The 2nd IP-Telephony Workshop

A Simulation Analysis of Aggregation Strategies in a WFQ+ Schedulers Network

Authors: R.G. Garroppo, S. Giordano, S. Niccolini, F. Russo

Topics and summary

The main topics of this presentation will be:

- DiffServ Architecture
- Aggregation Strategies in a DiffServ Environment
- WF^2Q+ Schedulers Network
- QoS parameters for Voice over IP

This presentation will show how we:

- Implemented a real DiffServ scenario (using Opnet Modeler 6.0)
- Set scheduler parameters
- Analyze and test different aggregation strategies in a DiffServ environment in order to provide the desired QoS to voice traffic

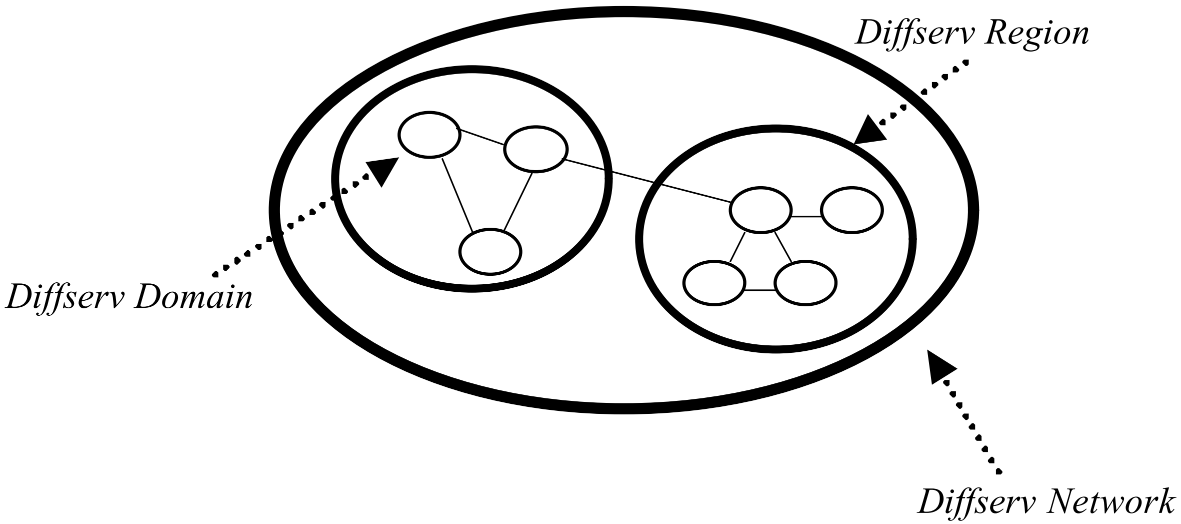
Finally we will show:

- The simulation results
- Our conclusion

DiffServ Architecture

• The DiffServ are still a matter of research because they are not completely standardized

Here is the reference DiffServ model architecture designed in order to provide scalability to the network



The IETF has standardized three service classes with different characteristic:

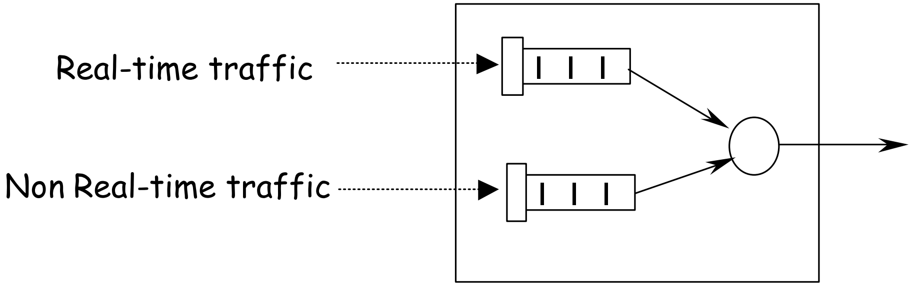
- Expedited Forwarding (EF)
- Assured Forwarding (AF)
- Best Effort Forwarding (BE)

