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Modeling and Characterization Traffic Voice, Video, Data and Telemetry under Pareto Distribution-Oriented Networks have on Power Line Communications

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Abstract

Background/Objectives: To date, it has not found a traffic model that allows characterizing different kinds of service, which describe one self-similar behavior HAN (Home Area Network) networks on PLC (Power Line Communications). The aim of the article is to propose a model to characterize traffic classes such as voice, video, data and telemetry, describing a behavior autosimilar supported on the Pareto distribution. **Methods:** The Pareto distribution is a distribution type that is well aligned with autosimilar type of traffic. Pareto distribution is used as a source of network traffic, which has a bursty behavior, which can be repeated continuously within the network. To estimate the parameters describing the Pareto distribution for each class of service, a theoretical and experimental analysis was performed to determine the minimum rate, average and maximum, required to describe a source of real traffic. **Topic Relevance:** Although there have been various PLC networks related to work, had not seen a traffic model that allows to represent different kinds of service HAN networks. Aspects that can be considered useful in scenarios requiring evaluate aspects related to QoS (Quality of Service). **Results:** The proposed model allowed mathematically adequately represent different kinds of service, looking for the best estimate of the self-similar behavior present in the networks have and model the behavior that packets may occur within the PLC network under an innovative multi-multiclass stage. **Application/Improvements:** The proposed model can be used in practical and research scenarios as a strategy to evaluate the performance of networks and estimating the bandwidth required for various classes of service, in order to provide adequate levels of QoS during the provision of services such as voice, video, data and telemetry. It is suggested to evaluate other probability distributions, with the aim of identifying new strategies for modeling the self-similar traffic.

Keywords: Multiservice Network, Pareto Distribution, Power Line Communications, Quality of Service, Service Classes, Traffic Model

1. Introduction

A traffic model corresponds to a mathematical model describing the demand that users of a communications network impose on network resources. Due to the unpredictable behavior of each of the sequences of traffic that can be generated, these modeled as stochastic processes, and generated models should represent the main features traffic statistics regarding their impact on network performance[1].

Over time they have proposed various traffic models, which have been quite successful so far. An example is the

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Poisson model whose assumptions were widely validated in various types of networks, enabling the development of design methods in order to ensure specific quality of service. This model considers arrival rates to exponentially distributed intervals with total independence between them, which allows use of a fairly easily in the analysis and design of communication networks[2]. Due to the success presented by the model Poissonian telephone networks during the 70s, it was applied in the context of data networks with equal success, even though his predictions were not as accurate[3].

However, due to the emergence of various types of services such as voice and video over IP, which are articulated together with data on the same network infrastructure, it has considered the need to define an alternative model to model Poisson fit better shape the type of traffic circulating in current communication systems. In view of the above, a new model called Markov Modulated Processes arises[4]. The voice can be represented as a Markovianamente modeling deterministic process (MMPP/ Markov Modulated Poisson Process), which, through the use of two states (ON, OFF) indicate the active off state (speech) or (unspecified) of this type traffic generated by the source[5]. On the other hand, traffic on a LAN network where data and sessions sporadically generated, can also be modeled by a Markov chain of two states, where each state arrival rates are Poisson and with different values intensity that can be described by a MMPP[6].

The MMPP describe more accurately the correlation between different types of traffic in the case of the Poisson model. MMPP model has been helpful in the traffic modeling video with a variable bit rate, as with the MPEG2 format in which the transition between types of frames I, P and B represent a Markov chain, each with its respective average rate Poisson arrival rate[7]. Other recent models include continuous flow schemes which have been very useful in cases where each piece of information (packets, bits, etc.) represents an infinitesimal load capacity for the transmission means. In[6] several studies where it is shown that the autocorrelation real rates of traffic declines hyperbolically over time, indicating that the arrival rate in an instant of time may depend on occurrences both long-time and short periods presented.

Self-similar traffic models were first reported in 1994 in the work of Leland[8]. Where the foundation for the study of this phenomenon were established and has since been demonstrated in various contexts supported in IP networks modern low using various transmission models[9]. Although there are a variety of traffic models, it can only be considered correct if the statistical inference techniques used on such traces of actual traffic indicate that these are consistent with the selected model. It is important to note that the MMPP model only considers an autocorrelation between intervals very close together time, which allows process analysis and design a simpler way to self-similar model, which favors preference. However, with increasing time scales, the autosimilar modeling shows greater accuracy[10].

