Measurements of Quality Differentiation

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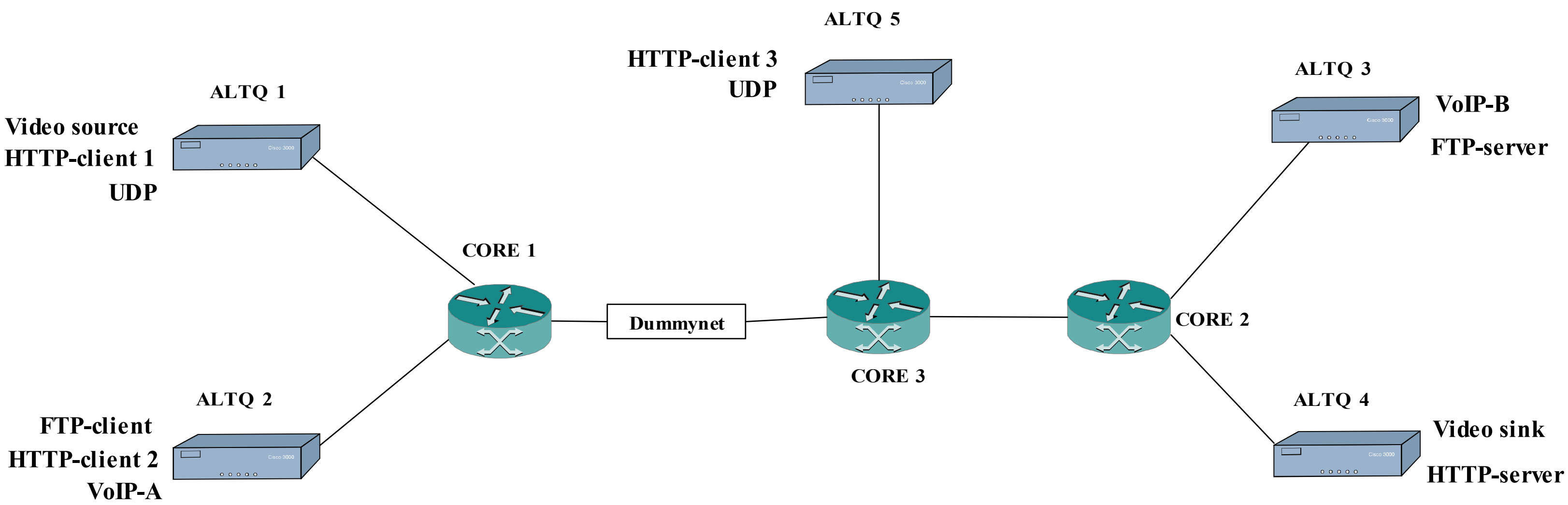
Contents

Network prototype

- Standard PC-hardware
  - AMD 1300 MHz/256 MB
  - 4 * 3Com 10/100 Ethernet NICs
- 3 core and 5 edge routers
- Dummynet network emulator
  - 30 ms extra delay (low and high delay paths)
- Several traffic generators



System configuration

- FreeBSD OS with ALTQ-package
  - QoS mechanisms (queueing, scheduling, shaping, metering, marking)
- All NICs configured to 10Base-Tx full duplex
- CBQ is used in traffic shaping
  - WRR as a general scheduler
- Static provisioning, no borrowing between classes
  - Capacity differentation (a predefined amount of link capacity for each traffic class)
- Queue management: tail-drop, RED, RIO
- Token Bucket (TB) and two rate Three Color Marker (TrTCM) used for metering/marking (color blind)

Traffic generation

- Applications with different properties
  - Real time / non-real time
  - Bandwidth sensitive / delay sensitive
  - TCP / UDP based
  - Constant bitrate / varying bitrate
  - Long “friendly” TCP flows / short “aggressive” flows
- How to carry all this traffic in a single network and same time provide quality of service?

Traffic generation (cont.)

