Experimental Study of Unicast and Multicast Video Traffic using WAN Test Bed

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***Abstract*— The deployment of IP multicast protocols in each router in the path is needed to access the benefits of the multicast. IP multicast exists today mainly in LAN networks and in small areas of highly controlled interconnected networks. In fact, some companies started to offer tailored solutions to try to overcome these limitations. On the other hand, over the last decade, the multimedia applications have expanded rapidly, and in particular in video applications. There are Internet sites that offer movies on-line; and it is common for users to upload and download videos with sites like YouTube (© Google Inc.). The video call via the Internet is common with applications, such as Skype (© Microsoft). The growth of video traffic should be taken into account when designing a network. Understanding the behavior of the video traffic and the requirements for the network helps network administrators to improve the traffic planning. In this work, a quantitative analysis is performed by experimentation, in order to evaluate the behavior and impact of video traffic on WAN networks. We propose a WAN test bed composed by a video traffic server and several client stations that allows injecting unicast and multicast video traffic, compressed with several codecs. From capturing video traffic, we identified several interesting unicast and multicast performance metrics, such as: throughput, number of frames, delays, jitter, and interframe spaces and frame sizes distributions. Several factors have been taken into account: the video resolution configuration, the type of video, and restrictions on the bandwidth, as in a corporate real WAN link of some few Mbps. This study facilitates the comparison of the results with those obtained from analytical and modelling studies for different contexts.**

***Resumen*— Para acceder a las ventajas de la multidifusión es necesario implementar protocolos de multidifusión IP en cada encaminador en la ruta. La multidifusión IP existe hoy principalmente en redes LAN y en pequeñas áreas de redes interconectadas altamente controladas. De hecho, algunas compañías han comenzado a ofrecer soluciones a la medida para tratar de resolver estas limitaciones. Por otro lado, en la última década, las aplicaciones multimedia se han expandido rápidamente, y en particular las aplicaciones de video. Hay sitios de Internet que ofrecen películas en línea; y es común que los usuarios puedan descargar videos en sitios como YouTube (© Google Inc.). Las llamadas de video vía Internet son comunes en las aplicaciones, tales como Skype (© Microsoft). El crecimiento del tráfico de video debería tenerse en cuenta cuándo se diseña una red. Entender la conducta del tráfico de video y los requerimientos para la red ayudan a los administradores de red a mejorar la planificación del tráfico. En este trabajo se ha realizado un análisis cuantitativo por experimentación con el objeto de evaluar la conducta y el impacto del tráfico de video en las redes WAN. Se propone un laboratorio de prueba WAN compuesto por un servidor de tráfico de video y algunas estaciones clientes, que permite inyectar tráfico de video unidifusión y multidifusión comprimido con diferentes codecs. Con la captura del tráfico de video se identificaron diferentes métricas de prestaciones undifusión y multidifusión, tales como: rendimiento, cuenta de tramas, retardos, jitter y distribuciones estadísticas de los espacios intertrama y de los tamaños de trama. Se han tenido en cuenta los siguientes factores: la configuración de la resolución del video, la clase de video, y las restricciones del ancho de banda, como en un enlace WAN corporativo de algunos pocos Mbps. Este estudio facilita la comparación de los resultados con aquellos obtenidos desde estudios analíticos y de modelación para diferentes contextos.**

I. Introduction

The traditional IP communication allows a host to send packets to another host (transmission through unicast) or to all hosts (transmission by diffusion). IP multicast provides a third possibility: it allow a host sending packets to a subset of all hosts as a group of transmission. IP multicast is a technology to conserve bandwidth, specifically designed to reduce traffic, transmitting a single stream of information potentially to thousands of recipients. In this way, it replaces multiple copies for all beneficiaries with the delivery of a single flow of information. Therefore, IP multicast is able to minimize the burden, both in source and destination hosts, and simultaneously the total traffic in the network.

Within a multicast network, routers are responsible to replicate and distribute the contents of multicast to all hosts that are listening to a specific multicast group. Routers use multicast protocols that build distribution trees to transmit content multicast, which ensure the greatest efficiency for sending data to multiple receivers. Any IP multicast alternative requires the source sending more than one copy of the data. The traditional unicast at application level, for example, requires that the source transmits a copy to each receiver of the group.

