Energy efficient rate control mechanism for multimedia delivery over wireless Internet



Sudipta Mahapatra, Ph. D Associate Professor Department of E & ECE IIT Kharagpur Sudipta@ece.iitkgp.ernet.in

M. Harikrishna harikrishna.mallisetty@gmail.com

Amareswararao K.

k.amareswararao@gmail.com

INSIDE THIS ISSUE: TECHNOLOGY QUARTERLY

The Economist

NWW. CODIES

EPTLASER 1714 -2340 2008

A very Japanese revolution After Katrina Mussis, 20 Mag 12-20 Tackling Latin American poverty 2045 13 Mg 14 Steve Jobs, resurrection man

How the internet killed the phone business



THE Internet touched our lives in terms of

✓ How we communicate
✓ How we promote our products
✓ How we teach our children
✓ How we invest our time

Growth.....50 million

Radio...... 40 years TV15 years Internet5 years

[Economist, September 2005]



Outline

> Wireless Internet (WI) and Multimedia Delivery

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- Major Challenges
- Need for Loss Differentiation
- RCM-BR Protocol
- Simulation study
- Conclusion



Wireless Internet (WI)

A service granting access to the World Wide Web or Internet via Wireless Networks. It consists of They host information

Fixed internet hosts. Mobile nodes. They host information resources as well as application software for providing network services.

They have wireless access card to access Internet. Ex. Wi-Fi card

Connectors like access points in between.

To provide wireless Access to the devices











MM Networking Applications

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Classes of MM applications:

- 1) Streaming stored audio and video
- 2) Streaming live audio and video
- 3) Real-time interactive audio and video

Fundamental characteristics:

- Typically **delay sensitive**
 - End-to-end delay
 - Jitter Delay variation
- But **loss tolerant**: infrequent losses cause minor glitches
- Normal data can tolerate delay tolerant, but not loss

Why Is Real-Time Transport Hard?

Internet is a best-effort network . . .

Insufficient rate to communicate Congestion Impairs perceptual quality Packet loss Impairs interactivity of services; Delay Telephony: one way delay < 150 ms [ITU-T Rec. G.114] Obstructs continuous media playout

Delay jitter



Challenges in Multimedia Delivery over Wireless Internet

- Different QoS requirements for different types of media.
- High packet loss rate and bit error rate.
- Bandwidth limitation and fluctuation.
- Low performance in traditional transport protocols.
- Heterogeneity among users and networks.
- Limited battery life.

Ref: Qian Zhang, Fan Yang, Wenwu Zhu,"Cross-layer Qos Support for Multimedia delivery over Wireless Internet", EURASIP Journal on Applied signal processing 2005, pp207-219.



Audio & Video Quality Requirements

Medium	Application	Degree of symmetry	Typical data rates	Key performance parameters and target values			
				One-way Delay	Delay variation	Information loss	Other
Audio	Conversational voice	Two-way	4-13 kb <i>i</i> s	< 150 ms preferred* < 400 ms limit*	< 1 ms	< 3% packet loss ratio (PLR)	
Audio	Voice messaging	Primarily one-way	4-13 kb/s	< 1 s for playback < 2 s for record	< 1 ms	< 3% PLR	
Audio	High quality streaming audio	Primarily one-way	32-128 kb/s	< 10 s	< 1 ms	< 1% PLR	
Video	Videophone	Two-way	32-384 kb/s	< 150 ms preferred < 400 ms limit		< 1% PLR	Lip-synch < 80 ms
Video	One-way	One-way	32-384 kb/s	< 10 s		< 1% PLR	

* Assumes adequate echo control

Ref: INTERNET-DRAFT Strategies for Streaming Media Using TFRC Jan 2008



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Problems: Wireless Losses

 Problem: Two types of losses in wireless Networks
 Packet drops due to congestion
 Packet drops due to bad channel conditions





Ref O. B. Akan and I. F. Akyildiz, "ARC: the analytical rate control scheme for real-time traffic in wireless networks," *IEEE/ACM Transactions on Networking*, vol. 12, no. 4, pp. 634–644, 2004



Fig:1 Steady state desired behavior of a TCP source in wireless link

"Equation based rate control has lower variation in throughput over time compared with TCP. That's why it is more suitable for applications where smooth sending rate is required".



Some Other Schemes:

RCS[2001], WMSTFP[2003], WLED[2006], DCCP[2006],

Equation Based Rate control in Cellular Networks[2009];





Energy potential

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- Congestion losses
 - Burst congestion loses
 - Transient congestion losses
- Wireless losses
 - Transient/Random losses
 - Burst wireless losses

In case of burst wireless losses

nlost ∝ rate energy wastage ∝ nlost



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RCM-BR protocol overview

- Estimate RTT
- Differentiate wireless and congestion losses
- Detect burst wireless loss
- Estimate loss probability
- Estimate available B.W.
- Adjust the rate





Adaptive Loss Differentiation

• Infers the cause of loss: The threshold is a function of the minimum RTT, and current sample RTT so that it may automatically adapt itself to current congestion level.

$$T > \overline{T} + T_{dev} \cdot \left(2 \cdot \left(\frac{T_p}{T}\right)^K - 1\right)$$
$$\overline{T} = \frac{7}{8}\overline{T} + \frac{1}{8}T$$
$$T_{dev} = \frac{3}{4}T_{dev} + \frac{1}{4}\left|T - \overline{T}\right|$$



