Chapter 1

Introduction

1.1 Introduction

Transfer of streaming media over large domain Mobile Ad-hoc Networks (MANETs) is highly sensitive to packet loss, with large round trip delay variations and jitter. As streaming media possess highly asynchronous traffic and being “bursty” in nature, the need for a mechanism to provide optimal Quality of Service (QoS) using adaptive scheduling methods is felt. Issues such as removal of corruptive effects such as noise, echo while varying traffic intensity lead to highly variable packet drop-out or loss, with increased delay for play-out at the receiver end, as well as other latency effects as jitter degrades the quality of media transmitted. Subjective studies [1, 2] have been conducted to access the impact of these effects, by analyzing the existing schemes and to identify viable solutions, which can provide an optimized scheduling approach to resolve this issue. An architectural approach for providing an end-to-end QoS for transfer of streaming media in MANETs using adaptive scheduling is designed and discussed using real time applications and analytical simulator.

In mobile ad-hoc networking scenario, the adoption of the Mobile - Internet Protocol (IP) protocol has provided a common service layer for compatibility between end-user applications and network devices such as wireless routers, Global System for Mobile communication (GSM) base-stations, wireless gateways and gatekeepers.
The major concern raised for providing quality in streaming media is demand for bandwidth and the need for intermediate supportive resources. While Asynchronous Transfer Mode (ATM) [3], Frame Relay and other networks believe in providing viable quality at higher cost, MANETs is in common readily acceptable for multiple heterogeneous mobile networking applications be it in business, research or public utility with a strong device protocol support.

1.2 Objectives

The aim of thesis work is of fivefold:

- To focus on the design and development of an adaptive congestion control mechanism based on service packet priority and load balancing control rate mechanism for MANETs.

- To minimize the average waiting time of packets in each transmission and hence control the latency effect.

- To analyze and create an effective traffic modeling approach for MANETs networks using fuzzy approaches.

- To develop an adaptive traffic scheduler scheme for variable type of packets and service architecture.

- To establish an end-to-end QoS when media packets are transferred over highly congested MANETs, thus providing effective management of wireless communication resources based on the phenomenon of context aware nature of variable service types and buffer management schemes.

The performance of the existing scheduler based QoS protocols has to be analyzed over the multicast routing protocols and variable service environments. The proposed
scheduler is evaluated with quantitative metrics such as jitter, packet delivery ratio and average end-to-end delay. Here each chapter provides an insight of the work done, the methods adopted, experimental setup, simulation results and discussion on performance analysis.

1.2.1 Definitions

- **Fault Tolerance Network** - It can be defined [4] as “the extent to which the network system can continue to work or operate correctly despite the maximum number of faults”. The system should work even under the maximum complex situations or maximum number of faults in the network. The system should provide acceptable results of a user under the worst condition.

- **MANETs** - Mobile Ad-hoc Networking concept enables users willing to communicate with each other to form a temporary network, without any form of centralized administration. Each node participating in the network acts both as host and mobile router and hence must be willing to forward packets for other nodes.

- **Media Stream Networks** - The transfer of media such as audio or video over the IP network (packet networks) is commonly termed as Media Transfer over IP Network. Transfer of any media was originally referred to as “Voice”, hence the term Voice over Internet Protocol(VoIP), but since the thesis discusses media in general (including any form of media types - audio, video), it is referred to as Media Transfer over IP networks. The transfer of real-time media is termed as “Media Streaming”.

- **QoS** - The term Quality of Service is defined [5] “as a concept for specifying how ‘good’ the offered networking services are”. Hereby, it is agreed on the following
understanding of the QoS notion: “QoS indicates the defined and controlling behavior of a service expressed through quantitative measurable parameter(s)”. Guarantee of quality in media streaming networks is gaining rapid acceptance in the current telecommunication scenario. QoS refers to the capability of a network to provide better service to selected network traffic over the technology.

• **Latency Time**- Latency is “the measure of time (in millisecond) taken to transfer media data from call originator to call receiver” or defined as “the difference in time noted between the media sent by call originator and received by call receiver terminals”. It is also defined as “end-to-end delay; it refers to the average time taken for a data to travel successfully through the network from the point of origin to the destination”. The amount of latency will affect the efficiency of applications running across the network. This will have more impact on time sensitive applications involving media transfer over IP network such as video on demand, which requires minimal latency between the end points to meet user’s expectations.

