Available Online at www.ijcsmc.com

International Journal of Computer Science and Mobile Computing

A Monthly Journal of Computer Science and Information Technology

ISSN 2320–088X

IJCSMC, Vol. 3, Issue. 12, December 2014, pg.567 – 573

RESEARCH ARTICLE

A Comparative Study and Survey on Broadcasting Multimedia Streaming Data Congestion Control Mechanisms

Ashish Parejiya[1], Dr. Vinod Desai[2]

[1]Ph.D Research Scholar, Mewar University, Rajasthan, India

[2]Computer Science Departments, Govt. Science College, Navsari, India

[1] ashishparejiya@gmail.com; [2] vinodl_desai@yahoo.com

Abstract— The real time transferring data of multimedia content (Multimedia streaming) come to be a projecting field with the advance of different networking technology and multimedia streaming technology. Most of the broadcasting applications such as video conferencing, Face-time, Daily-motion, Mobile video calling, Webinar, visual telephone, YouTube have become more and more popular etc. need multimedia communication techniques that send multimedia data from one end to another with enhanced productivity, strength and security. Multimedia broadcasting streaming service requires adequate bandwidth and delay dissimilarity to attain outstanding quality of service. Congestion occurs frequently owing to random fluctuations and burstiness of traffic flows within high speed networks. Congestion has to be resolved as it is the main cause for packet loss and long delay that affects the Quality of Service in the Networks. A number of researchers over last decade developed a number of congestion control mechanisms. But Congestion control for streamed media data traffic over Internet is still a great challenge. This paper presents A Comparative study, survey and points out Pros and Cons of various congestion control protocols and algorithms that were developed to suit media broadcasting traffic over Adhoc networks or internet.

Keywords— Congestion, Congestion control protocol, Broadcasting Multimedia Streaming Data, Quality of Service

I. INTRODUCTION

A. Multimedia streaming

Transmission of Multimedia streaming data from one end to another is Multimedia Streaming receiver. It enables users to view multimedia content immediately when the end user begins receiving the data. It is contrast to other schemes that require the user to wait for the entire media content to download before it can be viewed and hence minimizes the users waiting time. Its applications include video conferencing, Face-time, Daily-motion, Mobile video calling, Webinar, visual telephone, YouTube have become more and more popular etc. which requires Timeliness, Orderliness, and High Bandwidth. Multimedia data has to be displayed uninterruptedly at the receiver side which requires the network to deliver multimedia streaming data in a timely fashion. If the data does not arrive at the receiver before the play out time (life time) then they becomes unplayable. Hence delay should be very small to adapt to the real time constraint. The data size of multimedia ranges from a few megabytes to gigabytes. This large amount of data requires sufficient bandwidth. To minimize the effect of delay the buffering time increases when the delay variation is large.

If the buffering time increases users feel a low quality of multimedia. Also to maintain constant bandwidth delay variation has to be minimized. Hence to achieve good QoS Multimedia streaming requires sufficient bandwidth and proper delay variation characteristics with effective congestion control.

B. QoS

Quality of Service (QoS) refers to the capability of a network to provide better service to selected network traffic over various networking technologies. The primary goal of QoS is to provide priority including dedicated bandwidth, controlled Jitter, latency and loss characteristics. Depending upon the handling of network traffic different applications have different requirements. Those requirements are expressed using the following QoS related parameters.

- Bandwidth the rate at which application traffic must be carried by the network.
- Latency the delay that an application can tolerate in delivering a packet of data.
- Jitter variation in latency
- Loss- percentage of lost data.

This paper focuses on the last parameter Packet Loss which may be due to Congestion or wireless link errors. Several algorithms exist to control congestion thereby reducing packet loss and improving QoS.

C. Congestion control in Mobile Adhoc Networks

Most networks fail to tell applications about the availability of bandwidth at any given instant. As a result, an application doesn't have any information about how to control the transmission of data

Congestion occurs whenever the total input rate is greater than the output link capacity. When applications send more data than the network can handle buffering of data increases which lead to buffer overflows and packet losses [1]. Retransmission of the lost packet is done by the application. But it adds more traffic and further congests the network. Also retransmission may not be strictly necessary in multimedia streaming.

Static solutions are not sufficient to solve the Congestion as it is a dynamic problem. Congestion control is essential to guarantee that users get the good quality of service.

