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Performance Analysis of Rtp and DCCP for Video Traffic over WIMAX

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Abstract-In this project our focus is on the performance of two new protocols like Real Time Protocol(RTP) and Datagram Congestion Control Protocol(DCCP) is to determine which protocol can better meet the Quality of Service (Qos) for MPEG-4 video traffic over WiMAX network. Based on the our simulation results obtained in this analysis it is observed that both RTP and DCCP is better throughput than UDP. For video traffic load of 4Mbps, throughput achieved by both RTP and DCCP is almost 100% without expressing any packet loss. With the increase of video traffic load of 5Mbps and beyond RTP maintains its throughput and it shows minimum packet loss but DCCP losses its performance. Delay behavior of RTP also indicates that its performance is better than the DCCP. According to our simulation results RTP better satisfies the quality of service than DCCP for transport of multimedia (video) traffic. Mobile Multimedia is an attractive and promising application which allows immersive communication and discussion among people at different and distant places. However, its stringent delay and bandwidth requirements limit its scale and spread over current Internet. The Mobile multimedia application has increased enormously. Currently most of the multimedia application use as the main transport protocol, However UDP performance has not been satisfactory various factors, in multimedia application. A number of new protocols are being developed to meet the diverse needs of emerging multimedia applications. RTP and DCCP are two important developments are being considered in this regard. In this project, through simulation, performance of, RTP and DCCP protocols has been analyzed for the transport of MPEG_4 video traffic Over Mobile WiMax as underlying access technology. Considering single cell WiMax network, performances metrics such as throughput, delay, pack loss and jitter have been determined for each of the four protocols in varying WiMAX network topology.

Key words: MPEG-4 video traffic · RTP and DCCP protocols · Mobile WiMax

INTRODUCTION

There are number of wireless technologies available for mobile multimedia applications. Currently most of the multimedia application use UDP transport layer protocol [1]. However UDP performance has not satisfactory WiMax (Worldwide Interoperability for Microwave Access), RTP (Real Time Protocol), DCCP (Datagram Congestion Control Protocol), Qos (Quality of Service). QoS of diverse multimedia application. A variety of new protocols are being developed to meet the diverse needs of emerging multimedia applications [2-7]. DCCP and RTP are two important developments that are being considered in this project. In this project thought simulations, performance of DCCP and RTP protocols has been analyzed for the transport of MPEG-4 video traffic over WiMAX wireless access technology. WiMAX performance metrics such as throughput, delay and jitter has been determined for each of the two protocols in varying WiMAX network topologies. In this project through simulations, DCCP performance is better than RTP in terms of delay and jitter.The telecommunication industry has been through disruptive times, but data networking service revenue has continued to rise. The telecom industry is expected to continue to grow as demand increases for cable and high-speed internet in previously un serviced locations and as local telephone companies upgrade their line in response to increasing competition. In this project used wireless technology access medium is Interoperability for Microwave Access (WiMAX).

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What is WiMAX: WiMAX stands for **Worldwide Interoperability for Microwave Access**. It is a telecommunications technology providing wireless data over long distances in a variety of ways, from point-to-point links to full mobile cellular type access. It is based on the **Wireless MAN** (IEEE 802.16) standard. WiMAX is a highly scalable, long-range system, covering many kilometers using licensed spectrum to deliver a point-to-point connection to the Internet from an ISP to an end user. WiMAX can be used to provide a wireless alternative to cable and DSL for broadband access and to provide high-speed data and telecommunications services. WiMAX can also be used to Connect many Wi-Fi hotspots with each other and also to other parts of the Internet. When using WiMAX device with directional antennas, speeds of **10 Mbit/s at 10 km** distance is possible, while for WiMAX devices with omni-directional antennas only **10 Mbit/s over 2 km** is possible. There is no uniform global licensed spectrum for WiMAX, although three licensed spectrum profiles are being used generally – 2.3 GHz, 2.5 GHz and 3.5 GHz. With WiMAX enabled handsets and laptops coming into the market, people could connect to the fast broadband internet from anywhere, without having to depend on the slow rate mobile network data transfer. You can work on broadband, call friends and colleagues and watch real-time TV from the top of a forest hill station many kilometers away from the access point – without compromising on quality, speed or screen size. WiMAX could connect remote Indian villages to the Internet using broadband. This would avoid hassles in cabling through the forests and other difficult terrain only to reach a few people in remote places. Maintaining such system would also be easy [8]. WiMAX could provide Internet access, voice and IPTV to those areas. With WiMAX, WiFi-like data rates are easily supported, but the issue of interference is lessened. WiMAX operates on both licensed and non-licensed frequencies, providing a regulated environment and viable economic model for wireless carriers.

