Performance Analysis of Hybrid Codecs G.711 and G.729 over Signaling Protocols H.323 and SIP

Salma RATTAL
Faculty of sciences and techniques
Hassan 2nd university
Mohammedia-Casablanca
Box 146, 20650 Mohammedia
Morocco

Abdelmajid BADRI
Faculty of sciences and techniques
Hassan 2nd university
Mohammedia-Casablanca
Box 146, 20650 Mohammedia
Morocco

Mohammed MOUGHIT
National school of applied sciences
Hassan 1st university
Béni Amir Street, Box 77, Khouribga,
Morocco

ABSTRACT
Due to the rapid growth of the internet, customers request more applications over internet such as telephony and video. VoIP (Voice over Internet Protocol) is an emerging technology for voice communication which deploys many techniques to produce a high-quality service.
Signaling is one of success keys of Voice over IP; the two most promised approaches in this case are H.323 and SIP. This paper draws a comparison between the four combinations of H.323 and SIP (Session Initiation Protocol), G .711 and G.729 (each combination includes one protocol and one hybrid codec) using a powerful network simulation tool (OPNET Modeler) which enables network simulation by employing various protocols.
The combination H.323 and G.729 proved to be the most appropriate one in the connection process.

Keywords
VoIP, H.323, SIP, G.711, G.729, OPNET, QoS.

1. INTRODUCTION

VoIP requires a number of protocols in order to control connection establishment. In this case, two protocols are available; ITU-T H.323 and IETF SIP.
Several studies were conducted to differentiate between the two protocols which represent very different approaches to the same problem: H.323 embraces the more traditional circuit-switched approach to signaling, and SIP favors the more lightweight Internet approach based on HTTP; a deep study was conducted in order to explain their complexity, extensibility and scalability [1]. Other studies were carried out to analyze the performance of VoIP over digital communication; parameters of interest are the quality of service, the Mean Opinion Score, packet loss ratio and jitter. Also, the delays and distortion issues that VoIP might have, while increasing the traffic, load and generate a more realistic topology by adding extra models to the system and evaluate the impact to the overall Quality of service (QoS) [2].
This paper compares the two protocols over the most common hybrid codecs G.711 and G.729.

This paper tackles the following points. The first section provides definitions of VoIP signaling protocols and speech codecs, which are explained and compared separately. The second section demonstrates how to configure and simulate H.323 and SIP networks into OPNET and to study the comparative curves of QoS. The results are discussed in details in the third section. Finally, the conclusion determines which of the four combinations provides a high-quality service.

2. VoIP: SIGNALING & CODECS

2.1 H.323 architecture:

The H.323 recommendation covers the technical requirements for voice communication services over packet-switched networks. This standard defines four major components for a network-based communication system: Terminals, Gateways, Gatekeepers, and Multipoint Control Units.

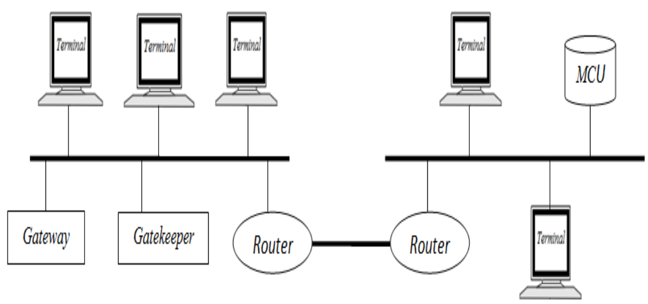


Fig 1: H.323 zone

Terminals are the client end-points that provide real-time one-way or two-way communications. All terminals must support voice communications. H.323 specifies the modes of operations required for different audio, video, and/or data terminals to work together.

Gateway is optional in an H.323 conference. Gateways provide many services, including a translation function between H.323 conferencing endpoints and other terminals. The Gateway also translates between audio and video codecs and performs call setup.

Gatekeepers perform a number of important functions that help preserve the integrity of the corporate data network. The first one is address translation from H 323 aliases for terminals and gateways to network addresses, The second function is bandwidth management, The fourth function is to manage a number of terminals, gateways, and MCUs as a single logical group known as the H.323 zone.

Multipoint Control Unit (MCU) supports conferences between three or more endpoints. [3]

2.2 SIP architecture:

SIP is an application-layer control protocol that can establish, modify, and terminate multimedia sessions (conferences) such as Internet telephony calls. SIP can also invite participants to already-existing sessions, such as multicast conferences. Media can be added to (and removed from) an existing session. SIP transparently supports name mapping and redirection services, which supports personal mobility

The main entities in SIP are:

Proxy Server is an intermediary entity that acts as both a server and a client for the purpose of making requests on behalf of other clients.

