International Journal of Innovations & Advancement in Computer Science
IJIACS
ISSN 2347 – 8616
Volume 4, Special Issue
March 2015

Comparative Survey on Reliable Video Transmission in MANET Using Node Stablization

Tanu Gupta[1]
M.Tech Student,
PDM College of Engineering & Technology, B'Garh,
Ajay Dureja[2]
Assistant Professor,
PDM College of Engineering & Technology, B'Garh,
Kavita Khanna[3]
Associate Professor,
PDM College of Engineering & Technology, B'Garh

Abstract
The purpose of this paper is to carry out study on techniques for reliable video transmission in MANET. Various algorithms are defined that were used by previous investigators. In this paper, we have reviewed, analyzed and a tabulated summary work is carried out. Related work shows various implementations carried out previously. This paper focuses on utilizing the bandwidth of the network so that when videos in the form of packets are transmitted in MANET then video packets does not get lost & a reliable video transmission is achieved. Reliable video transmission can be achieved by node stabilization parameters. Parameters will solve QoS problem even in low bandwidth.
Keywords: MANET, Packet, Bandwidth, streaming, congestion, TCP, video etc

1. INTRODUCTION

1.1 What is MANET?

MANET **(Mobile Adhoc Network)** is collection of communication node that wishes to communicate with each other, but has no any fixed infrastructure and pre defined topology of wireless links. Every node is free to move anywhere, anyplace, anytime. Any node can join and leave in network. Each of the nodes has a wireless interface to communicate with each other. These MANET networks are fully distributed, and can work at any place without the help of any fixed infrastructure as access points or base stations.

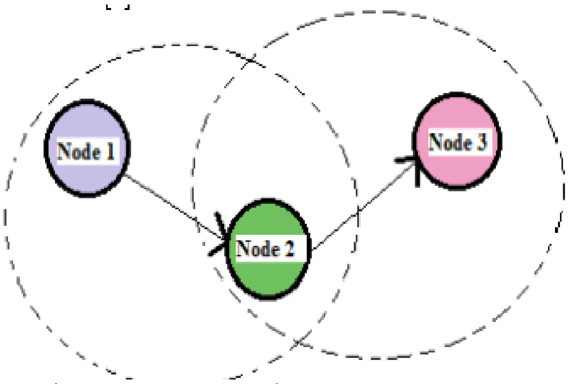

Fig. 1 MANET Network VIDEO STREAMING

Fig.1 shows a simple ad-hoc network with 3 nodes. Node 1 and Node 3 are not within range of each other. However the Node 2 can be used to forward packets between Node 1and Node 3. The Node 2 will act as a router and these three nodes together form an ad-hoc network.

Mobility is core functionality in network. In network, router perform task of routing. It is also different form infrastructure wireless network, in which special node known as an access point manages communication among other nodes. In network, topology can be dynamic and Unpredictable.

Mobile ad hoc networks (MANETs) are suitable for applications in battlefield communications, disaster rescue, and inimical environment monitoring, where fixed wired infrastructure is unavailable. In most of these scenarios, reliable data transfer is required. It is well known that transport control protocol (TCP) has been well tuned to provide such services in traditional wired network environment. Due to its wide use in the Internet, it is desirable that TCP remains in use to provide reliable data delivery for communications within MANETs and for those across MANETs and the Internet.

Types of MANET:

- Vehicular Ad hoc Networks (VANETs) are used for communication between vehicles and roadside equipment. Intelligent vehicular ad hoc networks (InVANETs) are a kind of artificial intelligence that helps vehicles to behave in intelligent manners during vehicle-to-vehicle collisions, accidents.
- Smart Phone Ad hoc Networks (SPANs) leverage the existing hardware (primarily Bluetooth and Wi-Fi) in commercially available smart phones to create peer-to-peer networks without relying on cellular carrier networks, wireless access points, or traditional network infrastructure. SPANs differ from traditional hub and spoke networks, such as Wi-Fi Direct, in that they support multi-hop relays and there is no notion of a group leader so peers can join and leave at will without destroying the network.
- Internet based mobile ad hoc networks (iMANETs) are ad hoc networks that link mobile nodes and fixed Internet-gateway nodes. For example, multiple sub-MANETs may be connected in a classic Hub-Spoke VPN to create a geographically distributed MANET. In such type of networks normal ad hoc routing algorithms don't apply directly.
- Military / Tactical MANETs are used by military units with emphasis on security, range, and integration with existing systems. Common waveforms include the US Army'sSRW, Harris's ANW2 and HNW, Persistent Systems' Wave Relay, Trellisware's TSM and Silvus Technologies' Stream Caster.
- A mobile ad-hoc network (MANET) is an ad-hoc network but an ad-hoc network is not necessarily a MANET.

1.2 Video Transmission in MANET

Mobile Video is a huge portion of the traffic on mobile networks today and is expected to increase to about 50-70% of the total traffic on a mobile network. The type of video content being consumed on the mobile network is improving as well, moving from the low quality UGC (User Generated Content) to high quality commercially produced content. The main reason for the increase in quality of video on the mobile network is because of the increased video processing capabilities of the mobile devices and the higher bandwidth available on the mobile networks to deliver the content.

Video is an essential media for communication. Streaming is the process of playing out a file even as it is downloading. It is a combination of video, voice and animation. Multimedia data has different characteristic as compared to data traffic. Its packet size is quite larger than data, so there are various purposes to built protocol for streaming over network. Streaming media may be either real-time or on-demand. On demand streams are stored on the server and based on the user requirement content is transmitted. Then, user may play video or may download the video for viewing purpose. Real time stream are only available at a particular time. For example, when the event is occurring and user can record the video. Video transmission over wireless network requires link reliability.

Reliable video Transmission:
Videos are sent in the form of packets. When packets are sent over the network completely or successfully without any loss or delay of packets then that video transmission is Reliable video transmission.
Issues regarding video streaming in MANET are:
Topology Changes: The node mobility leads to continuous changes in topology, which means that routes may be formed and broken rapidly. When a route breaks, the discovery of a new route will most likely introduce delays, which will affect the quality of an ongoing media stream. In addition, the topology change may introduce new bottleneck links in the network path, leading to a reduction in bandwidth.
Resource Constraints: The devices participating in a MANET will generally be small devices, which imply limited processing power, memory and storage capacity. Being small mobile devices, they will normally be battery. There are many challenges which are same as MANET network. So, it is required to overcome the challenges. Powered, which means energy consumption must be kept at a minimum. Wireless communication will often mean limited bandwidth, and as mentioned, the nature of wireless communications means that this bandwidth is shared by all devices in the surrounding area. Additionally, an increase in network traffic places additional load on the nodes in the network, which in turn increases energy consumption.
Lack of Fixed Infrastructure: The lack of a fixed infrastructure requires that nodes function as routers in the network. This can introduce large bottlenecks, if a lot of responsibility is assigned to a node with very limited resources.
Internet video is hugely popular across all devices – increasingly so on mobile devices. YouTube reported more than 1 trillion video views in 2011, over 20% of which were on mobile devices. In addition to Smartphone's, tablets have seen a large uptake by consumers in recent years with the success of Apple's iPad, and various manufacturers' equivalents. Even today's vehicular technology incorporates video streaming enabled devices as standard on most models of cars.

1.3 NODE STABLIZATION

Node stabilization means how much a node is stable to handle load. Node stabilization is important to manage congestion in a network. If a node is stable then it is reliable & it can handle load.
Node stabilization depends on various parameters:

- Response Time: It is time which a node takes to respond on load.
- Mobility Ratio: is movement of nodes in a fixed interval of time.
  Preferred mobility ratio is min for a node to be more stable.
- Load on a Node – Packet / Data load on a particular node

2. RELATED WORK

2.1 "Predicting mobile network bandwidth fluctuation to enhance video stream service quality".

