




**Worst-case latency in 802.1Qav
Ethernet bridges
(v2 – some early conclusions)**

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Change history

V	date	updates
1	31 jan 08	original version, class A only, no “observation interval”
2	11 may 08	validation of assumptions, where “class observation interval” is needed, more extensive conclusions

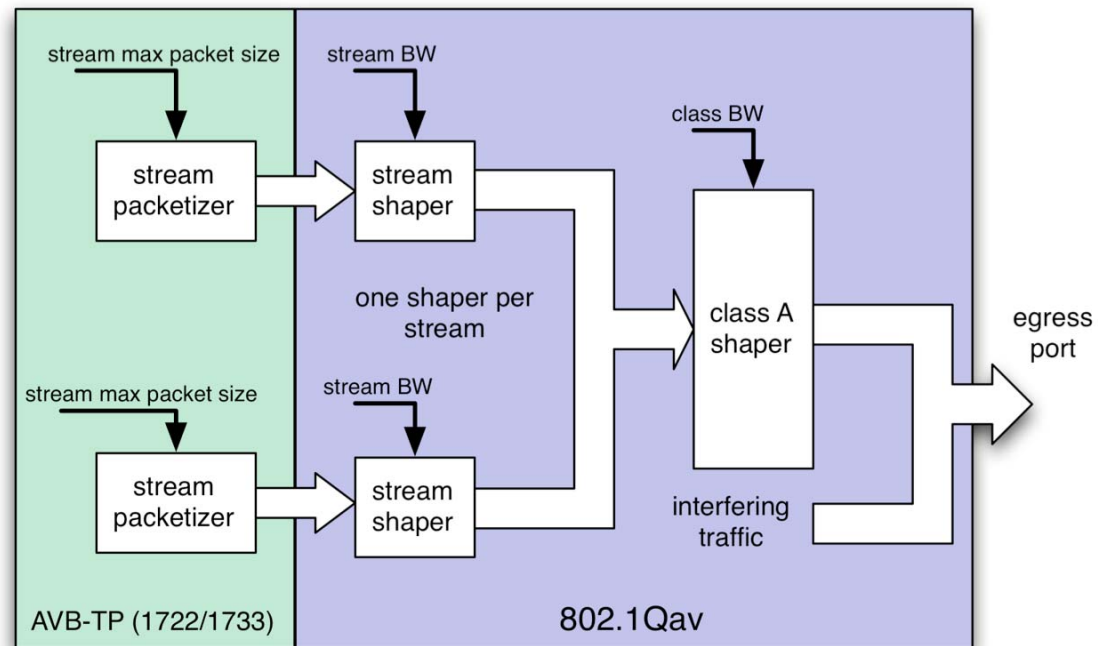
Notes

- This calculation is only for Class A
 - I want to make sure we understand the limits on a “2ms” latency network
 - Once we understand that, then I’ll add the Class B traffic to the analysis
- The parameters to be explored include:
 - Network topology (number of bridges and number of ports on each bridge)
 - Stream packet limitations (max packet size)
- All shapers are as described in Qav 0.3

Input, output & methodology

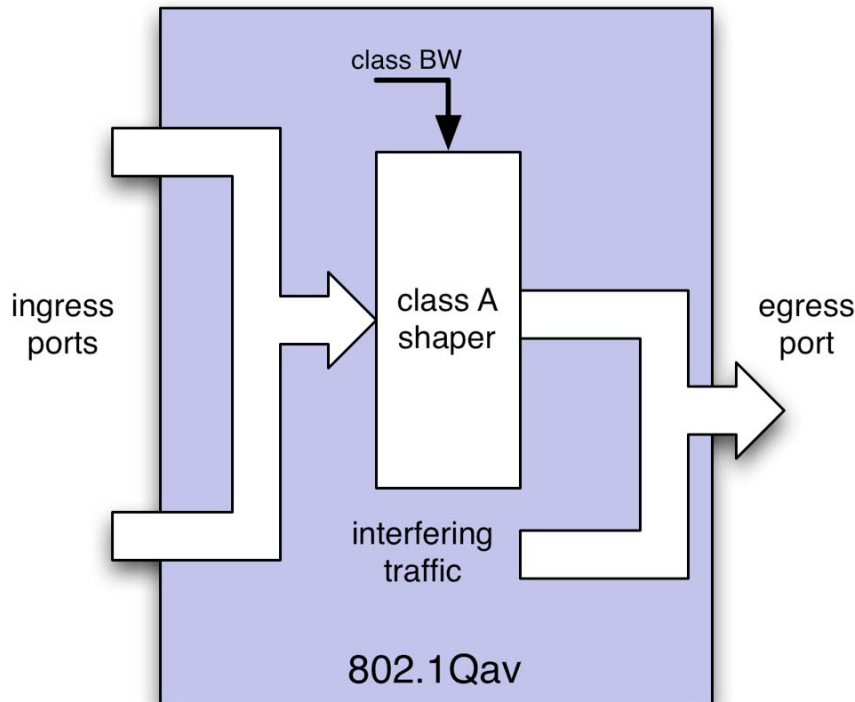
- The input parameters to be explored include:
 - Network topology (number of bridges and number of ports on each bridge)
 - Stream packet limitations (max packet size)
- Output is worst case delay
- Looking only at first order effects
 - mention will be made of 2nd order effects that are being ignored for now

