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SIMULATION OF QOS PARAMETERS FOR VARIOUS VIDEO TOPOLOGIES USING RLNC BASED RTP PROTOCOL IN VIDEO STREAMING OVER COGNITIVE RADIO NETWORKS

Neeta Moolani
Department of ECE, Dr. M.G.R. Educational and Research Institute, Chennai 600 095, India. natashamoolani@gmail.com

Dr. K. Senthil kumar
Department of ECE, Dr. M.G.R. Educational and Research Institute, Chennai 600 095, India. ksenthilkumar@drmgrdu.ac.in

Abstract: Nowadays video streaming extends its applications to various multimedia services. The extremely challenging Over-The-Top (OTT) multimedia application create enlarged challenges to Internet Service Providers (ISPs) in assuring a realistic Quality of Experience (QoE) to their clients due to a require of flexibility, quickness and scalability in established networks. Here, we focus on reducing the buffer size of video and its impact on QoS parameters such as Packet Delivery Ratio (PDR), Throughput, End-to-End Delay and Energy have been evaluated. The work presented in this paper has focused on the simulation of video streaming over three video topologies (MP4, HD720 and HD1080 pixels) using RTP protocol integrated with RLNC over a Cognitive radio network. The system has been implemented and tested under a simulation environment by using the NS2 simulating tool. The QoS metrics for each video topology are carried out by employing 20 to 300 numbers of nodes over some time.

Index Terms: Quality of Experience (QoE), Quality of Service (QoS), QoS parameters, Random Linear Network Coding (RLNC), Cognitive Network

1. Introduction

The consumption of multimedia services has increased speedily over the past few years and is expected to widen even more over the next years. According to Cisco Visual Networking Index (VNI) forecast[1], 82% of all IP traffic will be video by the end of future 2022. Mobile-connected devices (like Device to Device and Machine to Machine communications) are approximated to be 14.6 billion by the ending of 2022 and are surpassing the world's predicted population of 8 billion by 2022. This rampant growth, due to the growing popularity and use of video streaming services (like YouTube and Netflix etc) has stimulated and initiated new revenue potential for Internet Service Providers (ISPs), mobile operators and Over-The-Top providers (OTTP). Providing high video quality to the end users is very important and challenging for the continued success of such services [2]. Nowadays, the end-users are adapted to more resource-demanding services with better quality from ISPs. However, targeting good Quality of Experience (QoE) is a challenging task because of many factors such as QoE metrics like Packet Delivery Ratio, Throughput, End-to-End Delay and Energy, various client request patterns, media changing contents, different network conditions and significant spatial and temporal variation in the performance of Content Distribution Networks (CDNs). Significant efforts from both academia and industry sides have been made to enhance the video content delivery chain and strengthen end user's QoE. The most common techniques used for upgrading end-user's QoE are either based on network optimization (QoE-driven network resource allocation and QoE-driven routing) or client-driven adaptive video streaming.

Although these efforts, QoE management residue a challenging task suitable to several issues [3] and may be classified into four different criteria which are the changeability of network resources, the uneven temperament of wireless channels, the variety of fixed/mobile networks and heterogeneous environments of end-user devices. The first criterion is that crowded regions such as trains, shopping malls and stadiums need continuous variation of network resource allocation to various clients. The second one is the appearance of latest services (like virtual/augmented reality (VR/AR) and video gaming), the diversity of framework of use and the user's expectation). The third is accounting for a diversity of networks (like fixed and mobile), anywhere diverse measurements and assessment methods have to be working for QoE management considering resource constraints. The final criterion is associated with the familiarity and fast development of multimedia services over the Internet and the heterogeneity of end-user devices with different capabilities (such as screen size, computational power/resources and storage capabilities). This causes even more challenges when allocating resources among users with different QoE preferences. The QoS mainly impacts QoE, as several QoS parameters directly or indirectly contributed to their influence on the user-perceived QoS. The key QoS parameters to affect multimedia services are delay, bandwidth, packet loss rate and jitter [4]. In the case of multimedia video streaming services, the bandwidth be able to optimized by choosing the appropriate video codec, while the jitter can be there managed by configuring the buffer supervision of the UE's decoder.