- Traffic generators
  - SmartBits 600
  - PC hardware

- SmartBits 600
  - SmartMetrics 10/100 BaseT Ethernet module
  - SmartVoIPQoS
    - A test application to stress and analyze the networks ability to carry voice and data traffic simultaniously
    - Can generate multiple IP flows and simulate VoIP gateways and phones
    - Measure delay, jitter, throughput and packet loss (+define overall voice quality)

- PC hardware
  - Linux / FreeBSD TCP stack implementation
  - Synchronized by using Network Time Protocol (NTP)

Traffic tracing

- SmartBits 600 can trace the traffic it sends and give aggregated results
  - Transmit flows → Collect data from cards → Display results
- We perform also packet capturing using Tcpdump at client and server side
  - Analysis of TCP traffic
    - RTT, throuhput etc.
  - Packet captures are analyzed with Tcptrace tool (http://irg.cs.ohiou.edu/software/tcptrace/idex.html)

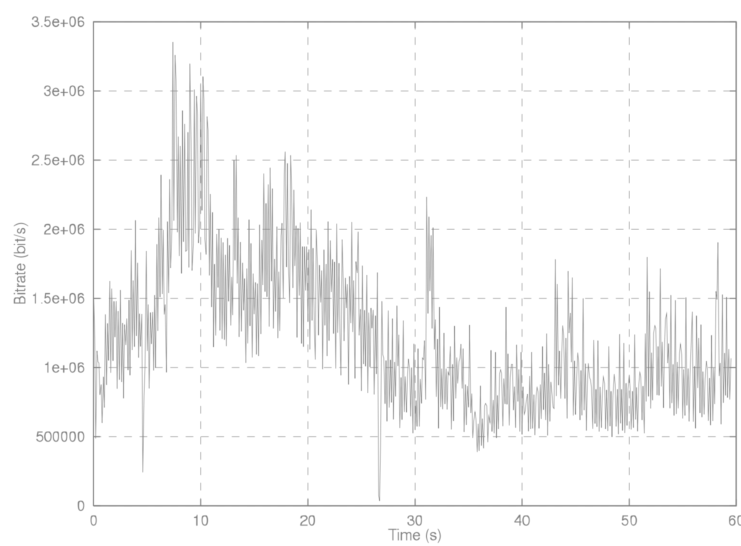
Applications

- VoIP
  - SmartBits 600 and SmartVoIPQoS software
  - G.711 μ-law voice coding with 20 ms framing
    - Packets size of 218 Bytes
  - We apply 20 flows per client, bi-directional
    - =40 VoIP flows
  - Silence detection is not being modelled
    - Constant bitrate application
  - Mapping to PSQM (Perceptual Speech Quality Measure) voice scoring system (0.4→6.5)
    - Comparing measurement results to a matrix that includes mappings between PSQM scores and the effect of impairments (jitter, packet loss)
    - Low PSQM value indicates of high voice quality
    - PSQM values are affected by:
      - Frame loss
      - Jitter
      - Type of codec used

Applications (cont.)

- Video streaming
  - Rude/Crude UDP traffic generator/receiver
    - Used with video trace files
  - MPEG-4 encoded video stream from a movie
  - 25 frames per second (40 ms interval)
  - Mean bitrate 1.029 Mbps, max 8.797 Mbps
    - Varying bitrate application

Video stream data rate profile



Applications (cont.)

- FTP
  - Client-server transactions
  - A modified version of Markus Peuhkuri's Kilent-Server application
  - Packets size of 1500 B (MTU)
  - 50 individual file transfers
  - File size modelled by geometric distribution (mean 500 kB)
  - Example of long lasting TCP connections

Applications (cont.)

- HTTP
  - Apache 2.0 www-server
    - Most popular web server in the Internet
  - Siege is used at the client side
    - http testing and benchmarking utility
    - Simulates a predefined number of users
    - Reports of response times, amount of data transferred, etc.
  - 2 clients with high delay paths and 1 client (client #3) with a low delay path
    - Study the effect of dissimilar RTT
  - HTTP 1.0
  - Simulating 165 users
  - Object size modelled by geometric ditsribution (mean 10 kB)
  - Reading time 12 s
  - Example of short, interactive TCP connections

Test procedure

- A one minute warm up period before actual measurements
  - To ensure that network is in stable condition
- We apply 15 seconds UDP bursts to congest the network
  - Packets of size 512 B (8 Mbit/s)
    - Does not aim to model any particular application
    - An aggressive, unresponsive source
  - VoIP flows
- We record all TCP traffic in the network for later analysis

