Synchronized Multicast Media Streaming

APAN Workshop (Fukuoka) – Multicast BOF

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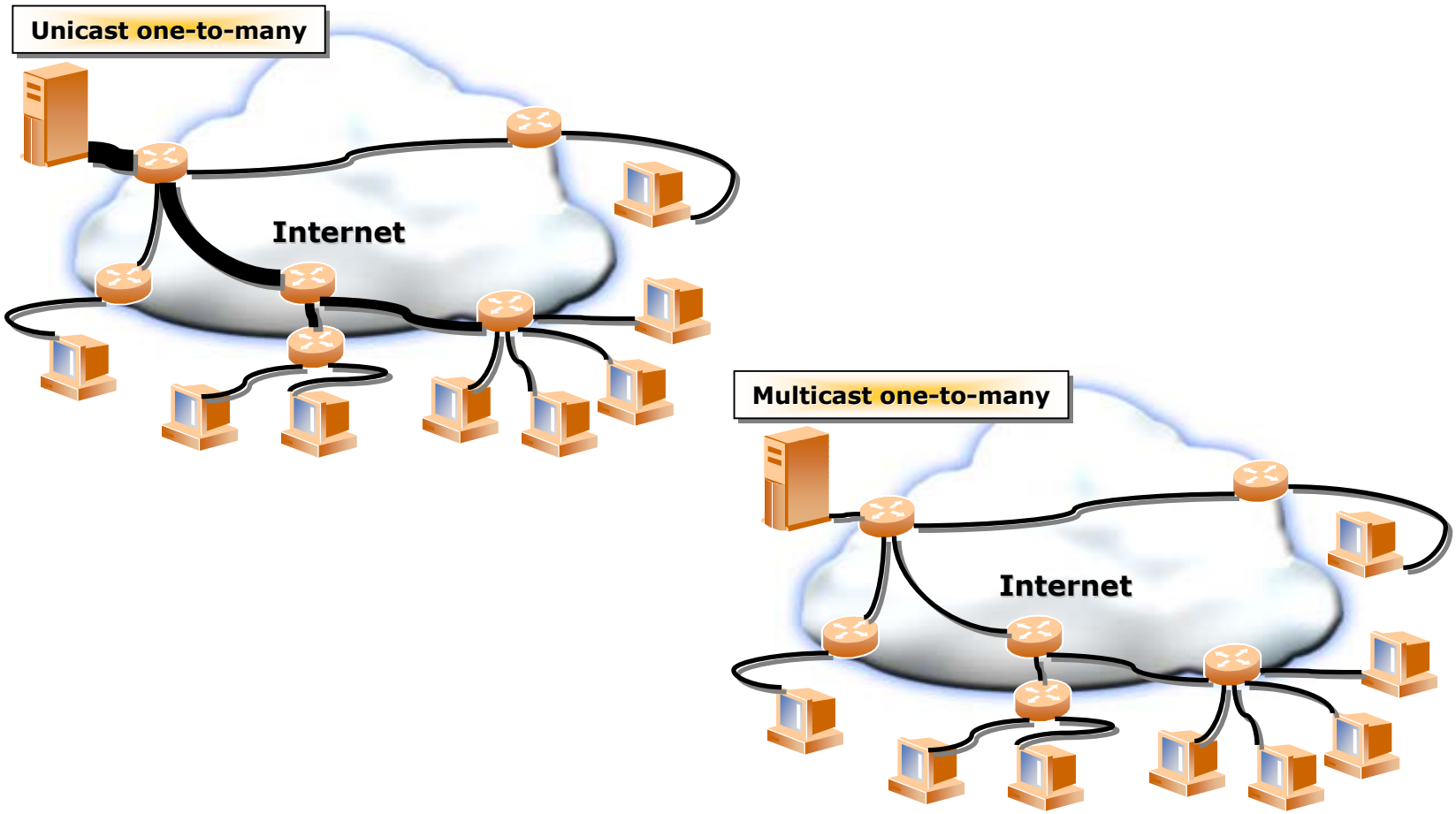
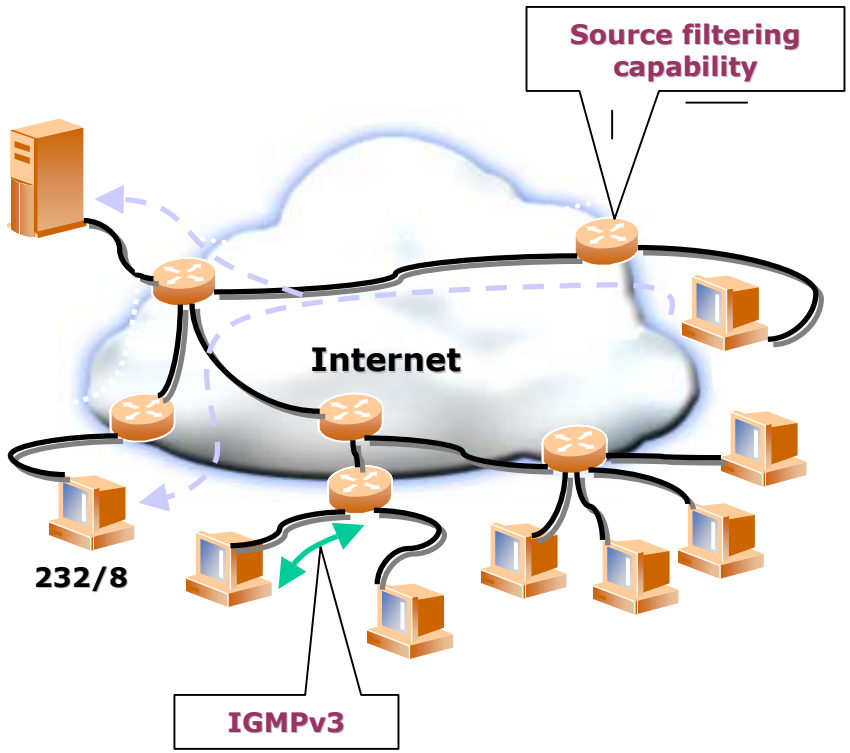
Unicast vs. Multicast One-to-Many



Source-specific Multicast

Source Specific Multicast



Source Specific Multicast

- One-to-many multicast services
- Source specific join (S, G)
- Require source filtering Functionality of IGMPv3
- IGMPv3 is currently under development

Address

Simplicity!

Feasible One-to-Many Multicast Service

End-to-end Quality of Service for Media Streaming

Media Compression

Maximize the compression Efficiency, the error Resiliency, and the delay tolerance.

Video CODEC:
JPEG, MPEG, H.263/261

Transport Protocol

Cope with bulky data and Real-time constraint.

Transport Protocol:
UDP, TCP, RTP/RTCP, RTSP

Quality of Service

Media Synchronization

Reconstruct temporal relationship (timing dependency) between media objects.

Synchronization:
Intra-media, Inter-media, Inter-client, Inter-object

Reliability

Mandatory to guarantee QoS.

Error Control:
ARQ, FEC, Hybrid ARQ/FEC
Congestion Control:
TCP-friendly Rate Control

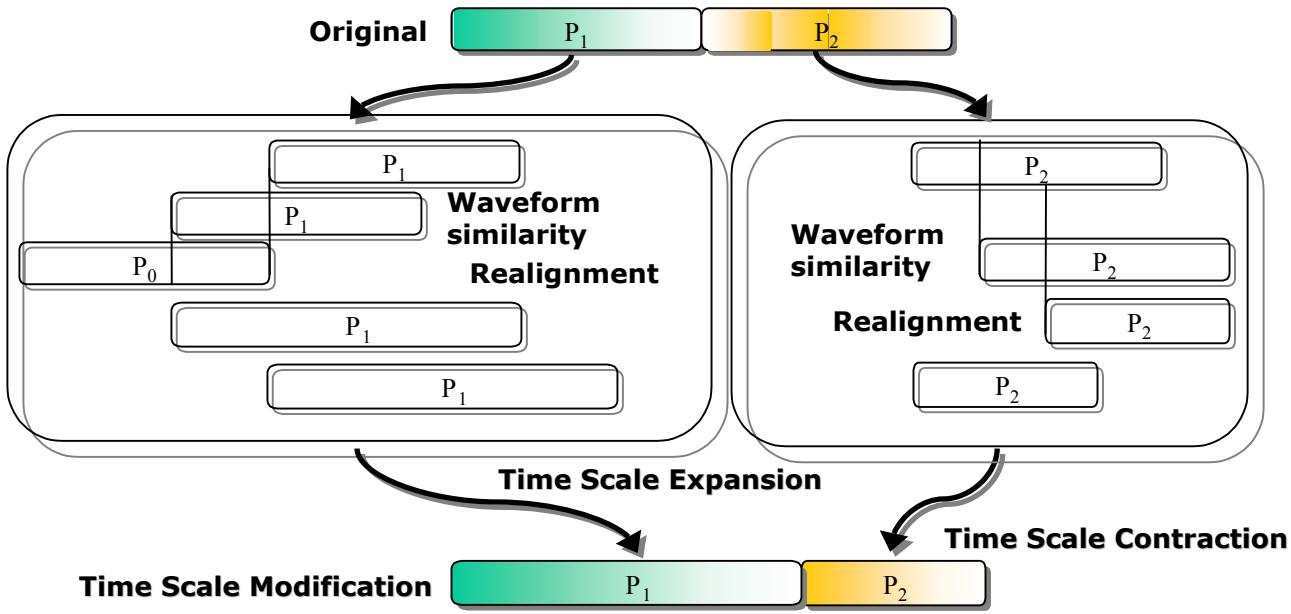
Application Adaptation

Adjust required degree of service quality to network fluctuation and system dynamic.