2. Random Self-Similar Processes

Assuming a random discrete time process X(n), which generates $\{X_0, X_1, \ldots, X_n\}$. Let X^m be an aggregate random process whose sampling values are:

$$X_0^{(m)} = \frac{1}{m}\left[X_0 + X_1 + \ldots + X_{m-1}\right]$$

$$X_1^{(m)} = \frac{1}{m}\left[X_m + X_{m+1} + \ldots + X_{2m-1}\right]$$

$$X_2^{(m)} = \frac{1}{m}\left[X_{2m} + X_{2m+1} + \ldots + X_{3m-1}\right]$$

$$\vdots$$

The random process is self-similar if it satisfies the following properties:

- The processes X and X^m are related by:

$$X^{(m)} = \frac{1}{m^{(1-H)}} X$$

  Where H is the Hurst parameter which can assume values between 0.5 and 1.
- The average value of X and X^m are equal:

$$E\left[X\right] = E\left[X^m\right] = \mu$$

- Autocovariance functions of X and X^m are equal:

$$E\left[\left(X(n+k)-\mu\right)X(n)-\mu\right)\right] = E\left[\left(X(n+k)^{(m)}-\mu\right)\left(X(n)^{(m)}-\mu\right)\right]$$

When the value of H → 1 increases the degree of self-similarity as well as the long-term dependence. On the other hand if H → 0.5 decreases the degree of self-similarity resembling a Brownian movement to a greater degree.

The distribution of heavy tail traffic comes from showing long-term dependency, such as in video traffic. This type of distribution has the following characteristics:

- It has a very high or infinite variance.
- The cumulative distribution function (CDF/Function Cumulative Distribution) has the following property:

$$1-F(x)=p(X>x)\sim\frac{1}{x^{b}}\quad x\rightarrow\infty,0<b<2$$

Where b is the shape parameter

3. Pareto Distribution

The Binomial and Poisson distributions have traditionally been used in traffic models for communications networks, where it is assumed that the nodes transmit low transmission rates steady. In[11] it mentioned that these models may be inadequate because in some cases traffic may present a behavior burst form, i.e. during a period of time can have high rates of packet transmission and in another period of low time transmission rates. Traffic describing this type of behavior is called "self-similar traffic"[12].

Traffic in communications network environments is similar to distributions that have a very high variance. In some cases, these distributions are known as "heavy tail", which can have large values in the probability distribution function (PDF/Probability Density Function) for x very far from the mean μ. In view of the above, this type of distribution is best suited to the density function for a data rate generated by a burst source[10].

Pareto distribution is a type of distribution is properly adjusted to the type of self-similar traffic.

The PDF for a Pareto distribution is:

$$f(x)=\frac{ba^{b}}{x^{b+1}}$$

Where a is the position parameter with $a < x < \infty$, b the shape parameter with $b > 0$.

Your CDF function is:

$$F(x)=1-\left[\frac{a}{x}\right]^{b}$$

The mean and variance values are: x

$$(\text{Average})\,\mu=\frac{ba}{b-1}$$

$$\sigma^{2}=\frac{ba^{2}}{(b-1)^{2}(b-2)}\quad(\text{Variance})$$

The variance value is only significant for $b > 2$.

The likelihood function is given by:

$$p(X>x)=\left[\frac{a}{x}\right]^{b}$$

Pareto distribution is used as a source of network traffic, which has a bursty behavior, which can be repeated continuously within the network.

Setting an appropriate value of b can satisfy all conditions defined by a heavy tail distribution, where:

$$p(X>x)=1-F(x)=\left[\frac{a}{x}\right]^{b}$$

To describe a real source, it is necessary to consider the following parameters:

- λ_{min}: Minimum arrival rate data.
- λ_{a}: Average arrival rate data.
- σ: Maximum data rate generated by the source.

Considering the above, a real source with Pareto distribution, can be described by describing the flow and description interarrival time.

4. Description Flow

This option allows you to specify the randomness of instantaneous data rate produced by the source. Start the process with the pdf for instantaneous data rate, which presents as follows[13]:

$$f_{\lambda}(\lambda)=\begin{cases}0 & \lambda<\lambda_{min}\\ \dfrac{ba^{b}}{x^{b+1}} & \lambda>\lambda_{min}\end{cases}$$

Where a is the position parameter and b is the shape parameter, which determines the slope of the exponential curve. The expressions that allow calculating parameters a and b are:

$$a=\lambda_{min}$$

$$b = \frac{\lambda_a}{\lambda_a - \lambda_{min}}$$

5. Description of Time Between Arrivals

This option allows you to specify the randomness of the periods between adjacent packages produced by the source. The pdf of an interarrival time following a Pareto distribution, description can be expressed by:

$$f_T(t) = \frac{ba^b}{t^{b+1}} \quad a \le t < \infty$$

A real source can be described under the use of the average rate λ_a and the sweep rate σ. In a period t, the maximum number of packages that could be generated by a source is:

$$N_m = \sigma t$$

In a period of time t, the average number of packets that can generate a source is given by:

$$N_a = \lambda_a t$$

The average time between two adjacent packets is:

$$T_a = \frac{t}{N_a} = \frac{1}{\lambda_a}$$

In a Pareto distribution, the average time between arrivals is given by:

$$T_a = \int_a^{\infty} t \frac{ba^b}{t^{b+1}} dt = \frac{ba}{b-1} [S]$$

Therefore, it can be established that:

$$\frac{1}{\lambda_a} = \frac{ba}{b-1}$$

6. Voiceover Internet Protocol

On the website of the Federal Communications Commission (FCC/Federal Communications Commission)[14]. Voice over IP is defined as a technology that allows telephone calls using a computer network over a data network like the Internet. VoIP converts voice signals from a telephone call into a digital signal through the Internet and at the other end converts they back into analog signals. This way, you can call someone located in another computer or a regular telephone[15].

The transport manager calls in a VoIP network used is the Internet Protocol (IP) belonging to the TCP/IP (Transmission Control Protocol/Internet Protocol). IP provides a set of facilities for managing the delivery of voice information over data networks, so that the voice from a call is digitized and segmented into packets which travel through multiple paths in computer networks and reassembled on the target computer, giving the feeling of zero interruptions.

6.1 Voice Coding

The human voice (analog), to be efficiently transmitted in data networks, a transformation needs to digital format by a process recognized as voice digitization and which is performed by a device called a CODEC (Coder/Decoder). The codecs can be characterized according to the following factors:

- The (of "waveform", of "speech synthesis") technology.
- The bit rate: Usually expressed in bits per second (bps).
- The resulting encoded audio quality.
- The complexity.
- The delay introduced: Expressed in milliseconds (ms).

According to the bandwidth of the input signal, the most representative codecs are classified to narrowband codecs and broadband codecs (See Tables 1 and 2).

Within the category narrowband codecs whose voice sampling rate is 8000 samples per second they are included. In the case of broadband frequencies they are much higher and range between 16000 and 32000 samples per second.

With wideband codec sound quality is increased and thus a larger amount of data is generated, which in turn require more bandwidth for transport and this makes them suitable only for VoIP systems in LANs and high-speed WAN[16].

According to its suitability and use for VoIP, the codec usually more related to this technology are respectively G711[17], G723[18] and G729[19] (See remarks in Tables 1 and 2). The process of digitizing voice respectively comprises three steps: Sampling, quantization and coding:

- **Sampling:** Samples of the voice signal are taken at periodic intervals. It is recommended that the intervals are consistent with the sampling theorem, which states that the minimum frequency at which a signal may be sampled before being reconstructed, must be twice the maximum frequency of said signal[20].

- **Quantification:** For each sampled value in the previous stage quantization is performed in discrete units. The total number of discrete values should minimize the possible quantization noise.
- **Coding:** The quantized values are coded in numbers that can be digitally processed and then transmitted. This feature is particularly performed by the voice codec used.

6.2 Bandwidth for Voice over IP

Given that, for transporting voice information over networks, it is necessary to assemble packets, the required bandwidth depends on the overhead generated by these packets. For sending voice over LAN networks PLC RTP (Real-time Transport Protocol/Transport Protocol-real time) is used. This protocol in turn is encapsulated on the transport protocol UDP (User Datagram Protocol/ Datagram Protocol user), which in turn is encapsulated on the Internet Protocol (IP) and travels over the PLC network (see Figure 1).

The sum of the different protocols used makes the bandwidth required for voice traffic over PLC is greater than the bandwidth of the original audio. Next, an example for the G.711 codec is presented.

For a window of 20 ms and G.711 audio encoding, voice 160 bytes per frame are obtained (see Figure 2).

The number of bytes of voice/frame can be calculated as follows:

$$\text{Voice bytes / frame} = 64\ \text{kbps} * 20\ \text{msec} / 8 = 160\ \text{bytes}.$$

The IP encapsulation (including RTP and UDP) adds 40 additional bytes.

$$\text{IP packet bytes} = 160 + 40 = 200\ \text{bytes}.$$

The PLC frame adds another 26 bytes:

$$\text{PLC frame bytes} = 200 + 26 = 226\ \text{bytes}.$$

For example, presented it shows that every 20 ms 226 bytes to be transported by the PLC LAN network are generated. This corresponds to a bandwidth of 90.4 Kbps, which is considerably higher than the audio stream (64 kbps).

$$\text{Bandwidth required} = 226 * 8 / 20\ \text{ms} = 90.4\ \text{kbps}.$$

It should be noted that the calculations were made for sending audio in a single direction. Silence suppression