Warm up → Start traffic sources/tracing → UDP bursts → END

- wait one minute

- transmit 15 seconds, wait about 10 s
- collect VoIP data during waiting time

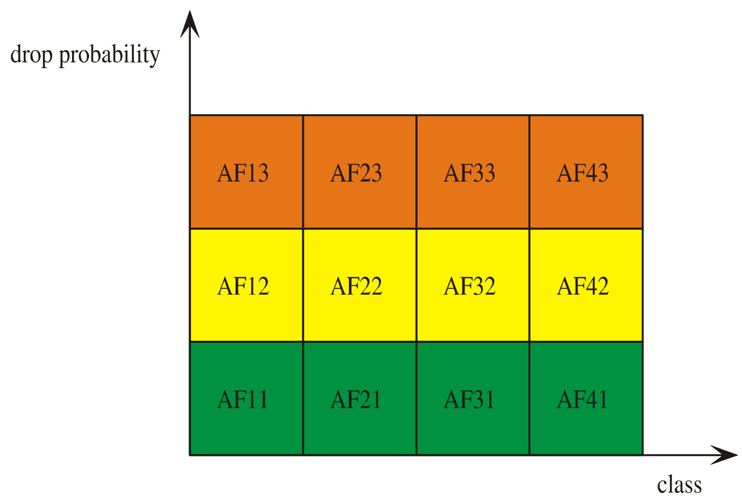
- terminate sources and tracing

Per Hop Behavior (PHB)

- Defines the treatment how traffic belonging to a certain behavior group is forwarded at the individual network node
- DiffServ codepoint of a packet (DSCP) is used to select the PHB
- Two standardized PHB groups
  - Assured Forwarding
  - Expedited Forwarding

Assured Forwarding (AF)

- Four independent forwarding classes with three drop precedences per class (RFC 2597)
- The forwarding assurance of an IP packet in a network node is determined by:
  - Resources allocated to the particular AF class
  - The current load of the AF class
  - Drop precedence of a packet



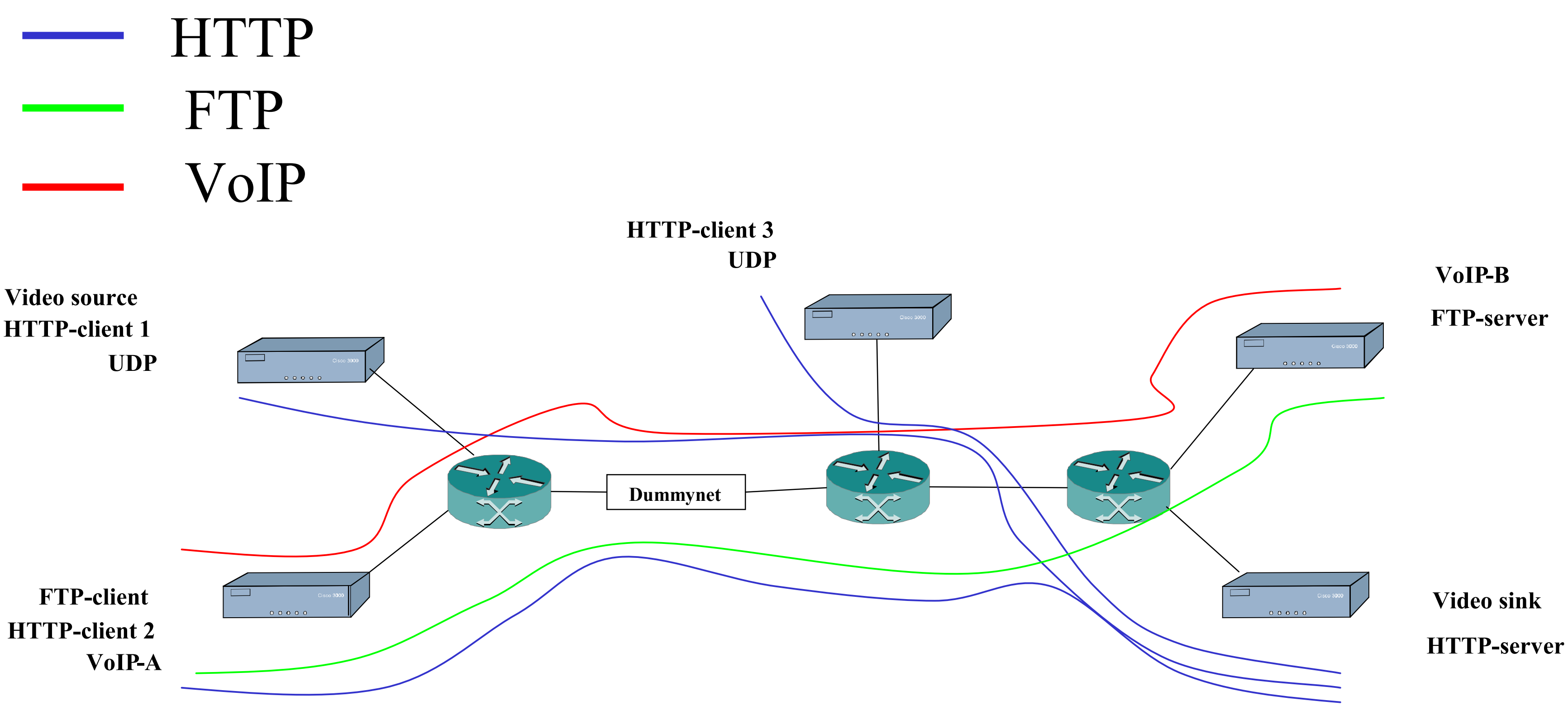
Expedited Forwarding (EF)