WiMAX can be used for wireless networking in much the same way as the more common WiFi protocol. WiMAX is a second-generation protocol that allows for more efficient bandwidth use, interference avoidance and is intended to allow higher data rates over longer distances.

Network Architecture: WiMAX has a flexible architecture. The Mobile WiMAX End to End network architecture is based on an All-IP platform, all packet technology and no circuit switch telephony. The open IP architecre gives network operators great flexibility when selecting that work with legacy networks or that use the most advanced technologies and in determining what functionality they want their network to support. They can choose from a vertically integrated vendor that provides a turnkey solution or they can pick and choose from a dense ecosystem of best of breed players with a more narrow focus. The architecture allows modularity and flexibility to accommodate a broad range of deployment options such as small scale to large scale, urban, suburban and rural coverage's, mesh topologies, flat hierarchical and their variant and finally, co- existence of fixed, nomadic portable and mobile usage models.

Tchnologies Employed By WiMAX: WiMAX operates in licensed frequency bands in the range of 2 to 6 MHz. OFDMA is perhaps the most important technology associated with WiMAX. OFDM is a form of Frequency Division Multiplexing, but it has higher spectral efficiency and resistance to multi path fading and path loss compare to other multiplexing methods. It divides the allocated frequency spectrum into sub carriers which are at right angles to each other. This reduces the possibility of cross channel interference thereby allowing the sub – carriers to overlap. This reduces the amount of frequency spectrum required, hence the high spectral efficiency. The reduced data rate of each stream reduces the possibility of inter [9].

Purpose: The purpose of this software requirements specification is to define all the requirements for "performance analysis for video traffic over WiMAX". It also describes design constraints and other factors necessary to provide a complete and comprehensive description of the requirements for the software. It's also identifiers the applications of this project.

Literature Survey

RTP: Real Time Protocol (RTP) is a very popular protocol and in the most widely used for video transmission in internet data application. RTP was developed by the Audio /video Transport working group of the IETF standards organization. RTP is used for transfer of multimedia data send control information and QoS parameters. The real-time transport protocol (RTP) provides end-to-end delivery services for data with real-time characteristics, such as interactive audio and video or simulation data, over multicast or unicast network services. Applications typically run RTP on top of UDP to make use of its multiplexing and checksum services; both protocols contribute parts of the transport protocol functionality. However, RTP may be used with other suitable underlying network or transport protocols. RTP supports data transfer to multiple destinations using multicast distribution if provided by the underlying network. RTP itself does not provide any mechanism to ensure timely delivery or provide other quality-of-service guarantees, but relies on lower-layer services to do so. It does not guarantee delivery or prevent out-of-order delivery, nor does it assume that the underlying network is reliable and delivers packets in sequence. The sequence numbers included in RTP allow the receiver to reconstruct the sender's packet sequence, but sequence numbers might also be used to determine the proper location of a packet, for example in video decoding, without necessarily decoding packets in sequence. RTP consists of two closely-linked parts:

The real-time transport protocol (RTP), to carry data that has real-time properties. The RTP control protocol (RTCP), to monitor the quality of service and to convey information about the participants in an on-going session. The latter aspect of RTCP may be sufficient for "loosely controlled" sessions, i.e., where there is no explicit membership control and set-up, but it is not necessarily intended to support all of an application's control communication requirements.

An RTP sender captures the multimedia data, which are then encoded as frames and transmitted as RTP packets, with appropriate timestamps and increasing sequence numbers. Depending on the RTP Profile in use, the Payload Type field is set. The RTP receiver, captures the RTP packets and may perform reordering of packets, which may have resulted because of the underlying IP network and the frames are decoded depending on the payload format and presented to the end user. The Real –time Transport Protocol (RTP) provides a real –time transport mechanism suitable for unicast or multicast communication between multimedia applications. Typical uses of RTP are for real –time or near real-time group communication of audio and video data streams. An important component of the RTP protocol is the control channel, defined as the RTP control Protocol (RTCP involves the periodic transmission of control packets between group members, enabling group size estimation and the distribution and calculation of session-specific information such as packet loss and round-trip time to other hosts. An additional advantage of providing a control channel for a session the that a third –party session monitor ca listen the traffic to establish network conditions and to diagnose faults based on receiver locations.

