Simulation Based Comparison of SCTP, DCCP and UDP Using MPEG-4 Traffic Over Mobile WiMAX/IEEE 802.16e

Muhammad Naveed Khalid
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Contact Information:
Author: Muhammad Naveed Khalid
E-mail: naveed_khalid_84@hotmail.com

University advisor:
Tahir Nawaz Minhas (Doktorand)
School of Computing

Examiner:
Dr. Patrik Arlos
School of Computing

School of Computing
Blekinge Institute of Technology
SE – 371 79 Karlskrona
Sweden

Internet : www.bth.se/com
Phone : +46 457 38 50 00
Fax : +46 457 271 25
ABSTRACT

With the advent of new multimedia applications the demand for in time delivery of data is increased as compared to the reliability. Usually the Transport Layer Protocols, User Datagram Protocol (UDP) and Transmission Control Protocol (TCP) are used to transfer the data over the IP based network like Internet. TCP provides a reliable mechanism to transfer the data but its reliable mechanism results in increase in delay. UDP lacks in providing any acknowledgment mechanism and it does not provide any congestion control mechanism also. However the unreliable behavior of UDP results in less delay in data transfer. Now a days one of the important issues is the Quality of Service (QoS) assurance as the behavior of transport layer protocols can affect the QoS. So in order to avoid these issues some new transport layer protocols have been developed by Internet Engineering Task Force (IETF). Two important transport layer protocols, Datagram Congestion Control Protocol (DCCP) and Stream Control Transmission Protocol (SCTP) are used in this study. DCCP is specially designed to avoid congestion in the network. DCCP is suitable for in time delivery of data and also for its congestion control mechanism. DCCP is an unreliable transport layer protocol, as the real time applications demands for in time delivery rather than reliability. SCTP is another transport layer protocol that provides reliable data transfer. In this research work performance of SCTP, DCCP and UDP has been evaluated using MPEG-4 video over Mobile WiMAX/IEEE 802.16e. The performance of these three transport layer protocols is analyzed in terms of performance metrics like packet loss, jitter, delay and throughput. By analyzing these performance measures it is found that the performance of DCCP and SCTP is much better as compared to UDP but DCCP gives much better performance then SCTP when compared in terms of throughput and packet loss. Comparing SCTP and DCCP with UDP in terms of delay and jitter shows that UDP has less delay and jitter as compared to SCTP and DCCP, but because of less throughput and large number of packet loss, UDP can badly degrade the video quality. So, it is found that the DCCP is the most suitable transport layer protocol for transportation of MPEG-4 traffic over Mobile WiMAX/IEEE 802.16e.

Keywords: IEEE 802.16e, DCCP, ns-2.
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1 INTRODUCTION

As the new access technologies are being introduced, the demand for multimedia applications is also increasing rapidly. These multimedia applications require QoS and better utilization of bandwidth during the transportation. Transport layer protocol is one of the important factor that can affect the QoS of multimedia applications while transportation over the network. Mainly, the transport layer protocol UDP is considered for the transportation of multimedia applications, as it gives less delay and the multimedia applications are delay intolerant. The only problem in UDP is lack of congestion control mechanism which can result in some packet loss. So in order to avoid these problems DCCP can be used for the transportation of video traffic. DCCP is an unreliable transport layer protocol but it has its own efficient congestion control mechanism to avoid congestion in the network, so DCCP can provide in time delivery of multimedia applications as these applications prefer in time delivery over reliability. SCTP is another transport layer protocol which provides reliable transfer of data with congestion control mechanism. In order to transmit the video traffic, different wireless access technologies can be used, as the wireless access technologies are easy to deploy and maintain as compared to wired access technologies. Different access technologies have different parameters like QoS, data rates, bandwidth etc. Mobile WiMAX is one of the access technologies that can be used for the transmission of multimedia traffic. Mobile WiMAX has different features like high data rates, scalability, QoS, security and mobility.

Multimedia applications comprise of great importance in wireless medium because of their challenging and demanding nature. The wireless technologies like WiMAX, UMTS, etc. are becoming popular worldwide resulting in demand of multimedia applications. Transport layer protocols are responsible for end to end communication. The transport layer protocols considered in this thesis are UDP, SCTP and DCCP. A brief overview of these protocols is given this section.

The User Datagram Protocol (UDP) provides an unreliable service with no guaranteed delivery and no congestion control mechanism [1]. So there is lack of QoS in UDP. But it is used in video streaming because of its less overhead. Stream Control Transmission Protocol (SCTP) is a reliable and connection oriented transport layer protocol that provides ordered data delivery and also there is no error in that data. Another important feature supported by SCTP is Multi-homing. In multi-homing, an end point can have support for multiple IP addresses or interfaces. SCTP also supports Multi-streaming feature in which, the data to be sent can be divided into streams. Multi-homing and Multi-streaming can provide better bandwidth utilization [2]. Datagram Congestion Control Protocol (DCCP) is an un-reliable transport layer protocol that provides congestion control on datagrams in network [3]. It has support for Explicit Congestion Notification (ECN). ECN supports for the notification of end-to-end network congestion [4].

Mobile WiMAX/IEEE 802.16e is a broadband wireless technology that provides support for both fixed and mobile broadband networks. Mobile WiMAX can support data rate theoretically up to 63 Mbps. A very important feature provided by Mobile WiMAX is QoS. It provides Service flows which enables end-to-end IP based QoS by using flow labels. In Mobile WiMAX scalability is provided in network architecture and also in radio access technology so providing better flexibility in network deployment and services offered [5].
1.1 Related Work

This section is based on a brief overview of the related work. In [6], the quality of VoIP sessions is evaluated by using WIMAX testbed and the transport protocols considered are UDP and DCCP. The performance of WiMAX technology has been explored in [7]. The network performance of WiMAX is compared with ADSL to check that which technology provides better video streaming. The performance of WiMAX and ADSL is evaluated in terms of packet loss, delay, jitter, and throughput. Modulation schemes in Mobile WiMAX have been evaluated in [8]. Different modulation schemes like BPSK, QPSK, 16-QAM and 64-QAM are compared with TCP and with different variants of TCP in terms of performance metrics like throughput and packet loss ratio.

In transportation of multimedia applications time is very important as compared to reliability. By using transport layer protocols like UDP, TCP and DCCP’s variants like CCID1, CCID2 and CCID3, the performance of MPEG-4 video is analyzed [9]. To analyze the performance of these transport layer protocols performance measures like throughput, packet loss, delay and jitter are used. The performance of MPEG-4 video has been analyzed by using SCTP as transport layer protocol over 802.11 wireless access medium in [10]. SCTP retransmission overhead delay has been evaluated by using computer simulations. The performance of Mobile WiMAX is analyzed by using traffic measurements in [11]. Transport layer protocols like, UDP and TCP have been used to analyze the link capacity and goodput performance. The Mobile WiMAX and High-speed Downlink Packet Access (HSDPA) are then compared on the basis of performances measured.

Throughput performance of SCTP is evaluated over IEEE 802.11WLAN in [12]. The performance is evaluated by using different number of hops between sender and receiver and with different SCTP window sizes at receiver side. In [13], the performance of SCTP is analyzed by using streaming video over CDMA2000 wireless network. The performance is evaluated by analyzing quality of video received using a specific buffer size. A study has been conducted to evaluate the performance of DCCP, SCTP and UDP for video traffic over Wi-Fi in [14]. Performance of Transport Layer Protocols for video traffic over fixed WiMAX is evaluated in [15].