However, the packet loss rate and delay are two major QoS parameters which are to be utilized effectively to recover the client-perceived QoE. Delay is an important parameter that can be rooted by dropping of video packet[5]. Normally, when a packet arrives after the end of the threshold timer, it cannot be accepted in real-time applications (like video streaming) and is considered to be lost. The packet loss and delay rate are being used as the major QoS parameters to evaluate the user's QoE[6]. The quality of video streaming service and QoS parameters are directly correlated.

In this paper, we are experimenting three different qualities of video topologies employing with the number of nodes to observe and measure the differences in QoS metrics of Packet Delivery Ratio, Throughput, End-to-End Delay and Energy [7]. We implemented our simulations by deploying the nodes in three different properties of video topologies and comparing these metrics in the three above-mentioned video network arrangements. Our work in this paper is to give a thorough analysis and simulation study of these three deployments of video topologies supported by experimental results. This paper is organized as follows. The section I start with is the introduction part. Section II reveals the preliminaries. Section III discusses modern tools used in this research work. Section IV outlines related works and section V encompasses simulation results and discussions based on PDR, Throughput, End-to-End Delay and Energy our simulation set-up includes description of typical video topologies, parameters and metricss to be used to calculate the performance of video topologies. are presented in Section IV. Finally, the paper is concluded in Section V.

2. Video topologies

Multimedia streaming is a widely popular service on the Internet and overall 31% of traffic is occupied for Internet video streaming. Internet speed is disparate due to bandwidth limitation, local network, Internet service provider's speed (ISP) and package plan of the user. So it is tedious to access high-definition (HD) videos over a slow network with different devices. Different video codecs (formats) were designed to compress the video and decrease the size for streaming on the internet with high-quality video [8]. In the past 2000s flash videos were utilized to play videos in browsers and adobe flash player was a tool to support playing video through real-time message protocol (RTMP) [9]. FLV files were encoded with codecs like the Sorenson Spark or VP6 video compression formats and includes video bit streams which are a proprietary variant of the H.263 video standard.

The long-ago of mobile phone devices did not maintain FLV appropriately, therefore in 2000 3GP video file format be created with International Telecommunication Union (ITU) to worldwide implement video file format for mobile streaming. 3GP has low-quality video due to its low bit rate, screen resolution and frame rate with high compression for support on the small display of mobile devices and low network speed of 2G. the 3GP file format stores video files as MPEG-4 version 2, H.263 or MPEG-4 version 10 advanced video codec (AVC/H.264)[10].HD videos take more time to load onto a webpage when compared to the low quality and the big size of the video and also utilize more storage space on the cloud/server and require more network bandwidth to transfer from cloud/server to client. Online live HD video streaming is still a worry for service providers (SPs) due to the file format of videos used by SPs and the network speed of user's connecion. SPs employed a client feedback system to get client reviews on video quality and provide optimum examine to them. The term quality of experience (QoE) is used for evaluating user's opinions about the video quality and service they receive. ''QoE brings an assessment of human feelings, perceptions, expectations cognition and satisfaction of delivering a particular product, service or application''.In general, QoE is defined as a measurement of customer satisfaction or customer performance depending on the objective or subjective measure of using any service or product[11]. The main work of this research paper is to assess the QoS metrics correlated to the QoE of the end-user on the different video file formats and quality to examine which file format is well suitable for video streaming over cloud/server video hosting website which satisfies the user needs. During the research work, we repeat experiments with video file formats having different quality MP4, 720P HD, and 1080P (Full HD) which were taken from YouTube and stored on the local system. Downloaded videos are played for the client to recognize videos to trace ratings and observe the attributes and their impact on the QoE of video quality

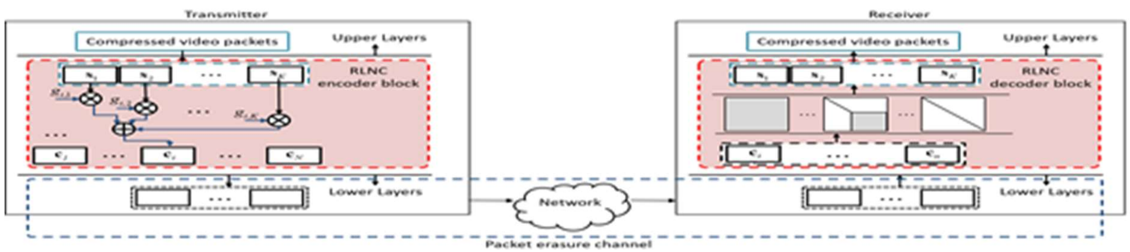
MP4 MPEG-4 (motion pictures expert group) is based on the quick time file format and it was first introduced in 2001 for public use [12]. MP4 file format was modelled by the International organization of standards (ISO) with variations such as .mp4, .m4a, m4p, m4b, .m4r and.m4v with standard number ISO/IEC 14496-14. Various versions of MP4 file format were provisioned with different compression codecs such as H.263, H.264 and H.265 for granting better video quality with compressed file size. The purpose of launching multiple versions with a different codec is to compress the file to transfer on the networks for multimedia streaming