Most of the research has focused on algorithm development of end to end control schemes such as TCP. Such schemes are adapted to Adhoc networks but not efficient for mobile Adhoc networks as dynamic mobility of nodes may frequently occur which causes unpredictable fluctuations.

The network congestion causes long transfer packet delay and very low throughput thereby providing low quality of service. Therefore congestion control mechanisms are very important for Mobile Adhoc networks as there is a growing need to support Quality of service.

II. CLASSIFICATION OF CONGESTION CONTROL SCHEMES

A. Open-loop control

A source depicts its traffic during call establishment process and if available the network reserves the corresponding resources. Source adjusts its traffic to match the traffic descriptor. The control decisions do not depend on any feedback and they do not monitor network state. This technique is for Congestion avoidance.

B. Closed-loop control

A source obtains information about the current availability of resources based on feedback from control points in the network and dynamically adjusts its flow. This scheme monitors the network state and Control decisions are based on feedback obtained. It is for Congestion detection and prevention.

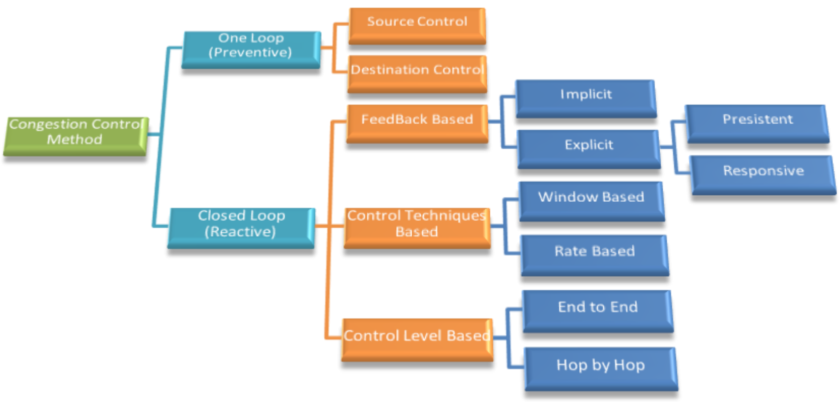

Figure I Classification of Congestion Control Schemes

Closed loop schemes can be further classified into three different ways.

1) *Feedback strategy:* The network traffic is controlled based on measurements (feedback).The Feedback can be explicit or implicit.

Explicit feedback - Control messages are sent as separate messages from the congestion point in the network to the end system (forward/backward) indicating the current status of the network. It can be persistent or responsive. In persistent constant feedback is provided even if there is no congestion and in responsive feedback is provided only if congestion occurs. Explicit feedback mechanism has more accurate control but comprises both communication and computation overhead.

Implicit feedback- No separate messages are sent compared to explicit strategy and only based on local observations, such as Round trip Time (RTT) or acknowledgement behavior the source deduces the existence of congestion. It has minimum overhead but less accurate control.

2) *Control technique:* The source is assigned a maximum rate of transmission (the capacity of the access link), and a limit on the length of time it can send data at that rate. The ability of the source to send a burst of data is controlled by these parameters.

Window (or credit) based control - In window-based control, the burst size is basically the window size. Each source has a maximum limit (called as window size) on the number of unacknowledged packets, which can be outstanding at a given time. Window size is the only parameter that is used to specify the control of data.

Rate based control - In the rate-based method, the maximum burst size can be set independently of the rates, thus allowing more control of the data flow. More than one parameter like average transmission rate, maximum rate of transmission (the capacity of the access link) are used to specify the control of data.

3) Control level strategy: This strategy depends on where the control was done either only in end systems or not.

End-to-End control - Congestion control mechanisms are designed into the end points (source and the destination) which respond to congestion and behave appropriately. It is very sensitive to RTT and may suffer from long control delay.

Hop-by-Hop control - Alternatively control was done by each successive node along the data path (i.e., between the source and destination). Here control delay is smaller but node complexity increases.

It is analyzed [2] that end to end congestion control approaches are simple than Hop by Hop approach as they require no changes to the network. But end system cannot acquire the actual available resources but only an estimation of the network characteristics. As Data loss or increased delay is the only sign of network congestion the end systems react to the congestion situation when it is too late. Hop-by-hop congestion control mechanisms [3] [4] have enhanced performance on congestion avoidance, link utilization and network delay. It is expected that better results may be obtained if both principles are combinely used. Research is going on in combining the end-to-end and hop-by-hop principles as well as the combination of AIMD-based and modeling-based schemes to combine the pros of all these schemes while avoiding their individual cons. However, the implementation of such schemes may be with some complexity that should be solved.

