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# Performance evaluation of an IP voice terminal

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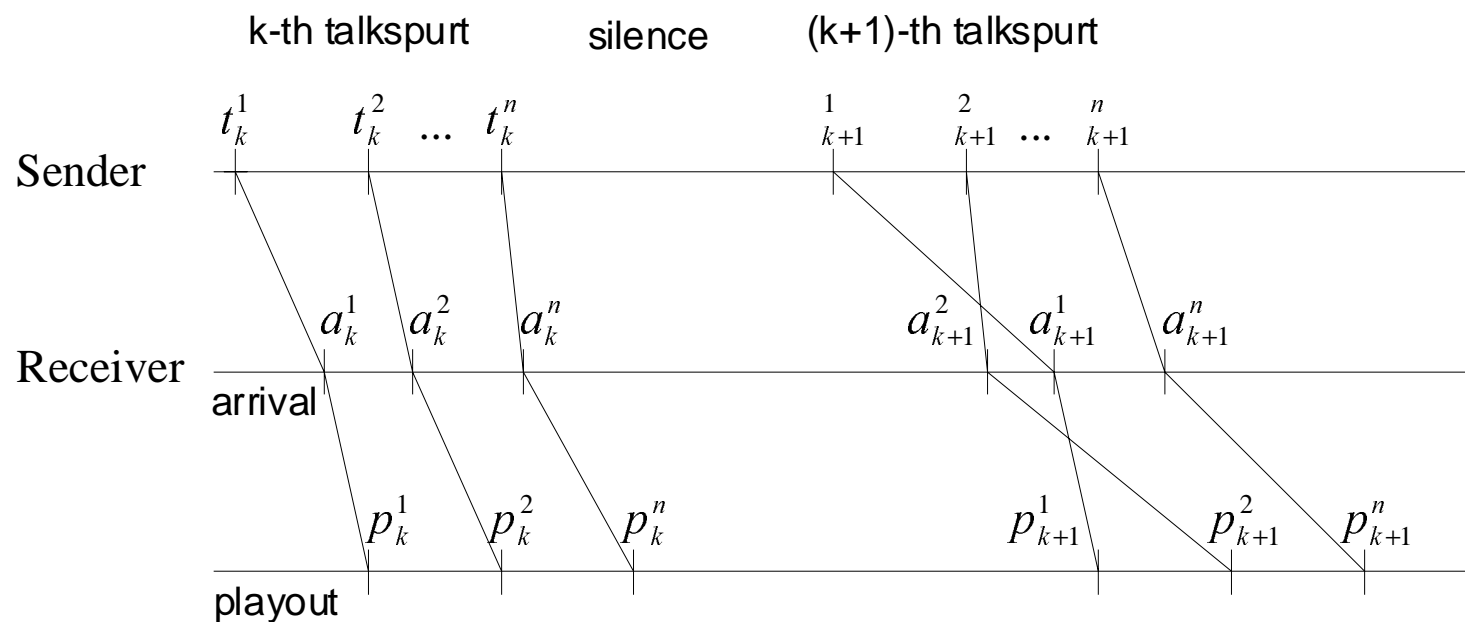
# *Delays in IP Voice*

- Framing delay (e.g. 20 ms)
- Processing delay:  
Coding & decoding, Dependent on used algorithm
- HW delay: A/D & D/A transform
- Network delay: Jitter
- Buffering
- Operating system: Scheduling



# *Playout delay adjustment*

- Fixed
- Adaptive: Per-talkspurt or Per-packet





# Playout algorithms

## Algorithm 1:

A linear predictor

$$\alpha = 0,998002$$

$$\hat{u}_k^i = \alpha * \hat{u}_k^{i-1} + (1-\alpha)\hat{d}_k^i$$

$$\hat{v}_k^i = \alpha * \hat{v}_k^{i-1} + (1-\alpha) | \hat{u}_k^i - \hat{d}_k^i |$$

