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CBT - VOIP: CODEC BASED TECHNIQUE TO REDUCE VOIP TRANSMISSION DELAY IN MOBILE AD-HOC NETWORKS

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Abstract: Mobile and wireless technologies provide Voice over Internet Protocol (VoIP) service to the users with the help of internet. However, there are some limitations with respect to bandwidth, delay, jitter, etc. While it meets congestion and heavy load on the network, it drastically consumes and wastes the resources unnecessarily. Likewise, Mobile Ad-hoc Networks (MANETs) is a part of ad hoc network to provide VoIP communication among the users without the aid of any internet facilities. Since the nodes are mobile in MANETs, it faces many challenges like dynamic topology, limited bandwidth and battery power in order to provide Quality of Service (QoS). This paper proposes a CBT-VoIP technique for reducing the VoIP transmission delay. Ultimately, it helps to improve QoS in MANETs.

Keywords: CODEC, Voice over Internet Protocol (VoIP), Bandwidth and Delay, TACA.

I. INTRODUCTION

In today's modern networking field, advancements provide a much needed task for all the technologies. Wireless communication is the one among them and has shown improved performance in attracting the society's needs. It offers a lot of benefits and also has several pitfalls. This makes the researches look for an overview with deeper knowledge. The Network is commonly classified into two methods. One is Infrastructure based and another is Infrastructure-less. Both have their own merits and demerits. In an Infrastructure, every user must be connected with the centralized access point or tower (base station). Access Point (AP) is providing services to the mobile users. Whereas in an Infrastructure less network, each user can take their own decisions and no common central server is placed. This network is configured quickly in an On-demand manner with the available nodes. Here, Collection of mobile nodes forms a network dynamically in order to exchange the information. This network offers different kinds of services. Among them, voice is a much needed application for interacting directly with participants. Codec is a hardware or software used to process analog speech (users' dialogue) samples into digital bits to save the bandwidth utilization. In MANETs, the voice based services are achieved through the Voice over Internet Protocol (VoIP). Codecs are available in different flavors. They are G.711, G.729, G.723, G.726 and Adaptive Multi Rate (AMR), etc., All of these codecs are varying with their characteristics like bandwidth, packetization delay and other things. Generally VoIP communication is done by using the internet but MANETs is not required for internet connectivity. Therefore, bringing the voice feature in MANETs is an important and challenging task.

This paper is structured as follows. Section 2 discusses the review of literature. Section 3 illustrates the proposed CBT-VoIP technique and Section 4 deals with the Results and Discussions. Section 5 shows the conclusion. Finally, references are listed.

II. RELATED WORKS

Several techniques have been proposed for improving the performance of real-time applications while on communication. However, only a few techniques focused on mitigating the delay of MANETs. Since, from source to destination end, the packets must be handled carefully. Connection-less approach does not care about the packet loss and other retransmissions, etc. Some of the literatures are discussed in the following section with respect to VoIP delay.

Shiraz et al., [1] analyzed the features of OLSR and TORA protocols for VoIP services to improve the QoS parameters like, throughput, jitter and delay, etc. They achieved the results through the OPNET simulator. Finally, they accomplished that, OLSR outperforms better than the TORA protocol for voice services in small networks. Uchenna et al., [2] discussed the characteristics and

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challenges (QoS) of implementing VoIP in MANETs. It was quite difficult to provide voice services in MANETs and so many protocols were poorly configured to do this. But, TORA protocol provided a certain kind of effort to obtain effective voice communication. They were also discussed under various mobility scenarios to know the about working functionality of VoIP.

Khushboo et al., [3] compared the AODV, OLSR and TORA routing protocols, and its performances about voice services in MANET environment. For this, they used the OPNET simulator to identify the QoS factors of those protocols. Calduwel et al., [4] state that seamless data transfer is not possible, since all the nodes are changing dynamically. The refined hamming distance (REHDIS) considered two important parameters such as refined hamming distance and delay. It selected the best optimum path from various paths. It transferred the data quickly. Though REHDIS has its merits, it also has its weaknesses like maintenance of tables.

Sheeba et al., [5] stated the importance of Session Initiation Protocol (SIP) in Voice services and proposed adapted SIP for voice enhancement in MANETs. For SIP adaptation, they introduced new fields in the header packet format and maintained the sessions cautiously with acceptable robustness. Vijayakumar et al., [6] developed a new algorithm for saving energy in buffer management of voice application in MANETs configured topology. For this, they used distance and energy as the main factors and they were simulated using the NS-2 tool.