User Agent (UA) is the endpoint entity. User Agents initiate and terminate sessions by exchanging requests and responses.

Registrar is a server that updates a location database with the contact information of the user.

Redirect Server is a server that maps the SIP address of the called party into zero or new addresses and returns them to the client. [4]

The message type of SIP is Request-Response message as illustrated on figure 2:

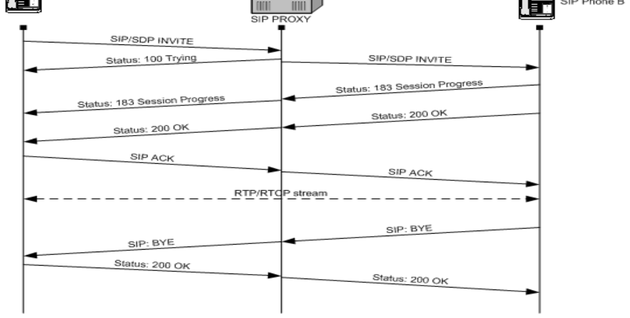


Fig 2: the message type of SIP

2.3 Signal transmission:

Two transport protocols are used to carry voice signals digitally encoded, so that all voice signals are converted from analogical into a digital form by transmitter device. In the first place, analogical signal is sampled based on a sampling rate of 8 KHz, then quantized: each sample is represented by 8 or 16 bits, after that, the output is encoded. Finally, the frames are encapsulated into an RTP/UDP/IP packet to be transmitted over IP network. These processes are established by one of various audio codecs, each one use his own different algorithm. [5][6].

2.4 Overview of speech codecs:

There are various codecs specified by the ITU-T. Codecs have different performance and impact on the voice quality due to different degrees of compression. High degree of compression results in higher compression delay and increases loss sensitivity compared to codecs with low or no compression. Contrary to this, codecs with high degree of compression have less bandwidth requirements, and thus have better performance in network congestion situations.

Therefore, it is necessary to select the appropriate codec to obtain best quality of voice with the lowest bandwidth requirements [7].

The G.711 codec does not use any compression; it has 8 kHz sampling rate, requires 64 Kbit/s of audio bandwidth and provides very good quality level. The G.729 codec is computationally complex, but provides significant bandwidth savings. It has 8:1 compression and requires just 8 Kbit/s of audio bandwidth. Main characteristics of the codecs mentioned are shown in Table 1 [8]

Table 1. Comparative table of G.711 & G.729

Codec	Bit Rate (Kbit/s)	Link Utilization (Kbit/s)	Delay (ms)	Loss (%)
G.711	64	87.2	0.125	7-10
G.729	8.0	31.2	15	<2

3. SIMULATION WITH OPNET

OPNET is a comprehensive network simulation tool with a multitude of powerful functions. It enables simulation of heterogeneous networks by employing a various protocols [9]. This paper constructs four scenarios; each one is used for studying results obtained by combining one signaling protocol (H.323 or SIP) with one of common hybrid codecs (G.711 or G729). In this case, there are four combinations (H.323/G.711, H.323/G.729, SIP/G.711 and SIP/G.729).

However, there are two types of topologies; the first one is an H.323 zone (gatekeeper, gateway and terminal) and the second is a SIP topology (proxy server and user agents).

Figure 3 demonstrates the simulation model for the first network, G.711 over H.323 protocol.

In this model, there are a number of terminals in order to generate the maximum number of simultaneous calls. A gatekeeper is implemented in each side of the network. Application configuration is a set of rules which has varieties of libraries to generate the traffic on the network according to the user requirement. In this simulation, different VoIP parameters are configured but the most important ones are the

Encoder Scheme set to G.711 and the Signaling Protocol set to H.323.

A subnet containing ten nodes in each side of the backbone is added in order to create Ftp and email traffics in the model, and show their influence on quality of voice because they will consume an important amount of bandwidth.

All network elements are connected using 100 Base-T links except routers; they are connected using PPP DS1 link of 1.5 Mbps.

