QoS on Low Bandwidth High Delay Links

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Agenda

- QoS Some Basics
- What are the characteristics of High Delay Low Bandwidth link
- What factors of applications are affected
- Behaviour of High Delay on Data Streams
- Behaviour of High Delay on RTP or voice Streams
- QoS considerations in such environment
- Test Setup & Some Test Results
- Implementation strategies
- Conclusions

QoS Basics

1. TDM Systems
   - Dedicated resources end to end
   - Whatever comes in has a guarantee to go out
   - No resource congestions
2. Data Systems
   - Packets come in & go out of different interfaces depending to lookup in Data devices
   - Oversubscriptions offers commercial advantages
   - Packet exit rate may be more than the physical capacity of the port
   - Resulting in resource congestions

QoS Basics …contd.

- Data devices use buffers to manage congestion within some limits
- FIFO (First In First Out) is one of the simplest buffering strategy
- Buffer size is function of
  - Link Speed
  - Protocol
- FIFO results in Best effort environment

QoS Basics …contd.

- What happens when some traffic needs to given special treatments on various vectors like
  - Bandwidth / Throughput
  - Delay
  - Loss
  - Jitter
- We come to QoS environment
- QoS is “Managed Unfairness”

QoS Basic … Throughput / Bandwidth

- Throughput / Bandwidth has different connotations for different people
  - Physical layer – Bit rate
  - Application Layer – FTP throughput – User view
  - Host to host aggregate flow – Administrator view
  - Network to network aggregate flow – Operator view

QoS Basics …. Delay

- Various contributors to delay are:
  - Serialization delay (fixed)
  - Propagation delay (fixed)
  - Queuing delay (variable)
  - Forwarding/processing delay (variable)
  - Shaping delay (variable)
  - Codec delay (fixed)
  - Compression delay (variable)

QoS Basics … Loss

- Loss of packets or information can be attributed to
  - Bad Link Quality
  - Resource crunch (Congestion)
  - Results in bad user experience
    - Application run slower
    - Lesser throughput

QoS Basics … Jitter

- Variation in the arrival rate of data packets that were transmitted in uniform manner
- Different from the Delay
- Mostly prominent in the voice application (isochronous traffic)

QoS Basics… Summary

- QoS involves, giving service deliveries on following vectors
  - Throughput / Bandwidth – guarantee of required availability
  - Delay – As minimum as possible
  - Jitter – As minimum as possible
  - Loss – No loss

High Delay Low Bandwidth links

- Transport of such links are characterized by high delay
- Satellite links
- Generally have low bandwidth
- Low bandwidth is subjective from end application perspective
- A 400 Kbps links is low bandwidth from SP perspective, but is very high bandwidth from Enterprise perspective
- End Application drive requirements

Affecting factors

- Impact of Delay
  - User experience
  - Pronounce on the TCP based application
- Impact of low bandwidth
  - Bandwidth to be efficiently used
  - Giving high priority to most critical applications

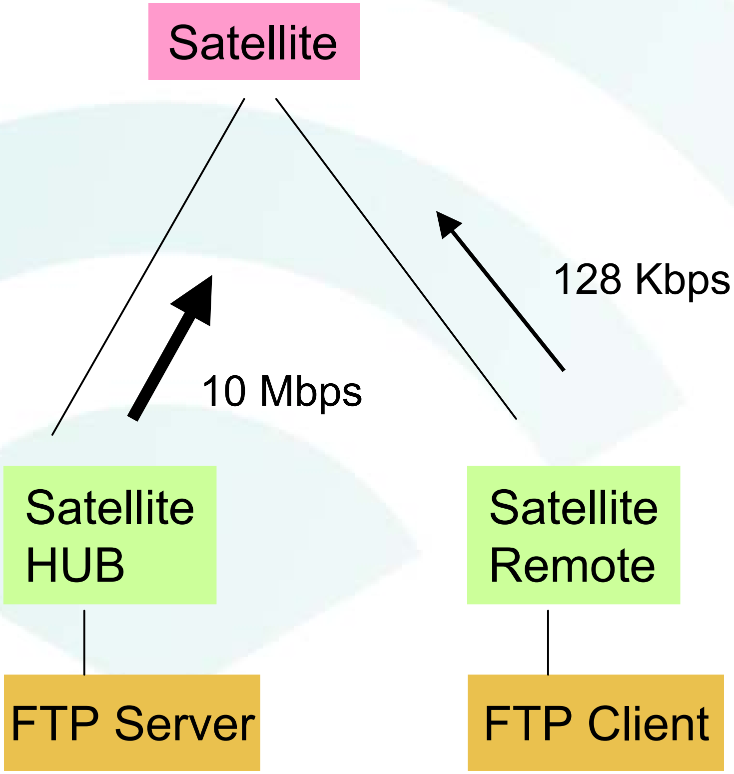
QoS Considerations …

- RTP or voice traffic streams
  - Should have zero drop rate
  - Minimum queuing delay
- TCP or Data traffic streams
  - TCP windowing mechanisms need to be fine tuned
  - Can have drop rates, as the end systems will allow retransmissions

Effect of Delay on Data or TCP streams

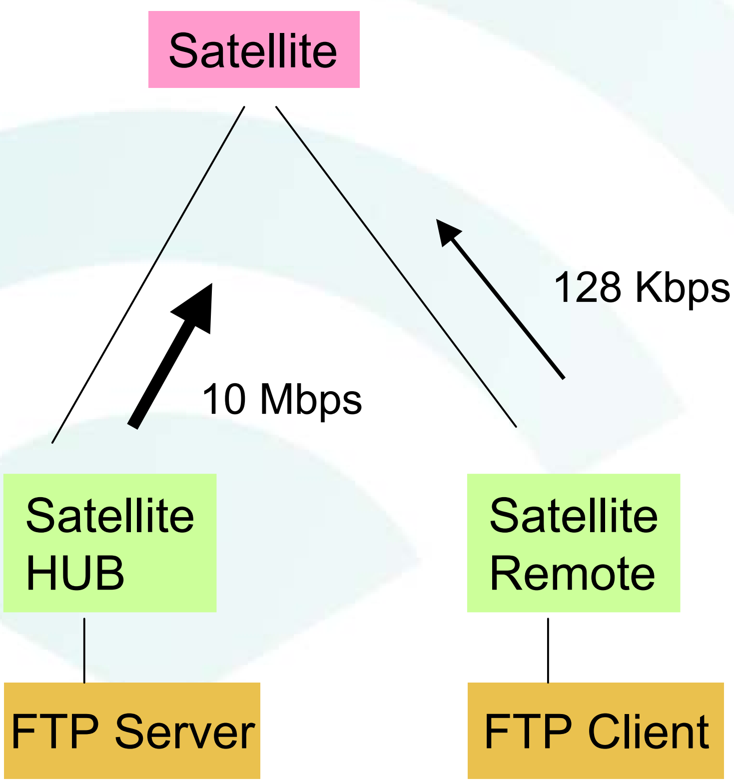
- TCP involves acknowledgment mechanism
- So the round trip time comes into picture
- On High Delay links the round trip time (RTT) will be higher
- Higher the RTT slower will be acknowledgment & feedback mechanism
- Will result in slower data transfer & lower application throughput
- Will consider an example

Illustration of lower throughput



- Forward bandwidth available in 10 Mbps
- Reverse – 128 Kbps
- If single hop RTT is 500 ms
- If the TCP window on the End TCP systems is 16 KB
- FTP client wants to download file from FTP server

Illustration of lower throughput (Contd…)



- Theoretical throughput formula is
  - T = (Window Size) / (RTT)
- Putting the values
  - T = 16 KB / 500 ms
  - T = 256 Kbps

