Comparative Analysis of Voice Quality on *iLBC* and *Speex* Codecs Server *PBX* with Voice Comparison Method

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Abstract— VoIP servers provide several types of codecs including Internet Low Bit Rate Codec (iLBC) and Speex, each type of codec has a different capacity and quality. So that in the process of choosing a codec implementation, VoIP it becomes one of the things that affect the quality of communication. The iLBC and Speex codecs are intended for high quality communication but at bit rate. By comparing iLBC and Speex communication, VoIP hoped that the better performance between the two codecs will be known by conducting a comparative analysis of VoIP based on Quality of Service (QoS) and analyzing the audio results of VoIP for voice quality analysis using the voice comparison method. using Matlab. Based on the results of the research between the iLBC and Speex parameter values QoS include delay, jitter, and packet loss. Speex Codec is smaller than the iLBC codec of 34.45 ms. The jitter of the iLBC codec is smaller than the speex codec of 0.00748 ms. The iLBC codec packet loss is smaller than the speex codec by 7.27%. Meanwhile, based on sound quality testing in MATLAB, it was found that the iLBC codec has an average value of delta amplitude lower than the speex codec with a value of 6.7E-06 volts in the AH Building, 3.6E-06 volts in the AI Building, and 1.8E-06 volts between the AH-AI Buildings.

Keywords — VoIP, iLBC Codec, Speex Codec, QoS, Voice Comparison

I. INTRODUCTION

The development of internet technology is advancing rapidly, resulting in new technologies that help human life in terms of communication. One of them is voice telephone technology that uses the internet network, this technology is known as Voice over Internet Protocol (VoIP). VoIP is a voice communication technology based on Internet Protocol (IP), where clients can perform voice communication only by using the internet network [1][2]. Voice data will be converted into digital code and streamed through the internet network by sending data packets, so that the data transmission process does not go through ordinary telephone circuits [3][4].

voIP technology has begun to be widely developed on the Raspberry Pi, because compared to computers, Raspberry Pi has the advantage of being small in size so it is easy to carry and has maximum power [2]. This is a major consideration in choosing a server for VoIP, because the server plays a role in managing conversational traffic so that it does not overlap each other and performs important tasks such as signaling, codecs, connect/disconnect, and databases – end [5].

Basically, codec is an algorithm that converts voice signals into data. The codec aims to reduce bandwidth usage in signal transmission on each call and at the same time to increase the number of calls [3][6]. VoIP servers provide several types of codecs, each type of codec has a different capacity and quality. So that in the process of choosing a codec implementation, VoIP becomes one of the things that affect the quality of communication [7].

Codec is short for compression/decompression, converts signal and compresses it into digital data form to be retransmitted and then returned to the form of audio signal such as data that is sent so that the codec plays an important role in the VoIP [4][8]. Codecs make changes by sampling the audio signal. Internet Low Bitrate Codec (iLBC) and speex codec using lossy compression techniques and low bitrate narrow band audio codecs [9][10].

Based on the description above, this study will compare the performance codec iLBC and Speex on the server FreePBX uses on a softphone. It is hoped that by comparing the two codecs using softphones, it is possible to know better performance between the codecs audio iLBC and Speech Communication VoIP in accordance with the Quality of Service (QoS) and analyzing the results of the audio signal generated using Matlab.

II. METHOD

A. Research Design

Stages of the research to be carried out are stated by the flow chart shown in Fig. 1.

