Enhanced Transport Scheme for Profitable Bandwidth Sharing in Network Traffics

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ABSTRACT

Modern networks integrate traffic to consist audio, data and video and Broadband Integrated Services Digital Network (B-ISDN) supports all types of traffic, including still or full motion image applications. With increasing number of wireless users, more traffic is added and networks become congested. Though personal computing facilitates easy access, manipulation, storage and exchange of information, network computing involves interconnections and resource sharing. As congestion remains the major challenge in broadband communication, transmission of multimedia data frames results in traffic levels overwhelming the network medium. Technically, to maximise bandwidth, all communication services may be provided on a single network to depict profitable economy of sharing thereby motivating an idea of providing integrated services over the access networks. Notably, implementation of Asynchronous Transfer Mode (ATM) technology as target technique for broadband sharing provisions profitable economic service continuity, even as users move around over wide area of coverage. Also, an enhanced transport scheme is necessary to meet users' negotiated Quality of Service (QoS). Such schemes, implemented via mathematical simulation techniques will help manage overlapping network traffics while allocated bandwidth is optimized. An integrated scheme of link-to-link rate-based traffic flow control implemented for Additive Increase Multiplicative Decrease (AIMD) algorithmic transmitted packets is proposed in this paper to provide for capacity maximization of bandwidth scarce resource to be utilized economically for all transmissions.

KEYWORDS: *Broadband Traffic, Congestion, TCP Flow, Bandwidth, ATM and QoS*

1.0 INTRODUCTION:

Modern communication including mobile broadband requires maximized bandwidth usage where services must be provided even as users move around. Congestion as a bottleneck results to network inability to tell application how much bandwidth is available at given instant of needs.

Broadband network offer of single telecommunication network for provision of multiple services to users due to economic reasons (Jones, 2008) suggests an assumed profitable economy of sharing. This is also motivated by availability of integrated service networks and optimal control of traffic for negotiated Quality of Service (QoS). Generally, broadband networks with listed characteristics of: broadband access, multimedia service, multi-point connection, multi-rate application and economically-implemented multiple-service platforms.

Congestion in packet-switching network offers performance degradation as saturation of network resources results to collapse. Resources such as *communication links, processor cycles, and memory buffers* saturation results to *long delay of message delivery, waste of system resources*, and e *network collapse* when communication links are completely disconnected. Network congestion in form of traffic jams in big cities, has become real threat to wireless applications.

Network congestion, a resource sharing problem in packet-switched network contributes to network overload as applications send more data than network can handle. Network buffer fills up and overflows as application retransmits data. Cost of retransmission is high. Kaufmann (2006) asserts that as application retransmits data, more data is added to traffic and further congestion arises from retransmission.

Communication transport services' recognizes three types of traffic – Constant Bit Rate (CBR), Variable Bit Rate (VBR) and Available Bit Rate (ABR). ATM networks performs congestion control on ABR traffic to efficiently share bandwidth among multiple request but this action is not guaranteed in either CBR or VBR traffics (Raj, 2006).

The major objective of this paper is to propose offer an integrated scheme to control (avoid) congestion by (1) *implementing a set of network congestion control* (to allow network protect all shared resources and allocate them fairly between network and end-devices), (2) *ensuring guaranteed flow control* (using sliding window protocol or 'traffic-shaping' algorithm to reduce traffic flow and (3) e*nforcing error control actions*, (using retransmission to allow cell-dropping). These congestion control actions are synergistic to flow controls.

2.0 CONCEPT OF CONGESTION IN NETWORKS

Congestion is concerned with the bottleneck routers in the packet switched network and its control can be distinguished from routing in that sometimes there is no way to 'route around' a congested router. Efforts must be made by network nodes to respond to overload conditions to achieve effective flow control such that the fast senders must be prevented (using appropriate strategies) from sending data into the network.

Congestion management is achieved on the network as congestion is controlled by providing each user with mechanisms to specify and obtain desired QoS (Vidya, 2010).