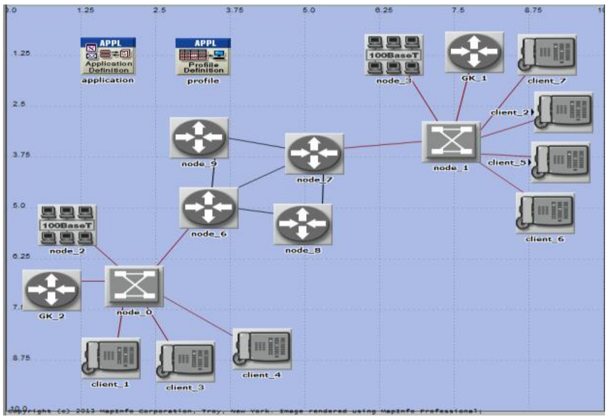
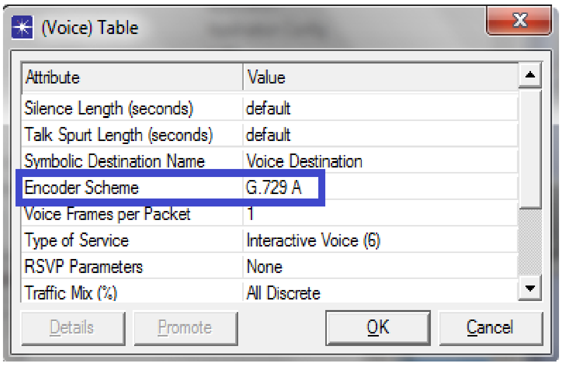


Fig 3: H.323 architecture simulated using OPNET

In the second case, the same H.323 network is established but the Encoder scheme is changed into G.729 as shown in figure 4.


(Voice) Table

Attribute	Value
Silence Length (seconds)	default
Talk Spurt Length (seconds)	default
Symbolic Destination Name	Voice Destination
Encoder Scheme	G.729 A
Voice Frames per Packet	1
Type of Service	Interactive Voice (6)
RSVP Parameters	None
Traffic Mix (%)	All Discrete

Details Promote OK Cancel

Fig 4: changing the used encoder

However, in SIP simulation a third scenario which contains a new SIP topology is created. For this model, in each side of the network, there are a number of SIP clients and SIP proxy server with a subnet containing 10 nodes to generate Ftp and email traffics as done in the first and second scenarios (figure 5).

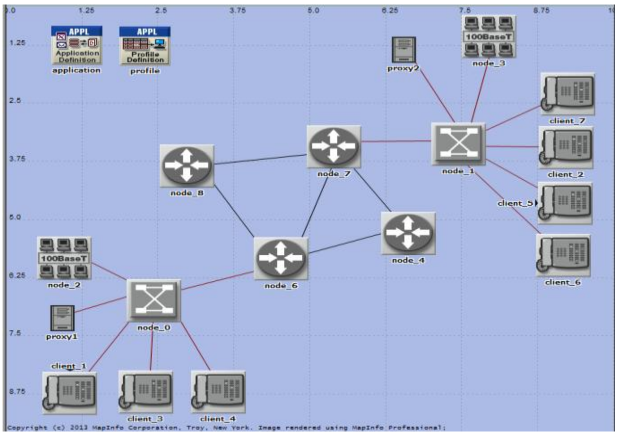


Fig 5: SIP architecture simulated using OPNET

Finally, a fourth model is established where the G.711 encoder is substituted by G.729 encoder.

4. DISCUSSION OF THE RESULTS

Analyzing the performance of every combination (H.323/G.711, H.323/G.729, SIP/G.711 and SIP/ G.729) requires calculation statistics of QOS parameters (MOS, Jitter, end-to-end delay and packet loss).

4.1 End to end delay:

End-to-end delay consists of end-system and network delay. The end-system delay occurs due to the encoding and decoding delay and de-jitter buffering delay.

Toll quality real-time communication is needed, which limits the maximal tolerable round-trip delay to 200-300 ms; that is, one-way delay must be in the range of 100-150 ms for adequate performance. [10]

figure 6 shows when both scenarios operate under the same SIP signaling protocol with two different coders G.711 and G.729, end to end delay is very high compared to the two other scenarios involving H.323 with G.711 or G.729 which take considerably less time.

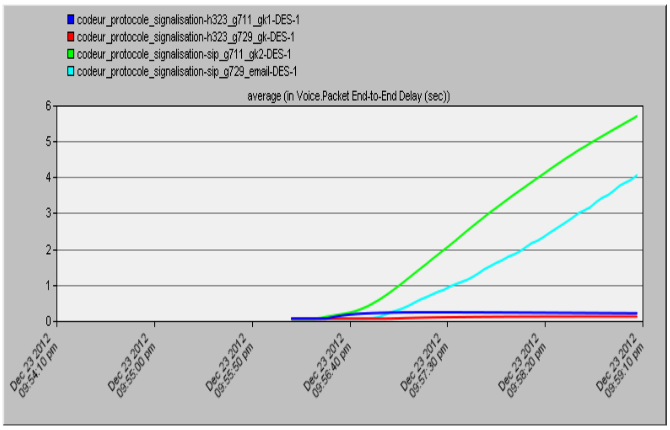


Fig 6: end-to-end delay

According to figure 6, SIP scenarios take a very long time to send packets from nodes to gateway. To resolve this problem, PPP DS3 links of 44.7 Mbps must be used to connect routers into SIP network simulations. This will clearly reduce RTP delay as shown in figure 7.

