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High Pass Filter Application to Reduce Voice Communication Delays on IP Phones

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Abstract: VoIP technology is equipped with several functions, namely, the signalling function, meaning the VoIP is in charge of receiving the network from the caller, after which the conversation delivered. This technology is capable to pass voice traffic in the form of packets over an IP network. Packets of sound undergo a long process or delay to get to the destination it can damage the voice's quality being heard. It happened because there is continuous delay in the communication between IP phone set on VoIP technology that causes echo in the receiver's voice. Echo can occur during communication with IP phone set with average delay capacity above 5ms. The delay on the delivery and reception as in the results of communication between telephones can be adjusted for sound frequency of 300 Hz to 1000 Hz as a cut off by adding High Pass Filter (HPF) application. HPF filter application is able to stabilize the amplitude of about -21 dBm from the set of transmission test on the receiver when there is weakening at low frequencies, but when the frequency is raised to 1500-2250 Hz the amplitude strengthens to -12,2 dBm or increases the value of about 8.8 dB. Meanwhile, for the lower frequency such as 300 – 950 Hz, the filter would not pass it since the frequency is designed to be cut-off at 1000 Hz. The value of delay is narrower to 0.08ms by HPF application at the frequency of sound upper limit received by any 1000-3400 Hz telephone set.
Keywords: VoIP technology, echo cancelling, HPF filter, delay, voice quality

I. INTRODUCTION

The VoIP (Voice over Internet Protocol) technology is converting sound data into digital code that is transmitted over data packets network instead of over analog telephone network. VoIP technology is equipped with several functions, among others, the signalling function which means that VoIP is functioning to receive the network from the caller, after which the conversation is delivered [1]. Then the caller and receiver can communicate with each other with the desired voice information. VoIP-owned database services are utilized to search final destination location, as well as interpreters of addresses utilized in two different networks [2]. A bearer control allows the users to receive, reject, or disconnect the phone calls. A coded operation is functioning to transmit or convert sound into data packets. VoIP technology utilizes an already available data network, so that the digital data packets can be compressed to reduce bandwith of about 8 kbps [3].However, VoIP has one weakness in terms of sound quality. The sound received is not as clear as on the Public Switched Telephone Network (PSTN) [4]. When the voice made over the VoIP network delayed means there is a pause when communication takes place [5]. The maximum delay recommended by ITU (International Telecommunication Union) for sound application is 150ms, while the maximum delay with acceptable sound quality to the user is 250ms. End to end delay is the number of analog to digital sound conversion delays, packetization time delays or referred to as packet length delays, and network delays at the point of time (t) [6]. Communication that is nailed VoIP in real time has weakness that need attention, namely, jitter, packet loss, and echo. The voice calls received on the VoIP network often result in echo [7]. The voice calls received on the VoIP network with echo result due to delays in the voice signals of about 400msecond. While the utilization of analog telephone connections through PSTN is about 10msecond that makes the sound does not experience echo. Some of the components that cause echo in VoIP calls include hardware utilized in telephone networks, such as broken wires, splitters, or poor quality analog headphones [8]. The intensity of the returning telephone volume is less strong than the signal sent resulting in an attenuation of 65dB. Electromagnetic interference from electronic devices such as computers, extension sockets, or connection on electrical lines with rather large electric field are also causing echo. Any inappropriate impedance in the analog system circuit from signal sent and receiver also results in sound echo. Echo results in user retention to VoIP telephone technology. The telephone system employs the phone handset to receive sound energy from the microphone by controlling the speaker's sound filtering from a strong signal to be received by the ear or referred to as "side tone" that experiences delays in the sound signal and is received by the ear of the phone receiver. The ratio of original sound signal and the sound signal with echo is calculated with decibel (dB) capacity. The echo-sounding values are lowest around 55 dB and highest around 15dB. When it is calculated in time the signal is coming and leaving from the path to the receiver set of about 30ms [9]. This time value is difficult to detect between side tone and echo.

When the time for the signal received or returned is increased by around 50ms, it becomes impossible to hear both the original sound from the speaker and the one heard by the receiver. A technique is expected for suppressing echo that involves filtering the sound signal received by the phone device with the sound amplitude variable set by the transmission test set device. The initial signal replacement device defines the range of sound frequencies that could be received by the phone, with the frequency of its settings causing echo in human voice. High pass filters, which are applied to sound signals, are a form of filter that misses high frequencies but disadvantages lower frequency amplitude than cutoff frequencies [10]. The study's objective based on the existing issues is to find voice echo cancellation by adding an HPF filter in suppressing echoes from IP phone devices linked to the internet that may send and receive voice data.

