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A novel probabilistic approach for analysis and planning of large capillarity broadband networks based on ADSL2+ technology

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1. Introduction

The increasingly spread of information through digital media raises new realities in the world's present scenario and, thus, new technologies have been emerging in order to streamline the process of disseminating information and providing quality access to such information by the population. The Next Generation Network (NGN) holds tremendous potential, with a promise to merge the transmission of data, voice, video and other media into a single network; unfortunately, several developing countries do not have the necessary infra-structure to implement NGN technology. The main concern in these networks is not the backbone or the transport layer, but in the last mile itself. Last mile has become a popular keyword to indicate the technology which connects the End User to the Network backbone.

In most of North America and Western Europe, Internet penetration is very high and nearly every citizen has access to the Internet. However, this is not true in many parts of the developing world, where only a small percent of the population has access, even if the bandwidth is significantly low and the cost is a substantial fraction of the user's income (Ambrosi et al., 2005). According to (IWS, 2009), more than 70% of the population of the developing countries does not have access to Internet due to lack of infrastructure ; furthermore, in countries like China, India and Brazil, with continental dimensions, the construction of a new telecommunications network, like optic fiber, becomes costly and impractical. In this context, new alternative technologies that can offer trade-off between performance and costs must be sought.

There are several approaches to deliver service to the end user (Xiao et al., 2007). An alternative with less time and cost would be to use a combination of existing infrastructures such as electrical grids or telephone networks, based on copper loops, which are widely available to end users in most developing countries (Papagianni et al., 2009). For areas

where the network has low penetration, wireless network can be a better solution; it, however, requires a basic infrastructure (base stations, antennas, etc.).

Telephone access networks were originally built for analog voice communication, carrying voice-band signals up to 4 KHz in the frequency bandwidth, and not for digital data communication. We considered here a large capillarity broadband network because it uses a combination of the existing copper infrastructure and digital subscriber line transmission technologies, thus enabling a universal broadband access at a fraction of the cost and in a fraction of the time required for others access networks.

DSL remains the dominant access technology with 65% of the worldwide subscribers for broadband, compared with 33% of fiber optics connection. It is in the developing countries that the number of DSL connections for last mile really stands out, such as in India, representing 83% of broadband connections; and China, which continues to grow, reaching 93,549,000 subscribers (Point Topic, 2009).

The DSL (Digital Subscriber Line) is considered as the dominant broadband access technology, not only in Europe but also in Latin America and developing countries like India (Olsen et al., 2006) (Arena et al., 2006) (Faudon et al., 2006). In Latin America, DSL technology accounts for 77% of all broadband access. At the end of 2005 there were nearly 5,300,000 subscribers of ADSL (Asymmetric Digital Subscriber Line) in Latin America (Arena et al., 2006).

Particularly, Brazil was marked by a major growth in broadband access. In 2005, 52.1% of home users had dial-up, 41.2% broadband and 6.7% both forms of access. By 2008, the statistics had changed considerably; Broadband access rose to 58% against 31% of dial-up access; whereas the DSL access accounts for 23% of the total broadband access (CETIC, 2008).

It is now a fact that the broadband access has been changing the user's needs, which was initially only for accessing websites. Now, users are keen to use services such as video, voice and data separately, one at a time. Customers enjoy the convenience of receiving all three services they need today from one service provider, increasing demand for triple play services. Thus, telephone companies (Telcos) offer triple play by providing television service using IP (i.e. Internet Protocol Television - IPTV) in order to compete more effectively with cable television companies that have entered the voice and Internet access markets. Therefore, it is imperative to study the computer applications in such infrastructures that were not designed with this goal.

The last mile network maintenance is another important factor as it is currently performed with a mixture of help systems, manual testing by technicians, and automated tests that are developed for plain-old telephone service (POTS) lines, which ignore DSL frequencies above 4 kHz. Provisioning is based on rough estimates of the loop length and does not account for individual loop characteristics. There will be more complications in the maintenance when the DSLs start supporting triple-play services: the Internet, Voice-over-Internet Protocols (VoIPs), and Internet Protocol TVs (IPTVs).

It is important to use a test system that can accurately identify and inform the source of a problem in the network; whether this problem comes from the Internet service provider (ISP), the telephone central office (CO), the outside plant, the modem, or the user's PC. DSL testing is not only limited to measuring electrical parameters on the copper pair but also to include the comparative analysis of extracted data with previously known limits as well as comparing assigned configuration with discovered configurations (Kerpez & Kinney, 2008).

The convergence between existing and emerging broadband technologies has been regarded as a major challenge, especially with respect to supporting multimedia content in these technologies. This is because new applications, such as IPTV, require high bandwidths, which are usually not available due to long distances of DSL links, noise, data congestion, lack of protocols implemented for this new need, among others.

The IP Protocol is considered as the standard protocol for communication among different network types. Unfortunately, the IP network presents issues on the provision of end-to-end QoS. For this matter, some services use the transport protocol TCP (Transmission Control Protocol), which has been considered as the main communication protocol. However, for networks with high packet loss, this protocol is flawed, for it enables the congestion control unnecessarily, since some applications can tolerate losses in communication.

It is then observed that the current data communication, which primarily uses the TCP/IP, is appropriate for applications such as HTTP traffic; for it maintains compatibility with existing networks (routers, gateways, etc.), and controls the flow and congestion. However, for present day applications, referred to as triple play with flows that are sensitive to QoS, TCP/IP is inefficient.

The division of the TCP/IP into layers facilitates the implementation of new applications. This feature, however, has become a negative aspect, since TCP/IP does not implement means for interaction between its layers. New applications require minimum levels of QoS for their operation, and this requires a greater interaction between its layers, in order to maintain these minimum levels for specific applications over others.

The Crosslayer model aims to implement an interaction between the layers of the protocol, providing more quality of service to the user, who is less interested on the technology details, but demands more in terms of quality. Using this technique, it is possible to, for example, adjust in real time the performance parameters of the application layer, such as throughput and jitter, given that the transport layer provides information about lost packets, allowing an adaptation of the application that is being used and the characteristics of the environment in which the transmission is carried out.

It is therefore essential to investigate the transmission technologies (last mile, and protocols used) in order to achieve a better strategy to expand telecommunications services in regions with little infrastructure available to the typical end user among the various possible scenarios. These inferences of possible scenarios, as a rule, are performed using a combination of prototyping and modeling for performance evaluation.

