

An Extended AODV Protocol for VoIP Application in Mobile Ad Hoc Network

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ABSTRACT

In this paper, the problem of voice calls over a mobile ad hoc network that employs IEEE 802.11b standard for the lowest communication layer is considered. The Distributed Coordination Function (DCF) is assumed as the basic medium access protocol. Two DCF access mechanisms for VoIP conversation are considered. A new metric for searching a stable routing based on an extended Ad-hoc On-Demand Distance Vector (AODV) protocol is proposed and investigated. The simulation results show that the new routing protocol can improve performance of VoIP over ad hoc network as compared to the standard AODV protocol.

Keywords: MANET, VoIP, DCF, EAODV, PDR, the E2E delay

1. INTRODUCTION

A mobile ad hoc network (MANET) consists of a collection of mobile wireless nodes, which dynamically form a temporary self-organizing mobile wireless multihop network. The advantage of this type of network is an easy and cost efficient way of its deployment and expansion.

In recent years, there has been a lot of interest in Voice over Internet Protocol (VoIP) operating on a mobile ad hoc network platform. The reason is that VoIP over MANETs is viewed as an important technology for voice communications without an infrastructure support. Potential applications of VoIP over MANETs concern scenarios such as groups of people communicating on a university campus or at a conference using laptop computers without involvement of any additional infrastructure between them. Other examples concern dynamic communication in emergency search, disaster rescue operations, and a battlefield.

Providing real-time VoIP services on MANET is a difficult task due to restrictions in device resources,

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adverse properties of the wireless channel, dynamic topology and the lack of central administration. Because of these limitations, there is a challenge to meet the Quality of Service (QoS) of VoIP in terms of packet delivery ratio (PDR) and end-to-end (E2E) delay that has to be within an acceptable range [1]. The limitations of VoIP using IEEE 802.11 Access Point (AP) network with respect to the number of users have been addressed in [2-4]. However, little has been reported with respect to the MANET environment. Due to a large overhead caused by transmitting small packets, VoIP applications make a very inefficient use of the wireless resources. In order to make their better utilization, an optimal routing in a MANET is required.

This paper reports on the use of VoIP on MANET and addresses the problems associated with achieving the real-time voice transmission using data packets over a digital network when there is lack or limited infrastructure.

In the presented investigations, it is assumed that the wireless interfaces rely on ad hoc configurations of IEEE 802.11b WLAN standard which has been widely used in the last few years. Two different Distributed Coordination Function (DCF) access mechanisms are considered in this ad hoc network. Because current ad hoc routing protocols are based on the shortest path strategy and neglect the issue of better channel utilization, a new routing protocol based on an Ad-hoc On-Demand Distance Vector (AODV) routing protocol, which uses a routing metric to discover a more stable routing, is proposed and investigated. This new routing protocol allows for a more efficient use of idle channels. It reduces the influence of channel interference and congestion and thus is capable to improve the QoS of voice conversation.

2. BACKGROUND

2.1 The Characteristics of VoIP

A VoIP session is considered as a real-time bi-directional, point-to-point connection between two users. Usually it takes place over a standard network infrastructure, and demands a very well-configured network to run smoothly. The following are its characteristics.

1. VoIP employs RTP protocol over UDP: Typical Internet applications use TCP/IP, whereas VoIP uses

RTP/UDP/IP [5]. While creating a voice packet, voice payload is time-stamped using RTP, which introduces a 12-byte header. The resulting segment is carried by a UDP datagram which further adds an 8-byte header. Finally, when encapsulated into an IP datagram, a 20-byte IP header is appended.

2. VoIP packets are small and packets intervals are short: Voice communication is sensitive to time delay. Hence, interval times for the sending of voice packets are shorter than the ones used for data packets, usually 10-30 ms, and the packets have a very small payload of between 20-160 bytes based on different codecs that are used. Table 1 lists the attributes of several commonly used codecs [5].

