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Qos and Voice Over IP

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As Voice over Internet Protocol (VoIP) technology matures, companies are increasingly adopting it to cut costs, improve efficiency and enhance customer service. Using the Internet as an existing network for integrating data and telecom systems through intelligent VoIP, a range of benefits results: lower long distance costs, cost cuts in cabling processes and more flexible telephony management. However, as voice over IP services grow in popularity, major threats arise: this rapid growth leads to traffic congestion, security is jeopardized and the poor quality of calls affects communication.
The objective of this article is to present all the elements that can affect voice quality in a VoIP network and to provide methods for solving them. A detailed analysis to minimize the impact of implementation of QoS will be made, and at the end solutions to management strategies will be proposed.

Keywords: *QoS, Jitter, MOS, PESQ*

Introduction

Due to a convergent network's characteristics, there have appeared several problems regarding the real-time transport of voice traffic, while maintaining a good reception.

The clarity of the audio signal and the lack of reception interferences are a priority when offering quality services. The listener needs to be able to determine the speaker's id as well as his momentary state.

The main factors that influence the voice quality are the echo, the jitter and the delay. The problems that these factors create are solved by implementing mechanisms that provide QoS guarantees. The factors that affect the voice signal are:

Fidelity The degree to which a system or a part of a system reproduces with accuracy at the exit the main characteristics of the signal that has entered. The band width of the transmission environment limits the human voice's characteristic width (100 Hz – 10 kHz). Still, 90% of the vocal signal components are between 100 Hz and 3000 Hz.

Echo It appears due to impedance difference on the transmission way. The echo is present in any network, even in the classic telephony, but at a lower level, that cannot be identified by the human ear. The two components that define echo are amplitude (signal level or echo intensity) and delay (the time difference between the direct and the reflected wave). To cancel the echo's effect different devices are implemented in the network (echo cancellers).

Jitter It is defined as the variation of the delay in the voice packages reaching destination. This variable time difference may determine interruptions in the voice signal. [7]

At the emission the packages are sent in a continuous flow, with a constant spacing between packages. Due to the blockages that may appear in the network, the traffic being slowed by transiting buffers or due to configuration errors, the packages flow may become inconstant. More precisely, the packages spacing, constant initially, becomes variable. This situation is presented in figure 1.

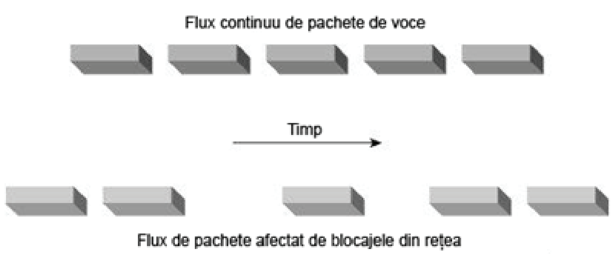

Figure 1: Jitter

When a router receives an audio package flow, it must call on different mechanisms to cancel the encountered jitter. Usually this delay is canceled by introducing buffer memories for the temporary stockage of voice packages (dejitter buffer). The procedure is receiving packages [2] in the order they arrive in the network and arranging them in a continuous flow with a constant spacing between packages. Thus arranged, the packages are received and processed by the digital signal processors (DSP) in order to obtain the analog voice signal.

Delay It is defined as the time that passes between the time that the voice signal is sent and the moment of arrival at destination. While projecting a network that transports voice packages, it is important to identify all the sources producing the delay. Also, there must be taken into account the predictable delays, in order to ensure an acceptable functioning of the network.

