Next steps on performance evaluation based on network simulation

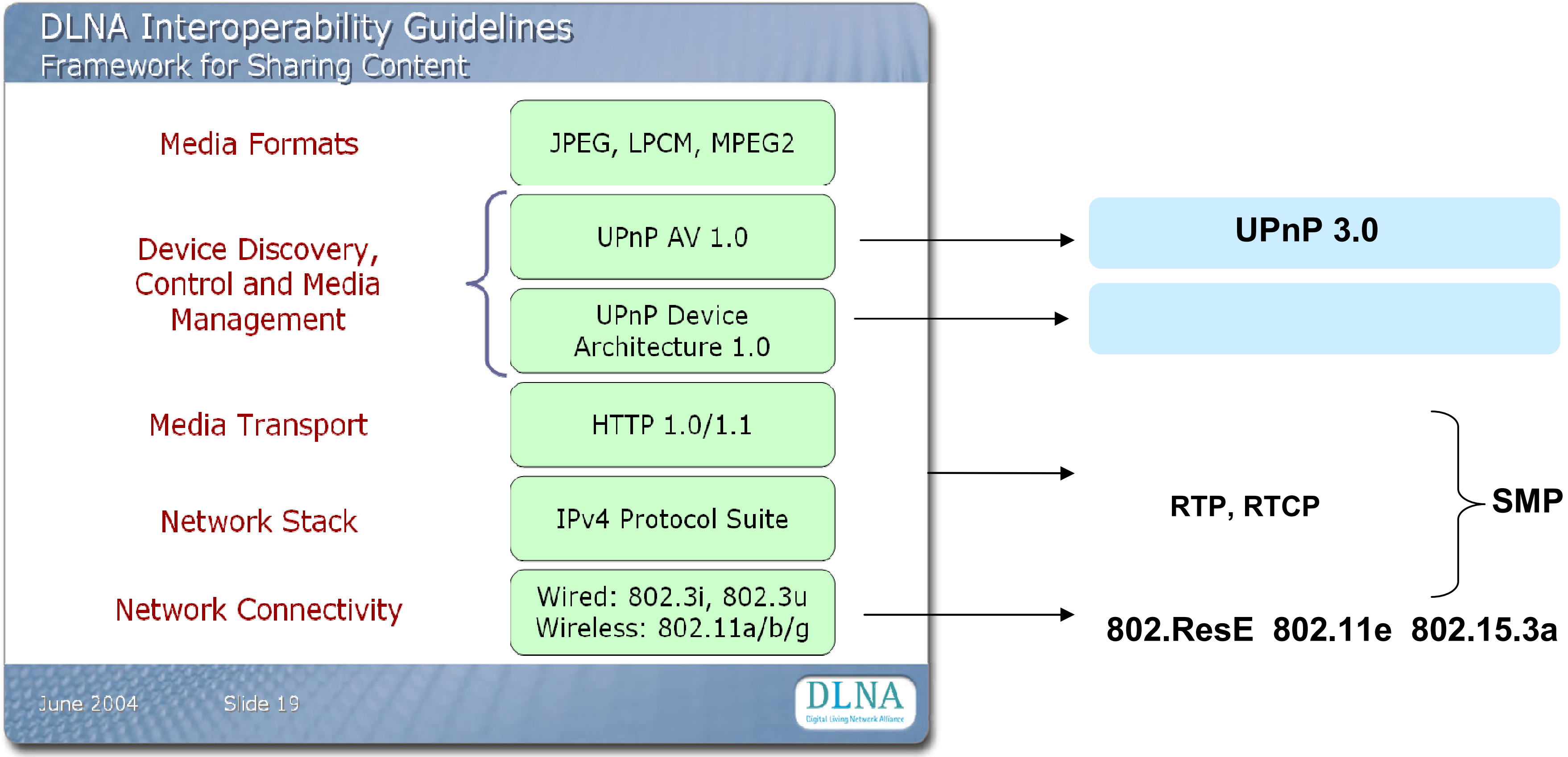
Eric HS Ryu / Samsung Electronics

Jinoo Joung/ Sang-Myung Univ.
Jong-Ho Youn / Aviation Univ.
Hong-Kyu Jung / ICU

IEEE 802 Plenary
July 2005

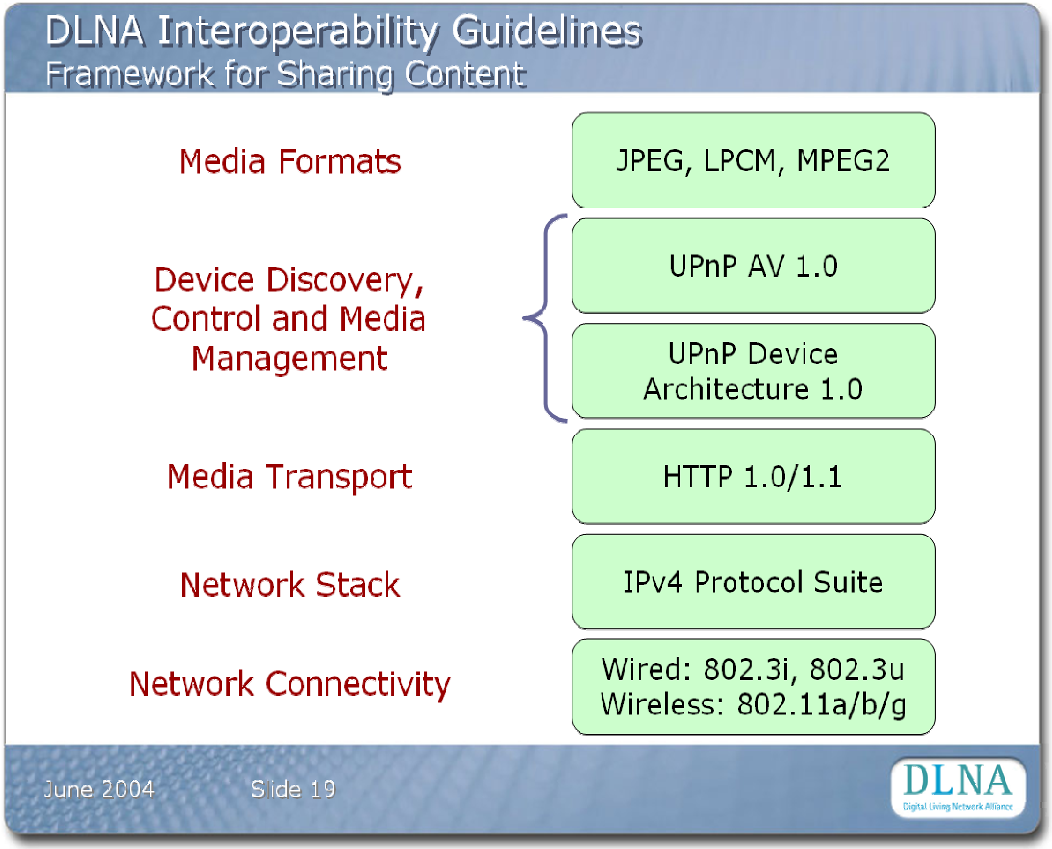
eric_ryu@samsung.com

QoS enabling of the protocol stack



- lower layers are enhanced and capable to provide on demand QoS guarantees
- higher layers should provide the hooks to allow applications to reserve the resources that they require

Shrinking of complex protocol stack



- By use of Bandwidth guarantee capability of new Layer-1/2 solutions, some of applications may access layer 1/2 directly
- Enables simplified solutions for low cost devices

Possible approaches and solutions

- Delay origin: nodes & links
  - Delay within a node is twofold: processing delay and queueing delay
  - Processing delay could be neglected because of current high-speed VLSI-based switching devices
  - Queueing delay is primary concern because delay within a link is bounded

- Better understanding of application traffic may give idea how to deal with the delay bounding problem for the simplified architecture

Description of MPEG-2 Traffic Model

❑ Traffic Characteristics

- Frame size distribution of I, P and B frame type
  - Candidates; Gamma, Weibull, **lognormal** pdf
- Video traffic modeling; pdf for lognormal distribution

$$f(z) = \frac{1}{z\sqrt{2\pi\sigma^2}} \exp[\frac{-(\ln z - u)^2}{2\sigma^2}], z > 0$$

$$f(z) = 0, otherwise$$

- Maximum Likelihood Estimator (MLE) parameters of fitting distributions

Frame Type	lognormal	
	μ	σ
I	5.1968	0.2016
P	3.7380	0.5961
B	2.8687	0.2675

Table 2: The MLE parameters of fitting distributions.

