Available online at https://asric.africa/engineering-sciences
ASRIC Journal on Engineering Sciences Vol.5(1)(2024) 124-132

Off-the-Shelf Infrastructure for Innovative Mobile Communication Services in Angola

Hermenegildo Marinha Cassoada[1], Ngombo Makaya Armando[2]

[1]Innovation Team - eSupport Telecom, Luanda-Angola, 1000012876@ucan.edu

[2]Department of Informatics Engineering - Kimpa Vita University, Bairro Kondo Benza 077 Uíge-Angola, ngombo@ieee.org

Received 09 April 2024; revised 08 July 2024; accepted 30 September 2024

Abstract

VoIP technology implementation over 802.11 standard in corporate environment is a challenge when it comes to ensuring Quality of Experience (QoE) before and during the handoff process. To this regard, we have developed a solution leveraging off-the-shelf enablers to insure high-quality mobile VoIP service with in a wireless Local Area Networks (LAN). Our case study has been assessed via simulative and laboratory experimentation methods. We also applied the MOS conversation opinion test method to 42 users, who evaluated the QoE of the system, according to the ITU-T P.800 recommendations. The tests carried out both in laboratory and corporate environments showed a satisfactory QoE (median MOS of 4.5 / 5.00), including cases of concurrent and conference calls. Such results guarantee a low-cost high-quality mobile communication service over the 802.11. In this context, our solution is a potential alternative to the conventional mobile networks, mainly in current days of coverage deficit and the galloping QoE degradation offered by Angolan mobile operators.

Keywords: VoIP; WLAN; ITU-T P.800; handoff; QoE.

I. Introduction

VoIP (Voice over Internet Protocol) is a service that allows voice to be transmitted over IP networks (Ribeiro & Mendes, 2008). To support VoIP calls, communication systems can be made over wired medium, wireless, or mixed access networks. However, in any of the aforementioned infrastructures, the voice signal can be affected by several degradation factors, which impact the quality at the reception point (Affonso et al., 2018). These degradation factors influence the subjective perception of such a real-time communication service i.e. voice calls (Holub et al., 2018). VoIP communication service has been widely adopted, providing an estimated market of USD 20+ billion (Taylor, 2023). Still, Quality of Experience (QoE) is not guaranteed for the quality of the voice signal is sensitive to degradations that may occur in the IP network (Affonso et al., 2018) . Companies adopt VoIP solutions aiming the reduction in communication costs and converging services, but it is generally implemented over wired access networks. This implementation approach is not designed to allow mobility of the user equipment (UE), which in some scenarios affects the quality of services provided by institutions. To provide mobility in a VoIP system, it must be implemented over a wireless access network, which in turn, must have mechanisms that guarantee the transition of the UE signal from one cell to another (Sánchez-Reyes et al., 2018) . This process occurs when a mobile device goes beyond the borders of a given cell and enters another, all the process being controlled by another Radio Base Station (ERB). The voice quality communication through a wireless network largely depends on the network quality (Affonso et al., 2018). When the handoff/handover occurs, the wireless network needs a certain quality standard so that voice traffic can be delivered under acceptable conditions, especially during the handoff.

Currently, improving the handover is still a challenge in the research community (AlShahwan et al., 2018). On the other hand, we know that VoIP technology is an effective and economical alternative to replacing conventional telephony. Therefore, one of the main challenges of this technology, when implemented over a WLAN, is to guarantee QoE during the handoff process. To this regard, the main contribution of the current study is to overcome these challenges, by designing, implementing and finally, evaluating a mobile VoIP system with good QoE. The rest of this paper is structured as follows: In Section 2 we present the related works. In section 3, we present the developed prototype and show the results obtained from its evaluation, before their analysis and discussion, in section 4. We conclude this paper in Section 5, identifying some opportunities for future research and an innovation-based IT business.

II. Methods, techniques, studied material and area descriptions

II.1. Related Works

The literature review was carried out according to PRISMA criteria (Page et al., 2021) with a time frame from 2017 to 2022. During the search process, a total of 208 records were found, among them 203 from selected scientific databases (see PRISMA flowchart in **Error! Reference source not found.**).