However, DS1 links of 1.5 Mbps are reasonable to obtain a transmission delay around 150 ms for H.323 zone.

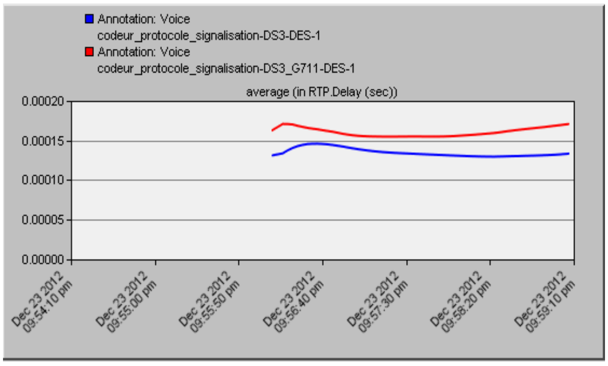


Fig 7: end to end delay with DS3 links

4.2 Jitter

The end-to-end delay variation between two consecutive packets is called jitter. A jitter of less than 50ms is acceptable for high quality VoIP call [11].

If the delay of transmissions varies too widely in a VoIP call, the call quality is greatly degraded. [12]

According to figure 8, jitter is obviously different depending on the scenarios.

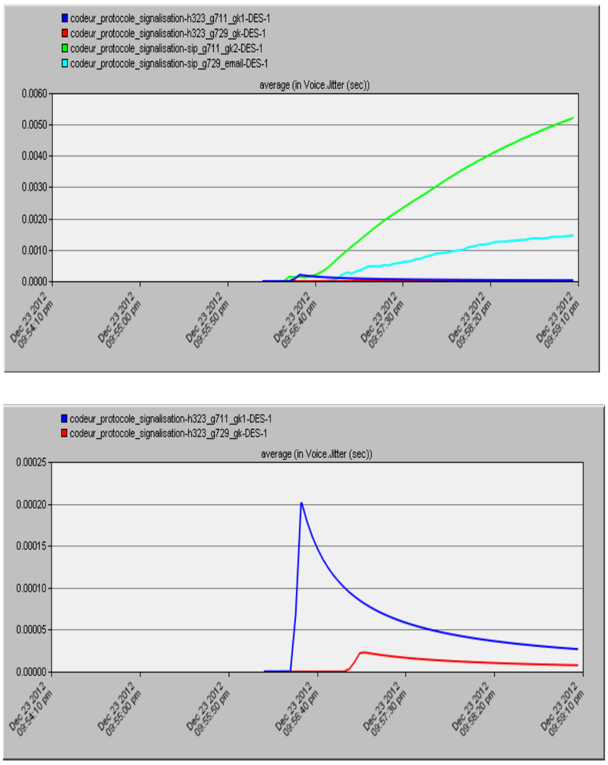


Fig 8: Voice jitter

4.3 Traffics sent/received:

As mentioned above, each encoder is characterized by its own bit rate: (G.711: 64 Kbit/s; G.729: 8 Kbit/s).

Sent traffic Figure demonstrates these values. However, the received traffic figure shows that, in parallel, a relatively large packet loss is present during communication.