Application Adaptation:
Delay, Delay jitter, Loss concealment using Adaptive Playout Control

Adaptive Playout

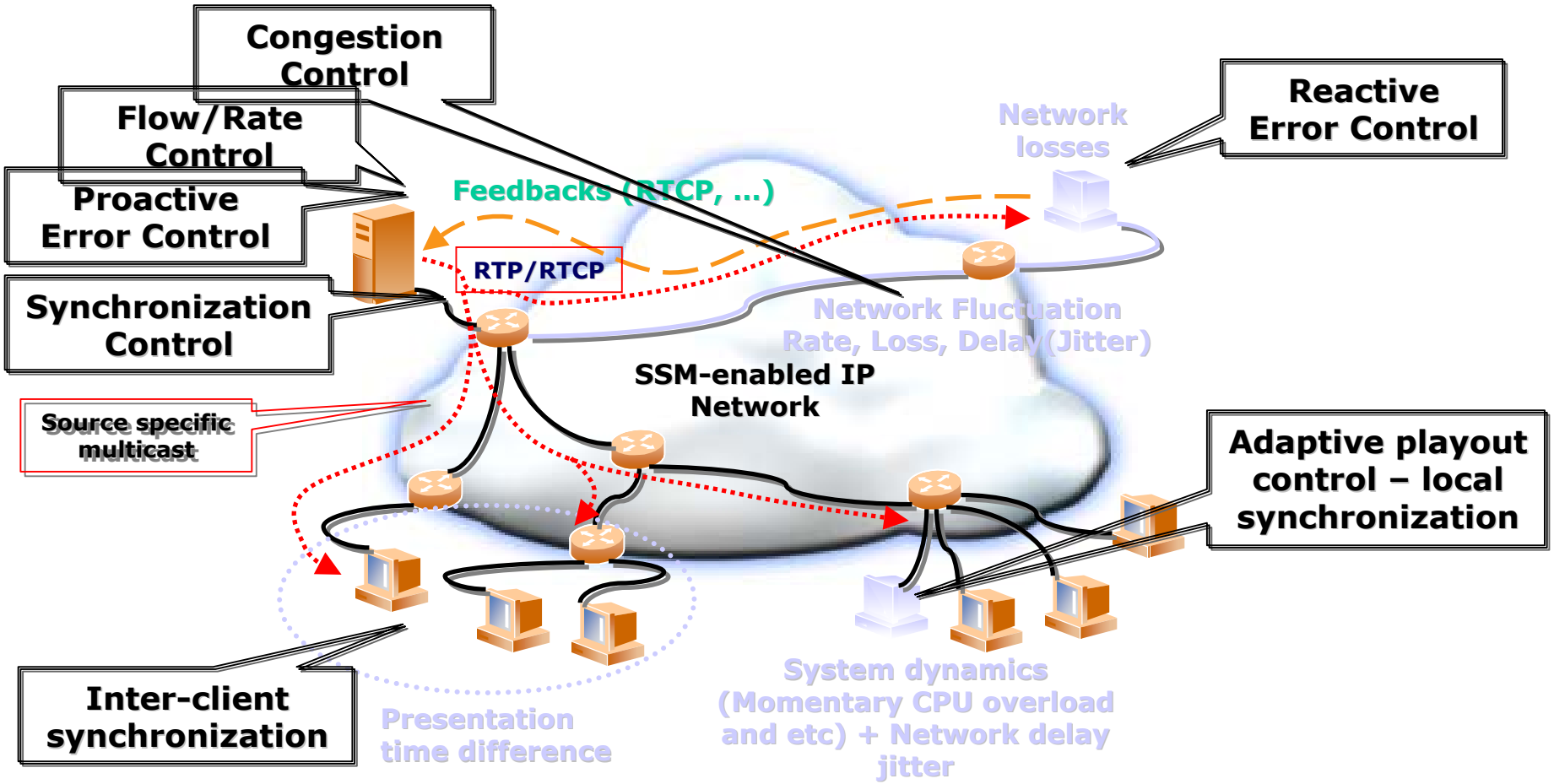
Waveform Similar and Add (WSOLA)[Verhelst93]



Samples)

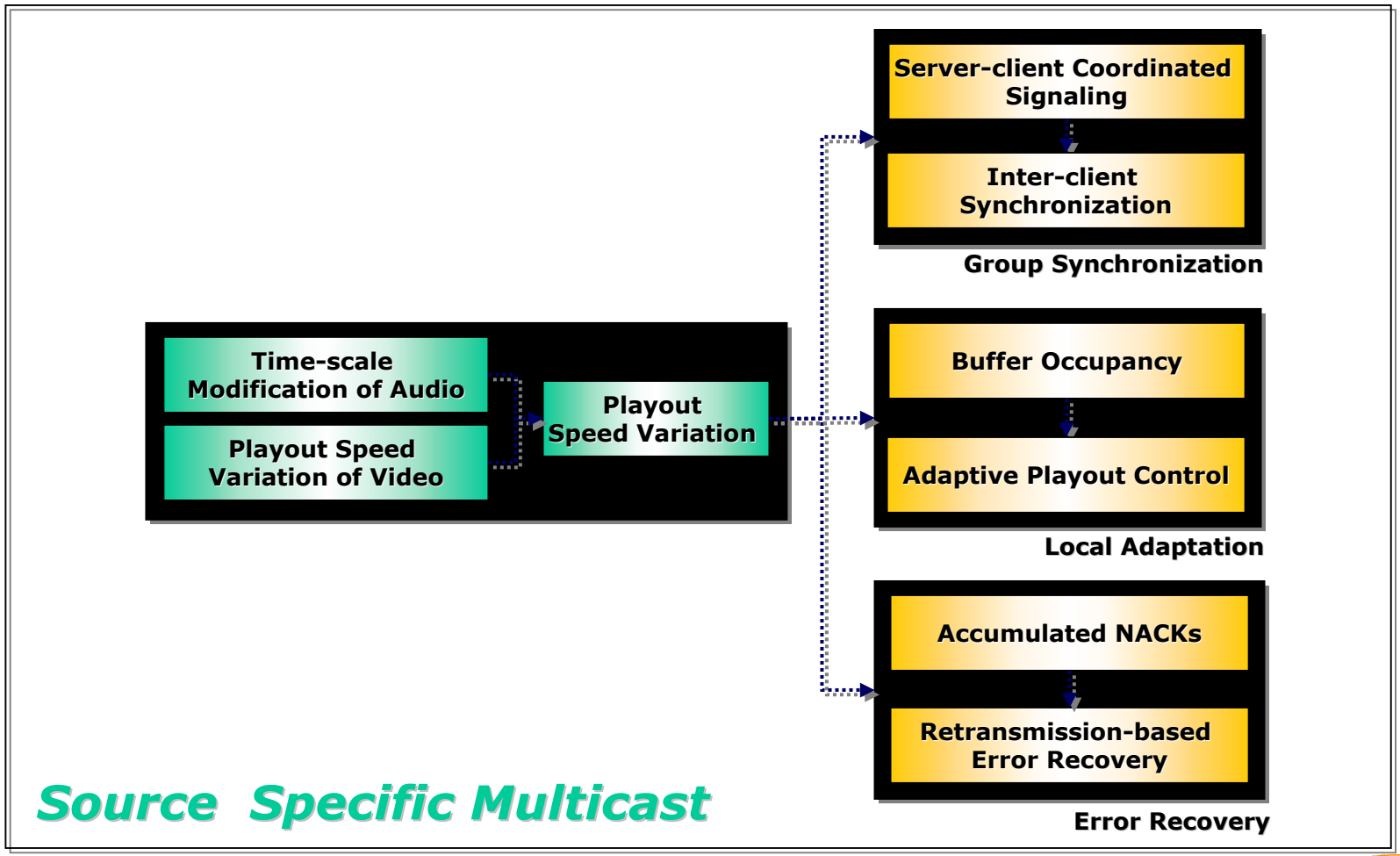
Original		Expansion	Contraction
	10%		
	20%		

Problem Scope & Approach



Server-client coordination of network adaptation tools (e.g., adaptive playout and error control) for enhanced streaming

Proposed Approach - How to



Related Works

Source specific multicast

[Holbrook01], [Chesterfield02]
: SSM, Unicast RTCP Feedback.

