Delay and Power Efficient Voice Transmission over MANET

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Abstract: *Increasing speed of hardware device and versatile functionalities of small equipments e.g. laptop, PDA etc. are introducing various voice oriented applications with mobility. Like other computer networks, in Mobile Ad-hoc Network (MANET) voice transmission is very much demanding and necessary. In this research paper we have a feasibility analysis of voice transmission over MANET. Since voice applications consume more energy than typical applications, we use an energy aware routing protocol known as WEAC for the study. We have a comparative study among several audio codecs (G.711, G.729 and G.723.1) and by simulation we show that the G.729 codec is suitable to use for voice transmission over MANET in terms of latency. We show that it is possible to launch voice transmission with acceptable quality over MANET using G.729 and WEAC protocol.*

I. Introduction

The evolution and deployment of wireless network since last few decades has raised the importance of mobile and decentralization concept in wireless communication era. This directly guides the trend to the advent to the MANET in wireless world. Every node in MANET uses a shared wireless channel standardized by IEEE 802.11. The wireless links in an Ad-hoc network are highly error prone and can go down frequently because of node mobility, interference, channel fading, and the lack of infrastructure. Voice over IP (VoIP) is the next technology to replace old and expensive Public Switched Telephone Network (PSTN) with the backbone of the internet, which makes integrated voice oriented applications over ad-hoc network very demanding and useful. Voice transmission over MANET is attractive for applications and environments, such as conference and convention center communications, as well as emergency response scenarios such as law enforcement and military operations. Ad-hoc networks are a good choice in marine operations where infrastructure is likely to be absent. Different types of audio oriented applications are increasing rapidly in small devices. Therefore transmitting voice over multi-hop ad-hoc network is set to make a revolutionary change in wireless communication.

All the nodes in a MANET are to operate using battery since there is no fixed infrastructure to support them. Therefore, consuming less energy is another important issue for launching a MANET. Among table driven and on-demand routing protocols, table driven is already proven to be non-suitable for routing in MANET in many literature studies since MANET is a frequently changing network [5]. On the other hand, on-demand routing protocols also show poor performance in large MANET because they flood the network with route request and route reply packets. Since all nodes in MANET can act as routers, a highly connected node can drain its battery soon. To improve the efficiency of routing, the use of cluster-based structure is introduced in ad-hoc network for the purpose of managing wireless transmission among multiple nodes and reducing channel contention by reducing the flooding of control packets. In cluster based network, the whole network is divided into several clusters depending on the connectivity. This structure of the cluster and its components are highly transient in the course of time in a MANET. In this study we use a cluster based as well as energy aware routing protocol known as Warning Energy Aware Clusterhead (WEAC) [1] to determine the feasibility of launching voice application over MANETs.

WEAC in Brief: It is a cluster based ad-hoc network topology creation protocol which combines both cluster-based and on-demand routing protocol concepts. A node is selected to perform as a base station for a period of time. This base station node is called the clusterhead. Since this is an energy aware routing protocol, the primary concern of the protocol is to minimize and utilize the energy of all the nodes homogeneously so that it increases the network lifetime. Unlike other typical traffic, voice traffic consumes a lot of power during compression and decompression processes. Therefore the utilization of power is a critical issue to increase the network lifetime while transmitting voice. The WEAC protocol shows quick response in topological changes and is scalable to an area where large population of nodes is expected. In this protocol the main criteria to classify a Mobile Terminal (MT) is the Energy Level (EL). Depending upon the EL, MTs are classified into four types which are shown in Fig.1.

