

# Sicsophone: A Low-Delay Internet Telephony Tool

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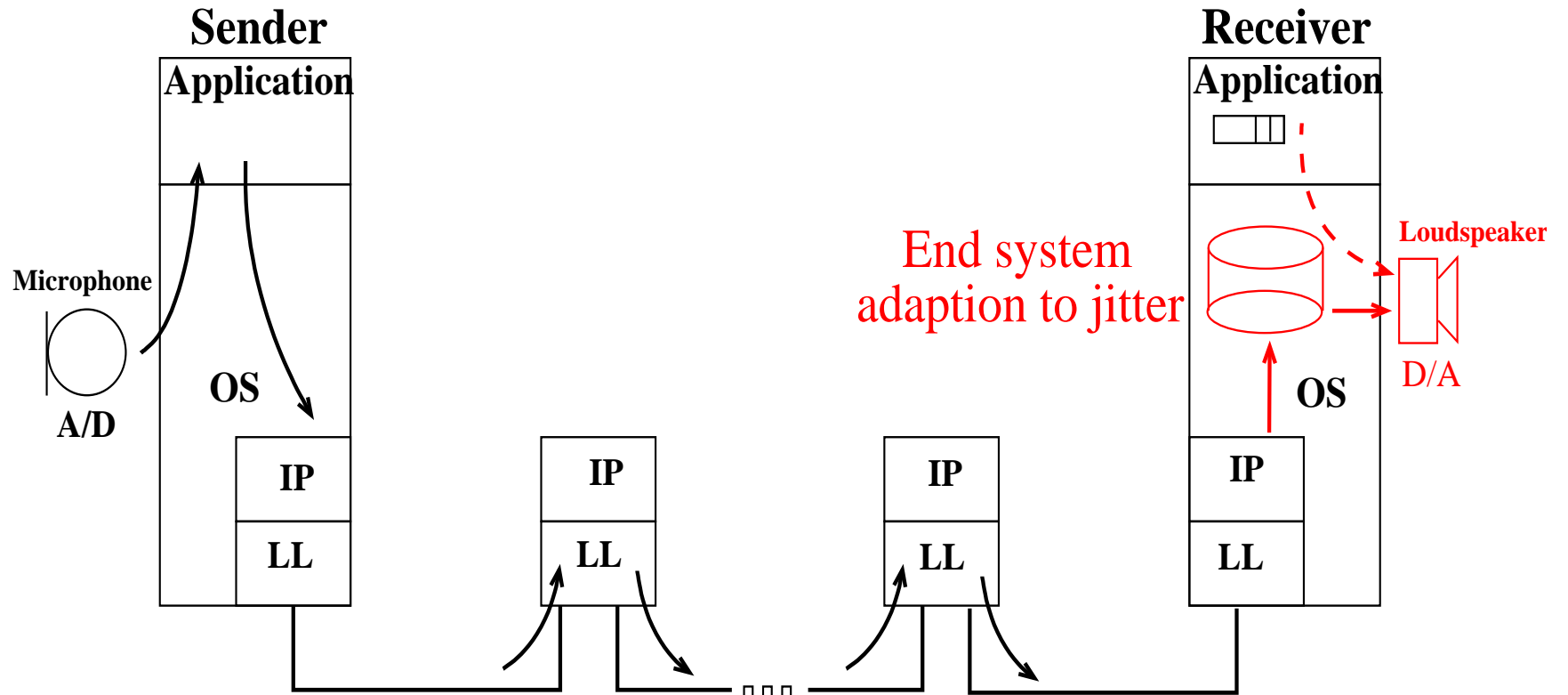
## Some words on quality

- Delay
  - Delay under 180 ms rated as good
  - Delay between 180 ms and 400 ms rated as acceptable
  - Delays over 400 ms rated as unacceptable
- Loss
  - Losses for simple audio coding e.g G.711 PCM
    - \* A loss rate of 1% with no packet loss concealment
    - \* A loss rate of 10% with packet loss concealment
- Jitter
  - Smoothing buffers used to deliver uninterrupted speech, therefore add delay

