The Key to Successful Deployment of VoIP?

If you're thinking of deploying voice over IP (VoIP) you might find it a bit more difficult and time consuming than you think, we believe the secret is to plan in as much detail as possible to avoid disappointment. This paper is designed to provide an overview of the considerations we should take into account before ripping out our telephone system and embarking on VoIP.

Examine your motives for going VoIP!

There is a good deal of hype in the industry claiming now is the time for VoIP, and its probably true that while it was the kiss of death for so many companies who had developed VoIP products over the last decade, now VoIP seems to be 'coming of age'. Certainly the hype is helped by vendors struggling to maintain revenue streams, and trying to push customers into making significant investments in new equipment. We continually see the trade press, stress how big VoIP is, but before we jump in headfirst we have to decide what benefits if any VoIP will bring our organisation.

We have seen a well-known vendor trying to persuade a customer with 3,000 plus internal extensions to replace their PBX and all their handsets with new IP telephony equipment. There was no business reason to rip out low cost analogue telephones, and a sophisticated PBX to replace them with a very expensive IP telephony system, when 90% of the traffic runs over the organisations existing internal wiring system. The only cost savings would be to the 100 extensions in a remote building a few miles away. However salesmen need to generate revenue for their companies and to earn their commission so they pushed the technology angle rather than what benefits the customer may gain, as it happened the customer was smart enough to know there was no justification at all for ripping everything out and starting again. This customer did however purchase two low cost Case Communications Viper, VoIP routers, which allowed the existing systems at each site to use IP to communicate over an IP link.

What is the difference between IP Telephony and VoIP?

IP telephony – means replacing your conventional PBX and handsets with an 'IP PBX' (usually software on a PBX on the network) and IP phones. Instead of having a dedicated phone on each desk, software on a PC can allow the user to make calls over the data network using IP.

To most organisations a telephone is even more vital than their PCs, and while users are happy to re-boot their PCs following a crash would they be as happy to loose their phones if they had a lock up or crash? Telephones are very low cost and highly reliable, and can be plugged in and forgotten for many years. Security issues have been in the press over the last year with IP Telephony systems being hacked and brought down, but we could also say tapping a standard phone may be just as easy.
Other considerations are the high cost of handset; while these are coming down in price they are still more expensive than standard analogue handsets.

Justifying IP telephony – if we are to spend a significant amount of money on a new IP telephony system we must look to see where we will recoup this investment, and the usual answer is from savings made over the cost conventional telephone bills. With the cost of calls coming down radically then this could be quite difficult to justify.

Voice Over IP – Is a generic term meaning transmitting voice over the Internet protocol. If you're happy with your existing PBX and handsets you can still make the same savings that you would have made with IP telephony by adding gateways or VoIP enabled routers in front of your existing PBXs.

This means the PBX believes it talking to the same Network, but in fact the router converts the voice to IP, send sit over the network and then out it back to a conventional format. This is the most cost effective way of implementing VoIP, and it gives you the same savings as a full IP telephony system, when it comes to reducing phone bills.

It's as easy as plugging in telephones?

Many of us are used to simply plugging in telephones at home, picking the handset up and making a call, one of the easiest pieces of equipment you could hope to install. However professional Voice over IP equipment is not as easy as that to install, and a number of factors need to be considered before venturing down the road of VoIP. The most important thing in the deployment is in careful planning, and pre-project homework.

Our own network or a third party network?

If we already own the network that we want to run the voice over, we have control of that network and must ensure the equipment is suitable to support our voice traffic. The sections below give an idea as to what we need to check to make sure our network is going to be capable of supporting voice traffic. We will expand on this further down.

If we are going to use a Carrier or third party network then we have to get a guarantee from the carrier that they will provide us with a Service Level Agreement. Their hardware should be capable of working with our equipment and of identifying voice packets as taking priority, in fact they would not be much use if they could not guarantee that, however if we are going into the network using Broadband, few vendors support QOS over broadband, as last months 'Newsletter' highlighted, even BT has postponed their QOS over broadband due to problems with their routers.

Derbyshire based 'Node4' are one of the few ISPs who can guarantee QOS over broadband. After you enter their IP network you stay on it and don't pass onto the Internet so they can guarantee QOS end to end.

MPLS in some ways reflects the days of Frame Relay and a Committed Information Rate (CIR), but for IP, and this does ensure trunks don't get over subscribed, that transit delays through the routers are minimal, (large delays give rise to echoes) and can give us additional confidence our chosen carrier can support our QOS requirements. ATM backbones can also do the same.

Which type of VoIP do you want to use – SIP or H.323?

Although many other VoIP signalling protocols exist, SIP is characterized by its proponents, as having roots in the IP community rather than the telecom industry. SIP has been standardized and governed primarily by the IETF (Internet Engineering Task Force) while the H.323 VoIP protocol has been traditionally more associated with the ITU. However, the two organizations have endorsed both protocols in some fashion.

