CHAPTER 4

MULTIMEDIA STREAMING TRANSMISSION IN MANET

4.1 INTRODUCTION

MANETs have been the subject of intense research for a number of years. In the past, many research works have been conducted on the relational database model which was postulated to the multimedia database model. More recently, the relational database model was speculated to the data streams model, due to the advancement in technology, wherein the data became massive and confounded in size due to the employment of ad hoc type of networks. This work considers one more steps of generalization by providing a multimedia stream transmission in MANET. The objective of this work is to define and to standardize the functionality of multimedia streaming transmission in ad hoc networks.

The work scrutinizes the usefulness and services that can be incorporated into the operations of wireless telecommunication networks for disseminating multimedia streams. Compared to nodes in networks with availability of infrastructure, the nodes in MANETs do not have a pre-existing infrastructure. This necessitates the nodes in MANET to perform operations using intermediate nodes, on behalf of other nodes.

The paramount objective in this work is to explore the appropriateness of mobile ad hoc networks for streaming multimedia files, using the basic structure as illustrated in Figure 4.1.
Figure 4.1 Structure of Mobile Ad hoc Networks

A subject of consideration, namely multimedia streaming transmission in MANET is the arrangement and interpretation of so-called forwarding data in multi-path manner over mobile ad hoc networks. As an alternative of sending data through a single communication channel, the data packet is breached up into multi-paths, which selects a different routing mechanism through the network. Goudarzi P. and Hosseinpour M. (2012) initially explained the basics of streaming of data and the introduction of multimedia streaming in the multi-path transmission over mobile ad hoc networks.

With the continuation of multi-path encrypted data security architecture, RDSA in Mobile Ad-hoc Network, an illustrative study and analytic method, to improve multiple paths routing efficiency in frequent communication failures due to channel interferences are introduced. Further, reliable data security architecture is planned to implement an application such as multimedia streaming to provide better bandwidth allocation and reduce transmission delay. Simulations are carried out with DSR and AODV protocols for efficient multi-path multimedia data transmission security scheme introduced in this study.
4.1.1 Basics of streaming of data

The basic principle behind the streaming of data is explained with the help of the diagram shown in Figure 4.2. The three items represented in streaming of data comprise “Begin”, “Slide”, “Tuples” and finally the actual data stream. The “begin” item represents when to begin emitting results, “tuples” item represents the resultant value at a particular interval of time and finally “slides” indicate the emitting of new sets. The tuples also contain the time stamping which specifies the implicit and explicit value. The implicit value is based on the time stamp given, when items arrive and explicit value is derived from data field in data. The representation of time may be either “physical” denoted by integer or “logical” denoted by date.

![Diagram of Streaming Data]

**Figure 4.2 Streaming Data**

4.1.2 Multimedia Streaming

Michele Nitti et al. (2012), referred to streaming media as multimedia that are received in a constant manner and frequently presented to an end-user with the help of the streaming provider. Streaming media transmission refers to the delivery method of the medium rather than to the medium itself. The distinction is used for different types of streaming that are distributed over
telecommunications networks, as most other delivery systems are either congenitally streaming or inherently non-streaming.

![Figure 4.3 Multimedia Streaming Data](image)

Multimedia streaming transmissions in MANET include services where continuous multimedia, audio and video data are presented by Tarek R. Sheltamia C. et al. (2011) to an end user as illustrated in Figure 4.3. Multimedia streaming transmission enables users to form perspective videos. The dependent system points out the delay once the end user starts perceiving the data. This is in converse to other methodologies which require the end user to stay tuned for an interval in order to ensure that the entire media files get downloaded before they are received and viewed by the end user. As they require several minutes and even more than that to download the entire multimedia files, multimedia streaming transmission over MANET offers the gratification of being able to illustrate the multimedia as soon as it begins to download.
4.1.3 Protocol for Multimedia Streaming

The services of multimedia streaming transmission in MANET are unbalanced between the sender and the receiver. This is because the streaming of video only flows in one direction, from the server to the clients. On the sender side, the multimedia streaming transmission services include the content formation and communication. The video stream is encoded from the video and audio signals while the receiving terminal decodes the data with the corresponding video and audio decoders.

The output acquired is then sent to be processed back at local audio and video devices for actual retrieval purpose. The multimedia streaming transmission in MANET includes system control protocols. It provides the setting up connections between clients. It involves the streaming transmission, adjudicating different options and capabilities. It controls and disseminates the various source codec that the multimedia streaming transmission in MANETs use as illustrated in Figure 4.4.

![Figure 4.4 Multimedia Streaming Transmission](image-url)
A provoking and challenging task when implementing Multimedia streaming transmission is to reinforce the preserved bandwidth required to display MST objects, which include audio, video objects. When compared to conventional data types, which include records, still images and text images, MST objects are generally large in size. For example, a three hour MPEG-2 encoded movie requires approximately 4.6 Giga Bytes of storage.

