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RESEARCH ARTICLE

Performability Evaluation of Voice Services in Converged Networks

Avaliação de Performabilidade de Serviços de Voz em Redes Convergentes

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Abstract: In the last years, the transmission of voice services in converged networks has experienced a huge growth. However, there are still some questions considering the ability of these networks to deliver voice services with acceptable quality. In this paper, we applied analytical modeling and simulation to analyze the quality of voice services using a new index, called MOS_a, which considers jointly the MOS index and the availability of the subjacent infrastructure. We consider the influence of different CODECs (G.711 and G.729), queuing policies (*Priority Queuing* and *Custom Queuing*), and the *warm standby* redundancy mechanism. Our goal is to analyze the quality of these services by taking into account overloading conditions in different architectures/scenarios. These scenarios were constructed using the modeling mechanisms Reliability Block Diagram and Stochastic Petri Nets in addition to a discrete event simulator. Experimental results indicate that the G.711 CODEC has a higher sensitivity both in terms of data traffic volume and allocated network resources in relation to the G.729 CODEC.
Keywords: VoIP — MOS — Availability — Reliability Block Diagram — Stochastic Petri Nets

Resumo: Nos últimos anos, a transmissão de serviços de voz em redes convergentes experimentou um enorme crescimento. Contudo, ainda existem algumas questões considerando a capacidade dessas redes de fornecer serviços de voz com qualidade aceitável. Neste artigo, aplicamos técnicas de modelagem analítica e de simulação para analisar a qualidade dos serviços de voz utilizando um novo índice, denominado MOS_a, que considera conjuntamente o índice MOS e a disponibilidade da infraestrutura subjacente. Consideramos a influência de diferentes CODECs (G.711 e G.729), políticas de enfileiramento (*Priority Queuing* e *Custom Queuing*) e do mecanismo de redundância *espera morna*. Nossa meta é analisar a qualidade destes serviços levando em consideração condições de sobrecarga em diferentes arquiteturas/cenários. Estes cenários foram construídos utilizando os mecanismos de modelagem diagrama de blocos de confiabilidade e redes de Petri estocástica além de simulação baseada em eventos discretos. Resultados experimentais indicam que o CODEC G.711 possui uma maior sensibilidade tanto em termos de volume do tráfego de dados quanto dos recursos de rede alocados em relação ao CODEC G.729.
Palavras-Chave: VoIP — MOS — Disponibilidade — Diagrama de Blocos de Confiabilidade — Redes de Petri Estocástica

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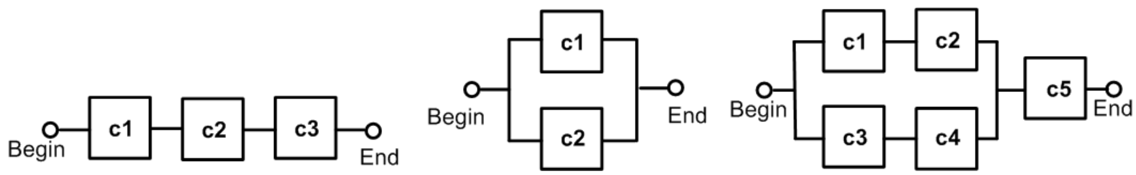
1. Introduction

Over the last few years, the use of voice transmission services in converged networks has significantly increased. This considerable growth is somewhat related to interoperability offered by voice and data transmission and its low cost. In many companies, *VoIP* (Voice over Internet Protocol) services are strategic and their degradation may lead to considerable losses. They should be continuously provided even when events like congestion, link malfunctioning, routing instabilities, sabotage, natural disasters, hardware or software failures happen [1]. Hence, either in normal operation or in case of network failure, *VoIP* services should be preserved.

An important factor that is used for the performance analysis of *VOIP* services is the *MOS* (Mean Opinion Score) [2] index. This index is directly impacted by performance factors and it can be calculated based on the voice packet loss rate [3, 4] of a voice connection. For a better perception of the voice services quality, it is important to consider both the performance aspects, reflected by the *MOS* index, as the



(a) Series (b) Parallel (c) Series-Parallel

Figure 1. Basic Structures

availability aspects of these services. Our work analyzes the behavior of voice services using a new index, called MOS_a, which considers jointly the MOS index and the availability [5] of the subjacent infrastructure. Several scenarios and architectures were constructed through different *CODECs* (Coder-Decoder), such as *G.711* [6] and *G.729* [7]; queuing policies, such as *Custom queuing* [8] and *Priority Queuing* [9]; and the *warm standby* redundancy mechanism [10]. Due to the system inexistence and costs of settings, analytical and simulation models are adopted as a strategy for a quantitative analysis.

