International Journal of Advanced Information Science and Technology (IJAIST) ISSN: 2319:2682
Vol.3, No.10, October 2014 DOI:10.15693/ijaist/2014.v3i10.109-116

QoS Evaluation of VOIP Codecs in Mobile Adhoc Network

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Abstract

Voice over Internet protocol has been proved to be an emerging popular technology and is an essential requirement in today's era as it provides real time communication and good services. Mobile Adhoc Network(MANET) provides a suitable platform for the deployment of VOIP service in many application scenarios. Combining these technology together, in this paper we estimated the performance of various VOIP codecs encapsulating Wi-Fi connectivity in different scenarios (sparse and dense) over MANET. The performance of any VoIP application is dominantly governed by the choice of a suitable voice codec. Moreover performance of several routing protocols i.e proactive(OSPF), reactive(DSR) and hybrid (ZRP) protocols are also analyzed for VOIP applications in order to determine their effect on various QoS metrics.

In this paper, the performance of seven voice codec i.e. G.711, G.723.1ar5.3, G.723.1ar6.3, G.726ar24, G.726ar32, G.728ar 16 and G.729are analyzed with some QoS metrics such as average MOS, average energy consumption, average throughput, average transmission delay and signal received with error using H.323 signaling protocol in order to determine most efficient network specific voice codec. Qualnet 6.1 simulator is used for the study of voice codec in order to determine their effect on QoS .

Keywords

MANET, VoIP, Voice codec, OSPF,DSR,ZRP

INTRODUCTION

The evolution of the Internet has facilitated the multimedia communications as the most prioritized communication means in today's world. Adhoc networks can be attached to the Internet, thereby integrate many different devices and while maintaining the connectivity with the rest of the world enables users in accomplishing tasks, accessing the communication services at anytime from anywhere from any device. As a consequence Mobile Adhoc network (MANET) is a significant part of mobile computing network which is infrastructure less network. In this mobile nodes are free to move and communicate with each other without the use of any centralized authority or access point. Because of this self-administering & self-organizing capabilities mobile adhoc network can be deployed with minimum user intervention and is highly suited for situations where no fixed nor too expensive infrastructure is required. Its intrinsic flexibility, auto-configuration makes it suitable for potential applications like for Tactical networks, Emergency services ,Search and rescue operations, Disaster recovery, Commercial and civilian, Vehicular services, home/office wireless networking, personal networks, wireless sensor network, data networks, device networks, etc. With many applications there are still some design issues and challenges to overcome inherent to wireless communication such as limited physical security, time varying channels, lower reliability, rapidly changing topology, battery constraint[1].VOIP is one of the significant application of MANET that is concerned for voice transmission over IP like tele-emergency system. Due to its gradually increasing performance efficiency the evolution of Voice over IP (VOIP) has been proved to be the most advanced and popular technology[2]. It is a technology that facilitates voice transmission over IP network such as Internet which benefits in obviating the toll charges by ordinary telephone services. It packetized the digital information and transmitted over packet switched network instead of circuit switched network.

The performance of any VoIP application is dominantly governed by the choice of a suitable voice codec. A suitable voice codec is required for providing better quality of service on VOIP networks which ensures the quality of voice communication with limited end to end delay and low packet loss rate. Voice digitization(analog/digital signal conversion) and compression are the primary functions of codec [3].

In this paper the performance of several voice codecs that are used in VOIP applications are analyzed along with the Proactive, Reactive and Hybrid routing protocols. The main objective of this paper is evaluate the performance of several voice codecs those are used in VOIP transmission over MANET. The main contributions of this work are as follows-

1.To study the performance of seven different voice codecs through different QoS metrics like like Mean opinion score(MOS), Throughput, Transmission delay, Energy Consumption, Signal received with error.

2. To design a model for MANET and encapsulating 802.11 connectivity using H.323 signaling protocol [4] for achieving high data rates and accurate simulations.

3. To evaluate the performance of proactive, reactive and hybrid routing protocols for VOIP applications.

4. Different voice codecs are analyzed under the effect of sparse and dense scenario environment using OSPF, DSR and ZRP routing protocol to determine their effect on various QoS metrics.