Any discrepancy from the real time requirement results in unsuitable artifacts, disruptions, and jitters, called as hiccups. The work introduces a new enhancement for Feedback-based Real-time Transport Control Protocol together with the conventional RTCP which provides the information required for feedback-based transmission control protocol.

The objective of this work is to present a RTCP feedback-based transmission control protocol. It includes RTCP feedback information containing the extension by conventional RTCP mechanism. It enhances the transmission rate at the server level and alleviates the aforementioned problems related to transmitting of multimedia wireless stream in MANET.

Real Time Control Protocol provides out-of-band statistics and control information for an RTP flow. It partners with RTP in the delivery and packaging of multimedia data, but does not transport any media streams by itself. Typically RTP will be sent on an even-numbered UDP port, with RTCP messages being sent over the next higher odd-numbered port. The primary function of RTCP is to provide feedback on the quality of service in media distribution by periodically sending statistics information to participants in a streaming multimedia session.

RTCP and MRTCP protocols use acknowledgements to take feedback data from network (e. g. QoS reports) to adapt the sender with network condition. It uses the acknowledgements to adapt the rate of Forward Error Correction data which will be sent together with the next outgoing packets. RTCP gathers
statistics for a media connection and information such as transmitted octet, packet counts, lost packet counts, jitter, and round-trip delay time. An application may use this information to control quality of service parameters, perhaps by limiting the flow of different codec.

Take two limits into consideration for adjusting the rate of returned ACKs. On the one hand, the dispatcher application prefers high rate of returned ACK to get knowledgeable of receiver status as soon as possible. But on the other hand, the high rate of packets in feedback channel allocates some part of valuable bandwidth to itself and the bandwidth could be used for the primary data itself to increase the quality of transmitted content. The circumstances get even worse in wireless ad hoc networks because the number of active flows will augment if each receiver wants to have an ACK flow towards its sender.

The more flow of data in ad hoc networks implies the more competition for acquiring the channel and in consequence there is more delay and overhead due to unsuccessful signal. Also because of channel interference property of MANET, the probability of the interference flanked by flows will be more when the number of flows increases in the network. Hence, even though the receiver sending ACKs back in higher rates should also consider the downside of increasing the rate.

**4.2 LINK RELIABILITY**

With the comprehensive utilization of Personal Digital Assistants and other devices, streaming multimedia content between mobile ad hoc networks is considered a prominent application. The streaming of video clips, drawings and articles from one client to another client is performed very well according to Yueh Min Huang et al. (2007) if the devices are close to each other. One of the confrontations in sharing multimedia stream among mobile peers is to distribute the content, commonly huge in size, over a mobile ad hoc network where link
possibility is invariably transposing. In order to achieve smooth media performance, link reliability is of crucial importance for media streaming in Mobile ad-hoc Networks. Thus, the timeless link availability compels the accomplishment of streaming multimedia files.

One frequently used mobility model called the Random Way Walk Mobility Pattern model over mobile ad hoc network is discussed in section 4.2.1. It uses the problem statement provided in 4.2.2 for known velocity for both the nodes as in Figure 4.5.

![Diagram](image)

**Figure 4.5 Multimedia compositions distributing amid PDAs**

When a user in mobile ad hoc network identifies that another user in his neighborhood has a picture he is interested in, the feasibility that he can stream that picture before the link breaks up is of special interest to him. For example, a typical size of picture is 3.5 x 5 or 4 x 6 (dimensions). If the available bandwidth is 256 kbps, a link has to be deliberately available for 130 seconds for a picture to be successfully drifted. If the file is available from multiple nodes in
an ad hoc nature, the user chooses the source that has the highest feasibility to
stay associated for 130 seconds.

As illustrated in Figure 4.5, the connectivity between A and B and that
between B and C are compelled by B’s communication range. If both node A and
C have the picture that B seeks, it is beneficial to select the node A as the source
as A and B are advancing in an analogous direction. Therefore, presenting a
concrete indication of repeated link availability helps the nodes in mobile ad hoc
networks by persuading the best choice among different number of sources.

4.2.1 Random Way Walk Mobility Pattern

Due to the inconsistent nature of human movement, computing the
future link availability is one of the major challenging tasks. When compared to
the previous work, as discussed by the author M. Qin et al. (2006), the research
work in focus introduces an insistent method for anticipating the future link
possibility based on the RWMP. The work multimedia streaming transmission in
MANET uses the Random Way Walk Mobility Pattern model to associate the
speed and the direction of change and the behavior of the nodes.

