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Performance Based Comparsion of Various VOIP CODECS over MR-MC WMNs

Shweta
Dept. of Computer Science and Engg.
Swami Devi Dyal Group of Professional Institutions
Barwala, Haryana, India

Kamal Tanwar
Dept. of Computer Science and Engg.,
Swami Devi Dyal Group of Professional Institutions
Barwala, Haryana, India

Abstract-

A VoIP system consists of an encoder-decoder pair and an IP transport network. The choice of voice codec is important because it has to fit the particularities of the transport network. compressing voice signals while keeping the quality perceived by users at acceptable levels represents a daunting challenge. The methods that have been proposed for the compression of audio signals, which are referred to as voice codecs. Voice codecs are the algorithms that enable the system to carry analog voice over digital lines. There are several codecs, varying in complexity, bandwidth needed and voice quality. The more bandwidth a codec requires, normally the better voice quality is. In this paper, we investigate the performance of different voip codecs, which are G.711, G.723.1,G.729 and GSM.AMR over multi-radio multi-channel wireless mesh network using various pause time and WCETT mesh routing protocol using ns2 simulator.

Keywords- WMN, WLAN, IEEE, ETX

I. WIRELESS NETWORKS

Wireless Communication is an application of science and technology that has come to be vital for modern existence. Wireless networking is an emerging technology that allows users to access information and services electronically, regardless of their geographic position. From the early radio and telephone to current devices such as mobile phones and laptops, accessing the global network has become the most essential and indispensable part of our lifestyle. Wireless communication is an ever-developing field, and the future holds many possibilities in this area. Now users are able to move freely and still have seamless, reliable and high-speed network connectivity. Portable computers and hand-held devices will do for data communication what cellular phones are now doing for voice communication. Traditional network mobility focused on roaming, which is characterized by hosts connecting to the fixed infrastructure internet at locations other than their well known home network address. Hosts can connect directly to the fixed infrastructure on a visited subnet through a wireless link or a dial-up line, these so called traditional (or fixed-infrastructure mobile) networks raise issues such as address management, but do not require significant, changes to core network functions such as routing.

1.1 Differences between Ad Hoc Wireless Networks and Wireless Mesh Networks

- If we consider mobility, many ad hoc wireless networks are high mobile networks and their network topologies change dynamically. On the other hand, WMNs have relatively static mobility where relay nodes are mostly fixed, so its network mobility is low compared to ad-hoc wireless networks.
- One bigger constraint in ad-hoc wireless network is energy because nodes in these networks may not have a constant power source, while WMNs nodes have better energy storage and power source due to the static topology, formed by fixed relay nodes.
- Deployment may be easy in ad-hoc networks, while in WMNs we may require planning.
- For the application scenario, most ad-hoc wireless networks are temporary, and WMNs are mostly semi-permanent or permanent. In addition, WMNs can be used for both military and civilian applications; an example is the provision of low cost Internet services in public places.

1.2. Wireless Mesh Networks

Wireless Mesh Network (WMN) [1] is a highly promising technology and it plays an important role in the next generation wireless mobile network. WMNs have emerged as important architectures for the future wireless communications. A wireless mesh network (WMN) is a communications network made up of radio nodes organized in a mesh topology. Wireless mesh networks often consist of mesh clients, mesh routers and gateways. A mesh network is reliable and offers redundancy. When one node can no longer operate, the rest of the nodes can still communicate with each other, directly or through one or more intermediate nodes. The animation below illustrates how wireless mesh networks can self-form and self-heal. Wireless mesh networks can be implemented with various wireless technology including 802.11, 802.15, 802.16, cellular technologies or combinations of more than one type. Wireless mesh architecture is a first step towards providing cost effective and dynamic high-bandwidth networks over a specific coverage area. Wireless mesh architectures infrastructure is, in effect, a router network minus the cabling between nodes.

