Transmission of Delay Sensitive Data over Non-QoS Enabled WANs

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***Abstract*—Vendors of Voice Over IP (VOIP) and other delay sensitive systems have always marketed their products with the stipulation that performance could be drastically affected unless end-to-end Quality of Service (QoS) mechanisms were in place to guarantee delivery under minimum latency and jitter conditions. For Wide Area Network (WAN) services, this usually means transmission over a single vendor, QoS enabled, Multiprotocol Label Switching (MPLS) private network.**

Broadband and dedicated Ethernet Internet connections provide a cost effective alternative to expensive, single vendor QoS enabled MPLS private networks. The caveat to using these services is that there is no QoS guarantee and all traffic is treated as best effort. Generally, companies use an encrypted tunnel to move their sensitive data over these networks. Over the past few years, providers of these connections have increased the bandwidth of their last mile, backhaul and backbone links. This growth in bandwidth has made these links a possible transport mechanism for delay sensitive traffic, such as voice and video.

This paper discusses the deployment of non-QoS enabled WAN links and the testing of VOIP and Video Teleconferencing (VTC) over those links. Quality metrics are presented that illustrate the capability of these networks to support transport of delay sensitive data. Quality metric comparisons between QoS enabled WAN links and non-QoS enabled WAN links are presented as well. Price savings and bandwidth information are also presented to illustrate the economic advantages of using non-QoS WAN links.

***Index Terms*— Assured Forwarding, Broadband Internet, CLEC, DSCP, dedicated Ethernet, Expedited Forwarding, ILEC, IPDV, Jitter, MPLS, metro-Ethernet, PHB, QoS**

I. INTRODUCTION

Transmission of delay sensitive data over Internet Protocol (IP) networks has been around for over a decade. VOIP and VTC vendors have always pressed their customers to deploy end-to-end QoS capability over their WAN before the vendor would validate operation of their VOIP or VTC systems. The solution to an end-to-end QoS enabled WAN normally meant a deployment scenario that consisted of a single vendor MPLS network that provides QoS service.

Service providers of these QoS enabled WANs normally transport the QoS marked traffic over a MPLS enabled private backbone network. These private networks normally perform as their Service Level Agreement (SLAs) warrant, but the caveat is the additional cost associated with each QoS enabled circuit. The charges for the QoS service may add an additional 30-50% to the monthly cost for each circuit.

The other issue Network designers face is that QoS hand-offs or peering between service providers is non-existent at the writing of this paper. Account representatives from four of our service providers indicated that there is no QoS hand-off or peering service offering. Three of these providers are considered tier one providers. The providers indicated that if the service exists, it is most likely a custom priced service that would tend to be higher than a single vendor private network.

Dedicated Ethernet, broadband-Ethernet, metro-Ethernet and other services delivered via an Ethernet hand-off on a dedicated circuit have become very attractive in their pricing over the past two to three years. Providers are often able to deliver a 4, 5 or even a 10 Mb/s service that is lower in cost than a QoS enabled MPLS T1 circuit. The cost effectiveness of these services has made it a worthwhile venture to determine if delay sensitive traffic can be transported over these non-QoS links while still maintaining an acceptable level of quality that is comparable to service delivered over a QoS enabled, MPLS private network.

For clarity purposes, this paper will refer to Ethernet services delivered over a dedicated circuit as ‘dedicated Ethernet’. Service delivered over a cable modem, Digital Subscriber Loop (DSL) or other circuit that has a shared bandwidth aggregation point in the backhaul network will be referenced as ‘broadband’ service.

The remainder of this paper discusses the design and deployment of a hybrid WAN that consists of non-QoS and QoS enabled WAN links along with the subsequent testing of VOIP and VTC services over that WAN. The results from measurements made on the non-QoS connections are directly compared to the measurements made on QoS enabled MPLS connections.

Organization of the paper is as follows: Section II Discusses the enterprise WAN architecture and presents test results from lab simulations that were made before deployment of non-QoS enabled service. Section III covers deployment of dedicated Ethernet service to branch offices. Section IV covers deployment of broadband service to small branch offices. Section V presents SLA quality measurements of QoS enabled MPLS links and non-QoS enabled links while section VI provides conclusions and future directions.

