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EFFICIENT WAY OF AVOIDING TIME DELAY AND CONGESTION FOR PERFORMING SECURE MULTIMEDIA TRANSMISSION

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ABSTRACT

The main objective of this paper is to make an efficient way of multimedia transmission using the Tcp based congestion control method.A Quality oriented multimedia transmission is done using the Media Tcp friendly congestion control (MTCC) which implements the process namely Finite Horizon Morkov Decision Process (FHMDP). This process contains algorithms for congestion control , but this process is very complex to implement and this process provides some difficult allocations like congestion window size allocation. Moreover there are some demerits in this process like increase in time delay , decrease in throughput. To overcome this demerits , the proposed method using the Stream Control Transmission Protocol (SCTP) , technique that implemented called as Highspeed Downlink Packet Access (HSDPA) in the proposed system.Using this technique the time delay can be decreased and the throughput can be increased when compare to the FHMDP.

Keywords: Finite horizon Markov Decision Process, HighSpeed Downlink Packet Access, Media Tcp friendly Congestion Control.

1) INTRODUCTION

MOTIVATION FOR THE PAPER

The Internet Protocol (IP) has been selected to provide the necessary interconnection among all wireless and wired existing technologies. However, the use of IP does not solve all drawbacks. Multimedia applications show that current transport protocols like TCP (Transmission Control Protocol) or UDP (User Datagram Protocol) are not good enough to meet the new quality requirements.

TRANSPORT LAYER PROTOCOLS

THE SCTP PROTOCOL

SCTP was originally designed for carrying telephony signaling in IP networks. The transport of SS7 signaling messages has strict requirement s for reliability and timely delivery , and the available transport protocol s before SCTP was developed (UDP and TCP) cannot meet these requirements. Similarly to TCP, SCTP provide s a connection-oriented, reliable transport service with flow control and congestion control.

Message oriented data delivery, which is mo re suitable for the transport of signaling messages, instead of byte-stream oriented delivery. o Multi-streaming, which allows application data to be multiplexed onto one association (the term used by SCTP t o represent a connection). This feature, when used to carry signaling messages, c an separate signaling message s for different calls into different message flows thus avoiding head-of-line blocking. This makes the transport timelier.

Multi-homing (i.e. multiple IP addresses), which allows an established SCTP association to be maintained when the either one of IP addresses used by the communicating endpoints changes. This feature enables endpoints have backup

network interfaces and also decreases the time require red for connection recovery from link failures. This later feature is useful for applications that require high availability.

Unordered delivery, which allows a data message marked as unordered to bypass the ordering mechanism s and be immediately delivered to the upper layer in the receiving endpoint, while all the other norm al data messages are delivered in order.

