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IMPROVING QUALITY OF VOICE OVER MULTI HOP MANETS

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Abstract - **Providing comfortable high voice quality communication in mobile ad hoc network (MANET) environment is challenging due to its characteristics like ad hoc nature of networks, multi-hop wireless environment with significant packet loss and variable delays. Voice over Internet protocol (VoIP) has originally been developed over the past two decades for infrastructure-depend networks. Quality of voice over infrastructure depend network is achieved by 802.11e Medium Access Control (MAC) protocol, but it is incapable of supporting QoS in Mobile ad hoc network environment due to absence of Access Points (APs) and distributed nature. One of the most important QoS parameter for voice communication is mouth to ear delay. In this work our primary aim is to reduce the mouth to ear delay, thereby reducing the packet loss of voice spell, that is, enhance the quality of voice over IEEE 802.11 e depend multi hop MANETs. This work proposes a new MAC protocol, capable of ensuring a Quality of voice over a multi path MANET and to manage a more number of voice spells in the system. I propose a dynamic voice spell mapping scheme (DVSM) to enhance the quality of voice over MANETs. In particular, an objective metric such as R-factor is considered and a flexibility index is determined, in order to extend the number of acceptable VoIP calls. Our mechanism dynamically map the voice spell to appropriate access category depend on the consequence of voice spell and their packet loss and delay values. Performance evaluation results shows that the proposed model enhances Quality of Service of MANETs by prioritization of voice communication.**

Index Terms - VOIP, MANETs, MAC, Mouth to Ear Delay.

I. INTRODUCTION

Mobile ad hoc network is a composed of mobile nodes without any pre established infrastructure, nodes are self organized themselves to form a network over radio communication links. If communicating nodes present with in a communication range of each other can communicate directly, otherwise makes the use of intermediate nodes for communication. Such type of arrangement in network known as multi hop network, where intermediate node function as routers [3]. Application of such network includes in an emergency operation, military services and law enforcement as it establish with a time and/or cost effective.

MANETs application needs to support real time traffic communication, Due to its wide speared deployment. Real time traffic such as voice communication requires a Quality of service support. One of the important parameter for voice communication is a mouth to ear delay [2]. Each packet in a voice communication needs to be reach destination within the certain value of time limit, after this limit packet become unacceptable. In a single-hop network, mouth to ear delay is not too large as every node is in a communication range of another. However, in multi hop network, mouth to ear delay is quite large as source and destination may be several hops away from each other. Packets from source node need to travel through intermediate nodes to reach the destination.

In this work, I consider the issue of communication voice traffic over CSMA-depend multi-hop MANETs, where the QoS parameter is mouth to ear delay. Excessive delay, jitter and packet loss parameters play a vital act in voice quality acceptance at the receiver end.. In a Wireless Local Area Networks, voice communication quality is improved with the help of access point, as it is a pre defined static central coordinator between communicating entities. However, achieving quality of voice communication in multi hop MANET [12-20] is very much challenging task as it is peer to peer network without central coordinator & packet needs to pass through the heterogeneous intermediate nodes to reach intended destination.

Our identified research issue is performance enhancement of voice spells over 802.11e depend multi hop network within a voice access category (AC's). Our proposed voice spell mapping scheme is depend on prioritization of voice spell traffic in order to meet the QoS goal, where priority to voice spells are depend on R-factor values defined by ITU-T recommendation G.107 standard E-model. This model is used to estimate the quality of vice session during the communication. R-factor value calculated by the: type of voice data (Rmax values), mouth-ear delay impairment factor and packet loss during voice communication. Our aim is to enhance the Packet Delivery Ratio (PDR), it is the

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measurement of the packets reach the destination within a specified duration.

The rest of the paper is structured as follow next section II will give background information of key concept of our research, Section III discuss about proposed work, Section IV gives the details about simulation and results along with future work.

II. BACK GROUND

The IEEE 802.11 [5, 1] standard specifies two types of medium access control mechanisms. One way is contention depend channel access mechanism i.e. Distributed co ordination function (DCF). Second way is controlled channel or contention free channel access mechanism i.e. Point co ordination function (PCF). DCF is highly distributed medium access mechanism depend on Carrier Sense multiple Access/Collision Avoidance and it is default method for Wireless local area networks. DCF is random access mechanism with all nodes in a network has identical priority & probability to access the wireless channel. Whenever the source node want to transmit the packets to destination ,it has to sense the medium if it finds channel as free, then source has to wait for DIFS (DCF inter-frame space) interval, after that interval source transmit the packets. if channel sensed not free, then source has to wait for random interval of time i.e. back off interval and sense the channel again until it finds the channel free. The second mechanism is PCF; it is a centralized mechanism to access the medium where central co coordinator makes the responsibility of medium access mechanism. Central coordinator polls other stations and allows them to access the channel with contention free. In both DCF and PCF methods medium access priority controlled by Inter frame Spacing i.e. length of time spacing between frames.