Time-scale modification

[Steinbach01],
: delay, loss con

Unicast
Environment,
Loss and delay

TCP's
Window-based,
Layered approach

cumulated NACK

[Poo01]
Cumulative NACK over
layered multicast

Buffer-based adaptive player

[Walpole97], [Rother
: Shed system dy
adaptive str
synchronizatior

System dynamics only,
Lack of
specific mechanisms

Inter-client synchronization

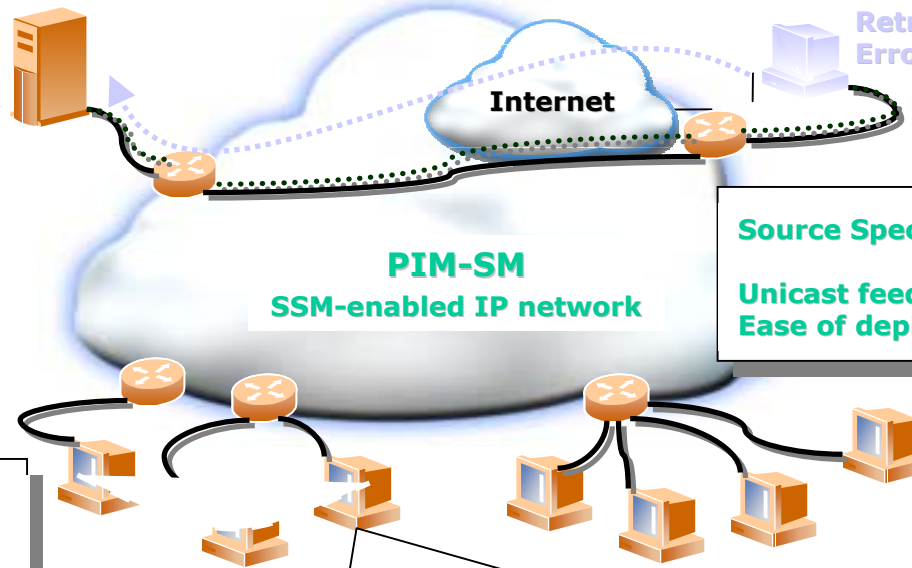
[Ishibashi01]
back-based Loosely coupled
roup synchronization

