Department of Computer Science
National Tsing Hua University

CS 5263: Wireless Multimedia Networking Technologies and Applications

Network-Adaptive Media Transport

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Acknowledgement: The instructor thanks Prof. Mohamed Hefeeda at Simon Fraser University for sharing his course materials

Outline

- Introduction
- Rate Distortion Optimized Framework
  - Basic Framework
  - Receiver Driven Streaming
- Rich Acknowledgements
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- Dependant Packet Delay
- Congestion Distortion Optimized Streaming
- Conclusion

Introduction

- **Internet Packet Delivery**
  - **Loss**
  - **Throughput**
  - **Delay**

- **Challenge : Maximize quality of audio and video considering transmission rate and latency constraints**

  ➡ **Network Adaptive**

Multimedia Streaming Systems

- Client Application
  - Error detection and concealment
- Transport Mechanism
  - Congestion control by retransmission and packet drops
- Encoder
  - Rate scalable coding
- Media Server
  - Intelligent transport by sending the right packet at the right time

Rate Distortion Optimized Framework

- Framework has been propose by Chao and Miao

- Goal : Compute which packets to send and when to minimize the reconstructed distortion
  - To maximize the video quality

Basic Framework

- Media Server has media streams packetized into data units

- Framework chooses optimal set of data units to transmit at successive transmission opportunities

- Scheduler decides based on an entire optimized plan

Parameters

- Data unit : *l*
- Size : *B(l)*
- Deadline :
  - The maximum arrive time to be useful for decoding
- ΔD_l : distortion reduction
  - which is decrease in distortion rate if *l* is decoded
- *N* : transmission opportunities
- π : transmission policy
  - It has *N* binary vector $\pi(l)$ for each data unit *l*
- *ε(l)*: error probability
  - data unit *l* received late or not at all
- *P(l)* :
  - number of times packet has been sent

Basic Framework

- Policy π wants to find the best tradeoff between expected transmission rate and distortion construction
- Formally , minimize Lagrangian function:

$$J(\pi) = D(\pi) + \lambda R(\pi)$$

# of transmissions

$$R(\pi) = \sum_{l} \rho(\pi_l) B_l$$

$$D(\pi) = D(0) - \sum_{l} \Delta D_l \prod_{l' \leq l} (1 - \varepsilon(\pi_l)$$

Maximum distortion

Quality improvement

Needs of RaDiO Framework

- Packet loss and delay are considered independently
  - Packet Loss → Bernoulli Function
  - Delay → Shifted τ−distribution
- Exhaustive Search is not useful. The search space grows exponentially

→ Chao and Miao proposed RaDiO framework

ISA Algorithm

- **The radio framework uses conjugate direction search**
- **The Iterative Sensitivity Algorithm minimizes the Lagrangian function**
  - **The policy for $\pi(l)$ is optimized while others are fixed.**
  - **It runs for every l in round robin fashion in order for π to achieve a (local) minimum**

ISA Algorithm (cont.)

