

Modeling and Simulation of Multicast IPTV and VoD Streams for a Multiservice IP Network

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Abstract :

This thesis work concerns the modelling and simulation of IPTV and VoD streams in an IP/MPLS network. Video traffic and their protocols are studied. We started from the $M/G/\infty$ model which allowed us to have images (I, B and P). These are the images that were used during the modeling process. Indeed, this model allows to characterize a video sequence by the following parameters: Slot or time to generate a packet; image level or GOP. Image size or GOP distribution: Mixing distribution. Type of correlation: Markov, short-term dependence or long-term dependence.

The protocols involved are: MPEG-TS; UDP; RTP; RTSP; IP; IGMPv2; MPLS. In order to find a packet output model, we created a "modeler" in a python environment to estimate multicast and VoD traffic. We have created four (4) functions to materialize this modeling: "Packer", TS, Sequencer and Transport. The results collected allowed us to determine the law and mathematical model of packet output. And it is the $M/M/\infty$ model that models a video source. We used the RStudio software for this purpose. All these theoretical results could be compared during the simulation of the IPTV and VoD servers.

Keywords: ICT, MPEG-TS, IPTV, VoD, Streaming, modeling, python, simulation, VoD server, mathematical laws, mathematical models.

I. INTRODUCTION

With the rapid paradigm shift from circuit switching (traditional networks) to packet switching, centered on the IP protocol (TCP/IP), networked multimedia applications have created a huge impact on computing and infrastructure networks. In addition, great progress has been made in the compression of digital data (source and channel coding) and the rapid growth in Internet use, which has led to the emergence of multimedia applications in all-IP networks.

This has led multimedia content providers such as IPTV or VoD to have their own infrastructure to deliver it. However, these types of services have raised the issue of quality of service (QoS) for these multimedia applications. Especially the services deployed are real-time applications that have a strict need in terms of latency and jitter. The evaluation of QoS is often carried out by stochastic studies of the flows provided to the networks

but also by a simulation of the model chosen for its validation. The reliability of the results obtained will be based on the reliability of the mathematical model that will model the traffic generated.

In this work, we address the issue of modelling and simulation of IPTV and VoD flows in a best-effort IP/MPLS network. We are mainly interested in the behaviour of video sources (IPTV and VoD) and how video traffic is delivered to the network in order to develop a mathematical model governing the queuing of packets from MPEG codecs and their encapsulation by transport protocols (MPEG-TS, RTP, UDP, IP and Ethernet). It is the most suitable model for a source of multicast and video-on-demand (VoD) video traffic.

II. CONTEXT, RESEARCH METHODOLOGY AND MULTIMEDIA SERVICES

1. Context and problematic issues

Due to the rapid development of Information and Communication Technologies (ICT), the Internet has evolved from a simple data network to a network that has seen the emergence of a new set of services. These include the emergence of virtual networks, Voice over Internet Protocol (VoIP) technologies, the streaming of video on demand (VoD) over packet networks and the emergence of Internet Protocol Television (IPTV).

However, the diversity of services results in a diversity of needs in terms of QoS, while IP support is a best effort service support that does not guarantee the QoS of multimedia applications. The introduction of innovative mechanisms for QoS control and management allows, in conjunction with protocols (TCP, RTP...), to transport multimedia applications over IP [1].

Before deploying multimedia applications on IP/MPLS networks, we must assess, estimate and define the long-term traffic and QoS requirements for optimal resource utilization. To do this, it is more than necessary to model the flows generated by these applications. It is a question of providing real models, based on the different flows offered to the network by a traffic source.

It should be noted that this source of multimedia applications can be very simple with a constant or exponential throughput. On the other hand, it can produce bursts of packets that strongly influence network resources. Thus two approaches emerge to model these multimedia applications, namely, the modelling of flows following a trace and the second approach consists in looking at the behaviour of the source and the packet generation process [2].

We must point out that only three fundamental types of mathematical models are available to us [3]:

- Deterministic models that do not take into account the uncertainties of certain events. These models implement known data through accurate analytical relationships and do not integrate the probabilities of events that could be considered random;
- Stochastic models that integrate the hazards of probable events and provide an optimal solution in an uncertain future;
- Heuristic models that seek optimal solution procedures and only take into account possible research areas and leave it to the decision-maker to choose the final solution.

We have chosen the second approach, where we will place ourselves at the source and carry out the modelling. This choice was made because we decided to offer a decision-making model before any deployment operation in an operational network.

2. Objectives

The objective of our study is to find a mathematical model based on queue theory to take into account the elementary video streams (IPTV and VoD) but also their association mode and the procedures for managing and routing traffic to the IP/MPLS network.

Then, we simulate these flows to evaluate network traffic, as well as their QoS parameters in a python programming environment. This will involve applying real input values to the models and comparing the results with the values actually observed in reality (statistical simulation). Finally, we will evaluate the performance of the network with multicast and VoD traffic.