4. The performance of the simulated model has been evaluated using Qualnet 6.1 simulator[5].

II. RELATED WORK

In this section numerous research work are described which has been done for the performance evaluation of VOIP applications. El Brak et al[6] has taken urban scenario for the performance evaluation of VOIP applications over VANET taking only urban scenario. S. Alshomrani et al[7]have discussed the evaluation of QoS of only three codecs over WiMAX under small scale scenario.Adhicandra [8] discussed about the data transferring andtelecommunication feature in the WiMAX network.Tucker [9] found the numerous factors that affect the performance of the network and also find out the ways to deals with them. Martelli et. al. [10] measured the VoIP performance over IEEE 802.11pin vehicular adhoc network to measure its performance.H.Zhang et al[11] has analyzed the Quality of voice streams over MANET through two QoS metrics i.e delay and loss but scalability issue was not taken into account.

The main goal of this research work is to analyze the performance of VOIP in MANET through QoS metrics. Different voice codecs are simulated under the effect of sparse and dense scenarios through both network level (such as loses) and user lever (MOS) metrics.

III. EVALUATION FRAMEWORK AND SYSTEM DESCRIPTION-

This section provides a brief overview of evaluation model, VOIP implementation and routing protocol.

A. Simulation Methodology- The following evaluation model has been proposed to evaluate the performance of different codecs of VoIP with Wi-Fi over MANET.

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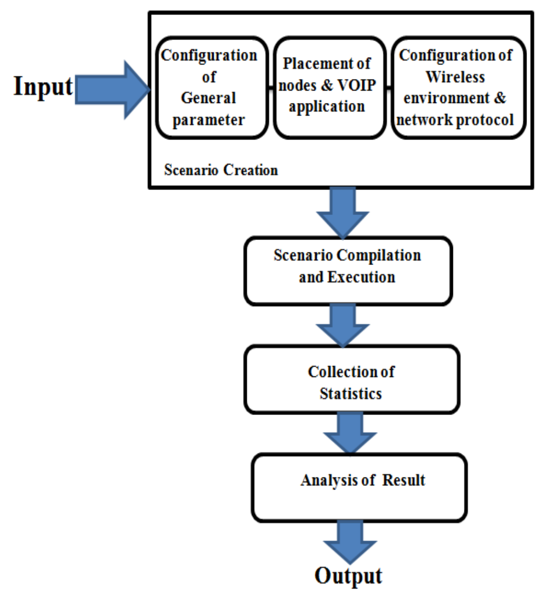

Figure 1

Step1- First step involves the creation of scenario which encapsulates the following stages-

a. Configuration of general parameters- This includes configuration of parameters like simulation time, terrain area and statistical & tracking property in which battery model is set.
b. Placement of nodes with mobility and subnet-It involves placement of nodes(set mobility according to the need), wireless subnets and connecting them to mobile nodes through links. Then VOIP applications are placed from node to node according to the requirement for transferring the multimedia data.

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c. Configuration of wireless environment and network protocol- In this stage properties of wireless subnet are configured from Table View tab of simulator in which PHY, MAC, NETWORK and Routing protocol are set. In Physical layer, network is set to 802.11n radio and Battery model is set to Generic.MAC layer is set to 802.11e and Configure Parameter 802.11n is set to Yes. Then in Network layer Routing Protocol OSPF,DSR,ZRP are set according to proactive, reactive and hybrid routing protocol.

Step2- In the Second step nodes are selected and their Mobility and Placement property is set to Random way point and Battery model is set to Linear via statistics and tracking property. Then VOIP nodes are selected and set to H.323 as multimedia signaling protocol in Application layer.

Step3- This involves the scenario compilation and execution which encapsulates real time and execution time. This turns the simulator from design mode to visualization mode.

Step4- After the executing the scenario statistics of various metrics are collected from Analyzer which are basically the simulation result according to the layer properties.

Step 4: The last step is the result analysis in which results of different statistics files with varying routing protocol are evaluated and concludes the output.Analyzer tool of Qualnet is helpful in comparing the result of different seven voice codecs with varying routing protocol in a particular time with values.Results with varying metric values are compared that concludes the best voice codec.