but the application to be forwarded on them are not specified

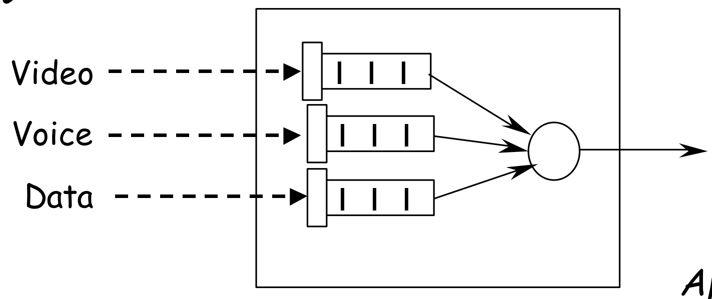
Aggregation Strategies

• **In order to provide somewhat relevant we focused on investigating the aggregation issue to follow in DiffServ environment**

• **It is clear that is necessary to divide the network traffic into two type of traffic (real-time and non real-time); this comes out considering the loss, delay and jitter characteristics of the two types of flows**



• **We decided to make a deeper investigation and we carried out a simulation scenario where we divided network traffic into three classes (video, voice and data)**

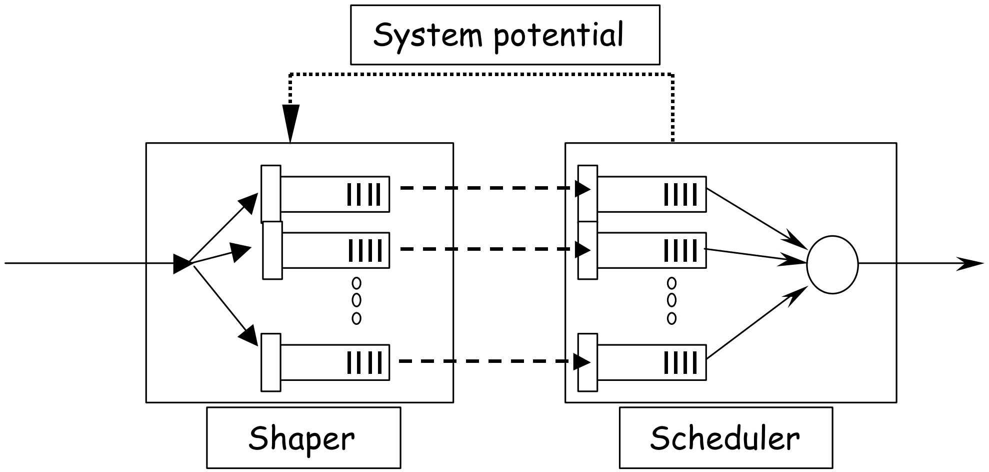


WF^2Q+ Scheduler

• WF^2Q+ (Worst Case Fair Weighted Fair Queueing) is a GPS (Generalized Processor Sharing) approximating service discipline with high fairness properties and relatively low implementation complexity

• WF^2Q+ uses a system potential function in order to schedule the packet trasmission (it belongs to Latency Rate servers class)

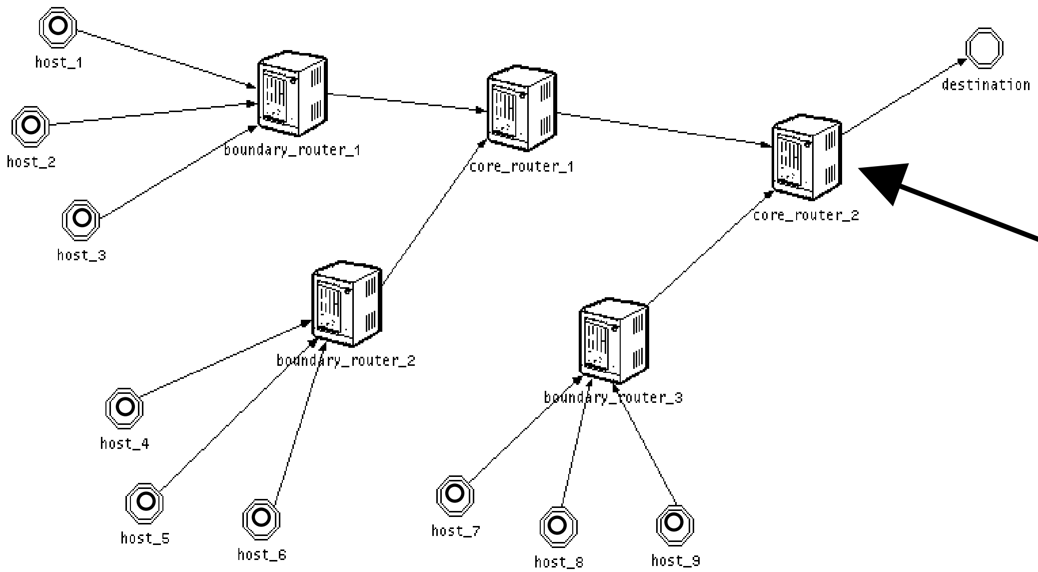
• The shaper here designed is used in order to improve scheduler's fairness (the target is to reach GPS fairness)