For clarity purposes, jitter, as defined in this paper, refers to the inter-packet delay variation (IPDV) referenced in Request for Comment (RFCs) 2679 and 3393 [1]. The term ‘jitter’ is commonly used by network equipment vendors to reference the measurement of IPDV and also used in network metric analysis tools. In this paper jitter does not refer to the variance in two timing sources of a synchronous transmission system. RFC 5481 [1] presents a formal applicability statement discussing packet delay variation

II WAN Architecture

Our WAN architecture is a very common one used by enterprises throughout the world and entails branch offices that are connected to hub sites (hub and spoke) where centralized storage, voice, email, Internet and other business services are provided. The WAN links are encrypted using encapsulating security protocol (ESP) or another method and the hub site quite often has a failover or back up site for continuity of operations (COOP). This typical WAN is shown in Figure 1. The aforementioned encryption is generally a router to router tunnel using Internet Security Association and Key Management Protocol (ISAKMP) for the key exchange and either a pre-shared key or a RSA key self-generated on each router. The underlying data encryption is usually Internet Security (IPSec). For COOP, each branch router has one tunnel to the Corporate Campus hub and second tunnel to the back-up site. The corporate campus and backup site also communicate over encrypted tunnels. Routing protocols, such as Open Shortest Path First (OSPF), Border Gateway Protocol (BGP) or Intermediate System to Intermediate System (IS-IS) can be used with varying metrics to force the branch offices to use either the Corporate Campus network or the back-up site as the default route to the Internet. A more detailed illustration of the dual tunnel concept is shown in Figure 2.

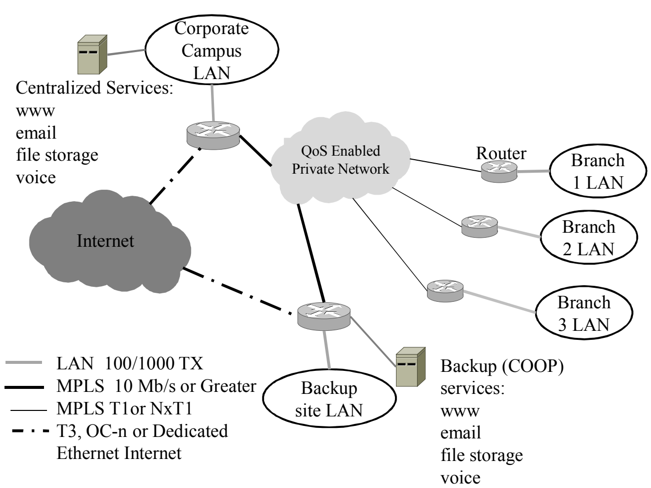

Fig 1: Typical WAN with Dual Homed Branch Offices

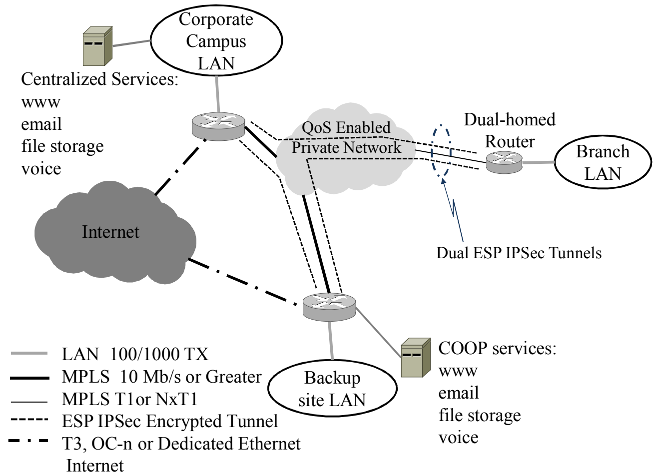

Fig 2: Typical WAN Branch Office with Dual Tunnels for COOP

The WAN architecture of interest consists of the aforementioned WAN, and additionally, branch offices with non-QOS enabled Internet links. The private network MPLS connections support site-to-site QoS capability while the non-MPLS connections support best effort traffic transport. The non-MPLS connections may consist of dedicated Ethernet connections or broadband connections from DSL or Cable operators. The private network MPLS connections consist of T1 for smaller offices, NxT1 for medium offices and T3, OC-n or dedicated Ethernet connections for larger offices. All private network links support end to end QoS. This ‘hybrid’ enterprise WAN is shown in Figure 3. All branch offices in this architecture employ dual encrypted tunnels. One tunnel connects to the corporate campus and one to the back-up site.

