Adaptive Joint Playout Buffer & FEC adjustment for Internet Telephony

&

Path Diversity with Forward Error Correction (PDF) System for packet switched networks

reviewed by

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Adaptive Joint Playout Buffer & FEC adjustment for Internet Telephony

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Objective

- Account for the impact of end to end delay on the perceived audio quality
- Discuss how to optimize the measurement of quality.

Introduction

Transport of real – time, interactive audio over IP networks often suffer from packet loss and delay variation that calls for correction.(Threshold effect of end to end delay around 150 ms)

- **FEC-** reduces the impact of packet losses but increases the end to end delay.
  To repair a packet loss, the receiver waits for redundant packets.
- **Adaptive Playout Buffer**– compensate for jitter at the receiver at the expense of end to end delay.

Current Approach: To optimize Quality

Adaptive rate/error control:	Play best received copy of a given packet.
	Side Effect: Buffer Overflow and End to end delay
Adaptive delay aware error control:	Play best received copy of a given packet + playout delay = delay (FEC absent) + delay (FEC used)
	If sender transmits additional redundant FEC then the destination waits for it.
Adaptive joint playout buffer + FEC:	Play first copy received correctly at designation.
	(Rather effective for non-delay aware FEC scheme but not clear about delay aware)

Background Materials

A. **Error Recovery**
   - Retransmission (ARQ)
   - **FEC** (Media Independent,Media Specific)
   - Error concealment(Compliment to FEC)

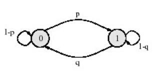


Fig. 1. *The Gilbert model*

B. **Loss Processor of Audio Packets**
   FEC is specially good when the consecutive loss is small.
   Gilbert Markov chain models the 2 statements.

C. **Playout Adjustment Algorithm**
   Jitter: In order to compensate for jitter, playback buffers is used.
   It's a tradeoff delay for loss.

Background Materials

D. Rate Control: Congestion control
   Audio application share resources with each other & with current TCP based applications. An approach that works with TCP friendly congestion control is used.
   equation-based mechanism
   It relies on the RTP and it's control part RTCP. The source performs equation-based congestion control based on feedback information contained in RTCP reports and adjust its sending rate by changing the packet size but maintaining the time interval packet.

E. An Audio Quality Measure: the E-model
   E-model predicts the subjective quality…

$$R= R_o - I_s – I_d – I_e + A$$

R_o= group of noise effect

I_s = Impairment due to quantization

I_d = mouth to ear delay

I_e = signal distortion (low bit rate codec & packet loss)

$I_{d=} I_{dte}(d, TEL) + I_{dle}(d, LEL)+ I_{dd}(d)$

Assuming echo loss is equal to both ends

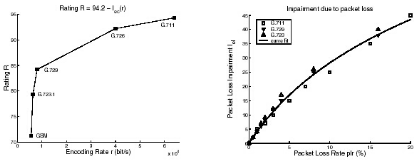
If $E_L = \infty$ (perfect echo control)

If $E_L = 51$(a simple yet efficient echo control)

Choice of Utility Functions which account for delay.

E-model – utility function → Rating = Encoding rate + packet loss rate; Different modes of interactivity

- **Influence of Distortion**
  Effect of packet loss increases the measured distortion



(a) Rating as a function of the encoding rate (for delay=0 and no loss)

(b) Impairment due to loss as a function of the packet loss rate.

Fig. 2. *Influence of distortion on quality.*

Choice of Utility Functions which account for delay

B. Influence of End to End Delay

Utility function responding to different mode interactively requires different switching speed and sensibilities to delay

Normal end to end delay = 150ms

For Telephone user with 300ms delay is like half duplex connection.

So utility function with a steep decrease around 150ms was proposed

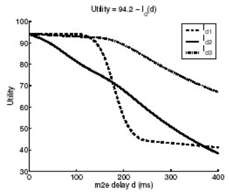


Fig. 3. *Utility as a function of the mouth-to-ear delay for different interactivity levels for $r = 64Kbits/s$ and no loss.*

C. Our Utility Functions

d – one way mouth to ear delay end to end delay

r - codec bit (reconstructed rate at the destination)

plr - the residual packet loss rate (after FEC is used)

d1 –strong interactivity

d2 - annoyed by delayed due to echo but with clear threshold effect

d3 – attaches a small importance to delay

Adaptive Joint Playout Delay & FEC Adjustment

Scenario: to maximize perceived utility

- **N1 : Partial method** - Adjusts the playout buffer at the receiver, the FEC scheme at the source is aware of playout delay computed at destination & adjust redundancy.

  If the source uses: **Signal Processing FEC or Media independent FEC** (RS FEC) a play first strategy is used.

