Quality aspects of real-time voice communication

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Vision

The Internet will carry a significant proportion of the world's voice traffic *and* the quality should be *no worse* than that offered by the traditional telephony system.

What does quality mean for a network researcher?

- Delay
  - Strictly it is not quality, rather interactivity
  - Under 180 ms rated as "good", between 180 ms and 400 ms "acceptable" and over 400 ms "unacceptable"
  - Note: once delay is in the system it cannot be removed
- Loss
  - For G.711 (POTS quality) losses can be 1-10% (if random)
  - Losses can be concealed, unlike delay
- Jitter
  - Jitter either results in delay, or loss, but not both

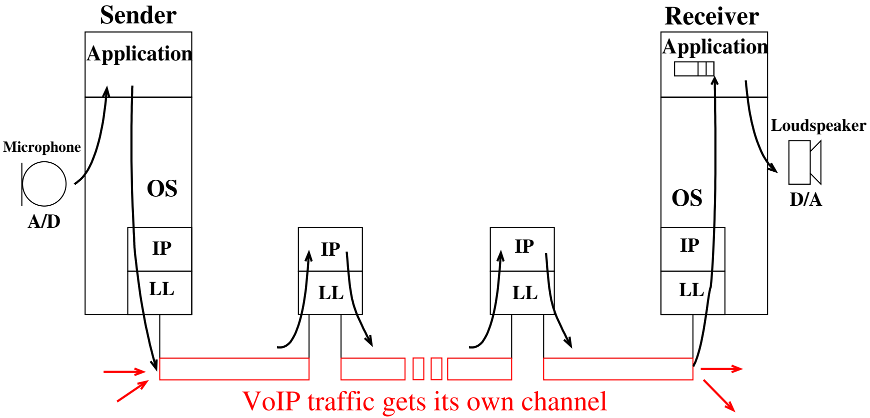
If you ask a subjective quality assessment person about quality they will tell you something very different!

Problems facing VoIP quality

First a *non-problem*. There is nothing inherently problematic with sending real-time voice across a packet-switched network. Some problems though:

- Disruption from competing TCP traffic
  - Individual flows can disturb telephony sessions
  - Large volume transfers too e.g. P2P media downloads
- Wireless environments
  - Interference, the environment & user movement can induce problems
- Other important obstacles
  - Poor quality infrastructures (e.g. in developing countries)
  - The end systems can add considerable delay
  - Human tolerances
  - Poor quality radio chip-sets in PDAs

Protecting VoIP traffic I



Protecting VoIP traffic II

- Problem becomes capacity allocation for a required quality
  - Telephony and ATM research fields have solutions
  - However they have been largely ignored by the IP community
- We investigated an existing proposal and applied it to IP networks
  - Implemented a computationally efficient model
  - We modeled the superposition of independent sources as a Markov Modulated Poisson Process (MMPP)
- Measured the loss probability through a finite buffer (for different loads and buffer sizes)
- Compared the model, simulation *and a* laboratory setup
- Quite unique to use three different evaluations