• Every queue has associated a weight (ϕ_i) which indicates the portion of the available bandwidth used by that queue

Simulation Scenario

Network Topology



Host 1,4,7:

- **15 voice source each**

Host 2,5,8:

- **Data traffic**

Host 3,6,9:

- **Video source**

Each network router has inside:

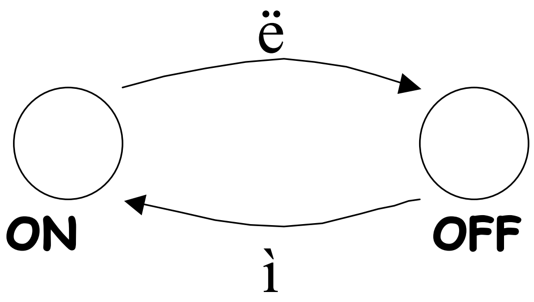
- **classifier**
- **marker (only in boundary)**
- **scheduler (WF^{2}Q+)**

The simulation scenario is implemented using OPNET Modeler vers 6.0 CAMAD (Computer Aided Modeling And Design) tool

Source Models

We adopted a model only for the voice sources (for other kinds of traffic we have considered actual traffic data)

Voice Model: "On-Off" (typical behavior of a voice source with Voice Activity Detection)



Mean values (exponential distribution)

mean_time_on = 1/ë = 0.35 sec

mean_time_off = 1/ì = 0.65 sec

R = 64 kbit/sec (during active periods)

Video Sources: output of a MPEG1 encoder loaded with different sequences of movies (Goldfinger, Asterix, Simpsons)

Video flow	Mean_rate (Mbps)	Peak_rate (Mbps)
GOLDFINGER	0.584	5.87
ASTERIX	0.537	3.54
SIMPSONS	0.446	5.77

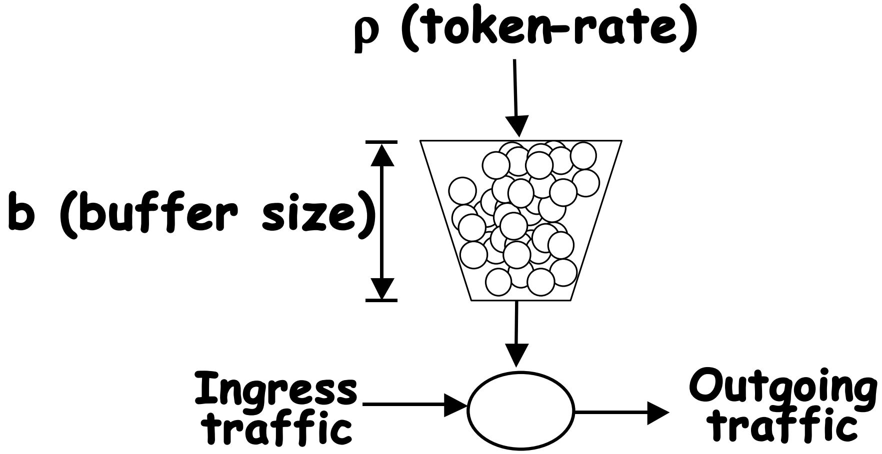
Data Sources: traffic exchanges by the Univerity of Pisa with the external world (recorded data traces)