Table 1: Attributes of Commonly Used Codecs

Codec	GSM 6:10	G.711	G.723.1	G.726.32	G.729
Bit rate(Kbps)	13.2	64	5.3/6.3	32	8
Framing interval(ms)	20	20	30	20	10
Payload(Bytes)	33	160	20/24	80	10
Packet/sec	50	50	33	50	50*

For all codecs except G.729, Packets/sec=1/Framing interval). For G.729, two frames are combined into one packet so that Packets/sec=1/(2 Framing interval).

3) The efficiency of transmission is low: Due to the high ratio between the header and payload size, the transmission of VoIP packets is not an efficient process. For transmitting 20-160 bytes a header of 40 bytes is needed.

2.2 QoS for Real-time Voice Conversations in IP Network

Voice quality is mainly governed by the choice of codec. However, the network performance also has a substantial impact. Two major factors associated with packet networks that have a significant influence on perceived speech quality are packet loss and delay [6, 7].

1) Packet loss: VoIP is a real-time audio service that uses UDP. Because of an unreliable protocol, the recovery of lost packets is not possible, so the codecs must be able to handle some packet loss, e.g. by interpolation. A loss of 5% or more is usually noticeable. For network performance metric, packet loss can be estimated by the packet delivery ratio (PDR). PDR is the ratio of the total number of data packets successfully delivered to the destinations to the total number of data packets generated by the sources. It is expressed in percentage.

2) Delay: VoIP delay or latency is characterized by an additional amount of time taken for a complete voice data to be transmitted, that is, the instant it is generated to the instant it is received. A delay of less than 200 ms is desirable for VoIP to operate in full-duplex mode. Anything longer requires switching it to a half-duplex conversation. In VoIP, delay

consists of three elements: accumulation delay, processing delay and network delay. The combined accumulation and processing delay generally is less than 50 ms, thus the major component of the total voice delay is attributed to the network delay. The network delay can be calculated by averaging the End-to-End (E2E) delay which is the time taken for a packet to be successfully delivered from the source to the application layer of the destination. Therefore, to achieve good transmission quality a network delay of no more than 150 ms is required.

2.3 VoIP Traffic over 802.11 Wireless Links

1) DCF in IEEE 802.11 Standards: IEEE 802.11 is a standard for the Physical (PHY) layer and Medium Access Control (MAC) sublayer [8]. IEEE 802.11 was initially designed for base station managed networks, which form a bridge between wireless and wired LANs. The DCF is the default 802.11 contention resolution mechanism for the MAC layer, which is based on the Carrier Sense Multiple Access/Collision Avoidance (CSMA/CA) mechanism. With DCF, ad hoc networks accompanied by suitable routing protocols can be built on top of IEEE 802.11. Most of the protocols designed for ad hoc networks assume that IEEE 802.11b is used for lowest-layer communications. There are two access mechanisms used for packet transmitting in DCF. One is a two-way handshaking basic mechanism, which uses Acknowledgement (ACK) to confirm the successful packet transmission. The other is a four-way handshaking mechanism, which uses the RTS/CTS technique to reserve the channel before data transmission takes place. This is used to solve the hidden terminal problem [9]. The basic operations of 802.11 DCF and DCF with the RTS/CTS [10-12] are shown in Fig.1 and 2.

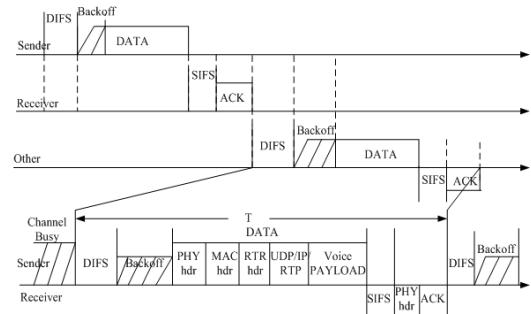


Fig.1: Basic DCF access mechanism

Using the basic DCF, when nodes are transmitting data, channel availability is checked first. If the channel is idle, the node may transmit data packets. However if the channel is busy, each node waits until transmission stops, and enters into a random back off procedure. This prevents multiple nodes from seizing the medium immediately after completion of the preceding transmission. On receiving the packet, packet