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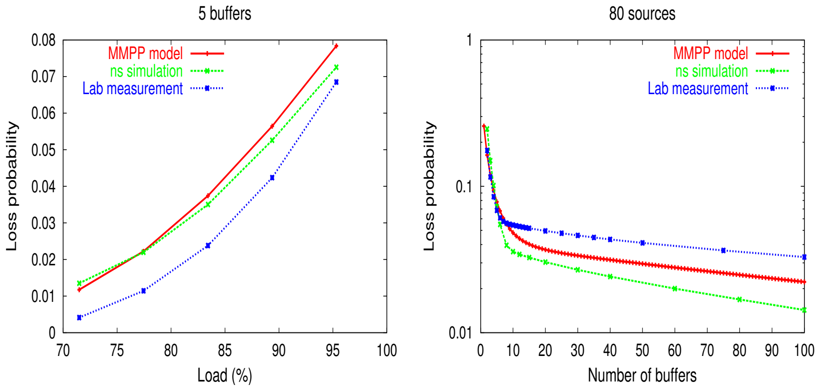
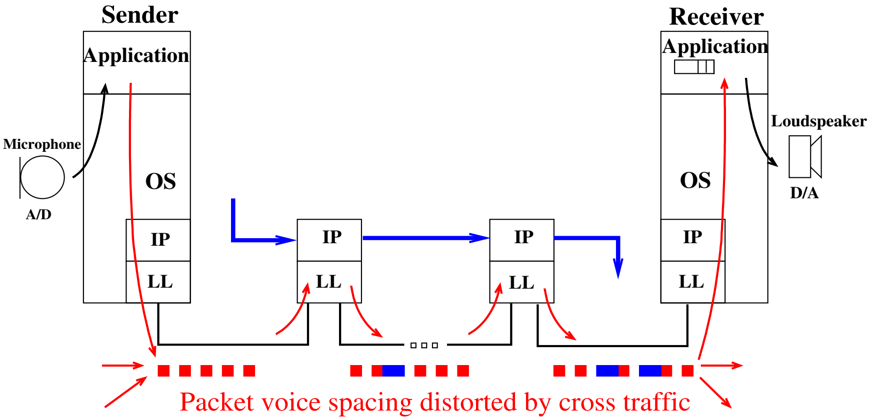
Loss probabilities of different loads & buffer sizes

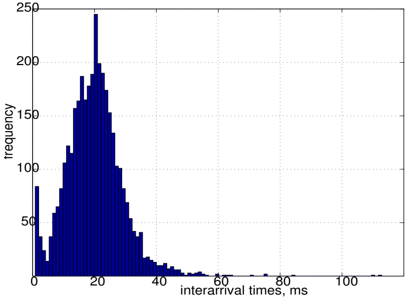


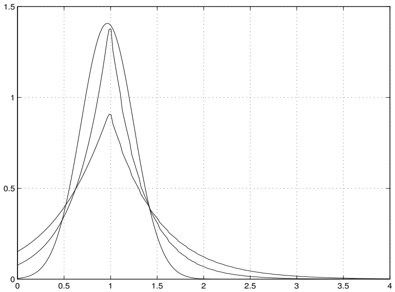
$$Load\ (\rho) = \frac{No.\ \ sources \quad \times \quad Sourceon_{prob} \quad \times \quad Rate_{peak}}{Link\ \ capacity}$$

Mixing VoIP and data traffic



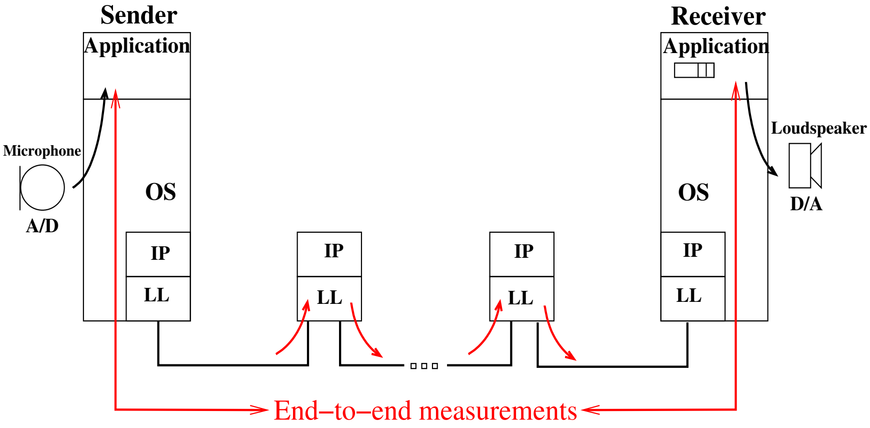
Measured and modeled VoIP interarrival histograms