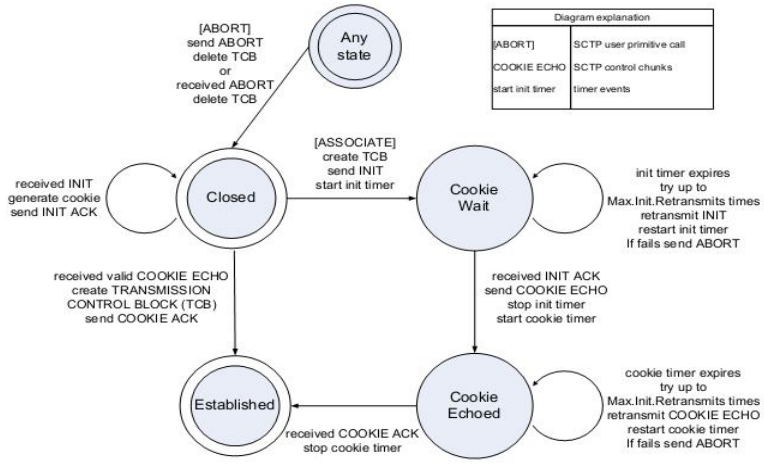

Figure 1a : chunk flow

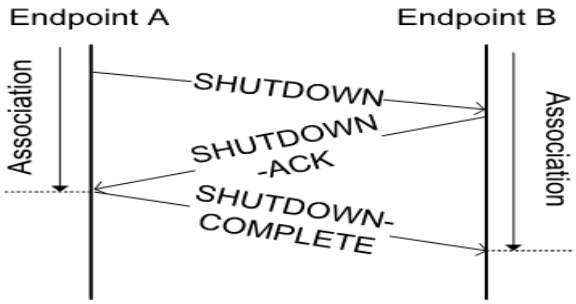

Figure 1b : chunk flow

A cookie mechanism to prevent denial-of-service attacks. Feature extensibility, which allows protocol developers to add ne w features to SCTP Feature extensions is done b y defining new chunk types. When two endpoints start an SCTP association, the variable-length parameters of the INIT messages allow them to negotiate the use of optional features. SCTP Partial Reliability Extension The SCTP Partial Reliability Extension (PR-SCTP) allows an SCTP endpoint to signal the other endpoint of the SCTP association to explicitly move the cumulative acknowledgement pointer forward to abandon one or more messages.

This extension provides a partially reliable data transport to the upper layer application. The use of this extension i s negotiated b y communicating endpoints during the SCTP association establishment and only when both endpoints agree with the use of this extension, t h e PR-SCTP data transport can be used. If both endpoints of an SCTP association agree to the use of this extension, then all the message stream s within the association are partial reliable.

From the perspective of the upper layer that uses the partially reliable data transport, although it is unreliable and some messages might be lost , all the messages except the unordered delivery messages are still delivered to the upper layer in order at the receiving endpoint. Therefore, excluding unordered delivery messages, the data transport provided by PR-SCTP is unreliable but ordered. This is different from the unreliable, unordered transport provided b y UDP or DCCP.

The PR-SCTP specification allows definition of different services provided by PR-SCTP data transport to the upper layer. One service that is already define d is the “timed reliability service”. The upper layer protocol that uses this service can specify the ‘lifetime’ of each message it sends. If by the time the transmission of a certain message in the SCTP protocol stack takes place the message has already ‘expired’ , then instead of sending the message down to the network layer, the S CTP protocol stack sends a ‘Forward-TSN’ control chunk to the other endpoint as a signal to abandon this expired message.

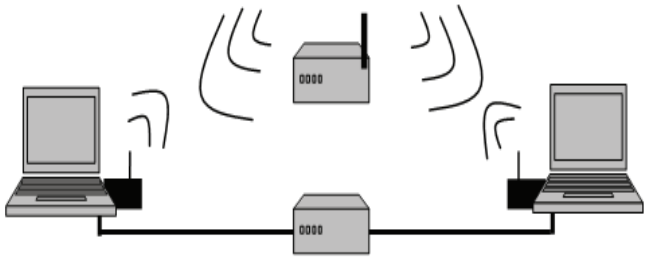

FIGURE 2: WIRELESS TOPOLOGY CONCEPT

SCTP IN WIRELESS NETWORKS:

Seamless mobility is one of the challenges in wireless networks. With the proliferation of new types of wireless access technologies (e.g., Wi-Fi, WiMAX, 3G, vehicular networks, etc.), a user, through his/her mobile device, should be able to change his/her location maintaining the Quality of Service (QoS) performance disregarding the roaming, either horizontal (under the same technology) or vertical (crossing different technologies). SCTP is a competitive solution for mobility due to its multihoming capability. Multimedia transmission is another challenge in wireless networks due to the higher likelihood of packet losses (error-prone channels). In this case, SCTP multistreaming improves the data rate throughput since streams are independently delivered; hence, the multimedia application is less sensitive to packet losses. Finally, some new modifications to SCTP have been presented in the related literature to increase its performance.