Group
Synchronization only

Proposed Approach - Framework

Roles of Server

Group coordination
Feedback suppression
Congestion control

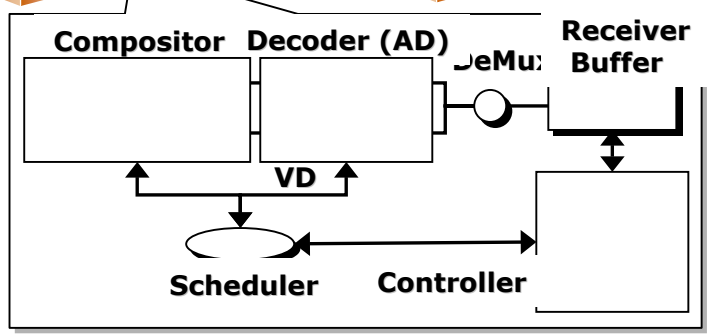


Source Specific Multicast

Unicast feedback
Ease of deployment

Inter-client Synchronization

Fair services
Mitigates clients heterogeneity
Assist error recovery

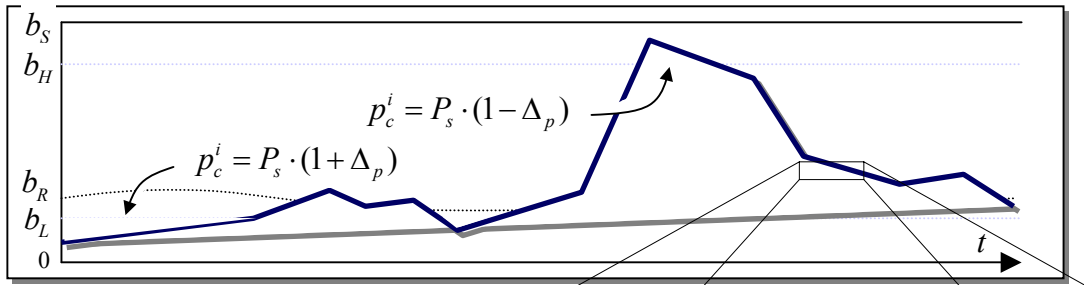


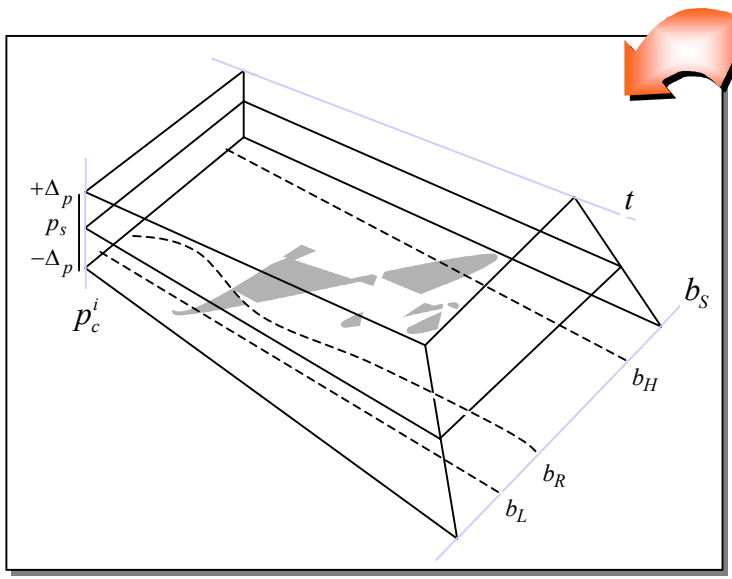
Adaptive Playout Control

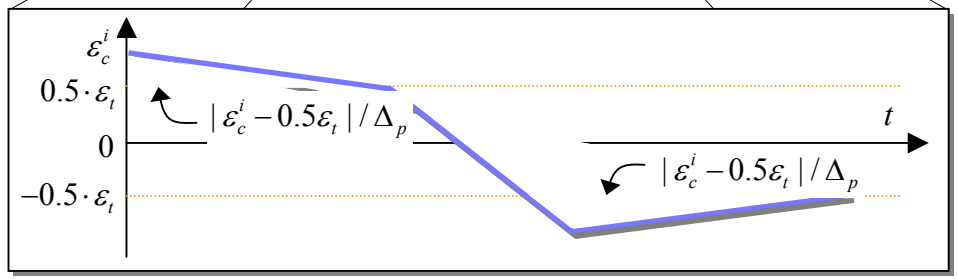
Reduce discontinuity
Loss concealment

Target of Adaptive Playout Control

Local Adaptation





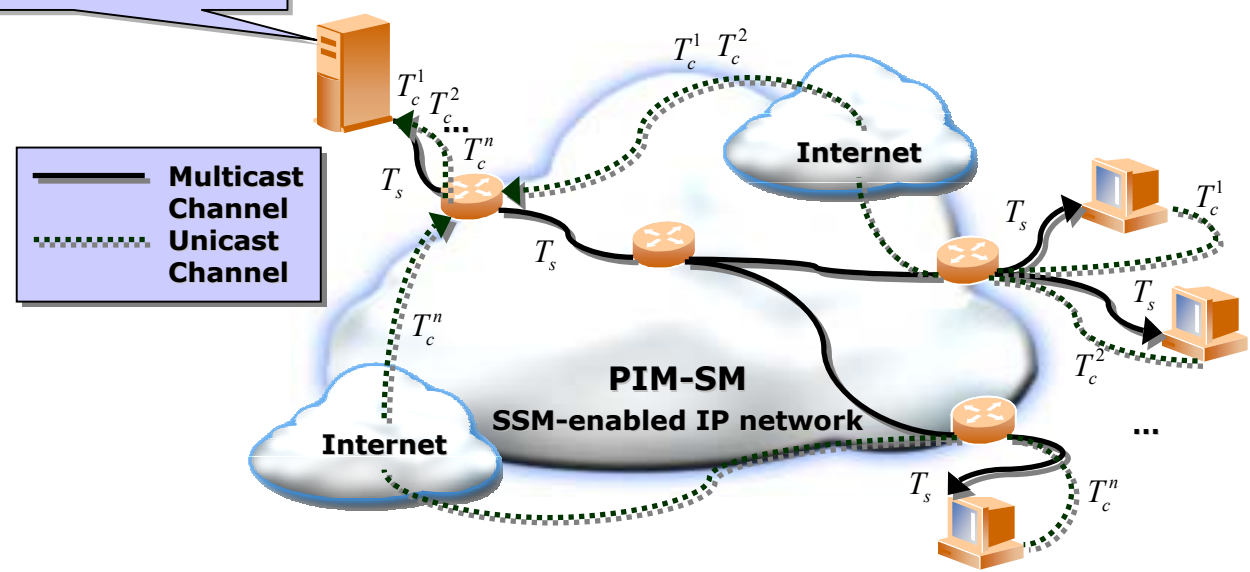


Server-aid Inter-client Synchronization

Server-client signaling for inter-client synchronization

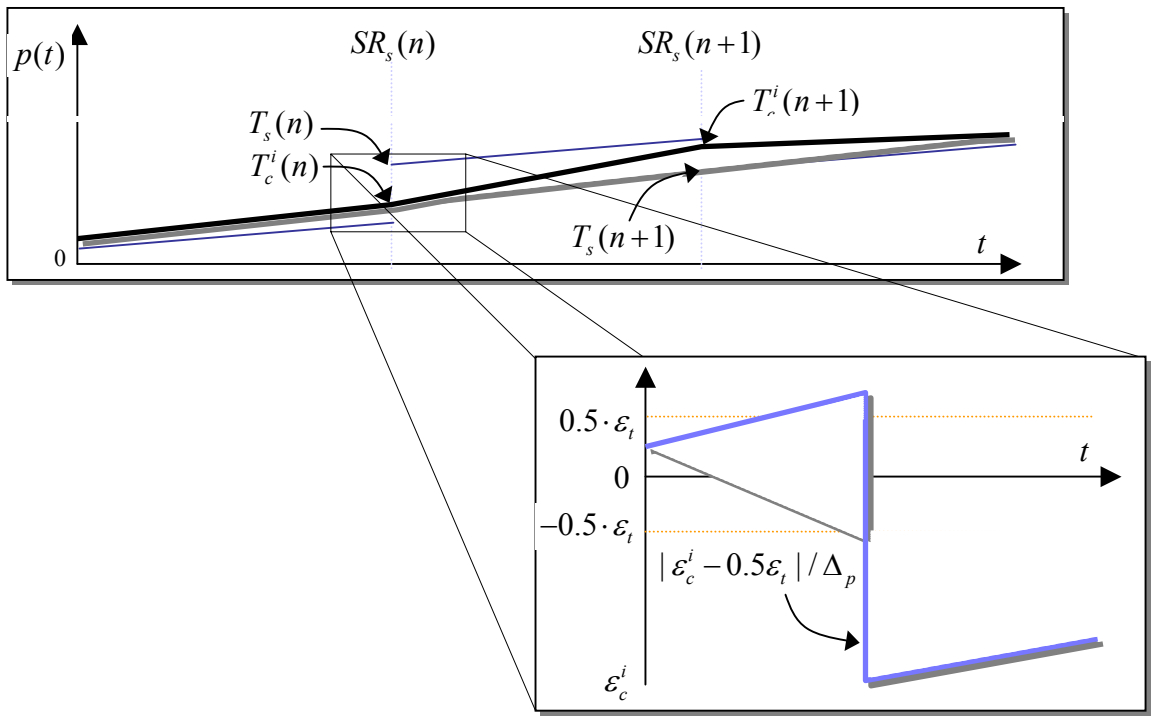
Arbitration Policy

$$T_s = F_{arbitration}\{T_c^i, d_s^i, RTT^i \mid i = 1, \ldots, N_c\}$$



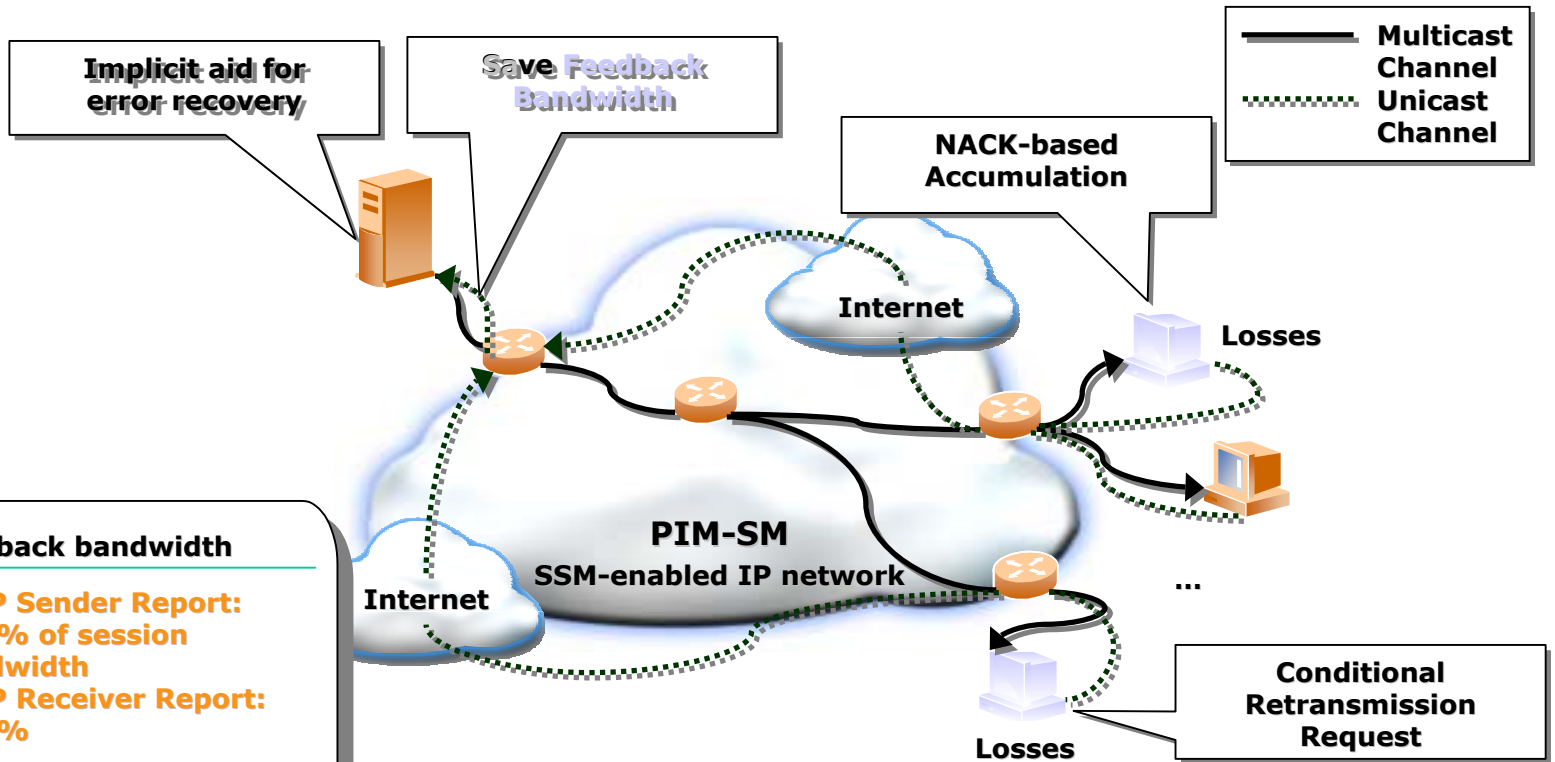
Server-aid Inter-client Synchronization

Main mechanism for Inter-client synchronization



Error Recovery

Accumulated NACK-based Error Recovery



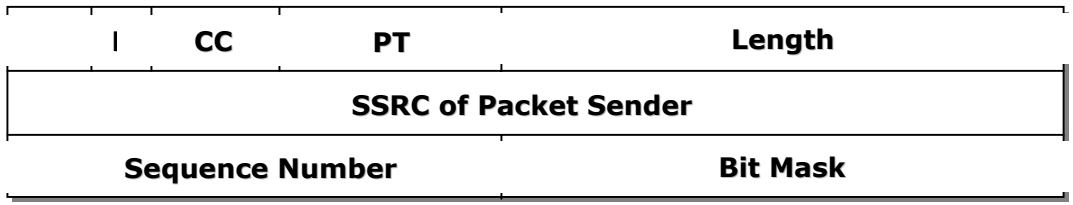
Feedback bandwidth

RTCP Sender Report: 1.25% of session Bandwidth
RTCP Receiver Report: 3.75%

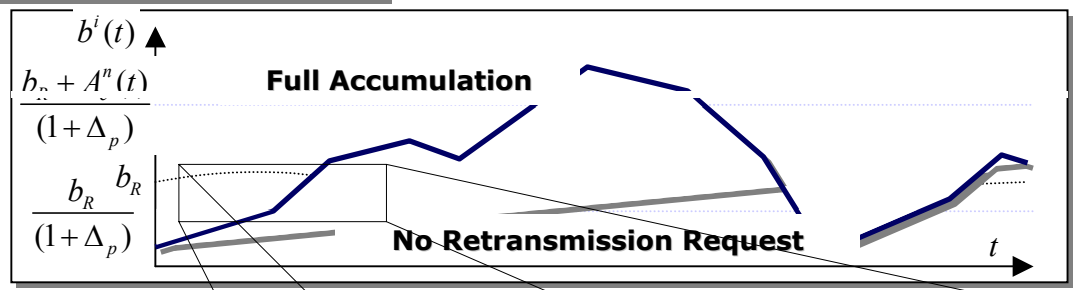
5% of session bandwidth can be used for RTCP signaling.