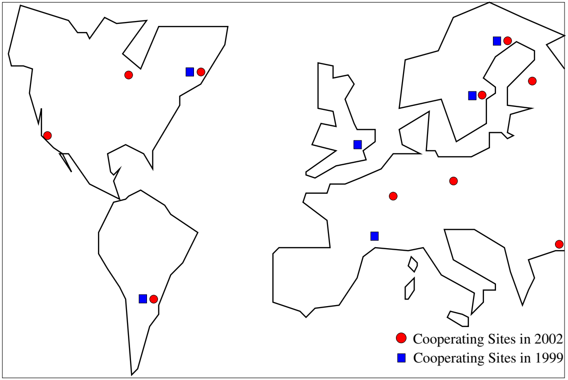


- Left histogram shows the interarrival times of a VoIP session (20ms)
- Right plot shows 3 choices of random network delays (2 Exps & 1 Gaussian)
  - $x = 0$ (origin) represents packets back to back
  - $x = 1$ (max) at the 20ms interval, dispersed for $x > 1$

Measuring wide-area VoIP quality



Co-operating test sites used in 1999 and 2002



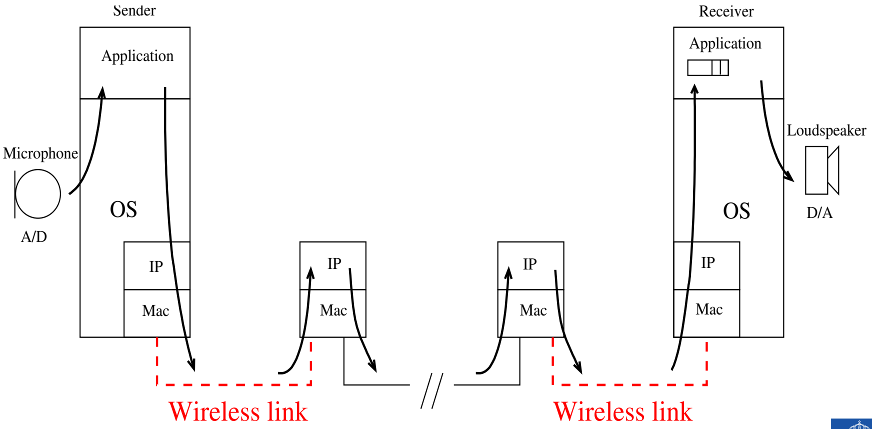
An active measurement method was used, a 70 second pre-recorded conversation between the marked sites was sent once per hour

Measurement goals and brief results

We conducted comprehensive loss, delay and jitter measurements in 2002:

- We found some trends by comparing the results from 1999
  - Improvements from 1999's measurements:
    - full-mesh topology, totally automatic invocation & more hosts
  - Also investigated some new items:
    - Asymmetry, time effects, packet sizes & silence suppression
- Quality has slightly improved over the past three years
- Most calls $<$ 2% loss & 10 ms jitter (delay is location dependent)
- Infra-structure not distance (or the number of hops) is important
- VoIP is still not usable on a global scale, 2 sites showed poor quality
- Collected 22,500 calls, data downloaded 100's of times & used in at least 6 publications

Moving onto wireless access...



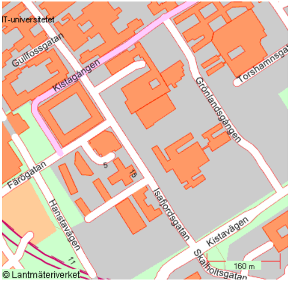
Wireless Measurements

Now we look at IEEE 802.11b access, and again use active measurements of the channel. In particular we look at the MAC layer behavior, the environment and the role of competing traffic.

