MODELING THE PERFORMANCE OF THE COMPUTER NETWORK APPLIED TO TELECOMMUNICATION

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ABSTRACT

This paper is the presentation of a mathematical modeling of the use of computer networks at the end of the main use of telecommunication, as we can see below telecommunication is at the origin the transmission of the voice and currently it is in the computer networks that this transport is done by the technology of the voice over IP the latter having a lot of parameters for a better service starting with the temporal constraint and the different parameters of the internet. These services do not offer guarantees without having to study and model their behavior..

Keyword : - *network,VoIP,mathematics,TCP,UDP*

1. INRODUCTION

The main use of computer applications in the field of telecommunication is VoIP , the latter having a strong time constraint are not supported by the Internet. The service offered by the latter does not offer a guarantee in terms of quality of service. Although widely used, the TCP protocol does not solve all the constraints of the voice because of its lack of flexibility vis-à-vis the latency. All these reasons imply that IP telephony should know how to operate dynamically with the non-connected mode offered by UDP. To measure the performance of VoIP with respect to these two protocols, performance parameter measurements such as packet loss, jitter, latency, and throughput are required. These points will be treated in this part.

2.IP Traffic Modeling

1.1 Model M / G / ∞

The model M / G / ∞ represents a process of occupation of a queue with clients arriving according to a Poisson distribution of parameter λ [7] (packet generation with inter-exponential arrivals), a law of service G (defined by its mean and its variance) and an infinite number of servers.

The occupation process noted with X_nn = 0,1,2, 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≥0is correlated but not stationary in general. If we note the service life of the i th client in the system at the date n, we have to choose the initial parameters as follows for the M / G / ∞ process to start in the steady state :

- X_0 which is the number of clients in the system at the date n = 0, is distributed according to a Poisson distribution with a parameter λ
- ∀ i, the random variable $\sigma_{0,i}$ is independent and identically distributed with a probability function :

$$P\left(\sigma_{0,i} = k\right) = \frac{P(\sigma \geq k)}{E(\sigma)}$$

These initial conditions the process of occupation (Xn) n≥0 verifies the following properties :

- ∀ n 0, the random variable is distributed according to a Poisson distribution with the parameter λE (σ).
- The correlation structure associated with the process (X_n) n≥0 is fully defined by the probability function (associated with the general law G of the M / G / ∞ process).
  Indeed, the autocorrelation function is given by :

$$p(k) = \frac{P(\sigma \geq k)}{E(\boldsymbol{\sigma})}$$

It follows that :

$$\sum_{k=0}^{+\infty} p(k) = \frac{1}{2} + \frac{E(\sigma^{2})}{2E(\boldsymbol{\sigma})}$$

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

$$p(\boldsymbol{\sigma} = k) = \frac{p(k+1) - 2p(k) + p(k-1)}{1 - p\left(\sigma_{0,i} = k\right)}$$

We estimate these correlations by an autocorrelation function p (k) mixture of two autocorrelation functions SRD and LRD defines as follows :

$$p(k) = \boldsymbol{\sigma}\rho_1(b1, k) + (1-\alpha).\rho_2(b2, k)$$

b1 and b2 are two strictly positive $\rho_1(k) = e^{-b1*\sqrt{k}}$ constants SRD and $\rho_2(k) = (k+1)^{-b2}$ for the component and for the LRD component.

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

$$f(x) = p * logn(\mu, \sigma) + (1-p) * gamma(\alpha, \beta)$$

1.2 Traffic Modes for Multimedia Applications.

An audio application can be modeled by a Markovian process with two states ON and OFF with a constant packet emission rate λ =1/T during the period ON (T is the inter-arrival of packet) [7] [13]. The measurements show that the transition ratio between the two states is exponentially distributed.

The cumulative distribution function (CDF) of the inter-arrivals of packets can be written :

$$1 - F(x) = \begin{cases} 1 \ pour\ 0 \leq x \leq T \\ (1-p)e^{\frac{-x-T}{Toff}} \ pour\ x \geq T \end{cases}$$

We are interested in this same distribution in the case of the superposition of N homogeneous application. The CDF function of the inter-arrivals of packets in this case takes the form :

$$1 - Fn\left(\frac{x}{N}\right) = \begin{cases} 1 - \lambda\frac{x}{N})^{N-1} \ pour\ 0 \leq x \leq T \\ \dfrac{1-p)e^{\frac{-x-T}{Toff}}}{\frac{T}{TOFF} + 1 - P} \ pour\ x \geq T \end{cases}$$

We can see that when N goes to infinity, we have $1 - Fn\left(\frac{x}{N}\right)$ which tends towards $e^{-\lambda x}$.