It's built of peer radio devices that don't have to be cabled to a wired port like traditional WLAN access points (AP) do. Mesh architecture sustains signal strength by breaking long distances into a series of shorter hops. Intermediate nodes not only boost the signal, but cooperatively make forwarding decisions based on their knowledge of the network, i.e. perform routing. This type of infrastructure can be decentralized (with no central server) or centrally managed (with a central server), both are relatively inexpensive, and very reliable and resilient, as each node needs only transmit as far as the next node. Nodes act as routers to transmit data from nearby nodes to peers that are too far away to reach in a single hop, resulting in a network that can span larger distances.

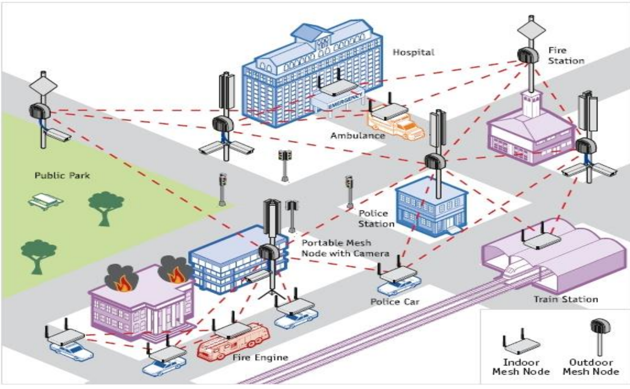

Fig 1: Wireless Mesh Network

The topology of a mesh network is also reliable, as each node is connected to several other nodes. If one node drops out of the network, due to hardware failure or any other reason, its neighbors can quickly find another route using a routing protocol.

1.3. Wireless Mesh Network Architecture

WMNs consist of two types of nodes: Mesh Routers and Mesh Clients.

- Wireless mesh router [1] contains additional routing functions to support mesh networking. To further improve the flexibility of mesh networking, a mesh router is usually equipped with multiple wireless interfaces built on either the same or different wireless access technologies .Compared with a conventional wireless router, a wireless mesh router can achieve the same coverage with much lower transmission power through multi-hop communications.In spite of all these differences, Moreover, the gateway/bridge functionalities in mesh routers enable the integration of WMNs with various existing wireless networks such as cellular, wireless sensor, wireless-fidelity (Wi-Fi) worldwide inter-operability for microwave access (WiMAX)[5] .
- Mesh Clients are the Conventional nodes (e.g., desktops, laptops, PDAs, Pocket PCs, phones, etc.) equipped with wireless network interface cards (NICs), and can connect directly to wireless mesh routers.

The architecture of WMNs can be classified into three main groups based on the functionality of the nodes:

1.3.1 Infrastructure/Backbone Wireless Mesh Networks

This type of WMNs includes mesh routers forming an infrastructure for clients that connect to them. The WMN infrastructure backbone can be built using various types of radio technologies, in addition to the mostly used IEEE 802.11 technologies. The mesh routers form a mesh of self-configuring, self-healing links among themselves. With gateway functionality, mesh routers can be connected to the Internet. This approach, also referred to as infrastructure meshing, provides backbone for conventional clients and enables integration of WMNs with existing wireless networks, through gateway/bridge functionalities in mesh routers. Conventional clients with Ethernet interface can be connected to mesh routers via Ethernet links. The architecture is shown in Fig 1 [1], where dash and solid lines indicate wireless and wired links, respectively.

