Department of Computer Science
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CS 5263: Wireless Multimedia Networking Technologies and Applications

Network-Adaptive Media Transport

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Outline

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- 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 Chou 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**

Variables

- **Data unit :** ***l***
- **Size :** ***B(l)***
- **Deadline :**
  - **The maximum arrive time to be useful for decoding**
- ΔD_l **: distortion reduction**
  - **The decrease rate of distortion if *l* is decoded**
- *N* **: transmission opportunities**
- π **: transmission policy**
  - **It has *N* binary vector *π(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

→ Chou 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.)

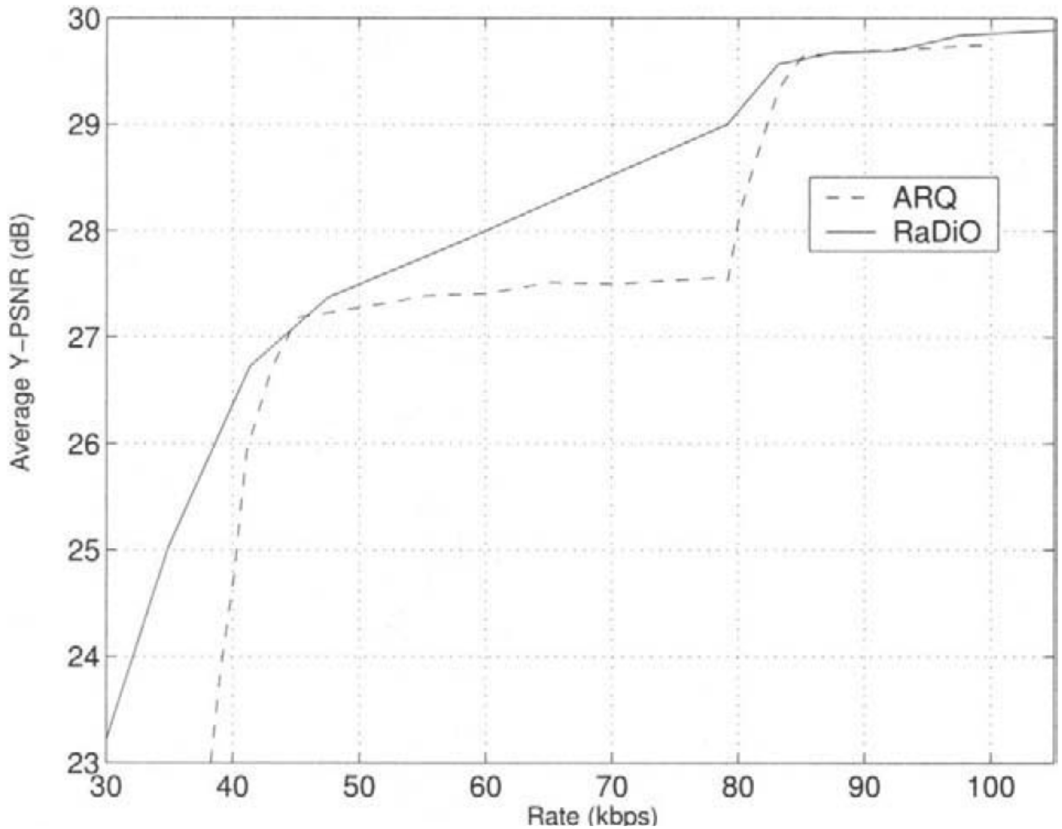
$$J_l(\pi_l) = \varepsilon(\pi_l) + \lambda' \rho(\pi_l)$$