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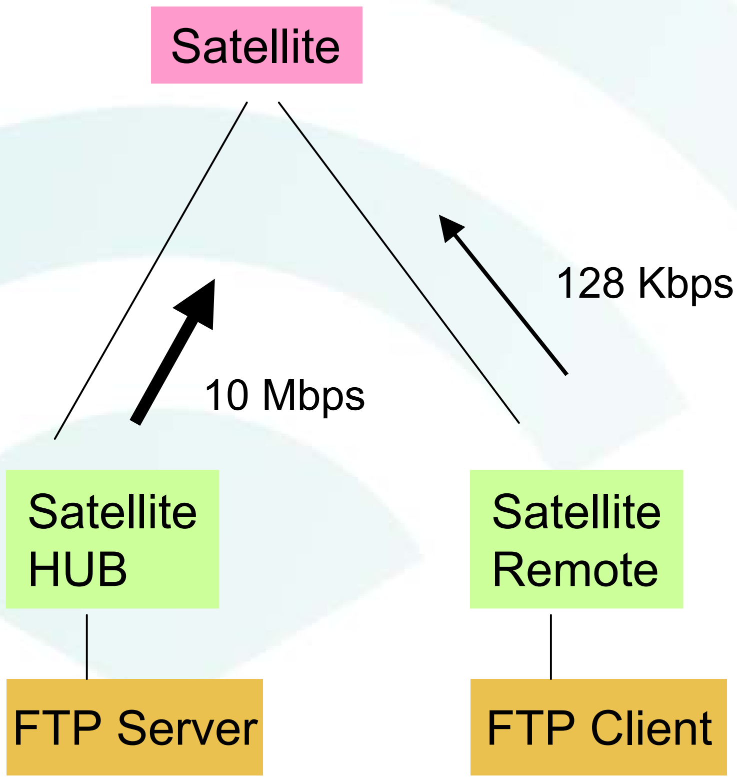
Illustration of lower throughput (Contd.)



- So Even if
  - the whole of 10 Mbps forward path is fully free
  - The servers & clients is fully idle (I.e with no cpu/memory crunch)
- A single FTP session cannot pump traffic beyond 256 Kbps, in the said environment
- Only 2-3 % of capacity being used

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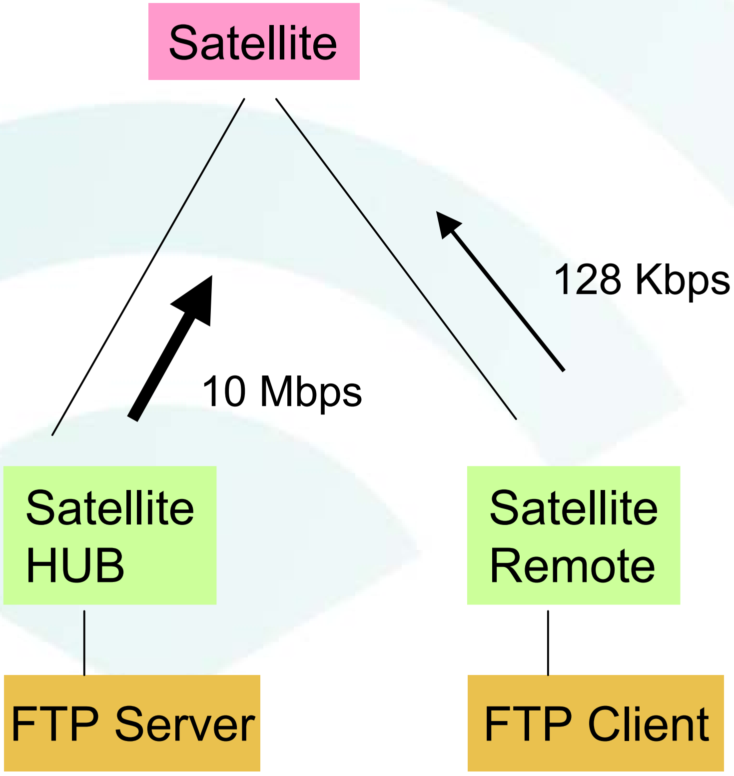
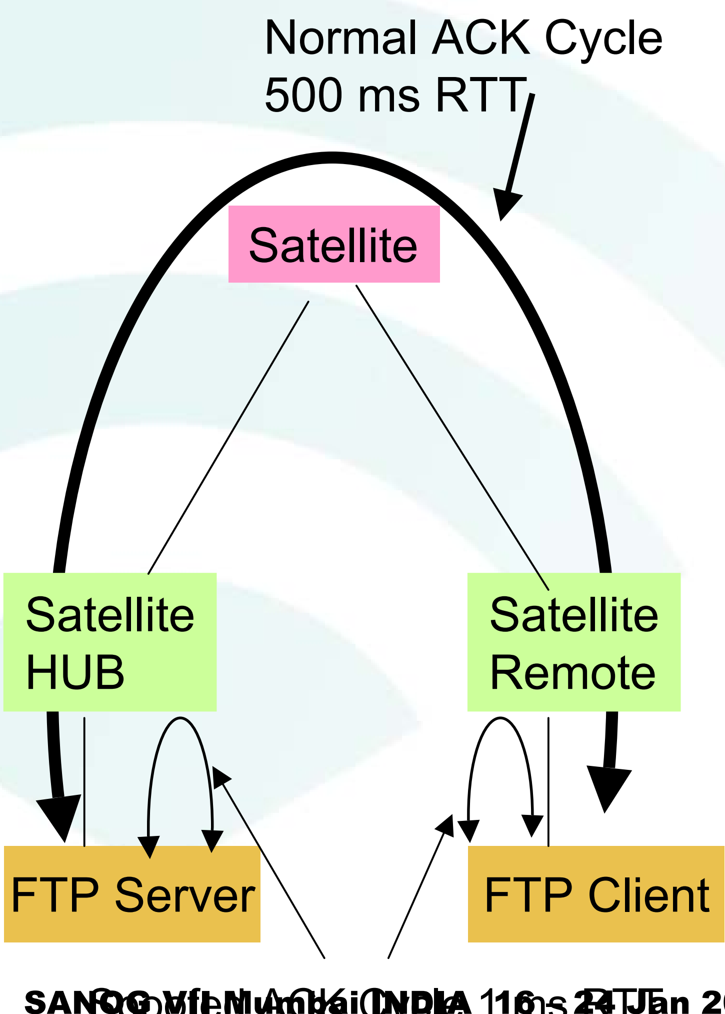
Illustration of lower throughput (Contd.)



So What is the solution?

- Have many such FTP sessions to spawn the bandwidth
  - Discouraged by users
- Increase windows size
  - Cumbersome, involves config tweak on end systems
  - Strongly discouraged by users
  - Window can be upto 64 KB only (on account of 16 bit counter in IP header)
  - Some Operating System like Windows use options field in IP packet to have higher size effective window

Illustration of lower throughput (Contd.)



- So What is the solution? continued…
  - Reduce the RTT ☺
    - Can’t go against the laws of physics
  - But we can fool the end systems
  - The local satellite interfaces acknowledge the packets
  - thereby considerably reducing the RTT & increasing the throughput
  - This is called spoofing

Spoofing …

- Thus we have seen that by Spoofing we can increase the throughput on single FTP session
- But the satellite systems should support the capability
- Major satellite system vendors like Hughes / Gilat support spoofing in their own ways

Another Challenge for high delay Data streams

- If the end application is chatty I.e. a lot of handshake happens between the client & server
- In the high delay environment a lot of the time goes in the above handshake
- Eventually application response suffers drastically giving a bad user experience
- No amount of spoofing helps
- Applications need to modified / rewritten to reduce the chatty nature

Summary : Data Streams on Satellite links

- Gives lesser TCP throughput on account of high delay
- The problem can be overcome by spoofing
- For Chatty applications, the chatty nature needs to be reduced

Considerations for the RTP traffic (Voice)

- RTP traffic or voice is real time characteristics
- Isochronous nature
- Small sized packets
- Low bandwidth per voice session
- Intolerable to Jitter / loss
- Should have as minimum latency as possible

Voice on Low Speed links

- Voice when put on IP inherently takes more bandwidth as can be seen as under (with the assumption of 50 pps)
- Majority of the bandwidth is taken by IP/RTP overheads
  - To transmit a payload of 20 bytes, the ultimate packet size becomes upto 80 bytes on Ethernet
  - Huge Waste
  - Scope of improvement
  - Concept of cRTP comes into picture

Sr.No.	Details	Pay Load	RTP Header	UDP Header	IP Header	L2 Header	Total Bytes	Total Kbps
1	On Ethernet	20	12	8	20	18	78	31.2
2	On WAN Link without cRTP	20	12	8	20	6	66	26.4
3	On WAN Link with cRTP	20	4			6	30	12

Voice on Low Speed links (Contd..)