III. CLASSIFICATION OF CONGESTION CONTROL PROTOCOLS

Congestion control protocols are categorized into four major classes [5] which are as follows.

A. Window -Based protocols

They are built based on congestion window-based technique. In this technique a congestion window is used at the sender or receiver side. For each packet a slot in that window is reserved. The slot becomes free only when the sent packet is acknowledged. Window-Based protocols allow transmission only when free slots are available. The size of window increases in the absence of congestion and upon presence it suddenly decreases to limit the amount of transmission.

B. Rate -Based protocols

They are built based on the rate adaptation technique. In this technique congestion is indicated by some feedback methods and the rate of transmission is adjusted accordingly.

C. Unicast congestion control protocols

The unicast congestion control protocols adopt Single rate congestion control mechanism. The rate of transmission can be adjusted depending upon the status of the receiver in unicast scheme. This Single-rate congestion control mechanism can also be used for multicast transmission if the sender sends the data with same rate to all receivers of the multicast group in the network.

D. Multicast congestion control protocols

Multi-rate congestion control uses the layered multicast approach where sender's data is divided into different layers, each layer corresponding to a group in a multicast. The receiver can join more number of multicast groups if permitted by the bottleneck in the path to sender. The more number of groups the better the quality of the data. For example in multicast video sessions the more the number of groups the receiver subscribes in, more layers the receiver receives, more better the quality of video is. With this mechanism, congestion control is attained by the group management and routing mechanisms of the basic multicast protocol.

IV. CONGESTION CONTROL PROTOCOLS

Various protocols exist for controlling congestion in multimedia streaming. Some of them are listed below.

A. TCP and UDP

Transmission Control Protocol (TCP), the most widely used transport protocol is not suitable for streaming applications in mobile ad hoc wireless network. This is because of the fact that TCP infers a missing packet as a sign of network congestion which is not always true for mobile ad hoc networks. Packet loss on wireless links are much higher than the wired links due to increased mobility of nodes, channel bit errors, medium contention and frequent path failures. It is acceptable fact that packet losses may be due to congestion or wireless link errors. Even though packet losses if not due to congestion TCP protocol invokes only congestion control algorithm and halves its transmission rate. Such a drastic change in transmission rate without any need affects the performance of the streaming applications. TCP provides strict reliable service and flow control mechanisms but TCP's retransmission scheme may not be needed for loss tolerant multimedia streaming applications.

User Datagram Protocol (UDP) is also used in multimedia streaming applications but not efficiently as it does not encompass any congestion control mechanism. As a result, the congested network may cause substantial performance degradation.

Thus basic TCP and UDP without any enhancement cannot be used to achieve good QoS in multimedia streaming applications. An enhancement in TCP, QoE aware POMDP-based congestion control algorithm, referred to as MOS-TCP [6], reveals an enhanced performance when transporting multimedia applications, specifically over a wireless path. This algorithm enhances TCP to suit for networks containing wireless links. The goal of the MOS-TCP algorithm is to control the end-to-end congestion and to maximize the QoE, where the packets lost due to congestion or randomly due to wireless link errors are distinguished effectively and resolved.

B. SSVP

Scalable Streaming Video Protocol SSVP [12] is an end-to-end protocol with the TCP-friendliness feature. It works as a payload on UDP. SSVP utilizes the AIMD mechanism as that of TCP and controls the sending rate by tuning the inter-packet gap IPG. SSVP handles both AIMD and IPG with great care of smoothness and achieves remarkable performance in transmitting real time video under several network variations. In case of clumsy network conditions a layered adaptation mechanism is applied using the receiver buffering capability. This mechanism adapts the video quality to the long-term variations in the available bandwidth. Change of layers may however provide an adverse effect on the end user perceived video quality. But based on the receiver buffer status and bandwidth availability SSVP sends a refinement layer and avoids unnecessary layer changes. Thus SSVP improves the viewer-perceived video quality. SSVP can perform better in transmitting a multimedia data even under limited bandwidth constraints.

