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Voice over Internet Protocol

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Voice over Internet Protocol

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Voice over Internet Protocol (VoIP)

Introduction

The human voice is still one of the most effective forms of communication. Therefore, it comes as no surprise that recent advances in voice-driven technology and the ever-increasing need to stay competitive in the business world are driving the development of Voice over Internet Protocol (VoIP) as a dependable union of telephony and data networks.

Many businesses, from large corporations to smaller e-commerce start-up companies, are hoping this emerging technology will ultimately lead to greater efficiency, productivity and cost savings. Not only will VoIP reduce redundant network expenditures, but information will transfer smoothly from: computer-to-computer, computer-to-phone, phone-to-computer, and fax-to-computer. While the benefits of integrating data and voice networks are numerous, the network infrastructure modifications required are often misunderstood. This is especially true when considering data cabling's role in bandwidth management.

What is VoIP?

VoIP is a type of network convergence. Specifically, it is the unification of voice and data onto a single network infrastructure. The idea behind VoIP is to successfully convert digital and analog communications (such as phone calls and faxes) into IP packets to be sent through one network instead of a separate telephony network. The voice signals first become digitized, then are split up into information packets and sent through an IP network.

Benefits of Integrating Data and Voice Networks

VoIP eliminates redundant networks and network resources, simplifies network management and facilitates communication.[1] Consequently, the major advantages of VoIP are cost savings and increased productivity.

- **Reduction of Redundant Networks**—Because typical PBXs (public branch exchanges) are also costly to operate, VoIP would result in substantial savings in infrastructure costs even within a single building or facility.[2]

- **Toll Bypass**—The corporate environment anticipates that convergence will result in savings on long-distance toll calls, especially international voice calls where a substantial part of the cost derives from regulatory fees. In most cases, these surcharges don't apply to circuits carrying data traffic. VoIP would result in a much less expensive way to make voice calls.[3]

- **Diverse Voice Call Routing**—Despite very reliable telephone service, large companies often purchase multiple, diverse circuits to the local telephone company's exchange to act as a reserve system. However, these circuits are rarely utilized and ultimately double the cost per month the company must pay for leased-line telephone access circuits. However, with VoIP, if a failure occurs on the primary telephone circuit at one location, the data network could be used to route calls temporarily to PBXs at other company locations. This eliminates the need for a redundant telephone circuit by leveraging the company's data network.[4]

- **Facilitation of Adds/Moves/Changes**—It's costly for a company to maintain a typical work area that includes data and telephone connectivity, especially when individuals move from one desk to another. With a DHCP (Dynamic Host Configuration Protocol, which enables dynamic assignment of IP addresses to devices on a network), moving a PC from one LAN to another will become simpler due to auto-configuration functionality. However, moving a phone extension from one desk to another (or to another building) is not as simple, usually requiring reconfiguration of the office PBX systems. With the increased popularity of VoIP, a new type of PBX and IP telephone station has entered the marketplace. An IP Phone can automatically configure itself as a DHCP client. Now a desk requires only a single data jack, and the IP Phone can act as a hub or switch to provide a data port for the user's PC. One or more PCs, or perhaps an IP fax machine, can be directly connected to the IP Phone, and all of them can utilize the local IP network. Moving employees from one desk to another is now less costly.[5]

- **Falling IP Equipment Costs**—Packetizing voice puts computing power for data networking equipment near the endpoints of the network, where the packet-switching equipment shows a faster improvement in price/performance than switched equipment. With many companies feverishly developing new features for IP equipment, the price/performance of packet-switching equipment should continue to improve. Packet-switching equipment technology is able to keep pace with the increase in demand for IP.[6]