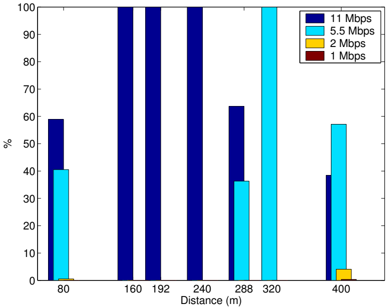
- Pure distance effects using line-of-sight between a single sender and receiver (outside)
- Distance effects with line-of-sight (office)
- Distance effects without line-of-sight (office)
- Competing traffic effect in ad-hoc mode (room)
- Competing traffic effect in infra-structure mode (room)
- IEEE specific bitrate selection, RTS/CTS (office)

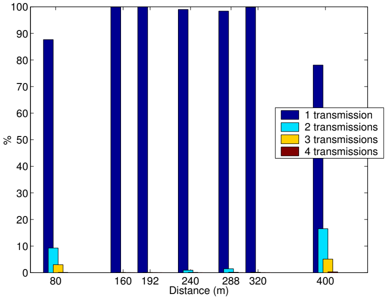
Line-of-sight setup

We took eight measurements at distances from 80 to 400 meters
We recorded the loss, delay, jitter, bitrate & no. of MAC transmissions The sender was stationary and the receiver moved



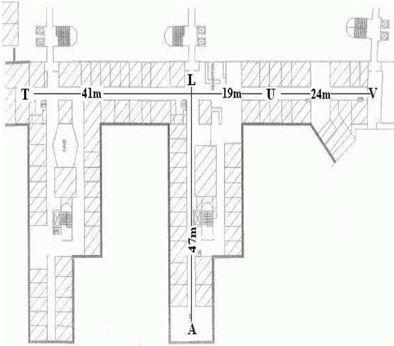
Line-of-sight results





The left histogram shows the rates at which each frame was sent, the right one is the number of retransmissions at each distance. VoIP quality in this scenario is generally good.

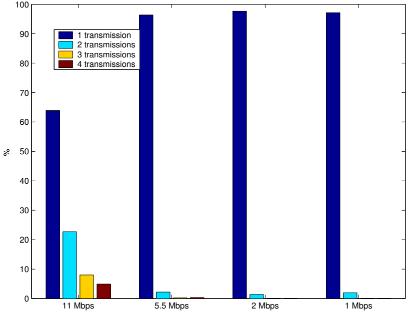
Line-of-sight in the office I



Locations	fraction of losses (%)	round-trip time (ms)	jitter (ms)
$A->L$ (47m)	[0.0, 0.0, 0.0]	[1.9, 2.0, 2.2]	[0.1, 0.1, 0.2]
$T->U$ (60m)	[0.0, 0.2, 0.9]	[1.9, 2.0, 2.9]	[0.2, 0.4, 0.9]
$T->V$ (84m)	[0.0, 0.4, 1.4]	[1.8, 2.2, 3.5]	[0.1, 0.2, 0.5]

Non line-of-sight office results

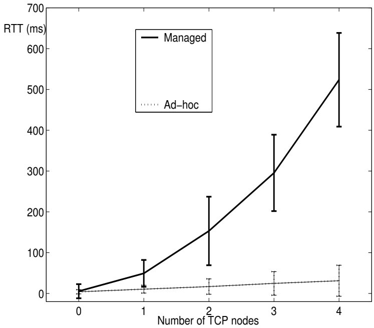
In this setup, the receiver is around a corner. This time we fixed the maximum bitrate and observed the rates selected.

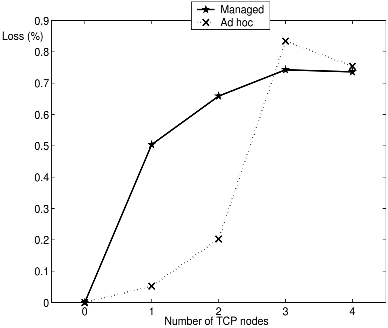


5.5 Mbits in this case would be a reasonable rate to send VoIP stream.

Voice quality and competing TCP traffic

If we now look at competing TCP traffic and its effect on ad-hoc and managed IEEE 802.11 modes