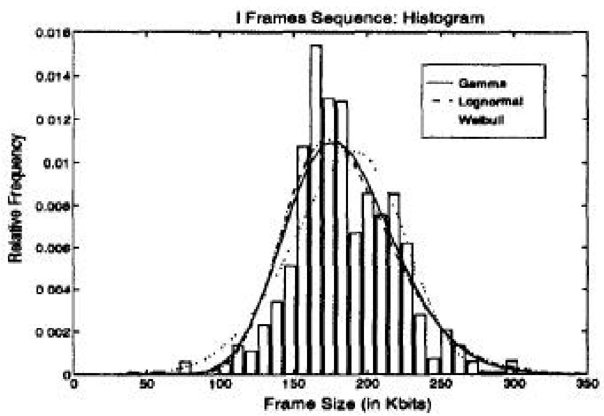


Figure 14: Histogram and fitting density functions for tl
I-frames subsequence.

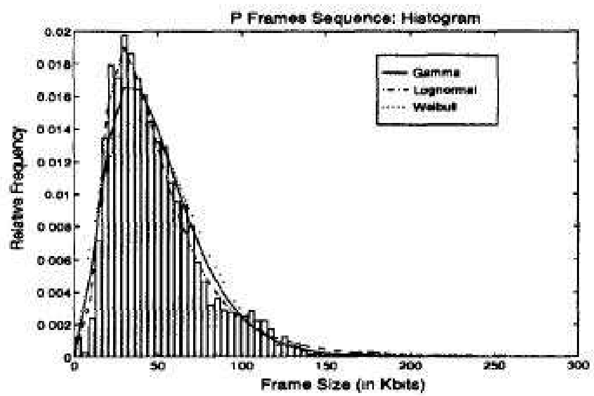


Figure 15: Histogram and fitting density functions for tl
P-frames subsequence.

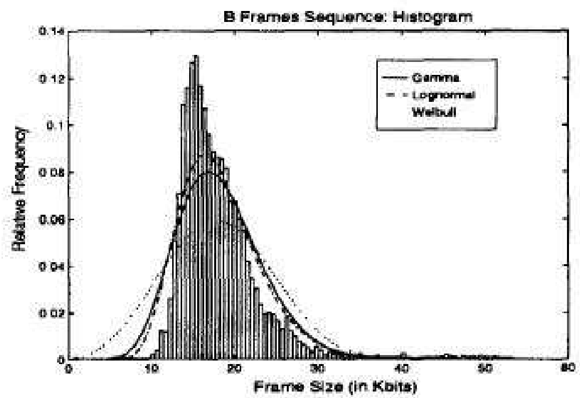


Figure 16: Histogram and fitting density functions for the
B-frames subsequence.

Specification of MPEG-2 Traffic Model

❑ MPEG-2 OpNet Module Descriptions

- GOP = Group of Pictures
- Frame rate = 30 frames/sec
- I frames are log-normally distributed
- B & P frames are iid log-normally distributed

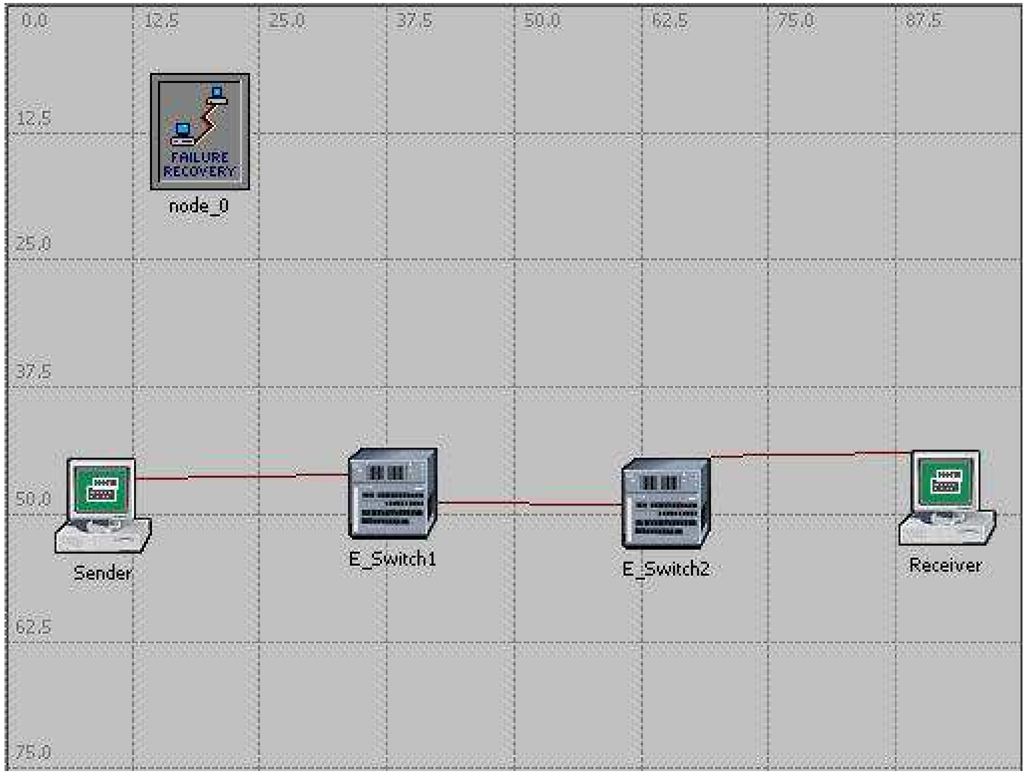
		SDTV Rate	HDTV Rate
Frame Rate	30 per sec		
Scene Size	d is geometrically distributed	Success Prob.: P = 0.1	Success Prob.: P = 0.1
GOP Size	Fixed	12 or 15	12 or 15
I Frames	Log Normal – iid/AR process	μ = 800 Kbits/ σ = 240 Kbits	μ = 3.2 Mbits/ σ = 960 Kbits
P Frames	iid – Log Normal	μ = 240 Kbits/ σ = 160 Kbits	μ = 960 Kbits/ σ = 640 Kbits
B Frames	iid – Log Normal	μ = 80 Kbits/ σ = 24 Kbits	μ = 24 Kbits/ σ = 96 Kbits
Description	μ and σ are the mean and variance of the log-normal distribution, respectively. a1 = 0.53, a2 = 0.15 and AREpsilon = normal (0, 392)		

