Retransmission Scheme for Voice , Data and Video using Voice Cell Jitter and Delay in ATM Networks

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Abstract: Multimedia is characterized by busty traffic and of fen stringent real time constraint multimedia traffic requires quality of service guarantees. This gives rise to traffic management issues in order to obtain high network utilization and qos guarantee to the multimedia stream. The aim of this article is to prevent an overview of the traffic management issues and their possible solutions for retransmission of multimedia over asynchronous transfer mode networks.

Keywords: Voice and Data Integration, Integration of Video

1. Introduction

Scalability especially in copying with increasing demand for bandwidth. The quality of service supports internetworking of heterogeneous networks. Simplication of the development process for multimedia networking applications. Characterizing multimedia networking application requirements by providing a taxanomy with respect to the transport architecture[7] and associated qos requirements. Studying the specific example of distributed video production which encomposes a variey of multimedia application charscteristics and qos requirements. This study focus on DVP qos requirements transport architecture and realtime network adaptation.

Developing implementation models for ATM adaptation models for ATM adaptation layer in order to ensure proper delivery of end to end qos for both CBR and VBR multimedia traffic[8]. Performance verification of the implementation model is achieved using cpmprehensive simulation models for an end to end connection. Proposing a generic realtime transport and adaptation protocol in order to enable IP networks to transport real time multimedia with a variety of QOS requirements under different network conditions. GRAP simplifies realtime application development and enables a smooth transition from best effort interest to more QOS aware transport models.

Developing a reference implementation model for GRAP on a select application on a socket application programming interface that simplifies the programming task for multimedia networking applications. Conducting experimental performance verification for GRAP in order to confirm the validity of the protocol and provide guidelines for the solution of associated design parameters.

2. ATM network and service categories.

Fair bandwidth sharing is based on sharing the link bandwidth assigned to connection during connection setup bandwidth scheduling assign a limited amount of bandwidth to a number of video connections according to specific scheduling time slots. This scheme[4] is easy to implement but suffers from lack of flexibility due to the time varying nature of the bandwidth requirement of multimedia streams.

Transcoding scheme is based on first decoding the compressed video and then reencoding the video at a lower bit rate{1}. The cost of a real time hardware encoder unit per multimedia stream makes this method very expensive for realtime multimedia applications. This advantage of this scheme are this straight forward and the bit rate of the encounter can easily be controlled by the feedback from the network.

Multirate coding scheme[11] encode and store video at a number of different bit rate. All the streams are played back simultaneously in a lockstep fashion and sent in a switch to the server which receives feedback from the network. Frame dropping scheme drops frames in order to match the reduced available bandwidth from the network. In the case of congestion in the network less important frames may be dropped by the server. The least important B-frames are dropped first followed by the dropping of P-frames if required.

Block dropping scheme blocks of frame are selectively dropped at the server to reduce the transmission bandwidth requirements{2}. At the client side the dropped blocks are interpolated from the neighbouring blocks of the frame. Layering coding the video is coded into a number of layer called the base layer and enhancement layer{3}. The enhancement layer combined with the base layer increase the quality of the video. In order to reduce the video bit rate during period of network congestion and the network congestion enhancement layer or not transmitted from the server.

3. Simulation study

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Physical channel transportation is the lowest layer with two win'32 file pipes are used to simulate the forward and feedback channel of each has a scrampling error generator attached. The transceiver on both the transmitter and receiver side are simulated through two win'32 task that interact with win'32 pipes. The built in file lock mechanism in win'32 platform enforces the synchronization of the concurrent access of the transceiver and error generator task. This layer provides asynchronous data exchanges between the two peers. Synchronization and delay control is the second lowest layer. The RTT delay and slight alignments are realized in this paper. IT also provides buffering of incoming and outgoing packet. This layer presents a synchronous transparent API to the upper layer which is protocol phase. The protocol phase is the processing logic are implemented in this layer. The sliding window structure[5] is also located inside. Upper layer simulation basically generate traffic and passes down to protocol layer on the transmitter side. It absorb data streams and maintain statistics on the receiver side.

The lost file containing an entry for each server accessed during the simulation. Each entry contains a unique ID and the DNS name the more window size for connection to the lost. The lost file contains and entry for each HTTP request. The entry contain the lost ID for the server request is fetched from the size of the HTTP request and response time taken in the US component of the network.

The simulation parameter includes the buffer size used by the proxies. The presence or absence of a US proxy and the number of connections between the proxities and the international link speed and delay in each direction.