Based on this property, we generate an equivalent exponential law with a bit rate given by :

$$A_{ia} = \frac{1}{N}\sum_{i=1}^{N} \frac{\lambda_{ia,i}}{ni}$$

The packets are generated according to a discrete distribution Ps of packet size given by :

$$Ps\ \{Ps_i, i = 1 \dots N\}$$
$$\Pr(Ps = Ps_i\ ) = \frac{n_i\ \lambda_{ia,i}}{\sum_{i=1}^{N} n_i\ \lambda_i\ a, i}$$

1.3 Differential Analytical Modeling of TCP / IP

The model developed by New Reno is an analytical model describing through equations the main characteristics of the network over time [7]. To do this, he began by observing the actual traffic of the TCP stream.

Flow rate λ (t) of the TCP source.
CWDN (t) size of the congestion window.
Number of packages in each router,
Trip Time RTT Router (t).

1.4 Lossless Slow Start Phase

Let RTT (t) be the Round Trip Time at time t, Let λ (t) be the bit rate of TCP source and CWDN (t) be its congestion window at time t, we have :

$$\frac{d\lambda(t)}{dt} = \begin{cases} \frac{\lambda(t)}{b.RTT(t) + k(t)\,\lambda(t)}\ si\ \lambda(t) < (b+1)\ \frac{\mu_{min}}{b}\ with\ \lambda(0) = 1 \\ 0\ \ if\ else \end{cases}$$

$$\frac{d\lambda(t)}{dt} = \begin{cases} \frac{\lambda(t)dRTT(t)}{dt} + \frac{1}{b})si\ \mathrm{CWND}(t) < (b+1)\ \frac{\mu_{min}}{b}\ RTT(t) \\ 0\ \ if\ else \end{cases}$$

Avec CWND(0)=1

Ou : $k(t) = \frac{RTT(t)}{2(b+1)}.\frac{1}{\mu_{min}}$

1.5 Phase Congestion Avoidance without losses :

$$\frac{d\lambda(t)}{} = \frac{1}{}\ \ avec\ \lambda(0) = \lambda(\text{previous step})/2$$

$$\frac{dCWND(t)}{dt} = \frac{1}{b.RTT(t)} + \lambda(t).\frac{dRTT(t)}{dt} \tag{1.01}$$

With CWND(0) = CWND(previous step)/2

1.6 The simplified load equation of the buffer i

$$\frac{\mathrm{dni(t)}}{dt} = ai(t) - di(t)\ ou$$

$$ai(t) = \begin{cases} \boldsymbol{\lambda}\mathrm{i}(\mathbf{t})\ \ \mathrm{si\ ni(t)} < ci \\ 0\ si\ ni(t) = Ci \end{cases}$$

$$di(t) = \begin{cases} \boldsymbol{\lambda}\mathrm{i}(\mathbf{t})\ \ \mathrm{si\ ni(t)} < 1 \\ \mu i\ si\ 1\ \leq \mathrm{ni(t)} \leq Ci \end{cases}$$

Let TA (t) be the end-to-end transit time of the packet

$$TA(t) = \sum\nolimits_{i=1}^{n} \frac{ni(t)}{\mu_i} + delay_i$$

Let TR (t) be the time that the acknowledgment packets (ACK) put back from the destination node to the sending node.

$$TA(t) = \sum\nolimits_{j=1}^{n} \frac{nj(t)}{\propto \mu_i} + delay_j$$

- Round Trip Time (RTT) :

$$RTT(t) = TA(t) + TR(t)$$

3. MODELING THE VOIP PERFORMANCE EVALUATION TECHNIQUE

The main factors that affect the performance of VoIP are the following parameters [12] [13]

- Loss of packets : collisions, errors, UDP
- Latency (end-to-end delay)
- Jitter
- The flow.

1.7 Loss of packets

Packet loss comes from the fact that some packets are lost by routers or switches when they are congested. Indeed, from the moment when the buffer of a network node is full, the new incoming packets are automatically discarded temporarily.

The following figure shows an illustration of this fact :

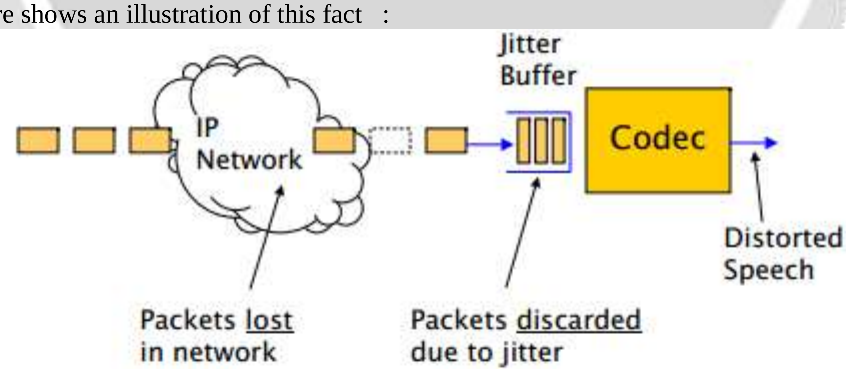

Fig -1 Packets lostt

When the buffers of the different IP network elements are congested, this makes it possible to send an implicit signal to the TCP terminals which decrease their bit rate in proportion to the negative acknowledgments sent by the recipient. Unfortunately, for voice packets, which are transported over UDP, no mechanism for controlling the flow or retransmission of lost packets is offered at the transport level. One of the solutions established is the implementation of the RTP and RTCP protocols which make it possible to determine the packet loss rate, and to act accordingly at the application level.