The main motivation of this work is the need to provide a high quality service to users, by ensuring that objectives of the network service level will be maintained; the need to correlate events from the lower layers of management, in order to determine strategies in the upper layers (crosslayer); the need to increase the sophistication required for diagnosis, given the greater complexity of the system; and the need to quickly detect network failures and, if possible, provide automatic recovery. In order to achieve these objectives, we apply a hybrid model, combining the qualities of genetic algorithms (GA) for space search with a Bayesian probability model for inference.

The correlation of events and attributes is important to analyse the behavior and functionality of applications, and reduce costs with respect to the network maintenance, improve availability and performance of its services. Raw data are interpreted and analyzed, taking into account a set of predetermined criteria, or defined dynamically according to the management process.

The tests were implemented over the backbone of the Brazilian Telecommunications System (Telebrás) and based on specific standards of DSL communication.

2. Test Bed Architecture

In order to evaluate the triple play communication in a DSL network, a standard model must be used, such as the ones suggested by (Papagianni et al., 2009) (Kerpez & Kinney, 2008) and (Sadri et al., 2007) (Figure 1), which define the sequence of connected elements:
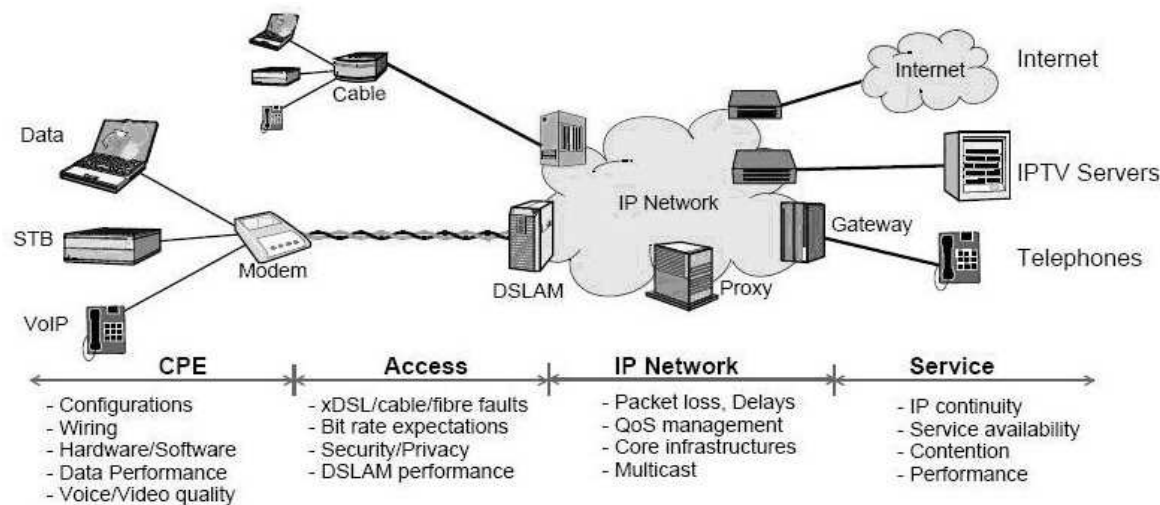

Fig. 1. Example of a Standard Test Bed

The architecture to be used will include the following items:
1. xDSL Modems, including ADSL, ADSL2, ADSL2+;
2. DSLAM (Digital Subscriber Line Access Multiplexer);
3. Simulator of cables for the European standard;
4. Protocols analyzer.

3. DSL Loop Length

With the recent rapid growth of high-speed DSL access subscriptions, there is a high demand in the telecommunication industry for equipment to accurately predict DSL access performance over a telephone subscriber line (also referred to as a local loop).

The subscriber line is a metallic twisted-pair network link between the customer and the telephone Central Office (CO). While some of the existing DSL analysis equipment is already capable of assessing the performance rate, it requires two-point operation (sending test signals from one end of the loop and measuring the signals at the other end) involving the dispatch of a service vehicle. This leads to expensive testing processes and it is therefore an undesirable solution for DSL access providers.

The use of DSL technology to transmit data at high speed enables quick service delivery, mainly due to the fact that there is an external network and cabling with twisted copper pair, with wide coverage in almost all areas and niche markets. This network is not homogeneous, co-existing with new systems and old networks of 30, 40, 50 years ago.

Particular items that can prevent the use of the service are: loop losses, bridge taps, specific noises of twisted pair links, long loops (distances from the central office to the user), among others, which can slow down the service. Usually the specifications are given for wiring 24 or 26 AWG (American Wire Gauge) over distances of 2090m (Telebras, 1997); it is, however, unknown exactly which parts are with each type of cable and their distances, making necessary for an extra effort to measure it. Such items are shown more specifically in the following section.

4. DSL Copper Impairments

As DSL uses relatively high spectrum frequencies, its signal is susceptible to external noise sources. Thus, the understanding about the behavior of different kinds of noise and their effects on network performance are extremely useful on the design of well established DSL systems (ADSL, ADSL2+) as well as those of upcoming generations (VDSL, VDSL2). During the past years, crosstalk has been considered the major impairment to DSL services. However, other types of noise have gained importance, such as radio frequency interference (RFI), impulsive noise (IN), repetitive electrical impulse noise (REIN), and isolated burst of electrical noise (IBEN) among others (Wallace et al., 2005) (Stolle, 2002).

Fundamental loop transmission impairments may not cause the highest number of DSL trouble calls; however, they can be very difficult to fix, and so, they result in a high DSL maintenance cost. Figure 2 illustrates the DSL copper impairments (Kerpez et al., 2003)(Starr et al., 2003), which are mainly loop and bridged tap loss, crosstalk, electromagnetic interference (EMI) radio ingress, impulse noise, harmonic distortion, and background noise.

Noise on phone lines normally occurs because of imperfect balance of the twisted pair. There are many types of noises that couple through imperfect balance into phone lines, the most common of which are crosstalk noise, radio noise and impulse noise.

Crosstalk is caused by electromagnetic radiation of other phone lines in close proximity, in practice, within the same cable. Such coupling increases with frequency and can be caused by signals traveling in the opposite direction, called near-end crosstalk (NEXT), and by signals traveling in the same direction, called far-end crosstalk (FEXT).

Radio noise is the remnant of wireless transmission signals coupling into phone lines, particularly AM radio broadcasts and amateur (HAM) operator transmissions.