Nowadays more and more network applications require the forwarding of packets from one or more sources to a group of receptors. These applications comprise the majority of the data transfer (for example, the delivery of software updates from developers to end users), the transmission of media (audio, video, text), the exchange of data (for example, a videoconference distributed among participants), the entry of data (for example, shares), the Web cache and updating of the interactive gaming (for example, virtual distributed environments or multiplayer games). A very useful abstraction for the representation of each one of these applications is the concept of multicast.

Nevertheless, the deployment of IP multicast protocols in each router in the path is needed to enjoy the benefits of the multicast. Therefore, IP multicast exists today mainly in LAN networks and in small areas of highly controlled interconnected networks. In fact, some companies started to offer tailored solutions to try to resolve these limitations.

The barriers that have so far prevented the widespread deployment of IP multicast are:

- Each routing device in the delivery route must support the IP multicast, which means that if the IP multicast is omnipresent, each service provider must decide to offer multicast service.
- In the network interconnections existing today, it is a great challenge to offer reliable multicast service in environments that are inherently better effort.
- Streaming video requires a flow of adaptive bit, which lacks the multicast.
- Nowadays streaming video formats, designed to use mechanisms of progressive download, are not inherently capable of multicast.
- The interdomain multicast or between pairs of Internet providers ISP requires business agreements and complex network configurations

However, according to the trends and traffic forecasts, the deployment of services of video is the one among all the applications that offers more incentives for the strengthening of the IP multicast platform of a supplier or between the suppliers of services, since it becomes the most efficient (depending on the average costs of supporting data traffic, voice and video).

According to a classification proposed in [1], the video traffic can be (among others):

- IPTV Broadcast: This requires delivery of a downlink for a track of traffic of latency-insensitive through broad channels of bandwidth (1 to 4Mbps SD or 6 up to 10 Mbps HD) for a few users per channel.
- Live transmission of video event (webcast): This requires delivery of a downlink for a track of traffic of latency-insensitive through a single channel of high bandwidth (1-4 Mbps) seen by all users.
- IP video surveillance: This requires delivery of an uplink/downlink traffic latency insensitive, with many channels of video feed of variable quality (500 Kbps to 2 Mbps) to a small set of viewers.
- Interactive videoconference: This requires the two tracks of delivery of interactive traffic affected by latency and jitter (maximum of 150 to 200 ms), but requires under symmetrical bandwidth (1 Mbps) between pairs.
- Video on demand (training, pre-recorded): This requires delivery of a downlink traffic insensitive to the latency with many channels of high-bandwidth (1 to 4 Mbps SD or 6 up to 10Mbps HD) consumed by a few concurrent users per channel.

These types of traffic illustrate the variables that need to be quantified for any deployment of video and multimedia: directionality, throughput, latency and jitter tolerance, as well as the number of channels and users. Another key metric is the tolerance for error.

The principal contributions of this article, that complements and extends the contributions and conclusions made in [2], are: i) to specify and experiment on a new WAN test bed as scenario offering metrics evaluation while using typical video codecs for network traffic in line with the expectations of a real WAN, ii) to compare metrics over the standard multicast mechanism that provides substantial improvements in relation to the unicast traffic in this context, and iii) to provide conclusions obtained using a WAN test bed with real equipment that complement, enrich, and facilitate comparison with precedents within WAN network knowledge obtained from analytical and modelling studies using mathematical tools for different real and hypothetical contexts.

The rest of this document is structured as follows. Section II introduces some background and related work. Section III provides a general view of video codecs. Section IV describes hardware and software resources of the WAN test bed. Section V presents the results obtained for the performance metrics and the statistical distributions. And section VI summarizes the most significant conclusions.

II. Background and Related Work

The projections for the next 5 years show a steady growth in the video applications, despite macroeconomic conditions in many parts of the world. Because of that fact, the contents of video have much higher bit rates than other types of content, video will generate much of the growth of future traffic. It is expected for the 2018 that 65% of the total will be associated with video applications [3].