3. Research methodology

As video traffic is considered a multimedia service, it will then be a question of highlighting the video characteristics and constraints, processing the digitization process with compression techniques and then processing the way the packets are encapsulated to form the video stream. The codec output of the digitized video has randomness, hence the need to define video traffic as a random model [3].

The following approach was adopted:

- An analytical study that deals with the stochastic process of video traffic and the mathematical theories of queuing that model multicast video streams and video-on-demand (VoD).
- A simulation of the models obtained using python programming. This simulation will allow to estimate multicast traffic and VoD and then evaluate the QoS parameters of the flows.

4. Multimedia services (IPTV et VoD)

4.1 IPTV service

IPTV (Internet Protocol Television) is the provision of programs by streaming encoded video data as a series of IP packets. TV programmes are delivered to receivers (TV sets, tablets, and computers) through a broadband connection, instead of being delivered via cable distribution or conventional broadcasting. It uses MPEG-TS which is a standard for the transmission, multiplexing and synchronization of audio and video.

Video and program channels are delivered to receivers via a broadband connection (2 to 8 Mbps). Video streams are encoded into a series of Internet protocol packets and then sent over the

Internet to be received by anyone with a set-top box and a subscription for the service. IPTV is generally provided or delivered with VoIP and Internet access: it can therefore be described as a "Triple Play" service.

IPTV is provided by the service provider using a closed network infrastructure. The flows pass through an ATM or MPLS network before arriving at the DSLAM and then being directed to the customer (decoder, IP Television or PC). A set-top box (STB) connected to a television set allows the user to choose programming using interactive menus.

The protocols used by IPTV are:

- IGMPv2 (Internet Group Management Protocol): for signalling during channel synchronization;
- UDP (User Datagram protocol);
- IP (Internet Protocole): for routing and addressing packets;
- MPLS (Multi-Protocol Label Switching): provides QoS in the operator's networks.

4.2 VoD service

Video on demand (VoD) is a technique for broadcasting digital video content. VoD service is offered or sold over cable networks, such as the Internet or wireless networks, such as 3G or 4G telephony. VoD uses the principle of streaming, a principle based on unicast broadcasting, and is a logical evolution of broadcast technologies, from the pay-per-view style. More flexible for customers who do not have to work on broadcast schedules. Using a digital decoder or computer, the user can order movies or television programs stored on servers.

The protocols used by VoD are:

- RTP (Real Time Transport Protocol): for the transport and time-stamping of datagrams;
- RTSP (Real Time Streaming Protocol): for delayed TV features (pause, resume, fast forward);
- UDP (User Datagram protocol);
- IP (Internet Protocole): for routing and addressing packets;
- MPLS (Multi-Protocol Label Switching): provides QoS in the operator's networks.

The architecture of the VoD network is the same as that of IPTV.

Figure 1 shows the architecture of the IPTV network and VoD [1].

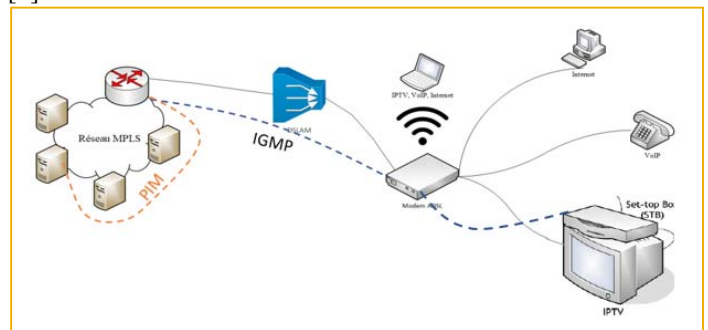


Figure 1: architecture of the IPTV network and VoD

III. VIDEO TRAFFIC AND MODELING

1. Video source coding and compression

A digital video consists of a sequence of digital images. Each image is formed by a matrix of pixels.

The human eye is only able to distinguish the refresh of images when the update rate is less than 25 fps (Frame Per Second). The fluidity of a video is characterized by its temporal resolution, i.e. by the number of frames per second (frame rate) expressed in fps (Frame Per Second). The latter is always greater than or equal to 25 fps and can reach 100 fps for some applications. In television uses 25 or 30 fps [1].

The coding algorithms can choose between inter-coding and intra-coding for the structural parts of each image [8]. Inter-coding uses kinetic vectors for internal block-based prediction to exploit statistical time dependencies between different images. Intracoding uses different spatial prediction modes to exploit spatial statistical dependencies in the source signal for a single image [10]. The residual prediction is then compressed using a transform to eliminate spatial correlation within the transformed block before it is quantified.

In a GOP: three types of images can be used: Images I for Intra, P for Predicted, B for Bidirectional interpolation. Each GOP starts with an image I.