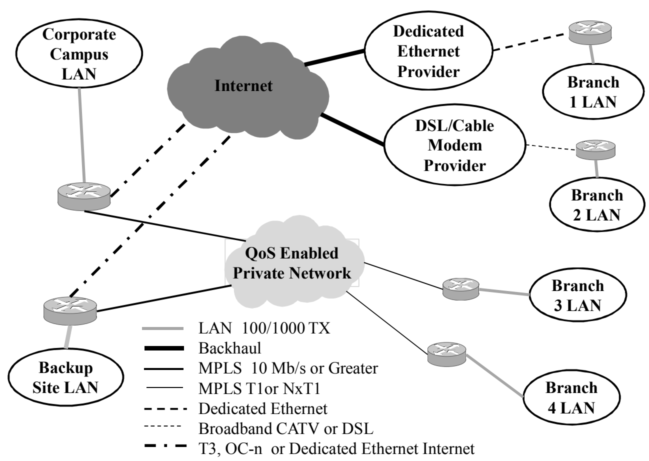

Fig 3: Dual Homed Hybrid WAN using QoS and Non-QoS Connected Branch Offices

For purposes of this paper, only tests between branch offices and the corporate site will be discussed. Presenting test results to the back-up site only serves to illustrate the back-up site effectiveness which is not the focus of this paper. The simplified diagram is shown in Figure 4.

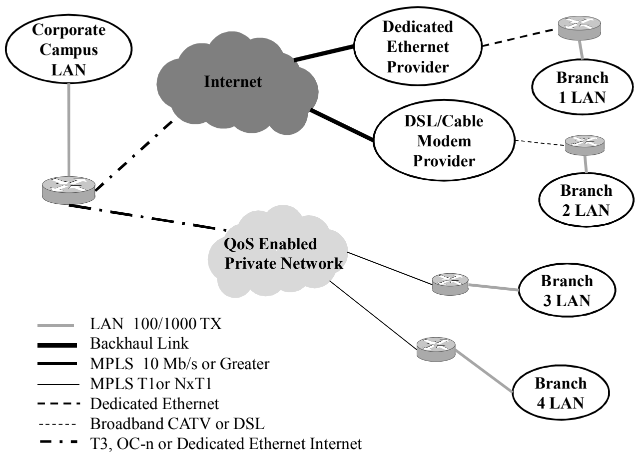

Fig 4: Hybrid WAN with Private Network and Internet Based WAN Links

Transporting delay sensitive services, such as voice and video, over non-MPLS links with no QoS parameters has always been frowned upon by vendors of the applications and supporting hardware of those services. For instance, the minimum requirements most IP phone vendors had dictated in the past was that for acceptable voice quality, the Round Trip Time (RTT) latency should be less than 300 ms along with jitter of less than 20 ms and packet loss less than 1%. ITU-T G.114 [2] recommends a one way transmission delay of 400 ms maximum and further stipulates that delays below 150 ms (RTT=300ms) should be acceptable for most applications.

In implementation of transporting delay sensitive services over the WAN, our private network provider guarantees a one-way latency of less than 40ms over their backbone with less than 0.05% packet loss and average jitter of less than 1ms. A four level QoS classification (class 1-4) queuing is used on our routers to match the QoS service offered by our service provider. Class based policy settings are used by our routers to properly classify and queue traffic to match the four class levels of our provider. Class 1 is reserved for delay sensitive voice RTP traffic while class 2 marks real time video and VOIP call set-up. Class 3 is reserved for Secure Shell (SSH) or other important management protocols while class 4 contains the majority of traffic and is classified as 'best effort'. Packets matching class 1 are placed in a strict priority queue that is sized for the maximum amount of VOIP traffic expected over that particular link. During link congestion, packets in the strict priority queue are guaranteed the bandwidth allocated for that queue while packets in the other queues may be dropped. If the strict priority queue bandwidth is exceeded, then those packets are dropped on a tail-drop first basis.

Packets are marked by the end device, such as an IP Telephone or VTC unit, using Differentiated Services Code Point (DSCP) assignments in the TOS byte of the IP header. When these QoS marked packets arrive at the router, they, along with other traffic, are matched to a service policy on the router for proper output queuing. Voice packets are set to a DSCP of 46 (Expedited Forwarding or EF) and placed in the class 1 strict priority queue. The VTC and call set up use a DSCP of 34 (Assured Forwarding or AF equal to 41) and are placed in the class 2 QoS queue. The class 3 queue matches DSCP 18 (AF21) while class 4 traffic is set to a DSP of zero and is classified as 'best-effort'. RFC 2597 covers DSCP and assured forwarding per-hop behavior (AF PHB) in more detail and RFC 3246 covers Expedited Forwarding (AF EF) or DSCP 46 [1]. Table 1 illustrates traffic classification for outbound traffic on each WAN connected router.