B. Voice Codec-Speech coding and speech compression are the main functions of voice codec in which voice is converted from analog signal to digital signal in order to produce the lowest bit rate stream. They can be classified as (1) Vocoders (2) waveform codecs (3) hybrid codecs .Vocoders requires limited bandwidth of only few Kbit/s due to which speech quality produced is not v good. Whereas Waveform codecs produces excellent speech quality and higher bandwidth requirement than that of vocoders. Eg- G.711, G.722,G.726 .On the other hand Hybrid codecs lie in somewhere in between the vocoders and waveform codecs regarding the bandwidth and speech quality achievement. G.723.1,G.729a etc. Quality of the voice depends mostly on the performance of the each voice codecs or vocoders and their respective performance metrics suchas throughput, delay, mean opinion score(MOS),energy consumption and signal received with error.

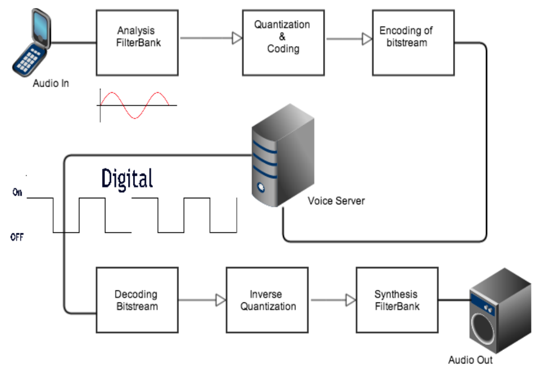


Figure 2

C. Routing Protocol- Routing Protocol establishes a route between the source and the destination without creating computational burden on the power constrained devices. In ad hoc network this process is done mainly by Proactive, reactive and hybrid routing protocol. Proactive routing protocol maintains up-to-date route to all other nodes at all times which produces a constant base load of the network to keep the routing tables of all nodes up to date like Fisheye state routing (FSR) ,Open shortest path first(OSPF).Reactive protocol discover route on demand basis by sending route request packet to destination node. This in-cures additional delay and high flooding of request message leads to high latency time. Example of such protocols are DSR(Dynamic State Routing), Ad-hoc on demand vector distance vector (AODV).On the other hand hybrid routing protocols combines the feature of both proactive and reactive routing protocols. Examples include Temporary Ordered Routing Algorithm (TORA), Zone Routing Protocol (ZRP).

IV.SIMULATION AND RESULT ANALYSIS-

A. Simulation Setup

We simulated two distinct scenarios (i) sparse scenario (ii) dense scenario to transmit audio file from source to destination and applied them over MANET architecture to analyze the voice codec performance by increasing the mobility space and node numbers.