to remote users but the limitation of bandwidth is the main consideration during the development [13]
720P Video: 720P (HD), well-known as widescreen high definition, is tiny for 1280px*720px resolution through 921,600 pixels in entire. It's the minimum obligation for HD. 720P is a progressive HDTV signal format with 720 lines/units/pixels vertically for horizontal progressive scan and 1280 lines/pixels horizontally that requirements vertical progressive scan [14]. Its aspect ratio is 16:9.

1080P Video: 1080P (Full HD) has a resolution of 1920px*1080px, and it's the cap of HD resolution. 1080P follows progressive scan(non-interlaced scan) with 1920 pixels/scan units horizontally and 1080 pixels/scan units vertically[15]. 1080P is also well-known as Full HD, FHD, BT.709, and Blu-Ray with 2,073,600 pixels.

3. Modern Tools

RLNC: The Random Linear Network Coding (RLNC) constitutes the RLNC encoder block in Fig.1



Generic RLNC sub-layers over the transmitter and the receiver side are connected via Packet erasure Channel Transmitter and RLNC decoder block at one or more receivers, connected via independent packet erasure channels [16]. An example of a single transmitter and a single receiver is illustrated in Figure 1.
RLNC encoder block: The input to the RLNC encoder is a source block s = [s1, s2, . . . , sK] includes K equal-length source packets, each containing L symbols of a finite field Fq of size q. RLNC encoder encodes the source block 's' into a stream of coded packets c = [c1, c2, . . . , cN], where each encoded packet ci represents a random linear combination of the source packets, i.e., ci = ∑Kgi,jsj, and is of the same length L symbols of Fq as source packets. Note that the transmitter can create arbitrarily many encoded packets N from K source packets in a rateless fashion. By fixing N, the set of encoded packets will be represented as c = s · GT, where coding matrix G represents an N × K random matrix over Fq. Systematic RLNC is also considered frequently when ci = si, 1 ≤ i ≤ K, i.e., the first K coded packets are the copy of source packets, thus the first K rows of G represent the IK×K identity matrix[17].

1. Packet Erasure channel:

Coded packets are transferred to single or many receivers through independent packet erasure channels. For the j-th receiver, the erasure probability of the corresponding channel is represented as ej, where 0 ≤ ej ≤ 1. While the focus is on a single receiver, we will neglect the index and use e as erasure probability.In this paper, we will only consider the integration of

RLNC as part of the higher protocol layers, without considering any details or information available from the physical (PHY) layer.

2. RLNC decoder block:

Due to possible packet losses, a receiver gets a subset of $n \leq N$ coded packets. Extracting the corresponding coding vectors, the receiver derives $c = s \cdot DT$, where $c = [c1, c2, \ldots, cn]$ indicates the set of received coded packets, while D denotes a random of $n \times K$ matrix over Fq. For recovering source packets, the receiver employs the Gaussian Elimination (GE) decoding method that successfully recovers the source blocks for the

Figure 1. Generic RLNC sub-layers over the transmitter and the receiver side are connected via Packet erasure channel rank r(D) of the decoding matrix D which holds that r(D) = K[18]. The complexity of GE decoding can be further diminished by exploiting systematic, sparse and tunable sparse

3. NS2 Tool: The simulated domain can be created using Network Simulator 2(NS2). NS2 employs programming languages like OTCL and C++ which motivates researchers and developers to include more functionalities and investigate new networking concepts by adding new protocols and applications to the object classes of the simulator. In addition, the TCL script is utilized to produce network topologies and connections as well as run simulations for predefined time stamps

4. Cognitive Network: The substantial growth of video-related services, which are bandwidth-hungry and delay-sensitive, and the need for quality of experience (QoE) granting for users pose a great challenge for the depleting radio spectrum. Cognitive radio (CR), as a powerful tool or technique for sharing spectrum, emerged as a promising solution to the above problem[19]. The basic principle is to allow CR users (CUs) to access the licensed spectrum, with restricted interference to primary users (PUs). Spectrum sharing can be attained over long timescales (as in the form of spectrum leasing) or over much shorter timescales (as in dynamic spectrum access).