Throughput = receive buffer size / Round Trip Time

Proportionate bandwidth = total bandwidth * weight / summation weight of all active connections

The packet arrival rate exhibits a piece wise constant bit rate behavior. A common network clock is available at both transmitter and receiver. Rate discontinuity occurring at the beginning of the connection or after a maximum silence period[6] . Rate change is smooth change due to a new frame or slice. Maximum silence period i.e. smooth rate change due to a maximum silence period is allowed and is directed. The initial rate cell such as for the first generator cell to the transmitted after a maximum silence period.

Severe jitter , we test the performance in the presence of severe jitter conditions by setting the CDV to 10MS. Supplying applications with a rich set of flexible and programmable realtime adaptation capabilities. Providing network transparency by abstracting network level detail for the applications.

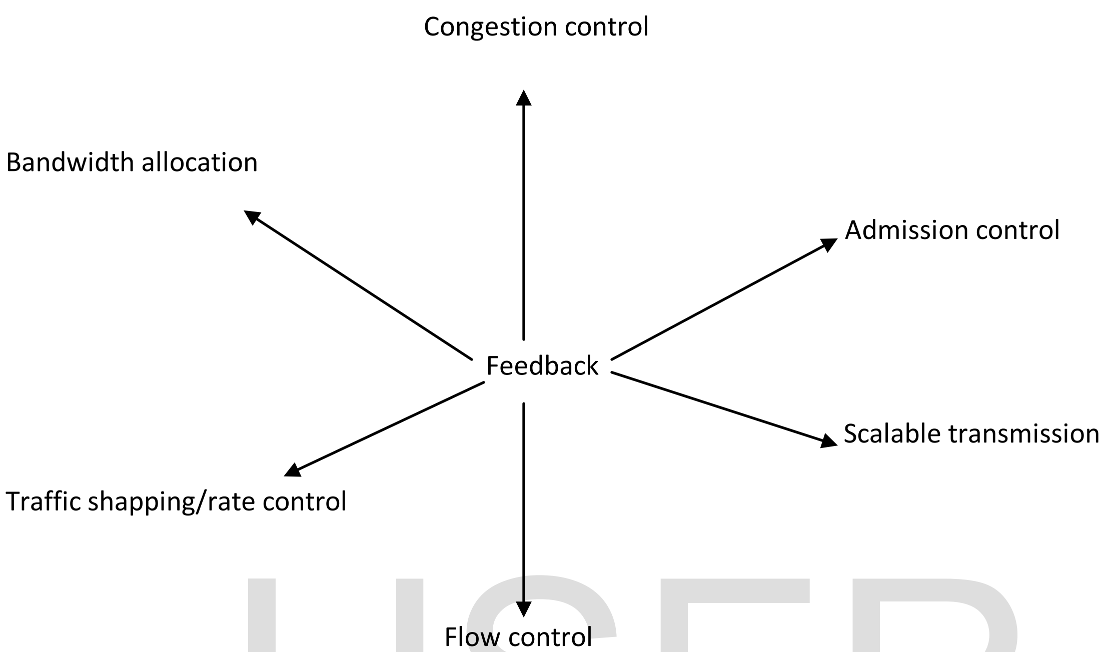

Fig : 1

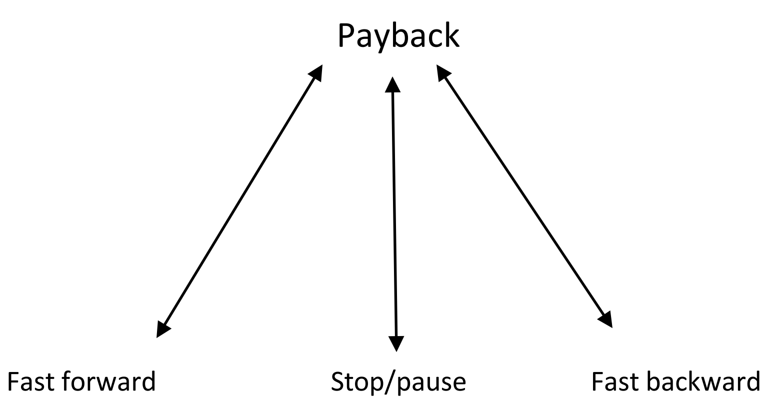

Fig : 2

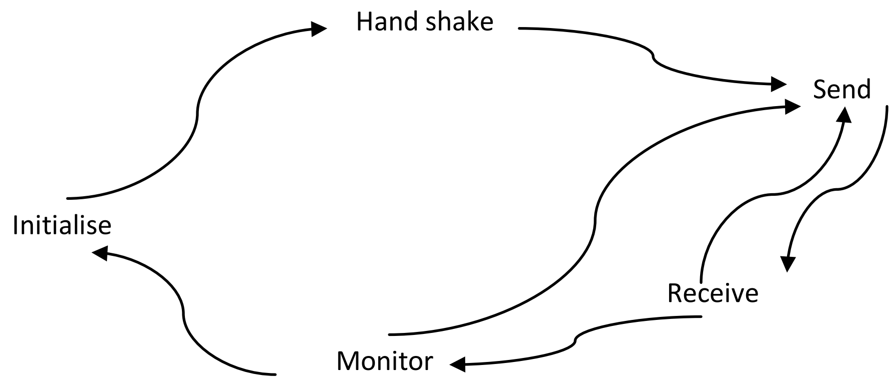

Fig : 3

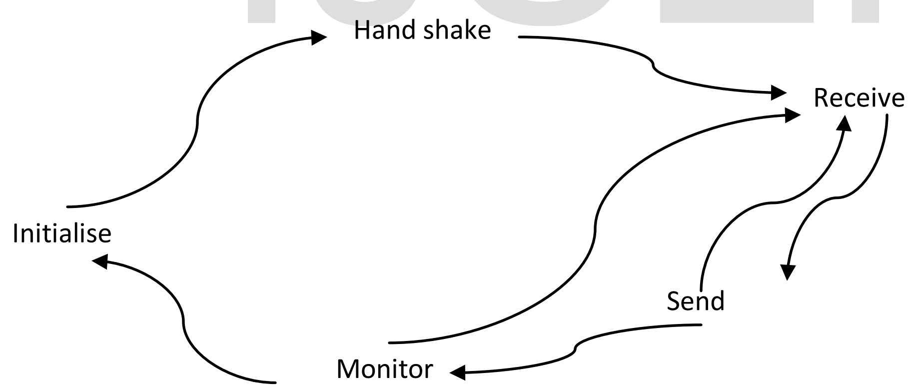

Fig : 4

ATM → RS outer encoder → Symbol inverter → Inner encoder → Channel interleaver ↓

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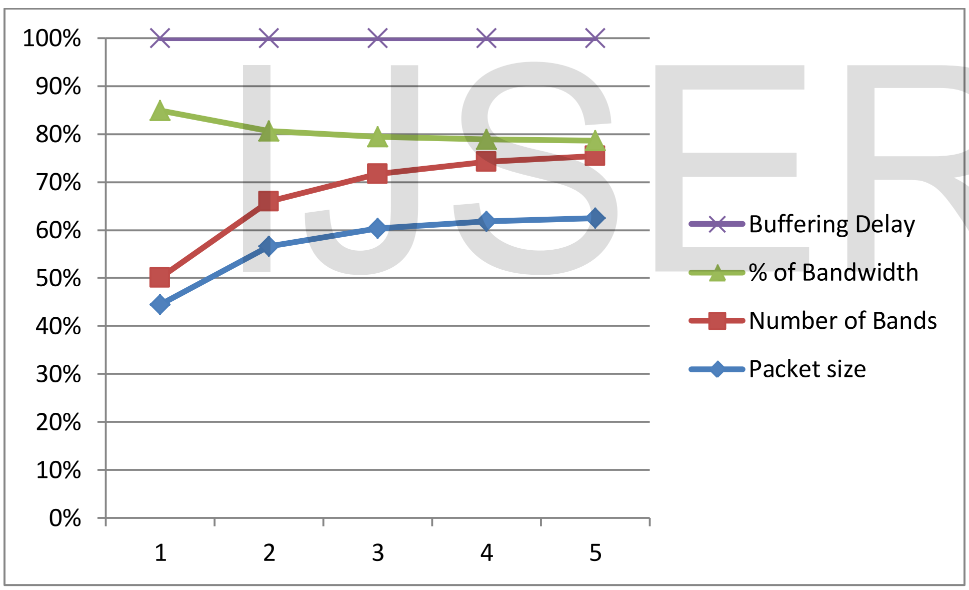
Cells

ATM ← Cells   RS outer encoder ←   Channel deinterleaver ←   Wireless channel

Real Time Traffic : Table1

Packet size	Number of Bands	% of Bandwidth	Buffering Delay
32	4	25	10.9
48	8	12.5	16.4
64	12	8.3	21.8
80	16	6.1	27.3
96	20	5	32.8