The time taken for direction change is not an input parameter model,
but the work purely depends on the speed of the nodes at which the shape of the
area of network changes. The movement epoch length is defined as the time
between two direction change events. An example of node movement in Random
Way Walk Mobility Pattern is illustrated in Figure 4.6.

The work proposed by M. Qin et al. (2006) proceeded with the
assumptions that all nodes in MANET involving speed and direction are
distributed in an uniform manner between $[0, V_{max}]$ and $[0,2\pi]$ respectively. As
such, if any one or both the nodes’ current velocity is determined, this algorithm
may not be used to predict the future link status of nodes in MANET. In this work, the researchers broaden the result obtained from the previous work as discussed by the author to provide an accurate model in order to predict the future link availability.

![Diagram](image)

**Figure 4.6 Example of node movement in Random Way Walk Mobility Pattern**

### 4.2.2 Problem Statement

For ease of reference, the proposed work proceeds with the assumption that two mobile peers of a link, consist of the same radio coverage, with a circle of radius r. The original distance between the two mobile nodes is proximate by \( D_0 \) \( (D_0 \leq r) \). An accurate estimation of \( D_0 \) is obtained through Derived Signal Stability and Time of Occurrence. As illustrated in Figure 4.7, let \( V_1 \) and \( V_2 \) represent the velocities of the two nodes A and B. The link will be available continuously from \( T_0 \) to \( T_0 + T \) when \( V_1 \) and \( V_2 \) are unknown.

In order to evaluate the velocities of single or both nodes represented in MANET, the work provides an enhancement by including the calculations from \( L(D_0, T) \) to \( L(D_0, T, A) \) and \( L(D_0, T, B) \) where \( L \) denotes the length. Let \( \gamma_A \) and \( \gamma_B \)
denote the movement epoch lengths of two mobile nodes A and B. The probability that the two nodes A or B do not change their velocity with T seconds is \( E^{\gamma AT} \) or \( E^{\gamma BT} \).

![Figure 4.7 Nodes current link status](image)

In order to determine the mobility of a mobile node, multimedia streaming transmission in MANET uses the random way walk mobility pattern. The basic principle behind the Random Way Walk Mobility Pattern is that the model is divided into a progression of intervals referred to as the mobility epochs. Each epoch represents the random period of time which is again exponentially distributed with mean \( \lambda_n^{-1} \). Meanwhile, during each mobility epoch, a node in MANET proceeds with a constant speed and direction. The speed consists of the random variable which is uniformly distributed between \([0, v_{\text{max}}]\) and the direction is uniformly distributed over \([0, 2\pi]\). In this way, a node proceeds in a straight manner for a longer period of time.

### 4.2.3 Known Velocity for both nodes

When both the velocity \( V_1 \) and \( V_2 \) are known, formulate a more accurate prediction of future link availability using the random way walk
mobility pattern. If neither node changes its velocity during \( (T_0, T_0 + T) \) the distance between A and B at \( T_0 + T \) will be:

\[
D(t) = \sqrt{(D_0 + V_1 T \cos \theta - V_2 T \cos \theta)^2 + (V_1 \sin \theta - V_2 \sin \theta)^2} T^2
\]  

(4.1)

Let \( T \) denote the time at which there is a break in link between node A and B when both of the nodes keep their current velocities. Let \( C_0(T) \) represent the probability that both nodes A and B are connected in a continuous manner and at the same time, an assumption follows that there is no velocity change to the link during \( (T_0, T_0 + T) \). The probability that there occurs no change in velocity within a period of \( T \) seconds is \( E^{(\gamma A - \gamma B)^T} \). Hence \( C_0(T) = E^{(\gamma A - \gamma B)^T} \)

### 4.3 DESIGN OF PROTOCOL – MULTIMEDIA STREAMING

Mobile ad hoc networks consist of a compilation of wireless mobile nodes which energetically exchange data among themselves without the reliance on a fixed base station or a wired backbone network. MANET nodes are normally illustrious by their restricted power, processing and memory resources as well as high degree of mobility. In such networks, the wireless mobile nodes may animatedly enter the network as well as go away from the network. Due to the incomplete broadcast range of wireless network nodes, multiple hops are typically required for a node to swap information with any other node in the network. Thus routing is a vital issue to the design of a MANET.

This section discusses the problems related to the uses of different protocols for supporting multimedia streaming transmission in MANETs and the protocol selected for the research work uses an illustration shown in Figure 4.8 it is discussed with the help of components and algorithm as in 4.3.1, 4.3.2 and 4.3.3 respectively.
Multimedia streaming transmission requires higher bandwidth when compared to the conventional applications like file transferring. Supporting multimedia streaming transmission over MANET poses varied number of practical problems. There are certain issues related to the design of a network protocol to support streaming media transmission in MANET. Some of the protocols comprise User Datagram Protocol, Real-time Streaming Protocol, Real-time Transport Protocol and the Real-time Transport Control Protocol and finally the Transmission Control Protocol.