Broadband and Congestion Control

The Longman dictionary of contemporary English defined *broadband* communications as a technical system of connecting computers to the Internet and moving information (messages, pictures) at a high speed while the *broadband* is a system of encoding radio signals with several messages to be sent at the same time.

Observed in Morgan (2007), congestion control scheme is intended to regulate how a fast sender sends data into the modern networks carrying integrated traffic. Insufficiency of required resources (including *available link capacity, router buffers* among others) in networks is satisfied by special provisions made by Broadband Integrated Services Digital Network (B-ISDN) specifications. Itemised in Floyd and Fall (1999), Raj (2006) and Jones (2008), B-ISDN evaluates broadband networks for performance by classifying network flow into data, video and audio traffics.

Mark (2006) defined congestion control as a technique for monitoring network utilization and manipulating transmission for data frames to keep traffic levels from overwhelming the network medium while Morgan (2007) defined congestion control as efforts made by network nodes to prevent or respond to *overload* conditions as traffic data is transmitted.

Concepts of Flows, flow control and Services

Flow is a sequence of packets sent between a source/destination pair. The pair follows the same route through the network. With connectionless flows (X.25) within TCP/IP models, there exists no state at the routers. This is because TCP is implemented as a connection at transport layer while IP provides a connectionless datagram delivery service.

The *flow control* scheme keeps fast senders from overrunning slow receivers in agreement with congestion control, which is intended to keep the fast senders from sending data to prevent router bottlenecks (buffer shortages).

Morgan (2006) explained QoS as a service model that supports some type of guarantee for a flow's service while Jones (2008) described the best-effort service as a service whereby the hosts are given no opportunity to ask for a guarantee on the flow's service.

To achieve flow control, Vidya (2010) defined congestion control algorithm as a set of instructions, which operates by sending message back to the various source(s) to *slow down* as network gets into trouble.

Traffic Management in ATM Networks

Traffic management is defined by Keshav (2011) as a set of policies and mechanisms that allow a network to efficiently satisfy a diverse range of service request. The fundamental aspect of traffic management is diversity in user requirements and efficiency in satisfying them.

Although, dealing with congestion after it is first detected is more effective than letting it gum up the works and then trying to deal with it. This observation lead to the idea of discarding packets before all the buffer space is really exhausted as practised in Random Early Detection (RED) algorithm (Floyd and Jacobson, 1993).

Algorithms used for controlling congestion are either **open-loop** (*retransmission, window, admission, acknowledgement, discarding* policies) or **close-loop** (*backpressure, choke, implicit and explicit* policies*)* and the various components of traffic management includes:

- scheduling - server deciding which request to serve next using an algorithm on the set of requests, sharing network resource *fairly* to and providing performance critical applications
- signalling – a process by which an end point requests the network to setup, tear down or regenerate a call
- renegotiation – where the guaranteed-service connection must specify its traffic descriptor at the time of connection establishment
- admission control – where the signalling network carries signalling messages and make the required reservations and
- congestion management – where each user is provided with mechanism specifying desired QoS from the network

Contributive factors of congestion

Congestion as a dynamic problem is justified not only to buffer space shortage, undesirable delay is still evidenced in slow links even with large buffers leading to short terms. Although high speed links available in recently advanced technology aggravates congestion problem at peak rates only and transmission of more data per unit time has resultant long queue and long delays.

Static solutions of increasing the buffer are not sufficient to solve dynamic problems of congestion, therefore, proper congestion management mechanisms are suggested in this study.

A *multimedia* transmission communicates audio, data, still images or full-motion video or any of its combinations and the multimedia network has to support a broad range of bit-rates demanded by connections. With *multi-rate* service requirement, the network should be able to allocate transmission capacity flexibly to connections.