Video streaming over CR networks (CRNs) has reached great interest in recent years, and significant technological advancements have been made. As illustrated in Fig.1, when there is a shortage of available spectrum for CUs, they can take advantage of the underutilized licensed and unlicensed bands to activate various video services such as video conferencing, video-on-demand streaming online gaming and various multimedia services. Even if a CU holds a licensed spectrum, the user can still employ CR to obtain more bandwidth to support a higher data rate to improve the QoE of the video session. By Comparing to users in traditional networks, CUs can adjust their QoE requirements according to the dynamic channel conditions in CRNs, or they can change their transmission parameters or metrics such as modulation, power, coding scheme and relay selection to adapt to varying network conditions. Since spectrum allocation is not fixed in CRNs, it is not impossible to optimize spectrum allocation whenever channel conditions change to achieve a global optimal QoE for multiple CUs[20]

4.1 Preliminaries for VCRN

Before streaming videos over a CRN, there are two major tasks to be executed which are identifying the available channels for video transmission and determining the QoE metric to be incorporated into the transmission strategy and for performance evaluation. Through spectrum sensing, the CR is more aware of the radio environment. The decision-making unit, usually in the form of a cognitive engine concludes to reconfigure the software-defined radio as the radio environment changes to maintain the QoE of CUs and avoid interfering with PUs.

Spectrum sensing is critical in QoE-driven VCRN in which it finds the possible transmission of opportunities. A reliable spectrum sensing result should not only disclose the available idle spectrum for CUs, but it should also not misidentify the unavailable spectrum as available spectrum for CU accesses to neglect interference to PUs [21]. In addition to that, spectrum sensing is coupled with the other aspects of QoE-driven VCRN.

The trade-off between Reliability and Delay: One of the most significant factors in determining QoE is the data rate. Video streaming is a bandwidth-hungry service, so a high data rate may be needed to achieve the target of QoE[22]. CUs have to find a sufficient amount of idle channels with spectrum sensing. However, trying to identify more idle channels usually requires a longer spectrum sensing time, which reduces the video transmission time and causes a larger transmission delay. Therefore, it is needed to reach a trade-off between spectrum sensing time and video transmission time. This feature differentiates VCRN from other applications that are not as bandwidth-hungry, such as voice, text messaging, and data file transmission.

Interaction with Other Factors: QoE is also correlated with other factors such as bit error rate and delay, which are impacted by choosing video modulation and coding scheme, channel conditions, and so on[23]. As discussed previously, a greater delay is going to occur as more time is spent on spectrum sensing, and therefore, QoE may be compromised.

Therefore, it is important to jointly optimize multiple design factors, such as spectrum sensing, modulation and coding scheme selection and relay selection for better QoE provisioning[24].

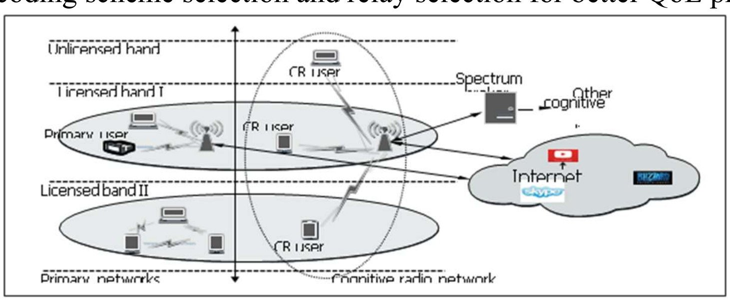


Fig.2.Configuration of video streaming over CRNs

4 .Related Works

Both industry and academia are acquiring SDN and NFV at an unparallel speed as future potential technologies to deliver service customization and better solutions concerning QoE control and management of multimedia services in current and future networks [25]. In addition to this, various works in the past were proposed to find a potential mechanism, use cases and architectures for QoE management of multimedia services in fixed and mobile networks.