LBAP Characterization

“Token-bucket”

- The traffic produced by a single source is upper bounded by the relation:

$$A(T) \le \rho T + b \ , \forall T$$



Upper bound for the end-to-end delay when passing through a Latency Rate Scheduler:

$$D_{\max} \le \frac{b_i}{\rho_i} + \Theta_i$$

where Θ_i is the latency term of the i-th flow

Extending the analysis to a network of KWF²Q+ schedulers:

$$D_i \le \frac{b_i}{\rho_i} + \sum_{j=1}^{k} \Theta_i^j$$

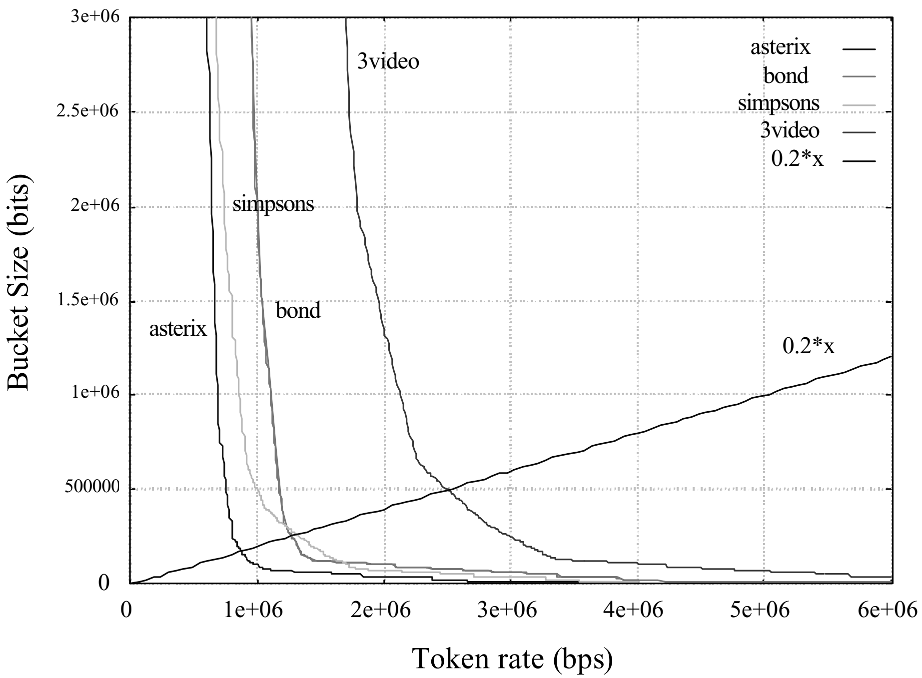
LBAP traffic characterization is conservative with respect of the statistical model approaches so:

$$D_i \le \frac{b_i}{\rho_i}$$

LBAP Curves

- **Starting from the previous presented results** $D_i \le \dfrac{b_i}{\rho_i}$

- **We characterize the sources setting a maximun delay bound D_i and finding where the LBAP curve intersect the straight line:** $b_i = \rho_i D_i$



Here it is the characterization we obtained:

Traffic flow	Rate (ρ) (Mbps)	Buffer (Kbit)	Dmax (msec)
GOLDFINGER	1.25	250	200
ASTERIX	0.83	160	193
SIMPSONS	1.27	260	205
15 VOICE sources	1.10	30	27
Data	0.40	400	1000

Simulation Results (1)

• The simulation analysis is mailnly focused on the evaluation of the impact of different aggregation strategies on the QoS parameters

(QoS parameters: whole set of properties which characterize the netwok traffic)

