Available online at www.elixirpublishers.com (Elixir International Journal)

Network Engineering

Elixir Network Engg. 87 (2015) 35547-35553

ISSN: 2229-712X

Quality of Service and Traffic Management in Mpls Network by Fuzzy Logic Computing

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ARTICLE INFO

Article history:

Received: 4 September 2015;

Received in revised form:

25 September 2015;

Accepted: 01 October 2015;

Keywords

QOS,

MOS,

LSP,

Absolute Delay,

E-Model,

Virtual links,

WAN,

Jitter,

LSR.

ABSTRACT

This research paper suggests a load balancing algorithm using fuzzy logic methodology so that maximum Quality of Service can be attained. Avoidance of jamming of packets is one of the key performance objectives of traffic management in MPLS networks. Load balancing can avoid the congestion caused due to inefficient allocation of network resources. Another feature of the network performance is Quality of Service (QOS). QOS in telecommunications jargon, is a measurement used to determine how well that network is satisfying the end user's requirements. The Mean Opinion Score (MOS) is an important feature in determining the QOS. MOS is a measurement of the quality delivered by the network based on human observation at the destination end. Precisely we can tell average opinion score (MOS) provides a numerical indication of the perceived quality of received media after compression and transmission.

Introduction

Traffic management is a procedure that improves overall network utilization by attempting to create a uniform or differentiated distribution of traffic throughout the network. An important result of this process is the avoidance of congestion on any one path. It is important to note that traffic engineering and management does not necessarily select the shortest path between two devices. It is possible that, for two packet data flows, the packets may traverse completely different paths even though their originating node and the final destination node are the same. This way, the less exposed or less-used network segments can be used and differentiated services can be provided. In MPLS, traffic engineering is inherently provided using explicitly routed paths. The Label-switched paths (LSPs) are created independently, specifying different paths that are based on user-defined policies. Multi-protocol label switching (MPLS) is a flexible solution to address the problems faced by present-day networks-speed, scalability, quality-of-service (QOS) management and traffic engineering. MPLS has emerged as an elegant solution to meet the bandwidth-management and service requirements for next-generation Internet protocol (IP)–based core networks. MPLS addresses issues related to scalability and routing (based on QOS and service quality metrics) and can exist over existing asynchronous transfer mode (ATM) and frame-relay networks. Avoidance of congestion is one of the major performance objectives of traffic engineering in MPLS networks.

Load balancing can prevent the jamming caused due to inefficient allocation of network resources. Another aspect of the network performance is Quality of Service (QOS). QOS in telecommunications jargon, is a measurement used to determine how well that network is satisfying the end user's requirements. The Average Opinion Score (MOS) is an important factor in determining the QOS. MOS is a measurement of the quality delivered by the network based on human observation at the destination end. Precisely we can tell average opinion score (MOS) provides a numerical indication of the perceived quality of received media after compression and transmission.

Theory

A Multi-Protocol Label Switching

In computer networking and telecommunications, Multiprotocol Label Switching (MPLS) refers to a mechanism which directs and transfers data between Wide Area Networks (WANs) nodes with high performance, regardless of the content of the data. MPLS makes it easy to create "virtual links" between nodes on the network, regardless of the protocol of their encapsulated data. The growing number of computer users on the Internet and intranets, as well as new bandwidth intensive applications such as those incorporating voice and video, are driving the need for guaranteed bandwidth and increased network reliability. The typical frame 8520/and packet-based networks lack the quality of service (QOS) and traffic shaping sophistication of the powerful yet expensive ATM networks. Furthermore, the proliferation of network protocols increases the complexity and reduces network capability and performance. In an effort to increase throughput, reduce network complexity in ATM networks, and bring advanced bandwidth shaping and QOS capabilities to non-ATM networks, the Internet Engineering Task Force (IETF) created Multiprotocol Label Switching (MPLS).MPLS combines the power of layer 2 switching with the flexibility and intelligence of layer 3 protocols; it operates independently of other network technologies but is fully capable of interoperating with them. MPLS brings non-ATM networks powerful QOS capabilities, the ability to route multiple network technologies (Ethernet, Frame Relay, ATM) over one infrastructure and the capability of

