Comparative Analysis and Tests of Intelligent Streaming Video on Demand for Next Generation Networks: Two Colombian Study Cases

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Abstract – **This document proposes to evaluate the transmission of Video on Demand using Intelligent Streaming and Next Generation Networks concepts. Intelligent Streaming is an adaptive methodology to take advantage of bandwidth and the network resources, the streaming system had been tested over the UDNET and RUMBO-RENATA networks during the peak data traffic. Jitter deviation, data packet losses, user datagram protocol efficiency and quality of service are some of the indicators measured. The Intelligent Streaming guarantees the quality of service matching the server with the client needs, without changing the network policies delivery systems.**

*Keywords – **Intelligent Streaming; Video on Demand; Next Generation Networks; TCP and UDP; Network Efficiency.***

I. Introduction

Communications have generated a change in people's daily habits since Internet is accessible to everyone. Internet has become the main tool for work, networking and leisure activities due to home use as well as the number of smart mobile terminals. Now, the way to share Internet content has changed with the Next Generation Networks (NGN) because the files are sent in data packets [1, 2]. An efficient way to broadcast information is video streaming which has become the main application for video-conferencing, Video on Demand (VOD) and video-aided distance learning [3, 4].

The streaming transmission over the web has been developed for many authors [5, 6], where the reliability of real time playing of video segments is shown. Some of them study the point of traffic in the network making a pre-release of information and broadcast at a defined time. If someone wants to access to specific video, the user has to subscribe and when the information is available the user has to authenticate the subscription [7]. One of the limitations for video streaming is the consumption resources when too many users are connected. Bandwidth efficient algorithms applications show the non - reduction of efficiency system when the level of interaction increases. The interaction available is not affected and the overheads are associated to external sources and not to an algorithm failure [8].

The development exposed in this paper is a transmission VOD using intelligent streaming, increasing the QoS (Quality of Service) and decreasing jitter. In this proposal the QoS is defined according to the features of the client's connection and the streaming server capability, keeping the policy delivery unmodified.

The advantages of intelligent streaming are playing the video while the information is being downloaded and analysing the client's speed connection in order to transmit the data packet in a bit rate appropriately in relation with the channel bandwidth, avoiding delays and getting hung up [10, 11]. The strategy for improving the systems quality and transmission rate is the use of dedicated servers for streaming instead of web servers. These servers are in charge of storing and operating the data packets [12].

The design requirements are a trade-off between latency, memory and the overhead, take advantage of bandwidth, availability of data network and decrease in data packet losses and jitter [13, 14].

This paper is organized as follows; the first part shows the features of a streaming system, the tool chosen for this development, the process for encoding files and the selection of encoding rates for multimedia contents. Next the streaming model proposed for being tested using VOD is described. Also the delivery polices and the operation of the server streaming sub-system. In the last section the test results over the UDNET and RENATA-RUMBO networks are compiled and analysed.

II. Streaming Study Features

A streaming system comprises the user with a specified bandwidth and transmission rate [1], a player that supports video and audio, an Internet Service Provider (ISP), a streaming server with a control, storage and communications subsystem, an encoding and compression unit and video and audio sources. The components are explained briefly next. Fig. 1 depicts a typical structure of streaming.

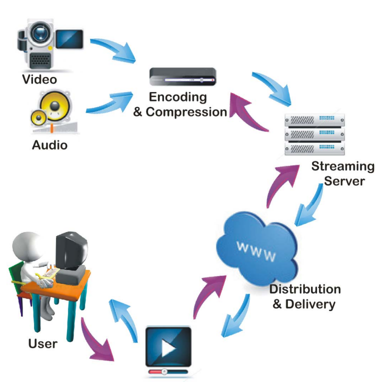

Fig. 1 Streaming system components

User: supports the multimedia contents reception and uncut viewing as well as VCR (Video Cassette Recorder) commands.

Player: is in charge of sending the user requests to the streaming server, the storage received contents in local buffers, encoding contents and synchronize the data for displaying.

Distribution and Delivery: Internet is the platform for exchanging information between *player* and *streaming server.*

Streaming server: the main functions are management of the policies delivery, system storage of the contents and to answer the requests made by the clients.

