Analysis Of Voice Quality on Aircraft Telephone Through Internet Telephony Gateway in Voice Over Internet Protocol

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***Abstract*—Along with the development of the era, analog telephones have been replaced by IP Phones whose communication range can be wider and more flexible. Therefore, analog telephones are almost forgotten because of the limited communication range. With the addition of ITG (Internet Telephony Gateway) then analog telephones can communicate in VoIP networks. This research will discuss the analysis of voice quality in the communication process between telephone sets via ITG in a VoIP network based on the shape of the input and output, the amplitude response to frequency changes and the delay. Data retrieval is done by using a transmission test set as a substitute for human voice and knowing the amplitude value generated every time there is a change in frequency and an oscilloscope as a tool to determine the shape of the input and output waves as well as the delay. The results of this study indicate that delay generated by communication between analog telephones is <1 ms. When viewed from the form of the information signal generated from communication between telephone sets, that is, the shape of the input and output information signals are both sines, so it can be said that the information signal has not decreased in quality. The amplitude response resulting from frequency changes in communication between analog telephones is, the greater the value of the communication frequency.**

***Keywords*— Amplitude response, delay, internet telephony gateway (ITG), VoIP, Waveform.**

I. INTRODUCTION

Voice over Internet Protocol (VoIP) is a voice communication service technology that carries IP (internet protocol) as a protocol in regulating the distribution of packaged voice signals through the transmission media used. Voice communication services with this technology are very effective, especially in terms of cost [1]. The use of IP networks allows cost savings because there is no need to create a new infrastructure for voice communication and the use of data width (bandwidth) is smaller than that of a regular telephone [2]. This is because VoIP can be assigned to any Ethernet and IP address, unlike traditional telephones which have to have their own ports in the Central or PBX. When planning a VoIP network [3], it is necessary to have a server that functions as an IP PBX [4].

In the era of globalization as it is today, the exchange of information is getting easier and the communication process is getting faster. This fast and easy communication process is supported by telecommunications equipment, such as telephones. However, along with the development of the analog telephone era, it has been replaced by the IPPhone whose communication range can be wider and more flexible [5]. Therefore, analog telephones are almost forgotten because of the limited communication range. With the addition of ITG (Internet Telephone Gateway), analog telephones can communicate in VoIP networks [6].

VoIP in its application uses a LAN network system and supports VoIP protocols such as standardization of SIP (Session Initiation Protocol) [7]. The use of more efficient VoIP technology will be simplified because it can be combined with the local PSTN (Public Switched Telephone Network) telephone network [8] there, using a VoIP gateway it will connect to PABX (Private Automatic Branch eXchange) [9]. The working principle of VoIP is that it will convert voice into code digitally over a packet data network [10]. The difference between VoIP and traditional telephones is the problem of infrastructure if VoIP uses a computer network while traditional telephones use telephone infrastructure that has been built by conventional telephone companies [11]. VoIP networks can be built using wireless and wired networks [12].

The author will analyze using several parameters, namely the shape of the information signal from the input and output signal waves, amplitude response to frequency changes, and the delay [13]. By using a measuring instrument in the form of a Transmission Test Set as a substitute for the human voice as well as knowing the amplitude value [14] produced every time there is a change in frequency and an oscilloscope as a tool to determine the shape of the input and output waves and the delay [15].

II. METHOD

System design from voice quality analysis in communication between telephone sets via ITG in a VOIP network. There are two systems design models which are.

A. Research Design

Fig. 1 describes the initial procedure in the research including problem identification and literature study to identify problems in detail and search for supporting theories, design, and test systems to design and implement tools according to designs that have been made, data collection and analysis to process data from test results, conclusions to make a final decision based on the analysis of the test results.