• **Throughput**- Throughput defines “the number of bits that can be transmitted over the network successfully over a period of time” [6]. The throughput of a network increases as load in network increases. But beyond a certain load level, the throughput stops increasing or starts decreasing.

• **Real Time Network Systems**- A real-time system is defined as (The German National Institute for Standards) “a process that delivers the results of processing in a given time-span” [8]. The overall system where each computing and communication service component in the end-to-end path must respond to the real-time network requirements to support the required service quality [7]. A real-time system must satisfy the required timing conditions, and real-time
programs for the deadline-driven processing of data must be available during the entire run-time of the system. The main characteristic of real-time systems is the correctness of the computation. This correctness does not only apply to data less computation, but also it depends on the time in which the result is presented or media data are played out at receiver end [9]. Hence, a real-time system can fail not only if a massive hardware or software failures occur, but also if the system is unable to execute its critical workload at the time.

1.3 Motivation and Problem Definition

Tremendous growth of MANETs and flexibility of Mobile IP-based frequency based networks have accelerated the convergence of interconnectivity between multiple mobile networks for data communications and voice / video based communications into a single “mobile network” based core architecture. MANETs offers “Best Effort” class of service [24] without any guarantee on media play-out at receiver, which induces play out delay, increase in packet loss, leading to reduced throughput and increase in latency effect. These effects have an adverse impact on the end-to-end performance of real-time network applications.

In general theory, media packets transmitted over the network using the Mobile IP protocol, which requires priority over data packets, since in a MANETs with high traffic intensity, media packets suffer from signal dropouts or excessive delays [11]. This problem leads to reduced latency, resulting in periodically interrupted and tone quality being worse than when calling over a conventional telephone line [12]. Mostly media streaming applications include real time constraints, which demand a certain degree of QoS. QoS guarantee in a network can be ensured, if the information about future behavior of interconnection topology is known in advance. But MANETs is
highly dynamic in nature. The topology changes frequently, which is a consequence of node mobility. Thus, the first step, towards QoS support, is finding stable paths and then assigning QoS. So, the ability of MANET to support QoS is directly related to the underlying routing protocol.

1.3.1 Introduction to Media Streaming in MANETs

The current MANETs is the convergence of mobile telephony networks (GSM / GRPS) with data network (EDGE) simply termed as MANETs (see in Fig. 1.1) and broadband network for transfer of streaming real time media, utilizing the existing data network infrastructure as the transport system for both services data and media [13]. To achieve this convergence, technology has been developed to convert the media signal (voice / video), which originates as an analogue signal and transports it within a digital medium. Media streaming networks over MANETs do not provide high tolerance where support for off-time media applications has minimal or negligible “quality” support. MANETs media networks consist of various resources such as Wireless Local Area Network (WLAN), Wi-Fi Personal Digital Assistant (PDA), Mobile Telephone, Wi-Fi Video-Conference, Wi-Fi Video On-demand devices which are interconnected over network components such as Wireless Gatekeepers, Mobile Gateway, and cellular GSM networks. All communication devices are interconnected together works on the MANET as backbone architecture.

1.3.2 Issues in Media Streaming Network

QoS is the basic and central focus of any multimedia real-time applications for transfer over MANET [14]. Media streaming networks require higher bandwidth and effective QoS measures for real time media transmission [15]. The media data transmitted through these networks require high reliable parameters, which require qualifying
techniques such as Data Checking, Higher Bandwidth on Demand, Packet Acknowledgment, Re-transmission of lost media packets and sequencing of media packets. The demand for higher bandwidth is the most vital parameter for solving most of the issues in media network. It has been found that video transferred over the IP network requires more than 1000Mbps of bandwidth [15]. In case of real time applications such as voice or video, re-transmission of lost media packets for play back at receiving end or handling out of sequenced media packets lead to delay. The total system together is termed as “Fault Tolerance” which is of great practical significance, more dominant and a mandatory behavioral feature for any real time network. Hence, the need to reexamine the QoS structure of Mobile IP-network to support a heterogeneous media, quality related media transfer over MANET is required.

