Voice Rules: The Practical Challenges of Voice on the LAN

White Paper

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1. Introduction

During the past year, virtually all of the network and telecommunications trade publications have run feature stories about the trend of convergence – voice, video, and data communications all being transmitted across a common infrastructure, typically based on IP (the Internet Protocol). This includes Internet telephony as a way to bypass long-distance toll charges, and various techniques such as voice-over-ATM or voice-over-frame relay for carrying voice traffic over the pre-existing enterprise data networks. Convergence of voice and data traffic in the wide area network, is becoming more common due to the immediate cost savings that can often be realized.

Convergence is also being driven by the ubiquity and characteristics of IP networking. IP networks are based on a totally distributed architecture where no single entity has to be in control; instead these networks are able to grow in a “bottom-up” fashion as individual organizations add their own subnets to the global Internet. IP networks are ideal for voice/data convergence because users can add intelligence at the network edge; unlike traditional carrier networks, users don’t have to wait for the carrier to add desired features.

But attention is now moving to the “last mile” of convergence – voice on the enterprise LAN. This implies not just the transmission of voice and data traffic over a common local transport network, but a radical shift in the way that organizations deploy, operate, and maintain their telephone systems. While there are many variations (described below) of voice over the LAN, one of the most-discussed alternatives is to use a combination of Ethernet phones with LAN-based call servers, as an alternative to the traditional PBX with its directly-wired telephone sets. These solutions are typically referred to as LAN PBXs or IP-PBXs.

Many vendors have begun to introduce LAN PBX products. These include large established companies such as Lucent, Nortel, Siemens, and Cisco, as well as a number of startup firms such as Sphere, E-Net, and NBX (acquired by 3Com). However, these early products may not measure up to the standards set by today’s PBX-based telephone systems. Customers need to understand the real practical challenges that can be encountered when implementing a voice communications system based on voice over the LAN today. Issues of interoperability, regulatory requirements, reliability, usability, scalability, and functionality must be addressed before these new technology alternatives will meet user expectations, without compromising the voice values that have been delivered by the traditional PBX solutions of the past decade or so.
Resolution of these issues isn’t a matter of “if,” but rather “when.” However, these issues must be addressed by the industry before LAN PBX solutions achieve the market penetration and success that the industry hype would have us believe is the case today.

Mitel is unquestionably an advocate of voice-over-the-LAN solutions. In fact, Mitel has been an early leader in voiceLAN educational programs at industry conferences, and has delivered some of the very first converged voice/data products that leverage LAN connectivity in voice communications systems - bringing voice and data together in solutions that deliver simplicity and cost savings in areas such a directory services, Windows NT-based call services, unified messaging, and CTI applications.

The purpose of this paper is to highlight the practical challenges of voice on the LAN today, as well as some of the potential solutions or alternatives. Because LAN PBXs are relatively new products, awareness of these issues will help IT and telecommunications managers to be better informed as they evaluate these product alternatives.

With over 25 years of voice communications experience, Mitel knows and understands how to build voice system products that meet customer expectations. Our 25-year voice pedigree also gives us an advantage in thoroughly understanding the voice rules associated with voice on the LAN solutions, as well as the potential solutions. Mitel is in the forefront of vendors developing LAN PBX solutions that will solve these challenges without compromising on the functionality, reliability and quality that businesses have come to rely on with traditional PBX solutions.
2. VoiceLAN

LAN PBXs are just a subset of an overall converged network vision that refers to VoiceLAN. VoiceLAN describes a group of technologies that unify all of the functional and physical levels of enterprise voice networks and local area networks. Within a VoiceLAN environment, the call control functionality normally provided by a PBX is instead provided by a call server on the LAN. The call server is often a Unix or Windows NT (Windows 2000) server running on standard PC server hardware. The call server may also operate as a gateway for calls traveling between the enterprise LAN environment and the public switched telephone network (PSTN). A typical VoiceLAN environment is shown in Figure 1 below:

Figure 1: VoiceLAN: Enterprise Convergence Model
A VoiceLAN environment encompasses a number of logical areas of convergence including systems management, applications, wide-area access, call control, switching fabrics, and desktop computing and communications devices. VoiceLAN allows cost savings, flexibility, uniformity, and remarkable service efficiency. For end users, this translates into greater ease of use, familiarity with system operations, and a clearer sense of what to expect from the combined services.

3. Voice-on-the-LAN Desktop Architectures

Before discussing the issues involved in implementing voice-on-the-LAN systems, it is important to understand the types of implementations that will be available in the marketplace, since they utilize different desktop network topologies and protocols. Depending on the vendor chosen, an enterprise might end up with an entirely separate wiring plant for its telephones, special hubs for PC versus phones, or different connectors for attaching desktop PCs to an IP-based phone or visa versa. This section explains some of the different techniques vendors use to connect telephone sets to a data network.