II. RESEARCH METHOD

This study is carried out to identify voice echo cancelling as a result of frequency limitations in human voices induced by spectral distortion when waves are reflected through the VoIP internet network. The test variable is delay as one of the causes of echo. In telephone transmission, echo might develop if the delay exceeds 5 milliseconds. Frequency range of human voice according to data of ITU-T P.861, the frequency range of 300 Hz - 3400 Hz is utilized as a frequency range in telephone communication [11]. High Pass Filter (HPF) is a filter that is used to reduce frequencies that are higher than the frequencies heard by telephone users. This filter may pass input signals with frequencies greater than or equal to the cut-off frequency (fc) while weakening input signals with frequencies less than or equal to the cut-off frequency (fc) [12]. HPF has two orders, high pass filter order 1 and high pass filter order 2, with variances in the transition band. An IP-phone is a telephone that is IP-based or that uses an internet connection to send and receive voice data. Rosettes are used as linking telephone devices with HPF filters to evaluate the quality of the signal generated by communication service providers who utilize intranet or internet networks, also known as IP PBX. The RJ9 cable to RJ11 links the phone to rosettes, filters, and measurement devices that monitor signals before and after filtering for the presence of echo. The RJ45 cable to RJ45 cable links the IP-PBX to the IP-phone, the transmission test set, the oscilloscope, the 50 kΩ potentiometer, the 10 NF capacitor, and the 5V power adapter DC as the filter's input.

The design of this study is as follows:

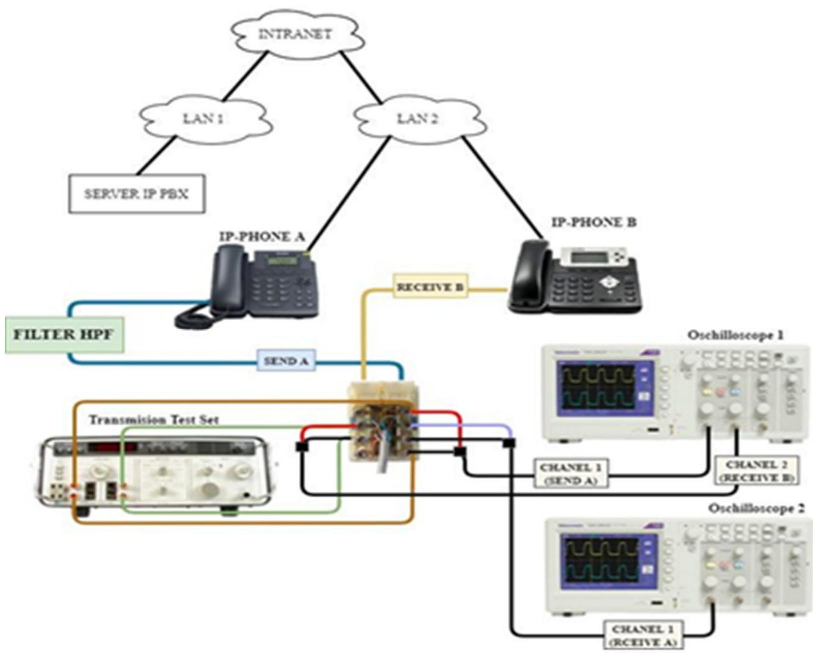

Fig. 1. Design of a System Suite and Echo Testing

The diagram block of system in figure 1 describes that Server (IP-PBX) serves as a telephone service provider for both IP phone A and IP phone B. The microphone on IP phone A is utilized as a transmitter (Tx - A), while the receiver (Rx-A) is utilized as a speaker. It goes the same with IP phone B where the receiver (Rx-B) is utilized as speaker B. HPF filters are linked to transmitters (Tx - A) then from transmitter Tx-A transmitted sound signals with frequencies up to the sound frequency limit of 300Hz - 3400Hz. HPF filters and receivers (Rx) are also connected to rosettes. Rosette is used to stabilize and filter telephone signals, with the transmission test set serving as a substitute source for information signals in the 300-3400 kHz telephone line frequency band.

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Tx- A is entered as input to transmission test set, while the oscilloscope is used as a display of the system's findings for the quality test of the sound signal and frequency established in the transmission test set. As an observation of the output of the observed sound signal with a shifting frequency range, the input transmission test set is linked to a rosette. When Tx-A interacts with Rx-B is, the echo frequency is limited by the HPF filter such that the sound received matches the original sound heard from Rx-A on the telephone devices. As a measurement device, the oscilloscope monitors the sound input signal and its output; the risk of delay arises from adjusting the phase of frequencies supplied and received by the telephone devices. The amplitude and delay time of the echo created by the telephone receiver during transmission are observable factors that interfere with the real audio conveyed. HPF filters are utilized as frequency restrictions when echo cancelling occurs in the received sound. The system's performance with the sound from the phone's speaker that is logged by the same phone's microphone may cause echo. The sound is evaluated from the entering sound flow while the flow of returning sound before being released again is monitored. The HPF filter is applied while passing through the rosette to reduce the excess noise entering the receiver, and the transmission test set's call parameters are utilized to stabilize the sound quality in the receiver. Based on the design of the cut-off frequency, the frequency is modified such that the HPF filter continues. Meanwhile, for the frequency below cut-off, the signal will be muffled. The HPF filter circuit employs a capacitor value of 10nF by calculating the value at the high pass value of order 1 to get resistance value of 15,915kΩ. Figure 2 depicts an HPF circuit.