Talker model



- Talker consists of data stream packetizers feeding into stream shapers feeding into class shaper
 - stream packetizer has a max packet size parameter
- Stream shapers have infinite “sendSlope”
 - has the effect of smoothing per-stream traffic, so there is no bunching within a stream
- Sum of all stream’s “idleSlope” is the class “idleSlope”
 - SRP bandwidth allocation is “idleSlope”

Bridge model



- Same as a talker with no stream shapers
 - conversely, a talker can be thought of as a bunch of single stream sources each with an infinitely fast link to a bridge

Talker delays

- Talker has just started to transmit a best effort frame of b bytes
- There are m streams, each with a max packet size of s_j bytes

- Egress port rate is e bytes/sec

- Delay is
$$\frac{\left(b + \sum_{i=1}^m s_i \right)}{e}$$

Bridge delays

- Bridge has just started to transmit a best effort frame of b bytes
- There are m ports, each routing class A traffic with a max packet size of s_j bytes through the egress port
- Egress port rate is e bytes/sec

- Delay is
$$\frac{\left(b + \sum_{i=1}^m s_i \right)}{e}$$

Network delays

- There are n bridges
 - so there are $n+1$ devices for queuing delays
- For each hop between devices there is no common stream
 - so it's possible for a stream to always be delayed by new interfering packets on each hop

- Delay is
$$\sum_{j=0}^n \left(\frac{\left(b_j + \sum_{i=1}^m s_{ij} \right)}{e_j} \right)$$

Simplification

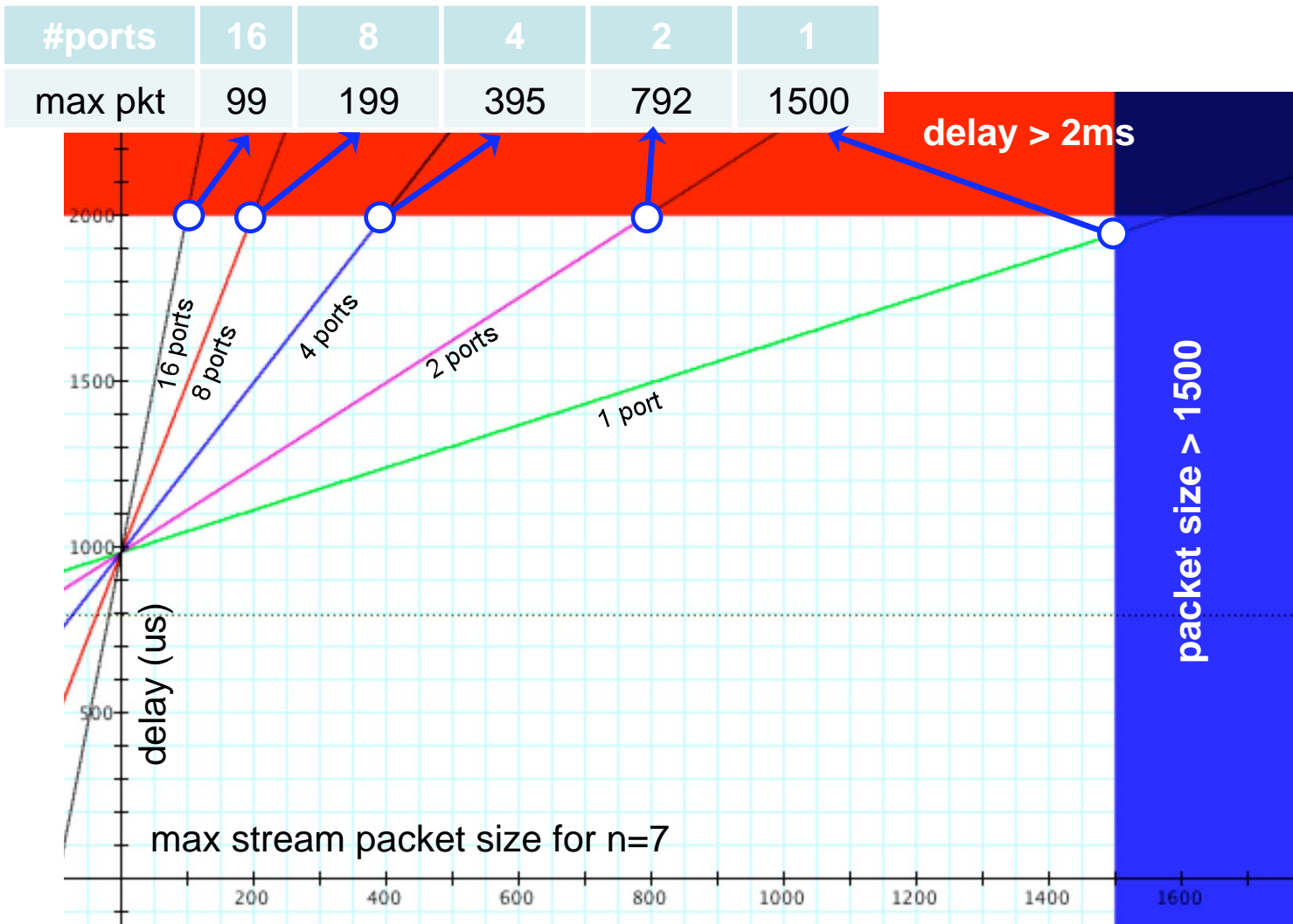
- all links are 100 Mbit/sec (e_j)
- worst case best effort interfering packet of 2000 bytes (b_j)
- all other class A packets are the same size (s_{ij})
- the talker launches m streams and each bridge has $m+1$ ports
- so, delay is