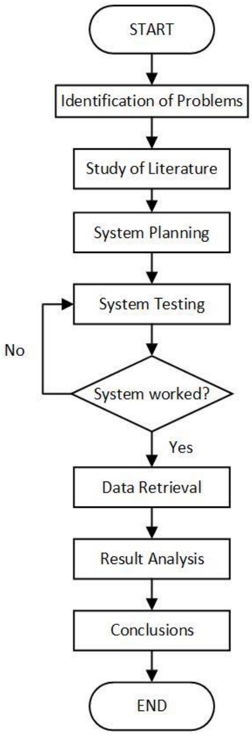

Figure 1. Diagram Research

B. Design System

Fig. 2 is a diagram block of the first system workflow system design that will be carried out during the research. An Intranet network is used as the main network for communication. Before communicating, ITG registration is carried out on the. ITG and IP Phone are connected to a LAN to get an IP to register on the server, then after ITG gets an IP, set up ITG via the web that has been provided by entering the IP number into a google search.

Next, phone 1 makes a call to phone 2, after having picked up the call (on hook) then the analogue phone is connected to the measuring instrument via a rosette and an RJ11 cable. After that, check the rosette to find out the send and receive. After obtaining the send and receive, the cable on the rosette is connected to the transmission test set. To find out the results of data collection for each parameter, an oscilloscope is used, where channel one is placed on a rosette that is connected to the transmission test set side to send, then channel two is placed on a rosette that is connected to the transmission test set side receive.

Fig. 3 is a diagram block of the system design regarding the workflow of the second system that will be carried out during the study. The workflow in the second block diagram has the same flow as the flow in the first diagram, the difference between the two lies in the phone being used as the sending side and the receiving side. In the second system diagram, phone two is used as the sending side and phone one is used as the receiving side. Next, check again on the rosette as explained in the first block diagram. After getting the results, phone two and phone one are connected to the transmission test set and an oscilloscope, the same as in the first diagram block.