Measuring the Subjective Performance of a Voice Service

The Quality of Service (QoS) is the minimum guaranteed performance of technologies and applications acceptable on the network through specific techniques with the aim of providing users with satisfactory experience (Adhilaksono & Setiawan, 2022; Boz et al., 2019; Mayor et al., 2022; Affonso et al., 2018). QoS indicates the qualities of service expressed in terms of delay, data rates and packet loss rates. QoS provides mechanisms to improve the performance of computer networks facilitating traffic prioritization, resource reservation, traffic shaping and policing, packet scheduling, and queue management operations (Hong et al.,2018; Salama & Saatchi, 2019). Adhilaksono & Setiawan presented a study on VoIP quality metrics, reviewing 38 works, with the aim of finding out which metrics are the most preferred for measuring the quality of VoIP applications. They found out that the most preferred Quality of Experience (QoE) metrics are: Mean Opinion Score (MOS), Perceptual Assessment of Speech Quality (PESQ) and Peak Signal-to-Noise Ratio (PSNR). Mayor et al.(2022) also evaluated VoIP QoS based on the MOS approach, while Boz et al. (2019) present QoE models that quantify the relationship between user experience and network service quality. MOS is a numerical scale defined by the ITU-T P.800 recommendation (Keller, 2011) to measure and evaluate calls quality whose values are obtained subjectively from a group of users who evaluate and classify the audio by assigning a score (Adhilaksono & Setiawan, 2022; Holub et al., 2018). MOS score for digital telephony on the PSTN ranges from 4.4 to 4.5. For a call to be considered of good quality, MOS score must be equal to or greater than 3.5 and generally the score for a cellular call varies from 3.8 to 4.0.(Keller, 2011).

VoIP Limitations

Although VoIP communications are widely adopted, its QoE is still not guaranteed since the quality of the voice signal depends on several factors and can be affected by different types of degradations that occur in network infrastructure (Affonso et al., 2018). In this work, authors propose a new non-intrusive model for evaluating speech quality, because the models proposed in Recommendation ITU-T P.862 are not suitable for real VoIP scenarios and ITU-T P.563 does not provide reliable results. Long-lasting calls have been appointed do present better call quality and consequently better QoE. To this regard, Holub et al. (2018) explored the dependence between the average duration and quality of VoIP calls and corroborate with Affonso et al.(2018) about various degradations that affect telephone calls carried over IP networks and their influence on users' subjective perception of call. Unnecessary handoffs can cause signal overload and increase energy consumption, especially in areas with intensive use of networks using the Wi-Fi access protocol. This overload negatively affects the system, as voice is very sensitive to delays (Lin et al., 2019). . Although every WLAN is an extension of a wired network allowing users mobility, signal may suffer losses during its path in free space (Villarreal et al., 2019; Holub et al., 2018; AlShahwan et al., 2018).

VoIP Quality Improvement Approaches

Because of VoIP low costs, this technology is being considered as the main means of voice transmission in cellular systems (4 generation onwards) and WLAN (Lin et al.(2019) e Ribeiro & Mendes, (2008), Sánchez-Reyes et al.(2018)). The literature also emphasizes that the handoff procedure in VoIP systems over WLAN is considered a basic QoS parameter, in such a way that it is guaranteed that an ongoing call is not interrupted when a user moves between cells. To obtain a quality service and minimize negative impact that network traffic can cause, it is very important to manage devices necessary for this service. (Neyra et al., 2017). To achieve such goal, it is not only necessary to maintain QoS parameters. Other criteria are beginning to be required that consider how the customer experiences the quality that is evaluated. At receiver level, some packets may be lost due to delays, congestion or transfer errors (Bakri et al., 2018). This packet loss causes loss of synchronization between the encoder and decoder (Affonso et al.(2018), Holub et al.(2018), AlShahwan et al.(2018) e Villarreal et al.(2019)).

Chakraborty et al.(2020) and Sánchez-Reyes et al.(2018) studied the unprecedented popularity of VoIP in mobile applications. Among the causes of this mass and growing adoption are its easy integration with IP-enabled services, low deployment and maintenance costs. VoIP is rapidly becoming the main communication means in 5G networks, so that ensuring improved QoS is a constant challenge. Chakraborty et al. (2020) develop a new technique to accurately predict and allocate channels with sufficient QoS guarantees for VoIP calls in heterogeneous networks. H.323 recommendation (ITU-T, 2022) describes terminals and other entities that provide multimedia communication services over packet-based networks without guaranteed QoS. This recommendation specifies what H.323 entities can provide, relatively to real-time audio, video and/or data communication. the recommendation also clarifies that support for audio is mandatory, while data and video are optional. However, when enabled to use a specific mode of operation, it is required that all terminals interconnect and support enabled media type.