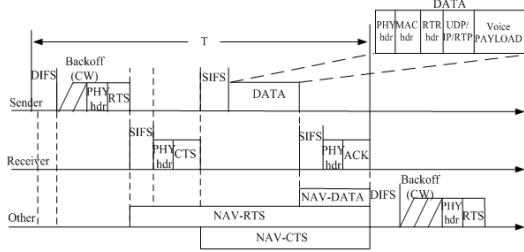


Fig.2: DCF with RTS/CTS access mechanism

*Network Allocation Vector (NAV), Short Interframe space (SIFS), DCF Interframe space (DIFS), Contention window (CW), Backoff = $\text{rand}[0, \text{CW}] \times \text{ShotTime}$

reception is acknowledged, however, if the acknowledgement is not received, then a failed transmission is assumed. In this case, the source node retransmits but when the limit of the retry counter is reached, further transmission attempts for the specific frame are abandoned.

DCF with the RTS/CTS can efficiently deal with the hidden terminal problem. A RTS frame is firstly sent by the source node before performing the transmission and a CTS is sent back by the receiver to notify its availability to receive the data. Once CTS confirmation is received, a data packet can begin transmission and on completion, an ACK will be received.

2) *VoIP Over IEEE 802.11b network*: Bandwidth utilization is limited in 802.11b network. The large overhead imposed by the DCF mechanism results in transmitting small data packets. The problem with small payloads is that most of the time is spent on sending headers and the RTS/CTS acknowledgments, waiting for separation DIFS and SIFS, and contentions for the medium. As a result, the achievable throughput for the 802.11b is far less than its maximum 11Mbps data rate. The values of the parameters for IEEE 802.11b DCF are listed in Table 2.

Equation (1) and (2) show the time spent on successfully transmitting one packet using the two different DCF access mechanisms. The actual voice data transmission time is shorter as compared to the whole packet transmission progress (nearly 0.77-11.1% in the basic method and 0.45-6.74% in the RTS/CTS method). The DCF with RTS/CTS has more overheads than the basic DCF mechanism. Hence, VoIP using wireless resources is inefficient.

There are limitations on the maximum number of VoIP calls that can be supported in a single cell of 802.11b network [2, 3, 13]. Different codecs support different number of VoIP calls. The capacity of voice calls is influenced by the packetization interval. As the packetization interval increases the capacity of voice calls increases [4]. If RTS/CTS handshake is used the capacity worsens. For a mobile ad hoc

Table 2: Parameter Values of 802.11b DCF

Characteristic	IEEE 802.11b		
SlotTime	20 us		
SIFS	10 us		
DIFS	50 us		
CWmin	32		
CWmax	1024		
RTS	160 bits+ PHY header	20 bytes+24 bytes	35.2us
CTS	112 bits+ PHY header	14 bytes+24 bytes	30.4us
Routing layer header	20	20	14.5us
PHY header	192 bits	24 bytes	192 us
MAC header	224 bits	28 bytes	20.4us
ACK	112 bits + PHY header	14 bytes +24 bytes	30.4us
RTP layer overhead	96 bits	12 bytes	29.1us
UDP layer overhead	64 bits	8 bytes	
IP layer overhead	160 bits	20 bytes	
Voice Data	80-1280bit	10-160 bytes	7.3- 116.4us
CTS_Timeout	300us		
ACK_Timeout	300us		
Average T_Backoff	310us		

*IEEE 802.11b initial 24 byte header (PHY), ACK, RTS & CTS is always transmitted at 1 mbps. Rest of the MAC protocol Data unit is transmitted at either 2, 5 or 11Mbps.