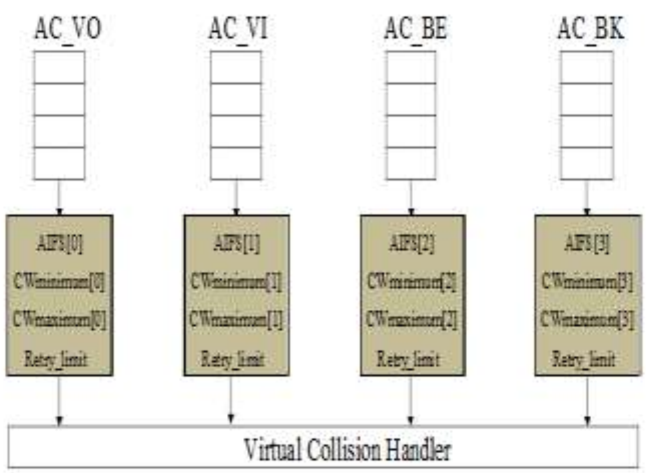

Figure.1 IEEE 802.11e MAC EDCA

The IEEE 802.11 e EDCA [6] proposed to enhance the 802.11 DCF by allowing distributed channel access, way to support service differentiation between different classes of traffic, shown in fig.1. EDCA categories traffic into four Access categories which include AC_VO for voice traffic, AC_VI for video traffic, AC_BE for best effort and AC_BK for background traffic. EDCA defines each access category with different MAC parameters settings, due to these parameters settings access categories have different priority. Parameter settings include AIFS (), CWmin(), CWmax, and retry-limit. Each AC has its own buffered queue and act as a self-sufficient back-off entity.

Voice traffic over Internet not only aims to maximize throughput as well as to meet the Quality of Service requirements by considering the characteristics of real time voice traffic. The VoIP system [15] comprises of analogue to digital conversion where analogue voice signals are digitized, compressed and encoded using a voice codec ITU-T G.711 with data rates of 64kbps, G.729 with 8kbps, G.723.1a with 5.3/6.3 kbps data rates, etc. into digital voice streams. The output encoded voice stream generates audio packets of constant bit rates within periodic time intervals. Construction of frame is carried out by Medium Access Control (MAC) layer as packets are delivered. The reverse process of decoding and de-packetizing occurs at the receiving side into analogue voice to reach its destination. [6] During the transmission process the voice packets are affected by differing latency that results in variation of time interval for packet arrival called Jitter. Packets are queued into a Di-jitter buffer at the receiver end to maintain a constant inter packet interval. Then VOIP packets are de-packetized and converted into analog voice, finally reach the destination [7].

VOIP code	G.711	G.729a	G.723.1	G.722
Bitrate(kb/s)	64	8	6.3	64
Tpk (ms)	20	10	30	20
Pvoice (bites)	1280	80	189	1280
Rmax (value)	93.2	93.2	93.2	129

TABLE 1. Voice parameters of different coding

III. PROPOSED WORK

In this work I propose a MAC protocol which aims to extend the Quality of voice in 802.11e depend multi hop MANETs depend on priority of voice spell. In our work i am considering the impact of multiple voice spell transmission over 802.11e network, with danger buffer underflow of an intermediate node. I take consideration of R factor values of voice spell depend on mouth to ear delay and packet loss. I assign the priority to voice spell depend on the calculated R factor value. According to voice spell priority voice traffic is initiated on medium.

Our contribution of work mainly as follows

1. Calculation of R factor value of voice spell depend on packet loss and delay

2. Priority of voice spell

3. Buffer sub categories

4. Voice communication depend on priority

1. Calculation of R factor value of voice spell depend on packet loss and delay

In order to calculate voice quality, most popular 'non intrusive 'objective calculation technique is E- model [9] is used in our work. It is a computation technique depends on assumption made to voice quality degradation with respect to damaging factor, without any additional traffic inject to network. E-model makes the use of transmission parameter of network to calculate the quality of voice. This model determines the base value to calculate the quality of voce depend on network factors. Each damaging factor to voice quality degradation is expressed in terms of certain value [9]. Both Base value and damaging factor values used to determine the quality of value. In our work I am considering damaging factors are mouth to ear delay and packet loss during the voice spell and base value is Rmax. The primary out come from E-model is "Rating Factor" R is used to evaluate the quality of voice. Calculations of R factor depend on active testing; depend on the generation of test traffic between points in the network to continually assess impairments.

$$R=R_{max} - I_{delay} - I_{loss}$$

Fig below shows different Rmax values of different voice codes [10].