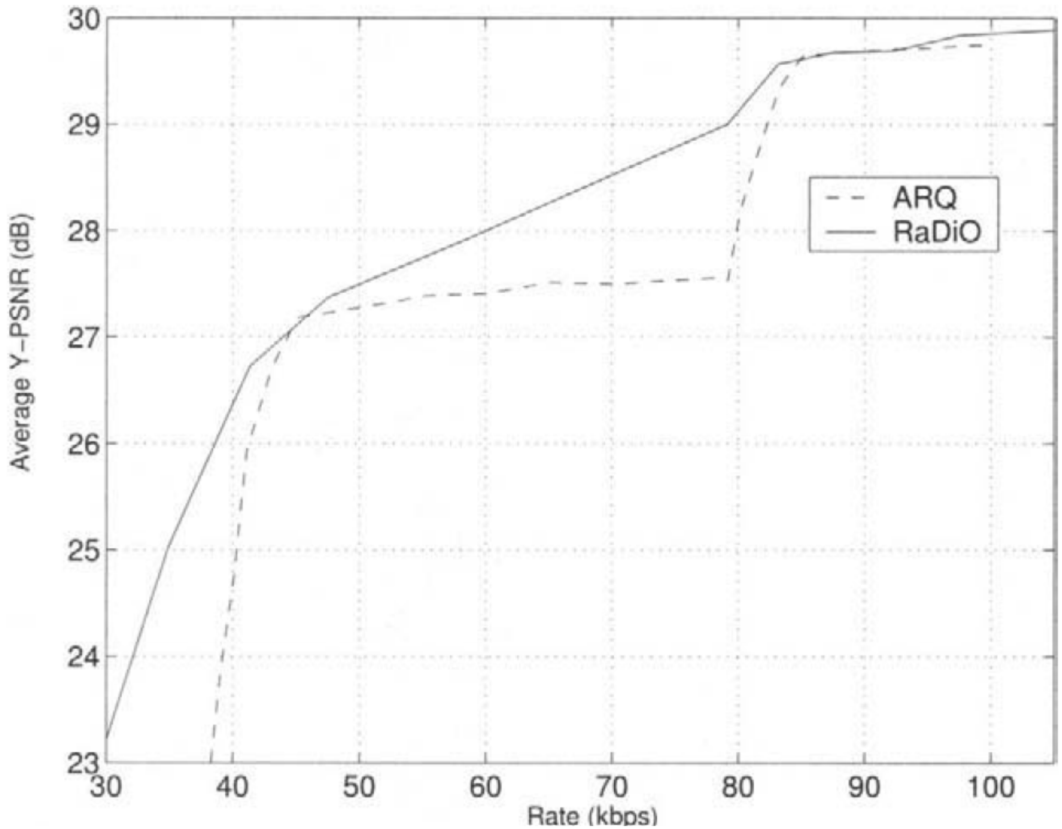
$$J_l(\pi_l) = \varepsilon(\pi_l) + \lambda' \rho(\pi_l)$$

where $\lambda' = \dfrac{\lambda B_l}{S_l}$ is the rate distortion tradeoff multiplier

B_l is the data unit size

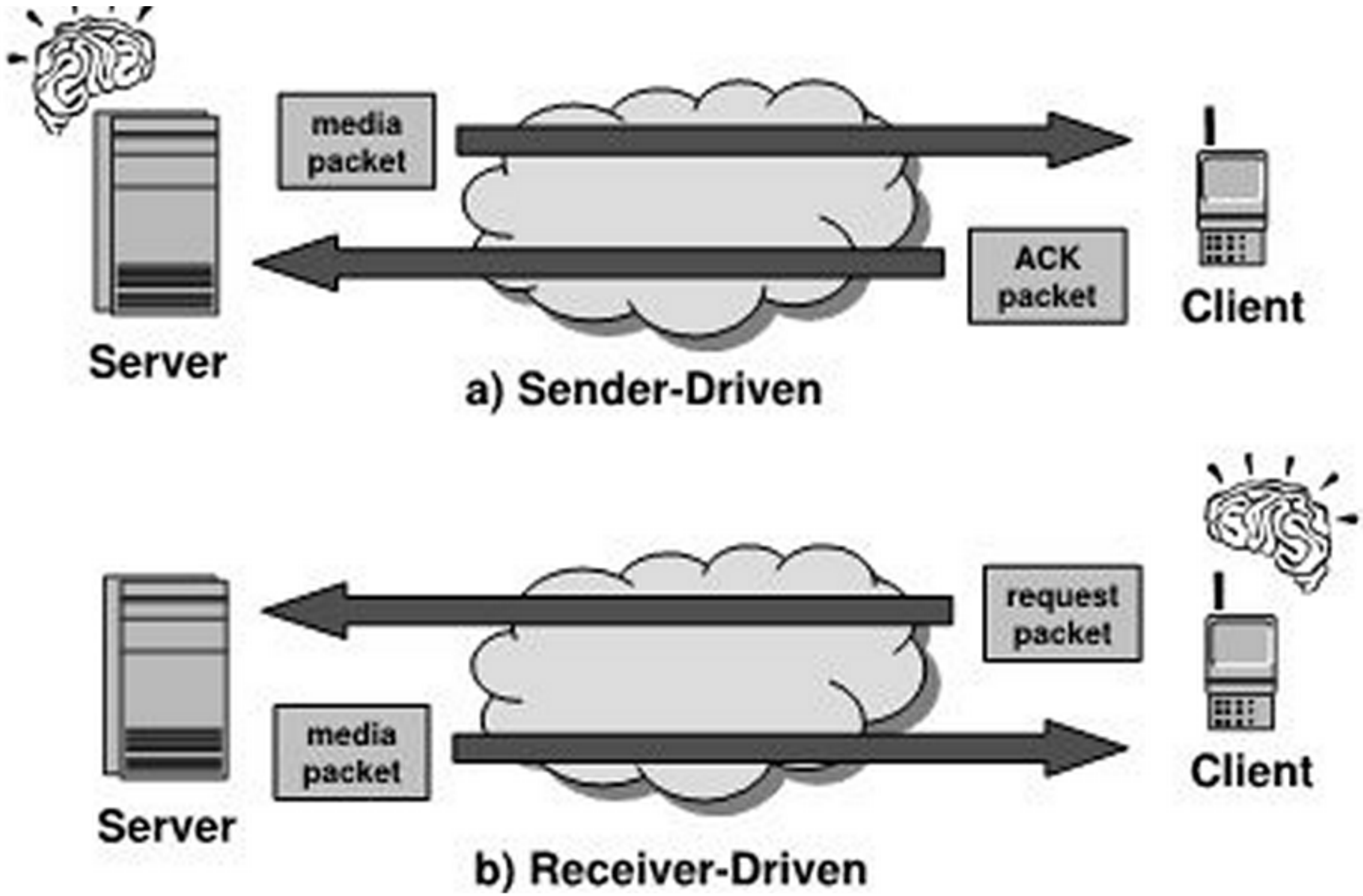
S_l is the sensitivity of overall distortion to error probability of data unit *l*