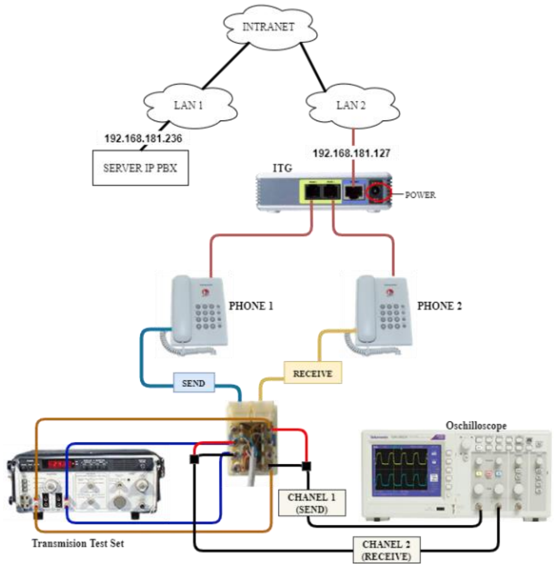

Figure 2. Diagram Block of Phone One to Phone Two

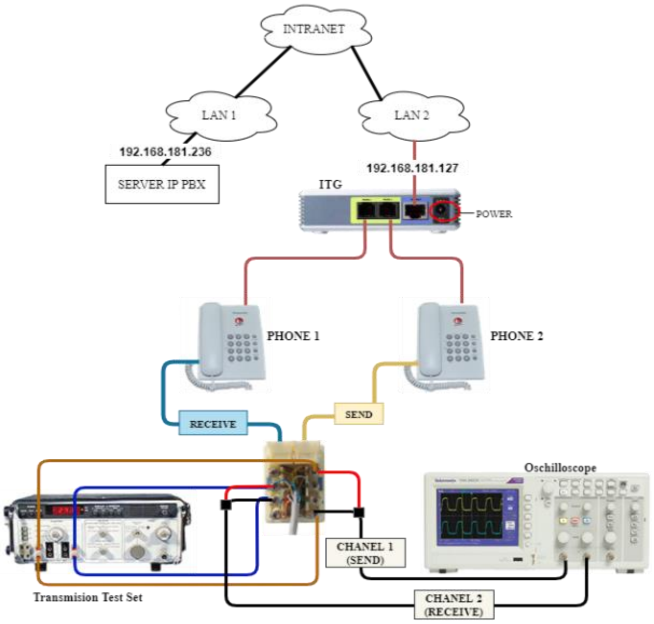

Figure 3. Diagram Block of Phone Two to Phone One

C. Determination of Procedure

According to Fig. 4, the first thing that needs to be done is to configure the measuring instrument transmission test set and oscilloscope. In addition, a cable skin is installed at the end of the rosette so that it is easy to use in conducting system testing. Next, connect the server with ITG by registering so that telephone devices can communicate. After the registration is complete, the rosette is checked to determine the location of the sending and receiving sides. Then, implement the system as a test to find out whether the implemented system is successful or not.

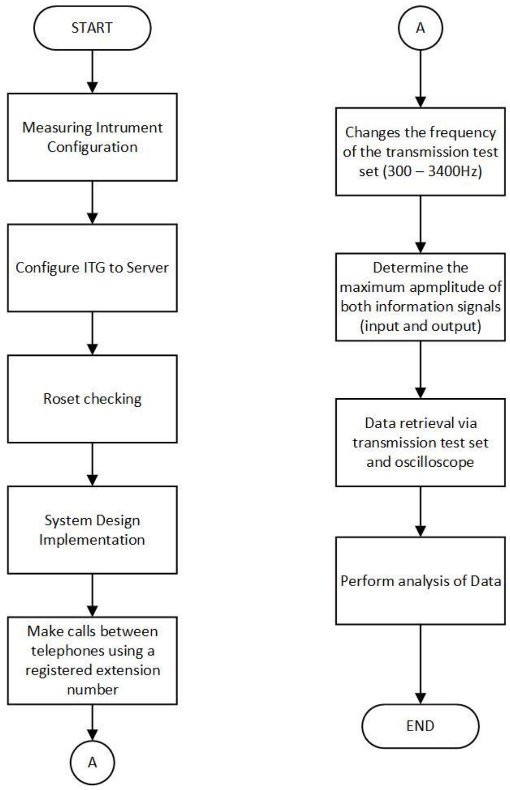

Figure 4. Flowchart of research procedures

After the system is successful, then communication is carried out between analog telephones that have been configured using an extension number that has been registered. measuring instrument *transmission test set* to change the frequency from 300Hz to 3400Hz. the *range* used was based on data from ITU-T P.861. Then, determine the maximum amplitude value from the sender and receiver sides on the *transmission test set*. Furthermore, data retrieval is carried out by recording the data results and storing the resulting images according to predetermined parameters. After the data is obtained, the data analysis process is carried out which will later be concluded as report writing.

III. RESULTS AND DISCUSSION

A. *Result of Input of Output Information Signal Waveform*

Is the result of a waveform information signal where the input is orange, and the output is blue. Based on these data results, it shows that communication from Phone 1 to Phone 2 and Phone 2 to Phone 1 for a range of 300 Hz-3400 Hz is obtained in the form of a sine information signal, both from the sender (input) and receiver (output) side. There is no difference between input and output information, it can be said that from the information signal obtained, communication between analogue telephones has good communication quality.

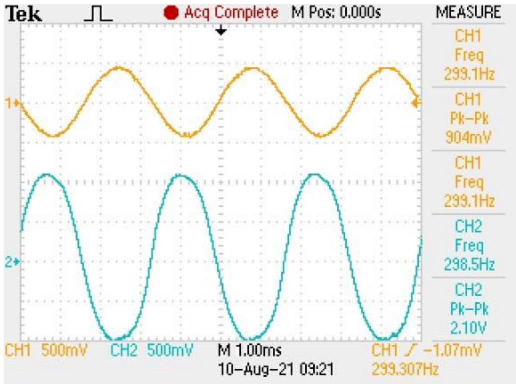

Figure 5. Information Signals from *Phone* 1 to *Phone* 2 Communications at a Frequency of 300 Hz