A Traffic Model based on reference : [Krunz]

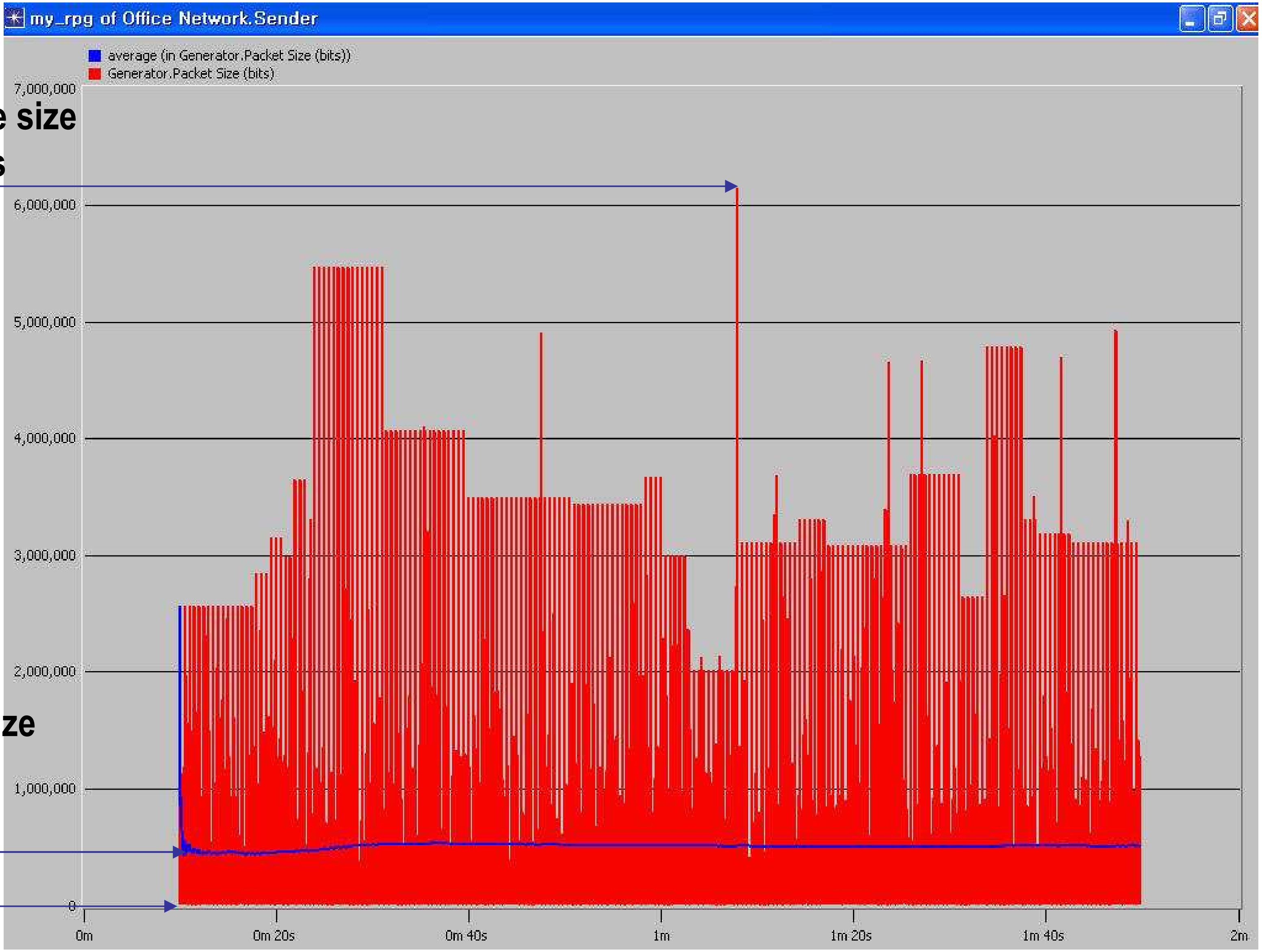
OpNet MPEG2 module developed by Sharp Laboratory of America