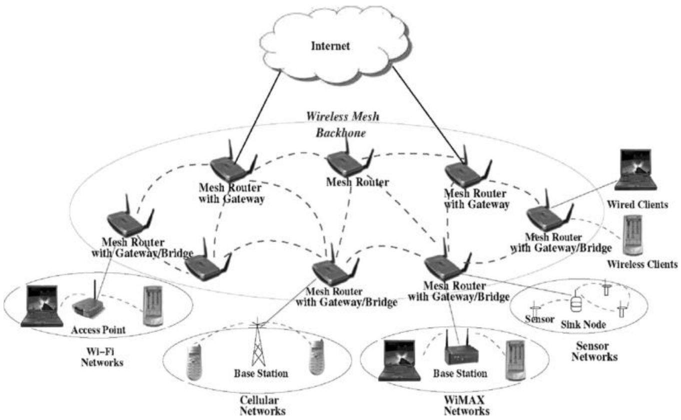

Fig 1: Infrastructure/backbone WMNs

1.3.2 Client Wireless Mesh Networks

In this type of Client WMN, nodes constitute peer-to-peer network, and perform routing and configuration functionalities as well as provide end-user applications to customers, mesh routers are not required. They support Multi-hop routing. Client nodes have to perform additional functions such as routing and self-configuration. The basic architecture is shown in Fig 2 [1].

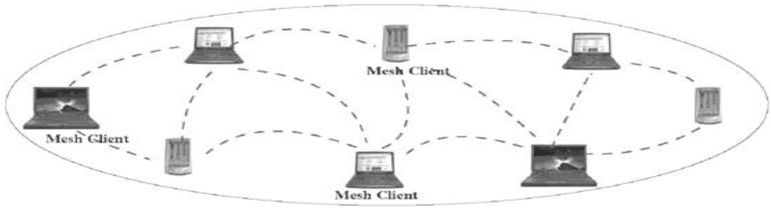

Fig 2: Client WMNs

1.3.3 Hybrid Wireless Mesh Networks

This architecture shown in Fig. 3 [1] is the combination of infrastructure and client meshing. Mesh clients can access the network through mesh routers as well as directly meshing with other mesh clients. While the infrastructure provides connectivity to other networks such as the Internet, Wi-Fi, WiMAX, cellular, and sensor networks and the routing capabilities of clients provide improved connectivity and coverage inside the WMN.

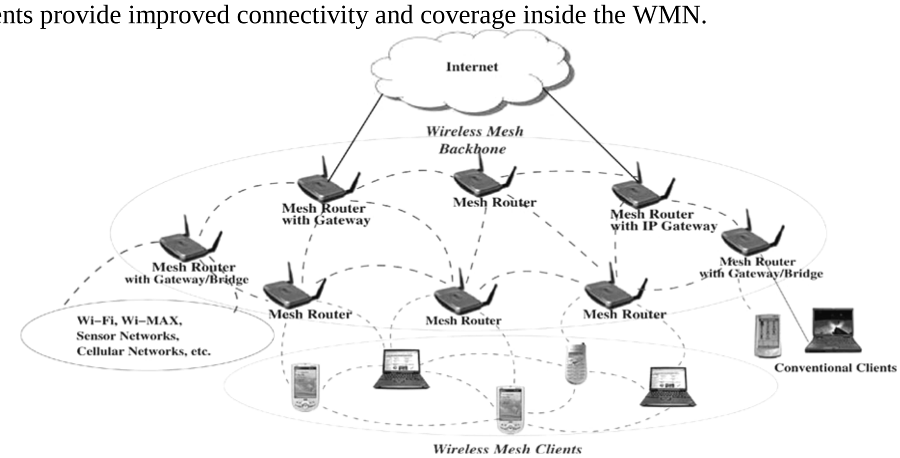

Fig. 3: Hybrid WMNs.

1.4 Routing and Channel Assignment protocols

Unlike ad hoc wireless networks, most of the nodes in WMNs are stationary and thus dynamic topology changes are less of a concern. Also, wireless nodes in WMNs are mostly access points and Internet gateways and thus are not subject to energy constraints. As a result, the focus is shifted from maintaining network connectivity in an energy efficient manner to finding high-throughput routes between nodes, so as to provide users with the maximal end-to-end throughput. In particular, because multiple flows initiated by multiple nodes may engage in transmission at the same time, how to locate routes that give the minimal possible interference is a major issue.The issue of locating interference free routes can roughly divided into two complimentary approaches.

