Achieving optimal performance and quality in LAN and WLAN for mission critical applications

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Abstract. Voice Over Internet Protocol (VoIP) properties are vital for its reliability in mission-critical applications. This research aims to find network topology, call signaling and voice codecs property combinations that meet reliability targets of VoIP communication in a Small Office Home Office (SOHO) environment where network resources may be limited but reliable and secured operation is essential. Local Area Network (LAN) and Wireless LAN (WLAN) scenarios are evaluated using Quality of Service (QoS) and Mean Opinion Score (MOS) measurements to find which property combinations satisfy predefined classes; best-quality and best-performance. The research extended Roslin et al. [1] on LAN VoIP to WLANs, and validated Khiat et al. [2]'s and Guy [3]'s work that argued SIP was effective in optimal set up. This research found that VoIP combinations offer some desirable characteristics, but at the cost of other properties required, leading to categorisation being based on interpretation of the results, concluding that though, not ideal for mission-critical applications, combinations functions well replicating real-world scenarios. Analysis also established VoIP's scalability for application-based configurations, impact of VoIP's modularity and ease-of-configuration in achieving user expectations. Further property testing can solidify VoIP's capabilities to function for mission critical environments.

Keywords: VoIP; LAN, WLAN, Security; Mission-critical ; QoS; MOS

1 Introduction

Internet-based Voice over Internet Protocol (VoIP) communication has been popular for interactive communication services like video and voice conferencing along with traditional dedicated, wired systems like public switched telephone network (PSTN) and Integrated Service Digital Network (ISDN) for many years. However, wired and dedicated infrastructure is expensive and challenging in the modern world, where mobility and flexibility have become increasingly important. Therefore, with the growth of the internet, most businesses opted for VoIP that relies only on an internet connection. As a result, PSTN and ISDN are likely to be phased out, if not globally, in the United Kingdom within the next five years [4]. Nevertheless, VoIP is susceptible to network conditions like packet loss, jitter and end-to-end delay, and implications can create critical reliability and security issues, especially in the Small Office Home Office (SOHO) Local Area Network (LAN) and Wireless LAN (WLAN) environments where network bandwidth may be limited. Poor network conditions directly impact availability in the confidentiality, integrity and availability (CIA) triad. Existing research suggests that using VoIP property combinations such as call signalling, voice codecs and encoders properties, network topology and type, network conditions can be managed, including its quality and performance [1, 5-8]. This research aims to find property combinations that have ideal characteristics to meet SOHO user quality or performance requirements by using Quality of Service (QoS) and Mean Opinion Score (MOS), respectively in LAN and WLAN environments. Additionally, results can indicate the most appropriate VoIP properties for mission critical applications like emergency services or military communications that require VoIP performance and reliability.

The experiment also enhance work of following; Roslin at al. [1] carried out similar experiment on LAN architecture. This research aims develop their research further including WLAN architecture. [2] argued that SIP was effective in optimal set up times when evaluating VoIP protocols in IEEE 802.11 networks . This experiment intends to validate if SIP is optimal in both LANs and WLANs. [3] findings show that end-to-end delays are high in VoIP over wireless networks. This research also seeks to examine their findings on how and why WLAN topologies introduce undesirable network conditions like end-to-end delay.

The organisation of this paper is as follows; next section presents the background and related research. Section 3 defines the experimental design, how the experiment is conducted and what is being simulated. The simulation results and discussions are presented in section 4. Significant findings and future work are outlined in section 5. Lastly section 6 concludes the paper.

2 Background and related research

QoS and MOS are methods for analysing the performance and quality of VoIP traffic. MOS can be both objective and subjective [9, 10]. QoS is objective when measuring network conditions, service performance and quality characteristics [7, 11-13]. [7] investigated VoIP QoS performance in Wireless mesh networks by testing different VoIP properties. When evaluating MOS and QoS results, they found combinations of 802.11g standard with G.711 and G.729 codecs using Hybrid Wireless Mesh Protocol (HWMP) decreased VoIP QoS performance. [5] looks at coupling signalling protocols and codecs scheme in achieving VoIP QoS over LAN. It concludes that G.723.1 codec had higher jitter variation when compared to G.711 and G.729A in their LAN based study when evaluating performance using MOS and QoS. Both studies do not investigate WLAN infrastructure topologies or compare WLAN to LAN topologies when considering VoIP QoS and MOS measurements. Infrastructure based WLAN combinations may offer desirable performance or quality characteristics ideal for user requirements and VoIP applications.

[14] finds that call signalling property H.323 is more complex than call signalling property SIP, and that SIP is excellent for development and is cheaper than H.323. It also finds that SIP is most suitable for internal use in large and mid-sized enterprises. [15] survey findings concludes that SIP is an alternative to H.323 due to the protocol's complexity of H.323, and SIP is more popular than H.323. [5] has a similar approach to this survey. Their study concluded that the combination of G.711 and signalling SIP produced the best jitter and call quality results when testing multiple codecs over SIP and H.323.

[16] looks at the combination of VoIP properties to improve QoS over two LAN topologies. They investigate network conditions for property combination performance characteristics like jitter and end-to-end delay. However, the study does not cover the set-up times over WLAN and LAN topologies for each combination. Set up times can contribute to the evaluation of VoIP performance. The call initiation speed can be a user requirement, especially in mission-critical applications.