In previous work, the performance of transport layer protocols is analyzed by using VoIP applications or MPEG-4 traffic over CDMA 2000, ADSL, WLAN and fixed WiMAX. The work has not been done yet by using Mobile WiMAX as the wireless access technology. So in this thesis work, the performance of DCCP, SCTP and UDP is analyzed by using MPEG-4 traffic over Mobile WiMAX.

1.2 Contribution

This thesis is based on the analysis of transport layer protocols using MPEG-4 traffic over Mobile WiMAX. Different scenarios are implemented in order to analyze the performance of these transport layer protocols. The effect of mobility is shown in the scenarios, that how the speed of mobile nodes affect the performance of transport layer protocols. The mobile nodes are moved to a certain point in simulations with different speeds. The performance metrics like packet loss, throughput, delay and jitter are used to analyze the performance of these transport layer protocols. The network simulator 2 (ns-2) is used to simulate the transport layer protocols using MPEG-4 traffic over Mobile WiMAX in order to find out the best transport layer protocol.
1.3 Thesis Outline

Chapter 2 includes description of transport layer protocols discussed in this thesis work. Chapter 3 is based on a description of Mobile WiMAX, its features, QoS classes and the layers (MAC and PHY) on which it operates. Chapter 4 describes the MPEG-4 video model and different attributes of MPEG-4 traffic. Chapter 5 includes the simulation setup, simulation scenarios and analysis of results. Chapter 6 is based on conclusions and the future work.
2 TRANSPORT LAYER PROTOCOLS

This chapter is based on the description of transport layer protocols considered in this thesis. These protocols are discussed one by one as follows.

2.1 Services and Features offered by DCCP, SCTP and UDP

The services and features provided by these transport layer protocols are shown in Table 2.1.

<table>
<thead>
<tr>
<th>Features and Services</th>
<th>DCCP</th>
<th>SCTP</th>
<th>UDP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Connection Oriented</td>
<td>Yes</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>Un-ordered Data Delivery</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Ordered Data Delivery</td>
<td>No</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>Reliable</td>
<td>No</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>Congestion Control</td>
<td>Yes</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>Flow Control</td>
<td>Optional</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>Multi-streaming</td>
<td>No</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>Multi-homing</td>
<td>No</td>
<td>Yes</td>
<td>No</td>
</tr>
</tbody>
</table>

The services and features offered by these transport layer protocols as shown in Table 2.1 are described in detail in next section.
2.2 Services and Features by DCCP

DCCP is a connection oriented and un-reliable transport layer protocol. This section is based on services and features provided by DCCP.

2.2.1 Packet Format of DCCP

DCCP packet format is shown below [16]:

<table>
<thead>
<tr>
<th>Bits 0-7</th>
<th>8-15</th>
<th>16-31</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Source Port</td>
<td>Destination Port</td>
</tr>
<tr>
<td>Data offset</td>
<td>CCVal</td>
<td>CsCov</td>
</tr>
<tr>
<td>0</td>
<td>Type</td>
<td>X</td>
</tr>
<tr>
<td>Options</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Data</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

The source port points to the sender of a particular packet while the destination port points to the receiver of that particular packet. Data offset is the offset which is from the beginning of a packet’s header to the beginning of the data part. It is represented in 32-bit words. CCVal is the Congestion Control Value. The Half Closed (HC) sender Congestion Control Identifier (CCID) uses the CCVal. CsCov is the Checksum Coverage. Parts of a packet which the checksum field covers, are specified by CsCov. The DCCP header and options are included in CsCov but some or none of the data part is included. Checksum is a value used to find out the data integrity. The checksum depends on CsCov. The field Type consists of some unique values. Each unique value represents an action which includes e.g. DCCP-Request, DCCP-Response, DCCP-Data etc. These values are then used by DCCP sender and receiver in order to perform an action. X is the Extended Sequence Numbers. In order to use an extended generic header along with 48-bit Acknowledgment and Sequence Number, the value of X is set to 1. The packets are identified uniquely on the basis of Sequence Number. The sequence number is increased by one as a packet is sent by the sender. Remaining is the data part of variable length.

2.2.2 Connection Orientation of DCCP

Connection establishment in DCCP is accomplished by using three-way hand shake. In this three-way hand shake a connection establishment request DCCP-Request is sent by client to the receiver then a DCCP-Response is sent by server to the client and then finally an acknowledgment via a DCCP-Ack is sent by client towards the server [17].
2.2.3 Un-Ordered Data Delivery of DCCP

DCCP is intended for the applications which require in-time delivery of data and reordering of data can cause delays which the multimedia applications cannot tolerate [18]. So in DCCP data is not reordered because reordering can result in delay.

2.2.4 Un-Reliability of DCCP

DCCP is an unreliable transport layer protocol. DCCP is aimed for the real time applications in which delays are intolerable, so that’s why DCCP does not retransmit dropped packets because the real time applications prefer to receive the most recent data then the older one [19].

2.2.5 Flow Control and Congestion Control in DCCP

DCCP is not a flow control protocol in fact it is a congestion control protocol [17]. The flow control limits can affect the transfer rates. So DCCP does not make use of flow control which is separate from congestion control. The flow control is optional in DCCP. If required, flow control mechanism can be implemented on top of DCCP [19].

DCCP is basically designed to provide congestion control. Two types of congestion control mechanisms are offered by DCCP named CCID 2: TCP-like congestion control mechanism and CCID 3: TCP Friendly Rate Control (TFRC) congestion control mechanism. In order to make choice between these two mechanisms, congestion control IDs (CCIDs) are used. A particular CCID is negotiated during the connection startup between two endpoints [19].

In this thesis, CCID 2: TCP-like congestion control mechanism is used. CCID 2 also uses some of the features used in Transmission Control Protocol (TCP) congestion control mechanism [20], that’s why it is called TCP-like congestion control mechanism. In case of packet loss in the network, as the packet loss takes place due to the congestion in the network, CCID 2 halves the congestion window to reduce the congestion in the network [21]. An ack vector option is used by the CCID 2 acknowledgements. Variables used by
CCID 2 congestions control mechanism are: CWND, slow-start threshold or ss-threshold and the number of outstanding data packets.

CCID 2 congestion control mechanism not only controls the forward path congestion but also reacts to the reverse path congestion. Congestion control on reverse path is also very important because high congestion on reverse path can slow down the traffic on forward path. So ideally this reverse-path congestion control should be used by modern protocols. So, a feature known as Ack Ratio is maintained by CCID 2. A rough ratio of data packets per acknowledgement is controlled by the ACK Ratio. Delayed-ack behavior as used in TCP is given by the default ack ratio. As the lost acknowledgments are detected by the CCID 2 sender, the Ack Ratio is manipulated in order to reduce the ack rate. Ack Ratio is an integer variable, so its value is at least set to 2 for congestion window of five or more packets in order to reduce the ack load.
2.3 Services and Features by SCTP

SCTP is a connection oriented and reliable transport layer protocol. A number of services and features are offered by SCTP which are discussed in this section.

2.3.1 Packet Format of SCTP

SCTP packet format is shown below [22].