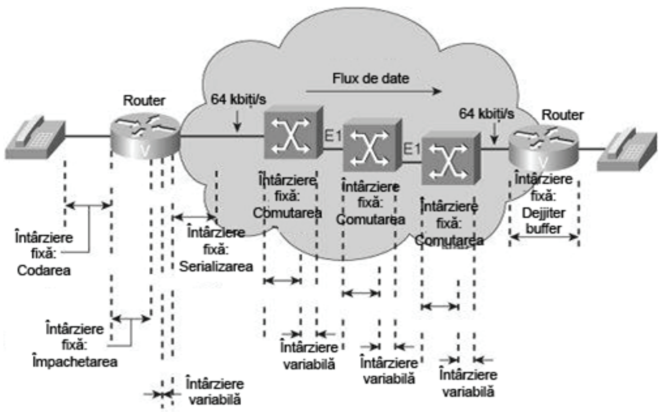

Figure 2: presents several types of delays

Fixed delay – represents a sum of predictable delays that are added directly to the total network delay. It includes the following components:

Coding – the time it takes for the transformation of the analog signal in digital signal

Packaging – the necessary time for the arrangement of the cadres in packages and the extraction of the information from the packages.

Serialization – the necessary time for the introduction of the information bytes in the transmission environment (the communication path);

Propagation – the necessary time for the voice packages to travel the communication path from one end to another.

Variable delay – caused by the delays that appear at the “lines” in the

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junction buffers that are located in the serial wan connection port. These buffers create a variable delay (jitter) in the network.

Quality Asessment Methods

In order to create a differencing of voice services based on quality, there have been developped three methods, by which there are granted qualifications based on the performance. These are:

MOS – Mean Opinion Score

MOS is one of the methods of assessing the voice quality. In order to grant MOS qualifications listeners assess several recorded phrases, that have been previously passed through different processes like compression algorithms. The phrases used in these evaluations contain a large array of sounds, for a correct assessment of the tested system performance. Listeners grant each phrase a value on a scale from 1 to 5, 5 being the maximum value.

An average is formed from the obtained degrees which sets the final points. MOS results are subjective, because they are based on listeners opinion. Also, the points indicated by one group cannot be compared to the same points accorded by another group.

PSQM* - P*erceptual* *Speech* *Quality* M*easurement

PSQM is a method for measuring voice quality in real time. PSQM software is usually found in the call management system, that are in their turn integrated in the SNMP system (Simple Network Management Protocol)

PSQM measurement is made by comparing original voice signal to the reception signal. Measurement is made during the call. This algorithm for automated testing has an over 90% accuracy compared to the subjective methods like MOS. Asessment is made on a scale from 0 to 6.5, where 0 is the best score and 6.5 is the lowest.

PSQM was projected for the voice networks with circuit commutation, and for this reason, its evaluation does not take into account jitter or the delay that appears in the voice networks with packaging commutation.

PESQ* - P*erceptual* E*valuation of* *Speech* *Quality

MOS and PSQM methods are not recommend for present VOIP networks. Both have been projected before the wide scale spread of VOIP technology and do not take into account specific problems like delay or jitter. Thus, a score of 3.8 can be obtained for a VOIP network while the delay in one way is larger than 500 ms, because a listener will asses only the quality of the audio material.

PESQ method, whose functioning is presented in figure 3.3 was initially developed by British Telecom, Psytechnics and KPN Research. Previously it evolved under the supervition of ITU in the P.862 standard.

PESQ takes into account the CODEC errors, filtering errors, jitter problems and specific delays of a VOIP network. PESQ is a combination of two methods: PSQM and PAMS (Perceptual Analysis Measurement System). The value sclae of PESQ is from 1 (very low) to 4.5 (very good), where 3.8 represents the acceptable quality in the traditional telephony network. PESQ measures a single aspect of voice quality. This method also does not reflect in its score the effects of two way communication like: loss of signal intensity, delay, echo. PESQ was designed such as its results will reflect as accurate as possible the points obtained at the MOS evaluation.

Objectives Of QoS Mechanisms

In order for VOIP technology to replace definitively the services probided by the public service telephony network (PSTN), it is necessary for the VOIP service users to be able to talk using a similar or better quality network, compared to the classic telephony network.