C. SCTP

Stream Control Transmission Protocol is a general purpose transport layer protocol. It provides reliable service and flow control mechanisms like TCP and also like UDP, it supports unreliable transmission, which is called SCTP partial reliability (PR-SCTP) [7]. It can make a distinction between the levels of reliability provided to messages. The main feature of SCTP is providing multi-streaming and multi-homing services for a single connection. Hence SCTP is selected for distribution of multimedia data over heterogeneous wireless networks.

Various SCTP-based Single Path Transfer (SPT) and Concurrent Multipath Transfer (CMT) solutions[8] improve the quality levels perceived by the end-user during multimedia delivery over wireless networks as analyzed [9][10]. Partially reliable concurrent multipath transfer protocol has been proposed in [11] which combine the features of SCTP partial reliable transmission with concurrent multi transfer and prioritized stream transmission to achieve better results in multimedia streaming applications. Area of research in SCTP focuses on the designing of a toolset to investigate the performance of Multimedia over SCTP.

D. SMCC

The Streaming Media Congestion Control Protocol [16] works on the basis of bandwidth estimation concept. The rate of data transmission in a connection is adjusted according to the dynamic share of bandwidth of that connection. It overcomes the limitation in TCP by avoiding oscillations in the rate of transmission as slow start phase is not followed here .It follows the algorithms like TCP Westwood for bandwidth estimation. SMCC does not use congestion window like TCP but adopts the linear bandwidth probing of TCP through adjusting its sending rate to avoid congestion. Hence it is suitable for streaming media applications as a congestion control protocol. SMCC also exhibits fairness as a number of SMCC flows can share the available bandwidth equally. Also it is friendly to TCP New Reno protocol. SMCC is also robust in responding to packet losses even if it is not due to congestion but due to random errors in wireless connections. Because of all these features SMCC seems to be advantageous in the growing Internet wireless access.

E. DCCP

Datagram Congestion Control Protocol (DCCP), [13] an unreliable transport protocol like UDP but incorporates end-to end congestion control. It implements a congestion-controlled unreliable flow of datagram for multimedia streaming applications. Also like SCTP it is enhanced to support partial reliability PR-DCCP. The attracting features are having different acknowledgement formats and supporting different congestion control mechanisms in two end points.ie. Data packets sent from one end (say x) to another end (say y) can use one congestion control identifier (CCID 2) and data packets sent from y to x can use other congestion control identifier (CCID 3).

A Data Dropped option makes one endpoint to declare that a packet was dropped either because of congestion or due to any wireless link errors. This facilitates research into more appropriate rate-control responses for the non-network congestion losses.

Existing congestion control algorithms as in TCP and DCCP are naturally data-oriented. They may not be suitable for video transmission as the characteristics of video data are not explicitly considered. It is analyzed that a new dimension of freedom would be open when the video codec (i.e., encoder rate control) and the transport layer (i.e., congestion control) [14] are jointly designed effectively for high-quality streaming videos.

F. TFRC

The TCP-friendly rate control TFRC [15], an enhanced version of TFRCP uses the complex TCP rate adaptation equation like TFRCP and also adopts a simpler method for gathering its required parameters. TFRC can support unicast communications and multicast communications with some enhancement in it. TFRC estimates the loss rate based on the loss intervals, covering the number of packets between successive loss events. In TFRC the sender passes through the slow start phase like TCP until maximum available share of bandwidth is reached without losses and when a loss event occurs this phase ends up. RTT is measured in TFRC in the standard way and for every RTT; the receiver updates its parameters and sends a state report to the sender. The sender computes its new fair rate and adjusts its sending rate accordingly. In case of some environments where complex TCP equation is not applicable TFRC can also use the delay-based congestion control model via tuning the IPG. It can also be run using linear throughput equation. But over wireless networks the performance of TFRC is found to be degrading and still research is going on in enhancing the performance of TFRC in wireless environment.

G. MTFRCC

Media and TCP-Friendly Rate based Congestion Control MTFRCC [17] is designed to provide scalable video streaming over Internet. MTFRCC applies utility-based model that uses the rate distortion function as the application utility measure for obtaining better video quality. It also applies a two timescale approach of rate averages (long-term and short term) for satisfying both media and TCP-friendliness. Even in different congestion levels it is found to be superior in smoothness. It has been tested in the situations of sudden changes in the available bandwidth regarding its responsiveness and aggressiveness. MTFRCC had lower oscillations in the rate of transmission. Compared with TFRC it is shown that MTFRCC performs better for various congestion levels, including an improvement of the overall video quality.

