VOICE OVER IP
a techno-regulatory view

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Voice over Internet Protocol (VoIP) is changing the way we communicate. Trefor Davies and Louise Lancaster examine how VoIP has become a mainstream proposition – to the extent that most users will have no idea that they are using it. There remain, however, a number of techno-regulatory challenges to overcome, which are affecting the evolution of Voice over IP and that communications future.

Voice over Internet Protocol (VoIP) began around 20 years ago as a rudimentary PC-to-PC solution. It was cool to be able to talk to friends using the internet. The experience wasn't great but that didn't matter. The technology was developing and by the late 1990s many companies had jumped on the bandwagon, partly out of fear that if they didn't they would be left behind in their communications markets. There were still many questions unanswered and few products and

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services that people would be willing to pay for.

Now, in 2010, the world has changed completely. VoIP technology has advanced to become a mainstream proposition to the extent that most users will have no idea that they are using it. It is part of our daily lives. This article drills into a number of VoIP subject areas that either have currency today or will have in the near future.

These include:

- The evolution of an old circuit switched feature set to one that is designed for Internet Protocol (IP);
- Quality of service (QoS);
- The regulation of VoIP services, in particular with respect to support for calls to emergency services;
- Number porting;
- Naked DSL (Digital Subscriber Loop);
- VoIP over next generation access.

THE EVOLUTION OF A CIRCUIT-SWITCHED FEATURE SET TO ONE THAT IS DESIGNED FOR INTERNET PROTOCOL

Back in the heyday of the Private Branch Exchange (PBX) – yes it had a heyday – there was a gold rush to sell phone systems with increasing numbers of features to businesses that, up until then, had used phones just to make and receive phone calls.

The highly competitive industry developed what became a common feature set that was satisfied by different manufacturers with different methods. Each vendor developed its own signalling protocol as its own competitive advantage and in doing so tied in its growing list of customers to a specific range of handsets, "servers" and peripherals.

This was not the world of standardization aside, perhaps, from interfaces such as Q.931[1] and Qsig[2]. Although a competitive market, it was actually quite a comfortable space for established vendors with existing customer bases locked in to their proprietary technologies.

As the end of the 20th Century approached a cloud called VoIP appeared on the horizon. VoIP was a threat to the natural order. It had disruptive potential. In particular VoIP brought with it standards such as H.323[3], Media Gateway Control Protocol (MGCP)[4] and Session Initiation Protocol (SIP)[5]. If standards became adopted then the comfortable existence of PBX vendors would end. Their customers would, in this brave new world, be able to select products, handsets for example, from different vendors. They could buy on value and choice rather than be hamstrung by a lack of alternatives.

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Most telephony vendors invested heavily in VoIP. Those that didn't became early casualties of the shakeout. These investors did have a problem though. Not only would VoIP, in an open standards-based world, expose their customers to the competition, but the standards available were not mature or functional enough to be able to replace their existing signalling protocols which had, by then, benefited from fifteen or twenty years of development.

What's more, which standard would be the one with longevity? MGCP was closest in architecture to existing protocols but had several flavours. Every MGCP-based service provider had their own proprietary extensions to make up for deficiencies in the standard. So a handset vendor would have to support several different types of MGCP and that at a time when it was not clear that any of the MGCP-based service providers would stay the pace. There was a similar story for H.323 which was likely to be replaced by SIP which, as a multimedia protocol, did things very differently from the old fashioned "master/slave" model of the existing order. SIP was so new that it definitely didn't cut the mustard when trying to replace the 400 or so features of the mainstream business PBX.

So the VoIP investments that were made sought exactly to replicate the features and functionality of existing equipment. The selling proposition was that removing the need for a separate time division multiplexing phone network in an organisation would in itself be a sufficient driver for customers to use VoIP. It was also maintained that customers liked their existing systems and would prefer

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to stay with a familiar environment. Same buttons to press for call transfer, last number redial, etc etc.