The Transmission Control Protocol, the most prevalent used protocol, is not significant for applications which include multimedia streaming in mobile ad hoc network. The reason behind is that the protocol depicts a missing packet as a symptom of network congestion which is not always the case mobile ad hoc networks. Due to the basic design as structured in TCP, the TCP protocol proceeds to the wireless losses in the same manner as it reacts to packet losses due to congestion.
Multimedia streaming transmission is also transported using User Datagram Protocol. The UDP works on the principle that it sends the media streams as a series of small packets. Though simple by mechanism, the protocols do not guarantee the delivery. It is up to the receiving application to detect loss or corruption and recover data using error correction techniques. If data is lost, the stream may suffer a dropout. At the same time, the problem behind UDP is that the protocol does not support any congestion control mechanism. The congested network may cause degradation on the performance of network.

The Real Time Streaming Protocol, Real Time Transport Protocol and the Real-time Transport Control Protocol were specifically designed by Wei W. and Zakhor A. (2004) to stream media over networks. Reliable protocols, such as the Transmission Control Protocol guarantee the correct delivery of each bit in the media stream. However, they accomplish this with a system of timeouts and retries, which makes them more complex to implement.

The propagation of wireless networks is compelling a subversive advancement in information sector. Due to the possibility of wireless integrations on mobile devices ad hoc networks are realizing and gaining popularity day by day. The ad hoc networks provide an extremely higher data rate which result in and make the multimedia applications accessible. The usage of these applications through mobile ad hoc networks is gaining immense popularity nowadays. A mobile ad hoc network consists of a scenario where the network consists of different mobile nodes associated by wireless links. The node present in the network not only acts as the end-system but also as a router. The nodes present in the MANET move in a random manner and the organization of the nodes is performed in an arbitrary manner.

This random management of ad hoc networks motivates rapid and unpredictable changes in the topology of the overall system. The motivation for
the research work is to design real time streaming protocol for multimedia streaming transmission over mobile ad hoc networks which can provide feedback with better results when compared to the conventional TCP, UDP and RTSP based protocol. The work presented here uses Real Time Streaming Transmission for designing Multimedia Streaming Transmission in MANET. The routing protocol provides the feedback through which the sender node differentiates between congestion and non-congestion loss.

4.3.1 Feedback-based RTSP for Multimedia Streaming Transmission

The Feedback-based RTSP for multimedia streaming transmission comprises a network control protocol designed for use in entertainment and communication systems to control streaming multimedia transmission. The protocol is used for authorizing and administering multimedia sessions between end points.

4.3.2 Feedback-based RTSP Components

The Feedback-based RTSP is a client-server multimedia streaming transmission-based protocol that enables controlled delivery of streamed multimedia transmission over mobile ad hoc networks. The Feedback-based RTSP provides mechanisms to obtain commands like play, pause, stop, fast-forward, rewind and so on and these are similar to the methodology provided by CD players. The mechanism in Feedback-based RTSP is explained below with the help of flow diagram in Figure 4.9.

(i) **Revival of media file from media server**: The client requests a description regarding the file to be presented for retrieval and asks the server to provide with a setup session to send the requested media file.
(ii) **Motivating a media server to a conference:** The media server is then invited to the conference to play back media file.

(iii) **Updating the media file to an existing presentation:** The server and the client now notify each other about any additional media becoming available for transmission or retrieval.

![Feedback-based RTSP diagram]

**Figure 4.9 Feedback-based RTSP mechanisms**

The formation, storage and transmission of ubiquitous, high performance multimedia streaming transmission services over a mobile ad hoc network present an imposing provocation to the indispensable computing framework. Computers with rapid processors and storage systems make it attainable to reinforce simultaneous displays for streaming multimedia services over mobile ad hoc networks. The research work provides an enhancement to the conventional RTSP protocol to provide feedback which is another driving force of the multimedia applications. The Feedback-based RTSP components are
illustrated in Figure 4.10 which comprises (i) Multimedia Server (ii) Content Description (iii) Content Base (iv) Feedback and (v) Actual Contents.

**Figure 4.10 Components of Feedback-based RTSP**

The objective of Multimedia servers is to store and manage multimedia objects and deliver multimedia stream transmissions in response to requests generated from the users. The paramount objective of a Multimedia Server (MS) is also to preprocess the information before delivering it to users. The content may range from long files to audio, video or image files.
In order to manage the objects, the MS provides computation capabilities comprising of storage, manipulation and finally distribution. Regardless of the capacity of multimedia server, the purpose of multimedia servers involves providing high throughput and a smooth display without any disruptions or jitters. The Content description contains the format, content and methods to access. The actual content is presented in the Base content.

