country that may affect their decisions as to where and when to adopt VoIP technology.

## 6.1 Types of Service Provider and their Motivation

For traditional telecommunications operators and indirect operators (e.g. Carrier Pre-Select operators[41]) the main motivation for adopting VoIP technology is typically potential savings in operational costs. Although large operators may have the most to gain in cost savings from VoIP, the economics of VoIP are often quite complicated. For example it may not be easy for operators to cut operational costs in other areas to fund investment in new VoIP technology. Furthermore, according to an ITU report (March 2002[42]) the cost of IP gateway equipment was four to five times higher than equivalent PSTN local exchange equipment. Technology developments and economies of scale are however likely to reduce significantly the absolute and relative cost of VoIP equipment over time.

Internet Telephony Service Providers (ITSPs) are typically driven by the prospects of capturing market from incumbent circuit switched telephony providers, and from other ISPs through the added feature of low cost voice services.

For some operators that provide international telephony services, particularly those operating from developing countries, VoIP can provide a way to bypass international settlement costs. According to a recent ITU report, in such countries it may however be detrimental to prohibit VoIP since national users would be subjected to higher cost telephony for outgoing calls, and incoming VoIP calls –which cannot be detected and therefore will be received – simply reduce the potential from international settlements[43],[44].

Further opportunities are likely to arise for telecommunications software developers in Ireland with the continuing proliferation of IP based telecommunications services[45].

## 6.2 International Regulatory Approaches

The European Commission has stated that Internet Telephony should not –at least for the moment[46] - be subject to regulation as it is not a substitute for voice telephony, arguing that the quality of service over the public Internet is not presently sufficiently good for voice services. Internet Telephony is therefore considered to be just another Internet application[47]. The new EU regulatory framework[48] is market based rather

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[41] For more information on CPS see www.comreg.ie – documents comreg0241, comreg0247, comreg0264, and comreg0307.

[42] 'IP Telephony, report by the group of experts on Internet Protocol (IP) Telephony / ITU-D'

[43] ITU, 'Summary: IP Telephony in practice'

[44] The new EU regulatory framework could effect this situation – see Section 6.3.1

[45] See the Communications Management Software forum – www.ida.ie .

[46] The European Commission is keeping this under review.

[47] EC consultation - http://europa.eu.int/comm/competition/liberalization/telecom/voice_over_internet/consultation/

[48] http://europa.eu.int/information_society/topics/telecoms/regulatory/new_rf/index_en.htm

than service based and aims to be technology neutral. Therefore any regulations that might apply to VoIP may stem from the markets that it is deemed to reside in.

Generally speaking European regulators have taken the approach to treat VoIP in the same way as PSTN voice (i.e. a technology neutral approach), so long as the service is a potential substitute for existing voice services (i.e. good quality, and it operates in real-time – not the Internet). The situation in Ireland is consistent with this. Variations in this approach to VoIP regulation are outlined in Annex 8.3. In other parts of the world VoIP has been fully permitted (e.g. China, Thailand, Hong Kong, Singapore, South Korea).

It is widely considered that VoIP can assist with the introduction of liberalisation and the fostering of competition. Although VoIP may be deemed a substitute for traditional voice communications in some cases, it is likely to be capable of providing additional functionality beyond what is available on traditional voice systems (see Section 3.2.1).

## 6.3 Potential Regulatory Issues

### *6.3.1 Interconnection*

Interconnection of IP networks is typically carried out on a bi-lateral basis without regulation (peering[49]), and is normally non-transparent. However, the new EU regulatory framework for interconnection (the Access Directive) extends to include the interconnection of all electronic communications networks including IP.

| **Interconnect** | **Circuit Switched Charging** | **IP/Internet Charging** |
|---|---|---|
| Terminating traffic | Charge per minute | No charge between IP networks of the same size<br><br>Large ISPs charge small ISPs on the basis of capacity of interconnect link |
| Transit traffic | Charge per minute | On basis of capacity of interconnect link |
| Interconnect link | Cost sharing on causation basis | Smaller IP networks usually bear the cost of the link in full |

Table 3: Typical interconnect charging approaches – circuit switched networks vs the Internet (source ITU/Ovum)

IP interconnection is usually carried out – directly or indirectly - with the large global ISP backbone providers (e.g. Worldcom, Cable and Wireless) and not national incumbents as is the case for circuit switched networks.

The adoption of VoIP technology instead of circuit switched technology may result in changes in the internal cost structures of telephony providers. This could require national regulators to revise their interconnect regulations which depend on such cost figures (see www.comreg.ie/_fileupload/publications/ComReg0316.pdf ).