network, the voice capacity of the whole network is smaller compared to an AP WLAN. When nodes forward each others' packets, they share the same wireless channel, thus causing interference and congestion with neighboring nodes (Note that most ad hoc networks assume the same channel in the whole network). In addition, it is influenced by dynamic topology [14].

$$\begin{aligned}
 T &= DIFS + T_{Backoff} + T_{DATA} + \delta + SIFS + T_{ACK} + \delta \\
 &= 50 + (310 + 192 + 20.4 + 14.5 + 29.1 + T_{Voice}) \\
 &\quad + \delta + 10 + 304 + \delta \\
 &= 930\text{us} + 2\delta + T_{Voice}
 \end{aligned} \quad (1)$$

$$\begin{aligned}
 T &= DIFS + T_{Backoff} + T_{RST} + \delta + SIFS + T_{CTS} \\
 &\quad + \delta + SIFS + T_{DATA} + \delta + SIFS + T_{ACK} + \delta \\
 &= 50 + 310 + 352 + \delta + 10 + 304 + \delta + 10 + (192 \\
 &\quad + 20.4 + 14.5 + 29.1 + T_{Voice} + \delta + 10 + 304 + \delta \\
 &= 1606\text{us} + 4\delta + T_{Voice}
 \end{aligned} \quad (2)$$

DCF with RTS/CTS access mechanisms

* δ represents the propagation delay

3. ROUTING PROTOCOL IN AD HOC NETWORK

In our previous work [15], a performance comparison of four routing protocols of VoIP communica-

tion was presented. The results revealed that reactive protocols have better performance for real-time VoIP conversation applications in an ad hoc network. However, most conventional ad hoc routing protocols use the minimum numbers of hops as the metric to make routing decisions. This approach does not take into account the wireless bandwidth and packet size constraints. A new route-discover strategy based on extended AODV protocol is proposed here to further improve the performances of VoIP over MANETs. The details are shown in the following.

3.1 Ad-hoc On-Demand Distance Vector Routing (AODV) Protocol

AODV is a reactive routing protocol, which typically minimizes the number of required broadcasts by creating routes on an on-demand basis. When a source node desires to send messages to a destination node whose route is unknown, it initiates a path discovery process to locate the destination node. It broadcasts a Route Request (RREQ) packet to its neighbors which then forward the request to their neighbors. This is done until the destination or an intermediate node with a current route to the destination answers with a Route Reply (RREP) unicast packet. The RREQ packet contains packet type, source and destination node IP addresses, broadcast ID, sequence numbers and hop count. AODV utilizes destination sequence numbers to ensure all routes are loop-free and contain the most recent route information. Each node maintains its own sequence number, as well as a broadcast ID. The broadcast ID is incremented for every RREQ and together with the node's IP address, uniquely identifies a RREQ. Once a node receives a RREQ, the source IP and broadcast ID is recorded. If the node IP equals source IP or the broadcast ID then the RREQ packet is dropped. Otherwise, a Reverse Route is created and preserved in route table. The node can only generate a RREP if the node IP address is equal to destination IP or it has a fresh enough route based on sequence number. Otherwise, the node forwards the RREQ to its neighbor and increments the hop count. The RREP is sent back along the reverse route and once the source node receives the RREP, it begins using the route for data packet transmissions. A local HELLO packet is periodically broadcast to neighbor nodes for maintaining the local connectivity. If there is a break in an active route, the node which detects a connectivity failure sends a Route Error (RERR) message to the next node in the reverse path. All nodes in the reverse path forward the RERR message until reaching the source node. When a source node receives a RERR message, it may re-initiate route discovery if necessary.

3.2 New Extended AODV (EAODV) Protocol

Most voice conversations in an ad hoc network only use a part of network while the remaining network capacity is not used and thus wasted. Therefore, a new routing method which transmits the voice packets utilizing path diversity (via the unused part of the network) can reduce channel interference and congestion which will improve the network and in particular VoIP QoS.