![Figure 2: Voice On LAN: Desktop Implementations](image-url)
a) Dual-port Ethernet Telephones

As shown in part (a) of Figure 2 diagram, one alternative for connecting IP-based phones to the data network is to use a telephone with dual Ethernet ports: the phone acts as a two-port switch through which the desktop PC accesses the LAN. This method is designed to use an existing single LAN port for both a desktop computer and an IP-based phone. In this configuration, because the Ethernet phone sits in front of the networked PC, crashing of the PC cannot affect phone service. However, multiple LAN ports in modern desktop offices may not be rare or costly anymore, so the advantage of sharing one Ethernet connection may no longer be an important factor.

Additionally, this approach may rely on the LAN cabling to deliver power to the phone, which may not have a separate AC adapter; DC electrical power is provided via an unused pair of LAN wires. Such an arrangement could significantly complicate cable installation and reconfiguration, and creates the risk that desktop computers may accidentally be plugged into a non-standard powered Ethernet jack (potentially damaging the computer). The phone must also be in place for the desktop PC to connect to the network.

This solution also requires separate Category 5 certification for the patch cables running between the Ethernet phone and the desktop PC. (This is not an issue for 10Base-T where Category 3 wiring is used.)

As a final consideration for this alternative, dual-port phones are normally more expensive than normal phone sets or other Ethernet phones, and can cost a few hundred dollars or more each. Dual port phones today typically have a mini hub or switch operating inside the phone plastics which drives up the cost of the phone. This needs to be balanced with the low cost today of Ethernet switch ports, which may justify utilizing multiple ports at the desktop.

Ultimately, notwithstanding the power issues that are covered as Key Issue 3 below, further integration within silicon will eliminate the need for the expensive componentry contained within the phone and will bring down the cost to an acceptable level.

b) Separate LAN Connectivity for Desktop PCs and Ethernet Telephones

In keeping with the traditional separate infrastructure for the telephone network and the data network, some vendors rely on independent LAN cable drops for desktop PCs and Ethernet telephones, as illustrated in the diagram part (b). While this approach seems like a logical way to use the existing cabling, it may have drawbacks. Ethernet phones require at least Category 3 cable and will not work over some traditional voice-grade wiring. While desktop work locations might have two or more Cat 3 or 5 drops, there might not be certified UTP (Unshielded Twisted Pair) LAN cable everywhere there is only a phone, such as within a building lobby or within a break room.
However in locations where there is existing wiring, using separate drops for the phone and desktop PC is a viable solution. For example, most modern buildings have at least two drops of Category 5 cable to each location. Note that both of these drops are usually certified for 100 MHz Category 5 UTP (Unshielded Twisted Pair), even if they are not currently in use. Switch ports are all coming down in price, so the cost addition to the wiring closet is not a big factor in using a separate wiring topology.

One reason for separating the wiring of the telephones and network clients is the use of different network transport architectures. Products such as Sphere’s Sphericall use ATM as the transport to connect the Ethernet phones to the LAN PBX. Since most enterprises are not running ATM to their desktop PCs, this technology drives the need for a separate wiring infrastructure for the ATM phones.

There are other reasons that support the use of separate wiring to the desktop. It makes troubleshooting problems easier for support staff by having separate traffic paths back to the switch. And some Ethernet switches can classify traffic into different quality of service (QoS) classes based upon switch port; this would permit voice and data traffic to receive different prioritization by the network.

c) Phone Attached to the Network via the Desktop PC

Some vendors market systems that make the desktop computer into a telephone. A soundboard in the networked computer, equipped with a digital signal processor, interfaces to a telephone handset or to a microphone and speakers. Alternatively, a telephone set may attach to the desktop PC via a USB (universal serial bus) or RS232 connection as shown in part (c) of the diagram. The most obvious drawback to this technique is the need for the desktop computer to be turned on and attached to the network in order to make and receive voice calls. If the desktop computer crashes or loses the network connection, the phone will not work.

Secondarily, the soundboard and microphone combination can be difficult to use and voice quality in this scenario can be compromised. There is also an issue with having to open a PC to add the necessary soundboard. This solution requires other desktop implementations for those locations where there may not be a PC present. While a PC can offer a rich screen-based GUI, optimized voice quality and phone features can be lost. Furthermore, people are familiar with their existing telephone sets and may find using a GUI application too much trouble for placing simple phone calls.

Implementation of this solution will likely be driven by multimedia/convergent applications rather than as an alternative to a standard telephone. Another driver may be cost – users may expect “soft phones” to be priced less than conventional handsets.
d) Voice Hub for Legacy POTs Devices

Moving to an IP-based environment should not mean having to throw out legacy analog or POTs (Plain Old Telephone Service) equipment such as analog phones for emergency use or in non-office areas like warehouses. Today, these account on average for at least 10% of total line stations in a business. Other items including modems, fax machines, recorded announcement devices for call centers, paging sources, answering machines, conference room speakerphones, etc., may also need continued connectivity. For the most part these are analog devices that are easily accommodated by traditional circuit-switched voice systems. Moving to an IP solution poses a challenge in supporting legacy analog devices.

