Video Streaming in Wireless Environments

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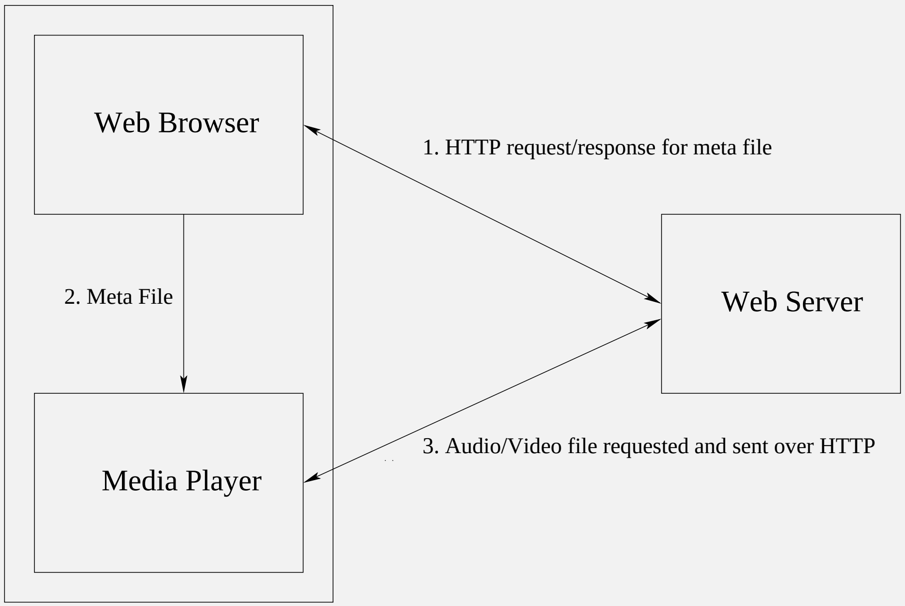
Motivation

- Refers to real-time transmission of *stored video*
- Has stringent bandwidth, delay and loss requirements
- Approaches
  - New protocols, router scheduling disciplines
  - Adapt output rate of video to available bandwidth
- *Rate Control Schemes* - employ feedback (loss, delay etc.,)
- Need to adapt *rate control schemes* to wireless environments
- **What is specific to Video Streaming ?**
  - Application layer QoS control
  - Affects user-perceived presentation quality

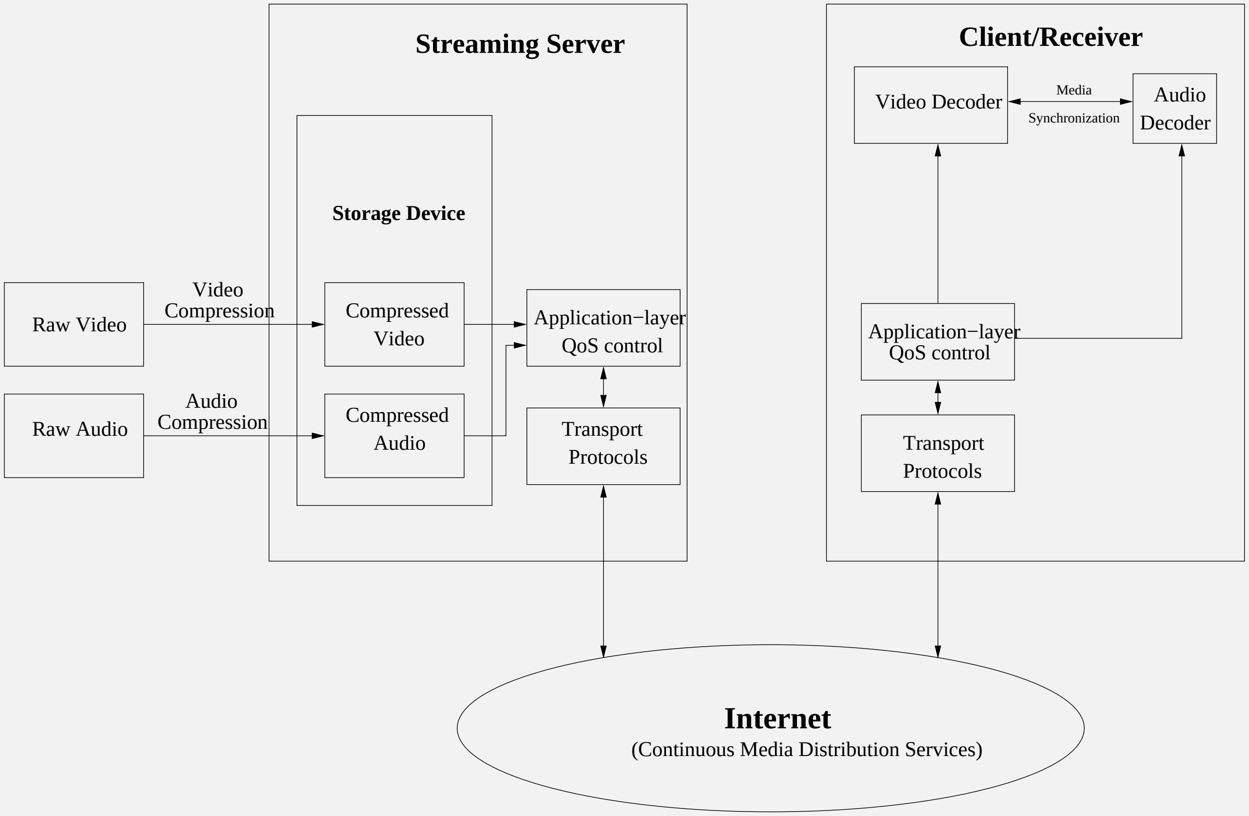
Architectures for Video Streaming

- **HTTP based Streaming**
  - Standard web servers used to deliver video content
  - Guaranteed-delivery protocols (like HTTP, TCP etc.,) not optimal for continuous media