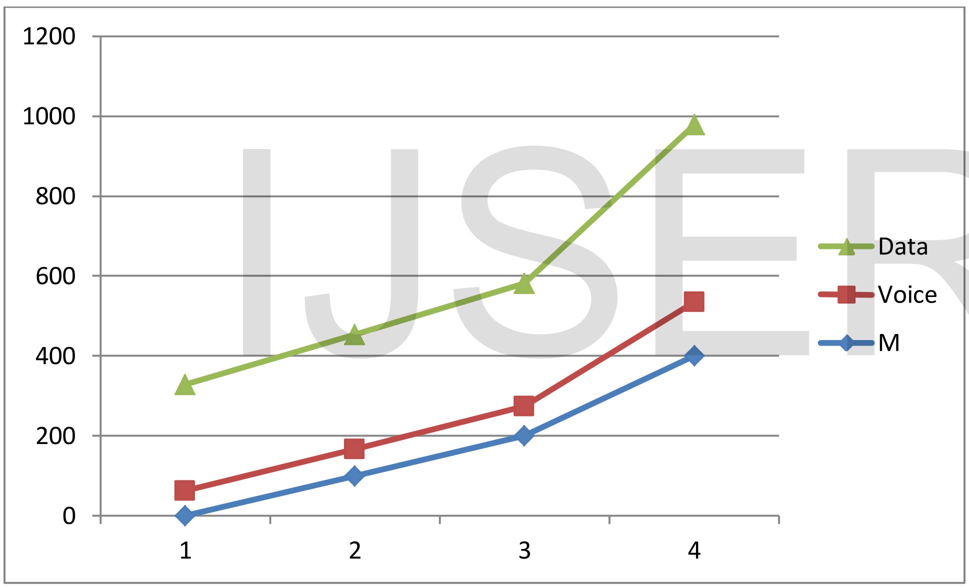


Compatibility requirement : Table2

Application Type	Bandwidth	Delay Time	Jitter Time	Crror Control
Offline video	NA	High	High	Low
Video on Demand	High	Medium	Low	Low
Live Broadcast	High	Medium	Low	Low
Joint Video Production	High	Low	Low	Low

Throughput : Table 3

M	Voice	Data
0	63	265
100	67	287
200	74	307
400	135	444



4. Performance Evaluation

Connections request from earth terminals to satellite. Enqueueing and dequeueing of cells at earth terminals. Minimum impact[9] on the design of the OBP most of the complexity i.e added to the satellite network is confined to the ground segment. Non adhoc scheduling approach to the strategy can be implemented in conjuction with any class based scheduling scheme. Changing applicvability of the proposed complexity strategy can also from part of the pricing changing architecture of the satellite network operator to maximize the revenue generation. A

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continuous flow indicating the buffer occupancy of cells from all connections excluding the real time video source. Weight and threshold levels are set to uniformly distributed small random numbers.

For voice

Mean on duration – 0.3521

Mean off duration – 0.6501

PCR – 150 cells/sec

For Data

Mean on duration – 0.052

Mean off duration – 1.8s

PCR – 1000 cells/sec

For video

PCR – 11578 cells/sec

MCR – 972 cells /sec

Conclusion and future work

The simulation results clearly shows that it is an efficient and simple control mechanism which adapt the ratios of voice and the data traffic on the network resulting in a fairly service among the voice and data traffic. Several scheme to scale the bit rate of the multimedia such as frame dropping and block duping can be used to scale down the bit rate of a precoded multimedia stream during period of network congestion.

As the channel becomes noisy with high BER[10]. Such as handoff high speed and sudden channel information to the powerful error control with lower coding rate should be used. The proposed scheme provides improved throughput over the general ones. Fig 1-4 and Table 1-4 throughput of hybrid ARQ and for different block size are compared. The performance of adaptive hybrid ARQ is better than ARQ techniques.

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The adaptive error control algorithm is further developed into a practical implementation which is computationally simple and suitable for using on low power wireless terminals. Mathematical model of the proposed real time ARQ. Some mathematical tools are available like burst error environment.

Abbreviations

UPC – Usage Parameter Control

VPI – Virtual Path Identifier

VCI – Virtual Channael Identifier

BER – Bit Error Rate

CDV – Cell Delay Variation

CLP – Cell Loss Priority

CLR – Cell Loss Ratio

CDR – Cell Delay Ratio

DFQ – Delayed Frame Queueing

HRR – Hierachial Round Robin

IPP – Interrupted Poison Process

RTD – Round Trip Delay

RTT – Round Trip Time

SDH – Synchronous Digital Hierachy

SNR – Signal Noise Ratio

ACR – Adaptive Check Recovery

ALF – Application Level Framing

ASF – Advanced Streaming Format

FEC – Forward Error Correction

GPC – Generic Packetization Component

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PSN – Packet Sequence NUmber

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