With continued users growth and increasing QoS improvements, WLANs have emerged as one of most dominant and influential wireless networks in human life (Qu et al., 2018). In this article, author discusses in detail most diverse implementation key scenarios of *IEEE 802.11ax* technologies, including improvements made to PHY layers and *MU-MAC* (*multi-user medium access control*), spatial reuse, power efficiency, standardization and latest academic studies.

Literature Gaps and Research Opportunities

The literature is abundant of contributions and discussion around improving communication systems quality, as well as the search for alternative communication systems that guarantee service quality and allow for cost reduction. However, we found a gap in literature analyzed regarding assessment of QoE in VoIP WiFi with handoff. This is the main motivation of our study i.e., clinging to this literary loophole and seeking to improve the quality of alternative voice services to public switched telephone network (PSTN). In the next section we present the prototype implemented over wireless network (802.11) with handoff. The objective of our solution is to guarantee mobility to users without reducing noticeable levels of QoE when compared to VoIP with fixed terminals. Finally, such a solution would provide not only cost reduction when replacing PSTN, but also network flexibility and terminal mobility.

II.2. Architecture of the Prototype

As displayer in Fig.1, our solution consists of an SDN or SD- Wi-Fi with AI roaming, constituted by two BSS and one ESS controlled by an Omada controller v5.13.23 TP- Link compatible with EAP 245. The desktop application was installed on an ASUS AMD E2 personal computer, with 8 Go of RAM and 256 Go SSD. Asterisk is a hybrid telephone exchange that implements both the functions of a traditional telephone exchange and VoIP protocols (Keller (2011). Asterisk central has ability to manage audio traffic in digital and analog communication channels and in networks TCP/IP. The server is also known as B2B (Back-to-

Back) User Agent, for establishing phone call and continuing monitor audio traffic between these points. Table 1 summarizes the main hardware components implemented in our solution.

