Improving VBR Voice and Cross Layer Resource Management QOS in ATM Networks

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Abstract - In this paper we investigate the issues of error control in the communication networks. In the network we receive the alternate error control scheme available for providing reliable end to end communication in the network environments. Through many studies the performance and tradeoffs of these schemes are discussed here. Based on the application environments and QOS requirements the design issues of error control are discussed to achieve the best solution.

1. INTRODUCTION
Delay is the elapsed time between a node sending a message and another node receiving the message. It is the measures of the amount of data held in transit in the network. The greater delay between the sender and receiver, the more intensive of the feedback loop becomes and therefore the end to end protocols becomes more intensive to short term dynamic changes in the network load. For intensive voice and video applications in the introduction of delay causes the system to appear introduction unresponsive in an internet telephony application there is some times a delay of up to several senders before the receiver can hear what the sender is saying. This linders interactive communication [4] between the two parties. Reliability is a property of the transmission medium and can be throughput of as a average error rate of the medium. An unreliable prone network is results of faulty channels that not only duplicate packets in transmit but also alter their order. Un reliability causes induced distortion as a original signal at the receiver side.

Jitter is the variation in end to end transmits delay. It is an abbreviation that occurs when video or voice is transmitted over a network and packet do not arrive at its destination in consecutive order or on a timely basis i.e. they vary in latency. High level of jitter in application is un acceptable in situation where the application is real time based such as an audio or video based. In such cases jitter causes the signal to be disturbed which is particularly damaging to multimedia traffic. The play back audio or video data may have a jitter or shaking quality. Bandwidth is a measure of data transmission capability. It is the maximal data transfer rate that can be sustained between end points. By increasing bandwidth we can transfer more data. Network bandwidth can be utilized as a pipe that transfers data. The layer the pipe the more data can be sent through it by increasing bandwidth we can always achieve QOS. This is not applicable because bandwidth is not cheap. Hence the issue here is to obtain certain level of QOS by using the main bandwidth required.

2. PROBLEM STATEMENT
First notice that the source and destination sites deals with the large packets while ATM backbone deal with small cells. While segmentation and reassembly functions can be added easily by proper adaptation layer protocols losing one or more cells of a packet may render the higher layer packet unless. The policing scheme along with rate enforcement. A necessary condition for FEC to be effective is for packet deletions to the dispersed over several blocks.

Different kinds of services have different requirements which improve error control such as BER delay. Complexity and availability of the shelf technologies. Channel interleaving to randomize burst error among different cells. FEC to reduce channel error rate[1]. It is very powerful FEC code with reasonable complexity available VLSI decoder chips. Increased importance of managing the network. The complexity including integrity, traffic difficulties, wireless access and heterogeneous data representation and to give the way of network and also the weakness in conventional management.

3. OBJECTIVES
Analyzing the performances of parameter jitter control error control delay and packet delay variation. Designing the network model for MPLS with different scenarios. Identification of the parameter which have the noticeable results can be performed based on voice traffic. Study out of the documents for understanding and determining the constraint and parameter that needed to be taken under consideration for the analysis. Determine the suitable model for supporting the real time application. Designing a model for the simulation using the simulation tools agent. Justify the research over using the simulated data as a measure for the given analysis.
4. RELATED WORKS

Reservation protocol request resources in only one direction i.e. it treats a sender separately from the receiver although the same application must be running at both the sender and the receiver side. RVSP makes resource allocation for both unicast and multicast applications. RVSP is the receiver oriented i.e. the receiver of the data flow initiates and maintains the resource reservation itself used for the flow. Resource reservation itself is not a routing protocol but it is designed to work with the existing routing protocols. RVSP supports both IPV4 and IPV6. The IP header contain a differentiated services code point indicating the level of service desired. The DSCP maps the packet to a particular mapping behavior by processing by DS-complaint router. The PHP provides a particular service level in accurate with network policy.

At the first hop router is the MPLS network the router makes a forward decision based on the destination address and then determine the approximate label value attaches the label to the packet and it forward to the next hop. At the next hop the router uses the label value as an index into a table that specifies the next hop and a new label. The LSR attaches the new label then forward the packet to the next hop. SBM manages the bandwidth over both legacy and newer LAN topologies[7] and takes advantages of the additional functionality as its between available in the new generation of switches and hubs or bridges. SBM specifies only a signaling method and portal for LAN based admission control over RVSP flows. It does not define any traffic control mechanism for the link layer. In the absence of any link layer traffic control or priority queue mechanism in the underlying LAN the SBM based admission control only limit the total amount of traffic load imposed by RVSP enabled flows on a shared LAN. Bandwidth allocation mechanism only limit the total amount of traffic load imposed by RVSP enabled flows on a shared LAN. Bandwidth allocation maintain state about mechanism of resources on the subnet and performs admission control according to the resources available and other administrator defined policy criteria. Request module resides in every end station and not in any switch. The RM maps between layer 2 priority heads and the higher level QOS protocol parameter according to the administrator defined policy.