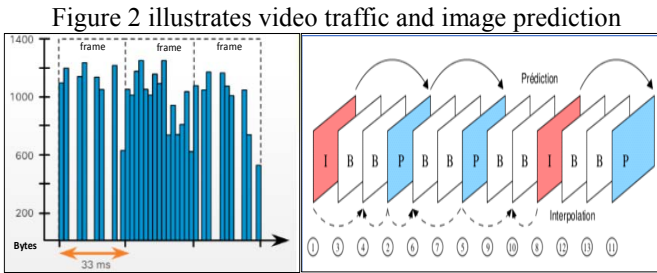


Figure 2: Video traffic and image prediction

2. Video traffic and random model

Video traffic as shown in Figure 2 is characterized by a number and size of video packets that vary with each frame (every 33 ms for a rasterization of 30 frames per second) and depending on the video content. Therefore, for each frame produced by the MPEG codec, the number of resulting packets and the size of these packets can each be associated with a probability law to describe their evolution over time.

Thus, considering all the frames generated by the video source, the number and size of the packets of the generated video stream are each associated with a stochastic process that allows to predict the statistics related to this traffic.

MPEG encoding also defines a hierarchy of "layers" (layers in Anglo-Saxon) within an image sequence. By following hierarchical levels, from the highest level, we distinguish the following categories:

- The sequence layer: random access unit;
- The image group (GOP): video coding. Random access point;
- The image layer: elementary coding unit;
- The slice: resynchronization unit;
- The macroblock layer: motion compensation unit;

- Block layer: an elementary image area subject to DCT coding.

It is the GOP layer (using the spatial and temporal redundancies that exist in video sequences is generated every 12/25 sec) that allows random access to the sequence with a first type I image. Thus we can note by $N(t)$ the number of images leaving the codec (figure below) at different time intervals.

We define a random variable X that counts 12 images = GOP so that $GOP = I B B P B B B B B B B B B B B B B B B B$. For each X counted, a GOP is associated. The law of probability is as follows:

$$P: X \rightarrow [0, 1]$$

$$x_i \rightarrow P(X = x_i) = p_i$$

$$\text{With } x_i = \text{GOP de rang } i \text{ et } p_i = \frac{12}{N(t)} \quad (1)$$

This property will be used to model video traffic using GoPs as traffic entities to model.

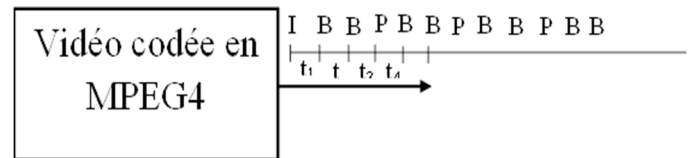


Figure 3 : MPEG 4 encoding image template

3. Procedure for coding and statistics of video traces

The study showed that video streams show a significant correlation between images or GOPs. MPEG traffic is divided over time by the GOP structure. The correlations are partly due to the periodic generation of GOPs. The idea is to recreate the observed generation using a correlated model generator. The $M/G/\infty$ process is very appropriate for recreating correlations between slots.

In MPEG, each slot represents the time required to generate an image (usually 1/25s or 1/30s) or a group of images (GOP) (generally 12/25s, 15/30s, 15/25s or 15/30s) depending on the codec used. The $M/G/\infty$ input model associated with a video sequence is therefore characterized by the following parameters:

- Slot or time to generate a packet; image level or GOP.
- Image size or GOP distribution: Mixing distribution.
- Type of correlation: Markov, short-term dependence or long-term dependence.

Due to its versatility and powerful correlation representation, we use the $M/G/\infty$ process to model video traffic using the same estimation techniques. However, the MPEG traffic structure is naturally divided into slots due to the periodicity of the generation GOP. We note that the GOP used is GOP12.

4. $M/G/\infty$ Model

The $M/G/\infty$ model represents a queue occupancy process with customers arriving according to a Fish law of parameter λ (packet generation with exponential inter-arrivals), a service law G (defined by its mean and variance) and an infinite number of servers.

The occupancy process noted X_n with $n = 0, 1, 2, \dots$ represents the number of clients in the system at the beginning of the time

interval $[n, (n + 1)]$. The resulting busy server process $(X_n)_{n \geq 0}$ is correlated but is not generally stationary. If we note $\sigma_{n,i}$ the service life of the second client in the system on date n , we must choose the initial parameters as follows for the $M/G/\infty$ process to start in the stationary mode:

- X_0 is the number of clients in the system on date $n = 0$, either distributed according to a Poisson distribution with a parameter $\lambda E(\sigma)$. λ being the parameter of the law M of the process $M/G/\infty$.
- $\forall i \geq 1$, the random variable $\sigma_{0,i}$ is independent and identically distributed with a probability function:

$$P[\sigma_{0,i} = k] = \frac{P[\sigma \geq k]}{E[\sigma]} \quad k=1,2,3, \dots \quad (2)$$

Under these initial conditions the occupancy process $(X_n)_{n \geq 0}$ checks the following properties:

- $\forall n \geq 0$, the random variable X_n is distributed according to a Poisson distribution with the parameter $\lambda E(\sigma)$.
- The correlation structure associated with the process $(X_n)_{n \geq 0}$ is entirely defined by the probability function σ (associated with the general law G of the process $M/G/\infty$).