As other applications that function in real time, VOIP applications are very sensible to problems caused by bandwidth and delay. To be certain that VOIP transmissions are audible for the final user, there must be avoided loosing packages, excessive delay or jitter.

VOIP guarantees a quality transmission of voice only if the signaling and voice packages have priority before other types of network traffic. A correct implementation of VOIP technology provides an acceptable level of voice quality, if there are respected the traffic conditions connected to bandwidth, delay and jitter.

By implementing QoS mechanisms, better qualitative services can be provided and also a higher predicatbility degree in the functioning of the network. For this, the following criteria needs to be observed:

• *Providing a dedicate bandwidth* – designing the network as well as it will permanently provide sufficient resources for voice and data transport

• *Minimizing losses* – projecting a Frame Relay network such as the priority in packets throwing will not affect the voice packages, maintaining the voice trafiic below the CIR threshold (Committed Information Rate)

• *Avoiding traffic bolckages in the network* – supposes the networks infracstructure will be able to take a large volume of information

• *Setting priorities for network traffic* – voice packages have priority before data traffic [56]

IntServ mechanism defines an architecture that demands controll of flow traffic in any network bump along the end-to-end path and a treatment of flows according to specific demanads imposed by ensuring quality for different types of applications. Differential treatment of flows imposes their classification according to QoS demands and sending these requirements over the network by a signaling protocol. The signling protocol employed by the QoS IntServ mechanism is RSVP (Resource Reservation Protocol). This is used for explicit signaling the requirements of each flow and for the controll of the reservation and maintaining the resource reservation for flows.

An IntServ flow is defined as a set of identifiable packages that are sent from one or more destinations, by which a common QoS treatment is demanded. A string of packages having the following identical parameters: source address, destination address, protocol identifier, source and destination port, is an example of IntServ flow. The IntServ flow is a stream of datagrams that result from one users activity and demands the same QoS. This is a simplex flow that goes only in one direction. For a video conference between N users there will be necessary N flows for each source to all participants.

IntServ imposes resource reservation between networks, that some users benefit privileged services and imposes authentication requirements: identification of users that make reservation demands and authentication of packages that require such resources [27.43]

Due to the fact that Internet is proposed to be used as a common infrastructure to support both real-time communications as well as non-real time, it must use a stave model of unified protocols, that imply only one protocol at the Internet level both for real-time as well as for other services.

IntServ model supposes that resources can be explicitly controlled. It also provides control of the accessibility to resources trough controll of admission, that can accept serving the application if there are available resources and secures their reservation, or due to lack of resources it will not allow serving the application.

Intermediary routers provide the resource reservation based on specific information, included in RSVP messages, that characterize the flow's demands.

In projecting IntServ model, it was established as work hypothesis, in order to satisfy the applications demands, the possibility of explicitly administrate the network resources. "Resource reservation" and "controlled access to resources" are key elements of this type of service. Resource reservation is aimed at providing for the user access to a service that has a quality specifications predictable enough that the application should run in an acceptable way, in a time that is desired by the user. [9,8]

IntServ architecture is based on reservation of individual flow resources.

To benefit from the providing of resources, an application must make a reservation before transmitting the network traffic. The application must make a specification of the traffic and demand for resources. The network uses a routing protocol to find a way that provides the demanded resources. Than a resource reservation protocol is used to install the reservation state along this path. In each bump, admission control checks if there are sufficient resources to accept the new reservation. After making the resource reservation the application may begin transmitting the traffic through the path that was reserved. For the traffic flow a classification of the packages is provided as well as mechanisms for planning in the network nodes used by the path previously defined [5,7]. The differentiation of services is aimed of providing the network's adapting to the diverse demands of applications and to the users expectancies. DiffServ also allows differential charging for internet services.

IntServ is a service model that allows guaranteeing QoS flow, but demands resource reservation on the flow and complex control architectures along each end-to-end path.

DiffServ appears as a reaction to the complexity of IntServ architectures and its end-to-end nature.