Providing Quality of Service (QoS) in networks is a considerable challenge for data networks with video traffic. Each new WAN scenario to inject unicast or multicast traffic, compressed with various codecs, introduces a new challenge due to the traffic levels, throughput, latency, and jitter and error tolerance. Another key metric is the number of simultaneous channels and users.

Various applications make use of growing video traffic in the LAN and WAN networks. Each one of them has its own special characteristics and demands to ensure an adequate level of QoS. There is a large amount of experimental work and simulation carried out that exhibits the behavior of each case, from an analysis of the capture of traffic. For example, in [4] the authors evaluate the performance of three state of the art video codecs on synthetic videos. An extensive number of experiments are conducted to study the effect of frame rate and resolution on codec's performance for synthetic videos. While in [5] a comparative assessment is presented for the two video coding standards: H.265/MPEG-HEVC (High-Efficiency Video Coding), H.264/MPEG-AVC (Advanced Video Coding), and also of the VP9 proprietary video coding scheme using an experimental test bed. In [6] the impact of the H.264 video codec on the match performance of automated face recognition in surveillance and mobile video applications is assessed. Other works evaluated the behavior with an experimental study of multimedia traffic performance in mesh networks, for performance evaluation and analysis of wireless networks [7]. On the other hand, [8] presents a comparison of the performance of H.264/AVC, Xvid and WebM/VP8 video codecs in wired and wireless networks. The codec performance is evaluated for different packet loss and delay variation values. There are also experiences to generate synthetic traffic, as in [9], where the QoS behavior of video traffic models for H.264 AVC video was evaluated. These models allowed the generation of synthetic packet traces which were used in a simulation model. In [10], the authors empirically analyze the video multicast operation, especially, along with power management operation in two aspects: (1) whether commercial WiFi devices correctly operate as defined in the standard and (2) what problem the standard-compliant operation can induce. While in [11], we specified and experimented on a new scenario offering metrics evaluation while maintaining a video intensive proportion for network traffic in line with the expectations of the Wi-Fi traffic future.

In these and other research articles [3-19] there are few papers available on simulation and experimental studies on WAN networks. And to the knowledge of the authors, there are no proposals that combine, in a WAN test bed, with constraints of these links, the problematic of the variants of the codecs for video traffic, with the use of unicast or multicast flow. This new experimentation scenario allowed us to obtain the required detail of several performance metrics for this context.

III. Brief Description of Codecs

Codec is an acronym for coder-decoder. Describes a specification developed in software, hardware, or a combination of both, able to transform a file with a data stream or a signal.

Codecs can encode the flow or the signal for transmission, storage or encryption, and recover or decrypt the same way for the reproduction or manipulation in a format more suitable for these operations. In our case study we have worked with the following 3 codecs:

A. H.264/MPEG-4 AVC

H. 264 / MPEG-4 AVC provides a significant advancement in the efficiency of compression to achieve a reduction of around 2 times in the bit rate compared to MPEG-2 and MPEG-4 simple profile. In the formal testing by the JVT (ITU), H. 264 gave an improvement of the efficiency of 1.5x or higher in 78% of the cases and 77% in those who showed improvement 2x or greater and up to 4x in some cases. The 2x improvement allowed H. 264 the creation of new market opportunities, such as: Video VHS-quality 600 Kbps. This can enable the delivery of video on demand via ADSL lines. Provides excellent clarity for the profiles covered by extensions of range of fidelity which extends the levels to "losseless" or very close to this and supports chroma 4:4:4 and bit depth of up to 12. MP4-AVC is more efficient than "Visual Coding" (part 2), MP4-AVC provides better quality at the same sampling rate or equal quality at lowest rates.

B. H.263/MPEG-4 Part 2

MPEG4 Part 2 called MPEG4 VISUAL, belongs to the family of standards MPEG-4 ISO/IEC. There are several implementations of this standard, being DIVX, Xvid, Nero Digital the most popular. MPEG4 VISUAL was put on the market with a family of configurations called parties. We are dealing with part 2 which supports three profiles, these are: Simple Profile, Advanced Simple Profile and Advanced Studio Profile. ASP was the profile used in our tests, this allows the use of the following types of visual objects: Simple (rectangular video that uses frames intra and predicted) and Simple Advanced (rectangular video, improved compression and bidirectional frames). Six compression levels are allowed (0 to 5). Levels 0 to 3 have data rates from 128 to 768kbps, levels 4 and 5 added interlaced encoding to achieve rate of 3 to 8 Mbps. MPEG4 VISUAL has a good support for moving image. As for quality, it is between moderate and very good, taking into account that sampling is limited to 4:2:0 and that MP4 is a lossy compression format. Both video interleaving and progressive are supported.