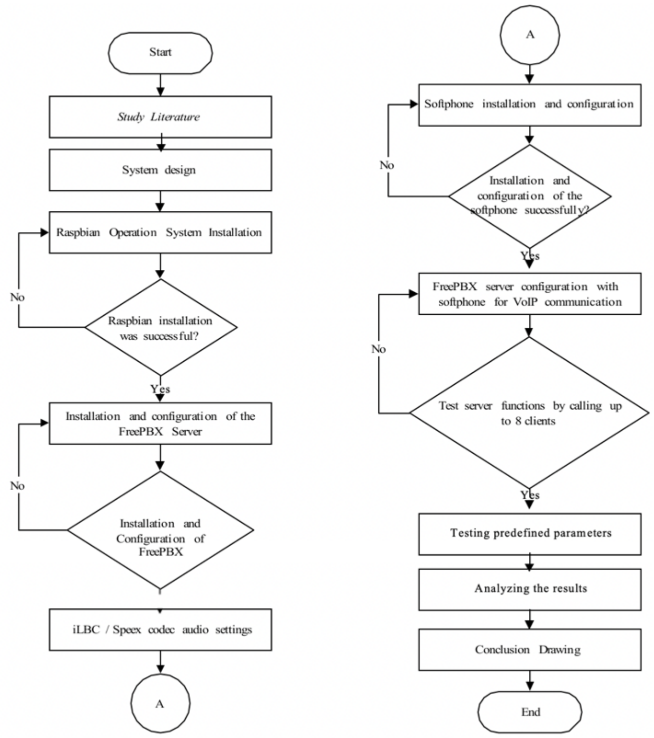

Figure 1. Flowchart of Research Stages

The explanation of the research design flowchart is as follows:

1. Making this system begins with a literature, namely studying previous research related to system design to be used and regarding the iLBC and Speex communications VoIP
2. The second stage is system planning. At this stage, the design of the test scheme is carried out using the local network Router TP Link which is placed in the Telecommunications Network Lab in the AI Building of the State Polytechnic of Malang.
3. The third stage is to install the Raspbian Operating System (OS) using the Python programming language. If the installation and configuration of the Raspbian OS fails, then do the installation and reconfigure.
4. The fourth stage is to install and configure FreePBX 16 as a Voice over Internet Protocol (VoIP) server. If the installation and configuration of FreePBX 16 fails, then do the installation and reconfigure.
5. The fifth stage is to configure the iLBC and Speex on FreePBX 16 as a VoIP server.
6. The sixth stage is to install and configure the softphone on Android communication VoIP If the softphone fails then do the installation and reconfigure.
7. The seventh stage is to configure FreePBX 16 as a VoIP server with softphone communication VoIP
8. The eighth stage is testing by conducting VoIP to 8 clients or 4 pairs of clients, after that the test data is obtained using the iLBC and Speex alternately on the softphone.
9. The ninth stage is to analyze the quality of the sound produced when communicating VoIP with the iLBC and Speex with the voice comparison method in Matlab using frequency and amplitude parameters. As well as using the parameters of QoS delay, jitter, and packet loss.
10. The last stage is drawing conclusions from the research.

B. System

Design The research design to be carried out is stated by the block diagram shown in Fig. 2 below.

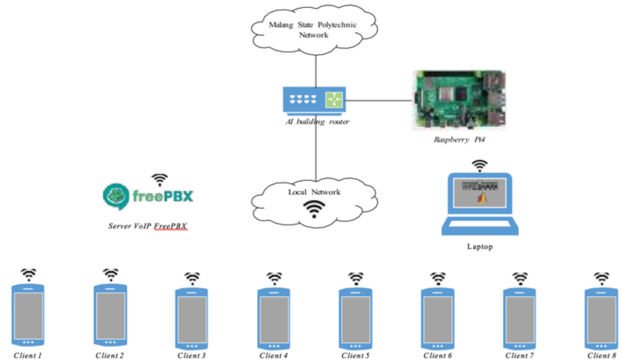

Figure 2. System block diagram

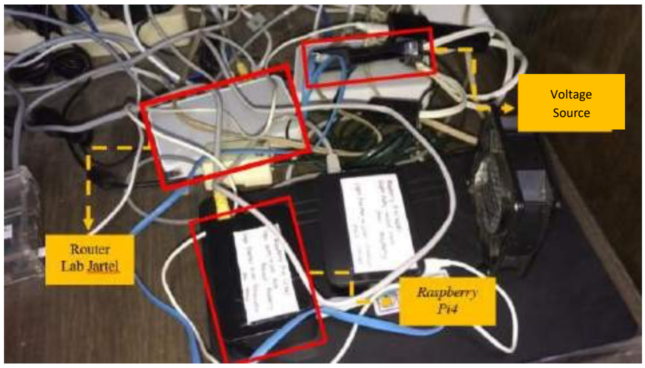

Figure 3. System Design Implementation

In the system block diagram shown in Fig. 2 above, it is explained about the flow carried out during the research, namely the Raspberry Pi4 as a VoIP server placed on the Malang State Polytechnic AI Building. On the client, 8 smartphones that have been configured with Linphone and Zoiper 1.36 softphones are used. In addition, a laptop is used to perform an analysis of the parameters that have been determined.

C. Matlab Program Design

The audio processing design to compare sound quality in MATLAB is stated by the flow chart shown in Fig. 4.