Performance Comparison



RaDiO improves video streaming performance

Receiver versus. Sender Driven Streaming



Receiver Driven Streaming

- Transmitting many video and audio make the server become computationally overwhelmed.

⇒ Shift Computation to the client as much as possible

Strategy

- **Client will be provided with information about size, distortion reduction values , interdependencies of data.**
  - **Such R-D overhead is small ← several bytes versus KB of video data**

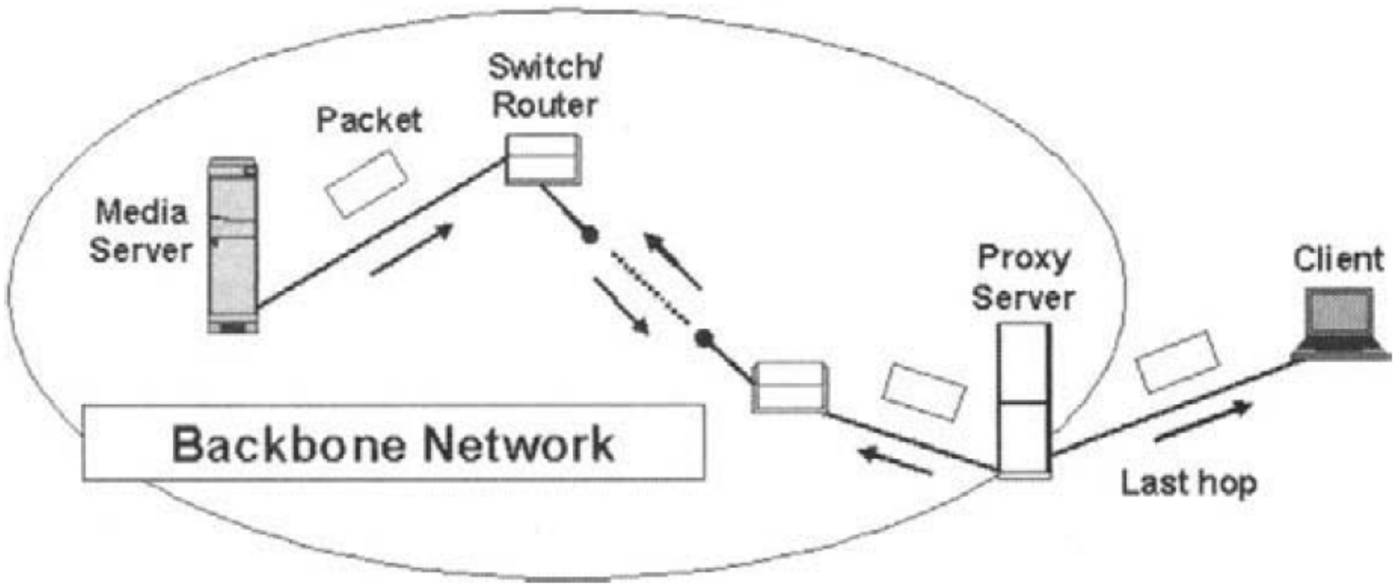
- **It computes optimized scheduler and compute sequence of requests that specify data units.**

Hybrid Solution

- Combining sender driven and receiver driven approach can be used to R-D optimized algorithm to diverse network topologies.