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

B_l is the data unit size

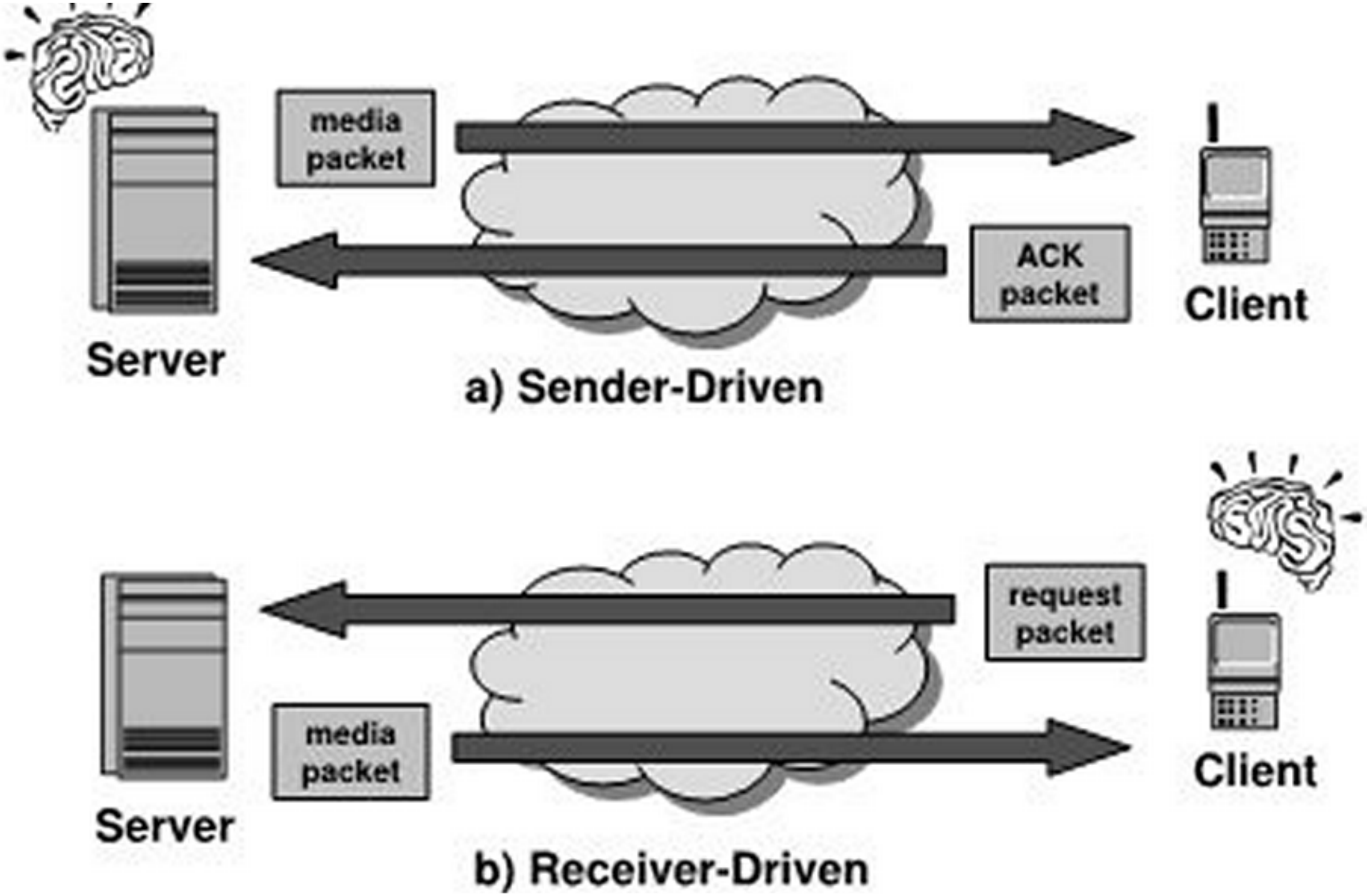
S_l is the sensitivity of overall distortion to error probability of data unit *l*

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

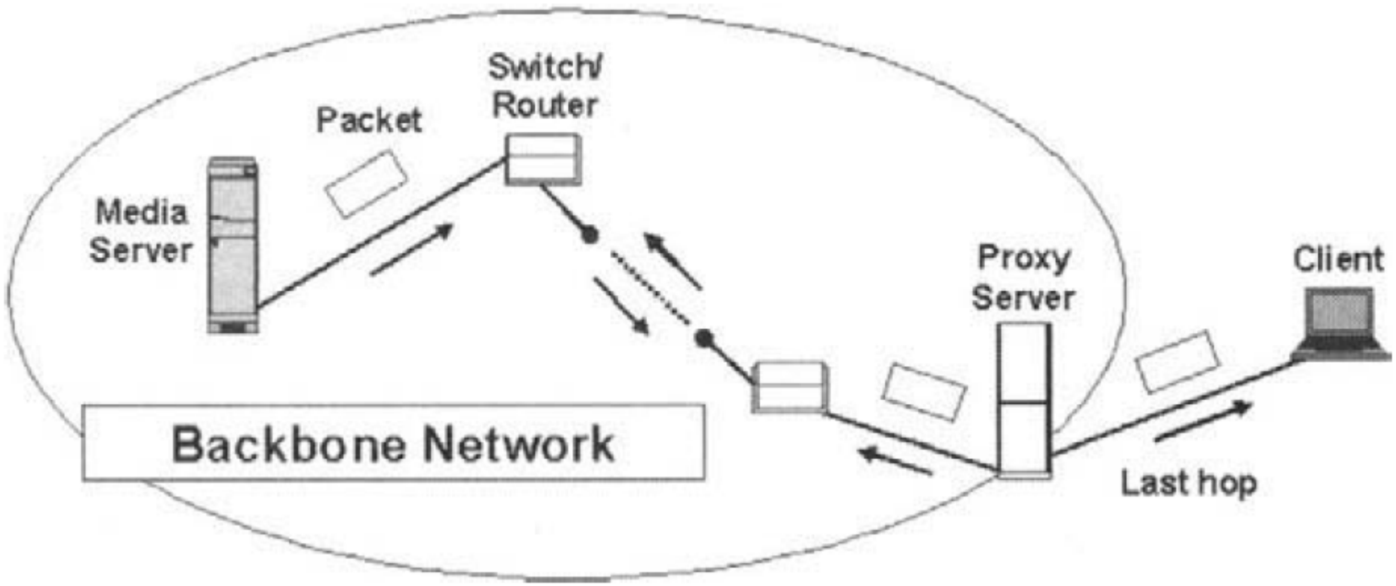
- It computes optimized schedule and compute sequence of requests that specify data units.

