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Analysis of Voice Quality in Communication Between IP Phones on the Intranet Network at the State Polytechnic of Malang

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***Abstract*— IP Phone is a Voice over Internet Protocol (VoIP) device that can transmit voice using IP. From communication between IP Phones there are waveforms of input and output information signals, the response of voice amplitude to changes in voice frequency and delay value from time shift at the beginning of the wave period of the input and output information signal using the Transmission Test Set and Oscilloscope measuring instruments to determine the character of the sound quality in communication between IP Phones according to the parameter test results. The system test results are that the waveform of the input and output information signal at the sound frequency has a sine wave shape. However, the output waveform of IP Phone A with a frequency of 300Hz to 1100Hz and IP Phone B with a frequency of 300Hz to 1300Hz is distorted, then from the results of the Peak to Peak wave is known to have a voltage gain, that is, the greater the frequency value at the input, the smaller the value of the resulting voltage gain.**

***Keywords*— Codec, IP PBX, IP Phone, Sound Quality, VoIP.**

I. INTRODUCTION

Voice over Internet Protocol (VoIP) is a technology that makes internet media to be able to perform direct long-distance voice communication. VoIP technology has many advantages, one of which is lower costs compared to cell phone tariffs [1]-[4]. From a survey result of network usage in companies based on the features used, teleconference activities via VoIP are still used in companies with 13.54%. This indicates that the use of VoIP in companies is still in demand [5]-[8].

IP Phone is one of the telephone terminals in Voice over IP (VoIP) that can transmit voice using internet protocol (IP). IP Phone is used to carry out the communication process in a building or institution, in contrast to the Analog Phone used by people in everyday life [9]-[13]. Over time, various IP Phones continue to appear with competitive prices in the homeland electronics market. This raises a question regarding how the comparison of IP Phone voice quality presented by these various types, considering that voice quality in telecommunications is very influential. Voice quality in a telecommunication device is a significant factor, and if a device has poor sound quality, it can cause misunderstandings during telecommunication [14].

Research on the comparison of voice quality using measuring tools in communication between IP Phones is still rarely done. This study will analyze how the character of voice quality in communication between IP Phones uses the Transmission Test Set and Oscilloscope measuring instruments. The parameters used in this study include the waveform, the response of the sound amplitude to the sound frequency and the delay of the time shift at the beginning of the wave period of the input and output information signal. The TBS1022 series Oscilloscope and Transmission Test Set model 3551A are used to test the system [15][16][17].

II. SYSTEM MODEL

The type of research used is quantitative research based on parameter tests. Two different types of devices observed input and output information waveforms, sound amplitude response to changes in sound frequency and delay value.

Two system design models are used in this study; first, communication is carried out using IP Phone type A as the sending side with testing on the microphone and IP Phone type B as the receiving side with testing on the speakers.

Fig. 1 and Fig. 2 are the sound quality testing system planning. The difference lies in the type of IP Phone used on the sending and receiving sides.

The microphone (TX) test is carried out on the sending side, and the speaker (Rx) test is performed on the receiving side.

Before communicating, the IP Phone is configured with the IP PBX server by opening the server IP address, and then several extension numbers will be displayed that the IP Phone can use to communicate. The selected extension number is used as the identity of the IP Phone device in communicating.

Then the IP Phone is connected to a switch already connected to the intranet network using an RJ45 connector cable so that the IP Phone gets an IP address to register with the server. When the IP Phone has successfully registered the extension number on the server, the IP Phone can be used for communication.