The proposed routing method is based on a simple modification to the existing AODV's route discovery mechanism allowing selection of an optimal path. By modifying the RREQ packet[16] and the method in selecting the route, the resultant routing is more stable and can increase the PDR. To implement the proposed method, firstly, two time recorders are used to estimate channel activity. One is used to monitor neighbour nodes in the MAC layer, while the other monitors the route layer. When the MAC layer receives a data packet that is not for the specific node, it discards the packet and the neighbour's time recorder records the current time. Otherwise, the packet is forwarded to the routing layer where its own time recorder registers the current time. Following this, an 8-bit channel activity counter is added to the RREQ header. It should be noted that the counter uses the existing 8 reserved bits in the AODV RREQ header which means it does not incur any additional byte overhead.

A time threshold is set to determine the channel activity. The function of the time threshold is to estimate the latest channel activity. It depends on the type of data traffic. For real-time voice conversation, the packet interval is used to define the threshold. In our simulation, the G.729 codec is employed with the packet interval of 20ms. The choice of the threshold greater than 20ms ensures that the protocol will select an idle channel in the network which has not other voice conversation in the transmitting time. This choice avoids channel interference and congestion. Note that if the time threshold is set to be less than 20ms, a free channel can be determined even if it is occupied by another voice conversation. If the threshold is too large, the channel may always be identified as busy even when there is the proper time space to transmit the data packet. By taking into account the above reasons, in the undertaken computer simulations the time threshold is set to 30ms. When the node forwards the RREQ, it first checks whether the current time minus the two prerecorded timings is less than the time threshold. If only one is less than the time threshold, the channel activity counter is incremented by 1; if both are less than the time threshold, it is incremented by 2; otherwise it keeps the record. Lastly, in the destination node a RREQ buffer, a RREQ-Timer and RREQ-Counter are used to select the stable routing. The whole process is

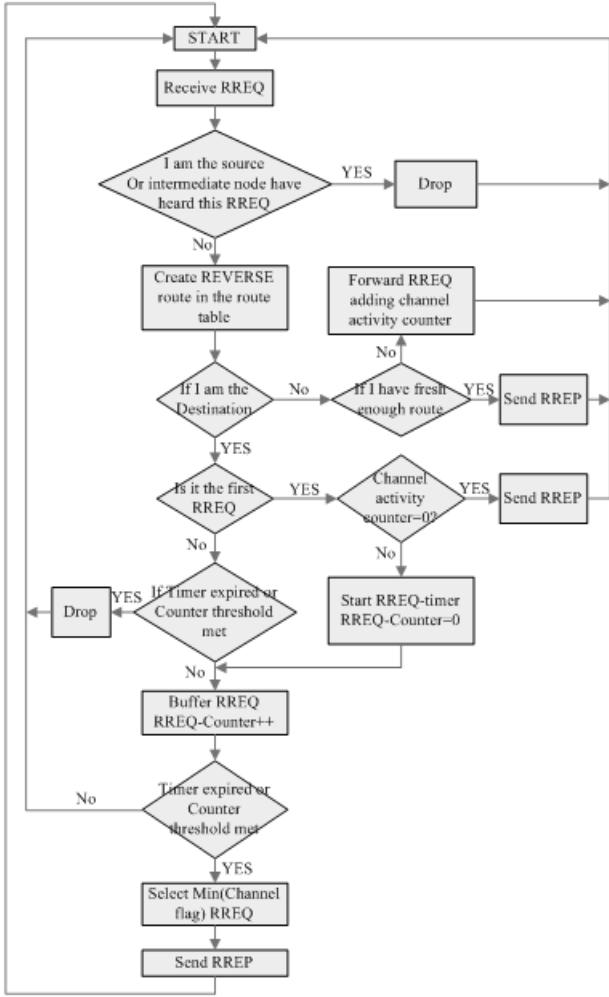


Fig.3: EAODV RREQ Process Flowchart

illustrated in Fig.3.