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

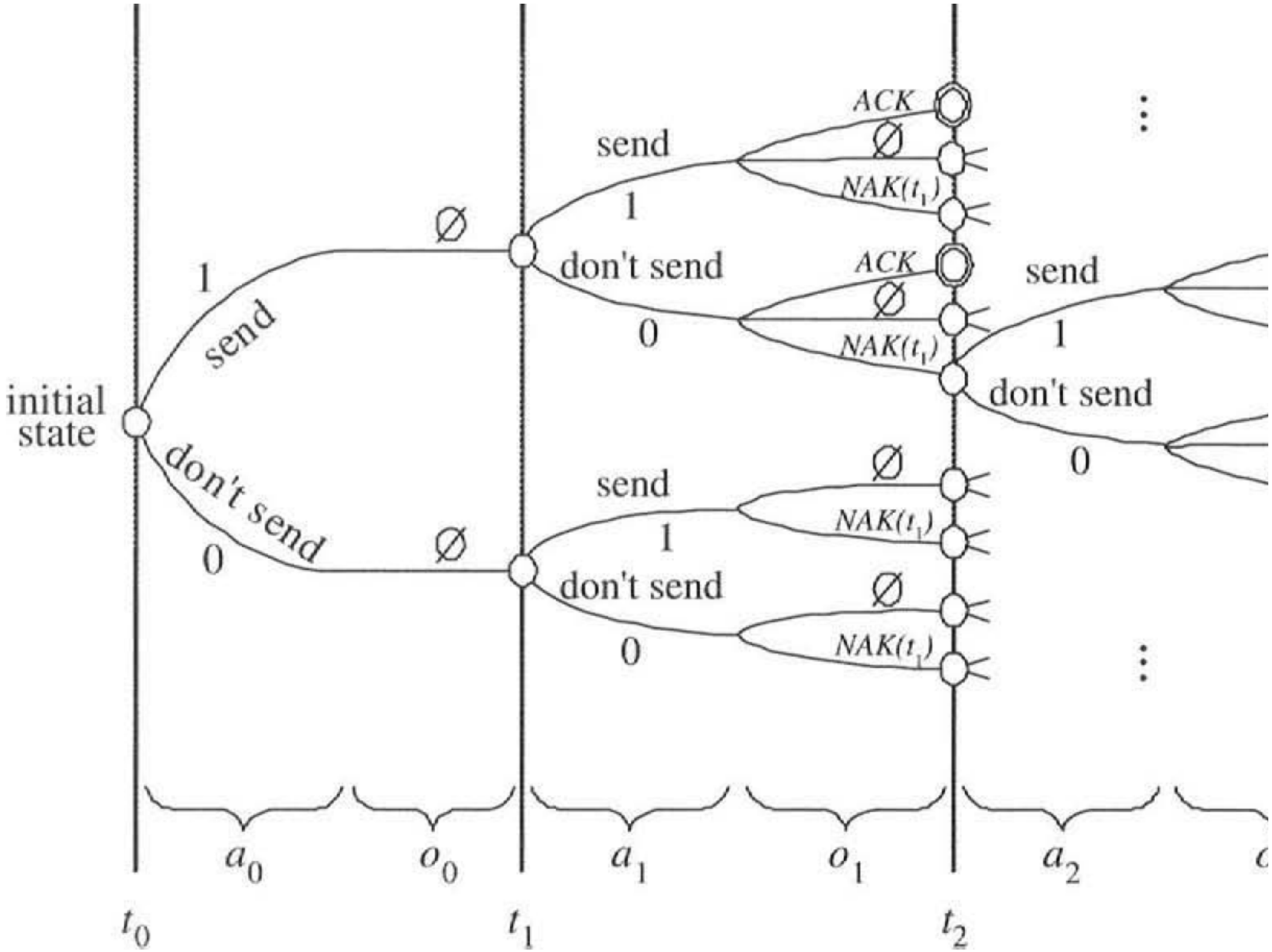
Markov Decision Process

- **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**

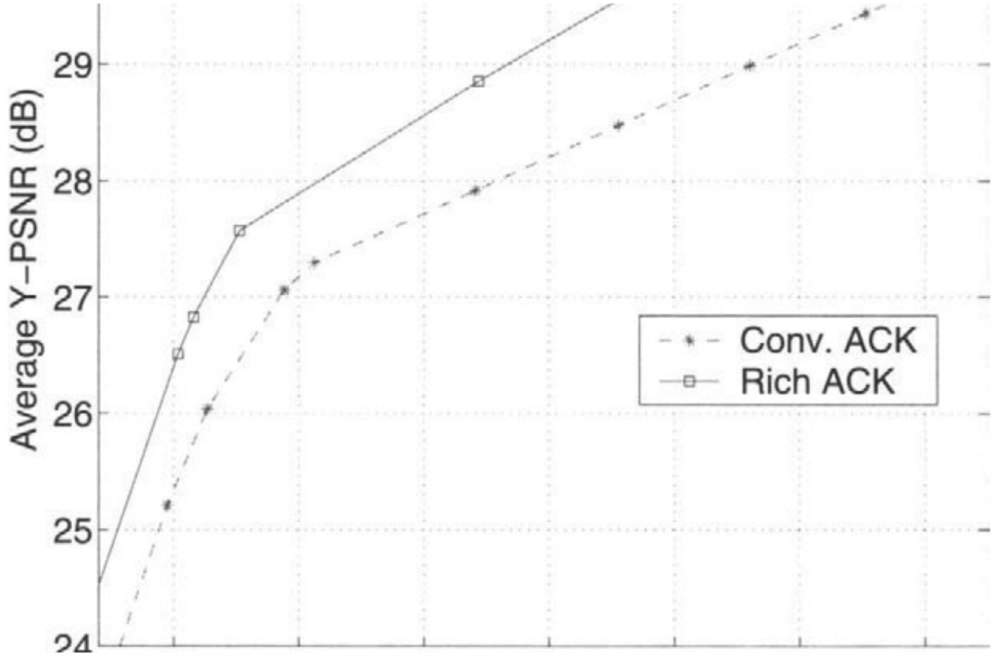