Encoding and Compression: is in charge of modifying the video and audio sources in a compatible format file with the user player.

Video and audio: are the sources for a streaming system. Their transmission can be on demand or live.

The aim of NGN and IPTV is convergence [15], for this reason the software for developing the application must satisfy this.

The tool Windows Media® has been chosen for its encoding system with a public domain license useful for files format supported by WMP® (Windows Media Player) like *wma* for audio files and *wmv* for video files. WME® (Windows Media Encoder) is a free multimedia contents player; the platform server is Windows Server 2008.

WME® offers a variety of possibilities for video edition and guarantees high quality and high compression file [16, 17]. The most important feature of WME is the capture contents in frames enabling the transmission of encoding sequences for different transmission rates, what is known as Intelligent Streaming. Besides it has other features: hardware acceleration, high definition video, multi-channel sound quality, segment-based encoding, easy to use because it has an intuitive interface, save contents and manage QoS delivery policies with the client. In addition to, it supports protocols for copyright like ISAN (International Standard Audiovisual Number), Ad-ID (Advertising Industry Standard Unique Identifier) and DRM (Digital Rights Management).

For encoding contents using WME it is necessary to get the source in a digital format, select the source, the distribution method and the characteristics of user connection, add the meta-data to the file and apply DRM. The process of encoding contents is:

File Conversion: this option allows for converting an *avi* or *mpeg* format file to compatible formats file with WME like *wmv, wma or wav*.

File Information: in this stage the meta-data are updated in the streaming file.

Direct Encoding: WME® allows it capture multimedia contents directly from the sound card or video card for encoding.

Select Source: the input file must be selected for encoding as well as the file name and the storage directory.

Distribution Method: is selected depending on the encoding formats and the kind of application. The streaming server for this development is WMS® (Windows Media Server).

Connection Characteristics: the transmission rate must be set, depending on the quality of connection. Also is possible set the transmission rate for multiple terminals that will be adjusted by intelligent streaming as is proposed in this document. In Table I. are showed the encoding rates selected.

TABLE I. SELECTION OF ENCODING RATES FOR MULTIMEDIA CONTENTS

TRANSMISSION RATE	FRAME RATE	FRAME SIZE	USE
1128 Kbps	29.97 fps	320 X 240	HIGH VIDEO DEFINITION BROADCAST OVER HIGH BANDWIDTH NETWORKS
548 Kbps	29.97 fps	320 X 240	STANDARD VIDEO DEFINITION BROADCAST OVER MODERATE BANDWIDTH NETWORKS
282 Kbps	29.97 fps	320 X 240	LOW BANDWIDTH NETWORKS
148 Kbps	29.97 fps	320 X 240	NETWORKS WITH DATA TRANSMISSION RATE NEAR TO 500 KBPS

III. MODEL SERVICE CHARACTERIZATION

The streaming systems are classified according to services offered like interactivity and information availability [17]. These characteristics make attractive the service increasing the interest and the desire to interact and learn about specific topics. The most common services offered are “Live” and “On Demand”.

Live: is a transmission that is being generated in real time [16]. On demand: is a transmission that is has its information stored in a buffer, allowing the user to download at any time video selected [4]. The model proposed and tested to VOD and applying Intelligent Streaming is presented in Fig. 2.