The current MANET is not well suited to support real-time media applications [16]. Real-time audio communication is bandwidth intensive, and has strict require-
ments for minimum throughput, network delay and jitter. MANET operates on a
best-effort basis using Mobile IP or Ad-hoc On-demand Distance Vector (AODV) pro-
tocol, which does not guarantee an upper bound on end-to-end delay or lower bound
on available bandwidth. Disaster recovery, military operations, home networking, and
conferencing are some potential examples. These applications can easily cause net-
work congestion and hence packet loss and increased delays can significantly impair
media quality. As such, applications become more widespread, and large number of
audio streams consume a considerable portion of the network load. Therefore, the
overall behaviour of the applications that include a large number of streams will have
significant impact on the network traffic and the quality of delivered media.

1.4 QoS in Media Streaming Network

QoS refers to a network’s ability, which reliably and consistently delivers the required
level of throughput and performance for terminal user. Reliable message transfer with
data control and notification of non-delivery is common in many modern communica-
tion systems. However, the recently much importance has been given to the ability to
specify timeliness, and the perceived quality of the data arriving, particularly where
more complex (multi) media are being used. The underlying concepts of bandwidth,
throughput, timeliness (including jitter), reliability, perceived quality and cost are
the foundations of what is known as QoS. QoS and resource management are basic
concepts and central focus on multimedia systems.

The primary goal of QoS is to provide priority, including dedicated bandwidth,
controlled jitter and latency (required by some real-time and interactive traffic), and
improved loss characteristics. Guarantee of QoS [15] for network is a complex method-
ology, but general internet has not been designed to offer guaranteed delivery or to
prioritize bandwidth during each transmission. QoS for conferencing typically involves network availability, bandwidth, end-to-end delay, jitter, and packet loss. The term QoS primarily refers to providing an acceptable level of 'quality' for end-users. The desired level of quality depends upon the end-user who is at the receiving end to receive the delivered media. Multiple variable resource parameters exist between the call originator terminals and call receiver terminals, which have to be controlled to provide the optimal quality level to the end-user. In general, an end-to-end QoS can be defined from application level to network level of Open Systems Interconnection (OSI) layer architecture as follows:

- Application level QoS characterizes the end user expectations such as clear voice, jitter free video, echo free play-out of the media.
- Network level QoS refers to tangible measurements such as controlled delay, latency effects, and required bandwidth and packet loss rate
- Traffic level QoS characterizes the method of packet transfer and related transport mechanisms involved in the transfer. Session layer deals with managing session for media data transfer between any multiple end points for multiple devices.

The bridging interface or relationship between the application layer and network level QoS discussed in earlier research works [16,17], does not provide any useful results. The goal of achieving the end-to-end QoS by statistical means is to achieve packet loss rate at 99% or at most less than 70ms end-to-end delay but not reserving dedicated bandwidth per session.

E-Model’s Mean Opinion Score (MOS) [18] based QoS routing setup in Fig. 1.2 shows the bandwidth and traffic intensity utilization factor of a media streaming network setup. It is found that the utilization factor is found to be higher for varying
Fig. 1.2: Graph predicting QoS values against bandwidth utilization percentage rate of media and data transferred through the network. Hence, it is understood that acceptable quality is obtained if bandwidth on demand is available, with controlled traffic intensity. The media-streaming network has to be re-engineered with variable parameters, which determine the quality of media. The inter-organization of variable parameters such as Traffic Intensity, Buffer Availability, Demand on Bandwidth provides optimized quality of media stream on transmit over a highly congested network.

For providing an end-to-end QoS setup, various parameters are put into activity initializing from User Level - QoS parameters, Device Level - QoS parameters, Network Level - QoS parameters, Application Level - QoS parameters, which work together in establishing an inter-organization among each entity. Mechanisms like negotiation, re-
source mapping, and management are performed. If maximum numbers of parameters are called into access, an effective and best QoS is achieved. If the variable parameters in activity are less, QoS value remains reduced. Hence, it is obvious that the need for high throughput demands high-speed network services and high performance end systems.

1.5 Issues in Providing QoS

Latency and throughput are the primary parameters determining the quality of media [19], which predicts the growth of the streaming of media technology. Latency can seriously affect the quality of the media transferred over the Internet as delays cause breaks and pauses during media transmission. Issues such as jitter, media blocking, inter-arrival time of media data, and play-out delay of media data at the receiver end attribute to the latency effect of the quality of media data. The throughput of the network depends on traffic intensity at routers or gateways, loss of media packets and play-out delay.