## Typical delays incurred by a receiver

Step	Process	Overhead (ms)	Depends On
1	De-packetisation	10 - 50	Packet Size
2	Decoding	10 - 50	Coding Used
3	Buffer Delay	5 - 200	Network
4	Delivery	5 - 120	End System

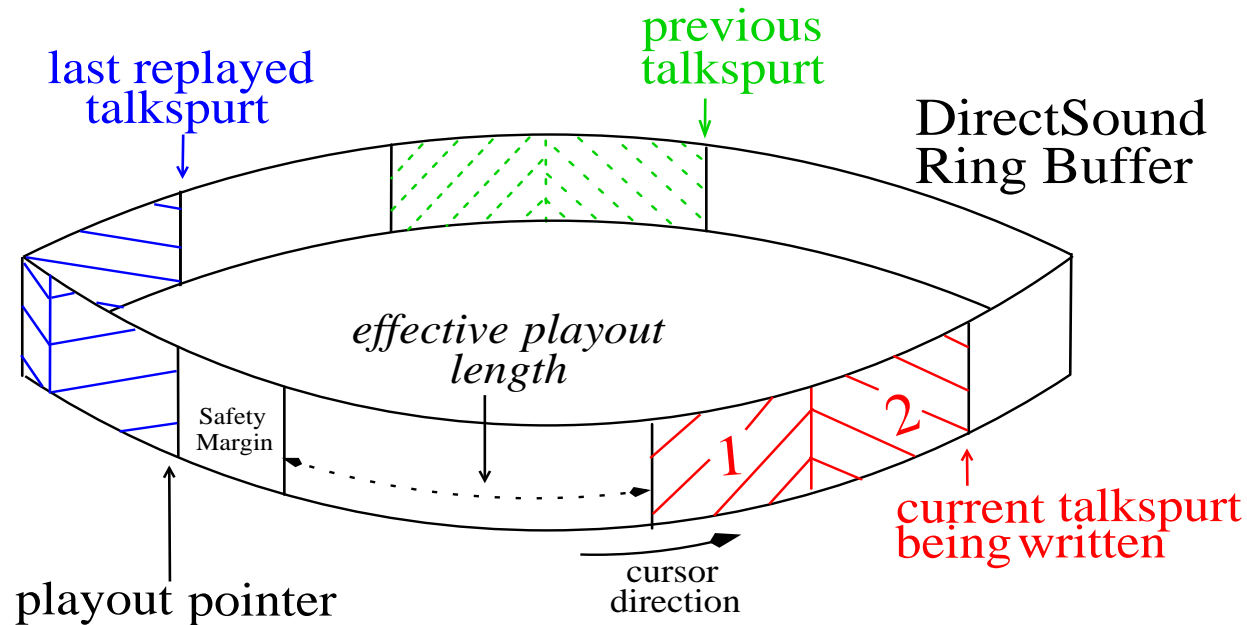
# End system adaption to jitter



# Three tier solution

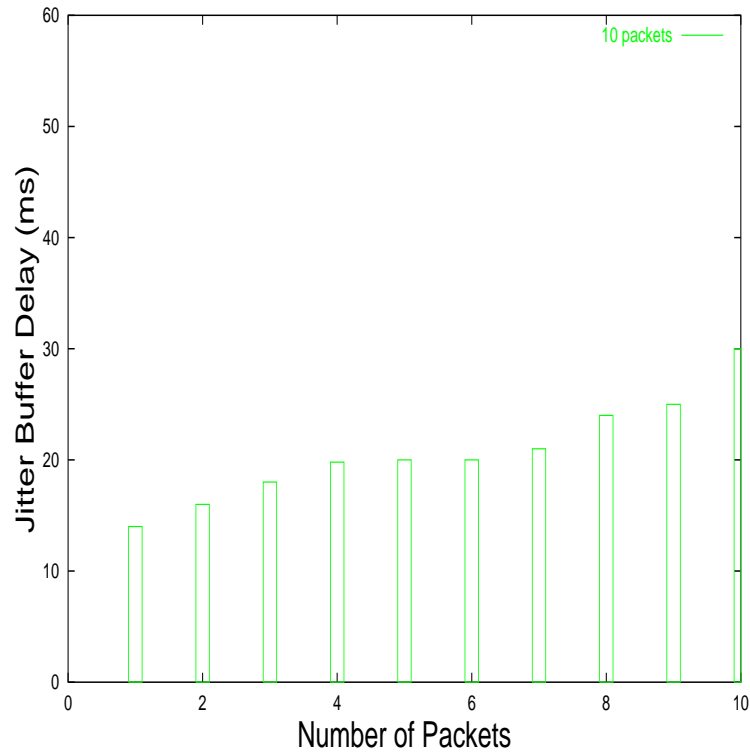
1. Use low-level functionality of the sound card
  - Memory in the audio card *becomes* the playout buffer
  - Sound card timers determine whether packets are early/late
2. Statistical per packet playout algorithm