[1] investigates QoS for VoIP property combinations over LAN based topologies; LAN H.323 topology and LAN SIP topology. Network conditions results show each combination's performance characteristics over their test model. However, the study does not record the amount of traffic data sent and received over the network topologies used for their simulation. Traffic loss is also a contributing factor to VoIP quality and performance. The amount of traffic loss could determine if the network topology is robust enough to be able to handle VoIP telephony and multiple services at the same time. Traffic loss can evidence the effect on network conditions.

VoIP can be applied both on infrastructure and ad hoc networks. This reflects positively on its ability to phase out PSTN and ISDN. However, traditional internet infrastructures' quality and performance are more superior to that of ad hoc networks. Ad hoc networks do have limitations but are likely to further integrate into society if VoIP becomes application based and newer standards of IEEE 802.11 are developed to ensure reliability [8].
[2] found that SIP was effective in optimal set up times when evaluating VoIP protocols in mobile 802.11 networks. [3] showcases high levels of end-to-end delay after additional calls are added over infrastructure and ad-hoc networks in their VoIP simulation.

[17] mentions that VoIP continues to be adopted into industry as it can offer features similar to Private Branch eXchange (PBX) when looking at VoIP application security issues. It is important to deploy VoIP appropriately, especially when meeting user requirements and subsequently finding ideal use case applications like industrial, public or private services and SOHO environments. VoIP properties should showcase ideal characteristics to achieve requirements which could encourage further VoIP adoption. Furthermore, VoIP application type can also play a part in the choice of property combination. It's possible that industry and public services are likely to opt for performance orientated characteristics as high traffic and multiple call handling requirements heavily increase packet loss and network congestion [18, 19].

End-to-end delay and jitter are important factor of VoIP QoS and MOS research. Table 1 outlines the acceptable ranges for network conditions defined by standards/proposed by research community;

Table 1. Network condition Recommended ranges

Network condition	Acceptable ranges
End-to-end delay	[20] defines acceptable End-to-end delay ranges as follows; 0 to 150ms is acceptable for most user applications, 150 to 400ms is acceptable if administrations are aware of the transmission time impact on the transmission quality of user applications. Above 400ms is unacceptable for general network planning purposes. However, it is recognised that in some exceptional cases this limit will be exceeded.
Jitter	[21] defines acceptable jitter should be between the values of 0ms and 50ms and unacceptable jitter is anything above this range.
ITUT MOS objective score	[1, 9] defines MOS user satisfaction based on • 4.3 – 5.0 = Very satisfied • 4.0 – 4.3 = satisfied • 3.6 – 4.0 = Some users satisfied • 3.1 – 3.6 = Many users dissatisfied • 2.6 – 3.1 = Nearly all users dissatisfied • 1.0 – 2.6 = Not recommended.
Packet loss	[22]defines that acceptable packet loss between 1% and 3%, and acceptable data loss between 0% and 1.5%. This experiment will use 3% as the higher threshold.

There are several popular simulators available like NS3 and NETSIM [17, 23]. However, Riverbed Modeler is an efficient simulator that provides both MOS and QoS analysis options which helps create more extensive results that better represent the performance and quality characteristics of VoIP property combinations [24]. However, Riverbed Modeler only supports up to WiFi 4 (802.11n). The newer protocol WiFi 6 (802.11ax) can handle multiple devices [25], ideal for the VoIP simulation environment.

3 Experiment design

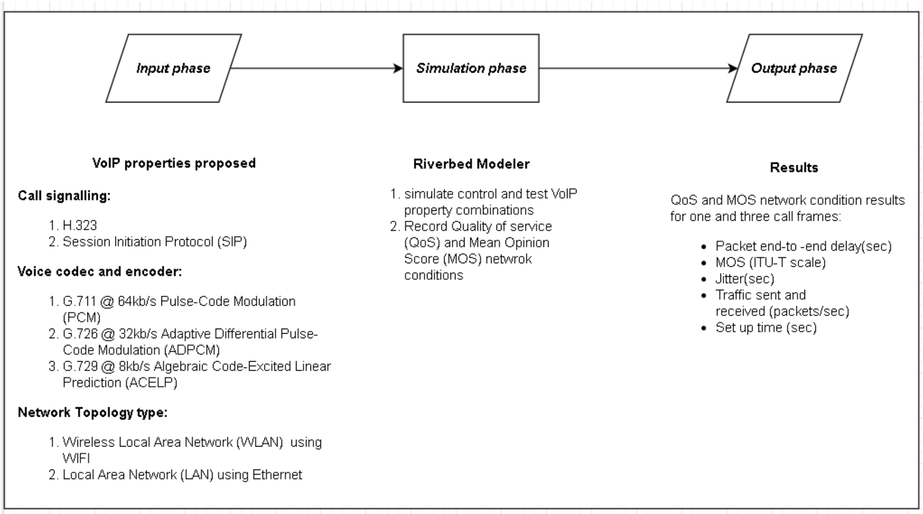


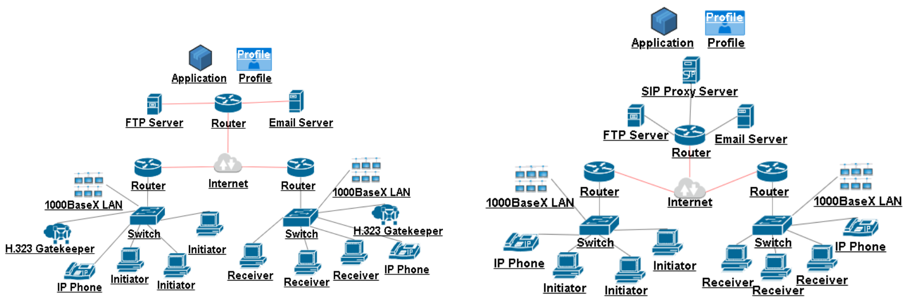
Fig 1. Experiment Design.