The proposed scenarios and architectures were constructed using Graphical Network Simulator-3 (*GNS-3*) [11] and *Mercury* [12] tools. For measuring performance, simulation environments using *GNS-3* tool are utilized. *GNS-3* is a software emulator for complex networks, launched in 2008. In turn, for dependability evaluation we used analytical models built with the *Mercury* tool. *Mercury* has been conceived to allow dependability/sustainability evaluation of general systems. The corresponding dependability models were created using the Reliability Block Diagram (*RBD*) [10] and Stochastic Petri Nets (*SPN*) [13] modeling mechanisms following a hierarchical approach.

The rest of this paper is organized as follows. Section 2 presents the state of the art in this area. Section 3 describes the network infrastructure models. Section 4 explains the adopted methodology in order to construct and analyze different scenarios. Section 5 presents the evaluation of the performance and dependability scenarios, concerning different *CODECs*, queuing policies and the *warm standby* redundancy mechanism. Finally, Section 6 concludes the paper and introduces ideas for future research works.

2. Related Work

In the last few years, much has been done to deal with issues relating to performance or dependability of real time applications, such as *VoIP* services [14, 15, 16, 17, 18, 19, 20, 21, 22]. Initially, Reference [14] provides an insight into the impact of *TCP* segmentation in *VoIP* monitoring and the solution that has been applied to face it. This study focuses on voice signaling messages which include relevant data during calls negotiation. The importance of such messages is justified because if they are lost, monitoring tools may produce erroneous statistics and lose the corresponding multimedia transmissions.

In turn, the work presented in [15] simulates the *VoIP* over IEEE 802.11b and 802.11e wireless local area networks (*WLAN*) with different *CODECs* by using the *NS-2* tool (Network Simulator-2). The main objectives were to analyze the Mean Opinion Score (*MOS*), throughput, and packet drop rate of *VoIP* traffic. It was found that the *CODEC* G.711 provides the best quality of *VoIP* calls for the network scenarios analyzed.

In [16], a systematic approach for quantifying the reliability of enterprise *VoIP* networks is presented. It identifies two key challenges for designing enterprise *VoIP* service infrastructure: i) there are no universally accepted objectives or standards for the reliability of the enterprise *VoIP* services; ii) there is no well known common set of reliability metrics to be used in enterprise *VoIP* service planning. [16] presents an enhanced method and procedure for reliability calculation, using a network matrix representation and *RBD*.

The authors of [17] focuse on network impairments that can seriously degrade *VoIP* performance, based on different queuing mechanisms. Three different mechanisms, DropTail (*FIFO*), *RED* and *DiffServ*, and their effects are analyzed. Measurements of jitter, end to end delay and packet loss are studied. Experimentation has proven that under burst traffic conditions up to a congestion level, *DiffServ* seems to perform better in all three categories examined.

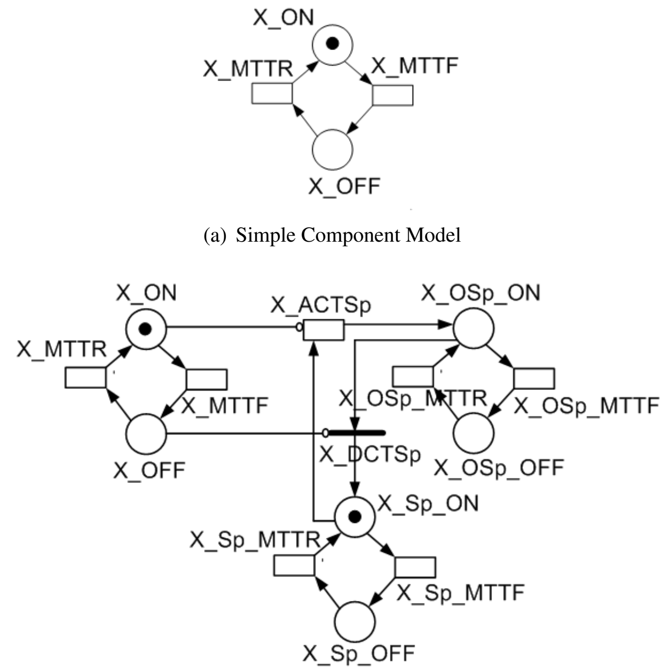
[18] evaluates, with measurements, the effect of the voice *CODECs*, G.711, G.729, and G.723.1, on the quality of *VoIP* calls over a *MANET* in a real indoor environment which includes a corridor within a university campus.