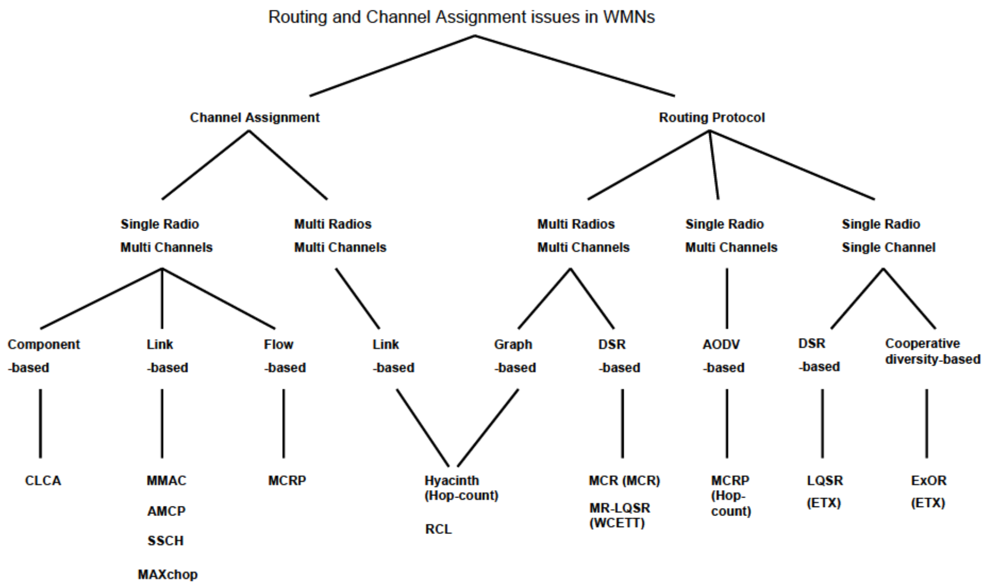

Fig. 4: A taxonomy of routing and channel assignment protocols for WMNs.

Expected Transmission Count (ETX)

This metric calculates the expected number of transmissions (including retransmissions) needed to send a frame over a link, by measuring the forward and reverse delivery ratios between a pair of neighboring nodes [9]. To measure the delivery ratios, each node periodically broadcasts a dedicated link probe packet of a fixed size. The probe packet contains the number of probes received from each neighboring node during the last period. Based on these probes, a node can calculate the delivery ratio of probes on the link to and from each of its neighbours. The expected number of transmissions is then calculated as

$$ETX = 1/df \times dr$$

where df and dr are the forward and reverse delivery ratio, respectively. With ETX as the route metric, the routing protocol can locate routes with the least expected number of transmissions. Note that the effects of link loss ratios and their asymmetry in the two directions of each link on a path are explicitly considered in the EXT measure. Measurements on wireless testbeds [9,10] show that, for the source-destination pairs that are with two or more hops, use of ETX as the route metric renders routes with throughput significantly higher than use of the minimum hop count.

Expected Transmission Time (ETT)

One major drawback of ETX is that it may not be able to identify high-throughput routes, in the case of multi-radio, multi-rate wireless networks. This is because ETX only considers the packet loss rate on a link but not its bandwidth. ETT has thus been proposed to improve the performance of ETX in multi-radio wireless networks that support different data rates. Specifically, ETT includes the bandwidth of a link in its computation [9],

$$ETT = ETX \times S/B$$

where S and B denote the size of the packet and the bandwidth of the link, respectively. ETT considers the actual time incurred in using the channel. In order to measure the bandwidth B of each link, a node sends two probe packets of different sizes to each of its neighbours every minute. The receiver node measures the difference between the instants of receiving the packets, and forwards the information to the sender. The bandwidth is then estimated by the sender node by dividing the larger packet size by the minimum of 10 consecutive measurements.

