Assessment of VoIP Quality over Internet Backbones

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Outline

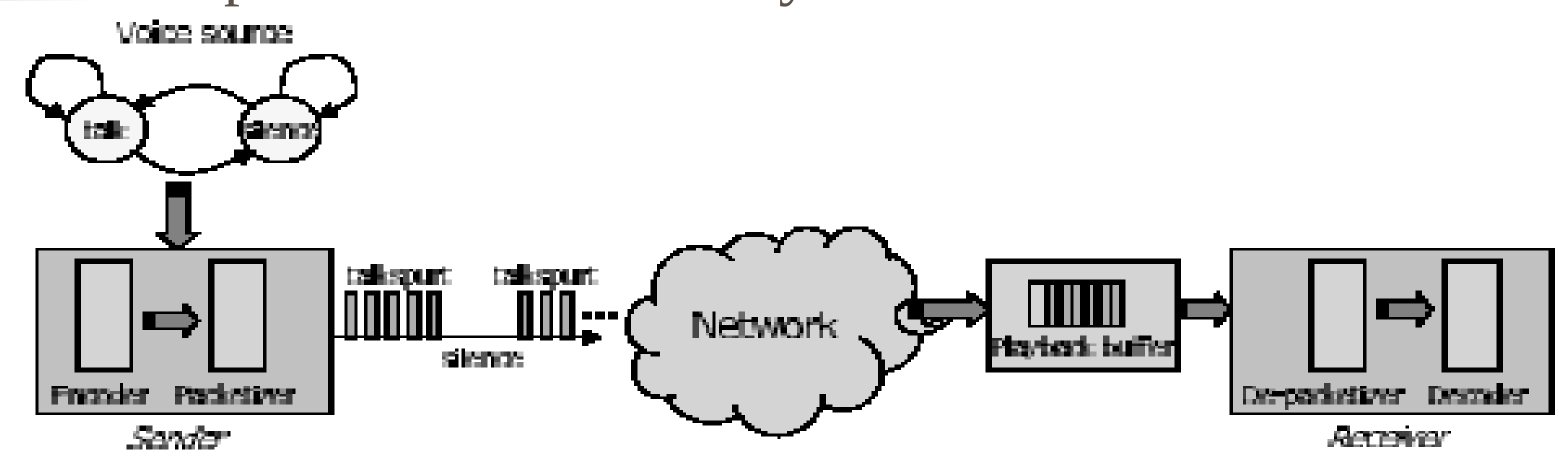
- Introduction
- VoIP System
- VoIP Quality Assessment Methodology
- Internet Measurements
- Numerical Results
- Conclusion

Introduction

- VoIP: via managed IP network or Internet?
- Objective: to assess to what extent today's Internet stands up to the toll-quality expectation
- Characteristics of the approach
  - Use delay and loss measurements collected by sending probes
  - Use appropriate voice quality measures that take into account various transmission impairments
  - Take into account the effect of different components of VoIP system, with emphasis on playback buffer