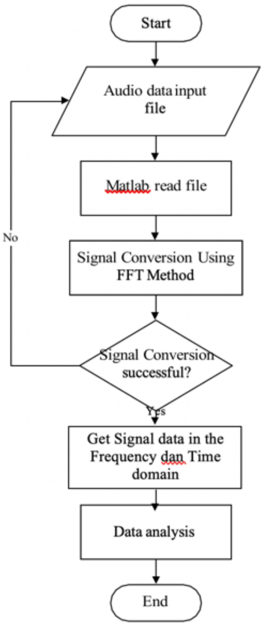

Figure 4. Audio Processing Design Flowchart

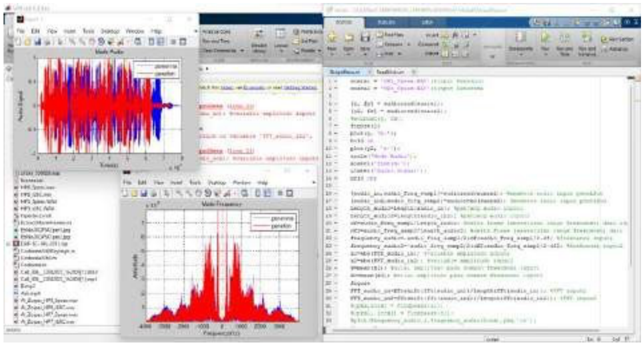

Figure 5. Simulation of sound quality comparison using matlab by way of sound comparison

In the audio processing design flow diagram shown in Fig. 4 above, it is explained about the flow carried out to perform sound comparisons on matlab. The sound processing process is carried out by inputting data from the sound record results from the test then running matlab to run the program, the input will be read by matlab and the input will be displayed in the time and frequency domains [11]. The analysis is done by comparing the audio of the caller and the receiver and then comparing the amplitude between the caller and the receiver from the VoIP.

D. *Test Parameters*

Voice quality parameters in VoIP communication that are measured based on the voice comparison method in Matlab are frequency and amplitude, based on the data obtained, which are recorded in the table. Meanwhile, based on QoS, namely delay, jitter, and packet loss, then the data obtained is recorded in a table for analysis and conclusions can be drawn.

III. RESULTS AND DISCUSSION

A. *Routing VoIP Call*

Routing is a routing process for sending packets and information from one network to another via the internet [12]. VoIP call routing in this study was carried out between the server installed in the AI building and calls made from the AH Polinema building. Before making a call, the routing setting process is carried out on the FreePBX server.

Figure 6. Routing Settings on the server

The figure shows that the routing settings are carried out on the FreePBX server by adding several IP Local Networks at the Malang State Polytechnic. The number of IPs added is 7 IP addresses including 192.168.130.0/23; 192.168.131.0/24; 192.168.181.0/24; 192.168.182.0/23; 172.16.18.0/24; 172.16.13.0/24; and 192.168.184.0/23.

B. *Testing VoIP Calls*

Calls VoIP using the iLBC softphone linphone codec can be seen in Fig.7 (a) and (b) for the Speex codec. Meanwhile, when using the Zoiper 1.36 iLBC codec can be seen in Fig. 8 and Fig. 9 for the Speex codec.

(a) (b)

Figure 7. (a) Test of VoIP codec iLBC using softphone linphone; (b) Test Call VoIP codec speex using softphone

Figure 8. Call Test iLBC Codec VoIP using Zoiper 1.36

Figure 9. iLBC Codec VoIP using Zoiper Softphone 1.36

C. Discussion

Testing of VoIP codec iLBC and speex calls was conducted in the AH Building, AI, and between the AH-AI Buildings using 4 pairs of mobile phones with 2 types of softphones, namely Linphone and Zoiper 1.36. The client pairs used are 1 with 2, 3 with 4, 5 with 6, and 7 with 8. Where clients 1, 3, 5 and 7 are the callers while clients 2, 4, 6 and 8 are the recipients. The following is an analysis of the results of testing VoIP calls from each parameter that has been determined.

1) Analysis of Delay Parameters

Table 1 until 3 show the delay obtained by the iLBC codec 29.69 – 144.06 ms, while the speex codec is 20.27 – 52.86 ms. This shows that the data has a very good standard based on TIPHON because it has a large delay of less than 150 ms [13]. Testing in the AH building, the average value of delay for the iLBC codec is 36.07 ms, while the speex codec is 37.81 ms, so the delay for the iLBC codec is smaller than the speex codec. This shows that testing in the AH codec iLBC building is neater in sending data than the speex codec. For testing in the AI building, the average delay value for the iLBC codec is 41.34 ms, while the speex codec is 24.42 ms so that the delay codec speex is smaller than the iLBC codec.