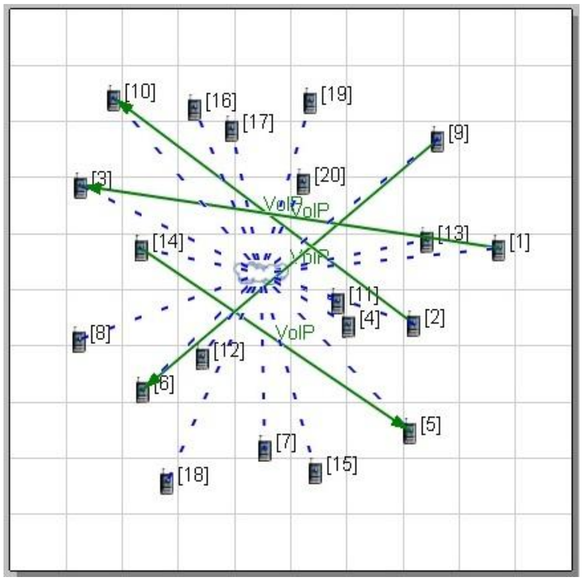


Figure 3

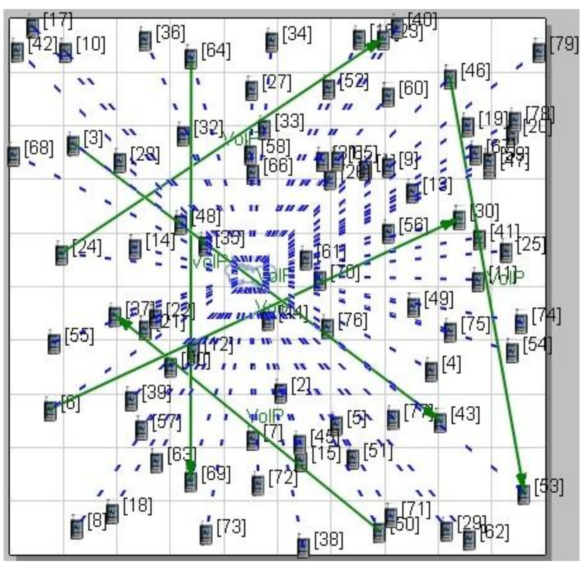


Figure 4

In consonance with the scenarios, we perform following activities to present our work.

Sparse Scenario Activity- The sparse scenario is illustrated in figure3. The 1000 x 1000 m^2 area is taken for creating the scenario in which 20 nodes are distributed .All the nodes are connected through wireless subnet with their node configuration and general properties set. The VoIP application are taken as traffic generator according to the requirement for transferring the multimedia data. We have taken three routing protocols to analyze the above with each category respectively, which correspondingly contains (OSPF), (DSR) and Zone Routing Protocol (ZRP).

Dense Scenario Activity-Dense network is illustrated in figure 4. The same area 1000 x 1000 m2 is taken for creating the scenario in which 80 nodes are distributed each connected with a group of nodes. VoIP is applied as traffic generator. The rest of the work is same as illustrated in section 3.

The simulation parameters which have been used in this paper, are tabulated as follows-

Table 1: Simulation Parameters

PARAMETER	DISCRIPTION
Simulation Tool	Qualnet 6.1
Simulation Time	300 sec
Bandwidth	20MHz
Transmission Power	20dBm
Antenna	Omni Direction
Physical Layer Protocol	802.11n
MAC Protocol	802.11e
Routing Protocols	OSPF,DSR, ZRP
Traffic Generator	CBR

Seven VoIP codecs are compared here: G.711, G.723.1ar5.3, G.723.1ar6.3, G.726ar24, G.726ar32, G.728ar 16 and G.729. IP telephony sessions are simulated by VOIP. We have taken VoIP traffic generator with 20 second average talking time and packetization of 20 millisecond interval. QualNet simulator tool has been used to estimate the performance and quality of audio codecs by using three different routing protocols OLSR,DSR & ZRP. By transmitting the multimedia data one by one through these protocols over both the sparse and dense scenario we concluded the best routing protocol for the optimal voice codec .After running the scenarios, network information and statistics of various metrics are collected from analyzer for the comparison analysis.

B. Result Analysis

The performance of several voice codecs those are used in VOIP transmission over MANET are analyzed through different Qos metrics. The result is divided according to sparse & dense condition of the network. So, we discuss both results one by one.

Sparse Scenario Result:

1. Mean Opinion Score (MOS): ITU-T P800 defines MOS as a subjective metric which needs sufficient number of listeners to rate the voice quality. It is measured by means of a score which scales from 1.0 (poor) to 5.0(best). It describes the human perception about QoS.So, high value of MOS is considered for the network and MOS below 3.6 does not results the good call quality [24].

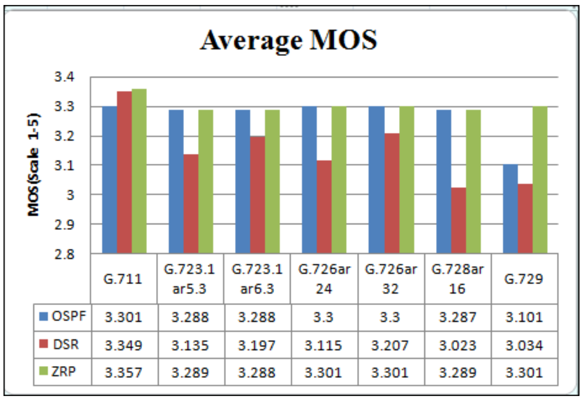


Figure 5

The figure 5 presents the graph of average MOS which shows that ZRP protocol has better MOS in sparse network environment. Furthermore in case of codec G.711 provides the maximum MOS with value of 3.349 than other codecs.