What we ended up with was a plethora of VoIP systems that were identical in all aspects to the old digital PBXs but with different network connections. This was, to the global community of VoIP pioneers, a big disappointment. A big opportunity to change the way people communicated was being missed and, moreover, the market still forced customers down the "sole source" locked-in route.

It has taken some time but this is beginning to change. The maturing of the VoIP market, perhaps characterised by the sales of VoIP systems overtaking old circuit-switched products, happened years ago. Today people think nothing of investing in VoIP. The old concerns over QoS and reliability are no more. In fact the old order of PBX vendors is also changing as more consolidation occurs and SIP-based solutions entering the market at the low end become established and push up into larger system sizes. You can now mix and match handsets with systems, and hosted sites with premises-based VoIP services.

This is all made possible by standards, and SIP in particular, which has now emerged as the protocol of choice. However, before we talk about how the SIP standards have evolved to make this new mix and match world possible it is worth taking a look at how the way we approach communications has changed.

VOIP FEATURE SETS ARE CHANGING

Time was when business people had the best in communications tools at their finger tips. Their employers would invest in systems that attempted to give them a competitive advantage, such as telephone systems, mobile phones, laptops, software and indeed intranets. Consumers would gaze in through the expensive plate glass windows of business and envy their toys.

The world has moved on. Your average consumer these days has access to all the tools that a business has. The average teenager probably has a communications capability far in advance of all but perhaps the most sophisticated few corporations.

It started with simple on-line facilities such as Instant Messaging and has moved on to video telephony, conferencing, collaboration and sharing, data storage and cloud computing. So much so that businesses are actually moving towards using the same services provided by the likes of Google and Skype.

VoIP vendors have also latched on to the fact that the functionality provided by notionally free consumer services is relatively simple to employ in their own systems. SIP, the very protocol that was avoided in the early days due to its perceived lack of functionality, is what underpins many consumer plays.

SIP does change the way people communicate. Instead of calling someone and finding their phone engaged, or reaching their voicemail, you don't call because you can see that they are on the phone due to their "presence" status. An Instant Message might be sent instead asking the person to call you when they are off the phone. In the old days you would leave a message or keep calling back to find that the other person was still on the phone. Very inefficient and also frustrating.

So VoIP systems that originally began life as proprietary sole-source products and which have eventually moved under market pressure to SIP are now beginning to integrate the advanced multimedia capabilities of SIP into telephony platforms. SIP video phones are becoming commonplace, with advanced functionality that actually emulates small PBX systems which, in practice, removes the need for such systems. As bandwidth becomes ever cheaper, video will become more prevalent, as will the use of high definition codecs which, in turn, will be an indication that we have left the old way behind for good.

As is the nature of the beast, the SIP standard has continued to develop and evolve in order to meet the practical needs of the communicating world. Although the development of such standards is conducted by the IETF, a good summary of the current activity can be seen at the SIP Forum website: http://www.sipforum.org/content/view/19/72/

QUALITY OF SERVICE

QoS has always been perceived as one of the potential problems of using VoIP. Whilst in reality few people suffer from QoS problems, coming to the UK market later this year is BT's "Real Time QoS" product. Although other service providers have offered a flavour of QoS over broadband in the past, the BT Real Time QoS service will in practice be the first commercially available service that employs prioritising technology as opposed to simply making sure that the pipes don't become congested. QoS might not be of interest to your everyday surfer. It does however have the potential to revolutionise the user experience when using the internet with better control of voice and gaming quality.

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QoS is a hot topic, bringing the 21CN network finally into the 21st Century and driving interest in the service from a variety of sources – including businesses wanting to use VoIP and communications providers wanting to sell consumers low cost additional home telephone lines.

WHAT IS QoS?

QoS in a network is usually the term used to describe the process whereby certain types of network traffic are prioritised above others. QoS is typically required in a network where time-critical applications are being supported. In most cases this means VoIP or video but can be applied to financial transactions and gaming (to improve the experience). There isn't a definitive list. By and large if you design a network with enough capacity to accommodate your bandwidth needs you don't have to implement QoS.