TABLE 1
BUILDING AH DELAY TEST RESULT

DELAY (ms)				
Client	Linphone		Zoiper 1.36	
to-n	iLBC	Speex	iLBC	Speex
1	61.69	29.8	29.69	49.25
2	60.79	43	29.78	49.64
3	30.88	44.79	29.74	29.63
4	33.02	41.47	29.79	38.52
5	33.02	47.73	29.81	30.34
6	32.39	24.04	30.06	41.31
7	50.18	30.73	30.24	41.45
8	35.94	25	30.23	38.33

This shows that testing in the AI codec speex building is neater in data transmission than the iLBC codec. For testing between buildings, AH-AI has an average delay value of 41.63 ms for the iLBC codec while the speex codec is 41.12 ms so that the delay codec speex is smaller than the iLBC codec. This shows that testing in the AI codec speex building is neater in data transmission than the iLBC codec. Based on the delay parameter testing, the highest average delay value is 41.63 ms in the iLBC codec testing between AH- AI buildings, while the lowest is 24.42 ms in the AI building speex codec testing. This is because VoIP calls pass through several routers at the Malang State Polytechnic including 192.168.130.1, 172.16.13.254, 172.16.18.18, and 192.168.181.250. Meanwhile, calls in the AI Building are directly directed to the server IP 192.168.181.250.

TABLE 2
BUILDING AI DELAY TEST RESULT

DELAY (ms)				
Client	Linphone		Zoiper 1.36	
to-n	iLBC	Speex	iLBC	Speex
1	35.65	20.28	30.34	25.01
2	35.66	20.27	30.24	25.07
3	33.17	24.02	31.21	20.94
4	33.16	23.99	31.33	20.96
5	34.9	36.72	31.39	21.46
6	36.12	34.67	31.39	21.36
7	144.06	20.81	30.45	26.26
8	62.23	20.74	30.08	28.11

TABLE 3
BETWEEN BUILDING AH - AI DELAY TEST RESULT

DELAY (ms)				
Client	Linphone		Zoiper 1.36	
to-n	iLBC	Speex	iLBC	Speex
1	63.52	38.74	31.89	52.34
2	62.84	43.81	31.97	52.86
3	34.67	45.13	32.19	38.38
4	36.39	41.36	32.58	39.43
5	36.66	47.96	33.65	37.26
6	37.68	30.39	33.37	44.71
7	70.54	32.75	32.55	41.47
8	62.98	30.34	32.62	40.95

2) Analysis of Jitter Parameters

Table 4 until 6 show the jitter obtained iLBC codec 4.12E-06 – 0.14435 ms, while the speex codec is 0.00106 – 0.03485 ms.

TABLE 4
BUILDING AH JITTER TEST RESULT

JITTER (ms)				
Client	Linphone		Zoiper 1.36	
to-n	iLBC	Speex	iLBC	Speex
1	0.14435	0.00924	4.05E-05	0.00201
2	0.02427	0.00862	0.00196	0.00843
3	0.06344	0.00255	0.00504	0.00629
4	0.00255	0.00349	4.97E-06	0.0051
5	0.00053	0.00513	6.28E-06	0.00169
6	0.0053	0.00385	0.00232	0.00114
7	0.0064	0.00399	0.00324	0.00106
8	0.00269	0.00979	4.12E-06	0.00539

This shows that the data has a good standard based on TIPHON because it has a jitter 0 to 75 ms [14]. Testing in the AH building the average value of the iLBC codec jitter is 0.01638 ms, while the speex codec is 0.00486 ms so that the speex codec jitter is smaller than the iLBC codec. This shows that when testing the speex codec, the network conditions used are quite stable compared to when testing the iLBC codec. For testing in the AI building, the average value of the iLBC codec jitter is 0.00354 ms, while the speex codec is 0.01392 ms so that the iLBC codec jitter is smaller than the speex codec.