[19] analyzes and plans the capacity of a real enterprise-class voice gateway system, which supports the voice communication of six university campuses. It was proposed the use of computer modeling and simulation for planning the critical resources of the investigated gateway system. Different scenarios are simulated to assess the current and future capacity of the system under study.

In turn, [20] investigates and compares the performance of the G.711, G.723 and G.729 *CODECs* on *SIP* architectures. It analyzes parameters such as jitter, throughput and *MOS*. The *Opnet* tool was used in order to compare the different *CODECs*.

The work in [21] utilizes simulation to determine the optimal values of Transmission Oportunity (*TXOP*) and Frame Aggregation (*FA*) that maximizes *VoIP* capacity. *TXOP* and *FA* are two important Medium Access Control (*MAC*) Layer enhancements provided by IEEE 802.11n standard. The voice capacity is calculated when optimal values of *TXOP* and *FA* are simultaneously used. Additionally, the *VoIP* capacity over User Datagram Protocol (*UDP*) and *TCP* Friendly Rate Control (*TFRC*) protocol in the presence of Transmission Control Protocol (*TCP*) traffic is determined. Finally, Reference [22] analyses the Quality of Service (*QoS*) of *VoIP* in 802.11ac networks under varying networking conditions. It measured various *QoS* metrics such as throughput, jitter, latency, loss ratio and Mean Opinion Score (*MOS*) using an Extended Service Set (*ESS*) testbed with *IPv4* and *IPv6* as network layer protocols. The result shows a degradation in *VoIP* quality when background traffic increases. *IPv6 VoIP* calls had a better Quality compared to *IPv4* calls.

Researchers have used different approaches, however note that none of them deal jointly with issues of performance and dependability of *VoIP* services. This research extends our preliminary works in [23, 24, 25] merging performance and dependability issues and proposing the MOS_a index that correlates MOS index and the availability of the subjacent infrastructure in different scenarios and architectures. The proposed index reflects, in a more realistic way, the effects of availability on the voice services quality. In addition, hierarchical models, which consider the main components of the infrastructure, are proposed to calculate its availability.



(a) Simple Component Model

(b) Warm Standby Model

Figure 2. SPN Building Blocks

3. Network Infrastructure Models

This section presents the base models for quantifying the availability metric of converged network infrastructures. The proposed models are generic enough to represent a wide variety of mechanisms found in real converged networks infrastructures.

3.1 RBD Models

The most common structures on converged network topologies are serial, parallel and both [26]. Figure 1 depicts three examples where the blocks (C_1, C_2, C_3, C_4 and C_5 components) are arranged through series (Figure 1(a)), parallel (Figure 1(b)) or series-parallel (Figure 1(c)) compositions. A series structure on a set of components means that for the whole subsystem to work, every component has to be functioning. Assuming a structure with n components in series, the availability (reliability) [10] is obtained by:

$$P_{(s)}(t) = \Pi_{i=1}^{n} P_i(t) \tag{1}$$

where $P_i(t)$ is the availability or the reliability of block i. On the other hand, a parallel structure means that the whole subsystem can function if at least one (or more) of the components is working. Taking into account n components in a parallel structure, the system availability (reliability) is:

$$P_{(p)}(t) = 1 - \Pi_{i=1}^{n}(1 - P_i(t)) \tag{2}$$

In order to calculate the availability (reliability) of a series-parallel structure, the serial results must be combined and placed into parallel equations. For other examples and closed-form equations, the reader should refer to [10].

3.2 SPN Models

This section briefly presents the *SPN* building blocks [27] for obtaining dependability metrics.

3.2.1 Simple Component

Figure 2(a) shows the dependability model of a generic simple component. This component is characterized by the absence of redundancy. Places *X_ON* and *X_OFF* are the model component's activity and inactivity states.

The *MTTF* (Mean Time To Failure) [1] and *MTTR* (Mean Time To Repair) [2] parameters represent the delay associated with the transitions *X_MTTF* (in our study, label "*X*" must be applied according to the component name) and *X_MTTR* respectively.

3.2.2 Warm Standby Model

Figure 2(b) shows the dependability model of the *warm standby* [10] redundancy approach. Places *X_ON*, *X_OFF* represent the activity and inactivity states of the components. Place *X_Sp_ON* represents the spare (*Sp*) component of *X* in non-operational state. In turn, place *X_OSp_ON* represents the spare component of *X* in operational (*OSp*) state. Initially, the spare component is in the non-operational state. As the main component fails, the transition *X_ACTSp* is enabled. Its

[1] This parameter is supplied by the manufacturer.

