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Voice Over IP - Is Your Network Ready?

Carrier Grade Service

When was the last time you called the phone company just to say, "I am just calling to say thank you for my phone service being so reliable?" People have come to take this reliability for granted. The truth is phone companies work to have an uptime of 99.999%. Matching this statistic is a unique challenge to the VoIP (Voice over IP) services being introduced in today's market. With everything moving towards IP, can this medium be used to actually create a carrier grade service?

What defines a carrier grade service? Of the uptime statistics, a true carrier grade service includes Reliability, Availability and Scalability. A "Five 9s" uptime means that the service can only be down 6 seconds per week. In a VoIP environment, one must also add security, manageability and interoperability. The additional factors are necessary as the voice transmissions are carried over network links in conjunction with data services. Voice grade services can not be sporadic. A voice communication must have the continuous available capacity from the beginning to the end of the conversation.

Need to Know Definitions

LATENCY represents a delay caused by the actual transmission. One can think of throwing a ball. The latency would be the time between when it was thrown to when it is caught at the receiving end. Long latency is not acceptable in voice grade transmissions. With high latency, a transmission would be more like an old radio voice transmission where a person would have to say "Over" at the end off each sentence, signaling the other person to speak. In voice networks, delay of less than 150ms is considered acceptable. This delay can be noticed if two people are using a VoIP system with high latency while looking directly at each other. The result is like watching an old Godzilla movie. You will hear the speaker slightly after you watch their lips move.

LOSS is another unacceptable factor in voice transmissions. In a typical voice transmission, a CODEC (Coder-Decoder) changes the voice waves into digitized packets for transmission. The packets are then formed for transmission on the network. VoIP packets carry very small samplings of the voice conversation, typically 20ms. Some loss will not greatly degrade the understandability of the conversation, however a loss over more than 1% will. Loss can occur in congested, overworked, or bursty networks.

JITTER is a hurdle for voice quality. Variations in packet delay cause packets to be received in sporadic patterns or out of order. When VoIP is run on a congested network with bursty traffic, jitter will be harder to compensate for sending and receiving stations. Voice traffic is buffered at the receiving end to help compensate for jitter. In effect, the conversation is stored until enough of the transmission is in memory to play for the receiver. TCP/IP delivers packets based on several routing algorithms. In order for VoIP to work properly, special attention must be paid to not only the paths that the packets take, but also the capacity and health of the network. Excessive jitter makes a conversation undecipherable. New layer 3 switches offer relief as they are able to understand prioritization.

SEQUENCE ERRORS occur when the packets arrive at the receiving station in a different order than they were sent. TCP/IP packets include keys to their position. A sending station, whether it be a PC or any other TCP/IP device, breaks the packets into datagrams for transmission. These datagrams are assigned a sequence number. If datagrams travel different paths on the network due to congestion, hardware failure, or cabling issues cause frequent retransmission, the packets will be received out of order. A high level of sequence errors will cause an audible degradation of the conversation.

Knowing the above factors, how does one plan for VoIP. Traffic is measured in Erlangs. The term is named after the A. K. Erlang, a Danish telephone engineer who pioneered queuing theory. An Erlang is expressed as the number of calls per hour times the duration of the calls divided by 60 (minutes in an hour). This figure also includes call attempts and hold time or any time the channel is busy with call activity. Digital voice signals are compressed which allows one channel to carry multiple lines of traffic. Statistics for Erlangs are generally obtained from the PBX or call manager currently in place. In sizing a network enough bandwidth must be present to address the maximum number of calls that will be loaded on the network. In analyzing network traffic for loading purposes, it is noted that traffic is highly dynamic. Different times of day carry different traffic loads. So how do you shoot this moving target?

Guaranteed Throughput

In a typical IP packet carrying data only, transmission through the network is fairly simple. The data packet is formed, sent and received. Small delays or retransmissions are accepted. But voice traffic can not tolerate errors. Voice packets need a mechanism to move at a higher priority so that their transmission is, in effect, guaranteed. Today, this is accomplished by setting the QoS (Quality of Service) bit in the IP header. All IP headers have a TOS or Type of Service byte. This attribute was built into the protocol several years ago and has more recently been redefined as the DiffServe Code Point (DSCP) field. Quality of Service is a term which refers to a set of parameters for both connection-mode (TCP) and connectionless-mode (IP) transmissions which provide for performance in terms of transmission quality and availability of service. It encompasses maximum delay, throughput and priority of the packets being transmitted. The first bits of the ToS byte reset with QoS information. The syntax is specified by the H.323 standard (ITU- International Telecommunication Union) which governs how audio, video and data communications travel within an IP network. These transmissions may be CCTV, Video Conferencing, VoIP, etc. New standards are being developed using the SIP (Session Initiation Protocol), Media Gateway Control Protocol (MGCP), Magaco/H.248 which will be explained later in this article.