Therefore a hardware device such as an analog or legacy voice hub needs to be part of any LAN PBX solution. Depending on the product, these hubs can attach directly to the Ethernet RJ45 jack that links to a standard 10/100 Mbps Ethernet hub/switch. Externally the phone hub has RJ11 ports for attaching standard analog devices. There may be issues with using analog fax/modem behind the hub because tones are very sensitive to lost packets and jitter. Using a phone hub for faxes can be problematical because of the number of analog/digital conversions required.

4. Key Issues

There are at least a dozen key issues that must be addressed before LAN PBX solutions can meet and exceed the expectations set by existing PBX products. The bar for acceptable voice system performance has been set for many years and the enterprise will not, and should not, expect anything less from a LAN PBX solution. In other words, moving to a LAN PBX environment should not force customers to compromise on what they have today.

Any successful voice over the LAN telephone system must address the following issues; these “rules” should be considered in any LAN PBX implementation:

a) Reliability, Redundancy, Resiliency

Issue: How can LAN PBX systems be made reliable?

For most enterprises, telephony represents the original “mission critical” application, implemented long before other data applications became critical. When the lights go out and their PCs shut down, users immediately reach for the phone to call someone regarding the outage. Likewise, many organizations never have to reset their proprietary PBXs, but are accustomed to rebooting their servers on a weekly or monthly basis. Before a business can seriously consider implementing a LAN PBX system, it needs to understand the reliability of such a system.
One way to achieve high reliability is to install redundant and fault-tolerant system components. Redundancy provides fault-tolerance because if a component fails, the hot spare will take over without any loss of service. Further fault tolerance can be achieved through the use of hot-swappable components. Even if a component fails, a truly reliable product will allow the technician to replace the faulty unit without powering down the entire system. These types of fault tolerant and redundant equipment techniques are expected in today’s PBXs, but are not necessarily included in PC servers, including any PC-based server that runs call server software in a LAN PBX environment. While many of the server’s components can be made redundant, this reliability comes at a cost that many small and medium sized businesses may not be able to afford.

Another well-known fact that must be addressed is that PC servers that run multiple applications (e.g., CTI applications in addition to the LAN PBX call server software) will be exposed to more failures due to the “misbehavior” of other applications. It is typically recommended that the organization run the call server software on a dedicated hardware server, at least for the near term.

In addition to hardware failure, server downtime can also be attributed to power loss. There are several UPS (Uninterruptible Power Supply) products on the market that will keep PC servers running for a limited amount of time. Unfortunately, most of these products are not designed for high server availability in the event of a significant power failure, but are only meant to allow users to save their current data and allow the administrator to gracefully shut down the server. This is not acceptable for a telephone system, so administrators will have to implement a battery backup system for all LAN PBX components -- a system that can maintain power to the call server, network connectivity devices (such as Ethernet switches and routers), and Ethernet phones in the advent of a power outage. All of this adds to the cost and complexity of the LAN PBX solution.

b) Voice Quality

Issue: How can “toll quality” voice at least as good as today’s PBXs or circuit-switched phone systems be obtained?

A common belief is that most LANs have enough excess capacity to handle the relatively small amount of traffic generated by voice applications. However, traffic loads are dynamic on a LAN, and only the use of specialized and sophisticated bandwidth allocation techniques can guarantee available bandwidth at all times of operation (see QoS discussion below). These factors coupled with the lower bandwidth of WAN connections means that LAN PBX systems must be prepared to deal with varying bandwidth without sacrificing quality.
The bane of any packetized voice stream is delay. Once a packet takes longer than 100 - 200 milliseconds to travel to its location, voice quality starts to become unacceptable. Human factors research has shown that most people greatly dislike long transmission delays in telephone calls. Subjectively, too much latency makes it difficult to carry on a very interactive dialog, which is typically the case in voice communications. This is particularly apparent when people try to interject comments or questions while someone else is talking. One party hears that the other has stopped speaking and begins to talk, only to find that the second party merely paused before continuing to say more. As latency increases, people tend to “step all over each other” in the conversation, not unlike collisions on an Ethernet LAN. See Figure 3 below.

Figure 3: Impact of Latency on Quality

There are several factors that can cause delay of voice traffic in an IP or other packet network, including message serialization delay, the time needed to process voice samples and compress/decompress them in a codec, network delays including queuing, and the need to accommodate delay variation at the receiver in a jitter buffer. Fortunately packet collisions have become a non-issue because they are eliminated in a switched Ethernet environment. Well-engineered LAN PBX systems will seek to minimize all of these elements in order to keep total end-to-end delays below 100 milliseconds.
Another factor that will determine voice quality is the particular codec (i.e., coder/decoder) being used. A user’s voice signal must be sampled and converted to a digital bit stream before it can be transmitted over the network; typically some sort of compression is also employed to reduce the amount of bandwidth required. Use of more compression generally results in worse voice quality (i.e., less voice fidelity), and adds more delay due to the additional processing required.