[2] This parameter is closely related to the maintenance policy adopted by organizations, which depend on equipment and site criticality as well as on the associated delays needed to reach remote locations.

firing represents the start of operation of the spare component. This time period is named Mean Time To Activate ($MTTA_w$). Immediate transition *X_DCTSp* represents the return to initial condition.

The *MTTR* parameter represents the delay associated with the *X_MTTR*, *X_Sp_MTTR* and *X_OSp_MTTR* transitions. The *MTTF* parameter of each component represents the delay associated to the *X_MTTF*, *X_Sp_MTTF* and *X_OSp_MTTF* transitions. In this work, it is considered that the transition *X_Sp_MTTF* has a delay 50% higher than transition *X_OSp_MTTF*. The timed transitions are exponentially distributed and have a single server concurrency policy.

4. Evaluation Methodology

This section presents the proposed methodology that aims the analysis of performance and dependability issues of converged networks. It consists of seven macro activities: *Problems and Components Definition*, *Obtaining Information*, *Model Generation and Metrics Mapping*, *Validation of the Model*, *Calculating the MOS_a index*, *Evaluation of the Scenarios* and *Results Analysis* (see Figure 3).

The first activity determines the problem together with its scope. At the end of this activity, two artifacts are obtained, namely: Problem Diagram and the considered Metrics. Problem Diagram is a document that contains aspects of the infrastructure under analysis such as: its topology, interconnection and component dependencies, component definitions and hardware settings. The second artifact, the set of metrics, defines the indicators used to analyze the dependability and performance aspects of the system. The main metrics are the voice packet loss rate, that it is used to calculate the *MOS* index of a voice connection [3], and the steady state availability of the subjacent infrastructure.

Information about the components is obtained in the *Obtaining Information* activity. For dependability models, information is related to the following parameters: *MTTF* and *MTTR*. In turn, the parameters of the performance models components are configured in accordance with each defined scenario and refers to different queuing policies, *CODECs* etc.

In the *Models Generation and Metrics Mapping* activity, the artifacts generated are the models (performance and dependability) and their metrics. It is accomplished by modeling each system component and composing them through composition rules. This activity also takes into account the metrics mapping that represent the metrics through expressions depicted by elements of each model. Regarding dependability models, this work adopts a hybrid modeling strategy [28] that considers the advantages of both *RBD* and *SPN*. Such a hierarchical approach is adopted to mitigate the complexity for representing large systems that can generate the state space explosion problem [29] for some state based models.

The activity *Validation of the Model* analyzes and carries out the adjustments when necessary. It concerns the evaluation of metrics considering performance and dependability scenarios. The checkpoint means that the models are mature enough to be evaluated according to established criteria. The end of this activity is reached when one or more models are obtained to solve the problems described.

After *Validation of the Model* activity, the *Calculating the MOS_a index* activity can be conducted. From the obtained metrics, it is calculated the MOS_a index (see Equation 3) that relates the MOS index of a scenario *i*, obtained through the voice packet loss rate, with the steady state availability (A) of the subjacent infrastructure.

$$MOSa_i = MOS_i \times A \tag{3}$$

Then, the *Evaluation of the Scenarios* activity can be conducted. For evaluation purposes of the MOS_a index, the parameters used for the *MOS* index will be considered. [30] says that a *MOS* rating above 4.0 matches the level of quality available in the current Public Switched Telephone Network (*PSTN*), a rating above 4.3 corresponds to the best quality, where users are very satisfied, and a rating between 4.0 and 4.3 corresponds to a high quality level, where users are satisfied. A *MOS* rating between 3.6 and 4.0 corresponds to a medium quality level, in which some users are dissatisfied. A *MOS* rating in the range between 3.1 and 3.6 corresponds to a low level of quality, where many users are dissatisfied. At a *MOS* rating in the range between 2.6 and 3.1, the level of quality is poor, and nearly all users are dissatisfied. Finally, a *MOS* below 2.6 is not recommended. Regarding availability, [31] states that a system with an availability of 99.999% is classified as *high availability* system. On the other hand, an availability of 90% and 99.0% classifies the system as unmanaged and managed respectively.

Finally, in the *Results Analysis* activity, the results obtained are interpreted and explained by adopting an appropriate vocabulary to the customer background and objectives.