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interoperating with modern routing protocols such as RIP, OSPF and BGP, while increasing efficiency and simplifying network organization. Multiprotocol label switching (MPLS) is a versatile solution to address the problems faced by present-day networks—speeds, scalability, quality-of service (QOS) management and traffic management. MPLS has emerged as an elegant solution to meet the bandwidth-management and service requirements for next-generation Internet protocol (IP)–based backbone networks. MPLS addresses issues related to scalability and routing (based on QOS and service quality metrics) and can exist over existing asynchronous transfer mode (ATM) and frame-relay networks. [1] and [2]. The components that participate in the MPLS protocol mechanisms can be classified into label edge routers (LERs) and label switching routers (LSRs). An LSR is a high-speed router device in the core of an MPLS network that participates in the establishment of LSPs using the appropriate label signaling protocol and high-speed switching of the data traffic based on the established paths.

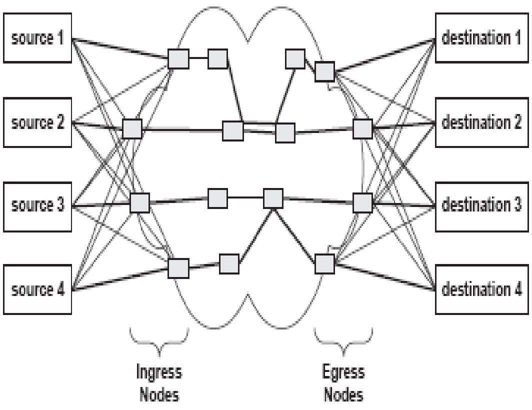


Figure 1

An LER is a device that operates at the edge of the access network and MPLS network. LERs support multiple ports connected to dissimilar networks (such as frame relay, ATM, and Ethernet) and forwards this traffic on to the MPLS network after establishing LSPs, using the label signalling protocol at the ingress and distributing the traffic back to the access networks at the egress. The LER plays a very important role in the assignment and removal of labels, as traffic enters or exits an MPLS network.

Forward Equivalent Class

The forward equivalence class (FEC) is a representation of a group of packets that share the same requirements for their transport. All packets in such a group are provided the same treatment and route to the destination. As contrasting to conventional IP forwarding, in MPLS, the assignment of a particular packet to a particular FEC is done just once, as the packet enters the network. FECs are based on service requirements for a given set of packets or simply for an address prefix. Each LSR builds a table to specify how a packet must be forwarded. This table, called a label information base (LIB), is comprised of FEC–to-label bindings.

Labels and Labels Binding in Mpls Network

A label, in its simplest form, identifies the path a packet should traverse. A label is carried or encapsulated in a Layer-2 header along with the packet. The receiving router examines the packet for its label content to determine the next hop. Once a packet has been labelled, the rest of the journey of the packet through the backbone is based on label switching. The label values are of local significance only, meaning that they pertain only to hops between LSRs.

Once a packet has been classified as a new or existing FEC, a label is assigned to the packet. The label values are derived from the underlying data link layer. For data link layers (such as frame relay or ATM), Layer-2 identifiers, such as data link connection identifiers (DLCIs) in the case of frame-relay networks or virtual path identifiers (VPIs)/virtual channel identifiers (VCIs) in case of ATM networks, can be used directly as labels. The packets are then forwarded based on their label value. Labels are bound to an FEC as a result of some event or policy that indicates a need for such binding. These events can be either data-driven bindings or control-driven bindings. The latter is preferable because of its advanced scaling properties that can be used in MPLS. Label assignment decisions may be based on forwarding criteria such as the following:

(1) Destination Unicast Routing
(2) Traffic Engineering and Management
(3) Multicasting
(4) Virtual private network (VPN)
(5) QOS

Label-Switched Paths (Lsps)

A collection of MPLS–enabled devices represents an MPLS domain. Within an MPLS domain, a path is set up for a given packet to travel based on an FEC. The LSP is set up prior to data transmission. MPLS provides the following two options to set up an LSP.

Hop-by-Hop Routing

Each LSR independently selects the next hop for a given FEC. This methodology is similar to that currently used in IP networks. The LSR uses any available routing protocols, such as OSPF, ATM private network-to-network interface (PNNI), etc.