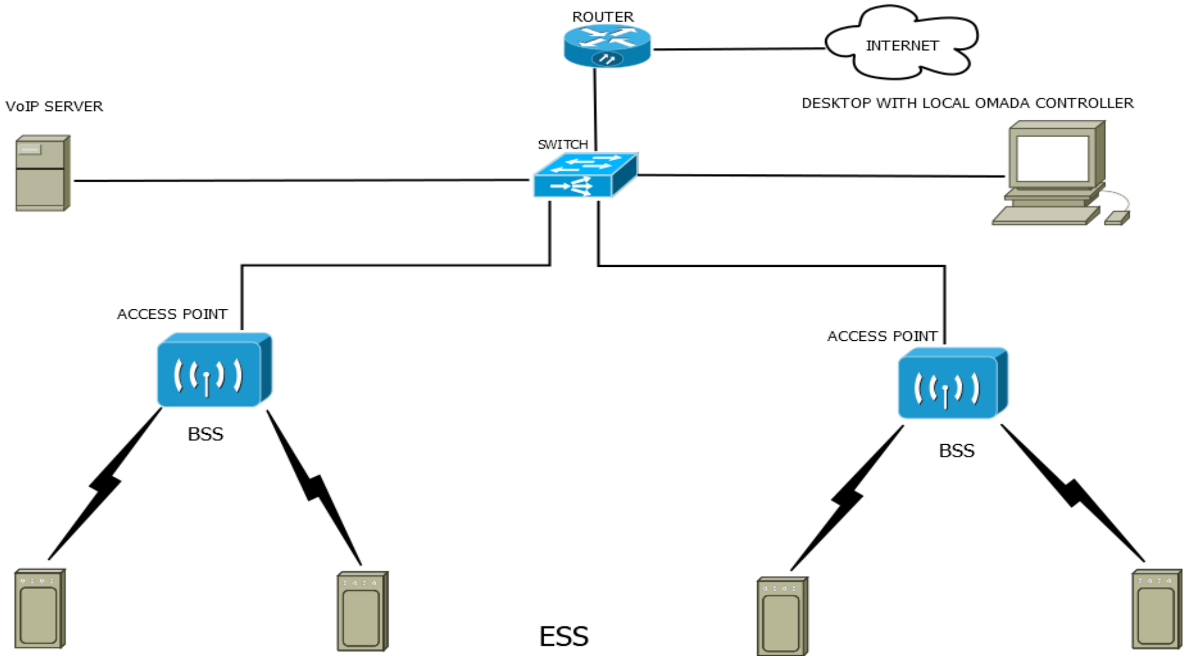


Figure 1-System architecture developed

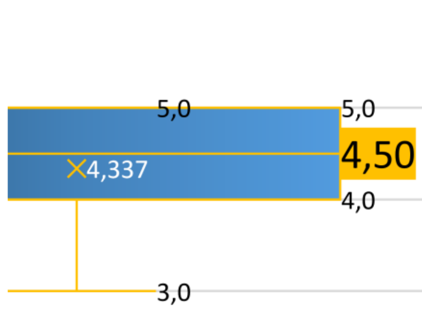
Table 1 *- Components used in implementing the solution*

Components	**Brand model**	**Function**	**Characteristics**
Router	**Cisco 2801**	**Network gateway** **DHCP Server**	**Manageable**
L2 Switch	**Cisco 2960**	**Create the ESS**	**Manageable**
VoIP Server	**Asterisk**	**Allocate asterisk services**	**Core I5, 4GB RAM**
Firewall	**Embedded in Asterisk**	**Protect the VoIP server**	**-**
Access points	**Tp link eap 245**	**Create the BSS's**	**802.11 ac dual band, MU-MIMO, WMM**
Controller	**Omada controller**	**Manage the SDN**	**Remote management and configuration**
Softphones	**Zoiper**	**end user software to access the VoIP service**	**Free (multipltaform)**
Codec	**G.711**	**Coding the calls**	**Ulaw**
Network cables	**twisted pair**	**Connect devices to the distribution network**	**UTP Cat. 6**

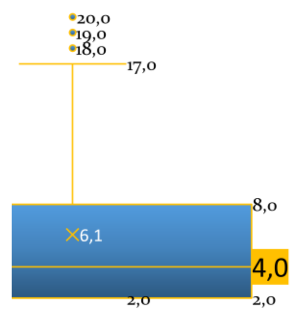
In the next sections we present the results obtained on both objective and subjective evaluations of our prototype. The evaluation has been done via simulative and experimental methods, according to the ITU-T P.800 recommendations. The experiments on the prototype included 103 calls made by 42 participants/testers for 33 continuous days. The array of calls evaluated were distributed as follows: 42 simultaneous, 35 non-concurrent and 26 conferences.

III. Results

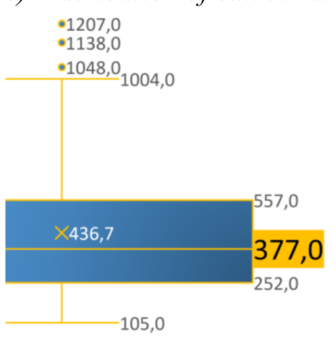
a) Distribution of the median MOS



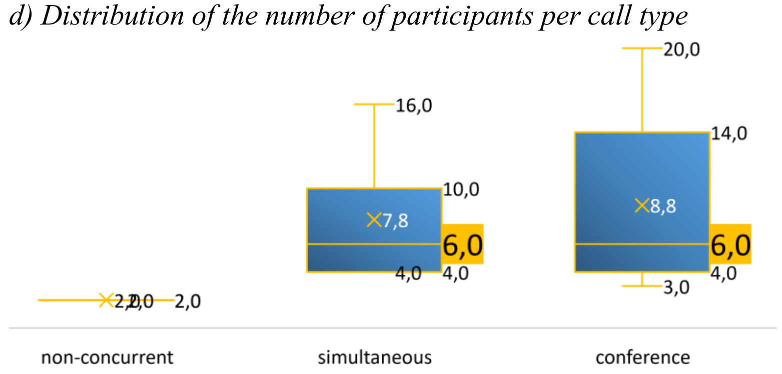
b) Distribution of connected UE per call