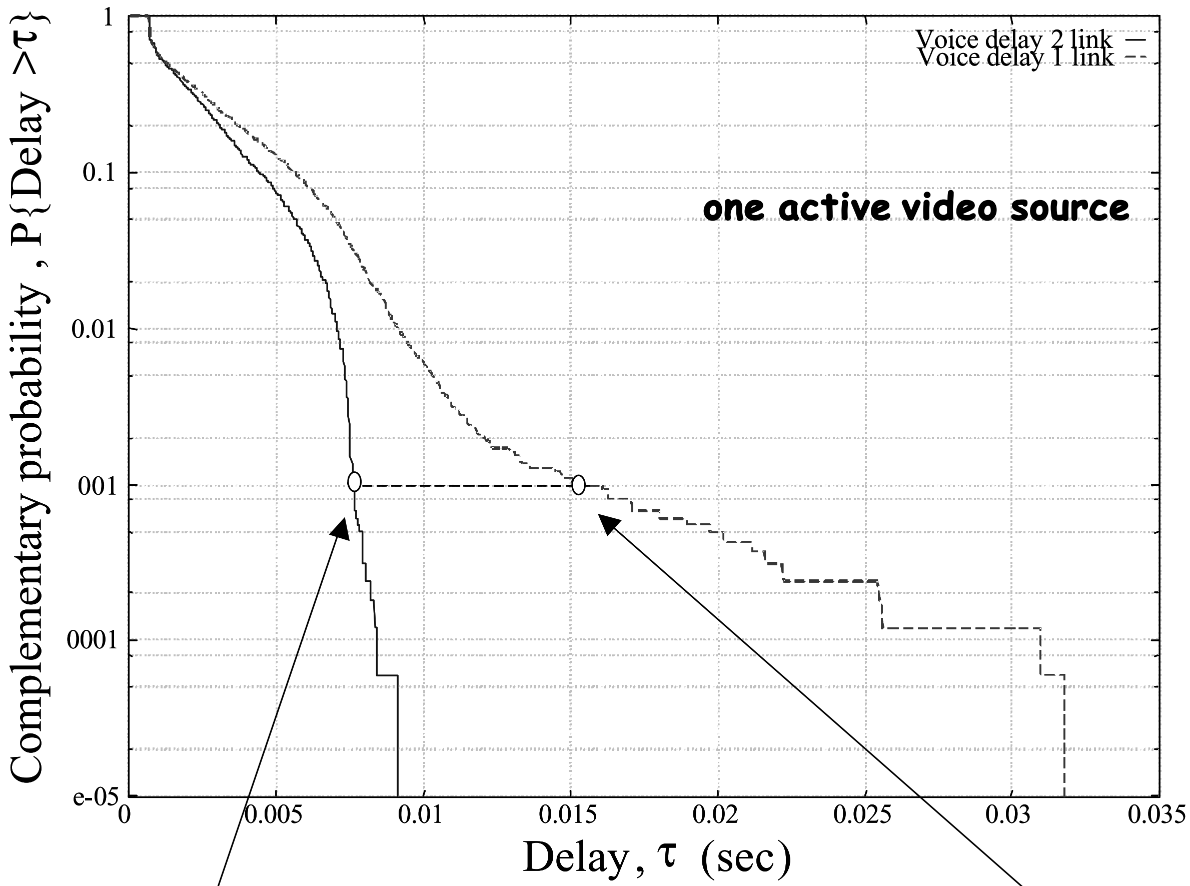
FIRST TEST

Scenario 1 link: One video source and voice traffic are carried toghether in EF class while data traffic is carried in a BE class

Scenario 2 link: One video source is carried in EF class while voice traffic is carried in AF class, data traffic is still carried in a BE class

Simulation Results(2)

Complementary Probability of Voice Delay



Here is possible to notice the goodness of our assumption in traffic characterization
Delay experimented is under 27 msec with a probability of 99.99% (1 link) and with a probability of 100% if video and voice are not merged in a single queue

Delay =~15msec; Prob{Delay>0.015}=0.1%

Delay =~8 msec; Prob{Delay>0.008}=0.1%

Simulation Results(3)