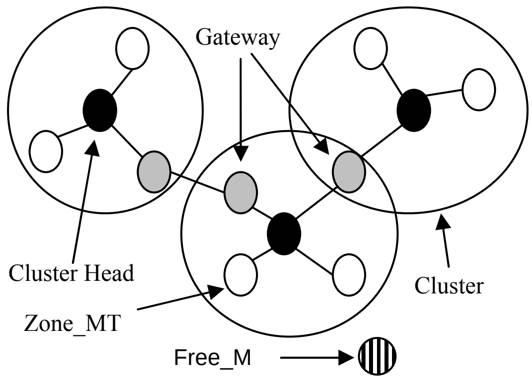

Fig.1: A typical cluster based MANET

• Clusterhead: This is the node which will act as the base station in its region. There will be a single clusterhead in a region. Usually an MT having more than a threshold amount of EL is selected as clusterhead.
• Zone_MT: This is a typical MT which is supervised by the clusterhead. Usually Zone_MTs have lower EL than the clusterhead.
• Free_MT: An MT which is not a member of any cluster.
• Gateway or Border MT: This is the MT which works as a bridge between two clusters. Location of a node is the criteria to become a gateway. A gateway can be a clusterhead or Zone_MT or Free_MT.

To keep track which kind of MT it is, a MT maintain a flag variable called myCH. The MT's myCH variable is set to the identity number (or id) of the clusterhead which supervises it. The myCH of a clusterhead is set 0. All the other MT's myCH is set to -1 to initially indicate that all of these are Free_MT. Within a short period of time the topology forms and the myCH variable changes as necessary.

Let us consider Fig.2, where the EL of a MT is classified into three threshold levels.

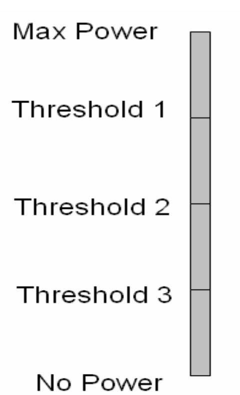

Fig.2: Different EL of MT's

All the nodes periodically broadcast hello messages to their neighbors. Along with this hello message they broadcast their identity, myCH, current EL, etc. After receiving the hello message different MTs operate differently.

Depending on the EL of an MT, the MT is classified into on of the following four categories:
• EL ≥ threshold 1: The MT has the ability to supervise other MTs. It accepts other MT's request to be a supervisor. If two MT's have same EL, the one with more neighbors will be selected as the clusterhead and the other MT's myCH variable will contain the value of the new clusterhead. An MT joins within a supervisee list of a clusterhead if it is a Free_MT or it receives a 'I am no longer your clusterhead' message from its clusterhead.
• Threshold 2≤ EL < threshold 1: The MT still has the ability to supervise, but does not allow other nodes to join in its supervisee list. That is, if an MT serving as a clusterhead it remains clusterhead but does not add more supervisees in its supervisee list so that it lasts for longer time.
• Threshold 3≤ EL ≤ threshold 2: In this energy level a node can not be a clusterhead rather an existing clusterhead with this EL sends a warning message to all its supervisees to look for another suitable clusterhead. Free MTs send a merge request to their neighbors whose EL is above or equal to Threshold 1. This way the clusterhead give time to the supervisee to find a new clusterhead instead of sudden hang up.
• EL ≤ Threshold 3: An MT ignores a merge request. The clusterhead will inform its supervisees that it is no longer their supervisor. The supervisee myCH will be -1 and they will find alternative clusterheads. The details of the WEAC protocol are described in [1].

In [9] it is stated that to satisfy the QoS of audio traffic, the maximum tolerable end-to-end delay and the delay jitter should be 250 and 150 millisecond, respectively. We use these parameters in our simulation study. The structure of the paper is as follows: The next section contains the related work. In section 4 and 5 we describe our approach to the problem and the simulation model, respectively. Finally we conclude our work in section 6.

II. Related Work

So far a lot of research has focused on audio transmission for a typical network. But very rare studies considered audio transmission over a MANET. Since MANET has strict bandwidth and power constraints, the voice coding technique and the power management of a MANET must be sophisticated to provide an acceptable voice service.