C. MPEG-2

MPEG-2 was published as ISO standard 13818. MPEG-2 is typically used to encode audio and video for broadcasting signals, including digital terrestrial TV, satellite or cable. MPEG-2 with some modifications, is also the coding format used by commercial discs and DVD SVCD movies. MPEG-2 is similar to MPEG-1, but also provides support for interlaced video (the format used by the TV). MPEG-2 introduces and defines Transport Streams, which are designed to carry digital video and audio through unpredictable and unstable environments, and are used in television broadcasts. With some enhancements, MPEG-2 is also the current standard for HDTV broadcasts.

IV. Experimentation Scenario and Resources

In the present work we experimentally evaluated the performance of streaming of video stored on a WAN test bed.

A. Working Topology

Fig. 1 shows the working topology. It uses a PC as server and 9 PC's as clients. In this topology the links indicated with continuous line are of type FastEthernet with a transmission rate of 100 Mbps, while the links indicated with dashed line are dedicated with serial interfaces to a transmission rate of 2Mbps.

B. Hardware Resources of Experimentation

For the scenario shown in the topology of work we used desktop computers with the following features:

- Processor: AMD Athlon(tm) II X2 250 at 3GHz with 2GB of RAM
- Operating System: Windows 7 Professional 32-bit.

And

- Routers R1, R2, R3 and R4 were Cisco Model 2811
- Routers R5 and R6 were Cisco Multilayer Model WS-CS3750 switches.

Finally for the connection of the routers to the PCs Cisco Layer 2 Catalyst Model WS-2950-24 switches were used.

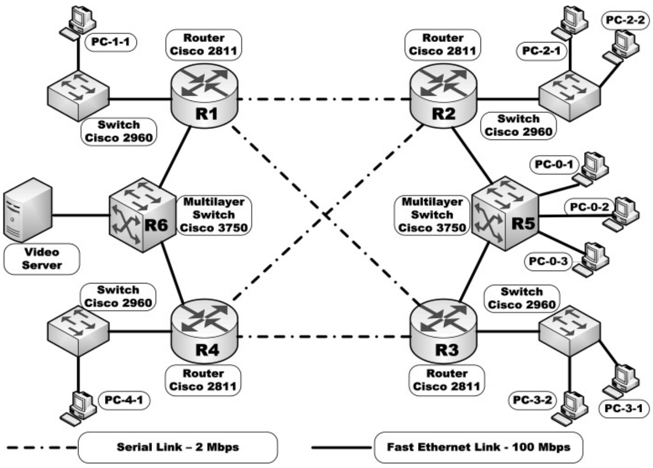


Fig. 1. View of the topology of the WAN test bed.

Our scenario assumes certain conditions, such as for example, the non-existence of voice traffic or general traffic (both, usually, in a very low proportion with respect to the video traffic), in order to facilitate the comparison and impact of different codecs on the multicast traffic.

C. Software Resources of Experimentation

For the experimental development Unreal Media Server [20] v.11.0 was used, and media player v.6.1 as streaming client software. It is a multi-protocol, high performance and small resources footprint software platform for streaming live and on demand audio video content over IP networks.

It streams with variety of streaming protocols to deliver content to Flash Player, Silverlight, Windows Media Player, Unreal Media Player, mobile devices and Set-Top boxes. The server supports UMS (proprietary, DirectShow-based, codec-independent) protocol for streaming to Unreal Media Player in unicast and multicast modes, and any multimedia file format, encoded with any codec.

Supported container formats include but are not limited to: MP4, ASF, AVI, MKV, MPEG, WMV, FLV, OGG, MP3, 3GP, MOV. Unreal Live Server supports any possible capture device attached to a PC.