Indeed, the autocorrelation function is given by:

$$\rho(k) = \frac{P[\sigma \leq k]}{E(\sigma)} \quad (3)$$

It follows from this that
$$\sum_{k=0}^{\infty} \rho(k) = \frac{1}{2} + \frac{E[\sigma^2]}{2E[\sigma]} \quad (4)$$

Thus the process $(X_n)_{n \geq 0}$ can generate a SRD (Short Range Dependence) or LRD (Long Range Dependence) behavior depending on the value of $E(\sigma^2)$. In other words, the occupancy process $(X_n)_{n \geq 0}$ is SRD if $E(\sigma^2) < \infty$ or LRD if $E(\sigma^2) = \infty$.

Conversely, the choice of the correlation structure makes it possible to characterize the distribution of service time. So we have:

$$P[(\sigma = k)] = \frac{\rho(k-1) - 2\rho(k) + \rho(k+1)}{1 - \rho(1)} \quad k=0, 1, 2, \dots \quad (5)$$

This property allows the definition of the service law of the $M/G/\infty$ process based on the estimated autocorrelation function of the data size over a time interval. The model of the autocorrelation function depends on the structure of the existing correlation. We estimate these correlations by an autocorrelation function $\rho(k)$ mixing two autocorrelation functions SRD and LRD defines as follows:

$$\rho(k) = \alpha \cdot \rho_1(\beta_1, k) + (1 - \alpha) \cdot \rho_2(\beta_2, k) \quad (6)$$

β_1 et β_2 are two strictly positive constants and $\rho_1(k) = e^{-\beta_1 \sqrt{k}}$ for the RDU component and $\rho_2(k) = (k + 1)^{-\beta_2}$ for the LRD component.

In addition, the marginal distribution of observed data sizes over time slots is generally better estimated with a mixture distribution $f(x)$ defined by:

$$f(x) = p \cdot f_1(x) + (1 - p) \cdot f_2(x). \quad (7)$$

Once the correlations between the images have been determined, the image sizes must be characterized. For this purpose, the Gamma and Lognormal laws were used because these two laws are very appropriate. Distribution mixtures (Gamma + Pareto) can also be used.

So we have $f(x) = p \cdot \text{lognormal}(\mu, \sigma) + (1 - p) \cdot \text{gamma}(\alpha, \beta)$

With p : the weighting of the distribution of the Lognormal function and $(1 - p)$ the Gamma one [6].

3.1.2.1 Modeling video sequences

Using the $M / G / \infty$ model, we target a general case where correlations can be both SRD and LRD and the slot size distribution is a mixed distribution. Thus we define a mixed correlation function $\rho(k)$ as follows:

$$\rho(k) = \alpha \cdot \rho_1(b_1, k) + (1 - \alpha) \cdot \rho_2(b_2, k) \quad (8)$$

The distribution function is:

$$f(x) = p \cdot \text{lognormal}(\mu, \sigma) + (1 - p) \cdot \text{gamma}(\alpha, \beta). \quad (9)$$

Remember that the $M / G / \infty$ process is a fish-based slot generation process. However, it is necessary to transform the fish distribution into a Lognormal-Gamma distribution.

Note by $F(x)$ the distribution function of the Lognormal-Gamma distribution. Thus, we have $F(x) = G(\text{Fish}(x))$. Where $\text{Fish}(x)$ is the distribution function of the fish distribution of the parameter slots $\lambda_{\text{poisson}} = \lambda E(\sigma)$. [B6]

Log normal-Gamma is defined by the distribution function:

$$f(x) = p \cdot f_1(x) + (1 - p) \cdot f_2(x) \quad (10)$$

With p : the weighting of the distribution of the Lognormal function and $(1 - p)$ the Gamma distribution.

Note by U the uniform random variable in the interval $[0, 1]$. Thus the general law $G(x)$ is determined according to the algorithm:

$$\begin{aligned} &\text{If } p < U \text{ then} \\ &G(x) = F_1^{-1}(x) \\ &\text{Otherwise} \\ &G(x) = F_2^{-1}(x) \end{aligned} \quad (11)$$

Where $F_1^{-1}(x) = e^{\Phi^{-1}(x) + \mu}$ is the opposite of the Lognormal distribution function.