Error Recovery

Packet format of accumulated NACK



Main mechanism for accumulated NACK

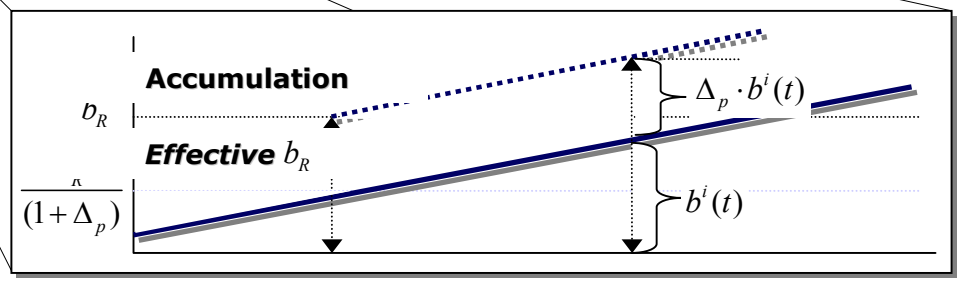


Accumulation time

$$\min[(1+\Delta_p)\cdot b^i(t) - b_R, A_c^n(t)]$$

$$where, \quad \frac{b_R}{(1+\Delta_p)} \le b^i(t) \le b_R + A_c^n(t)$$

The player should slow down Its playout speed for A_c/Δ_p ***ms.***



Playout coordination

Coordination

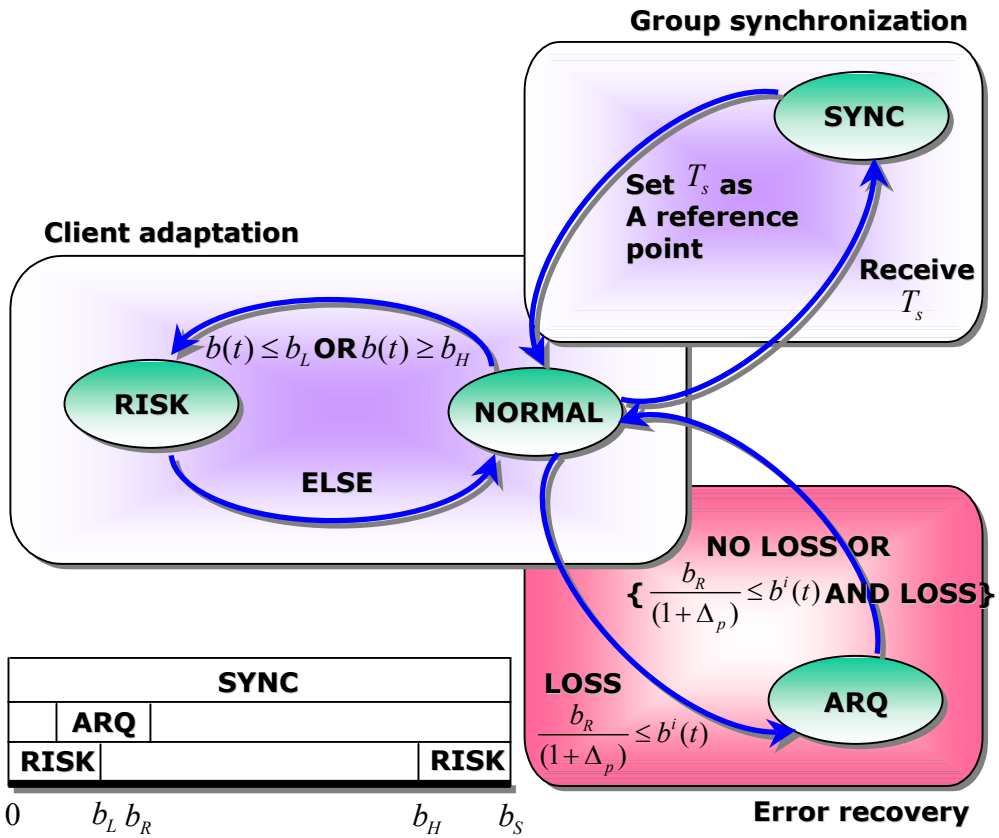
Four States

NORMAL

RISK
High risk of buffer underflow or overflow

ARQ
The aid of adaptive playout control is required to request retransmission

SYNC
Clients receive a synchronization message



Simulation Model

Simulation Topology and Model

Source-specific Multicast (SSM) over Protocol Independent Multicast – Sparse Mode (PIM-SM)

One Multicast streaming server and 16 session participants

Use modified RTP/RTCP over UDP (Unicast RTCP feedback)

Traffic Source: MPEG-II 5Mbps with 30 f/s.

Group of Picture (GOP) with 15 I-frame and 3 P-frame distance

Join interval : 11 s (fixed)

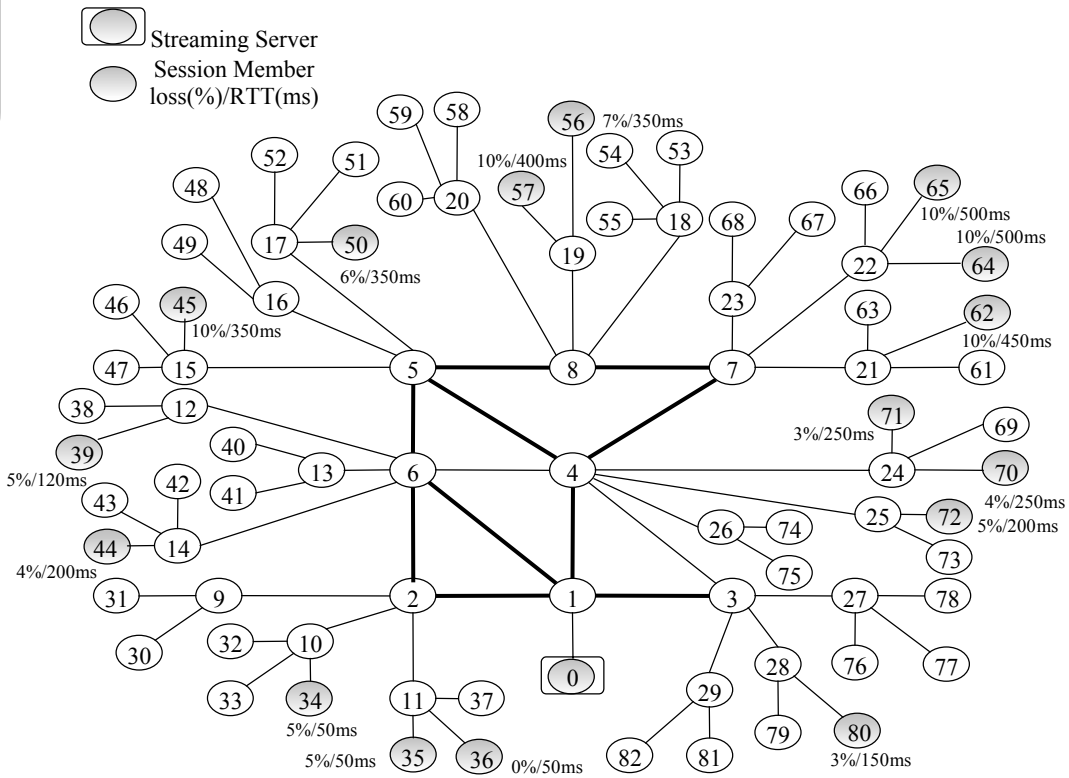
Momentary CPU overload Interval : 10 s (exponential di.) Strength : less than 50 ms (uniform di.)

Playback factor : 0.25
: 100 ms
ε_t rver, client buffer size : 1s

$$b_H : 0.9\ s,\ b_L = \frac{b_R}{(1+\Delta_p)}$$

Average loss rate	Minimum	Maximum
6 %	0 %	13 %

Assumption : Compound RTCP RR is 120 bytes and RTCP ARQ is 12 bytes

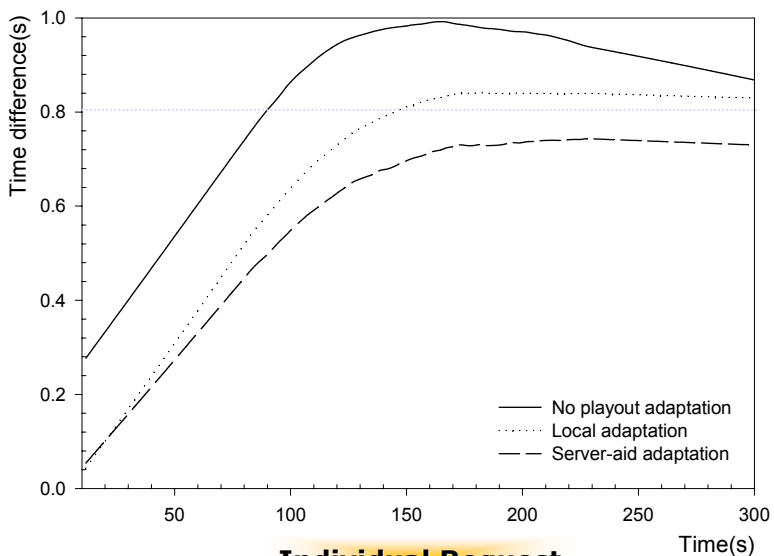


Simulation Results

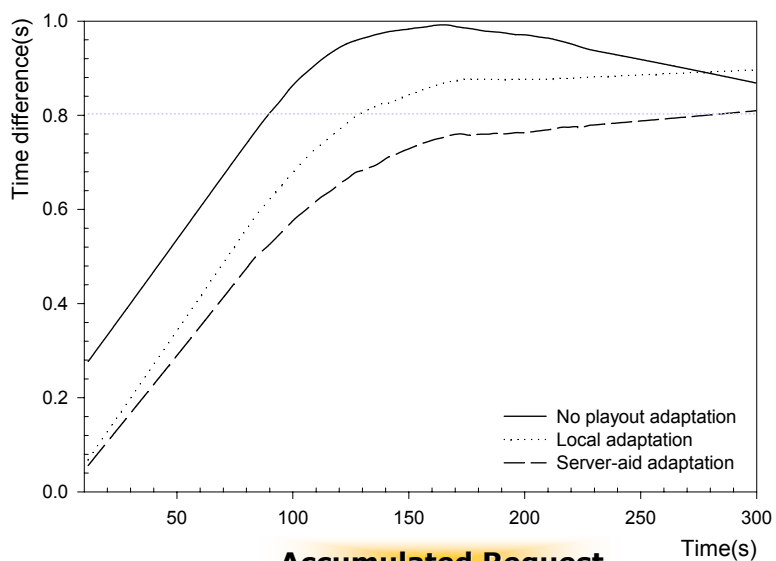
Three different playout cases

1. ***No playout adaptation:*** *No playout control, No server-aid, NACK-based individual retransmission request*
2. ***Local adaptation:*** *Playout control, No server-aid, NACK-based individual or accumulated retransmission request*
3. ***Server-aid adaptation:*** *Playout control, Server-aid, NACK-based individual or accumulated retransmission request*