1.7.3.3 Weighted Cumulative ETT (WCETT)

What ETX and ETT have not explicitly considered is the intra-flow interference. WCETT was proposed [9] to reduce the number of nodes on the path of a flow that transmit on the same channel. Specifically, let Xc be defined as the number of times channel c is used along a path. Then WCETT for a path is defined as the weighted sum of the cumulative expected transmission time and the maximal value of X_c among all channels, i.e.,

$$WCETT=(1-\beta)\ \sum_{i=0}^{n} ETT_i\ +\ \beta\ \sum_{1}^{C} MAX_c\ X_c$$

where β (0 <β<1) is a tunable parameter. Moreover, the two terms also represent a trade-off between achieving low delay and high throughput. Reducing the first term reduces the delay, while reducing the second term increases the achievable link throughput. The tunable parameter β is used to adjust the relative importance of the two objectives. Modified Expected Number of Transmissions (mETX) and Effective Number of Transmissions (ENT) Another issue which ETX does not consider is the effect of short-term channel variation, i.e., ETX takes only the average channel behavior into account for the route decision. In order to capture the time-varying property of a wireless channel, the metrics mETX and ENT were proposed in [11] which took into account both the average and the standard deviation of the observed channel loss rates. Specifically, mETX is expressed as

$$mETX = exp\ (\mu_{\sum} + 1/2\alpha_{\sum}^{2})$$

where $\mu_{\sum}$ and $\alpha_{\sum}^{2}$ are the average and variability of the channel bit error probability. In some sense, mETX incorporates the impact of physical layer variability in the design of routing metrics. On the other hand, when the problem of maximizing aggregate throughput with the packet loss rate constraint is considered, mETX may not be sufficient since the links which mETX selects may achieve the maximum link-layer throughput but incur high loss rates at the same time. The ENT metric is devised to meet both objectives.

II. RELATED WORK

According to [1], the authors considers the issues with the MR-MC architecture, existing communication protocols, ranging from routing, MAC, and physical layers, need to be revised and enhanced. In physical layer, techniques mainly focus on three research areas: enhance transmission rate, enhancing error resilience capacity, and increasing re configurability and software controllability of radios. In order to improve the capacity of wireless networks, many high-speed physical techniques, such as OFDM, UWB, and MIMO, have been discovered.

According to [2], In MAC layer, depending on which network node take responsibility of the coordination of medium access, MAC can be categorized into two major types: centralized MAC and distributed MAC. In WMNs, due to its distributed nature, distributed MAC is preferred. The MAC protocols for WMNs can be classified into two types: single-channel and multi-channel MAC protocols .

According to [3], To select a routing path in WMNs, the routing algorithm requires to consider network topology, and the routing path selection is to twist with resource allocation, interference reduction and rate adaptation in multiple hops. An MR-MC routing protocol not only require to select a path between different nodes , but it also require to select the most effective channel or radio node on the path.