Explicit Routing

Explicit routing is similar to source routing. The ingress LSR (i.e., the LSR where the data flow to the network first starts) specifies the list of nodes through which the ER–LSP traverses. The path specified could be non-optimal, as well. Along the path, the resources may be reserved to ensure QOS to the data traffic. This eases traffic engineering throughout the network, and differentiated services can be provided using flows based on policies or network management methods.

Label Distribution Protocol (LDP)

The LDP is a new protocol for the distribution of label binding information to LSRs in an MPLS network. It is used to map FECs to labels, which, in turn, create LSPs. LDP sessions are established between LDP peers in the MPLS network (not necessarily adjacent). The peers exchange the following types of LDP messages:

1. Discovery messages--announce and maintain the presence of an LSR in a network.

2. Session messages--establish, maintain, and terminate sessions between LDP peers. 3. Advertisement messages **-** create, change, and delete label mappings for FECs.

4. Notification messages**--**provide advisory information and signal error information.

Label Stacks in Mpls Network

The label stack mechanism allows for hierarchical operation in the MPLS domain. It basically allows MPLS to be used simultaneously for routing at the fine-grain level (e.g., between individual routers within an Internet service provider [ISP] and at a higher domain-by-domain level). Each level in a label stack pertains to some hierarchical level. This facilitates a tunnelling mode of operation in MPLS.

Table 1. Audio Visual Quality Classes

MOS	User Satisfaction
4.3 to 5	Very satisfied
4 to 4.3	Satisfied
3.6 to 4	Some users dissatisfied
3.1 to 3.6	Many users dissatisfied
2.6 to 3.1	Nearly all users dissatisfied
1 to 2.6	Not recommended

Table 2. Relation between R, MOS and User Satisfaction

R-value(lower limit)	MOS(lower limit)	User Satisfaction
90	4.34	Very satisfied
80	4.03	Satisfied
70	3.60	Some users dissatisfied
60	3.10	Many users dissatisfied
50	2.58	Nearly all users dissatisfied

Traffic Engineering and Management

It is the process that enhances overall network utilization by attempting to create a uniform or differentiated distribution of traffic throughout the network. An important result of this process is the avoidance of congestion on any one path. It is important to note that traffic engineering does not necessarily select the shortest path between two devices. It is possible that, for two packet data flows, the packets may traverse completely different paths even though their originating node and the final destination node are the same. This way, the less exposed or less-used network segments can be used and differentiated services can be provided. In MPLS, traffic engineering is inherently provided using explicitly routed paths. The LSPs are created independently, specifying different paths that are based on user-defined policies. However, this may require extensive operator intervention. RSVP and CR–LDP are two possible approaches to supply dynamic traffic engineering and QOS in MPLS. The Current Internet Gateway Protocols (IGP) uses the shortest paths to forward traffic. Using shortest paths conserves network resources, but it causes some resources of the network to be over utilized while the others remain underutilized. The shortest paths from different sources overlap at some links, causing congestion on those links. The traffic from a source to a destination exceeds the capacity of the shortest path, while a longer path between these two routers is under-utilized. The purpose of traffic engineering [3, 4] is to enhance network utilization and to improve the architecture (topology and link capacity) of a network in a systematic way, so that the network is robust, adaptive and easy to operate. An efficient traffic engineering solution shares the data traffic load with the routers, nodes and switches in the network, making none of its individual components either over utilized or underutilized, thus assuring satisfactory service delivery and optimizing resource efficiency.

MPLS has the extended routing capability that supports applications, which requires more than destination-based forwarding. Figure 2 illustrates that MPLS provides an efficient control of network traffic by easing congestions and spreading the load over the layer 2 links. There exist services where some links are reserved for certain classes of traffic or for particular set of users.There are two ways to set up a Label switch path within a MPLS network, control driven or explicitly routed. Control driven LSP can be set up by hop-by-hop routing or LDP (Label distribution protocol), which involves setting up a connection thru UDP and TCP. Second approach is ER-LSP.