- ” Leased line emulation”
  - Low loss, low latency and assured bandwidth service
- Defined in RFC 2598
- Strict queueing treatment
- Arrival rate < minimum service rate
  - Requires powerful forwarding from the router

Test cases

- Case 1: Best Effort (BE): No differentation between traffic flows
- Case 2: Expedited Forwarding (EF): EF service for VoIP flows.
- Case 3: Assured Forwarding (AF): Four independent forwarding classes.
- Case 4: EF+AF
- Case 5: AF: Different provisioning

Connection pairs



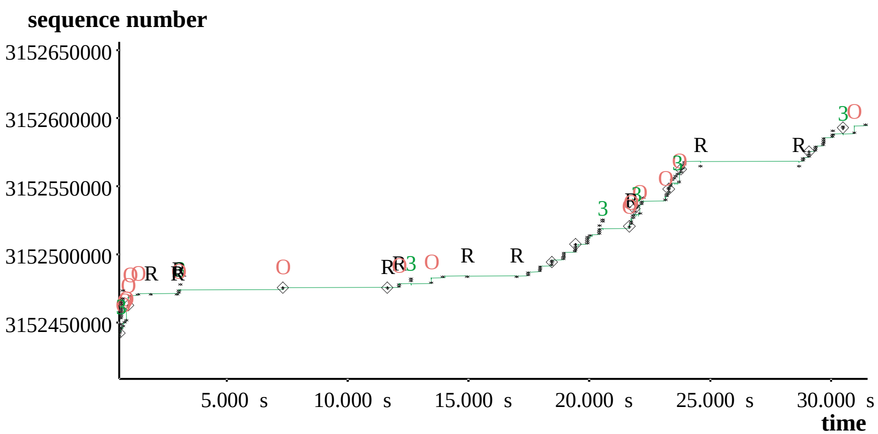
Case 1: BE

- Baseline (reference) for our studies
- No differentiation
  - Traffic sources are competing of the resources
  - Situation in today's Internet

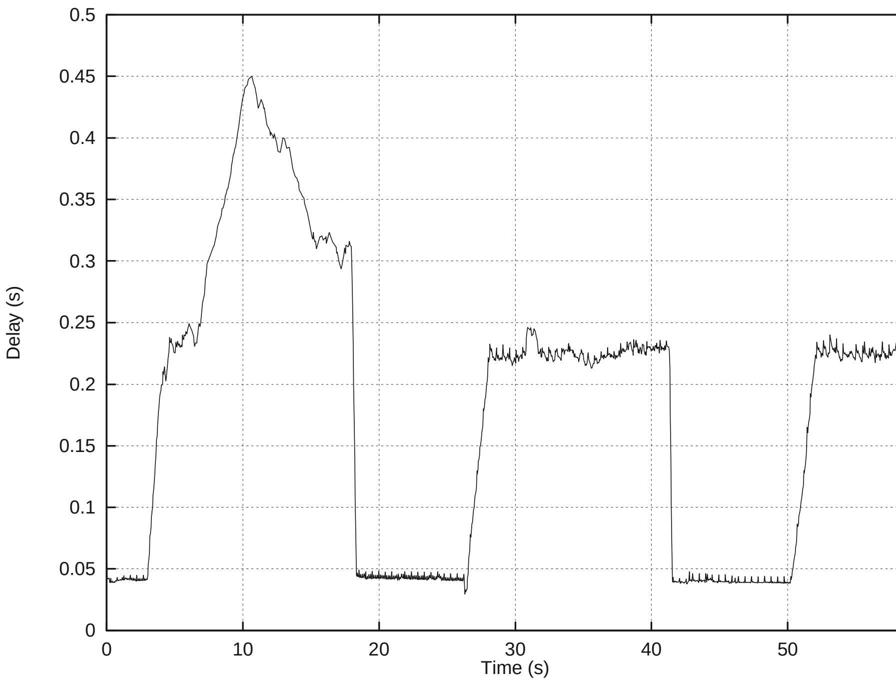
Case 1: BE (cont.)