Because of this tradeoff between voice quality versus bandwidth savings due to compression, there are different ITU (International Telecommunications Union) codec standards that may be used. For example, some older standards such as G.711 are designed to code and decode voice at an uncompressed rate of 64 Kbps, but with very high quality and minimal delay. Use of this codec may be appropriate if there is considerable available bandwidth. Other codecs such as G.723 can compress voice down to 5 or 6 Kbps, but are more computationally intensive, provide a delay of up to 30 ms for compression/decompression processing, and the resulting speech quality may not sound as good. The best solution may be to use a codec capable of compression, but only compress on a call-by-call basis. For example, a call to another user on the LAN may only need to use a G.711 codec, but a call to a remote user across the WAN may need the added compression of a G.723 codec.

Other factors that may affect voice quality in a LAN PBX system are echo and packet loss. Echo becomes a problem in most voice over packet networks because the round-trip delay through the network is almost always greater than 50 ms – the threshold of human perception. Thus, LAN PBX systems must address the need for echo control and may need to implement some means of echo cancellation.

Lost or dropped packets can be a significant problem affecting voice quality, although this is likely to be more significant on bandwidth-constrained wide-area links than LANs. Because IP networks do not guarantee packet delivery, under peak loads and congestion, voice frames will normally be dropped in a manner similar to data frames. Most data frames, however, are not time-sensitive and dropped packets can be recovered through the process of retransmission by the TCP (Transmission Control Protocol) end-to-end protocol. However, this approach doesn’t work with dropped interactive voice packets, since they are encapsulated in UDP instead of TCP; there isn’t enough time to retransmit lost or errored voice packets.

Some methods used by voice over IP systems to address the problem of lost frames include interpolation for lost speech packets by replaying the last packet received during the interval when the lost packet was supposed to be played out, or transmission of redundant information at the expense of bandwidth utilization. However, this last approach uses more bandwidth and creates greater delay.
c) Power to Ethernet Phones

Issue: How do vendors plan to provide electrical power to Ethernet/IP telephones?

As with any other electronic device, these phones need a supply of power -- an easily overlooked requirement that many businesses may not consider when evaluating a LAN PBX solution. Most traditional analog or digital phones can continue to operate in the event of an electric utility/building power failure, with power generated by the PBX. In a LAN PBX environment, not only must uninterruptible power be supplied to the LAN PBX call server, but each desktop Ethernet phone may also need similar power protection in order to operate. UPS power aside, the need for a separate AC adapter for each Ethernet phone set is not as attractive as traditional analog or digital phones that don’t need a separate power connection to work.

In an effort to eliminate the large number of AC adapters, LAN PBX vendors are coming up with different solutions. One of the most interesting approaches is centralized power – sending DC power across a spare pair (wires 7 and 8) of Category 3 or 5 UTP cable. This eliminates the need for an AC adapter and battery backup (and moves this problem of reliable power back to the wiring closet). This method, currently used by Cisco (Selsius) and NBX, may be acceptable as long as the spare pair is not in use, and the phone is placed between a networked PC and the LAN. If an unsuspecting user plugs their computer into a powered RJ45 jack, the DC current may damage the PC. (Some Ethernet NICs may terminate unused pairs, potentially shorting out the power supply as well). However, as new categories of UTP are developed, and technologies such as Gigabit Ethernet over UTP (which requires four copper pairs) become more prevalent, there may not be any spare wires available in the UTP drops. As this happens, vendors will need to look for a better solution.

Another solution is to use a technique called phantom feed to send a small amount of DC power across the Ethernet wiring. Based on the now-defunct IEEE 802.9F committee proposal, phantom feed would allow power to be supplied over the same pairs used by Ethernet. In this scenario, the Ethernet hub/switch provides power via the unused center taps on the transformers normally used on Ethernet transmit and receive pairs; at an Ethernet phone, a DC voltage can be derived from these same transformers in the LAN interface. While this seems like a good solution, it will require vendors to develop special Ethernet phone-capable Ethernet hubs-switches and to agree on a standard.

Widespread deployment of Ethernet phones may not happen until the LAN infrastructure industry solves the problem of reliable centralized power from the wiring closet, which makes them comparable to existing PBX solutions. Ideally standards bodies such as the IEEE 802 group will address this, but in the short term, individual vendors may be able to act unilaterally to set some de facto standards.
In the absence of industry standards to deliver centralized power, an interim solution is to use an adjunct power module. This resides in the closet to deliver spare pair power, and acts as an adjunct to a pre-existing LAN hub or switch. After leaving the LAN hub or switch, cabling passes through the adjunct power module, which applies power on a port-by-port basis using a powered spare pair scheme. This is relatively easy to implement, particularly in a typical patch panel configuration.