SIP (Session Initiation Protocol) is a signalling protocol used to create, manage and terminate sessions in an IP based network. A session could be a simple two-way telephone call or it could be a collaborative multi-media conference session. This makes it possible to implement services like voice-enriched e-commerce, web page click-to-dial or Instant Messaging with buddy lists in an IP based environment. SIP has been the choice for services related to Voice over IP (VoIP) in the recent past. It is a standard (RFC 3261) put forward by Internet Engineering Task Force (IETF). SIP is still growing and being modified to take into account all relevant features as the technology expands and evolves. But it should be noted that the job of SIP is limited to only the set-up of sessions. The details of the data exchange within a session e.g. the encoding or codec related to an audio/video media is not controlled by SIP and is taken care of by other protocols.

Voice Encapsulation

With traditional telephone networks your home phone is usually an analogue device, but once it gets into the telephone network and is transmitted over the carriers backbone, its transmitted using a technology called PCM (Pulse Code Modulation) which samples the analogue voice and converts it to a digital stream, taking 64,000 bps of bandwidth. PCM is the standard by which all other voice technologies are compared, and to date none of the compressions algorithms have managed to achieve the same levels of quality as PCM. For voice over IP we mainly use G.729 as it is quite highly compressed and has low bandwidth requirement. Standard G.729 operates at 8 Kbit/s, but there are extensions, which provide also 6.4 Kbit/s and 11.8 Kbit/s rates for marginally worse and better speech quality respectively. Also very common is G.729a which is compatible with G.729, but requires less computation. This lower complexity is not free since speech quality is marginally worsened. One thing to bear in mind are the IP overheads to be taken into account with VoIP, typically we might expect an 8K calls to take around 11-12 Kbps. If your only running voice over your network, bear in mind Time Division Multiplexing technology managed to get reasonable quality speech down to 4800 bps, so you would be better off with TDM rather than IP. The table below provides details of the various

algorithms and their required bandwidth. The first Bandwidth column is without any compression or silence suppression. The second bandwidth column is with header compression only and the third bandwidth column is using header compression and silence suppression.
The following assumptions are made

- The IP/User Datagram Protocol (UDP)/RTP headers are 40 bytes.
- RTP header compression reduces the IP/UDP/RTP headers to 2 or 4 bytes.
- Multilink Point-to-Point Protocol (MLPPP) or Frame Relay Forum (FRF).12 adds 6 bytes of layer 2 header.

Compression Technique Codec Bit Rate Kbps	Payload Size Bytes	Bandwidth MLPPP or FRF.12 Kbps	Bandwidth with header compression MLPPP or FRF.12 Kbps	Bandwidth with header compression and silence suppression MLPPP or FRF.12 Kbps
G.711 (64)	240	76	66	50
G.711 (64)	160 (Default)	83	68	54
G.726 (32)	120	44	34	29
G.726 (32)	80 (Default)	50	35	33
G.726 (24)	80	38	27	25
G.726 (24)	60 (Default)	42	27	27
G.728 (16)	80	25	18	17
G.728 (16)	40 (Default)	35	19	23
G.729 (8)	40	17.2	9.6	11.2
G.729 (8)	20 (Default)	26.4	11.2	17.2
G.723.1 (6.3)	48	12.3	7.4	8.0
G.723.1 (6.3)	24 (Default)	18.4	8.4	12.0
G.723.1 (5.3)	40	11.4	6.4	7.4
G.723.1 (5.3)	20 (Default)	17.5	7.4	11.4

PBX Signalling

Digital PBXs -Where existing PBXs use digital trunks (E1 or ISDN for example) we must consider the signalling, systems used by those PBXs to communicate with each other. Some of the more sophisticated PBXs use timeslot 16 to communicate with multiple other PBXs and this is unlikely to be supported within any IP Routers that are being used to convert the PBX E1 trunk to IP.

However what is possible to provide virtual paths between PBXs each one having its own E1 port into the IP router, which then carries the sophisticated PBX signalling untouched within timeslot 16, to the PBX at the remote end.

This is one of the simplest and lowest cost ways to use VoIP, as well as providing the greatest cost saving, allowing us to remove an expensive 2 Mbps circuit.

PBXs with ISDN interfaces – For PBXs with basic and primary rate ISDN Voice Over IP Routers such as the Case Communications Vipers, can emulate Primary Rate and Basic Rate ISDN, taking in ISDN calls, and converting them to IP before transmitting them over the network, and putting them back as they were or into another format. The Router recognises the incoming telephone number (or routes on part of the number) and and converts it to an IP address, making an IP call over the network to the target destination. Therefore a call plan must be established to match dialled numbers coming in, to destination IP addresses.

Analogue PBXs – Some of the older PBXs use analogue signalling, like E&M or AC15, and it's also possible to transport these over IP and to convert them into another signalling system at the remote end of the link.

AC15 is one of the more difficult systems to work with, as there are variances in different vendors interpretation of the system. Here the only way to ensure success is to try the PBX on the network and if it fails to work, it maybe necessary to buy in converters to translate AC15 into another signalling system such as E&M.