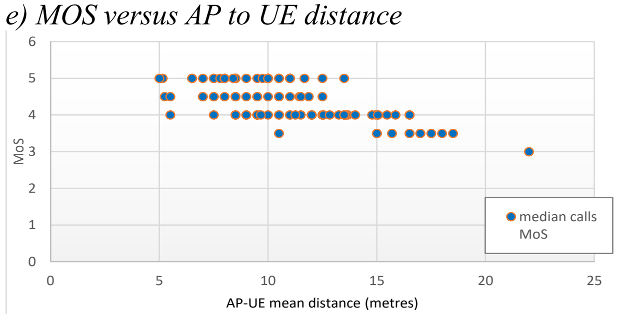
c) Distribution of call duration



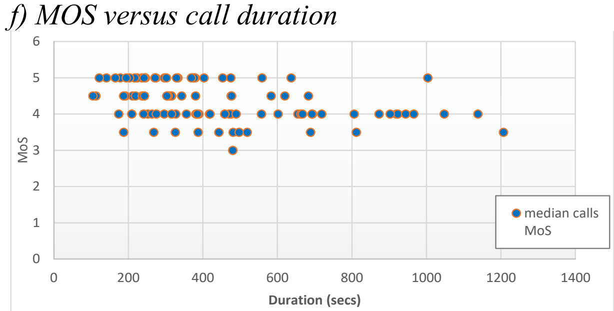
d) Distribution of the number of participants per call type



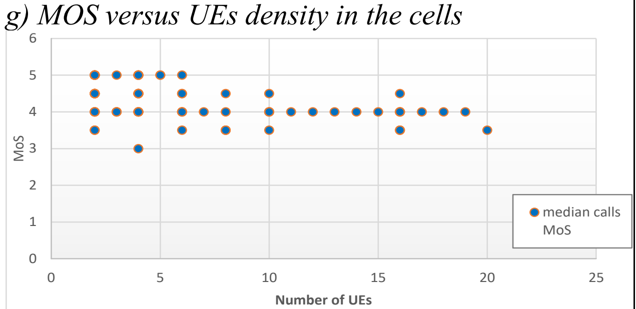
e) MOS versus AP to UE distance



f) MOS versus call duration



g) MOS versus UEs density in the cells

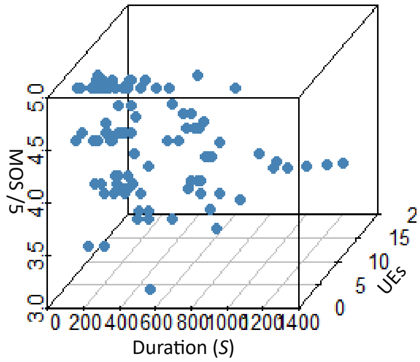


h) Pearson coefficients of correlations

	Duracao	Qtd de ramais em linha	distancia média	MoS médio da chamada
Duracao	1.00	0.58	0.38	-0.46
Qtd de ramais em linha	0.58	1.00	0.64	-0.56
distancia média	0.38	0.64	1.00	-0.71
MoS médio da chamada	-0.46	-0.56	-0.71	1.00

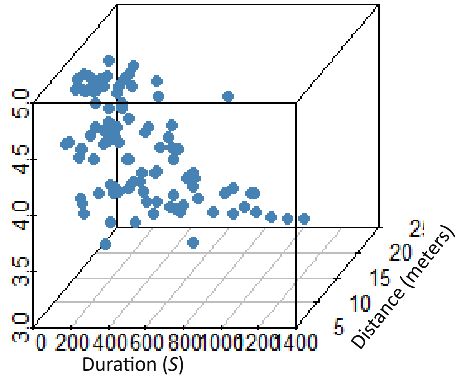
__Figure 2__- QoE and correlation pattern

a) *Impact of Call duration and Conneted UE on MOS*



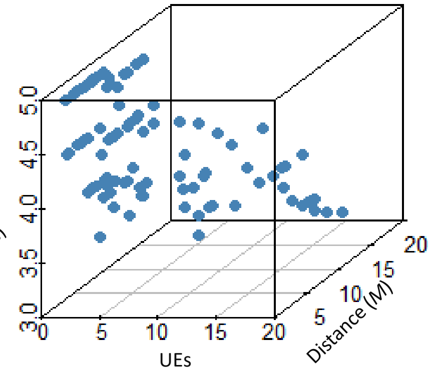
X axis: MOS
Y axis: Call duration
Z axis: Connected UEs

b) *Impact of Call duration and AP-UE distance on MOS*



X axis: MOS
Y axis: Call duration
Z axis: AP-UEs distance

c) *Impact of Conneted UE and AP-UE distance on MoS*



X axis: MOS
Y axis: Connected UEs
Z axis: AP-UEs distance

__Figure 3__- Linear Regression patterns

Both raw data collected and further details on the obtained results of the evaluation are available at the following page: https://github.com/gildo1996/merged.dataset/blob/main/dataSet.%20of%20the%20shelf%20infrastructure%20for%20inovative%20mobile%20communications.pdf