S(I) = Timestamp from RTP data packet I

R(I) = Time of arrival for RTP data packet I, expressed in RTP timestamp units, The receiver must use the same clock frequency (increment interval) as the source but need not synchronize time values with the source

D(I) = The difference between the interarrival time at the receiver and the spacing between adjacent RTP data packets leaving the source

J(I) = Estimated average interarrival jitter up to the receipt of RTP data packet I

The value of D(I) is calculated as

$$D(I) = (R(I) - R(I -1)) - (SI) - S(I-1))$$

Thus, D(I) measures how much the spacing between arriving packets differs form the spacing between transmitted packets. In the absence of jitter, the spacing's will be the same and D(I) will have a value of 0. The interarrival jitter is calculated continuously as each data packet I is received, according to the formula

$$J(I)= 15/16\ J(I - 1) +1/16\ | D(I) |$$

Audio and Video Conference: If both audio and video media are used in a conference, they are transmitted as separate RTP sessions. That is, separate RTP and RTCP packets are transmitted for each medium suing two different UDP port pairs and /or multicast addresses. There is no direct coupling at the RTP level between the audio and video sessions, except that a user participating in both sessions should use the same distinguished name in the RTCP packets for both so that the sessions can be associated [10].

DCCP: The Datagram Congestion Control Protocol (DCCP) is a message- oriented Transport Layer protocol. DCCP implements reliable connection setup, teardown, ECN, congestion control and feature negotiation. DCCP is useful for applications with timing constraints on the delivery of data that may become useless to the receiver if reliable in order delivery combined with congestion avoidance is used. The Datagram Congestion Control Protocol (DCCP) is a transport protocol that provides bidirectional unicast connections of congestion – controlled unreliable datagram. DCCP is suitable for applications that transfer fairly large amounts of data and that can benefit from control over the tradeoff between timelines and reliability.

The Datagram Congestion Control Protocol (DCCP) is a transport protocol that implements bidirectional, unicast connections of congestion-controlled, unreliable datagram. Specifically, DCCP provides the following: Unreliable flows of datagram. Reliable handshakes for connection setup and teardown. Reliable negotiation of options, including negotiation of a suitable congestion control mechanism. Mechanisms allowing servers to avoid holding state for unacknowledged connection attempts and already-finished connections. Congestion control incorporating Explicit Congestion Notification (ECN) and the ECN Nonce. Acknowledgement mechanisms communicating packet loss and ECN information. Acks are transmitted as reliably as the relevant congestion control mechanism requires, possibly completely reliably. Optional mechanisms that tell the sending application, with high reliability, which data packets reached the receiver and whether those packets were ECN marked, corrupted, or dropped in the receive buffer. Path Maximum Transmission Unit (PMTU) discovery. A choice of modular congestion control mechanisms. Two mechanisms are currently specified: TCP-like Congestion Control and TCP-Friendly Rate Control (TFRC). DCCP is easily extensible to further forms of unicast congestion control. DCCP is intended for applications such as streaming media that can benefit from control over the tradeoffs between delay and reliable in-order delivery. TCP is not well suited for these applications, since reliable in-order delivery and congestion control can cause arbitrarily long delays. UDP avoids long delays, but UDP applications that implement congestion control must do so on their own. DCCP provides built-in congestion control, including ECN.

DCCP is intended for applications such as streaming media that can benefit from control over the tradeoffs between delay and reliable in-order delivery. TCP and UDP both are not suited for these applications as in case of TCP reliable in-order delivery and congestion control can cause arbitrarily long delays, UDP avoids long delays, but UDP lack congestion control. DCCP provides built-in congestion control, including ECN and ECN Nonce ability. DCCP is suitable for applications that transfer fairly large amounts of data and that can benefit from control over the tradeoff between timeliness and reliability. For example in applications such as streaming media, Internet telephony, videoconferencing and games, all share a preference for timeliness over reliability. That is given a chance to retransmit an old packet or to transmit a new packet arrived; it would choose the new packet. By the time the old packet arrived, it would have been useless anyway: in media applications, users often prefer bursts of static to choppy rebuff ring delay; in games, only the latest position information matters.