If a new RREQ is received (determined by the source IP and RREQ ID), the channel activity counter is firstly checked. If the counter is zero it means the channel is free and immediately replies a RREP. If the channel is not zero it means the channel in the reverse path may be occupied by other nodes for data transmitting. The RREQ-Timer and the RREQ-Counter are initialized and this RREQ is cached in the RREQ buffer. The RREQ-Counter is used to record the number of buffered RREQs. Following the first RREQ, subsequent RREQs are directly placed in the buffer until either the RREQ-Timer expires or the RREQ-Counter reaches a certain threshold. The optimal values for the RREQ-Timer and Counter are primarily dependent upon the average node density. In case where the density is high, the RREQ-Counter will reach its threshold value well within the RREQ-Timer. However, in case of a sparse network, the RREQ-Counter may never reach its threshold value before the RREQ-Timer expires. (Note: For computer simulations, we set the RREQ-Timer at 30 ms and the RREQ-

Counter threshold at 8.) Once the RREQ-Timer expires or the RREQ-Counter reaches the threshold, the node begins selecting an optimal routing by choosing the RREQ which has the minimum channel activity counter. Thereafter, a reply RREP is sent to the source node along the reverse route indicated by the selected RREQ. Other later arrived RREQ with same RREQ ID is dropped immediately. In this way, the selected optimal routing has minimum influence on channel congestion by other nodes. As a result, the EAODV protocol looks to be advantageous for real-time applications as it relies on a stable link to guarantee successful data transmission.

4. SIMULATION SETTINGS, RESULTS AND ANALYSIS

Computer simulations are performed using the Network Simulator (ns-2.29) under the following conditions. Radio propagation model is based on the two-ray ground propagation model; antennas are assumed to have omni-directional radiation patterns; media access control protocol uses the IEEE802.11 DCF with two different access mechanisms; nominal data rate is 11Mbps. The node transmission radius is 100 m. The interfering range is assumed to be equal to the effective transmission range.

With reference to [14], the node can interfere with packet reception at another node even when two nodes are set too far apart for successful transmission. In order to overcome this problem, the signal strength threshold (Carrier Sense threshold) is set to be equal to the receive power threshold and therefore interference can only occur in the successful transmission range.

Since many VoIP codecs send data at a constant rate, we use the normal UDP/ Constant Bit Rate (CBR) assumption to simulate VoIP. VoIP in the ad hoc network operates in full-duplex mode. Consequently, in a VoIP session, two nodes are chosen to respectively generate a CBR data stream to emulate two users talking to each other using mobile devices. In the presented simulations, the VoIP session assumes the G.729 codec with the voice payload of 20 Bytes and packet intervals of 20ms. The QoS criterion for PDR is not less than 95% and the average E2E delay is not more than 150ms as mentioned before.

4.1 Representative EAODV Scenario

A representative grid topology scenario is used to analyze the EAODV routine. The static scenario is considered including 33 users located as shown in Fig.4. Two VoIP conversations are generated in node 25-26 (VoIP 1) and node 32-33 (VoIP 2). The scenario is configured using the RTS/CTS mechanism. When the two VoIP sessions are respectively initiated, VoIP 1 chooses the dashed line (25-16-26) path while VoIP