Capturing network streams over RTSP, RTMP, MPEG2-TS, HLS and MMS protocols is supported as well. Unreal Live Server encodes / transcodes captured audio-video with H.264, VC1, AAC, MP3, WMA codecs and streams it over UMS protocol to Unreal Media Server.

Measurements were taken with Wireshark [21] sniffer software on the server and on each client PC, to obtain unicast and multicast frames for the metrics.

D. Codecs and video for Experimentation

We used three different codecs, those described in section III, and a commercial video [22] of 29 s configured with a resolution of 176x164 pixels.

V. Evaluating the Experimentation Scenario

The video was codified in the 3 codecs and was reproduced individually from the server. Catches were taken of traffic on the server as well as each client PC, in both configurations of traffic (dissemination and multicast). With these catches several metrics of benefits were identified, such as: performance, average interframe space, average frame size, and the count of the number of frames. Partial results concerning multicast were presented in [2]. In this paper, we present the results of subsequent studies in which the contrast between the unicast and multicast traffic for each one of the codecs selected on the server and the PCs have been analyzed in a better detail.

The results are presented in Table I.

TABLE I
DIRECT AND AVERAGE METRICS OF UNICAST AND MULTICAST TRAFFIC

Metrics	Traffic type	MPEG-4 AVC Codec	MPEG-4 Visual Codec	MPEG-2 Codec
Packet Number	Multicast	906	913	608
	Unicast	906	913	608
Video bytes [Mbytes]	Multicast	0,7041	0,3819	0,6952
	Unicast	0,7041	0,3819	0,6952
Average Packet Size [bytes]	Multicast	777,1015	418,1073	1142,9739
	Unicast	777,5655	417,6547	1139,9965
Average Interframe Spaces [s]	Multicast	0,03156	0,03164	0,04542
	Unicast in PCs	0,03119	0,03089	0,04476
	Unicast in Server	0,00375	0,00391	0,00616
Average Bit Rate [Mbps]	Multicast	0,1896	0,10191	0,18996
	Unicast	0,18955	0,18196	0,19002

Both direct and average metrics collected and the statistical distributions have very similar values, regardless of the location of the PCs and if the traffic is unicast or multicast (for the configuration set in the laboratory).

It also stresses that the metrics depend almost exclusively on the type of codec used, and not on the network configuration or where they were undertaken catches. The exception of these measurements is the average interframe space; there is a significant reduction in this unicast metric on the server, (approximately 83%) due to the fact that there is a traffic for each customer and more packets must be sent (Fig. 2).

In addition, we quantify the average, minimum and maximum delay of traffic, and the maximum jitter of unicast and multicast video from the server to the stations. These are very important parameters for the QoS. Fig. 3 and Fig. 4 show metrics in both cases. It is noted that the multicast traffic has a better behavior in all cases.

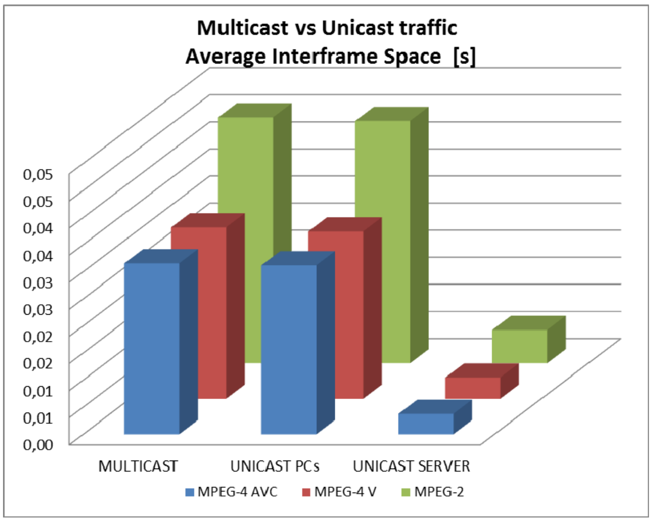
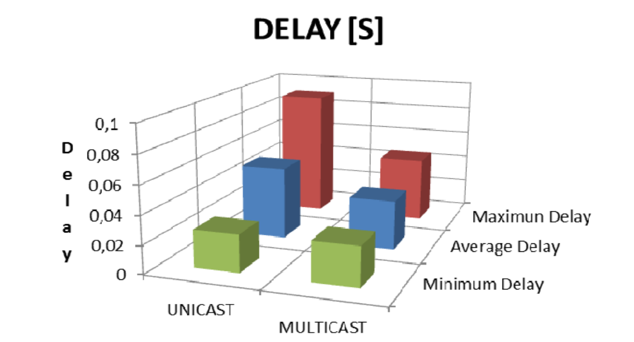
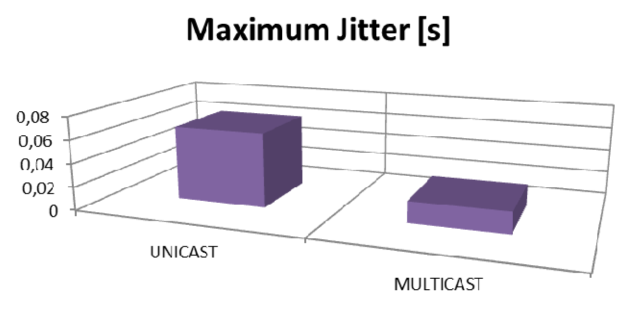