Table 1. Codecs narrow band

codec	Firstname	Bit Rate (kbps)	Delay (ms)	Observations
G.711	PCM: Pulse Code Modulation	64, 56	0.125	Codec "base", uses two possible compression laws: μ-law and A-law
G.723.1	Hybrid MPC-MLQ and ACELP	6.3, 5.3	37.5	Originally developed for video on the PSTN, it is highly used in VoIP systems
G.728	LD-CELP: Low-delay code excited linear prediction	40, 16, 12.8, 9.6	1.25	Designed for applications DCME (Digital Circuit Multiplex Encodig)
G.729	CS-ACELP: Structure Algebraic Codebook Conjugate Excited Linear Prediction	11.8, 8, 6.4	fifteen	Widely used in VoIP applications 8 kbps
AMR	Adaptive Multi Rate	12.2 to 4.75	twenty	Used in GSM cellular networks

Table 2. Wideband codecs

codec	Firstname	Bit Rate (kbps)	Delay (ms)	Observations
G.722.1	Transform Coder	24, 32	40	Audio and video useful
G.722.2	AMR-WB	6.6 to 23.85	25.9375	Common standard 3 GPP and high immunity to background noise especially in harsh environments (e.g. cell)
G.711.1	G.711 Wideband	64, 80, 96	11,875	Wide bandwidth of 711 Codec, optimizing its use for VoIP
G.729.1	G.729 Wideband	8-32	<49	Wide bandwidth of G.729, optimizing its use for VoIP
RTAudio	Real Time Audio	8.8, 18	40	Microsoft's proprietary codec, used in unified communications (OCS)

techniques, in which no packets are sent when no audio is present, can be employed. In these cases, the bandwidth in each direction may be slightly more than half of the previous calculation.

In the Table 3 bandwidths required for the most representative codec in VoIP network management LAN PLC (G.711, G.723 and G.729) are presented. As there shown, the bandwidth required can vary considerably depending on codec used and the selected window.

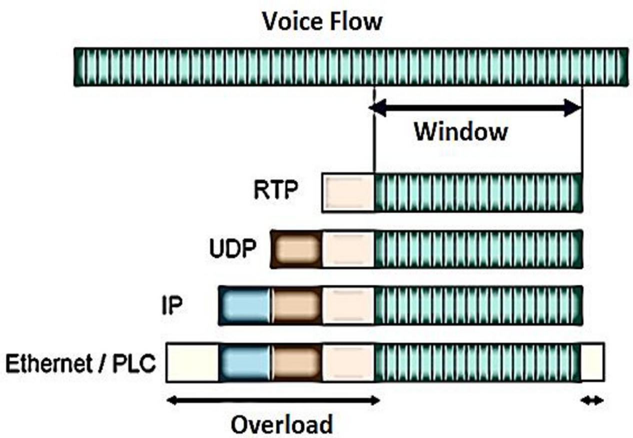

Figure 1. Transport network voice PLC.

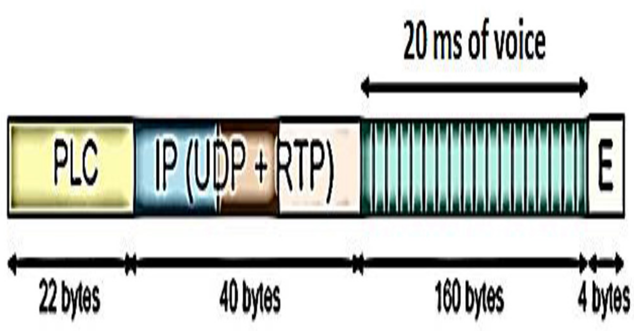

Figure 2. Plot for G.711 codec.

7. Video Transmission over IP

Like voice, video is another service that has now become very important in the development of IP-supported solutions. However, the video requires more resources both channel capacity and delay a number of measures, to be transmitted in a data network due to the large volume of information and the degree of sensitivity while this service requires.

Video sequences (Elementary Streams) are packaged in units called PES (Packetized Elementary Streams), consisting of a header and up to 8 bytes of data sequence. These PES turn are fragmented into small packets of 184 bytes, which together with a 4-byte header (188 bytes in total) form the MTS (MPEG Transport Stream) and can be transmitted by various means including PLC.

In IP networks, video transport is done via the RTP protocol (Real-time Transport Protocol) and RTCP (RTP Control Protocol). RFC 2250 standard establishes the procedures for transporting MPEG-1 and MPEG-2 video over RTP. Several MTS packets of 188 bytes can be transmitted in a single RTP packet in order to improve service efficiency[21]. Some systems do not use the RTP protocol but include MTS packets directly into a UDP packet, which in some cases is used to reduce delays in the system. However, the use of the RTP protocol has several advantages:

- Sending video within the RTP protocol standardized by the RFC 2250 and the DVB-IP group[22], Ensuring interoperability.
- The RTP protocol uses RTCP as a control mechanism.
- RTP includes default clock information, which will be used by the decoder at the receiver for proper reconstruction of multimedia flows.