H. WCMT-SCTP

A new transport protocol [19] that solves the receiver buffer blocking problem during congestion. This SCTP based CMT protocol transmits data chunks of the same stream on the same path in multi hop networks. Even though it improves the throughput by taking advantage of multipath resources it doesn't deal with prioritized stream transmission and time constrained data.

I. PR-CMT

This protocol PR-CMT [20] overcomes the drawback of WCMT-SCTP by combining the techniques of CMT concurrent Multipath transfer, PR-SCTP partially reliable transmission and prioritized stream transmission. QoS is guaranteed on a per stream basis.

J. RTP/RTCP

RTP protocol guarantees real time data transmission where RTCP provides reliable delivery mechanism for transmitting data. LDA (loss detection algorithm) utilizes the delay from transmitting end to receiving end to distinguish the status of packet loss. It is shown [18] that LDA makes use of RTP/RTCP to overcome the defect produced when TFRC is used to transport video in wireless network

The limitation is it is not adapted to deal with high handoff frequency and high packet loss rate for data with limited lifetime and different importance.

V. LIMITATIONS IN EXISTING CONGESTION CONTROL

SOLUTIONS FORMULTIMEDIA STREAMING Multimedia streaming over IP networks is eminent, yet challenging problem. Various solutions have been proposed to attain good performance in supporting delay sensitive, loss-tolerant multimedia applications. But still several limitations exist in current approaches. [21]

Optimization of one QoS parameter may adversely affect the other QoS parameters. For example Model based approaches tends to optimize bandwidth utilization but they don't consider distortion impact, delay deadline thereby resulting in poor video quality. Flow based approaches apply rate distortion optimization, forward error correction, frame dropping ,considering only the average and peak rate of the flow and not on the delay deadline of each packet or interdependencies existing among packets of multimedia applications. They fail to maximize the multimedia application performance by providing only suboptimal solutions for multimedia transmission. Moreover in many existing approaches control decisions are taken based on the current network status considering only the instantaneous multimedia quality and not on the long term multimedia quality in a foresighted manner.

Still efficient loss detection and recovery techniques which are robust to change over is lagging.[22] Research is going on in investigating new loss detection and recovery techniques, dynamic shared bottleneck detection techniques for congestion control and algorithms for scheduling traffic on multiple paths. Cross-layer congestion control strategy [23] is efficient where the MAC layer is video-coding aware and through congestion/distortion optimization it adjusts its transmission parameters. This approach even with the limited information available at each node is providing a reliable estimation of congestion and distortion. Current research is focusing on a cross-layer paradigm, which has been pointed out as a needed shift of perspective in protocol design for multimedia transmission over wireless networks.

VI. CONCLUSION AND FUTUREWORK

Several algorithms for controlling congestion in streaming media traffic over internet have been implemented and tested in last few years. In this paper we presented a survey on recent trends and progression in the area of congestion control for multimedia streaming.

We discussed the classification of congestion control algorithm 0

Rhythms based on the network awareness, various media content aware congestion control protocols with its pros and cons and finally presented the limitations in existing congestion control solutions for multimedia streaming. With an evolving research area there are many issues to be solved in the area of multimedia streaming congestion control. There is no standard method for comparing the congestion control protocols .There is no benchmark for the protocols used in media traffic. It seems that at present there is no single protocol exists that can act as the standard protocol for multimedia streaming. An algorithm that can solve all the problems of congestion control in streaming applications is still in the stage as castle in the air. More research work is needed to solve these issues efficiently. Our future work is to develop a novel protocol that can reach the stage of maturity to be the standard protocol for media streaming.

REFERENCES

[1] Shilpa Kaura, A. K. Vatsa, “A Novel Architecture and Mechanism form Congestion Control in High Speed Network”, International Journal of Next-Generation Network, March-2011.

[2] Qian Wang, Keping Long, Shiduan Chang, Runtong Zhanf, “TCP Friendly Congestion Control Shemes in the Internet”, IEEE Conference 2001.

[3] P.P Mishra and H Kanakia, “A Hop by Hop rate based Congestion Control scheme”, ACM Sigcomm, 1992.