Amanda Peart, Andrew Lockett Mo Adda in their paper focuses on bandwidth fluctuation & QoS (Quality of Service) problem. It describes a solution to the problem of sudden drop in bandwidth of a mobile network when any mobile network user moves from Strong Signal area (SSA) to Weak Signal Area (WSA) & QoS problem in video downloading. Proposed solution is based on predictions i.e. location of active mobile user & based on dynamic limiting technique [1].

2.2 "On the Influence of Network Impairments on YouTube Video Streaming".

Arkadiusz Biernacki, Florian Metzger, Kurt Tutschku conducted an experimental evaluation of HTTP based video transmission focusing on how they react to packet delay & loss. This shows that how long video playback is stalled & how often re-buffering events take place. They revealed threshold levels for the packet delay, packet losses & network throughput which should not be exceeded in order to preserve smooth video transmission [2].

2.3 "Network friendly transmission control for progressive download over TCP".

Hiroyuki Hisamatsu , Go Hasegawa , Masayuki Murata first investigate the data transfer mechanisms of the current video streaming services using TCP & show that they perform data transfer at much higher rates than the video playback rate. Then, a new transfer mechanism for video streaming over TCP is proposed that controls the data transfer rate based on the network congestion level & the amount of buffered video data at the receiver. Results show a low frequency of buffer underflow at the receiver & a lack of excessive bandwidth "stealing" from competing traffic [3].

2.4 "Design, Implementation and Evaluation of Congestion Control Mechanism for Video Streaming".

Hiroki Oda, Hiroyuki Hisamatsu, Hiroshi Noborio introduced a new transport-layer protocol, called TCP Stream that solves the problem of TCP in video streaming. TCP Stream performs a hybrid congestion control that combines the loss-based congestion control, which uses packet loss as an index of congestion, and the delay-based congestion control, which uses delay as an index of congestion. Results show that TCP Stream transmits data at the adjusted rate, unlike TCP NewReno, and does not steal bandwidth from of other network traffic [4].

2.5 "Measuring the Quality of Experience of HTTP Video Streaming".

Ricky K. P. Mok, Edmond W. W Chan & Rocky K. C Chang describes relationship among three levels of quality of service (QoS) of HTTP video streaming: network QoS, application QoS & user QoS (i.e. QoE). This describes the correlation between the application QoS & the network QoS using analytical models & empirical evaluation. It concludes that the frequency of rebuffering is the main factor responsible for the variations in the QoE [5].

2.6 "Application Flow Control in YouTube Video Streams".

Shane Alcock , Richard Nelson presents the results of an investigation into the application flow control technique utilized by YouTube. A block sending algorithm is used & it examine how the block sending algorithm interacts with the flow control provided by TCP and reveal that the block sending approach was responsible for over 40% of packet loss events in YouTube flows in a residential DSL dataset and the retransmission of over 1% of all YouTube data sent after the application flow control began. Results shows that changing YouTube block sending to be less bursty would improve the performance and reduce the bandwidth usage of YouTube video streams [6].

2.7 "Trickle: Rate Limiting YouTube Video Streaming".

Monia Ghobadi, Yuchung Cheng, Ankur Jain, Matt Mathis introduces TRICKLE, a server side mechanism that uses TCP to rate limit **You** Tube video streaming. This paces the video stream by placing an upper bound on TCP's congestion window as a function of the streaming rate & the round – trip time. Trickle in Europe & India analyzed the impact on losses, bandwidth, RTT & video buffer under run events. Results were quite good as it reduces the average TCP loss rate by up to 43% & the average RTT by up to 28% while maintaining the streaming rate requested by the application [7].

2.8 "Bandwidth Estimation Schemes for TCP over Wireless Networks".
Antonio Capone, Luigi Fratta, Fabio Martigon in their paper analyzes the problem faced by every bandwidth estimation algorithm implemented at the sender side of a TCP connection. In this a new bandwidth estimation scheme TIBET (Time Interval based Bandwidth estimation Technique) is introduced that can be implemented within the TCP congestion control procedure, modifying only at the sender side of the connection. New Algorithm enhances the TCP source performance over wireless links [8].