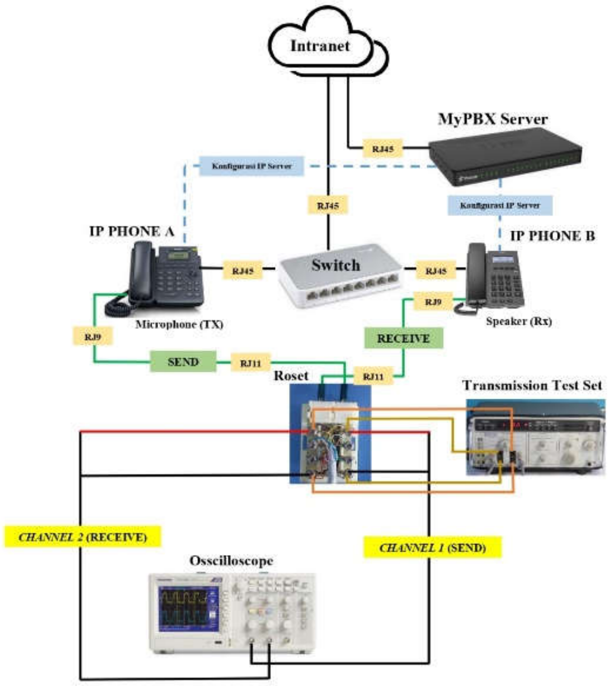


Fig. 1. The proposed system model

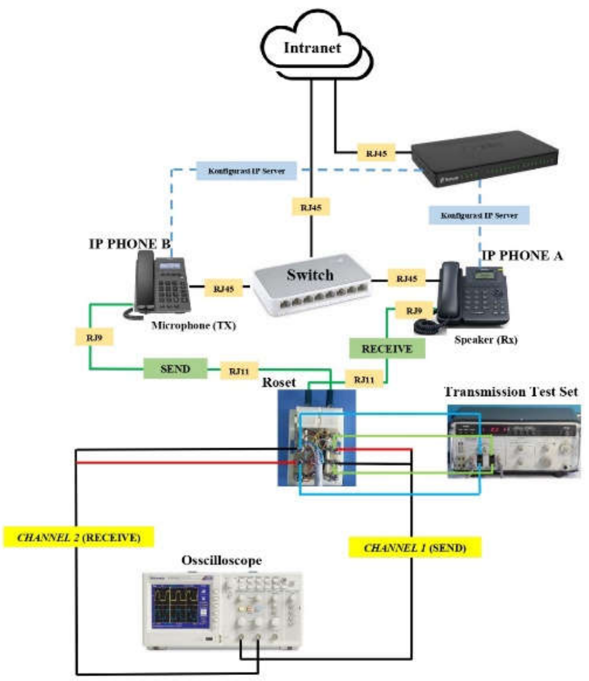


Fig. 2. The proposed experiment measurement

The next stage is sending IP Phone calls to the receiving IP Phone using the extension number that was previously registered. When the recipient's IP Phone has picked up the call (on-hook condition), the two IP Phones are connected with a rosette. The connection uses an RJ11 cable on the rosette port and an RJ9 cable on the handset port of both types of IP Phones. Each type of IP Phone has a different layout of the microphone and speaker on the rosette, which will be connected to the Transmission Test Set, so check the rosette using a multimeter to determine the layout of the microphone used for the sending side and the speaker used for the receiving side on the rosette.

After knowing the layout of the microphone as the sender and the speaker as the receiver, the Transmission Test Set and the Oscilloscope are connected to a rosette so that system testing can be carried out.

In the Transmission Test Set, the sound frequency is changed from 300Hz to 3400Hz as the input frequency and set the amplitude value of -23.5 dBm as a fixed amplitude input value. The results displayed on the Transmission Test Set are the input and output frequency values and the input and output sound amplitude values.

On the oscilloscope, there are two channels used in testing. Channel 1 is connected to the rosette on the microphone side of the sending IP Phone, while channel two is connected to the rosette on the speaker side of the receiving IP Phone. The results displayed on the oscilloscope are the input and output frequency values, input and output waveforms, peak to peak input and output waves and time shifts at the beginning of the input and output information signal wave periods.

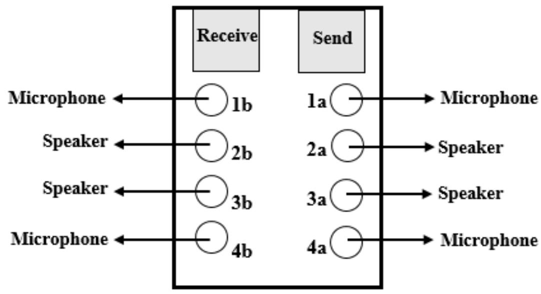


Figure 3. Layout of Microphone Testing and Type A IP Phone Speakers in Rosette

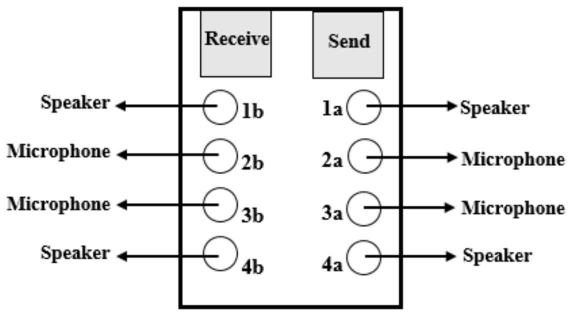


Figure 4. Layout of Microphone Testing and Type B IP Phone Speakers in Rosette

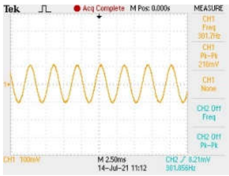
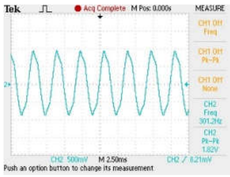
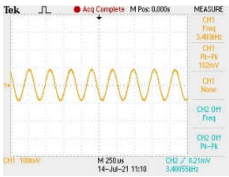
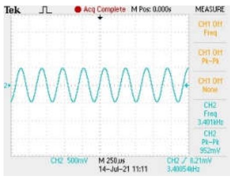
III. RESULTS AND DISCUSSION

This section describes the experimental results of the proposed system and discusses the results of each parameter that has been carried out.