In DiffServ architecture processing and complex decisions (eg. Classifcation) are made at a network's border, in the "edges" nodes, and the QoS controll is provided by using a restraing set of guidance classes in the Core routes. These DeffServ characteristics determine the appearance of faster Core routers and the reduction of signaling, processing and memorizing states (QoS characteristics can be expressed in the terms of a small number of guidance classes).

Contrasting to the flow-orientation used by the IntServ mechanism, QoS controll through DiffServ is made by grouping packages in a small number of flows combined depending on the type of service (TOS) and destination. These combinations will be classified in classes of guidance that will have associated a DiffServ code called DSCP (DiffServ Code Point), placed in the heading of IP packages. This is known as Behavior Aggregate Classification (BA).

In the core network DiffServ domains are defined, which can control QoS inside the domain (Edge-to-Edge QoS).

The marking of packages through defining DSCP in the border nodes of the network allows the mapping of a large variety of traffic in a reduced set of behaviours (BA) supplied by the core network. Border routers may also provide different traffic control policies, providing a control of the temporal characteristics of the traffic classes from the whole core network.

DiffServ divides the trafic in a small number of groups called forwarding classes. A forwarding class contains all the packages that have the same DSCP code in the header of the IP packet. [3]Each guidance class has a guiding treatment associated, defined by its priority in the throwing of packages (drop priority) and in the bandwidth allocation.

Implementing QoS. Minimizing impact and enhancing voice quality

The purpose for implementing mechanisms that provide QoS guarantees is mainly that of minimizing the impact that jitter and delay have on the network's functioning. We will be presenting a series of QoS functions implemented in the routers operating system.

For the coordination of packages at the buffer exit, there can be used the following planning disciplines:

CBWFQ (Class-based weighted fair queuing) – extends the functioning allowing to define traffic classes, based on the transported information.

LLQ (Low latency queuing) – together with CBWQ ensures the strict priority of voice packages. LLQ establishes the priority for classes defined in CBWFQ.

WFQ (Weighted fair queuing) – cumulates the different sorts of trafic into an information flow and allocates it depending on the available bandwidth

WRED (Weighted random early detection) – ensures differentiated performance characteristics for different classes of services. WRED especially renounces at the lower-priority traffic in front of the higer priority traffic when the exit interface gets crowded.

WAN specific protocols can use the following QoS mechanisms:

Class-based selection – provides the transfer rate limiting function based on the available band width. In the same time, the distribution policy is established for traffic that may pass over the allocated bandwidth

Traffic-modeling – intentionally delays traffic excess by using buffers (to withhold packages), in this way modeling the data flow when the source transmission rate is larger than expected.

FRF. 12 (Frame Relay Forum 12) – enhances traffic for low speed Frame Relay connections (<768 kbytes/s) by interposing voice packages with fragments from larger frames.

MLP (Multilink PPP) – allows large packages to be encapsulated separately for more connections, thus being fragmented and interrelated with other packages. By fragmenting there have been obtained smaller packages that can respect delays imposed in the real-time traffic.

VOIP traffic can use the following QoS mechanisms

cRTP (Compressed Real-Time Transport Protocol) – RTP is a protocol that ensures real-time transport of all types of traffic. RTP uses an extended heading that incorporates time information (timestamp) for each package.

CRTP function provides a compression for the extended heading. The advantage is the reduction of the necessary bandwidth and a corresponding reduction of the dealy

RSVP (Resource Reservation Protocol) – ensures the network resources conservation, allowing the users to employ those resources for complying with the QoS criteria. For VOIP networks, RSVP together with the mechanisms that ensure traffic modulation and call signaling secure the network access (CAC) [7]

In order to be able to sustain a data transmission in real time, a commutation packages network must fulfill several requirements, all of them grouped generically under the name of Quality of Service – QoS)
For voice traffic, the most important characteristics that must be fulfilled are network delay and packages loss.