  - Substantial fluctuations in delivery times of packets due to re-transmission, available bandwidth variations etc.,

- **UDP/RTP based Streaming**
  - Streaming Server retrieves media components in a synchronous fashion
  - Video sent over UDP using application-layer protocols tailored for video streaming (e.g., RTP)



(Figure from Wu et al, Streaming in the Internet : Approaches and Directions)

Transport Protocols (RTP/RTCP)

- **RTP**
  - Provides end-end transport functions for supporting real-time applications
  - Functions for media streaming like
    - sequence numbering
    - time-stamping
    - payload identification
- **RTCP**
  - Works in conjunction with RTP
  - Designed to provide QoS feedback to participants

Application-Layer QoS Control : Rate Control

- Minimizes network congestion by adjusting the output rate of the video coder to estimated available bandwidth
- Classified into
  - Source-Based Rate Control
  - Receiver-Based Rate Control
- Source based rate control schemes may use
  - Probe-Based Approach

    Example : AIMD, MIMD Algorithms etc.,
  - Model-Based Approach

    Example : TFRC, RAP Algorithms etc.,

Rate Control in Wireless Environments

- Characteristics of Wireless Channels
  - Limited Bandwidth
  - High Error Rates
  - Burst Errors
- Loss based rate control schemes may inaccurately estimate the available bandwidth
- AIMD based on packet loss fraction during each interval
- In *TCP Friendly Rate Control (TFRC)*,

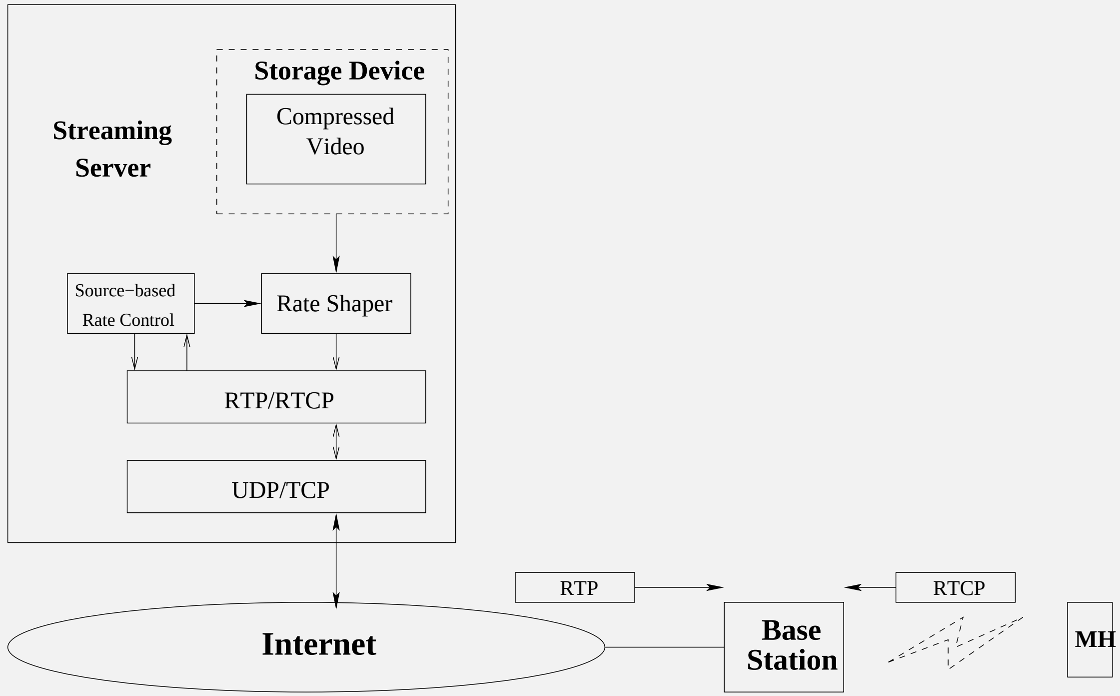
$$\lambda = \frac{1.22 \times MTU}{RTT \times \sqrt{p}} \tag{1}$$

- Assuming MTU and RTT constant,

$$\lambda \propto \frac{1}{\sqrt{p}} \tag{2}$$

The Problem

- During bad channel conditions, loss rate reported by receiver may be high
- Sender may inaccurately assume the network to be congested and decrease the output rate
- Hence, quality of video delivered to the receiver affected