The experiment is designed to simulate VoIP property combinations of WLANs and LANs to determine which property combinations satisfy predefined best-quality and best- performance conditions. In section 2, the research established that; VoIP mainly uses SIP and H.323 call signalling. Different voice codecs have varying attributes suited for different use cases, such as high bit-rate quality or low bandwidth requirements. The research incorporates VoIP properties of LAN topology used in Roslin et al. [1] as the control set whilst VoIP WLAN topology is defined by this research based on most common ways that VoIP infrastructures are set up. This experiment follow Roslin at al. and record one and three frames for each combination. Test set of property combinations are populated using the use-cases. Fig 1outlines the overview of the experiment design;

The inputs to the simulation are call signalling, voice codec, encoder properties and network topology information of both WLANs and LANs, which are defined as below;

- Topologies: LAN H.323 and LAN SIP (Fig 2: (a) and (b), topology from [1]), WLAN H.323 and WLAN SIP (Fig 3: (a) and (b) – topology designed based on most common ways that VoIP infrastructures are set up
- Call signalling properties: H.323 and SIP dataset (Fig 2, dataset from [1])
- Signalling and codec types (Table 2, dataset from [1])

3.1 LAN topology for H.323 and SIP



(a) (b)

Fig 2. Signaling topology for (a) H.323 LAN and (b) SIP LAN from dataset [1]

Fig 2 shows the LAN architecture proposed by Roslin at al. [1] for H.323 and (b) SIP. This experiment uses data for the control set, and extends their work by simulating LAN topologies to collect more QoS results like set-up times and packet loss that are not recorded in [1]. Details of devices are explained in Appendix A.

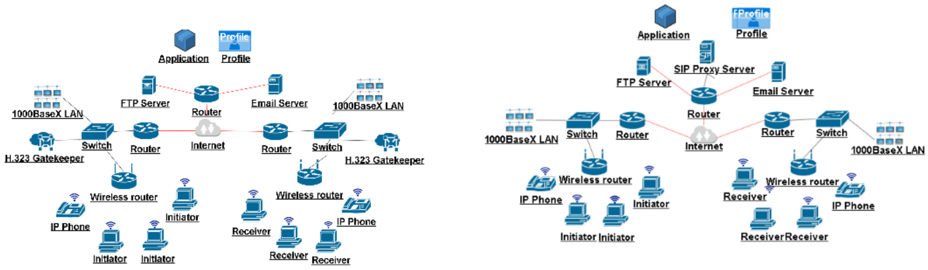
3.2 Control VoIP Properties for call signalling and codec type.

The VoIP property combinations proposed by [1] are used as the control set in this experiment. Test combinations use similar call signalling and voice codecs for both LAN and WLAN. Properties simulated are listed in Table 2.

Table 2. Control VoIP Properties for call signalling and codec type.

Signalling Type	Voice Codec Type
H.323 ITU-T [26]	G.711 @ 64Kb/s PCM Narrowband ITU-T [28]
	G.726 @ 32Kb/s ADPCM Narrowband ITU-T [29]
SIP [27]	G.729 @ 8Kb/s ACELP Narrowband ITU-T [30]

3.3 WLAN topologies for H.323 and SIP



(a) (b)

Fig 3. Signaling topologies (a)WLAN H.323 and (b) WLAN SIP for test variables.

Fig 3, has two WLAN topologies simulated and evaluated in this experiment against the control combinations. Details of devices are explained in Appendix A.

During a call, the receiver and sender will alternate roles. The simulation will not include external interferences like radio waves to help remove the bias towards isolated application scenarios. This will aid in getting a more general depiction of combinations tested. This experiment uses Roslin et al. [1] data including codec frame size, speech detection, packet loss concealment, and topologies used to construct the control variables.

3.4 Network conditions to be simulated

Network conditions are used for analysis when determining the ideal VoIP property combinations for user requirements criteria best performance and quality. Table 3 lists QoS and MOS network conditions recorded in the experiment for control and test combinations. The condition of calculations are listed in Appendix B.

Table 3. Network Condition

Network conditions
Packet end-to-end delay (sec)
Mean Opinion Score (MOS)
Jitter (sec)
Packets sent and received (packets/sec)
Set up time (sec)
Network conditions

3.5 Classification of user requirement categories

The results are classified into two categories; Best performance and best quality. The best performance is geared towards network conditions that improve the performance characteristics of VoIP and focuses on applications that are based on user functional requirements like speed. The best quality is geared towards the quality characteristics of VoIP and focuses on applications that are based on non-functional user requirements like sound quality.

The two proposed categories give a general overview of VoIP use case scenarios where the availability of quality or performance is a requirement. The categories will also act as a guide to identifying use case applications that benefit from the combinations. The classification focuses on each combination's network condition value. Table 4 outlines the classification categories.

Table 4. Classification of categories

Category	Classification
Best Performance combination	The combination that offers: · The least amount of jitter · The fastest set-up time · The least amount of packet end to end delay.
Best Quality combination	The combination that offers: · The least amount of jitter · The highest Mean Opinion Score · The least amount of traffic loss with the highest amount of traffic received

Table 5 shows the proposed use case criteria utilised to identify possible use cases; best performance and quality. In this experiment, use case applications of [1] which is similar to SOHO is used with one and three VoIP frames combinations.