Barakovic´ and Skorin-Kapov conferred a survey of QoE management in wireless networks by focusing on QoE modelling, QoE measurement and QoE adaptation. Skorin-Kapov et al. in [26] analyzed the emerging concepts and challenges related to managing QoE for multimedia services. By accounting for the parameters for QoE optimization, surveys on QoE-based scheduling strategies for wireless systems were shown by Sousa et al. [27]. A survey on QoE in HTTP adaptive video streaming was provided by giving a comprehensive survey of various QoE models for HTTP Adaptive Streaming (HAS) applications. Petrangeli et al. [28] have recently surveyed the QoE-centric management of adaptive video streaming services [29]. A comprehensive survey of existing work on QoE modelling and assessment (including both subjective and objective) and QoE management of video deliverance over various types of networks is provided by Alcardo Alex Barakabitze et al.[30]. However, the survey papers cited above do not explain how QoE can be correlated with QoS metrics parameters. Eventually, these studies do not exhibit insights into how networks or service providers can utilize the QoE management system to meet the end users' quality demands.

Even though the recent efforts enhance the QoE management aspects using SDN and/or NFV, we recognize that these works are limited in at least one of the following: 1) limited review and discussion of the standardization activities related to SDN and/or NFV; 2) lack of comprehensive interpretation of current research projects, state-of-the-art efforts, challenges as well as concrete research directions in SDN or NFV and emerging trends of multimedia services and applications in future networks; 3) concerning scope, they could not provide various architectural proposes, implementations and deployment innovations for QoE management system using SDN and/or NFV; 4) they do not have an answer on how QoE can be managed/measured or controlled in emerging technologies (such as cloud/fog computing, MEC or Information-Centric Networking (ICN)) and new domains with new constraints (e.g., delay-sensitive applications such as AR/VR or video gaming applications).

5. Simulation Results & Analysis

The following metrics were employed for performance analysis of protocols:

Throughput: It can be defined as the average rate of completed data transmissions in a streaming setup over a specific period

Packet Delivery Ratio (PDR): It is the ratio of actual packets delivered to total packets sent.

End-to-End Delay: End-to-end delay or one-way delay (OWD) is referred to the time duration for a packet to be transferred over a network from source to destination. Only way in one direction when source to destination is measured.

Energy: Energy consumed for streaming video. Generally, streaming a Netflix video in 2019 typically consumed around 0.077 kWh of electricity per hour, some 80 times lesser than the original estimate by the Shift Project (6.1 kWh) and 10 times less than the estimated (0.78 kWh)

The simulations were carried out with an NS-2 simulator on three different kinds of videos for the RTP protocol. To understand the effect of a varying number of nodes, the various efficiency parameters of Packet Delivery Ratio, Throughput, End-to-End Delay and Energy are calculated. The result was carried out for each number of nodes which are 20, 60, 100, 150, 200, 250, and 300. In fact, it has been observed that the packet delivery ratio improved by some margin when the number of nodes is increased Likewise the differences are measured for the parameters throughput, end-to-end delay and energy

Since the approach focuses on achieving efficiency through employing a different number of nodes on various video topologies, QoS parameters are simulated to validate the methodology. Targeting optimum QoS parameters for different quality of videos at various nodes will support enhancing the QoE on live video streaming networks. The QoS metric parameters have been analyzed thoroughly and the values at each node have been tabled for comparison. The X-graph simulator, which is based upon NS2, authorizes is judged by varying simulation time. The type of traffic considered for this network scenario is the constant bit rate. The number of connections considered is 20. The simulation time period ranges from 100 to 600 seconds, with a node velocity of 0–20 m/s. Every data packet has a size of 512B and the packet rate in time period ranges from 100 to 600 seconds, with a node velocity of 0–20 m/s. Every data packet has a size of 512 B, and the packet rate is4 P/s. The interpretation analysis of accomplishment for the recommended algorithm. In an NS2 simulator, the X-graph is utilized to plot the distinct network parameters such as Packet Delivery Ratio, Throughput, End-to-End Delay (E2D) and Energy. The X-graphs are believed to provide effective results including interactive plotting and graphing. An n number of nodes are dispensed over each video network topology to measure the parameters. The specified nodes are varied and their performances are measured. PDR=Ri/Si, where Ri is the number of nodes received by the receiver, and Si is the number of nodes actually sent by the sender. Simulation results show that the MP4 104 MB (fig.3)and HD 720-pixel (fig.4) videos can have higher PDR at a lower number of nodes and will get decreased at higher number of nodes. On the other hand, the HD 1080 pixel (fig.5) video delivers higher PDR at the increased number of nodes. Eventually, the PDR shows lesser values at the increased number of nodes.