➡ Example : Using radio framework in a proxy between network backbone and last hop link.

Architecture



Proxy Driven RaDiO Streaming

Benefits

- **Proxy uses hybrid of sender and receiver streaming.**
- **It improves the end-to-end performance. The traffic caused by retransmission of lost packets is not traversed to server and stays in last hop**

Rich Acknowledgements

- **Instead of sending ACK for each received packet , send the state of received packets periodically that ACKs received packets and NACKs lost packets.**

➡ **Needs changes in the basic framework**

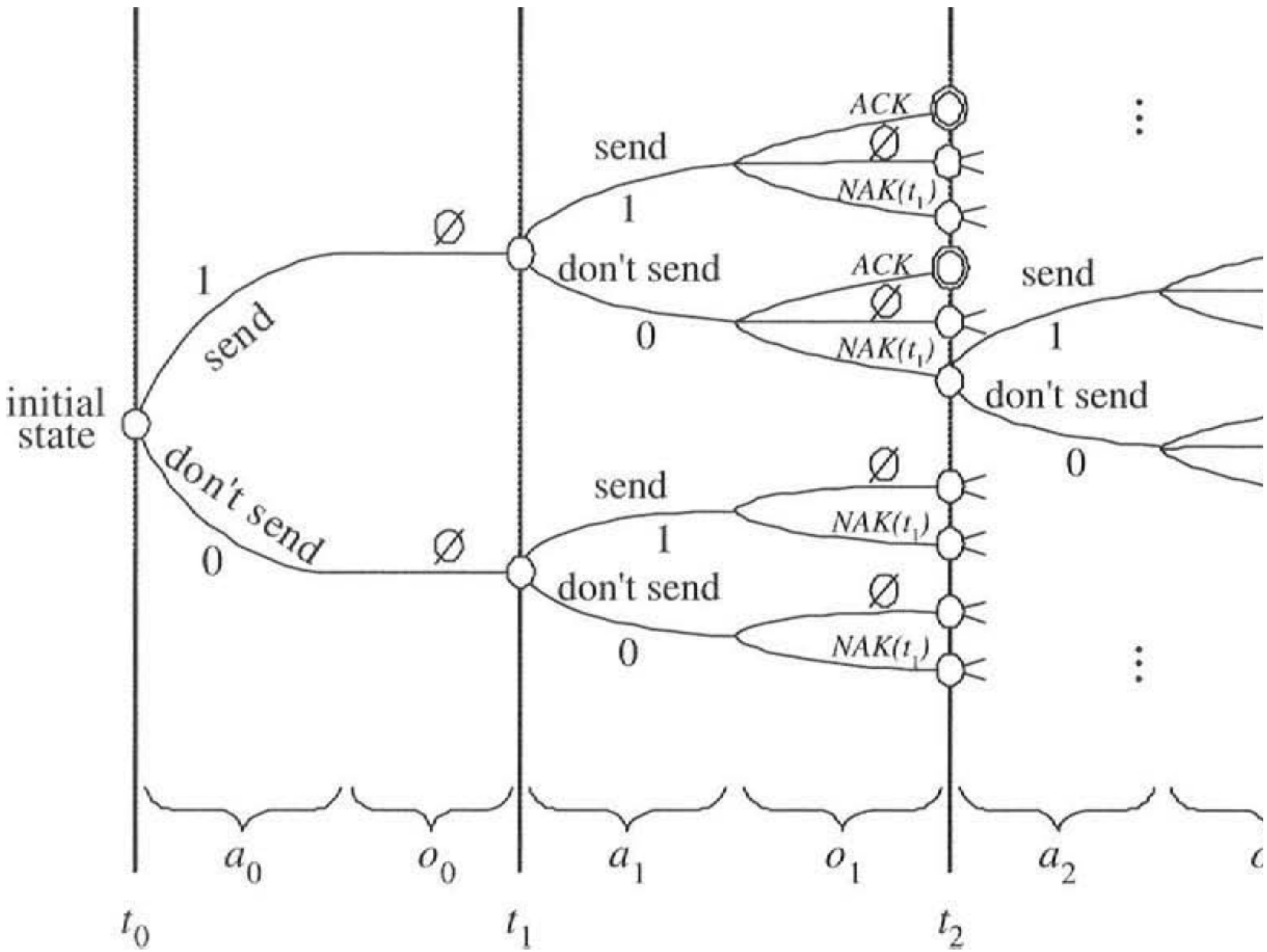
Markov Decision Process

- **Transmission policy of a data unit can be modeled by Markov Decision Process.**
- **At time *t(i)*, server makes observation *o(i)* and takes action *a(i)* which is send or don't send .**
- **The sequence of <*o(i)* , *a(i)*> forms a Markov decision tree.**

Actions and Observations