VoIP System

- Components of the VoIP system



Encoding & packetization delay

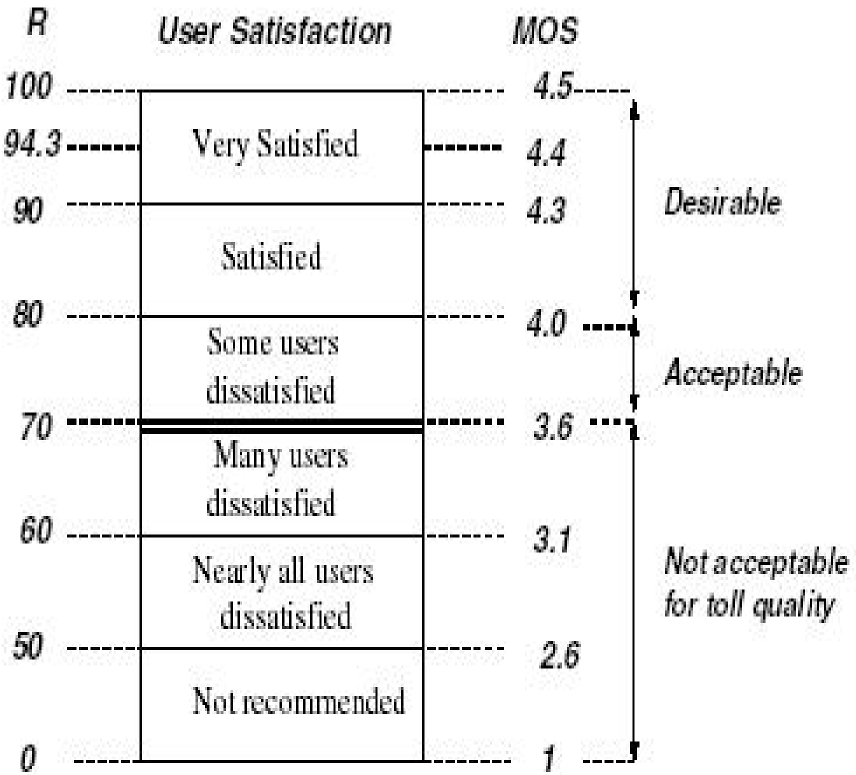
Propagation, transmission & queueing delay

Buffering & decoding delay

- Encoder: G.711, G.723.1, G.726, G.729
  - Voice Activity Detection (VAD)
- Packetizer: encapsulate speech samples into packet of RTP/UDP/IP
- Playback buffer: two operation modes
  - Fixed scheme
  - Adaptive scheme
    $p = d_{av} + 4v$
- Depacketizer
- Decoder
  - Packet Loss Concealment (PLC)

Assessment Methodology

- Choose voice quality measures
  - Network performance
  - Mean Opinion Score (MOS)
  - E-model (R-value)
- Combine recent studies to rate an entire voice call, consisting of multiple short speech segments