<table>
<thead>
<tr>
<th>Bits 0-7</th>
<th>8-15</th>
<th>16-31</th>
</tr>
</thead>
<tbody>
<tr>
<td>Source port</td>
<td>Destination Port</td>
<td></td>
</tr>
<tr>
<td>Verification tag</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Checksum</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Chunk N type</td>
<td>Chunk N flags</td>
<td>Chunk N length</td>
</tr>
<tr>
<td>Chunk N data</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

The fields, source port and destination port are already described in section 2.1.1. The field verification tag is used by the receiver of a particular packet in order to validate the sender of that particular packet. The checksum field contains the checksum value of a particular SCTP packet. The Chunk type is used to identify the type of data available in Chunk data. The Chunk flags bits are dependent on chunk type, they are set to zero if not specified by the chunk type. The chunk length specifies the chunk size in terms of bytes. It includes size of chunk data, chunk flags, chunk type and chunk length fields. The field chunk data is of variable length.

2.3.2 Connection Orientation of SCTP

SCTP is connection oriented transport layer protocol. In SCTP, a connection is known as an association. When a process at host A wants to exchange the data from a process at host B, then first of all an association is established between each other [23]. After that data is exchanged between them, the association is terminated. This process is shown in Figure 2.2.
2.3.3 Ordered data delivery of SCTP

Unordered data delivery is allowed in SCTP as shown in Figure 2.3. In case a stream is affected in SCTP then only the affected stream is blocked temporarily while the other streams will be allowed to pass. SCTP does not wait for the messages to be ordered numerically, in fact, it process them on their arrival, no matters what order they have. These messages are transferred reliably by SCTP. This feature of SCTP of simply passing the data minimizes the overhead of message reordering on the server [24].

2.3.4 Reliability in SCTP

SCTP offers a reliable transmission service like TCP [20]. SCTP ensures that the data is transmitted to the network in sequence and without error. Reliable transmission in SCTP is also achieved by detecting when the data is corrupted, duplicated or discarded. This feature also transmits the damaged data if necessary [2].

2.3.5 Flow Control and Congestion Control in SCTP

SCTP offers a process to process flow control mechanism similar to one provided by TCP [20]. The receiver by defining the receiver window size controls the rate by which the sender side sends the data. Receiver then also sends the value of receiver window with all
selective acknowledgement (SACK) chunks. The sender also maintains a variable called as Congestion Window (CWND). The bytes to be sent before they are acknowledged are controlled by the congestion window. All the data received should be acknowledged, the receiver can wait for a certain time usually it is 200 ms [25].

The SCTP congestion control mechanism is applied to the whole association not to the individual streams. SCTP congestion control mechanism is similar to that of TCP [20]. In SCTP there are different modes for the congestion control like slow start and congestion avoidance. The congestion is controlled by using the CWND. The CWND is maintained at the sender side. A variable known as slow-start threshold or ss-threshold is maintained by the sender and it is used to distinguish that which type of congestion control mechanism should be used to avoid congestion [26]. In congestion-avoidance mode the CWND increases by one Maximum Transmission Unit (MTU) per Round Trip Time (RTT). In case of slow start mode, CWND increases faster that is roughly one MTU per SACK chunk that a sender receives. [25].

2.3.6 Selective Acknowledgement (SACK) in SCTP

To acknowledge the data, SCTP makes use of selective acknowledgement (SACK) mechanism. This mechanism points out gaps, if any, in the transmission like missing data blocks. SCTP can report larger amounts of missing data packets in a SACK [27].

2.3.7 Half-Closed Connection in SCTP

The condition of half-closed connection takes place when one end of the association believes that the association is closed but the other end still believes that the association is open. In SCTP this possibility can be removed by making use of a three-way shutdown mechanism, which consists of SHUTDOWN, SHUTDOWN-ACK and SHUTDOWN-COMPLETION. After this shutdown procedure both ends close the communication. A new connection establishment is required to send more data or information [24].

2.3.8 Preservation of Message Boundaries

Preservation of the message boundaries is applicable in SCTP. If a sender sends 200 bytes and 100 bytes towards a receiver then the information received by the receiver will be with preserved message boundaries that is, these bytes will not merge during the communication. So, in this way the application does not need to split the messages as the message boundaries are preserved [24]. The preservation of message boundaries in SCTP is shown in Figure 2.4

![Figure 2.4: Preserved Message Boundaries](image)
2.3.9 Multi-Streaming function in SCTP

SCTP allows multiple data streams in an association or connection. All the streams in an association are related to that specific association but they are independent. A stream number is given to each stream, which is encoded in to the packets flowing through the association. If there is some packet loss within a stream then only that specific stream will be blocked (until the lost packets are re-transmitted) without affecting other streams in an association. This problem is known as head-of-line blocking [28].

![Figure 2.5: Multiple streams within an SCTP association](image)

2.3.10 Multi-Homing function in SCTP

Multi-homing is one of the core functions of SCTP. In multi-homing, a single SCTP end point can maintain multiple IP addresses or multiple interfaces [2]. In SCTP multi-homing feature, one interface is selected as the primary interface while other becomes the secondary as shown in Figure 2.6. The communication starts through primary interface but in case if the primary interface becomes inactive or fails, the communication is shifted to the secondary interface. When the primary interface is available again, the communication is shifted back to the primary interface. A heartbeat acknowledgment mechanism is used to check and monitor the primary and secondary interfaces. This heartbeat mechanism indicates which interface is slow and which one is fast, if primary interface is slower as compared to the secondary interface then the communication is shifted to the secondary interface [24].

![Figure 2.6: Multi-Homing feature in SCTP](image)
2.4 Services and Features by UDP

User Datagram Protocol (UDP) is a connectionless and unreliable transport layer protocol. In this section UDP is described on the basis of different services and features.

2.4.1 Packet Format of UDP

The UDP header consists of four fields and after that the data is added [1]. The UDP data header format is shown in Table 2.4.

<table>
<thead>
<tr>
<th>Table 2.4: Packet Format of UDP</th>
</tr>
</thead>
<tbody>
<tr>
<td>0-15</td>
</tr>
<tr>
<td>16-31</td>
</tr>
<tr>
<td>Source port</td>
</tr>
<tr>
<td>Destination Port</td>
</tr>
<tr>
<td>Length</td>
</tr>
<tr>
<td>Checksum</td>
</tr>
<tr>
<td>Data Octets …</td>
</tr>
</tbody>
</table>

The fields, source port, destination port and checksum are described already in section 2.1.1. The field Length indicates the total size of UDP header and data octets in bytes.

2.4.2 Connection Orientation of UDP

UDP is responsible for process to process communication. UDP is a connectionless transport layer protocol. A datagram can be sent at any time without any prior negotiation between the sender and receiver. Sender just sends the datagram with the hope that receiver is ready to receive it [29].

2.4.3 Unordered Data Delivery of UDP

In UDP there is no ordered data delivery and duplicate packets can also arrive at the receiver. So it means that a packet with Packet ID-2 can arrive before a packet with Packet ID-1. Also a packet with same ID can be received twice which is called packet duplication [29].

2.4.4 Unreliability of UDP

UDP is an unreliable transport layer protocol. There is no guarantee that the datagrams transmitted by the sender will be received by the receiver. The receiver does not send any acknowledgment to the sender that the packet has arrived. There is no packet recovery mechanism in UDP, so if a packet is lost then it cannot be re-transmitted [29].
2.4.5 Flow Control and Congestion Control in UDP

Flow control is the process of adjusting the data flow from sender to the receiver in order to ensure that the receiver side can handle or receive all the incoming data [30]. As UDP is a connection less protocol, so flow control mechanism is not implemented in it.