Markov Decision Tree



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!**

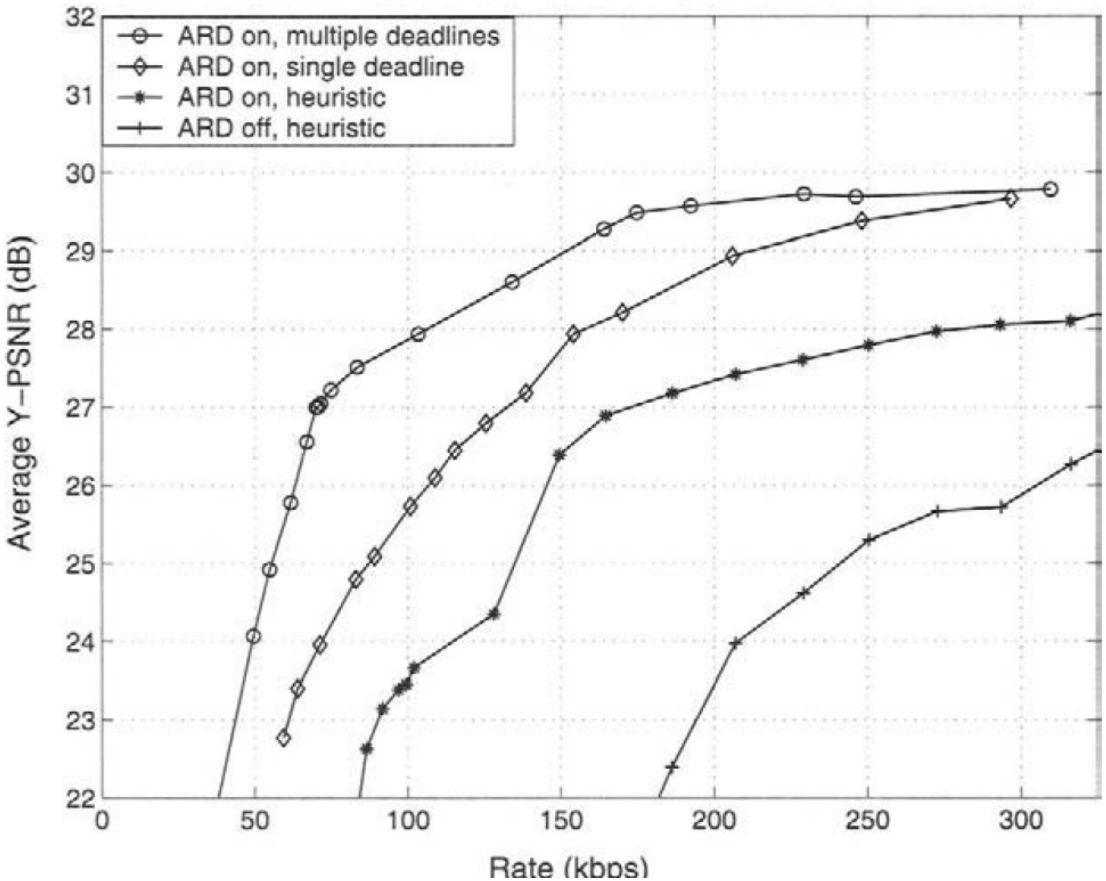
Formulation

- **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

