Phase Correction System Using Delay, Phase Invert and an All-pass Filter

University of Sydney
DESC 9115 Digital Audio Systems
Assignment 2
31 May 2011

Daniel Clinch
SID: 311139167
The Problem

Phase is one of the big issues in audio recording and reproduction. Current recording studio techniques often involve large multi-microphone setups as engineers try to create unique sounds by combining the different characters of microphones and their placements in a room. This requires great care to be taken with the placement and physical alignment of microphones to prevent certain frequencies being cancelled out when the signals are mixed together.

However, with the rise of the untrained engineer and the home recording studio, this care and attention to detail in the recording process is being lost. Consequently, inexperienced engineers often complain about thin sounds when mixing multiple microphones together even though each individual microphone may sound great. More and more, people are looking to their digital audio workstations to fix the problems with their recordings after the fact.

When correcting phase alignment issues resulting from a time delay between two microphones (for example) on a single source, one or more of the following techniques are usually used;

- A 180 degree phase inversion to one of the signals
- A simple delay to the earlier of the two signals
- or by moving one of the microphones to an “in phase” position.

While these can be effective they do have their limitations.

- A 180 degree phase inversion may improve the sound to some degree, but the best sounding result may lie somewhere between zero and 180 degrees.
- A delay may be effective for correcting errors caused by differences in arrival times at the transducers. However, timing may be unacceptably altered if the required corrective delay is too long and any desirable artefacts from the difference between the arrival of the two signals are now lost.
- Moving one of the microphones to an “in phase” position may work, but the sound will change and may not be desirable. This may also be impractical if not impossible in some circumstances.

These methods all occur in the time domain, but we can also adjust the phase response of a signal in the frequency domain by using an all-pass filter. The all-pass filter is somewhat under represented in the current software market and when combined with a phase invert and a delay line you have a complete tool for correcting phase issues.

Specification

This system consists of three parts, a delay line, a phase invert and an all-pass filter all combined in the one convenient tool.

![Fig 1. System block diagram](image-url)
The delay line, as its name suggests, delays the input signal by a given amount allowing it to be aligned in the time domain with another later occurring signal. As most adjustments will usually be small, the delay time is adjusted in milliseconds. Using milliseconds as the unit allows for easy calculation of the delay time if the distance between the microphones is known.

A 180 degree phase inversion is included to perform the phase flip that is commonly found on mixing consoles and most digital audio workstations. It simply inverts the polarity of the input signal.

At the heart of this system is the all-pass filter. The all-pass filter has a unique response compared to other filters. There is no change in gain across the frequency spectrum when it is applied to a signal, however, the phase relationships of the frequency components are affected. When the phase response of a first order all pass filter is plotted, there is a constantly increasing phase shift from zero degrees at 0 Hz, approaching 180 degrees at higher frequencies. In other words, the higher the frequency, the greater the phase shift. The all-pass filter performs phase shift in the frequency domain without effecting the time domain.

**Implementation**

**Delay Line**
A value for the delay given in millisecond units is converted into seconds and calculated as samples with the formula;

\[
\text{delay (samples)} = \frac{fs \times (\text{delay (ms)} / 1000)}{fs = \text{sampling frequency in samples/second}}
\]

Eq. 1

An array of zeros equal to the length of the delay is then generated and concatenated to the front of the input signal, delaying the output. The delay will always be entered in milliseconds, but the number of samples required to represent that delay will always be dependent on the sampling frequency of the system.

**180 degree Phase inversion**
The magnitude values of a sampled periodic waveform are represented by positive and negative integers. A phase inversion is simply achieved by multiplying these magnitude values of the signal by -1, changing the sign of every integer.

\[
\text{Inverted signal} = \text{signal} \times -1
\]

Eq. 2

**All-pass Filter**
The first order all-pass filter is essentially a combination of FIR and IIR comb filters with a single sample delay line. The FIR filter produces sharp notches in the signal's frequency response while the IIR filter produces broad notches with sharp peaks. The IIR filter's sharp peaks are centred on the FIR filter's sharp notches. When combined, the peaks and notches cancel out and result in a flat magnitude response across all frequencies and an ever increasing phase shift with frequency.

Since the phase shift of a first order all-pass filter increases continuously from zero degrees to 180 degrees with frequency there is a frequency at the midpoint where the phase shift is 90 degrees. This frequency which I have called \( f_{90} \) is the all-pass filter parameter that the operator will adjust.
The first order all-pass filter is given by the equation;

\[ y(n) = cx(n) + x(n - 1) - cy(n - 1) \]

Eq. 3

Where \( c \) tunes the filter to the desired f90;

\[ c = \frac{(\tan (\pi * f90 / fs) -1)}{(\tan (\pi * f90 / fs) + 1)} \]

Eq. 4

fs = sampling rate of the system (Hz)

f90 = frequency (Hz) where the phase shift is 90 degrees

As indicated in Equation 2, c has a positive sign as the feed forward coefficient and negative sign as the feedback coefficient. This ensures that the centres of the sharp notches of the FIR filter and the centres of the sharp peaks of the IIR filter are perfectly lined up. For the system to remain stable c must have a value of less than one. If this value exceeds one the system will feedback with ever increasing gain resulting in distortion of the signal.

The first order all-pass filter can be drawn as a block diagram;

Fig 2. All-pass filter block diagram with a delay (T) of 1 sample

**Evaluation**

The system has been tested in its basic form and has been shown to function as expected. For clarity the tests have been done with a 5ms long, 1kHz sine wave. The accompanying figures clearly show the effects of the system.

Figure 3 is a simple 5ms delay applied to the input signal. As expected the signal has been delayed with no other effects.

Figure 4 is the phase inversion. Only the polarity of the waveform has been inverted.

Figure 5 is the all-pass filter with f90 set to 1kHz. As expected the result is a phase shift of 90 degrees and this is most clear at the end of the sample plot.

Figure 6 applies the 5ms delay, phase invert and all-pass filter (with f90 of 1kHz) to the input signal. All parts of the system function as expected when combined.
Fig 3. Effect of a 5ms delay

Fig 4. Effect of phase invert
Fig 5. Effect of the all-pass filter with f90 of 1kHz

Fig 6. Effect of 5ms delay, phase invert and all-pass filter with f90 of 1kHz
Further Evaluation

While the sine wave signals used in the initial testing show that all of the functions work, they are not representative of real world scenarios. This system will find a lot of use in music recording and that is where further evaluations should be undertaken. The following scenarios are representative of real world situations;

Bass guitar – A signal from a D.I. Box and a signal from a microphone on the cabinet. Compared to the D.I. Signal, the microphone signal will be delayed slightly (amount dependent on the distance from the speaker) and the amplifier/speaker system will produce significant phase shift to the bass signal.

Electric guitar – Multiple microphones (2-3) at different distances from the guitar amplifier with one microphone wired 180 degrees out of phase. The different arrival times at the microphones will cause cancellations as will the incorrectly wired microphone.

Acoustic guitar – Multiple microphones (2-3) at different distances from the guitar with one microphone wired 180 degrees out of phase.

Drums – A bass drum with two microphones, one close and one distant.

Drums – A snare drum with two microphones, one on top and one on the bottom.

Test users will be presented with these scenarios and will be given the opportunity to correct them firstly with conventional tools (delay, manually moving the audio files, phase invert) and then using only the phase correction system. At the completion of each scenario they will complete a survey to gather their impressions of the effectiveness, functionality and quality of the phase correction system. The data gathered will be used for further development of the system.

References