Calduwel et al., [7] proposed a new technique for increasing the admission of real-time users and along with that packet per second also increased. In order to achieve this, they used Analytic Hierarchy Process (AHP) which arranges the available users in an alphabetical order. It ultimately enhances the Quality of Service (QoS) in Mobile Networks. Malik et al., [8] prepared a selection scheme for identifying which codec suits to which networking environment and their performance levels. Finally, they confirmed that WiMAX G.729 is the best, G.711 suits to WLAN and GSM-FR is provided well for UMTS, etc. The QoS functional description of each codec was verified by using the OPNET simulator.

Mohamed et al., [9] analyzed the performance of voice codecs in WiMAX networks and proved that G.723 offers richer services than the other codecs in terms of delay and throughput along with the speech quality. It is estimated using Mean Opinion Score (MOS) method. OPNET simulator was used here for analyzing the results. K.Ramkumar et al., [10] developed a new algorithm for adaptively adjusting the buffer size. It allows the buffer to capture the packets at an appropriate time. The outcome of this scheme is to increase the speech quality of VoIP users.

Shweta et al., [11] implemented the mesh based topology for analyzing the capacity of various codecs. They used weighted cumulative ETT protocol for multi channel operations and proved that GSM based AMR yields good results when compared with the other codecs. Many stability oriented routing algorithms were discovered for discovering a stable route for transferring data packets via intermediate nodes [12,13]. Attention was given to discover a stable route that should flood only the minimum number of control packets.

Maja Sulovic et al., [14] developed an algorithm for selecting the best codec based on the network conditions. For this, bandwidth is taken as a major parameter and OPNET was used for simulation. Calduwel et al., [15] introduced priority based procedure for reducing the delay in the waiting time of the voice users. It also classifies the users as real-time and non real-time users for assigning high priority to the real-time users.

III. CBT-VOIP: A PROPOSED TECHNIQUE

In the proposed CBT-VoIP, the nodes are classified according to their codec type and this classification is done by considering the bandwidth rate. Proposed CBT-VoIP is elaborated as follows.

Step 1: Determine the Source and Destination for routing VoIP packet transmission.

Step 2: Identify the VoIP traffics with respect to packet sizes.

Step 3: Classify each node based on the codec types and its neighbours.

Step 4: Do the following:

If Packet Size is greater than the Available Bandwidth

Select the CODEC nodes having greater Bandwidth (G.711)

Else If the Packet Size is less than the Available Bandwidth

Select CODEC node’s having lesser Bandwidth (G.729)

Else If the Packet Size is equal to the Available Bandwidth

Select codec node’s path which has dissimilar codec types (AMR and others)

End If

Step 5: Repeat the steps 2 to 4 until all node codec types are classified

Step 6: Resume the data transfer

IV. RESULTS AND DISCUSSIONS

The above section explained CBT-VoIP of the proposed technique for VoIP. For example, source node ‘S’ has to identify the suitable path among the number of available paths. It is depicted by considering a network scenario as shown in Figure 1. There are 14 nodes that are considered in the network scenario. They are S, N1, N2, N3, N4, N5, N6, N7, N8, N9, N10, N11, N12 and D. Here, S denotes source, ‘D’ is a destination and the other nodes N1 to N12 are considered as intermediate nodes to route the packets. The number on the edges denotes the delay in milliseconds.

Network Scenario

In the above scenario, the initial path (i.e. S→N1→N2→N3→D) is selected based on maximum bandwidth and minimum hop count. If the selected path is congested, an alternate path is chosen based on more transmission delay and maximum hop count. According to the existing OLSR-VA technique, it yields the first preference to the path which does not involve in any previous data transmission and with less congested when compared to OLSR. Here, the alternate path is selected (i.e. S→N4→N5→N6→N7→D) since, it has the least number of VoIP calls. In another case, the selected path may get delayed due to less bandwidth and maximum delay. So, there is a need for an alternate path. The CBT-VoIP

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technique identifies the optimum path based on different types of CODEC in the path. In the existing OLSR-VA they did not consider about the type of codec. For example, if the type of node has AMR codec it carries only 4.75Kbps. Likewise, G.711 and G.729 carry 64Kbps and 8Kbps respectively. Assume that, as per OLSR-VA the selected path has a distinct AMR codec in all links. It gets delayed for transmitting VoIP packets. So, the proposed CBT-VoIP considers dissimilar codec types in order to deliver the VoIP packets with minimum transmission delay. Table 1 shows the different codec types with respect to bandwidth and packet size.