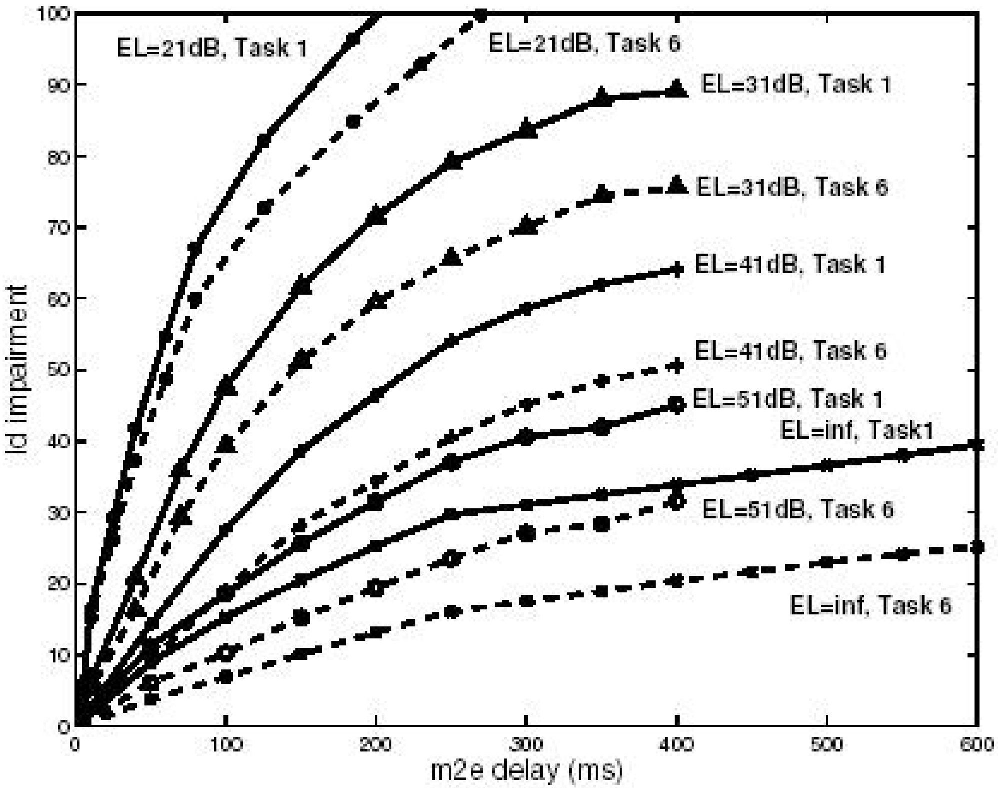


E-model

- A computational model, ITU-T G.107/G.108
- Represented by psycho-acoustic scale
  $R = (R_o - I_s) - I_d - I_e + A$
  - R_o: effect of noise; I_s: loudness and quantization; intrinsic to voice signal itself and not depend on the transmission over the network
  - I_d : capture the effect of delay, quality degradation due to one-way or "mouth-to-ear" (m2e) delay
  - I_e : capture the effect of signal distortion, called the "Special Equipment Impairment factor"
  - A : advantage factor

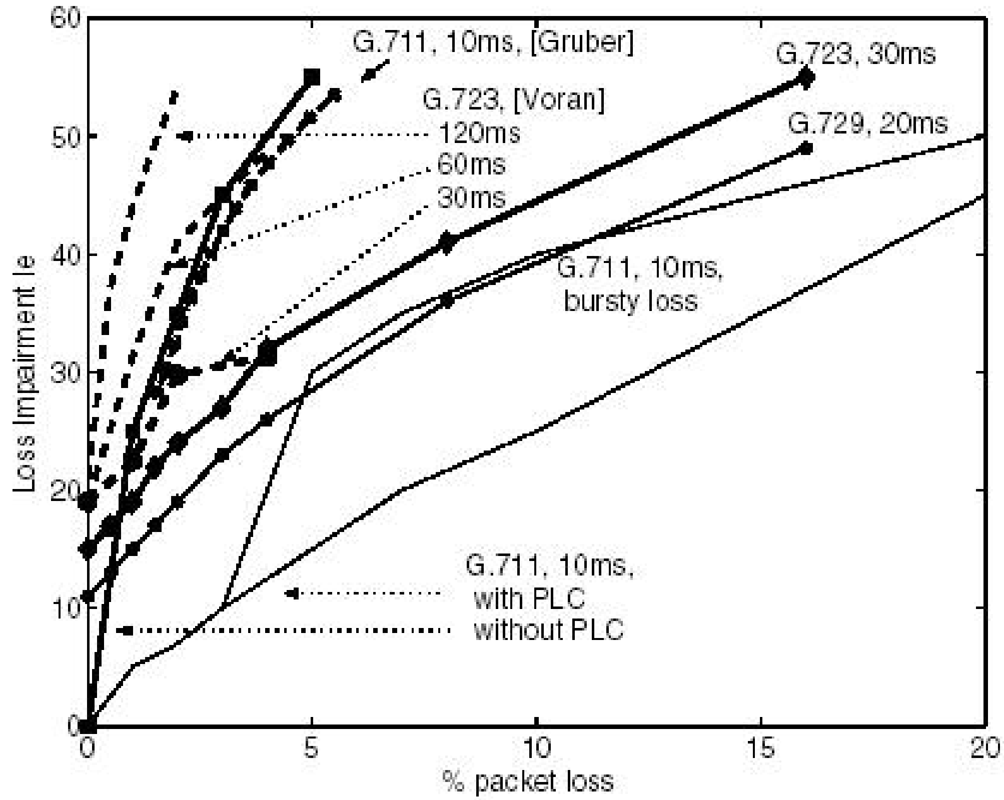
Delay impairment I_d

- $I_d = I_{dte}(m2e, EL_2) + I_{dle}(m2e, EL_1) + I_{dd}(m2e)$
  - I_{dte}, I_{dle} : impairments due to talker and listener echo
  - EL_1, EL_2 : echo losses at the point of reflection
  - Tasks (not captured in I_d): types of conversation, "Task 1" (most stringent) "Task 6" (most relaxed)