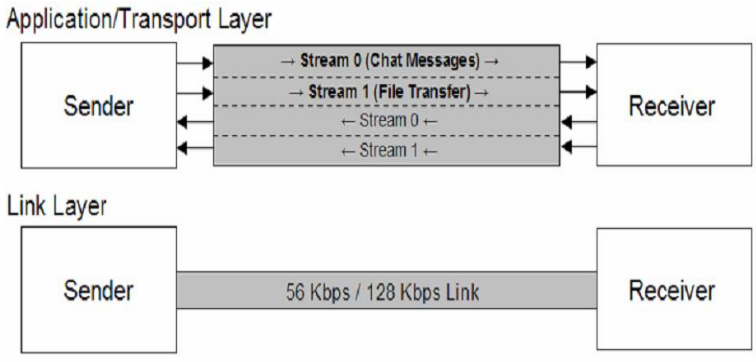

Figure 5 - Simulation Network

Figure 3 : Simulation Network

HIGHSPEED DOWNLINK PACKET ACCESS (HSDPA):

The HSDPA is one of the 3G technology that used in the cellular networks. It is the latest and current technique that implemented in the field of the networks and it has the some efficient features like high data rate transfer and the high increase of bit rate in downlink of sender and receiver. This HSDPA is used in the effective multimedia transmission because due to the high bit rate of data transfer in this cellular networks. Due to this high data transfer the downlink between the sender and the receiver is high and it helps to reduce the time delay and increases the throughput.

It supports Hybrid Automatic Repeat Request which consists of 2 schemes namely called Chase Combining and the Incremental Redunancy.

Chase Combining :

1) Coding is applied to transmission packets

2) Soft combining of original and retransmitted signals is done at receiver before decoding

3) self decodable, time diversity, path diversity are the advantages.

Incremental Redunancy:

Reducing the effective data throughput/bandwidth of a user and using this for another user

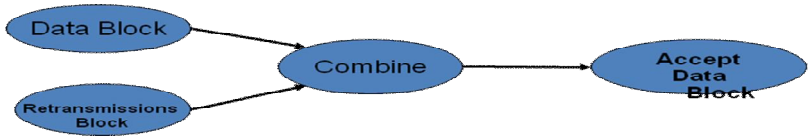

Figure 4 : Chase Combining

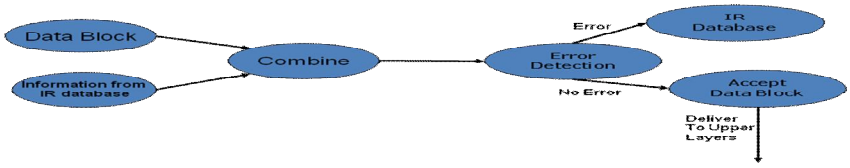

Figure 5 : Incremental Redunancy

2) PAPER DESCRIPTION

The Main Objective of this paper is to efficient way of avoiding the time delay and congestion in the multimedia transmission. The Quality oriented multimedia transmission should done in the Tcp based congestion control activities. For this concept, the example for the Tcp based multimedia transmission are taken from the paper called, "A Quality centric Tcp friendly congestion control for multimedia transmission",IEEE transactions, june 2012. In this paper , Media Tcp friendly congestion control (MTCC) which implements the process namely Finite Horizon Morkov Decision Process (FHMDP). This process contains algorithms for congestion control , but this process is very complex to implement and this process provides some difficult allocations like congestion window size allocation. Moreover there are some demerits in this process like increase in time delay , decrease in throughput.To overcome this demerits , the another method using the Stream Control Transmission Protocol (STCP) , technique that implemented called as Highspeed Downlink Packet Access (HSDPA) in the proposed system. The SCTP is a suitable transport protocol for many connection establishment between the sender and receiver nodes. This protocol supports the multi address (multi homing) and therefore an endpoint may be reached via different and possibly redundant network paths. High speed Downlink Packet Access is one of the 3G method that used in the cellular networks and this technique is very effectively used for transmiting the video streaming and all multimedia files. It has the high rate of data transfer and increases the bit rates in the downlink. This HSDPA technique helps to reduce the time delay and the throughput are increased due to the fast transmission of datas downlink in the sender and the receiver.

It supports the Hybrid Automatic Repeat Request, it consists of two schemes called as chase combining and the Incremental Redunancy.

PROPOSED SYSTEM

In this proposed system , the Stream Control Transmission Protocol (SCTP) is used , in which Highspeed Downlink Packet Access (HSDPA) technique is used , using this method the time delay can be reduced and the throughput will be increased due to the effective features like fast downlink of datas between the sender and receiver. Moreover this technique is effective in transmitting the video streaming and other multimedia files and also the SCTP is a multi streaming and its fully message oriented , it helps to avoid the congestion easily.