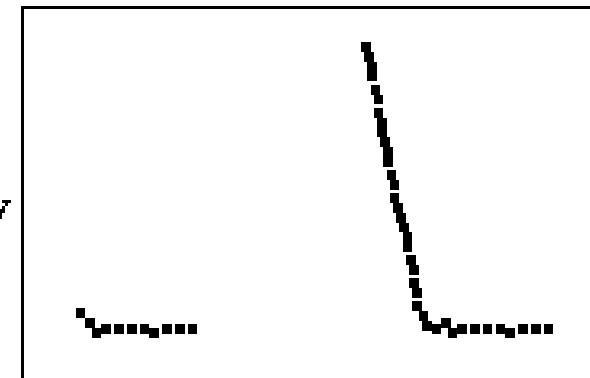
$$\hat{p}_k = \hat{u}_{k-1}^{n_{k-1}} + \beta * \hat{v}_{k-1}^{n_{k-1}}$$

## Algorithm 2:

Two operational modes

- Normal mode
- Spike mode

Network delay



Time of arrival

A typical spike



## **Algorithm 3:**

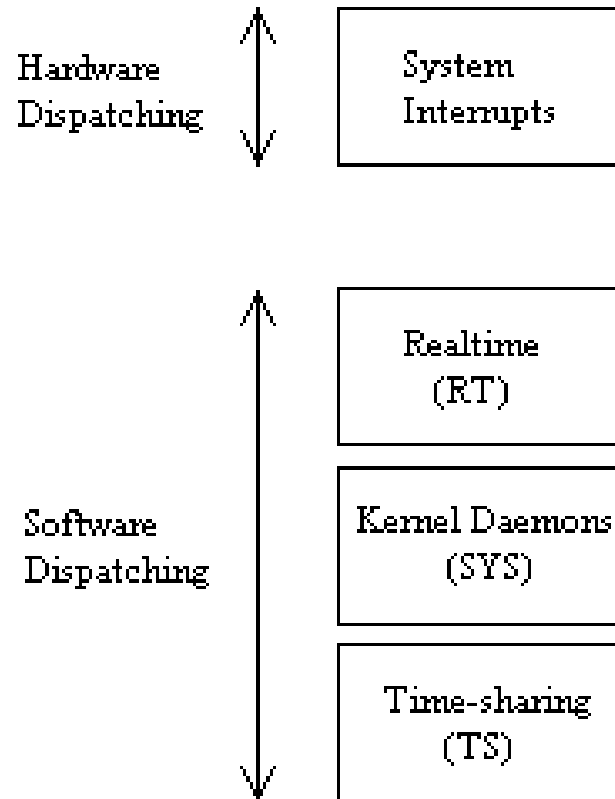
- Playout delays are calculated by using collected delay statistics
- Normal mode & spike mode
- Takes some time to first collect the statistics

## **Algorithm 4:**

- A combination of algorithms 1 and 3



# *Unix process scheduler*



Dispatch priorities for scheduling classes



# Measurements and results

<i>Used CPU time</i>	<i>PCM</i>	<i>ADPCM</i>	<i>GSM</i>	<i>LPC</i>
<i>in ms</i>	0,26	0,49	1,14	1,83
<i>in percent</i>	1,3	2,5	5,7	9,1

**Used CPU time in ms and % of used CPU time from total CPU time with different audio codings**

<i>Delay component</i>	<i>Delay in ms</i>
Framing delay	20,0
Processing delay	0,5 - 3,7
HW delay	2,2
Network delay	0,5
OS delay	0 - 2,9
Buffering delay	8,2 - 8,8