- **Possible actions :**
  - Send
  - Don't send
- **Possible observations :**
  - $\varnothing$ no relevant feedback has arrived
  - ACK feedback packet acknowledged received data unit
  - NACK feedback packet indicate lost data unit

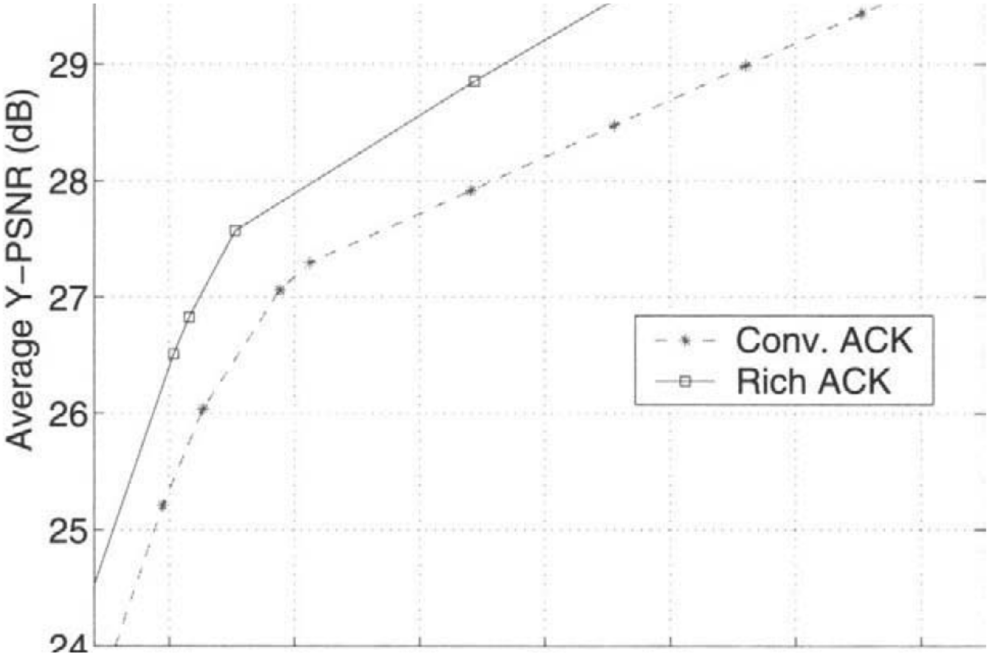
Markov Decision Tree



Optimization Algorithm

- Optimization algorithm calculates the probabilities through each path given the policy and find the best tradeoff between expected number of transmission $p(l)$ and loss probability $e(l)$

Versus Conventional Acknowledgement



Sample results from QCIF foreman

Reasons of Improvements

- Effect of lost ACK packet is mitigated because subsequent feedback packets contain same information
- Less ambiguity for server by having NACKs

Multiple Deadlines

- Instead of discarding the frame arrive later than the associated deadline, we consider that frame will be useful for decoding other frames or at least itself.