PROPOSED SYSTEM DESIGN

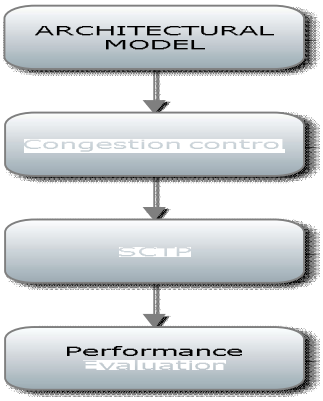

Figure 6 : Proposed System Design

FLOW DIAGRAM

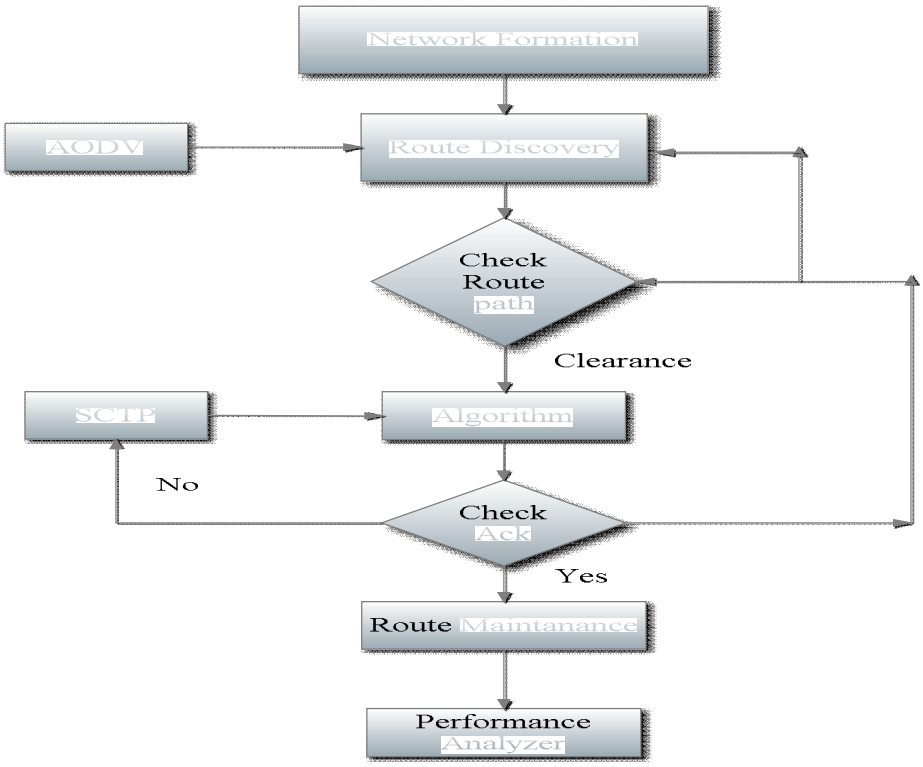

Figure 7 : Flow Diagram of the paper

3) MODULES DESCRIPTION:

The modules included are:
1. Architecture Model.
2. Congestion Control
3. SCTP
4. Performance evaluation.

ARCHITECTURE MODEL

Wireless nodes are created in 100 numbers.
We used ns2 simulator on Linux machine. Because, we focus on the link stability and route lifetime, no route overhead was considered in our simulation. In 870 X 870 m^2area, mobile nodes exist. IEEE 802.11 MAC layer is used. We used square area to increase average hop length of a route with relatively small nodes. Every mobile node is moving based on mobility data files that were generated by mobility generator module.
The transmission range is fixed at 250 units. 20 nodes of them have destinations and try finding routes to their destination nodes. Maximum speed of node is set to 10 m/sec. All nodes do not stop moving, and the simulation time is 500 sec. The number of nodes is varying from 50 to 100.