A. Input and output waveform comparison

The test results of the waveform parameters of the information signal at the input and output when communication occurs between IP Phones are obtained from the Oscilloscope in the form of an information signal waveform on the sending and receiving sides with peak to peak values for each waveform.

TABLE I
SAMPLE DATA FORM OF INFORMATION WAVES FREQUENCY 300HZ AND 3400HZ ON IP PHONE TYPE A AND IP PHONE TYPE B

No	Freq. (Hz)	Information Signal Waveform (Pk-Pk)	
		Sent	Received
IP Phone type A as sender and IP Phone type B as receiver			
1	300	 Figure 5. Information Signal Waveform Input Frequency 300Hz (IP Phone Type A – IP Phone Type B)	 Figure 6. Information Signal Waveform Output Frequency 300 Hz (IP Phone Type A – IP Phone Type B)
2	3400	 Figure 7. Signal Waveform Input Frequency 3400Hz (IP Phone Type A – IP Phone Type B)	 Figure 8. Information Signal Waveform Output Frequency 3400Hz (IP Phone Type A – IP Phone Type B)

B. Response of Sound Amplitude to Changes in Sound Frequency.

The results of the response value of the sound amplitude to changes in the sound frequency, from the Transmission in the frequency range of 300Hz to 3400Hz and the limit of the input amplitude value of -23.5dBm. The input and output sound amplitude values will be displayed on the Transmission Test Set.

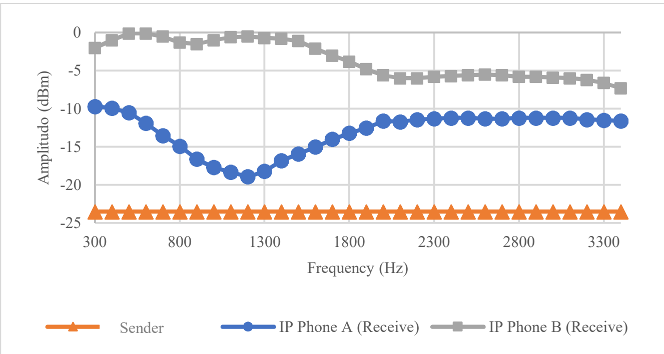


Figure 9. Graph of Amplitude Value Results Against Frequency Changes

In IP Phone type A, the largest amplitude value is at a frequency of 300Hz with a value of -9.7 dBm, while the smallest amplitude value is located at a frequency of 1200Hz with a value of -18.9 dBm. While on IP Phone type B, the largest amplitude is located at 500Hz and 600Hz with an amplitude value of -0.1 dBm, while the smallest amplitude value is located at a frequency of 3400Hz with an amplitude value of -7.3 dBm.

From the results of the amplitude response test to changes in frequency and is accompanied by a calculation of the power gain using the formula:

$$\text{Power Amplification (dB)} = dBm_{output} - dBm_{input}$$

It is stated that IP Phone type A has a lower amplification value than IP Phone type B on the reception side. The gain on IP Phone type A ranges from 13.79 dB to 11.9 dB, while the gain on IP Phone type B ranges from 21.5dB to 16.2 dB.

C. Delay Value for Each Change in Sound Frequency

The results of the delay value test of the time shift at the beginning of the wave period of the input and output information signal at each change in the sound frequency from 300Hz to 3400Hz.

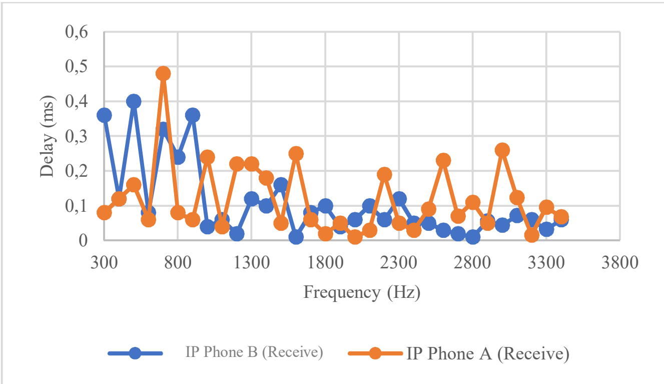


Figure 10. Graph of Delay Value Results Against Changes in Frequency

The results of the measurement and calculation of the delay parameter from the time shift at the beginning of the input and output, also output information signal wave period on IP Phone type B as the receiver has the largest delay value of 1.6 ms at a frequency of 300Hz, while the smallest delay value is 0.04 ms at a frequency of 1800 Hz. Furthermore, IP Phone type A as the receiver has the largest delay value of 0.48 ms at a frequency of 700Hz, while the smallest delay value is 0.16 ms at 3200 Hz.