In practical terms, where an Asymmetric Digital Subscriber Line is concerned, this usually means providing a dedicated broadband connection for sensitive applications such as VoIP so that, in this case, voice traffic doesn't have to compete with email and other traffic. Even if a shared connection is used, typically a router can control the up-link QoS, at least so that sending emails doesn't interfere with the voice service.

However, it is not possible to control the quality at the local exchange. Traditionally, traffic at the Digital Subscriber Line Access Multiplexer (or Multi-Service Access Node in BT's 21st Century Network (21CN) parlance) is on a best efforts basis. There is rarely a problem for VoIP but quality is not guaranteed and this does, to some extent, take the shine off the "new world" network offering.

BT's 21CN has been designed to support QoS and it is expected that this service improvement will be productised and offered to customers this year. QoS is really the icing on the cake from the networking perspective. It brings the 21CN network finally into the 21st Century. Although we don't know how much it is going to cost yet one would expect there to be significant interest in the service from a variety of sources.

These include businesses wanting to use VoIP, communications providers wanting to sell consumers low cost additional home telephone lines and people wanting to improve the experience when playing interactive on-line games.

HOW DOES IT WORK?

Although BT is not the only game in town when it comes to network offerings in the UK it will, however, be the first to make available true QoS as a generic technology, as opposed to using QoS to manage its own voice services. The BT network essentially supports a standard communications protocol called Diff Serv. Diff Serv allows network providers to "Tag" their traffic to tell the network what priority to give this traffic. In a multimedia VoIP network the highest priority will typically be given to the signalling packets. These are the messages that tell the voice packets where to go. Voice will then have second priority and video lowest. Priorities could be different in different networks depending on the type of service most important to particular users of the network. QoS is going to play an important part for all of us in moving the infrastructure of UK plc truly into the 21st Century.

THE REGULATION OF VOIP SERVICES AND THE PROVISION OF ACCESS TO EMERGENCY SERVICES

As VoIP services have moved into the mainstream, the regulatory authorities have been keen to ensure that consumers of those services are protected. The prevailing view in the noughties was that anything which looked like a regular phone service should perform all the functions that consumers had come to expect from a phone service. Chief among UK regulator Ofcom's concerns was that users should be able to access the emergency services.

Ofcom recognised that there was strong public interest in widespread reliable access to emergency services, so they did not want disproportionate regulatory burdens to discourage providers from offering 999 access. So in 2004 Ofcom announced [1] that, as an interim measure, VoIP providers would be allowed to offer access to emergency services without having to meet all the regulatory obligations that go with being a provider of Publicly Available Telephony Services ("PATS"). PATS is defined by the following criteria:

- A service available to the public;
- For originating and receiving national and international calls;
- Through a number or numbers in a national or international telephone numbering plan;
- Providing access to emergency services.

However, the European Commission subsequently made it clear that, where an operator meets all four criteria of PATS, it automatically becomes a PATS service, with all the additional regulatory obligations that come with that. Ofcom therefore published a statement in 2006 [2] which withdrew its policy of "interim forbearance". VoIP providers still had the choice of whether or not to provide access to emergency services but, if they did, they would have to comply with all the General Conditions which apply to PATS operators, including General Condition 3, which imposes an obligation on the provider to maintain the "proper and effective functioning of the Public Telephone Network provided by it at fixed locations at all times". At first sight, this is a daunting prospect for a small VoIP provider.

With providers able to elect not to support 999 calls, Ofcom was still concerned about the possibility of consumer harm where a service that appeared to be a normal telephone service could not to offer access to emergency services. So in March 2007 it published a statement [3] which introduced a new code of practice [4] requiring service providers to notify consumers at the point of sale and in user guides of any potential limitations in the service as regards reliability (for example if the service would not function in the event of a power failure), access to emergency services or inability to port numbers.

This statement also included guidelines on the application of PATS obligations to VoIP service providers. This was intended to help VoIP providers to comply with requirements about the reliability of calls to 999 / 112 and the requirement to provide caller location information for use by the emergency services.