Simulation Case – Single Sender & Receiver

❑ Topology



Packet Size fluctuation

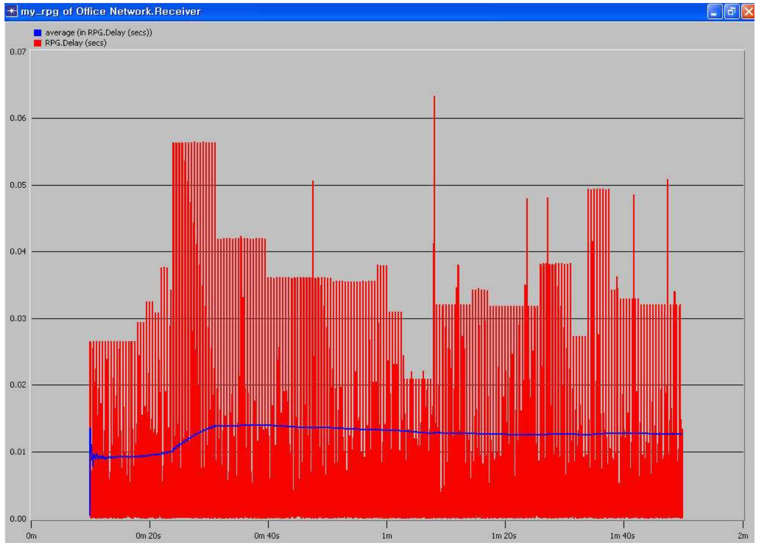


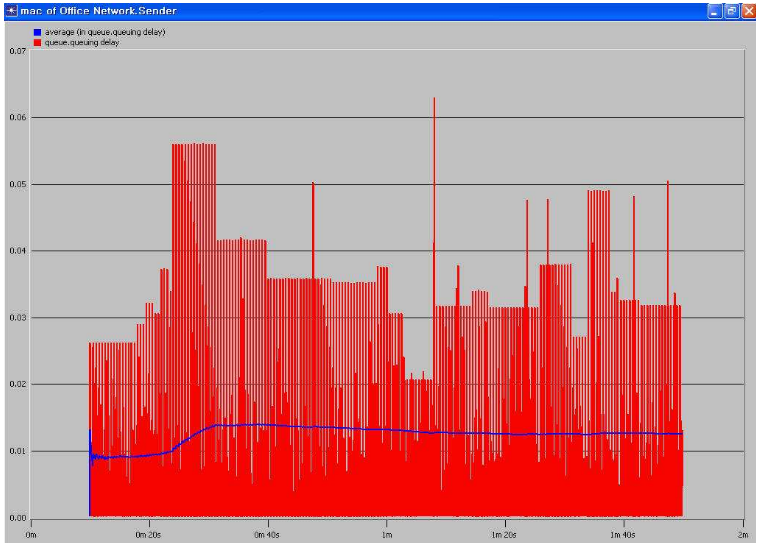
Maximum frame size
6.141 Mbits

Average frame size
461kbit/frame
→ 13.85 Mbps

Minimum
30 bits

ETE Delay and Queueing Delay





- Delay variation occurs because of various frame sizes of MPEG-2 traffic
  - Especially the difference of I, P and B frame size makes huge delay variation
  - Average delay is approximately 0.01266 sec
  - Maximum delay is approximately 0.06332 sec