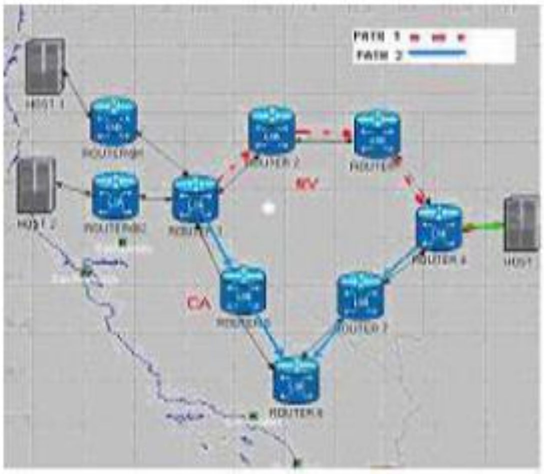


Figure 2. Routing Functionality in MPLS

An Explicit route is a small sequence of hops from ingress to egress LSRs to set up an LSP. This explicit route can contain several hops within the set of many nodes, within an MPLS environment before emerging to the next hop specified in the Explicit Route. Explicit routing helps in diversion of network traffic around failed links and helps in providing already set up LSP-backup to maintain uninterrupted flow. Explicit routing has its significance to force an LSP, which differs from the one offered by the routing protocol. Constraint-based Routing (CBR) [8] computes routes that are subject to constraints such as bandwidth and administrative policy. Using a combination of the metrics defined for traffic engineering and the capabilities of routers, constraint-based routing substantially reduces the requirements for operator activity necessary to implement TE. Because Constraint-based Routing considers more than network topology in computing routes, it may find a longer but lightly loaded path better than the heavily loaded shortest path. Network traffic is hence distributed more evenly.

Quality of Service (QOS)

QOS is the overall performance of the system from the point of view of the users. It is the measure of end-to-end performance at the service level from the user perspective and an indication of how well the system meets the user’s needs. Quality of Service (QOS) also refers to a set of technologies (QOS mechanism) that enable the network administrator to manage the effects of congestion as well as providing differentiated service to selected network traffic flows or to selected users. In order to deliver acceptable service quality, QOS targets should be established for each service and be included early on in system design and engineering processes.

QOS for the end user is essential and will be a key differentiator with respect to competing service offerings. Subscribers to network services don't care how service quality is achieved. What matters to them is how well a service meets their goals and expectations their Quality of Service (QOS).To achieve this end, the QOS engineering process involves the following steps.

- Defining the QOS matrices and targets.
- Identify QOS contributing factors and dependencies (delay, jitter and loss).
- Traffic engineering and resource allocation. This involves a control policy for routing, budget allocation etc. In our case we will introduce a load balancing algorithm to optimize the impairments and jitter in order to maximize the QOS.

As an important measure of the end-to-end performance at the services level from the user's perspective the QOS is an important metric for the design of systems and engineering processes. This is particularly relevant for video services because bad network performance may highly affect the user's experience, mainly because these services are compressed and have low entropy. So, when designing systems the expected output, i.e. the expected QOS, is often taken into account also as a system output metric. This QOS metric is often measured at the end devices and can conceptually be seen as the remaining quality after the distortion introduced during the preparation of the content and the delivery through the network until it reaches the decoder at the end device. There are several elements in the video preparation and delivery chain and some of them may introduce distortion. This causes the degradation of the content and several elements in this chain can be considered as "QOS relevant" for video services. These are the encoding system, transport network, access network, home network and end device. The concept of QOS in engineering is also known as Perceived Quality of Service (PQOS), in the sense of the QOS as it is finally perceived by the end-user.

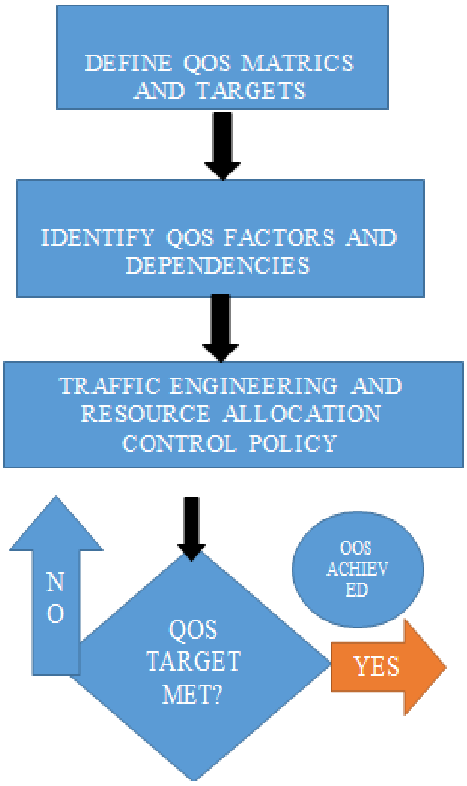
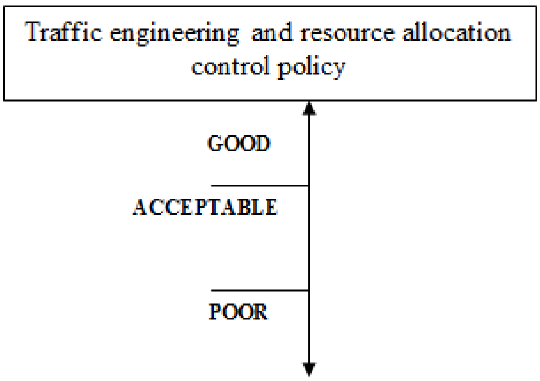