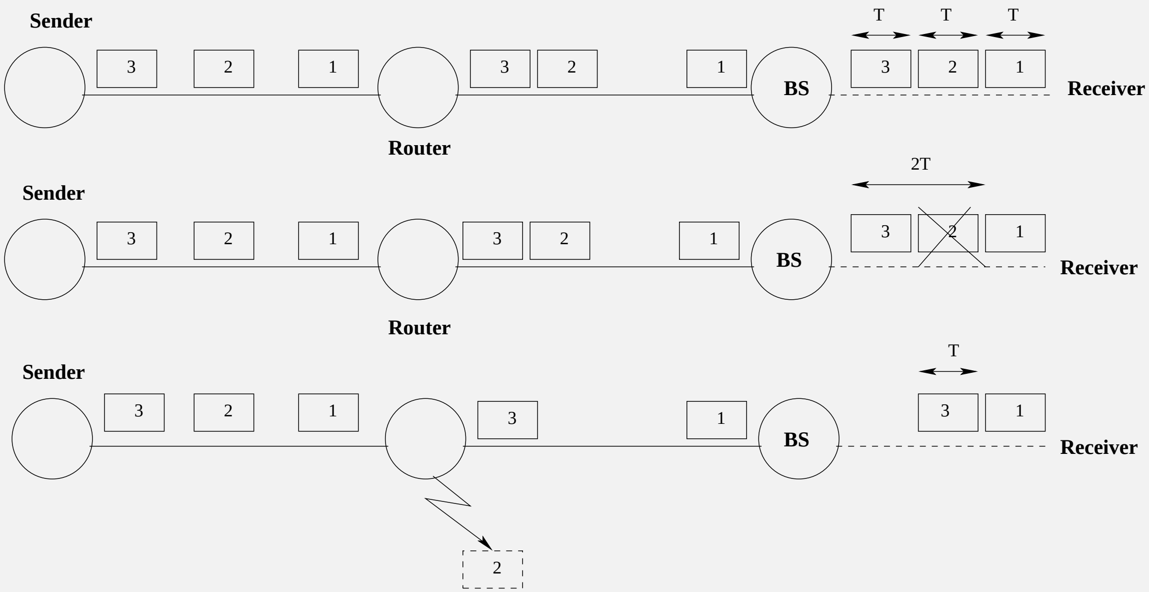


Solution Scheme(s)

- Prime reasons for the problem
  - Inability of receiver to distinguish between congestion and wireless packet losses
  - Sender estimates state of network using *loss rate* as principal feedback parameter
- Two Schemes proposed
  - Report Only Congestion Losses (ROCL)
  - Report Correlation of Loss and Delay (RCLD)

Report Only Congestion Losses (ROCL)

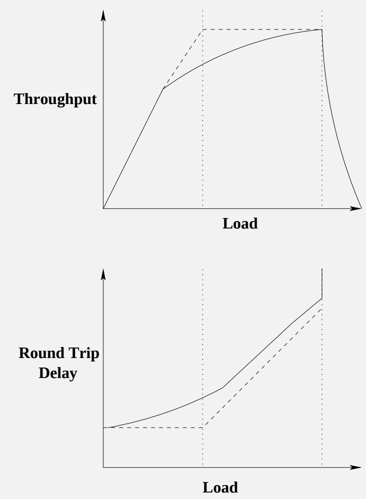
- Receiver enabled to report loss rate only due to congestion
- Uses heuristic proposed by Saad Biaz et al to discriminate congestion and wireless losses
- Heuristic based on inter-arrival times of packets at the receiver



(From Nitin Vaidya et al, Discriminating Congestion Losses and Wireless Losses Using Inter−arrival times at the Receiver)

Report Correlation of Loss and Delay (RCLD)

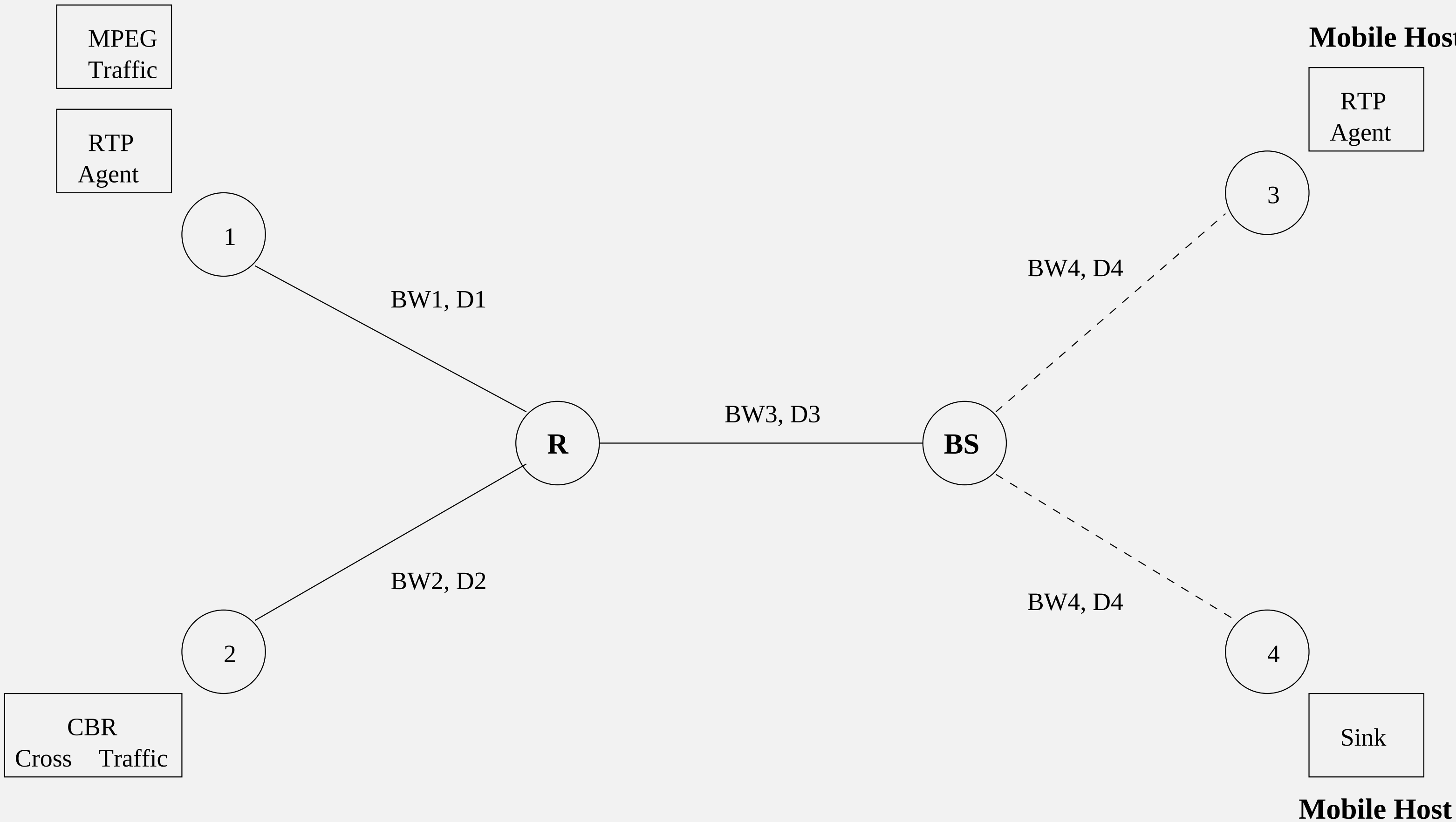
- Based on general patterns of throughput and response time as a function of load
- Besides loss rate, sender reports correlation between the packet loss and delay curve