- TCP is not able to send new data during congestion
- High delay and packet loss

TCP sequence number congestion (FTP connection)



Video streaming delay



	Packet loss (%)	PSQM	Avg latency (ms)	Avg jitter (ms)
A->B	14.57	3.16	220.87	1.95
B->A	18.15	3.49	231.16	2.82
Total avg	16.36	3.3	225.56	2.37

VoIP

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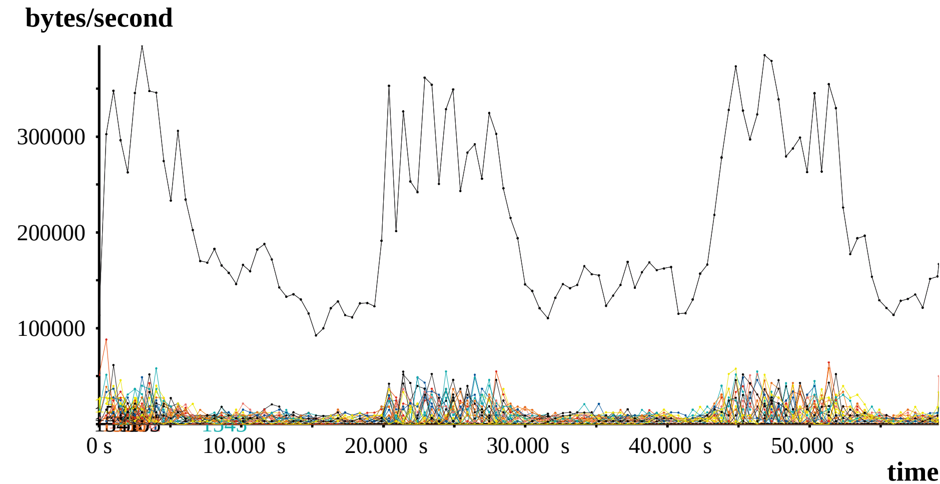
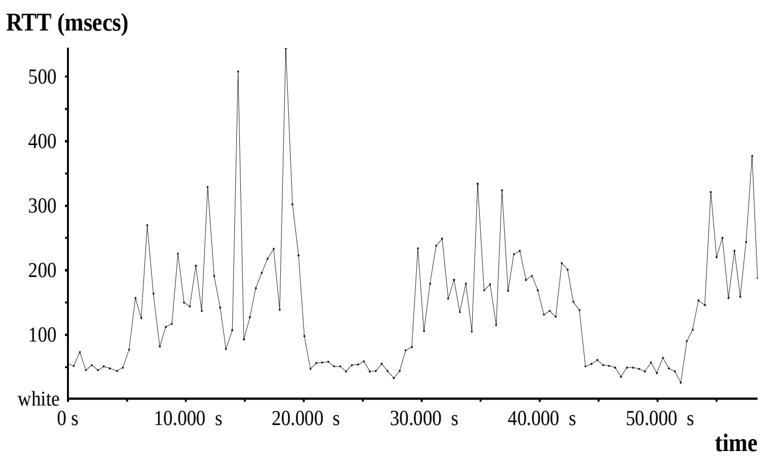
Case 1: BE (cont.)

- Delays not acceptable, heavy packet loss
- http client (client 3) with low delay path is dominating

FTP throughput



HTTP



HTTP

Source	Data transferred (B)	Response time (s)	Throughput (kbps
Client 1	2751418	3.23	365.33
Client 2	2616671	2.84	350.17
Client 3	4272803	0.11	570.75

Case 2: EF

- Two scenarios
  - 20 % of the link capacity provisioned for VoIP traffic
  - 30 % of the link capacity provisioned for VoIP traffic
- Other traffic sources get BE service
- Max delay set to 20 ms
- Leased line emulation

Case 2: EF (cont.)