Management Strategies

According to the IT specialists' reports in the last 4 years, there is an increased trend of MSP technology usability. Like in any activity domain, there will always be organizations choosing for software monitoring and an internal system management. Thus, if opting for MSP to assure consistent and solid solutions at a high quality level, it is predictable for the results to show rapidly. Many companies, to achieve success, start from creating a multidisciplinary project that can establish the parameters of this implementation process. It is of highly importance to determine from the beginning of the evaluation the impact that would suppose to have the new applications towards the performances of the existing network. This way it is easier to investigate success (or the failure, depending on the case). On strict terms of implementation, the first step is indentifying a group of beta users inside the company to serve as a test cell for initial development, afterwards for the migration of the users, the identification of problems that appear and, at the end, the development of the communicational global solution.

To chose the vendor that provides the perfect solution, it should be taken into consideration the next aspects:

• Selecting a partner who can understand the characteristics of the organization's activities;

• Proceeding with a further test of the vendor;

• Communicating the results expected after implementing the chosen solution and determining the total cost of the investment;

• Establishing the parameters for success;

• Evaluating the vendor regarding certifications and training processes;

• Understanding the sub-contractual policies of the contract;

• Evaluating the data security policies;

• Establishing a list of names and contact information to assure maintenance and repair processes;

• Understanding and negotiating advantageous terms of service level agreements.

In order to fulfill each and every one of these aspects, it is of high importance for the company to know the tools used for implementing the technological solution, and the method of conciliation with the work environments of the other vendors.

Although the development and implementation of VoIP is still at the beginning, the managers of the companies want more and more the deployment of a management based on the tools and services that these technologies offer. Even without mentioning about their standardization, the real-time management of integrated applications is based on 4 components: planning, reporting, the operational part and the troubleshooting reports.

The high developed management tools should take into consideration each of these components with well defined characteristics and solutions for every situation to appear. Moreover, it should allow the integration with the other management systems from the organization and the given solutions should be based on friendly user interfaces. Furthermore, it is necessary to be taken into consideration the implementation of the IPAM notion (IP address management), too, in the context of developing the technology VoIP.

This area is intentionally forgotten many times, based on the fact that it is rather linked to the networks or data, although the development of the VoIP technology, by itself, is a current factor for updating IP addresses.

Nowadays, almost every company manually manages IP addresses. For a continuous and complex development, this primary form of management would be harder to achieve.

IPAM protocol is an important gate for complex notions of voice and data in a network, using the recent defined ENUM mechanism with the usability of DNS structures and the numbering plan areas. Consequently, the IT staff need to manage IPAM technology as a complex method of exploitation of resources that should allow the development of data security area, of capacity planning, validity of information and managerial growth.

Implementing the IP address management notion doesn't mean only finding an action plan, but finding specific tools that allow exploiting the DNS architectures, assuring DNS services for different mobile devices of the company and also assuring a correct configuration for DHCP technology.

A proactive and aggressive management IP address determines a better usability of the security components like firewall, security systems and traffic report analysis in a network.

It is presumed that using IPAM at a complex level, with the ENUM services, would determine a significant loading of the DNS servers and a loaded DNS and DHCP traffic.

Conclusions

The QoS mechanism called Integrated Services or IntServ was the first major attempt of enhancing the QoS capacity.

IntServ is an extension of the fundamental model for the internet service that traditionally provides the best effort type delivery of IP packages. Integrated Services takes into account the completation of the Internet network such as the new network, called IP-QoS network, will be able to provide the integration of the services in real time. The integration of these services is done by controlling the link sharing both for multicast and unicast functioning.

Initserv architecture is used as support for:

• Real time Internet services (audio, video), that will secure predictable or guaranteed services

• Controlled link portaging, which allows network services providers to allocate band on demand and to control the treatment that is applied to packages belonging to different classes of services.

In each DiffServ router, packages are guided according to PHB (Per Hop Behavior). DiffServ eliminates the necessity of flow processing and provides a simpler solution from an implementing and development point of view.

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