Many audio coding techniques already exist, however, not all of these are equally suitable for MANET Authors in [2] evaluate the performance of different audio codecs for real time voice transmission over an IP network using the Mean Opinion Score (MOS) method. The study shows that audio codec G.723.1 and G.729A outperform other codecs in terms of quality.

Since the power issue in ad-hoc network is also critical, many studies have focused on this area. Authors in [10, 11, 12, 13] propose energy aware routing protocols to maximize the lifetime of network. On the other hand some authors in [3, 4, 7] focus on the transmission mechanism and the path loss model of an ad-hoc network to optimize and reduce transmission power.

A lot of work has been done on audio traffic over conventional networks and especially the Internet. Different kinds of network protocols such as ATM, MPLS, have been devised to support the quality of service required for the voice traffic. Voice traffic is deemed of higher priority compared to typical data traffic. But very few studies have introduced audio packets over wireless network especially for ad-hoc network. In [14], the authors propose a priority mechanism to provide QoS for voice traffic in an ad-hoc network scenario using an on-demand routing protocol. To improve the end-to-end delay for real time traffic over ad-hoc network, of the study in [6] proposes a new protocol named Differentiated Services-Stateless Wireless Ad Hoc Network (DS-SWAN).

III. Approaching the problem

In this research work, we analyze and measure the maximum number of hops a MANET can support for voice traffic, which is very important to know before launching a MANET. We also focus on the efficiency of the network for different number of nodes. To transmit voice a node needs to capture the audio signal by vocoder. Then it needs to compress the data before transmission so that it maximizes the bandwidth utilization. To measure the end-to-end delay of a voice packet being transmitted from sender to destination we consider following types of delays and performance matrices in our simulation program:

- *Routing delay*: Routing delay is the average delay per packet, which is required to find the path from the source to the destination.
- *Compression and Decompression Delay*: Any node wishes to transmit an audio file to other node will have to compress the file by some compression techniques to reduce bandwidth requirement. On the other hand the receiver needs to decompress the data using reverse technique of compression. These two kinds of delays will be imposed on the sender and receiver node respectively.
- *Processing Delay*: the delay that occurs in the node while processing the packet for transmission. Per packet delay is packet size over the link speed.
- *Propagation Delay*: the time required to propagate one bit through the wireless media.
- *Media Access Delay*: the defer delay, the delay that results from competing for the medium.
- *Acknowledgment and Retransmission delay*: the delay that occurs due to the overhead to guarantee the error free reception of packets by the receiver.
- *End-to-end Delay*: the total time required for one bit to traverse from the sender to the receiver.
- *Delay Jitter*: the fluctuation or variation of end-to-end delay from one packet to the next packet within the same flow of packets. The reason of the delay jitter is due to the large waiting time in queue because of network congestion. More unstable network has more delay jitter.

The less the end-to-end delay and delay jitter are the better the quality of the voice is.

Optimization is required to support voice traffic over MANET. We use power optimization approach in power equations to optimize power consumption.

Path loss model and power minimization: Let us consider P_t and P_r are the transmitted and received power respectively. Then the ratio of the power is:

$$\frac{P_r}{P_t} = G_t G_r \left(\frac{\lambda}{4\pi d}\right)^{\alpha} \tag{1}$$

where G_t and G_r are the transmitter and receiver antenna gains, respectively. λ is the wavelength of the radio signal, α is the path loss gradient, and d is the distance between the transmitter and the receiver. If P_0 is the signal strength received at one meter distance then:

$$P_0 = P_t G_t G_r \left(\frac{\lambda}{4\pi}\right)^{\alpha} \tag{2}$$

Therefore, one can write

$$P_r = \frac{P_0}{d^{\alpha}} \tag{3}$$

We assume α to be equal to 2 as in the free space path loss model. The power required to transmit one bit of information in HCB model is defined by the equation (4), which was also used in [7].

$$T_x = P_t \times d^{\alpha} + c \tag{4}$$

where c is an additional receiver power, assumed to be 2, required in the relay node. We calculate value of P_t from (2) considering SNR=18dB, background noise =-120dBm. Therefore, P_0=18+ (-120) = -102 dB= $10^{-10.2}$ mW. If we use G_t=G_r=1 from (2) we get P_t=6.382x10^{-7} mW = -61.95 dBm.