Impulse noise is a nonstationary crosstalk from temporary electromagnetic events (such as the ringing of phones on lines sharing the same binder, and atmospheric electrical surges) that can be narrowband or wideband and that occurs randomly. Impulse noises can be tens of millivolts in amplitude and can last as long as hundreds of microseconds (Cioffi, 1999)(Starr, 1999).
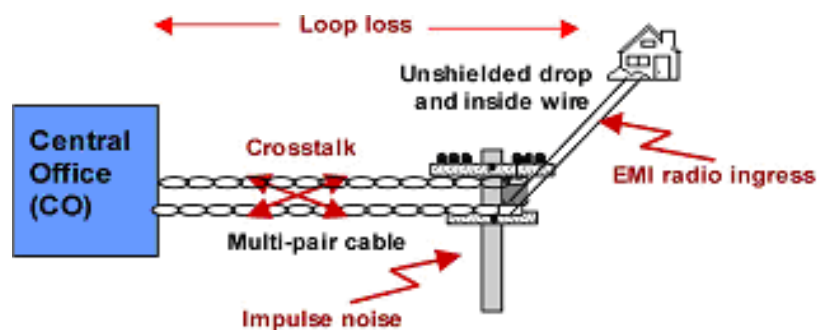

Fig. 2. DSL Copper Impairments (Kerpez & Kinney, 2008)

DSLs are generally provisioned to withstand a worst-case level of crosstalk; however, provisioning systems are approximate, and some older cables have poor crosstalk isolation. Moreover, after several DSLs are activated, some small percentage will actually exceed fundamental worst-case crosstalk engineering rules—however, this small percentage can translate into a high number of troubles.

In spite of conducting several investigations about the impact of non-stationary noise in DSL systems, just few studies have been conducted addressing their impact in terms of experimental analysis. This may be credited to the inaccessibility to a proper infrastructure to handle practical experiments.

5. Planning Methodology and Performance Results

This work implements, through crosslayer techniques, strategies for planning and evaluating the performance of ADSL2+ networks, which implement minimum levels of QoS for Triple Play applications. This approach will be achieved through a set of techniques such as: data measurement, modeling, optimization, simulation, etc. So this will enable creating an information framework that will guide the implementation of triple play applications and/or infrastructure for broadband networks.

The strategy and methodology to be used in the tests are divided into the following topics:
- Definition of architecture and equipment ;
- Definition of variables to be analyzed;
- Implementation of the testbed;
- Set up of equipments and preparation for the tests;
- Empirical tests;
- Analysis of the results;
- Correlation study using Bayesian networks.

With the performance measures and the help of a domain specialist, conjectures can be taken about the behavior and functionality of applications. This study, however, is not complete without considering factors such as the influence and correlation of all the attributes involved.

The correlation of events is important to reduce costs with respect to the network maintenance, improve availability and performance of network services. Raw data are interpreted and analyzed, taking into account a set of predetermined criteria, or defined dynamically according to the management process.

Among the computational intelligence techniques available for correlation analysis and uncertainty, we implement for this analysis the algorithm of Bayesian networks. Known for their models as components with a qualitative (representing the dependencies between the nodes) and quantitative (conditional probability tables – CPTs of the nodes) structures, evaluating, in probabilistic terms, these dependencies (Korb & Nicholson, 2003)(Chen, 2001). Together, these components provide an efficient representation of the joint probability distribution of the variables in a given field.

Bayesian networks are probabilistic graphical models for knowledge representation and reasoning in domains with uncertainty. Its unified nature makes it possible to compare different scenarios about the data, and the intuitive nature of its graphical formalism makes it one of the best analytical methods available for decision making (Rocha et al., 2007).

With Bayesian networks, the behavior of the attributes can be studied; propagating and evaluating hypothesis given certain evidences. Thus, from the Bayesian networks, one could predict how the triple play flow will behave in last mile networks or what are the physical characteristics that the network should have to meet this new need. This should provide a quantified indication in order to enable telecommunications companies invest safely, given the user's need for quality and efficiency in the service provision.

5.1 Definition of architecture and equipment
The test was implemented in the Laboratory of Technological Innovation in Telecommunications (LABIT), with a scenario consisting of modems, DSLAM, telecommunication cables, noise generator, and computers.

The generation of noise is made by the DSL 5500, a noise generator from Spirent Communications, in the operating range of ADSL2+ (4.3125 kHz to 2.208 MHz). A protocol

analyzer from RADCOM (Radcom, 2009) was also used to filter the packets that will travel in the network, isolating specific flows to generate performance metrics.

DSLAM/EDA (Ethernet DSL Access) is the equipment available in the telephone central office, allowing the data communication via a DSL link. The computer connected to the DSLAM is responsible for generating video flows to be distributed to the clients via multicast.

A Wireline Simulator of ADSL2+ ETSI DLS 410E3 from Spirent Communications was used. It reproduces the AC and DC characteristics of twisted pair copper telephony cable using passive circuitry (R, L & C).

The methodology applied is conventionally used for benchmarking of high protocol layers, considering all types of data that can be transmitted; where the data to be changed are specific of the DSL technology, they are the loop length and the applications that are used.

5.2 Definition of variables to be analyzed

The performance measures obtained for this case study are divided by application:

- Voice flow: Jitter (Jitter_VoIP), loss of IP packets (Loss_VoIP), MOS - Mean Opinion Score (MOS_VoIP), number of successful attempts (Attempts_VoIP).
- Video flow: Jitter (Jitter_Video), vídeo throughput (Throughput_Video) and loss of IP packets (Loss_Video).
- Data flow (FTP): Delay (Delay_FTP), jitter (Jitter_FTP), loss of IP packets (FTP_Loss) and throughput (Throughput_FTP).

Where:

Def.1: We call it "throughput" the maximum bit rate, that allows end-to-end IP packet transmission without occurring any packet loss during the test (retransmission is not provided).

Def.2: The one-way IP packet delay is the time an IP packet (of a certain size) needs to travel from source to destination.

Def.3: The IP packet loss is the ratio between the number of lost packets and transmitted packets between source and destination over a long period of time.

Def.4: We have repeated the measurements for different loop distances (2500m, 3000m, 3500m, 4000m and 4500m) and cable type Ø=0.4mm PE. Simulating scenarios without any noise (named Case0), level of White Noise W= -140 dBm and 24 DSL (ISDN) Impairment (named Case1), level of White Noise W= -130 dBm, and 24 DSL (ISDN) Impairment (called Case2) and level of White Noise W= -120 dBm, and 24 DSL (ISDN) Impairment (called Case3). All noises recommended for (TR-048, 2002) and (ITU-T, 2005).

All tests, for each loop length, were repeated 10 times, with duration of 120 seconds.