Loss impairment I_e

- I_d captures the distortion of the original signal due to low-rate codec and packet loss
- Impairment increase
  - 4 unit in R scale per 1% packet loss for codec with PLC
  - 25 units per 1% packet loss for code without PLC

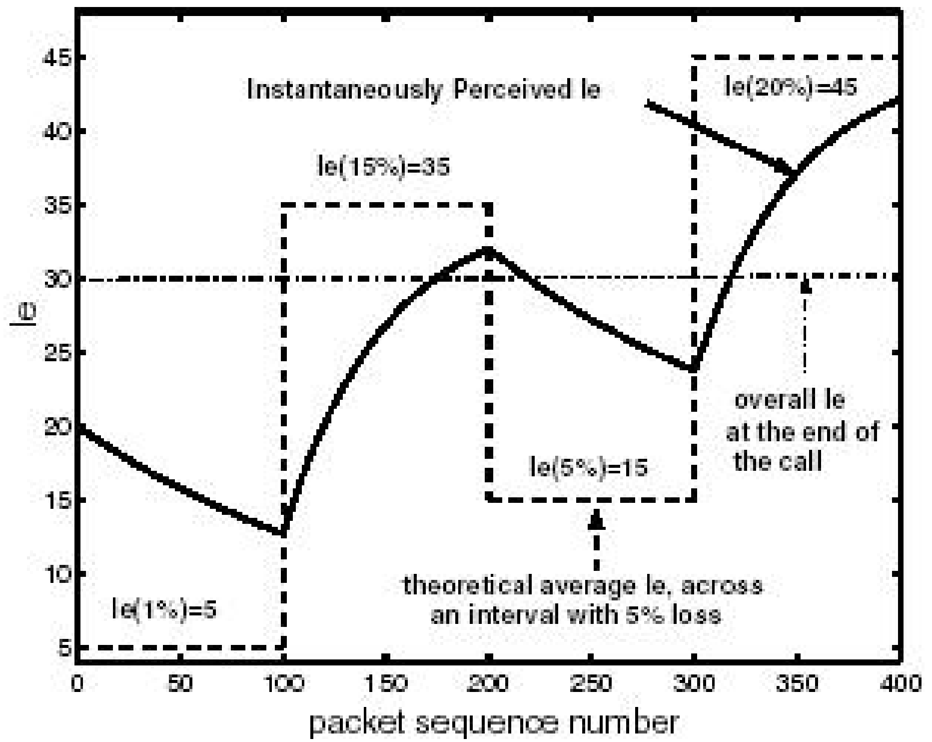


VoIP call quality

- Calculating the average loss rate and the average delay only give a rough estimate
- Approach
  - Divide the call duration into fixed time interval and assess the quality of each interval independently
  - Independent MOS(t) rating of each short interval [9]
  - Take into account the “recency effect”[10]
  - Use variable length intervals to handle the burstiness of packet loss, defined bursts and gaps [4] [7]
  - Assessment of an entire call use both final rating [4] and the worst quality experienced during a call