According to [4], TC is considered as an additional protocol layer between the routing and MAC layer in the protocol stack . The routing layer is required for finding and maintaining the paths between source/destination pairs in the network, and for routing packets toward the destination at the intermediate nodes on the route. Two-way interactions may occur between the routing protocol and TC protocol. The TC protocol, which create and maintains the list of the all immediate neighboring nodes, can send a route update in case it detects that the neigh borers list is considerably changed, and hence leading to a faster response time to topology changes and to decrease packet-lost rate.
According to [5], Converged IP networks seek to incorporate voice, data, and video on the same infrastructure. However, the integration of all types of traffic onto a single IP network has several advantages as well as disadvantages. While reducing cost and increasing mobility and functionality, VoIP may lead to reliability concerns, degraded voice quality, incompatibility, and end-user complaints due to changing network characteristics. The authors have described the various codecs in VoIP implementation and analysed three commonly used narrow band codecs namely G.711, G.723 and G.729 using peer-to-peer network scenario. It can be analysed from the simulation results that G.711 is an ideal solution for PSTN networks with PCM scheme. G.723 is used for voice and video conferencing however provides lower voice quality. Music or tones such as DTMF cannot be transmitted reliably with G.723 codec. G.729 is mostly used in VoIP applications for its low bandwidth requirement.
According to [6], Internet Protocol is used as a communication platform for Voice over Internet Protocol (VoIP). Before the transmission of VoIP it is required to do signal processing for reliable & efficient performance over the computer and the communication platform. The authors have analyzed the performance of the narrowband wireless VoIP system under varying packet loss for different modulations schemes such as DBPSK, DQPSK, and QAM64. The VoIP simulations are conducted for G.729A and AMR-NB speech coders at different packet loss rates. The forward error correction scheme was proposed for narrowband VoIP coders based wireless VoIP system. The performance of the system was analyzed wireless fading environment with DQPSK, DBPSK & QAM64 modulation schemes. The authors found from the VoIP simulation results that DBPSK modulation worked well with proposed packet loss concealment scheme, since Mean Opinion Scores (MOS) scores were best in case of DBPSK as compared to DQPSK and QAM64 for both narrowband coders.
According to [7], use of codecs appropriately is very important in the implementation of VoIP to generate maximum QoS value. The result shows a selection of G.729A codec in a simulation gives a significant result for the performance of VoIP that codec G.729A has acceptable MOS value and less deviation of received to transmit packet as compared to G.711 and G.723.1 also average delay like end to end delay and Voice jitter is lesser in codec G.729A as compared to the other two referenced codecs.
According to [8], The algorithm of ITU-T G.711.1, a wideband scalable codec of G.711 proposed by ETRI, France Telecom, Huawei Technologies, Voice Age and NTT, was enhanced. G.711.1 is designed to achieve a very short delay and low complexity. The bit stream has an embedded structure where the core layer is generated by a G.711 compatible codec utilized with a noise shaping feedback. On top of the core layer, there are two enhancement layers: a lower band enhancement layer for the refinement signal encoded with a dynamic bit-allocation, and another one for higher band encoded with an interleaved CSVQ in MDCT domain. The emphasis in the codec design was on complexity.
According to [9], investigate the performances of the most common VoIP codecs, which are G.711, G.723.1 and G.729 over a WiMAX network using various service classes and NOAH as a transport protocol. Voice Over Internet Protocol is a promising new technology which provides access to voice communication over internet protocol based network, it becomes an alternative to public switched telephone networks due to its capability of transmission of voice as packets over IP networks. Therefore VoIP is largely intolerant of delay and hence it needs a high priority transmission protocol. To analyse the QoS parameters, the popular network simulator ns-2 was used. Various parameters that determine QoS of real life usage scenarios and traffic flows of applications is analysed. The objective is to compare different types of service classes with respect to the QoS parameters, such as, throughput, average jitter and average delay.
According to [10], authors proposed a Next generation networks with multiple technologies offer different multimedia services to the user. Next Generation Wireless Networks (NGWNs) focus on convergence of different Radio Access Technologies (RATs) providing good Quality of Service (QoS) for applications such as Voice over IP traffic (VoIP) and video streaming. To meet the demand of providing high-quality of VoIP at any time and from anywhere, it is imperative to design suitable QoS model. Simulation results over OPNET simulator show that WiMAX outscores the UMTS with a sufficient margin, and is the better technology to support VoIP applications compared with UMTS. It also provides the luxury of utilizing the best available technology for the required service to a user, companies and business organizations.
According to [11], VoIP performance is evaluated in terms of both throughput and jitter through on-the-road measurements, investigating also the effects of inter-vehicle distance and speed on VoIP performance. Simulation Results of their study show that VoIP performance is quite poor, especially due to an excessively large number of lost packets under all proposed scenarios.

III. PROPOSED METHODOLGY

Multiradio Multichannel Wireless Mesh Network uses multiple network interfaces per node allows simultaneous transmission and reception on different interfaces tuned to different channels, which can substantially improve multihop throughput. A VoIP system consists of an encoder-decoder pair and an IP transport network. The choice of voice codec is important because it has to fit the particularities of the transport network. compressing voice signals while keeping the quality perceived by users at acceptable levels represents a daunting challenge. The main objective of this paper is to experimentally analyze the impact of different voice codecs (compression of sudio signals) on wireless multihop multichannel mesh network.