We associate multiple deadlines to one frame

Accelerated Retroactive Decoding (ARD)

- **Example :**
  - **We have a set of frames IBBBP. If frame p arrives later than deadline, it still can help in decoding the next B frames.**
  - **Decoders that allow *Accelerated Retroactive Decoding* (ARD)**
    - **It allows many streaming clients to decode video faster than real time. When the late frame arrives, it goes back to dependant frames and decode the dependent frames up to play-out time. ← in contrast to waiting for late frames!**

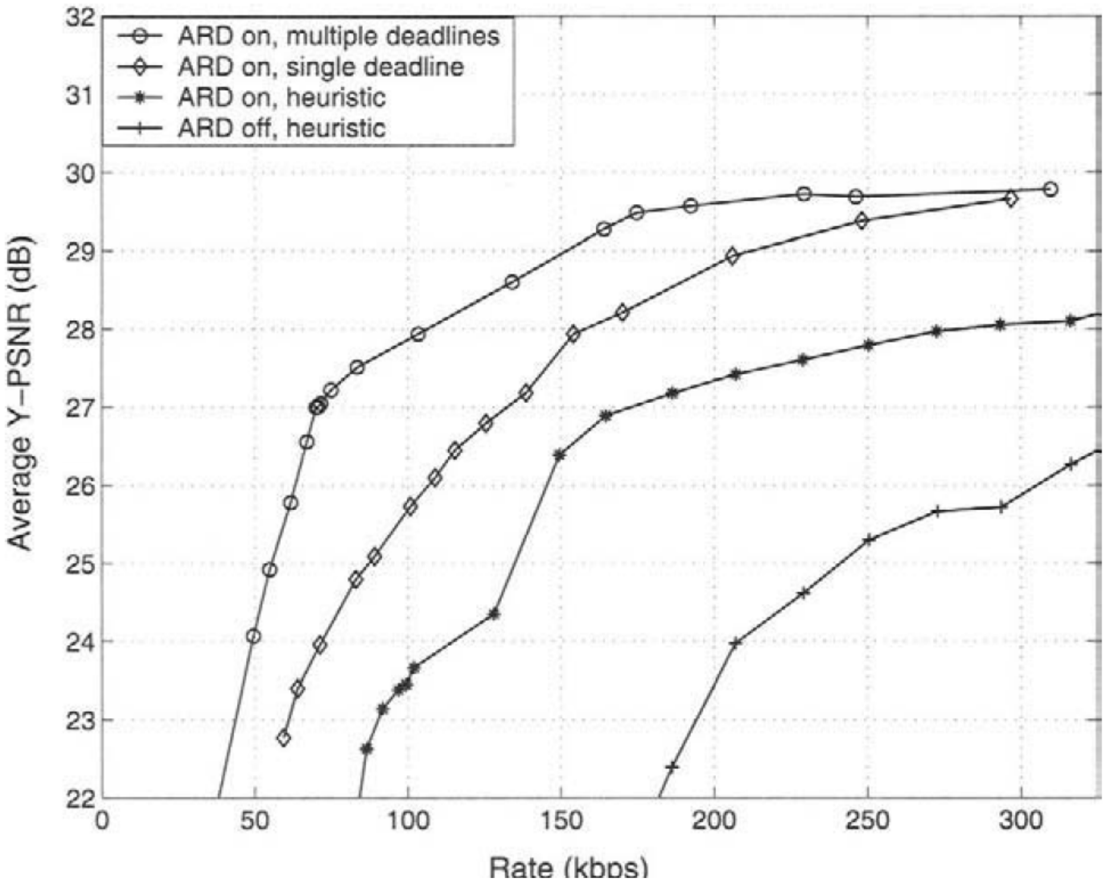
Formulation

- **Need changes in the formulation :**
  - **We have to add the error probability for each deadline**

$$J_l(\pi_l) = \rho(\pi_l) + \sum_{i \in W_l} v_{t_i}\, \varepsilon(\pi_l, t_i)$$

  - **In the above equation, $v_{t_i} = \frac{S_{l,t_i}}{\lambda B_l}$ is the sensitivity factor that depends on each deadline and is the sensitivity of overall distortion if data unit l arrives in t_i.**