TABLE 5
BUILDING AI JITTER TEST RESULT

JITTER (ms)				
Client	Linphone		Zoiper 1.36	
to-n	iLBC	Speex	iLBC	Speex
1	0.0019	0.01424	0.00038	0.00502
2	0.0019	0.0142	0.00038	0.00502
3	0.00167	0.00392	1.26E-05	0.01467
4	0.00168	0.00391	7.37E-06	0.01468
5	0.00243	0.03485	0.01767	0.01027
6	0.00243	0.03481	0.01767	0.0103
7	0.00386	0.00317	0.00038	0.02519
8	0.00386	0.00317	0.00038	0.02526

TABLE 6
BETWEEN BUILDING AH - AI JITTER TEST RESULT

JITTER (ms)				
Client	Linphone Zoiper 1.36			
to-n	iLBC	Speex	iLBC	Speex
1	0.00226	0.00893	0.00248	0.00782
2	0.00255	0.00438	0.00281	0.00825
3	0.00285	0.00957	0.00281	0.00535
4	0.00298	0.00684	0.00295	0.00783
5	0.00127	0.00765	0.00288	0.00852
6	0.00283	0.00368	0.00181	0.00725
7	0.00228	0.00321	0.00293	0.00686
8	0.00323	0.00667	0.00126	0.00775

This shows that when testing the iLBC codec, the network conditions used are quite stable compared to when testing the speex codec [15]. For testing between AH-AI buildings, the average value of the iLBC codec jitter is 0.00251 ms, while the speex codec is 0.00691 ms, so the iLBC codec jitter is smaller than the speex codec. This shows that when testing the iLBC codec, the network conditions used are quite stable compared to when testing the speex codec.

3) Analysis of Packet Loss Parameters

Table 7 until 9 show the packet loss obtained by the iLBC codec 1.68% - 10.06%, while the speex codec 2.96% - 10.9%. This shows that the data has a very good and good standard because it has a packet loss 0% - 2% and 3% - 14% [16]. Testing in the AH building, the average value of the iLBC codec packet loss is 6.16%, while the speex codec is 7.1% so that the iLBC codec packet loss is smaller than the speex codec.

TABLE 7
BUILDING AH PACKET LOSS TEST RESULT

PACKET LOSS (%)				
Client	Linphone		Zoiper 1.36	
to-n	iLBC	Speex	iLBC	Speex
1	3.26	2.96	5.8	7.52
2	3.26	3.3	6.42	7.62
3	7.28	8.26	6.42	7.3
4	8.24	8.66	6.38	7.1
5	6.62	7.58	5.74	7.68
6	6.76	7.08	6.58	7.94
7	6.08	7.8	6.64	7.34
8	6.04	7.5	7	7.98

This shows that the test in the AH codec iLBC is more stable in sending data packets because fewer packets are lost or unread, so calls using the iLBC codec are smoother than when using the speex codec. For testing in the AI building, the average value of packet loss for the iLBC codec is 6.2%, while the speex codec is 5.93% so that the packet loss of the speex codec is smaller than the iLBC codec.

TABLE 8
BUILDING AI PACKET LOSS TEST RESULT

PACKET LOSS (%)				
Client	Linphone		Zoiper 1.36	
to-n	iLBC	Speex	iLBC	Speex
1	7	6.28	7.14	6.38
2	6.88	7.18	6.9	6.66
3	8.1	5.22	6.38	6.94
4	7.94	5.54	6.08	7.58
5	7.36	3.34	6.34	6.98
6	7.74	3.28	5.52	6.1
7	1.68	6.14	6.3	5.26
8	1.96	6.46	5.9	5.54

TABLE 9
BETWEEN BUILDING AH - AI PACKET LOSS TEST RESULT

PACKET LOSS (%)				
Client	Linphone Zoiper 1.36			
to-n	iLBC	Speex	iLBC	Speex
1	8.88	10.76	9.44	10.14
2	9.78	10.7	10.06	9.92
3	9.98	10.56	9.56	10.34
4	9.22	10.74	9.66	9.98
5	9.4	9.94	9.58	10.22
6	9.4	10.66	9.2	10.7
7	8.8	9.76	9.84	10.56
8	8.98	10.9	9.38	9.92

This shows that testing in the AI codec speex more stable in sending data packets because fewer packets are lost or unread, so calls using the speex codec are smoother than when using the iLBC codec. For testing between buildings, AH-AI has an average packet loss value of 9.45% for the iLBC codec while the speex codec is 10.36% so that the iLBC codec packet loss is smaller than the speex codec. This shows that the inter-building testing of the AH-AI iLBC codec is more stable in sending data packets because fewer packets are lost or unread, so calls using the iLBC codec are smoother than when using the speex codec.