Scheduler plays a major role in providing QoS at Data Link Layer, Network Layer
and Transport Layer. In Fig. 1.3 shows the mechanism of scheduling adopted in MANET which is followed in this research work. Any MANET node which transmits packet route traffic queuing based on resource requests. The network traffic scheduler identifies the traffic and schedules the flow accordingly based on a traffic marker algorithm which marks based on service type. The gateway manager authorizes the flow and shapes the traffic base on packet weighing approach and priority for inflow of packets. The call authorizer routes the packets based on policy control mechanism.

1.5.1 Packet Loss

The phenomenon of dropping packets or losing packets due to collision or a corrupted packet is termed as Packet Loss in network glossary [20]. The Mobile IP packets from multiple sources are queued for transmission to an outgoing link in a wireless router, where packets are transmitted over the queue one at a time. An arriving packet is lost or dropped, if there is no space in the router queue, the packet has to be discarded. When the routers become too much congested, then the possibility for a packet collision takes place, resulting in packet loss. Loss of packets can cause severe damage to the quality of voice that is being transmitted.

1.5.2 Jitter

The delay introduced by the call receiver to hold one or more media packets, creating variations in arrival times is termed as Jitter [9]. The event of Jitter is due to a unavailable buffer at the call-receiver that causes a flutter in receiving the media packets to be played out and jitter can be avoided using the inter-packet timing sequencing parameter.
1.5.3 Delay

The quality of media for play-out at receiver is based on the delay of playing out the media stream and the time for receiving the media packets [21]. Call receivers in the media streaming network, identify negligible delay of around 180ms. Most receivers identify delay when it is more than 150ms. If it exceeds 200ms, the voice quality is defined as poor. End-to-End delay is the time media data takes to get data across the network between call originators and call receivers. Even though there are packet loss and traffic at gateways, congestion can be considered as external attributes that contribute an end-to-end delay. Other internal delay parameters such as Accumulation Delay, Propagation Delay, Processing Delay, Transport Delay, Play-out Delay, Network Delay, and Access Delay are defined.

1.5.4 Congestion

Congestion is the state of sustained network overload where the demand for network resources is close to or exceeds the required capacity. Similarly congestion on the internet would lead to high packet loss rates, increased delays and collapse of complete system [18]. This state of an increased in offering load leads to decrease in throughput. Another form of congestion collapse is undelivered packets where the available bandwidth is wasted by dropping packets before the packets reach their destination. Congestion defines the maximum number of packets waiting at the gateway or router to get serviced. The parameter invariably depends upon the bandwidth and buffer capacity of the gateway. The buffer to hold packets in the gateway is proportional to the incoming flow of packets. Congestion takes place when the inter-arrival flow of packets is higher to the servicing capacity of the gateway and the capacity to hold. Any increase in traffic intensity leads to congestion, and increase in congestion leads
1.6 Re-engineering of Media Streaming Network

Guarantee of QoS insists on multiple variable run time media parameters. The media-streaming network has to be re-designed to provide a reliable, fault-tolerant network oriented for delivery of required media quality. The focus has to be provided for minimizing media packet loss, handling media traffic, avoiding congestion at routers / gateways such that jitter is minimized. Effective control would lead to effective throughput and minimized latency effect. Analysis of video on demand application [18] executed over the network shows that the QoS parameters of media stream are variable in nature. It has been found that a guarantee of QoS relies on traffic in terms of packet drop priority, queuing delay, average delay variation and peak data flow rate. Hence, resource allocation has to be tailored to meet user’s acceptable quality. Effective management of resources over the session between call originator and call receiver delivers quality.

Re-engineering of the network to provide quality can be done by two major approaches to provide Quality Service, which can be applied either separately or simultaneously. The most flexible and simple adjustment of multimedia streams onto a given computing and communication environment can be achieved by Scaling and Adaptation of media quality. Individual service components and their corresponding resources are identified which are needed for multimedia processing and communication, and reserves the required resources before the processing and communication of multimedia stream starts. This concept, called Resource Reservation, includes all resources along the end-to-end multimedia stream path. This concept can also be applied to parts of an application, which processes continuous media.
Table 1.1: Current QoS values and optimal QoS values.