Average Delay Comparison

- ❑Queueing delay in the MPEG adoption layer MAC of sender also delay variation because of the fluctuated incoming traffic rate
  - Average queueing delay of sender is 0.01258 sec
  - Maximum delay is approximately 0.06297 sec
- ❑ Longer delay performance is cause by queueing delay (Ethernet frame segmentation from long MPEG-2 frames)

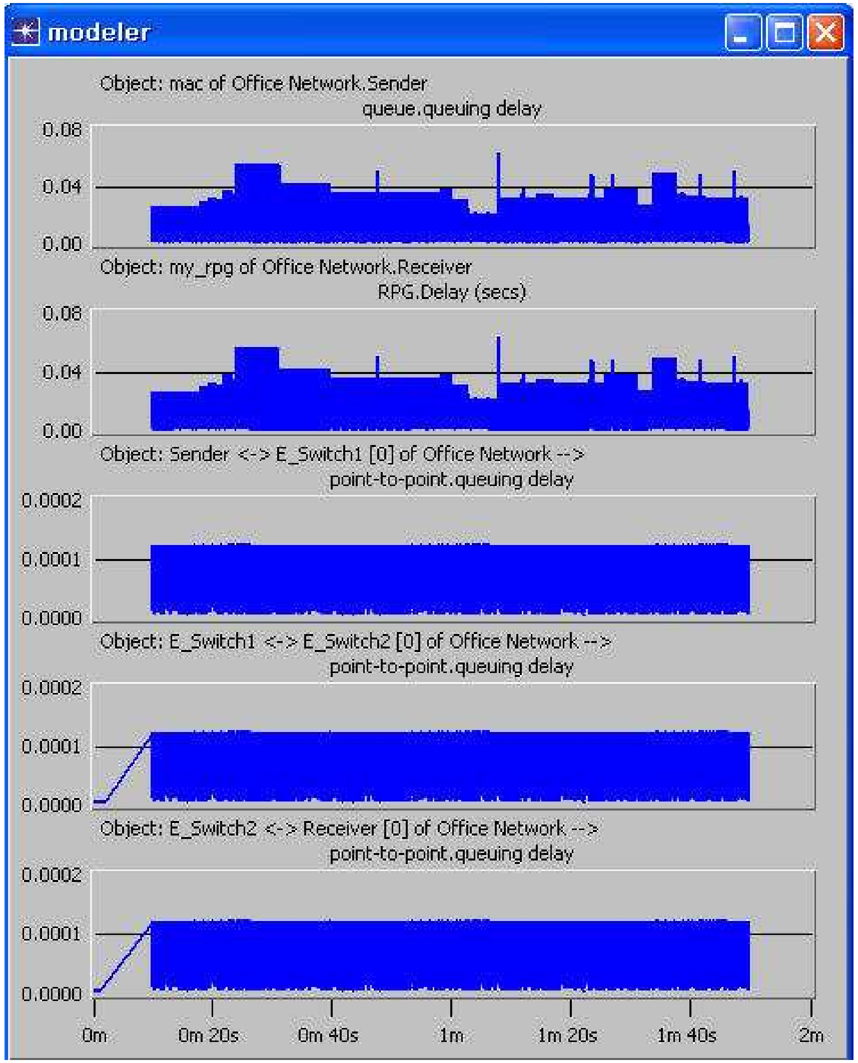


Table of approaches: QoS Control for Video and Audio communication in Conventional and Active Networks

Scheme	Operation	Advantages	Disadvantages
WFQ	Packets are sent in the increasing order of their service finishing times	•Provides end-to-end delay bound •Provides a fair share of bandwidth to each flow •Is applicable to various size packets	•Involves complex computation due to updating virtual finishing times upon each packet arrival •Increases the waiting time for each flow queue when the number of flows is increased
WF^2Q	The packet that has started service and has the smallest finishing times is transmitted first.	•Provides end-to-end delay bound •Provide worst-case fairness •Is more fair than WFQ, but has lower delay bound than WFQ	Involves complex computation of finishing and starting times of the packets
SCFQ	Packets are sent in the increasing order of their service finishing times. The calculation of finishing time is simplified through the use of an approximation algorithm, as compared to WFQ.	•Provides end-to-end delay bound •Is less complex than WFQ and WF^2Q	•Is less fair than WFQ •Degrades delay bound of WFQ
WRR	A rotation is used where each flow is served in relation to their weights, and the weight corresponds to the number of packets.	•Provides end-to-end delay bound •Is simpler than WFQ	•Is less fair than WFQ, W^2FQ, and SCFQ •Involves a higher latency than WFQ, W^2FQ, and SCFQ •Cannot distribute bandwidth fairly in systems which have variable packet length
Delay-EDD	Packet with earliest deadline is transmitted first.	Guarantees worst-case delay	Tends to increase end-to-end jitter over long paths
Jitter-EDD	Like Delay-EDD, packets are scheduled based on their deadline. Unlike the Delay-EDD, an input regulator is added, which holds the packet until the time it is expected to arrive at the node. At that time the packet is eligible to be scheduled.	Ensures the minimum and maximum delay	Is more complex than Delay-EDD
$FIFO^+$	Jitter is minimized by sharing jitter among hops, i.e., when a packet is scheduled before its deadline at one hop it is delayed at the next hop.	Provides end-to-end jitter bound	Is more complex than Delay-EDD

