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Research paper

Behavior analysis of VoIP performances in next-generation networks

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Abstract

In the last decade, the telecommunications industry has evolved. Traditional networks cannot meet customers' incessant demand for new multimedia and interactive services. Next-generation networks (NGN) have been proposed to improve the quality of service (QoS) of the transported traffic as well as to make the network evolving, thus supporting future applications. Multi-Protocol Label Switching (MPLS) and IPv6 are used mainly on transport layer of NGN. These provide higher reliability, QoS, and security levels for the user's applications. Voice over IP (VoIP) is one of those applications that gaining momentum in our daily lives. Individuals or organisations tend toward this type of services given their various benefits. Several factors can degrade the quality of VoIP, such as the number of established sessions, the Call Signalling Protocol (SIP or H.323), the used transport protocols (IPv4 or IPv6) and especially the VoIP codecs chosen (GSM, G71, and G729).
According to our research, several works have been carried out evaluating the performance of VoIP on traditional networks. However, in this paper, we will assess the performance of the VoIP application in NGN architecture with MPLS as the transport protocol. The evaluation takes into consideration: (i) the impact of transport protocol on call establishment duration, (ii) the effect of user number and used codec on VoIP quality. The evaluation parameters used are the duration of the establishment of the call by the SIP protocol, the MOS score, latency, loss rate, and CPU. The results obtained showed that the call signalling duration depends mainly on the transport protocol in addition to the network architecture. The number of clients influences the quality of VoIP. However, the used codec may allow a higher number of VoIP sessions to be established.

Keyword: *VoIP, next-generation network, IPv6, codec, SIP..*

1. Introduction

1.1. Next-Generation Network

Telecommunications industry seeks to focus its technology for helping of Internet Service Providers (ISP) to stay competitive in an environment of increased competition and deregulation.
Next Generation Networks (NGN), with their distributed architecture, are fully exploiting advanced technologies to deliver sophisticated new services and increase ISP revenues while reducing capital expenditures and costs operating. NGNs are defined as a packet-based transport network that converges new services (video, voice, and data) into wired and wireless networks. These networks offer the same services from different access networks.
The topology of the NGN network is structured around six layers (Figure 1):

- Terminal layer: it contains all the terminals allowing the user to make and receive calls.
- Access layer: it connects users to the network and consolidates their traffic.
- Transport layer: it transports the traffic to the destination. The transport layer uses Internet Protocol (IP), MPLS (Multiprotocol Label Switching), or Metro Ethernet.
- Adaptation layer: it regulates the traffic for its transport on the network. This layer contains gateways (MGW - Media Gateways) for interworking between the access layer and the transport layer.
- Control layer: it provides call intelligence. This layer decides which service a user will receive. It also indicates the treatment to be applied to traffic.
- Application Layer: provides value-added services through application servers.

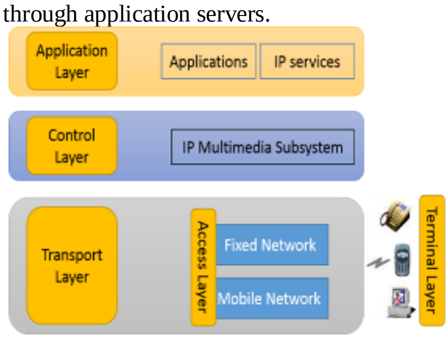

Fig1. The architecture of Next-Generation Network

1.2. Voice over IP

After the success of the Internet network and the emergence of new multimedia services such as VoIP, video conferencing or streaming video, researchers have faced a significant challenge, and it is to measure and ensure the quality of service of these applications [1][2][3][4][5]. According to the report Number 6506 published in 2016 by the National Agency for the Regulation of Telecommunications (ANRT) [6] of the Kingdom of Morocco, the penetration rate to data networks of the Moroccan population is 42.75% and almost 14.5 million subscribers by the end of 2015. With the advent of digital and the convergence of telecommunications and Internet, the kingdom of Morocco is actively engaged in a cycle full of challenges, and it is about the digital transformation of the society and the economy. As a result, several companies with several branches or partners located at different sites deploy communication solutions between them through Internet data network.
VoIP is a technology that allows the routing of digitised voice packets through the data network. Subsequently, these packets must be routed in the correct order and within a reasonable time for the voice to be reproduced appropriately.
VoIP technology offers several benefits for both customer and service provider, among them we note:

- Cost reduction: companies can reduce their cost of communication especially in the context of international communication.
- Standardization and interoperability between providers: the used network architecture is unique as the telephone network is integrated into the data network to form a single communications network.
- Mobility: the phone number can be kept no matter the geographic position of the customer.