IV. Discussion

IV.1. QoE and Correlations

As displayed in *Fig. 2-a)*, the results show a satisfactory QoE (median MOS =4.5, mean > 4.3 /5.00). This performance orresponds to a high quality mobile audio communication solution as defined by Keller (2011) and low cost, for its IEEE 802.11 RAN-based infrastructure. Calls were made with an average of 6 UEs connected per call, with 50% of these being made with 4+ UE online and 20 was the maximum number of extensions online *(see Fig. 2-b)*. Calls lasted an average of 437 seconds, 75% of these lasted >= 252 seconds, while the longest call lasted 1207 seconds and the shortest lasted 105 seconds *(see Fig. 2-c)*. 22 users made non-competing calls and during competing calls, an average of 8 extensions were online. The maximum number of extensions online during this type of call was 16. An average of 9 UEs participated in conferences, with a minimum and maximum users' number of 3 and 20, respectively *(see Fig. 2-d)*.

Overall, long distances presented lower QoE, contrary to short distances (*see Fig. 2-e)*. This result is a classic one as comes to telecommunications due to the attenuation of the communication signal over distance. The solution for such phenomenon is usually an increase the number of cells or the installation of APs with greater capabilities both in range and number of connected UEs. Short call durations presented lower QoE *(see Fig. 2-f)*. We therefore conclude that call duration does not objectively influence QoE. What can happen is that over time, depending on the interest of the conversation, users tend to show a certain annoyance and a consequent degraded evaluation of the system. The increment of extensions/UEs online tends to negatively influence QoE *(Fig. 2-g)*. Finally, *Fig. 2-h* displays the Pearson correlation coefficients between the studied variables, with the following analysis:

1. There is an almost irrelevant positive correlation (0.38) between the duration of calls and the average distance of users, not allowing us to conclude anything about it.

2. There is a weak negative correlation (-0.46) between call duration and average call MOS. Generally short durations presented lower QoE. However, we cannot conclude about the reciprocity of the phenomenon because the correlation factor is weak.
3. There is a moderate positive correlation (0.64) between the number of extensions online and the average distance between users and APs. This result informs that during the tests the distances between users and APs and the number of extensions online were directly proportional.
4. There is a moderate negative correlation (-0.56) between the number of extensions online and the average MOS of calls, which allows us to conclude that the increase in extensions online negatively affects QoE.

IV.2. Multiple Linear Regression

As a complement to the the performance tests presented above, we also looked for Multiple Linear Regression between the studied variables. We tested the normality of the model's waste with the *Shapiro.test* function. As shown in Appendix II.a, the P-value is greater than 50%, meaning that the distribution of our model is approximately normal. The most important assumption for multiple linear regression analysis is the lack of multicollinearity between the independent variables. In fact, the opposite influences the standard error of the coefficients, making it difficult to estimate the parameters, distorting the model results and making them ambiguous for interpretation. There is multicollinearity between two variables of a model when the correlation coefficient between the two is higher than 0,8 or their inflation value factor (VIF) is greater than 10 (Miloca & Conejo, 2017). To verify the existence of multicollinearity, we use the correlation panel between the variables of the model or check their VIF. There is an absence of multicollinearity between the independent variables of the study, since no correlation coefficient between them is higher than 0.8 (see Appendix III). After verifying the VIF values of the independent variables, the absence of multicollinearity between them is confirmed, since they are well below 10 (see Appendix II.b).

By analyzing the multiple regression of the model, we found that a unit of distance has the greatest impact on QoE than a unit of the other independent variables. More precisely, we verified that for the prototype implemented, each second of call duration degraded -0.000436 the QoE, while each line extension degraded -0.006598 and each meter -0.095135 of the QoE (see Appendix II.c).