In Gilbert's model [7], we note the possession of the two states of the transmission channel, one with which the transmission is perfect, and the other with which the probability of error is equal to d. The simplified model (Figure 1) [7] eliminates this probability of loss. Each packet is lost when the string is in state 1. Comparisons were made with respect to [24], and this allowed to finely simulate the packet loss of the network.

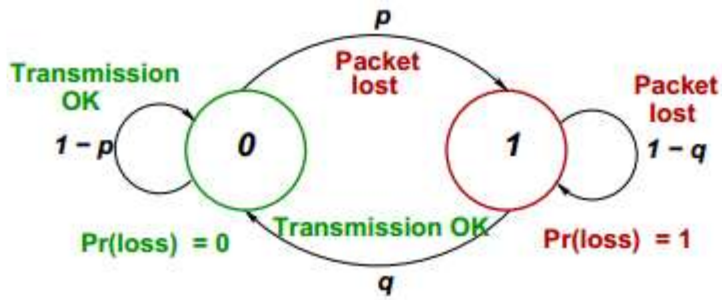

Fig -2 Complex modeling of packet loss

For the figure above, when one is in state 0, the transmission is without error. In state 1, all packets are lost, the transition from state 0 to state 1 constitutes the loss of a quantity of data.

suppose that :

The successful arrival is encoded by " 0 "And the loss by" 1 ".

Suppose $X\ n$ is the result of n packet transmission with $n \geq 1$

The sequence X = Xn, $n \geq 1$ is a discrete stochastic process of discrete time Markov with {0,1} value.

The following formula denotes this hypothesis.

$$\Pr(Xn_{+1}=1 \mid X_n=0) = p,$$

$$\Pr(Xn_{+1}=0 \mid Xn = 1) = q.$$

With, p the probability of losing a packet when the previous transmission was correct.

And 1-q, the probability of losing a packet, when the previous transmission was not correct.

Gilbert's model also assumes the following notation:

With LBS the size of the lost packet stream, we then have the formula:

$$\Pr(LBS = k) = (1 - q^{k-1})\, q,\ k \geq 1$$

- Si MLBS (Most Lower Bit Size), we have : MLBS = q^{-1}

Noting TEB, the bit error rate, we have the followi

$$TEB = \Pi_0 p + \Pi_1(1-q),$$

With $((\pi 0, \pi 1)$, the stationary distribution of X. we have $\pi 0 = q(p+q)^{-1}$ et $\pi 1 = p(p+q)^{-1}$,

$$TEB = \frac{q}{p+q}p + \frac{p}{p+q}(1-q)$$

$$TEB = \Pi 1$$

With this formula we deduce that:

$$p = q\,\frac{TEB}{1-TEB} = \frac{1}{MLSB}\frac{TEB}{1-TEB}$$

We can now calibrate the model. The bit error rate can then be measured with the average loss rate :

$$p = \frac{1}{MLSBm}\frac{TEBm}{1 - TEBm}\ et\ \ q = \frac{1}{MLSBm}$$

We can therefore conclude that if there are losses (at least one) and if all the packets are not lost, then necessarily :

$TEBm$ > 1 et 0<$TEBm$<1, ce qui implique 0<p,q<1.

4. NOTION ON QOS IN VOICE OVER IP

The quality of service corresponds to the possibility of transporting in the best possible conditions all types of information that do not have the same constraints.

It is mostly used for networks carrying voice or video type data, so that this type of traffic has priority over computer data.

4.1Setting up

QoS applies to OSI Level 3 equipment. To differentiate the traffic, the IP (type of service) ToS field is used, as does the Diffserv model.

The implementation of the QoS consists first of all in differentiating the network traffic by carrying out a recognition of the packets passing through :

- By IP
- By level 3 or 4 protocol (TCP, UDP, ICMP, ...)
- By port.
- By date and time.
- Depending on the congestion.
- Depending on the bandwidth.
- Depending on the latency time.

Most of the time is set up traffic smoothing, which reserves a certain amount of bandwidth to each service and avoids too much bandwidth. The Token Bucket or Leaky Bucket algorithms are used for this purpose (token bucket or pierced bucket).

5. CONCLUSION

In our research, we were able to mathematically model the performance evaluation of computer networks that use VoIP technology. This modeling starts with IP traffic, then the multimedia traffic and the different performance parameters of VoIP.

Thus we have been able to deduce some mathematical formulation of the problems related to the parameter of performance of these networks to finish with the quality of service perceived by the user. The latter is very important to have an approximate value of the quality of the offers at the level of the users of the Internet, one can conclude that the VoIP is difficult to put in place and that the architectures necessary need to be well designed so that the quality of service is obtained at a very low cost for an IP convergence of telecommunication.

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