Case Communications

Voice and data integration

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Welcome,

Welcome to the Case Communications March Newsletter. This month we look at implementing Voip, BT and its Global MPLS network and Cisco moving away from Windows to Linux for its Call manager.

The Key to Successful Deployment of VoIP?

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If your thinking of deploying Voice over IP 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 overview is designed to provide an overview of the considerations we should take into account before ripping out our telephone system and embarking on VOIP.

[More]

BT to Integrate Voice into its Global MPLS Network

As part of its 21CN, or 21st century network, British Telecomm plan’s to completely replace its current time division-multiplexing (TDM) infrastructure with a MPLS network spanning more than 30 countries.
[More]

Lets go for a SPIN?

In the 1960’s, fictional character Reginald Moleshusband shot to fame as the star of a Public Information film featuring bad parking. Poor Reggie struggled to manoeuvre his Austin 1100 into any available space.
[More]

Cisco sways away from Windows with Linux based 'Call Manager'

New PBX software will be appliance and open source-based
Cisco is switching its PBX software to Linux with the latest release of Call Manager.
[More]

Case Communications win contract to supply high-speed routers.

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Cisco moves to SIP

Last week Cisco announced it was to embrace the Session Iniation Protocol (SIP) into its IP telephony products, making it one of the last of the mainstream vendors to adopt the technology
[More]

An overview of Wimax

Case Communications development director Roger has been sticking his nose into Wimax, technology, and ever eager to impart knowledge has has provided us with the following overview.
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The Key to Successful Deployment of VoIP?

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If your thinking of deploying Voice over IP 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 overview 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.

Its 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 ISP’s who can guarantee QOS over broadband. After you enter their IP network you stay on it and dont 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.

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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 PBX 's -Where existing PBX's use digital trunks (E1 or ISDN for example) we must consider the signalling, systems used by those PBX's to communicate with each other. Some of the more sophisticated PBX's use timeslot 16 to communicate with multiple other PBX's 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 PBX's 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.

PBX's with ISDN interfaces - For PBX's 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 IP address 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 PBX's - Some of the older PBX's 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 queueing 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:

Queueing

- Fair-queueing
- First in first out (FIFO) ·
- Weighted round robin,
- WRR · class based weighted fair queueing
- 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 PBX's 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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BT to Integrate Voice into its Global MPLS Network

BT says that the new network will allow it to handle large volumes of traffic while delivering new advanced services to customers. The new 21C network will support a wide range of hosted or premises-based voice and multimedia applications, while providing ease of integration with existing servers and a standard application interface. A spokesman from BT has reported that the network is already available in 12 countries. BT are promoting the single integrated voice platform with MPLS as being something customers can leverage the benefits of globalisation from, and allowing them to avoid the complexities and overheads of a patch-worked networking environment. Cynics might say that the greatest winner from this integration is BT, allowing them to dispose of networks which provide better quality voice than VOIP, and putting everything onto a single platform, saving BT a fortune, and downgrading the quality of their services. Its rumoured that once we have all gone Voice Over IP, we will be charged by the byte, thus the costs of the next generation network may be far more expensive than the existing networks as far as subscribers are concerned.

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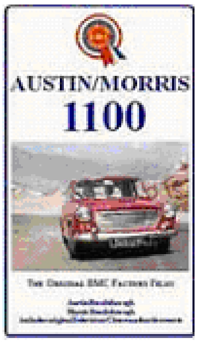

Over 40 years on, and Nissan have just previewed the PIVO at the International Motor Show Geneva 2006 – a Reggie dream mobile!

Promoted as the town car of the future, if your fantasy vehicle is compact, with sci-fi features, this car is for YOU. Electrically driven, the 'cabin' seats three people, driver in the middle, with a passenger either side. It is awash with hi-tech wizardry, including strategically mounted cameras, relaying a 360° view of external surroundings to a dashboard monitor. An Infared (IR) Commander responds to hand or finger signals to operate controls. It can even project information on to the windscreen. The pièce de résistance, however, is that the cabin can be rotated 360°. You would never have to reverse again. Problems with parallel parking and reversing round corners but distant memories. Other benefits afforded by such versatility are left to the readers' imagination. For the moment, no production is planned.

Even if it were, doubtless you would need to part with a shed load of cash before becoming a proud owner. Nevertheless, it will be a seminal moment for the motorist when just such a 'head turner' finally does arrive in car showrooms. The phrase 'let's go for a spin' will be immortalised.

Tilly Tattle 6.03.06

Links: Reginald Moleshusband

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Cisco sways away from Windows with Linux based 'Call Manager'

Its move away from a Windows platform follows heightened concerns over the security of IP telephony systems.