5.3 Empirical Tests

For the analysis of this last mile technology, a typical scenario for IPTV transmission will be used, where services of voice, video and data will be available (Papagianni et al., 2009).

One of the challenges that VoIP carriers deal with early in their network planning is to choose the most appropriate voice coding standard in order to provide good voice quality and adequate network efficiency. From uncompressed G.711 at 64 kbps to G.726 at 16kbps, G.729 at 8kbps and the highly compressed G.723.1 at 5.3 kbps, the VoIP service providers can choose the level of voice compression that will be applied to their customers. In this particular study the G.711 codec is employed. G.711 is the international standard for encoding telephone audio on a 64 kbps channel.

For the voice transmission, Callgen (VoIP tool developed by the OpenH232 project) (OpenH323, 2007) was used. Besides being widely used for testing, Callgen supports the G.711 codec (Papagianni et al., 2009).

The video codec H.264 standard, which is being adopted by all major video service operators, is utilized in the performance evaluation of triple play service over xDSL access network (ITURec.H.264 & ISO/IEC14496-10, 2007). It was jointly developed by ITU Video Coding Experts Group (VCEG) and ISO Moving Picture Experts Group (MPEG). H.264 is used in fixed and wireless network environment.

H.264 has proven to be more resilient to error prone networks through the use of flexible macroblock ordering, slice interleaving and data partitioning. In addition, it attains enhanced compression performance; therefore it is a "network-friendly" standard. It is capable of providing good video quality at substantially lower bit rates than other standards. Compared to MPEG-2 video, it cuts down transmission bit rate by half, while the coding gain over H.263 and H. 263+ is in the range of 24% up to 47% (Kamaci & Altunbasak, 2003).

VLC (VideoLAN Client) (VLC, 2009) was used to generate the video traffic. VLC is a multimedia player that supports various video formats and streaming protocols, the RTP (Real Time Protocol) was used for the video transmission. The codec H264 was used for the video flow, with rate of 1.2 Mbps; for the audio, the AAC was used with rate of 192 Kbps.

IPERF tool (IPERF, 2009) was used to simulate the FTP traffic continuously, aiming to occupy the total available bandwidth. This tool is dedicated to the performance analysis of networks and widely used in testing (Rao et al., 2009) (Primet et al., 2002).

The tests performed are divided into: tests of the network capacity (Basic Connection); and tests of the applications behavior (Network Capacity Services), where the behavior of the protocols involved are studied.

5.4 Results Obtained

The Mean Opinion Score (MOS) is a numerical pattern (proposed by the ITU-T P.800) used to measure the quality of voice after the compression and/or transmission. Figure 3 shows

the behaviour for the quality of the VoIP communication by enlarging loop lengths and inserting noise; we considered MOS equals to zero when the connection cannot be made or maintained. We can see that the noise of -120 dBm and 24D (case3) has a negative influence in the communication; and that the noise impact in the voice communication is higher in distances above 4000m, with quality loss up to 69% over a distance of 2500m.
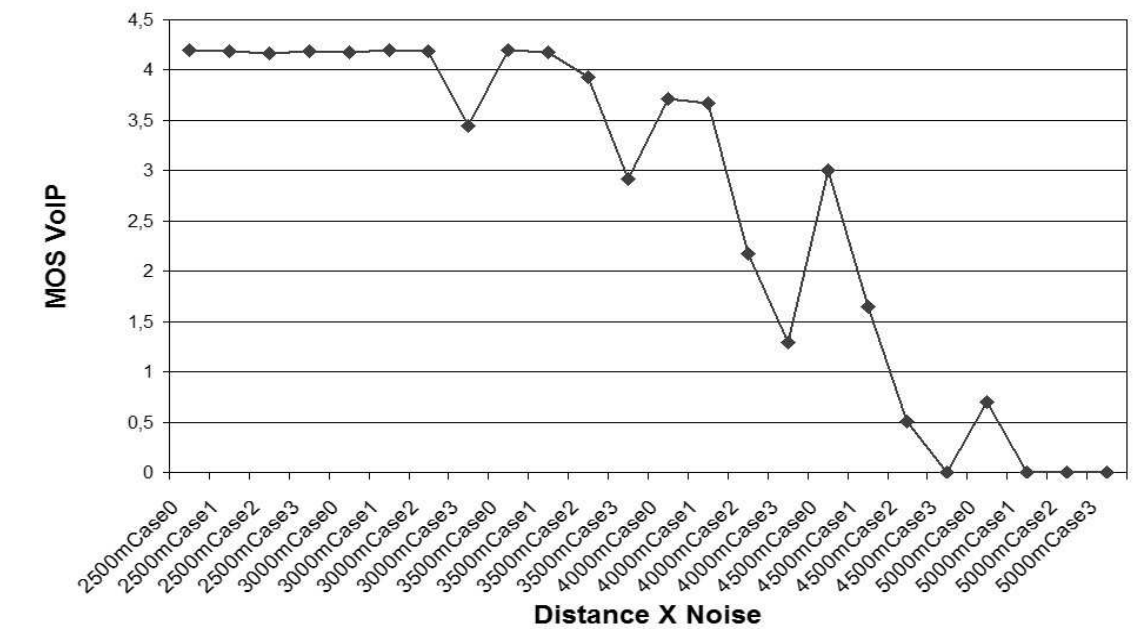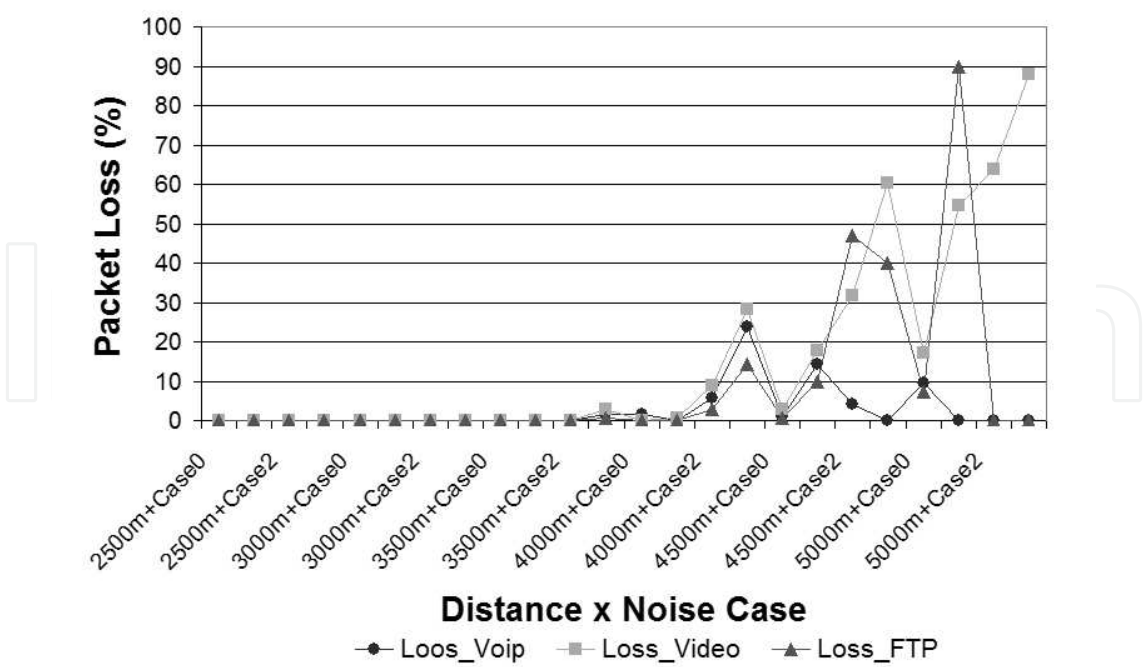