2. Throughput**:** It is defined as the average number of packets that can be transferred successfully on communication network from source to destination in a specified interval of time. It is measured in bits per sec.So high throughput is desirable in any communication network.

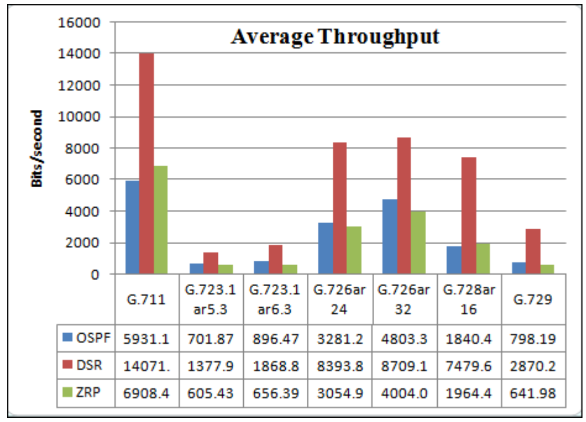


Figure 6

Figure 6 represents the graph of average throughput which signifies that DSR gives high throughput. In comparison of codecs highest throughput in sparse network is given by G.711 codec.

3.Average Transmission Delay: It specifies the time required by the data packet to put all bits of the packet into the wire across the network from source to destination and is directly proportional to the packet's length. It is usually measured in seconds. It can be described as follow-

$T_{d=N/Rt}$

where

T_{d}=Transmission delay in secs

N=Total bits

Rt= Rate of transmission

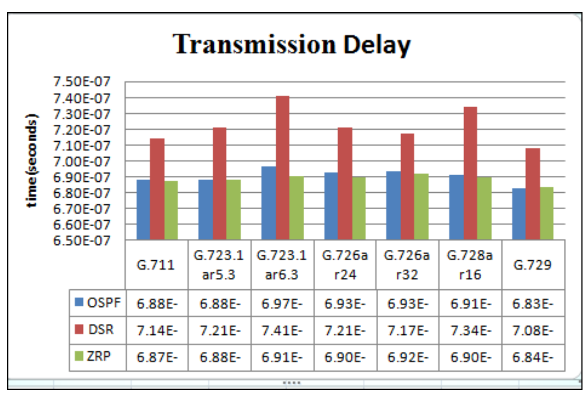
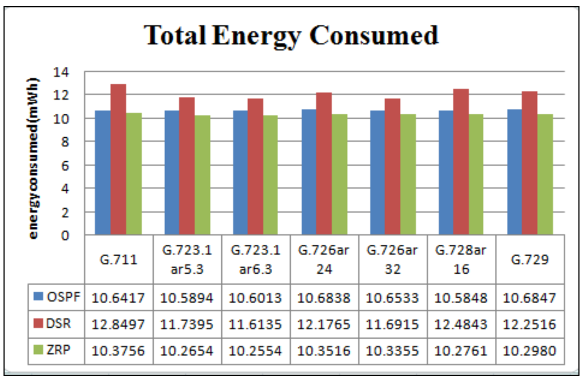


Figure 7

The above graph of transmission delay signifies that OSPF experiences less delay with value of 6.83E compared to other protocol. Furthermore codec G.729 experience lowest delay in sparse network and G.723.1ar6.3 consumes highest delay in sparse network environment.

4. Energy Consumption: It refers to the amount of energy consumed by a system which in total is the summation of energy consumed in Transmit mode, Receive mode, idle mode and Sleep mode. It is calculated under physical layer and measuring unit is mWh.