Maximum playout time difference between leading and trailing clients



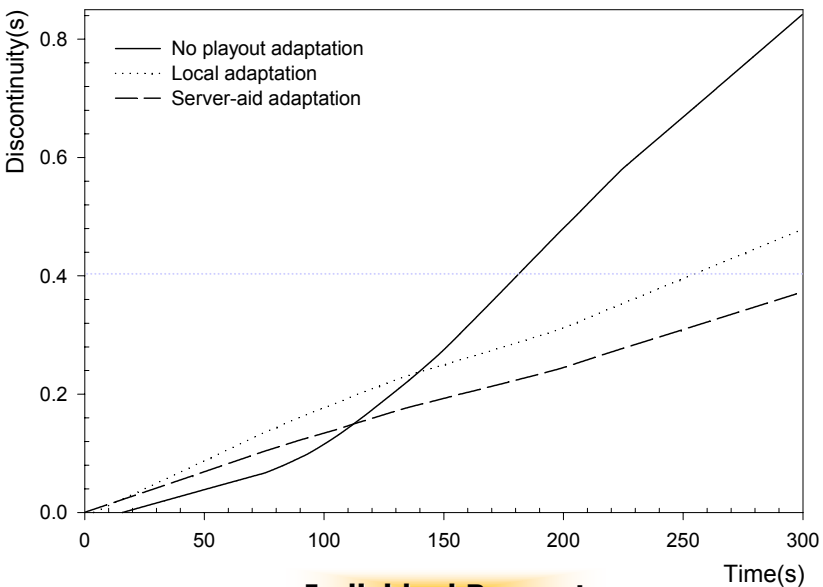
Individual Request



Accumulated Request

Simulation Results

Accumulated playout discontinuity averaged per each client



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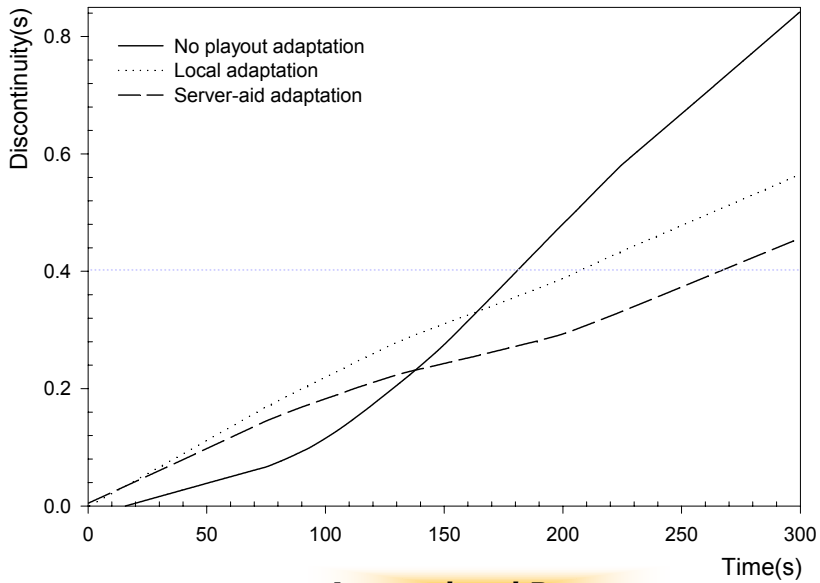
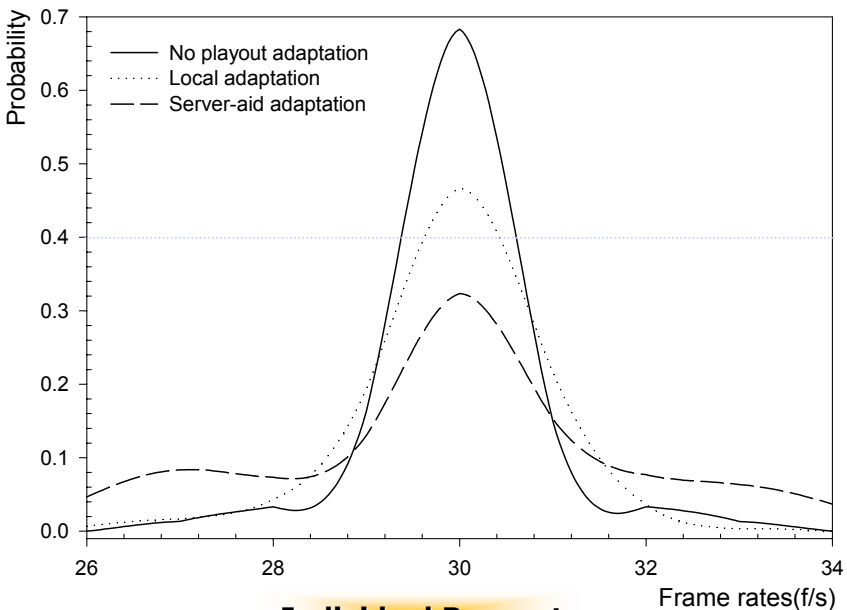


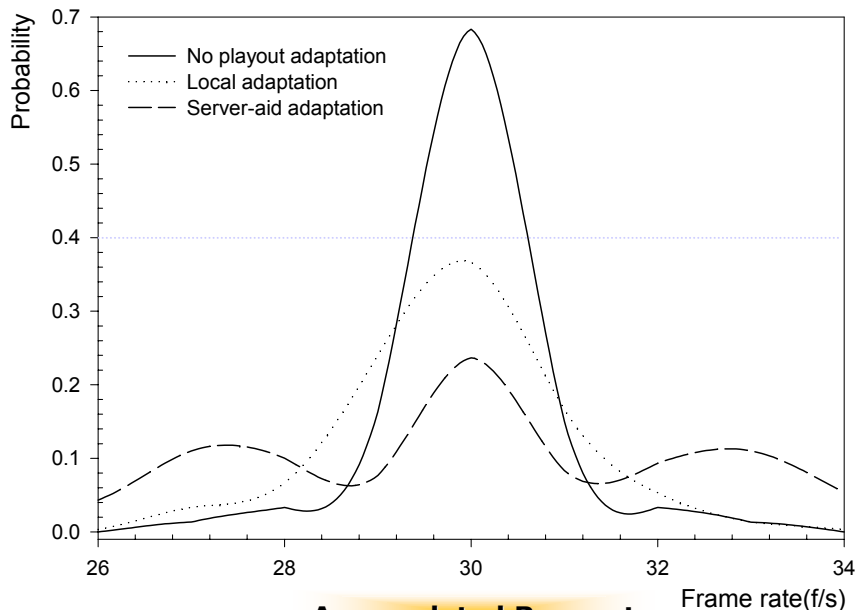
Accumulated Request

Simulation Results

Playout speed variation (a long-run client)