3.1 Simulation Methodoly

In this paper, we have used a detail simulation model based on NS-2 has been used in the evaluation, and in order to perfectly evaluate the effect of different voice codecs while WCETT routing protocol is used under different pause time scenarios over MRMC WMN. The Wireless mesh network comprising of 30 mobile nodes is constructed in NS-2 simulator with the use of TCL Script in the topological boundary area of 1500 m*1000 m. Antenna chosen is Omni Antenna: Omni direction antenna is antenna which radiates radio power uniformly in all direction in one plane. UDP agent is as a transport layer agent. With UDP agents CBR traffic is attached. Propagation model is two ray ground. This model is mathematical formulation for the characterization of radio wave propagation as a function of frequency, distance and other conditions. The simulation runs for 150 Seconds. Table 1 shows the important simulation parameters used in the simulation process.

Table 1:Important Simulation Parameters

Parameter	Value
Simulation area	1500m x 1000m
Antenna	Omni antenna
No. of nodes	30
No. of interfaces/ node	2
No. of channels/ node	5
Voice codecs	G.711, G.723.1, G.729A and GSM.AMR
Max queue length	50
Routing protocol	WCETT
Transport Layer	UDP
Channel Assignment Strategies	Common Channel assignment

- **Effect of VOIP codecs on Throughput**

  Throughput is the ratio of total number of delivered or received data packets to the total duration of simulation time. Figure 5, shows the impact of narrow band voice codecs on the throughput when the pause time is varied over MRMC MWN. Simulation results shows WCETT routing protocol that is specifically designed for WMN, gives better performance for G.729A and GSM.AMR. Simulation graph shows that as pause time of mesh client nodes increase the performance metric throughput decreases

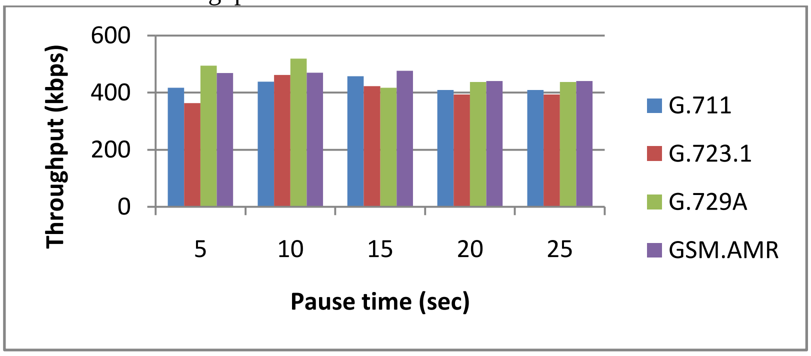

Fig 5: Throughput vs pause time.

- **Effect of VOIP codecs on PDR**

  PDR also known as the ratio of the data packets delivered to the destinations to those generated by the CBR sources. Figure 6 shows the effect of audio compression schemes on packet delivery ratio against WCETT routing protocol when the pause time is varied. Simulation results shows that for higer traffic load, WCETT routing protocol gives better performace for GSM.AMR.

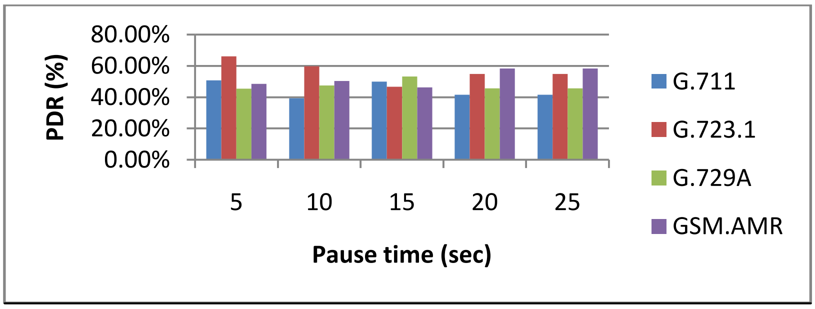

Fig 6: PDF vs pause time.