VoIP

20 % provisioning

	Packet loss (%)	PSQM	Avg latency (ms)	Avg jitter (ms)
A->B	0.080	0.46	109.60	1.019
B->A	0.067	0.45	98.56	1.271
Total avg	0.073	0.46	103.94	1.144

30 % provisioning

	Packet loss (%)	PSQM	Avg latency (ms)	Avg jitter (ms)
A->B	0	0.4	30.870	1.123
B->A	0	0.4	33.992	1.572
Total avg	0	0.4	32.431	1.348

- Increasing provisioning improves the quality of VoIP calls
  - –However, there is some oskillation in jitter

Case 2: EF (cont.)

- Low delay path more dominant when more resources allocated for BE traffic
- EF can be used to provide a leased line emulation

HTTP

80 % for BE service

Source	Data transferred (B)	Response time (s)	Throughput (kbps)
Client 1	2848275	2.68	380.66
Client 2	3149460	2.35	421.12
Client 3	4256773	0.14	565,5

70 % for BE service

Source	Data transferred (B)	Response time (s)	Throughput (kbps)
Client 1	2814368	2.89	374.56
Client 2	3352512	1.85	445.37
Client 3	4088847	0.46	543.01

Case 3: AF

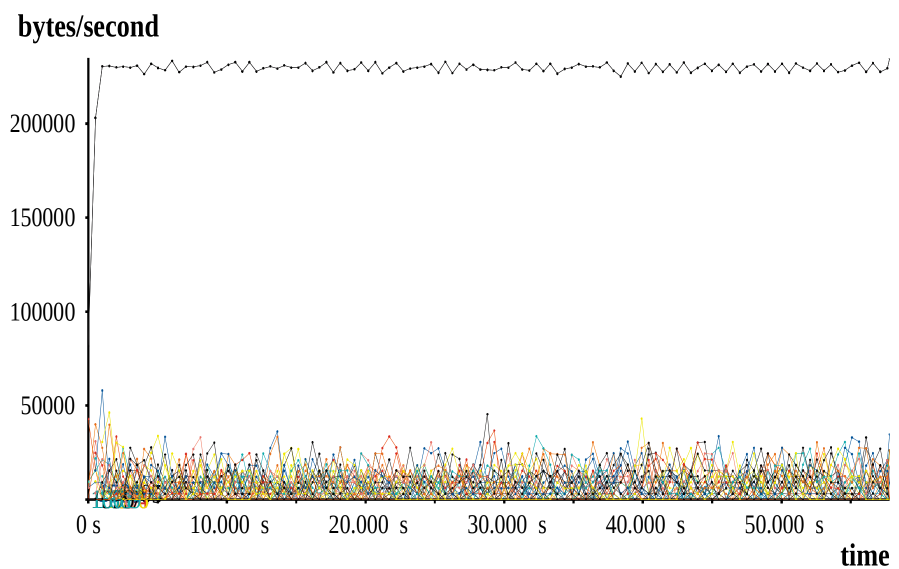
Class	Application	Bandwidth %	Buffer size	Conditioner	CIR (Mbps)	Peak rate (Mbps)
AF1	VoIP+Video	50	20 ms	TB	5	N/A
AF2	HTTP	20	60 ms	trTCM	4	5
AF3	FTP	18	120 ms	trTCM	2	3
AF4	Other	10	360 ms	trTCM	2	3

- Class AF1 for real time applications
- Class AF2 for interactive TCP traffic (HTTP)
- Class AF3 for non-interactive TCP traffic (FTP)
- Class AF4 for the traffic that does not conform to AF1, AF2 or AF3

- 2 % of resources is assigned for the control traffic

Case 3: AF (cont.)