The authors in [7] state that cooperative routing is sometime more efficient than direct transmission. From equation (1) we see the transmission power is proportional to the square of the distance, hence it more energy efficient to transmit using intermediate nodes. To depict this using an example, assume there are two nodes A and D that are neighbors of a third node S as shown in Fig.3.

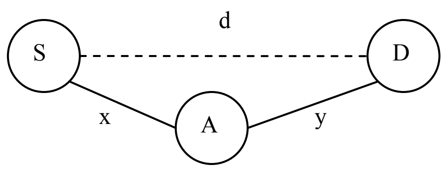

Fig.3: Use of intermediate node to conserve energy

If S has to send packets to D and since both of A and D are neighbors, S either sends directly to D or alternatively it could send via A. To reduce the required transmission power, S uses following algorithm according to [7]:

If $d \geq \left(\dfrac{c}{P_t(1-2^{1-P_t})} \right)^{\frac{1}{2}}$

If $(P_t{*}x^2+c) + (P_t{*}y^2+c) < (P_t{*}d^2+c)$

Transmit using node A

Else transmit directly to node D

End if

Else transmit directly to node D

End if

This would reduce the transmission power at the expense of increased delay. Since the number of hops increases, the delay also increases due to additional receiving and queuing delays. In our study we focus on the behavior of the protocol with voice traffic which is very much sensitive to delay. On the other hand, the protocol is also concerned of energy. Therefore, as a trade off we use this energy saving enhancement only if a node sends some packets to its neighbor. This would increase delay to one hop distance traffic only, which will still remain far smaller than 250 milliseconds.

Mobility Model: Among many other mobility models we chose random waypoint mobility model since it is the most suitable mobility model for Ad-hoc network. Here pause time and random direction of the nodes are used to calculate the next position after every pause time. The direction θ of the node is chosen randomly between 0 to 2π. The speed of a node is also selected randomly between minSpeed and maxSpeed which are 0 and 5 km/hr, respectively. Fig.4 shows an example of traveling pattern of mobile nodes used in this mobility model where a node starting position is (415,150).

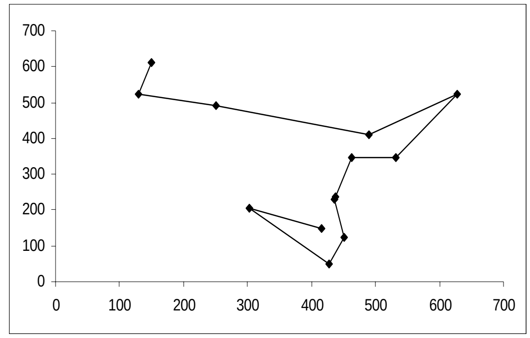

Fig.4: Traveling pattern of node in random waypoint mobility model.

If a node comes to the boundary of the simulation area, that node is forced to remain on the boundary unless it changes its direction of movement away from the border.

Voice Codec: Selection of the suitable compression technique is very crucial especially when the bandwidth is limited. There are many audio compression techniques proposed in the literature and among the commonly used by International Telecommunication Union (ITU) are those listed in Table 1.

Codecs	Bit Rate	Delay
G.711	64 kbps	Negligible
G.729	8 kbps	25 mS
G.723.1	6.4 kbps	67.5 mS

Table 1: Specification of different codecs

Among these, G.729 has reasonable bit rate with toll quality voice and it is one of the most tested standard for applications including wireless networks. The vendor interoperable G.729 has different Annexes to support variable bit rates. Moreover to determine appropriate voice codec we performed a simulation based comparative study (described in the next section) among these codecs.