  - Standard algorithm as proposed by Van Jacobson (1994)
  - Keep a (sorted) running history of the arrival times
  - If loss can be tolerated, the delay can be reduced
3. Limit the frequency of buffer size changes
  - Jitter calculations are bounded to upper and lower limits
  - Alleviates unnecessary changes in the system

# Directsound playout support

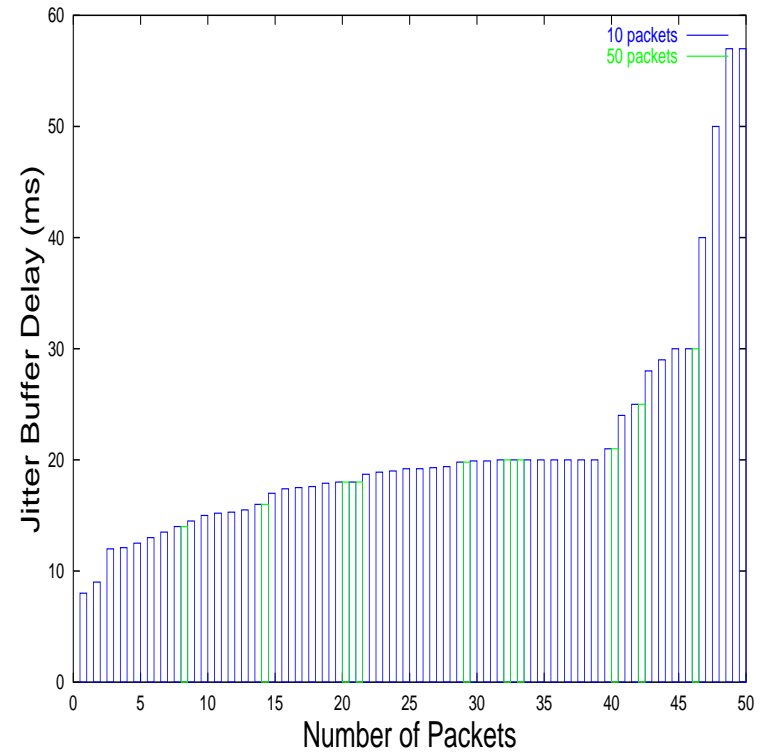


- Implemented using DirectX interface by Microsoft
- Circular buffer, pointers rotate anti-clockwise
- Talkspurts written contiguously, adapt during silence periods