Table 1: Codec Types, Bandwidth and Packet Size [7]

Codec	Bandwidth	Packetization Delay	Compression Ratio	Total Packet Size	Packets Per Second (PPS)	Bandwidth Per Call
G.711 μ law	64Kbps	20ms	31.6%	206(160)	50	82.4 Kbps
G.711 A law	64Kbps	30ms	31.6%	206(160)	50	82.4 Kbps
G.711(a& μ)	64Kbps	40ms	31.6%	206(160)	50	82.4 Kbps
G.729a	8Kbps	20ms	76%	66(20)	50	26.4 Kbps
G.729b	8Kbps	30ms	76%	66(20)	50	26.4 Kbps
G.729c	8Kbps	40ms	76%	66(20)	50	26.4 Kbps
G.723	6.3Kbps	30ms	63.5%	76(30)	33	20.4 Kbps
AMR	4.75Kbps	20ms	73.4%	66(20)	30	15.8 Kbps
AMR	7.4Kbps	20ms	73.4%	66(20)	46	24.2 Kbps
AMR	12.2Kbps	20ms	73.4%	66(20)	76	40.3 Kbps
G.726	40Kbps	10ms	70.5%	66(20)	250	132 Kbps
G.726	32 Kbps	10ms	70.5%	66(20)	200	105.6Kbps
G.726	16 Kbps	10ms	70.5%	66(20)	100	52.8 Kbps
iLBC	15Kbps	20ms	48.8%	76(30)	62	37.7 Kbps
iLBC	13 Kbps	40ms	48.8%	76(30)	55	33.4 Kbps
EVRC	16 Kbps	50ms	61.7%	66(20)	100	52.8 Kbps

Figure 1 shows the MANETs topology with multiple paths between source and destination. According to the existing technique, the path S→N4→N5→N6→N7→D is chosen based on a less congested path when compared with S→N1→N2→N3→D path. Although these two paths are not optimal, the selected path gets delayed due to maximum transmission delay. Hence, the proposed CBT-VoIP chooses S→N11→N12→N7→D since it has a dissimilar Codec type of nodes as shown in Table 2.

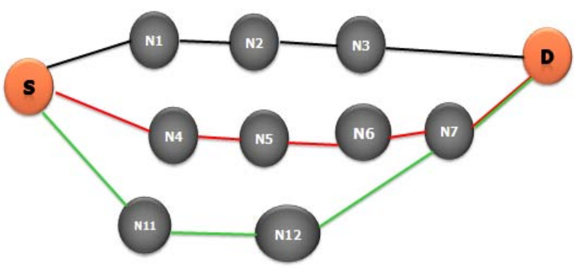

Figure 1: Available Paths in MANETs Topology

Table 2: Information about Available Paths, CODEC Types, and CODEC's Bandwidth

Packet Size (in kb)	OLSR-VA Delay (in ms)	CBT-VoIP Delay (in ms)
100	99.63	75.81
200	199.39	151.73
300	299.09	227.59

Table 3 shows the comparison of OLSR-VA and CBT-VoIP in terms of different data packet size with transmission delay. When 2MB data is sent, the OLSR-VA takes 20ms whereas proposed CBT-VoIP takes only10ms.

Table 3: Comparison of OLSR-VA and CBT-VoIP with respect to Data Size

Available Path(s)	CODEC Type(s)	CODEC's Bandwidth on Each Node (in kbps)
S→N1→N2 →N3→D	G.711, G.711, G.711 and G.711	64, 64, 64 and 64
S→N4→N5 →N6→N7→D	AMR_1, AMR_2, AMR_3 and AMR_4	1.8, 5.15, 6.7 and 10.2
S→N11→N12 →N7→D	G.711, G.729, G.726 and AMR_4	64, 8, 16 and 1.8

Figure 2 shows a graphical representation of OLSR-VA and CBT-VoIP along with different data packet size.

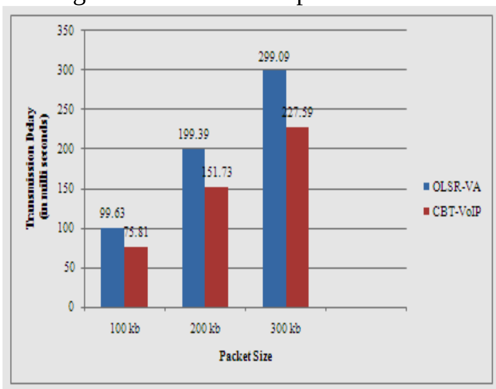

Figure 2: OLSR-VA Vs CBT-VoIP

V. CONCLUSION

In this paper, the CBT-VoIP technique is proposed for providing VoIP service in MANETs. The CBT-VoIP works based on codec with respect to different bandwidth levels. The result shows that the proposed technique reduces the VoIP transmission delay better than the existing technique. In future work, different parameters would have been considered for checking the voice quality to provide seamless communication in order to evaluate throughput, jitter, etc., The demerit of this paper is that did not consider the link failure issue during voice transmission.

VI. ACKNOWLEDGEMENT

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