$$\frac{(n + 1)(b + ms)}{e} = \frac{(n + 1)(2048 + ms)}{100}$$

2nd order effects

- Cumulative “bunching”?
 - Does not appear to be an effect, at least for networks carrying a single AVB class
- Remember, no analysis given to multiple classes!

Max network delays



Conclusions (part 1)

- Class 5 max packet size directly effects the latency, as does the number of bridges in a path from talker to listener, as does the number of ports on those bridges
 - We have been assuming 7 hops is a good limit for class A at 2ms max delay.
 - So we need to assume limits for the number of ports on the bridges and the max packet size
- For a 7 hop 100 Mbit/sec Ethernet configuration, we should perhaps assume 8 port bridges are a maximum
 - If so, then class 5 packets need to be no larger than about 200 bytes
- SRP *can* allow larger packets, but it will have to be ready to deny requests even when there is bandwidth available on a egress port
 - because the latency budget of “250 usec/bridge” is used up

Changing the simplifications

- This is all a bit too restricting
 - We end up with very large latencies unless we restrict everyone's packet sizes

- It's also not necessary

- The important value is the amount of interfering traffic:

$$\left(b_j + \sum_{i=1}^m s_{ij} \right)$$

- We can't limit the best effort packet length, but the sum of the stream packet lengths could be under control

So

Using a measurement interval (part 1)

- “Class measurement interval” is useful to determine the max packet size parameter of the stream packetizer
 - packet size for stream $s_{ij} \leq a_{ij}c$ where
 - a_{ij} is the allocated bandwidth for a stream
 - c is the class measurement interval
- Since the total allocated bandwidth must be less than 75% of the egress rate: $\sum_{i=1}^m a_{ij} \leq 0.75e_j$
- We then know that $\sum_{i=1}^m s_{ij} \leq 0.75e_jc$ or about 1180 bytes for 100Mb/s

Using a measurement interval (part 2)

- Using a class measurement interval allows stream bandwidth (available to SRP) to be used directly as an analog for stream-induced delays so that SRP can relate max delay to max bandwidth

Alternate simplification

- all links are 100 Mbit/sec (e_j)
- worst case best effort interfering packet of 2000 bytes (b_j)
- all class A packets uses the same class measurement interval of 125us

- the talker launches m streams and each bridge has $m+1$ ports

- so, delay is
$$\frac{(n + 1)(b + \sum_{i=1}^m s_{ij})}{e}$$
$$= \frac{(n + 1)(2000 \times 8 + .75 \cdot 100 \times 10^6 \cdot 125 \times 10^{-6})}{100 \times 10^6}$$
$$\cong (n + 1)0.000254us = 2.03ms \text{ for } n = 7$$

Conclusions

- If a class measurement interval is used to force frame sizes to smaller size, then the max delay is not dependent on the number of ports on a bridge, just on the number of hops
- A 125us measurement period for class A will guarantee ~ 2ms worst case delay over 7 hops