5. Case Study

In this section our goal is a analysis of the voice services considering the influence of different *CODECs*, queuing policies and redundancy mechanisms on the their quality through different architectures and scenarios that were built to obtain performance and dependability results. The evaluation scenarios are similar to those obtained in small and medium enterprise networks.

Our work is based on two architectures (see Figure 4) [27]. Figure 4(a) shows the simulation environment of the first architecture (architecture A_1). If a component (Router R_1, Link L_1 or Router R_2) fails, the system goes down. Figure 5(a) shows the respective dependability model that adopts a hierarchical approach. In the upper layer, each component is modeled as a block in a series structure. The lower level models represent the routers (Router R_1 or Router R_2) in a series structure.

In turn, Figure 4(b) shows the simulation environment of the Architecture A_2. This architecture is similar to A_1

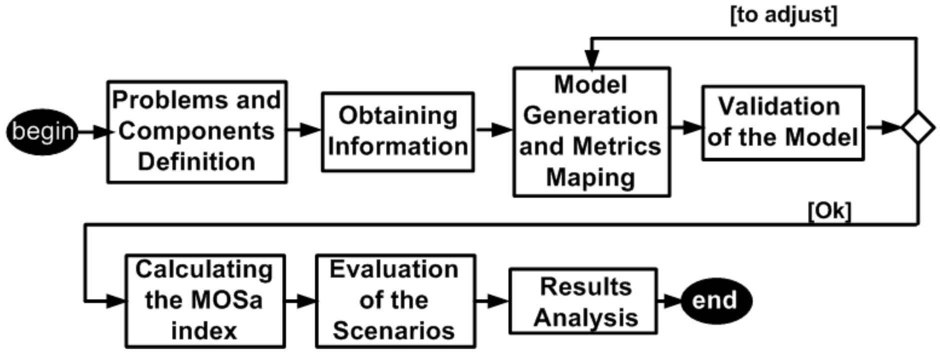


Figure 3. Proposed Methodology

and was obtained through the insertion of a redundant link (Link L_2) that is configured as a spare that adopts the *warm standby* redundancy mechanism, such that when the primary link (Link L_1) fails, the spare link assumes the role of the primary one. After recovery, the system returns to the initial state. If a router (Router R_1 or Router R_2) or both links fail, the system goes down. The dependability model adopts a hierarchical approach (see Figure 5(b)). The upper level model is a *RBD* and the lower level models are *RBD* and *SPN*. The *SPN* model represents the *warm standby* redundancy approach between the primary and spare links (see Figure 2(b)). Table 1 shows the parameters used in the dependability models of the architectures A_1 and A_2.

In the simulation environments, the *IP* phones are responsible for the voice connections and the machines generate the data traffic. The *IP* phones generate voice traffic with different *CODECs* (G.711 and G.729). Table 2 shows that G.711, titled Pulse Code Modulation (*PCM*), uses the most bandwidth, whereas it introduces the least amount of latency. These factors have made *PCM* inexpensive to deploy and highly effective in transporting voice traffic over long distance. By contrast, G.729, described as coding of speech at 8Kbit/s using code-excited linear prediction speech code (*CS-ACELP*), provides significant bandwidth savings at the expense of increased latency.

It will be used six, eight, ten or twelve machines in different scenarios in order to analyze the behavior of voice traffic with the increase of data traffic. Each machine, connected on router R_1, generates *ICMP* packets with *1432 bytes* at a rate of one packet per second.

The serial interfaces 1 (Architectures A_1 and A_2) and 3 (Architecture A_2) were chosen for the configuration of the queuing policies custom queuing (*CQ*) and priority queuing (*PQ*) due to the fact that they are the critical points where the voice and data traffic will dispute the access to the network resources. The analysis of each performance scenario will be done in these interfaces considering the resulting *MOS* index from the voice packet loss ratio in a time of one minute. This index is calculated considering different *CODECs* (G.711 and G.729), number of machines generating data traffic and network resources configured.

The different network resources were attributed through the queuing policies *CQ* and *PQ*. *CQ* allows a custom configuration of resources in an interface to different applications. It allows configuration of a specified number of *bytes* to be forwarded from a queue each time that queue has been serviced. It has sixteen possible queues. A maximum number of packets per queue can be specified [8]. In our work, *CQ* was configured with two queues, one queue for voice and one queue for data and attributed different network resources for each one.