- **N2 : Complete method** - Uses playout delay & the redundancy at the source. Signal processing FEC is used, both first play & play best strategy are considered.

Five Combination of FEC and playout method and decoding strategy

- N1, SP FEC, Play First
- N2, SP FEC, Play First
- N2, SP FEC, Play best
- N1, RS FEC
- N2, RS FEC

Adaptive Joint Playout Delay & FEC Adjustment

Voice packets are transmitted over an unreliable network

Control Scheme

- A **packet loss process** modeling Gilbert process
- A **stationary delay process**: Network delays of voice packets are identically distributed.
- **Independence loss delay**: Packet losses & network delays are mutually independent

To maximize the quality of voice call.

Given

R_{max} - rate available for audio flow due to TCP friendly rate control scheme & is updated upon reception of an RTCP receiver report.

D - Playout delay for each talkspurt

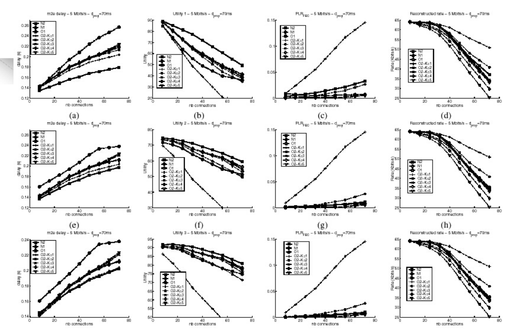
K_{max} - max copies of each voice packet sent.

Find- The optimal number of copies to send & optimal encoding rate for each copy + playout delay.

Simulation Results

Scenario Considered

- N audio sources share a bottleneck link win 3n sack TCP
- ON/OFF source(CBR 500kbits/s when ON) with ON & OFF periods exponentially distribute.
- Average ON/OFF times of 3seconds
- Packet loss rate is varied artificially by changing the number of TCP & audio connections sharing the link.



(a) (b) (c) (d)

(e) (f) (g) (h)

Simulation Results

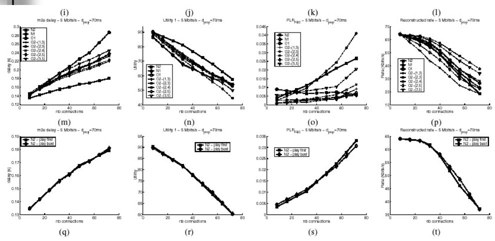


Fig. 4. *Performance of the different methods N1, N2, O1 and O2 (with different FEC parameter settings). Mouth-to-ear delay, utility measured at the destination, residual packet loss rate and reconstructed rate of audio for SP FEC with utlity f_1 from (a) to (d), f_2 from (e) to (h), f_3 from (i) to (l) and for Reed-Solomon with utility f_2 from (m) to (p). Comparison of play first and play best strategies from (q) to (t).*

Simulation Results

Fig 5a using utility **f2**

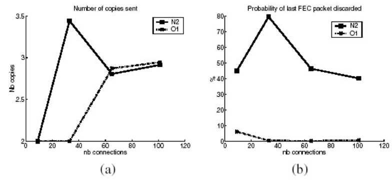


Fig. 5. *The optimal solution (N2) does not always wait for all FEC packets to arrive.*

Advantages / Disadvantages

- The method performs better than others especially where end to end delay is of importance.
- It is worth using joint playout adjustment algorithm when a delay FEC is used.
- With joint adaptive, it is best to use playfirst (simple) as the playfirst and playbest work similarly.
- Improvement brought by delay aware FEC cannot be obtained if the delay aware FEC control is just a piggybacked on the existing playout control method.

Evaluation of Paper

- Although, delay is still present, in cases where delay matters, there is a real benefit in using a joint method.
- Need for a more sophisticated method to reduce delay.

Path Diversity with Forward Error Correction (PDF) System for packet switched networks.

Thin Nguyen & Arideh Zakhor

reviewed by

Olufunke Olaleye

Objective

- Reduce packet loss and end to end delay on delay sensitive application over the internet by using PDF
- Disjoint path from a sender to a receiver using a collection of relay nodes.

Introduction

Delay sensitive applications are challenged by:

* *High bit rates* * *Delay* * *Loss sensitivity*

<u>Proposed solution</u><u>: Single Path Scheme</u>

- Source-coding: Layered and error resilient video codec. It adapts it's bit/sec to the available bandwidth.
- Channel Coding: FEC - reduce delay but at the expense of bandwidth.
- Lossy environment: Retransmission - introduces delay of RTT between sender and receiver thereby exceeding the allowable delay 150ms for interactive applications
- Protocol: Equation based rate control - to compete fairly with other TCP traffic for bandwidth stabilizing the throughput and reducing jitter.