3. COMPARISION OF VARIOUS SOLUTIONS PROPOSED PREVIOUSLY
A tabulated summary is described for various implementation proposed previously. Each paper worked on a specific algorithm & a specific solution is defined for each conclusion of various implementations. Main focus will be on the future work of the base paper.

Table1

S.No	Paper Name	Year	Author	Worked On	Algorithm Used	Tools Used	Conclusion	Future Work
1	**Predicting mobile network bandwidth fluctuation to enhance video stream service quality**	2013	Amanda Peart , Andrew Lockett , Mo Adda	It proposed a solution that helps counter this mobility problem by attempting to foresee a user entering a WSA, and dynamically rate-limiting other nearby best-case users to increase available bandwidth to said user. Predictions are based on active user location information, and Mobile Network Coverage Map (MNCM) queries.	Location Based Algorithm-Client Side Algorithm , Server side Algorithm	....	A new video stream service framework is proposed which uses dynamic rate limits to impose limitations on best-case users to improve the QoS for users predicted to enter a WSA.	The work can be extended to consider multiple users in a given area predicted to enter a WSA at a similar time, and how to calculate more appropriate rate-limits for such cases. One possible solution for this would be to look at other nearby SSAs for more best-case users in order to further disperse the required rate-limitation.
2	**Network friendly transmission control for progressive download over TCP**	2012	Hasegawa , Masayuki Mu	A new transfer mechanism called non bandwidth inrusive video streaming over TCP is introduced that controls the data transfer rate based on the network congestion level & the amount of buffered video data at the receiver.	....	ns - 2 simulator	It results into a mechanism has two characteristics lacked by current video streaming over TCP i.e. 1) a low frequency of buffer underflow at the receiver. 2) a lack of excessive bandwidth "stealing" from competing traffic.	Plan can be extended to operate solely by sender - side application. Different data transfer control can be preferred to increase in transfer rates for continuous video playback acts to worson the network cpngestion & exacerbates video playback interruptions.

3	**On the influence of Network Impairments on Youtube Video streaming**	2012	Arkadiusz Biernacki, Florian Mezger	An experimental evaluation of HTTP based video transmission focusing on how they react to packet delay & loss.	Firefox playback (re)-start decision algorithm	....	It shows how long video playback is stalled & how often re-buffering events take place. It revealed threshold levels for the packet delay , packet losses & network throughput which should not be exceeded in order to preserve smooth video transmission.	....
4	**Measuring the Quality of Experience of HTTP Video streaming**	2011	Ricky K.P Mok , Edmond W.W. Chan	Investigation of relationship between three levels of quality of service of HTTP video streaming : network Qos , application Qos , user Qos. It works on coorelation b/w application & network Qos using empirical evaluation & analytical models and to evaluate the relationship b/w application Qos & QoE.	....	....	It reveals that the frequency of rebuffering is the main factor responsible for the variations on the QoE.	....
5	**Design , Implementation & Evaluation of congestion control mechanism for video streaming**	2011	Hiroki Oda , Hiroyuki Hisamatsu	A new TRANSPORT LAYER PROTOCOL called TCP STREAM is introduced to solve the problem of TCP in video streaming. TCP stream performs a hybrid congestion control that combines loss based congestion control & delay based congestion control.	....	ns - 2 simulator	TCP stream can utilize open bandwidth when a network is not in a congestion state. When network is in congestion state ,TCP stream transmits data at a adjusted rate & does not steal bandwidth from of other network traffic.TCP need not to change the configuration of operating system or modify the kernel at receiving end, it only need changes at sender side only.	TCP stream can be implemented over the internet to generate new simulation results.
6	**Application Flow Control in YouTube Video Streams**	2011	Shane Alcock,Richard Nelson	Basic properties of YouTube application flow control like block sending interacts with the flow control provided by TCP and reveal that the block sending approach was responsible for over 40% of packet loss events in YouTube flows in a residential DSL dataset and the re transmission of over 1% of all YouTube data sent after the application flow control began.	Block Detection Algorithm	....	Changing YouTube block sending to be less bursty would improve the performance and reduce the bandwidth usage of YouTube video streams.	....