MATLAB is used in this simulation program. We use 2km by 2km simulation area where nodes can move in promiscuous mode. The RF-range of every node is 250 meter. The compression and decompression delay are 25 milliseconds. The size of the control packets are 1KB. Every node sends hello message after every 1 sec periodically. Initially all the nodes are charged with full power capacity which is equal to 100 joule. When this power goes bellow 30 m-joules the nodes are recharged immediately. There are three threshold power mentioned in the protocol that we use. Those are threshold1, threshold2 and threshold3 value of which are 3/4th, 1/2nd and 1/3rd of the full power respectively. We consider the worst case scenario where all the nodes are transmitting voice to other nodes randomly. The service rate is proportional to the link speed which is 2 Mbps.

To obtain realistic value for coding power, we consider the low cost voice processing engine AC4880X produced by AudioCodes Ltd. The AC4880X voice over packet processor combines one or two channels of toll quality low bit rate voice compression and provides G.729 compression/decompression using a small amount of power equal to 180 mW.

IV. Result and Discussion

To determine which codec among G.711, G.729 and G.723.1 performs better with the WEAC protocol we use a simulation program and Fig.5 and Fig.6 show the end-to-end delay and delay jitter of audio packets in a MANET, respectively.

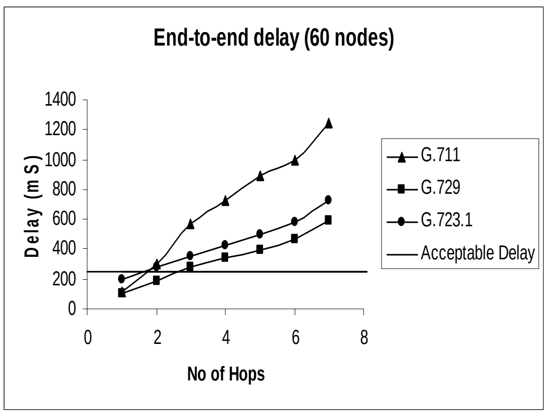

Fig.5: End-to-end delay of different codecs

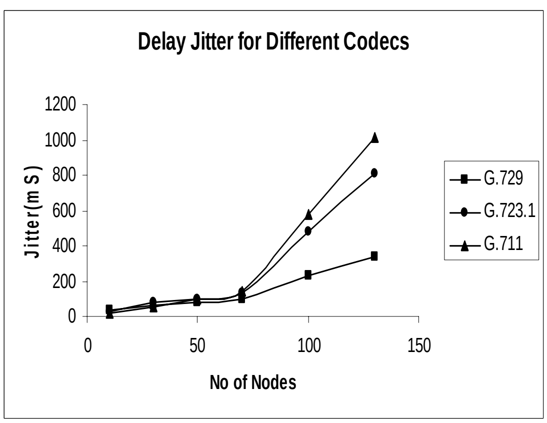

Fig.6: Delay jitter of different codecs

G.711 has negligible delay. But since its bit rate is very high, lots of audio packets are injected in the network to support the same arrival rate. Consequently it causes more delay. On the other hand G.723.1 has lowest bit rate. But due to its high compression decompression delay it does not perform well too. Therefore G.729 in fact performs well because of its reasonable bit rate and latency. Though all the codecs have almost same delay jitter for small network, but for larger networks G.729 performs better. Therefore we conclude that G.729 outperforms the other codecs.

For the second experiment, we transmit voice over MANET using G.729 and WEAC protocol. Fig.7 depicts the end-to-end delay versus the no of hops traversed by the packet. It is noted that for longer pause time i.e. for less mobility, the maximum number of hop supported is four. However, the number of hops decrease as mobility increases.