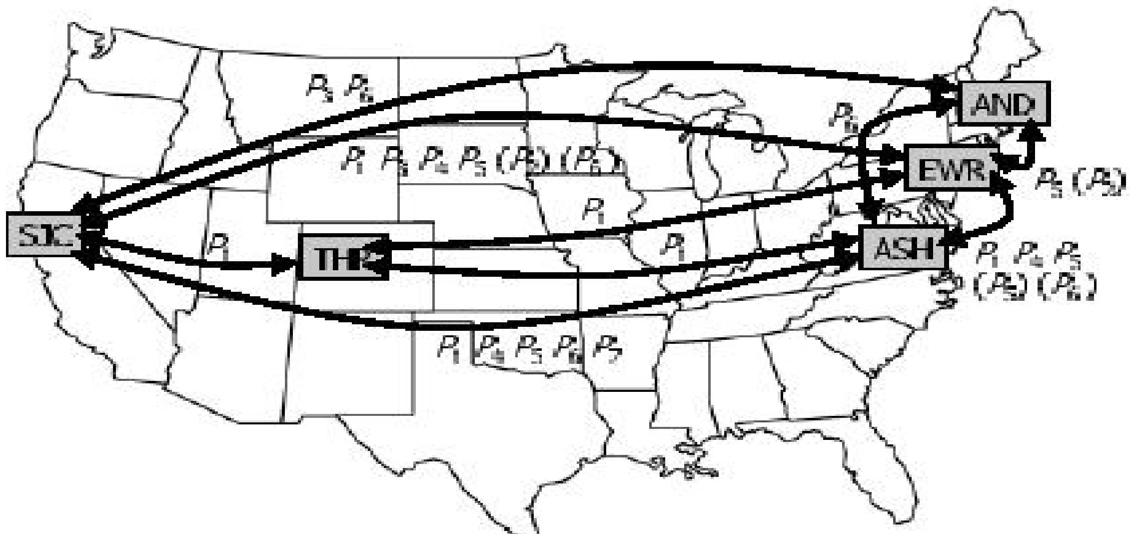
Instantaneously perceived I_e

- I_e*(loss)* for a gap or burst, exponential curve with
  - T_{bad} = 5 *sec* for the high loss periods
  - T_{good} = 15 *sec* for the low loss periods



Measurements Experiments

- Delay and loss measurements by probes
  - Probes were sent by and collected at measurement facilities in 5 major US cities, 43 paths by 7 providers



  - Probes were 50 bytes each and sent every 10ms from 2001/6/27 19:22~2001/6/29 00:50

Trace Description (1/2)

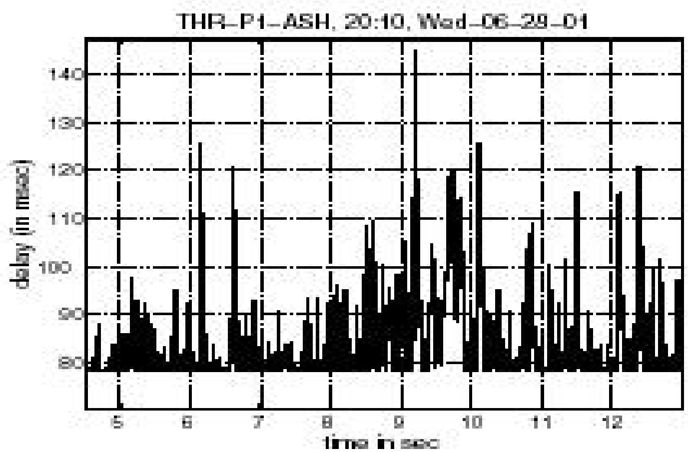
- 43 paths are classified into 5 types

Type	Num	Example path				Delay (in msec)				Loss			
										usual loss events		long outages	
		From	Prov.	To	Dist.	min	50%	98%	99%	avg clip (msec)	#clips per hour	duration (sec)	times per day
A	11	EWR	P_8	ASH	short	3.4	3.6	3.7	3.72	20	1-5	5-12	1-2
B	2	ASH	P_4	EWR	short	6.8	7.2	120	200	20	2-3	12-25	1
C	16	SJC	P_5	EWR	long	32.7	32.8	33.5	34.5	0	0	2	1
D	4	SJC	P_4	ASH	long	45.1	45.4	170	225	10	1-2	15-25	2-3
E	10	THR	P_1	ASH	long	77.8	78.2	100	210	10	2-20	1	1