A feedback mechanism is used to provide the feedback to the sender or the receiver regarding the stream data. Dynamic content-based mechanism is incorporated in the feedback-based RTSP model over Mobile ad hoc network. To demonstrate how the objective Feedback-based RTSP can be used to provide content-based adaptation, a prototype client-server system was developed. The client returns feedback-based RTCP comprising information about loss, delay occurred and the details regarding the congestion. As soon as the server receives the feedback from the client, the server acts according to the FRTSP algorithm.

**4.3.3 Feedback-based Real Time Streaming Transmission Algorithm**

The FRTST algorithm as shown in Figure 4.11 uses RTSP as a means of real time streaming of multimedia data over mobile ad hoc networks. Initially, the client sends the SETUP requests. When the server receives the feedback regarding the candidate lists from the client, the server sets up the media and starts connectivity. After the reception of PLAY request from the client, the server starts to send the media file.
Step 1: The RTSP client sends SETUP requests.

Step 2: After receiving the candidate lists from the clients the RTSP server associates its own clients.

Step 3: The RTSP server once gets ready for reception, responds to the SETUP request.

Step 4: The server starts the connectivity.

Step 5: The client receives the SETUP response and identifies the candidate’s address.

Step 6: When the connectivity check reaches from client to the server, it results in a triggered check.

Step 7: Now the client issues the PLAY request.

Step 8: Upon reception of the PLAY request from the client, the server plays the stream.

Figure 4.11 Feedback-based Real Time Streaming Transmission Algorithm

4.4 MULTIMEDIA STREAMING TRANSMISSION IN MANET

While multimedia streaming transmission in ad hoc type of network is one of the most important services for mobile nodes, obtaining competent quality from an end user point of view is a significant work due to the involvement of differing conditions like wireless links and the mobile terminals limited capabilities in an ad hoc network. Figure 4.12 demonstrates the multimedia streaming transmission scenario. The server holding the content sends
multimedia streaming data to the client through an IP network, using UDP as transport protocol. The IP network is divided into three elements consisting of

(i) A public network
(ii) The core network of the mobile operator and
(iii) Finally the wireless link.

After the multimedia data are being transmitted through this wireless link, packets are received in the client buffer.

This diagram describes the multimedia transmission. The group of multimedia stream data is present in a server. This stream of data contains in the public network and it then sends to the core network through the wireless link. The data sent through the wireless link are also present in the client buffer.

Figure 4.12 Multimedia Streaming Transmission Scenario
The common file system measurements for streaming media storage size are measured in the range of mebibytes, megabytes, gigabytes, terabytes, and so on. These are calculated from streaming bandwidth and length of the media using the following formula:

\[
\text{Storage size (in mebibytes)} = \text{length (in seconds)} \times \text{bit rate (in bit/s)} / (8 \times 1024 \times 1024)
\]

(4.2)

Where Mebibyte = \(8 \times 1024 \times 1024\) bits.

One hour of video encoded at 300 kbit/s (this is a typical broadband video and it is usually encoded in a 320×240 pixels window size) will be:

\[
(3,600 \text{ s} \times 300,000 \text{ bit/s}) / (8 \times 1024 \times 1024) \text{ give around 128 MiB of storage.}
\]

If the file is stored on a server for on-demand streaming and if this stream is viewed by 1,000 people at the same time using a Unicast protocol, the requirement is:

\[
300 \text{ kbit/s} \times 1,000 = 300,000 \text{ kbit/s} = 300 \text{ Mbit/s of bandwidth}
\]

(4.3)

This is equivalent to around 135 GB per hour. Of course, using a multicast protocol the server sends out only a single stream that is common to all users. Hence, such a stream would only use 300 kbit/s of serving bandwidth. The following sections provide more information on these protocols.

### 4.4.1 Multimedia Multi-path Transmission

The basic structure of Mobile ad hoc networks is designed in such a way that the nodes in MANETs move randomly at a faster interval of time which does not have a predetermined infrastructure or a fixed based station to monitor the movement of nodes in network. The nodes in MANET are typically
distinguished by their limited memory resources, limiting power and processing
capabilities as well as high degree of mobility.

In such types of networks, the mobile nodes dynamically enter into the
network as well as leave the network. Due to the availability of limited
transmission range of wireless network nodes, multiple hops are usually required
for a node to exchange information with any other node in the network. Thus
routing is a crucial issue to the design of a MANET.

The work examines the issues of multi-path routing in MANETs. Multi-path routing allows the establishment of multiple paths between a single
source and single destination node. It is typically proposed in order to increase
the reliability of data transmission (i. e. fault tolerance) or to provide load
balancing. Load balancing is of special importance in MANETs because of the
limited bandwidth between the nodes. The multipath routing supports application
constraints such as reliability, load-balancing and Quality-of-Service.