- During congestion, delay curve increases with loss curve hence will have positive correlation
- If loss rate high, sender decreases rate only if correlation is positive

Simulation Experiments

- Network Simulator
  - ns from UC Berkeley (version 2.1b8a)
- Simulation Model



- **Experiment 1**
  - Network set in an uncongested state so that only wireless losses occur
  - Simulation parameters

$$\begin{aligned} BW_1 = BW_2 = 1Mbps \quad &, \quad D_1 = D_2 = 2ms \\ BW_3 = 256kbps \quad &, \quad D_3 = 10ms \\ BW_4 = 64kbps \quad &, \quad D_4 = 1ms \end{aligned}$$

- **Experiment 2**
  - Network set in a congested state using cross traffic generated from *Traffic/Expo*
  - Simulation parameters

$$\begin{aligned} BW_1 = BW_2 = 128kbps \quad &, \quad D_1 = D_2 = 2ms \\ BW_3 = 80kbps \quad &, \quad D_3 = 10ms \\ BW_4 = 64kbps \quad &, \quad D_4 = 1ms \end{aligned}$$

Results

Experiment 1

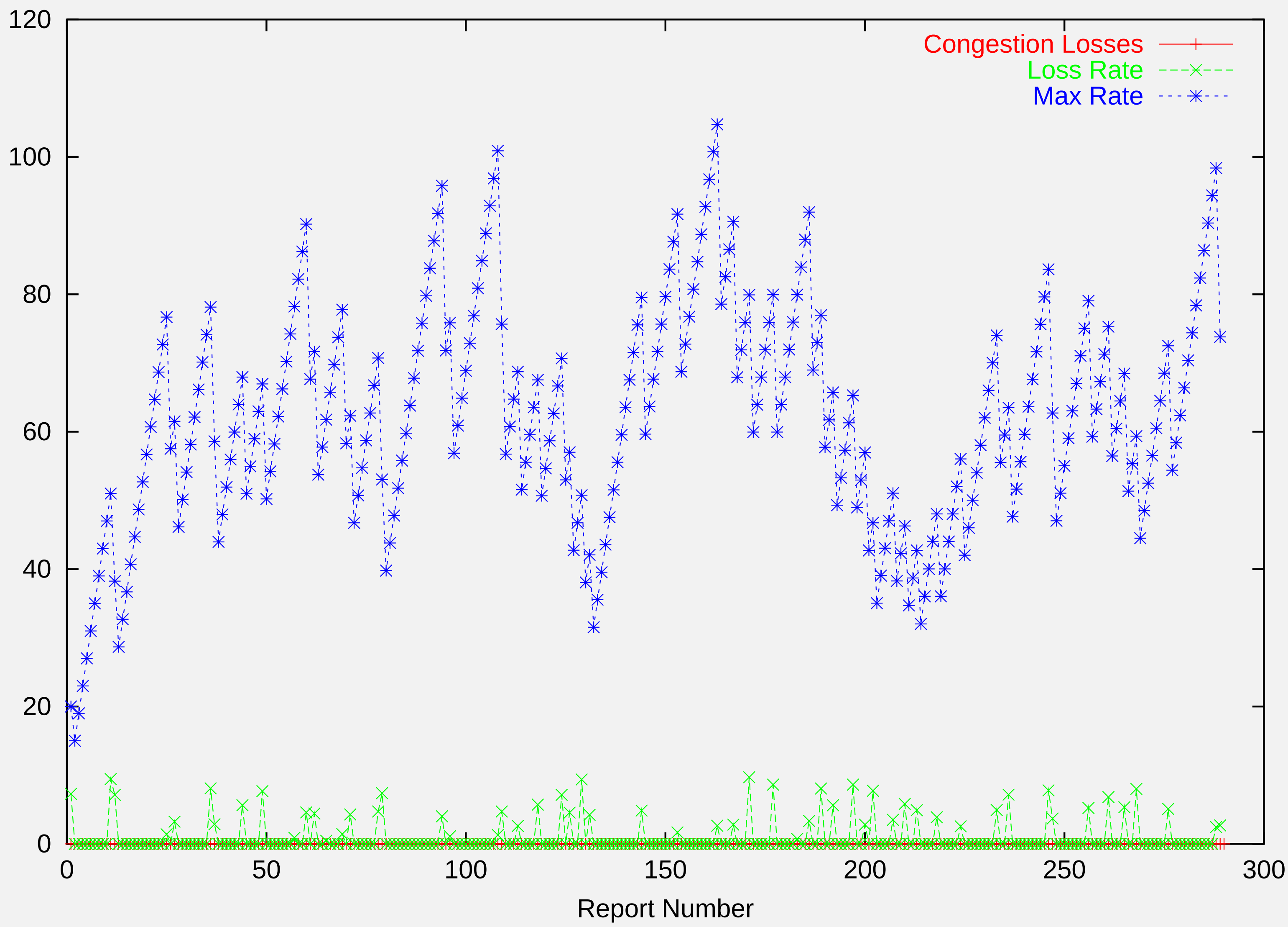


Figure 1: Original scheme without proposed modification

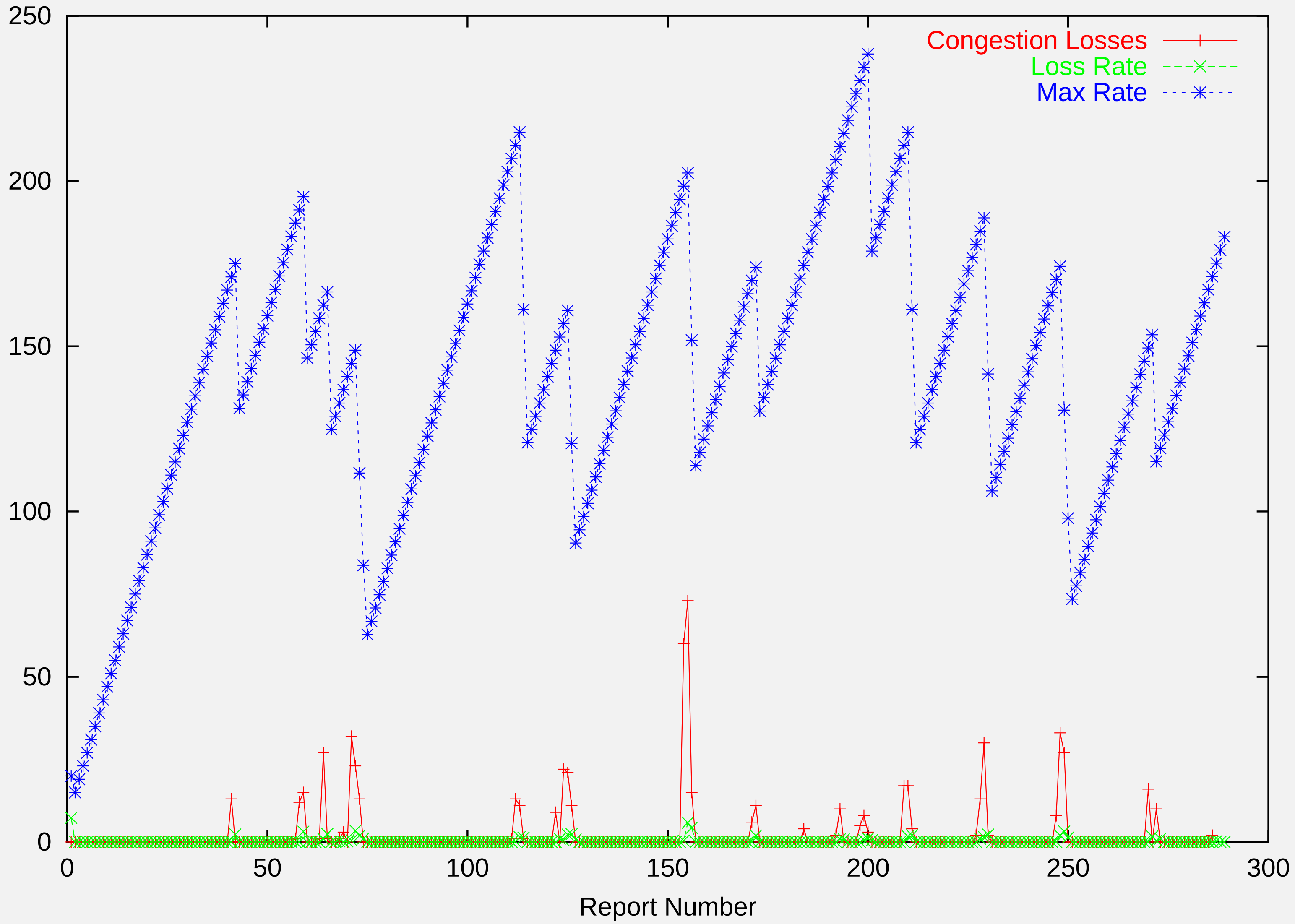