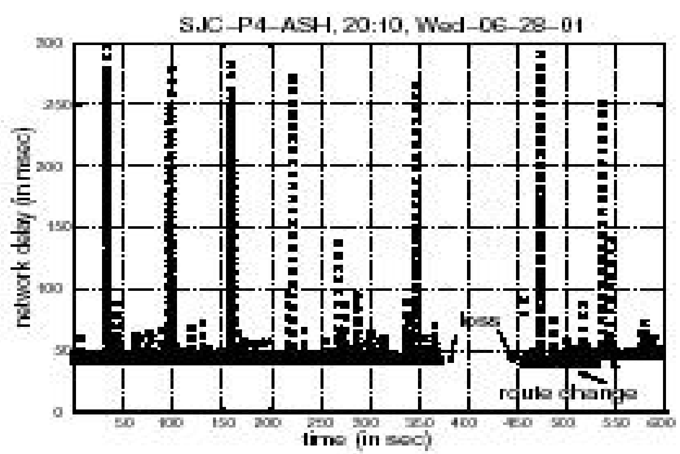
- Observation of loss events
  - Only 3 out of 43 paths had consistently no loss
  - 6 out of 7 providers experienced outage periods 10-220 sec for 1-2 times per day
  - 0.5-2sec loss durations were correlated with delay spikes
  - The number of out-of-order packets was negligible

Trace Description (2/2)

- Observation of delay
  - Paths of type A and C have practically no queueing
  - Paths of type B and D have in general low queueing, except for clustered delay spikes
  - Paths of type E: queueing component is high and delay varies