CONGESTION CONTROL

When a new connection is established with a host on another network, the congestion window is initialized to one segment (i.e., the segment size announced by the other end, or the default, typically 536 or 512). Each time an ACK is

received, the congestion window is increased by one segment. The sender can transmit up to the minimum of the congestion window and the advertised window. cwnd gets larger after every new ACK
MaxWindow :: min (CongestionWindow , AdvertisedWindow) EffectiveWindow = MaxWindow – (LastByteSent - LastByteAcked)
The congestion window is flow control imposed by the sender, while the advertised window is flow control imposed by the receiver. The former is based on the sender's assessment of perceived network congestion; the latter is related to the amount of available buffer space at the receiver for this connection. The sender starts by transmitting one segment and waiting for its ACK. When that ACK is received, the congestion window is incremented from one to two, and two segments can be sent. When each of those two segments is acknowledged, the congestion window is increased to four. cwnd get smaller when loss is detected At some point the capacity of the internet can be reached, and an intermediate router will start discarding packets. This tells the sender that its congestion window has gotten too large.

SCTP
The Stream Control Transmission Protocol (SCTP) is a new IP transport protocol, existing at an equivalent level with UDP (User Datagram Protocol) and TCP (Transmission Control Protocol), which provide transport layer functions to many Internet applications. SCTP has been approved by the IETF as a Proposed Standard .The error check algorithm has since been modified . Stream Control Transmission Protocol (SCTP) is a reliable datagram-oriented IP transport protocol, specified by RFC 2960. It provides the layer between an SCTP user application and an unreliable end-to-end datagram service such as IP. The basic service offered by SCTP is the reliable transfer of user datagrams between peer SCTP users. It performs this service within the context of an association between two SCTP hosts.
The HSDPA is one of the 3G technology that used in the cellular networks. It is the latest and current technique that implemented in the field of the networks and it has the some efficient features like high data rate transfer and the high increase of bit rate in downlink of sender and receiver. This HSDPA is used in the effective multimedia transmission because due to the high bit rate of data transfer in this cellular networks. Due to this high data transfer the downlink between the sender and the receiver is high and it helps to reduce the time delay and increases the throughput.

PERFORMANCE EVALUATION :
The Comparison between the performance of the techniques that used in the both existing and proposed system. The different parameters that compared for the performance are time delay,throughput,packet delievery ratio.Each parameter should be compared in this performance evaluation.

4) CONCLUSION AND FUTURE ENHANCEMENTS
Thus in this paper, the implememtation of the SCTP (StreamControl Transmission Protocol) the technique used is the HSDPA(Highspeed Downlink Packet Access) is very effectively implemented and some parameters are improved when compare to the Finite Horizon Markov Decision Process(FHMDP) which involved in the MTCC , the time delay is decreased and the throughput is increased when compare to the FHMDP in the MTCC.