Table 3. Bandwidth associated with codecs for VoIP on PLC

TypeCodec	Frameduration (ms)	Voice Bytes / Plot	IP packet bytes	Bytes frame PLC	PLC LAN Bandwidth (kbps)
G.711 (64 kbps)	10	80	120	146	116.8
	20	160	200	226	90.4
	30	240	280	306	81.6
G.723.1 (6.3 kbps)	30	24	64	90	23.9
G.723.1 (5.3 kbps)	30	20	60	86	22.9
G.729 (8 kbps)	10	10	50	76	60.8
	20	20	60	86	30.4
	30	30	70	96	25.6

- Network equipment identified by the RTP protocol packets that require prioritization and thereby maintain adequate levels of QoS, which does not happen if encapsulated directly over UDP.

RFC 3016[23] and RFC 3640[24] establish procedures for transporting audio streams and MPEG-4 video. RFC 3984[25] it establishes procedures for transporting video streams encoded in H.264.

7.1 Compression Factor

The process of digitizing video using techniques that transform a sequence of pixels to the domain of spatial frequency (DCT/Discrete Cosen Transform), quantizing values, discarding possibly high frequency components and using prediction techniques and motion compensation. This generates "quantization noise" which can degrade the original image to detectable levels. It is this "quantization noise"[26], Which it generates the classic degradations Viewable images and videos with high compression, including the known "block effect", which makes view the image as a set of small blocks.

Compression algorithms currently used in digital video encoding introduced various types of degradations, which can be classified according to their main characteristics[27]. This classification is useful to understand the causes of degradation and the impact on perceived quality. Due to the large bandwidth required to send video signal uncompressed, using compressed codecs it is extremely common. This puts a limit on the received image quality, which is independent of the transmission medium over which the signal travel.

The main degradations introduced by the use of compression codecs are:

- Effect of blocks (blocking).
- Effect of base image (image basis).
- Fuzziness or blur (blurring)
- Color bleeding (color bleeding)
- Staircase effect and Ringing.
- Mosaic patterns (mosaic patterns).
- Contours and false edges.
- Errors Motion Compensation (MC mismatch).
- Mosquito effect.
- Fluctuations in stationary areas.
- Chrominance errors.

8. IP Bandwidth for Video

The digital video coding using compression algorithms which generate frame lengths and variable band width. The bandwidth depends on the type of encoding (MPEG-1, 2, 4, H264, etc.), image resolution (SD, CIF, QCIF, etc.), the type of quantification and movement and texture image. To this must be added the overhead of the IP, UDP and RTP headers. Video encoding is statistically and depends on the transmitted image, so that the calculations are approximate. Normally the video allows you to set the bit rate, or desired bandwidth and codec dynamically varies its encoding parameters to achieve the bandwidth established at the expense of changing the quality of it.

In IP networks, the overhead depends on the form of encapsulation used. As mentioned above, the use of IP, UDP and RTP protocols incorporate 40 bytes and the PLC header another 26 bytes. Taking into account that in the case of MPEG-2, through the use of MTS packages encapsulated in RTP, up to 7 MTS packages can be included in the same IP packet. Each MTS has 4 bytes of header and 184 bytes of content. Therefore, an IP packet with MPEG-2 would be shaped as shown in Figure 3:

In an IP packet may include 7 x 184 = 1288 bytes of MPEG-2 content, and then there are 40 + 4 x 7 = 68 bytes of headers at layer 3 (IP) and 94 bytes of headers at layer 2 (Ethernet). (Table 4) corresponding to the bandwidth that can be required for a particular compression format video value is presented.

Based on the above expressions, and the data recorded in Tables 3 and 4, they have been calculated parameters for a PDF of a interarrival time following a Pareto distribution, for voice or video over a PLC network, which are recorded in Tables 5 and 6.

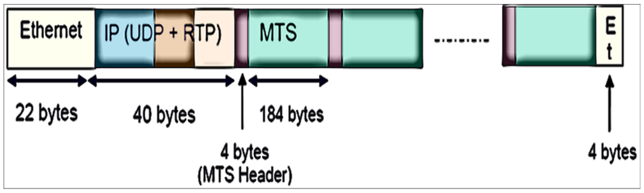

Figure 3. Structure of an IP packet with MPEG-2.