Based on the packet loss parameter testing, the highest average packet loss value is 10.36% in the AH-AI inter-building speex codec test, while the lowest is 5.93% ms on the AI building's speex codec test. This is because VoIP calls pass through several touters at the Malang State Polytechnic including 192.168.130.1, 172.16.13.254, 172.16.18.18, and 192.168.181.250. Meanwhile, calls in the AI Building are directly directed to the server IP 192.168.181.250.

4) Analysis of Voice Quality Parameters

This sound quality test uses 5 different people's voice samples. Where A's voice is a female voice with an amplitude of 0.00000632 volts, B's voice is a female voice with an amplitude of 0.00003554 volts, C's voice is a female voice with an amplitude of 0.00003321 volts, D's voice is a male voice with an amplitude of 0.00006957 volts, while E's voice is a male voice. male with an amplitude of 0.0008277 volts.

In addition to using 5 different people's voice samples, each voice sample is recorded with varying durations of 3 seconds, 6 seconds, and 9 seconds. The sentence used for a duration of 3 seconds is "Friday, the twenty-second of April, our two thousand two", for the duration of 6 seconds, "Friday, the twenty-second of April, two thousand two-two we conducted a voice call test at the AH Building", while for the duration of 9 seconds, namely "Friday, the twenty-second of April two thousand two-two, we conducted a voice call test at the AH Building, State Polytechnic of Malang".

The sound quality analysis process is carried out mathematically using Matlab by reading the audio from the VoIP call test results which are displayed in the time and frequency domains to more easily compare the amplitude of the receiver and caller. The analysis is carried out by the voice comparison method where the results of voice recordings from VoIP calls are compared to determine the difference in the amplitude of the caller and receiver. From each call, the amplitude value will be seen from the caller and recipient side. To observe the sound quality resulting from audio readings in frequency mode, namely the delta amplitude of the caller and receiver. Delta amplitude is the difference between the amplitude of the caller and the receiver, where the higher the average delta amplitude, the lower the sound quality will be because there is a mismatch between the receiver and the caller [17]-[19].

Table 10 and 11 show the average value of the delta amplitude from the iLBC and Speex with 5 sound samples and a duration of 3 seconds, 6 seconds, 9 seconds, testing in the AH Building using the Linphone softphone, obtained from 5 voice samples with a duration of 3 seconds, the average delta amplitude iLBC codec is lower at 9.3E-07 volts. At a duration of 6 seconds, the average delta amplitude iLBC codec is lower, namely 1.45E- 06 volts. Meanwhile, for a duration of 9 seconds, the average delta amplitude iLBC codec is lower, namely 2.04E-06 volts.

TABLE 10
VOICE QUALITY TEST RESULT USING LINPHONE
VOICE COMPARISON METHOD IN THE AH BUILDING

Samples of Sound	Average Delta Amplitude (v)					
	3 sec		6 sec		9 sec	
	iLBC	Speex	iLBC	Speex	iLBC	Speex
A	1.26E-06	1.36E-06	1.94E-06	2.73E-06	3.02E-06	4.22E-06
B	8.80E-07	1.10E-06	2.02E-06	1.96E-06	3.08E-06	3.07E-06
C	8.70E-07	1.26E-06	1.14E-06	1.58E-06	1.49E-06	2.08E-06
D	8.10E-07	1.18E-06	1.06E-06	1.37E-06	1.29E-06	1.62E-06
E	8.50E-07	1.24E-06	1.09E-06	1.52E-06	1.34E-06	1.81E-06
Average	9.30E-07	1.23E-06	1.45E-06	1.83E-06	2.04E-06	2.56E-06

This shows that the iLBC codec has better sound quality when tested in the AH Building using a linphone softphone. Meanwhile, for testing in the AH Building using the Zoiper 1.36 softphone, it was obtained from 5 sound samples with a duration of 3 seconds, the average delta amplitude codec iLBC was lower, namely 1.04E-06 volts. At a duration of 6 seconds, the average delta amplitude of the iLBC codec is lower, namely 3.09E-06 volts. Meanwhile, for a duration of 9 seconds, the average delta amplitude of the iLBC codec is lower, namely 3.17E-05 volts. This shows that the iLBC codec has better sound quality when tested in the AH Building using the Zoiper 1.36 softphone.