- cRTP compresses the redundant 40 bytes IP/RTP header to 4 bytes
- Reduces the per voice call bandwidth requirement considerably

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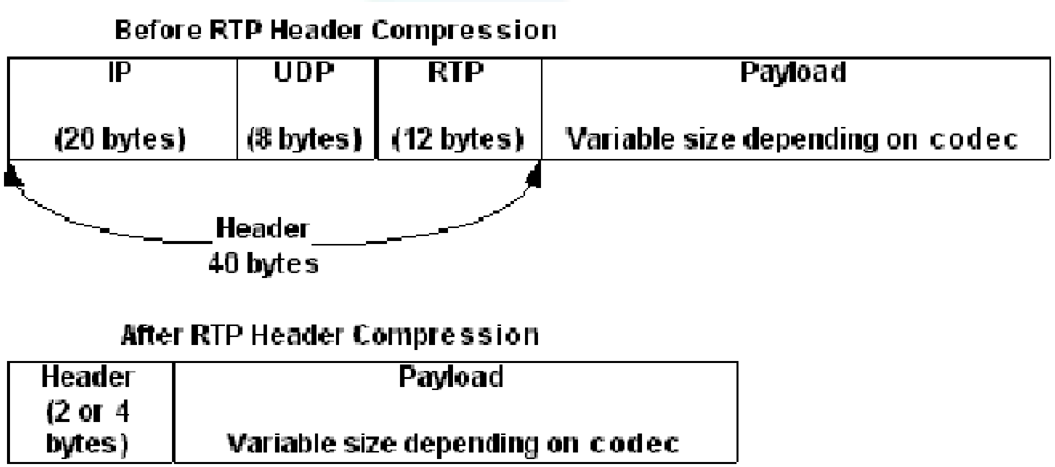


Concept of Serialization Delay

- Router takes some finite time to serialize the packet
- The time is inverse function of bandwidth on serial link
- And direct function of packet size to be transmitted
- The table shows the amount of delay
- On account of serialization delay the link is hogged up for a finite amount of time

Serialization Delay Table

	1 Byte	64 Bytes	128 Bytes	256 Bytes	512 Bytes	1024 Bytes	1500 Bytes
56 kbps	143 us	9 ms	18 ms	36 ms	72 ms	144 ms	214 ms
64 kbps	125 us	8 ms	16 ms	32 ms	64 ms	126 ms	187 ms
128 kbps	62.5 us	4 ms	8 ms	16 ms	32 ms	64 ms	93 ms
256 kbps	31 us	2 ms	4 ms	8 ms	16 ms	32 ms	46 ms
512 kbps	15.5 us	1 ms	2 ms	4 ms	8 ms	16 ms	32 ms
768 kbps	10 us	640 us	1.28 ms	2.56 ms	5.12 ms	10.24 ms	15 ms
1536 kbps	5 us	320 us	640 us	1.28 ms	2.56 ms	5.12 ms	7.5 ms

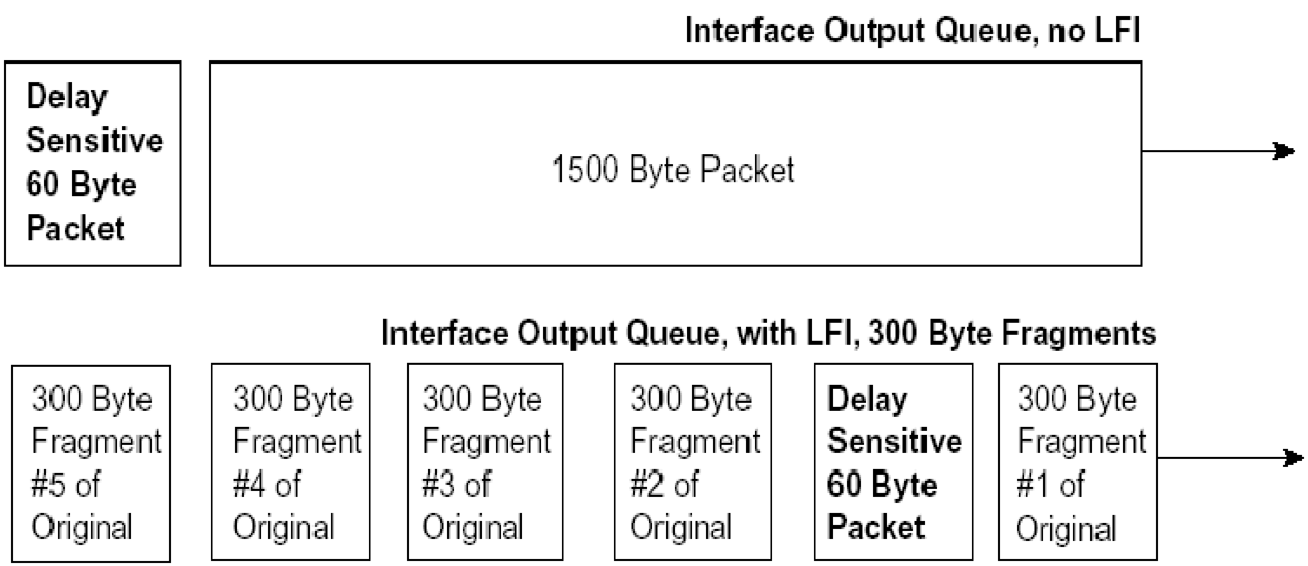
Effect of Serialization Delay

- When Data & Voice application are simultaneously sharing the low bandwidth link
- Because of serialization delay jitter will be introduced inspite any kind of qos
- What's the solution, to break Bigger size packets into smaller
- This is nothing but the concept of LFI
  - Link Fragmentation & interleaving
  - Illustrated as under

LFI Illustration

- A giant 1500 byte packet will hog up the link (equivalent to serialization delay)
- Such high bytes traffic can potentially kill low bytes traffic
- So can be called as killer traffic
- Will create jitter issues for the RTP streams
- With LFI, the bigger size packets are cut into smaller size packets, reducing the jitter

The following image illustrates the operation of LFI:



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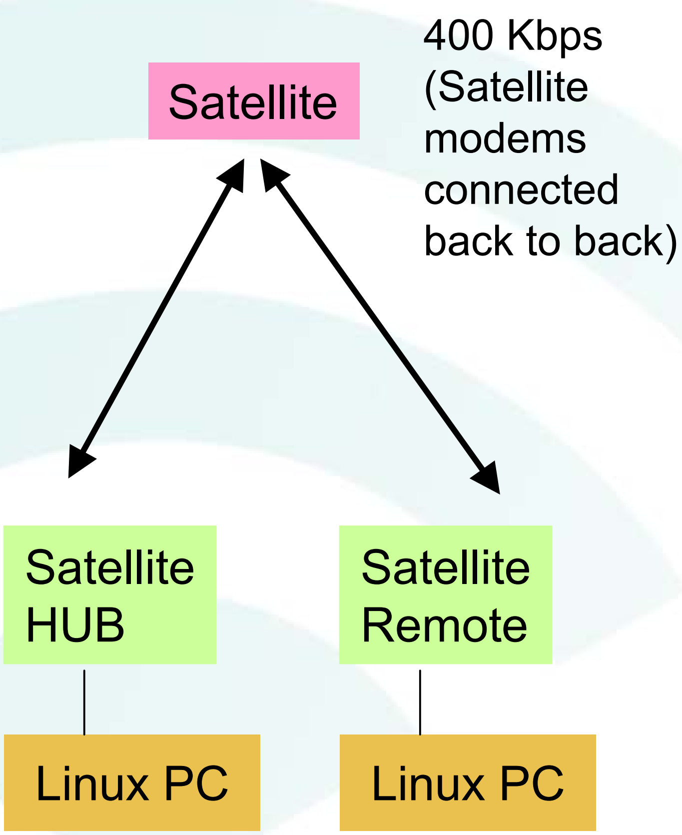
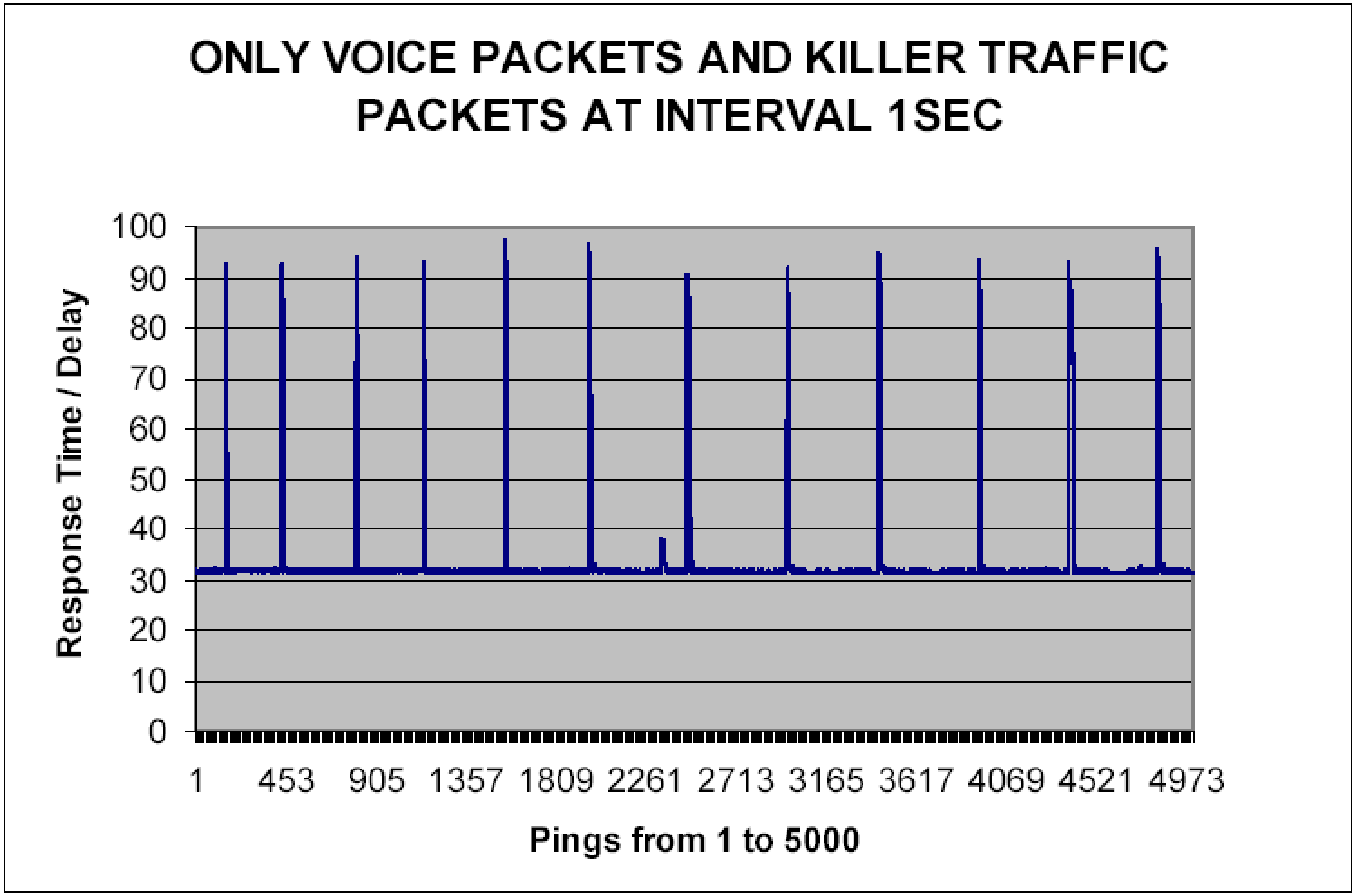
LFI Illustration Test Setup

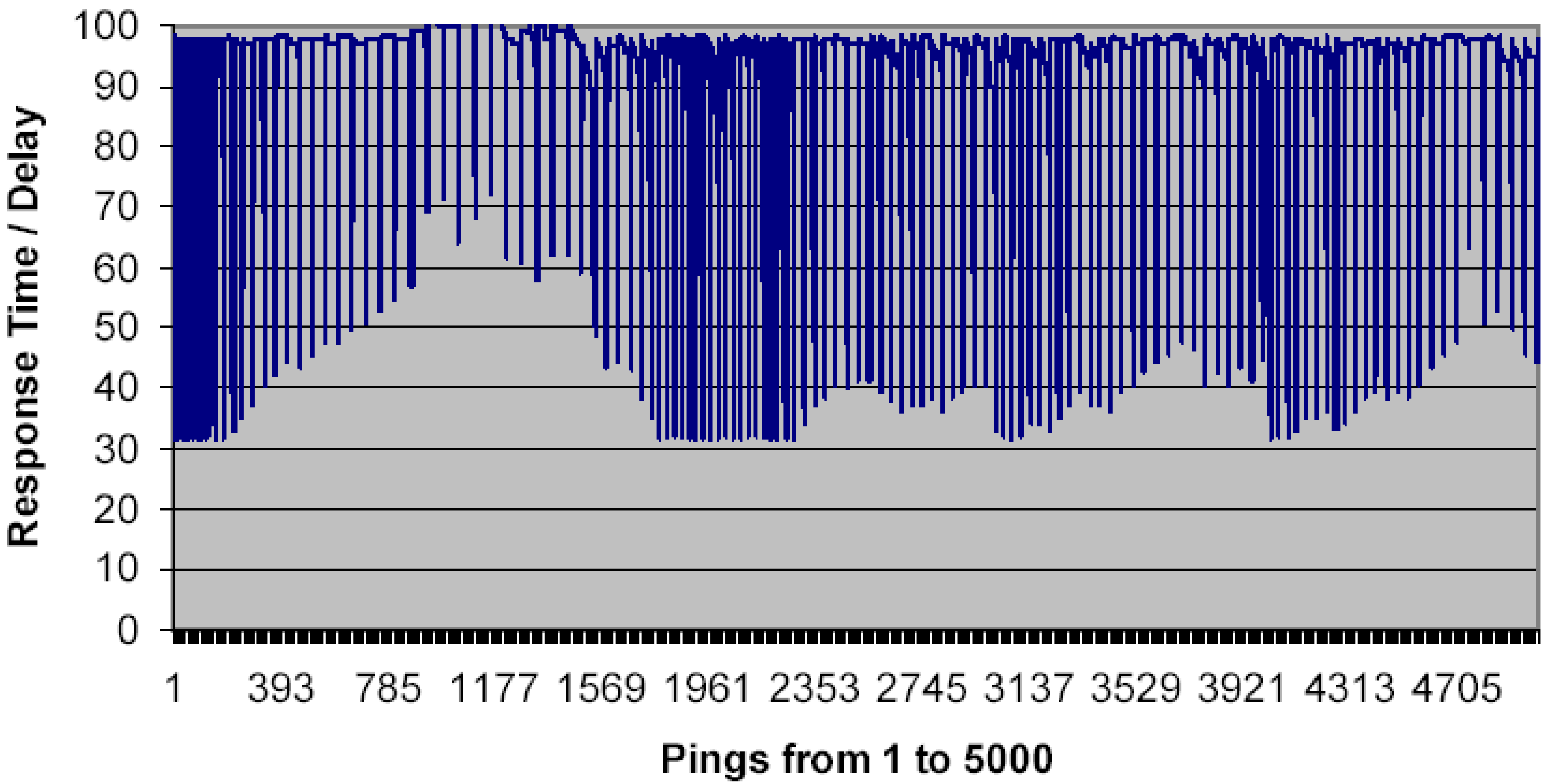


- LFI Illustration can be done by two ICMP ping streams on Linux Machines
- 1500 Byte ping – Stream denoting high byte killer traffic
- 40 Byte ping stream denoting voice traffic
- Both are put in separate queues with 40 bytes given higher priority
- A constant 5000 ping sequence were run & it response was noted with the injection of 1500 byte ping stream simultaneously
- The response of 40 byte ping streams simulates how the voice traffic will behave

Observation (No QoS – No LFI – Killer traffic at 1 sec)



Observation (No QoS – No LFI – Killer traffic at 0.1 sec I.e 100ms)