FTP throughput

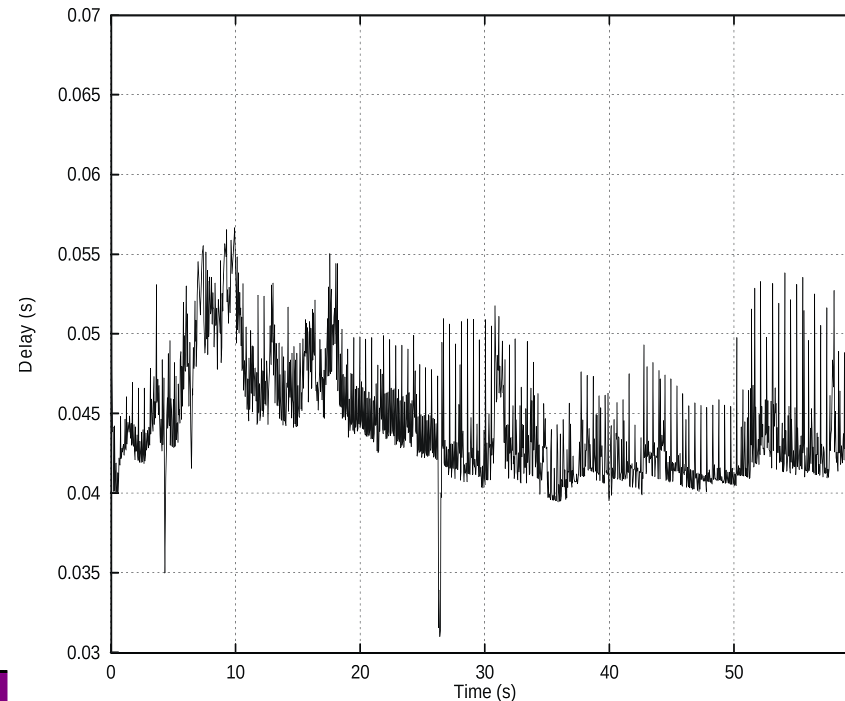


- FTP throughput is bounded to about 1.8 Mbit/s (the target rate for class AF3)

Case 3: AF (cont.)

- Fairness between HTTP clients
  - AF helps to diminish the effect of RTT
- Real time applications get decent quality of service
- TCP sources achieve (almost) their target rate

Video streaming delay



VoIP

	Packet loss (%)	PSQM	Avg latency (ms)	Avg jitter (ms)
A->B	1.06	1.02	32.27	2.28
B->A	0.0	0.4	34.77	3.91
Total avg	0.53	0.7	33.48	3.10

HTTP

Source	Data transferred (B)	Response time (s)	Throughput (kbps)
Client 1	3742733	1.26	500.20
Client 2	3744647	1.27	501.21
Client 3	3931651	0.79	522.31

Case 4: EF+AF

Class	Application	Bandwidth %	Buffer size
EF	VoIP	20	20 ms
AF1	Video	30	20 ms
AF2	HTTP	20	60 ms
AF3	FTP	18	120 ms
AF4	Other	10	360 ms

- A finer differentiation between real time applications
  - EF PHB for VoIP flows
- Otherwise same as in previous case

Case 4: EF+AF (cont.)

HTTP

Source	Data transferred (B)	Response time (s)	Throughput (kbps)
Client 1	3613647	1.45	483.75
Client 2	3760971	1.29	495.84
Client 3	3922848	0.82	525.32

VoIP

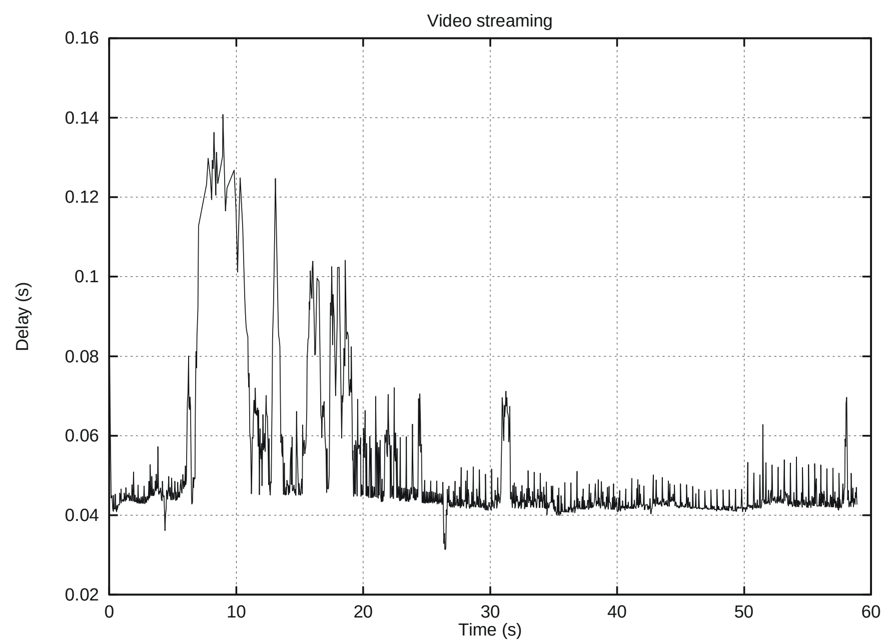
	Packet loss (%)	PSQM	Avg latency (ms)	Avg jitter (ms)
A->B	0	0.4	32.199	1.807
B->A	0.040	0.43	34.386	3.665
Total avg	0	0.4	33.293	2.736