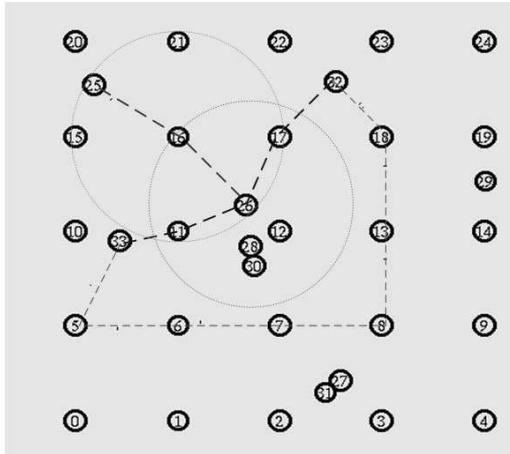


Fig.4: Representative static scenario with two VoIP

2 follows the path (32-17-26-11-33). Both VoIP sessions demonstrate good behavior in PDR and E2E delay. When VoIP 1 is initiated at 0 sec and VoIP 2 is initiated at 20 sec, both sessions finish at 120 sec. Using the AODV routing protocol, the calculated PDR is 97.1% and the average E2E delay is 0.0355 sec. A review of the Ns-2 output trace file reveals that when VoIP 2 initially uses the routing, 32-17-26-11-33, the path is not stable, as the two VoIP flows interfere with each other in node 26. The packet collision and channel competition cause packet drop. This is shown as packet out of queue or MAC collision in the trace file. These results in routing interrupt. Both of the two VoIP flows have to rediscover a new routing. These processes persist until VoIP 2 finds a new stable routing solution (shown in Fig.4 by the dashed line, 32-18-13-8-7-6-33). Concurrently, at approximately 35 sec, the remaining VoIP 1 selects the path 25-16-26. Then, stability is achieved and the two routes remain unchanged until the simulation is finished. However, using our EAODV protocol, the stable path is found immediately and used when VoIP 2 starts at 20sec, while VoIP 1 maintains the initial routing. Both sessions exhibit improved PDR of 99.1% and the average E2E delay of 0.0136 sec. In this scenario, the stable paths for the two VoIP are the same if both AODV and EAODV are used. EAODV, however, find the stable paths at the beginning of the routing discovering and therefore manages to maintain a significant improvement over AODV.

4.2 General mobile ad hoc network Scenario

Here a small-scale mobile ad hoc network which consists of 50 users in an area of 500x500m² is considered. In this scenario, a matrix of 25 fixed nodes is uniformly distributed to ensure all mobile nodes have existing routes with the others within the coverage area. The rest of the 25 nodes (mobile nodes) are randomly positioned. Every node supports ad hoc routing protocols and can communicate with the

fixed nodes or mobile nodes within its transmission radius. Two or three VoIP sessions are generated in the simulations at times of 0, 20 and 30 sec and stop at 120 sec. The conversation nodes are chosen from the 25 mobile nodes.

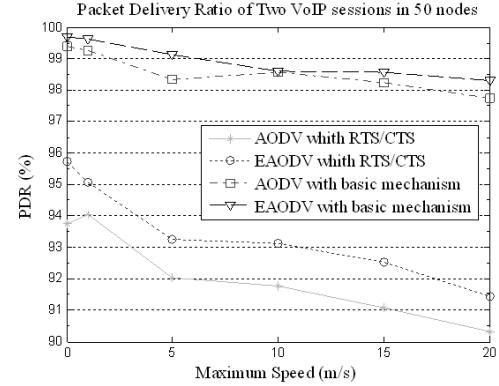


Fig.5: PDR of two VoIP sessions

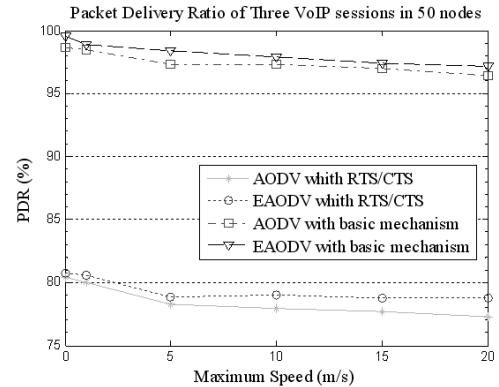


Fig.6: PDR of three VoIP sessions

For each evaluated routing protocol, several simulations are performed, varying the maximum mobile speed and the number of VoIP sessions with two DCF access mechanisms. Maximum mobile speeds of 0, 1, 5, 10, 15, 20m/s are considered. The adopted mobility model uses pause times and random mobile speeds. In particular, all the mobile nodes randomly choose mobile speeds between 0 and the maximum speed. An additional pause time where nodes must wait after reaching their target locations is configured as n sec (e.g. n=10). After running 50 iterations, the calculated average result is shown in Fig. 5-8. (Note that for a fair comparison, same numbers of iterations are used for each scenario and at the same maximum speed.)