Fig. 3. VoIP MOS Behavior.

Fig. 4. Triple Play Packet Loss per flow.

Packet loss is one of the main aspects that affect the quality of triple play flows, particularly for applications not using reliable communication protocols (primarily voice and video). Data based applications, which uses reliable protocols that implement retransmissions, guarantee the arrival of information with integrity, even at low transmission rates. Figure 4 shows the behavior of applications considering the packet loss. The results illustrate the direct relationship between distance, noise and degradation of flows, especially at distances from 3500m to 4500m, which are more susceptible to noise. These distances measures are widely used in countries with large geographical area (such as Brazil, India and China) and an already established telephony infrastructure, which should now be adapted for digital transmission of data.

This fact can be better identified in Tables 1, 2 and Figure 5, which represent the behavior of applications (in percentage levels) when compared with a communication without noise. The voice application did not suffer packets loss in the entire range of noise, however, for a white noise with -130 dBm and 24D (Case2), a growth of 44% was seen in the jitter, which directly impacted on the MOS, causing a degradation of 6% in the quality of the communication. This impact was even greater when combining the white noise -120 dBm and 24D (Case3), which led to a degradation of 40% in the quality of communication, sending the MOS from, initially, 4.2 to an average of 2.9.

The video and data applications were barely impacted, with small variations (up to 3%), as shown in Table 2; with exception of the data flow, which suffered a drop of 2 Mbps (without noise in communication) to about 317 Kbps (white noise of -120 dBm and 24D – Case3), that is, a decrease of 600%.
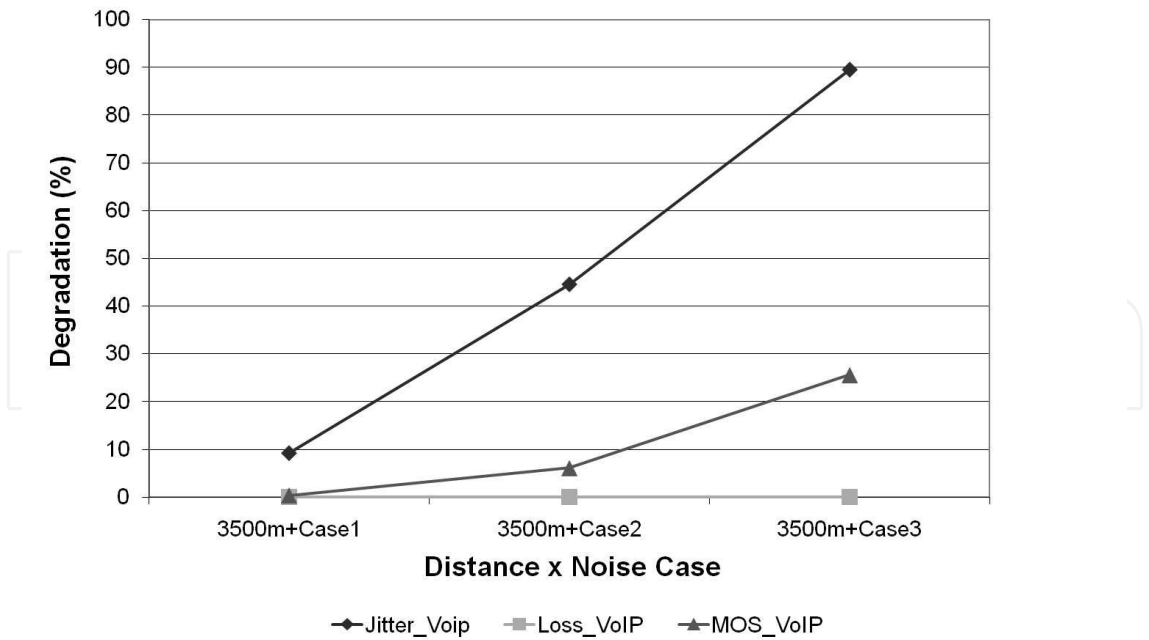

Fig. 5. VoIP Behavior for 3500m.

Metric / Impairment&Distance	Jitter Video (ms)	Loss Video (%)	Throughput Video (Mbps)	Jitter FTP (ms)	Loss FTP (%)	Delay FTP (ms)	Throughput FTP (Mbps)
3500m+(Case01)	7.9	0.02	1.41	5.07	0.11	0.073	2.06
3500m+(Case02)	7.39	0.03	1.42	5.37	0.18	0.079	1.98
3500m+(Case03)	4.55	2.7	1.39	12.01	0.56	0.28	0.317

Table 1. Video and Data metrics for 3500m.

For the distance of 4000, it is observed that VoIP and FTP applications do not vary much with the insertion of a noise -130 dBm and -120 dBm; this is due to the inability to maintain the FTP transfer with these noises, enabling other applications to use all the available bandwidth. As the video application has a greater need for bandwidth, it is expected to show a greater variation than other applications, as shown in Table 2.