Table 5. Use cases

Use case (user requirement) application	Proposed criteria
Small Office Home Office (SOHO)	• Light load services running like FTP and Email servers. • Allows for more bandwidth availability with fewer conservation concerns. • The availability of bandwidth enables higher quality requirements for user satisfaction.

Industrial and Commercial	• Simultaneous VoIP call handling capabilities. • Low VoIP impact for bandwidth conservation. • Ability to operate VoIP over large amounts of service data traffic like high load FTP and Email servers.
Public and private services	• Service availability and performance is mission-critical • Communication quality is ideal

In the simulation environment, devices are configured and connected to form the network topologies WLAN and LAN in which the VoIP property combinations are tested through. Background traffic for FTP and email server simulates a network load when VoIP is used [1]. The devices used in the simulation environment for control and test combinations create the topologies outlined in Fig 2 (a) and (b), Fig 3 (a) and (b). These device nodes are listed in Appendix A.

3.6 Simulation and user requirement classification

The simulation is iterative and process both control and test property combinations to obtain QoS and MOS results. The Discrete Event Simulation (DES) method is used to create the effect of real-world system in the simulated environment. QoS and MOS results are generated at the end of the DES events. Then, the network condition results like end-to-end delay, jitter and packet loss are evaluated to identify if they are in acceptable recommended ranges outlined in Table 1 . The results are also evaluated to classify the ideal combinations for the quality and performance categories. The voice attributes are configured in the simulation environment. Each combinations details are implemented as seen in Table 6.

Table 6. Simulation conditions

Attribute	Value
Silence Length (sec)	Exponential (0.65) (default)
Talk Spurt Length (sec)	Exponential (0.352) (default)
Encoder Schemes	G.711 64kb/s PCM, G.726 32kb/s ADPCM, G.729 8kb/s CS-ACELP
Voice Frames per Packet	1 and 3
Signalling	H.323 and SIP
Type of Service	Interactive Voice
Frame Size/duration	5ms, 10ms

Voice Activity Detection	Comfort Noise Generation (CNG), No

Once the simulation is completed, the resulted network conditions are sent to the classifier. Classifier analyses the network conditions and classify them based on the best combination for user requirement; quality and performance. The classification of categories are explained in Table 4. Pandas in Python is used to automate the sorting of network condition results. The process uses ‘Max’ and ‘Min’ to sort the values in line with optimal network conditions according to the predefined classification criteria.

4 Results and discussion

4.1 Simulation output

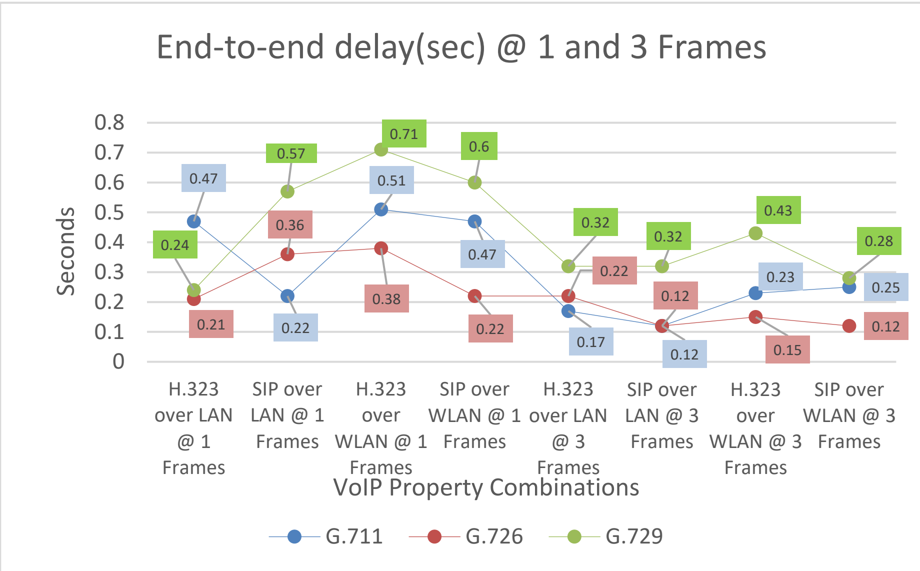


Fig 4. Packet end-to-end delay(sec) 1 and 3 Frames

Fig 4 shows the packet end to end delay(sec) recorded for property combinations. Overall, the highest end-to-end delay 0.71 seconds is shown in G.729 when using H.323 over WLAN at 1 frame. This is considered “unacceptable” by the network condition recommended ranges (Table 1). End-to-end delay ranged from 0.12 to 0.43 seconds shown for combinations using three frames. They performed considerably better than one frame combinations that ranged from 0.21 to 0.71 seconds showcasing that frame size does improve delay. [3] finds that an increase in calls negatively affected results when simulating VoIP over WLAN infrastructure. [18, 19] also mention that heavy traffic with multiple concurrent communication sessions can heavily increase packet loss rate. This experiment has a total of eight VoIP devices for each combination tested in the simulation environment. These devices run alongside other FTP and email services which may have caused high end-to-end delay seen in fig 4.