## **Used workstation:**

Sun Ultra Enterprise 1

Operating system: SunOS 5.5.1

## **Used software:**

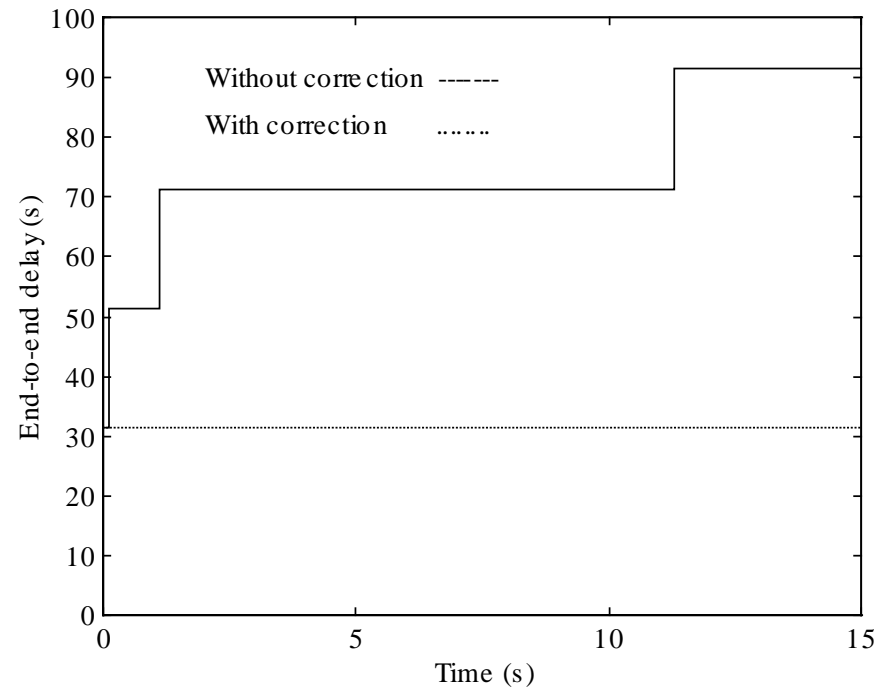
Nevot v.3.35

**Components of the end-to-end delay**





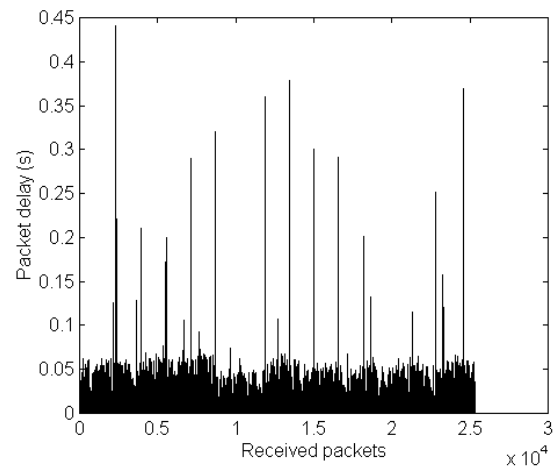
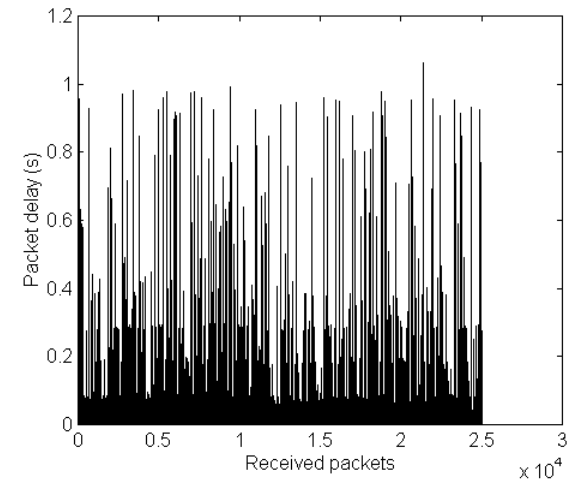
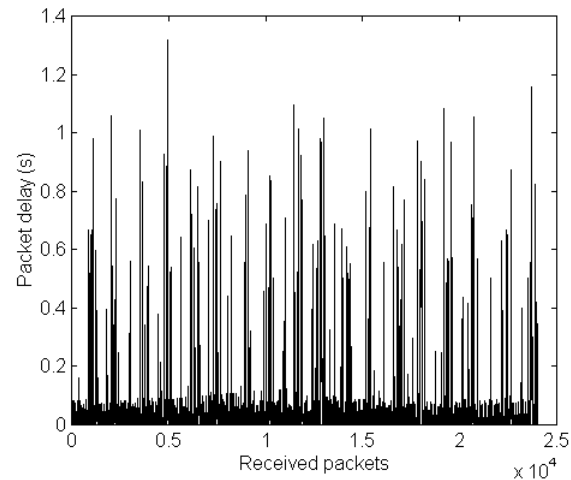
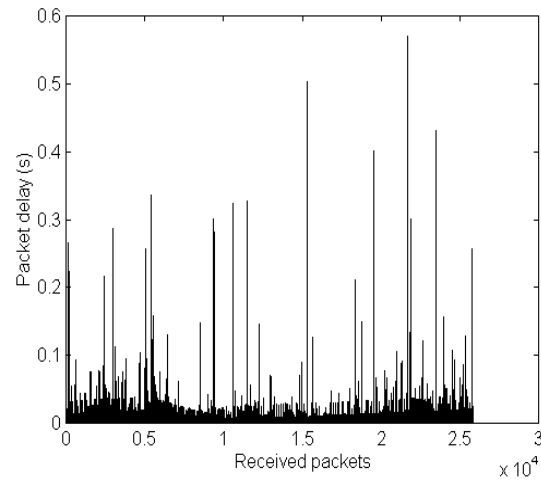
# Measurements and results