Et $F_2^{-1}(x) = \frac{1}{\lambda} \Gamma^{-1}(a, \Gamma(a, 0)(1 - x))$ is the opposite of the Gamma distribution function. With

$$\Phi(x) = \int_{-\infty}^x \frac{1}{\sqrt{2\pi}} e^{-\frac{t^2}{2}} dt \quad \text{et } \Gamma(a, x) = \int_x^{+\infty} e^{-t} t^{a-1} dt \quad (12)$$

IV MODELING AND SIMULATION OF THE MULTISERVICE IP TRANSPORT NETWORK

1. Characteristics of IPTV

We will propose a model that will characterize multicast traffic, namely IPTV traffic. To do this, let us recall the protocols used by IPTV:

- MPEG-TS: is an MPEG standard that defines the transport aspects across networks for digital television.
- UDP (User Datagram Protocol).
- IP: packet addressing and routing.
- IGMPv2 (Internet Group Management Protocol version2): for signalling when tuning channels.
- MPLS (Multi-Protocol label Switching): ensures QoS in the operator's network.

2. Characteristics of VoD

In the same way as with IPTV, we will define the VoD traffic model. We recall the protocols that are used by VoD, these are:

- MPEG-TS: is an MPEG standard that defines the transport aspects across networks for digital television.
- UDP (User Datagram Protocol).
- RTP (Real Time Protocol): the transport and time-stamping of datagrams.
- RTSP (Real Time Streaming Protocol): for delayed TV functionalities (pause, resume, fast forward).
- IP: packet addressing and routing.
- MPLS (Multi-Protocol label Switching): ensures QoS in the operator's network

As we can see, both multicast and VoD have similar approaches when processing their packets, the only difference is in RTP encapsulations at the VoD level.

Hence we present the following algorithm that will allow us to simulate IPTV and VoD traffic. We will define the following functions:

- The "packer" function;
- The TS function ;
- The sequencer function;
- The transport function (where we will model UDP, RTP, IP and Ethernet encapsulations).

Figure 4 below illustrates the simulator where we see that it is composed of the following functions: "Packer", TS, Sequencer and Transport.

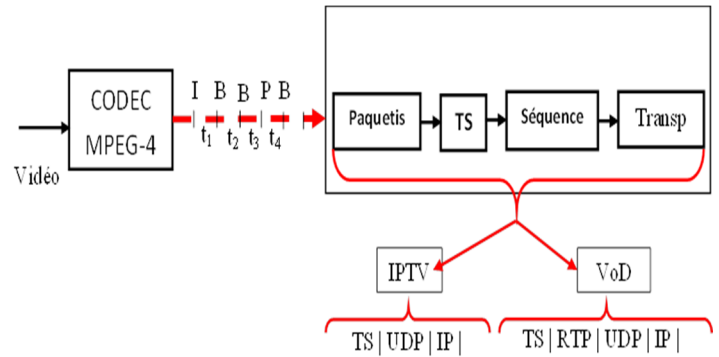


Figure 4: MPEG coding and transport process

3. Function « Packer »

In this function, the packets received are randomly divided. This function is modeled as follows:

```
// Fonction packetiser
ENTIER FONCTION packetiser (DONNER: ENTIER k, ENTIER moy,
1 ENTIER)
DEBUT_FONC
    VARIABLE alea, tab [], nPaquet [] : ENTIER
    VARIABLE i, j, som: ENTIER
    tab = []
    nPaquet = []
    alea = 0
    som = 0
    POUR i = 0 JUSQU'A 1 FAIRE
        alea = random(0....moy)
        tab[i] = alea
        som = som + alea
    FINPR
    POUR t = 0 JUSQU'A tab.length FAIRE
        nPaquet[t] = tab[t]*(moy/som)
    FINPR
    RETURN nPaquet
FIN_FONC
```

We have defined a random variable that represents the number of splits that the packet received from the source will undergo. It should be noted that this division is done randomly.

4. Transport Stream Function (TS)

A TS packet is 188 bytes. Each packet sent by the "packer" function will be split into 184-byte packets by the TS function. In case the packet sent by the "packer" function is smaller than 184 bytes, the TS function will process it by adding non-zero information called jam packets. After this processing, the packet will be accompanied by a 4-byte header. Hence we present the following algorithm:

```
// Function TS
ENTIER FONCTION ts (DONNEE : ENTIER p [])
DEBUT_FONC
    VARIABLE bourage, i, j, paquetTS[], allPaquetTS[], TS[]:
    ENTIER
```

```

bourage = 184
paquetTS = []

POUR j = 0 JUSQU'A p[i].length FAIRE
    SI (P[i] < bourage)
        ALORS
            paquetTS[i] = P[i] + (bourage - P[i])
        SINON
            paquetTS[i] = P[i]
    FINSI
FINPR
RETURN paquetTS
FIN_FONC

```

At the end, we count the total number of TS packets generated by the TS function.