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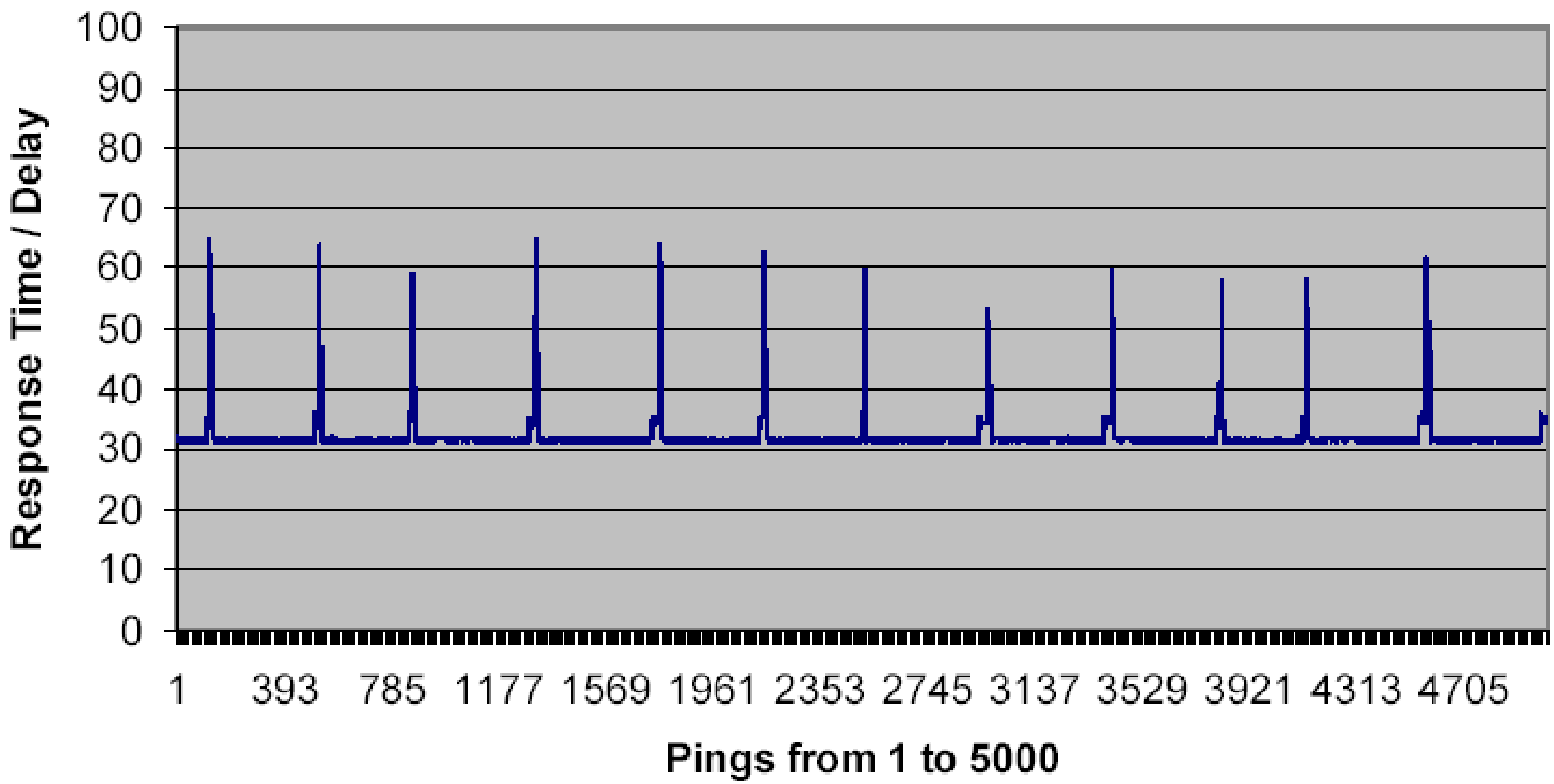

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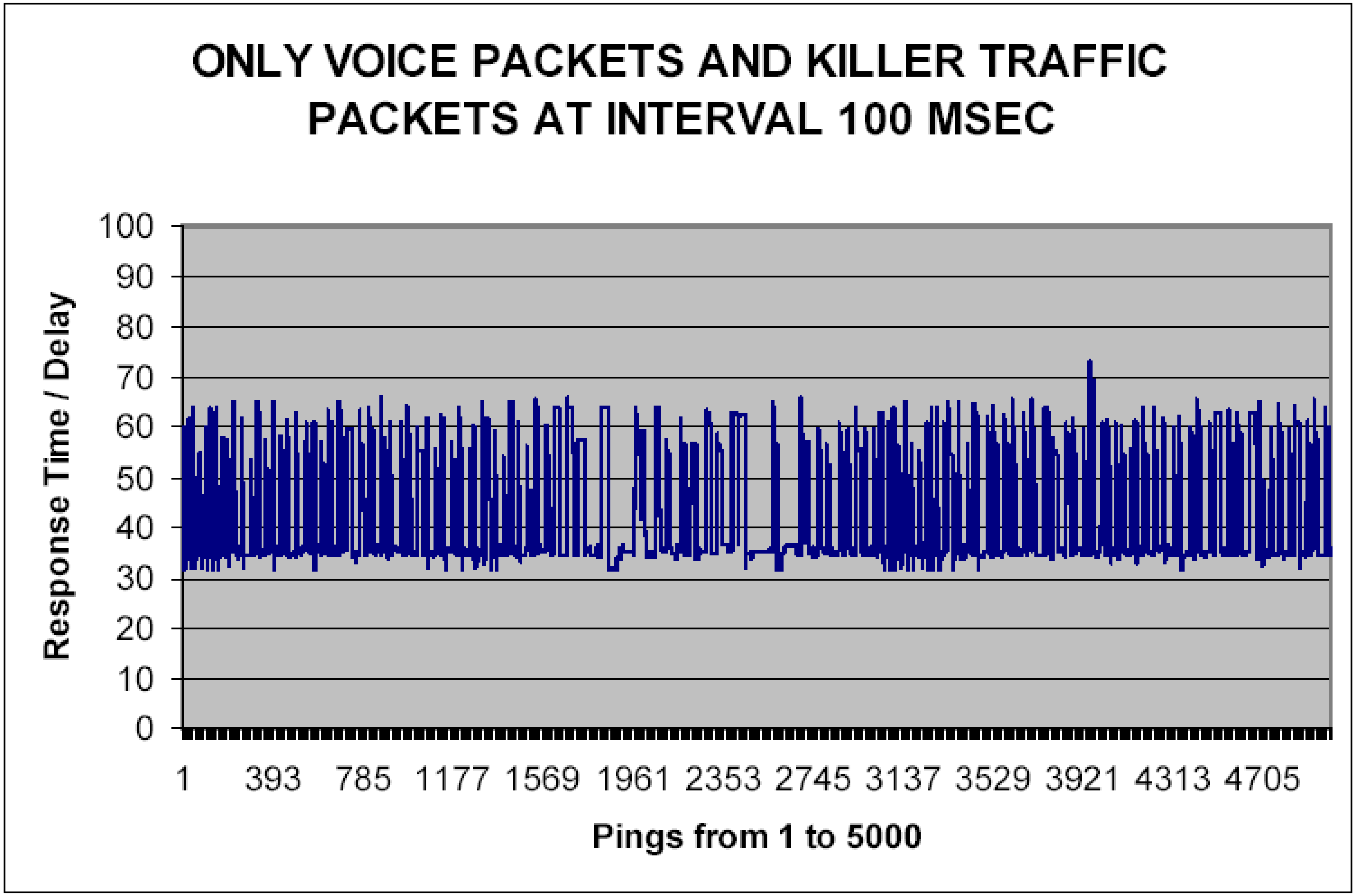
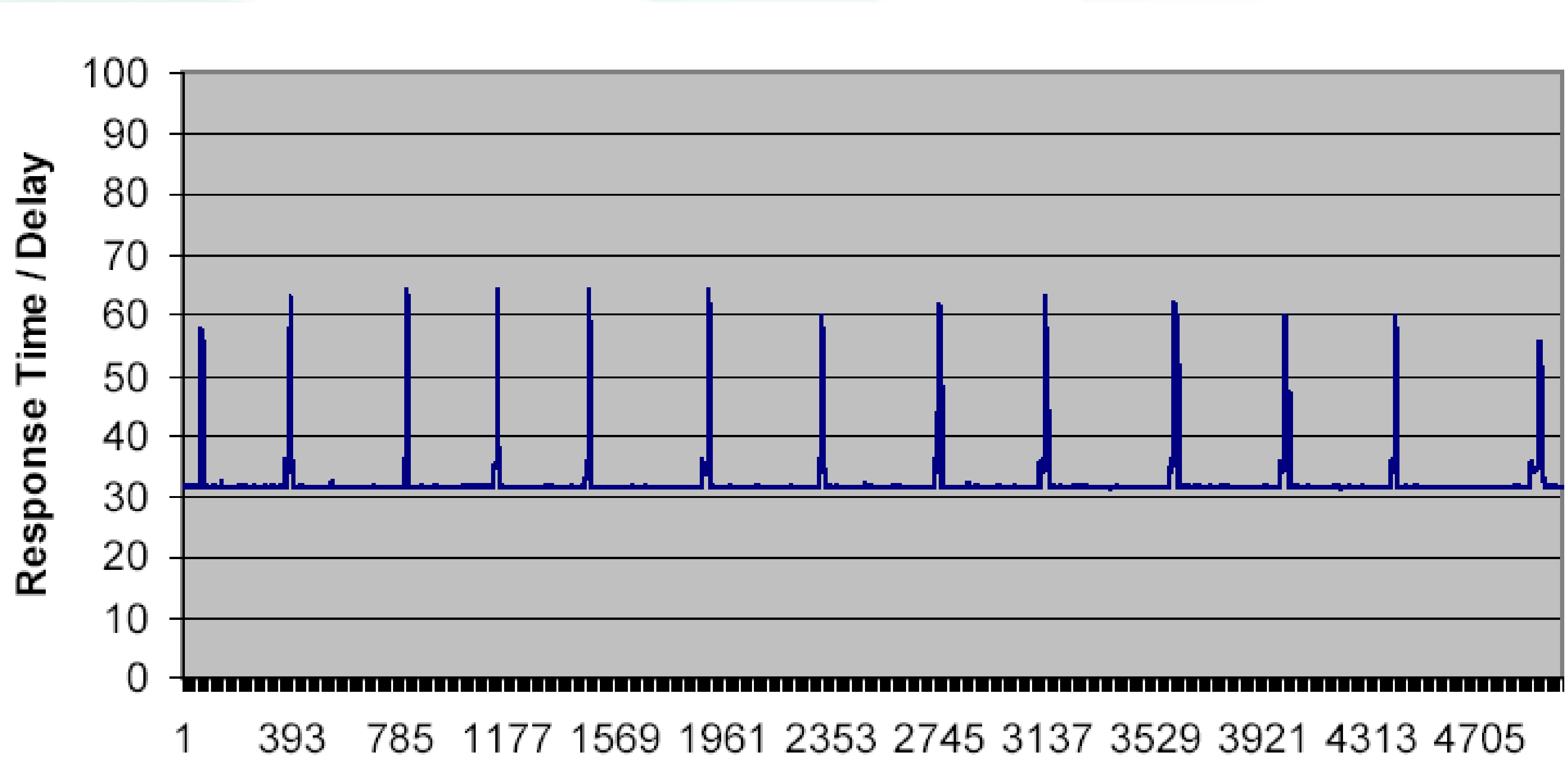