VoIP technology has also several drawbacks:

- Architecture: since VoIP is based on an internet connection, it will be affected by the quality and reliability of the broadband Internet service and sometimes by the limitations of the terminal.
- Quality and reliability: data streams (voice) use an existing network that already has problems that can affect the quality of the Telephone Service. Indeed, the latency issues, delays, loss of packages can significantly reduce the quality and reliability of the service.

The rest of the paper is organised as follows: Section two deals with a brief description of VoIP constraints. In the third section, we will present the most recent related works. The experimentation model and used simulator will be described in the fourth section. Methods of the simulation are shown in the fifth section. And we conclude in the last part.

2. VoIP operation and signalling

Quality of Service (QoS) is a management concept for optimising network resources and ensuring the best performances for critical business applications. QoS allows offering users speeds and response delays differentiated by services according to the implemented Service Level Agreement (SLA).
It is interesting to note that the quality of service is only discussed if there is a degradation in the performance of a network. Among these constraints, we observe flow rate, latency, jitter, packet loss and MOS score.
The call setup is the first step to establish a VoIP communication. Call setup is performed by several signalling protocols, these protocols are responsible for defining the data formats, the dialogue, and control methods of the infrastructure and terminals, as well as the identification of the interlocutors. Several signalling protocols can be used to set up a VoIP call such as SIP, H.323, and MGCP. SIP protocol is the most suitable and used signalling protocol on NGN [7]

2.1. SIP protocol

SIP is a signalling protocol standardised by the IETF in RFC3261 applied for the establishment, modification, and termination of multimedia sessions on the internet such as multimedia conferences and VoIP. SIP architecture includes two types of user agents: User Agents Client (UAC) and User Agent Server (UAS), the first initiates the session and the second response.
The SIP architecture relies mainly on three servers:

- Recording servers: receive updates about the current location of users.
- Proxy servers: a proxy server receives requests from UAS and redirects them to another proxy server, UAS, or redirect server.
- Redirection servers: when a redirection server receives a request, instead of redirecting it, it sends the address of the next server to the client who will have to contact him directly.

Figure 2 illustrates the sequences of a session establishment by the SIP protocol.

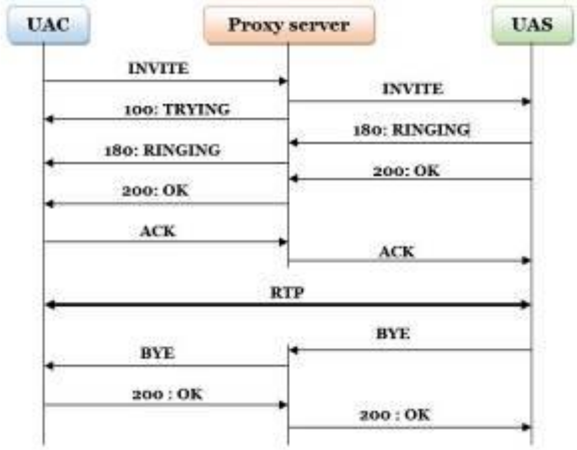

Figure 2: Call establishment using the SIP protocol.

2.2. VoIP Codec

Codecs serve to convert an analog voice signal into a digital signal.
G.711: The G.711 codec has a high bit rate (64 Kbps). It is the native codec used for voice transport over the Public Switched Telephone Network (PSTN) or ISDN. This codec is available under two versions: A-law and U-law. U-law is inherited from the T1 standard used commonly in Japan and North America. A-law is inherited from the E1 rule used in the rest of the world. The difference lies in the sampling of analog signals. In both cases, the signal is sampled in a logarithmic manner.
G.729: G.729 is a standard codec defined by the ITU. The main argument of this codec is that it offers a good ratio between the quality of voice and the used bandwidth. Indeed, the G729 needs a relatively low bit rate 8Kbps. However, it is a somewhat "expensive" codec regarding CPU processing time, so some VoIP phones and adapters cannot handle more than one G.729 call at a time. It can cause call failures if the user tries to use the three-way conference, or make multiple simultaneous requests.
GSM: Global System for Mobile Communications (GSM) (historically "Mobile Special Group" 1) is a second-generation digital standard for mobile telephony. It has been specified and developed by the ETSI (European Telecommunications Standard Institute) for the 900 MHz frequency range. A variant called the Digital Communication System (DCS) uses the 1800 MHz range. This standard is mainly used in Europe, Africa, the Middle East and Asia. Two other variants, 850 MHz and 1900 MHz PCS (personal communications services), are also used. Data protection is ensured by the encryption algorithms A5 / 1 and A5 / 2.