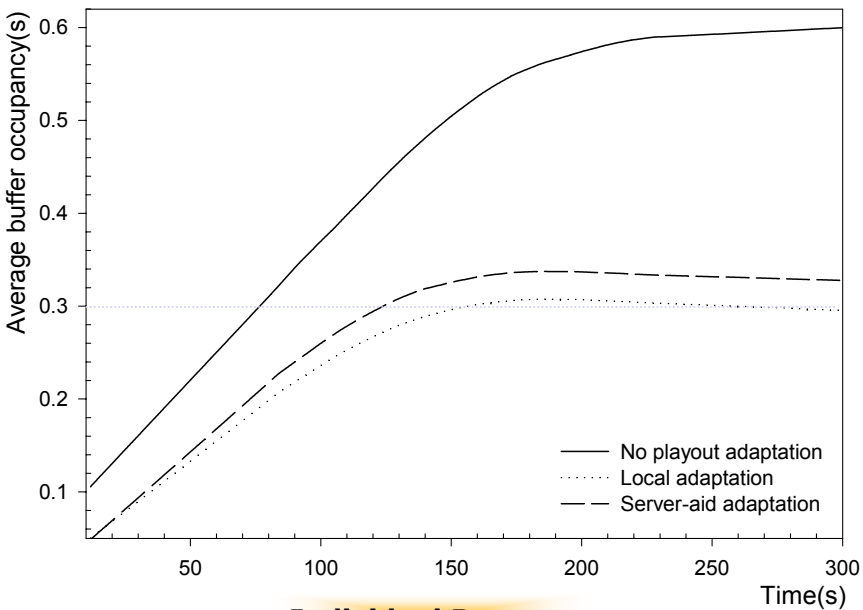
Individual Request



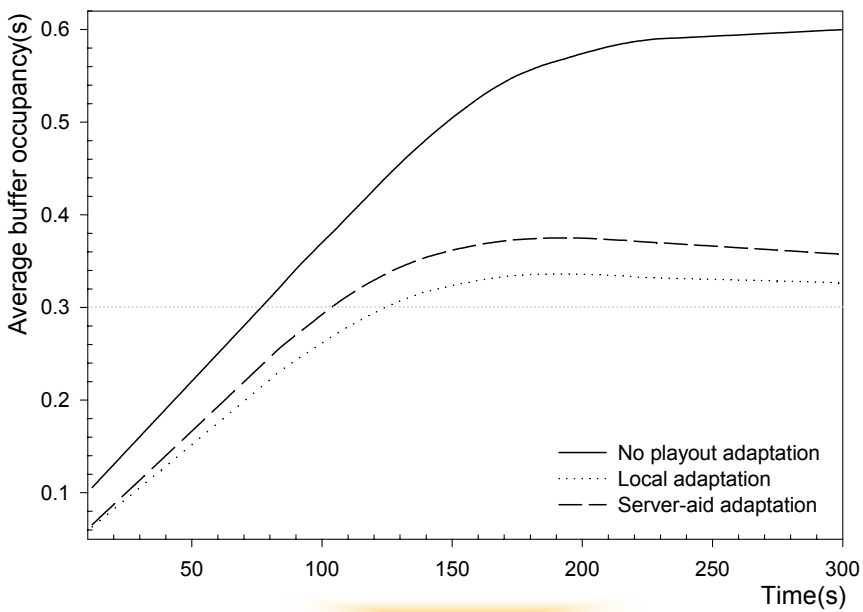
Accumulated Request

Simulation Results

Average buffer occupancy at time t in a group



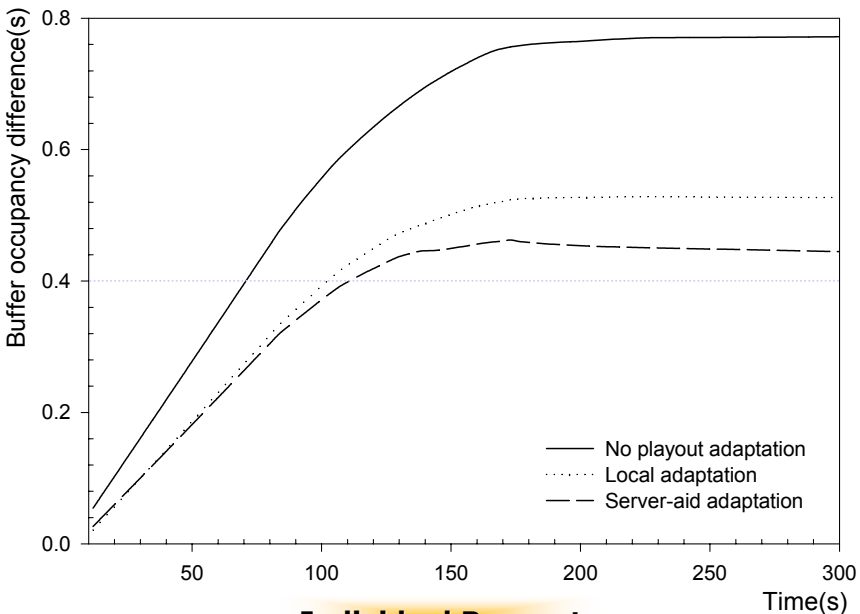
Individual Request



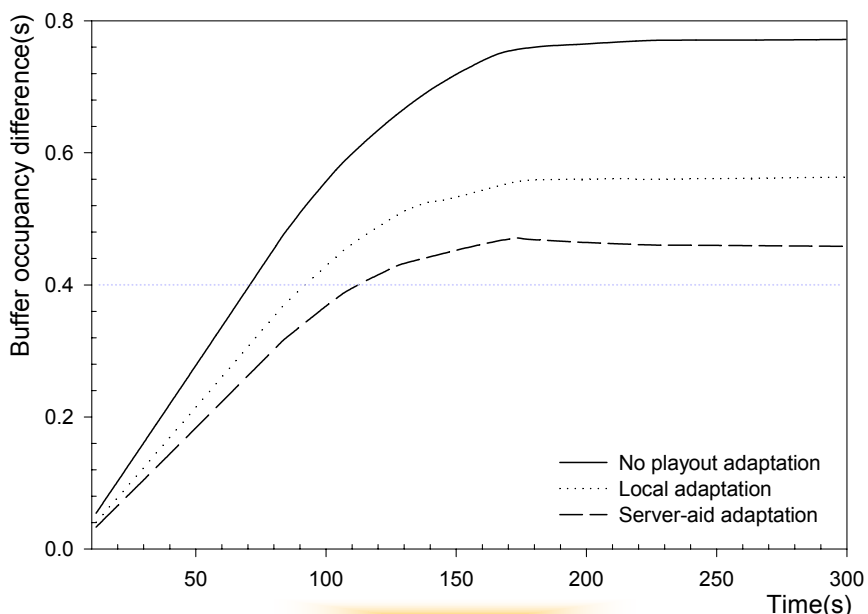
Accumulated Request

Simulation Results

Maximum buffer occupancy difference at time t in a group



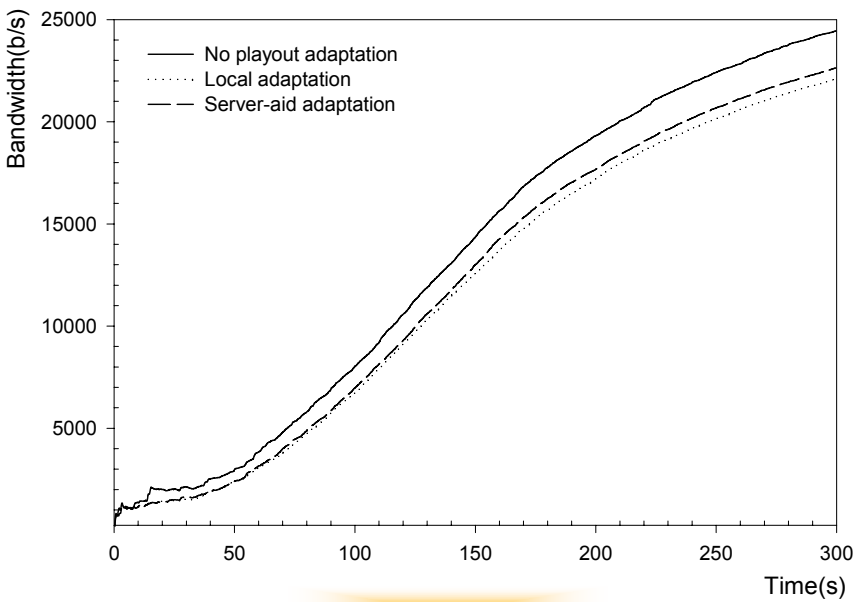
Individual Request



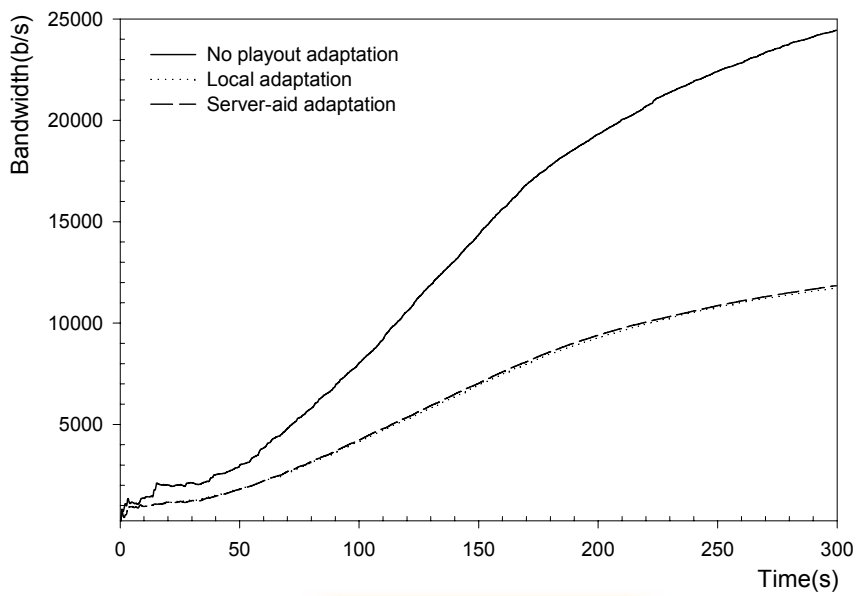
Accumulated Request

Simulation Results

Required feedback bandwidth for retransmission request and receiver report



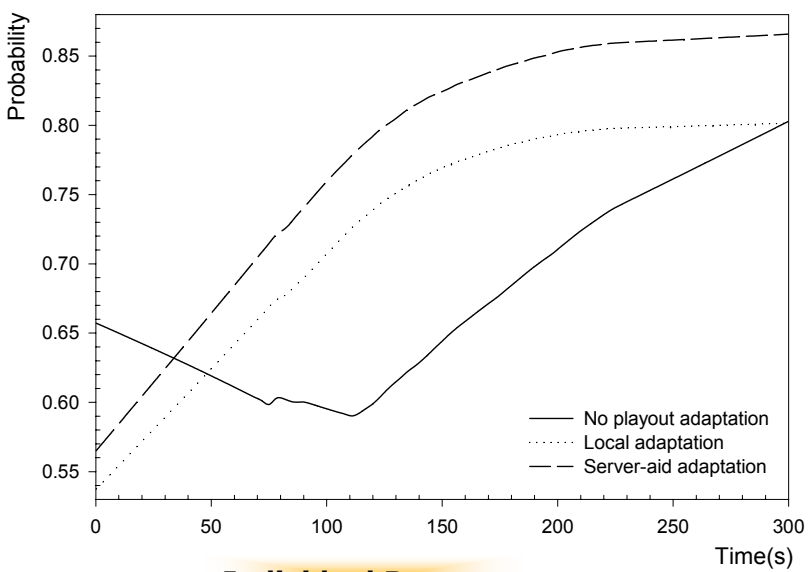
Individual Request



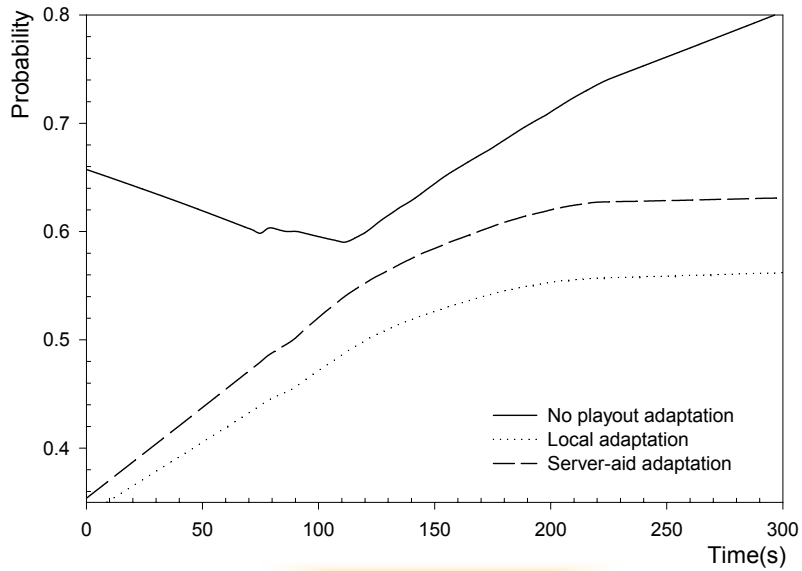
Accumulated Request

Simulation Results

Averaged lost packet repair probability



Individual Request



Accumulated Request

Impact of Ts arbitration policy on the performance

Ex) $T_s = median\ (T_c^1 + d_s^1 + RTT^1, \ldots, \ldots, T_c^{N_c} + d_s^{N_c} + RTT^{N_c})$