- **Data Manipulation**—A single network connection on the Web (via VoIP) also allows the Internet to host multiple forms of communication such as voice mail, e-mail and video images instead of using a conventional (and costly) telephone line. Information will be able to be accessed and delivered in several forms. Internet-based information (such as e-mail and e-commerce) can now be accessed by telephone by using voice commands. A desktop computer (in addition to receiving e-mail) can now function as a business telephone and fax machine. Therefore, because of VoIP, organizations now recognize the potential for closer interaction with their customers (and potential customers). For example, instead of breaking a phone connection to transfer a potential buyer to customer service (or worse, losing them to a competitor's customer service rep or Web site), a sales representative can access the appropriate customer service rep or Web page and provide the information while maintaining the original connection.[7]

Converting Voice to Data

In order for an existing TCP/IP network to successfully carry VoIP traffic, several modifications must be made to make the network act like a circuit-switched network.[8]

Traditional voice networks are circuit-switched. They use a dedicated communications path for every call, with a constant bit rate of 64 kbps. However, silence consumes more than half of the average telephone call and if no calls are made, this bandwidth remains unavailable for other traffic. This unused capacity is an inefficient use of bandwidth.[9]

However, this inefficiency is also what makes telephony networks reliable. Compared to a telephony system, a data network is more likely to experience problems such as server crashes or other similar delays. According to Lucent Technologies, telephony problems are rare because 80 percent of voice network issues are resolved *without* human intervention. This is partly the result of telephony networks circuit-switched design and the dedicated bandwidth for each line. Since there's a fixed amount of bandwidth used during a circuit-switched voice call, voice data is never received out of *sequence* at the end of the connection. When data travels over the network it's divided up and transmitted through information packets. If these packets are lost, they can end up being transmitted late or out of order because they took different paths. Today, one of the most important design considerations in implementing voice is minimizing one-way, end-to-end delay. Retransmission is not an option with voice and video traffic flows. It needs to stream in real-time; if there is too long a delay in packet delivery, the data is unrecognizable.

Sequence and real-time transmission make bandwidth management crucial. With VoIP, the data network infrastructure will need to be able to handle more traffic—consistently—than ever before.

Data compression, routing and processing are crucial.

Compression—The bit rate for an uncompressed telephone call approaches 64 Kbps (and uses a single Digital Service, Level 0 channel). This degree of bandwidth is considered excessive, and is usually reduced for delivery over a typical data network. Several encoding and compression algorithms are available to reduce the bandwidth consumed by a telephone call. These compression mechanisms are usually found at the VoIP gateways and not within the data network routers or switches.[10]

Network Quality of Service—A TCP/IP network must have mechanisms in place to prioritize VoIP traffic above all other traffic on the network (except other real-time application traffic such as video). A protocol called Resource Reservation Protocol (RSVP) has been designed to reserve resources across the network for real-time transmissions. Quality of Service (QoS) mechanisms within TCP/IP have also recently been implemented by a number of TCP/IP router and switch vendors. ATM networks and, to a lesser degree, Frame Relay networks have QoS functionality already built into them. Generally, TCP/IP routers and switches will use a priority queuing system to buffer non-VoIP packets and send them only after all of the VoIP packets have been transferred to the next network element. Large IP packets (non-VoIP) are buffered to the side so that the smaller VoIP packets can be sent on time. Other mechanisms will predict times of congestion over the wide area link and throttle back bandwidth demands from non-real-time applications.[11]

IP Packet Precedence—IP precedence bits should be set at the edge of the network, with VoIP traffic given the highest possible precedence. Data networks with protocols other than TCP/IP running on them are not as well suited for VoIP as a purely TCP/IP network because it's more difficult to give traffic priority when it is not a TCP/IP packet. Whenever possible, all bridged traffic should all be segregated from the TCP/IP WAN links where VoIP will flow. Bridging of any sort over the wide area will hinder TCP/IP QoS implementations.[12]

Weighted Fair Queuing—Weighted fair queuing (WFQ) is a buffering mechanism that will buffer TCP/IP packets, classify them based on a number of different criteria, and then de-buffer the packets based on IP precedence or traffic flow. The classifications available are: source and destination address, protocol, and session identifier. During the de-queuing procedure, packets are given privilege based on the three IP precedence bits in the packet's IP header.[13]