In a telephone conversation over the PSTN (Public Switched Telephone Network) when a person dials a number, a circuit is set up between the sending telephone and the receiver. This circuit is available for the entire conversation until the telephone is placed on the cradle. The circuit is maintained even through silence on the line. In IP transmissions, there is not a circuit, per se. The packets are routed over a network to the receiving workstation. If the call is to be placed outside of the network the conversation will still be routed to the point that the PBX or Gateway puts the call on the PSTN. The gateway typically is accompanied by a gatekeeper that handles the prioritization of voice and video traffic where transmissions are assigned a higher priority to address the necessity for quality at the receiving end.

If a network had unlimited bandwidth potential, Quality of Service would not be an issue. In today's networks, however, there are transmissions of documents, applications, email, backups and a myriad of other data. Adding voice services requires some attention. To build QoS into a system one must address construction of the framework. Applications and appliances must be able to set and understand the QoS bit, or recognize the other transfer mechanisms if using another standards. It would be rather pointless to build the capability into a system if a network's edge devices such as routers did not understand or prioritize these requirements.

Traffic Cops

There are several applications called traffic shapers on the market today. The applications work by acting much like a traffic cop. They will slow "less important" traffic and allow priority movement of datagrams that have a higher priority. The problem with these applications, however, is that they offer a single point of failure in a network. They may have issues with scalability for networks over a certain number of devices. Routers and Layer 2 switches throughout a network that understand this bit will provide both redundancy and in the near future multicast abilities. In Layer 2 type switches, IEEE has developed two standards (802.1p and 802.1q) to address quality of service.

A Layer 2 802.1p compliant switch has the ability within the MAC layer to group LAN packets according to their traffic class. There are eight classes defined which network managers map to their specific applications. Priority seven is the highest value and is generally used for router to router communication and path information within the network. VoIP, videoconferencing and other delay-sensitive applications will use values five or six. The lower numbers are used for other types of traffic from multimedia all the way down to what is classified as "loss eligible" traffic. These switches also must understand multicast traffic. In the application mapping, a value of zero indicates that no value has been set.

It is up to LAN administrators to determine the settings of these bits in their applications. With new layer 3 switches and routing capabilities, this task can be done on an address level within the router tables reducing or eliminating the need for traffic shapers and complicated mapping tables. The 802.1q standard allows network administrators to break up larger LANs into smaller segments or VLANs (Virtual Local Area Networks).

DiffServe-Another QoS Technique

Another means of identifying QoS is through setting the DiffServe (Differentiated Services) bit (this is actually a better mechanism by design). The first bits of the TOS (type of service) byte or DiffServe Code Point (DSCP) field in the IP header are set using one of three per hop behavior mechanisms. A hop is defined as packet movement from forwarding point to another (router to router, router to switch, etc). The advantage of this bit is that it is understood by all Layer 3 devices throughout the internet including routers and Layer 3 switches. It will allow traffic to travel via various paths until it reaches its destination. This bit is set as a request by the initiating application. In order to assure end to end quality, all routers and switches within the given paths must adhere to this setting and forward the packets accordingly. The bit can be reset at each domain as the packet enters that domain.

There are five categories of service covered in the DiffServe Code Points:

1. Relative Priority Marking
2. Service Marking
3. Label Switching
4. Integrated Services/Resource Reservation Protocol
5. Static per-hop Classification

Other Standards

SIP - Session Initiation Protocol

SIP proposed by the IETF (Internet Engineering Task Force in RFC 2543) is an application layer protocol to overcome the limitations of H.323 QoS and DiffServe as described above. H.323 operates in a connectionless mode where no end to end session or "circuit" is created for the duration of the conversation. SIP handles setting up the sessions and encompasses user location, availability, redirection, and multiparty (conference) abilities in Layer 7 (the application layer) of the OSI protocol stack. Further, it allows VoIP gateways, PBX's and other communication systems devices to interoperate making it far more scalable. This is accomplished by means of the following:

- RSVP (Resource Reservation Protocol)
- RTP (Real-time Transport Protocol)
- RTSP (Real-time Streaming Protocol)
- SAP (Session Announcement Protocol)
- SDP (Session Description Protocol)

This protocol has less overhead as it reuses the same header information from HTTP. Also, because it is connection oriented, multicast, unicast and other connection dependent conversations are more reliable than they would be with dependency on QoS only. Name mapping and redirection are built into the protocol making it possible to have one URL (Universal Resource Locator). It is now possible to dial a person's phone by keying in their email address.