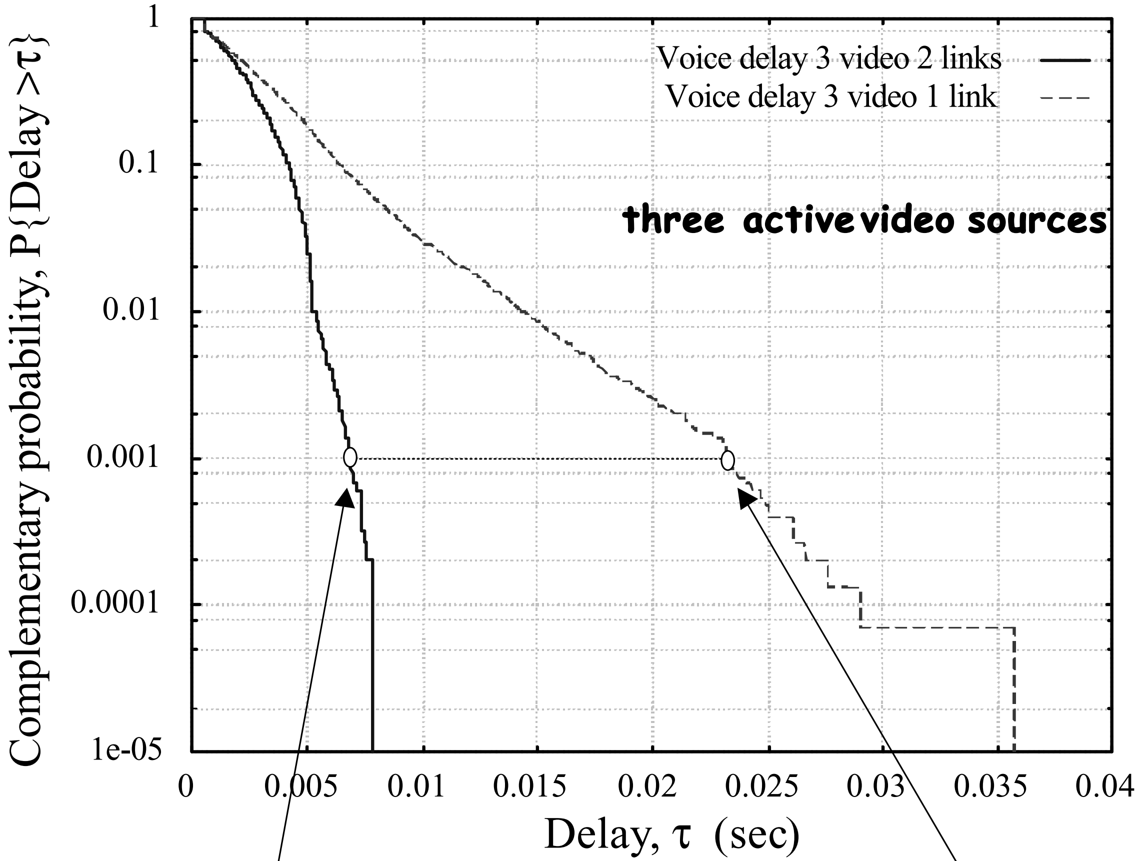
SECOND TEST

Scenario 1 link: Three video source and voice traffic are carried toghetherin EF class while data traffic is carriedin a BE class

Scenario 2 link: Three video source are carried in EF class while voice traffic is carriedin AF class, data traffic is still carriedin a BE class

Simulation Results(4)