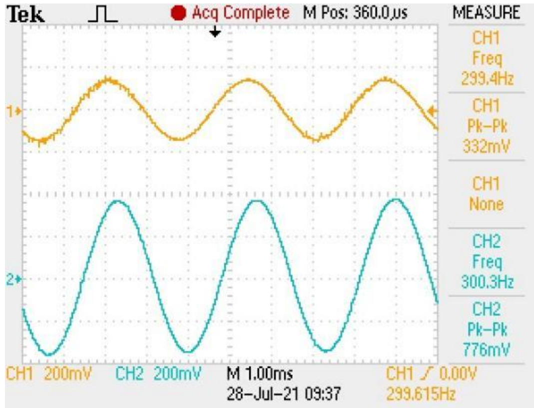

Figure 6. Information Signals from *Phone* 2 to *Phone* 1 Communications at a Frequency of 300 Hz

B. *Amplitude Response Result to Frequency Changes*

Based on the data in Table 1 and Fig. 7, it can be seen that the receive phone one with a frequency of 300 Hz – 2100 Hz there is a gain of 5 dB – 0.1 dB. Then at a frequency of 2200 Hz-3400 Hz, there is a weakening of 0.2 dB-3.3 dB. As we can be seen in Fig. 7, the largest gain is at a frequency of 300 Hz with a gain of 5 dB and the smallest gain at a frequency of 2100 Hz. Then, the largest attenuation is at a frequency of 3400 Hz, which is 3.3 dB, and the smallest attenuation is at a frequency of 2200 Hz with an attenuation of 0.2 dB. Furthermore, on receiving phone 2, at a frequency of 300 Hz-3400 Hz there is a gain of 10.3 dB-4.1dB.

The largest gain is at a frequency of 300 Hz, which is 10.3 dB and the smallest gain is at a frequency of 3400 Hz, which is 4.1 dB. As can be seen that in Fig. 7, the amplitude response to changes in the frequency of communication between analog telephones. The attenuation that occurs almost forms a straight line, and the greater the frequency of information. The smaller the received amplitude value is generated.

C. *Result of Testing Delay Value for Each Change in Frequency*

In table 1 at a frequency of 300 Hz – 3400 Hz the delay value varies from the smallest to 0.012 ms. At a frequency of 2900 Hz the largest gun is 0.32 ms at 600 Hz. Furthermore, in table 4 the smallest delay value is at a frequency of 3100 Hz and

3400 Hz, namely 0.04 ms. While the largest delay is at the lowest frequency, which is 300 Hz.

TABLE I
COMMUNICATION AMPLITUDE DATA TELEPHONE

No.	Frequency (Hz) Send and receive	Amplitude (dBm)		
		Send (phone 1 & 2)	Receive phone 1	Receive phone 2
1.	300	-10	-5	0.3
2.	400	-10	-6.3	0.3
3.	500	-10	-6.2	0.4
4.	600	-10	-5.8	0.4
5.	700	-10	-5.7	0.4
6.	800	-10	-6.4	0.3
7.	900	-10	-6.7	0.2
8.	1000	-10	-7.3	0.2
9.	1100	-10	-8	0.1
10.	1200	- 10	-6.7	0
11.	1300	-10	-7	-0.2
12.	1400	-10	-7.5	-0.3
13.	1500	-10	-8	-0.4
14.	1600	-10	-8.7	-0.5
15.	1700	-10	-8.8	-0.7
16.	1800	-10	-9.1	-0.8
17.	1900	-10	-9.4	-1.1
18.	2000	-10	-9.5	-1.3
19.	2100	-10	-9.9	-1.3
20.	2200	-10	-10.2	-1.4
21.	2300	-10	-10.5	-1.7
22.	2400	-10	-10.8	-1.8
23.	2500	-10	-11.1	-2.2
24.	2600	-10	-11.4	-2.3
25.	2700	-10	-11.7	-2.8
26.	2800	-10	-11.8	-2.9
27.	2900	-10	-12.1	-3.4
28.	3000	-10	-12.6	-3.4
29.	3100	-10	-12.5	-3.6
30.	3200	-10	-13.2	-3.4
31.	3300	-10	-13.1	-3.9
32.	3400	-10	-13.3	-5.9

