# EVALUATING PERFORMANCE OF AUDIO CONFERENCING ON REACTIVE ROUTING PROTOCOLS FOR MANET

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**Abstract**

Mobile ad hoc network (MANET) represents a system of wireless nodes that can freely and dynamically self-organize into arbitrary and temporary network topologies, allowing people and devices to seamlessly internet-work in areas without any preexisting communication infrastructure. The routing protocols of this network elapsed much time in route discovery and route maintenances. In this research work we have studied the performance of routing protocols DSR and AODV when the nodes involved in audio conferencing. Simulation studies show that there are not any significant differences between these protocols. However DSR is better than AODV in coverage area with acceptable delay and packet loss.

**Keywords:** Ad hoc Network, Routing Protocols, H.323, Audio conferencing, Packet Loss

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## 1. Introduction

In a mobile ad hoc network (MANET), mobile nodes communicate using wireless links without a fixed infrastructure such as base stations (access points) or centralized control. A typical mobile ad hoc network is a group of hosts or nodes operating in wireless ad hoc mode. Each mobile node acts as a router to enable multihop communication. A node is free to move around randomly and as a result, the topology formed by the nodes is highly dynamic and unpredictable. A MANET can operate in a stand-alone fashion, or can be connected to a fixed internetwork, for example, the Internet. Ad hoc networks are very useful in emergency search-and-rescue operations, meetings or conventions in which persons wish to quickly share information, and data acquisition operations in inhospitable terrain [1].

In this network when a node sends data to a destination node who is not an immediate neighbor, it must first find a route to that destination. Intermediate nodes co-operate to forward packets from the source to the destination. Due to the absence of dedicated network infrastructure, participating devices should allocate their resources for routing. The prime objective of routing protocols of ad hoc network is to find an efficient route from sending node to receiving node. As the nodes have to administer themselves to manage the network which will decrease the network ability in maintaining its quality of service [2]. Besides, ad hoc networks must cope with other wireless problems, such as low transmission rate, high Bit Error Rate (BER), and significant variations in physical medium conditions. This complexity makes transmission of real-time traffic on ad hoc networks a great challenge due to Quality of Service (QoS) requirements [3].

Many different protocols have been proposed to solve the multihop routing problem in ad hoc networks, each based on different assumptions and intuitions. When real-time audio or video packets transmit in ad hoc network, the performance of these protocols is needed to study. It is required to investigate how long the protocols satisfy the QoS requirements. In this paper we have studied the routing protocols performance of ad hoc network when packets carry audio data. Audio data is generated when nodes participate in audio conferencing. For example when students use laptop computers to participate in an interactive lecture, business associates share information during a meeting, emergency disaster relief personnel coordinating efforts after a hurricane or earthquake. In this work we assume that all ad hoc nodes have hardwire, codecs and software to generate and packets the audio information. The contribution of this paper is to find the audio conferencing area of ad hoc network for routing protocols of DSR and AODV.

This paper is structured as follows: Section 2 gives a brief overview about the routing protocols. We discuss the conferencing standards in section 3. Section 4 presents the simulation methodology and discusses the simulation results. Finally, we conclude this paper in section 5.

## 2. Routing in Ad Hoc Network

Routing protocols in ad hoc networks create and maintain routes between pairs of communicating nodes. Routing must deal with unpredictable node mobility patterns, radio transmission errors, the entrance and exit of nodes. The main types of routing protocol used in ad hoc networks are table driven, on demand (source and hop by hop) and hybrid routing [1].

### 2.1 DSR

Dynamic Source Routing (DSR) [4] is an on-demand source routed protocol designed for mobile multi-hop wireless ad hoc networks. DSR provides two main functions – route discovery and route maintenance. Route Discovery is performed when a node wishes to send a packet to a destination for which it does not have a route. The route maintenance function identifies link failure on an active route. The source, whose packet precipitated the discovery of the link failure is informed of the link failure and updates its routing cache appropriately. Both route discovery and route maintenance operate on an on-demand basis. No data is exchanged between nodes in a periodic manner. DSR is not capable of monitoring or handling congestion. Because of this, should congestion occur, packet losses are dealt with by a higher layer, such as TCP.

### 2.2 AODV

The AODV [5] routing protocol uses the periodic beaconing and sequence numbering procedure of DSDV [6] and a similar route discovery procedure as in DSR. However, there are two major differences between DSR and AODV. The most distinguishing difference is that in DSR each packet carries full routing information, whereas in AODV the packets carry the destination address. This means that AODV has potentially less routing overheads than DSR. The other difference is that the route replies in DSR carry the address of every node along the route, whereas in AODV the route replies only carry the destination IP address and the sequence number. The advantage of AODV is that it is adaptable to highly dynamic networks. However, node may experience large delays during route construction, and link failure may initiate another route discovery, which introduces extra delays and consumes more bandwidth as the size of the network increases.