Observation (QoS – LFI of 10ms – Killer traffic at 1 sec)



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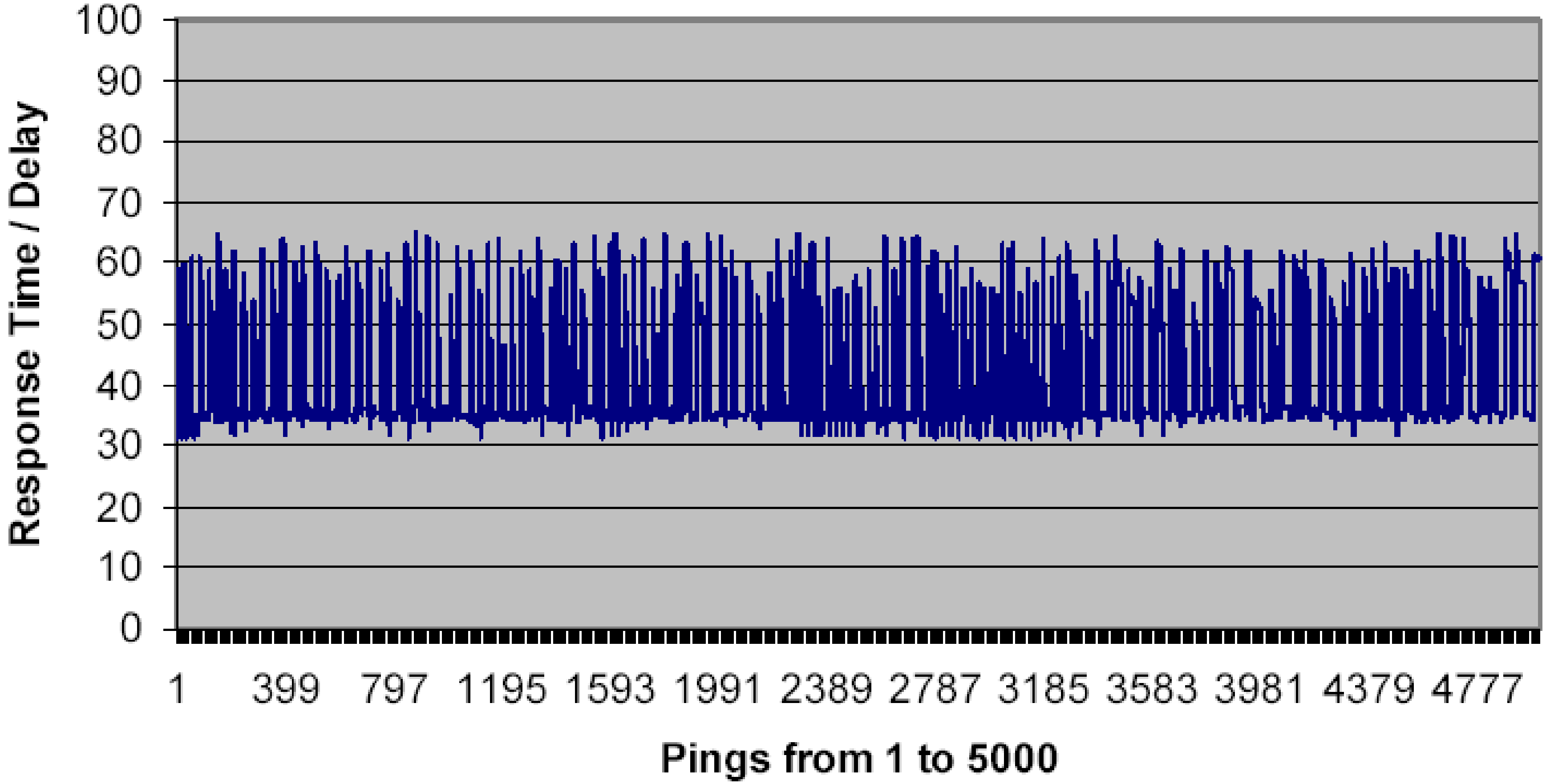
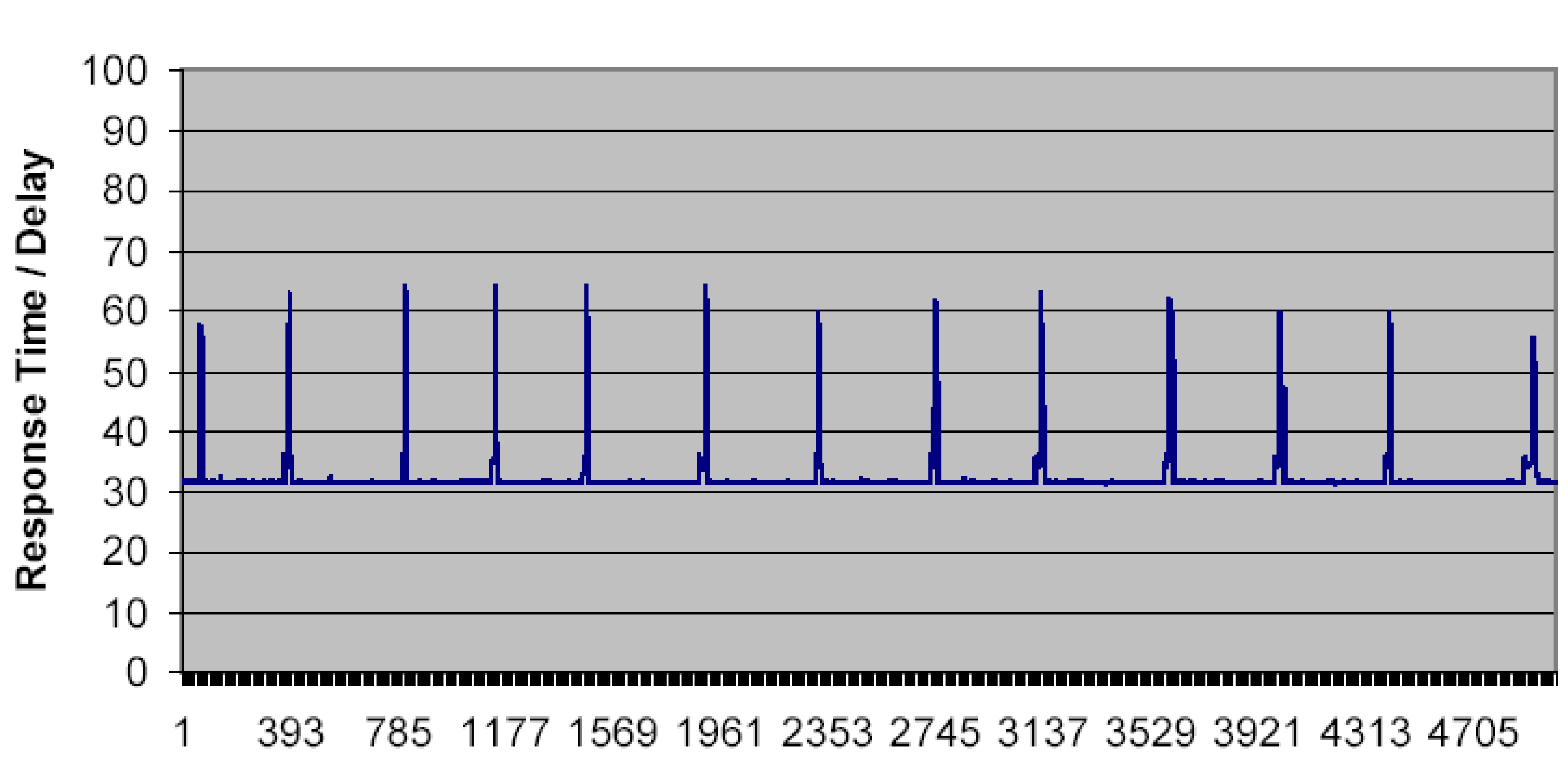