Typically UDP is used in case of real-time video and audio streams because in these applications, the recovery of lost packets is not needed. As UDP is an unreliable transport layer protocol so the receiver does not send any acknowledgement (ACK) signals to the sender [31]. As no ACK mechanism is implemented in UDP, so the sender cannot determine the link capacity and thus cannot apply the congestion control mechanism.

2.4.6 Half-closed Connection in UDP

Half closed connection condition applies when one end of the connection assumes that the connection is open while the other end assumes that the connection is closed. This condition does not apply in UDP as there is no connection establishment in UDP [24].

2.4.7 Preservation of Message Boundaries

Preservation of the message boundaries is applicable in UDP. If a sender sends 200 bytes and 100 bytes towards a receiver then the information received by the receiver will be with preserved message boundaries that is, these bytes will not merge during the communication. So, in this way the application does not need to split the messages as the message boundaries are preserved [24]. The preservation of message boundaries in UDP is shown in Figure 2.4.
3 MOBILE WiMAX/IEEE 802.16e

3.1 Introduction

Mobile WiMAX/IEEE 802.16e is a broadband wireless technology that provides support for both fixed and mobile broadband networks. Mobile systems in Mobile WiMAX are configured based on a similar baseline feature resulting in full inoperability between base stations and terminals. Some optional elements are also specified in base station profiles in order to give extra flexibility to specific deployment scenarios as they might need different configurations that can be coverage or capacity optimized [5].

In Mobile WiMAX, Base Station (BS) provides air interface to the Mobile Station (MS). It may provide other functions to MS that are; QoS policy enforcement, radio resource management, traffic management etc. Communication between mobile stations is done via BS [32]. The Mobile WiMAX system is shown in Figure 3.1.

3.2 Salient Features of Mobile WiMAX

Mobile WiMAX provides a broadband wireless solution. In Mobile WiMAX, scalability is provided in network architecture and also in radio access technology so providing better flexibility in network deployment and services offered. Mobile WiMAX can
support peak download rates up to 63Mbps and peak upload rates up to 28 Mbps [5]. Some of the salient features provided by Mobile WiMAX are as follows:

1. High data rate.
2. Quality of Service.
4. Scalability.
5. Mobility.

### 3.3 OSI Model Layers used by Mobile WiMAX

The Open Systems Interconnection (OSI) model comprises of seven layers in which the last two layers are Data Link and Physical Layers. The Data Link Layer is further divided into two sub layers known as Medium Access Control (MAC) and Logical Link Control (LLC) layers. The OSI model is shown in Figure 3.2.

The IEEE 802.16e uses the MAC Layer and the Physical Layer as shown in figure 3.3. The responsibility of MAC layer includes connection establishment and maintenance, while the Physical Layer is responsible for Physical connection establishment among two communicating peer entities [33]. These layers are explained in detail in next section.

![OSI Model](image1.png)

**Figure 3.2: OSI Model, First two layers are defined for Mobile WiMAX**

![Mobile WiMAX Layers](image2.png)

**Figure 3.3: Mobile WiMAX Layers**
3.3.1 Medium Access Control (MAC) Layer

Mobile WiMAX MAC layer consists of three sub-layers:

1. Service Specific Convergence Sublayer (CS)
2. Medium Access Control Common Part Sublayer (MAC CPS)
3. Medium Access Control Security Sublayer

These sub-layers are described as following:

1. Service Specific Convergence sublayer (CS)

The Convergence Sublayer (CS) lies above the MAC CPS. The MAC CPS sub-layer provides services to the CS through MAC Service Access Point (SAP) [33]. The functions performed by CS are as follows:

- The CS layer accepts the Protocol Data Units (PDUs) from the upper layers. The CS specifications are provided for two different types of upper layers, these layers are Packet CS and asynchronous transfer mode (ATM) CS.
- The CS layer classifies and maps the MAC Service Data Units (MSDUs) into proper or suitable Connection Identifiers (CIDs).
- The upper layer PDUs are processed (if required) depending on the classification.

2. Medium Access Control Common Part Sublayer (MAC CPS)

The MAC CPS is located in the center of MAC layer. This is the most important sublayer in MAC layer [33]. The responsibilities of this sub-layer include:

- Bandwidth allocation by the BS to MS.
- Connection establishment and maintenance between the BS and MS.

Through the MAC SAP, the CPS receives the information from CS.

3. Medium Access Control Security Sublayer

The MAC layer comprises of another important sub-layer which is the Security Sublayer. The Security Sublayer provides with authentication, secure key exchange, encryption and integrity control [33].

3.3.2 Physical (PHY) Layer

The responsibility of Physical Layer is the physical connection establishment between transmitter and receiver. It transmits data in the form of bits. The services provided by Physical Layer are; the type of modulation and demodulation being used, the type of signal transmitted or received the transmission power of the signal etc. [33].

The frequency band considered in 802.16 is 2-66 GHz [33]. The 2-11 GHz is used in fixed WiMAX while 2-6 GHz is used in Mobile WiMAX [32].

The techniques used for the multi-carrier modulation are Orthogonal Frequency Division Multiplexing (OFDM) and Orthogonal Frequency Division Multiple Access (OFDMA). The sub-carriers can be adaptively modulated by using these two techniques. Techniques used for modulation are QPSK, 16-QAM and 64-QAM. Selection of these modulation techniques is based on noise and distance [34].
OFDM applies interleaving and coding to the information around the sub-carriers before the transmissions, so in this way OFDM uses frequency diversity of the multipath channels. The Inverse Fast Fourier Transform (IFFT) can be used in order to achieve the OFDM modulation. With less complexity a large number of sub-carriers can be enabled by using IFFT (up to 2048). In the frequency domain the resources are available in terms of sub-carriers while in time domain the resources are available in terms of OFDM symbols. The frequency and the time resources can be allocated to individual users by organizing them into sub-channels. In OFDMA, data streams are multiplexed form multiple users to the downlink sub-channels and it also uplinks multiple access by making use of uplink sub-channels [5].

Physical layer also includes QoS mechanisms i.e. Forward Error Control (FEC), Time Division Duplex (TDD), Frequency Division Duplex (FDD) etc. [34].

In FDD, data can be transmitted on different sub-bands. There is no interference between the sub-bands as each sub-band use different set of frequencies [34].

In TDD, downlink and uplink bandwidth can be dynamically allocated. For example, if there is more downlink traffic then more bandwidth can be allocated to downlink traffic and bandwidth can be taken back in case of less downlink traffic [34].

3.4 Mobile WiMAX Quality of Service (QoS) Classes

The MAC layer of Mobile WiMAX base station (BS) provides five QoS classes. These QoS classes are as following [35]:

- Unsolicited Grant Services (UGS).
- Real Time Polling Services (rtPS).
- Non Real Time Polling Services (nrtPS).
- Extended Real Time Polling Services (ertPS).
- Best Effort (BE).

3.4.1 Unsolicited Grant Service (UGS)

The UGS is designed for the real time data streams. These real time applications generate fixed-size data. Applications supported by UGS are Voice over IP (VoIP) without silence suppression, T1/E1.