Complementary Probability of Voice Delay



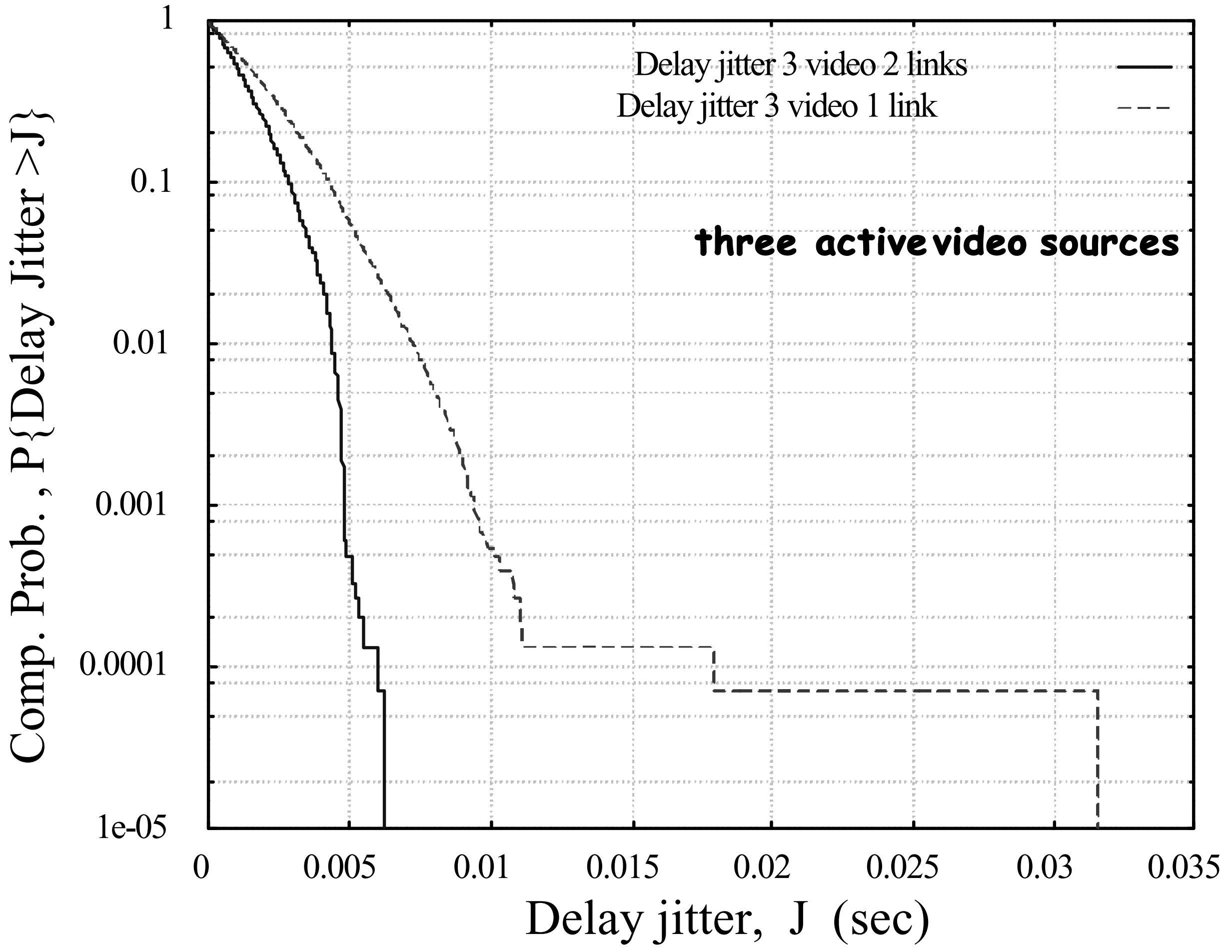
Once again it is possible to observe the performance degradation when voice and video are carried in a single class (here the degradation is more evident)

Delay =~6 msec; Prob{Delay>0.006}=0.1%

Delay =~23msec; Prob{Delay>0.023}=0.1%

Simulation Results(5)

Complementary Probability of Voice Delay Jitter



Performance worsening can be observed as we consider the jitter parameter, too.

Analyzing results

• Degradation in terms of end-to-end delay when the video flows are merged in the same queue with the voice traffic

Due to "lock-out" phenomenon? (video packets are much greater than voice ones) When voice traffic is carried in a his own queue the voice packets have not to wait behind video ones and the delay can be reduced

• Degradation is amplified when the number of video sources is increased

• Video performance takes benefit from aggregation (even if the rate allocated to the three video toghether is less than the sum of the three single rate charaterization).

• Performance worsening affects not only end-to-end delay but delay jitter, too.

Conclusion

- Wrong aggregation of traffic flows with different statistical features (such as video and voice) may lead to performance worsening
- Achieving desired QoS is not only matter of aggregation issue, also "fair scheduling" has to taken into account
- "Right aggregation" and "fair scheduling" has to be followed by an adequate setting of sheduler parameters (as proposed)

THAT'S ALL