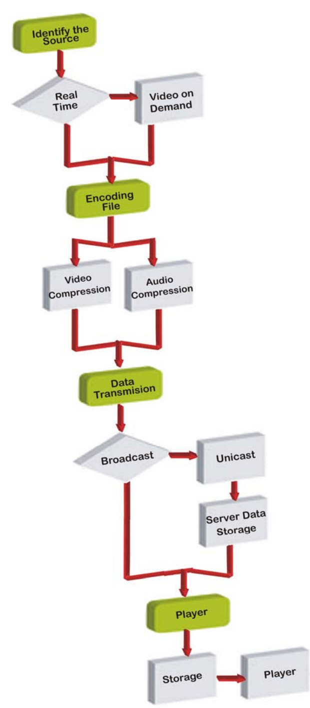

Fig. 2 Intelligent streaming structure

Intelligent streaming is a feature of WMS that scaling the data rate transmission through multiple video tracks [18]. The process for intelligent streaming begins with the request from the user, this request defines if the data source is live or VOD. Then the streaming server start the negotiation of rate transmission defining the QoS policies. In all streaming transmission the audio has priority over the video, because of overlapping is less noticeable in video than audio. Next, media content can be broadcasted in any of the 4 coding rates explained and the requests are analyzed to broadcast the video preserving the audio quality. The next steps are encoding separately audio and video and synchronize them on the client player. Data transmission is the stage that define if it is necessary storage the encoding files in a storage server, it happens when the request is unicast, while a broadcast request send the information to all terminals and each client decide what information wants to watch [16]. The advantage of broadcast is the traffic reduction over the net, the decreasing bandwidth needed and the streaming server processes are steady. The player has a storage system that save a part of files and the playing begins, this stage is a dynamic system that is storing and playing the information [8].

Streaming server delivery policies are the basis to synchronize data transmission ensuring data flow along all network and are related with a control sub-system [19], a communications sub-system and a storage sub-system as is depicted in Fig. 3.

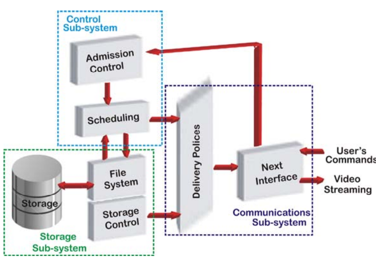

Fig. 3 Server Streaming Sub-systems

The storage sub-system has three sub-modules, storage, file system and storage control. All the encoding files are saved on data storage and the sub-module file system is in charge of enabling the data saving or transmission depending on the control sub-system requests. The sub-module storage control is communicated with the sub-module delivery policies of communications Sub-system, and its function is informing the capabilities of system during the transmission [19].

The control sub-system has the sub-modules of admission control and scheduling. The admission control must ensure the QoS and the resources availability as disk bandwidth, processing bandwidth, bandwidth network and space for storage, for the client. The scheduling sub-module has two sub-modules, disk Scheduling and network scheduling their function is planning how must be the data transference from the Storage sub-system to the memory buffers and from the memory buffers to the network. Communications sub-system has the sub-modules, delivery policies and net interface. The main function is responded to user requests and offers the services according to client capabilities.

IV. NETWORK ANALYSIS MEASUREMENT RESULTS

With the known benchmark program GNU Iperf [20], which measures some TCP/IP network features, two networks are studied as study case with the same video data and are evaluated with different performance indices in order to show the effectiveness of the intelligent streaming model proposed.

Fig.4 shows two pictures of the same video from a server to a client – server, the software mentioned evaluate the TCP and UDP performance.


Fig. 4 Real Time Video Streaming Implemented

A. *Measurements on the UDNET Network*

The first study case is the UDNET Network (Universidad Distrital Network) which is LAN (Local Area Network) and its theoretical transmission bandwidth is of 100 Mbps/s, the evaluation indicates that this network has more efficiency with large amount of information without fragmenting; the first test shows the TCP window rate with 8, 20000, 100000 and 1000000 Kbytes and is observed how the efficiency increase with those windows rates. Fig. 5 shows this network test.

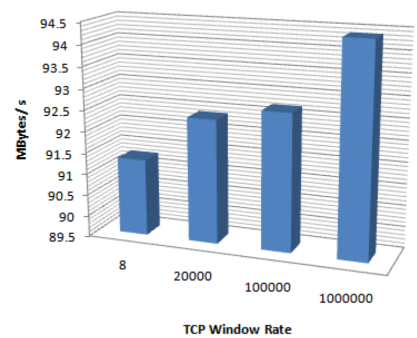

Fig. 5 TCP Effective Bandwidth of UDNET Network.

Data packets have not always the same delay [14], this associated to the jitter effect which relates the time expected when packets arrive, with the time of packets are delivered.