Connecting handsets – If we are attaching traditional telephones to remote branch offices then we need a call plan, where we recognise traditional PSTN (Public Switched Telephone Network) numbers and convert them into IP addresses, or if they are local numbers, possibly pass them out of the router into the local PSTN. Or we may simply want long line extensions where a user picks up their handset and gets dial tone from the remote PBX, this is one of the easiest ways to implement VoIP for handsets, as the extensions are simply mapped port to port.

Ensuring your network is ready for VoIP.

If you have an existing IP network then its going to be necessary to check the equipment in your network to make sure its capable of supporting VoIP. The following sections are designed to be a top-level checklist to ensure your network is ready to accommodate voice.

Quality of Service

In packet-switched networks, the traffic engineering term Quality of Service (QoS) refers to the probability of the telecommunication network meeting a given traffic contract, or in many cases is used informally to refer to the probability of a packet succeeding in passing between two points in the network. Having determined that your calls are to be converted to IP and that they will be passed over your network, it's necessary to ensure all the routers in the network are capable of recognising that packets containing 'Voice' (and voice signalling) are to take priority.

The currently accepted approach to Quality of Service is "DiffServ" or differentiated services. In the Diffserve model, packets are marked according to the type of service they need. In response to these markings, routers and switches use various queuing strategies to tailor performance to requirements.

At the IP layer, differentiated services code point (DSCP) markings use the 6 bits in the IP packet header. At the MAC layer, VLAN IEEE 802.1p and IEEE 802.1D can be used to carry essentially the same information.

Routers supporting 'Diffserve' use multiple queues for packets awaiting transmission from wide area interfaces. Router vendors provide different capabilities for configuring this behaviour, to include the number of queues supported, the relative priorities of queues, and bandwidth reserved for each queue. In practice, when a packet must be forwarded from an interface with queuing, packets requiring low jitter (e.g. VoIP or VTC) are given priority over packets in other queues. Typically, some bandwidth is allocated by default to network control packets (e.g., ICMP and routing protocols), while best effort traffic might simply be given whatever bandwidth is left over. Additional mechanisms may be used to further engineer performance, to include:

Queuing

- Fair queuing
- First in first out (FIFO)
- Weighted round robin (WRR)
- Class-based weighted fair queuing
- Weighted fair queuing
- Buffer tuning

Congestion avoidance

- RED, WRED – Lessens the possibility of port queue buffer tail-drops and this lowers the likelihood of TCP global synchronization
- Policing and Traffic shaping

Identifying your routers capabilities

Identifying whether or not your router supports these service may not be that straight forward. Some vendors have multiple versions of software and an enhanced version might be required to handle the necessary QOS, therefore some detective work maybe required to see if the operating system in your products has the necessary QOS facilities. If not then an upgrade may be necessary, which in turn could require more FLASH or RAM. On a Cisco router the command 'show ver' will show you the routers IOS.

Packet Fragmentation

Another consideration is the fragmentation of packets when running voice over IP. Under normal circumstances the network carries all sorts of traffic from large files to interactive traffic. A problem can occur when a voice packet arrives at a router just after a large file transfer has arrived. The voice packet does not want to wait for the file to be passed through the router, so its necessary for the router to 'fragment' the file (this means break one large frame into several smaller frames) so the voice can be slipped in between the file transfer frames. Luckily most routers automatically fragment files when they detect voice packets. However it now means a previously efficient link where large packets shared small overheads has now become a less efficient link, with large numbers of small packets mingled in with voice, so does the bandwidth need increasing, or do the routers need upgrading?

Is your existing network equipment powerful enough?

When one of the world major router vendors was asked to speak publicly about the low performance of his company's products, he simply said 'Our customers are not interested in performance'. However today with Voice and streaming video, router performance is far more of an issue and well know brand names will not be enough to keep poorly performing products in place. As voice packets require the routers to fragment the packets on the link, the number of packets passing through a router are going to be much greater and performance is far more important than in the traditional data only networks.

Pilot Network

Once the true cost of migrating to VoIP has been established its important to issue an RFI or and to run a Pilot network to not only sort out equipment configuration but to identify problems and resolve them before you go to far. Routers with insufficient power may create echoes within the network, or lock up, analogue PBX' using mature signalling systems, such as AC 15 may well not be able to set up calls or two different vendors PBXs may both support a common signalling system such as Q.931, but due to differences in the specification fail to communicate, with each other As we witnessed between a few years ago trying to get a well known German PBX vendors product to talk to a well known American PBX, both claimed to be Q.931 compliant, but they failed to communicate, because of their different interpretations of Q.931.

Summary

While we see VoIP everywhere and are being told it's the panacea but before we jump headlong into this world we really ought to examine our motives for doing so, and to make rational business decisions not emotive decisions based on a desire to run with new technology. Certainly the carriers can save a fortune by integrating their networks into a single platform, but we need our own reasons to invest in VoIP.

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