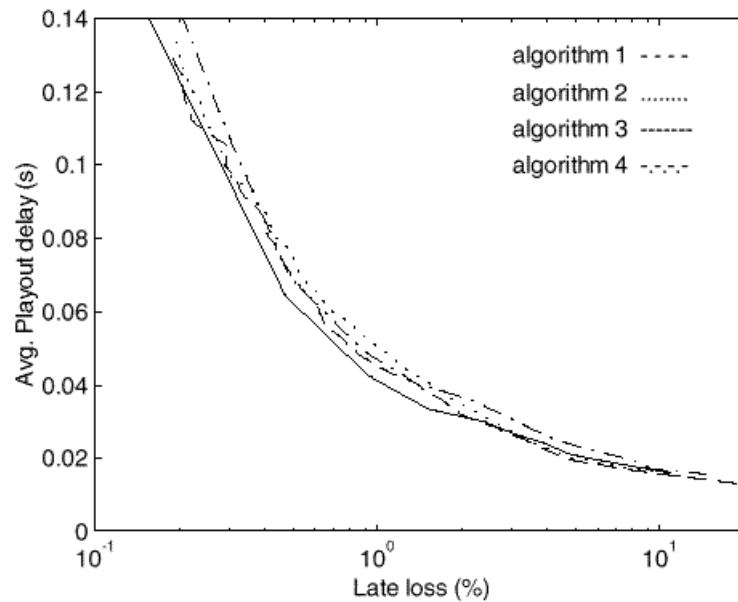
End-to-end delays with and without correction