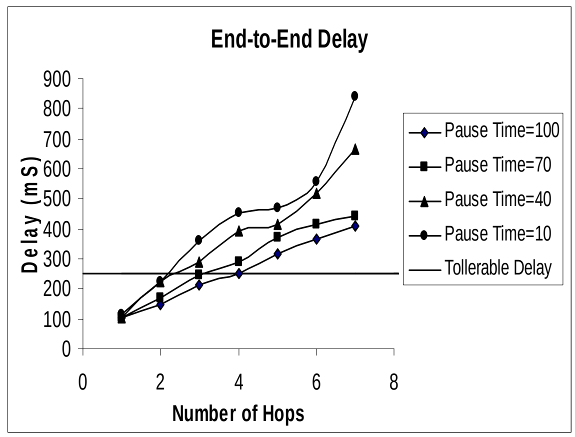

Fig. 7: End-to-end delay for packets vs. no of hops traverse for 60 nodes for different mobility

The ratio of number of packets received to number of packets sent shows the efficiency of the overall network. Fig.8 plots this ratio as a function of the number of nodes. We see that this ratio increases with the increase of number of nodes in the network, because in case of low number of nodes, the connectivity is low and lot of packet drops occur. When the number of nodes increases beyond 80, the packet drops increases because of delay constraint packet drop increases and the efficiency again decreases.

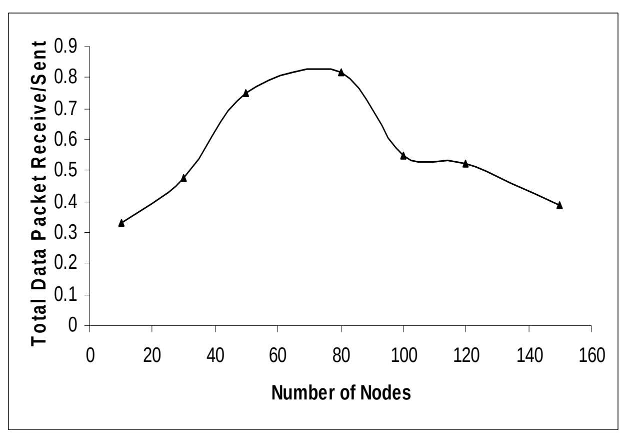

Fig.8: The Efficiency of the network

Finally, Fig.9 shows the number of nodes in the network versus the standard deviation of remaining power in the individual nodes. The standard deviation shows that the protocol utilizes the power of nodes uniformly.

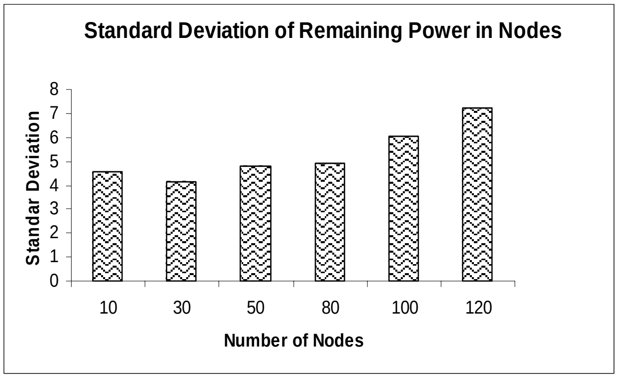

Fig.9: Standard Deviation of the remaining energy in the nodes of the network.

V. Conclusion

Our study of the delay analysis for voice traffic over MANET reveals that if we use an ad-hoc network with typical configuration and compression technique with the utilization of the WEAC routing protocol, it is possible to launch ad-hoc network with reasonable voice quality and service. For an RF-range of 250 meter, the network can transmit voice traffic up to 1km away utilizing at most four hops. Moreover, the study considered the worst case scenario where all nodes communicate all the time. Therefore we conclude that when mobility is low and the size of the network is small, it is possible to launch voice over MANET using the codec G.729 and WEAC protocol with efficient power utilization.

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