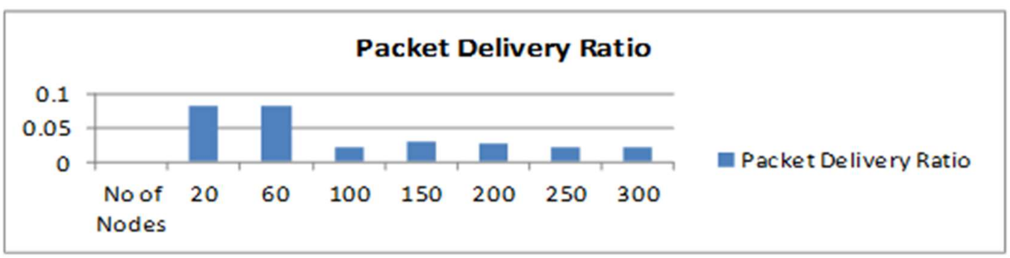


Fig.3. Video-1 104 MB (MP4)

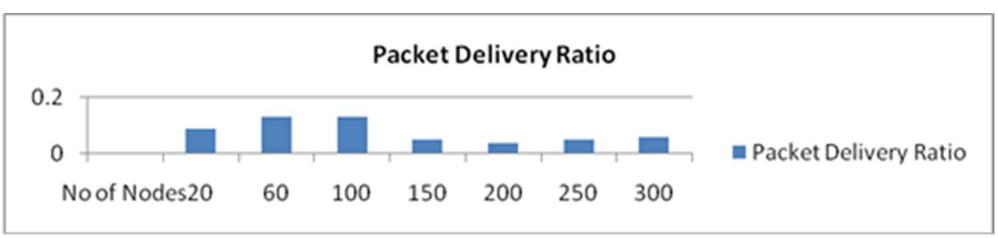


Fig.4. Video-2 966.86 MB (HD- 720 Pixels)

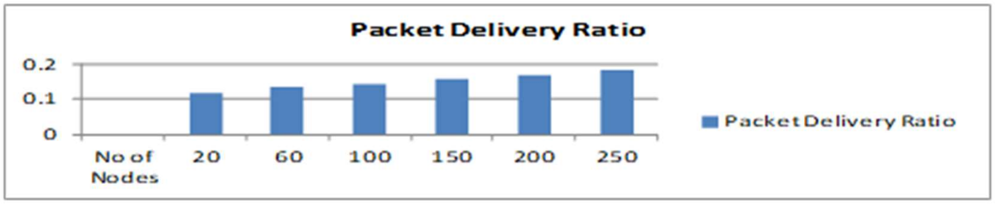


Fig.5. Video-3 1.59 GB (HD- 1080 Pixels)

Throughput can be measured in bits per second with the given function: Throughput=Dp×Ps where Dp symbolizes the number of delivered packets, and Ps stand for the size of a packet. Throughput reached a higher value at 60 numbers of nodes and attained a lower value at 150 numbers of nodes while applying a different number of nodes from nodes 20 to 300 on simulating MP4 104 MB (fig.6) videos whereas, on HD720 pixel (fig.7) and HD1080 pixel (fig.8) videos, the lower and higher throughputs were at 200 and 100 numbers of nodes respectively on simulation.

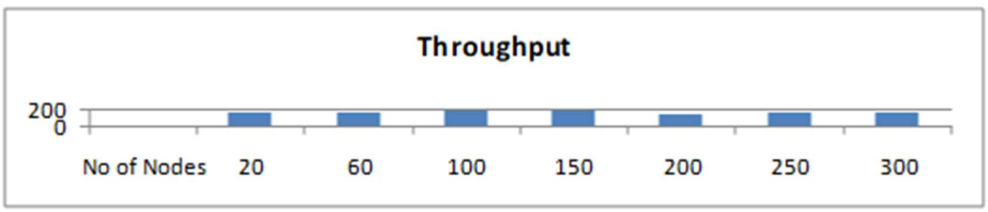


Fig.6. Video-1 104 MB (MP4)