Introduction (contd)

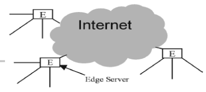


Fig. 1. Edge server architecture.

<u>Proposed solution</u><u>: Single Path Scheme</u>

Content Delivery Network: Edge architecture- load balancing, lower latency and higher throughput.
This is achieved by moving content to the edge of the network to reduce RTT and avoid congestion.

<u>Proposed solution: Multiple Path Scheme</u>

Sending packets simultaneously over multiple path to overcome the unpredictability and congestion of the internet

Related Work

- Wired network:
  - Path diversity for security
  - load balancing
- Simultaneous Downloading- from multiple mirror sites to achieve path diversity.TCP connections to different sites for non- delay sensitive data.
- Overlay network: sender $\xrightarrow{\text{Packet}}$ relay node $\xrightarrow{\text{Packet}}$ receiver

  using a redundant path, apart from the physical default path.

  e.g RON probes for best path, then send all packet through that link.

Motivation

Model the network using

- 2 state continuous time Markov chain that models a busty loss environment.

  Chain changes from good to bad

Packet Loss { small - Good

{ large - Bad (i.e aggregate traffic exceed link capacity)

Example: Rather than sending packets through a default path only, divide them into 2 paths → default / → disjoint redundant path

If congestion occurs in one path, then #successive lost packets will reduced. Thus, transforming the busy loss into a uniform loss, increasing the efficiency of FEC technique.

Motivation

Given #independent paths with different loss behavior, source bit rate, total amount of FEC protection. There should be an optimum partition of sending rates for each path to minimize the irrecoverable loss probability

Irrecoverable loss probability - The prob. that FEC cannot recover lost packet in a FEC block.

For a Reed – Solomon code RS(N,K)

K data packet, N - K redundant packet

Irrecoverable loss probability = Prob. **>** N - K is lost

Consider: 2 disjoint path A and B
Packet size = 500 bytes, sending rate = 800 Kbps
Protected by RS(30,23)

	Average good time	bad time
Path A	1 sec	10millisec
Path B	1 sec	10 – 50millisec

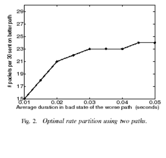


Fig. 2. Optimal rate partition using two paths.

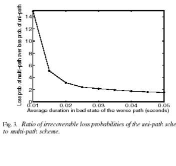


Fig. 3. Ratio of irrecoverable loss probabilities of the uni-path scheme to multi-path scheme.

System Description

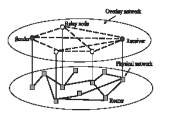


Fig. 4. System architecture; bold dashed and solid lines denote virtual and physical paths; The thin vertical solid lines connecting circles and squares represent the correspondence between virtual nodes and physical routers.

At any instance, a node can act simultaneous as receiver, sender or relay node

Sender Video stream → Physical path → **Receiver**; **Sender** → Virtual path → **Receiver**

By choosing an appropriate node, the packet transverse through the virtual path.

A participating node that is neither a receiver or sender may receive and forward packet on behalf of others.

Setting Up A Communication Channel

Sender $\xrightarrow{\text{trace route}}$ Participating Nodes & Receiver

Sender $\xleftarrow{\text{Link latencies, router's name along default path from sender to receiver \& all path between sender and participating node}}$ Participating Nodes & Receiver

Sender $\xrightarrow{\text{Setup packet to execute trace route from themselves to receiver}}$ Participating Nodes

Sender $\xleftarrow{\text{Path information between themselves and receiver}}$ Participating Nodes

Based on the path information, the redundant path is selected

Sender $\xrightarrow{\text{Setup packet \{ flow ID, IP address, port no of Receiver\}}}$ Selected Relay Node $\xrightarrow{\text{Store the information in a table and forward packet to based on flow ID}}$ Receiver

Redundant Path Selection

•BGP - only exchange and update summary information between ASes.

•OSPF – Good but not scalable , only used within a AS.

•Probing Tool – Good but at the expense of bandwidth expansion.

•Passive Probing – Probing packet are the application packet. Accuracy is dependent on application sending rate.

•RON – execute complex topologies and routing algorithm but have scalability problem. (only limited to 50 nodes)

Trace Route: Compute a set of relay node that result in minimum number of joint links between the default internet path and all the redundant through an exhaustive search

It finds a set of redundant paths that are disjoint as possible from the default path , within this set, it select the one with the minimum latency

But if the dependent path latency exceed the desired delay, it is dropped and step 1 and 2 are repeated

Note After each iteration, the number of shared link for redundant path increase

Simulation Result

- **A: Simulation Topologies**

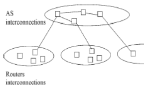


Fig. 5. *Two-level hierarchical topology*

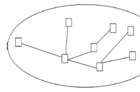


Fig. 6. *Flat topology*

Models	No. Nodes	No. Edges
Flat *Albert-Barabasi*	1500	2967
H-Albert-Barabasi I	1500	2997
H-Albert-Barabasi II	1500	4377

TABLE I

Information for various topologies

To estimate average latency and hop count and the degree of disjointness between redundant path and default path.