Basic characteristic of packet-based information transport in B-ISDN/ATM networks is sharing of critical network resources (*memory, processor real-time, transmission bandwidth*) between large numbers of information flows. Controls are needed to protect such shared resources from depletion, allocate them fairly and as well increase *transport* speeds.

Usually, at broadband speeds, three main issues relates to control architectures in broadband. These includes (1) *level of control algorithm complexity* to be implemented in individual devices (limited to costs associated with specialized high-speed hardwares); (2) *effectiveness of sharing explicit control* and (3) *status information between various ATM cell-handling devices along transport path*s. to trigger continuous, real-time control actions (this is limited by large propagation-delay-to-cell-queue-time-constant ratios) and *complete transport integrity* (this is limited by the large and expensive buffering memories needed to eliminate ATM cells losses in the network).

Although, large buffering memories would add significantly to end-to-end delays implying that ATM cell loss during broadband transport is a phenomenon somehow difficult to 'design away'. Therefore, it becomes important that error control strategies be incorporated with congestion and flow control strategies.

3.0 ARCHITECTURAL DESIGN PRINCIPLE FOR BROADBAND USAGE

Effective congestion control, simultaneously at network and transport layer reduces the need to retransmit data in communication (Jones, 2008). When one part of the subnet involving one or more routers in an area becomes overloaded, congestion results because the router received packets faster than it can forward. The usual option of either - the subnet preventing additional packets from entering congested region (to prevent *enqueuing*) until those already present are processed or the congested router discarding (to allow *dequeuing*) queued packets to make room for new arrivals is available for designs.

In this research, simulation of combined algorithm is implemented as technique to adjust speed of message flow (in packets) to match available end-to-end network capacity. To adapt the speed of transmission to match available network capacity, additive increase multiplicative decrease (AIMD) algorithm is adopted. Also, fair queue (FQ) system implemented within AIMD algorithm, consists of *arrivals*, *waiting* element, *unit served*, *service facility* and *unit leaving* queue after service.

Integrated scheme: AIMD and LRQ algorithm

Conventionally, all networks are guided by Transmission Control Protocol (TCP). An integrated scheme is hereby proposed to assist the sliding window protocol measure accurate window size in bytes during transmission. Capacity is determined as stated in equation (1), where sliding window ensures full utilization of the link, having implemented required full utilization both at data link layer and transport layer.

$$capacity = bandwidth \times delay\ of\ path \qquad ....(1)$$

Finding the link capacity, initial window size W_{init} for slow start is estimated at one packet per RTT for safe options of ACK. ACK of new data increasing window size by 1packet per RTT. When the packet is lost, slow start technique (Jacobson, 1988) stop increasing window size. Slow start (SS) finds correct window for path as new connection starts. With two or more packets generated per ACK, window size doubles and SS rapidly finds correct window size for path/link. As connection is established, changes of packet size to available capacity effects congestion avoidance.

4. DISCUSSION

When network becomes congested, packets are lost unless retransmitted. Rate-based link-by-link technique reduces transmission rates enabling notification by incoming application but with credit-based technique, application gets fewer credits (available bandwidth) thereby leading to increasing packet loss. In rate-based, as network congestion clears, transmission rate increases, to maximize network capacity and applications thereby are enabled to transmit at full speed (Raj, 2006).

Link-by-link Rate-based Queued (LRQ) Control

ABR traffics implements variety of techniques of end-to-end, link-by-link, rate-based and credit-based traffic control in ATM networks. Rate-based traffic flow constantly tells the application what transmission rate (currently allowed rate) the sending device uses in a round-trip time (RTT) but credit-based technique requires network periodically replenishing application's credit (available bandwidth). Until network indicates amount of buffer space (credits) available to sending device, there would be no transmission.

Using link-by-link technique, each link along the network connection controls its traffic flow independently. The beauty of this scheme is that each link buffers data as needed to adjust the incoming speed to the outgoing speed and this augurs well with rate-based traffic-flow, which constantly monitors the application and the transmission rate simultaneously.