5. Sequencer function

In this function, we ensure a group of 7 TS which will be conveyed by the transport function. We do this grouping because it is the maximum number of TS that can be encapsulated in an IP frame. In case the number of TS packets does not reach 7, it will be encapsulated because we are not at the limit of an IP frame. The function is characterized by the following algorithm:

```

// function sequencer
ENTIER FONCTION sequencer (DONNEE : ENTIER TS [])
DEBUT_FONC
    VARIABLE seq [], groupe, somTS, k : ENTIER
    groupe = 0
    somTS = 0
    k = 0
    POUR j = 0 JUSQU'A TS.length
        SI TS[j] <= 184
            ALORS
                groupe = groupe + 1
                SI (groupe < 8)
                    ALORS
                        somTS = somTS + TS[j]
                    SINON
                        seq[k] = somTS
                        groupe = 1
                        somTS = 0
                        somTS = somTS + TS[j]
                        k = k + 1
            FINSI
        SINON
            a = (TS[j]/184)
            POUR t JUSQU'A a FAIRE
                groupe = groupe + 1
                SI (groupe < 8)
                    ALORS
                        somTS = somTS + 184
                    SINON
                        seq[k] = somTS
                        groupe = 1
                        somTS = 0
                        somTS = somTS + 184
                        k = k + 1

```

```

FINSI
FINPR
FINSI
FINPR
seq[k] = somTS
RETURN seq
FIN_FONC

```

At the end we will have groups of 7 TS packets that will be one by one encapsulated by the UDP, RTP, IP and Ethernet protocols.

6. The Transport function

This function ensures that the service is the type of protocol encapsulation. Hence we made the service choice option (1: for IPTV, 2: VoD). The algorithm is as follows:

```

//function paquet Transport
// 1 --> IPTV
// 2--> VoD
ENTIER FONCTION paquetTrans (DONNEE: ENTIER seq [], ENTIER type)
DEBUT_FONC
    VARIABLE UDP, IP, RTP, i, seq[], r, a : ENTIER
    UDP = 8
    IP = 20
    RTP = 12
    ETH = 14
    TS = 28
    a = 0
    r = 0
    SI (type == 1)
        ALORS
            POUR i = 0 JUSQU'A seq.length FAIRE
                SI (i == 1228)
                    ALORS
                        a = i + UDP + IP + ETH + TS
                        pTrans.append(a)
                    SINON
                        r = i/184
                        a = i + UDP + IP + ETH + r*4
                        pTrans.append(a)
            FINPR
            RETURN pTrans
        SINON SI (type == 2)
            POUR i = 0 JUSQU'A seq.length FAIRE
                SI (i == 1228)
                    ALORS
                        a = i + UDP + IP + ETH + RTP
                        + TS
                        pTrans.append(a)
                    SINON
                        r = i/184
                        a = i + UDP + IP + ETH + RTP
                        + r*4
                        pTrans.append(a)
            FINPR
            RETURN pTrans
        SINON

```

RETURN 0
FINSI
FIN_FONC

7. Presentation of the results of the modelling

We note that we send empty TS between the intercomings of the images. Figure 5 shows the results of the transport function for multicast stream

1358.0	1358.0	1358.0	1358.0	1358.0	1358.0	1358.0	1358.0
1358.0	1358.0	1358.0	1358.0	1358.0	1358.0	1358.0	1358.0
1358.0	1358.0	1358.0	1358.0	1358.0	1358.0	1358.0	1358.0
1358.0	1358.0	1358.0	1358.0	1358.0	1358.0	1358.0	1358.0
1358.0	1358.0	1358.0	1358.0	1358.0	1358.0	1358.0	1358.0
1358.0	1358.0	1358.0	1358.0	1358.0	1358.0	1358.0	1358.0
1358.0	1358.0	1358.0	1358.0	1358.0	1358.0	1358.0	1358.0
1358.0	1358.0	1358.0	1358.0	1358.0	1358.0	1358.0	1358.0
1358.0	1358.0	1358.0	1358.0	1358.0	1358.0	1358.0	1358.0

Figure 5: transport function for multicast flow

We make the following observations:

- A 1358 packet size is equivalent to 7 TS (1316 bytes) + UDP (8 bytes) + IP (20 bytes) + Ethernet (14 bytes).
- However, the last one we can read on the capture is 418 bytes. It is equivalent to 2 TS (376 bytes) + UDP (8 bytes) + IP (20 bytes) + Ethernet (14 bytes).
- The transport function for VoD flow

1370	1370	1370	1370	1370	1370	1370	1370
1370	1370	1370	1370	1370	1370	1370	1370
1370	1370	1370	1370	1370	1370	1370	1370
1370	1370	1370	1370	1370	1370	1370	1370
1370	1370	1370	1370	1370	1370	1370	1370
1370	1370	1370	1370	1370	1370	1370	1370
1370	1370	1370	1370	1370	1370	1370	1370
1370	1370	1370	1370	1370	1370	1370	1370
1370	1370	1370	1370	1370	1370	1370	1370

Figure 6: Transport function for VoD flow

V INTERPRETATION OF SIMULATION RESULTS

1. Mathematical model of packet output

We collected the results of our simulation output on csv files. We remind you that the results are the packages and the time of the interarrivals.