The test data results show that the frequency of 300Hz to 3400Hz has a delay value of less than 150ms, both in communication from IP Phone type A to IP Phone type B and vice versa. Referring to the TIPHON standard issued by ETSI TR 101 329 V2.1.1 regarding "General aspects of Quality of Service (QoS)", the delay value of less than 150 ms is included in the excellent category with an index value of 4, meaning that in communication between two types of IP The phone already has a delay value that meets the standard in the excellent category.

IV. CONCLUSION

From the results of research and discussion of the system that has been tested, it is concluded that the test results of the input and output information signal waveforms show that both types of IP Phones have a sine wave shape on the input side, but on the output side on IP Phone type A the frequency is 300Hz to 1100Hz. Moreover, IP Phone type B frequency 300Hz to 1300Hz has noise, so it does not form a complete sine. The amplitude response value to changes in sound frequency has the highest amplitude value of -9.7 dBm on IP Phone type A and -0.1 dBm on IP Phone type B; from the calculation of decibels, the gain on IP Phone A tends to be small at a frequency of 300Hz to 1200Hz and at the frequency of 1300Hz to 3400Hz tends to be large, while on IP Phone B the gain value tends to decrease when the frequency value increases. The delay value of the difference in input and output waves on IP Phone type A and IP Phone type B has a delay value of <150ms with the highest delay value on IP Phone type A 0.48 ms and the highest delay value on IP Phone type B 1.6 ms.

REFERENCES

[1] R. Abrar Muhammad Saleh, "Implementasi dan Analisa Kinerja VoIP Server Pada Jaringan Wireless LAN Menggunakan SmartPhone," J. Elektr. Borneo, vol. 5, no. 1, pp. 1–5, 2019.

[2] M. R. Syahrial, "Analisa Quality of Service IP Telephony dengan Metode Low Latency Queuing," J. Telekomun. dan Komput., vol. 5, no. 1, p. 69, 2014.

[3] I. Salsabila, S. Anwar, and R. Radhiah, "Perbandingan Kualitas Suara Smartphone Menggunakan Metode Dynamic Time Warping (DTW)," J. RESTI (Rekayasa Sist. dan Teknol. Informasi), vol. 5, no. 1, pp. 82–90.

[4] F. L. Witi and A. Mude, "Analisis Jaringan Intranet Di Universitas Flores Menggunakan Quality Of Service (QoS)," vol. 01, pp. 7–12, 2020.

[5] S. M. L. Tobing, Tugas akhir Aplikasi VoIP Pada USUNET (Universitas Sumatera Utara). Universitas Sumatera Utara, 2011.

[6] M. Exsan and U. Fadlilah, "Pembangunan Infrastruktur Voice Over Internet Protocol di Organisasi Perangkat Daerah Boyolali menggunakan Server Elastix," Emit. J. Tek. Elektro, vol. 17, no. 2, pp. 39–47, 2017.

[7] Sutarti, Siswanto, and A. Subandi, "Implementasi Dan Analisis QoS (Quality of Service) Pada VoIP (Voice Over Internet Protocol) Berbasis Linux," J. PROSISKO, vol. 5, no. 2, pp. 92–101, 2018.

[8] A Hariyadi et al 2021 IOP Conf. Ser.: Mater. Sci. Eng. 1073 012050

[9] D. Anggraini, "Analisis FXS (Foreign Exchange Subscriber) Port Pada IP PBX (Internet Protocol Private Branch Exchange)," Politeknik Negeri Malang, Malang, 2019.

[10] F. Group, "Datasheet FANVIL X1/X1P," Cina, 2018.

[11] K. Features, "Datasheet YEALINK T19P-E2," Cina.

[12] K. Wallace, Implementing Cisco Unified Communications Voice over IP and QoS (CVOICE) Foundation Learing Guide. 2011.

[13] E. T. S. Institute, "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON); General aspects of Quality of Service (QoS)," vol. V2.1.1, no. ETSI TR 101 329, 1999.

[14] ITU-T Study Group 12, "ITU-T P.861 - Objective quality measurement of telephone- band (300-3400 Hz) speech codecs," vol. 861, 1998.

[15] D. R. J. Coolen, Electronic Communications, Third Edit. Canada: Erlangga, 1984.

[16] S. Waluyanti, "Dasar-dasar sinyal audio 1.1.," 2000.

[17] M. H. Weik, "Telephone Network," Comput. Sci. Commun. Dict., pp. 1752–1752, 2000, doi: 10.1007/1-4020-0613-6_19250.