V. Conclusion

The present work consisted of a case study leveraging off-the-shelf technologies to insure high-quality mobile VoIP service with handoff in WLANs. The evaluation tests have been carried out according to the ITU-T P.800 recommendation. We obtained a 90% satisfactory QoE (median MOS = 4.5 / 5.00). Our prototype guarantees a high-quality mobile audio communication solution and low cost. Regarding the negative impact of the AP-UEs distance on the MOS, we will study the portability of the system in 802.16 networks. In a context of a galloping increase in personal mobile terminal devices, the challenge of applied research on VoIP systems in 802.16 networks will ensure quality audio communications, beyond short-range coverage areas. Finally, we plan to implement an industrial version of the prototype, labelled *AngoVoIP*. To this regard, we intend to allocate the VoIP distributed and cloud-based server with a commercial version of asterisk. Concerning RAN, we will leverage a synergy of 802.11/802.16-based access points for pervasive, long-distance coverage. For all these aspects, we believe our project can be an alternative to conventional mobile networks, with the potential to contribute to improving communication in the country as well as reducing their costs.

VI. Acknowledgment(s)

The authors are grateful to *eSupport Telecom, Inc* workers and all the participants that anonymously took part of the MOS assessment tests.

VII. References cited

Adhilaksono, B., & Setiawan, B. (2022). A study of Voice-over-Internet Protocol quality metrics. *Procedia Computer Science*, *197*, 377–384. https://doi.org/10.1016/j.procs.2021.12.153

Affonso, E. T., Nunes, R. D., Rosa, R. L., Pivaro, G. F., & Rodriguez, D. Z. (2018). Speech Quality Assessment in Wireless VoIP Communication Using Deep Belief Network. *IEEE Access*, *6*, 77022–77032. https://doi.org/10.1109/ACCESS.2018.2871072

AlShahwan, F., Alshamrani, M., & Amer, A. A. (2018). Dynamic Novel Cross-Layer Performance Enhancement Approach for SIP over OLSR. *IEEE Access*, *6*, 71947–71964. https://doi.org/10.1109/ACCESS.2018.2880917

Bakri, A., Amrouche, A., Abbas, M., & Bouchakour, L. (2018). Automatic Speech Recognition for VoIP with Packet Loss Concealment. *Procedia Computer Science*, *128*, 72–78. https://doi.org/10.1016/j.procs.2018.03.010

Boz, E., Finley, B., Oulasvirta, A., Kilkki, K., & Manner, J. (2019). Mobile QoE prediction in the field. *Pervasive and Mobile Computing*, *59*, 101039. https://doi.org/10.1016/j.pmcj.2019.101039

Chakraborty, T., Ghosh, S., Barik, S., Kar, S., & Chatterjee, S. (2020). VoIP-HDK – A novel channel allocation technique for QoS aware VoIP communication over heterogeneous networks. *Procedia Computer Science*, *171*, 62–71. https://doi.org/10.1016/j.procs.2020.04.007

Holub, J., Wallbaum, M., Smith, N., & Avetisyan, H. (2018). Analysis of the Dependency of Call Duration on the Quality of VoIP Calls. *IEEE Wireless Communications Letters*, *7*(4), 638–641. https://doi.org/10.1109/LWC.2018.2806442

Hong, E.-K., Baek, J.-Y., Jang, Y.-O., Na, J.-H., & Kim, K.-S. (2018). QoS-guaranteed scheduling for small cell networks. *ICT Express*, *4*(4), 175–180. https://doi.org/10.1016/j.icte.2017.11.017

ITU-T. (2022). *SERIES H: AUDIOVISUAL AND MULTIMEDIA SYSTEMS Infrastructure of audiovisual services – Systems and terminal equipment for audiovisual services*.

Keller, A. (author name) (2011) (year) Introdução ao VoIP e ao Asterisk (chapter name) In: Prates R (editor of the book) *Asterisk na prática* (book name), 2nd edn (edition), NovaTec (publisher house), São Paulo.