Alternatively, some customers may choose to install a separate power supply and feed power through the punch down block. While this scheme can be accommodated without any additional specialized equipment, it is more labor intensive and subject to a higher failure rate.

d) Feature Sets and Functionality.
Issue: How do organizations obtain the approximately 500 features of their existing PBX, given that on average, new LAN PBX or UN-PBX systems deliver about 50 features?

Probably no business uses all of the features and functions of their traditional PBX, but each business likely uses a different set of features. Unfortunately, that makes it hard for vendors to decide which features to implement first in their new LAN PBX products. Notably, many of these hundreds of PBX features are just minute variations of each other; they do have special usage for different businesses and vertical industries. The need to pick specific features makes it difficult for the IT or telecom manager, who has not had to worry about features for the past 10 years. Now, these managers may fear giving up features that only a few users use and value. Overall, this decision can cause numerous headaches.

Currently most LAN PBX vendors are only implementing the basic features that almost everyone uses, such as call transfer and conferencing. This unfortunately causes many customers to wonder why they would want to step backwards and accept fewer features than they have with their pre-existing PBXs.

This issue of reduced functionality applies to telephone sets as well. Some LAN PBX vendors today support only analog phones, which poses usability issues and a step backwards for users who are accustomed to display feature phones.

Existing Ethernet phones today typically don’t provide the customer with many choices. For example, if a user needs a special feature phone with a LCD display, he/she may have to use a phone hub to adapt an analog phone or proprietary digital phone set to the LAN PBX system, since there may be no equivalent LAN PBX feature phone. Currently there are few options for choosing among a limited range of different Ethernet phones, and the costs are extremely high for the features that are included. Another issue is familiarity and the existing investment in training on existing proprietary sets. Ideally, voice vendors need to offer both traditional and IP variants of their telephone sets.
In considering a LAN PBX system, users need to ask themselves: How does this compare to what I have today? What am I giving up? What are the plans for future enhancements? How many types of IP feature phones are available?

e) E911 Compliance

Issue: When a person dials 911, how can emergency workers locate that particular telephone user in a LAN PBX environment?

There is an emerging requirement in North America to make voice systems more compliant with E911 emergency calls. For example, if a person is located in a large building or campus environment and places a 911 emergency call, the only information the emergency call center currently can derive is the main address for that building or campus.

The problem of E911 compliance is being addressed in two phases: first local notification, and eventually remote location notification. PBX vendors such as Mitel are developing methods to alert a local administrative center at the campus or in the building when someone makes an emergency call.

Here is an example: At the same time that an actual 911 call is being made, the PBX will send information to a CTI application that provides a "screen pop" to an attendant or security staff relating the caller’s physical location on the campus with their extension number. When an emergency team (e.g., police or fire department) arrives, an employee will be able to meet the team and direct them to the scene of the emergency. In the future, if this location information is to make it all the way to a remote emergency call center, standardized signaling or the use of common applications must be implemented in every region in North America.

If an enterprise decides to deploy a LAN PBX telephone system, the issue of E911 compliance becomes more complicated. The primary difference between wired and Ethernet phones is one of mobility, in that an Ethernet phone can be moved without any centralized administrative intervention.

PBXs simply rely on ports to determine the location of a phone. The closest relationship to a physical port in the IP world is an address. Most hubs and switches can use embedded SNMP (Simple Network Management Protocol) agents to report the IP address of each port. In turn, this information can be used to generate a map of all of the IP addresses and assign them to a location in a campus or building. However, a general trend has emerged for many organizations to rely on DHCP (Dynamic Host Configuration Protocol) to assign IP address leases on a dynamic basis. As IP addresses change and are reused by different physical devices, the system will be required to maintain up-to-the-minute mappings.
Another solution for tracking user/device locations may be to use the user’s login name. This will require special computer telephony (CT) applications and in cases where the Ethernet phone is not connected to the desktop PC, the user may have to login to the telephone set.

f) Distance Limitations
Issue: How can phones overcome the 100-meter maximum distance of Ethernet over unshielded twisted pair wiring?

Most proprietary PBXs allow organizations to locate phones as far as 1 or 2 kilometers away from the PBX. However, because they are standard Ethernet devices, Ethernet phones are limited to 100 meters between the phone set and the hub/switch. This could be a limiting factor for widespread deployment of LAN PBX systems in large buildings or factories where phones are a necessity, but PCs are not particularly widespread.

The options for solving this problem are the same as for expanding the reach or coverage of a LAN. The most basic solution is to use an Ethernet repeater to extend the distance between an Ethernet phone and a hub/switch located in a wiring center. However the number of allowable repeaters and maximum Ethernet diameter restrictions ultimately limits this solution. Another solution is to extend the backbone by utilizing fiber and a pair of Ethernet fiber converters. The bottom line is that these Ethernet solutions are somewhat cumbersome and expensive relative to simply installing a longer run of copper cabling between a PBX and a traditional phone set. The upside is that they are consistent with and use common LAN technology with which customers are familiar.