Ofcom recognised that there was strong public interest in widespread access to emergency services that was reliable and included location information

Concerns remained, however, that VoIP providers could choose not to offer 999 access so, in December 2007, Ofcom published further regulations which made it compulsory for certain types of voice service, namely those that enable the user to make calls to the PSTN, to provide access to emergency services [5]. This means that some VoIP services which would not otherwise qualify as PATS (because, for example, they offer outbound but not inbound calling) are still required to provide 999 access. Only PC to PC (internet) services, "click to call" and inbound-only services are exempted.

Location information must be provided "where technically feasible". In practice, this means that "nomadic" VoIP services, namely any service where the customer logs on from a location other than their registered address, do not have to provide the actual location of the caller. Instead, VoIP calls present to the emergency call handling centres with a "flag", which prompts the operator to ask the caller for their location. Meanwhile, work has begun to enable the Calling Line Identity (CLI) to link with the IP address to provide the caller's physical location, which would benefit both the emergency service providers and authorities working on crime prevention and detection.

VoIP services which are effectively indistinguishable from PSTN services are now regulated in the same way as other electronic communications services.

NUMBER PORTING

In 2004 Ofcom made both geographic number ranges and a new 056 number range available for VoIP services. Whilst the 056 number range proved unpopular, the availability of geographic numbers has made it easier for VoIP providers to compete against mainstream service providers. VoIP providers are, however, hampered by the deficiencies in the UK's system for number portability. The method adopted in the UK, known as "onward routing", requires that a call to the customer's new service provider must first transit the network of the original number range holder. This entails bilateral porting agreements between all communication providers who are potential "donors" and "recipients", creates routing inefficiencies and acts as a barrier to competition for new entrants. It can lead to problems when range holders go out of business and it means that many service providers are, in reality, unable to offer full number portability to their customers.

Many communications providers are pressing for the introduction of a more recipient-led porting process with direct routing from the call originator to the current service provider, achieved by the automatic interrogation of an "all call query" database. The recently amended Universal Service Directive requires that the telephone numbers of customers in the European Union should be able to be ported within one working day. Without a major overhaul of porting processes in the UK, this target is unachievable.

NAKED DSL

DSL, as readers will know, is provided using an analogue telephone line as a bearer. Many broadband lines are now provisioned exclusively for use as a VoIP carrier, particularly for businesses. The analogue line rental, however, carries a portion of cost to cover the plain old telephony voice service that such a line would normally be used for. Most Internet telephony service providers would prefer to offer data-only broadband lines that do not include the voice element of the analogue line bearer. This is partly because it could be slightly cheaper to provide, and partly because provisioning of broadband only, rather than broadband plus voice line rental (two separate products), would be less complicated.

Such services, known somewhat alluringly as "Naked DSL", are available in some countries but not in the UK. Here, despite requests from the Internet telephony service provider community, the incumbent BT has pushed back, maintaining that such services are not economic to provide. Clearly it is not in BT's interest to provide a service that could allow competitors to erode its traditional voice business. Ofcom has not seen the need to intervene as it tends to favour bundled service provision in the mass market.

Tying broadband provision to line rental has other unforeseen consequences. When a customer on a PSTN line ports their number to a VoIP provider, the line should remain in place with the service provider from whom the customer rents the line. However, BT processes do not currently allow for this. Instead, when the number is ported, the line is ceased because BT currently associates the line rental with the CLI. To compound the problem, when the voice line rental is ceased this, in turn, disconnects the broadband service, so that the VoIP service can't function either. With the advent of IP services, BT needs to adopt a new way of thinking about service provision.

VOICE OVER NEXT GENERATION ACCESS

Now that network operators are rolling out Fibre-to-the-Cabinet and Fibre-to-the-Premises, the big question is how voice services will be provided on those networks. . Although Fibre-to-the-Cabinet retains a copper connection to the customer premise, it is unlikely to be sold in an unbundled form. So if the wholesale products are going to change (for example to Ethernet services), then this makes it possible for a new, data-only wholesale product to be offered.