3.4.2 Real Time Polling Services (rtPS)

This QoS class supports the streaming Audio and Video applications. These applications consist of variable sized data. The QoS features provided in this class are traffic priority, minimum reserved rate and latency tolerance.

3.4.3 Non Real Time Polling Services (nrtPS)

In this QoS class, unicast polls are provided to the service flows, so that they can get request opportunities even in case of congestion in the network.
3.4.4 Extended Real Time Polling Service (ertPS)

In this QoS class, unsolicited unicast grants are provided by the BS, so in this way the latencies caused by the bandwidth requests are removed.

3.4.5 Best Effort (BE)

This service class does not offer guaranteed delivery. Web browsing and data transfer are supported by the BE.
4  MPEG-4 VIDEO MODEL AND TRAFFIC

Moving Picture Expert Group version 4 (MPEG-4) gives better performance in terms of video streaming application for Internet by comparing with MPEG-1 and MPEG-2. MPEG-4 makes use of video codes which reduces the bit rate but still keep the same video quality. So in this way MPEG-4 is much better then MPEG-1 and MPEG-2. MPEG-4 video stream consists of three types of frames, namely; I-frame (Intra-coded frame), P-frame (Predicted frame) and B-frame (Bidirectional frame). I-frame is independently encoded from the other frames. P-frame is encoded with reference to I-frame or P-frame, it considers the closest time-preceding frame. B-frames are coded with reference to I or P frames, these are the time adjacent frames [36].

4.1 Video Traffic

In this thesis MPEG-4 video traffic is used at the application layer. If MPEG-4 is compared to MPEG-1 and MPEG-2, it is more suitable for transmission over the internet. The MPEG-4 video traffic used in this thesis has been taken from [37]. This video traffic is generated by a traffic generator, which consist of six statistical files. These statistical files are used to generate I, P and B frames. As discussed already, I, P and B frames are used to construct the MPEG-4 video traffic. These frames are shown in Figure 4.1.

![Figure 4.1: The dependency of I, P and B frame](image)

The bit rate provided by MPEG-1 and MPEG-2 is up to 1.5 Mbps and 15 Mbps respectively while the bit rate provided by MPEG-4 ranges between 4.8 kbps to 64 kbps. Video Object Plane (VOP) is the basic object in MPEG-4 [38]. The VOP represents a video frame with a rectangular shape. A VOP is a combination of I, P and B frames while a combination of VOP is called Group of VOP (GOV). A GOV always starts with an I-frame [36].
4.2 Video Formats

Resolution is the number of pixels in a single video frame. Pixels in a frame are represented as (pixels in a row) x (pixels in a column) [39]. Different video formats have different video resolutions. Obviously the video with higher resolution provides better visual quality. Video resolution ranges from 176x144 to 1920x1080 [40]. HD DVD has the best resolution i.e. 1920x1080.

4.3 Pixel color depth

Color depth is the amount of pixel information stored in bits. More pixel information results in better picture quality. More memory will be required for the storage of image in case of high color depth [41].

4.4 Video compression scheme

Video compression is very important because in case of streaming video over internet, there are limitations like bandwidth etc. Video compression is applied when there are redundancies in the video stream. MPEG-4 is a compression standard for web and multimedia compression [42].

4.5 Frame inter-arrival rate

Frame inter-arrival rate is very important aspect of multimedia traffic. It plays an important role as increase in the number of video frames also increases the video quality. Frame inter-arrival rate is the rate at which the receiver receives the video frames and playback [7].
5 SIMULATIONS

This chapter is based on simulation scenarios, performance measures and the final results. The performance of SCTP, DCCP and UDP is evaluated by using the MPEG-4 traffic over Mobile WiMAX. The results of the simulation will explore that which transport layer protocol delivers better performance. The performance measures used in these scenarios are packet loss, throughput, jitter and delay.

5.1 Ns-2 Simulator

The simulator used in this thesis is Network Simulator version 2 (NS-2) [43]. Ns-2 is an open source and discrete event simulator. The basic purpose of Ns-2 is to provide virtual environment of a network. Ns-2 provides support for the simulation of routing, transport protocols and multicast protocols over wireless and wired networks. Ns-2 is based on two main languages i.e. Object Oriented Tool Command Language (OTcl) and C++.

5.2 IEEE 802.16e Implementations for Ns-2

The WiMAX module used in this thesis is based on IEEE 802.16 and mobility extension IEEE 802.16e [44]. The features provided by this module are as following:

1. Time Division Duplexing (TDD).
2. Management messages for node entry in the network.
3. Round robin scheduler is used as the default scheduler.
4. Fragmentation and re-assembly of the downlink and uplink frames.
5. IEEE 802.16e extensions for mobility.

User data rate can be controlled by using different type of parameters provided in this module e.g. type of modulation, uplink to downlink ratio, frequency bandwidth etc.

5.3 Performance Measures

The performance measures used in this thesis are briefly described in this section. These performance measures are calculated at the transport layer.

- **Packet Loss**

  Packet loss is the total number of packets lost during the video transport between BS and MS. Packet loss is calculated by taking the difference between total number of packets transmitted and received.

\[
\text{Packet Loss} = \text{Total number of packets transmitted} - \text{Total number of packets received} \quad (1)
\]
• **Throughput**

   Throughput is the amount of data that a mobile station or a subscriber station receives from the server or sender in a specific time interval.

   \[
   \text{Throughput} = \frac{\text{Total amount of data received}}{\text{Total time}} \quad \ldots \quad (2)
   \]

   The total time is the difference between last packet and first packet sent time.

• **Delay**

   Delay is the total time taken by a packet during the transmission in a network from one communicating entity to the other. Delay is calculated by taking the difference of sending time and receiving time of a packet.

   \[
   \text{Delay} = \text{Packet (A) receive time} - \text{Packet (A) send time} \quad \ldots \quad (3)
   \]

   Mean delay is calculated in the experiments conducted.

• **Jitter**

   Jitter is the time difference between two consecutive delays. Formula for calculating the jitter is given by

   \[
   \text{Jitter} = \text{Delay (A)} - \text{Delay (B)} \quad \ldots \quad (4)
   \]

   Delay (A) is the delay of a current packet while Delay (B) is the delay of a previous packet. Mean jitter is calculated in the experiments conducted.

### 5.4 Simulation Scenarios and Results

The simulation setup is based on a single cell Mobile WiMAX network. The Mobile Station (MS) and Base Station (BS) are connected to each other by means of a wireless link. The Server is connected to the BS through a wired link. This setup is shown in figure 5.1. The server sends the MPEG-4 traffic towards BS and then BS sends the traffic towards MS. The MPEG-4 traffic is generated at the application layer which flows towards the transport layer and then from the transport layer towards the Mobile WiMAX layers as shown in Figure 5.2.

Different scenarios have been considered to transport the MPEG-4 traffic between MS and the server. Performance of SCTP, DCCP and UDP is analyzed by using packet loss, throughput, delay and jitter as the performance metrics.