9. Characterization of Traffic Control Services and Telemetry over IP

Currently there are several technologies aimed at the provision of control and telemetry over IP such as X10,

Table 4. Bandwidth for various video compression formats

Compression format	Video Stream	BW Min [Mbps]	Max BW [Mbps]
H.263	video Conference	0,064	0.384
3GP	video Phone	0.016	0,128
MPEG-1	video Standard	1.5	3
MPEG-2	Standard TV	3.5	5.25
	DVD	3	9.8
	HDV	19	25
MEPG-4	HDTV	8	fifteen
	AVCHD	24	30

A10, Zigbee, LonWorks, HomePlug GP, which in most cases do not require high transmission rates for optimal performance. Each control device can generate transmission rates ranging from 60bps (X10) to the order of 200Kbps (Zigbee)[28].

In order to be able to characterize the traffic for the provision of IP control services under the use of PLC technology and taking into account that in the market it is not possible to easily acquire equipment that meets these conditions, because these are found restricted for large companies, under demands for corporate solutions and high costs, it was decided to develop a prototype of IoT control, supported by PLC technology; which allows to perform various functions such as: On/Off, Dimmer and power consumption measurement.

The use of the IoT prototype resulted in an average rate of 2 Kbps and a maximum transmission rate of 150 Kbps, during an average time of 10 minutes. In view of the above, the traffic generated by the proposed device for the provision of control and telemetry services can be characterized by a Pareto distribution with values of 1 and 2 Kbps corresponding to parameters a and b respectively.

10. Data Traffic Characterization

Again, using the Wireshark tool traffic generated by a computer is captured for a time of 10 minutes, yielding results in an average rate of 2.2 Mbps and a maximum transmission rate of 20.5 Mbps. In view of the above, the data traffic generated by a computer, can be characterized by a Pareto distribution with values of 1.1 and 2 Mbps corresponding to the parameters a and b respectively.

11. Description of the Proposed Scenario

Figure 4 shows the general scheme of the proposed network topology, which is made up of N nodes. Each node is

Table 5. Parameters Pareto voice over IP

codec	BW Min [kbps]	$\overline{BW}$ [Kbps]	Max BW [kbps]	L [bytes]	λ_a [Packages / s]	λ_σ [Packages / s]	Pareto parametersfor $f_T(t)=\frac{ba^b}{t^{b+1}}\ a\le t<\infty$			
							to	b	Halfμ	variance σ^2
G.711	81.6	99.2	116.8	160.0	77.5	13.8	0.0086	3.0	1,2903E-02	5,2893E-03
G.723.1	22.7	23.2	23.7	21.0	138.1	3.0	0.0048	3.0	7,2414E-03	1,1290E-01
G.729	25.6	43.2	60.8	20.0	270.0	110.0	0.0025	3.1	3,7037E-03	8,2645E-05

Table 6. Pareto parameters for video services over IP

codec	BW Min [kbps]	$\overline{BW}$ [Kbps]	Max BW [kbps]	L [bytes]	λ_a [Packages / s]	λ_σ [Packages / s]	Pareto parameters for $f_T(t)=\frac{ba^b}{t^{b+1}}\ a\le t<\infty$			
							to	b	Halfμ	variance σ^2
H.263	0,064	0.224	0.384	1500	18.7	13.3	.0370	3.2	5,3571E-02	5,6250E-03
3GP	0.016	0,072	0,128	1500	6.0	4.7	.1156	3.3	1,6667E-01	4,5918E-02
MPEG-1	1.5	2.25	3	1500	187.5	62.5	0.0036	3.1	5,3333E-03	2,5600E-04
MPEG-2	3	14	25	1500	1166.7	916.7	0.0006	3.3	8,5714E-04	1,1901E-06
MEPG-4	8	19	30	1500	1583.3	916.7	0.0004	3.2	6,3158E-04	1,1901E-06

made up of a PLC adapter and a traffic source. The node N will be considered as the main node of the PLC network.

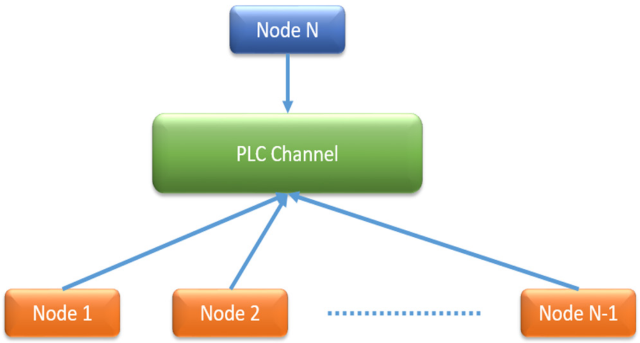

Figure 4. Stage proposed network under a PLC environment.