TABLE 11
VOICE QUALITY TEST RESULT USING ZOIPER 1.36
VOICE COMPARISON METHOD IN THE AH BUILDING

Samples of Sound	Average Delta Amplitude (v)					
	3 sec		6 sec		9 sec	
	iLBC	Speex	iLBC	Speex	iLBC	Speex
A	3.14E-06	3.20E-06	1.20E-05	1.20E-05	1.53E-04	1.60E-04
B	7.30E-07	1.37E-06	1.41E-06	2.43E-06	2.22E-06	3.69E-06
C	4.00E-07	8.40E-07	5.90E-07	1.10E-06	1.00E-06	1.52E-06
D	3.90E-07	9.90E-07	6.30E-07	1.22E-06	9.30E-07	1.48E-06
E	5.20E-07	9.00E-07	8.20E-07	1.15E-06	1.03E-06	1.42E-06
Average	1.04E-06	1.46E-06	3.09E-06	3.57E-06	3.17E-05	3.36E-05

Table 12 and 13 show the average value of the delta amplitude from the iLBC and Speex codecs with 5 sound samples and a duration of 3 seconds, 6 seconds, 9 seconds, testing in the AI Building using the Linphone softphone, obtained from 5 voice samples with a duration of 3 seconds, the average delta amplitude the iLBC codec is lower at 2.0E-08 volts. At a duration of 6 seconds, the average delta amplitude of the iLBC codec is lower, namely 6.0E-08 volts. Meanwhile, for a duration of 9 seconds, the average delta amplitude of the iLBC codec is lower, namely 1.1E-07 volts.

TABLE 12
SOUND QUALITY TESTING RESULTS USING LINPHONE
THE COMPARATIVE SOUND TESTING METHOD IN THE AI BUILDING

Samples of Sound	Average Delta Amplitude (v)					
	3 sec		6 sec		9 sec	
	iLBC	Speex	iLBC	Speex	iLBC	Speex
A	0	0	1.0E-08	2.0E-08	1.5E-07	1.0E-08
B	2.0E-08	6.0E-08	1.4E-07	1.5E-07	1.8E-07	1.9E-07
C	1.0E-08	6.0E-08	5.0E-08	1.3E-07	9.0E-08	2.3E-07
D	2.0E-08	5.0E-08	3.0E-08	9.0E-08	6.0E-08	1.5E-07
E	4.0E-08	7.0E-08	6.0E-08	9.0E-08	9.0E-08	1.3E-07
Average	2.0E-08	5.0E-08	6.0E-08	1.0E-07	1.1E-07	1.4E-07

This shows that the iLBC codec has better sound quality when tested in the AI Building using the linphone softphone. Meanwhile, for testing in the AI Building using the Zoiper 1.36 softphone, it was obtained from 5 sound samples with a duration of 3 seconds, the average delta amplitude codec iLBC was lower, namely 0.00000454 volts. At a duration of 6 seconds, the average delta amplitude iLBC codec is lower, namely 0.00000689 volts. Meanwhile, for a duration of 9 seconds, the average delta amplitude iLBC codec is lower, namely 0.00000982 volts. This shows that the iLBC codec has better sound quality when tested in the AI Building using the Zoiper 1.36.

TABLE 13
SOUND QUALITY TESTING RESULTS USING ZOIPER 1.36
THE COMPARATIVE SOUND TESTING METHOD IN THE AI BUILDING

Samples of Sound	Average Delta Amplitude (v)					
	3 sec		6 sec		9 sec	
	iLBC	Speex	iLBC	Speex	iLBC	Speex
A	7.48E-06	3.59E-05	1.24E-05	5.63E-05	1.80E-05	8.13E-05
B	6.25E-06	1.74E-05	8.63E-06	2.51E-05	1.36E-05	3.28E-05
C	3.67E-06	1.35E-05	5.76E-06	1.56E-05	8.21E-06	1.72E-05
D	3.11E-06	1.07E-05	4.82E-06	1.42E-05	5.56E-06	1.68E-05
E	2.17E-06	9.27E-06	2.88E-06	1.12E-05	3.77E-06	1.44E-05
Average	4.54E-06	1.74E-05	6.89E-06	2.45E-05	9.82E-06	3.25E-05

Table 14 and 15 show the average value of the delta amplitude of the iLBC and Speex codecs with 5 sound samples and a duration of 3 seconds, 6 seconds, 9 seconds, testing between AH-AI buildings using the Linphone softphone, obtained from 5 sound samples with an average duration of 3 seconds. The delta amplitude codec of the iLBC is lower at 0.00000097 volts. At a duration of 6 seconds, the average delta amplitude iLBC codec is lower, namely 0.00000159 volts. Meanwhile, for a duration of 9 seconds, the average delta amplitude iLBC codec is lower, namely 0.00000212 volts.