- VoIP and HTTP flows experience similar QoS as in AF case

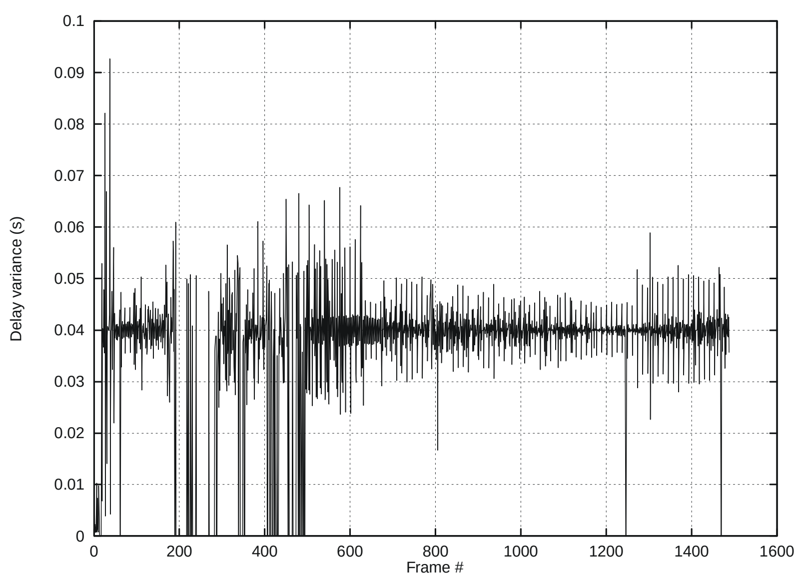
Case 4: EF+AF (cont.)

- Resources are exhausted during the first UDP burst in AF1 class (video streaming)
  - Bad provisioning (overload in the class)

Video streaming delay



Video streaming delay variance / packet loss



Case 5: AF different provisioning

Class	Application	Bandwidth %	Buffer size	Conditioner	CIR (Mbps)	Peak rate (Mbps)
AF1	VoIP+Video	30	20 ms	TB	5	N/A
AF2	HTTP	40	60 ms	trTCM	4	5
AF3	FTP	10	120 ms	trTCM	2	3
AF4	Other	18	360 ms	trTCM	2	3

- More bandwidth allocated for TCP traffic

Case 5: AF different provisioning (cont.)

	Packet loss (%)	PSQM	Avg latency (ms)	Avg jitter (ms)
A->B	14.62667	3.21	32.026	1.768
B->A	0	0.4	33.666	2.656
Total avg	0	1.81	33.846	2.212

VoIP

Source	Data transferred (B)	Response time (s)	Throughput (kbps)
Client 1	4148934	0.40	554.67
Client 2	4136732	0.45	549.64
Client 3	4251938	0.21	565.70

HTTP

Summary

	VoIP	HTTP response time (s)			Video streaming
Case	**PSQM**	**Client 1**	**Client 2**	**Client 3**	**Packet loss %**
BE	3.3	3.23	2.84	0.11	15.1
EF (30%)	0.4	2.89	1.85	0.46	56.3
EF (20%)	0.46	2.68	2.35	0.14	23.2
AF	0.7	1.26	1.27	0.79	2.46
EF+AF	0.4	1.45	1.29	0.82	7.69
Case 5	1.81	0.40	0.45	0.21	32.2

Conclusions

- In Best Effort network the traffic sources are interfering with each other
- EF can be used to provide premium service
- AF helps to reduce the effect of RTT for TCP connections
  - Better fairness
- CBQ is not a perfect solution
  - Need for adaptive schedulers?
- It’s all about provisioning
  - The problem of provisioning the resources
  - Need for dynamic provisioning
- Tuning the parameters is not easy

Future work

- Test different mechanisms
  - Borrowing, different schedulers, queuing algorithms etc.
- Measurements using dynamic provisioning
- Adding more flows / connections per class
- More realistic traffic distribution
  - More TCP connections
- Deeper analysis on data
- Measurements using Adtech AX4000
- Development of centralized management platform
  - Easier to manage the network and traffic generators

Thank you

Questions ?