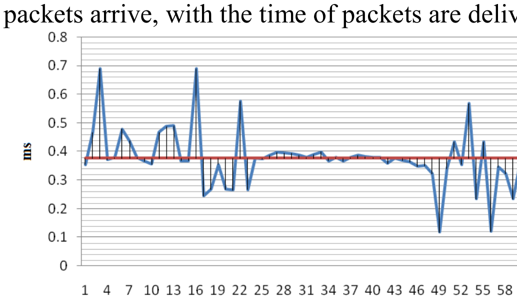

Fig. 6 In blue: UDP Jitter derivation in UDNET network. In red: UDP Jitter average.

Jitter measurement in this application is presented in Fig. 6; the jitter difference between the average is minimal; it shows

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that there is no degradation in video transmission. The average level of jitter deviation is 0.38ms, with a maximum value of 0.689ms and a minimum of 0.118ms.

It is well known that packets of data do not have a correct order in delivery and the rate transmission has losses and is also inconstant, however the audio requires this rate to be constant. Jitter buffer at the receiver compensates the effects by its function of trade off between delay and loss [10], these jitter buffers have a variations of 30ms and manage the audio transmission at a constant rate.

If the rate of transmission is slower than supported in buffer, is presented with high losses in packets spoiling the transmission quality, the maximum limit of losses should not exceed 1% [10, 12], due to these data losses which are noticeable in the final user and the service is demoted. The data video on the network studied shows the next behaviour according to the amount of information transferred.

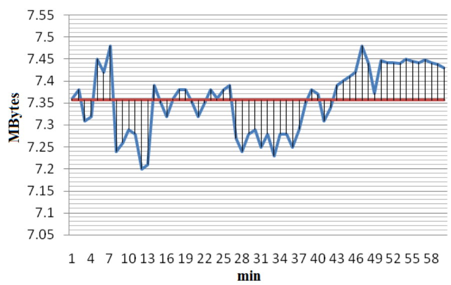

Fig. 7 In blue: Amount of Information Deviation Transferred. In red: Amount of Information Average.

According to the effective average of data transmitted, which is of 7.36 Mbytes, 3600 samples per hour are sent (sample per minute), Fig. 8 shows the variability of losses, obtaining a minimum deviation of average reference.

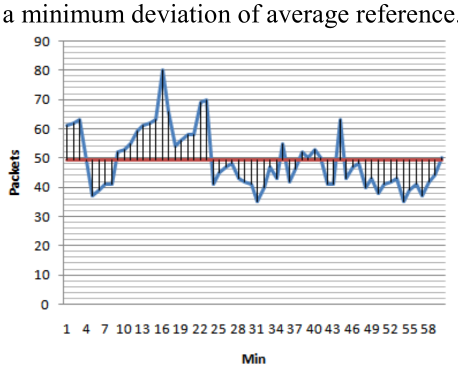

Fig. 8 In blue: Data Packet Sent Losses on the UDNET Network. In red: Information Losses Average.

The packet average lost is of 49 packets per 5351 packets sent, with a maximum lost of 80 and a minimum of 35 packets. In percentage terms the value of data packet losses is about of 0.91%, which is suitable to guarantee a QoS (Quality of Service QoS) in voice and video standard demand of lower value of 1%. Tests on UDNET network shows this fulfillment demanded in this kind of service. Fig. 9 shows the efficiency measurement mentioned.

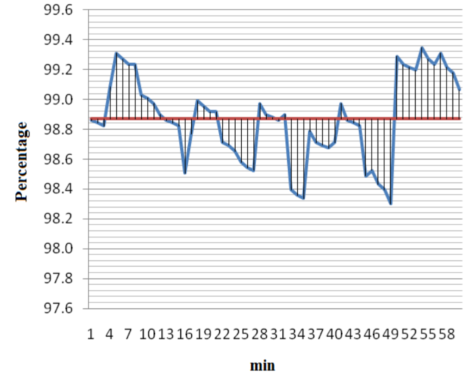

Fig. 9 In blue: UDP Efficency on the UDNET Network. In red: UDP Efficency Average.

The efficiency average in the network faced with observed variations, has a measured value of 99.1% which is in the within margins in a correct service.

Table II shows the main measurements performance indices evaluated in the UDNET Network, where TB, EB and TCP WR mean Theoretical Bandwidth, EB Effective Bandwidth respectively (Mbytes/s) and TCP Window Rate (Kbytes).