<table>
<thead>
<tr>
<th>QoS parameters</th>
<th>Current setup(average)</th>
<th>Optimal expected values</th>
</tr>
</thead>
<tbody>
<tr>
<td>Latency</td>
<td>≤ 230ms</td>
<td>≤ 140ms</td>
</tr>
<tr>
<td>Jitter Delay</td>
<td>≤ 75ms</td>
<td>≤ 40ms</td>
</tr>
<tr>
<td>Packet Loss</td>
<td>≤ 5 %</td>
<td>≤ 1 %</td>
</tr>
</tbody>
</table>

This thesis aims to design, build and validate media service that efficiently support applications that allow simultaneous media streaming as a natural pattern of conference activity among participants, and can adapt to, network conditions as well as smooth out the effects of the network on the end user quality. Such a service is essential for efficient management of the Internet and the quality of delivered media.

Re-engineering insists on the following:

- Increase throughput of media applications for transfer over IP networks
- Provide efficient, quality for multiple, interoperable implementations of heterogeneous networks
- Should scale well over different service schemes
- Optimal quality of play-out on receiving end (see in Table 1.1)
- Possible reduction in Latency effect
- Work with end host operating systems and middleware

An optimum network settings should provide the following effective QoS value [22] based on adaptive scheduling approach.
1.7 Objectives and Scope

This thesis primarily focuses on various scheduling issues to provide an effective QoS for MANETs. The research work takes the nodes of traffic load into consideration during the process of adaptive scheduling. By means of simulation studies, the performance of this algorithm is compared with that of the Weighted Fair Queue (WFQ) algorithm. Simulation results show that this algorithm performs better. The scheduling scheme of this chapter mainly deals with best-effort traffic, but the QoS provision is becoming more and more important to the deployment of MANETs. The future work of this research needs to focus on:

- Mechanisms to handle session based scheduling for MANETs, since the establishment of a session and using a session for a complete transmission in MANET is a complex task.

- The need for an associative rule based fuzzy approach can be surveyed and analyzed which can suggest an effective method for agent based scheduling approach.

It support, for minimizing media packet loss during multicast process. This scheme uses active queue flow management and monitoring between multiple flow paths to minimize packet loss with reduced delay. Improved scheduling schemes such as I-LABS works on the basis of adaptive buffer mechanism which provides effective buffer management and variable queue support for media packets in a multicast process. This scheme works in providing adaptive session based buffer allocation between multiple endpoints of MANET node involved in a session. COAAS works on identification of context aware variables involved in group communication with rule based session management to identify optimal QoS between multiple heterogeneous networks based
on policy management structure.

COAAS works as an interface process in linking between I-LABS and COAAF, which together achieves optimal QoS for optimal time delay for play-out of media packets at the receiver end. In best-case situation this scheme provides no packet loss while in the worst-case situations the percentage of packet loss and round trip delay achieves the optimal level, which is minimal when compared with the existing schemes. This scheme is heuristic in its approach, which adopts adaptive scheduling strategy traffic scheduling approach. This scheme achieves an optimal level for a round-trip delay time and media play out at the receiver end, thus effectively minimizing media packet loss, jitter by avoiding congestion at routers and providing controlled session management for handling the media call.

COAAS possess is a context aware queue controlled architecture to regulate distributed end-to-end resource provisioning over scalable MANETs. This method meets heterogeneous demand of media streaming networks. Such control architecture is an essential component for delivering QoS and coordination of network management policies in the future Internet.

1.8 Organization of thesis work

The main aim of our thesis is to investigates the channel based routing in MANETs. This thesis consists of seven chapters and they are organized as follows:

In Chapter 2 focuses on a complete study of the existing methods and schemes such as AODV, TORA, ADiffServ and DiffServ. The nature of network traffic intensity for different media applications under various network conditions is studied. Similarly, various test cases carried out are also analyzed. A new scheme LBCA-AOMDV for minimizing media packet loss is proposed in Chapter 3. This scheme is also tested for
media packet loss and latency time at different time intervals using NS-2 simulator and real time experimental approach. I-LABS routing is proposed in Chapter 4, which aims at achieving optimal QoS with the context aware nature of the queue. This rule based analytical model focuses on achieving optimal QoS between endpoints. This scheme is tested with test-bed analysis and results are discussed.

Chapter 5 deals with proposed a COAAF scheme to provide session based variable queue management for the entire transmission session using priority assignment for service and management algorithm. Chapter 6 deals with the proposed COAAS to provide an adaptive network management for QoS supportive network-aware applications. Chapter 7 provides the summary of various schemes discussed in this thesis and major contribution towards research. Suggestions for future research work are also discussed. The execution Codes are given in Appendix A. A list of references and list of papers published based on this research work are attached at the end of this thesis report.