Table 1: Comparison between various codecs

Codecs	Algorithm	Throughput (Kb/s)	MOS
G.711	PCM	64	4.10
G.729e	CS-ACELP-LPC	11.8	4.00
G.729d	CS-ACELP	6.4	3.80
G.729a	CS-ACELP	8	3.70
GSM-FR	RPE-LTP	13	3.60
GSM-HR	VSELP	5.6	3.50
GSM-EFR	ASELP	12.2	4.10

Table 1 illustrated a short comparison between cited codecs regarding throughput, the MOS score and used the algorithm.

3. Related Works

The VoIP performance evaluation is a vibrant research field and exceptionally up-to-date. The published work deals in general with: (i) the impact of the network architecture on the performance of VoIP. (ii) the influence of codecs on the quality of VoIP. (iii) the impact of scalability on the transmission quality of VoIP.
The work [8] carried out a study on the impact of MPLS on VoIP performance. The authors compared the effect of the following networks on VoIP: MPLS, MPLS VPN, IP, and encrypted MPLS VPN. The evaluation criteria studied are jitter, latency, the MOS score and loss rate. As results, the authors showed that the MPLS technology offers the most optimal results, followed by MPLS VPN with a small difference compared to the MPLS. As for the IPsec protocol, the latter significantly degrades the quality of the VoIP. Similar work has resulted in the same result [9] [10] [11].
The study conducted at work [12] was done in a Wi-Fi wireless network and WiMax in both vertical and horizontal handover cases. The results obtained showed that the number of standard equipment radically influences the quality of VoIP, so WiMax technology is the most suitable for ensuring quality transmission. Similar work has led to the same result [13] [14].
Concerning Codecs and their impact on end-to-end VoIP quality, work [15] has been conducted. This work evaluates the effect of the different G.711, G.729, and G.723 codecs according to the end-to-end delay, the MOS score, the jitter and the amount of traffic received. The authors showed that the G.711 codec has the highest MOS score, whereas G.729 offers the lowest jitter compared to other codecs. The previous study was done in a wireless network consisting of Wi-Fi technology. The work [16] performed in a WiMax network found the same result. Other similar results have been presented in work [17] [18] [19].
Raising load or number, in other words, scalability, is used to evaluate the behaviour of VoIP technology in critical states where several calls are generated. Few works have been one about this problem. The work [20] was performed in a Mobile Ad-hoc Network (MANET) evaluating the efficiency of Ad-hoc routing protocols by increasing the number of VOIP clients. The work [21] is done by increasing the number of VoIP calls generated in the same scenario. The number of clients or calls influence the quality of transmission. However, the authors have not studied the most important constraints of VoIP which are latency; loss rate and throughput.
All the previous work has been carried out in traditional IP networks. In this paper, we will evaluate the performance of the VoIP application in NGN architecture with MPLS as the transport protocol. The evaluation takes into consideration: (i) the impact of transport protocol on call establishment duration (MPLS, MPLS VPN, and IPv6). (ii) the effect of user number and used codec on VoIP quality.

4. Experimentation

a) Model of experimentation

Before proceeding with the simulation, we must first set up an experimental model (modelling) in which a set of criteria must be well defined. These criteria are [22]:

- Inputs: they represent all the elements to be evaluated, they can be applications (video conference, VoIP, etc.), technologies or architectures (Number of nodes).
- Outputs: represent the objectives of the simulation, namely the degree of scalability of the inputs, the impact of the QoS on the data, the impact of the mobility protocols on the contributions.
- Controllable factors: represent the technical choices of simulation (use of different codecs).
- Uncontrollable factors: these are the factors that vary independently of the technical choice of the simulation (frequency of the generation of the flows, the rate of the streams and the trajectory of the nodes).