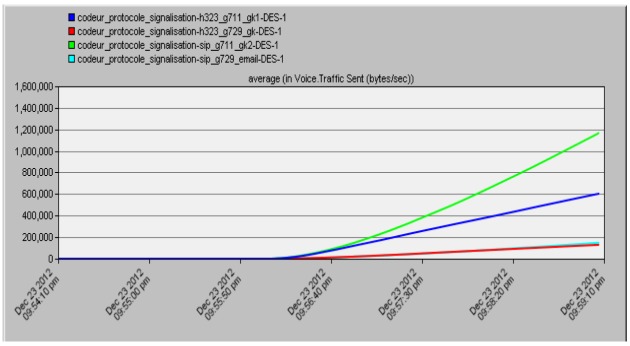


Fig 9: Sent traffic

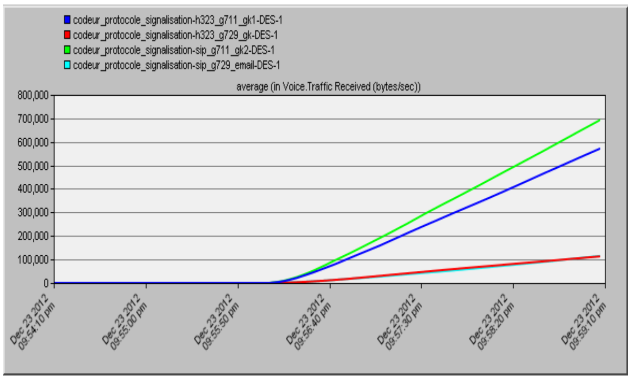


Fig 10: received traffic

4.4 MOS (Mean Opinion Score):

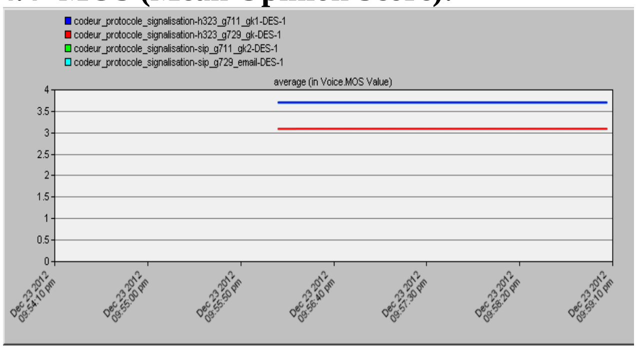


Fig 11: MOS calculation

The calculated MOS is a standard ETSI and ITU-T. a mathematical calculation of this value could be realized by considering the characteristics of the communication.

This recommendation is based on the fact that the damage is added together on a scale of predetermined quality. If a signal through multiple devices, damage equipments add up.

As shown in figure 11, the MOS value is equal to 3.10 for both combinations H.323/G.729 and SIP/G.711. This means that the voice quality is low in contrast to the MOS value of the other two combinations H.323/G.711 and H.323/G.729 which is equal to 3.7, indicating a better quality according to the following table [13]:

Table 2. Relationship of MOS values to the Quality of Voice Rating

Quality of Voice Rating	MOS
Best	4.34 - 4.50
High	4.03 – 4.34
Medium	3.60 – 4.03
Low	3.10 – 3.60
Poor	4.58 – 3.00

This study is conducted to find out the different characteristics of the scenarios established and to select the best scenario for VoIP communication, Figure 12 draws a comparison, in the third minute of communication, between the four scenarios against the following characteristics: jitter, packet loss, MOS and end to end delay.

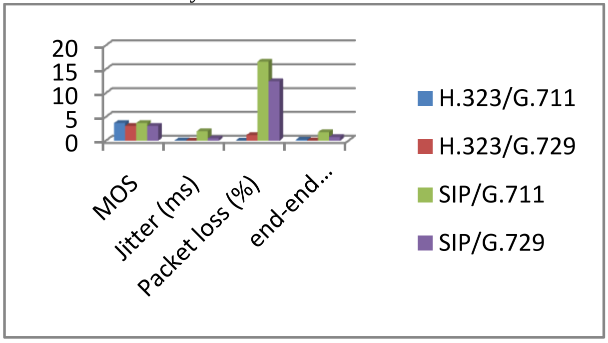


Fig 12: comparison of scenarios performances

In other words, Table 3 illustrates this comparison clearly. It demonstrates which characteristics are positive in the four scenarios. The table consequently helps in choosing the most successful scenario in VoIP communication.

Table 3. Determination of the most appropriate scenario.

	MOS	*jitter*	*Packet loss*	*End-end delay*
H.323/G.711	+		+	
H.323/G.729		+	+	+
SIP/G.711	+			
SIP/G.729				

As shown in the table 3, H.323/G.729 scenario obtains the maximum positive points to ensure a high-quality service.

5. CONCLUSION

This paper studies the performances of SIP and H.323 protocols with hybrid codecs G.711 and G.729.

After a deep study of these protocols and codecs, it was necessary to simulate every scenario using Opnet which makes it simple to calculate QoS performances (jitter, RTP delay, end to end delay, packet loss and MOS).

Accordingly, it was absolutely clear that the most appropriate combination of codec and signaling protocol is H.323 and G.729. This combination offers the minimum delays (RTP delay/end to end delay) with an acceptable jitter value for the best call quality.

In the future work, both signaling protocols H.323 and SIP will be integrated together in the same equipment it means that gatekeepers will operate in the network with an integrated proxy. This will be accomplished by reprogramming some entities in node and process models of Opnet.

6. ACKNOWLEDGEMENTS

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