5) OUTPUT SNAPSHOTS

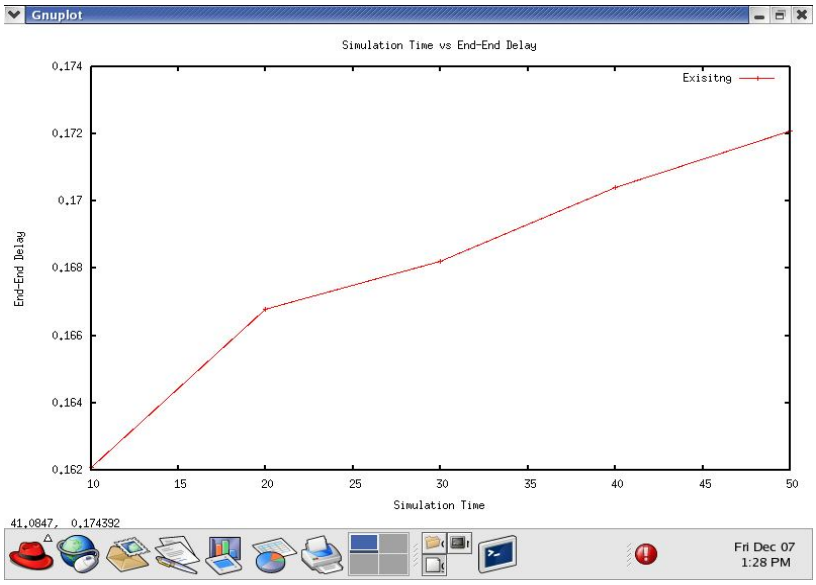

Figure 7a : End to End time delay in Existing System

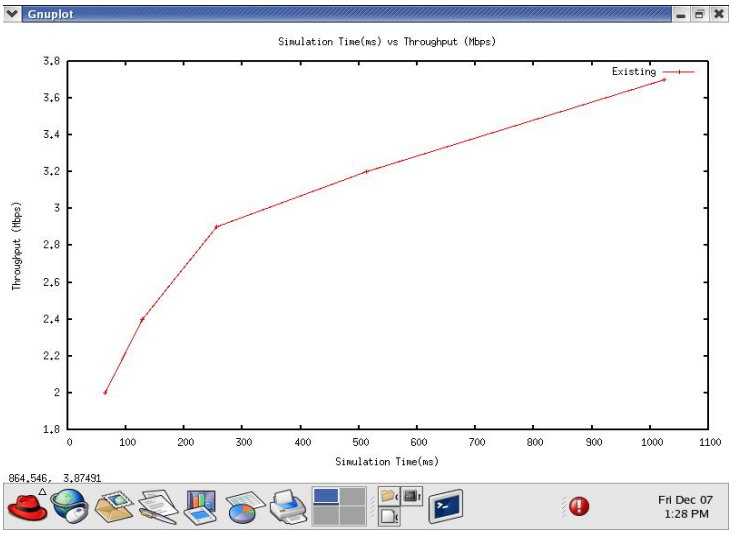

Figure 7b : Throughput in Existing System

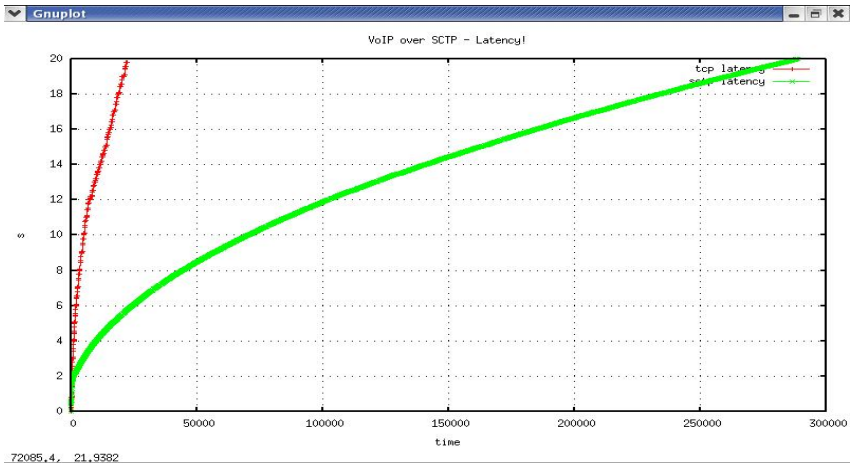

Figure 8a : End to End time delay in proposed system

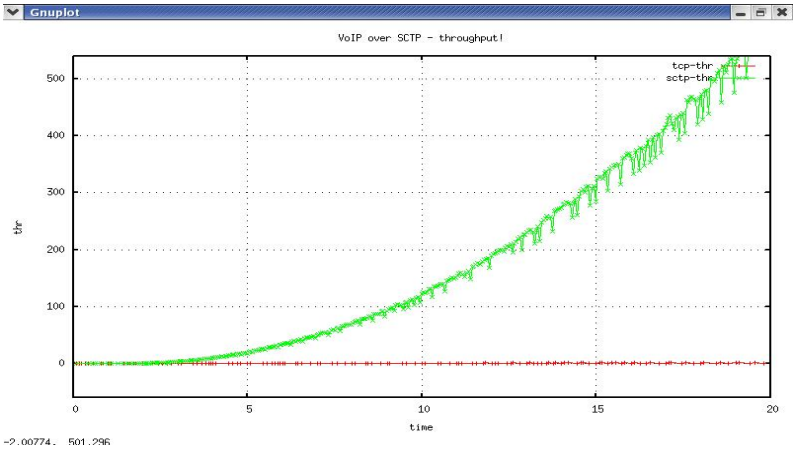

Figure 8b : Throughput in proposed system

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