Figure 3. Engineering Process of QOS

The evaluation of the PQOS for audio-visual content will provide a user with a range of potential choices, covering the possibilities of low, medium or high quality levels. Moreover the PQOS evaluation gives the service provider and network operator the capability to minimize the storage and network resources by allocating only the resources that are sufficient to maintain a specific level of user satisfaction. The evaluation of the PQOS is a matter of objective and subjective evaluation procedures, each time taking place after the encoding process (post encoding evaluation). Subjective quality evaluation processes (PQOS evaluation) require large amount of human resources, establishing it as a time-consuming process. Objective evaluation methods, on the other hand, can provide PQOS evaluation results faster, but require large amount of machine resources and sophisticated apparatus configurations. Towards this, objective evaluation methods are based and make use of multiple metrics.

Quality of Service (QOS) Metrices

There are two popular methods to accesses audio-visual quality: Subjective quality assessment and objective quality assessment. Subjective quality assessment means playing a sample audio-visual clips to a number of participants. Their judgment of the quality of the clip is collected and used as a quality metric. Objective quality assessment does not rely on human judgment and involves automated procedures such as signal-to-noise ratio (SNR) measurement of original and reconstructed signals and other sophisticated algorithms such as Mean Square Error (MSE) distortion, Frequency weighted ASE, Segmented SNR and E-model to determine quality metrics. The method we are going to adopt in our work is the E-model. The problem with Subjective quality assessment techniques is that human perception of quality is based on individual perception, which can vary significantly between a given set of individuals. The problem with objective quality assessment technique is that they may not necessarily reflect the actual end user experience, in either case the output of these measurements is AVERAGE OPINION SCORE (MOS) which ranks the audio-visual quality on a scale of 1 to 5.here our objective is to maximize the MOS which is our metric. The table 1 below shows the audio-visual quality classes for different MOS values [4].



Factors Affecting (QOS) Metrices

An important aspect in our problem is identifying the network parameters that affect QOS the most and knowing the relative impact of these parameters on the QOS. Extensive studies have been carried out in [4] [5] and [13] where it has been established by objective and subjective experiments on audio-visual traffic, that the variables which affect the MOS ranking the most, are the dynamic network changes caused by route fluctuation, competing traffic and congestion .This network dynamics can be characterized by 3 network metrics namely delay jitter and loss.Delay is defined as the amount of time that a packet takes to travel from sender's application to the receiver's destination application. It is recommended by [6] and also verified by [4] [5] that delay bounds for the various grades of perceived performance in terms of Human interaction can be defined as: GOOD (0ms –150ms), ACCEPTABLE (150ms-300ms), POOR (>300ms). Jitter is defined as the variation in the delay of the packets arriving at the receiving end. It is caused due to congestion at various point in the network,varying packet sizes that result in irregular processing times of the packets and other such factors. Loss is defined as the percentage of transmitted packet that never reach the intended destination due to deliberately discarded packets or non – deliberately by intermediate links, nodes, and end–systems. It is suggested that loss more than 1% can severely affect audio-visual quality.

E-Model

The algorithm we will use exploits the E –model as recommended in ITU-T G.107 [6] which returns a value for —Rating factor R which offers an estimate of the user option called the QOS. The E- model is a well established computational model that uses the transmission parameter to predict the subjective quality .It uses a R scale whose value is from 0 to 100 and can be mapped to MOS rankings and user satisfaction as shown in the table 2 [6]

The purpose of the model is:

- to predict the subjective effect of combinations of impairments using stored information on the effects of individual impairments
- to help network planners design networks
- to replace hierarchical models and apportionment, which are difficult to apply in a liberalized market.