Fig. 2 Average interframe spaces in the unicast o multicast traffic



	UNICAST	MULTICAST
Minimum Delay	0,026402167	0,027813
Average Delay	0,051341086	0,034965219
Maximun Delay	0,090483083	0,04618225

Fig. 3 Average, minimum and maximum delay
in the unicast o multicast traffic



	UNICAST	MULTICAST
Maximun Jitter	0,064080917	0,01836925

Fig. 4 Maximum jitter in the unicast o multicast traffic

Finally, we also analyzed the statistical distributions of frame sizes and of interframe spaces for each codec by overlaying the responses of the codecs in a single figure (in blue the response of MPEG-4/AVC, in red of MPEG-4/V and green of MPEG2). These distributions depend exclusively on the codec used and not on the topology, equipment neither on the network configurations.

In Fig. 5, significant differences are observed in the distribution by frame size. MPEG-4/V presents a significant concentration of small frames. MPEG-4/AVC and MPEG2 have a concentration of large frames in the order of 1500 bytes. MPEG-4/AVC shows a lower concentration and the rest of the sizes distributed more intensely in different lengths. And in Fig. 6, we show differences observed in the distribution by interframe space. These distributions in unicast and multicast, on the side of the customer, are virtually equal and depend exclusively on the codec. MPEG-4/V presents a greater concentration of frames in the order of 50 ms. Again MPEG-4/AVC and MPEG2 share a common behavior, with a greater concentration of interframe spaces below 5 ms, and some peaks in times greater.

Finally, Fig. 7 represents the distribution for unicast at the server side. Note the concentration of frames in all codecs below 15 ms, in accordance with what is stated in Table I.

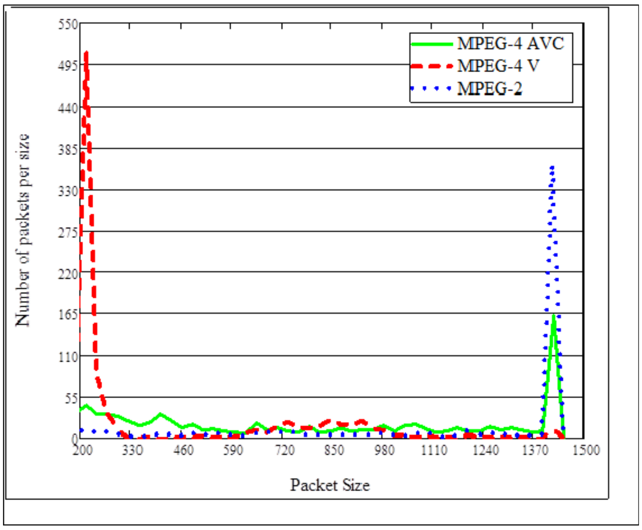


Fig. 5 Distribution of the packets grouped by size
for codecs for unicast or multicast traffic in PCs