![Figure 5.1: MS and Server Connected through BS](image)
5.4.1 Simulation Scenarios

Three scenarios have been configured for the performance analysis of SCTP, DCCP and UDP using MPEG-4 traffic over Mobile WiMAX. The perfect radio conditions are considered in these scenarios. These scenarios are as following:

1. **Capacity of the WiMAX downlink.**
2. **Effect of video send rate on SCTP, DCCP and UDP.**
3. **Effect of multiple Mobile Stations on SCTP, DCCP and UDP.**

1. **Capacity of the WiMAX downlink**

   In this scenario total capacity of WiMAX downlink have been evaluated. The Subscriber Station (SS) downloads the Constant Bit Rate (CBR) traffic from the server through the BS. Figure 5.3 shows the setup.

   The parameters considered in this scenario are: Different downlink to uplink ratios (dl_ratio) like 0.4, 0.5 and 0.6. The downlink to uplink ratio is the difference of speeds in downloading and uploading the data. In TDD, with the increase in the amount of downlink traffic, more resources are allocated to it accordingly. Similarly, when the uplink traffic increases more resources are allocated to it. The other parameters used are: Frame duration of 4 ms. Basically a frame consist of a downlink sub-frame and an uplink sub-frame. For example, when dl ratio is 0.4, it means that 40% of the frame is for downlink while 60% is
for uplink. Packet size is of 1200 bytes. An increase in the frequency bandwidth results in an increase in throughput. In this scenario the frequency bandwidth chosen is 5 MHz. Different modulation schemes can be used like BPSK, QPSK, 16 QAM and 64 QAM. 64 QAM gives better throughput as compared to other modulation schemes so the modulation scheme used is 64 QAM. The bandwidth of wired link is 100 Mb, the propagation delay is 1 ms and the queuing mechanism is drop tail. In drop tail queuing mechanism, if the queue is full, a new packet is dropped on arrival to the queue. The total simulation time is 55 s while the CBR traffic starts at 25 s and stops at 50 s. The send rate and received rate are calculated at the transport layer.

<table>
<thead>
<tr>
<th>Send Rate (Mbps)</th>
<th>Received Rate (Mbps) when dl_ratio=0.4</th>
<th>Received Rate (Mbps) when dl_ratio=0.5</th>
<th>Received Rate (Mbps) when dl_ratio=0.6</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>0.98</td>
<td>0.98</td>
<td>0.98</td>
</tr>
<tr>
<td>2</td>
<td>1.97</td>
<td>1.97</td>
<td>1.97</td>
</tr>
<tr>
<td>3</td>
<td>2.95</td>
<td>2.95</td>
<td>2.95</td>
</tr>
<tr>
<td>4</td>
<td>3.93</td>
<td>3.93</td>
<td>3.93</td>
</tr>
<tr>
<td>5</td>
<td>4.92</td>
<td>4.92</td>
<td>4.92</td>
</tr>
<tr>
<td>6</td>
<td>5.23</td>
<td>5.9</td>
<td>5.9</td>
</tr>
<tr>
<td>7</td>
<td>5.23</td>
<td>6.88</td>
<td>6.88</td>
</tr>
<tr>
<td>8</td>
<td>5.23</td>
<td>7.11</td>
<td>7.86</td>
</tr>
<tr>
<td>9</td>
<td>5.23</td>
<td>7.11</td>
<td>8.85</td>
</tr>
<tr>
<td>10</td>
<td>5.23</td>
<td>7.11</td>
<td>9</td>
</tr>
<tr>
<td>11</td>
<td>5.23</td>
<td>7.11</td>
<td>9</td>
</tr>
<tr>
<td>12</td>
<td>5.23</td>
<td>7.11</td>
<td>9</td>
</tr>
</tbody>
</table>

From Table 5.1, it can be observed that when dl_ratio is 0.4 the throughput or the received rate increases up to 5.23 Mbps, after that the link saturates and a constant throughput of 5.23 Mbps is observed even if send rate is increased. The reason is that the UDP is used as the transport protocol, so when the traffic exceeds the limit of UDP, it starts dropping the packets. Similarly a constant throughput of 7.11 Mbps is observed when dl_ratio is 0.5 and a constant throughput of 9 Mbps is observed when the dl_ratio is 0.6. The results have also been analyzed over fixed WiMAX [15], in order to show the capacity of WiMAX downlink. The results are also depicted in the graphical form in Figure 5.4.

Figure 5.4: Capacity of the WiMAX downlink
2. Effect of video send rate on SCTP, DCCP and UDP

Figure 5.5: Downloading video traffic

Figure 5.5 shows a downloading scenario. MS downloads the video traffic from the server through the BS. This scenario is conducted for all the transport layer protocols considered in this Thesis. Packet Loss, delay, jitter and throughput is examined for SCTP, DCCP and UDP. Whenever the server wants to send the data to the MS, first it sends to the BS and then BS routes the data to the MS. Rate factor is adjusted at the application layer, while the performance measures are calculated at transport layer.

The Parameters used in this scenario are frequency bandwidth of 5 MHz, downlink to uplink ratio is 0.7, means 70 % of the frame is for downlink while 30% for uplink. Frame duration is 5 ms and packet size is of 1200 bytes. Modulation technique used is 64 QAM. The bandwidth of wired link is 100 Mb, the propagation delay is 1 ms and queuing mechanism is drop tail. The total simulation time is 55 s while the MPEG-4 traffic starts at 25 s and stops at 50 s. The MS is generated near a location closed to BS then it linearly starts moving away from the BS with a speed of 5 m/s. The performance measures calculated in this scenario for SCTP, DCCP and UDP are shown in Table 5.2.

<table>
<thead>
<tr>
<th>Rate Factor</th>
<th>DCCP</th>
<th>SCTP</th>
<th>UDP</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Send Rate (Mbps)</td>
<td>Throughput (Mbps)</td>
<td>Packet Loss</td>
</tr>
<tr>
<td>5</td>
<td>1.11</td>
<td>1.09</td>
<td>0</td>
</tr>
<tr>
<td>10</td>
<td>1.98</td>
<td>1.95</td>
<td>0</td>
</tr>
<tr>
<td>15</td>
<td>3.28</td>
<td>3.23</td>
<td>0</td>
</tr>
<tr>
<td>20</td>
<td>4.16</td>
<td>4.1</td>
<td>0</td>
</tr>
<tr>
<td>25</td>
<td>4.59</td>
<td>4.53</td>
<td>0</td>
</tr>
<tr>
<td>30</td>
<td>4.57</td>
<td>4.5</td>
<td>17</td>
</tr>
<tr>
<td>35</td>
<td>4.5</td>
<td>4.43</td>
<td>2</td>
</tr>
</tbody>
</table>

Rate factor is a parameter used to increase or decrease the flow of traffic. Rate factor directly affects the send rate of the sending node, as the variation in rate factor accordingly increases or decreases the send rate. The increase or decrease in send rate affects the corresponding throughput or received rate.
From Table 5.2, it is observed that, as the send rate increases the throughput also increases. In case of UDP, the maximum throughput achieved is 6.34 Mbps but there is a large number of packet loss. In SCTP, it can be observed that the send rate increases up to 4.59 Mbps but after that a minor decrease in send rate is observed. As SCTP is a reliable protocol so when it detected the congestion in the network it reduced the send rate to avoid the packet loss, so it can be observed that the packet loss is again decreased when rate factor is 35. It can be observed that there is very less packet loss in SCTP as compared to UDP, and better throughput is received if compared to the send rate. In case of DCCP maximum throughput is achieved, this is because of the congestion control mechanism of DCCP as the main purpose of DCCP is to control the congestion in the network. Also there is no packet loss in DCCP. Graphical representations of send and received rate are shown in Figure 5.6 and 5.7 respectively.