Metric / Impairment& Distance	Jitter Voip (ms)	Loss Voip (%)	MOS voip	Throughput Voip (Kbps)	voip Calls	Loss Video (%)	Jitter Video (ms)	Throughput Video (Mbps)	Jitter FTP (ms)	Loss FTP (%)	Delay FTP (ms)
4000m_sr	8.58	1.7	3.71	170	4	2.73	4.43	1.38	4.952	0.12	79.56
4000m rb140dBm&24D	19.56	0.14	3.67	170	4	17.81	5.18	1.15	8.093	0.13	163.58
4000m rb130dBm&24D	43.63	5.55	2.17	160	4	31.73	6.9	0.94	17.475	2.88	376.92
4000m rb120dBm&24D	39.26	23.87	1.29	141.66	4	60.33	12.7	0.54	30.729	14.3	364.19

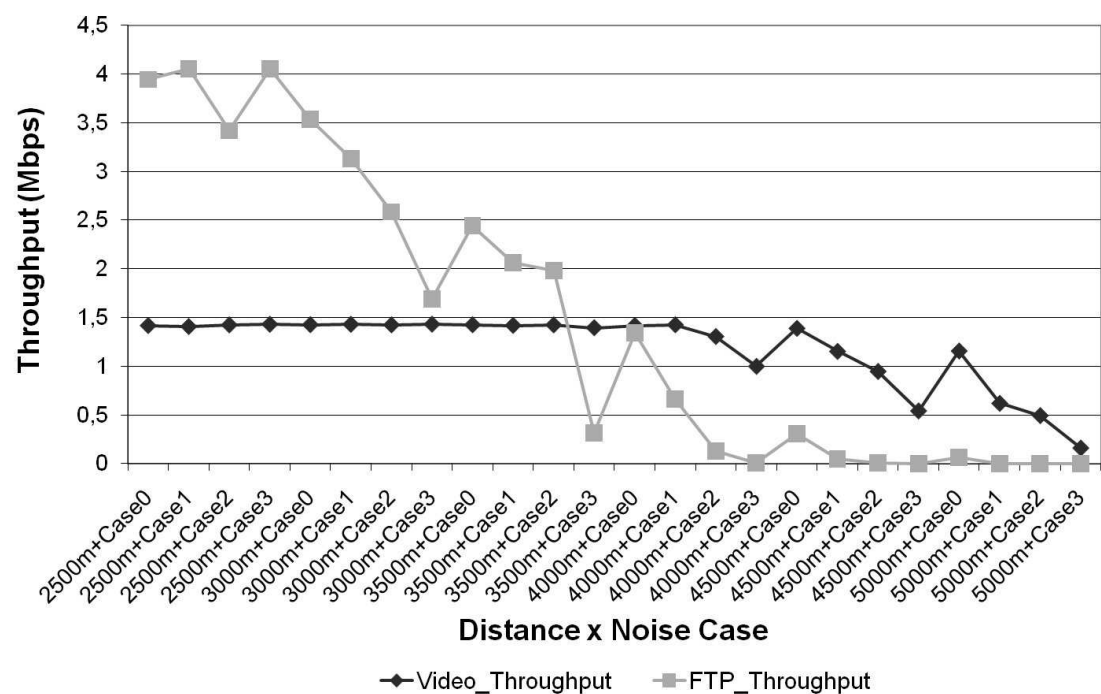Table 2. Video and Data metrics for 4000m.

Fig. 6. Video and Data throughput behavior.

The impact of the loop distance and noise in the applications are shown in Figure 6; it shows the throughput behavior of the data and video applications. In case of video application, throughput remains constant until 4500m with noise -140 dBm and 24D (Case1), where a sharp drop is observed. The data application has an unstable behaviour due to the use of the TCP protocol, which is adaptive and uses the available bandwidth for transmission.

5.5 Bayesian Correlation study

Figure 7 shows the Bayesian network (BN) with all the attributes obtained from the empirical testing (see section 5.2). Each node has a conditional probability table associated with it (e.g. Delay_FTP); with the nodes, the dependencies are also represented, given the direction of their connecting arrows (e.g. the existence of noise in the communication influences the likelihood of a jitter variation in the VoIP application, and, in turn, in the VoIP MOS).

In the BN, all the attributes were discretized in twenty states, according to the frequency of their values, allowing us to verify the probability associated to each one of them, as well as the conditional probabilities existing among the variables.

When inferences are made in the network (e.g. it is evidenced from the occurrence of a white noise of -140 dBm and 24D in the communication), the impacts of these events are propagated, as a chain reaction, throughout the network, updating the probability values of the remaining nodes, in order to reflect their behavior; thus predicting how the network would perform given the occurrence of the instantiated event.
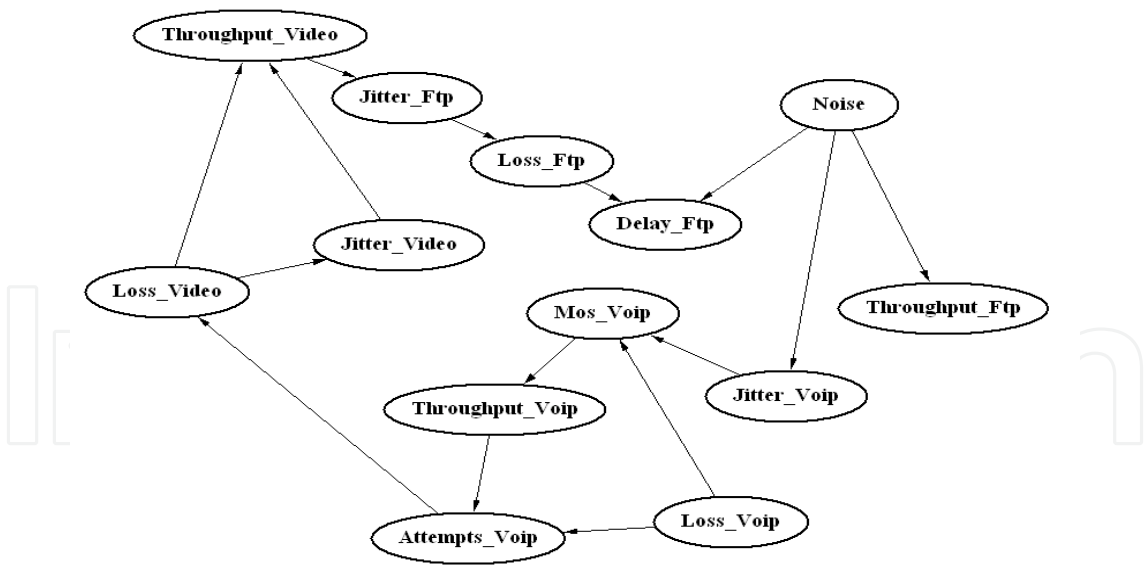

Fig. 7. Bayesian network for Triple Play applications over DSL last mile.