Cisco said in February that its CallManager software could be subject to a denial of service attack. Hacking attempts are far more common on Windows platforms than open source alternatives, largely because of their popularity.

The new version of Call Manager, version 5.0, is currently on "controlled release", Cisco said, which means customers can order it, but delivery will take a couple of months. Version 5.0 will be based on Red Hat Linux, and it will be offered for the first time as an appliance, as well as the traditional software.

Cisco executives said that the move to Linux would make Call Manager "easier to install", but they only reluctantly accepted that the move was made for security reasons. The vendor will continue to sell Call Manager 4.2, which operates on the Windows platform. However, it will merge versions 4.2 and 5.0 into Call Manager 5.1 in due course, offering both Windows and Linux based versions in version 5.1. It will also continue to sell its CallManager Express and Survivable Remote Site Telephony, which are built into its Integrated Services Router. Miercom, a test house which has recently evaluated a range of IP PBXs, said in its report on the subject that the security of IP telephony systems has been bolstered, but it said that "there are still security holes to be dealt with." Five large vendors, including Siemens, Alcatel and Avaya, offered their systems to Miercom for testing late last year, but Cisco declined, saying Call Manager was not ready for review. Linux has become the call control platform of choice, Miercom said, with few PBXs running from Windows. A small number, such as Siemens' HiPath, run on Unix and Unix derivatives.

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Case Communications win contract to supply high-speed routers.

March 2006 Following several months of intensive testing, Case Communications has won a contract to supply their range of high-speed Linux routers, running Packet Over SONET and equipped with monitoring cards, to a government in the Middle East.

Case Communications Business development manager Massey Monfared, commented, ‘Our customer tested routers from the usual router vendors but found a lot of these organisations just could not run the circuit at wire speed, and were significantly more expensive than the Case Routers.’

The biggest challenge we face is brand awareness, a lot of organisations simply say ‘We only buy equipment from X,Y Z, and use this as an excuse to stick their head in the sand. If we are talking to a customer who knows what they are doing and can test the equipment, we rarely loose a deal.’

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Cisco moves to SIP

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In an announcement last week, Cisco said Unified CallManager 5.0, Express 3.4 and Survivable Remote Site Telephony (SRST) 3.4 products will now support SIP natively, "effectively opening up the system to an emerging standards-based developer community while retaining the current security and resiliency features." SIP is now the most commonly used protocol to initiate connections between Voice over IP (VoIP) enabled devices, and also allows for the implementation of enhanced technologies such as presence, mobility and video calling capabilities.

It is thought Cisco has been reluctant to adopt SIP as it paves the way for customers to integrate Cisco VoIP kit into a multi-vendor infrastructure, rather than being locked in to a Cisco offering

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An overview of Wimax

WiMAX is an acronym that stands for **W**orldwide **I**nteroperability for **M**icrowave **A**ccess, a certification mark for products that pass conformity and interoperability tests for the IEEE 802.16 standards.

WiMAX is a standards-based wireless technology that provides high-throughput broadband connections over long distances. WiMAX can be used for a number of applications, including "last mile" broadband connections, hotspots and cellular backhaul, and high-speed enterprise connectivity for business.

Products that pass the conformity tests for WiMAX are capable of forming wireless connections between them to permit the carrying of internet packet data. It is similar to WiFi in concept, but has certain improvements that are aimed at improving performance and should permit usage over much greater distances. A WIMAX wireless internet map of coverage is being publicly developed now.

IEEE 802.16 is working group number 16 of IEEE 802, specializing in point-to-multipoint broadband wireless access.

Technical advantages over WiFiBecause IEEE 802.16 networks use the same LLC layer (standardized by IEEE 802.2) as other LANs and WANs, it can be both bridged and routed to them. An important aspect of the IEEE 802.16 is that it defines a MAC layer that supports multiple physical layer (PHY) specifications. This is crucial to allow equipment makers to differentiate their offerings. This is also an important aspect of why WiMAX can be described as a "framework for the evolution of wireless broadband" rather than a static implementation of wireless technologies.

Enhancements to current and new technologies and potentially new basic technologies incorporated into the PHY (physical layer) can be used.

A converging trend is the use of multi-mode and multi-radio SoCs and system designs that are harmonized through the use of common MAC, system management, roaming, IMS and other levels of the system. WiMAX may be described as a bold attempt at forging many technologies to serve many needs across many spectrums. The MAC is significantly different from that of IEEE 802.11 Wi-Fi (and Ethernet). In Wi-Fi, the MAC uses contention access-all subscriber stations wishing to pass data through an access point are competing for the AP's attention on a random basis. This can cause distant nodes from the AP to be repeatedly interrupted by less sensitive, closer nodes, greatly reducing their throughput.