Figure 3 represents the model followed in all of our experiments:

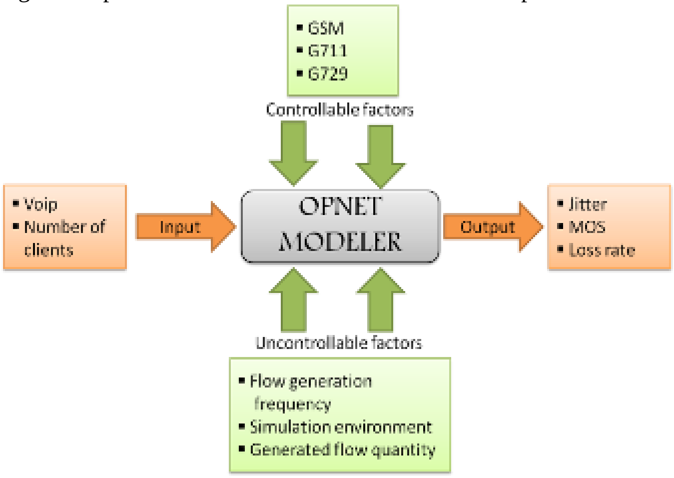

Figure 3: Model of experimentation

b) Choice of simulator

Simulation of network communications is an essential task in the process of design, planning, and optimisation of architectures. Thanks to the tools of simulation, we can study some problems like scalability and high availability, which are difficult to evaluate in a real environment because of their very high costs.
Several simulators of scientific research can be used. We choose the OPNET Modeler tool [23]. The latter is an extensive network simulation program supporting a vast number of wireless network features. It makes it possible to study the performances of the systems under variable conditions. It also contributes to the modelling of new protocols and the analysis of the achievements of emerging technologies.
OPNET is an efficient and straightforward simulation environment that offers an easy-to-use Human Machine Interface. The simulation under OPNET Modeler undergoes crucial steps, through which the user specifies the parameters of the network on which the tests will take place, then configure the applications to evaluate and of course determine the statistics to obtain.
Among its features, we quote:

- OPNET is a highly scalable tool for simulation engines: for wired and wireless network models, it allows to use simulation runtimes that use acceleration techniques;
- Hierarchical network models support complex topologies with an unlimited number of nested subnets;

- Full support for the realisation of protocols. More than 1000 functions are included, and the libraries provide the support for the understanding of the protocols.
- Point-to-Point and Multipoint Wireless Networks: the behaviour of network links is open and programmable. The characteristics of delays, availability, bit errors, and link rates are modifiable. This includes the physical level and changes in features depending on the environment. The Longley-Rice library is included as standard in Wireless Modeler. The same goes for TIREM and Free Space libraries.

5. Methods

In this section, we present and describe the network on which our measurements were made.

a) Network Testbed

The example network consists of three OSPF areas: one backbone area and two standard areas. The border networks represent customer sites; each LAN contains multiple users that we vary from one scenario to another. The proxy servers are located in the backbone of the operator: (i) Proxy- Call Session Control Function (P-CSCF), (ii) Interrogating-Call Session Control Function (I-CSCF), and (iii) Serving- Call Session Control Function (S-CSCF).
Figure 4 illustrates network testbed performed under OPNET Modeler simulator.