Table 3. *Summary of packet scheduling schemes in IS networks.*

[Yan Bai and M. R. Ito, "QoS Control for Video and Audio communication in Conventional and Active Networks: Approaches and Comparison", IEEE Communications, 1st Quarter 2004, Vol. 6, No. 1

Aggregated flow based scheduling

- [Bernet00] suggested "IntServ over DiffServ", where the admission control is based on individual flows, yet the scheduling is based on the class, the aggregated flow.

- [Cobb02] showed that if the aggregation of flows is performed *fairly*, then an upper bound on end-to-end delay is guaranteed to the constituent flows. The end-to-end delay with flow aggregation is smaller or equal to the end-to-end delay without the aggregation.

Taxonomy of schedulers for Delay bound guarantee

- Work-conserving
  - Individual flow based (Latency-Rate server class)
    - Sorted-priority based
    - Round robin based
  - Aggregated flow based [Cobb02], [Sto99],[SCED+]

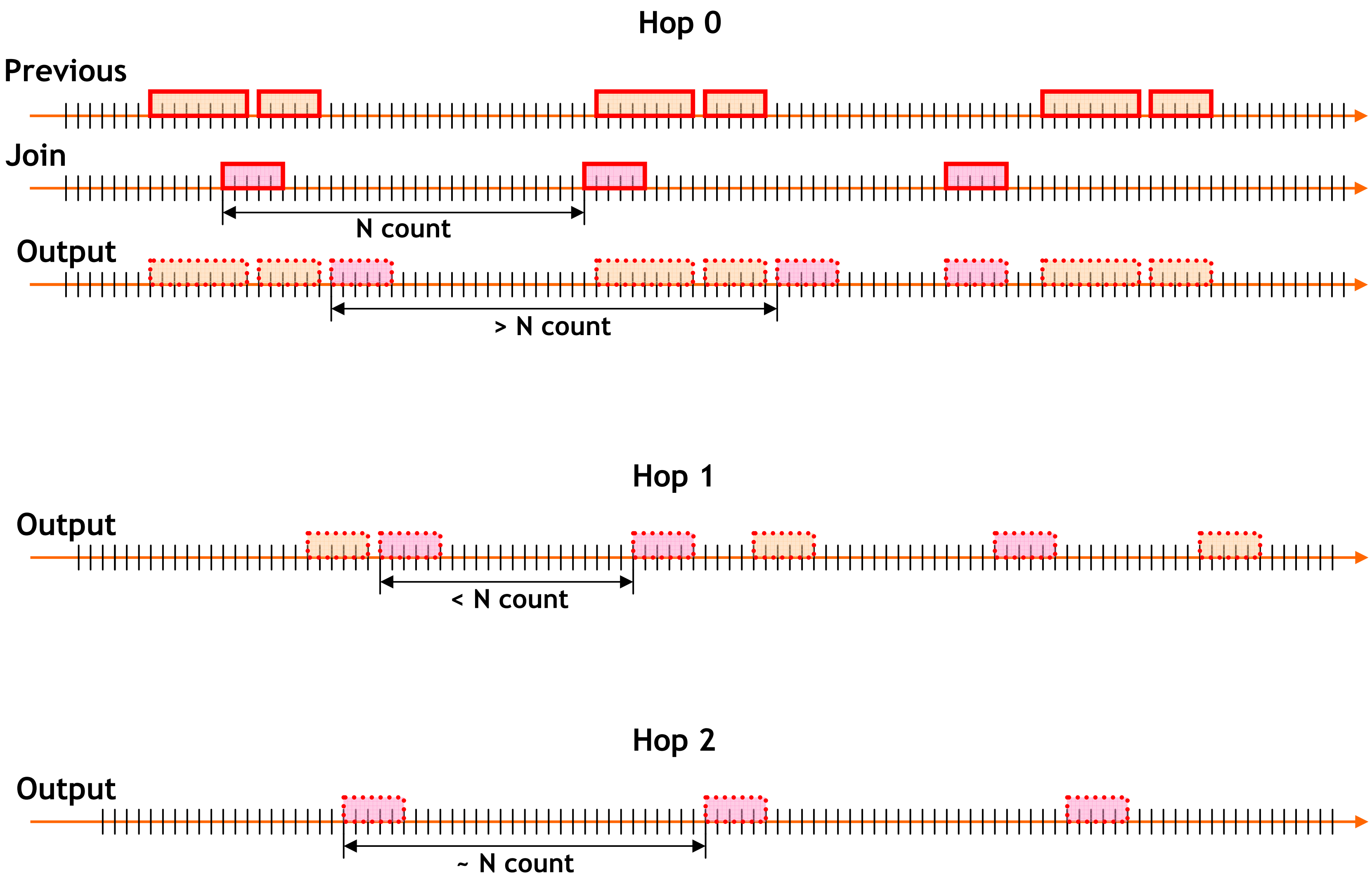
Achievement through intensive research on Packet Data Networks