The idea behind calculation of R factor comes from the standard proposed by ITU (Rec.862, Feb.2001) [11] for the objective evaluation of the coded speech signal that goes through the telephone network. PESQ [11] adds new factors and procedures to calculate signal distortion of speech samples. This model is recommended procedure for calculating of a codec quality and network testing. It is applied for the evaluation of Packet loss, and delay [4].

2. Priority of voice spells

I prioritize the voice spell packets depend on packet loss and mouth to ear delay. As Excessive delay, jitter and packet loss can all makes the quality of voice to end users unacceptable. I use the measure R factor to decide the priority of packet. According to ITU-T [2] recommendation the R factor has a value between 0 and 100, with 100 representing the best quality and 0 the worst quality. However values of below 50 are generally unacceptable and the value between 90-100 are Very satisfied and 80-90 are satisfied Some users satisfied with 70-80.This value gives us to rough estimation of voice spell condition. Hence, i am assign the priority to voice spell, the packets of voice spell with R factor more than 90 gives to lower priority, voice spell with R factor between 80 to 90 medium priority and voice spell with R factor between 50 to 80 is higher priority.

3. Buffer sub categories

Fig-2 shows the proposed model Access categories architecture. 802.11e AC architecture, AC-VO has divide into three sub access categories and each type of voice spell is mapped to the appropriate sub AC-VO. Sub access categories including namely high priority voice sub access category AC_VO_1, medium priority voice sub access category AC_VO_2, low priority voice sub access category AC_VO_3. Sub access category AC-VO_1 has higher priority to access the medium whenever AC_VO capture the channel, then AC_VO_2 and AC_VO_3.

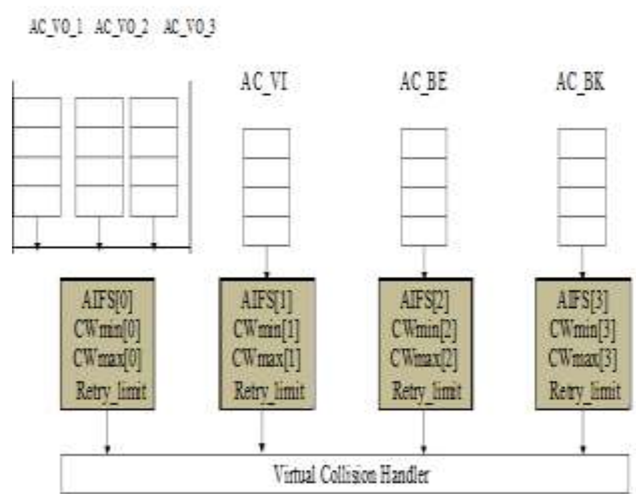

Figure2. IEEE 802.11e MAC EDCA

4. Voice communication depend on priority

Proposed scheme is depend on existing 802.11e MAC protocol with minimal modification. The channel access technique is same as in 802.11e, each AC compute for channel and any one of four (AC_VO, AC_VI, AC_BE, AC_BK) is capture the channel. Whenever MAC layer receive the packet from network layer it will store some of the

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information related to packet while it inserting the packet into on its four access categories, information like transmission rate need to transmit the packet ,whether the packet need to encrypt or not , encryption key and preamble needs to precede packet. Adding the additional information does not impact on storage memory. Thus i add the information related to priority of packets and i modify the queue of access category of AC_VO into three sub access categories. The relevant additional information is R factor of voice spell of packet. If AC_VO of 802.11e capture the channel the packets are transmitted as FIFO (first in first out) in existing 802.11e. But in our model I have assign the priority voice spell packets and sub divide the access category of 802.11e AC_VO into sub access categories and assign them also priority . voice spell packet with R factor more than 90 mapped to AC_VO_3,voice spell with R factor between 80 to 90 mapped to AC_VO_2 and between 50 to 80 mapped to AC_VO_1.When AC-VO gain access to the wireless channel, spell which has R factor between 50 to 80 get high priority and access the channel ,after the completion of this spell next priority gained by R_{factor} between 80 to 90 and then R_{factor} higher than 90.Our mapping scheme enhance high priority to spell with R_{factor} between 50 to 80 without throughput reduction of other AC's by only considering AC_VO to map the voice spells.