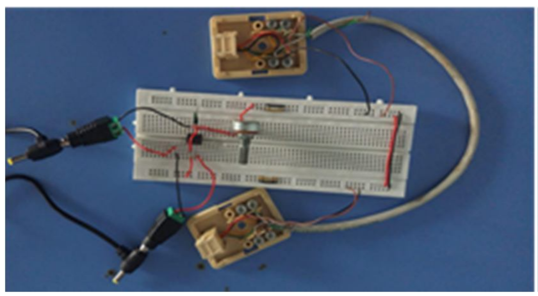

Fig. 2. Circuit of HPF Filter

An echo can occur when sound enters the receiver after a delay of more than 25 milliseconds. The PSTN, or Public Switched Telephone Network, contains impedance discrepancies, resulting in echoes in the receiver. When there is a delay in the delivery of sound, the HPF circuit is employed to reduce impedance.

III.RESULTS AND DISCUSSION

There are strengthening and weakening occurrence based on the voice quality test between IP Phones by displaying the amplitude response of frequency variations in the transmission test set. After the HPF filter, the delay values that result in echo are examined. The results of data retrieval on communication between telephone devices A and B, which function as Tx-A and Rx-B, shows that the value of changes in the range of sound frequencies tested using transmission test set as the input setting before going through the HPF filter observes the delay from its frequency changes. The HPF filter is designed by using simulations on multisims as the basis for voice echo cancelling mapping on telephone devices from data transmission test sets as a test with a frequency setting of 300 Hz to 1000 Hz with observation on an average delay on the receiver of about 1.01ms. HPF with a cut-off frequency of 1000 Hz and using capacitors of 10 nF. The data before filtering is about 550Hz with a delay of 2.16ms in the low frequency zone.

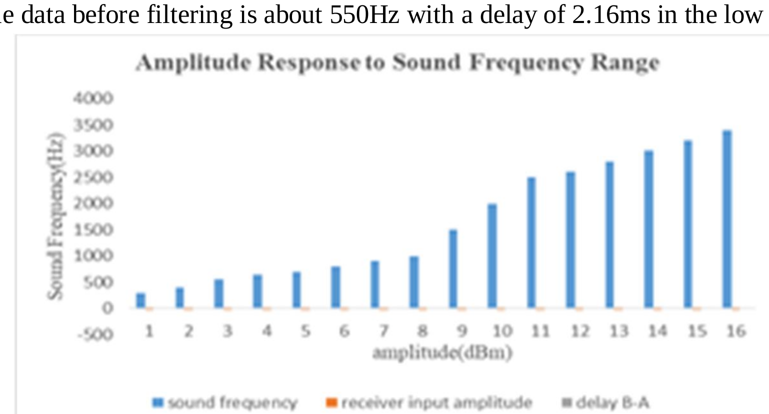

Fig. 3. Amplitude Response to Frequency Changes With Delay Restrictions

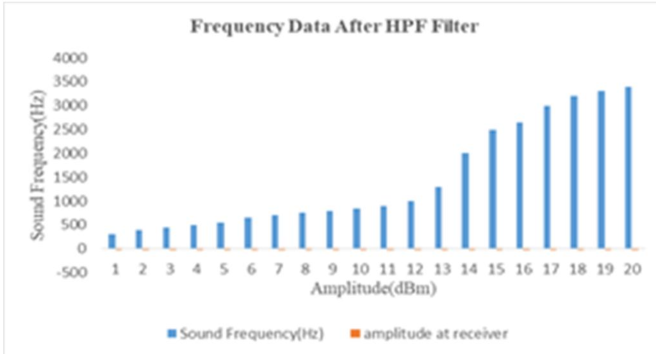

Fig. 4. Sound Amplitude Response with HPF Filter

Based on the graph analysis of the delay research results obtained on the telephone devices with the frequency input set from the transmission test set of around 5ms. When the receiver's phone has a lengthy delay limit but still in the ITU standard of 25ms, then the value is included within the regular standard limit. The amplitude stability of around -21dBm from the transmission test set setting on the receiver appears to be deteriorating for low frequencies, but when the frequency is raised to 1500-2250 Hz, the amplitude increases to -12.2 dBm which is a value increase of about 8.8 dB. An enhancement of 8.5 dB is seen with a HPF filter sound frequency in the range of 1000-3400Hz, while filters do not pass the frequency intended cut-off at 1000Hz at frequencies below 300-950 Hz. At the highest limit frequency of sound received by telephone devices, 1000-3400Hz, the delay value is 0.08ms.

IV. CONCLUSIONS

According to the findings of the research, delays caused on communication between telephone devices with a time period of roughly 5ms are still within ITU requirements. Echo in telephone device receivers that restricted the frequency range of the transmission test set has not created interference to the original sound on the receiver information with a high delay of 2.16ms held by the lowest frequency, which is between 500-950 Hz. HPF filters are employed to pass frequencies below the cut-off frequency of 1000Hz, resulting in a minor delay of 0.02ms for frequencies ranging from 2400 to 3250Hz.

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