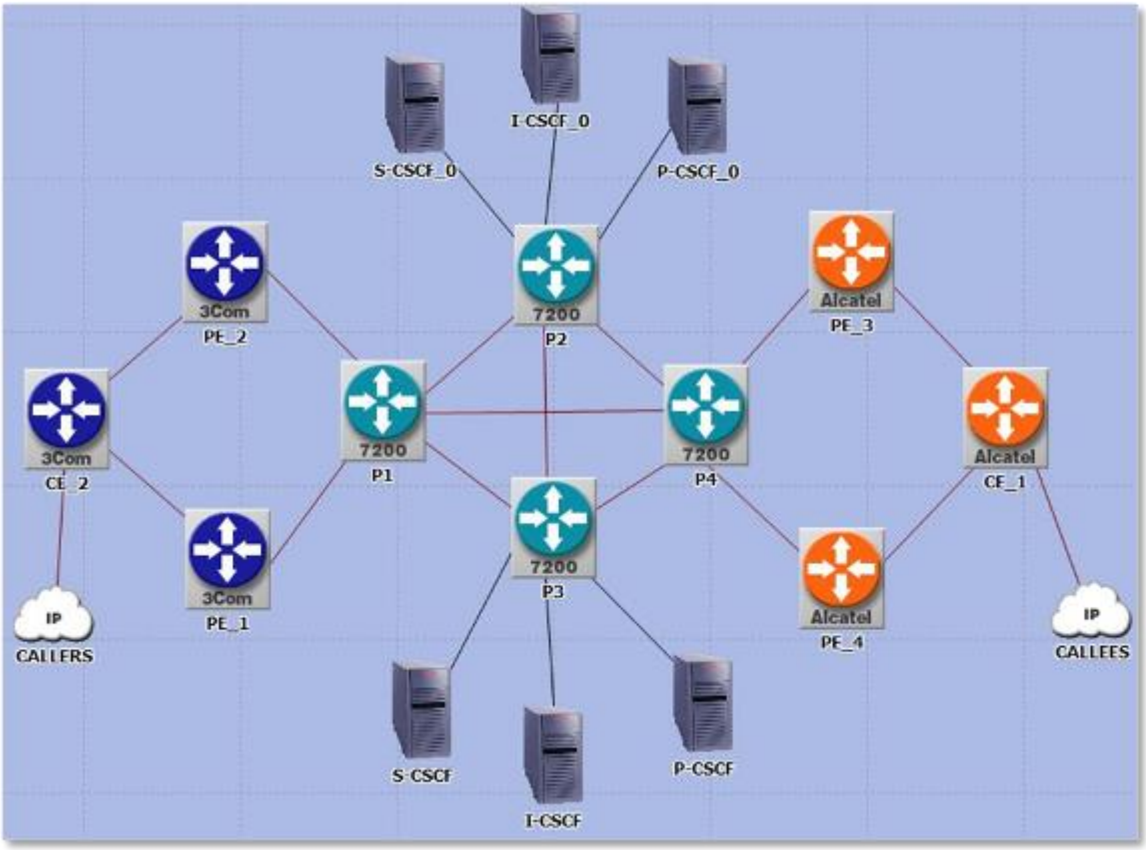

Figure 4: Network Testbed

b) Evaluation parameters

Links used on network testbed are summarised in table 2.

Table 2. Used link on simulation

Link	Definition
100BT	The 100BaseT duplex link represents an Ethernet connection operating at 100Mbps. It can connect any combination of the following nodes (except hub-to-hub, which cannot be connected)
DS1	1.544Mbps between CE and PE
DS3	44.736Mbps on Provider routers

5. Results and discussions

In this section, we will present the obtained results.

a) SIP session establishment duration

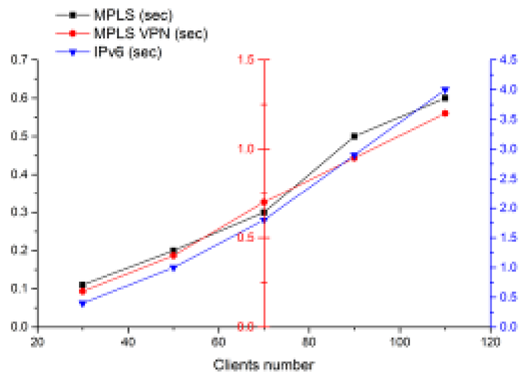

Figure 5: SIP session establishment duration.

Figure 5 illustrates the duration of the establishment of SIP sessions by different technologies of the transport layer: MPLS;

MPLS VPN, and IPv6 only. The results obtained showed the effectiveness of MPLS technology compared to other techniques. Establishing a SIP session with MPLS requires 110 milliseconds, while MPLS VPN and IPv6 require 200 and 400 milliseconds respectively. This means an increase factor of 81.81% compared to MPLS VPN and 263% compared to IPv6. This order of preference is noticed in all scenarios, even for the situation of 110 clients per site by the MPLS technology. The difference saw between MPLS and MPLS VPN is justified by the fact that the latter requires additional routing in the tunnel. Moreover, the encapsulation process is carried out by two labels. The rest of the results will be carried out using MPLS as a protocol of transport layer.