[4] Anita kanavalli, Jayshree M, P Deepa Shenoy, Venogopal K R, L M Patnaik, “Hope by Hope Congestion Control Algorithm for Adhoc Networks”.

[5] Dr E. Chandra, B. Subramani, “A Survay of Congestion Control” Global Journal of Computer Science and Technology Vol. 9 Issue 5 (Var 2.0), January 2010.

[6] Oussama Habachi, Yusuo Hu, Mihaela van der Sachar, Yezekael Hayel, and Feng Wu, “MOS-based Congestion Control for Conversational Services in Wireless Environments”, IEEE Journal on Selected Areas in Communication, VOL. 30, No 7, August 2012.

[7] T. Maed, M. Kozuka, and Y. okabe, “Reliable steaming transmission Using PR-SCTP”, in 9th Annu. Int. Symp. Appl. And Internet, Washing, DC, USA, Jul. 2009.

[8] Changqiao Xu, Enda Fallon, Yuansong Qiao, lujie Zhong, and Gabriel-Miro Muntean, “Performance Evaluation of Multimedia content Distribution Over Multi-Homed Wireless Network”, IEEE Transections on Broadcasting, Vol 57, No 2 June 2011.

[9] C. Q. Xu, Y. S. Qiao, E. Fallon, and G.-M. Muntean at all., “Comparative study of real-time multimedia transmission over multihoming transport protocols”, in 11th IFIP/IEEE Int. Conf. Manage. Multimedia Mobile Netw. Serv.(MMNS), Samos Island, Greece, Sep. 2008.

[10] C. Q. Xu, E. Fallon, Y. S. Qiao, and G.-M. Muntean et al., “Performance evaluation of distributing real-time video over concurrent multipath”, in Porc. IEEE Wireless Communication. Netw. Conf. (WCNC), Budapest, Hungary, Apr. 2009.

[11] C.M.Huang and M.S.Lin, “Multimedia streaming using partially reliable concurrent multipath transfer for multihomed networks”, IET Commun., 2011

[12] Panagiotis Papadimitriou, Vassilis Tsaoussidis, “Ssvp: A Congestion control scheme for real-time video streaming”, Computer Networks, vol. 51, no. 15, 2007.

[13] E. Kohler UCLA M.Handley UCL S. Floyd ICIR, “Datagram Congestion Control Protocol (DCCP)”, the Internet Society (March 2006).

[14] Yingsong Huang, Shiwen Mao, Scott F. Midkiff, “A Control-Theoretic approach to Rate Control for Streaming Video”, IEEE Transactions on Multimedia, Vol. 11, No. 6, October 2009.

[15] S. Floyd, M. Handley, J. Padhye, and J. Winder, “Equation-based congestion control for unicast applications,” in Proc. ACM SIGCOMM, Stockholm, Sweden, Aug.2000,pp. 43-56.

[16] N.Aboobaker, D. Chanay, M. Gerla, M. Y. Sanadidi, “Streaming media congestion control using estimation”, MMNS 2002, California, Oct.2002.

[17] Jinyao Yan, Kostas Katrinis, Martin May, and Bernhard Planttner, “Media- and tcp-friendly congestion control for scalable video streams ”,0 IEEE Transactions on Multimedia, vol. 8,no.2,2006.

[18] Zhou Zurong Chen Rongzhen, “Research of congestion control in wireless video transmission”, IEEE 2010.

[19] B.Wang W.Feng S.D.Zhang H.K.Zhang,” Concurrent Multipath transfer protocol used in Adhoc Networks”, IET Commun., 2010.

[20] C.M.Huang M.S.Lin,” Multipath Streaming Using Partially reliable Concurrent Multipath transfer for multimedia Transmission “, IEEE June 2012.

[21] Hsien-Po Shiang and Mihaela van der Schaar, “A Quality centric TCP-Friendly Congestion Control For Multimedia Transmission”, IEEE June 2012

[22] Preethi Nararajan and Fred Baker, Paul D. Amer and Jonathan T. Leoghton “SCTP: What Why How”, IEEE 2009

[23] Claudio Greco, Marcho Cagnazzo, Beatric Pesquet-Popescu, “Low-Latency Video Streaming With Congestion Control in Mobile Ad-Hoc Network”, IEEE Transection on Multimedia, Vol 14, No 4, August 2012