Voice Data Processing—Because of voice portals and voice sites in VoIP, voice recognition is now processed on the network server and not on the telephone itself. This allows the system to support millions of calls and to also recognize the various ways callers may state or ask for information. VoIP is handled by a voice gateway dedicated to PC-grade equipment outfitted especially with the hardware necessary for Internet telephony. The gateway is placed between the public switched telephone network (PSTN) and the Internet protocol (IP) network. The gateway assists with the signals between the phone networks, reception of telephone numbers, conversion between telephone numbers and the IP addressing in the IP network and the conversion of voice to packets.[14]

While these applications will certainly reduce data traffic problems, it is doubtful that they alone will address increasing bandwidth demands.

Efficient and Reliable Network Infrastructures

Presently, more than 50 percent of data network problems are caused by the infrastructure.[15] With VoIP, the data network infrastructure will need to be able to handle more data traffic consistently than ever before. Successful implementation of VoIP applications require an efficient and reliable data network infrastructure.

Fortunately efficient and reliable data cabling solutions are available to help manage data traffic better.

The true gauge of a network's performance is its throughput efficiency. *Throughput* refers to the amount of data that is transferred from server to user. If the network is supposed to transmit data at 100 Mbps, but it's transferring data at a lesser rate, it isn't working at optimum levels, and that network's efficiency is compromised as a result. These delays in data transference are usually the result of a network infrastructure that degrades a signal to the point it is no longer intelligible to the receiver. This requires a retransmission of the signal and results in delays and inefficient network performance.

Today, this scenario is relatively common, but with VoIP it won't be acceptable. Retransmissions are not an option with voice and video traffic flows. It needs to stream in real-time—in fact, a delay of more than 200 milliseconds is considered unacceptable. While packet prioritizing and routing applications will help, compression programs often make data signals more susceptible to degradation. This issue will be exacerbated by the increased data traffic and bandwidth requirements VoIP will bring.

Choosing the right data cabling solution requires not just an understanding that throughput efficiency is a necessity, but also that reliability is directly related to headroom. Once the expected data traffic has been estimated, the level of reliability—the amount of headroom—needs to be determined. These two factors determine which high-performance cabling solution is required.

Certain network infrastructures are designed to maximize data throughput well beyond minimum standard requirements. Unfortunately, no data cabling standard adequately addresses the performance requirements associated with VoIP. However there are other sources for information, like Anixter's Levels® Program, which can provide guidance. Anixter's Levels Lab® tests network infrastructures by running real-world applications in real-world networking environments to determine which solutions provide enough data throughput and headroom. Taking the time to ensure that the cabling infrastructure can consistently provide the necessary bandwidth, is crucial to implementing VoIP successfully.

Conclusion

Many businesses are hoping VoIP will ultimately lead to greater efficiency, productivity and cost savings. VoIP eliminates redundant networks and network resources, simplifies network management and facilitates communication.

However, for an existing TCP/IP network to successfully carry VoIP traffic, several modifications must be made to the network infrastructure. While data compression and QoS mechanisms will help reduce data traffic, an efficient and reliable data cabling solution is needed.

Data network reliability concerns are not insurmountable. There are data cabling solutions available that provide efficient throughput and sufficient headroom to ensure that even bandwidth-hungry applications will run smoothly.

Migration and Adoption

Some businesses are already in the process of implementing VoIP.[16] Other companies are preparing for the eventual integration of voice and data networks.

In both cases identifying the right high performance data cabling system can be difficult. No data cabling standard adequately addresses the performance requirements associated with VoIP.

Fortunately, there are other sources for information on throughput efficiency and headroom performance, like Anixter's Levels Program. Taking the time to ensure that the data cabling can provide the necessary throughput and headroom is crucial to implementing VoIP successfully.

Sources

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