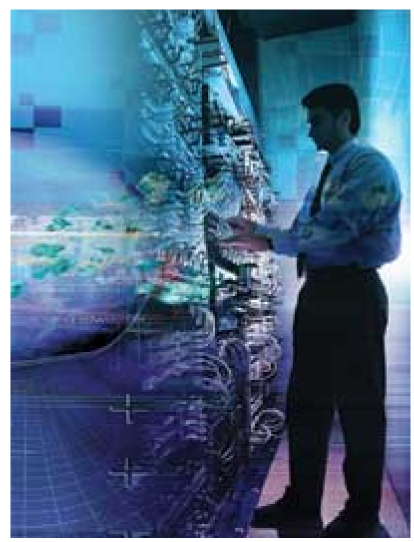
MGCP - Media Gateway Control Protocol

This standard is proposed by the IETF in RFC 2705 for the convergence of audio signals carried over the Public Switched Telephone Network to datagrams which travel over the internet. It addresses the way Media Gateway Controllers (MGC) and media gateways communicate using SGCP (Simple Gateway Control Protocol) combined with IPDC (IP Device Control). It actually works in a master/slave configuration. This standard is in line to be replaced by Megaco/H.248.

Megaco/H.248

This standard is co-sponsored by the ITU-T Study Group 16 and a subset of the IETF called Megaco. This standard specifies specific protocols for all devices and divides the gateway functions into subcomponents. This standard also allows for devices to function with "clocked" type devices found in the PSTN (Public Switched Telephone Network) to provide more affordable call switching techniques.

Voice Over IP - Is Your Network Ready ?



There is no fast answer for this question. At this point, the first thing to do is plan. Determine your requirements, the health of your network and infrastructure, the vendor with whom you will be working, and the priorities of your applications on your network. Moving to VoIP is much like building a house. It won't be worth much if the blueprints aren't well designed. Even the best plans can suffer when the hardware is not speaking the same language. It is not just as simple as putting a VoIP system on an existing network. There may be a need to replace older electronics, certainly if hubs exist in the network due to their lack of intelligence and antiquated technology. Older versions of Layer 2 and Layer 3 switches may need to be replaced if they do not understand how to route your packets properly.

Other key factors to consider are whether your systems are open enough to allow you a variety of phones for later connections to your network. There are certainly cost advantages, particularly when a company has multiple offices in multiple states. IP Telephony is a non-tariffed service, meaning there is no toll charge for the calls. This makes the return on investment for these systems quite attractive. It would be in your best interest to call your phone service provider and see what VoIP services they are planning to offer in the future months. The latest trend on the carrier side is to become an ELEC (Ethernet Local Exchange Carrier) which would eliminate your need to supply the gateway to the PSTN, as this would be provided by your phone company. If they plan to offer this service, find out what standard they will be supporting prior to making your decision.

Revisit Your Infrastructure

Years ago, anything would run on CAT5 cabling, phones would run on CAT3 or worse. The infrastructure became an installed and "forgotten" entity in the network environment, unless of course an expansion or failure occurred. Electronics vendors will request a certain category of copper or fiber, but what people often forget is that all connectivity and network cabling is not equal. Further, a network can be crippled by an improperly or poorly installed infrastructure. Someone gave me a great analogy to this problem, "You can buy faster, more expensive cars, but if the road is in disrepair they won't last very long." And certainly, they won't be able to drive at the speeds to which they are capable. This is also true for your network.

New installations are generally CAT6 or Augmented Category 6, and CAT7 is already here. 10G over copper standards will be published in 2006 and 10G over fiber is already approved. For more information on these standards, look at our 10G *ip*™ product information.

There are many more applications that play on your network playground, with the numbers growing daily. The cost savings to companies are making these applications increasingly popular. What steps do you take? First, you should have a certified installer or certified infrastructure auditor look at your cable plant. Terminations, cabling runs and labeling should be compliant with ALL of the appropriate standards. A good protocol analyzer or network analyzer can point out potential problems. It is also important to make sure that your cabling meets today's standards. Companies that invested early in the emerging cabling standards may not have cabling that meets current standards.

Next, you will want to examine your electronics and the general health of your traffic patterns. Make sure that your electronics will understand QoS, DiffServe, or whatever mechanisms used by your VoIP system. Your vendor should be able to assist you with this configuration. Get rid of unnecessary protocols that are using up your bandwidth. Finally, and most importantly, monitor your service after it is in place. There is no substitution for proper planning and network administration. It makes sense to put in the best cabling, connectivity and devices available to assure smooth operations for the future.

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