5.6 Scenario Analysis

Case studies were implemented to demonstrate the usability of this approach for network planning.

In the first case study minimum QoS parameters were used; these parameters are those of international standards, which define the quality of these applications and the expected performance measures. The objective is to find the maximum loop length and the set of noise that will enable us to effectively accomplish the quality in transfer of flows. So, Telcos could assess whether their links can support these applications. The results are compared with loop samples obtained from the Brazilian telecommunications networks (Telebras, 1997).

In the second case study, the inverse process is made; a certain loop distance is given as input (inferred), and the impact of this evidence is observed in the performance measures of the applications, by comparing the estimated results with their standards.

Test Case 1: VoIP Application

According to (Papagianni et al., 2009), a VoIP communication should have a minimum quality parameters, without which the VoIP communication is not feasible. Using the G.711 VoIP protocol, a VoIP communication is seen as feasible if it shows measures such as a 60ms jitter, 10% packet loss and up to 150 ms of delay. G.711 is the standard protocol for 64kbps communication, hence its use in the testing analyses.

VoIP communication requires a low bandwidth, but it is very susceptible to communication problems like bottlenecks, delays, losses and noise in communication, making it a major concern in a triple play communication. Therefore, from the inferences made in the BN, which considers and propagates the correlations between all the attributes of the domain, it was verified which behavior other applications (video and data) would have to present in order to maintain the quality required for voice communication.

Initially, with regard to the physical layer, as shown in Figure 8, the distances that enabled these parameters to be maintained were 3000m with white noise of -130 dBm and -24D, 3500 without any noise, or white noise of -140 dBm and 24D (Case1).

Since the average distance of telephone links, according to standard (Telebras, 1997) is 2090 m, it is observed that there is a possibility of extending that distance in 66.9%, i.e. 1404 m. Allowing smaller investments from Telcos in repeaters (bridge taps) which create new interference with communication or the exact definition of the distances necessary to meet these needs.

The video application has 88.7% probability of the bandwidth to be between 1.2 Mbps and 1.4 Mbps, with 99% jitter to be in the range below 20 ms and 87.9% of loss up to 10%. The implementation of FTP, which uses the adaptive TCP, has 91.2% chance of presenting average delays up to 100ms, 90.7% loss under 10% and flow rate between 1.6 Mbps and 2.4 Mbps. All levels are acceptable according to international standards (TR-126, 2006).

Test Case 2: Loop length

Here, the inverse analysis will be used, by setting a specific distance and analyzing the behavior of the triple play flow for this situation. The distance of 4500m was used, with

white noise -120 dBm and 24D; that is, the influence of a high-intensity noise at a distance representing 11% (4 to 4.5 Km) of the existing loops of the telecommunication network (Telebras, 1997).

For this distance and noise scenario, it was noticed, from the correlations, that the VoIP communication was impossible due to a high competition for the channel and the lack of dedicated channels for each application. The TCP/IP protocol defines the dispute over the channel was modeled to be fair, in which all applications have the same chance to secure the channel; in this scenario, the video application manage to occupy, almost entirely, the available bandwidth, even with precariously.

The video application can obtain bandwidths from 400 to 550 Kbps, but will suffer a packet loss from 50% to 60% of the total, which is above the maximum stipulated by the TR-126 standard (TR-126, 2006), which is set to 10%. The use of FTP presents an unstable performance, with a throughput rate up to 400kbps and 10% of packet loss.

A solution to this situation is to implement QoS (Quality of Service) on the last mile, which would allow to establish routing priority for the packets; so the VoIP flow can be transmitted, even if other applications suffer from performance and/or from quality losses. The video application would have to be adapted to a new quality of both picture and sound, thus achieving the available bandwidth; codecs such as H.264 (Xiao et al., 2007) enable video compression with high quality and transmission with bandwidths from 256Kbps to 10Mbps.

Optimal State Configuration Search

Here we present the model used to search for the best strategy for implementing or expanding telecommunications services in regions with little infrastructure available to the typical end user, providing a high quality service to users by ensuring that objectives of the network service level will be maintained and correlating events from lower layers of management for determining strategies in the upper layers.

By applying a hybrid model developed using GAs and BNs (Rocha, 2009) we introduce an approach for implementing strategies for capacity planning of networks that were not originally designed for triple play applications. We show that the use of real measures and probabilistic analysis will enable the planning of communication networks, considering logical and physical parameters, such as noise, protocols and triple play applications.

The model characterizes the process of discovering scenarios that can lead to achieving a specific goal. It is aimed at identifying the best configuration, among the possible values (states of nodes in a BN) of variables in the domain, corroborating the achievement of a target value for one (or more) variable(s) in the domain in question.

The interaction between these two computational intelligence techniques (GA and BN) occurs as follows. As can be seen in Figure 8, the process of scenario discovery starts with supplying the BN, generated from the data, and its parameters; then, a GA is applied using as fitness function for the individuals (characterizing the possible scenarios available) the

actual inference engine of the BN; at the end of its iterations, the optimal scenario to achieve a particular goal is obtained.
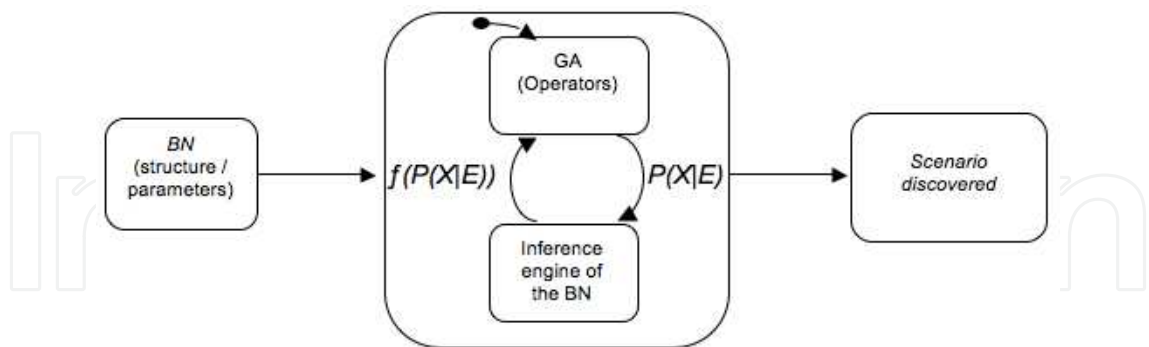

Fig. 8. Representation of the method for discovery of scenarios (Rocha, 2009).