Table I: Traffic Classification

Traffic Type	DCSP (AF PHB) Value	Provider QoS Queue
VOIP RTP	46 (EF)	1
VTC and VOIP Call Set-up	34 (41)	2
Management	18 (21)	3
Normal (best effort)	0 (0)	4

Before attempting to convert a private network office to a dedicated Ethernet or broadband service, we simulated some worst case conditions in a lab environment. The test consisted of running bi-directional TCP stream tests in conjunction with a VOIP RTP IP phone call over a simulated WAN connection. The simulated WAN consisted of a bandwidth controllable T1/FT1 and a latency/packet loss server. The T1/FT1 bandwidth was controlled via an integrated router with a T1/FT1 CSU card. The IPSec tunnel from the lab router terminates on the same router as all other branch offices thus providing a test environment very similar to a branch office WAN link. The latency/packet loss server is a desktop PC running Nistnet, a WAN/LAN emulation application developed by NIST. Figure 5 illustrates the lab network.

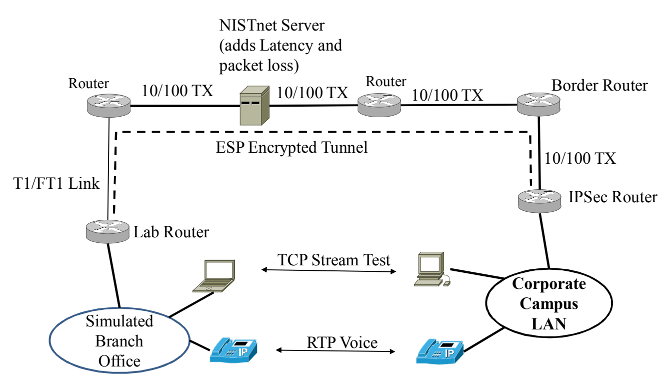

Fig 5: Simulated Branch Office Network

While the simulated network does not account for congestion experienced at aggregation points and Points of Presence (POPs) on the Internet, it does provide the ability to simulate a congested link and the ability to determine if packet classification and prioritization is working correctly. Most importantly, it aids in determining to what degree RTP voice and video streams can exist in a poor WAN environment.

Our worst test case came as a result of an overseas office in an underdeveloped country where our employees desired the ability to talk to family and friends back home. The office WAN link is a 17 hop transit with the last hop being a 256 Kb/s satellite link. The WAN link has 0.5% to 1% packet loss and a one way latency of between 330 ms and 430 ms. Before agreeing to deploy an IP phone in the office, a simulation test was conducted by setting the T1/FT1 to 256 Kb/s (4 channels on a T1) and programming the NISTnet server to high values of RTT latency and 0.5% packet loss. Table II illustrates the results of two tests.

Table II Voice Test on simulated 256 Kb/s over a Satellite Link

One way Latency	Packet loss	Jitter (IPDV) avg/max (ms)	MOS* avg/min
350 ms	0.5%	2/120	4.2/3.9
400ms	0.5%	3/167	3.9/3.4

* Mean Opinion Score Listening Quality (MOS-LQ)

Even with the high latency, average jitter and MOS scores were acceptable for VOIP. The one caveat on the actual voice conversation was that the latency created conditions where the talkers were 'stepping' on each other. After using it a few times, the users became accustomed to the delay and adjusted their conversation accordingly. Presently, there are two G711 IP phones deployed at the office. Future plans are to replace them with G722 capable IP phones. The employees can be reached via a 4-digit extension from anywhere on the enterprise network plus they have the ability to call loved ones back home by dialing 9 to get local telco dial-tone and then dialing 7 or 10 digit numbers according to the North American Numbering Plan (NANP) or country code and number for International calling..

III: Dedicated Ethernet Service Deployment

Dedicated Ethernet services have become readily available in many areas as a result of large fiber deployments by competitive local exchange carriers (CLECS), incumbent local exchange carriers (ILECs), cable operators and other third party suppliers. The resulting competition over the past few years has made dedicated Ethernet pricing very attractive. In many instances, the cost per megabit per second is much less than private network MPLS T1 costs from ILECs and traditional long distance carriers. Quite often the best value service is offered by the CLEC, cable operator or other 3rd party provider. The ILECs have a tendency to be higher priced and require a longer lead time for service installation when compared to the CLECs and other 3rd party providers. For our WAN, every dedicated Ethernet non-QOS enabled WAN connection is provided by a CLEC, cable operator or 3rd party provider. In all instances, the installation interval was less than 45 calendar days from the agreement signature date.