- Non work-conserving
  - "Reshaped at each node" scheme [Zhang92]
    - Variations: Stop-and-Go, HRR, Jitter-EDD
  - IEEE1394-type Isochronous scheduling
  - TDMA (why not, like 802.15.3 and 4)

Large delay, small delay variations

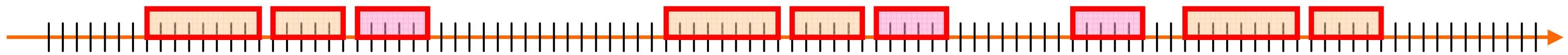
Not even based-on the principles of Packet Data Networks

An example of scheduling: *Counter based isochronous packet scheduling*

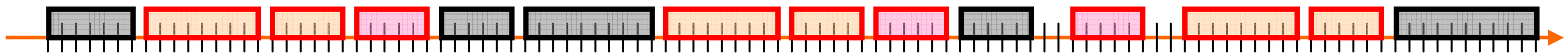


An example of scheduling (2): *Counter based isochronous packet scheduling*

First, Isochronous packets are mapped



Second, asynchronous packets are mapped and segmented based on the empty slots status



- ❑By use of holding, frame segmentation of asynchronous packet may enhance the utilization and reduce the maximum delay of those packets.

- ❑But to do this, end-device or some switching node that located on the border of ResE cluster should have reassembly functionality

Considerable parameters for performance evaluations

#	Category	parameter
1	delay related parameters	a. short term delay variation (packet jitter, delay variation, etc)
		b. long term delay variation (wander)
		c. absolute delay (latency)
2	network time synchronization	a. Time synchronization accuracy
		b. synchronized time duration
		c. MTIE mask
3	resource usage efficiency	resource usage efficiency
4	service (application) response time	service (application) response time
5	scalability for	a. span length
		b. future survivability
		c. upgradeability (how easily can add new features)

Suggestions

- Use of same version of simulator.
- Share a simulation module and testing by the ResE group members, to get more reliable results.
- Collaborate work for modeling and designing of reference model, use cases, scenarios.
- MPEG2 traffic model should need to be verified in order to define commonly used model.

References

- [Krunz]
  - Marwan Krunz and Herman Hughes, “A Traffic Model for MPEG-Coded VBR Streams”, pp.47-55, SIGMETRICS’95
- [Bernet00]
  - Y. Bernet, et. al., “A framework for integrated services operation over DiffServ networks,” RFC 2998, Nov. 2000.
- [Cobb02]
  - J. A. Cobb, “Preserving Quality of Service Guarantees in Spite of Flow Aggregation”, IEEE/ACM Transactions on Networking, vol.10, no.1, Feb. 2002
- [Sto99]
  - I. Stoica and H. Zhang, “Providing guaranteed services without per flow management,” in Proc. ACM SIGCOMM, Cambridge, MA, Aug. 1999, pp. 81–94.
- [SCED+]
  - R. L. Cruz, “SCED+: Efficient management of quality of service guarantees,” in Proc. IEEE INFOCOM, vol. 2, Mar. 1998, pp. 625–634.
- [Zhang92]
  - H. Zhang, “Service disciplines for packet-switching integrated services networks,” Ph.D. dissertation, Univ. of California, Berkeley, 1992.