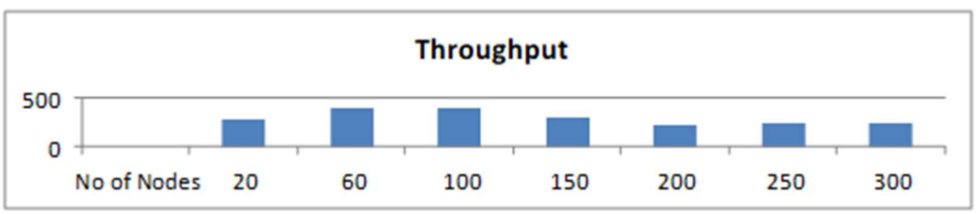


Fig.7.Video- 2 966.86 MB (HD- 720 Pixels)

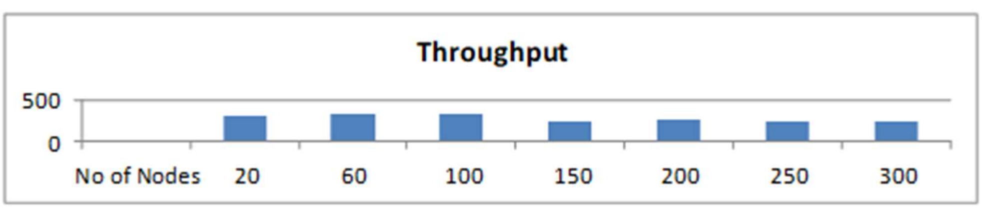


Fig.8.Video-3 1.59 GB (HD- 1080 Pixels)

The E2E Delay includes all the possible delays which can be caused by temporary storage of the data during path discovery, latency, and retransmission of the data by intermediate nodes, processing delay, and propagation delay. It can be determined by Di=(Tr−Ts), where Tr represents the receiving time and Ts represents the sent time of the packet. Generally, the end-to-end delay increases while increasing the pixel quality of the video. In this simulation, when the number of nodes is increased from 20 to 300 over three video topologies, on simulating MP4 104 MB (fig.9) videos whereas, on HD720 pixel (fig.10) and HD1080 pixel (fig.11) videos, the delay is increased.

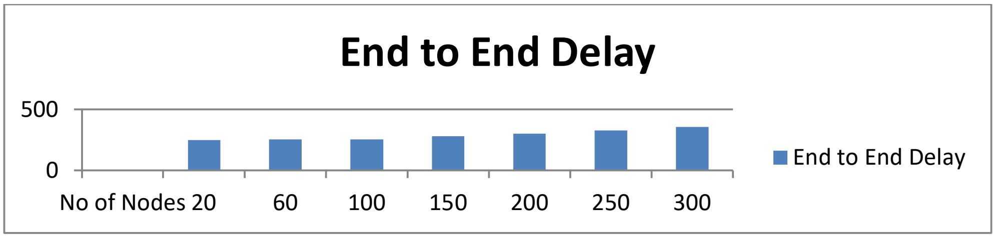


Fig.9.Video-1 104 MB (MP4)

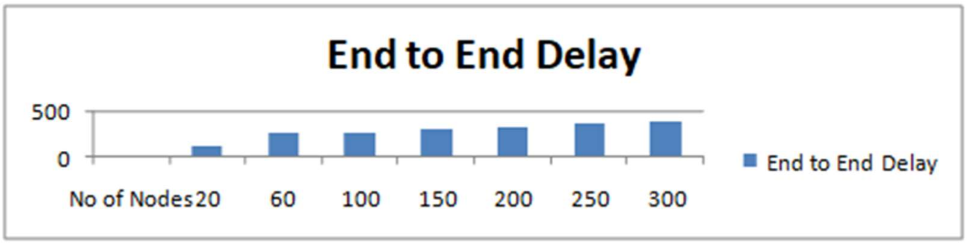


Fig.10.Video- 2 966.86 MB (HD- 720 Pixels)

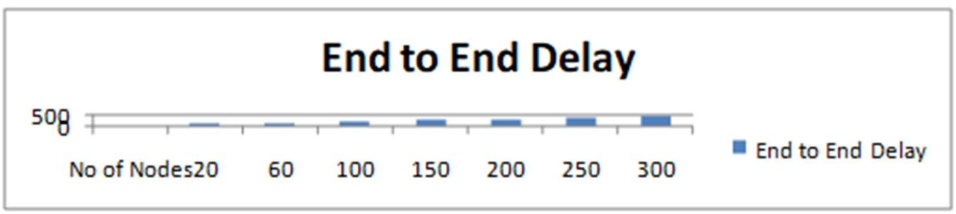


Fig.11.Video-3 1.59 GB (HD- 1080 Pixels)

As the number of nodes increases, so does the energy. Based on the simulation results, the lowest energy level has reached no nodes and when each node is added like from 20 to 300 numbers of nodes on each video network, on simulating MP4 104 MB (fig.12) videos whereas, on HD720 pixel (fig.13) and HD1080 pixel (fig.14) videos, the energy increases from lower to higher.