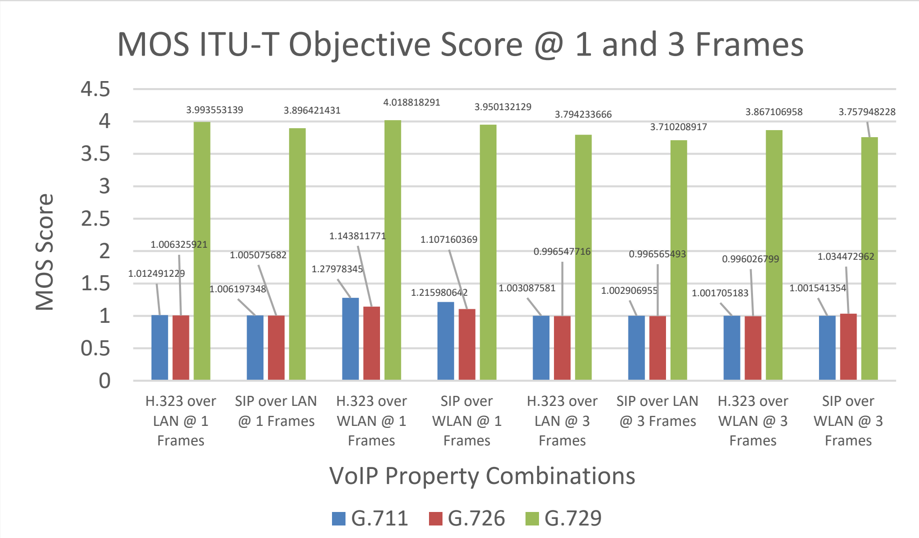


Fig 5. Voice MOS Value (ITU-T) 1 and 3 Frames

Fig 5 shows the MOS (ITU_T score). The MOS score is outlined in Table 1. Across all combinations simulated, the codec G.729 shows the highest overall MOS score results around 4 “satisfied” when compared to G.711 and G.726 which average around 1 “not recommended”. Overall, the codec G.711 generally performed marginally better than G.726 but still in the MOS range of “not recommended”.

This experiment findings contradict Roslin et al. [1]’s MOS results as the codecs G.711 and G.726 performed poorly, averaging around 1 “not recommended“ in this experiment when compared to “satisfied” and “nearly all users dissatisfied” in Roslin et al. results. However, following results agree with Roslin et al. [1] : the lower bit rate codec G.729 at 8kb/s had a similar MOS score averaging 3.81 “some users satisfied” in their results and 3.87 in this experiment also resides in the same range. This could be due to the network topology’s inability to withstand higher demand codecs like G.711 at 64kb/s and G.726 at 32kb/s whilst supporting running other services. This also could explain the low MOS scores displayed by the higher bit rate codecs. Alternatively, it is also possible that unmentioned simulated factors and/or software version could have caused inconsistencies. The academic edition used in this study limits the number of simulated events, Roslin et al. [1]’s work has no mention of specific version used.

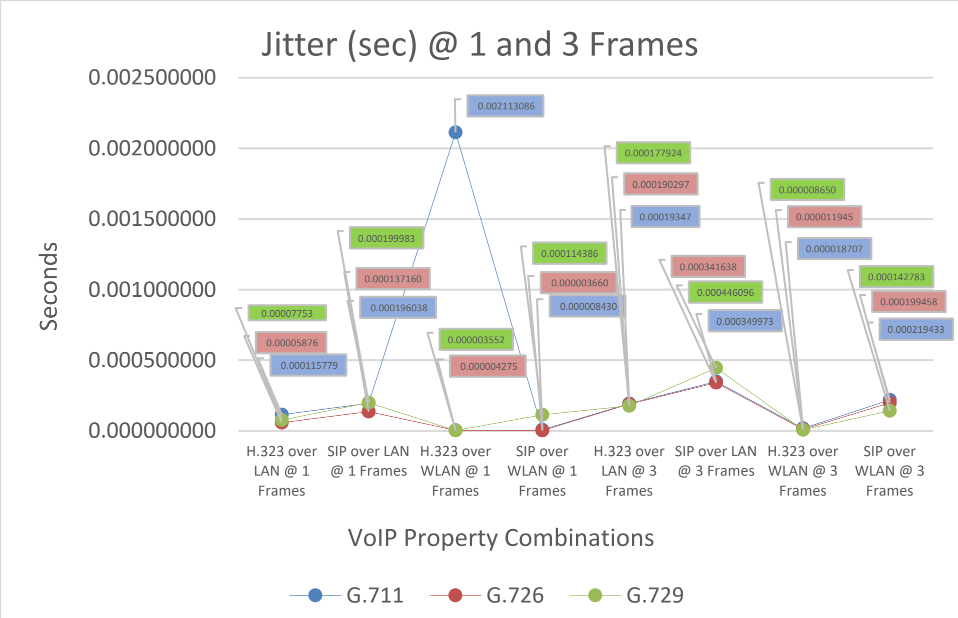


Fig 6. Jitter(sec) 1 and 3 Frames

Fig 6 shows the Jitter(sec) results. Acceptable jitter should range between 0ms and 50ms as outlined in Table 1 . All combinations showcased jitter within the acceptable margins with H.323 over WLAN using codec G.711 at 1 frame showing the highest jitter at 2.113086 ms and WLAN H.323 G.729 at one frame having the lowest jitter at 0.003552 ms. Roslin et al. [1]'s work in LANs recorded different jitter levels compared to this experiment with jitter results ranging from 0.00040 sec to 0.00000 sec for G.729 over SIP at 3 frames. Overall, both experiment results were within the acceptable range outlined in Table 1. Differences between combinations may be indistinguishable, but further research can be carried out to confirm.