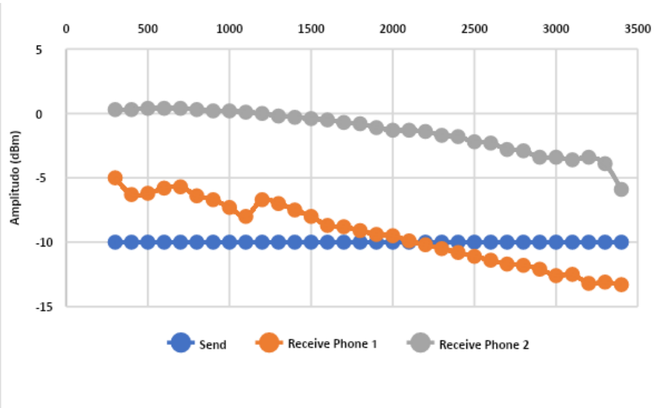

Figure 7. Amplitude response against changes in communication frequency between telephones

Based on the data in Fig. 8 and Fig. 9, it shows that from the frequency of 300 Hz to 3400 Hz, the delay value is said to be very good. This is due to the ETSI standard which states that the delay value <150 ms is classified in the very good category. So, the delay in communication from phone 1 to phone 2 and phone 2 to phone 1 has met the applicable standards. With the results of a straight-line graph up, which means that the frequency greatly affects the size of the delay.

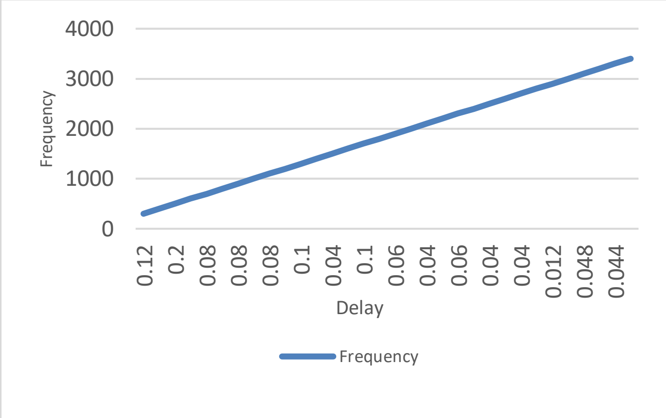

Figure 8. Testing delay value.

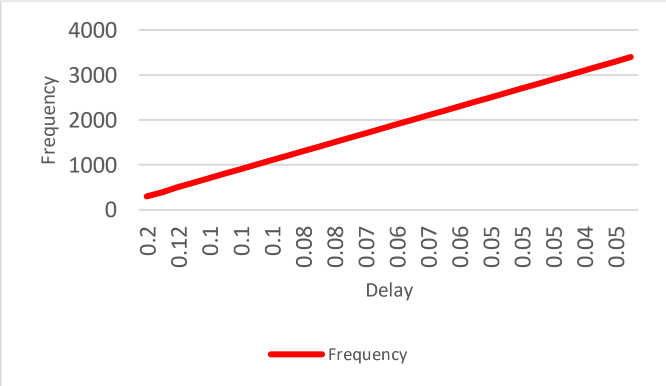

Figure 9. Graph of results of testing delay propagation.

IV. CONCLUSION

The form of the information signal generated from communication between telephone sets, namely, the transmitter and receiver are both sines, so it can be said that the information signal on communication does not experience a decrease in quality. The amplitude response resulting from frequency changes in communication between analog telephones is, the greater the value of the communication frequency, the smaller the received amplitude value is. Delay is <1 ms, based on the TIPHON standard which states that the delay <150 ms is classified in the very good category. So that delay in communication between analog telephones already meets the applicable standards.

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