7	**Trickle: Rate Limiting YouTube Video Streaming**	2008	Yuchung Cheng, Ankur Jain, Matt Mathis	Trickle, a server-side mechanism that uses TCP to rate limit YouTube video streaming. Trickle paces the video stream by placing an upper bound on TCP's congestion window as a function of the streaming rate and the round-trip time.	Trickle Algorithm in throttling phase.	....	Trickle reduces the average TCP loss rate by up to 43% and the average RTT by up to 28% while maintaining the streaming rate requested by the application.	....
8	**Bandwidth estimation Schemes for TCP over wireless networks**	2004	Antonio Capone, Fabio martignon	An algorithm TIBET (Time Interval based Bandwidth Estimation Technique) is introduced that can be implemented within the TCP congestion control procedure , modifying only the sender side of a connection.	Estimation algo's- Packet -Pair Algo , TCP vegas Algo , TCP testwood Algo	....	TIBET enhances TCP source performance over wireless networks. The performance of TIBET is analyzed & compared & it provide an upper bound to performance of all possible schemes based on different bandwidth estimates.	....

4. CONCLUSION

This paper concludes to utilize bandwidth to avoid packet loss & congestion problem. Related work shows various implementations carried out previously This study can be used to make a conclusion to utilize the bandwidth of the network so that different packets does not get lost when any user moves from high bandwidth area to low bandwidth area. Each packet in a video will be assigned a bandwidth that is sufficient to send that packet even in weak signal area. Hence, congestion can be controlled & no packet loss occurs when any user moves from a strong signal to area to weak signal area. Proposed work will be based on node stabilization parameters like Response time of a node, Mobility ratio, and Load on a node.

5. FUTURE WORK

This work can be enhanced by working on Network Layer of TCP. Proposed work parameters Mobility ratio, Load on a node can be implemented on Network Layer of TCP. An implementation can also be done by considering more than 1 person moving from SSA to WSA.

6. REFERENCES

[1]. Amanda Peart, Andrew Lockett, Mo Adda, “Predicting mobile network bandwidth fluctuation to enhance video stream service quality”, Science and Information Conference, October 7-9, 2013.

[2]. Arkadiusz Biernacki, Florian Metzger, and Kurt Tutschku, “On the Influence of Network Impairments on YouTube Video Streaming”, Journal of Telecommunications & Information Technology, 2012.

[3]. Hiroyuki Hisamatsu , Go Hasegawa ,Masayuki Murata ,”Network friendly transmission control for progressive download over TCP” , Journal of communications,Vol.7.No.3 ,March 2012.

[4]. Hiroki Oda, Hiroyuki Hisamatsu, Hiroshi Noborio, “Design, Implementation and Evaluation of Congestion Control Mechanism for Video Streaming”, International Journal of Computer Networks & Communications, Vol 3, No. 2, May 2011.

[5]. Ricky K. P. Mok, Edmond W. W. Chan, and Rocky K. C. Chang, “Measuring the Quality of Experience of HTTP Video Streaming”, Department of Computing, The Hong Kong Polytechnic University, 2011.

[6]. Shane Alcock, Richard Nelson, “Application Flow Control in YouTube Video Streams”, 2011.

[7]. Monia Ghobadi, Yuchung Cheng, Ankur Jain, Matt Mathis, “Trickle: Rate Limiting YouTube Video Streaming”, University of Toronto, 2008.

[8]. Antonio Capone, Luigi Fratta, “Bandwidth Estimation Schemes for TCP over Wireless Networks”, IEEE Transactions on mobile computing, Vol. 3. No.2, April-June 2004.