Algorithm

1. AC_VO Capture the channel
2. Voice Spell[i] mapped to AC_VO_1
3. If calculation interval= true then
4. For Voice spell[i]
5. Calculate the value of R_{factor}=R max – I delay – I loss
6. End for
7. Else wait for calculation interval become true
8. While tuning interval = = true
9. If 50 << R_{factor} << 80
10. Voice Spell[i] mapped to sub access categoryAC_VO_1
11.
12. Else If 80<< R_{factor} >>90
13. Voice Spell[i] mapped to sub access category AC_VO_2
14. Else Voice spell[i]= AC_VO_3
15. End if
16. End if
17. Wait for tuning interval to become true
18. End if

The above algorithm works whenever all AC compute for channel and AC_VO capture the channel. Then voice spell is mapped to AC_VO_1, all the packets of voice spell are buffered at buffer AC_VO_1, as i assigned higher priority assigned access to channel to sub AC_VO_1 followed by AC_VO_2 and AC_VO_3. The packets in the sub access categories relocated depend on their priority depend on R factor value. If packets at buffer AC_VO_1become empty, AC_VO grant the channel access to AC_VO_2 followed by AC_VO_3.However no modification may occur at a TI- Tuning interval of time and values for R-factor are computed at every CI- Calculation interval

IV. PERFORMANCE CALCULATIONS

I investigate the performance of proposed DASM MAC protocol using the ns-2 simulator. I implemented the DASM protocol with the necessary extension to NS-2 simulator and compared it with existing 802.11e MAC protocol in a same network condition with respect to Throughput and PDR. In our simulations, I use a fixed transmission range of 250 meters, which is supported by most of real time and current network interface cards. All nodes to be equipped with IEEE 802.11 network interface card and data rates of 2 Mbps. I am considering the static environment as dynamic environment impact on routing path and network layer. Our work is depend on MAC layer and is independent of routing protocol. Traffic spells are generated randomly on selected different source-destinations.

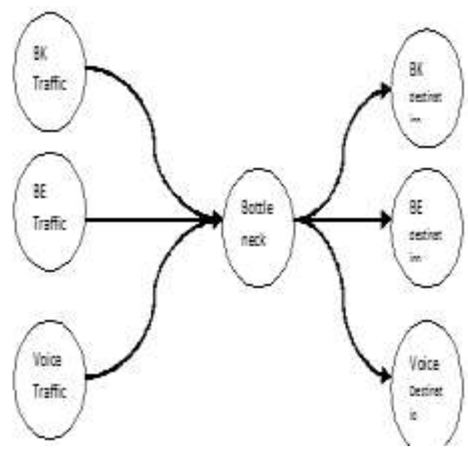

Figure3 . An example of bottleneck bode

Every node in a network has to run our algorithm whenever it becomes an intermediate node to forward the information of source nodes. Each simulation was run for the duration of 10 minutes and sampled data i collected from simulation is average of 3 times. Simulation parameters are shown in table.2.Topology of our simulation is random and traffic generation is also random. In our work i are considering the impact of multiple voice spell transmission over 802.11e network, with danger buffer underflow of an

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intermediate node. According to it i built the topology as shown in fig.3 which accommodated multiple Background, Best-effort traffic and G.711 VOIP calls which modelled as constant bit rate (CBR) of 160 bytes payload and 40 bytes RTP/UDP/IP header overhead. I have taken multiple voice spell claims for medium access in presence of TCP. I define three types of voice spell; include high priority voice spell, medium priority voice spell, low priority voice spell. Calculation of PDR in our work is the ratio of number of voice packets received by application layer of source station to the number of voice packets received by the application layer of destination station. And throughput is computed by total number of bytes received by destination and is divided by total end to and delay.

Network Parameters	**Values**
Number of Nodes	50 to 100
Link layer type	Logical Link (LL)
MAC Type	802.11e, DSMA
Routing	AODV
Traffic	CBR, FTP
Network Area	1500mx1500m
Transmission Range	250m

Table 2. Network parameters used in Simulation

V. RESULT ANALYSIS

Calculation of PDR and throughput is done for voice spell in the presence of other traffic of BK- Background, BE-Best effort. Simultaneously three different traffics are flowing through bottleneck node i.e. Transmission of CBR traffic from the first node, TCP transmission from second node and UDP transmission from the third node and I calculated the throughput for voice traffic shown in fig, represents throughput calculation of voice traffic in presence of other traffic such as best effort and back ground. Results of comparison are shown by the 802.11e with our proposed approach (DASM). Eventually, conclusions from the fig.4, 5 shows that the proposed mechanism enhanced the throughput of voice spells comprising real time traffic in the presence of background and best-effort traffics because the flows which use low priority AC share the channel access probability with voice flow.