Unlike end-to-end rate-based congestion control scheme, where the network must measure the minimum available bandwidth along the connection, communicate the amount measured to application, thereby introducing additional overhead, each link in link-by-link simply forward their data as fast as it receives it, giving way to improved throughput and reduced delay.

Principle of conserving packets is maintained by AIMD algorithm when the sending rate is adjusted to incorporate faster reduction than increment. This is achievable in two steps:

- starting slowly and increasing gradually to find equilibrium. At equilibrium, one packet can be sent for each acknowledgement (ACK) received by
  - adding small amount to the sending speed each time interval without loss using window-based algorithm $W_i = W_{i-1} + \alpha\, wi$ *each RTT, where* α *= 1*
- respond to congestion rapidly by
  - multiplying the sending window by some factor B<1 each interval loss seen using $W_i = W_{i-1} + \beta\, wi$ *each RTT, where* β *= ½.*

Findings

Projected AIMD algorithm offer faster reduction than projected increase leading to stability in a network. As TCP offer required slow starts, congestion is avoided and equal share of bandwidth is approximately apportioned to each flow on the network sharing link or same traffic (Jacobson, 1988). The link-by-link queue (LRQ) rate-based algorithm enables efficient transmission. Furthermore, the sliding window protocol (i) acknowledges each packet and only sends new data when an ACK received and (ii) adjust the window size, based on AIMD rules. Slow, additive increase in window: $W_i = W_{i-1} + 1$ increases window until congestion observed make the scheme respond rapidly with embedded slow start. This is the required *congestion avoidance,* guided by random early detection (RED) transport scheme.

4.0 EVALUATION OF PROPOSED SOLUTION (AIMD + LRQ)

Combining AIMD and LRQ algorithms, link-by-link rate-based traffic flow and AIMD techniques enables ATM traffic monitor and offer precise congestion management than using only end-to-end rate-based traffic flow schemes. Fair queue algorithm is also embedded in link-by-link algorithm as it simply performs end-to-end flow control in end-device monitored (via application). This strategy avoids congestion collapse. As router communicates with node (end-device), network preserves existing equipment and provides for future growth of the link size (Jones, 2008).

Evaluation criteria used for deciding network *effectiveness* and *fair* allocation of resources include measurement of *throughput, efficiency, delay, queue length, goodput* and *power usage.* Relationship between these phenomena expressed for throughput as a measure of efficiency in resource utilization in given in (2) .

$$power = \frac{throughput}{delay} \quad \text{...........(2)}$$

Throughput, expressed as a function of end-to-end delay, is given in (3) as

$$throughput = power \times delay \quad \text{..........(3)}$$

Finally, Available Bit Rate (ABR) traffic for ATM networks is implemented with reduced overheads because traffic demands are stochastic. This is obtainable from reduced retransmission, full acknowledgement and prioritized admission, strongly supported by queuing algorithm. Packets are buffered, transmitted and marked or dropped. Indirect delay at router determines how congestion is controlled. Therefore, modern network equipped with high speed information processing, provided an integrated traffic (voice, video and data) to offer a more profitable economy of bandwidth sharing.

5.0 CONCLUSION

Rapid advances in communication technologies led to evolution of new techniques for networking. Topmost among these is software defined networking (SDN), which is shaping up future of data communication . ISDN frame-relay and ATM networks as two most-influential developments of networking advances the growing concern for profitable traffic management. Generally, congestion control in ATM networks is addressed using proposed strategies, especially as communication of multimedia packets now requires *multi-point* (implementation of many calls per cell) and *multi-rate* (implementation of ABR, CBR and FBR) traffics for efficiency. High profit is obtained in traffic sharing using combined strategies of additive increase multiplicative decrease and of

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Components used in multimedia transmission require various demands for improved communication qualities. *Bandwidth, signal latency* and *transmission fidelity* are traffic parameters for improved service delivery. Investigation of these parameters is suggested for future research.

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