- 1. Randomly select a set of participating nodes.
- 2. Randomly select a pair of receiver an sender in the set of participating nodes
- 3. Find the redundant and default path for a given configuration of sender, receiver and participating nodes.

Assumption: The default path is the path with the smallest latency or equivalent shortest path

Repeat step 1 to 3 over 5000 times.

Simulation Result

A: Simulation Topologies (contd)

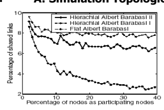


Fig. 7. *Percentage of shared links between the redundant and the default paths.*

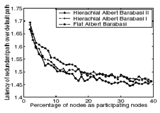


Fig. 8. *Latency of redundant path over the latency of default path.*

Shared link % ↓ as the #participating node ↑

This is because a redundant path is created via participating node.

Latency ↓ as the #participating node ↑

Shared link and latency decreases { rapidly with the participating nodes less than 20%

{ slowly with the participating nodes greater than 20%

This is an indication that increasing the #participating node beyond a certain value will be of no benefit.

Simulation Result

A: Simulation Topologies (contd)

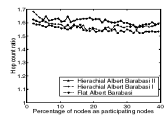


Fig. 9. *Number of hops of the redundant path over the number of hops of the default path.*

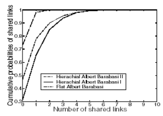


Fig. 10. *Cumulative distribution of shared links for various network topologies.*

The ratio decreases slightly at first & then stay relatively constant

Indicating that reduction in latency is not dependent on fewer link but shorter latency

Redundant part with few shard link can be found with high probability.

Hence, PDF can deployed effectively

Simulation Result

B: NS Simulation

Packet Loss Reduction

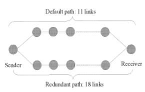


Fig. 11. *Simulation configuration for two disjoint redundant and the default paths.*

To simulate busty packet loss

Peak rate = 1.8Mbs Average idle period = 8 s

Packet size = 500 bytes Burst period = 40 ms

Protected with Reed- Solomon RS(30, 23)

Compare Packet (Scenario)

- 1 Sender streams the video to the receiver at 800Kps on default path
- 2 Sender streams the video to the receiver at 400Kps on default path using **2 paths**
- 3 Same as 2 but **one** shared link between redundant and default path

Simulation Result

B: NS Simulation

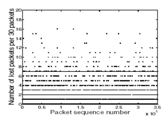


Fig. 12. *Sending packets using traditional default path.*

Point above the horizontal line represent irrecoverable loss event

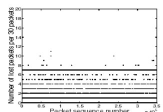


Fig. 13. *Sending packets using both redundant and default paths without shared link between them.*

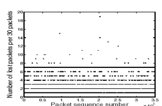


Fig. 14. *Sending packets using both redundant and default paths with one shared link between them.*

As bustiness ↓ recovery probability ↑

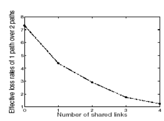


Fig. 15. *The ratio of average loss rates using one path over that of using both redundant and default paths with various number of shared links between them.*

Recovery Probability of FEC ↓ as #shared link ↑

Conclusion

- For various internet-like topology, only 10% of participating node are required for propose path redundant selection to effectively find a redundant path sharing 2 or fewer with the default path.
- This effectively result in a factor of 3 in irrecoverable packet loss.

Advantage/ Disadvantage

- Busty Loss in a sending packet over a single path can be reduced to uniform loss when sent over disjointed multiple paths.

- **Drawback:**
  - The performance of PDF is dependent on the information provided by tracreroute. The information could be incomplete or inaccurate.

  - Some Ases does not do not report accurately or deliberately hide information about networks

Evaluation Of Paper

- The author did not specify which topology is Hierarchical Albert I or II
- The incomplete or inaccuracy of trace route will definitely affect the performance of PDF over the actual internet.
- Interested in the implementation of the PDF system for delay sensitive video application over the actual internet.

Questions

- Explain what media independent and media specific FEC technique.
- What is playout delay and the role of playout adjustment algorithm.
- What are the advantage and disadvantage of media independent technique.
- Explain how the redundant path is chosen in PDF.
- What type of transport protocol was discussed in PDF paper and why is that protocol used.