The basic equation for the model is:

R = Ro – Is– Id – Ie + A

Where:

Ro = Basic signal-to- noise ratio

Is = Impairments simultaneous to voice signal

Id = Impairments delayed after voice signal

Ie = Effects of special equipment e.g. codecs

A = Advantage factor (to take account of user advantages such as mobility)

The R factor can be mapped to MOS by the formula,

For R>0: MOS=1

For 0<R<100: MOS=1+0.035R+R(R-60) (100-R) 7.10-6

For R>100: MOS=4.5

As this quantity only affects Id, the objective function is characterized assuming default values for all other impairments except Id and that leads to the following expression:

$$Rd = 94 - Id \qquad \boxed{1}$$

$$Id = 25\left\{\left(1+X^6\right)^{\frac{1}{6}} - 3\left(1+\left(\frac{X}{3}\right)^6\right)^{\frac{1}{6}} + 2\right\} \qquad \boxed{2}$$

$$X = \frac{\lg\left(\frac{Ta}{100}\right)}{\lg 2} \qquad \boxed{3}$$

Where Ta is the delay, assuming M/M/1 model, the average delay is expressed by the following

$$Ta = \sum_i \left(\frac{\lambda i}{\lambda s}\right) / \left(\frac{\mu}{\lambda i}\right) \qquad \boxed{4}$$

Where i is the index number of the ith LSP operating between the ingress and the egress router.

By substituting (4) and (2) in (1), the rating factor can be characterized as the function of the traffic rate (λ).The problem is now a maximization problem:

Maximize,

$$f(\lambda) = 94 - 25\left[\left(1+\left(\left(\log\left(\sum_i (\lambda_i/\lambda_s)/(\mu-\lambda_i)\right)/100\right)/\log 2\right)^6\right)^{1/6} - 3\left(1+\left(\left(\log\left(\sum_i (\lambda_i/\lambda_s)/(\mu-\lambda_i)\right)/100\right)/\log 2\right)/3\right)^6\right)^{1/6}\right] \qquad \boxed{5}$$

$$\text{Subject to} \quad \lambda s = \sum_i \lambda i \qquad \boxed{6}$$

$$\forall i \in \text{the IE pair and } \lambda i \geq 0$$

Results and Discussion

In the simulation process, we tried to compare the graphs and results obtained with the defined standards of ITU-T