DCCP meets all features of modern TCP congestion control protocol, including selective acknowledgments, explicit congestion notification (ECN), acknowledgment verification and so forth, as well as obvious extensions hard to port to TCP, such as congestion control of acknowledgements. Currently DCCP uses TCP-like congestion control mechanism. As in TCP, DCCP implementation should be able to manage congestion control without application aid. DCCP receivers must detect congestion events without application intervention; DCCP senders must calculate and enforce fair sending rates without application cooperation. Any API for sending DCCP packets will support some buffering, allowing the operating system to smooth out scheduling bumps. However, when the buffer overflows-the application's send rate is more than congestion control allows smart application may want to decide exactly which packets should be sent. Some packets are more valuable than others (for example audio data might be preferred to video), or newer packets preferred to older ones. As explained above DCCP provide bidirectional connection: data and acknowledgements flow in both directions. However, many DCCP applications will have fundamentally asymmetric data flow. For example in streaming media almost all data flows from server to client; after the initial connection setup, the client's packets are all acknowledgements.

Simulation Module: In this simulation module we are developing all the nodes are like mobile nodes Or Wireless nodes. This project basically depends on wired cum wireless network. So all the nodes are directly communicates with the other nodes with the help of Base station. And also the nodes should be movement node.

First create the three nodes, like one is sender and other one is receiver. Sender will transmit the packet to destination. And the receiver will retransmit the ack signal to the source. It will communicate with each other depends upon the distance routing. Connect the channel between the two nodes. Otherwise the nodes cannot able to communicate with each other and also not able to transmit the packet with exact destination path. Attach the agent to the sender side. The agent like DCCP/ RTP.Attach the sink node to the receiver side [11].

Now the two nodes are ready to communicate with each other.

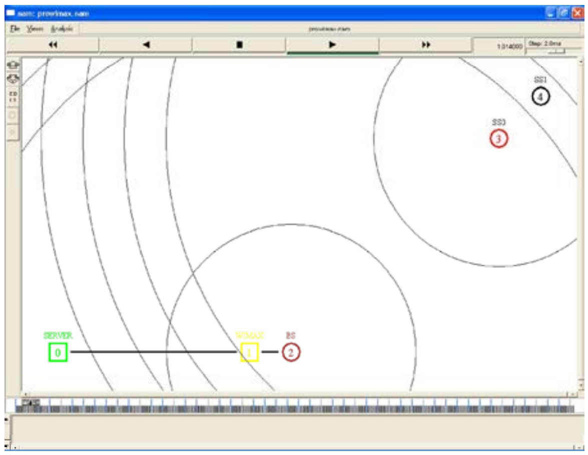

Fig. 1: WiMAX Connection Between variable Subcribers(SS)

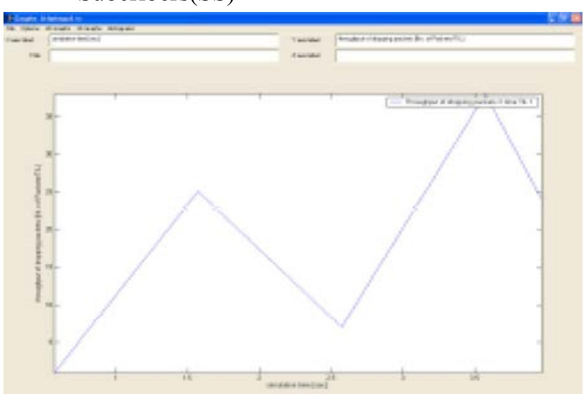

Fig. 2: Throughput Dropping packet

Table 1: node configuration

Nodes type	Wireless node
Number of Nodes	4
Agent	RTP DCCP
Queue Algorithm	Drop Tail
Simulation Time	120 Seconds
Mac Protocol	802.16
Antenna Type	Omni – Directional
Network interface Type	OFDM

Table 2: Throughput Comparison of Dccp and Rtp

	Throughput rate in Mbps	
Video send rate	DCCP	RTP
1.5Mbps	2.00	2.00
2Mbps	2.00	2.00
3Mbps	3.00	3.00
4Mbps	4.00	4.00
5Mbps	4.11	4.77
6Mbps	4.11	5.729
7Mbps	4.11	6.088

User Characteristics: The user in the network needs less packet loss rate during packet transmission and increasing the throughput.