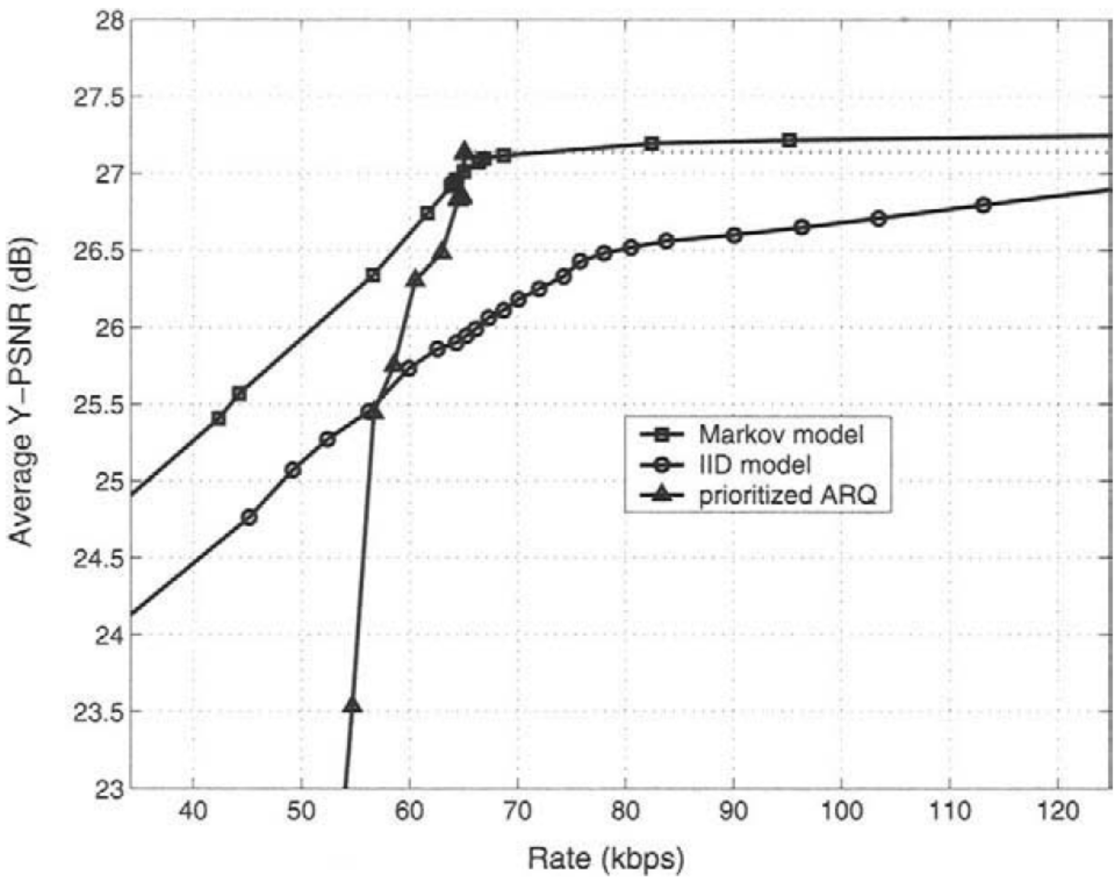
- **Multiple deadline approach take the benefit of using the information of late arrived packet, therefore they improve PSNR compared to single-deadline scheme.**