- **Effect of VOIP codecs on Average End to End Delay**

  Average End to End delay is the average time taken by a data packet to reach from source node to destination node. It is ratio of total delay to the number of packets received. Figure 7 shows the Average delay for two different channel assignment strategies uner WCETT routing protocol when the taffic load is varied. Simulation results shows that audio G.723.1 and G.729A compression scheme takes longer time than G.711 and GSM.AMR schemes for MRMC wireless mesh network.

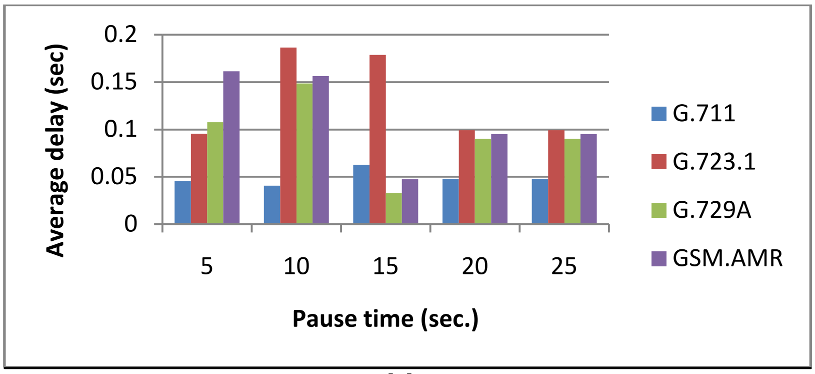

Fig 7: Average delay vs pause time.

- **Effect of VOIP codecs on Routing Overhead**

  The sum of all transmissions of routing packets sent during the simulation. For packets transmitted over multiple hops, each transmission over one hop, counts as one transmission. Routing overhead is important to compare the scalability of the routing protocols.. Figure 8 shows the routing overhead for different audio compression strategies under WCETT routing protocol when the pause time is varied. Simulation results shows that for higher pause time, WCETT routing protocol gives better performace for GSM.AMR.

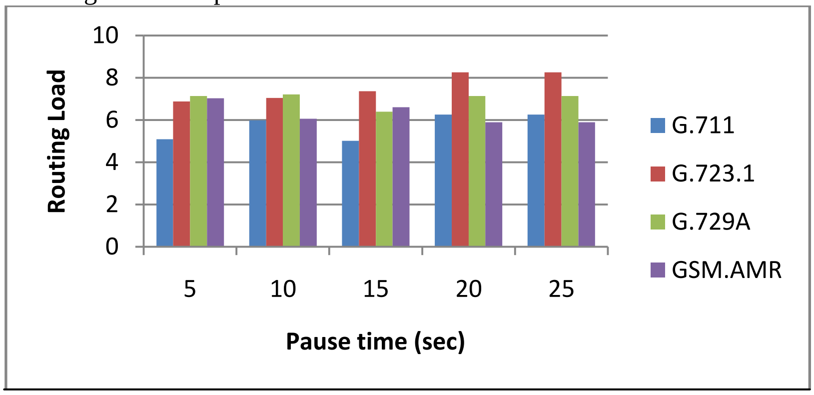

Fig 8: Routing overhead vs pause time.

IV. CONCLUSION

In this paper, the effect of four different VOIP audio compression strategies namely G.711, G.723.1, G.729A and GSM.AMR is examined on to evaluate the performance Weighted Cumulative ETT (WCETT) routing protocol under different mesh clients pause time scenarios over multiradio multichannel wireless mesh network. we have identified the key challenges associated with VOIP is to choice efficient voice codec for radio interfaces in a multi-radio wireless mesh network. Performance in multihop wireless networks is known to degrade with the number of hops for UDP traffic. From simulation results, it is observed that VOIP audio compression scheme namely GSM.AMR has best all-round performance under all pause time scenarios considered. ITU G.711 codec in terms of average end to end delay and routing overhead performs almost similar to GSM.AMR codec but for packet delivery ratio ITU G.723.1 performs better than GSM.AMR over MRMC WMNs.

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