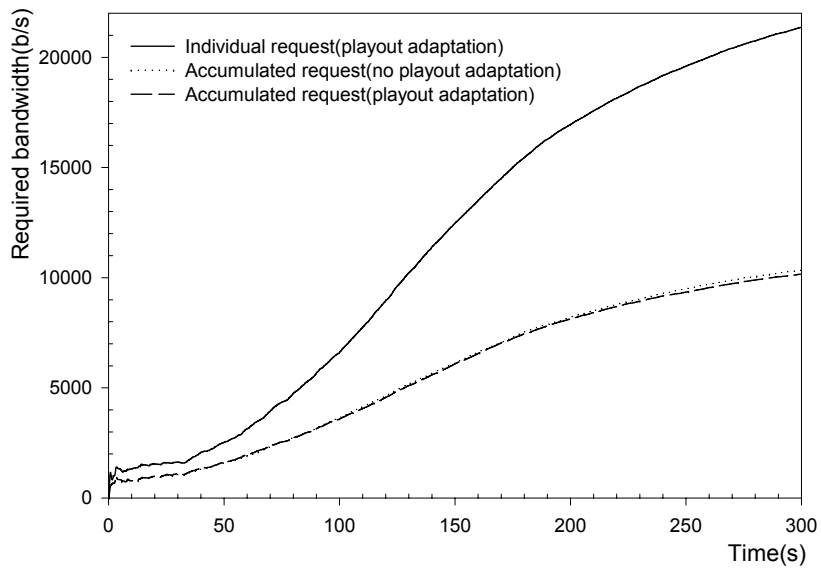
Ts Selection	Playout discontinuity(s)	Playout speed variation(f/s)
Minimum	0.3875	3.090097
Median	0.5291	3.014417
Maximum	0.5479	2.306243
Average	0.4916	2.922741

Simulation Results

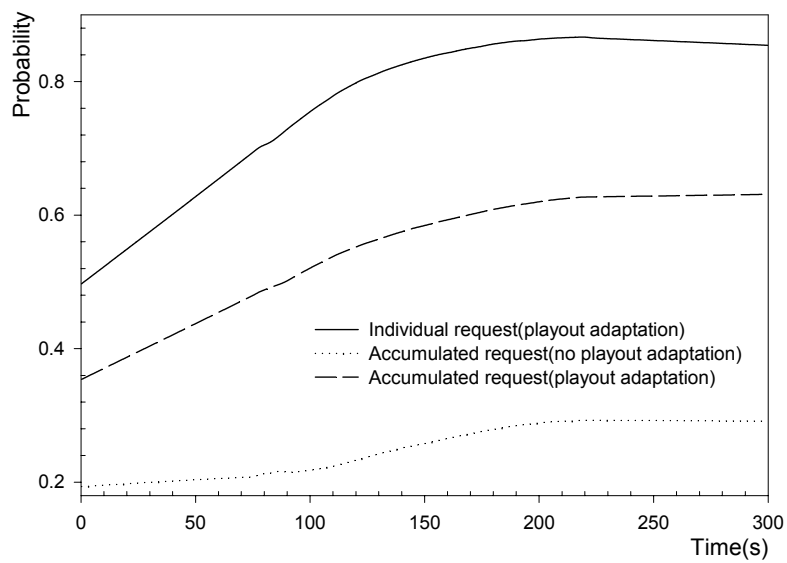
Three different error recovery cases

1. ***Individual retransmission request***
   : *Playout control, Server-aid, NACK-based individual retransmission request*
2. ***Accumulated retransmission request without playout aid***
   : *No playout control, No server-aid, NACK-based full accumulation*
3. ***Accumulated retransmission request with playout aid***
   : *Playout control, Server-aid, NACK-based accumulated retransmission request*

Required feedback bandwidth



Averaged lost packet repair probability



Conclusion and Discussion

Frameworks suitable for one-to-many multicast media streaming

- ***adaptive playout control (local adaptation): to reduce discontinuity from buffer under-/overflows.***
- ***Server-aid adaptation: to synchronize presentation time of session members and to assist the loss recovery at each client.***
- ***Cumulative NACK-based error recovery and the role of adaptive playout control: to reduce feedback bandwidth occurred by individual retransmission request and retransmission fails (late reply)***

The overall frameworks can reduce the playout discontinuity, and thus mitigate the client heterogeneity

Require precise unit conversion from buffer volume to its corresponding time quantity.

Require proper congestion control mechanisms.

Appendix

Notation

Notation	Description
P_s, P_c^i, Δ_p	Normal playout speed, Current playout speed at a client i, Playback factor
$\varepsilon_t, \varepsilon_c^i$	Target synchronization range, Skew between target and actual presentation time
T_s, T_c^i	Target and Actual presentation time
b_L, b_R, b_H, b_S	Lower limit, Retransmission limit, Higher limit, Buffer size
$p(t), SR(n)$	Presentation time, n^{th} Sender Report
$b(t), b^i(n)$	Buffer occupancy level at t, b(t) when the first packet i comprising accumulated packet is lost
$A_c, A_c^n(n)$	Accumulated time, required time to accumulate n packets

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Appendix

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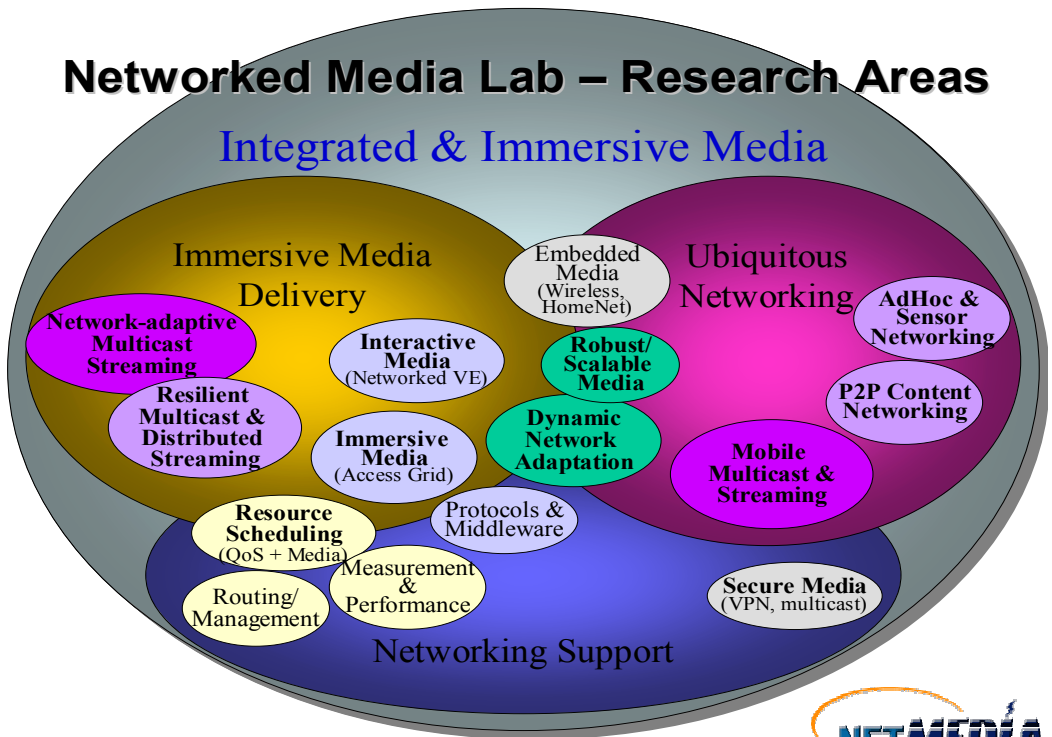
Thank You!

Networked Media Lab.

- **Networked Media Lab (Since Sept. 2001)**
  - Faculty: JongWon Kim, Ph.D.
  - Members: 1 PostDoc / 2 Ph.D/7 M.S. candidates
  - http://netmedia.kjist.ac.kr

Networked Media Systems and Protocols focusing "Reliable and Flexible Delivery System for Integrated Media over Wired/Wireless Networks"



Networked Media Lab.

- Dynamic Network Adaptation Framework
- Standard-based Internet/Mobile Media Streaming
- Reliable/Secure/P2P Multicasting + Content Dist. Networking
- Resource Management and Scheduling support (E2E QoS)
- Ubiquitous Networking (Mobility, Multicast, QoS) Support
- Immersive Media Delivery (Access GRID, Networked VE)
- Media Networking Middleware Support (GRID, Home Net)
- Robust and Scalable media for Universal Media Access