The GA starts with the random generation of an initial population I (where each gene corresponds to a node in the BN), consisting of a set of candidate scenarios, which are then evaluated by the method of inference of the BN; in order to obtain the fitness of the scenarios, the probability of obtaining the target value for the queried variable X, given a particular configuration of states (scenario) of the variables of evidence E is calculated. The process continues with the selection of individuals, through the method of roulette. Next, we apply the operators of crossover, with crossover rate Tc; and mutation, with a mutation rate Tm. The process is repeated for n generations.

With this model we can obtain the best scenario that can derive in a specific target (or set of targets), as well as pointing the main variables and quantifying their contribution for achieving the given goal.

Using the variables obtained from the empirical tests and the BN structure defined, all the attributes were discretized in twenty states, according to the frequency of their values. We follow to apply the method described earlier to search for the best scenario, based on the network attributes, to achieve a desired behaviour for a given attribute.

Here, each of the individuals of the GA represents an inference configuration of the BN, generated randomly. Each individual is then, for its classification, submitted to the Bayesian inference module in order to verify the probability for the chosen behaviour to manifest; obtaining, at the end of the iterations, the best possible scenario of inferences on the BN to achieve desired behaviour for the chosen attribute(s).

Instead of a cost function to validate the individuals of the population, a Bayesian inference algorithm is implemented; that is, the BN is used as a cost function. This way, each of the individuals of the genetic algorithm represents an inference configuration of the BN, generated randomly (e.g. evidencing the variables noise with state 18, Jitter_VoIP with state 1, Loss_VoIP with 7 and Throughput with 4 generates the individual 2-1-7-4). Each individual is then, for its classification, submitted to the Bayesian inference module in order to verify the probability in which the chosen attribute(s) would be maximized, obtaining, at

the end of the iterations, the best possible configuration of inferences on the BN for the maximization of the chosen attribute(s).

At the end of this step (after the genetic algorithm analysis) are obtained only the respective states for this maximization (for each attribute).

We point only one simulated scenario, where the loop length that connects the user to the CO has 4500m with level of noise W= -140 dBm and 24 DSL (ISDN) impairment; which needed a VoIP communication with an acceptable quality (MOS values from 3 to 4). Based on these needs, the attributes Noise and MOS_VoIP were defined with states 18 (4500m plus White Noise of -140 dBm and 24 DSL (ISDN) impairment) and 7 (MOS from 3 to 4).

Using the technique presented was obtained the results above:

Attribute	States
Jitter_VoIP	378 to 425 ms
Loss_VoIP	2.5 to 5 %
Throughput_VoIP	153 to 170 kbps
Attempts_VoIP	2
Loss_Video	0 to 8.7 %
Jitter_Video	28.5 to 30.78 ms
Throughput_Video	0.547 to 0.675 Mbps
Jitter_FTP	0 to 0.1 ms
Loss_FTP	54.6 to 63.7 %
Delay_FTP	0 to 0.1 ms
Throughput_FTP	1.6 to 2.02 Mbps

Table 3. Values of the attributes for the maximization of noise and MOS_VoIP.

The results obtained showed that the inference was possible, but defined some restrictions. For the VoIP communication, only 2 of the 4 VoIP calls made can be successfully maintained. For this, the video application will have a top available bandwidth of 600Kbps. With this, only videos with standard resolution can be transmitted. The FTP application will have 1.6 to 2.02 Mbps of available bandwidth, however with a packet loss from 54 to 63%, considered very high. The results showed the difficult or, in the worst scenario, impossibility of maintaining these applications, unless some kind of QoS (hardware or software) or adjustment in the loop is implemented. With this, the diagnosis needed for complex systems and quickly detection of network failures can be improved and an automatic recovery provided.

6. Final Remarks

DSL (Digital Subscriber Line) technology enables a universal broadband access at a reduced cost and time for implementation required for others access networks since it is considered a large capillarity broadband network, using a combination of the existing telephony infrastructure and digital subscriber line transmission technologies, which are widely available to end users in most developed countries. The environment and the flow to be

transmitted must be analyzed and evaluated, given that the data obtained in this stage can prove applications to be infeasible or, at the very least, to require for an increased investment in infrastructure.

For this reason, the implementation of planning methods to aid in this process, and that take into account the current needs of applications (voice, video and data) are of major importance.

This paper implemented, with the use of crosslayer techniques, strategies for the planning and evaluation of ADSL2+ networks, which implement minimum levels of QoS for Triple Play applications.

The main contribution of this work was to apply computational intelligence methods to extract patterns in last mile DSL networks, studying the behaviour of Triple Play applications on future or already existing networks; especially those with long distances, common in countries with wide geographic area. It then becomes possible to establish more suitable contracts and/or investments with greater security; and provide government managers, in partnership with Telecommunications suppliers with subsidies to better formulate government programs for digital/social inclusion; since the expansion in the provision of Internet access, particularly when it comes to the Amazon region, which still has many areas with no basic communication infrastructure, is an essential factor of development.

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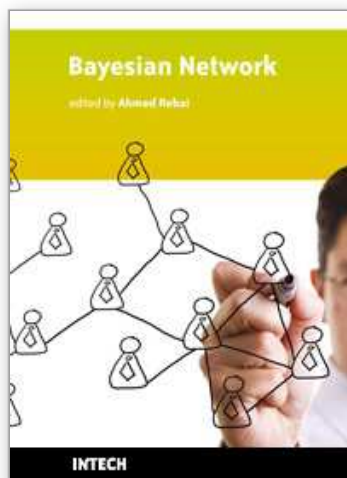
Bayesian Network

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Bayesian networks are a very general and powerful tool that can be used for a large number of problems involving uncertainty: reasoning, learning, planning and perception. They provide a language that supports efficient algorithms for the automatic construction of expert systems in several different contexts. The range of applications of Bayesian networks currently extends over almost all fields including engineering, biology and medicine, information and communication technologies and finance. This book is a collection of original contributions to the methodology and applications of Bayesian networks. It contains recent developments in the field and illustrates, on a sample of applications, the power of Bayesian networks in dealing the modeling of complex systems. Readers that are not familiar with this tool, but have some technical background, will find in this book all necessary theoretical and practical information on how to use and implement Bayesian networks in their own work. There is no doubt that this book constitutes a valuable resource for engineers, researchers, students and all those who are interested in discovering and experiencing the potential of this major tool of the century.

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