Another solution is to deploy a legacy analog voice hub to provide analog phones to these locations beyond the 100-meter limit.

g) Network Quality of Service
Issue: How can businesses know that their (pre-existing) LAN infrastructures will deliver service quality adequate for voice over IP?

Compared to most data applications, voice packets are much more susceptible to any kind of delay, jitter, or dropped packets; these transmission impairments will be readily noticeable and result in degraded voice quality (see voice quality discussion above). Thus while a preexisting network may be acceptable for data transmission, it may not be adequate for sending voice packets.
How do enterprises determine if their LAN or WAN infrastructures provide acceptable quality of service (QoS) for voice? This is not something that can easily be tested, since problems may only occur during unusual or peak data traffic periods. Instead, network managers need to consider how their networks will respond during such periods.

There is a perception that if average LAN utilization is sufficiently low (or conversely, there is an abundance of bandwidth), than voice quality will be acceptable. However, this is not necessarily true. It is the nature of most IP data applications to burst to the full capacity of a circuit, even for very short periods of time. Thus even in a lightly loaded Fast Ethernet LAN, whenever an application is transmitting, it will grab all of the available bandwidth. If voice packets happen to be transmitted during the same time period, contention for bandwidth will occur and some of the voice packets will be dropped or delayed. Thus “best effort” service for voice may not be good enough, even over high-capacity LAN links.

Fortunately there are several techniques for improving the service delivered to voice traffic. The most basic QoS technique is prioritization. Network administrators will need to rely on some type of prioritization technique, such as differentiated services (DiffServ) in their routers and switches in order to provide voice packets with the guaranteed bandwidth and services they need.

LAN PBX vendors such as Mitel are looking closely at different lightweight QoS techniques to better insure that voice traffic can be transmitted with “toll quality.” One of the most promising QoS techniques for LAN PBX vendors is the IEEE 802.1p/Q standard for prioritization in Layer 2 Ethernet switches. When combined with standards such as RSVP and DiffServ in LAN Layer 3 switches and routers, these QoS techniques may be used to implement a voice-capable LAN infrastructure.

Unfortunately the replacement of pre-existing Layer 2/3 switches and routers with newer QoS-capable devices is not inexpensive. The need for such a network may negate any operational or implementation cost savings used to justify a LAN PBX system installation. On the other hand, QoS is also becoming more important for mission-critical data applications; organizations may be faced with the need to upgrade to a QoS-capable network infrastructure anyway.

Until QoS is implemented, engineered LAN configurations may mitigate QoS issues but this is not trivial undertaking. Customers need to be able to determine how lightly loaded the LAN must be in order to ensure that voice packets don’t get dropped when bandwidth contention occurs. Is the limit 5% utilization? 10%? 20%? What happens to voice during peak traffic periods? Even if the utilization limit is understood, this approach requires constant monitoring of LAN traffic utilization to ensure it stays below this limit.
h) Business Value Proposition

Issue: *Where is the value to permit an organization to justify the installation of a LAN PBX system?*

Making “free” long distance calls over the Internet may be justification enough for implementing a voice over IP in the WAN, but why would a business want to run its voice communications over the LAN (instead of using a conventional PBX)? The unfortunate situation is that it may be difficult to create a business case today for LAN PBX installation.

The most common argument for LAN PBX is the replacement of two separate voice and data infrastructures with a single converged data network. While this sounds nice in theory, in practice most organizations have already invested in separate building wiring infrastructures, PBXs, traditional analog or digital phone sets, and people/skill sets for maintaining their circuit-switched phone systems. Thus it is difficult to justify replacement of an existing PBX with a LAN PBX system. Note that this is the biggest challenge for larger businesses with substantial investments in voice communications systems; smaller businesses are less constrained by preexisting systems.

Moves and changes are another argument for installing a LAN PBX system. Any user can move his/her desktop to another location on the LAN and login with the same privileges and access to resources. Moving a phone set to a new jack without losing services and features is not as easy. In fact, Computer Telephony magazine recently quoted Microsoft executives who claimed that the company could save $3 million dollars a year if they could move telephones as easy as their networked PCs.

Most LAN PBX vendors will agree that there is no new “killer application” yet enabled by LAN PBX systems and Ethernet phones. Customer demand for multimedia applications is generally not here yet. Part of the reason is that computer telephony integration (CTI) applications work well with today’s PBX systems; the converged infrastructure adds little value in that regard. Virtually the same application program interfaces (APIs) are available in both LAN PBX and more-traditional environments. There is an expectation that LAN PBX systems will be justified due to lower prices compared to the proprietary systems offered by traditional PBX vendors; to date, this has not happened. But as WAN links and carrier networks migrate to voice over IP, it should eventually become simpler and less expensive to create an end-to-end voice over IP system that includes desktop LAN devices.