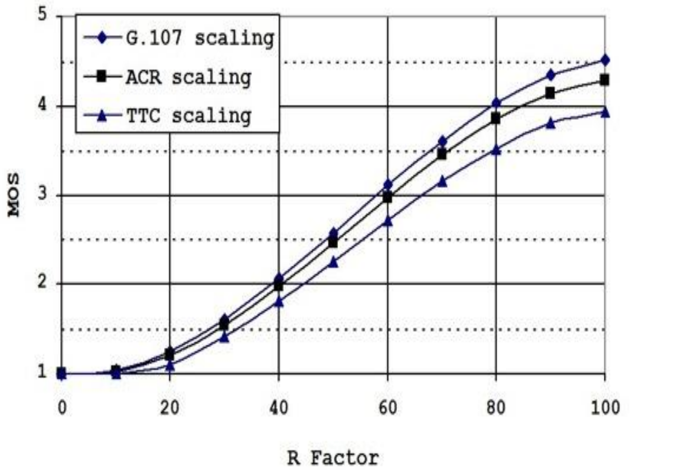


Figure 4. ITU-T G.107 relation between MOS and R Factor

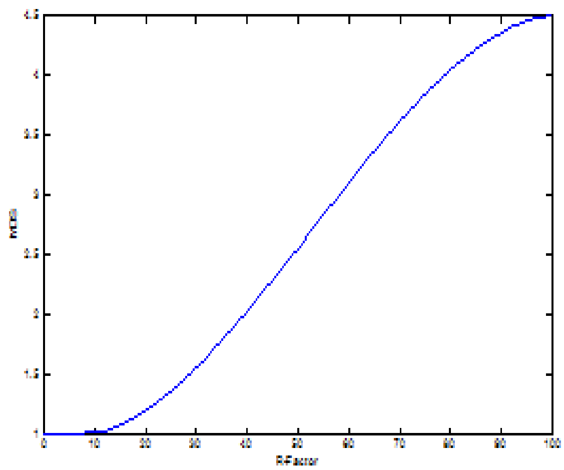


Figure 5. Simulation Relation between MOS and R

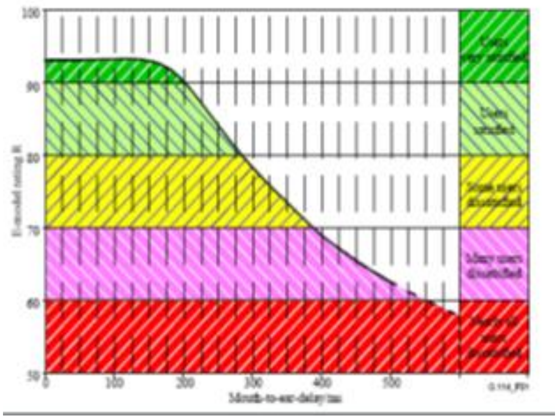


Figure 6. ITU-T G.108 Relation between Absolute Delay and R Factor

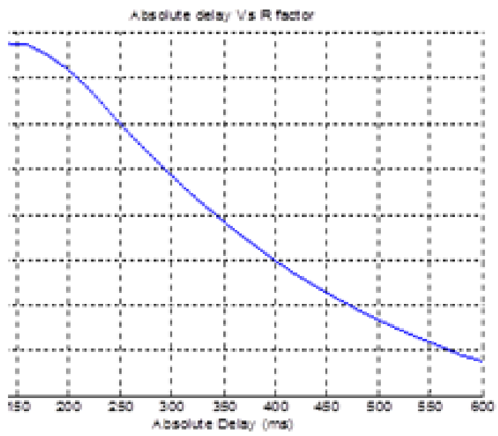


Figure 7. Plot shows Relation between Absolute delay and R Factor

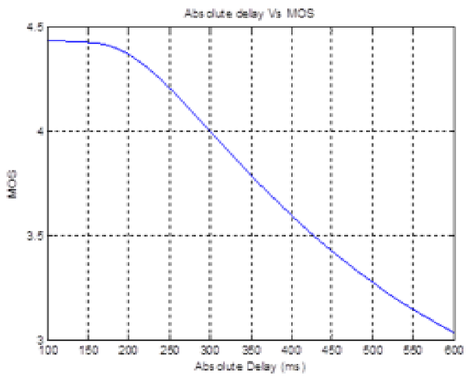


Figure 8. Plot shows Relation between Absolute Delay and MOS

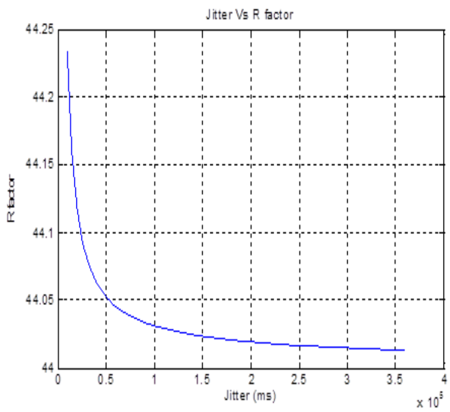


Figure 9. Plot shows Relation R factor and Jitter

Conclusion

Since it is difficult to develop as many communication solutions as possible applications, the scientific and technological communities aim towards providing general services allowing to give to each application or user a set of properties now-a days called —Quality of Service (QOS), a terminology lacking a precise definition. This QOS concept takes different forms according to the type of communication service and the aspects which matter for a given application: for performance it comes through specific metrics (delays, jitter, throughput), for dependability it also comes through appropriate metrics: reliability, availability; vulnerability for instance in the case of WAN (Wide Area Network) topologies, etc. At the conclusion we can tell that QOS is a subjective measure of performance in a system. QOS relies on human opinion and differs from quality of service (QOS), which can be precisely measured. For example, a person's reaction to listening to music through headphones is based not only on the frequency response of the system and the speakers, but the comfort of the unit and the individual's hearing sensitivity.

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