4.4.2 Streaming flow through Multiple-path

The whole multimedia stream is transmitted through a single path or
the content is divided into multiple minor flows and then streamed through the
available paths. In the general case, it might have N paths and M minor flows
(M >= N). The benefit of having integrated path is that the experienced delay by
packets is the same approximately. Hence, the receiver senses less jitter and as a
consequence, smaller buffers at the receiver and sender are required.
Furthermore, reordering in TCP triggers unwanted fast-retransmission events and
considerably degrades the throughput. One drawback of integrated path is
providing with the delay of switching to a new path when the primary route
breaks.
This delay might be intolerable for real-time applications. To address the switching delay, the work proposes transmission of a copy of data in each path. Obviously, the redundancy of this approach is accurate and hence not quite efficient. There is a need to propose alternate paths separated from the primary path in the middle of the way. In the case of failure, content could be switched to the next available alternate path by the intermediate node in order to reduce the loss overhead.

4.4.3 Flow Bit Rate and Stale Route

In solitary flow approaches, the other paths will be worn as backup, when the primary path brakes. The difficulty that arises is that one could not be sure that the backup route itself has not broken yet. Some alternate approach comes up with the solution to provide with two routes whereas the infrequent fresh packets are sent through the option paths to make sure that their sate (path brakes, transmission) is up to date. Some others divide the main traffic through available paths in a round Robin manner.

The alternate approach in providing with the multi-flow streaming algorithms is that how much bit-rate to be linked to each flow. The easy solution is to assign the same bit-rate is to all of the flows. The alternate solution is to allocate the bit-rate based on the excellence of the link. This quality could be intended using alternate multi path routing to produce multiple paths. A more complicated move is based on the updating of bit-rate. The flow is based on the bandwidth of the path.

Stale route often contains route caches information for an extended period of time. The erased stale routes are possibly un-erased due to in-light carrying of data packets. When a node has invalid route in its route cache, it attempts to transmit a number of data packets while consuming energy without any success. While the main cause of the stale route problem is node mobility, it
is the unconditional overhearing that dramatically aggravates the problem. While
the primary route is checked for its validity during data communication between
the source and the destination, alternate routes remain in route cache unchecked
even after they become stale. This is the case not only for the nodes along the
alternative routes, but also for all their neighbors because of unconditional
overhearing.

4.4.4 Channel Interferences in Multi-path Routing

The Feedback-based RTCP provides out-of-band statistics and control
information for an RTP flow. It provides a hand-shake method with RTP in the
delivery and packaging of multimedia data, but does not transport any media
streams itself. Typically, the packet in RTP is sent on an even-numbered UDP
port, with RTCP messages being sent over the next higher odd-numbered port.
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other hand, the high rate of packets in feedback channel allocates some part of valuable bandwidth to itself and the bandwidth could be used for the primary data itself to increase the quality of transmitted content. The circumstances get even worse in wireless ad hoc networks because the number of active flows will augment if each receiver wants to have an ACK flow towards its sender.

In ad hoc networks, more flow of data implies more competition for acquiring the channel and in consequence there is more delay and overhead due to unsuccessful signaling. Also because of channel interference property of MANET, the probability of the interference flanked by flows will be more when the number of flows increases in the network. Hence, even though the receiver sends ACKs back at higher rates, it should also consider the downside of increasing the rate.

4.5 RESULTS AND DISCUSSIONS

The proposed algorithm simulated an ad hoc network with varying densities and transmission ranges. Each node had a random location in a 1000 x 1000 area. Node also had a random amount of initial power within a range of 700k to 1500K. The thresholds were varied from 700k to 1000K.

The life of the network depends on various factors, which include the initial threshold, density of the network and range of the transmitters. These parameters need to be adjusted for each network. The proposed method presented here studied the various power saving techniques employed by mobile devices. The initial results show the various parameters affect the lifetime of the network. At this point, the nodes’ personal tasks are not considered. But in the future, the power will take into account to see its impact on the lifetime of the network.
In this proposed method, the video is split into ‘n’ parts and transmitted in multi-path based on the availability of the nodes. The source and the destination for the transmission are visible. Eventually, the video is multicast from the source to all nodes. Throughput is the number of useful bits per unit of time forwarded by the network from a certain source address to a certain destination, excluding protocol overhead and excluding retransmitted data packets. Throughput is the amount of digital data per time unit that is delivered over a physical or logical link or that is passing through a certain network node.

Delivery Ratio = \frac{(\text{Number of Packets Received})}{(\text{Number of packets Sent})} \quad (4.4)

Delay is defined as the average time taken by the packet to reach the server node from the client node.