Figure 2: “Report Only Congestion Losses (ROCL)” Scheme

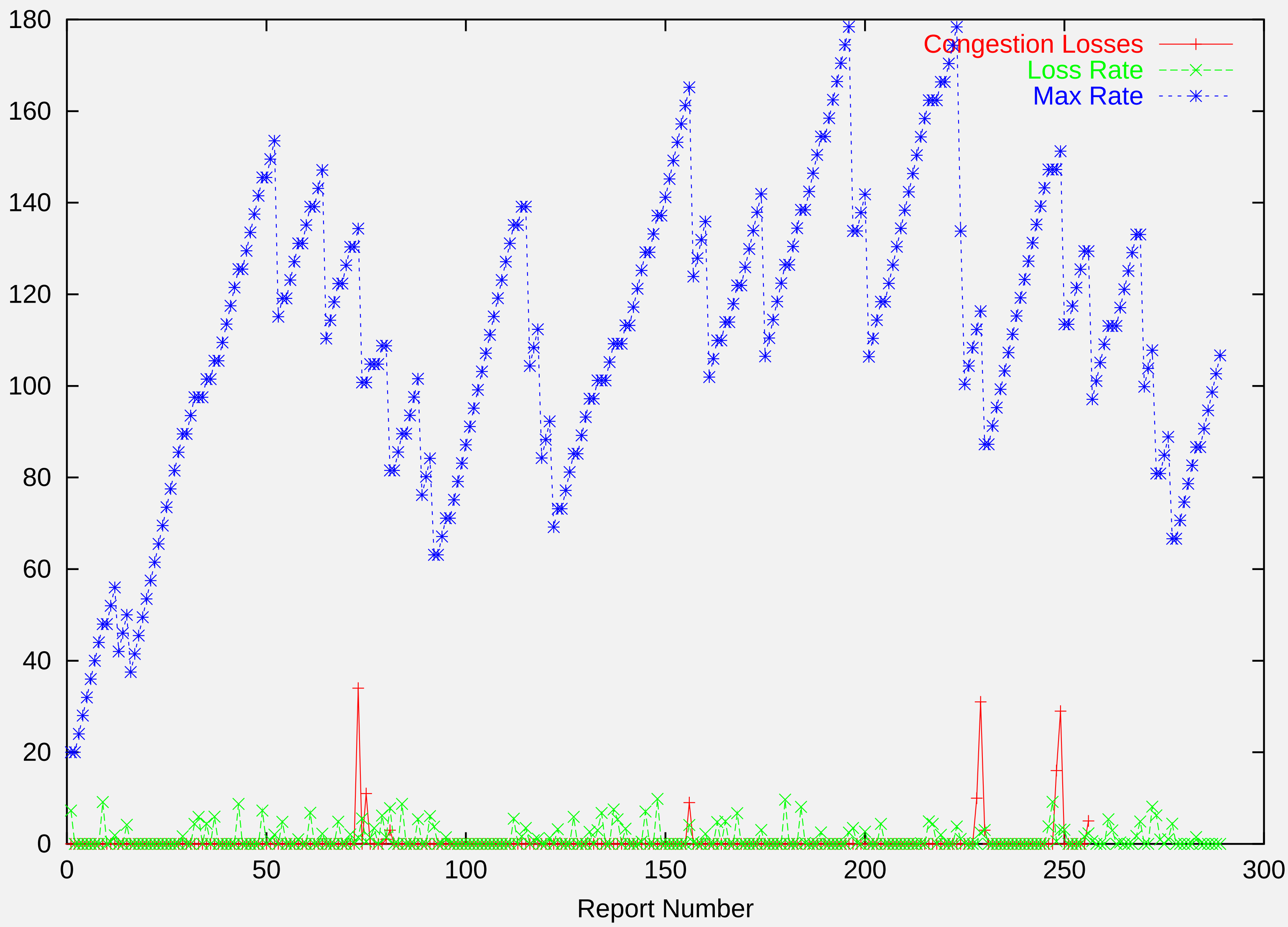


Figure 3: “Report Correlation of Loss and Delay (RCLD)” Scheme

Results

Experiment 2

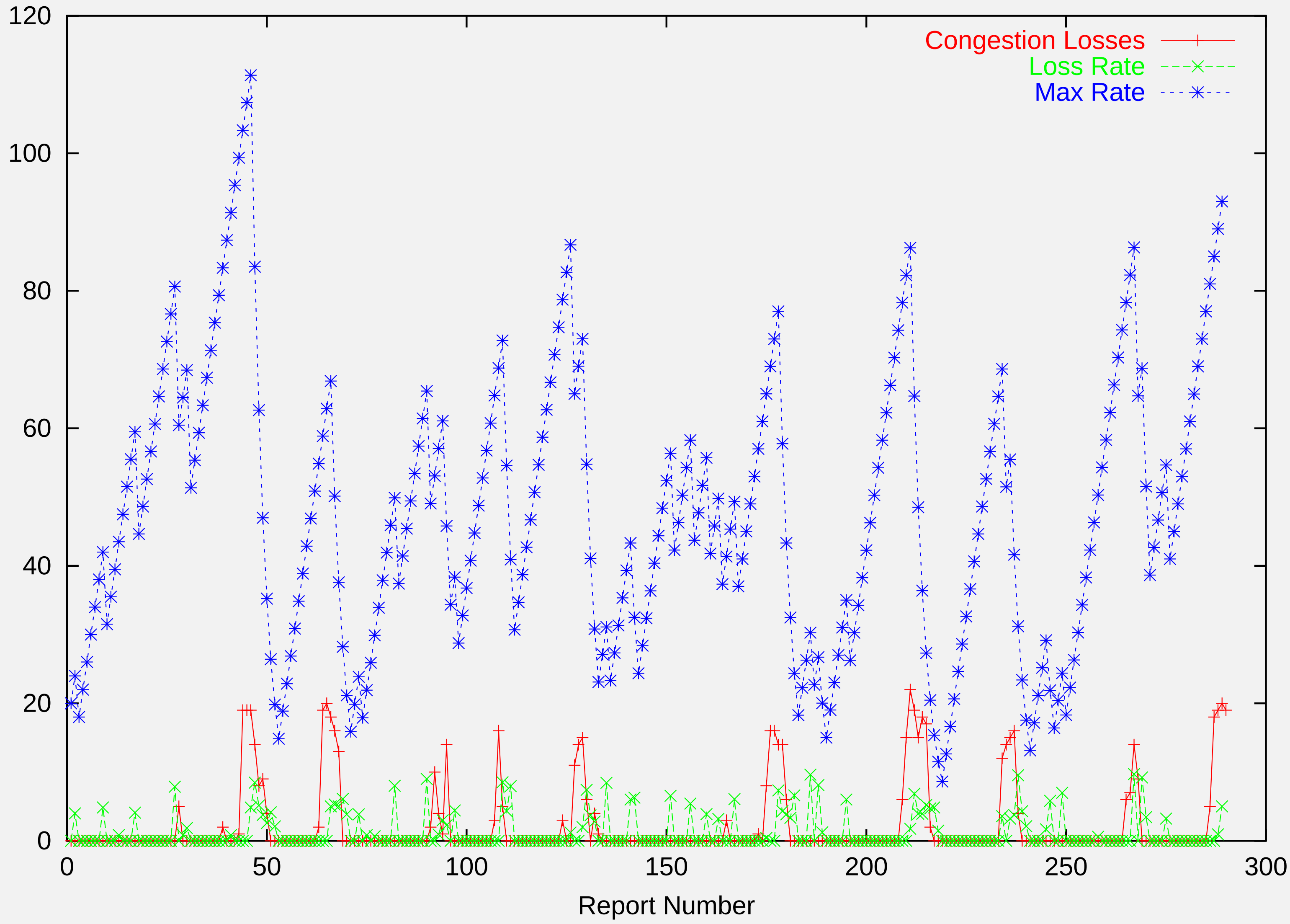


Figure 4: Original scheme without proposed modification

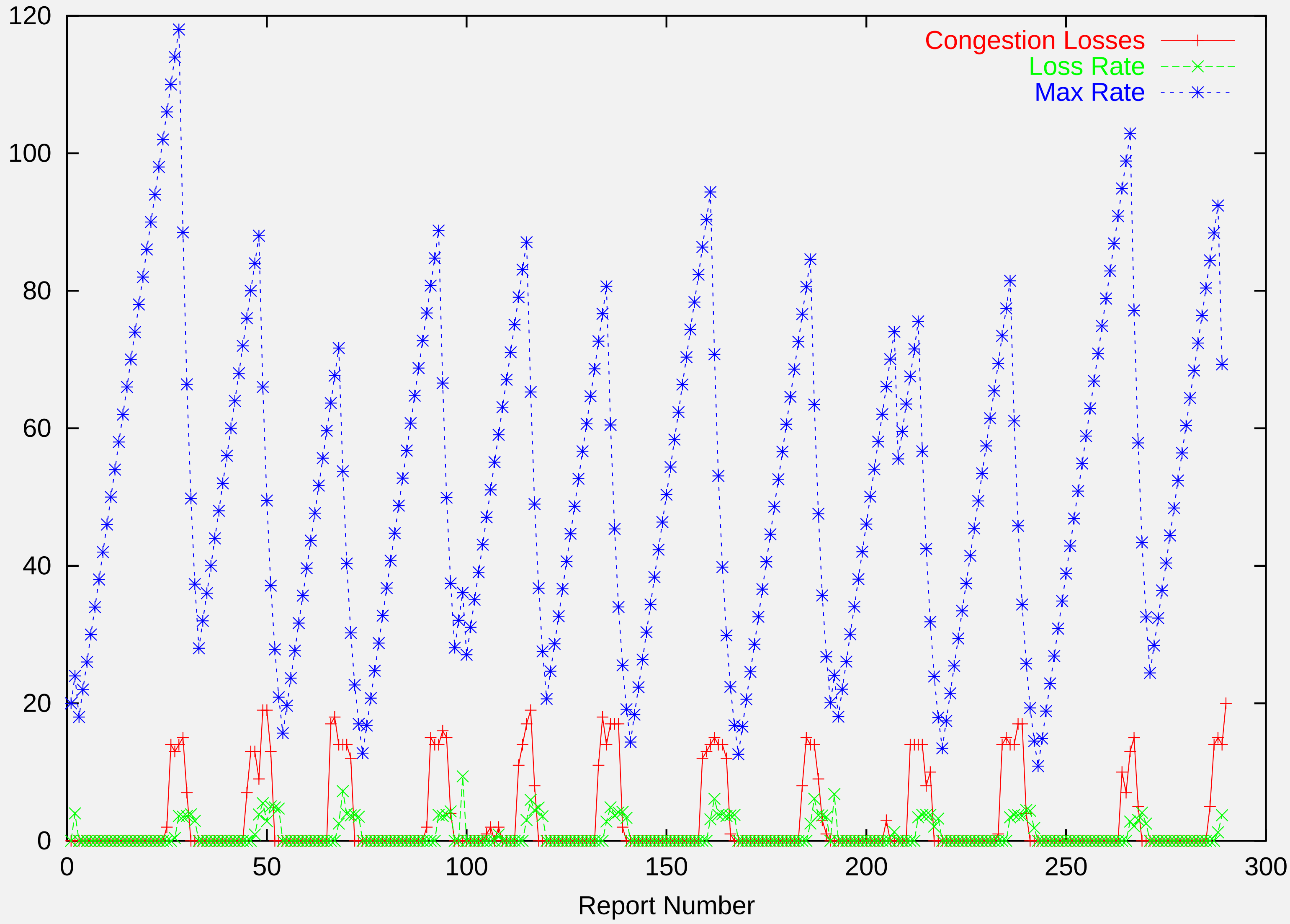


Figure 5: “Report Only Congestion Losses (ROCL)” Scheme

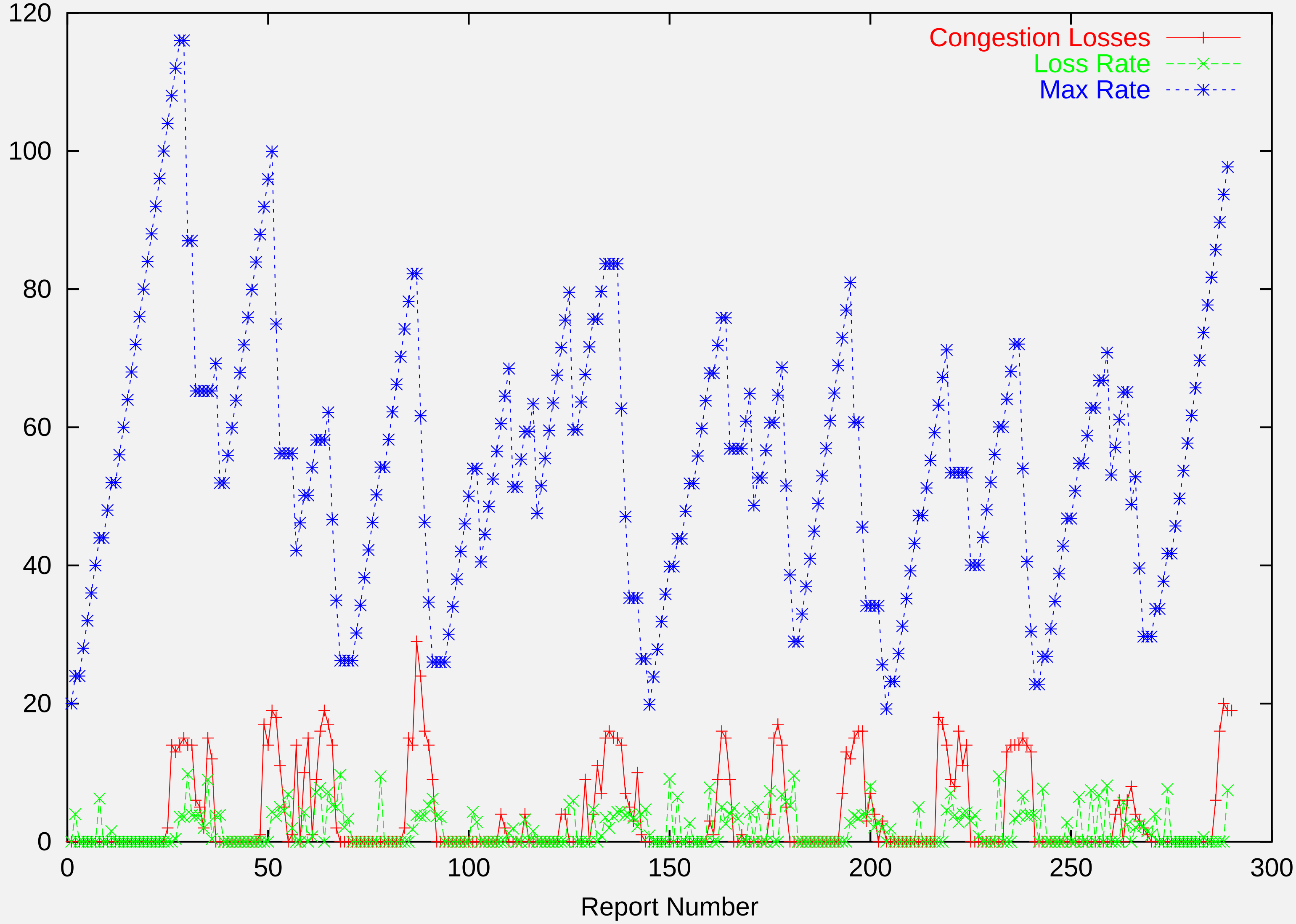
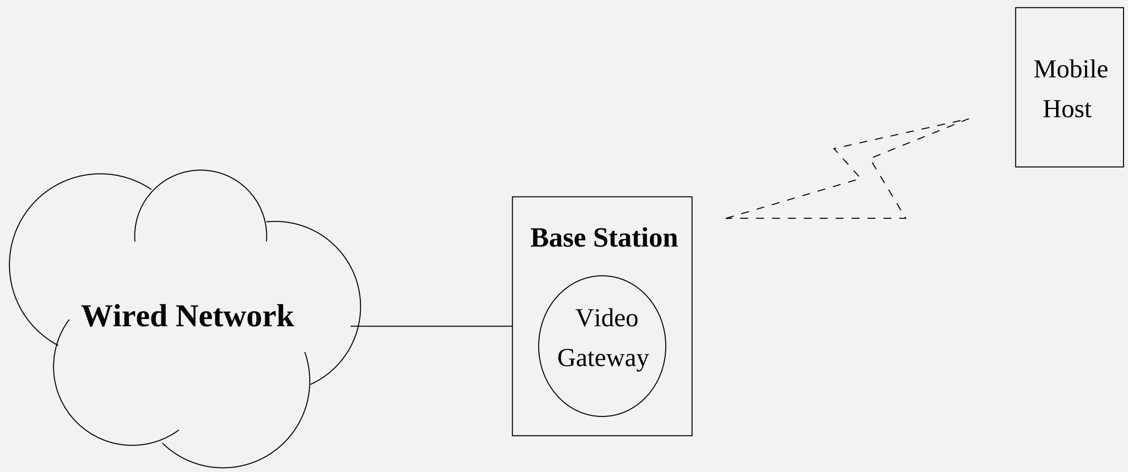


Figure 6: “Report Correlation of Loss and Delay (RCLD)” Scheme

Related Work

- Elan Amir et al proposed "Application Level Video Gateway"
- Employs split-connection approach
- Transcodes video stream from server to lower bandwidth



- Problems
  - Increase in end-end delay due to transcoding
  - Transcoding difficult when packets are encrypted

Conclusion & Future Work

- ROCL and RCLD try to decrease the output rate only in response to congestion
- Simulation experiments using ROCL and RCLD show significant increase in the output rate of video during bad channel conditions
- **Future Work**
  - Investigate appropriate functions to replace loss event rate p in model-based schemes
  - Maintain network state at the sender to aid in making adaptation decisions

References

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[2] S. Biaz and N. Vaidya. Discriminating congestion losses from wireless losses using inter-arrival times at the receiver. *IEEE Symposium ASSET'99, Richardson, TX, USA*, 1999.

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