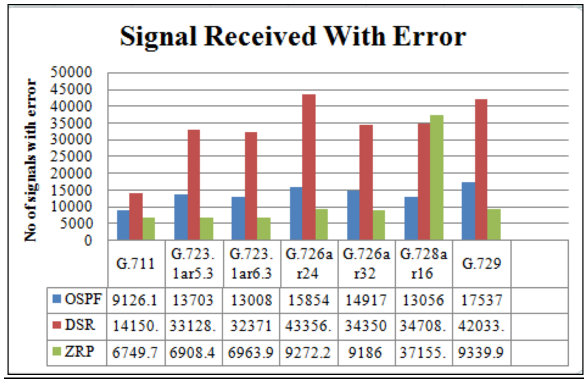


	G.711	G.723.1 ar5.3	G.723.1 ar6.3	G.726ar 24	G.726ar 32	G.728ar 16	G.729
OSPF	10.6417	10.5894	10.6013	10.6838	10.6533	10.5848	10.6847
DSR	12.8497	11.7395	11.6135	12.1765	11.6915	12.4843	12.2516
ZRP	10.3756	10.2654	10.2554	10.3516	10.3355	10.2761	10.2980

Figure 8

Figure 8 represents that ZRP consumes lowest energy compared to OSPF and DSR protocol.Whereas in comparison of codec G.723.1ar6.3consumes lowest energy and G.711 consumes highest energy in sparse network scenario.

5. Signal Received With Error: It is defined by the no of signals that are unsuccessful to reach to the destination .It is measured under Physical layer.



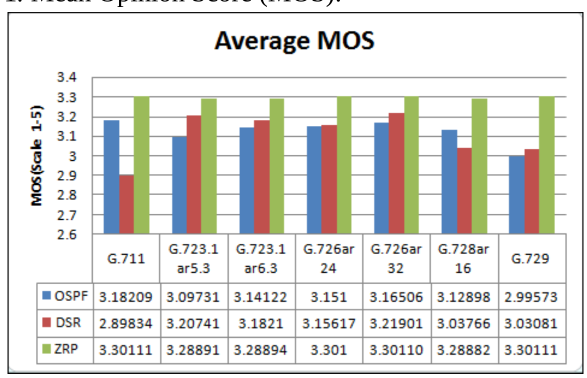
	G.711	G.723.1ar5.3	G.723.1ar6.3	G.726ar24	G.726ar32	G.728ar16	G.729
OSPF	9126.1	13703	13008	15854	14917	13056	17537
DSR	14150.	33128.	32371	43356.	34350	34708.	42033.
ZRP	6749.7	6908.4	6963.9	9272.2	9186	37155.	9339.9

Figure 9

Figure 9 represents graph of signal received with error which signifies that ZRP receives less no of signals with error. Furthermore lowest no of signals with error is received by codec G.711whereas codec G.726ar24 receives highest no of signals with error in sparse network environment.

Dense Scenario Result-

1. Mean Opinion Score (MOS):



	G.711	G.723.1 ar5.3	G.723.1 ar6.3	G.726ar 24	G.726ar 32	G.728ar 16	G.729
OSPF	3.18209	3.09731	3.14122	3.151	3.16506	3.12898	2.99573
DSR	2.89834	3.20741	3.1821	3.15617	3.21901	3.03766	3.03081
ZRP	3.30111	3.28891	3.28894	3.301	3.30110	3.28882	3.30111

Figure 10

The figure 10 of average MOS shows that ZRP protocol has best MOS in dense network environment. Furthermore in case of codec G.711 provides the maximum MOS with value of 3.30111 as compared to other codecs.

2. Throughput:

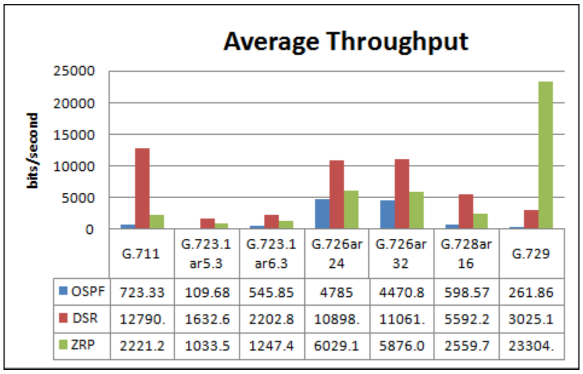


Figure 11

From Figure 11 which represents the graph of average throughput it is clear that ZRP gives maximum throughput. While in comparison of codecs G.729 codec gives the highest throughput in dense network scenario.