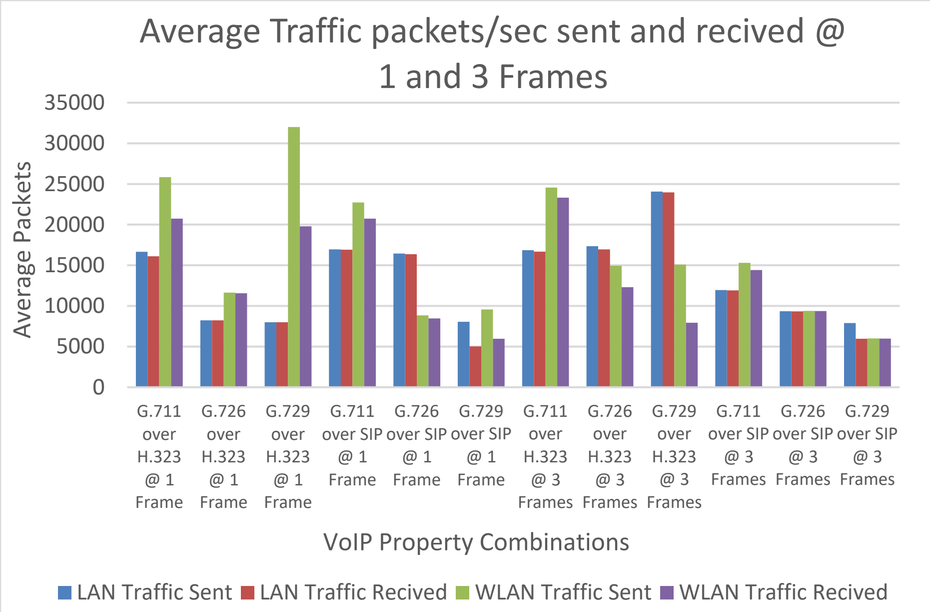


Fig 7. Traffic sent (packets/sec) 1 and 3 Frames

Fig 7 shows traffic sent and received (packets/sec). Acceptable data loss calculated from the difference between traffic sent and traffic received should be between 0% and 3% as outlined in Table 1. WLAN combinations show higher amounts of traffic loss when compared to LAN combinations. For example WLAN using G.729 over H.323 at 1 frame had 30021.7 packets lost. Combinations like G.729 over SIP at 1 frame had around 5000 to 10000 packets sent and received in comparison to combinations like G.711 over H.323 at 1 frame which have an average of 15000 to 25000 packets sent and received. The reduction in traffic sent and received could indicate some network congestion.

For data sent and received, the expectation was that G.711 operating at a bandwidth of 64kbps would consistently send more data than the codecs G.726 with a bandwidth of 32kbps and G.729 at 8kbps data rate. Results indicated that while different bit rate codecs are simulated, some had very similar amounts of data sent. For example, H.323 operating over WLAN using codecs G.726, G.729 and SIP over WLAN using the codec G.711 sent around 8000 to 10000 packets. It is possible that the higher bit rate codecs caused bottlenecks when queuing during the transmission of packets from sender to receiver leading to a reduction in packets sent. This is apparent by results like LAN, H.323 and G.729 at three frames compared to LAN H.323 G.711 at three frames. Newer types of WiFi protocols like WiFi 6 IEEE 802.11ax could help mitigate the amount of data loss. WiFi 6 is the new version of WLAN that is reliable and can support multiple devices, making it ideal for the simulated environments throughput requirements [25].

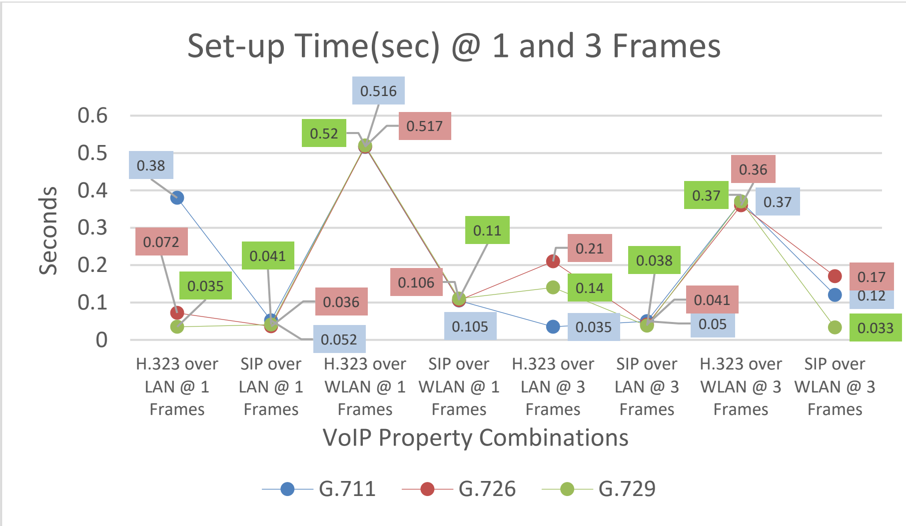


Fig 8. Set-up Time (sec) 1 and 3 Frames.