Operational Environment: This project is mainly used in wireless network at the place of where more number packets sending from one node to other node. This project is mainly used to avoid the packet loss while transmitting time and getting maximum throughput in any wireless network using wimax network.

Design and Implimentation Constraints: Develop the project in offline process only. Because this project is basically execute under simulation. Some implementation constraints are followed.

Number of nodes to be mentioned.

Using fixed wireless network.

Nodes to be communicated in 2 – ray mechanism.

Using Drop-Tail queuing algorithm.
Applied the concept of Forward and Reverse Flow.
Topology Setting.

Performance Analysis

Throughput: In this project for video traffic load of 2 Mbps to 4Mbps throughput or receive bit rate is equal to the send rate and packet loss is equal to zero. Up to 4 Mbps of video send rate, DCCP and RTP show an excellent performance for video traffic. But throughput achieved by DCCP is 22% less than DCCP. This is because percentage of packet loss in DCCP is very high and due to unreliable performance of this protocol. As traffic further increased from 4Mbps onwards, DCCP experience more packet drop and its throughput decrease. For 5Mbps of video traffic rate, throughput achieve By RTP is 25% more than DCCP. RTP achieves 4.77 Mbps of throughput and DCCP achieves 4.11Mbps. There is no further increase in DCCP throughput beyond 4.11Mbps for an increase in video send rate. But maximum throughput experience by RTP is up to 6.099Mbps of for video send rate up to 7Mbps.

Packet Loss: DCCP experiences is very small packet loss as compared to RTP. For instance for 5Mbps to video traffic load, packet loss experience by DCCP is 30% and of RTP is 4.46%. These results show that performance of RTP is better than SCTP in terms of packet loss and throughput.

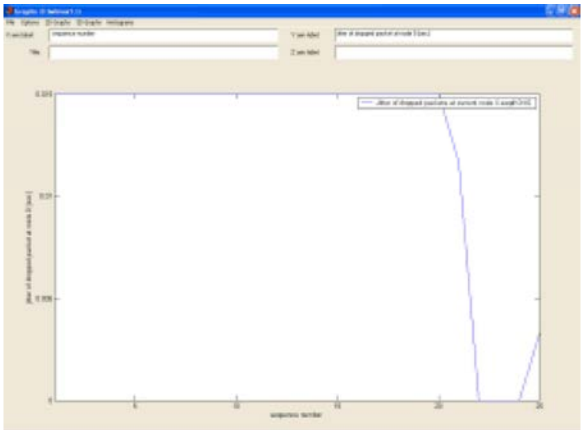

Fig. 3: Jitter of dropped at current node

Delay: Video traffic on delay when DCCP is used as a transport protocol. When 16.66% of background traffic shares the bandwidth with video traffic, video rate varies as 2Mbps, 3Mbps, 4Mbps and 5Mbps and their respective average delay are 18.925ms 31.204 ms. This shows that there is an increase of average delay as the video send rate increases [12].

Jitter: Behavior for DCCP as found in RTP where jitter values increases for an increase in the video rate for different percentages as background traffic

CONCLUSION

Simulation result for Performance Analysis of two protocols DCCP and RTP, video traffic in WiMAX RTP is better than the DCCP protocol throughput, packet loss and delay. We vary the send rate for video traffic from 2Mbps to 7 Mbps and determine the throughput, packet loss, delay and jitter. Based on simulation results we concluded that up to 4Mbps of traffic load DCCP and RTP show an equitant performance that is throughput is equal to sent rate and no packet loss is observed. As traffic load is further increased beyond 4 Mbps the RTP maintains it throughput and its packet loss is minimum but DCCP losses its performance. Packet loss experienced by DCCP is better than SCTP in terms of packet loss and throughput. Also RTP experiences smaller jitter and minimum average delay as compared to the DCCP and RTP [13-17].

Future Enhancement: This work is being extended to include other multimedia application and some new wireless technology such as WIBRO, LTE(long Term Evaluation).

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