Observation (QoS – LFI of 10ms – Killer traffic at 1 sec)



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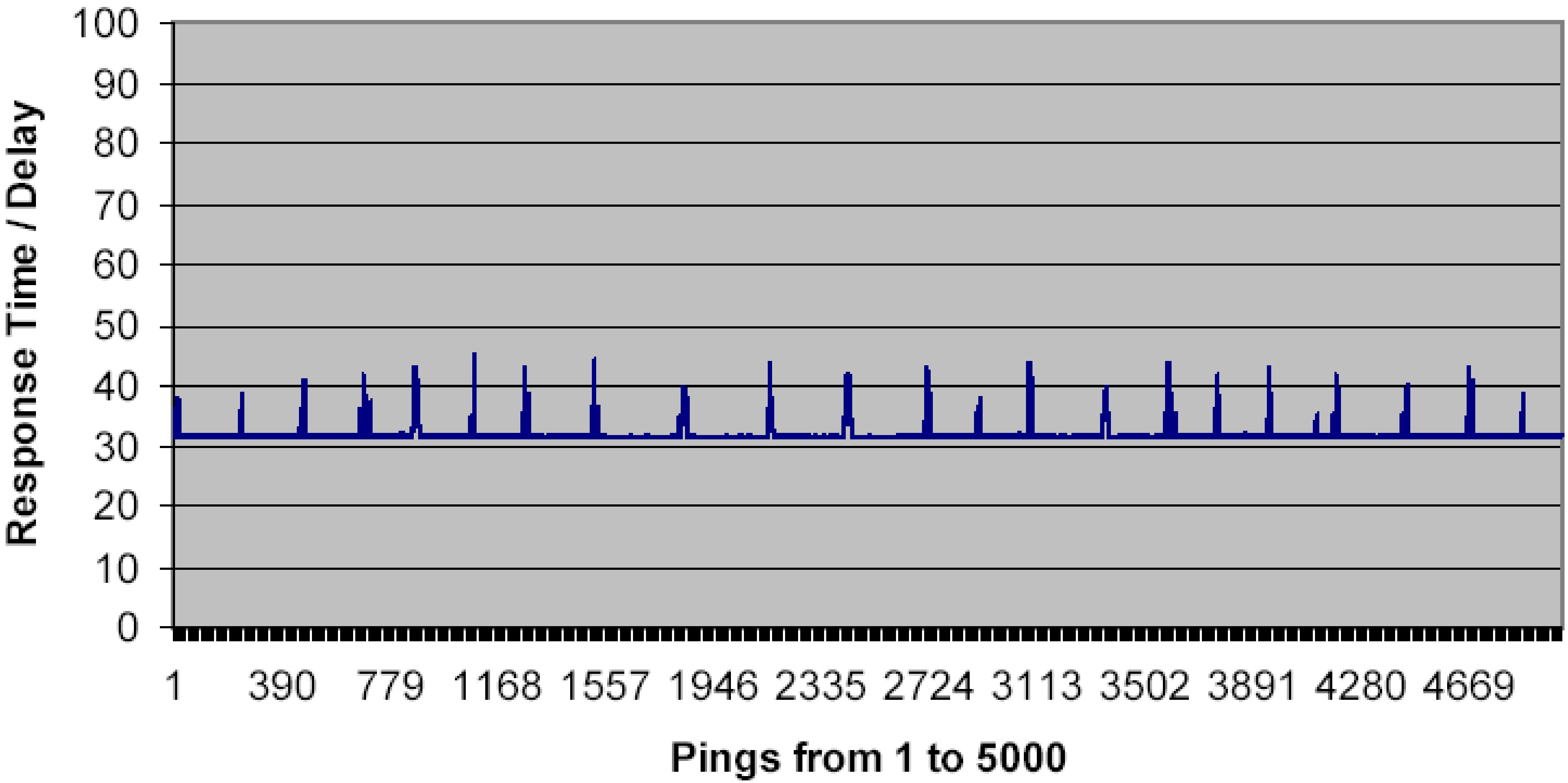
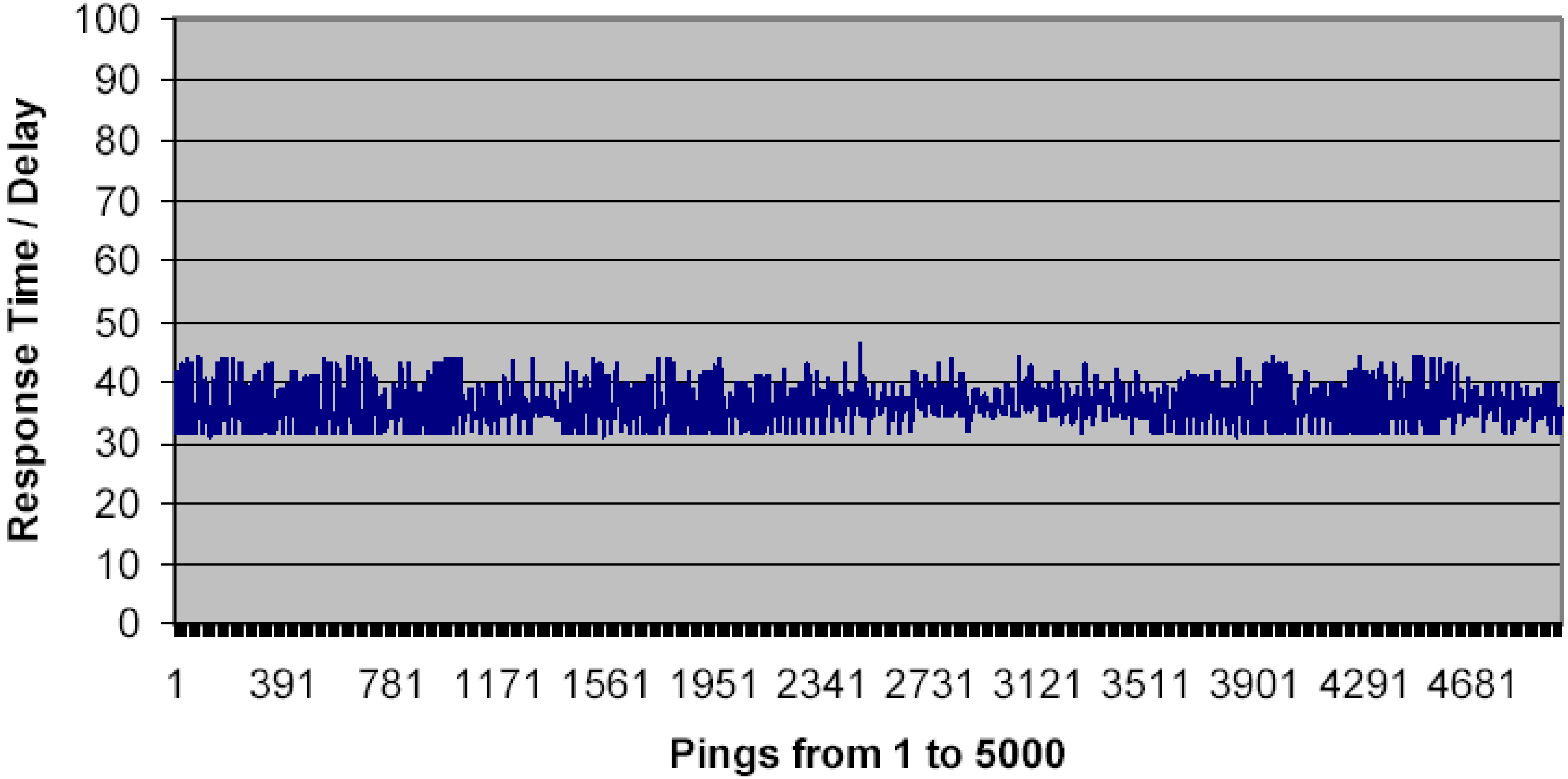
What is Difference Between 10 ms LFI & 2 Ms LFI