The major obstacle to overcome with non-QoS enabled WAN connections is determining if delay sensitive traffic can be transported across the connection while still providing the user with a reasonable expectation of quality and performance. The key was to determine if the link could support VOIP and VTC before signing a long term (12-24 months) agreement. The process that we used was to verbally negotiate and perform a test with the provider's technical support center before establishing an agreement for service. Most of the prospective providers were cooperative in regards to the test. Generally, they provided a public IP address of an edge router that would either terminate our service, or an aggregation router that was one or two hops from the edge router.
In cases where we could not perform tests, the prospective providers produced network information that aided in determining if VOIP and VTC could be run across the network.

The network tests were straight-forward and often used by network administrators on a daily basis. . Traceroute was used to measure hop count and User Datagram Protocol (UDP) latency while Internet Control Message Protocol (ICMP) echo request (ping) was used for latency, latency deviation and packet loss. These tests were run from several hosts on the corporate LAN as well as one host from the back-up site. The ping tests were done with small packets and large packets to obtain an idea of latency for small voice packets and large video packets respectively.

Exact jitter measurements could not be made since there were no VOIP or VTC endpoints in place and we could not pull statistics from the provider's router. Instead, jitter was estimated using ping statistics. Although this method is rudimentary, since ping statistics are based on round trip information and affected by possibly two paths, it did provide an estimate of jitter. To gain a level of confidence in the provider's network, we specified to the provider that we were going to transport VOIP over their system and that jitter was a concern. The provider normally responded with internal jitter statistics and or a number of companies currently running VOIP over their network.

True jitter measurements were only able to be made after deployment of the non-QoS enabled link. Before moving forward with test results, a brief discussion of jitter, or inter-packet delay variance as referenced in RFCs 2679 and 3393 [1] is in order as shown in the following:

If packet one is transmitted at t_1 and received at time r_1 and packet two is transmitted at time t_2 and received at time r_2 then their transport delay (latency) time, td_i is given by:

$td_1 = r_1 - t_1$: latency of packet one (1)
$td_2 = r_2 - t_2$: latency of packet two (2)

The instantaneous jitter between the packets would be the difference in packet latency times or:

$$J = td_1 - td_2 \quad (3)$$
$$= [(r_1 - r_2) - (t_1 - t_2)] \quad (4)$$

Note that $(t_1 - t_2)$ is the constant inter-packet transmit rate, TR, and is normally 20ms for VOIP packets. The term, $(r_1 - r_2)$, is the instantaneous jitter and is equal to the deviation in arrival times from the normal inter-packet transmit rate of 20ms or:

$$J = r_1 - r_2 + TR, \quad \text{where } TR = 20\text{ms for VOIP} \quad (5)$$

Since TR is constant, then the only effective variable causing differences in arrival times at the receiver is the network path. The instantaneous jitter may then be referenced as the difference in arrival times between two successive packets.

$$J = r_1 - r_2 \quad (6)$$

For transmission of delay sensitive information, such as VOIP, it is important that receive jitter buffers account for $r_1 - r_2$ but not so large as to interfere with the real time interactivity. These same criteria also hold true for video tele-conferencing over IP.

To gain an understanding of the network path's characteristic and relation to jitter, a brief review of random variables is necessary. In the following, a stable network path refers to the physical links an IP packet will traverse when being routed from sender to receiver.

The one way latency over a stable network path may be described by a Gaussian distribution with mean μ and standard deviation of σ. It follows that packets transmitted over that network path would exhibit a similar Gaussian distribution describing the latency with a mean of μ and standard deviation of σ. The arrival time of the packet at the receiver would have a 99.7% probability of arriving in the time interval given by:

$$[\mu - 3\sigma, \mu + 3\sigma] \quad [3], [4], [5] \quad (7)$$

So that the arrival time of any particular packet, r_i has a 99.7% probability of having a latency in the range of:

$$\mu - 3\sigma \le r_i \le \mu + 3\sigma \quad (s) \quad [3] \quad (8)$$

For two standard deviations, packet r_i has a 95.4% probability of arriving in the range given by:

$$\mu - 2\sigma \le r_i \le \mu + 2\sigma \quad (9)$$

For a stable network path, the latency of any packet is described by a random variable with a Gaussian distribution as noted above. Successive packets can also be described by the same random variable with mean μ, and standard deviation, σ. The difference in time between two successive packets would be described by the difference between the two random variables. Since they have the same distribution function, they have the same mean and that difference is zero, however the standard deviation is additive and this gives an instantaneous jitter equal to twice the standard deviation as shown by (10):

$$\mu - 6\sigma \le J \le \mu + 6\sigma \quad (10)$$

Expression (10) indicates that a large standard deviation in arrival time differences could adversely affect the play-out buffer in a typical VOIP call. A deviation of 10ms could equate to a jitter of at least 120ms (± 6 σ) and affect the ability for the audio or video stream to be played back at a continuous rate at the receiving end point

Receiver jitter buffers are normally designed to account for jitter as well as the inter-packet transmit rate. It is important that receiver jitter buffers for VOIP and video conferencing equipment be small enough to allow for quick play-out of the received packet as well as be large enough to account for jitter as described above.

The aforementioned discussion on jitter also underscores the importance of outbound service policies on routers that place delay sensitive information in priority queues. Even if the service provider is not giving precedence to QoS marked packets, outbound service policies on the routers at each end of the path can minimize latency and jitter by giving transmission precedence to delay sensitive packets. Our design follows these criteria. All VOIP and VTC endpoints mark their traffic according to previously discussed QoS values. All WAN connected routers prioritize the marked packets such that VOIP packets are given priority. VOIP call set-up and VTC sessions receive the next highest classification. Important network management protocols such as SSH are given the next priority level and the last classification holds best effort traffic, such as Web browsing, file transfer and other non-real time protocols.

Our present enterprise WAN configuration consists of a total of 15 WAN connected offices. Seven of these offices have dedicated Ethernet connections that replaced QoS enabled, private network MPLS T1 or NxT1 service. As previously mentioned, dedicated Ethernet service not only provided significant bandwidth increases, but also provided cost savings ranging from 10% to over 50% per month compared to a QoS enabled MPLS T1 from our existing service provider.

The measures of success for the non-MPLS connections were the SLA statistics that were recorded and compared with the QoS enabled MPLS connections. Table III lists SLA measurements of latency and MOS for non-MPLS connections. Further QoS and non-QoS comparison data is presented in section IV.

Table III: Dedicated Ethernet Cost Savings and SLA Information for VOIP

Office	Connection Bandwidth Mb/s	Monthly Savings (US $)	Percent Bandwidth Increase	RTT Latency (ms)	MOS
8	10	$771	64%	34	4.3
10	5	$10	40%	38	4.1
11	4.5	$111	63%	42	4.1
12	4	$344	63%	37	4.2
13	5	$346	70%	27	4.0
15	5	$903	10%	84	4.0

Thus far seven offices have been converted to a dedicated Ethernet service. As indicated in Table 3, the cost savings have been considerable while also realizing a significant bandwidth increase.

IV: Broadband Deployment

Broadband service has only recently been marketed as a business solution. Up until the past few years, DSL and Cable modem service were marketed as residential consumer solutions. With the upgrade to Data Over Cable Service Interface Specification (DOCSIS) 3.0 in the cable TV market and larger geographical penetrations of high speed flavors of DSL, Cable companies, Telcos and some 3rd party providers are marketing broadband service as a small business solution. Another catalyst in this movement has been the improvement in the backhaul link from the aggregation points to IP PoPs. These links have been improved from T1 and NxT1 service to Ethernet, packet over Synchronous Optical Network (SONET) and other high speed backhaul technologies.

The same type of test procedure used for dedicated Ethernet deployment was also used for broadband deployment. One of the difficulties in determining the ability of a broadband link to support VOIP or VTC is the unknown backhaul bandwidth and the number of shared subscribers on the same backhaul link. Testing to these sites was usually done over the course of several days and at different times during the business day. The tests were relatively short, but they helped in determining any wide variation in SLA performance. All broadband offices were able to support IP telephony using G711 or G722 CODECS.

VTC testing focused on the quality of conference room VTC units running H.263 with frame rates up to 30 frames/sec and a bandwidth of approximately 480 Kb/s (audio and video combined).