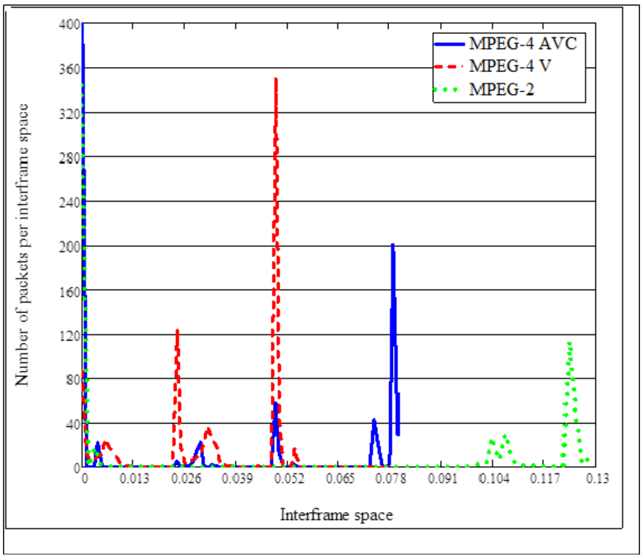


Fig. 6 Distribution of the packets grouped by interframe spaces by codecs
for unicast or multicast traffic in PCs

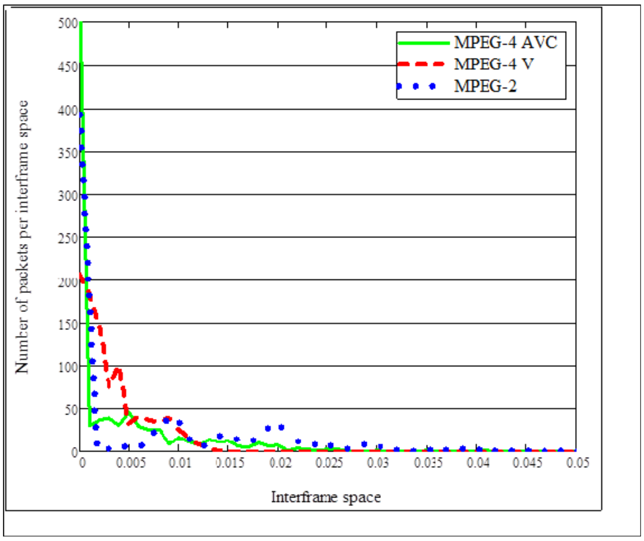


Fig. 7 Distribution of the packets grouped by interframe spaces by coded
for unicast traffic in the server

For the foregoing reasons, for the unicast traffic, it is noted that there is a commitment in the available bandwidth on the segment in which the server is located, and that depends on the number of end devices served. And in addition, in the available bandwidth on the WAN link that depends on the amount of flows that traverse it, according to the distribution of the PCs in the scenario, and the resolution to make routing protocols and multicast groups. In this experimentation, and with a proper configuration, the bandwidth of the WAN links was never committed.

VI. CONCLUSION AND FUTURE WORK

This study used a new WAN test bed as scenario offering metrics evaluation while using typical video codecs. Metrics were exhaustively analysed for unicast and multicast network traffic in line with the expectations of a real WAN.

Direct metrics and their averages, and the statistical distributions were quantified over real equipment.

Obtained figures show that the multicast traffic provides the QoS and the performance that is expected over each station using different types of codecs. Differences in the behavior of the multicast traffic are given by differences between codecs, and not by the multicast traffic in itself. Certainly, statistical analysis for this scenario shows that different codecs display a different behaviour in the distribution of packet lengths and of interframe spaces.

The impact on the overall traffic of a WAN link depends on the codec used, the number of clients to serve, and the applied configuration of multicast protocols of the network (OSFP and PIM). Logically, the unicast traffic is more sensitive than the multicast to the number of end users, especially in the segment where the application server is and where links or segments are shared for the clients.

We foresee future studies offering a quantitative behaviour evaluation at different 802.11 physical layers. These studies would precisely determine the best general network behaviour for higher Wi-Fi velocities. Finally, a new line of research could be carried out regarding the impact of queue length on maximum throughput for each codec.

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