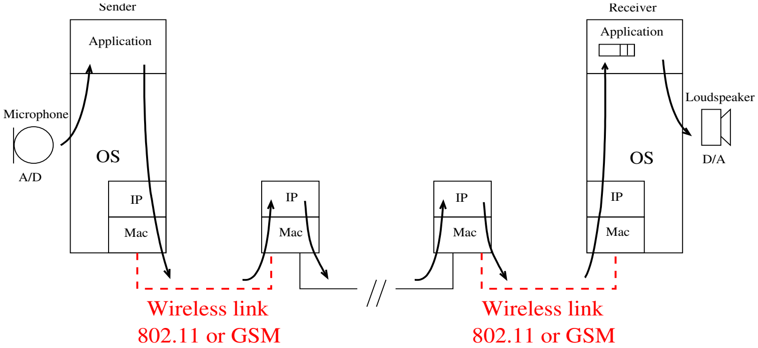


Left plot shows the delay, & the right plot loss.

Conclusions for real-time voice and 802.11b access

- Generally good quality, 802.11b performs better than we expected (or reported)
- However even line-of-sight vulnerable to interference, a trend that will increase with more base stations being deployed
- Loss is generally caused by intervening obstacles
- Delay and jitter generally arise from competing traffic
- Access point can add high delays (both scheduling & queuing)
- We combined information measured from both the MAC and application layers (so called "cross-layered" approach)
- Problems: asymmetric antennas, physical obstacles

802.11-based voice with alternative access

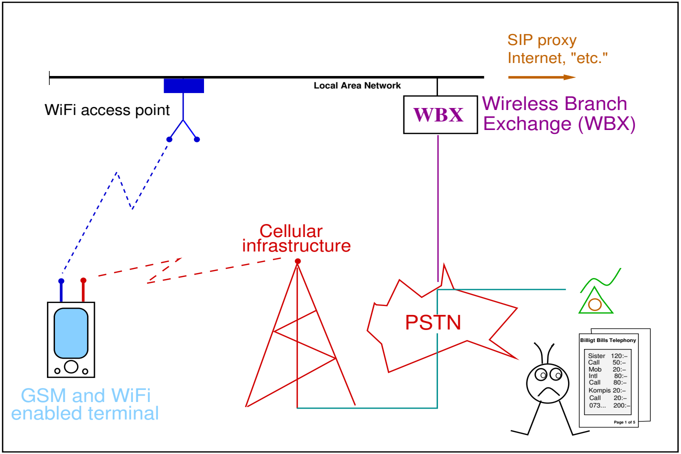


- Use WiFi network for voice whilst in "good" conditions and switch to the cellular network when the quality is not
- Switch with minimum disruption to the user

Seamless voice roaming

- The idea is to roam (seamlessly) between different network types
- For example between WiFi and GSM
- A problem is the 5 second setup needed to initiate a ring signal in the GSM network
- One needs to initiate a handover to the GSM network before poor quality is experienced in the WiFi network
- But not to switch for short-term audible but tolerable sound "glitches", GSM users are accustomed to this
- When leaving/entering WiFi networks a handover is nearly always needed
- But users don't want calls to move backwards and forwards between the network types unnecessarily