To determine the mathematical law (see mathematical model), we will analyze the packets and the inter-arrival between the packets. Table 1 below shows the results of the interarrivals between the TS packets.

Inter-arrivée		Inter-arrivée		Inter-arrivée	
0,00199962	0,002644	0,002255	0,000173	0,000866	0,000218
0,00150108	0,002647	0,00202	0,001773	0,001273	0,001667
0,00200129	0,002642	0,002487	0,000411	0,000842	0,001657
0,00100064	0,002651	0,001787	0,001531	0,001051	0,001674
0,00200152	0,002664	0,002171	0,000649	0,001046	0,001673
0,00200105	0,002619	0,002544	0,00131	0,001432	0,001678
0,00200152	0,002653	0,001688	0,000873	0,000664	0,001673
0,00150108	0,002637	0,002318	0,001108	0,001359	0,001675
0,00150156	0,002644	0,001964	0,001955	0,000754	0,001685
0,00150085	0,002669	0,002534	0,000213	0,001049	0,001759
0,00250149	0,002627	0,001741	0,002721	0,001182	0,001595
0,00200176	0,002638	0,002216	0,003014	0,000919	0,001669
0,00150037	0,002631	0,002058	0,002877	0,001079	0,001688
0,00150013	0,00264	0,002501	0,002905	0,001022	0,001678
0,002002	0,002649	0,001774	0,002069	0,001489	0,001675

Table 1: Time of inter-arrivals between TS packets

To determine the mathematical laws that the results of our simulation follow, namely the output of the packets of the IPTV and VoD streams, we used RStudio software to perform our analysis.

RStudio software is a statistical software created by Ross Ihaka & Robert Gentleman. It is both a computer language and a working environment: commands are executed using instructions coded in a relatively simple language, results are displayed in text form and graphics are displayed directly in a window of their own. It is a clone of the S-plus software that is based on the S object-oriented programming language, developed by AT&T Bell Laboratories in 1988. This software is used to manipulate data, plot graphs and perform statistical analyses on these data.

We used VoD data because the analysis of multicast traffic results also gives the same result of the packet output law. We also processed this analysis with the R software. The results are as follows:

The average m is the parameter λ the average arrival rate.

EnvironmentHistoryConnections

Import Dataset

List

Global Environment

Data

donnee

542075 obs. of 1 variable

values

m

0.00189866783474427

Figure 7: average arrival rate

First, we present the graph of the simulation of inter-arrival times.

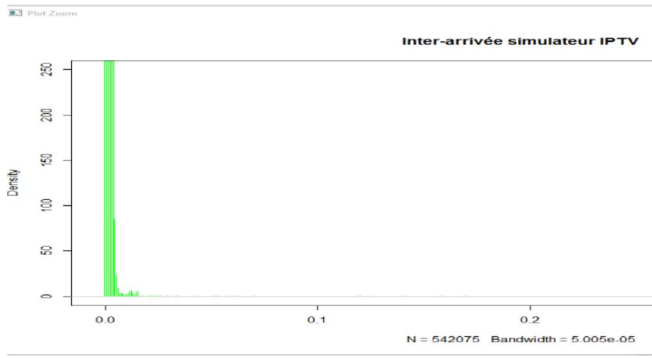


Figure 8: Inter-arrival graph

Then we present the one of the exponential function.

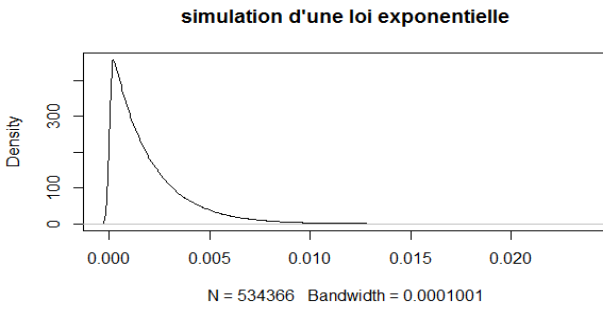


Figure 9: exponential function graph

Overlay of the two graphs for confirmation or not.

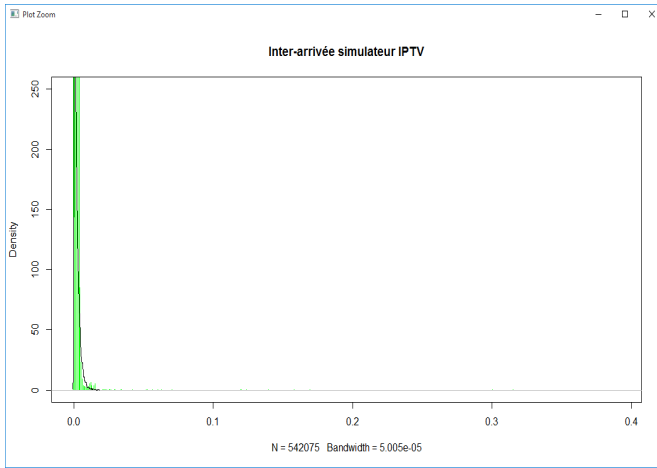


Figure 10: Superposition of the two graphs

We have just verified through this superposition of the two graphs that the inter-arrivals follow an exponential law.