Results



ARD ←multiple deadlines

Discussion

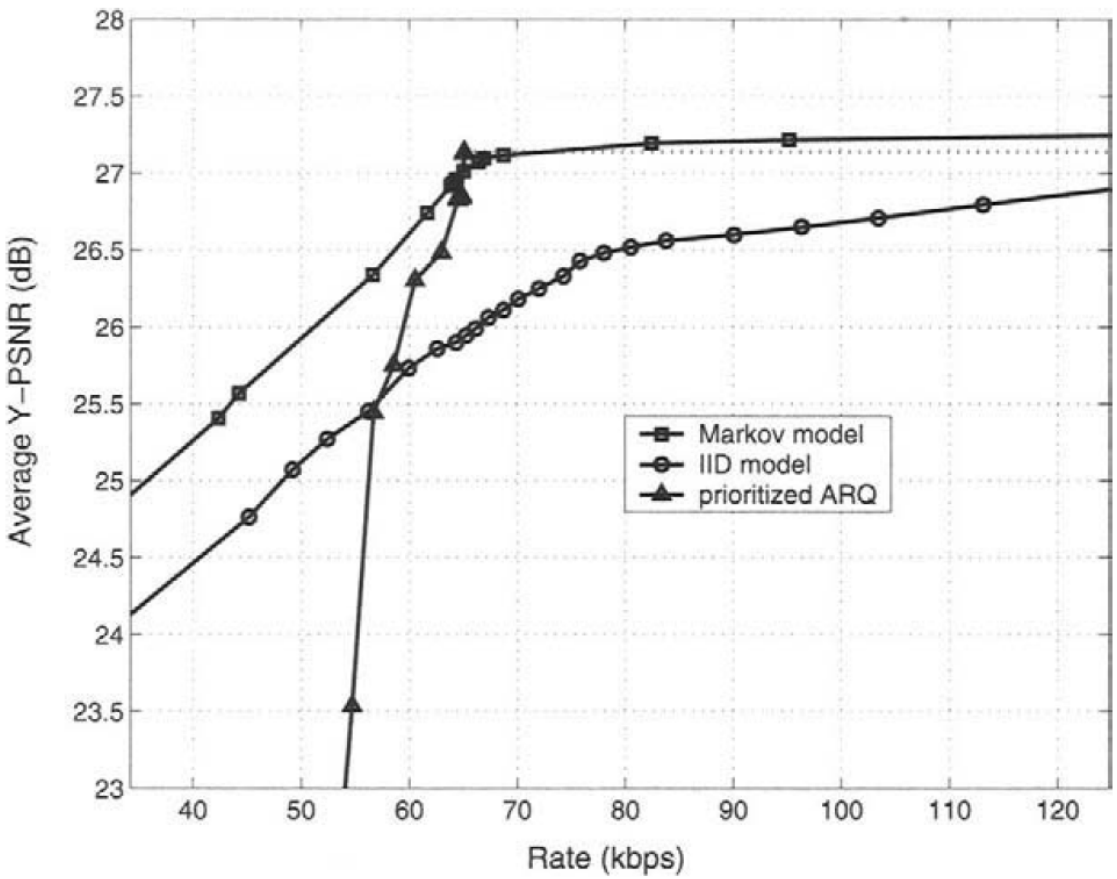
- **Multiple deadline approach take the benefit of using the information of late arrived packet, therefore they improve PSNR compared to single-deadline scheme.**

Dependent Packet Delays

- **In the original framework :**
  - **Packet Delay : Shifted Γ-distribution**
  - **Loss : Bernoulli Model**
- **Packet delays are assumed to be independent of each other which simplifies the calculation of error probability**

This is not realistic **Suboptimal performance**

Problem with i.i.d. Assumption



Rate distortion performance scheduler for Foreman sequence streamed over measured Internet delay trace

Benefits of ARD

- **In higher rate, heuristic ARD (multi-deadline) will outperform i.i.d.- model:**

The algorithm mistakenly believes that if a data unit arrives late, other data units will arrive on time or earlier. Therefore, it sends packets multiple times even though packet loss in low (0-14%)

Improvement

- **Model the delay at successive transmission time slots as a first order Markov random process.**

➡ **Feedback packets will inform server about the delay over channel in the recent past.**