Scenario 1: In this first scenario a network consisting of five (5) PLC nodes is proposed. Each node is made up of a PLC adapter and a traffic source. Each traffic source can generate more than one traffic class r simultaneously (Voice, Data, Video and Telemetry). Table 7 shows the traffic classes and the established codec (applicable only to Voice or Video), for each node i and class r that are part of scenario 1.

Table 8 shows the estimated bandwidth per traffic class; according to the terms established under a Pareto distribution for scenario 1 is presented.

Table 7. Traffic class for each node i class r scenario 1

Node i	Trafficclasses			
	Voice (r = 1)	Video (r = 2)	Telemetry (r = 3)	Data (r = 4)
1			X	X
2	G.723.1			X
3		MPEG-1	X	
4	G.729			X
5		MPEG-2		X

Table 8. BW required for each node i and r class scenario 1

node i	Trafficclasses			
	Voice [Kbps]	Video [Mbps]	Telemetry [Kbps]	Data [Mbps]
1			2.75	9.19
2	23,50			6.14
3		2,32	1,38	
4	44.72			5.45
5		4.07		6.61

Scenario 2: In this second scenario a network consisting of seven (7) PLC nodes is proposed. Each node is made up of a PLC adapter and a traffic source. Each traffic source can generate more than one traffic class r simultaneously (Voice, Data, Video and Telemetry). Table 9 shows the traffic classes and the established codec (applicable only to Voice or Video), for each node i and class r that are part of the proposed scenario.

Table 10 shows the estimated bandwidth per traffic class, according to the terms established under a Pareto distribution for scenario 2 is presented.

To perform an analysis on the result of the data, will use a whisker diagram in order to identify levels of dispersion for each service class. Boxplots-Whiskers (or box and whisker boxplots) are a display depicting several important characteristics, while such as dispersion and symmetry. Its construction is necessary to calculate the three quartiles and minimum and maximum data values, which are aligned horizontally or vertically.

This type of graph is a rectangular box, where the longest sides are the interquartile range. The box is divided by a segment, which indicates the position of the medium, which coincides with the second quartile (Q2). Additionally, the pack is located on a scale segment whose ends the minimum and maximum values of the analyzed data. In view of the above, the lines protruding from the box are called whiskers. These whiskers have a limit of extension, so that any data that is not within this range is marked and individually identified.

In Figure 5 the distribution diagram width corresponding to voice packets band appears. It can be seen that the lower whisker (Xmin, Q1), is quite short compared to the top. Additionally, the bottom of the box (Q1, Q2) is smaller than the top (Q2, Q3); This means that most of the estimated band width for the provision of voice services will range from values below 50% of the population (Xmin, Q2). Those presenting greater dispersion bandwidth estimates that are between 50% and 75% (Q2, Q3). Finally, the interquartile range = Q3 - Q1, reflects 50% of the estimated bandwidth for voice packets is between 81.6Kbps 23.5Kbps and located within a range of 58.1 Kbps dispersion.

Figure 6 shows the distribution diagram of the bandwidth corresponding to the Video packages. In it you can see that the lower mustache (Xmin, Q1), is shorter compared to the upper one. Additionally, the lower part of the box (Q1, Q2) is much smaller than the upper part (Q2, Q3); this means that most of the estimated bandwidth for the provision of video services will be between the values below 50% of the population (Xmin, Q2). With a greater dispersion, those bandwidth estimates that are between 50% and 75% (Q2, Q3). In this case, the interquartile range = Q3 - Q1, reflects that 50% of the estimated bandwidth for video packets is between 3.19 Mbps and 16.89 Mbps, located within a dispersion interval of 13.7 Mbps.

In Figure 7 the distribution diagram width corresponding to the data packets band appears. It can be seen that the lower whisker (Xmin, Q1), is shorter compared to the upper. Additionally, the bottom of the box (Q1, Q2) is symmetrical with the top (Q2, Q3); this means for the provision of data, bandwidth requirements are more balanced and less scattered compared to voice and video services. In this particular case, the interquartile range = Q3 - Q1, reflects 50% of the estimated bandwidth for data packets is between 5.96 Mbps and 11.87 Mbps, located within a range of 5.91 Mbps dispersion.

Figure 8 shows the distribution pattern width corresponding to packets Telemetry band appears. It can be observed very similar to reflected in the data packets from the point of view of the symmetrical distribution on the box (Q1, Q3) situation. However, in the case of whiskers if it can be seen that the lower whisker (Xmin, Q1) is much shorter compared to the upper. Finally, the interquartile range = Q3 - Q1, reflects 50% of the estimated bandwidth for packet telemetry is between 1.5 Kbps and 2.8 Kbps, located within a range of 1.3 Kbps dispersion.