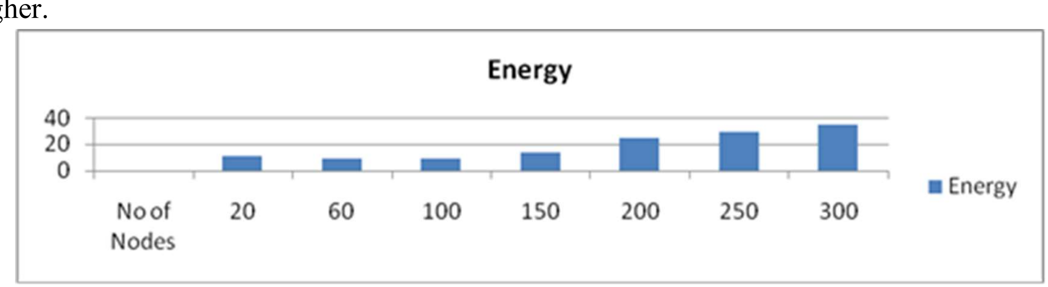


Fig.12.Video-1 104 MB (MP4)

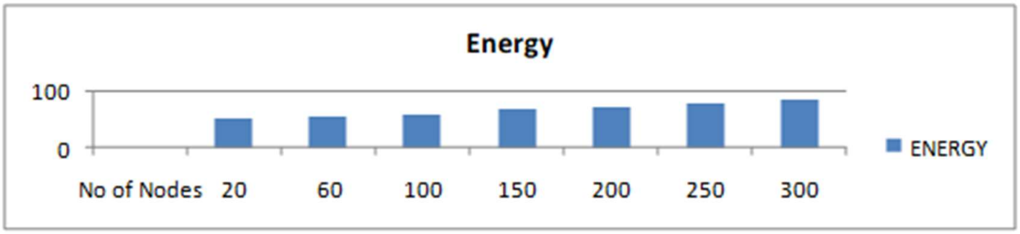


Fig.13.Video-2 966.86 MB (HD- 720 Pixels)

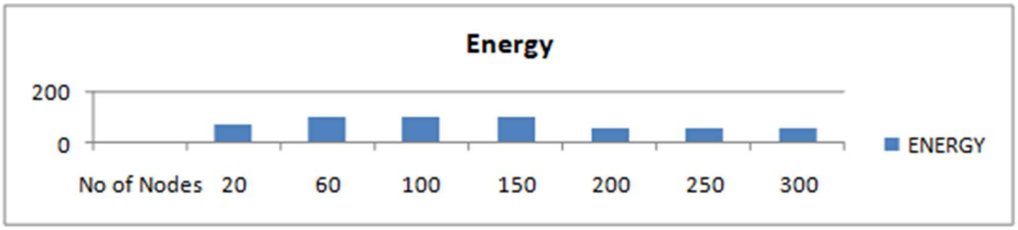


Fig.14.Video-3 1.59 GB (HD- 1080 Pixels)

6. Conclusion

Here, we have simulated and compared the performance of QoS parameters for different video pixel topologies using RTP protocol integrated with RLNC network coding technology for multimedia services that can be implemented in various streaming applications. The four performance metrics packet delivery ratio, throughput, end-to-end delay and energy are analyzed during this research work. It is found in our study that a simple video topology design does not offer better streaming performance in multimedia services. Simulation results show that the HD 1080 pixel quality of video performs better over other video topologies in streaming applications. The HD1080 pixel video topology could be used as the last possible option available for streaming networks as it causes low packet delivery along with higher end-to-end delay. The selection of an appropriate video type is significantly important in the deployment of multimedia streaming networks. The only disadvantage that is observed in this kind of video topology is the low throughput, as the packet drop ratio of this video type is much higher. Although the parameters of QoS metrics concerning the HD1080 pixel video topology reach an optimum level at an increasing number of nodes, it is analyzed that with more sink nodes, a higher HD pixel video design does not guarantee an improvement in the performance of QoE which is correlated with QoS metrics.

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