V: QoS SLA Comparison Measurements

Measurements of SLA information in the form of RTT, Jitter, Packet loss, MOS and user experience were recorded for comparison of QoS enabled, MPLS switched, private links versus non-QoS service delivered over dedicated and broadband links. In the case of VTC, user visual experience was also recorded.

Test data was created from the following sources:

- SNMP data pulled from network equipment running SLA tests once per minute
- Network Performance applications that measure Jitter (IPDV), MOS, Latency, traffic profile, and other SLA information
- IP Phone per-phone call information (latency, jitter and MOS) available on each phone screen display during an active call.
- Lab testing that involved using TCP stream tests to create congestion on a T1 WAN link and the subsequent affect on traffic policing.

Table IV presents data from the SLA measurements made on several QoS enabled MPLS private WAN links while Table V shows data on non-QoS enabled links consisting of dedicated and broadband connections. Note that in some instances, performance over the non-QoS links was better than QoS enabled links. Jitter measurement data is from the hub site to the branch site. Jitter measurements from the branch sites to the hub site produced similar results, even in the case of asymmetric bandwidth found in broadband links.

Table IV: QoS Office SLA Data over 1 month period

Office	Link	RTT (ms)	Jitter (ms)	MOS
1	T1	27	1.6	4.3
2	T1	55	1.7	4.3
3	T1	33	1.6	4.3
4	T1	47	1.9	4.3
5	T1	48	2.2	4.3
6	T1	45	1.6	4.3
7	T1	28	1.6	4.3

Table V: Non-QoS Office SLA Data over 1 month Period

Office	Link (Mb/s)	RTT (ms)	Jitter (ms)	MOS
8	10	32	1.6	4.3
9	5dwn/3up*	40	1.6	4.2
10	5	38	2.1	4.3
11	4.5	44	1.7	4.3
12	4	39	1.5	4
13	50dwn/10up*	42	1.8	4.3
14	5	28	1.6	4.3
15	5	79	1.8	4.2

* Broadband office with cable modem

MOS, jitter and latency statistics were also recorded for individual calls on QoS enabled and non-QoS enabled links. This active call information is available on many models of IP phones. This information can be compared to the network equipment statistics in tables IV and V. The first series of tests were performed over the simulated lab branch office network where latency and packet loss were controlled via NISTnet. Further data was recorded from calls on QoS enabled and non-QoS links. Generally, the calls were left up over a 30 minute period. Average and worst case call statistics were recorded. The data appear in Table VI.

Table VI: Audio Calls over Simulated Lab, QoS enabled MPLS and Non-QoS Links (Statistics from remote office)

Office	Avg Latency (ms)	Max Latency (ms)	Avg Jitter (ms)	Max Jitter (ms)	Avg MOS	Min MOS
Lab 160*	164	169	11	54	4.4	3.3
Lab 60**	64	70	6	44	4.2	3.9
QoS office 1	25	76	6	46	4.8	4.2
non-QoS office 12	52	158	7	73	4.4	3.8

* Simulated Lab link with 160 ms delay, 0.5% packet loss

** Simulated Lab link with 60 ms delay, 0.5% packet loss

VTC tests were a combination of subjective observations plus SLA statistical information. There are many variables to consider and measure in a VTC comparison. For our tests, a standard VTC conference unit using H.263 video and G722 audio was used at each end. The tests were done with one person slowly moving at each end during a 10 minute session. The idea behind the continuous movement was to maintain a higher frame rate and provide a degree of network 'stress' to the VTC session. Table VII illustrates the VTC test results.

Table VII: VTC Comparison QoS vs non-QoS

Office	Audio avg jitter (ms)	Video avg jitter (ms)	Frame/s (fps)	*Subjective measurement 5=HD 1=Poor
3 (QoS))	0.8	10.9	23.3	3
15 (non-QoS)	5.3	11.9	29	3

* Subjective measurements are done on a scale similar to MOS (1 is poor, 5 is broadcast quality TV at 30fps))

VI: Conclusion and Future Efforts

Based on the test data presented in this paper and actual deployment of non-QoS enabled dedicated Ethernet and broadband links, transmission of delay sensitive data over non-QoS enabled links can be deployed effectively in the corporate enterprise. Corporate users may expect to receive the same level of service on these WAN links as they have come to expect on QoS enabled private links provided proper outbound queuing design, as discussed in this paper, is implemented.