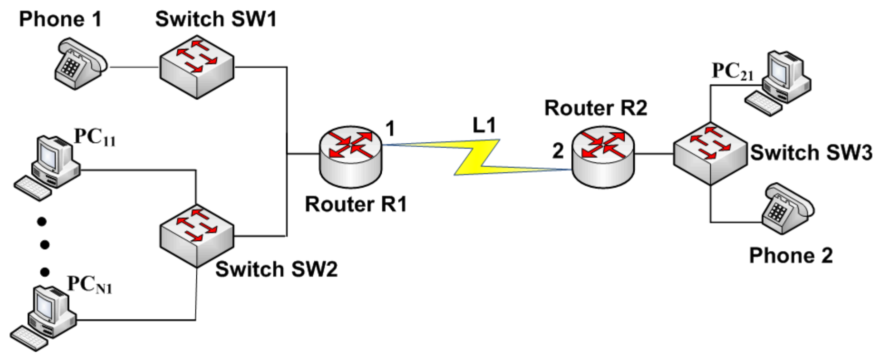
PQ provides static allocation of network resources in an interface. It has a set of queues and a scheduler that empties them in priority sequence. When asked for a packet, the scheduler inspects the highest priority queue and, if there is data present, returns a packet from that queue [9]. There are four output queues: *High*, *Medium*, *Normal* and *Low*. The network component places traffic in these queues, based on predefined filters. *PQ* was configured with *High* and *Medium* queues being used.

Figure 6 shows the simulation results for the *PQ* queuing policy. This figure depicts the influence of the number of machines (6, 8, 10 and 12) and the influence of the queue (*High* or *Medium*) where voice traffic is configured on the MOS_a index that considers the performance and dependability aspects.

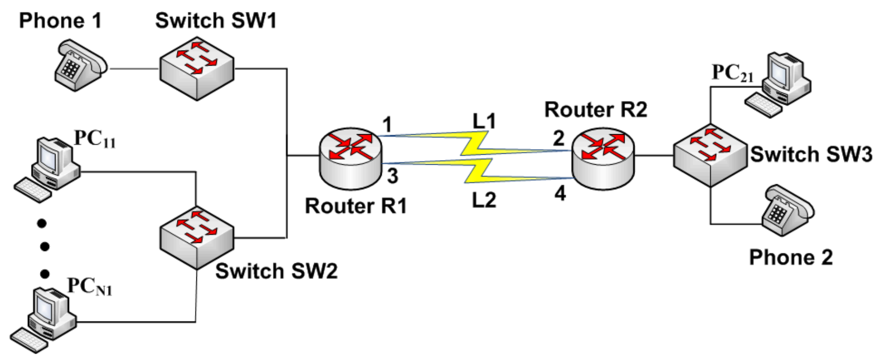
In the Figure 6(a), the G.711 *CODEC* was used. When the voice traffic is configured to the *high* queue, the MOS_a index is not influenced by the number of machines in both A_1 and A_2 architectures. This is due to the fact that the *high* queue has the highest priority. In turn, when the voice service is configured to the *medium* queue, the MOS_a index is sensitive to the increase in the number of machines. In both cases, there is a small variation of this index due to the effect of the availability of each architecture.

Figure 6(b) shows the behavior of the G.729 *CODEC*. This *CODEC* provides a stability of the MOS_a index values for both *high* and *medium* queues. In turn, this index presents a small variation resulting from the effect of the availability of each architecture.

Considering *CQ* queuing policy, Figure 7 shows both the influence of the allocated network resources (25,00%, 50,00% and 75,00%) to the voice traffic and the influence of the number of machines (6, 8, 10 and 12) on the MOS_a index.



(a) Architecture A_1



(b) Architecture A_2

Figure 4. Architectures A_1 and A_2

Table 1. Resulting Availability and Dependability Parameters of the Architectures A_1 and A_2

Architecture	Availability	MTTF(R1)	MTTR(R1)	MTTF(R2)	MTTR(R2)	MTTF(L1)	MTTR(L1)	MTTF(L2)	MTTR(L2)
A_1	0.9608	105,000	12	105,000	12	296	12	–	–
A_2	0.9985	105,000	12	105,000	12	296	12	296	12

Table 2. Encoding and MOS

Coding	ITU-T Standard	Bit Rate (kbps)	Coding Delay (ms)
PCM	G.711	64	0.125
CS-ACELP	G.729	8	20

In the Figure 7(a), the G.711 *CODEC* was used. The voice traffic has a greater sensitivity to the number of machines generating data traffic when it has 25% or 50% of the allocated network resources. On he other hand, this traffic is less sensitive when it has 75% of the allocated resources. This is demonstrated by the MOS_a index variation. In both cases, the availability of each architecture causes a small variation of this index.