Fig 8 shows network setup times of frames. Overall combinations with 3 frames has a set-up time ranges from 0.033 sec to 0.37 sec which performed faster than 1 frame variants that ranged from 0.035 sec to 0.52 sec. Combinations using H.323 over WLAN, both tested frames had the longest set up times when compared to other combinations, averaging around 0.518 sec. Codecs using SIP offered faster set-up times ranging from 0.003 sec to 0.17 sec compared to H.323 combinations ranging from 0.035 sec to 0.52 sec. Similarly [2] recorded that SIP had the most optimal set up time, suggesting that SIP is ideal for set up time depending on requirements. They proposed that this could be due to the fewer messages exchanged at the establishment of the session as opposed to H.323.

4.2 Classification of simulation output

After collecting the QoS and MOS results as illustrated in Fig 4-8, the simulation output of the property combinations are classified into one of the user requirement class; best quality or performance combination.

Table 7 shows the best performance combination for each network condition when considering the criteria listed in Table 4. The classification output shows that the, SIP over WLAN using G.726 at one frame is the ideal performance combination for the best performance. It has acceptable end to end delay, quick set up time and low jitter. The combination offers network condition mitigation and performance characteristics best suited for use in industrial and commercial applications where performance is the key to ensuring service availability. This may not be a major concern in a SOHO environment where bandwidth is less conserved. Industrial applications can have many services running concurrently, including multiple VoIP call handling. This can be mission-critical, so characteristics like fast set-up time and lower packet end-to-end delay are essential. The research assume fast set up time is desirable but not essential in a SOHO environment. Network congestion is more likely in industrial settings, making VoIP jitter and end-to-end delay mitigation essential alongside bandwidth conservation which this combination can offer. These characteristics are less of a requirement in SOHO setups but are desirable, especially when implementing redundancy and future network expansion capabilities.

Table 7. Best performance combinations.

Condition	The best combination for condition	Condition
Lowest Jitter (Fig 6)	WLAN H.323 G.729 @ 1 Frame	Lowest Jitter (Fig 6)
Quickest Set-up time (Fig 8)	WLAN SIP G.729 @ 3 Frames	Quickest Set-up time (Fig 8)
Lowest End-to-end delay (Fig 4)	LAN SIP G.726 and WLAN SIP G.726 @ 3 Frames	Lowest End-to-end delay (Fig 4)
Overall best performance combination	SIP over WLAN using G.726 at one frame	Overall best performance combination

Table 8 shows the best quality combination for each network condition when considering the criteria listed in Table 4. The classification output shows that H.323 over WLAN using G.729 at one frames was the ideal quality combination. It offered low jitter, “fair” to “good” MOS score, high traffic throughput, and low traffic loss. This combination is best suited for consumer or small business applications where user satisfaction and non-functional requirements like quality are preferable. This could include SOHO environments where bandwidth is available, and quality is the main requirement. Applications that would benefit from this combination are unlikely to support high load services or many simultaneous VoIP calls meaning bandwidth conservation is not a priority over quality. The combination had higher traffic sent when compared to some of the other combinations but suffered from a considerable amount of traffic loss, but a simplified SOHO network set-up could solve this, for example, lower load server applications. A high MOS indicates better sound quality characteristics. Minimising jitter is also an essential quality requirement for increasing call quality and clarity. It is vital to note that some industrial applications could have quality requirements that outweigh performance requirements. In this case, the infrastructure needed for VoIP to deliver the best quality characteristics needs to be very robust, primarily if other services like high load traffic such as FTP or email servers use the same available bandwidth. Quality can also be a mission-critical requirement. Some applications may rely on call quality as a functional requirement, for example, communication of sensitive data over VoIP in specialised services like military or emergency service scenarios.

Table 8. Best quality combinations

Condition	The best combination for condition	Condition
Lowest Jitter (Fig 6)	WLAN H.323 G.729 @ 1 Frame	Lowest Jitter (Fig 6)
Highest MOS (Fig 5)	WLAN H.323 G.729 @ 1 Frame	Highest MOS (Fig 5)
The least amount of traffic loss with the highest amount of traffic received (Fig 7)	LAN H.323 G.729 @ 3 Frame	The least amount of traffic loss with the highest amount of traffic received (Fig 7)
Overall best quality combination	H.323 over WLAN using G.729 at one frames	Overall best quality combination

Further analysis revealed that using objective QoS and MOS measurement method adequately captured each combination's performance and quality characteristics. However, it could have also been beneficial to get a subjective measurement as human perception of VoIP performance as then quality would be more realistic. The differences in results between the combinations could be indistinguishable through a subjective test. The use case categorisation technique worked well to represent a general set VoIP of use cases but could be improved and expanded. There are many use cases applicable to VoIP, making it challenging to simulate and classify all of them.

Overall results do not conclusively highlight a superior best performance or best quality combination that fulfils all the classification criteria proposed for both conditions. Instead, classification is based on the interpretation of results to classify the most ideal combination for each criteria. This was not ideal but functioned well at replicating a real-world selection scenario.

Lastly, the experimental process ran well. Riverbed modeler had the correct tools and performed as expected to carry out this project which agrees with [1, 24]. The simulation environment functioned well when implementing the devices into the topologies LAN and WLAN. The network condition data selection and collection process simplified analysis tasks particularly when exporting the MOS and QoS results for automated sorting.