### *6.3.2 Tariffs*

The notion of distance based costing may not be relevant in IP networks where traffic between two points may follow different and often circuitous routes. IP based networks are widely viewed as being more suited to volume based or subscription/flat-rate style tariffs. Such schemes are significantly different from how most telephony services are currently charged and regulated (see Table 3 above). According to an ITU

[49] Peering describes the mutual interconnection of IP networks, typically without charge, e.g. the Internet Neutral EXchange (INEX – www.inex.ie ).

study[50] (March 2002), the cost of IP telephony was typically between 30% and 50% less than PSTN for end users.  The reduced costs could potentially stimulate demand and usage, thus averting any potential losses in revenue.

[50] 'IP Telephony, report by the group of experts on Internet Protocol (IP) Telephony / ITU-D', March 2002.

# 7 Conclusion

Using IP technology to carry voice communications can potentially yield cost savings for existing services and help with the introduction of new services and applications. Operational cost savings can be achieved through more efficient operation of telecommunications networks, by combining voice and data networks into a single network. These operational cost savings should ultimately translate into reduced costs for end users. New voice based applications enabled by VoIP technology could create new telecommunications services for users and new revenue generating opportunities for network operators. VoIP will also enable greater integration between voice services and the Internet, which should bring about even more developments in new applications and services.

Migration of voice services onto IP technology is unlikely to occur overnight. Existing operators have made significant investments in digital circuit switched technology over the past 20 years and will probably look to extract value from these networks. This is particularly true in the current market climate where operators are seeking to cut costs through reduced capital investment. It is likely that VoIP will operate in tandem with circuit switched telecommunications networks for some years to come.

The widespread use of VoIP could potentially alter the way in which network operators provide voice services and the way in which they interoperate with one another. Changing pricing structures and interconnection arrangements will undoubtedly create new opportunities for operators as well as disrupt established business models. So the extent to which operators can extract value from circuit switched networks may be constrained to some extent by the levels of competition from IP-focused players in the marketplace.

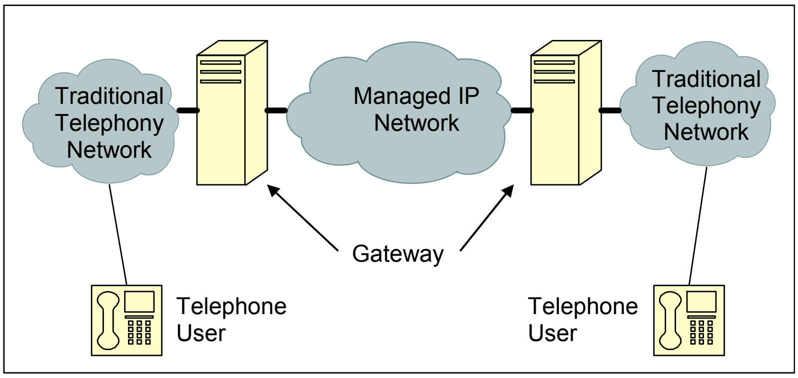
VoIP is currently being deployed in networks in Ireland and around the globe. Whilst it is becoming an established technology in core national and international telecommunications networks, reliable solutions are now also available for public access networks and we are beginning to see deployment over technologies such as DSL, Cable and FWA. VoIP is likely to be an important part of the increasing reliance of telecommunications services on IP based systems and the move towards Next Generation Networks.

# 8 Annexes

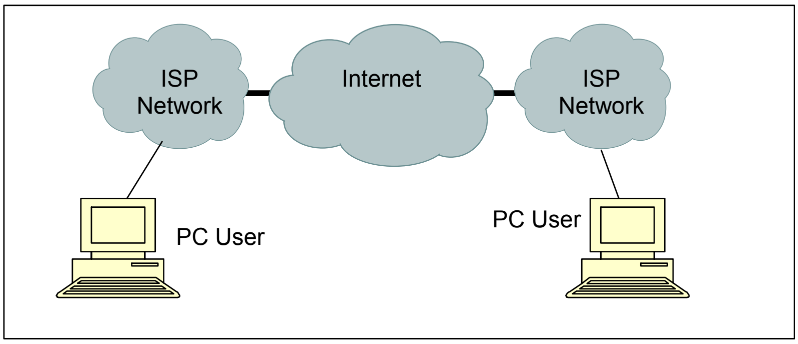
## 8.1 IP Telephony Types

Below are diagrams illustrating typical examples of the three main types of VoIP as described in Section 3.1.