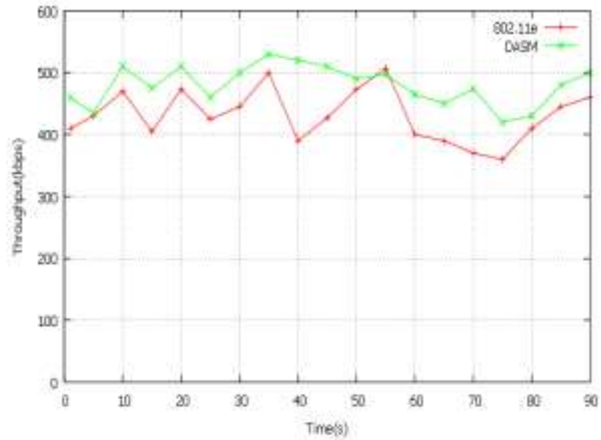

Figure 4. Throughput Comparison of Flows in AC_BK

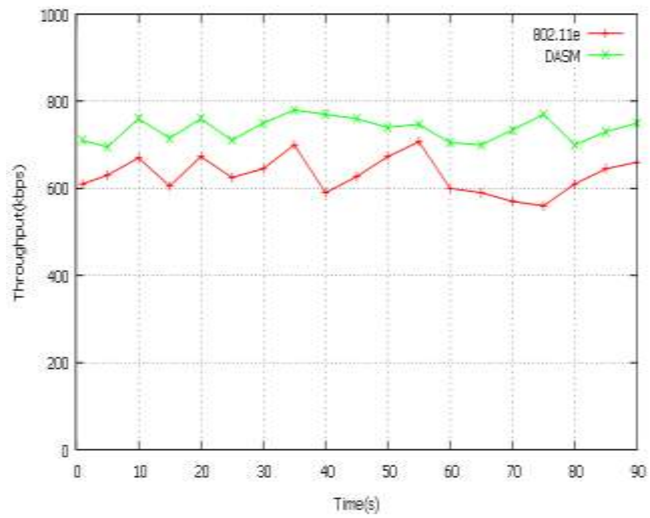

Figure 5. Throughput Comparison of Flows in AC_BE

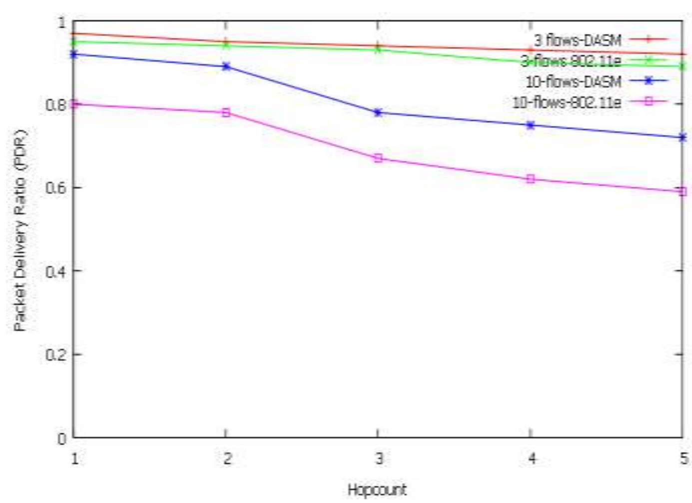

Figure 6. Variation of PDR Vs hop count for voice traffic

Fig.6 shows clearly as load increases the PDR decreases due to congestion at bottleneck node. At lower load there is no observable difference between 802.11e and DASM as there is no congestion. As the hop count increases DASM performance is better than 802.11e. This is due to the fact I assign the priority to packet depend on one of the factor delay. However performance of DASM not decreases drastically as that of 802.11e

VI CONCLUSION

A dynamic voice spell mapping scheme for voice traffic is developed to enhance the VoIP quality over the IEEE 802.11e depend mobile ad hoc network. Enhancement of QoS voice over MANETs is achieved by the calculation of R factor of each voice spell in communication. Priority to voice spell packets are given depend on packet loss and mouth to ear delay value of voice spell. Depend on priority these packets are suitably mapped to secondary access categories of AC_VO. An enhancement in voice traffic quality is shown in contrast with the 802.11e with PDR and throughput.

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