As previously indicated there are currently no ISPs performing QoS hand-offs with other ISPs. Without QoS handoffs between ISP networks, enterprise WAN connectivity over the Internet will continue to be based on best effort transport for the near future. CLEC and other 3rd party entities that offer dedicated Ethernet service also provide an economic replacement for expensive private WAN links. Enterprise WAN planners may consider these non-QoS enabled connections as a possible replacement for expensive, QOS enabled, private network links.

New technologies designed for use on multiple broadband links that monitor link latency and jitter through continuous types of SLA measurements are finding applications in offices where several broadband technologies may be available. Instead of purchasing the more expensive dedicated service, multiple broadband links are used to transport office traffic. VOIP and VTC packets are sent over the link with the lowest latency and jitter while data insensitive to latency and jitter, such as Web and file transfer traffic can be queued and sent as best effort on any of the broadband links. Also, with the advent of 4G and Long Term Evolution (LTE) networks, consideration for their use as a broadband alternative for office connectivity will most likely depend on cost and availability.

In networks where traffic volume may have a cost basis, such as those found on cellular networks, or burstable wired links with average data limits, the link that provides the best QoS is selected for data and voice. This link is most likely to be the more expensive path as well. Delay insensitive traffic, such as Web browsing, messaging and file transfer are selected for transport over less expensive links that most likely have higher latency. New methods for link selection based not only on the available networks, but the end user criterion for each transaction have been proposed [6]. Under this scenario, QoS is no longer a static policy implemented in a router or MPLS switch, but a dynamic process where user based needs determine the QoS for each particular data transaction.

Other studies that go beyond static QoS implementation have proposed the use of application aware transport equipment so that routing decisions for path selection can be made using application based processing. Information Transfer Data Services (ITDS) is a suggested future Internet architecture that proposes the transport network is application aware instead of just processing raw data at the network and transport layers [7][8]. Although QoS policies allow hop-by-hop checks at the network and transport layer of the OSI model, future architecture such as ITDS would allow the ability to provide end-to-end QoS through layer 7. This is unlike today's architecture where only the end devices have service intelligence and reliance on static QoS policies at the network and transport layer dictate how traffic is transported.

Unfortunately most Internet traffic is still transported on a best effort basis. Cisco predicts that by the end of 2015, 90% of all Internet traffic will be video [9]. Although not all of this video is necessarily real time, even streaming video needs higher precedence than file transfer or Web browsing traffic and most everyone has experienced a poor video teleconference with poor audio and pixilated video. QoS mechanisms other than best effort transport are needed for future Internet communications.

References:

[1]. RFCs:

2597: “Assured Forwarding PHB Group”, Standards Track, June 1999.

2697: “A One Way Delay Metric for IPPM”, Standards Track, September, 1999.

3246: “An Expedited Forwarding PHB (Per Hop Behavior)”, Standards Track, March, 2002.

RFC 3393: “IP Packet Delay Variation for IPPM”, Standards Track, Nov. 2002.

RFC 3550, “RTP A Transport Protocol for Real Time Applications”, Standards Track, July 2003.

RFC 5481, “Packet Delay Variation Applicability Statement”, Informational, March, 2009.

[2] International Telecommunication Union Recommendation G.114, *One Way Transmission Time;* (2003) pp 2.

[3] Oleg Berzin, “Jitter and Some Math”, found at: http://www.ccieflyer.com/pdf/Jitter%20and%20Some%20Math.pdf

[4] Hugh D. Young, *Statistical Treatment of Experimental Data*, New York, McGraw Hill, 1962, pp 75.

[5] B.P. Lathi, *Modern Digital and Analog Communication Systems*, 2nd ed., Saunders College Publishing, 1989, pp 385 to 387

[6]Amit Sehgal and Rajeev Agrawal, “QoS Based Network Selection Scheme for 4G Systems”, IEEE Transactions on Consumer Electronics, Vol. 56, No. 2, May 2010, pp560 to 565.

[7] S. Paul, J. Pan and R. Jain, “Architectures for the Future Networks and the Next Generation Internet: A Survey,” Comp. Communications., vol. 34, no. 1, 15 Jan. 2011, pp2-42

[8] S. Balasubramaniam, K. Leibnitz, P. Lio, D. Botvich and M. Murata, “Biological Principals for Future Internet Architecture Design”, IEEE Communications Magazine, No. 7, Vol. 49, July 2011, pp 44-52

[9] Cisco Systems White Paper, “Entering The ZettaByte Era”, Cisco Systems, San Jose, CA, June 1, 2011, Paper can be found at http://www. cisco.com

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