Lin, C.-Y., Chang, H.-L., Wu, M.-T., Cai, Y.-Z., Wang, Y.-T., Chen, L.-A., Tsai, M.-H., & Liu, R.-S. (2019). Reducing Signal Overload by Disconnection Tolerant Voice Service in Heterogeneous Networks. *IEEE Access*, *7*, 332–346. https://doi.org/10.1109/ACCESS.2018.2885361

Mayor, V., Estepa, R., Estepa, A., & Madinabeitia, G. (2022). Deployment of UAV-mounted Access Points for VoWiFi Service with guaranteed QoS. *Computer Communications*, *193*, 94–108. https://doi.org/10.1016/j.comcom.2022.06.037

Miloca, S. A., & Conejo, P. D. (author name), 2017 (year). *Multicolinearidade em Modelos de Regressao* (article name),Colegiado do Curso de Matematica - Centro de Ciências Exatas e Tecnológicas da Universidade Estadual do Oeste do Parana (journal name), XXII SEMANA ACADÊMICA DA MATEMÁTICA.

Neyra, J. M., Irizar, C. A., & Calderón, C. A. (author name) 2017(year). *Evaluación de QoE en servicios IP basada en parámetros de QoS* (article name), RIELAC (journal name), 38 (volume Number) (3) (Issue): 36-46 (pages).

Page, M. J., McKenzie, J. E., Bossuyt, P. M., Boutron, I., Hoffmann, T. C., Mulrow, C. D., Shamseer, L., Tetzlaff, J. M., Akl, E. A., Brennan, S. E., Chou, R., Glanville, J., Grimshaw, J. M., Hróbjartsson, A., Lalu, M. M., Li, T., Loder, E. W., Mayo-Wilson, E., McDonald, S., … Moher, D. (2021). The PRISMA 2020 statement: An updated guideline for reporting systematic reviews. *BMJ*, n71. https://doi.org/10.1136/bmj.n71

Qu, Q., Li, B., Yang, M., Yan, Z., Yang, A., Yu, J., Gan, M., Li, Y., Yang, X., & Chen, K.-C. (sem data). *Survey and Performance Evaluation of the Upcoming Next Generation WLAN Standard—IEEE 802.11ax*.

Ribeiro, N. J., & Mendes, L. A. M. (author name) 2008 (year) . *VoIP – Tecnologia de Voz sobre IP* (article name), Enfoque UTE (journal name), 10 (volume Number) (4) (Issue): 28-44 (pages)..

Sánchez-Reyes, S., Rivero-Angeles, M. E., & Torres-Cruz, N. (2018). Teletraffic Analysis for VoIP Services in WLAN Systems with Handoff Capabilities. *Computación y Sistemas*, *22*(3). https://doi.org/10.13053/cys-22-3-2749

Taylor, P. (2023). *Mobile VoIP market size worldwide by region 2014-2024*. https://www.statista.com/statistics/691602/global-mobile-voip-market-size-by-region/

Villarreal, M. A. R., Arroyo, J. B. M., Camarena, J. F. M., & Lira, L. A. N. (author name) 2019 (year). *Evaluación de una red inalámbrica de banda ancha para VoIP* (article name), UNIPAC (journal name).

List of abbreviations

5G	5th Generation
AI ROAMING	Artificial Intelligence Roaming
AP	Access Point
B2B	Back-To-Back

BSS	Basic Service Set
DHCP	Dinamic Host Configuration Protocol
EAP	Extension Authentication Protocol
ESS	Extended Service Set
ERB	Estação Rádio Base
IEEE	Institute Of Electrical And Electronic Engineers
IP	Internet Protocol
ITU-T	Telecommunication Standardization Sector
IT	Information Technology
LAN	Local Area Network
MOS	Mean Opinion Score
MU MAC	Multi-User Medium Access Control
MU MIMO	Mu-Mac-Multi-User Medium Access Control
PESQ	Avaliação Perceptiva Da Qualidade Da Fala
PRISMA	Preferred Reporting Items For Systematic Reviews And Meta-Analyses
PSNR	Relação Sinal-Ruído De Pico
PSTN	Public Switched Telephone Network
QOE	Quality Of Experience
QOS	Quality Of Service
RAM	Randon Access Memory
RAN	Radio Access Network
SDN	Software Defined Network
SSD	Solid State Drive
TCP	Transmission Control Protocol
TP LINK	Twisted Pair Link
UDP	User Datagram Protocol
EU	User Equipament
UTP	Unshielded Twisted Pair
VIF	Inflation Value Factor
VOIP	Voice Over Internet Protocol
WI-FI	Wireless Fidelity
WLAN	Wireless LAN
WMM	Wi-Fi Multimedia