b) MOS Score

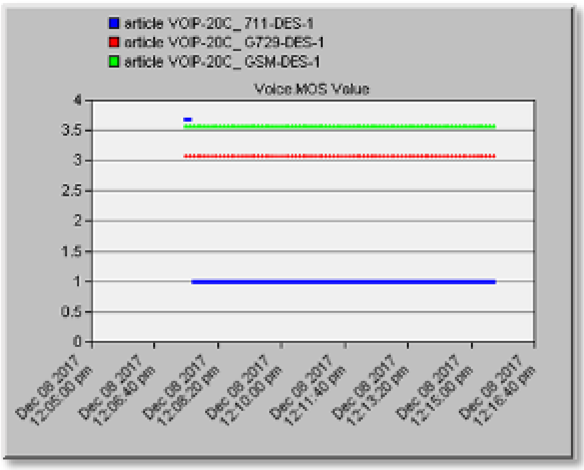

Figure 6: Effect of the different codecs on the MOS score for 20 clients.

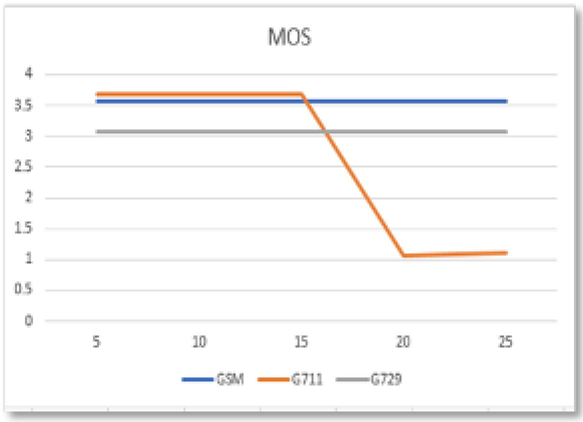

Figure 7: Comparison between codecs by varying the number of clients.

Figure 6 and 7 illustrate the results obtained from the MOS score of the G711, G729, and GSM scenarios by varying the number of clients.

We notice that the G711 codec offers a quality of telephony up to the scenario of 15 customers, beyond this scenario the quality of the VoIP is degraded remarkably thus condemning the score to be mediocre (MOS = 1).

It's clearly shown also that the quality of the VoIP remains excellent for all the scenarios configured with the G729 and GSM codecs, their qualities did not change even in the situation of 25 customers.

As an interpretation, we can say that the fact of asking a large size of the voice payload eventually saturates the band of the link between the CE1 router and the PE1.

As a summary, the GSM is the perfect codec that takes into consideration the high number of customers and fixed and limited bandwidth. However, its quality remains acceptable and above all understandable.

c) Loss rate

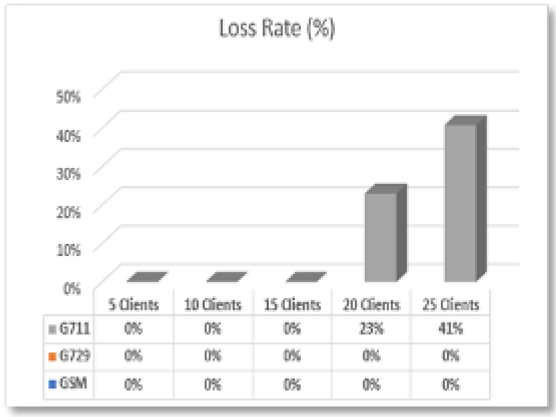

Figure 8: Effect of the loss rate on the G711 codec.

From Figure 8 we see that the loss rate remains zero for the G729 and GSM codecs up to the scenario of 25 customers. On the other hand, the loss rate increases significantly for the G.711 codec from the situation of 20 customers with a value of 23% and 41% for 25 customers. It is justified by the useful voice charge required by each of the codecs; G.711 involves a load of 160 bytes, and G.729 consists of a capacity of 20 bytes and GSM 32 bytes. Generally, if we increase the number of users, this will result in exceeding 1.544 Mbps (ds1). The first one that will undergo a saturation will be the G.711, and the other codecs can continue the extension. The question that arises is "Beyond which some clients will both codecs answer on quality? "

Figure 9: Comparison of the loss rate using GSM and G729.

