Making the Case for Testing Ethernet Links

Introduction

This paper describes the parameters to test on an Ethernet link and explains why they must be tested. Specifically it describes the four key parameters that affect any Ethernet or Internet Protocol (IP) link: throughput, frame loss, latency, and jitter. Then it explores the issues that cause these problems as well as the proper testing methodologies used to ensure successful deployment of an Ethernet metropolitan (MAN) or wide area network (WAN) link.

Ethernet Testing Parameters

Whether it is a 10 G link across core routers, a 1 G link across an enterprise, or a 512 kbps link over a satellite, the Ethernet link requirements remain the same. The customer wants a reliable link at an affordable price without any disruptions to the service. To guarantee that link, these three key parameters always must be maintained:

Throughput

Tests whether the full pipe is allocated. For example, if one deploys a 100 MB pipe but can only get a committed information rate (CIR) of 50 MB through it, a problem exists. If one has a 10 G link but can receive only 1 G through the link reliably, the rate is unacceptable.

Latency (Round-Trip Delay)

Refers to the time it takes for a packet to reach its destination and return. This type of testing has been popularized by the ping test—an inefficient method for testing the link, the reasons for which will be discussed later. In any event, it is clear that long delays will negatively affect the end-user experience (though sometimes unavoidable, as is typically the case with satellite links).

Frame Loss

Occurs when frames sent from one end are not received at the other end. Because Ethernet runs on a physical layer, the reliability of the Ethernet link is important for the higher layers of the OSI model and can drastically impact them. For example, a packet loss of 1 percent can degrade the Transmission Control Protocol (TCP) utilization by as much as 80 percent; thus, a small proportion of frame loss can drastically affect the quality of experience (QoE) for the end user.

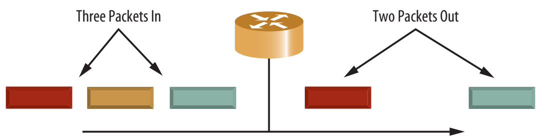

Figure 1 Frame loss

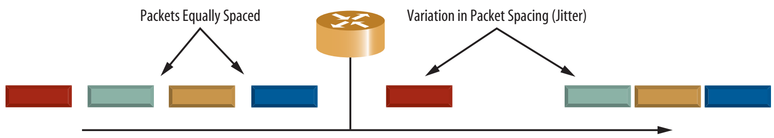

Figure 2 Packet jitter

The last parameter to test is the presence of *packet jitter* on the network. Packet jitter has little effect on a data-only network; however, excessive packet jitter will disrupt service on a network carrying IP television (IPTV) or voice over IP (VoIP) traffic. Higher layer protocols cannot use late arriving packets. For example, a phone call with packets that arrive too late to reach the speaker's ear are dropped, resulting in degraded voice quality.

Testing the Key Parameters

It is important to conduct all tests *at the maximum CIR* when testing an Ethernet link. Problems rarely occur when running the link at 0.1 percent utilization or less (as a continuous ping test often does) but rather occur at 100 percent utilization. Improper prioritizations, wrong path setups, auto-negotiation problems, poor network planning, duplex issues, and even poor cabling or network equipment rarely displays problems at low utilizations. True evaluation of the link quality can be determined only when a network is properly stressed. Therefore, the first step for testing is to ensure that the committed information rate is being delivered properly. While a customer is happy to purchase a 10 Mbps link and receive 100 Mbps, network planners and traffic management individuals do not like giving away bandwidth. Similarly, the customer who purchases a 100 Mbps link and only receives 10 Mbps is certainly going to be unhappy. Therefore, the first step of any type of testing is to determine whether the link is delivering the proper throughput.

Once the throughput has been determined, the next step is measuring the frame loss rate of erred frames traveling across the network to determine whether the traffic is going through error-free. Because frames with an erred frame check sequence (FCS) are dropped by a switch or a router, erred frames going across a network will be dropped before they arrive at the end destination. Therefore, to determine the reliability of the link each frame must be generated with a sequence number. Incoming frames are then checked for the proper sequence number and dropped or erred frames are detected, thus determining the frame loss ratio of the link.

Figure 3 Packet error rate

The final two tests, latency and jitter, can determine whether the arriving valid frames will be useful for the application layer and whether the buffering and prioritization qualities of service (QoS) are properly set up in a live network. Network elements will buffer traffic in excess of the CIR; for example, from A to B span in Figure 3 the buffers fill up (A to B in Figure 4). Once the traffic falls below the CIR (span B to C in Figure 4), the buffers gradually empty; however, if the buffer fills to the maximum (at D in Figure 5), the extra traffic is dropped and packets are lost. More importantly, as these buffers fill and empty, the jitter and round-trip times should be tested to ensure that they remain within acceptable levels. Where QoS policing exists, this test is conducted using traffic with differing QoS (VLAN Priority bits or type of service [TOS]/Differentiated Services Code Point [DSCP] bits) to determine if some traffic types are successfully prioritized over other types.

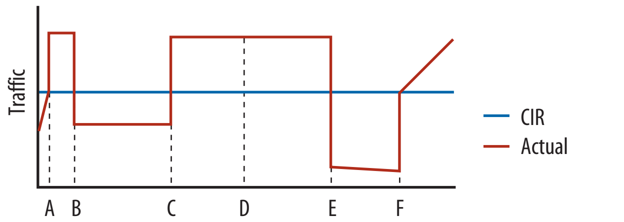

Figure 4

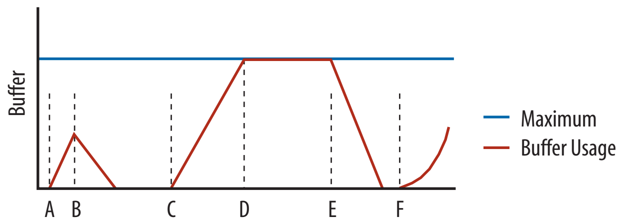

Figure 5

Summary

As mentioned previously, the first step is to run all tests at the maximum throughput expected for the link. For example, if the maximum throughput is 10 Mbps, run the test at 10 Mbps and any lower value will signify a problem. As for acceptable packet loss, round-trip time and jitter results depend on the network under test or the service level agreement being measured. Some standard results are a packet loss of <0.01 percent; a round-trip time of <16 ms on a short link, <100 ms on a transcontinental link, and <550 ms on a satellite link; and jitter of <20 ms when carrying VoIP traffic. Therefore, requirements for networks such as wireless backhaul are often more stringent, whereas networks such as T1 emulation over DOCSIS® (data-over-cable service interface specification) are often less so. In the end, it is imperative for each network to be tested to its unique requirements using the proper methodology described above.

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 30162666 000 0209 TESTETHERLINKS.WP.ACC.TM.AE February 2009