![Diagram of MPLS signaling, Constraint Based Routing, IGP Routing Protocol, Link state Database, Functions of Access Point, FEC push out policy, New Traffic Load Monitor, Decodable Formal Rat, Packet Loss Monitor, Priority Checking, Packet Block]
Traffic Model

CBR

VBR

Deterministic

Stochastic

Leaky Bucket Traffic Model

Self Similar Traffic Model

EC scheme

Outer code Encoder

Outer Inter Leaver

Inner code encode

Inner Inter Leaver

Modules

Inner Code Decoder

Inner De Inter Leaver

Demodulator

Burst Error Channel

Outer de inter leaver

Outer de coder

Multi Media Network System

User

Applications

System

Multimedia Design

Network
5. SIMULATION RESULTS

Voice quality should be comparable to what is available using PSTN even over networks with variable level of Quality of Service. The underlying output network must meet strict performance criteria including minimum call refusals network latency packet loss and disconnect. The goal should be met even during congested provides or when multiple user must share network resources. Call control must make the telephone process transparent so that callers are unaware of the technology they are using PSTN/VOIP service internetworking should be provided frequently consolidated with the PSTN operation support system. Maintaining and examining over increasing error loss. Tracking fault back to devices that have to fixed location in the network i.e. personal router devices networked via temporary connections. Multimedia network tend to generate congestion errors that cannot be considered faults. The destination between genuine errors and fault tolerance and the error must be more difficult to make. The gathering of fault related data in more resource intensive. Improved distribution of the processing capabilities and/or storage facilities is needed. Keeping track of all virtual connections and their rerouting. Monitoring of resources as compared to QOS offered to terminals. In terms of uplink or downlink and control being able to change things on the fly. Dual stack confirmation and protocol tuning you might want to configure your device so that it can load a particular stack depending on the type of connection required. At the moment your monitor want to able to fine tune your stack configuration according to the configuration of the available access point. Configuration of alarm thresholds and the multimedia application require alarm setting function capable of dealing with a more sophisticated mix of traffic on the network. Monitoring functions to spot check quality of service delivery satisfying commitments made. Monitoring and reporting of errors on the wireless link control technology flow error connection retransmission adaptive and error control schemes. Production of report to aid on the planning of additional cache servers. Proxy servers are other high congestion system in the network monitoring of response time alarm thresholds and policies. Monitoring QOS degradation during peak times with the view of findings using optimal algorithms [4]. In the case of wireless connections production of statistics as the air interface including link usage by connection results and connection rejects [5]. Keeping track of network usage by difficult traffic classes . This information is useful for network planning. Keeping track of user based and application based accounting profile and statistics. Discount or credit schemes for unsatisfied QOS combined with performance management. Management of special tariff according to the character of the particular connections. Changing for special routing connections and performing the security checks when crossing domains and a looser check when moving in the same domain. Mechanism to perform power on authentication mechanism to generate privileges on network resource usage. By using the simplified point to point protocol on the computational at the thin client will be drastically reduced. This also results in faster access of management information and low power consumption as the thin client.

Simple and easy to use and understand due to its GUI. Packet level simulation which produces low error in simulation with real time. Portable as its run in most of the platforms such as linux and windows. Being commercial the support and updates are frequent. Flexible to edit the flow of event at any node by altering the codes on the given nodes. Packet formats are also editable depending upon the requirements. It able to map real time scenario such as world map office. Here the table and diagram shows the entire report and details to solve any type of problems using the all the formulas and its relevant results.
Propagation Delay = Required Application Delay – Queuing Delay * Hop Count
T.upper = Source Satellite Distance / Speed of Signal
T.lower = Destination Satellite Distance / Speed of Signal
T.min = Signal Length / Speed of Signal

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TABLE 1: COMPARISON OF VOIP

TABLE 2: SATELLITE DELAY
Buffer utilization defined that as the ratio of buffers utilized to the maximum size of buffer available at the base station. Hand of calls accepted defined as the ration of hard calls accepted to the total hard off calls arrived at the base station. New calls accepted is the ratio of new calls arrived at the base station. Dropping the probability is defined as the ratio of packets depend on the total number of packets arrived at the base station.

CONCLUSION
In the OSI architecture provides QOS in the network layer and some enhancement in the transport layer. The OSI 95 object considered integrated QOS specifications and negotiations in the transport probability [3]. Internet protocol stack based on IP protocol provide resource reservation if RVSP protocol [6] is used. QOS handling and management is provided in end point architecture at the application and transport subsystem [2] where QOS on the end to end control management protocol implements QOS handling over both sub systems and relies on control management in ATM networks.

ACRONYMS
RVSP – RESOURCE RESERVATION PROTOCOLS
BRAN – BROADBAND RADIO ACCESS NETWORKS
NUPT – NORMALIZED USER PAYLOAD THROUGHPUT
OFDM – ORTHOGONAL FREQUENCY DIVISION MULTIPLEXING
WAND – WIRELESS ATM NETWORK DEMONSTRATOR
APSK – AMPLITUDE PHASE SHIFT KEYING
AWGN – ADDITIVE WHITE GAUSSIAN NOISE
BPSK – BIARY PHASE SHIFT KEYING
DAMA – DEMAND ASSIGNMENT AND MULTIPLE ACCESS
DBRA – DYNAMIC BANDWIDTH AND RESOURCE ALLOCATION
HD TV – HIGH DEFINITION TELEVISION
HTTP – HYPERTEXT TRANSFER PROTOCOL
ICMP – INTERNET CONTROL MESSAGE PROTOCOL
IMSI – INTERNATIONAL MOBILE SUBSCRIBER IDENTITY
LDPC – LOW DENSITY PARITY CHECK
MPDV – MAC PACKET DATA UNIT
MPEG – MOVING PICTURE EXPERT GROUP
PSTN – PUBLIC SWITCHED TELEPHONE NETWORK
QPSK – QUADRATURE PHASE SHIFT KEYING
RBDC – RATE BASE DYNAMIC CAPACITY
RCST – RETURN CHANNEL SATELLITE TERMINAL
SMAC – SATELLITE MEDIUM ACCESS CONTROL
SNIR – SIGNAL TO NOISE INTERFACE RATION
TDMA – TIME DIVISION MULTIPLE ACCESS
VBDC – VOLUME BASE DYNAMIC CAPACITY

REFERENCES
[3]. Cross Layer QOS for IP Based Hybrid Satellite Terrestrial Networks, December 2011