	Fragmentation will happen after how many bytes	Fragmentation will happen after how many bytes
Bandwidth (Kbps)	LFI (10ms)	LFI (2ms)
400	500	100
800	1000	200
1500	1875	375
2000	2500	500

Observation (QoS – LFI of 2 ms – Killer traffic at 1 sec)



Observation (QoS – LFI of 2ms – Killer traffic at 0.1 sec)



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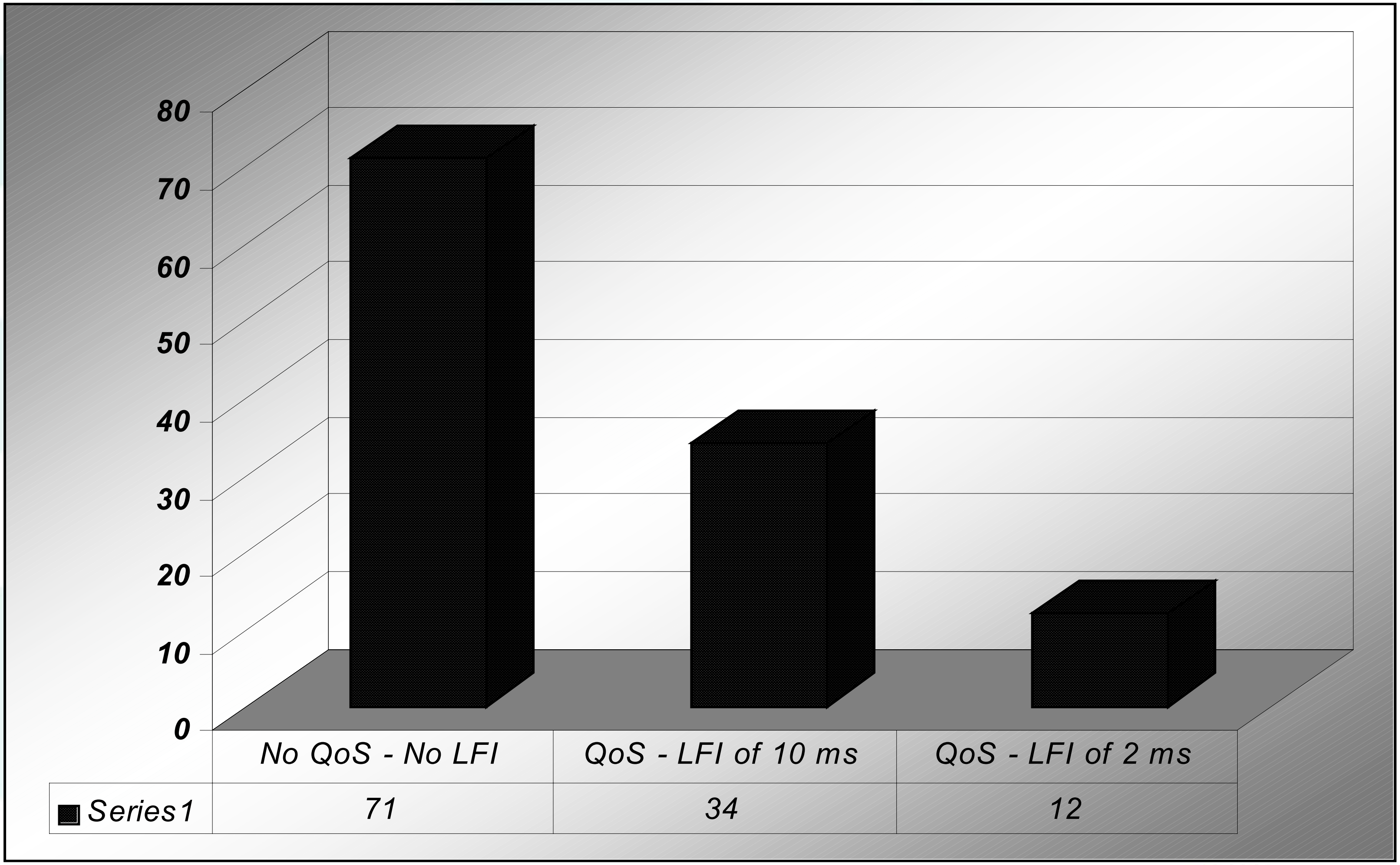
LFI Test results (summary)



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Summary of RTP stream behaviour

- With cRTP the bandwidth can be reduced
- With LFI the voice performance can be guaranteed in the Data + voice mix scenario
- LFI should be used if the bandwidth on the link is low
- Off course enabling cRTP & LFI feature will means additional workload for the network devices
- So careful understanding & Engineering is must

Implementation Strategies ….

- Fully Understand the end requirements / expectations from the user
- List down the applications expected to run across the network & their performance expectations
- Based on the above inputs device strategies to
  - Classify the packets
  - Mark them accordingly
  - Apply policies depending on the user requirements
- Test all the traffic profile
- Based on the test results fine tune the configurations to move towards the final customer expectations

Conclusions

- On Low bandwidth & High Delay satellite links
  - QoS is very important on account of the characteristics like high delay
  - Understanding of the traffic profile is very important
  - Based on the traffic profile QoS strategies need to be finalized to meet the customer experience
    - Keeping in mind the Engineering & commercial considerations

References …

- QoS Presentation
  - Jeff Doyle – Juniper Networks
- IP Telephony & QoS Guide –
  - Wendell Odom / M J Cavanaugh – Cisco Press
  - Cisco Website
- Spoofing –
  - Sunil Janardhanan - Hughes

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