## 3 Conferencing

Conferencing systems in computer technology allow individuals at two or more different geographical locations to see and hear each other, exchange data, and work together using interactive video, audio and computer technologies. The core technology used in a conferencing system is digital compression of audio and video streams by hardware or software called codec (coder/decoder) [7]. International Telecommunications Union (ITU) has produced a number of international standards for real time digital multimedia communication, including audio, video and data conferencing such as H.320 for ISDN, H.321 for ATM/B-ISDN, H.323 for packet networks, H.324 for conferencing over the general telephone network etc.

In this work we consider H.323 standards [8] because this standard creates specifications for packet based networks. It was designed with multipoint voice and video conferencing capabilities. H.323 standard which has Audio codec standard: G.711, G.722, G.723.1, G.728, G.729 [8]. In this work we consider the audio codec G.723.1. G.723.1 transmits audio data with 5.3 and 6.3 Kbps. We choose such low data rate because wireless signals faces high attenuation, bit error, fading etc.

## 4. Performance evaluation

In this section we describe the performance matrices and simulation in details. We also discuss our simulation results.

### 4.1 Performance Metrics

To measure performance of audio data packets in ad hoc network we follow End to End delay that includes all possible delays caused by buffering during route discovery latency, queuing at the interface queue, retransmission delays at the MAC, and propagation and transfer time. Calyam [15] has suggested that for audio conferencing audio delay should not over 10ms ($d_{acc}$) [15] for good performance. For wireless medium packet losses have great impact on real time data. Calyam also suggested that the packet loss should be 0%-0.5% for good quality of conferencing. 0.5%-1.5% packet loss is acceptable but if it more than 1.5% the performance become poor. In this work, the maximum acceptable packet loss is 1% ($l_{acc}$).

### 4.2 Simulation Parameters

In this research work, we assume that all ad hoc nodes have required hardware, codecs and software. The system supports multi-party conferencing. This work considers the requirements of codec G.723.1 (6.3kb). We use network simulator NS-2 [9]. The link layer of NS-2 simulator implements the IEEE 802.11 standard Medium Access Control (MAC) protocol Distributed Coordination Function (DCF) in order to accurately model the contention of nodes for the wireless medium. The radio model used for simulation is based on the Two-Ray Ground Propagation Model and the standard 802.11 b.

For simulation we consider the followings:

*Nodes* – There are 20 nodes in the network with two scenarios – 10 connections and 5 connections

*Simulation time* – 100 seconds for each simulation

*Mobility* – Average movement of nodes is 10 meter/second.

*Simulated Area* – The field configurations are as follows: begin with (300m×300m) square field. Then increase *x* axis and *y* axis by 100 meter. Dimension of ad hoc field is increased to maximum (800m×800m) if packet loss and end to end delay below acceptable value.

*Pause time* – For pause time seconds each node of the simulation remains stationary. Here we take pause time 0 second i.e. nodes are in continuous motion.

*Data packet size* – Packet size 512 byte

*Data Packet* – Data packet format of 802.11 MAC is given below [13]. The MAC overhead is almost constant per packet. For 512 byte packet, each packet contains 7% overhead.

| Frame Control | Duration ID | Address1 (source) | Address2 (destination) | Address3 (rx node) | Sequence Control | Address4 (tx node) | Data | FCS |
|---|---|---|---|---|---|---|---|---|
| 2 | 2 | 6 | 6 | 6 | 2 | 6 | 0 - 2,312 | 4 |

Fig. 1: 802.11 Frame Format

*Data transfer rate* – With 7% packet overhead data transfer rate is 6.74kb for G.723.1 codec.

*No of packets* – To transmit G.723.1 codec data we need 2 packets.

Table 1 represents summary of parameters used in simulation.

**Table 1: Simulation parameters**

| | |
|---|---|
| Routing Protocols | DSR/ AODV |
| No. of Nodes | 20 |
| Area | $300m^2$ to $800m^2$ |
| Traffic type | Audio and Video |
| Codecs | G.723.1 |
| Mobilty | 10 m/sec |
| Nodes Position | Random |
| Simulation Time | 100 sec |

| | |
|---|---|
| MAC/ Phy | 802.11 b |
| Packet Size | 512 Bytes |
| Packet Loss | Maximum 1% |
| End to End delay | Maximum 10 ms |

### 4.3 Simulation results

First we setup our simulation for audio codec G.723.1. To evaluate the performance we first set simulation area (300m × 300m) and measure the packet loss for two ad hoc routing protocols AODV and DSR. The constraint in this case is time delay. Each time area is increased by 100m in *x*-direction and 100m in *y*-direction. We do not continue our simulation when either time-delay more than acceptable audio delay ($d_{acc}$)or packet loss more than ($l_{acc}$).