The bottom line today is that the industry needs to deliver on the promise of incremental applications and strengthen the value proposition of IP-based solutions.
i) Investment Protection and Legacy Migration

Issue: How can investment protection be provided for legacy voice systems?

From the perspective of many organizations, their attitude towards replacing a PBX with a LAN PBX system is often: “If it’s not broken, then don’t fix it.”

To date, LAN PBX vendors are not selling their product predominately to companies with existing PBXs, but to smaller companies that have no PBX and have outgrown their key systems or Centrex service. There are several reasons for this.

First, if the PBX has adequate capacity, and users and administrators know how to program it, why settle for a LAN PBX system that has fewer features, and different implementations of these features? Second, switching to a LAN PBX system may represent a “fork lift” upgrade that requires all new phones and some re-wiring. Third, the average life span of a PBX is shrinking but still typically 5 to 10 years (compared to 18 months to 3 years for a PC). Finally, companies have a significant “soft investment” in training users and administrators on how to utilize and administer the PBX -- the thought of training users on a new system with new desktop interfaces, new ways of implementing features, or even doing without key functionality to which they’ve become accustomed is not particularly appealing.

Thus any organization that considers implementation of LAN PBX systems will need to consider asset protection for its preexisting systems (and people). Ideally LAN PBX solutions should work in a “hybrid” implementation along with existing legacy proprietary systems to protect both hard and soft voice investments. Examples are branch offices that are networked to the head office, or departmental adjuncts within larger organizations. Hybrid implementations need to deliver seamless integration, feature and functionality transparency between the IP and traditional systems, common telephone set interfaces, etc. Also, vendors need to demonstrate a clear migration path between legacy voice equipment and next generation platforms. This should define the product roadmap, what products will migrate forward (and what will not,) and how legacy systems will inter-operate with new platforms.

j) Management

Issue: How can LAN PBX systems be managed with the same ease as traditional PBX systems?

PBX telephone systems such as those offered by Mitel are typically managed by physical port; each port can be assigned a phone number, name, location, and calling access and restrictions. This management and configuration is performed using a graphical utility that communicates directly to the PBX hardware.
With LAN PBX systems, there is no concept of a PBX port. Instead, administration will need to be associated with a particular user name or some other differentiator, such as the MAC or IP address of an Ethernet phone.

Ideally a LAN PBX system would be managed based upon uniquely defined user names. With such a system, a user would have the same telephone number and access rights even if he/she was using a different Ethernet phone. By tying this information into a directory service [potentially based on DEN (Directory Enabled Networks) or LDAP (Lightweight Directory Access Protocol)] the administrator would be able to globally manage every Ethernet phone in the entire enterprise from a central location.

It is important to note that the LAN PBX system could act as a proxy for the Ethernet phones, thus allowing the administrator to “manage” these devices to some extent. This area is somewhat more complex than alluded to above, as there are many voice-centric features in a full featured LAN PBX system that need to be administered, including class of service, hunt groups, and ACD groups. The level of management an administrator can provide is similar to that for any SNMP-managed IP device, as per standard MIB (Management Information Base) entries. There may be proprietary MIB extensions needed to manage aspects of a phone that are within the domain of the voice call-control application residing in the LAN PBX.

Another potential solution is to manage each phone based on the IP address of that phone set. While many large enterprises rely on DHCP to administer IP addresses dynamically, managing Ethernet phones based on their IP address would require the use of static addresses. This could complicate the administration of the entire IP addressing scheme for data and voice network devices.

An issue that is unique to LAN PBX systems is policy-based management – maintenance of access control lists and quality of service settings per user or Ethernet phone. As phones move onto the data network, the same management techniques that are used for controlling PCs and users will need to be implemented for LAN PBX components and users.

**k) Security**

**Issue: How can the privacy of voice calls traversing the LAN be assured?**

Anyone with a LAN protocol analyzer can easily access network data as it travels across the LAN. While network switches make it harder to intercept data that is on a different desktop LAN segments, an intruder can easily connect a monitoring device on or close to the network backbone.

It may be argued that voice packets will be harder to decode because an eavesdropper would have to use the correct codec along with a protocol analyzer to listen to a call. While this task is not easy, it is also
not impossible. Because the standard solution for protection from packet eavesdropping on a data network is encryption, vendors will have to consider the same solution for LAN PBX systems.

Most organization only encrypt important data as it travels across a public network and usually do not encrypt LAN traffic. The reasoning behind this practice is the slow performance of software-based encryption and the high cost of faster hardware-based solutions. However, some organizations may decide that voice calls over the LAN are more important to secure than data packets. Interestingly, most LAN PBX vendors will probably leave the packet encryption up to the customer or reseller to implement. This will keep the costs lower and let customers make the decision on whether their voice traffic needs extra security. However, adding security in the form of encryption can increase delay and potentially affect QoS (since LAN switches and routers can’t decode encrypted portions of IP packets, and hence can’t classify traffic properly).