![Figure 5.6: Rate Factor vs. Send Rate](image)

![Figure 5.7: Rate Factor vs. Received Rate](image)

From Figure 5.8, it is obvious that in UDP there is a large number of packet loss, because no congestion control mechanism is implemented in UDP. While in the case of SCTP a very minor packet loss is observed and in case of DCCP there is no packet loss. This is due to the congestion control mechanism of DCCP as whenever it detects the congestion in network it reduces the size of its congestion window, thus avoiding the packet loss.
Table 5.3: Rate Factor vs. Delay and Jitter

<table>
<thead>
<tr>
<th>Rate Factor</th>
<th>DCCP</th>
<th></th>
<th>SCTP</th>
<th></th>
<th>UDP</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Delay</td>
<td>Jitter</td>
<td>Delay</td>
<td>Jitter</td>
<td>Delay</td>
<td>Jitter</td>
</tr>
<tr>
<td>5</td>
<td>0.00573</td>
<td>0.0034</td>
<td>0.00599</td>
<td>0.00119</td>
<td>0.00542</td>
<td>0.00108</td>
</tr>
<tr>
<td>10</td>
<td>0.00916</td>
<td>0.00523</td>
<td>0.0082</td>
<td>0.00142</td>
<td>0.00721</td>
<td>0.0011</td>
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<tr>
<td>15</td>
<td>0.01284</td>
<td>0.00653</td>
<td>0.01028</td>
<td>0.00167</td>
<td>0.01002</td>
<td>0.00136</td>
</tr>
<tr>
<td>20</td>
<td>0.01665</td>
<td>0.00821</td>
<td>0.01478</td>
<td>0.00173</td>
<td>0.01189</td>
<td>0.00143</td>
</tr>
<tr>
<td>25</td>
<td>0.02049</td>
<td>0.01036</td>
<td>0.02053</td>
<td>0.00175</td>
<td>0.01407</td>
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</tr>
<tr>
<td>30</td>
<td>0.02559</td>
<td>0.01249</td>
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<td>0.00177</td>
<td>0.01528</td>
<td>0.0015</td>
</tr>
<tr>
<td>35</td>
<td>0.03041</td>
<td>0.01339</td>
<td>0.02017</td>
<td>0.00178</td>
<td>0.01639</td>
<td>0.00152</td>
</tr>
</tbody>
</table>

From Table 5.3, it is obvious that delay and jitter in case of UDP are decreased as compared to DCCP and SCTP. The reason is that UDP just keeps on sending the packets without checking the network condition. The packet is either received by the receiver or it is dropped. While SCTP and DCCP checks the network condition, if there is congestion in the network they reduce the send rate until the network condition gets better due to which the delay and jitter are more in SCTP and DCCP as compared to UDP.

3. Effect of multiple Mobile Stations on SCTP, DCCP and UDP

This scenario is based on multiple mobile stations which downloads the MPEG-4 traffic from the server through BS, as the server is connected to the BS through the wired link while the mobile stations are connected to the BS through the wireless link. This scenario is shown in Figure 5.9. This scenario is conducted for two different speeds. First the results are shown when mobile stations are moving with a speed of 3 m/s and then the results are shown when mobile nodes are moving with a speed of 5 m/s.
3.1. Mobile Stations moving with a speed of 3 m/s

Parameters used in this scenario are frequency bandwidth of 5 MHz. Downlink to uplink ratio is 0.7, means 70% of the frames are for downlink while 30% for uplink. The frame duration is 5 ms and the packet size is of 1200 bytes. The Modulation technique used is 64 QAM. The bandwidth of wired link is 100 Mb, the propagation delay is 1 ms and queuing mechanism is drop tail. The total simulation time is 55 s while the traffic starts at 25s and stops at 50 s. All the Mobile Stations have same parameters and they are generated on a location close to the BS, and then they linearly start moving away from the BS with a speed of 3 m/s.

Video traffic is multicasted towards the mobile stations with a constant rate factor of 5. The effect of increasing the mobile stations on performance of transport layer protocols is analyzed by using delay, jitter, packet loss and throughput. The performance measures calculated are shown in Table 5.4.

Table 5.4: Number of Mobile Stations vs. Send Rate, Throughput and Packet Loss

<table>
<thead>
<tr>
<th>Number of Mobile Stations</th>
<th>DCCP</th>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Send Rate (Mbps)</td>
<td>Send Rate (Mbps)</td>
<td>Throughput (Mbps)</td>
<td>Packet Loss</td>
<td>Throughput (Mbps)</td>
<td>Packet Loss</td>
<td>Throughput (Mbps)</td>
<td>Packet Loss</td>
<td>Throughput (Mbps)</td>
<td>Packet Loss</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>1</td>
<td>1</td>
<td>1</td>
<td>0</td>
<td>1.1</td>
<td>1.09</td>
<td>0</td>
<td>0.77</td>
<td>0.75</td>
<td>8</td>
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<td></td>
<td></td>
</tr>
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<td>0</td>
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<td>2.06</td>
<td>0</td>
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<td>3.03</td>
<td>0</td>
<td>3.15</td>
<td>3.1</td>
<td>0</td>
<td>3.05</td>
<td>2.98</td>
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<td></td>
<td></td>
<td></td>
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<tr>
<td>4</td>
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<td>0</td>
<td>4.1</td>
<td>4.03</td>
<td>2</td>
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<td>144</td>
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<td>5.1</td>
<td>7</td>
<td>5.02</td>
<td>4.8</td>
<td>366</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>6</td>
<td>6.19</td>
<td>6.16</td>
<td>0</td>
<td>6.06</td>
<td>5.9</td>
<td>25</td>
<td>6.08</td>
<td>5.71</td>
<td>697</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>7</td>
<td>7.05</td>
<td>7.02</td>
<td>0</td>
<td>7.3</td>
<td>7.15</td>
<td>81</td>
<td>6.77</td>
<td>6.24</td>
<td>1148</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>8</td>
<td>8.08</td>
<td>8.04</td>
<td>0</td>
<td>8.25</td>
<td>8.06</td>
<td>132</td>
<td>8.06</td>
<td>7.25</td>
<td>1845</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

From Table 5.4, it can be observed that increase in the number of MS increases the received rate or the throughput accordingly. By analyzing the table above it is noticed that performance of SCTP and DCCP is almost same in terms of throughput. It can be seen that
the overall performance of SCTP and DCCP is much better than UDP in terms of throughput as shown in Figure 5.10.

Figure 5.10: Number of Mobile Stations vs. Received Rate

UDP has large number of packet loss as compared to the SCTP and DCCP. The packet loss in SCTP is also observed when the number of MS is equal to 4 and so on. In case of DCCP zero packet loss is observed. Packet loss in these transport layer protocols is shown in Figure 5.11.