Delay = \frac{(\text{Number of packets Received})}{(\text{Simulation Time})} \quad (4.5)

Pause-time is the time for which a packet stops in when it reached a destination after the travel from the place of origination. The unit of pause-time is seconds. Mobility is the velocity with which a node moves from the source to a destination. It is usually specified in m/s. Dropped packets are the number of packets dropped due to the effect of link breaks and they may be control packets or data packets.

Simulation result shows that the video quality of multiple path multicast video communication is significantly higher than that of single path multicast video communication, with similar routing overhead and forwarding efficiency. The video multicasting technique applied in this regard is much superior to the existing technique. By the extensive use of AODV protocol, the QoS parameters, namely delay and throughput have significantly improved. From the simulation results the delay has been reduced by 0.5 s and the
throughput has been increased by 5%. Wireless multicast is required for a range of emerging wireless applications employing group communication among mobile users.

### 4.5.1 Routing Overhead

Routing overhead, defined as congestion occurs in the route during the transmission of packets from a source to a destination path. Here the routing overhead refers to the packet transfer rate.

**Table 4.1 Number of groups vs Routing overhead**

<table>
<thead>
<tr>
<th>Number of groups (100 nodes per group)</th>
<th>Routing overhead(%)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Multimedia multipath Transmission</td>
</tr>
<tr>
<td>1</td>
<td>7</td>
</tr>
<tr>
<td>2</td>
<td>10</td>
</tr>
<tr>
<td>3</td>
<td>12</td>
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<tr>
<td>4</td>
<td>15</td>
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<td>5</td>
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<td>6</td>
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<td>7</td>
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<td>8</td>
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<tr>
<td>9</td>
<td>25</td>
</tr>
<tr>
<td>10</td>
<td>27</td>
</tr>
</tbody>
</table>

Table 4.1 describes the routing overhead arising while the members in the group leave the domain in MANET. A larger routing overhead arises, when a member leaves the domain of the MANET. In the multimedia multipath transmission, routing overhead is less compared to single path multipath
transmission scheme. The table compares the routing overhead of the proposed Multimedia multipath Transmission with an existing multimedia single path transmission method.

![Graph](image)

**Figure 4.13 No. of groups vs Routing overhead**

Figure 4.13 describes the process of routing overhead formed in MANET. In the Multimedia multipath Transmission, the packet data have been transferred using data security process and so there is less chance in routing overhead.

The performance graph of the proposed Multimedia Multipath Transmission in routing overhead is shown in Figure 4.13. When the number of groups in the MANET increases, the routing overhead decreases gradually in the Multimedia multipath Transmission when compared to the existing system. The variance in the routing overhead for routing the packet data from source to destination would be 15-25% low in the proposed Multimedia multipath Transmission and produces good results.
4.5.2 Throughput

Throughput is defined as the average rate of successful message delivery over a communication channel from one group to another group in MANET. The throughput is generally calculated in bits per second (bit/s or bps) and sometimes in data packets per second or data packets per time slot.

Table 4.2 Number of messages in groups vs Throughput

<table>
<thead>
<tr>
<th>Number of messages in groups</th>
<th>Throughput (%)</th>
<th>Multimedia multipath Transmission</th>
<th>Multimedia single path Transmission</th>
</tr>
</thead>
<tbody>
<tr>
<td>10</td>
<td>70</td>
<td>10</td>
<td></td>
</tr>
<tr>
<td>20</td>
<td>75</td>
<td>19</td>
<td></td>
</tr>
<tr>
<td>30</td>
<td>78</td>
<td>25</td>
<td></td>
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<tr>
<td>40</td>
<td>82</td>
<td>32</td>
<td></td>
</tr>
<tr>
<td>50</td>
<td>86</td>
<td>35</td>
<td></td>
</tr>
<tr>
<td>60</td>
<td>90</td>
<td>38</td>
<td></td>
</tr>
<tr>
<td>70</td>
<td>92</td>
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<td>95</td>
<td>52</td>
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<tr>
<td>90</td>
<td>96</td>
<td>59</td>
<td></td>
</tr>
<tr>
<td>100</td>
<td>99</td>
<td>60</td>
<td></td>
</tr>
</tbody>
</table>

Table 4.2 describes the throughput for the successful delivery of messages over MANET. The table compares the throughput of the Multimedia multipath transmission with Multimedia single path transmission.
Figure 4.14 No. of messages in groups vs Throughput

Figure 4.14 shows the throughput for successful delivery of messages in MANET using Multimedia multipath transmission. In the Multimedia multipath transmission, the delivery rate of packet data is less. In the Multimedia multipath Transmission the packet data have been transferred using data security and the routing discovery process is made at first.