2. Mathematical model of the source

The packets from the simulator have Poisson distribution inter-arrivals (exponential inter-arrival times). This source has the queue nomenclature M/M/∞ with the property of no memory ("Markovian" property).

We saw to model the video, first the source was modeled with the M/G/∞ model and then go through a transformation of the Poisson distribution into a Lognormal-Gamma distribution. We can conclude that this source can be modelled directly by the M/M/∞ model. We present the study of the M/M/1 queue with the output discipline S, a single server.

The M/M/1 queue is the one with Markovian arrivals and services with a single station and infinite capacity and FIFO (First In First Out) discipline.

The performance study of such a queue consists in calculating the characteristics:

L_s : the average number of clients in the system;

L_q : the average number of customers in the queue;

W_s : the average duration of a client in the system;

W_q : the average duration of a client in the queue;

μ : the service rate or average number of servers used;

$1/\mu$: the average duration of a service;

λ : the average arrival rate or the number of clients per unit of time;

P_n : the probability of having n clients in the system;

P_0 : the probability of having 0 clients in the system.

$$P_n = \left(\frac{\lambda}{\mu}\right)^n P_0 \quad (13)$$

$$\text{As in } \sum_{n=0}^{\infty} P_n = P_0 + P_1 + \dots + P_{\infty} = 1 \quad (14)$$

$$\text{Si } \frac{\lambda}{\mu} < 1 \quad \text{we find } P_0 = 1 - \frac{\lambda}{\mu}$$

Based on the above:

$$L_s = \sum_{n=0}^{\infty} n P_n = \frac{\lambda}{\lambda - \mu} \rightarrow L_s = \frac{\lambda}{\lambda - \mu} \quad (15)$$

$$L_s = \lambda W_s \rightarrow W_s = \frac{1}{\lambda} L_s$$

$$W_s = \frac{1}{\lambda - \mu} \quad (16)$$

$$L_s - L_q = \frac{\lambda}{\mu} \rightarrow L_q = L_s - \frac{\lambda}{\mu} = \frac{\lambda}{\lambda - \mu} - \frac{\lambda}{\mu}$$

$$L_q = \frac{\lambda^2}{\mu(\mu - \lambda)} \quad (18)$$

$$L_q = \lambda W_q \rightarrow W_q = \frac{1}{\lambda} L_q$$

$$W_q = \frac{\lambda}{\mu(\mu - \lambda)} \quad (19)$$

The parameters λ and μ of our simulation are as follows:

λ is 0.0018986678783474427

Determination of the μ parameter. To do this we collected the packet processing time which is 12.450520 ms and the total number of packets is 7713. Let's calculate the $1/\mu$ which is the average length of service.

$$1/\mu = \frac{12,450520}{7713} = 0,00161422 \text{ ms}$$

VI CONCLUSION

The development of the Internet network and the diversification of the services offered make it an extremely complex network. One of the most complex aspects will be the use of the Internet network to transport mass flows with critical service qualities such as the broadcasting of video films. The design of such networks requires the development of methodologies and modelling tools to accurately assess its performance.

Today, the use of multimedia services such as VoD, IPTV, Audio Streaming, and Voice over IP is increasingly increasing. The end user of these services expects his content provider to provide him with quality services and based on good mathematical modelling of the different traffic sources. It is in this context that we decided to conduct our study. The objectives have been achieved. It was a very enriching experience and a real opportunity to put into practice a body of knowledge acquired during our training.

The definition of the inter-arrivals of the images (I, B and P) allowed us to choose the $M/G/\infty$ model to code these images (I, B and P). This facilitated the modeling of the video traffic source through our modeling model which includes the following functions: "Packer", TS, Sequencer, and Transport. From these functions, we collected the processed data to determine the mathematical model of packet output.

We used the RStudio software, which is a statistical tool to define the mathematical laws that govern the correlations between packets. The time of packet arrivals follows an exponential law, and this has facilitated the determination of the mathematical model of packet output which is the $M/M/\infty$ model. This confirms that the video source can be modeled by the $M/M/\infty$ model.

In terms of perspective, the data, which we used, were coded with GOP 12. However, this data can also be encoded with GOP 15, GOP 25 or GOP 50, what would be the output model of the source packets, if the size of the GOP is other than 12. When coding the images, we also used the model where the B images were also coded. The modelling work can be repeated with a GOP containing only I and P images and evaluate the mathematical model that suits the source.

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