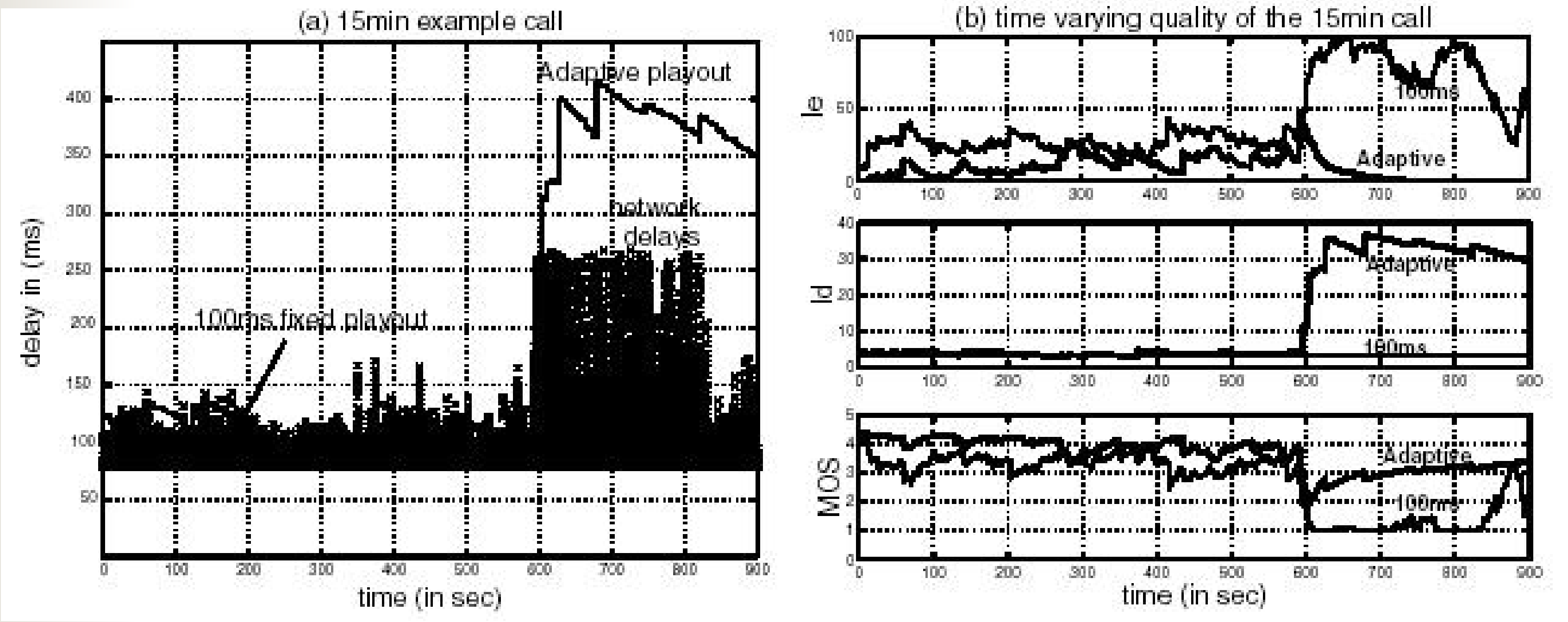
(a) Example trace, type E



(b) Example trace, type D

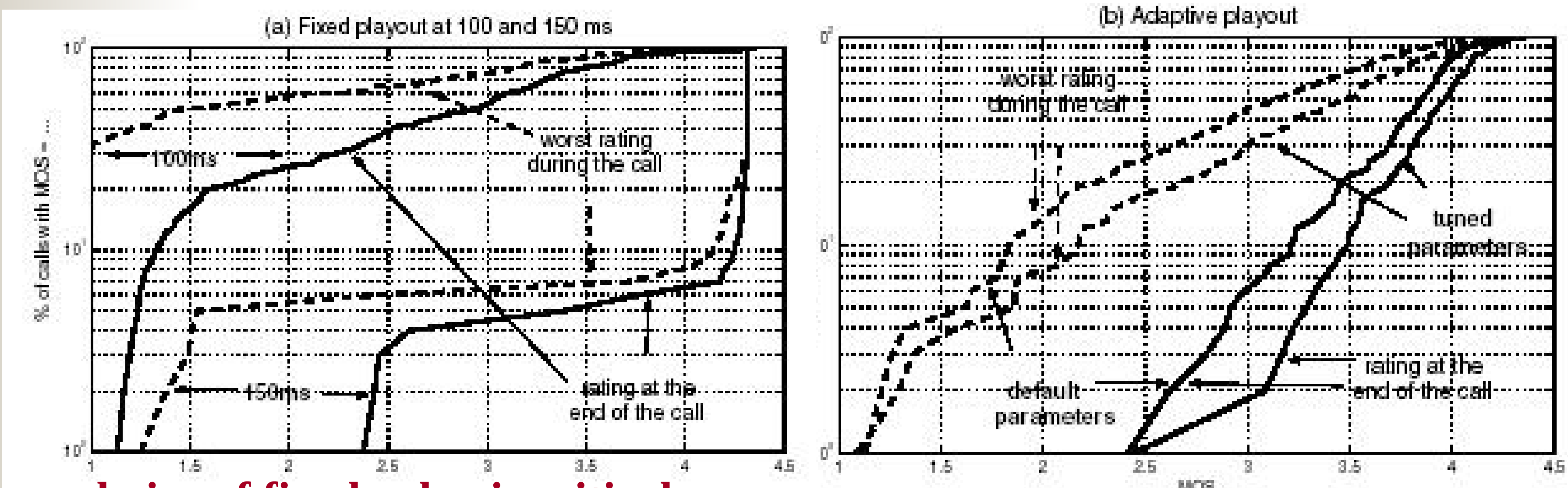
Numerical Results (1/3)

- An example call trace of type E
  - Large delay variation and a period of sustained loss
  - There exists a tradeoff between loss and delay



Numerical Results (2/3)