Figure 7(b) shows the behavior of voice traffic when the G.729 *CODEC* was used. It is interesting to note that the voice traffic is less sensitive to variations in both allocated network resources and data traffic. The availability of each architecture also causes a small variation of this index.

6. Conclusion

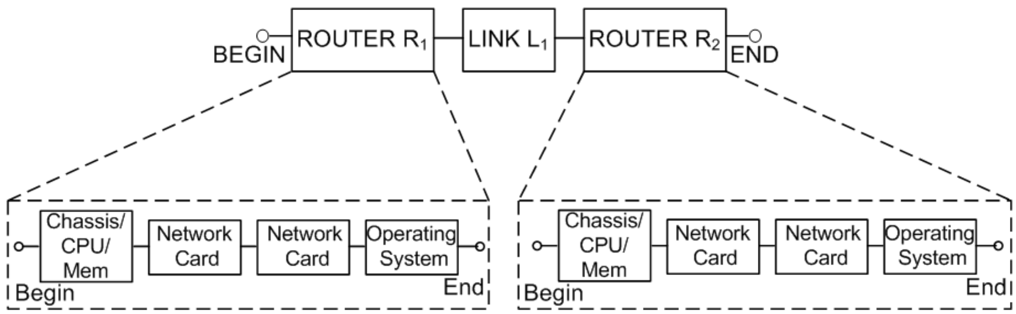
This work proposed simulation and analytical models to evaluate performance and dependability aspects of voice services. Several architectures and scenarios were constructed combining different number of machines, queuing policies, *CODECs* and a redundancy mechanism.

The merging of performance and dependability issues into the same problem is one of our contributions. Another relevant contribution is the proposed MOS_a index that aims to quantify the effects of the infrastructure availability on the MOS index, associated with each voice connection. This index showed that the G.711 *CODEC* has a higher sensitivity both in terms of data traffic and allocated network resources.

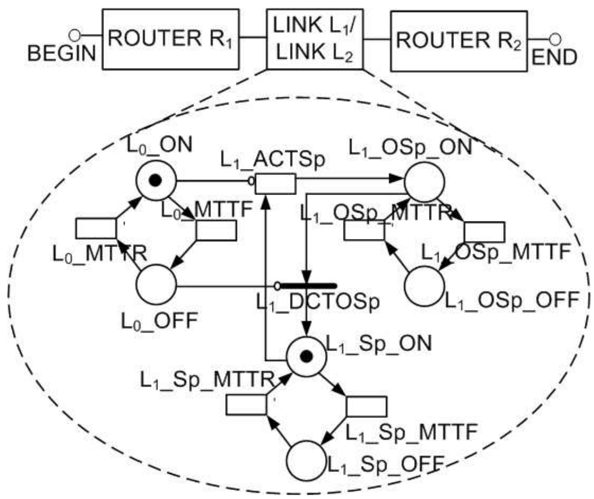
For future work, we plan to extend these models to include the effects of different maintenance policies, multiple repairing units as well as taking into account different recovery strategies.

Author contributions

The authors contributed equally to this work.



(a) Hierarchical Model - A_1



(b) Hierarchical Model - A_2

Figure 5. Hierarchical Models - A_1 and A_2

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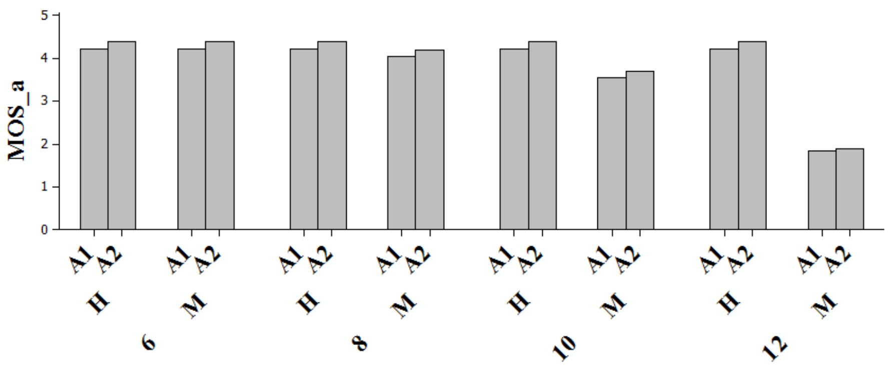
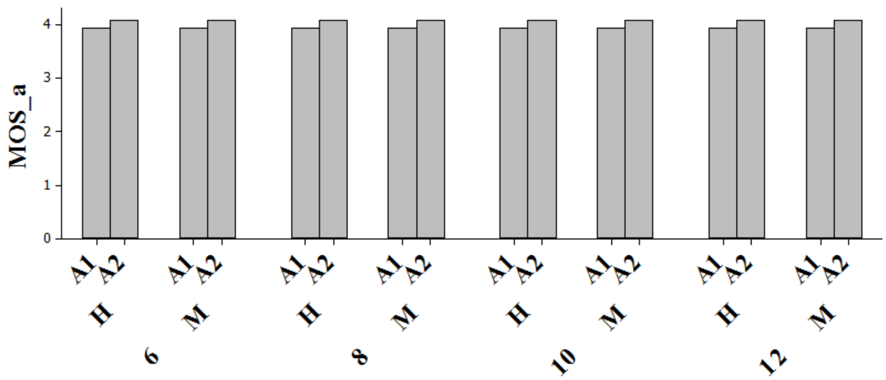