3. Average Transmission Delay:

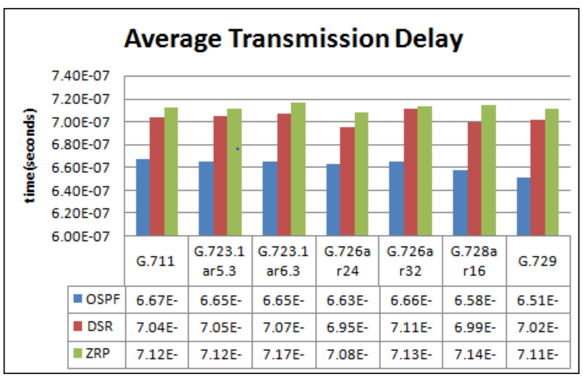


Figure 11

Figure 12 represents the graph of transmission delay that signifies that OSPF experiences less delay with value of 6.51E compared to other protocol. Furthermore codec G.729 consumes lowest delay in dense network environment.

4. Signal received with error:

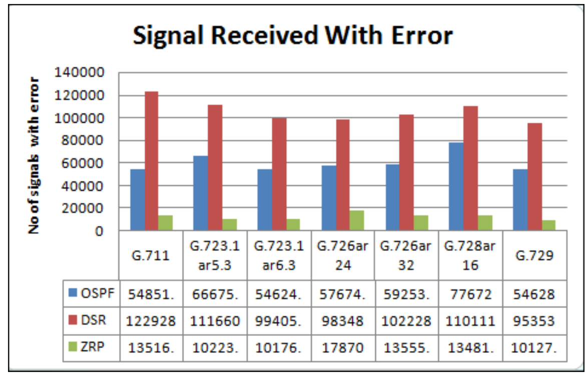


Figure 1 2

The above graph represents signal received with error graph figure which state that ZRP receives less no of signals with error compared to OSPF and DSR routing protocol. Furthermore in case of codecs lowest no of signals with error is received by codec G.729whereas codec G.711 receives highest no of signals with error in dense network scenario.

5. Energy Consumption:

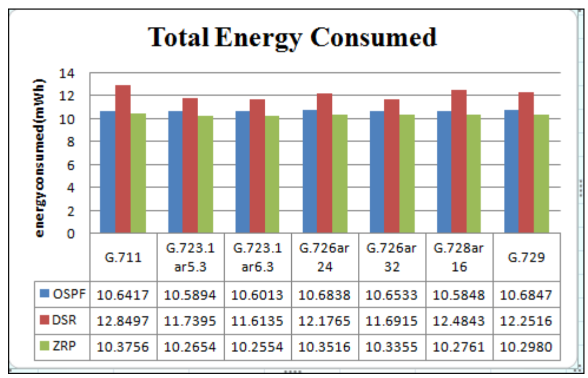


Figure 14

The above figure 14 represents total energy consumed graph which shows that ZRP consumes minimum energy compared to OSPF and DSR routing protocol. Furthermore in case of codecs lowest energy is consumed by codec G.723.1ar6.3 whereas codec G.726ar32 maximum energy in dense network scenario.

Conclusion

After the assessment of complete result, we have analyzed the codec performance in sparse and dense scale scenario. On the basis of contents of this paper, conclusion can be drawn as follows:

Result in accordance with the routing protocol concludes that the hybrid routing protocol (ZRP) performs better in most of the metrics in both the scenarios because it uses zone concept due to which high signal strength is dispersed to all the nodes of its zone. Whereas OSPF has uncontrollable flooding problem due to which more bandwidth is consumed. DSR requires large no of time to rediscover a broken link and high processing power to establish an alternate route.

On the basis of network condition, G.711 codec founds to be the most optimal codec in sparse scenario of 20 nodes due to maximum average MOS, optimal average throughput and low signal received with error. On the other hand among all codecs G.729 has been found best in dense scenario due to its optimal average throughput, minimum transmission delay and low signal received with error. From the above observation it is concluded that low data rate codec are more suited for dense scenario due to the long route distance between source and the destination which creates large internode transmission space .This long route distance & mobility leads to inconsistent link which facilitates the need of low bandwidth consumption codec. High data rate codecs performs optimal in sparse environment due to the short internode space between the source and destination node which facilitates low packet loss and high data rate.

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