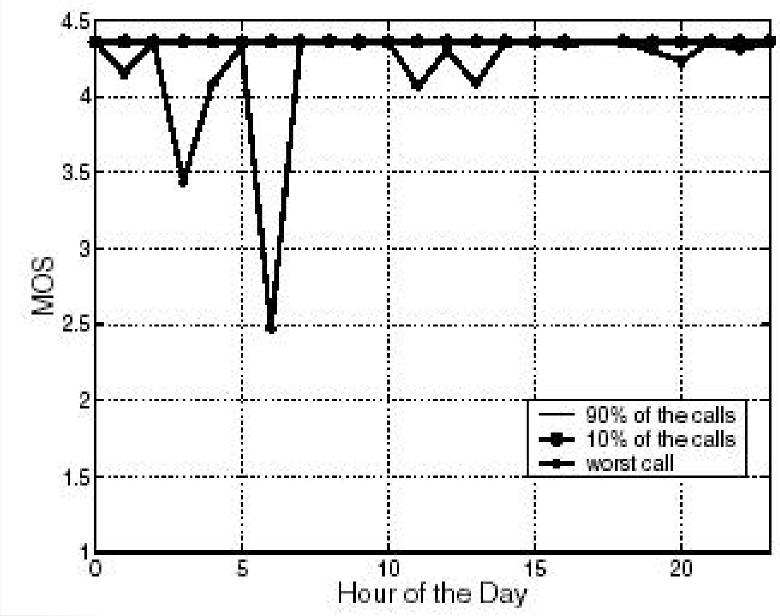
- Many calls uniformly spread over an hour
  - 150 short (3.5 min mean): business calls
  - 50 long (10 min mean): residential calls

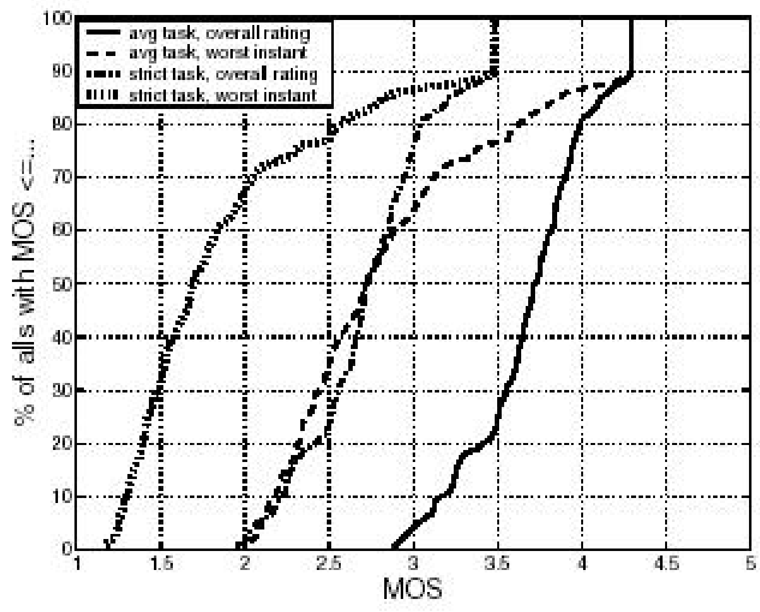


choice of fixed value is critical:
150 ms is acceptable(6% below 3.6)

Numerical Results (3/3)

- All other paths
  - Both short (type A) and long (type C) distance achieve an excellent MOS at all times except for rare cases when network drop occurs.
  - Path of type B and D exhibit periodically clusters of high spikes: packets are dropped at the playback buffer, whether fixed or adaptive





Discussion

- On the performance of the backbone networks
  - Highly loaded paths (type E) as well as some over-provisioned paths exhibiting frequent delay spikes (type B and D) have poor VoIP performance
- On the playout buffer
  - An appropriate choice of the fixed playout buffer is the one that leads to maximum MOS
  - The need of adaptive playout comes when
    - The delay is high and no margin for overestimating
    - Delay is unknown and receiver doesn’t know how to select an appropriate fixed value
  - The adaptive playback was useful on loaded network (type E) that exhibited high and slowly varying delays but fails on others.

Conclusion

- This paper assess the ability of Internet backbones to support VoIP considering realistic configuration.
- In general, backbone networks are over-provisioned, but poor VoIP performance on a large number of ISP backbone networks is observed.
- Action for improving today’s VoIP performance to reach toll-quality standards can be taken
  - Inside the network: marking the voice traffic to give it preferential treatment
  - At the receiver, playout buffer scheme should be carefully chosen to match the delay pattern