TABLE II. UDNET NETWORK MEASUREMENTS SUMMARY

SECONDS	MBYTES	TB	EB	TCP WR	EFFICIENCY
10	109	100	91.3	8	91.3
12	133	100	92.4	20000	92.4
19.3	213	100	92.7	100000	92.7
11.3	127	100	94.4	1000000	94.4
				AVERAGE	92.7

B. *Measurements on the RUMBO-RENATA Network*

Now, the same tests exposed above are applied to the second study case, the RUMBO – RENATA network. Fig. 10 shows the measurement of the network with the same window rates of the first case, this presents a better efficiency transmission packet that is about of 20000 Kbytes and its theoretical transmission rate is of 60Mbytes/s.

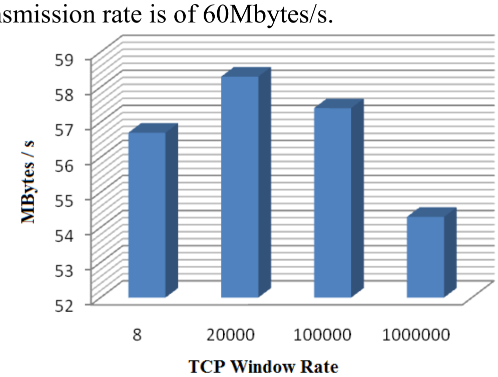

Fig. 10 TCP Effective Bandwidth for RUMBO-RENATA Network.

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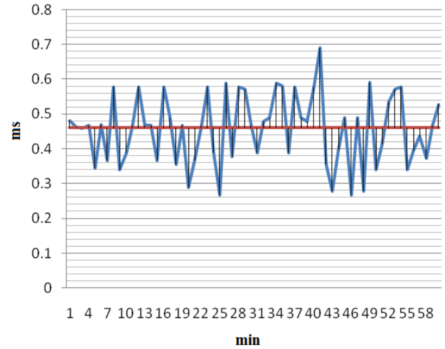

Fig. 11 In blue: UDP Jitter derivation in the RUMBO-RENATA Network. In red: UDP Jitter Average.

Jitter average deaviation measured is 0.46ms, with a maximum of 0.690ms and a minimum of 0.267ms, these values are in the margins to bring a suitable service to VOD, this value is according to the maximum value allowed in buffer jitter mentioned before.

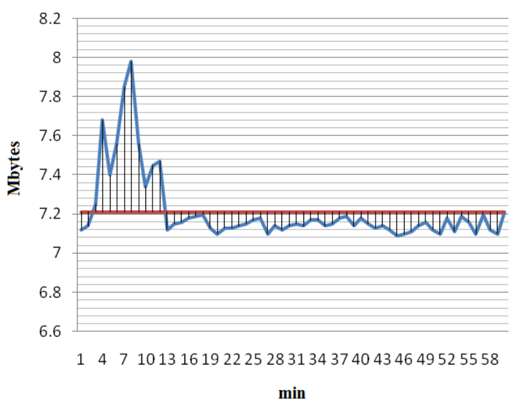

Fig. 12 In blue: Amount of Information Variability Transferred on RUMBO-RENATA Network. In red: Amount of Information Variability Average Transfered.

Fig. 12 shows the effective transference in the network, the average data transferred is 7.21 Mbytes with some variations observed, like in the first case 3600 samples has been sent in one hour.

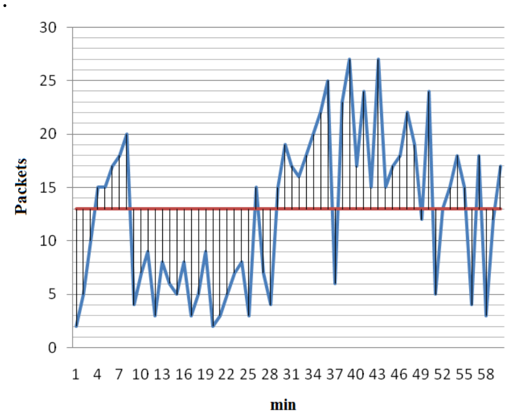

Fig. 13 Network. In blue: Data Packet Sent Losses on the RUMBO-RENATA Network. In red: Information Losses Average.