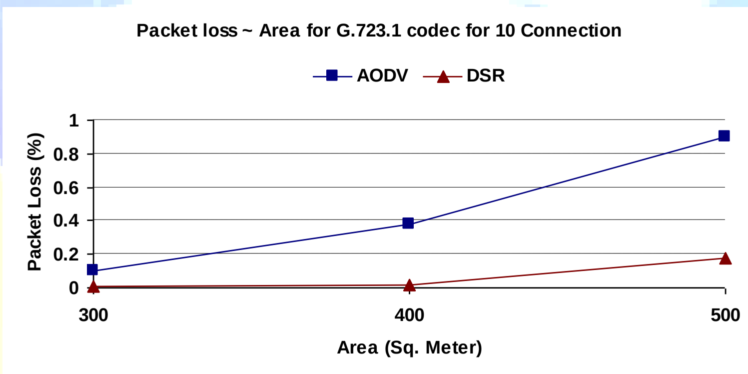


Fig. 2: Packet loss for 20 nodes with maximum 10 connections for G.723.1

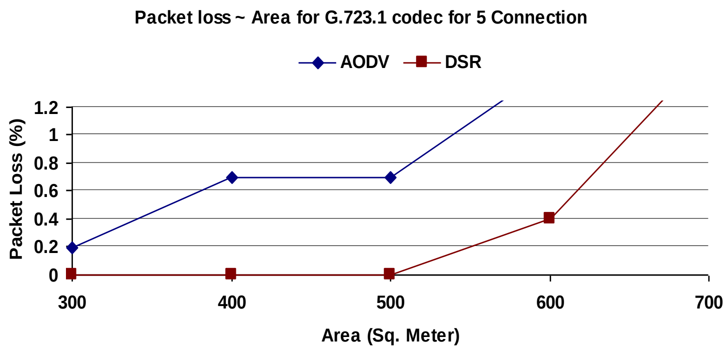


Fig. 3: Packet loss for 20 nodes with maximum 5 connections for G.723.1

Fig. 2 represents the maximum area for G.723.1 when there are maximum 10 connections between the nodes. For 20 nodes with maximum connection 10, both protocols AODV and DSR wrap within 500 meter$^2$. Within this area DSR has low packet loss than AODV. After this size of filed the packet losses are more than 1% which indicates poor quality audio conversation between nodes. Decreasing the maximum connection from 10 to 5 with 20 nodes AODV can cover the area nearly 600 meter$^2$ while DSR can cover approximately 650 meter$^2$ (fig. 3) within acceptable time delay. To measure the end to end delay we apply the constraint maximum 1% packet loss. When all nodes participate in conferencing, maximum area is 500 meter$^2$ for both AODV and DSR (fig. 5). When half of them participate in conferencing, maximum coverage area is 600 meter$^2$ under AODV protocol and 700 meter$^2$ for DSR protocol. It is interesting that the value of end to end delay is much lower than acceptable delay. But packet loss limits the coverage area.

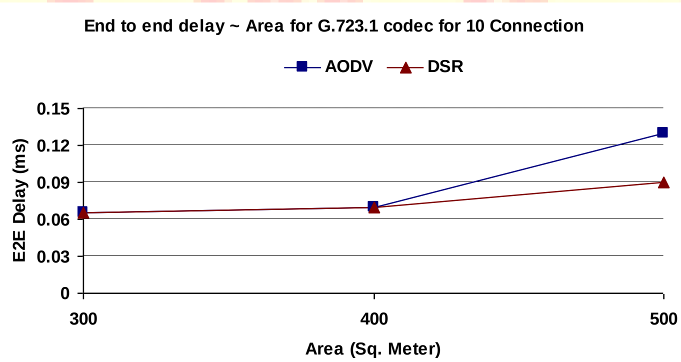


Fig. 4: End to End delay for 20 nodes with maximum 10 connections for G.723.1

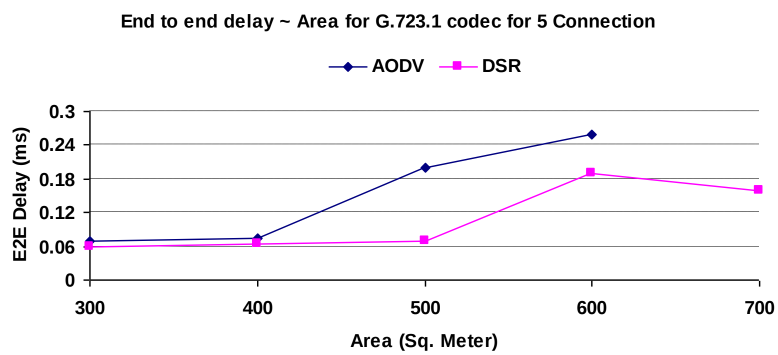


Fig. 5: End to End delay for 20 nodes with maximum 5 connections for G.723.1

## 5. Conclusion

Ad-Hoc network is an emerging field in networking area. Transmission of voice over such network makes it more applicable in real world. In this paper we investigate how voice transmission is influenced by wireless multi-hop network. We have performed a study of routing protocols DSR and AODV when they carry real time audio data. Main aim of this study is to find the coverage area. In large area nodes frequently experience route breaking and elapse more time to find an efficient route. So packet loss and end to end delay are increased. These are not major problem for data transmission but it has serious influence on real time packets. For full connections between nodes, there are significant packet losses that not tolerable. Simulation studies demonstrate area coverage by both protocols is almost same but DSR has slightly better performance than AODV. Coverage area of DSR is more than AODV with low end to end delay.

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