Handover infra-structure



Infra-structure by Optimobile AB, we added a prediction module

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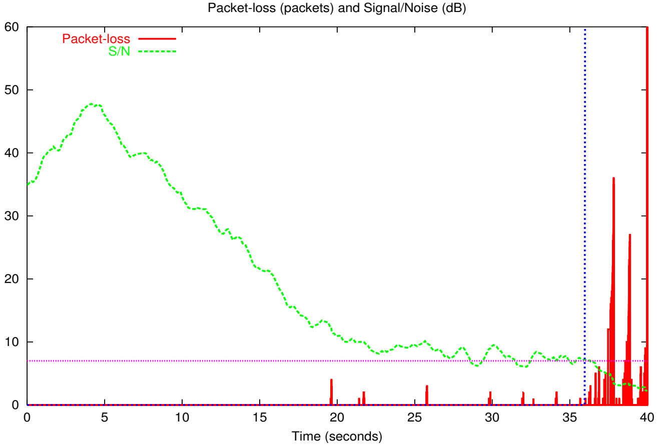
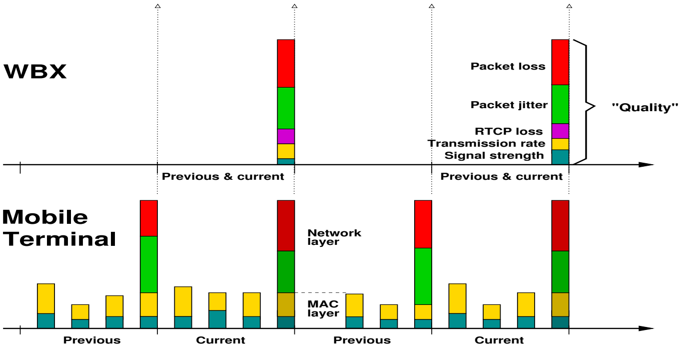
Loss and signal to noise ratio



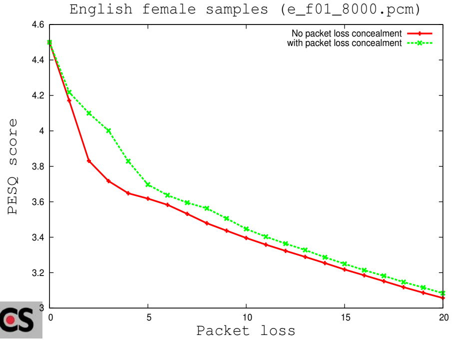
Combining quality and link-layer parameters



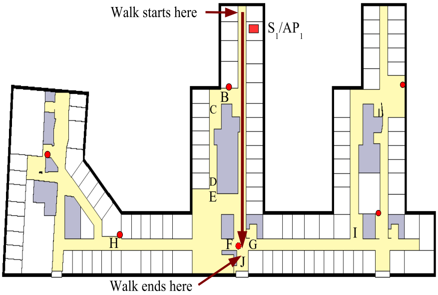
Component	interval 1	interval 2	both intervals
Loss ($> 4\%$)	-1	-1	-1
Jitter ($> 85ms$)	-1	-1	-1
SNR ($> -70db$)	-	+1	-
SNR ($< -90db$) & rate ($< 2Mbits$)	-	-1	-
Rate ($> 3Mbits$)	-	+2	-
2 lost messages	-	-	-1
Total	$\sum X$	$\sum Y$	≥ 0 do handover

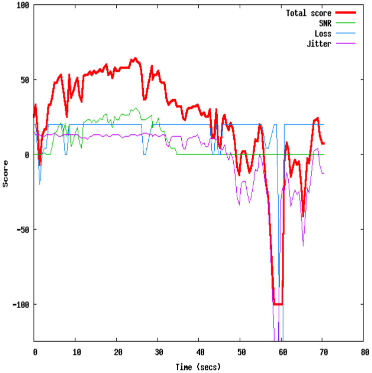
Small diversion - human tolerance to loss

- How much loss do we know is acceptable?
- Well, the ITU have developed a psychoacoustic model for human speech quality evaluation, called PESQ
- No need for human listeners, the model estimates the quality close to a humans rating 4.5 (excellent) to 1 (very poor)
- 7 consecutive losses are approximately a lowering of one rating



