Feedback remover on the sound system using inverse phase method

A Hariyadi*, R H Y Perdana, N Hidayati and Y Ratnawati

Electrical Engineering Department, State Polytechnic of Malang, Indonesia

*aadhariyadi@gmail.com

Abstract. Currently, almost all activities carried out by all parties are inseparable from the sound system. With the sound system, it is not necessary to speak loudly. There are several types of sound systems or models, ranging from the cheapest to the most expensive, with the usual quality up to good. However, there is a problem that can be obtained by a sound system that is of good quality or the usual quality, namely feedback from the speaker to the mic. Feedback will be very annoying because of the entry of sound from the speaker to the mic, which should be avoided. By using the inverse phase method and narrow bandpass filter, the simulation results can eliminate the feedback signal that is 100%. However, there are side effects on the band that occurs feedback, which is the reduction in the sound signal by the feedback power level that occurs between 30% - 100% that are in the frequency band range.

1. Introduction

Sound processing systems, or better known as sound systems, have become a device inherent in humans today, starting from television, mobile phones, home theatre etc. Large-scale sound systems are used when there are events such as weddings, meetings, meetings and so on. However, sometimes there is a small but very annoying problem, namely the emergence of feedback. Feedback occurs due to incorrect placement of sound equipment or improper sound system settings. So that the waves emitted by the loudspeaker go back into the microphone and enter back into the sound system, it is repeated continuously so that the signal will get bigger. With the feedback signal getting bigger, it will disturb the listener even more. Currently, the process of eliminating the feedback signal is still using manual methods, namely by finding the frequency of the feedback signal and reducing it.

The latest technology in sound systems already uses digital signal processing [1,2]. With the implementation of digital signal processing, it will further improve the quality of the sound signal. The use of simulations in digital signal processing is carried out to simplify calculations [3]. By using the Fourier transformation, which is the basis of digital signal processing, the signal value from the frequency domain will be obtained [4]. The frequency domain is the basis for the filter circuit design. Filters that are often used in a sound system are low pass filters, bandpass filters and high pass filters. The old type sound system still uses analogue filters, while the new sound system uses digital filters [5]. The advantage of digital filters is that the processing of values is easier and more precise. Because the filter is digital, the data that is processed is also in the form of digital data with the processing hardware, namely FPGA [1,6]. In digital signal processing, there is a method for removing a signal, which is called the inverse phase. This method uses an input signal with the phase reversed and then added to the input signal again so that the output signal is an addition to the input signal with a reversed signal.
In this study, a simulation is proposed to eliminate feedback that occurs in a sound system, by using the inverse phase method and combined with the use of bank filters. Hopefully, it can eliminate the feedback signal accurately.

2. Methods
The design method used in this study uses six filters, consisting of 1 low pass filter, four bandpass filters, and one high pass filter. Filter manufacturing begins using the order 64, 126, 190 and 254. The system design in this study is shown in Figure 1.

Figure 1 shows that the x[n] input signal will enter the filter bank with a different cut off frequency. With a speech signal bandwidth of 20-20KHz, the bandwidth used in the filter design is 2KHz wide. Then the signal after passing through the filter series will enter the processing unit. In this section, the signal that contains feedback will be selected, and the phase will be reversed using the inverse phase method. At the end of the signal that has been reversed will be added again with the input signal. The mathematical process in this study is shown in formula 1.

\[ Y[n] = X[n] + X_1[n] \, \ldots \, X_n[n]' \]  

(1)

In the formula 1, Y[n] is the output of the system, with the input being X[n] and X1[n]' is the input signal from the first filter to Xn[n]' which is the input signal from the nth filter that has been an inverse phase. The filter design process used is shown in Figure 2. Figure 2 is a picture showing six bandpass filters arranged to form a filter bank.
3. Results and discussion

Filters are made with orders ranging from 64, 126, 190 and 254. The results of filter design are shown in Figure 3 (a) for filters with order 64, Figure 3 (b) for filters with order 126, Figure 4 (a) for filters with an order: order 190 and filter with order 254 in figure 4 (b).