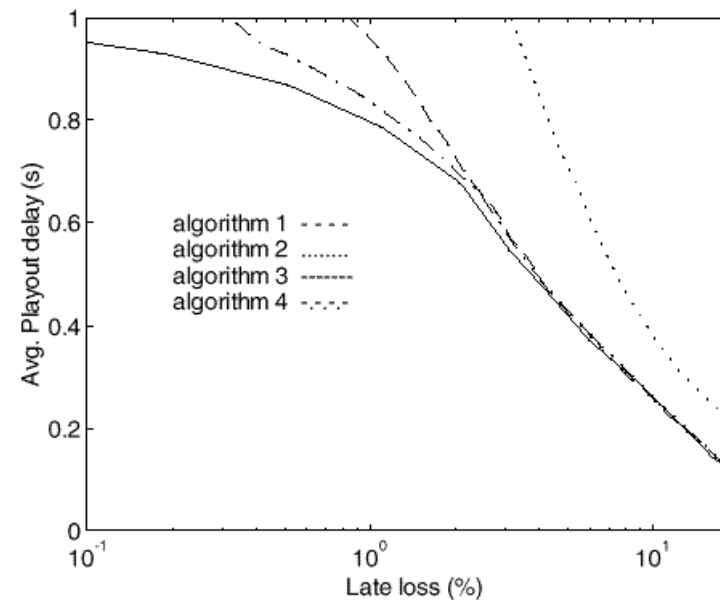
# Comparison of playout algorithms



<i>Trace</i>	<i>Traffic description</i>	<i>Frames/s</i>	<i>Bytes/frame</i>	<i>Load/Mbps</i>	<i>Length/packets</i>
Trace 1	Small packets	3000-5000	160	3,936-6,560	25830
Trace 2	Small packets, high load	2000-3500	320	5,184-9,072	25023
Trace 3	Large packets, bursty load	100-860	1450	1,163- 10,000	24048
Trace 4	Variable size packets, bursty load	200-860	160-1450	0,2624- 10,000	25337



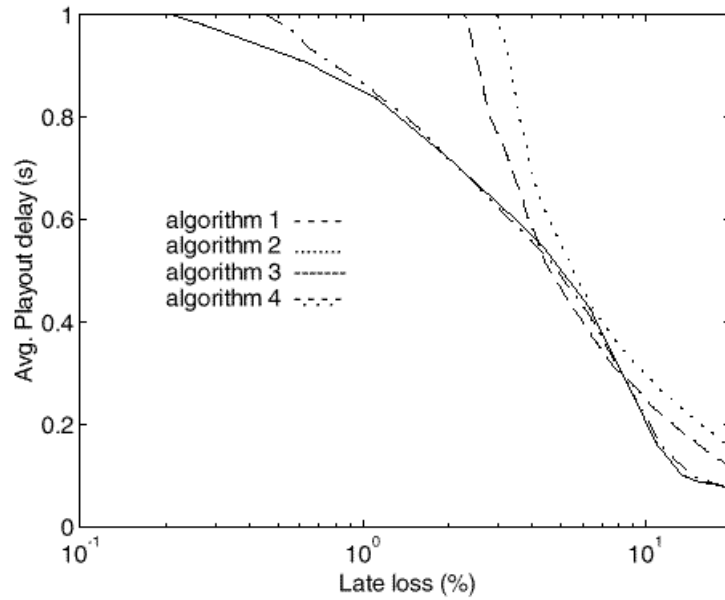
Algorithms 1-4 on trace 1



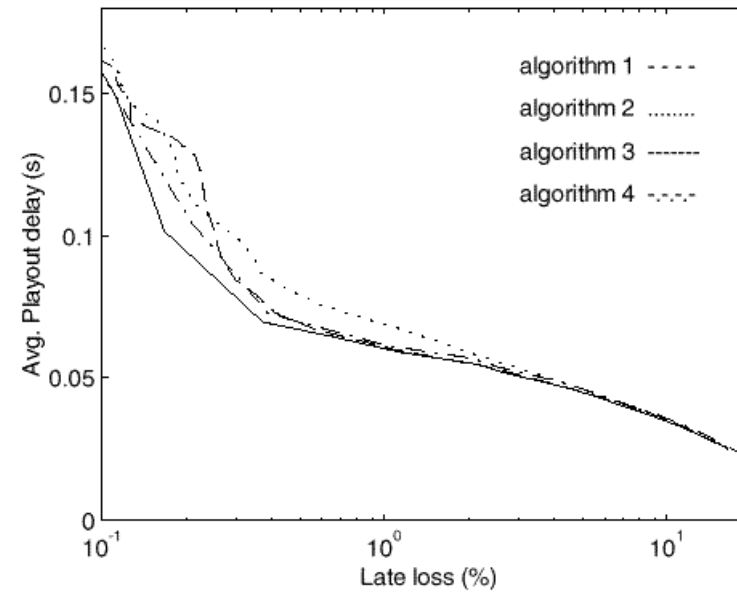
Algorithms 1-4 on trace 2



# Comparison of playout algorithms



Algorithms 1-4 on trace 3



Algorithms 1-4 on trace 4



## *Conclusions*

- Significant part of the delay generated by the VoIP client software (Nevot)
- Other delays quite small in an unloaded LAN
- A new playout algorithm was presented as a combination of two existing algorithms. Best performance is obtained during calls over 250 seconds

### **Future work:**

- Simulations with actual network traces and comparison with results presented in this paper