From Fig.5 and 6, it can be observed that all PDRs decrease with an increased maximum speed. This is because there are more topologies changes. Using basic DCF, the PDR for both routing protocols is acceptable for two to three voice conversations, which are always above 95%. When a four-way handshak-

ing RTS/CTS mechanism with the AODV protocol is used the PDRs of these VoIP are unacceptable. In turn, using the EAODV with the same DCF mechanism, an improved PDR is achieved compared to AODV. This is because there is a frequent link failure in finding stable routings when the AODV protocol is used.

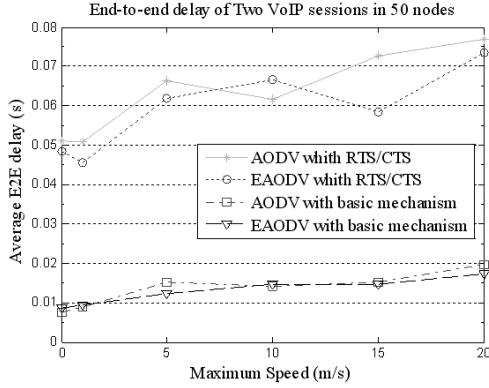


Fig. 7: Average E2E delay of two VoIP sessions

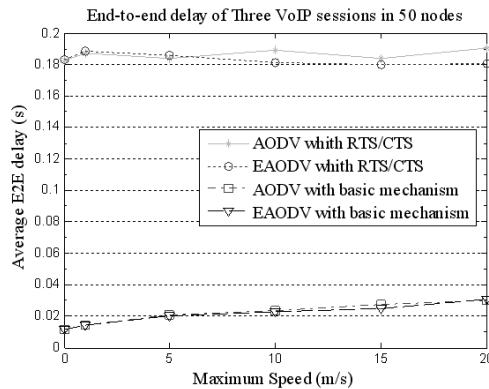


Fig. 8: Average E2E delay of three VoIP sessions

As shown in Fig. 7 and 8, the E2E delay increases in response to an increase in the user mobility and the number of VoIP sessions. It can be observed that the DCF access mechanism influences the average E2E delay. The results show that the basic DCF mechanism has a shorter delay than the RTS/CTS mechanism. In addition, three VoIP sessions with the RTS/CTS mechanism is unacceptable (the E2E delay is approximately 0.18 sec longer than the acceptable 150ms criterion).

It is further noted that both AODV and EAODV have almost the same E2E delay. Due to the fact that standard AODV selects the first incoming RREQ, whereas, EAODV choose the more stable routing and may introduce a longer latency in determining the stable routing step. However, since EAODV can reduce the trend of rediscovery routing, it can make up the latency in rediscovery and therefore achieves almost the same E2E delay comparable to AODV.

5. CONCLUSION

In this paper, the performance of VoIP in an IEEE 802.11b wireless ad hoc network has been considered. A new EAODV routing protocol for ad hoc routing has been proposed and investigated. Opposite to the current one, the extended ad hoc routing protocol utilizes the unused part of the ad hoc network and employs a new metric to discover more stable routings. In addition, the influence of two DCF access mechanisms for VoIP conversation has been investigated. The DCF two-way handshaking basic mechanism offers a better performance in VoIP application in ad hoc network because it has less overhead and is more efficient than the RTS/CTS mechanism. The presented computer simulations have shown that the proposed EAODV allows for a more efficient use of the idle channel routing. In addition, it reduces the adverse effects of channel interference and congestion. The obtained results indicate that the new EAODV protocol can be applied to improve the QoS of real-time VoIP in wireless ad hoc networks.

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