Data packet loses average is of 13 per 5351 packets sent which represents 0,24%, with a minimum of 2 packets and a maximum of 27, this fits with the parameters established.

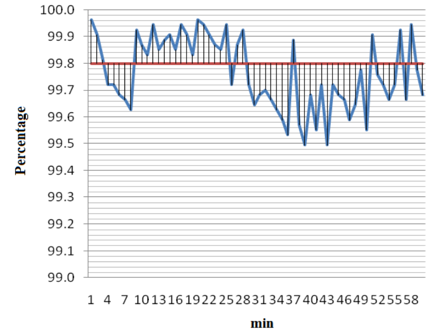

Fig. 14 UDP Efficiency on the RUMBO-RENATA Network. In blue: UDP Efficency on the RUMBO-RENATA Network. In red: UDP Efficency Average.

Average efficiency conexión has a value of 99.8% which implicates a well performance in video transmission.

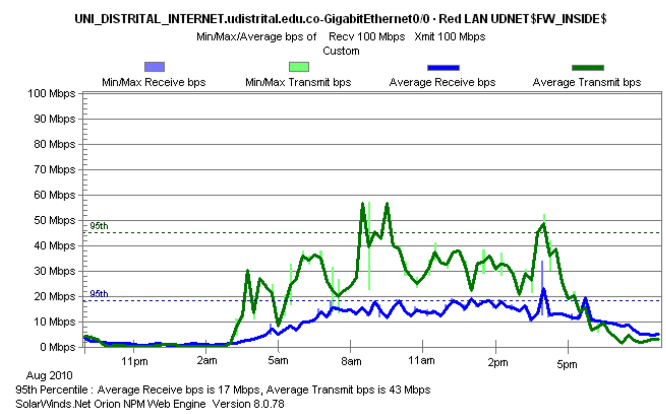

Fig. 15 Transmit and Receive Rate on the UDNET Network. In blue: Average Receive bps on the UDNET Network. In green: Average Transmit bps on the UDNET Network.

TABLE III. RUMBO RENATA NETWORK MEASUREMENTS SUMMARY

SECONDS	MBYTES	TB	EB	TCP WR	EFFICIENCY
10	111	100	56.7	8	94.50
12	130	100	58.3	20000	97.17
19.3	220	100	57.4	100000	95.67
11.3	118	100	54.3	1000000	90.50
				AVERAGE	94.46

From measurement Table is concluded that RENATA-RUMBO Network presents a high level efficiency in video packets transmission using intelligent streaming

V. CONCLUSIONS

Next Generation Networks have become a viable option for firms and has impacted positively in clients which demand them, tests performance in this document demonstrate the simplicity to broadcast multimedia contents with intelligent streaming in networks where it is not apparently supported.

A high efficiency performance of TCP and UDP is obtained in both Intranet networks (92.7%), even though the peak hour traffic (8:00 am – 9:00 am) and it validate the implementation QoS in the two study network cases developed here.

This real time application shows how it is improved the quality of service applying intelligent streaming in some local study cases networks, without a change in the policies or access network, only requiring network negotiations between client and the server.

It has found that the peak hour is between the 8 and 9 of morning because in this hour the University starts the most Network activities, start to work the administrative area and a big part of University´s computers request access to the servers, as shown in Fig. 15.

Multimedia streaming has been gradually gaining ground from peer to peer networks used to download and make play lists contents, due to streaming technology it does not require high storage capacity of streams, on the other hand, peer to peer requires first, a complete download to reproduce the content.

The TCP Protocol is used only for the service website and the connection control, the pair of protocols RTP/UDP was used for the transmission of audio and video data.

SCTP Protocol was no used because their greatest features as RTO and heartbeats to declare inactive or failed connections would not be exploited, also for this specific application was unlikely that the computer, laptop or smartphone to have 3 or more IP address to take advantage of another great feature of this protocol as the “multihoming”.

This paper shows a relatively easy and efficient way to transmit audio a video through a typical network and an academic network of advanced technology using techniques such as intelligent streaming with support for IPv6 using Windows Server 2008, and comparing the results of efficiency in both networks.

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