RTT TECHNOLOGY TOPIC
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Packet shaping / Traffic shaping protocols Part 1

There has always been a substantial gap between PSTN performance and Internet performance. PSTN end to end delay is typically 35 to 40 milliseconds, internet end to end delay is typically 350 to 400 milliseconds - ten times the time.

This does not matter when delivering delay tolerant content. It does matter when delivering time sensitive content - voice or real time image, video and application exchange.

Packet network delay is the delay introduced by packet capture, routing and switching, buffering and queuing. Packet capture is the time taken to process an entire packet before forwarding to a router. Routing and switching delay is the time taken to check the header and routing table, queuing and buffer delay is the time spent by packets waiting in router buffers while routers deal with other packets - a process that typically takes 20 to 30 milliseconds.

One of the reasons for moving to IPV6 is to improve router performance. IPV4 has a variable length header and variable length payload. This is flexible and minimises address overhead but means that a router never quite knows what to expect.

IPV6 defines a standard header size of 40 bytes. The 40 bytes are divided down into 8 fields rather than the 14 fields used in IPV4. The defined header size and reduced number of fields means router latency can be reduced and packet shaping/traffic shaping protocols more rigorously applied.

There are three primary packet shaping/traffic shaping protocols - RSVP sits at the edge of the network (for instance in Windows 2000) and defines four levels of service, **high quality/application driven** for applications that can declare their resource requirements - for example MP4 encoded 'declarative content'; **medium quality** where the application identifies the type of traffic flow needed, for example isochronous or non-isochronous but letting the network determine priority; **low quality network driven** - basic latency bounds and a minimum bandwidth guarantee and **best effort**. Multi Protocol Label Switching (MPLS) is then used to break the packet stream into fixed length cells, grouping packets within an IP session into a single flow which can be tagged to optimise router throughput. (A definition of a flow is a sequence of packets treated identically through a 'possibly complex' routing function - the idea is to pass down long lived flows, for example multi-media rich media streams to be switched by hardware). Finally, Diffserv is used to define four levels of service - platinum, gold, silver, bronze - and used as a mechanism for grouping traffic flows sharing similar QoS attributes. MPLS and Diffserv used together will typically reduce queuing delay from 20 to 30 milliseconds to 5 milliseconds.

Delay is however only part of the story. An additional parameter is delay variability (also known as jitter). Delay variability is a consequence of packet loss triggering send again protocols and is therefore related to the provisioning of buffer bandwidth. As traffic has become more asynchronous (increasingly bursty), buffer bandwidth has become increasingly harder to dimension. Essentially, bursty traffic can be accommodated by over provisioning buffer bandwidth - a 2 Mbyte buffer at 60% usage delivers a 4×10^{-2} packet drop rate, a 64 Mbyte buffer reduces the drop rate to 3×10^{-6}. Note that you could use UDP (user datagram protocol) to hide the packet loss (UDP allows packets to drop while TCP/IP requires dropped packets to be sent again) but dropped packets are bad news for differentially encoded rich media.

And there's the snag - packet routed networks promise greater bandwidth efficiency but need to deliver similar dynamic range to existing network topologies, ie to support real time rich media, packet shaping protocols have to deliver an order of magnitude improvement on existing Internet latency performance. The jury is still out as to where this is achievable. Even if it is, bandwidth efficiency will be little better than existing circuit switched networks.

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