(a) G.711 CODEC



(b) G.729 CODEC

Figure 6. MOS_a index in accordance with the availability of the subjacent infrastructures (architectures A_1 and A_2)

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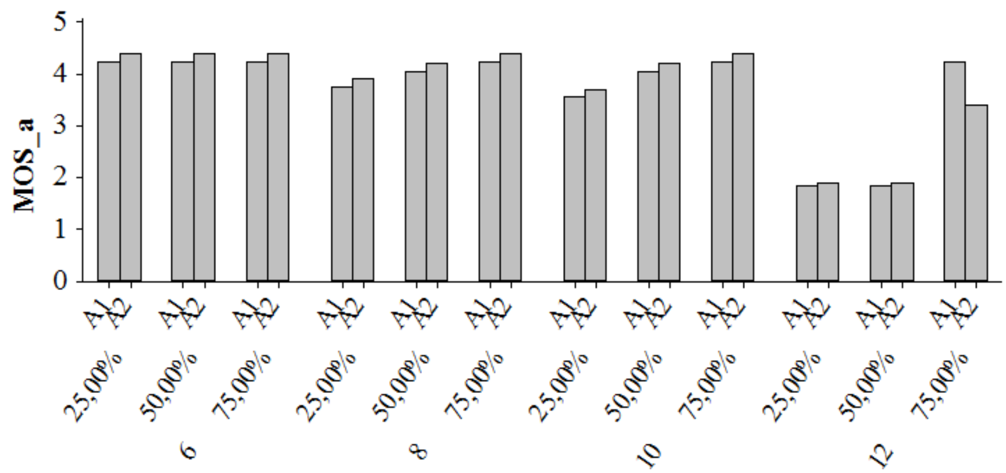
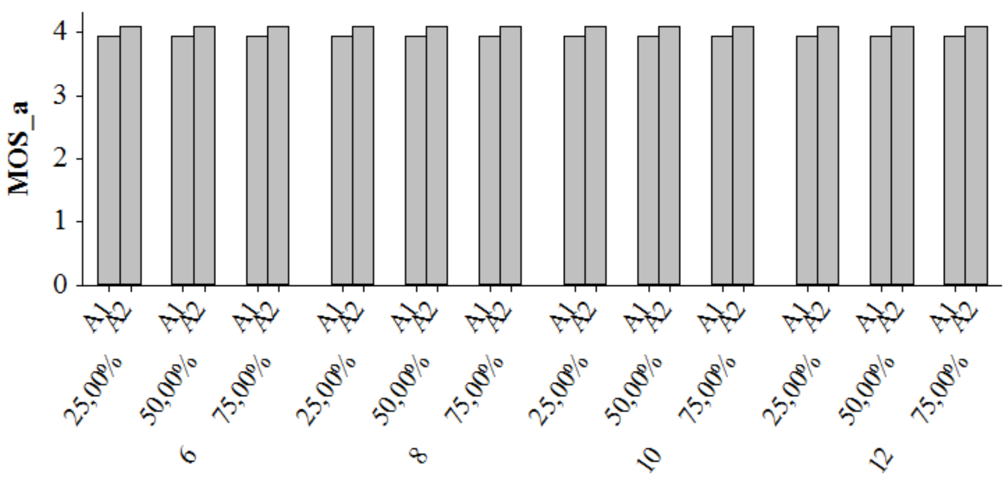


(a) G.711 CODEC



(b) G.729 CODEC

Figure 7. MOS_a index in accordance with the availability of the subjacent infrastructures (architectures A_1 and A_2)

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