The performance graph of the Multimedia multipath Transmission in throughput is shown in Figure 4.10. As the messages in the group increased, the throughput for the packet delivery also increased drastically. The variance in the throughput for delivery of packet data from source to destination would be approximately 50% high in the Multimedia multipath Transmission, compared to Multimedia single path Transmission and produces the best result.
4.5.3 Security Group

Security Group is termed as members of group and it can access only their authorized data and none other group in MANET can access those data which are authorized to that group alone. The security level of the group is high.

Table 4.3 Security level of group in MANET

<table>
<thead>
<tr>
<th>Method</th>
<th>Security level of group in MANET (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Multimedia multipath Transmission</td>
<td>90</td>
</tr>
<tr>
<td>Multimedia single path Transmission</td>
<td>40</td>
</tr>
</tbody>
</table>

Table 4.3 indicates the security level of group for data communication in MANET. The table shows the comparison of security level of group in MANET using the multimedia single path transmission. Comparison shows that the security level of multimedia multipath transmission is higher when compared to multimedia single path transmission.

Figure 4.15 Security level of group in MANET
Figure 4.15 exhibits the security level of the group formed by Multimedia multipath transmission in MANET. In the Multimedia multipath transmission the packet data have been transferred using multimedia stream and there is less chance for accessing the data to attacker/ unauthorized user.

The performance graph of the Multimedia multipath transmission in security level is shown in Figure 4.15. As the security level increases, the proposed system performs better when compared to the Multimedia single path Transmission in MANET. The variance in the security level of the group for routing the packet data from source to destination would be 50-70% higher in the Multimedia multipath transmission compared to the Multimedia single path transmission.

4.5.4 Bandwidth

**Bandwidth** refers to the maximum data transfer rate of a network. It measures how much data can be sent over a specific connection in a given amount of time.

Table 4.4 depicts the bandwidth efficiency obtained when the number of nodes increases in the mobile ad hoc network environment. The outcome of the Multimedia multipath transmission in MANET is compared with a Multimedia single path transmission.
Table 4.4 No. of Nodes vs Bandwidth efficiency

<table>
<thead>
<tr>
<th>Number of Nodes</th>
<th>Bandwidth Efficiency (%)</th>
<th>Multimedia multipath transmission</th>
<th>Multimedia single path transmission</th>
</tr>
</thead>
<tbody>
<tr>
<td>100</td>
<td>75</td>
<td>42</td>
<td></td>
</tr>
<tr>
<td>200</td>
<td>80</td>
<td>45</td>
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<tr>
<td>300</td>
<td>82</td>
<td>48</td>
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<tr>
<td>400</td>
<td>83</td>
<td>51</td>
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<tr>
<td>500</td>
<td>86</td>
<td>54</td>
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<tr>
<td>600</td>
<td>84</td>
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<tr>
<td>700</td>
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<tr>
<td>800</td>
<td>86</td>
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<tr>
<td>900</td>
<td>89</td>
<td>57</td>
<td></td>
</tr>
<tr>
<td>1000</td>
<td>92</td>
<td>59</td>
<td></td>
</tr>
</tbody>
</table>

Figure 4.16 shows the simulation results for the bandwidth efficiency compared with the Multimedia multipath transmission and Multimedia single path transmission. In ad hoc networks, the link level bandwidth is used in multimedia provisioning for end-to-end flows. If an end-to-end flow traverses numerous hops in the link layer, then the bandwidth assigned to such flow is measured by the ability of the bottleneck link. To provide multi-path routing and to carry out tasks such as data control, an end-to-end flow’s requested bandwidth
is verified against the link layer bandwidth hop-by-hop to discover a possible path.

![Graph showing Bandwidth Efficiency vs No. of nodes]

**Figure 4.16 No. of Nodes vs Bandwidth efficiency**

The performance graph of the Multimedia multipath transmission in finding the bandwidth efficiency is shown in Figure 4.16. Hence, the Multimedia multipath routing relies on the ability of the system in quantifying link layer bandwidth. It can be seen that the Multimedia multipath transmission achieves much better bandwidth efficiency compared with the Multimedia single path transmission. Multimedia multipath transmission achieves approximately 50% to 65% Bandwidth efficiency, being more than that of Multimedia single path Transmission.
4.6 SUMMARY

Secured Multi-path multimedia data transmission has served as a basis for future routing algorithms due to its fault tolerance, less overall delay and better resource utilization in the wireless context. The major issues challenging this aspect are interrogated in this presentation. They are tradeoff between path disjointing and the packet reordering among different paths, while multiple disjoint paths are used for different packets, tolerating the burst of packet losses in the case of route breakage due to channel interferences. The work presented here is a secure reliable framework for multi-path multimedia streaming over wireless network using Feedback-based RTSP. The proposed framework is designed efficiently for security against not only ad hoc routing but also real-time data forwarding.