Similarly with Fibre-to-the-Premises, it makes sense for the provision of network access by the network operator to the service provider to be completely service agnostic. The wholesale product can be raw, so the service provider can choose what to use it for. This means that traditional retail models, such as those based on wholesale line rental, could be forced to evolve. It means that the cost of the maintenance of the fibre connection (a utility) could be split amongst multiple services, such as voice, broadband, TV content, and healthcare.

With the demise of copper and its concomitant line powering, questions remain over consumer protection and whether or not battery back-up should be a regulatory requirement for voice services on fibre. Ofcom's position at the moment remains that if consumers are notified at the point of sale and in marketing literature of the fact that service could be disrupted in the event of a power failure, then battery back-up is not necessarily required. Over time, as Digital Enhanced Cordless Telecommunications (DECT) phones become the norm and fibre becomes commonplace, it is likely that consumers will no longer expect their phones to work in a power cut.

CONCLUSIONS

VoIP has come a long way over the last few years to the point that most users these days will be unaware that their telephone conversation is actually being conducted over a packet network. Regulation of services using the technology is an endorsement of its mainstream nature. VoIP is also changing the way we communicate and the UK's telecommunications infrastructure is moving inexorably towards becoming one exclusively based on VoIP. There are some challenges that prevent the wholesale migration away from the PSTN and to VoIP, not the least being fiscal. However the technological challenges have mostly been overcome and the stage is set for a rich communications future.

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ABBREVIATIONS

21CN	21st Century Network (BT)	PATS	Publicly Available Telephony Services
CLI	Calling Line Identity	PBX	Private Branch Exchange
DSL	Digital Subscriber Loop	QoS	Quality of Service
IETF	Internet Engineering Task Force	Qsig	Q-Signalling protocol (European Computer Manufacturers Association)
IP	Internet Protocol	SIP	Session Initiation Protocol
ISDN	Integrated Services Digital Network	VoIP	Voice over Internet Protocol
MGCP	Media Gateway Control Protocol		

FOOTNOTES

1 Q.931 is the signalling protocol recommended by the ITU for signalling between a terminal (including a PBX) and the ISDN.

2 QSIG is the signalling protocol specified by the European Computer Manufacturers Association and adopted by ETSI for signalling between PBXs. It is largely based on Q.931 but can support more advanced features required by large private networks.

3 H.323 is a signalling protocol recommended by the ITU for controlling real-time multimedia (eg video conferencing) applications over packet-based networks. It is based on Q.931 and has a call model similar to that of ISDN. It can therefore ease the introduction of VoIP into existing networks of ISDN-based PBX systems including the transition to IP-based PBXs.

4 MGCP is a signalling protocol defined by the Internet Engineering Task Force (IETF) for controlling the media gateway between an IP-based network and a traditional PSTN/ISDN. It enables PSTN/ISDN calls to be transported over an IP network. The call control, which signals to the media gateway, is generally centralised, for example in a soft switch.

5 SIP is a signalling protocol defined by the IETF for controlling multimedia (e.g. voice, video) communication sessions end-to-end over IP networks. It is a text-based protocol familiar to those who develop web-based applications enabling the communication sessions to be integrated with other web applications.

ABOUT THE AUTHORS

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He is co-founder and CTO of Timico, deemed one of the fastest growing technology companies in the UK for three years running (Sunday Times Techtrack100). In 2009 the company was recognised by Deloittes as the seventh fastest growing tech company (based on five years compound annual growth). Tref now serves on the Industrial Panel of the University of Wales School of Informatics. He is also on the council of the UK Internet Telephony Service Providers Association (ITSPA) and the Internet Services Providers' Association (ISPA). He writes an independent blog at www.trefor.net

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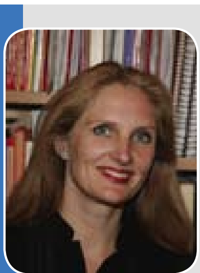

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