# Benefit of history

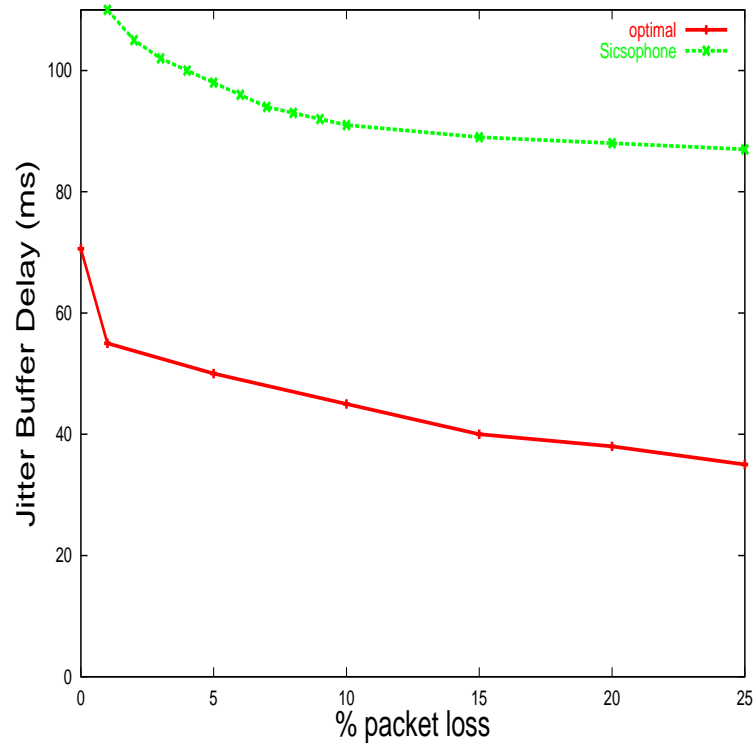


10 packets

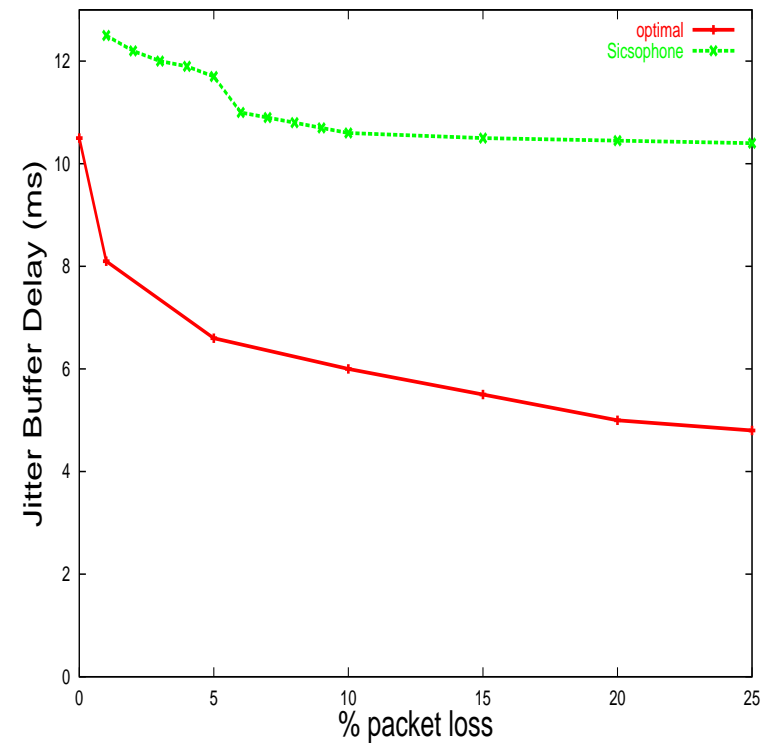


Original 10 plus 40 new arrivals

# Sicsophone compared to optimal playout



California to France



Amherst to Berlin

## Mouth-to-ear delay measurements

Audio Tool	Latency (ms)
Sicsophone prototype	25-100
Vocal Internet Phone 4.5	450-550
NetMeeting 2.1	620
VAT 3.4 (Solaris)	1200
RAT 3 (Solaris)	1500

Windows (98, NT) operating systems with SoundBlaster audio cards  
Simple square wave used, easier triggering and delay calculation

## **End system adaption to jitter conclusions**

- Other tools were not tuned, just used out of box
- Precious time can be saved by careful buffer management
- Inter-layer copying slows down delivery of audio
- Standard algorithm used, further delay reduction is possible

## Further work and references

- Add SIP signalling
- Integrate iLBC coder (GlobalIPsound)
- Lower delay jitter playout buffer

Further reading:

- Licentiate thesis “Quality aspects of audio communication”
- QoFIS 2003 paper “Wide area measurements of VoIP quality”
- Available from <http://www.sics.se/~ianm>