By contrast, the 802.16 MAC is a scheduling MAC where the subscriber station only has to compete once (for initial entry into the network). After that it is allocated a time slot by the base station. The time slot can enlarge and constrict, but it remains assigned to the subscriber station meaning that other subscribers are not supposed to use it but take their turn. This scheduling algorithm is stable under overload and oversubscription (unlike 802.11). It is also much more bandwidth efficient. The scheduling algorithm also allows the base station to control Quality of Service by balancing the assignments among the needs of the subscriber stations. A recent addition to the WiMAX standard is underway which will add full mesh networking capability by enabling WiMAX nodes to simultaneously operate in "subscriber station" and "base station" mode. This will blur that initial distinction and allow for widespread adoption of WiMAX based mesh networks and promises widespread WiMAX adoption.

The original WiMAX standard, IEEE 802.16, specifies WiMAX in the 10 to 66 GHz range. 802.16a added support for the 2 to 11 GHz range of which most parts are already unlicensed internationally and only very few still require domestic licenses. Most business interest will probably be in the 802.16a standard, as opposed to licensed frequencies. The WiMAX specification improves upon many of the limitations of the Wi-Fi standard by providing increased bandwidth and stronger encryption. It also aims to provide connectivity between network endpoints without direct line of sight in some circumstances. The details of performance under non-line of sight (NLOS) circumstances are unclear as they have yet to be demonstrated. It is commonly considered that spectrum under 5-6 GHz is needed to provide reasonable NLOS performance and cost effectiveness for PtM (point to multi-point) deployments. WiMAX makes clever use of multi-path signals but does not defy the laws of physics.
Uses for WiMAX

WiMAX is a wireless metropolitan area network (MAN) technology that can connect IEEE 802.11 (Wi-Fi) hotspots with each other and to other parts of the Internet and provide a wireless alternative to cable and DSL for last mile (last km) broadband access. IEEE 802.16 provides up to 50 km (31 miles) of linear service area range and allows connectivity between users without a direct line of sight. Note that this should not be taken to mean that users 50 km (31 miles) away without line of sight will have connectivity. Practical limits from real world tests seem to be around "3 to 5 miles" (5 to 8 kilometers). The technology has been claimed to provide shared data rates up to 70 Mbit/s, which, according to WiMAX proponents, is enough bandwidth to simultaneously support more than 60 businesses with T1-type connectivity and well over a thousand homes at 1Mbit/s DSL-level connectivity. Real world tests, however, show practical maximum data rates between 500kbit/s and 2 Mbit/s, depending on conditions at a given site.

It is also anticipated that WiMAX will allow interpenetration for broadband service provision of VoIP, video, and Internet access-simultaneously. Most cable and traditional telephone companies are closely examining or actively trial-testing the potential of WiMAX for "last mile" connectivity. This should result in better pricepoints for both home and business customers as competition results from the elimination of the "captive" customer bases both telephone and cable networks traditionally enjoyed. Even in areas without preexisting physical cable or telephone networks, WiMAX could allow access between anyone within range of each other. Home units the size of a paperback book that provide both phone and network connection points are already available and easy to install. There is also interesting potential for interoperability of WiMAX with legacy cellular networks. WiMAX antennas can "share" a cell tower without compromising the function of cellular arrays already in place. Companies that already lease cell sites in widespread service areas have a unique opportunity to diversify, and often already have the necessary spectrum available to them (i.e. they own the licenses for radio frequencies important to increased speed and/or range of a WiMAX connection). WiMAX antennae may be even connected to an Internet backbone via either a light fiber optics cable or a directional microwave link.

Some cellular companies are evaluating WiMAX as a means of increasing bandwidth for a variety of data-intensive applications. In line with these possible applications is the technology's ability to serve as a very high bandwidth "backhaul" for Internet or cellular phone traffic from remote areas back to a backbone. Although the cost-effectiveness of WiMAX in a remote application will be higher, it is definitely not limited to such applications, and may in fact be an answer to expensive urban deployments of T1 backhauls as well. Given developing countries' (such as in Africa) limited wired infrastructure, the costs to install a WiMAX station in conjunction with an existing cellular tower or even as a solitary hub will be diminutive in comparison to developing a wired solution. The wide, flat expanses and low population density of such an area lends itself well to WiMAX and its current diametrical range of 30 miles. For countries that have skipped wired infrastructure as a result of inhibitive costs and unsympathetic geography, WiMAX can enhance wireless infrastructure in an inexpensive, decentralized, deployment-friendly and effective manner.