From Figure 9 we note that the loss rate increases significantly for the G729 codec from the 40-client scenario and the GSM codec from the 60-client situation. As a conclusion, the GSM codec is the most adaptable for a more or less high number of customers.

d) Latency

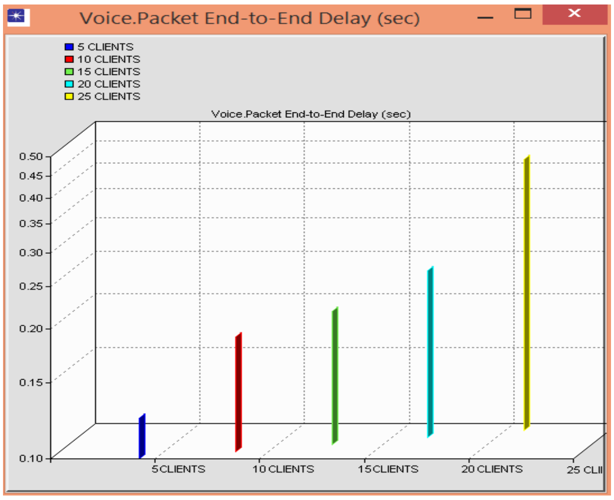

Figure 10: VoIP latency.

Figure 10 illustrates the VoIP latency with the G.729 codec. This delay represents the difference between the time of sending and receiving the same end-to-end packet. Referring to ITU Recommendation G411, a latency exceeding the 300 msec value results in a poor telephony quality. Obtained results show that until 20 clients the quality of VoIP remains acceptable. However, the scenario of 25 clients undergoes a latency of 420 milliseconds, which is higher than the tolerable threshold. For five clients, the lag is 120 milliseconds. This time increased by a factor of 50% almost compared to 10 clients and 100% compared to 20 clients.

e) CPU

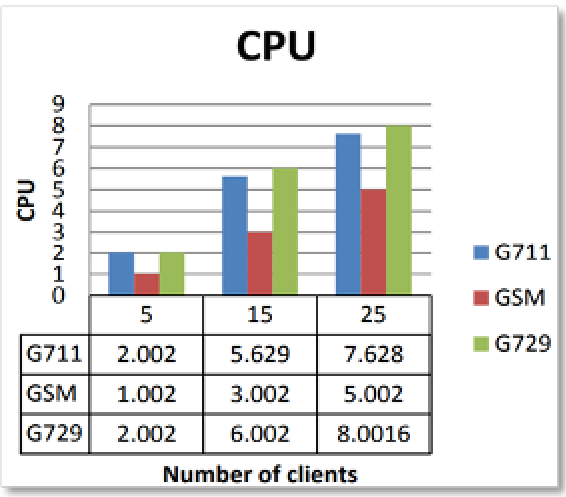

Figure 11: CE CPU performances.

The following figures show the CPU cycles of the CE1 router.
The results obtained from figure 11 that illustrate the CPU in all scenarios for the GSM, G711, and G729 codecs. We note that the G729 codec uses the CPU performance the most, it reaches a rate of 8.
The G711 codec is positioned in the second place and reaches a maximum rate of 7.7. G.729 codec offers a high call quality while consuming a low bit rate (8Kbps). It means that G.729 can provide more calls compared to G. 711 Codec. A consideration before implementing the G.729 codec is that, while the bandwidth used to transmit the call data is small, it needs much more CPU processing delay to generate a call.

6. Conclusion

NGN is an ideal solution for separating the transport and application layers to allow better scalability of the network. NGN aims to have a single system for all services. It brings many advantages to subscribers and operators, such as flexibility, cost reduction, and quality of service for all types of applications. VoIP is one of the most used applications until today. In this paper, we evaluated the performances of VoIP in Next-Generation Networks taking into account various codecs and transport protocols while increasing the number of users. The codecs discussed in this paper are G.711, G.729, and GSM. The evaluation criteria chosen are SIP session establishment duration, the MOS score, loss rate and the CPU cycle. From obtained results, we concluded that the most suitable protocol for the transport layer is the MPLS protocol followed by MPLS VPN and IPv6. The results obtained showed the effectiveness of the G.711 protocol in scenarios where the number of clients is limited. However, GSM has demonstrated its ability to respond to problems related to scalability. The G.729 codec on its part remains an excellent choice for quality telephony.

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