I) Channel Credibility and Experience

Issue: Will VARs and resellers be able to install, troubleshoot, and maintain LAN PBX systems?

Let’s face it, IT networking staff aren’t particularly knowledgeable about phone systems, and telephony technicians don’t usually know much about data networks. This is why most organizations today employ separate staffs for operation of their voice versus data network infrastructures. If a VAR or reseller offers LAN PBX systems, finding and retaining staff members who are familiar with both data networks and telephone systems will not be an easy task.

Because LAN PBX phones are usually IP devices, and the LAN PBX call server software runs on an NT or UNIX-based server, potential integrators of these technologies could be data networking consultants or staff who come from a data communications background. Unfortunately, many data network administrators have a different concept of system availability than voice system administrators. Whereas the need to take down a server on a weekly basis or reboot a client PC during the day is normal for networking professionals, the idea of rebooting the phone or taking the PBX down for maintenance is not something users will likely get used to. Voice isn’t just another network application that can be installed and maintained along with e-mail, web, and database programs. Voice has its own vocabulary and technology.

Voice resellers are obviously more comfortable with voice technology and understand the resiliency and reliability needed for any phone system. They also have the mentality and business infrastructure in place to support mission critical applications. However, they need to develop skills and competencies in data technologies such as packet networks, bandwidth concerns, switching versus routing, or QoS technologies. They also have the mentality and business infrastructure in place to support mission critical applications. Thus it will be up to the vendors to educate their resellers about IP data networks.
Voice resellers have a long history of service, maintenance, and support of their existing customer phone systems, but it is going to take some time before they can offer the same level of expertise for LAN PBX solutions. In fact, LAN PBX products are sufficiently new that both data networking and voice systems professionals will have some development ahead of them.

Fortunately, there are a small but growing number of VARs from both the voice and data sides that have recognized the opportunity to differentiate themselves by developing and/or acquiring the missing skill sets. These pioneers are the forerunners of a hybrid voice and data channel that is capable of providing and maintaining solutions that bridge the voice and computing environments.

In order to increase the number of VARs with this “hybrid” capability, vendors must educate the VARs on IP technology, identify the skill set requirements, and facilitate their transition. The dilemma facing all vendors and VARs is that this effort must commence immediately, while the return on that investment may be some time off yet.
5. Conclusions

The network and telecommunications equipment industry has significant challenges in driving widespread adoption of LAN PBX solutions, but the momentum towards IP-based network convergence is inevitable. Proof of the value of voice/data convergence in the wide area will eventually cause it to happen in the enterprise. It's a matter of when, not "if" LAN PBXs will become popular.

In order for LAN PBX solutions to compete on an equal footing with traditional PBXs and be accepted by the mainstream market, they must address the practical challenges described in this paper. In particular, organizations evaluating LAN PBX solutions should ask the following questions of the solution vendor:

What compromises in voice system functionality, feature sets, quality, and/or reliability will be required?
Will the proposed LAN PBX solution meet the expectations of my users today?

How will my legacy voice investments be protected through migration, familiarity, and legacy support?

Who is a credible experienced supplier and channel for LAN PBX systems? What are their credentials, and what are their competencies and skill bases?

How will this investment contribute to a reduction in the total communications system cost of ownership, and how will it add value to my organization or business?

Mitel is in the forefront of vendors developing LAN PBX solutions which will solve the challenges described in this paper, without compromising on the functionality, reliability and quality that business users will continue to demand.

We will continue to build on our track record of delivering convergence solutions that simplify, save costs and add value to businesses, and provide an evolutionary approach to convergence that helps protect existing customer technology investments and move towards this goal at a pace that makes sense for their business.
Notes
About Mitel Corporation

Mitel Corporation (NYSE:MLT) (TSE:MLT) is an international voice communications supplier, and a recognized world leader in creating solutions that provide exceptional value to its customers. Mitel’s product portfolio has evolved from world leading voice communications systems, desktop peripherals and semiconductor devices to voice, LAN and WAN networking solutions and a portfolio of Computer Telephony Integration (CTI) products, including micro-electronics, voice operating systems, applications and platforms. The company’s products serve customers across the globe in industries such as finance, education, health care, manufacturing and lodging.

Mitel’s vision is to facilitate enriched and effective communications between people. Mitel foresees the day when communications across distance, time, media and language are facilitated as easily as face-to-face contact.

Founded in 1973, the company’s core competency continues to be real time voice communications. With more than twenty-five years experience, Mitel ranks among the world’s leading suppliers of real-time call control solutions. To meet business demands for network simplicity and cost reduction, voice, as a communications medium, will eventually share the same network as data traffic, becoming just another application within the enterprise. As an early pioneer in computer telephony integration, Mitel has been able to leverage its extensive experience in real-time call control and voice communications to lead the industry in the growing trend toward voice and data network convergence.