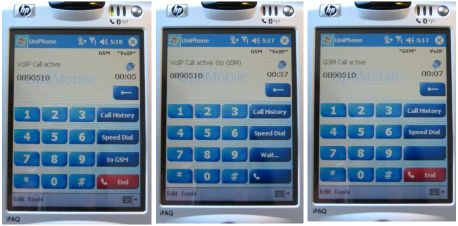
Single test case



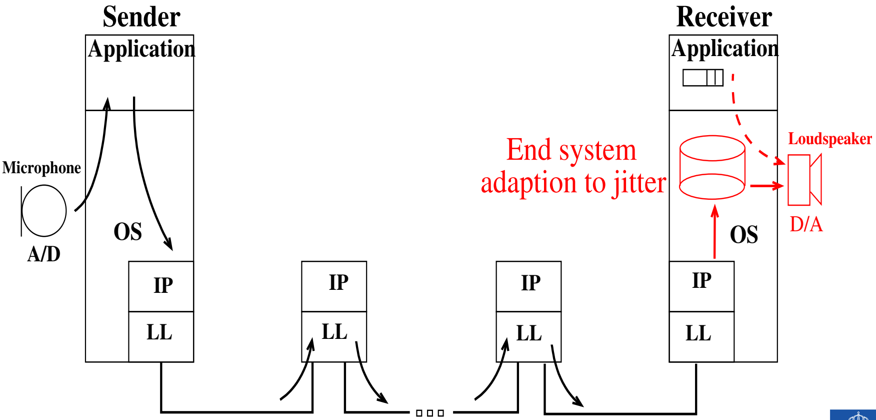


Working demonstration

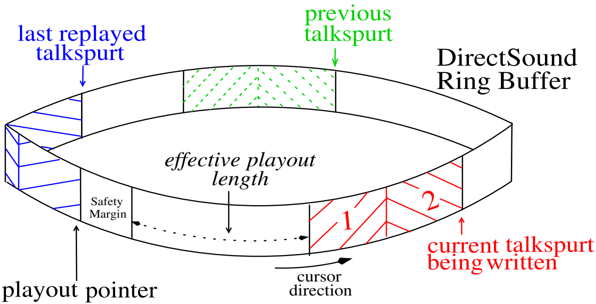
- Use Optimobile's system to implement our quality module
- We have tested in a single cell with one access point (AP)
- They have conducted tests in Kista Gallerian
- Implemented the module in two kinds of PDA's (HP/Qtek)
- Possible to "choose" different levels of handover actions
  - More conservative - handover earlier, less risk of disconnecting
  - More aggressive - handover later, hence saving money



End-system adaption to network jitter



Directsound playout support



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

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 rather than a speech sample for easier triggering and delay calculations
- Point here is to show the end-systems should not be neglected, and one can change them relatively easily (unlike the network!)

Wireless VoIP challenges

- WiFi voice is not yet a true competitor to the cellular network
- Lack of true handsets (currently only PDAs), also people *like* their phones as fashion accessories, cameras etc.
- Quality issues related too:
  - PDAs are not voice optimized
  - WiFi primarily a data communication technology
  - Quality related to the environment, walls, windows etc.
  - Use of unlicensed spectrum (2.4 Ghz)
  - Other interfering devices: access points, users on the same channel, Bluetooth devices, microwaves ovens and so on
  - User mobility is unpredictable

Voice quality conclusions

- We have looked at the complete system in separate studies
- We do not assume any network QoS support, codec type etc.
- Simply ask "How is the system performing today?" we answer through measurements
- Good quality is possible on todays Internet (as you know)
- Operators still want to isolate VoIP traffic or give it higher priority, especially in upcoming multi-service networks
- We found well-provisioned links are almost as satisfactory
- Certain infrastructures need upgrading for reasonable VoIP
- Wireless should work but there limitations
- Handovers are an intermediate/cost saving solution
- The end-systems should not be ignored

A pot-pourri of possible collaboration items I

- Measurements (New COST action)
  - WG1 Monitoring and data collection technologies
  - WG2 Measurements and Traffic Analysis
  - WG3 Applications and practical exploitation
- Handovers
  - Better score determination
  - Roaming between WLANs or *into* a WLAN from cellular
  - CRAFT proposal on handovers in progress (JOSHWA)
- Ambient
  - I was in task 3 (Composition) in phase 1
  - Current INESC work?
- Others:
  - 7th framework programme ?

The end ?