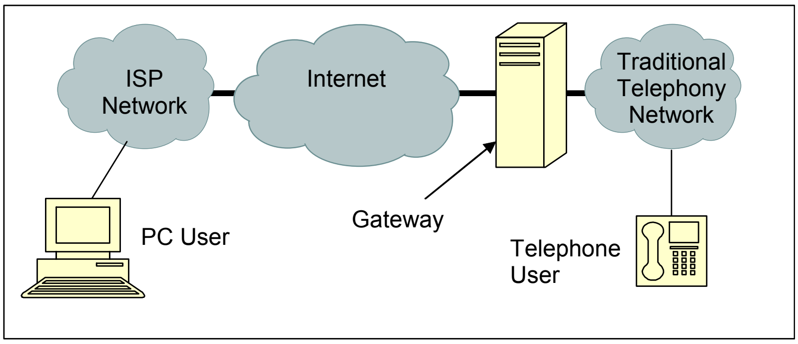
### *8.1.1 Phone to Phone*




### *8.1.2 PC to PC*




### *8.1.3 PC to Phone*



## 8.2 VoIP Related Standards and Protocols

To ensure interoperation between PSTN networks and IP networks technical standardisation is needed. There are two main standards for VoIP signalling between end users and telecommunications networks; IETF - Session Initiation Protocol (SIP) and ITU H.323. H.323 emerged first, but SIP has since gained popularity due primarily to its Internet style approach. SIP is suitable for services such as web services. ITU T H.248 is a VoIP signalling protocol used between points in an IP network e.g. a softswitch and a gateway node (see Annex 8.4). Proprietary standards and proprietary versions of standards also exist. Brief descriptions of some key protocols are listed below.

### *8.2.1 DiffServ*

DiffServ (Differentiated Services) is an IETF protocol that allows different types of traffic to be treated differently in congested situations. In the case of congestion packets marked with lower priorities will be discarded first. IP routers must be DiffServ enabled for the service to work.

### *8.2.2 RSVP*

Resource Reservation Protocol (RSVP) is a way of achieving quality of service on IP networks by requesting that certain packets are prioritised above others in individual routers.

### *8.2.3 SIP*

Session Initiation Protocol is a protocol from the IETF designed to handle voice and other multi-media services over IP networks. It operates on a client-server model and enables voice and multimedia calls to be initiated, modified and terminated.

### *8.2.4 H.323*

H.323 is an ITU standard that can handle voice and multi-media calls on IP networks.

### *8.2.5 H.248*

H.248 is an ITU standard for signalling between elements on IP networks such as softswitches and gateways. Megaco is another older protocol.

### *8.2.6 IntServ*

An IETF protocol used to help implement quality of service on IP networks by reserving resources for individual traffic flows.

### *8.2.7 MPLS*

MPLS allows different traffic types to be prioritised, and for virtual paths to be set up during voice calls ensuring that quality is maintained. MPLS can work across different operators’ networks once their IP routers have been MPLS enabled (in some cases a software up-grade).

## 8.3 European Regulatory Approaches to VoIP

| | |
|---|---|
| Belgium | Contractual agreement between ISPA Belgium and Telecommunications and Justice Ministries |
| France | Public consultation. Technology neutral approach |
| Czech Republic | Most forms of Internet Telephony are permitted |
| UK | VoIP operators must be licensed if it is a substitute for public voice telephony |
| Hungary | Licence required for use of IP for voice. Sound quality limitations imposed to differentiate VoIP |
| Germany | No explicit regulations on Internet telephony – not considered to be a voice telephony service |
| Malta | Public consultation. VoIP must be licensed where it is a public telephony service |
| Poland | Provision of VoIP is liberalised |
| Russia | Licence required for service provision |
| Switzerland | Permitted based on a case-by-case basis, once it has been determined as public telephony |

Table 3: Some European regulatory approaches to VoIP (Source: Yankee Group)

## 8.4 VoIP Network Elements

Some of the main VoIP network elements are described below.

### *8.4.1 Gateway*

A gateway is a device that enables voice traffic to pass between two different types of network (e.g. IP to circuit switched).

### *8.4.2 Softswitch*

A softswitch allows for the separation of call processing functions from physical switching functions. Physical switching is carried out by a media gateway and call control is carried out by a media gateway controller.

### *8.4.3 IP PBX (iPBX)*

A private branch exchange (PBX) for VoIP. This allows users to utilise their Local Area Network – previously only used for data – for voice communications also.

### *8.4.4 SIP Server*

A SIP Service is a device that carries out SIP communications with other SIP enabled devices (clients), such as controlling a VoIP call.