Estimating loss probabilities

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• Wireless loss

$$A = \begin{bmatrix} 1-p & p \\ q & 1-q \end{bmatrix} \qquad p = P(q_t = B|q_{t-1} = G);$$
$$q = P(q_t = G|q_{t-1} = B);$$

$$\pi_{B} = \frac{p}{p+q} \qquad \pi_{G} = \frac{q}{p+q}$$

$$p_E = (1 - P_G) \pi_G + (1 - P_B) \pi_B;$$

Congestion loss

 $p_c = (1 - \Pr) \pi_r + (1 - \Pr_l) \pi_l;$

Total loss

 $\Pi = 1 - (1 - p_E)(1 - p_c)$





Accuracy of Classification

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Type of algorithm	C/C	W/W	C/W	W/C
Spike	100%	0%	100%	0%
Zig-Zag	85%	60%	40%	15%
Adaptive LDA	82.5%	85.38%	14.62%	17.5%
SVM	88%	84%	16%	12%



Detecting burst wireless loss

burst-thrust function

$$g(k) = \begin{cases} 1, k \le \theta \\ \sqrt{k - \theta}, k > \theta \end{cases}$$

Where, $\theta = \max(4, \text{ previous rate*rtt/2})$ k is the no .of packets lost due to wireless loss

$g(k)>1 \Rightarrow$ Burst wireless loss





Algorithm

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Probe()

While(t<=RTT) Send PROBE_PACKET with S_{target} End; Estimate initial rate end;

STEADY()

```
Calculate RTT;

Calculate g(wlost)

If (g(wlost)=1)

Calculate \pi, \omega;

Calculate T;

Rf=(now-lastchange)/RTT

If(T>rate_)

rate=rate+Rf*(1-\pi)*T;
```

else

Burst recovery()
calculate g(k)
if (g(k)>1)
 rate=(1-w)*rate;
else
 rate=prev_g(k)*rate
 steady()

end;

else

```
rate=(1-w)*rate
burst recovery()
end;
```

Where R_f is the rate change factor that detects how many times the sending rate is changed with respect to RTT. 'now' represent the current time and 'last change' represents the last time when the rate was changed.



Probe period

- Initially sender sends probe packets with target rate till RTT.
 - Low priority dummy packets
- Receiver sends acks to received probe packets
- At 2*RTT, sender sets initial transmission rate

$$\mathbf{S}_{\text{init}} = \frac{\max\{1, n_{\text{ack}}\}}{\text{RTT}}$$





calculate RTT Calculate g(k) If Burst wireless loss is not detected Calculate Π, ω Estimate available B.W $T = \frac{s}{4.RTT} \left(3 + \sqrt{25 + 24 \left(\frac{1 - \omega}{\pi - \omega} \right)} \right)$

Rf=(now-lastchange)/RTTIf(T>rate) $rate=rate+Rf*(1-\pi)*T;$ else

rate=*a* * *T*+(*1*-*a*) **rate*



Burst recovery

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calculate g(k)
if (g(k)>1)
 rate=max(minrate,(1-w)*rate);
else
 rate=prevg(k)*rate



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- Pareto traffic to generate congestion losses in wired bottleneck
- Three state markov model for wireless losses



Performance Metrics

- Throughput or Goodput.
- Delay or latency.
- Delay Variation Jitter.
- Packet loss rate.

• Fairness Index,

$$F.I = \frac{\left(\sum_{i=0}^{n} Throughput_{i}\right)^{2}}{n\left(\sum_{i=0}^{n} Throughput_{i}^{2}\right)}$$

Ref: RFC5166,S.Floyd,March 2008.



GoodPut Vs Time for WP=0.1.

GoodPut Vs Time for WP=0.3.











• With wp=0.2







• Energy efficiency Vs Wireless loss probability



• Throughput Vs Wireless loss probability



- With wp=0.1
- Starts between 1 and 3 sec



• FI Vs No. of flows



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Conclusion

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- RCM protocol augmented with burst recovery(RCM-BR) improves the utilization and fairness of the transport protocol, while achieving high energy efficiency.
- Improve the end-to-end loss differentiation algorithm
- Improve burst detection mechanism









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Back-up slides

55)

Goodput?

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- **Goodput** is the application level throughput, i.e. the number of useful data bits delivered by the network per unit of time.
- Excludes protocol overhead bits as well as retransmitted data packets.

RCM

- "now" is the current time and "lastchange" is the time when RTT was changed last.
- RTT is round trip time calculated using exponential weighted moving average.
- In the equation S_init=max(1, n_ack)/RTT,
- "n_ack" is the number of acknowledgements received by the sender for the probe packets sent in second RTT time.
- Initially the probe packets will be sent with a rate of "S_target", which will be given by the application to achieve highest quality (maximum rate needed). In the simulation it is given manually in the TCL file with the variable like target_rate.

Goodput?

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- For example, if a file is transferred, the goodput that the user experiences corresponds to the file size in bits divided by the file transfer time. It is lower than the throughput.
- Examples of factors that cause lower goodput than throughput are:
- *Protocol overhead,* such as transport layer, network layer and sometimes datalink layer protocol overhead is included in the throughput, but is excluded from the goodput.
- *Transport layer flow control and congestion avoidance*, e.g. TCP slow start, may cause a lower goodput than the maximum throughput.
- Retransmission of lost or corrupt packets due to transport layer automatic repeat request (ARQ), caused by bit errors or packet dropping in congested switches and routers, is included in the datalink layer or network layer throughput but not in the goodput.

Performance of Loss differentiation algorithm

• Accuracy of Adaptive loss differentiation without link layer ARQ

C/C	W/W	C/W	W/C
82.5%	85.38%	14.62%	17.5%

Accuracy of Adaptive loss differentiation with link layer ARQ

	W/W	C/W	W/C
C/C			
95.3	0%	0%	4.7



Combination of wired and wireless networks