Congestion Distortion Optimized Streaming

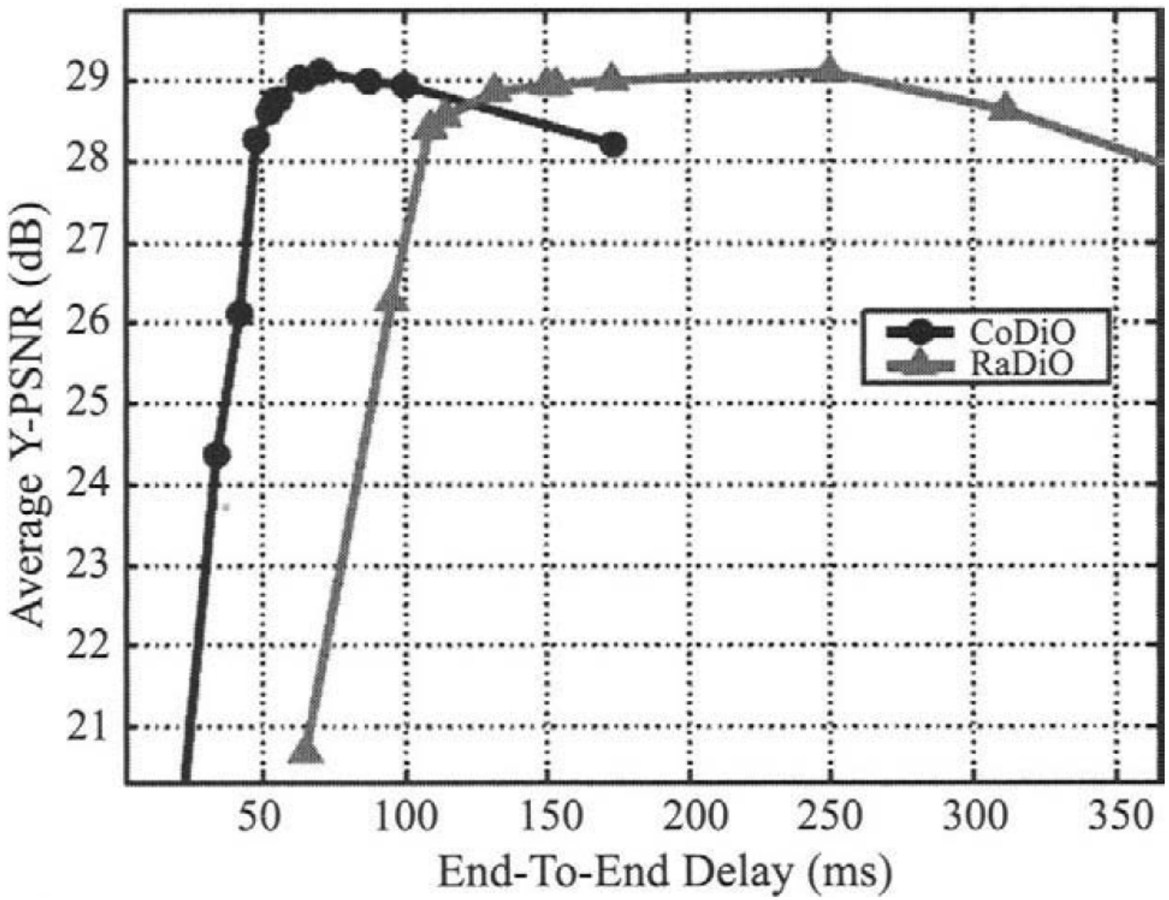
- In RaDiO streaming approach, packet delay is not affected by transmitted packets.

- Delay is a random variable with parameterized distribution that adapts slowly according to feedback information.
  - Media steam transmitted at negligible rate to bandwidth , model is acceptable. But, in higher rates it is not.

Improvement

- **Congestion distortion optimized streaming (CoDiO)**
  - **Effects of transmitted packets is considered. It gets an optimal tradeoff between congestion and reconstructed distortion.**
  - **It assumes a succession of high bandwidth link followed by a bottleneck last hop used by media streams.**

Performance Comparison: RaDiO and CaDiO



Performance Comparison of Codio and Radio Streaming for video streaming over a bottleneck link

Observation

- **CoDiO outperforms Radio:**
  - **It transmits packets as late as safely possible. This reduces the congestion in backlog and therefore end-to-end delay.**

Conclusion

- To get better quality of audio and video streams , media streaming should be network adaptive

- Media server can decide which packets and when to send to optimized the distortion of decoded video

- Radio Framework proposed to avoid exhaustive search

- There are several extensions to the basic framework which improve performance
  - Rich Acknowledgement
  - Multiple Deadline
  - Dependant Packet Delay
  - Congestion Distortion Optimized Streaming