Figure 5.11: Number of Mobile Stations vs. Packet Loss
Table 5.5: Number of Mobile Stations vs. Delay and Jitter

<table>
<thead>
<tr>
<th>Number of Mobile Stations</th>
<th>DCCP</th>
<th>SCTP</th>
<th>UDP</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Delay</td>
<td>Jitter</td>
<td>Delay</td>
</tr>
<tr>
<td>1</td>
<td>0.00573</td>
<td>0.00339</td>
<td>0.00599</td>
</tr>
<tr>
<td>2</td>
<td>0.00827</td>
<td>0.00408</td>
<td>0.00733</td>
</tr>
<tr>
<td>3</td>
<td>0.01038</td>
<td>0.00462</td>
<td>0.01054</td>
</tr>
<tr>
<td>4</td>
<td>0.01219</td>
<td>0.00594</td>
<td>0.01156</td>
</tr>
<tr>
<td>5</td>
<td>0.01499</td>
<td>0.00669</td>
<td>0.01419</td>
</tr>
<tr>
<td>6</td>
<td>0.01858</td>
<td>0.00863</td>
<td>0.01669</td>
</tr>
<tr>
<td>7</td>
<td>0.02145</td>
<td>0.01043</td>
<td>0.01995</td>
</tr>
<tr>
<td>8</td>
<td>0.02498</td>
<td>0.01316</td>
<td>0.02011</td>
</tr>
</tbody>
</table>

By analyzing Table 5.5, it can be found that UDP gives less delay and jitter as compared to SCTP and DCCP. UDP has no idea about network condition and it keeps on sending the packets even in case of congestion in network, so either the packet is received by the receiver or it is dropped. While SCTP and DCCP keeps on checking the network condition, if there is congestion in the network, they decrease the send rate due to which delay and jitter are comparatively larger than UDP.

3.2. Mobile Stations moving with a speed of 5 m/s

Parameters used in this scenario are frequency bandwidth of 5 MHz. Downlink to uplink ratio is 0.7, means 70% of the frames are for downlink while 30% for uplink. The frame duration is 5 ms and the packet size is of 1200 bytes. The Modulation technique used is 64 QAM. The bandwidth of wired link is 100 Mb, the propagation delay is 1 ms and queuing mechanism is drop tail. The total simulation time is 55 s while the traffic starts at 25 s and stops at 50 s. All the Mobile Stations have same parameters and they are generated on a location close to the BS, and then they linearly start moving away from the BS with a speed of 5 m/s.

Video traffic is multicasted towards the mobile stations with a constant rate factor of 5. The effect of increasing the mobile stations on performance of transport layer protocols is analyzed by using delay, jitter, packet loss and throughput. The performance measures calculated are shown in Table 5.6.

Table 5.6: Number of Mobile Stations vs. Send Rate, Throughput and Packet Loss

<table>
<thead>
<tr>
<th>Number of Mobile Stations</th>
<th>DCCP</th>
<th>SCTP</th>
<th>UDP</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Send Rate (Mbps)</td>
<td>Throughput (Mbps)</td>
<td>Packet Loss</td>
</tr>
<tr>
<td>1</td>
<td>1.11</td>
<td>1.09</td>
<td>0</td>
</tr>
<tr>
<td>2</td>
<td>2.13</td>
<td>2.09</td>
<td>0</td>
</tr>
<tr>
<td>3</td>
<td>3.17</td>
<td>3.12</td>
<td>0</td>
</tr>
<tr>
<td>4</td>
<td>4.12</td>
<td>4.04</td>
<td>0</td>
</tr>
<tr>
<td>5</td>
<td>5.14</td>
<td>5.26</td>
<td>4</td>
</tr>
<tr>
<td>6</td>
<td>6.14</td>
<td>5.98</td>
<td>18</td>
</tr>
<tr>
<td>7</td>
<td>7.07</td>
<td>6.91</td>
<td>62</td>
</tr>
<tr>
<td>8</td>
<td>8.23</td>
<td>8.08</td>
<td>125</td>
</tr>
</tbody>
</table>
From Table 5.6, it can be observed that increase in the number of MS increases the throughput accordingly. By analyzing the table above it is noticed that performance of SCTP and DCCP is much better than UDP in terms of throughput. But overall, the performance of DCCP is better than SCTP in terms of both throughput and packet loss. By comparing Table 5.6 (when the speed of mobile stations is 5 m/s) with Table 5.4 (when the speed of mobile stations is 3 m/s), it can be noticed that overall there is some difference between the performance of SCTP and DCCP. It can be observed that in table 5.6 the throughput of DCCP is increased as compared to Table 5.4, while the packet loss is zero in both cases. The throughput of SCTP is almost same in both tables with a very minor change, while the packet loss in Table 5.6 is decreased. Comparison of UDP in both tables shows that its performance is same in both cases. The throughput of these transport layer protocols is shown in Figure 5.12.

![Figure 5.12: Number of Mobile Stations vs. Received Rate](image1)

Figure 5.12: Number of Mobile Stations vs. Received Rate

Figure 5.13 shows that UDP has large number of packet loss as compared to the SCTP and DCCP. The packet loss in SCTP is also observed when the number of MS is equal to 5 and so on. In case of DCCP zero packet loss is observed.

![Figure 5.13: Number of Mobile Stations vs. Packet Loss](image2)

Figure 5.13: Number of Mobile Stations vs. Packet Loss
By analyzing Table 5.7, it can be observed that UDP gives less delay and jitter as compared to SCTP and DCCP. UDP has no idea about network condition and it keeps on sending the packets even in case of congestion in network, so either the packet is received by the receiver or it is dropped. While SCTP and DCCP keeps on checking the network condition, if there is congestion in the network, they decrease the send rate due to which delay and jitter are comparatively larger than UDP.
6 CONCLUSIONS

6.1 Conclusion and Future work

In this thesis work, performance of SCTP, DCCP and UDP is evaluated by using MPEG-4 traffic over Mobile WiMAX/IEEE 802.16e. In order to analyze the performance of these transport layer protocols different performance metrics are used i.e. packet loss, delay, jitter and throughput. In each scenario the performance of DCCP is much better than UDP and SCTP.

By analyzing the results it is observed that the performance of DCCP and SCTP is much better as compared to UDP. The performance of SCTP and DCCP is almost same up to some extent but after that the performance of SCTP degrades while the performance of DCCP still remains constant.

By comparing SCTP and DCCP in terms of throughput it is observed that the performance of DCCP is better than SCTP because in case of SCTP packet loss is observed. While in case of DCCP zero packet loss is seen. The performance of UDP was also better in terms of throughput but the packet loss was maximum, which can badly affect the MPEG-4 video quality.

By comparing the performance of these transport layer protocols in terms of jitter and delay it is observed that delay and jitter in UDP are much small as compared to DCCP and SCTP. But because of the bad performance in terms of packet loss and throughput UDP is not the suitable transport layer protocol for the transport of video traffic as it can badly affect the quality.

After comparing three protocols, SCTP, DCCP and UDP it is concluded that DCCP gives better QoS requirements in terms of jitter, delay, throughput and packet loss and thus it can be used for efficient transportation of the MPEG-4 video traffic over IEEE 802.16e. The reason is that it gives much better throughput and less packet loss as compared to SCTP and UDP. The reason behind better performance of DCCP is the congestion control mechanism which avoids the congestion in network.

In future, this research work can be extended by including multiple cells and analyzing the effect of handovers on the performance of these transport layer protocols for MPEG-4 traffic over Mobile WiMAX. Furthermore, the performance of SCTP, DCCP, UDP and other related transport layer protocols can be evaluated by transporting MPEG-4 traffic and VoIP application over the wireless access technologies like LTE, HSDPA, and WCDMA etc.
REFERENCES