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

Performance of ARD

- **In higher rate, heuristic ARD (multi-deadline) outperforms 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%)

Solution Approach

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

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

Congestion Distortion Optimized Streaming

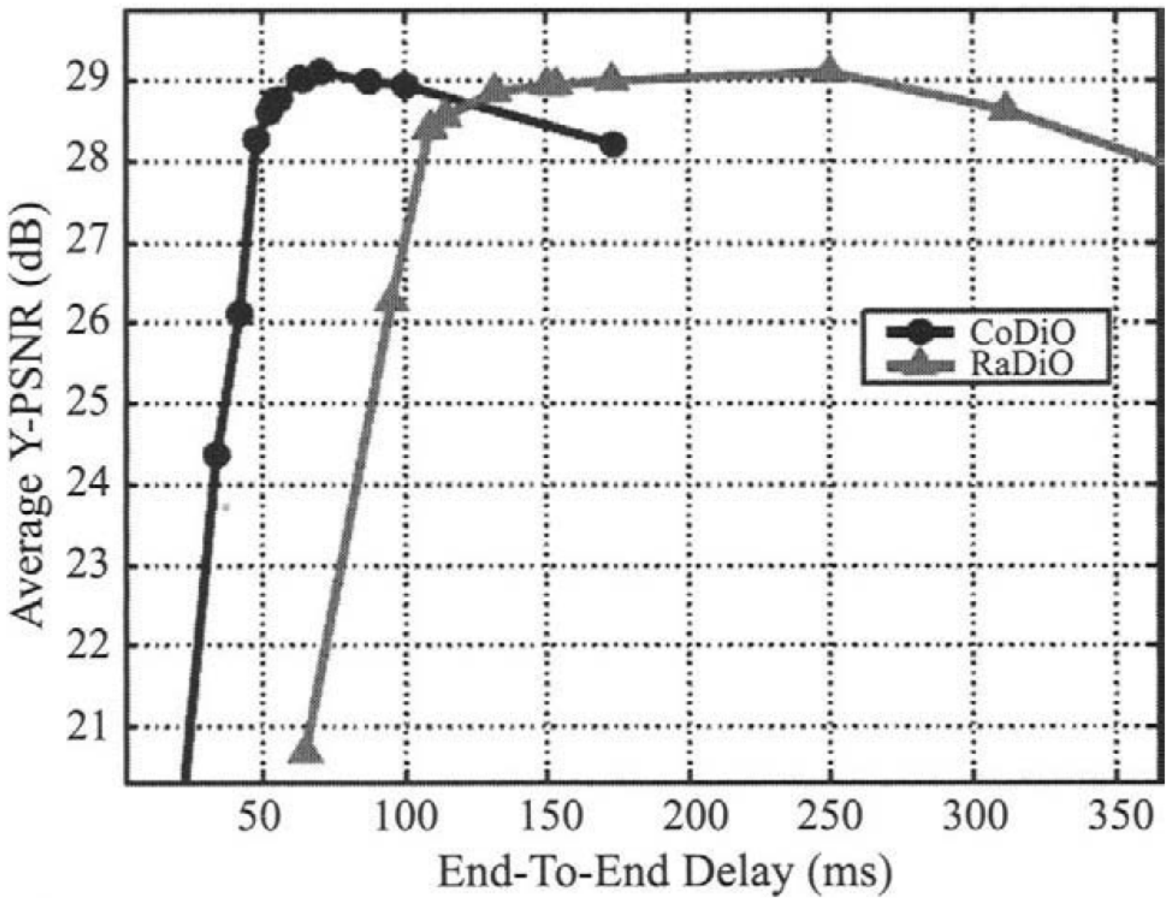
- In RaDiO streaming approach, packet delay is not affected by transmitted packets.

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

Congestion-Aware Streaming

- **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