![Diagram](image.png)

**Figure 2.** Bandpass filter bank.

![Graphs](image.png)

**Figure 3.** (a) Filter with order 64, (b) Filter with order 126.
In a filter with a small order, it can be seen that the resulting filter response is not too sharp, but on orders above 190, the filter response is getting sharper. The process of designing and testing filters with higher orders will burden the processor because the mathematical process is very long and long. Filters are designed from 0Hz to 20KHz with six filter banks.

The system testing process is carried out by entering a speech signal that has been mixed with a feedback signal. Speech signals in the time domain and frequency domain are shown in Figures 5 (a) and (b). Furthermore, the feedback signal is shown in figures 6 (a) and (b).

**Figure 4.** (a) Filter with order 190, (b) Filter with order 254.

**Figure 5.** Voice signal (a) time domain, (b) frequency domain.
Figure 6. Feedback signal (a) time domain, (b) frequency domain.

In Figure 5, the time domain signal has a duration of 2.4 seconds and has a frequency ranging from 0Hz to 5KHz. Whereas in Figure 6, the feedback signal has a duration of about 1.3 seconds with a frequency of 1.3KHz and 2.1KHz to 2.3KHz. The result of adding the two signals produces a new signal, as shown in Figure 7.

Figure 7. Voice signal + Feedback signal (a) time domain, (b) frequency domain

It is shown in Figure 7, a speech signal that has added a feedback signal. The feedback signal has a very large amplitude compared to the speech signal so that in the frequency domain, it can be seen that the speech signal has a very small amplitude.

The results of the phase reversal process on the filtered signal can be seen in Figure 8.
In Figure 8 (a), a filtered feedback signal is shown then the phase of the signal is inverted to produce the signal shown in Figure 8 (b). Not visible difference from the inverse signal or not, but when enlarged you will see that the signal phase has been reversed.

After obtaining the inverse signal, then adding to the original signal in the form of a speech signal is still mixed with the feedback signal, resulting in an output signal $y[n]$ that has no feedback in it. The $y[n]$ signal is shown in Figure 9.

Figure 9 shows an output signal that has no feedback signal in it. It appears that the feedback signal is no longer visible in the signal in the frequency domain. However, the process of removing this feedback signal has a weakness. Namely, the original signal is also weakened at the frequency where the feedback signal is 30% - 100% of the original signal.

4. Conclusion
From the experiments that have been done, it can be concluded that using the inverse method to eliminate feedback on the sound system is running very well. With a percentage of the accuracy, eliminating the feedback signal reaches 100%. However, based on the test, it was found that the speech signal attenuation is at the same frequency as the feedback signal, so that the speech signal has
attenuation by 30%-100%. Overall, the use of the inverse method is successful in eliminating the feedback contained in a sound system.

References
[1] Lv M, Gao T, Yan D and Qiao L 2020 Design of Audio Signal Analyzer Based on MCU and FPGA J. Phys. Conf. Ser. 1544
[2] Ye Y, Wang K, Hu W, Li H, Yang K, Sun L and Chen Z 2019 A Wearable Vision-To-Audio Sensory Substitution Device for Blind Assistance and the Correlated Neural Substrates J. Phys. Conf. Ser. 1229
[3] Zhou N, Quan X and Wang Y 2019 Research on Digital Signal Processing Experiment System Based on Virtual Instrument Technology J. Phys. Conf. Ser. 1288
[4] Hidayat D, Syafei N S, Wibawa B M and Tumbelaka B Y 2018 Fourier transform of high frequency ultrasonic waves propagated with a transmission mode J. Phys. Conf. Ser. 1080
[5] Thabit A A 2020 Design and Software Implementation of Analog and Digital Filters for Coherent Receivers under Noisy Channels IOP Conference Series: Materials Science and Engineering vol 765
[6] Xin W, Zhi-Qiang H, Da-Qing J, Zhi-Xiao S, Yu Z and Yong L 2019 Several Implementation Methods of Signal Processing Algorithm Based on FPGA IOP Conference Series: Materials Science and Engineering vol 565