TABLE 14
RESULTS OF SOUND QUALITY TEST USING LINPHONE
THE COMPARISON OF SOUND TESTING BETWEEN AH-AI BUILDING TESTING

Samples of Sound	Average Delta Amplitude (v)					
	3 sec		6 sec		9 sec	
	iLBC	Speex	iLBC	Speex	iLBC	Speex
A	9.00E-07	6.30E-07	1.68E-06	1.10E-06	2.59E-06	2.61E-06
B	1.06E-06	2.36E-06	2.43E-06	3.06E-06	3.02E-06	3.84E-06
C	4.10E-07	2.30E-06	6.90E-07	2.92E-06	1.07E-06	4.13E-06
D	1.81E-06	1.83E-06	2.62E-06	2.38E-06	2.99E-06	3.07E-06
E	6.70E-07	2.71E-06	5.20E-07	2.52E-06	9.20E-07	3.39E-06
Average	9.70E-07	1.96E-06	1.59E-06	2.40E-06	2.12E-06	3.41E-06

TABLE 15
RESULTS OF SOUND QUALITY TEST USING ZOIPER 1.36
THE COMPARISON OF SOUND TESTING BETWEEN AH-AI BUILDING TESTING

Samples of Sound	Average Delta Amplitude (v)					
	3 sec		6 sec		9 sec	
	iLBC	Speex	iLBC	Speex	iLBC	Speex
A	9.90E-07	1.25E-06	1.28E-06	1.71E-06	1.59E-06	2.18E-06
B	1.81E-06	3.00E-06	1.95E-06	2.85E-06	2.47E-06	3.69E-06
C	1.87E-06	3.02E-06	2.34E-06	3.15E-06	2.92E-06	3.64E-06
D	1.73E-06	2.33E-06	2.00E-06	2.73E-06	2.59E-06	3.72E-06
E	2.18E-06	1.79E-06	2.47E-06	2.38E-06	3.26E-06	2.72E-06
Average	1.72E-06	2.28E-06	2.01E-06	2.56E-06	2.57E-06	3.19E-06

This shows that the iLBC codec has better sound quality when tested in the AI Building using the linphone softphone. Meanwhile, for testing between AH-AI buildings using the Zoiper 1.36 softphone, it was obtained from 5 sound samples for a duration of 3 seconds, the average delta amplitude codec iLBC was lower, namely 0.00000172 volts. At a duration of 6 seconds, the average delta amplitude of the iLBC codec is lower, namely 0.00000201 volts. Meanwhile, for a duration of 9 seconds, the average delta amplitude of the iLBC codec is lower, namely 0.00000257 volts. This shows that the iLBC codec has better sound quality when testing between AH-AI buildings using the Zoiper 1.36 softphone.

IV. CONCLUSION

Based on the design, observation and testing that has been carried out, conclusions can be drawn according to the Qos parameters including delay, jitter and packet loss and voice quality in VoIP communication which is measured based on the voice comparison method in Matlab based on the resulting delta amplitude.

Based on the design, the FreePBX server can be implemented as a VoIP server using a raspberry pi4 with Raspbx OS and an asterisk server version 16. The server is installed in the Lab. The AI Building Jartel so that the server IP is 192.168.181.250 while the client IP when testing the AH Building is 192.168.130.94 for the client IP when testing the AI Building, which is 192.168.183.55. So as long as the client uses the Polynema network, the client can be registered with the server.

Based on QoS testing including delay, jitter and packet loss, it was obtained for the delay parameter the average delay result for the codec speex is smaller than the iLBC codec of 34.45 ms. For the jitter parameter, the average jitter of the iLBC codec is smaller than the speex codec of 0.00748 ms. The packet loss parameter for the iLBC codec is smaller than the speex codec of 7.27%.

Meanwhile, based on the sound quality test on matlab using linphone softphone and Zoiper 1.36 in the AH, AI, and AH-AI buildings, the iLBC codec has an average value of delta amplitude lower than the speex codec with a value of 6.7E-06 volts. in the AH Building, 3.6E-06 volts in the AI Building, and 1.8E-06 volts between the AH-AI Building.

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