Table 9. Traffic class for each node i class r scenario 2

Node i	Trafficclasses			
	Voice (r = 1)	Video (r = 2)	Telemetry (r = 3)	Data (r = 4)
1	G.711		X	X
2	G.723.1		X	X
3		MPEG-4		X
4	G.729		X	X
5		MPEG-2	X	
6				X
7	G.711			X

Table 10. BW required for each node i and r class scenario 2

Node i	Trafficclasses			
	Voice [kbps]	Video [Mbps]	Telemetry [Kbps]	Data [Mbps]
1	81.6		1.5	3.5
2	23.2		2.2	10.2
3		18,4		8.4
4	35.8		2.7	12.7
5		15.38	1.8	
6				16.5
7	102.6			11.6

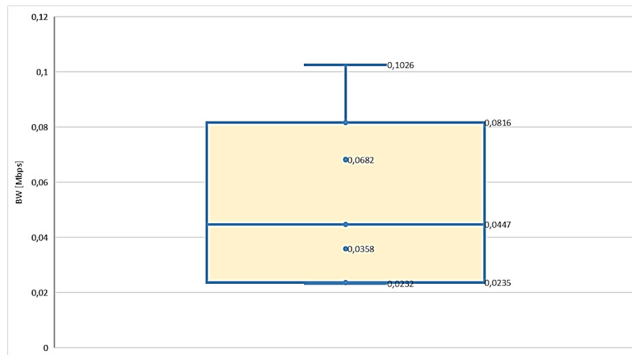

Figure 5. Diagram distribution BW for voice packets.

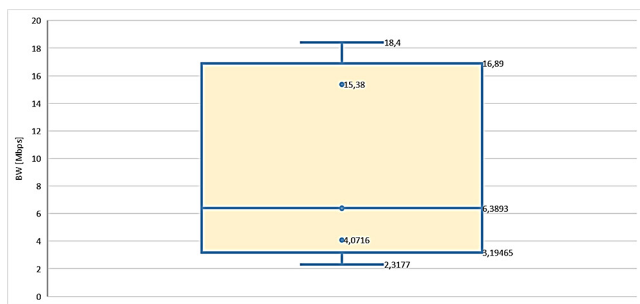

Figure 6. Diagram distribution BW for packages video.

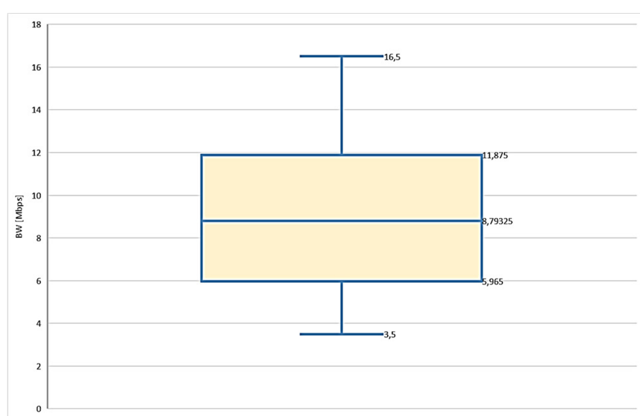

Figure 7. Diagram distribution BW for packet data.

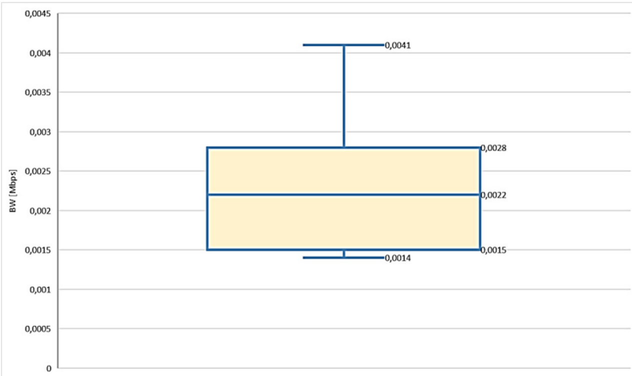

Figure 8. Diagram distribution BW for packages telemetry

12. Conclusions

During the evolutionary processes of networks, the need to have traffic models that describe the behavior of the different kinds of service that may be present in the closest way to reality has arisen. In view of the above and taking into account that in recent studies it has been demonstrated that the traffic in LANs describes a self-similar behavior, the strategy of characterizing each of the classes of services such as voice, video, data and the recent one was considered. Telemetry associated to the traffic product of IoT devices, as a Pareto distribution, with specific parameters for each class and estimated for its modeling within networks supported in Power Line Communications (PLC). Based on the results obtained, it was possible to demonstrate that the proposed models for each class of service can be used as a strategy for the analysis of quality of service within PLC networks, as well as for the generation of traffic in other scenarios that require estimating the bandwidth needs that may be required during the provision of any of the four services mentioned above.

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