5 Significant findings and future work

This paper enhanced the study of Roslin et al. [1] work on LAN, extending it to WLAN topologies. The reconstruction of [1]'s work on MOS for G.729 over LAN SIP produced similar results however, the QoS results differed. Codecs using SIP over LAN offered the fastest set-up times, which agrees with the study of Khiat et al. [2]. This suggest that SIP is ideal for set up time requirements instead of H.323. This experiment also further examined Guy et al. [3] findings to understand how and why WLAN topologies introduce undesirable network conditions and concluded that packet loss and reduced throughput could contribute to the network conditions found in WLAN combinations. This study found VoIP property combinations that offered ideal characteristics to match the best performance and quality user requirements.

This experiment found characteristics of VoIP property combinations that are ideal for the proposed use cases. Results indicated that VoIP is versatile and can suit many applications, including SOHO, industrial and emergency services, which could be mission-critical. SOHO environments could benefit from a hybrid blend of multiple network types and a property combination focused on performance and quality requirements. The overall experiment findings support VoIP as a popular telephony method for its versatility and ease of configuration. Moreover, both SOHO and industrial user requirements met availability, performance and quality expectations. This may contribute to the broader adoption of VoIP as the primary mode of telephony communication.

The simulation was limited to WiFi 4 in the simulation model . This study can be enhanced by implementing WiFi 6 in WLAN scenarios which could help mitigate the high end-to-end delay and data loss amounts. It can also support more significant amounts of client end devices than WiFi 4 or WiFi 5. This study was non-human based and therefore, objective MOS was used. This experiment can be further enhanced by subjective MOS experiment where the end-user can give more realistic measurements of VoIP voice quality. Furthermore, using both subjective and objective measurements could help find the optimal VoIP property combinations to match the user requirements better than just objective measurement.

Further testing and development would be beneficial as VoIP has an extensive range of property combinations and configurations that could offer more desirable characteristics for SOHO and other use case environments. To better meet user requirements especially in adaptive environments where network configurations need to be adaptable. An AI or machine learning module could be utilised to analyse network conditions and alter VoIP property combinations in real-time to ensure user requirements. This can help mitigate the bottlenecks and traffic loss highlighted by the results to help maintain functional or non-functional requirements, especially in mission-critical applications. It is also possible that independent basic service set (IDSS) or Ad-Hoc networks could mitigate WLAN traffic loss.

6 Conclusions

This experimental research successfully found optimal VoIP property combinations that showcased ideal QoS and MOS characteristics for user requirements; best performance and quality. By expanding on existing research, good comparisons between LAN and WLAN connectivity types for VoIP in the simulated environment were highlighted. The proposed mission-critical use cases helped visualise the best applications for VoIP when achieving the defined user requirements. Both SOHO and industrial applications showed promising results on availability, performance and quality for VoIP deployment. However, user requirements can be better achieved by implementation of newer WiFi technology and further property testing, which can solidify VoIP's capabilities to function for mission critical environments. VoIP qualities like availability, versatility and reliability were apparent throughout the experimental processes which support existing research for VoIP's future.

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Data Availability Statement: In this study a publicly available dataset from a published peer-reviewed paper [1] was used to construct the control test VoIP properties. This data set can be found

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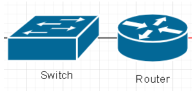
Conflicts of Interest: The authors declare no conflict of interest.

7 Appendix

7.1 Appendix A

Appendix A. Simulation device legend

Node icon	Node name	Description
	Ethernet workstation (IEEE 802.3)	IEEE 802.3 ethernet standard workstation running VoIP profile. Uses SIP User Agent Client (UAC) and H.323 Gatekeeper routed signalling (GKRCS)

	WLAN Workstation (IEEE 802.11g)	IEEE 802.11g workstation running VoIP profile. Uses SIP User Agent Client (UAC) and H.323 Gatekeeper routed signalling (GKRCS). Set to non-roaming.
	IP phone and Wireless IP phone	Client devices for VoIP telephony. IP phone used for an ethernet connection and wireless phone used for the WIFI connection. It also uses SIP UAC or H.323 (GKRCS)
	Ethernet Router and Ethernet switch (IEEE 802.3)	This router supports PPP to IP cloud and ethernet links to end devices and services.
	WLAN Ethernet Router (IEEE 802.11)	IEEE 802.11 with Basic Service Set (BSS) identifier allows wireless end devices to communicate to the correct wireless router. In addition, it supports one ethernet link to connected devices.
	H.323 Gatekeeper	H.323 management tool to facilitate communication between clients. The gateway is set to gatekeeper routed signalling to accomplish this.
	SIP Proxy Server	Proxy is used to facilitate communication between addresses or nodes. This enables the devices to talk to each other after the proxy initiates the communication channel.

7.2 Appendix B

Appendix B. Network conditions and condition information.

Network Condition	Condition calculation

Packet end-to-end delay (sec)	delay = network_delay + encoding_delay + decoding_delay + compression_delay + decompression_delay + dejitter_buffer_delay
Mean Opinion Score (MOS)	Global statistic captures the minimum MOS value collected in the network.
Jitter (sec)	If two consecutive packets leave the source node with time stamps t1 & t2 and are played back at the destination node at time t3 & t4, then: jitter = (t4 - t3) - (t2 - t1) Negative jitter indicates that the time difference between the packets at the destination node was less than that at the source node.
Packets sent and received (packets/sec)	Traffic sent: Average number of packets per second submitted to the transport layers by all voice applications in the network. Traffic received: Average number of packets per second forwarded to all Voice applications by the transport layers in the network.
Set up time (sec)	H.323: This statistic holds the average set-up time (in seconds) for a H.323 call in the network. SIP: Time to set-up a call.

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